An Investigation into Transport Protocols and Data Transport Applications Over High Performance Networks

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Abstract

The future of Grid technology relies on the successful and efficient movement of control information and raw data around the world to interconnect computers that form the backbone of the Internet. It has been estimated that Petabytes of data will be replicated and consumed across many sites worldwide in projects such as the LHC and AstroGrid. This thesis focuses on the performance issues related to replicating such large volumes of data across the Internet for the successful deployment of Grid in the near future.

More specifically, the main content of the thesis focuses on the reliable transport protocols that are necessary for checkpoint and replication data. It is shown that the existing Transmission Control Protocol (TCP) algorithms are insufficient to make use of the increased network capacities of high speed long distance networks. Many new proposals have been put forth to solve this problem. They are described and extensively explored in a set of simulated, lab-based and real-life tests in order to validate the theoretical models and experimental results from real-world application.

As the issues of new transport protocols are not solely a matter of the total achievable throughput for a single user, but a problem of network stability and fairness, a set of performance parameters based on these metrics are also investigated.

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A special 'thank you' goes to Ben Waugh who was kind enough to plough through the entirety of this thesis.

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Last, but not least, I wish to thank Pi-Hsien Chang and my parents for their continued support and compassion during this degree.

Publications And Workshops

Some of the work described in this dissertation has also been published and or presented elsewhere:

- Yee-Ting Li, Stephen Dallison, Richard Hughes-Jones and Peter Clarke, Systematic Analysis of High Throughput TCP in Real Network Environments, In *Second International Workshop on Protocols for Long Distance Networks*, February 2004
- Andrea Di Donato, Yee-Ting Li, Miguel Rio and Peter Clarke, Using QoS fo High Throughput TCP Transport Over Fat Long Pipes, In *Second International Workshop on Protocols for Long Distance Networks*, February 2004
- Yee-Ting Li, Experience with Loss Based Congestion Algorithms, In *First International Grid Networking Workshop*, March 2004
- Richard Hughes-Jones, Steven Dallison, Nicola Pezzi and Yee-Ting Li, In *Bringing High-Performance Networking to HEP users, Computing in High Energy and Nuclear Physics 04*, September 2004
- Yee-Ting Li, Doug Leith and Robert Shorten, Experimental Evaluation of TCP Protocols for High-Speed Networks, submitted to and accepted by *IEEE/ACM Transactions on Networking*, June 2005

Chapter 1

Introduction

The growth of the Internet has enabled the sharing of data and has improved the collaboration of major science projects between academics and researchers around the world.

The deployment of data Grids and compute Grids will further increase the productivity of large scale compute and data intensive applications by distributing processing and storage subsystems around the world. Through the implementation of *middleware*, Grids will rely on existing hardware solutions such as RAID storage systems and Internet communications.

1.1 Research Motivation

This dissertation begins by looking into data grids where the storage and retrieval of large data sets are important - where jobs are limited primarily by the retrieval of data over the Internet rather than the processing of it.

Several key aims are pursued:

- Identification that the key performance bottleneck is the replication of large scale data across the Internet and the understanding of the mechanisms that limit data transfer rates.

- Several new congestion control algorithms that govern the rate of data transfer are investigated and systematically tested across laboratory and dedicated wide-area networks, and across the Internet to determine the benefits and disadvantages of each.

It is the aim of this dissertation to use systematic testing to highlight the performance bottlenecks and the benefits and disadvantages of new transport algorithms for the replication of bulk data. Furthermore the research conducted here will help refine suitable design and implementation of high speed Internet transport protocols.

1.2 Research Scope

This dissertation assumes that the predominant transport protocol in use on the Internet will continue to be the Transmission Control Protocol. Also, it is assumed that the data to be replicated is sufficiently large that the TCP flows are 'long lived' and that they spends most of their time in the 'congestion avoidance' phase of TCP's transmission behaviour. As such, the improvement of congestion control algorithms will aid the performance and fairness of all flows competing along a bottleneck link.

The effects of network topology, (multiple) bottleneck location and policing mechanisms are not investigated.

The dissertation also takes the view that the deployment of new transport protocol algorithms should be evolutionary rather than revolutionary. This is

especially important as the distributed nature of the Internet makes switched deployment difficult.

1.3 Contributions

The contributions made by this research are as follows, all work was conducted by myself, unless otherwised noted:

- The identification of TCP as a bottleneck in Internet communications and the investigation of performance enhancing techniques for TCP.
- The implementation of several New-TCP algorithms in Linux kernels (with SACK and Linux networking optimisation code from Baruch Evans and Doug Leith of Hamilton Institute).
- The evaluation of several New-TCP algorithms in laboratory conditions (i.e. dummynet [Riz98]) across varying network conditions.
- The evaluation of several New-TCP algorithms across MB-NG (See Appendix C.2) and DataTAG (See Appendix C.3) research-use wide area networks.
- The Identification of severe unfairness experienced by New-TCP flows under asymmetric network conditions.
- The identification of negative consequences for existing legacy traffic with New-TCP algorithms.
- The implementation of DiffServ with New-TCP algorithms (conducted with Andrea Di Donato of University College London).

- Identification of Linux specific limitations upon TCP transport performance.
- Identification of *ack* and SACK processing bottlenecks on TCP transport performance.
- The evaluation of single flow bulk transport over the Internet using New-TCP algorithms.
- The identification of important network metrics for the successful utilisation of TCP transport for Large Hadron Collider (LHC) applications, with suggestions for New-TCP usage.

1.4 Dissertation Outline

Scope of the work in terms of the Large Hadron Collider (LHC) project is presented in Chapter 2 and data replication requirements are discussed. Preliminary tests are performed in Chapter 3 which investigates the performance of the UDP and TCP protocols across the Internet. It is identified that a performance gap exists between the two - suggesting that TCP has performance problems not associated with raw hardware performance.

An in-depth review of TCP and the needs for congestion avoidance is provided in Chapter 4 and the performance limitations are explored in Chapter 5. In particular, it is identified that the fundamental Additive Increase Multiplicative Decrease (AIMD) congestion avoidance algorithm of TCP imposes an absolute performance limit under realistic network conditions for the High Energy Physics (HEP) research community.

A survey of amendments to the TCP congestion control algorithm is presented and discussed in Chapter 6, a testing framework where new congestion

control algorithms can be comparatively analysed is presented in Chapter 7 and a series of systematic tests involving this framework is presented in Chapter 8.

Chapter 9 presents results of running these algorithms under dedicated cross-UK and trans-Atlantic networks and highlights the hardware and software limitations of protocol implementation. A network Quality of Service solution using DiffServ is also presented, and the benefits of Active Queue Management are demonstrated. A series of network tests involving New-TCP algorithms is conducted across the Internet and results analysed.

Finally the applicability of these TCP algorithms are then presented in Chapter 10 and recommendations are made for the deployment of these algorithms on the Large Hadron Collider (LHC) project. A summary of the contributions made and areas for further research is also presented.

Chapter 2

High Energy Particle Physics

The high energy physics experiments in the forthcoming decades will provide insight into the nature of fundamental interactions, structures and symmetries that define space-time.

The application of computer processing and information retrieval for such large-scale science projecys has lead scientists to an increasing need to transfer petabytes of data across the Internet in order to realise the goal of world wide collaborative science.

This Chapter looks into one such application where there is an increasing demand to meet the requirements of transferring many petabytes of data per second across geographically distributed locations. The network requirements of new High Energy Physics experiments are described here, taking the Large Hadron Collider (LHC) experiment as an example.

2.1 Overview

High energy physics experiments are designed to accelerate and collide charged particles at such high energies that new particles of matter may be found in the remnants of the collision. By the analysis of these remnants in terms of velocity, charge, energy and decay rate, physicists are able to deduce whether the experimental results have lead to the verification of theoretical predictions of new particles.

Different types of detectors are used to characterise each and every particle that is involved both before and after such a collision (referred to by physicists as an *event*) and are arranged radially from the beam line of the initially accelerated particles. Silicon vertex detectors are used to reconstruct tracks and identify a particle decay; drift chambres are used to determine a particle's momentum and particle energies are measured inside calorimeters.

Each detector has independent readout channels that will produce output when a particle has been detected.

Due to the large velocities and the small sizes of the particles, groups of particles are typically used (such as in the LHC) in order to maximise the possibility of a collision which may result in the desired event. Typically, most interactions are of no or little interest and therefore decisions must be made at runtime to either keep or reject the data (which must be done carefully in order to reduce the risk of rejecting interesting collisions). As such, the detectors are designed to continuously output huge amounts of data, only some of which will be retained for storage.

The process filtering this data is called a *trigger*. A combination of hardware and software triggers are used to reduce the amount of data, which will be appropriately buffered in order to minimise the amount of time processing

the data in order to increase collider efficiency.

2.2 Analysis Methods

The search for new exotic particles such as the Higgs boson becomes a statistical study of the probabilities that a particular event occurs. As such, large volumes of data must be generated in order to determine with high confidence that these rare events do actually occur.

Also, Monte Carlo simulations must be produced in order to reduce statistical errors to an acceptable level. Generally, the size of such datasets are equivalent to that of the filtered data from the detector.

At present, a typical experiment will store all raw detector data centrally in tape archives, from which the data will be staged to disk when required. When an institute wishes to run analysis on a particular dataset, the data is typically copied (replicated) from the central facility to that of the local farm at the institute. Collaborating institutes on the experiment operate more or less independently, in as much as should another institute also want to analyse the same dataset, they will typically also replicate it to their local facilities - unaware that an exact replica may be physically closer than that at the detector.

Access patterns for datasets vary. Experimental data files typically have a single creator from which the initial production period will last several weeks and will be modified as new information is added. However, metadata is also created, which describes the information about the experiment. This may be created by multiple individuals and may be modified and or augmented over time, even after the creation of the experimental data. The size of meta data is typically smaller than that of the experimental data.

2.3 Data Volumes and Regional Centres

The total output of data from the LHC experiment is expected to be of the order of ten petabytes per year. Not only must this data be stored, but it must be made available to sites distributed around the world, and processing power must be available in order to analyse it. The estimated data set sizes of the LHC experiment are an order of magnitude greater than that of any previous experiment. The increased size of LHC experimental data poses an increased burden on individual institutions to house and manage large compute and storage farms in order to analyse the data. With tens of terabytes of data produced each day, both storage and compute resources at any one site would soon be exhausted.

Assuming that each event will occupy approximately 1.5MB of storage space [Bau03], then an experiment such as the LHC will produce approximately 13TB [Bau03] of raw data in a day. This is an order of magnitude greater than the data rates produced by detectors such as the Tevatron at Fermilab [ea02].

Therefore, the rate at which data can be analysed is typically limited by both the compute and network resources available to each institution. With the increased volume of data that will become available with the introduction of the LHC, a new computing model that spans the world in order to store and process such large amounts of data will be required.

An example of the increasing bandwidth demands of science and academia are: US Energy Science Network (ESNet) traffic has grown 70% per year between 1992 and 1999, and 100% growth upto 2004. Stanford Linear Accelerator Center's (SLAC) network traffic has been growing at a average annual rate of 80% with a prediction of reaching 2 Terabits/sec by 2014. LHCNet

Year	Production	Experimental
2001	0.155 Gbps	0.622-2.5 Gbps
2002	0.622 Gbps	2.5 Gbps
2003	2.5 Gbps	10 Gbps
2005	10 Gbps	2-4×10 Gbps
2007	2-4×10 Gbps	40-100 Gbps
2009	40-100 Gbps	5×40 Gbps or 20-50×10 Gbps
2011	5×40 Gbps or 20×10 Gbps	25×40 Gbps or 100×10 Gbps
2013	Terabit	Multi-Terabit

Table 2.1: The expected traffic volumes of the ATLAS LHC Experiment compared to that of commodity Production volumes [New05].

which connects the US and Centré Européenne pour la Recherche Nucléaire (CERN) has grown from a 9.6Kb/sec link in 1985 to a 10Gbit/sec link today.

The predicted network resource requirement of the ATLAS experiment is shown in Table 2.1. It shows that within ten years of expected data collection at the LHC, the amount of data expected to be transfered per year will grow from petabytes to extabytes of traffic per year.

The flow of the data from the detector to physicists must be well coordinated. The MONARC [LN00] group defines a hierarchy of Tiers from which the data stored at the facilities at a detector should be distributed around the world for storage and processing of these huge datasets.

The member sites of LHC experiment are categorised into Tiers, depending on their ability to supply data to the user. The intention is that the total resources at each Tier will be approximately the same. The figures quoted here are for ATLAS, and similar resources will be needed by the other LHC experiments. The Tiered hierarchy is shown in Figure 2.1, which also shows the expected traffic volumes between Tiers.

- Tier-0

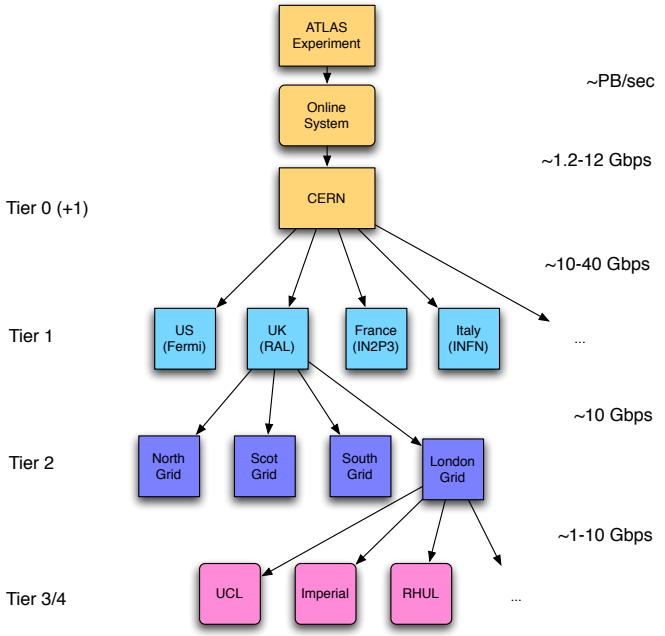


Figure 2.1: The MONARC hierarchy of Tiers for Atlas, with estimated initial data transfer rates between the different tiers.

There is only one Tier-0 site. For LHC this is CERN, where the data is acquired from the experiment, and initially stored. The first data reconstruction occurs here, and CERN shares the work of the Tier-1 sites.

- Tier-1

  Tier-1 Regional Centres will service a nation, or a group of nations. They are expected to replicate as much of the data stored at CERN as possible in order to facilitate access to the data with approximately 10 sites worldwide. Typically, the network provision between the Tier-0 site and Tier-1 sites should be high, deploying either multi-gigabit links and or DiffServ and or MPLS solutions to guarantee flow protection.

- Tier-2

  Tier 2 centres will service single nations or regions, caching popular data in their local storage. There will be approximately 60 sites worldwide.

- Tier-3 / 4

  The local computing resources at member institutions make up Tier-3 of the model, with Tier-4 consisting of individual machines.

The initial storage and processing of event data will be located at the Tier-0 site. In the case of the LHC, this will be CERN. The data is then replicated to various Tier-1 centers around the world where they are further processed. Approximately sixty Tier-2 regional centers, each serving a medium-sized country, or one region of a larger country (e.g. USA, UK, Italy, etc), will then replicate data from the Tier-1 centers. Physicists will then typically access and further analyse data using one of hundreds of Tier-3 and Tier-4 workgroup servers and/or desktops.

The design of such a system ensures that a physicist should not need to wait more than 10 minutes [LN00] to transfer relevant data for analysis.

The estimated data rates between Tiers are based on having a very well-ordered, group orientated and scheduled approach to the transfer of data. As such, the data transfer rates are expected to actually be higher in practice.

An important development in being able to realise the transport requirements of such data intensive science applications is that of a shift from the typical shared 'best-effort' services of the Internet to dedicated connection oriented end-to-end paths to facilitate the separation (and hence impact) of such large scale traffic to that of standard Internet traffic. This is apparent in projects such as Terapaths [Gib04], OSCARS [Guo04] and LHCnet

[BMMF05] where new technologies such as MPLS [RVC01, AMA+99] and DiffServ [BBC+98] are being implemented across production local and wide area networks.

This will allow the provision of network resource to form Hybrid Networks, whereby end-users (typically large collaborations such as the LHC) will form their own private (virtual) networks for the movement of large scale datasets.

However, this paradigm comes at at price as the overheads of installation, configuration, support and maintenance must now be conducted by the institutions rather than leased from providers. Therefore the utilisation of such network resources are very important in order to provide a sufficient return on investment.

2.4 Data Transfer Requirements

2.4.1 Tier-0 to Tier-1

The typical network connections between the Tier-0 and Tier-1 sites are meant to be well provisioned to support efficient replication of data across to different geographical Tier-1 sites. As Tier-1 sites will be located worldwide, latencies will range from pan-European (around 20msec) to world wide distances (around 200msec). As most of the infrastructure and investment will be located between the Tier-0 and Tier-1 sites, there will be (at least initially) a lot of spare capacity, and technologies such as MPLS/Diffserv may be utilised to offer a greater Quality of Service (QoS) than ordinary traffic [Gib04, Guo04] and also provide class protection to reduce the affect of such traffic upon ordinary Internet traffic. Similarly dedicated light paths may be leased for the sole purpose of continuous data transfer.

2.4.2 Tier-1 to Tier-2

Traffic between Tier-1 and Tier-2 sites are likely to be relatively local, with latencies in the range of 10msec to 50msec. Typical data transfers are likely to pass through existing infrastructures such as that of Janet, Abilene etc between the Tier-1 and Tier-2 sites. Currently most of these networks are built upon 10Gb/sec backbones with relatively light loading (typically around 10% [ST]). If the network is shared (Internet), then it will be important to maintain fairness between the LHC and commodity traffic.

2.4.3 Tier-2 to Tier-3/4

The 'last hop' of the LHC data transport will be from the Tier-2 centres to that of local workgroup machines and desktop computers where physicists will perform the analysis of data. As such, this will typically be what users will perceive as the raw performance of the data transfer. Due to the locality of the Tier-2 to Tier-3/4 sites, latencies are likely to be relatively small (less than 20msec). However, the networks at this edge are generally likely to be bottlenecked by the local MAN and campus networks, and will likely need to be shared between all users on the campus and or local area and thus fairness becomes an important issue. Therefore, is will be important to sustain a balance between affecting other network users (and the network in general) and achieving high throughput to enable the users to do their work.

2.4.4 Summary

Table 2.2 shows a matrix of TCP performance metrics according to the type of end-to-end connection with a relation to that of the Tiers in the MONARC architecture.

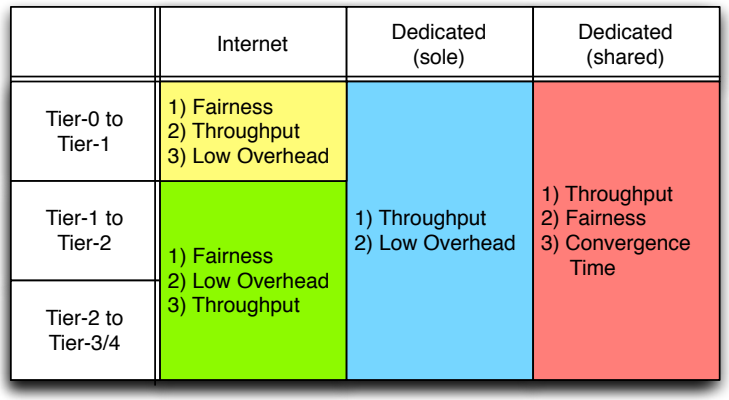


Table 2.2: Overview of Transport requirements for different Tiers based upon the type of end-to-end connection.

For an in-depth discussion of the definitions of the various metrics presented, please refer to Chapter 7.

When the large scale data transfers are competing over the commodity Internet, it will be very important that the transfer will not adversely affect the existing traffic on the network, therefore the most important metric to consider is that of Fairness. In order to aid efficient replication to the Tier-1 and Tier-2 locations, throughput becomes the next important factor to consider such that very large scale caching will not be required at the Tier-0 and Tier-1 sites respectively. It will also be important to minimise the amount of data that needs to be retransmitted as such traffic on the commodity Internet may lead to congestion collapse [FF99]. As such it is important for the transport algorithms to have low overhead. This is especially important at the bottom of the hierarchy due to the high possibility of bottlenecked systems which are more likely to be located in Tier-3 and Tier-4 locations (due to less hardware and or staff investment in tuning and optimisation).

When considering the move to dedicated circuits for the transport of LHC

data, the most important metric will be to obtain high throughput in order to maximise return on investment. Associated with this will be the need to maximise throughput by sustaining a low overhead as retransmissions will reduce goodput. As there will be no competing traffic on dedicated circuits the order of importance of the metrics is the same throughout all Tiers.

If the dedicated link were to be shared between a few network users (e.g. parallel streams, or simultaneous replication of data to a few sites across the same dedicated network[1]), then the sharing of the throughput between the flows is important. As such, fairness and to a lesser degree, the *convergence times* become a factor.

2.5 Summary

An outline of the methods and data rates of particle physics experiments, such as the LHC were given. The ATLAS experiment was presented and the MONARC hierarchy was discussed. In particular, the conservatively estimated data rates required for the replication and analysis of such data is expected to be at least an order of a magnitude greater than current experiments. As such, the idea of Hybrid networks was presented in order to facilitate high transfer rates between Tiered sites, yet maintain segregation between flows in order to prevent/reduce the impact upon commodity Internet Traffic.

The relative importance of various evaluation metrics were also presented within the LHC application and four unique areas were identified.

[1]Due to the continued development of reliable multicast, the HEP community does not currently consider it as a viable option.

Chapter 3

Background

Grid technologies [FKNT02, Fos01, FKT02] is considered as a major component of enabling world-wide collaborative science, and of course, the Large Hadron Collider (LHC) project.

Grid technologies are built upon existing technologies. They use Core Middleware [FK97] that acts solely as a way of communicating to the underlying Fabric [Fos01] to conduct Grid operations.

This Chapter looks into the medium on which Grid communication relies: the Internet, and identifies that the TCP protocol imposes a bottleneck in terms of high throughput data transfer.

3.1 Data Intercommunication

The need to transport data stored on computer systems around the world is of fundamental importance in the application of the Grid. Without the

movement of data to be processed, there would be very little point in having a distributed Grid.

Fortunately, with the standardisation of the Internet [IET], much work has already been done to enable seamless transfer of data from one system to another. In this section, the paradigm of data transfer across the Internet are discussed and the performance of the various layers that enable data inter-communication are investigated.

TCP/IP

The TCP/IP protocol suite is commonly used by all modern operating systems. TCP/IP [Ste94] is designed around a simple four-layer scheme. The four network layers defined by the TCP/IP model are as follows.

Layer 1 - Link This layer defines the network hardware and device drivers that refer to the physical and data link layers of OSI [Tan96].

Layer 2 - Network This layer is used for basic communication, addressing and routing. TCP/IP uses IP and ICMP protocols at the network layer and encompasses the network layer of OSI.

Layer 3 - Transport Handles communication among programs on a network. TCP and UDP falls within this layer and hence is also equivalent to the transport layer in OSI.

Layer 4 - Application End-user applications reside at this layer and represents the remaining layers of OSI: the session, presentation and application layers.

There are two main transport protocols that are commonly used today: User Datagram Protocol (UDP) [Pos80] and Transmission Control Protocol

(TCP) [Pos81b]. The former offers a unreliable way of transferring information, i.e. it does not know explicitly that a sent packet has been received, whilst the latter offers a reliable service - for every packet of information that is sent and consequently received by the receiving host, some kind of acknowledgment is sent by the receiver and received by the sending host.

UDP can offer a greater raw performance as it does not require the extra overhead of acknowledging information and maintaining state. However, as UDP offers no kind of signaling from the network to discover the current network conditions, it can be potentially dangerous if used maliciously (Denial of Service attacks [MVS01, HHP03]).

As the Internet is based mainly on the connectionless communication model of the IP protocol, in which UDP and TCP segments are encapsulated before transfer across the internet, IP has no inherent mechanisms to provide delivery guarantees according to traffic contracts and hence mechanisms to reserve network resources have to be implemented via other means (See Section 9.2). Because of this, IP routers on a given data path from source to destination may suffer from *congestion* when the aggregated input rate exceeds the output capacity.

3.2 Network Monitoring

3.2.1 Networking Metrics

Networking performances are broadly classified into both latency and throughput. Moreover, other metrics such as the *internet path* (e.g. with `traceroute`) [Jac89], the *connectivity* (whether the Internet host is reachable) [MP99] and the *jitter* (AKA interpacket-delay variance) [DC02] may be important.

Whilst latency is an important metric, especially when applied to applications such as voice or video conferencing, it is essentially constrained by the physical (relative) location of the two hosts[1]. As the majority of the Internet is a shared resource with Best Effort [CF98] scheduling of resources, consecutive `pings` may experience different latencies and hence cause jitter. Similarly, the traversal of packets through different paths[2] may also increase the jitter, so much so that packets may arrive 'out-of-order'.

The transfer of bulk amounts of data involve the movement of many packets of data. As such the microscopic effects of jitter and delay will also affect the macroscopic effects of bulk data transport. Also, as the raw rate at which data can be transfered is often a useful and readily measurable metric, the issue of bandwidth monitoring has become an important indication of the performance of many Internet applications. Unlike latency, which is limited by physical constraints, such as the speed of light, the capacity (optimal throughput) of hardware is limited by the clock frequency and protocols used to control the medium - as defined by OSI Layer 2.

Two bandwidth metrics that are commonly associated with a path are the *capacity* and the *available throughput* [LTC+04]. The capacity is the maximum throughput that the Internet path can provide to an application when there is no competing traffic load (cross traffic). The available throughput, on the other hand, is the maximum throughput that the path can provide to an application, given the path's current cross traffic load. Measuring the capacity is crucial in calibrating and managing links. Measuring the available bandwidth, on the other hand, is of great importance for predicting the end-

[1]Delays caused by queuing at router and switches may also affect latency.

[2]Both between routers and within routers themselves depending on router design/configuration.

to-end performance of applications, for dynamic path selection and traffic engineering. A more user centric metric is the *achievable throughput* which defines the actual throughput through the Internet of a real application.

For example, `FTP` [PR85] is a popular application level protocol designed to transfer files across the Internet. Assume that the end-to-end path capacity is 10Mbit/sec; because of competing users (because it's a busy link) the achievable throughput may only be 3Mbit/sec (as 7Mbit/sec is being used by other users). However, upon initiating the `FTP`, the user may only manage a transfer rate of 1Mbit/sec in steady state - which is the achievable throughput due to the complex interaction of network buffers and user traffic patterns.

3.2.2 Test Methodology

Assuming adequate hardware and driver configurations, back-to-back tests of the transfer of data should be capable of high speeds. However, the TCP/IP stack is composed of two different transport protocols that regulate the way in which data is injected into the network. In order to determine the performance limitations on inter-communication, the upper layers of the OSI and TCP/IP stacks are investigated and these transport protocols are tested.

As mentioned previously, UDP is a stateless protocol that allows fine control of packet data control. However, the transfer of data using UDP is not recommended due to various reasons as outlined in Chapter 4. Therefore, as most Internet applications work with the TCP transport protocol, it is important to compare and contrast the performance of these two protocols.

Two hosts were identified to be the sender and the receiver of active network traffic in order to determine the achievable throughput through a real

Internet environment. The hosts were equipped with Gigabit NIC cards and the network path consisted of 15 hops as shown in Figure D.1 with a path bottleneck capacity of 622Mbit/sec. The machines were chosen to give a representative transfer rates achievable by real physics users wishing to transfer data from CERN, in Switzerland, to Rutherford Appleton Laboratory, in England, UK.

In order to generate UDP throughput traffic, `UDPMon` [HJ] was used and configured to a inter-packet wait time of zero seconds (i.e. back-to-back packets). For the TCP traffic, `iperf` [TQD+03] was used to simulate the transfer of data and appropriate provisions for a large socket buffer size was used with a sufficiently large socket buffer size of 8MB at both ends (See Section 5.1.1).

Tests were run over an entire week, with randomised start times. UDP flows were conducted with 1,000 full sized packets and TCP was ran for 30 seconds. These values were found to be sufficiently long to reach steady state.

3.2.3 Results

Figure 3.1 shows the results of the tests and shows that whilst UDP was able to achieve almost line rate with a mean throughput of 493Mbit/sec, TCP was only able to achieve about half of that with a mean throughput of 195Mbit/sec. Therefore, there is a clear discrepancy in the performance at the transport protocol layer.

Hardware performance bottlenecks between the two end-hosts can be assumed to be negligible due to the successful transmission of UDP packets at high speeds.

Similar tests to CERN from Manchester showed only marginal improve-

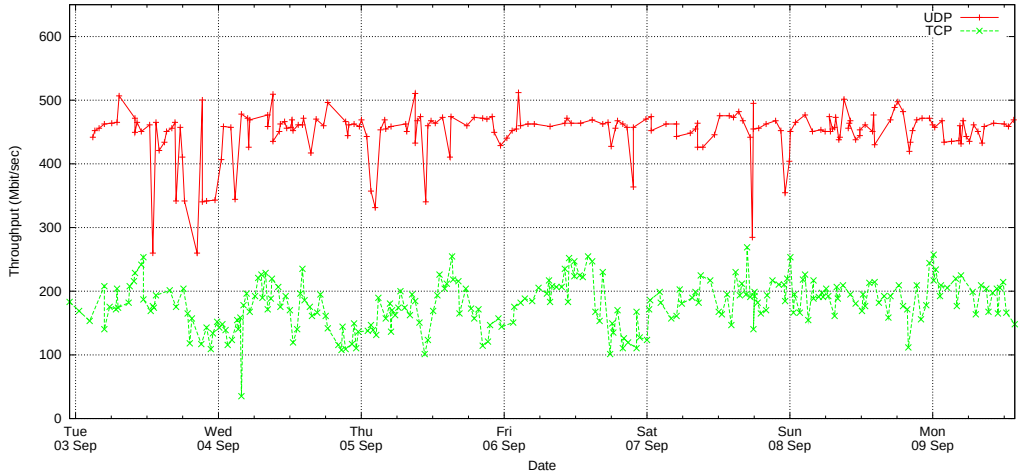


Figure 3.1: Throughput across production networks using UDP and TCP Transport Protocols between CERN and RAL in September 2002.

ments in the TCP throughput (approximately 30Mbit/sec more) with approximately the same UDP throughput. This suggests that there is a performance problem with TCP in being able to achieve the complete utilisation of the path from CERN to the UK.

As such tests are intrusive and disruptive to network stability (See Section 4.1), `UDPMon` [HJ] was only configured to transfer 1000 packets in order to calculate the UDP throughput. Whilst this is sufficient to determine the transient state of the network, it may not provide a deterministic indication of the state of the network for the 30 seconds that the TCP flows were transferred for. As such, the longer duration of the TCP tests (compared to the UDP tests), may be more prone to the effects of network cross traffic. However, shorter TCP would not reach steady state due to the dynamics of slow-start (See Section 4.3).

However, the transfer of UDP packets may provide an indication of the router queue occupancy along the network path. As packets are typically

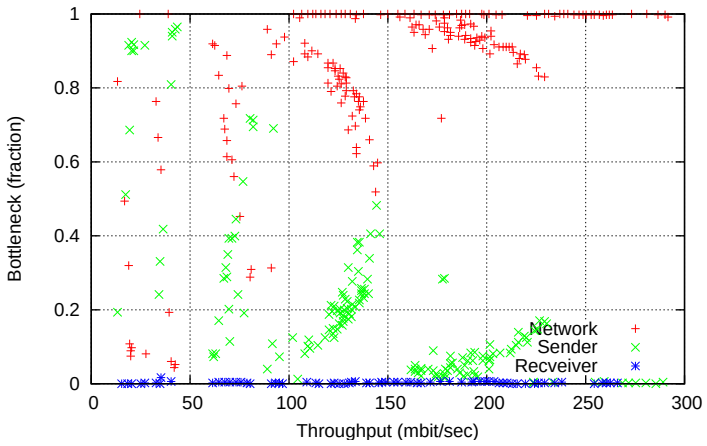


Figure 3.2: Bottleneck limits of TCP WAN transfers between CERN and RAL as reported by Web100.

only lost when the network queues overflow, the consistent high throughput of the UDP packets suggest that the average queue occupancy is low. Given this information, it is expected that TCP should also be able to sustain high throughput (on average).

To investigate the details of TCP transfers, all TCP results were passively monitored using the Web100 [MHR03] framework in order to determine the possible bottlenecks on the system. Web100 implements a facility whereby the performance of TCP can be determined to be bottlenecked by either the receiver, the sender or by the path in between. These variables are presented in a 'triage' whereby the fraction between the three bottlenecks are logged.

Web100 is a Linux kernel patch that enables the in-depth logging of TCP parameters. Figure 3.2 shows the result of the Web100 triage [MHR03] for numerous network transfers (including those not presented in Figure 3.1) for the network path shown in Figure D.1. The triage determines the bottleneck associated with each transfer in terms of a ratio between the network, the sender, and the receiver.

It clearly shows that that the network component is the overriding performance bottleneck for TCP.

The structure of the graph show that there are certain operating regions from the results at which the points 'cluster'. This was found to be a result of effect of socket buffers (See Section 5.1.1) which limit the physical memory allocation to TCP, and hence the achievable throughputs.

3.3 Summary

Grid fabric components of storage and network hardware are investigated and presented in Appendix A. In summary, the tests demonstrate that the hardware potential for transferring data at greater than Gbit/sec rates is perfectly feasible.

Test transfers across real production networks were conducted were conducted using both the stateless UDP protocol and the standard TCP protocol using memory-to-memory data copying between RAL in the UK and CERN in Switzerland.

It was discovered that the UDP protocol consistently performed a lot better than TCP, achieving on average about a factor of 2 difference in throughput. As much of the Internet traffic relies on the reliable replication of data across sites, TCP is a fundamental protocol in the performance of all Internet-based applications. As such, further investigation as to why TCP is incapable of achieving high throughput transport is warranted.

Chapter 4

Transmission Control Protocol

The movement of bits across a network requires both hardware and software to co-ordinate the copying of data from one machine to another. The dynamics and performance of this interaction is presented in the previous chapter and in Appendix A. Real world transfers of data were conducted. It was found that the performance of TCP from Switzerland to England was only half that of the UDP protocol.

The Transmission Control Protocol (TCP) [Pos81b, Bra89, APS99] is the most widely used transport protocol in the Internet and provides applications with reliable, byte-oriented delivery of data on top of the stateless Internet Protocol. In this chapter we look more closely at TCP to find out why it has become the de-facto transport protocol on the Internet.

4.1 Overview

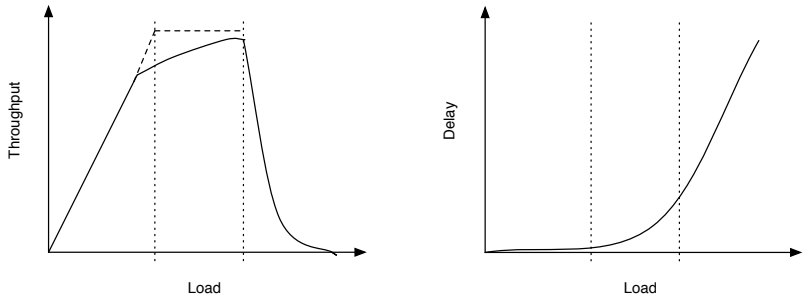
The Transmission Control Protocol was originally devised by Postel in 1981 [Pos81b]. Designed to enable inter-communication between disparate systems, it is encapsulated above the stateless Internet Protocol [Pos81a].

IP routers connected on multiple network paths are shared resources which may suffer from *congestion* when the aggregated input rate at any networking node exceeds the output capacity. As there is no global signalling mechanism to control the rate at which end-nodes send data into the network, congestion can occur. Consequently, flows tend to experience varying network conditions that affect the original traffic profile from the source. This is known as Best-Effort servicing as each individual node tries its best to forward packets without any service guarantees upon its performance.

Should all Internet users be using UDP as their primary transport protocol, the aggregated input rate at any router would be greater than the maximum output rate and overloading of Internet resources would take place and potential Congestion Collapse [Nag84] would occur.

When congestion occurs, the amount of useful work done is diminished to such a degree that the throughput declines and the network is termed as undergoing congestion collapse. [FF99] and [MR91] outline the need for mechanisms to prevent congestion collapse.

Congestion collapse is shown in Figure 4.1. It shows that as the sending rate increases the useful work done increases linearly up to a point where it reaches the network bandwidth. Past this point the throughput saturates and the link pipes are filled and packets get queued at the routers. Similarly, the delays experienced by the traffic increase due to the buildup of routers' queues, and soon after the flows starts to lose packets. The loss can be re-



(a) Throughput (b) Delay

Figure 4.1: Congestion Collapse

covered by the sending hosts retransmitting them, but part of the bandwidth will be wasted due to retransmissions.

At the turning point where the flow starts losing packets the amount of useful work done by the network decreases dramatically and connectivity diminishes completely; congestion collapse has occurred.

There are many types of congestion collapse [FF99]. Two of the most common forms is congestion from the retransmission of packets already in transit or that have been received (*classical collapse*), and that arising from the transmission of packets that are subsequently dropped by the network (*congestion collapse from undelivered packets*).

Originally devised to simply transport data reliably across networks [Pos81b], TCP was adapted in 1988 by Van Jacobson [Jac88] to avoid network congestion collapse and to provide network fairness between network users. Much research, development and standardisation since its incarnation has led TCP to be widely used in the Internet and it is considered as the de-facto standard transport-layer protocol.

The achievements of TCP are most evident by the fact that the protocol has changed little over the last two decades. With the advent of fibre optics

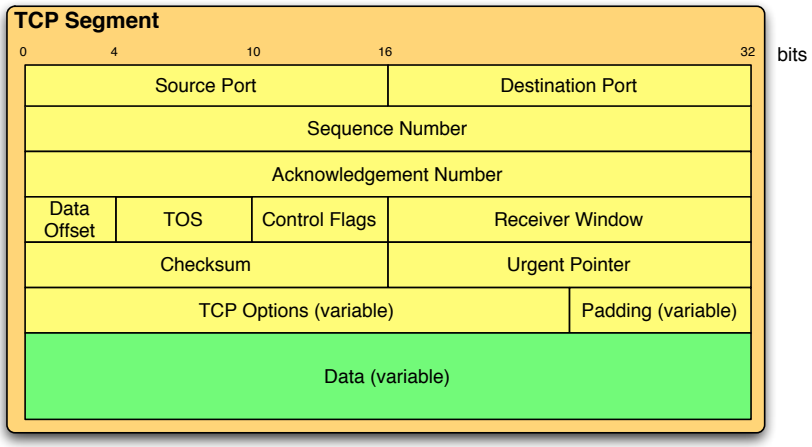


Figure 4.2: TCP header format.

and satellite communications, technological development has increased the size of the Internet by orders of magnitude in terms of size, speed, load and connectivity - and all this in the last couple of decades [CO01, CO99]. Even with the wide range of network conditions present on the Internet, TCP has been able to utilise and effectively share network resources between Internet Users. More importantly, it is generally believed to have prevented severe congestion collapse during this time [FF99, Nag84].

4.2 Protocol Description

The TCP header is shown in Figure 4.2. The source and destination ports allow the hosts to identify multiple concurrent TCP flows at once. The sequence and acknowledgment numbers facilitate reliable transport (See Section 4.2.2). The Receiver Window field is used for flow control (See Section 4.2.3). The Data Offset field provides a pointer to the start of the Data field as TCP Options [Pos81b] are variable in size. The size of the Data field

is dependent upon the Maximum Segment Size (MSS) of the TCP segment and constrained by the MTU of the network path. The variable Padding field at the end of the TCP Options is used to ensure that the TCP header ends at a 32-bit boundary. Control Flags determine the type of TCP segment (e.g. SYN, ACK, SYN-ACK, FIN, reset, or urgent) and denoted by either a true or false bit for each type of segment (e.g. a SYN packet will have the SYN bit marked, whilst a SYN-ACK will have both SYN and ACK fields marked). The TOS field determines quality of service and enables packet level service differentiation. The entire TCP segment is check-summed to facilitate the detection of header and/or payload corruption.

4.2.1 Connection Initialisation

TCP connections are established between two hosts via a three-way handshake. According to [Jac88], the sending and receiving of data after the initialisation of the TCP connection is bounded by a packet conservation principle; when this principle is obeyed, congestion collapse from excessive data being put into the Internet will become the exception rather than the rule.

The principle of packet conservation utilises the idea that data packets cannot be placed into the network any faster than that at which acknowledgments (*ack*'s) by the sender of the data are received. This is called *ack*-clocking and is an important property in maintaining a stable and even data flow [VH97, Jac88, BPS99].

In the following we will give an overview of the features and implementations that have made TCP such a success.

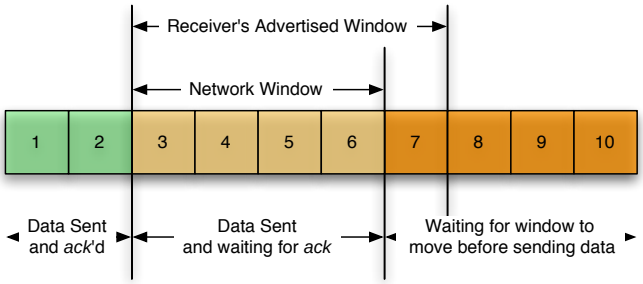


Figure 4.3: Sliding Window of TCP.

4.2.2 Reliable Data Replication

To ensure reliable in-order data replication between machines, a TCP sender must maintain sufficient history of what data has been received by the TCP receiver. The implementation of an acknowledgment system in TCP offers such a mechanism by explicitly notifying the sender of what the receiver has received.

This history is provided by implementing a window of bytes on the sender which are currently 'in-situ' or packets in flight. Each byte of data is assigned a unique sequence number and the sequence number corresponding to the start of the data payload of each TCP packet is embedded in the TCP header.

The left-hand side of the window, `snd_una`, is advanced when the sender receives an acknowledgment from the receiver. *ack*s are cumulative in the sense that the acknowledgment number in the header of each *ack* confirms all bytes up to the given sequence number.

As data is not always delivered to the TCP receiver in a continuous way (the network[1] can lose, duplicate or re-order packets) any data packets that are not sequentially received at the receiver are called out-of-order data.

[1] And potentially any layer directly underneath the transport layer.

As a response to out-of-order segments, the TCP receiver sends duplicate acknowledgments, *dupack*s, that carry the same acknowledgment number as the previous *ack* it had sent - hence this does not change `snd_una`.

A sequential receipt of a number of *dupack*s signifies that a data segment has potentially been lost in the network, and therefore the sender should send the data segment (the left most side of the sliding window) again.

In combination with retransmission timeouts (RTO) (See Section 4.5.1) that will resend segments should no *ack*s that advance `snd_una` be received within this RTO, *ack*s provide reliable data delivery [Bra89] by informing the sender of when data has (and implicitly, has not) been received by the receiver. Similarly, when new data is sent into the network, the right most pointer, `snd_nxt`, advances to the right. Therefore, the window 'slides' across the range of sequence numbers as the transfer takes place. Through the evolution of the data transfer, this window will *slide* across the entire byte range of the relevant data to transfer and hence is known as a 'sliding window'.

The macroscopic result of this is that for each and every TCP connection, this window has to be maintained as part of the connection state in memory in order to be able to keep track of the number of packets in-flight. The Bandwidth Delay Product (BDP) defines the relation between the throughput b, the round-trip latency T, and the size of the sliding window w required.

$$b = \frac{w}{T} \tag{4.1}$$

The relation between w and the number of packets is defined by:

$$w = n \times s \tag{4.2}$$

Where n is number of packets in flight, and s is the size of each packet as limited by the MTU and MSS. As such, should the physical memory assigned to the sliding window be restricted, then the number of packets in flight is restricted and consequently the throughput is also restricted.

Another way of thinking about the BDP is that for a network capacity of C (i.e. $b = C$), and an end-to-end latency of T, the required window in order for a TCP flow to saturate the path is $C \times T$. This is only valid when there are no competing flows as the presence of network cross traffic is that the available bandwidth of the network is less than C. However, the presence of cross traffic may also increase the latency of the flow, and therefore there is a balancing effect (of the decrease in bandwidth, but an increase in latency). A full account of this is given in [JPD03].

4.2.3 Flow Control

To prevent a fast sender from overflowing a slow receiver, and hence causing losses at the receiver (and wasting network resources), TCP implements flow control [Tan96]. In order to prevent such an occurrence, in every *ack* the receiver advertises to the sender its 'receiver window' ($rwnd$) which represents the number of bytes available for buffering at the TCP receiver before it is processed and sent up to the receiver Application.

Therefore, if the TCP receiver cannot process the incoming data at the speed which is being sent, the value of $rwnd$ sent will get smaller as the socket buffer at the receiver fills.

Through the advertisement of $rwnd$, the TCP sender can prevent overflowing the receiver by limiting its maximum allowed window size for transmission. This mechanism provides a simple, yet elegant way of preventing

excessive transmission of data which will be lost at the receiver.

4.3 Congestion Control

The implications of congestion collapse led to the first proposal for a transport protocol which will help prevent congestion collapse [Jac88].

TCP utilises another window to determine the dynamic flow control required of the heterogeneous networks and hence an estimation of the available capacity in the network.

This congestion window (*cwnd*) is defined as the number of allowed segments (or bytes) sent but not yet acknowledged (packets or bytes in transit). As such, it is analogous to controlling the rate at which packets are allowed into the network along the bottleneck of a given network path.

Whilst congestion control was not originally part of the original TCP specification [Pos81b], it has become part of the standard TCP specification since the introduction of TCP Tahoe (See Section 4.5.1).

The two specific algorithms that control the evolution of *cwnd* are defined as follows:

4.3.1 Slow Start

The purpose of the slow start algorithm [Jac88] is to get the *ack* clock running and to determine the available capacity in the network. It is a mechanism to quickly find an operating point at which TCP can work. Therefore it is invoked at the start of every TCP connection after the initial TCP handshake.

Starting with a conservative *cwnd* of one segments[2], the TCP connection

[2]Recent suggestions state that TCP *may* set its initial *cwnd* to two segments in light of the increase in link speeds [AFP98].

sends out as many segments as it can ensuring that the number of packets in flight does not exceed *cwnd*. After approximately one RTT, the TCP sender should receive an acknowledgment informing receipt of the data from the receiver. For every non-duplicate *ack* received,

$$cwnd \leftarrow cwnd + 1 \tag{4.3}$$

During slow start, this algorithm is equivalent to an exponential increase in the *cwnd* size as it effectively doubles every RTT as one more packet is sent out for every non-*dupack* received. This increase of *cwnd* is quite aggressive. However, it enables the TCP connection to quickly reach a stable point for congestion avoidance (See Section 4.3.2) to take place.

Slow start ends when a segment loss is detected or when the congestion window reaches the slow start threshold, *ssthresh* (See Section 4.3.3).

4.3.2 Congestion Avoidance

Under the packet conversation principle, a TCP connection at equilibrium will send new packets with the rate at which it receives *ack*s. Under ideal circumstances, this rate should reflect the rate packets are served by the slowest link of the entire path, also known as the 'bottleneck link'.

When a sender is operating at equilibrium, i.e. in steady state, it should be able to adapt to changes in the condition of the path. More importantly, if more connections (say from other users) start to use part of the path, any one TCP flow should 'back-off' its transmission rate and be fair to other flows. Similarly, should a connection terminate then the sender should attempt to utilise the extra bandwidth. This is exactly what congestion avoidance [Jac88] does, and is implemented through a simple algorithm such that it will

probe for extra bandwidth by increasing its sending rate slowly, falling back to a 'safe' sending rate should loss or excessive delay of an *ack* (an implicit indication congestion) occur.

The implementation of such an algorithm is given below:

$$\mathrm{A}CK: \quad cwnd \leftarrow cwnd + \frac{1}{cwnd} \tag{4.4}$$

$$\mathrm{L}OSS: \quad cwnd \leftarrow cwnd - \frac{1}{2} \times cwnd \tag{4.5}$$

Upon receipt of an acknowledgment, when in congestion avoidance, the sender increases the *cwnd* by $\frac{1}{cwnd}$. Therefore, after about *cwnd* of *ack*s, the sender will be able to send one more packet due to this increase. This mechanism allows TCP to 'probe' the network to determine if it has more resources; if it does, then it will make use of the extra bandwidth by continuing to increase *cwnd*; if not, after approximately RTT the sender should either be able to determine packet loss from *dupack*s sent by the receiver or through a RTO soft timer expiration (See Section 4.5.1).

Should a loss be detected, then the congestion avoidance algorithm of TCP reduces *cwnd* by half[3] and hence reducing the number of packets in-flight from the connection. This will reduce the resources used by this TCP connection and helps to prevent congestion collapse from occurring.

The value of *cwnd* is never allowed to be less than 1 packet.

It must be noted that the requirement to prevent congestion collapse is that all end-to-end connections should have the same form of reactive congestion control behaviour. Therefore, the requirement of preventing congestion

[3][APS99, Pos81b] actually specify that $cwnd \leftarrow FlightSize/2$, where $FlightSize$ is the amount of data sent, but yet unacknowledged. For simplicity, it is assumed that the $FlightSize$ is the same as the *cwnd*. This assumption is valid only when TCPs *ack* clock is not broken.

collapse is a distributed one whereby all transport protocol control algorithms should react upon indication of congestion by decreasing their sending rates.

The algorithm that governs Congestion Avoidance is referred to as Additive Increase Multiplicative Decrease, (AIMD) [CJ89], due to the form of Equations 4.4 and 4.5.

4.3.3 Slow Start Threshold

The general characteristics of the TCP algorithm are an initial, relatively fast scan of the network capacity using Slow Start, followed by a cyclic adaptive behaviour of Congestion Avoidance that reacts to congestion by reducing its sending rate, and then slowly increasing the sending rate in an attempt to stay within the area of maximal transfer efficiency.

In order to determine the balance between the aggressive algorithm of Slow Start and the steady-state congestion probing algorithm of Congestion Avoidance, another variable, *ssthresh* is introduced and is used as follows:

```
if  cwnd < ssthresh
   do slow start
else
   do congestion avoidance
```

This slow start threshold, *ssthresh*, is typically set to a large initial value and updated upon loss detection. In other words, after loss, the value of *ssthresh* is set to half of the number of packets in-flight.

Through continuing updates of the value of *ssthresh*, the TCP connection is able to recover to a stable operating region of congestion avoidance quickly using slow-start when a *RTO* occurs and then continue its conges-

tion avoidance regime from a 'safe' operating at half the rate at which loss occurred.

4.4 Retransmission Timeout

As TCP requires the receipt of acknowledgments in order to slide the TCP window and thus maintain the flow of data between the sender and receiver, should *ack*'s not be received then the TCP sender will indefinitely stall.

In order to rectify this, TCP implements a Retransmission Timeout (RTO) which is a soft timer which is re-initialised after the receipt of an *ack*. It is assumed that an *RTO* expiry (i.e. a period where no *ack*s are received) is equivalent to packet lost and therefore the missing segment should be re-transmitted to facilitate reliable delivery and progress movement of the TCP window.

The delivery of each segment of data may experience differing latencies; it is therefore important to have a good RTT estimator in order to determine an appropriate RTO.

[Jac88] presented a calculation of *RTO* by applying a filter such that it takes into account the variation of the average RTT, in fact this exponentially weighted estimator devised for Tahoe is still in use today. The RTO is calculated using a smoothed estimate of the RTT, SRTT, and a variance estimate, RTTVAR [PA00].

$$SRTT \leftarrow SRTT + \frac{1}{8}(RTT - SRTT) \tag{4.6}$$

$$RTTVAR \leftarrow VRTT + \frac{1}{4}(|RTT - SRTT| - RTTVAR) \tag{4.7}$$

$$RTO \leftarrow SRTT + 4 \times RTTVAR \tag{4.8}$$

This enables spurious packet delays to be absorbed into the time-out to facilitate a good estimation of the *RTO*.

RTT measurements resulting from retransmitted segments are not included [KP87] to avoid false positive RTO calculations.

Standard TCP implementations (such as TCP Tahoe - See Section 4.5.1) only time-stamp a single packet per RTT to reduce the CPU load, and thus only a single RTT measurement can be gathered per RTT which will limit the robustness of the above algorithm. More accurate RTT samples can be gathered by using using the TCP Timestamp option (See Section 5.2.1) to facilitate a more robust RTO calculation.

4.5 TCP Variants

Whilst the basic design of TCP has changed little since its incarnation, various adaptations to the basic flow recovery mechanism have enabled TCP to adapt as the Internet has evolved.

The original specification in [Pos81b] implements only the sliding window and flow control of TCP. Also a very basic RTO estimator was used. This section describes the major implementations and alterations of the TCP algorithm to the current state.

4.5.1 TCP Tahoe

Designed in 1988, TCP Tahoe by Van Jacobson [Jac88] extends the original TCP by Postel [Pos81b] with five new mechanisms that have become de-facto TCP mechanism: slow start, *ack* clocking, window dynamic adjustment, fast retransmit and round-trip time variance estimation. All of these new

mechanisms are still widely in use today in the all TCP implementations.

TCP Tahoe assumes that congestion signals are represented by lost segments and that losses due to packet corruption are much less probable than losses due to buffer overflows on the network.

As there is no explicit notification of packet loss through the *ack* mechanism, TCP Tahoe introduces the notion whereby packet loss can occur if:

A TCP soft time-out occurs such that no *ack*s have been received in a period *RTO* since the last 'normal' *ack* was received. Therefore, the *ack* clock has stopped. TCP Tahoe implements a more accurate calculation of *RTO*.

Receipt of 3 *dupack*s The TCP connection is still active and there is still *ack* clocking. A packet may have been lost but all successive packets were delivered successfully.

The refinement of an accurate round-trip time estimator was also introduced with TCP Tahoe (See Section 4.4).

The occurrence of an *RTO* timeout is taken as a signal of severe network congestion and therefore the rate at which the TCP should send more segments should be decreased to prevent network collapse. As such, the TCP flow reinitiates the TCP connection in slow start so that the network conditions can once again be probed to re-initialise the *ack* clock. By setting *cwnd* to 1 packet, TCP Tahoe starts sending from a very low rate and only if the network is capable of the extra traffic will the *ack* clocking reinitiate.

Fast Retransmits [Jac88] were proposed to reduce the long idle periods of time during which the TCP on the sending host waits for a timeout to occur. Fast Retransmit is a mechanism that sometimes triggers the retransmission

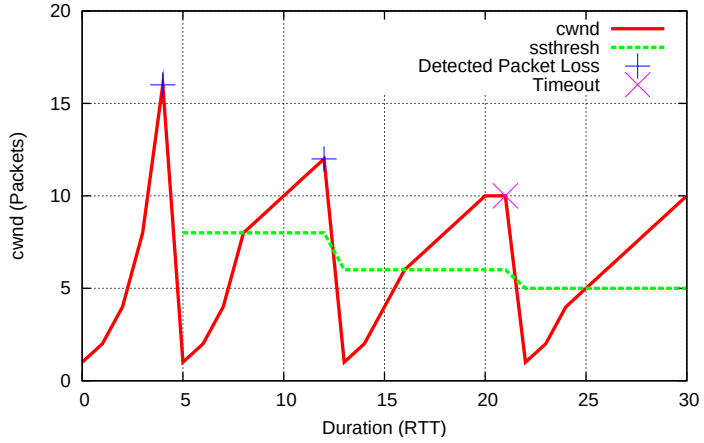


Figure 4.4: Time Evolution of Example TCP Tahoe Trace.

of a dropped packet sooner than the regular timeout mechanism and hence improves packet loss detection. It does not replace regular timeouts; it just enhances that facility and keeps the *ack* clock from failing.

Fast Retransmits are implemented via the detection of *dupack*s at the sender. It suggests that either a packet has been lost, or received out of order[4]. Upon the occurrence of three consecutive *dupack*s, TCP Tahoe decreases the sending rate and the inferred lost packet is resent. The main difference between TCP Tahoe and variants such as TCP Reno (see Section 4.5.2) is that upon loss detection through *dupack*s, *cwnd* is set to one packet rather than half of the previous value. As the mechanism to set *ssthresh* still holds, the TCP connection always restarts in slow start (after loss detection) until *cwnd* reaches *ssthresh* and then the linear additive increase of the congestion avoidance mechanism occurs until the whole process is repeated or the connection is closed.

Figure 4.4 shows a typical time based evolution of TCP Tahoe. The TCP

[4]Later implementations such as TCP SACK can calculate the exact missing segment based on extra information provided by the receiver to prevent the need to retransmit already received data.

connection initially starts off in slow start, until in the 4^{th} RTT packet loss is detected and *ssthresh* is calculated as half of the number of packets in flight. As TCP Tahoe always restarts the connection in slow start, *cwnd* is set to 1 packet and *cwnd* slow starts up to *ssthresh*. When $cwnd \geq ssthresh$ the TCP connection enters congestion avoidance until packet loss is again detected. The cycle continues until the 21^{st} RTT when a network timeout occurs whereby *ssthresh* is recalculated and *cwnd* is set to 1 packet.

4.5.2 TCP Reno

In 1990, Van Jacobson refined the congestion control algorithm. He noted that congestion control should do two things: prevent the pipe from going empty after a loss (as if it doesn't go empty, the TCP flow does not have to spend time refilling it again) and correctly account for all data that is actually in the pipe (as congestion avoidance should be estimating and adapting to the rate).

As a consequence of this chain of thought, TCP Reno [Jac88] introduces major improvements over Tahoe. It still maintains all of the features of TCP Tahoe (such as the improved *RTO* calculation and slow start after a timeout) but changes the way in which it reacts when a loss is detected upon duplicate acknowledgments.

The concept is that should the network be heavily congested, the receipt of *ack*s will stop (as the receiver will not receive any data packets to respond acknowledgments to). Therefore it is highly likely timeouts only occur when the network undergoes heavy congestion, and that the receipt of *dupack* signifies that the network is only moderately congested.

Therefore, the implementation of slow start after a Fast Retransmit in

TCP Tahoe completely destroys the *ack* clock unnecessarily and is too conservative in its rate reduction.

Given this assumption, under moderate loads the sender should be able to keep on sending data since the flow still exists (as the equilibrium has not been completely destroyed). However, the sender should be sending with less vigour to utilise less resources in order to prevent congestion collapse.

As such, Reno introduces a mechanism called Fast Recovery [Ste97] which should be activated after a Fast Retransmit. Recall that Fast Retransmits will cause the sender to retransmit the lost packet after receiving three *dupacks*. However, under Reno, it will not fall back to slow start, but instead, it should take advantage of the fact that the flow that currently exists should keep on sending, but using less resources.

By using Fast Recovery, the sender calculates half the number of packets in-flight just before loss detection. It sets *ssthresh* to this value and then subsequently also sets *cwnd* to the same value, rather than to one packet as in Tahoe[5]. As such, this forces TCP Reno to send fewer packets out and therefore it has indeed reduced its utilisation of the network.

As *ssthresh* is also updated upon congestion, TCP Reno is therefore also able to quickly revert back to congestion avoidance *should* a *RTO* expiration occur (causing the TCP flow to enter slow start).

But the shrinking of *cwnd* means that the TCP flow has already sent out all the packets that are contained within the window and therefore there is no way of sending out new packets unless the window is forced to include the new ones. As the receipt of *dupack*s under Tahoe does not move this window, then the only way to shift the window is to (artificially) inflate the size of

[5]Actually, it is set to half of the the number of packets in flight before the loss *plus* 3 packets to signify the data segments that have been acknowledged by the 3 *dupack*s.

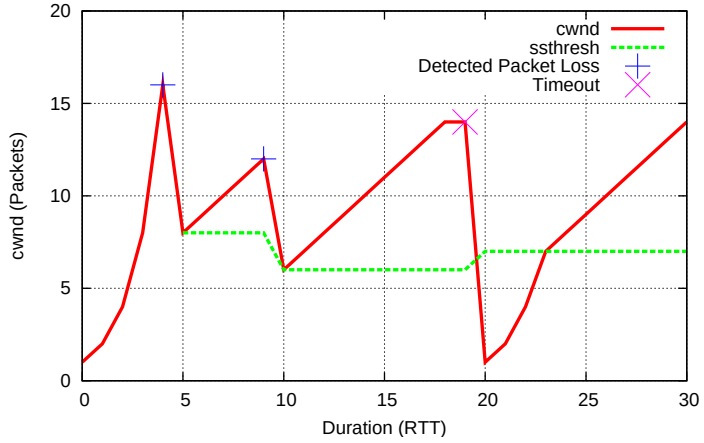


Figure 4.5: Time Evolution of Example TCP Reno Trace.

cwnd upon each *dupack* received as each represents another data segment which has left the network. Therefore, upon the receipt of each *dupack*, the *cwnd* is inflated by one segment as each *dupack* signals the fact that another packet has left the network.

When all the *dupack*s have been received ($cwnd/2$ packets corresponding to the number of of packets that were in flight prior to the loss detection), the next *ack* should be that caused by the retransmission and therefore should acknowledge all packets. Upon the receipt of this normal *ack*, the sender exits Fast Recovery and set its *cwnd* to *ssthresh* to maintain sending at a rate which does not destroy the *ack* clock.

Therefore, under Reno, slow start is only invoked at the start of a TCP connection and when a packet is timed out by the *RTO*, i.e. only when the *ack* clock of the TCP flow has been destroyed.

Figure 4.5 shows a typical time evolution of TCP Reno. Slow start is initiated whereby at the 4^{th} RTT packet loss is detected and *ssthresh* is recalculated. Unlike TCP Tahoe, TCP Reno effectively sets *cwnd* to the value of *ssthresh* after Fast Recovery. After this, congestion avoidance is initiated

until packet loss is detected at the 9^{th} RTT, where Fast Retransmission and Fast Recovery is again initiated. At the 19^{th} RTT, a timeout occurs and again *ssthresh* is recalculated, but *cwnd* is set to 1 packet whereby slow start is initiated until $cwnd \geq ssthresh$.

4.5.3 Motivation for Improving Loss Detection

A fundamental problem is that TCP *acks* are cumulative; an *ack* confirms reception of all data up to a given sequence number, but provides no information whether any bytes beyond this number were received. Therefore upon loss, the Fast Retransmit and Fast Recovery algorithms assume that only the segment at `snd_una` is lost per window. This can result in the loss of *ack* clocking and timeouts if more than one segment is lost.

Due to the prevalence of FIFO queues in the internet, losses often occur in bursts [JS00]. As link speeds increase and the internet becomes more geographically distributed, with more hops, TCP *cwnd* sizes are increased to make use of available capacity. The result of this amalgamation is that single packet loss within a window is rare and multiple losses within the same window are more likely to be the observed effect.

When Tahoe and Reno experience multiple packet losses in a window of data (usually in the order of half a window under heavily congested links), Fast Retransmissions and Fast Recovery are invoked several times in succession leading to multiplicative decreases of *cwnd* and *ssthresh*.

As this happens several times in succession, the left edge of the sending window advances only after each successive Fast Retransmit and the amount of data in-flight (sent but not yet *ack*ed) eventually becomes more than the *cwnd* (halved by the latest invocation of Fast Retransmit). As there are

no more *ack*s to receive then the sender stalls and recovers from this deadlock only through a timeout - which destroys the *ack* clocking and is only recoverable through slow start. This has the impact of severely reducing the throughput of the TCP flow.

Another problem of inducing multiple Fast Retransmits is that the source retransmits packets that have already been correctly received by the receiver which increases the potential for *classical congestion collapse* [FF99].

A more in-depth discussion with diagrams is presented in [FF96].

4.5.4 TCP NewReno

A modification of Reno led to TCP NewReno [FH99] which shows that Reno can be improved without the addition of SACKs (See Section 4.5.5) but still suffers without it. Here, the wait for a retransmit timer is eliminated when multiple packets are lost from a window.

NewReno modifies the Fast Retransmit and Fast Recovery. These modifications are intended to fix the Reno problems of multiple Fast Retransmits and are wholly implemented in the sender side, in contrast to TCP SACK.

NewReno is the same as Reno but with more intelligence during fast recovery. It utilises the idea of partial *ack*s; when there are multiple packet drops, the *ack*s for the retransmitted packet will acknowledge some, but not all the segments sent before the Fast Retransmit.

In TCP Reno, the first partial *ack* will bring the sender out of the Fast Recovery phase and will require a timeout when there are multiple losses in a window to recover from the losses, thus stalling the TCP connection. In NewReno, a partial *ack* is taken as an indication of another lost packet and as such the sender retransmits the first unacknowledged packet. Unlike Reno,

partial *ack*s do not take NewReno out of Fast Recovery and it retransmits one packet per RTT until all the lost packets are retransmitted thus avoiding the multiple fast retransmits from a single window of data [Flo94b].

The downside of this is that it may take many RTT's to recover from a loss episode, and enough new data must be present in order to keep the *ack* clock running. Otherwise, re-initiation of the *ack* clock through a *RTO* timeout is still necessary.

The modifications to Reno to enable NewReno are as follows:

Multiple Packet Loss

When first entering Fast Retransmit, the highest sequence number sent so far is saved in the variable `recover`. Normal retransmission of data and the Fast Recovery algorithm are run as normal. However, when a new *ack* arrives, an additional check is performed to ensure that this *ack* covers the value of `recover`. If the sequence number of the *ack* is less than that of `recover`, then this *ack* is a partial *ack* and signals that another segment was lost from the same window of data.

As such, TCP can retransmit the segment reported as expected by the partial *ack*. There are two versions of NewReno algorithm which differ in the way that the *RTO* is updated under this scenario [FHG04]. Under the 'slow-and-steady' variant, the *RTO* is reset on every partial *ack*. Whilst with the 'impatient' variant the *RTO* is only set after the first partial *ack*. Due to this *RTO* update, the impatient variant is more likely to be able to recover quickly under very lossy conditions (or large *cwnd*s) by resorting to timeouts, whilst the 'slow-but-steady' variant will take approximately n RTTs to recover (where n is the number of lost packets in the window)

[Hoe96]. In both cases, the TCP connection only exits Fast Recovery if the new *ack*'s sequence number is larger than `recover`.

False Retransmits

In order to prevent the retransmission of segments that have already been received by the receiver[6], the same `recover` variable as per the previous discussion is used[7]. With the initial call into Fast Retransmit after a loss is detected, the *RTO* is reset. Upon consecutive *RTO*'s, the highest sequence number transmitted so far is also recorded into `recover`. By checking the received *ack*s against `recover`, the TCP sender is able to prevent having to retransmit unnecessary data.

The sender comes out of Fast Recovery and Fast Retransmits only after all outstanding packets (at the time of first loss) have been *ack*ed.

4.5.5 Selective Acknowledgments (SACKs)

With Selective Acknowledgment, SACK [MMFR96], the data receiver can inform the sender about all segments that have arrived successfully, so the sender need retransmit only the segments that have actually been lost. However, TCP-SACK requires that both TCP senders and receivers be modified to support SACKs.

In order to be backwards compatible with existing non-SACK TCP receivers, SACK capability is negotiated during the TCP handshake. If neither of the TCP hosts implement the TCP SACK option, then TCP-SACK behaves the same as TCP Reno - with its associated problem with multiple

[6] As retransmitted packets may also be dropped.

[7] Note that the original 'experimental' draft of NewReno does not implement the check against recover.

drops within a window. To avoid this TCP NewReno was proposed (see Section 4.5.4).

TCP-SACK has been shown to perform well even at a high level of packet losses in the network [MM96].

SACK Blocks

SACK attaches detailed information called SACK Blocks in TCP Options [Pos81b] on each and every *ack*. Each SACK block specifies a contiguous and isolated block of data not covered by the TCP Cumulative Acknowledgment field. Therefore cumulative SACK Blocks reports in the *ack* segments help the sender to identify 'holes' in the sequence numbers at the receiver; on the receipt of an *ack* containing the SACK Block, the sender can infer which segments are lost and hence retransmit only those segments. Therefore, SACKs enables a TCP sender to maintain an image of the receiver's queue and enables recovery from multiple losses per window within roughly one RTT.

Each SACK block is described using absolute sequence numbers, with two sequence numbers describing each block, each taking 4B:

1st Left edge of the block, i.e. the first byte of the block successfully received.

2nd Right edge of the block; i.e. the sequence number of the byte immediately following the last byte in the block.

Given that the TCP options field is limited to 40B, the maximum number of SACK blocks per ack is limited[8] to four. However, as it is expected that SACK will often be used in conjunction with other TCP Options such as

[8]Each TCP option has a necessary associated overhead.

TCP Timestamps [JBB92] (See Section 5.2.1), typically only 3 SACK blocks are expected in each *ack*.

The extra redundancy in having more than one SACK Block in each ack is necessary due to the lossy nature of network paths and enables 'robustness' when *ack*s are lost. It is therefore also important that the SACK option always reports the block containing the most recently received segment, as this provides the sender with the most up-to-date information about the state of the network and the data receiver's queue.

However, the unacknowledged segments can be treated in different ways when accounting for outstanding data. The approach promoted by the IETF is to consider all unacknowledged data to be outstanding in the network. Holes within SACK blocks indicate potentially lost packets. However, they may also indicate reordered packets. The method by which SACK is implemented can be conservative or aggressive in the way it treats the holes within the SACK blocks. The Forward Acknowledgments (FACK) algorithm [MM96] takes the more aggressive approach and considers the unacknowledged holes between the SACK blocks as lost packets and therefore retransmits the data. The more conservative approach suggested by the IETF is to consider all unacknowledged segments to be outstanding in the network. Although the FACK approach can result in better TCP performance it may increase the retransmission of unnecessary data.

How the sender uses the information provided by SACK is implementation-dependent. For example, Linux uses a Forward Acknowledgment (FACK) algorithm [MM96] Another implementation is sometimes referred to as Reno+SACK [MMFR96, MM96].

D-SACK

As an extension of the SACK mechanism, if a duplicate segment was received, then the receiver should send an *ack*, containing as its first SACK block, a reference to the duplicate data. Using SACK information and the D-SACK extensions [FMMP00] it is possible to infer the amount of re-ordering on a path and it is possible to make TCP more robust to re-ordering by using this information to set the fast retransmit threshold depending on the reordering detected on a path [BA02, ZKFP02].

4.6 Summary

The functions and implementation of the Transmission Control Protocol (TCP) were presented and some of the key changes to its algorithms in order to better utilise network resources upon congestion were given.

As the Internet is a shared resource of many end-nodes sending traffic into a 'black-box' network, it is important that users do not send data into the network such that congestion collapse occurs. Should the transport algorithms not support a mechanism that is aware of the need for appropriate use of these resources, then congestion collapse will invariably occur. An important aspect of TCP is that it introduces the idea of 'congestion control' in order to regulate the rate at which data is sent (and retransmitted) into the Internet. Because of the success of the Slow Start, *RTO* calculation, Fast Retransmit and Fast Recovery mechanisms, it is widely believed that TCP in all of its variants has prevented the occurrence of Internet collapse.

TCP implements the notion of a congestion window, *cwnd*, which is the protocols' estimation of the rate at which the network can handle network

traffic for a particular connection. Depending on the network state determined from the inference of packet loss through the heuristics of *ack*s, TCP adapts its *cwnd*. By increasing *cwnd* such that the rate at which new data is put into the network is increased TCP is able to take advantage of extra bandwidth available. And upon indications of network congestion, TCP is able to quickly free up network resources.

However, in order to reach this 'steady state', TCP requires a mechanism to quickly adapt to network conditions from which Additive Increase Multiplicative Decrease (AIMD) can take place. This is implemented via a slow start regime where the sending rate is doubled every RTT (as long as there is no loss). In conjunction with a *ssthresh* variable, TCP is able to switch between the aggressive slow start and congestion avoidance algorithms.

As TCP is a reliable protocol, TCP needs to deal with potential loss of packets. It also assumes that loss due to packet corruption is rare[9] TCP infers the difference between the types of congestion by the arrival (or absence) of *ack*s from the receiver. Under heavy congestion, it is likely that no data will get delivered, and therefore there will be no *ack*s in response. To detect packet loss because of this, TCP implements a *RTO* soft-timer that will initiate the retransmission of the packet that it believes is lost. Under moderate congestion, it is likely that only some segments will be lost, and this is represented in the arrival of *dupack*s from the receiver.

Much of the developments of TCP in the past have been related to the way in which it responds to packet loss detection. TCP Tahoe reacts the same to both heavy and moderate congestion by reverting the TCP connection back

[9]TCP performance suffers in environments where there exists high Bit Error Rates (BER) (e.g. wireless/radio). However, the kind of BER considered in this dissertation for the HEP community typically include high quality links optical interconnects which have low raw BER of 10^{-12}.

into slow start. However, this will result in the breaking of the *ack* clock which is required for stable TCP connections and therefore would take a few RTTs to get back into congestion avoidance. Therefore, TCP Reno was devised to be able to differentiate moderate congestion and react by only halving *cwnd*. This is called AIMD in reference to increase of *cwnd* by one segment per RTT when probing and the halving of *cwnd* upon loss.

However, as connection bandwidths and end-to-end latencies of the Internet increase, *cwnd* has to increase to retain enough state to enable the sliding window mechanism required for reliable delivery of data. As this occurs, the chance of multiple losses of segments in a single window increases.

Unfortunately, TCP Reno and TCP Tahoe do not handle such situations well and each will successively call its corresponding loss detection mechanism. The effect of this is the stalling of the TCP connection as it attempts to retransmit what it thinks is lost in the network.

Two primary proposals were developed to facilitate the recovery of TCP under multiple packet losses per window. Of the two, Selective Acknowledgments (SACKs) allow the TCP sender to recover from the losses in the order of just one RTT. However, it requires that both the sender and receiver specially implement SACKs in order to operate. TCP NewReno, however, only requires a sender side modification, but it will take an amount of time related to the number of lost packets in order to recover. Otherwise, it will require that a *RTO* timeout occur at the sender.

Overall, it can be seen that many changes have been made over time in order to improve TCP's behaviour with respect to congestion. In the next Chapter, more recent modifications are considered with the aim of improving performance in high-speed networks.

Chapter 5

TCP For High Performance

The role of TCP in network transport is an important one. Not only does it provide reliable replication of data across a multitude of different networks, but it also prevents congestion collapse.

This chapter looks into the current application of the TCP protocol, real world implementation difficulties and the effect upon TCP performance.

5.1 TCP Hardware Requirements

TCP is often implemented in software. However, the design of the TCP protocol requires certain constraints on physical hardware in order to operate efficiently.

This section investigates the effect of physical hardware upon TCP performance.

5.1.1 Memory Requirements

The reliable data transport design of TCP requires that a sliding window's worth of data be stored in memory in order to maintain state about the TCP connection. Without such knowledge, it would be impossible for TCP to be able to replicate data reliably across a network. The amount of physical memory allocated to this sliding window is called the *socket buffer* memory.

Should the assigned value of this socket buffer memory be less than the required value of the sliding window for optimal TCP transfers, then the performance of TCP suffers as it is starved of physical memory space to keep history for the number of packets in flight.

This is true of both the sending and the receiving TCP end-points; the TCP receiver must also have a large enough assigned socket buffer memory, or else TCP's flow control at the sender will restrict the throughput to maintain a rate sustainable by the TCP receiver (See Section 4.2.2).

The amount of memory that needs to be allocated to the socket buffer memory, on both the sending and receiving hosts, is related to the Bandwidth Delay Product (See Section 4.2.2) [CJ89].

The effect on a TCP connection of having limited memory is shown in Figure 5.1. The graph shows the statistically multiplexed result of running various socket buffer sizes from the production network between CERN and RAL using the machines and configuration as shown in Section 3.2.2. The test consisted of 490 individual bulk transfers using `iperf` [TQD+03] with the socket buffer sizes configured as shown. The theoretical prediction of the estimated throughput is shown. It can be seen clearly that with small socket buffers, the throughput is proportionally restricted by the amount of memory available to the TCP socket.

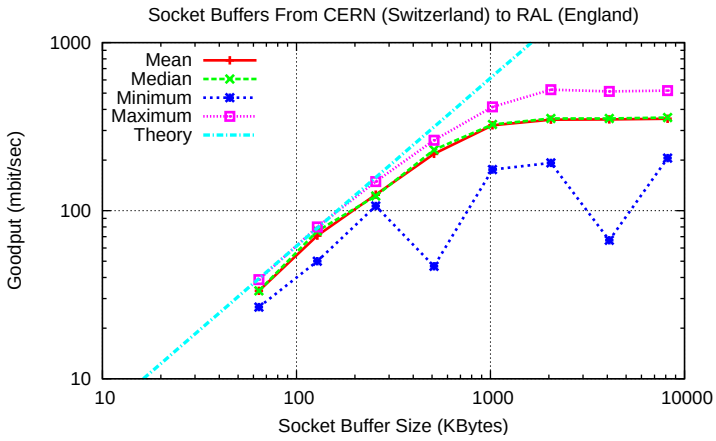


Figure 5.1: Effect of limiting the socket buffer size on TCP goodput.

The typical socket buffer memory size for Linux and BSD systems is set to 64KB. Depending on the latency and stable throughput of the TCP path, one can see that TCP would perform abysmally over the example link and that the throughput would be capped so that the TCP window fits within the 64KB.

To address this problem, grid and network researchers continue to manually optimise socket buffer sizes to keep the network pipe full, and thus achieve increases of transport throughput by many orders of magnitude.

The amount of memory allocated to each TCP connection on Unix systems can be changed using the standard Unix `setsockopt()` [Ste98] call. However, under Linux systems, the kernel also imposes a limit on the maximum size of the socket buffer memory that can be allocated for each connection. As such system administrator privileges are often required to alter this limit such that userspace applications such as FTP can utilise large(r) socket buffer sizes. This is achieved using the standard `sysctl` variables `net.core.rmem_max` and `net.core.wmem_max` for the read and write socket buffers respectively. Similarly, the default allocated socket buffer memory

for systems that do not implement the `setsockopt()` call can be given default socket buffer settings as defined in the `net.core.rmem_default` and `net.core.wmem_default` `sysctl` variables.

One may be tempted to assign large default and maximum values to the socket memory to enable high speed TCP transfers over high BDP paths. However, this could lead to performance degradation on servers as physical memory may need to be swapped to and from disk if many such connections are used. Servers can often have hundreds to thousands of simultaneous network connections.

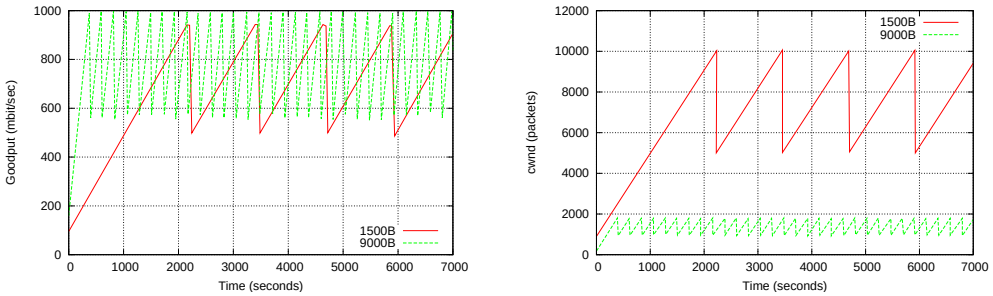
5.1.2 Network Framing and Maximum Segment Size

As the TCP/IP paradigm requires the transport of data packets across the Internet, a fundamental question arises upon the size of each packet or segment. However, there is an overhead cost due to framing and encapsulation upon of each packet.

As the IP, TCP and underlying Layer 2 headers are often static in size, by imposing a larger packet size it is possible to increase the raw data throughput due to the relative decrease in header overheads. Large packets also result in fewer interrupts at then end systems which can result in higher throughput (See Section A.2.4).

However, with larger packet sizes, and with constant Bit Error Rate (BER), the probability of packet corruption increases - although this issue only becomes serious with wireless networks [BPSK96, Che01, CKPC03].

The Maximum Segment Size (MSS) [Pos83] is the size of the raw data which TCP is allowed to store in each packet. To prevent problems due to fragmentation at the IP level, the Maximum Transfer Unit (MTU) [MD90]



(a) Goodput (b) CWND

Figure 5.2: Effect of TCP performance with different MTU sizes.

is used to impose an upper limit on MSS.

These values do not prevent senders from placing larger packet sizes into the network. However, large segments packets will be 'fragmented' [Cla82a, ZRT95] into smaller packets to overcome the size difference. This imposes the extra overhead of increased packet header sizes and a higher chance of packets lost to BER. At the receiver, the individual fragments are then reassembled to form the original packet. To over come the problems of packet fragmentation, Path MTU Discovery is often conducted to determine the appropriate segment sizes on the network [MD90, Lah00].

The effect of using large MTU's is shown in Figure 5.2 based on tests run on the DataTAG testbed (See Appendix C.3). TCP counts acknowledgments by segments rather than by bytes. As such, in order to achieve the same throughput, larger MSS flows only require a smaller value of *cwnd* as more bytes are transferred per unit value of *cwnd*.

On the other hand, the transport of each large MSS segments requires an increase in latency in order for routers and switches to transport the segment. This could lead to a larger *RTO* which may have a negative influence on TCP timeouts and hence lead to a deterioration in average throughput. However,

this effect should be negligible on long latency paths where the extra latency of processing the larger segment is small compared to the end-to-end latency.

Further information can be found in [Ste94, MDK+00].

5.1.3 CPU Requirements

A context switch requires that the CPU state needs to be stored and restored. The transport of segments to and from a PC requires the context switching of hardware and software control such that separate compute processes of the operating system can gain access to the CPU resources.

Gigabit speeds require the transport of $\frac{1{,}000{,}000{,}000}{8\times1500} = 83,333$ packets per second using standard MTU sized packets. This means that a packet needs to be processed about every 12μsec.

As TCP uses feedback from acknowledgments, the TCP sender must also process all incoming *ack*s. This means that a TCP sender must send out data and receive *ack*s in order to enable reliable transport. Also, TCP requires the extra processing overhead of the contents of *ack*s in order to determine what data needs to be retransmitted[1].

At higher speeds, more packets need to be handled by the end systems which results in higher CPU utilisation.

The overheads required in the context switching of interrupts can be mitigated via interrupt coalescing (See Section A.2.4) and the use of advanced queue management techniques such as NAPI (See Section 5.2.3). Using larger sized packets (See Section 5.1.2) may also help reduce the CPU utilisation as long as fragmentation does not occur.

Figure 5.3 shows the CPU load during individual TCP transfers between

[1]Also SACKs prove to be a serious overhead in computational terms.

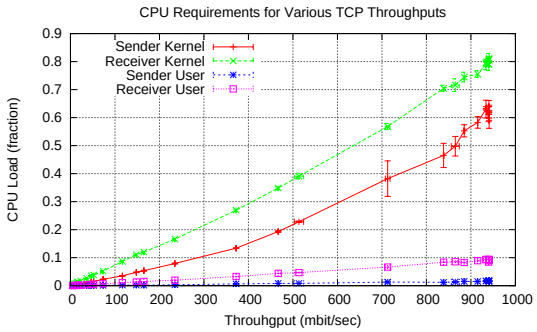


Figure 5.3: CPU utilisation with different TCP throughputs.

identical Dual Xeon 2.0Ghz PCs over the MB-NG private WAN (See Appendix C.2). The configuration of both machines is presented in Table B.2.

The receiver is configured to simulate packet loss by dropping every n^{th} packet to achieve a desired throughput as determined by the response function which limits the throughput of TCP based upon the loss rate experienced (See Equation 5.7). As each TCP connection under Linux is handled under a single processor thread, the benefits of SMP systems are limited, and this is demonstrated by nearly 100% utilisation of a single processor at high speeds.

It was observed that the TCP receiver requires a higher CPU load than that of the sender, with an almost linear relation. The sender, on the other hand, requires more CPU at higher speeds (with a higher gradient at higher throughputs), most likely to due to influx of *ack* packets that require extra processing overheads (such as duplicate *ack*s and SACKs).

Other research [tePM] confirms the approximate calculation that 1Ghz CPU is required for each Gigabit/sec throughput when the performance is restricted to end-node hardware rather than Internet performance.

5.2 TCP Tuning & Performance Improvements

5.2.1 Standardised Changes to TCP

Inadequate Window

The 16-bit TCP Sequence Number and Acknowledgment fields as proposed in [Pos81b] imposes a maximum TCP window size of 2^{16} = 64KB. Without the Large Window extensions [JBB92], the maximum throughput of a TCP connection is limited by the RTT as given by Equation 4.1. On a typical long distance link with a RTT of 100ms, the maximum throughput of the TCP connection is therefore limited to $\frac{64KB}{100ms}$ = 5.12Mbit/sec.

The TCP extension for large windows [JBB92] is implemented using a 3-byte TCP Option that defines an implicit scale factor to multiply the window size as reported in each TCP packet. It increases the size of the TCP advertised window to 32-bits and then uses a scale factor to carry this 32-bit value in the 16-bit window field of the TCP header. It is sent only in a SYN segment at the TCP handshake and hence the scaling factor is fixed in each direction when a new connection is opened. The maximum window guaranteed is 2^{30}-bytes which imposes a limit of 1GB.

The use of scaling the window imposes problems of sequence numbers wrapping around and therefore the possibility of *ack*ing incorrect segments in the window. This is dealt with internally to ensure that the sequence number is within 2^{31}-bytes of the left edge of the window. Old *ack*s are also dealt with using PAWS and through the IP TTL field [Pos81a].

TCP Timestamps and PAWS

The Timestamp Option [JBB92] provides for timing of every packet in-flight, rather than only one per RTT as defined in the original specifications [Pos81b]. This improves the RTT estimation and enables keeping a more dynamic and accurate value for RTO and hence slightly quicker response to TCP timeouts.

TCP Timestamps occupy 10-bytes and require the sender to place a current timestamp into each packet sent out (including retransmissions) which is then echoed by the TCP receiver in the *ack*.

The implementation of the Timestamps option is especially useful for environments with very variable latencies such as wireless networks [LK02]. However, especially in wireless environments, the implementation has to be balanced against the extra overhead of header information.

The implementation of a timestamp also enables extra protection against potential wrap around due to the reuse of sequence numbers for long duration transfers. [JBB92] defines the Protection Against Wraparound Sequence Numbers (PAWS) algorithm that enables the checking to ensure that arriving packets are valid in time for the connection.

The inclusion of the Timestamp option also enables low-level improvements in processing TCP data packets at the receiver. This is referred to as 'Header Prediction' [Jac90] and enables the receiver to quickly process a TCP segment provided that the timestamp is larger than the previous one recorded and that the sequence number is the next in sequence.

Initial Window Size

For short TCP connections that only transfer a small amount of data, the rate of transfer is limited primarily by the amount of time spent in slow start. TCP is currently defined to start a connection with an initial *cwnd* of one. Therefore, by increasing this value to three segments rather than one, the amount of data transferred at the start of the connection is improved by a factor of three and the connection can therefore terminate earlier for the same amount of data sent. However, due to the larger initial burst, a connection across links with very small router buffers may experience loss sooner and therefore resort back into congestion avoidance.

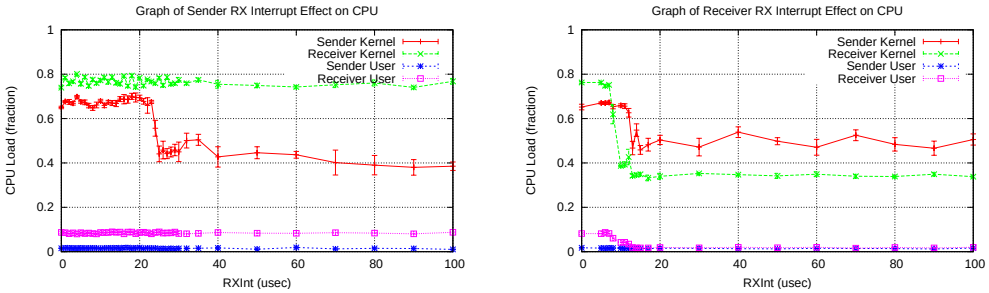
[AFP98] defines an experimental extension that allows an increase of the initial window to three or four segments. However, the number of segments sent after an *RTO* is still fixed at one segment.

However, for large file transfers that take a long time, the time saved by having a larger initial window is irrelevant as TCP will most likely spend most of its time in congestion avoidance.

5.2.2 Host Queues

The `txqueuelen` is a Linux `sysctl` variable that determines the maximum number of packets that can be buffered on the egress queue of the kernel before it enters the NIC's device driver [ABL+03].

Higher queues sizes means that more packets can be buffered and hence not lost. This is especially important for TCP in the Linux kernel as the overflow of this queue will cause immediate congestion control to be instantiated, even though the packet is lost locally rather than on the network. The Linux Web100 patch [MHR03] determines such action as `SendStalls` and



(a) Sender (b) Receiver

Figure 5.4: Effect of varying RX Interrupt values on CPU load.

can be deactivated using Web100's `net.ipv4.WAD_IFQ` `sysctl`.

However, imposing a large `txqueuelen` may add to extra delays in the transport of TCP segments due to extra queuing latencies that may be experienced. This could result in large variations of RTT and hence result in poor *ack*-clocking and spurious timeouts [LK02, GL03].

The default value of 100 for `txqueuelen` in the Linux 2.4 kernels has been reported to be insufficient to maintain high throughput transport [ABL+03]. Newer Linux 2.6 kernels implement a default of 1,000.

Similarly, at the receiving end of transport, packets will be queued, once past the device driver, in a queue called `max_backlog`.

5.2.3 Driver Modifications

Interrupt Coalescing

By reducing the number of context switches required to process a packet, CPU overheads can be reduced at the expense of increased latency upon the incoming/outgoing packets.

Intel e1000 network interface card drivers enable interrupt coalescing in

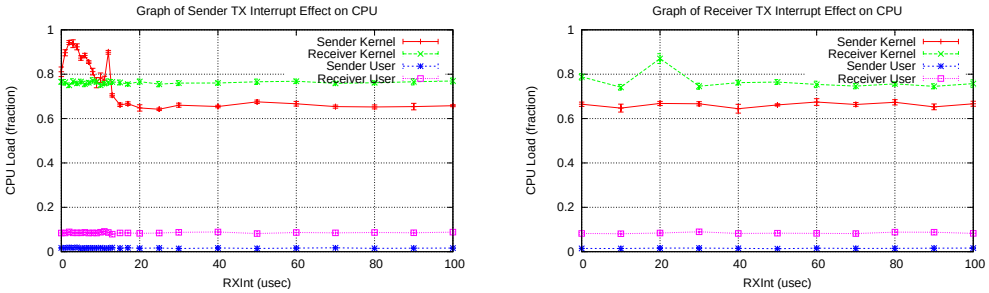
granularity of $1.024\mu sec$ time slices. This value was varied to investigate the effect upon CPU utilisation which was measured using the Unix `time` command. Figure 5.4 shows the effects of varying the interrupt values on the Intel e1000 (version 5.2.20) driver on back-to-back tests at 1Gb/sec using TCP and `iperf` on the MB-NG testbed network (See Appendix C.2). In all tests, it can be seen that the most intensive component of CPU utilisation is that of the kernel process, which involves both the memory handling and packet interrupts.

Figure 5.4(a) shows the effect of changing the coalescing values on the sender only, with the receiver set to interrupt on every packet (`RXIntValue=0`). It clearly shows the reduction of CPU load on the sending machine as it approaches 23μsec - which is equivalent to approximately 2 packets (in this case *ack*s) that are received back-to-back due to the use of ABC (See Section 5.2.4) which causes two data packets to be sent back-to-back. As the receiver is constantly interrupting on every (data) packet, its CPU utilisation remains constant.

Figure 5.4(b) shows the effects of adjusting the receiver's `RXIntValue`. It can be seen that a small change of this value to approximately the time taken to process one packet can reduce the CPU utilisation of the kernel by about half. As it is now the sender which is interrupting on each packet, it can be seen that there is also the benefit of lower CPU utilisation on the sender at the same settings.

Figure 5.5 shows similar tests but varying the transmitting `TXIntValue` coalesce value rather than the receiving.

It was observed that altering the `TXIntValue` on the receiver has no effect on the CPU utilisation of either system. This is due to the fact that even though *ack* clocking is important to TCP, it does not affect the number



(a) Sender (b) Recevier

Figure 5.5: Effect of varying TX Interrupt values on CPU load.

of data packets that are injected into the network if the *ack*s are slightly delayed. The effect of varying the transmit interrupt on the sender is that there is an initially high CPU requirement as the the driver attempts to interrupt for every packet ($<11\mu$sec) and beyond that the CPU utilisation remains constant.

New API (NAPI)

NAPI is a system implemented into the NIC driver that facilitates the reduction in CPU loads due to context switching at high speeds. Implemented on the receiving queue, the API works by switching from interrupts to polling when the RX ring is reasonably occupied. This has the benefit of decreasing processing loads, at the expense of increased latency (upto the inter-poll time between polls).

5.2.4 Delayed Acknowledgments & Appropriate Byte Sizing

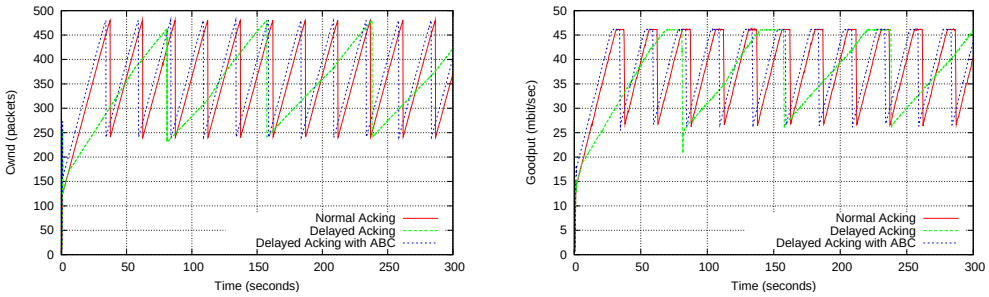
At high throughput rates, TCP receivers would send a lot of acknowledgement packets on the reverse path of the bulk transfer to facilitate *ack*-clocking and to slide the TCP window. Under bulk one-way transport, *ack*s are often composed of an empty TCP packet (with only header information), and are therefore normally quite small (56B). As demonstrated in Section A.2.3, small packets can pose a serious processing burden on both the end-host and the intermediate switches and routers.

As *ack*s cover all segments prior to the sequence number within the *ack*, the advancement of the TCP window is not diminished by reducing the number of *ack*s sent by the TCP receiver [Cla82b].

Under delayed acknowledgments [Cla82b, APS99], a TCP receiver does not acknowledge a received segment immediately, but waits for a certain time (typically 500msec). If a data segment is sent from the receiver during this time (i.e. as with two way communication), the acknowledgment is piggy backed into it. Alternatively, if another data segment arrives from the TCP sender, then the receiver will send a single *ack* that confirms both segments at once.

It is suggested by the IETF that delayed *ack*s *should* be incorporated into TCP implementations [Cla82b]. However, the consequence may be that 'stretched' *ack*s result that will acknowledge more than two full-sized segments. This would lead to potentially large line-rate bursts of traffic [Pax97] (which can also occur with large amounts of *ack* loss).

Typically each arriving *ack* at the sender advances the sliding window and increases the congestion window by one segment; thus, a connection with



(a) CWND (b) Goodput

Figure 5.6: Effect of delayed *ack*s upon *cwnd* dynamics.

delayed *ack*s is less aggressive, and increases *cwnd* at half the rate compared to a TCP connection without delayed acknowledgments. This results in reduced throughput over short timescales after each congestion event for any TCP connection where the receiver is using delayed acknowledgments. This is shown in Figure 5.6 for machines on a private dummynet network (See Appendix A.5).

This is especially noticeable in the slow start phase, because in slow start each arriving *ack* increases the congestion window by one segment, thus effectively doubling the rate at which packets are sent.

The Linux kernel implements delayed *ack*ing by default [Bra89], but imposes a less conservative delayed *ack*ing time of up-to 200ms. Linux introduces a feature called quick *ack*s [SK02] which helps to reduce the problems of reduced responsiveness when using delayed *ack*ing during slow start. The idea is to disable delayed *ack*s at the receiver for the first n packets of the connection, where n is a configurable parameter, acknowledging every packet at the beginning of the connection and thus achieving the equilibrium state (congestion avoidance) in shorter time.

On the other hand, such a policy would increase the probability of net-

work congestion on the reverse path as more *ack*s are put into the network. However, as quick *ack*s are designed primarily to reduce the time required for slow start to complete, the number of quick *ack*s is calculated as $n = \frac{rwnd}{2 \times MSS}$ where *rwnd* is the reciever advertised window and MSS is the maximum segment size of the connection. This value, n, corresponds to the number of packets that the TCP connection should take to exit slow start.

Appropriate Byte Counting (ABC) is a technique to eliminate the slow growth of *cwnd* with delayed *ack* receivers and is outlined in [All03]. It gives details of how TCP senders should behave when updating their congestion windows through byte counting rather than the more common packet counting. This has the benefit of improving TCP responsiveness without increasing the amount of *ack* traffic (relative to the throughput) upon the reverse path.

Figure 5.6 shows the effect of running an ABC enabled sender when transferring data to a normal delayed *ack* receiver. It can be seen that the dynamics of using ABC is similar to that of a TCP sender sending data to a receiver with delayed *ack*s disabled.

It can also be observed that the queueing is apparent as the goodput remains constant as the *cwnd* continues increasing until finally the buffer overflows and fast retransmit and fast recovery takes place.

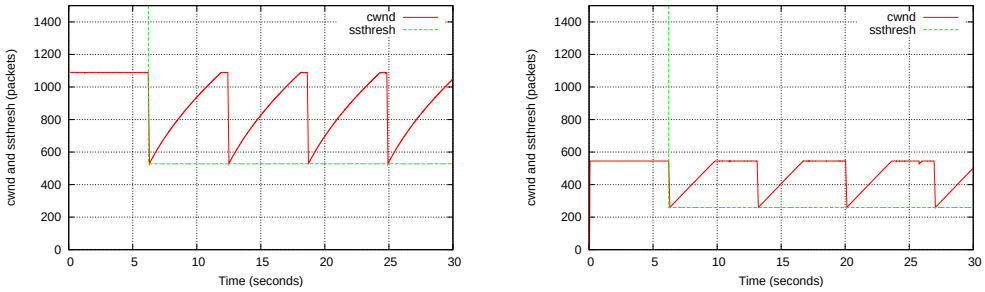
A potential consequence of implementing ABC is an increase in the burstiness of the TCP traffic as two back-to-back data packets are sent for every delayed *ack* [All99]. In order to reduce the effects of multiple packet bursts, especially during slow start, a limit `L`, is introduced that limits the number of full sized segments that the *cwnd* will advance by for each delayed or stretched *ack* [All03]. The value of `L`= 2 means that a maximum of 2 new segments are sent for every one delayed *ack* received.

5.2.5 Dynamic Right Sizing & Receiver Window

When the TCP sender's window is not constrained by the system socket buffer size nor by the congestion window, it will be throttled only by the receiver's advertised window (as per TCP flow control). The size of the receiver window is a standard control parameter of TCP [Pos81b]. By advertising a smaller window the receiver can control the number of segments that the sender is allowed to transmit (similar to imposing a small socket buffer at the sender).

[FF01] describe modifications to the Linux kernel that allow the kernel to tune the buffer size advertised by the TCP receiver. Called Dynamic Right Sizing (DRS), the receiver's TCP kernel estimates the bandwidth from the amount of data received in each round-trip time and uses that estimation to derive the receiver's window to advertise back to the sender. Assuming that the sender is limited only by the network condition (i.e. by its *cwnd* value), then the amount of data transferred in approximately one RTT is equal to one windows worth of data. Therefore, by the TCP receiver measuring the time taken to receive one windows worth of data, it can determine the RTT of the connection. This value of RTT is used as an upper bound on the latency of the link and therefore the appropriate receive window size to advertise in following TCP *ack*s.

Therefore, the effect of DRS on TCP senders is that their *cwnd* value is never allowed to grow beyond what is actually achievable through that connection and is clamped to the value as calculated by the TCP receiver. As such, the TCP flow never actually imposes the additive increase of congestion control and hence does not impose the necessary loss that is typical of the oscillatory behaviour of TCP.



(a) Normal (b) DRS

Figure 5.7: Effect on cwnd with Dynamic Right Sizing.

This behaviour is especially important for long-distance, high throughput environments where the Multiplicative Decrease of TCP upon congestion/loss causes a halving in throughput. Another benefit of using DRS is that with appropriate socket buffer sizing (see Section 5.1.1) it also keeps the amount of memory utilised to the minimum required for each flow.

Figure 5.7 shows the behaviour of the sender *cwnd* with and without DRS along the MB-NG testbed (See Appendix C.2) with periodic loss on the forward path. MB-NG has a RTT of 6.2ms with 1Gb/sec connectivity. As such, the Bandwidth Delay Product (BDP - Equation 4.1) is approximately 535 packets.

Figure 5.7(a) shows the effect of queueing resulting in the curved *cwnd* graph as the router queues begin to fill[2]. The initial plateau of *cwnd* in Figure 5.7(a) is caused by socket buffer limitations at the sender which enable flow control by capping the maximum value of *cwnd*. Figure 5.7(b) shows the same network and dropping probability, but this time with DRS enabled at the receiver. It can be clearly seen that the *cwnd* value is capped to that of

[2]Note that in this case, the actual loss is imposed by the receiver selectively dropping packets rather than the network.

the BDP of the network at just under 600 packets, which is the maximal rate at which the receiver is able to receive packets as the sender is transferring data at line rate.

It is clearly visible that the oscillatory behaviour of TCP is minimised such that the sustained throughput is higher. However, it should also be noted that due to the drop in *cwnd* upon loss, the utilisation of a DRS'd flow is not maximal as with the case of normal TCP. The advantage of imposing such an algorithm is two-fold: with appropriate measures to minimise the socket-buffer size to that actually used by the flow (i.e. approximately the value of *cwnd*) memory is not wasted; and induced loss through sending too many packets into the network is also minimised, and therefore should the buffer provision be small along the path, a DRS'd flow should be able to obtain higher utilisation than a normal flow.

5.2.6 Socket Buffer Auto-Tuning & Caching

[SMM98] describes modifications to a NetBSD kernel that allow the kernel to automatically resize the sender's buffers. It enables a host that serves many clients (such as a web server) to fairly share the available kernel buffer memory and to provide better throughput than manually configured TCP socket buffers. This is called *socket buffer autotuning* and is important as the assignment of large default socket buffer sizes for all TCP connections would result in the reduction of physical memory.

For applications that do not explicitly set the TCP send buffer size via the `setsockopt()` options, socket buffer autotuning allows the sender socket buffer to grow with *cwnd* and utilise the available bandwidth of the link (up to the receiver's advertised window). As the number of flows increases,

consuming physical memory, the kernel also facilitates the reduction of the sender's socket buffer window sizes to reduce the burden of memory swapping.

The modification of socket buffer auto-tuning algorithm were ported to the Linux 2.4 kernel in 2001 and are controlled by new kernel variables `net.ipv4.tcp_rmem` and `net.ipv4.tcp_wmem` defining the minimum, default and maximum socket buffer memory allocated to each TCP connection for the read and write buffers respectively[3].

Independent of auto-tuning, the Linux 2.4 and 2.6 kernels implement a system where connection performance histories of TCP transfers are cached for up-to 10 minutes. This 'retentive' TCP stores details of the measured / experienced smoothed RTT and RTT variance, *ssthresh* and the path reordering information for previously connected pairs of hosts. These values are stored for up-to 10 minutes and will prevent a consecutive TCP flow for that pair to reach congestion avoidance quickly without the excessive packet losses of a typical slow start. It will also aid in preventing early spurious timeouts. The cache can be prematurely cleared through the use of the `net.ipv4.route.flush sysctl`.

5.3 Network Aid in Congestion Detection

Due to the transient nature of network utilisation, TCP gently probes the network with increasing number of segments to determine whether it can achieve more throughput. Therefore, in a sense, TCP has to cause congestion in order to determine the capabilities of the network path.

Most current routers in TCP/IP networks do not detect incipient conges-

[3]Note, however, that the maximal tunable socket buffer memory is still limited by the maximum allocated socket buffer size in the kernel under net.ipv4.tcp_mem, net.core.rmem_max and net.core.wmem_max.

tion. When a queue overflows packets are dropped, and reliable transport protocols, such as TCP, will attempt to retransmit missing data.

The design of router queue provisioning is often not to increase the utilisation of a single flow going through their network, but that of the aggregate. Given this and the limited resources of physical memory, it is often not possible to provision large buffers required for high throughput, high latency transport as required by Equation 4.1 and Equation 5.4.1. Also, with the increased popularity of VoIP and similarly delay sensitive flows [KBS+98, JS00], it may not be feasible to utilise a large buffer, which is necessary for efficient high throughput transport due to the increased end-to-end latencies of large router queues. More important, however, is that a small queue is insufficient to handle the bursts of packets that are commonplace on the Internet [LTWW94].

This section looks at providing feedback and queueing disciplines from routers that may improve the performance of TCP.

5.3.1 Explicit Congestion Notification

In TCP, a packet loss is an implicit notification of congestion. Through the signaling of an Explicit Congestion Notification (ECN) [RFB01] packet, the TCP sender should reduce the sending rate to mitigate the possibility of congestion.

Designed to be complemented with Active Queue Management techniques (see Section 5.3.2) on routers, ECN is implemented as the last two bits of the IPv4 TOS octet in the TCP header and implemented such that TCP can respond to congestion without actually suffering from packet losses.

When experiencing the incipient stages of congestion, an ECN enabled

router should mark the TCP packets with a Congestion Experienced (CE) bit pattern and then process it normally. Upon receipt of a packet containing the ECN codepoints, the TCP receiver should echo back the same bit pattern in its *ack*s. Should the received packet contain a CE-bit marked, the TCP receiver should also set an internal flag to indicate that all subsequent *ack*s should also have the CE-bit set to 1.

When the TCP sender receives a CE marked *ack*, it should immediately reduce the transmission rate according to congestion control principles. However, the signaling of many CE packets must be handled such that TCP reacts at most once per window (or approximately once every RTT) to prevent successive reductions in *cwnd* [Flo94a].

The TCP sender, after reducing its transmission rate as a consequence of a CE packet, marks the Congestion Window Reduced (CWR) flag in all subsequent data packets on the first bit of the ECN codepoint. The TCP receiver will only stop sending CE marked *ack*s once it has received a packet with the flag CWR.

The use of this field requires that all routers, switches and firewalls must not modify packets with ECN-bits unless the node is under congestion. Also, nodes that do not implement ECN should still pass ECN bits unaltered and should not consider packets with ECN bits set as malformed and drop the packets.

ECN has the obvious advantages in avoiding unnecessary packet drops due to lack of memory. It also avoids the excessive delays necessary for duplicate acknowledgments and retransmission timeouts. Indeed, experimental validations of ECN have shown that TCP achieves moderately better throughput [H.K98, SA00]. It also enables the efficient usage of bandwidth on the forward path of the data transfer and hence improves the responsiveness

of the TCP flow when a high fraction of losses is less likely.

5.3.2 Active Queue Management & Random Early Detection

The prevalence of drop-tail routers in the Internet can result in 'lock-up' [FJ92] and the unfair sharing of network resources. The former occurs when a high throughput flow starves other competing (possibly lower throughput) flows of any queue occupancy due to the fact that a higher throughput flow is more likely to have a larger number of packets in an ingress queue. When 'lock-up' does not occur, higher throughput flows still occupy a majority of the queue, hence being unfair to other flows sharing the bottleneck.

The Random Early Detection (RED) [FJ93] algorithm solves the problems of over-buffering and fair sharing of buffer resources. [BCC98] recommends that RED is used as the default queue management algorithm on the Internet as there apparently no disadvantages and numerous advantages [FJ93].

Active queue management attempts to prevent queue overflow by selectively dropping packets in a queue. It also has the advantage that TCP flows do not have to induce a buffer to overflow as an indication of the congestion.

RED is termed an Active Queue Management solution as a RED enabled router detects the incipient stages of congestion by observing the exponentially weighted moving average of the queue size.

Defined with `minth` and `maxth`, the two values define the thresholds from which no packets should be dropped and where all packets should be dropped respectively. Therefore, the average queue occupancy, `q`, should fall within `minth` and `maxth`. This enables the unavoidable bursts of packets in the Inter-

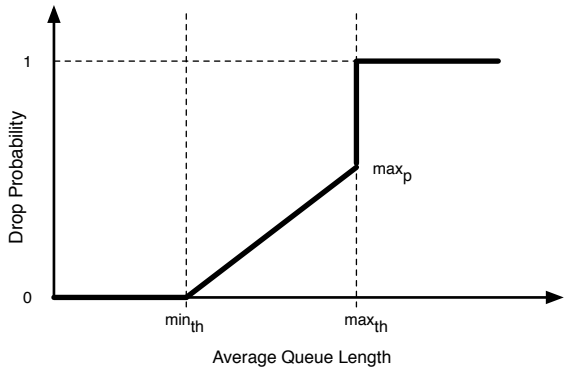


Figure 5.8: Random Early Detection Parameters.

net [BCC98] to be accommodated without any bias. The above parameters are combined with `maxp` which sets the maximum probability that a packet is to be dropped at the router. It is defined as a function of the average queue size and is set to a linear dropping probability from 0 to `maxp`,whereby arriving packets are marked depending on `q`. Should `q` be less than `minth` upon an arriving packet, the packet will be processed normally. However, if `q` is greater than `maxth`, then the packet will be dropped. Otherwise, the incoming packet will be dropped according to the probability related to the average queue occupancy. This is shown in Figure 5.8.

Therefore, the router selectively drops packets from the queue based on the various parameters and occupancy of the queue. RED is able to achieve fairness between flows as higher throughput connections will have a greater occupancy in the queue, and hence a greater chance of packet drops.

The use of RED can also reduce the possibility of TCP synchronisation whereby all flows experience drops at the same time and then simultaneously increases their rates, only to cyclically have all flows experience packet loss. The occurrence of synchronisation results in poor throughput and low link utilisation [ASA00].

RED is wholly implemented on routers and therefore requires no modification on TCP hosts. The exception is when RED is used in conjunction with ECN (see Section 5.3.1) whereby a marked packet will be set to include the CE-bit rather than be physically dropped by the router in order to mitigate congestion.

5.4 Analysis of AIMD Congestion Control

For bulk transport of data, the dominant factor in the rate at which data is transported is the congestion avoidance algorithms of TCP. Unfortunately, the congestion control mechanisms of TCP constrain the congestion windows that can be achieved by TCP in realistic environments, and hence the achievable throughput.

The performance of a very long bulk transfer TCP connection is dependent on the packet loss rate, bandwidth and round trip time, and of a connection. These are considered in turn:

Packet Loss Rates Standard TCP implementations can only reach the large congestion windows necessary to fill a an end-to-end path with a high bandwidth delay product when there is an exceedingly low packet loss rate. As standard TCP implementations cannot distinguish between real network loss (e.g. corruption) and losses due to network congestion[4], the underlying physical network has to be able to maintain very low loss rates in order for TCP not to invoke congestion avoidance when a packet is lost through corruption rather than congestion.

[4]Without network feedback such as ECN which would require deployment throughout bottleneck routers.

For example, in real life systems, a packet loss rate of about 10^{-7} is comparable to the random losses that occur in long haul fibres, routers, switches and end-systems. This physical limit imposes an absolute limit on the throughput that a TCP connection can achieve - even without competing traffic on the end-to-end path (see Section 5.4.2).

Bandwidth The higher throughput that the TCP connection has to sustain, the larger its *cwnd* for the connection. With the implementation of the current Additive Increase of 1 segment per round-trip time, TCP probes for extra network capacity slowly, and therefore is unable to effectively use available network resources that are present on high bandwidth networks that are not fully utilised.

Also, the detection of packet loss through congestion or network corruption in TCP results in a halving of its *cwnd*. When high speeds are reached this mechanism of Multiplicative Decrease by half leads to a significant decrease in the throughput of the TCP connection.

As TCP congestion control is cyclic (i.e. it will continue to slowly increase its sending rate until congestion/loss is detected, where upon it will halve the rate) the overall utilisation of the link by a single TCP flow could be low compared to the capacity of the bottleneck link on the path.

Round-Trip Time TCP utilises acknowledgments to update its window based algorithms. This provides a feedback loop that drives the linear increase of *cwnd* to increase its throughput. As the round-trip latency increases, the time required for this feedback also increases. This results in a slower rate of increase (in absolute time) of *cwnd* and therefore a

slower rate of throughput increase.

This is especially evident after congestion as TCP halves *cwnd*. Therefore, when a TCP connection runs over a low loss, un-congested link it is unable to achieve high throughput in high-speed, wide area networks due to the long recovery times and large value of optimal *cwnd* required after congestion detection when *cwnd* is halved.

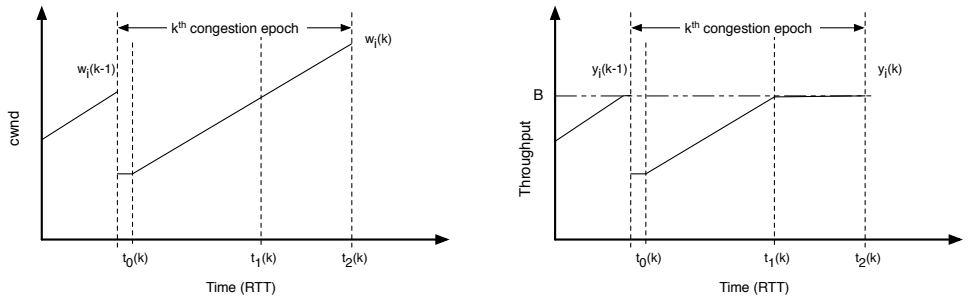
Therefore, as the geographical distance of the TCP connection increases, the time required to reach congestion (assuming static networks conditions) is proportionally increased and its average utilisation is decreased.

To give context to these problems, a standard TCP connection with 1500B packets and a 100ms round-trip time (typical trans-Atlantic connection), would require an average congestion window of 83,333 segments and a packet drop rate of at most one congestion event every 5 billion packets to achieve a steady-state throughput of 10Gbit/sec. This requires at most one congestion event every 1 hour and 40 minutes. This loss rate (which is assumed to be related directly to the Bit Error Rate) is well below what is possible today with the present optical fibre and router technology.

5.4.1 Throughput Evolution

A model of the algorithmic dynamics of TCP can be gathered through simple geometry when considering the steady state behaviour of TCP as a cyclic iteration over k congestion epochs.

Figure 5.9 shows the evolution of *cwnd* over time. Assuming that the time for recovery and the time required to signal the loss is negligible, the



(a) *cwnd* (b) Throughput

Figure 5.9: Evolution of TCP congestion window *cwnd* and throughput w.r.t. time.

cwnd dynamic of the i^{th} flow at the end of the k^{th} congestion epoch, $w_i(k)$, can be represented as:

$$w_i(k+1) = \beta w_i(k) + \alpha T(k) \tag{5.1}$$

where α and β are the increase and decrease parameters of Additive Increase Multiplicative Decrease (AIMD) such that $\alpha = 1$ and $\beta = 0.5$ (See Section 4.3).

Let $t_0(k)$ be the start time of the k^{th} congestion epoch. At time $t_1(k)$ the queue begins to fill, and at a subsequent time $t_2(k)$ congestion occurs and a packet is lost. For the period $t_1(k) - t_0(k)$, the number of packets placed into the network is related to the increase parameter $\alpha \times T$ where T is the RTT of the flow. After $t_1(k)$, putting extra packets into the network is simply buffered in the bottleneck queue of size q, whilst the throughput of the flow remains at the line rate B. At time $t_2(k)$ the queue overflows and congestion is assumed to be signaled immediately, causing the TCP flow to back-off its throughput such that the new throughput is $\beta \times P$ where P is the "pipe" $P = BT + q$, i.e. the Bandwidth Delay Product (See Section 4.2.2) worth of

packets traversing along the network, plus the packets buffered at the queue.

Given this, the time taken to just fill the pipe after the k^{th} epoch can be determined by considering the rate at which packets are put onto the network:

$$\begin{aligned}(\beta P) + \frac{\alpha}{T}(t_1(k) - t_0(k)) &= BT \\ t_1(k) - t_0(k) &= \frac{T}{\alpha}(BT(1-\beta) - \beta q)\end{aligned} \tag{5.2}$$

At period $t_1(k)$ to $t_2(k)$, all subsequent packets are buffered at the queue such that:

$$\begin{aligned}\frac{\alpha}{T}(t_2(k) - t_1(k)) &= q \\ t_2(k) - t_1(k) &= \frac{Tq}{\alpha}\end{aligned} \tag{5.3}$$

And the time between congestion epochs is defined as,

$$\begin{aligned}\frac{\alpha}{T}(t_2(k) - t_0(k)) &= P - \beta P \\ t_2(k) - t_0(k) &= \frac{T}{\alpha}(P - \beta P)\end{aligned} \tag{5.4}$$

Defining the provision parameter $\gamma = q_{max}/BT$ and that $1/1+\gamma \leq \beta$, i.e. the queue empties on backoff, the throughput, y, can be calculated as the area under the graph between $t_0(k)$ and $t_2(k)$ where the area of the triangle before the queue begins to fill, $(t_1(k) - t_0(k) \times \beta B$, is subtracted from the area for the entire epoch, $(t_2(k) - t_0(k)) \times B$.

$$y = \frac{B^2T^2(\beta-1)(1+\gamma)}{\alpha} - \frac{B^2T^2(\beta\gamma+\beta-1)}{2\alpha} \tag{5.5}$$

Normalising by the total number of packets sent out (i.e. the total area

of the graph) provides the utilisation of the flow:

$$\eta = \frac{\beta^2(1+\gamma)^2 - 2\gamma - 1}{2(\beta-1)(1+\gamma)} \tag{5.6}$$

As such, Equation 5.6 states that the utilisation of the link of capacity B is dependent on the queue size provision γ and the back-off factor β of the TCP flow *only*. This assumes that the only form of congestion signal is that from the overflow of the buffer from this flow only (i.e. zero packet loss) and that the number of congestion epochs experienced by the flow is sufficiently large.

Therefore, substituting the default decrease parameter of $\frac{1}{2}$ and a queue provision of zero gives 75% utilisation of the bandwidth B. Similarly, with a queue provision γ of 1 (i.e. $q = BT$, the Bandwidth Delay Product), the flow will experience 100% utilisation with $\beta = 1/2$.

5.4.2 Response Function

Mathematical models of the evolution of a TCP flow have been developed in order to provide researchers with tools to accurately design new models and to modify new congestion control models.

One of the most popular is the response curve [PFTK00] that defines the bulk transfer throughput of TCP Reno for different loss conditions. Unlike previous models, Equation 5.7 predicts the throughput of TCP under both timeouts and congestion avoidance and is given by,

$$B = \frac{s}{T\sqrt{\frac{2p}{3}} + T_0\left(3\sqrt{\frac{3p}{8}}\right)p(1+32p^2)} \tag{5.7}$$

where the throughput B is related to the RTT, T, the timeout, T_0, under loss rates, p using packets of size s. It therefore demonstrates that the throughput is increased for low latency flows experiencing low loss conditions.

Considering the case of large file transport where the TCP flow will spend most of the time in congestion avoidance, Equation 5.7 can be approximated to,

$$cwnd = \sqrt{\frac{3}{2p}} \text{ packets} \tag{5.8}$$

where the Bandwith Delay Product, $cwnd = B \times T$ (Equation 4.1) is used to relate the throughput and the *cwnd* by the RTT. This is equivalent to the models presented in [Flo91, LM97, MSMO97].

The importance of this model is that the throughput is inversely related to the loss rate experienced by the TCP flow. Therefore, a fundamental limit on throughput of a TCP flow exists even without congestion due to the physical existence of bit error rates on the Internet. Given a typical link loss rate of 10^{-7}, and a typical long distance latency of 100ms, TCP is only capable of achieving approximately 450Mbit/sec.

5.4.3 TCP Generalisation

[JDXW03] considers the evolution of *cwnd* as a stability problem whereby *cwnd* oscillates around an equilibrium position. They specify that the increase of 1 packet per RTT results and the decrease of half results on a flow level description of the *cwnd* (w) as:

$$\frac{dw_i(t)}{dt} = \frac{1}{T_i(t)} - \frac{w_i(t)}{T_i(t)} p_i(t) \frac{1}{2} \frac{4}{3} w_i(t) \tag{5.9}$$

Where the first term comes from the linear increase of w and the second

term is derived from the congestion measure[5] $p_i(t)$ and a drop of half from the peak *cwnd* value of $\frac{4}{3}w_i(t)$.

It can be shown that Equation 5.9 readily reduces to Equation 5.8 when $\dot{w}_i(t) = 0$.

By defining that:

$$\kappa_i(T_i) = \frac{1}{T_i} \text{ and } u_i(w_i(t), T_i(t)) = \frac{3}{2w_i^2} \tag{5.10}$$

[JDXW03] stipulate that all the design of congestion algorithms can be generalised by the following:

$$\dot{w} = \kappa_i(t) \times \left(1 - \frac{p_i(t)}{u_i(t)}\right) \tag{5.11}$$

where $\kappa(t) := \kappa(w_i(t), T_i(t))$ and $u(t) := u(w_i(t), T_i(t))$ and is defined as the *gain function* and the *marginal utility* function respectively.

By choosing an appropriate value of $u(t)$ which is equivalent to the loss probability, the time variance of w_i can be minimised and therefore enable a more 'stable' flow. Meanwhile, the choice of $\kappa(t)$ facilitates a quick adaptation to the equilibrium value of w.

Therefore, the choice of the variables $\kappa(t)$ and $u(t)$ play an important role in the stability and responsiveness, and the equilibrium properties respectively.

[5]Which for this discussion is equivalent to the the indication of packet loss.

5.5 Summary

Even though TCP has been a fundamental part of the success of the Internet, enabling reliable data transfers and preventing congestion collapse, it was never designed to be scalable in terms of throughput. As such TCP has problems with long distance, fast networks which will form the backbone of future grids in order for computers to interoperate and share data files.

There has been a recent interest in utilising the new backbone speeds available in today's Academic and Research networks, especially in terms of large scale data replication that is required for projects such as the LHC [CFK+01, SRGC00]. As such, network researchers have pushed the boundaries on tuning host hardware and software to facilitate high utilisation of network resources. Many of the techniques used have been studied in detail in this Chapter.

However, [ST] states that 90% of bulk transports (i.e. those over 10MB) typically achieve transfer speeds of less than 5Mbit/sec. Furthermore, Equation 5.6 shows that the maximum utilisation of a single TCP flow has a theoretical limit of 75% when there is zero buffering. Full utilisation is only achieved when the queue size is equal to the BDP of the TCP flow.

Loss is also a fundamental limit on the throughput achievable with Standard TCP. It was shown that the maximum throughput of a typical trans-Atlantic transfer *without competing flows sharing the network path* is only 450Mb/sec - because of physical limits upon bit error rates.

Therefore, it has been demonstrated both experimentally in Section 3.2.2 and theoretically that TCP is simply incapable of high throughput in long distance, high capacity paths - regardless of the amount of tuning performed outside of the AIMD algorithms.

Chapter 6

New-TCP Transport Algorithms

TCP is the most widely used transport protocol with 'network friendly' congestion control mechanisms. It has been extensively studied, extended, and refined in the last two decades and has proven to be reliable under a variety of network conditions and applications requirements. Increasingly, however, the standard TCP implementations are becoming inadequate to support high-performance distributed applications emerging in the science community. These science applications are expected to generate petabits per second of traffic to be distributed to scientists in different geographical location [CFK+01, SRGC00].

The previous chapters have shown that the practical throughput of TCP's default AIMD parameters over paths with a large Bandwidth Delay Product (BDP) are far below the supportable link speeds of the underlying network. Therefore, further research and development efforts are required to enhance

TCP so that it will deliver and sustain many gigabits per second for scientific applications.

The requirement for adapting the TCP algorithms, rather than inventing and deploying a completely new transport protocol are two fold:

- TCP has been proven to be remarkably robust over the evolution of the internet and has prevented network collapse. It already offers many facilities that are deemed essential for most data transport applications such as reliable data delivery, flow control and congestion control.

- It is very likely that TCP will play a continued role as the transport protocol of choice due to the requirements of web, file and mail delivery on the Internet. The development of new transport protocols will also require the phased introduction of new applications to support such protocols.

This chapter gives an overview of different proposed changes to the TCP AIMD algorithms to enable high speed transport.

6.1 Survey of New-TCP Algorithms

Several proposals have been made that offer solutions to obtaining high throughput transport in high-speed and high delay environments. The survey presented here gives an overview of the many different approaches to enable higher speed communications through the adaptation of the TCP congestion control algorithms.

Amongst those, only algorithms that require sender side modifications to the TCP stack are presented. This is important as it allows compatibility with existing TCP implementations and would ease deployment.

6.1.1 HighSpeed TCP

The HighSpeed TCP for Large Congestion Windows was introduced in [Flo03] as a modification of TCP's congestion control mechanism to improve performance of TCP connections with large congestion windows.

The strategy under HSTCP is to slowly improve the performance of the TCP flow as a function of the current *cwnd* such that when the *cwnd* is larger, the increase parameter is increased and reduction of *cwnd* upon loss detection is decreased. Therefore, as the HSTCP flow deviates from the expected achievable *cwnd* of standard TCP, both the reduction in the change of throughput upon congestion is decreased, and the rate of increase at which HSTCP probes for extra bandwidth increases.

HSTCP is based on the adaptation of the response function in Equation 5.7 such that the high throughputs are achievable under realistic loss conditions. The HSTCP response function is defined using three parameters: W_{switch}, W_{High} and P. They are used to establish a point of transition from standard TCP behaviour to HighSpeed TCP behaviour.

As such, HighSpeed TCP's response function is the same as that of the Standard TCP when the current congestion window, *cwnd*, is less than or equal to W_{Switch}. HighSpeed TCP uses an alternative response function when the current congestion window is greater than W_{Switch}. W_{High} and P are used to specify the upper end of this response function, where the packet drop rate P is that needed to achieve an average congestion window of W_{High} under the HighSpeed TCP regime. Also, $0 < H_{decr} \leq 0.5$ is specified such that when $cwnd = W_{High}$ and a congestion event occurs, *cwnd* will be decreased by a factor of H_{decr} such that $cwnd \leftarrow cwnd - H_{decr} \times cwnd$.

The HighSpeed TCP response function is therefore represented by new

additive increase and multiplicative decrease parameters which change according to the current value of *cwnd*. These translate to new additive increase and multiplicative decrease parameters α and β as such:

$$\text{ACK:} \quad cwnd \leftarrow cwnd + \frac{\alpha(cwnd)}{cwnd} \tag{6.1}$$

$$\text{LOSS:} \quad cwnd \leftarrow \frac{\beta(cwnd)}{cwnd} \tag{6.2}$$

where $\alpha(cwnd)$ is a function of cwnd and $\alpha = 1$ when $cwnd \leq W_{Switch}$ and $\alpha > 1$ when $cwnd > W_{Switch}$. $\beta(cwnd)$ is also a function of *cwnd* and $\beta = 0.5$ [1] when $cwnd < W_{Switch}$ and $\alpha > 1$ when $cwnd \geq W_{Switch}$ with $0.5 \leq \beta < 1$:

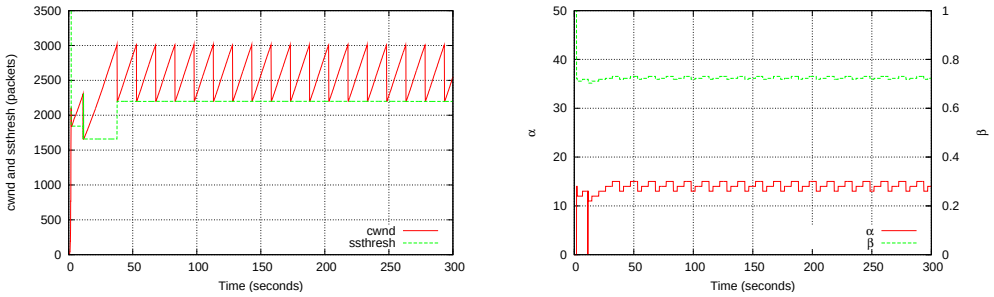
$$\alpha(w) = \frac{W_{High}^2 \times P \times 2 \times (1 - \beta(w))}{2 - (1 - \beta(w))} \tag{6.3}$$

$$\beta(w) = 0.5 - \frac{(H_{decr} - 0.5) \times (log(w) - log(W_{Switch}))}{log(W_{High}) - log(W_{Switch})} \tag{6.4}$$

The recommended settings of W_{Switch}, W_{High}, P and H_{decr} are 38, 83,000, 10^{-7} and 0.1 respectively. The use of these settings for HighSpeed TCP will be henceforth be referred to as HSTCP.

Figure 6.1 shows a typical *cwnd* trace of a single flow of HSTCP. Due to the compute cost of calculating a continuous function for the values of $\alpha(cwnd)$ and $\beta(cwnd)$, it is suggested that implementations should use a look-up table to determine the appropriate values [Flo03]. The dynamic change of the AIMD parameters is shown in Figure 6.1(b) - it clearly shows the adaptation of the increase and decrease parameters to make the HSTCP algorithm more aggressive as the value of *cwnd* increases.

[1] Traditionally, represented as $cwnd \leftarrow cwnd - \frac{b(cwnd)}{cwnd}$. However, for consistent comparision against other New-TCP algorithms, define $\beta(cwnd) = 1 - b(cwnd)$.



(a) Congestion Window (b) AIMD Parameters

Figure 6.1: *cwnd* dynamic of HSTCP (Single flow on Dummynet, link capacity 200Mbit/sec, RTT 150ms, queue size 500 packets.)

6.1.2 ScalableTCP

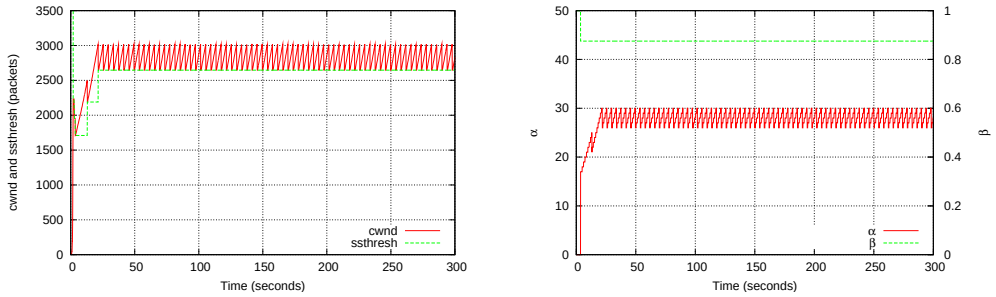
ScalableTCP [Kel03] defines a mechanism by which the number of RTTs to recover from a loss event is independent of the of the link bandwidth. The design of ScalableTCP ensures that the recovery time required at any throughput is approximately a constant regardless of the RTT of the connection. It therefore ensures a degree of scalability of the algorithm regardless of the link capacity of the path that enables ScalableTCP to outperform standard TCP.

The generalised ScalableTCP window update algorithm responds to each acknowledgment received in which congestion has not been detected with the update:

$$\text{ACK:} \quad cwnd \leftarrow cwnd + \alpha \tag{6.5}$$

$$\text{LOSS:} \quad cwnd \leftarrow \beta \times cwnd \tag{6.6}$$

where α is a constant with $0 < \alpha < 1$ and β is a constant with $0.5 < \beta < 1$. The scaling property applies for any choice of the constants α and β. The



(a) Congestion Window (b) AIMD Parameters

Figure 6.2: *cwnd* dynamic of ScalableTCP (Single flow on Dummynet, link capacity 200Mbit/sec, RTT 150ms, queue size 500 packets.)

recommended values of $\alpha = \frac{1}{100}$ and $\beta = \frac{7}{8}$ are used.

In order to provide compatibility with Standard TCP, a variable W_{Switch} is defined to determine the mode switch point from Standard TCP behaviour to Scalable TCP behaviour. It is defined in terms of the *cwnd*, so that when $cwnd \leq W_{Switch}$ Standard TCP is invoked. The recommended value of $W_{Switch} = 16$ segments is used.

Figure 6.2 shows the *cwnd* evolution of ScalableTCP as a function of time. The small decrease of *cwnd* upon packet loss guarantees high throughput for individual flows. Meanwhile, the multiplicative increase means that approximately a constant number of RTTs would be required to reach the value of *cwnd* just before congestion.

The scaling property of ScalableTCP means that it implements an exponential increase in the sending rate when no loss is detected. This kind of increase, with the associated multiplicative decrease, is known as Multiplicative Increase Multiplicative Decrease (MIMD). [CJ89] states that MIMD based algorithms do not converge to fairness in drop-tail environments.

6.1.3 H-TCP

H-TCP [LS04a] was designed to address the issues of the slow transient convergence between similar flows of HSTCP and ScalableTCP.

H-TCP is based on the same idea of HSTCP and ScalableTCP such that the rate at which TCP allows packets into the network should reflect the prevailing network conditions. What is unique about H-TCP is the fact that it incorporates a mode-switch between a Standard TCP regime and that of a faster mode for each and every congestion epoch (whereas HSTCP and ScalableTCP incorporates a switch from Standard TCP mode to their relevant 'high-speed' modes if the *cwnd* is larger than some value).

The H-TCP can be encompassed in the following algorithm:

$$\text{ACK:} \quad cwnd \leftarrow cwnd + \frac{2(1-\beta) \times \alpha(\Delta)}{cwnd} \tag{6.7}$$

$$\text{LOSS:} \quad cwnd \leftarrow \beta \times cwnd \tag{6.8}$$

with

$$\alpha(\Delta) = \begin{cases} 1 & \Delta \leq \Delta^L \\ max(\bar{\alpha}(\Delta)T_{min}, 1) & \Delta > \Delta^L \end{cases} \tag{6.9}$$

$$\beta = \begin{cases} 0.5 & |\frac{B(k+1)-B(k)}{B(k)}| > \Delta_B \\ min(\frac{T_{min}}{T_{max}}, \beta_{max}) & otherwise \end{cases} \tag{6.10}$$

Δ is the (real) time since the last congestion epoch and Δ^L is a constant parameter. T_{min} and T_{max} are the flows' minimum and maximum experienced latencies and $B(k)$ and $B(k+1)$ are the measured goodputs of the H-TCP flow at the moment of congestion for the previous and current epoch respectively.

Δ_B is a design parameter.

By increasing the value of $\alpha(\Delta)$ such that it slowly increases as a function of time, H-TCP is able to switch in to an aggressive mode which under heavy congestion is fair with Standard TCP flows, but is able to utilise available bandwidth quickly. This transition into H-TCP's high-speed mode is effected a certain time Δ^L after the last congestion event such that the Standard TCP update algorithm is used while $\Delta \leq \Delta^L$. The suggested value of Δ^L is 1 second.

A quadratic increase function $\bar{\alpha}$ is suggested in [LS04a], such that:

$$\bar{\alpha}(\Delta) = 1 + 10(\Delta - \Delta^L) + 0.25(\Delta - \Delta^L)^2 \tag{6.11}$$

Therefore, the growth of *cwnd* is such that H-TCP operates similarly to Standard TCP in conventional networks where the period between congestion events is small. Under high speed long distance networks, H-TCP switches to a polynomial increase in *cwnd* to take advantage of available capacity where Standard TCP is incapable of high performance. As the mode switch is based on time rather than on the current value of *cwnd*, sources already in a high speed mode do not gain a long term advantage over new flows starting up and therefore guarantee fairness.

The calculated increase parameter $\bar{\alpha}(\Delta)$ is multiplied by the minimum experienced latency, T_{min}, of the flow in order to facilitate fairness between competing flows with different latencies [LS04b]. This feature is called RTT Scaling.

In order to utilise the capacity effectively, H-TCP defines a back-off factor $\beta(k)$ derived from the desire to maintain a queue occupancy after congestion such that the queue is not empty for too long. Under the assumption of the

validity of Equation 4.1, [LS04a] states that this is successful when the maximum queue size is equal to the maximum RTT measured and the minimum (zero) queue size occurs when the flow experiences the minimum RTT.

The bandwidth delay products before and after congestion are:

$$\text{Before congestion:} \quad R(k)^{-} = \frac{cwnd}{RTT_{max}} \tag{6.12}$$

$$\text{After congestion:} \quad R(k)^{+} = \frac{\beta \times cwnd}{RTT_{min}} \tag{6.13}$$

A simple way of ensuring full utilisation of the network by this single H-TCP flow would be to equate $R(k)^{-}$ and $R(k)^{+}$. As such, this constraint can only be achieved by enforcing that:

$$\beta(k) \geq \frac{RTT_{min}}{RTT_{max}} \tag{6.14}$$

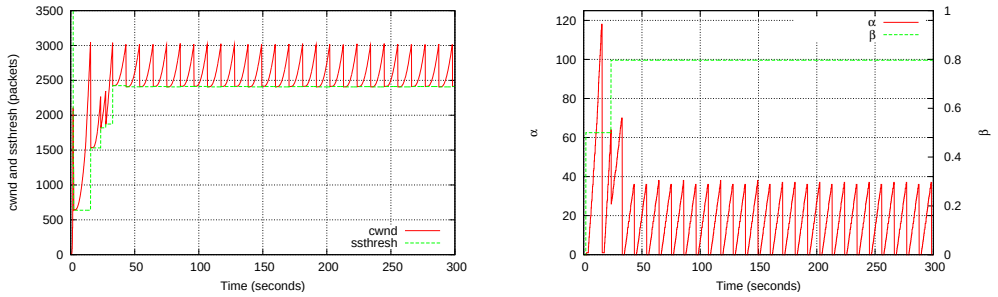
where $\beta(k)$ is the value of β at the k'th congestion epoch. The calculation prevents the queue size from emptying upon loss/congestion detection by the TCP flow and therefore maintain maximum throughput for the TCP flow.

However, high values of β would cause slow convergence between flows. Therefore, a parameter β_{max} specifies the largest value of β such that $0.5 \leq \beta \leq \beta_{max}$ to enable compatibility with Standard TCP whilst being able to keep high utilisation of the network path.

To further aid the fair sharing of network bandwidth, H-TCP specifies that should the measured throughput of a flow at the next congestion epoch be greater than the parameter Δ_B, then β should default back to 0.5.

Δ^B and β_{max} have suggested values of 0.2 and 0.8 respectively.

Figure 6.3 shows the H-TCP algorithm in action. It shows the slow 'TCP-friendly' α increase immediately after loss, and then the switch into the high



(a) Congestion Window (b) AIMD Parameters

Figure 6.3: *cwnd* dynamic of H-TCP (Single flow on Dummynet, link capacity 200Mbit/sec, RTT 150ms, queue size 500 packets.)

speed mode a period Δ_i later. After this period, the growth of *cwnd* is polynomial in nature, reaching very high values before loss is induced and the cycle begins again. The adaptive backoff is also shown, with an initial value of 0.5 and later set to 0.8 due to the (relatively) large queue which means that the ratio of the minimum and maximum latencies is capped by the algorithm.

6.1.4 BicTCP

BicTCP [XHR04] attempts to address an issue with HSTCP and ScalableTCP whereby competing flows would starve the throughput of a Standard TCP flow due to the faster growth of *cwnd*.

The implementation of BicTCP is based around two separate algorithms; A binary search to determine the optimal *cwnd* size, and an additive increase to ensure fast convergence and RTT-fairness.

Binary search is used to determine whether the current sending rate (or window) is larger than the network capacity. This is achieved with a variable w_{target} that holds a value halfway between the values of *cwnd* just before and

just after the last loss event. The *cwnd* update rule seeks to rapidly increase *cwnd* when it is beyond a specified distance S_{max} from w_{target}, and update *cwnd* more slowly when its value is close to w_{target}.

The second algorithm complements the binary search algorithm by providing a mechanism to maintain high values of *cwnd* (and hence high throughput) without incurring large values of packet loss. This is implemented by an aggressive additive increase of *cwnd* after congestion such that it can quickly reach w_{target}. The additive increase is controlled by a parameter S_{max} that defines the rate of increase of *cwnd* such that temporal equality of *cwnd* and w_{target} is reached so that *cwnd* does not induce loss as quickly as pure linear increase alone. This also provides a plateau which enables an almost constant value of *cwnd* around the binary searched value of w_{target} and therefore high throughput can be maintained.

Similar to ScalableTCP, a constant multiplicative backoff factor β is used that is greater than Standard TCP's 0.5.

The BicTCP algorithm can be represented as follows:

$$\text{ACK:} \quad \begin{cases} \delta & = \frac{w_{target} - cwnd}{B} \\ cwnd & \leftarrow cwnd + \frac{\alpha(\delta, cwnd)}{cwnd} \end{cases} \tag{6.15}$$

$$\text{LOSS:} \quad \begin{cases} w_{target} & = \begin{cases} \frac{1+\beta}{2} \times cwnd & cwnd < w_{target} \\ cwnd & otherwise \end{cases} \\ cwnd & \leftarrow \beta \times cwnd \end{cases} \tag{6.16}$$

$$\alpha(\delta, cwnd) = \begin{cases} \frac{B}{\sigma} & (\delta \leq 1, cwnd < w_{target}) or (w_{target} \leq cwnd < w_{target} + B) \\ \delta & 1 < \delta \leq S_{max}, cwnd < w_{target} \\ \frac{w_{target}}{B-1} & B \leq cwnd - w_{target} < S_{max}(B-1) \\ S_{max} & otherwise \end{cases} \tag{6.17}$$

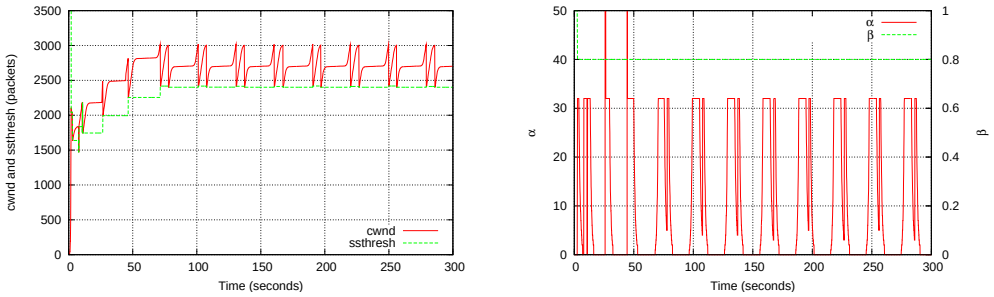
where the parameter B is set to 4.

If BicTCP is successful in maintaining a larger $cwnd$ than w_{target} (as after a while the difference between w_{target} and $cwnd$ will be larger than S_{max} if no losses are experienced), BicTCP initiates slow-start to probe for new parameters to maximise throughput utilisation. During this timed regime, the pre-calculated value of w_{target} may be wrong, and it is therefore recalculated as the maximum $cwnd$ value at the time of packet loss. Therefore, upon the next iteration, $cwnd$ will rise quickly to the new $value$ of w_{target} at the point of loss.

[XHR04] argue that as the design specifies a logarithmic increase in $cwnd$ as it approaches the targeting window sizes, the need to retransmit packets due to losses from aggressive probing is reduced.

Similar to the regimes utilised by HSTCP and ScalableTCP, BicTCP imposes fairness with Standard TCP by utilising the standard AIMD parameters when $cwnd$ is less than 14 packets.

Many varying versions of the BicTCP implementation exist [XHR04, LL04] where $\beta = \frac{7}{8}$ or 0.8 and other features such as 'Low Utilisation Detection' whereby upon detection of low network utilisation, a more aggressive algorithm is implemented to increase utilisation. The version used throughout this study is based on version 1.1 of the official BicTCP implementation and uses a value of $\beta = 0.8$ with Low Utilisation Detection.



(a) Congestion Window (b) AIMD Parameters

Figure 6.4: *cwnd* dynamic of BicTCP (Single flow on Dummynet, link capacity 200Mbit/sec, RTT 150ms, queue size 500 packets.)

Figure 6.4 shows the binary search and increase algorithm of BicTCP. It demonstrates the aggressive probing with the additive increase algorithm followed by the almost zero increase in *cwnd* as *cwnd* approaches w_{target}. There is then consecutive slow start which induces loss and a subsequent aggressive additive increase is initiated to equalise *cwnd* to the value before loss. As a second probing does not facilitate the discovery of a new w_{target}, w_{target} is calculated as the mid-point between the minimum and maximum *cwnd*'s during the second probe and the cycle begins again. At the start of the connection, successful slow start attempts to find a new w_{target} can be seen where the slow start cycle is initiated until loss is detected.

6.1.5 FAST

TCP Vegas [BP95] uses end-to-end delay as a signal to adapt a source's sending rate. Increased end-to-end delay indicates congestion, leading to a reduction in *cwnd*; decreased delay is associated with less congestion and leads to an increase in *cwnd*. The result is a congestion control which attempts to avoid packet loss and has bandwidth allocation properties that

differ from those with traditional congestion control.

FAST TCP [JWL04] is an adaptation of the Vegas algorithm to enable high speed data transport. Its primary difference is that it utilises RTT estimation as its primary feedback from the network. The assumption used is that the queuing delays conveys the 'right congestion information' to maximise the network utilisation and that it scales with network capacity [JDXW03, CL03, PWLD03]. Their argument for a delay-based congestion avoidance algorithm is that at high speeds, loss based algorithms will necessarily change their windows to maintain stable state. Under such circumstances, the evolution of the window through time is defined in Equation 5.9. Using only binary packet loss as an indication of congestion is insufficient to maintain equality between the utility function $u_i(t)$, and the end-to-end congestion feedback $p_i(t)$.

The design of FAST incorporates the utility function such that the window adjustment is small when close to equilibrium, and largely otherwise *independent* of where the equilibrium is. Therefore, the slow convergence properties due to the necessary oscillation of loss based algorithms are eliminated.

On a packet level implementation, the evolution of *cwnd* (w) is restricted such that the calculated optimal window is defined by:

$$\text{ACK:} \quad w \leftarrow min\{2w, (1-\gamma)w + \gamma\left(\frac{T_{min}}{\bar{T}}w + \alpha(w, T_q)\right)\} \quad (6.18)$$

$$\text{LOSS:} \quad w \leftarrow \frac{w}{2} \quad (6.19)$$

where $T_q = \bar{T} - T_{min}$, $\bar{T}$ is the exponentially averaged instantaneous RTT of the TCP connection and T_{min} is the minimum experienced latency. To

eliminate the variance of T_q, it is smoothed such that,

$$\bar{T}(k+1) = \eta(t_k)T(k) + (1 - \eta(t_k))\bar{T}(k) \tag{6.20}$$

where $\eta(t_k) = 50$ is a design paramter of FAST.

Window changes are implemented over two RTTs and adjusted such that at maximum *cwnd* doubles in each RTT (i.e. slow start) such that $\alpha(w, T_q) = aw$ when the queuing delay T_q is zero. However, this growth only occurs until $T_q > 0$, after which $\alpha(w, T_q) = \alpha$ where α is a constant which determines the number of packets per flow that is (attempted to be) maintained in the network buffers at equilibrium.

This rudimentary definition of α means that the bottleneck queue size needs to scale with the number of competing FAST flows. Therefore, in order for n competing FAST flows to reach equilibrium and stabilise, a queue-size of at least $n \times \alpha$ is required. In the current [JWL04] version of FAST, a simple algorithm to tune α based on the achieved throughput is used.

Throughout this study, the default parameters of $\gamma = 50$ is used, and that $\alpha(w, T_q)$ is calculated from the measured goodput of the FAST flow. More specifically, the value of $\alpha(w, T_q)$ changes such that it is set to 8, 20 or 200 when the measured goodput is less than 10Mbit/sec, less than 100Mbit/sec and greater than 100Mbit/sec respectively.

6.2 Discussion and Deployment Considerations of New-TCP Algorithms

The optimisation of high performance, high throughput network transport protocols is very important. There is a clear need to maximise the net-

work transport performance and efficiency in order to maximise the network resource utilisation. As the Standard TCP AIMD algorithms have been proven to be very robust, replacement algorithms for TCP should also be as robust. Furthermore, the development of new transport protocols must be scalable with network capacities and adaptable to different types of transmission medium and different types of applications.

Congestion control can be interpreted as a distributed optimisation problem where sources and links carry out a distributed computation to maximise the sum of individual source utilisations [LPD02, LS03].

As high speeds networks become more prevalent on today's Internet, it is important to be able to utilise the spare capacity quickly. This inherently implies that the deployment of New-TCP algorithms should be compatible with the standard windowing mechanisms of TCP [Pos81b]. All of the algorithms in this survey are sender side modifications to the congestion control algorithm and therefore only requires the modification of the sending host to facilitate high throughput transfer - with sufficient network resources and tweaking as outlined in Chapter 5.

6.2.1 Scalability and Network Utilisation

One problem of Standard TCP being able to attain high speeds is that a very low loss rate is required in order to maintain large values of *cwnd*. As Standard TCP cannot differentiate between packets lost to congestion and that lost to transmission errors this becomes a serious problem on lightly loaded networks.

However, the primary reason for requiring a replacement congestion control algorithm is that Standard TCP, with its linear *cwnd* increase, simply

does not scale to utilise available bandwidth.

As higher capacities and latencies become more common, it is important the dynamic by which TCP determines that there is spare capacity is sufficiently scalable. A trivial solution to this problem is to shorten the congestion epoch time. This can be achieved by increasing the value of α such that the rate of growth of the *cwnd* is increased. Therefore, large increases in the probing rate will facilitate the utilisation of spare capacity over short timescales. This is most evident with ScalableTCP and H-TCP; ScalableTCP with its multiplicative increase ensures that the next congestion epoch is reached quickly.

However, especially for loss-based algorithms, this large increase in *cwnd* will result in a higher probability multiple lost segments within the same window which have to be retransmitted. The result of this retransmission may result in much burstier flow and could lead to congestion collapse. It is therefore prudent to minimise the amount of data that needs to be retransmitted when congestion occurs [KRB05].

FAST minimises this requirement by slowly adjusting its optimal *cwnd* value according to multi-bit feedback from network delay. Meanwhile, BicTCP increases the utilisation of network resources by maintaining a similar plateau for its *cwnd* dynamic as it finds its w_{target} value.

However, as defined in Equation 5.6, the actual time independent (i.e. sufficiently long) utilisation of a flow depends upon not the increase parameter, but the algorithm's decrease parameter β assuming a linear increase in α.

Obviously, the decrease of half used in Standard TCP is too severe at high speeds. The simple solution would be to minimise this decrease to say, $\frac{7}{8}$ as with ScalableTCP. This would therefore result in a network utilisation

of 93.75%[2], rather than 75% for Standard TCP with zero buffering along the network path.

However, with an increase in the value of the decrease parameter to maximise utilisation, comes the problem of interaction between similar flows. As this decrease applies only every congestion event, a pair of flows with the same end-to-end latency and sharing the same bottleneck will take a greater number of congestion events before they equalise to a fair share of the network capacity.

The matter in which the capacity is shared by the flows is broadly known as fairness [CJ89, BG87, Kel97, LS04b]. Two generally used fairness criteria are *max-min* fairness and *proportional* fairness.

The max-min fairness concept [BG87, Jaf81] emerged from fair queuing models whereby routers utilise a separate queue per flow and implement the round-robin model to serve the flows in turn. Therefore, under this model, the flows that send less data receive higher priority than flows that send bulk data. Conversely, proportional fairness favours the flows that have more packets over those with fewer packets. Proportional fairness is useful for designing pricing models for the Internet [Kel97].

[XHR04] points out that more aggressive TCP leads invariably to unfairness between the flows, unless care is taken.

Should the competing flows be synchronised in terms of network congestion events, then this time variant nature of converging to a fair share is directly related to the value of β; if β is large, then many congestion events are required before convergence; if β is small, then convergence is reached quickly - but at the expense of reduced utilisation.

H-TCP directly considers this convergence to fairness by measuring the

[2]Actually less due to the increased growth rate of α with respect to time.

throughput at congestion: should the change of throughput at consecutive congestion points be larger than 80%, then it sets β to 0.5 (as with Standard TCP) to aid faster convergence to fairness.

Another important factor of scalability is the adaptation of the algorithms at different network capacities (or more accurately with large *cwnd* values). The design of HSTCP is explicitly such that you have to 'tune' the increase and decrease parameters depending on the required loss rates against the response function (See Section 5.4.2). The current specification of HSTCP suggests tuning appropriate for 10Gb/sec network flows. Whilst Section A.2.3 suggests that it is currently not possible to reach such speeds, even with the best currently available hardware, with time this is sure to change. As such, when it becomes possible to enable network speeds greater than 10Gb/sec (or in the unlikely even that the Internet becomes more lossy), then all deployed versions of HSTCP will need to be 're-tuned' to suit the new networks available.

All of the other algorithms have no such problem in terms of utilisation. However, with the larger *cwnd* values, an aggressive increase parameter may require extra buffering to accommodate the extra packet bursts due to aggressive algorithms, which may lead to a fundamental limit to the scalability of the algorithm. This feature is most evident for ScalableTCP which has a consistent large increase parameter.

FAST has a theoretical utilisation of 100% [JWL+03, JWL04]. However, a problem with the use of latency measurements to determine the incipient stages of congestion is that unless one-way delay is used, both the forward and reverse paths of the connection's latency is measured. As such, should the reverse path be congested, rather than the forward path, then FAST (and TCP Vegas) suffers from low throughput as it cannot differentiate between

the congestion points [GM, BC]. Another problem related to this is the prevalence of asymmetric links on the Internet [BPK99] which means that the actual calculation of the bandwidth delay product could be wrong and hence also affect the calculation of the appropriate window size. Indeed, it is paramount that TCP Vegas and FAST maintain some indication of the RTT through the injection of packets in order to successfully operate, otherwise persistent congestion may occur [LWA98, JWL+03].

However, the same problem of avoiding the induction of packet loss on a network also prevents algorithms like Vegas and FAST from being able to compete with other loss-based algorithms on the same link. As loss based algorithms only decrease their sending rates in response to loss, in order for FAST to compete comparatively it will have to induce loss. As TCP Vegas is incapable of inducing packet loss, it is unable to compete with aggressive loss based flows [KRB05].

6.2.2 Buffering and Packet Bursts

[KRB05] state that the losses experienced by a TCP flow under drop tail conditions are related to the algorithm's increase parameter. This is important as it implies that the aggressiveness of any algorithm is limited by the provision of network buffers available.

ScalableTCP as shown in Figure 6.2 has a continuously high increase value as a result of its multiplicative increase parameter. H-TCP, on the other hand, experiences only periods of very high α. It should be noted however, that the increase is only temporary: with its time based increase factor, the extra buffering after congestion is reduced due to the reversion of H-TCP back to $\alpha = 1$.

A similar approach to H-TCP is taken with BicTCP, with almost zero increase in *cwnd* as it approaches its w_{target}. FAST, however, with its design to stabilise the value of *cwnd* means that it is scalable with any buffer size. But, this only applies when it is stable. In order to reach equilibrium, the (up-to) exponential increase in *cwnd* can cause serious implications for network traffic. Again, this increase is determined by the value of α in the utility function.

One way of preventing large losses as a result of small buffer allocations is to utilise packet pacing [ASA00] whereby packets are not sent back-to-back but spread out over each RTT to prevent the large number of sequential packets drops due to drop-tail queuing mechanisms. However, findings in [ASA00] suggest that applying global TCP pacing would result in large amounts of TCP synchronisation due to the higher average queue occupancy of intermittent routers. Synchronisation occurs when flows competing along a link experience congestion at the same time, and as a result they all backoff together and ramp-up together. Depending on the backoff factor of the flows (as defined by H-TCP's adaptive backoff factor), the link may be under-utilised as a result of synchronisation.

The size of network queues is also important. As defined previously, network queues are necessary to buffer the increase in packets caused by the various algorithms' increase parameters. Equation 5.6 states that for Standard TCP, a queue provision equal to the bandwidth delay product is necessary for 100% utilisation. Indeed, this requirement is often stipulated in simulations involving network bulk transport [VS94]. However, [AKM04] states that the inclusion of such large amounts of buffer memory is not feasible due to physical size of Static RAM and the long latencies of DRAM. This problem raises concerns for fast long distance networks as the required

amount of buffering increases substantially. Fortunately, [AKM04] also states that due to the lack of synchronisation between competing flows, the queue size provision (for Standard TCP flows) requirement is related to the inverse square root of the number of flows.

Another method to reduce synchronisation is to implement RED on the bottleneck queue to cause non-synchronised drops from competing flows. It also has the benefit of being able to actively prevent congestion by the dropping of packets before a potential buffer overflow and therefore the ability to maintain smaller sized buffers and provide lower-delay interactive services [BCC98].

6.2.3 Interaction with Legacy and other Traffic

As phased deployment of any Internet protocol is likely, New-TCP algorithms should be able to coexist fairly with Standard TCP flows. Whilst the investigation of interaction with legacy traffic is more suited to a discussion of the self-similar nature of TCP aggregate flows [LTWW94, PKC97], the interaction between the bulk transport of data using Standard TCP and New-TCP algorithms should be studied.

This co-existence with other traffic is considered trivially by nearly all of the algorithms by simply defining a threshold at which the New-TCP algorithms will switch from their low-speed modes to their high speed modes. This is implemented by defining an arbitrary value of *cwnd* such that they mimic the standard AIMD algorithm below this value. This ensures that the algorithms can co-exist with Standard TCP under low *cwnd* environments, yet are able to achieve high throughput on high-BDP paths.

H-TCP, however, proposes a radically different way of determining how

algorithms should be friendly. It implements the idea that upon congestion the H-TCP flow should always revert back to Standard TCP and hence maintain at least temporary fairness with Standard TCP. Only after a period of time, after congestion, should the high speed mode be applied.

However, the balance between being able to co-exist with legacy traffic, and being able to achieve high throughput seems difficult. On one hand, it is a necessity to increase the aggressiveness of the congestion control algorithm in order to utilise spare bandwidth. On the other hand, this potentially means starving less aggressive flows of network resources.

This also applies to flows of the same algorithm as it is unlikely that two flows on the Internet sharing a bottleneck will also have the same end-to-end latency. Therefore, a question arises over how fair these algorithms are in sharing bandwidth when competing flows have different end-to-end latencies. The problem of this RTT unfairness stems from the fact that low latency flows are more responsive, and hence more aggressive, than flows on longer latency links. Therefore, it is possible that the flows on a low latency path completely starve the less responsive flow of throughput.

H-TCP implements a 'RTT scaling' function in order to maintain fairness between competing flows of different RTT's. This is implemented by altering the value of its increase parameter α such that it is scaled by the ratio of the experienced minimum latency and the average latency for a flow [LS04b].

It is also possible to help impose fairness between flows using AQM (See Section 5.3.2). Using RED, for example, would drop a higher proportion of packets from a more intensive flow; and therefore reduce its throughput and equalise fairness [Has89]. This also has the beneficial effect of preventing lock-out [FJ92] by aggressive flows that prevent less aggressive flows from getting any throughput.

6.3 Summary

The recent rise of high performance networks and long latency links has lead to the visible deterioration of TCP performance in real life situations. As such, there has been much interest in the transport community in the need to replace the standard TCP congestion control algorithms with something that is capable of attaining reasonable throughput under realistic network conditions.

A question arises over the manner in which deployment should occur. As these protocols have never been widely tested in real network environments, it is difficult to determine whether there are potential hazards to running these New-TCP algorithms on the Internet. Whilst running a single flow through the Internet is unlikely to cause problems, the aggregation of many flows may. The problem lies in the possibility of lock-out and severe unfairness that will prevent competing users from sharing network resources. However, the modeling of such systems prove very difficult to do realistically [AF99] due to the heterogeneity of the Internet.

Many of the proposals presented often cite the inclusion of AQM to mitigate the potential problems that aggressive algorithms may pose. [FJ93] states that AQM *should* be implemented, however, this is often far from the truth as most networks on the Internet still employ drop tail queuing. Therefore, when it comes to the question of the near term deployability of such protocols, special attention should be given to ensure that it is not dangerous to use such congestion control algorithms now.

The other question is which New-TCP algorithm should people use? Each protocol appears to have distinct advantages and disadvantages; for example, ScalableTCP ensures that high capacity can be attained regardless of network

capacity - however, at what cost? [CJ89] states that MIMD based algorithms may have serious convergence, and hence unfairness issues. FAST has a theoretical network utilisation of 100% - but is based on TCP Vegas which, despite its various advantages, was never widely deployed for general usage due to its inability to compete against aggressive flows, and hence the user perception of throughput was lower. Also TCP Vegas suffers from poor performance when the path is asymmetric [FLcL01].

So why cannot users simply choose their own algorithm to use? Indeed, this appears to be the situation at present as new congestion control algorithms appear all the time. However, very little research has been conducted to gauge the effectiveness of these algorithms *against each other* in similar network conditions. Furthermore, most of the analysis of New-TCP algorithms are centred around simulation based studies. Whilst the modeling of new algorithms in simulations is no doubt necessary, there may be physical limitations of running TCP protocols at high speeds - which may have negative consequences on network stability. For example, the aggressive increases of some of the algorithms may cause global synchronisation which will in fact lower network utilisation rather than increase it [ASA00].

Therefore, there is a clear need to test these algorithms on real hardware in realistic environments, and possibly on real Internet networks. The goal of such an investigation should not only focus on the performance in terms of throughput achieved, but also on other factors such as fairness, how quickly algorithms converge to fairness and the potential for congestion collapse.

Chapter 7

Testing Framework for the Evaluation of New-TCP Algorithms

Details of the methodology and definitions of performance metrics are given whereby the performances of New-TCP algorithms can be compared and contrasted in order to gain important insights into the benefits and disadvantages of each algorithm.

From this information, systematic testing of the new and old flavours of TCP algorithms will result in the development of 'capable' New-TCP algorithms.

7.1 Metrics

In this section, metrics which are considered important for the evaluation of bulk data transport are presented and discussed.

7.1.1 Goodput

Define *goodput*, $\bar{x}_i$, to be the mean rate of useful application layer data transferred per unit time, such that for the i^{th} flow, $x_i(t)$ is the rate of of useful data sent in the time period t. More formally it is defined as the throughput less losses.

$$\bar{x}_i := \lim_{T\to\infty} \frac{1}{T} \int_0^T x_i(t)dt, \tag{7.1}$$

Another consideration is that as the number of flows increases, this can be extended to define the *aggregate goodput*, $\bar{X}$, when there are n flows competing along the same bottleneck to be:

$$\bar{X} := \sum_{i=1}^{n} \bar{x}_i \tag{7.2}$$

Define the short-term average goodput as the exponentially weighted moving average,

$$\hat{x}(t)_i = \int_0^t exp[-\lambda(t-s)]x_i(s)ds \tag{7.3}$$

where λ is typically selected proportional to the bandwidth-delay product so that the averaging window scales with the congestion epoch duration.

In order to convey the New-TCP dynamics over a range of *cwnd* values, a range of network capacities and latencies should be investigated in order to obtain a general perspective of this particular performance metric. The effect of algorithmic mode switches from low to high speed regimes of the

various New-TCP algorithms should also be investigated.

7.1.2 Stability

In order to gauge the stability of a TCP flow, it must understood that TCP flows are inherently unstable as they strive to always probe for extra bandwidth and adapt to network conditions.

However, under static environments, the characteristics of the long term stability of a TCP flow in steady state can be determined. Define the following simple metric for the stability, S_i, of a flow i for a period $s = 1, ..., t$:

$$S_i := \frac{1}{\bar{x}_i} \sqrt{\frac{1}{n-1} \sum_{s=1}^{t} (x_i(s) - \bar{x}_i)^2} \tag{7.4}$$

The *stability*, S, of n competing flows is therefore:

$$S := \frac{1}{n} \sum_{i=1}^{n} S_i \tag{7.5}$$

S as defined is the standard deviation of the average goodput of n flows, each normalised by its mean goodput, and can be used as an indication of *goodput predictability* of n flows. It is therefore preferable to have high average goodput, $\bar{x}_i$, and low standard deviation, S_i, and hence low S.

7.1.3 Convergence Time

On long time-scales, such as with bulk transfer, the performance of TCP is dependent on the responsiveness of TCP flows to changes in the network state. In order to take advantage of increases in network capacity or to reduce sending rates to enable fair share of network traffic, it is important that the

TCP flows are responsive to these network changes.

In dynamic environments where flows join or leave the system often, the sending rate of each flow will adapt (i.e. lower) to enable sharing of bandwidth for new incoming flows. Due to this dynamic, the efficiency and fairness of the system may not always be optimal.

Define ε-*convergence time* following start-up of a new flow to be the time before the short-term average throughput $\hat{x}(t)_i$ of the new flow is within a factor ε of its long-term average value. Arbitrarily choose $\varepsilon = 0.8$ yielding the 80% convergence time.

7.1.4 Fairness

As the Internet is a shared resource, congestion avoidance algorithms should ensure that network users obtain a reasonable allocation of network resources depending on network conditions and the state and condition of competing users. This concern is embodied into the metric of *fairness* and depends upon the state and conditions of the competing flows.

Define fairness as a ratio of goodputs experienced by competing flows. As the dynamics of congestion control of competing flows with different RTTs will result in different growth rates of *cwnd*, the effect of having users with different RTTs upon the fairness of each flow should also be considered.

The dynamics of the Standard TCP congestion control algorithm can be estimated from an analysis of *cwnd* geometry such that at any time k, the *cwnd* value $w_i(k)$ is dependent upon the decrease fraction $\beta_i(k)$ and the increase value α_i for the congestion epoch k with round trip time $T_i(k)$, and

assuming equilibrium such that $w_i(k) = w_i(k+1)$ and $T_i(k) = T_i(k+1)$:

$$w_i(k) = \beta_i w_i(k) + \alpha_i T_i(k) \tag{7.6}$$

Assuming that queuing delay is negligible and where α_i and β_i are constant values, define that the expectation values of w_i and T_i are $E(w_i)$ and $E(T_i)$, Equation 7.6:

$$E(w_i) = \frac{\alpha_i E(T_i)}{\lambda_i(1-\beta_i)} \tag{7.7}$$

where λ_i is the probability that a congested system will result in immediate back-off of flow i. Similar to [CJ89], define that *cwnd* fairness as the ratio of *cwnd* between the two AIMD TCP flows, i and j, such that:

$$\frac{w_i}{w_j} = min\left(\frac{\frac{\alpha_i}{\lambda_i(1-\beta_i)}}{\frac{\alpha_j}{\lambda_j(1-\beta_j)}}, \frac{\frac{\alpha_j}{\lambda_j(1-\beta_j)}}{\frac{\alpha_i}{\lambda_i(1-\beta_i)}}\right) \tag{7.8}$$

where the quantity $\frac{\lambda_j}{\lambda_i}$ defines the *synchronicity* between the two flows as defined by the ratio of drops experienced between the two competing flows.

Using the BDP (Equation 4.1) to relate goodput and latency for a single flow, the *fairness*, $F_{i,j}$ between two flows, i and j with $\alpha_i = \alpha_j$ and $\beta_i = \beta_j$ (i.e. both flows have the same AIMD parameters) is:

$$F_{i,j} = \frac{\bar{X}_i}{\bar{X}_j} = min\left(\frac{\lambda_j}{\lambda_i}\left(\frac{RTT_j}{RTT_i}\right)^2, \frac{\lambda_i}{\lambda_j}\left(\frac{RTT_i}{RTT_j}\right)^2\right) \tag{7.9}$$

Therefore, the fairness between two competing Standard TCP flows depends upon the synchronicity and the square of the ratio between the latencies of the flows. As the growth rate of *cwnd* of the short RTT flow is greater than that of the long RTT flow, the increased growth rate in time results in

higher throughput achieved by the lower latency flow.

Also, under symmetric network conditions, two flows when perfectly synchronised and experiencing the same end-to-end latencies, should have a fairness of unity for all environments.

It is therefore important to investigate the effects of RTT unfairness between New-TCP algorithms such that long latency flows do not experience severe unfairness because of the increased aggressiveness of shorter latency flows.

7.1.5 Friendliness

The deployment of New-TCP is unlikely to be sudden, and therefore gradual deployment whereby New-TCP algorithms have to co-exist with Standard TCP on the Internet is expected. As such, the dynamic between New-TCP stacks and legacy Standard TCP should be studied.

New-TCP algorithms should be able to utilise spare capacity of the network; however, it should also not starve legacy TCP flows of bandwidth. Conversely, in low-speed networks, they should share bandwidth fairly with legacy TCP.

Therefore, define *friendliness* to be (the value of) the fairness metric, $F_{i,j}$ when the i^{th} flow is that of Standard TCP.

7.1.6 Overhead

Loss based congestion control algorithms must generate packet loss in order to probe for free bandwidth. The increase in the rate of probing will increase the number of congestion epochs per unit time, and as a result, the rate of packet loss also increases. This is especially important as the large number

of packets in flight as a result of large values of *cwnd* leads to a larger number of packets that may be lost due to packet bursts and having a too aggressive increase in α [KRB05].

It is important in order to prevent classical congestion collapse and improve goodput that the fraction of retransmitted packets is kept low [FF99]. Therefore, given that, the i^{th} TCP flow retransmits $r_i(k)$ bytes of data in the duration s, define the ratio of retransmitted packets for $s = 1, ..., t$ as:

$$\epsilon_i = \frac{\sum_{s=1}^{t} r_i(s)}{\sum_{s=1}^{t} u_i(s)} \tag{7.10}$$

where $u_i(s)$ is the amount of data transfered during the period s of the i^{th} flow.

In order to give a comparable metric between stacks that achieve different goodputs under identical network conditions, normalise all ϵ_i by the total goodput of all flows, $\bar{X}$, to give the total *overhead*, ξ:

$$\xi = \frac{\sum_{i}^{n} \epsilon_i}{\bar{X}} \tag{7.11}$$

Therefore a low value of ξ is desirable and should not be much greater than that of Standard TCP to facilitate the stability of the Internet.

7.2 Environment

The goal of this investigation is to be able to compare and contrast the performance metrics of New-TCP algorithms against that of Standard TCP in order to gauge the advantages and disadvantages of each.

In order to get a representation of the performance of these congestion

control algorithms over a wide range of environments, the metrics defined in Section 7.1 are applied to a series of different experiments designed to show the differences between the New-TCP algorithms.

The following network conditions can effect the performance of TCP transport. It is therefore desirable to test over a range of different network environments encompassing these conditions to obtain a good representation of New-TCP performance.

Network Capacity & Link Latency Greater link capacities result in larger values of *cwnd* and hence the number of packets in flight. Similarly, longer latencies also result in larger values of *cwnd*. It is also important to understand the affects of how these algorithms will 'switch' into their high speeds modes and how that may adversely affect the performance metrics outlined in Section 7.1.

Bottleneck Queue Size As mentioned in Section 5.4.1, the buffering of packets is important to absorb packet bursts and to increase bulk transport efficiency. The dynamic of the queue provision is also important as competing web traffic may affect the metric dynamics of the New-TCP flows. Whilst the short term dynamics of web traffic is not well understood [BPS+98, CB95], it is argued that the bulk transfer of data will play an important part in future networks.

Number of Flows (and type) It is important to gauge the scalability of these flows in terms of how they share bandwidth. In terms of friendliness, it is also important to measure the unfairness of New-TCP flows against Standard TCP. Related to the bottleneck queue dynamics, the burstiness of TCP flows may also affect the various performance metrics such as convergence time and overhead.

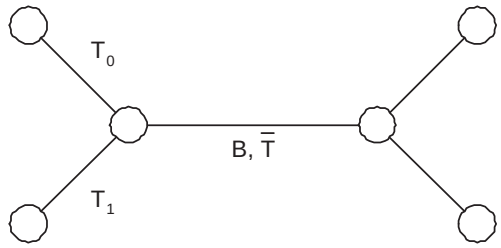


Figure 7.1: Topology of Framework tests for the Evaluation of New-TCP algorithms.

Using the topology illustrated in Figure 7.1, a suite of simple tests was implemented that exercises the performance of New-TCP algorithms. Even with the simplicity of this topology, the advantages and disadvantages of the various New-TCP algorithms can be demonstrated when applied to the performance metrics oulined in Section 7.1:

Queue-size The importance of how these algorithms perform over a range of bottleneck queue-sizes is especially important as large BDP's are reached due to the hardware limits on the amount of physical memory on intermediary routers and switches.

Symmetric Network It is expected that all New-TCP algorithms are fair when identical TCP flows are competed against each other across a range of different network environments. As a sanity check, it is expected that all algorithms will achieve higher goodputs than that of Standard TCP in high bandwidth delay environments, yet should be able to compete equivalently under low bandwidth delay environments.

Asymmetric Network As higher capacities are introduced into the Internet, it is important that New-TCP algorithms can scale such that a flow with larger *cwnd* does not starve a larger latency (and hence less

responsive) flow and hence cause lock-up [FJ92]. It is therefore important that the unfairness between flows of different end-to-end latencies is not (much) greater than that of Standard TCP.

Friendliness The phased deployment of New-TCP requires that immediate performance benefits are achieved, without causing significant performance penalties for existing Standard TCP flows.

7.3 Summary

A generic framework was presented such that comparative testing of New-TCP algorithms can be accomplished. The comparative testing of algorithms is important to understand why each New-TCP algorithm performs better in each network scenario so that better algorithms can be developed.

The framework consists of a set of metrics such as goodput, convergence time and fairness which must be applied to a range of network conditions/environments. Whilst the parameter space for such an analysis is vast, it is important that such large changes to the TCP congestion control algorithms should be well understood and potential problems identified prior to widespread deployment.

Chapter 8

Systematic Tests of New-TCP Behaviour

Implementations of the New-TCP protocols and algorithms outlined in the Chapter 6 are now tested and evaluated under controllable network environments using the framework outlined in Chapter 7.

The focus of this Chapter is not to determine a specific 'winner' or the 'best' New-TCP algorithm, but to understand the specific environments and reasons why different New-TCP stacks perform better or worse.

8.1 Methodology

The testing of network protocols is often difficult due to the changing characteristics, unpredictable loads of Internet traffic and uncontrollable determinants of performance. As such, an initial investigation into the dynamics of New-TCP algorithms was performed under repeatable network conditions

where all network factors such as bandwidth, end-to-end latency and background loads could be controlled and altered.

This investigation focuses upon the real-life implementation of these algorithms and as such relies upon real hardware and software on which the tests are performed. Careful selection of network conditions were selected such that hardware limitation, such as the lack CPU horse power, did not dominated the results.

8.1.1 Dummynet

All tests in this Chapter were performed in simulated environments in order to better understand the performance of these protocols using real equipment. Experiments were conducted on a dummynet [Riz98] testbed in order to present results that should be indicative of real-world performance without the implications on the necessary analysis of real network cross-traffic. Dummynet provides the facility to emulate the effects of network queues, bandwidth limitations and communication delays and also enables the repetition of tests under controllable conditions.

Dummynet utilises the FreeBSD [UCB] IP firewall, *ipfw* [AKN+], to allow the selection of IP packets based on a combination of source and destination addresses, ports and protocols types (TCP, UDP, ICMP, etc), interface, and direction (in or out). This selection allows the packets arriving at a dummynet router to be forwarded onto dummynet *pipes* which simulate the effects of a network link.

The topology of the testbed is shown in Figure 7.1 with further details in Figure C.1. It consists of a simple dumb-bell topology with two pairs of high-end commodity PCs connected to gigabit ethernet switches which are

fed into the dummynet router via two separate NIC's (Intel e1000's). The dummynet router is of a similar hardware specification to the testbed PCs (See Table A.5), but runs FreeBSD 4.8 instead of Linux 2.6.

The dummynet set-up is an idealised network in the sense that all aspects of the network variables are configurable and is perfect for validating the performance of New-TCP algorithms against the theoretical performance.

The chosen latency values[1] of 16ms to 162ms give a good indication of the trans-European to trans-Atlantic transmission times on today's Internet. Bottleneck capacities of upto 250Mbit/sec were chosen as hardware limitations resulted in unrepeatable results at higher capacities. Relatively small bottleneck link speeds of 10Mbit/sec were used to investigate the algorithmic switching of various New-TCP algorithms into their 'high-speed' modes.

8.1.2 altAIMD Kernel

The various New-TCP algorithms have their independent patches publicly available. However, certain deficiencies in the Linux networking stack [Lei04] suggest that testing of these algorithms using their publicly available implementations would most likely yield the testing of the various Linux performance patches rather than the actual New-TCP algorithms. As the various patches either do not address the issues at all, or do so in different ways, it was necessary to build the congestion control algorithms into a common kernel for the purpose of this investigation so that only the algorithmic differences of New-TCP algorithms were comparatively analysed.

This kernel will hereby be referred to as the *altAIMD* kernel. It is based on the Linux 2.6.6 kernel and incorporates the following features:

[1]The minimal limit of 16ms is imposed by Dummynet as tests performed at lower latencies exhibited strange unaccountable effects upon throughput.

Implementation of New-TCP Algorithms A single `sysctl` [Rub97] is used to switch between New-TCP algorithms without the requirement of rebooting the machine.

Appropriate Byte Sizing (RFC3465) [All03] The counting of *ack*'s by the number of bytes acknowledged rather than the number of *ack*'s received to counter the problems of reduced *cwnd* growth under delayed *ack*'s.

SACK Processing Improvements The current implementation of SACK processing in the Linux kernels requires a processing time which is $O(cwnd)$ [Lei04]. This has serious performance implications when dealing with a large number of packets in flight which is common with large BDP's. A more robust algorithm was implemented with complexity of $O(\text{lost packets})$ (which is equivalent to number of un-sacked packets).

Cap Sets *cwnd* to minimum of packets in packets in-flight and *ssthresh*, and therefore reduces the influence of miscalculations in the network stack which helps prevent packets bursts.

Throttle Patch A build-up of *ack* packets at the sender can cause an overflow in the Linux network ring buffers which causes all packets to be dropped. A modification to prevent this behaviour was implemented.

Undo Patch The Linux implementation of undo's [SK02] was modified to suit the appropriate New-TCP network stack.

Web100 Instrumentation of the TCP stack were recorded and logged via Web100 Linux kernel patch [MHR03] in order to validate and check all results gathered.

Various other constraints on TCP network performance were also removed, such as the default low limits on the TCP socket buffer sizes. `txqueuelen` and `max_backlog` values were left at acceptable default values (under Linux 2.6) of 1,000 and 300 respectively. Default device driver settings were also used which were found to be sufficiently provisioned for these tests.

8.1.3 Overview

In order to minimise the effects of local hosts' queues and flow interactions, only a single TCP flow was injected from each source machine into the testbed through the dummynet router using `iperf` [TQD+03] . All flows were run with slow-start enabled upon the start of each transfer to give a representative test of real users competing for bandwidth when conducting bulk transport along the network path.

A second flow was initiated at a random time after the first flow such that the perturbation of the second flow's slow start covered a range of times within a congestion epoch of the first flow. This was to ensure that a representative range of convergence times and *cwnd* dynamics was captured.

As defined in [Cla82b], all TCP receivers were run with delayed *ack*ing on. As Linux 2.6 kernels were used, the effects of quick-*ack*ing (See Section 5.2.4) were left enabled. All TCP senders were configured with Appropriate Byte Sizing (See Section 5.2.4).

Each individual test was run for at least ten minutes. This duration gave a good representation of the number of congestion epochs required for all of the New-TCP algorithms. In the case of tests involving Standard TCP, individual tests were run for up to an hour as the long epoch time of Standard TCP, especially at large BDPs, requires more time to reach steady state.

In order to obtain a good representation of the range of performance metrics, all tests were repeated 5 times and the arithmetic mean taken. The error on each measurement was taken as the standard error from this mean [Boa83] and is presented in all plots.

Furthermore, due to the prevalence of FIFO queuing mechanisms in use on the Internet, the study is limited to drop-tail environments to gauge the worse case scenario whereby network feedback is limited only by buffer overflows.

8.2 Test Calibration

8.2.1 Response Function

The validation of the *altAIMD* algorithm and the representative New-TCP algorithms is shown in Figure 8.1. It shows the theoretical and measured response function of a single New-TCP flow (See Section 5.4.2) under the dummynet test network. The loss rate was controlled by the dummynet `p` variable that defines the average random loss rate experienced through a dummynet pipe.

It can be seen that the measured results match very well to the theoretical predictions as specified by the respective New-TCP authors. The most notable deviation is that for both ScalableTCP and HSTCP, where there is a gradual, rather than immediate switch from their low speed modes to their high speed modes; this is represented by their increased goodputs over the regions of the switch. This was found to be due to the mode switching from Standard TCP into their representative high speed modes (this does not apply to FAST or H-TCP) as *cwnd* values approach their relevant thresholds.

For HSTCP (Figure 8.2(a)), it was observed that the actual average *cwnd*

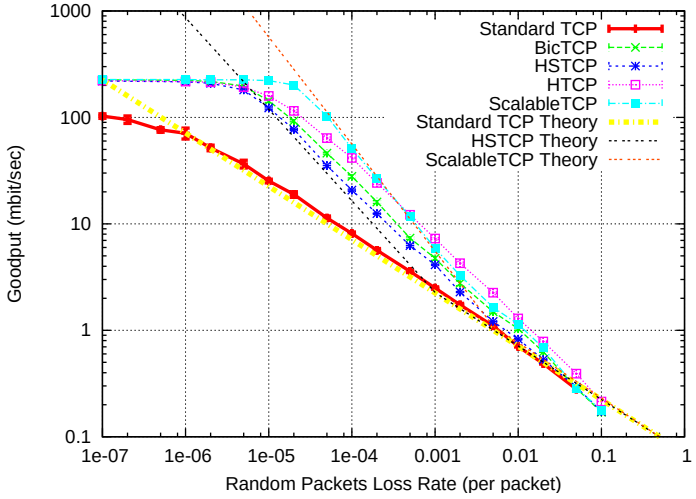
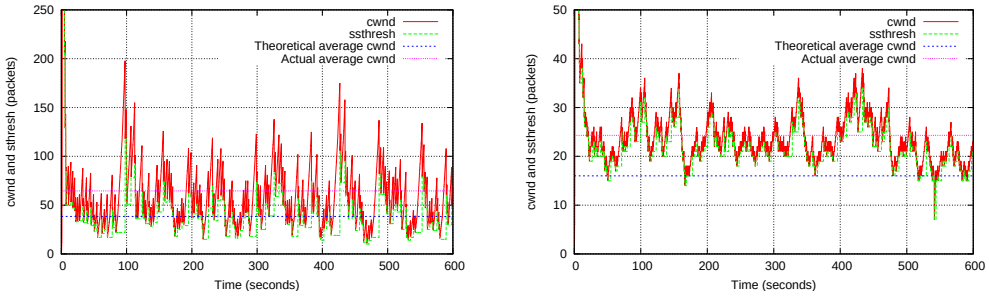


Figure 8.1: Effect of varying the dummynet random loss probability on a single TCP flow (250Mbit/sec, 162ms RTT, Bottleneck queue-size 20% BDP).



(a) HSTCP at $p = 0.001$

(b) ScalableTCP at $p = 0.005$

Figure 8.2: Magnification of New-TCP mode switch from low-speed into high-speed modes (250Mbit/sec, 162ms RTT, queue-size 20% BDP).

value of 65 packets is actually a lot higher than the theoretical prediction of 38 packets. As the *cwnd* is directly related to the goodput achieved by the TCP flow (Equation 4.1), it is unsurprising from this that the goodput is higher than the prediction allows. A factor that contributes to the larger value of *cwnd* is that *cwnd* is held as an integer under Linux; this causes a

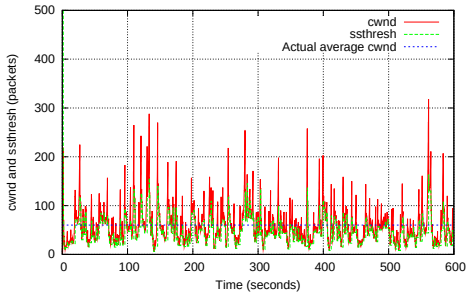


Figure 8.3: H-TCP *cwnd* dynamics showing the mode switch from low-speed into high-speed modes (250Mbit/sec, 162ms RTT, queue-size 20% BDP).

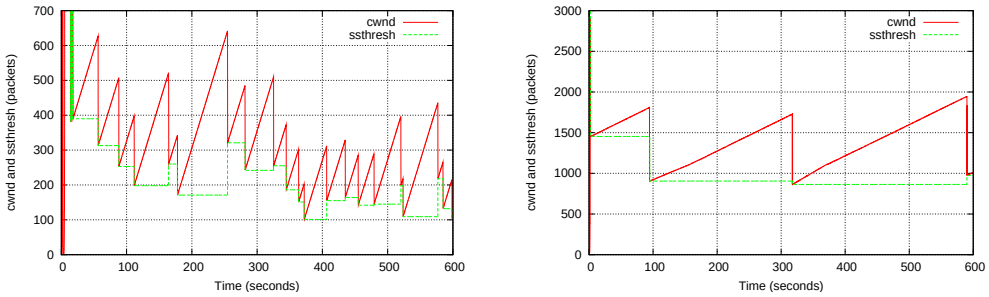
higher value of *cwnd* upon loss as the calculation of β is overestimated[2].

However, the main cause of the high *cwnd* values is due to the random loss nature of the dummynet packet drops; there are occasions where the value of *cwnd* reaches the next 'stage' in the HSTCP algorithm where the increase and decrease parameters are higher; making the *cwnd* algorithm grow more aggressively, and fall back to a relatively higher value upon loss.

Figure 8.2(b) shows the *cwnd* trace of ScalableTCP at 0.5% loss rate. Again, it was observed that the actual average *cwnd* value of 24.3 is much larger than the predicted 16 packets. It is worth noting that 16 packets is actually the threshold where ScalableTCP starts to engage its high-speed mode, and the effects of this can be clearly seen as almost none of the congestion events result in the Standard TCP *cwnd* drop to a half. As the decrease parameter β of 0.875 is also implemented in the same way as HSTCP, it was observed that the decrease in *cwnd* upon loss is actually less than theory due to the quantisation of the values of *cwnd* (especially at such low values of *cwnd*).

Figure 8.1 clearly shows that H-TCP is able to achieve very high utilisation across the range of loss environments. Figure 8.3 shows the *cwnd* trace

[2]As it is implemented traditionally where $cwnd \leftarrow cwnd - b \times cwnd$ where $b = 1 - \beta$.



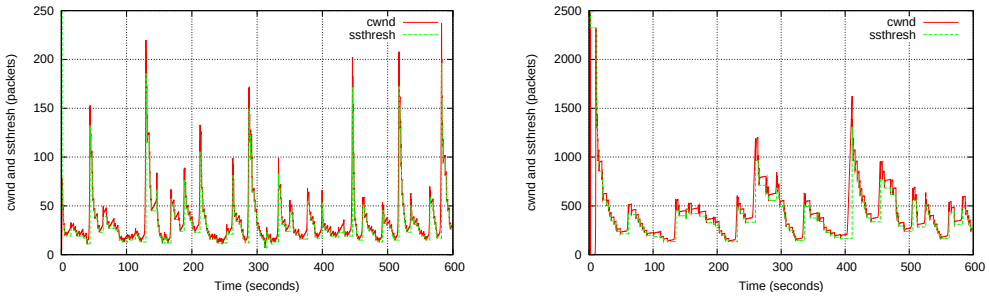
(a) Standard TCP at $p = 0.00002$ (b) Standard TCP at $p = 10^{-7}$

Figure 8.4: Standard TCP *cwnd* dynamic (250Mbit/sec, 162ms RTT, queue-size 20% BDP).

of H-TCP at 0.2% loss rate. It was observed that there are periods whereby the *cwnd* reaches (relative to the test) very high values; where it takes a few congestion events before the *cwnd* decreases to a more 'stable' value. The large values of *cwnd* are compounded by H-TCP's increase parameter which is polynomial with respect to time. As a direct consequence of the large *cwnd* values, H-TCP is able to achieve high goodput, and the longer the congestion epochs, the greater the achievable value of *cwnd*. Whilst the authors of H-TCP do not provide the calculation of H-TCP's response function, they claim that it is comparable to that of HSTCP [LS04a].

As Standard TCP approaches low loss rates, higher values of *cwnd* are observed; however, so too is the very slow growth of AIMD. This is observed in Figure 8.4(b) where the slow increase and the dramatic decrease of *cwnd* results in the low utilisation of the link over time-scales less than a single congestion epoch, after detected loss. Also, it is this slow growth which prevents Standard TCP from reaching high values of *cwnd* under a specific loss rate compared to the other New-TCP algorithms.

BicTCP argues that the correct response function to balance between



(a) BicTCP at $p = 0.002$ (b) BicTCP at $p = 10^{-4}$

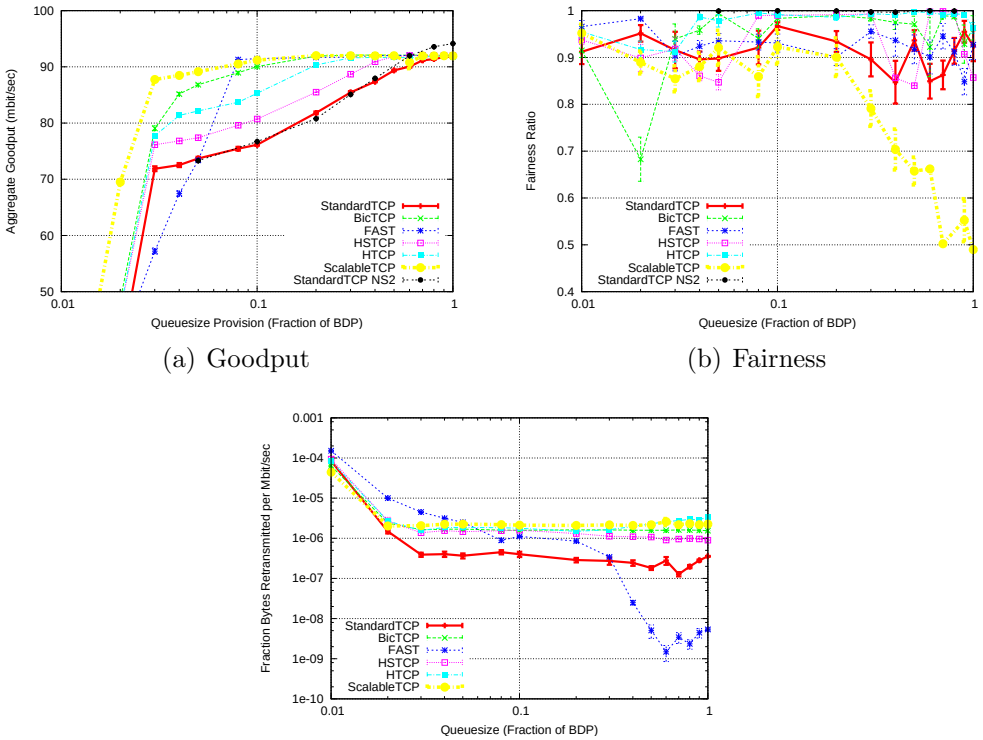
Figure 8.5: BicTCP *cwnd* dynamic (250Mbit/sec, 162ms RTT, queue-size 20% BDP).

friendliness and scalability lies within a region between HSTCP and ScalableTCP [XHR04]. Figure 8.5(a) shows the *cwnd* trace of BicTCP at $p = 0.002$ and $p = 10^{-4}$. It is visible from Figure 8.5(b) that the aggressive additive increase enables BicTCP to reach large values of *cwnd*, whilst the low decrease parameter of 0.8 maintains the high value of *cwnd* even under consistent loss. The plateau of *cwnd* which BicTCP uses to maximise throughput is less visible at such (relatively) high loss rates. This is especially true in the case of $p = 0.002$ where the *cwnd* growth after loss undergoes additive increase.

FAST has been excluded from these results due to its dependence upon network delay rather than network loss.

8.2.2 Bottleneck Queue Size

As bottleneck queue-sizes are important in being able to accommodate packet bursts, it is important to provide sufficient memory at ingress and egress queues. The larger values of the congestion avoidance increase α of New-TCP algorithms are likely to impose a greater need for larger queue-sizes as



(a) Goodput

(b) Fairness

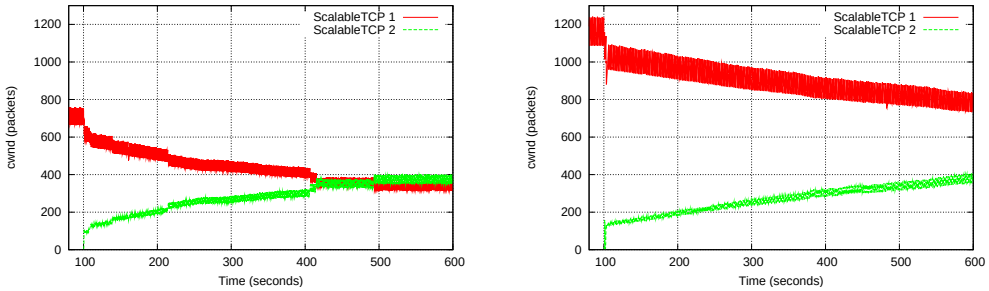
(c) Overhead

Figure 8.6: Effect of varying the bottleneck queuesize on two symmetric competing flows (100Mbit/sec, 82ms RTT, BDP 683 packets).

more packets will be sent out for each *ack* received [KRB05].

Figure 8.6 shows the effect of two competing TCP flows under symmetric conditions sharing a bottleneck link with various queue size allocations. It shows that the experimental results of goodput match closely to that of simulation results from *ns*2 [ns2] for Standard TCP.

With a queue provision of less than about 5% of the BDP, all algorithms perform badly due to packet bursts which flood the network queue. Between this latter area and the area where the algorithms reach line rate, it can be seen that all New-TCP algorithms perform better than Standard TCP; with ScalableTCP and BicTCP achieving the highest aggregate goodput.



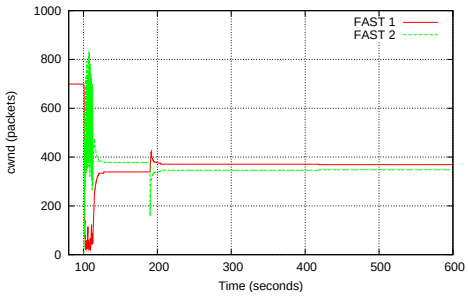
(a) Queuesize 10% BDP (b) Queuesize 80% BDP

Figure 8.7: *cwnd* trace of ScalableTCP at various queue-size provisions under symmetric network conditions (100Mbit/sec, 82ms RTT, queue-size as shown).

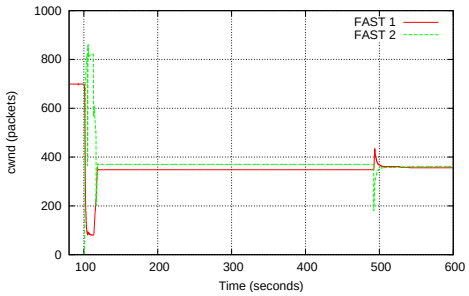
It is worth noting that even though Equation 5.6 specifies that 100% utilisation of Standard TCP should only occur at 100% BDP queue provision, the calculations assume that only one flow exists in the system or that all flows are perfectly synchronised. As such, the experimental results demonstrate the lack of synchronisation between competing Standard TCP flows which increases bottleneck utilisation.

Fairness between two flows of the same New-TCP algorithm across the range of queue provisions is shown in Figure 8.6(b). It shows the difficulty in being able to achieve consistent fairness results from the New-TCP algorithms, with relatively large errors shown. However, the results do show that all of the algorithms, with the exception of ScalableTCP, are reasonably fair across the difficult queue-size provisions. The problems of achieving fairness with ScalableTCP is shown in Figure 8.7 where the long convergence times required for steady state are clear which (combined with the method of measuring fairness) contribute to low fairness.

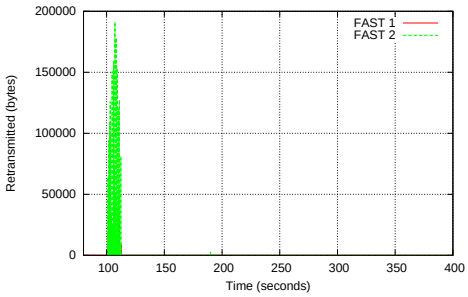
As all algorithms share the same SACK code base, the *overhead* gives an indication of aggressiveness of the underlying congestion control algorithm



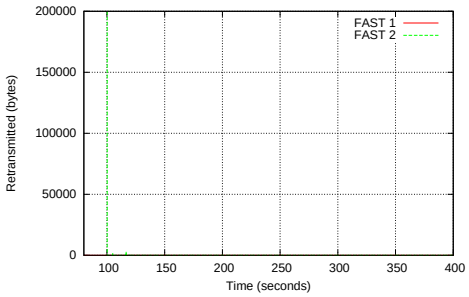
(a) Congestion Window at 10% BDP Queuesize



(b) Congestion Window at 80% BDP Queuesize



(c) Data Retransmissions per 100ms at 10% BDP Queuesize



(d) Data Retransmissions per 100ms at 80% BDP Queuesize

Figure 8.8: *cwnd* and retransmissions dynamic of FAST at various queue size provisions under symmetric network conditions (100Mbit/sec, 82ms RTT, Queuesize as shown).

upon network bursts. This is shown in Figure 8.6(c). It was observed that all New-TCP algorithms achieve higher throughput at the expense of increased fractional losses. However, even at large queue provisions, all of the New-TCP algorithms induce much more packet loss, with no obvious benefit in extra goodput. This is demonstrated by the differences between the overhead of Standard TCP and nearly all of the New-TCP algorithms. The exception to both points is FAST, which manages to avoid excessive losses, but only when sufficiently large queue-sizes of $> 30\%$ of BDP are provisioned.

Figure 8.8 shows the number of bytes retransmitted for the experiments.

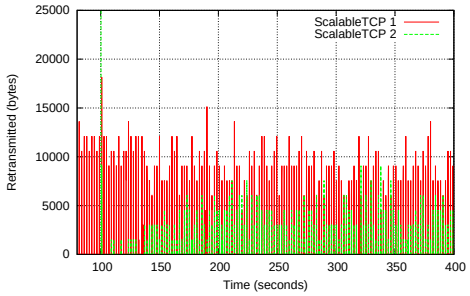
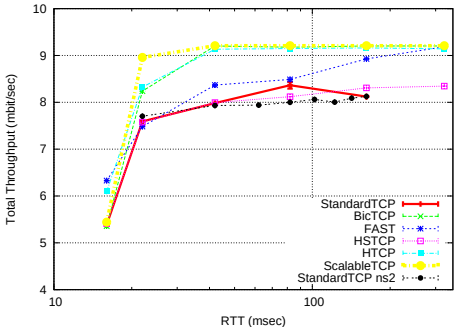


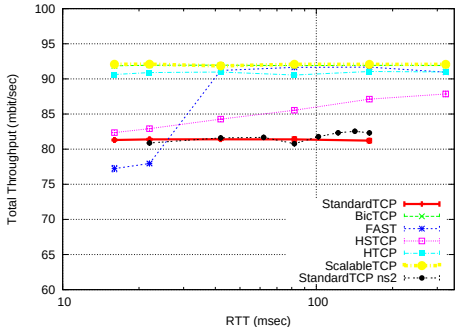
Figure 8.9: Instantaneous retransmissions of ScalableTCP every 100ms at 80% queue size provision under symmetric network conditions. (100Mbit/sec, 82ms RTT).

It can be seen that the *cwnd* of FAST is flat, suggesting that it is able to utilise its RTT calculation effectively to settle into a constant *cwnd* value. As such, there are almost no losses and retransmissions during the course of the test. However, upon start-up, FAST's slow start algorithm (which is different from standard slow start) induces much loss and hence retransmissions. This effect is much more pronounced in the small queue size provisions (where Figure 8.8(d) shows almost no retransmissions) than in the large queue size provisions, which result in the much lower overhead of FAST with larger queue sizes.

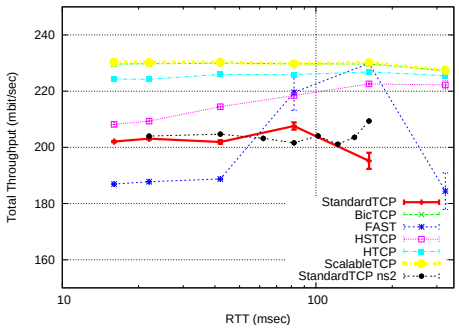
In contrast, loss based algorithms, such as ScalableTCP as shown in Figure 8.9 *have* to induce loss continuously for the entire duration of the test. Figure 8.9 corresponds to the *cwnd* trace presented in Figure 8.7(b) and it can be seen that the larger *cwnd* flow (Scalable 1) experiences greater loss than that of the second flow. This is due to the aggressive nature of ScalableTCP's algorithm with its decrease value β which favours the higher throughput (and hence larger *cwnd*) flow. This is common to all loss-based algorithms, and implies that with large capacities, much more data will have to be retransmitted in order to maintain congestion avoidance.



(a) 10Mbit/sec Bottleneck Capacity



(b) 100Mbit/sec Bottleneck Capacity



(c) 250Mbit/sec Bottleneck Capacity

Figure 8.10: Aggregate Goodput of two competing New-TCP flows under symmetric network conditions (Bottleneck Queuesize set to 20% BDP).

8.3 Results

This section focuses upon the dynamics of only two competing flows along a bottleneck link. Even though the topology and set-up of the tests are relatively simple, it is observed that even with these straightforward tests much information can be gathered and assessments of the various New-TCP algorithms can be made.

8.3.1 Symmetric Network

Goodput

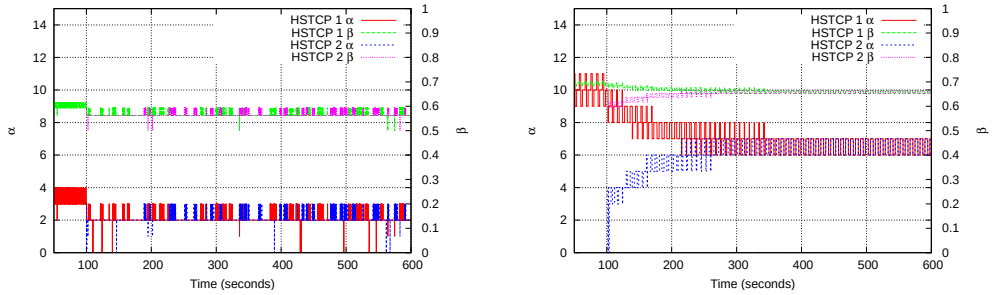
Figure 8.10 shows the aggregate goodputs of two competing flows under symmetric environments. All New-TCP algorithms are able to achieve greater goodput than Standard TCP, with ScalableTCP and BicTCP achieving the highest utilisation. The poor performance by all algorithms at 16ms in the 10Mbit/sec environment is attributed to bottleneck queue-size provision of only two packets, which results in complete queue flooding with just a single flow (as ABC bursts two packets per *ack* under Standard TCP).

FAST appears to have erratic goodput performance across the range of different bandwidths and latencies; for example its goodput performance is comparable to all the other algorithms at 10Mbit/sec. However, it actually achieves lower goodput than Standard TCP for low latencies in the higher speed environments.

HSTCP is able to achieve greater goodput as the BDP increases. At the largest BDP tested, it was observed that HSTCP has similar goodput performance to that of the other New-TCP algorithms. The change in aggression of HSTCP over the range of latencies is shown in Figure 8.11. It clearly shows that with increased latency (and hence BDP) α and β values are also increased which enables HSTCP to become more aggressive and hence able to utilise more of the available bandwidth.

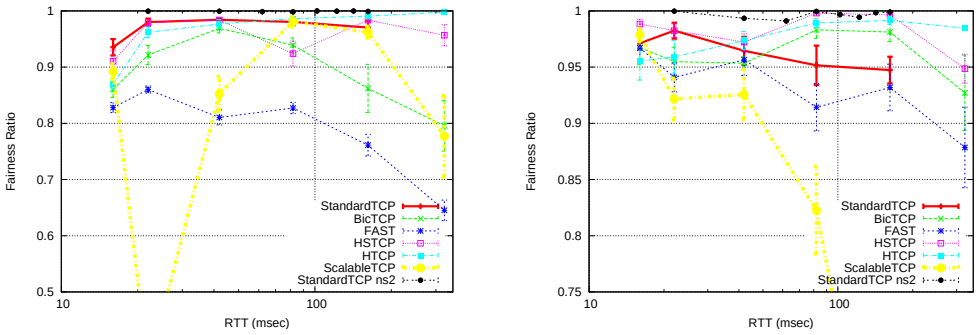
Fairness

Figure 8.12 shows the fairness achieved between two New-TCP flows. ScalableTCP appears to have serious fairness problems at all capacities, being the most unfair for long latencies and all capacities. An example *cwnd* trace

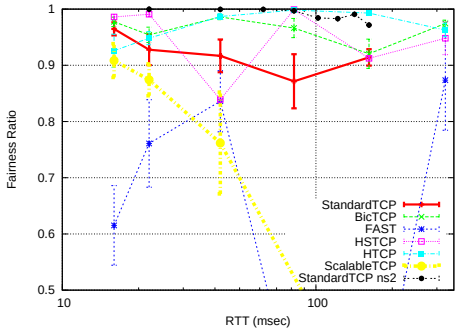


(a) 22ms (b) 162ms

Figure 8.11: AIMD parameters of HSTCP of competing flows under symmetric network conditions (100Mbit/sec, Bottleneck Queuesize set to 20% BDP).

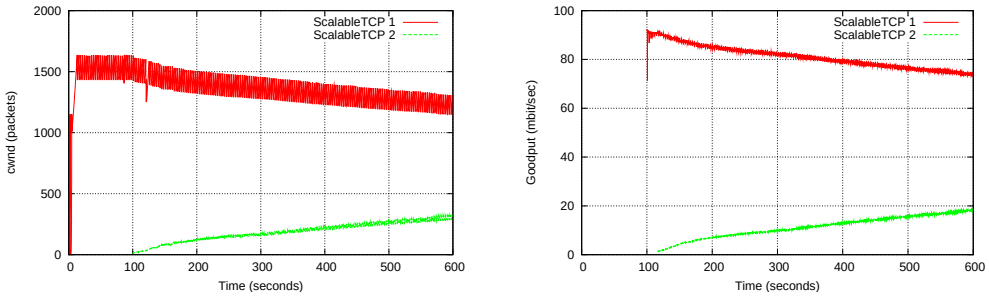


(a) 10Mbit/sec Bottleneck Capacity (b) 100Mbit/sec Bottleneck Capacity



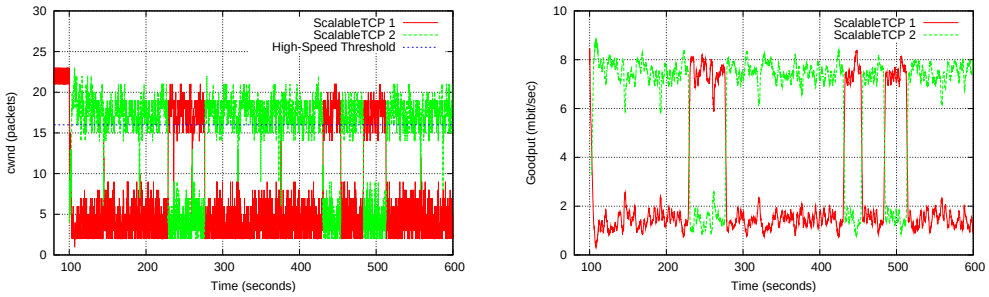
(c) 250Mbit/sec Bottleneck Capacity

Figure 8.12: Fairness between two competing TCP flows with symmetric network conditions (Bottleneck Queuesize set to 20% BDP).



(a) Congestion Window (b) Goodput

Figure 8.13: Unfairness between ScalableTCP flows (Symmetric network, 100Mbit/sec, 162ms RTT, queuesize 20% BDP).

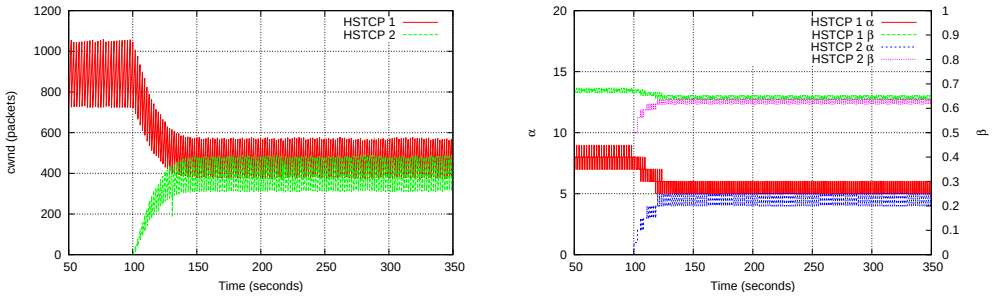


(a) Congestion Window (b) Goodput

Figure 8.14: Unfairness between ScalableTCP flows (Symmetric network, 10Mbit/sec, 22ms RTT, queuesize 20% BDP).

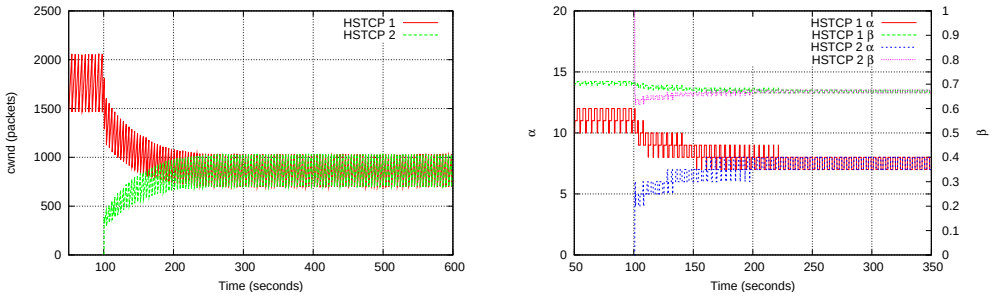
of the unfairness experienced by ScalableTCP is shown in Figure 8.13. Similar to the effects of having large queue size provisions, it can be seen that the unfairness between the two ScalableTCP flows is caused by very slow convergence. Similar results were gathered for all cases where ScalableTCP showed unfairness.

ScalableTCP also demonstrates unfairness at 22ms and 42ms under the 10Mbit/sec environment. This was found to be due to its low/high speed mode switch when *cwnd* is equal to or more than 16 packets and the vastly different *cwnd* update algorithms as shown in Figure 8.14.



(a) Congestion Window (b) AIMD Parameters

Figure 8.15: Unfairness between HSTCP flows (Symmetric network, 250Mbit/sec, 42ms RTT, queuesize 20% BDP).

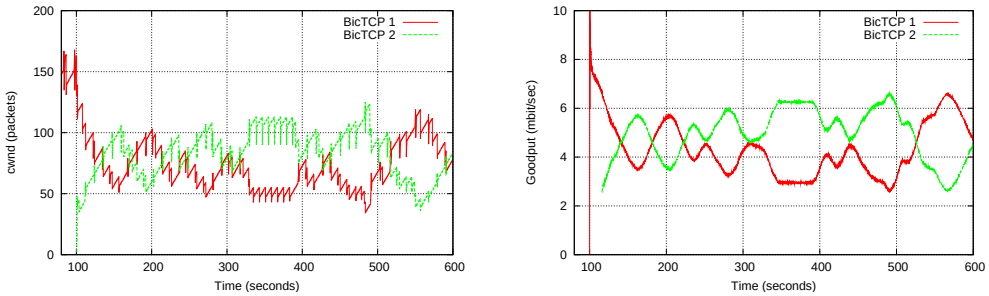


(a) Congestion Window (b) AIMD Parameters

Figure 8.16: Unfairness between HSTCP flows (Symmetric network, 250Mbit/sec, 82ms RTT, queuesize 20% BDP).

There are also fairness problems with HSTCP at certain latencies, caused by the regions of mode switching between the different α and β regimes. This is visible at 82ms with 10Mbit/sec, 42ms at 100Mbit/sec and 42ms at 100Mbit/sec. An example of this *cwnd* and AIMD dynamic is shown in Figures 8.15 and 8.16 and shows that the unfairness appears to be caused by the quantisation of HSTCP's AIMD parameters of the two flows. This is caused by the implementation of the look-up table as defined in [Flo03] with discrete values of AIMD for the various operating regions of HSTCP.

In Figure 8.12(a), it can be seen that BicTCP appears to have prob-



(a) Congestion Window (b) Goodput

Figure 8.17: Unfairness between BicTCP flows (Symmetric network, 10Mbit/sec, 162ms RTT, queuesize 20% BDP).

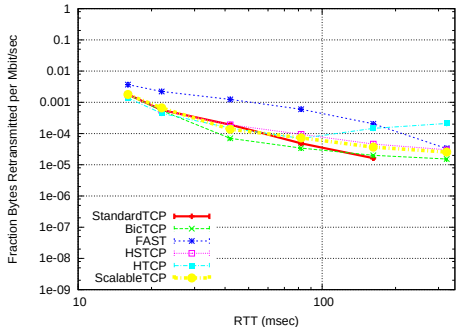
lems achieving fairness at low capacities and high latencies, although it is more capable at higher speeds. The problems at low capacities are shown in Figure 8.17 which shows that at such small values of *cwnd*, the plateau of BicTCP is almost non-evident, resulting in a simple aggressive additive increase algorithm.

H-TCP appears to have to no problems achieving fairness across all network conditions, and was observed to be more fair than Standard TCP in these tests.

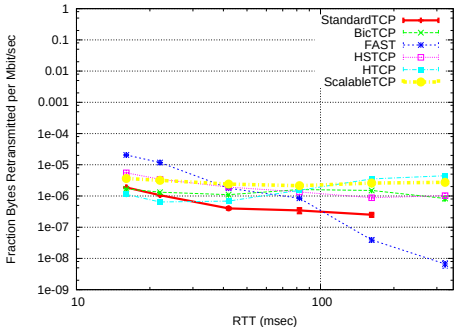
Overhead

The overhead of the New-TCP algorithms is shown in Figure 8.18. It can be seen that Standard TCP has the clear advantage in terms of this metric.

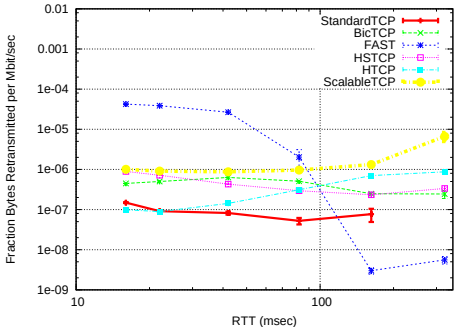
At high latencies, FAST with its delay based congestion control manages to retransmit less data per unit of goodput than Standard TCP. But at lower capacities and latencies, the overhead of FAST is actually approximately two orders of magnitude worse than Standard TCP. Also, at the most extreme setting of 250Mbit/sec with 324ms delay, FAST's goodput performance is



(a) 10Mbit/sec Bottleneck Capacity



(b) 100Mbit/sec Bottleneck Capacity

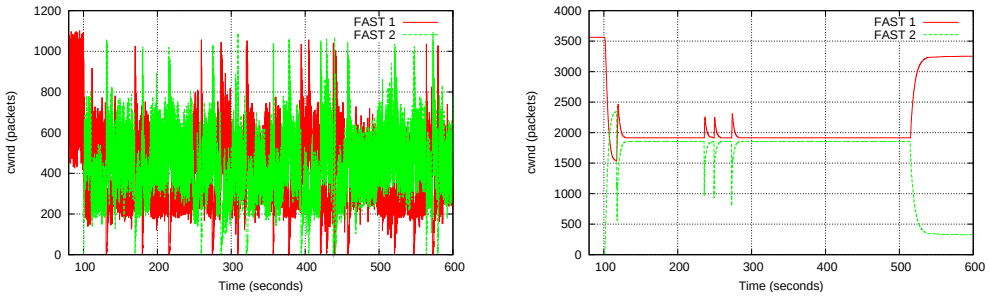


(c) 250Mbit/sec Bottleneck Capacity

Figure 8.18: Overhead of New-TCP algorithms with two competing TCP flows with symmetric network conditions (Bottleneck Queuesize set to 20% BDP).

actually quite poor in relation to the other algorithms. This was found to be due to FAST requiring sufficiently provisioned bottleneck queue sizes in order to provide the multi-bit network information for FAST to be able to settle into a constant value of *cwnd*. Otherwise, FAST has to resort to an aggressive probing of bandwidth with large values of α, which induces loss and hence decreases efficiency. This is confirmed by [KRB05] which states that the queue has to be sized to enable the bursts of packets caused by this increase.

The *cwnd* dynamic of FAST is shown in Figure 8.19. Under the lower latency network of 42ms, it can be seen that FAST is continuously attempting

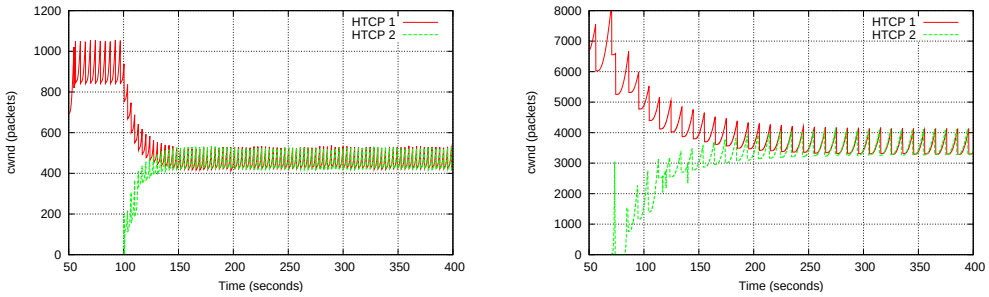


(a) 42msec (b) 162msec

Figure 8.19: FAST *cwnd* dynamics (Symmetric network, 250Mbit/sec, queuesize 20% BDP).

to increase its *cwnd* value through very aggressive probing, whilst under the longer latency (and larger buffer size) of 162ms, the *cwnd* is able to stabilise very quickly (after a very noisy stabilisation period at the start) to equal sharing of the network path, with almost no losses induced by either flow; however, towards the end of the test, the two flows' *cwnd* diverge. It is worth noting that these dynamics were typical of measurements across a range of network environments and are not selected as worse cases behaviours. This divergence was found to be due to the sensitivities of FAST to changes in throughput which are only propagated into a change in *cwnd* dynamic a period of time after the bandwidth change is detected [JWL+03].

Indeed, the experimental results also agree with [KRB05]. FAST incorporates switching between different values of α depending on the throughput (currently measured by the rate of packets) such that in low speed environments $\alpha = 8$. When performing upto 100Mb/sec $\alpha = 20$ and above this it is 200. In the experimental results, FAST only manages to stabilise *cwnd* (similar to that as presented in Figure 8.19(b)) when the latency is 162ms, 42ms and 82ms for network capacities of 10, 100 and 250Mb/sec respectively.



(a) 42msec (b) 324msec

Figure 8.20: H-TCP *cwnd* dynamics (Symmetric network, 250Mbit/sec, queuesize 20% BDP).

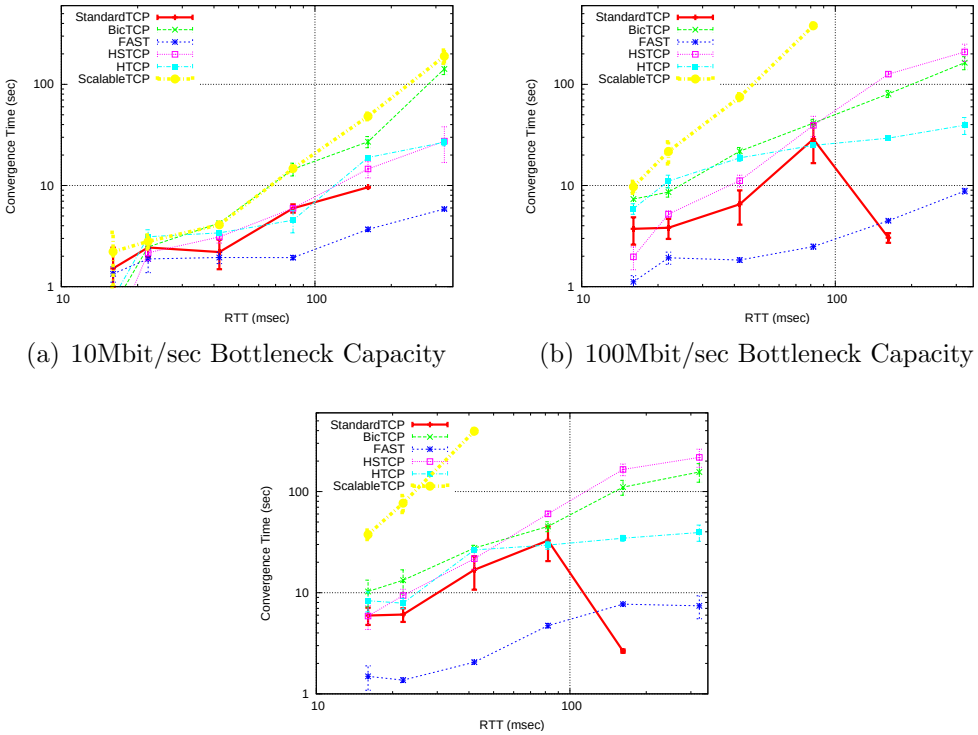
These tests correspond to network buffer sizes of 27, 70 and 342 packets receptively which correspond approximately to the sum of α for both competing FAST flows respectively.

At high latencies, H-TCP increases its overhead by almost an order of magnitude; however at low latencies the results are comparable to those of Standard TCP. A zoomed figure of the *cwnd* trace of H-TCP is shown in Figure 8.20 and shows the complex synchronisation between the two competing flows.

As expected ScalableTCP has the highest overhead due to its very aggressive exponential algorithm (with the exception of FAST at low speeds). Although it should be noted that at 10Mbit/sec and 100Mbit/sec with very long latencies, H-TCP was observed to have a higher overhead than ScalableTCP.

Convergence Time

Figure 8.21 shows the convergence times between the two symmetric New-TCP flows. The small decrease of *cwnd* upon loss in ScalableTCP results in a



(a) 10Mbit/sec Bottleneck Capacity

(b) 100Mbit/sec Bottleneck Capacity

(c) 250Mbit/sec Bottleneck Capacity

Figure 8.21: Convergence time to 80% throughput between two competing flows under symmetric network conditions (Bottleneck Queuesize set to 20% BDP).

visibly increased amount of time for convergence to fairness (See Figure 8.13).

FAST, on the other hand, manages fast convergence, although this calculation arises from the interplay between flows before it converges to a stable operating region. Also, FAST appears to have problems maintaining a stable equilibrium (See Figure 8.19).

BicTCP and HSTCP have very similar profiles for convergence, although it should be noted that the scales are in log form in Figure 8.21. To put context to this, HSTCP and BicTCP spend almost a third of a ten minute test converging to stable state for the longer latency tests.

H-TCP is able to converge quickly even in high latency environments.

This is due to the adaptive back-off (See Figure 6.1.3) whereby upon sudden changes in bandwidth at congestion, a more conservative back-off of 0.5 is utilised. This is aided by the short congestion epoch time as a result of its polynomial increase. Even with a RTT of 324ms, two H-TCP flows converge to 80% throughput ratio within approximately half a minute under all tested conditions; this is about four times faster than BicTCP or HSTCP.

The low convergence times of Standard TCP for the latencies of 162ms were found to be caused by the second Standard TCP flow's *cwnd* to grow beyond the of the first flow as a consequence of slow start; which causes immediate convergence between the two competing flows.

8.3.2 Asymmetric Network

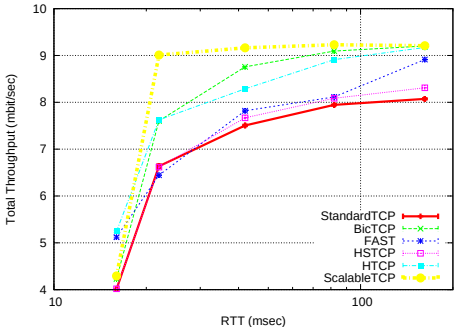
The interaction between New-TCP flows of different latencies was investigated. This was accomplished by keeping the initial flow at a latency of 162ms and varying the latency of the second flow from 16ms to 162ms.

In order to determine the effects of buffering upon the competing flows, two values were use for the queue-size 20% of the BDP of the short latency flow (small queue-size) and that of 20% of the long latency flow (large queue-size).

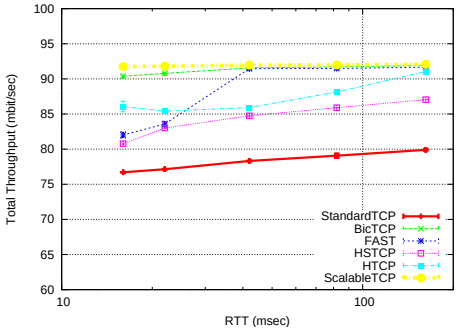
Small Queue-size

Figure 8.22 shows the aggregate throughput of both flows under asymmetric network conditions with a queue size buffer set to 20% BDP of the smallest latency flow which is marked on the abscissa.

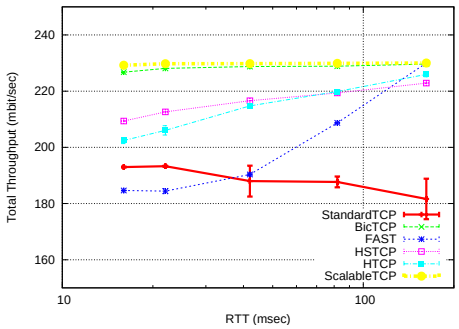
Similar to the results shown in Figure 8.10, ScalableTCP and BicTCP managed to achieve the highest throughput out of all the New-TCP algo-



(a) 10Mbit/sec Bottleneck Capacity



(b) 100Mbit/sec Bottleneck Capacity

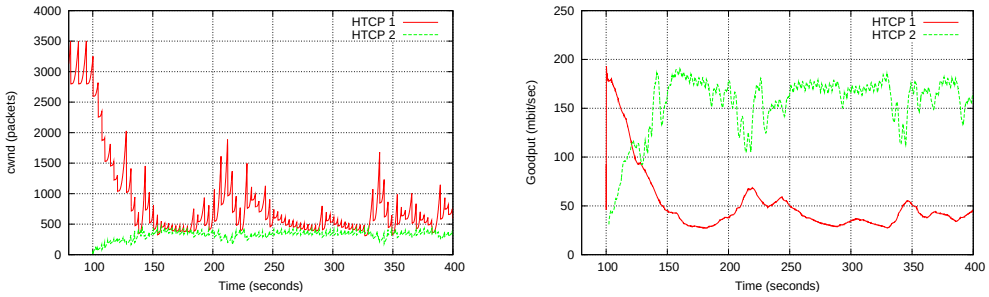


(c) 250Mbit/sec Bottleneck Capacity

Figure 8.22: Aggregate Goodput of two competing TCP flows with asymmetric network conditions. The first flow is set to 162ms RTT and the second flow to that as shown (Bottleneck Queuesize set to 20% BDP of the low latency flow).

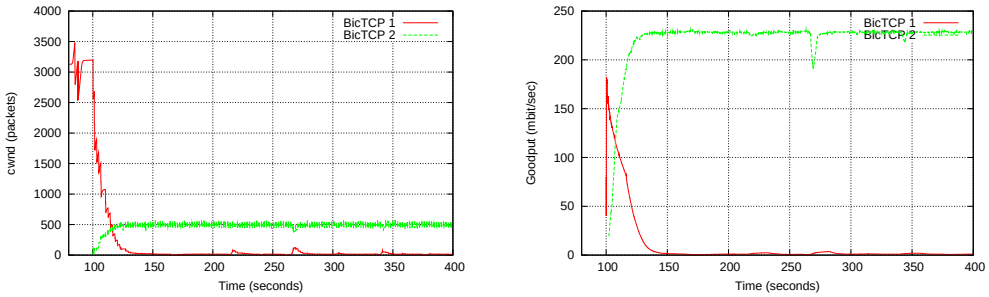
rithms. Also FAST appears to have problems with some of the lower BDP network conditions which is most evident with the small bottleneck queuesizes of the low latency network with 250Mbit/sec capacity.

What is most notable under this test is that H-TCP's utilisation is notably decreased with larger network capacities. Figures 8.23 and 8.24 shows the *cwnd* and goodput histories for H-TCP and BicTCP respectively. A comparison against BicTCP was chosen as it achieves very close to line rate in this set of tests for all capacities. Due to the inverse relation between throughput and latency (See Equation 4.1), the shorter latency flow is able to consume



(a) Congestion Window (b) Goodput

Figure 8.23: H-TCP under asymmetric network conditions (250Mbit/sec capacity, with Flow 1 experiencing 162ms RTT and Flow 2 experiencing 22ms RTT, queuesize 20% BDP of 22ms).

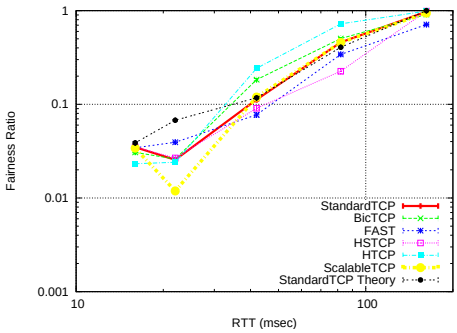


(a) Congestion Window (b) Goodput

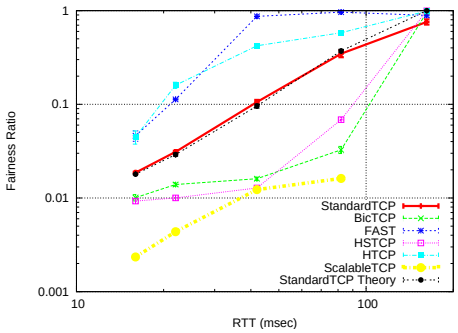
Figure 8.24: BicTCP under asymmetric network conditions (250Mbit/sec capacity, with Flow 1 experiencing 162ms RTT and Flow 2 experiencing 22ms RTT, queuesize 20% BDP of 22ms).

more of the network resources which results in a higher throughput compared to the longer latency flow. It can be seen that the first BicTCP flow consumes almost all of the available throughput. H-TCP, however, is able to maintain (almost) *cwnd* fairness, and as such the longer latency H-TCP flow is still capable of a achieving a respectable proportion of the network throughput.

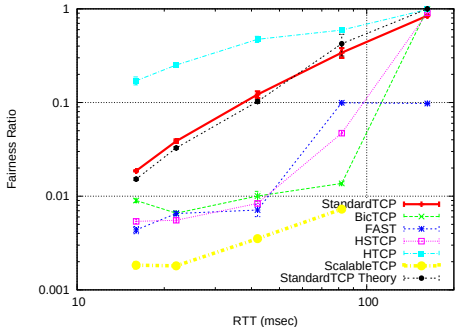
The fairness between the New-TCP algorithms is shown in Figure 8.25.



(a) 10Mbit/sec Bottleneck Capacity



(b) 100Mbit/sec Bottleneck Capacity



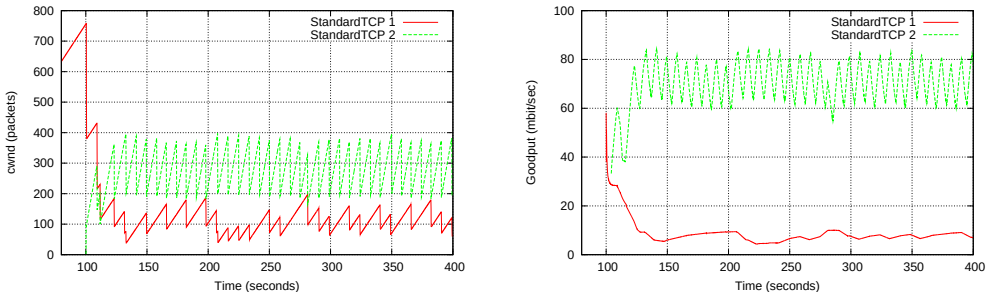
(c) 250Mbit/sec Bottleneck Capacity

Figure 8.25: Fairness between two competing TCP flows with asymmetric network conditions. The first flow is set to 162ms RTT and the second flow to that as shown (Bottleneck Queuesize set to 20% BDP of the low latency flow).

A typical example of the performance of Standard TCP under asymmetric networks is shown in Figure 8.26. The lack of synchronisation between Standard TCP flows have been captured in the theoretical calculations of fairness as show in Figure 8.25.

Whilst the unfairness experienced by all algorithms is similar to Standard TCP under low speed 10Mbit/sec tests, at higher capacities it was observed that there is a marked difference in the unfairness properties of each New-TCP algorithm.

ScalableTCP has the most severe unfairness being upto two orders of



(a) Congestion Window (b) Goodput

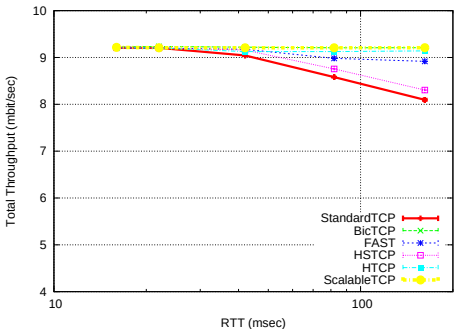
Figure 8.26: Standard TCP under asymmetric network conditions (100Mbit/sec capacity, with Flow 1 experiencing 162ms RTT and Flow 2 experiencing 42ms RTT, queuesize 20% BDP of 42ms).

magnitude more unfair than Standard TCP in identical test environments. HSTCP, BicTCP and FAST have similar unfairness properties at across the board. Out of all the algorithms, only H-TCP shows fairness across the range of RTT ratios that is more than that of Standard TCP. This suggests that on a long distance link, H-TCP flows will not be starved of throughput by shorter latency H-TCP flows, whilst the other New-TCP algorithms are likely to cause lock-up of the competing New-TCP flow.

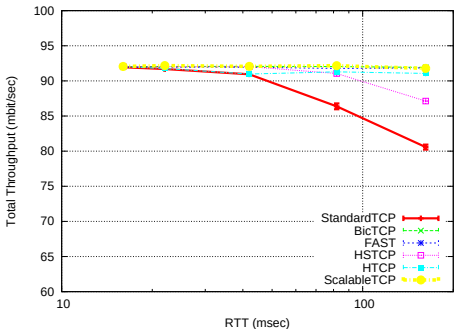
Large Queuesize

The effect of a larger queue size clearly shows an almost full utilisation of the network capacity for all algorithms as shown in Figure 8.27. However, Standard TCP still suffers from reduced goodput with long latencies. This effect was also observed with HSTCP (although not as much), and is expected as it only gradually deviates from that of Standard TCP with larger BDP at such low *cwnd* values.

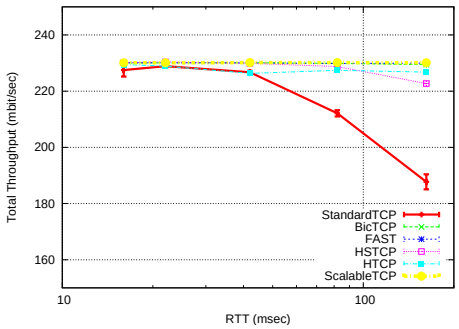
The fairness between the flows is shown in Figure 8.28. The effect of differing queue sizes upon the fairness between two asymmetric ScalableTCP



(a) 10Mbit/sec Bottleneck Capacity



(b) 100Mbit/sec Bottleneck Capacity

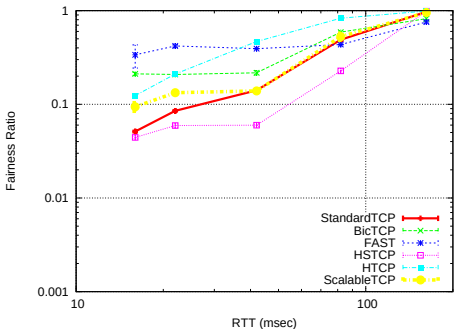


(c) 250Mbit/sec Bottleneck Capacity

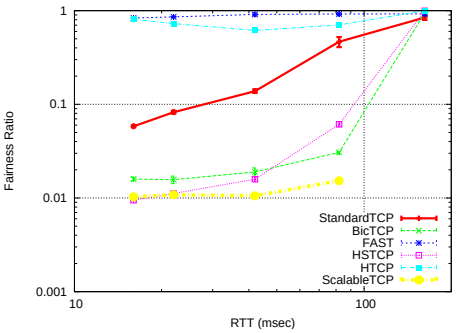
Figure 8.27: Aggregate Goodput of two competing TCP flows with asymmetric network conditions. The first flow is set to 162ms RTT and the second flow to that as shown (Bottleneck Queuesize set to 20% BDP of the high latency flow).

flows is shown in Figure 8.29. In both cases the long latency flow is severely handicapped by the aggressiveness of the second flow.

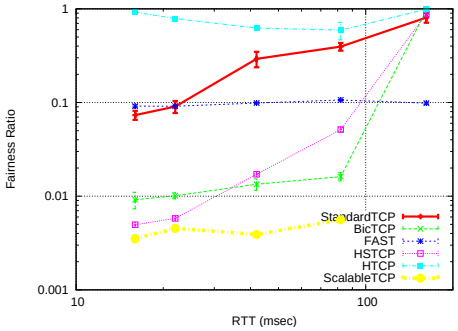
H-TCP is much more fair with bandwidth allocation between flows as shown in Figure 8.30. It shows that even with large latency differences between flows, the longer, and hence less responsive flow is able to maintain a sufficiently large *cwnd* to enable high throughput transport, and hence achieve fairness. As the bandwidth is inversely proportional to the latency of the flow, a larger *cwnd* is required by a longer latency flow in order to remain fair. This is shown in Figure 8.30(b).



(a) 10Mbit/sec Bottleneck Capacity

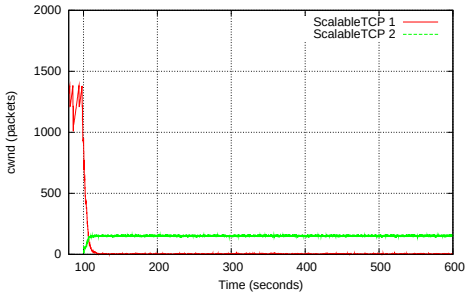


(b) 100Mbit/sec Bottleneck Capacity

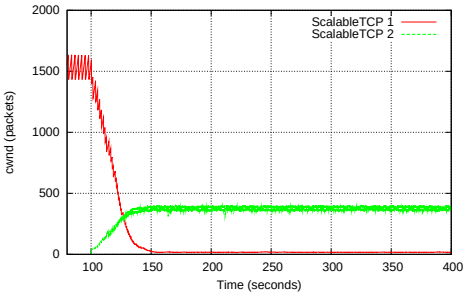


(c) 250Mbit/sec Bottleneck Capacity

Figure 8.28: Fairness between two competing TCP flows with asymmetric network conditions. The first flow is set to 162ms RTT and the second flow to that as shown (Bottleneck Queuesize set to 20% BDP of the high latency flow).

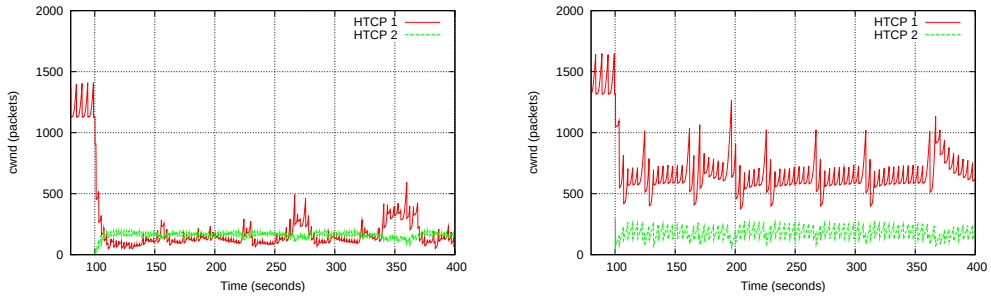


(a) 26 packets (20% of 16ms BDP)



(b) 270 packets (20% of 162ms BDP)

Figure 8.29: ScalableTCP under asymmetric network conditions (100Mbit/sec capacity, with Flow 1 experiencing 162ms RTT and Flow 2 experiencing 16ms RTT).



(a) 36 packets (20% of 16ms BDP) (b) 270 packets (20% of 162ms BDP)

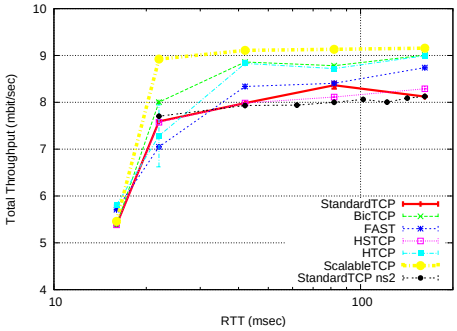
Figure 8.30: H-TCP under asymmetric network conditions (100Mbit/sec capacity, with Flow 1 experiencing 162ms RTT and Flow 2 experiencing 22ms RTT. Bottleneck queue size at 20% BDP of the 162ms flow).

8.3.3 Friendliness

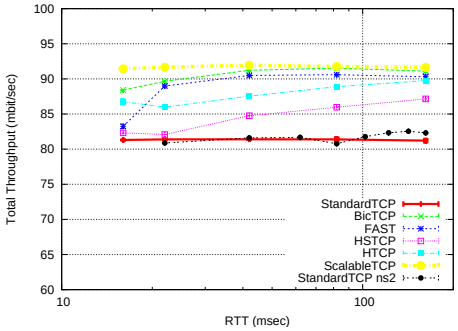
Figure 8.31 shows the aggregate throughput between a Standard TCP flow and a New-TCP flow under symmetric network conditions.

As the New-TCP algorithms are more aggressive, especially in high BDP environments, the question is by what ratio the goodput is shared between the legacy TCP flow and the New-TCP flow. Figure 8.32 shows the friendliness of New-TCP algorithms against Standard TCP under identical network conditions across a range of different network environments.

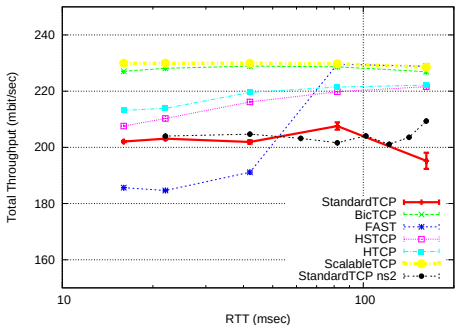
The aggressiveness of a ScalableTCP flow against Standard TCP can be seen in Figure 8.33. Even under environments where Standard TCP is still capable of high throughput, the *cwnd* dynamic of ScalableTCP forces the goodput of Standard TCP to well below fair sharing of the link. It can be seen that all loss-based congestion algorithms predictably become less fair to Standard TCP as the bottleneck capacity increases. At 10Mbit/sec, however, ScalableTCP and FAST are still more aggressive to the Standard TCP flow and impose an unfairness of approximately 1:5.



(a) 10Mbit/sec Bottleneck Capacity



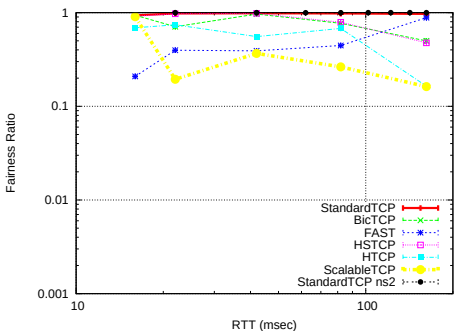
(b) 100Mbit/sec Bottleneck Capacity



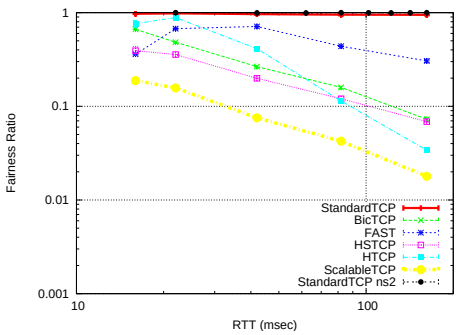
(c) 250Mbit/sec Bottleneck Capacity

Figure 8.31: Aggregate Goodput of a Standard TCP flow competing against a New-TCP flow with symmetric network conditions (Bottleneck Queuesize set to 20% BDP).

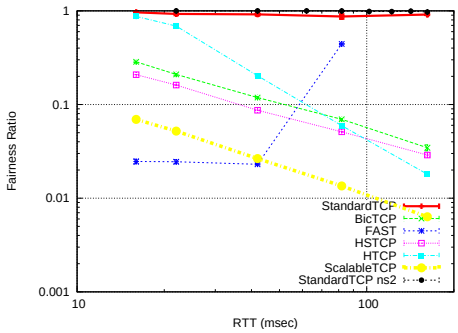
The interaction between Standard TCP and FAST is shown in Figure 8.34 and shows that the increase in average queue occupancy of the Standard TCP flow causes the FAST flow to back away due to the delay-based response to congestion control of FAST. The effect of this approach is that FAST exhibits an almost opposite representation of the Standard TCP goodput. The result of this is that FAST is especially friendly under environments where Standard TCP is incapable of high goodput, but also implies that the FAST flow actually achieves less goodput at the cost of increased fairness with Standard TCP.



(a) 10Mbit/sec Bottleneck Capacity

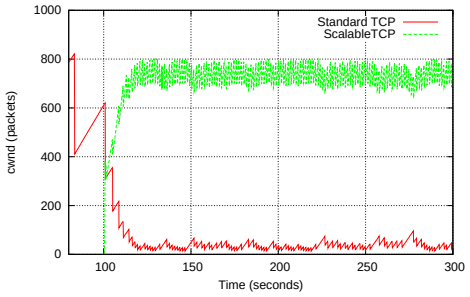


(b) 100Mbit/sec Bottleneck Capacity

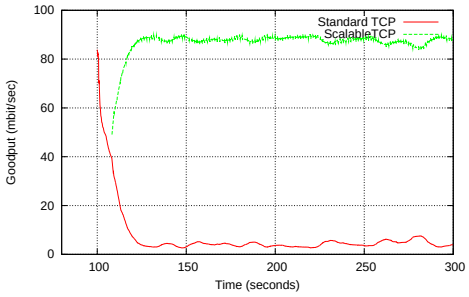


(c) 250Mbit/sec Bottleneck Capacity

Figure 8.32: Friendliness between a Standard TCP flow competing against a New-TCP flow with symmetric network conditions (Bottleneck Queuesize set to 20% BDP).

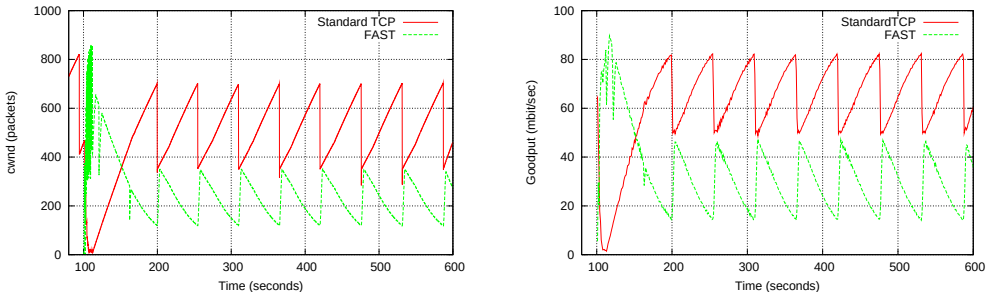


(a) Congestion Window



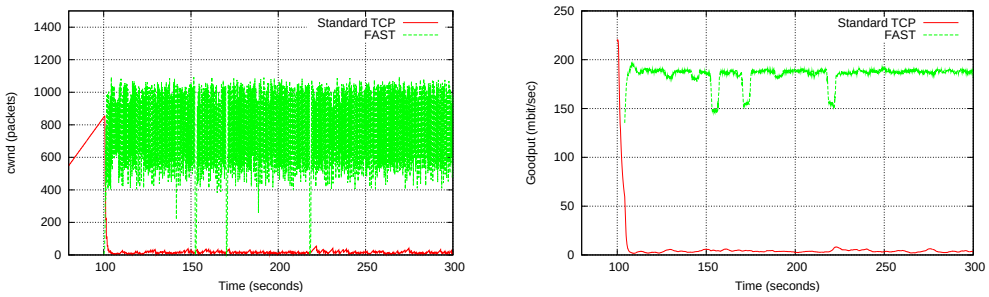
(b) Goodput

Figure 8.33: Standard TCP competing against ScalableTCP under symmetric network conditions (100Mbit/sec capacity, 82ms RTT, 20% BDP queuesize).



(a) Congestion Window (b) Goodput

Figure 8.34: Standard TCP competing against FAST under symmetric network conditions (100Mbit/sec capacity, 82ms RTT, 20% BDP queuesize).

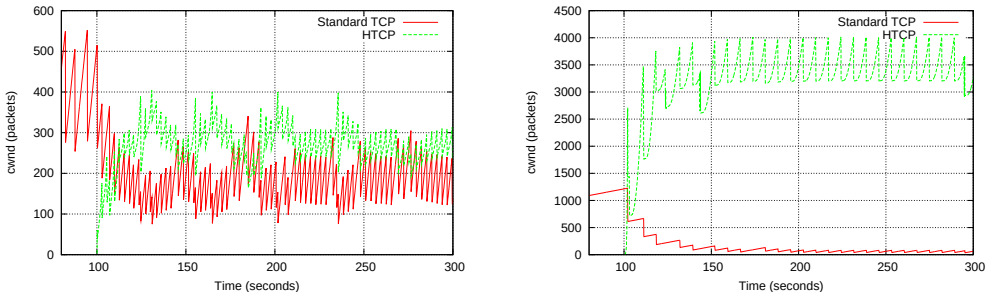


(a) Congestion Window (b) Goodput

Figure 8.35: Standard TCP competing against FAST under asymmetric network conditions (250Mbit/sec capacity, 42ms RTT, 20% BDP queuesize).

The failure of the FAST dynamic upon insufficiently provisioned queuesizes is shown in Figure 8.35. In contrast to the dynamic as shown in Figure 8.34, FAST continually induces an aggressive ramp up without being able to stabilise into its congestion control algorithm. The result is unfairness between the Standard TCP flow and the FAST flow that is actually more aggressive than that of ScalableTCP. The utilisation under such environments is also lower than that of just two Standard TCP flows, due primarily to flooding of packets into the network by the FAST flow at 250Mbit/sec.

Conversely to the other loss-based algorithms, H-TCP at high speeds

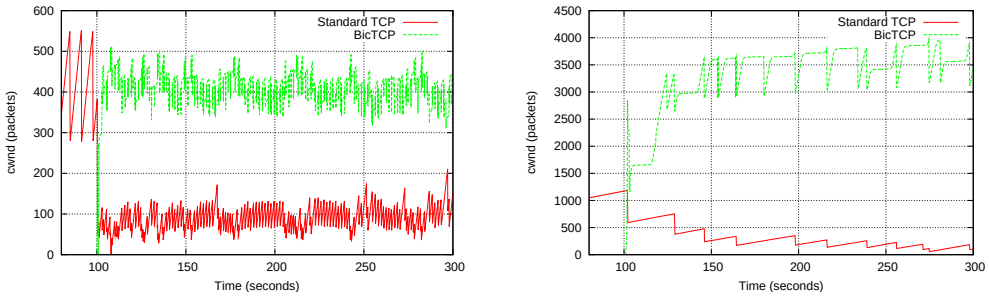


(a) 22ms (b) 162ms

Figure 8.36: Standard TCP competing against H-TCP under asymmetric network conditions (250Mbit/sec capacity, 20% BDP queuesize).

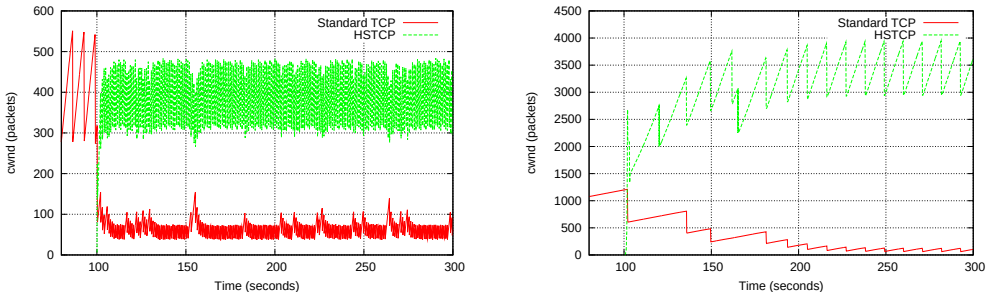
shares its goodput very fairly in low latency environments, and becomes more aggressive as the RTT increases; utilising more of the path capacity in environments where Standard TCP would be incapable of high transport. This is shown in Figure 8.36. This dynamic is due to the time based adaptive congestion control that becomes more aggressive as the congestion epoch time becomes larger, and indicates that Standard TCP is incapable of utilising the network resources. However, the use of a polynomial increase function results in a very aggressive algorithm that results in less friendliness to Standard TCP compared to that of HSTCP and BicTCP in high bandwidth delay product environments.

It was observed that the friendliness of BicTCP and HSTCP follow similar trends to each other. Figure 8.37 shows the interaction between BicTCP and Standard TCP for 22ms and 162ms with a bottleneck capacity of 250Mbit/sec. It can be seen that the plateau'ing of BicTCP's *cwnd* facilitates the growth of Standard TCP's *cwnd* and hence improves fairness upon long latency environments - whilst still maintaining a high value of *cwnd* to enable high utilisation of the network capacity. However, under short latency environ-



(a) 22ms (b) 162ms

Figure 8.37: Standard TCP competing against BicTCP under asymmetric network conditions (250Mbit/sec capacity, 20% BDP queuesize).



(a) 22ms (b) 162ms

Figure 8.38: Standard TCP competing against HSTCP under asymmetric network conditions (250Mbit/sec capacity, 20% BDP queuesize).

ments, the plateau has negligible effect and effectively becomes an aggressive linear increase causing unfairness against the Standard TCP flow.

The performance of HSTCP's friendliness is shown in Figure 8.38. Compared to that of BicTCP (Figure 8.37), it is observed that the period between (synchronised) congestion events is less, therefore forcing Standard TCP into a slightly lower average *cwnd* causing less friendliness of HSTCP against Standard TCP than that of BicTCP. At lower latencies, the opposite is true; HSTCP induces congestion less frequently and therefore is more fair against the Standard TCP flow.

New-TCP	Symmetric	Asymmetric		Friendliness
		Small Buffer	Large Buffer	
Standard TCP	207.6±1.3	187.7±1.9	212.1±1.1	207.6±1.3
BicTCP	229.7±0.0	228.9±0.2	229.9±0.2	228.7±0.1
FAST	219.6±6.4	208.7±0.2	229.8±0.1	229.6±0.1
HSTCP	218.4±0.0	219.3±0.1	229.7±0.6	219.8±0.1
H-TCP	225.9±0.0	219.8±0.1	227.4±0.7	221.4±0.4
ScalableTCP	229.7±0.5	229.9±0.1	230.2±0.0	229.7±0.1

Table 8.1: Summary goodput (in Mbit/sec) of two competing New-TCP flows at 250Mbit/sec bottleneck and 82ms RTT.

8.4 Discussion of Results

Even simple tests can show the very different performances of the different New-TCP algorithms. This section describes the various congestion control algorithms and how they affect the performance metrics.

8.4.1 Goodput

All New-TCP algorithms fulfill the primary goal of being able to achieve high goodput across the range of network environments under investigation as shown in Table 8.1.

Amongst the algorithms, ScalableTCP and BicTCP are consistently able to achieve the highest utilisation of the all of the algorithms tested.

FAST also has good prospects of utilising all of the capacity of the path. However, it appears to be hampered by the requirement to have a queue size buffers suited to the value of its α parameter in order to remain stable. When the network buffer size is approximately less than the sum of FAST's α values for *every FAST flow* then the performance is seriously degraded as the flows cause the bottleneck queue to overflow, which induces large oscillations

New-TCP	Symmetric	Asymmetric		Friendliness
		Small Buffer	Large Buffer	
Standard TCP	0.87±0.05	0.34±0.04	0.40±0.04	0.87±0.05
BicTCP	0.97±0.02	0.01±0.00	0.02±0.00	0.07±0.00
FAST	0.31±0.14	0.10±0.00	0.11±0.00	0.44±0.00
HSTCP	1.00±0.00	0.05±0.00	0.05±0.00	0.05±0.00
H-TCP	1.00±0.00	0.59±0.00	0.59±0.13	0.06±0.00
ScalableTCP	-	0.01±0.00	0.01±0.00	0.01±0.00

Table 8.2: Summary fairness of two competing New-TCP flows at 250Mbit/sec bottleneck and 82ms RTT.

in FAST's throughput which can be *less* than that of Standard TCP.

As expected, HSTCP shows very little deviation from Standard TCP goodput at low speeds, but the increase of BDP shows HSTCP to be as aggressive as the other algorithms as larger *cwnd* values are reached.

H-TCP has similar goodput performance to that of HSTCP.

8.4.2 Fairness and Friendliness

Whilst the fairness between symmetric flows was not expected to be a difficult test for the New-TCP algorithms, results showed otherwise. This was most noticeable for ScalableTCP and FAST and is demonstrated in Table 8.2. In the former case, unfairness resulted primarily because of the long convergence times required due to the small back-offs experienced by the high throughput flow which requires many congestion epochs to result in fair share between flows.

FAST, on the other hand, appears to have instability issues across the tested environments which result in both short term and long term unfairness issues. Unlike the loss-based algorithms, the dynamics of FAST are very dependent upon the queue size allocation. At small sizes, the aggres-

sive probing of FAST causes severe unfairness against Standard TCP and other FAST flows. However, with sufficiently provisioned buffers, sized to the sum of α value(s) of the FAST flow(s), long term fairness is *optimal*, even when competing against Standard TCP. The term optimal is used as it is not aggressive against the competing flow, but facilitates the usage of all of the spare bandwidth on the link. It is therefore optimal in the sense that it attempts to not induce loss on the competing flow and hence enables the perturbative flow to achieve as much goodput as it can without inducing losses on the network path that could result in unfairness. This is especially important in high-speed, high-delay networks as there is plenty of spare bandwidth, and FAST is capable of using the throughput without being too aggressive.

H-TCP also shows high friendliness at low latencies, however becomes less friendly with high latencies due to the increased congestion epoch times.

More revealing are the asymmetric tests which show that all algorithms exhibit lower fairness than that of Standard TCP with the exception for H-TCP. The implication of RTT unfairness upon all of the loss-based algorithms is that short latency flow almost completely prevents the long latency flow from achieving any throughput. The way in which fairness scales is of vital importance as higher network capacities (and BDPs) result in a greater degrees of unfairness between flows of different latencies as the high throughput flow becomes more aggressive. This is most evident with ScalableTCP and BicTCP which almost consistently prevent another flow from attaining sufficiently large values of *cwnd* to enable fairness.

More interestingly, whilst ScalableTCP and BicTCP perform the best with regards to goodput performance, they are the worst performers under the fairness metrics.

New-TCP	Convergence Time
Standard TCP	32.8±12.3
BicTCP	45.2±5.1
FAST	4.7±0.2
HSTCP	60.2±3.7
H-TCP	29.6±2.7
ScalableTCP	-

Table 8.3: Summary convergence times (seconds) of two competing New-TCP flows at 250Mbit/sec bottleneck and 82ms RTT under Symmetric network conditions (20% BDP queue-size).

HSTCP also shows RTT unfairness which is caused by the relatively low increase parameters and small decrease factors of low *cwnd* flows. This results in a flow that is not as aggressive as the low latency flow (which gain large *cwnd* values quickly) and therefore incapable of achieving high throughput.

H-TCP, with its RTT Scaling, enables fairer sharing of asymmetric links; so much so that it is actually fairer than Standard TCP. This therefore prevents lock-out and will ensure that even very long latency flows will not be starved of goodput.

8.4.3 Responsiveness/Convergence Time

It can be seen in Figure 8.13 that the convergence times of ScalableTCP are very slow (it does not converge to fairness as shown in Table 8.3). Even though ScalableTCP has a small congestion epoch time, the decrease factor of 0.875 means that the drop in throughput of the first flow at the moment of congestion is small, and is therefore slow to react to sudden decreases in available bandwidth.

This effect is also evident with HSTCP during the time of the test (see Figure 8.21) and can readily be seen that the convergence time is in the

order of hundreds of seconds for a symmetric latency of 162ms. In HSTCP's case, even though the decrease factor is not static, but dependent upon the *cwnd* value, a high capacity flow has smaller a back-off per congestion, and therefore many more congestion epochs are required for the flows to converge to fairness. Meanwhile, HSTCP flows with small *cwnd*s experience large decreases upon congestion, and therefore are much more likely to take a longer time under periodic losses to reach a large value of *cwnd*. [NY04] specifies an adaptation to the back-off times of HSTCP such that the decrease factor is the same as that of Standard TCP ($\beta = 0.5$) under a detected downward trend of *cwnd* at consecutive congestion events.

Slow convergence is also evident with BicTCP, where even though the decrease parameter is static, the time between congestion epochs is dependent upon the increase parameter. Under low BDP's, BicTCP reverts to an almost 'additive increase only' regime which aids short congestion epoch times and therefore improves convergence times. However, as larger *cwnd* values are reached by BicTCP, the result of this plateau is that the congestion epoch time is larger, and as such the convergence time increases.

FAST, unlike all of the loss-based algorithms, always results in the perturbing flow achieving larger *cwnd* values upon start-up than that of the existing flow. As a result, the convergence times for FAST is very small - however, this does not include the time required to actually reach stability between competing flows.

H-TCP takes the approach whereby under highly variable environments, the back-off factor is reduced to that of Standard TCP. Also, as H-TCP's α is polynomial in time, the period between congestion epochs is always small. The combination of these two factors results in a fast convergence between competing H-TCP flows where for long latency environments, the

New-TCP	Symmetric	Asymmetric		Friendliness
		Small Buffer	Large Buffer	
Standard TCP	0.52±0.01	2.09±0.15	1.57±0.25	0.53±0.01
BicTCP	5.08±0.00	6.44±0.70	5.05±0.67	3.25±0.40
FAST	20.2±10.9	29.2±0.61	0.36±0.04	0.01±0.00
HSTCP	2.95±0.01	7.63±0.65	6.53±0.77	3.98±0.21
H-TCP	3.11±0.04	4.99±0.09	4.47±1.00	3.99±0.13
ScalableTCP	9.64±0.15	12.3±0.64	11.45±0.83	22.4±0.08

Table 8.4: Summary overhead (Bytes/Mbit/sec × 1e-7) of two competing New-TCP flows at 250Mbit/sec bottleneck and 82ms RTT.

convergence time is almost constant.

8.4.4 Overhead

H-TCP is good at being fair and switching into aggressiveness to utilise spare capacity as shown in Table 8.4. However, the polynomial increase with respect to time means that a lot of packets can be lost in one window, and efficiency decreases as a result. This is most evident for network environments which result in large periods of congestion epochs, whilst the overhead is similar to Standard TCP under environments with short congestion epoch times.

FAST, with its delay based approach to congestion control, is also very capable of low loss rates, but requires that the bottleneck queue sizes are provisioned such that they are at least equal to the number of flows multiplied by their α values. Under such environments, the overhead of FAST can be as much as two orders of magnitude better than Standard TCP. However, under environments with low levels of queue provisioning along the path, the loss rates of FAST are much higher resulting in two orders of magnitude less efficiency compared to StandardTCP. This is caused primarily by the

aggressive increase of FAST as it attempts to determine an optimal rate to send data at.

BicTCP does very well in terms of reducing the number of losses due to the use of its plateau as this helps maintain a high throughput and increases the amount of time between congestion epochs (and hence loss events). The subsequent additive increase after loss also facilitates the quick probing of capacity and determination of a new plateau for *cwnd*, without substantially increasing the overhead associated with the subsequent probing.

ScalableTCP, whilst very aggressive, does not have that much more of an overhead than that the other algorithms under test (except FAST with low BDP queue-size allocations).

8.5 Summary

The deployment of New-TCP is important in order for network applications to be able to utilise future network resources effectively. However, due to the requirement of large values of *cwnd* in order to maintain sufficient amounts of data on the network to facilitate high throughput transport, the dynamic of growing and maintain these large *cwnd* values plays a vital role in the various performance traits of New-TCP algorithms.

All algorithms tested have very different characteristics in achieving large *cwnd* values and the way they respond to loss events. However, a careful choice of α and β is required to maintain a balance between flows such that neither slow convergence nor lock-out occurs. There is a similar argument for friendliness; the slow increase and large decrease of Standard TCP means that it will nearly always result in low goodput performance when competing with New-TCP flows which result in low fairness/friendliness.

The importance of analysing the performance of each New-TCP with more than just a single metric was demonstrated. Whilst ScalableTCP and BicTCP show the highest goodput of all of the New-TCP algorithms, they have problems with slow convergence which results in short term unfairness between competing flows.

Also, whilst FAST appears to have problems competing with another FAST flow, it shows benefits in short term deployability with its reactive approach to congestion by ensuring high levels of friendliness with a competing Standard TCP flow.

One problem which is exhibited by all New-TCP algorithms, except for H-TCP, is RTT unfairness. At lower capacities, the level of unfairness is much lower. However, at much higher capacities the increased aggressiveness of most of the algorithms means that lower latency flows are much more aggressive and therefore cause almost complete lock-out of the longer latency flow.

H-TCP also has the advantage of fast convergence between competing flows, exhibiting almost constant convergence times at longer latencies. However, H-TCP also has problems with increased overhead with long latencies due to its time based approach to congestion control.

Chapter 9

Transport Over Wide Area Networks

The understanding of New-TCP protocols through laboratory networks is important to understand the deviations from their theoretical dynamics. However, the application on real life networks may provide insight into unexpected performance problems. This section investigates the limits of New-TCP in terms of hardware and software constraints and analyses the performance of these protocols in real high speed network environments.

9.1 Transfer Tests Across Dedicated Private Wide Area Networks

The application of tests on real network hardware will demonstrate the potential problems that users will face in providing high throughput transport across the Internet.

Two projects that have aided in the facilitation of understanding high speed network transport are the cross-UK MB-NG development network (See Appendix C.2) and the transatlantic DataTAG testbed (See Appendix C.3). Both networks were configured to provide a link capacity of 1Gbits/s and the performance of New-TCP algorithms was investigated.

9.1.1 Methodology

A series of tests were conducted to analyse the performance of New-TCP flows at 1Gb/sec on the DataTAG and MB-NG testbeds. The two extremes of latencies available with these two networks enable analysis of the *cwnd* dynamics of New-TCP. DataTAG provides the high capacity, high latency networks at which Standard TCP fails and these algorithms are designed to operate, whilst the low latencies of MB-NG can provide information regarding the switch between the low and high speed mode of these algorithms.

At the time of the testing only HSTCP, ScalableTCP and H-TCP had been proposed and were therefore available for testing. As such, the results within this section are limited to only these New-TCP algorithms.

`iperf` [TQD+03] was used to initiate TCP flows that simulate bulk transfer using standard IP packet sizes of 1500B. All machines used in the tests were installed with the altAIMD kernel version 0.3 and were run with Appropriate Byte Counting (See Section 5.2.4) with local host buffers set to sufficiently large values as shown in Table B.2 to ensure that performance bottlenecks did not exist in the end-host systems.

All flows were also configured with sufficient socket buffer allocation to prevent host-based flow control and each test was repeated at least three times to determine an appropriate standard error.

A series of different network conditions were created in order to monitor the performance of New-TCP algorithms. They are described as follows:

Variable UDP Background Load

Given a static queue-size, a path may experience varying loads that would represent the background traffic of the path. Therefore, an experiment was devised to test the capability of these algorithms as the average queue occupancy is increased as a result of an increase in background traffic load.

Whilst the buffer dynamic of running constant bit rate (CBR) traffic would not be representative of the complex interactions of self-similar traffic [CB95, LTWW94, PKC97], UDP CBR background traffic is used to simplify understanding. As such, a custom made tool, `FIP`, was developed to generate UDP based background traffic to guarantee packet delivery times of a specific traffic profile based on the number of packets that should be sent out per time interval.

Tests were conducted with both single and ten concurrent New-TCP flows competing against various loads of CBR background traffic. For the case of multiple concurrent TCP flows, the flows were distributed evenly over the available testbed PCs in order to reduce hardware and scheduling constraints imposed by running many flows on a single machine.

Number of Concurrent Flows

As New-TCP protocols are designed to be replacement protocols for the Standard TCP algorithm, it is important to understand the affects of running many flows on the network. Therefore, an experiment was devised to run many concurrent 'parallel' New-TCP flows on the test networks to determine

the goodput and fairness characteristics of the aggregate traffic.

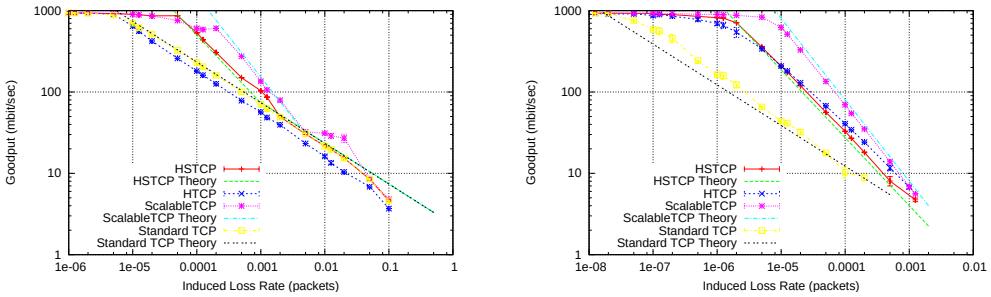
Impact

The primary aim of New-TCP proposals is to utilise the available link bandwidth better. However, the result of this increased utilisation is that it will disrupt the traditional 'background' traffic sharing the link by 'stealing' bandwidth from Standard TCP flows and causing unfriendliness (See Section 8.3.3). The amount by which New-TCP algorithms affect varying numbers of legacy Standard TCP bulk transport flows can be defined through a *impact* which provides an interpretation of fairness and an indication of how intrusive a New-TCP algorithm is:

$$Impact = \frac{B(n) - B(i)}{B'(n) - B'(i)} \tag{9.1}$$

Given a pipe with n TCP flows, define $B(n)$ to be the aggregate goodput of n Standard TCP flows and $B(i)$ to be the goodput of i Standard TCP flow in the presence of the aggregate. Under identical network conditions, also define $B'(n)$ to be the aggregate goodput of n flows, of which $n - i$ flows are that of Standard TCP flows in the presence of i New-TCP flows. Furthermore, define that $B'(i)$ is the aggregate goodput of i New-TCP flows.

As such, the *impact* provides the ratio of achieved goodput of the Standard TCP aggregate with and without the inclusion of i flows of New-TCP. Therefore, this represents the impact of i New-TCP flows upon the goodput achieved by n Standard TCP bulk transport users compared to that of Standard TCP flow. A value of $UIF > 1$ means that the goodput experienced by a legacy TCP flow as a result of being run concurrently with a New-TCP flow is diminished due to the aggressiveness of the New-TCP algorithm.



(a) MBNG (b) DataTAG

Figure 9.1: Response Function of New-TCP over dedicated test networks.

9.1.2 Calibration

A calibration test was conducted to ensure the appropriate implementation of the New-TCP algorithms under the altAIMD 0.3 kernel. This was achieved by comparing experimental results with that of theory for the response function (See Section 5.4.2).

This was performed between a single pair of end systems where packets were selectively dropped at the receiver in a regular fashion making use of a kernel modification to achieve the drop. The implementation of this packet drop relies on the counting of all received data packets at the receiver whereby the checksum for every n'th packet is declared invalid in order to simulate bit error on the network path and at packet loss rate of $p = 1/n$.

Figure 9.1 shows that the introduction of a single packet drop per window is handled well in the kernel, and that the experimental results match very closely to theory for both networks. And in agreement with theory, ScalableTCP is consistently able to obtain better or equivalent throughput than both HSTCP and H-TCP over the range of the drop rates.

It can be seen that the mode switch from low to high speeds matches closely to theory and is handled well in ScalableTCP and HSTCP in the low

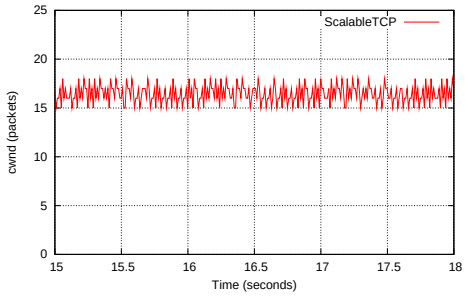


Figure 9.2: Quantisation of *cwnd*.

latency environment of MB-NG. The exception is the introduction of a slight glitch just before the mode switch of ScalableTCP at approximately $p = 0.02$ and the slightly higher goodput of HSTCP at approximately $p = 0.001$. In both cases, this was due to the switch between the respective low and high speed modes, which due to the constant packet loss rate results in larger values of *cwnd* and hence high goodput (See Section 8.2.1).

The slightly higher goodputs experienced in the experimental tests compared to the theory for MB-NG for ScalableTCP and HSTCP are due to quantisation effects as the Linux kernel stores the value of *cwnd* as integers[1]. As the implementation of the decrease calculation of *cwnd* upon congestion in Linux is $cwnd \leftarrow cwnd - b \times cwnd$, the calculation of *cwnd* upon loss is such that the rounding error is less than what it should be and hence the final value of *cwnd* is in fact higher.

This is shown for ScalableTCP at 0.02 p in Figure 9.2 where the maximum value of *cwnd* is 18 packets. As the back-off b of ScalableTCP is 1/8, the corresponding value of *cwnd* upon packet loss should be 15.75; however, due to the quantisation of *cwnd*, the actual value used is 16. This results in the

[1]In order to determine when to update *cwnd*, a separate integer counter *cwnd_cnt* which is increased per valid *ack* and checked to see if it is larger than *cwnd* before *cwnd* is increased by a whole number. Thus maintaining an increase of 1 *cwnd* per RTT under Standard TCP.

observed slight increase in the goodput in Figure 9.1.

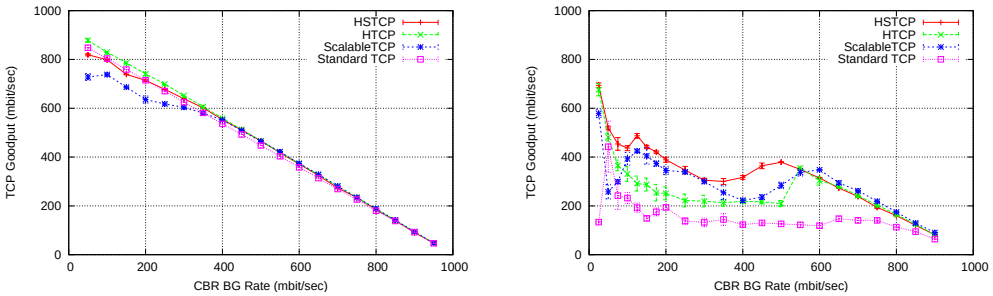
Conversely, the decrease factor of H-TCP is programmed such that $cwnd \leftarrow \beta \times cwnd$ and therefore, the calculation upon loss actually results in a lower *cwnd* and goodput. The effect of this is most notable on the MB-NG network.

The observed higher goodput achieved with Standard TCP on the DataTAG link is attributed to the method by which packets are dropped by the receiver and the limited duration of the test. As the counting of packets also include those under slow start, a large initial value of *cwnd* occurs due to the exponential increase of slow start. Therefore, with lower loss rates, a high average for the goodput is achieved due to the goodput bias as a result of slow start. This effect is less apparent for the New-TCP algorithms due to the fast(er) increases in α which ensures a fair number of congestion epochs and therefore stable operation of congestion avoidance. A solution to this bias would be to run the tests of Standard TCP for a prolonged period of time. However, due to to practical reasons this was not feasible.

On both networks, as lower loss rates result in goodputs reaching 1Gb/sec, the New-TCP flows are no longer limited by the dynamics of *cwnd*, but by physical limitations of the NIC which causes saturation of goodput at the physical line rate.

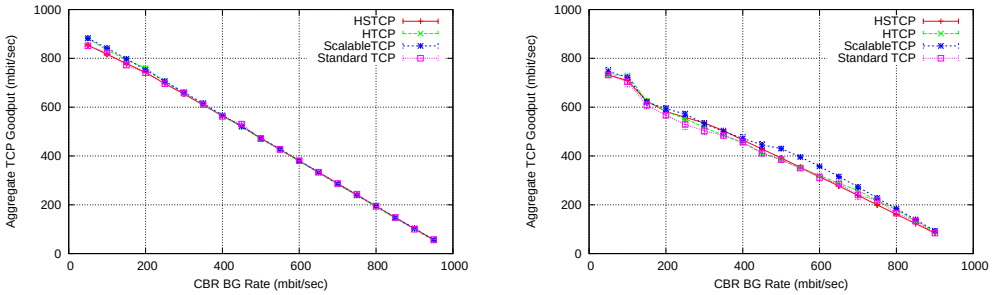
9.1.3 Results: CBR Background Traffic with 1 New-TCP Flow

Figure 9.3 shows the goodput achieved by a single TCP flow competing against various loads of CBR traffic along both MB-NG and DataTAG. Under MB-NG, all of the algorithms achieve roughly the same throughput; with H-TCP actually being able to get a little more bandwidth than all of the other



(a) MBNG (b) DataTAG

Figure 9.3: Goodput of a single TCP flow against various Background Loads.



(a) MBNG (b) DataTAG

Figure 9.4: Goodput of 10 New-TCP flows against various CBR Background Loads.

algorithms. ScalableTCP (and less so with HSTCP) flows apparently achieve lower goodput for regions of low competing background rates.

However, tests on the DataTAG testbed shows very different goodput profiles for each New-TCP algorithm which are discussed in depth in Section 9.1.7.

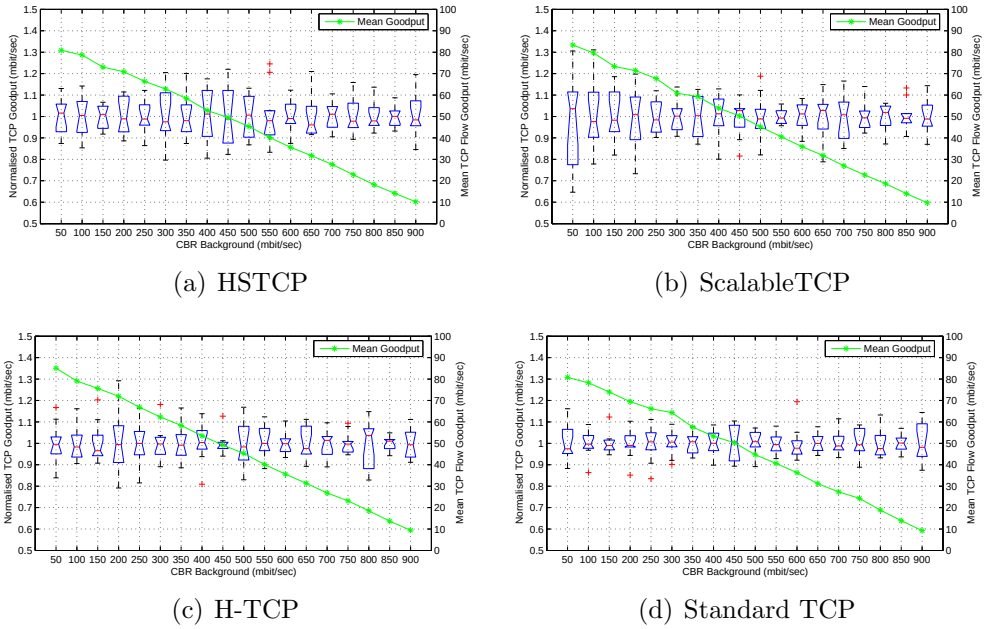
9.1.4 Results: CBR Background Traffic with 10 New-TCP Flows

Figure 9.4 shows the goodput of running 10 New-TCP flows against varying amounts of CBR traffic and shows that on both networks, the aggregate goodput of all algorithms were almost identical - including that of Standard TCP on the DataTAG network. Generally, ScalableTCP and H-TCP are able to squeeze a little more out of the available bandwidth than the other algorithms on MB-NG, yet achieved similar rates to HSTCP on DataTAG. ScalableTCP was also able to achieve a little more combined goodput at around 500Mbit/sec CBR background on the long latency link of DataTAG.

Figures 9.5 and 9.6 gives a representation of the fairness between the 10 New-TCP flows on the MB-NG and DataTAG respectively. They show box-plots representing the range and quartiles of the achieved mean goodput of the 10 competing New-TCP flows under each CBR background rate. In order to give a comparative analysis of the fairness in each network environment, the box-plots are normalised by the mean goodput of the flows which is plotted on the secondary y-axis. While the aggregate goodput results gathered represent the means and standard deviations on the mean of many iterations of the tests, the figures shown only present that of a single iteration.

Under MB-NG the fairness distribution between H-TCP flows was similar to that of Standard TCP. It can be argued that ScalableTCP shows a greater variation in the fairness distribution between flows with less competing background traffic. However, ScalableTCP shows similar fairness to that of the other algorithms at higher background rates (i.e. lower average TCP goodputs).

However, it was observed that there was a wider distribution of goodput



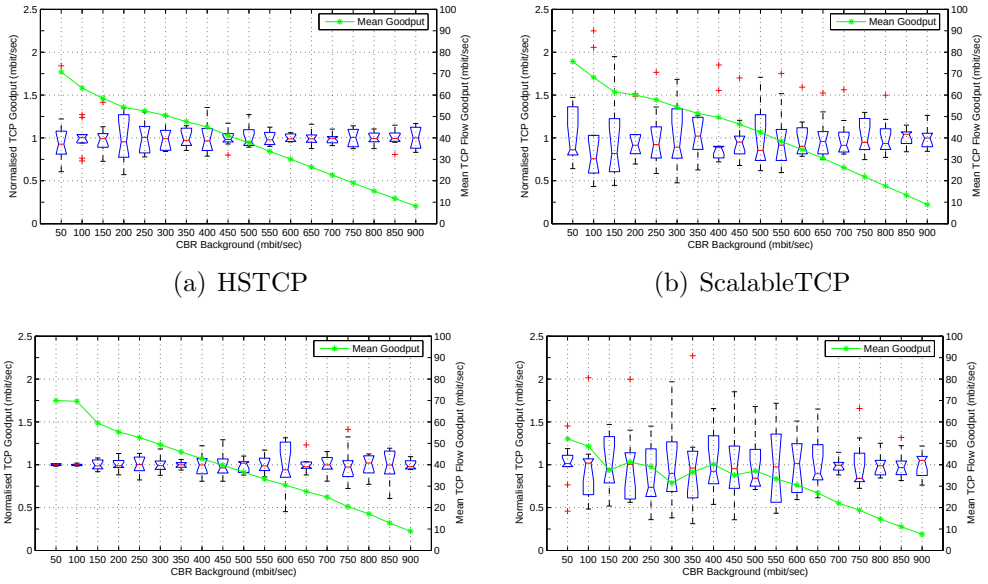
(a) HSTCP (b) ScalableTCP

(c) H-TCP (d) Standard TCP

Figure 9.5: Fairness distributions between 10 New-TCP flows against various Background Loads on MB-NG.

under DataTAG for Standard TCP compared to that of HSTCP and H-TCP. This is due to the slow growth of *cwnd* under this environment which results in a lack of congestion epochs which are required for flows to back-off and converge to fairness. Therefore, the unfairness between Standard TCP flows in the DataTAG environment is more dependent upon the exit value of *cwnd* after slow start (rather than the dynamic of AIMD) due to the relatively short duration of the tests. These results demonstrate the short term unfairness experienced between Standard TCP flows under high BDP environments.

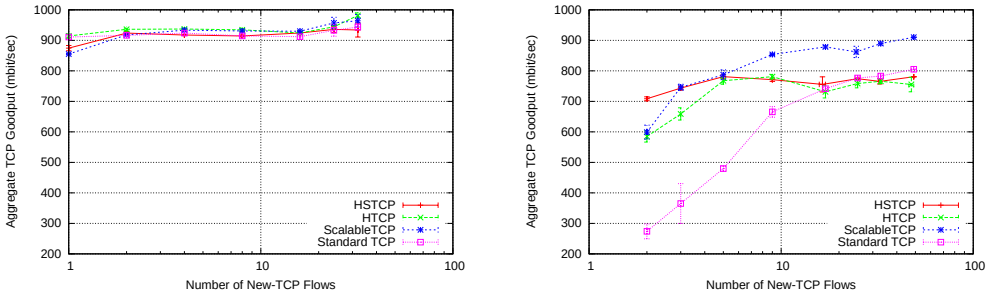
It was observed that ScalableTCP is mostly fair, except for stray flows which achieves approximately 1.5 to 2 times the mean goodput of the other flows. This is demonstrated in the box-plots by the consistent outlier point in each test. However, ScalableTCP shows a much greater variation in average flow goodputs than that of both HSTCP and H-TCP.



(a) HSTCP (b) ScalableTCP

(c) H-TCP (d) Standard TCP

Figure 9.6: Fairness distributions between 10 New-TCP flows against various Background Loads on DataTAG.

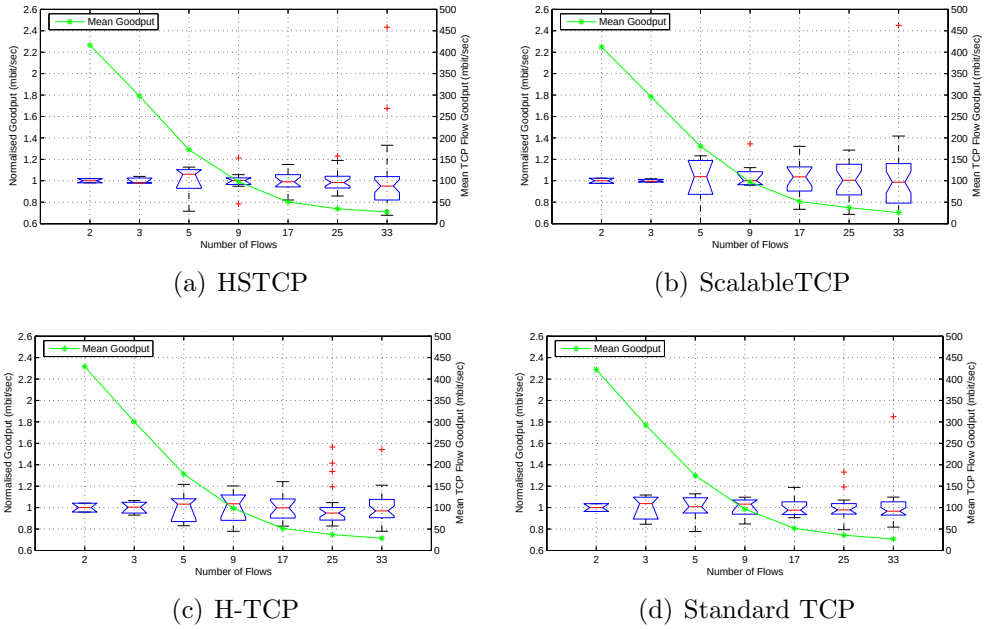


(a) MBNG (b) DataTAG

Figure 9.7: Aggregate goodput of n New-TCP flows.

9.1.5 Results: Multiple Flows

These sets of experiments demonstrate the fairness between identical New-TCP flows as the number of competing flows is increased. There is no competing traffic and all flows in each test are distributed between the testbed



(a) HSTCP (b) ScalableTCP

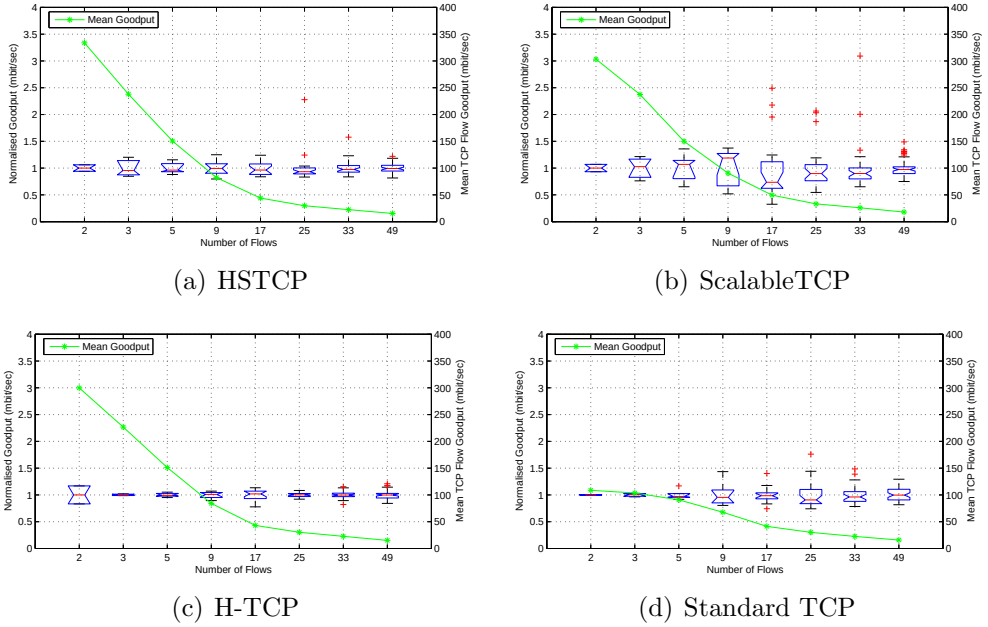
(c) H-TCP (d) Standard TCP

Figure 9.8: Fairness between various number of concurrent parallel New-TCP flows on MB-NG.

PCs.

Figure 9.7 shows the achieved aggregate goodput of n flows of each stack. A slight increase in the total achieved goodput of all flows on MB-NG was observed with an increase in the number of 'parallel' flows. All New-TCP algorithms on MB-NG also exhibit similar aggregate goodputs, with ScalableTCP and H-TCP achieving a greater goodput than that of HSTCP. This is unsurprising due to the relatively large queue size allocations of the network which facilitates high utilisation of the network.

On the DataTAG link, it was observed that Standard TCP bulk goodput scales poorly on the long latency link, with an almost linear increase in performance with the number of flows up to about 10 flows. Beyond this value, the differences in the aggregate goodput between HSTCP, H-TCP and Standard TCP are less apparent, showing a maximum utilisation of about



(a) HSTCP (b) ScalableTCP

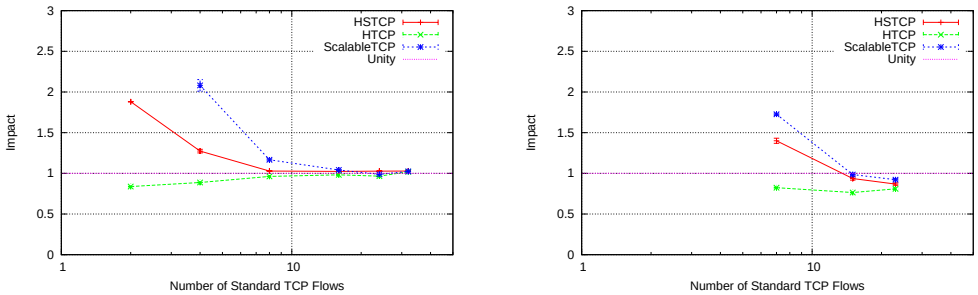
(c) H-TCP (d) Standard TCP

Figure 9.9: Fairness between various number of concurrent parallel New-TCP flows on DataTAG.

80%. ScalableTCP achieves similar goodputs to that of the other New-TCP algorithms with a small number of flows, but is able to achieve utilisation with more flows.

Figures 9.8 and 9.9 show the fairness distributions of running numerous concurrently parallel flows of each algorithm. Under the low latency network of MB-NG, it was observed that ScalableTCP appears to have poor fairness for 5 or more flows, with 33 flows demonstrating severe unfairness. However, this is also true of the other algorithms - including Standard TCP. It can be argued that the fairness performances of HSTCP, H-TCP and Standard TCP are comparable under this low latency environment.

With the longer latencies of DataTAG, the lower mean goodput of each Standard TCP flow is apparent. Out of all of the algorithms, H-TCP appears to have the best fairness, with the smallest distribution of goodputs for all



(a) 1 New-TCP Flow (b) 10 New-TCP Flow

Figure 9.10: Impact of New-TCP flows on Standard TCP Bulk Traffic on MB-NG.

number of flows. HSTCP shows similar fairness profiles to that of Standard TCP. This is expected due to the deviation towards standard AIMD parameters with smaller *cwnd* values of each flow. The unfairness experienced between Standard TCP flows is primarily due to the lack of congestion epochs between flows which keeps the flows at fixed unfairness for a longer period.

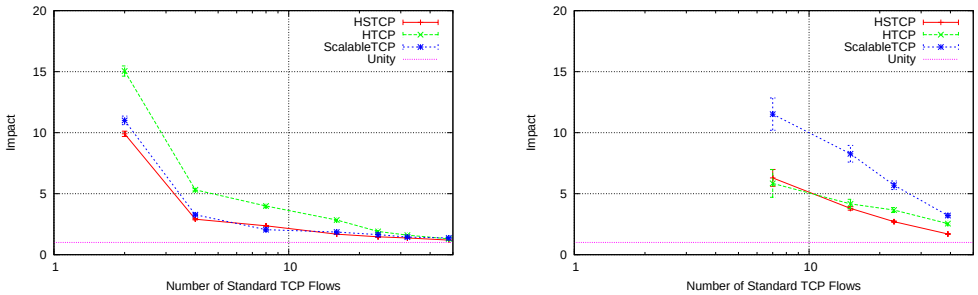
ScalableTCP has clearly the largest spread of goodputs and is therefore the least fair of the algorithms shown. It also suffers from consistently high number of outlying points suggesting that a small number of flows always achieves large values of goodput compared to the other flows.

9.1.6 Results: Impact

Figures 9.10 and 9.11 show the *impact* of running 1 and 10 New-TCP flows upon various numbers of Standard TCP bulk transport flows. As noted before, an impact greater than unity implies the degree of unfairness between the Standard TCP flows and New-TCP flows.

In both networks, impact tends to unity due to the decreased average goodput per flow due to sharing of the network capacity.

Under MB-NG, it was observed that these New-TCP algorithms are rela-



(a) 1 New-TCP Flow (b) 10 New-TCP Flow

Figure 9.11: Impact of New-TCP flows on Standard TCP Bulk Traffic on DataTAG.

tively fairer under the low latency network due to the low *cwnd* values associated with the bandwidth delay product of the MB-NG network. H-TCP, on the other hand, appears to be 'too fair' with the competing background traffic, by actually facilitating a higher goodput of the background traffic. This was found to be due to the slower *cwnd* increase of H-TCP while the congestion epoch time was small combined with the overly aggressive decrease implementation as a result of *cwnd* quantisation.

Under DataTAG, it was readily observed that the larger *cwnd* values associated with the large bandwidth delay products increased the impact of the New-TCP flows. Surprisingly, H-TCP appears to be the most aggressive under single flows. Upon closer inspection of the *cwnd* traces, it was found that this was due to the severity of the *cwnd* decreases due to SACK processing and `moderate_cwnd()` which lowered goodput of both ScalableTCP and HSTCP. H-TCP was aided by the relatively faster convergence times which facilitate higher goodput, and hence also increased impact.

With a larger number of New-TCP flows, these problems were alleviated due to smaller bandwidth delay products per flow, and clearly shows the

aggressive impact of ScalableTCP. Both HSTCP and H-TCP appear to have similar impact properties, with H-TCP being a little bit more aggressive with larger numbers of Standard TCP flows.

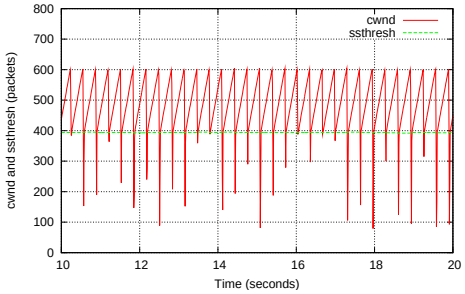
9.1.7 Discussion of Results

In order to better understand the lack of bandwidth utilisation of a single New-TCP flow when competing against CBR traffic (See Figure 9.3), Figure 9.12 shows the *cwnd* and goodput traces of the various New-TCP algorithms with 200Mbit/sec and 600Mbit/sec CBR background loads. For HSTCP, at low background loads and after congestion, the value of *cwnd* diminished below the value of *ssthresh*. It is this decrease in *cwnd* after congestion that lowers the goodput even though extra bandwidth is available.

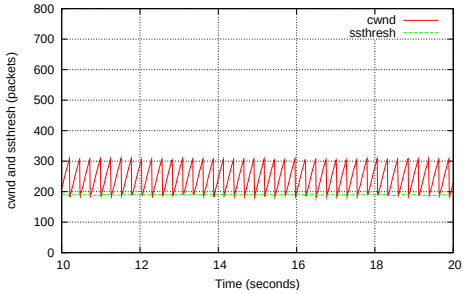
Similar results were observed for ScalableTCP. However, unlike HSTCP, there are intermittent periods whereby the *cwnd* actually drops to very low values and ScalableTCP has to grow *cwnd* all the way from very low values of *cwnd* upto the point of congestion. At lower goodputs, ScalableTCP is able to function appropriately, with a very small back-off per congestion event which results in a slightly higher goodput compared to the other algorithms.

It was observed that H-TCP's dynamic is not as severely affected under large bandwidths as that of ScalableTCP or HSTCP, which explains the slightly higher value of H-TCP under MB-NG.

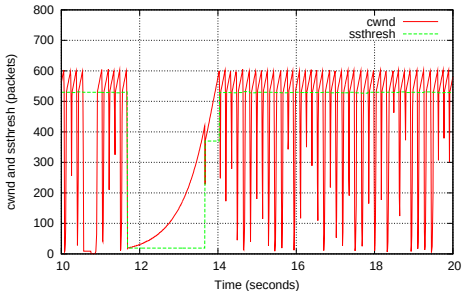
As the TCP specifications [Jac88] state that the *cwnd* should be set to the calculation of *ssthresh* upon congestion (plus 3 packets for the duplicate acknowledgments), is it surprising that the values of *ssthresh* for these tests show that there is no recalculation. This suggests that these drops in *cwnd*



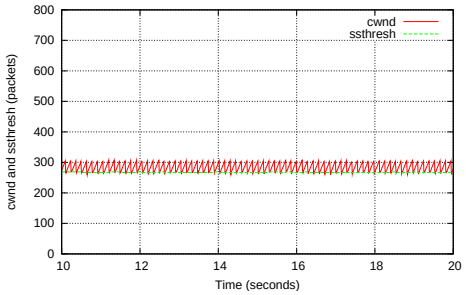
(a) HSTCP with 200Mbit/sec BG



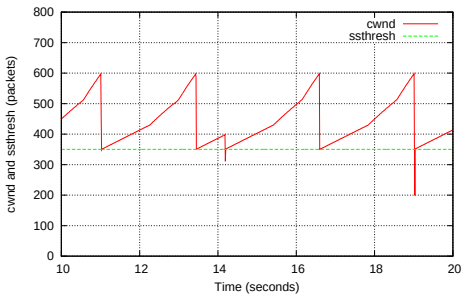
(b) HSTCP with 600Mbit/sec BG



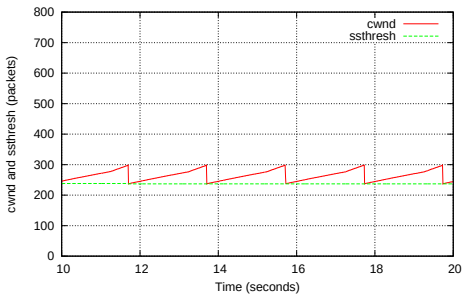
(c) ScalableTCP with 200Mbit/sec BG



(d) ScalableTCP with 600Mbit/sec BG



(e) H-TCP with 200Mbit/sec BG



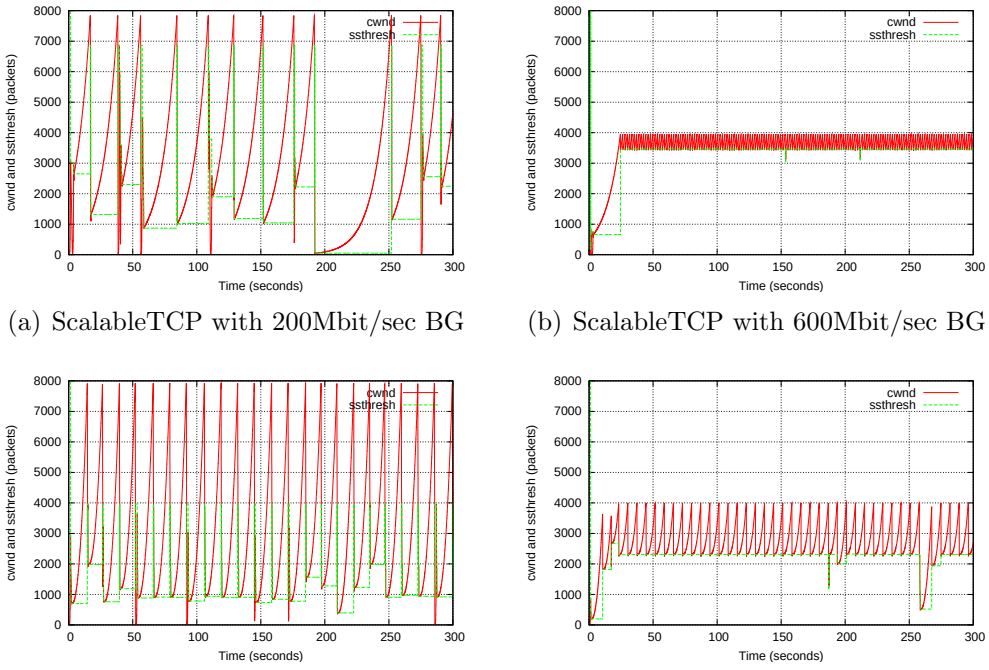
(f) H-TCP with 600Mbit/sec BG

Figure 9.12: *cwnd* trace of New-TCP algorithms with various CBR Background Traffic loads on MB-NG.

were not associated with time-outs.

This effect is more pronounced with the DataTAG results (See Figure 9.3(b)), whereby a large dip in goodput was experienced for all algorithms in the region between 100Mbit/sec and 400Mbit/sec CBR background.

Figure 9.13 shows the large variation of *cwnd* dynamics of New-TCP al-



(a) ScalableTCP with 200Mbit/sec BG

(b) ScalableTCP with 600Mbit/sec BG

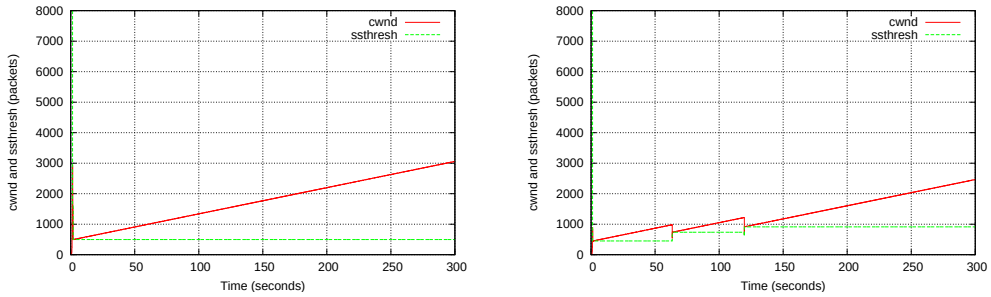
(c) H-TCP with 200Mbit/sec BG

(d) H-TCP with 600Mbit/sec BG

Figure 9.13: *cwnd* trace of New-TCP algorithms with various CBR Background Traffic Loads on DataTAG.

gorithms on the DataTAG testbed. With 600Mbit/sec background, the *cwnd* dynamic of ScalableTCP is as expected which results in high utilisation of the path (with the exception of an early packet drop which forces the flow to undergo ScalableTCP congestion avoidance in order to fill the pipe). However, with 200Mbit/sec background, a different *cwnd* dynamic was observed whereby consecutive sequential recalculations of *ssthresh* causes *cwnd* to reach small values which takes much time before *cwnd* reaches its previous maximum. Similar dynamics were observed (but not shown) with HSTCP.

As observed in Figure 9.13, the amount of time required for H-TCP to recover from these low values of *cwnd* is much faster than that of ScalableTCP - it is this responsiveness of H-TCP that allows it to achieve a greater goodput



(a) 200Mbit/sec CBR Background Traffic (b) 600Mbit/sec CBR Background Traffic

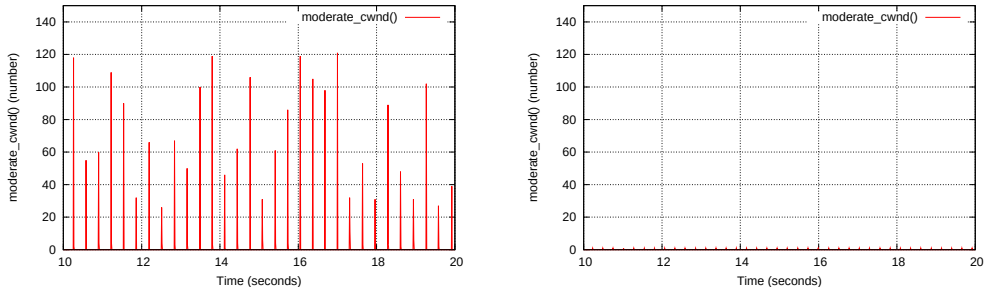
Figure 9.14: *cwnd* trace of Standard TCP with various CBR Background Traffic Loads on DataTAG.

than ScalableTCP on the DataTAG testbed.

The problems of Standard TCP along high BDP environments are demonstrated on the DataTAG link in Figure 9.14 and shows the slow growth of Standard TCP's *cwnd* which causes the low goodputs achieved in the tests. It was observed that Standard TCP achieves a goodput almost independent of the competing background traffic rate (unless there is over 700Mbit/sec background). It was found that this profile is determined primarily by the exit value of slow start when the TCP flow is initiated because of the slow *cwnd* growth in congestion avoidance.

TCP Acknowledgments

Further investigation into the Web100 [MHR03] traces showed that the drops of *cwnd* below *ssthresh* as observed on the MB-NG testbed were not caused by time-outs but rather through a call of the function `moderate_cwnd()` which causes the recalculation of *cwnd* under 'dubious' circumstances. This effect is not due to the algorithmics of TCP congestion control, but rather through the way in which the basic TCP features are implemented.



(a) 200Mbit/sec CBR Background Traffic (b) 600Mbit/sec CBR Background Traffic

Figure 9.15: Number of instance calls of moderate_cwnd() of HSTCP with various CBR Background Traffic Loads on MB-NG.

Figure 9.15 shows the correlation of these calls with HSTCP and 200Mbit/sec background traffic. When there is only 600Mbit/sec background traffic it was observed that there is a negligible number of calls of `moderate_cwnd()` which in turn results in clean *cwnd* traces. Similar results were gathered from the other algorithms.

`moderate_cwnd()`, when called, sets the current *cwnd* value to that of the number of packets in flight. Therefore, under normal operation, *cwnd* should not be affected (much) when this function is called as the process of *ack* clocking ensures that the number of packets in flight is the same as the *cwnd* value [Jac88]. However, when *ack* clocking is broken, these two values can stray such that the *cwnd* value is larger than that of the number of packets in flight.

This is most noticeable under congestion where *ack* clocking efficiency is reduced. Under such circumstances, the 'dubious' conditions whereby `moderate_cwnd()` is called, are prevalent. A dubious condition, as defined by the code of the Linux kernel, is when an *ack* is received by the sender which acknowledges more than 3 data packets. Should the TCP receiver not

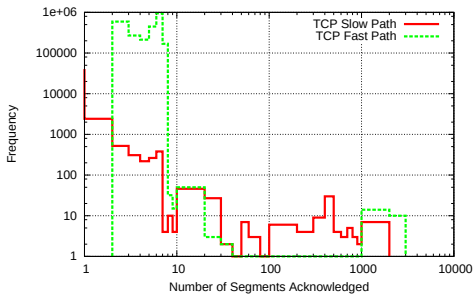


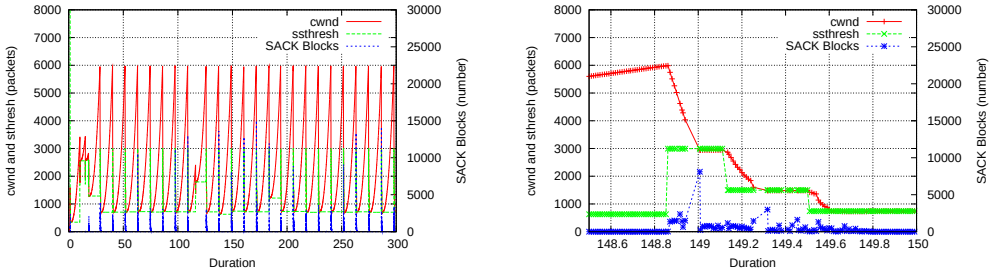
Figure 9.16: Histogram of the Number of Segments acknowledged per *ack* under a Dummynet Link of 600Mbit/sec and 40ms RTT.

obey delayed *ack*ing where an acknowledgment is generated for every other data packet, then a dubious circumstance occurs. Similarly, if the reverse path of the TCP data connection is congested, then acknowledgments may be lost which will also generate a dubious event. Similarly, if *ack* packets are reordered, a dubious event will also occur. As the testbed networks did not have any reverse traffic, the latter two possibilities are unlikely.

Figure 9.16 shows the number of segments acknowledged by each *ack*. It clearly shows that there is a high number of *ack*s that acknowledge more than just the two packets expected for delayed *ack*ing. Under such circumstances, dubious conditions will be prevalent and calls to `moderate_cwnd()` *will* occur and therefore goodput performance diminishes.

SACK Processing

The reductions of *cwnd* due to `moderate_cwnd()` do not account for the double or triple reductions in *ssthresh* upon congestion which are evident under the DataTAG tests. Should TCP Reno or NewReno be used for these tests, the multiple reductions can be explained by the multiple drops per congestion window which Reno and NewReno would require to exit from loss recovery (See Section 4.5.3). However, as all tests were run with SACKs,



(a) *cwnd* (b) SACK Blocks

Figure 9.17: *cwnd* and SACK Block trace of H-TCP with 400Mbit/sec CBR Background Traffic Loads on DataTAG.

recovery should occur completely within one RTT.

However, the use of SACKs also imposes an extra processing burden for the TCP sender. This is shown in Figure 9.17[2] which shows a Web100 trace of H-TCP on DataTAG with 400Mbit/sec background CBR traffic. The zoomed plot on the right shows a magnified section of the same experiment. Each measurement point for each variable is indicated with a marker. It is apparent that shortly after the initial congestion notification that there are periods where no measurements were taken. This is demonstrated with the lack of measurement markers and the large increases of the number of SACK Blocks as soon as measurement continues.

This can be explained with the fact that Web100 is composed of two parts; a kernel part which actually increments all the relevant variables, such as the number of SACK Blocks; and a user-land library which polls the `/proc` structure to retrieve the TCP measurements. Therefore, the periods of silence shown in Figure 9.17 are due to the lack of servicing by the system of the user-land application. Meanwhile, the counters are still incremented

[2]Graph courtesy of Baruch Evans of Hamilton Institute.

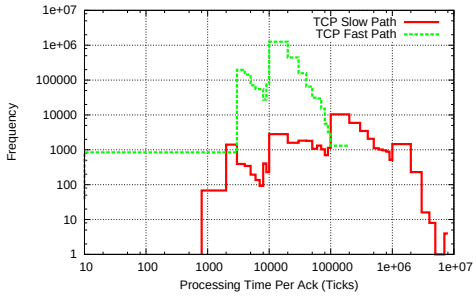


Figure 9.18: Histogram of SACK Processing Time in processor ticks of Standard TCP under Dummynet with 600Mbit/sec link capacity and 40ms RTT.

as they work in kernel space.

The number of packets in flight is roughly equal to the bandwidth delay product (See Section 4.1). Therefore, the TCP sender, assuming perfect delayed *ack*ing, will need to process half of the number of packets in flight; with each *ack* containing a maximum of 3 SACK Blocks. As the processing of each SACK Block results in a linear traversal of the unacknowledged packets in transmission buffers (and subsequent updates to the TCP Control Block) such that data segments that need to be retransmitted can be identified, there is a huge processing burden with large bandwidth delay products. Therefore, under large fluxes of SACK Blocks, the system processor is completely saturated by the need to process SACK information.

The processing time required to process both normal (TCP Fast Path) and duplicate (TCP Slow Path) acknowledgments are shown in Figure 9.18. It clearly demonstrates that even though there is a higher number of normal *ack*s, duplicate *ack*s can take up-to 10 times longer to processes.

9.1.8 Summary

The main conclusion from these sets of experiments is that there are implementation problems associated with the deployment of New-TCP algorithms. The problem stems not directly from the dynamics of the *cwnd* update algorithms, but with the actual TCP protocol when used at high speeds.

For typical trans-Atlantic link such as DataTAG, as much as 10,000 packets worth of segments must be kept in memory to facilitate TCP's sliding window. Under delayed acknowledgments, this could result in flux of 5,000 packets per RTT.

The processing burden of having to scoreboard all these *ack*s is increased with SACK information that is supposed to aid faster recovery upon congestion detection under multiple packet losses per window. It was demonstrated that a high flux of SACK Blocks can cause processing lock-out, and therefore a stall in the TCP *ack* clocking, and more importantly, a serious decrease in the goodput performance of the TCP stack.

Another concern is that Linux TCP receivers do not strictly adhere to delayed *ack*ing, and as result may generate many 'stretched *ack*'s' which also break the *ack* clock and may skew the calculation of RTT/RTO and *cwnd* to the number of packets in flight.

The result of this is under these 'dubious' circumstances, the TCP sender will recalculate the *cwnd* to be the number of packets in flight (as per [Pos81b, Jac88]) which will result in deviations from the theoretical *cwnd* dynamics and hence reduced goodput performance.

The interaction of many New-TCP flows were also compared and contrasted. Due to the reduced bandwidth delay product of having many flows, the effects of the above mentioned implementations are often mitigated.

It can be argued that ScalableTCP, especially under long latency environments experiences unfairness between competing flows. This was often observed with either a large distribution of per flow goodputs and or frequent outlying points which show that a small number of flows often achieved much higher goodput than that of the other flows.

9.2 Preferential Flow Handling Using DiffServ

The recent decrease in price to link capacity ratio has meant that it is much more cost effective to upgrade the capacity of the network rather than to engineer a lower speed one. This acceleration is much faster than the observed bandwidth usage from traditional Best Effort (BE) traffic. Thus in the short to medium-term there is excess capacity available, especially in the core.

However, as shown in Section 9.1, Standard TCP is incapable of utilising such spare capacity due to slow growth of its congestion windows and the requirement for very low loss rates.

Another problem is that due to the implementation issues of the Linux kernel and the processing overheads of SACKs, transport of New-TCP algorithms in Best Effort (BE) environments at high speeds is difficult.

One way in which it is possible to utilise this spare capacity is to implement some form of mechanism in the network whereby network users can be differentiated to offer better (or worse) service, depending on a number of factors; such as their price plan, types of traffic or network usage.

The different service qualities can be addressed by a variety of mechanisms. However, the actual deployment of those mechanisms in the current

structure of the Internet is rare. Network layer mechanisms such as ATM assume a homogeneous infrastructure between two end-systems, which for economic reasons may not exist. The Integrated Services (IS) Architecture [BCS94] does not depend on a homogeneous infrastructure but often requires that the end-systems have specialised software installed and that every network device is capable of per flow handling of traffic - which has serious implications upon scalability and manageability.

Differentiated services (DiffServ) [BBC+98] defines a new packet-handling framework that addresses the concerns of scalability. The DiffServ architecture defines ways that a packet should be treated by interior routers based on associated classes called per-hop-behaviour (PHB) [HFB+99, JNP99]. It requires that edge routers are capable of identifying flows and marking them with this appropriate class of service whereby core routers can provide the related PHB.

However, in contrast to previous Quality of Service (QoS) architectures such as RSVP [BCS94], DiffServ focuses on the behaviour of aggregates rather than individual flows. Packets are identified by simple markings that indicate which aggregate behaviour they should be assigned to. By implementing the complex task of identifying flows at the edges of the network, core routers need only provide the service for each aggregate based on these markings.

Packets are marked either by applications or by edge routers. However, marking by applications will often require policing by the edge routers to enforce that the correct 'codepoints' are marked on the packets. If edge routers mark packets, they may choose to do so on a per-flow basis or on any other criteria. These definitions should be agreed between different providers through Service Level Agreements (SLAs).

The DiffServ architecture offers new opportunities for building preferential macro-flow handling for end-to-end networks. However, it also introduces new challenges. [FKSS98, YR99b, YR99a] show that it is difficult to guarantee requested throughputs for TCP flows, because of the flow and congestion control mechanisms used by TCP, which result, for example, in bursty traffic.

Two recent developments can significantly contribute to tackle the problems of high capacity service differentiation: Proposals for high throughput TCP algorithms and the availability of DiffServ enabled networks at gigabit speeds.

9.2.1 Methodology

Sections 9.1.3 and 9.1.6 and suggests that a mechanism for segregating traffic sharing the same packet-switched path would be beneficial. The reasons for this are two fold:

- To protect traditional traffic from the possible aggression of the New-TCP algorithms and hence be more 'fair' to existing Internet users.
- To provide a guarantee on the level of end-to-end service predictability for the New-TCP proposals.

As such, DiffServ provides a facility to offer protection of the background traffic whilst maintaining a predefined pipe whereby the high-speed New-TCP flow can fully utilise the link.

Two PHB classes were configured: BE for traditional Best Effort [CF98] traffic and Assured Forwarding (AF) [HFB+99] to provide network resources for the New-TCP flows. Traffic was marked with an appropriate Diffserv codepoint (DSCP) [BBC+98] at the sending host and each class is put in

a physically different router output queue. Each queue is then assigned a minimum guaranteed bandwidth using weighted round robin.

Background CBR UDP traffic was injected into the network using `iperf` [TQD+03] and marked to be treated as Best Effort traffic. A subsequent time after, a single flow of New-TCP traffic is started using `iperf` and marked to the AF class to give higher priority over the BE background traffic.

Various allocations percentages of the AF and BE classes were investigated and the *cwnd* dynamic and goodput performances were measured.

Metrics

As DiffServ is only active when there is congestion on the network such that the traffic breaches the defined DiffServ handling, it is important to measure how effectively the bandwidth allocations work. This is especially important for TCP traffic as the congestion avoidance algorithms of TCP cause a lower utilisation of available resources.

This utilisation, stability and class protection of both the background BE traffic and the AF TCP flow can be quantified through the computation of two parameters which are defined as the BE QoS efficiency factor (QEF_{BE}) and the AF QoS efficiency factor (QEF_{AF}) for the BE and AF class, respectively.

$$QEF_{BE} = \frac{B_{UDP}}{B_{BE}} \tag{9.2}$$

$$QEF_{AF} = \frac{B_{TCP}}{B_{AF}} \tag{9.3}$$

Where, B_{UDP} and B_{TCP} are the achieved (measured) goodputs of the UDP and TCP flows respectively with allocated bandwidth allocations of B_{BE} and B_{AF} for the corresponding BE and AF classes. Therefore, QEFs

give an indication of the bandwidth utilisation of the BE and AF classes.

Test Set-up

The experiments were performed on the DataTAG testbed (See Appendix C.3) and DiffServ was enabled on the Juniper M10. This testbed is unique in providing a DiffServ network with high propagation delay with Gigabit per second bandwidth capacity.

The testbed PCs of DataTAG as shown in Table B.2 are used and results are obtained for a TCP connections with congestion moderation turned on - i.e. the code of `moderate_cwnd()` are left as per the Linux defaults. The experiments were conducted by continuously running a 1Gb/sec UDP BE flow running and by switching on and off different New-TCP algorithms for different bandwidth allocation tuples.

The UDP flow was injected into the network using `iperf` and was set to inject packets into the network at line rate, i.e. 1Gb/sec. As such, and because the background BE UDP traffic does not back off in the event of congestion, the BE class is always oversubscribed. Therefore, whenever AF under utilises its allocated bandwidth, (i.e. $QEF_{AF} < 1$) the BE throughput should show a proportional gain ($BE > 100\%$).

Calibration

The bandwidth scheduling and DiffServ systems of the Juniper router were initially tested with UDP CBR in both the AF and BE classes in order ensure correct calibration of the system. Figure 9.19 shows the achieved bandwidth allocations of the BE and AF flows as the router configuration was changed on-the-fly between the bandwidth allocations of 90:10, 70:30, 50:50, 30:70

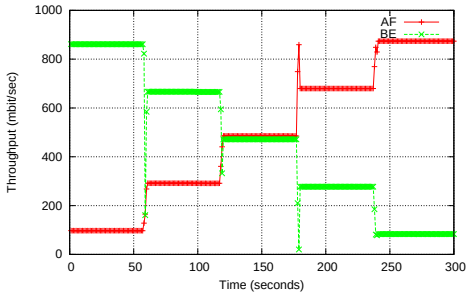


Figure 9.19: Dynamic switching of BE:AF bandwidth allocations every 60 seconds from 90:10, 70:30, 50:50, 30:70 and 10:90.

and 10:90 between the AF class and BE classes.

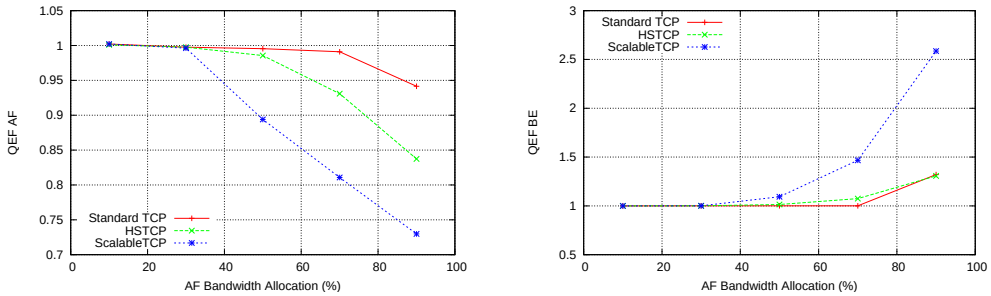
The results show that the bandwidth scheduler works well as the allocations are changed approximately every minute. The peaks and drops of throughput at each change are a result of the dynamic switching whereby the bandwidth scheduler is over providing the AF flow.

9.2.2 Results

Figure 9.20 shows the measurement of QEF for the different allocations of AF:BE ratios. Compared to the results of Figure 9.3, where in the absence of differentiated traffic, Standard TCP performs very poorly, the improvement in goodput of single TCP flows when the TCP flow is protected is dramatic - utilising at least 95% of the pipe allocation (so at 90% AF Standard TCP is able to achieve approximately 85.5% of the pipe).

Surprisingly, however, is that the other algorithms perform relatively badly with large AF allocations with ScalableTCP showing the lowest utilisation of its AF pipe. At the lowest AF allocations, all algorithms perform identically.

The dynamic bandwidth allocations between the AF and BE classes is



(a) TCP (AF) (b) UDP (BE)

Figure 9.20: QoS Efficiency Factors for various AF:BE allocations.

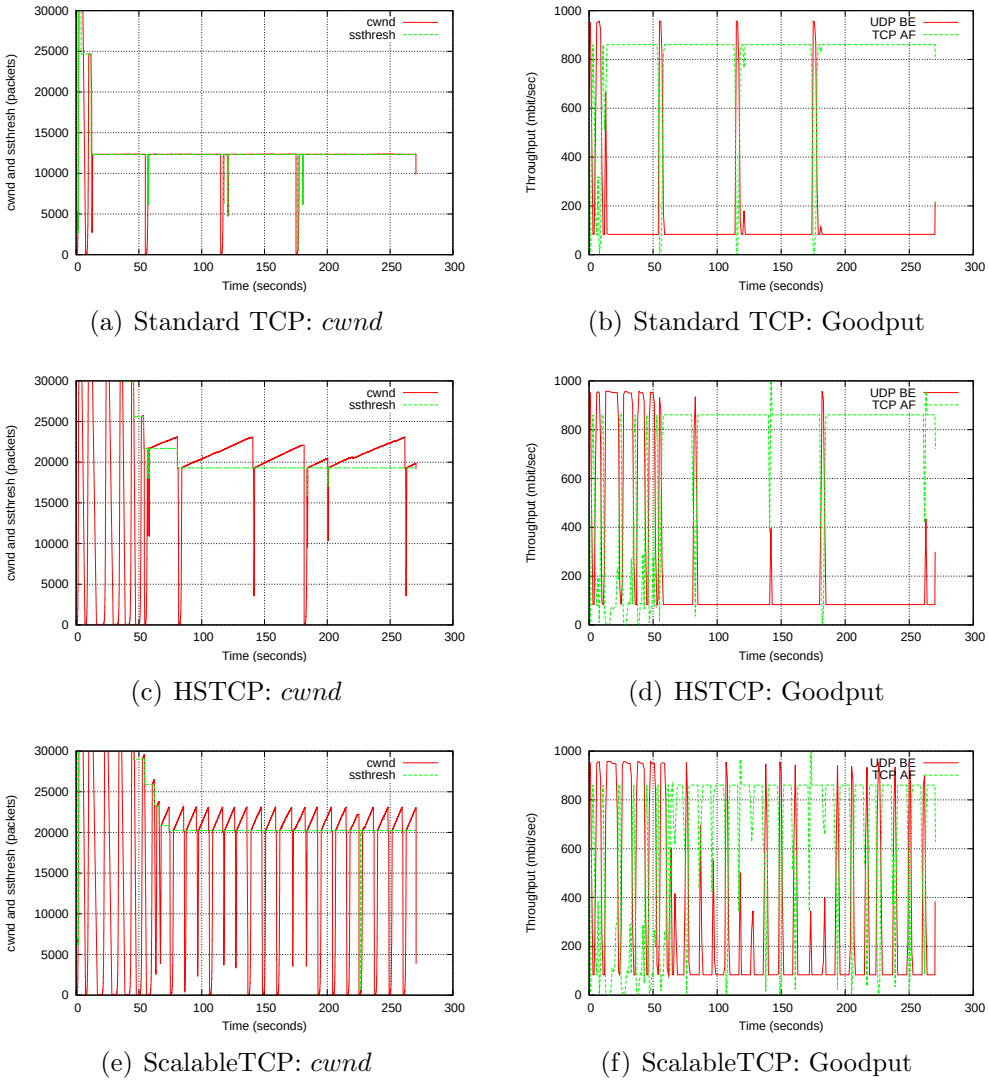
demonstrated by the over utilisation of the BE class's traffic when the AF class is under-provisioned due to the failure of the New-TCP flows in being able to achieve 100% utilisation.

9.2.3 Discussion

Given the dynamics of *cwnd*, it would be expected that the performance of Standard TCP should be poor under high bandwidth, long latency environments. However, Figure 9.20 clearly indicates that HSTCP and ScalableTCP appear to have problems under DiffServ environments - not Standard TCP. As a DiffServ enabled network should provide protection for AF flows, the goodput performance of New-TCP flows were expected to be greater.

At small AF allocations, i.e. small available bandwidths, it is expected that the performance differences between algorithms are negligible due to the sizes of *cwnd*. However, as the bandwidth, and hence *cwnd* increases due to a larger AF provision, one would expect HSTCP and ScalableTCP to be able to utilise the extra capacity more effectively.

Figure 9.21 shows the *cwnd* and goodput trace of Standard TCP with 90% AF, 10% BE allocation. Apart from an initial burst of traffic which



(a) Standard TCP: *cwnd*

(b) Standard TCP: Goodput

(c) HSTCP: *cwnd*

(d) HSTCP: Goodput

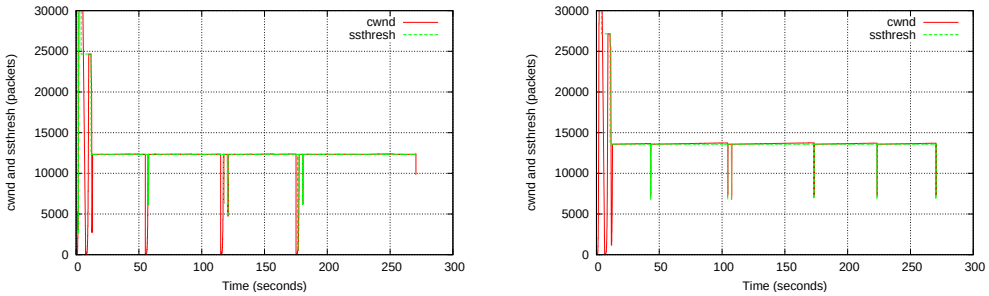
(e) ScalableTCP: *cwnd*

(f) ScalableTCP: Goodput

Figure 9.21: New-TCP *cwnd* and throughput dynamic with 90% AF allocation.

is captured by the router, it can be seen that the *cwnd* and hence goodput of the Standard TCP flow is relatively static in value as the test progresses. This is due to the very small growth rate of Standard TCP which is incapable of inducing packet loss at the bottleneck queue.

The differences in *cwnd* dynamic between Standard TCP, HSTCP and



(a) With Congestion Moderation (b) Without Congestion Moderation

Figure 9.22: Effect of Congestion Moderation on Standard TCP with 90% AF allocation.

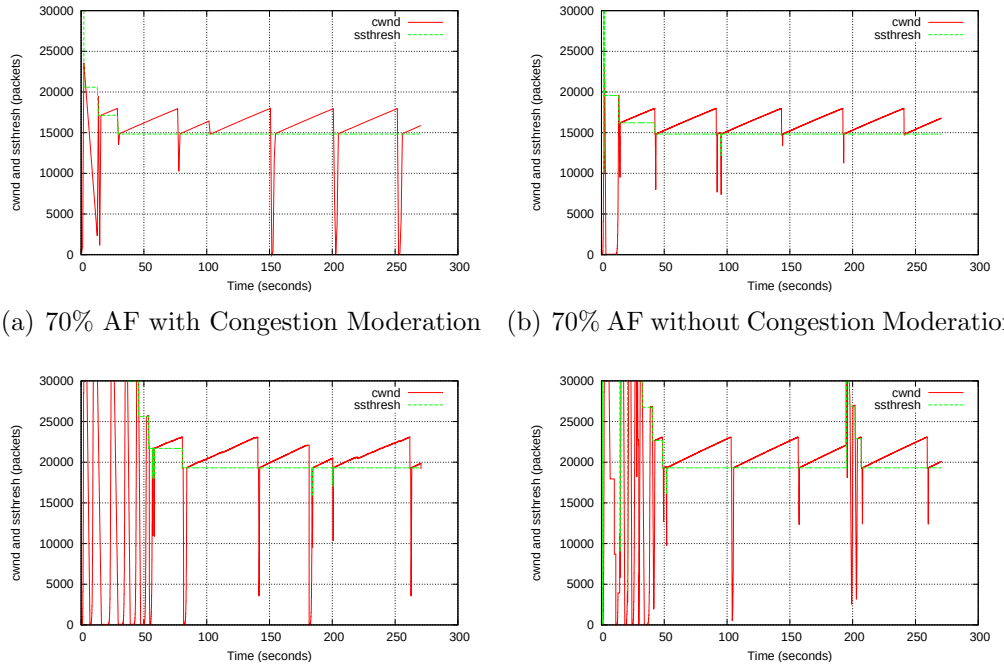
Scalable TCP is shown in Figure 9.21. It can be argued that the main difference between the three results was the high frequency with which ScalableTCP appears to re-enter its slow start-like phase, with HSTCP only occasionally also doing so.

Similar to the previous single flow tests without DiffServ (See Figures 9.1), the *cwnd* histories showed that these drops in *cwnd* value were *not* associated with slow-starts, but with a large number of function calls of `moderate_cwnd()`.

9.2.4 Congestion Moderation

An investigation into the performance of New-TCP algorithms with and without the facilitation of *congestion moderation* via the Linux `moderate_cwnd()` function was conducted. Note this is would be in violation of the TCP protocol specifications [Pos81b, Jac88], but would provide a simpler fix than rectifying the problems of Linux receivers imposing stretched *ack*s.

With Standard TCP, the difference with and without congestion moderation is shown in Figure 9.22. Standard TCP with congestion moderation turned off allows the *cwnd* to grow without the frequent reductions in *cwnd*



(a) 70% AF with Congestion Moderation (b) 70% AF without Congestion Moderation

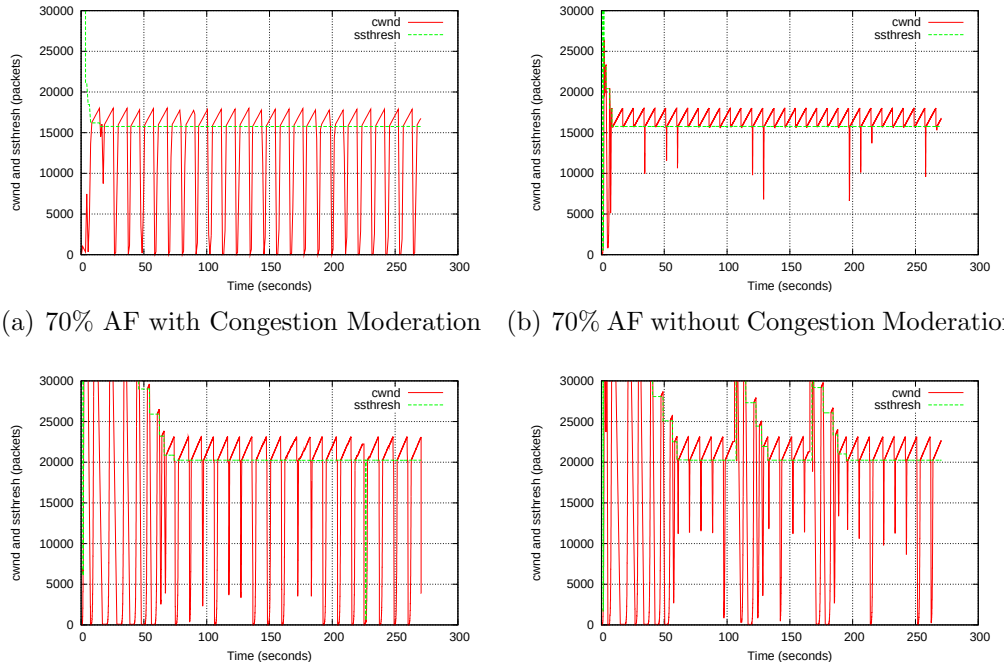
(c) 90% AF with Congestion Moderation (d) 90% AF without Congestion Moderation

Figure 9.23: Effect of Congestion Moderation on HSTCP with 70% and 90% AF allocation.

to very low relative values.

Even without congestion moderation, it was also observed that there were a few drops in *cwnd* and goodput, which were briefly raised (not through slow start nor congestion avoidance) to their previous values before the drop. The cause of this action was found to be due to Linux undo's [SK02] during which the *ssthresh* and *cwnd* values are reduced due to an indication of congestion. However, shortly afterwards, DSACK (See Section 4.5.5) and/or *ack* information infers that the congestion event was in fact not necessary and therefore the TCP *cwnd* and *ssthresh* values are readjusted to the previously recorded values.

Without congestion moderation, HSTCP shows an appreciable difference



(a) 70% AF with Congestion Moderation (b) 70% AF without Congestion Moderation

(c) 90% AF with Congestion Moderation (d) 90% AF without Congestion Moderation

Figure 9.24: Effect of Congestion Moderation on ScalableTCP with 70% and 90% AF allocation.

in the *cwnd* dynamic for the 70% and 90% AF bandwidth allocations as the probability of dubious events that cause `moderate_cwnd()` function calls with is higher due to the increased bandwidth delay product with larger AF allocations. The absence of congestion moderation appears to be more beneficial for the 700Mbit/sec AF bandwidth allocation, whereas the problem is only partially solved for a 900Mbit/sec AF allocation.

Figure 9.24 shows the *cwnd* dynamics of ScalableTCP with and without congestion moderation at 70% and 90% AF bandwidth allocation. Similar to the results observed for HSTCP, the improvement gained without congestion moderation under 70% AF allocation is dramatic based on reduction of the large decreases in *cwnd* after congestion detection. However, the results

under 90% AF allocation still show serious problems with the *cwnd* dynamic which consequently results in reduced goodput performance.

The main difference between ScalableTCP and HS-TCP flows are that the occurrence of the slow start like events for HS-TCP are much lower, making the effects of the congestion moderation much less noticeable. As such, the HS-TCP *cwnd* dynamics are much cleaner and nearer to the theoretical traces than those of Scalable TCP. However, having congestion moderation turned off has less overall effect on the *cwnd* dynamics when AF is allocated 900Mbit/sec - regardless of the TCP algorithm used.

The inference is that even without the implications of congestion moderation, SACK processing is still a dominant factor which causes *cwnd* dynamic deviations from theory that result in the observed decreased goodput performance. When the AF allocation is 700Mbit/sec, it was observed that the dominant factor *is* congestion moderation - as the removal of the function call completely resolves the problem of deviations of the *cwnd* dynamic from theory. But when the AF allocation is 900Mbit/sec, the flux of SACKs is so high that removing the congestion moderation code only partially resolves the problems of *cwnd* dynamics entering low values.

9.2.5 Active Queue Management

In this section, the negative effects of SACK processing are mitigated using an active queue management solution. The objective was to lower the rate of SACK events after multiple drops and therefore allow the sending host to process the SACK Blocks appropriately.

Neither Standard TCP nor HSTCP experienced as many slow-starts as ScalableTCP. Therefore, an AQM solution was implemented for ScalableTCP

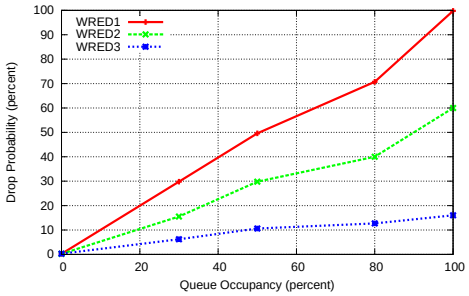


Figure 9.25: WRED profiles.

only and for the most challenging AF allocation of 900Mbit/sec.

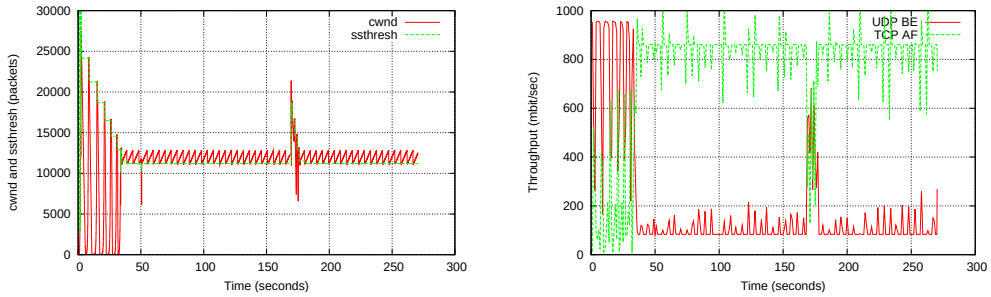
The aim of these tests is to smooth out the observed *cwnd* deviations from theory by means of configuring different Weighted Random Early Detection (WRED) [FJ93] drop profiles for the AF class.

Methodology

As the New-TCP algorithms under investigation in this section are loss based protocols, the *cwnd* dynamic of each necessarily require the oscillation of *cwnd* below the point of congestion. Therefore, it is important not to impose too great of a random loss upon the AF class as this would correlate to high(er) loss rates for the single ScalableTCP flow and thus a decrease in goodput (See Section 5.4.2). As such, a gentle WRED drop profile was used.

From a packet-level perspective, the justification for using WRED is based on the belief that a smoother distribution of the loss-pattern in the AF queue will help lower the burst length of SACKs and therefore it will help avoid deep stalls of *cwnd* in the TCP sending host.

Using the same network topology and hardware and software configurations as used previously, three WRED drop profiles are configured on the Juniper M10 router as shown in Figure 9.25.



(a) CWND (b) Throughput

Figure 9.26: Effect of a gentle WRED upon ScalableTCP with 90% AF allocation.

Congestion moderation was kept disabled during the test to simplify the interpretation of the SACK processing results.

Results

The improvement of *cwnd* dynamic when WRED2 is employed is dramatic as shown in Figure 9.26 which demonstrates that the *cwnd* dynamic of ScalableTCP no longer exhibits the large drop of *cwnd* values that result in low goodput performance as seen without WRED.

This result validates that gently redistributing the loss pattern in order to avoid the large flux of SACK Blocks is beneficial to the goodput performance of TCP flow at high speeds. This was necessitated from the processing burden of large bursts of SACKs which cripple the sending host.

9.2.6 Summary

This section has shown that the performance penalties due to software and hardware limitations on the transport of TCP data across large bandwidth delay networks can be alleviated with both Linux TCP stack modifications and hardware intervention from network nodes.

Also, with the successful deployment of these faster New-TCP algorithms, the interaction with Standard TCP will cause unfairness that may cripple the bulk transport of data using Standard TCP.

As such, network traffic separation and protection via DiffServ was deployed to guarantee the protection of traditional BE traffic against the possible aggression of these New-TCP algorithms. DiffServ would also guarantee the full utilisation of the pipe reserved and, as a consequence, of the whole link. Moreover, it guarantees a certain level of service predictability for network transfers.

Surprisingly, Standard TCP not only performed well under the DiffServ environment but also shows the best performance, followed by HSTCP and then, at some distance, ScalableTCP. Standard TCP was able to maintain high bandwidth utilisation due to the slow growth of *cwnd* whereby the Standard TCP did not induce self loss, and also due to the lack of packet loss as a result of low bit error rates encountered on the testbed.

However, for longer transfers where there are higher probabilities of having Standard TCP packets lost due to the physical limit of bit error rates, the halving of *cwnd* under AIMD would also result in the halving of goodput. As shown previously in Section 5.4.1, the rate at which Standard TCP probes for extra bandwidth is also slow and therefore would require much time in order to fully occupy the AF pipe (assuming no further packet losses).

Therefore, there is still a requirement for the use of high speed TCP algorithms under DiffServ enabled long distance, high capacity networks. HSTCP and ScalableTCP attempt to solve the problem of low goodput by growing *cwnd* more aggressively that Standard TCP; this self induction of loss instead leads to worse performance than Standard TCP. This self-induced loss invariably requires the processing of a large flux of SACK Blocks which

can cause stalls in the data transfer which lowers the achievable goodput.

Due to Linux specific implementation TCP features that were designed to lower the processing overhead on the receiver, many 'dubious events' were observed which causes the TCP sender to recalculate *cwnd* to that of the number of packets in flight which caused a reduction in the observed goodput.

In order to reduce the processing burden of SACKs, an AQM solution (WRED) was implemented. Specifically, a drop probability in the router AF queue with a gentle gradient was proven to be extremely effective in maintaining high goodput for ScalableTCP. This validates the belief that the SACK processing is a dominant reason for having poor goodput performance.

This AQM solution can relax the requirement of any further optimisation of the SACK code. However, it is believed that the main limitations in using aggressive TCP protocols under high BDP IP DiffServ-enabled paths lies in not having enough CPU speed to cope with both the sheer number of packets both coming in and out and extra burden of multiple packet drops in the same *cwnd* that they induce.

9.3 Internet Transfers

The transport of New-TCP protocols across real-life wide area networks is investigated in this section. The goal is to to determine the real-world performance of the various New-TCP proposals using single flow transfers across Transatlantic and pan-European academic networks.

9.3.1 Test Methodology

The variable loads and unpredictable performance of Internet traffic makes obtaining consistent results across the Internet difficult. A long duration single transfer test of each New-TCP algorithm to each host would yield insufficient comparison between measurements as the network load may have changed sufficiently such that it is a measurement of the variability of the network, rather than that of the performance improvements of the New-TCP algorithm.

However, running many short tests may not provide sufficient time for the TCP flows to reach a steady state whereby the flow spends enough time in congestion avoidance instead of slow start.

A simple solution would be to run the tests for a fixed number of bytes. However, the variability of Internet performance may result in tests having to run for a very long time due poor network goodput.

By imposing completely random start times and random sequences in which the the various locations and New-TCP algorithms were run, unfair comparison between algorithms due to the time based patterns of network load is minimised. Therefore, many 5 minute TCP transfers were conducted over the Internet from CERN in Switzerland to various locations in the United States of America and Europe. The Internet link from CERN to the Internet consisted of a 1Gb/sec connection. The hardware and software specification of the test machine is defined in Table B.3.

All tests were run with the Linux 2.6.6 altAIMD-0.6 kernel using `iperf` to conduct the TCP bulk transfers.

Table 9.1 shows the number of individual measurements that where taken for each destination host and for each New-TCP algorithm. The tests were

	Stanford	LBL	Dublin	Rutherford
Standard TCP	220	28	145	168
HSTCP	246	75	151	208
ScalableTCP	249	173	149	172
H-TCP	248	206	143	234
FAST	249	221	65	238
BicTCP	247	135	152	240

Table 9.1: Number of individual New-TCP measurements to each site.

run over the period of 2 months between October and December of 2004 and were ran over a 24 hour period, 7 days a week.

All traffic was either routed through Dante's GEANT network if the destination was European, or through the DataTAG network via its production link over to the U.S. whereby it was locally routed onto Internet2's Abilene network. All network paths are shown in Appendix D.

There are problems associated with the way that Linux accounts for the number of packets in flight whereby there is no artificial inflation of *cwnd* during loss recovery. The result of this is that upon a TCP time-out (when *cwnd* is set to 1 segment) it will limit the number of packets that can be retransmitted each RTT. Because there is no artificial inflation of *cwnd*, loss recovery occurs at a rate of 1 packet per RTT after a TCP time-out. Therefore the loss recovery time is proportional to the number of packets lost. For large *cwnd* values where many packets can be lost, this recovery time can be very long for long distance networks. This is documented in [HLW+04].

This effect was especially noticeable during the initial slow-start period of tests and caused many occurrences whereby the TCP flow spent nearly all of the test time in loss recovery. As such, the initial slow start was disabled such

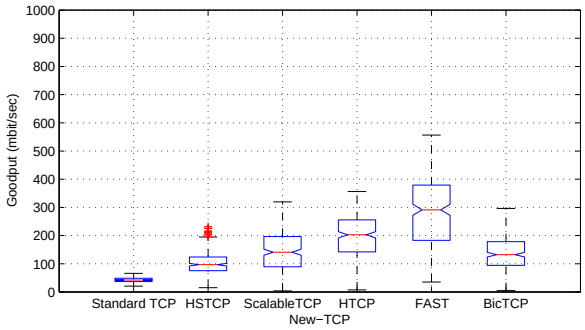


Figure 9.27: Goodput from CERN to Stanford.

that results could be gathered. This was achieved by setting the initial value of *ssthresh* upon a TCP connection to 2 segments rather than `max(INT)`. The effect of disabling slow-start would be similar to the effects of large bursts of transient cross traffic which would similarly cause slow-start to exit early.

9.3.2 Results: CERN to Stanford

Goodput

Figure 9.27 shows the goodput performance distributions of the various New-TCP algorithms from CERN to Stanford, California, U.S.. It clearly shows the consistent inability of Standard TCP to achieve reasonable goodput over such long distance paths.

For the other New-TCP algorithms, the results show a large range of measured goodput performances with symmetric distribution of goodputs. Most notably all algorithms have a minimum which is near zero, suggesting that that the network load can be high, thus preventing high performance network transfers.

Whilst it is clear that FAST is able to achieve the highest median goodput performance, it also has the widest distribution. The high maximum also

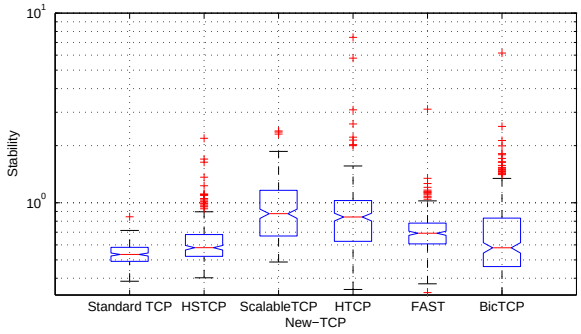


Figure 9.28: Stability from CERN to Stanford.

suggests that FAST is very capable of reaching higher speeds.

Out of all the New-TCP algorithms, HSTCP has the lowest median goodput, and generally was unable to achieve goodputs much greater than 200Mbit/sec, with typical results around 100Mbit/sec (which is still about twice that of Standard TCP).

ScalableTCP and BicTCP, although they performed very well in artificial tests, do not achieve as high goodputs across this real-life environment. They both provide a similar distribution, with ScalableTCP having a slightly higher deviation towards higher goodputs.

H-TCP exhibited goodput performance higher than ScalableTCP, but not much more in terms of absolute maximum range.

Stability

Figure 9.28 shows the stability (See Section 7.1.2) distributions of the various New-TCP algorithms to Stanford for the goodput performance results shown in Figure 9.27.

The large number of outlier points suggests that all New-TCP algorithms have problems maintaining a low variance and high goodput. Most notably,

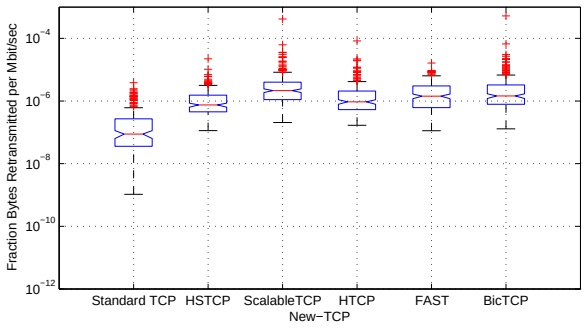


Figure 9.29: Overhead from CERN to Stanford.

H-TCP and BicTCP have a high number of outliers which cause very unstable results. However, BicTCP has a lower median and inter-quartile range, suggesting generally better stability when compared to H-TCP.

Whilst ScalableTCP does not have the large number of outliers, it has the largest inter-quartile range of results. FAST also maintains a large range of values, but has a very narrow inter-quartile range; making it the most consistent TCP protocol, bar Standard TCP.

Standard TCP demonstrates low coefficient of variance due to the slow growth of *cwnd* which over this long latency path is unable to frequently induce packet loss and hence multiplicative decrease.

Overhead

The overhead inefficiency is shown in Figure 9.29. It shows that all New-TCP algorithms perform similarly (although note the log scale), with approximately a magnitude greater overhead compared to Standard TCP. It is arguable that ScalableTCP has the highest overhead. However, BicTCP also maintains many outlying points, with as much range as that of ScalableTCP.

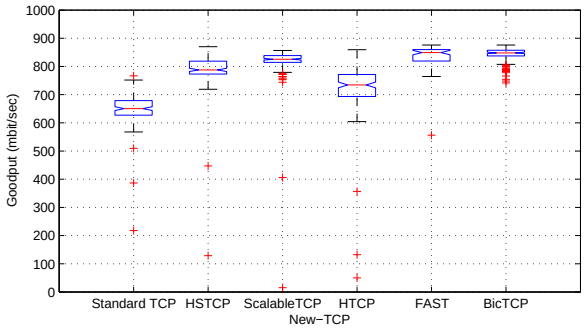


Figure 9.30: Goodput from CERN to Dublin.

9.3.3 Results: CERN to Dublin

Goodput

Figure 9.30 shows the transfer results from CERN to Dublin. All algorithms, including Standard TCP, are able to transfer at high rates due to the relatively low latencies of the network path.

Unlike previous results with Stanford which show a few outliers above the median goodput, the Dublin link shows that there are a few occasions where the measured goodput of the New-TCP algorithms were lower. Whist the number of these points are small, they suggest rare instances of competing traffic on this network path which reduces the goodput performance.

Out of the protocols, BicTCP appear to have the highest goodput, with also the smallest inter-quartile range. However, FAST maintains similar median goodput values to that of BicTCP, but shows a larger inter-quartile and normal range of values. ScalableTCP maintains a very narrow goodput range, however, maintains a slightly lower goodput when compared to FAST and BicTCP.

HSTCP, even with its gently aggressive deviation from standard AIMD at low *cwnd* values, demonstrates higher goodput performance with larger

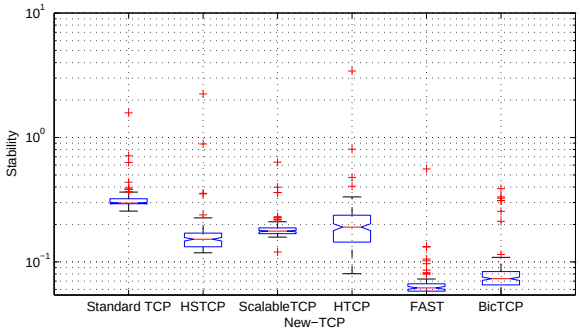


Figure 9.31: Stability from CERN to Dublin.

cwnd values. It also shows a similar range of values to that of FAST - but with an even lower median value.

Unlike the test to Stanford, H-TCP has the worse goodput of all of the New-TCP algorithms, with both a low median value and a large range of goodput measurements. However, the measured quartiles of H-TCP's results suggest that it has better goodput performance than Standard TCP.

Stability

The stability of the New-TCP algorithms is shown in Figure 9.31. As demonstrated with the goodput results, even with these relatively low latencies, the *cwnd* values are sufficiently large enough to demonstrate the increased stabilities of these New-TCP algorithms with their smaller decreases in *cwnd* upon loss detection.

The results show that FAST has the lowest coefficient of variance per measurement, followed by BicTCP. HSTCP, ScalableTCP and H-TCP perform similarly, with H-TCP being the most variable with the largest inter-quartile and normal range.

As FAST utilises the standard multiplicative decrease of 50% upon loss,

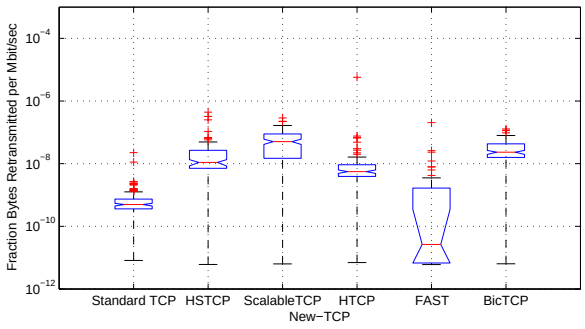


Figure 9.32: Overhead from CERN to Dublin.

the increased stability of FAST suggest that the delayed based congestion control algorithm of FAST is functioning well. Similarly, the large coefficient of variation of Standard TCP is consistent with its large decrease values of *cwnd* upon loss.

Overhead

FAST has the clear advantage in terms of lowering the overhead as shown in Figure 9.32. This correlates well with previous results suggesting that the inability of FAST to stabilise will result in large overheads.

Even though FAST has the largest inter-quartile range of overhead of all the New-TCP algorithms, it is consistently below that of the other New-TCP algorithms which show very similar results.

9.3.4 Results: CERN to LBL

Goodput

Figure 9.33 shows the goodput results to LBL. The performance boost by the New-TCP protocols is clearly visible, with FAST having the highest median goodput - with about a 10 times improvement over that of Standard

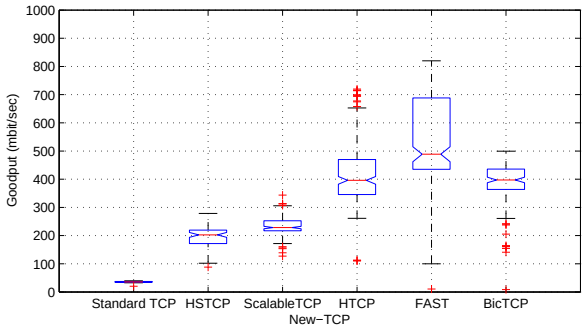
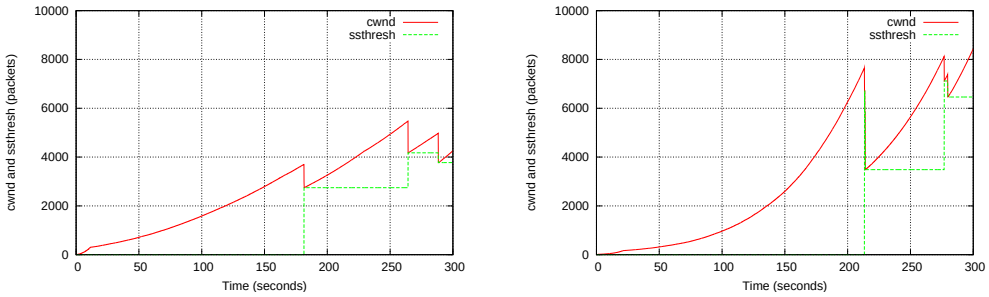


Figure 9.33: Goodput from CERN to LBL.

TCP. FAST also has the largest inter-quartile and normal range of all the algorithms which gives it the least consistent goodput performance. However, its inter-quartile range suggest that it consistently gains greater goodput than the other New-TCP algorithms.

BicTCP and H-TCP have similar goodput performances. However, H-TCP has the larger ranges of values - with a significantly larger goodput range. Both algorithms also show that the minimum goodput is likely to be higher than that of FAST's, although BicTCP does have a significant number of outlying points.

HSTCP and ScalableTCP both have relatively lower goodputs than that of the other New-TCP algorithms with comparable performance. This was found to be due to the slow growth of *cwnd* from the small *cwnd* values due to the absence of slow start. Figure 9.34 shows that in the 5 minute tests, HSTCP and ScalableTCP actually spends half of the time to grow *cwnd* to sufficiently large enough values in order to fill the available bandwidth of the path. Notable, however, is that the worst performing New-TCP algorithm (HSTCP) under this network path still achieves approximately five times higher goodput than that of Standard TCP.



(a) HSTCP (b) ScalableTCP

Figure 9.34: Slow Convergence of HSTCP and ScalableTCP over long distance high capacity Internet links.

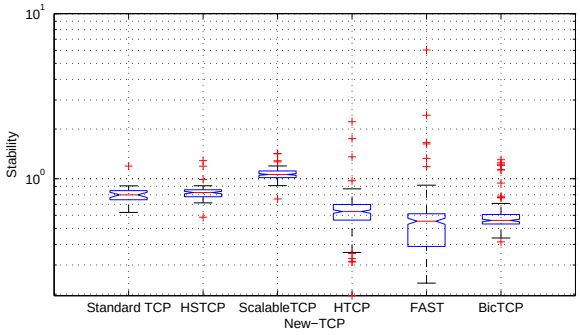


Figure 9.35: Stability from CERN to LBL.

Stability

Figure 9.36 shows that FAST, H-TCP and BicTCP have similar median stabilities, with FAST showing a wider range of stability values.

The other New-TCP algorithms show similar stability profiles, mostly because of the lack of congestion epochs as shown in Figure 9.34.

Overhead

HSTCP has the lowest transfer overhead as shown in Figure 9.36. However, these results have to be taken with care due to the lack of congestion epochs

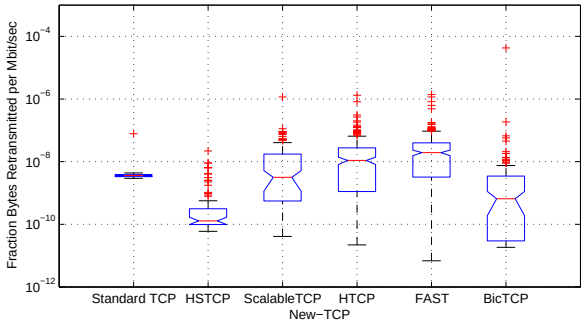


Figure 9.36: Overhead from CERN to LBL.

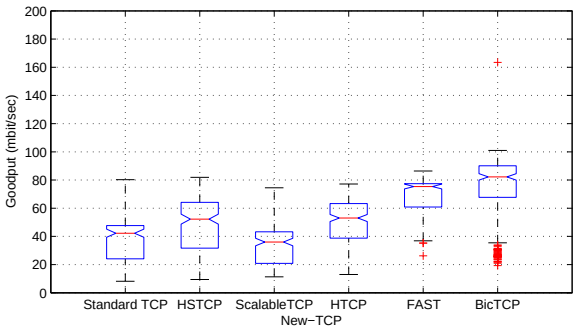


Figure 9.37: Goodput from CERN to RAL.

of the HSTCP and ScalableTCP flows.

BicTCP has the lowest inter-quartile range of all the algorithms and shows that it is able to achieve the least overhead of all of the New-TCP algorithms.

FAST, unlike the tests to Dublin, exhibited the highest overhead with the greatest fraction of bytes retransmitted per unit goodput. However, the results are comparable to that of H-TCP.

9.3.5 Results: CERN to RAL

Goodput

The goodput performance to RAL is show in Figure 9.37. Even though the machines and networks are capable of 1Gb/sec, it demonstrates that the

achieved goodput is much below this maximum. This is likely to be due to the the large amounts of traffic that RAL receives and sends due to its pivotal role as a scientific data centre in the UK.

BicTCP is able to achieve the highest median value, with also the largest number of low goodput outlier points. It was also able to achieve a very high value of over 160Mbit/sec demonstrating the capability of the RAL site under extra-ordinary conditions.

FAST is also able to maintain a high goodput, with a lower quartile which is almost greater than the upper quartiles of H-TCP and HSTCP. FAST also has a high minimum range suggesting that the goodput performance would be greatest with these algorithms.

HSTCP and H-TCP perform very similarly, with HSTCP having a slight tendency for lower goodputs.

Surprisingly, the goodput of ScalableTCP under this very busy environment is actually less than that of Standard TCP, although it does have a slightly higher minimum goodput than that of both HSTCP and Standard TCP.

Stability

Similarly to the goodput results, Figure 9.38 shows that the stabilities between all of the TCP algorithms (Standard TCP included) have similar characteristics.

FAST has the smallest inter-quartile range, suggesting the most predictable stability performance. However, it also has the highest median value - suggesting that this predictability is not beneficial. It also maintains the largest number of outlier points.

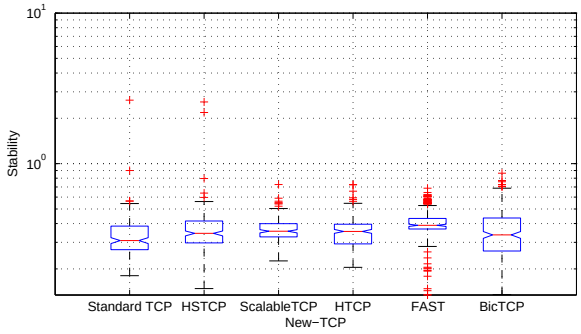


Figure 9.38: Stability from CERN to RAL.

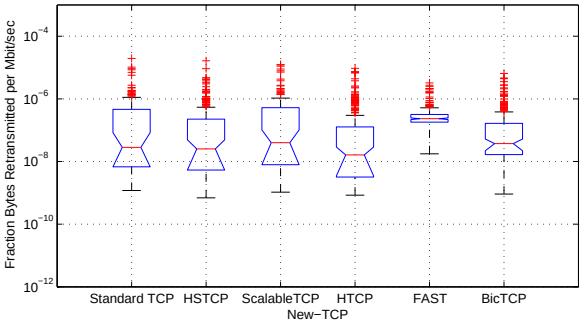


Figure 9.39: Overhead from CERN to RAL.

Overhead

In terms of overhead, all of the New-TCP flows show similar distributions to that of Standard TCP - except for FAST which again maintains a narrow distribution with the highest overhead. This is shown in Figure 9.39. One could argue that H-TCP has the best (i.e. lowest) overhead. However, the notches indicate that the large 95 percentile range between all algorithms overlap.

9.3.6 Summary

Whilst the included network paths are in no way indicative of the typical network topology and network load, they offer an indication of the expected

performance of single flow New-TCP performance over the Internet.

Unsurprisingly, all of the New-TCP algorithms are able to achieve higher goodputs than that of Standard TCP. In certain circumstances substantially higher goodput was achieved.

FAST is consistently able to achieve high goodput performance across all network paths investigated, followed by BicTCP and H-TCP (although the differences between the latter two are negligible).

Under the lower latency destinations investigated, the performance difference between the New-TCP algorithms and Standard TCP are not that great. This is expected due to the TCP algorithm design requirement to be comparable to Standard TCP under low latency, low bandwidth environments.

The slow growth of *cwnd* by HSTCP and ScalableTCP with small initial *cwnd* values are apparent under the CERN to LBL tests, and serves to demonstrate that under highly variable network conditions where sudden flash traffic may occur, they will be incapable of quickly utilising available bandwidth.

In general, the stability and overhead of all of the algorithms are comparable, with mixed results from network to network and are influenced primarily by the number of congestion events experienced.

9.4 Summary

The investigation on real life networks initially began with the analysis of these New-TCP algorithms on the dedicated wide-area test networks of MB-NG and DataTAG.

The experiments demonstrated that there are serious implementation

problems in being able to run these congestion control algorithms effectively as single high bandwidth flows. The problems arise primarily through the limited resources of hardware and implementation issues with software.

The problems in particular are related to the large number of packets that need to be processed by both the sender and receiving TCP hosts. In particular, it was identified that the TCP receiver was not always sending acknowledgments for *at least* every other data packet. This results in *ack* stretching which increases the likelihood of buffer overflows due to larger bursts of data packets on the forward path. More evident, however, was that it causes Linux specific performance penalties due to the way in which *cwnd* and the number of packets in flight are calculated.

A more serious problem, especially for larger values of *cwnd* required for higher speed, long distance networks, is that the processing of SACK packets was demonstrated to also be a bottleneck upon TCP transfer rates.

As demonstrated, the former problem can be rectified with modifications to the Linux kernel. The latter problem can also be resolved with newer and faster hardware; however, a network-based solution was implemented to smooth the loss pattern and hence reduce the processing burden required.

A limited study was conducted to show the interaction of multiple flows of New-TCP traffic. The results demonstrate and corroborate prior results (in Chapter 8) that ScalableTCP has problems with fair and equal sharing of network resources. It was also demonstrated that these New-TCP algorithms can have a serious negative impact upon existing network traffic flows, reducing the amount of traffic utilised by bulk Standard TCP flows by several orders of magnitude. However, as a direct consequence of the implementation problems aforementioned, this impact was reduced due to the inability for the New-TCP flows to be able to achieve high goodputs.

Nevertheless, a DiffServ demonstration was devised to show that with appropriate design and allocation of Best Effort and Assured Forwarding Per Hop Behaviours, the network bottleneck utilisation can be optimised to both provide high performance for the New-TCP flow, whilst protecting the traditional 'existing' traffic from the impact of sharing the bottleneck link.

A separate study to inspect the performance of these New-TCP algorithms over the Internet was also conducted. It showed that for single flow experiments, FAST performed the best in terms of high goodput performance with high stability. BicTCP and H-TCP also performed well. Meanwhile, HSTCP and ScalableTCP were found to offer lower goodput (but still considerably higher than that of Standard TCP) due to the slow rate at which they grow *cwnd* when *cwnd* is small.

Chapter 10

Conclusion

This chapter gives a summary of the work conducted and the conclusions that can be drawn from this dissertation. It also provides a discussion of the areas of further research.

10.1 Summary

This dissertation focused on the performance bottlenecks that are associated with reliable large scale unicast data replication over the Internet.

Appendix A gives a broad overview of the performance of various components required for data replication. In terms of data storage, read and write performance can be improved with the combination of hardware multiplexing with RAID solutions and tuning of kernel parameters. For the transport of data between disparate computers, Network interface Cards at speeds of 10Gb/sec are investigated. However, PCI-X and computational processing are discovered to be the primary bottlenecks at such speeds on server class

PCs. But at 1Gb/sec speeds, tests showed almost flawless performances with CBR UDP traffic.

Chapters 3 through 5 showed that TCP, the primary transport protocol in use on today's networks, is incapable of achieving high throughput transport across high latency, high capacity network environments due to the standard AIMD congestion control algorithm. Also Chapter 9 showed that performance limits exist in the TCP protocol due to the processing of *ack* and SACK packets.

Several New-TCP congestion control algorithms (Chapter 6) were tested under various networks and network conditions (Chapters 8 and 9) using a framework outlined in Chapter 7 which provides a means to comparatively analyse the advantages and disadvantages of each. Tests specifically investigated the raw throughput performance achievable and the interaction of these algorithms both amongst themselves and with legacy Standard TCP.

Even with these New-TCP algorithms, the aforementioned performance bottlenecks due to hardware and protocol design were apparent and it was found that the memory to memory transport of data across long distance high capacity networks is limited to a few hundreds megabits a second under realistic conditions.

The New-TCP algorithm tests showed that:

- All algorithms are capable of achieving greater raw goodput performance than that of Standard TCP.

- The interaction between New-TCP flows of various latencies may cause serious problems in terms of fair sharing of network resources that is much greater than that of Standard TCP. H-TCP - with its RTT Scaling, and FAST - with sufficient network buffer are the only exceptions

to this.

- The slow convergence due to aggressive α and β parameters will lead to long periods of transient unfairness between competing flows. This was especially evident with ScalableTCP.

- The delay based congestion control approach of FAST has several benefits, most notably very good friendliness and high performance. However, it also suffers from the need for network queue provisions to be suitably high for efficient performance.

- Under the assumption that all traffic flows are long lived bulk transport, all New-TCP algorithms impose a specific cost to existing network traffic which can be lowered with network aid such as Quality of Service using DiffServ or Dedicated Circuits. Also the goodput performance of New-TCP flows can be enhanced by offering higher levels of protection against network congestion.

10.2 New-TCP Suitability Study

Table 2.2 in Chapter 2 shows four different application areas for the role of transport on the LHC. In this section, each application area will be discussed in turn with regards to the metrics and performance analysis from Chapter 8 and Chapter 9.

10.2.1 Area 1

The requirement for transport over the Internet requires backwards compatibility with existing Internet traffic flows. However, due to the necessity to get

data from the Tier-0 site to the Tier-1 sites very quickly, the raw throughput is also an important issue.

For a small number of flows (or such that the sum of the number of flows's γ value does not exceed the bottleneck queue size), FAST is able to both sustain high throughput, and maintain good friendliness with StandardTCP. However, this requirement for 'large' queue sizes seriously affects FAST's performance profile. As the infrastructure of the Internet cannot be centrally managed, tuning of such buffers is problematic.

Other New-TCP algorithms capable of high throughput, such as ScalableTCP, maintains very bad RTT fairness between flows. This may also become an issue as different destinations may share the same network bottleneck - therefore TCP algorithms such as ScalableTCP is not advised for deployment under this scenario. BicTCP and HSTCP maintains similar RTT unfairness, however, it is much greater than that of StandardTCP. H-TCP, however, is able to maintain relatively high throughput with very good RTT fairness. Similarly, FAST - given sufficient buffering on the network path, also maintains good RTT fairness.

As superfluous retransmissions may cause Internet collapse, maintaining a low overhead is also important. H-TCP has a noticeable number of retransmissions with longer latency paths when compared to the other algorithm; whilst FAST is able to maintain significantly lower overhead by 2 orders of magnitude.

10.2.2 Area 2

Area 2 has a requirement similar to Area 1, however, poses a greater risk of the TCP transport needing to be shared with potentially high amounts of

local site traffic. As such, the overhead and friendliness becomes much more important under such scenarios.

As the logical latencies between Tier-1 to Tier-2 and Tier-2 to Tier-3/4 are likely to be small due the region design of the MONARC system, the issues of H-TCP having higher overheads at very long latencies becomes less of an issue.

However, there is a higher chance of insufficient network buffering due to the traversal through many autonomous systems and the requirement for the LHC traffic to share with normal site traffic. Also should many LHC flows be sharing the infra-structural links, the requirement for FAST to maintain a queue size proportional to the number of connections makes provision of such buffers difficult.

Due to the unfriendliness and RTT unfairness constraints of ScalableTCP, BicTCP and HSTCP, it is not recommended for use in this application area.

10.2.3 Area 3

Dedicated circuit usage in this application area only requires that the single New-TCP flow maintains high throughput. Under such scenario's, BicTCP, ScalableTCP and FAST are able to achieve very high utilisation of the network pipes.

Due to the loss-based nature of BicTCP and ScalableTCP, FAST is able to maintain a lower overhead, which may give it the advantage in being able to sustain higher throughputs. However, this is only the case when there are sufficiently sized buffers available on the network path. Depending on the type of dedicated circuit (e.g. and entire SONET link, or DiffServ/MPLS provisioned) and more importantly who it is leased from, it may or may not

be possible to manually set the intermittent devices queuesize to facilitate FAST TCP usage.

The ability for the TCP flows to quickly 'grab' available throughput on such empty pipes is also important. Section 9.3 demonstrates that FAST, H-TCP and BicTCP are able to maintain high throughput by virtue of having aggressive throughput gradients on the Internet - which is also applicable on dedicated circuits. ScalableTCP, on the other hand, has a very slow growth rate when throughputs are relatively small - as demonstrated in Figure 9.34.

H-TCP also implements an adaptive backoff algorithm whereby the measured latency can be used to provide maximal utilisation of the network path.

10.2.4 Area 4

In this application area, the sharing dedicated circuits will require that New-TCP algorithms are both fair and have quick convergence times to maximise utilisation.

Both FAST and H-TCP is able to quickly converge to fairness between competing flows, whilst ScalableTCP has undesirable properties of very long convergence times and hence is also transiently unfair. The issue of RTT unfairness will also seriously limit the rate at which sites further away from their parent Tiers can transfer data. It is therefore preferable for each site to sustain equal goodput; irrelevant of the geographical distance. As such, ScalableTCP, BicTCP and HSTCP are not recommended.

Similar to the arguments for Area 3, as long as there is sufficient buffering for the FAST flows, FAST is able to maintain very good properties for high thoughput and fairness. Under insufficient buffering, the loss-based nature of

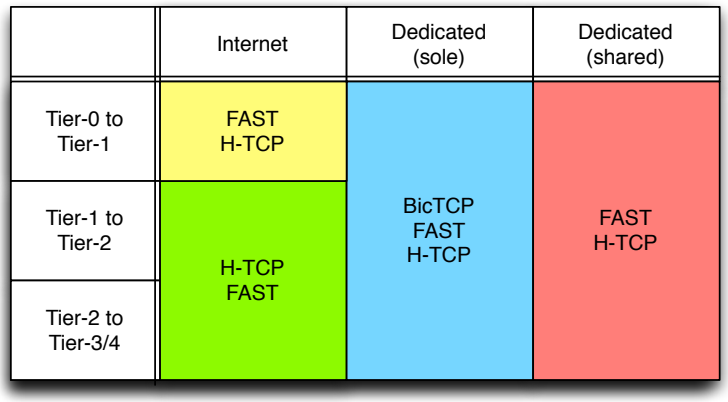


Table 10.1: Overview of Recommended New-TCP algorithms for different Tiers based upon the type of end-to-end connection.

H-TCP will have distinct advantages, whilst with large buffer provisioning, the RTT fairness of H-TCP becomes better.

10.2.5 Summary

Table 10.1 summarises the recommended New-TCP algorithm for use in each application area. Generally, FAST and H-TCP are the recommended New-TCP algorithms for use for the majority of the LHC MONARC design.

However, it should be noted that FAST is only recommended where sufficient network buffering is available. Across multi-domain network paths such as that of the Internet, such tuning may not be possible; especially when there are a large number of FAST TCP flows. This also applies to dedicated network paths - depending upon who supplies the dedicated circuit this tuning may not be possible.

BicTCP, whilst maintaining very unfriendly and relatively long (and hence undesirable) convergence times, does provide high throughput with relatively

low overhead rates and is therefore recommended for dedicated circuits when there is no other traffic.

FAST and H-TCP are recommended for use on dedicated circuits for regardless of whether it is a shared circuit or solely dedicated. Their advantage lies in the ability to maintain high throughput, yet being fair and facilitating fast convergence times between competing flows.

10.3 Future Directions

It is expected that many new TCP congestion control algorithms will come to light both from the natural evolution of the existing New-TCP algorithms and from future designs and ideas. A natural question arises from this: would future networks impose a homogenous or heterogeneous mix of these algorithms? Whilst it is not within the scope of this dissertation to answer this question, it is worth noting that the work in Chapter 7 sets a foundation from which *comparative* testing has been identified as important in being able to understand and evolve congestion control algorithms. It should therefore be important to be able to expand upon these ideas in to a consistent framework from which disparate research groups can effectively contribute to a common shared goal. More importantly it will enable the focus of effort in order to evolve the more promising algorithms.

The expansion of this work is summarised in the following sections.

10.3.1 New-TCP Algorithms

The research conducted in this dissertation suggests that the algorithmic implementation of H-TCP has certain advantages; in particular the polyno-

mial increase of *cwnd* facilitates fast utilisation of network resources, whilst the adaptive backoff and resetting of β to half upon consecutive drops in throughput due to competing traffic greatly aids fast convergence and fairness. However, the polynomial increases also results in a large overhead - especially at long latencies (unsurprising as the polynomial increase is time based). As such, some tuning of this parameter and matching to real networks will be beneficial.

The delayed-based FAST algorithm also demonstrates similar attributes to H-TCP. However, the major disadvantage of such delayed-based algorithms is the requirement of network buffering to be proportional to the number of FAST flows sharing the bottleneck link. As demonstrated throughput Chapter 8, when the network buffers are small, FAST has some very undesirable throughput, fairness and friendliness performances. Therefore, further research into the choice of γ is required. Also a mechanism whereby FAST does not cause these large fluctuations in goodput when there is insufficient buffering should be developed.

Several other TCP congestion control algorithms have been proposed but were chosen to either not be within the scope of the research or were not incorporated due to time constraints.

- TCP Limited Slow-Start [Flo04] is an adaptation of the slow start algorithm to prevent large numbers of retransmits due to the exponential growth of *cwnd* by switching to an aggressive linear increase of *cwnd* when *cwnd* is larger than a threshold value.

- TCP Westwood [CGM+02, GM, WVS+02] utilises RTT to determine the appropriate bandwidth delay product in order to determine the appropriate *cwnd* size for the TCP flow. Unlike FAST, it still incor-

porates standard additive increase of 1 packet, and will back-off *cwnd* upon loss to the determined bandwidth delay product value. It also utilises a filter on the RTT to enable the detection of losses due to BER rather than real network congestion.

- TCP AFRICA [KRB05] was devised to incorporate the benefits of HSTCP with that of delayed based congestion control such that the flow is more stable and fair against competing flows.

- XCP [KHR] is an extension of the ECN methodology whereby the state of network congestion is determined not by end-hosts but from information gathered directly through intermittent routers. The basic idea is that each XCP enabled router determines the appropriate window sizes that each host-pair should use based on internal calculations. This enables the adaptation of sending rates to optimal fair-share allocations in the order of one RTT. Whilst offering potentially the most efficient form of global congestion control, it also requires much infrastructure in the form of XCP capable routers, and XCP capable hosts in order to fully appreciate its benefits.

10.3.2 AQM/RED

Whilst the deployment of AQM solutions is currently limited, it is suggested that AQM *should* [BCC98, FJ93] be implemented to aid fair sharing of network resources.

This dissertation only focused upon the use of drop tail queuing disciplines to study the effects of New-TCP algorithms, and only touched upon a WRED solution to optimise SACK processing. There is therefore a large area of

research, especially within the framework of fairness implications, in using AQM solutions.

10.3.3 Implementation of New-TCP Algorithms

Experimental results in Chapter 9 were hampered by the implementation difficulties related to software and hardware limits of dealing with the flux of incoming and outgoing packets.

An obvious avenue to pursue is the optimisation of SACK processing code, and the investigation of the other performance bottlenecks that may exist within the Linux (and other operating systems) kernels.

An orthogonal approach is to mitigate these problems into dedicated hardware subsystems as with TCP Offload Engines [Gwe01, SUV03] in order to lower the burden of dealing with network activity in a similar way to RAID subsystems for disk activity.

10.3.4 Other Forms of Transport

Due to the various constraints outlined in this dissertation with TCP, several research groups have investigated into non-TCP (unicast) based transport.

Specifically, due to the constraints of requiring congestion control, most implementations offer alternative congestion control strategies, but more importantly offer the solution of using negative *ack*s in order to reduce the reverse traffic burden by signaling losses rather than successful data receipt.

Such research offers insight as to whether it may be beneficial to completely replace TCP with an alternative protocol for data replication. However, as most Internet software relies upon TCP for communication, much work will need to be done in terms of updating such software.

A more immediate problems to the deployment of non-TCP based transport are those associated with firewalls due to usage caps on UDP traffic and non-regular port numbers.

10.3.5 Further Tests

Web Traffic

To make the tests more realistic, and as defined in Chapter 7, the systematic series of tests presented in Chapter 8 should be extended to incorporate varying levels of web traffic. The effect of this would be that the changes of the bottleneck queue size may affect the dynamic between New-TCP flows.

One specific example would be how FAST TCP will scale under such scenario's and whether the 'noise' imposed by the queue dynamics will have an adverse effect on the delayed-based portion of the algorithm. The effect upon the other algorithms should be less noticeable; however, algorithm problems involving large bursts of packet due to large α values may become more apparent due to the queuing dynamics imposed with web traffic.

Impact of Reverse Path Topologies

With the increased aggressiveness of New-TCP algorithms in order to attain large *cwnd* values required for high speed transport, it is important to understand the effects and causes of packet bursts in order to better design new transport algorithms [BA02, BPS99]. This is also important in the provisioning of network router queues and the designation of AQM parameters.

A direct area of interest related to packet bursts is the use of packet pacing whereby the sending of packets is spread over the period of a single RTT in order to prevent clustering of data packets, and thus also improve

ack clocking. Whilst simulations studies in [ASA00] suggest that there may be negative global consequences under Standard TCP, the consequences of large packet bursts as a result of large α parameters may be mitigated.

Large Scale Simulations

Whilst this dissertation focuses on a small number of flows, a more difficult and pressing question is the large scale effect of the phased deployment of these New-TCP algorithms.

As covered by the *impact* factors, it is important that the implementation of these New-TCP algorithms do not adversely affect existing network traffic dramatically. Meanwhile, it is also vital to trust that complete deployment (100% usage) of such algorithms is globally stable and fair.

A specific area of research is that of the interaction between short lived and long lived flows [BC02] as bulk transport flows do not constitute all of the Internet's traffic.

10.4 Conclusion

There is much work that needs to be done in order to keep the growth and expansion of the Internet at its current rate. This is important in order to effectively collaborate and share ideas on a truly world wide scale.

A realisation of this is through the collaboration of world wide science projects of which a major component is the technology and middleware required for the operation of future data and compute grids. An integral part of information retrieval is the rate at which data can physically be transferred from one place to another.

This dissertation investigated the performance bottlenecks, tuning and modifications of hardware and software components to maximise this metric.

More specifically, this dissertation studied the performance bottlenecks of TCP and congestion control algorithms. It investigated new congestion control algorithms in a series of systematic tests and discovered potentially serious implications upon fairness and impact upon competing traffic. As such, a Quality of Service solution using DiffServ was investigated which mitigated these effects and demonstrated significant performance improvements due to the protection of the flow.

In conclusion, whilst the short term goal of achieving faster achievable goodputs is possible, the long term issues of Internet stability and fairness are still unclear.

Appendix A

Hardware Performance Tests

A.1 Data Storage

With the growth of Data Grids becoming important to enable processing of large sets of data across the world, a fundamental question is where and how the data is stored. In its simplest form, data may be stored on Desktop PC's which act as temporary scratch space for local processing. On the other hand, regional centres and the like may have multiple storage options depending on the importance of the data in question. As such, archived data may be stored on cheap tape based systems, whilst more immediately required data may be stored on large storage clusters such as RAID farms.

This section gives an overview of RAID technologies and gives examples of existing performance limitations on such systems.

While it may be possible to make fast transfers across high bandwidth networks, transfers of real data are limited by the rate at which end hosts can read and write data to and from disk. This section demonstrates that

with the choice of appropriate hardware with optimised kernel and hardware settings the effect of this bottleneck can be significantly reduced.

A.1.1 Overview

RAID stands for 'Redundant Array of Independent Disks' and is a method of distributing the storage and retrieval of data across many, usually identical hard disks. Depending on the configuration of the RAID, one may achieve increased performance in terms of data access time and transfer rates, or one may seek protection against data loss in case of hard drive failure.

The latter is important to ensure constant and continual availability of storage systems as restoration from backups will often involve taking down the system. Data stored in data protected RAID systems can also be recovered with minimal hassle.

RAID controller solutions can be hardware or software implemented and can be configured for different storage needs. As such, the performance of several types of hardware RAID controllers, mounted on high performance end systems, and types of RAID configuration are investigated.

Originally developed by D. Patternson [PGK88] in 1988, a series of RAID configurations were defined which manufacturers have adopted[1]. Some of the current RAID standards are outlined below.

- RAID-0: Commonly referred to as *striping*. It implements a striped disk array whereby the data is broken down into blocks which are written to separate hard drives in parallel. This solution provides no data redundancy and therefore has very little overhead. It therefore offers

[1]RAID-0 was not part of the original specifications, but presented later in 1994 in [CLG+94].

absolutely no fault tolerance and the failure of just one drive will result in all data being lost.

- RAID-1: Also known as *mirroring*. Total redundancy is provided by writing identical data to multiple disks, and therefore requires at least two hard drives. As the data is written just once (and mirrored) there is no improvement in write times over a single hard disk. However, as the data is distributed over pairs of disks an improvement of two times the single drive read speed can be achieved. As each pair of disks is mirrored, there is no performance increase as the number of disks used increases. However, as there is 100% redundancy of data, no rebuild is necessary in case of a disk failure and a simple copy of data to the replacement disk is required.

- RAID-10: Combines *striping* and *mirroring*, thus providing high performance read write speeds and total redundancy. It is implemented as a striped array who segments are RAID-1 arrays. As such it has the benefits of performance and fault tolerance of both RAID-0 and RAID-1 systems. However, the use of so many disk drives means that the cost is high as storage capacity is low. In many ways RAID-10 is an ideal solution for high speed, fault tolerant data transfer and storage.

- RAID-5: Writes data in stripes similarly to RAID-0, but also checksums the data and creates parity blocks in the same rank on writes. These are distributed across the total number of disks alongside the ordinary data stripes. It requires a minimum of three disks to implement. In the event of a single disk failure the ordinary data can be recovered using the parity data. This configuration gives high redundancy to-

gether with high capacity and storage efficiency. Due to the inclusion of the parity data total write speeds are slower than for RAID-0. Total read speeds are also slower as the parity stripes must be skipped over in order to reach the ordinary data stripes.

A.1.2 Test Methodology

A set of tests were performed to compare the read and write speeds of a selection of RAID controllers. RAID configurations were chosen to compare and contrast the optimal performance configuration of RAID-0, and an optimum real life implementation with RAID-5.

Only the transfer of large data-sets is considered where it is assumed that the data is stored in a roughly linear manner whereby hard disk seek times may be neglected[2].

All tests were performed on 'server-class' PC's as shown in Table A.1 with the associated RAID controller cards as shown in Table A.2. The disks used in all tests are shown in Table A.3 and were formatted with the `ext2` file system that is representative of most Linux based systems which are prevalent in research communities.

Two separate programs were created to perform the reading and writing of data to and from user space. They implement the standard `read()` and `write()` Unix calls. Given a predefined file or file size, the `read()` program reads data from the RAID partition into a circular buffer in memory. The `write()` program writes a defined size of random data from memory onto disk.

The `vm.max-readahead sysctl` [Rub97] in Linux affects how early the

[2]It is also assumed that the presence of file fragmentation is negligible.

	Description
CPU	Dual Intel(R) Xeon(TM) CPU 2.40GHz
Memory	512MB
Motherboard	Supermicro P4DP8-G2
Front-side bus speed	400Mhz
PCI Word Size	64 bit
PCI Speed	66/100/133Mhz
OS	Redhat Linux 7.2

Table A.1: Hardware and software configuration of PCs used for RAID performance tests.

Raid Controller	Controller Type	PCI Interface	Driver Version
ICP GDT8546RZ	SATA	64bit 33/66Mhz	2.03
3Ware 7505-8	ATA	64bit 33Mhz	1.02.00.031
3Ware 7506-8	ATA	64bit 66Mhz	1.02.00.031
3Ware 7506-8	SATA	64bit 66Mhz	1.02.00.031

Table A.2: RAID Controllers under investigation.

Manufacturer	Controller Type	Size (Gb)	Spindle Speed (rpm)	Cache Size
Maxtor	ATA	160	7500	8MB
Maxtor	SATA	160	7500	8MB

Table A.3: Hard disk models used in RAID tests.

Linux VFS (Virtual File System) fetches the next block of a file from memory. For the retrieval of data stored on disk in a uniform linear fashion (such as a large file) transfer rates can be increased with larger values of `vm.max-readahead`. However, increasing the value of `vm.max-readahead` may result in excess and often unnecessary memory usage. Conversely, lowering this value will result in a decrease in memory usage at the cost of reduced performance.

The virtual memory setting in the Linux kernel `vm.max-readahead` was varied to determine if any improvement in read and write speeds could be achieved. The variable `vm.min-readahead` which controls the floor of the `vm.max-readahead` settings was also investigated but was found to have negligible effect and the results are therefore are not presented.

The read and write speed tests were performed for a distribution of file sizes ranging from 100MB to 2GB with increments of 100MB in order to give a realistic range of useable data file sizes.

Each measurement was repeated five times from which the arithmetic mean value was calculated. This mean was used as the central value from which to determine the standard deviation for the five measurements to determine the standard error on the mean [Boa83].

A.1.3 Results

Figures A.1 to A.5 show a series of graphs for read and write speeds versus file sizes for different settings of the Linux variable `vm.max-readahead`. Due to hardware restrictions, only RAID-5 configurations with an array of 4 disks were investigated.

Figure A.6 summarises the RAID-5 results achieved by the various controllers with `vm.max-readahead` settings of 1200 and also includes 8-disk arrays. Figure A.7 is the equivalent summary graph for RAID-0. The results for 2GB files are also displayed in Table A.4 .

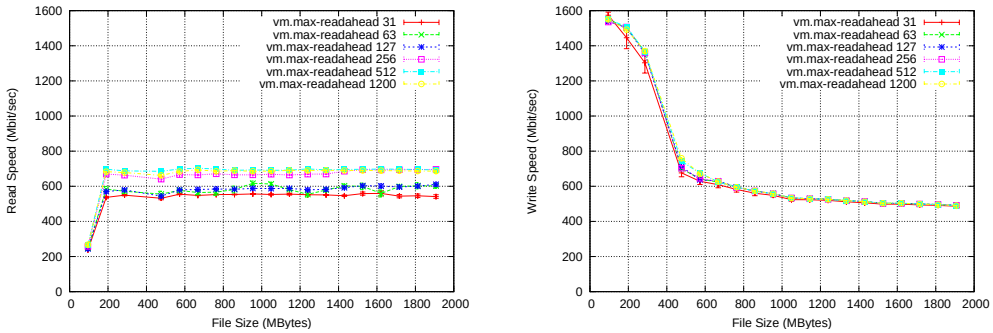
All graphs shown for read and write speeds show similar shapes: write speeds typically begin at a relatively high value for small file sizes then drop quite suddenly at around 400MB for all RAID-5 configurations. The graphs flatten off to a relatively low value - approximately 500 to 600Mbit/sec for

the ICP cards and about 400Mbit/sec for the 3Ware cards. The trend of the drop can be partially explained by file caching and also by the extra overheads introduced by the writing of the parity information required for RAID-5. The RAID-0 write distributions (Figure A.7) are typically much flatter as parity information is not created.

The read speeds remain essentially constant for all file sizes, only decreasing to lower speeds with file sizes smaller than 200MB. This can be explained through the overheads of file seek and buffer management. It was observed that the read 3Ware cards were able to achieve about 50% greater read performance than that of the ICP cards at optimal `vm.max-readahead` settings.

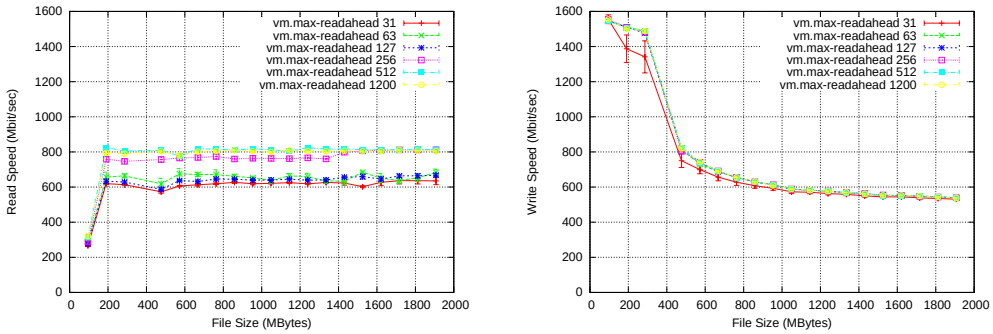
For all configurations, it was observed that an increase in the `vm.max-readahead` value also increased disk read performance. As `vm.max-readahead` continues to be set to higher and higher values the relative improvement in disk performance diminishes. No significant improvements in disk performance were achieved for `vm.max-readahead` values greater than 1200 and hence larger values are not included. There appears to be no conclusive increase in the write performance with a change in `vm.max-readahead`.

Figure A.1 and Figure A.2 show results for the ICP SATA RAID controller configured as RAID-5 with PCI interface jumper settings of 33MHz and 66MHz respectively. The read speed clearly improves as `vm.max-readaheadis` increased. The write speeds remained largely unaffected, although there was a larger standard deviation of values at small file sizes for a `vm.max-readahead` of 31. In both cases, there is no discernible increase in performance until the `vm.max-readahead` was set above 127. The use of a 66Mhz PCI bus clearly leads to an increase in performance in both read and write speeds. However, it should be noted that a tuned 33Mhz ICP card is able to get the



(a) Read Performance (b) Write Performance

Figure A.1: Performance of ICP SATA (RAID 5 with 4 Disks and 33Mhz PCI).

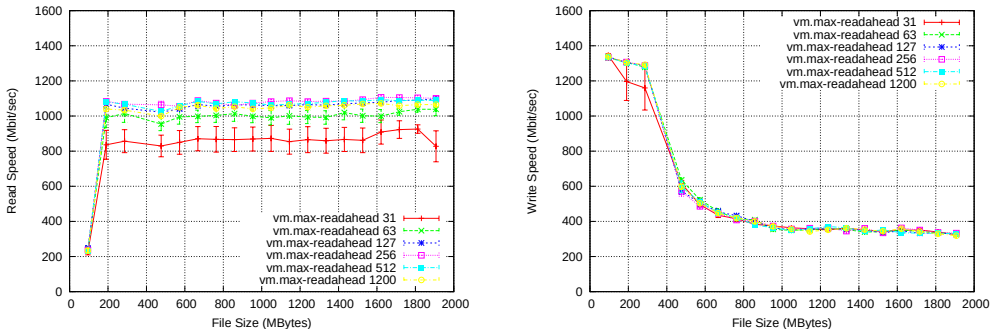


(a) Read Performance (b) Write Performance

Figure A.2: Performance for ICP SATA (RAID 5 with 4 Disks and 66Mhz PCI).

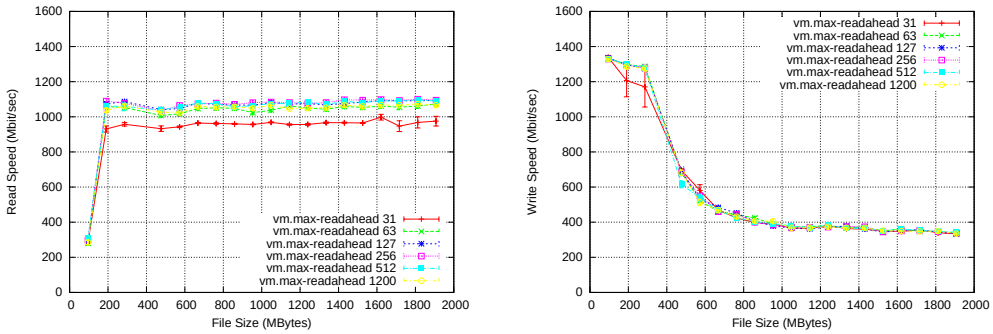
same read performance as that of an un-tuned 66Mhz ICP card under these tests. A slightly higher write speed was achieved with the 66 MHz PCI mode compared with the 33 MHz mode.

Figure A.3 and Figure A.4 show results for the 3Ware parallel ATA RAID controllers configured as RAID-5. Figure A.3 refers to the older controller with a 33 MHz PCI bus speed, whilst Figure A.4 refers to the newer model with a 66 MHz PCI bus speed. The read speeds improve with the `vm.max-readahead` setting for both cards, with the newer card with the 66 MHz PCI bus showing greater stability in terms of the variation in speed



(a) Read Performance (b) Write Performance

Figure A.3: Performance for 3Ware ATA (RAID 5 with 4 Disks and 33Mhz PCI).



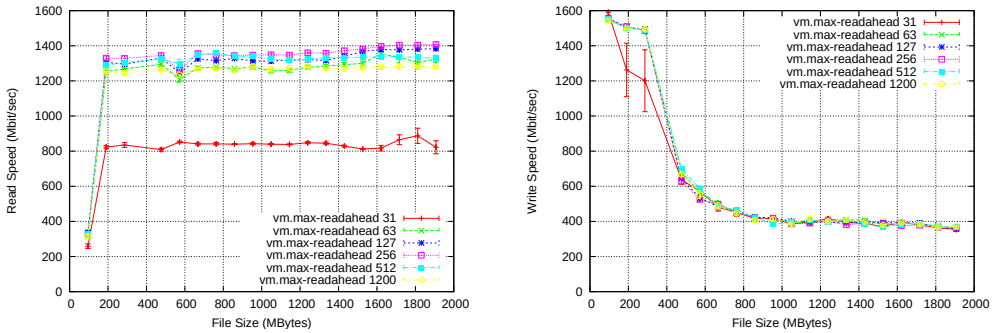
(a) Read Performance (b) Write Performance

Figure A.4: Performance for 3Ware ATA (RAID 5 with 4 Disks and 66Mhz PCI).

for small values of `vm.max-readahead`. However, there was no observable increase in the read speed for higher settings of `vm.max-readahead`.

However the overall read performance for both the 3Ware parallel cards is over 200Mbit/s faster than for the ICP card. The write speeds for the two 3Ware cards are also very similar with the 66 MHz card having marginally better performance. On the other hand, the overall write performance of the 3Ware cards is significantly less than that of the ICP card, being over 200Mbit/s slower.

The particular ICP card under test had a maximum disk array size of 4



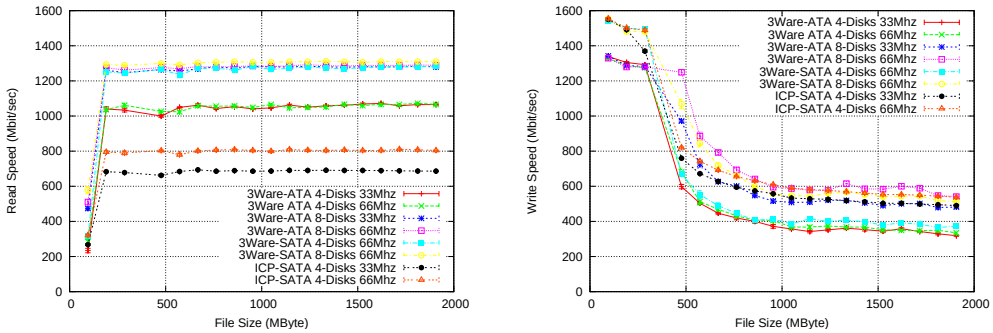
(a) Read Performance (b) Write Performance

Figure A.5: Performance for 3Ware SATA (RAID 5 with 4 Disks and 66Mhz PCI).

disks. The 3Ware controllers are able to hold up to 8 disks. The read and write speeds for the 3Ware cards were greatly enhanced when 8 disks were included in the array as demonstrated in Figure A.6 as well as Table A.4. An ICP GDT8586RZ 8 channel array could not be obtained, and therefore equivalent testing was not possible.

Figure A.5 shows results for the 3Ware-SATA RAID controller configured as RAID-5. This card has a 66 MHz PCI bus; an equivalent 33 MHz version was not available for testing. The read speeds for this controller were the fastest of all the cards in these tests, being marginally faster than the 8-disk array for the 3Ware parallel ATA controllers. Also the 3Ware-SATA read speeds did not significantly improve when 8 disks were included in its array as opposed to 4 (See Table A.4). The write speeds achieved are comparable to the equivalent speeds achieved with the parallel cards, being only slightly faster with a 4-disk array and slightly slower with an 8-disk array.

Figure A.6 shows a comparison of the maximum (`vm.max-readahead` 1200) read and write speeds achieved for the various controllers and RAID-5 configurations. Table A.4 shows detailed read and write speeds for 2 GB transfers. The greatest overall read speeds (approximately 1300Mbit/s) were



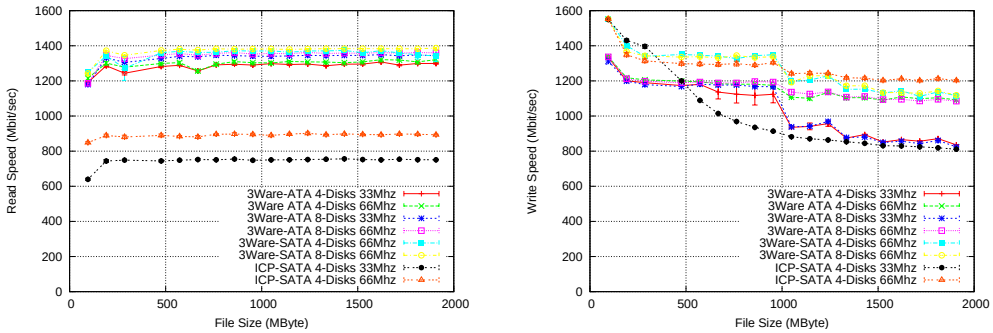
(a) Read Performance (b) Write Performance

Figure A.6: Performance of Various Controllers in RAID-5 Configuration.

achieved by the 3Ware-SATA card mounted with 8 disks, closely followed by the 3Ware parallel ATA running at 66 MHz with 8 disks and then the 3Ware parallel ATA running at 33 MHz. The 3Ware controllers perform less well when running with only 4 disks, each reading at approximately 1050Mbit/s.

The read performance the ICP cards in RAID-5 were not as high as expected. When running in 66 MHz mode typical write speeds were approximately 800Mbit/s, whilst the 33 MHz mode only read at around 700Mbit/s. The greatest overall write speeds were achieved by the 3Ware-ATA 66 MHz card mounted with 8 disks. The 3Ware-SATA, 3Ware-ATA 33 MHz and ICP 66 MHz controllers performance was approximately equal. However the ICP cards only ran with 4 disks. Out of all the 4 disk array tests the ICP 66 MHz controller achieved by far the greatest write speeds.

Figure A.7 shows a comparison of the maximum (`vm.max-readahead` 1200) read and write speeds achieved for the various controllers and RAID-0 configurations. Again, the 3Ware cards show better read performance than that of the ICP cards and are comparable to that of the RAID-5 performances. However the read speeds are approximately 50 to 100Mbit/sec faster in RAID-0 over RAID-5 configurations. This is expected as the read-



(a) Read Performance (b) Write Performance

Figure A.7: Performance of Various Controllers in RAID-0 Configuration.

ing of data under RAID-5 requires the skipping of parity data which would result in a light performance decrease. The main performance increase in using RAID-0 over RAID-5 is that the write speeds are significantly higher as parity data does not need to be generated for each block of data written. As such, write speeds of approximately twice the rate of that of RAID-5 were observed. The pattern of write speeds is also different; with all cards showing a drop in write performance at approximately 1GB file sizes. The ICP card, at 33Mhz, however, shows the worst write performance with larger file sizes, not showing the stepped performance decrease associated with the other cards. However, at 66Mhz, the ICP card performs the fastest with RAID-0 write speeds with large files, and only marginally slower than the 3Ware SATA cards with files less than 1GB in size.

A.1.4 Summary

Table A.4 shows the read and write performances observed using various RAID cards in both RAID-0 and RAID-5 configurations using optimally tuned Linux kernel parameters using `vm.max-readahead`.

Controller Type	Number of Disks	RAID-0 (Mbit/sec) Read	RAID-0 (Mbit/sec) Write	RAID-5 (Mbit/sec) Read	RAID-5 (Mbit/sec) Write
ICP 33	4	751	811	686	490
ICP 66	4	893	1,202	804	538
3W-ATA 33	4	1,299	835	1,065	319
3W-ATA 66	4	1,320	1,092	1,066	335
3W-ATA 33	8	1,344	824	1,280	482
3W-ATA 66	8	1,359	1,085	1,289	541
3W-SATA 66	4	1,343	1,116	1,283	372
3W-SATA 66	8	1,389	1,118	1,310	513

Table A.4: Summary of Read and Write Speeds of 2GB Files for controllers configured as RAID 0 and RAID 5 with vm.max-readahead=1200.

Generally, the performance of both read and write speeds is aided with the following configurations:

Tuning of Kernel Parameters It was observed that setting `vm.max-readahead` to 1200 helped in all cases. The default value of 31 reduced the performance of RAID-5 devices by as much as 75% in the case of the 3Ware SATA tests.

Number of Disks An increase in the number of channels being used in the RAID helps with the striping of data across hard disks and improves the read speed as each separate hard disk is able to retrieve a separate block of the file.

PCI Bus Speed A higher PCI bus speed is recommended, with the greatest increase in performance achieved with the write performance of the RAID devices. The performance increase of the 3Ware cards with increased PCI Bus speed was not as noticeable as that of the ICP cards.

File Size Whilst the read performance remains constant for all file sizes

larger than 100MB, the optimal write file size is less than 400MB under RAID-5. The decrease in RAID-0 write performance with file size is not as pronounced, but all cards observe a stepped decrease in performance with files larger than 1GB.

A major factor that should be considered in the feasibility of these tests is that the test programs are only single threaded and only read and/or write a single file at any one time. This type of access to files is unlikely in real world scenarios - especially as RAID systems are often deployed on busy server systems which are shared by many users accessing different files at the same time. The result of this would be a marked decrease in the performance achieved in 'real-world' environments. However, in-depth study of user access patterns would be required to give a realistic representation of expected multi-user performance.

However, the tests do serve to demonstrate the optimal performance achievable by the various RAID systems tested and demonstrates that the raw performance achievable under RAID-0 is capable of reading and writing data in the order of a gigabit/sec. However, the requirement of fault protection and data integrity is often more important than transfer speeds, and as such it is very unlikely that RAID-0 systems would be deployed in real-world environments. It was shown that this trade off for file integrity and protection would result in a decrease in write performance of up-to 50%.

A.2 Network Interface Cards

The need for intercommunication between computer systems is often handled using Network Interface Cards (NICs). These are defined by Layer 1 and

Layer 2 of the ISO/OSI model and Layer 1 of the TCP/IP network model.

Typically, most NICs are Ethernet (IEEE 802.3) based [MB76] due to its low cost. It is also worth noting that the interconnects used to construct the Layer 1 communications are also cheap - utilising twisted-pair copper cabling[3]. The viability of utilising Ethernet as a generic hardware networking solution is also increasing as computer hardware manufacturers and suppliers have also been integrating NICs into their products as standard.

As defined in the OSI and TCP/IP models, the NIC provides the base and hence limiting factor in the communication speeds between networked devices. It is therefore important to benchmark the performance at which data can be physically and logically placed into the network.

A.2.1 Test Methodology

As the NIC is the primary subsystem that communicates data into a format which can be sent over any network, it is important to investigate its performance without the intervention of Routing and Switching which may impose further bottlenecks/performance deterioration.

The performance of a Network Interface Card is defined as follows:

Speed The absolute rate at which data can be transmitted to and from the NIC. Due to header size constraints and the relation between the packet sizes used and overhead required per packet, smaller packet sizes are more likely to cause a performance burden.

Latency The amount of time that it takes for the system to transmit a packet. More importantly, the change in latency with packet size can

[3]More expensive fibre optic based Ethernet cards are also available.

determine how close to theoretical hardware limits data transport is capable of getting [HJS00].

In general, the the transfer of data between networked computers must involve the interaction of both hardware and software. In the case of the hardware requirements, the data must be transferred from memory to the network interface before the data can be placed onto the network. Therefore, the interaction and co-operation between the memory subsystem, the CPU, the bridge to the input-output bus, the input-output bus (in this case PCI/PCI-X) and finally the NIC are important [HJS00].

In terms of software, the design and implementation of application layer programs, the network stack, and network drivers is also of importance to be able to both schedule and regulate data flows into and out of the hardware subsystems. An overview of some of these major components is presented below.

TCP/IP Kernel Implementation The translation of data flows from user-space into transport layer datagrams (with relevant source-destination and checksum information) are performed here. The encapsulation into IP packets and then consequently Ethernet frames is also performed here.

NIC driver The driver provides the translation between the kernel level data and bit formats which the underlying layer 1 hardware can understand. Scheduling both input and output data are also managed here by managing interrupts [PJD04].

NIC hardware The chipset of the network interface card actually does the work of forming/receiving the electrical, photonic or RF signals for

Layer 1 propagation.

Similarly, the transport of data also relies upon the co-ordination of other hardware components within the PC subsystems.

CPU Speed The rate at which data from Applications can be processed and the techniques of scheduling and the processing of hardware through the kernel are determined by the CPU chipset and clock frequencies.

Memory Speed and Bus Speed The copying of data to and from main memory into and out of the CPU can be crucial factors when a lot of data is being copied to and from the application.

PCI Word Length and Frequency The number of bits that can be transferred at once across the PCI subsystems (the Word Length) [Gro00] and the rate at which data can be placed into the PCI bus (the frequency) imposes a fundamental limit on the rate at which the CPU and chipset architecture can communicate with NIC hardware.

UDP packets were used to determine the performance characteristics of a selection of NICs. This is important as the various features of TCP may impose a performance limit and as such would not give a determination of the NIC performance, but the performance of the TCP stack itself. The use of UDP also enables finer control of packet sizes and the rates at which data is generated and sent to the NIC from userspace without modification to any underlying systems. As such, the use of UDP packets should give an indication of the raw performances achievable by suitably programmed applications.

As such, a series of tests were developed to measure and determine the performance of the various components that contribute to the performance

of data transport. For all tests, server-class rack-mount PCs were used as defined in Table A.5 and Table A.6 using the NICs listed in Table A.7.

	Description
CPU	Dual Intel(R) Xeon(TM) Presontia 2.20GHz HT[4]
Memory	1GB
Motherboard	Supermicro P4DP6
Chipset	Intel E7500 (Plumas)
Front-side bus speed	400Mhz
PCI Word Size	64 bit
PCI Speed	66/100/133Mhz
Kernel	Linux 2.4.19

Table A.5: Hardware and Software Configuration of PCs used for NIC performance testing.

	Description
CPU	Dual Itanium 2.10GHz (64-bit)
Memory	2GB
Motherboard	HP RX2600
Front-side bus speed	800Mhz
PCI Word Size	64 bit
PCI Speed	133Mhz
Kernel	Linux 2.5.72

Table A.6: Hardware and Software Configuration of High Performance Itanium PCs used for NIC performance testing.

NIC	Chipset	Driver Version
Intel Pro (Onboard)	Intel 82546	e1000 4.4.12
Syskonnect	SK9843	sk98lin v6.0 04
Intel PRO/10GbE LR	-	Ixgb 1.0.45

Table A.7: Hardware and Software Configuration of NICs tested.

A.2.2 Latency

In order to determine the limits of network interface card performance, the implications of having data transferred through the PCI bus onto the NIC should also be considered. The rate at which data can be transferred through the PCI bus is determined by both the word size and the frequency of the bus. Table A.8 shows the theoretical latencies expected from transferring data through the PCI bus.

In order to determine the speed at which packets can be processed by the PCI subsystem and the NIC, a single UDP packet (with relevant encapsulation) were sent direct from userspace using the unix `send()` function using various sized data packets. This 'request' packet solicits the generation of a 'response' packet from the server application on the receiving machine, which then sends a constant sized UDP packet back to the sender. The time measured between the initial send and the receipt of the 'response' packet is used to determine the Round-Trip Latency.

It is assumed that the processing of a UDP packet to be sent out requires the copy of data into memory, and then the consequent transfer (after encapsulation etc.) of the data through the PCI subsystem into the NIC hardware. The rate at which data is to be transferred to the receiving system is then limited to the rate at which the Layer 1 and Layer 2 hardware and data link components operate. The receipt of the 'response' packet also requires the traversal of the PCI subsystem followed copying of the data into memory before being delivered to the application.

The theoretical inverse data rates of the PCI bus and NIC speeds are given in Table A.8 and A.9 respectively. Table A.10 gives the memory data rates for the PCs used in the tests. Therefore, define the minimal rate at

PCI Bus Speed (Mhz)	PCI Word Length (bits)	Data Rate (Gbit/sec)	Inverse Data Rate (μsec/byte)
33	32	1.056	0.00758
66	32	4.224	0.00379
33	64	2.112	0.00379
66	64	4.356	0.00189
133	64	8.512	0.00094

Table A.8: Theoretical data rates of PCI bus speeds.

NIC Speed (Mbit/sec)	Inverse Data Rate (μsec/byte)
10	0.80000
100	0.08000
1,000	0.00800
10,000	0.00080

Table A.9: Theoretical data rates of various NIC speeds.

Model	Processor	Front Side Bus (Mhz)	Data Rate (Gbit/sec)	Inverse Data Rate (μsec/byte)
Supermicro P4DP8-G2	32-bit	400	12.8	0.000625
HP RX2600	64-bit	800	51.2	0.000156

Table A.10: Theoretical memory data rates of PCs.

which data can be transferred, τ. to be:

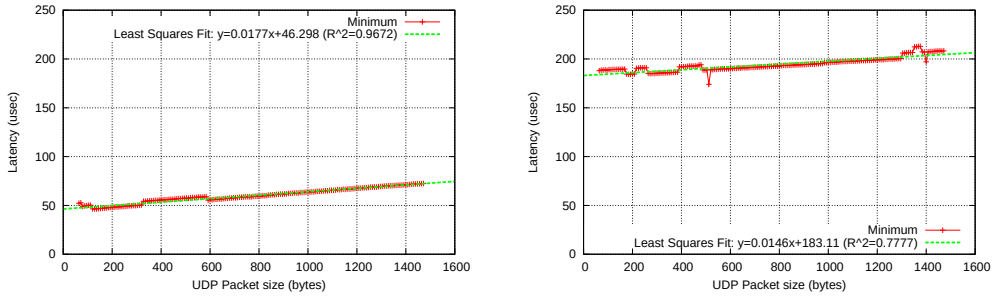
$$\frac{1}{\tau} = \frac{2}{t_m} + \frac{2}{t_p} + \frac{1}{t_n} \tag{A.1}$$

Where t_m is the rate at which data is copy to or from memory, t_p is the rate at which data is transferred over the PCI bus and t_n is the rate at which data is transferred over the network (assuming no losses nor congestion).

By varying the size of the 'request' packet, and keeping the 'response' packet size constant, the rate at which packets are processed by the various subsystems can be calculated. Also the actual propagation delay through the

Model	Processor	Front Side Bus (Mhz)	PCI Word Length (bits)	PCI Bus Speed (Mhz)	NIC Speed (Mbit/sec)	Inverse Data Rate (μsec/byte)
Supermicro	32-bit	400	64	133	1,000	0.011130
Supermicro	32-bit	400	64	133	10,000	0.003930
HP RX2600	64-bit	800	64	133	10,000	0.002992

Table A.11: Theoretical combined data rates of PCs



(a) Syskonnect (b) Intel

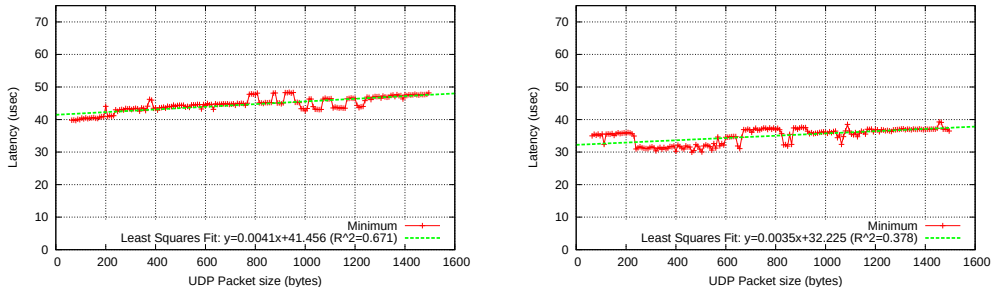
Figure A.8: Ping-pong latency of 1Gb/sec NIC Cards (66Mhz in 64-bit PCI buses).

sum of the hardware components can be calculated by the y-intercept of a latency/packet-size graph [HJS00]. Table A.11 shows the expected/theoretical data transfer rates of various hardware combinations.

For each packet size, the request-response was initiated 1,000 times in a serial fashion and the minimum values are reported. The spread of the latencies experienced by the 1,000 packets for all packet sizes and testbed configurations were not found to be significant and therefore is not presented.

Figure A.8 shows the performance of two popular 1Gb/sec Ethernet cards. The smooth function of the Syskonnect indicates that the driver-NIC management works well with a low latency suggesting that the driver interrupts once for every packet received. The recorded slope of 0.177μsec/byte is also within reasonable agreement with the theoretical value of 0.0111μsec/byte.

The Intel NIC, however, has much higher latency, suggesting that some



(a) Supermicro P4DP8-G2 Dual Xeon (b) HP RX2600 Dual Itanium

Figure A.9: Ping-pong latency of 10Gb Intel NIC Cards (PCI-X at 133Mhz).

form of interrupt coalescing is occurring whereby the number of software interrupts by the driver (to notify the kernel of data availability in the NIC) is decreased by not triggering a software interrupt for every hardware interrupt or by interrupting once in each period of time.

This has the benefits of lowering CPU utilisation as the number of context switches required is reduced at the cost of higher latency of data receipt. The very low gradient of 0.0146μsec/byte observed for the Intel makes good agreement against theory (0.0111μsec/byte).

Figure A.9 shows the latency performance of the Intel PRO/10GbE LR Server Adaptor on two different high performance server PCs. The Xeon-based system (same as that used in the 1Gb/sec NIC tests) very good agreement between the measured values of 0.0041μsec/byte and the theoretical slope of 0.00393μsec/byte. The Itanium-based system also shows good correlation between the measured results of 0.0035μsec/byte and theory of 0.00299μsec/byte.

Note in both cases for the 10Gb/sec NIC cards, the least squares fit of the measured results are not as good as that of the 1Gb/sec NIC tests which would account for discrepancies in the measured gradients.

A.2.3 Throughput

Due to the simplicity of the UDP datagram format and stateless nature of UDP packets, the use of UDP packets to determine throughput can determine the raw absolute throughput achievable by appropriately designed programs.

The `UDPMon` [HJ] program was used to transmit a stream of UDP packets at regular intervals. The rate at which the packets are received can be used to determine the throughput achieved.

The rate at which data is transferred is determined by two factors:

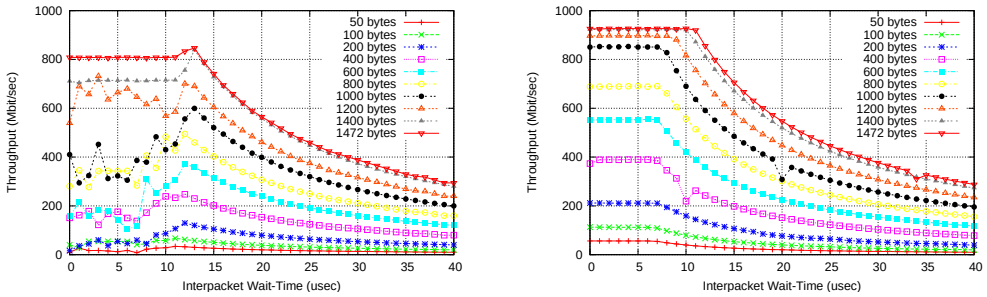
Packet Size The amount of data in each packet; each packet requires the necessary overhead of UDP, IP and Ethernet information, each Ethernet frame must have at least 60B[5] for data communication to occur. As the amount of application data in each packet is increased, the headers require a lower relative overhead and performance improves. In all tests, the 'wire rate' is shown which includes all the necessary header overheads for comparison against the published Ethernet rates[6].

Inter-packet Time By adjusting the time between the injection of packets into the network, and assuming constant packet size, the rate at which data is sent into the network can be increased or decreased. Therefore, a $\frac{1}{t}$ relation between the throughput and the inter-packet time, t, is expected.

Therefore higher throughputs will be achieved with an increase in the packet size and by increasing the amount of data that is sent per time unit.

[5]Includes: 8B for the inter-packet gap, 6B for the preamble, 18B for the Ethernet header and CRC, 20B for the IP header and 8B for the UDP header.

[6]The actual transfer rates achievable for real data exclude the header overheads.



(a) Syskonnect (b) Intel

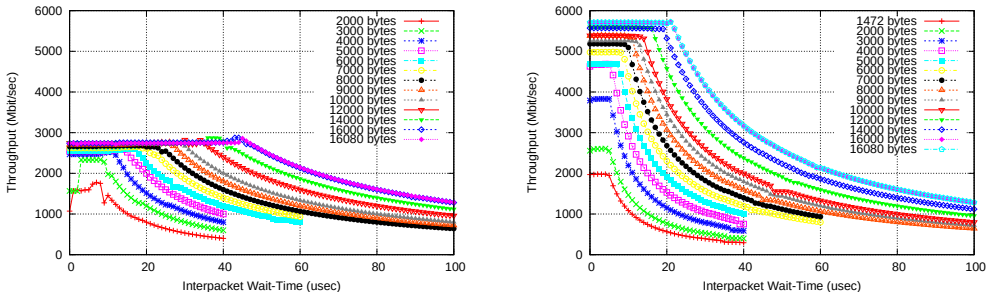
Figure A.10: Throughput performance of 1Gb/sec NIC cards (66Mhz in 64-bit PCI buses).

Figure A.10 shows the throughput achieved by flooding the NIC with UDP packets with various packet sizes and inter-packet times. As with the tests involving latency, identical computer systems were connected back-to-back via their NICs.

Figure A.10 shows the throughput achieved by the same two cards that were used for the latency tests. They show the $\frac{1}{t}$ relation as measured by the `UDPMon` [HJ] program.

It was noticed that as small inter-packet wait times were used, a plateau on the rate at which data is transferred was reached. The length of the plateau of each curve is defined by the amount of time required to put a packet on the wire. More specifically, this is defined by packet size divided by the line rate, i.e. 'standard' IP packets of size 1500B would require $1,500 \times 8/1,000,000,000 = 12\mu$sec per packet. As the packet processing is First-In-First-Out (FIFO), the packets are simply queued if a inter-packet wait time that is less than the physical time required to put the packet on the wire.

Figure A.10 also shows that more stable performance was achieved for low wait times between packets with the Intel card compared to that of the Syskonnect. This was found to be due to high CPU utilisation of the



(a) Supermicro P4DP8-G2 Dual Xeon (b) HP RX2600 Dual Itanium

Figure A.11: Throughput performance of 10Gb/sec Intel NIC cards (PCI-X at 133Mhz).

Syskonnect card and is discussed later.

Figure A.11 shows the performance of the Intel 10GbE LR NIC on the high-end Xeon and Itanium PCs. In order to reduce CPU overheads and to increase the data transfer rates, Jumbo frames were used with packets up-to 16,080B[7] for these tests[8]. The following characteristics can be determined from the comparison:

- The Xeon-based PC is only capable of delivering up-to 2.9Gbit/sec, whilst the Itanium-based PC observed 5.7Gbit/sec. The former limit is due to CPU limitations of the Xeon-based PCs, whilst the latter bottleneck was found to be due to the the theoretical maximum capacity of PCI-X which is 8.5Gb/sec.

- Using a MTU of 2000B cannot achieve anywhere near the optimal performance of larger MTUs.

- The maximal throughput achieved by the maximal MTU does not reach the claimed 10Gb/sec NIC speed.

[7] 16,114B including IP and UDP headers.

[8] As Jumbo frames are non-standard, hardware re-configuration was required on both PCs.

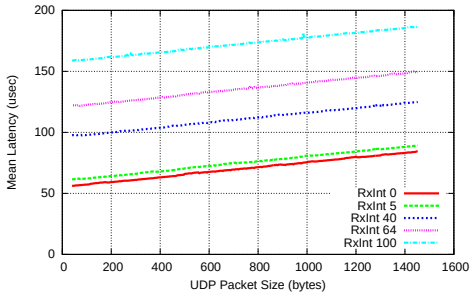


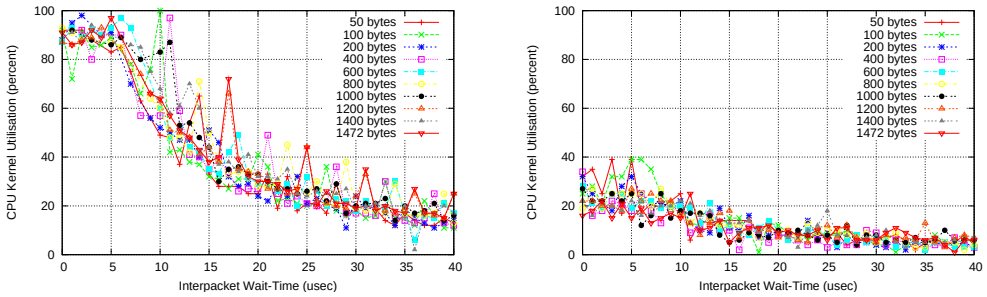
Figure A.12: Effect of Interrupt Coalescing upon packet latency.

A.2.4 Interrupt Coalescing

Depending on the configuration set by the driver, a modern NIC can interrupt the host for each packet sent or received or it can continue to buffer packets for a certain period of time. This is known as *interrupt coalescing* [PJD04] and the details and options are hardware and NIC driver dependent. The NIC may generate interrupts after a fixed number of packets have been processed or after a fixed time from the first packet transferred after the last interrupt. In some cases, the NIC dynamically changes the interrupt coalescence times dependent on the packet receive rate [Cor]. Separate parameters are usually available for the transmit and receive functions of the NIC.

The effect of interrupt coalescing are most apparent upon packet latencies as shown in Figure A.12. It shows that as the packet size is increased, a predictable increase in the latency for each interrupt coalesce setting is achieved. More importantly, higher values of interrupt coalesce result in proportionate increases in the experienced latency.

The effect of a high value of coalescing often lowers throughput as packets could be dropped due to the lack of buffer space. When set too low, high CPU utilisation occurs due to the necessary context switching between kernel processes to absorb/transmit the packet .



(a) Syskonnect (b) Intel

Figure A.13: CPU Kernel utilisation of 1Gb/sec NIC cards (66Mhz in 64-bit PCI buses).

Therefore, to achieve the same throughput with small packet size more interrupts per unit time are required unless interrupts are generated as a function of time. This would explain the decrease in throughput as a result of using smaller packet sizes.

Figure A.13 shows the CPU utilisation of the kernel (networking) thread for the 1Gb/sec Syskonnect and Intel cards as tested in Figure A.10. It shows that the cleaner throughputs achieved by the Intel card also correlate to a much lower CPU utilisation compared to that of the Syskonnect card. This, combined with the high latencies experienced suggests that interrupt coalescing is indeed being utilised by the Intel driver, whilst the high utilisation of the Syskonnect driver affects stable high capacity throughput due to the lack of CPU cycles to service the packets.

A.2.5 Summary

With suitable hardware, it was shown that Network Interface Cards are capable of delivering close to the performance expected. Related work [GB02] also demonstrates the capability of delivering Gigabit rates with suitable

hardware and software configurations.

An important consideration in the deployment of high speed NICs is the CPU overheads associated with the increased rate of packets and the associated rate of hardware interrupts. It was demonstrated that the CPU overhead can be reduced through the use of interrupt coalescing that is available on most NIC drivers. However, this comes at the cost of increased latency of packet receipt (or transfer).

It was also demonstrated that the use of large packet sizes is preferable as it combines the benefit of reducing the number of interrupts generated per unit data and also increases the fraction of data being transferred per packet due to the necessity of encapsulation and the associated protocol headers.

Appendix B

Hardware and Software Configurations

B.1 Systematic New-TCP Tests

	Testbed PC's	Dummynet Router
CPU	Intel Xeon CPU 2.80GHz	
Memory	256MB	
Motherboard	Dell PowerEdge 1600SC	
NIC	Intel 82540EM Gigabit Ethernet Controller	
Kernel	2.6.6 altAIMD-0.6	FreeBSD 4.8
NIC Driver	Linux e1000 5.2.39-k2	Default

Table B.1: Hardware and Software Configuration of Dummynet Testbed PCs router.

	DataTAG	MB-NG
No. Machines	2 × 6	2 × 3
CPU	Intel Xeon CPU 2.2GHz	Intel Xeon CPU 2.0Ghz
Motherboard	Supermicro P4DP8-G2	Supermicro P4DP6
NIC	Syskonnect SK9843	Intel Pro 1000
Kernel	2.4.20 altAIMD-0.3	2.4.20 altAIMD-0.3
NIC Driver	Linux sklin 6.18	Linux e1000 4.4.12
TXqueuelen	50,000	2,000

Table B.2: Hardware and Software Configuration of MB-NG and DataTAG Testbed PCs

B.2 Wide Area Network Tests

B.3 Internet Tests

	Description
CPU	Intel Xeon CPU 2.80GHz
Memory	256 MB
Motherboard	Dell PowerEdge 1600SC
NIC	Intel 82540EM Gigabit Ethernet Controller
Kernel	2.6.5 altAIMD-0.5
NIC Driver	Linux e1000 5.2.39-k2

Table B.3: Hardware and Software Configuration of PC used for Internet transfers.

Appendix C

Network Topologies

C.1 Dummynet

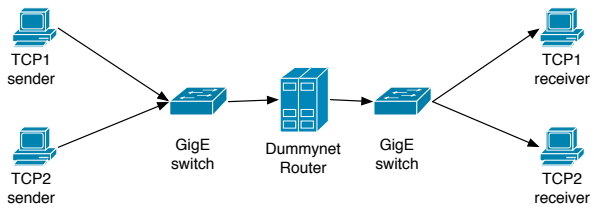
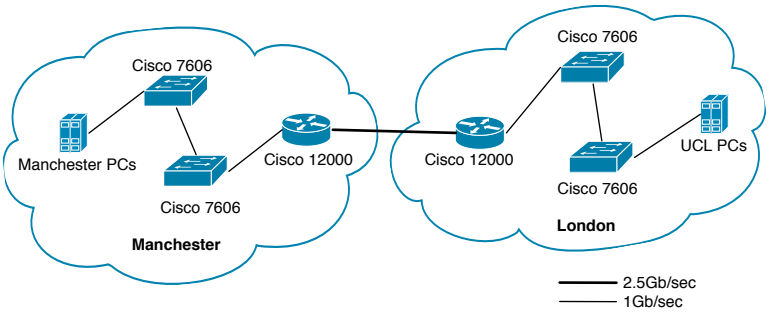


Figure C.1: Topology and Experimental set-up of Dummynet network tests.

C.2 MB-NG

MB-NG part of the Janet Research network run by UKERNA. It is defined by three end points forming a triangle connecting ULCC in London, to



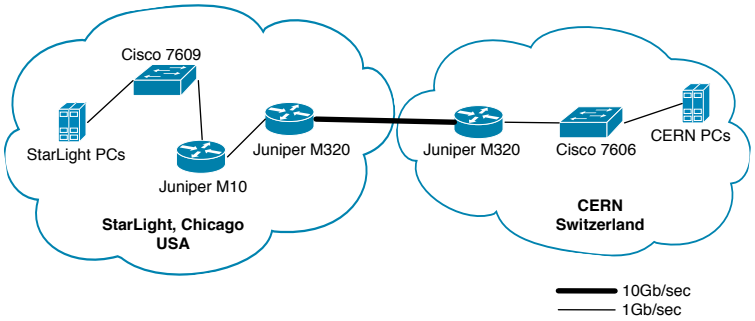
(a) MB-NG Testbed

Figure C.2: Logical representation of the MB-NG Testbeds.

Manchester Computing Centre in Manchester and Rutherford Apple Laboratory in Warrington. For this investigation, is was sufficient to test along only the London to Manchester parts of the MB-NG network. The three end-points are connected via three Cisco 12000 GSR backbone routers. Each site has two Cisco 7600 in which 3 high-end PCs were directed connected onto. A physical bottleneck of 1Gb/sec was provided by a single 1Gb/sec fibre between the two Cisco 7600 co-located at each site.

C.3 DataTAG

DataTAG is composed of a leased fibre connecting CERN in Switzerland to StarLight in Chicago, United States of America. The infrastructure of the DataTAG testbed consists of two Juniper T320 which aggregate the traffic at either sites onto a dedicated STM64 link across the Atlantic Ocean. The traffic from the testbed PCs are connected through Cisco 7600 series routers which are connected to the Juniper T320's at either side of the DataTAG network. As the testbed PCs are limited by their 1Gb/sec NICs, the testbed was configured to provide a bottleneck capacity of 1Gb/sec through a virtual



(a) DataTAG Testbed

Figure C.3: Logical representation of the DataTAG Testbed.

LAN which is diverted through a Juniper M10 router located in StarLight in Chicago.

Appendix D

Network Paths of Internet Network Tests

```
pcgiga.cern.ch:192.91.245.29
cernh4-vlan4:192.91.245.1
cernh7:192.65.185.17
swice2-g3-2.switch.ch:192.65.184.221
switch.ch1.ch.geant.net:62.40.103.17
ch.fr1.fr.geant.net:62.40.96.30
fr.uk1.uk.geant.net:62.40.96.90
janet-gw.uk1.uk.geant.net:62.40.103.150
london-bar4.ja.net:146.97.37.81
po6-0.lond-scr.ja.net:146.97.35.129
po4-0.read-scr.ja.net:146.97.33.74
po2-0.ral-bar.ja.net:146.97.35.158
146.97.40.74:146.97.40.74
192.100.78.2:192.100.78.2
192.100.78.2:192.100.78.2
```

Figure D.1: Internet path as reported by traceroute between CERN and RAL during September 2002.

```
ham02gva.datatag.org:192.91.239.55
r04chi-v-570.caltech.datatag.org:192.91.239.56
192.84.86.210:192.84.86.210
lax-hpr--caltech.cenic.net:137.164.30.225
svl-hpr--lax-hpr-10ge.cenic.net:137.164.25.13
hpr-stan-ge--svl-hpr.cenic.net:137.164.27.162
rtr-dmz1-vlan401.slac.stanford.edu:192.68.191.85
*
iepm-resp.slac.stanford.edu:134.79.240.36
```

Figure D.2: Internet path as reported by traceroute between CERN and Stanford during October-December 2004.

```
ham02gva.datatag.org:192.91.239.55
r04gva-v-570:192.91.239.62
cernh7-r04.cern.ch:192.65.184.38
switch-bckp.ch1.ch.geant.net:62.40.103.181
ch.fr1.fr.geant.net:62.40.96.30
fr.uk1.uk.geant.net:62.40.96.90
uk.ie1.ie.geant.net:62.40.96.137
heanet-gw.ie1.ie.geant.net:62.40.103.230
hyperion-gige5-2.bh.core.hea.net:193.1.195.130
mantova-gige3-1.bh.access.hea.net:193.1.196.174
portia-po1.bh.access.hea.net:193.1.196.218
ham02dub.may.ie:193.1.31.106
```

Figure D.3: Internet path as reported by traceroute between CERN and Dublin during October-December 2004.

```
ham02gva.datatag.org:192.91.239.55
r04chi-v-570.caltech.datatag.org:192.91.239.56
192.84.86.210:192.84.86.210
198.32.11.41:198.32.11.41
snvang-losang.abilene.ucaid.edu:198.32.8.95
snvcr1-pos-abilene.es.net:198.129.248.86
lbl-snv-oc48.es.net:134.55.209.6
lbnl-ge-lbl2.es.net:198.129.224.1
ir1000gw.lbl.gov:131.243.128.210
net100.lbl.gov:131.243.2.93
```

Figure D.4: Internet path as reported by traceroute between CERN and LBL during October-December 2004.

```
ham02gva.datatag.org:192.91.239.55
r04gva-v-570:192.91.239.62
cernh7-r04.cern.ch:192.65.184.38
switch-bckp.ch1.ch.geant.net:62.40.103.181
ch.fr1.fr.geant.net:62.40.96.30
fr.uk1.uk.geant.net:62.40.96.90
janet-gw.uk1.uk.geant.net:62.40.103.150
po2-2.lond-scr3.ja.net:146.97.35.137
po0-0.read-scr.ja.net:146.97.33.38
po3-0.warr-scr.ja.net:146.97.33.54
po1-0.manchester-bar.ja.net:146.97.35.166
gw-nnw.core.netnw.net.uk:146.97.40.202
gw-fw.dl.ac.uk:193.63.74.233
alan3.dl.ac.uk:193.63.74.129
rtlin1.dl.ac.uk:193.62.119.20
```

Figure D.5: Internet path as reported by traceroute between CERN and RAL during October-December 2004.

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