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Improving explicit congestion notification with the mark-front strategy ☆

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Abstract

Delivering congestion signals is essential to the performance of networks. Current TCP/IP networks use packet losses to signal congestion. Packet loss not only reduces TCP performance, but also adds large delay. Explicit congestion notification (ECN) delivers a faster indication of congestion and has better performance. However, current ECN implementations mark the packet from the tail of the queue. In this paper, we propose the mark-front strategy to send an even faster congestion signal. We show that mark-front strategy reduces buffer size requirement, improves link efficiency and provides better fairness among users. Simulation results that verify our analysis are also presented. © 2001 Elsevier Science B.V. All rights reserved.

Keywords: Explicit congestion notification; Mark-front; Congestion control; Buffer size requirement; Fairness

1. Introduction

Delivering congestion signals is essential to the performance of computer networks. In TCP/IP, congestion signals from the network are used by the source to determine the load. When a packet is acknowledged, the source increases its window size. When a congestion signal is received, its window size is reduced [1,2].

TCP/IP uses two methods to deliver congestion signals. The first method is timeout. When the source sends a packet, it starts a retransmission timer. If it does not receive an acknowledgment within a certain time, it assumes that congestion has happened in the network and the packet has been lost. Timeout is the slowest congestion signal because the source has to wait a long time for the retransmission timer to expire.

The second method is loss detection. In this method, the receiver sends a duplicate ACK immediately on reception of each out-of-sequence packet. The source interprets the reception of three duplicate acknowledgments as a congestion packet loss. Loss detection can avoid the long wait of timeout.

Both timeout and loss detection use packet losses as congestion signals. Packet losses not only increase the traffic in the network but also add large transfer delay. The explicit congestion notification (ECN) proposed in [3,4] provides a

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light-weight mechanism for routers to send a direct indication of congestion to the source. It makes use of two experimental bits in the IP header and two in the TCP header. When the average queue length exceeds a threshold, the incoming packet is marked as *congestion experienced*, with a probability calculated from the average queue length. When the marked packet is received, the receiver marks the acknowledgment using an *ECN-Echo* bit in the TCP header to send congestion notification back to the source. Upon receiving the ECN-Echo, the source halves its congestion window to help alleviate the congestion.

Many authors have pointed out that marking provides more information about the congestion state than packet dropping [5,6], and ECN has been proven to be a better way to deliver congestion signals and to exhibit better performance [4,5,7].

In most ECN implementations, when congestion happens, the congested router marks the incoming packet that just entered the queue. When the buffer is full or when a packet needs to be dropped, as in random early detection (RED), some implementations, such as the *ns* simulator [8], have the "drop-from-front" option, as suggested by Yin [9] and Lakshman [10]. A brief discussion of drop-from-front in RED can be found in [11]. However, for packet marking, these implementations still pick the incoming packet and not the front packet. We call this policy "mark-tail".

In this paper, we propose a simple marking mechanism – the "mark-front" strategy. This strategy marks a packet when the packet is going to leave the queue and the queue length is greater than a predetermined threshold. The mark-front strategy is different from the current mark-tail policy in two ways. First, since the router marks the packet at the time when it is sent, and not at the time when the packet is received, a more up-to-date congestion signal is carried by the marked packet. Second, since the router marks the packet in the front of the queue and not the incoming packet, congestion signals do not undergo queueing delay as do the data packets. In this way, a faster congestion feedback is delivered to the source.

The implementation of this strategy is extremely simple. One only needs to move the marking action from the enqueue procedure to the dequeue procedure and choose the packet leaving the queue instead of the packet entering the queue.

We justify the mark-front strategy by studying its benefits. We find that, by providing faster congestion signals, the mark-front strategy reduces the buffer size requirement at the routers; it avoids packet losses and thus improves the link efficiency when the buffer size in routers is limited. Our simulations also show that mark-front strategy improves the fairness among old and new users and alleviates TCP's discrimination against connections with large round-trip time (RTT).

The mark-front strategy differs from the drop-from-front option in that, when packets are dropped, only implicit congestion feedback can be inferred from timeout or duplicate ACKs; when packets are marked, explicit and faster congestion feedback is delivered to the source.

Gibbons and Kelly [6] suggested a number of mechanisms for packet marking, such as "marking all the packets in the queue at the time of packet loss", "marking every packet leaving the queue from the time of packet loss until the queue becomes empty", and "marking packets randomly as they leave the queue with a probability so that later packets will not be lost". Our mark-front strategy differs from these marking mechanisms in that it is a simple marking rule that faithfully reflects the up-to-date congestion status, while the mechanisms suggested by Gibbons and Kelly either do not reflect the correct congestion status or need sophisticated probability calculations for which no sound algorithm is known.

It is worth mentioning that the mark-front strategy is as effective in high speed networks as in low speed networks. Lakshman and Madhow [12] showed that the amount of drop-tail switches should be at least two to three times the bandwidth-delay product of the network in order for TCP to achieve decent performance and to avoid losses in the slow start phase. Our analysis in Section 4.3 reveals that, in the steady-state congestion avoidance phase, the queue size fluctuates from empty to one bandwidth-delay product. So the queueing delay experienced by packets when

congestion happens is comparable to the fixed round-trip time. [1] Therefore, the mark-front strategy can save as much as a fixed round-trip time in congestion signal delay, independent of link speed.

This paper is organized as follows. In Section 2, we describe the assumptions for our analysis. The dynamics of queue growth with TCP window control is studied in Section 3. In Section 4, we compare the buffer size requirements of mark-front and mark-tail strategies. In Section 5, we explain why mark-front is fairer than mark-tail. The simulation results that verify our conclusions are presented in Section 6. In Section 7, we remove the assumptions made to facilitate the analysis and apply the mark-front strategy to the RED algorithm. Simulation results show that mark-front has advantages over mark-tail, as revealed by the analysis.

2. Assumptions

ECN is used together with TCP congestion control mechanisms like slow start and congestion avoidance [2]. When the acknowledgment is not marked, the source follows existing TCP algorithms to send data and increase the congestion window. Upon the receipt of an ECN-Echo, the source halves its congestion window and reduces the slow start threshold. In the case of packet loss, the source follows the TCP algorithm to reduce the window and retransmit the lost packet.

ECN delivers congestion signals by setting the *congestion experienced* bit, but determining when to set the bit depends on the congestion detection policy. In [3], ECN is proposed to be used with average queue length and RED. The goal is to avoid sending congestion signals caused by transient traffic and to desynchronize sender windows [13,14]. In this paper, to allow analytical modeling, we assume a simplified congestion detection criterion: when the *actual queue length* is smaller than the threshold, the incoming packet will not be marked; when the *actual queue length* exceeds the threshold, the incoming packet will be marked.

We also make the following assumptions: (1) Receiver windows are large enough so that the bottleneck is in the network. (2) Senders always have data to send and will send as many packets as their windows allow. (3) There is only one bottleneck link that causes queue buildup. (4) Receivers acknowledge every packet received and there are no delayed acknowledgments. (5) There is no ACK compression [15]. (6) The queue length is measured in packets and all packets have the same size.

3. Queue dynamics with TCP window control

In this section, we study the relationship between the window size at the source and the queue size at the congested router. The purpose is to show the difference between mark-tail and mark-front strategies. Our analysis is made on one connection, but with small modifications, it can also apply to multiple connection cases. Simulation results of multiple connections and connections with different round trip time will be presented in Section 6.

In a path with one connection, the only bottleneck is the first link with the lowest rate in the entire route. In the case of congestion, the queue builds up only at the router before the bottleneck link. The following lemma is obvious:

Lemma 1. *If the data rate of the bottleneck link is d packets per second, then the downstream packet inter-arrival time and the ACK inter-arrival time on the reverse link cannot be shorter than* $1/d$ *seconds. If the bottleneck link is fully loaded (i.e., no idling), then the downstream packet inter-arrival time and the ack inter-arrival time on the reverse link are* $1/d$ *seconds.*

Denoting the source window size at time t as $w(t)$, we then have

Theorem 1. *Consider a path with only one connection and only one bottleneck link. Let the fixed round-trip time be r seconds, the bottleneck link rate*

[1] The fixed round-trip time is the round-trip time under light load, i.e., without queueing delay.

d packets per second, and the propagation and transmission time between the source and bottleneck router be t_p. If the bottleneck link has been busy for at least r seconds, and a packet just arrived at the congested router at time t, then the queue length at the congested router is

$$Q(t) = w(t - t_p) - rd. \quad (1)$$

Proof. Consider the packet that just arrived at the congested router at time t. It was sent by the source at time $t - t_p$. At that time, the number of packets on the path and outstanding acks on the reverse link was $w(t - t_p)$. By time t, $t_p d$ acks are received by the source. All packets between the source and the router have entered the congested router or have been sent downstream. As shown in Fig. 1, the pipe length from the congested router to the receiver and then back to the source is $r - t_p$. The number of downstream packets and outstanding acks are $(r - t_p)d$. The rest of the $w(t - t_p)$ unacknowledged packets are still in the congested router. So the queue length is

$$\begin{aligned} Q(t) &= w(t - t_p) - t_p d - (r - t_p)d \\ &= w(t - t_p) - rd. \end{aligned} \quad (2)$$

This finishes the proof. □

Notice that, in this theorem, we did not use the number of packets between the source and the congested router to estimate the queue length, because the packets downstream from the congested router and the acks on the reverse link are equally spaced, but the packets between the source and the congested router may not be.

The analysis in this theorem is based on the assumptions in Section 2. The conclusion applies to both slow start and congestion avoidance phases. In order for Eq. (1) to hold, the router must have been congested for at least r seconds.

4. Buffer size requirement and threshold setting

When ECN signals are used for congestion control, the network can achieve zero packet loss. When acknowledgments are not marked, the source gradually increases the window size. Upon the receipt of an ECN-Echo, the source halves its congestion window to reduce the congestion.

In this section, we analyze the buffer size requirement for both mark-tail and mark-front strategies. The result also includes an analysis on how to set the threshold.

4.1. Mark-tail strategy

Suppose P was the packet that increased the queue length over the threshold T, and it was sent from the source at time s_0 and arrived at the congested router at time t_0. Its acknowledgment, which was an ECN-Echo, arrived at the source at time s_1 and the window was reduced at the same time. We also assume that the last packet before the window reduction was sent at time s_1^- and arrived at the congested router at time t_1^-.

In order to use Theorem 1, we need to consider two cases separately: when T is large and when T is small, compared to rd.

Case 1. If T is reasonably large (about rd) such that the buildup of a queue of size T needs r time, the assumption in Theorem 1 is satisfied and we have

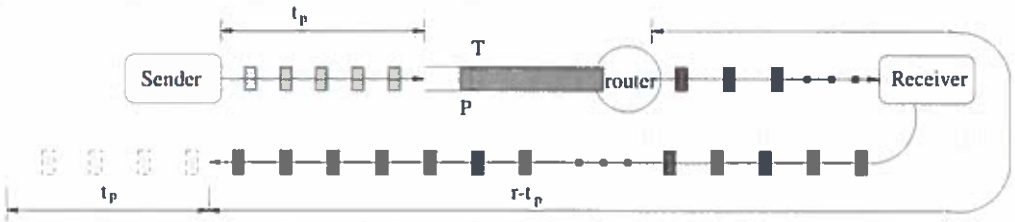


Fig. 1. Calculation of queue length.

$$T = Q(t_0) = w(t_0 - t_p) - rd = w(s_0) - rd, \quad (3)$$

so

$$w(s_0) = T + rd. \quad (4)$$

Since the time elapse between s_0 and s_1 is one RTT, if packet P were not marked, the congestion window would increase to $2w(s_0)$. Since P was marked, the congestion window before receiving the ECN-Echo was

$$w(s_1^-) = 2w(s_0) - 1 = 2(T + rd) - 1. \quad (5)$$

When the last packet sent under this window reached the router at time t_1^-, the queue length was

$$\begin{aligned} Q(t_1^-) &= w(s_1^-) - rd \\ &= 2w(s_0) - 1 - rd \\ &= 2T + rd - 1. \end{aligned} \quad (6)$$

Upon the receipt of the ECN-Echo, the congestion window was halved. The source cannot send any more packets before half of the packets are acknowledged. So $2T + rd - 1$ is the maximum queue length.

Case 2. If T is small, rd is an overestimate of the number of downstream packets and acks on the reverse link.

$$\begin{aligned} w(s_0) &= T + \text{number of downstream} \\ &\quad \text{packets and acks} \\ &\leqslant T + rd. \end{aligned} \quad (7)$$

Therefore,

$$\begin{aligned} Q(t_1^-) &= w(s_1^-) - rd \\ &= (2w(s_0) - 1) - rd \\ &\leqslant 2(T + rd) - 1 - rd \\ &= 2T + rd - 1. \end{aligned} \quad (8)$$

So, in both cases, $2T + rd - 1$ is an upper bound of the queue length that can be reached in the slow start phase.

Theorem 2. *In a TCP connection with ECN congestion control, if the fixed round trip time is r seconds, the bottleneck link rate is d packets per second, and the bottleneck router uses threshold T for congestion detection, then the maximum queue length that can be reached in slow start phase is less than or equal to* $2T + rd - 1$.

As shown by Eq. (6), when T is large, the bound $2T + rd - 1$ can be reached with equality. When T is small, $2T + rd - 1$ is just an upper bound. Since the queue length in the congestion avoidance phase is smaller, this bound is actually the buffer size requirement.

4.2. Mark-front strategy

Suppose P was the packet that increased the queue length over the threshold T, and it was sent from the source at time s_0 and arrived at the congested router at time t_0. The router marked the packet P' that stood in the front of the queue. The acknowledgment of P', which was an ECN-Echo, arrived at the source at time s_1 and the window was reduced at the same time. We also suppose that the last packet before the window reduction was sent at time s_1^- and arrived at the congested router at time t_1^-.

Consider two cases separately: when T is large and when T is small.

Case 1. If T is reasonably large (about rd) such that the buildup of a queue of size T needs r time, the assumption in Theorem 1 is satisfied. We have

$$T = Q(t_0) = w(t_0 - t_p) - rd = w(s_0) - rd, \quad (9)$$

so

$$w(s_0) = T + rd. \quad (10)$$

In the slow start phase, the source increases the congestion window by one for every acknowledgment it receives. If the acknowledgment of P was received at the source without the congestion indication, the congestion window would have been doubled to

$$2w(s_0) = 2(T + rd).$$

However, when the acknowledgment of P' arrived, $T - 1$ acknowledgments corresponding to packets prior to P were still on the way. So the window size at time s_1^- was

$$w(s_1^-) = 2w(s_0) - (T - 1) - 1 = T + 2rd. \quad (11)$$

When the last packet sent under this window reached the router at time t_1^-, the queue length was

$$\begin{aligned} Q(t_1^-) &= w(s_1^-) - rd \\ &= T + 2rd - rd \\ &= T + rd. \end{aligned} \tag{12}$$

Upon the receipt of the ECN-Echo, the congestion window is halved. The source cannot send any more packets before half of the packets are acknowledged. So $T + rd$ is the maximum queue length.

Case 2. If T is small, rd is an overestimate of the number of downstream packets and acks on the reverse link.

$$\begin{aligned} w(s_0) &= T + \text{number of downstream} \\ &\quad \text{packets and acks} \\ &\leqslant T + rd. \end{aligned} \tag{13}$$

Therefore,

$$\begin{aligned} Q(t_1^-) &= w(s_1^-) - rd \\ &= (2w(s_0) - T) - rd \\ &\leqslant 2(T + rd) - T - rd \\ &= T + rd. \end{aligned} \tag{14}$$

So, in both cases, $T + rd$ is an upper bound of the queue length that can be reached in the slow start phase.

Theorem 3. *In a TCP connection with ECN congestion control, if the fixed round trip time is r seconds, the bottleneck link rate is d packets per second, and the bottleneck router uses threshold T for congestion detection, then the maximum queue length that can be reached in slow start phase is less than or equal to $T + rd$.*

Again, when T is large, Eq. (12) shows that the bound $T + rd$ is tight. Since the queue length in the congestion avoidance phase is smaller, this bound is actually the buffer size requirement.

Theorems 2 and 3 estimate the buffer size requirement for zero-loss ECN congestion control.

4.3. Threshold setting

In the congestion avoidance phase, the congestion window increases roughly by one in every RTT. Assuming that the mark-tail strategy is used, using the same timing variables as in the previous sections, we have

$$w(s_0) = Q(t_0) + rd = T + rd. \tag{15}$$

The congestion window increases roughly by one in an RTT,

$$w(s_1^-) = T + rd + 1. \tag{16}$$

When the last packet sent before the window reduction arrived at the router, it saw a queue length of $T + 1$:

$$Q(t_1^-) = w(s_1^-) - rd = T + 1. \tag{17}$$

Upon the receipt of the ECN-Echo, the window was halved:

$$w(s_1) = (T + rd + 1)/2. \tag{18}$$

The source may not be able to send packets immediately after s_1. After some packets were acknowledged, the halved window allowed new packets to be sent. The first packet sent under the new window saw a queue length of

$$\begin{aligned} Q(t_1) &= w(s_1) - rd \\ &= (T + rd + 1)/2 - rd \\ &= (T - rd + 1)/2. \end{aligned} \tag{19}$$

The congestion window was fixed for an RTT and then began to increase. So $Q(t_1)$ was the minimum queue length in a cycle. In summary, in the congestion avoidance phase, the maximum queue length is $T + 1$ and the minimum queue length is $(T - rd + 1)/2$.

In order to avoid link idling, we should have $(T - rd + 1)/2 \geqslant 0$ or, equivalently, $T \geqslant rd - 1$. On the other hand, if the minimum queue is always positive, the router keeps an unnecessarily large queue and all packets suffer a long queueing delay. Therefore, the best choice of threshold should satisfy

$$(T - rd + 1)/2 = 0, \tag{20}$$

or

$$T = rd - 1. \tag{21}$$

If the mark-front strategy is used, the source's congestion window increases roughly by one in every RTT, but congestion feedback travels faster than data packets. Hence

$$Q(s_1^-) = T + rd + \varepsilon, \tag{22}$$

where ε is between 0 and 1, and depends on the location of the congested router. Therefore,

$$Q(t_1^-) = w(s_1^-) - rd = T + \varepsilon, \tag{23}$$

$$w(s_1) = (T + rd + \varepsilon)/2, \tag{24}$$

$$\begin{aligned} Q(t_1) &= w(s_1) - rd \\ &= (T + rd + \varepsilon)/2 - rd \\ &= (T - rd + \varepsilon)/2. \end{aligned} \tag{25}$$

For the reason stated above, the best choice of threshold is $T = rd - \varepsilon$. Compared with rd, the difference between $rd - \varepsilon$ and $rd - 1$ can be ignored. So we have the following theorem:

Theorem 4. *In a path with only one connection, the optimal threshold that achieves full link utilization while keeping queueing delay minimal in congestion avoidance phase is* $rd - 1$. *If the threshold is smaller than this value, the link will be under-utilized. If the threshold is greater than this value, the link can be fully utilized, but packets will suffer an unnecessarily large queueing delay.*

Combining the results in Theorems 2–4, we can see that the mark-front strategy reduces the buffer size requirement from about $3rd$ to $2rd$. It also reduces the congestion feedback's delay by one fixed round-trip time.

5. Lock-out phenomenon and fairness

One of the weaknesses of the mark-tail policy is its discrimination against new flows. Consider the time when a new flow joins the network, but the buffer of the congested router is occupied by packets of old flows. In the mark-tail strategy, the packet that just arrived will be marked, but the packets already in the buffer will be sent without being marked. The acknowledgments of the sent packets will increase the window size of the old flows. Therefore, the old flows, which already have a large share of the resources, will grow even larger. However, the new flow with a small or no share of the resources has to back off, since its window size will be reduced by the marked packets. This causes a "lock-out" phenomenon, in which a single connection or a few flows monopolize(s) the buffer space and prevent(s) other connections from getting room in the queue [16]. Lock-out leads to gross unfairness among users and is clearly undesirable.

Contrary to the mark-tail policy, the mark-front strategy marks the packets in the buffer first. Connections with large buffer occupancy will have more packets marked than connections with small buffer occupancy. Compared with the mark-tail strategy that let the packets in the buffer escape the marking, the mark-front strategy helps prevent the lock-out phenomenon. Therefore, we can expect the mark-front strategy to be fairer than the mark-tail strategy.

TCP's discrimination against connections with large RTT is also well known. The cause of this discrimination is similar to that against new connections. If connections with small RTT and large RTT start at the same time, the connections with small RTT will receive their acknowledgments faster and therefore grow faster. When congestion happens, connections with small RTT will take more buffer room than connections with large RTT. With the mark-tail policy, packets already in the queue will not be marked but only newly arrived packets will be marked. Therefore, connections with small RTT will grow even larger, but connections with large RTT have to back off. Mark-front alleviates this discrimination by treating all packets in the buffer equally. Packets already in the buffer may also be marked. Therefore, connections with large RTT can have larger bandwidth.

6. Simulation results

In order to compare the mark-front and mark-tail strategies, we performed a set of simulations with the *ns* simulator [8]. We modified the RED algorithm in the *ns* simulator to deterministically mark the packets when the real queue length exceeded the threshold. The basic simulation model is shown in Fig. 2. A number of sources $s_1, s_2, \ldots, s_m$ are connected to the router r_1 by a 10 Mbps links, router r_1 is connected to r_2 by a

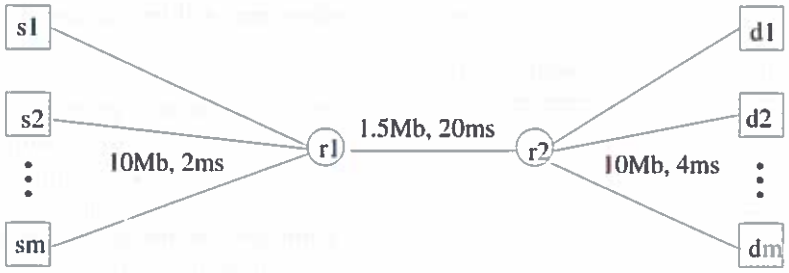


Fig. 2. Simulation model.

1.5 Mbps link, and destinations $d_1, d_2, \ldots, d_m$ are connected to r_2 by 10 Mbps links. The link speeds are chosen so that congestion will only happen at router r_1, where mark-tail and mark-front strategies are tested.

With the basic configuration shown in Fig. 2, the fixed round trip time, including the propagation time and the transmission time at the routers, is 59 ms. Changing the propagation delay between router r_1 and r_2 from 20 to 40 ms gives an RTT of 99 ms. Changing the propagation delays between the sources and router r_1 gives us configurations of different RTT. An FTP application runs on each source. Reno TCP and ECN are used for congestion control. The data packet size, including all headers, is 1000 bytes and the acknowledgment packet size is 40 bytes. With the basic configuration,

$rd = 0.059 \times 1.5 \times 10^6$ bits

$= 11062.5$ bytes

≈ 11 packets.

In our simulations, the routers perform mark-tail or mark-front. The results for both strategies are compared.

6.1. Simulation scenarios

In order to show the difference between mark-front and mark-tail strategies, we designed the following simulation scenarios based on the basic simulation model described in Fig. 2. If not specified, all connections have an RTT of 59 ms, start at the zeroth second and stop at the 10th second.

1. One connection.
2. Two connections with the same RTT.
3. Two overlapping connections with the same RTT, but the first connection starts at the zeroth second and stops at the ninth second, while the second connection starts at the first second and stops at the 10th second.
4. Two connections with RTT equal to 59 and 157 ms, respectively.
5. Two connections with the same RTT, but the buffer size at the congested router is limited to 25 packets.
6. Five connections with the same RTT.
7. Five connections with RRT of 59, 67, 137, 157 and 257 ms, respectively.
8. Five connections with the same RTT, but the buffer size at the congested router is limited to 25 packets.

Scenarios 1, 4, 6 and 7 are mainly designed for testing the buffer size requirement. Scenarios 1, 3, 4, 6, 7 and 8 are for link efficiency, and scenarios 2, 3, 4, 5, 6 and 7 are for fairness among users.

6.2. Metrics

We use three metrics to compare the two strategies. The first metric is the *buffer size requirement* for zero-loss congestion control. This is the maximum queue size that can be built up at the router in the slow start phase before the congestion signal takes effect at the congested router. If the buffer size is greater than or equal to this value, no packet loss will happen. This metric is measured as the maximum queue length in the entire simulation.

The second metric, *link efficiency*, is calculated from the number of acknowledged packets (not counting the retransmissions) divided by the possible number of packets that can be transmitted during the simulated time. Because of the slow start phase and possible link idling after window

reduction, the link efficiency is always smaller than 1. Link efficiency should be measured with long simulation time to minimize the effect of the initial transient state. We tried different simulation times from 5 to 100 s. The results for 10 s show the essential features of the strategy, without much difference from the results for 100 s. So the simulation results presented in this paper are based on 10-s simulations.

The third metric, the *fairness* index, is calculated according to the formula in [17]. If m connections share the bandwidth, and x_i is the number of acknowledged packets of connection i, then the *fairness* index is calculated as

$$\text{fairness} = \frac{(\sum_{i=1}^{m} x_i)^2}{m \sum_{i=1}^{m} x_i^2}. \tag{26}$$

The fairness index is often close to 1. In our graphs, we draw the *unfairness* index

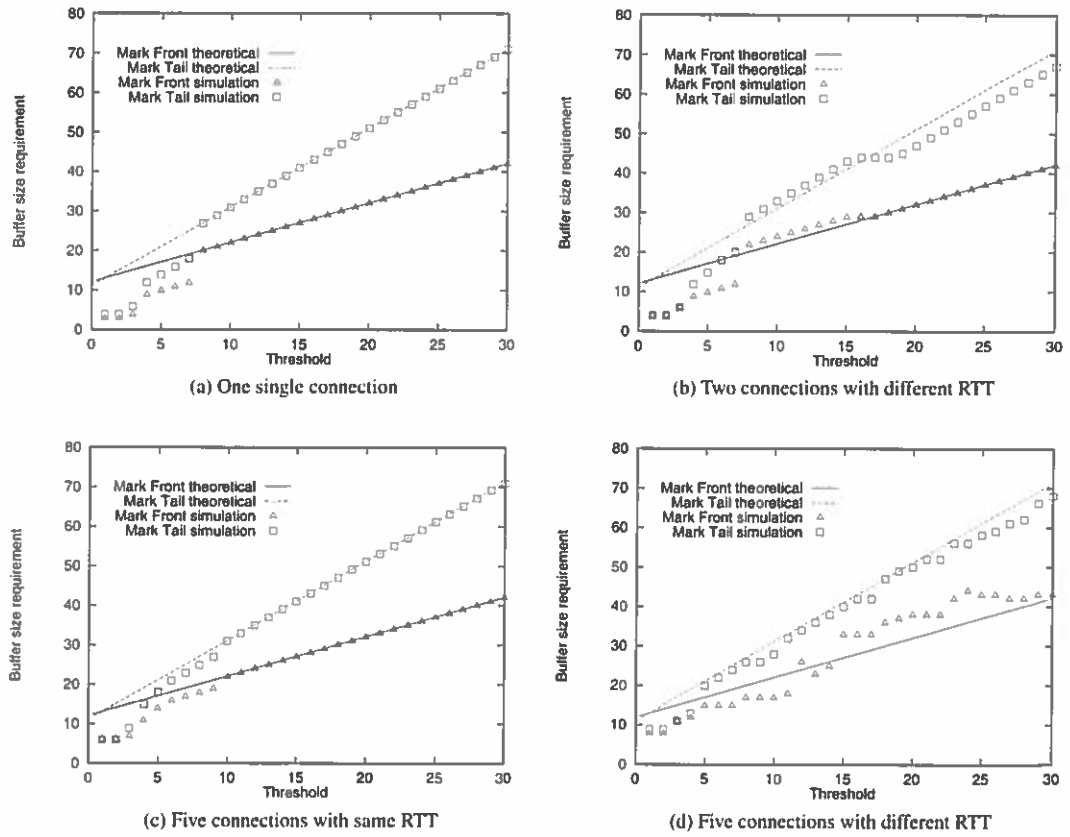
$$\text{unfairness} = 1 - \text{fairness}. \tag{27}$$

The performance of ECN depends on the selection of the threshold value. In our results, all three metrics are drawn for different values of threshold.

6.3. Buffer size requirement

Fig. 3 shows the buffer size requirement for mark-tail and mark-front. The measured maximum queue lengths are shown with "□" and "△". The corresponding theoretical estimates from Theorems 2 and 3 are shown with dashed and solid lines. In Figs. 3(b) and (d), where the connections have different RTT, the theoretical estimate is calculated from the smallest RTT.

From the simulation, we find that, for connections with the same RTT, the theoretical estimate of buffer size requirement is accurate. When threshold T is small, the buffer size requirement is



(a) One single connection

(b) Two connections with different RTT

(c) Five connections with same RTT

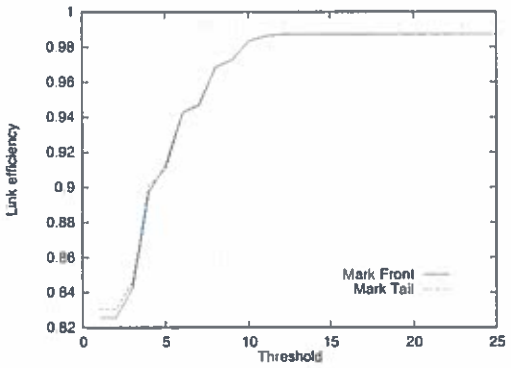
(d) Five connections with different RTT

Fig. 3. Buffer size requirement in various scenarios.

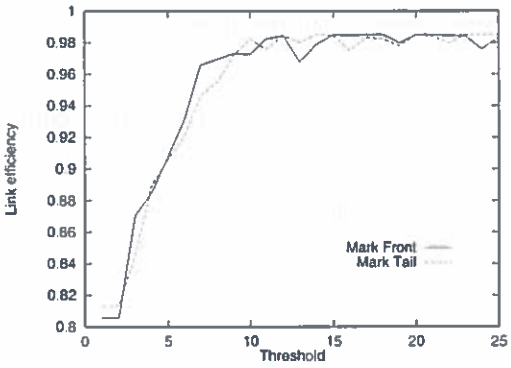
an upper bound; when $T \geqslant rd$, the upper bound is tight. For connections with different RTT, the estimate given by the largest RTT is an upper bound, but is usually an overestimate. The estimate given by the smallest RTT is a closer approximation.

6.4. Link efficiency

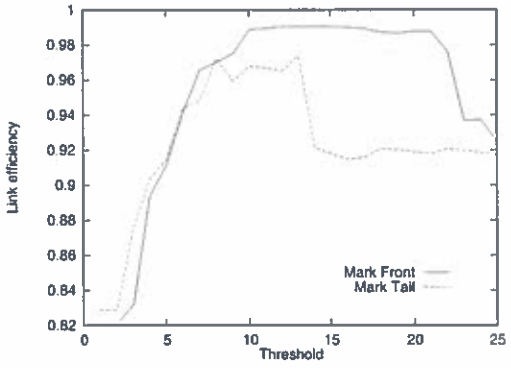
Fig. 4 shows the link efficiency for various scenarios. In all cases, the efficiency increases with the threshold, until the threshold is about rd, where the link reaches almost full utilization.



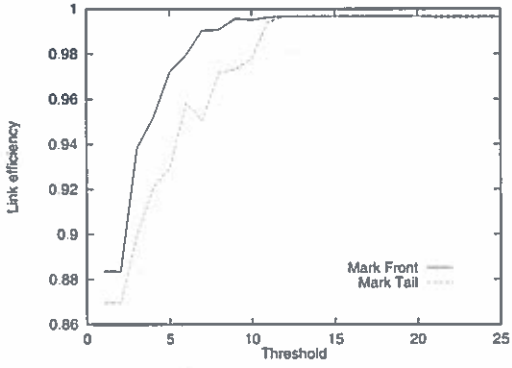
(a) One single connection



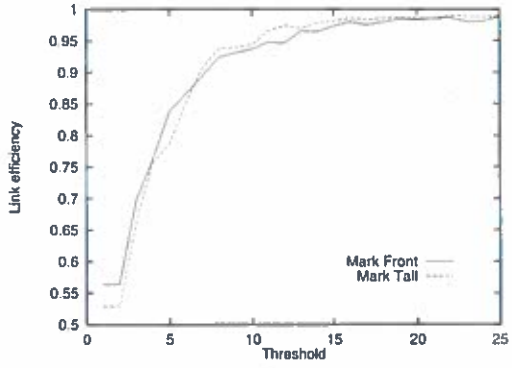
(b) Two overlapping connections



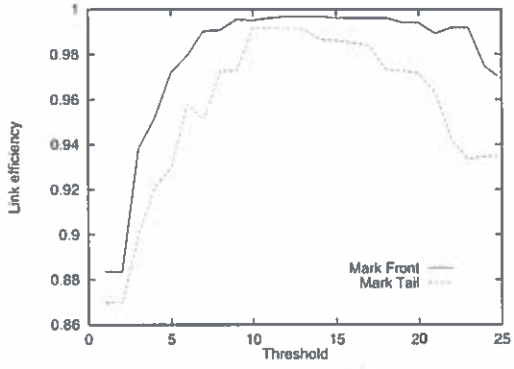
(c) Two same connections, limited buffer



(d) Five connections with same RTT



(e) Five connections with different RTT



(f) Five same connections, limited buffer

Fig. 4. Link efficiency in various scenarios.

Small threshold results in low link utilization because it generates congestion signals even when the router is not really congested. Unnecessary window reduction actions taken by the source lead to link idling. The link efficiency results in Fig. 4 verify the choice of threshold stated in Theorem 4.

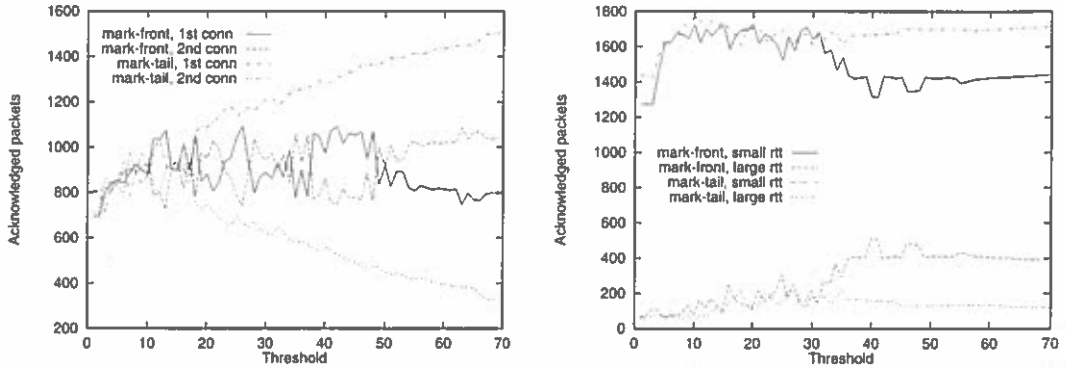
In the unlimited buffer cases (a), (b), (d) and (e), the difference between mark-tail and mark-front is small. However, when the buffer size is limited, as in cases (c) and (f), mark-front has much better link efficiency. This is because, when congestion happens, the mark-front strategy provides a faster congestion feedback than mark-tail. Faster congestion feedback prevents the source from sending more packets that will be dropped at the congested router. Multiple drops cause source timeout and idling at the bottleneck link, and thus low utilization. This explains the drop of link efficiency in Figs. 4(c) and (f) when the threshold exceeds about 10 packets in mark-tail and about 20 packets in mark-front.

6.5. Fairness

Scenarios 2–7 are designed to test the fairness of the two marking strategies. Fig. 5 shows the lock-out phenomenon and alleviation by the mark-front strategy. With the mark-tail strategy, old connections occupy the buffer and lock out new connections. Although the two connections in scenario 3 have the same time span, the number of acknowledged packets in the first connection is much larger than that in the second connection, Fig. 5(a). In scenario 4, the connection with the large RTT (157 ms) starts at the same time as the connection with the small RTT (59 ms), but the connection with the small RTT grows faster, takes over a large portion of the buffer room and locks out the connection with the large RTT. Of all of the bandwidth, only 6.49% is allocated to the connection with the large RTT. The mark-front strategy alleviates the discrimination against large RTT by marking packets already in the buffer. Simulation results show that the mark-front strategy improves the portion of bandwidth allocated to connection with large RTT from 6.49% to 21.35%.

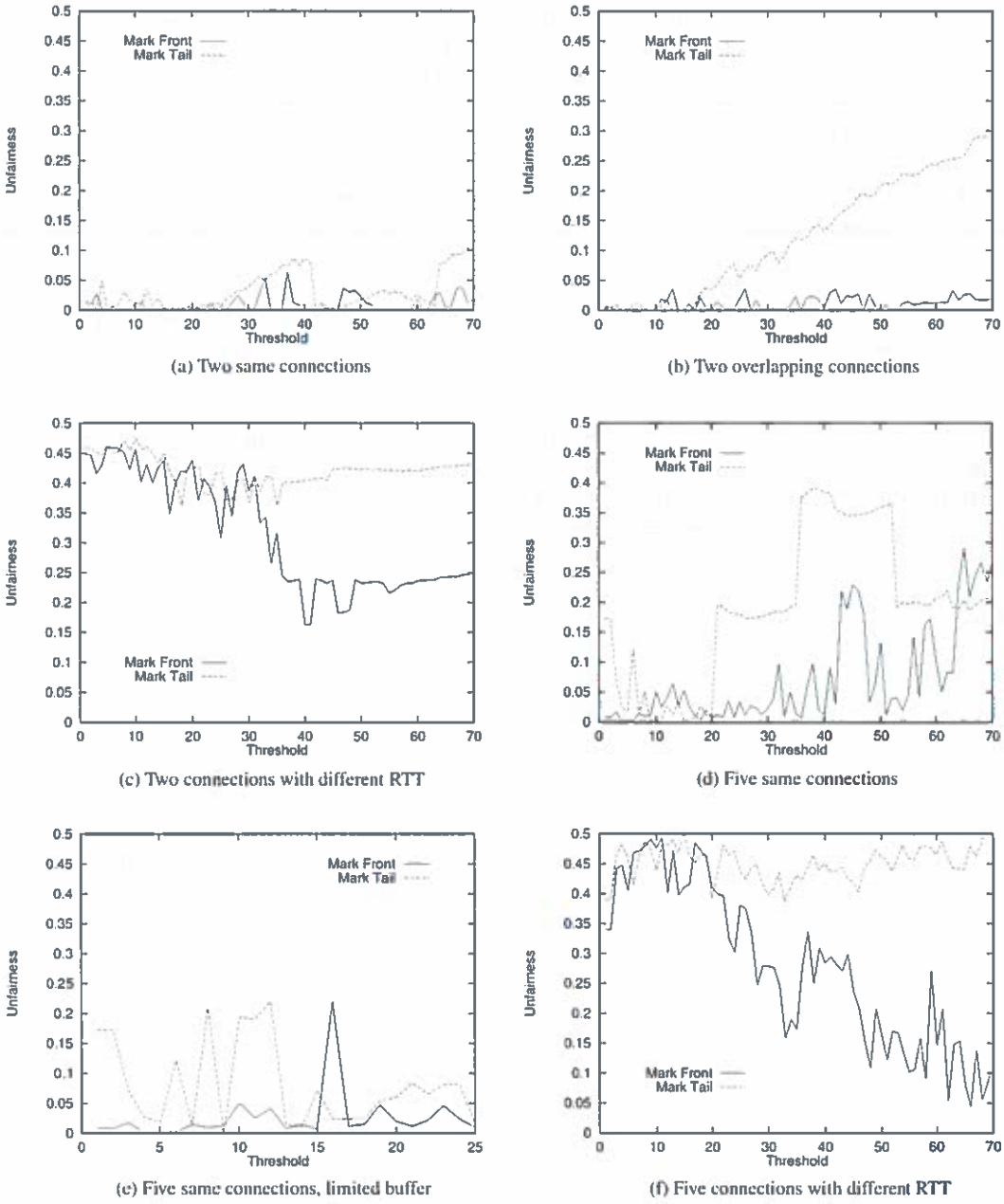
Fig. 6 shows the unfairness index for the mark-tail and mark-front strategies. In Fig. 6(a), the two connections have the same configuration. Which connection receives more packets than the other is not deterministic, so that the unfairness index seems random. However, in general, mark-front has a smaller unfairness index than mark-tail.

In Fig. 6(b), the two connections are different: the first connection starts first and takes the buffer room. Although the two connections have the same time span, if the mark-tail strategy is used, the second connection is locked out by the first and, therefore, receives fewer packets. Mark-front avoids this lock-out phenomenon. The results



(a) Two overlapping connections (b) Two connections with different RTT

Fig. 5. Lock-out phenomenon and its alleviation by the mark-front strategy.



(a) Two same connections

(b) Two overlapping connections

(c) Two connections with different RTT

(d) Five same connections

(e) Five same connections, limited buffer

(f) Five connections with different RTT

Fig. 6. Unfairness in various scenarios.

show that the unfairness index of mark-front is much smaller than that of mark-tail. In addition, as the threshold increases, the unfairness index of mark-tail increases, but that of mark-front remains roughly the same, regardless of the threshold.

Fig. 6(c) shows the difference between connections with different RTT. With the mark-tail

strategy, the connections with small RTT grow faster and therefore lock out the connections with large RTT. Since the mark-front strategy does not have the lock-out problem, the discrimination against connections with large RTT is alleviated. The difference between the two strategies is obvious when the threshold is large.

Fig. 6(e) shows the unfairness index when the router buffer size is limited. In this scenario, when the buffer is full, the router drops the packet in the front of the queue. Whenever a packet is sent, the router checks whether the current queue size is larger than the threshold. If yes, the packet is marked. The figure shows that mark-front is fairer than mark-tail.

Similar results for five connections are shown in Figs. 6(d) and (f).

7. Application to RED

The analytical and simulation results obtained in the previous sections are based on the simplified congestion detection model that a packet leaving a router is marked if the actual queue size of the router exceeds the threshold. However, RED uses a different congestion detection criterion. First, RED uses average queue size instead of actual queue size. Second, a packet is not marked deterministically, but with a probability calculated from the average queue size.

In this section, we apply the mark-front strategy to the RED algorithm and compare the results with those of the mark-tail strategy. Because of the difficulty in analyzing RED mathematically, the comparison is carried out by simulations only.

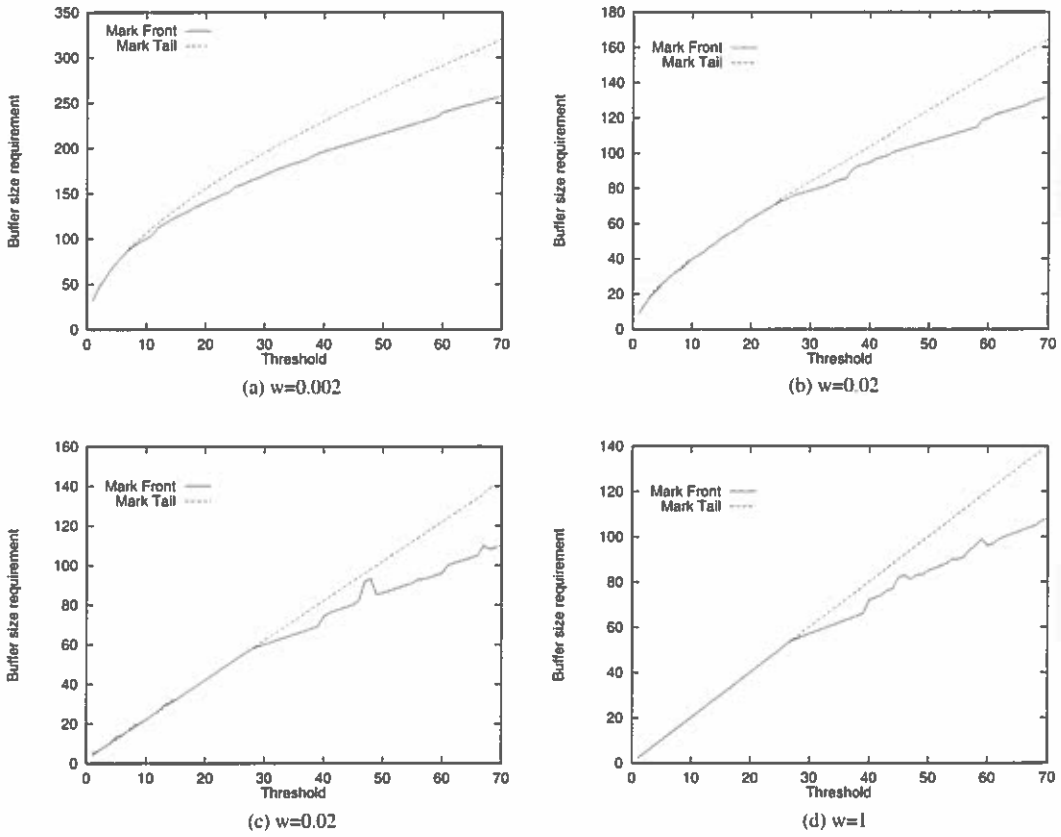


Fig. 7. Buffer size requirement for different queue weights, $p_{max} = 0.1$.

The RED algorithm needs four parameters: queue weight w, minimum threshold th_{min}, maximum threshold th_{max} and maximum marking probability p_{max}. Although determining the best RED parameters is out of the scope of this paper, we have tested several hundreds of combinations. In almost all these combinations, mark-front has better performance than mark-tail in terms of buffer size requirement, link efficiency and fairness.

Instead of presenting individual parameter combinations for all scenarios, we focus on one scenario and present the results for a range of parameter values. The simulation scenario is the scenario 3 of two overlapping connections described in Section 6.1. Based on the recommendations in [13], we vary the queue weight w for four values: 0.002, 0.02, 0.2 and 1, vary th_{min} from 1 to 70, fix th_{max} as $2th_{min}$, and fix p_{max} as 0.1.

Fig. 7 shows the buffer size requirement for both strategies with different queue weights. In all cases, the mark-front strategy requires a smaller buffer size than the mark-tail. The results also show that queue weight w is a major factor affecting buffer size requirement. Smaller queue weights require larger buffers. When the actual queue size is used (corresponding to $w = 1$), RED requires the minimum buffer size.

Fig. 8 shows link efficiency. For almost all values of threshold, mark-front provides better link efficiency than mark-tail. Contrary to common belief, the actual queue size (Fig. 8(d)) is no worse than the average queue size (Fig. 8(a)) in achieving higher link efficiency.

The queue size trace at the congested router shown in Fig. 9 provides some explanation for the smaller buffer size requirement and higher

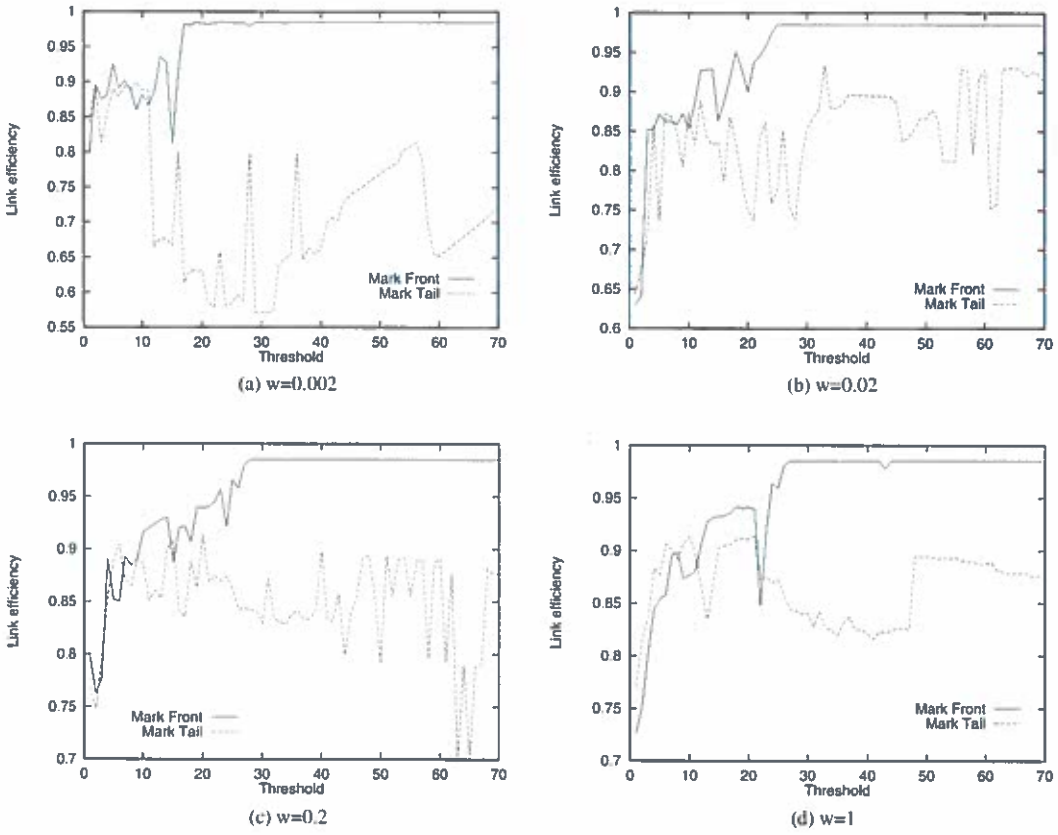


Fig. 8. Link efficiency for different queue weights, $p_{max} = 0.1$.

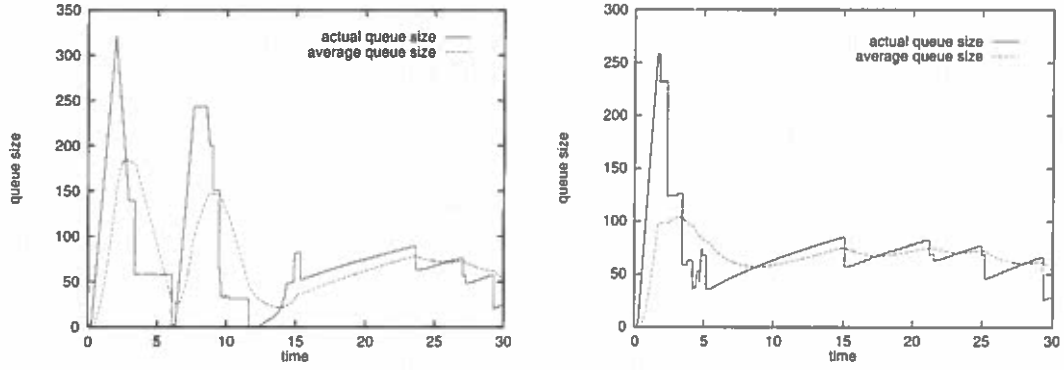


Fig. 9. Change of queue size at the congested router, $w = 0.002, th_{\min} = 70, th_{\max} = 140, p_{\max} = 0.1$.

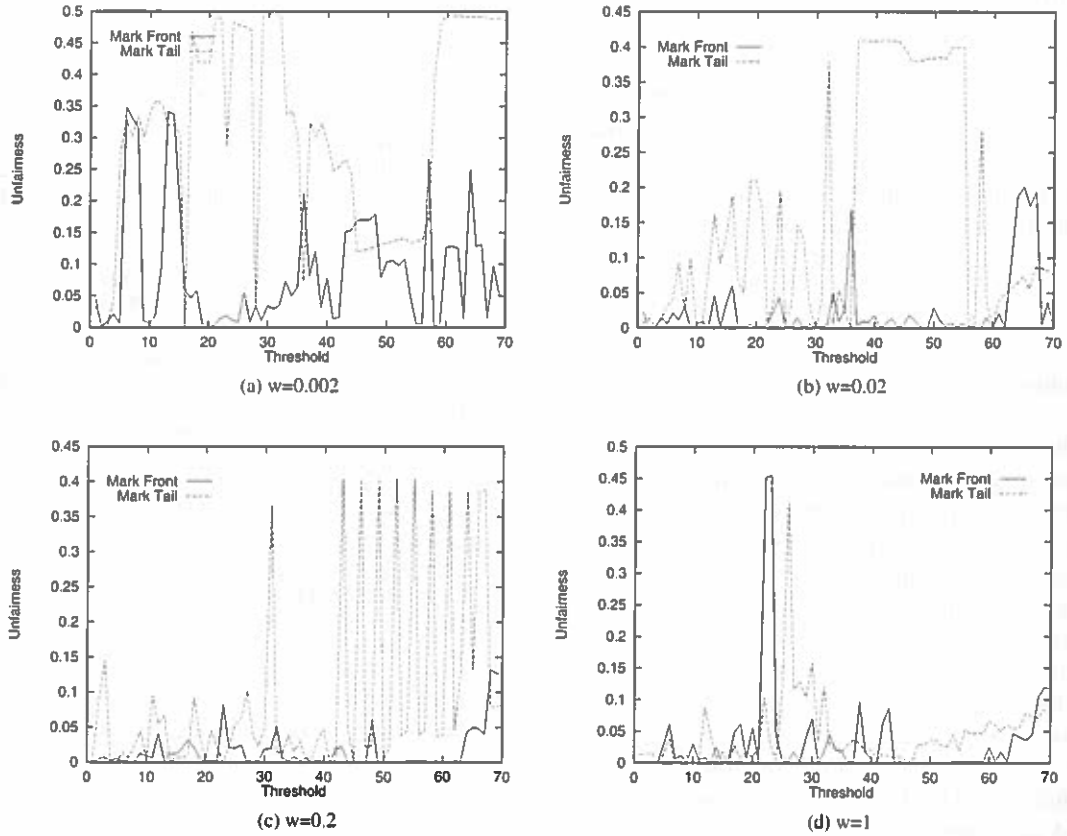


Fig. 10. Unfairness for different queue weights, $p_{\max} = 0.1$.

efficiency of the mark-front strategy. When congestion happens, mark-front delivers faster congestion feedback than mark-tail, so the sources can stop sending packets earlier. In Fig. 9(a), with the mark-tail signal, the queue size stops increasing at 1.98 s. With the mark-front signal, the queue size stops increasing at 1.64 s. Therefore the mark-front strategy needs a smaller buffer.

On the other hand, when congestion is relieved, mark-tail is slow in reporting the change of congestion status. Packets leaving the router still carry the congestion information set at the time when they entered the queue. Even if the queue is empty, these packets still tell the sources that the router is congested. This out-dated congestion information is responsible for the link idling around the sixth and 12th seconds in Fig. 9(a). As a comparison, in Fig. 9(b), the same packets carry more up-to-date congestion information to tell the sources that the router is no longer congested, so the sources send more packets in time. Thus, the mark-front signal helps avoid link idling and improve efficiency.

Fig. 10 shows the unfairness index. Both mark-front and mark-tail have big oscillations in the unfairness index when the threshold changes. These oscillations are caused by the randomness of how many packets of each connection get marked in the bursty TCP slow start phase. Changing the threshold value can significantly change the number of marked packets of each connection. In spite of the randomness, mark-front is fairer than mark-tail in most cases.

8. Conclusion

In this paper, we analyze the mark-front strategy used in ECN. Instead of marking the packet from the tail of the queue, this strategy marks the packet in the front of the queue and thus delivers faster congestion signals to the source. Compared with the mark-tail policy, the mark-front strategy has three advantages. First, it reduces the buffer size requirement at the routers. Second, it provides more up-to-date congestion information to help the source adjust its window in time to avoid packet losses and link idling and thus improves link efficiency. Third, it improves fairness among old and new users and helps alleviate TCP's discrimination against connections with large round-trip time.

With a simplified model, we analyze the buffer size requirement for both mark-front and mark-tail strategies. Link efficiency, fairness and more complicated scenarios are tested with simulations. The results show that the mark-front strategy achieves better performance than the current mark-tail policy. We also apply the mark-front strategy to the RED algorithm. Simulations show that the mark-front strategy used with RED has similar advantages over mark-tail.

Based on the analysis and the simulations, we conclude that mark-front is an easy-to-implement improvement that provides better congestion control and therefore helps TCP achieve smaller buffer size requirement, higher link efficiency and better fairness among users.

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