

Evaluation of Advanced TCP Stacks on Fast Long-Distance Production Networks*

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Abstract

With the growing needs of data intensive science, such as High Energy Physics, and the need to share data between multiple remote computer and data centers worldwide, the necessity for high network performance to replicate large volumes (TBytes) of data between remote sites in Europe, Japan and the U.S. is imperative. Currently, most production bulk-data replication on the network utilizes multiple parallel standard (Reno based) TCP streams. Optimizing the window sizes and number of parallel stream is time consuming, complex, and varies (in some cases hour by hour) depending on network configurations and loads. We therefore evaluated new advanced TCP stacks that do not require multiple parallel streams while giving good performances on high speed long-distance network paths. In this paper, we report measurements made on real production networks with various TCP implementations on paths with different Round Trip Times (RTT) using both optimal and sub-optimal window sizes.

We compared the New Reno TCP with the following stacks: HS-TCP, Fast TCP, S-TCP, HSTCP-LP, H-TCP and Bic-TCP. The analysis will compare and report on the stacks in terms of achievable throughput, impact on RTT, intra- and inter-protocol fairness, stability, as well as the impact of reverse traffic.

We also report on some tentative results from

tests made on unloaded 10Gbps paths during SuperComputing 2003.

1 Introduction

With the huge amounts of data gathered in fields such as High Energy and Nuclear Physics (HENP), Astronomy, Bioinformatics, Earth Sciences, and Fusion, scientists are facing unprecedented challenges in managing, processing, analyzing and transferring the data between major sites like major research sites in Europe and North America that are separated by long distances. Fortunately, the rapid evolution of high-speed networks is enabling the development of data-grids and super-computing that, in turn, enable sharing vast amounts of data and computing power. Tools built on TCP, such as bbcp [11], bbftp [4] and GridFTP [1] are increasingly being used by applications that need to move large amounts of data.

The standard TCP (Transmission Control Protocol) has performed remarkably well and is generally known for having prevented severe congestion as the Internet scaled up. It is well-known that the current version of TCP - which relies on the Reno congestion avoidance algorithm to measure the capacity of a network - is not appropriate for high speed long-distance networks. The need to acknowledge packets sets a limit for the throughput for Reno TCP to be a function of $1/RTT$ where RTT is the Round Trip Time. For example, with 1500-Byte packets and a 100ms RTT, it 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

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to achieve a steady-state throughput of 10Gbps (or equivalently, at most one congestion event every 100 minutes)[8]. This loss rate is typically below what is possible today with optical fibers.

Today the major approach to improve the performance of TCP is that of adjusting the TCP window size to the bandwidth (or more accurately the bitrate) * delay (RTT) product (BDP) of the network path, and using parallel TCP streams.

In this paper, we will analyze the performance and the fairness of various new TCP stacks. We ran tests in 3 network configurations: short distance, middle distance and long distance. With these different network conditions, our goal is to find a protocol that is easy to configure, that provides optimum throughput, that is network friendly to other users, and that is responsive to changes in available bitrates. We tested 7 different TCP stacks (see section 2 for a brief description of each): P-TCP, S-TCP, Fast TCP, HS-TCP, HSTCP-LP, H-TCP and Bic-TCP. The main aim of this paper is to compare and validate how well the various TCPs work in real high-speed production networks.

Section 2 describes the specifications of each advanced protocol we tested. Section 3 explains how we made the measurements. Section 4 shows how each protocol: affects the RTT and CPU loads, and behaves with respect to the `txqueuelen` (the number of packets queued up by the IP layer for the Network Interface Card (NIC)). This section also shows: how much throughput each protocol can achieve; how responsive is each protocol in the face of “stiff” sinusoidally varying UDP traffic; and the stability of each protocol. Section 5 moves on to consider the effects of cross-traffic on each protocol. We consider both cross-traffic from the same protocol (intra-protocol) and a different protocol (inter-protocol). We also look at the effects of the reverse traffic on the protocols. Section 6 reports on some tentative results from tests made during SuperComputing 20003 (SC03). Section 7 talks about possible future measurements and section 8 provides the conclusion.

2 The advanced stacks

We selected the following TCP stacks according to two criteria in order to achieve high throughput on long distance:

Software change Since most data-intensive science sites such as our research center are end-users of networks - with no control over the routers or infrastructure of the wide area network - we required that any changes needed would only apply to the end-hosts. Thus, for production networks, protocols like XCP [15] (router assisted protocol) or Jumbo Frame (e.g. MTU=9000) are excluded. Furthermore, since our research center is a major generator and distributor of data, we wanted a solution that only required changes to the sender end of a transfer. Consequently we eliminated protocols like Dynamic Right Sizing [5], which required a modification on the receiver’s side.

TCP improvement Given the existing software infrastructure based on file transfer applications such as `bbftp`, `bbcp` and `GridFTP` that are based on TCP, and TCP’s success in scaling up to the Gbits range [6], we restricted our evaluations to implementations of the TCP protocol. Rate based protocols like SABUL [9] and Tsunami [21] or storage based protocols such as iSCSI or Fiber Channel over IP are currently out of scope.

We call *advanced stacks* the set of protocols presented below, except the first (TCP Reno). All of these stacks are improvements of TCP Reno apart from Fast TCP that is an evolution from TCP Vegas. All the stacks only require to be used on the sender’s side. Further all the advanced stacks run on GNU/Linux.

2.1 Reno TCP

TCP’s congestion management is composed of two major algorithms: the slow-start and congestion avoidance algorithms which allow TCP to increase the data transmission rate without overwhelming the network. Standard TCP cannot inject more than *cwnd* (congestion window) segments of unacknowledged data into the network. TCP Reno’s congestion avoidance mechanism is referred to as AIMD (Additive Increase Multiplicative Decrease). In the congestion avoidance phase TCP Reno increases *cwnd* by one packet per packet of data acknowledged and halves *cwnd* for every window of data containing a packet drop. Hence the following equations:

Slow-Start

$$ACK : new_{cwnd} = old_{cwnd} + c \quad (1)$$

Congestion Avoidance

$$ACK : new_{cwnd} = old_{cwnd} + \frac{a}{old_{cwnd}} \quad (2)$$

$$DROP : new_{cwnd} = old_{cwnd} - b \cdot old_{cwnd} \quad (3)$$

Where $a = 1$, $b = 0.5$, $c = 1$.

2.2 P-TCP

After tests with varying maximum window sizes and numbers of streams, from our research center to many sites, we observed that using the TCP Reno protocol with 16 streams and an appropriate window size (typically the number of streams * window size \sim BDP) was a reasonable compromise for medium and long network distance paths. Since today physicists are typically using TCP Reno with multiple parallel streams to achieve high throughputs, we use this number of streams as a base for the comparisons with other protocols. However:

- It may be over-aggressive and unfair
- The optimum number of parallel streams can vary significantly with changes (e.g, routes) or utilization of the networks.

To be effective for high performance throughput, the best new advanced protocols, while using a single stream, need to provide similar performance to P-TCP (parallel TCP Reno) and in addition, they should have better fairness than P-TCP.

For this implementation, we used the latest GNU/Linux kernel available (2.4.22) which includes SACK and New Reno. This implementation still has the AIMD mechanism shown in (2) and (3).

2.3 S-TCP

Scalable TCP changes the traditional TCP Reno congestion control algorithm: instead of using Additive Increase, the increase is exponential and the Multiplicative Decrease factor b is set to 0.125. It was described by Tom Kelly in [16].

2.4 Fast TCP

The Fast TCP protocol is the only protocol which is based on Vegas TCP instead of Reno TCP. It uses both queuing delay and packet loss as congestion measures. It was introduced by Steven Low and his group at Caltech in [14] and demonstrated during SC2002 [13]. It reduces massive losses using pacing at sender and converges rapidly to an equilibrium value.

2.5 HS-TCP

The HighSpeed TCP was introduced by Sally Floyd in [7] and [8] as a modification of TCP's congestion control mechanism to improve the performance of TCP in fast, long delay networks. This modification is designed to behave like Reno for small values of $cwnd$, but above a chosen value of $cwnd$ a more aggressive response function is used. When $cwnd$ is large (greater than 38 packets), the modification uses a table to indicate by how much the congestion window should be increased when an ACK is received, and it releases less network bandwidth than $\frac{1}{2}cwnd$ on packet loss. We were aware of two versions of High-Speed TCP: Li [18] and Dunigan [3]. Apart from the SC03 measurements, we chose to test the stack developed by Tom Dunigan which was included in the Web100¹ patch.

2.6 HSTCP-LP

The aim of this modification, that is based on TCP-LP [17], is to utilize only the excess network bandwidth left unused by other flows. By giving a strict higher priority to all non-HSTCP-LP cross-traffic flows, the modification enables a simple two-class prioritization without any support from the network. HSTCP-LP was implemented by merging together HS-TCP and TCP-LP.

2.7 H-TCP

This modification has a similar approach to High-Speed TCP since H-TCP switches to the advanced mode after it has reached a threshold. Instead of using a table like HS-TCP, H-TCP uses a heterogeneous AIMD algorithm described in [24].

¹<http://www.web100.org>

2.8 Bic-TCP

In [26], the authors introduce a new protocol whose objective is to correct the RTT unfairness of Scalable TCP and HS-TCP. The protocol uses an additive increase and a binary search increase. When the congestion window is large, additive increase with a large increment ensures linear RTT fairness as well as good scalability. Under small congestion windows, binary search increase is designed to provide TCP friendliness.

2.9 Westwood+ TCP

This protocol continuously estimates the packet rate of the connection by monitoring the ACK reception rate. The estimated connection rate is then used to compute congestion window and slow start threshold settings after a timeout or three duplicate ACKs. This protocol was described in [20].

2.10 GridDT

This protocol allows the users to tune AIMD parameters which can reproduce the behavior of a multi-stream transfer with a single stream and can virtually increase the MTU as described in [22].

Due to some time delay in the kernel availability, we were unable to test Westwood+ TCP and GridDT. We hope to test and report on them in a future paper.

3 Measurements

Each test was run for 20 minutes from our research center to three different networks: Caltech for short-distance (RTT of 10 ms), University of Florida for middle distance (RTT of 70 ms) and University of Manchester for long-distance (RTT of 170 ms). We duplicated some tests to DataTAG² Chicago (RTT of 70 ms) and DataTAG CERN (RTT of 170 ms) in order to see if our tests were coherent.

The throughputs on these production links go from 400 Mbps to 600 Mbps which was the maximum we could reach because of the OC12 (622

Mbps) links to ESnet and CENIC at our research center. The route for Caltech uses CENIC from our research center to Caltech and the bottleneck capacity for most of the tests was 622 Mbps. The route used for University of Florida (UFL) was CENIC and Abilene and the bottleneck capacity was 467 Mbps at UFL. The route to CERN was via ESnet and Starlight and the bottleneck capacity was 622 Mbps at our research center. The route used for University of Manchester is ESnet then Geant and JANET.

At the sender side, we used three machines:

Machine 1 runs ping.

Machine 2 runs Advanced TCP.

Machine 3 runs Advanced TCP for cross-traffic or UDP traffic.

Machines 2 and 3 had 3.06GHz dual-processor Xeons with 1 GB of memory, a 533MHz front side bus and an Intel GigabitEthernet (GE) interface. Due to difficulties concerning the availability of hosts at the receiving sites, we usually used only two servers on the receiver's side (Machines 1 and 2 at the sender side send data to the same machine at the receiver side).

After various tests, we decided to run **ping** and **iperf** in separate machines. With this configuration we had no packet loss for **ping** during the tests.

We used a modified version of **iperf**³ in order to test the advanced protocol in a heterogeneous environment. Following an idea described by Hacker [10], we modified **iperf** to be able to send UDP traffic with a sinusoidal variation of the throughput. We used this to see how well each advanced TCP stack was able to adjust to the varying "stiff" UDP traffic. The amplitude of the UDP stream varied from 5% to 20% of the bandwidth with periods of 60 seconds and 30 seconds. Both the amplitude and period could be specified.

We ran **iperf** (TCP and UDP flows) with a report interval of 5 seconds. For the ICMP traffic the interval, that was used by the traditional **ping** program, is of the same order as the RTT in order to gain some granularity in the results. The tests were run mostly during the weekend and the night in order to reduce the impact on other traffic.

On the sender's side, we used the different kernels patched for the advanced TCP stacks. The different kernels are based on vanilla GNU/Linux 2.4.19

²Research & Technological Development for a TransAtlantic Grid: <http://datatag.web.cern.ch/datatag/>

³<http://dast.nlanr.net/Projects/Iperf/>

	TCP Reno	P-TCP	S-TCP	Fast TCP	HS-TCP	Bic-TCP	H TCP	HSTCP-LP
Caltech 256KB	238+-15	395+-33	226+-14	233+-13	225+-17	238+-16	233+-25	236+-18
Caltech 512KB	361+-44	412+-18	378+-41	409+-27	307+-31	372+-35	338+-48	374+-51
Caltech 1MB	374+-53	434+-17	429+-58	413+-58	284+-37	382+-41	373+-34	381+-51
UFL 1MB	129+-26	451+-32	109+-18	136+-12	136+-15	134+-13	140+-14	141+-18
UFL 4MB	294+-110	428+-71	300+-108	339+-101	431+-91	387+-52	348+-76	382+-120
UFL 8MB	274+-115	441+-52	281+-117	348+-96	387+-95	404+-34	351+-56	356+-118
Manchester 4MB	97+-38	268+-94	170+-20	163+-33	171+-15	165+-26	172+-13	87+-61
Manchester 8MB	78+-41	232+-74	320+-65	282+-113	330+-52	277+-92	323+-64	118+-111
Manchester 12MB	182+-66	212+-83	459+-71	262+-195	368+-161	416+-100	439+-129	94+-113
Avg. thru Size 1	154	371	178	177	177	179	185	155
Avg. thru Size 2	244	357	384	343	356	345	336	292
Avg. thru Size 3	277	362	422	341	346	367	388	277
Avg. thru size 2 & 3	261	360	403	342	351	356	362	294
Std. dev. size 2 & 3	113	107	49	53	54	49	41	125

Table 1: Iperf TCP throughputs for various TCP stacks for different window sizes, averaged over the three different network path lengths.

through GNU/Linux 2.4.22. The TCP source code of the vanilla kernels is nearly identical. On the receiver’s side we used a standard Linux kernel without any patch in the TCP stack.

For each test we computed different values: throughput average and standard deviation, RTT average and standard deviation, stability and fairness index. The stability index helps us find out how the advanced stack evolves in a network with rapidly varying available bandwidth.

With `iperf`, we can specify the maximum window size the congestion window can reach. The optimal window sizes according the bandwidth delay product are about 500KBytes for the short distance path, about 3.5MBytes for the medium distance path and about 10MBytes for the long distance path. We used 3 main window sizes for each path in order to try and bracket the optimum in each case: for the short-distance we used 256KBytes, 512KBytes and 1024KBytes; for the middle distance we used 1MBytes, 4MBytes and 8MBytes; and for the long-distance we used 4MByte, 8MByte and 12MByte maximum windows. In this paper, we note these three different window sizes for each distance as: size 1, 2 and 3.

4 Results

In this section, we present the essential points and the analysis of our results. The whole data are

available on our website⁴.

4.1 RTT

All advanced TCP stacks are “fair” with respect to the RTT (i.e. do not dramatically increase RTT) except for P-TCP Reno. On the short distance, the RTT of P-TCP Reno increases from 10 ms to 200 ms. On the medium and long distances, the variation is much less noticeable and the difference in the average RTTs between the stacks is typically less than 10ms.

For the other advanced stacks the RTT remains the same except with the biggest window size we noticed, in general, a small increase of the RTT.

4.2 CPU load

We ran our tests with the `time` command in order to see how each protocol used the cpu resource of the machine on the sender’s side. The MHz/Mbps utilization averaged over all stacks, for all distances and all windows was 0.93+-0.08 MHz/Mbps. The MHz/Mbps averaged over all distances and window sizes varied from 0.8+-0.35 for S-TCP to 1.0+-0.2 for Fast. We observed no significant difference in sender side CPU load between the various protocols.

⁴<http://www-iepm.slac.stanford.edu/bw/tcp-eval/>

4.3 txqueuelen

In the GNU/Linux 2.4 kernel, the `txqueuelen` enables us to regulate the size of the queue between the kernel and the Ethernet layer. It is well-known that the size of the `txqueuelen` on the NIC can change the throughput but we have to use some optimal tuning. Some previous tests [19] were made by Li. Although use of a large `txqueuelen` can result in a large increase of the throughput with TCP flows and a decrease of `sendstall`, Li observed an increase of duplicate ACKs.

Scalable TCP by default used a `txqueuelen` of 2000 but all the others use 100. Thus, we tested the various protocols with a `txqueuelen` sizes of 100, 2000 and 10000 in order to see how this parameter could change the throughput. In general, the advanced TCPs perform better with a `txqueuelen` of 100 except for S-TCP which performs better with 2000. With the largest `txqueuelen`, we observe more instability in the throughput.

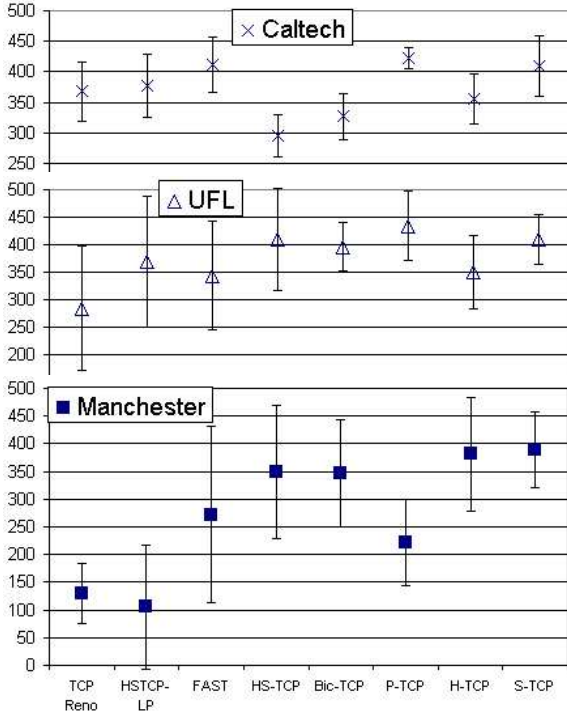


Figure 1: Average of the 1200 second averages for maximum window sizes 2 and 3 shown for three network distances and various TCP stacks. The y axis is the throughput achieved in Mbps.

4.4 Throughput

Table 1 and Figure 1 show the iperf TCP throughputs averaged over all the 5 seconds intervals for each 1200 second measurement (henceforth referred to as the 1200 second average) for the various stacks, network distances and window sizes. Also shown are the averages of the 1200 second averages for the three network distances for each window size. Since the smallest window sizes were unable to achieve the optimal throughputs, we also provide the averages of the 1200 second averages for sizes 2 and 3.

- With the smallest maximum window sizes we were unable to achieve optimal throughputs except when using P-TCP.
- Depending on the paths, we could achieve throughputs varying from 300 to 500 Mbps.
- There are more differences in the protocol achievable throughputs for the longer distances.
- For the long distance (Manchester), the BDP predicts an optimum window size closer to 12MBytes than 8Mbytes. As a result S-TCP, H-TCP, Bic-TCP and HS-TCP perform best for the Manchester path with the 12MByte maximum window size.
- The top throughput performer for window sizes 2 and 3 was Scalable-TCP, followed by (roughly equal) Bic-TCP, Fast TCP, H-TCP, P-TCP and HS-TCP, with HSTCP-LP and Reno single stream bringing up the rear.
- The poor performance of Reno single stream is to be expected due to its AIMD congestion avoidance behavior.
- Since HSTCP-LP deliberately backs off early to provide a lower priority, it is not unexpected that it will perform less well than other more aggressive protocols.
- P-TCP performs well at the short and medium distances, but less well on the long-distance path, possibly since the $\text{windows} \times \text{streams}$ product was \gg the BDP.

It needs to be noted that the standard deviations of these averages are sufficiently large the ordering should only be regarded as a general guideline.

4.5 Sinusoidal UDP

The throughput of a protocol is not sufficient to describe its performance. Thus, we analyzed how the protocol behaves when competing with a UDP stream varying in a sinusoidal manner. The purpose of this stream is to emulate the variable behavior of the background cross-traffic. Our results show that in general, all protocols converge quickly to follow the changes in the available bandwidth and maintain a roughly constant aggregate throughput - especially for Bic-TCP. Fast TCP, and P-TCP to a lesser extent have, some stability problems on long-distance and become unstable with the largest window size. Figure 2 shows an example of the variation of Bic-TCP in the presence of sinusoidal UDP traffic.

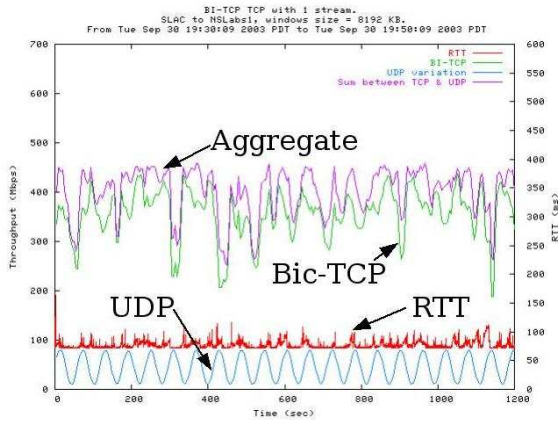


Figure 2: Bic-TCP with sinusoidal UDP traffic.

4.6 Stability

We compute the stability index as the standard deviation normalized by the average throughput. If we have few oscillations in the throughput, we will have a stability index close to zero.

Without the UDP cross-traffic, all stacks have better stability indices (factor of 1.5 to 4 times better) with the smallest window sizes. S-TCP has the best stability (index ~ 0.1) for the optimal and larger than optimal window sizes, this is followed closely by H-TCP, Bic-TCP and HS-TCP. Single stream Reno and HSTCP-LP have poorer stabilities (> 0.3).

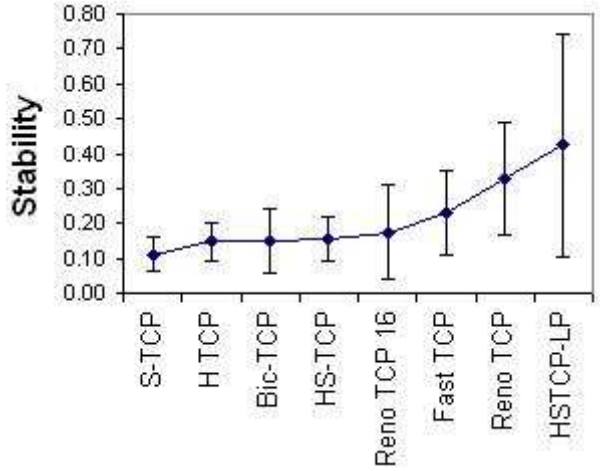


Figure 3: Stability index averaged over the optimal and largest window sizes and the 3 different network path lengths.

With the sinusoidal UDP traffic better stability is achieved once again with the smallest window sizes. For the other window sizes there is little difference (0.01) between the two UDP-frequency stabilities for a given stack. The throughputs with the UDP cross-traffic are generally larger (15%) than those without the UDP cross-traffic. Bic-TCP closely followed by the two more aggressive protocols, P-TCP and Scalable-TCP, have the best stability indices (< 0.2). H-TCP and HS-TCP have stability indices typically > 0.2 and Fast TCP and HSTCP-LP have stability indices > 0.3 . We noticed that, in general, the protocols are more stable with the largest window (size 3) than with the optimal window (size 2).

5 Cross-traffic

5.1 Intra-protocol fairness

The cross-traffic tests are important and help us to understand how fair a protocol is. At our research center, we wanted to know not only the fairness of each advanced protocol against TCP Reno, but also how fairly the protocols behave towards each other. It is important to see how the different protocols compete with one another since the protocol that our research center will adopt shortly must coexist

	P-TCP	S-TCP	Fast TCP	HS-TCP	Bic-TCP	H TCP	HSTCP-LP
short distance	0.16	0.24	-0.1	-0.28	0.01	-0.02	-0.47
middle-distance	0.78	0	-0.01	-0.06	0.15	-0.12	0
long-distance	0.19	-0.08	0.04	-0.38	-0.03	0.25	-0.56
Average	0.37	0.05	-0.02	-0.24	0.04	0.04	-0.34

Table 2: Average asymmetry of the marked protocol vs all others (see the comments for details).

harmoniously with existing protocols and with advanced protocols chosen by other sites. Of course, we cannot avoid a future protocol being unfair only with our chosen one.

In this paper we consider a fair share per link metric. If there are n flows through a bottleneck link, each flow will take $1/n$ of the capacity of the bottleneck link. We measure the average bandwidth x_i of each source i during the test then we compute the fairness index as described in [2] by Jain and al.:

$$F = \frac{(\sum_{i=1}^n \bar{x}_i)^2}{n \sum_{i=1}^n \bar{x}_i^2}$$

A fairness index of 1 corresponds to a perfect allocation of the throughput between all protocols.

There are other definitions of the concept of fairness. For example, in [25] the authors describe and extend the concept of “ F_a fairness”. However, we chose to use the definitions of Chiu and Jain which are the ones mainly used in the networking literature concerning a simple model of a single bottleneck.

The intra-protocol fairness is the fairness between two flows of the same protocol. Each flow is sent from a different sending host to a different receiving host.

In general, all the protocols have a good intra-fairness ($F \geq 0.98$) except HS-TCP ($F \leq 0.94$) on the middle distance, and P-TCP, H-TCP and HSTCP-LP on the long-distance with window sizes greater than the optimal. We noticed that on the middle distance, the two HS-TCP flows will switch with one another instead of maintaining a constant share of the bandwidth. The first flow will decrease after a certain time and leave the available bandwidth to the second flow. As a result, we observe a large instability in these HS-TCP flows. We do not notice this HS-TCP behavior on the short-distance path.

5.2 Inter-protocol fairness

For the inter-protocol fairness we sent two different flows on the link from two different machines. The aim of this experience was to see how each protocol behaves with a competing protocol. We desired that the protocol would neither be too aggressive nor too gentle (non-aggressive) towards the other protocols. The fairness computation described earlier does not tell us how aggressive or gentle the protocol is, only that it is not taking/getting a fair share of the achievable throughput. Hence we introduce the following formula, which is the asymmetry between two throughputs:

$$A = \frac{\bar{x}_1 - \bar{x}_2}{\bar{x}_1 + \bar{x}_2}$$

where \bar{x}_1 and \bar{x}_2 are the throughput averages of streams 1 and 2 in the cross-traffic.

Table 2 shows the result of the cross-traffic between different stacks. A value near one indicates that the protocol is too aggressive towards the competing protocol. A value near minus one indicates a too gentle protocol. The optimal is to have a value near 0 that indicates that the protocol is fair against the other protocols.

Our results show that Bic-TCP, Fast TCP, S-TCP and H-TCP have small absolute values of the fairness asymmetry. It is normal for HSTCP-LP to be too gentle (and have a large negative value of the asymmetry) since it uses only the remaining bandwidth and is deliberately non-intrusive - thus we removed it from our calculation of the average asymmetry of the other protocols for the middle-distance and long-distance. On the short-distance, we can see that as expected P-TCP is very aggressive - thus, we do not include P-TCP in the average asymmetry of the other protocols for the short-distance - and that the other advanced TCP stacks compete like a single stream of Reno. Only Bic-TCP is sufficiently aggressive to compete with P-TCP in this case, but it appears too aggressive for

the other protocols. Our results show that S-TCP, which is very aggressive in short-distance, becomes quite gentle in the long-distance. On the other hand, H-TCP, which is gentle in the short and middle distances, becomes aggressive in long-distance. HS-TCP is too gentle in our tests.

5.3 Reverse-traffic

Reverse-traffic causes queuing on the reverse path. This in turn can result in the ACKs being lost or coming back in bursts (compressed ACKs). So, we tested the protocols by sending TCP traffic from our research center to UFL using an advanced stack and from UFL to our research center using P-TCP with 16 streams. Normally, the router, the path and the Ethernet card are full-duplex and should not be affected by the reverse-traffic but actually the reverse-traffic affects the forward throughputs by its modification of the ACK behavior. For example, HS-TCP never reaches the limit at which the AIMD behavior changes from Reno to HS. Also, our tests show that Fast TCP - which is based on TCP Vegas that measures RTT - is more heavily affected by heavy reverse-traffic that affects (usually increases) the reverse path delays and hence the RTTs. The net effect is that, for the tested version of Fast TCP, throughput is 4 to 8 times less than the other stacks.

6 10Gbit/s path tests

During SuperComputing 2003⁵, we made some tentative TCP performance measurements on 10Gbps links between hosts at the SLAC/FNAL booth at the Phoenix convention center and a host at the SLAC/Stanford Point of Presence at the Palo Alto Internet eXchange (PAIX) and hosts at StarLight in Chicago. Due to the short amount of time we had access to these links (~ 3 days) and the emphasis on demonstrating the maximum throughput for the SC03 Bandwidth Challenge these measurements are necessarily incomplete, however some of the tentative results are felt to be worth reporting.

⁵SC2003: <http://www.sc-conference.org/sc2003/>

6.1 Setup

All the hosts at Phoenix and PAIX were Dell 2650s with dual Xeon CPUs, a 533MHz front side bus, each with an Intel PRO/10GbE LR Network Interface Card (NIC) plugged into the 133MHz 64 bit PCI-X bus slot. There were 3 hosts at the SLAC/FNAL booth at SC03, two with 3.06GHz CPUs, and the third with 2.04GHz CPUs. The host at PAIX had dual Xeon 3.06GHz CPUs, a 533MHz front side bus, and an Intel PRO 10GbE Ethernet connection (level 2) to a Cisco HPR router in LA. From the HPR router the signal was transmitted via an OC192/POS circuit to a Juniper router managed by SCInet at the Phoenix Convention Center. From the Juniper the path went via a Force 10 E1200 router to the Cisco 6509 in the SLAC/FNAL booth.

The StarLight/Chicago path from the booth went via the Force 10 router through a second Juniper router at 10Gbps to the Abeline core and thence via Abilene routers at Los Angeles, Sunnyvale, Kansas City, and Indianapolis to Chicago.

6.2 Methodology

We set up the sending hosts at SC03 with the Caltech Fast TCP stack, and the DataTAG altAIMD stack⁶ that allowed dynamic (without reboot) selection of the standard Linux TCP stack (New Reno with Fast re-transmit), the Manchester University implementation of the High Speed TCP (HS-TCP) and the Cambridge University Scalable TCP stack (S-TCP). By default we set the Maximum Transfer Unit (MTU) to 9000Bytes and the transmit queue length (txqueuelen) to 2000 packets.

We started the measurements just before midnight on Wednesday 19th November. This was a reasonably controlled set of measurements with little cross-traffic. Each test was for 1200 seconds, with a given stack, and fixed maximum window size.

6.2.1 Tests to PAIX (17ms RTT)

On the Phoenix to PAIX link we used maximum window sizes of 8MBytes, 16MBytes and

⁶Available at: <http://www.hep.uvl.ac.uk/~ytl/tcpip/linux/altaimd/>

32MBytes⁷. This bracketed the nominal optimum window size calculated from the BDP of 17ms * 10Gbps ~ 20MBytes. For the PAIX link, all the tests were made with a single TCP stream. For Reno, HS-TCP and S-TCP there was little observable differences between the four stacks in the achievable bandwidth behavior.

- For an MTU of 9000Bytes, a window size of 8MBytes was inadequate and resulted in throughput being limited to about 2.9Gbps for Reno single stream, HS-TCP and S-TCP.
- For an MTU of 9000Bytes with window sizes of 16MBytes or 32MBytes for Reno single stream, HS-TCP and S-TCP the throughput increased to about 4.3Gbps.
- The throughputs with an MTU of 9000Bytes were very stable. Typical values of the Stability Index (standard deviation / average throughput) were < 0.4%. The larger values for Reno with a single stream and an 8MByte window and HS-TCP with a 16MByte window were each caused by a single sudden drop of throughput for one 5 second period.
- Reducing the MTU from 9000Bytes to 1500Bytes reduced the throughput for a 16MByte window from 4.3Gbps to about 700Mbps, and for an 8MByte window from 2.9Gbps to ~ 360Mbps. Also the throughputs were less stable with the 1500Byte MTU (Stability Index > 10%).
- With an MTU of only 1500Bytes, Fast TCP gave similar performance to HS-TCP and S-TCP when they ran with 1500Byte MTUs.

The 4.3Gbps limit was slightly less than the ~ 5.0Gbps achieved with UDP transfers in the lab at SLAC between back to back 3.06GHz Dell PowerEdge 2650 hosts. On the other hand it is less than that calculated from the expected data transfer rate for a 10GE NIC with a 64 bit 133 MHz PCI-X bus⁸.

⁷More explanations are available on: <http://www-iepm.slac.stanford.edu/monitoring/bulk/sc2003/hiperf.html> and <http://www-iepm.slac.stanford.edu/monitoring/bulk/sc2003/stack-cf.html>

⁸See Figure 3.1 of Performance of Gigabit and 10 Gigabit Ethernet NICs with Server Quality Motherboards by R Hughes-Jones, P. Clarke, S. Dallisom and G. Fairey, to be published

The limitation in throughput is believed to be due to CPU factors (CPU speed, memory/bus speed or the I/O chipset). The relative decrease in throughput going from 9000Byte MTU to a 1500Byte MTU was roughly proportional to the reduction in MTU size. This maybe related to the extra cpu speed / memory bandwidth required to process the 6 times as many, but 6 times as small MTUs. Back to back UDP transfers in the lab at SLAC between 3.06GHz Dell PowerEdge 2650 hosts achieved about 1.5Gbps or about twice the 700Mbps achieved with the SC03 long distance TCP transfers. Further work is required to understand this discrepancy.

7 Future experiments

In the near future, we plan on repeating the tests on higher speed networks, in particular on the emerging 10Gbps testbeds. We also plan to test other promising TCP stacks such as Westwood+, and rate based protocols such as RBUDP, SABUL and UDT, and compare their performances with the TCP based protocols. Also we are planning to work with others to compare our real network results with those from simulators such as ns-2⁹ or emulators such as Dummynet [23].

In the future, we would like to test a similar topology as described in [12] where the authors indicate that it may be beneficial for long RTT connections to become slightly more aggressive during the additive increase phase of congestion avoidance. In this paper we only made cross-traffic tests with two protocols having the same RTT. It means that all the senders' servers were at the same place. It was the same for the receiver. Thus, we have to check how each protocol behaves with different RTTs on the same link. The increase between the different protocols on the path will be different and it may affect the fairness.

We should also test the different protocols with more than one stream to see how aggressive or gentle a protocol is on this case. Finally, we plan to test other promising TCP stacks and rate based protocol and compare their performance with the TCP based protocols.

⁹"The Network Simulator - ns-2", available at <http://www.isi.edu/nsnam/ns/>

8 Conclusion

In this paper we presented the results of a two-month experiment to measure the performance of 7 TCPs from SLAC over various network paths. If we compare the various TCPs for the more important metrics (throughput achievable, impact on RTT, aggressiveness, stability and convergence) we observe for the set of measurements:

- The differences in the performances of the TCP stacks is more noticeable for the longer distances.
- TCP Reno single stream, as expected, is low performance and unstable on longer distances.
- P-TCP is too aggressive. It is also very unfair with the RTT on short distance.
- HSTCP-LP is too gentle and, by design, backs-off too quickly otherwise it performs well. It looks very promising to use to get Less than Best Effort (LBE) service without requiring network modifications.
- Fast TCP performs as well as most others but it is very handicapped by the reverse traffic.
- S-TCP is very aggressive on middle-distance and becomes unstable with UDP traffic on long distance but achieves high throughput.
- HS-TCP is very gentle and has some strange intra-fairness behavior.
- Bic-TCP overall performs very well in our tests.

It is also very important to choose a TCP stack that works well with and will not decrease the performance and efficiency of TCP Reno used all around the world. Moreover, we will always prefer an advanced TCP which has the recommendation of the IETF and which will be used by everybody.

Finally, it must be borne in mind that the transport needs can be very different whether you transfer a huge amount of data during several minutes or you need the connection for a one-hour video conference, or just want to surf the web.

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