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Research Article

Rate-Based Active Queue Management for TCP Flows over Wired and Wireless Networks

Jun Wang and Min Song

Department of Electrical and Computer Engineering, Old Dominion University, 231 Kaufman Hall, Norfolk, VA 23529, USA

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Current active queue management (AQM) and TCP protocol are designed and tuned to work well on wired networks where packet loss is mainly due to network congestion. In wireless networks, however, communication links suffer from significant transmission bit errors and handoff failures. As a result, the performance of TCP flows is significantly degraded. To mitigate this problem, we analyze existing AQM schemes and propose a rate-based exponential AQM (REAQM) scheme. The proposed REAQM scheme uses the input rate as a primary metric and queue length as the secondary metric. The objectives of REAQM are to stabilize networks with low packet loss, low packet delay, and high link utilization regardless the dynamic of network conditions. We prove the global asymptotic stability of the equilibrium based on Lyapunov theory. Simulation results suggest that REAQM is capable of performing well for TCP flows over both wired and wireless networks, and has comparable implementation complexity as other AQM schemes.

1. INTRODUCTION

The essence of network congestion control is that a sender adjusts its transmission rate according to the congestion measure of the underline networks. There are two components to accomplish this. One is a source algorithm that dynamically adjusts the transmission rate in response to the congestion; the other one is a link algorithm that implicitly or explicitly conveys information about the current congestion measure to sources using that link. In the current Internet, the source algorithm is carried out by TCP, and the link algorithm is performed by active queue management (AQM) schemes at routers. Examples of AQM schemes are RED [1], REM [2], AVQ [3], and Yellow [4]. TCP defines how the source rates are adjusted while AQM schemes define how the congestion measure is defined and updated.

According to the type of metrics used to measure congestion, AQM schemes can be classified into three catalogs: queue-based, rate-based, and schemes based on concurrent queue and rate metrics. In queue-based schemes, congestion is observed by average or instantaneous queue length and the control aim is to stabilize the queue length. The drawback of queue-based schemes is that a backlog is inherently necessitated. Rate-based schemes predict the utilization of the link, determine the level of congestion, and take actions based on the packet arrival rate. Rate-based schemes can provide early feedback for congestion [3]. Other AQM schemes deploy a combination of queue length and input rate to measure congestion and achieve a tradeoff between queue stability and responsiveness.

Current TCP/AQM algorithms assume that packet loss is mainly due to network congestion. However, these TCP/AQM algorithms are insufficient for the hybrid wired and wireless networks. In wireless networks, communication links have intrinsic characteristics that affect the performance of transport protocols including variable bandwidth, corruption, channel allocation delays, and asymmetry [5]. All of these cause significant noncongestion packet loss. The generic TCP protocol notifies sources to reduce their transmission rates only when facing packet losses due to network congestion. Therefore, the TCP protocol in hybrid wired and wireless networks needs not only to detect packet losses, but also to detect the reason of packet losses. Moreover, the degree of statistical multiplexing number of flows over wireless links is different from that over wired links [5]. For example, a wireless link only has several flows instead of hundreds of flows over a wired link. While existing TCP/AQM schemes are tuned well for wired links which have high degree of statistical multiplexing, they may not work well for wireless links with only several flows.

In this paper, we present a new rate-based exponential AQM (REAQM) with explicit congestion notification (ECN) marking, and use REAQM to enhance the TCP performance over wireless links. The main idea of REAQM is to use the mismatch between input rate and link capacity as the primary metric, which exploits the early feedback benefit of rate-based marking. Furthermore, REAQM uses queue length to compute a coefficient of rate mismatch and use it as a secondary metric. The rationale of the secondary metric is to achieve a tradeoff between queue stability and responsiveness of the system. Using ECN marking to notify senders of incipient congestion, REAQM overcomes the packet loss problem of burst data and errors over wireless links, and also performs well for high degree of multiplex of flows over wired links.

The rest of the paper is structured as follows. In Section 2, we introduce some related work, including the analytical model for TCP/AQM algorithms, the state-of-the-art research results on AQM schemes, and the characteristics of wireless links. We present the design of REAQM and analytical results in Section 3. In Section 4, we discuss simulation design and results, and compare our REAQM scheme with other AQM schemes. Finally, we provide concluding remarks in Section 5.

2. RELATED WORK

We start this section by introducing three representative AQM schemes. RED is a queue-based, most popular AQM schemes used [1]. In RED, packets are randomly dropped before the buffer is completely full, and the drop probability increases with the average queue length. As source rates increase, queue length grows and more packets are marked/dropped. This prompts the sources to reduce their rates. RED configuration has been a problem since its first proposal, and many studies have tried to address this issue [6]. While the presence of a persistent queue indicates a congestion, its length gives very little information as to the severity of congestion. Therefore, decoupling queue length from congestion management has some benefits.

REM is a queue-and-rate-based scheme. The objective of REM is to stabilize both the input rate around link capacity and the queue length around a predetermined target, regardless of the number of users sharing the link [2]. In REM, each output queue maintains a *price* function as a congestion measure. The price is updated based on rate mismatch (i.e., difference between input rate and link capacity) and queue mismatch (i.e., difference between queue length and the predetermined target). Correspondingly, REM uses an exponential marking probability function. REM decouples congestion measure from performance measure such as packet loss, queue length, or packet delay. Since REM tries to maintain a target queue length regardless the number of flows, it limits its ability to handle burst traffic.

AVQ is primarily a rate-based scheme, as opposed to queue length or average queue length-based marking [3, 7]. AVQ maintains a virtual queue whose capacity is less than the actual capacity of the link. When a packet arrives in the real queue, the virtual queue is also updated to reflect the new arrival. Packets in the real queue are marked/dropped when the virtual buffer overflows. The virtual capacity at each link is then adapted to ensure that each link achieves a desired utilization of the link. Another rate-based scheme is called Yellow [4]. Yellow uses the load utilization as a primary metric to manage congestion, and a queue control function as a secondary metric to improve congestion control performance. According to Yellow, a load factor is calculated from link incoming rate and virtual available capacity, where the virtual available capacity is updated based on the queue control function. The queue control function is introduced to achieve a stable and smooth response. The main idea of Yellow is to perform queue management based on the link load factor, and to predict incipient congestion timely and accurately with controlled queuing delays, stability, and robustness. Packet marking probability is based on the load factor to achieve high link utilization and avoid high packet loss ratio.

Rate-based marking provides early feedback and responds fast when there exists a rate mismatch between input rate and link capacity. The advantages of this approach have been explored in [8]. The early feedback is more robust to the presence of extremely short flows or variability in the number of long flows in the network. When utilization is close to 100%, the variance introduced by short flows seems to lead to an undesirable transient behavior where excessively large queue lengths persist over long periods of time. Since queue length is a cumulative difference value of rate mismatch, queue metric is insensitive to current queue arrival and drain rates. Low [7] proposed a duality model for TCP/AQM algorithms to explain the equilibrium properties of a large network under TCP/AQM control, such as throughput, delay, queue length, loss probability, and fairness. Duality model considers the process of congestion control as a distributed computation by sources and links over a network to solve a global optimization problem in real time.

Gurtov and Floyd [5] consider the interplay between wireless links and transport protocols, and present the rules to appropriately model wireless links. Wireless links suffer from significant packet losses due to bit errors and handoffs. A TCP flow can only reduce its sending rate on losses due to network congestion, not on those due to wireless effects. New AQM algorithms have been proposed specifically for wireless links. Sagfors et al.use instantaneous queue length to measure congestion and propose a deterministic dropping strategy, packet discard prevention counter, which is tailored to an estimate of the pipe capacity and TCP's rate halving policy [9]. Li and Liu present an explicit feedback scheme with AQM to handle large bandwidth delay product and burst packet losses over wireless links [10]. The main problem in such scenarios is preventing the slow start overshoot of TCP connections. The overshoot happens because TCP detects congestion up to one RTT after filling the buffer and the first packet drop. The TCP's sending rate at this point can be twice the available bandwidth of the path and thus generate many packet drops.

Most wireless networks suffer from bursty errors. A two-state Markov error model is used to capture the packet loss characteristics [11]. The two states represent a *good* state with a low-bit-error-rate (BER) value and a *bad* state with a high BER value. The link switches between good and bad states, and the lifetime of each state is exponentially distributed with a different mean value at each time. Moreover, the degree of statistical multiplexing number of flows over wireless links is different from that over wired links. For example, a wireless link only has several flows instead of hundreds of flows over a wired link. When the degree of statistical multiplexing is low and the buffer size is small, existing AQM algorithms may not perform sufficiently well. While many link technologies include forward error correction (FEC) and local retransmission for addressing corruption at the link layer, these mechanisms can introduce their own complications, and result in a high variability of bandwidth and delay on wireless links.

3. REAQM SCHEME

3.1. Scheme description

The objectives of REAQM scheme are to regulate the link utilization and stabilize the networking system regardless the changing network conditions. Similar to REM, REAQM maintains a variable, *price*, as a congestion measure and uses an exponential marking probability function. On the other hand, REAQM differs from REM in the definition of congestion measure. Denote $p_l(t)$ as the price at queue l in period t. The marking probability function of REAQM is

$$m_l(t) = 1 - \phi^{-p_l(t)}, \tag{1}$$

where ϕ is a constant larger than 1. The parameter ϕ determines the range of loss or marking probability, which also depends on the range of price $p_l(t)$. Ideally, ϕ should be chosen in a way that the end-to-end probability observed at hosts is small, especially for the AIMD algorithm of TCP Reno and its variants.

At each packet arrival epoch, the price is updated according to the following equation:

$$p_l(t+1) = \left[p_l(t) + \gamma\left(f\left(q_l(t)\right)y_l(t) - c_l\right)\right]^+. \tag{2}$$

Here, $p_l(0) = 0$, and $[z]^+ = \max\{z, 0\}$. $q_l(t)$ is the aggregate queue length at queue l in period t, $y_l(t)$ is the aggregate input rate to queue l in period t, and c_l is the link capacity at queue l. The smoothing parameter $\gamma > 0$ is a step size which determines the speed of convergence of the algorithm. A larger γ results in a faster convergence, but it also incurs a higher risk of oscillatory queue. The coefficient of rate mismatch is a function of queue mismatch at queue l in period t;

$$f\left(q_l(t)\right) = \frac{\alpha_l q_l(t)}{c_l} + 1, \tag{3}$$

where weight parameter, α_l, is a constant. Obviously, $f(q_l(t)) \geq 1$, and it is equal to 1 only if $q_l(t) = 0$. If the value of α_l is small, the queue length has less effect on the coefficient. Substituting (3) into (2), we get

$$p_l(t+1) = \left[p_l(t) + \gamma\left(\alpha_l q_l(t)\frac{y_l(t)}{c_l} + y_l(t) - c_l\right)\right]^+. \tag{4}$$

Price is updated, periodically or asynchronously, mainly based on rate mismatch. The price is incremented if rate mismatches is positive, and decremented otherwise. Rate mismatch is positive when the input rate exceeds the link capacity and negative otherwise. Queue length is used to compute the coefficient of rate mismatch and adjust the scale of rate mismatch. If queue length is small, the impact of rate mismatch decreases and REAQM is less aggressive. Otherwise, REAQM is more aggressive. The rationale behind price update is that the smaller queue length, the more rate mismatch is allowed. This makes a tradeoff between the system stability and utilization. If current input rate exceeds link capacity, the packet marking probability will increase; otherwise, it will decrease. Second, the larger the current queue length, the larger the coefficient of rate mismatch is; thus the larger the price value increases or decreases as well. While rate mismatch is positive, that is, $y_l(t) > c_l$, the price always increases for any value of queue length. If α_l is larger enough, frequently positive rate mismatch results in a small queue size and may lower system throughput. We modify the price update equation in this case as

$$p_l(t+1) = \left[p_l(t) + \gamma\left(\alpha_l' q_l(t)\frac{\left(y_l(t) - y_0\right)}{c_l} + y_l(t) - c_l\right)\right]^+, \tag{5}$$

where y_0 is target queue length while rate mismatch is positive, and α_l' is relative weight parameter larger than α_l. REAQM has comparable implementation complexity as REM, since the mainly difference of REAQM and REM is the price function.

When the number of TCP flow increases, the rate mismatches grow and thus push up the price and hence marking probability. This increases the intensity of congestion signal to sources, which then reduce transmission rates. When transmission rates are too small, the mismatches will be negative. This lowers the price and marking probability and raises transmission rates, until eventually the mismatches are driven to zero yielding high utilization and negligible loss and delay in equilibrium. Recall that the mean queue length steadily increases as the number of flows increases in RED. In contrast, the price steadily increases while rate mismatches grows and the mean queue length is stabilized under REAQM.

Generally, ECN bit is used to inform the source which losses are due to congestion or wireless effects [2, 7]. If a packet is marked by setting its ECN bit, the mark is carried to the destination and then conveyed back to the source via acknowledgments. We set ECN bit to 1 while packets are probabilistically marked according REAQM scheme, and drop packets only when they arrive at a full buffer. Although this method cannot completely prevent buffer from overflowing and thus some packets are dropped by congestion, it differentiates most of error losses and congestion losses.

We conclude this section by stating that AQM must be more aggressive to avoid buffer overflow for high degree of statistical multiplexing and less aggressive to avoid underutilization for low degree of statistical multiplexing. Consider a single link accessed by many TCP sources with the same round-trip time (RTT). Assume fixed packet size and the link is equally shared amongst n TCP long-live flows. A congestion notification to one flow reduces the offered load by a factor of $(1 - 1/2n)$. Therefore, the larger n is, the less impact of individual marking.

3.2. Stability analysis

In this section, we present a global stability analysis for REAQM based on the duality model for TCP/AQM system [7]. The global asymptotic stability of the equilibrium for REAQM scheme can be proved based on Lasalle's invariance principle applied to a suitable Lyapunov function as in [12].

Let (q^*, p^*) be an equilibrium of our system, q_s^* and x_s^* be the equilibrium source price and rate for source s, respectively. Further, let y_l^* be the equilibrium link rate for queue l. The queue length is taken as follow:

$$q_l(t+1) = \left[q_l(t) + y_l(t) - c_l\right]^+. \tag{6}$$

From (2) and (6), the price and queue dynamic are taken to be

$$\dot{p}_l(t) = \begin{cases} \gamma\left(f\left(q_l(t)\right)y_l(t) - c_l\right), & p_l(t) > 0, \\ \left[\gamma\left(f\left(q_l(t)\right)y_l(t) - c_l\right)\right]^+, & p_l(t) = 0, \end{cases}$$
$$\dot{q}_l(t) = \begin{cases} \left(y_l(t) - c_l\right), & q_l(t) > 0, \\ \left[y_l(t) - c_l\right]^+, & q_l(t) = 0. \end{cases} \tag{7}$$

Theorem 1. *Given the system defined by* (7), *assume $f_s(q_s)$ is strictly decreasing in $q_s > 0$ and that R is of full row rank, then the unique equilibrium point $q^* = 0$, p^* is globally asymptotically stable.*

First, we introduce the candidate Lyapunov function $V(q, p)$:

$$V(q, p) = \sum_{l=1}^{L}\left[\gamma\alpha_l\frac{q_l^2}{2} + \left(c_l - y_l^*\right)p_l\right] + \sum_{s=1}^{S}\phi_s\left(q_s\right), \tag{8}$$

where $\phi_s(q_s) = \int_{q_s^*}^{q_s}(x_s^* - f_s(\sigma))d\sigma$. Function V is nonnegative and radially unbounded.

Take the derivative of $V(q, p)$ along trajectories of our system:

$$\dot{V} = \sum_{l=1}^{L}\left[\gamma\alpha_l q_l\dot{q}_l + \left(c_l - y_l^*\right)\dot{p}_l\right] + \sum_{s=1}^{S}\left(x_s^* - f_s\left(q_s\right)\right)\dot{q}_s. \tag{9}$$

The last term above can be rewritten as

$$\begin{aligned}\sum_{s=1}^{S}\left(x_s^* - f_s\left(q_s\right)\right)\dot{q}_s &= \left(x^* - x\right)^T\dot{q} = \left(x^* - x\right)^T R^T\dot{p} \\ &= \left(y^* - y\right)^T\dot{p} = \sum_{l=1}^{L}\left(y_l^* - y_l\right)\dot{p}_l.\end{aligned} \tag{10}$$

Substituting back, we have

$$\dot{V} = \sum_{l=1}^{L}\left[\gamma\alpha_l q_l\dot{q}_l + \left(c_l - y_l\right)\dot{p}_l\right] = \sum_{l=1}^{L}v_l, \tag{11}$$

where $v_l = \gamma\alpha_l q_l\dot{q}_l + (c_l - y_l)\dot{p}_l$.

We will show that $\dot{V} \leq 0$ for each l. We apply the dynamic equation (7), and discuss the four cases:

(1) $q_l > 0$, $p_l > 0$. Here

$$\begin{aligned} v_l &= \gamma\alpha_l q_l\left(y_l - c_l\right) + \left(c_l - y_l\right)\gamma\left(f\left(q_l\right)y_l - c_l\right) \\ &= \gamma\left(y_l - c_l\right)\left[\alpha_l q_l - \left(f\left(q_l\right)y_l - c_l\right)\right] \\ &= \gamma\left(y_l - c_l\right)\left[\alpha_l q_l - \left(\frac{\alpha_l}{c_l}q_l y_l + y_l - c_l\right)\right] \\ &= -\gamma\left(\frac{\alpha_l q_l}{c_l} + 1\right)\left(y_l - c_l\right)^2 = -\gamma f\left(q_l\right)\left(y_l - c_l\right)^2. \end{aligned} \tag{12}$$

(2) $q_l > 0$, $p_l = 0$. Here

$$v_l = \left(y_l - c_l\right)\left(\gamma\alpha_l q_l - \left[\gamma\left(f\left(q_l\right)y_l - c_l\right)\right]^+\right). \tag{13}$$

If $[\gamma(f(q_l)y_l - c_l)]^+ = 0$, $y_l \leq c_l/f(q_l) < c_l$, $v_l = \gamma\alpha_l q_l(y_l - c_l) < 0$.

Note that $f(q_l) > 1$ while $q_l > 0$.

Otherwise,

$$\begin{aligned} v_l &= \left(y_l - c_l\right)\left[\gamma\alpha_l q_l - \gamma\left(f\left(q_l\right)y_l - c_l\right)\right] \\ &= -\gamma f\left(q_l\right)\left(y_l - c_l\right)^2. \end{aligned} \tag{14}$$

(3) $q_l = 0$, $p_l > 0$. Here $f(q_l) = 1$, so

$$v_l = \left(c_l - y_l\right)\gamma\left(f\left(q_l\right)y_l - c_l\right) = -\gamma\left(y_l - c_l\right)^2. \tag{15}$$

(4) $q_l = 0$, $p_l = 0$. Here $f(q_l) = 1$, so

$$\begin{aligned} v_l &= \left(c_l - y_l\right)\left[\gamma\left(f\left(q_l\right)y_l - c_l\right)\right]^+ \\ &= \left(c_l - y_l\right)\left[\gamma\left(y_l - c_l\right)\right]^+. \end{aligned} \tag{16}$$

If $y_l - c_l < 0$, $[\gamma(y_l - c_l)]^+ = 0$, so $v_l = 0$.
If $y_l - c_l \geq 0$, $v_l = -\gamma(y_l - c_l)^2$.

We thus confirm that $v_l \leq 0$ for every l, thus $\dot{V} \leq 0$. Based on Lyapunov's stability theorem, the trajectory $(q(t), p(t))$ must remain bounded over the time, and the equilibrium point (q^*, p^*) is stable in the sense of Lyapunov. Note that $\dot{V} = 0$ when either $y_l - c_l < 0$, or $y_l - c_l < 0$ and $q_l = p_l = 0$, which is the desired stable zone. The set of states (q, p) where the Lyapunov derivative is zero is as same as that in [12]. The system is globally asymptotically stable by means of Lasalle's invariance principle.

4. SIMULATIONS AND ANALYSIS

Simulations are conduced using *ns*-2 simulator. Figure 1 shows the network topology, where n TCP flows share a bottleneck link that marks or drops packets according to some AQM scheme.

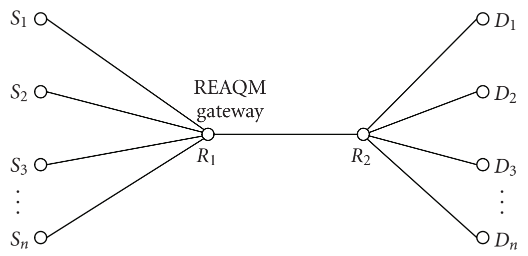
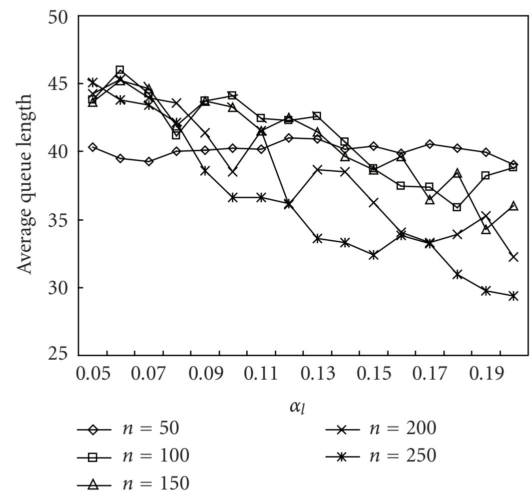


Figure 1: Network topology.

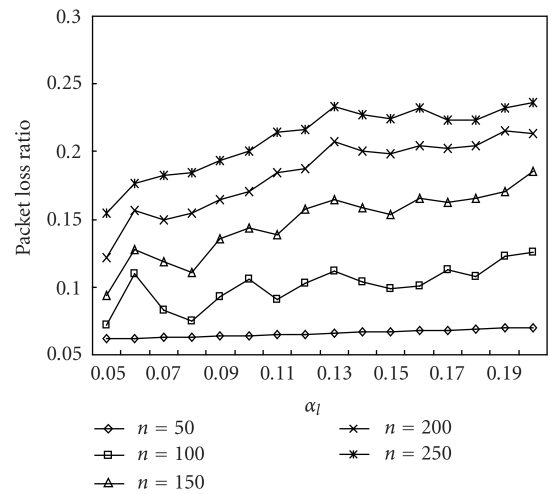
In Figure 1, the two routers (R_1 and R_2) are connected by a link with a capacity of 10 Mbps, which could be a wired or wireless link. The capacity of all other links is 100 Mbps. The propagation delay of the bottleneck link (between two routers) is set to be 10 milliseconds, and those of the other links are set to be uniformly distributed between 10 milliseconds and 30 milliseconds. NewReno is used as default transport protocol with the TCP data packet size 1000 bytes. The performance metrics are average queue length, link utilization, and packet loss ratio.

In the first experiment, we study the parameter setting of REAQM, especially, for the parameters α_l. We assume that the bottleneck link is reliable and run the simulation for different number of TCP flows. We start the experiment with 50 FTP flows in the system, and new 50 FTP flows are added every 50 seconds until the total number of flows reach 250. The queue size is 50. The objective is to study the sensitivity for the parameter α_l and the thumb of rule for parameter configuration.

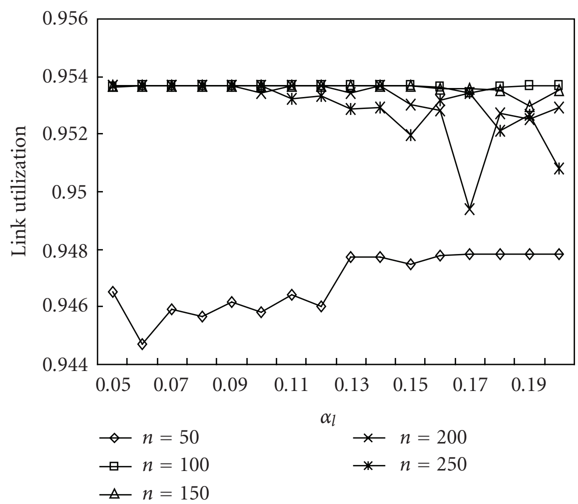
The performance influence of parameter α_l is given in Figure 2. The value of α_l determines the prominence given to the queue length when determining the level of congestion. As can be seen the average queue length (Figure 2(a)) is reduced with increasing α_l. The smaller α_l is, the smaller the coefficient of rate mismatch is. Therefore, queue length has less effect on the rate mismatch. If only (4) is used, the price is always increased with positive rate mismatch. While small α_l is chosen, average queue length will be increased. The coefficient is small while we use small α_l. The price increases lightly even with large queue length and positive rate mismatch. Eventually, it results in very large queue length. On the other hand, large α_l indicates that even a small queue length has relative effect on the coefficient of rate mismatch. If both the number of flows and α_l are large enough, average queue length is very small and link utilization is also low. Therefore, price is updated based on (5) in the experiment. For the same α_l, average queue length decreases mostly while the number of flows increases. The reason is larger number of flows results in a higher probability of positive rate mismatch, which reduces the allowable queue length. It should be noticed that packet loss ratio (Figure 2(b)) is increased lightly with increasing α_l. For different α_l, the larger average queue length is, the smaller packet loss ratio is. Therefore, choosing different value of α_l makes a tradeoff between queue length and loss ratio. Packet loss ratio is increased with the increasing of number of flows. Link utilization (Figure 2(c)) is very stable



(a) Average queue length.



(b) Packet loss ratio.



(c) Link utilization.

Figure 2: Performance using different α_l.

for all α_l while the number of flows is larger than 50, and has oscillation while number of flows is 50. We can use different values of α_l to achieve small queue length or maintain small packet loss ratio.

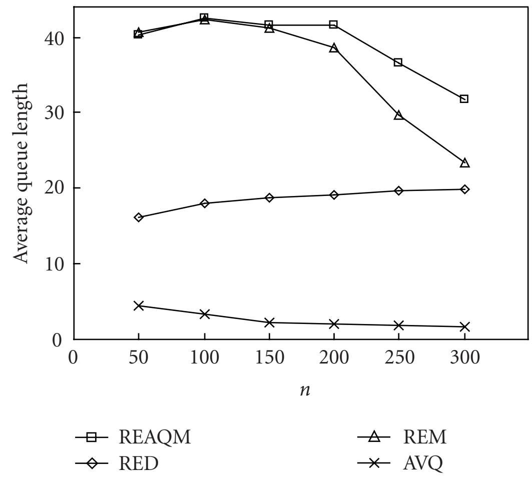
In the following experiments, we compare REAQM with other AQM schemes over wired and wireless networks. In these experiments, we use the "gentle" version of RED. The parameters of REM were chosen as recommended in [2], and the parameters of AVQ were chosen as recommended in [3]. The queue size is 50 for all AQM schemes. The parameters of REAQM are set as follows: $\alpha_l = 0.11$, $\gamma = 0.001$, and $\Phi = 1.001$.

In the second experiment, we consider the bottleneck link is reliable, and FTP flows are added or dropped to the network. We start the experiment with only small number of FTP flows in the system, and new 50 FTP flows are added every 50 seconds until the total number of flows reach the maximal value, 300. Then 50 flows are dropped every 50 seconds until the total number of flows is equal to the initial number. For every 50 seconds, we calculate each performance metric. A long transient period is always with an increasing average queue length while new flows are added before the scheme is able to converge at new network condition. Therefore, average queue length over each 50 seconds interval captures persistent transients.

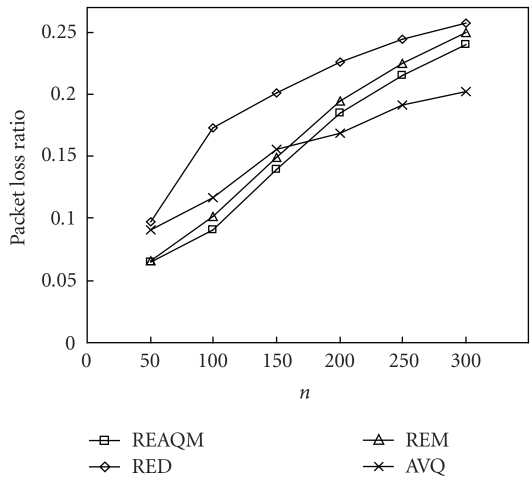
Figure 3 shows the performance of each AQM scheme under varying degree of flow multiplexing over the wired link. The overall performance of REAQM is comparable with REM and RED. Average queue length (Figure 3(a)) of REAQM is comparable with that of REM, and is larger than that of RED and AVQ. Packet loss ratio (Figure 3(b)) of REAQM is smaller than other schemes. Link utilization (Figure 3(c)) of REAQM is comparable with that of REM, and is higher than RED and AVQ. The average queue length and link utilization of REAQM are also very stable for different number of flows. REAQM uses rate mismatch as main metric to update the price, which makes it very robust to the present of variability in the number of flows. The performance of AVQ is different with other schemes. It has the smallest average queue length and link utilization.

In the third experiment, we compare the properties of various AQM schemes while the bottleneck link is wireless. We modify NewReno so that a source halves its window when it receives a mark or detects a loss through timeout, but retransmits without halving its window when it detects a loss through duplicate acknowledgments. The number of flows added or dropped at each time interval is 10 due to the low degree of flow multiplexing over wireless link.

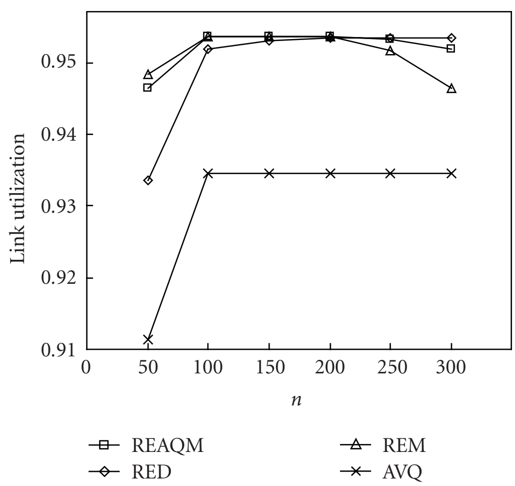
Figure 4 shows the performance of each AQM schemes under varying degree of flow multiplexing over the wireless link. The average queue length (Figure 4(a)) of REAQM is slight smaller than that of REM, and higher than that of other two schemes. Packet loss ratios (Figure 4(b)) of all schemes are comparable. The link utilization (Figure 4(c)) of REAQM and REM are better than RED and AVQ. While the number of flows is small, link utilization of AVQ is lower than that of other schemes.



(a) Average queue length.

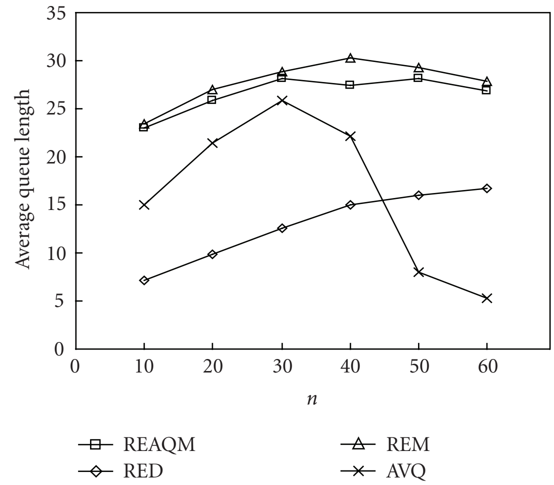


(b) Packet loss ratio.

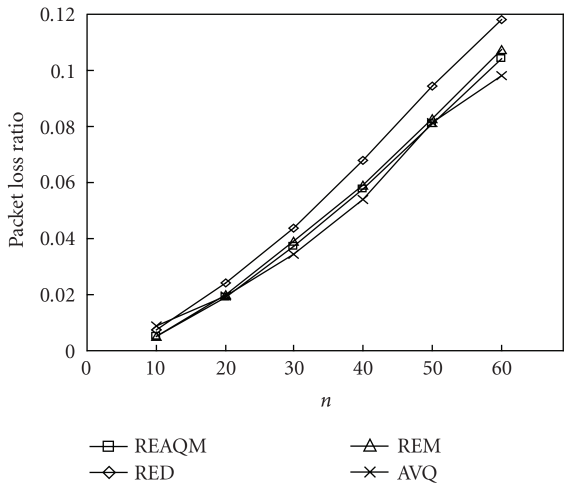


(c) Link utilization.

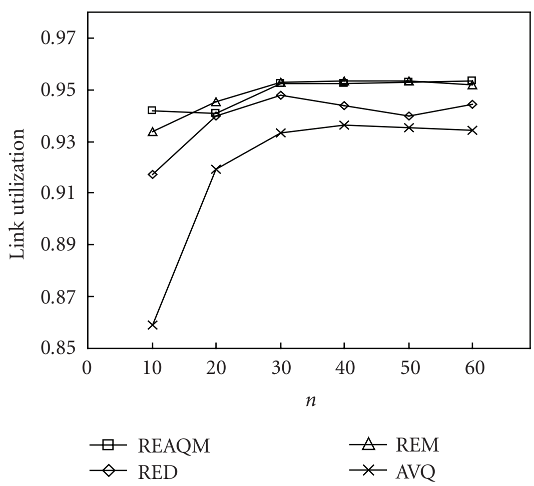
Figure 3: Performance test versus number of FTP flows for the different AQM schemes over wired link.



(a) Average queue length.



(b) Packet loss ratio.



(c) Link utilization.

Figure 4: Performance test versus number of FTP flows for the different AQM schemes over wireless link.

5. CONCLUSIONS

In wireless networks, packets are lost mainly because of bit errors and intermittent connectivity. In this paper, we have presented a new rate-based exponential AQM (REAQM) with explicit congestion notification (ECN) marking, and use REAQM to enhance the TCP performance over both wired and wireless links. The main idea of REAQM is to use the mismatch between input rate and link capacity as the primary metric, which exploits the early feedback benefit of rate-based marking. Furthermore, REAQM uses queue length to compute a coefficient of rate mismatch and uses it as a secondary metric. The rationale of the secondary metric is to achieve a tradeoff between queue stability and responsiveness of the system. The global asymptotic stability of REAQM has been proved using Lyapunov theory. Simulation results suggest that REAQM is capable of performing well for TCP flows over both wired and wireless networks, and has comparable implementation complexity as other AQM schemes.

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