Department of Computer Science

Per Hurtig

Fast retransmit inhibitions for TCP

Master's Thesis

2006:xx

Fast retransmit inhibitions for TCP

Per Hurtig

This thesis is submitted in partial fulfillment of the requirements for the Masters degree in Computer Science. All material in this thesis which is not my own work has been identified and no material is included for which a degree has previously been conferred.

Per Hurtig

Approved, 2006-02-07

Opponent: Stefan Lindberg

Opponent: Fredrik Strandberg

Advisor: Johan Garcia

Examiner: Tim Heyer

Abstract

The Transmission Control Protocol (TCP) has been the dominant transport protocol in the Internet for many years. One of the reasons to this is that TCP employs congestion control mechanisms which prevent the Internet from being overloaded. Although TCP's congestion control has evolved during almost twenty years, the area is still an active research area since the environments where TCP are employed keep on changing. One of the congestion control mechanisms that TCP uses is fast retransmit, which allows for fast retransmission of data that has been lost in the network. Although this mechanism provides the most effective way of retransmitting lost data, it can not always be employed by TCP due to restrictions in the TCP specification.

The primary goal of this work was to investigate when fast retransmit inhibitions occur, and how much they affect the performance of a TCP flow. In order to achieve this goal a large series of practical experiments were conducted on a real TCP implementation.

The result showed that fast retransmit inhibitions existed, in the end of TCP flows, and that the increase in total transmission time could be as much as 301% when a loss were introduced at a fast retransmit inhibited position in the flow. Even though this increase was large for all of the experiments, ranging from $16 - 301\%$, the average performance loss, due to an arbitrary placed loss, was not that severe. Because fast retransmit was inhibited in fewer positions of a TCP flow than it was employed, the average increase of the transmission time due to these inhibitions was relatively small, ranging from $0,3 - 20,4\%$.

Contents

1 Introduction **1**

1.1 Scope of work . . . 2

1.2 Disposition . . . 3

2 Background **5**

2.1 TCP - Transmission Control Protocol . . . 5

2.1.1 TCP Areas . . . 6

2.1.2 Data Transfer . . . 7

2.1.3 Reliability . . . 7

2.1.4 Flow Control . . . 9

2.1.5 Multiplexing . . . 10

2.1.6 Connection Management . . . 10

2.1.7 TCP Segments . . . 12

2.2 Congestion Control Detail . . . 15

2.2.1 Slow Start . . . 16

2.2.2 Congestion Avoidance . . . 17

2.2.3 Fast Retransmit . . . 18

2.2.4 Fast Recovery . . . 18

2.2.5 Congestion Control Summary . . . 20

2.3 Network Emulation . . . 21

2.3.1 Dummynet 22
2.4 Related Work 23
2.5 Summary 26

3 Experimental Design & Environment **29**
3.1 Problem Description 29
3.2 Method 31
3.2.1 Overview 31
3.2.2 Details 33
3.2.3 Parameters 34
3.3 Environment Overview 38
3.4 Environment Details 39
3.4.1 FreeBSD 6.0B3 40
3.4.2 FreeBSD 6.0B5 41
3.4.3 Client & Server applications 41
3.4.4 Dummynet & Loss patterns 41
3.4.5 Tcpdump 43
3.4.6 Script 43
3.5 Summary 44

4 Results and Analysis **45**
4.1 Short flows 45
4.1.1 General performance loss 45
4.1.2 Positional dependencies 48
4.1.3 Bad RTO Estimation 51
4.1.4 Link utilization implications 55
4.1.5 Receive buffer fluctuations 57
4.1.6 TCP packet bursts and their implications 58

4.2 Long flows 60
4.2.1 General performance loss 60
4.2.2 Positional dependencies 62
4.2.3 TCP implementation issues 64
4.2.4 Irregularity analysis 68
4.3 Results related to earlier work 71
4.4 Summary 72

5 Future work 73
5.1 Other approaches 73
5.2 SCTP 74
5.3 Dummynet 74
5.4 FreeBSD Initial window size 74
5.5 FreeBSD Slow Start issue 75
5.6 FreeBSD recovery issue 75
5.7 Performance gain with buffer tuning 76
5.8 Lowering the fast retransmit threshold 76

6 Conclusions 77

References 79

A List of Abbreviations 83

B Additional Software 85
B.1 Tcptrace 85

C FreeBSD Kernel details & modifications 89
C.1 TCP Implementation 89
C.1.1 Initial RTO Calculation 96

C.2 Deactivating the TCP host cache 97

D Source code & Scripts **99**

D.1 Loss pattern generator 99

D.1.1 Usage 99

D.1.2 Source code 100

D.2 Experiment script 106

D.2.1 Source code 107

D.3 Client & Server applications 113

D.3.1 Client source code 114

D.3.2 Server source code 117

List of Figures

2.1 TCP Communication 6

2.2 Acknowledgment 8

2.3 Sliding window 9

2.4 Three Way Handshake 11

2.5 TCP Connection Termination 12

2.6 TCP Header 14

2.7 TCP Congestion Control 20

2.8 Real operational network 21

2.9 Network emulator 22

2.10 Dummynet pipes 23

3.1 Fast Retransmit inhibitions 31

3.2 Experiment overview 32

3.3 Environment overview 39

3.4 Environment detailed 39

3.5 Dummynet Configuration 42

3.6 Loss pattern 42

3.7 Tcpdump 44

4.1 Performance loss for the short flows 46

4.2 Average performance loss for the short flows 47

4.3 Transmission time graph 40 Kbit/s 48
4.4 Transmission time graph 80 Kbit/s 49
4.5 Transmission time graph 160 Kbit/s 50
4.6 Transmission time graph 320 Kbit/s 51
4.7 Transmission time graph 500, 1000, 2000, 4000, 8000, and 10000 Kbit/s . . 52
4.8 Transmission time graph 1000 Kbit/s, low delays 53
4.9 Bad RTO estimation 54
4.10 Acknowledgment misinterpretation 56
4.11 Server with 320 Kbit/s, 300ms delay, no losses 59
4.12 Server with 320 Kbit/s, 300ms delay, loss on position eight 59
4.13 Server with 320 Kbit/s, 300ms delay, loss on position nine 59
4.14 Server with 320 Kbit/s, 300ms delay, loss on position ten 60
4.15 Performance loss for the long flows 60
4.16 Average performance loss for the long flows 61
4.17 Transmission time graph 40, 80, and 160 Kbit/s 63
4.18 Transmission time graph 500 Kbit/s 64
4.19 Transmission time graph 1000, 4000, and 10000kbps 65
4.20 Server with 80 Kbit/s, 300ms delay, loss on position 12 68
4.21 Server with 80 Kbit/s, 300ms delay, loss on position 50 68
4.22 Server with 80 Kbit/s, 300ms delay, loss on position 84 69
4.23 Server with 500 Kbit/s, 300ms delay, loss on position 41 70
4.24 Server with 500 Kbit/s, 300ms delay, loss on position 50 70
4.25 One controlled loss, 10 ms 71

B.1 Tcptrace Symbols 85
B.2 Tcptrace Example 88

List of Tables

2.1 TCP Flags 15

3.1 Test case parameters 35

B.1 Tcptrace symbol explanation 87

C.1 TCP parameters 94
C.2 Default values of TCP parameters 95
C.3 TCP constants 98

Listings

3.1 Experiment detailed . . . 33
4.1 sys/net/inet/tcp_input.c, lines 1983–1999 . . . 66
C.1 sys/net/inet/tcp_input.c, lines 2744-2746 . . . 96
C.2 sys/net/inet/tcp_input.c, lines 2762-2763 . . . 96
C.3 sys/net/inet/tcp_timer.h, lines 129-135 . . . 97
C.4 sys/net/inet/tcp_var.h, lines 341-344 . . . 97
C.5 sys/net/inet/tcp_hostcache.c, lines 130-134 . . . 98
C.6 sys/net/inet/tcp_hostcache.c, lines 130-134, modified . . . 98
D.1 Loss pattern generator . . . 100
D.2 Configuration script . . . 107
D.3 Experiment execution script . . . 113
D.4 Client application source code . . . 114
D.5 Server application source code . . . 117

Chapter 1

Introduction

For many years TCP [29] has been the most commonly used transport protocol in the Internet. TCP is currently used by a large set of network applications like web browsers, FTP clients, and online computer games.

Because of recent years' network evolution, including the mobile communication revolution, the performance improvements of networking hardware, and the continuing growth of the Internet, the demands that a transport protocol is facing have become more and more complex. In order to meet these demands a lot of research and refinements have been done on TCP, making it one of the most complex transport protocols of today.

One of the reasons to why TCP is so popular is that it guarantees reliable data transfer to the applications using it. Other transport protocols, like UDP [28], do not provide this guarantee and are therefore unsuitable for applications that require that data arrives safely at the receiver.

Another reason to the popularity of TCP is that it contains mechanisms that automatically try to prevent networks from being congested. As computer networks, especially the Internet, began to grow large in the late 80's the problem of network congestion was discovered. Basically this problem, congestion, occurs when parts of the network infrastructure becomes over-utilized by large amounts of traffic and, therefore, are unable to deliver any data.

Since the incorporation of congestion control in TCP, the reliability mechanisms have been combined with some of the mechanisms that provide the congestion control. This is done because of the fact that possible unreliability of a network in most cases is due to congestion, and therefore it is natural that some of these mechanisms are combined. The exact behavior and interaction between various TCP mechanisms can potentially have a large impact on performance. In this work we focus on the examination of one mechanism, fast retransmit.

Fast retransmit is both a reliability mechanism as well as a congestion control mechanism. Its primary purpose is to resend data that has been lost in the network, but it also gives TCP a hint about the current congestion status in the network. As the name indicates fast retransmit is a fast way of retransmitting data, much faster than the other reliability mechanism, i.e. timeout, that TCP uses. But one of the problems regarding the fast retransmit mechanism is that it can not be employed under certain circumstances, due to factors that have to do with the TCP specification. This phenomenon is called fast retransmit inhibitions, and the primary goal of this thesis is to investigate when these inhibitions occur, and how much they affect the performance of a TCP session.

1.1 Scope of work

This thesis presents an experimental evaluation of fast retransmit inhibitions in an emulated network environment, using the FreeBSD 6 TCP implementation. The congestion control mechanisms in this TCP implementation are based on NewReno [9]. Furthermore, this TCP version employs a bandwidth limiting feature called “Bandwidth Delay Product Window Limiting” which was disabled for the experiments.

We consider a simple model of a single TCP flow between a client and a server. By controlling the existence and placement of data loss, within this flow, our goal is to investigate where fast retransmit inhibitions occur, in a flow, and how much they affect the performance of the TCP session.

For the experiments we have used a large set of network parameters, such as different bandwidths, end-to-end delays, and flow sizes. In order to conduct this evaluation a real-time network emulator was used to represent the network that the client and the server communicate over. Furthermore, a number of existing applications & scripts was modified and some new developed to automate the experimental process, in terms of configuring the network emulator according to desired network parameters, and executing the actual experiments.

Although the main intention of this work was to examine the existence and impact of fast retransmit inhibitions, other, interesting, results that are related to the behavior and performance of TCP are also presented in the thesis.

1.2 Disposition

The rest of the thesis is structured as follows. Chapter 2 presents background information on TCP, network emulation, and other work that in some ways are related to the work presented in this thesis. The first two sections in the chapter give an overview of how TCP manages connections between different hosts, transfers data, and how the reliability and the congestion control mechanisms are designed. The following two sections give a quick overview of the concept of network emulation and how it can be used, and a summary of some related work that have been conducted in this area.

In Chapter 3 the problem of fast retransmit inhibitions is discussed in more detail. In addition to this, the experiment design, environment, and the tools needed to do the experiments are presented. In the first part of this chapter some theoretical background on the fast retransmit inhibitions are provided along with an extended problem description and the method that was used for the experiments. The rest of this chapter contains more detailed information about the parameters of the experiments, and the environment that the experiments were conducted in.

Chapter 4 presents the results of the experiments. This chapter is divided into two separate parts; one for short TCP flows, and one for long ones. Each of these parts begins with an overview of how much the performance suffers in the presence of fast retransmit inhibitions. The parts then continue with a more detailed view of how the positioning of data loss affects the total transmission time of a TCP flow. This is followed by a number of sections that analyzes particularly interesting results, and finally, in the end of this chapter some of the results are compared to results that have been published in a related study.

In Chapter 5, further work that might be conducted are presented. This chapter contains ideas on how to use different evaluation techniques to investigate fast retransmit inhibitions. It also contains suggestions of work that might be done given the results in this thesis.

Finally, in Chapter 6, the work that is described in this thesis is summarized. This chapter provides a quick resumé over the work, a summary of the most important results, and mentions some interesting phenomena that were discovered during the work.

To aid the reader of this thesis, a list of abbreviations is provided in Appendix A.

Chapter 2

Background

This chapter provides background information relevant to this project's area. The focus of this chapter is on TCP details, but an introduction to network emulation is also provided, as well as a section with related work.

2.1 TCP - Transmission Control Protocol

The Transmission Control Protocol [3, 29] (TCP) is the most used transport protocol in the Internet today. It is a part of the TCP/IP protocol suite which allows computers, regardless of operating system and hardware, to communicate with each other. One of the major properties of TCP is that it is able to provide a connection-oriented data transfer service that is reliable to applications who require that no data is lost and/or damaged in the communication process.

TCP is used in conjunction with the Internet Protocol [28] (IP) which only provides an unreliable conectionless data transfer service between different hosts. To be able to provide conection-oriented reliable communication, TCP needs to implement mechanisms on top of IP. Figure 2.1 shows how two different processes, P and Q, located on two different hosts use TCP to communicate with each other.

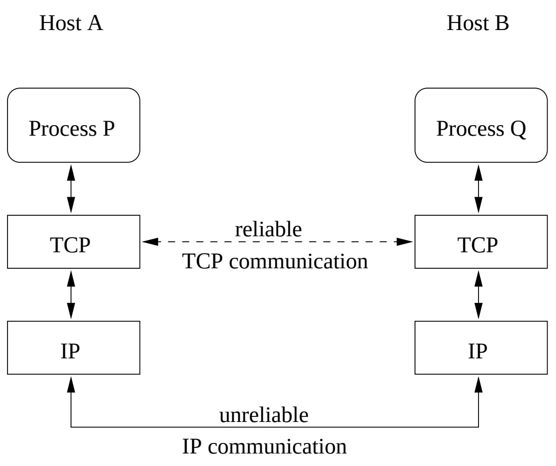


Figure 2.1: TCP Communication

2.1.1 TCP Areas

As mentioned before, the primary purpose of TCP is to provide a connection-oriented reliable data transfer service between different applications. To be able to provide this service on top of an unreliable communication systems TCP needs to consider the following areas [3, 29]:

- Data Transfer
- Reliability
- Flow Control
- Multiplexing
- Connection Management
- TCP Segments
- Congestion Control

These areas are discussed below, with special attention given to congestion control.

2.1.2 Data Transfer

TCP is able to transmit a continuous byte stream in each direction between its users. To achieve this TCP packages the data that is about to be sent into segments and then transmits them to the other end.

The sending TCP is permitted to send data that is submitted by the user at its own convenience, but in some cases the user wants to make sure that all data submitted to the TCP have been transfered to the receiver. For this purpose a push operation is available. When this operation is used, the sending TCP sends all the remaining data to the receiver, which in turn must pass the data immediately to the receiving process. To make it possible for the receiving TCP to know if the data received is to be delivered immediately TCP specifies a `PUSH` flag.[1]

2.1.3 Reliability

To be able to provide a reliable data transfer TCP must recover from data that is lost, damaged, received out-of-order, or duplicated during the end-to-end transfer. This is made possible by assigning each transmitted segment a sequence number, and requiring an acknowledgment from the receiving TCP. This sequence number is sent along with the actual data. Figure 2.2 shows how a sender (`host1`) transmits data to a receiver (`host2`) which in turn acknowledges this data.

At the receiving TCP, the sequence numbers are used to eliminate duplicate segments, and to reorder segments that have been received out-of-order. If a segment arrives out-of-order, the receiving TCP sends a duplicate acknowledgment back to the sender. A duplicate acknowledgment contains information about which segment that was supposed to be received, and by its presence it also tells the sender that another segment was received by the other side.[2]

[1]The TCP header flags are covered in Section 2.1.7

[2]Duplicate acknowledgments can only be generated if a segment arrives at the receiver.

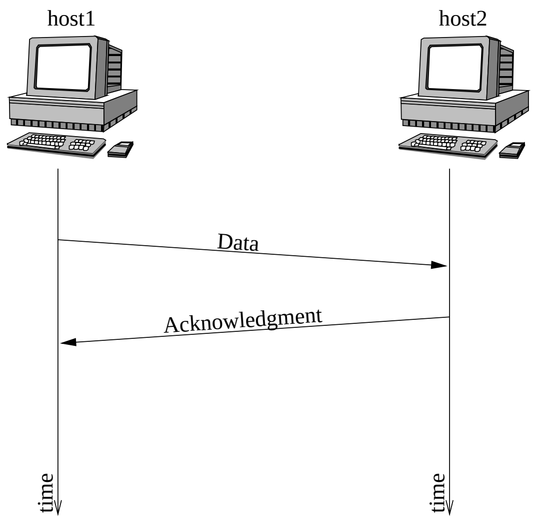


Figure 2.2: Acknowledgment

To implement the actual reliability TCP uses two different techniques;

1. If an acknowledgment to a transmitted segment is not received within a certain time interval, the segment is retransmitted. This event is often called Retransmission TimeOut (RTO) and the timer that keeps track of the interval is recalculated, on a regular basis, according to delay in the network. To effectively estimate the delay the round-trip times[3], RTTs, of transmitted segments are used as a basis for the calculation.

2. If three, consecutive, duplicate acknowledgments are received the corresponding segment is retransmitted. This technique is called fast retransmit, and is described in more detail in Section 2.2.3. This technique is faster than the previous, because TCP is not required to wait for the RTO timer to expire before retransmitting the segment.

TCP must also handle damaged segments. This is done by calculating a checksum of

[3]One round-trip time is equivalent to the time that it takes for one segment to be sent and then acknowledged.

each transmitted segment which the receiver must verify. Damaged data is discarded by the receiver and the recovery relies on the two retransmission techniques mentioned earlier.

2.1.4 Flow Control

In some situations data is received faster than it is consumed by the application using TCP. When this happens TCP buffers the incoming data so that the application can read the data when it needs it. To avoid that the buffer runs out of space TCP has a flow control mechanism. This mechanism provides means for the receiving TCP to control the amount of data that is sent by the sending TCP by including a "window" with every acknowledgment. This window, called the receiver window, contains an acceptable range of sequence numbers (beyond the last successfully received segment) that may be sent by the sender. Figure 2.3 shows how the sender uses this window in the sending process. Outside the window, on the left, we can see three segments that have been sent and acknowledged by the receiver. In the first part of the window there are three segments that have been sent but not yet acknowledged. The second part of the window contains segments that can be sent right away, and outside the window, on the right hand, we have segments that can not be sent before the window slides. As long as the window offered by the receiver is constant the receipt of an acknowledgment will make the window slide one position to the right for each acknowledged segment.

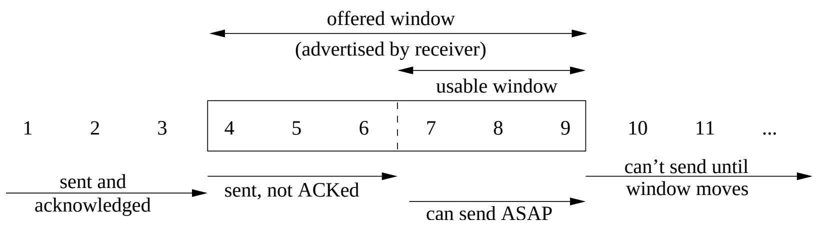


Figure 2.3: Sliding window

There are situations when the TCP receiver has nothing to acknowledge but still needs to send an update of the receiver window. If the last advertised receiver window had zero size, i.e. the receiver's buffer is full, and the buffer space is beginning to be freed the receiver must have some way of reporting this to the sender (to prevent deadlock). This is solved by the use of window updates. A window update is simply an acknowledgment that does not acknowledge the receipt of any new data, only advertises a new receiver window, and is sent when the buffer opens up.

2.1.5 Multiplexing

To allow more than one application per computer to use TCP simultaneously TCP uses a multiplexing service. This service enables TCP to demultiplex incoming data so that every application using TCP receives its' "own" data. To accomplish this TCP specifies that every process using TCP must be assigned a port. This port concatenated with the network address of the host forms a socket. A socket may simultaneously form pairs with a number of different sockets but every pair of sockets uniquely identifies a connection between two processes. The binding of ports to processes is handled independently by each host machine, but some ports are often bound to a specific type of process. Examples of this is FTP client-/server processes which often use a port number of 21.

2.1.6 Connection Management

As described above the unique identifier of a connection is a pair of sockets, but there is more to it. Although sockets can be used to identify a connection, a connection also consists of certain status information. TCP is required to initialize and maintain this information, including sockets, sequence numbers, and window sizes. This information is used by the reliability, flow control, and congestion control mechanisms.

To realize the communication between two processes, their TCP's must establish a connection. The connection establishment is the initialization of the status information (on each side). Later, when the communication is complete, their TCP's must close the connection, freeing the resources for other use.

Because of the fact that the connection establishment is undertaken over a potentially unreliable communication system, a certain form of handshake must be conducted between the two TCP's. This handshake, called a three way handshake, synchronizes the status information that needs to be shared between the two TCP's (including window sizes, sequence numbers). An example of a connection establishment is shown in Figure 2.4.

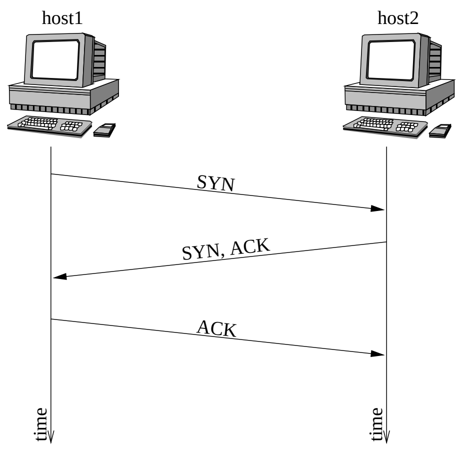


Figure 2.4: Three Way Handshake

1. `host1` sends a segment with the `SYN` flag set. This flag tells `host2` that the sender wants to initiate a connection (synchronize connection information).

2. `host2` accepts the connection invitation by also sending a segment with the `SYN` flag set. It also acknowledges the `SYN` from `host1`.

3. `host1` acknowledges the response.

After these three segments have been correctly received, a "full duplex"[4] connection between the two is established.

To terminate a TCP connection, both sides must issue a termination request to the other. This is illustrated in Figure 2.5.

[4]Full duplex means that data can flow in both directions simultaneously.

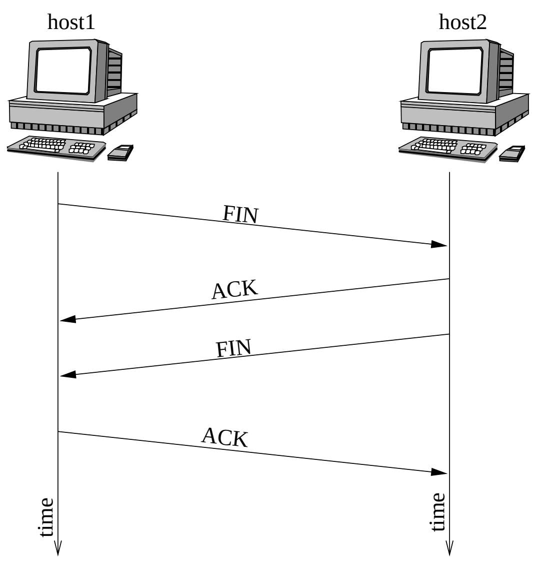


Figure 2.5: TCP Connection Termination

1. `host1` sends a segment with the `FIN` flag set. This flag tells `host2` that the sender wants to terminate the connection.

2. `host2` acknowledges the `FIN` and sends a segment with `FIN` flag set. This tells `host1` that `host2` is also ready to terminate the connection.

3. `host1` acknowledges the `FIN`.

2.1.7 TCP Segments

Each TCP segment that is transmitted contains a header and, in most cases, data. Figure 2.6 shows the format of a TCP segment. The first two fields (source and destination ports) are used in order to determine which process that should receive/has transmitted the data (see Section 2.1.5). The third and fourth field contain the sequence number information that is used by the reliability mechanism of TCP (see Section 2.1.3). The 4-bit Data Offset field is used to indicate where the actual data is located in the segment. This is necessary because the Options field in TCP segments can be of different lengths. The Reserved field has a length of 6-bits and is not currently used as it is intended for future use. The small fields, occupying one bit each, are called flags and are summarized in Table 2.1. The 16-bit window field is the receiver window that was described in Section 2.1.4. The next field, the 16-bit checksum field, contains checksum information that can be used to determine if the segment has been damaged (see Section 2.1.3). The 16-bit urgent pointer field contains a pointer to data that is urgent. This field is only used if the `URG` flag is set. The Options field can, for example, contain time stamps, and/or information about window scaling [12].

Because the Options and Data fields are not necessarily used, the minimum size of a TCP segment can be 20 bytes. The maximum segment size (MSS), however, is determined in the connection establishment by a negotiation between the different hosts. This negotiation is normally based on the maximum transfer unit of the underlying IP protocol at both hosts. The maximum transfer unit, or MTU, is the maximum size of a IP datagram, and if a TCP segment is larger than this value it will be fragmented into multiple IP datagrams. To avoid fragmentation issues[5] hosts normally calculate their maximum segment size based on the MTU, and the smallest segment size that one of the hosts wants to use, to avoid fragmentation, will be used as the maximum segment size (MSS) throughout the connection (by both hosts).[6]

[5]One issue that comes with fragmentation is the re-assembly of fragmented datagrams which can be time consuming.

[6]The negotiation is conducted by the use of the Options field.

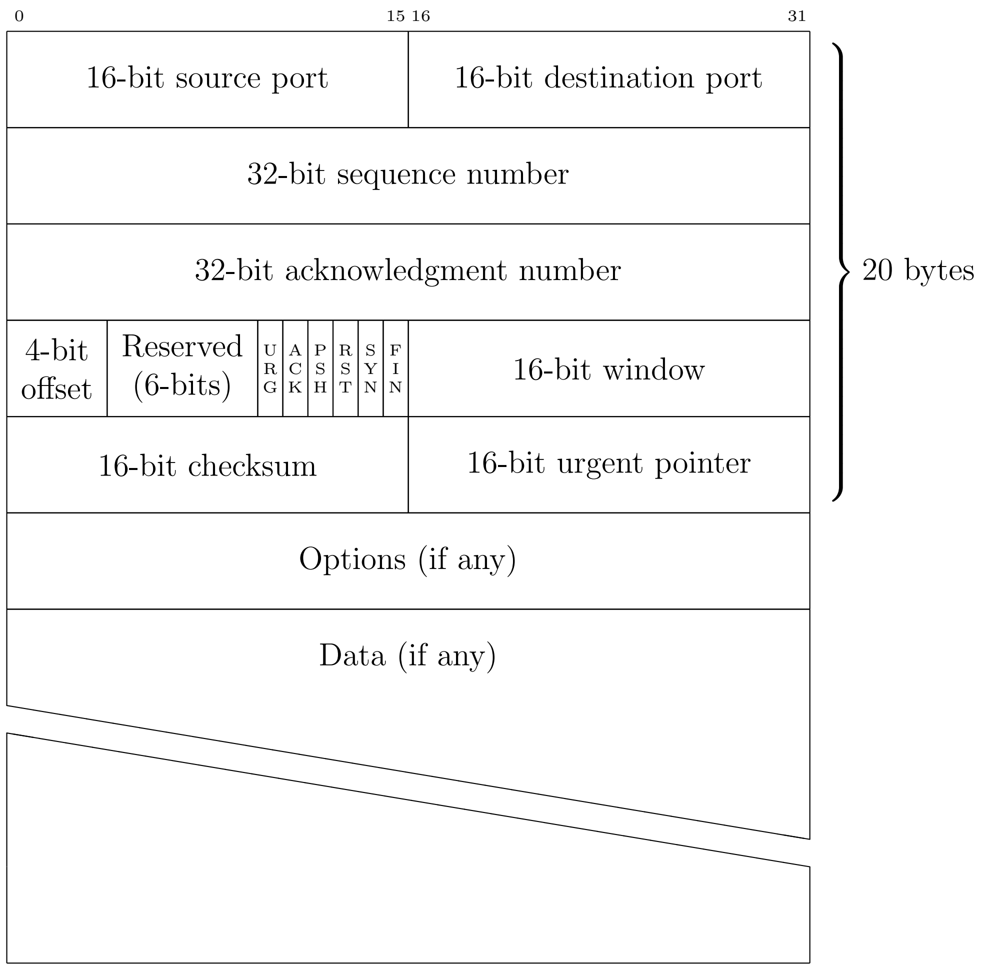


Figure 2.6: TCP Header

Name	**Description**
`URG`	Urgent pointer. Indicates that the segment contains urgent data which is pointed at by the urgent pointer.
`ACK`	Acknowledgment. Indicates that the acknowledgment number field contains the sequence number of the next expected segment.
`PSH`	Push. Indicates that the data should be delivered immediately to the receiving process.
`RST`	Reset. Resets the connection.
`SYN`	Synchronize. Synchronize sequence numbers.
`FIN`	Finish. No more data from sender.

Table 2.1: TCP Flags

2.2 Congestion Control Detail

When a router in a network can not distribute packets as fast as it receives them, it starts filling up its internal buffers. If the arrival rate continues to be higher than the sending rate, the router will eventually start to drop packets. This is called congestion. The network itself has no means of telling the hosts on the network that congestion has occurred, so the TCP protocol must rely on other information to detect congestion. TCP solves this by interpreting packet loss as a sign of network congestion,[7] and as already mentioned (in Section 2.1.3) TCP uses two different mechanisms for detecting packet loss. One mechanism is timeout, which occurs when sent data is not acknowledged by the receiver in time. The other mechanism is the reception of three duplicate acknowledgments.

In order to deal with the problem of congestion TCP uses two different mechanisms; Slow Start which is described in the next section, and Congestion Avoidance introduced in the section thereafter.

[7]Packet loss could also be due to damage, but normally only a small amount of packet losses are due to damage, so the assumption of congestion is considered to be safe.

2.2.1 Slow Start

In early TCP implementations, a sender was allowed to send multiple segments (up to the advertised receiver window size) in the start of a connection in order to quickly reach the capacity of the link. The situation is similar to the situation of pouring water, fast, through a regular pipe. No one would consider the idea of pouring a small fraction of the water and wait and then pour a little more, and so on. The approach that most people would choose is to simply pour the water as fast as possible through the pipe. For computer networking, this approach may work very well if the two hosts share a LAN, or if the links and routers between them are underutilized. If, on the other hand, it exists slower links or routers which are under high utilization between them, this approach can lead to congestion.

To avoid this TCP uses an algorithm called slow start. Slow start is conducted in the beginning of every TCP connection and its main purpose is to find the maximum available bandwidth at which it can send data without causing the network to be congested. To realize this, slow start forces the TCP sender to transmit at a slow sending rate and then rapidly increasing it until the available bandwidth between the hosts is believed to be found.

The slow start mechanism introduces a new window to the sender's TCP: the congestion window (often referred to as `cwnd`). When a new connection is established the `cwnd` is initialized to $0 < \texttt{cwnd} \leq \min(4 * \texttt{MSS}, \max(2 * \texttt{MSS}, 4380 \text{ bytes}))$ [5].

Each time an acknowledgment is received the size of `cwnd` is increased by one segment, allowing the sender to transmit two new segments. This approach will lead to an almost exponential growth of the `cwnd`. Even though this strategy causes the `cwnd` to grow large very quickly the sender is never allowed to transmit more data than the receiver advertised window allows (see Section 2.1.4), even if the `cwnd` is larger.

Eventually some intermediate router will not be able to handle this growing traffic flow, without dropping packets. When this happens TCP will interpret the lost packets as a sign of congestion and enter congestion avoidance.

2.2.2 Congestion Avoidance

If the receiver window is large enough, the slow start mechanism described in the previous section will eventually increase the congestion window (`cwnd`) to a point where one or more routers in between the hosts will start discarding packets. As mentioned earlier (see Section 2.2) TCP interprets packet loss as a sign of congestion, and when this happens TCP invokes the Congestion Avoidance mechanism.

Even though slow start and congestion avoidance is two different mechanisms they are more easily described together. In the joint description below a new TCP variable is introduced. This variable, `ssthresh`, is the slow start threshold which TCP uses to determine if slow start or congestion avoidance is to be conducted.

1. When establishing a new connection `cwnd` is initialized to

$$0 < \texttt{cwnd} \leq \min(4 * \texttt{MSS}, \max(2 * \texttt{MSS}, 4380 \text{ bytes}))$$

2. The sender side TCP sends a maximum of

$$\min(\texttt{cwnd}, \texttt{rwnd})\text{bytes}$$

3. When congestion occurs

$$\texttt{ssthresh} \leftarrow \min\left(\frac{\min(\texttt{cwnd}, \texttt{rwnd})}{2}, 2 * \texttt{MSS}\right)$$

   If congestion was due to a timeout slow start is conducted.

4. When new data is acknowledged by the other end `cwnd` is increased. The way in which TCP increases the `cwnd` depends on if we are doing slow start ($\texttt{cwnd} < \texttt{ssthresh}$), or congestion avoidance. The increase of `cwnd` in slow start was described in the

previous section, and if we are doing congestion avoidance then $\texttt{cwnd} \leftarrow \texttt{cwnd} + \frac{1}{\texttt{cwnd}}$, which results in a linear increase of the `cwnd`.

2.2.3 Fast Retransmit

If an out-of-order segment is received TCP generates a so called duplicate acknowledgment (see Section 2.1.3). This duplicate acknowledgment is sent immediately from the receiver to the sender indicating that a segment arrived out-of-order, and which segment that was supposed to be received.

Since it is not possible to know whether the duplicate acknowledgment was caused by a lost segment or just reordering of segments, the sender waits for three duplicate acknowledgments before retransmitting the segment. If this limit would have been lower, this would increase the chance of reordered segments causing duplicates to be created, and transmitted needlessly.

The advantage of this mechanism is that TCP does not have to wait for the retransmission timer to expire, it simply assumes that three duplicate acknowledgments is a good indicator of a lost segment.

2.2.4 Fast Recovery

After fast retransmit is conducted, congestion avoidance and not slow start is performed. This behavior is called Fast Recovery. Fast recovery is an algorithm allows for higher throughput under congestion, especially when using large congestion windows.

Receiving three duplicate acknowledgments tells TCP more than the expiration of the retransmission timer. Since the receiving TCP only can generate duplicate acknowledgments when it is receiving other segments it is an indication that data still flows between the different hosts, and that the congestion is not that severe. By using this approach, skipping the slow start, the TCP does not reduce the transfer rate unnecessarily much.

When implemented, the two algorithms (fast retransmit and fast recovery) works in the following way (all actions are described from the sending TCPs point of view).

1. For the first two consecutive duplicate acknowledgments received the Limited Transmit algorithm [4] is used:
   a) If the receiver's advertised window allows; send one new segment, but do not change the `cwnd`.
2. Third consecutive duplicate acknowledgment received:
   a) $\texttt{ssthresh} \leftarrow \max(\frac{\texttt{cwnd}}{2}, 2 * \texttt{MSS})$.
   b) Retransmission of the missing segment is performed.
   c) $\texttt{cwnd} \leftarrow \texttt{ssthresh} + 3 * \texttt{MSS}$.[8]
3. More duplicate acknowledgments arrive:
   a) Increment `cwnd` by the segment size for each arriving duplicate acknowledgment.
   b) If the new value for `cwnd` permits, send a new segment.
4. The acknowledgment of the previously lost (or assumed lost) segment arrives.[9]
   a) $\texttt{cwnd} \leftarrow \texttt{ssthresh}$.

The `cwnd` is in a sense "inflated" during the error recovery in steps 2c and 3a. This is done to have the possibility to send more segments in step 3b. When the error recovery finishes, the "inflated" `cwnd` is "deflated" back in step 4.

[8]This inflates the congestion window by the number of segments that have left the network and which the other end has cached (3).

[9]This acknowledgment should also acknowledge all the intermediate segments that were sent.

2.2.5 Congestion Control Summary

The mechanisms described earlier in this section; slow start, congestion avoidance, fast retransmit, and fast recovery are summarized in Figure 2.7. Please see [3, 5, 9] for the complete specification of the TCP congestion control.

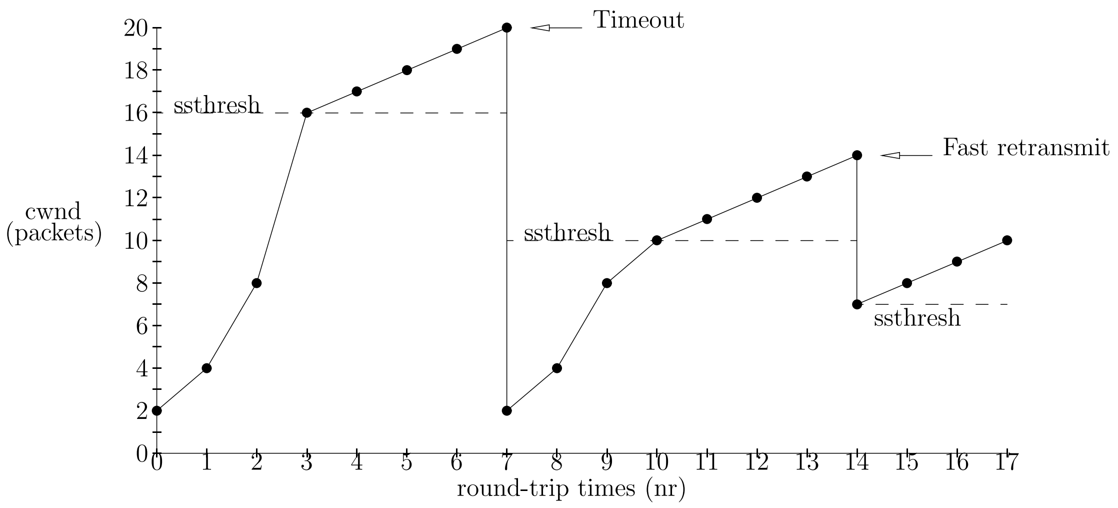


Figure 2.7: TCP Congestion Control

The first part of the graph shows how slow start is conducted until the slow start threshold, `ssthresh`, is reached. TCP then enters congestion avoidance. According to the figure a retransmission due to timeout is experienced at round-trip time number 7. When this happens the `ssthresh` variable is set to half its current value and slow start is performed. At round-trip time number 14 fast recovery is entered due to the fast retransmit.

2.3 Network Emulation

Because of the complexity of todays networks, especially the Internet, transport protocols have become more and more complex. Because of the increase in complexity it is nowadays hard, if possible at all, to fully grasp the behavior and performance issues of a protocol without evaluating it under controlled conditions. Luckily, there are several methods of evaluating transport protocols. These methods include analysis, simulation and experiments with real protocol implementations. Even though analysis and simulation can be used with advantage, experiments with real protocol implementations are attractive because they more accurately reflect how transport protocols that are used in real operational networks behave. Experiments with real protocol implementations can be done in two different ways, using a real operational network or using network emulation.

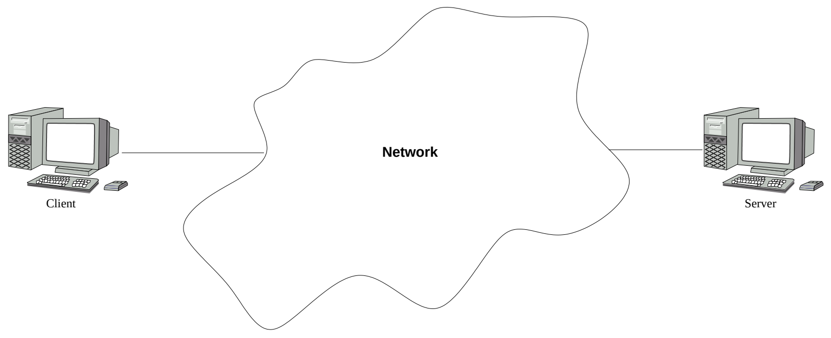


Figure 2.8: Real operational network

In a real operational network, illustrated in Figure 2.8, it is often hard (if possible at all) to monitor and control the network parameters that can have impact on the experiments. These parameters include delays, bandwidths, queue sizes, packet losses and external traffic sources. If the relationship between an experimental result and a collection of network parameters is to be investigated, there can be problems if one, or more, of these parameters are unknown or uncontrollable. It is much easier to investigate such a relationship if the parameters are available. In addition to this it is also very hard to control and reproduce experiments if no control over the network parameters are provided.

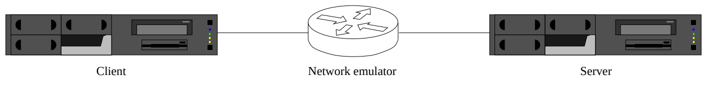


Figure 2.9: Network emulator

By using a network emulator (see Figure 2.9) to emulate various network conditions, instead of using a real operational network, a large subset of these network parameters can be controlled and evaluated. Network emulators typically provide facilities for setting bandwidth restrictions, introducing delays, reordering packets, dropping packets, and managing queues.

The next section describes one such network emulator, called Dummynet, which was used for the work described in this thesis.

2.3.1 Dummynet

Dummynet [34, 35] is a network emulator software that is included as a part of the FreeBSD kernel [10]. Even though Dummynet originally was developed to work solely as network emulator software it has been used for other purposes as well (one popular use is bandwidth management). To be able to emulate the conditions of a real operational network, Dummynet intercepts the network communication of the FreeBSD protocol stack to simulate queue and bandwidth limitations, delays, packet loss, and multi-path effects.[10]

Dummynet can be configured with the FreeBSD firewall configuration program `ipfw`. Using this program one can create so called Dummynet pipes. By using the IP filtering mechanisms of the FreeBSD firewall the user can specify that certain traffic, such as all TCP traffic from a certain IP address destined for another IP address, is placed in one of these pipes. The pipes can then be configured with Dummynet parameters in order to emulate certain network characteristic such as bandwidth limitations, delays, packet loss, and so on. Figure 2.10 shows two Dummynet pipes that are configured to carry the TCP traffic between two hosts applying some emulation effects.

[10]One of the most common multi-path effects is packet reordering.

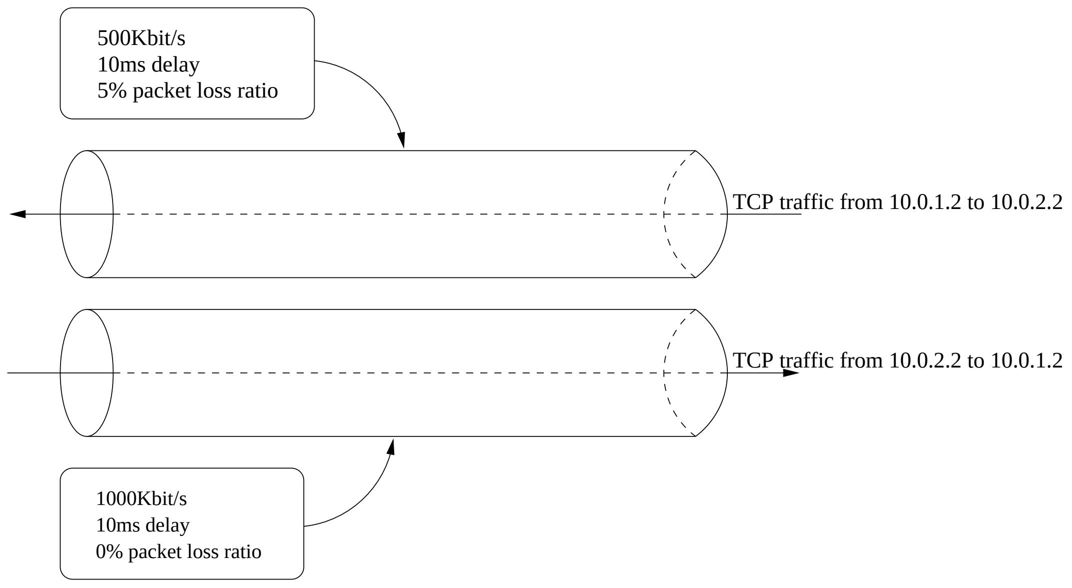


Figure 2.10: Dummynet pipes

2.4 Related Work

Ever since the original TCP congestion control was specified by Van Jacobson [13] researchers have proposed enhancements to it, and some of these have become Internet standards adopted by the IETF.[11] One of the major enhancements was the Reno/NewReno versions of TCP, which includes the fast retransmit mechanism [3, 9], that today is believed to be used in a majority of the web servers on the Internet [2].

As mentioned in Section 2.1.3, fast retransmit is triggered by the receipt of three duplicate acknowledgments which in turn are generated by reordered TCP packets. In [15] experiments show that packet reordering that is not caused by packet loss is a common phenomenon in todays networks. In [15] it is argued that this can lead to performance issues, as unnecessary fast retransmissions wastes bandwidth and quickly decreases the congestion window. This problem is addressed in [7] and [8] which both propose mechanisms for detecting, and recovering from, false fast retransmissions. These studies also

[11]IETF, or the Internet Engineering Task Force, is a large open international community of people that are engaged in the development of standards concerning the Internet.

propose mechanisms for increasing the duplicate acknowledgment threshold dynamically if reordering of segments is detected. Other work that has been conducted in the same area [27] argues that even though packet reordering is a common phenomenon the effects of it is not that severe and therefore the duplicate acknowledgment threshold could be lowered to two in order to gain more fast retransmit opportunities. A lowering of the duplicate acknowledgment threshold is also proposed in [19], but in this case the lowering is dynamic and happens in two different scenarios; when the amount of data in flight is not enough to enable the fast retransmit mechanism[12], or when the sender is idle.[13]

Other work related to increase the fast retransmit opportunities is done in [20, 21] which specifies a TCP segmentation mechanism called TCP-SF. By fragmenting the packets so that at least three packets always are in flight, TCP-SF aims to enable fast retransmit even when the congestion window is too small to normally allow this.

Even though fast retransmit is the preferred way of retransmitting lost data, retransmissions due to timeout are still common. To gain performance a large number of TCP hosts [2] uses TCP implementations which allow a minimum RTO that is less than the specified standard of one second [26]. This is a potential problem, especially for TCP hosts that operate over low bandwidth links and have large initial windows [5], as it can result in spurious retransmissions which considerably lower the performance of the connection. Even if the minimum RTO is set according to the standard, spurious retransmissions can occur due to different path characteristics. To avoid that these, unnecessary, retransmissions lower the performance, changes to the TCP protocol have been proposed in [6, 32][25]. In [6, 32] the Eifel Algorithm is presented. This algorithm aims to improve TCP performance by restoring the TCP congestion state after a spurious retransmission is detected. It also adapts the RTO timer in order to prevent further spurious retransmissions. F-RTO, described in [25], is a proposed sender side modification which is similar to the limited

[12]To generate three duplicate acknowledgments at least three packets must be on the way to the receiver so that three duplicate acknowledgments can be generated.

[13]Due to receiver window limitations or that no unsent data is ready for transmission.

transmit algorithm, but instead applied to the recovery from timeouts.

Besides the work on avoiding timeouts, in favor of fast retransmit, and recovering from spurious retransmissions, a lot of research has been conducted due to the increase of network paths with high bandwidth and delay. One of the major problems with TCP is that it performs poorly when both the bandwidth and the delay increases. To solve this problem different techniques have been proposed; Adaptive start (Astart) [33] which is a proposed modification of the TCP slow start phase to better utilize the link with the help of bandwidth estimation techniques, and Scalable TCP [16] which is a modification to the congestion control of TCP that allows for better link utilization by allowing a faster growth of the congestion window.

Even though the congestion window nowadays can be very large [12], thus allowing a large amount of data in flight, some TCP implementations are unable to benefit from it. The size of the TCP send buffer can effectively limit the amount of data in flight, as copies of unacknowledged segments must be kept in the buffer, in case retransmissions are necessary. The effective congestion window for TCP implementations that have statically sized send buffers are therefore given by

$$\texttt{cwnd} = \min(\texttt{cwnd}, \texttt{buffersize})$$

To avoid that the size of the buffer lowers the performance of a TCP connection several proposals have been made. In [30] a method for dynamically changing the socket buffer size is described. This method tries to calculate the bandwidth that is available for the connection, and then adjust the buffer size so that it is appropriate. In [38] a daemon called Work Around Daemon (WAD) is described. WAD tries to optimize the performance of different TCP flows by manipulating the buffer sizes. WAD also has mechanisms for tuning other parameters, like the maximum slow start threshold. In [14] kernel modifications for supporting dynamic send and receive buffer tuning are presented. This method has three mechanisms for determining the appropriate size of the buffers; the first mechanism determines the buffer size according to the current network conditions, the second takes the memory management into account[14], and the third asserts a memory usage limit to prevent that too much memory is used. One operating system that includes dynamic buffer sizing is Linux [39] which automatically increases the send buffers when there is need for it.

Although a lot of research has been conducted on TCP, little has been done with a deterministic emulation approach. Common network emulators, like Dummynet, usually provide means for generating packet losses, but only in non-deterministic ways like randomly induced losses and overflow of emulated buffers.

According to [11] non-determinstic evaluation is not enough to cover all aspects of transport protocol behavior, especially for short-lived flows the position of the loss within the flow has an impact on the throughput. This statement is proved by experiments done with an extended, deterministic, version of the Dummynet network emulator. This version allows for position based losses, and the results presented clearly show that the position of a loss has impact on the total transmission time.

Another feature of the deterministic emulation approach is, according to [11], that experimental results are reproducible. Using a typical network emulator it is not possible to place losses at the exact same location as in previous experiments, but with this deterministic approach losses can be placed in the exact same location every time. This makes results easier to reproduce and the variance in the results is less.

2.5 Summary

In this chapter the basic behavior of TCP has been described, in terms of data transmission, reliability mechanisms, and the congestion control mechanisms that are used. We have seen that TCP is a reliable transport protocol, which also tries to prevent over-utilization of the

[14]So that the memory is fairly shared between different TCP flows.

network it operates in. The chapter also included a short introduction to the concept of network emulation, and explained the basics of the Dummynet network emulation software. Finally, in the last part of the chapter, some work that is related to this thesis was presented. In the next chapter the experiment design is defined, and the environment in which the experiments were conducted is described.

Chapter 3

Experimental Design & Environment

In this chapter detailed descriptions of the experimental design, and the environment in which the experiments were conducted, are provided. The chapter starts with a description of the fast retransmit inhibition problem, and continues with details on how the experiments were designed. In the last part of this chapter the different parts of the experimental environment are described.

3.1 Problem Description

As described in Section 2.2, TCP congestion control consists of a number of different, and interrelated, mechanisms. One of these mechanisms is the fast retransmit. The purpose of the fast retransmit mechanism is to allow a sender to retransmit lost packets before they are regarded as lost by the TCP retransmission timer. Fast retransmit uses the receipt of three duplicate acknowledgments as an indicator of packet loss. Duplicate acknowledgments can be generated by the receiver for two different reasons; packet reordering in the network, or packet loss. Which of these reasons that causes the duplicate acknowledgment is hard for the sender to decide. To ensure that the fast retransmit mechanism does not mistakenly assume that a packet that has been reordered in the network was lost, and makes an unnecessary retransmission of it, a duplicate acknowledgment threshold is used. For TCP this threshold is set to three.

While the use of the duplicate acknowledgment threshold helps TCP to avoid retransmitting packets that have been mildly reordered in the network it also has negative consequences. One of these negative consequences is evident at the end of a connection when the application has written all its data to the transport layer. Since the fast retransmit mechanism requires the receipt of three duplicate acknowledgments as an indication of packet loss, it cannot work if there are less than three packets to send after the packet that has been lost. Thus, at the end of connections when there are not enough packets to send to generate three duplicate acknowledgments, the lost packet must be recovered using a retransmission due to timeout. A similar problem is also evident at the beginning of a connection when the three-way-handshake is conducted (see Section 2.1.6). If one of the first packets in this procedure is lost it is not possible to perform a fast retransmit because packets needed for duplicate acknowledgment generation simply does not exist. Allthough this problem can cause performance problems as well, it is not considered as a fast retransmit inhibition. This because it is not an effect of the duplicate acknowledgment threshold. An illustration of a client and a server that experience these problems is shown in Figure 3.1.

A timeout is intended to occur only if the network suffers from heavy congestion, therefore the recovery from such a timeout includes a considerable reduction of the sending rate[1] in order to reduce the (over)utilization of the network. Fast retransmit, on the other hand, uses a recovery mechanism that lowers the sending rate less compared to the timeout recovery. The reason why the sending rate is lowered less is that the receipt of the three duplicate acknowledgments indicates that packets in fact have left the network, indicating that the congestion is not that severe.

So, even if fast retransmit is suitable for the current network conditions, it is not possible

[1]By employing the slow start mechanism.

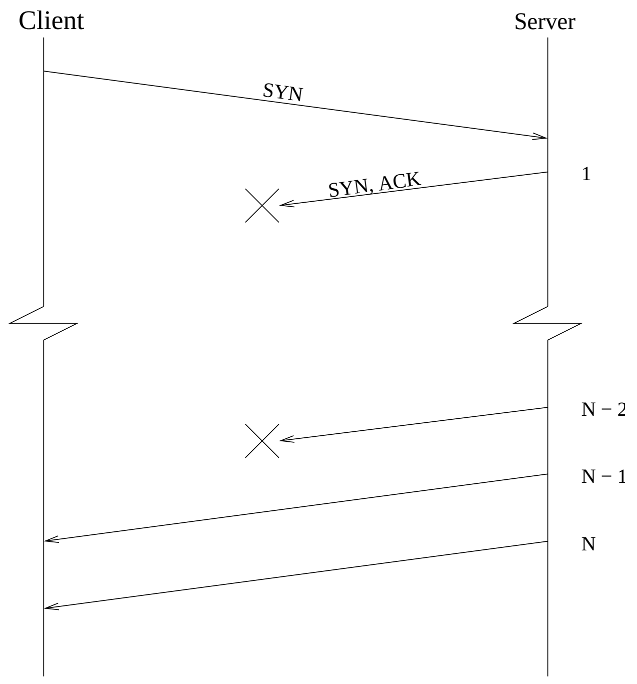


Figure 3.1: Fast Retransmit inhibitions

to use it if the packet loss occurs late in the connection. This fast retransmit inhibition may have impact on the total performance, in terms of transmission time, and the experiments described in this chapter are designed to reveal this, possible, performance issue.

Experiments on this phenomenon has been conducted earlier but only for a relatively small set of network parameters, and with an old TCP implementation [11]. The intention of these experiments is to investigate if this problem still is relevant (for a newer TCP implementation), and if so, investigate if the performance issue is relevant as well.

3.2 Method

3.2.1 Overview

In order to conduct the experiments a client, a server, and a network emulator with the ability to lose packets based on their position in a TCP flow were used. These, along with other components that were used for building the experimental environment are described in the next two sections.

As shown in Figure 3.2, the client initiates a connection with the server (via the network emulator), which in turn sends N packets back to the client and then terminates the connection. The client then logs the total transmission time that was required for receiving these N packets.

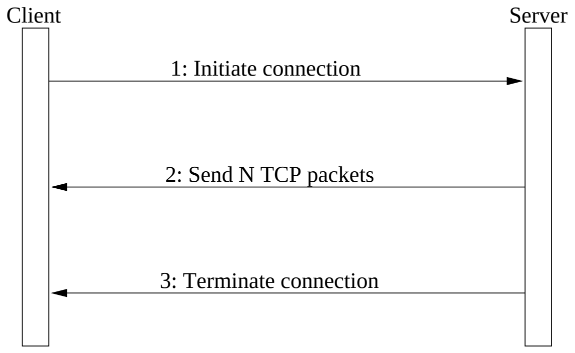


Figure 3.2: Experiment overview

In the first run no packets were lost, this in order to get a reference on how fast the transmission would be without any losses. In the second run the different procedures ($1-3$ in Figure 3.2) were repeated, but this time the first packet from the server to the client was lost. This behavior ($1 - 3$) then repeats itself until a packet loss has been imposed, individually, on all N positions. This was done in order to reveal how the total transmission time was affected by the placement of the loss and, if so, how much.

The procedure as a whole, losing a packet individually on all N positions, was then repeated for different combinations of bandwidths, end-to-end delays, and other parameters. All these parameters, and their values, are described and specified in Section 3.2.3.

3.2.2 Details

To provide a more detailed description about the experiments that were conducted, a pseudo version of the experiment script is shown in Listing 3.1. This script was used to initialize and configure the different parts of the experimental environment (which is described in the next section) and also to execute the actual experiments. The real version of this script can be found in appendix D.2. The variables that are present in the listing are all described and argued for in the following subsection.

```
foreach flowsize:
  start_server(flowsize);
  foreach bandwidth:
    foreach delay:
      foreach loss_position:
        foreach replication:
          configure_tcp(server, tcp_parameters);
          configure_tcp(client, tcp_parameters);
          config_emulator(bandwidth, delay, loss_position, queue);
          start_packet_sniffer(server);
          start_packet_sniffer(client);
          start_client(server);
          log_transmission_time();
```

Listing 3.1: Experiment detailed

- The server application was started and configured according to the flow size of the experiment. For each combination of bandwidth and delay the experiment was then conducted as follows;

- For every position within the TCP flow (loss_position) the following was done (with replication repetitions);
    1. The client and server machines were configured with TCP parameters.
    2. The network emulator was configured with bandwidth, delay, position of packet loss, and queue size parameters.
    3. Traffic loggers on the client and server machines were started.
    4. The client application was started (starting the experiment).
    5. The total transmission time of the experiment was logged.

3.2.3 Parameters

TCP parameters

As mentioned earlier, in Section 2.2, TCP makes use of two different phases during a connection; slow start and congestion avoidance. In both these phases the congestion window is steadily increasing until a packet loss is detected, then the congestion window is decreased and the procedure repeats itself. Instead of letting TCP cause congestion when probing for available bandwidth some TCP versions implement bandwidth limiting functionality that is supposed to decrease the sending rate when the available bandwidth is believed to be reached.

To determine the available bandwidth different TCP implementations use different techniques. The TCP implementation of FreeBSD 6 [10] (which was used for the experiments) has such a feature, called TCP bandwidth-delay product window limiting, which is enabled by default. The implementation of this bandwidth limiting functionality is similar to TCP/Vegas [17], which also tries to prevent congestion losses by adapting the throughput to the network conditions.

Even though such features may be useful to prevent network congestion from occurring, the primary goal of this work was to study what actually happens, in terms of performance and behavior, when packet loss do occur. Furthermore, this bandwidth limitinh feature is not a proposed TCP standard which results in that different TCP implementations that incorporates this kind of feature, almost certainly, will have a unique way of implementing it. Thus, having it enabled would make the experiments lack in generality.

For these reasons the bandwidth limiting feature of FreeBSD was disabled. This was done by changing the value of the `sysctl`[2] parameter `net.inet.tcp.inflight.enable` from its original value of 1 to 0.

Network & Application parameters

In Table 3.1 the different test case parameters that were used for the experiments are shown. These values are argued for in the remainder of this section.

Flow size (packets)	Bandwidth (Kbit/s)	Delay (ms)	Queue size (packets)
20	40, 80, 160, 320, 500, 1000, 2000, 4000, 8000, 10000	5, 10, 20, 40, 60, 80, 100, 150, 200, 250, 300	99
100	40, 80, 160, 500, 1000, 4000, 10000	5, 10, 20, 40, 60 100, 200, 300	99

Table 3.1: Test case parameters

Two different flow sizes were used for the experiments; 20 and 100 packets. Due to the fact that a large portion of the Internet traffic today is web traffic, which typically is composed of short-lived flows, it is interesting to investigate the possible effect of fast retransmit inhibitions on short-lived flows (20 packets). The other flow size, 100 packets, were chosen to see how important the size of the flow was when considering these inhibitions. As shown in Table 3.1 there are less combinations of bandwidths and delays for the longer flows, this was done because of the large amount of experiments that a flow size of 100 generates.

[2]For more information on sysctl, please see Section 3.4.1

The server application that were used for the experiments, described in Section 3.4.3, is designed to send N buffers containing 2500 bytes of data to the client. To be able to send exactly 20 packets we were forced to take several things into account. The MTU of the IP packets in the experimental environment was set to 1500 bytes which yields a maximum TCP packet size of $1500 - 20 - 20 - 12 = 1448$ bytes, where the IP and TCP header is consuming 20 bytes each, and the TCP options used by the FreeBSD TCP implementation require 12 bytes of data. Due to the nature of TCP the server application also needs to send 2 packets that do not contain any data; one of these packet is the `SYN-ACK` packet that is used in the three-way-handshake, and the other one is the final acknowledgment sent in the connection termination process. By solving difference 3.1, we can conclude that 10 buffers (x) of data must be sent in order to get the additional 18 packets.

$$17 < \frac{x \times 2500}{1448} \leq 18, \quad x \in \mathbb{Z}^+ \tag{3.1}$$

By the same argumentation we concluded that 56 buffers of data was required to generate a flow of 100 packets.

The bandwidths at which the experiments were conducted, 40−10000 Kbit/s, is believed to be a range of bandwidths that frequently exists within real networks. The lower limit, 40 Kbit/s, may not be a common bandwidth in todays high-speed networks, but it is included as some people still use modems to connect to the Internet. The upper limit, 10000 Kbit/s, is quite low but we were restrained to use it as it was the theoretical maximum for the network emulation software that we used.

In todays networks a wide range of delays exist. In a local area network, LAN, the physical distance between different hosts is, typically, small and they are often interconnected using few, if any, routers. The short distance and the limited processing of the traffic[3] lead to small end-to-end delays between the hosts. A wide area network, or WAN,

[3]In terms of routing.

can be defined as a set of interconnected LANs. Depending on the size of the WAN, and where the different hosts are located, in terms of physical distance and number of routers in between, the delays can vary considerably. While two hosts that both share WAN and LAN have a small end-to-end delay, two hosts that are separated by thousands of miles and numerous intermediate routers have a considerably longer delay. To cover most of these scenarios delays between $5 - 300$ ms were used.

Another reason for choosing the bandwidths and delays in the way that was done was to create bandwidth-delay products, or BDPs, that are equal (or almost equal). BDP is a measurement of the maximum link capacity that can be utilized by TCP, and it is therefore interesting to see if any similar effects, between different experiments, occur when the BDP is equal.

$$\text{BDP (bytes)} = \text{Bandwidth (Kbytes/s)} * \text{Delay (ms)} \tag{3.2}$$

Worth to note is that the delay in Equation 3.2 is not the end-to-end delay, but the round-trip time.[4]

For example; using equation 3.2, the BDP for a connection with a bandwidth of 40 Kbit/s and an round-trip time of 20 ms is exactly the same as for the case of 80 Kbit/s and 10 ms.[5]

The queue size was set to 99 packets for both of the different flows sizes. This number is sufficiently high to avoid buffer overflows. Normally, as mentioned in Section 2.2, congestion occurs when a router in a network has a full buffer and the incoming rate of the traffic keeps exceeding the outgoing, thus making the router start discarding incoming packets. Due to the fact that we want do decide exactly when to loose packets in the flow that behavior, losing packets due to buffer overflows, is not preferred.

[4]The time required for a segment to be sent and acknowledged.

[5] $\frac{40*10^3}{8} * 20 = \frac{80*10^3}{8} * 10 = 100$ Kbytes. Where the division with 8 is done to go from bits to bytes.

Other parameters

The number of replications was set to three, this in order to account for possible variance in the results.

3.3 Environment Overview

To be able to conduct the experiments that were described in the previous section, an experimental environment was constructed.

There were two major requirements for the experimental environment. The first requirement was that the TCP communication between a client and a server (located in the environment) should not differ much from TCP communication between a client and a server over a real network. This requirement was fulfilled by using a network emulator that could emulate network parameters that are common in real operational networks. The other requirement was that the environment should be able to produce position based losses within a TCP communication flow. With position based losses it would be possible to make positional dependencies of the losses visible and it would also be easier to reproduce results with small variance. To deal with this requirement the network emulator was equipped with a network emulation software that was able to produce losses according to given positions.

In Figure 3.3 the setup of the experimental environment is illustrated. The solid lines show the links that carry the experiment traffic, i.e. the traffic between the client and the server (via the network emulator). The dashed lines show the separate control network that the three computers were connected in. This control network was created in order to let the computers to be automatically configured with experiment parameters without interfering with the experiment traffic.

The computers used for building this environment were Dell Optiplex GX260 with 3Com 100Mbit/s network cards. The client and the server was running FreeBSD 6.0B3,

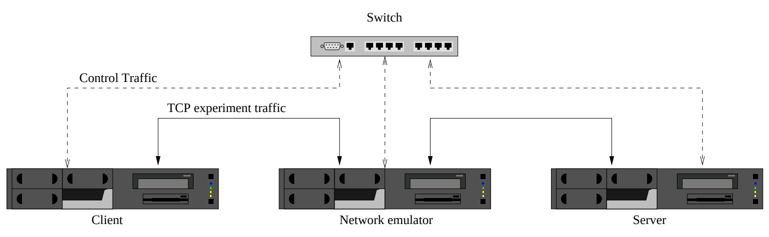


Figure 3.3: Environment overview

and the network emulator FreeBSD 6.0B5.

3.4 Environment Details

Figure 3.4 shows the different computers along with the software that they use in order to realize the experimental environment. The communication and dependencies between the different software components in the environment are illustrated with lines and arrows in the illustration.

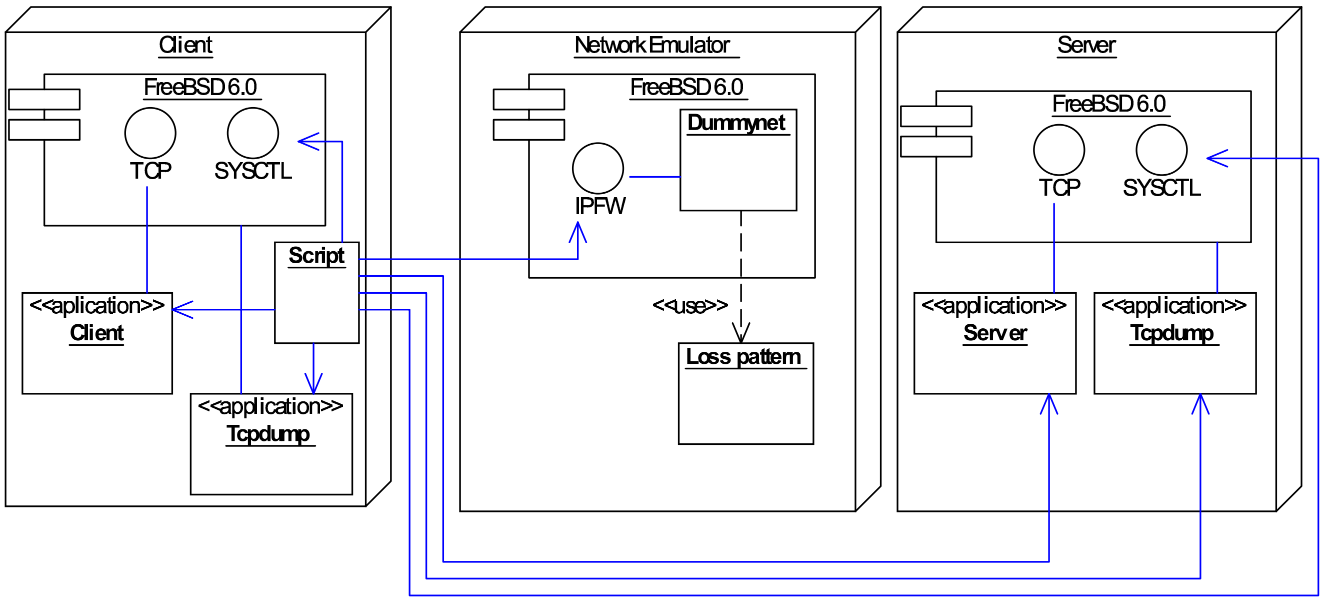


Figure 3.4: Environment detailed

With the help of this illustration the different components of the framework are described in the following subsections.

3.4.1 FreeBSD 6.0B3

The client and the server were both configured to run FreeBSD 6.0B3, this to make sure that the TCP implementations of the client and the server were the same. If different implementations of TCP is used in an experiment there can be a problem determining if the result depends on one of the implementations, or on the other, or on the combination of the two. The easiest way to eliminate this risk is simply to run identical implementations on the different hosts.

FreeBSD 6.0B3 has a feature that is called host caching. This feature, TCP host caching, caches host observations from TCP. This allows the TCP to reuse round-trip times, congestion window size, slow start threshold, and bandwidth estimates from previous connections in order to optimize new TCP connections to the same host. This was not acceptable for the experimental environment because every experiment was, of course, required to be independent from previous experiments. To disable this feature a slight modification to the kernel was required. Details about this kernel modification can be found in Section C.2 of the appendix.

FreeBSD has a command that is called `sysctl`. This command acts as an interface to the kernel and can be used to list and modify a large number of kernel parameters. Some of these parameters can be used in order to configure the TCP implementation. Two of these parameters, `net.inet.tcp.slowstart_local_flightsize`, and `net.inet.tcp.slowstart_flightsize`, are used to control the allowed amount of outstanding TCP packets during the slow start phase. Because network communication between computers on a local network (same subnet) are more likely to have a large amount of available bandwidth, the values of these two parameters differed.[6] The machines that were used as client and server in the environment were on the same subnet, so we were

[6]Allowing a larger amount of outstanding data in a local network.

required to change the value of `net.inet.tcp.slowstart_local_flightsize`, in order to treat the traffic in the network as “non-local”.

For more information on the FreeBSD TCP implementation see appendix C.1.

3.4.2 FreeBSD 6.0B5

The operating system on the network emulator computer was FreeBSD 6.0B5. The kernel of this operating system was configured to run a modified version of Dummynet. More information about Dummynet and the modified Dummynet can be found in Section 3.4.4.

3.4.3 Client & Server applications

The client and server applications in this environment work like a sink and a source. When the server is started it takes as argument the port number to listen for connections at and how much data to transmit to connecting clients. The client is started by passing the server host name and port number. When a successful connection is established between the two applications, the server application sends the specified amount of data to the client and closes the connection. When the client has received all the data sent by the server, indicated by a close from the server, the client tries to close the connection and returns the elapsed time from connection to the receipt of the final data.

The sources to these applications are provided in Appendix D.3.

3.4.4 Dummynet & Loss patterns

As described in Section 2.3.1 Dummynet is a network emulator software. The Dummynet version that was used for this environment is the extended version mentioned in Section 2.4. To be able to use the extended version we were required to port it from FreeBSD 4.6.x to FreeBSD 6.0B5.[7] By using this version we could generate positional losses in TCP

[7]Unfortunately this porting is too extensive to be included/described within the thesis.

data flows by simply specifying which packets that should be lost. A Dummynet example configuration is shown in Figure 3.5.

```
ipfw add 1 pipe 100 tcp from <server> <server-port> to <client> in
ipfw add 2 pipe 200 tcp from <client> to <server> <server-port> out
ipfw add 3 pipe 300 tcp from <server> <server-port> to <client> out

ipfw pipe 100 config packlfile <pattern>
ipfw pipe 200 config bw <bw> delay <delay> queue <queue>
ipfw pipe 300 config bw <bw> delay <delay> queue <queue>
```

Figure 3.5: Dummynet Configuration

In Figure 3.5 we can see that three Dummynet pipes are created and configured.[8] One pipe for traffic that is coming into the network emulator (from the server), and two pipes for the outgoing traffic (traffic from both client and server).[9] The two pipes that is configured for outgoing traffic is configured to emulate bandwidth limitations, delays, and queue-size limitations. The pipe for the incoming server traffic is configured with a "loss pattern". This loss pattern is a file that contains a binary pattern that tells Dummynet exactly which packets in the TCP data flow that should be lost.

Let us say that we want to lose the fourth packet in a TCP data stream, then the loss pattern simply looks like in Figure 3.6 where N is the period of the loss pattern, i.e. after Dummynet processed N packets it will start over from the beginning of the pattern again.

$$\underbrace{\begin{array}{|c|c|c|c|c|c|c|c|}\hline 0 & 0 & 0 & 1 & 0 & 0 & \dots & 0 \\ \hline\end{array}}_{N \text{ positions}}$$

Figure 3.6: Loss pattern

[8]This is done by the help of the program ipfw, that was mentioned in Section 2.3.1

[9]No pipe is created for incoming traffic from the client, which is perfectly legal if no emulation effects is to be applied on that traffic.

To simplify the creation of such patterns, a pattern generation application was constructed. This application is described in appendix D.1.

3.4.5 Tcpdump

To be able to log and analyze TCP traffic flowing between the client and the server Tcpdump [37] was used. This tool, Tcpdump, can be used to capture traffic on a network interface according to a given expression. An example usage of Tcpdump is;

```
tcpdump -i eth0 -w log.pcap tcp port 80
```

where we instruct Tcpdump to capture all TCP traffic, with a source or destination port of 80, on the network interface `eth0`. The `-w` switch tells Tcpdump to store the captured data to a file (`log.pcap` in this example).

Applications like Tcpdump are often called packet sniffers. These sniffers are usually made up of two major components; a packet analyzer and a packet capture. The packet capture component is responsible for the actual traffic capturing, which takes place in kernel space. Here the packet capture component receives copies of all the link frames that are sent and/or received (in our case Ethernet frames). The other component, the packet analyzer, is an application executing in user space. This application is able to interpret the captured frames in terms of higher-level protocols and to perform various actions on the captured traffic. In Figure 3.7 we can see that Tcpdump, which actually is just a packet analyzer, uses the BSD Packet Filter (BPF) [23], with the help of the pcap library [18], in order to capture network traffic. These components, together, form a packet sniffer.

3.4.6 Script

The script is the "brain" of the experimental environment. As illustrated in Figure 3.4, and described in Section 3.2.2, this script was used to start the client and the server application,

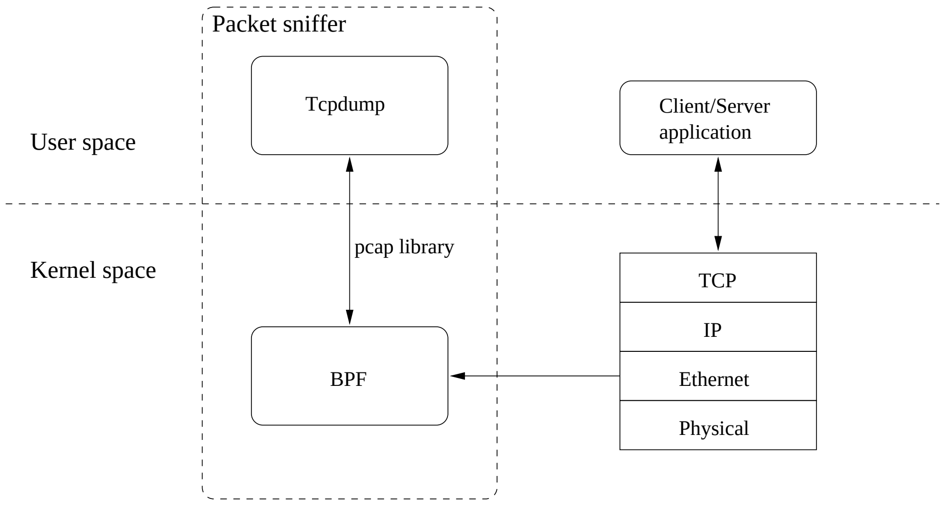


Figure 3.7: Tcpdump

and to configure all parts of the environment according to the desired parameters of the experiment at hand. The source code to this script can be found in Appendix D.2.

3.5 Summary

In this chapter we have presented the problem with fast retransmit inhibitions, the experimental design that was required to reveal these inhibitions were presented, and the environment in which the experiments were conducted was described in detail. In the next chapter the results of the experiments are presented and, in some cases, analyzed.

Chapter 4

Results and Analysis

This chapter presents the results and the analysis of the experiments. The chapter is divided into two different parts; one for the short flows (20 packets), and one for the long flows (100 packets). Each part begins with a section where the performance loss due to the fast retransmit inhibitions are presented. In the following section a more detailed view is presented. Here we can see how losses on all different positions in the flows affects the total transmission time. All interesting phenomena and anomalies that are present in this section are then discussed and analyzed in the subsequent sections, where the flows that contain these interesting results are studied on packet level.

4.1 Short flows

4.1.1 General performance loss

Figure 4.1 shows how much additional time that was required in order to complete a transmission if the introduced packet loss was placed in the end of the flow, where fast retransmit should be inhibited, as opposed to a loss where fast retransmit was used. The y-axis of the graph show the additional time, in percent, that was required for a flow with a delay according to the x-axis (ms). The figure shows that for combinations of high

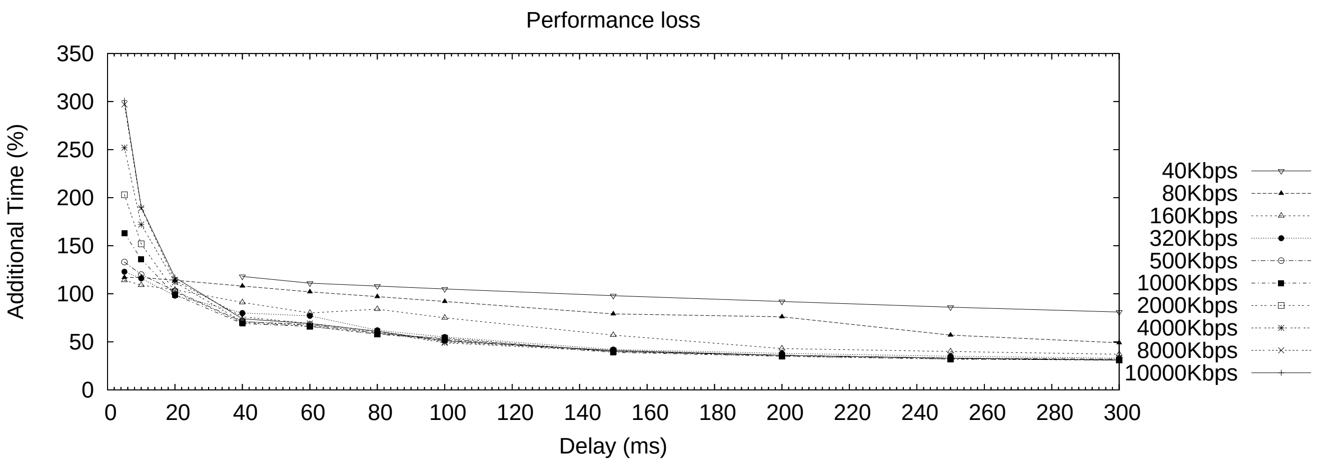


Figure 4.1: Performance loss for the short flows

bandwidth and low delay the performance consequences of a late loss are dramatic. For a bandwidth of 10000 Kbit/s and a delay of 5 ms 301% of additional time is required. This can be compared with a bandwidth of 80 Kbit/s and a delay of 5 ms where "only" 117% of additional time is required. Another interesting thing to note is that the additional time seems to decrease more rapidly for higher bandwidths, as the delay increases, than for the lower bandwidths. For lower bandwidths the performance impact seems to be more stable. In the case of 80 Kbit/s the additional time required ranges from 117% to 47% for the different delays, whereas a bandwidth of 10000 Kbit/s ranges from 301% to 31%. The reason why this graph does not contain any results for the three lowest delays (5, 10, and 20 ms) of 40 Kbit/s has to do with a defect in the FreeBSD TCP implementation. This defect causes the transmission time of flows with low bandwidth and delay to increase dramatically. To be certain of that these results are not mistakenly associated with the problem of fast retransmit inhibitions they were not included in the graph. The defect in the FreeBSD implementation is discussed more throurghly in the next two sections.

In Figure 4.2 the average additional time that was required due to fast retransmit inhibitions is shown. In the previous figure, Figure 4.1, it was assumed that the packet loss occurred at a position where fast retransmit was inhibited. In this figure, however, the

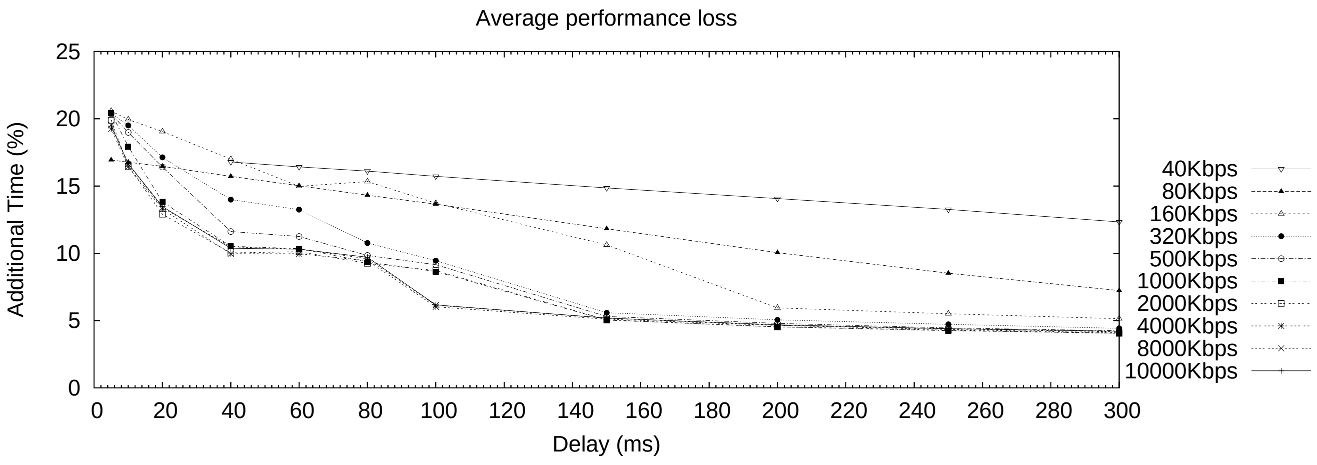


Figure 4.2: Average performance loss for the short flows

assumption is only that a packet loss has occurred at an arbitrary position within the flow. Because of the fact that fast retransmit is employed more often than it is inhibited, it is natural that the average performance loss is less severe. The layout of this graph, Figure 4.2, is the same as for the previous one; showing the additional required time on the y-axis (in percent) and the different delays on the x-axis. Looking at this figure we can see that all of the bandwidths, included in the experiments, had a considerably high performance drop for low delays, as almost all of them required about 20% of additional time when the delay was 5 ms. Worst was 1000 Kbit/s which required 20.4% of additional time. As the delays increased, the performance was less affected, and most of the bandwidths only suffered from $5 - 10\%$ of additional time required, when the delay was 300 ms. It is also apparent that for lower bandwidths the performance impact is quite stable for all the different delays. For 40 Kbit/s, for example, the additional time required was between $10 - 20\%$.

This graph also lacks the results for the three lowest delays (5, 10, and 20 ms) of 40 Kbit/s, and the reason why is covered in the next sections.

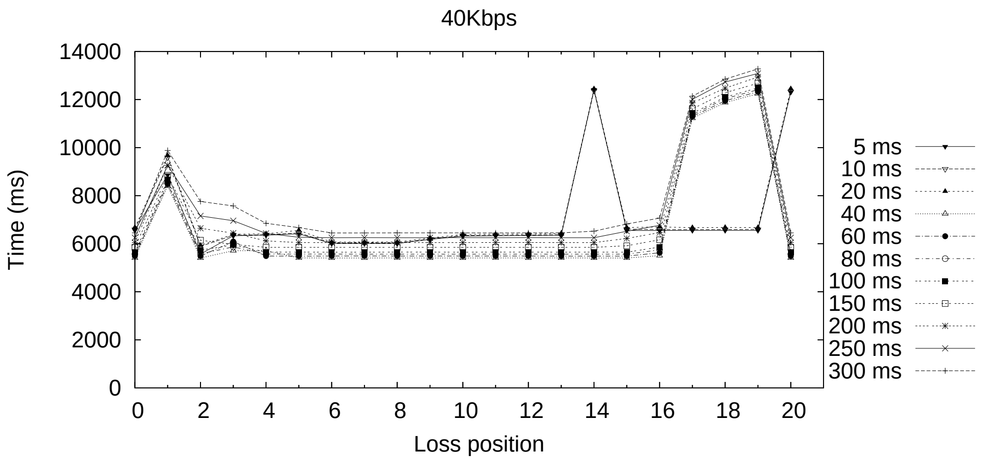


Figure 4.3: Transmission time graph 40 Kbit/s

4.1.2 Positional dependencies

Figure 4.3 shows, for a bandwidth of 40 Kbit/s, the total transmission time of the whole flow when a packet loss is introduced at a certain position in the flow. The y-axis of the graph shows the total transmission time, in milliseconds, that was required for a flow with packet loss at a certain position (x-axis). Position 0 on the x-axis corresponds to no loss. As we can see in this figure the expected positional dependencies among the losses exists. When the first packet was lost, the `SYN-ACK` packet in the three way handshake (see Section 2.1.6), the transmission time was increased by approximately three seconds which corresponds to the initial value of the TCP retransmission timer [26]. When the loss was positioned in the middle of the flow the results indicate that the fast retransmit algorithm was used. For losses placed at the end of the flow the transmission time shows that fast retransmit was not conducted.[1]

However, some of the results, shown this figure, are strange and unexpected. For the three lowest delays (5, 10, and 20 ms) the average transmission time is higher than for

[1]The 20th packet is an acknowledgment to the connection termination packet from the client to the server. When this packet is sent the total transmission time is already calculated by the client, which implies that a loss of this packet does not affect the total transmission time.

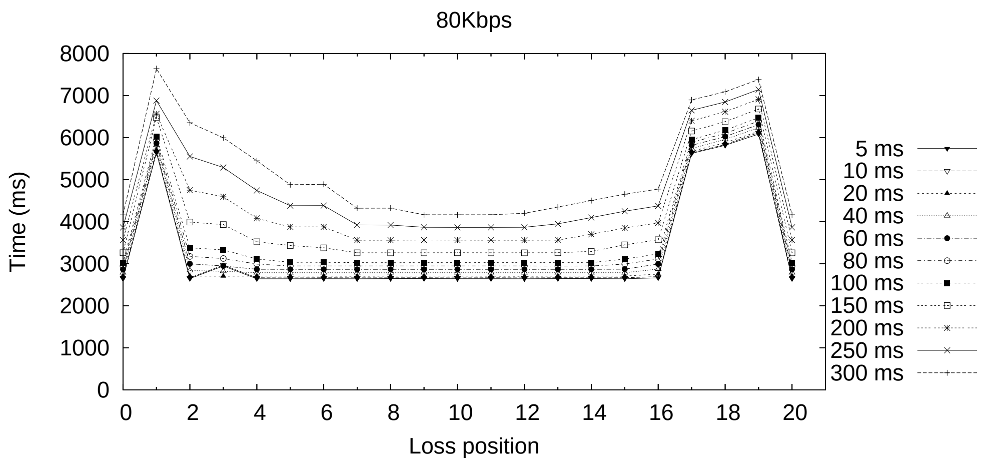


Figure 4.4: Transmission time graph 80 Kbit/s

larger delays, and these three delays also contains two strange peaks for losses at positions 14 and 20. This strange peak result is analyzed in Section 4.1.3.

If we continue with the results for the higher bandwidths we can see that the strange peaks that existed in the case of 40 Kbit/s, Figure 4.3, are gone. The results for 80 Kbit/s, Figure 4.4, confirms the expected positional dependencies among the losses. For the lowest delays we can clearly see that there exist three separate regions in the graph; the beginning where the first packet (the `SYN-ACK`) is lost, the middle section where the fast retransmit algorithm is used, and the end where fast retransmit is inhibited. For higher delays, however, the middle section of the graph becomes smaller, making the total transmission time considerably longer for a loss early in the flow than in the middle. This phenomenon is even more visible in the case of 160 Kbit/s, Figure 4.5, where the total transmission time when a loss occurs at position four in the flow is greater than the transmission time when one of the last packets in the flow is lost, even though fast retransmit is possible at the fourth position. This is analyzed in Section 4.1.4.

Another interesting phenomenon is that the end section starts to expand for low delays. For 160 Kbit/s with low delay, Figure 4.5, the results indicate that the fast recovery algorithm is not used when losses are placed at positions 16, 17, 18, and 19 in the flow.

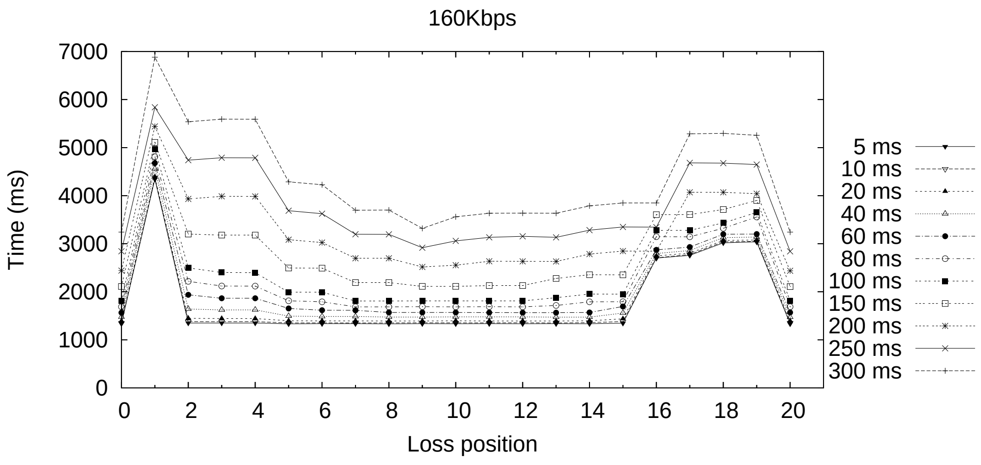


Figure 4.5: Transmission time graph 160 Kbit/s

In the previous results this only applied to positions 17, 18, and 19. This is analyzed in Section 4.1.5.

Moving up to a bandwidth of 320 Kbit/s, Figure 4.6, we can tell by the results that all the previous phenomena and dependencies are visible. A new and interesting thing is that a “dip” in the transmission time is visible when a packet loss is introduced at position nine in the flow. For 320 Kbit/s with a delay of 300 ms the required transmission time is about 14% less for a loss at position nine than for losses at positions eight and ten. This is analyzed in Section 4.1.6.

In Figure 4.7 the results for the bandwidths 500, 1000, 2000, 4000, 8000, and 10000 Kbit/s are shown. These results do not contain any new interesting phenomena, even though some of the previously analyzed phenomena, such as the impact of an early loss, seem to have a greater impact on the total transmission time as the bandwidth and delay increase. In Figure 4.8 the results for 1000 Kbit/s with delays ranging from $5-100$ ms are shown, to better give a view of the results for the lower delays. In this figure we can see most of the previously mentioned dependencies and phenomena.

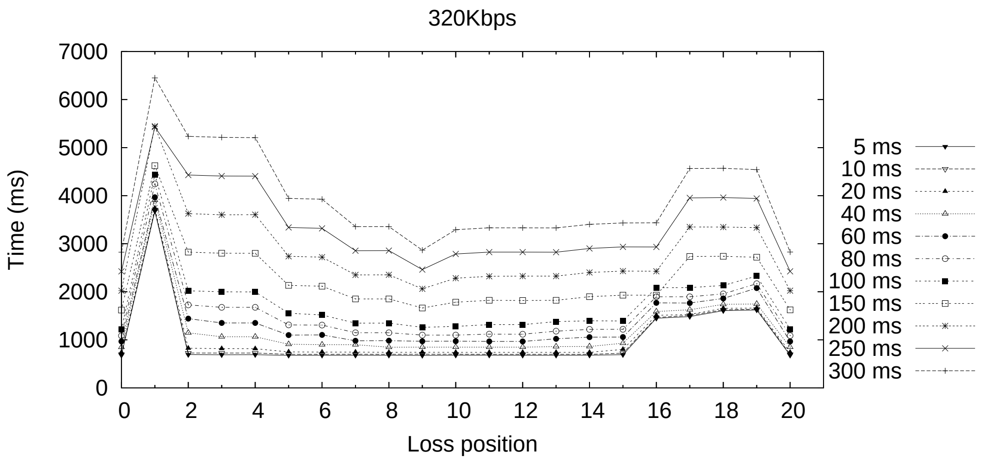


Figure 4.6: Transmission time graph 320 Kbit/s

4.1.3 Bad RTO Estimation

The reason to why the average transmission time is longer for delays of 5, 10, and 20 ms in the case of 40 Kbit/s bandwidth than for other, larger, delays seems to be a problem with the RTO calculation in the FreeBSD TCP implementation. Figure 4.9(b) shows all the different actions and events at the server during the initial transmission phase of the flow. The flow in this graph is the flow width 40 Kbit/s bandwidth and 5 ms delay. The y-axis of the graph represents the sequence number space, and the x-axis shows the time. For a more thorough description of the structure of the graph, and the symbols that are used please see Appendix B.1. Looking at the figure we can see that the server starts by sending four packets of data, this is followed by a timeout and retransmissions of the four packets. The client on the other hand, illustrated in Figure 4.9(a), successfully receives the packets and also acknowledges them, but these acknowledgments are received by the server after the timeout has occurred.[2]

The RTO calculation in the FreeBSD TCP implementation (explained in Appendix C.1.1) uses incoming acknowledgments to (re)calculate the RTO timer. For the server,

[2]The reason why the time differs between the two figures is that the clocks of the client and server machines were not synchronized.

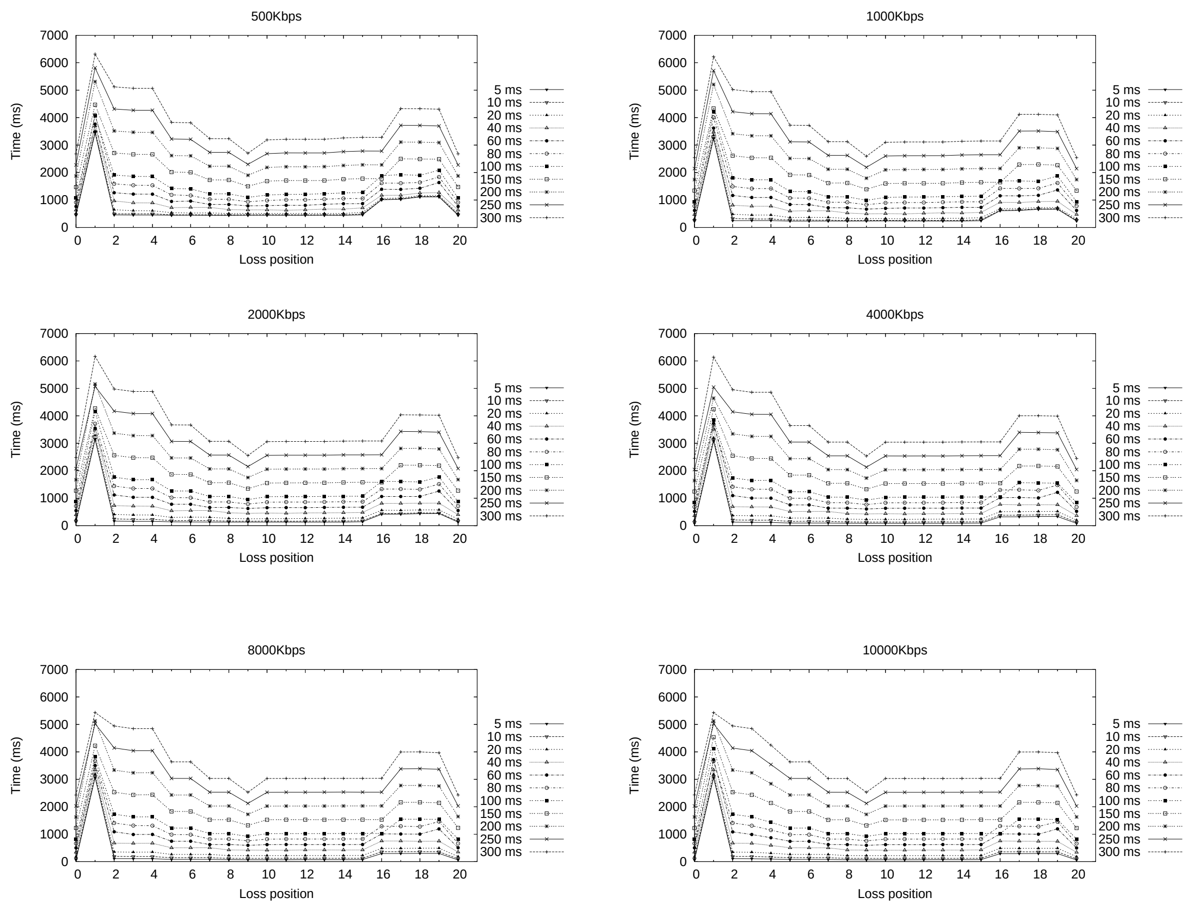


Figure 4.7: Transmission time graph 500, 1000, 2000, 4000, 8000, and 10000 Kbit/s

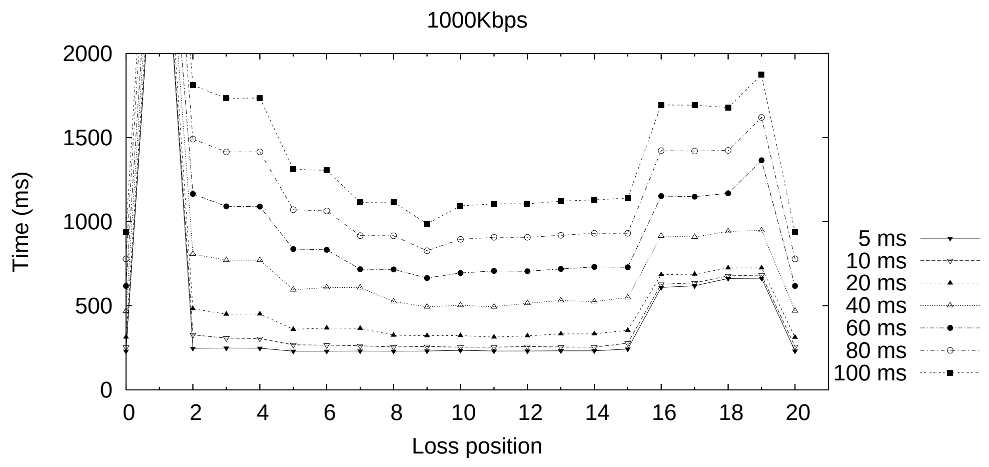
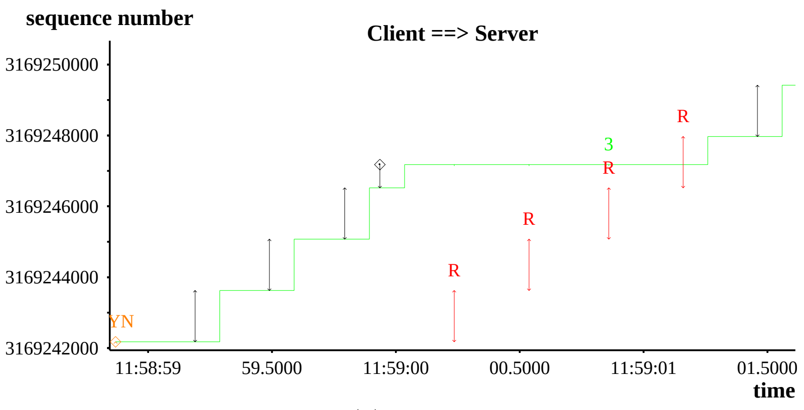


Figure 4.8: Transmission time graph 1000 Kbit/s, low delays

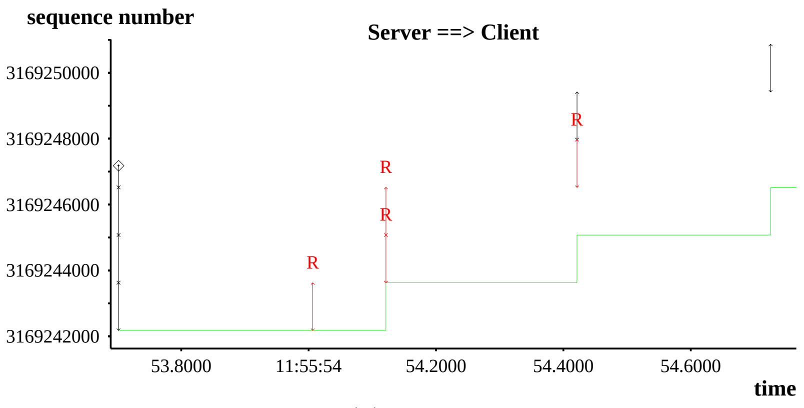
Figure 4.9(b), the only acknowledgment that was used as a base for the RTO timer calculation (before the retransmissions started) was the acknowledgement to the SYN-ACK packet in the three-way handshake (see Section 2.1.6). These packets had the sizes 78 and 66 bytes which gives a round-trip time, without considering the delay, of approximately $\frac{78*8}{40*10^3} + \frac{66*8}{40*10^3} = 0,0156 + 0,0132 = 0,0288$ seconds. This sets, if we follow the steps in C.1.1, the RTO timer to 287 milliseconds. If we consider a full-sized packet, 1500 bytes, with an acknowledgment of 66 bytes the round-trip time, without considering the delay, becomes $\frac{1500*8}{40*10^3} + \frac{66*8}{40*10^3} = 0,3132$ seconds. Thus, when the first full-sized packet is sent the round-trip time will be greater than the RTO timer which will result in a retransmission. Even though no delay was included in this example, our results show that this problem seems to exist for delays that are ≤ 20 ms.

The strange peaks located at positions 14 and 20 in Figure 4.3 are related to the inaccurate RTO calculations. In Figure 4.10(a) we can see another strange behavior that is conducted by the FreeBSD TCP implementation. When acknowledgments for new packets, packets sent after the retransmissions, are received by the server they[3] are regarded as

[3]The acknowledgments.



(a) Client



(b) Server

Figure 4.9: Bad RTO estimation

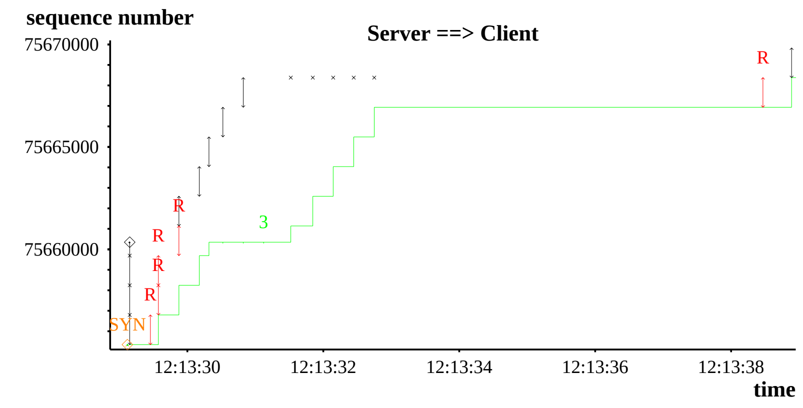
already received and duplicate acknowledgments is sent to the client in response.[4] Because of this (mis)behavior there are not enough packets in flight to trigger the fast retransmit mechanism, instead the server waits for the retransmission timer to expire (because of the four initial losses the retransmission timer is set very high) and then resends packet number 14. For a loss at position 20 the problem is the same.

The reason why these peaks do not exist for losses at other positions in the flow is illustrated in Figures 4.10(b) and 4.10(c), which shows how losses on position 13 and 15 are treated by the server. In 4.10(b) we can see that packet 14 triggers a retransmission of the lost packet. This behavior indicates that the server believes that it still recovers from the four initial losses, otherwise it would have taken three duplicate acknowledgments to trigger a retransmission. In Figure 4.10(c) the 15'th packet is lost, but this packet is one of the strange duplicate acknowledgments that was sent from the server to the client, so no harm seems to be done. The reason why the server behaves in this way is, unfortunately, unclear.

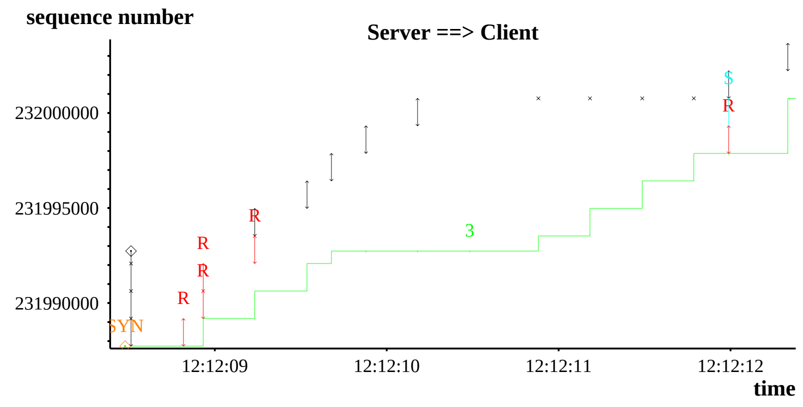
4.1.4 Link utilization implications

The reason why losses at early positions causes the transmission time to increase significant as the bandwidth and the delay increases, as in Figure 4.5, is due to the fact that the bandwidth-delay product increases. The bandwidth-delay product, or BDP, is a measurement of how much data that can be in flight at the same time. Earlier, in Section 2.2.1, the transfer of data was compared with pouring water through a pipe. The fastest way to do this is to pour all the water into the pipe at once, making the pipe completely filled. The same argument holds for data transfer. The BDP, which can be compared to the volume of a pipe, is calculated according to Equation 3.2 in Section 3.2.3, which is repeated here for convenience.

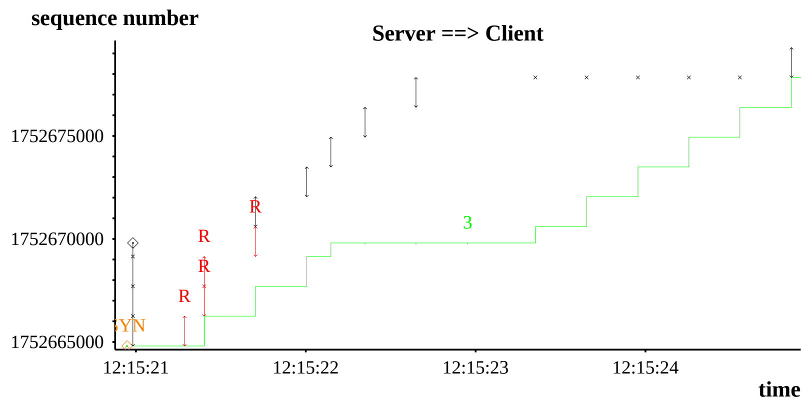
[4]Indicated by the small x's in the figure.



(a) Peak



(b) Before peak



(c) After peak

Figure 4.10: Acknowledgment misinterpretation

$$\text{BDP (bytes)} = \text{Bandwidth (Kbytes/s)} * \text{Delay (ms)}$$

Because of the possible problem with network congestion TCP does not send all data that is submitted for transmission right away, to utilize the full capacity of the link without considering concurrent traffic, but uses the slow start mechanism (see Section 2.2.1) to rapidly increase the utilization of the link to an acceptable level.

At the beginning of a TCP connection, the slow start mechanism has not had the chance to increase sending rate to fully utilize the link, so if a loss occurs early the congestion avoidance phase, which increases the sending rate very slow, is entered earlier. For connections with high BDP an early loss can be a problem, as the utilization of the link will be lower.

4.1.5 Receive buffer fluctuations

In Figure 4.5, 160 Kbit/s, we can see that the "end section" of the graph, the section where fast retransmit is not possible to perform, grows larger for low delays. The reason to why the section where fast retransmit is no longer possible grows is related to the use of window updates. As mentioned in Section 2.1.4 a TCP receiver (the client in our case) uses window updates to inform the sender (server) when the receiver window opens up. Unfortunately a TCP sender has no way of determining if a duplicate acknowledgment with an updated receiver window is an "ordinary" duplicate acknowledgment or if it is a window update. Therefore all duplicate acknowledgments with an updated receiver window are treated as window updates and not duplicate acknowledgments. This behavior seems to have consequences on the fast retransmit mechanism, especially for connections with low delay.

For connections with a low end-to-end delay the TCP client is forced to buffer data, due to the fast delivery. The buffering makes the advertised receiver window to shrink, and when a duplicate acknowledgment is sent in response to our introduced loss this duplicate acknowledgment also contains an updated receiver window size and is therefore treated as a window update. The reason why this phenomenon is not visible for lower bandwidths and/or higher delays is that the client, in most cases, does not receive data faster than it reads it, therefore it does not buffer any data which leads to an unchanged receiver window.

4.1.6 TCP packet bursts and their implications

The reason why a dip in the total transmission time can be seen when a loss is introduced at position nine in flows with high delay is a consequence of the data transmission behavior of TCP in combination with the length of the flow. The transmission behavior of TCP is, basically, to send a number of packets, wait for them to be acknowledged, and then send more packets. Figure 4.11 shows the communication from the servers point of view when the bandwidth was 320 Kbit/s, the delay 300 ms, and no losses were introduced. As we can see the packet sending was partitioned into three distinct groups (or bursts); the initial group that was sent upon connection establishment, and then two groups that were sent in response to the acknowledgments of the previous. Looking at Figure 4.12 we can see that the loss that was at position eight in the flow caused the server to send all the data in four groups, instead of three, which adds approximately one extra round-trip time to the total transmission time. A loss on position nine, Figure 4.13, did not cause an extra group of packet sending as the congestion window had opened up slightly more than in the previous case, and therefore both the retransmission and the final packet were sent in the same group. In Figure 4.14 we can see that a loss on position ten also caused an extra group of packet sending, but in that case it was due to the fact that the packet loss was not detected until the acknowledgments of the packets in the final group arrived.

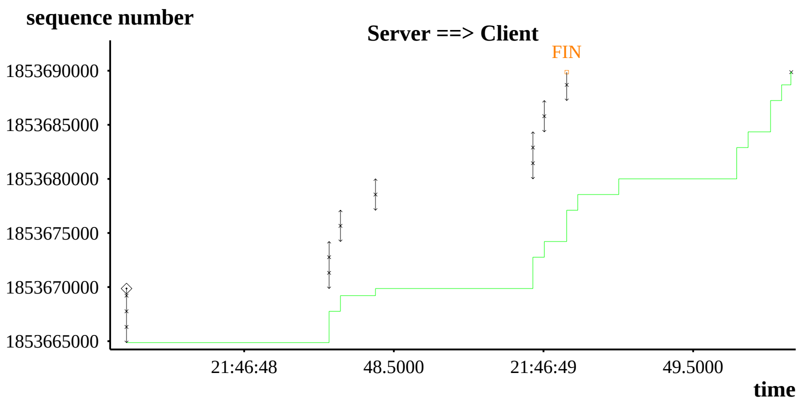


Figure 4.11: Server with 320 Kbit/s, 300ms delay, no losses

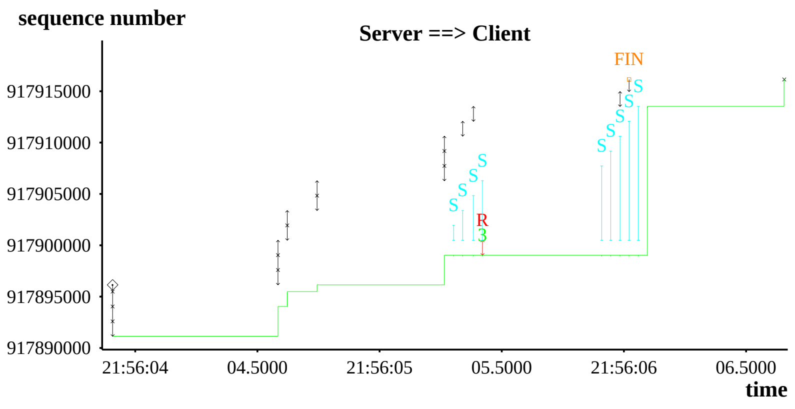


Figure 4.12: Server with 320 Kbit/s, 300ms delay, loss on position eight

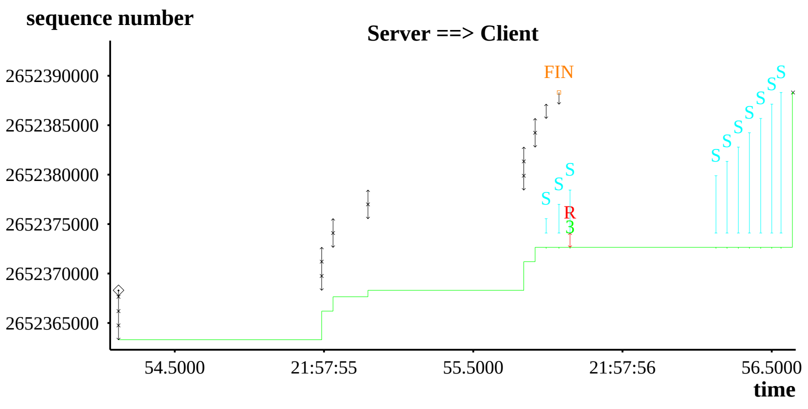


Figure 4.13: Server with 320 Kbit/s, 300ms delay, loss on position nine

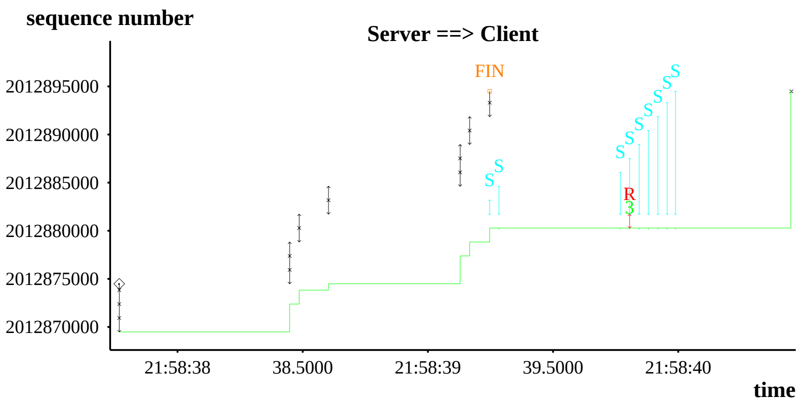


Figure 4.14: Server with 320 Kbit/s, 300ms delay, loss on position ten

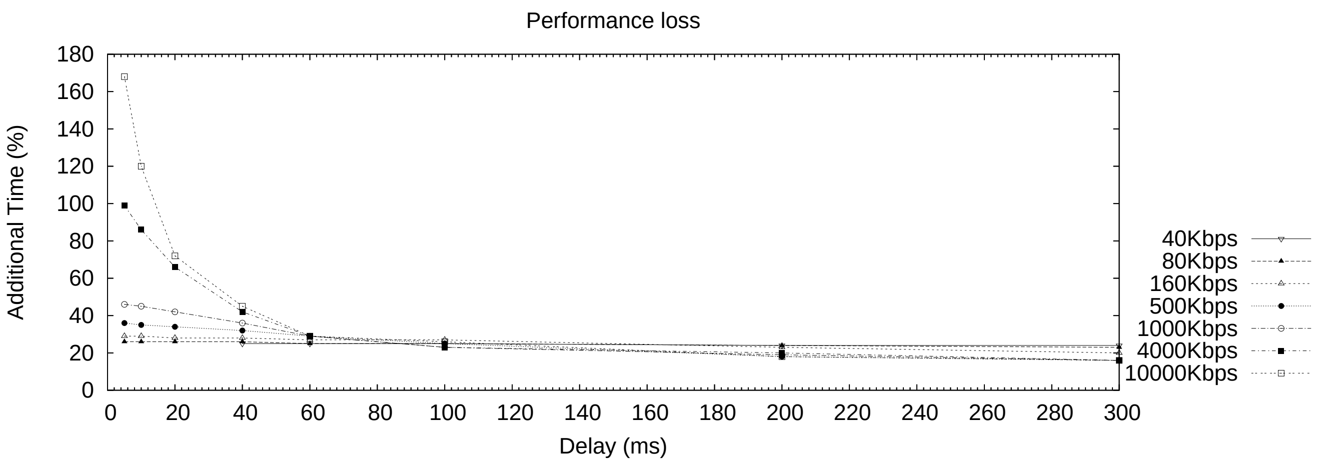


Figure 4.15: Performance loss for the long flows

4.2 Long flows

4.2.1 General performance loss

Figure 4.15 shows how much additional time that was required in order to complete a transmission if the introduced packet loss was placed at a position where fast retransmit should be inhibited, as opposed to a position where it should be employed. The y-axis of the graph show the additional time, in percent, that was required for a flow with a delay according to the x-axis (ms). This figure, 4.15, shows that for combinations of high bandwidth and low delay the performance consequences of a late loss are significant. For

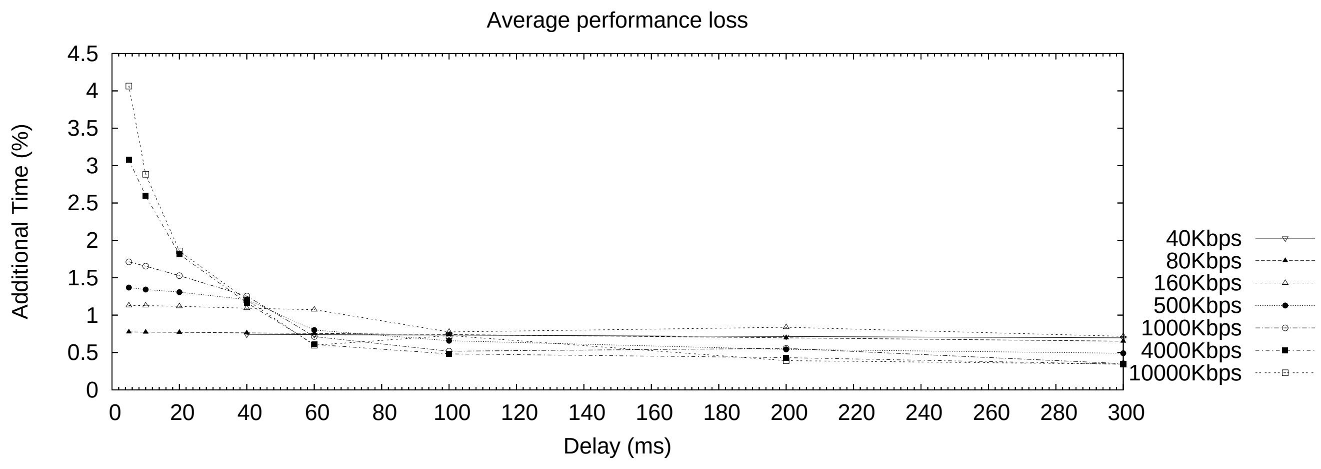


Figure 4.16: Average performance loss for the long flows

a bandwidth of 10000 Kbit/s and a delay of 5 ms 168% of additional time is required. Compared to the results of the short flows, where all the bandwidths that had a delay of 5 ms had over 100% of additional time required, these longer flows seem to generally suffer less from a late loss. As we can see in the figure the results converge fast, and from 60 ms of delay all of the bandwidths require less than 40% of additional time.

In Figure 4.16 the average additional time that was required due to fast retransmit inhibitions is shown. In the previous figure, Figure 4.15, it was assumed that the packet loss occured at a position where fast retransmit was inhibited. In this figure, however, the assumption is only that a packet loss has occurred at an arbitrary position within the flow. The layout of this graph is the same as for the previous one; showing the additional required time on the y-axis (in percent) and the different delays on the x-axis. The reason to why the average performance loss for these longer flows are so insignificant (compared to the short flows discussed in Section 4.1.1) is partly due to the fact that the number of occasions when fast retransmit is inhibited is very small compared to the number of times when it is employed. Another reason to why the effects of the fast retransmit inhibitions are so small has to do with an implementation issue in FreeBSD which causes the average transmission time to be relatively high, even if fast retransmit is conducted. This problem is identified in the next section, and is more deeply analyzed in Section 4.2.3. If we take a look at Figure 4.16 we can see that the additional time that is required is 4% or less for all different combinations of bandwidths and delays. For the largest delay, 300 ms, all of the bandwidths require an additional time that is less than one percent.

The reason why these graphs, Figures 4.15 and 4.16, do not contain any results for the three lowest delays (5, 10, and 20 ms) of 40 Kbit/s has to do with a defect in the FreeBSD TCP implementation (discussed in Section 4.1.3). This defect causes the transmission time of flows with low bandwidth and delay to increase dramatically. To be certain of that these results are not mistakenly associated with the problem of fast retransmit inhibitions they were not included in the graphs.

4.2.2 Positional dependencies

Figure 4.17 shows, for the bandwidths of 40, 80, and 160 Kbit/s, the total transmission time of the whole flow when a packet loss is introtuced at a certain position in the flow. The y-axis of the graph shows the total transmission time, in milliseconds, that was required for a flow with packet loss at a certain position (x-axis). Position 0 on the x-axis corresponds to no loss. Looking at the top of this figure (40 Kbit/s), we can see that positional dependencies among the losses are clearly visible in the results. The three lowest delays (5, 10, and 20 ms) have the same strange peaks, and high average transmission time, as the corresponding short flows had. This is due to the same problems that were discussed in Section 4.1.3.

Compared to the short flows the results for these, longer flows, seem to contain a section in the middle where the total transmission time is unexpectedly high. For 80 Kbit/s, in the middle of Figure 4.17, this middle section becomes more visible and for 160 Kbit/s with a delay of 300 ms, bottom of Figure 4.17, 30% of additional transmission time is required for a loss at position 50 compared to a loss at position 80, even though fast retransmit should have been used in both cases. This is analyzed in Section 4.2.3.

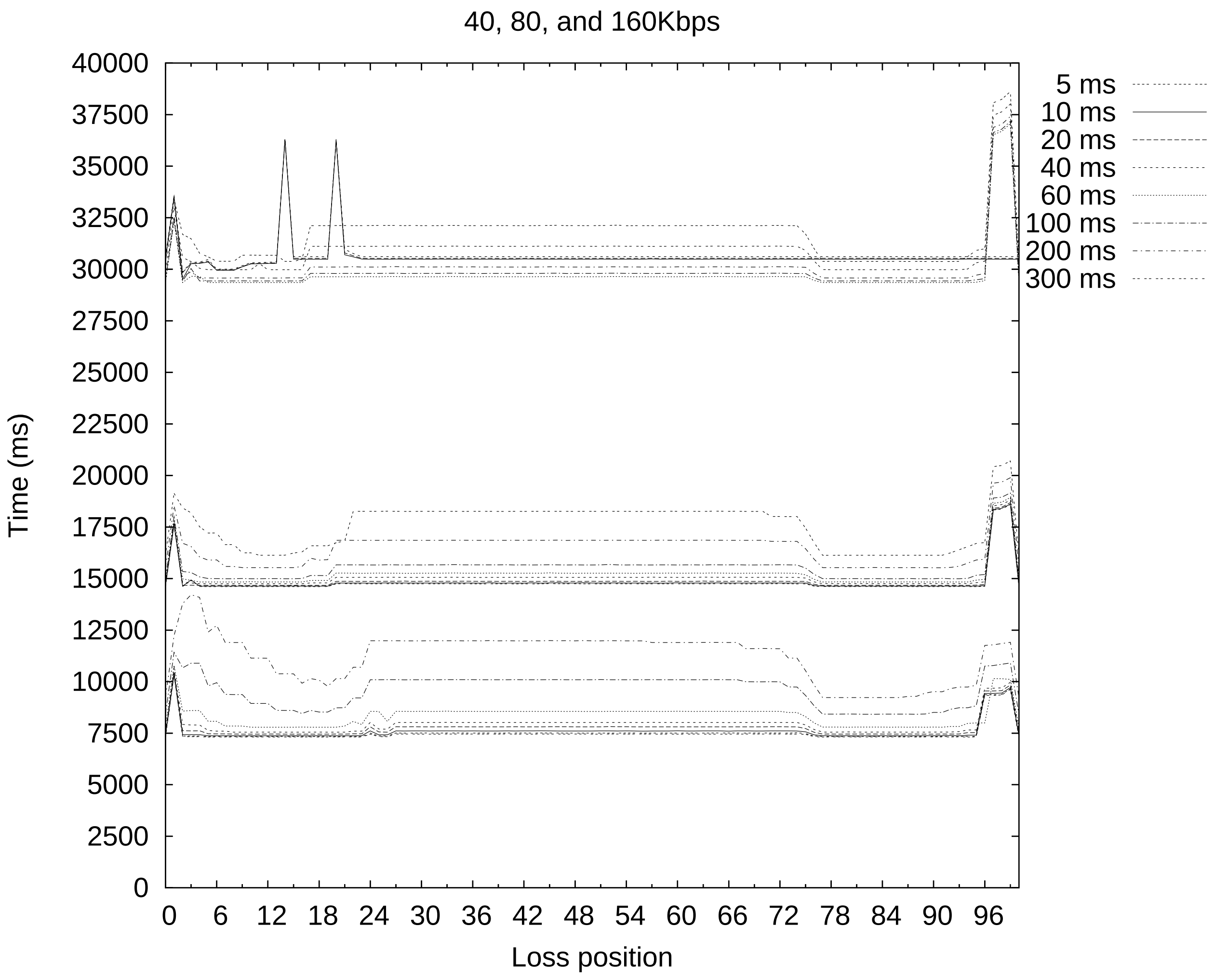


Figure 4.17: Transmission time graph 40, 80, and 160 Kbit/s

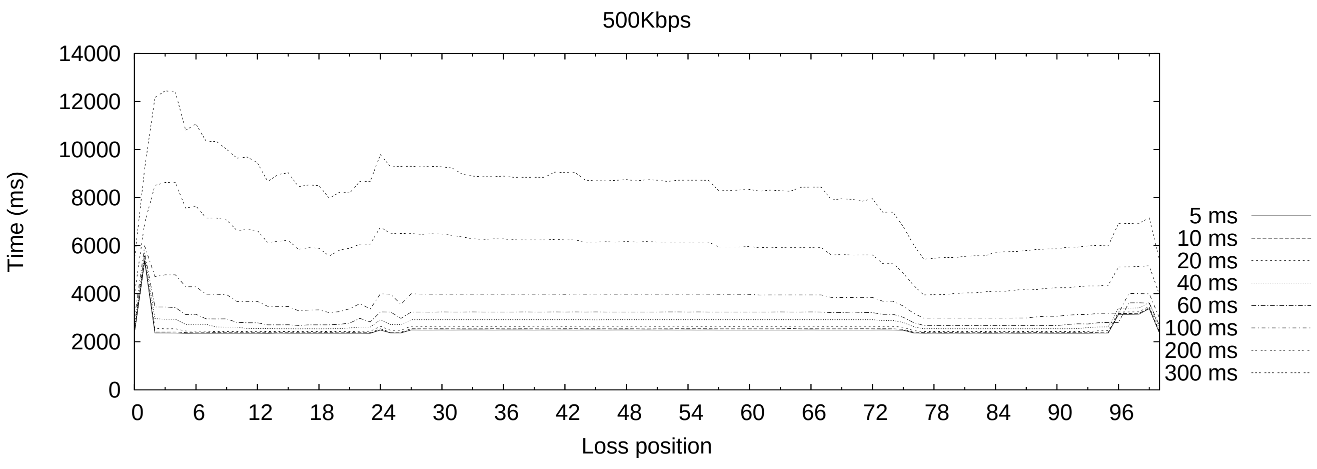


Figure 4.18: Transmission time graph 500 Kbit/s

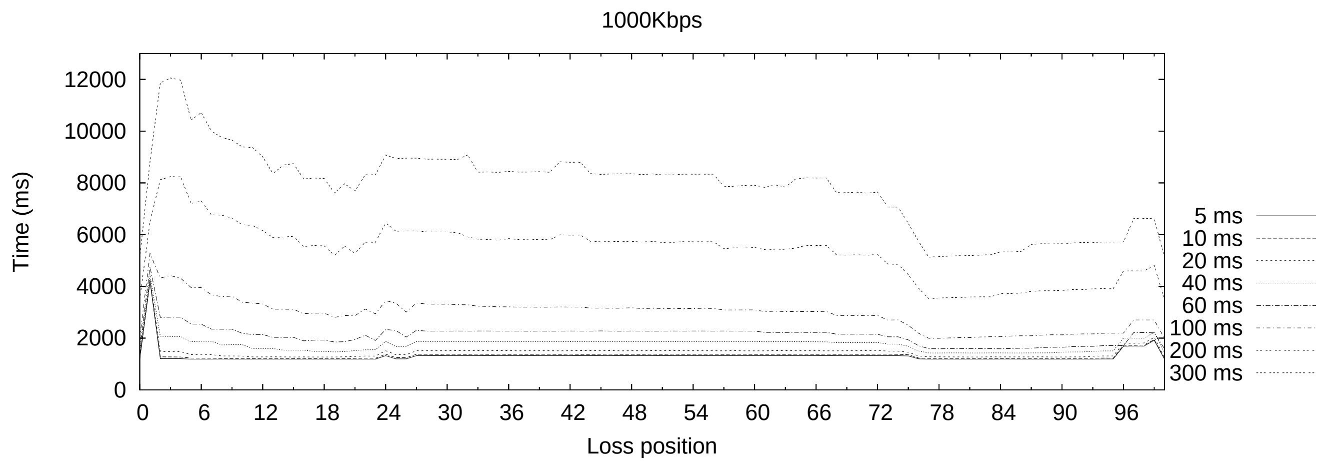
Another interesting thing that is visible in the 160 Kbit/s results, shown in the bottom of Figure 4.17, is that the total transmission time becomes very high, for flows with high delay, if a loss is introduced at an early position. This behavior was also found for the shorter flows, and was analyzed in Section 4.1.4.

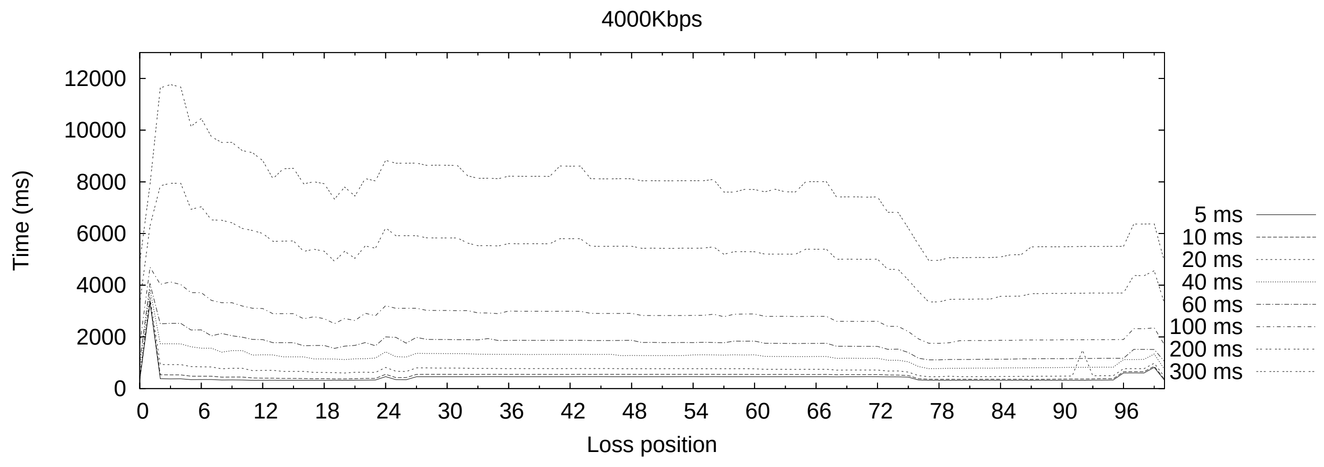
For a bandwidth of 500 Kbit/s, Figure 4.18, we can see that the first section of the graph continue to grow, as the bandwidth and the delay increases. Another interesting observations is that the middle section of the graph becomes more and more irregular as the delay increases. This behavior is discussed in Section 4.2.4.

The rest of the results, shown in Figure 4.19, do not contain any new interesting aspects. The only difference from the previous results is that the impacts of an early loss seem to increase as the bandwidth and the delay increase.

4.2.3 TCP implementation issues

For a loss at an early position ($\approx 2 - 20$) in the case of 80 Kbit/s, and with a delay of 300 ms, Figure 4.17, fast retransmit seems to work. This is also confirmed by Figure 4.20. This figure shows all the different actions and events at the server when a loss was introduced at position 12 in the flow. The y-axis of the graph represents the sequence number space,





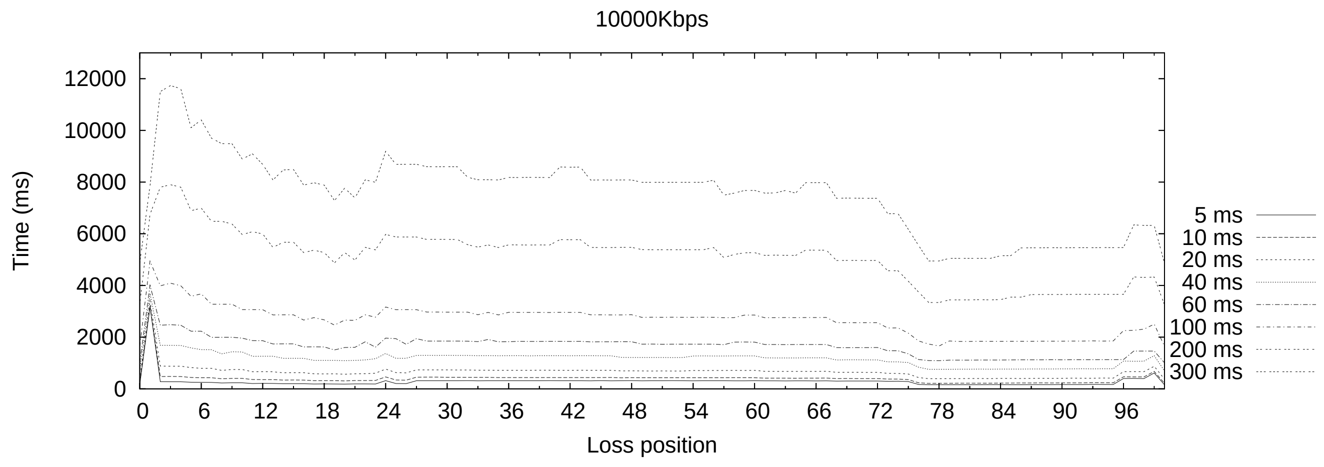


Figure 4.19: Transmission time graph 1000, 4000, and 10000kbps

and the x-axis shows the time. For a more thorough description of the structure of the graph, and the symbols that are used please see Appendix B.1. In this figure, 4.20, we can see that the server was retransmitting the lost segment after receiving three duplicate acknowledgments, and that limited transmit and window inflation was performed in the recovery phase. However, for a loss positioned in the middle section of the flow, position 50, the server acted suspiciously. As mentioned before, in Section 2.2.4, fast retransmission is followed by congestion avoidance which slowly increases the congestion window. But in this case, Figure 4.21, we can see that slow start is conducted, as the server rapidly injects packets into the network (sending two packets for every incoming acknowledgment).

The reason to why this is done has to do with an implementation detail in the FreeBSD kernel.

```
/*
 * Out of fast recovery.
 * Window inflation should have left us
 * with approximately snd_ssthresh
 * outstanding data.
 * But in case we would be inclined to
 * send a burst, better to do it via
 * the slow start mechanism.
 */
if (SEQ_GT(th->th_ack +
                tp->snd_ssthresh,
            tp->snd_max))
        tp->snd_cwnd = tp->snd_max -
                        th->th_ack +
                        tp->t_maxseg;
else
```

```
        tp->snd_cwnd = tp->snd_ssthresh;
```

Listing 4.1: sys/net/inet/tcp_input.c, lines 1983–1999

Listing 4.1 shows an extract of the kernel source code. In the listing `th_ack` is the number of the received acknowledgment, `snd_ssthresh` is the slow start threshold variable, `t_maxseg` is the MSS, and `snd_cwnd` is the congestion window. This piece of code actually suggests that slow start should be used, instead of congestion avoidance, under certain circumstances. To verify that it was this piece of code that actually caused this behavior, it was modified to print out the value of the acknowledgment number, in order to compare it with the traffic log.

Another thing that TCP implementations also should do is to perform slow start until a packet loss has occurred.[5] Looking at Figure 4.21 again we can see that this is not done either, as the packet transmission rate prior to the loss is linear.[6] In addition to this, the server did not use limited transmit or window inflation when the duplicate acknowledgments arrived.

The explanation to this behavior is related to the size of the TCP send buffer. To ensure reliability TCP must buffer copies of transmitted segments, in case that some of them are lost and must be retransmitted. The size of this buffer has a considerable effect on the performance, as the maximum number of packets that can be in flight is implicitly limited by it. In FreeBSD the default size of this buffer is set to 32 Kbytes (please see Appendix C.1), which limits the amount of data that can be in flight to exactly 32 Kbytes. When the congestion window has grown equal to (or greater than) this number, the send buffer is full and for every segment that is acknowledged, TCP is only able to send one new segment.[7] The reason why this phenomenon is not visible in Figure 4.20 is that the

[5]The slow start phase is also to be abandoned when the ssthresh variable is reached. However, for this version of FreeBSD, ssthresh has the same default size as the maximum allowed congestion window, and therefore the slow start phase is not abandoned until the congestion window is at its maximum value, or a packet loss is detected.

[6]If slow start had been conducted, the transmission rate would have increased considerably more.

[7]When transmitted segments are acknowledged they are removed from the buffer.

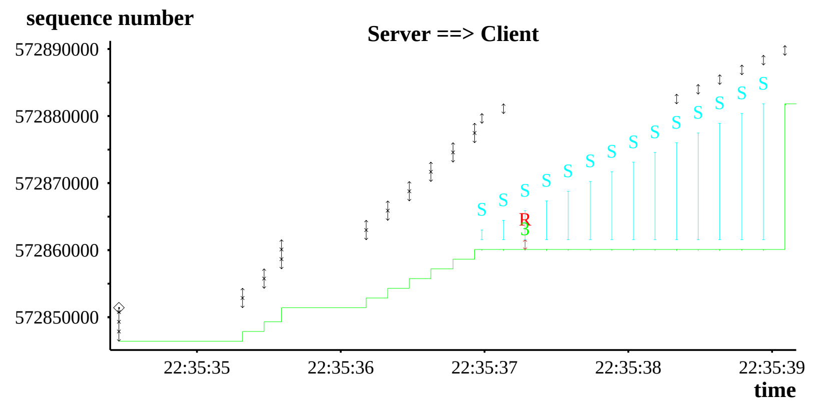


Figure 4.20: Server with 80 Kbit/s, 300ms delay, loss on position 12

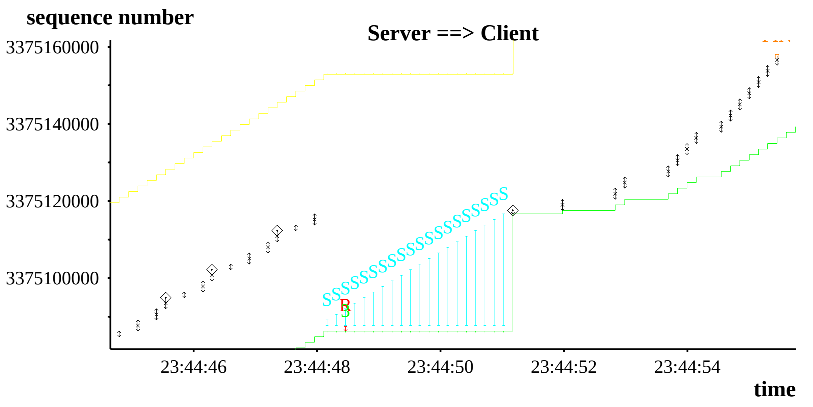


Figure 4.21: Server with 80 Kbit/s, 300ms delay, loss on position 50

congestion window has not had the chance to grow beyond 32 Kbytes yet.

For a packet loss late in the flow, position 84, the 32 Kbyte send buffer did not seem to effect the total transmission time as much as a loss in the middle region did. This is due to the fact that no recovery were needed as all “original” packets already had been sent and the only action the server needed to do was to retransmit the lost packet. This is shown in Figure 4.22.

4.2.4 Irregularity analysis

The reason to why the middle section, shown in Figure 4.18, becomes more irregular as the bandwidth and the delay increases is related to burstiness of TCP. As mentioned in Section

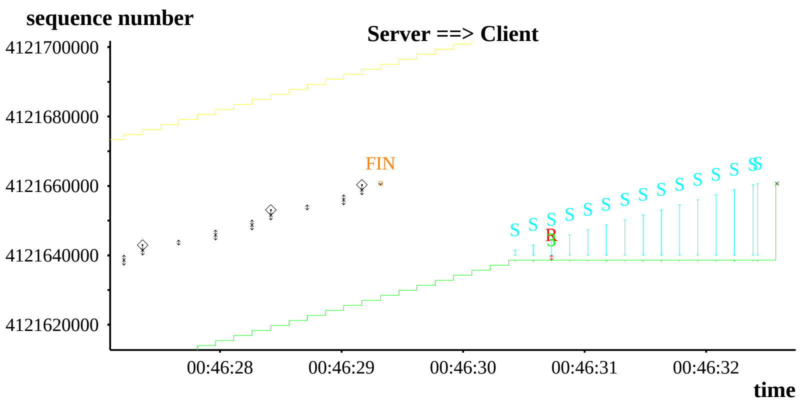


Figure 4.22: Server with 80 Kbit/s, 300ms delay, loss on position 84

4.1.6 the data transfer behavior of TCP makes the sender send packets in partitions, or bursts, and depending on where the loss is introduced the number of such bursts can vary. If we take a look at Figure 4.23, which is the flow with 500 Kbit/s and 300 ms delay, we can see the server when a packet loss was introduced at the 41'th position of the flow (where an irregularity clearly is visible in Figure 4.18). If we, in this figure, count the number of distinct partitions, or bursts, we get the number 12. If we, on the other hand, take a look at Figure 4.24 which shows the same flow, but with a loss on position 50, we can see that there are only 11 bursts. As we can see in Figure 4.18 the difference in the number of bursts, clearly affects the total transmission time (making a slight bump in the graph where the extra burst is). For flows with low delay, an extra burst (which approximately corresponds to an extra round-trip time) is not as significant, in terms of additional time, as for flows with higher delays. That is the reason why the irregularities are not visible, though they exist, for lower delays.

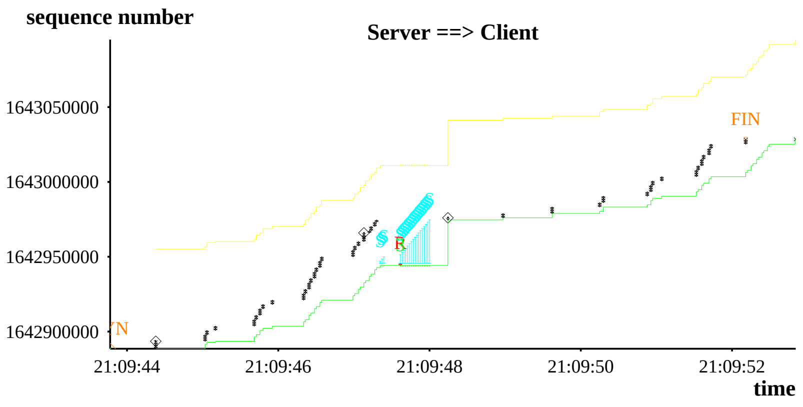


Figure 4.23: Server with 500 Kbit/s, 300ms delay, loss on position 41

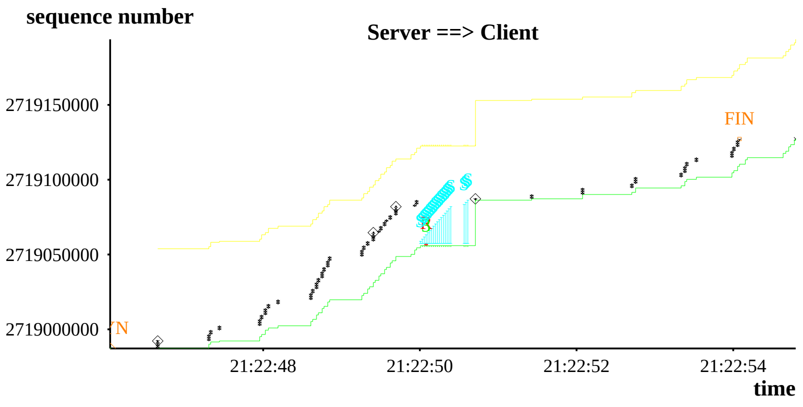


Figure 4.24: Server with 500 Kbit/s, 300ms delay, loss on position 50

4.3 Results related to earlier work

As mentioned earlier, in Section 2.4, other work that has been conducted [11] has also revealed the existence, and to some extent the implications, of fast retransmit inhibitions. One of the results of that work, shown in Figure 4.25, clearly shows that the position of a loss has impact on the total transmission time.

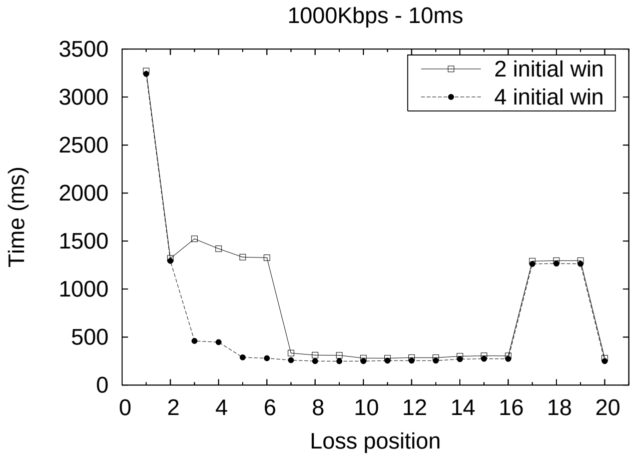


Figure 4.25: One controlled loss, 10 ms

Figure 4.25 shows that the total transmission time for a short-lived flow (20 packets) were considerably longer when the loss was positioned in the beginning or the end of a flow. If we take a look at Figure 4.8 in Section 4.1.2, which contains the result of an experiment with corresponding network parameters[8] (1000 Kbit/s, and 10 ms delay) we can clearly see some differences. First of all we can see that the end section, where fast retransmit is inhibited, of the graph shown in Figure 4.25 is considerably higher than the one shown in Figure 4.8. This is due to the fact that newer versions of FreeBSD break the proposed standard of 1 second minimum RTO [26], thus allowing a much faster timeout to occur. While this behavior seems to work very well in this case, it can also have negative consequences. In Section 4.1.3 we could see that combinations of low bandwidth and low

[8]For our experiment an initial window size that approximately corresponds to 4 was used.

end-to-end delay could cause the RTO timer to expire prematurely, thus causing spurious retransmissions. This could have been avoided if the standard of 1 second minimum RTO had been used.

Another interesting thing worth to note is that the results shown in Figure 4.25 do not contain the enlarged end section that is visible in Figure 4.8 and analyzed in Section 4.1.5. Why this difference exists is not clear, but it may be due to differences in the TCP implemenations.

4.4 Summary

In this chapter we have seen the results of the experiments, and some analysis regarding them. We have seen that fast retransmit inhibitions are present, and that they in most cases affect the performance very much. The problem of fast retransmit inhibitions showed to be greater for short flows, as the risk of experiencing a late loss is much higher.

In addition to this, many interesting and unexpected phenomena have been discovered and discussed. To some of these phenomena we were able to give a complete analysis, but in some cases more work is needed in order to gain a complete understanding.

The next chapter presents some interesting ideas of what can be done in the future.

Chapter 5

Future work

The results that were achieved by the experiments did not only reveal the existence and implications of fast retransmit inhibitions. Several, other, interesting phenomena were discovered and analyzed in the previous chapter. However, some of these phenomena, like the strange peaks that were discussed in Section 4.1.3, were somewhat out of scope for this thesis, and were therefore not fully analyzed. The following sections provide some interesting ideas on future work that is possible.

5.1 Other approaches

The fast retransmit inhibitions could also be investigated using other evaluation techniques, such as

- Mathematical analysis. This approach uses models that describe TCP behavior in mathematical terms. An initial effort will have to be made to classify the different models and select the most appropriate. The model must capture the startup behavior of TCP and the effects of timeouts.

- Simulations. This approach uses the TCP implementation in the ns2 network simulator [1]. The use of simulation allows a large parameter space to be explored, and since the TCP implementation in ns2 is much used in the network research community the results are easy to relate and compare to other simulation studies.

- Implementation examination. When insights into the behavior of the fast retransmit inhibitions have been obtained through one or more of the above approaches, an implementation examination study can also be performed. The purpose of this study would be to examine to what extent current implementations of TCP are sensitive to the inhibitions, and how large the differences between different implementations are.

5.2 SCTP

The same experiments could be conducted for the Stream Control Transmission Protocol [31] (SCTP). SCTP is in many ways similar to TCP, in terms of reliability and congestion control, and it would be nice to investigate if fast retransmit inhibitions exist here as well.

5.3 Dummynet

The extended version of the Dummynet network emulation software, that was used for this work, could be developed further. This in order to support dynamically varying delays and bandwidths.

5.4 FreeBSD Initial window size

Even though the `sysctl` parameter that controls the initial slow start size was manipulated (see Section 3.4.1), and the host caching feature that FreeBSD uses was effectively disabled, the TCP implementation allowed more data to be transmitted, initially, than the TCP standard proposes [5]. In fact, the TCP implementation allowed us to send 4 full-sized segments upon connection establishment, where the TCP standard only allows about 3. It is unclear why the current implementation does this, but it might be possible to find out why using some different approaches.

- By inspecting the source code of the FreeBSD TCP/IP implementation.
- By doing experiments with computers that do not reside on the same subnet, we might get a different behavior.

5.5 FreeBSD Slow Start issue

In Section 4.2.3 it was found that the FreeBSD TCP implementation, under certain cirumstances, chooses to employ the slow start mechanism instead of congestion avoidance. The reason for this, according to the comments in the source code, is that it is better to send a "burst" via the slow start mechanism than the congestion avoidance mechanism. Further experiments could be conducted in order to investigate if this behavior improves the performance, or if the TCP standard way is more efficient.

5.6 FreeBSD recovery issue

In Section 4.1.3 it was observed that the TCP implementation was unable to exit the recovery phase correctly. Due to a bad RTO estimation the server retransmitted a number of segments prematurely. When the server started to send new segments, the acknowledgments to these (sent by the client) triggered the server to send duplicate acknowledgments back, instead of leaving the recovery phase and continue with regular transmission. The reason why the server behaved in that way is unclear, and it would be interesting to investigate this strange behavior further.

5.7 Performance gain with buffer tuning

As the results for the long flows showed, Section 4.2, the performance was affected by the small default size of the TCP send buffer. For servers conducting large file transfers over network paths with high delays, a 32 Kbyte send buffer size is totally unacceptable, as it will limit the link utilization considerably. Some interesting work that could be done is

- Measuring the performance gain as the buffer size increases. This would be relatively easy, and straightforward, as the size of the send buffer in FreeBSD can be controlled with a `sysctl` variable.
- Implement buffer auto-tuning as it is done in the Linux kernel [39]. This would, of course, be a much more complex task to perform, but definitely interesting.

5.8 Lowering the fast retransmit threshold

In the TCP implementation that was studied, the duplicate acknowledgment threshold that fast retransmit uses was three (according to TCP standards). By lowering this threshold fast retransmit opportunities would be gained. However, as mentioned in 2.4, studies show that this could, possibly, make TCP vulnerable to network reordering. Work that could be done here is to investigate if a good estimation technique of network reordering could be developed and used for dynamically lowering the threshold.

Chapter 6

Conclusions

In this thesis the existence of and performance loss due to fast retransmit inhibitions, which usually are evident in the end of TCP connections, have been investigated.

By setting up an experimental environment consisting of a client, a server, and a network emulator which had the ability to lose packets according to their position in a flow, we were able to design and execute experiments that concluded that these inhibitions exist in modern TCP implementations and that they, considerably, can reduce the performance of a TCP connection, especially for short-lived flows.

The experiments that were conducted covered a large number of combinations of network parameters, such as different bandwidths, end-to-end delays, and flow sizes. All together, over 20.000 experiments were performed in order to gain the results. In order to perform this large number of experiments, in an efficient manner, scripts that automated the configuration and execution of the experiments were used. Furthermore, we have ported network emulation software in order to run it on newer versions of FreeBSD, and developed a number of scripts and applications that aided us in the experimental process.

It was shown that the increase in total transmission time, of a short-lived TCP flow, could be as much as 300% when a loss was introduced at a fast retransmit inhibited position in the flow. For larger flows, the worst case scenario resulted in an increase of transmission time with about 170%. Even though the increase of the total transmission time, in precense of a loss where fast retransmit was inhibited, were considerably large in all of the experiments, ranging from $16 - 301\%$, the average performance loss, due to an arbitrarily positioned loss, was not that severe. Because of the fact that fast retransmit is inhibited in fewer positions of a TCP flow than it is employed, the average increase of the transmission time due to these inhibitions was not that large. For the short flows the average increase was at most 20.4%, and for the longer flows, where the risk of having a loss at a fast retransmit inhibited position is less likely, the maximum increase was only 4% at the most.

Not only the fast retransmit inhibitions of TCP affected the results of the conducted experiments. One reason to why the average increase, in terms of additional transmission time, of the long flows was so small had to do with the default buffering capabilities of the TCP implementation that was used. This buffering issue caused the total transmission time to be large, even when fast retransmit was possible to perform, and therefore the difference in performance was smaller. In addition to the buffering issues, some other implementation details, that were related to the behavior and performance of TCP were discovered. One example of this was the management of the RTO timer, which allowed a RTO that is smaller than the standard of 1 second to be used. While this lead to positive performance effects in some cases, it was also shown that it could cause spurious retransmissions, which considerably lowered the performance. We also found a number of other interesting things during the work. Some of them were described and analyzed within the thesis, while others were out of the scope and therefore suggested as future work.

References

[1] The network simulator - ns-2. http://www.isi.edu/nsnam/ns.

[2] Sally Floyd Alberto Medina, Mark Allman. Measuring the evolution of transport protocols in the internet. *ACM SIGCOMM Computer Communication Review*, 35(2), April 2005.

[3] M. Allman. RFC 2581: TCP congestion control. Technical report, April 1999.

[4] M. Allman. RFC 3042: Enhancing TCP's loss recovery using limited transmit. Technical report, January 2001.

[5] M. Allman. RFC 3390: Increasing TCP's initial window. Technical report, October 2002.

[6] Reiner Ludwig Andrei Gurtov. Responding to spurious timeouts in TCP. In *Proc. IEEE INFOCOM*, pages 2312–2322, March 2003.

[7] Ka-Cheong Leung Changming Ma. Improving TCP robustness under reordering network environment. In *Proc. IEEE GLOBECOM,2004*, page 2072, December 2004.

[8] Mark Allman Ethan Blanton. On making TCP more robust to packet reordering. *ACM SIGCOMM Computer Communication Review*, 32(1), January 2002.

[9] S. Floyd. RFC 3782: The newreno modification to TCP's fast recovery algorithm. Technical report, April 2004.

[10] FreeBSD 6.0. http://www.freebsd.org.

[11] Johan Garcia. *Improving Performance in Heterogeneous Networks: A Transport Layer Centered Approach.* PhD thesis, Karlstad University, 2005.

[12] V. Jacobson. RFC 1323: TCP extensions for high performance. Technical report, May 1992.

[13] Van Jacobson. Congestion avoidance and control. In *ACM SIGCOMM '88*, volume 18, pages 314–329, August 1988.

[14] Matthew Mathis Jeffrey Semke, Jamshid Mahdavi. Automatic TCP buffer tuning. In *ACM SIGCOMM*, volume 28, pages 315–323, October 1998.

[15] Nicholas Shectman Jon C. R. Bennett, Craig Partridge. Packet reordering is not pathological network behavior. *IEEE/ACM Transactions on Networking (TON)*, 7(6), December 1999.

[16] Tom Kelly. Scalable TCP: improving performance in highspeed wide area networks. *ACM SIGCOMM Computer Communication Review*, 33(2), April 2003.

[17] Larry Peterson Lawrence Brakmo, Sean O'Malley. TCP vegas: New techniques for congestion detection and avoidance. In *SIGCOMM*, volume 24, pages 24–35, October 1994.

[18] Libpcap. http://www.tcpdump.org.

[19] U. Ayesta J. Blanton M. Allman, K. Avrachenkov. Early retransmit for TCP and SCTP. IETF,Draft,work in progress, August 2003.

[20] Claudio Casetti Marco Mellia, Michela Meo. TCP Smart-Framing: using smart segments to enhance the performance of TCP. In *Proceedings of IEEE GLOBECOM 2001*, pages 1708–1712, April 2001.

[21] Claudio Casetti Marco Mellia, Michela Meo. TCP smart framing: a segmentation algorithm to reduce TCP latency. *IEEE/ACM Transactions on Networking (TON)*, 13(2):316–329, April 2005.

[22] M. Mathis. RFC 2018: TCP selective acknowledgment options. Technical report, October 1996.

[23] S. McCanne and V. Jacobson. The bsd packet filter: a new architecture for user-level packet capture. In *Proceedings of the Winter 1993 USENIX Conference*, pages 259–269, San Diego, California, USA, January 1993.

[24] Shawn Ostermann. Tcptrace. http://www.tcptrace.org.

[25] Kimmo Raatikainen Pasi Sarolahti, Markku Kojo. F-RTO: an enhanced recovery algorithm for TCP retransmission timeouts. *ACM SIGCOMM Computer Communication Review*, 33(2):51–63, April 2003.

[26] V. Paxson. RFC 2988: Computing TCP's retransmission timer. Technical report, November 2000.

[27] Vern Paxson. End-to-end internet packet dynamics. *IEEE/ACM Transactions on Networking (TON)*, 7(3), June 1999.

[28] J. Postel. RFC 791: Internet protocol. Technical report, September 1981.

[29] J. Postel. RFC 793: Transmission control protocol. Technical report, September 1981.

[30] R. Prasad, M. Jain, and C. Dovrolis. Socket buffer auto-sizing for high-performance data transfers. *Journal of Grid Computing*, 1(4):361–376, 2004.

[31] et al. R. Stewart. RFC 2960: Stream control transmission protocol. Technical report, October 2000.

[32] Randy H. Katz Reiner Ludwig. The eifel algorithm: making TCP robust against spurious retransmissions. *ACM SIGCOMM Computer Communication Review*, 30(1), January 2000.

[33] Kenshin Yamada M. Y. Sanadidi Mario Gerla Ren Wang, Giovanni Pau. TCP startup performance in large bandwidth delay networks. In *Proc. IEEE INFOCOM*, pages 795–804, March 2004.

[34] Luigi Rizzo. Dummynet: a simple approach to the evaluation of network protocols. *ACM Computer Communication Review*, 27(1):31–41, 1997.

[35] Luigi Rizzo. Dummynet and forward error correction. In *Freenix 98*, New Orleans, June 1998.

[36] Tim Shepard. Xplot. http://www.xplot.org.

[37] Tcpdump. http://www.tcpdump.org.

[38] Brian Tierney Tom Dunigan, Matt Mathis. A TCP tuning deamon. In *Proceedings of the 2002 ACM/IEEE conference on Supercomputing*, November 2002.

[39] Linus Torvalds and The Free Software Community. The Linux Kernel. http://www.kernel.org, September 1991.

Appendix A

List of Abbreviations

ACK	Acknowledgment
cwnd	Congestion Window
BDP	Bandwidth-delay Product
FIN	Finish
FTP	File Transfer Protocol
IP	Internet Protocol
MSS	Maximum Segment Size
MTU	Maximum Transfer Unit
RTO	Retransmission TimeOut
RTT	Round-Trip Time
rwnd	Receiver window
ssthresh	Slow start threshold
SCTP	Stream Control Transmission Protocol
SYN	Synchronize
TCP	Transmission Control Protocol

Appendix B

Additional Software

B.1 Tcptrace

Tcptrace [24] is a piece of software that can produce graphs based on traffic logs generated by Tcpdump (see Section 3.4.5). These time sequence graphs can then be viewed by Xplot [36] to show all the actions and events that occur during a lifetime of a TCP connection. The y-axis in these graphs represents the sequence number space, and the x-axis shows the time. The symbols that appears in the graphs are presented in Figure B.1, and described in table B.1.

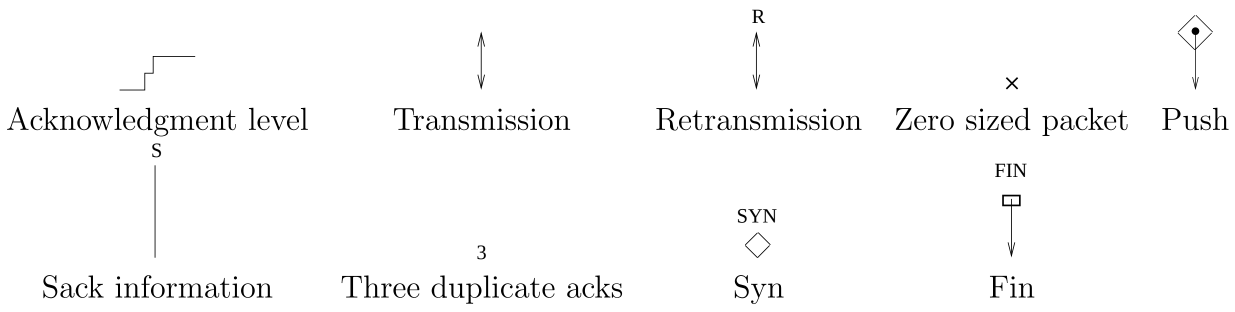


Figure B.1: Tcptrace Symbols

In Figure B.2 an example of a time sequence graph is shown. In this example a connection between a client and a server is shown from the servers point of view. If we start from the beginning of the connection (at time 16:29:12) and walk forwards we can see the following events:

1. A packet with the `SYN` flag has been received by the server, thus a TCP connection has been established.
2. The server sends four TCP segments to the client. The last of these packets has the `PUSH` flag set.
3. When the acknowledgment to the first segment arrives at the server (indicated by the growing acknowledgment level) it transmits two new segments.
4. The behavior continues; acknowledgments $\Rightarrow$ more segments sent.
5. Duplicate acknowledgments with SACK information arrives at the server. When three of these are received (indicated by the symbol “3” in the graph) a retransmission is conducted by the server (indicated by the symbol “R”). Apparantly the fifth segment that was sent from the server was lost on its' way to the client.[1]
6. The arrival of duplicate acknowledgments continues until the retransmission is acknowledged.
7. The server tries to close the connection by sending a segment with the `FIN` flag set.
8. Acknowledgments to transmitted segments arrive.
9. The connection is terminated.

[1]This can be concluded by comparing the arrowheads of the retransmitted segment and segment number five.

Symbol	**Description**
Acknowledgment level	This line keeps track of the acknowledgments received from the other end. If a graph contains two of these lines the upper one represents the advertised receiver window.
Transmission	Represents sent packets. The down arrow represents the sequence number of the first byte of the packet, and the up arrow represents the last.
Retransmission	Represents retransmitted packets. The up and down arrows has the same meaning as for regular transmissions.
Zero sized packet	Packet without data sent.
Push	Packet with the `PUSH` flag set.
Sack information	Duplicate acknowledgment with SACK [22] information received. The line shows the sequence range for the SACK blocks.
Three duplicate acks	Indicates that three duplicate acknowledgments has been received.
Syn	A packet with the `SYN` flag has been sent.
Fin	A packet with the `FIN` flag has been sent.

Table B.1: Tcptrace symbol explanation

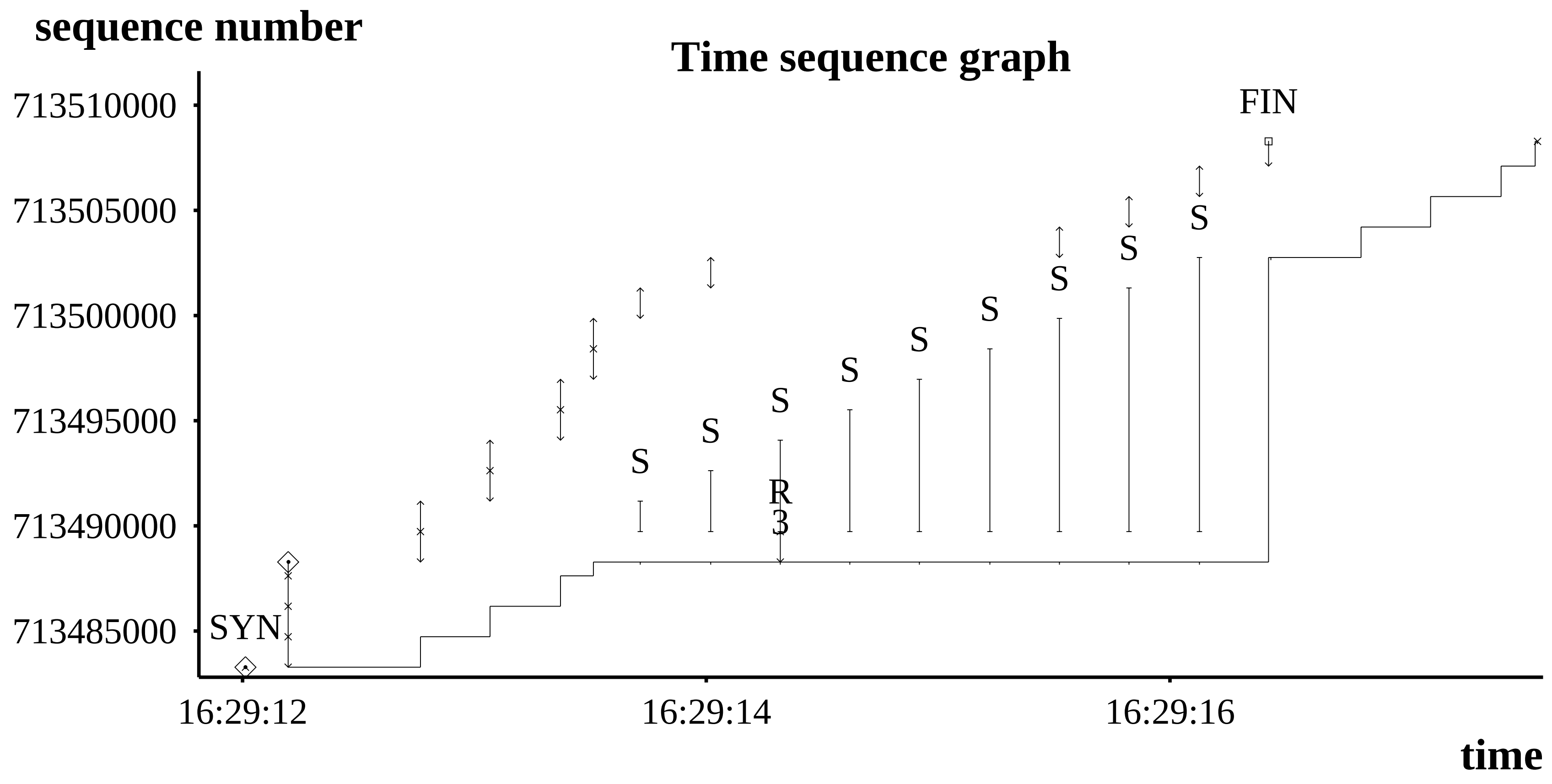


Figure B.2: Tcptrace Example

Appendix C

FreeBSD Kernel details & modifications

In this appendix some relevant information about the FreeBSD 6 TCP implementation is provided. This information includes a subset of the different parameters that the TCP implementation uses, along with descriptions of them and their default values. The appendix also provides information on how the initial RTO timer is calculated, and how the TCP implementation was modified in order to disable the host caching feature.

C.1 TCP Implementation

The TCP implementation of FreeBSD 6 is a modern TCP implementation which supports a wide range of the proposed RFC TCP standards. The implementation is based on the NewReno congestion control, and it also supports a modern technique, bandwidth-delay product limiting, for preventing network congestion.

A large amount of parameters that controls the behavior of the FreeBSD TCP implementation is available via the `sysctl` command. `sysctl` is a command that is used to modify, and inspect, kernel parameters during runtime. In table C.1 a subset of these parameters is shown and described.[1] The default values of the parameters listed in these tables are presented in table C.2.

Parameter (net.inet.tcp.)	**Description**
rfc1323	Implement the window scaling and timestamp options of RFC 1323.
rfc3042	Enable the Limited Transmit algorithm as described in RFC 3042. It helps avoid timeouts on lossy links and also when the congestion window is small, as happens on short transfers.
rfc3390	Enable support for RFC 3390, which allows for a variable-sized starting congestion window on new connections, depending on the maximum segment size. This helps throughput in general, but particularly affects short transfers and high-bandwidth large propagation-delay connections. When this feature is enabled, the slowstart_flightsize and local_slowstart_flightsize settings are not observed for new connection slow starts, but they are still used for slow starts that occur when the connection has been idle and starts sending again.
sack.enable	Enable support for RFC 2018, TCP Selective Acknowledgment option, which allows the receiver to inform the sender about all successfully arrived segments, allowing the sender to retransmit the missing segments only.
sendspace	Maximum TCP send window.
recvspace	Maximum TCP receive window.

[1]The table is extracted from the FreeBSD TCP manual pages.

slowstart_flightsize	The number of packets allowed to be in-flight during the TCP slow-start phase on a non-local network.
local_slowstart_flightsize	The number of packets allowed to be in-flight during the TCP slow-start phase to local machines in the same subnet.
msl	The Maximum Segment Lifetime, in milliseconds, for a packet.
delayed_ack	Delay acknowledgment to try and piggyback it onto a data packet.
delacktime	Maximum amount of time, in milliseconds, before a delayed acknowledgment is sent.
newreno	Enable TCP NewReno Fast Recovery algorithm, as described in RFC 2582.
path_mtu_discovery	Enable Path MTU Discovery.

rexmit_min, rexmit_slop	Adjust the retransmit timer calculation for TCP. The slop is typically added to the raw calculation to take into account occasional variances that the SRTT (smoothed round-trip time) is unable to accommodate, while the minimum specifies an absolute minimum. While a number of TCP RFCs suggest a 1 second minimum, these RFCs tend to focus on streaming behavior, and fail to deal with the fact that a 1 second minimum has severe detrimental effects over lossy interactive connections, such as a 802.11b wireless link, and over very fast but lossy connections for those cases not covered by the fast retransmit code. For this reason, we use 200ms of slop and a near-0 minimum, which gives us an effective minimum of 200ms (similar to Linux).

inflight.enable	Enable TCP bandwidth-delay product limiting. An attempt will be made to calculate the bandwidth-delay product for each individual TCP connection, and limit the amount of inflight data being transmitted, to avoid building up unnecessary packets in the network. This option is recommended if you are serving a lot of data over connections with high bandwidth-delay products, such as modems, GigE links, and fast long-haul WANs, and/or you have configured your machine to accommodate large TCP windows. In such situations, without this option, you may experience high interactive latencies or packet loss due to the overloading of intermediate routers and switches. Note that bandwidth-delay product limiting only effects the transmit side of a TCP connection.
inflight.debug	Enable debugging for the bandwidth-delay product algorithm.
inflight.min	This puts a lower bound on the bandwidth-delay product window, in bytes. A value of 1024 is typically used for debugging. 6000-16000 is more typical in a production installation. Setting this value too low may result in slow ramp-up times for bursty connections. Setting this value too high effectively disables the algorithm.

inflight.max	This puts an upper bound on the bandwidth-delay product window, in bytes. This value should not generally be modified, but may be used to set a global per-connection limit on queued data, potentially allowing you to intentionally set a less than optimum limit, to smooth data flow over a network while still being able to specify huge internal TCP buffers.
inflight.stab	The bandwidth-delay product algorithm requires a slightly larger window than it otherwise calculates for stability. This parameter determines the extra window in maximal packets / 10. The default value of 20 represents 2 maximal packets. Reducing this value is not recommended, but you may come across a situation with very slow links where the ping(8) time reduction of the default inflight code is not sufficient. If this case occurs, you should first try reducing inflight.min and, if that does not work, reduce both inflight.min and inflight.stab, trying values of 15, 10, or 5 for the latter. Never use a value less than 5. Reducing inflight.stab can lead to upwards of a 20% underutilization of the link as well as reducing the algorithm's ability to adapt to changing situations and should only be done as a last resort.

Table C.1: TCP parameters

Parameter (net.inet.tcp.)	**Default value**
rfc1323	Enabled
rfc3042	Enabled
rfc339	Enabled
sack.enable	Enabled
sendspace	32768 bytes
recvspace	65536 bytes
slowstart_flightsize	1 packet
local_slowstart_flightsize	4 packets
msl	30000 milliseconds
delayed_ack	Enabled
delacktime	100 milliseconds
newreno	Enabled
path_mtu_discovery	Enabled
rexmit_min	3 milliseconds
rexmit_slop	200 milliseconds
inflight.enable	Enabled
inflight.debug	Disabled
inflight.min	6144 bytes
inflight.max	1073725440 bytes
inflight.stab	20

Table C.2: Default values of TCP parameters

C.1.1 Initial RTO Calculation

In FreeBSD 6 the RTO timer is (re)calculated each time an acknowledgment arrives. The calculation is triggered on lines 1188 – 1199 in the `sys/net/inet/tcp_input.c` file. A function named `tcp_xmit_timer` is called with the difference between the current time (measured in ticks) and the time when the packet that triggered the acknowledgment was sent (also in ticks) plus one tick.[2] This is the actual round-trip time of the packet that the acknowledgment acknowledges.[3]

For the listings in this section there are some variable who's values do not show. These values are listed in Table C.3.

The function `tcp_xmit_timer` starts by determining whether a smoothed round-trip time has been previously calculated or not, if this is not the case the function calculates the smoothed rtt (`t_srtt`), the rtt variance (`t_rttvar`) and the best rtt (`rttbest`) with the help of the given round-trip time (`rtt`);

```
tp->t_srtt = rtt << TCP_RTT_SHIFT;
tp->t_rttvar = rtt << (TCP_RTTVAR_SHIFT - 1);
tp->t_rttbest = tp->t_srtt + tp->t_rttvar;
```

Listing C.1: sys/net/inet/tcp_input.c, lines 2744-2746

After these steps have been conducted the retransmission timer (`t_rxtcur`) is set;

```
TCPT_RANGESET(tp->t_rxtcur, TCP_REXMTVAL(tp),
  max(tp->t_rttmin, rtt + 2), TCPTV_REXMTMAX);
```

[2]If the timestamp option (RFC 1323 [12]) is turned off the RTO calculation is not called from here, and not with the same parameters.

[3]On most architectures one tick corresponds to one millisecond

Listing C.2: sys/net/inet/tcp_input.c, lines 2762-2763

where the macro `TCPT_RANGESET` is defined as

```
#define TCPT_RANGESET(tv, value, tvmin, tvmax) do { \
  (tv) = (value) + tcp_rexmit_slop; \
  if ((u_long)(tv) < (u_long)(tvmin)) \
    (tv) = (tvmin); \
  else if ((u_long)(tv) > (u_long)(tvmax)) \
    (tv) = (tvmax); \
} while(0)
```

Listing C.3: sys/net/inet/tcp_timer.h, lines 129-135

and the macro `TCP_REXMTVAL` as

```
#define TCP_REXMTVAL(tp) \
  max((tp)->t_rttmin, (((tp)->t_srtt >> (TCP_RTT_SHIFT - \
  TCP_DELTA_SHIFT)) + (tp)->t_rttvar) >> TCP_DELTA_SHIFT)
```

Listing C.4: sys/net/inet/tcp_var.h, lines 341-344

C.2 Deactivating the TCP host cache

In order to disable the host caching feature of the FreeBSD TCP implementation, the source code shown in Listing C.5, was changed to look like the code in Listing C.6. When this change was performed the only thing that was required to be done for the host cache to be emptied, was to change the value of the `sysctl` variable `net.inet.tcp.hostcache.purge`

Name	Value
`TCP_RTT_SHIFT`	5
`TCP_RTTVAR_SHIFT`	4
`TCPTV_REXMTMAX`	64 * hz
`TCP_DELTA_SHIFT`	2
`hz`	1000
`t_rttmin`	3
`tcp_rexmit_slop`	200

Table C.3: TCP constants

from 0 to 1 and wait for 5 seconds. Originally a purge of the host cache only resulted in an empty host cache within 5 minutes, which was too long.

```
/* Arbitrary values */
#define TCP_HOSTCACHE_HASHSIZE        512
#define TCP_HOSTCACHE_BUCKETLIMIT     30
#define TCP_HOSTCACHE_EXPIRE          60*60     /* one hour */
#define TCP_HOSTCACHE_PRUNE           5*60      /* every 5 minutes */
```

Listing C.5: sys/net/inet/tcp_hostcache.c, lines 130-134

```
/* Arbitrary values */
#define TCP_HOSTCACHE_HASHSIZE        512
#define TCP_HOSTCACHE_BUCKETLIMIT     30
#define TCP_HOSTCACHE_EXPIRE          60*60     /* one hour */
#define TCP_HOSTCACHE_PRUNE           5         /* every 5 seconds */
```

Listing C.6: sys/net/inet/tcp_hostcache.c, lines 130-134, modified

Appendix D

Source code & Scripts

This appendix provides the source codes to some of the applications and scripts that were used in this work. The first section in the appendix presents the loss pattern generator mentioned in Section 3.4.4, the second section describes the experiment script that was used in order to automatically configure the environment and run the experiments, and, finally, the third section in this appendix describes the client & server applications that were used for the experiments.

D.1 Loss pattern generator

This application was developed in order to simplify the creation of the loss patterns that were used by the Dummynet network emulator. In the two following subsections the usage of the application, and its source code are provided.

D.1.1 Usage

The usage of the loss pattern generator is as follows;

```
loss_pos <sequence factor> <positions> <filename> <mode> [<start value>]
```

where the argument `<sequence factor>` specifies the length of the pattern. Let us say that a sequence factor of 2 is entered, then the length of the pattern is $2 * 16 = 32$, which means that after 32 processed packets Dummynet will start over from the beginning of the pattern again. The argument `<positions>` is a comma separated list of positions where losses should occur in the flow. A list of `1,4`, would make Dummynet to lose the first and the fourth packet. The `<filename>` argument is simply the filename of the pattern that should be created. This is the file that should be loaded into Dummynet. The argument `<mode>` is used to control how the comma separated list, that was mentioned earlier, is interpreted. If `<mode> == time` then the numbers in the list specifies when to switch from non-loss to loss. For example, if `<positions> == 4,8` and `<mode> == time`, then there will be no losses on positions $0 - 3$, losses on positions $4 - 7$, and no losses from position 8 and on. If `<mode> == time` then the optional argument `<start value>` can be used. By default `<start value> == 0`. If we change this value to 1, however, then the previous example would result in a pattern that specifies; losses on positions $0 - 3$, no losses on positions $4 - 7$, and losses from position 8 and on.

D.1.2 Source code

```
#include <stdio.h>
#include <stdlib.h>
#include <strings.h>
#include <string.h>

#define SEQUENCE_LENGTH_ARG    1
#define LOSS_POSITIONS_ARG     2
#define LOSS_PATTERN_NAME      3
#define LOSS_MODE              4
#define TIME_DRIVEN_START      5
```

```
#define REQUIRED_ARGUMENTS   5


int sequence_length = 0;

int determine_chunk(int position) {
  return position/16;
}

int determine_position(int position) {
  return 15-(position%16);
}

void set_pattern(int chunk, int position, short int* patt) {
  if(chunk < sequence_length)
    patt[chunk] |= (unsigned short)(1 << position);
  else
    fprintf(stderr,"Warning: One of the specified loss positions
       is out of range for the given sequence.\n");
}

void set_zero(int chunk, int position, short int* patt) {
  patt[chunk] = (patt[chunk] & (~(1 << position)));
}

int is_one(int chunk, int position, short int* patt) {
  return patt[chunk] & (1 << position);
}

int is_digit(char arg) {
  return (arg >= '0' && arg <= '9');
}

int is_integer(const char* arg) {

  int result = 1;
  int i;

  for(i=0; i < strlen(arg); i++) {
    result &= is_digit(arg[i]);
  }

  return result;
}

int set_losses(const char* loss_list, short int* patterns) {
  if (is_integer(loss_list)) {
    int tmp = atoi(loss_list);
    set_pattern(determine_chunk(tmp),determine_position(tmp),
       patterns);
    return 1;
  } else {
    const char* list_element = strtok(loss_list,",");
    if(list_element != NULL && is_integer(list_element)) {
      int tmp = atoi(list_element);
      set_pattern(determine_chunk(tmp),determine_position(tmp),
         patterns);
      return 1 &
         set_losses(loss_list+strlen(list_element)+1,patterns);
    }
  }
  return 0;
}


int get_list_element(char* list) {
  const char* list_element = strtok(list,",");
  list = list + strlen(list_element) + 1;
  printf("list: %s\n", list);
  if (list_element != NULL)
    return atoi(list_element);
  return -1;
}

int main(int argc, char** argv) {

  /* Validate command line arguments (START) */
  if(argc < REQUIRED_ARGUMENTS ||
     !is_integer(argv[SEQUENCE_LENGTH_ARG])) {
```

```
    fprintf(stderr, "Error: Either the number of arguments is
        insufficient or the loss position list is invalid.\n");
    exit(0);
  }
  if(atoi(argv[SEQUENCE_LENGTH_ARG]) < 1) {
    fprintf(stderr, "Error: sequence factor must be greater than
        1. Aborting.\n");
    exit(0);
  }

  sequence_length = atoi(argv[SEQUENCE_LENGTH_ARG]);
  unsigned short patterns[sequence_length];
  bzero(patterns,sizeof(unsigned short)*sequence_length);

  if(strcmp("data", argv[LOSS_MODE]) == 0) {
    if(set_losses(argv[LOSS_POSITIONS_ARG],patterns) == 0) {
      fprintf(stderr,"Error when setting losses according to loss
          position list.\n");
      exit(0);
    }
  }
  else if(strcmp("time",argv[LOSS_MODE]) == 0) {
    int mode = 0;
    if(argc > 5)
      mode = atoi(argv[TIME_DRIVEN_START]);

    if(set_losses(argv[LOSS_POSITIONS_ARG], patterns) == 0) {
      fprintf(stderr,"Error when setting losses according to loss
          position list.\n");
      exit(0);
    }

    if(mode == 0 || mode == 1) {
      int i,j;
      for(i=0; i < sequence_length; i++) {
  for(j=15; j >= 0; j--) {
    if(is_one(i,j,patterns)) {
      mode = ( mode ? 0 : 1 );
    }
    if(mode) {
      set_pattern(i,j,patterns);
    }
    else {
      set_zero(i,j,patterns);
    }
  }
      }
    }
    else {
      fprintf(stderr,"Error: Start value inccorect.\n");
      exit(0);
    }

  }
  else {
    fprintf(stderr,"Error: Mode does not exist.\n");
    exit(0);
  }

  /* Write pattern to file */
  FILE *outfile;
  outfile = fopen(argv[LOSS_PATTERN_NAME],"w");
  if (outfile == NULL) {
    fprintf(stderr,"Error: Could not open %s for writing.
        Aborting.",argv[LOSS_PATTERN_NAME]);
    exit(0);
  }
  if(fwrite(&patterns,sizeof(patterns),1,outfile) == 0) {
    fprintf(stderr,"Error: Could not write to file %s. Aborting",
         argv[LOSS_PATTERN_NAME]);
    exit(0);
  }
  fclose(outfile);

  return 0;
}
```

Listing D.1: Loss pattern generator

D.2 Experiment script

In this section the script that was used to configure the environment and to execute the experiments is described. In fact, it is two scripts; one script, written in Perl, that takes care of the environment configuration, and one shell script that executes the actual experiment. The shell script is called by the configuration script when the configuration of the environment is completed.

The scripts provided in this appendix are designed for the experiments with the short flows, but comments are provided within the code to show how the experiments with the longer flows were conducted as well.

These scripts are modified versions of the scripts that were used for the experiments in [11].

D.2.1 Source code

Configuration script

```
#!/usr/bin/perl
# Define general options
# 041105 Johan Garcia
# Modified 050830 Per Hurtig
#-----------------------------
$replikat=3;

# name template for the output files
$outfilnamn="/home/per/experiment/test_";

# Machine identities
# This script is designed to be run on the client
$server = "10.0.2.1";
$client = "10.0.1.1";
$rctrl  = "root\@3g.carl.kau.se";
```

```
$sctrl  = "per\@3b.carl.kau.se";

$portnr = "1234";

# Network Parameter
#------------------------------------

# Define the test case parameters
@bwdown =
    ("40Kbit/s","80Kbit/s","160Kbit/s","320Kbit/s","500Kbit/s","1000Kbit/s","
@bwup    =
    ("40Kbit/s","80Kbit/s","160Kbit/s","320Kbit/s","500Kbit/s","1000Kbit/s","

@delay = (5,10,20,40,60,80,100,150,200,250,300);

@queue = (99);

# Set the filesizes to be retrieved * 2500
@size = (10);

$patterns=20;
$patternprefix="patterns/pattern";

# For flows with length 100

#@bwdown =
    ("40Kbit/s","80Kbit/s","160Kbit/s","500Kbit/s","1000Kbit/s","4000Kbit/s"
#@bwup     =
    ("40Kbit/s","80Kbit/s","160Kbit/s","500Kbit/s","1000Kbit/s","4000Kbit/s"

#@delay = (5,10,20,40,60,100,200,300);

#@queue = (99);

# Set the filesizes to be retrieved * 2500
#@size = (56);

#$patterns=100;
#$patternprefix="patterns/pattern";


# Private adresses are considered local,
# so fix initial window
system("sudo sysctl net.inet.tcp.local_slowstart_flightsize=1");
system("ssh $sctrl sudo sysctl
    net.inet.tcp.local_slowstart_flightsize=1");

# Disable inflight bandwidth limiting
system("sudo sysctl net.inet.tcp.inflight.enabled=0");
system("ssh $sctrl sudo sysctl net.inet.tcp.inflight.enabled=0")

$tcidx = 0;
$testcount = 1;
```

```
# Print outfile header
open (FDR, ">>$outfilnamn");
print FDR "Nr      BandwDown   BandwUp    Del Que Fsize   DrpCnt
   Lossdiff   Time     \n";
close (FDR);

# Remove possible old server & tcpdump
system ("ssh $sctrl killall tcp_server");
system ("ssh $sctrl killall tcpdump");

foreach $size  (@size){
 system("ssh -f $sctrl ./tcp_server_ny $portnr $size");
 $tcidx=0;
 while($tcidx < @bwdown){
  foreach $queue (@queue) {
   foreach $delay (@delay) {
    for($pattcount = 0; $pattcount <= $patterns; $pattcount++) {
     $repcount=$replikat;
     while($repcount--) {
      # Reset route ssthresh delay etc memory between runs
      system("ssh $sctrl sudo route delete $client");
      system("ssh $sctrl sudo route add -host $client 10.0.2.2");

      # Reset host caching.
      system("sudo sysctl net.inet.tcp.hostcache.purge=1");
      system("ssh $sctrl sudo sysctl
         net.inet.tcp.hostcache.purge=1");
```

```
# Remove old Dummynet configuration
system("ssh $rctrl ipfw -f flush");
system("ssh $rctrl ipfw -f pipe flush");

# To block the icmp source quench dummynet sends when
    buffer becomes full
system("ssh $rctrl ipfw add drop icmp from any to any out
   icmptypes 4");

# Create Dummynet pipes
system("ssh $rctrl ipfw add 2 pipe 200 tcp from $server
   $portnr to any in");
system("ssh $rctrl ipfw add 3 pipe 300 tcp from any to
   $server $portnr out");
system("ssh $rctrl ipfw add 4 pipe 400 tcp from $server
   $portnr to any out");

$packlfile=$patternprefix.$pattcount.$plr.".pep";

# Configure Dummynet pipes
system("ssh $rctrl ipfw pipe 200 config packlfile
   $packlfile");
system("ssh $rctrl ipfw pipe 300 config bw $bwdown[$tcidx]
   delay $delay queue $queue");
system("ssh $rctrl ipfw pipe 400 config bw $bwup[$tcidx]
   delay $delay queue $queue");
```

```
      open(FDR, ">>$outfilnamn");
      printf FDR("%4i %10s %10s %3d %3d %3d
         %s",$testcount,$bwdown[$tcidx],$bwup[$tcidx],$delay,$queue,$size,$
      close(FDR);



      # Serverside logging
      system("ssh -f $sctrl sudo tcpdump -i xl0 -w /tmp/temp.pcap
           tcp port $portnr &");
      system("sleep 5");

      # Execute test
      system ("./runtest.sh $server $portnr >> $outfilnamn");
      system("sudo gzip -f /tmp/temp.pcap");
      system("cp /tmp/temp.pcap.gz
         ".$outfilnamn.$testcount.".pcap.gz");
      system ("ssh $sctrl sudo killall tcpdump");
      system("ssh -f $sctrl sudo gzip -f /tmp/temp.pcap");
      system("ssh -f $sctrl cp /tmp/temp.pcap.gz
         ".$outfilnamn."srv".$testcount.".pcap.gz");

      $testcount++;
    }
   }
  }
 }
```

```
  ++$tcidx;
 }
}
```

Listing D.2: Configuration script

Execution script

```
#!/usr/local/bin/bash
# Usage: runtest.sh server serverport

# Clientside logging
sudo tcpdump -w /tmp/temp.pcap -i xl0 tcp port $2 &
sleep 1

# Start client
./tcp_client $1 $2 100 100 unused > tmp.tmp
sleep 2
sudo killall tcpdump

# Return elapsed time
cat tmp.tmp
```

Listing D.3: Experiment execution script

D.3 Client & Server applications

This section contains the source codes of the client & server applications. These applications are modified versions of the applications used in [11].

D.3.1 Client source code

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


int main(int argc, char **argv) {
  char              *remhost;
  char               read_buf[50000], c;
  u_short            remport;
  int                sock, nread, fd;
  int                len;
  int                buffer;
  long               sum = 0;
  long               tid;
  struct sockaddr_in remote;
  struct hostent *h;
struct timeval  tv1, tv2;



if (argc != 3) {
  fprintf(stderr, "usage: tcp_client <host> <port>\n");
  exit(1);
}

remhost = argv[1];
remport = atoi(argv[2]);

/* Create TCP socket */
if ((sock = socket(AF_INET, SOCK_STREAM, 0)) < 0) {
  perror("test_client:socket");
  exit(1);
}

/* Initialize sockaddr_in structure for remote host */
bzero((char *)&remote, sizeof(remote));
remote.sin_family = AF_INET;

if ((h = gethostbyname(remhost)) == NULL) {
  perror("test_client:gethostbyname");
  exit(1);
}
```

```
bcopy((char *)h->h_addr, (char *)&remote.sin_addr,
   h->h_length);
remote.sin_port = htons(remport);



/* Start the timer before connecting to remote host */
gettimeofday(&tv1, NULL);

if (connect(sock, (struct sockaddr *) & remote,
   sizeof(remote)) < 0) {
  perror("test_client:connect");
  exit(1);
}

/*
 * Read data from socket. Count number of bytes received and
     measure
 * the time
 */
while ((nread = read(sock, read_buf, sizeof(read_buf))) > 0) {
  sum += nread;
}

gettimeofday(&tv2, NULL);
tid = ((tv2.tv_sec - tv1.tv_sec) * 1000 + (tv2.tv_usec -
   tv1.tv_usec) / 1000);
close(sock);
```

```
  printf("%7ld\n", tid);
}
```

Listing D.4: Client application source code

D.3.2 Server source code

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


char send_buf[25000];

int main(int argc, char **argv) {
  int              listener, i, loops;
  u_short          myport;
  int              conn;
  struct sockaddr_in s1, s2;
  int              length;
  char             ch, tecken;
```

```
struct timeval  tv1, tv2;
long            tid;
long            bytes;

tecken = '0';
for (i = 0; i < sizeof(send_buf); i++) {
  send_buf[i] = tecken;
  if (tecken == '9')
    tecken = '0';
  else
    tecken++;
}

if (argc != 3) {
  fprintf(stderr, "usage: tcp_server <port> <nmb_loops>\n");
  exit(1);
}
if ((listener = socket(AF_INET, SOCK_STREAM, 0)) < 0) {
  perror("test_server:socket");
  exit(1);
}
myport = atoi(argv[1]);

bzero((char *)&s1, sizeof(s1));
s1.sin_family = AF_INET;
s1.sin_addr.s_addr = INADDR_ANY;
s1.sin_port = htons(myport);
```

```
if (bind(listener, (struct sockaddr *) & s1, sizeof(s1)) < 0) {
  perror("inet_rstream:bind");
  exit(1);
}
length = sizeof(s1);
if (getsockname(listener, (struct sockaddr *) & s1, &length) <
   0) {
  perror("test_server:getsockname");
  exit(1);
}
listen(listener, 1);
length = sizeof(s2);

while (1) {
  if ((conn = accept(listener, (struct sockaddr *) & s2,
     &length)) < 0) {
    perror("test_server:accept");
    exit(1);
  }
  printf("\nConnection established, sending data");
  loops = atoi(argv[2]);

  gettimeofday(&tv1, NULL);
  if (loops > 1) {
    write(conn, send_buf, 5000);
    loops -=2;
  }
```

```
    for (i = 0; i < loops; i++)
      write(conn, send_buf, 2500);
    gettimeofday(&tv2, NULL);
    tid = ((tv2.tv_sec - tv1.tv_sec) * 1000 + (tv2.tv_usec -
       tv1.tv_usec) / 1000);
    printf("\nBytes sent: %ld\n", bytes = (loops *
       (sizeof(send_buf))));
    printf("Elapsed time: %ld ms\n", tid);
    printf("Send rate: %lf bps\n", (bytes * 8 * 1000.0) / tid);
    fflush(stdout);
    close(conn);
  }
}
```

Listing D.5: Server application source code