

# TCP Vegas: New Techniques for Congestion Detection and Avoidance

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## **Abstract**

Vegas is a new implementation of TCP that achieves between 40 and 70% better throughput, with one-half to one-fifth the losses, as compared to the implementation of TCP in the Reno distribution of BSD Unix. This paper motivates and describes the five key techniques employed by Vegas, and presents the results of a comprehensive experimental performance study—using both simulations and measurements of the actual Internet—of the Vegas and Reno implementations of TCP.

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# 1 Introduction

Few would argue that one of TCP's strengths lies in its adaptive retransmission and congestion control mechanism, with Jacobson's seminal paper [5] providing the cornerstone of that mechanism. This paper attempts to go beyond this earlier work—to provide some new insights into congestion control, and to propose modifications to TCP that exploit these insights.

The tangible result of this effort is a new implementation of TCP that we refer to as TCP *Vegas*. This name is a take-off of earlier implementations of TCP that were distributed in releases of 4.3 BSD Unix known as Tahoe and Reno; we use Tahoe and Reno to refer to the TCP implementation instead of the Unix release.<sup>1</sup> Note that Vegas does not involve any changes to the TCP specification; it is merely an alternative implementation that is able to interoperate with any other valid implementation of TCP. In fact, all the changes are confined to the sending side.

The main result reported in this paper is that Vegas is able to achieve between 40 and 70% better throughput than Reno. Moreover, this improvement in throughput is not achieved by an aggressive retransmission strategy that effectively steals bandwidth away from TCP connections that use the current algorithms. Rather, it is achieved by a more efficient use of the available bandwidth. Our experiments show that Vegas retransmits between one-half and one-fifth as much data as does Reno.

This paper is organized as follows. Section 2 outlines the tools we used to measure and analyze TCP. Section 3 then describes the techniques employed by TCP Vegas, coupled with the insights that led us to the techniques. Sections 4 and 5 present a comprehensive evaluation of Vegas' performance; the former reports simulation results and the latter gives preliminary numbers for measurements of TCP running over the Internet. Finally, Section 6 makes some concluding remarks.

## 2 Tools

This section briefly describes the tools we used to implement and analyze the different versions of TCP. All of the protocols were developed and tested under the University of Arizona's *x*-kernel framework [4]. Our implementation of Reno was derived by retrofitting the BSD implementation into the *x*-kernel. Our implementation of Vegas was derived by modifying Reno.

### 2.1 Simulator

Many of the results reported in this paper were measured on a network simulator. Even though several good simulators are available—e.g., REAL [8] and Netsim [3]—we decided to build our own simulator based on the *x*-kernel. In this environment, actual *x*-kernel protocol implementations run on a simulated network. Specifically, the simulator supports multiple hosts, each running a full protocol stack (TEST/TCP/IP/ETH), and several abstract link behaviors (point-to-point connections and ethernet). Routers can be modeled either as a network node running the actual IP protocol code, or as an abstract entity that supports a particular queuing discipline (e.g., FIFO).

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<sup>1</sup>We limit our discussion to Reno, which is both newer and better performing than Tahoe. Also note that in terms of the congestion-related algorithms, Reno is roughly equivalent to the BNR2 implementation of TCP.

The  $x$ -kernel-based simulator has several nice features. First, it provides a realistic setting for evaluating protocols; each protocol is modeled by the actual C code that implements it, rather than some more abstract specification. Second, it automatically makes all the protocols we have implemented in the  $x$ -kernel available for analysis in the simulator. Third, it is trivial to move protocols between the simulator and the real world, thereby providing a comprehensive protocol design, implementation, and testing environment.

One of the most important protocols available in the simulator is called TRAFFIC—it implements TCP Internet traffic based on *tcplib* [2]. TRAFFIC starts conversations with interarrival times given by an exponential distribution. Each conversation can be of type TELNET, FTP, NNTP, or SMTP, each of which expects a set of parameters. For example, FTP expects the following parameters: number of items to transmit, control segment size, and the item sizes. All of these parameters are based on probability distributions obtained from traffic traces. Finally, each of these conversations runs on top of its own TCP connection.

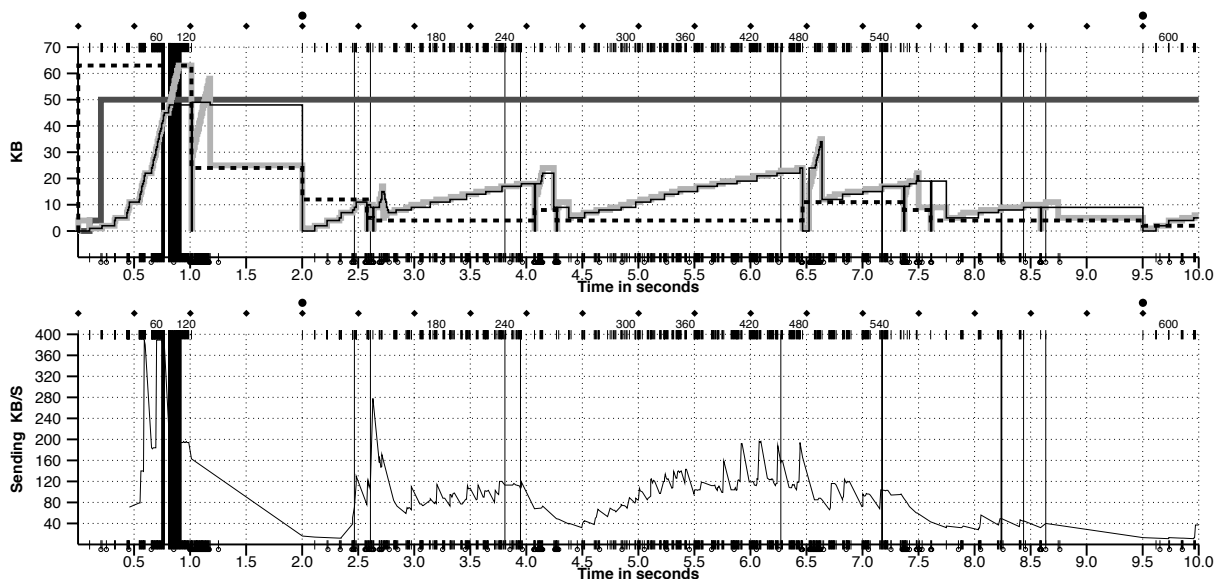


Figure 1: TCP Reno Trace Examples

## 2.2 Trace Facility

Early in this effort it became clear that we needed good facilities to analyze the behavior of TCP. We therefore added code to the  $x$ -kernel to trace the relevant changes in the connection state. We paid particular attention to keeping the overhead of this tracing facility as low as possible, so as to minimize the effects on the behavior of the protocol. Specifically, the facility writes trace data to memory, dumps it to a file only when the test is over, and keeps the amount of data associated with each trace entry small (8 bytes).

We then developed various tools to analyze and display the tracing information. The rest of this section describes one such tool—it graphically represents relevant features of the state of the TCP connection as a function of time. This tool outputs multiple graphs, each of which focuses on a specific set of characteristics

of the connection state. Figure 1 gives an example. Since we use graphs like this throughout the paper, we now explain how to read the graph in some detail.

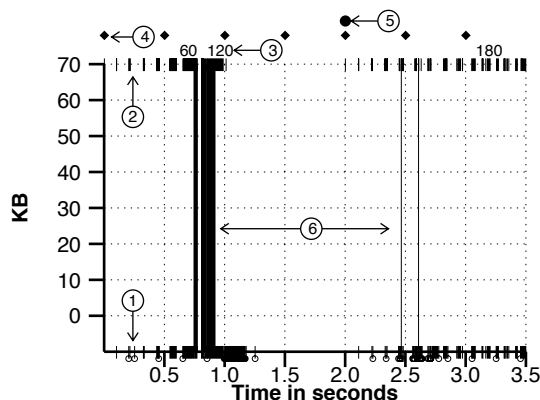


Figure 2: Common Elements in TCP Trace Graphs

First, all of the TCP trace graphs have certain features in common, as illustrated in Figure 2. The circled numbers in this figure are keyed to the following explanations:

1. Hash marks on the  $x$ -axis indicate that an ACK was received at that time.
2. Hash marks at the top of the graph indicate that a segment was sent at that time.
3. The numbers on the top of the graph indicate the time when the  $n^{th}$  kilobyte (KB) was sent.
4. Diamonds on top of the graph indicate when a coarse-grained timer event happens. This does not imply a timeout, just that the clock facility woke up and checked if any timeouts should happen.
5. Circles on top of the graph indicate the time when a coarse-grained timeout occurred, causing a segment to be retransmitted.
6. Solid vertical lines running the whole height of the graph indicate the time when a segment that is eventually retransmitted was originally sent, presumably because it was lost.<sup>2</sup> Notice that several consecutive segments are retransmitted in the example.

In addition to this common information, each graph depicts more specific information. The bottom graph in Figure 1 is the simplest—it shows the average sending rate, calculated from the last 12 segments. The top graph in Figure 1 is more complicated—it gives the size of the different windows used by TCP for flow and congestion control. Figure 3 shows these in more detail, again keyed by the following explanations:

1. The dashed line gives the threshold window. It is used when doing slow-start, and marks the point at which the congestion window growth changes from exponential to linear.
2. The dark gray line gives the send window. It is the minimum of the sender's buffer size and receiver's available buffer space.

<sup>2</sup>For simplicity, we sometimes say a segment was lost, even though all we know for sure is that the sender retransmitted it.

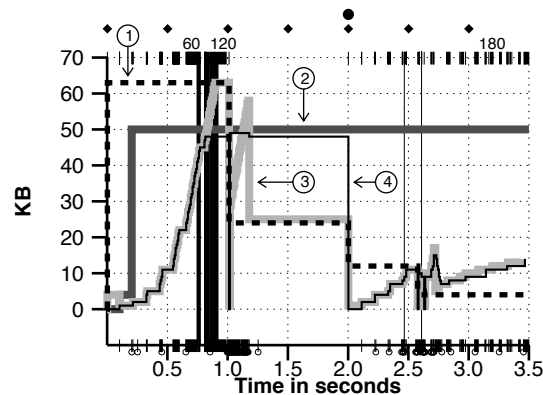


Figure 3: TCP Windows Graph

3. The light gray line gives the congestion window. It is used for congestion control, and is an upper limit to the number of bytes sent but not yet acknowledged.
4. The thin line gives the actual number of bytes in transit at any given time, where by in transit we mean sent but not yet acknowledged.

The graphs just described are obtained from tracing information saved by the protocol, and are thus available whether the protocol is running in the simulator or over a real network. The simulator itself also reports certain information, such as the rate, in KB/s, at which data is entering or leaving a host or a router. For a router, the traces also save the size of the queues as a function of time, and the time and size of segments that are dropped due to insufficient queue space.

### 3 Techniques

This section motivates and describes five techniques that Vegas employs to increase throughput and decrease losses. The first two of these techniques are simple extensions that result in a more timely decision to retransmit a dropped segment. The third technique is an adjustment to TCP's congestion window resizing algorithm; it is required because of Vegas' more accurate timeout mechanism. The fourth technique is designed to better space transmissions. The fifth technique is the most novel. It gives TCP the ability to anticipate congestion, and adjust its transmission rate accordingly. The relationship between our techniques and those proposed elsewhere are discussed at the appropriate places throughout this section.

#### 3.1 More Accurate RTT Calculation

In Reno, round trip time (RTT) and variance estimates are computed using a coarse-grained timer (around 500 ms), meaning that the RTT estimate is not very accurate. This coarse granularity influences both the accuracy of the calculation itself, and how often TCP checks to see if it should time out on a segment. For example, during a series of tests on the Internet, we found that for losses that resulted in a timeout—usually due to two or more dropped segments in a RTT, the exact figure depends on the number of segments in

transit—it took Reno an average of 1100ms from the time it sent a segment that was lost until it timed out and resent the segment, whereas less than 300ms would have been the correct timeout interval had a more accurate clock been used.

In contrast, Vegas reads and records the system clock each time a segment is sent. When an ACK arrives, Vegas reads the clock again and does the RTT calculation using this time and the timestamp recorded for the relevant segment. The proposed extension to TCP includes a Timestamp option that could be used to accomplish the same effect [6], although our approach obviously needs no cooperation from the receiver.

Using an accurate RTT estimate serves two purposes. First, it leads to a more accurate timeout calculation. Second, it is used in conjunction with the mechanism described in the next subsection to decide to retransmit a dropped segment in a more timely fashion.

### 3.2 New Mechanism for Deciding to Retransmit

Reno not only retransmits when a coarse-grained timeout occurs, but also when it receives  $n$  duplicate ACKs ( $n$  is usually 3). Reno sends a duplicate ACK whenever it receives new data that it cannot acknowledge because it has not yet received all the previous data. Vegas extends the duplicate ACK mechanism as follows; a simple example is given in Figure 4.

- When a duplicate ACK is received, Vegas checks to see if the difference between the current time and the timestamp recorded for the relevant segment is greater than the timeout value. If it is, then Vegas retransmits the segment without having to wait for  $n$  (3) duplicate ACKs. In many cases, losses are either so great or the window so small that the sender will never receive three duplicate ACKs, and therefore, Reno would have to rely on the coarse-grained timeout mentioned above.
- When a non-duplicate ACK is received, if it is the first or second one after a retransmission, Vegas again checks to see if the time interval since the segment was sent is larger than the timeout value. If it is, then Vegas retransmits the segment. This will catch any other segment that may have been lost previous to the retransmission without having to wait for a duplicate ACK.

In other words, Vegas treats the receipt of certain ACKs as a trigger to check if a timeout should happen.<sup>3</sup> It still contains Reno's coarse-grained timeout code in case these mechanisms fail to recognize a lost segment.

### 3.3 Modified Window Sizing on Congestion

The congestion window should only be reduced due to losses that happened at the current sending rate, and not due to losses that happened at an earlier, higher sending rate. In Reno, it is possible to decrease the congestion window more than once for losses that occurred during one RTT interval. In contrast, Vegas only decreases the congestion window if the retransmitted segment was previously sent *after* the last decrease. Any losses that happened before the last window decrease do not imply that the network is congested for the *current* congestion window size, and therefore do not imply that it should be decreased again.

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<sup>3</sup>An obviously more accurate thing to do is schedule a timer event for each segment as it is transmitted. However, our experience is that this is too expensive.

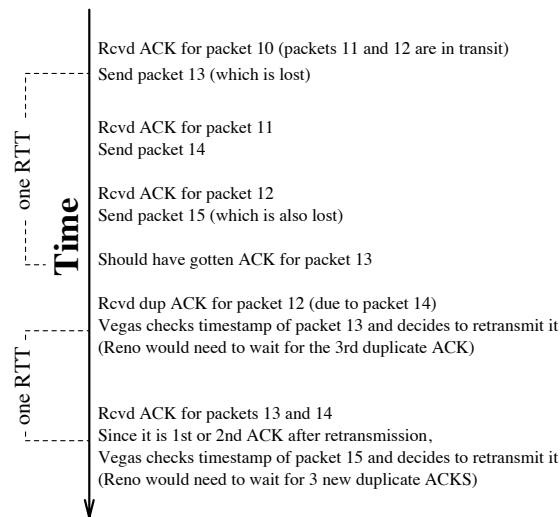


Figure 4: Example of Retransmit Mechanism

This policy is needed because Vegas is much better than Reno at detecting losses. If Vegas used Reno's window sizing policy, it would decrease its congestion window *more* than Reno simply because it detected losses sooner than Reno.

### 3.4 Spike Suppression

TCP's self-clocking feature depends on the arrival of evenly spaced ACKs, each of which acknowledges roughly a segment's worth of data. Based on traces of Reno, we concluded that many Reno losses are due to a failure in the self-clocking mechanism that result in *spikes* in the sending rate. By a spike in the sending rate, we mean that the sender transmits multiple segments back-to-back; Figure 1 at time 2.7 seconds illustrates an example of a spike. There are two main reasons that spikes occur:

1. Large cumulative ACKs—a single ACK that acknowledges several kilobytes due to segments arriving out-of-order at the receiver.
2. ACK compression [13, 10]—a sequence of ACKs arrive closer together than they were sent.

In the case of the large ACKs, the main reason for segments arriving out-of-order is that segments are lost and then retransmitted. Thus, one often sees large spikes in the sending rate right after certain losses, those for which the congestion window is large enough to allow a lot of data to be sent. Spikes that lead to losses are commonly seen during slow-start; more on this in Section 3.5.

Although the idea behind the congestion window is to limit the number of bytes sent during one RTT, the congestion window does not control how those bytes are spaced over the RTT. Ideally, one would want to space the bytes (segments) as evenly as possible to prevent overrunning the buffers on the intermediate nodes of the connection. If the maximum segment size, such that there is no fragmentation, is given by *MaxSeg*, and *WindowSize* denotes the size of the congestion window, then the ideal segment spacing is given by:

$$SegSpacing = RTT \times MaxSegment / WindowSize$$

Vegas' spike suppression mechanism consists of only allowing twice *MaxSegment* bytes on each *SegSpacing* interval; the total number of bytes transmitted per RTT is still bounded by the congestion window. It allows for transmitting twice the ideal number of bytes per *SegSpacing* so the current number of bytes in transit can *catch up* with the allowed number given by the congestion window. The spike suppression mechanism is turned off during slow-start because transmission spikes are necessary to keep up with the exponentially increasing transmission rate of the slow-start mechanism.

### 3.5 Congestion Detection and Avoidance

TCP Reno's congestion detection and control mechanism uses the loss of segments as a signal that there is congestion in the network. It has no mechanism to detect the incipient stages of congestion—before losses occur—so they can be prevented. Reno is reactive, rather than proactive, in this respect. As a result, Reno *needs* to create losses to find the available bandwidth of the connection. This can be seen in Figure 5, which shows the trace of a Reno connection sending 1MB of data over the network configuration seen in Figure 6, with no other traffic sources; i.e., only Host1a sending to Host1b. In this case, the router queue size is ten and the queuing discipline is FIFO.

#### 3.5.1 Previous Work

There are several previously proposed approaches for proactive congestion detection based on a common understanding of the network changes as it approaches congestion (an excellent development is given in [7]). These changes can be seen in Figure 5 in the time interval from 4.5 to 7.5 seconds. One change is the increased queue size in the intermediate nodes of the connection, resulting in an increase of the RTT for each successive segment. Wang and Crowcroft's DUAL algorithm [12] is based on this increase of the round-trip delay. The congestion window normally increases as in Reno, but every two round-trip delays it checks to see if the current RTT is greater than the average of the minimum and maximum RTTs seen so far. If it is, then it decreases the congestion window by one-eighth.

Jain's CARD (Congestion Avoidance using Round-trip Delay) approach [7] is based on an analytic derivation of a socially optimum window size for a deterministic network. The decision as to whether or not to change the current window size is based on changes to both the RTT and the window size. The window is adjusted once every two round-trip delays based on the product  $(WindowSize_{current} - WindowSize_{old}) \times (RTT_{current} - RTT_{old})$  as follows: if the result is positive, decrease the window size by one-eighth; if the result is negative or zero, increase the window size by one maximum segment size. Note that the window changes during every adjustment, that is, it oscillates around its optimal point.

Another of the changes seen as the network approaches congestion is the flattening of the sending rate. Wang and Crowcroft's Tri-S scheme [11] takes advantage of this fact. Every RTT, they increase the window size by one segment and compare the throughput achieved to the throughput when the window was one segment smaller. If the difference is less than one-half the throughput achieved when only one segment was in transit—as was the case at the beginning of the connection—they decrease the window by one segment. Tri-S calculates the throughput by dividing the number of bytes outstanding in the network by the RTT.



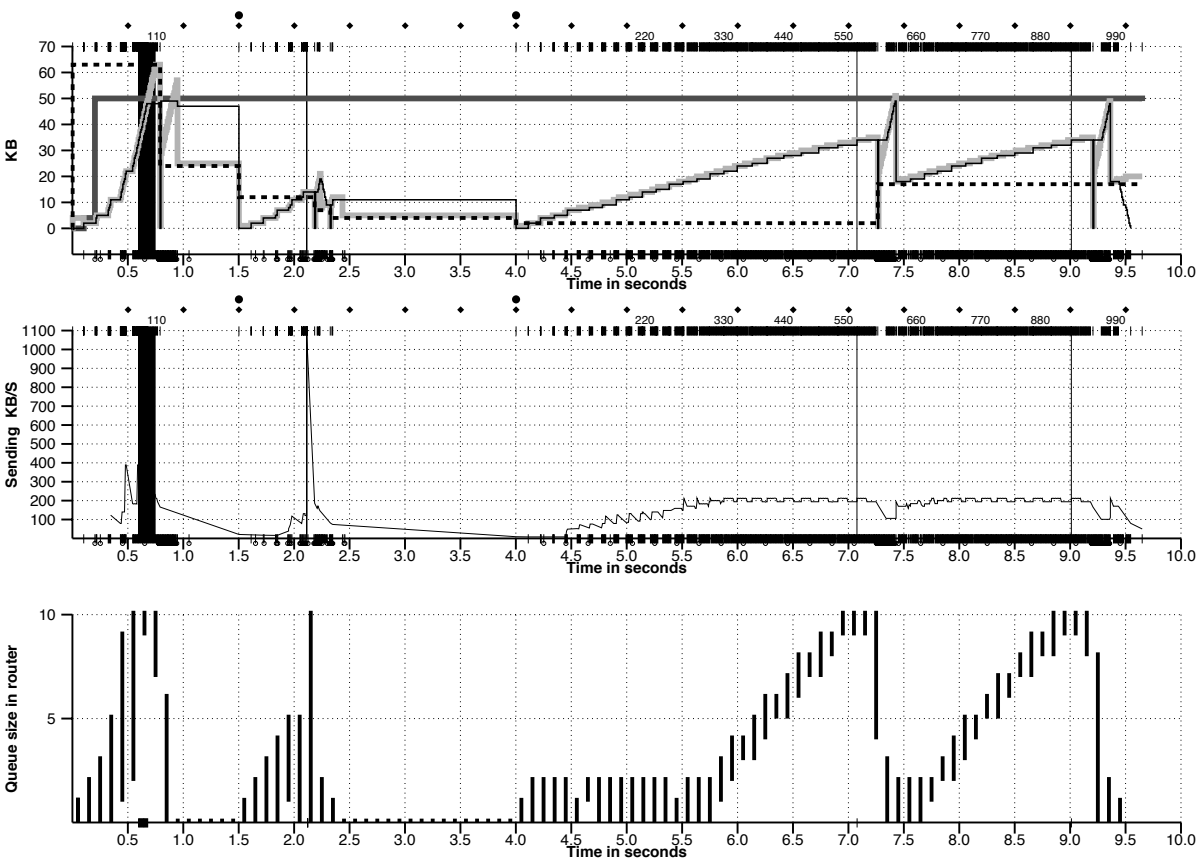


Figure 5: TCP Reno with No Other Traffic (Throughput: 105 KB/s)

As a final example, Keshav’s Packet-Pair rate probing technique sends two back-to-back segments and gets an estimate of the available bandwidth from the delay observed between the ACKs [9].

### 3.5.2 Congestion Detection in Vegas

Vegas’ approach is most similar to Tri-S, in that it looks at changes in the throughput rate. However, it differs from Tri-S in that it calculates throughputs differently, and instead of looking for a change in the throughput slope, it compares the measured throughput rate with an expected throughput. The simple idea that Vegas exploits is that the number of bytes in transit is directly proportional to the expected throughput, and therefore, as the window size increases—causing the bytes in transit to increase—the throughput of the connection should also increase.

Vegas uses this idea to measure and control the amount of *extra* data this connection has in transit, where by extra data we mean data that would not have been sent if the bandwidth used by the connection exactly matched the available bandwidth of the network. The goal of Vegas is to maintain the “right” amount of extra data in the network. Obviously, if a connection is sending too much extra data, it will cause congestion. Less obviously, if a connection is sending too little extra data, it cannot respond rapidly enough to transient

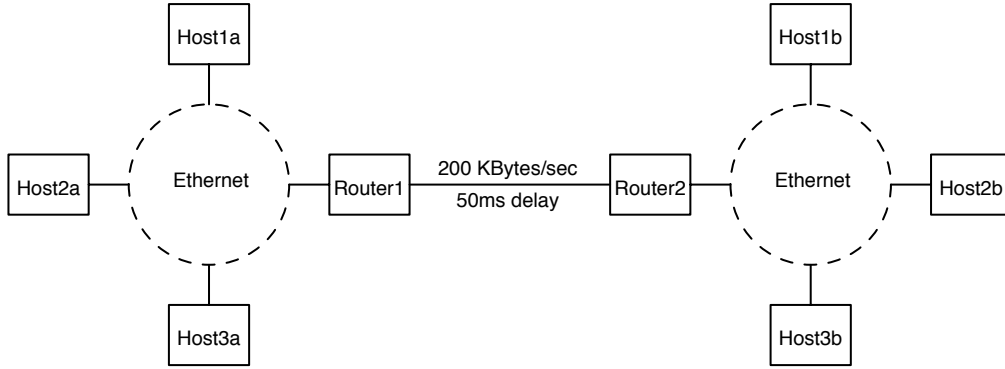


Figure 6: Network Configuration for Simulations

increases in the available network bandwidth. Vegas' congestion avoidance actions are based on changes in the estimated amount of extra data in the network, and not only on dropped segments.

We now describe the algorithm in detail. Note that the algorithm is not in effect during slow-start. Vegas' behavior during slow-start is described in the following subsection.

First, define a given connection's *BaseRTT* to be the RTT of a segment when the connection is not congested. In practice, Vegas sets *BaseRTT* to the minimum of all measured round trip times; it is commonly the RTT of the first segment sent by the connection, before the router queues increase due to traffic generated by this connection. If we assume that we are not overflowing the connection, then the expected throughput is given by:

$$Expected = WindowSize / BaseRTT$$

where *WindowSize* is the size of the current congestion window, which we assume for the purpose of this discussion, to be equal to the number of bytes in transit.

Second, Vegas calculates the current *Actual* throughput. This is done by recording the sending time for a distinguished segment, recording how many bytes are transmitted between the time that segment is sent and its acknowledgement is received, computing the RTT for the distinguished segment when its acknowledgement arrives, and dividing the number of bytes transmitted by the sample RTT. This calculation is done once per round-trip time.<sup>4</sup>

Third, Vegas compares *Actual* to *Expected*, and adjusts the window accordingly. Let  $Diff = Expected - Actual$ . Note that *Diff* is positive or zero by definition, since  $Actual > Expected$  implies that we need to change *BaseRTT* to the latest sampled RTT. Also define two thresholds,  $\alpha < \beta$ , roughly corresponding to having too little and too much extra data in the network, respectively. When  $Diff < \alpha$ , Vegas increases the congestion window linearly during the next RTT, and when  $Diff > \beta$ , Vegas decreases the congestion window linearly during the next RTT. Vegas leaves the congestion window unchanged when  $\alpha < Diff < \beta$ .

Intuitively, the farther away the actual throughput gets from the expected throughput, the more congestion there is in the network, which implies that the sending rate should be reduced. The  $\beta$  threshold triggers this

<sup>4</sup>We have made every attempt to keep the overhead of Vegas' congestion avoidance mechanism as small as possible. To help quantify this effect, we ran both Reno and Vegas between SparcStations connected by an Ethernet, and measured it to be less than a 5%. This overhead can be expected to drop as processors become faster.

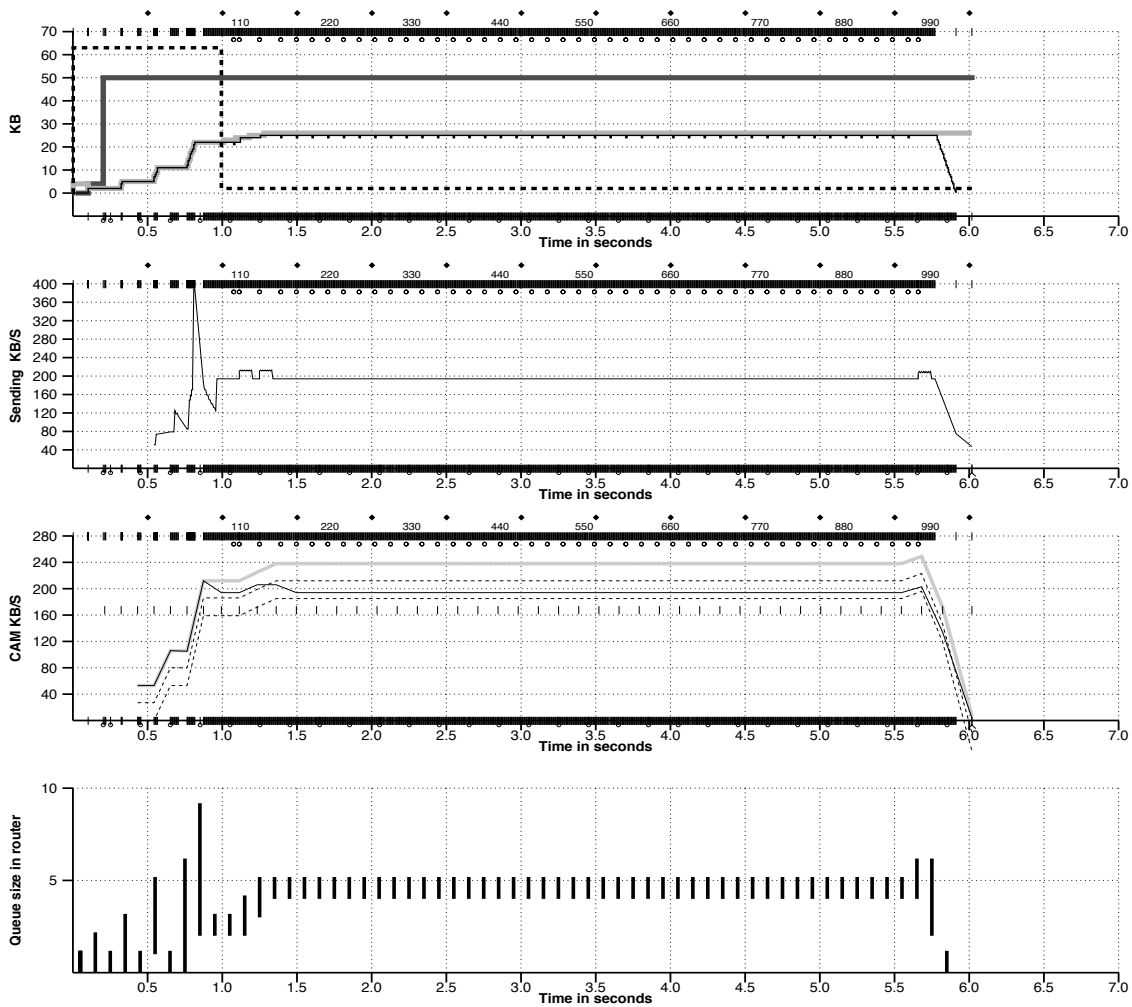


Figure 7: TCP Vegas with no Other Traffic (Throughput: 169 KB/s)

decrease. On the other hand, when the actual throughput rate gets too close to the expected throughput, the connection is in danger of not utilizing the available bandwidth. The  $\alpha$  threshold triggers this increase. The overall goal is to keep between  $\alpha$  and  $\beta$  extra bytes in the network.

Because the algorithm, as just presented, compares the difference between the actual and expected throughput rates to the  $\alpha$  and  $\beta$  thresholds, these two thresholds are defined in terms of KB/s. However, it is perhaps more accurate to think in terms of how many extra *buffers* the connection is occupying in the network. For example, on a connection with a *BaseRTT* of 100ms and a segment size of 1KB, if  $\alpha = 30\text{KB/s}$  and  $\beta = 60\text{KB/s}$ , then we can think of  $\alpha$  as saying that the connection needs to be occupying at least three extra buffers in the network, and  $\beta$  saying it should occupy no more than six extra buffers in the network.

In practice, we express  $\alpha$  and  $\beta$  in terms of buffers rather than extra bytes in transit, and we determine a suitable value experimentally. During linear increase/decrease mode—as opposed to the slow-start mode described below—we set  $\alpha$  to two and  $\beta$  to four. This can be interpreted as an attempt to use at least two,

but no more than four extra buffers in the connection.

Figure 7 shows the behavior of TCP Vegas when there is no other traffic present; this is the same condition that Reno ran under in Figure 5. There is one new type of graph in this figure, the third one, which depicts the congestion avoidance mechanisms (CAM) used by Vegas. Once again, we use a detailed graph (Figure 8) keyed to the following explanation:

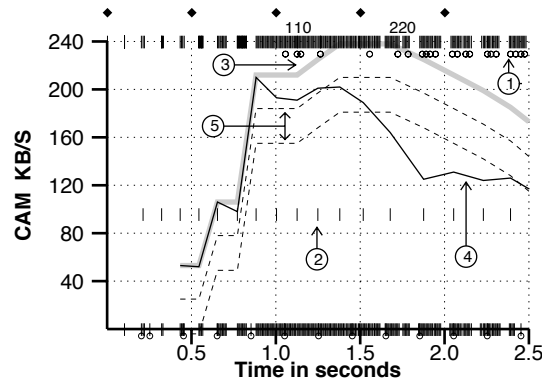


Figure 8: Congestion detection and avoidance in Vegas

1. The small circles below the send segment marks on top of the graph are indications that Vegas' spike suppression mechanism prevented segments from being sent.
2. The small vertical line—once per RTT—shows the times when Vegas does congestion control decision; i.e., computes *Actual* and adjusts the window accordingly.
3. The gray line shows the *Expected* throughput. This is the throughput we should get if all the bytes in transit are able to get through the connection in one *BaseRTT*.
4. The solid line shows the actual sending rate *Actual*. We calculate it from the number of bytes we sent in the last RTT.
5. The dashed lines are the thresholds used to control the size of the congestion window. The top line corresponds to the  $\alpha$  threshold, and the bottom line corresponds to the  $\beta$  threshold.

Figure 9 shows a trace of Vegas sharing the bottleneck router with *tcplib* traffic. The bottom graph shows the output produced by the TRAFFIC protocol simulating the TCP traffic. The thin line is the sending rate in KB/s as seen in 100ms intervals; the thick line is a running average (size 3). The graph clearly shows Vegas' congestion avoidance mechanisms at work and how its throughput adapts to the changing conditions on the network.

### 3.5.3 Modified Slow-Start with Congestion Detection

Reno's slow-start mechanism is very expensive in terms of losses when it is not limited by a small send buffer or a slow host. Since it doubles the size of the congestion window every RTT while there are no

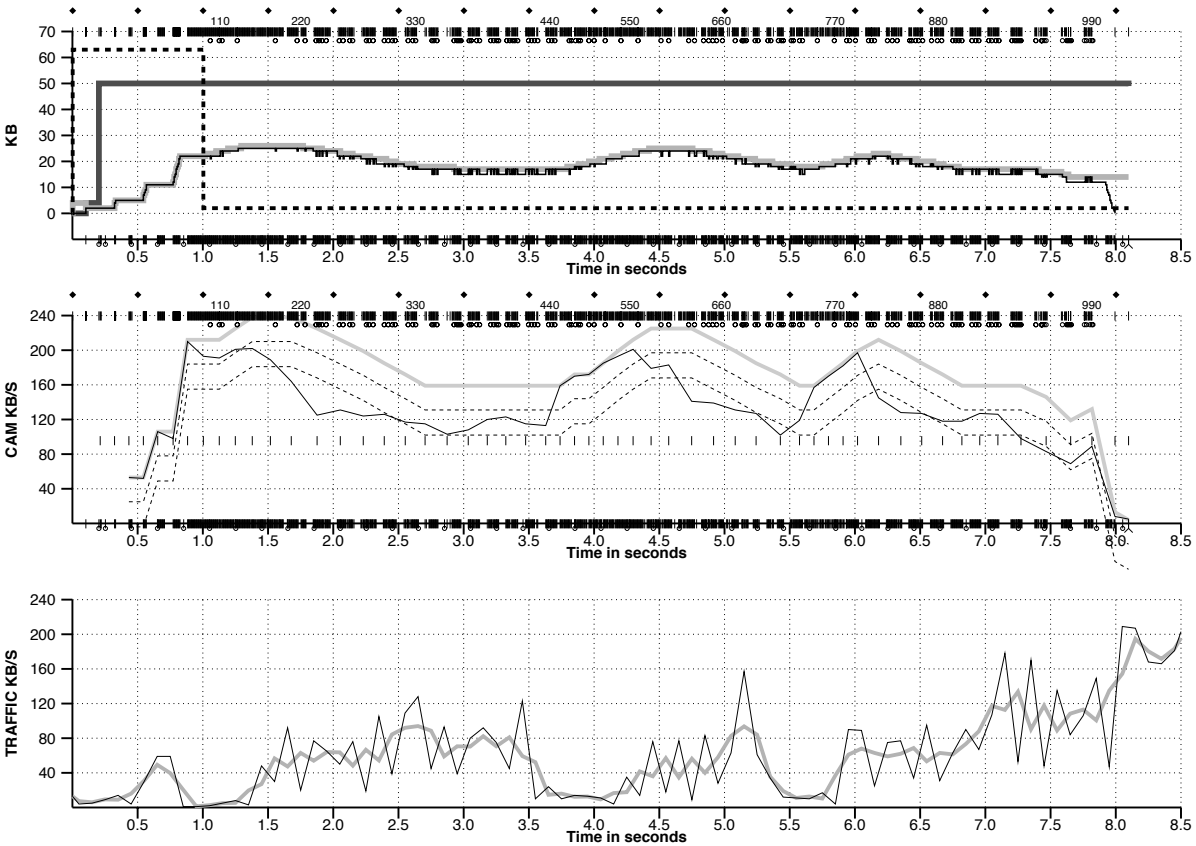


Figure 9: TCP Vegas with *tcplib*-Generated Background Traffic

losses—which is equivalent to doubling the attempted throughput every RTT—when it finally overruns the connection’s bandwidth, we can expect losses in the order of *half* the current congestion window, more if we encounter a burst from another connection.

As network bandwidth increases, we can anticipate a similar increase in the expected losses of slow-start. We need a way to find a connection’s available bandwidth that does not incur this kind of loss. Towards this end, we incorporated our congestion detection mechanism into slow-start with only minor modifications. To be able to detect and avoid congestion during slow-start, Vegas allows exponential growth only every other RTT. In between, the congestion window stays fixed so a valid comparison of the expected and actual rates can be made. When the actual rate falls below the expected rate by a certain amount—we call this the  $\gamma$  threshold—Vegas changes from slow-start mode to linear increase/decrease mode. As with the  $\alpha$  and  $\beta$  thresholds,  $\gamma$  was determined experimentally.

The behavior of the modified slow-start can be seen in Figures 7 and 9. The reason that we need to measure the actual rate with a fixed congestion window is that we want the actual rate to represent the bandwidth allowed by the connection. Thus, we can only send as much data as is acknowledged in the ACK. (During slow-start, Reno sends two segments for each ACK received.)

## 4 Simulation Results

This section reports the results of simulated runs of TCP Vegas and Reno using the  $x$ -kernel based simulator. Although we have also measured Vegas and Reno on the actual Internet—see Section 5—the simulator allows us to better control the experiment, and in particular, gives us the opportunity to see whether or not Vegas gets its performance at the expense of Reno-based connections. Note that all the experiments used in this section are on the network configuration shown in Figure 6.

### 4.1 One-on-One Experiments

We begin by studying how two TCP connections interfere with each other. To do this, we start a 1MB transfer, and then after a variable delay, start a 300KB transfer. The transfer sizes and delays are chosen to ensure that the smaller transfer is contained completely within the larger.

	Reno/Reno	Reno/Vegas	Vegas/Reno	Vegas/Vegas
Throughput (KB/s)	60/109	61/123	66/119	74/131
Throughput Ratios	1.00/1.00	1.02/1.13	1.10/1.09	1.23/1.20
Retransmissions (KB)	30/22	43/1.8	1.5/18	0.3/0.1
Retransmit Ratios	1.00/1.00	1.43/0.08	0.05/0.82	0.01/0.01

Table 1: One-on-One (300KB and 1MB) Transfers

Table 1 gives the results for the four possible combinations, where the column heading Reno/Vegas denotes a 300KB transfer using Reno contained within a 1MByte transfer using Vegas. For each combination, the table gives the measured throughput and number of kilobytes retransmitted for both transfers; e.g., in the case of Reno/Vegas, the 300KB Reno transfer achieved a 61KB/s throughput rate and the 1MByte Vegas transfer achieved a 123KB/s throughput rate.<sup>5</sup> The ratios for both throughput rate and kilobytes retransmitted are relative to the Reno/Reno column. The values in the table are averages from 12 runs, using 15 and 20 buffers in the routers and the delay before starting the smaller transfer increasing from 0 to 2.5 seconds.

The main thing to take away from these numbers is that Vegas does not adversely affect Reno’s throughput. Reno’s throughput stays pretty much unchanged when it is competing with Vegas rather than itself—the ratios for Reno are 1.02 and 1.09 for Reno/Vegas and Vegas/Reno, respectively. Also, when Reno competes with Vegas rather than itself, the combined number of kilobytes retransmitted for the pair of competing connections drops significantly. The combined Reno/Reno retransmits are 52KB compared with 45KB for Reno/Vegas and 19KB for Vegas/Reno. Finally, note that the combined Vegas/Vegas retransmits are less than 1KB on the average—an indication that the congestion avoidance mechanism is working.

### 4.2 Experiments with Background Traffic

We next measured the performance of a distinguished TCP connection when the network is loaded with traffic generated from *tcplib*. That is, the protocol TRAFFIC is running between Host1a and Host1b in

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<sup>5</sup>Comparing the small transfer to the large transfer in any given column is not meaningful. This is because the large transfer was able to run by itself during most of the test.

Figure 6, and a 1MByte transfer is running between Host2a and Host2b. In this set of experiments, the *tcplib* traffic is running over Reno.

	Reno	Vegas-1,3	Vegas-2,4
Throughput (KB/s)	58.30	89.40	91.80
Throughput Ratio	1.00	1.53	1.58
Retransmissions (KB)	55.40	27.10	29.40
Retransmit Ratio	1.00	0.49	0.53
Coarse Timeouts	5.60	0.90	0.90

Table 2: 1MByte Transfer with *tcplib*-Generated Background Reno Traffic

Table 2 gives the results for Reno and two versions of Vegas—Vegas-1,3 uses  $\alpha = 1$  and  $\beta = 3$ , and Vegas-2,4 uses  $\alpha = 2$  and  $\beta = 4$ . We varied these two thresholds to study the sensitivity of our algorithm. The numbers shown are averages from 57 runs, obtained by using different seeds for *tcplib*, and by using 10, 15 and 20 buffers in the routers.

The table shows the throughput rate for each of the distinguished connections using the three protocols, along with their ratio to Reno’s throughput. It also gives the number of kilobytes retransmitted, the ratio of retransmits to Reno’s, and the average number of coarse-grained timeouts per transfer. For example, Vegas-1,3 had 53% better throughput than Reno, with only 49% of the losses. Also note that there is little difference between Vegas-1,3 and Vegas-2,4.

These simulations tell us the expected improvement of Vegas over Reno: more than 50% improvement on throughput, and only half the losses. The results from the one-on-one experiments indicate that the gains of Vegas are not made at the expense of Reno; this belief is further supported by the fact that the background traffic’s throughput increases by 20% when Reno is competing for resources with Vegas, as compared to when Reno is competing with itself.

We also ran these tests with the background traffic using Vegas rather than Reno. This simulates the situation where the whole world uses Vegas. The throughput and the kilobytes retransmitted by the 1MByte transfers didn’t change significantly—less than 4%—but this time the throughput of the background traffic was unaffected by the type of protocol running the 1MB transfer. These results are summarized in Table 3.

traffic over	1MB Transfer	
	Reno	Vegas
Reno (KB/s)	68	82
Vegas (KB/s)	84	85

Table 3: Throughput of Background Traffic When Competing with a 1MB Transfer

### 4.3 Other Experiments

We tried many variations of the previous experiments. On the whole, the results were similar, except for when we changed TCP’s send-buffer size. Below we summarize these experiments and their results.

- *One-on-one tests with traffic in the background.* The results were similar. Again, Reno did better when running against Vegas than against itself, but this time, its losses increased by only 6% (versus 43%) in the Reno/Vegas case.
- *Two-way background traffic.* There have been reports of change in TCP's behavior when the background traffic is two-way rather than one-way [13]. Thus, we modified the experiment in Section 4.2 by adding *tcplib* traffic from Host3b to Host3a. The throughput ratio stayed the same, but the loss ratio was much better: 0.29. Reno resent more data and Vegas remained about the same. The fact that there wasn't much change is probably due to the fact that *tcplib* already creates some 2-way traffic—TELNET connections send one byte and get one or more bytes back, and FTP connections send and get control packets before doing a transfer.
- *Different TCP send-buffer sizes.* For all the experiments reported so far, we ran TCP with a 50KB send-buffer. For this experiment, we tried send-buffer sizes between 50KB and 5KB. Vegas' throughput and losses stayed unchanged between 50KB and 20KB; from that point on, as the buffer decreased, so did the throughput. This was due to the protocol not being able to keep the pipe full.

On the other hand, Reno's throughput initially *increased* as the buffers got smaller, and then it decreased. It always remained under the throughput measured for Vegas. We have previously seen this type of behavior while running Reno on the Internet. If we look back at Figure 5, we see that as Reno increases its congestion window, it uses more and more buffers in the router until it loses packets by overrunning the queue. If we limit the congestion window by reducing the size of the send-buffer, we may prevent it from overrunning the router's queue.

If we knew the connection's available bandwidth, then we could set Reno's send-buffer size appropriately. The problem, of course, is that we don't have this knowledge at the beginning of the connection and the connection's available bandwidth may change with time.

## 5 Preliminary Results on the Internet

This section measurements of TCP over the Internet. Because it is simple to move a protocol between the simulator and the "real world", the numbers reported in this section are for exactly the same code as in the previous section. Our results must be considered preliminary because they were limited to transfers between the University of Arizona (UA) and the National Institutes of Health (NIH). The connection consists of 17 hops, and passes through Denver, St. Louis, Chicago, Cleveland, New York and Washington DC.

The results are derived from a set of runs over a seven day period from January 23-29. Each run consists of a set of seven transfers from UA to NIH—Reno sends 1MB, 512KB, and 128KB, Vegas-1,3 sends 1MB, 512KB, and 128KB, and Vegas-2,4 sends 1MB. We inserted a 45 second delay between each transfer in a run to give the network a chance to settle down, a run started approximately once every hour, and we shuffled the order of the transfers within each run.

Table 4 shows the results for the 1MB transfers. Depending on the congestion avoidance thresholds, it shows between 37% and 42% improvement over Reno's throughput with only 51% to 61% of the retransmissions. Note that Vegas out-performed Reno on 92% of the transfers, and across all levels of congestion; i.e., during both the middle of the night and during periods of high load.



	Reno	Vegas-1,3	Vegas-2,4
Throughput (KB/s)	53.00	72.50	75.30
Throughput Ratio	1.00	1.37	1.42
Retransmissions (KB)	47.80	24.50	29.30
Retransmit Ratio	1.00	0.51	0.61
Coarse Timeouts	3.30	0.80	0.90

Table 4: 1MByte transfer over the Internet

	1024KB		512KB		128KB	
	Reno	Vegas	Reno	Vegas	Reno	Vegas
Throughput (KB/s)	53.00	72.50	52.00	72.00	31.10	53.10
Throughput Ratio	1.00	1.37	1.00	1.38	1.00	1.71
Retransmissions (KB)	47.80	24.50	27.90	10.50	22.90	4.00
Retransmit Ratio	1.00	0.51	1.00	0.38	1.00	0.17
Coarse Timeouts	3.30	0.80	1.70	0.20	1.10	0.20

Table 5: Effects of transfer size over the Internet

Because we were concerned that Vegas' throughput improvement depended on large transfer sizes, we also varied the size of the transfer. Table 5 shows the effect of transfer size on both throughput and retransmissions for Reno and Vegas-1,3. The first thing to see is that Vegas does better relative to Reno as the transfer size decreases; in terms of throughput, we see an increase from 37% to 71%. The results are similar for retransmissions, as the relative number of Vegas retransmissions goes from 51% of Reno's to 17% of Reno's.

The next thing to notice is how the number of kilobytes retransmitted by Reno starts to flatten out as the transfer size decreases. When we decreased the transfer size by half, from 1MB to 512KB, we see only a 42% decrease in the number of kilobytes retransmitted. When we further decrease the transfer size to one-fourth its previous value, from 512KB to 128KB, the number of kilobytes retransmitted only decreases by 18%. This indicates that we are approaching the average number of kilobytes retransmitted due to Reno's slow-start losses. From these results, we conclude that there are around 20KBs retransmitted during slow-start, for the conditions of our experiment.

On the other hand, the number of kilobytes retransmitted by Vegas decreases almost linearly with respect to the transfer size. This indicates that Vegas eliminates nearly all losses during slow-start due to its modified slow-start with congestion avoidance.

## 6 Concluding Remarks

We have introduced several techniques for improving TCP, including a novel approach to congestion avoidance that tries to control the number of extra buffers the connection occupies in the network. Experiments on both the Internet and using a simulator show that Vegas achieves 40 to 70% better throughput, with one-half to one-fifth as many bytes being retransmitted, as compared to the implementation of TCP in the

Reno distribution of BSD Unix. In many respects, this work is still preliminary, with the following issues requiring more attention.

First, we believe that adding congestion detection to slow-start is important, and will become more important as network bandwidth increases. Vegas offers a beginning, but there is a problem that still needs to be addressed. During slow-start, the TCP sends segments at twice the rate supported by the connection; i.e., two segments are sent for every ACK received. If there aren't enough buffers in the bottleneck router, Vegas' slow-start with congestion detection may lose segments before getting any feedback that tells it to slow down. We have several solutions in mind that we plan to implement in the near future. One is to use rate control during slow-start, using a rate defined by the current window size and the BaseRTT. Another is to slow down as we reach the bandwidth available to the connection (Vegas has enough information to figure this out).

Second, Vegas' congestion detection algorithm depends on an accurate value for *BaseRTT*. If our estimate for the *BaseRTT* is too small, then the protocol's throughput will stay below the available bandwidth; if it is too large, then it will overrun the connection. Our experience is that the the protocol does well with its current choice of *BaseRTT*. However, we plan to study this more carefully.

Third, the question of whether or not Vegas' linear increase/decrease mechanism is fair must still be answered. On the one hand, we have no reason to believe that our approach is unfair, and we plan to do further analysis to attempt to show this. On the other hand, we plan to experiment with an additive increase and multiplicative decrease algorithm [1].

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