Understanding Asymmetric Links, Buffers and Bi-Directional TCP Performance

by

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Understanding Asymmetric Links, Buffers and Bi-Directional TCP Performance

Thesis directed by Professor Timothy X Brown

Asymmetric uplink and downlink rates are common in many broadband access networks, particularly DSL, DOCSIS, and cellular. With the advent of peer-to-peer protocols and pervasive devices supporting multimedia capabilities, bi-directional TCP traffic has also become commonplace [1]. Under conditions where TCP connections are sending data in both directions simultaneously, performance, as we will show, can be severely degraded.

This thesis seeks to determine how buffers, combined with elementary queueing strategies like FIFO, influence bi-directional TCP performance across asymmetric network links. It also seeks to determine if there is a way to size buffers such that the throughput for a small number of bi-directional connections is optimized.

To answer this question, fundamental system components and algorithms will be reviewed. Following this, research pertaining to observations, optimization techniques and design spaces will be analyzed. Next, a set of real world experiments will be analyzed. Finally, a buffer sizing model will be proposed that optimizes throughput for the bi-directional, long lived, TCP traffic under a FIFO queueing strategy.

Practically speaking, this thesis seeks to determine if current consumer grade customer premise equipment is designed to provide adequate performance for bi-directional Internet traffic across typical asymmetric access technologies like DSL and DOCSIS.

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Introduction

Asymmetric uplink and downlink rates are common in many broadband access networks, particularly ADSL, DOCSIS, and cellular. With the advent of peer-to-peer protocols and pervasive devices supporting multimedia capabilities, bi-directional TCP traffic has also become commonplace [1]. Under conditions where TCP connections are sending data in both directions simultaneously, performance, as we will show, can be severely degraded, in some cases as low as 20%.

This thesis presents an analysis of how packet-based buffers, combined with FIFO queueing, affect bi-directional TCP performance across asymmetric network links. We also go on to present a buffer sizing strategy for these links that provides performance of at least 68% across three different combinations of asymmetric links.

As we will show, the main interaction between bi-directional traffic and buffer size is round trip time (RTT) variability. In TCP, RTT is the cumulative time required to send data and receive a corresponding acknowledgement. This time includes propagation delay, insertion time, processing, and during periods of congestion, queueing delay in buffers. Improperly sized buffers can lead to large amounts of RTT variability, which in turn inhibits the TCP sender's ability to effectively size its CWND to the bandwidth delay product (BDP) of the path. As we will show, these problems occur when buffers are sized too *large*. Our sizing strategy provides a way to estimate a maximum buffer size such that RTT variability is limited, thus allowing TCP sender's to accurately track the

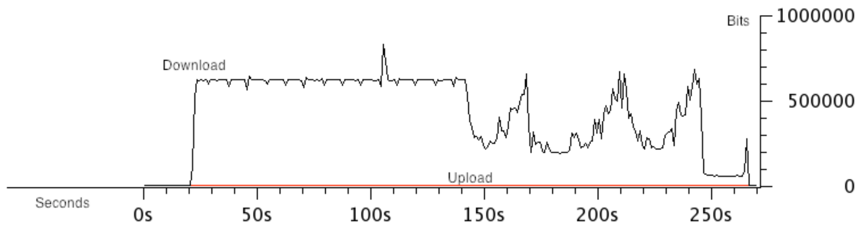
BDP of the path over the lifetime of the connection.

To illustrate the scenario of bi-directional TCP traffic across an asymmetric link with poorly sized buffers, consider the following scenario: A user is streaming a video from Hulu across a typical DSL link. In the middle of the video, the user decides to upload a personal movie clip to Facebook. During the period where the movie clip is uploaded, the quality of the Hulu video is degraded to the point where it is not viewable. We determine why this occurs and then go on propose a method for avoiding it.

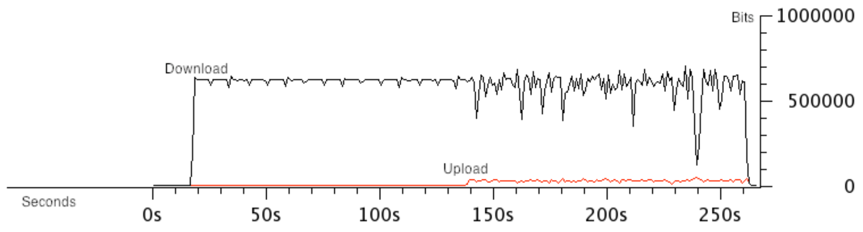
To begin to understand why this phenomenon occurs, we also present this scenario through a graphical representation. In Figure 1.1(a) we show throughput over time for two connections, a download in black and an upload in red. In this system, the uplink and downlink buffers are 50 packets. The upload is launched around time 150. Once the upload is present in the system, the download throughput is degraded by about 50%. We contrast this performance with Figure 1.1(b). Here we show the same scenario, but this time with the uplink buffer sized using our proposed model. When the upload is started around time 150s, the download performance is hardly changed. In this scenario the download throughput does not vary as widely as in Figure 1.1(a). It is clear that uplink buffer size plays a dominant role in determining performance for two-way TCP traffic.

Through these examples, it is clear that this problem has real significance in networks. To understand why this problem occurs, we first characterize the system by defining a set of terms and metrics. We will then then go on to analyze the dynamics, present our buffer sizing strategy, and measure its effectiveness.

Asymmetry occurs in a network when the link rates in each direction of a path are not equal. This asymmetry can take on many forms. We highlight a set of three different configurations as examples. First, a link can be asymmetric if the amount of physical bandwidth allocated to each direction on the link is unequal. This type of configuration is common in access networks served by xDSL technology. In ADSL, links to each subscriber are provisioned with asymmetric levels



(a) 50 Packet



(b) 15 Packet

Figure 1.1: Throughput with 50 and 15 Packet Uplink Buffers for 2.4Mbps Downlink and 196Kbps Uplink

of bandwidth in the downstream and upstream directions. It is common for downstream links to be provisioned with more bandwidth than the upstream. This bandwidth provisioning scheme is deployed to align with a common traffic pattern where large quantities of data are downloaded and small quantities are uploaded. Next, asymmetry can occur across links that are shared. In shared medium, access to the channel is typically controlled by an access control protocol whereby each subscriber is allocated a portion of the total link bandwidth in either direction. Again, it is typical for access to the channel to be asymmetric for the reasons mentioned previously. Examples of this type of link are found in wireless technologies like IEEE 802.16 or DOCSIS. Finally, it is also possible to encounter an asymmetric link where routing protocols or other traffic engineering technologies have selected paths or rate controlled the traffic in downstream and upstream directions such that they operate at different rates. Often times, routing protocols like BGP or OSPF and traffic engineering techniques like MPLS-TE are used to implement these types of paths. It is important to understand in this final example that the asymmetry does not occur because of an asymmetric physical link. It occurs because either the rates allocated to the forward and reverse paths, which can consist of many links, are different.

The level of asymmetry is expressed as a ratio of the downlink and uplink bit rates,

$$\text{Asymmetry Ratio} = C_D/C_U, \tag{1.1}$$

where C_D is the capacity of the downlink and C_U is the capacity of the uplink. It is typical that the downlink operates at a higher rate than the uplink.

Two way, or bi-directional TCP traffic is comprised of two TCP connections sending data in opposite directions. The TCP senders are on opposite sides of the asymmetric link operating over separate sockets. In practice, two distinct pairs of hosts could terminate the TCP traffic in each direction. In the experiments conducted in this research, two individual hosts were used to originate and terminate the two way TCP traffic.

We use the minimum of the normalized goodputs of the TCP connections sending data across the asymmetric link to measure the performance of the system and our buffer sizing model. As we will show later, by changing the size of the uplink buffer, it is possible to manipulate the performance of the system. We choose this metric because it provides fairness to both TCP connections, as compared to other metrics, such as normalized net goodput, that favors one link more than the other based on the level of asymmetry between the downlink and uplink. In (1.2) let G_D and G_U equal goodput for the download and upload respectively. Let C_D and C_U equal capacity of the downlink and uplink.

$$\text{Performance} = \min(G_U/C_U, G_D/C_D) \tag{1.2}$$

Using data collected from the experiments we estimate a maximum buffer size. As we will go on to show in more detail, this estimation model is derived from work by Zhang et al. in [2]. To verify our model's accuracy, we compare the performance realized near the estimated buffer size to the performance of all other measurements in the link configuration set. In all cases, the buffer size estimated through our model yields values for all tested asymmetric link configurations of at least 68%, whereas any larger measured buffer size yields a further decrease of at least 10% and in some cases up to 30%.

1.1 Contestability and Significance

The problem of maximizing performance of bi-directional TCP traffic across asymmetric links still remains widely unsolved [3] [4]. Researchers have taken many different approaches to solving the issue. Some have chosen to focus on end-to-end solutions while others propose localized solutions that are implemented on intermediate nodes. In each of these approaches, researchers use many different methodologies ranging from queueing, to new TCP variants. In any case, a well agreed upon strategy to solve this problem still remains at large.

Regardless of the approach taken, the problem is well known throughout industry. RFC 3449 details performance problems experienced across asymmetric links. It presents strategies for both localized and end-to-end solutions. That is, strategies that can be implemented at intermediate nodes as well as at the hosts. In intermediate nodes, the authors recommend queueing techniques such as random early detection (RED) and acknowledgement segment (ACK) prioritization. At end hosts, sender pacing, compression and window prediction are recommended methodologies to improve TCP performance across an asymmetric link [5].

1.2 Scope

This thesis is written for network designers, researchers and software engineers with sufficient knowledge of basic network concepts including packet switching, the layering or encapsulation model, the TCP/IP protocol suite [6], and basic queueing theory concepts. As we will show later, our experiments focus on TCP/IPv4 traffic, however our solution is independent of the network layer protocol and can support any packet based, connectionless network protocol encapsulating TCP segments. This includes IPv6.

We aim to solve the problem of poor bi-directional TCP performance across asymmetric links by limiting the solution to one that is localized. We further focus our solution towards access networks where asymmetric access links are common, and, as we will show in reviewing prior work, where the majority of potential delay in a network is stored.

Within the localized solution set, we choose to focus on buffering and FIFO queueing versus TCP proxy or a complex queueing strategy for two main reasons. First, choose this strategy because we assume that typical CPE nodes can easily support simple FIFO queueing and adjusting buffers, whereas the other two solutions may be more difficult to implement on current platforms for both technical and economical reasons. We base this assumption on specifications and pricing for typical CPE nodes listed in [7] and [8]. We also argue that because the slow uplink is close to the affected

TCP endpoint, the user will be motivated to resolve this problem; as opposed to the slow uplink being located in the network core and far away from any of the affected users.

We choose a localized approach versus an end-to-end approach because the problem is isolated to asymmetric links and two-way TCP traffic. Because of this and the fact that most end devices running TCP are now mobile and attach to many different types of networks, changing TCP to address this specific problem at the possible expensive of introducing new ones does not make sense.

1.3 Approach

We first analyze the existing literature in detail and develop a simple buffer sizing strategy in Chapter 2. In Chapter 3 we go on to discuss the TCP system dynamics with and without our strategy. To test our strategy we take an empirical approach and conduct a set of experiments on a test network containing two hosts that send data to each other across an asymmetric link. The experiments are organized as follows: We define a set of asymmetric links with asymmetry values of 30, 12, and 3. For each asymmetry configuration, we vary uplink buffer size from 5 packets up to the size of the downlink buffer, which is 50 packets, in increments of 5 packets. For each of the buffer configurations we first launch a download and allow it to run without the presence of an upload for 2 minutes. After the 2 minutes, we launch an upload and allow both the download and upload to run across the network for an additional 2 minutes. Throughout the life of each of the connections, we gather TCP state information using software that we have developed and describe in Chapter 4. This TCP state information is then used to estimate maximum uplink buffer size via our buffer sizing model. We then go on to verify its effectiveness at maintaining a high level of performance compared to all other measured buffer sizes. These results are described in Chapters 5 and 6. We draw conclusions and describe future work in Chapter 7.

Chapter 2

Background and Prior Work

2.1 TCP

TCP is the de-facto standard for reliably transmitting information over IP based networks. The protocol can be classified as a reliable byte-stream based on the concept of a sliding window. The designers created TCP such that it utilizes the maximum available bandwidth in a system, while at the same time sharing it with others. It is assumed that the underlying network is reliable and yields low levels of packet loss. Based on this assumption, any packet loss experienced by TCP is assumed to be due to congestion somewhere along the path between the sender and receiver. To avoid congestion, it also employs adaptive algorithms to control the rate at which information is injected into the network [9].

Three Way Handshake: At the genesis of a TCP connection, the sender and receiver perform a three way handshake to open the connection and negotiate transmission options. The initiator of the connection sends a SYN segment to the receiver. The receiver then responds to the SYN with a SYN-ACK. The initiator follows this with an ACK. During this phase of a TCP connection initial window size, options like maximum segment size (MSS), window scaling, and selective

Table 2.1: TCP State Variables

TCP Variables		
RTT	SSTHRESH	CWND

acknowledgement are negotiated [9].

Slow Start: Slow start is performed following the genesis of a connection. During this phase of the connection, the sender begins transmitting data such that after each received ACK segment, the window is increased by one MSS. This type of increase leads to exponential growth of the sender window [10].

Congestion Avoidance: The congestion avoidance phase of TCP occurs once the CWND has grown larger than the so-called slow start threshold, (SSTHRESH). In this phase TCP is trying to utilize any additional bandwidth available on the link while at the same time avoiding packet loss. Window growth in this phase is controlled via the additive increase algorithm. That is, after every round trip time (RTT), the CWND is increased by one MSS [10].

Fast Retransmission: Fast retransmission is an algorithm to reduce the time it takes TCP to recover from a lost segment. Upon detection of a lost segment by receiving 3 duplicate ACKs, TCP immediately sends the lost segment without waiting for the retransmission timeout (RTO) value to expire [10].

Fast Recovery: Following fast retransmission of the lost segment, TCP remains in congestion avoidance phase and sets SSTHRESH to 1/2 of CWND. Next, CWND is set equal to SSTHRESH + 3·MSS. For each duplicate ACK received, the CWND is incremented by a MSS. If allowed, new data is sent into the network. Finally, when the ACK for the first lost segment is received, CWND is set equal to SSTHRESH [10].

Bandwidth Delay Product: The slow start and congestion avoidance algorithms in TCP are used to accurately identify and track the bandwidth delay product (BDP) of a network path. The BDP can be defined as the number of bytes in transit. The TCP sender window should track this value to keep the utilization of the link maximized:

$$BDP = C \cdot RTT/8, \tag{2.1}$$

where C is the bottleneck link speed in bits per second and RTT is the round trip time in seconds. TCP depends on a steady BDP in order to utilize the link efficiently. In TCP, the congestion avoidance algorithm cannot tolerate large amounts of variability in this value because it only adjusts its window by a single MSS per RTT. As we will go on to show, when large amounts of variable delay are introduced into the network, TCP connections in congestion avoidance mode cannot accurately track the BDP of the link. This poor tracking leads to poor performance. We will carefully examine how buffer size effects BDP tracking and go on to recommend a methodology for sizing buffers such that a TCP can track the BDP accurately.

Selective Acknowledgement: Selective acknowledgement (SACK) allows the receiver to inform the sender that certain amounts of discontiguous data have been received. When the receiver acknowledges discontiguous data it does not advance the window via an ACK. Rather, it simply acknowledges the blocks of data within the current window that have been received by sending a SACK segment. This allows the sender to retransmit only the lost segments and not any duplicate data.

2.2 TCP Phenomena and Performance Impacts

TCP can experience performance problems that result in degraded goodput when fundamental assumptions of its algorithm break down. These include stable RTT, timely arrivals of ACK segments, low packet loss, and constant bandwidth.

ACK Compression: ACK compression is a phenomena observed when the inter-arrival time of ACK segments changes such that periodically a small burst of ACK segments arrives with small inter-ACK spacing. Following this, a longer period of time lapses before another burst of ACKs arrive. This type of phenomenon leads TCP to become bursty, which in some cases can overflow buffers and lead to packet loss. Packet loss in turn, reduces the CWND and slows down transmission of data. ACK compression is induced by variability in RTT. Assuming the path through the network does not change, the primary source of variability in RTT is the amount of buffers within the network.

Variable RTT: A variable RTT leads to a variable BDP. In steady state, or congestion avoidance mode, TCP requires a relatively static BDP to operate at peak performance. If the BDP increases rapidly over time, the TCP window will only grow by a single MSS per RTT. This leads to a scenario where the window size is smaller than the BDP of the link. If RTT changes rapidly in either direction, but remains at the new value for a relatively longer period of time, the sender window will eventually catch up. If the RTT changes periodically, with large variance per period, congestion avoidance in TCP cannot adapt fast enough to changes and ultimately breaks down. This break down, or mis-sized CWND, can lead to inefficient use of the link.

Synchronization and Congestion Collapse: Under certain network conditions, TCP global synchronization can occur. This phenomenon appears when many TCP connections are traversing a link that experiences losses. After the period of packet loss, the TCP connections that lost data become synchronized and perform error recovery algorithms in unison. This in turn leads to a feedback loop whereby each of the TCP connections continues to remain in sync and perform slow start indefinitely. Under slow start, the network periodically receives large waves of segments followed by periods of small waves of segments. This loop produces poor TCP goodput for all the connections traversing the congested link. While, RFC 896 formally acknowledges this phenomenon, modern day TCP variants implement features like fast retransmit and recovery to avoid it [11]. Additionally, drop strategies that we will discuss in Section 2.5 have also been created to avoid this problem.

2.3 Buffers

Packet switched systems require buffers to store packets during periods of processing and congestion within the network element. Process time can be comprised of, among other things, table look-ups, traffic classification, security functions, and queueing and scheduling delays. Congestion occurs as data arrives into a network element faster than it can leave. Buffers play a critical role in preventing packet loss and maintaining efficient throughput for traffic that transits the element.

2.3.1 Architectures

Buffer architectures can take on one of three forms, packet based, byte based or a hybrid of the two. In IP based equipment, buffers are typically implemented in full size IP packets. That is, a buffer is broken up into segments that are 1500B long. This permits packets less than or equal to full size to be stored in the buffer. When packets that are not full size are placed in a queue, the excess memory is sacrificed.

Buffers that are managed in bytes allow for more efficient use of the space. Variable sized packets are placed in queue at variable offsets based on packet size. While the memory efficiency of the buffer is increased under this model, it comes at the cost of added software complexity and the risk of buffer fragmentation. This complexity arises due to the fact that the locations of the beginning and end of packets are not static. Therefore a dynamic list of memory pointers must be maintained.

Finally, a hybrid of the two architectures can be implemented where a buffer is broken up into bins that are larger than a byte, but smaller than a full size packet. Packets are fragmented and placed in the bins and reassembled before being placed in the transmit ring buffer and ultimately onto the physical media.

2.3.2 Components

Buffers have two main components, physical memory and the software that controls the memory. The memory that is allocated to the buffers must be of sufficient speed to match the line rate of the transceivers being used in the network element. Generally, high speed SRAM and DRAM is used as buffer memory. System scalability becomes an issue when line rates begin to exceed memory speeds. Furthermore, as line rates increase, larger amounts of memory can be required, thus leading to increased cost [12].

2.3.3 Sizing

The sizing of buffers in TCP/IP based network elements has evolved with capacity and utilization. It was first proposed that a buffer at a bottleneck link equal to the BDP of the connection was sufficient to keep the connection busy even during periods of loss. When TCP detects a segment loss it reduces CWND by one half. The reduction in window size can cause TCP to stop transmitting if the connection has more data inflight than its current CWND size. Upon receiving the ACK, the window is inflated and a new data segment is sent. During the period of reducing CWND and waiting for the arrival of an ACK, a full RTT has lapsed. Therefore, in order for the link to remain busy during idle periods, a full RTT of TCP data must be buffered. This general rule of thumb was first described in [3].

A buffer following the rule of thumb using a full size IP packet architecture can be described as,

$$B = RTT \cdot C/(8 \cdot 1500), \tag{2.2}$$

where C is the capacity of the bottleneck link, RTT is the round trip time for the TCP connection, and B is the size of the buffer in full size packets.

This sizing strategy works well for small numbers of TCP connections on relatively low bandwidth links. However, when link speeds and the number of TCP connections increase, buffers tend to become too large. That is, the amount of memory in a high speed network element is so large that it becomes a bottleneck in the overall system. Appenzeller et al. discovered that for large numbers of TCP connections across high capacity links, the buffers do not need to follow this general rule of thumb. This is because it is assumed that each connection is sufficiently desynchronized from one other due to other factors in the network such as access technology or end point. Using this new data, buffers are sized for high capacity core routers using small buffers, on the order of ten or twenty packets [13]. This sizing strategy is described as,

$$B = C \cdot RTT/(\sqrt{N} \cdot 8 \cdot 1500), \tag{2.3}$$

for a buffer architected using 1500 byte packets. N in (2.3) is equal to the number of active TCP connections transiting the buffer while C and RTT are equal to the capacity of the link and round trip time of the connection respectively.

2.4 Queueing Strategies

Queueing strategies used in network elements control how classified traffic is placed in memory and how it is removed and sent to an interface for transmission. Although there are many types of queueing strategies, there are four common types that appear in IP networks. Each strategy has strengths and weaknesses that are exposed based on utilization and underlying technologies [14].

FIFO: FIFO, or first-in first-out, is the simplest of queueing strategies. As each packet enters the network element, it is placed in an outgoing queue bound to the appropriate interface on the device. The queue is emptied in the order of which packets were queued. This type of queueing works well when the size of the buffer is small with respect to time and when there are not strict quality of service requirements for the different types of traffic transiting the queue [14].

Priority Queueing: Priority queueing is a strategy whereby classified traffic is serviced from a queue before all other queues until the queue is empty. This strategy was designed to support delay sensitive traffic such as VoIP. Priority queueing is typically used across converged networks where bearer traffic, signaling traffic and data traffic all share the same infrastructure and compete for resources [14].

Fair Queueing: Fair queueing strategies combat the side effects of a large FIFO queue. Instead of servicing the queue of packets in the order of arrival, fair queueing inspects each packet and establishes a set of flows. Each flow represents the exchange of data between two unique IP addresses. Each flow is serviced such that during periods of congestion level of fairness is achieved by sending packets from each flow regardless of where they located in the queue. Large buffers can be used with fair queueing to hold data for many flows while at the same time not inducing large amounts of latency for any of the connections [14].

Class-Based Weighted Fair Queueing: Class-based weighted fair queueing (CBWFQ) builds upon fair queueing by establishing multiple classes of traffic. Each traffic class is given a queue serviced using fair queueing. Each class is also assigned a percentage of the available bandwidth such that a scheduler removes packets from each class in a weighted fashion. The selected packets are then dispatched to a transmit ring buffer where they are placed on the physical media [14].

2.5 Drop Strategies

Tail Drop: Tail drop is the simplest of all strategies. It simply drops packets that do not fit into the queue. For TCP based traffic, this can lead to the previously discussed effect called global synchronization. This phenomenon leads to congestion collapse in a network [14].

Random Early Detection: Random early detection algorithms (RED) were created to reduce the occurrence of global synchronization. The algorithms are designed to selectively drop packets from different TCP connections traversing the network link at different times based upon queue length and a utilization threshold. This type of drop strategy preemptively drops segments from different connections at different times before and during periods of congestion to avoid global synchronization. Global synchronization is avoided using this type of drop strategy by dropping packets from each TCP connection at different times [14].

2.6 Prior Work

Extensive research has been conducted to analyze the effects of buffer size on TCP throughput across a variety of networks including symmetric, asymmetric and rate controlled links. The research can be classified into five main areas, observation; queueing and dropping, TCP layer inspection and adaptation, host based techniques and finally, buffering techniques. This section summarizes this work.

Zhang et al. detail the traffic patterns and TCP characteristics of two way traffic across a bottleneck link. They define synchronization according to when the CWNDs grow for each connection. Connections that are synchronized in-phase have windows that grow during the same time period, while out of phase, or asynchronous, connections have windows that grow while the opposite connections window shrinks. This research identifies ACK compression, synchronous and asynchronous congestion patterns and buffering methodologies that reduce the goodput of the flows. Specifically, the authors make note that under two-way flows across a bottleneck link, common characteristics on either side of the link are asynchronous. Buffers fill, windows increase and decrease as do RTT values reciprocally. Furthermore, the authors identify that increasing buffer size under two-way flows does not improve performance, rather, it degrades performance [2].

They also state that asynchronous and synchronous phases can be estimated using:

$$W_D \;\leq\; W_U + 2 \cdot BDP \;\; \text{(in phase)} \tag{2.4}$$

$$W_D \;>\; W_U + 2 \cdot BDP \;\; \text{(out of phase)} \tag{2.5}$$

where W_D and W_U are the fixed congestion window sizes for the download and upload respectively. When (2.4) holds true, the connections are out of phase and use the downlink most efficiently since they do not simultaneously send at full rate. When (2.5) holds true, the connections are in phase and the downlink is not utilized as efficiently. We will build upon this work to construct a model for sizing packet based buffers.

As shown by Zhang et al., and further described in [15], goodput degradation occurs for two-way TCP traffic across any poorly sized bottleneck link buffer. The authors derive efficiency relationships between the two flows that model utilization of the link in either direction. The drop in goodput is stated to be a result of ACK compression, where large numbers of ACK segments are queued behind large numbers of data segments. Ultimately they show that TCP connections traversing rate-controlled channels see better performance than unshaped channels because smoother inter-ACK spacing is achieved. Rate-controlled links are ones that employ the use of a buffer to store traffic that arrives faster than it can be sent. By buffering bursts of traffic exceeding the outgoing rate, the transmission of packets is smoothed. A rate-limiter simply drops packets that exceed a pre-defined rate. While the authors show that rate-controlled traffic reduces the effects of ACK compression, they do not address of the problem of ACK compression itself. As we will show next, new research conjectures that ACK compression is a byproduct of poorly sized link buffers.

In [4], Heusse et al. observe that there is a dynamic relationship and efficiency ratio between bi-directional TCP flows passing across an asymmetric link. The efficiency ratio is simply $CWND_D/CWND_U$ of the two flows, where $CWND_D$ is the CWND of the download and $CWND_U$ is the CWND of the upload. Furthermore, the authors state that ACK compression is not the reason for poor goodput across an asymmetric link. Rather, the true reason for poor goodput is directly related to the size of the buffers on either end of the asymmetric, bottleneck link. Additionally, it is conjectured that a buffer architected in bytes provides better performance than a buffer architected in packets. We build upon these findings to construct a model for sizing packet based buffers.

To combat the side effects of poorly sized buffers and ACK compression, [16] proposes three methodologies for improving two way throughput across asymmetric links. The first two strategies focus on drops and queueing while the last re-writes TCP headers in certain flows. The first strategy discussed, named ACK congestion control, uses RED on both data and ACK segments of TCP connections. As a second strategy, the authors propose ACK filtering. This strategy drops certain ACK segments to limit the constant throughput of the ACKs to be below a certain value. Because ACK segments are cumulative, lost ACK segments do not impact throughput. This strategy is similar to the delayed ACK strategy implemented at the receiver. Finally, TCP sender adaptation is proposed. Under this proposal, the sender window is grown or reduced based on the number of ACK segments received, not the number of bytes acknowledged. While this set of solutions does in fact increase performance of the system, it requires complex software implementations at intermediate nodes on either side of the asymmetric link. We avoid these types of solutions due to the complexity and requirements it poses on CPE nodes.

Another queue based approach to the asymmetric link problem is described in [17]. Here the authors propose an adaptive class-based queueing strategy that employs separate queues for ACK and data segments. Queue size, scheduling and servicing of each queue are dynamically adjusted based upon an algorithm that estimates the throughput rates of the TCP flows in either direction. Because of the intelligent queueing and drop algorithms implemented in the network element, buffer sizes are not as relevant and are made arbitrarily large, and symmetric on either side of the asymmetric link. Again, we steer away from a complex queueing based approach to solving the problem because of software complexity and increased hardware requirements at CPE nodes.

While queueing and drop strategies have been shown to be effective at improving goodput under the asymmetric bottleneck link scenario, they are not the only methods that have been developed. In [3], Villamizar and Song discuss a strategy for optimizing throughput across high speed WANs by adjusting the window size of TCP receivers and senders. By adjusting the window to reflect the BDP product of the link, the link can be filled to capacity and optimum goodput can be obtained in both directions. They make note of conditions where this adjustment is not optimal, particularly in situations where bottleneck links do not have enough buffering available. They show that inadequate buffering at these bottlenecks leads to congestion collapse. To steer around this problem, a tradeoff is proposed where window sizes are set to values slightly less than the BDP. In addition to these recommendations, the authors also declare the original rule of thumb for sizing buffers as mentioned earlier and shown in equation (2.2). This solution solves the problem at the expense of limiting the host's throughput capabilities on other higher speed networks. While the host based adjustments proposed in this work do not lend themselves to mobility, the buffering recommendations provide the foundation for further research. We go on to recommend a more precise sizing recommendation for asymmetric bottleneck links.

As much of the research discussed so far has shown, bi-directional TCP flows across asymmetric links interfere with one another. Kalampoukas et al. conjecture that interference is due to inducing large amounts of queueing delay not only in intermediate nodes, but in the hosts themselves [18]. They propose a set of three strategies to minimize the queueing delay. First, a queue dedicated to ACK segments and serviced with a higher frequency than other queues is developed. Second, a back pressure algorithm is proposed where the TCP sender is signaled to stop sending data based upon IP queue size at the sender. Finally, they propose connection level bandwidth allocation similar to the resource reservation protocol (RSVP) [19]. While these strategies are all possible ways to improve performance, the theory that interference arises at the hosts themselves is not entirely correct. Their conjecture holds true when the bottleneck link is directly attached to the host. When the bottleneck link is somewhere more than one hop away, their theory does not hold true. This is because regardless of how the data segments and ACK segments are queued in the host, they are transmitted at a rate faster than the bottleneck link. Furthermore, this research only addresses the use case where the TCP connections are originated and terminated on the same two hosts. Under scenarios where the TCP connections are spread across different hosts, this conjecture does not apply.

In [20], Appenzeller et al. investigate the effects of buffer size on TCP goodput in a network with a large number (100,000) of TCP connections. Through simulation, the authors show that traditional buffer sizing compared to small sized buffers, offer similar throughput. Additionally, the authors note that when the edge link speed is increased and core buffer sizes are decreased, unfairness emerges between connections traversing the core and those that do not. In this thesis, we examine buffer sizing in asymmetric links at the edge of the network for a small number (2) of TCP connections. We also assume that all connections transit a core to reach their destinations. Finally, this work does not consider asymmetric links at the edges of the network. This work does assist in validating the assumption that RTT variability lives at the edge of a network. This is because the researchers show that for links carrying large numbers of TCP connections, both large and small buffers yield the same amount of goodput.

In [13], Vishwanath et al. review recent work on the sizing of buffers carrying TCP traffic. As previously mentioned, it was discovered that core router buffer sizes could be reduced from the general rule of thumb sizes to a fraction of this size and still maintain high levels of link utilization when switching large numbers of long lived TCP connections. Additional work has shown that further reducing the size of router buffers via what has come to be known as the *tiny model* reduces link utilization to 80%–90%. Under some conditions, primarily with optical network devices operating at 40Gbps and higher, this level of utilization is acceptable. We use these findings as part of the foundation for our assumption that RTT variability lives at the edges of a network.

The authors in [21] describe the tradeoffs between buffer size and two types of TCP behavior, bursty and smooth. Bursty behavior in TCP occurs during slow start when the window size grows quickly and when ACK compression occurs. Smooth behavior occurs during the congestion avoidance phase when ACK segments are transmitted at a constant rate and arrive at the sender evenly spaced. Based upon either phase, the authors note that buffer size plays an important role in the overall stability of the system. It is determined that with small buffers, smooth TCP traffic is more prone to becoming bursty when compared to large buffers. This may seem to contradict this thesis, however it does not address two-way traffic, nor does it examine TCP behavior across links that are asymmetric and that carry smaller numbers of TCP connections. In addition, the authors do not define how much TCP traffic is transiting the link, nor do they define what large and small buffers are relative to link rates. In our research we define these characteristics and go on to show appropriate sizing of buffers based on link rates and TCP window sizes.

Bursty TCP traffic has been shown to produce negative effects on goodput in networks with small buffers. In order to reduce this effect during slow start the authors in [22] propose a scaled pacing algorithm. By limiting the growth rate during this initial phase of TCP, the size of the bursts can be reduced, thus limiting the amount of loss experienced in the network, and potentially allowing TCP to estimate the BDP faster. The scaling factor controls the growth of the CWND by reducing how fast it grows. The paper goes on to show that this approach to window growth out performs all TCP variants under single and multiple flows when buffers are small in the network. As we have mentioned previously, we avoid an end-to-end approach to resolving this problem because the adjustments to TCP have the potential to introduce different performance problems in other types of networks.

As the Internet was extended to mobile devices operating over cellular networks, the challenge of optimizing TCP over lossy links emerged. TCP Westwood is a newer variant of TCP that aims to improve TCP goodput over this type of link [23]. To achieve the improved gooput, new RTT estimation algorithms are developed and used to provide what the authors deem a “faster recovery”. Communication theory and filter design are used to develop an RTT estimation algorithm that is used in turn to accurately estimate the true BDP of a link over time. When a loss occurs Westwood uses this algorithm to adjust the CWND at the sender rather than simply cutting it in half. They go on to show that using this augmentation provides significant increases in goodput over lossy links when compared to traditional TCP variances like Reno and New Reno [23].

To verify the performance of Westwood over other types of links the authors in [24] detail the mathematical analysis, simulations and experiments that compare and contrast the performance characteristics of TCP Reno and Westwood across symmetric links with varying buffer sizes. They go on to show that Westwood achieves a better goodput regardless of buffer size compared to the Reno variant.

The research in [23] and [24] focuses on lossy channels and 10Gbps symmetric core links. The research also excludes two-way TCP traffic from analysis. We focus our work on edge networks with asymmetric access links operating at typical broadband access rates.

The authors in [25] show that by using a coarse retranmission timeout (RTO) estimator, goodput can be increased over networks with variable delay. According to their research, the primary reason for this performance increase is because a coarse RTO allows TCP to stay out of the slow start phase during periods of increased congestion and delay. By remaining in congestion avoidance mode, the sender window can remain large and continue to fill the network path with data. This is presumably because the RTO value remains at a larger value and as a result, produces less timeouts.

A hybrid queueing and TCP proxy methodology for dealing with variable delay in 3G networks is presented in [26]. The strategy accounts for varying channel conditions and the corresponding impacts on goodput for both short and long lived TCP connections. The system improves goodput for short TCP flows by reducing latency using a priority queue while for long flows, by adjusting CWND values. Long and short flows are identified by performing higher layer protocol inspection and statically defining certain protocols as long or short lived.

Extensive analysis of real world Internet traffic has been performed by a number of researchers. In [27], eight hours of packet traces containing Internet traffic are analyzed. The analysis produces a set of descriptive statistics showing that the median RTT for Internet TCP traffic is 110 milliseconds while the range spans from less than a second to over 200 seconds. This data was gathered using a capture device running tcpdump placed on the edge of the network. This information is useful for determining appropriate buffer sizes in edge network elements.

This research has shown that TCP goodput is a function of many different network characteristics. However, these characteristics can all be classified into three main categories, goodput, variable delay and packet loss. Table 2.2 summarizes this work and shows that while a significant amount of research has been conducted in the area of TCP performance, only a single paper addresses the exact set of requirements for this thesis. We build upon the work in [4] and, as mentioned previously, focus only on packet based buffers. The remainder of this thesis aims to study the effects of variable delay induced by two-way traffic, buffers and asymmetric links on performance.

Table 2.2: Prior Work Summary

Paper	Type of Problem	Type of Solution	Network Block	Link Type	Traffic Type
[13]	Goodput	Observation	Core	Symmetric	Unidirectional
[20]	Goodput	Buffering	Core	Symmetric	Unidirectional
[21]	Goodput	Buffering	n/a	Symmetric	Unidirectional
[22]	Goodput	TCP Adaptation	n/a	n/a	Unidirectional
[23]	Loss	TCP Adaptation	Access	n/a	Unidirectional
[25]	Delay	TCP Adaptation	n/a	n/a	Unidirectional
[26]	Loss	TCP Inspection and Adaptation	Access	Symmetric	Unidirectional
[3]	Goodput	TCP Adaptation, Buffering	Access	Symmetric	Bi-directional
[2]	Observation	n/a	Access	Symmetric	Bi-directional
[15]	Goodput	Queueing	n/a	Symmetric	Bi-directional
[17]	Goodput	Queueing	Access	Symmetric	Bi-directional
[27]	Observation	n/a	Access	Symmetric	Bi-directional
[16]	Delay	Queueing, TCP Adaptation	Access	Asymmetric	Bi-directional
[18]	Delay	Queueing, TCP Adaptation	Access	Asymmetric	Bi-directional
[4]	Goodput	Buffering	Access	Asymmetric	Bi-directional
This Thesis	Goodput	Buffering	Access	Asymmetric	Bi-directional

Chapter 3

System Dynamics

The dynamics of bi-directional TCP in an asymmetric system have four components that contribute to the goodput realized by either connection; RTT, CWND, link asymmetry, and buffering. The rate at which RTT and CWND grow are related to the size of the buffers in the system.

Consider the system where there are two hosts, A and B that are connected to each other as shown in Figure 3.1. Let B_{R1} represent the link buffer in R_1 connecting R_1 to R_2. Let L_{12} represent the link connecting R_1 to R_2. Let L_{21} represent the link connecting R_2 to R_1. Let B_{R2} represent the link buffer in R_2 connecting R_2 to R_1. Let TCP_{AB} represent a TCP connection where host A is sending data to host B. Let TCP_{BA} represent a TCP connection where host B is sending data to host A.

The remainder of this thesis will reference this system and corresponding connections and buffers.

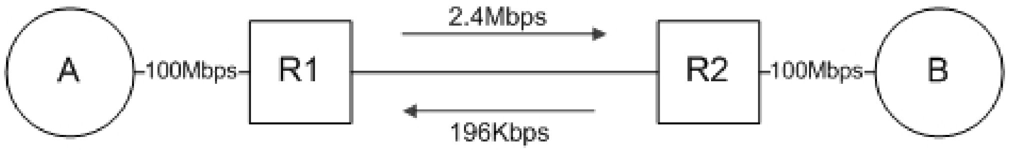


Figure 3.1: Test Network

3.1 Round Trip Time

The round trip time in TCP is measured and statistically smoothed over time. This measurement, in conjunction with the advertised receive window, controls how fast the sender window grows. As previously reviewed, when TCP is in congestion avoidance mode, the window grows additively by, at most, a single MSS per RTT. Said another way, the sender window grows by, at most, a single MSS every time an ACK is received. RFC 2581 specifies that the sender window grows additively at the rate of $MSS \cdot MSS/CWND$ for every ACK segment received. This equates to the window growing by roughly a single MSS per RTT [10]. This implies that large increases in RTT can slow down the growth of the window.

3.2 Windows

Window size controls how much data is injected into the network during an RTT. If the window is large and the sender injects a large amount of data into the network such that congestion occurs, link buffers can be filled. Depending on how large the buffer is and how large the window is for the opposite connection, ACK segments might be dropped. If all ACK segments are dropped or delayed long enough in a single RTT, the corresponding TCP connection will detect packet loss through retransmission timeout (RTO) expiration and enter slow start.

3.3 Asymmetry

Asymmetry has many affects on bi-directional TCP. If asymmetry is large, the ACK stream traversing the slower uplink can overwhelm the link. Furthermore, as we will show, either TCP connection can periodically control the link and induce packet loss on the other. That is, one connection can fill the buffers in the network and force the opposite connection's packets to be dropped.

3.4 Buffering

Buffering controls the amount of RTT variability in the network. Large buffers imply that there is a large amount of RTT variability within the system. An important, somewhat counterintuitive point to note is that a TCP connection will only fill link buffers in one direction, assuming that the opposite link operates at a rate faster than the resulting stream of ACK segments generated by the receiver.

In many systems, including the test system described in this thesis, buffers are allocated on a per packet basis. In this case, an N packet buffer is full with N packets regardless of their size. That is to say a 1500 byte packet and 50 byte packet both occupy the same amount of space in a packet based buffer.

3.5 Challenge of Asymmetric Links

Consider the following scenario as depicted in Figure 3.2. In step 1, TCP_{AB} is sending data in congestion avoidance mode. B_{R1} contains some TCP_{AB} data segments while B_{R2} is completely empty because the ACK segments generated by host B arrive at a rate slower than the L_{21}.

In step 2, host B creates TCP_{BA} and begins sending data. During the slow start phase of TCP_{BA}, a large amount of data is injected into the network, filling B_{R2}. If B_{R2} is sufficiently large, a number of the TCP_{AB} ACK segments are either delayed or dropped while B_{R2} empties. Subsequently, the RTT and BDP of TCP_{AB} is increased. Because TCP_{AB} is in congestion avoidance, its window can only be inflated by a single MSS per, the now much larger, RTT. Therefore, TCP_{AB}'s window cannot be inflated to a value large enough to compensate for the increase in BDP, leading to a decrease in goodput.

In steps 3 and 4, when B_{R2} is sufficiently large, B_{R1} is emptied and the downlink becomes

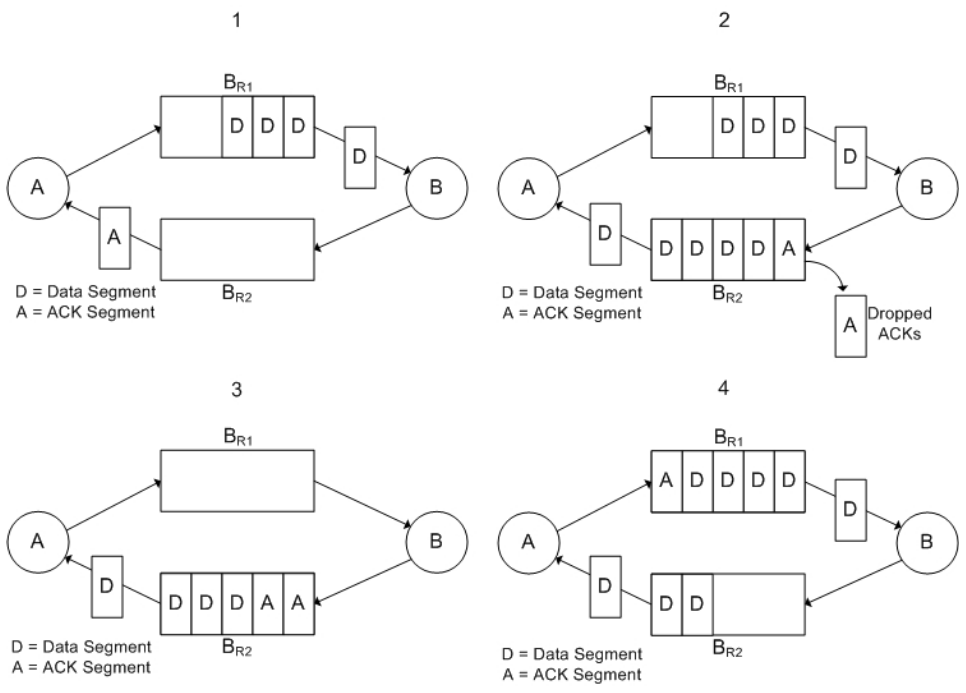


Figure 3.2: Sequence of Events

idle. Following this, a burst of ACK segments are delivered to TCP_{AB} after the data segments in B_{R2} are emptied, which in return, causes TCP_{AB} to emit a burst of data segments and its CWND to grow. At this point, B_{R1} is filled with data segments from TCP_{AB}, while at the same time delaying ACK segments for TCP_{BA}.

After B_{R1} empties, the ACK segments for TCP_{BA} are delivered in a burst. As a result, TCP_{BA} emits a burst of data segments, filling B_{R2}, thus causing the same sequence of events to repeat, beginning at step 3. It is important to note that the BDP of each link changes periodically from relatively small values to large values. This rate of change is faster than the congestion avoidance algorithm can change. As a result, the BDP of each link cannot be tracked accurately.

This phenomenon has been described in [2] and [15]. As we will see, this sequence is exaggerated for asymmetric links with poorly sized buffers.

This sequence of events is undesirable, cannot be avoided and more importantly, can occur in real networks. It is easy to imagine a DSL user watching an online video over a TCP connection and then sending a large email, subsequently initiating this sequence of events. To minimize the affects this scenario has on the corresponding TCP connections, the system must be designed such that appropriate boundaries are placed on the RTT of either connection.

Chapter 4

Experiments

4.1 Introduction

In order to gain a detailed, empirically based understanding of TCP dynamics across asymmetric links, a test network was created. The network consisted of two hosts and two routers connected via an asymmetric bottleneck link as shown in Figure 3.1. A client-server TCP program was developed to gather TCP state information over the course of a connection. Using this test network and software, three sets of experiments were conducted on asymmetric link configurations with asymmetric ratios of 30, 12, and 3. For each asymmetric link configuration, a series of ten experiments were conducted where B_{R1} was fixed at 50 packets and B_{R2} was varied in intervals of 5 packets, starting at 5 and leading up to a maximum of 50. All packets were captured on each

Table 4.1: Experiment Configuration

Experiment	R1 Buffer	R2 Buffer
1	50	5
2	50	10
3	50	15
4	50	20
5	50	25
6	50	30
7	50	35
8	50	40
9	50	45
10	50	50

Table 4.2: Equipment Setup

Node	Make	Model	Software
R_1	Cisco	2821	IOS 12.4(24)T2
R_2	Cisco	2811	IOS 12.4(24)T2
A	Apple	Macbook	OS X 10.6.2, VMWare Fusion 3.01, FreeBSD 7.2
B	Apple	Macbook	OS X 10.6.2, VMWare Fusion 3.01, FreeBSD 7.2

host using tcpdump. TCP state variables were probed and written to a log file for post processing and analysis of each experiment run. This setup is detailed in the following sections.

4.2 Network and Host Equipment

The test network was configured using hardware and software listed in Table 4.2 as shown in Figure 3.1. R_1 was a Cisco 2821 router. R_2 was a Cisco 2811 router. Both were running IOS release `12.4(24)T2`. R_1 and R_2 were configured to shape traffic according to Table 4.2 via a 100Mbps Ethernet link. The Ethernet connections to hosts A and B operated at 100Mbps. Hosts A and B ran the FreeBSD version 7.2 operating system.

4.3 Router Buffer and Queueing Configuration

R_1 and R_2 were configured with traffic shapers using FIFO queueing to limit the Ethernet interfaces to the committed information rates previously mentioned in Table 4.2. The shapers on R_1 and R_2 were configured with packet-based buffers. The routers used a token bucket shaping algorithm where the buffer was serviced by the scheduler for a certain burst of packets. The burst

Table 4.3: Traffic Shaper Configuration

CIR Kbps (Downlink/Uplink)	B_c Bits (Downlink/Uplink)	T_c msec (Downlink/Uplink)	Asymmetry Ratio
3000/1000	12000/4000	4/4	3
2400/196	9600/784	4/4	12
3000/100	12000/400	4/4	30

Table 4.4: TCP Options

Node	TCP	Options
A	Reno	SACK, Delayed ACK
B	Reno	SACK, Delayed ACK

of packets was then sent across the link at 100Mbps. The periodic bursts transmitted at 100Mbps average to the committed information rates of each link over the burst time interval. That is, for every time interval, a fixed number of packets are sent according to $CIR = B_c/T_c$, where CIR is the committed information rate, B_c is the committed burst in bytes, and T_c is the time interval. The values for T_c and B_c are calculated internally by the router's shaper algorithm based on physical link speed and the desired CIR [28].

4.4 TCP Configuration

Hosts A and B were configured with the same TCP configuration and options as listed in Table 4.4. All TCP settings were left at default values except for the inflight option [29]. The inflight option was disabled after boot time using the sysctl utility in FreeBSD. Inflight configures TCP to operate using a TCP Vegas style variant by placing limits on maximum window size, this actually inhibits performance due to the algorithm placing limits on maximum size. Under our scenario, as we will show, the window must actually be set larger, not constrained.

4.5 Software

A client-server TCP application was developed to log TCP state information [30]. The server application simply listens on a port and absorbs segments from the client. The client application,

Table 4.5: Collected TCP Variables

RTT	RTT Variance	SSTHRESH	CWND

or TCP sender, opens a socket and emits full size data segments. After each write to the TCP socket, the client application queries the kernel and gathers TCP state variables listed in Table 4.5 via the `TCP_INFO` data structure. With each query, a timestamp with microsecond resolution is sampled and recorded. In addition to state variables, throughput is calculated based on the total number of bytes written to the socket divided by the total elapsed time.

The client algorithm below details the order of operations used to record TCP state information from the client.

```
while ($true$)
    Record time
    Send data
    Get TCP state information
    Calculate throughput
    Write time, state variables, and goodput to log
```

4.6 Time Synchronization

R_1 was configured as an NTP master server [31]. R_2 was then configured as a peer. Hosts A and B used R_1 and R_2 as NTP servers respectively. This allowed for relative time synchronization such that TCP state variables could be analyzed accurately for each connection at the same time. This configuration also minimized the majority of NTP traffic from traversing the asymmetric link [32].

4.7 Post Processing

Each log file was comprised of a number of tab delimited lines; each containing a timestamp, and values for RTT, RTT variance, SSTHRESH, CWND, receiver window, sender window, and goodput. The log files for hosts A and B under each run of the experiment were fed into three Python scripts [33]. The scripts computed a number of values that allowed each connection to be characterized over time. These values were then run through Gnuplot for observation and further analysis.

4.7.1 Averages and Ratios

For each run of the experiment, averages of TCP characteristics and throughput for the two connections were generated using Python scripts. For the download sourced from host A, averages were generated for the entire connection and for the slice of time where both TCP_{AB} and TCP_{BA} were active.

Ratios for CWND and throughput were calculated for each buffer size using the previously calculated averages of the portion of the experiment where both TCP_{AB} and TCP_{BA} were sending information.

4.7.2 Tcpdump, Wireshark, Mergecap

Tcpdump was run on each host during each experiment. Tcpdump was configured to capture the first 100B of each frame on the network interface card. After each experiment was finished, mergecap was run on the capture files from hosts A and B capture files to produce a single pcap file containing both sessions [34]. Using the merged file, plots of goodput over time could then generated for TCP_{AB} and TCP_{BA} across a single time axis using Wireshark.

4.7.3 Gnuplot

Gnuplot was used to generate all the plots used to explain the interactions between the TCP connections and buffer sizes in the network [35].

Chapter 5

Results

5.1 Observations

Figure 5.1 shows performance for each asymmetric link combination. It is clear that as buffer size increases, performance decreases for all asymmetric link configurations. The remainder of our discussion in this chapter we will focus on the asymmetric link configuration of 12 where the downlink operates at 2.4Mbps and the uplink operates at 196Kbps. Figures 5.2, 5.3, 5.4 and 5.5 clearly display the goodput degradation of TCP_{AB} and marginal goodput increases in TCP_{BA} as uplink buffer increases. This behavior is due to TCP_{AB} sizing its window assuming a slowly changing BDP value and meanwhile TCP_{BA}, in conjunction with the size of B_{R2}, adding large periodic changes in RTT. The fluctuations in RTT result in BDP changes such that under larger sizes of B_{R2}, the BDP tracking characteristic of the congestion avoidance algorithm in TCP_{AB} cannot adapt fast enough to reflect the actual BDP of the link. This leads to the poor goodput of TCP_{AB}. The concepts are discussed in more detail below.

5.1.1 Connection Synchronization

Connection synchronization for the experiments is shown in Figures 5.6 and 5.7. For small sizes of B_{R2}, the connections remain out of phase. As B_{R2} is increased, the synchronization between TCP_{AB} and TCP_{BA} drifts from out-of-phase towards in-phase. Figures 5.6 and 5.7 display a detail

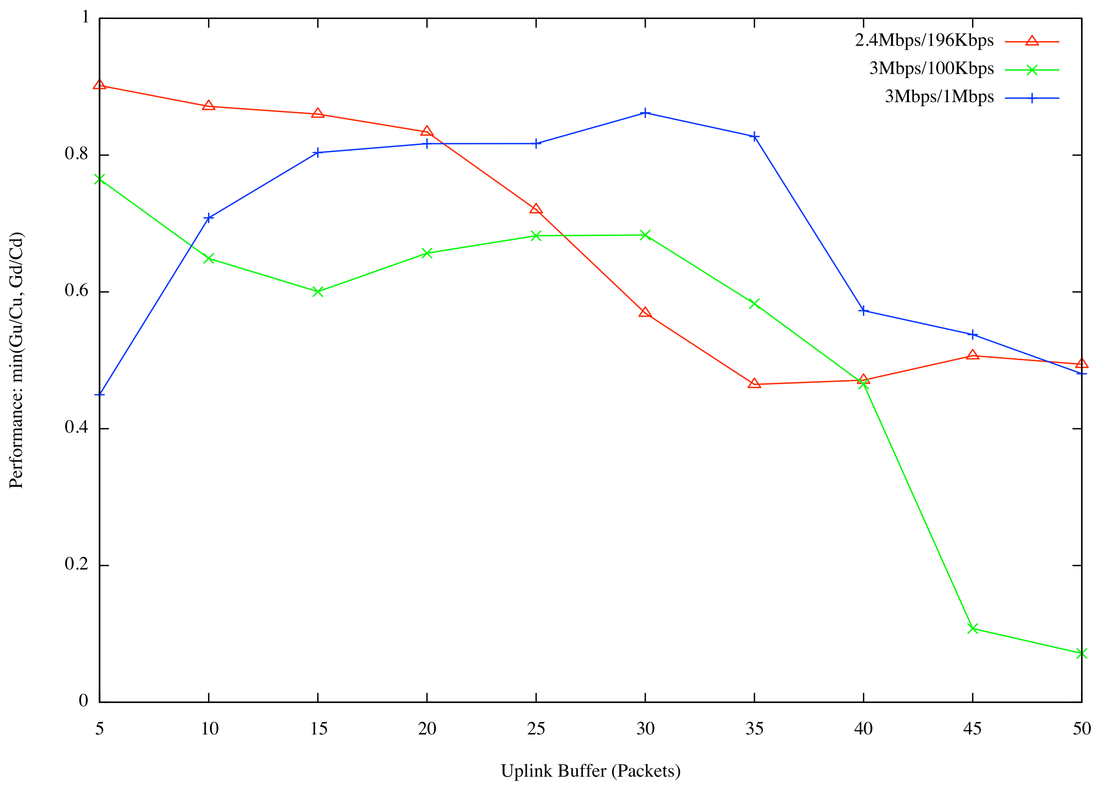


Figure 5.1: Performance for Asymmetric Link Combinations of 3, 12, 30

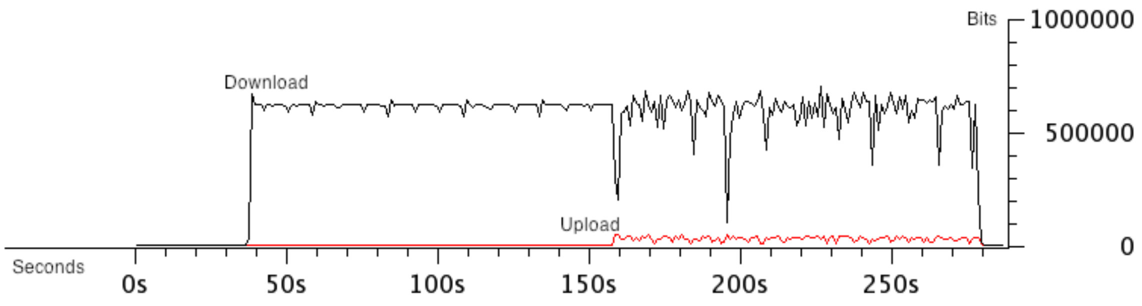


Figure 5.2: Throughput with 5 Packet Uplink Buffer for 2.4Mbps Downlink and 196Kbps Uplink

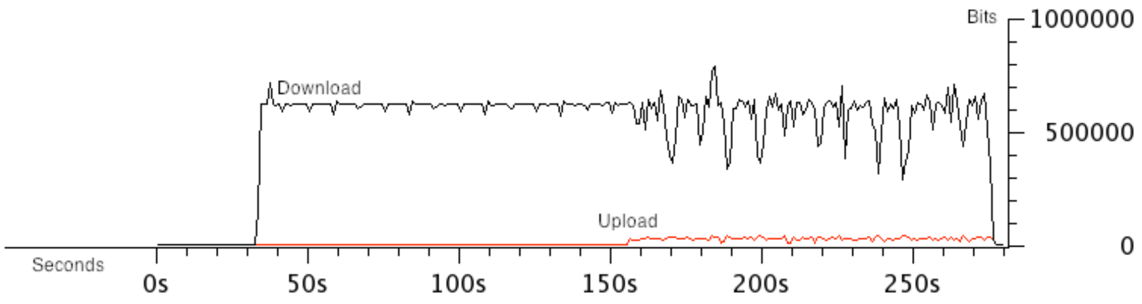


Figure 5.3: Throughput with 20 Packet Uplink Buffer for 2.4Mbps Downlink and 196Kbps Uplink

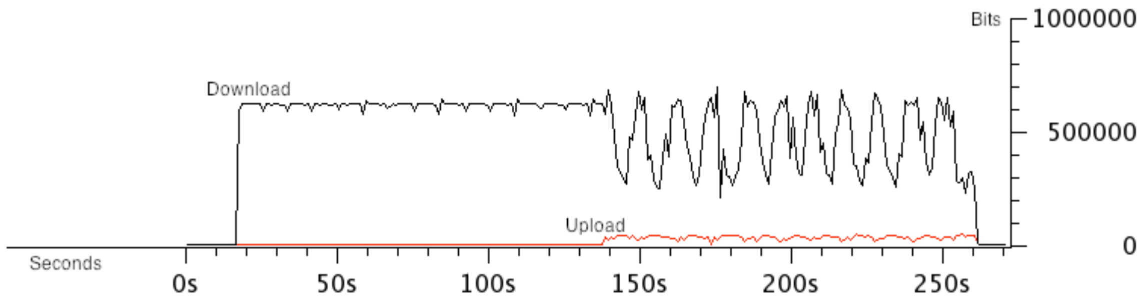


Figure 5.4: Throughput with 25 Packet Uplink Buffer for 2.4Mbps Downlink and 196Kbps Uplink

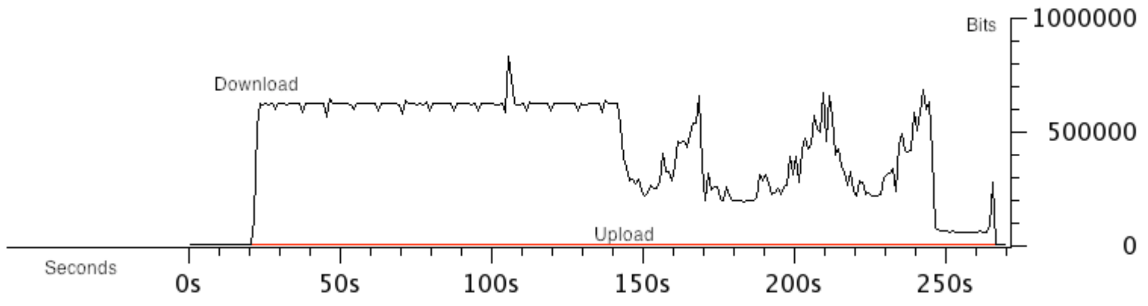


Figure 5.5: Throughput with 50 Packet Uplink Buffer for 2.4Mbps Downlink and 196Kbps Uplink

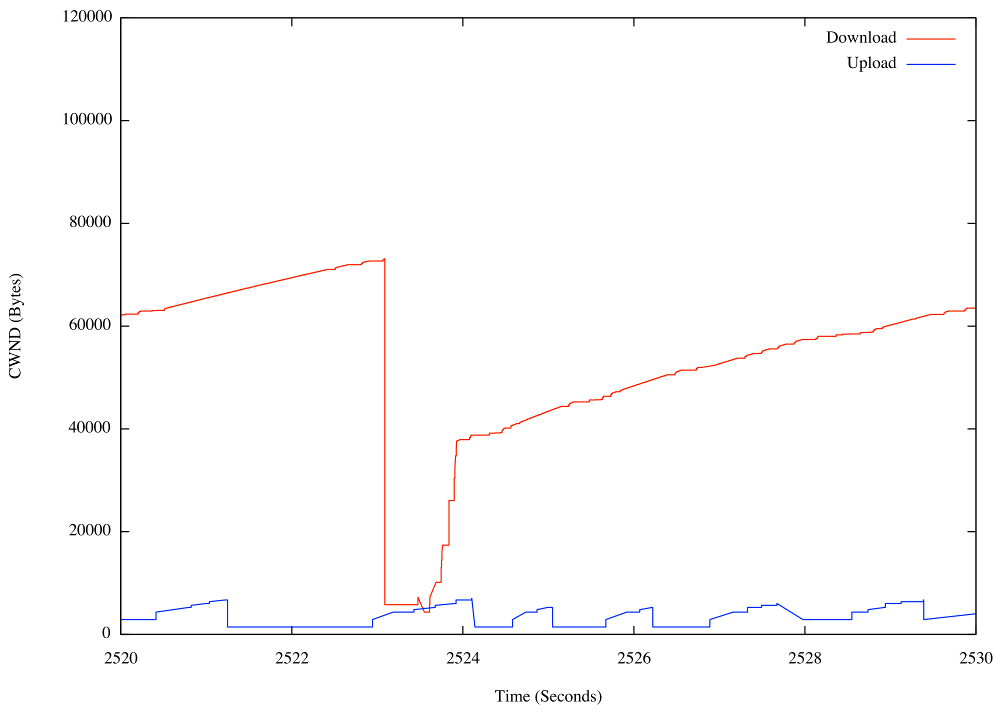


Figure 5.6: 10 Packet Uplink Buffer - Out of Phase CWND for 2.4Mbps Downlink and 196Kbps Uplink

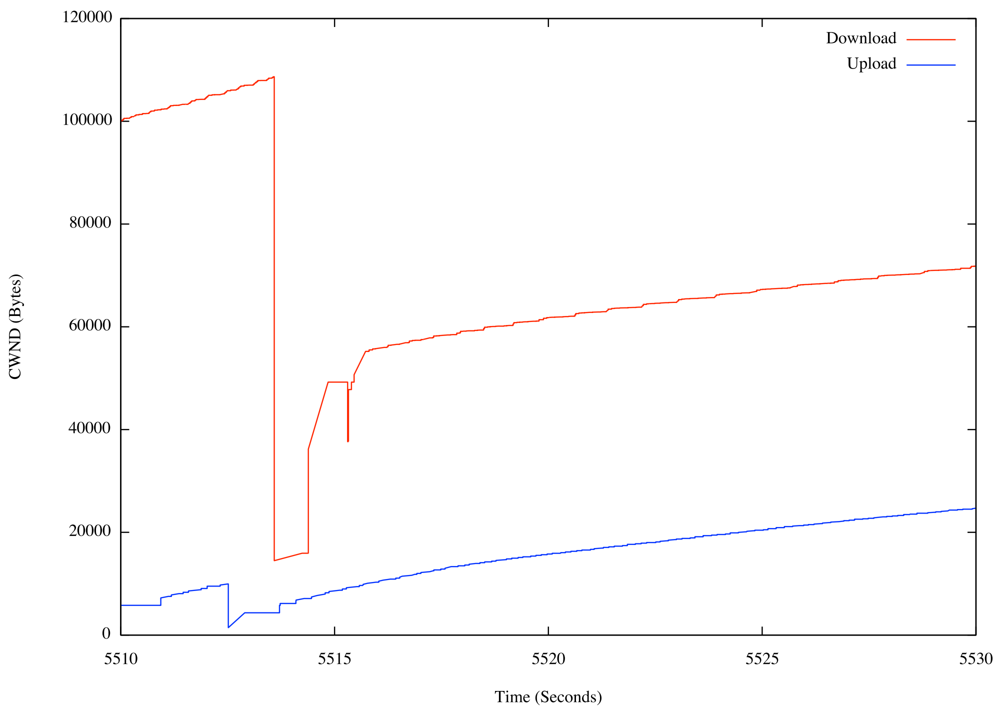


Figure 5.7: 50 Packet Uplink Buffer - In Phase CWND for 2.4Mbps Downlink and 196Kbps Uplink

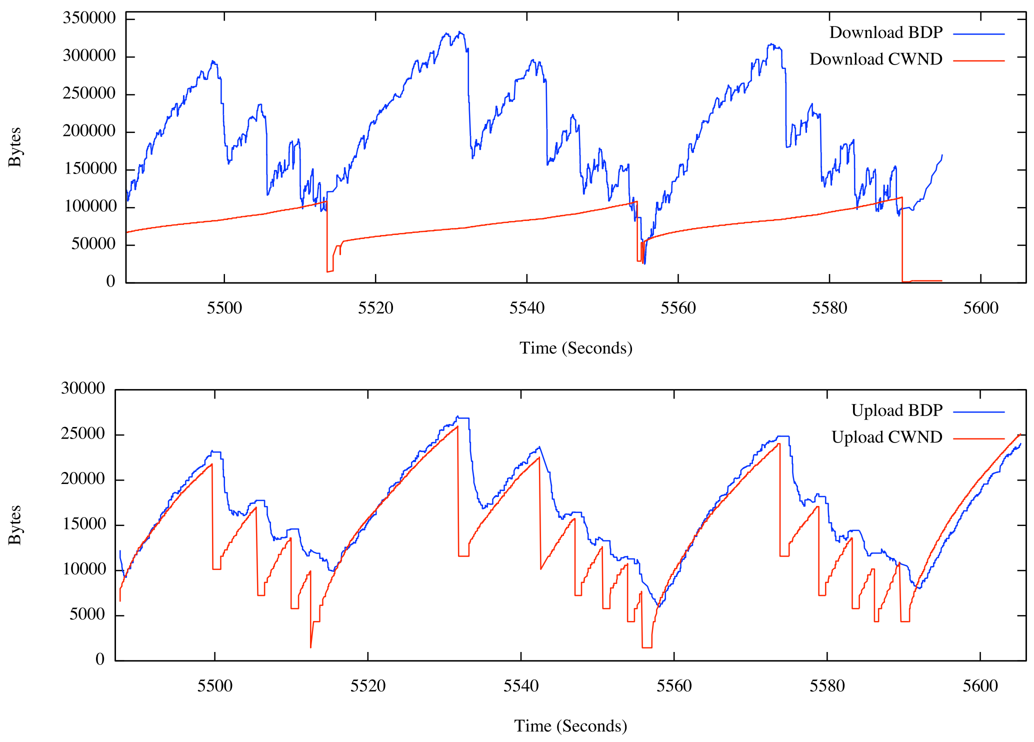


Figure 5.8: 50 Packet Uplink Buffer for 2.4Mbps Downlink and 196Kbps Uplink

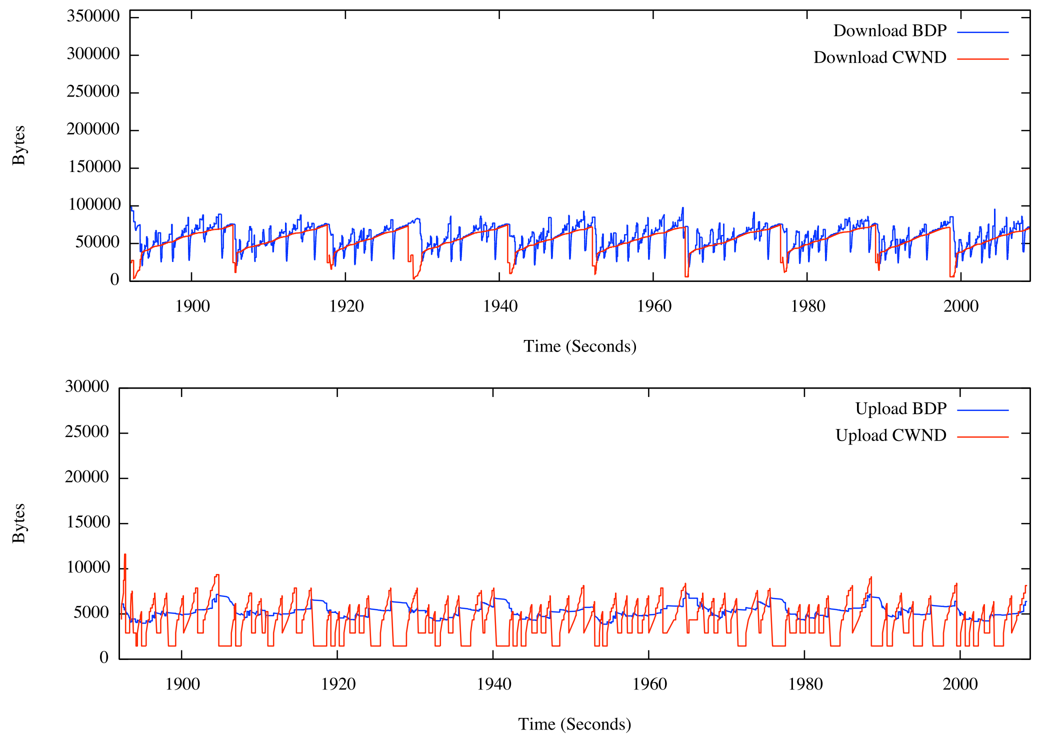


Figure 5.9: 5 Packet Uplink Buffer for 2.4Mbps Downlink and 196Kbps Uplink

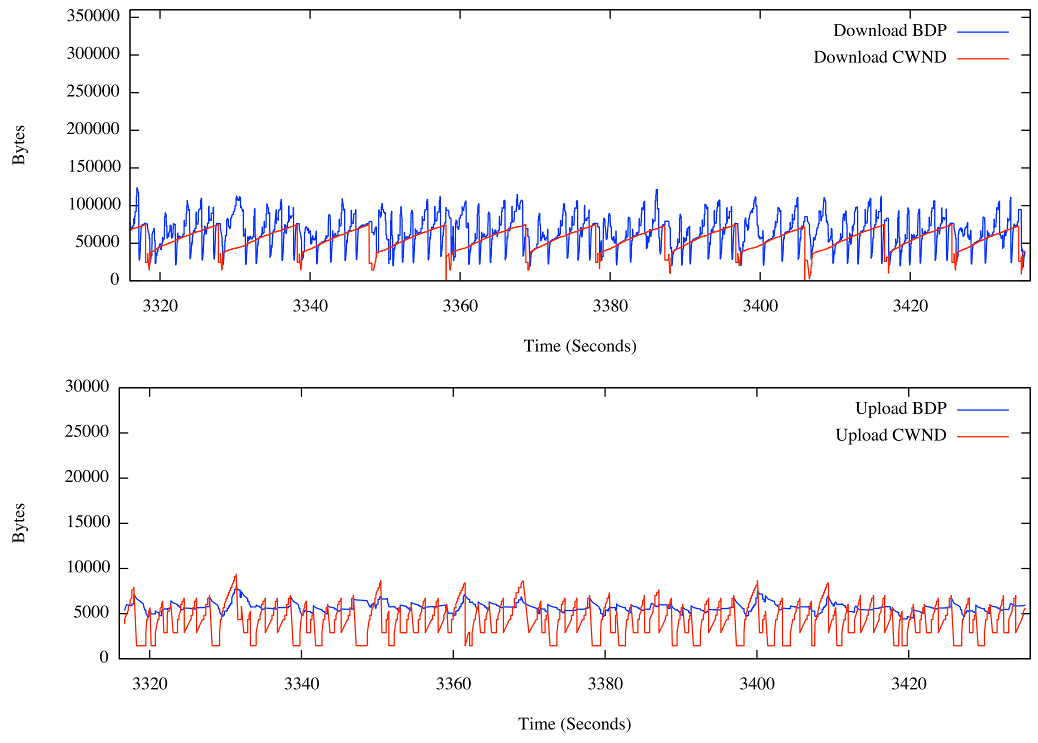
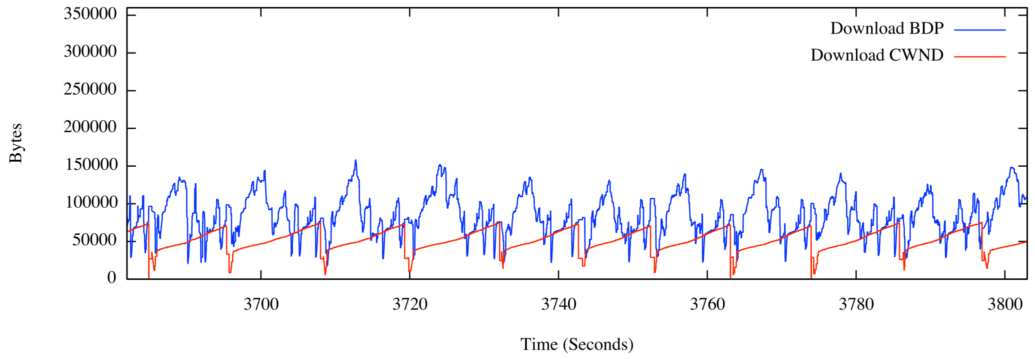


Figure 5.10: 20 Packet Uplink Buffer for 2.4Mbps Downlink and 196Kbps Uplink



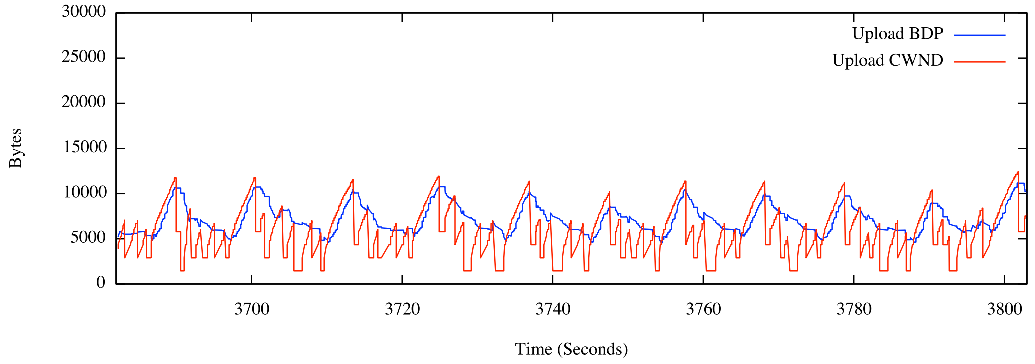


Figure 5.11: 25 Packet Uplink Buffer for 2.4Mbps Downlink and 196Kbps Uplink

of this behavior for B_{R2} sizes of 10 and 50 respectively. For B_{R2} equal to 10, the CWND value for TCP_{AB} drops while the CWND value for TCP_{BA} increases. For B_{R2} equal to 50, both CWND values for TCP_{AB} and TCP_{BA} change in the same direction around the same time. Out of phase synchronization yields higher goodput for TCP_{AB}.

5.1.2 ACK Compression

ACK compression is a phenomenon where a series of ACK segments arrive at a rate that is larger than the rate at which the corresponding data segments were transmitted. ACK compression occurs in connection TCP_{BA}. As the B_{R2} buffer size is increased, ACK compression becomes more pronounced because a larger amount of ACK segments are queued for longer periods of time and then subsequently transmitted back to back in a burst. In Figure 5.8 it can be seen that small sawtooth patterns occur on top of a larger sawtooth pattern. The smaller sawtooth patterns indicate periods of ACK compression. This is indicated by periods of window growth, followed by periods of zero transmission.

ACK compression induces bursty TCP behavior. What follows is intermittent periods where the network is filled with too many packets, resulting in packet loss, TCP throttling, and ultimately, reduced goodput.

5.1.3 Random and Periodic Connection Interference

When the two connections are active on the network, each experiences interference from the other. As the size of B_{R2} increases, the amount of interference also increases, and transforms from what appears to be a random effect, in Figures 5.2 and 5.3, into a periodic effect, in Figures 5.4, and 5.5. This periodicity is presumably a result of B_{R2} filling with TCP_{BA} data segments, thus delaying enough TCP_{AB} ACKs to stall the connection for periods of time. Table 5.1 lists B_{R2}'s

Table 5.1: Uplink Packet Buffer in Packets and Seconds

Packets	Seconds
5	0.306
10	0.612
15	0.918
20	1.224
25	1.531
30	1.837
35	2.143
40	2.449
45	2.755
50	3.061

size in packets and corresponding value in seconds, assuming 1500 byte packets and a 196Kbps link rate. This implies that when TCP_{AB} is sending in congestion avoidance while not in the presence of an upload from TCP_{BA}, there is an additional amount of potential delay within the network equal the buffer size. Intuitively, the larger the uplink buffer is, the larger the potential delay is. As we will show, the key is to size this amount of potential delay such that both TCP_{AB} and TCP_{BA} can exist together without inducing large amounts of RTT variaiblity with one another.

5.1.4 BDP Tracking

Figures 5.9, 5.10, 5.11 and 5.8 show that as B_{R2} is increased, the BDP becomes much larger than CWND of TCP_{AB}, indicating that the CWND stops accurately tracking the BDP of the system. This is due to the large periodic increases in RTT. The increase can be correlated to the queueing delay in B_{R2}. Table 5.1 lists B_{R2}'s size in packets and corresponding value in seconds for the asymmetric link configuration of 2.4Mbps down and 196Kbps up.

When B_{R2} is kept to small values, the interference between TCP_{AB} and TCP_{BA} is minimized. That is, the variability in RTT as measured by the connections remains small. As B_{R2} increases, TCP_{BA} fills it with data segments during slow start, thus delaying the ACKs from TCP_{AB} and increasing the RTT and BDP. As a result, the CWND used by TCP_{AB} is not sufficiently large enough to keep the downlink busy when TCP_{BA} is active. We show RTT variability as seen by host A in terms of standard deviation in Figure 5.12 for each asymmetric link configuration and uplink buffer size. It is clear that RTT variability increases as buffer size increases. The rate at which RTT variability increases with respect to buffer size is a function of link asymmetry where the larger the asymmetry the larger the growth in RTT variability.

5.1.5 Summary

These experiments and results show the impact of buffer size on bi-directional TCP traffic across asymmetric links, and also begin to demonstrate that a simple solution does in fact exist. We notice that for buffers of up to a certain size, performance across all of our asymmetric link combinations remain at high levels. However, once this size is exceeded, performance rapidly drops. The reason for the drop is related to the large fluctuations in RTT that are induced when the two TCP connections operate in phase of one another. Therefore, a simple solution exists where as long as a buffer is less than the threshold, the system will yield significantly higher performance than it would otherwise. In our next chapter, we will derive an uplink buffer estimation model based on previous work in [2] that can be used to determine an approximate value for this threshold and thus, preserve overall system performance.

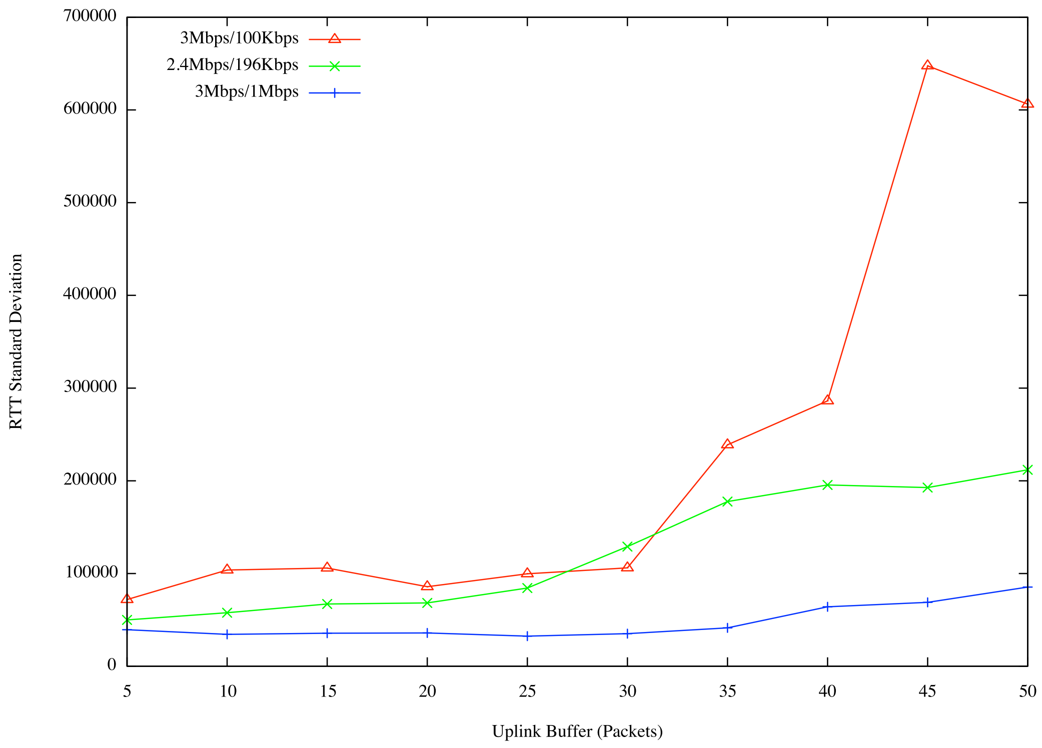


Figure 5.12: Uplink Buffer vs. RTT Variability

Chapter 6

Optimization

6.1 Introduction

In this chapter, a buffer sizing strategy is proposed that constrains maximum RTT and improves goodput for TCP_{AB} while at the same time providing acceptable goodput for TCP_{BA}. This strategy is beneficial because it does not involve complex queueing algorithms in network elements and can potentially be implemented locally at the bottleneck link without requiring changes to de facto TCP implementations in existing operating systems widely used on the Internet. First we briefly review the factors of bi-directional TCP performance that are influenced by buffer size and contribute to degraded performance. We then go on to show our buffer sizing strategy and review its accuracy using data from Section 5.1. Finally we present a procedure for network designers or operators that can be used on real world networks to ensure uplink buffers are sized such that the system yields performance values similar to our test network as shown in Table 6.2.

6.2 Review of System Dynamics

As we have previously reviewed, a TCP sender must size its CWND to the BDP of the path between itself and the receiver in order to fully utilize the link. BDP is calculated via Equation (2.1) and is composed of bottleneck link line rate and RTT as seen by the sender. As we have

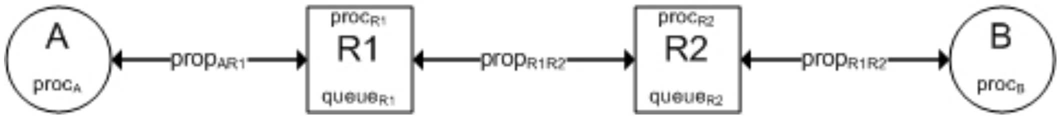


Figure 6.1: Test Network RTT Components

shown in Section 5.1, in the scenario where bi-directional TCP traffic is present, the BDP of the path changes periodically over time and is directly related to uplink buffer size. We have also reviewed prior work that details the dynamics of bi-directional traffic, specifically how window sizes of the TCP connections influence the overall performance of the system. The work shows that if TCP connections remain out of phase, only one direction of the bottleneck link is fully utilized. In the case of asymmetric links, it is preferred to use more of the downlink than uplink in order to maximize the performance of the system. As we will show with respect to our test system, sizing the uplink buffer appropriately limits what phase the two connections operate in, which in turn limits the variability in RTT, shown in Figure 5.12, and preserves the TCP sender's ability to track the BDP of the path while in congestion avoidance mode.

In our system we calculate RTT using Equation (6.1) as shown in Figure 6.1. In Equation (6.1), let t_{prop} equal total propagation delay, t_{proc} equal total processing delay, and t_{queue} equal total queueing delay. Each component of delay can be detailed further as shown in Equations (6.2) - (6.4).

$$\mathrm{RTT} = t_{prop} + t_{proc} + t_{queue} \tag{6.1}$$

$$t_{prop} = 2 \cdot (prop_{AR_1} + prop_{R_1R_2} + prop_{BR_2}) \tag{6.2}$$

$$t_{queue} = queue_{BR_1} + queue_{BR_2} \tag{6.3}$$

$$t_{proc} = 2 \cdot (proc_A + proc_{R1} + proc_{R2} + proc_B) \tag{6.4}$$

In Equation (6.2), let $prop_{AR_1}$, $prop_{R_1R_2}$, and $prop_{BR_2}$ equal the propagation delays of the links between host A and router R_1, R_1 and R_2, and R_2 and host B respectively. In Equation (6.3), let $queue_{B_{R1}}$ and $queue_{B_{R2}}$ equal the queueing delay of B_{R1} and B_{R2} respectively. In Equation (6.4), let $proc_A$, $proc_B$, $proc_{R_1}$, $proc_{R_2}$ equal the processing delays at host A, host B, router R_1, and router R_2 respectively. These delays are shown in Figure 6.1.

6.3 Uplink Buffer Sizing Model

The dynamics of bi-directional TCP connections involve the RTT and bottleneck rate of the path and TCP window size. TCP operates in steady state when the RTT of a link is steady. This allows the congestion avoidance algorithm to accurately track the BDP of the link. If RTT changes by large amounts, periodically, TCP congestion avoidance mode cannot accurately size its CWND to reflect the BDP of the path. Therefore, in order to preserve this characteristic, the amount by which the RTT can change must be limited. As have we shown in Table 5.1, much of the potential RTT variability is located in empty buffers within the network, B_{R2} in our network. This implies that buffers must be sized with care. This has been the topic of significant research, and has led the *tiny model* for sizing core router buffers. However, a packet-based buffer sizing strategy for bi-directional TCP traffic and asymmetric links has yet to be produced.

Beginning with Equation (2.5), a maximum threshold for the uplink buffer is be derived under which both connections are out of phase, and both can accurately track the BDP of downlink and uplink. Let W_D equal the maximum CWND for TCP_{AB} and W_U equal the maximum CWND for TCP_{BA}. Also, let C_D equal the capacity of L_{21} in bytes, and RTT equal to the total delay excluding the queueing delay at B_{R2} in seconds. Finally, let B_{R2} equal the size of R_2's buffer in packets.

$$W_D > W_U + 2 \cdot (RTT + (B_{R2} \cdot 1500)/C_U) \cdot C_U \tag{6.5}$$

$$B_{R2} < 1/1500 \cdot ((W_D - W_U)/2 - RTT \cdot C_U) \tag{6.6}$$

To obtain accurate values, we use our software to determine maximum CWND values for both TCP_{AB} and TCP_{BA} as well as RTT for each asymmetric link combination. Maximum CWND values for TCP_{AB} and TCP_{BA} are obtained from the log file for each connection under each asymmetric link combination and experiment run. We measure the RTT of our network at host A using log file data collected by our software. We take the average RTT as seen by TCP_{AB} without the presence of TCP_{BA} across each asymmetric link combination to avoid queueing delay at R_2. For each link configuration, we only take RTT measurements from one buffer configuration because the ACK stream generated by host B always requires less bandwidth than what is available on the uplink, thus avoiding any queueing delay at R_2. The results of the RTT measurements are listed in Table 6.1.

Table 6.1: Average RTT

Downlink (Kbps)	Uplink (Kbps)	Asymmetry	Average RTT (msec)
3000	1000	3	140
2400	196	12.24	200
3000	100	30	160

Using Equation (6.6) and the previously the mentioned RTT and CWND values, the maximum size for B_{R2} is estimated for each asymmetric link configuration. Figure 6.2 shows the actual values of B_{R2} as configured in our experiments versus the estimated value of B_{R2} for each asymmetric link ratio. For each asymmetric link configuration, the intersection of the actual buffer and estimated buffer represents the maximum uplink buffer threshold. Once this value is exceeded, system performance begins to degrade rapidly while buffer sizes less than this value yield performance values as high as 93%. We list maximum buffer size estimations and corresponding performance measurements in Table 6.2.

Using asymmetric link combination of 2.4Mbps down and 196Kbps as an example, we show that the uplink buffer is sized below the maximum threshold, up to a queue size of 17 packets. Beyond this value, the buffer is sized inappropriately, and as a result, leads to the breakdown in BDP tracking in TCP_{AB}'s congestion avoidance algorithm as previously reviewed in Section 5.1.4.

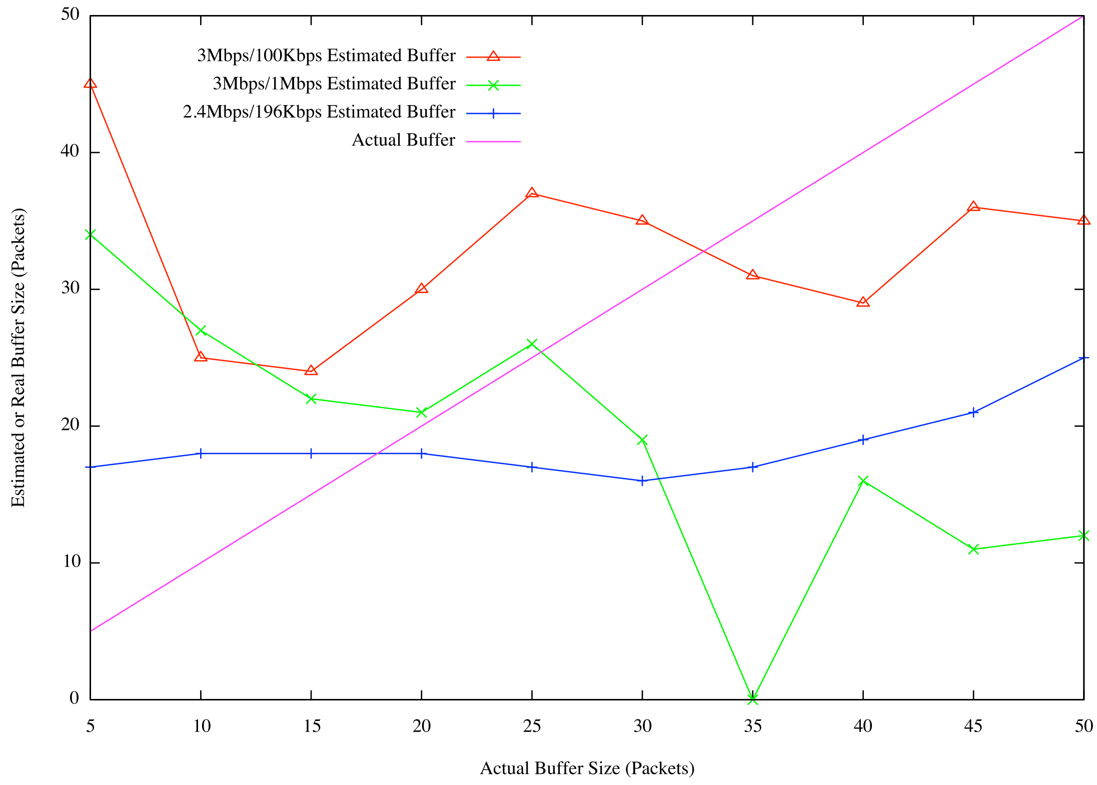


Figure 6.2: Actual Buffer vs. Estimated Buffer for Asymmetric Link Combinations

Table 6.2: Results Summary

Downlink (Kbps)	Uplink (Kbps)	Asymmetry	Estimated Uplink Buffer (Packets)	Performance
3000	1000	3	25	0.80
2400	196	12.24	17	0.86
3000	100	30	30	0.68

As shown in Figure 5.1, this results in a continuous drop in system performance, from 86% at our estimated threshold, to 55% at the largest measured size of B_{R2}.

6.4 Use Case and Procedure

Network designers will find our work useful when incorporating asymmetric links into the access layer of a network because we present a model and results based on a small number of TCP connections in either direction, which is a typical traffic pattern found at CPE nodes. Furthermore, our work presents a simple solution that can be implemented in software on CPE nodes without introducing additional hardware requirements.

Our work can be used by network designers and operators in two ways. One way is to use our test methodology, software and estimation model in conjunction with a real world network and set of hosts as shown in Figure 6.3 to determine appropriate uplink buffer size such that bi-directional TCP traffic performs at levels shown in our work. In Figure 6.3, let A and B represent TCP hosts. Let CPE represent the network element terminating the asymmetric link. Let X represent the downlink and Y represent the uplink. The CPE implements FIFO queueing using a packet based buffer on its uplink. In the following procedure we refer to this system and demonstrate how to estimate maximum uplink buffer size.

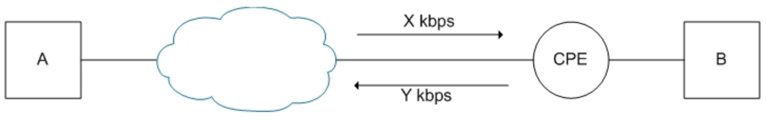


Figure 6.3: Real World Network with Asymmetric Link Ratio of X / Y

(1) RTT: Measure the RTT at a typical TCP sender, A in Figure 6.3, on the network while the uplink at the CPE, link Y in Figure 6.3, is not congested, thus excluding uplink queueing delay at the CPE.

To determine the RTT, a download should be launched across the downlink where host B downloads data from host A. The duration of the download is somewhat arbitrary. The session should last long enough to determine an appropriate average value. During the download, RTT measurements should be collected. We present two different ways to measure RTT.

First, RTT can be measured by capturing the download traffic using Wireshark. In Wireshark, set a filter to retain only the TCP download. Once the download has completed, save the file. Next, browse to the IO Graphs section under the Statistics menu. From here, select the Y Axis Unit: Advanced option from the drop down menu. Next, choose AVG(*) from the Calc: section in Graph 1. In the text field next to Calc:, enter `tcp.analysis.ack_rtt`. Finally, click the Graph 1 button to generate the plot.

A second way to determine RTT is by using our software and its resulting log file. After the download between hosts A and B, where A is the TCP sender, has been completed, the average of all RTT values listed in the second column of the log file can be calculated. This can be automated using the Download Averaging Script we provide in the Appendix.

(2) Maximum CWND: Next, the uplink buffer on the CPE must be configured to an arbitrary value. After the buffer has been configured, a download must be launched using our software, where host A is the TCP sender. After host A's TCP connection has entered congestion avoidance, an upload must be launched where host B is the TCP sender. Because it is not easy to determine what phase a TCP connection is operating in, we recommend letting the download run for 30 seconds before starting the upload. We then recommend letting the upload and download both run for an additional 30 seconds before terminating the connections. After both connections are terminated, the maximum CWND values should be extracted from the resultant log files generated by both connections. The Maximum CWND Script is provided in the Appendix to generate maximum CWND values.

(3) Uplink Buffer Estimation: Now substitute average RTT calculated in step 1 and maximum CWND values calculated in step 2 into Equation (6.6) to generate a maximum uplink buffer value. If the uplink buffer provisioned on the CPE is less than this value, it can be used to keep bi-directional TCP connections out of phase, which in turn, keeps RTT variability minimized and system performance higher than it would be if the uplink buffer was sized any larger. If the actual buffer is larger than the estimated buffer, the buffer should be set to less than the estimated buffer produced by Equation (6.6).

A second way, one that we have not validated and present as future work, is to use data presented previous work to estimate RTT and CWND values for downloads and uploads. As shown in previous work, variable delay primarily exists in the access layer of a network because of the *tiny model* for buffer sizing in core network elements. This means network designers can accurately estimate how much RTT variability is in the network based upon buffer sizes and link speeds in the access network. In addition to this, research also shows that there is an average RTT for TCP connections traversing the Internet of about 110 msec [27]. Using these two points, it is possible to estimate an average BDP which in turn can be used to estimate a maximum CWND value. The estimated RTT and CWND values can then be inserted into inequality (6.6) to determine an appropriate uplink buffer size for the particular asymmetric link.

6.5 Summary

In this chapter we derive our sizing model and demonstrate its accuracy by comparing its recommended size to performance results in Section 5.1. In all cases, if buffer size is sized any larger than our model recommends performance decreases. Under the worse case, where the buffer is sized as large as the uplink, performance drops from 68% at our recommended size to 7% at 50 packets. We also present a use case scenario and accompanying procedure to be used by network designers and operators to size buffers using our model in asymmetric links for the now common bi-directional TCP traffic pattern.

Chapter 7

Conclusions

This thesis investigates the effect of buffers on bi-directional TCP connections over asymmetric links. Previous work focuses on sizing buffers for symmetric links, deriving relationships for the dynamics of connections and applying novel queueing strategies and TCP adjustments in order to maintain high levels of goodput. This thesis presents a new approach that solves the problem by appropriately sizing buffers in asymmetric links.

Experiments were performed on a real network where the uplink buffer, B_{R2}, was varied from 5 packets to 50 packets in increments of 5. Packet traces and TCP state variables were collected for connection traversing the asymmetric, bottleneck link and then analyzed to show the effect of buffer size on goodput for both connections.

In summary, by properly sizing buffers, changes in link characteristics that TCP depends on, such as low packet loss, constant RTT and a steady ACK stream, can be reduced for both connections, thus optimizing goodput in both directions.

Future work will include additional data on queue length and RTT over time to more clearly highlight the behavior of each TCP connection. It will also consider how to estimate the desired buffer size in an online setting. Finally, it will use data reviewed in prior work to validate whether or not a general case buffer configuration for asymmetric links can be created that aligns with a common Internet based TCP connection characteristics.

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Appendix A

Router Shaper Configuration

```
R1 Configuration

policy-map R1

 class class-default

  shape average 2400000

  queue-limit 50

!

interface FastEthernet0/0

 ip address 192.168.1.1 255.255.255.252

 speed 100

 duplex full

 service-policy R1 out

!

R2 Configuration

policy-map R2

 class class-default

  shape average 196000

  queue-limit <5-50>
```

```
!

interface FastEthernet0/0

 ip address 192.168.1.2 255.255.255.252

 speed 100

 duplex full

 service-policy R2 out
```

Appendix B

TCP Server

```
#include <unistd.h>
#include <stdio.h>
#include <stdlib.h>
#include <string.h>
#include <netinet/in.h>
#include <sys/types.h>
#include <sys/socket.h>
#include <errno.h>
#include <netdb.h>
#include <netinet/tcp.h>
#include <arpa/inet.h>

#define PORT 5000

int
main(int argc, char *argv[]) {
    int tcpsock, sockopt, newsock, pkts, i = 0;
    int len = sizeof(struct tcp_info);
    struct sockaddr_in myaddr, tcpaddr;
        struct tcp_info tcpinfo;
    char buf[1460];

    bzero(&myaddr, sizeof(myaddr));
    myaddr.sin_family = AF_INET;
    myaddr.sin_addr.s_addr = htonl(INADDR_ANY);
    myaddr.sin_port = htons(PORT);

    bzero(&tcpaddr, sizeof(tcpaddr));
    tcpaddr.sin_family = AF_INET;

    if ((tcpsock = socket(AF_INET, SOCK_STREAM, IPPROTO_TCP)) < 0)
        printf("socket: %s\n", strerror(errno));

    if (setsockopt(tcpsock, SOL_SOCKET, SO_REUSEADDR, &sockopt, sizeof(sockopt)) < 0)
        printf("couldnt set options\n");

    if (bind(tcpsock, (struct sockaddr *)&myaddr, sizeof(myaddr)) < 0)
```

```
        printf("bind: %s\n", strerror(errno));

    if (listen(tcpsock, 5) < 0)
        printf("listen: %s\n", strerror(errno));

    if ((newsock = accept(tcpsock, (struct sockaddr *)&tcpaddr, &len)) < 0)
        printf("accept: %s\n", strerror(errno));

    for (;;) {
        if (read(newsock, buf, sizeof(buf)) <= 0) {
            printf("read: %s\n", strerror(errno));
                        close(newsock);
            return 0;
        }
    }
    return 0;
}
```

Appendix C

TCP Client

```
#include <unistd.h>
#include <stdio.h>
#include <stdlib.h>
#include <string.h>
#include <netinet/in.h>
#include <sys/types.h>
#include <sys/socket.h>
#include <arpa/inet.h>
#include <netinet/tcp.h>
#include <errno.h>
#include <netdb.h>
#include <time.h>
#include <sys/time.h>


#define PORT 5000
#define EXTRA_TIME 1267500000

int
main (int argc, char *argv[]) {
    int tcpsock, sockopt, pkts, i, bytes = 0, totalbytes = 0;
    int len = sizeof(struct tcp_info);
        double starttime = 0, curtime = 0, reltime = 0;
    float throughput;
    struct sockaddr_in myaddr, tcpaddr;
    struct tcp_info tcpinfo;
    struct timeval tv;
    char buf[1460];

    FILE *fd;
    fd = fopen("tcpstats.txt", "a+");

        if (argc == 3)
                pkts = atoi(argv[2]);
        else
                pkts = 0;
```

```
bzero(&tcpinfo, sizeof(tcpinfo));
bzero(&tv, sizeof(tv));

bzero(&myaddr, sizeof(myaddr));
myaddr.sin_family = AF_INET;
myaddr.sin_addr.s_addr = htonl(INADDR_ANY);
myaddr.sin_port = htons(0);

bzero(&tcpaddr, sizeof(tcpaddr));
tcpaddr.sin_family = AF_INET;
tcpaddr.sin_port = htons(PORT);
if (inet_pton(AF_INET, argv[1], &tcpaddr.sin_addr) <= 0) {
            printf("could not copy ip to address structure\n");
            return 1;
    }

if ((tcpsock = socket(AF_INET, SOCK_STREAM, 0)) < 0)
    printf("socket: %s\n", strerror(errno));

if (setsockopt(tcpsock, SOL_SOCKET, SO_REUSEADDR, &sockopt, sizeof(sockopt)) < 0)
    printf("couldnt set options\n");

if (connect(tcpsock, (struct sockaddr *)&tcpaddr, sizeof(tcpaddr)) < 0)
    printf("connect: %s\n", strerror(errno));

fprintf(fd, "%s\t%s\t%s\t%s\t%s\t%s\t%s\t%s\t%s\n",
                    "Time",
                    "RTT",
                    "RTT Var",
                    "SSTHRESH",
                    "CWND",
                    "Rcv Wnd",
                    "Snd Wnd",
                    "Snd BW Wnd",
                    "Throughput");

if (pkts != 0) {
    for (i = 0; i <= pkts; i++) {
        gettimeofday(&tv, NULL);
        curtime = (double)tv.tv_sec + (double)tv.tv_usec/1000000 - EXTRA_TIME;

                    if (bytes == 0)
                            starttime = curtime;

                    reltime = curtime - starttime;

        if ((bytes = write(tcpsock, buf, sizeof(buf))) < 0)
            printf("write: %s\n", strerror(errno));

        totalbytes += bytes;
        throughput = (totalbytes * 8)/reltime;
```

```
            if (getsockopt(tcpsock, IPPROTO_TCP, TCP_INFO, (void *)&tcpinfo, (socklen_t *)&len) == -1)
                printf("getsockopt:␣%s\n", strerror(errno));

                        fprintf(fd, "%f\t%u\t%u\t%u\t%u\t%u\t%u\t%u\t%f\n",
                                curtime,
                                tcpinfo.tcpi_rtt,
                                tcpinfo.tcpi_rttvar,
                                tcpinfo.tcpi_snd_ssthresh,
                                tcpinfo.tcpi_snd_cwnd,
                                tcpinfo.tcpi_rcv_space,
                                tcpinfo.tcpi_snd_wnd,
                                tcpinfo.tcpi_snd_bwnd,
                                throughput);
        }
    }
    else {
        for (;;) {
            gettimeofday(&tv, NULL);
            curtime = (double)tv.tv_sec + (double)tv.tv_usec/1000000 - EXTRA_TIME;

                        if (bytes == 0)
                                starttime = curtime;

                        reltime = curtime - starttime;

            if ((bytes = write(tcpsock, buf, sizeof(buf))) < 0)
                printf("write:␣%s\n", strerror(errno));

            totalbytes += bytes;
            throughput = (totalbytes * 8)/reltime;

            if (getsockopt(tcpsock, IPPROTO_TCP, TCP_INFO, (void *)&tcpinfo, (socklen_t *)&len) == -1)
                printf("getsockopt:␣%s\n", strerror(errno));

            fprintf(fd, "%f\t%u\t%u\t%u\t%u\t%u\t%u\t%u\t%f\n",
                                        curtime,
                                        tcpinfo.tcpi_rtt,
                                        tcpinfo.tcpi_rttvar,
                                        tcpinfo.tcpi_snd_ssthresh,
                                        tcpinfo.tcpi_snd_cwnd,
                                        tcpinfo.tcpi_rcv_space,
                                        tcpinfo.tcpi_snd_wnd,
                                        tcpinfo.tcpi_snd_bwnd,
                                        throughput);
        }
    }
    close(tcpsock);
    fclose(fd);
    return 0;
}
```

Appendix D

Bandwidth Delay Product Estimation Script

```
#! /usr/bin/env python

import sys

bwdStats = open(sys.argv[3], "a")
myFile = open(sys.argv[1], "rU")
linkSpeed = int(sys.argv[2])

for aRow in myFile:
        if aRow.split('\t')[0] == "Time":
                continue
        else:
                bwd = linkSpeed * float(aRow.split('\t')[1]) / 8000000
                myList = [float(aRow.split('\t')[0]), bwd]
                for i,j in enumerate(myList):
                        bwdStats.write(repr(j)+'\t')
                bwdStats.write('\n')

myFile.close()
bwdStats.close()
```

Appendix E

Download Averaging Script

```
#! /usr/bin/env python

import sys

tcpStats = open(sys.argv[5], "a")
myFile = open(sys.argv[1], "rU")
cwnd = 0.0
rtt = 0.0
i = 0.0
j = 0.0
downcwnd = 0.0
downrtt = 0.0

for aRow in myFile:
        if aRow.split('\t')[0] == "Time":
                continue
        elif float(aRow.split('\t')[0]) > int(sys.argv[2]):
                cwnd+=float(aRow.split('\t')[4])
                rtt+=float(aRow.split('\t')[1])
                i+=1
        else:
                downcwnd+=float(aRow.split('\t')[4])
                downrtt+=float(aRow.split('\t')[1])
                j+=1

avgDownCWND = downcwnd/j
avgDownRTT = downrtt/j
avgDownThroughput = (avgDownCWND * 8) / (avgDownRTT / 1000000)

avgCWND = cwnd/i
avgRTT = rtt/i
avgThroughput = (avgCWND * 8) / (avgRTT / 1000000)
difThroughput = avgDownThroughput - avgThroughput
difRTT = avgRTT - avgDownRTT
avgBWD = (avgRTT / 8000000) * int(sys.argv[3])
diffWND = avgBWD - avgCWND
diffThpt = (diffWND * 8) / (avgRTT / 1000000)
```

```
print "Average_CWND:\t%f_" % avgCWND
print "Average_Down_CWND:\t%f" % avgDownCWND
print "Average_Throughput:\t%f" % avgThroughput
print "Average_Download_Throughput:\t%f" % avgDownThroughput
print "Throughput_Difference:\t%f" % difThroughput
print "Average_RTT:\t%f" % avgRTT
print "Average_Download_RTT:\t%f" % avgDownRTT
print "RTT_Difference:\t%f" % difRTT
print "Average_BW*D:\t%f" % avgBWD
print "Window_Difference:\t%f" % diffWND
print "Ideal_Thpt_Difference:\t%f" % diffThpt

myList = [int(sys.argv[4]), avgCWND, avgDownCWND, avgThroughput, avgDownThroughput,
                difThroughput, avgRTT, avgDownRTT, difRTT, avgBWD]

for i,j in enumerate(myList):
        tcpStats.write(repr(j)+'\t')
tcpStats.write('\n')
myFile.close()
tcpStats.close()
```

Appendix F

Upload Averaging Script

```
#! /usr/bin/env python
import sys
tcpStats = open(sys.argv[3], "a")
myFile = open(sys.argv[1], "rU")

i = 0
cwnd = 0
rtt = 0

for aRow in myFile:
        if aRow.split('\t')[0] == "Time":
                continue
        else:
                cwnd += float(aRow.split('\t')[4])
                rtt += float(aRow.split('\t')[1])
                i+=1

avgCWND = cwnd/i
avgRTT = rtt/i
avgThroughput = (avgCWND * 8) / (avgRTT / 1000000)

myList = [int(sys.argv[2]), avgCWND, avgThroughput, avgRTT]
for i,j in enumerate(myList):
        tcpStats.write(repr(j)+'\t')
tcpStats.write('\n')

tcpStats.close()
myFile.close()
```

Appendix G

Maximum CWND Script

```
#!/usr/bin/env python

#1 infile
#2 outfile
#3 buffer

import sys

myFile = open(sys.argv[1], "rU")
tcpStats = open(sys.argv[2], "a")

cwnd = 0.0

for aRow in myFile:
        if aRow.split('\t')[0] == "Time":
                continue
        else:
                if cwnd < float(aRow.split('\t')[4]):
                        cwnd = float(aRow.split('\t')[4])

print "Max CWND: %f" % cwnd

myList = [int(sys.argv[3]), cwnd]

for i,j in enumerate(myList):
        tcpStats.write(repr(j)+'\t')

tcpStats.write('\n')
myFile.close()
tcpStats.close()
```

Appendix H

Buffer Estimation Script

```
#! /usr/bin/env python

#1 max cwnd file
#2 outfile
#3 bwd in bytes of uplink

import sys

maxCwnd = open(sys.argv[1], "rU")
out = open(sys.argv[2], "a")

bwd = int(sys.argv[3])

for aRow in maxCwnd:
        if aRow.split('\t')[0] == "buffer":
                continue
        else:
                data = (int(aRow.split('\t')[1]) - int(aRow.split('\t')[2])) / 2 - bwd
                packets = data / 1500
                myList = [int(aRow.split('\t')[0]), packets]
                for i,j in enumerate(myList):
                        out.write(repr(j)+'\t')
                out.write('\n')
```

Appendix I

Performance Script

```
#! /usr/bin/env python

#open a file with both dowload and upload throughputs in columns
#1 infile
#2 outfile
#3 downlink rate
#4 uplink rate

import sys

avgGoodput = open(sys.argv[1], "rU")
out = open(sys.argv[2], "a")

Cd = float(sys.argv[3])
Cu = float(sys.argv[4])

for aRow in avgGoodput:
        if aRow.split('\t')[0] == "buffer":
                continue
        else:
                netPerformance = (float(aRow.split('\t')[1]) + float(aRow.split('\t')[2])) / (Cd + Cu)
                downPerformance = float(aRow.split('\t')[1]) / Cd
                upPerformance = float(aRow.split('\t')[2]) / Cu
                myList = [int(aRow.split('\t')[0]), netPerformance, downPerformance, upPerformance]
                for i,j in enumerate(myList):
                        out.write(repr(j)+'\t')
                out.write('\n')
```

Appendix J

RTT Variability Script

```
#!/usr/bin/env python

#calculate the variance of RTT from an input file by taking the RTT from a file, building a list
#then calculating the variance using np.var(rtt)
#1 infile
#2 time
#3 buffer
#4 outfile

import numpy as np
import sys

myFile = open(sys.argv[1], "rU")
tcpStats = open(sys.argv[4], "a")

rttList = []

for aRow in myFile:
        if aRow.split('\t')[0] == "Time":
                continue
        elif float(aRow.split('\t')[0]) < float(sys.argv[2]):
                continue
        else:
                rttList.append(float(aRow.split('\t')[1]))

rttVar = np.var(rttList)
rttStd = np.std(rttList)
print rttVar
print rttStd

myList = [int(sys.argv[3]), rttVar, rttStd]

for i,j in enumerate(myList):
        tcpStats.write(repr(j)+'\t')
tcpStats.write('\n')
myFile.close()
tcpStats.close()
```