Internet Engineering Task Force
INTERNET-DRAFT
draft-floyd-tcp-highspeed-03.ps
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ICSI
29 June 2003
Expires: December 2003

HighSpeed TCP for Large Congestion Windows

Status of this Document

Abstract

This document proposes HighSpeed TCP, a modification to TCP's congestion control mechanism for use with TCP connections with large congestion windows. The congestion control mechanisms of the current Standard TCP constrains the congestion windows that can be achieved by TCP in realistic environments. For example, for a Standard TCP connection with 1500-byte packets and a 100 ms round-trip time, achieving a steady-state throughput of 10 Gbps would require an average congestion window of 83,333 segments, and a packet drop rate of at most one congestion event every 5,000,000,000 packets (or equivalently, at most one congestion event every 1 2/3 hours). This is widely acknowledged as an unrealistic constraint. To address this limitation of TCP, this document proposes HighSpeed TCP, and solicits experimentation and feedback from the wider community.

TO BE DELETED BY THE RFC EDITOR UPON PUBLICATION:

Changes from draft-floyd-tcp-highspeed-02.txt:

* Added a section on "Deployment issues."

* Added a short section on "Implementation issues."

* Added a section on "Limiting burstiness on short time scales".

* Added to the discussion on convergence times.

* Clarified that "log" is "log base 10".

* Clarified that W = Low_window and W_1 = High_window, in the equation for b(w).

Changes from draft-floyd-tcp-highspeed-01.txt:

* Added a section on "Tradeoffs for Choosing Congestion Control Parameters".

* Added mention of Scalable TCP from Tom Kelly.

Changes from draft-floyd-tcp-highspeed-00.txt:

* Added a discussion on related work about changing the PMTU.

* Added a discussion of an alternate, linear response function.

* Added a discussion of the TCP window scale option.

* Added a discussion of HighSpeed TCP as roughly emulating the congestion control response of N parallel TCP connections.

* Added a discussion of the time to converge to fairness.

* Expanded the Introduction.

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1. Introduction.

This document proposes HighSpeed TCP, a modification to TCP's congestion control mechanism for use with TCP connections with large congestion windows. In a steady-state environment, with a packet loss rate p, the current Standard TCP's average congestion window is roughly 1.2/sqrt(p) segments. This places a serious constraint on the congestion windows that can be achieved by TCP in realistic environments. For example, for a Standard TCP connection with 1500-byte packets and a 100 ms round-trip time, achieving a steady-state throughput of 10 Gbps would require an average congestion window of 83,333 segments, and a packet drop rate of at most one congestion event every 5,000,000,000 packets (or equivalently, at most one congestion event every 1 2/3 hours). The average packet drop rate of at most 2*10^(-10) needed for full link utilization in this environment corresponds to a bit error rate of at most 2*10^(-14), and this is an unrealistic requirement for current networks.

To address this fundamental limitation of TCP and of the TCP response function (the function mapping the steady-state packet drop rate to TCP's average sending rate in packets per round-trip time), this document describes a modified TCP response function for regimes with higher congestion windows. This document also solicits experimentation and feedback on HighSpeed TCP from the wider community.

Because HighSpeed TCP's modified response function would only take effect with higher congestion windows, HighSpeed TCP does not modify TCP behavior in environments with mild to heavy congestion, and therefore does not introduce any new dangers of congestion collapse. However, if relative fairness between HighSpeed TCP connections is to be preserved, then in our view any modification to the TCP response function should be addressed in the IETF, rather than made as ad hoc decisions by individual implementors or TCP senders. Modifications to the TCP response function would also have implications for transport protocols that use TFRC and other forms of equation-based congestion control, as these congestion control mechanisms directly use the TCP response function [RFC3448].

This proposal for HighSpeed TCP focuses specifically on a proposed change to the TCP response function, and its implications for TCP. This document does not address what we view as a separate fundamental issue, of the mechanisms required to enable best-effort connections to *start* with large initial windows. In our view, while HighSpeed TCP proposes a somewhat fundamental change to the TCP response function, at the same time it is a relatively simple change to implement in a single TCP sender, and presents no dangers in terms of congestion collapse. In contrast, in our view, the problem of enabling connections to *start* with large initial windows is inherently more risky and structurally more difficult, requiring some form of explicit feedback from all of the routers along the path. This is another reason why we would propose addressing the problem of starting with large initial windows separately, and on a separate timetable, from the problem of modifying the TCP response function.

2. The Problem Description.

This section describes the number of round-trip times between congestion events required for a Standard TCP flow to achieve an average throughput of B bps, given packets of D bytes and a round-trip time of R seconds. A congestion event refers to a window of data with one or more dropped or ECN-marked packets (where ECN stands for Explicit Congestion Notification).

From Appendix A, achieving an average TCP throughput of B bps requires a loss event at most every BR/(12D) round-trip times. This is illustrated in Table 1, for R = 0.1 seconds and D = 1500 bytes. The table also gives the average congestion window W of BR/(8D), and the steady-state packet drop rate P of 1.5/W^2.

TCP Throughput (Mbps)	RTTs Between Losses	W	P
1	5.5	8.3	0.02
10	55.5	83.3	0.0002
100	555.5	833.3	0.000002
1000	5555.5	8333.3	0.00000002
10000	55555.5	83333.3	0.0000000002

Table 1: RTTs Between Congestion Events for Standard TCP, for 1500-Byte Packets and a Round-Trip Time of 0.1 Seconds.

This document proposes HighSpeed TCP, a minimal modification to TCP's increase and decrease parameters, for TCP connections with larger congestion windows, to allow TCP to achieve high throughput with more realistic requirements for the steady-state packet drop rate. Equivalently, HighSpeed TCP has more realistic requirements for the number of round-trip times between loss events.

3. Design Guidelines.

Our proposal for HighSpeed TCP is motivated by the following requirements:

* Achieve high per-connection throughput without requiring unrealistically low packet loss rates.

* Reach high throughput reasonably quickly when in slow-start.

* Reach high throughput without overly long delays when recovering from multiple retransmit timeouts, or when ramping-up from a period with small congestion windows.

* No additional feedback or support required from routers:

For example, the goal is for acceptable performance in both ECN-capable and non-ECN-capable environments, and with Drop-Tail as well as with Active Queue Management such as RED in the routers.

* No additional feedback required from TCP receivers.

* TCP-compatible performance in environments with moderate or high congestion:

Equivalently, the requirement is that there be no additional load on the network (in terms of increased packet drop rates) in environments with moderate or high congestion.

* Performance at least as good as Standard TCP in environments with moderate or high congestion.

* Acceptable transient performance, in terms of increases in the congestion window in one round-trip time, responses to severe congestion, and convergence times to fairness.

Currently, users wishing to achieve throughputs of 1 Gbps or more typically open up multiple TCP connections in parallel, or use MulTCP [CO98,GRK99], which behaves roughly like the aggregate of N virtual TCP connections. While this approach suffices for the occasional user on well-provisioned links, it leaves the parameter N to be determined by the user, and results in more aggressive performance and higher steady-state packet drop rates if used in environments with periods of moderate or high congestion. We believe that a new approach is needed that offers more flexibility, more effectively scales to a wide range of available bandwidths, and competes more fairly with Standard TCP in congested environments.

4. Non-Goals.

The following are explicitly *not* goals of our work:

* Non-goal: TCP-compatible performance in environments with very low packet drop rates.

We note that our proposal does not require, or deliver, TCP-compatible performance in environments with very low packet drop rates, e.g., with packet loss rates of 10^-5 or 10^-6. As we discuss later in this document, we assume that Standard TCP is unable to make effective use of the available bandwidth in environments with loss rates of 10^-6 in any case, so that it is acceptable and appropriate for HighSpeed TCP to perform more aggressively than Standard TCP is such an environment.

* Non-goal: Ramping-up more quickly than allowed by slow-start.

It is our belief that ramping-up more quickly than allowed by slow-start would necessitate more explicit feedback from routers along the path. The proposal for HighSpeed TCP is focused on changes to TCP that could be effectively deployed in the current Internet environment.

* Non-goal: Avoiding oscillations in environments with only one-way, long-lived flows all with the same round-trip times.

While we agree that attention to oscillatory behavior is useful, avoiding oscillations in aggregate throughput has not been our primary consideration, particularly for simplified environments limited to one-way, long-lived flows all with the same, large round-trip times. Our assessment is that some oscillatory behavior in these extreme environments is an acceptable price to pay for the other benefits of HighSpeed TCP.

5. Modifying the TCP Response Function.

The TCP response function, w = 1.2/sqrt(p), gives TCP's average congestion window w in MSS-sized segments, as a function of the steady-state packet drop rate p [FF98]. This TCP response function is a direct consequence of TCP's Additive Increase Multiplicative Decrease (AIMD) mechanisms of increasing the congestion window by roughly one segment per round-trip time in the absence of congestion, and halving the congestion window in response to a round-trip time with a congestion event. This response function for Standard TCP is reflected in the table below. In this proposal we restrict our attention to TCP performance in environments with packet loss rates of at most 10^-2, and so we can ignore the more complex response functions that are required to model TCP performance in more congested environments with retransmit timeouts. From Appendix A, an average congestion window of W corresponds to an average of 2/3 W round-trip times between loss events for Standard TCP (with the congestion window varying from 2/3 W to 4/3 W).

Packet Drop Rate P	Congestion Window W	RTTs Between Losses
10^-2	12	8
10^-3	38	25
10^-4	120	80
10^-5	379	252
10^-6	1200	800
10^-7	3795	2530
10^-8	12000	8000
10^-9	37948	25298
10^-10	120000	80000

Table 2: TCP Response Function for Standard TCP. The average congestion window W in MSS-sized segments is given as a function of the packet drop rate P.

To specify a modified response function for HighSpeed TCP, we use three parameters, Low_Window, High_Window, and High_P. To ensure TCP compatibility, the HighSpeed response function uses the same response function as Standard TCP when the current congestion window is at most Low_Window, and uses the HighSpeed response function when the current congestion window is greater than Low_Window. In this document we set Low_Window to 38 MSS-sized segments, corresponding to a packet drop rate of 10^-3 for TCP.

To specify the upper end of the HighSpeed response function, we specify the packet drop rate needed in the HighSpeed response function to achieve an average congestion window of 83000 segments. This is roughly the window needed to sustain 10 Gbps throughput, for a TCP connection with the default packet size and round-trip time used earlier in this document. For High_Window set to 83000, we specify High_P of 10^-7; that is, with HighSpeed TCP a packet drop rate of 10^-7 allows the HighSpeed TCP connection to achieve an average congestion window of 83000 segments. We believe that this loss rate sets an achievable target for high-speed environments, while still allowing acceptable fairness for the HighSpeed response function when competing with Standard TCP in environments with packet drop rates of 10^-4 or 10^5.

For simplicity, for the HighSpeed response function we maintain the property that the response function gives a straight line on a log-log scale (as does the response function for Standard TCP, for low to moderate congestion). This results in the following response function, for values of the average congestion window W greater than Low_Window:

W = (p/Low_P)^S Low_Window,

for Low_P the packet drop rate corresponding to Low_Window, and for S as following constant [FRS02]:

S = (log High_Window - log Low_Window)/(log High_P - log Low_P).

(In this paper, "log x" refers to the log base 10.) For example, for Low_Window set to 38, we have Low_P of 10^-3 (for compatibility with Standard TCP). Thus, for High_Window set to 83000 and High_P set to 10^-7, we get the following response function:

W = 0.12/p^0.835. (1)

This HighSpeed response function is illustrated in Table 3 below. For HighSpeed TCP, the number of round-trip times between losses, 1/(pW), equals 12.7 W^0.2, for W > 38 segments.

Packet Drop Rate P	Congestion Window W	RTTs Between Losses
10^-2	12	8
10^-3	38	25
10^-4	263	38
10^-5	1795	57
10^-6	12279	83
10^-7	83981	123
10^-8	574356	180
10^-9	3928088	264
10^-10	26864653	388

Table 3: TCP Response Function for HighSpeed TCP. The average congestion window W in MSS-sized segments is given as a function of the packet drop rate P.

We believe that the problem of backward compatibility with Standard TCP requires a response function that is quite close to that of Standard TCP for loss rates of 10^-1, 10^-2, or 10^-3. We believe, however, that such stringent TCP-compatibility is not required for smaller loss rates, and that an appropriate response function is one that gives a plausible packet drop rate for a connection throughput of 10 Gbps. This also gives a slowly increasing number of round-trip times between loss events as a function of a decreasing packet drop rate.

Another way to look at the HighSpeed response function is to consider that HighSpeed TCP is roughly emulating the congestion control response of N parallel TCP connections, where N is initially one, and where N increases as a function of the HighSpeed TCP's congestion window. Thus for the HighSpeed response function in Equation (1) above, the response function can be viewed as equivalent to that of N(W) parallel TCP connections, where N(W) varies as a function of the congestion window W. Recall that for a single standard TCP connection, the average congestion window equals 1.2/sqrt(p). For N parallel TCP connections, the aggregate congestion window for the N connections equals N*1.2/sqrt(p). From the HighSpeed response function in Equation (1) and the relationship above, we can derive the following:

N(W) = 0.23*W^(0.4)

for N(W) the number of parallel TCP connections emulated by the HighSpeed TCP response function, and for N(W) >= 1. This is shown in Table 4 below.

Congestion Window W	Number N(W) of Parallel TCPs
1	1
10	1
100	1.4
1,000	3.6
10,000	9.2
100,000	23.0

Table 4: Number N(W) of parallel TCP connections roughly emulated by the HighSpeed TCP response function.

We do not in this document attempt to seriously evaluate the HighSpeed response function for congestion windows greater than 100,000 packets. We believe that we will learn more about the requirements for sustaining the throughput of best-effort connections in that range as we gain more experience with HighSpeed TCP with congestion windows of thousands and tens of thousands of packets. There also might be limitations to the per-connection throughput that can be realistically achieved for best-effort traffic, in terms of congestion window of hundreds of thousands of packets or more, in the absence of additional support or feedback from the routers along the path.

6. Fairness Implications of the HighSpeed Response Function.

The Standard and Highspeed Response Functions can be used directly to infer the relative fairness between flows using the two response functions. For example, given a packet drop rate P, assume that Standard TCP has an average congestion window of W_Standard, and HighSpeed TCP has a higher average congestion window of W_HighSpeed. In this case, a single HighSpeed TCP connection is receiving W_HighSpeed/W_Standard times the throughput of a single Standard TCP connection competing in the same environment.

This relative fairness is illustrated below in Table 5, for the parameters used for the Highspeed response function in the section above. The second column gives the relative fairness, for the steady-state packet drop rate specified in the first column. To help calibrate, the third column gives the aggregate average congestion window for the two TCP connections, and the fourth column gives the bandwidth that would be needed by the two connections to achieve that aggregate window and packet drop rate, given 100 ms round-trip times and 1500-byte packets.

Packet Drop Rate P	Fairness	Aggregate Window	Bandwidth
10^-2	1.0	24	2.8 Mbps
10^-3	1.0	76	9.1 Mbps
10^-4	2.2	383	45.9 Mbps
10^-5	4.7	2174	260.8 Mbps
10^-6	10.2	13479	1.6 Gbps
10^-7	22.1	87776	10.5 Gbps

Table 5: Relative Fairness between the HighSpeed and Standard Response Functions.

Thus, for packet drop rates of 10^-4, a flow with the HighSpeed response function can expect to receive 2.2 times the throughput of a flow using the Standard response function, given the same round-trip times and packet sizes. With packet drop rates of 10^-6 (or 10^-7), the unfairness is more severe, and we have entered the regime where a Standard TCP connection requires at most one congestion event every 800 (or 2530) round-trip times in order to make use of the available bandwidth. Our judgement would be that there are not a lot of TCP connections effectively operating in this regime today, with congestion windows of thousands of packets, and that therefore the benefits of the HighSpeed response function would outweigh the unfairness that would be experienced by Standard TCP in this regime. However, one purpose of this document is to solicit feedback on this issue. The parameter Low_Window determines directly the point of divergence between the Standard and HighSpeed Response Functions.

The third column of Table 5, the Aggregate Window, gives the aggregate congestion window of the two competing TCP connections, with HighSpeed and Standard TCP, given the packet drop rate specified in the first column. From Table 5, a HighSpeed TCP connection would receive ten times the bandwidth of a Standard TCP in an environment with a packet drop rate of 10^-6. This would occur when the two flows sharing a single pipe achieved an aggregate window of 13479 packets. Given a round-trip time of 100 ms and a packet size of 1500 bytes, this would occur with an available bandwidth for the two competing flows of 1.6 Gbps.

Next we consider the time that it takes a standard or HighSpeed TCP flow to converge to fairness against a pre-existing HighSpeed TCP flow. The worst case for convergence to fairness occurs when a new flow is starting up, competing against a high-bandwidth existing flow, and the new flow suffers a packet drop and exits slow-start while its window is still small. In the worst case, consider that the new flow has entered the congestion avoidance phase while its window is only one packet. A standard TCP flow in congestion avoidance increases its window by at most one packet per round-trip time, and after N round-trip times has only achieved a window of N packets (when starting with a window of 1 in the first round-trip time). In contrast, a HighSpeed TCP flows increases much faster than a standard TCP flow while in the congestion avoidance phase, and we can expect its convergence to fairness to be much better. This is shown in Table 6 below. The script used to generate this table is given in Appendix C.

RTT	HS_Window	Standard_TCP_Window
100	131	100
200	475	200
300	1131	300
400	2160	400
500	3601	500
600	5477	600
700	7799	700
800	10567	800
900	13774	900
1000	17409	1000
1100	21455	1100
1200	25893	1200
1300	30701	1300
1400	35856	1400
1500	41336	1500
1600	47115	1600
1700	53170	1700
1800	59477	1800
1900	66013	1900
2000	72754	2000

Table 6: For a HighSpeed and a Standard TCP connection, the congestion window during congestion avoidance phase (starting with a congestion window of 1 packet during RTT 1.

The classic paper on relative fairness is from Chiu and Jain [CJ89]. This paper shows that AIMD (Additive Increase Multiplicative Decrease) converges to fairness in an environment with synchronized congestion events. From [CJ89], it is easy to see that MIMD and AIAD do not converge to fairness in this environment. However, the results of [CJ89] do not apply to an asynchronous environment such as that of the current Internet, where the frequency of congestion feedback can be different for different flows. For example, it has been shown that MIMD converges to fair states in a model with proportional instead of synchronous feedback in terms of packet drops [GV02]. Thus, we are not concerned about abandoning a strict model of AIMD for HighSpeed TCP.

7. Translating the HighSpeed Response Function into Congestion Control Parameters.

For equation-based congestion control such as TFRC, the HighSpeed Response Function above could be used directly by the TFRC congestion control mechanism. However, for TCP the HighSpeed response function has to be translated into additive increase and multiplicative decrease parameters. The HighSpeed response function cannot be achieved by TCP with an additive increase of one segment per round-trip time and a multiplicative decrease of halving the current congestion window; HighSpeed TCP will have to modify either the increase or the decrease parameter, or both. We have concluded that HighSpeed TCP is most likely to achieve an acceptable compromise between moderate increases and timely decreases by modifying both the increase and the decrease parameter.

That is, for HighSpeed TCP let the congestion window increase by a(w) segments per round-trip time in the absence of congestion, and let the congestion window decrease to w(1-b(w)) segments in response to a round-trip time with one or more loss events. Thus, in response to a single acknowledgement HighSpeed TCP increases its congestion window in segments as follows:

$$w \leftarrow w + a(w)/w.$$

In response to a congestion event, HighSpeed TCP decreases as follows:

$$w \leftarrow (1-b(w))w.$$

For Standard TCP, a(w) = 1 and b(w) = 1/2, regardless of the value of w. HighSpeed TCP uses the same values of a(w) and b(w) for w <= Low_Window. This section specifies a(w) and b(w) for HighSpeed TCP for larger values of w.

For w = High_Window, we have specified a loss rate of High_P. From [FRS02], or from elementary calculations, this requires the following relationship between a(w) and b(w) for w = High_Window:

$$a(w) = High\_Window^2 * High\_P * 2 * b(w)/(2-b(w). \quad (2)$$

We use the parameter High_Decrease to specify the decrease parameter b(w) for w = High_Window, and use Equation (2) to derive the increase parameter a(w) for w = High_Window. Along with High_P = 10^-7 and High_Window = 83000, for example, we specify High_Decrease = 0.1, specifying that b(83000) = 0.1, giving a decrease of 10% after a congestion event. Equation (2) then gives a(83000) = 72, for an increase of 72 segments, or just under 0.1%, within a round-trip time, for w = 83000.

This moderate decrease strikes us as acceptable, particularly when coupled with the role of TCP's ACK-clocking in limiting the sending rate in response to more severe congestion [BBFS01]. A more severe decrease would require a more aggressive increase in the congestion window for a round-trip time without congestion. In particular, a decrease factor High_Decrease of 0.5, as in Standard TCP, would require an increase of 459 segments per round-trip time when w = 83000.

Given decrease parameters of b(w) = 1/2 for w = Low_Window, and b(w) = High_Decrease for w = High_Window, we are left to specify the value of b(w) for other values of w > Low_Window. From [FRS02], we let b(w) vary linearly as the log of w, as follows:

$$b(w) = (High\_Decrease - 0.5) (\log(w)-\log(W)) / (\log(W\_1)-\log(W)) + 0.5,$$

for W = Low_window and W_1 = High_window. The increase parameter a(w) can then be computed as follows:

$$a(w) = w^2 * p(w) * 2 * b(w)/(2-b(w)),$$

for p(w) the packet drop rate for congestion window w. From inverting Equation (1), we get p(w) as follows:

$p(w) = 0.078/w^{1.2}$.

We assume that experimental implementations of HighSpeed TCP for further investigation will use a pre-computed look-up table for finding a(w) and b(w). For example, the implementation from Tom Dunigan adjusts the a(w) and b(w) parameters every 0.1 seconds. In the appendix we give such a table for our default values of Low_Window = 38, High_Window = 83,000, High_P = 10^-7, and High_Decrease = 0.1. These are also the default values in the NS simulator; example simulations in NS can be run with the command "./test-all-tcpHighspeed" in the directory tcl/test.

8. An alternate, linear response functions.

In this section we explore an alternate, linear response function for HighSpeed TCP that has been proposed by a number of other people, in particular by Glenn Vinnicombe and Tom Kelly. Similarly, it has been suggested by others that a less "ad-hoc" guideline for a response function for HighSpeed TCP would be to specify a constant value for the number of round-trip times between congestion events.

Assume that we keep the value of Low_Window as 38 MSS-sized segments, indicating when the HighSpeed response function diverges from the current TCP response function, but that we modify the High_Window and High_P parameters that specify the upper range of the HighSpeed response function. In particular, consider the response function given by High_Window = 380,000 and High_P = 10^-7, with Low_Window = 38 and Low_P = 10^-3 as before.

Using the equations in Section 5, this would give the following Linear response function, for w > Low_Window:

$W = 0.038/p$.

This Linear HighSpeed response function is illustrated in Table 7 below. For HighSpeed TCP, the number of round-trip times between losses, 1/(pW), equals 1/0.38, or equivalently, 26, for W > 38 segments.

Packet Drop Rate P	Congestion Window W	RTTs Between Losses
10^-2	12	8
10^-3	38	26
10^-4	380	26
10^-5	3800	26
10^-6	38000	26
10^-7	380000	26
10^-8	3800000	26
10^-9	38000000	26
10^-10	380000000	26

Table 7: An Alternate, Linear TCP Response Function for HighSpeed TCP. The average congestion window W in MSS-sized segments is given as a function of the packet drop rate P.

Given a constant decrease b(w) of 1/2, this would give an increase a(w) of w/Low_Window, or equivalently, a constant increase of 1/Low_Window packets per acknowledgement, for w > Low_Window. Another possibility is Scalable TCP [K03], which uses a fixed decrease b(w) of 1/8 and a fixed increase per acknowledgement of 0.01. This gives an increase a(w) per window of 0.005 w, for a TCP with delayed acknowledgements, for pure MIMD.

The relative fairness between the alternate Linear response function and the standard TCP response function is illustrated below in Table 8.

Packet Drop Rate P	Fairness	Aggregate Window	Bandwidth
10^-2	1.0	24	2.8 Mbps
10^-3	1.0	76	9.1 Mbps
10^-4	3.2	500	60.0 Mbps
10^-5	15.1	4179	501.4 Mbps
10^-6	31.6	39200	4.7 Gbps
10^-7	100.1	383795	46.0 Gbps

Table 8: Relative Fairness between the Linear HighSpeed and Standard Response Functions.

One attraction of the linear response function is that it is scale-invariant, with a fixed increase in the congestion window per acknowledgement, and a fixed number of round-trip times between loss events. My own assumption would be that having a fixed length for the congestion epoch in round-trip times, regardless of the packet drop rate, would be a poor fit for an imprecise and imperfect world with routers with a range of queue management mechanisms, such as the Drop-Tail queue management that is common today. For example, a response function with a fixed length for the congestion epoch in round-trip times might give less clearly-differentiated feedback in an environment with steady-state background losses at fixed intervals for all flows (as might occur with a wireless link with occasional short error bursts, giving losses for all flows every N seconds regardless of their sending rate).

While it is not a goal to have perfect fairness in an environment with synchronized losses, it would be good to have moderately acceptable performance in this regime. This goal might argue against a response function with a constant number of round-trip times between congestion events. However, this is a question that could clearly use additional research and investigation. In addition, flows with different round-trip times would have different time durations for congestion epochs even in the model with a linear response function.

The third column of Table 8, the Aggregate Window, gives the aggregate congestion window of two competing TCP connections, one with Linear HighSpeed TCP and one with Standard TCP, given the packet drop rate specified in the first column. From Table 8, a Linear HighSpeed TCP connection would receive fifteen times the bandwidth of a Standard TCP in an environment with a packet drop rate of 10^-5. This would occur when the two flows sharing a single pipe achieved an aggregate window of 4179 packets. Given a round-trip time of 100 ms and a packet size of 1500 bytes, this would occur with an available bandwidth for the two competing flows of 501 Mbps. Thus, because the Linear HighSpeed TCP is more aggressive than the HighSpeed TCP proposed above, it also is less fair when competing with Standard TCP in a high-bandwidth environment.

9. Tradeoffs for Choosing Congestion Control Parameters.

A range of metrics can be used for evaluating choices for congestion control parameters for HighSpeed TCP. My assumption in this section is that for a response function of the form w = c/p^d, for constant c and exponent d, the only response functions that would be considered are response functions with 1/2 <= d <= 1. The two ends of this spectrum are represented by current TCP, with d = 1/2, and by the linear response function described in Section 8 above, with d = 1. HighSpeed TCP lies somewhere in the middle of the spectrum, with d = 0.835.

Response functions with exponents less than 1/2 can be eliminated from consideration because they would be even worse than standard TCP in accomodating connections with high congestion windows.

9.1. The Number of Round-Trip Times between Loss Events.

Response functions with exponents greater than 1 can be eliminated from consideration because for these response functions, the number of round-trip times between loss events decreases as congestion decreases. For a response function of w = c/p^d, with one loss event or congestion event every 1/p packets, the number of round-trip times between loss events is w^((1/d)-1)/c^(1/d). Thus, for standard TCP the number of round-trip times between loss events is linear in w. In contrast, one attraction of the linear response function, as described in Section 8 above, is that it is scale-invariant, in terms of a fixed increase in the congestion window per acknowledgement, and a fixed number of round-trip times between loss events.

However, for a response function with d > 1, the number of round-trip times between loss events would be proportional to w^((1/d)-1), for a negative exponent ((1/d)-1), setting smaller as w increases. This would seem undesirable.

9.2. The Number of Packet Drops per Loss Event, with Drop-Tail.

A TCP connection increases its sending rate by a(w) packets per round-trip time, and in a Drop-Tail environment, this is likely to result in a(w) dropped packets during a single loss event. One attraction of standard TCP is that it has a fixed increase per round-trip time of one packet, minimizing the number of packets that would be dropped in a Drop-Tail environment. For an environment with some form of Active Queue Management, and in particular for an environment that uses ECN, the number of packets dropped in a single congestion event would not be a problem. However, even in these environments, larger increases in the sending rate per round-trip time result in larger stresses on the ability of the queues in the router to absorb the fluctuations.

HighSpeed TCP plays a middle ground between the metrics of a moderate number of round-trip times between loss events, and a moderate increase in the sending rate per round-trip time. As shown in Appendix B, for a congestion window of 83,000 packets, HighSpeed TCP increases its sending rate by 70 packets per round-trip time, resulting in at most 70 packet drops when the buffer overflows in a Drop-Tail environment. This increased aggressiveness is the price paid by HighSpeed TCP for its increased scalability. A large number of packets dropped per congestion event could result in synchronized drops from multiple flows, with a possible loss of throughput as a result.

Scalable TCP has an increase a(w) of 0.005 w packets per round-trip time. For a congestion window of 83,000 packets, this gives an increase of 415 packets per round-trip time, resulting in roughly 415 packet drops per congestion event in a Drop-Tail environment.

Thus, HighSpeed TCP and its variants place increased demands on queue management in routers, relative to Standard TCP. (This is rather similar to the increased demands on queue management that would result from using N parallel TCP connections instead of a single Standard TCP connection.)

10. Related Issues

10.1. Slow-Start.

An companion internet-draft on "Limited Slow-Start for TCP with Large Congestion Windows" [F02b] proposes a modification to TCP's slow-start procedure that can significantly improve the performance of TCP connections slow-starting up to large congestion windows. For TCP connections that are able to use congestion windows of thousands (or tens of thousands) of MSS-sized segments (for MSS the sender's MAXIMUM SEGMENT SIZE), the current slow-start procedure can result in increasing the congestion window by thousands of segments in a single round-trip time. Such an increase can easily result in thousands of packets being dropped in one round-trip time. This is often counter-productive for the TCP flow itself, and is also hard on the rest of the traffic sharing the congested link.

[F02b] proposes Limited Slow-Start, limiting the number of segments by which the congestion window is increased for one window of data during slow-start, in order to improve performance for TCP connections with large congestion windows. We have separated out Limited Slow-Start to a separate draft because it can be used both with Standard or with HighSpeed TCP.

Limited Slow-Start is illustrated in the NS simulator, for snapshots after May 1, 2002, in the tests "./test-all-tcpHighspeed tcp1A" and "./test-all-tcpHighspeed tcpHighspeed1" in the subdirectory "tcl/lib".

In order for best-effort flows to safely start-up faster than slow-start, e.g., in future high-bandwidth networks, we believe that it would be necessary for the flow to have explicit feedback from the routers along the path. There are a number of proposals for this, ranging from a minimal proposal for an IP option that allows TCP SYN packets to collect information from routers along the path about the allowed initial sending rate [J02], to proposals with more power that require more fine-tuned and continuous feedback from routers. These proposals all are somewhat longer-term proposals than the HighSpeed TCP proposal in this document, requiring longer lead times and more coordination for deployment, and will be discussed in later documents.

10.2. Limiting burstiness on short time scales.

Because the congestion window achieved by a HighSpeed TCP connection could be quite large, there is a possibility for the sender to send a large burst of packets in response to a single acknowledgement. This could happen, for example, when there is congestion or reordering on the reverse path, and the sender receives an acknowledgement acknowledging hundreds or thousands of new packets. Such a burst would also result if the application was idle for a short period of time less than a round-trip time, and then suddenly had lots of data available to send. In this case, it would be useful for the HighSpeed TCP connection to have some method for limiting bursts.

We do not in this document specify TCP mechanisms for reducing the short-term burstiness. One possible mechanism is to use some form of rate-based pacing, and another possibility is to use maxburst, which limits the number of packets that are sent in response to a single acknowledgement. We would caution, however, against a permanent reduction in the congestion window as a mechanism for limiting short-term bursts. Such a mechanism has been deployed in some TCP stacks, and our view would be that using permanent reductions of the congestion window to reduce transient bursts would be a bad idea [Fl03].

10.3. Other limitations on window size.

The TCP header uses a 16-bit field to report the receive window size to the sender. Unmodified, this allows a window size of at most 2**16 = 65K bytes. With window scaling, the maximum window size is 2**30 = 1073M bytes [RFC 1323]. Given 1500-byte packets, this allows a window of up to 715,000 packets.

10.4. Implementation issues.

One implementation issue that has been raised with HighSpeed TCP is that with congestion windows of 4MB or more, the handling of successive SACK packets after a packet is dropped becomes very time-consuming at the TCP sender [S03]. Tom Kelly's Scalable TCP includes a "SACK Fast Path" patch that addresses this problem.

The issues addressed in the Web100 project, the Net100 project, and related projects about the tuning necessary to achieve high bandwidth data rates with TCP apply to HighSpeed TCP as well [Net100, Web100].

11. Deployment issues.

11.1. Deployment issues of HighSpeed TCP

We do not claim that the HighSpeed TCP modification to TCP described in this paper is an optimal transport protocol for high-bandwidth environments. Based on our experiences with HighSpeed TCP in the NS simulator [NS], on simulation studies [SA03], and on experimental reports [ABLLS03,D02,CC03,F03], we believe that HighSpeed TCP improves the performance of TCP in high-bandwidth environments, and we are documenting it for the benefit of the IETF community. We encourage the use of HighSpeed TCP, and of its underlying response function, and we further encourage feedback about operational experiences with this or related modifications.

We note that in environments typical of much of the current Internet, HighSpeed TCP behaves exactly as does Standard TCP today. This is the case any time the congestion window is less than 38 segments.

Bandwidth	Avg Cwnd w (pkts)	Increase a(w)	Decrease b(w)
1.5 Mbps	12.5	1	0.50
10 Mbps	83	1	0.50
100 Mbps	833	6	0.35
1 Gbps	8333	26	0.22
10 Gbps	83333	70	0.10

Table 9: Performance of a HighSpeed TCP connection.

To help calibrate, Table 9 considers a TCP connection with 1500-byte packets, an RTT of 100 ms (including average queueing delay), and no competing traffic, and shows the average congestion window if that TCP connection had a pipe all to itself and fully used the link bandwidth, for a range of bandwidths for the pipe. This assumes that the TCP connection would use Table 12 in determining its increase and decrease parameters. The first column of Table 9 gives the bandwidth, and the second column gives the average congestion window w needed to utilize that bandwidth. The third column show the increase a(w) in segments per RTT for window w. The fourth column show the decrease b(w) for that window w (where the TCP sender decreases the congestion window from w to w(1-b(w)) segments after a loss event). We note that the actual congestion window when a loss occurs is likely to be greater than the average congestion window w in column 2, so the decrease parameter used could be slightly smaller than the one given in column 4 of Table 9.

Table 9 shows that a HighSpeed TCP over a 10 Mbps link behaves exactly the same as a Standard TCP connection, even in the absence of competing traffic. One can think of the congestion window staying generally in the range of 55 to 110 segments, with the HighSpeed TCP behavior being exactly the same as the behavior of Standard TCP. (If the congestion window is ever 128 segments or more, then the HighSpeed TCP increases by two segments per RTT instead of by one, and uses a decrease parameter of 0.44 instead of 0.50.)

Table 9 shows that for a HighSpeed TCP connection over a 100 Mbps link, with no competing traffic, HighSpeed TCP behaves roughly as aggressively as six parallel TCP connections, increasing its congestion window by roughly six segments per round-trip time, and with a decrease parameter of roughly 1/3 (corresponding to decreasing down to 2/3-rds of its old congestion window, rather than to half, in response to a loss event).

For a Standard TCP connection in this environment, the congestion window could be thought of as varying generally in the range of 550 to 1100 segments, with an average packet drop rate of 2.2 * 10^-6 (corresponding to a bit error rate of 1.8 * 10^-10), or equivalently, roughly 55 seconds between congestion events. While a Standard TCP connection could sustain such a low packet drop rate in a carefully controlled environment with minimal competing traffic, we would contend that in an uncontrolled best-effort environment with even a small amount of competing traffic, the occasional congestion events from smaller competing flows could easily be sufficient to prevent a Standard TCP flow with no lower-speed bottlenecks from fully utilizing the available bandwidth of the underutilized 100 Mbps link.

That is, we would content that in the environment of 100 Mbps links with a significant amount of available bandwidth, Standard TCP would sometimes be unable to fully utilize the link bandwidth, and that HighSpeed TCP would be an improvement in this regard. We would further contend that in this environment, the behavior of HighSpeed TCP is sufficiently close to that of Standard TCP that HighSpeed TCP would be safe to deploy in the current Internet.

11.2. Deployment issues of Scalable TCP

We believe that Scalable TCP and HighSpeed TCP have sufficiently similar response functions that they could easily coexist in the Internet. However, we have not investigated Scalable TCP sufficiently to be able to claim, in this document, that Scalable TCP is safe for a widespread deployment in the current Internet.

Bandwidth	Avg Cwnd w (pkts)	Increase a(w)	Decrease b(w)
1.5 Mbps	12.5	1	0.50
10 Mbps	83	0.4	0.125
100 Mbps	833	4.1	0.125
1 Gbps	8333	41.6	0.125
10 Gbps	83333	416.5	0.125

Table 10: Performance of a Scalable TCP connection.

Table 10 shows the performance of a Scalable TCP connection with 1500-byte packets, an RTT of 100 ms (including average queueing delay), and no competing traffic. The TCP connection is assumed to use delayed acknowledgements. The first column of Table 10 gives the bandwidth, the second column gives the average congestion window needed to utilize that bandwidth, and the third and fourth columns give the increase and decrease parameters.

Note that even in an environment with a 10 Mbps link, Scalable TCP's behavior is considerably different from that of Standard TCP. The increase parameter is smaller than that of Standard TCP, and the decrease is smaller also, 1/8-th instead of 1/2. That is, for 10 Mbps links, Scalable TCP increases less aggressively than Standard TCP or HighSpeed TCP, but decreases less aggressively as well.

In an environment with a 100 Mbps link, Scalable TCP has an increase parameter of roughly four segments per round-trip time, with the same decrease parameter of 1/8-th. A comparison of Tables 9 and 10 shows that for this scenario of 100 Mbps links, HighSpeed TCP increases more aggressively than Scalable TCP.

Next we consider the relative fairness between Standard TCP, HighSpeed TCP and Scalable TCP. The relative fairness between HighSpeed TCP and Standard TCP was shown in Table 5 earlier in this document, and the relative fairness between Scalable TCP and Standard TCP was shown in Table 8. Following the approach in Section 6, for a given packet drop rate p, for p < 10^-3, we can estimate the relative fairness between Scalable and HighSpeed TCP as W_Scalable/W_HighSpeed. This relative fairness is shown in Table 11 below. The bandwidth in the last column of Table 11 is the aggregate bandwidth of the two competing flows given 100 ms round-trip times and 1500-byte packets.

Packet Drop Rate P	Fairness	Aggregate Window	Bandwidth
10^-2	1.0	24	2.8 Mbps
10^-3	1.0	76	9.1 Mbps
10^-4	1.4	643	77.1 Mbps
10^-5	2.1	5595	671.4 Mbps
10^-6	3.1	50279	6.0 Gbps
10^-7	4.5	463981	55.7 Gbps

Table 11: Relative Fairness between the Scalable and HighSpeed Response Functions.

The second row of Table 11 shows that for a Scalable TCP and a HighSpeed TCP flow competing in an environment with 100 ms RTTs and a 10 Mbps pipe, the two flows would receive essentially the same bandwidth. The next row shows that for a Scalable TCP and a HighSpeed TCP flow competing in an environment with 100 ms RTTs and a 100 Mbps pipe, the Scalable TCP flow would receive roughly 50% more bandwidth than would HighSpeed TCP. Table 11 shows the relative fairness in higher-bandwidth environments as well. This relative fairness seems sufficient that there should be no problems with Scalable TCP and HighSpeed TCP coexisting in the same environment as Experimental variants of TCP.

We note that one question that requires more investigation with Scalable TCP is that of convergence to fairness in environments with Drop-Tail queue management.

12. Related Work in HighSpeed TCP.

HighSpeed TCP has been separately investigated in simulations by Sylvia Ratnasamy and by Evandro de Souza [SA03]. The simulations in [SA03] verify the fairness properties of HighSpeed TCP when sharing a link with Standard TCP.

These simulations explore the relative fairness of HighSpeed TCP flows when competing with Standard TCP. The simulation environment includes background forward and reverse-path TCP traffic limited by the TCP receive window, along with a small amount of forward and reverse-path traffic from the web traffic generator. Most of the simulations so far explore performance on a simple dumbbell topology with a 1 Gbps link with a propagation delay of 50 ms. Simulations have been run with Adaptive RED and with DropTail queue management.

The simulations in [SA03] explore performance with a varying number of competing flows, with the competing traffic being all standard TCP; all HighSpeed TCP; or a mix of standard and HighSpeed TCP. For the simulations in [SA03] with RED queue management, the relative fairness between standard and HighSpeed TCP is consistent with the relative fairness predicted in Table 5. For the simulations with Drop Tail queues, the relative fairness is more skewed, with the HighSpeed TCP flows receiving an even larger share of the link bandwidth. This is not surprising; with Active Queue Management at the congested link, the fraction of packet drops received by each flow should be roughly proportional to that flow's share of the link bandwidth, while this property no longer holds with Drop Tail queue management. We also note that relative fairness in simulations with Drop Tail queue management can sometimes depend on small details of the simulation scenario, and that Drop Tail simulations need special care to avoid phase effects [F92].

[SA03] explores the bandwidth 'stolen' by HighSpeed TCP from standard TCP by exploring the fraction of the link bandwidth N standard TCP flows receive when competing against N other standard TCP flows, and comparing this to the fraction of the link bandwidth the N standard TCP flows receive when competing against N HighSpeed TCP flows. For the 1 Gbps simulation scenarios dominated by long-lived traffic, a small number of standard TCP flows are able to achieve high link utilization, and the HighSpeed TCP flows can be viewed as stealing bandwidth from the competing standard TCP flows, as predicted in Section 6 on the Fairness Implications of the HighSpeed Response Function. However, [SA03] shows that when even a small fraction of the link bandwidth is used by more bursty, short TCP connections, the standard TCP flows are unable to achieve high link utilization, and the HighSpeed TCP flows in this case are not ‘stealing’ bandwidth from the standard TCP flows, but instead are using bandwidth that otherwise would not be utilized.

The conclusions of [SA03] are that "HighSpeed TCP behaved as forseen by its response function, and appears to be a real and viable option for use on high-speed wide area TCP connections."

Future work that could be explored in more detail includes convergence times after new flows start-up; recovery time after a transient outage; the response to sudden severe congestion, and investigations of the potential for oscillations. We invite contributions from others in this work.

13. Relationship to other Work.

Our assumption is that HighSpeed TCP will be used with the TCP SACK option, and also with the increased Initial Window of three or four segments, as allowed by [RFC3390]. For paths that have substantial reordering, TCP performance would be greatly improved by some of the mechanisms still in the research stages for robust performance in the presence of reordered packets.

Our view is that HighSpeed TCP is largely orthogonal to proposals for higher PMTU (Path MTU) values [M02]. Unlike changes to the PMTU, HighSpeed TCP does not require any changes in the network or at the TCP receiver, and works well in the current Internet. Our assumption is that HighSpeed TCP would be useful even with larger values for the PMTU. Unlike the current congestion window, the PMTU gives no information about the bandwidth-delay product available to that particular flow.

A related approach is that of a virtual MTU, where the actual MTU of the path might be limited [VMSS,S02]. The virtual MTU approach has not been fully investigated, and we do not explore the virtual MTU approach further in this document.

14. Conclusions.

This document has proposed HighSpeed TCP, a modification to TCP's congestion control mechanism for use with TCP connections with large congestion windows. We have explored this proposal in simulations, and others have explored HighSpeed TCP with experiments, and we believe HighSpeed TCP to be safe to deploy on the current Internet. We would welcome additional analysis, simulations, and particularly, experimentation. More information on simuations and experiments is available from the HighSpeed TCP Web Page [HSTCP]. There are several independent implementations of HighSpeed TCP [D02,F03] and of Scalable TCP [K03] for further investigation.

We are bringing this proposal to the IETF to be considered as an Experimental RFC.

15. Acknowledgements

The HighSpeed TCP proposal is from joint work with Sylvia Ratnasamy and Scott Shenker (and was initiated by Scott Shenker). Additional investigations of HighSpeed TCP were joint work with Evandro de Souza and Deb Agarwal. We thank Tom Dunigan for the implementation in the Linux 2.4.16 Web100 kernel, and for resulting experimentation with HighSpeed TCP. We are grateful to the End-to-End Research Group, the members of the Transport Area Working Group, and to members of the IPAM program in Large Scale Communication Networks for feedback. We thank Glenn Vinnicombe for framing the Linear response function in the parameters of HighSpeed TCP. We are also grateful for contributions and feedback from the following individuals: Les Cottrell, Mitchell Erblich, Jeffrey Hsu, Tom Kelly, Jitendra Padhye, Andrew Reiter, Stanislav Shalunov, Alex Solan, Paul Sutter, Brian Tierney, Joe Touch.

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18. Security Considerations

This proposal makes no changes to the underlying security of TCP.

19. IANA Considerations

There are no IANA considerations regarding this document.

20. TCP's Loss Event Rate in Steady-State

This section gives the number of round-trip times between congestion events for a TCP flow with D-byte packets, for D=1500, as a function of the connection's average throughput B in bps. To achieve this average throughput B, a TCP connection with round-trip time R in seconds requires an average congestion window w of BR/(8D) segments.

In steady-state, TCP's average congestion window w is roughly 1.2/sqrt(p) segments. This is equivalent to a lost event at most once every 1/p packets, or at most once every 1/(pw) = w/1.5 round-trip times. Substituting for w, this is a loss event at most every (BR)/12D)round-trip times.

An an example, for R = 0.1 seconds and D = 1500 bytes, this gives B/180000 round-trip times between loss events.

B. A table for a(w) and b(w).

This section gives a table for the increase and decrease parameters a(w) and b(w) for HighSpeed TCP, for the default values of Low_Window = 38, High_Window = 83000, High_P = 10^-7, and High_Decrease = 0.1.

w	a(w)	b(w)
38	1	0.50
118	2	0.44
221	3	0.41
347	4	0.38
495	5	0.37
663	6	0.35
851	7	0.34
1058	8	0.33
1284	9	0.32
1529	10	0.31
1793	11	0.30
2076	12	0.29
2378	13	0.28
2699	14	0.28
3039	15	0.27
3399	16	0.27
3778	17	0.26
4177	18	0.26
4596	19	0.25
5036	20	0.25
5497	21	0.24
5979	22	0.24
6483	23	0.23
7009	24	0.23
7558	25	0.22
8130	26	0.22
8726	27	0.22
9346	28	0.21
9991	29	0.21
10661	30	0.21
11358	31	0.20
12082	32	0.20
12834	33	0.20
13614	34	0.19
14424	35	0.19
15265	36	0.19
16137	37	0.19
17042	38	0.18
17981	39	0.18
18955	40	0.18
19965	41	0.17
21013	42	0.17
22101	43	0.17
23230	44	0.17
24402	45	0.16
25618	46	0.16
26881	47	0.16
28193	48	0.16

29557	49	0.15
30975	50	0.15
32450	51	0.15
33986	52	0.15
35586	53	0.14
37253	54	0.14
38992	55	0.14
40808	56	0.14
42707	57	0.13
44694	58	0.13
46776	59	0.13
48961	60	0.13
51258	61	0.13
53677	62	0.12
56230	63	0.12
58932	64	0.12
61799	65	0.12
64851	66	0.11
68113	67	0.11
71617	68	0.11
75401	69	0.10
79517	70	0.10
84035	71	0.10
89053	72	0.10
94717	73	0.09

Table 12: Parameters for HighSpeed TCP.

This table was computed with the following Perl program:

```
$top = 100000;
$num = 38;
if ($num == 38) {
  print "     w  a(w)  b(w)\n";
  print "  ----  ----  ----\n";
  print "    38     1  0.50\n";
  $oldb = 0.50;
  $olda = 1;
}
while ($num < $top) {
  $bw = (0.1 -0.5)*(log($num)-log(38))/(log(83000)-log(38))+0.5;
  $aw = ($num**2*2.0*$bw) / ((2.0-$bw)*$num**1.2*12.8);
  if ($aw > $olda + 1) {
     printf "%6d %5d  %3.2f0, $num, $aw, $bw;
     $olda = $aw;
   }
  $num ++;
}
```

Table 13: Perl Program for computing parameters for HighSpeed TCP.

C. Exploring the time to converge to fairness.

This section gives the Perl program used to compute the congestion window growth during congestion avoidance.

```
$top = 2001;
$hswin = 1;
$regwin = 1;
$rtt = 1;
$lastrtt = 0;
$rttstep = 100;
if ($hswin == 1) {
  print "  RTT  HS_Window Standard_TCP_Window0;
  print "  ---  --------- -------------------0;
}
while ($rtt < $top) {
  $bw = (0.1 -0.5)*(log($hswin)-log(38))/(log(83000)-log(38))+0.5;
  $aw = ($hswin**2*2.0*$bw) / ((2.0-$bw)*$hswin**1.2*12.8);
  if ($aw < 1) {
      $aw = 1;
  }
  if ($rtt >= $lastrtt + $rttstep) {
    printf "%5d %9d %10d0, $rtt, $hswin, $regwin;
    $lastrtt = $rtt;
  }
  $hswin += $aw;
  $regwin += 1;
  $rtt ++;
}
```

Table 14: Perl Program for computing the window in congestion avoidance.

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This draft was created in August 2002.