Router Active Queue Management for Both Multimedia and Best-Effort Traffic Flows

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Abstract

In this paper, a novel active queue management scheme is proposed to employ in the routers that deal with both the best-effort traffic flows and multimedia traffic flows. Most of the available active queue management schemes consider only TCP flows, which results in unexpected congestion when it deals with multimedia flows. The queue size and packet receiving rate are the common parameters to calculate the marking probability in the queue. The use of Round Trip Time (RTT) in the process of marking probability calculation has the advantages over other approaches. We discuss the applicability of our approach when the framework in our earlier paper is used for rate adaptation in multimedia flows. With the use of RTT in packet marking probability calculations, we can assure rate reduction in both best-effort and multimedia flows before a queue gets exhausted. The most influential feature of the proposed approach is that it can work with the existing systems. The soundness of the proposed approach is inferred from the simulation results.

1. Introduction

The recent immense improvements in the Internet based applications have generated high demands on high-quality multimedia transmission systems. To meet such demands, it is critical for the multimedia transmission systems to have the capability to manage the available resources. Adaptive transmission rates are essential in multimedia communication to achieve the required service quality while utilizing the available resources [6][7]. A framework for rate adaptation depending on the required playback rates, client buffer occupancy, and network delays was discussed in [6].

The most prevailing factor in the service degradation is the packet being lost at a router due to congestion [4]. In the traditional router queue management approaches, there is no action taken to administrate the router queue size. This results in the increment of the router queue until it gets overflow, and drops all packets thereafter. One solution to the incontinent increase of the router queue size was the adjustment of the transmission rate depending on the number of packets being lost [2][4]. On the other hand, Explicit Congestion Notification (ECN) is the mechanism used to provide feedback from the router to the sender about impending congestions. In this mechanism, an active queue management (AQM) scheme in the router predicts about impending congestions and marks the packets with a probability determined by its AQM scheme. Now the sender is able to adjust its transmission speed depending on the number of marked packets, in order to restraint possible congestion. This approach is significant because of its ability to mitigate the unnecessary packet losses [4]. Several AQM schemes were proposed to calculate the packet marking probability in the router. Among them, the Random Early Detection (RED) is the most popular one.

In this paper, we propose an AQM scheme that is based on the instantaneous queue size together with the queue size increasing rate and RTT to suit the routers that are handling both best-effort and multimedia traffics. The proposed approach is based on the assumption that all multimedia flows have a common end-to-end rate adaptation framework proposed in [6]. The simulation results attest the soundness of our approach over RED [2] and REM [1].

2. Available active queue management schemes

2.1. Random early detection (RED)

RED is capable of predicting future congestions of the gateway. The average queue size is considered as the only parameter of predicting congestions even though impending congestions depend on several factors. Marking probability variations against the average queue size is shown in Figure 1. Service differentiation is unfeasible in RED since it responds to every packet in the same way.

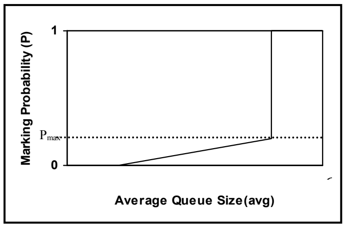

Figure 1. Marking probability for RED

2.2. Random exponential marking (REM)

REM maintains a variable called *price* as a congestion measure. This variable is used to determine the marking probability and is updated periodically based on rate mismatch and queue mismatch [1]. REM uses an exponential curve instead of a liner curve to determine the packet marking probability against *price*. Even though REM considers the queue size increasing rate indirectly in the process of marking probability calculations, it ignores the RTT in the marking probability calculation.

3. The proposed approach

Our proposed approach has several novel ideas. First, it differentiates the best-effort packets from the multimedia packets depending on one bit of the packet header, which is one of the most common methods to achieve service differentiation [5]. Hence, our proposed approach maintains only one queue for both types of flows, but with the use of two different algorithms, it can create two virtual queues. Other novel ideas include the calculation of q_{max} (size of the queue where the router starts to mark all packets) to reflect RTT and the queue size increasing rate information in the marking probability, and the marking probability calculation curves to reduce the unnecessary rate reductions while minimizing congestions. The use of RTT in the process of the marking probability calculation facilitates the proposed approach to assure that the router receives the packets with a reduced rate before it gets exhausted. The ability of the proposed approach to work with the existing TCP nodes is also influential.

Backward approximation is used to calculate the queue increment rate since it does not involve delays as in other approaches. We must neglect bursty traffics and transient congestions in calculating the marking probability [2]. The Finite impulse Response (FIR) digital filter is proposed to filter out these short-term changes in our proposed approach, since FIR can calculate the filtered output without any delay.

3.1. Estimation of q_{max}

Let T_f be the time taken to overflow after reaching q_{max}, $I_r(t)$ be the input rate, and $O_r(t)$ be the output rate, we have the following equation:

$$T_f = \frac{Q - q_{\max}}{dq(t)/dt} = \frac{Q - q_{\max}}{I_r(t) - O_r(t)} \tag{1}$$

To prevent the queue from overflows, the queue must start to mark all packets (i.e., q_{max}) of a particular stream when the time remaining to overflow the queue is equal to the RTT of that particular stream. Therefore, T_f is selected to be equal to RTT of the stream.

$$RTT = \frac{Q - q_{\max}}{\frac{dq(t)}{dt}} \tag{2}$$

$$q_{\max} = \begin{cases} Q - RTT \times \dfrac{dq(t)}{dt} & ;\ if\ \dfrac{dq(t)}{dt} > 0 \\ Q & ;\ if\ \dfrac{dq(t)}{dt} \le 0 \end{cases} \tag{3}$$

With the selection of q_{max} according to Equation (3), the possibility of getting the packets with the reduced rate before the queue gets exhausted is one, for both TCP and multimedia flows. Hence, the possibilities of congestions are very low with the proposed approach.

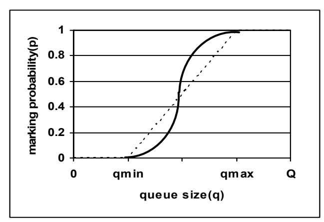

Figure 2. Marking probability for multimedia traffics

3.2. Estimation of marking probability

Once the q_{max} value is available, the next step is to calculate the marking probability based on the q_{max} value and the instantaneous queue size. Two separate algorithms are used for the multimedia traffic flows and the best-effort traffic flows.

3.2.1. Multimedia traffics

The marking probability is zero for the queue values below predetermined value, q_{min}. The marking probability must be one for the queue values greater than or equal to q_{max}. To meet the above requirements, the marking probability curve in Figure 2 is proposed. As can be seen from Figure 2, the marking probability and the rate reduction in our proposed approach is lower than those of the linear case up to the midpoint between q_{max} and q_{min}. Then the marking probability is comparatively high since it is near congestion. This can reduce the unnecessary rate reductions and achieve an optimal rate. The vigorous idea of our proposed approach is that it can prevent the queue from being overflow while maintaining the transmission rate at a higher value. Hence, the marking probability (*P*) at the router is given by:

$$\text{if } q < q_{\min} \text{ then } P = 0$$

$$\text{if } q_{\min} \le q < \left(\frac{q_{\min}+q_{\max}}{2}\right) \text{ then } P = \frac{2K_m\left(q^2/2 - q q_{\min}\right) + K_m q_{\min}^{\;2}}{\left(q_{\max}-q_{\min}\right)}$$

$$\text{if } \left(\frac{q_{\min}+q_{\max}}{2}\right) \le q < q_{\max} \text{ then } P = \frac{2K_m\left(q_{\max} q - q^2/2\right) + \left(q_{\max} - q_{\min} - K_m q_{\max}^{\;2}\right)}{\left(q_{\max}-q_{\min}\right)}$$

$$\text{if } q_{\max} \le q \text{ then } P = 1$$

$$\text{Where } K_m = 2/\left(q_{\max}-q_{\min}\right)$$

3.2.2. Best-effort traffics

To meet the requirements of the TCP congestion control mechanism, we propose a marking probability curve similar to RED (shown in Figure 1) but with dynamic q_{max} values to reflect the queue size increasing rate and RTT. q_{min} is not predetermined and its value depends on the calculated q_{max} value. q_{min} is defined to be one third of the q_{max} value as in RED [2].

3.3. Rate adaptation with marking probability for multimedia traffics

For multimedia streams, the sender first calculates the transmission rate based on the framework in [6] and then weights the calculated transmission rate by the marking probability. The receiver counts the number of marked packets out of the total number of packets over the feedback period. Based on the total number of packets and the number of marked packets, the receiver reconstructs the marking probability (P_m). Then the receiver sends this information to the sender via the feedback information. The sender uses P_m to weight the transmission rate using Equation (4).

$$next\_rate = \frac{next\_rate}{\left(1 + K_{mark} \times P_m\right)} \tag{4}$$

3.4. Round trip time (RTT) estimation

The estimation of RTT for individual streams is a challenge. One approach is to use RTT information in the IP header of each packet. However, our proposed approach must be compatible with the existing systems that use the established IP header. Hence, an alternative approach is proposed.

A handshaking procedure is proposed to calculate RTT for individual flows [3]. In handshaking, first the sender sends a request and the receiver acknowledges. Then the sender starts to transmit after it receives the acknowledgement. So the time period between the first two packets of the flow from the sender to the receiver is equal to RTT. This method is reasonably accurate for use in the AQM. For multimedia flows, the above method is applicable to calculate the RTT. However, in the framework in [6], the feedback information is sent periodically with a fixed interval and hence our proposed approach relies on the end system to include the feedback interval into the RTT.

4. Simulations and results

4.1. Simulation environment

We conducted our simulations on ns-2 environment. Four approaches including our proposed approach were considered: (1) RED queues in all gateways (denoted as RED); (2) REM queues in all gateways (denoted as REM); (3) RED for best-effort packets and the proposed approach for multimedia packets (denoted as PRO+RED); and (4) The proposed approach for every packet (denoted as PRO). Throughout the simulations, 100 packet queues were used, q_{min} = 20 packets, and P_{max} = 0.04. The selected parameters for RED, are q_{max} = 80 packets and w_q = 0.002. For PRO+RED, q_{max} = 60 packets and w_q = 0.008. For PRO, the update time = 0.005 sec, cut-off frequency of the short-term change filter = 0.125 of the update frequency and the FIR filter with five weights. K_{mark} is the variable used to weight the transmission rate with the detected marking probability. First, we did the simulation with one gateway to find the most suitable value for K_{mark}, and from the result, 0.4 was selected as the K_{mark} value for all simulations. The network protocol we used in our simulations is shown in Figure 3.

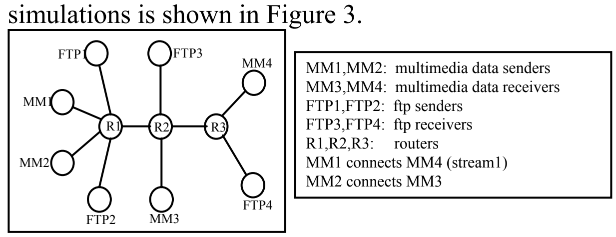

Figure 3. Simulation protocol

4.2. Simulation results

The instantaneous queue sizes are shown in Figure 4. It can be easily seen that our proposed approach has a very stable queue size. The rate of queue overflow is very low in comparison to the others. These slow queue size variations are the direct outcomes of our proposed approach's ability to monitor the queue size increasing rate, and its ability to send the congestion information to the senders before it gets exhausted.

The packet losses for stream 1 are shown in Figure 5. The number of packets being lost is very low in our proposed approach in comparison to all the other three approaches, which is directly resulted from the stable queue size. The average packet losses are only 0.126 packets per feedback interval in our proposed approach; while it is 1.158, 1.21, and 0.906 in PRO+RED, REM and RED, respectively.

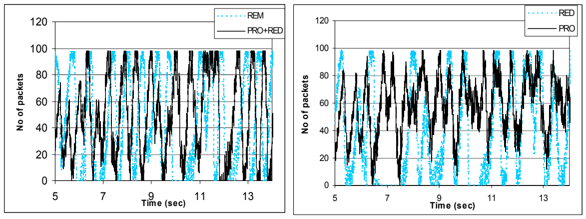

Figure 4. Queue size for the queue between R1 and R2

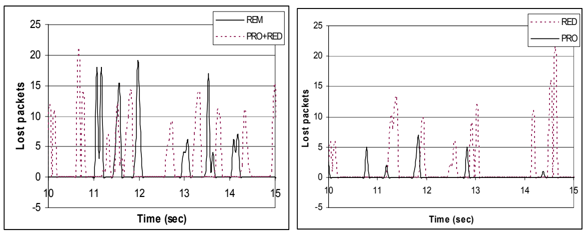

Figure 5. Packet loss for multimedia stream 1

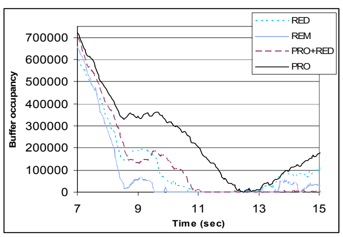

Figure 6. Buffer occupancy for multimedia stream 1

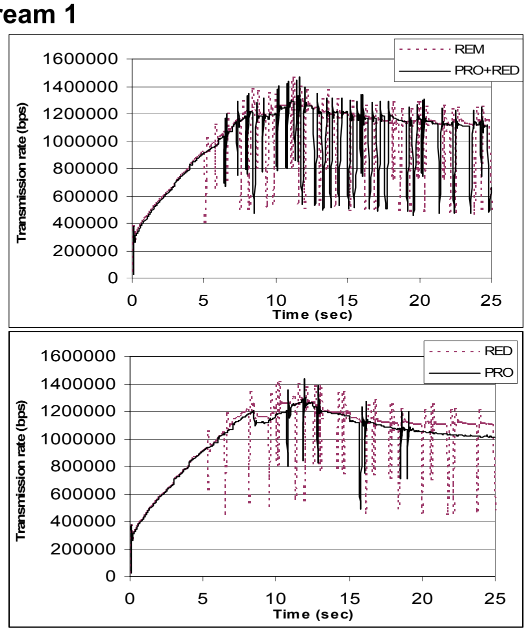

Figure 7. Transmission rate for multimedia stream 1

Figure 6 shows the buffer occupancy. It can be easily seen that our proposed approach has a significant improvement in comparison to the other approaches including REM. The reason behind this improvement is that our proposed approach always tries to minimize unnecessary rate reductions.

The transmission rates of multimedia stream 1 are shown in Figure 7. As shown in the figure, the proposed approach has a much stable transmission rate than the other three approaches. The average transmission rate is 1,000,896.4 bps for the proposed approach; while they are 969,171.489, 967,442.746, and 981,356.475 for PRO+RED, REM, and RED, respectively.

5. Conclusions

In this paper, a router active queue management scheme that can deal with both multimedia traffic flows and best-effort traffic flows is proposed. Our proposed approach considers the instantaneous queue size, queue size increasing rate, and round trip time (RTT) in calculating the marking probability. The handshaking procedure of each stream is adopted to calculate RTT. It can differentiate multimedia packets from the best-effort packets and treat them differently. The simulations under the ns-2 environment were conducted. The performance of our proposed approach was compared with some other approaches including RED and REM under the instantaneous queue sizes, packet losses, client buffer occupancy, and the transmission rates. The simulation results attest the superiority of our proposed approach.

6. References

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