On Individual and Aggregate TCP Performance

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Abstract

As the most widely used reliable transport in today's Internet, TCP has been extensively studied in the past decade. However, previous research usually only considers a small or medium number of concurrent TCP connections. The TCP behavior under many competing TCP flows has not been sufficiently explored.

In this paper we use extensive simulations to investigate the individual and aggregate TCP performance for large number of concurrent TCP flows. We have made three major contributions. First, we develop an abstract network model that captures the essence of wide-area Internet connections. Second, we study the performance of a single TCP flow with many competing TCP flows by evaluating the best-known analytical model proposed in the literature. Finally, we examine the aggregate TCP behavior exhibited by many concurrent TCP flows, and derive general conclusions about the overall throughput, goodput, and loss probability.

1 Introduction

TCP is the most widely used reliable transport today. It has used to carry a significant amount of Internet traffic, including WWW (HTTP), file transfer (FTP), email (SMTP) and remote access (Telnet) traffic. Due to its importance, TCP has been extensively studied in the past ten years. However, previous research usually only considers a small or medium number of concurrent TCP connections. The TCP behavior under many competing TCP flows has not been sufficiently explored.

In this paper, we use extensive simulations to explore the performance of TCP-Reno, one of the most commonly used TCP flavors in the current Internet. We first develop a generic model that abstracts an Internet connection by exploring the Internet hierarchical routing structure. Based on the abstract model, we study the behavior of a single TCP connection under many competing TCP flows by evaluating the TCP analytical model proposed in [13]. We also investigate the aggregate behavior of many concurrent TCP flows, and derive general conclusions about the overall throughput, goodput, loss and probability.

The rest of the paper is organized as follows. Section 2 presents our abstract network model for wide-area Internet connections. Section 3 studies the performance of a single TCP flow under cross traffic by evaluating the TCP analytical model proposed in [13]. Section 4 further simplifies our network model in order to study the aggregate behavior of TCP connections. Section 5 presents our simulation results and analysis of the aggregate TCP performance, including the characteristics of the overall throughput, goodput, and loss probability. Section 6 gives a summary of related work. We end with concluding remarks and future work in Section 7.

2 Network Abstraction

Network topology is a major determinant of TCP performance. To systematically study how TCP behaves, we need to have a simple network model, which is able to characterize real Internet connections. The complexity and heterogeneity of the current Internet make it very challenging to come up with such a general network model. Yet carefully examining the hierarchical structure of the Internet gives us valuable insights to build an abstract model capturing the essence of Internet connections.

2.1 Connections from a single domain

Today's Internet can be viewed as a collection of interconnected routing domains [18], which are groups of nodes that are under common administration and share routing information. It has three levels of routing. The highest level is the Internet backbone, which interconnects multiple autonomous systems (AS's). The next level is within a single AS, which is from the routers in an enterprise domain to the gateway. At the lowest level, we have routing within a single broadcast LAN, such as Ethernet or FDDI [7]. The upper portion of Figure 1 shows an abstract view of a cross-domain connection:

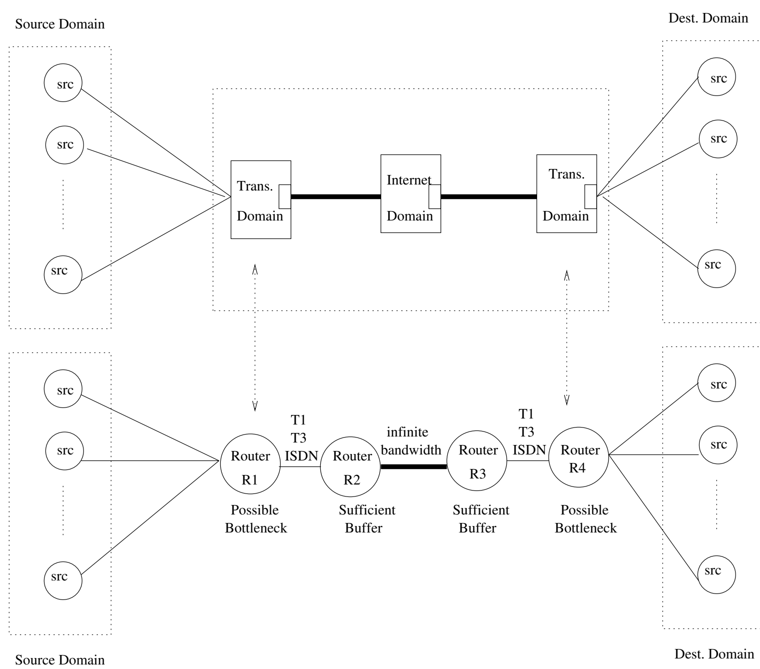


Figure 1: Abstract network topology for connections from a single domain

The connection from a source domain first reaches a transient backbone, which is linked to the Internet backbone via an access link. The other end of the Internet backbone is connected to a transient domain closest to the destination domain. The link between an enterprise domain and the transient backbone is typically dedicated to the enterprise, and each enterprise usually has enough bandwidth to carry its own traffic. The Internet backbones generally have large enough bandwidth, though sometimes it can also get congested. In contrast, the access link is shared among multiple domains, and its bandwidth is usually very limited. Therefore, we can reasonably assume **the access link is the bottleneck. (Assumption 1)** for wide-area Internet connections.

With the assumption that bottlenecks are located at access link, the topology can be further abstracted as shown in the lower portion of Figure 1, where the interconnection between transient backbones and internet backbones are abstracted as routers connected by access links and one high capacity link. The routers at both access links can become the bottleneck. Which one is bottleneck depends on the traffic condition at a given time.

2.2 Connections from multiple domains

As we know, the access links are usually shared by multiple domains. This is a much more complicated scenario, as shown in Figure 2:

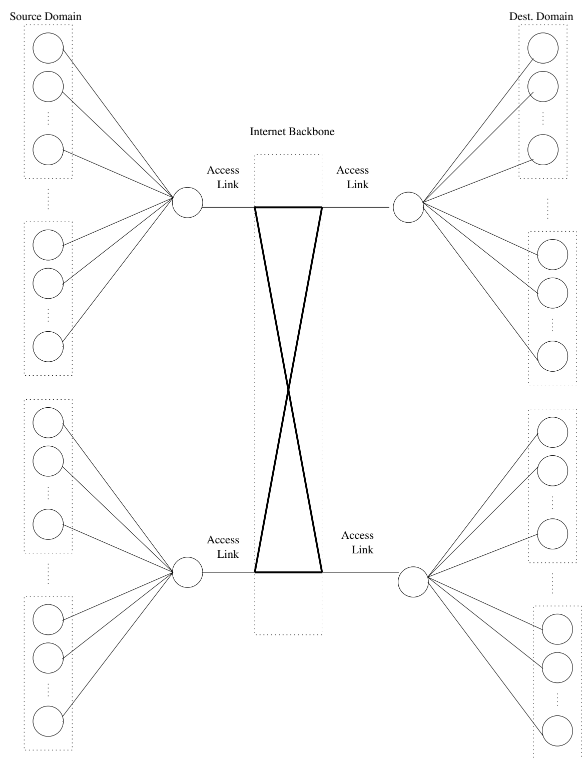


Figure 2: Abstract network topology for connections from multiple domains

At the first glance, it seems that all connections are intermixed with one another. However, according to the assumption that only access links can be bottlenecks, two connections not sharing any access links are thus independent. Using the above insight, we can partition all the connections into independent groups as follows:

1. Map the network connection into a graph, where each access link is represented as a node in the graph, and there is an edge between any two nodes if and only if there is at least one connection going through both access links denoted by the two nodes;
2. Find all connected components in the graph. The connections in two different connected components have no effect on one another.

After decomposition, we can now focus on studying a single connected component. The single connected component looks exactly the same as multiple cross-domain connections as shown in Figure 2, except that in the single connected component all the connections compete with each other, whereas before decomposition some connections can be independent of others.

In today's Internet, there are three common types of access links: ISDN (64kbps), T1 (1.5Mbps), and T3 (45Mbps), which we label as $type_1$, $type_2$, and $type_3$ link respectively. Then the whole system can be characterized by the following 12 parameters: $Propagation\ Delay$, $BufferSize_{R1}$, $BufferSize_{R4}$, C_{11}, C_{12}, C_{13}, C_{21}, C_{22}, C_{23}, C_{31}, C_{32}, C_{33}, where $C_{i,j}$ stands for the number of connections with incoming link of $type_i$ and outgoing links of $type_j$.

To summarize, in this section we have developed a simple generic model that can characterize wide-area Internet connections. This model allows us to study TCP behavior in a realistic yet manageable fashion.

3 A Single TCP Flow Under Cross Traffic

The behavior of a TCP connection is very complicated under cross traffic. There has been lots of previous work in this area. [13] is the best-known analytical model for the steady state performance of TCP-Reno. It captures the essence of TCP's congestion avoidance behavior by taking into account of fast retransmission, timeout, and the impact of window limitation. According to their model, the steady state TCP throughput can be approximated as follows:

$$B(p) = min(\frac{W_{max}}{RTT}, \frac{1}{RTT\sqrt{\frac{2bp}{3}} + T_0 min(1, 3\sqrt{\frac{3bp}{8}})p(1+32p^2)})$$

where p is the loss probability, $B(p)$ is the long-term steady-state TCP throughput, b is average number of packets acknowledged by an ACK, W_{max} is the maximum congestion window, RTT is the average roundtrip time experienced by the connection, and T_0 is the average duration of a timeout without back-off.

They also have a more accurate full model, whose formula is omitted here for brevity. [13] empirically validate the models using real traces. We are very interested in further investigating their models through simulations. We believe evaluation through simulations has its unique value:

- Simulations can accurately determine the variables in the expression, some of which can only be approximated in real traces, such as dropping probability p and RTT. (Dropping probability can only be approximated in real traces by loss indications. RTT estimation can be inaccurate due to coarse-grained timer.)
- Simulations using a generic model can cover a wider variety of scenarios.
- There are different implementations of TCP-Reno, which can lead to a surprisingly large range of behavior [12].
- In the real traces there are a number of unknown factors, such as processing time and system overhead, whereas all these factors are well under control in simulations.

We evaluate their models through a large number of simulations in different variations of the abstract network topology proposed in Section 2. More specifically, we use ns [11] and REAL [14] to simulate the network topology shown in Figure 3 and Figure 4, where the topologies are labeled according to Table 1.

Table 2 and Table 3 summarize the accuracy of their approximate model and full model respectively, where the percentage are based on over 10000 data points. (For example, one simulation run of 100 concurrent TCP connections gives us 100 data points.) In simulation settings other than those we present here, we have obtained similar results.

As shown in Table 2 and Table 3, most of their estimations are within a factor of 2, which is reasonably good. The approximate model, though simpler, is no worse than the full model. This may be due to the fact that the derivation of the full model requires estimating additional terms, which are not required in the approximate model.

We have identified a number of reasons that contribute to the divergence between their model and the simulation results:

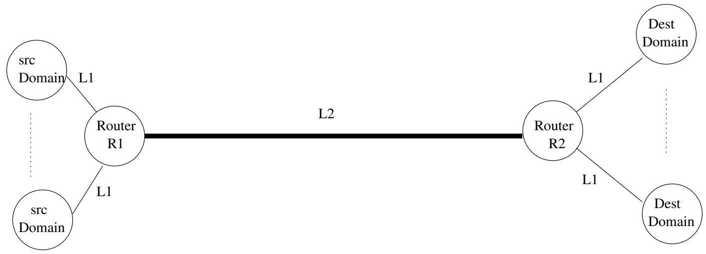


Figure 3: Simulation settings for topology 1 - 4

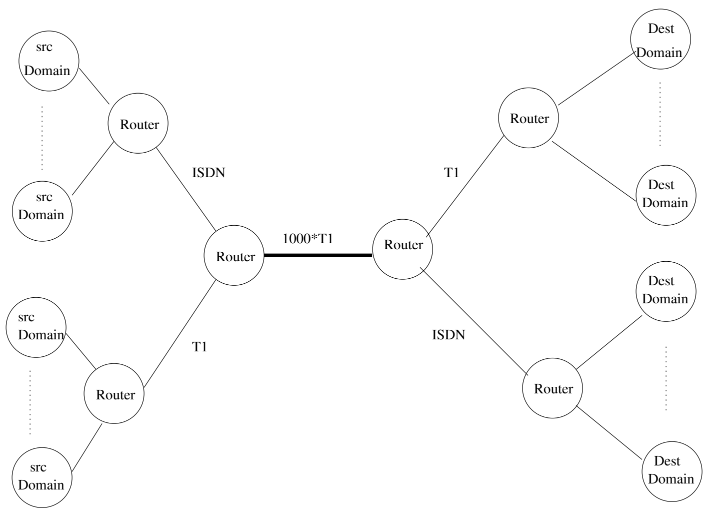


Figure 4: Simulation Topology 5, where all the links unlabeled have bandwidth of 28.8 kbps.

Topology	L1 propagation delay	L1 Bandwidth	L2 propagation delay	L2 bandwidth	Buffer size at all the routers (pkts)	Total Connections
1	0.001 ms	10 Mbps	50 ms	64 kbps	100 - 500	1 - 100
2	0.001 ms	10 Mbps	50 ms	64 kbps	100 - 500	110 - 600
3	0.001 ms	10 Mbps	50 ms	1.6 Mbps	100 - 500	1 - 100
4	0.001 ms	10 Mbps	50 ms	1.6 Mbps	100 - 500	110 - 600
5	as shown in Figure 4				100 - 500	1 - 600

Table 1: Simulation topologies

Topology	% within a factor of 1.5	% within a factor of 2	% within a factor of 3	% within a factor of 4
1	21.04	78.48	98.83	99.31
2	31.65	58.70	87.23	94.29
3	69.02	78.92	88.66	91.77
4	77.01	91.51	97.58	98.61
5	58.60	78.60	90.90	95.42

Table 2: The accuracy of the prediction based on the approximate model proposed in [13]

Topology	% within a factor of 1.5	% within a factor of 2	% within a factor of 3	% within a factor of 4
1	7.00	70.87	99.03	99.54
2	36.11	71.96	95.56	99.29
3	67.59	79.55	89.20	91.86
4	39.25	79.25	92.75	95.59
5	74.36	89.31	95.85	97.56

Table 3: The accuracy of the prediction based on the full model proposed in [13]

- The simplified assumption that packet losses are correlated in such a way that once a packet in a given round is lost, all remaining packets in that round are lost as well. (Each round begins with a window size of packets being sent back-to-back and ends with reception of the first ACK.)
- Ignoring the packets sent in the slow start phase.
- Several mathematical simplifications can introduce distortion, such as $E[f(x)] \approx f(E[x])$, where f is a nonlinear function, and the number of rounds in two consecutive triple duplicate acks (TD) is considered to be independent of the window size at the end of the TD period, which is not true.
- Other simplifications that can introduce distortion: ignoring the possibility of losing ACKs; ignoring timeout could occur before triple duplicate ACKs; ignoring packet reordering, that is loss is the only cause of duplicate ACK.

Of course, it is very hard if ever possible to come up with a general TCP model taking into account of all the details. On the other hand, if we know the distinguishing feature of a TCP connection, we can explore it to improve the accuracy of approximation. For example, if the connection is short, we need to consider the slow start phase [2].

In summary, the model is successful in that its analytical results are usually close to simulation results, and its development also gives us insights about on the performance of a single TCP connection under cross traffic.

4 Aggregate Behavior of TCP Connections - Model Simplification

Section 3 shows the analytical model proposed in [13] approximates a single TCP connection's steady state throughput reasonably well. This gives us some good insights into individual TCP behavior. However the model does not capture the aggregate behavior of TCP connections, such as the overall TCP throughput, goodput, loss rate, and fairness. These aggregate behaviors are sometimes more interesting. For example, from ISP's point of view, they are more interested in the aggregate throughput and loss rate of all the connections rather than a particular flow, because the aggregate performance is often more valuable for network provisioning.

Our approach is to use simulations to study TCP aggregate behavior. However, in order to make simulation an effective approach, it is necessary to have a simple simulation model with small parameter space so that we can identify the underlying relationship of how the performance varies with different parameters. The network model proposed in Section 2 is useful in that it captures the essence of Internet connections. However it is still too complicated (12 parameters) to obtain comprehensive understanding of TCP behavior. We have to further simplify the model.

The main reason for such a large parameter space is that the locations of the bottlenecks in the system are very hard to determine. Some connections have bottlenecks at the left side (before entering the backbone), whereas other connections' bottlenecks lie at the right side (after exiting the backbone). In order to take into account of the possibility of different locations of the bottleneck, we need all the 12 parameters. A natural simplification is to **assume the bottleneck will eventually stabilize, and different connections sharing the same access link congest at the same place. (Assumption 2)** The simulation models used in most literature are one-bottleneck models, which implicitly use this assumption.

Under this assumption, the abstract network model can be simplified as an one-bottleneck model, as shown in Figure 5. The whole system can now be characterized by the following four parameters: $propagation\ delay$, $BufferSize_S$, $Conn$, and $Type_{accesslink}$, where the access link is the bottleneck according to our assumption. By collecting and analyzing the simulation results with different sets of parameters, we can understand TCP behavior for a variety of network topologies, and identify the relationship between underlying topology and TCP performance.

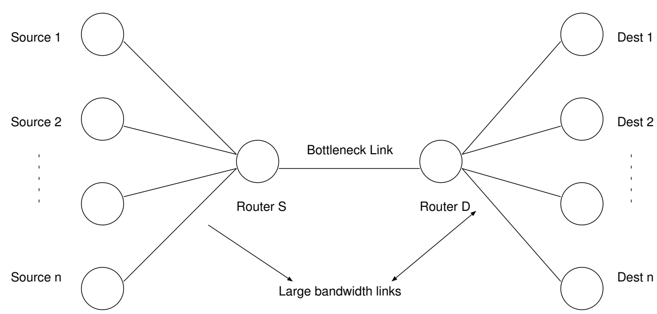


Figure 5: Simplified abstract network model

5 Aggregate Behavior of TCP Connections - Simulation and Analysis

In this section, we study the aggregate TCP performance through extensive simulations using *ns* network simulator [11]. We use the abstract model (shown in Figure 5) derived in Section 4 as our simulation topology, where the large bandwidth link is set to 10 Mbps Ethernet bandwidth with 0.001 ms delay. We vary each of the four parameters in the model: $propagation\ delay$, $BufferSize_S$, $Conn$, and $Type_{accesslink}$ to see how each of them affects TCP performance. More specifically, we consider both ISDN and T1 links, with the link delay of either 50 ms (typical for terrestrial WAN links) or 200 ms (typical for geostationary satellite links). We also vary the buffer size and the number of connections for each scenario.

The bottleneck link router uses FIFO scheduling and drop-tail buffer management since they are most commonly used in the current Internet. The TCP segment size is set to 500 bytes. As [10] points out, it is very common to have hundreds of concurrent connections competing for the bottleneck resource in today's Internet, so we are particularly interested in investigating the TCP behavior under such a large number of connections.

We use the following notations throughout our discussions:

- Let $W_{opt} = propagation\ delay * bottleneck\ bandwidth$, which is the number of packets the link can hold.
- Let $W_c = W_{opt} + B$, where B is the buffer size at the bottleneck link. W_c is the total number of packets that the link and the buffer can hold.

5.1 TCP behavior for flows with the same propagation delay

Our study of TCP flows with the same propagation delay shows TCP exhibits wide range of behaviors depending on the value of $\frac{W_c}{Conn}$, where $Conn$ denotes the number of connections. Based on the capacity of the pipe (measured as $\frac{W_c}{Conn}$), we classify our results into the following three cases: large pipe ($W_c > 3 * Conn$), small pipe ($W_c < Conn$), and medium pipe ($Conn < W_c < 3 * Conn$).

5.1.1 Case 1: $W_c > 3 * Conn$ (Large pipe case)

Previous studies have shown a small number of TCP connections with the same RTT can get synchronized [15]. We originally thought adding more connections introduces more randomness, and makes synchronization harder to take place. Surprisingly, our simulation results show the synchronization persists even in the case of large number of connections.

Figure 6 depicts the synchronization behavior. **In all the graphs we sort the connection ID's by the total number of packets each connection has received, since such sorting reveals synchronization behavior more clearly.** As shown in the figure, the buffer occupancy periodically fluctuates from half to full, which implies all connections halve their congestion windows in synchrony. The global synchronization behavior can be further demonstrated by the periodic white stripes in the scatter plot of ACK arrival time, which imply all the connections start and end loss recovery in a synchronized manner.

The explanation for the synchronization behavior is similar to the case for small number of connections. Suppose at the end of the current epoch the total number of outstanding packets from all the connections is equal to W_c. During the next epoch all connections will increment their window. All the packets that are sent due to window increase will get dropped. Thus all the connections will incur loss during the same RTT. This makes all connections adjust window in synchrony. When $W_c > 3 * conn$, most connections have more than 3 outstanding packets before the loss. So they can all recover the loss by fast retransmissions, and reduce the window by half, leading to global synchronization. In contrast, when $W_c < 3 * Conn$, though all the connections still experience loss during the same RTT, they react to the loss differently. Some connections whose $cwnd$ is larger than 3 before the loss can recover the loss through fast recovery, whereas the others will have to use timeout to recover the loss. Since the set of connections recovering loss using fast recovery and the set using timeout will change over time, global synchronization cannot be achieved.

Due to global synchronization, all the connections share the resource very fairly: in the steady state they experience the same number of losses and send the same number of packets. We can aggregate all the connections as one big connection, and accurately predict the aggregate loss probability. All the original connections will have the same loss probability. More specifically, if W_c is a multiple of $Conn$, let $W = \frac{W_c}{Conn}$ and b be the average number of packets acknowledged by an ACK. During congestion avoidance phase, in each epoch other than the first and last ones, each connection's window size starts

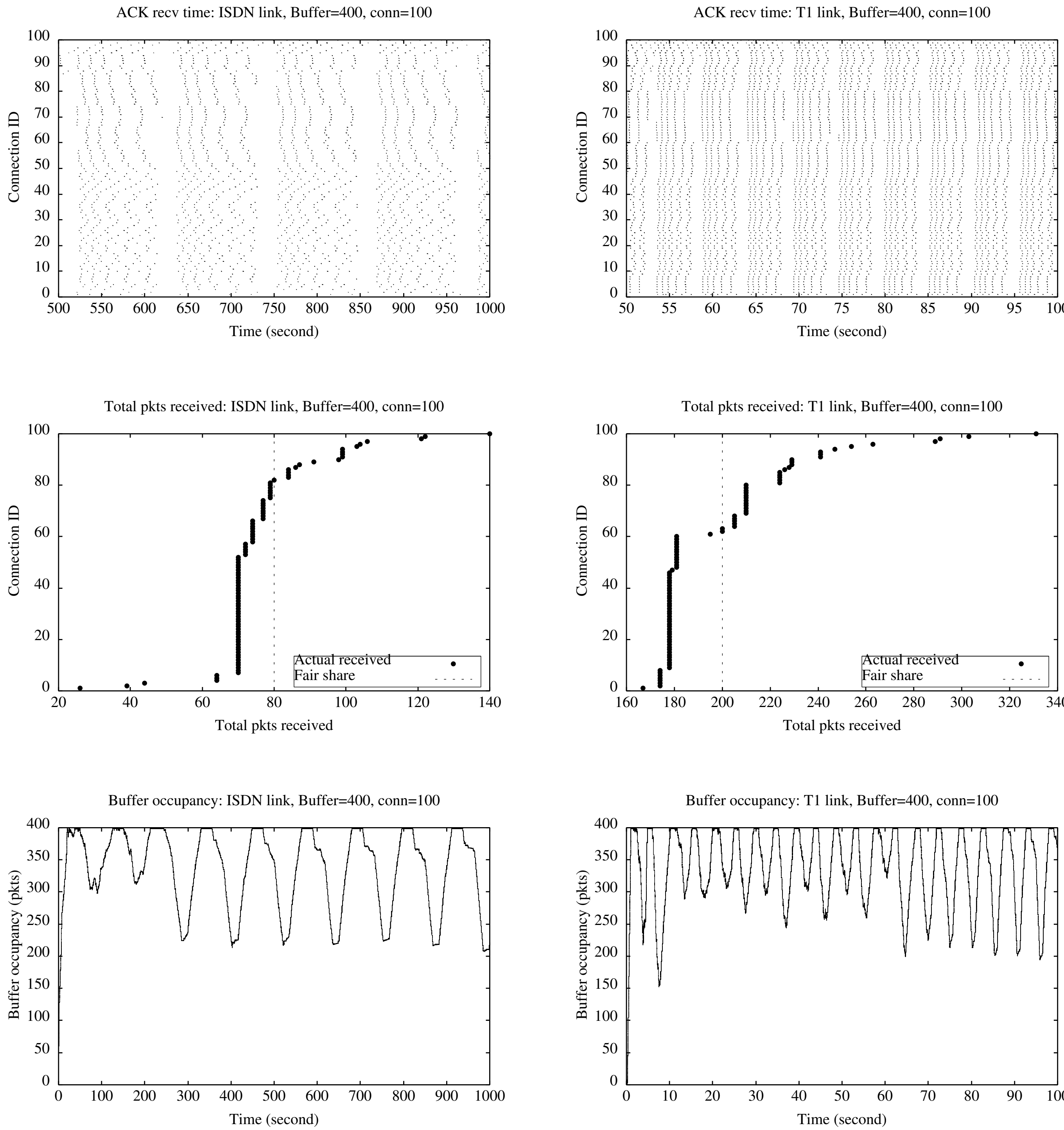


Figure 6: Large pipe case leads to global synchronization: 100 connections sharing the bottleneck link of either ISDN or T1. In both cases, the oneway propagation delay = 50 ms and bottleneck buffer size = 400 packets. As shown in the figure, the buffer occupancy fluctuates periodically from half to full, which implies all connections halve their congestion windows in synchrony. The global synchronization behavior can be further demonstrated by the periodic white stripes in the scatter plot of ACK arrival time, which imply all the connections start and end loss recovery in a synchronized manner.

from $\lfloor \frac{W+1}{2} \rfloor$ and increases linearly in time, with a slope of $\frac{1}{b}$ packet per round trip time. When the window size reaches $W+1$, a loss occurs. Before the loss is detected by duplicated ACKs, another W packets are injected into the network. Then the window drops back to $\lfloor \frac{W+1}{2} \rfloor$ and a new epoch begins. Therefore, the total number of packets sent during an epoch can be computed as $S(T) = b * (\sum_{x=\lfloor \frac{W+1}{2} \rfloor}^{W} x) + 2 * W + 1$. Every connection incurs one loss during each epoch, so

$$Loss\ Probability = \frac{1}{S(T)} = \frac{1}{b * (\sum_{x=\lfloor \frac{W+1}{2} \rfloor}^{W} x) + 2 * W + 1}$$

When $b = 1$,

$$Loss\ Probability = \begin{cases} \frac{8}{3*W^2+20*W+9} & \text{if W is odd} \\ \frac{8}{3*W^2+22*W+8} & \text{otherwise} \end{cases}$$

So we can approximate $Loss\ Probability$ as

$$Loss\ Probability \approx \frac{8}{3 * W^2 + 21 * W + 8}.$$

Figure 7 shows our prediction matches very well to the actual loss probability.

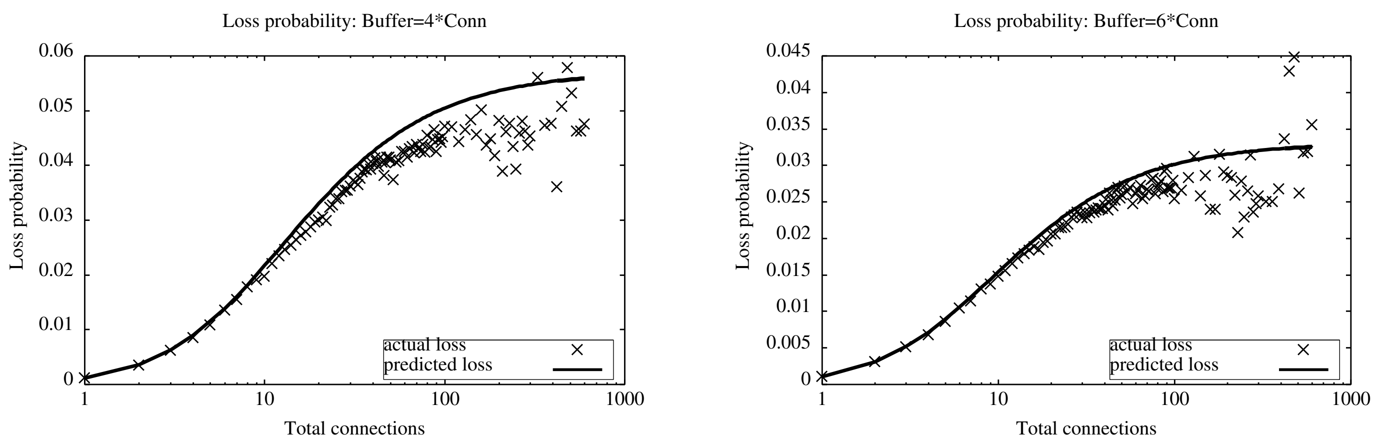


Figure 7: Loss prediction for large pipe case: Varying number of connections sharing T1 link with one-way propagation delay of 50 ms, and the bottleneck buffer is either 4 times or 6 times the number of connections.

Furthermore, as expected, global synchronization leads to periodical fluctuation in buffer occupancy as shown in Figure 6. When the total buffer size is less than $\frac{W_c}{2}$, halving $cwnd$ in synchrony leads to under-utilization of bottleneck link.

5.1.2 Case 2: $W_c < Conn$ (Small pipe case)

When $W_c < Conn$, we have found TCP connections share the path very unfairly: only a subset of connections are active (i.e. their goodput is considerably larger than 0), while the other connections are shut-off due to constant timeout as shown in Figure 8. The number of active connections is close to W_c, and the exact value depends on both W_c and the number of competing connections. When $Conn$ exceeds the number of active connections the network resource can support, adding more connections to the already overly congested network only adds more shut-off connections. Almost all the packets sent from the shut-off connections get dropped. The remaining active connections are left mostly intact. This explains the curves in Figure 9: when $Conn$ is larger than the number of active connection the network resources can support, the total number of packets sent and lost grows linearly with the number of connections. The linear increase in the number of packets sent and lost mainly comes from the increase in the number of inactive (mostly shut-off) connections, each of which sends a constant number of packets before it gets completely shut off.

5.1.3 Case 3: $Conn < W_c < 3 * Conn$ (Medium pipe case)

As shown in Figure 10, TCP behavior in this case falls in between the above two cases. More specifically, as explained earlier (in Section 5.1.1), since $1 < \frac{W_c}{Conn} < 3$, the connections respond to loss differently: some connections whose $cwnd$ is larger than 3 before the loss can recover the loss through fast recovery, whereas the others will have to use timeout to recover the loss. Since the set of connections recovering loss using fast recovery and the set using timeout will change over time, no global synchronization occurs, and the network resources are not shared as fairly as $W_c > 3 * Conn$. On the other hand, there is still local synchronization, as shown Figure 10, where some groups of connections are synchronized within the groups. Furthermore, since $\frac{W_c}{Conn} > 1$, all the connections can get reasonable amount of throughput. In contrast to the small pipe case, there are almost no connections getting shut-off.

5.1.4 Aggregate Throughput

We define *normalized aggregate TCP throughput* as the number of bits sent by the bottleneck link in unit time normalized by the link capacity. Our results are as follows:

- As shown in Figure 11(a), when the number of connections is small and the buffer size is less than W_{opt} ($W_{opt} = 160$ packets in this case), the normalized TCP throughput is less than 1. The degree of under-utilization depends on both the number of connections and the ratio of the buffer size to W_{opt}. The smaller the number of connections and the lower the ratio, the lower the network utilization is.
- As shown in Figure 11(b), when the buffer size is larger than W_{opt} ($W_{opt} = 40$ packets in this case), the normalized TCP throughput is close to 1, regardless of the number of connections.
- When the number of connections is large, even if the buffer size is small (smaller than W_{opt}), the normalized TCP throughput is close to 1. This is evident from Figure 11(a), where the throughput is close to 1 for large number of connections under all the buffer sizes considered.

5.1.5 Aggregate Goodput

We define *normalized aggregate goodput* as the number of *good* bits received by all the receivers (excluding unnecessary retransmissions) in unit time normalized by the link capacity. As shown in Figure 12,

- There is a linear decrease in goodput as the number of connections increases.
- The slope of the decrease depends on the bottleneck link bandwidth: the decrease is more rapid when the bottleneck link is ISDN, and is slower when T1 is used as the bottleneck link.

These results can be explained as follows. The difference between the throughput and goodput is the number of unnecessary retransmissions. As the number of connections increases, the loss probability increases, which in turn increases the number of unnecessary retransmissions. Therefore the more connections, the lower the goodput is. On the other hand, since

$$loss\ in\ the\ normalized\ goodput = \frac{total\ unnecessary\ retransmissions}{link\ capacity}$$

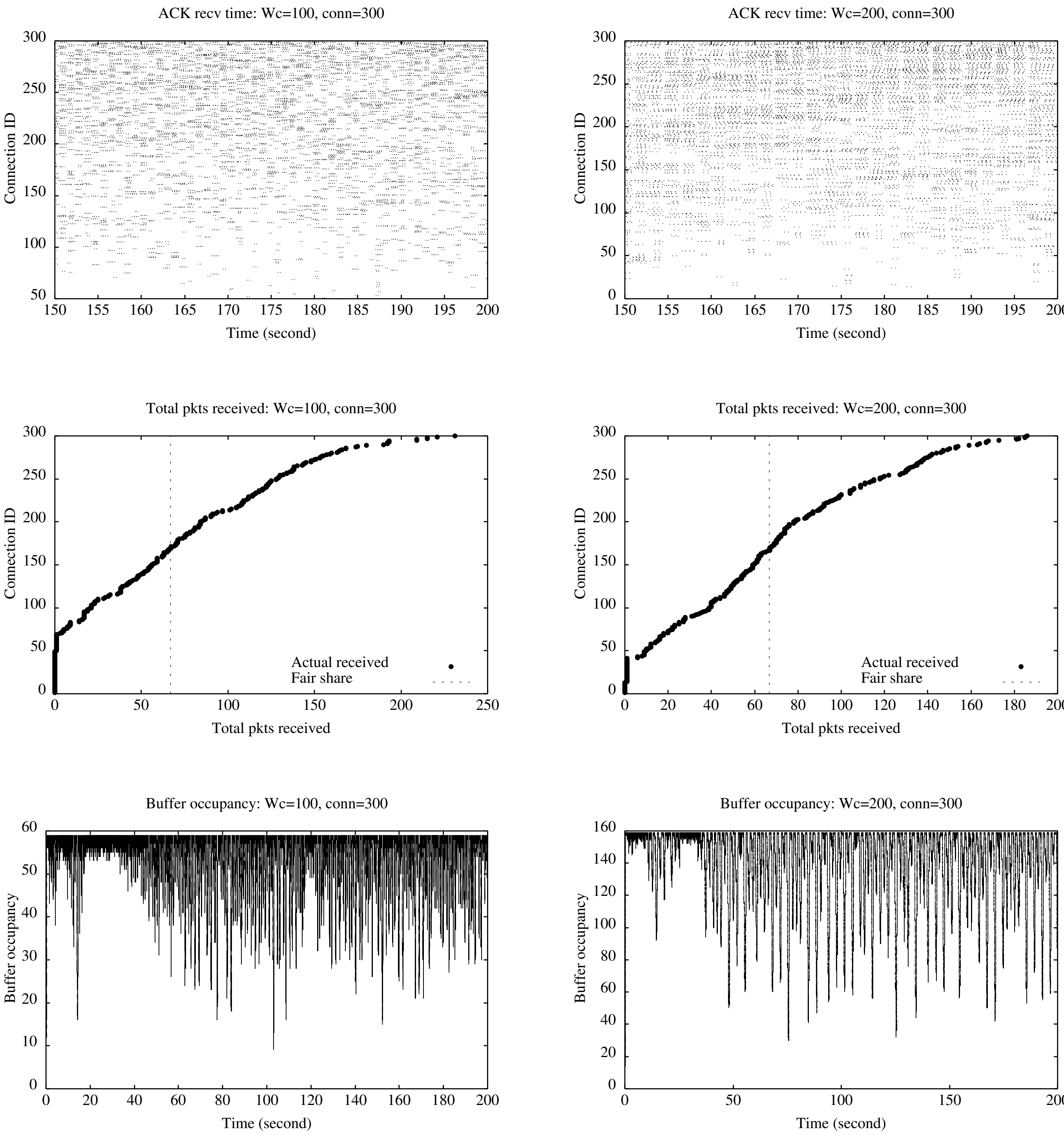


Figure 8: Small pipe case: 300 concurrent connections compete for T1 link with one-way propagation delay of 50 ms and bottleneck buffer size of 60 (or 160) packets ($W_c = W_{opt} + Buffer =$ 100 or 200 packets). Note that the buffer occupancy fluctuates widely. Moreover part of connections receive little goodput as shown in the two graphs in the middle.

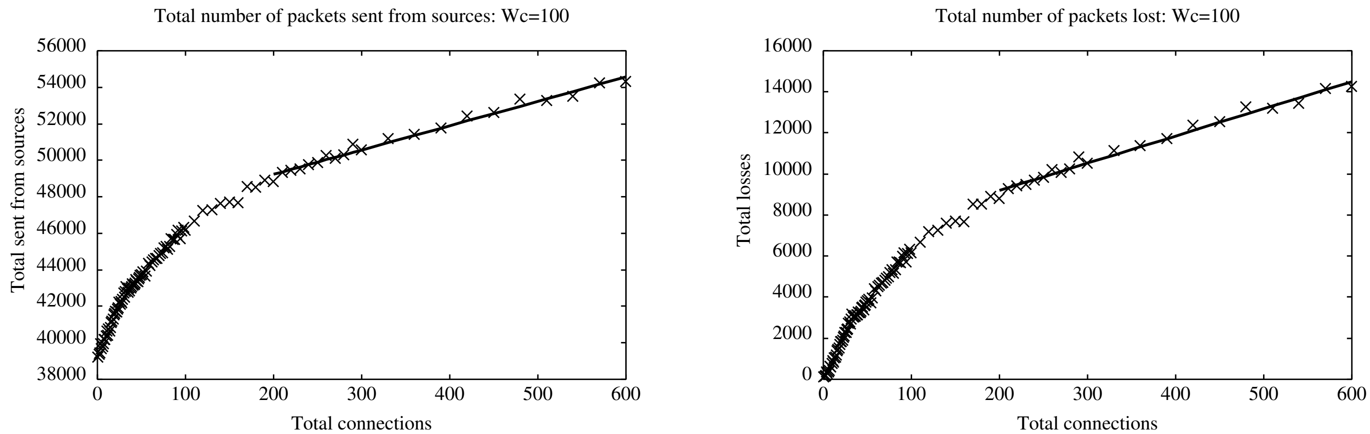


Figure 9: Varying number of connections compete for T1 link with oneway propagation delay of 50 ms: the total number of packets sent and dropped increases linearly with the number of connection when the connection number is large.

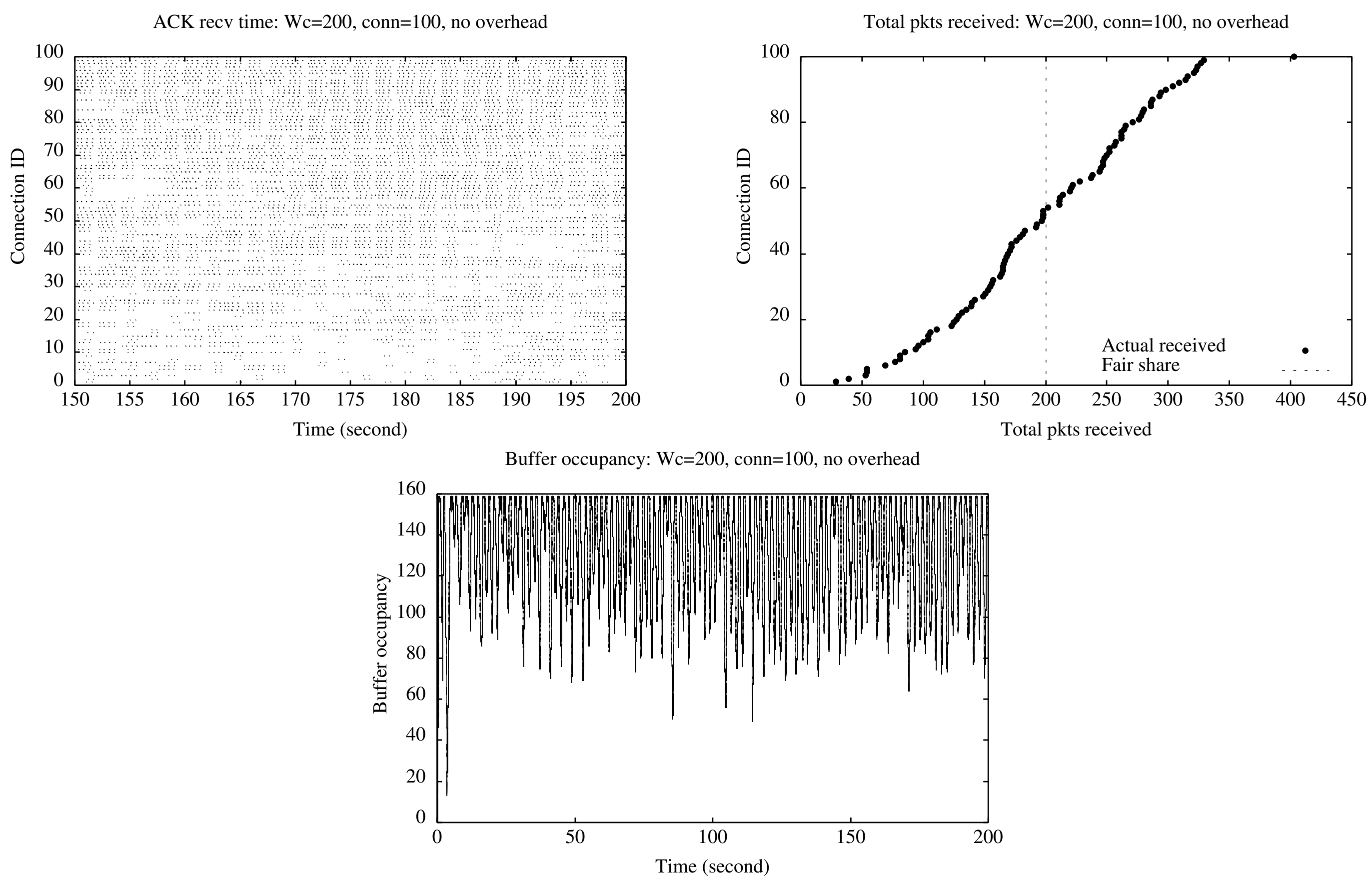
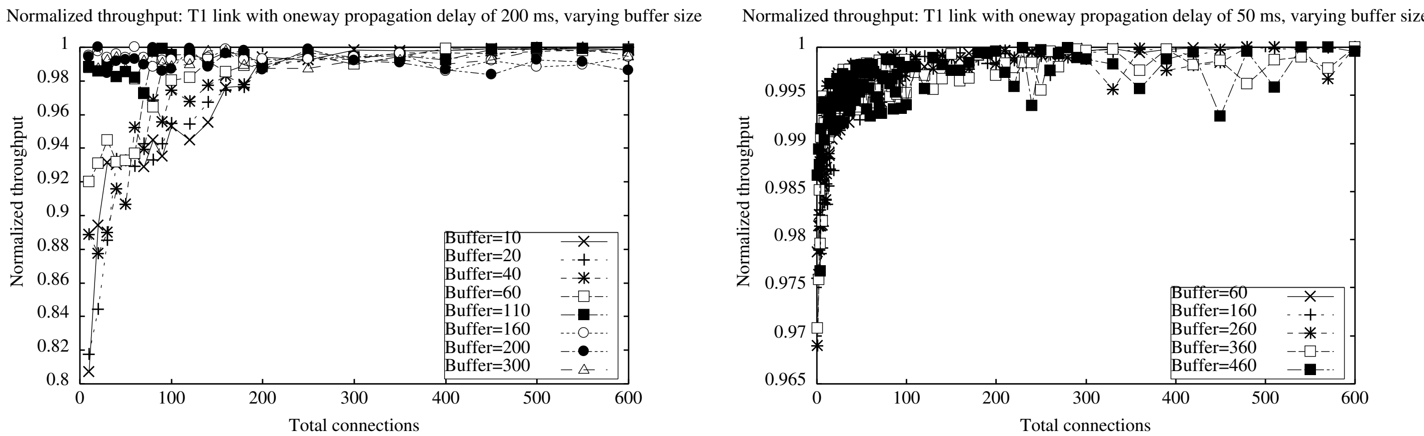


Figure 10: Medium pipe case: 100 concurrent connections compete for T1 link with one-way propagation delay of 50 ms and bottleneck buffer size of 160 packets ($W_c = W_{opt} + Buffer = 200$ packets). Buffer occupancy fluctuates widely, and there is local synchronization within some groups.



(a) (b)

Figure 11: Throughput: varying number of connections compete for the bottleneck link T1 with oneway propagation delay of either 200 ms or 50 ms

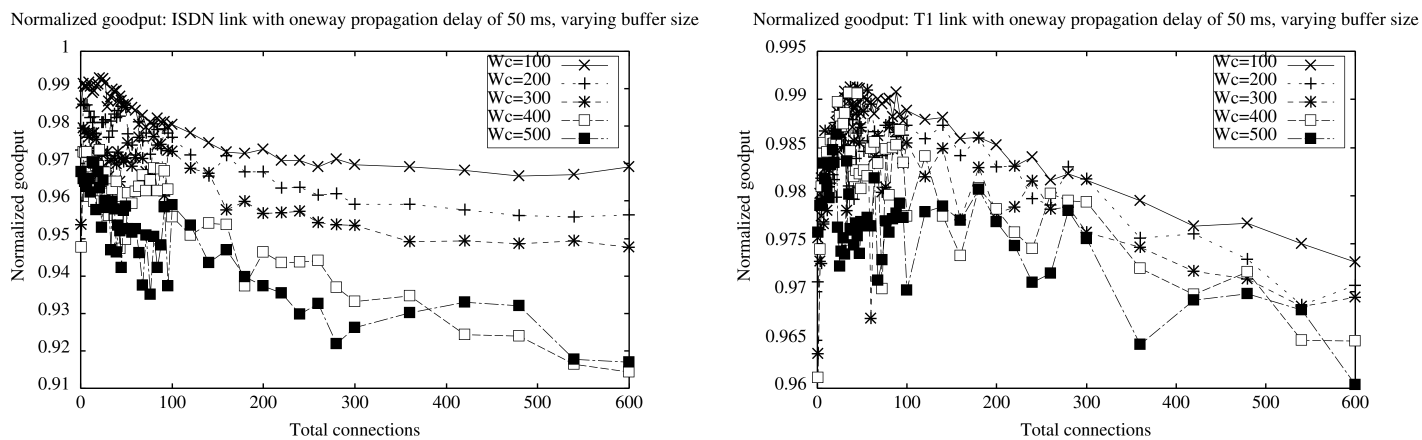


Figure 12: Goodput: varying number of connections compete for the bottleneck link of either ISDN or T1. In both cases, the oneway propagation delay=50 ms.

the decrease in the goodput is more substantial with slower bottleneck (e.g. ISDN), and less significant with faster bottleneck (e.g. T1), which is evident from Figure 12.

To summarize, when the bottleneck link is fast or the loss probability is low (less than 20%), the number of unnecessary retransmissions is negligible so that the normalized goodput is close to the normalized throughput. Otherwise (i.e. when the bottleneck link is slow and the loss probability is high) the loss in the goodput due to unnecessary retransmissions becomes significant. The decrease in the goodput depends on both the bottleneck link bandwidth and the loss probability.

5.1.6 Loss Probability

Our simulation results indicate when the W_c is fixed and the number of connections is small, the loss probability grows quadratically with the increasing number of connections as shown in Figre 13. The quadratic growth in the loss probability can be explained as follows. When $\frac{W_c}{Conn} > 3$, TCP connections can recover loss without timeouts. Every connection loses one packet during each loss episode. So altogether there are $Conn$ losses every episode. Meanwhile the frequency of loss episode is proportional to $Conn$. Therefore for small number of connections, the loss probability is proportional to $Conn^2$. Such quadratic growth in loss probability is also reported in [10] for routers with RED dropping policy.

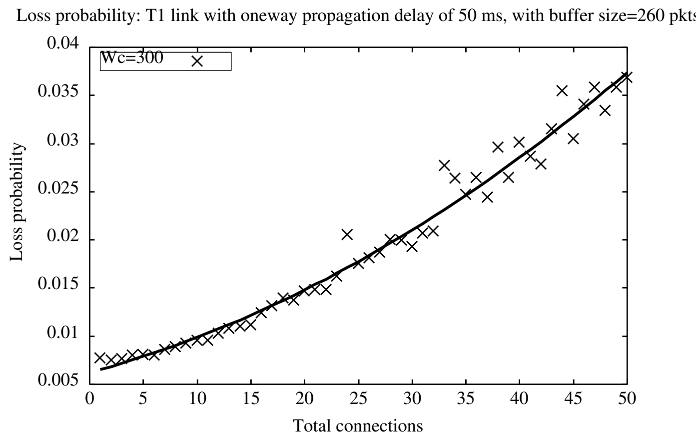
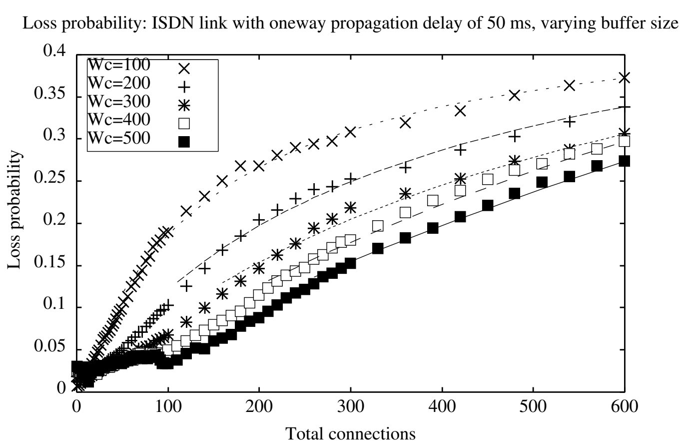


Figure 13: Loss probability for small number of connections: varying number of connections compete for the bottleneck link of T1 with oneway propagation delay of 50 ms. The loss probability grows quadratically when the number of connections is small.

As the number of connections gets large (larger than $\frac{W_c}{3}$), the growth of loss probability with respect to the number of connections matches impressively well with the following family of hyperbolic curves represented by

$$y = \frac{b * x}{x + a}$$

as shown in Figure 14. Table 4 gives the parameters of the hyperbolic curves used in Figure 14. As part of our future work, we will investigate how to predict the parameters of the hyperbolic curves given the network topology.



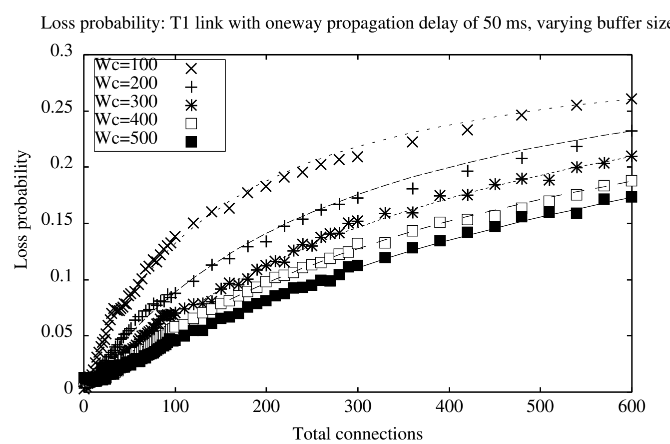


Figure 14: Loss probability for large number of connections: varying number of connections compete for the bottleneck link of either ISDN or T1. In both cases, the oneway propagation delay = 50 ms. The loss probability curves match very well with hyperbolic curves when the number of connections is large, where the parameters of the hyperbolic curves are given in Table 4

ISDN			T1		
Wc	a	b	Wc	a	b
100	149.2537	0.4646	100	144.9275	0.3237
200	333.3333	0.5262	200	285.7143	0.3429
300	588.2352	0.6059	300	454.5454	0.3682
400	1250.000	0.9170	400	526.3158	0.3517
500	2000.000	1.1852	500	769.2308	0.3948

Table 4: Parameters for the hyperbolic curves used for fitting loss probability as shown in Figure 14

5.2 TCP behavior with random overhead

5.2.1 $W_c > 3 * Conn$ (Large pipe case)

Our discussions in the previous section (Section 5.1) focus on the macro behavior of concurrent TCP connections with the same propagation delay. In order to explore properties of networks with Drop Tail gateways unmasked by the specific details of traffic phase effects or other deterministic behavior, we add random packet-processing time in the source nodes. This is likely to be more realistic. The technique of adding random processing time was first introduced in [4]. However we have different goals. In [4], Floyd and Jacobson are interested in how much randomness is necessary to break the systematic discrimination against a particular connection. In contrast, we are interested in how much randomness is sufficient to break the global synchronization. Consequently, the conclusions are different. [4] concludes that adding a random packet-processing time ranging from zero to the bottleneck service time is sufficient; while we find that a random processing time that ranges from zero to $10\% * RTT$ (usually much larger than a random packet service time) is required to break down the global synchronization. This is shown in Figure 15, where the global synchronization is muted after adding the random processing time up to $10\% * RTT$.

The performance results in the non-synchronization case also differ from the global synchronization case. As shown in Figure 16, when the number of connections is less than 100, the loss probability in the two cases are almost the same; as the number of connections increases further, the gap between the two opens up: the non-synchronization case has higher loss probability than the synchronization case. Nevertheless, using the prediction based on global synchronization gives a reasonable approximation (at least a lower bound) of loss probability for non-synchronized case. However since in the non-synchronization case, the connections do not share the bandwidth fairly. As shown in Figure 15, there is a large variation in the throughput of different connections, so we can no longer predict the bandwidth share for each connection in this case.

A few comments follow:

- There is a general concern that synchronization is not good since it may lead to under-utilization of the bottleneck bandwidth. However with the use of TCP-Reno and sufficient bottleneck buffer provisioning, this is unlikely to be a problem in practice.
- There have been several attempts to break synchronization in order to avoid under-utilization. However our simulation results indicate breaking down the synchronization using random processing time increases the unfairness and loss probability.

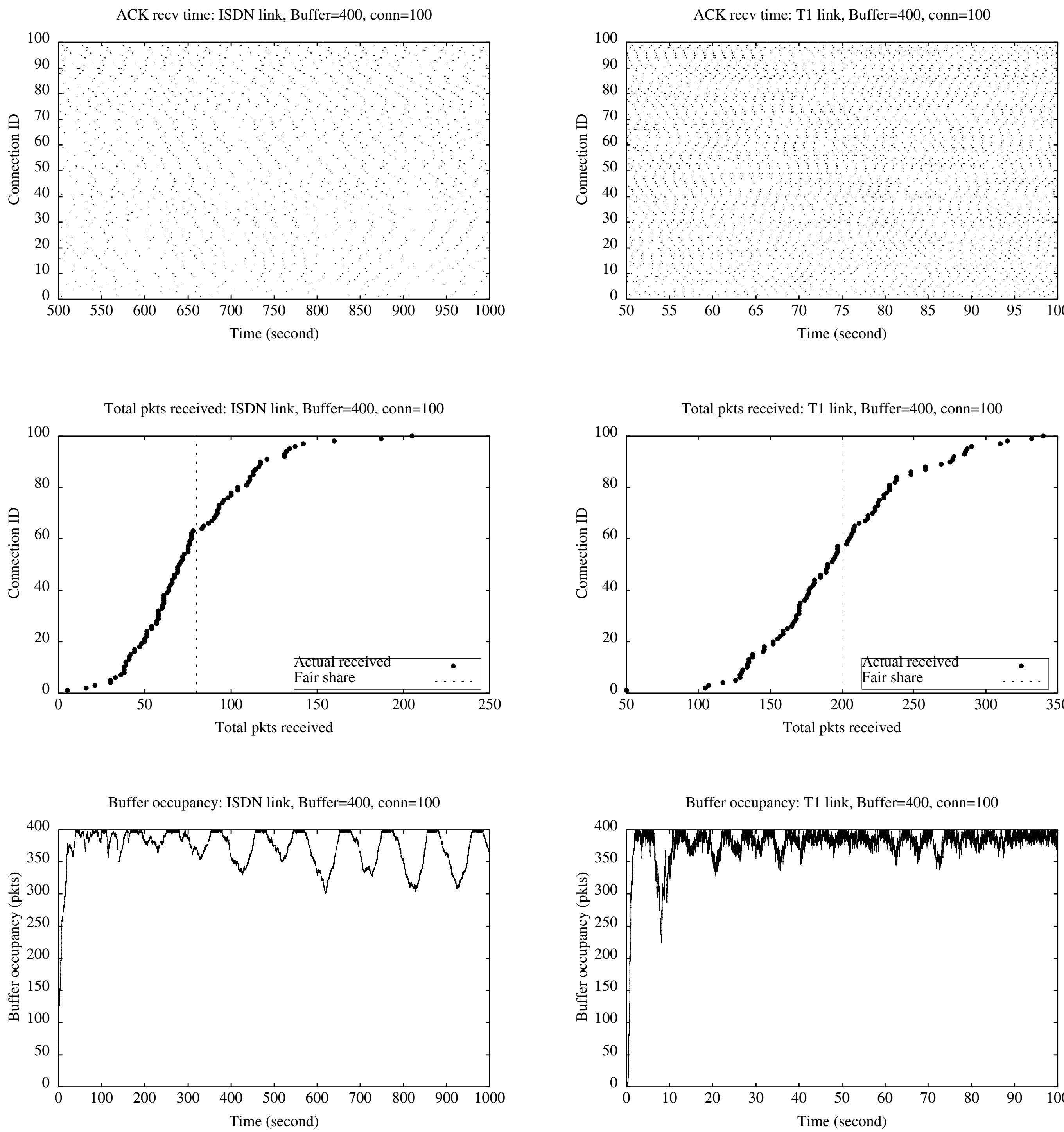


Figure 15: Adding random process time in large pipe case breaks down the global synchronization: 100 connections sharing the bottleneck link of either ISDN or T1 with one-way propagation delay of 50 ms and bottleneck buffer size of 400 packets. Compared to the case of without random processing time, the buffer occupancy is quite stable. Moreover global synchronization disappears as shown in the scatter plot for ACK arrival.

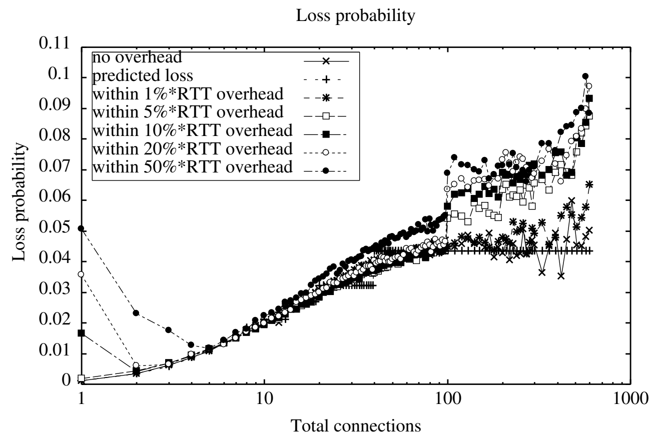


Figure 16: Compare the loss probability by adding different amount of random processing time: varying number of connections compete for T1 link with oneway propagation delay of 50 ms

5.2.2 $W_c < Conn$ (Small pipe case)

Adding random processing time also affects the case when $W_c < Conn$. Without random processing time, we find there is consistent discrimination against some connections, which end up totally shut off due to constant time-out. After adding random processing time, there is still discrimination against some connections, but much less severe than before. As shown in Figure 17, the number of shut-off connections is considerably smaller than before. Furthermore, the buffer occupancy is mostly full and stable, whereas without random processing time, the buffer occupancy is quite low, and fluctuates a lot.

5.2.3 $Conn < W_c < 3 * Conn$ (Medium pipe case)

As shown in Figure 18, adding random processing time does not have much impact on TCP behavior in the case of medium size pipe: as before, most connections get reasonable goodput, though not synchronized. On the other hand, the buffer occupancy now becomes mostly full and stable in contrast to without random processing time, where the buffer occupancy is low, and fluctuates a lot. In addition, even local synchronization disappears after adding random processing time.

5.2.4 Aggregate Throughput & Goodput

Adding random processing time has little effect in the overall throughput and goodput as shown in Figure 19 and Figure 20.

5.2.5 Loss Probability

For a small number of connections, adding random processing time makes the loss probability grow mostly linearly as the number of connections increases. This is evident from Figure 21, which compares the loss probability curves before and after adding random processing time.

For the large number of connections, adding random processing time does not change the general shape of the loss probability curve: as before, the growth of loss probability with respect to the number of connections matches impressively well with hyperbolic curves, as shown in Figure 22. On the other hand, for the same configuration (the same number of connections and buffer size), the loss probability becomes larger after adding the random processing time. So the parameters

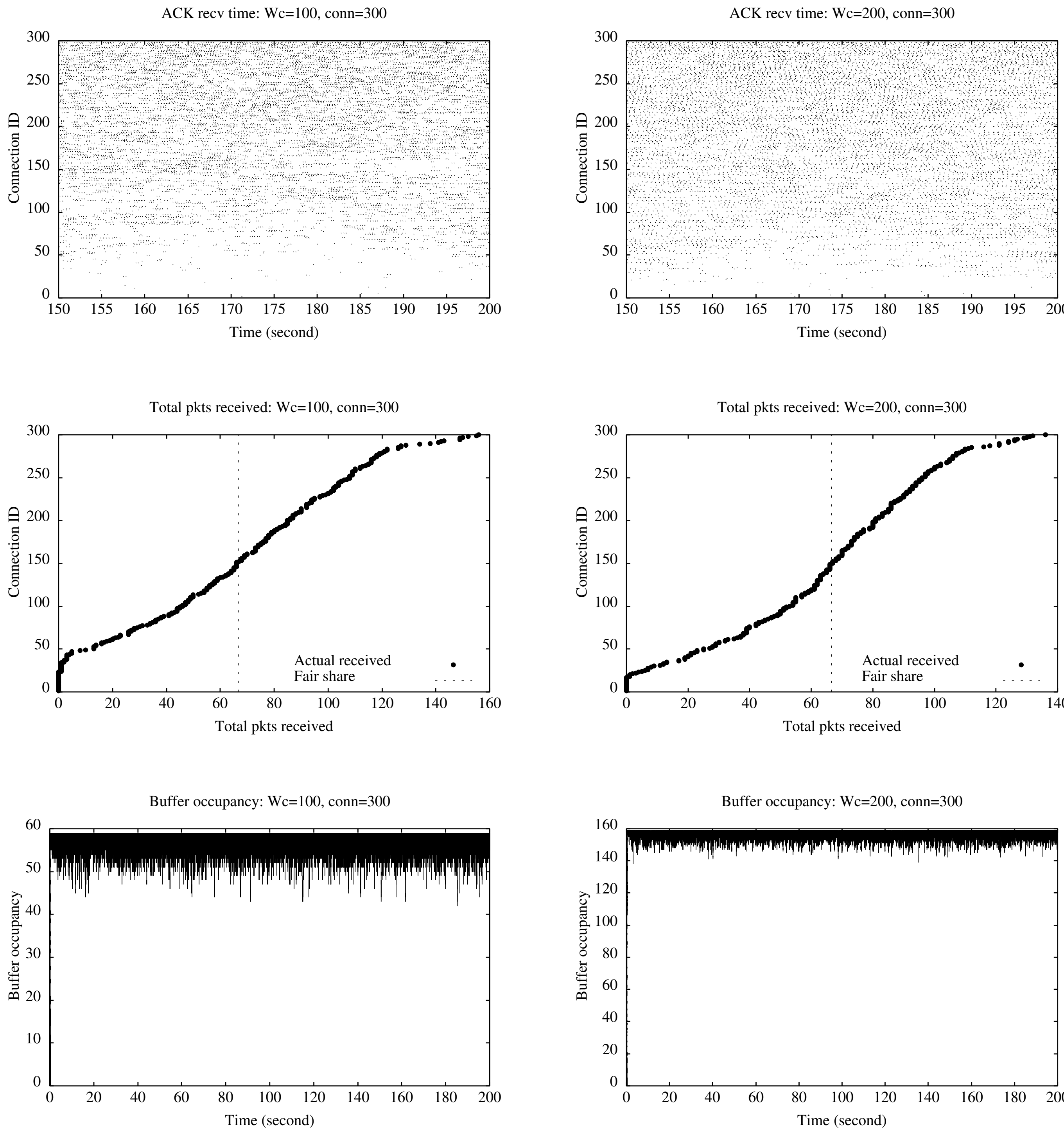


Figure 17: Adding random processing time in small pipe case: 300 concurrent connections compete for T1 link with one-way propagation delay of 50 ms and buffer size of 60 (or 160) packets ($W_c = W_{opt} + Buffer$=100 or 200 packets). The buffer occupancy is quite stable, and the consistent discrimination is not as severe as without adding random processing time.

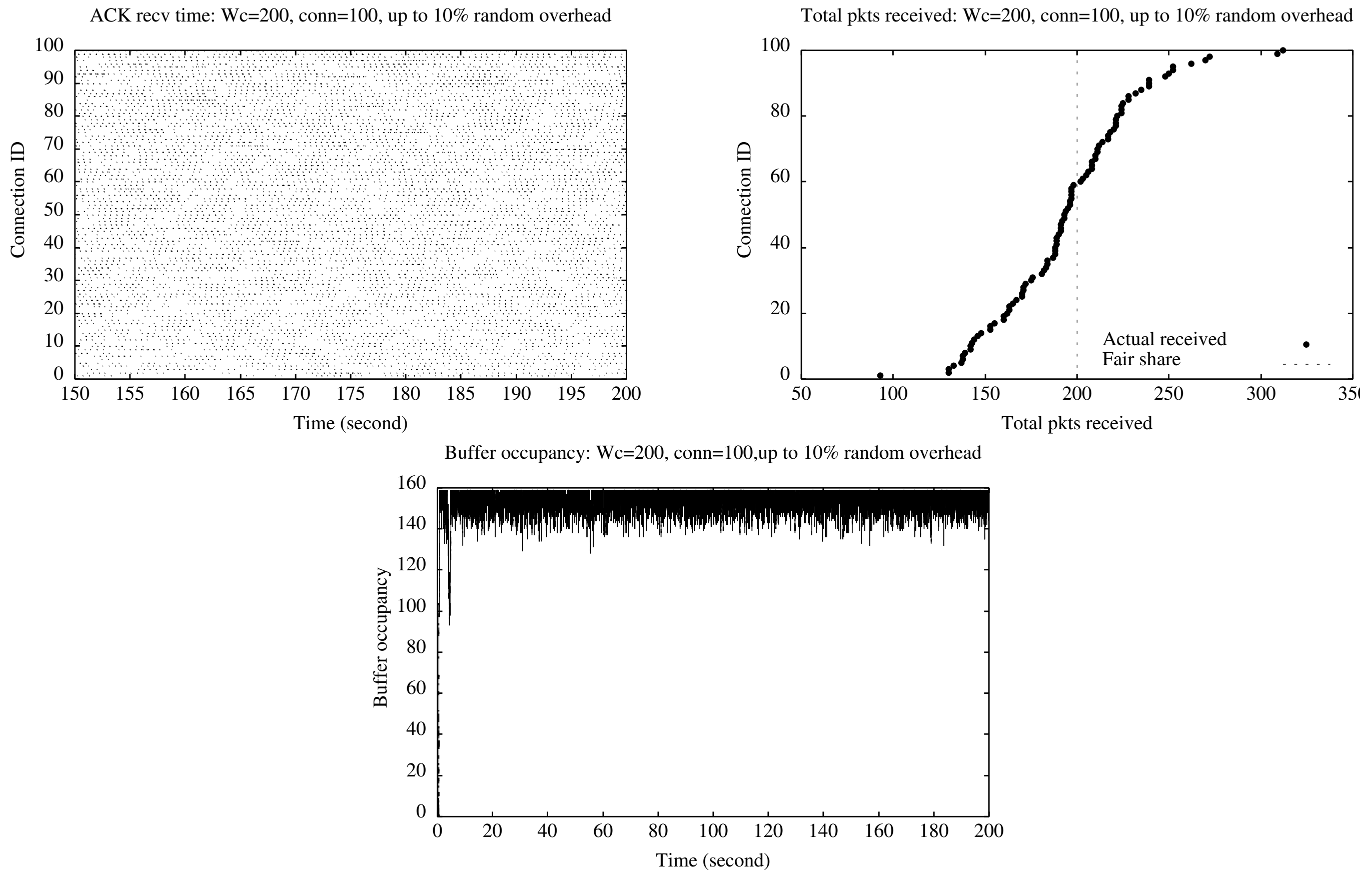


Figure 18: Adding random processing time in medium pipe case: 100 concurrent connections compete for T1 link with one-way propagation delay of 50 ms and buffer size of 160 packets ($W_c = W_{opt} + Buffer = 200$ packets). The buffer occupancy is quite stable. In contrast to without random processing time, there is no local synchronization.

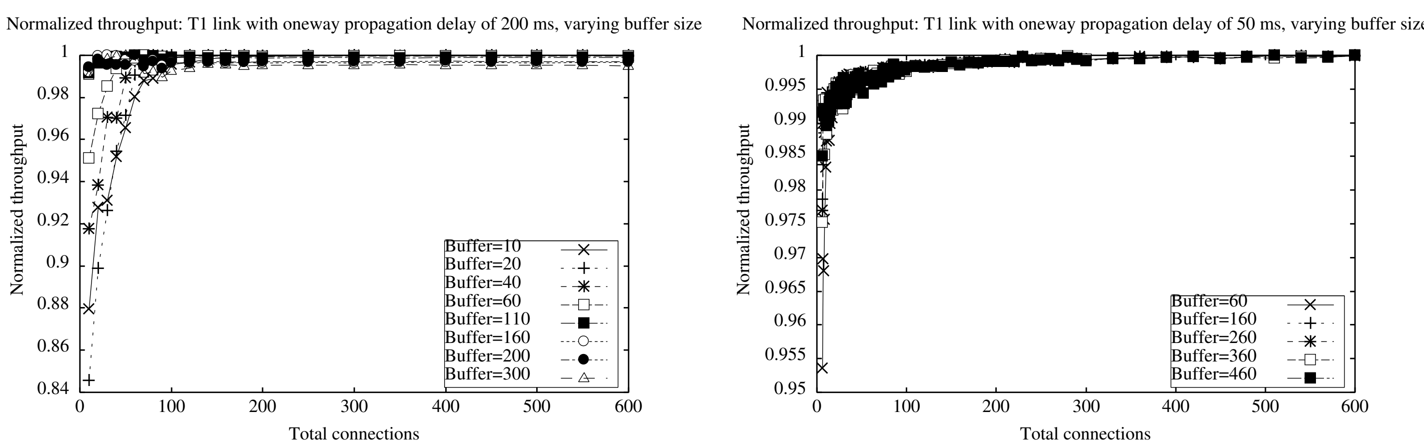


Figure 19: Throughput after adding random processing time of up to $10\% * RTT$ at TCP source: varying number of connections compete for the bottleneck link of T1 link with propagation delay of either 200 ms or 50 ms

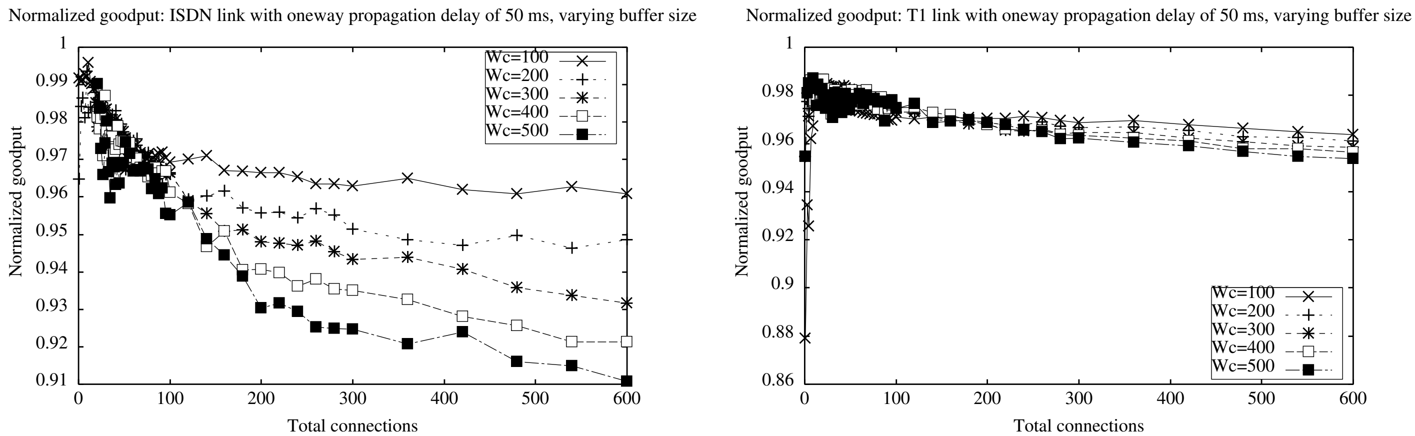


Figure 20: Goodput after adding random processing time of up to $10\% * RTT$ at TCP source: varying number of connections compete for the bottleneck link of either ISDN or T1. In both cases, the oneway propagation delay=50 ms.

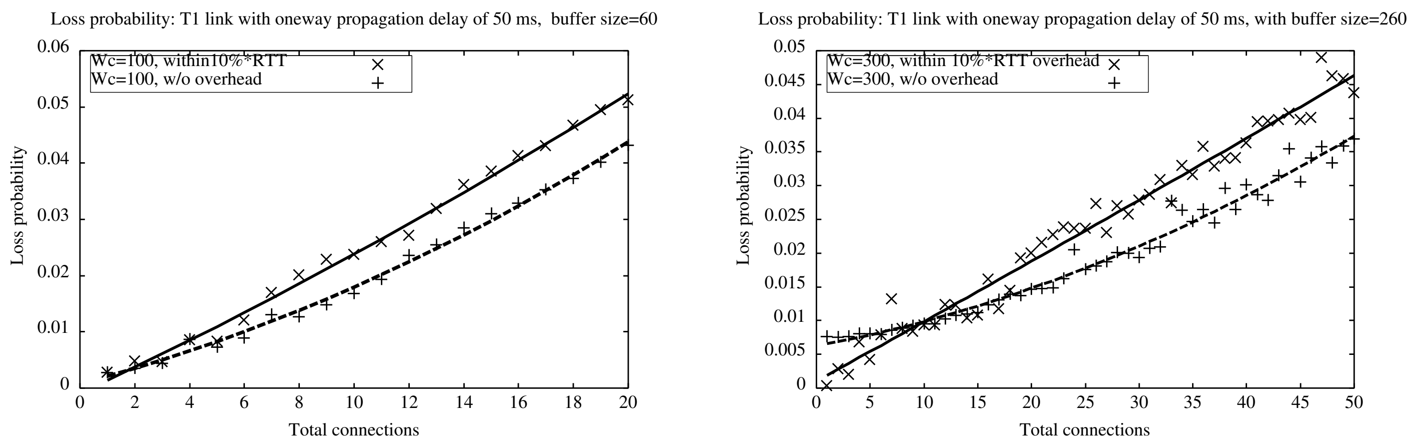


Figure 21: Loss probability for small number of connections after adding random processing time of up to $10\% * RTT$ at TCP source: varying number of connections compete for the bottleneck link of T1 link with oneway propagation delay of 50 ms. The loss probability grows linearly when the number of connections is small.

of hyperbolic curves are different as shown in Table 5.

5.3 TCP behavior with different RTT

It is well-known that TCP has bias against long roundtrip time connections. We are interested in quantifying this discrimination through simulations. Our simulation topology is similar to Figure 5 (in Section 4), except that we change the propagation delay of the links. More specifically, we divide all the connections into two equal-size groups, where one group of connections has fixed propagation delay on the large bandwidth links, and the other group of connections has varying propagation delay on the large bandwidth links. As suggested in [4], we add a random packet-processing time in the source nodes that ranges from zero to the bottleneck service time to remove systematic discrimination. Our goal is to study how the throughput ratio of two groups changes with respect to their RTT's.

Our simulation results are summarized in Figure 23, which plots the throughput ratio vs their RTT ratio both in **log2**

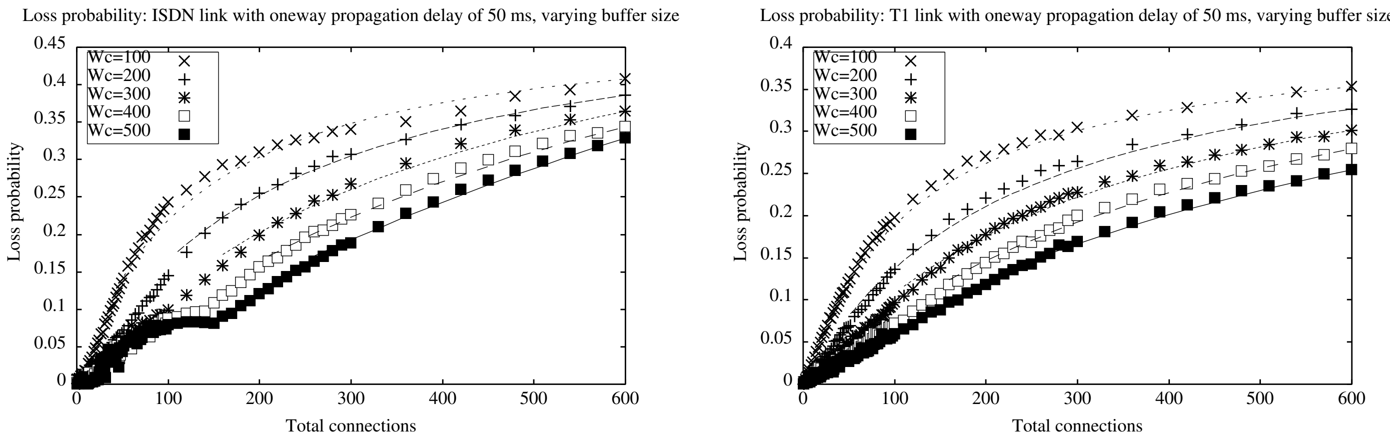


Figure 22: Loss probability for large number of connections after adding up to $10\% * RTT$ at TCP source: varying number of connections compete for the bottleneck link of either ISDN or T1. In both cases, the oneway propagation delay = 50 ms. The loss probability curves match very well with hyperbolic curves when the number of connections is large, where the parameters of the hyperbolic curves are given in Table 5

ISDN			T1		
Wc	k	Scale	Wc	k	Scale
100	125.0000	0.4930	100	125.0000	0.4263
200	217.3913	0.5257	200	227.2727	0.4505
300	400.0000	0.6071	300	333.3333	0.4690
400	714.2857	0.7536	400	500.0000	0.5123
500	1428.5714	1.1110	500	666.6666	0.5366

Table 5: Parameters for the hyperbolic curves used for fitting loss probability as shown in Figure 22

scale. As shown in Figure 23, the throughput ratio is bounded by two curves. More specifically, when $RTT_1 \leq RTT_2$,

$$(\frac{RTT_2}{RTT_1})^2 \leq \frac{Throughput_1}{Throughput_2} \leq 2 * (\frac{RTT_2}{RTT_1})^2 \qquad A(1)$$

where RTT_1 and RTT_2 are the average RTT the connections in group 1 and group 2 experience respectively. Since we can swap the labels for groups 1 and 2, so the throughput ratio is symmetric as shown in Figure 23(a).

Now let's try to explain the relationship (A1). For ease of discussion, we aggregate all the connections in one group as a big connection. So in the following we just consider two connections compete with each other. Moreover, due to symmetry, we only need to consider the case when $RTT_1 \leq RTT_2$.

Figure 24 depicts roughly how the congestion windows evolve during congestion for two connections with different RTT. As shown in the figure, during every epoch the $cwnd$ of connection i grows from W_i to $W_i * 2$. So the average length of epoch, denoted as E_i, is roughly equal to $RTT * W_i$. Therefore

$$Throughput_i = \frac{3 * W_i^2}{2} * \frac{1}{Ei} = \frac{3 * W_i}{2 * RTT_i} \qquad (A2)$$

.

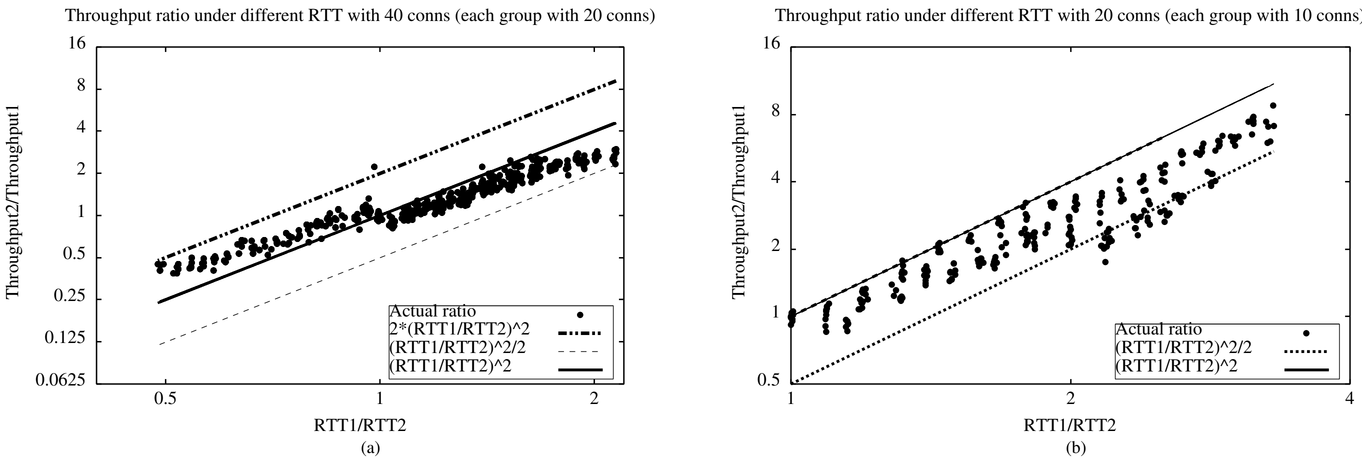


Figure 23: Two groups of TCP connections compete for T1 link with oneway propagation delay of 50 ms

Now let x denote $\frac{E_1}{E_2}$. Using (A2), we have

$$\frac{Throughput_1}{Throughput_2} = (\frac{RTT_2}{RTT_1})^2 * x.$$

Applying the equality of $A(1)$, we obtain $1 \leq x \leq 2$. This means the average epoch length of connection 1 is usually no larger than twice the epoch length of connection 2. That is, for every two losses in connection 2, on average there is usually at least one loss in connection 1. This implies there is no consistent discrimination against any particular connection, which is likely to be the case after adding random processing time [4].

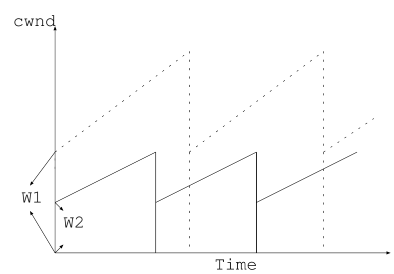


Figure 24: Window evolution graph for two connections with different RTT's

The roundtrip time bias in TCP/IP networks has received lots of attention. There have been a number of studies on analyzing such bias. [8] gives analytical explanation for this, and concludes the ratio of the throughput of two connections (i.e. $\frac{Throughput_i}{Throughput_j}$) is proportional to $(\frac{RTT_j}{RTT_i})^2$. The analysis is based on TCP-Tahoe window evolution. As pointed out in [8], since in TCP-Reno the number of times the window is halved at the onset of congestion equals the number of lost packets, and since phase effects can cause one connection to systematically lose a large number of packets, it is possible that a connection gets almost completely shut off. Therefore the throughput ratio of two connections using TCP-Reno is unpredictable: the connection with smaller propagation delay can sometimes get lower throughput due to systematical discrimination. *Our simulation study shows that though the throughput ratio of two connections may fluctuate a lot, the aggregate throughput ratio of two groups of connections is relatively stable when the group size is reasonably large.* Furthermore, the analysis in [8] is based on the assumption that two connections with different RTT's are synchronized. This does not always hold.

Therefore the throughput ratios $\frac{Throughput_2}{Throughput_1}$ do not match very well with the curve $(\frac{RTT_1}{RTT_2})^2$ as shown in Figure 23. Instead our results indicate *the throughput ratio is clustered within a band close to* $(\frac{RTT_1}{RTT_2})^2$*, and the width of the band is usually one unit in log2 scale*.

As part of our future work, we will investigate how TCP behaves when the number of different RTT groups increases. Also we are interested in examining how the performance is affected when the number of connections in each group is unequal.

6 Related Work

Large scale performance analysis has been an active research area recently. Many researches are currently focused on building scalable simulators, such as [1, 6, 16].

Analyzing simulation results to estimate TCP performance, as done in this project, is a very different approach from building a scalable simulator. The strategy taken by [10] is the closest to ours. It studies how TCP throughput, loss rates, and fairness are affected by changing the number of flows. Their work differs from ours in that we have created a generic abstract model of Internet connection, and focused on studying how TCP performance changes by varying parameters of the model, whereas they focus on studying how the TCP behaves as one of the parameters - the number of flows changes, while keeping all the other parameters to be something reasonable. Furthermore they study TCP tahoe assuming RED dropping policy at the routers, whereas we study TCP reno using drop-tail, since they are more widely deployed in today's Internet. RED dropping policy is not sensitive to instantaneous queue occupancy, so it is relatively easy to obtain the steady state performance. A number of analytical models have been developed for studying the steady state TCP throughput when routers use RED dropping policy [9, 17]. However modeling multiple connections sharing a bottleneck link with drop-tail policy is much more challenging, since such policy is very sensitive to the instantaneous queue occupancy, and loss is non-randomized. Simulation approach, as employed in this paper, proves to be an effective approach for studying TCP performance under drop tail policy.

7 Conclusion and Future Work

In this paper, we have investigated the individual and aggregate TCP performance. We first develop a generic network model that captures the essence of wide area Internet connections. Based on the abstract model, we study the behavior of a single TCP connection under other competing TCP flows by evaluating the TCP analytical model proposed in [13]. We also examine the aggregate behavior of many concurrent TCP flows. Through extensive simulations, we have identified how TCP performance is affected with changing parameters in the network model. These results give us valuable insights into how TCP behaves in diverse Internet.

There are a number of directions for future work. First, we have shown the loss probability curves can be approximated quite well with simple analytical functions. As part of our future work, we will investigate how to quantitatively determine the parameters in the functions. Second, we plan to further explore TCP performance under different RTT's. In particular, we want to consider the following two extensions: (i) when the two different RTT groups are not equal size; and (ii) with different number of RTT groups. Finally, we plan to use Internet experiments to verify some of the results in the paper.

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