Improved Algorithms for TCP Congestion Control

by

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Abstract

Reliable and efficient data transfer on the Internet is an important issue. Since late 70's the protocol responsible for that has been the *de facto* standard TCP, which has proven to be successful through out the years, its self-managed congestion control algorithms have retained the stability of the Internet for decades. However, the variety of existing new technologies such as high-speed networks (e.g. fibre optics) with high-speed long-delay set-up (e.g. cross-Atlantic links) and wireless technologies have posed lots of challenges to TCP congestion control algorithms. The congestion control research community proposed solutions to most of these challenges. This dissertation adds to the existing work by: firstly tackling the high-speed long-delay problem of TCP, we propose enhancements to one of the existing TCP variants (part of Linux kernel stack). We then propose our own variant: TCP-Gentle. Secondly, tackling the challenge of differentiating the wireless loss from congestive loss in a passive way and we propose a novel loss differentiation algorithm which quantifies the noise in packet inter arrival times and use this information together with the span (ratio of maximum to minimum packet inter arrival times) to adapt the multiplicative decrease factor according to a predefined logical formula. Finally, extending the well-known drift model of TCP to account for wireless loss and some hypothetical cases (e.g. variable multiplicative decrease), we have undertaken stability analysis for the new version of the model.

Keywords: Congestion Control, Congestion Avoidance, TCP Algorithms, End-to-End Flow Control, High-Speed Networks, Packet Loss Differentiation.

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I would like to thank Dr Guan, for supporting me by buying the books that I needed for my research, for her feedback on my research and reviewing the research papers that I have contributed to. I would also like to thank Dr George Oikonomou for his technical support and assistance in LAB experiments and also in rereading some of the research papers that I have contributed to.

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I would also like to thank Loughborough University in general and the Department of Computer Science in particular for funding my PhD studies, without this support, this work could not have been done.

This work is dedicated to my parents who have made the impossible for me from the day that I was born up to this moment and for all the loyal people in this world.

Talal A.Edwan

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Listings

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List of Abbreviations

AIAD	Additive Increase Additive Decrease
AIMD	Additive Increase Multiplicative Decrease
AQM	Active Queue Management
ARQ	Automated Repeat Request
ATM	Asynchronous Transfer Mode
BDP	Bandwidth Delay Product
CA	Congestion Avoidance
CC	Congestion Control
DCCP	Datagram Congestion Control Protocol
ECN	Explicit Congestion Notification
ELN	Explicit Loss Notification
FEC	Forward Error Correction
IEEE	Institute of Electrical and Electronics Engineers
IETF	Internet Engineering Task Force
IIAD	Inverse Increase Additive Decrease
IP	Internet Protocol
ISDN	Integrated Service Digital Network
ITU	International Communication Union
LDA	Loss Differentiation Algorithms
MIAD	Multiplicative Increase Additive Decrease

MIMD	Multiplicative Increase Multiplicative Decrease
PIT	Packet Inter arrival Times
QCN	Quantised Congestion Notification
RFC	Request For Comment
SACK	Selective ACK
SCTP	Stream Control Transmission Protocol
TCP	Transmission Control Protocol
TFRC	TCP Friendly Rate Control
VSAT	Very Small Aperture Terminal
XCP	Explicit Control Protocol

Chapter 1

Introduction

1.1 Motivation

TCP/IP is the name given to a protocol suite which provides the mechanism for implementing the Internet. It consists of dozens of different protocols but only a few protocols define the core operation of the suite. Of these key protocols, two are usually considered important: The Internet Protocol (IP) a primary OSI network layer protocol which provides addressing, datagram routing and other functions in an internetwork. The Transmission Control Protocol (TCP) a primary transport layer protocol which is responsible for connection establishment, management and reliable data transport between various software processes.

The protocol suite has over the years continued to evolve to meet the needs of the Internet and other smaller networks that uses the protocol suite. As part of this, testing and development of TCP has been a continuing process since 1973. For example, in October 1986 the Internet had the first of what became a series of congestion collapses. The data rate from Lawrence Berkeley Laboratories to 400 yards away UC Berkeley, dropped from 32Kbps to 40bps. The problem was solved at that time (1988) by adding two algorithms (slow start and congestion avoidance) to control the transmission when a congestion is detected. Now, the heavily patched old protocol is facing more and more challenges imposed by new technologies such as those in wireless, mobile and high-speed long-delay networks. Recently, some of these algorithms have shown problems when working over some underlying technologies, one example is the problem of congestion avoidance algorithm not being able to efficiently utilise the capacity of long-delay high-speed pipe. Another example is packet[1] loss caused by link errors which disturb the operation of congestion avoidance that relies on this information to detect congestion.

[1]Packets and segments are used interchangeably in this thesis

Since the TCP protocol is still in use, all these problems, and others have formed a motivation for the networking research community and for me as part of this community. From the beginning of 90's until now, we have witnessed a series of algorithms/changes/ideas which address the aforementioned problems, despite that, there is no one panacea for all problems, in fact this has turned the problem one of a compromise and trade-offs. For example, it is challenging to have an algorithm that can efficiently utilise a high-speed long-delay pipe, have a high responsiveness to network changes, fast convergence and at the same time be fair to other flows and immune to link errors. This by itself has posed a challenge and formed another motivation for me to study the subject and contribute to it.

1.2 Objectives

1. *Suggest Improvements to TCP Congestion Control (CC) Algorithms*: the fruit of the work in this thesis is to contribute to existing TCP congestion control algorithms through a cycle of study, solutions suggestions and systematic evaluation.

2. *Suggest a solution(s) to increase TCP CC algorithm's immunity against error prone links*: develop a sender side passive technique to increase the immunity of TCP CC algorithms against packet losses that are not caused by congestion; and extend TCP model(s) to account for such losses wherever possible.

1.3 Contributions

This dissertation has the following contributions:

1. We approached the end-to-end Internet congestion control from a theoretical perspective, represented by the optimisation framework, and from practical perspective, represented by the implementation of congestion control in the Internet via the well known transport layer protocol: TCP. We provided a detailed discussion from the two perspectives.

2. A key element in the performance evaluation of TCP congestion control is to have clear and well defined metrics. There have been many definitions of metrics in the literature. We classified congestion control metrics, provided a detailed definitions.

3. We proposed an improvement to one of TCP congestion control variants, TCP-Illinois [69], which we call *TCP Illinois*^{n}. A conference paper written

by the author and a number of colleagues in the Computer Science Department at Loughborough University gives detailed description of this research work. The author has been the main contributor to the paper. The author conducted all comparative analysis experiments using `ns2` based on the congestion control metrics mentioned in this dissertation. The author also implemented the new modifications in the relevant `TCP/Linux` module for `ns2`. The new modifications in the form of patches (for both `ns2` and the Linux kernel) along with an extensive list of post simulation scripts written by the author to manipulate metrics computation are publicly available.

4. We proposed a new TCP congestion control algorithm [2], which we call *TCP Gentle*, the proposal is an incremental development based on the latest proposal [9] in Linux kernel up to the date of writing this dissertation. The author is the inventor of the new ideas of this TCP congestion control algorithm, the author implemented the algorithm both in `ns2` and Linux kernel, the source code is publicly available. The author also conducted all simulation experiments and real test bed experiments at the laboratories of Loughborough University. The author is the main contributor of a ready to submit research paper [3] which gives details of this research work.

5. One of the challenges for TCP congestion control has been non-congestive loss, which has great impact on the throughput. We proposed an idea to discriminate congestive loss from non-congestive loss, in what is known in literature as: *Loss Differentiation Algorithms (LDAs)*, the idea is a novel algorithm which tries to differentiate between the two different types of loss, mainly based on the *noise* in packet inter-arrival times. A conference paper written by the author and a number of colleagues in the Computer Science Department at Loughborough University gives detailed description of this research work. The author has been the main contributor to the paper.

6. We extended the TCP mathematical model [88] by including non-congestive packet loss and variable multiplicative decrease parameters. We linearised the model, applied *Laplace Transforms* and analysed the stability. We have classified non-congestive loss as *disturbance* in the context of control theory, contrary to congestive loss, which is within TCP's control. We have shown in the same context that the stability range for a variable multiplicative decrease is larger than traditional fixed value for TCP. A workshop paper written by the author and a number of colleagues in the Department of

[2] A history time line is available in the Appendix.
[3] TCP-Gentle: An "Accordion-Bellows" Congestion Window for YeAH-TCP

Computer Science at Loughborough University gives details of the analysis. The author has been the main contributor to the paper. An extended version of the work will appear in a Journal paper.

1.4 Structure of the Thesis

The structure of the dissertation is as follows: chapter 2 gives a background of the congestion problem in computer networks and different source behaviours that act on this problem. This is followed by background of TCP protocol and two broad areas of challenges: Wireless networks and High-Speed networks, followed by an examples of existing approaches to alleviate some of the challenges. Chapter 3 elaborates on current proposals for solving specific problems in these two broad areas, mainly the High-Speed Long-Delay problem. Chapter 4 describes the congestion problem in computer networks from an optimisation perspective, and highlights our modification/analysis of the TCP drift model. Chapter 5 classifies and defines the metrics used in performance evaluation of congestion control algorithms. Chapter 6 describes one of our proposals to enhance one of the high-speed long-delay TCP variants. Chapter 7 describes our new algorithm: TCP-Gentle which is another high-speed long-delay TCP variant. Chapter 8 describes an algorithm which we have developed to differentiate the wireless packet loss from congestive loss. Chapter 6 - 8, show our contribution in the two broad areas mentioned at the beginning of the section. Finally, chapter 9 concludes our work and gives potential research directions for future.

Chapter 2

Background

This chapter gives the reader a background of Internet congestion problem, end-to-end solutions to it, currently most deployed standard end-to-end protocols and some of their challenges. The chapter has a historical flavour to provide a "bottom-up" development of the subject and is divided as follows: in section 2.1 we discuss congestion control definitions and terminologies, in sections 2.2 - 2.4 we focus on source behaviour in end-to-end congestion control. We then look at TCP protocol in section 2.5 and its congestion control in section 2.6 and section 2.7. We discuss some challenges to TCP congestion control in section 2.8 and section 2.9. In section 2.10 we discuss a general approach that can be used to solve TCP performance problems followed by network assistance for TCP in section 2.11. Finally, we summarise the main ideas in section 2.12.

2.1 Congestion Phenomenon

The Internet is a best effort service, this implies that packets[1] send across the network might reach the other end quickly, slowly or never make it. There are several reasons for this: unavailable links and packets re-routing, environmental issues like the effect of wireless links and network congestion.

Network congestion degrades the quality of this best effort service and the treatment of this problem should start by understanding the root cause of it and the result of it. Congestion occurs when resource demands exceed the capacity [100]. Another simplified definition from a user perspective [62]: "A network is said to be congested from the perspective of a user if the service quality noticed by the user decreases because of an increase in network load". It is a phenomenon that is tightly related to the pattern of users usage of the network." It is also

[1]The term packet was first coined in 1967 by Donald Watts Davies at (NPL) National Physical Laboratory in Middlesex, England. Another fancy name for it is (PDU) Protocol Data Unit [90, p.362]

related to link capacities and topological issues, for example congestion can occur when traffic arrives on a high capacity link and gets sent out on a low capacity link, or when multiple input flows arrive at a node whose output capacity is less than the sum of the inputs [90]. The result in this case is an excess packets or traffic spikes. The node has two choices to deal with these packets: either drop them or buffer them. Which one to choose is not really an easy task . Typically routers are designed to buffer such spikes, based on the assumption that they last for short time and thus the router acts as an ample device that absorbs these spikes, however the choice of the optimal buffer size is puzzling: big buffers can handle traffic spikes, reduce packet loss; but at the same time increase the delay, cause TCP time-outs, and might rise a feasibility issue, however; some see that in general, queues should be kept short [100], and short queues lead to short delay and high throughput [45] for an ACK-based protocol (in the sense that large queues lead to large round trip time which increse the delay of returned ACKs and in in turn reduce the growth of the congestion window i.e. forward path rate.)

Now, this is the cause and result of congestion. There are three ways to treat it: congestion avoidance , congestion control, and/or over provision the resources. Before discussing the three ways, let us define some terms: solutions to congestion can be grouped into two classes [94]: open-loop and closed-loop. An open-loop solution is a preventive solution, prevention policies can be used in data link, network and transport layers to prevent congestion from happening, examples of such policies are: retransmission, out-of-order caching, acknowledgement, packet queueing and service, packet discard, packet lifetime, routing algorithm, time-out determination, etc. Resource reservation in some connection-oriented protocols is an example of open-loop solution, however, one problem that might arise is bandwidth under-utilisation. A closed-loop solution, treats congestion after it happens or just before it happens, therefore it is more difficult to tackle.

Figure 2.1, shows the big picture of a closed-loop solution:[2]

1. This is the congestion control loop.

   (a) If the detector is proactive, this means it prevents congestion before it happens and the process is referred to as congestion avoidance. If the source behaviour is conservative the process is also called congestion avoidance.

   (b) By network we mean multiple nodes between sender and receiver and the following applies :

   * Fairness issues, since the path is likely to be used by other sources.

[2]This is based on my conclusions

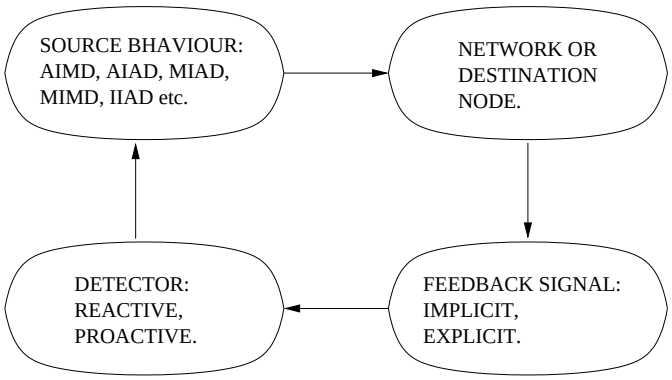


Figure 2.1: Bird-eye view of congestion control loop.

* Intermediate nodes and end node can generate feedback signals.

(c) If there are no nodes between sender and receiver the following applies :

* Destination node can generate feedback signals.

* The process is called flow control.

2. Congestion Control: Note the word "control": it is not only prevention, but also utilising the capacity of the network fairly and efficiently. (not exceeding and not underutilising the capacity). These days, networks are often over provisioned, and the underlying question has shifted from "How to eliminate congestion" to "How to efficiently use all the available capacity". Efficiently using the network means answering both of these questions at the same time; this is what good congestion control mechanisms do.

3. Congestion Avoidance & Congestion Control: Parallel to the above discussion, a graphical illustration in figure 2.2 aims to help in distinguishing between congestion control (sometimes referred to as recovery [45]) and congestion avoidance. Congestion control's goal is to stay left of *Cliff*[3] while congestion avoidance's goal is to stay left of *Knee*.

4. Resource Over provision: This is basically increasing the capacity of the network. In these days, congestion has, in general, moved into the access links (at the edges of the network, not at the core). The reasons for this are of a purely financial nature [100]:

(a) Cheap Bandwidth. It pays off to over provision a network if the excess bandwidth costs significantly less money than that an Internet Service Provider (ISP) could expect to lose in case a customer complains.

[3]The name is inspired from the fact that after exceeding a cliff there is collapse

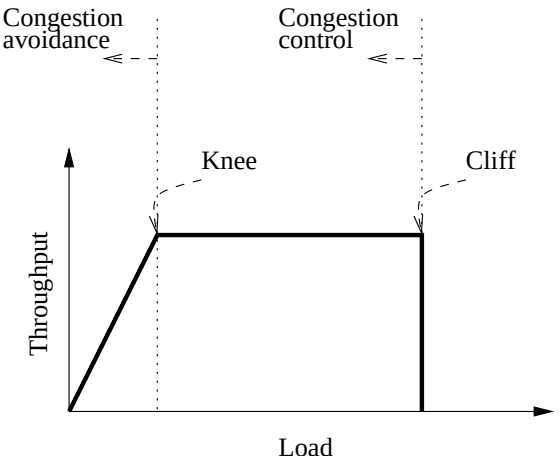


Figure 2.2: Basic definitions of congestion avoidance and congestion control as appeared in Chui-Jain paper in 1989.

(b) It is more difficult to control a network that has just enough bandwidth than an over provisioned one, the former needs skilled network administrators which demands additional costs for training.

(c) There is an increased risk of network failures, which once again leads to customer complaints.

(d) Scalability for future.

As a historical note, access speeds were higher than the core capacity in the late 1970s, but changed in the 1980s, when ISDN (56 kbps) technology came about and the core was often based upon a 2 Mbps Frame Relay network. Where in the 1990s ATM , with 622 Mbps along with 100 Mbps Ethernet connections were dominant. And nowadays high-speed networks (Gigbit, optical fibber, some wireless technologies) are widely deployed. So the development of new technologies can be considered as another encouraging factor for the over provisioning choice.

Having said that, we focus our attention on the source behaviour in the congestion control loop (upper left block in figure 2.1) for two reasons: i) We believe that a source-based solution can be easily deployed compared to a network-based (or hybrid-based) solution e.g. a patch as part of an upgrade to an operating system can be easily distributed among the hosts running a protocol that has a congestion control algorithm. This can be easier than providing a solution that requires changes in Internet routers. ii) A source-based solution treats the network as a black-box, . If the solution is a rigorously defined congestion control algorithm, changing the network equipment e.g. wired router to wireless router, or a router with high speed capabilities or even using a completely congestion-unaware router will not prevent control of congestion.

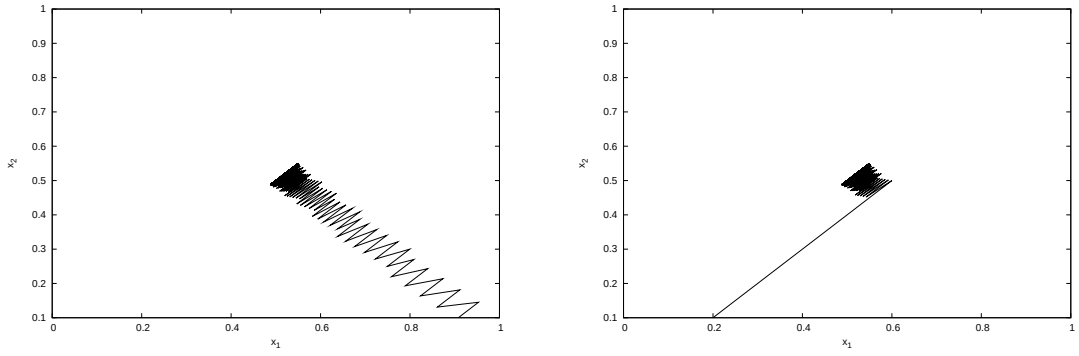
2.2 Linear Algorithms

As it can be seen from figure 2.1, whether targeting CA or CC; a source can take an action (this is basically increase or decrease rate) based on feedback from network/destination. Nowadays, since the capacity of links has significantly increased; efficient use of links has become a critical issue i.e. working between the "Knee" and "Cliff", therefore targeting CC. Having said that, there are many issues to consider in addition to the efficient use of capacity, like fairness among flows, response to changes, convergence, magnitude of rate oscillation etc. Such issues influence the increase/decrease laws of a CC algorithms.

Considering source behaviour, one famous class of CC algorithms is linear algorithms. They are given this name because they have one algebraic term involved in the increase/decrease rule. To further illustrate this point we list the rules of four types of this class, the source rate at time t is denoted by $x(t)$:

$$\begin{aligned} AIMD: &\quad x(t+1) = x(t) + \alpha I_{\leq c} - x(t)\beta I_{>c} \\ AIAD: &\quad x(t+1) = x(t) + \alpha I_{\leq c} - \beta I_{>c} \\ MIMD: &\quad x(t+1) = x(t) + x(t)\alpha I_{\leq c} - x(t)\beta I_{>c} \\ MIAD: &\quad x(t+1) = x(t) + x(t)\alpha I_{\leq c} - \beta I_{>c} \end{aligned}$$

$I_{\leq c}$ is an indicator of whether or not the source rate has increased beyond capacity, in other words, $I_{\leq c} = 1$ when link capacity is not exceeded, $I_{\leq c} = 0$ otherwise. And $I_{>c} = 1$ when link is exceeded, $I_{>c} = 0$ otherwise. Here if α and β are constants then $x(t)$ is the only algebraic term. Despite its 'classic' [31] content; Chui-Jain's



(a) x1=0.9, x2=0.1 (b) x1=0.2, x2=0.1

Figure 2.3: Vector diagram: two AIMD sources with different initial rates

work [24] highlights the dynamics of linear algorithms. To review these dynamics we ran a test for discrete forms of linear algorithms. Figure 2.3 and figure 2.4 show a vector diagram plot of two sources. It has been shown [24] that an AIMD converges to fair point in an environment with synchronised congestion events (i.e. all losses happen at the same time for different flows), this can be seen from

figure 2.3. A well-known but interesting point is that, MIMD and AIAD do not show same properties: both converges to a non-fair points and MIAD oscillates of from the optimal share point i.e. one source will take all the capacity and deprive the other source from bandwidth: this is clearly illustrated in figure 2.4.

However, others argued that these results do not apply to asynchronous environments (i.e. losses happen at different frequencies for different flows), like the Internet for example [31]. Their argument can be supported by the fact that MIMD converges to fairness in a model with proportional instead of synchronous packet loss [50]. Nevertheless, the AIMD characteristic of convergence has made it a favourable choice among other types. Next, we briefly discuss a super-class of CC algorithms, of who linear algorithms are a sub-class.

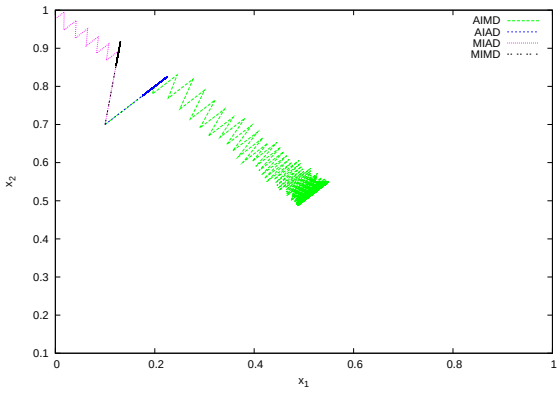
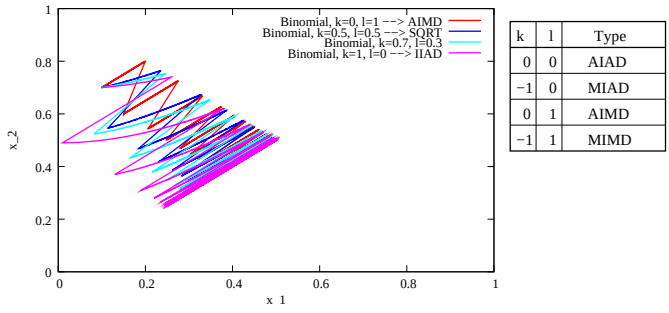


Figure 2.4: Convergence properties of linear algorithms, initial rates: x1=0.1, x2=0.7

2.3 Non-Linear Algorithms



k	l	Type
0	0	AIAD
−1	0	MIAD
0	1	AIMD
−1	1	MIMD

Figure 2.5: Convergence properties of binomial algorithms, initial rates: x1=0.1, x2=0.7

Because of it's convergence and fairness merits; the AIMD principle has been adopted for developing safe and stable CC algorithms for the Internet (since 1987). However the fixed increase and decrease parameters of an AIMD used by a CC

algorithm have reduced its flexibility. For example, considering a total deployment of CC in the Internet. Streaming audio and video applications do not react well to abrupt rate reductions because of the degradation of user-perceived quality. This has formed a motivation to find (let me call it) a super-principle that extends the features of AIMD.

Binomial CC algorithms [11] was first introduced in 2001 as a non-linear generalisation of linear CC algorithms and subsequently a generalisation of AIMD. The general rule for increase and decrease is:

$$x(t+1) \quad = \quad x(t) + \frac{\alpha}{x^k(t)} I_{\leq c} - x^l(t)\beta I_{>c}$$

Where, $\alpha > 0$, $0 < \beta < 1$. They are called binomial because they have two algebraic terms involved in their increase/decrease rule, x and x^k in the increase rule. And x and x^l in the decrease rule.

Obviously, linear algorithms rules are special cases of this rule. Also, smaller k results in large aggressiveness[4] and smaller l results in smaller reductions in the rate when congestion is experienced. There is a trade-off between k and l, for example; a choice of large l result in higher reduction which necessitate a small k to substantially increase the rate after this large reduction. There is also a rule that restricts the choices in order to maintain friendliness to existing standard protocols. However, the key point here is the advantage of this general rule; which is the many choices (flexibility). For example; for $k = 1, l = 0$ we obtain a rule that is less aggressive than AIMD and has fixed MD, and a choice of $k = 0.5, 0.5$ lies between the two. Figure 2.5 shows a vector diagram plot of two sources for a set of choices of k and l obtained by discrete version of the algorithm[5]. We note that all choices converge to a fair point.

2.4 Equation-Based Algorithms

A different approach for adapting the source rate is by the adherence to a reference equation. An equation giving the average throughput as a function of packet loss rate and round trip time could be used calculate the rate which the source rate can be adapted to. By adapt we mean: if actual rate is above the calculated rate; the actual rate is reduced and if the actual rate is below the calculated rate; the actual rate is increased. The advantage of such an approach is that the algorithm using this equation will be fair to any other algorithm using or working according to it. A well known example of this approach is the equation-based TFRC mechanism [37]

[4]This is defined in chapter 5.
[5]Written by the author

which uses TCP's equation [78].

Since the values used in the equation are typical average values, it is unlikely to have abrupt increase/decrease in the rate i.e. low level of oscillation. In fact this merit makes this approach an alternative to binomial approach when considering streaming audio/video applications, however as we will see in chapter 5; there is always a trade-off for each advantage. The penalty for the smoothness in rate is slow responsiveness to change in available bandwidth.

Theoretical formulation and analysis are important when studying CC, however the practical side has the final word, especially when considering a complex environment like the Internet. Practically speaking the aforementioned ideas can be embedded in any protocol that needs to have a CC functionality, however; since the Internet is controlled by standards shepherd by bodies like IETF , IEEE , ITU , it is more interesting to see what algorithms are implemented and where they are implemented in standard protocols. This leads us to the topic of the following section.

2.5 TCP Protocol

Regardless of telecommunication infrastructure, most traffic in the Internet is controlled by the transport layer protocol, TCP. It is the de facto standard for reliable protocols and the most dominant control protocol in the Internet. A relatively recent study conducted during 1998-2003 on one of the few sources publicly available: the NLANR PMA (National Laboratory for Applied Network Research - Passive Measurement and Analysis Project)[6] showed that TCP traffic percentages of total bytes, packets and flows respectively were: 72%-94%, 63%-87% and 41%-71% [43]. It is worth to mention that although TCP is the dominant control protocol; it is not the only standard protocol that uses CC. Other protocols such as SCTP borrows TCP CC. Another example is DCCP which provides a modular CC mechanisms [65]. Two mechanisms are available: TCP-like CC [40] and TFRC [41].

TCP and other Internet protocols are specified in RFCs. There are currently 5841 RFCs since 1969[7]. The protocol specification was documented in RFC-793 [80] in 1981 based on the original paper of TCP written in 1974 (where TCP stands for Transmission Control Program at that time) by two scientists (Prof. Vint Cerf and Prof. Robert Kahn). However there are a number of TCP-related

[6]The Passive Measurement and Analysis (PMA) Project is one of two research projects that form the core of the NLANR Measurement and Network Analysis Group's Network Analysis Infrastructure (NAI). The other is the Active Measurement Project (AMP)

[7]RFC-5841: TCP Option to Denote Packet Mood. R. Hay, W. Turkal. April 1 2010

RFCs [18, 71, 5, 42, 81, 38].

TCP is a connection-oriented protocol, i.e. initially any two communicating hosts should establish a connection in what is generally known as three-way handshaking. The protocol was designed to provide a reliable service to higher applications when they connect over a network, this entails; application layer data being broken to best size segments, each segment has sequence number by which the receiving host can resequence any out of order segments. Received segments are acknowledged by sending ACK segments back to sender. A window based mechanism was adopted to aid flow and congestion control.

For each sent segment; TCP maintains a timer waiting for an ACK for the received segment. If an ACK is not received (timeout) the segment is retransmitted, this is timer-driven recovery mechanism. There is another data-driven mechanism called Fast Retransmit [89], in this mechanism a three duplicate ACKS trigger the retransmission of what is believed to be a lost segment, it was assumed that one or two duplicate ACKs are usually caused by out of order (or duplicate) received segments so TCP waits for a third one to make the decision. Another relatively recent loss recovery mechanism is SACK [71]: it can efficiently recover from multiple losses per window in one round trip time. The idea is that the receiver can inform the sender about all segments that have arrived successfully, so the sender need retransmit only the segments that have actually been lost. For this to take place, two options need to be enabled in the TCP header: SACK-permitted (in the SYN segments) and SACK option. In case of segments loss; the receiver specify non-contiguous blocks (usually three) of data in the SACK options and send it back to the sender in the ACK segments.

In the context of SACK discussion, it is worthwhile to mention an extension to SACK called D-SACK [42] which aims to help the sender to distinguish whether the duplicate ACKS were generated due to a lost segment or a duplicate segments. The problem addressed here is when the threshold of three segments are not enough to determine lost segments (it has been highlighted [85, p.21] that this is not uncommon), the D-SACK can be used to resolve this ambiguity. It does so as follows: when duplicate segments are received, the first block of the SACK option field is used to report the sequence numbers of the segment that triggered the ACK and thus the sender can determine if the cause of duplicate ACK is duplicate segments (same sequence number) or not.

In the scope of this thesis, these are the protocol ideas needed to familiarise the reader with TCP, however the protocol is complex and there are lots of literature covering its details.

2.6 TCP Congestion Control

Since 1988, the use of congestion avoidance algorithm in TCP is mandated [18]:

> "Recent work by Jacobson [TCP:7] on Internet congestion and TCP retransmission stability has produced a transmission algorithm combining *slow start* with *congestion avoidance*. A TCP MUST implement this algorithm."

We also came across the same point during implementation of CC modules in GNU/Linux kernel, where the use of two functions: `ssthresh()` and `cong_avoid()` was mandated for standard purposes.

The window based mechanism of TCP can be summarised as follows: maintain two windows, a congestion window which represents flow control imposed by the sender, based on the sender's assessment of perceived network congestion and an advertised window which is flow control imposed by the receiver, related to the amount of available buffer space at the receiver for the connection, then TCP sends the minimum of both windows. The way the congestion window is adapted critically affects both the connection and the network. The standard TCP CC uses a slow start mechanism at the beginning of a connection (or after time-out) and an AIMD mechanism with $\alpha = 1$ and $\beta = 0.5$ in congestion avoidance (after slow start threshold is reached) and keeps using this mechanism after packet loss (not time-out), these algorithms are sometimes referred to as Jacobson's algorithms [96] figure 2.6 illustrates the congestion window evolution of standard TCP CC.

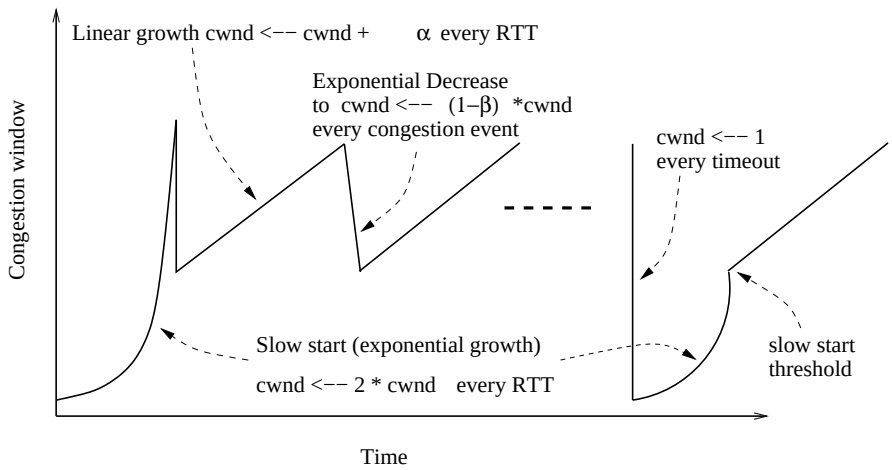


Figure 2.6: Standard TCP congestion window evolution

$$\begin{aligned} \text{Ack}: &\quad cwnd \leftarrow cwnd + \frac{\alpha}{cwnd} \\ \text{Loss}: &\quad cwnd \leftarrow cwnd - \beta \times cwnd \end{aligned}$$

Where, $\alpha = 1$ $\beta = 0.5$. This approach was sufficient to solve the congestion problem at that time, but this did not prevent the research community from questioning some issues, for example, whether the oscillatory behaviour can be reduced while maintaining the fairness merits of AIMD. One such interesting attempt appeared in 1991 [98] trying to minimise the oscillation of Jacopson's algorithms through the use of the so-called *Normalised Throughput Gradient (NTG)*, the basic idea is to adapt the congestion window based on the change in throughput. In the initial mode, the algorithm increases exponentially, after a time-out it enters a decrease mode and starts from one packet (i.e. the unit of adjustment), but this time for each received ACK it checks the NTG, if a threshold is exceeded it increases exponentially otherwise it enters an increase mode where it increases linearly and checks the NTG each round trip time, if NTG is below another threshold it reduces by one packet (i.e. the unit of adjustment) otherwise it keeps the window unchanged.

The algorithm [98] fits the network speeds of that time, however in today's networks (e.g. large bandwidth delay product pipes) the trade-off between responsiveness and smoothness becomes more critical, for example the additive decrease may not be sufficient to decrease quickly when a sudden increase in traffic occurs. Another issue is that the thresholds need to be selected carefully, for example one choice may underestimate the link capacity, while another may delay the action of exponential increase when traffic load decreases, i.e. the change in NTG has to exceed the threshold. There are other issues like for example when competing with other greedy (Jacobson's algorithms) flows this approach gives the rate to the competing flows. Finally, reverse path bottlenecks can also force this approach to stop increasing its rate, thus underutilise its uncongested forward path. As we will see in chapter 7, one of our proposals overcome the responsiveness issue by reacting almost immediately (in one round trip time) once an empty queue is detected.

2.7 Multipath TCP Congestion Control

The basic idea of multipath congestion control is to let a multipath-capable flow shift its traffic from a congested path to an uncongested path, by doing so the packet drop rates due to congestion on the congestion path is reduced while that on the uncongested path is increased,this attempts to balance the load on the network by making the network perform *as if* its resources are grouped and shared among the flows. This last idea is referred to as *Resource Pooling* [26].

Considering multipath TCP congestion control, algorithms [25] have been proposed in order to achieve this form of load balancing. The basic idea of AIMD

was used, however for the case when the AI and MD parameters of flow are selected based on the sum of all congestion windows on different paths, a flappiness between paths were reported i.e. the flow spends more time on one path than the other, then it flips to the other path when packet drop rates on the current path increases. On the other hand when selecting the AI and MD separately based on the individual congestion window. The flappiness disappear but the main objective of load balancing (or resource pooling) is not achieved. Obviously, a trade off appears between the two mechanisms. A compromise solution is to select the AI parameter based on the sum and the MD based on the individual. However the choice of AI parameter has to be carefully selected.

2.8 TCP & Wireless Networks

Wireless networks impose several challenges on TCP performance, in this section we summarise some of the common challenges.

One challenge is link errors in wireless networks, this is usually blamed for unnecessary throughput reduction. Other problems are: packet reordering, effect of asymmetric paths (bandwidth, loss-rate, latency, etc) and low congestion window due to low link speeds in some wireless systems [10] (like cellular systems for example). It has been shown that wireless loss and packet re-ordering are not uncommon [56], and they are caused by inherited properties in the wireless technology, e.g. signal fading, hand-off and mobility.

The Internet approach usually deals with error control at higher end-to-end layers [51], [83]. Applications have different degrees of reliability, some could be error intolerant (cannot rely on link layer error recovery, they need more reliability) others could be error tolerant. The end-to-end approach gives more flexibility to applications to accept or refuse the error recovery overhead. However this does not eliminate the need for link layer error recovery in case of high error rate links. Here, lower layers error recovery can be fast and more adaptable. But we should note that they are not perfect solutions, they usually recover from error using FEC, or ARQ mechanisms. FEC may be unable to correct too many bits if the frame error rate is high, and with ARQ some of the protocols at link layer trades off reliability for delay variance, frames not received after few retransmissions are dropped, higher layer protocols can provide additional recovery, if needed.

One point to note here is that, applications and Internet protocols which implement there own error recovery schemes may interact adversely with link layer mechanisms [51]. The question that rise here is, where and how to mitigate the problems of high error rates? This is part of the system design problem, and depends on system's usage requirements, for instance, in cellular systems consid-

erable processing is required in order to reduce the high error rate of the link, leading to significant processing delays which in turn needs to be considered as part of the design. On the other hand, in WLAN systems, where the error rate is lower, error recovery is usually left to higher protocol layers [51], e.g. TCP-SACK, this reduces the dependency on lower layers, this also should be considered in the design.

Many attempts were made to solve the problem at higher layers, mainly the transport layer (we refer the reader to the history time line in the Appendix) it is beyond the scope of this thesis to provide a complete discussion of TCP performance in wireless networks, but rather to summarise the main challenges and to focus on certain problems and provide a solution for them. For more details the reader can refer to [12], [51].

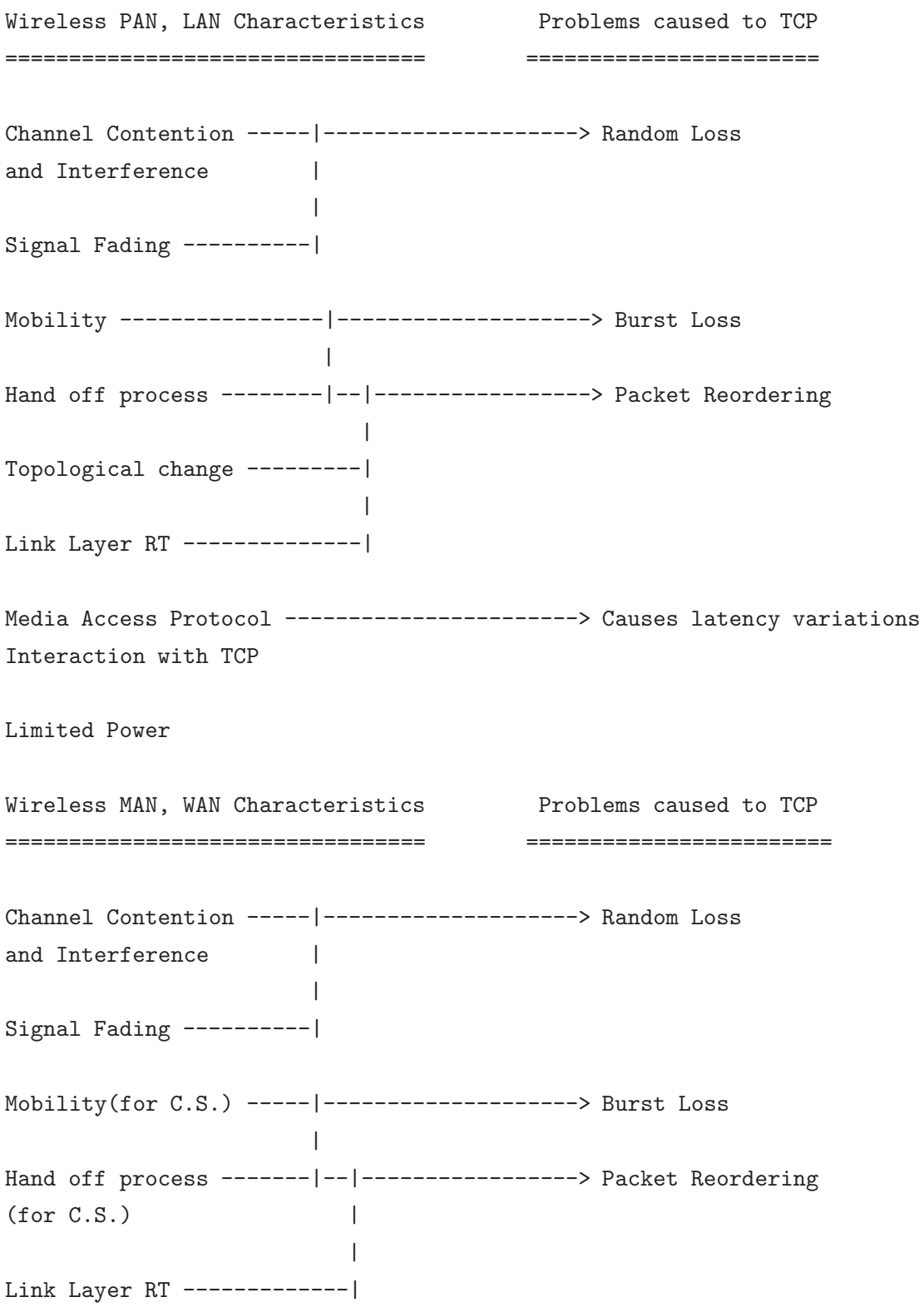
To see how these problems affect TCP performance, we complied a list of different wireless technologies and their potential problems to TCP. Below is a list of the most common standards in wireless technologies, followed by a list of the most common problems caused to TCP when it works over wireless networks:

```
Types of Wireless Networks           Standards
==========================         ===========
PAN System interconnection   ----->|-IEEE 802.15
                                    |
Wireless LANs                       |-IEEE 802.11
[Infrastructure and Ad-Hoc]

Wireless WANs                ----->|-IEEE 802.16
[Infrastructure]                    |
[High speed and Low speed]          |-Cellular Systems
[Satellite Communications]            2.5G, EDGE+GPRS
                                      3G, ITU IMT
                                      W-CDMA+UMTS
```

- Standards:

- IEEE 802.11 Wireless LAN & Mesh (Wi-Fi certification), (Wireless Local area network-WLAN)
- IEEE 802.15 Wireless PAN, (Wireless Personal area network-WPAN)
- IEEE 802.15.1 (Bluetooth certification)
- IEEE 802.15.4 (ZigBee and Mi-Wi certification)
- IEEE 802.16 Broadband Wireless Access, (WiMAX certification), (Wireless Metropolitan area network-WMAN), IEEE 802.16e (Mobile) Broadband Wireless Access, (M-WiMAX).
- IEEE 802.18 Radio Regulatory TAG
- IEEE 802.19 Coexistence TAG
- IEEE 802.20 Mobile Broadband Wireless Access, (Wireless Mobility)
- IEEE 802.21 Media Independent Hand off (Hand-off/Interoperability Between Networks)
- IEEE 802.22 Wireless Regional Area Network


Wireless PAN, LAN Characteristics
Problems caused to TCP
Channel Contention and Interference
Random Loss
Signal Fading
Mobility
Burst Loss
Hand off process
Packet Reordering
Topological change
Link Layer RT
Media Access Protocol
Causes latency variations
Interaction with TCP
Limited Power
Wireless MAN, WAN Characteristics
Problems caused to TCP
Channel Contention and Interference
Random Loss
Signal Fading
Mobility(for C.S.)
Burst Loss
Hand off process (for C.S.)
Packet Reordering
Link Layer RT
Low speeds (for C.S.)
Small Congestion Window
Affect Data-driven loss recovery + Increase time-outs.
Media Access Protocol
Causes latency variations

Interaction with TCP

Asymmetric paths ----------> Affects Reverse Path ACK
(bandwidth, loss-rate,latency) feedback Affects Forward
performance.

Limited Power (for C.S. and Satellites)

- Link Layer behaviour can also increase number of TCP time-outs and retransmissions.
- 802.16 family of standards is also called Wireless MAN, Broadband Wireless and WiMAX.
- Satellites can also be considered as mobile devices because they move relative to earth.
  and the whole system is designed to give full coverage, this reduces the effect of hand off.
- It is worth to note that FEC (Forward Error Correction) techniques like Hamming codes are used in the physical layer in addition to check sums in upper layers in the case of broadband wireless, because so many transmission errors are expected in this case, this can add to latency variations.
- Hand off process can increase latency.
- Too many retransmissions affects the limited power.

Table 2.1: TCP Throughput over LAN and WAN connections [51]

Network Type	Nominal Bandwidth	Actual TCP Throughput	Achieved (%)
LAN	1.5 Mbps	0.70 Mbps	46.66
WAN	1.35 Mbps	0.31 Mbps	22.96

Table 2.2: TCP Throughput over IEEE 802.11 connections [51]

Standard	Nominal Bandwidth	Actual TCP Throughput	Achieved (%)
802.11	2 Mbps	0.98 Mbps	49
802.11b	11 Mbps	4.3 Mbps	39.1

We focus our attention on the loss problem (which is due to high transmission error rates) and how it affects TCP performance. Having mentioned that lower layer error recovery cannot recover from all errors, this may lead to packets being corrupted and thus discarded and not handed to TCP. TCP in turn makes a tacit assumption that the lost packets are due to congestion and thus reacts by reducing its congestion window drastically (multiplicative decrease). This results in an unnecessary *throughput* reduction. We adopt some figures mainly for WLANs [51] to show how severe the problem could be. Table 2.1 depicts TCP throughput over a WLAN path and a WAN path consisting of a single WLAN plus 15 wired links. The nominal bandwidth is the bandwidth in the absence of any losses, the actual throughput is the throughput when the WLAN suffers from a frame error rate of 2.3% for a frame size of 1400 bytes. Table 2.2 shows the results for IEEE 802.11. Note that high speed links are affected more since TCP drastically reduces its throughput after each loss event (multiplicative decrease) and thus it takes longer to reach the peak throughput supported by higher speeds [51]. Unnecessary TCP throughput loss also occurs in cellular systems. In their voice mode the residual frame error rate is 1-2% *after* low level error recovery. Another point to mention here is that TCP works in the forward and reverse directions (data and ACKs), in wireless links this could lead to undetected collisions which in turn increases the frame error rate.

The SACK mechanism in TCP helps in recovering from multiple packet loss quickly (i.e. in one round trip time), however, non-congestive packet loss is problematic for standard TCP CC because fast recovery is invoked and the congestion window is halved.

2.9 TCP & High-Speed Networks

2.9.1 Long Delay

The evolution of high-speed networks has facilitated the transfer of huge amounts of scientific data like that gathered in Astronomy, Bioinformatics, Earth Sciences, Physics etc. Nowadays, cross Atlantic data transfer between research institutes is not uncommon. Such transfers need a reliable and efficient protocol that can handle multiple simultaneous transfers and TCP is the most used reliable protocol.

TCP CC armed with its fixed AIMD mechanism may have a problem utilising the full bandwidth of a high-speed long-delay pipe (this is usually referred to as: high BDP). To see how TCP substantially underutilises network bandwidth over high BDP connections; let us consider this case: suppose we have TCP flow running over a 10 Gbps with RTT = 100ms, packet size = 1500 bytes = 12000 bits. This gives a BDP = $10^{10} \times 0.1 = 10^9$ bits, $= 83,333.33$ packets. If TCP is running in congestion avoidance phase, which means that the congestion window is increasing by 1 packet each RTT, upon a packet loss event the window is halved. So: $\Delta cwnd = (1/RTT)\Delta t$, $\Delta t = \Delta cwnd \times RTT = 416666.67RTT = 4166.67$ seconds $= 1.16$ hours to reach the peak i.e. full utilisation again. If packet size is 10000 bits, it takes ≈ 1.5 hours. In other words TCP needs a low packet loss rate to achieve full utilisation and such low loss rates are not realistic especially with the spread of new technologies like fibre optics and wireless networks, and even at these low loss rates, it takes too long to fully utilise the link after a back off.

It has been shown [88, p.72], that TCP is not stable for large RTT. Another way to look at this nonstability is by comparing the growth of the congestion window for two flows, one with large RTT and another with small RTT. Recall that the slope of a congestion window growth function over time is: α/RTT where $\alpha = 1$ for TCP. In fact, the flow with the large RTT (low slope) spends most of its time increasing from the point at which packet loss happened to the peak, during the same period; the flow with the small RTT (high slope) must have reached the peak several times (reached steady state faster).[8]

2.9.2 Short Delay

The high-speed long-delay pipes are not the only potential problem for TCP. A pathological behaviour of TCP in some high-speed low-delay environments has been also identified. The problem can be seen when TCP works in a certain communication pattern which is known as *Incast*. In this pattern a receiver issues a request to multiple senders, the senders upon receiving the request concurrently

[8] This is my understanding.

respond to the receiver. The sender traffic traverses a bottleneck link in a many-to-one fashion and a potential for a congestion problem arise. In fact, a worst case of congestion collapse can occur and the problem is usually referred to as *TCP-Incast*.

Such communication pattern is not uncommon, it arises in typical data centre applications: *cluster-storage* when data is stripped on multiple servers (for reliability and better use of bandwidth) and servers need to respond to a request, *web-search* when many workers respond to a search query. In these set-ups, the data usually traverse an Ethernet switches which typically have small buffers of the range 32KB-256KB; thus a high chance that they overflow in case of congestion. Large companies (e.g. Google, Microsoft, Amazon, etc) use data centres for web search, storage, e-commerce and large-scale computations. Thinking business; the use of existing technologies is more cost effective, therefore the vast majority of data centres use TCP for communication [23].

Technically (and historically) speaking, some of TCP parameters were tuned for typical WAN environments, for example in Linux the intial retransmission time-out timer is set (to a reasonable value for WAN) of $200ms$, i.e. the sender can make a decision of a time-out (lost packet) after $200ms$. However, in an data centre environment with low latency (or round trip time) this is considered long and result in throughput degradation. To illustrate this point, suppose the sender sends a number of packets (say a window) and they are all lost, then it will take $200ms$ to realise that they are lost and starts retransmitting. During this period no packets are sent. Suppose that the retransmission time-out timer is close to the round trip time i.e. a smaller value, then the sender will realise that the packets are lost in nearly one round trip time and respond by retransmitting the lost packets, thus we have more packets sent in the same period of time. It has been found that using TCP with its current set-up and increasing the number of servers beyond a certain number result in a huge drop in the receiver's goodput[9], this is due to TCP large retransmission time-out value, window halving and time-outs.

One quick solution is to use large switch buffers. While this can delay the onset of Incast, it comes at the price of substantial increase in the cost, e.g. switches with 1MB packet buffering per port may cost $500,0000, TCP improvements e.g. NewReno, SACK, RED, ECN, Limited Transmit and modification to slow start, mitigate the problem but do not solve it. Ethernet flow control (when the sender is sending too much traffic, the overwhelmed node can send a PAUSE frame to throttle the sender) is effective when all nodes are on the same switch and less effective when nodes are on different switches, this is due to inter-trunk head-of-

[9] Application-level throughput, we define this in chapter 5

line blocking[10]

It has been shown that an effective solution is to reduce TCP's retransmission time-out to microsecond granularity, specifically; a value of $200\mu s$ achieves full goodput for as many as 47 servers in real world cluster environment [7]. Deviating from this point, their are other attempts to solve the problem at lower layers, particularly the use of the so-called Quantised Congestion Notification(QCN) [82] in Ethernet (see for example [4] for a modified QCN to alleviate the problem). However we are not intending to elaborate on that in this thesis, our concern is to focus on TCP solutions.

2.10 TCP-PEP Approach

One approach used to compensate for TCP performance degradation when working in different environments is the approach of TCP-Performance Enhancing Proxies (TCP-PEP) [17, p.5]. In general, PEP can be integrated i.e. implemented on a single node, or distributed i.e. implemented on multiple nodes. However, a common mechanism used in TCP-PEP is to split[11] a TCP connection into three parts, where the first and last run standard TCP and another compensating protocol runs in the middle, this could be a different protocol e.g. XCP or an optimised TCP algorithm e.g. high-speed long-delay TCP algorithm.

Three years ago[12] an ISP used to have an asymmetric path (a shared E1 for upload traffic and a higher bandwidth satellite link for download traffic) which likely to make TCP perform badly! One such problem that may arise is ACK compression at the upload link which results in undesirable bursts which in turn are reflected as a bursts in the forward direction. Such bursts are not good for the network. A TCP-PEP may be used to alter the ACK spacing to mitigate the effect of bursts and thus smoothing TCP throughput.

Another example is the use of TCP-PEP to alter the behaviour of TCP connection in a high BDP environment by generating local ACKs which make the congestion window evolve faster (affects forward performance) and thus enhance throughput. As said in the first paragraph of this section, this can be accompanied by another protocol/algorithm which takes the responsibility of sending the data over a high BDP pipe. There are other examples, like the use in VSAT and

[10] In a cross-bar switch fabric, when two ports have packets in their input queues destined to same output port, a contention may appear; which blocks the rest of packets in a FIFO input queue.

[11] Some may argue if this breaks the end-to-end argument [83] i.e. functionality is restricted at end-hosts. There is no functionality replacement at end hosts, PEPs adds performance optimisation to a subpath of the end-to-end path and that agrees with the end-to-end argument [17].

[12] This is based on the author's experience.

WLAN environments. A typical example of the use of TCP-PEP in WLAN is TCP-Snoop [10], this is briefly discussed in chapter 3.

We end this section with a science fiction note. In 1781 Sir William Herschel announced the discovery of Uranus (the seventh planet from the Sun). Later on, in 1986 the spacecraft Voyager-2 (unmanned interplanetary space probe) reached Uranus, four days before that scientists noticed jittered sent images caused by a software bug. The software bug was resolved and a piece of code was sent over 3.2×10^9km to Voyager-2 at the speed of light (300×10^3km/s) in three hours. The problem was solved and clear images were received.

Indeed there is no need to use a protocol with congestion control for the case in the previous paragraph, but we build an imaginary case based on it, it might appear in one of science fiction movies and may become true one day. The case is: having a multiple nodes (probes) in space, where these nodes are supposed to capture some scientific data and send it back to the Earth. Depending on the size of data and the number of nodes and the capacity of links and the hardware used in future, a congestion control problem may appear. If TCP is still used by that time, TCP-PEP may be considered as a solution. Since for example based on the previous paragraph, a round trip time of six hours may not fit a congestion control algorithm, for instance, one choice is to split a connection (multiple connections) and run a space version (e.g. aggressive rate increase) of TCP in the middle.

Alternatively, in literature we found three ways (at network layer) for packet networks to deal with congestion [94]: i) Warning bit (ECN is an example), ii) Choke packets, iii) Hop-by-hop choke packets. The hop-by-hop choke packets can also be considered as another choice to mitigate the previous problem.

2.11 Explicit Feedback Issues

TCP follows an end-to-end approach and treats the network as a black box, however the network can play an explicit role in mitigating the congestion problem. In this section we look at two examples[13] which we believe are related in the context[14] of TCP CC. The first one is Explicit Congestion Notification (ECN) [81] appeared in 2001 and the second one is Quick-Start mechanism [35] appeared in 2007.

ECN assumes that the network has some sort of Active Queue Management

[13] A third example is ICMP source Quench message sent back to the sender to reduce the rate in case of congestion. This technique is rarely used in the Internet: consumes bandwidth, due to being ineffective and unfair [45]

[14] There are network assisted protocols which need special routers, these are believed to be alternatives to TCP CC, an example is Explicit Control Protocol XCP [57] , these are not discussed in this thesis.

(AQM) which usually detects congestion proactively through (for example) average queueing delay, and reports this as a binary (congestion,no congestion) feedback information in the packets. There is no point in reporting this if the senders are not cooperating by changing their rates, thus ECN requires support from transport layer protocol(s). Information about ECN-Capability and binary feedback information can be conveyed via four code points i.e. two bits in the IP header of a packet, where two code points indicate if the transport protocol is capable of using the ECN information e.g. has headers to send congestion information to the other end, these two code points are called: ECN-Capable Transport (ECT). The other two code points are: non-ECT and Congestion Experienced (CE).

TCP cooperates with ECN by dedicating two flags in its header: ECN-Echo (ECE) by which the receiver can inform the sender when a CE packet has been received and a Congestion Window Reduced (CWR) by which the sender can inform the receiver that it responded by reducing the rate. Typically the ECN-Capability of TCP is negotiated during the connection set-up and the ECT is set.

A typical sequence of events is that a queue threshold is exceeded, congestion experienced bit(s) in the IP header field are set. In response to that when an ECN-Capable TCP receiver receives this information it echos it back by setting ECE in an ACK packet, the sender then backoffs exactly as it does when a packet drop happens and inform the reviver about this by setting the CWR in a forward packet. The approach reduces delay by preventing buffer overflow and reduces packet drop rates (if marking is used).

Another example of network cooperation, is Quick-Start which tries to find the appropriate initial congestion window size (and thus the available bandwidth) quickly i.e. in one round trip time. Opposed to Slow-Start available bandwidth probing mechanism, Quick-Start suggests an explicit feedback from all the routers along the path, for instance; the TCP sender would set its desired rate in bytes per seconds in a Quick-Start option in the IP header of a packet. Each router along the path can the either: i) Approve the requested rate, ii) Reduce the requested rate, iii) Unapprove the requested rate. The TCP receiver communicates this information back the the sender in an answering packet and the rate (congestion window) is adjusted accordingly. Subsequent transmissions are then governed by the congestion control algorithm. However if the request is not approved the default congestion control algorithm is used. An interesting point is that the range of the requested rate is: 80 Kbps - 1.3 Gbps. Which means that TCP CC can significantly benefit from this technique when working in a high speed long delay environments.

2.12 Summary

In this chapter we discussed a number of definitions of congestion control problem. We discussed the difference between terms like "congestion avoidance" and "congestion control" and how the goal of congestion control changes in time. First it was to prevent congestion, and nowadays to efficiently utilise high capacity links. We focused on end-to-end solutions and because these solutions rely on adapting the source rate, we discussed a number of approaches used to adapt the source rate, we classified these into three classes: i) Linear algorithms, i) Non-linear algorithms, iii) Equation-based algorithms. Then we looked at the practical side of some of these solutions, mainly the most deployed protocol, TCP, then Multipath TCP. We steered the discussion towards some challenges to TCP in high-speed low-latency networks (e.g. TCP Incast), high-speed long-delay networks and wireless networks, this is followed by a general approach (TCP-PEP) that can be used to mitigate some of these problems. Finally, we mentioned that TCP can cooperate with other approaches which gather information from network, we provided two examples: i) ECN as an explicit congestion notification, ii) Quick-Start mechanism to determine the initial window size. However, solutions for the challenges that we left open are discussed in the next chapter with more elaboration.

Chapter 3

TCP Congestion Control Variants

This chapter sheds the light on a number of TCP congestion control algorithms aimed at optimising the performance of TCP in high-speed long-delay networks, it also mentions other variants designed to enhance the performance of TCP in different set-ups e.g. wireless networks. The chapter is divided as follows: in section 3.1 we briefly discuss a number of high-speed long-delay TCP variants, we used the notation: TCP-X, (where X is the name of that variant) to refer to the variant under discussion in each subsection. We elaborate more on TCP-Illinois and TCP-YeAH variants in chapter 6 and chapter 7 respectively. In section 3.2 we mention a number of other TCP variants. Finally we summarise in section 3.3 the ideas in this chapter.

3.1 TCP for High-Speed Long-Delay Networks

In the last decade, the research community responded with a number of congestion control algorithms aimed to moderate the problem of bandwidth under-utilisation of high BDP pipes [69, 102, 47, 32, 61, 64, 27, 93, 9], we visualise some of these in figure 3.1 and provide a briefing of some of them in this section.

We have noticed that a common thing between all these algorithms is that their designs are influenced by the following points:

- Scalability or utilisation, efficient use of bottleneck link.
- RTT fairness
- TCP friendliness, backward compatibility with TCP when deployed in low-speed short-distance networks.
- Fairness, equal number of packets in flight at each congestion event.

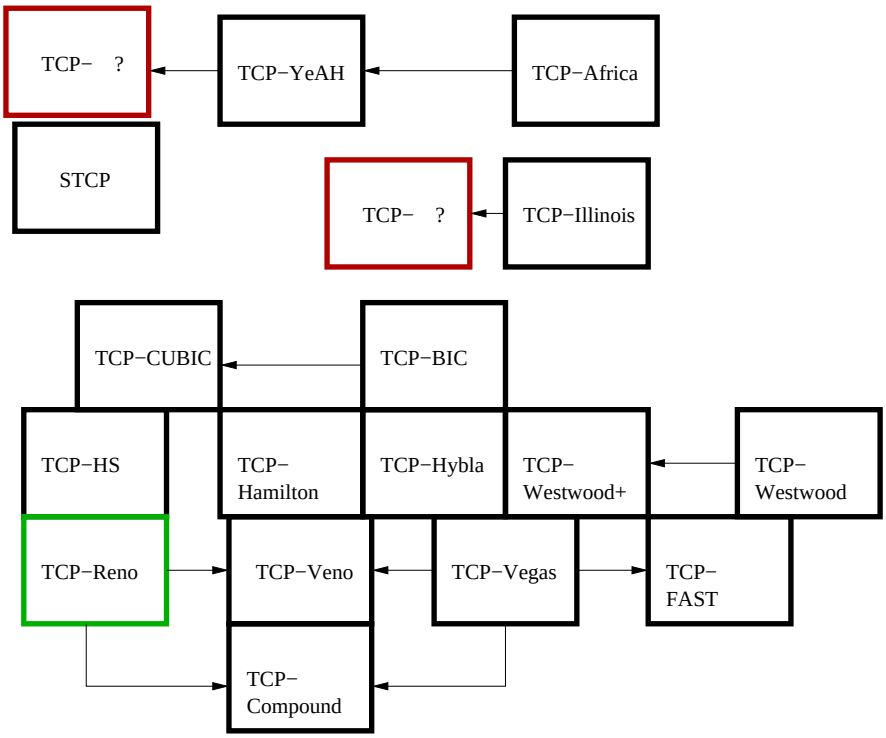


Figure 3.1: A number of TCP congestion control stacks

- Responsiveness in terms of convergence, respond quickly to changes in available bandwidth.

We also found that a simple way to characterise an algorithm and reveal many of these points is by using the so-called *Response Function*, this is basically the average throughput as a function of the back off probability (just to link ideas, this is the same equation that we mentioned in chapter 2, when we discussed equation-based algorithms, some refer to it as throughput equation). Some information can be directly revealed from a response function:

- Scalability
- RTT fairness
- TCP friendliness

The well-known expression of TCP-Reno in steady state average rate for small packet loss rates was found to be:

$$X \propto \frac{1}{RTT}\sqrt{\frac{1}{\beta p}}$$

A closed form was derived [78], we mention the simplified result here:

$$X_{TCP} \approx \frac{1}{RTT}\sqrt{\frac{3}{2bp}}$$

And for a deterministic AIMD decrease model [36]:

$$X_{AIMD} \approx \frac{1}{RTT}\sqrt{\frac{\alpha(2-\beta)}{2\beta p}}$$

For a stochastic model [103] and also mentioned in other literature [47]:

$$X_{AIMD} \approx \frac{1}{RTT}\sqrt{\frac{\alpha(1+\beta)}{2b(1-\beta)p}}$$

Where, α and β are constants, b is the number of packets acknowledged by each ACK (in normal operation $b = 1$, for delayed acknowledgement $b = 2$). Note that $\beta = 1/2$ here.

The deterministic AIMD model assumes that a packet is dropped each time the congestion window reaches a certain fixed number of packets. In contrast, an AIMD stochastic model assumes that packet is dropped randomly [36]. In fact this is the AIMD model mentioned in [103]. The advantage of the deterministic model is its simplicity and usefulness for focusing on the role of the α and β in AIMD congestion control, however the stochastic model gives more accurate model under certain assumptions.

If the response function has the form: c/p^d where, c and d are constants then by taking a log scale (i.e. the slope becomes the absolute value of d), the following properties can be deduced:

- As d increases the slope of the function increases and the scalability of the algorithm increases (i.e. large windows at low loss rates).
- RTT unfairness is roughly proportional to $(RTT_2/RTT_1)^{d/(1-d)}$, so if d increases the slope of the function increases and the RTT unfairness increases [102].
- TCP friendliness can be seen from the point where the response function of an algorithm crosses that of standard TCP. If it crosses TCP at lower loss rates then its more TCP friendly. The reasoning behind that ([102]) is since TCP does not consume too much bandwidth at low loss rates (remember, this is the problem with the TCP algorithm) other algorithms could apply their scalable techniques and consume what is left from TCP, so here we say, they are not working like TCP in this region. After this point (i.e. the crossing point) they start working like TCP. In other words it becomes more TCP friendly if an algorithm function crosses TCP's function as low as loss rate as possible.

We recreated a response function plot of a number of TCP variants. Figure 3.2

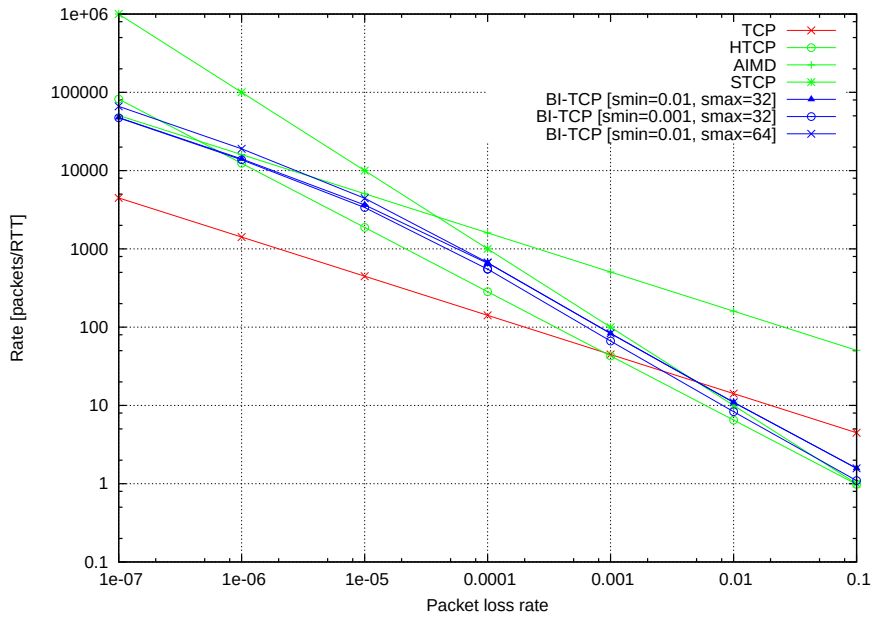


Figure 3.2: Response functions

depicts a log scale of this plot. The value of d for TCP, AIMD,TCP-HS and STCP are 0.5,0.5,0.82 and 1 respectively. An interesting feature of TCP-BIC is that this value is variable, it takes values $0.5 < d < 1$, as the window size increases d decreases from 1 to 0.5, this makes the algorithm RTT fair (with low d) for large windows and TCP friendly (high d, high slope thus shifting the crossing point to the left) for small windows.

What we are trying to emphasise here is that response functions can be used to understand the behaviour of congestion control algorithms and thus helps in the design process. In the following sections, we provide a discussion of a number of TCP variants and see how they considered the aforementioned issues in their design.

3.1.1 TCP-BIC

The basic idea behind this loss-based algorithm[1] [102] is that it uses a combination of additive increase, binary search, multiplicative decrease and slow start (the former two are collectively referred to as binary increase) to find the right size of congestion window that matches the pipe capacity. Two windows are maintained, one holds the value of congestion window when loss happens max_window and the other holds the value after multiplicative decrease takes action min_window. The mid point between the two is called the target window and is held in another variable w_{target}. After that for each received acknowledgement, if $w_{target} - cwnd < S_{max}$ increase directly to w_{target} (binary

[1]Default in Linux kernel

search) otherwise increase by S_{max} (additive increase[2]), update the following: $min\_window = cwnd, w_{target} = \frac{max\_window+min\_window}{2}$. Perform slow start any time the current window grows past max_window (since the maximum window is unknown the above sequence is effective-less, so probing for max window stars). To aid convergence, the algorithm use a *Fast Convergence* technique that exploits the down trend of congestion window when another flow starts competing for bandwidth. The previous max_window is memorized in $prev\_window$ and when loss happens, if $prev\_window > max\_window$ the algorithm computes the mid value but this time it sets the max_window to this value *then* computes a mid value again (based on the new maximum window) and sets the w_{target} to this value. To achieve TCP friendliness, the algorithm operates like standard TCP if the congestion window is below certain threshold. Figure 3.3 depicts the congestion window evolution of the two flows using the algorithm and competing for bottleneck link using a Drop-Tail queue.

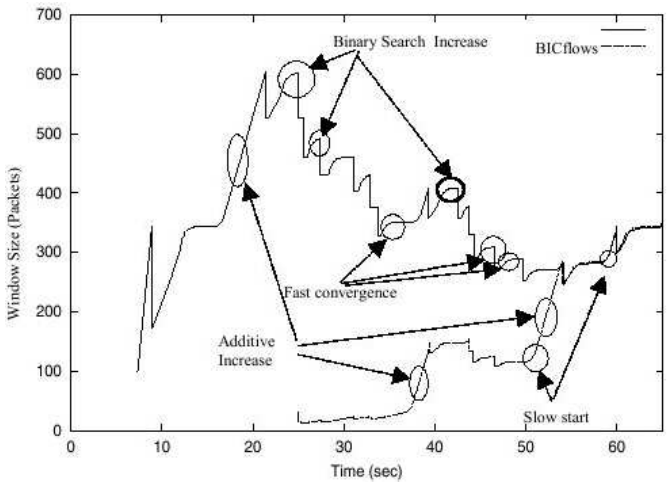


Figure 3.3: Congestion windows: two TCP-BIC flows competing for bandwidth [102]

3.1.2 TCP-CUBIC

This algorithm [47] is an improvement over TCP-BIC, it tries to achieve the same congestion window evolution of TCP-BIC while making it less dependent on round trip time in an attempt to achieve RTT-fairness between flows. The idea is to adapt the congestion window in real time according to a function. The function is shown below, t is the elapsed time since last back off

$$cwnd = \begin{cases} cwnd_{tcp} & = & cwnd_{max}\beta + 3\frac{1-\beta}{1+\beta}\frac{t}{RTT} & \text{if } cwnd_{tcp} > cwnd_{cubic} \\ cwnd_{cubic} & = & C(t-K)^3 + cwnd_{max} & \text{otherwise} \end{cases}$$

[2]Binary search and additive increase are collectively referred to as: binary increase

If $cwnd_{tcp}$ is larger than the $cwnd_{cubic}$ this means that the the algorithm is likely to be working in a short round trip time network, therefore the TCP window is selected, however if $cwnd_{cubic}$ is larger, then the algorithm likely to be working in a long round trip time network and the CUBIC window is selected. Recent studies [91] have shown that the algorithm causes collateral damage to real-time traffic on broadband networks.

3.1.3 TCP-Illinois

As it can be seen from figure 3.3, the concavity feature of congestion window is desirable, this actually leads to larger area under the curve which translates to more packets being transmitted. Another loss-delay based algorithm with this feature is TCP-Illinois [69]. The algorithm uses additive increase and multiplicative decrease (upon loss):

$$\begin{aligned} Ack: \quad cwnd &\leftarrow cwnd + \frac{f_\alpha(d_a)}{cwnd} \\ Loss: \quad cwnd &\leftarrow (1 - f_\beta(d_a)) \times cwnd \end{aligned}$$

But it adapts the AI and MD parameters according to estimated average queueing delay, $f_\alpha(d_a) \propto \frac{1}{d_a}$ and $f_\beta(d_a) \propto d_a$. The basic idea is to estimate the maximum average queueing delay as the difference between the maximum and minimum average round trip times seen so far by the flow during a connection. Then to estimate the current average queueing delay (estimation is done each round trip time i.e. each time a window is sent). If the estimated average queueing delay is close to the estimated maximum delay then it sets small AI and large MD, if the it is far it sets large AI and small MD. The algorithm falls back to standard TCP operation if the congestion window is below a threshold. One advantage of this algorithm is that, adapting the MD parameter in this way increases the immunity against non-congestive packet loss, a disadvantage; however, is that a receiver can cheat by increasing the round trip time, thus the sender thinks that there is a large queue and overestimates the maximum round trip time, subsequently the current average delay is likely to be far from the maximum delay and the AI is large. We address other issues of this algorithm and propose solutions in chapter 6.

3.1.4 TCP-HS

Another algorithm that uses AIMD is TCP-HS [32], the algorithm adapts AI and MD as a functions of the congestion window size.

$$\begin{aligned} Ack: \quad cwnd &\leftarrow cwnd + \frac{f_\alpha(cwnd)}{cwnd} \\ Loss: \quad cwnd &\leftarrow f_\beta(cwnd) \times cwnd \end{aligned}$$

Where, $f_\alpha(.) \uparrow$ and $f_\beta(.) \downarrow$ as $cwnd \uparrow$. Again, the algorithm engages after a window threshold. The idea is to adjust α and β so that the response function gives certain W/RTT at certain high $p(W)$ (low window), and higher W_1/RTT at certain low $p(W_1)$ (high window).

3.1.5 STCP

Scalable TCP [61] algorithm adopts a multiplicative increase and multiplicative decrease approach, while this is very scalable it has fairness and RTT unfairness issues, however the rule is simple simple to implement and scalable for high-speed long-delay networks,

$$\begin{aligned} Ack: \quad & cwnd \leftarrow cwnd + \alpha \\ Loss: \quad & cwnd \leftarrow \beta \times cwnd \end{aligned}$$

Where, $\alpha = 0.01$, $\beta = 0.875$ and the algorithm engages after a window threshold.

3.1.6 TCP-Hamilton

This algorithm updates the congestion window as a function of elapsed time since last congestion event, the function was chosen to achieve a response function similar to that of TCP-HS.

$$\begin{aligned} Ack: \quad & cwnd \leftarrow cwnd + \frac{2(1-\beta)f_\alpha(\Delta)}{cwnd} \\ Loss: \quad & cwnd \leftarrow f_\beta(B) \times cwnd \end{aligned}$$

$$f_\alpha(\Delta) = \begin{cases} 1 & \text{if } \Delta \leq \Delta_L \\ \max(\bar{f}_\alpha(\Delta) * T_{min}, 1) & \text{if } \Delta > \Delta_L \end{cases}$$

$$f_\beta(B) = \begin{cases} 0.5 & \text{if } |\frac{B(k+1)-B(k)}{B(k)}| > \Delta_B \\ \min(\frac{T_{min}}{T_{max}}, 0.8) & \text{otherwise} \end{cases}$$

Where, Δ is the elapsed time since the last congestion event, $\Delta_B = 0.2$, $B(k+1)$ is the maximum achieved throughput during the last congestion epoch and: $\bar{f}_\alpha(\Delta) = 1 + 10(\Delta - \Delta_L) + (\frac{\Delta-\Delta_L}{2})^2$

3.1.7 TCP-FAST

This algorithm [54] is a delay-based algorithm, it adapt the AI parameter according to a predefined function based on the congestion window and average queueing delay. The congestion window is updated as a smooth average:

$$\begin{aligned} Every\ RTT: \quad & cwnd \leftarrow (1-\gamma)(cwnd) + \gamma\left(\frac{T_{min}}{T} cwnd + f_\alpha(cwnd, d_a)\right) \\ Loss: \quad & cwnd \leftarrow \beta \times cwnd \end{aligned}$$

Where, $\beta = 0.5$, T_{min} and $\bar{T}$ are minimum and average observed round trip times respectively, d_a is the average queueing delay. The algorithm also reacts to loss by multiplicatively decreasing the congestion window. Note that the congestion window update rule has a component that is inversely proportional to the average round trip time i.e. it decreases every round trip time (gradually) as the queue builds up (congestion), rather than waiting for packet loss to happen, in other words this is a CA rather than a CC approach (to link ideas, please see figure 2.2).

3.1.8 TCP-YeAH

Yet Another High speed algorithm [9], tries to exploit the network bandwidth efficiently while putting less stress on the network compared to standard TCP. The basic idea is to have two different modes of operation: *Fast* and *Slow* modes. In the fast mode the congestion window is incremented according to an aggressive rule (STCP rule was used for ease of implementation but any other rule can be used). In the slow mode the congestion window is incremented like standard TCP and a precautionary decongestion algorithm is implemented.

Switching between slow and fast modes is decided according to the estimated number of packets in the bottleneck queue. This number is estimated as follows [9]: Let RTT_{base} be the minimum RTT measured by the sender (estimate of propagation delay) and RTT_{min} the minimum RTT estimate in the current data window of *cwnd* packets and thus is updated once per window of data (i.e. each RTT). The current queuing delay can be estimated as $RTT_{queue} = RTT_{min} - RTT_{base}$. This RTT_{queue} is then used to infer the number of packets enqueued at the bottleneck as:

$$Q = RTT_{queue}.G = RTT_{queue}.\left(\frac{cwnd}{RTT_{min}}\right)$$

Where, G is the goodput. The ratio[3] between the queuing delay and the propagation delay $L = RTT_{queue}/RTT_{base}$ is used to indicate network congestion level.

If $Q < Q_{max}$ and $L < 1/\varphi$, the algorithm is in the fast mode, otherwise it is in the slow mode. Q_{max} and φ are tunable parameters; Q_{max} is the maximum number of packets a single flow is allowed to keep into the buffers. $1/\varphi$ is the maximum level of buffer congestion with respect to BDP. The precautionary decongestion algorithm is to diminish the congestion window by Q whenever $Q > Q_{max}$ and the to set the slow start threshold to half the congestion window.

The fact that the queue estimate keeps exceeding Q_{max} even after applying the decongestion algorithm is logically coupled with the fact that there is a greedy competing source and this is the rational for falling back to slow mode. In other

[3]This can be thought of as the ratio between the number of packets enqueued and the pipe's BDP

words if TCP-YeAH spends more time in the slow mode that it does in the fast mode then it is more likely that it is competing with a greedy source. However TCP-YeAH does not continue this polite behaviour of diminishing its congestion window by Q when competing with greedy sources. It maintains count (this count in fact represents an estimate of the competing standard TCP congestion window, since it increase by one packet each RTT), the count is incremented each time the algorithm finds itself in the slow mode. If the congestion window reaches this count or less, it stops decreasing for not to let the window drop below the estimated standard TCP window, then it starts increasing increasing like standard TCP. When packet loss happens the count is halved.

One point to note here is that the queue estimate (Q) is done each RTT, i.e. this is the granularity of decongestion, thus considering flows with different RTTs, each flow reaches Q_{max} at different time and thus diminish its excess packets in the queue at different time without affecting link utilisation. Therefore, in theory the algorithm achieves RTT fairness. In chapter 7 we proposed an increase rule for this algorithm which automatically assigns small increase for flows with small RTT and large increase for flows with large RTT, thus improving RTT fairness.

3.1.9 TCP-Compound

This algorithm [93][4] is a loss-delay based algorithm, it maintains a compound congestion window: loss-based congestion window and delay-based congestion window, so TCP's window becomes: $min\{cwnd + dwnd, awnd\}$. The basic idea is to use an aggressive binomial rule (to link ideas please see figure 2.5, IIAD as an example) in the delay-based congestion window, after reaching a certain number of packets determined by a threshold γ, the delay-based congestion window diminishes by $\zeta\, diff$[5]. The loss-based congestion window increases normally according to standard TCP rule. However note that even after reaching the threshold and the delay-based congestion window decreases, there are still packets in the queue due to the loss-based congestion window, as a result the delay-based congestion window keeps decreasing until it disappears, effectively leaving only the loss-based congestion window. The overall congestion window evolves according to the following rules [93]:

$$\begin{aligned} Ack: \quad & cwnd \leftarrow cwnd + \frac{\alpha}{cwnd+dwnd} \\ Loss: \quad & cwnd \leftarrow (1-\beta) * (cwnd + dwnd) \end{aligned}$$

[4]Default in Windows XP and Vista SP1. Discontinued in Linux kernel code base since 2.6.17
[5]Note how this component is similar to what was mentioned about TCP-YeAH

$$
\begin{array}{ll}
dwnd \leftarrow dwnd + \alpha((cwnd + dwnd)^k - 1)^+ & \text{if } diff < \gamma \\
dwnd \leftarrow dwnd - \zeta\, diff & \text{if } diff \geq \gamma \\
dwnd \leftarrow ((cwnd + dwnd)(1 - \beta) - \frac{cwnd}{2})^+ & \text{if Loss/ECN}
\end{array}
$$

diff is the estimated number of backlogged packets:

$$
\begin{array}{lcl}
Expected & = & \frac{cwnd+dwnd}{baseRTT} \\
Actual & = & \frac{cwnd+dwnd}{RTT} \\
diff & = & (Expected - Actual) * baseRTT
\end{array}
$$

The parameters were selected so that the algorithms gives a response function similar to that of TCP-HS (i.e. by comparing both response functions). They are found to be: $k = 0.8$, $\alpha = 0.125$, $\beta = 0.5$, for implementation purposes k was set to 0.75. (easier through the use of fast integer algorithm of square-root). An auto-tuning algorithm (*gamma auto-tuning*) was proposed for empirically adjusting the γ value.

3.2 Other TCP Variants

There are so many TCP variants customised to achieve different goals, in this section we only mention some which are related in the context of this thesis. One of the earliest proposals was TCP-Vegas [19] which is totally a delay-based algorithm, the idea is very similar to that of the algorithm ([98]) mentioned in chapter 2. TCP-Vegas algorithm calculates the difference between actual rate and expected rate and if this is less than a threshold (α) it increases linearly, if the difference is larger that another threshold(β) it decreases linearly and if the difference is larger than α and less than β then the congestion window remains unchanged. The algorithm achieves better throughput compared to standard TCP, however, the additive decrease nature makes it less responsive and rise fairness issues.

A hybrid algorithm which combines features from standard TCP (Reno) and TCP-Vegas is TCP Veno [44], the algorithm estimates the number of backlogged packets at the bottleneck using TCP-Vegas parameters($N = Actual * (RTT - baseRTT)$), if this number is less than a threshold (β) a non-congestion state is assumed and the increase rule of standard TCP is adopted but the multiplicative decrease parameter is set to a high value (4/5). However, if the number is greater than or equal the threshold, a congestion state is assumed and the congestion window is increased by one packet every other round trip time instead of one packet every round trip time (i.e. less aggressive and more concave window) and the multiplicative decrease parameter is set to 1/2.

TCP-Westwood [20] alters the congestion window upon congestion i.e. when receiving duplicate ACK or time-out. The idea is measure the flow rate by monitoring and filtering the rate of ACK at the source. By doing so, the congestion window can be deduced by multiplying the rate by the minimum round trip time RTT_{min}, the slow start threshold and the congestion window are then set to this values (instead of halving the window). In case of time out, only the slow start threshold is set to this value and the congestion window is to 1 packet. All other increase rules of standard TCP are maintained. An advantage of this approach is that it is immune against non-congestive packet loss, particularly; due to the increase rules of the algorithm, it fits wireless environments.

Finally, TCP-Snoop [10] tries to confine packet retransmissions over the wireless link of the path only by snooping inside TCP connections and transparently transmit corrupted packets. The basic idea is to run a snoop agent at wireless gateway, the agent maintains a state for each flow traversing the gateway and packets sent from the wired host (sender) to the wireless host (receiver) are cached locally. If duplicate ACKS are received the agent perform local retransmission and suppress the duplicate ACKS, thus prevents the sender from doing retransmission and invoking congestion control algorithm(s). It has been mentioned that if the direction of flow is reversed i.e. the sender on the wireless side, acknowledgements may take long time and this may have impact on performance [51]

3.3 Summary

In this chapter we discussed a number of TCP congestion control proposals that are mainly optimised to overcome the high-speed long-delay problem. Their are a number of important factors in the design of such proposals: efficiency, RTT fairness, TCP friendliness, fairness and responsiveness. We also showed that lots of these factors can be inferred directly from a response function plot.

We found that the proposals are either loss-based (reactive approach, CC) or delay-based (proactive approach, CA), or hybrid approach (CC,CA). They are trying to achieve similar goals but in different ways, however they are experimental and operating systems have different default algorithms. In fact some default proposal in certain operating systems are discontinued in other operating systems. Recent studies showed that some of these algorithms have adverse effects on real time traffic.

We showed examples of other TCP congestion control proposals that try to solve other problems, for example flatten and increase the throughput, increase the immunity against non-congestive packet loss and thus increase the throughput.

Chapter 4

Optimisation & Congestion Control

It is valuable to understand the formulation of congestion control problem as a resource allocation problem, before actually tackling any of the problems mentioned in the previous chapters. For this reason we provide a discussion of this formulation and some stability issues related to TCP congestion control.

This chapter is structured as follows: in section 4.1 we discuss the formulation of congestion control problem as a resource allocation problem. Section 4.2 highlights a three different types of algorithms that can be used to solve the resource allocation problem, we finalise this section by relating this to TCP congestion control. In section 4.3 we contribute and extend the current work by analysing a non-linear case with non-congestive packet loss and a version of TCP with a variable MD. We further explore the dynamics of TCP in section 4.4 by linearising the model around an equilibrium point. Finally we conclude in section 4.5.

4.1 Congestion Phenomenon & Resource Allocation

In this section we discuss the formulation of congestion control problem as a resource allocation problem, the intention is to give the reader a general idea about how the problem is formulated and how stability techniques can be applied, without going into details. A simple case of single link single source was used in all cases. However the extension to multiple sources and multiple links can be done using algebraic techniques (e.g. matrices etc). In fact a comprehensive analysis has been done on that in literature [88]. In spite of that we analysed some special/hypothetical cases which up to our knowledge have not been analysed in the same context.

4.1.1 Resource Allocation

The resource allocation problem is that of fairly assigning rates to users who share a common link. In this case the resource is the capacity of the link. Note that this semantic also agrees with the congestion control (using the network efficiently). However this problem can be viewed as an optimisation problem [88] this is basically to maximise or minimise a function subject to fixed outside conditions or constraints. The method of Lagrange multipliers is usually used for solving this class of problems. An example of this type of problems is: How can we minimise the aluminium used to make a "can" while making sure it can hold 100 ml of liquid?

In general, if $f(x)$ is a differentiable concave function, and H is a matrix of appropriate dimension (could be constant), then the problem can be formulated as:

$$\max_{\mathbf{x}} f(\mathbf{x}), \tag{4.1}$$

Subject to:

$$H\mathbf{x} \leq \mathbf{c}$$

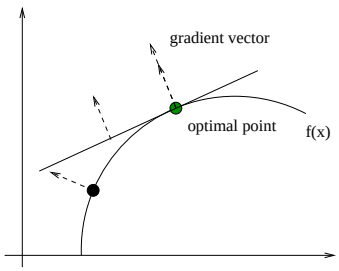


Figure 4.1: Graphical illustration of the Lagrange multiplier method.

Where c is the constraint. Figure 4.1 illustrates the graphical inspiration of the problem. Note that at the optimal point the gradient vectors (normal to the curve / surface / ...) are in the same direction but could have different magnitudes. We could equate both gradient to find the optimal point. However a more formal way is to define a function then differentiate it and equate it to zero. This function is usually referred to as the *Lagrangian function*. The Lagrangian function L can be written as:

$$\begin{aligned} L(\mathbf{x}, \lambda) &= f(\mathbf{x}) - \lambda^T(H\mathbf{x} - c), \\ \tfrac{\partial L}{\partial \mathbf{x}} = 0, &\Longrightarrow \bigtriangledown f(\mathbf{x}) - H^T\lambda = 0. \end{aligned} \tag{4.2}$$

And,

$$\lambda^T(H\mathbf{x} - \mathbf{c}) = 0. \tag{4.3}$$

Where $\lambda \geq 0$ is the Lagrange multiplier. And $\bigtriangledown f(\mathbf{x}) = (\partial f/\partial x_1, \partial f/\partial x_2, \ldots, \partial f/\partial x_n)$ is the gradient vector. If $\mathbf{x}$ satisfies 4.2 and 4.3, then it solves 4.1.

Another formal way to look at the problem is through the so-called *duality* [88]. This is basically putting the function of our interest in another way. This can be done by fixing the the Lagrangian function at a certain point of $\mathbf{x}$ and varying the multiplier λ to reach the optimal value (instead of fixing $\vec{\lambda}$ and varying $\mathbf{x}$ in first method, which is also called the primal method). Obviously λ should be tuned to find the minimum of the new function (which is now a function of the multiplier only). To illustrate this point we will write the new function which is called the *dual function* [88], as:

$$D(\vec{\lambda}) = \max_{\mathbf{x} \in C} L(\mathbf{x}, \vec{\lambda}) \tag{4.4}$$

Where C is the set of all $\mathbf{x}$. We can also write:

$$\begin{aligned} f(\mathbf{x}) &\leq f(\mathbf{x}) \underbrace{-\vec{\lambda}^T(H\mathbf{x} - c)}_{\geq 0} \\ \max_{\mathbf{x} \in C} f(\mathbf{x}) &\leq \max_{\mathbf{x} \in C}\{f(\mathbf{x}) - \vec{\lambda}^T(H\mathbf{x} - c)\} = D(\vec{\lambda}) \\ f(\hat{\mathbf{x}}) &\leq \min_{\vec{\lambda} \geq 0} D(\vec{\lambda}). \end{aligned}$$

For concave objective function and linear constraints, the equality of the above equation holds, which means that the primal and dual objectives are the same. Figure 4.2, depicts a graphical interpretation of the primal and dual methods. Both solves 4.1 but from different perspectives.

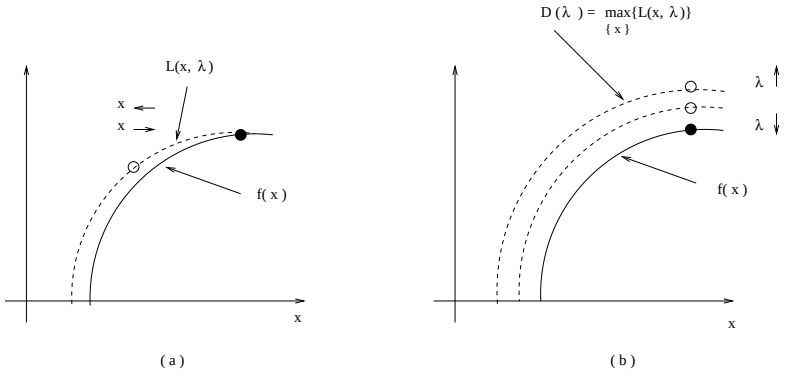


Figure 4.2: Optimal objectives: (a) Primal method (b) Dual method.

To relate this to the resource allocation problem, we will adopt the approach mentioned in [88] and assume we have r users sharing a route from a source to a destination. Each user has a rate x_r bps, and drives a utility function $U_r(x_r)$.

Then the optimisation problem is:

$$\max_{\{x_r\}\in S} \sum_r U_r(x_r) \tag{4.5}$$

With the constraints of:

$$\sum_{r:\ell\in r} x_r \leq c_\ell, \ell \in L, \tag{4.6}$$

$$x_r \geq 0, r \in S,$$

Where:

c_ℓ: is the capacity of the link;

L: is the set of all links in the route;

S: is the set of all sources in the network;

$U_r(x_r)$: for each r, is a contiguously differentiable, non-decreasing, strictly concave function, and for all r, $U_r(x_r) \to -\infty$ as $x_r \to 0$[1].

Depending on the type of the utility function, we define the following types of fairness:

1. Weighted proportional fairness: when $U_r(x_r) = w_r \log x_r$, and because of the property of convex functions the following holds: $\sum_{r\in S} U'_r(\hat{x}_r)(x_r - \hat{x}_r) \leq 0$, thus $\sum_{r\in S} w_r \frac{x_r - \hat{x}_r}{\hat{x}_r} \leq 0$. Were $\hat{x}_r$ is the optimal user rate. To illustrate this property of convex functions we do the following: if we take the first order Taylor series expansion of a convex function $f(y)$ at point x, this gives $f(x) + (y - x)f'(x)$, where $f'(.)$ is first derivative. This is an approximation of the function around the point x and the equation is a line equation. If the function is convex, any point on this line is less or equal than any point on the actual function, i.e. $f(x) + (y - x)f'(x) \leq f(y)$. A special case appears when $y = \hat{x}$, since $f(\hat{x}) \leq f(x)$ we get: $(\hat{x} - x)f'(x) \leq f(\hat{x}) - f(x) \leq 0$[2]

2. Minimum potential delay fairness: when user with rate x_r attempts to transfer a file of size w_r, the requires transfer time is w_r/x_r, let $U_r(x_r) = -w_r/x_r$, note that the minus sign is to make the function concave, in this context we say that the problem is to maximise the sum of the utility functions and we take the absolute value of the final result. This is equivalent of taking a convex function $U_r(x_r) = w_r/x_r$ and minimising the sum of utility functions. Both cases have the same meaning of minimising the total file transfer time.

3. Max-Min fairness: Here the capacity of the link is divide among the users in such a way that the only way to increase a user rate is to decrease the rate

[1] We will see later that this does not hold for TCP congestion control algorithm

[2] We add this illustration for sake of clarity.

allocated to some other source. It can also be shown that the users rates allocation is max-min fair if and only if every user has a bottleneck [88]. In the case of single bottleneck in the network, this implies an equal share of the resource among the user flows through it [60]. An approximate utility function for this case could be: $U_r(x_r) \approx 1/(\alpha {x_r}^{\alpha})$ for large values of α

A general class of utility function mentioned in [88] is:

$$U_r(x_r) = \begin{cases} -w_r \frac{x^{1-\alpha_r}}{1-\alpha_r}, & \alpha_r > 0, \alpha_r \neq 1 \\ w_r \log x_r, & \alpha_r = 1 \end{cases} \tag{4.7}$$

Where:
Proportional fairness $\longrightarrow \alpha_r = 1, \forall r$
Minimum potential delay fairness $\longrightarrow \alpha_r = 2, \forall r$
Minimum- Maximum fairness $\longrightarrow w_r = 1, \alpha_r = \alpha, \forall r, \alpha \to \infty$

We will give a simple example to illustrate the basic idea, a similar example can be found in [88]. Consider the network in figure 4.3, there are three sources, sources 0 and 1 share link A's capacity, C_a, sources 0 and 2 share link B's capacity, C_b. Substituting in the above equations we get:

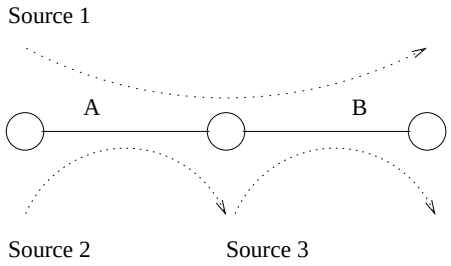


Figure 4.3: Example to illustrate the basic idea

$$\max_{\{x_r\}} \left(\frac{x_1^{1-\alpha}}{1-\alpha} + \frac{x_2^{1-\alpha}}{1-\alpha} + \frac{x_3^{1-\alpha}}{1-\alpha} \right)$$

subject to:

$$\begin{aligned} x_1 + x_2 &\leq C_a, \\ x_1 + x_3 &\leq C_b, \\ x_1, x_2, x_3 &\geq 0, \end{aligned}$$

Using the general class utility function, the Lagrangian function is:

$$L(x_1, x_2, x_3, \lambda_a, \lambda_b) = \frac{x_1^{1-\alpha}}{1-\alpha} + \frac{x_2^{1-\alpha}}{1-\alpha} + \frac{x_3^{1-\alpha}}{1-\alpha} - \lambda_a(x_1 + x_2 - C_a) - \lambda_b(x_1 + x_3 - C_b).$$

Then:

$$\begin{aligned}
\frac{\partial L}{\partial x_1} = 0 \rightarrow \quad & \frac{1}{x_1^\alpha} = \lambda_a + \lambda_b, \\
\frac{\partial L}{\partial x_2} = 0 \rightarrow \quad & \frac{1}{x_2^\alpha} = \lambda_a, \\
\frac{\partial L}{\partial x_3} = 0 \rightarrow \quad & \frac{1}{x_3^\alpha} = \lambda_b.
\end{aligned}$$

Since C_a, C_b and α are known, we only have five unknowns and five equations:

$$\begin{aligned}
x_1 + x_2 - C_a &= 0, \\
x_1 + x_3 - C_b &= 0, \\
x_1^\alpha(\lambda_a + \lambda_b) - 1 &= 0, \\
x_2^\alpha \lambda_a - 1 &= 0, \\
x_3^\alpha \lambda_b - 1 &= 0.
\end{aligned}$$

Tables 4.1 and 4.2 show the solution of the equations using a computer program[3]. Note that for high values of α the solution exhibits max-min fairness.

Table 4.1: $C_a = 10$ and $C_b = 2$

α	x_1	x_2	x_3	λ_a	λ_b
1	0.94495	9.05505	1.05505	0.11044	0.94782
2	0.996916	9.003084	1.003084	0.012337	0.993860
10	1.0000e-00	9.0000e+00	1.0000e+00	2.8680e-10	1.0000e-00

Table 4.2: $C_a = 15$ and $C_b = 5$

α	x_1	x_2	x_3	λ_a	λ_b
1	4.226497	15.773503	5.773503	0.063397	0.173205
2	2.4751e+00	1.2525e+01	2.5249e+00	6.3746e-03	1.5686e-01
10	2.5000e+00	1.2500e+01	2.5000e+00	1.0737e-11	1.0486e-04

If each link could compute the Lagrange multipliers corresponding to its capacity constraint, and the information can be conveyed to the sources, then each source can make use of the sum of the Lagrange multipliers on its route to compute its optimal rate, since the optimal rate equations are known, the sources can adjust their rates in a fast and dynamic way according to an *algorithm* which makes use of the features of the utility function (concave) to converge to a unique optimal point and stays there. Based on this idea and on the following assumptions, the resource allocation problem can be solved precisely if [88], [60]:

[3]We used GNU Octave, version 2.9.9

1. Each user knows its own utility function.

2. Each router knows the total arrival rate (resource usage) on its links.

3. There is a protocol that conveys the required information about the resource usage on a user's route to the user, this could be done by having a field in the packet header accessible by the routers to put the required information in it, each router can adds its resource usage information to this field then when the packet reaches the destination, the receiver can send back this information in the acknowledgement packets. One disadvantage in this approach according to [88], is that the field would have to be large to convey the information accurately.

4.2 Congestion Control Algorithms

From an optimisation point of view, there are two main classes of congestion control algorithms: The primal algorithm, and the dual algorithm [60][4], as we will see later it is also possible to have primal-dual algorithm which uses the features of both algorithms. These algorithms are non-linear time variant feedback control systems and they need special treatment in the analysis phase. The proof of convergence and global asymptotic stability using Lyapanov stability theory for the algorithms in this section are mentioned in [88], [60].

The basic idea is nearly the same, solving the optimisation problem. However the primal algorithm tries to solve it by adapting the sources rates (source side), while the dual algorithm tries to solve it by adapting the prices at each link (network or router side). In this section we will have a look at both, then see how TCP's congestion control algorithm fits in this context. We start by defining some useful terms: R is the routing matrix and is defined as a matrix of size $|L| \times |S|$ where the $(\ell, r)^{th}$ entry is 1 if the source rate passes through link ℓ and 0 otherwise. y_ℓ denotes the total arrival rate of traffic at link ℓ, and the following holds:

$$\underbrace{\begin{bmatrix} 1 & 1 & 0\cdots \\ 1 & \ddots & \\ \vdots & & \end{bmatrix}_{|L|\times|S|}}_{R} \times \underbrace{\begin{bmatrix} x_1 \\ x_2 \\ \vdots \end{bmatrix}_{|S|\times 1}}_{\mathbf{x}} = \underbrace{\begin{bmatrix} y_1 \\ y_2 \\ \vdots \end{bmatrix}_{|L|\times 1}}_{\mathbf{y}}$$

[4]The names for each of them came from the theory of optimisation, see [13] or [88, p.17]

4.2.1 Primal & Exact Primal Algorithms

Instead of solving the optimisation problem directly, an approximate solution could be obtained by defining a function that has similar characteristics to the Lagrangian function, suppose we put it as:

$$V(\mathbf{x}) = \sum_{r \in S} U_r(x_r) - \sum_{\ell \in L} \int_0^{\sum_{s:\ell \in S} x_s} f_\ell(y) dy, \tag{4.8}$$

Where $f_\ell(.)$ is interpreted as the *price* of sending traffic at rate $\sum_{s:\ell \in S} x_s$ on link ℓ [88] or the penalty function [13] in [88]. The penalty function is assumed to be a non-decreasing contiguous function i.e. $\int_0^y f_l(x)dx \to \infty$, as $y \to \infty$, which means as the load on the link increase the price does not decrease. Furthermore $V(\mathbf{x})$ is assumed to be strictly concave function, the proof is mentioned in [88], this should be expected since as we mentioned before, the utility function is contiguously differentiable, non-decreasing, strictly concave. Also $V(\mathbf{x}) \to -\infty$ as $\|\mathbf{x}\| \to 0$ and $V(\mathbf{x}) \to \infty$ as $\|\mathbf{x}\| \to \infty$. Now the problem reduces to:

$$\frac{\partial V}{\partial x_r} = 0, r \in S.$$

$$U_r'(x_r) - \sum_{\ell:\ell \in r} f_\ell \left(\sum_{s:\ell \in S} x_s \right) = 0, r \in S. \tag{4.9}$$

If the router is able to compute its resource usage at each link, i.e y_ℓ, then the price can be computed according to the following formula:

$$p_\ell = f_\ell(y_\ell(t)).$$

and the route price:

$$q_r = \sum_{\ell:\ell \in r} p_\ell, \tag{4.10}$$

In matrix format:

$$\underbrace{\begin{bmatrix} 1 & 1 & \cdots \\ 1 & \ddots & \\ 0 & & \\ \vdots & & \end{bmatrix}_{|S| \times |L|}}_{R^T} \times \underbrace{\begin{bmatrix} p_{\ell 1} \\ p_{\ell 2} \\ p_{\ell 3} \\ \vdots \end{bmatrix}_{|L| \times 1}}_{\mathbf{p}} = \underbrace{\begin{bmatrix} q_1 \\ q_2 \\ q_3 \\ \vdots \end{bmatrix}_{|S| \times 1}}_{\mathbf{q}}$$

Now equation 4.9 can be rewritten as:

$$U_r'(x_r) - q_r = 0, r \in S.$$

And the candidate congestion control algorithm to solve this problem is:

$$\frac{dx_r}{dt} = k_r(x_r)(U_r'(x_r) - q_r(t)), r \in S. \tag{4.11}$$

$k_r(x)$ is an arbitrary non-decreasing contiguous function such that $k_r(x) > 0$ for any $x_r > 0$. The algorithm is globally asymptotically stable. The proof of convergence is provided in the appendix.

In order to have an exact solution to the optimisation problem, $f_\ell(.)$ should be chosen such that equation 4.8 can reflect the resource allocation problem *exactly*. In fact we shall find a way to tune $f_\ell(.)$ to the desired value. One solution inspired from the Queueing theory, uses the steady-state blocking probability with $y_\ell < \tilde{c}_\ell$ a buffer threshold B_ℓ in an *M/M/1* queue is:

$$p_\ell = f_\ell(y_\ell, \tilde{c}) = \left(\frac{y_\ell}{\tilde{c}}\right)^{B_\ell}. \tag{4.12}$$

In other words another parameter was added, $\tilde{c}$. It can be shown this parameter can be adaptively chosen such that the price solves the resource allocation problem exactly. Particularly, it can be adapted according to the following equation:

$$\frac{d\tilde{c}}{dt} = \alpha_\ell(c_\ell - y_\ell)^+_{\tilde{c}_\ell} \tag{4.13}$$

$$(g(x))^+_x = \begin{cases} g(x), & x > 0, \\ \max(g(x), 0), & x = 0, \end{cases}$$

Where $\alpha_\ell > 0$. Because the solution is inspired from the Queueing theory $\tilde{c}$ is usually referred to as the *virtual capacity*, for not to be confused with the original capacity of the link c_ℓ. One way to understand it is through its proportionality to the available bandwidth, for example if the arrival rate is greater than the link capacity (no available bandwidth) the virtual capacity should be decreased to increase the price and thus signal the source to reduce its rate. In contrast, if the arrival rate is less than the link capacity (there is an available bandwidth) the virtual capacity should be increased to signal the source to increase its rate.[5]

4.2.2 Dual Algorithm

Now we will discuss an algorithm that solves the optimisation problem directly and exactly, the dual algorithm views the problem from the network side, i.e.

[5]This is usually called adaptive virtual queue AVQ [88]

it dynamically adapt the prices at each router. Here we let the $V(\mathbf{x}, \vec{\lambda})$ be the Lagrangian function, so we can write:

$$\begin{array}{lcl} L(\mathbf{x}, \vec{\lambda}) & = & \sum_{r\in R} U_r(x_r) - \sum_{\ell\in L} \lambda_\ell \left(\sum_{r:\ell\in r} x_r - c_\ell\right), \\ \frac{\partial L}{\partial x_r} & = & 0, \\ \frac{\partial L}{\partial x_r} & = & U_r'(x_r) - \sum_{\ell\in L} \lambda_\ell = 0. \end{array} \tag{4.14}$$

Since the Lagrange multiplier is equivalent to the price i.e. $\lambda_\ell = p_\ell \geq 0$, and using 4.10, we can also write:

$$\begin{array}{lcccl} \frac{\partial L}{\partial x_r} & = & U_r'(x_r) - \sum_{\ell\in L} p_\ell & = 0, \\ U_r'(x_r) & = & \sum_{\ell\in L} p_\ell & = q_r, \\ U_r'(x_r) - q_r & = & 0, & & \\ x_r & = & {U'}_r^{-1}(q_r). & & \end{array} \tag{4.15}$$

And using 4.3 per link (not matrix),

$$\begin{aligned} p_\ell(\sum_{s:\ell\in s} x_s - c_\ell) = 0. \\ p_\ell(y_\ell - c_\ell) = 0. \end{aligned} \tag{4.16}$$

The candidate algorithm to solve for the price p_ℓ (or the Lagrange multiplier) and for the rate x_r, i.e. solving 4.15 and 4.16 is:

$$\frac{dp_\ell}{dt} = h_\ell(p_\ell)(y_\ell - c_\ell)^+_{p_\ell}. \tag{4.17}$$

Where $h_\ell(p_\ell) > 0$ is a non-decreasing contiguous function. Note that when $y_\ell = c_\ell \rightarrow \frac{dp_\ell}{dt} = 0$ i.e when the sum of all user rates going through link ℓ is equal to its capacity, the price adaptation stops and the optimal point is reached[6]. Since this algorithm adapts the price (or the Lagrange multiplier), it adopts the duality approach discussed in section 4.1.1 hence the name "dual algorithm". In fact the problem now is to minimise the duality function, which can be written as:

$$\begin{array}{lcl} D(\mathbf{p}) & = & \max_{x_r} \sum_r U_r(x_r) - \sum_\ell p_\ell \left(\sum_{s:\ell\in s} x_s - c_\ell\right) \\ & = & \max_{x_r} \sum_r \left(U_r(x_r) - x_r \sum_\ell p_{\ell:\ell\in r} p_\ell\right) + \sum_\ell p_\ell c_\ell. \\ & \therefore & \min_{p\geq 0} D(\mathbf{p}) \end{array}$$

[6]From stability point of view, we could ask whether the algorithm converges if it started from any initial condition? And how long would it take to reach the optimal point?

4.2.3 Primal-Dual Algorithm

As the name states, this is an algorithm that combines the features of both primal and dual algorithms, i.e. solving the problem from the network side and the source side at the same time:

$$\frac{dx_r}{dt} = k_r(x_r)(U_r'(x_r) - q_r), \tag{4.18}$$

And,

$$\frac{dp_\ell}{dt} = h_\ell(p_\ell)(y_\ell - c_\ell)^+_{p_\ell}. \tag{4.19}$$

It is worth to mention that various attempts were made to analyse different flavours of the above algorithms, for example, the behaviour of wireless links in a mobile environment has been represented as a a time varying capacities, i.e. by making c_ℓ function of time $c_\ell(t)$, in a dual algorithm [105], stability analysis has also been provided for the algorithm with a logarithmic utility function. We have not touched discrete-time versions, however it has been mentioned [88, p.28] that others have proven stability for a discrete-time versions of the dual algorithm with logarithmic utility functions.

4.2.4 Simplified Model for TCP Congestion Control Algorithm

Several models were mentioned in literature to mathematically approximate and formulate the behaviour of TCP, a summary of the most common approaches in TCP modelling is mentioned in [51]. The common thing between these models is that they try to capture the dynamics of TCP and to certain extent the statistics of the environment, the former point needs good understanding of the operations of TCP (the reader can refer to [90] for details of the operation of TCP protocol), the later point usually requires assumptions of stochastic events, such as the type of loss process. One of the most common models, is the detailed packet loss model [78], the periodic model [72]. Others are the stochastic model [6], network system model [60], control system model with stochastic differential equations [74]. Markov chain models [101] in [51].

A detailed control system models is provided by [74], it takes into account both phases of TCP operation: *slow start* and *congestion avoidance*, the effect of time-out back off taken by TCP and the limitation of maximum window size (advertised by the receiver). In this section we will mention a simplified model that describes the operation of the TCP in steady state congestion avoidance phase and later on in this chapter we will build on that by modifying the model and using it to analyse the stability of TCP in the presence of non-congestive loss. We use the

following equation to describe the original TCP-Reno congestion control:

$$\frac{dW_r}{dt} = \overbrace{\frac{x_r(t-RTTr)}{W_r}(1-q_r(t))}^{additive-increase} - \overbrace{\beta W_r(t)x_r(t-RTT_r)q_r(t)}^{multiplicative-decrease} \tag{4.20}$$

Where, q_r is the probability of packet loss at a function of time, W_r,x_r,RTT_r are the congestion window, rate and round trip time of flow r respectively. Note the time shift in the quantities by RTT_r, this is due to packet loss being detected after one RTT from the time when it actually happened. Equation 4.20 can be re-written in terms of the rate using the fact that $x_r = W_r/RTT_r$:

$$\frac{dx_r}{dt} = \left(\frac{x_r(t-RTT_r)}{RTT_r^2 x_r(t)} + \beta x_r(t-RTT_r)x_r(t)\right)\left(\frac{1}{\beta RTT_r^2 x_r^2(t)+1} - q_r(t)\right) \tag{4.21}$$

At equilibrium there is no change in the average rate, i.e. $dx_r/dt = 0$, thus equation 4.21 reduces to:

$$\hat{x}_r = \frac{1}{RTT_r}\sqrt{\frac{1-\hat{q}_r}{\beta\hat{q}}}. \tag{4.22}$$

For small values of q_r:

$$\hat{x}_r \approx \frac{1}{RTT_r}\sqrt{\frac{1}{\beta\hat{q}}}.$$

This is the well known expression of TCP-Reno average throughput (or user rate), which is approached by different methods and assumptions in literature.

It is easy to see that equation 4.21 is similar to equation 4.11, this result is mentioned in literature [88], we note that the first derivative of the utility function with respect to the source rate is:

$$U_r'(x_r) = \frac{1}{\beta RTT_r^2 x_r^2(t)+1}. \tag{4.23}$$

Using the fact that: $\int \frac{1}{x^2+a^2}dx = \frac{1}{a}tan^{-1}\frac{x}{a}$, we get:

$$U_r(x_r) = \frac{tan^{-1}\left(x_r(t)RTT_r\sqrt{\beta}\right)}{\sqrt{\beta}RTT_r}. \tag{4.24}$$

In the next section we show our contribution of using this model to represent a modified TCP-Reno which we also analyse its stability in the presence of non-congestive loss.

4.3 TCP-Model Revisited: Non-Linear Case

We start by investigating the effect of general variable MD on the stability of a modified version of the original TCP model. It has been shown that the convergence attribute of TCP congestion control algorithms is inherited from the stability of the primal algorithm [88]. The same work ([88]) proves the convergence of the primal algorithm using the *Lyapunov stability theory*.

Consider a user's TCP flow r, which has a round trip time[7] of RTT_r, a probability of window reduction incident due to congestion of $q_{rc}(t)$, and rate of $x_r(t)$. In this work, we add a probability of window reduction incident due to link error $q_{re}(t)$. Note that $q_{re}(t) = 0$ for a congestion control algorithm that does not reduce its rate in response to non-congestive packet loss. This, however, is not the case for TCP therefore in our context $q_{re}(t) \neq 0$.

The equation which describes TCP's dynamics in the steady-state congestion avoidance phase has two parts: i) linear AI of one segment each round trip time ($1/RTT$) and ii) MD by the new factor β' (in case of packet loss). Taking into consideration the new factor for link errors ($q_{re}(t)$), this expression can be rewritten as:

$$\begin{aligned} \frac{dx_r}{dt} &= \frac{x_r(t-RTT_r)(1-(q_{rc}(t)+q_{re}(t)))}{RTT_r^2 x_r(t)} - \\ & x_r(t-RTT_r)(q_{rc}(t)+q_{re}(t))\beta' x_r(t) \end{aligned} \tag{4.25}$$

And β' takes values $0 < \beta' \leq 0.5$. The primal algorithm is:

$$\frac{dx_r}{dt} = k_r\left(U_r' - q_r(t)\right) \tag{4.26}$$

Where, $k_r(.)$ and $U_r(.)$ (the utility function) are functions of x_r. Rewriting equation 4.25 to make it similar to the equation of the primal algorithm:

$$\begin{aligned} \frac{dx_r}{dt} = & \left(\frac{x_r(t-RTT_r)}{RTT_r^2 x_r(t)} + x_r(t-RTT_r)\beta' x_r(t)\right) \\ & \left(\frac{1}{RTT_r^2 x_r(t)*\beta' x_r(t)+1} - (q_{rc}(t)+q_{re}(t))\right) \end{aligned} \tag{4.27}$$

From equation 4.27, the first derivative of the utility function with respect to the source rate is:

$$U_r'(x_r) = \frac{1}{RTT_r^2 x_r(t) * \beta' x_r(t) + 1} \tag{4.28}$$

It is trivial to show that, if β' is constant, equation 4.23 is reduced to the result

[7] A constant RTT was assumed

mentioned in [88] and [70].

Following the proof of the primal algorithm, at steady-state the algorithm converges to a unique, non-zero solution. We note that the existence of the term q_{re} in the equations disturbs the algorithm's convergence; equilibrium is reached at lower rates. Assuming that loss is due to congestion, the algorithm keeps decreasing transmission rate in an attempt to converge to an equilibrium point. We believe that equation 4.27 reveals little information about the dynamics of the algorithm in the presence of non-congestive packet loss. For this reason, we linearise the model around an equilibrium point to take the analysis further.

4.4 Linearisation & Stability Analysis

We considered a simple, single link - single source scenario. We linearised the TCP congestion avoidance model in the presence of i) delay, ii) window reduction incidents due to link errors, iii) window reduction incidents due to congestion. We then repeated the analysis for a general variable MD version of the model. For a more general case, we included a constant κ that takes into account the AI factor as well. For standard TCP, $\kappa = 1/RTT^2$. In this context of analysis, RTT is assumed to be constant. The reason behind this assumption is that for proper operation of TCP, packet loss must occur. In other words, the queue is nearly full most of the time and thus the RTT is nearly constant [88]. One obstacle in linearisation is that it is usually assumed that TCP operates in the vicinity of an equilibrium point i.e. near a certain congestion event probability. While this is true when considering error-free links, it is less accurate for wireless environment. To overcome this, we assumed that TCP also operates in the vicinity of certain non-congestion event probability.

4.4.1 TCP - Constant Multiplicative Decrease

We used the following equation to describe the source rate evolution of TCP during the congestion avoidance phase:

$$\frac{dx}{dt} = \kappa x(t-T)\left(\frac{1-p(t-T)}{x(t)} - \frac{\beta}{\kappa}x(t)p(t-T)\right) \tag{4.29}$$

Where T is the RTT. Because we have window reduction incidents due to link errors as well as due to congestion, we denoted two different probabilities. Thus $p = p_c + p_e$, where p_c is the probability of a window reduction incident due to congestion and p_e is the probability of window reduction due to a link error. There

is a relationship between the source rate and p_c, we represent this by $p_c = f(x)$ (this depends on the queueing process). Substituting for p in the above equation:

$$\begin{aligned} \frac{dx}{dt} &= \kappa x(t-T)\left(\frac{1-p_c(t-T)-p_e(t-T)}{x(t)}\right. \\ &\quad \left. -\frac{\beta}{\kappa}x(t)p_c(t-T) - \frac{\beta}{\kappa}x(t)p_e(t-T)\right) \end{aligned} \tag{4.30}$$

When $\frac{dx}{dt} = 0 \rightarrow \frac{\beta}{\kappa} = \frac{1-(\hat{p}_c+\bar{p}_e)}{\hat{x}^2(\hat{p}_c+\bar{p}_e)} \rightarrow \hat{p}_c + \bar{p}_e = \frac{1}{1+\hat{x}^2\frac{\beta}{\kappa}}$.

Where, $\hat{p}_c$ and $\bar{p}_e$ are equilibrium probabilities for a window reduction incident due to a congestion and link error respectively. $\hat{x}$ is the steady-state rate.
If TCP operates near a steady-state point, we can define:
$x(t-T) = x(t) = x = \hat{x} + \delta x$,

$x^2 = \hat{x} + 2\hat{x} + \cancelto{0}{(\delta x)^2}$,

$p_c(t-T) = \hat{p}_c + \delta p_c(t-T)$,

$p_e(t-T) = \bar{p}_e + \delta p_e(t-T)$.

Substituting in equation 4.30 and neglecting higher-order terms:

$$\begin{aligned} \frac{dx}{dt} = \quad & \kappa\Big(\frac{-\delta x}{\hat{x}}(1-(\hat{p}_c+\bar{p}_e)) - \delta p_c(t-T) - \frac{\beta}{\kappa}\hat{x}^2 \\ & \delta p_c(t-T) - \delta p_e(t-T) - \frac{\beta}{\kappa}\hat{x}^2\delta p_e(t-T)\Big) \end{aligned} \tag{4.31}$$

We linearise the relationship $p_c = f(x)$ using the chain rule $\delta p_c = f'(\hat{x})\delta x$, where $f'(.)$ is the first derivative of $f(.)$ with respect to x. Then, taking the *Laplace*

Transform, $\mathscr{L}\{.\}$ yields: $p_c(s) = \hat{p}'_c X(s)$. And for equation 4.31:

$$\begin{aligned}
sX(s) - x_0 &= \tfrac{-2\kappa}{\hat{x}}(1 - (\hat{p}_c + \bar{p}_e))X(s) - \kappa e^{-Ts} p_c(s) - \kappa e^{-Ts} p_e(s) \\
&\quad -e^{-Ts}\beta\hat{x}^2 p_c(s) - e^{-Ts}\beta\hat{x}^2 p_e(s) \\
&= \tfrac{-\kappa}{\hat{x}}(1 - (\hat{p}_c + \bar{p}_e))X(s) - \tfrac{\kappa}{\hat{x}}(1 - (\hat{p}_c + \bar{p}_e))X(s) \\
&\quad -\kappa e^{-Ts} p_c(s) - \kappa e^{-Ts} p_e(s) - e^{-Ts}\beta\hat{x}^2 p_c(s) \\
&\quad -e^{-Ts}\beta\hat{x}^2 p_e(s) \\
&= \tfrac{-\kappa}{\hat{x}}(1 - (\hat{p}_c + \bar{p}_e))X(s) - (\hat{p}_c + \bar{p}_e)\hat{x}\beta X(s) \\
&\quad -\kappa e^{-Ts} p_c(s) - \kappa e^{-Ts} p_e(s) - e^{-Ts}\beta\hat{x}^2 p_c(s) \\
&\quad -e^{-Ts}\beta\hat{x}^2 p_e(s) \\
x_0 &= (s + \tfrac{\kappa}{\hat{x}}(1 - (\hat{p}_c + \bar{p}_e)) + (\hat{p}_c + \bar{p}_e)\hat{x}\beta)X(s) \\
&\quad +\kappa e^{-Ts}(1 + \tfrac{\beta}{\kappa}\hat{x}^2) \underbrace{p_c(s)}_{controlled} + \kappa e^{-Ts}(1 + \tfrac{\beta}{\kappa}\hat{x}^2) \underbrace{p_e(s)}_{disturbance}
\end{aligned}$$

Rearranging terms and using $p_c(s) = \hat{p}'_c X(s)$, we end up with a system with a *characteristic equation*:

$$1 + G(s)H(s) = 0$$

Where,

$$\begin{aligned}
G(s) &= \tfrac{1}{s + \frac{\kappa}{\hat{x}}(1 - (\hat{p}_c + \bar{p}_e)) + (\hat{p}_c + \bar{p}_e)\hat{x}\beta} \\
H(s) &= \kappa \tfrac{e^{-Ts}\hat{p}'_c}{\hat{p}_c + \bar{p}_e} \\
G(s)H(s) &= \tfrac{\kappa e^{-Ts}\hat{p}'_c}{(\hat{p}_c + \bar{p}_e)(s + \frac{\kappa}{\hat{x}}(1 - (\hat{p}_c + \bar{p}_e)) + (\hat{p}_c + \bar{p}_e)\hat{x}\beta)}
\end{aligned}$$

Figure 4.4 shows the arrangement of $G(s)$ and $H(s)$ in a block diagram. Note that only window reduction incidents due to congestion are part of the feedback and within TCP's control, while window reduction incidents due to link errors are considered external or *disturbance*. Concerning stability, we can apply the *Nyquist Criterion* to $G(j\omega)H(j\omega)$. In other words $G(j\omega)H(j\omega)$ should not encircle the point -1. We can rely on the following lemma [88]:

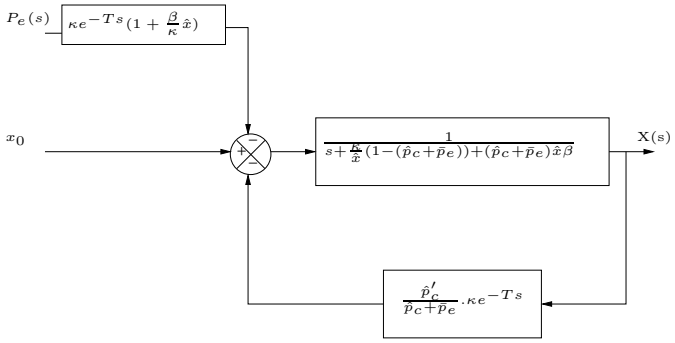


Figure 4.4: TCP with constant β, after linearisation.

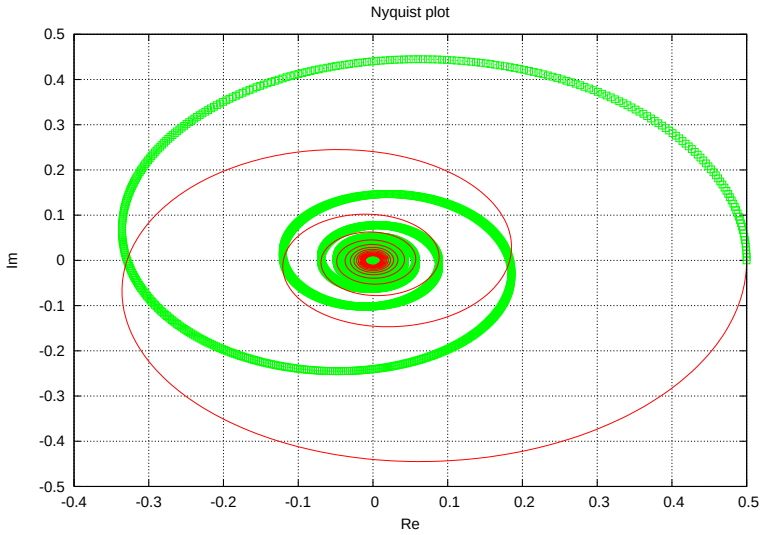


Figure 4.5: Nyquist Plot for $h(\omega)$, as $\omega \to -\infty$ (upper plot) $\omega \to \infty$ (lower plot), $K = 1, \alpha = 2$.

LEMMA 1. *As ω is varied from $-\infty$ to ∞, $h(\omega) := \frac{e^{-j\omega T}}{\alpha + j\omega T}$ does not encircle the point $-2/\pi$.*

Figure 4.5 is a visualisation of this condition. Let:

$$\begin{aligned} K &= \frac{\kappa \hat{p}'_c}{(\hat{p}_c + \bar{p}_e)} \\ \alpha &= \frac{\kappa}{\hat{x}}(1 - (\hat{p}_c + \bar{p}_e)) + (\hat{p}_c + \bar{p}_e)\hat{x}\beta \end{aligned}$$

Substituting $s = j\omega$, the stability criterion becomes:

$$\begin{aligned} \Re\{G(j\omega)H(j\omega)\} &> -K\frac{2}{\pi} \\ -K\frac{2}{\pi} &< 1 \; (Nyquist - condition) \\ T\kappa\frac{\hat{p}'_c}{\hat{p}_c + \bar{p}_e} &< \frac{\pi}{2} \\ T\kappa(1 + \hat{x}^2\frac{\beta}{\kappa})\hat{p}'_c &< \frac{\pi}{2} \\ (1 + \hat{x}^2\frac{\beta}{\kappa}) &< \frac{\pi}{2\kappa\hat{p}'_c T} \\ T &< 1/(\frac{2\kappa\hat{p}'_c}{\pi}(\frac{\beta}{\kappa}\hat{x}^2 + 1)) \end{aligned} \tag{4.32}$$

In other words, the system is stable if the RTT (T) is lower than the right term of inequality 4.32. The importance of this result becomes clearer in the next subsection.

4.4.2 Modified TCP - Variable Multiplicative Decrease

Let $b = \beta' x(t)$. Equation 4.30 can then be rewritten as:

$$\begin{aligned}\frac{dx}{dt} = \; & \kappa x(t-T)\left(\frac{1-p_c(t-T)-p_e(t-T)}{x(t)} - \frac{b}{\kappa}p_c(t-T)\right. \\ & \left. -\frac{b}{\kappa}p_e(t-T)\right)\end{aligned} \tag{4.33}$$

When $\frac{dx}{dt} = 0 \rightarrow \frac{\bar{b}}{\kappa} = \frac{1-(\hat{p}_c+\bar{p}_e)}{\hat{x}(\hat{p}_c+\bar{p}_e)} \rightarrow \hat{p}_c + \bar{p}_e = \frac{1}{1+\hat{x}\frac{b}{\kappa}}$.

Recall that TCP operates near a steady-state point, thus we can define:

$$\begin{aligned} x(t-T) &= x(t) = x = \hat{x} + \delta x \\ p_c(t-T) &= \hat{p}_c + \delta p_c(t-T) \\ p_e(t-T) &= \bar{p}_e + \delta p_e(t-T) \\ b(t-T) &= \bar{b} + \delta b(t-T) \end{aligned}$$

Substituting in equation 4.33 and neglecting higher order terms:

$$\begin{aligned}\frac{dx}{dt} = \; & \kappa\Big(- \delta p_c(t-T) - \delta p_e(t-T) - (\hat{x}\frac{\bar{b}}{\kappa}\delta\hat{p}_c \\ & \delta p_c(t-T) + \hat{x}\hat{p}_c\delta b(t-T) + \frac{\bar{b}}{\kappa}\hat{p}_c\delta x + \\ & \hat{x}\frac{\bar{b}}{\kappa}\delta p_e(t-T) + \hat{x}\bar{p}_e\delta b(t-T) + \frac{\bar{b}}{\kappa}\bar{p}_e\delta x)\Big)\end{aligned} \tag{4.34}$$

Taking the Laplace Transform of both sides, $\mathscr{L}\{.\}$ yields: $sX(s) - x_0 =$

$$\begin{aligned}& \kappa\Big(- (1+\hat{x}\frac{\bar{b}}{\kappa})\hat{p}'_c e^{-Ts}X(s) - (1+\hat{x}\frac{\bar{b}}{\kappa})e^{-Ts}p_e(s) - \\ & \frac{(\hat{p}_c+\bar{p}_e)}{\kappa}\hat{x}e^{-Ts}b(s)\Big)\end{aligned}$$

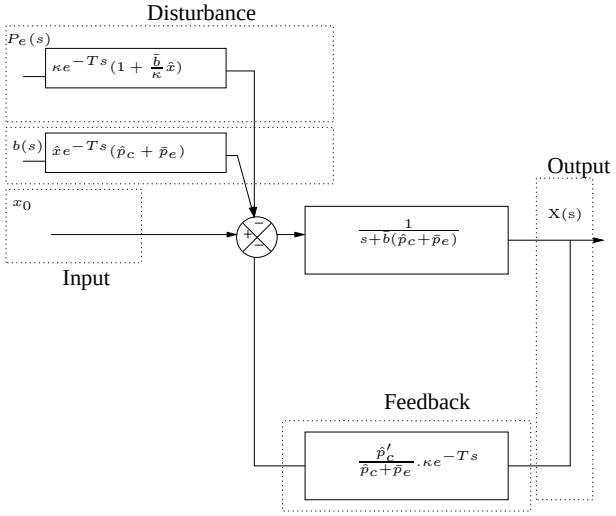


Figure 4.6: TCP with variable β', after linearisation.

$$x_0 = \Big(s + \kappa(1+\hat{x}\frac{\bar{b}}{\kappa})\hat{p}'_c e^{-Ts} + (\hat{p}_c+\bar{p}_e)\bar{b}\Big)X(s) +$$

$$\kappa(1+\hat{x}\frac{\bar{b}}{\kappa})e^{-Ts}p_e(s) + (\hat{p}_c+\bar{p}_e)\hat{x}e^{-Ts}b(s)$$

Rearranging terms and using $p_c(s) = \hat{p}'_c X(s)$, we end up with a system with a *characteristic equation*:

$$1 + G(s)H(s) = 0$$

Where,

$$\begin{aligned} G(s) &= \kappa(1+\hat{x}\tfrac{\bar{b}}{\kappa})\hat{p}'_c e^{-Ts} \\ H(s) &= \tfrac{1}{s+(\hat{p}_c+\bar{p}_e)\bar{b}} \\ G(s)H(s) &= \tfrac{\kappa(1+\hat{x}\frac{\bar{b}}{\kappa})\hat{p}'_c e^{-Ts}}{s+(\hat{p}_c+\bar{p}_e)\bar{b}} \end{aligned}$$

Figure 4.6 depicts a block diagram of $G(s)$ and $H(s)$. We note the existence of the additional term $b(s)$, which is not considered as disturbance because it is part of the input. Regarding stability, we again apply the *Nyquist Criterion* to $G(j\omega)H(j\omega)$,
Let:

$$\begin{aligned} K &= \kappa(1+\hat{x}\tfrac{\bar{b}}{\kappa})\hat{p}'_c \\ \alpha &= (\hat{p}_c+\bar{p}_e)\bar{b} \end{aligned}$$

Substituting $s = j\omega$, the stability criterion becomes:

$$\begin{aligned}
\Re\{G(j\omega)H(j\omega)\} &> -K\tfrac{2}{\pi} \\
-K\tfrac{2}{\pi} &< 1\ (Nyquist - condition) \\
T\kappa\tfrac{\hat{p}'_c}{\hat{p}_c+\bar{p}_e} &< \tfrac{\pi}{2} \\
T\kappa\tfrac{\hat{p}'_c}{\hat{p}_c+\bar{p}_e} &< \tfrac{\pi}{2} \qquad (4.35) \\
T\kappa(1+\hat{x}\tfrac{\bar{b}}{\kappa})\hat{p}'_c &< \tfrac{\pi}{2} \\
(1+\hat{x}\tfrac{\bar{b}}{\kappa}) &< \tfrac{\pi}{2\kappa\hat{p}'_c T} \\
T &< 1/(\tfrac{2\kappa\hat{p}'_c}{\pi}(\tfrac{\bar{b}}{\kappa}\hat{x}+1))
\end{aligned}$$

The expected value of MD factor ($\bar{b}$) in inequality 4.35 is lower than the original fixed value (β) used in inequality 4.32 (i.e. $\bar{b} \leq \beta$). Therefore, the right-hand term of inequality 4.35 is greater than the right-hand term of inequality 4.32. Thus, the system will be stable for a wider range of RTT (T) values compared to the system discussed in subsection 4.4.1. We view this as a *relax* in the stability condition, which leads to the conclusion that TCP can achieve better link utilisation if the MD factor is adjusted according to the type of loss (congestion or wireless link errors).

4.5 Summary

In this chapter we provided a discussion of the congestion control problem formulated as a resource allocation problem, we briefly mentioned the types of algorithms used to solve this resource allocation problem and how this is related to TCP congestion control. Our contribution in this chapter is the stability analysis of two versions of TCP-Reno in the presence of non-congestive packet loss, such losses can be caused by error-prone links e.g. optical fibre or wireless links. The first version is the original algorithm, the other version takes variable MD into account. We modified the TCP-Reno model and analytically investigated the stability of the TCP congestion control mechanism. Our analysis extends existing work using control-theoretic techniques and taking into account two additional factors: i) link errors and ii) general MD. We showed that, in the context of control theory, link errors are considered *disturbance*. Afterwards, by examining the *Nyquist Stability*

Criterion we showed that adapting the MD factor according to the type of loss (congestion or wireless link errors) makes the system (single-flow single-link case) more stable at higher delays. To be more specific, the right-hand term of the inequality of the final result in subsection 4.4.1 is lower than the right-hand term of the final result in subsection 4.4.2. This relaxes the stability condition for an algorithm that uses a variable MD. In other words, one such algorithm is expected to be more stable at higher delay values than an algorithm using a fixed MD.

However, this result is general and theoretical, because of the assumptions made in section 4.4. The analysis also assumes the existence of an ideal algorithm, capable of differentiating between loss due to congestion and loss due to other reasons and then adjusting lower MD in the former case and higher MD in the later case. However, under those two commonly encountered assumptions, our work provides better understanding of TCP's congestion control.

Chapter 5

Congestion Control Metrics

Performance evaluation of congestion control algorithms is an important issue, especially in judgement of new proposals which are to be integrated into existing protocols e.g. TCP; before being actually put into action in real environments. In order to achieve this, researchers have used a number of metrics. It is important to mention that there is not necessarily a consensus with the research community about the metrics that a congestion control algorithm should be designed to optimise [34]. However, according to the same source, there is a clear consensus that congestion control algorithms should be evaluated in terms of 'trade-offs' between certain metrics rather than in terms of optimising for a single metric. In light of that, this chapter provides a discussion of metrics relevant to evaluation of congestion control algorithms. Some of the metrics mentioned in this chapter were used in our experiments.

The rest of the chapter is divided as follows: in sections 5.1 - 5.3 we present definitions of some elementary metrics. In section 5.4 we discuss a number of convergence metrics, followed by a discussion of fairness metrics in section 5.5. We discuss metrics used to measure backward compatibility with standard protocols with focus on TCP in section 5.6. In section 5.7 we divide the metrics used to measure response to change into: i) Transient response metrics, ii) Response to packet loss metrics and discuss both sets separately. We briefly discuss metrics related to oscillation and robustness in section 5.8 and section 5.9 because they received little attention in our experiments. Finally, we sum-up by discussing trade-offs between a number of desired characteristics in section 5.10 followed by a summary in section 5.11.

5.1 Throughput

Throughput can be measured as the rate in bytes per second, it can be a network-based metric e.g. aggregate rate of flows, or flow-based metric, e.g. per-flow rate. It can also be measured at different time scales or as an average. A good example of measuring the throughput as a running average also called weighted moving average (i.e. low pass filtering) or short-term average, is mentioned in [68, p.3], we used the same formula in our experiments. Equation 5.1 shows a similar expression, where α is a smoothing factor. As we have seen in chapter 4, maximising the throughput subject to network constraints is of concern in computer and communication networks. Another way at looking at this, is by minimising the file transfer time per connection.

$$(Smoothed\ \ Thr.)_{new} = \alpha \times \frac{Number\ \ of\ \ bytes\ \ sent}{Period\ \ of\ \ time} + (1-\alpha) \times (Smoothed\ \ Thr.)_{old} \tag{5.1}$$

A related metric, which is usually thought of as the effective throughput and used in performance studies of error-prone environments, is the goodput. Goodput is a subset of throughput, it excludes dropped and duplicate packets, etc,[34]. It is also referred to as application layer throughput [10]. Thus goodput is a measure of useful (for the application) bytes per second. Equations 5.2 - 5.4, are expressions for long-term average values. If an accurate measure of goodput is to be considered, packet headers should be excluded, since they are part of lower layers communication overhead. However, in our experiments we did not take this accurate measure. It is also possible to combine the two metrics as a goodput to throughput ratio which may give a good indication of efficiency. For example, in wireless networks this may indicate the efficient use of radio spectrum.

$$Throughput = \frac{Total\ \ number\ \ of\ \ sent\ \ bytes}{Total\ \ transfer\ \ time} \tag{5.2}$$

$$Goodput = \frac{Total\ \ number\ \ of\ \ useful\ \ (to\ \ application)\ \ bytes}{Total\ \ transfer\ \ time} \tag{5.3}$$

$$Goodput\ \ ratio = \frac{Total\ \ number\ \ of\ \ useful\ \ (to\ \ application)\ \ bytes}{Total\ \ number\ \ of\ \ bytes\ \ transmitted\ \ by\ \ the\ \ sender} \leq 1 \tag{5.4}$$

5.2 Delay

Delay in communication networks can be caused by several reasons: propagation time, processing time, etc. However, this could be a flow-based metric measured as of per-packet transfer time, or network-based metric measured as queueing delay over time. The former can of significant concern in both, bulk-transfer (affects the throughput) and streaming media [34]. Similar to throughput, the delay can be measured at different time scales or as an average. Besides that, as we have seen in chapter 3, maximum delay or worst-case delay could be a useful metric.

In our experiments, we considered the network-based delay metric, we used the round trip time to estimate the queueing delay from the receiver side, in an approach similar to the approaches mentioned in [69, 9]. Note that in the absence of any congestion i.e. queueing, the round trip time forms the base round trip time, on the other hand larger round trip times, can carry information about queueing delay and the difference between the two can be used to estimate the queueing delay. In some of our `ns2` simulation experiments, we probed the bottleneck queue directly.

5.3 Packet Loss Rates

Packet loss rates are usually measured as:

$$p = \frac{Total\ number\ of\ lost\ packets}{Total\ number\ of\ sent\ packets} \tag{5.5}$$

And can be network-based or flow-based metrics [34].

Packet loss rate gains its importance as a metric from the following two points: firstly, it can be used as a direct metric which indicates the efficiency of the environment e.g. efficient use of the medium, or for example the number of retransmissions and its effect on the sender's energy in a wireless environment. Just to link ideas, we mention that the goodput ratio ideally should be equal to $1 - p$, where p is the loss rate [10]. Packet loss rate is usually expressed as a fraction of sent packets (equation 5.5), Note that in this case, both metrics indicates the efficiency of environment as mentioned before when the goodput ratio was discussed.

Secondly, packet loss rates can affect per-connection transfer times, in the sense that congestion control algorithms usually use packet loss/mark information as an indication of congestion and react by reducing their rates, and thus reducing their throughput. Although different algorithms react differently upon packet loss/mark and after that when they increase their rates, packet loss rates can be used indirectly to indicate per-connection transfer times or throughput, especially

when comparing flows running the same algorithm. In chapter 8, differentiating packet loss type was considered, our simulation scripts calculate two different types of packet loss rates.

5.4 Convergence

An important type of congestion control metric is that of convergence times. This metric is of special concern for flows in large bandwidth delay product pipes, it concerns the time for convergence to fairness between an existing flow and a newly starting one; or the convergence to fairness after a sudden change, e.g. network path, competing cross traffic, wireless-link characteristics [34]. Convergence times could be measured between flows running the same congestion control algorithm or between flows running different algorithms [34].

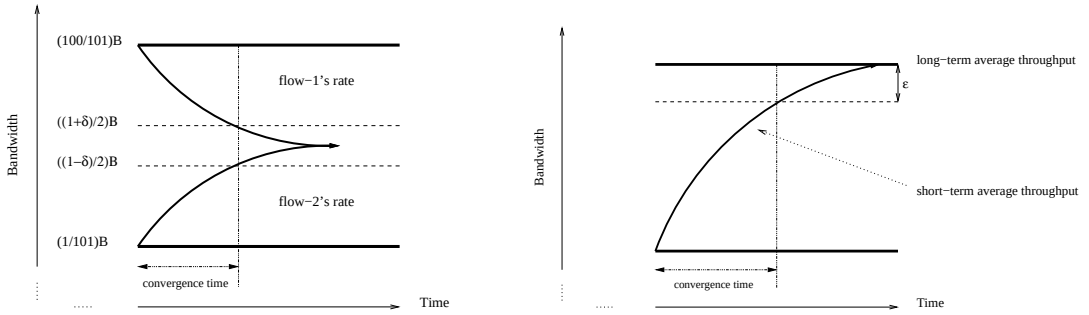
From literature, we focused on two ways to measure convergence times: Delta-Fair convergence [34] and epsilon convergence times [68].

1. Delta-Fair convergence: is defined as the time taken for two flows with the same round trip time to go from shares of 100/101 -th and 1/101-th of the link bandwidth, to having close to fair sharing with shares of $(1+\delta)/2$ and $(1-\delta)/2$ of the link bandwidth. Figure 5.1a illustrates this definition, where B is the link bandwidth, flow rates can be measured as short-term average throughputs. The choice of δ is an arbitrary small number, we choose the values between 0.1 and 0.2, this maps to 80%-90% of the fair share, these numbers were concluded from Epsilon-Fairness (discussed below) experiments [68].

2. Epsilon convergence: following the start up of a new flow, is defined as the time before a short-term average throughput of the new flow is within a factor ϵ of its long-term average throughput value. A value of ϵ between 0.8 and 0.9 yields 80%-90% convergence times. Figure 5.1b illustrates this definition.

5.5 Fairness

It is important to consider the case of multiple flows when developing congestion control algorithms, whether running the same algorithm (Intra-protocol fairness) or different algorithms (Inter-protocol fairness), the question here is: how a flow affects other flows and how other flows affect a flow in terms of throughput? The



(a) Definition of delta convergence time (b) Definition of epsilon convergence time

Figure 5.1: Convergence Times

fairness metric is usually used as an index of how fair a resource (bandwidth) is used among a number of flows.

Depending on the resource requirements, fairness measures can be classified into two categories [34](discussed below)

5.5.1 General Fairness Between Flows

Fairness measures here do not take path characteristics into account, e.g. the number of links in a path, or different round trip times. But this does not mean that they cannot be used to measure the effect of different path characteristics on fairness between flows. For example, if an experiment was conducted using flows running the same congestion control algorithm, all path characteristics are the same and the fairness measure indicates unfairness, then one possible interpretation is that the algorithm is not sharing the resource efficiently. However, if for example the flows have different round trip times, then the fairness measure will indicate unfairness, but the interpretation will not be clear i.e. whether the unfairness is completely caused by the difference in round trip times or both difference in round trip times and the algorithm. The same applies if an experiment of flows running different algorithms were run, in fact the situation may be worse in this case, since there are three possible choices for the cause of unfairness.

On the other hand, if a congestion control algorithm is known to share resources equally between flows having similar path characteristics, then changing one path characteristic measures the fairness/unfairness with respect to that parameter e.g. if RTT is changed this measures RTT-unfairness.

We mention three measures. In the discussion the j^{th} throughput is denoted by x_j.

1. Jain fairness index:

$$f(x_1, x_2, \ldots, x_n) = \frac{\left(\sum_{j=1}^{n} x_j\right)^2}{n \sum_{j=1}^{n} (x_j)^2} \qquad 0 \leq f \leq 1 \tag{5.6}$$

Where,
n : number of simultaneous connections.
x_j: j^{th} connection throughput.
f : fairness index

From equation 5.6, it can be seen that, when k users equally share a resource and the other $n - k$ receive nothing of the resource, the index is k/n, thus, when all flows receive the same share, Jain's index is at maximum with a value of 1.

2. Network power: is defined as the product measure, however because x_j is sometimes referred to as the power of the j^{th} flow, the measure can be referred to as the network power and expressed as:

$$n_p = \prod_{j=1}^{n} x_j \tag{5.7}$$

We note that if one flow has zero throughput the metric drops to zero, on the other hand the measure is maximum when all flows have the same throughput [34]. We can stop here and take this as a fact, however; although this may seem to be trivial, we find this a good situation to test our understanding of Lagrange multiplier method mentioned previously in chapter 4. To see how the product measure is maximum when all flows are the same, we formulate the problem as an optimisation problem as follows: assume we have three flows with rates of, x_1, x_2, x_3 competing for bandwidth of a link with a capacity C then we can write,

$$\max(x_1 \times x_2 \times x_3)$$

subject to:

$$x_1 + x_2 + x_3 \leq C$$

And:

$$L(x_1, x_2, x_3, \lambda) = x_1 \times x_2 \times x_3 - \lambda \times (x_1 + x_2 + x_3 - C).$$

Then:

$$\begin{aligned} \frac{\partial L}{\partial x_1} = 0 \rightarrow \quad & x_2 \times x_3 - \lambda = 0, \\ \frac{\partial L}{\partial x_2} = 0 \rightarrow \quad & x_1 \times x_3 - \lambda = 0, \\ \frac{\partial L}{\partial x_3} = 0 \rightarrow \quad & x_1 \times x_2 - \lambda = 0. \end{aligned}$$

Thus it follows that $x_1 = x_2 = x_3 = C/3$. This can be extended to n flows as: $\max \prod_{j=1}^{n} x_j$, subject to $\sum_{j=1}^{n} x_j \leq C$ and $x_j \geq 0$. The solution is then $x_j = C/n$.

3. Epsilon Fairness: is measured as the ratio between minimum and maximum throughputs, a rate allocation among a set of flows is considered ϵ-fair if:

$$\frac{\min_j x_j}{\max_j x_j} \geq (1 - \epsilon) \tag{5.8}$$

This could be thought of as measuring the worst-case ratio. Figure 5.2 illustrates the idea graphically.

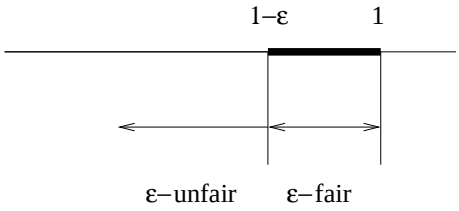


Figure 5.2: Epsilon fairness

4. Asymmetry index: this measure gives more information about the competing flows than the aforementioned measures, particularly, it gives information about which flow larger share of bandwidth, it is formulated for two flows and originally defined by [49], and has been used in performance studies (e.g. [99]):

$$A = \frac{\hat{x}_1 - \hat{x}_2}{\hat{x}_1 + \hat{x}_2} \tag{5.9}$$

Where, $\hat{x}$ is the average throughput. Note that when $A = 0$ both flows have the same share, $A = 1$ first flow has larger share, $A = -1$ second flow has larger share

This could be thought of as measuring the worst-case ratio. Figure 5.2 illustrates the idea graphically.

5.5.2 Flows with Different Resource Requirements

There are a number of fairness measures that account for flows with different path characteristics, like for example the number of links in a path or different round trip times

1. Max-Min fairness: informally, this can be defined as each user's throughput is at least as large as that of all other users which have the same bottleneck [45]. Max-min fairness[1] could be explained by the progressive filling algorithm, where all flows start at zero and all the rates grow at the same pace, each flow rate stops growing only when one or more links on the path reach its capacity [34]. From literature [45, p.18] we found a good way to understand this is to examine this fairness macroscopically along the path (global view), or on per-link basis (local view) and to translate from locally fair allocation to globally fair one, each flow should limit its resource usage to the smallest locally fair allocation along its path; this result in a globally fair allocation.

   To have a closer look at max-min fairness we implemented the algorithm and run a simple test: 48 sources using 9 links as defined in table 5.1. The output is shown in table 5.2, where U indicates that the link is underutilised and B indicates bottleneck for the listed sources. We note from table 5.2 that each source has a bottleneck link, this agrees with the Lemma mentioned in [88, p.11], which states that a set of rates is max-min fair if and only if every source has a bottleneck link.

2. Proportional fairness: A feasible allocation, $\hat{\mathbf{x}}$, is defined as proportional fair if, for any other feasible allocation $\mathbf{x}$ the aggregate of proportional changes is less than or equal to zero:

$$\sum_{j=1}^{n} \frac{x_j - \hat{x}_j}{\hat{x}_j} \leq 0$$

   What does this mean from an an optimisation[2] point of view? If we have a convex utility function say: $U_r(x_r)$, then from the property of convex functions [88, p.9]:

$$\sum_{r \in S} U'_r(\hat{x}_r)(x_r - \hat{x}_r) \leq 0$$

[1] A generalization of is weighted max-min fairness where the weight reflects the right for relative resource share for each user

[2] chapter 4 notations apply.

Table 5.1: Max-Min Fairness Test

Link	C_0	C_1	C_2	C_3	C_4	C_5	C_6	C_7	C_8
Capacity	3	4	8	10	6	7	8	11	6
Sources Used	1 8-10	1,3-4, 6,8-10	1,3-4, 6,8-10, 26	1,3-4, 6,8,10 12,44	1,3,5-6,8, 11,13,18, 25,27,30, 44-45,48	1,3,5-7,11 13-14,18, 27,31,34, 44-45,48	1-3,5-7, 11,13-14, 16,18-19, 28,33-34, 36,42, 44-45,48	2-3,6, 11,14, 16-21,24, 28-29,32-36, 42-44,46-48	2,11,14-23, 32-43,46-48

Table 5.2: Link utilisation, U: under-utilised, B: bottleneck

Link	C_0	C_1	C_2	C_3	C_4	C_5	C_6	C_7	C_8
Status	U	B	B	B	B	B	B	B	B
Share	–	0.69	4.00	6.20	0.48	1.01	1.16	2.08	0.22
Sources	–	4,9-10	26	12	1,3,5-6,8, 13,25-27, 30,44,45	7,31	28	24,29	2,11, 14-23, 32-43, 46-48

$\{\hat{x}_r\}$: set of optimal rates (optimal solution), $\{x_r\}$: any other set of feasible rates (feasible solution). If we choose a logarithmic function with arbitrary weight say: w_r the inequality becomes:

$$\sum_{r \in S} w_r \frac{x_r - \hat{x}_r}{\hat{x}_r} \leq 0$$

The corresponding resource allocation is weighted proportional fair or just proportional fair in case $w_r = 1$.

As we have seen in chapter 4, when solving problems related to proportional fairness, the resultant rate allocation is different from max-min fairness. It is reasonably easy to feel the fairness in max-min fairness, but why proportional fairness is considered fair? To answer this question, we take a simple example of three sources using two links link A, $C_A = 2$ and link B, $C_B = 1$ as follows: source 0 uses link A and link B, source 1 uses link A, source 2 uses link B. Assume the optimal rate allocation is: $\{0.485, 1.515, 0.515\}$. We choose a feasible solution i.e. a solution that satisfies the capacity constrains condition, say: $\{0.5, 1.5, 0.5\}$. Since: $(0.5 - 0.485)/0.485 + (1.5 - 1.515)/1.515 + (0.5 - 0.515)/0.515 = -0.02$, the optimal solution is proportional fair. This set could be considered fair if for example source 2 needs *at least* 0.515 for its application while the added benefit of getting more than 0.485 is *negligible* for source 0. In fact, this is another way of looking at fairness concept, even in general terms, I do not want to distract the reader or lead him/her outside the context, but this simple example shows that equally dividing a resource among users who share this resource may not be fair, but rather giving each user its requirement of the resource could be more fair.

3. Minimum potential delay fairness: this can be thought of as minimising the average download time of a set of flows downloading an equal-sized file. Formally, a rate allocation $\hat{\mathbf{x}}$ is minimum potential delay fair if (subject to the capacity constrains and positiveness of rate) it satisfies:

$$\max_j \sum_{j=1}^{n} \frac{-w_j}{x_j}$$

Where, w_j is the j^{th} file size. That is the average download time is smaller than any other feasible allocation.

4. Round Trip Time fairness: RTT fairness between two flows running the same congestion control algorithm, is often defined in terms of throughput

ratio. The flow with small RTT will usually get higher throughput than the flow with the large RTT, this will starve the flow with large RTT from bandwidth [39, 102]. In other words, TCP has a bias against long flows (traverse large number of links)[3] and this problem is sometimes referred to as RTT-unfairness.

Since all algorithms suffer from this problem, researchers tend to compare RTT-unfairness of congestion control algorithms with that of AIMD (as a reference), the reason behind that is that AIMD exhibit a quadratic increase in throughput ratio as RTT ratio increases which is interpreted as linear RTT-unfairness[4] i.e. like standard TCP. Hence the goal has been to make the RTT-unfairness of a congestion control algorithm as close as possible to that of AIMD, although their were attempts (e.g. TCP-Hamilton, TCP-CUBIC) to make it better by making the congestion window function independent from RTT and instead; adapting it in real time, we restrict our discussion to the algorithms that adapt as standard TCP[5]. It can be shown[6] that for a response function of the form: c/p^d, where c is constant and p is the steady state packet loss probability, the throughput ratio is:$\frac{Th_1}{Th_2} = (\frac{RTT_2}{RTT_1})^{\frac{1}{1-d}}$. We concluded from the discussion in [31, 102] that $0.5 \leq d \leq 1$, where $d = 0.5$ for AIMD. When $d = 1$ this gives a linear response function with fixed congestion epoch (period between losses) regardless of the packet loss rate. When $d > 1$ the congestion epoch is inversely proportional with the average window, for example we conclude from [31] that for $d = 2$ it should be: $1/\sqrt{wc}$ (where w is the average window). Between 0.5 and 1; as d gets smaller the congestion control algorithm operation gets closer to AIMD. The same result can be achieved by studying a `log-log` plot of the response function, where d is the slope. Small ds give an operation similar to AIMD, large ds give more scalability (could be a good choice for high-speed networks).

It is worth mentioning that there is no clear consensus in the networking community about the desirability of this goal in general [34]. In addition to that, the desire that flow with large RTT share equally resources on its end-to-end path (even though it uses more network resources along its end-to-end path) with small RTT flow may not always be considered a pragmatic approach [73], in fact this is one argument against this goal [34]

[3]There is a subtle conclusion here that TCP is proportionality fair based on our previous proportional fair example which has a bias against source 0. This is not necessary [45].

[4]The reader is to refer to appendix for the derivation of the expression.

[5]It is easy to forget a benefit of making the congestion window function of RTT: throughput decrease when congestion increases, since RTT increases and its in denominator

[6]The result has been mentioned in [31, 102], the reader is to refer to appendix for our workout of the equations.

5.5.3 Fairness in Optimisation Framework

Recently, there has been extensive work in analysing fairness in optimisation framework, [70, 8, 97] are just examples. The aim of this subsection is to link fairness metrics discussed in this chapter with the definitions mentioned in chapter 4, we are not targeting an in-depth discussion of current literature in this direction but rather an understanding of the concepts and how they are measured in the context of performance evaluation. We mentioned several fairness concepts in chapter 4, and we already mentioned some optimisation terms in our discussion of fairness in this chapter, in fact optimisation framework is one way to analyse and understand fairness concepts [34, 88]. For sake of relevance to context and in order to link ideas, we will summarise some of these results here.

The main resource allocation problem can be described by equation 4.5; using different utility functions result in different optimal solution i.e. different set of optimal rates. In other words, the choice of utility function determines what set of rates is optimal – proportional fair, minimum potential delay fair, max-min fair. A choice of logarithmic utility function, $U_r(x_r) = w_r \log x_r$, where x_r denotes the rate of user r and w is an arbitrary weight for user r, gives a weighted proportional fair set of rates. Another choice is $U_r(x_r) = -w_r/x_r$, where w_r can be interpreted as the file size for user r, and the problem degenerates to minimising the average download time i.e. rates are allocated as minimum potential delay fair. A choice of $U_r(x_r) \approx 1/(cx_r{}^c)$ for large values of the positive constant c, shows a max-min fair behaviour.

5.6 Backward Compatibility

Compatibility with existing protocols is another vital issue that needs attention, mainly existing predominant protocols such as TCP. Particularly; researchers are usually concerned about *fairness* between a flow running a congestion control algorithm with a flow running standard TCP algorithm in the so-called 'well-behaving' range of TCP, i.e loss rates of $10^{-2} - 10^{-4}$. This is sometimes referred to as 'TCP-friendliness' [102]. In other words, we do not expect standard TCP to work well (scale) at small loss rates (large windows), but we are concerned about how the a new algorithm will affect standard TCP at high loss rates (small windows). This is one primary concern when developing scalable congestion control algorithms for high-speed networks.

The term 'TCP-fairness' has been described as a new definition [34, p.11], in this definition a set of TCP fair flows do not cause more congestion than a set of TCP flows would cause, where congestion is defined in terms of queueing delay,

queueing delay variation, congestion event rate and packet loss rate.

5.7 Response to Change

We divide this into two parts: transient response and response to packet loss.

5.7.1 Transient Response

Sudden congestion in a network can occur due to bandwidth or routing changes or from a burst of competing traffic. Sudden bandwidth delay product changes are also possible, in fact, sudden changes (sometimes by several orders of magnitude [34]) in bandwidth and RTT are becoming more frequent and inevitable with the spread of mobile and heterogeneous wireless networks (WiFi, WiMAX, WCDMA, Bluetooth,...) along with the use of some techniques like vertical handover for example in mobile networks, a question that rises here is: how fast should a congestion control algorithm respond to all these changes?

A congestion control algorithm is expected to respond quickly (low response times) to sudden congestion, while at the same time avoid severe response to changes that last less than the connection RTT (e.g. sudden increase in delay) [34]. Some of the key metrics in evaluating the response to sudden or transient changes are: smoothness, responsiveness and aggressiveness. For an individual connection[7] these can be defined as:

1. Smoothness: in a deterministic environment, is defined as the largest reduction in the sending rate in one round trip time. The sending rate here can be measured each RTT or as short-term average throughput.

2. Responsiveness: is defined as the number of round trip times of sustained congestion required for the sender to halve the sending rate. Another possible measure of responsiveness is through convergence times following a sudden congestion caused by change in bandwidth. For example, epsilon convergence time of a flow following the start up of second flow could measure responsiveness. In both cases we measure the sending rate as short-term average throughput.

3. Aggressiveness: is defined as the maximum increase in the sending rate in one round trip time in packets per second, in the absence of congestion. Sending rate here can be measured each RTT or as short-term average throughput.

[7] Responsiveness and smoothness can be measured as network-based metrics.

The asymmetry index [49] (please see 5.9) can be used to measure aggressiveness, when $A = 0$ both flows have the same aggressiveness, $A = 1$ first flow is more aggressive, $A = -1$ first flow is more gentle.

5.7.2 Response to Packet Loss

Another measure for response to change is *Response function*. Response function describes the response of congestion control algorithm to packet loss, particularly; it is the average throughput as a function of packet loss. The function could be derived analytically, there are several examples mentioned in literature: for standard TCP with random packet loss [78], for deterministic analysis [36], for deterministic analysis of one of TCP high speed variants [102]. The function can also be obtained experimentally by measuring the average throughput for a span of random packet loss rates, some use this approach to measure the response to packet loss [68, p.3].

5.8 Oscillations

Two major signs of instability are: under-utilisation of link capacity and synchronised loss (as we will see later in this chapter, synchronised loss is blamed for a set of problems). These two reasons are the result of queue overflow which is caused by fluctuations in queue size due to rate fluctuations. Thus throughput or queueing delay fluctuation (or oscillation), measures stability [34].

Throughput and queueing delay fluctuations (throughput fluctuation can be measured as flow-based metric or network-based metric), can be measured by: coefficient of variation and standard deviation, both can be used at different time scales. For throughput fluctuations the smoothness metric can be used to indicate the level of oscillation. Concerning minimising oscillation in throughput or queueing delay (thus enhancing stability), relevant flow-based metrics of minimising jitter in round trip time and loss rates can be considered [34].

5.9 Robustness

Robustness of congestion control algorithm when working in different environments can be important, for example; robustness has gained more importance in the Internet architecture compared to efficiency [34]. We quote from the same source two important metrics, one of which was discussed before:

1. In mobile environment: energy consumption metric, this metric is affected by the transport protocol.

2. In wireless environment: goodput ratio metric, high values indicate efficient use of radio spectrum.

5.10 Other Issues & Trade-Offs

We first look at the impact of synchronised loss on congestion control metrics. Synchronised loss happens when packet loss occurs across multiple competing flows simultaneously [102] and it is a sign of instability [34]. Synchronised loss encourages RTT unfairness [102, 39], prolong convergence times (slower convergence) [102, 34], hurts fairness [102], cause long-term oscillation of data rates [102, 34], affects delay [34] and causes under-utilisation [102].

The problem can get even worse when considering congestion control algorithms designed for high-speed networks, for example, concerning window size, as the window gets larger, the probability it loses at least one packet grows exponentially, this is fairly simple to observe, let p be the packet loss probability, then the probability that a window gets delivered without errors is: $(1-p)(1-p)(1-p)$ ... *end of window* or $(1-p)^w$, where w is the window size. Then the probability that a window is not delivered (i.e. has at least one packet loss) is: $1-(1-p)^w$. In other words, synchronised loss can occur more frequently at large windows. It is well-known that queue management policy plays a role in mitigating synchronised loss, it has been shown that Drop Tail policy exhibits substantial synchronised loss, while for example RED policy exhibits less [102]. As a result, all problems associated with synchronised loss are mitigated when using RED policy.

On the other end of the spectrum, implementation issues also need to be considered. Practical wise, deploy-ability metrics such as: protocol overhead, ease of diagnosis and the added complexity at nodes are to be considered when developing a new congestion control algorithm. Two main aspects have been mentioned in [34] related to implementation, first the range of deployment needed i.e. sender side implementation; receiver side implementation; or both, router involvement, etc. Second is the complexity of code.

As mentioned in the first paragraph of the chapter, understanding trade-offs between congestion control metrics is important in the evaluation of congestion control metrics, in some situations the relation between metrics is not clear and depends on many factors e.g. throughput is proportional to fairness in certain topologies while inversely proportional in others [34]. Our interest however is in the effect of source behaviour on these metrics, a simple way to have a broad view of the effect of AIMD mechanism on these metrics is to look at the relation between AI (α) and MD (β) on one side and the affected metrics on the other side. Figure 5.3, illustrates this broad view. A high AI and low MD provide better

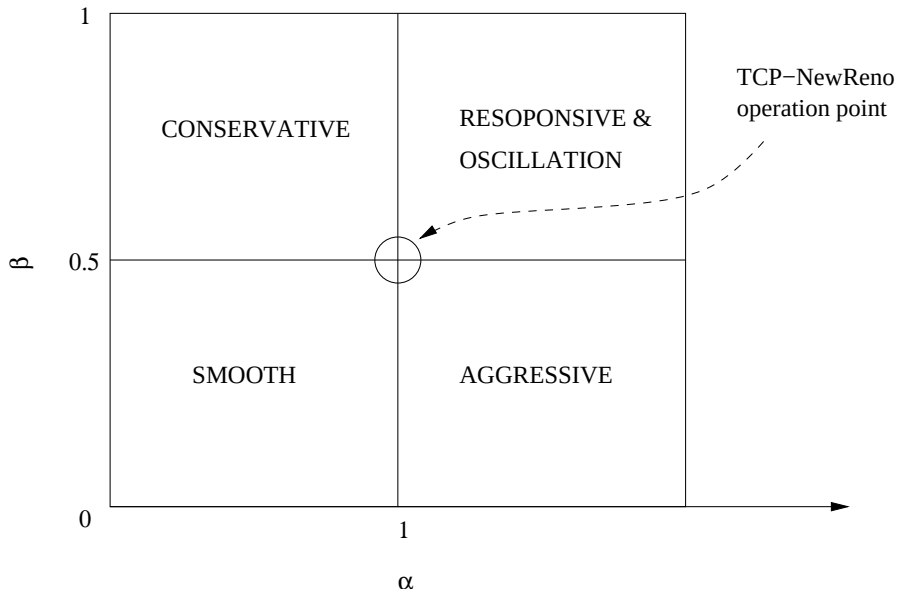


Figure 5.3: AIMD operational space

aggressiveness, this is a trade-off with conservativeness which can be achieved by low AI and high MD. On the other hand, a low AI and low MD achieve a smooth rate, this is a trade-off with high responsiveness which can be achieved by high AI and high MD. Also note that there is a trade-off within the upper right block: we achieve high responsiveness at the price of high oscillation.

We conclude from this discussion that a CC algorithm with fixed AI and MD has limitations in adapting to different working environments. A CC algorithm that varies AI and MD according to a fixed relation or function is more flexible yet the flexibility depends on this used function, thus some limitation can occur. A CC algorithm that uses multiple AI and MD that can be adapted according to different network conditions is indeed more flexible, this can be thought of as having multiple points in the four blocks in figure 5.3.

5.11 Summary

The meaning of congestion control metrics is important in performance evaluation and comparison between different congestion control algorithms. For this reason we took the time to understand the metrics mentioned in literature. We started with what we called 'elementary' metrics: throughput, delay and packet-loss rates. We show different ways to measure these metrics. We called these elementary because other metrics can be derived from them, for instance the convergence and fairness metrics which we discussed after that; depends on the throughput metric. It is quite hard to define 'fairness', however we divided the discussion into: i) General fairness metrics which are used to indicate whether the flows are

equally sharing the resources or not, ii) A definition of a number of fairness types when flows differ in resource requirements, e.g. traversing different links. Common fairness types in this case are: max-min fairness, proportional fairness, minimum potential delay fairness and RTT-fairness. iii) We view the definitions of 'ii' from an optimisation perspective, mainly a briefing of related issues from chapter 4.

We discussed issues related to backward compatibility with standard TCP, and metrics for transient response: smoothness, responsiveness and aggressiveness and how to capture response to packet loss. We briefly mentioned oscillations and robustness metrics and finalise by a discussion of some problematic issues that might hurt the performance e.g. synchronised packet loss, followed by various trade-offs for AIMD algorithms.

Chapter 6

Proposed Changes to TCP Illinois

This chapter highlights our contributions to high-speed TCP Congestion Control; we suggest an idea to improve some shortcomings in TCP-Illinois [69]. We show that higher order versions of the delay functions employed to adjust AI and MD used in [69] can provide better scalability and responsiveness.

The chapter is divided as follows: in section 6.1 we discuss our proposal for the new delay functions. In section 6.2 we define the notion of relative aggressiveness and relative responsiveness which we use to understand the improvement of the new proposal. In section 6.3 we discuss our simulation methodology which we use to validate our work, then in light of that we discuss our simulation results. Finally, we summarise in 6.4 the ideas discussed in this chapter

6.1 Higher Order Delay Functions

It has been observed that both TCP-Illinois [69](part of Linux kernel stack) and TCP-Compound [93](Default in Windows Vista SP1) can exhibit poor scaling behaviour as path BDP increases [29]. For TCP-Illinois, the reason behind that is believed to be related to linear increase which limits its scalability, and to its relatively long congestion epochs. It is well known that convergence times are proportional to congestion epochs duration, since long congestion epochs cause long convergence times which translates into prolonged unfairness. In addition to that long congestion epochs lead to sluggish responsiveness [29].

In this section we propose an alternative mechanism for varying AI and MD parameters for TCP-Illinois [69]. Particularly we consider higher order versions of the delay functions used to adjust AI and MD. The motivation behind that is to see whether or not this choice of functions provide better aggressiveness, responsiveness, Inter/Intra protocol fairness and convergence and if yes, by what magnitude.

The original algorithm [69] addresses the low throughput performance problem of TCP when working in high-speed long-delay networks. It has an adaptive AI and MD parameters that engage after a window threshold, $cwnd \geq cwnd_{thresh}$, and falls back to TCP NewReno operation (i.e. fixed AI and MD) if the window is below this threshold. The algorithm keeps tracking RTT for each ACK and average the measurements over the last window ACK to compute the average RTT, average queueing delay, maximum average queueing delay, and finally α according to the following function:

$$\alpha = f(d_a) = \begin{cases} \alpha_{max} & \text{if} \quad d_a \leq d_1 \\ \frac{\kappa_1}{\kappa_2 + d_a} & \text{otherwise} \end{cases} \tag{6.1}$$

And β according to:

$$\beta = f(d_a) = \begin{cases} \beta_{min} & \text{if} \quad d_a \leq d_2 \\ \kappa_3 + \kappa_4 * d_a & \text{if} \quad d_2 < d_a < d_3 \\ \beta_{max} & \text{otherwise} \end{cases} \tag{6.2}$$

Where, T_a: average RTT for one window,
T_{min}: minimum average RTT seen so far,
T_{max}: maximum average RTT seen so far,
d_m: maximum average queueing delay,
d_a: average queueing delay,
$d_m = T_{max} - T_{min}$,
$d_a = T_a - T_{min}$,
$d_i = \eta_i d_m \quad (i = 2, 3)$ and $0 \leq \eta_2 \leq \eta_3 \leq 0.5$,
$0 < \beta_{min} \leq \beta_{max} \leq 0.5$,
$\kappa_i, \ i = \{1, 2, 3, 4\}$,
$\kappa_1 = \frac{(d_m - d_1) * \alpha_{min} * \alpha_{max}}{\alpha_{max} - \alpha_{min}}$,
$\kappa_2 = \frac{(d_m - d_1) * \alpha_{min}}{\alpha_{max} - \alpha_{min}} - d_1$,
$\kappa_3 = (\beta_{min} * d_3 - \beta_{max} * d_2)/(d_3 - d_2)$,
$\kappa_4 = (\beta_{max} - \beta_{min})/(d_3 - d_2)$.

We note here that, for each window there should be a computed value of β and α; regardless of any window reduction incident. However, multiplicative decease takes action when the algorithm decides a window reduction incident (i.e. congestion event). Because the function uses average delay, the algorithm is believed to be immune to non-congestive packet loss.

As a step towards studying different functions, we changed the original functions mentioned in [69] and used higher order versions. The functions are expressed

as:

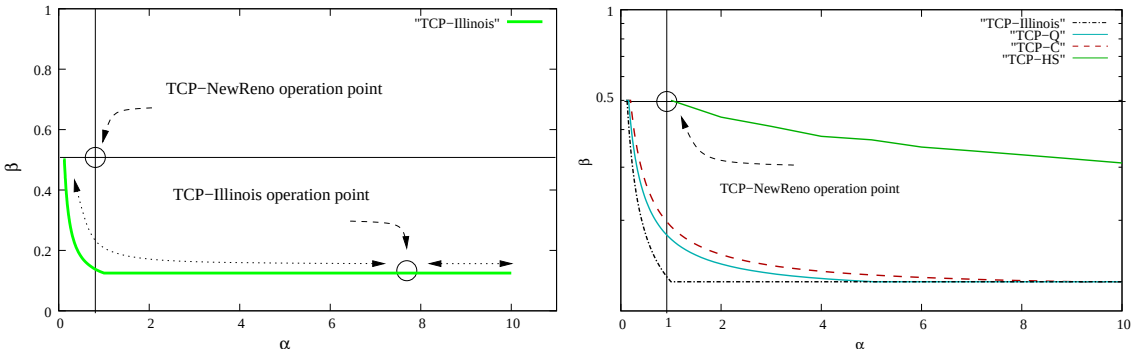
$$\alpha = f_n(d_a) = \begin{cases} \alpha_{max} & \text{if} \quad d_a \leq d_1 \\ \frac{\kappa_1(n)}{\kappa_2(n)+d_a^n} & \text{otherwise} \end{cases} \tag{6.3}$$

Where,
$n > 1$,
$\kappa_1(n) = \frac{(d_m^n - d_1^n) * \alpha_{min} * \alpha_{max}}{\alpha_{max} - \alpha_{min}}$,
$\kappa_2(n) = \frac{(d_m^n - d_1^n) * \alpha_{min}}{\alpha_{max} - \alpha_{min}} - d_1^n$.

$$\beta = f_n(d_a) = \begin{cases} \beta_{min} & \text{if} \quad d_a \leq d_2 \\ \kappa_3(n) + \kappa_4(n) * d_a^n & \text{if} \quad d_2 < d_a < d_3 \\ \beta_{max} & \text{otherwise} \end{cases} \tag{6.4}$$

Where,
$n > 1$,
$\kappa_3(n) = (\beta_{min} * d_3^n - \beta_{max} * d_2^n)/(d_3^n - d_2^n)$,
$\kappa_4(n) = (\beta_{max} - \beta_{min})/(d_3^n - d_2^n)$.

We now use a graph argument to illustrate the responsiveness[1] characteristic



(a) Original, (n=1, TCP-Illinois) (b) Higher orders, (n=2,TCP-Q, n=3, TCP-C)

Figure 6.1: Adaptive AI and MD

of higher order functions. Figure 6.1 shows the relationship between α and β for TCP-NewReno, TCP-Illinois, TCP-Q (power of 2), TCP-C (power of 3) and TCP-HS [32]. Considering TCP-NewReno as a neutral fixed point; TCP-Illinois point "slides" according to the curve in figure 6.1a, from the aggressive domain to the smooth domain. Referring to figure 6.1b, a point on the dashed or solid (the one below the dashed) curves slides faster (where "faster" translates to responsiveness in this context) from right to left compared to a point sliding on the semi-dashed (bottom) curve. Note also that the difference between the first order and second order is less that between the third and second order which makes the second order function a good representative of higher order functions[2].

[1] Where responsiveness is defined in terms of response to changes in network.
[2] End of graph argument

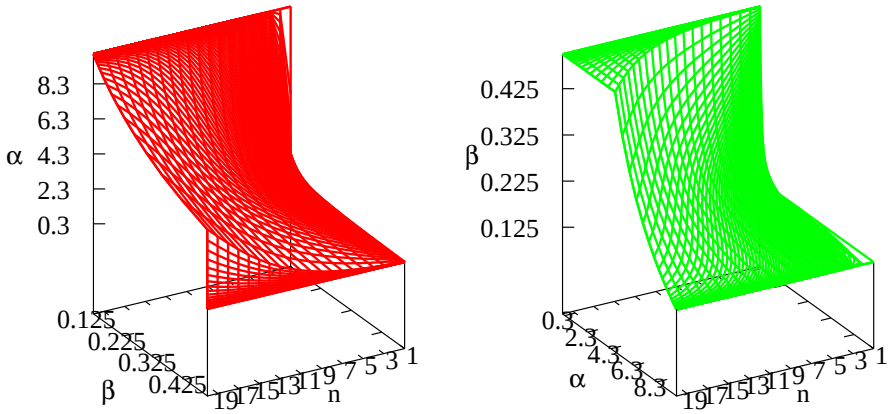


Figure 6.2: AI functions

6.2 Formal Definitions

In this section we layout our definitions of relative aggressiveness and relative responsiveness, we also aid our proofs with a visualisation of the new functions. Figure 6.2 illustrate the AI and MD as functions of normalised delay and n. It can be seen that the functions are non-decreasing, it can also be seen that all functions are upper and lower bounded by the original values of the algorithm.

6.2.1 Relative Aggressiveness

For an AIMD with variable AI and MD; both the AI parameter and the MD parameters affect aggressiveness (increase when AI is high and MD is low), however the AI parameter is updated more often especially in loss-based AIMD, thus aggressiveness is tightly linked to the AI parameter, this also agrees with the definition of aggressiveness [34]. Based on that, we state the following definition of relative aggressiveness:

DEFINITION 1. *A loss-based AIMD CC algorithm with a variable α_i and β_i compared to another loss-based AIMD CC algorithm with a variable α_j and β_j:*

- *has more aggressiveness, if for $\beta_i = \beta_j$, $\alpha_i > \alpha_j$;*
- *has equal aggressiveness, if for $\beta_i = \beta_j$, $\alpha_i = \alpha_j$.*

It follows from equations 6.1, 6.3 and figure 6.1 that:

COROLLARY 1. *Adapting the AI parameter according to equations 6.1, 6.3 achieves variable aggressiveness.*

OBSERVATION 1. *The maximum aggressiveness achieved by higher order delay functions of TCP-Illinois's algorithm is upper bounded by the maximum aggressiveness of TCP-Illinois's algorithm, α_{max}. In this case both have the same aggressiveness (i.e. $\beta_{ill} = \beta_n = \beta_{max}$ and $\alpha_{ill} = \alpha_n = \alpha_{max}$).*

OBSERVATION 2. *The minimum aggressiveness achieved by higher order delay functions of TCP-Illinois's algorithm is lower bounded by the minimum aggressiveness of TCP-Illinois's algorithm, α_{min}. In this case both have the same aggressiveness (i.e. $\beta_{ill} = \beta_n = \beta_{min}$ and $\alpha_{ill} = \alpha_n = \alpha_{min}$).*

LEMMA 2. *Compared to TCP-Illinois, higher orders of power $n \in \mathbb{N}^*$ of the delay functions, achieve higher aggressiveness for $\alpha \in (\alpha_{min}, \alpha_{max})$ and equal aggressiveness for $\alpha \in \{\alpha_{min}, \alpha_{max}\}$.*

Proof. Let $\kappa_i(n)$, $i = 1, 2, 3, 4$, be the algorithm constants and $n \in \mathbb{N}^*$. If $\alpha(n)$ is a non-decreasing function in n, then $\alpha(n2) > \alpha(n1)$ for all $n2 > n1$.

$$\alpha(n, \beta) = \frac{\kappa_1(n)\kappa_4(n)}{\beta - \kappa_3(n) + \kappa_2(n)\kappa_4(n)}$$

It can be shown that $\frac{\partial \alpha}{\partial n} > 0$. Therefore, the function is indeed non-decreasing between the following two values:

$$\alpha(1, \beta) = \tfrac{\kappa_1(1)\kappa_4(1)}{\beta - \kappa_3(1) + \kappa_2(1)\kappa_4(1)}$$

$$\lim_{n\to\infty} \alpha(n, \beta) = \alpha_{max}$$

$$\therefore \alpha(n2, \beta) > \alpha(n1, \beta)$$

From Observations 1, 2:

$$\alpha(n2, \beta) \geq \alpha(n1, \beta)$$

$\therefore$ higher orders of power $n \in \mathbb{N}^*$ achieves better aggressiveness. □

6.2.2 Relative Responsiveness

The responsiveness characteristic of higher order functions can also be approached from a different angle. Formally speaking we state the following definition of relative responsiveness:

DEFINITION 2. *An AIMD CC algorithm 'A' with a variable α_a and β_a compared to another AIMD CC algorithm 'B' with a variable α_b and β_b:*
has better responsiveness if $\beta_a \geq \beta_b$ and $\alpha_a \geq \alpha_b$ except when, $\beta_a = \beta_b$ and $\alpha_a = \alpha_b$ where it has equal responsiveness.

LEMMA 3. *Compared to TCP-Illinois, higher orders of power $n \in \mathbb{N}^*$ of the delay functions achieve better responsiveness for $\beta \in (\beta_{min}, \beta_{max})$, $\alpha \in (\alpha_{min}, \alpha_{max})$ and equal responsiveness for $\beta \in \{\beta_{min}, \beta_{max}\}$, $\alpha \in \{\alpha_{min}, \alpha_{max}\}$.*

Proof. Using similar argument as in lemma 2, it can be shown that $\beta(n2, \alpha) > \beta(n1, \alpha)$. From Observations 1, 2:

$$\beta(n2, \alpha) \geq \beta(n1, \alpha)$$

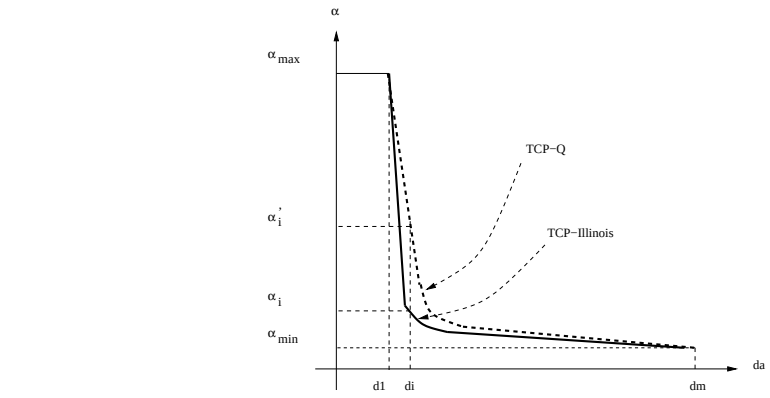
From this and lemma 2, higher orders of power $n \in \mathbb{N}^*$ achieves better responsiveness. □

6.2.3 Three TCP-Illinois Variants

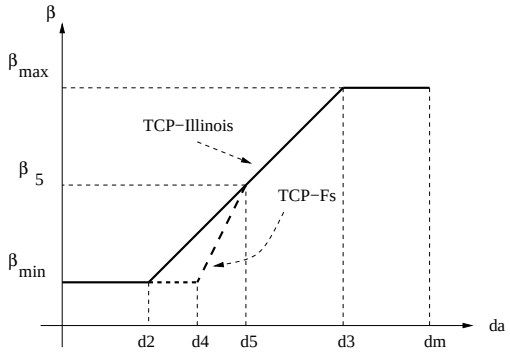
Based on the conclusions of our previous formal analysis, we experimented with three specific TCP-Illinois variants. The first variant (TCP-Q) employs a second order (power two - quadratic) AIMD function. Additionally, we experimented with two more variants by modifying the MD function only, while leaving the AI function unaltered: TCP-Fs uses a sub-linear MD function and TCP-Fq uses a quadratic MD function. In total we have three different modified versions of the algorithm: TCP-Q (quadratic MD and AI), TCP-Fs (sub-linear MD), TCP-Fq (quadratic MD). Figures 6.3a and 6.3 plot the AI (α) and MD (β) factors as a function of delay. It is important to note that TCP-Fs and TCP-Q have two different aims, the former aims to be more robust to non-congestive loss compared to TCP-Illinois by selecting a lower MD below mid range of average delay and keeping the same MD above mid range, i.e. this value of mid range separates the congestive from non-congestive loss actions. On the other hand TCP-Q has different aims: increase aggressiveness and responsiveness, for this reason the MD has different function for all the delay range, this function is more sensitive to delay in the sense that a small change in delay will result in a large change in MD compared to TCP-Illinois. In addition to that TCP-Q varies also varies AI in a similar sense to achieve these aims. We conducted the experiment of TCP-Fq just for curiosity i.e. to compare it with TCP-Fs, we have no aims for this version.

We observe that these variants have a rapid increase and small β for low delay values. Since the algorithm uses $cwnd \leftarrow (1 - \beta) \times cwnd$ for back off, our new functions yield higher window values. With regards to the sub-linear function, the slope gets steeper below midway between d_2 and d_3. Equation 6.5 shows our new definition of this function:

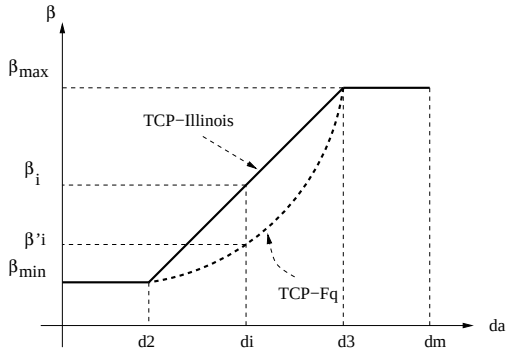
$$\beta = f_s(d_a) = \begin{cases} \beta_{min} & \text{if} \quad d_a \leq d_4 \\ \kappa'_3 + \kappa'_4 * d_a & \text{if} \quad d_4 < d_a \leq d_5 \\ \kappa_3 + \kappa_4 * d_a & \text{if} \quad d_5 < d_a < d_3 \\ \beta_{max} & \text{otherwise} \end{cases} \tag{6.5}$$



(a) AI functions



(b) Sub-linear



(c) Quadratic (n=2)

Figure 6.3: MD functions

Where,
$d_5 = (d_2 + d_3)/2$: the arithmetic mean of d_2 and d_3,
$d_4 = (d_2 + d_5)/2$: the arithmetic mean of d_2 and d_5,
$\beta_{mid} = (\beta_{max} + \beta_{min})/2$: the arithmetic mean of β_{max} and β_{min},
$\kappa'_3 = (\beta_{min} * d_5 - \beta_{mid} * d_4)/(d_5 - d_4)$,
$\kappa'_4 = (\beta_{mid} - \beta_{min})/(d_5 - d_4)$.

The conclusion of the previous discussion is that, compared to TCP-Illinois, we expect TCP-Q to have better aggressiveness and better responsiveness (and thus convergence times and inter-fairness).

6.3 Simulation Experiments & Results

By modifying the TCP-Illinois code in the `Linux` kernel, we obtained a module for each of our variants and used them for our simulations using the `TCP/Linux` patch for `ns2`. Based on the definitions of congestion control metrics [34], we conducted a comparative analysis between our variants and a number of relevant high speed TCP algorithms.

The simulation topology is a simple dumb-bell (figure 6.4) with only two TCP

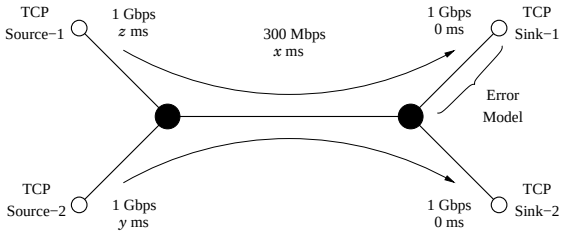


Figure 6.4: Topology

flows and one bottleneck with a fixed capacity of 300 Mbps. We used fixed MSS of 1000 bytes and drop-tail queue policy. The buffer size is fixed to 5% of BDP (when the total propagation delay is $46ms$, roughly a round trip time of $100ms$), the reason for using this buffer size was to limit the amount of data generated by the simulator for high-speed setup. We found that $300s$ of simulation time is sufficient for TCP to reach equilibrium under our setup.

We studied responsiveness in terms of transient response (response to change) and in terms of response functions (response to loss). For the later, we used an error model generating uniformly distributed random loss. The remaining variables in figure 6.4 had values set on a per experiment basis.

6.3.1 Intra-Protocol Fairness & RTT-Unfairness

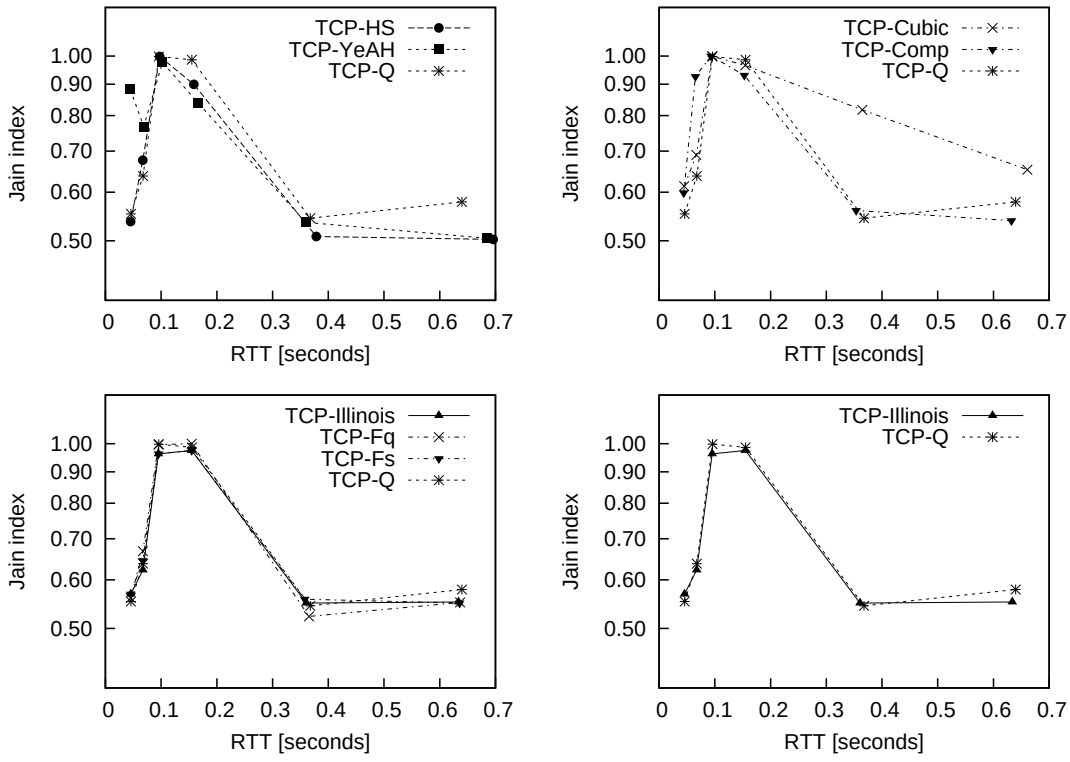
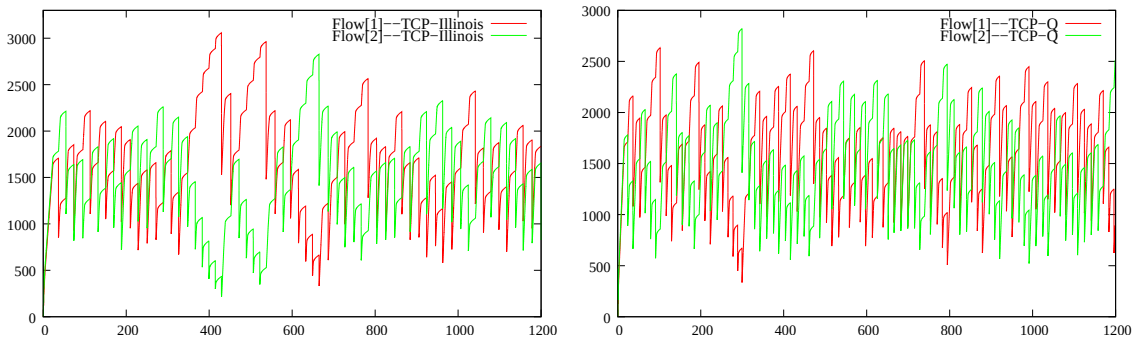


Figure 6.5: Algorithm fairness & RTT-Unfairness



(a) TCP-Illinois (b) TCP-Q

Figure 6.6: Congestion window for two flows running the same algorithm,$x = 16ms$,$y = z = 30ms$

To evaluate RTT-Unfairness we used Jain index[3]. Referring to figure 6.4; two flows have the same starting time, bottleneck link propagation delay is fixed at $x = 16ms$, first flow propagation delay is fixed at $z = 30ms$ and second flow propagation delay is varied as $y = 5ms, 16ms, 30ms, 60ms, 160ms, 300ms$, when $y = z$ we measure fairness, otherwise we measure RTT-Unfairness. For each run, two flows use the same TCP congestion control algorithm. We used the following relevant algorithms: TCP-YeAH, TCP-HS, TCP-Compound, TCP-Cubic, TCP-Illinois and our modified versions: TCP-Fs, TCP-Fq, TCP-Q.

Figure 6.5 shows Jain's fairness index versus second flow RTT for a set of TCP congestion control algorithms. Each trace, shows the result of two flows operating the same algorithm, where in all experiments $x = 16ms$ and first flow propagation delay is fixed at $z = 30ms$, second flow propagation delay is varied as $y = 5ms, 16ms, 30ms, 60ms, 160ms, 300ms$. Note that when $y = z$ with a RTT $\approx 100ms$ for both flows, fairness index and throughput ratio are maximum for all except for TCP-Illinois and TCP-Fs. We believe that the reason TCP-Fs operates like TCP-Illinois is because average delay must have taken values above midway between maximum and minimum values (please see figure 6.3b). On the other hand, both TCP-Fq and TCP-Q have better fairness.

For RTT values above $200ms$ and below $100ms$, RTT-unfairness is clear for all algorithms, this is indicated by drop in the fairness index. Note that the drop is severe for large RTTs for all algorithms except for TCP-Cubic and some values of TCP-Q.

[3] $f(x_1, x_2, \ldots, x_n) = \frac{\left(\sum_{j=1}^{n} x_j\right)^2}{n \sum_{j=1}^{n} (x_j)^2}$ $0 \leq f \leq 1$

6.3.2 Aggressiveness & Smoothness

The aggressiveness and smoothness characteristics of our new proposals compared to TCP-Illinois can be seen from AI and MD rules of the algorithms: for instance; if we consider smoothness in a deterministic environments to be the largest reduction in sending rate in one RTT. Then this means that losses happen at certain times, because this usually happens around maximum average delay, and since AI is the same for TCP-Fs, TCP-Fq, TCP-Q and TCP-Illinois, they all have the same reduction, β_{max}. In parallel, if we consider aggressiveness as the maximum increase in the sending rate in one RTT, then this is held constant for TCP-Fs,TCP-Fq,TCP-Q and TCP-Illinois. This can be seen from figure 6.3a to be α_{max}. In other words these characteristics are upper bounded by the original values of TCP-Illinois.

6.3.3 Transient Response

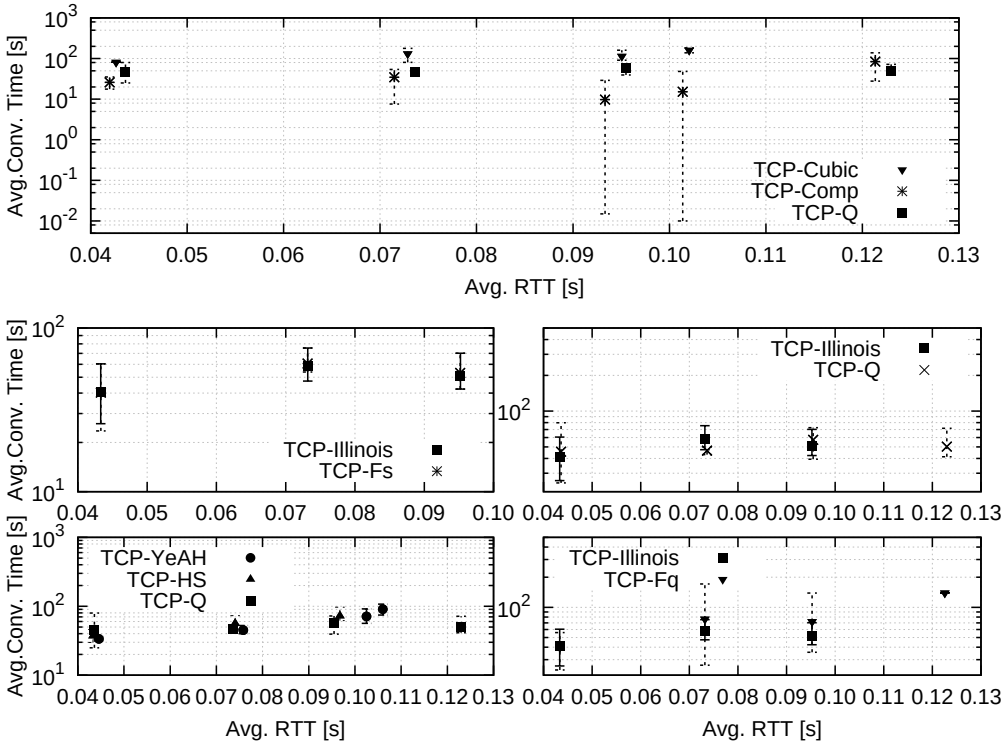


Figure 6.7: 20%-fair convergence, s-factor=0.005

We studied transient by investigating responsiveness, where responsiveness is defined in terms of convergence. In particular we used Delta-Fair convergence, where we define this as the time taken for two flows with same RTT to go from shares of 100/101-th and 1/101-th of the bottleneck bandwidth, to having close share of $(1 \pm \delta)/2$ of it.

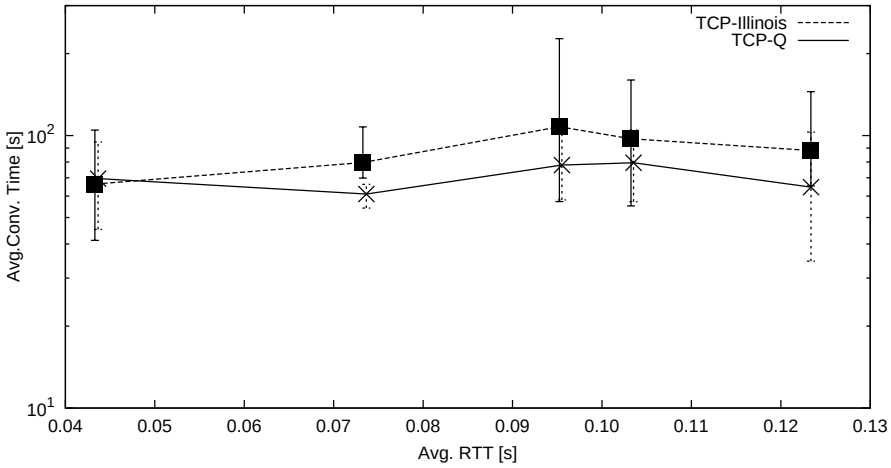
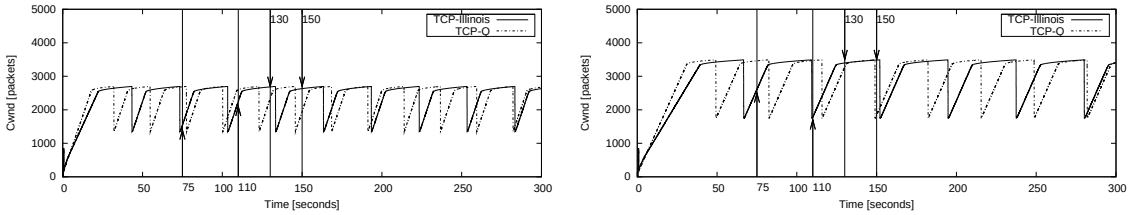


Figure 6.8: 20%-fair convergence, s-factor=0.0001



(a) $x = 35ms$,$y = z = 0ms$ (b) $x = 46ms$,$y = z = 0ms$

Figure 6.9: Bottleneck = 300Mbps, buffer size = 5%BDP

Referring to figure 6.4, we let $z = y = 0ms$ and varied the bottleneck propagation delay as $x = 20ms, 35ms, 46ms, 50ms, 60ms, 90ms, 190ms, 330ms$, thus for each run the RTT is the same for both flows. In addition to that, for each RTT run we used four different starting times for the second flow: $75s, 110s, 130s, 150s$. and calculated the throughput for each flow at a granularity of $0.2s$; then we short-term averaged with a smoothing factor of 0.005 and with a $\delta = 0.2$ which gives 20%-fair convergence.

Figure 6.7 shows average convergence times versus RTT (same for both flows) for a set of TCP congestion control algorithms. The bars above and below each point are the minimum and maximum values (not confidence intervals), and missing points indicate no convergence in the duration of simulation (300s). Figure 6.8 shows the result for TCP-Illinois and TCP-Q when we used smaller smoothing factor i.e. increased the low pass filtering.

Note that TCP-Fq has slightly higher average convergence times compared to TCP-Illinois at all RTTs, while TCP-Fs has almost identical values. For TCP-Illinois and TCP-Q; also note that when $x = 20ms$ ($RTT \approx 40ms$), both have nearly same average convergence times. If we look at the other two points in more detail, when $x = 35ms$ ($RTT \approx 70ms$); TCP-Q has less convergence time, to understand this we refer to figure 6.9a, the figure shows the congestion window

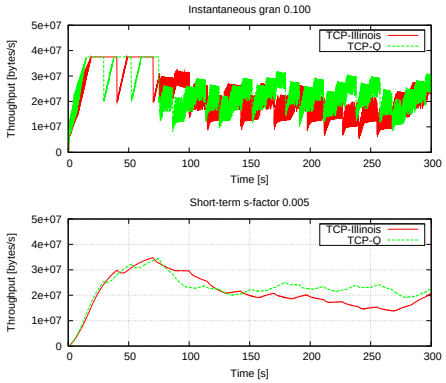
evolution of a single flow in the absence of any other competing flows, the two traces are from two different experiments. Vertical lines with arrows show the entry point in time of the second flow. We observed from the experiments that if the second flow enters at the beginning of congestion epoch (just after a loss) better convergence time is achieved, and if it enters at the end of it; convergence times are longer. We refer to this as *good entry* and *not good entry* respectively, we also consider any value below the middle as good entry and not good entry otherwise. The reason for that is that if the second flow enters at lower window values the reduction in window after a short period of that will result in significant drop in rate which will in turn enhance convergence times.

Referring to figure 6.9a, $75s$ and $110s$ are good entries for TCP-Illinois and not good entries for TCP-Q, $130s$ and $150s$ are good entries for TCP-Q and not good entries for TCP-Illinois. Thus having the same number of good and not good entries for both algorithms which makes the experiment unbiased to certain algorithm. We note that the entry point at $75s$ for TCP-Q is similar to entry point at $130s$ for TCP-Illinois. It would be more meaningful if we compare these two points.

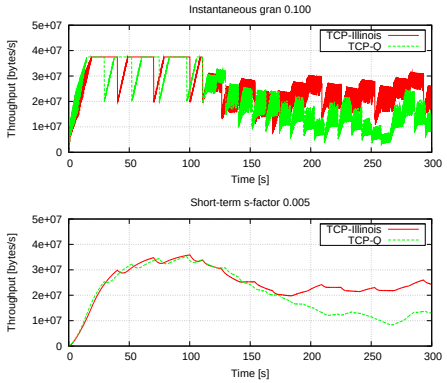
Now, figures 6.10a - 6.10d show the throughput versus time for $x = 35ms$, by comparing the dashed trace of figure 6.10a with the solid trace of figure 6.10c we see that TCP-Q has less convergence time. Similarly, entry point at $110s$ for TCP-Q is similar to entry point at $150s$ for TCP-Illinois, and by comparing the dashed trace of figure 6.10b with the solid trace of figure 6.10d we see that TCP-Q has also lower convergence times.

In a parallel argument, when $x = 46ms$ ($RTT \approx 92ms$) and by referring to figure 6.9b, we see that $75s$ and $110s$ are good entries for TCP-Illinois and not good entries for TCP-Q, $130s$ is the same for both and not a good entry, while $150s$ is a good entry for TCP-Q and not good entry for TCP-Illinois. In other words, TCP-Illinois has two good entries and two bad entries and TCP-Q has one good entry and three bad entries which makes the experiment biased towards TCP-Illinois. From figure 6.10h, it can be seen that TCP-Q converged to 20% fair share better that other entry points. In addition to that, we note that entry point at $150s$ for TCP-Q is similar to entry point at 110 for TCP-Illinois.

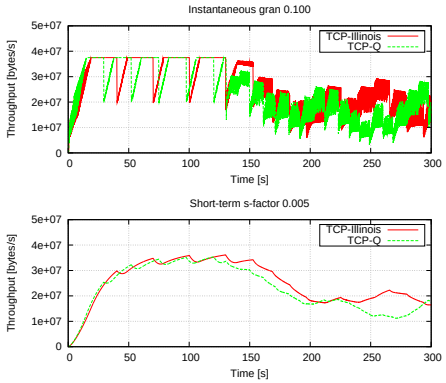
Figures 6.10e - 6.10h show the throughput versus time for $x = 46ms$, by comparing the dashed trace in figure 6.10h with the solid trace of figure 6.10f, we see that TCP-Q has less convergence time. However, because of the bias in this experiment the total average convergence time was larger for TCP-Q (please see figure 5.1).



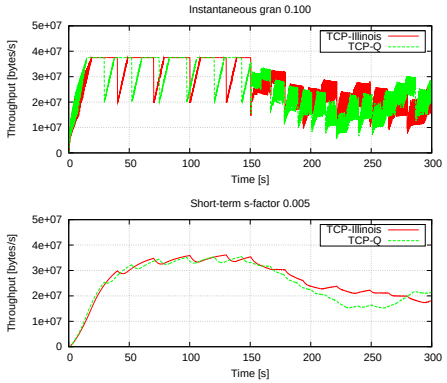
(a) Starting time: 75 seconds, x=35ms



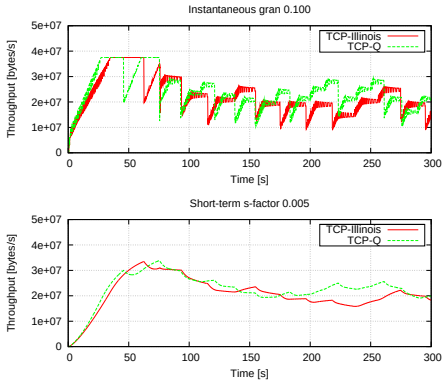
(b) Starting time: 110 seconds, x=35ms



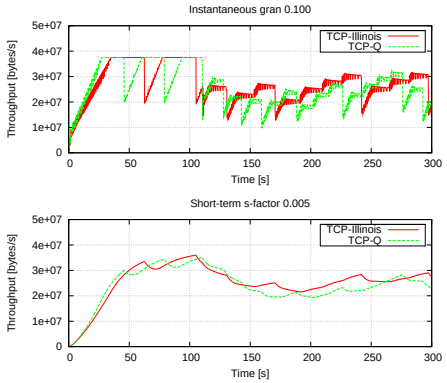
(c) Starting time: 130 seconds, x=35ms



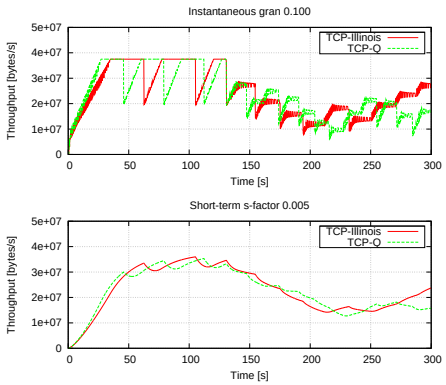
(d) Starting time: 150 seconds, x=35ms



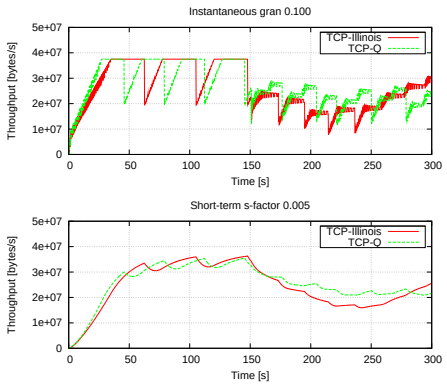
(e) Starting time: 75 seconds, x=46ms



(f) Starting time: 110 seconds, x=46ms

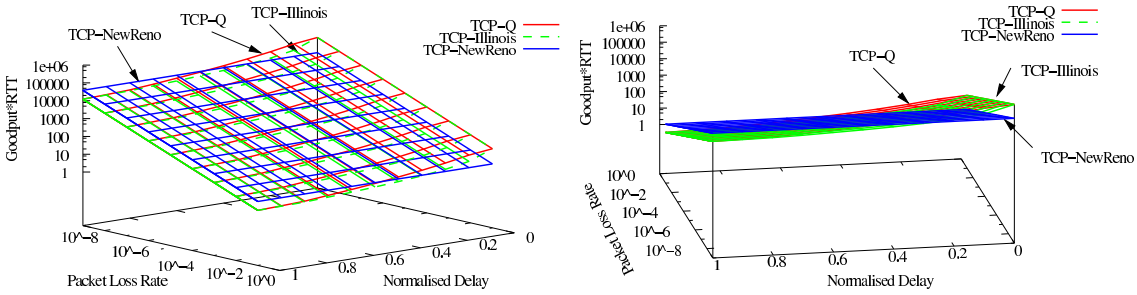


(g) Starting time: 130 seconds, x=46ms



(h) Starting time: 150 seconds, x=46ms

Figure 6.10: First flow throughput for different second flow starting times



(a) First view (b) Second view

Figure 6.11: Theoretical response function

6.3.4 Response to Packet Loss

We first considered the theoretical *Response function* shown in figure 6.11. It can be seen that TCP-Q has higher goodput compared to TCP-Illinois at low loss rates and low delay values i.e. more scalable.

Experimentally we used only one flow with propagation delays $z = 0ms, x = 46ms$, and set an error model to produce a uniformly distributed packet loss on the final link with a set of packet loss probabilities $10^{-5} - 10^{-1}$. We used bottleneck link capacities of 100Mbps, 300Mbps with buffer sizes of 5,16% BDP respectively and computed the average throughput for each packet loss probability.

Figure 6.12 shows the empirical response functions for a set of TCP congestion control algorithms, the y-axis is the average throughput of one flow trying to utilise a bottleneck link of 300Mbps with a buffer size of 5%BDP and having a propagation delays $z = 0ms, x = 46ms$ under the influence of an error model which produces a uniformly distributed packet loss on the final link. The x-axis represents the measured loss probability, we tried many packet loss probabilities in the range of $10^{-5} - 10^{-1}$; however in some experiments I were not able to produce the exact loss probabilities as other experiments, but they are all in the same range, this explains the different points for each trace. For low loss probabilities ($< 10^{-4}$) loss happens nearly at fixed interval when the delay is $\approx$ maximum (i.e. deterministic environment) and in steady state all achieve the same average throughput, this is expected for TCP-Fs and TCP-Fq, since they have the same AI as TCP-Illinois and will have the same MD if the delay is maximum. However, for TCP-Q at very low loss probabilities where the algorithm nearly utilises the capacity of the pipe, the congestion window evolution is different, this can be seen from figure 6.13a, the dark shaded areas represent the difference in the number of packets sent when TCP-Illinois is sending more than TCP-Q, and the light shaded areas represents the difference when TCP-Q is sending more than TCP-Illinois. If both areas are approximately equal (through same period of time) both have approximately the same average throughput.

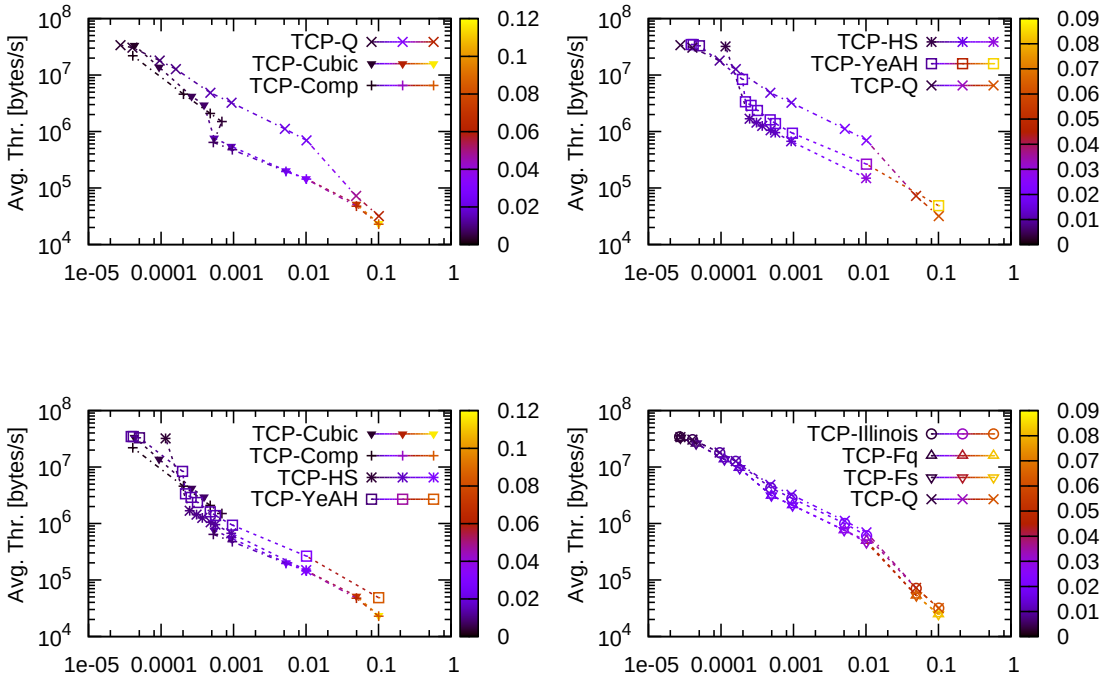
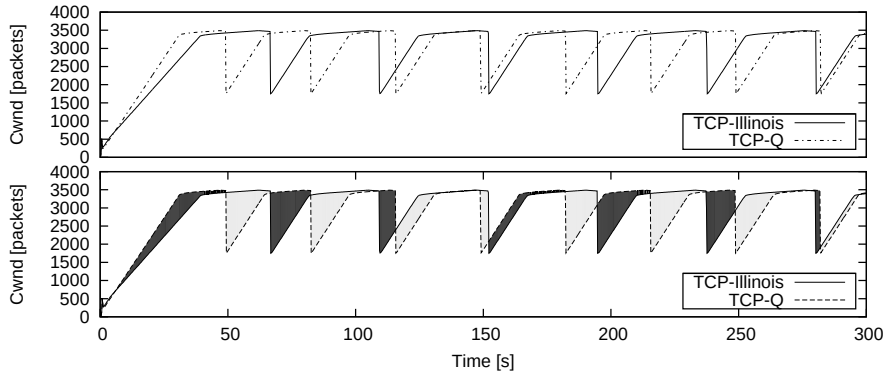


Figure 6.12: Bottleneck capacity = 300Mbps, buffer size = 5%BDP. Thermal bars represent average queueing delay in seconds.

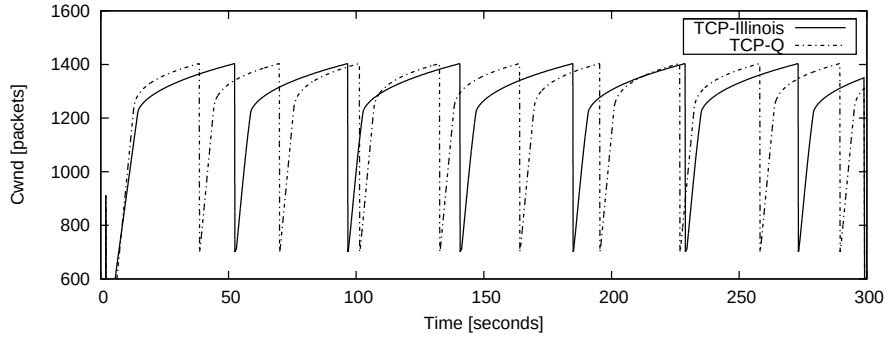
For loss probabilities in the range of $10^{-4} - 10^{-3}$ we note that TCP-Q achieves better average throughput compared to TCP-Illinois, while TCP-Fs and TCP-Fq are nearly identical to TCP-Illinois. We view this as a minor advantage due to short range of loss probabilities, we also believe the reason for this is the fast additive increase of TCP-Q compared to TCP-Fs, TCP-Fq and TCP-Illinois; this result in larger light shaded total area and thus larger average throughput in the duration of simulation.

For loss probabilities ($> 10^{-3}$), TCP-Illinois, TCP-Q, TCP-Fs and TCP-Fq work much like standard TCP, since they all fall-back to standard TCP when the window is below $cwnd_{thresh}$. We note that the average delay at high loss probability of 10% is due to loss happening in the slow start phase, this is basically a reset; as TCP starts again from one segment. We blame slow-start's aggressive bandwidth probing for this increase in average delay.

Finally, for illustrative purposes we show in figure 6.13 the congestion window evolution for TCP-Illinois and TCP-Q in absence of any competing flows, each trace is from different experiment. Figure 6.13a illustrates the difference in sent packets (area under the curve) at different times. Figure 6.13b shows the result when we used different bottleneck capacity. Note that the congestion epoch for TCP-Q is less than that for TCP-Illinois, and shorter congestion epochs usually translates to better convergence/responsiveness.



(a) Bottleneck capacity = 300Mbps, buffer size = 5%BDP



(b) Bottleneck capacity = 100Mbps, buffer size = 16%BDP

Figure 6.13: $x = 46ms$,$y = z = 0ms$

6.4 Summary

In this chapter we proposed a new idea for improving the sluggish responsiveness of TCP-Illinois, the idea is simple to implement and is based on generalising the delay functions used to adjust the additive increase and multiplicative decrease. We have shown that this can yield improvements, while operating within the same range of values for both MD and AI. More specifically, the higher order functions that we use enhance the algorithm's responsiveness (in terms of convergence). Experiments on special case quadratic variant show that the delay-loss based algorithm (TCP-Q) exhibits better fairness and convergence compared to the original algorithm (TCP-Illinois). Additionally, our new proposal is very easy to deploy since it only requires modifications on the sender side, which can potentially lead to a performance leap. According to our preliminary investigation, we believe that TCP-Fs and TCP-Fq (where only MD is changed) will exhibit better immunity in the case of error-prone links. This is mainly because of their lower MD values compared to TCP-Illinois. As part of our future work, we aim to run experiments in order to validate this assumption. Additionally, we will examine all three variants in more realistic, larger-scale environments.

The next chapter sheds the light on a new TCP congestion control algorithm, the algorithm is believed to combine the good merits of a number of existing

ones, especially targeting the feature of efficient utilisation of high bandwidth delay product pipes and contrast to TCP-Illinois, keeping the network load to minimum. We believe this is one of the challenges in the TCP congestion control field of research.

Chapter 7

TCP-Gentle

Our view of an ideal CC algorithm is to have an easy-deployable algorithm that has a perfect link utilisation, responsiveness, convergence to a fair share in zero time, robust (to environmental effects, e.g. link errors) has no additional load on the network. This formed the basis for the second contribution on this track. The approach that we adopted was to use a number of principles rather than to adhere to one to solve all problems. Each principle targets a problem without affecting other principles. In fact this lead us to the idea of *modes* which appeared in 1991 [98] and has been recently used in some relatively recent proposals [64], [9]. Having said that, we propose TCP-Gentle, a high speed TCP CC based on TCP-YeAH [9] (the latest in GNU/Linux kernel stack at the time of writing this thesis). Unlike loss-based algorithms, TCP-YeAH algorithm focuses on high link utilisation, high responsiveness while maintaining low network load, it achieves this by dividing the problem and adopting the idea of modes. However, we argue that better results can be achieved by dividing the problem differently. Our argument is supported by the fact that TCP-Gentle (our new proposal) is able to break some of the trade-offs, mainly, being smooth and having high responsiveness while being gentle to network and potentially more safe. Our interest in smoothness is to keep high link utilisation; since low smoothness (high oscillation) has undesirable effects on link utilisation (many large backoffs throughout the connection).

The rest of the chapter is divided as follows: in section 7.1 we discuss TCP-Gentle algorithm, its operational modes/phases and provide a a full pseudo-code for TCP-Gentle and TCP-YeAH. We also derive an expression for the average throughput which acts as a deterministic model for the algorithm. In section 7.3 we validate the new algorithm through simulation experiments, we validate the basic concepts for operation in high-speed long-delay network, study friendliness to TCP-NewReno and also study the effect of web traffic. In section 7.4 we present some of our real test-bed experiments, mainly congestion window evolution and link utilisation in high-speed long-delay network and compare this to TCP-YeAH.

We also study response to different random packet loss probabilities, intra-protocol fairness and RTT-unfairness and compare this to TCP-YeAH and TCP-CUBIC and examine friendliness to TCP-NewReno. Finally, we summarise in 7.5 the ideas discussed in this chapter.

7.1 TCP-Gentle Algorithm

We described the operation of TCP-YeAH in chapter 3, in this section we describe TCP-Gentle: a new proposal based on TCP-YeAH. The objective of this new algorithm is to get as close as possible to our view of ideal CC behaviour. This translates to the following aims:

1. Intra-protocol fairness: Address the potentially unsafe MI rule of TCP-YeAH in fast mode where an MIMD is used. As mentioned in chapter 2 this type of operation may not converge to a fair share, and when considering RTT-unfairness it is completely unfair. Instead, TCP-Gentle uses an adaptive AIMD rule (a safe rule from fairness perspective) which adapts according to different network circumstances.

2. Smoothness & Responsiveness: Break the trade-off between smoothness and responsiveness while being gentle to the network. Thus have higher smoothness and high responsiveness. By gentle to network we mean when TCP-Gentle is competing with other traffic, e.g. web traffic (many sources) it leaves a proportion of the queue rather than occupying additional proportion.

3. TCP-Friendly: Maintain friendliness to standard TCP-NewReno.

4. Link utilisation: Keeps high link utilisation and minimum queue size.

5. Network load: Keeps network load as minimum as possible, i.e. keep queue size as minimum as possible.

Each of of these aims maps to a principle, and principles are then grouped in modes. Instead of operating TCP-YeAH fast and slow modes, TCP-Gentle uses two different modes:*gentle mode* and *reno mode*.

The reno mode maintains TCP-YeAH slow mode idea of switching to NewReno operation when competing with a NewReno flow, and also has the same decongestion mechanism i.e. back off when a queue threshold is exceeded. This means, when a queue threshold is exceeded, the algorithm backs off by reducing the congestion window by the estimated queue size, and increment a counter. If in the next round trip time it finds that its queue estimate is still above threshold, it

Table 7.1: Breakdown of TCP-Gentle ideas

Aim	Principle	Mode of Occurrence	Phase
Intra-protocol fairness	AIMD	Gentle	T/D
	Increase responsiveness: fast detection of change	Gentle	D
	Increase responsiveness: fast response to change	Gentle	T
Smoothness	Damping AI rule	Gentle	D
	Decongestion	Beginning of Reno	-
TCP-Friendliness	Reno AI rule	Reno	-
Link utilisation	Damping AI rule	Gentle	D
	Decongestion	Beginning of Reno	-
Network load	Damping AI rule	Gentle	D
	Decongestion	Beginning of Reno	-

reduces the congestion window by the queue estimate again, and also increment the counter, and so on. When the counter exceeds a predefined number, the algorithm gives up and stops its non-greedy behaviour and assumes that the other competing source is greedy (NewReno-like) and switches to reno mode i.e. using AI specified by $\alpha_{reno} = 1$ and MD upon loss or ECN by half. TCP-Gentle adds to this by setting two different AI variables after each back off (upon exceeding queue threshold), these are: α_{gh} and α_{gh}/NP^v. This is explained later on in this chapter, for now we treat these as two parameters. These two parameters are used when the algorithm backs off after exceeding the queue threshold and in the next round trip time it finds that the queue threshold is not exceeded, i.e. the network responded to the decongestion mechanism, in this case TCP Gentle enters its gentle mode, use these two parameters and keeps its non-greedy behaviour. The gentle mode is described in details in following paragraphs.

While not in reno mode, TCP-Gentle is in gentle mode, this mode of operation consists of two phases: *thrust phase* and *damping phase*. The thrust phase is enabled only when the queue is empty. In this phase the AI is adapted according to an aggressive mechanism which we use to replace slow start and limited slow start [33], this mechanism is explained in detail in the following subsection. On the other hand the damping phase is enabled if the queue is not empty and below a threshold, in this phase the two AI variables set after decongestion back off (in reno mode) are used as boundaries i.e. maximum and minimum, and the AI is adjusted according to a formula each RTT; starting from the maximum value α_{gh} until the minimum value α_{gh}/NP^v is reached at end of decongestion epoch (when a back off is needed again). Table 7.1 summarises the relationship between the aims, principles, modes and phases.

7.1.1 Gentle Mode: Thrust Phase

In the thrust phase of gentle mode we use an aggressive rule which we call *rocket mechanism*. This mechanism is a realisation of our principle for fast response to change, it replaces slow start and limited slow start. The basic idea is illustrated below:

Slow Start, every RTT:

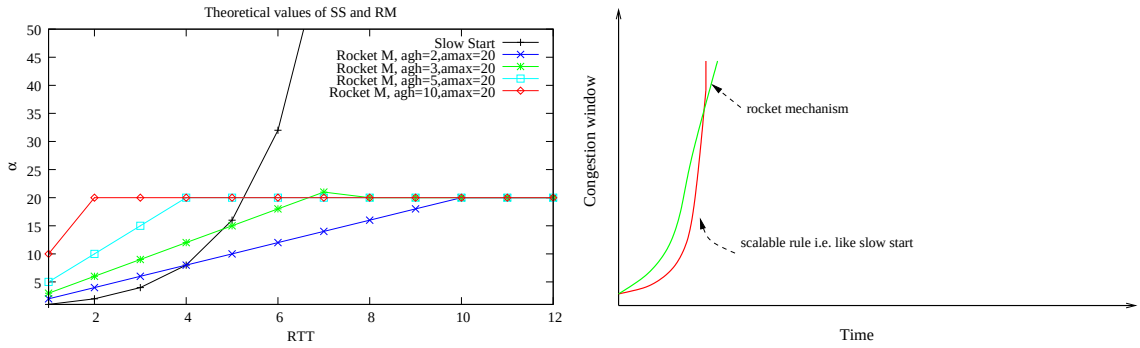
$$cwnd \leftarrow 2 \times cwnd$$

Rocket Mechanism, every RTT:

$$\begin{aligned} &if\ Q = 0\ then \\ &\alpha \leftarrow fuel \times \alpha_{rocket} \\ &cwnd \leftarrow cwnd + \alpha \\ &fuel \leftarrow fuel + 1 \\ &else \\ &fuel \leftarrow 0 \\ &\ldots \end{aligned}$$

The mechanism is enabled only when the the queue size estimate [1] is zero. The increase starts by a specified AI denoted by α_{rocket} instead of one segment. The AI keeps increasing until it reaches an upper bound α_{max}, after that the increase is linear. The name of the mechanism was inspired from the shape of congestion window evolution being similar to that of a real rocket path. The rational behind using this mechanism is three folds: i) We are more concerned in increasing responsiveness than probing the network like slow start, ii) Since the queue is empty we can be less conservative and increase by more than one segment from the beginning, iii) After certain number of RTT the growth is linear, while in an MI approach e.g. slow start, the window can grow to large values and put load on the network just before it reaches the full capacity (when it is not needed). We are aware of new techniques like limited slow start which can mitigate the large congestion window growth problem, however our technique treats the congestion window evolution at the beginning of the phase (more increase) and at the end of it (limited linear increase) and maintains consistency of using AI mechanism in the whole algorithm. While limited slow start limit the growth after a threshold and indeed keeps the traditional slow start before that in order to probe the network for available bandwidth.

[1] Queue size estimate is done each RTT



(a) SS RM (b) Congestion window evolution

Figure 7.1: Theoretical Values

Figure 7.1a shows the theoretical values of both slow start and rocket mechanism with different values of α_{rocket}, for simplicity we used $\alpha_{rocket} = \alpha_{gh}$. It can be seen from the figure that instantly the rocket mechanism inflates the congestion window more than slow start; thus allowing more packets to be transmitted, after a number of RTT, the upper bound is reached and the increase is fixed, while slow start keeps increasing the congestion window. The evolution of congestion window is shown in figure 7.1b.

7.1.2 Gentle Mode: Damping Phase

In the damping phase of gentle mode we make the algorithm adapt the AI according to a formula, the formula has been chosen to realise a set of principles (the relevant set of principles are mentioned in table 7.1). The damping phase is enabled only when the queue estimate is larger than zero and less or equal a threshold Q_{max}, during which the AI is adapted according to the following formula:

$$\alpha = \frac{\alpha_{gh}}{NP^{v}} \times (\frac{Q_{max}}{Q})$$

Upper bounded by α_{gh}

$$\alpha = \min\{\alpha, \alpha_{gh}\}$$

Lower bounded by α_{gl}

$$\alpha = \max\{\alpha, \alpha_{gl}\}$$

Where, α_{gh} is the initial value of AI and is set to same value in the thrust phase, NP is a counter which is incremented each decongestion epoch. The purpose of this counter is to damp the target minimum AI in a decongestion epoch as time progress since the phase is entered. The parameter v is used to control the speed of damping. The AI is inversely proportional to the queue estimate and in each decongestion epoch the AI is adapted between the maximum AI, α_{gh} (from

α_{gh}/NP^v when $NP = 1$) and the target minimum AI in that decongestion epoch α_{gh}/NP^v. The maximum value of NP is α_{gh}/α_{gl}, where α_{gl} is the minimum AI in the algorithm.

To further control the damping of AI, we used a parameter which allows us to use the same target minimum AI for a number of decongestion epoch before actually incrementing NP, we named this parameter alpha age, AG.

7.1.3 Reno Mode

The Reno mode has no phases, this mode is a realisation of the principle mentioned in the third row of table 7.1, the mode is enabled only when[2] $Q > Q_{max}$ we are in Reno mode and we use this rule.

$$\alpha = \alpha_{reno}$$

7.1.4 Complete Version

In this subsection we put all the ideas together to summarise TCP-Gentle algorithm. The final expression of AI is as follows:

$$\text{Each RTT:} \begin{cases} cwnd \leftarrow cwnd + \alpha & \\ \alpha_{reno} & \text{if } Q > Q_{max} \text{ OR } L > \frac{1}{\phi} \\ \max\{\min\{\frac{\alpha_{gh}}{NP^v} \times \frac{Q}{Q_{max}}, \alpha_{gh}\}, \alpha_{gl}\} & \text{if } 0 < Q \leq Q_{max} \\ \min\{fuel \times \alpha_{rocket}, \alpha_{max}\} & \text{if } Q = 0 \end{cases}$$

The final expression of MD is the same as TCP-YeAH and is as follows:

$$\text{Back off} \begin{cases} cwnd \leftarrow cwnd - Q \text{ if } Q > Q_{max} \\ cwnd \leftarrow cwnd - Q \text{ if loss or ECN} \\ cwnd \leftarrow \max\{\min\{cwnd, \frac{cwnd}{8}\}, \frac{cwnd}{2}\} \end{cases}$$

In the following we show a pseudo code of both TCP-YeAH and TCP-Gentle for more clarification of the differences between the two algorithms. We also show the code for a version of TCP-Gentle that allows the thrust phase to take one period (the first period) of the damping phase, we believe that this gives better performance, in terms of convergence/responsiveness during transient state (when traffic enters/exits) instead of sensing for zero queue to stop the thrust phase and start the damping phase. This may result in a conservative behaviour of the algorithm, since any time the queue is not empty the thrust phase is stopped,

[2]There is another condition adopted from YeAH: $L > 1/\phi$, where L is an estimate of the ratio of packets in flight to the BDP

alternatively we let the algorithm stops after one period of the damping phase. We call this version TCP-Gentle-2 compared to the original version, TCP-Gentle-1.

Algorithm 1: YeAH

```
        Initialisation:
        lastQ ← 0
        reno count ← 2 // end of initialisation
no loss: if packet loss is FALSE then
per ack:    foreach ACK do
                if cwnd < ssthresh then
                    SlowStart
                end
                if doing reno now is FALSE then
scalable:           cwnd ← cwnd + a // Scalable Rule
                else
reno:               cwnd ← cwnd + 1/cwnd // Reno Rule
                end
            end
per rtt:    foreach RTT do
slow mode:      if (Q > Q_max) ∨ (L > 1/φ) then
                    if (Q > Q_max) ∧ (cwnd > reno count) then
                        reduction ← min{Q/γ, cwnd/2^ε}
                        cwnd ← cwnd − reduction
                        cwnd ← max{cwnd, reno count}
                        ssthresh ← cwnd
                    end
                    if reno count ≤ 2 then
                        reno count ← max{cwnd/2, 2}
                    else
                        reno count ← reno count + 1
                    end
                    doing reno now ← doing reno now + 1
                else
fast mode:          fast count ← fast count + 1
                    if (fast count > ζ) then
                        reno count ← 2
                        fast count ← 0
                    end
                    doing reno now ← 0
                end
                lastQ ← Q
            end
        else
loss:       if (doing reno now < ρ) then
                reduction ← lastQ
                reduction ← min{reduction, max{cwnd/2, 2}}
                reduction ← max{reduction, cwnd/2^δ}
            else
                reduction ← max{cwnd/2, 2}
            end
            fast count ← 0
            reno count ← max{reno count/2, 2}
            cwnd ← cwnd − reduction
        end
```

Algorithm 2: Gentle-1: No loss

init: Initialisation:
- $lastQ \leftarrow 0$
- $virtual\ reno \leftarrow 2$
- $\alpha \leftarrow \alpha_{gh}$ // reset gentle mode --> start
- $NP \leftarrow NP_{min}$
- $AG \leftarrow AG_{min}$ // reset gentle mode --> end
- $fuel \leftarrow 0$
- $gmode\ reset\ count \leftarrow 0$
- $do\ reno \leftarrow 0$ // end of initialisation

per ack: **foreach** ACK **do**
- **if** $cwnd < ssthresh$ **then**
  - $SlowStart$
- **else**
  - $cwnd \leftarrow cwnd + \frac{\alpha}{cwnd}$ // AI Rule
- **end**

end

per rtt: **foreach** RTT **do**
- $currentQ \leftarrow Q$ // To distinguish from Q used in YeAH
- $lasQ \leftarrow Q$ // Same as YeAH, for loss reduction

reno mode:
- **if** $(Q > Q_{max}) \vee (L > \frac{1}{\phi})$ **then**
  - **if** $(Q > Q_{max}) \wedge (cwnd > virtual\ reno)$ **then**
    - $reduction \leftarrow \min\{\frac{Q}{\gamma}, \frac{cwnd}{2^\epsilon}\}$
    - $cwnd \leftarrow cwnd - reduction$
    - $cwnd \leftarrow \max\{cwnd, virtual\ reno\}$
    - $ssthresh \leftarrow cwnd$
    - $gmode\ reset\ count \leftarrow$
    - $gmode\ reset\ count + 1$
    - **if** $(gmode\ reset\ count > RESET\ THRESHOLD)$ **then**
      - $\alpha \leftarrow \alpha_{gh}$ // reset gentle mode --> start
      - $NP \leftarrow NP_{min}$
      - $AG \leftarrow AG_{min}$ // reset gentle mode --> end
    - **end**
    - **if** $(AG < AG_{max})$ **then**
      - $AG \leftarrow AG + 1$ // this is decongestion start counting
    - **else**
      - **if** $(NP < NP_{max})$ **then**
        - $NP \leftarrow NP + 1$
      - **else**
        - $AG \leftarrow AG_{min}$
      - **end**
    - **end**
  - **end**
  - **if** $virtual\ reno \leq 2$ **then**
    - $virtual\ reno \leftarrow \max\{\frac{cwnd}{2}, 2\}$
  - **else**
    - $virtual\ reno \leftarrow virtual\ reno + 1$
  - **end**
  - $do\ reno \leftarrow do\ reno + 1$
- **else**

gentle mode:
  - $gmode\ reset\ count \leftarrow 0$
  - $gmode\ count \leftarrow gmode\ count + 1$
  - **if** $(gmode\ count > \zeta)$ **then**
    - $virtual\ reno \leftarrow 2$
    - $gmode\ count \leftarrow 0$
  - **end**
  - $do\ reno \leftarrow 0$
- **end**
- **if** $do\ reno\ is\ FALSE$ **then**

thrust phase:
  - **if** $(currentQ = 0)$ // Gentle mode --> Thrust phase
  - **then**
    - $fuel \leftarrow fuel + 1$
    - $\alpha \leftarrow \min\{fuel * \alpha_{rocket}, \alpha_{max}\}$
  - **else**
    - $fuel \leftarrow 0$
  - **end**

damping phase:
  - **if** $(currentQ > 0) \wedge (currentQ \leq Q_{max})$ // Gentle mode --> Damping phase
  - **then**
    - $\alpha \leftarrow \min\{\frac{\alpha_{gh}}{NP^\nu} * \frac{Q_{max}}{currentQ}, \alpha_{gh}\}$
  - **end**
- **else**
  - $\alpha \leftarrow \alpha_{reno}$
- **end**

end

Algorithm 3: Gentle-2, Without slow start: No loss

: Initialisation:
- $lastQ \leftarrow 0$
- $virtual\ reno \leftarrow 2$
- $\alpha \leftarrow \alpha_{rocket}$ // reset gentle mode --> start
- $NP \leftarrow NP_{min}$
- $AG \leftarrow AG_{min}$
- $fuel \leftarrow 0$
- $max\_cwnd \leftarrow 2$ // reset gentle mode --> end
- $do\ reno \leftarrow 0$ // end of initialisation

: **foreach** ACK **do**
- $cwnd \leftarrow cwnd + \frac{\alpha}{cwnd}$ // AI Rule

end

: **foreach** RTT **do**
- $currentQ \leftarrow Q$ // To distinguish from Q used in YeAH
- $lasQ \leftarrow Q$ // Same as YeAH, for loss reduction
- $max\_cwnd \leftarrow max(max\_cwnd, cwnd)$

:
- **if** $(Q > Q_{max}) \vee (L > \frac{1}{\phi})$ **then**
  - **if** $(Q > Q_{max}) \wedge (cwnd > virtual\ reno)$ **then**
    - $reduction \leftarrow \min\{\frac{Q}{\gamma}, \frac{cwnd}{2^\epsilon}\}$
    - $cwnd \leftarrow cwnd - reduction$
    - $cwnd \leftarrow \max\{cwnd, virtual\ reno\}$
    - $ssthresh \leftarrow cwnd$
    - $fuel \leftarrow 0$
    - **if** $(cwnd < max\_cwnd - \frac{max\_cwnd}{2^\epsilon})$ // instead of RESET THRESHOLD
    - **then**
      - $\alpha \leftarrow \alpha_{rocket}$ // reset gentle mode --> start
      - $NP \leftarrow NP_{min}, AG \leftarrow AG_{min}$
      - $fuel \leftarrow 0, max\_cwnd \leftarrow 2$
      - // reset gentle mode --> end
    - **end**
    - **if** $(AG < AG_{max})$ **then**
      - $AG \leftarrow AG + 1$ // this is decongestion start counting
    - **else**
      - **if** $(NP < NP_{max})$ **then**
        - $NP \leftarrow NP + 1$
      - **else**
        - $AG \leftarrow AG_{min}$
      - **end**
    - **end**
  - **end**
  - **if** $virtual\ reno \leq 2$ **then**
    - $virtual\ reno \leftarrow \max\{\frac{cwnd}{2}, 2\}$
  - **else**
    - $virtual\ reno \leftarrow virtual\ reno + 1$
  - **end**
  - **if** $cwnd \leq virtual\ reno$ **then**
    - $do\ reno \leftarrow do\ reno + 1$
  - **end**
- **else**

:
  - $gmode\ count \leftarrow gmode\ count + 1$
  - **if** $(gmode\ count > \zeta)$ **then**
    - $virtual\ reno \leftarrow 2$
    - $gmode\ count \leftarrow 0$
  - **end**
  - $do\ reno \leftarrow 0$
- **end**
- **if** $do\ reno\ is\ FALSE$ **then**

:
  - $fuel \leftarrow fuel + 1$ // Gentle mode --> Thrust phase
  - $\alpha \leftarrow \min\{fuel * \alpha_{rocket}, \alpha_{max}\}$

:
  - **if** $(currentQ > 0) \wedge (currentQ \leq Q_{max}) \wedge (NP \geq (NP_{min} + 1))$ // Gentle mode --> Damping phase
  - **then**
    - $\alpha \leftarrow \min\{\frac{\alpha_{gh}}{NP^\nu} * \frac{Q_{max}}{currentQ}, \alpha_{gh}\}$
    - $fuel \leftarrow 0$
  - **end**
- **else**
  - $\alpha \leftarrow \alpha_{reno}$
- **end**

end

Algorithm 4: Gentle-1: Loss

if ($do\ reno < \rho$) **then**
  $reduction \leftarrow lastQ$
  $reduction \leftarrow$
  $\min\{reduction, \max\{\frac{cwnd}{2}, 2\}\}$
  $reduction \leftarrow \max\{reduction, \frac{cwnd}{2^{\delta}}\}$
else
  $reduction \leftarrow \max\{\frac{cwnd}{2}, 2\}$
end
$gmode\ count \leftarrow 0$
$virtual\ reno \leftarrow \max\{\frac{virtual\ reno}{2}, 2\}$
$\alpha \leftarrow \alpha_{gh}$ `// reset gentle mode-->start`
$NP \leftarrow NP_{min}$
$AG \leftarrow AG_{min}$ `// reset gentle mode-->end`
$cwnd \leftarrow cwnd - reduction$

Algorithm 5: Gentle-2: Loss

if ($do\ reno < \rho$) **then**
  $reduction \leftarrow lastQ$
  $reduction \leftarrow$
  $\min\{reduction, \max\{\frac{cwnd}{2}, 2\}\}$
  $reduction \leftarrow \max\{reduction, \frac{cwnd}{2^{\delta}}\}$
else
  $reduction \leftarrow \max\{\frac{cwnd}{2}, 2\}$
end
$gmode\ count \leftarrow 0$
$virtual\ reno \leftarrow \max\{\frac{virtual\ reno}{2}, 2\}$
$\alpha \leftarrow \alpha_{rocket}$ `// reset gentle mode-->start`
$NP \leftarrow NP_{min}$
$AG \leftarrow AG_{min}$
$fuel \leftarrow 0$
$max\_cwnd \leftarrow 2$ `// reset gentle mode-->end`
$cwnd \leftarrow cwnd - reduction$

One advantage of TCP-Gentle, is the ability to adapt its congestion window by controlling the parameters. For example, if the alpha age parameter AG is set to a large value $\rightarrow \infty$, the congestion window evolves according to fixed additive increase rule, α_{gh}. Another example is to consider a negative power of v, in this case the algorithm increase rule gains a multiplicative increase component in steady-state. Next, we derive the throughput expression for TCP-Gentle.

7.2 Throughput Expression

We are interested in the theoretical average throughput expression of the new algorithm. We focus on: i) Steady-state for gentle mode / damping phase, ii) Rocket mechanism. We follow an approach similar to some approaches mentioned in literature [11], the steps are as follows: i) Obtain an expression for the congestion window as a function of time, ii) Determine the the length of decongestion epoch in time, iii) Determine the number of packets sent i.e. the area under the curve, iv) Divide the total number of packets sent by the length of congestion epoch to obtain the average throughput, v) Write this in terms of probability of back off (loss for loss-based algorithms). We show later in this chapter, that the theoretical values calculated herein agrees with experimental results values, we consider this a good validation of the expression.

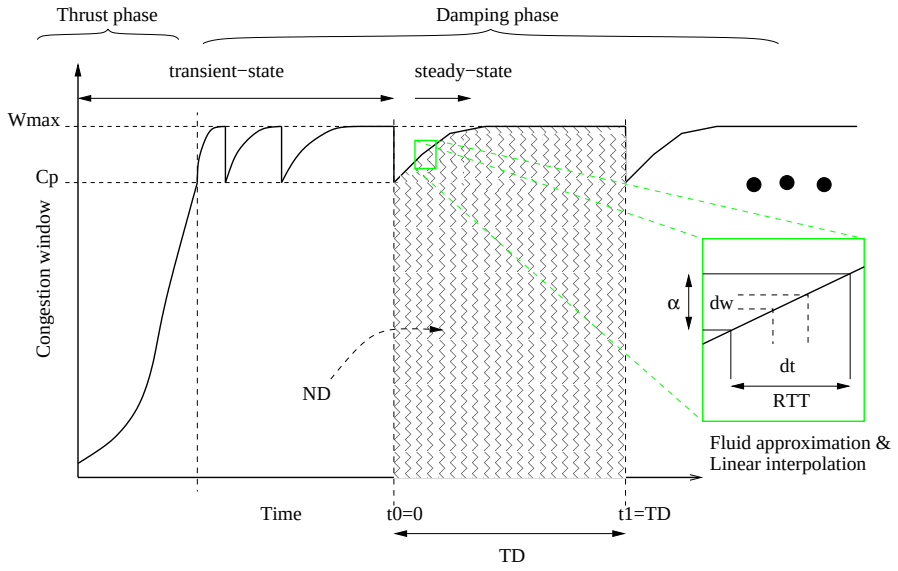


Figure 7.2: Congestion window curve

7.2.1 Steady-state average throughput:

The increase rule of TCP-Gentle can be expresses as:

$$w(t+1) \leftarrow w(t) + \alpha$$

Which can be written as:

$$\Delta w = \alpha$$

We know from TCP congestion control, that this change in congestion window takes place each RTT, thus we can write the slope of the congestion window function as:

$$\frac{\Delta w}{\Delta t} = \frac{\alpha}{RTT}$$

Using fluid approximation and linear interpolation, we can generalise this to any infinitesimal sample on the curve (see figure 7.2), thus

$$\frac{dw}{dt} = \frac{\alpha}{RTT} \tag{7.1}$$

In steady-state, $NP = NP_{max}$. Let C_p be the pipe capacity in packets, then the queue size can be written as

$$Q = w - Cp, \quad Q \geq 0$$

Thus the increase parameter in the steady-state is:

$$\alpha = \frac{\alpha_{gh}}{NP^v_{max}} . \frac{Q_{max}}{Q}$$

Substituting for Q:

$$\alpha = \frac{\alpha_{gh}}{NP^v_{max}}.\frac{W_{max} - C_p}{w - C_p}$$

Thus, equation 7.1 becomes:

$$\frac{dw}{dt} = \frac{\alpha_{gh}}{NP^v_{max}}.\frac{W_{max} - C_p}{w - C_p}.\frac{1}{RTT}$$

Separating variables and integrating both sides:

$$\begin{aligned} \int (w - C_p).dw &= \int \tfrac{\alpha_{gh}}{NP^v_{max}}.\tfrac{W_{max}C_p}{RTT}.dt \\ \tfrac{w^2}{2} - C_p w &= \tfrac{\alpha_{gh}}{NP^v_{max}}.\tfrac{W_{max}C_p}{RTT}.t + C \end{aligned}$$

Where, C is the integration constant and can be found from the initial condition. At t_0, $w = C_p$, thus $C = -C_p^2/2$. Rearranging terms of the last result:

$$\frac{w^2}{2} - C_p w - (\frac{\alpha_{gh}}{NP^v_{max}}.\frac{W_{max}C_p}{RTT}.t - \frac{C_p^2}{2}) = 0$$

Solving for w:

$$\begin{aligned} w &= C_p \pm \sqrt{\tfrac{2\alpha_{gh}(W_{max}-C_p)}{NP^v_{max}RTT}.t} \\ &= C_p + \sqrt{\tfrac{2\alpha_{gh}(W_{max}-C_p)}{NP^v_{max}RTT}.t} \end{aligned} \tag{7.2}$$

The negative sign solution was rejected because the window in steady-state is always larger or equal to the capacity of the pipe. Note that the congestion window is a square-root function of time.

To find TD, we note that the congestion window reaches W_{max} after a period TD, thus using equation 7.2:

$$\begin{aligned} W_{max} &= C_p + \sqrt{\tfrac{2\alpha_{gh}(W_{max}-C_p)}{NP^v_{max}RTT}.TD} \\ TD &= \tfrac{NP^v_{max}RTT(W_{max}-C_p)}{2\alpha_{gh}} \end{aligned} \tag{7.3}$$

To find ND, we find the area under the curve during TD:

$$\begin{aligned} ND &= \tfrac{1}{RTT}\int_0^{TD} C_p + \sqrt{\tfrac{2\alpha_{gh}(W_{max}-C_p)}{NP^v_{max}RTT}}.\sqrt{t}.dt \\ &= \tfrac{1}{RTT}\left(C_pTD + \sqrt{\tfrac{2\alpha_{gh}(W_{max}-C_p)}{NP^v_{max}RTT}}.TD^{3/2}.\tfrac{2}{3}\right) \end{aligned} \tag{7.4}$$

To find the average throughput X, we divide the ND by TD, thus:

$$X = \frac{ND}{TD} = \frac{1}{3RTT}(2W_{max} + C_p) \tag{7.5}$$

Finally, to write the average throughput in terms of the probability of back off p_b, we need to do the following steps: i) Use equation 7.3 to substitute for TD in equation 7.4:

$$ND = \frac{NP_{max}^{v}(W_{max} - C_p)(C_p + \frac{2}{3}(W_{max} - C_p))}{2\alpha_{gh}} \tag{7.6}$$

ii) Use the relationship, $ND = 1/p_b$:

$$p_b = \frac{2\alpha_{gh}}{NP_{max}^{v}(W_{max} - C_p)(C_p + \frac{2}{3}(W_{max} - C_p))} \tag{7.7}$$

iii) Solve equation 7.7 for W_{max}:

$$W_{max} = \frac{C_p + 3\left(\sqrt{C_p^2 + \frac{16\alpha_{gh}}{3NP_{max}^{v}p_b}}\right)}{4} \tag{7.8}$$

iv) Substitute for W_{max} in equation 7.5:

$$X = \frac{ND}{TD} = \frac{1}{2RTT}\left(\sqrt{C_p^2 + \frac{16\alpha_{gh}}{3NP_{max}^{v}p_b}} + C_p\right) \tag{7.9}$$

This final result shows that the average throughput in steady-state is $\propto \frac{1}{\sqrt{p_b}}$. However, it is very important to note that for large p_b the average throughput does not reduce below the link capacity. In other words the average throughput is nearly flat. This can be seen from equation 7.17 and also is understood from the algorithm operation since it only reduces by the amount of excess packets in the network. As we will see later in this chapter, the limitation of this expression is that it does not take time-outs into account.

To illustrate the use of equation 7.9, we give a numerical example. Given the algorithm parameters: $\alpha_{gh} = 2, NP_{max} = 4, v = 2, Q_{max} = 50$ pkts, and link bandwidth of 300 Mbps, with large buffer and a RTT = 100ms. Let the packet size be 1000 bytes. We calculate probability of back off and the average throughput. First note that, the capacity of the pipe without the buffer is: 3750 pkts ($300\times10^6\times0.1/8000$) and with 50 pkts in the buffer, the capacity is 3800 pkts, i.e. the maximum throughput in the steady-state is 304 Mbps ($3800 \times 8000/0.1$).

Using equation 7.7 we find that the probability of back off is $p_b = 1.32 \times 10^{-6}$, using this information and equation 7.9 we find that the average throughput is

302.67 Mbps (X = 37833.7 pkts/s). The algorithm values used in this example are the same values used in our experiments.

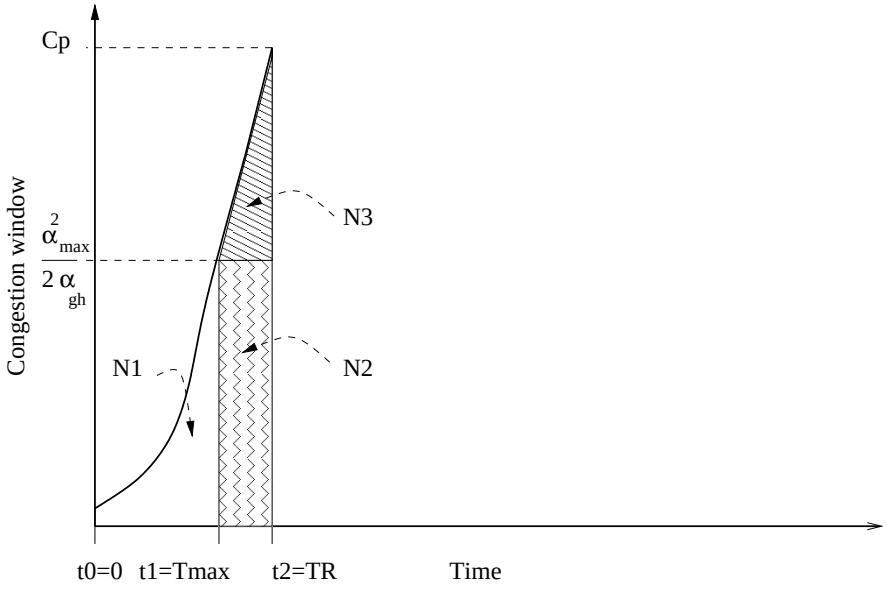


Figure 7.3: Congestion window curve for thrust phase

7.2.2 Initial average throughput:

Now we derive and expression for the average throughput in the thrust phase. In this phase the rocket mechanism is enabled and the increase rule of TCP-Gentle can be expresses as:

$$w(t+1) \leftarrow w(t) + \alpha$$

Using the same fluid approximation and interpolation:

$$\frac{dw}{dt} = \frac{\alpha}{RTT}$$

But:

$$\alpha(t+1) \leftarrow \alpha(t) + \alpha_{gh}$$

Similarly:

$$\frac{d\alpha}{dt} = \frac{\alpha_{gh}}{RTT}$$

We solve for α by separating variables and integrating both sides:

$$\alpha = \frac{\alpha_{gh}}{RTT}.t + C_1$$

Where C_1 is an integration constant and can be found from the initial condition at t_0. At t_0 $\alpha = 0$, thus, $C_1 = 0$. The the window equation becomes:

$$\frac{dw}{dt} = \frac{\alpha_{gh}}{RTT^2}.t \tag{7.10}$$

We solve for w by separating variables and integrating both sides:

$$w = \frac{\alpha_{gh}}{RTT^2}\int t.dt = \frac{\alpha_{gh}}{RTT^2}.\frac{t^2}{2} + C_2 \tag{7.11}$$

Where C_2 is an integration constant and can be found from the initial condition at t_0. At t_0 $w = 0$, thus, $C_2 = 0$. Note that the window is a quadratic function of time at the beginning of this phase. However, after reaching α_{max} at time T_{max}, the window growth is linear and the increase is fixed and determined by αmax. T_{max} can be found from equation 7.10:

$$T_{max} = \frac{RTT\alpha_{max}}{\alpha_{gh}} \tag{7.12}$$

We can find the window at T_{max} by substituting for T_{max} in equation 7.11:

$$w_m = \frac{\alpha_{max}^2}{2\alpha_{gh}}$$

Using the same approach, we need to find the window function after T_{max}:

$$\frac{dw}{dt} = \frac{\alpha_{max}}{RTT}$$

solving again for w by separating variables and integrating both sides:

$$w = \frac{\alpha_{max}}{RTT}.t + C_3$$

Where C_3 is an integration constant and can be found from the initial condition at t_1. At t_1, $w = w_m$, thus, $C_3 = -\alpha_{max}^2/(2\alpha_{gh})$. The the window equation becomes:

$$w = \frac{\alpha_{max}}{RTT}.t - \frac{\alpha_{max}^2}{2\alpha_{gh}} \tag{7.13}$$

From equation 7.11 and equation 7.13 we can write a full form for the window function in this phase:

$$w = \left(\frac{\alpha_{gh}}{RTT^2}.\frac{t^2}{2}\right)u(T_{max} - t) + \left(\frac{\alpha_{max}}{RTT}.t - \frac{\alpha_{max}^2}{2\alpha_{gh}}\right)u(t - T_{max}) \tag{7.14}$$

Where, $u(.)$ is the unit step function. Note that the window growth function is first quadratic, then linear, then square-root (in the damping phase), which we believe that this strongly supports a congestion avoidance approach and thus safe to deploy in the Internet.

To find the average throughput we need to find each of the sub-areas shown in figure 7.3, $N1, N2, N3$, find their sum and divide it by T_R. First we find T_R, by substituting $w = C_p$ in equation 7.13 and rearranging terms:

$$T_R = \left(C_p + \frac{\alpha_{max}^2}{2\alpha_{gh}}\right)\frac{RTT}{\alpha_{max}} \tag{7.15}$$

The we find the sub-areas:

$$N1 = \frac{1}{RTT}\int_0^{T_{max}} \frac{\alpha_{gh}}{RTT^2}\frac{t^2}{2}.dt = \frac{\alpha_{gh}T_{max}^3}{6RTT^3}$$

$$N2 = \frac{\alpha_{max}^2}{2RTT\alpha_{gh}}\left(T_R - T_{max}\right)$$

$$N3 = \frac{1}{2RTT}\left(T_R - T_{max}\right)\left(C_p - \frac{\alpha_{max}^2}{2\alpha_{gh}}\right)$$

The average throughput can be expressed as:

$$X = \frac{N1 + N2 + N3}{T_R}$$

$$= \frac{\alpha_{ghT_{max}^3}}{6RTT^3T_R} + \frac{1}{RTT}\left(1 - \frac{T_{max}}{T_R}\right)\left(\frac{C_p}{2} + \frac{\alpha_{max}^2}{4\alpha_{gh}}\right) \tag{7.16}$$

We can write the average throughput in terms of the algorithm parameters and the capacity C_p by substituting for T_{max} and T_R in equation 7.16:

$$X = \frac{1}{RTT}\left(\frac{\alpha_{max}^2}{6\alpha_{gh}}\left(\frac{\alpha_{max}^2}{\alpha_{gh}C_p - \frac{\alpha_{max}^2}{2}}\right) + \left(1 - \frac{\alpha_{max}^2}{\alpha_{gh}C_p - \frac{\alpha_{max}^2}{2}}\right)\left(\frac{C_p}{2} + \frac{\alpha_{max}^2}{4\alpha_{gh}}\right)\right) \tag{7.17}$$

As a numerical example, we consider the the same values mentioned when discussing the steady-state, however in this case substituting the values in equation 7.17 we get 146.9 Mbps ($X = 18367.05$ pkts/s). This means that on average approximately 50% of the pipe capacity (in bits/s) can be achieved in the thrust phase.

In the following sections, we show through simulation and real experiments that for a set of algorithm parameters, the desired aims can be achieved.

7.3 Simulation Experiments & Results

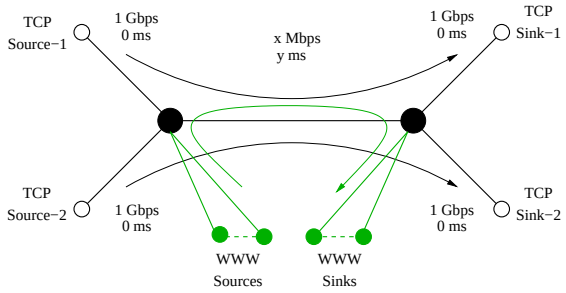


Figure 7.4: Topology

In this section we discuss our simulation experiments used to evaluate TCP-Gentle. We implemented TCP-Gentle based on TCP-YeAH module and used `TCP/Linux` patch[3] for `ns2`. We have found that the following choice for the set of TCP-Gentle parameters achieve our aims: $\alpha_{gh} = 2, \alpha_{gl} = 0.1, \alpha_{rocket} = 2, \alpha_{max} = 20, NP = 7, AG = 8, v = 2$. We used the same queue size threshold in TCP-YeAH (80 packets), to be able to compare TCP-Gentle and TCP-YeAH under the same conditions. The choice of 80 packets gives reasonable performance of TCP-YeAH for high-speed long-delay environment [9]. A simple dumbbell topology (please see figure 7.4) with only two TCP flows and one bottleneck with a Drop-Tail queue policy and a capacity of 300 Mbps and 100Mbps was used in most of the experiments. The MSS was fixed to 1000 bytes, buffer size was fixed to 5% of BDP[4] (when the total propagation delay is $46ms$, roughly a round trip time of $100ms$), the reason for using this buffer size was to limit the amount of data generated by the simulator for high-speed setup. The WWW sources, are used only when studying the effect of web traffic. We found that in most cases; $300s$ of simulation time is sufficient for TCP to reach equilibrium under this setup. However, a large buffer size (100 % of BDP) and longer simulation time, $1200s$ were used in a number of cases when studying steady-state behaviour.

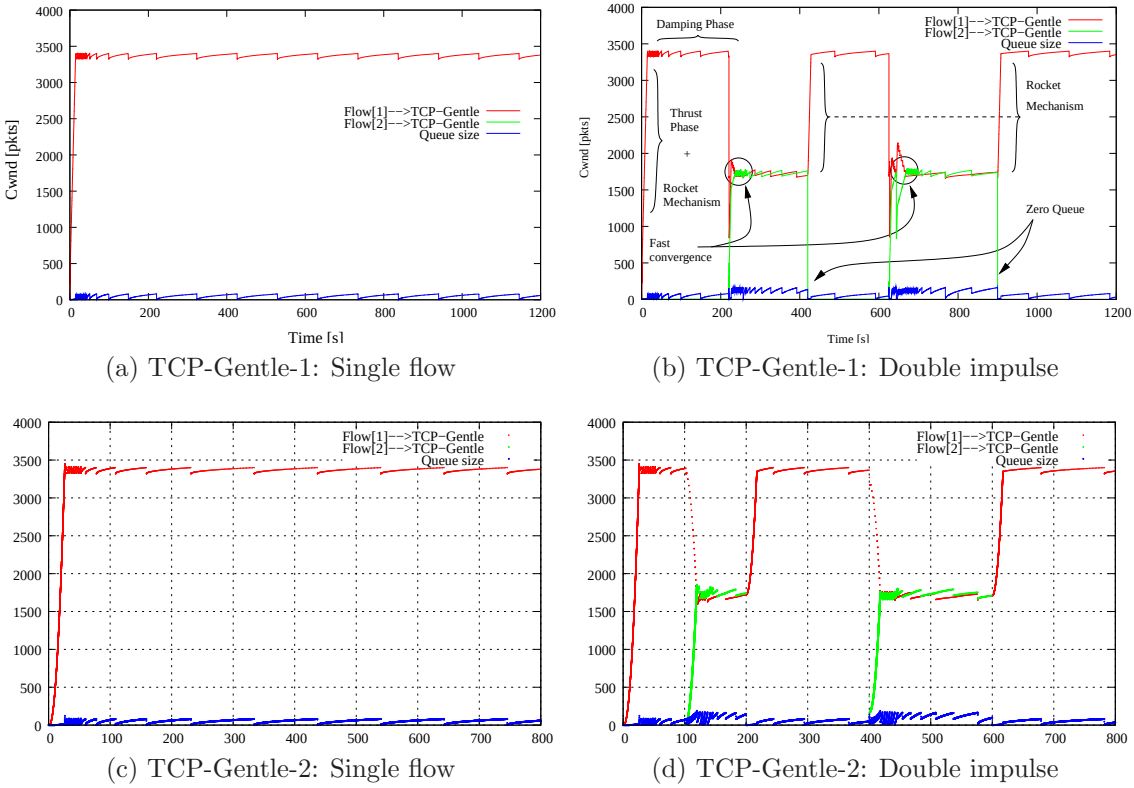
7.3.1 High BDP Operation

In order to evaluate TCP-Gentle basic concepts, we used two flows running TCP-Gentle. In the absence of any other competing traffic, the flows try to utilise the bottleneck link in figure 7.4.

First, we let the second flow start at $\rightarrow \infty$, and studied the operation of a single flow. Figure 7.5 shows the congestion window evolution of TCP-Gentle flow

[3]TCP/Linux is now part of ns2 code.

[4]Unless otherwise mentioned, these values of bottleneck link capacity and buffer size are default values in this section



(a) TCP-Gentle-1: Single flow

(b) TCP-Gentle-1: Double impulse

(c) TCP-Gentle-2: Single flow

(d) TCP-Gentle-2: Double impulse

Figure 7.5: Bottleneck = 300 Mbps, RTT = 92 ms, Buffer = 5%, BDP = 172 pkts

and the queue size of the bottleneck link. As can be seen from figure 7.5(top), when the connection starts, the algorithm is in the gentle mode. Since the queue is empty, the thrust phase is enabled and the congestion window evolves according to the rocket mechanism rule until it senses a non-empty queue. In the RTT where a non-empty queue is sensed, the algorithm de-couple the rocket mechanism and enters the damping phase with an initial increase of α_{gh} per RTT. After that, the congestion window keeps increasing until it senses that the queue threshold is exceeded. In the RTT where the sender estimate of the bottleneck queue exceeds this queue threshold, the algorithm reduces the congestion window by the queue estimate (this is decongestion) and switches reno mode, however because the network responds to this reduction, the next RTT, the sender sees a queue less than the threshold and exits reno mode and goes back to gentle mode / damping phase. Note that the reno mode is entered only once per decongestion epoch and the algorithm exits in the next RTT i.e. the algorithm spends most of its time in gentle mode / damping phase, which is the steady-state mode in this case.

Second, we study a double impulse response. We let the second flow start at $t = 200s$ and exit at $t = 400s$, then start again at $t = 600s$ and finally exit at $t = 900s$. The congestion window evolution of this case is illustrated in figure 7.5(bottom). One point to note is the fast convergence when the second

flow enters, this is because the first flow reset its gentle mode after a number of subsequent reductions ($RESET\ THRESHOLD$), and thus two flows start with the high α_{gh} which facilitates the fast convergence according to the concept of AIMD (chapter 2). Another point is the automatic enabling/disabling of rocket mechanism. For example, when the second flow exits at $t = 400s$ and $t = 900s$, the queue size estimate in the next RTT becomes zero and the first flow relieves that the second flow exits and thus enables the rocket mechanism. Note that even if the queue was not completely empty when the second flow exits, the algorithm still respond aggressively due to its damping phase rule being inversely proportional to the queue estimate ($\propto 1/Q$. The square-wave-like shape of the first flow congestion window is desirable, since on one side it indicates the trade-off break of high responsiveness and smoothness, from another side it increases the area under the curve, i.e. number of packets sent during time interval. Historically, increasing this area was indeed a desirable thing, and was achieved by means of using a concave congestion window growth, see for example [102], [69].

7.3.2 Friendliness to TCP-NewReno

Considering TCP-NewReno friendliness, TCP-Gentle differs from TCP-YeAH in two points: firstly, TCP-Gentle has different implementation of TCP-NewReno increase rule, a stanza from both codes is shown in Listing 7.1 and Listing 7.2. When competing with a TCP-NewReno flow, both increment by one packet each RTT, however this implementation of of TCP-Gentle rule fits our needs, since α is variable and can take different values (in reno mode, $\alpha = \alpha_{reno} = 1$) while in TCP-YeAH the increment is always by one.

Listing 7.1: TCP-Gentle AI rule

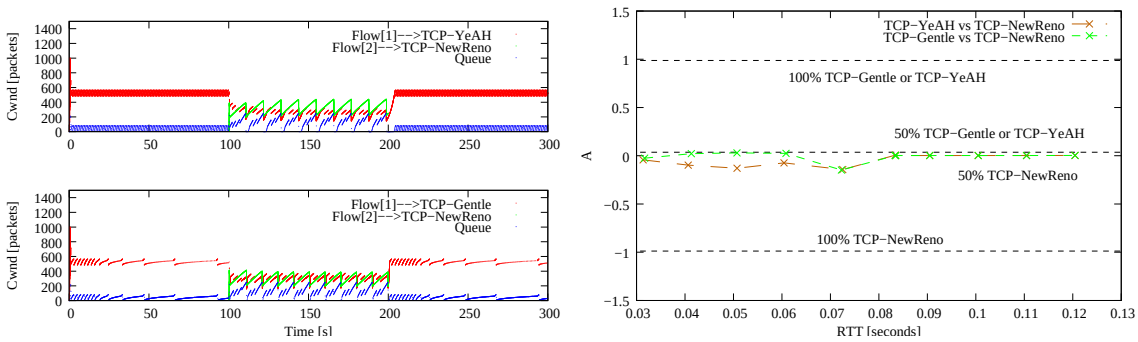
```
u32 delta;
tp->snd_cwnd_cnt+=gentle->pkts_acked;
delta = (tp->snd_cwnd_cnt * gentle->alpha) >> ALPHA_SHIFT;
if (delta >= tp->snd_cwnd){
tp->snd_cwnd = min(tp->snd_cwnd + delta /tp->snd_cwnd,
(u32) tp->snd_cwnd_clamp);
tp->snd_cwnd_cnt = 0;
}
```

Listing 7.2: Reno AI rule used in TCP-YeAH

```
if (tp->snd_cwnd_cnt >= w){
if (tp->snd_cwnd < tp->snd_cwnd_clamp)
tp->snd_cwnd++;
tp->snd_cwnd_cnt = 0;
```

```
}
else
{
tp->snd_cwnd_cnt++;
}
```

Secondly, in steady-state TCP-Gentle and TCP-YeAH have totally different modes of operation; TCP-Gentle steady-state operation is in gentle mode, while TCP-YeAH steady-state operation is fast mode. When a competing TCP-NewReno enters both exist from different modes and after a the competing TCP-NewReno exits, TCP-Gentle enters gentle mode, while TCP-YeAH enters fast mode. In order to study the impact of these two points on TCP-NewReno, we used two flows that have the same starting time, bottleneck link propagation delay varied as $x = 10ms, 15ms, 20ms, 25ms, 30ms, 35ms, 40ms, 45ms, 50ms, 55ms, 60ms$, $z = y = 0ms$. For each run, one flow use TCP-Gentle or TCP-YeAH and other flow is is TCP-NewReno. Then we used the Asymmetry index [49] to get an indication about which flow is having more/less share than the other flow.



(a) Bottleneck = 100 Mbps , RTT = 40 ms, Buffer = 50%, BDP = 250 pkts

(b) Asymmetry index

Figure 7.6: Friendliness to TCP-NewReno

Figure 7.6 shows the result of the experiments. It can be seen from figure 7.6b that over this range of RTTs, TCP-Gentle AI rule is more friendly to TCP-NewReno, this can be seen from the close-to-zero values of asymmetry index, which indicates better fair share of bandwidth when competing with TCP-NewReno.

Figure 7.6a shows the congestion window evolution for one of the RTT experiments, but instead of starting at the same time, the TCP-NewReno flow was started at $t = 100s$ and stopped at $t = 200s$. We note that at $t = 200s$ TCP-Gentle responds better than TCP-YeAH case, this is due to the rocket mechanism. Interestingly, the use of initial value of $\alpha_{rocket} = \alpha_{gh}$ in the rocket mechanism exhibits an s-shape congestion window evolution in the thrust phase, this is because at the end of this phase the algorithm enters to the damping phase with an initial

value of α_{gh}. Again this concave behaviour is desirable for the reasons mentioned in the previous subsection. We also run longer experiments, e.g. TCP-Gentle 20 minutes flow interrupted by a TCP-NewReno. The congestion window evolution is shown in figure 7.7.

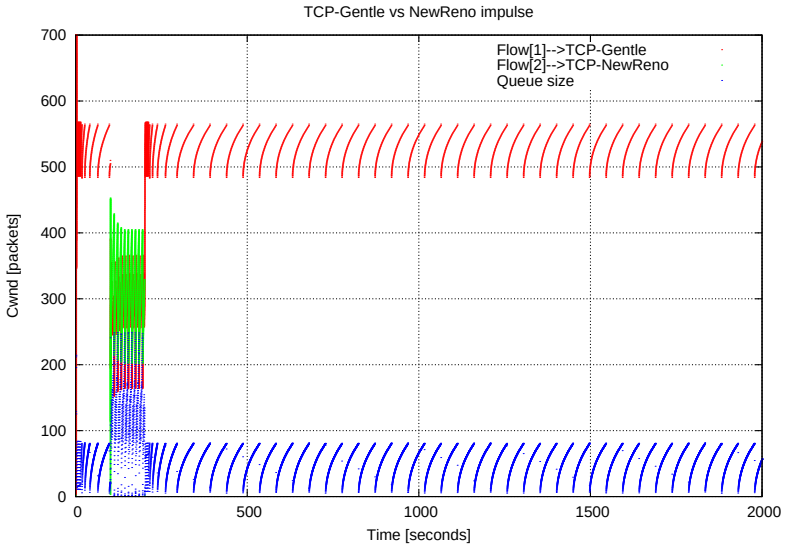


Figure 7.7: Bottleneck = 100 Mbps, RTT = 40 ms, Buffer = 50%BDP = 250 pkts

7.3.3 Effect of Web Traffic

In order to study the effect of web traffic on TCP-Gentle long-lived flows and queue size, we used an HTTP/1.0 background traffic generator code [52] for `ns2`. The generator's WWW Model implements a client and server sources to emulate the request and response traffic processes from typical web browsers and servers. The client object initiates a variable length request; after a random processing time, the server responds with a random number of connections of varying length. After a random viewing time (called "think time"), the client issues another request. The same empirical distributions were used to dictate the various random quantities. However, we modified the code to use GNU/Linux agents, this allowed us to have a typical traffic from a GNU/Linux sources, we used 340 sources giving roughly 4Mbps of web traffic load. We added this traffic to a long-lives TCP-Gentle flow and used a bottleneck capacity of 300 Mbps, RTT of 92ms and buffer size of 5% BDP (172 pkts). Note that the web traffic shares about 1.3% of the bottleneck capacity and the rest 98.67% is used by the bulk transfer of TCP-Gentle flow.

Figure 7.8 depicts the queue size for a single flow TCP-Gentle, with and without the web traffic load, left upper plot. The left bottom plot, shows the case when the algorithm is replaced by TCP-YeAH. The right vertical plot shows average values over 50 seconds. We note that, when web traffic load is added the

average queue size reduces in the case of TCP-Gentle and increases in the case of TCP-YeAH. The reason for the increase is indeed TCP-YeAH aggressive rule. On the other hand the reason for the decrease is because the added traffic increase the number of backoffs of TCP-Gentle and because TCP-Gentle additive rule in the damping phase, it takes longer time to reach the threshold after a back off, this results in less number of TCP-Gentle packets in the buffer.

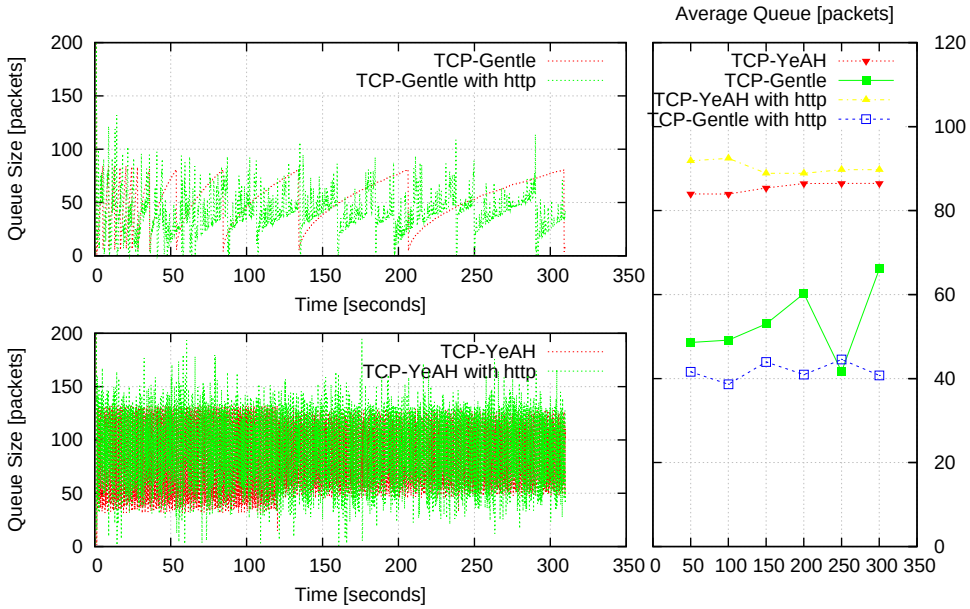


Figure 7.8: Effect of Web Traffic

One interesting observation is the drop in the average queue in the case of TCP-Gentle at $250s$. The reason behind that is because the α value is damping and the lowest appears in the last decongestion epoch $(200s - 300s)$, so we expect a drop in the queue size in this epoch. In addition to that, it is more likely to be in the first half of the epoch $(200s - 250s)$ just after the back off of the previous epoch which would have probably emptied the queue, and unlikely to be in the second half of the epoch $(250s - 300s)$ where the queue is most probably about to reach the threshold.

7.4 Real Test-Bed Experiments & Results

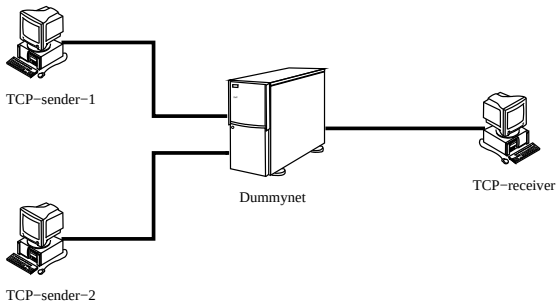


Figure 7.9: Topology

Parallel to our simulation experiments, we run experiments[5] on a real test bed. The aim was to test TCP-Gentle in high-speed long-delay environment, particularly the objectives of our experiments are:

Single flow testing:

1. Examine basic concepts of single TCP-Gentle flow in a high-speed long-delay environment;
2. Measure TCP-Gentle response function.

Double flow testing:

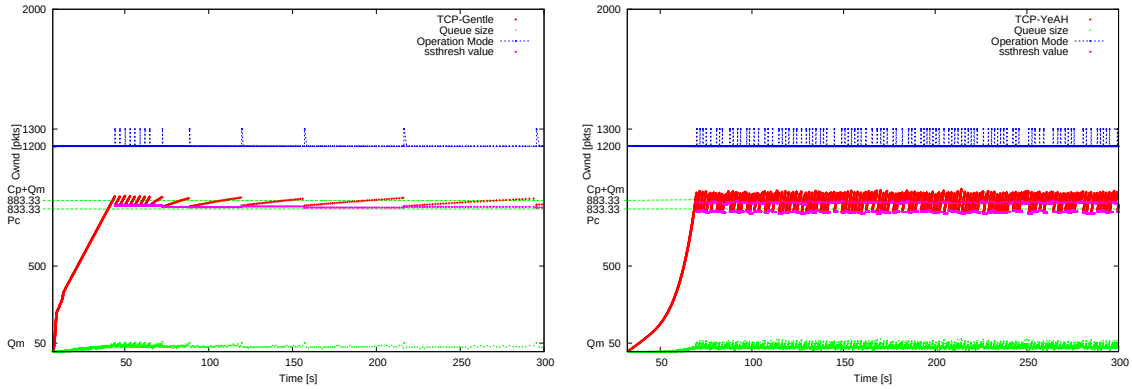
1. Measure Intra-protocol fairness/RTT-unfairness;
2. Measure Inter-protocol fairness (TCP-NewReno).

Figure 7.12 depicts the arrangement of the test bed. The experimental test bed consists of four Linux boxes, one of which is used as a router running `dummynet`, the other three PCs running `Iperf` (two TCP senders and a receiver). All PCs have two processors each is Intel(R) Pentium(R) 4 CPU clock frequency 3.20GHz. 1GB RAM. The network interface cards used are all 1 Gigabit Ethernet. All PCs have Linux-2.6.33.3 installed on them with TCP-Gentle enabled on the TCP senders. We used `tcpprobe` module to probe the congestion window and the slow start threshold values, we also modified the code to probe the sender size queue estimate and the `do_reno` flag (to identify the mode of operation). We used a bottleneck capacity of 100 Mbps and a packet size of 1500 bytes in all experiments in this section, we also changed the queue size threshold (i.e. Q_{max}) to 50 packets instead of 80 packets, we did that to study the effect of reducing the queue threshold, since we already studied a value of 80 packets in the previous section. In addition

[5] A step by step procedure is mentioned in the appendix

to changing the queue size threshold, we disabled slow start algorithm for TCP-YeAH and TCP-Gentle in all experiments in this section, we did that to study the operation of newly introduced ideas without the effect of any other congestion control algorithms operating inside our algorithm. In the following subsections we elaborate on findings related to each of the aforementioned objectives.

7.4.1 High BDP Operation



(a) TCP-Gentle (b) TCP-YeAH

Figure 7.10: Bottleneck = 100 Mbps , RTT = 100 ms, large buffer

In the first experiment, we used a single flow running TCP-Gentle, then repeated the experiment for TCP-YeAH. We emulated a high BDP environment by setting the bottleneck capacity to 100 Mbps and the propagation delay to $50ms$ i.e. RTT of $100ms$, this gives a BDP of 833.33 packets[6]$(100 \times 10^6/(8 \times 1500))$. This is the pipe capacity and we denote it by C_p. The algorithm is allowed to keep Q_m in the buffer, where $Q_m = 50$ packets, thus we have a margin of 50 packets after reaching the pipe capacity, in other words, in gentle mode the congestion window can grow to a maximum value of $C_p + Q_m$ before a decongestion back off takes place.

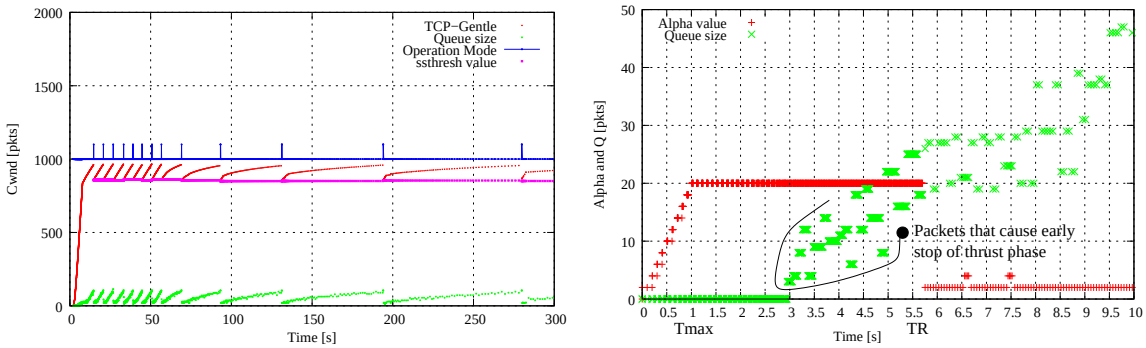
In figure 7.10 we show a plot of the congestion window evolution for both experiments,the status of the `do_reno` flag is indicated as pulses (the top line), if `do_reno == 0` the algorithm is in the gentle mode, otherwise its in the reno mode. To plot this information on the same plot of congestion window, we scaled the flag value, i.e. if `do_reno == 0`, the operation mode[7] is set to 1200 and the algorithm is in gentle mode, otherwise the operation mode is 1300 and the algorithm is in reno mode. The resultant pulse train gives information about the frequency of

[6]BDP is usually expressed in bits, but we express it in terms of packets for convenience, since we use packets in the context of the discussion

[7]This is the value is represented by top line and it is unit less

switching between the two modes and when it happens. Obviously, TCP-Gentle has less number of pulses due to its gentle rule of increase, while TCP-YeAH has more pulses due to its more aggressive rule in the fast mode. In other words the queue size threshold is exceeded less number of times in TCP-Gentle compared to TCP-YeAH, this supports our argument about TCP-Gentle rule being less aggressive than TCP-YeAH.

Considering link utilisation, both algorithms efficiently utilise the pipe capacity, this can be seen from figure 7.10. We also run longer experiments of 20 minutes for TCP-Gentle, TCP-CUBIC and reported the average throughput from `Iperf`. TCP-Gentle transmitted 13.0 Gbytes in 1200.0 seconds; which maps to 93.2 Mbps, while TCP-CUBIC transmitted 13.1 Gbytes in 1200.2 seconds; which maps to 93.9 Mbps.



(a) Congestion window (b) Alpha & Queue size

Figure 7.11: Bottleneck = 100 Mbps , RTT = 100 ms, large buffer, Qmax=100 pkts, TPQ= 25 pkts

To study the initial average throughput, we used a single TCP-Gentle flow and stopped the flow at the end of the thrust phase, we repeated the process for ten times and reported the average value of the throughput, we compared the theoretical results calculated using equation 7.15 and equation 7.17 with the experimental results. We found that if thrust phase stops when $Q \geq 0$ (the current algorithm setup), the rocket mechanism stops early i.e. before reaching the capacity of the pipe. We found that when this happens the congestion window is $\approx \frac{C_p}{2}$. According to equation 7.17, this achieves 25% of the pipe capacity (in pkts/s) The reason behind that is the existence of some packets in the queue even when the link is not fully utilised, these packets are not persistent, they appear to be random, persistent packets only appear when the link is fully utilised.

To overcome this problem, we let the thrust phase stop after a higher threshold $Q \geq TPQ$ and set $TPQ = 25$ pkts, and set $Q_{max} = 100$ instead of 50 pkts, this prevents early stop of the rocket mechanism, in fact it stops only when the threshold (TPQ) is exceeded, when this happens the congestion window is $= C_p$

Table 7.2: Initial state average throughput when $TPQ = 25$ pkts, $Q_{max} = 100$

Experiment #	$T_R = 5$s Throughput [Mbps]	$T_R = 6$s Throughput [Mbps]	$T_R = 7$s Throughput [Mbps]
1	32.7	41.2	48.9
2	33.9	41.1	48.3
3	32.5	42.3	47.8
4	32.9	42.3	48.9
5	32.5	42.2	45.8
6	32.6	42.4	49.0
7	33.9	40.0	45.3
8	32.6	42.3	48.4
9	33.9	42.1	49.5
10	32.6	41.2	48.3

and 50% of the pipe capacity is achieved. However we note that there is no problem of stopping the rocket mechanism early, the only issue is that the algorithm will be more conservative. at the end of the thrust phase, which agrees with the gentle behaviour of the algorithm..

Figure 7.11 shows the congestion window evolution and the AI increase value of one of our experiments, it can be seen from the congestion window evolution plot, that the thrust phase stops when the capacity of the pipe is reached, unlike the original case, for example figure 7.10a. Theoretical values of T_{max} and T_R are calculated using equation 7.12 and equation 7.15, and they are: $T_{max} = 1$ second, $T_R = 4.67$ seconds. Experimentally we found that the values are: $T_{max} = 1$ second, $T_R = 5.68$. The average throughput calculated using equation 7.17 is 41.8 Mbps, table 7.2 summarises the average throughput of the of the initial state for ten runs for durations of T_R and near it.

Over all, TCP-Gentle has the advantage of putting less load on the network while achieving the high performance of its sibling TCP-YeAH and other algorithms like TCP-CUBIC. We finish this subsection by two remarks, the first is that TCP-CUBIC is a loss-based algorithm i.e. it follows a reactive approach, while TCP-YeAH and TCP-Gentle follow a proactive approach (they react before loss). The second is that in the original case for TCP NewReno the slow start threshold *sshthresh* is changed upon back off (loss or time out) and considering TCP congestion control algorithms, algorithms adjust this value differently. However, Linux TCP code adaptively change this value even for the original case of TCP-NewReno , it sets an infinite initial value (`0x7fffffff`) and if $cwnd > ssthresh$, Linux may rise the *sshthresh* to half-way to *cwnd*, the exception is rate halving phase, when *cwnd* is decreased towards *ssthresh* (practically this happens exponentially, although it is theoretically referred to as MD). Details of this implementation is

mentioned in the Linux source code tree: `/include/net/tcp.h`.

7.4.2 Response Function

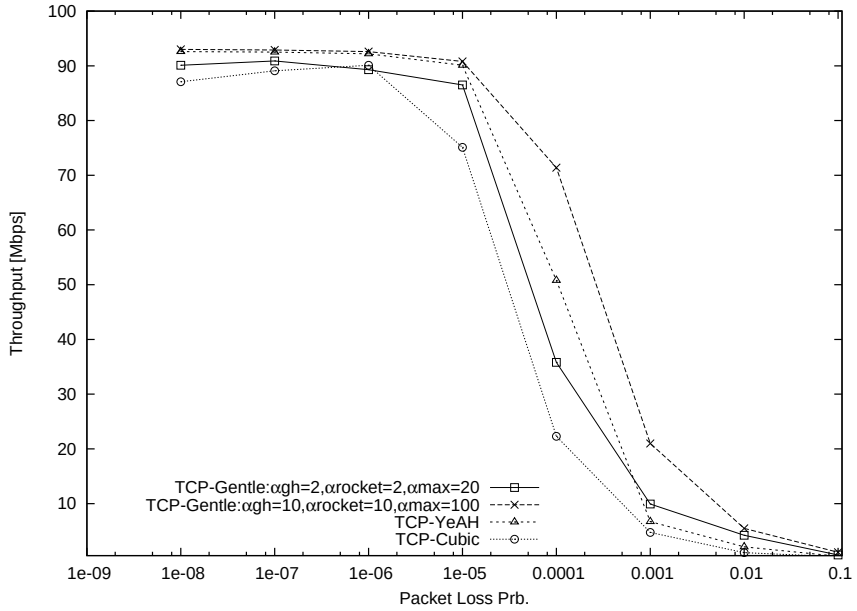


Figure 7.12: Response functions: bottleneck capacity = 100 Mbps, RTT = 100 ms, large buffer size

In this subsection we study the response of TCP-CUBIC, TCP-YeAH and TCP-Gentle to range of random packet loss probabilities.In our experiments we considered empirical *Response function*. In the setup in figure 7.9 we used a single flow running one of the algorithms, a bottleneck capacity of 100 Mbps, a RTT of $100ms$ and used the uniform packet loss functionality provided by `dummynet`, we used a range of packet loss probabilities [10^{-08},10^{-01}] and run a 5 minutes experiment for each probability in this range then reported the average throughput.

Figure 7.12 shows a plot of average throughput versus packet loss probability for the algorithms. We note that the response of TCP-Gentle is very close to that of TCP-YeAH, when we tried different values for TCP-Gentle parameters, we noticed a better response, this is indicated by increase in throughput at higher loss rates, however the price for that is indeed more aggressiveness. Since our aim is to reduce network load as much as possible, we would trade a tolerable decrease in responsiveness (response to packet loss) for reducing network load. We would like to mention that upon packet loss all algorithms use the SACK mechanism for retransmitting the lost packets, Linux implementation of SACK usually spend time to locate the missing segment, and this time is sufficient to cause time-outs which will result in less average throughput. We have experienced this in our experiments, we also found that this has been reported as a problematic issue for

Linux when working in high-speed long-delay networks. We did not disable SACK in our experiments and the reason was that we wanted to take into account all implementation aspects from a worst case point of view.

7.4.3 Intra-Protocol Fairness & RTT-Unfairness

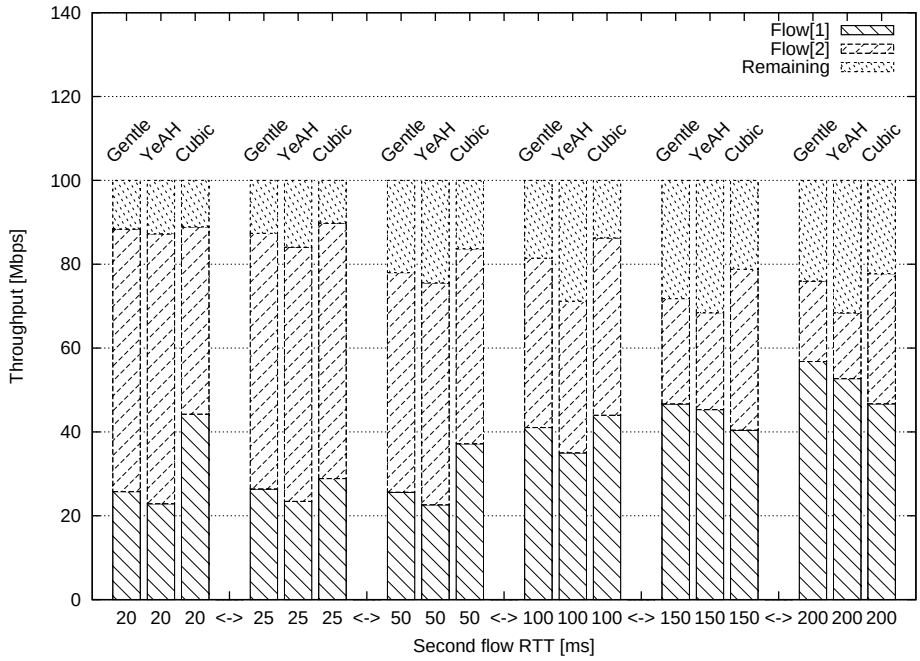


Figure 7.13: Algorithm fairness & RTT-Unfairness

We study the intra-protocol fairness of TCP-Gentle compared to TCP-YeAH and TCP-CUBIC, by intra-protocol fairness we mean fairness between flows running the same algorithm. Referring to figure 7.9, we used two flows running the same algorithm and one bottleneck with a capacity of 100 Mbps. The RTT for the first flow is fixed at $100ms$. The RTT for the second flow is varied as: $20ms, 25ms, 50ms, 100ms, 150ms, 200ms$. The reason for that is to study the impact of RTT-unfairness. At $100ms$ both flows have the same RTT. Each experiments is repeated three times and the average value of the throughput is reported. We summarise our results in figure 7.13.

As can be seen from figure 7.13, all algorithms under test underutilise the pipe capacity, this is indicated by the upper shaded areas. The reason for that is each flow tries to get half the the pipe capacity which is in this case 50 Mbps. We found that trying to utilise this bandwidth using the RTTs in this experiment result in excessive time-outs. In addition to that we found that, when SACK is enabled, after a time-out the congestion window is halved instead of going back to one packet and when SACK is disabled the congestion window goes back to one packet after each time-out (original case). In other words the upper shaded

areas in the figure would have been larger if the SACK is disabled. We also note that this underutilisation gets worse as the RTT increases i.e. the network gets sluggish and time-outs increase. However using, larger bottleneck capacity e.g. 300 Mbps makes each flow struggle for 150 Mbps which would indeed show less time-outs and thus better link utilisation i.e. less upper shaded areas.In parallel, it is easy to forget that `dummynet` is also queueing packets in the network (this is way the emulator works), this can create increased delay and eventually time-outs, we also found that such cases are also reported in literature for high-speed long-delay networks.

TCP-YeAH and TCP-Gentle exhibit clear RTT-unfairness as the RTT of the second flow is varied and it is worse as the the RTT increase difference increase, however TCP-Gentle perform better than TCP-YeAH, and the reason behind that is believed to be the MI rule of TCP-YeAH. TCP-CUBIC is very immune against RTT-unfairness and the reason behind that is that increase rule is not a function of RTT and it is updated as a function of real time, in fact the algorithm was optimised to achieve this goal, although the desirability of this goal is arguable in literature we quote [34, p.11]:

> "We note that there is not a consensus in the networking community about the desirability of this goal, or about the implications and interactions between this goal and other metrics [FJ92] (Section 3.3). One common argument against the goal of fairness between flows with different round-trip times has been that flows with long round-trip times consume more resources; this aspect is covered by the previous paragraph. Researchers have also noted the difference between the RTT-unfairness of standard TCP, and the greater RTT-unfairness of some proposed modifications to TCP [LLS05]."

When both flows have the same RTT, we note that TCP-Gentle performs better than TCP-YeAH and its performance is similar to that of TCP-CUBIC. We note that TCP-YeAH substantially underutilise the pipe capacity compared to the other two algorithms. As a result, TCP-Gentle unique AI rule provides better fairness and RTT-unfairness compared to TCP-YeAH and thus has the potential to be an improvement from this perspective.

7.4.4 Friendliness to TCP-NewReno

In order to study the interaction of TCP-Gentle with standard TCP-NewReno, we used a bottleneck capacity of 100 Mbps and a RTT of $14ms$ in the setup shown in figure 7.9. This setup is a typical setup for TCP-NewReno normal operation and

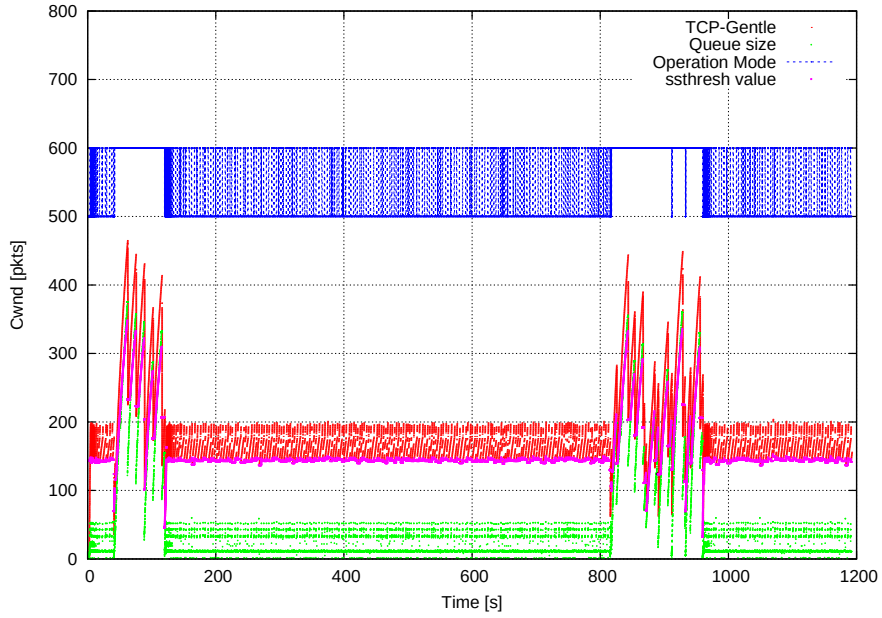


Figure 7.14: Bottleneck = 100 Mbps, RTT = 14ms, large buffer size

it is not a high speed long delay setup. We used a single long-lived TCP-Gentle running for 20 minutes, during this period we let another TCP-NewReno flow start at a random time ($\approx$ 40 seconds) and last for 1 minute, then we repeated the process again at random time ($\approx$ 815 seconds) but this time we let the flow last for 2 minutes. We also reported the `do_reno` flag, but this time we scaled it between 500 and 600 to be able to plot it on the congestion window plot.

The result of the experiment is shown in figure 7.14. The figure shows the double impulse response of the algorithm, only the congestion window of TCP-Gentle is plotted, we note the dark areas on of the congestion window at the beginning of the connection and when TCP-NewReno exits, these are resets of the gentle mode and thus the algorithm starts with the thrust phase then enters the damping phase (with α_{gh}) after the queue size threshold is reached, following that it quickly reaches steady-state. Considering the `do_reno` flag, if the value is 600 this means that the algorithm is in reno mode and if the value is 500 this means that the algorithm is in the gentle mode, thus the upper plot shows the frequency of switching between the two modes of operation. We note that the algorithm spends more time in the reno mode when it is competing the NewReno [8] flow. The reason for the large values of congestion window when competing with a NewReno flow is the large buffer size. And since NewReno and TCP-Gentle in reno mode are loss-based algorithms, the buffer is highly occupied, this can also be seen from the sender size queue size (the bottom plot).

[8] This mechanism of switching is borrowed from TCP-YeAH

7.5 Summary

In this chapter we discussed in details our new TCP congestion control algorithm: TCP-Gentle. The new algorithm is an incremental development over TCP-YeAH, which is an algorithm optimised for high link utilisation while maintaining minimum network load and at the same time being friendly to standard TCP-NewReno. Our algorithm uses different increase rules depending on network conditions. It uses an adaptive additive increase rule instead of multiplicative increase. It is believed to be more gentle to the network and has better fairness properties than YeAH-TCP, has a nearly flat rate in steady-state while at the same time maintains high responsiveness, high link utilisation and friendliness to TCP-NewReno.

We derived a deterministic model for the new algorithm, this is basically a theoretical expression for the initial and steady-state average throughputs. We implemented the new algorithm as a Linux kernel module and showed through theoretical analysis and simulation/real experiments that the algorithm is even more gentle to the network and shares bandwidth more fairly and thus is a competitive algorithm to existing algorithms. We discussed some of our experience dealing with the Linux TCP code wherever it was needed. This finalise the contributions of this dissertation on this track. In the next chapter we move to another TCP problem related to non-congestive packet loss which is another challenge to TCP protocol.

Chapter 8

A Loss Differentiation Algorithm

In this final chapter, we address another TCP congestion control problem, the problem arises from the fact that TCP makes a tacit assumption that each packet loss is caused due to congestion and based on that TCP reduces the congestion window. Due to the existence of new technologies such as optical fibre, wireless networks, etc, coupling the packet loss with congestion is no more accurate. To elaborate on this point, some studies have shown that such non-congestive loss are not uncommon in WLANs and can have tragic impact on TCP throughput, for example; a test on a single WLAN with 15 wired links showed that the throughput in the absence of loss was 1.5 Mbps, while with independent frame loss of 2.3%, frame size of 1400 bytes the actual throughput was 0.7 Mbps and for a WAN case the actual throughput was 0.3 Mbps, a test for IEEE 802.11 in the absence of loss showed that the throughput was 2 Mbps which dropped to 0.98 Mbps when the same percentage of loss was introduced [51]. Applying these figures to IEEE 802.11g bandwidths (up to 54 Mbps) which are extensively used nowadays, the 46.66% and 22.96% and 49% drop in throughput can be significant. In addition to that the problem can get even worse with the existence of clustered errors where multiple errors in a congestion window usually result in a time-out forcing TCP to effectively reset its congestion window. Another complication, is the way TCP operates, in a shared WLAN medium forward TCP traffic contends with reverse ACKS leading to undetectable collisions which significantly increase the Frame Error Rate (FER) visible to higher layers [51]. Considering wireless networks in general, there are other TCP congestion control problems associated with cellular communication systems (e.g. GSM, satellite system) and even the heterogeneity of wireless systems, but these are out of the scope of this thesis.

Techniques such as ECN can help in mitigating the problem, however they do not provide a complete solution [81, p.10]:

> "... if a CE packet is dropped later in the network due to corruption

> (bit errors), the end nodes should still invoke congestion control, just as TCP would today in response to a dropped data packet. This issue of corrupted CE packets would have to be considered in any proposal for the network to distinguish between packets dropped due to corruption, and packets dropped due to congestion or buffer overflow. In particular, the ubiquitous deployment of ECN would not, in and of itself, be a sufficient development to allow end-nodes to interpret packet drops as indications of corruption rather than congestion."

There have been many attempts to solve the non-congestive loss problem and the proposed algorithms are usually referred to as Loss Differentiation Algorithms (LDAs), some do not use this term but rather integrate the solution as part of the algorithm/protocol design, like for example the explicit loss notification (ELN) [10], which relies on the queue length at the wireless link gateway to differentiate between the type of losses. Some of the High-Speed TCP algorithms [69] use the queueing delay.

In this chapter, we present a novel heuristic which is based on sensing and quantifying what we call *noise* and *span* in packet inter arrival times (PITs) of duplicate ACKs in the *Fast Recovery* phase. After that the multiplicative decrease factor is adapted according to a logical formula.

The rest of the chapter is organised as follows: in section 8.1 we provide a discussion of the assumptions used to build this heuristic for loss differentiation. Then we provide a prolonged theoretical discussion of our thoughts about the behaviour of PITs under different network scenarios, first we look at different PITs distributions in section 8.2 then we look at the PITs in presence of packet drops in 8.3. In section 8.4 we present our logical formula which we use to adapt the multiplicative decrease based on the noise and span in PITs. In 8.5 we discuss our simulation results and finally we summarise in section 8.6.

8.1 Assumptions for a New Algorithm

In [14] and [2], a heuristic was developed for discriminating congestive loss from wireless loss based on PITs. The heuristic works under several assumptions: the sender is on a wired network and the receiver is connected via a wireless link, the wireless link is the bottleneck link for the connection (often valid in cellular environments), processing time at the receiver is negligible, only the last link in the path is wireless, and finally the sender performs a bulk data transfer.

In [59], PITs were analysed in a totally different context and were used in detecting bottleneck in a passive way. We look at the problem of discriminating

congestive loss from wireless loss from a different angle, combine some of the previous work and add a new technique that attempts to discriminate loss even if the bottleneck link is not the last link in the connection (implies other wireless environments not just cellular).

We analyse a sequence of packets only when they arrive out of order, in the following manner: let SQ_0, SQ_1, SQ_2 be the first, second and third packet sequence number of the first three captured packets and all belong to the same flow (TCP connection). If $SQ_1 \neq SQ_0+1$ and $SQ_2 \neq SQ_0+1$ then capturing continues to $np+1$ of packets with SQ_{np} being the last packet. On the sender side this is equivalent to capturing the first three duplicate acknowledgements, with the exception that if three duplicate acknowledgements are received capturing continues for the rest of the duplicate acknowledgements until receiving the acknowledgement of the retransmitted packet. In other words, np is determined by the number of duplicate acknowledgements. This means that the captured samples contain at least one out of order sample which implies at least one packet loss event. From sender side implementation point of view this can help us implementing the new technique in TCP *Fast Recovery* algorithm [89] and adjusting the congestion window according to type of loss.

We maintain all the assumptions mentioned in [14], except that the wireless link is not necessarily the bottleneck link. Now, depending on the path traversed, we assume that the above PITs, have the following cases:

A) Congestive loss case: PITs for out of order packets have *no pattern* compared to wireless loss case and the difference between the maximum PIT and the minimum PIT in our samples is *less* than the difference in wireless loss case.

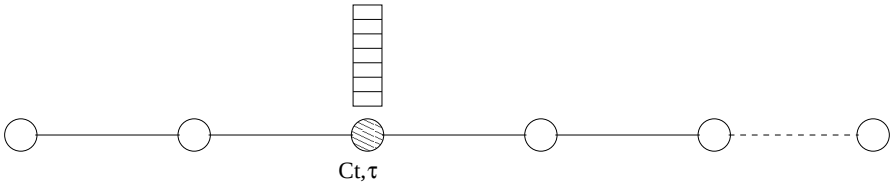
B) Wireless loss case: PITs for out of order packets have a *pattern* compared to congestive loss case and the difference between the maximum PIT and the minimum PIT in our samples is *greater* than the difference in the congestive loss case.

By "pattern" we mean that PITs take certain finite values. Our discrimination approach can be informally stated as follows: if the difference between maximum and minimum PITs is small then make the assumption that this is a congestive loss, if the difference is large then an ambiguity arise (could be non-congestive or congestive plus large cross traffic interference causing this large difference), in this case; use the randomness in PITs as another signal, if randomness is high then make the assumption that this is a congestive loss, if randomness is small then make the assumption that this is non-congestive loss.

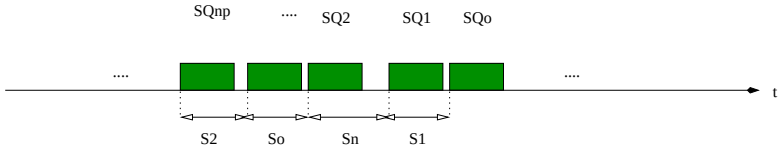
To see how these assumptions can be valid, let us consider the following two cases in a multi hop network, a typical arrangement for a multi hop network is a

hybrid network (wired-wireless), obviously this is not the only arrangement, there could be a complete wireless, but hybrid networks are common infrastructure networks.

8.1.1 Single Flow without Cross Traffic



(a) Typical single flow scenario with path capacity C_t and transmission time τ



(b) Out of order packets at the receiver

Figure 8.1: Single flow without cross traffic

It is possible for a congestion to occur without having any cross packets between our single flow packets, this could happen for example if the sender rate overloads the capacity of the bottleneck link. In this case packets spacing are still relatively close, because the spacing is approximately the transmission time of the bottleneck link. Figure 8.1a depicts a typical arrangement for this scenario. We denote by C_t the capacity of the path and by τ the transmission time. Solid lines means a wired link and dashed lines means a wireless link. Each circle represents a node (typically a router), the shaded circle represents the node that has the minimum capacity link attached to it from the right, we call this node the *bottleneck node*. In this case if a packet drop happens due to congestion (drop from the queue[1]) the difference between the maximum PIT and the minimum PIT in our samples can be relatively close because packets that survive from loss need to be transmitted by the bottleneck link, so the spacing should be $\approx \tau$. On the other hand, if a packet drop happens due to wireless link the difference between the maximum PIT and the minimum PIT in our samples, can be much greater because out of order packets will not traverse any link after the last link. In addition to that, bursty loss is not uncommon in wireless links. Figure 8.1b shows our single flow out of order packets, SQ_0 is the sequence of the first packet and SQ_{np} is the last one, $np+1$ is number of captured packets. Packets inter arrival times are denoted

[1]Can have active or passive management policy

by S, where $S_0 \leq S_1 \leq S_2 \ldots \leq S_n$ and the number of samples is $n+1$. In this case $S \approx \tau$, if we plot the distribution of the PITs we get a single major spike at the transmission time of the bottleneck capacity, τ. In our discussion a "major spike" is synonym to a "pattern". So in this situation we have small difference between the maximum and minimum PITs and a pattern. The discriminator decision based on the difference between the maximum and minimum PITs appears to be valid in this case.

8.1.2 Single Flow with Cross Traffic

It is possible to have cross traffic joining our single flow packets during their travel from source to destination across the multi hop path and exit the path at some point later, cross traffic interference during congestion (queue build-up) adds randomness to our single flow PIT, this is due to queue build-up and queue empty processes, because each cross traffic packet that interferes with the single flow packets represent a time gap in future time (when the cross traffic packet exits the path) these time gaps are subject to variations along the path, starting from the random arrival of cross traffic then bottleneck node processing time, any other future queue build-up, or other cross traffic, etc. In addition to that cross traffic packets can have different sizes (e.g. ACK packets which have smaller size).

If packet loss happens in this situation (likely due to congestion), an ambiguity appears: is the difference between the maximum and minimum PITs due to cross traffic interference or is it the large difference that appears in a non-congestive loss (when there is at least one large gap in PITs and the rest are small i.e. back-to-back)?

To help resolving this, we use another signal which we believe it is coupled with the existence of cross traffic, we expect the single flow PITs in our captured sample to have *significant* randomness due to cross traffic interference during congestion. In contrast, if there is no congestion (queue build-up) and a packet drop happens (likely due to wireless link transmission error), the single flow PITs in our captured sample are expected to have *less* randomness, since the packets are dropped at the last link, the time gaps have less chance to change and the PITs are approximately multiples of the bottleneck link transmission time (depending on how many packet drops happened at the last link). Therefore, this randomness in PITs can help our discriminator to make the discussion about the type of packet loss in this case.

Two points to note here: Firstly, lower layer wireless protocols adds randomness to both situations. Secondly, an intuitive question that stems from the above discussion is: What if a packet drop due to wireless link transmission error happens during congestion? Since the overall pattern is random in a congestion situation

the packet drop will be assumed to be due to congestion. In fact it is better to be conservative in our assumptions, it has been shown [15] that it is much more costly to mistakenly identify a congestive loss as a non-congestive one than the other way round.

Another point to mention is that it is possible to have a cross traffic interference without having a congestion, this means that either there is no queue build-up or the queue build-up and empty between two single flow packets (before the second packet arrive at the node), both situations have limited effect on the pattern. Any packet drop in this case will be recognised as wireless link transmission error based on case B. Note that the network is not congested here, this can be viewed as temporary or transient congestion that disappears after a short time.

8.2 PITs Distributions

In this section we examine different PITs distributions in existence and absence of cross traffic. Where spikes are considered patterns, the areas between these spikes, in fact; constitute the randomness[2] that we were talking about. Hence, the wider the spectrum, the more likely to have randomness in PITs. Our aim in this section is to show that cross traffic interference can have this wide spectrum.

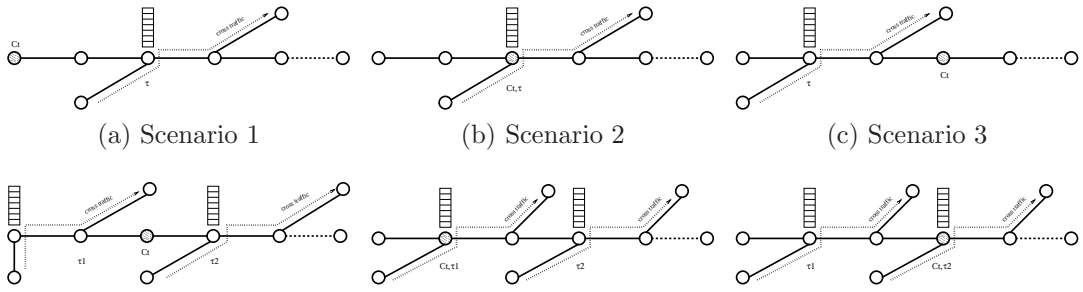
8.2.1 Single Flow without Cross Traffic

Referring to figure 8.1, in normal operation; our single flow packets arrive back-to-back at the receiver separated by the bottleneck transmission time, maintaining the assumption that the sender performs bulk data transfer, the PIT distribution is expected to have single spike at the bottleneck transmission time, in this case τ.

8.2.2 Single Flow with Cross Traffic

There are many scenarios where this could happen, we break these into six scenarios (other scenarios can be combinations of these six scenarios). In figure 8.2 all the conventions in figure 8.1a apply. It is possible to have a queue build-up at C_t, but we are not showing this queue to avoid confusion with the queue build-up due to cross traffic. The bottleneck transmission time is denoted by T_c, the transmission time of the link attached to right to the node which has significant queueing due to cross traffic is denoted by τ, in case there are more than one, they

[2]In reality, there are minor i.e. very small spikes between the major spikes, we refer to these as "areas" without drawing them, so the term "spike" can retain the meaning of a pattern. Plots from real experiments are available in literature [59]



(a) Scenario 1 (b) Scenario 2 (c) Scenario 3

(d) Scenario 4 (e) Scenario 5 (f) Scenario 6

Figure 8.2: Single flow with cross traffic

are denoted by $\tau_1, \tau_2, \ldots$. The flow of packets is from the left to the right, with our single flow packets arriving with an initial time spacing denoted by T_i at the first node from the left and determined by the minimum transmission time in the path traversed so far (not shown in the figure) and packet size, however if the first node from the left is the sender, then T_i is determined only by sender rate and packet size. In scenarios 1-6, $T_c \geq T_i, T_c \geq \tau$

In the analysis of these scenarios we are concerned in a three major points:

- The location of the bottleneck node;
- The location of the queue due to cross traffic;
- The packet drop event (discussed in the next section).

We leave the last point to the next section. Let us start by looking at scenario 1, our single flow packets get re-spaced from T_i to T_c at the bottleneck node (first from left in figure 8.2a), then they are queued at the third node where cross traffic starts joining our packets in their journey to the destination. Due to cross traffic interference and queue build-up, PITs get expanded or squeezed[3] by multiples of τ depending on the degree of congestion. Some practical cases that illustrate this behaviour are shown in [59]. As a result, the distribution of PIT is expected to have a major spike at T_c (because queueing is transient and most packets will arrive spaced approximately by the bottleneck transmission time T_c in the absence of the queue at node three) and minor spikes at the multiples of τ. Please see figure 8.3a, scenario 1. It is not necessary for the spikes to be symmetrical, this could depend on the severity of cross traffic interference but several symmetrical cases were reported in practical measurements made in [59].

Scenario 2 can be analysed similarly, but now cross traffic joins our single flow packets at the the bottleneck node, thus $T_c = \tau$, PITs get expanded but cannot get

[3]Note that squeezing could happen if single flow packets queue behind cross traffic packets and is possible because τ is less than T_c

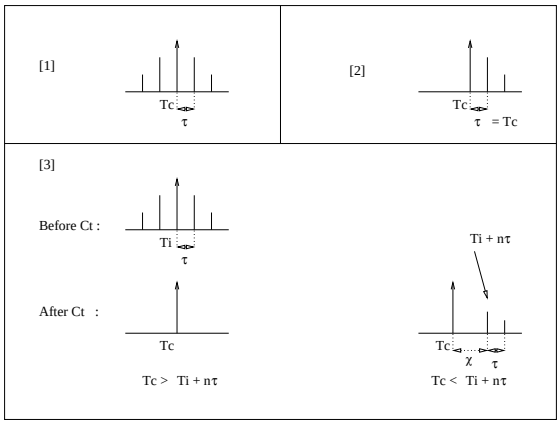
squeezed (less than T_c), the distribution of PIT is expected to have a major spike at T_c and minor spikes at the multiples of it only from the right. See figure 8.3a, scenario 2.

In scenario 3, the cross traffic joins our single flow packets before the bottleneck node, so the queueing at the second node disturbs the initial PIT, T_i by expanding and squeezing at multiples of τ. However, after passing the bottleneck link things change depending on the amount of of expansion in T_i. If $T_c > T_i + n\tau$, where n is the number of cross traffic in between, the bottleneck link re-spaces the packets to T_c. If $T_c < T_i + n\tau$, then packets can escape from the bottleneck link without being re-spaced. The distribution of PIT is expected to have only one major spike at T_c when $T_c > T_i + n\tau$ and a major spike at T_c followed by one minor spike on the right at a random time determined by n in $T_i + n\tau$, we denote this by X. After that, minor spikes at multiples of τ could appear on the right. See figure 8.3a, scenario 3.

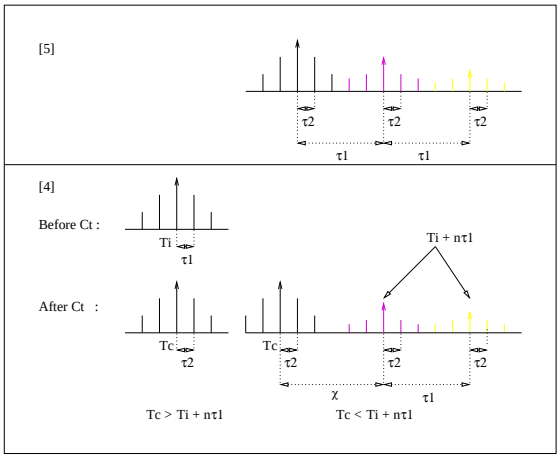
Scenario 4 can be seen as a combination of scenario 1 and scenario 3, in this case we have two queues due to cross traffic, one at node one and the other at node four, each followed by two links with transmission times of τ_1, τ_2 respectively, and the bottleneck is in between. We analyse the case where $\tau_1 > \tau_2$. Before the bottleneck node, the initial packets spacing T_i gets expanded and squeezed by τ_1 at node one, again squeezing is possible because τ_1 is less than T_i. If no packets escape from the bottleneck link re-spacing (as in scenario 3) then all packets leave the bottleneck link re-spaced by T_c. After the bottleneck node our single flow packets still have one queue barrier at node four where again their spacing gets expanded and squeezed by τ_2, squeezing is possible because τ_2 is less than T_c. However if some packets escaped with a spacing of $T_c > T_i + n\tau_1$ each can get further expanded or squeezed by τ_2, squeezing is possible because τ_2 is less than their spacing. See figure 8.3b, scenario 4. figure 8.3c scenario 4, depicts the case when $\tau_1 < \tau_2$.

Scenario 5 can be seen as a combination of scenario 1 and scenario 2, in this case we have two queues due to cross traffic, one at the bottleneck node (node two) and the other at node four, each followed by two links with transmission times of $\tau_1 = T_c$, τ_2 respectively. We analyse the case where $\tau_1 > \tau_2$. Before the second queue, the single flow packets are spaced by τ_1 or expanded by multiples of it (similar to scenario 2), each of these spacings can get squeezed or expanded by multiples of τ_2 (note that they might remain unchanged in the absence of the second queue, but here we are considering a queue build-up). See figure 8.3b, scenario 5. figure 8.3c scenario 5, depicts the case when $\tau_1 < \tau_2$.

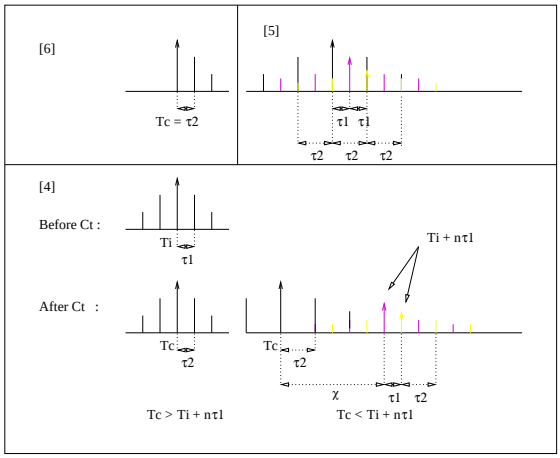
Finally, scenario 6 can be seen as a combination of scenario 2 and scenario 3, since $T_c = \tau_2 > \tau_1$, the second queue is expected to dominate the packet spacing



(a) Scenarios 1-3.



(b) Scenarios 4-6, $\tau_1 > \tau_2$.



(c) Scenarios 4-6, $\tau_1 < \tau_2$.

Figure 8.3: Hypothetical PIT distributions for scenarios 1-6.

in this scenario, most packets are expected to arrive spaced by the bottleneck transmission time T_c along with other expanded (but not squeezed) in time at multiples of τ_2. figure 8.3c scenario 6, depicts the PITs distribution.

As mentioned before, there are other scenarios, for example having two or more queues (due to cross traffic) before the bottleneck, or after it. This adds more time spacing variations and the analysis of such cases will not add to the discussion; since we have already seen similar time spacing variations in the six scenarios. Most of these scenarios have a wide spectrum of PITs and the areas between their distribution spikes are potential values for PIT, i.e. PIT can take values between these spikes in reality, see for example [59]. We believe that this supports our assumption about randomness in PITs during cross traffic interference.

8.3 Analysis with Packet Drop

In this section we analyse the case of single flow without cross traffic and scenario 2 of the other case, we also explain a new technique which we use for extracting the

noise from the samples in order to quantify it.

8.3.1 Single Flow without Cross Traffic

The cases mentioned before are ideal cases, they are mentioned for pedagogical purposes to illustrate the concept. In reality PITs are not separated exactly by fixed time spaces, this is due to variations in the delay along the path: propagation delay, queueing delay, packet re-spacing, node processing times, in addition to packet re-spacing that can take place along the path, lower layer protocols behaviour (e.g. frame retransmission, not significant in wired links compared to wireless links), in addition to cross traffic interference during congestion, where a packet could arrive while the another is being served[4]. To take this into account, we assign a noise factor $\pm\delta$ to each sample. We looked at single flow out of order packets in figure 8.1b, we refer the same figure in our discussion. There are $np+1$ packets and their sequence numbers are denoted by: $SQ_0 \dots SQ_{np}$ and captured upon a packet loss trigger. Samples are denoted by: $S_0 \leq S_1 \leq S_2 \dots \leq S_n$. Each sample is represented as: $xT \pm \delta$, where:
T: Packet transmission time of the minimum capacity link in the path (all packets arrive back to back spaced by this time in normal operation);
x: Number of multiples of T, $\in \mathrm{Z}^+$;
$x_0 \leq x_1 \leq x_2 \dots \leq x_n$, $x_i \leq x_{i+l}$, $0 < x_0 \leq x_i \leq x_n$, $0 \leq i \leq n$;
$S_{min} = x_0T \pm \delta_0$ and $S_{max} = x_nT \pm \delta_n$;
δ: Noise factor, $\delta \ll T$;
np: Maximum number of packets. $\in \mathrm{Z}^+$;
$n\ell_i$: Number of lost packets in each sample (between two out of order packets),$\in$ Z^+, $0 \leq i \leq n$;
n: Maximum number of PITs (samples) to be analysed. $n \in \mathrm{Z}^+$;
$n = np - 1$;
$x_i = n\ell_i + 1$.
Because out of order packets arrive spaced by $\approx T$ in case of congestion, and $\approx xT$, $x > 1$ in case of wireless loss, we rely on the span $1 - S_{min}/S_{max}$ (we explain this formula later on in this chapter) to discriminate the type of loss in this case. A large span indicates a wireless loss and a small one indicates congestive loss.

8.3.2 Single Flow with Cross Traffic

This section includes the analysis of scenario 2 in the presence of packet drop. Scenario 2 was chosen for its simplicity, since most of the events happen at the

[4]This affects the time spacing between packets [59], which we will rely upon in discriminating the type of packet loss.

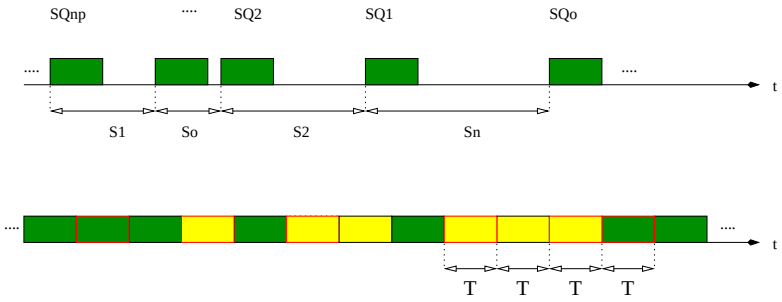


Figure 8.4: scenario 2, bottom: cross traffic and packet drop, top: after cross traffic leaves the path.

same location (the bottleneck node).

Each sample is represented as $xT \pm \delta$, where $T = T_c = \tau$. Figure 8.4 depicts a hypothetical PIT pattern for this scenario. Single flow packets have the green colour, and cross traffic have the yellow colour. Lost packets from both have a red contour. The samples are:

$$\begin{aligned} S_0 &= x_0 T \pm \delta_0 \\ S_1 &= x_1 T \pm \delta_1 \\ S_2 &= x_2 T \pm \delta_2 \\ &\vdots \\ S_n &= x_n T \pm \delta_n \end{aligned} \tag{8.1}$$

If the packet drop happens due to congestion, the noise factor is expected to have large values compared to the case where it happens due to wireless link errors. In subsection 8.3.1 we used a quantified measure (the span) we shall find a way to quantify the noise factor in order to make a decision. Our first goal is to extract the noise factor (δs) from our samples, then as a second goal, to find a way to quantify the collected δs.

In order to extract the noise factor from the samples, we propose a recursive technique and call it a "Difference Canceller", the basic idea is to subtract all possible different combinations of the samples from each other, because xT takes finite values, after repeating the process for a certain number of iterations, the term xT diminishes at the same time the technique dumps residual terms (those less than T) each iteration. The block diagram in figure 8.5 depicts the operation of the recursive technique. The process continues until sufficient T terms are cancelled; leaving the residual noise factor terms. It can be shown that the maximum sample

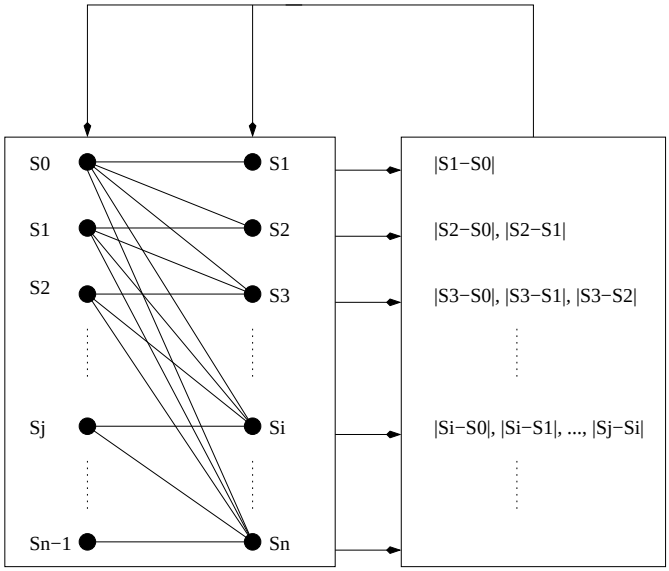


Figure 8.5: Difference Canceller

of the next iteration is always less than the maximum sample of the previous iteration, which means that the technique converges towards T. Now, we apply the technique to scenario 2, let k be the number of iterations, recall that $T = T_c = \tau$, we define $d_{i,j}$ to be the absolute difference between two samples, by taking all different possible combinations of samples we obtain a set Δ_k for each iteration, where:

$$d_{i,j} = |S_i - S_j| \qquad \text{, when } k = 1, 0 \leq j \leq n-1. \tag{8.2}$$

$$\Delta_k = \{d_{i,j}\}_{i>j} \qquad \text{, when } k = 1, j < i \leq n. \tag{8.3}$$

We need the number of samples in each iteration to calculate all possible different combinations, for example; for the first iteration:

$$\binom{n+1}{2} = \frac{(n+1)!}{2!(n-1)!} = \frac{(n+1)n}{2}$$

All output samples in Δ_k are considered inputs for the next iteration. The number of the new samples is $n' + 1$, where $n' + 1 = (n+1)n/2$, therefore there are $S'_0, S'_1, S'_1 \ldots S'_{n'}$

The samples for the next iteration are:

$$\begin{aligned} S'_0 &= |(x_1 - x_0)T \pm \delta_1 \mp \delta_0| \\ S'_1 &= |(x_2 - x_0)T \pm \delta_2 \mp \delta_0| \\ &\vdots \end{aligned}$$

$$\begin{aligned} S'_{n-1} &= |(x_n - x_0)T \pm \delta_n \mp \delta_0| \\ S'_n &= |(x_2 - x_1)T \pm \delta_2 \mp \delta_1| \\ &\vdots \\ S'_{n'-1} &= |(x_n - x_{n-2})T \pm \delta_n \mp \delta_{n-2}| \\ S'_{n'} &= |(x_n - x_{n-1})T \pm \delta_n \mp \delta_{n-1}| \end{aligned} \tag{8.4}$$

Below is the first and second iterations, the number to the right is the number of samples of the j^{th} set. One point to note here is that some noise factors can get cancelled in a sample due to the subtraction process but this doesn't mean that they are lost, because they appear in other samples, thus the information of this noise factor is reserved by the additive term in other samples, this can be seen for example in the second iteration when $j' = 0$, δ_0 is cancelled in some terms, but it sill appear in other samples, like the last one. In other words there is a redundancy and the noise factor information is conveyed throughout the iterations, unlike the the xT terms which diminish throughout the iterations:

---

k=1

$$j = 0 \begin{cases} d_{1,0} = |(x_1 - x_0)T \pm \delta_1 \mp \delta_0| \\ d_{2,0} = |(x_2 - x_0)T \pm \delta_2 \mp \delta_0| \\ d_{3,0} = |(x_3 - x_0)T \pm \delta_3 \mp \delta_0| \\ \vdots \\ d_{i,0} \\ \vdots \\ d_{n,0} = |(x_n - x_0)T \pm \delta_n \mp \delta_0| \end{cases} \qquad n$$

$$j = 1 \begin{cases} d_{2,1} = |(x_2 - x_1)T \pm \delta_2 \mp \delta_1| \\ d_{3,1} = |(x_3 - x_1)T \pm \delta_3 \mp \delta_1| \\ d_{4,1} = |(x_4 - x_1)T \pm \delta_4 \mp \delta_1| \\ \vdots \\ d_{i,1} \\ \vdots \\ d_{n,1} = |(x_n - x_1)T \pm \delta_n \mp \delta_1| \end{cases} \qquad n-1$$

$$\begin{cases} \vdots \\ d_{i,j} = |(x_i - x_j)T \pm \delta_i \mp \delta_j| & n-j \\ \vdots \end{cases}$$

$$j = n-2 \begin{cases} d_{n-1,n-2} = |(x_{n-1} - x_{n-2})T \pm \delta_{n-1} \mp \delta_{n-2}| \\ d_{n,n-2} = |(x_n - x_{n-2})T \pm \delta_n \mp \delta_{n-2}| \end{cases} 2$$

$$j = n-1 \begin{cases} d_{n,n-1} = |(x_n - x_{n-1})T \pm \delta_n \mp \delta_{n-1}| & 1 \end{cases}$$

---

k=2

$$j' = 0 \begin{cases} d_{1,0} = |(x_2 - x_1)T \pm \delta_2 \mp \delta_0 \mp \delta_1 \pm \delta_0| \\ d_{2,0} = |(x_3 - x_1)T \pm \delta_3 \mp \delta_0 \mp \delta_1 \pm \delta_0| \\ d_{3,0} = |(x_4 - x_1)T \pm \delta_4 \mp \delta_0 \mp \delta_1 \pm \delta_0| \\ \vdots & n' \\ d_{i',0} \\ \vdots \\ d_{n',0} = |(x_n - x_{n-1} - x_1 + x_0)T \pm \delta_n \mp \delta_{n-1} \mp \delta_1 \pm \delta_0| \end{cases}$$

$$j' = 1 \begin{cases} d_{2,1} = |(x_3 - x_2)T \pm \delta_3 \mp \delta_0 \mp \delta_2 \pm \delta_0| \\ d_{3,1} = |(x_4 - x_2)T \pm \delta_4 \mp \delta_0 \mp \delta_2 \pm \delta_0| \\ d_{4,1} = |(x_5 - x_2)T \pm \delta_5 \mp \delta_0 \mp \delta_2 \pm \delta_0| \\ \vdots & n'-1 \\ d_{i',1} \\ \vdots \\ d_{n',1} = |(x_n - x_{n-1} - x_2 + x_0)T \pm \delta_n \mp \delta_{n-1} \mp \delta_2 \pm \delta_0| \end{cases}$$

$$j' = n-1 \begin{cases} d_{n,n-1} = |S_n - S_{n-1}| = |(x_n - x_0 - x_2 + x_1)T \pm \delta_n \mp \delta_0 \mp \delta_2 \pm \delta_1| \\ \vdots \\ d_{n',n-1} = |S_{n'} - S_{n-1}| = |(x_n - x_0 - x_n + x_{n-1})T \pm \delta_n \mp \delta_0 \mp \delta_n \pm \delta_{n-1}| & n'-n+1 \\ \qquad = |(x_{n-1} - x_0)T \pm \delta_n \mp \delta_0 \mp \delta_n \pm \delta_{n-1}| \end{cases}$$

$$\begin{cases} \vdots \\ d_{i',j'} = |(x_{i'} - x_{j'})T \pm \delta_{i'} \mp \delta_{j'}| & n'-j' \\ \vdots \end{cases}$$

$$
\begin{array}{l}
j' = n'-2 \left\{ \begin{array}{rl}
d_{n'-1,n'-2} &= |(x_n - x_{n-2} - x_{n-1} + x_{n-2})T \pm \delta_n \mp \delta_{n-2} \mp \delta_{n-1} \pm \delta_{n-2}| \\
&= |(x_n - x_{n-1})T \pm \delta_n \mp \delta_{n-2} \mp \delta_{n-1} \pm \delta_{n-2}| \\
d_{n',n'-2} &= |(x_n - x_{n-1} - x_{n-1} + x_{n-2})T \pm \delta_n \mp \delta_{n-1} \mp \delta_{n-1} \pm \delta_{n-2}| \\
&= |(x_n - 2x_{n-1} + x_{n-2})T \pm \delta_n \mp \delta_{n-1} \mp \delta_{n-1} \pm \delta_{n-2}|
\end{array} \right. \quad 2 \\
j' = n'-1 \left\{ \begin{array}{rl}
d_{n',n'-1} &= |(x_n - x_{n-1} - x_n + x_{n-2})T \pm \delta_n \mp \delta_{n-1} \mp \delta_n \pm \delta_{n-2}| \\
&= |(x_{n-1} - x_{n-2})T \pm \delta_n \mp \delta_{n-1} \mp \delta_n \pm \delta_{n-2}|
\end{array} \right. \quad 1
\end{array}
$$

It can be shown using simple algebra that the finite term xT in our samples cancel as the number of iterations increase. We refer to this as "convergence of the algorithm". We state the proof of convergence as follows:

Proof. Let S_{max} be the initial maximum sample and $S'_{max}, S''_{max}, \ldots$ be the maximum samples of the first, second, ... iterations respectively. Similarly, S_{min} be the initial minimum sample and $S'_{min}, S''_{min}, \ldots$ be the minimum samples of the first, second, ... iterations respectively. We already know that:

$$
\begin{aligned}
S_{max} &= x_n T \pm \delta_n \\
S_{min} &= x_0 T \pm \delta_0
\end{aligned}
\tag{8.5}
$$

And using equation 8.2:

$$
\begin{aligned}
S'_{max} &= |S_{max} - S_{min}| \\
S''_{max} &= |S'_{max} - S'_{min}| \\
&\vdots
\end{aligned}
\tag{8.6}
$$

Given that $T \gg \delta$ and $0 < x_0 < x_n$, then:

$$
\begin{aligned}
x_0 &> 0 \\
-x_0 &< 0 \\
x_n - x_0 &< x_n \\
|(x_n - x_0)T| &< |x_n T|, \quad T > 0 \\
|(x_n - x_0)T \pm \delta_n \mp \delta_0| &< |x_n T \pm \delta_n|, \quad T \gg \delta
\end{aligned}
$$

$$\therefore \quad S'_{max} \quad < \quad S_{max} \tag{8.7}$$

Following a similar semantic for the next iteration, here S'_{min} is not precisely known, only its range is known:

$$\begin{aligned} S'_{max} &= |(x_n - x_0)T \pm \delta_n \mp \delta_0| \\ S'_{min} &= |(y_p - y_q)T \pm \delta_p \mp \delta_q| \end{aligned} \tag{8.8}$$

Where, $0 < x_0 \leq y_q \leq y_p \leq x_n$. Now using equation 8.2 and equation 8.6 we get:

$$\begin{aligned} |y_p - y_q| &> 0 \\ -|y_p - y_q| &< 0 \\ |x_n - x_0| - |y_p - y_q| &< |x_n - x_0| \\ (|x_n - x_0| - |y_p - y_q|)T &< (|x_n - x_0|)T, \quad T > 0 \\ |(|x_n - x_0| - |y_p - y_q|)T \underbrace{\pm\delta_n \mp \delta_0 \mp \delta_{y_p} \pm \delta_{y_q}}_{\ll T}| &< |(|x_n - x_0|)T \pm \delta_n \mp \delta_0|, \quad T \gg \delta \\ \therefore \quad S''_{max} &< S'_{max} \end{aligned} \tag{8.9}$$

And:

$$\ldots < S^{(k+3)}_{max} < S^{(k+2)}_{max} < S^{(k+1)}_{max} < S^{(k)}_{max} < S_{max} \tag{8.10}$$

□

Note, that for this proof to hold, the condition: $T \gg \sum \delta$ in each iteration is necessary.

Example (scenario 2):

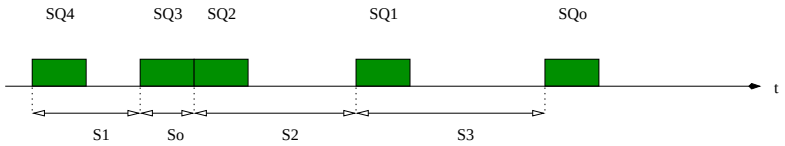


Figure 8.6: Scenario 2 example, PIT of five packets.

In this example, $\{SQ_1, SQ_2\} \neq SQ_0 + 1$, and we captured five packets, thus $np + 1 = 5, np = 4$ and $n = np - 1 = 4 - 1 = 3$. The samples to be analysed in the first iteration ($k = 1$) are:
$S_0 = T \pm \delta_0$
$S_1 = 2T \pm \delta_1$
$S_2 = 3T \pm \delta_2$
$S_3 = 4T \pm \delta_3$
Calculating the differences using equation- 8.2 and equation- 8.3:
$d_{i,j} = |S_i - S_j|$, when $k = 1, 0 \leq j \leq 2$.
$\Delta_1 = \{d_{i,j}\}_{i>j}$, when $k = 1, j < i \leq 3$.

$$
\begin{aligned}
d_{1,0} = |S_1 - S_0| = \quad & |2T \pm \delta_1 - T \mp \delta_0| = |T \pm \delta_1 \mp \delta_0| \\
d_{2,0} = |S_2 - S_0| = \quad & |3T \pm \delta_2 - T \mp \delta_0| = |2T \pm \delta_2 \mp \delta_0| \\
d_{3,0} = |S_3 - S_0| = \quad & |4T \pm \delta_3 - T \mp \delta_0| = |3T \pm \delta_3 \mp \delta_0| \\
d_{2,1} = |S_2 - S_1| = \quad & |3T \pm \delta_2 - 2T \mp \delta_1| = |T \pm \delta_2 \mp \delta_1| \\
d_{3,1} = |S_3 - S_1| = & |4T \pm \delta_3 - 2T \mp \delta_1| = |2T \pm \delta_3 \mp \delta_1| \\
d_{3,2} = |S_3 - S_2| = \quad & |4T \pm \delta_3 - 3T \mp \delta_2| = |T \pm \delta_3 \mp \delta_2|
\end{aligned}
$$

Now, the number of input samples for the next iteration ($k = 2$) is: $n' + 1 = n(n+1)/2 = 3 \times 4/2 = 6, n' = 5$.
$S'_0 = T \pm \delta_1 \mp \delta_0$
$S'_1 = T \pm \delta_2 \mp \delta_1$
$S'_2 = T \pm \delta_3 \mp \delta_2$
$S'_3 = 2T \pm \delta_2 \mp \delta_0$
$S'_4 = 2T \pm \delta_3 \mp \delta_1$
$S'_5 = 3T \pm \delta_3 \mp \delta_0$

Calculating the differences for the new iteration using equation 8.2 and equation 8.3:

$$
\begin{aligned}
d'_{1,0} &= |S'_1 - S'_0| = |T \pm \delta_2 \mp \delta_1 - T \mp \delta_1 \pm \delta_0| = \quad |\pm \delta_2 \mp \delta_1 \mp \delta_1 \pm \delta_0| \\
d'_{2,0} &= |S'_2 - S'_0| = |T \pm \delta_3 \mp \delta_2 - T \mp \delta_1 \pm \delta_0| = \quad |\pm \delta_3 \mp \delta_2 \mp \delta_1 \pm \delta_0| \\
d'_{3,0} &= |S'_3 - S'_0| = |2T \pm \delta_2 \mp \delta_0 - T \mp \delta_1 \pm \delta_0| = \quad |T \pm \delta_2 \mp \delta_0 \mp \delta_1 \pm \delta_0| \\
d'_{4,0} &= |S'_4 - S'_0| = |2T \pm \delta_3 \mp \delta_1 - T \mp \delta_1 \pm \delta_0| = \quad |T \pm \delta_3 \mp \delta_1 \mp \delta_1 \pm \delta_0| \\
d'_{5,0} &= |S'_5 - S'_0| = |3T \pm \delta_3 \mp \delta_0 - T \mp \delta_1 \pm \delta_0| = |2T \pm \delta_3 \mp \delta_0 \mp \delta_1 \pm \delta_0| \\
d'_{2,1} &= |S'_2 - S'_1| = |T \pm \delta_3 \mp \delta_2 - T \mp \delta_2 \pm \delta_1| = \quad |\pm \delta_3 \mp \delta_2 \mp \delta_2 \pm \delta_1| \\
d'_{3,1} &= |S'_3 - S'_1| = |2T \pm \delta_2 \mp \delta_0 - T \mp \delta_2 \pm \delta_1| = \quad |T \pm \delta_2 \mp \delta_0 \mp \delta_2 \pm \delta_1| \\
d'_{4,1} &= |S'_4 - S'_1| = |2T \pm \delta_3 \mp \delta_1 - T \mp \delta_2 \pm \delta_1| = \quad |T \pm \delta_3 \mp \delta_1 \mp \delta_2 \pm \delta_1| \\
d'_{5,1} &= |S'_5 - S'_1| = |3T \pm \delta_3 \mp \delta_0 - T \mp \delta_2 \pm \delta_1| = |2T \pm \delta_3 \mp \delta_0 \mp \delta_2 \pm \delta_1| \\
d'_{3,2} &= |S'_3 - S'_2| = |2T \pm \delta_2 \mp \delta_0 - T \mp \delta_3 \pm \delta_2| = \quad |T \pm \delta_2 \mp \delta_0 \mp \delta_3 \pm \delta_2| \\
d'_{4,2} &= |S'_4 - S'_2| = |2T \pm \delta_3 \mp \delta_1 - T \mp \delta_3 \pm \delta_2| = \quad |T \pm \delta_3 \mp \delta_1 \mp \delta_3 \pm \delta_2| \\
d'_{5,2} &= |S'_5 - S'_2| = |3T \pm \delta_3 \mp \delta_0 - T \mp \delta_3 \pm \delta_2| = |2T \pm \delta_3 \mp \delta_0 \mp \delta_3 \pm \delta_2| \\
d'_{4,3} &= |S'_4 - S'_3| = |2T \pm \delta_3 \mp \delta_1 - 2T \mp \delta_2 \pm \delta_0| = \quad |\pm \delta_3 \mp \delta_1 \mp \delta_2 \pm \delta_0| \\
d'_{5,3} &= |S'_5 - S'_3| = |3T \pm \delta_3 \mp \delta_0 - 2T \mp \delta_2 \pm \delta_0| = |T \pm \delta_3 \mp \delta_0 \mp \delta_2 \pm \delta_0| \\
d'_{5,4} &= |S'_5 - S'_4| = |3T \pm \delta_3 \mp \delta_0 - 2T \mp \delta_3 \pm \delta_1| = |T \pm \delta_3 \mp \delta_0 \mp \delta_3 \pm \delta_1|
\end{aligned}
$$

8.4 A Modified Multiplicative Decrease Factor

In this section we present a brief discussion of our noise quantification technique and where it fits in the TCP congestion control algorithm. We define a new TCP multiplicative decrease as follows:

$$\beta' = \mu\beta \; 0 \leq \mu \leq 1 \tag{8.11}$$

And upon a packet loss: $W \leftarrow W - \beta' W$. Let μ be function of the noise and span according to a predefined logic,we denote by ρ the residual noise, which is the number of noise factors at the end of the difference canceller last iteration. And we define the residual noise level as N_s:

$$
\begin{aligned}
N_s(\rho) &= sgm(\rho), \qquad 0 \leq N_s \leq 1, \rho \geq 0, \\
N_s(\rho) &= \tfrac{2}{1+e^{-a\rho}} - 1.
\end{aligned}
\tag{8.12}
$$

where: $sgm(.)$ is a sigmoid function and a is the residual noise level sensitivity, or noise sensitivity for short. In addition we define the sample span S_p as:

$$S_p = 1 - \left(\tfrac{S_{min}}{S_{max}}\right) \quad 0 \leq N_s \leq 1. \tag{8.13}$$

The reasons behind this choice of functions are: i) We try to avoid an abrupt decision of either 0 or 1, thus we need weighted values between 0 and 1. ii) We

need a a tunable function for the noise factor since the amount of noise that is needed to make a decision is still vague and a tunable function is desirable for experimental purposes, an attractive choice of functions is the sigmoid, where we can change the value between 0 and 1 and tune it using a. iii) On the other hand the amount of span needed to make a decision depends on the ratio of minimum to maximum values of a sample thus it is logical to use a simple linear function of the ratio, also note that taking the ratio (not the difference) makes the function's output unit less (i.e. not in seconds).

Let $\mu(N_s, S_p)$. We now find the relation between μ and N_s, S_p using the logic in table 8.1. This logic is in fact a translation of our assumptions.

Table 8.1: Relation between Noise level [N_s], Samples Span [S_p] and Output [μ]

S_p	N_s	Output [μ]	Type of Loss
$\downarrow$	$\downarrow$	$\uparrow$	congestive loss
$\downarrow$	$\uparrow$	$\uparrow$	congestive loss
$\uparrow$	$\downarrow$	$\downarrow$	wireless loss
$\uparrow$	$\uparrow$	$\uparrow$	congestive loss

$$\begin{aligned} \mu &= \neg S_p + N_s, \\ &= 1 - S_p + N_s, \end{aligned} \tag{8.14}$$

Substituting for N_s S_p using equations 8.12, 8.13: $\mu(\rho, S_{min}, S_{max}) =$

$$= \frac{S_{min}(1 + e^{-a\rho}) - S_{max}(1 + e^{-a\rho}) + 2}{S_{max}(1 + e^{-a\rho})} \tag{8.15}$$

8.5 Simulation Experiments & Results

In this section we present some of our simulation results. Nodes were arranged as in scenario 2 (please see figure 8.2b), with link capacities from left to right: 4 Mbps, 5 Mbps, 3 Mbps, 4 Mbps and 5 Mbps. Each having 10ms propagation delay. We used single TCP flow, where the source performs a bulk transfer for one minute of simulation time. A group of: 10, 50 and 100 UDP cross traffic sources enter the network route randomly and interfere with the TCP flow during the one minute. Each cross traffic source sends data at rate of 0.3 Mbps and each cross traffic link has a capacity of 1 Mbps with randomly assigned propagation delays. We used an error model at the last link to simulate the behaviour of a lossy wireless link. At the bottleneck link we used ordinary Drop-Tail queue with buffer size of 5 packets. We did not take the effect of variable packets size of cross traffic into account in our experiments and used a fixed packet size of 1000 bytes

for both TCP flow and UDP cross traffic.

Table 8.2: Loss probabilities used in Group 2 and Group 3 experiments

Experiment	Congestive loss (%)	Wireless loss (%)
1	0.465371	0.634537
2	0.826583	0.942725
3	0.911279	1.00585
4	0.79264	0.942725
5	0.80507	0.942725
6	0.379205	0.535438
7	1.05028	1.32505
8	0.541859	0.743615
9	0.827519	0.942725
10	1.04483	1.32505

The aim is to validate our assumptions and the logical formula that we have used, in order to do that, we run three groups of simulation experiments, 10 experiments for each group, Group 1: no cross traffic, no wireless loss, Group 2: cross traffic without wireless loss, Group 3: wireless loss without cross traffic. By examining the resultant values of the new multiplicative decrease factor β', noise and span, we can achieve our aim as follows: the TCP flow in Group 2 suffers only from congestive loss and thus its β' values are expected to have large values. In contrast, the TCP flow in Group 3 suffers mainly from wireless loss and thus its β' values are expected to have small values. The arguments for noise and span and their assumptions are mutatis mutandis to that of the new multiplicative decrease factor β'.

In the absence of any cross traffic and wireless loss (Group 1) we noticed that the network suffered from a packet loss probability = 0.00950841% (caused by standard TCP CC operation). Table 8.2 shows the packet loss probabilities used in the 10 experiments of Group 2 (second column) and the 10 experiments of Group 3 (third column). These probabilities were obtained as follows: we first run Group 2 experiments, we select a number of cross traffic sources, run the first experiment, calculate the packet loss probability, then run the second experiment and do the same, and so on until the tenth experiment. Second, we run Group 3 experiments, but in this group we need to set the random loss model, we used the values obtained from Group 2 experiments in this model. We set the loss probability to the value obtained from the first experiment from Group 2 and run the first experiment then calculate the loss probability, then we set the loss probability to the value obtained from the second experiment from Group 2 and run the second experiment then calculate the loss probability and so on until the tenth experiment.

Steps in bullet points are:

- Nodes were arranged as in scenario 2
- Link capacities from left to right: 4 Mbps, 5 Mbps, 3 Mbps, 4 Mbps and 5 Mbps
- Each having 10ms propagation delay
- Single TCP flow, one minute bulk transfer
- A group of: 10, 50 and 100 UDP randomly interfere with the TCP flow
- Cross traffic rates are 0.3 Mbps and each link has a capacity of 1 Mbps with randomly assigned propagation delays
- We used an error model at the last link to simulate the behaviour of a lossy wireless link
- At the bottleneck link we used drop tail queue discipline with buffer size of 5 packets
- Fixed packet size of 1000 bytes for both TCP flow and UDP cross traffic
- We did not take the effect of variable packets size and lower layers into account in our experiments
- We run three groups of simulation experiments, 10 experiments for each group
- Group 1: no cross traffic, no wireless loss
- Group 2: cross traffic without wireless loss
- Group 3: wireless loss without cross traffic

After that we run a non-implemented version of the modified TCP-Reno which is part of our TCP post-simulation analysis tool that we developed to test our work. The tool does packet drop statistics, TCP parameters calculations, runs the new technique and generates 15 graphs representing the results. We set the Difference Canceller to work for one iteration i.e. $k = 1$ and the noise sensitivity $a = 1$. Steps in bullet points:

- Non-implemented version of the modified TCP-Reno as part of our TCP post-simulation analysis tool
- We set the Difference Canceller to work for one iteration i.e. $k = 1$ and the noise sensitivity $a = 1$

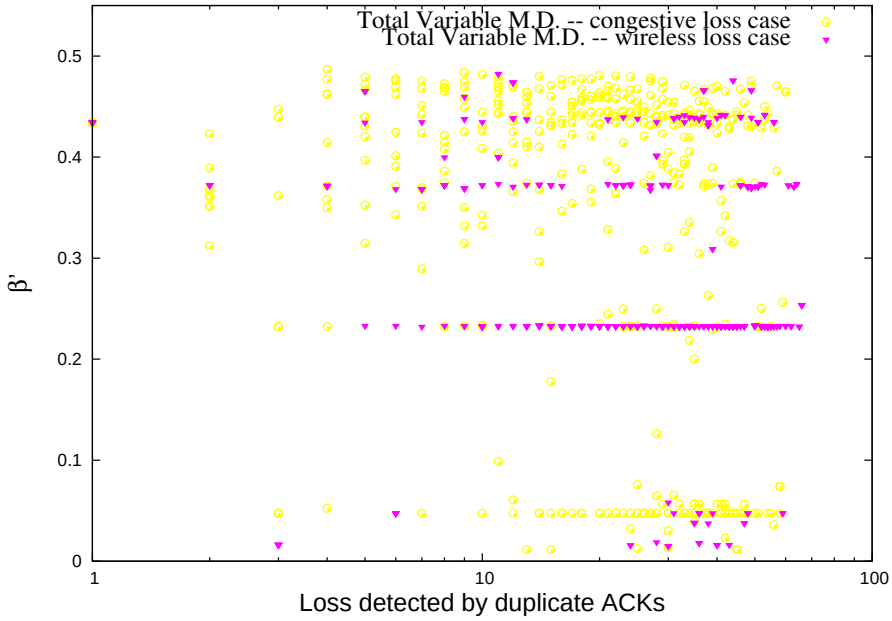


Figure 8.7: Accumulated values for 10 experiments: new multiplicative decrease factor

- The tool does packet drop statistics, TCP parameters calculations, runs the new technique
- Generates 15 graphs representing the results

8.5.1 Total Values of Variables

Figures 8.7, 8.8 and 8.9 depict the accumulated values over the 10 experiments for the new multiplicative decrease factor β', noise and span respectively for Group-2(congestive loss case) and Group-3(wireless loss case). The figures where generated as follows: we select a group, say Group-2, we run an experiment, we find the total number of congestion window cuts (only these detected by duplicate ACKs), then we ask: what are the calculated values of β', noise and span for each congestion window cut? For example suppose we have 5 congestion window cuts and the values of β' are: 0.5, 0.3, 0.1, 0.5, 0.2. We plot this as: (5,0.5),(5,0.3),(5,0.1),(5,0.5),(5,0.2). Then we run another experiment, say this time we have 15 congestion window cuts, we do the same and plot that on top of the previous plot, then we run another experiment and so on. Say we have two experiments with the same number of congestion window cuts but with different values of β', we also plot these on the same plot. When we finish 10 experiments, we move to another group, say Group-3 and we do the same and plot all on the same plot. Generating the plots in this way, makes it easier for us to figure out whether our assumptions were right or wrong, for example, when we run 10 experiments for Group-2, there are no wireless loss and all loss detected by duplicate

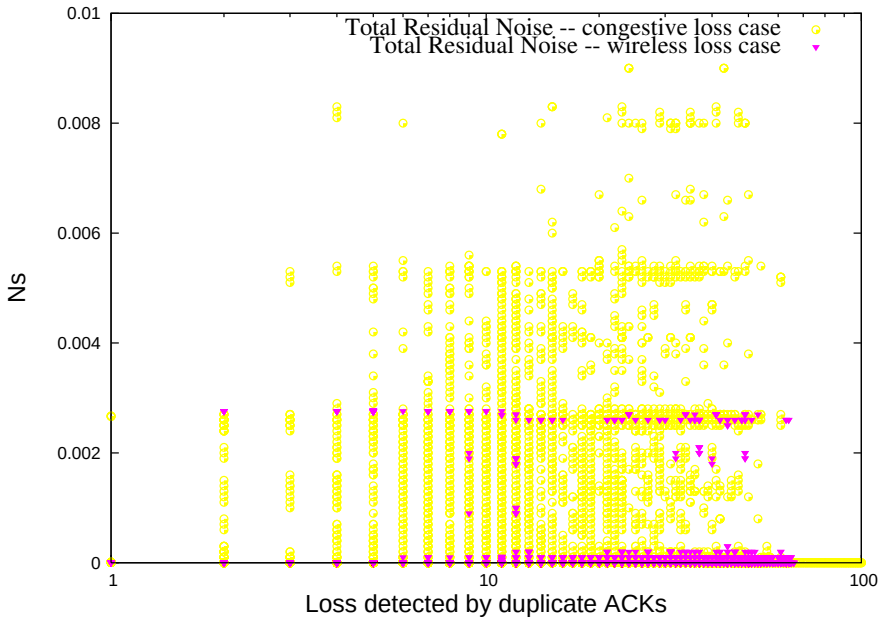


Figure 8.8: Accumulated values for 10 experiments: noise

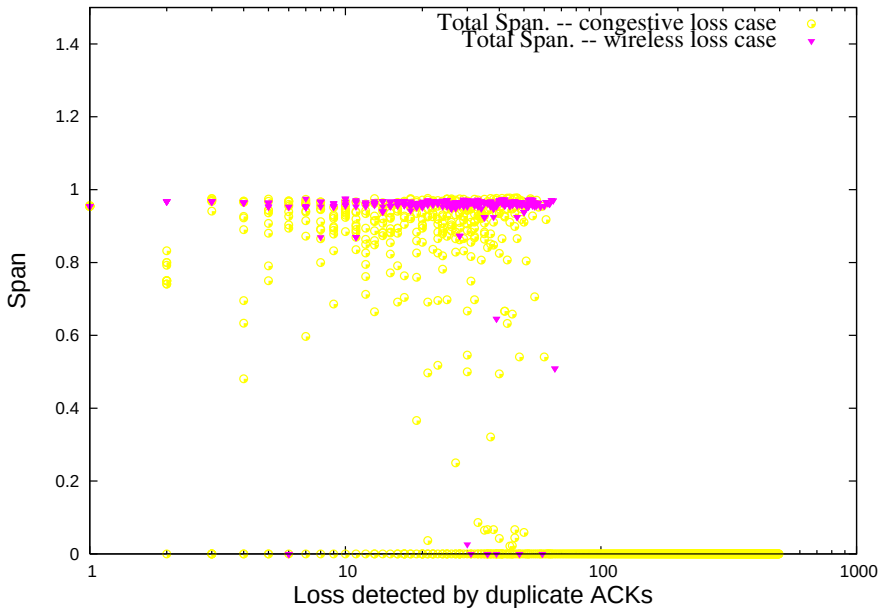


Figure 8.9: Accumulated values for 10 experiments: span

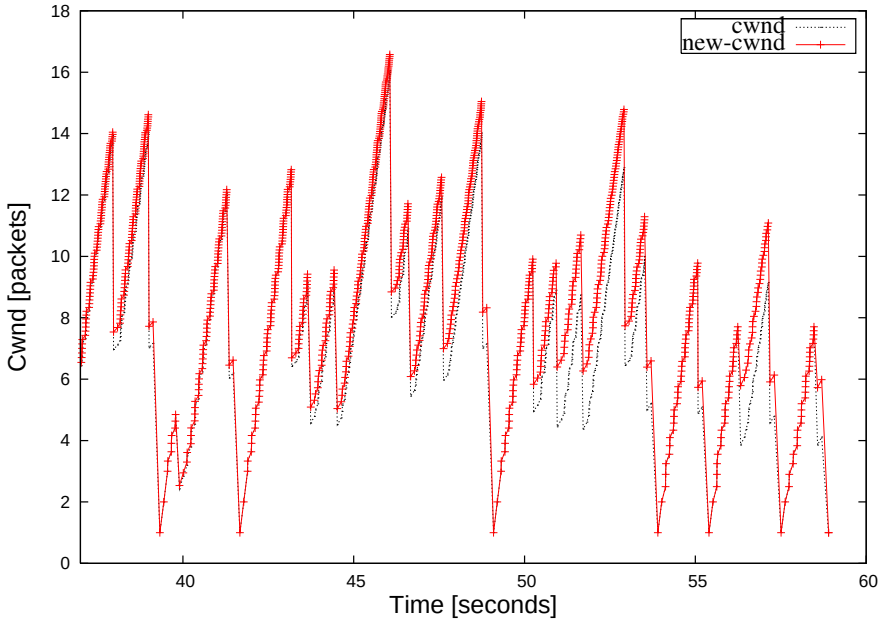


Figure 8.10: congestive loss case

ACK are due to congestion, so if our assumption was right; the figure should show high values of β' for all cuts in all experiments in this group.

As it can be seen from figure 8.7, in case of congestive loss the decrease factor tends to move to the original value of 0.5 and the operation falls back to TCP-Reno operation. However, in case of wireless loss most values are less than the original factor, with some values around 0.5. Similarly residual noise levels tend to be larger and more variable in a congestive loss case compared to a wireless loss case. On the other hand most span values take higher values in the case of wireless loss.

Figures 8.10 and 8.11 show the congestion window evolution of the modified version of TCP-Reno compared to the original TCP-Reno for experiment number one from Group 2 and Group 3 respectively (space requirements limit us to displaying results from only one experiment).

8.5.2 Congestion Window Evolution

One point to note here is the drop of the congestion window to one segment which in fact acts as a reset for both. This is due to segments time-outs, another point is the variable drop of the new version which is more significant when the wireless loss is more dominant.

Figure 8.12 and 8.13 show the effect of the noise sensitivity parameter on the evolution of congestion window. We noticed that large values of a make the algorithm more sensitive to noise and thus to interpreting the loss as congestive loss, while small values let the algorithm feel that it is working in a non-congested

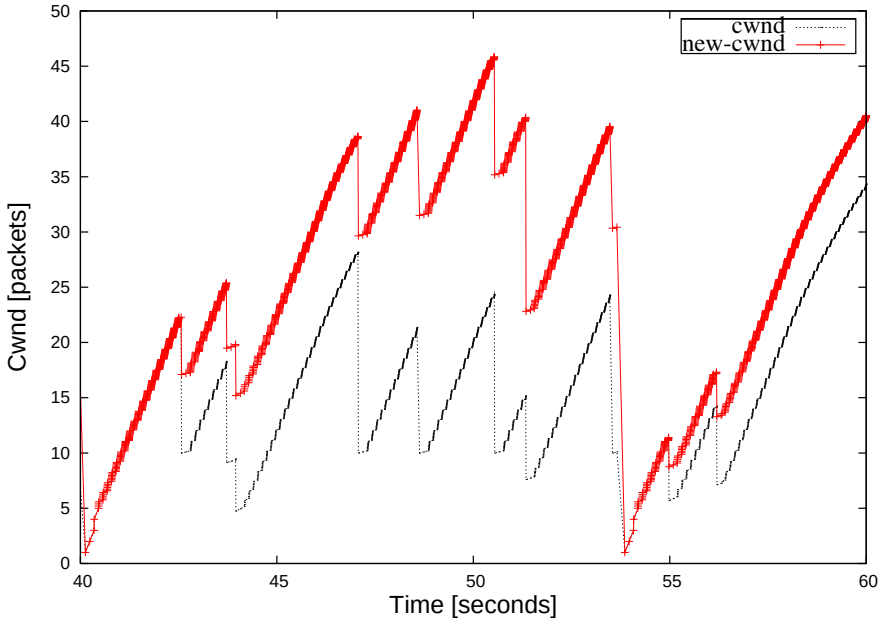


Figure 8.11: wireless loss case

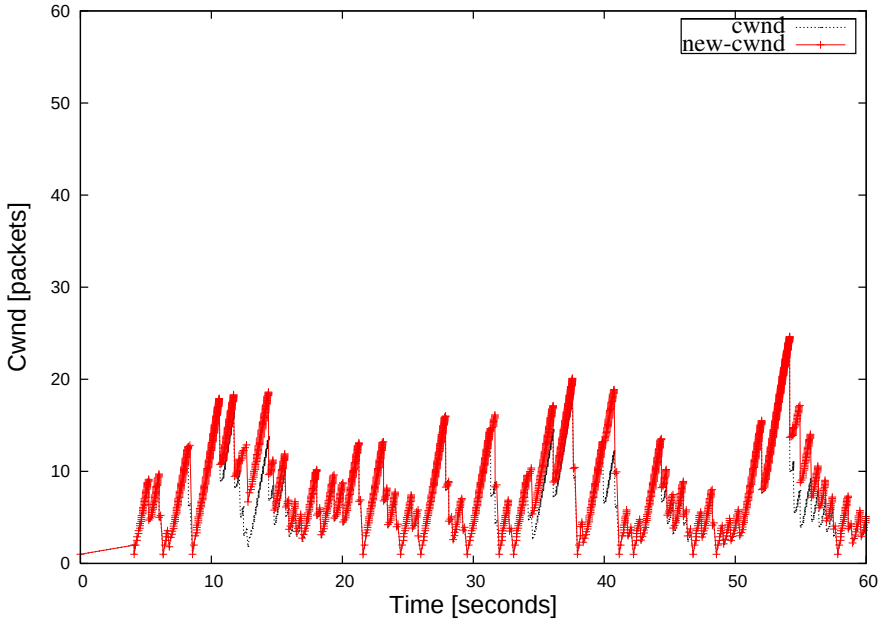


Figure 8.12: with noise sensitivity, a = 10, wireless loss case – Loss = 1.04483 %

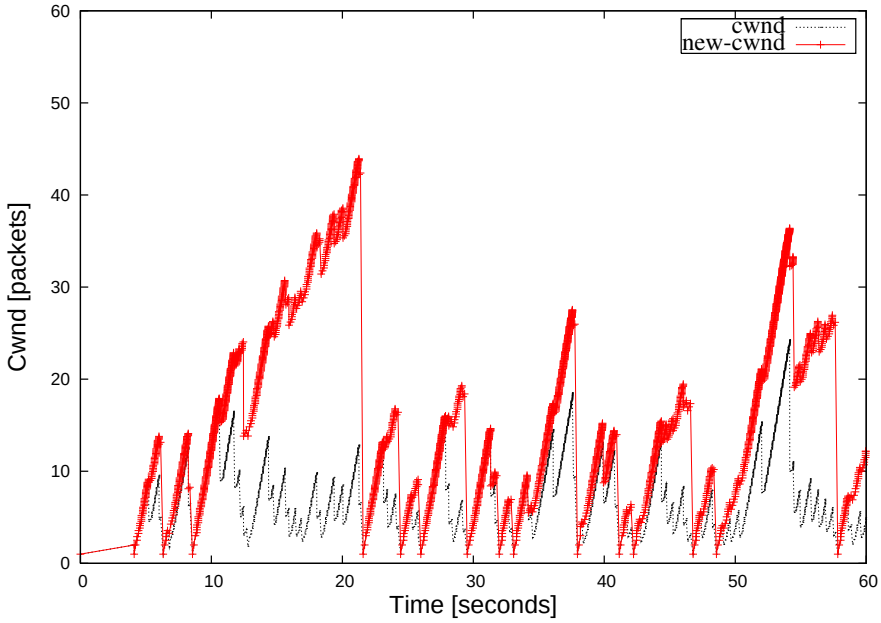


Figure 8.13: with noise sensitivity, a = 0.1, wireless loss case – Loss = 1.04483 %

lossy environment.

8.6 Summary

We presented a new technique for discriminating congestive packet drop from non-congestive packet drop based on the noise level and span in out of order PITs. The importance of this technique is that it is passive and can help TCP congestion control algorithms in the judgement of congestion window reduction when a packet drop happens. The technique can run along side other techniques working to resolve the same problem and can help in taking the decision whenever an ambiguity about the source of packet drop appears, for example; not all network routers cooperate by setting the ECN bit(s). Also reverse traffic may affect delay-based approaches. When such ambiguity appears, our technique can be consulted especially if the network setup is similar to that mentioned in this chapter.

The approach that we proposed in this chapter is orthogonal to other approaches mentioned in literature, whether those which rely on queueing delay or queue size or even network assisted techniques like ECN[5]. However; although it has intersection with some of PIT-based techniques, we believe that the approach is new and for this reason we provide a prolonged theoretical discussion of the ideas and assumptions.

We suggested six general scenarios for evaluation of the new technique and focused on one scenario where we mathematically formulated and proved the tech-

[5] Also uses average queue size.

nique. Finally we validated a non-implemented version in a controlled simulated environment and showed that this technique can augment other available techniques in helping TCP to differentiate between the two different types of packet drop.

Chapter 9

Conclusions & Future Work

Most of the Internet protocols designs were influenced by the technologies which existed at the time when the Internet was born. It was difficult at that time to predict the evolution of technologies for the next decades. Since the birth of the Internet, TCP, the most important Internet protocol that aims at providing reliable data transfer, flow and congestion control has performed well. Its self-managed congestion control algorithms preserve the stability of the Internet and the protocol has become the most used protocol for data transfer on the Internet. However, the spread of new technologies such as fibre optics and wireless networks poses many challenges to TCP in general and to TCP's congestion control algorithms in particular.

This dissertation focused on two main challenges to TCP congestion control. The first challenge is link underutilisation when working in a high-speed long-delay networks. Such networks are characterised by high bandwidth-delay product, standard TCP congestion control algorithms have a problem utilising these pipes, this is due to its fixed and small amount of congestion window increase per round trip time, which is one packet per round trip time. The research community, however; responded quickly in the last decade with a number of proposals in the form of modular congestion control algorithms. Most of these proposals are experimental and some of them have disadvantages and shortcomings. Therefore this thesis makes two contributions which we believe that the congestion control research community would benefit from: i) Improvements to one of the existing algorithms (part of Linux kernel stack), show us through simulation that our suggestions improves the responsiveness and aggressiveness shortcomings of this algorithm. ii) TCP-Gentle, an incremental development over the latest proposed algorithm (also part of Linux kernel stack) addressing this problem. Unlike all other proposals, TCP-Gentle uses different increase rules depending on network conditions. The thesis presents a throughput expression for the new algorithm based on a deterministic model and shows through simulation and real test bed

experiments that the new algorithm is gentle to networks in the sense that it keeps a small number of a TCP-Gentle flow packets in the queue compared to all other TCP congestion control algorithms, it also has good fairness properties, nearly flat rate in steady-state while at the same time maintains high responsiveness, high link utilisation and friendliness to standard TCP. Therefore it is competitive to existing algorithms.

The second challenge is the ability to differentiate between congestive loss and non-congestive loss (e.g. wireless loss), this has been a problem for standard TCP congestion control algorithms, since TCP makes a tacit assumption that each packet loss is due to congestion and reduces its congestion window drastically. The thesis proposes a novel loss differentiation algorithm which quantifies the noise in packet inter arrival times and use this information together with the span (ratio of maximum to minimum packet inter arrival times) to adapt the multiplicative decrease factor according to a predefined logical formula. We show through simulation experiments that in certain topologies the randomness in packet inter arrival times and the span can be exploited to differentiate between the two types of packet loss, to the best of our knowledge, this is a unique approach in packet loss differentiation which is orthogonal to approaches mentioned in literature. The thesis also suggestes that this algorithm can work cooperatively with other existing loss differentiation algorithms (especially in topologies were the algorithm is efficient) and resolve any ambiguities that may arise in other algorithms.

On the modelling side of TCP congestion control, the thesis extends the well-known drift model of TCP to account for wireless loss and some hypothetical cases (e.g. variable multiplicative decrease). It shows us a stability analysis for the new version of the model after linearising it around an equilibrium point. The thesis mathematically shows how the wireless loss is considered *disturbance* and it is beyond the control of standard TCP congestion control algorithms. Following that, it analytically shows that this can be controlled via a variable multiplicative decrease.

Because TCP protocol is still in use and technologies are evolving, there is a potential for other problems to arise in future. There is also an interest to improve the performance of networks (e.g. Multipath TCP). One part of our future work is to investigate the performance of TCP algorithms in different network set-ups including the Internet.

The thesis treated the end-to-end TCP congestion control approach. Despite the advantages of this approach like scalability, ease of deployment, etc, there are limitations for improvements, most work is done by end-hosts which try to infer things about the network. One such limitation is the reverse path traffic which affects ACK-based protocols that use estimates like delay and queue size based

on round trip time measurements as part of their congestion control algorithms. TCP-Gentle is subject to this problem, however our argument against this is that if the reverse traffic is persistent the algorithm is tricked and assumes a case of congestion then takes a sequence of events which eventually leads to a fall back to standard TCP congestion control algorithm operation, i.e. the worst case is to work like the standard TCP congestion control algorithms (increase until a packet loss happens then decrease).

Having said that, another part of our future work will focus more on network assisted approaches hoping to investigate the limitations of totally depending on end-hosts, interesting examples are: XCP protocol, lower layer QCN approach. Performance evaluation and analysis of these are a of research interest.

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Appendix A

History Time Line for TCP-Related Issues

History Notes: TCP/IP and Transport Layer Protocols
Compiled by: Talal A.Edwan.
Date: 04-March-2008. Update date: 23-July-2010.

TCP/IP Overview and History[1]

The best place to start looking at TCP/IP is probably the name itself. TCP/IP in fact consists of dozens of different protocols, but only a few are the main protocols that define the core operation of the suite. Of these key protocols, two are usually considered the most important. The Internet Protocol (IP) is the primary OSI network layer (layer three) protocol that provides addressing, datagram routing and other functions in an internetwork. The Transmission Control Protocol (TCP) is the primary transport layer (layer four) protocol, and is responsible for connection establishment and management and reliable data transport between software processes on devices.

Due to the importance of these two protocols, their abbreviations have come to represent the entire suite: TCP/IP. (In a moment we'll discover exactly the history of that name.) IP and TCP are important because many of TCP/IP's most critical functions are implemented at layers three and four. However, there is much more to TCP/IP than just TCP and IP. The protocol suite as a whole requires the work of many different protocols and technologies to make a functional

[1] This text is quoted from {source:http://www.tcpipguide.com/free/t_TCPIPOverviewandHistory.htm}

network that can properly provide users with the applications they need.

TCP/IP uses its own four-layer architecture that corresponds roughly to the OSI Reference Model and provides a framework for the various protocols that comprise the suite. It also includes numerous high-level applications, some of which are well-known by Internet users who may not realise they are part of TCP/IP, such as HTTP (which runs the World Wide Web) and FTP. In the topics on TCP/IP architecture and protocols I provide an overview of most of the important TCP/IP protocols and how they fit together. Early TCP/IP History

As I said earlier, the Internet is a primary reason why TCP/IP is what it is today. In fact, the Internet and TCP/IP are so closely related in their history that it is difficult to discuss one without also talking about the other. They were developed together, with TCP/IP providing the mechanism for implementing the Internet. TCP/IP has over the years continued to evolve to meet the needs of the Internet and also smaller, private networks that use the technology. I will provide a brief summary of the history of TCP/IP here; of course, whole books have been written on TCP/IP and Internet history, and this is a technical Guide and not a history book, so remember that this is just a quick look for sake of interest.

The TCP/IP protocols were initially developed as part of the research network developed by the United States Defence Advanced Research Projects Agency (DARPA or ARPA). Initially, this fledgling network, called the ARPAnet, was designed to use a number of protocols that had been adapted from existing technologies. However, they all had flaws or limitations, either in concept or in practical matters such as capacity, when used on the ARPAnet. The developers of the new network recognised that trying to use these existing protocols might eventually lead to problems as the ARPAnet scaled to a larger size and was adapted for newer uses and applications.

In 1973, development of a full-fledged system of inter networking protocols for the ARPAnet began. What many people don't realise is that in early versions of this technology, there was only one core protocol: TCP. And in fact, these letters didn't even stand for what they do today; they were for the Transmission Control Program. The first version of this predecessor of modern TCP was written in 1973, then revised and formally documented in RFC 675, Specification of Internet Transmission Control Program, December 1974.

Modern TCP/IP Development and the Creation of TCP/IP Architecture:

Testing and development of TCP continued for several years. In March 1977,

version 2 of TCP was documented. In August 1977, a significant turning point came in TCP/IPs development. Jon Postel, one of the most important pioneers of the Internet and TCP/IP, published a set of comments on the state of TCP. In that document (known as Internet Engineering Note number 2, or IEN 2), he provided what I consider superb evidence that reference models and layers aren't just for textbooks, and really are important to understand:

"*We are screwing up in our design of internet protocols by violating the principle of layering. Specifically we are trying to use TCP to do two things: serve as a host level end to end protocol, and to serve as an internet packaging and routing protocol. These two things should be provided in a layered and modular way. I suggest that a new distinct internetwork protocol is needed, and that TCP be used strictly as a host level end to end protocol.*"

– Jon Postel, IEN 2, 1977

What Postel was essentially saying was that the version of TCP created in the mid-1970s was trying to do too much. Specifically, it was encompassing both layer three and layer four activities (in terms of OSI Reference Model layer numbers). His vision was prophetic, because we now know that having TCP handle all of these activities would have indeed led to problems down the road.

Postel's observation led to the creation of TCP/IP architecture, and the splitting of TCP into TCP at the transport layer and IP at the network layer; thus the name TCP/IP. (As an aside, it's interesting, given this history, that sometimes the entire TCP/IP suite is called just IP, even though TCP came first.) The process of dividing TCP into two portions began in version 3 of TCP, written in 1978. The first formal standard for the versions of IP and TCP used in modern networks (version 4) were created in 1980. This is why the first real version of IP is version 4 and not version 1. TCP/IP quickly became the standard protocol set for running the ARPAnet. In the 1980s, more and more machines and networks were connected to the evolving ARPAnet using TCP/IP protocols, and the TCP/IP Internet was born.

Key Concept: TCP/IP was initially developed in the 1970s as part of an effort to define a set of technologies to operate the fledgling Internet. The name TCP/IP came about when the original Transmission Control Program (TCP) was split into the Transmission Control Protocol (TCP) and Internet Protocol (IP). The first modern versions of these two key protocols were documented in 1980 as TCP version 4 and IP version 4.

Important Factors in the Success of TCP/IP:

TCP/IP was at one time just one of many different sets of protocols that could be used to provide network-layer and transport-layer functionality. Today there are still other options for inter networking protocol suites, but TCP/IP is the universally-accepted world-wide standard. Its growth in popularity has been due to a number of important factors. Some of these are historical, such as the fact that it is tied to the Internet as described above, while others are related to the characteristics of the protocol suite itself. Chief among these are the following:

- Integrated Addressing System: TCP/IP includes within it (as part of the Internet Protocol, primarily) a system for identifying and addressing devices on both small and large networks. The addressing system is designed to allow devices to be addressed regardless of the lower-level details of how each constituent network is constructed. Over time, the mechanisms for addressing in TCP/IP have improved, to meet the needs of growing networks, especially the Internet. The addressing system also includes a centralised administration capability for the Internet, to ensure that each device has a unique address.

- Design For Routing: Unlike some network-layer protocols, TCP/IP is specifically designed to facilitate the routing of information over a network of arbitrary complexity. In fact, TCP/IP is conceptually concerned more with the connection of networks, than with the connection of devices. TCP/IP routers enable data to be delivered between devices on different networks by moving it one step at a time from one network to the next. A number of support protocols are also included in TCP/IP to allow routers to exchange critical information and manage the efficient flow of information from one network to another.

- Underlying Network Independence: TCP/IP operates primarily at layers three and above, and includes provisions to allow it to function on almost any lower-layer technology, including LANs, wireless LANs and WANs of various sorts. This flexibility means that one can mix and match a variety of different underlying networks and connect them all using TCP/IP.

- Scalability: One of the most amazing characteristics of TCP/IP is how scalable its protocols have proven to be. Over the decades it has proven its mettle as the Internet has grown from a small network with just a few machines to a huge internetwork with millions of hosts. While some changes have been required periodically to support this growth, these changes have taken place as part of the TCP/IP development process, and the core of TCP/IP is basically the same as it was 25 years ago.

- Open Standards and Development Process: The TCP/IP standards are not proprietary, but open standards freely available to the public. Furthermore, the

process used to develop TCP/IP standards is also completely open. TCP/IP standards and protocols are developed and modified using the unique, democratic RFC process, with all interested parties invited to participate. This ensures that anyone with an interest in the TCP/IP protocols is given a chance to provide input into their development, and also ensures the world-wide acceptance of the protocol suite.

- Universality: Everyone uses TCP/IP because everyone uses it!

This last point is, perhaps ironically, arguably the most important. Not only is TCP/IP the underlying language of the Internet, it is also used in most private networks today. Even former competitors to TCP/IP such as NetWare now use TCP/IP to carry traffic. The Internet continues to grow, and so do the capabilities and functions of TCP/IP. Preparation for the future continues, with the move to the new IP version 6 protocol in its early stages. It is likely that TCP/IP will remain a big part of inter networking for the foreseeable future.

Key Concept: While TCP/IP is not the only inter networking protocol suite, it is definitely the most important one. Its unparalleled success is due to a wide variety of factors. These include its technical features, such as its routing-friendly design and scalability, its historical role as the protocol suite of the Internet, and its open standards and development process, which reduce barriers to acceptance of TCP/IP protocols.

For Brief History of Wireless Technologies see: Wireless Data Technologies. Vern A. Dubendorf 2003 John Wiley & Sons, Ltd ISBN: 0-470-84949-5. Notice that in 802.11(b) are popular in 2000.

Note: the aim of the following history time line is to:

- See the effect of various telecommunication technologies (e.g. wireless, long fat pipes,...) on the evolution of transport layer reliable protocols. (trend of research).
- Show the need for reliable protocols (the topic is still evolving).
- Help in tracking the changes and learning from previous algorithms.
- Provide short reading list for certain topics. (organised research or pointers for research papers).

1957: USSR launches Sputnik, first artificial earth satellite. The start of global telecommunications. Satellites play an important role in transmitting all sorts of data today. In response, US forms the Advanced Research Projects Agency (ARPA) within the Department of Defence (DOD) to establish US lead in science and technology applicable to the military. {source: Gromov's Timeline, inthistory file}

1967: ARPANET design discussions held by Larry Roberts at ARPA IPTO PI meeting in Ann Arbor, Michigan (April)

ACM Symposium on Operating Systems Principles in Gatlinburg, Tennessee (October).

- First design paper on ARPANET published by Larry Roberts: "Multiple Computer Networks and Inter computer Communication.
- First meeting of the three independent packet network teams (RAND, NPL, ARPA).

National Physical Laboratory (NPL) in Middlesex, England develops NPL Data Network under Donald Watts Davies who coins the term packet. The NPL network, an experiment in packet-switching, used 768kbps lines.

1969: Birth of Internet. ARPANET commissioned by DOD (US Dept. of Defence) for research into networking. First node at UCLA (Los Angeles) closely followed by nodes at Stanford Research Institute, UCSB (Santa Barbara) and U of Utah (4 Nodes) {source: Gromov's Timeline, inthistory file}

1970: ALOHANET developed at the University of Hawaii. {source: Gromov's Timeline, inthistory file}

1971: People communicate over a network. 15 nodes (23 hosts) on ARPANET. E-mail invented.
{source: Gromov's Timeline, inthistory file}

1970-1973: The ARPANET was a success from the very beginning.

1973: Need of development of fully fledged system for ARPAnet.

1974: TCP v.1, Vint Cerf and Bob Kahn publish "A Protocol for Packet Network Interconnection" which specified in detail the design of a Transmission Control Program (TCP). {IEEE Trans Comm} BBN opens Telenet, the first public packet data service (a commercial version of ARPANET).
Formally documented in RFC 675 December 1974.

1976: Elizabeth II, Queen of the United Kingdom sends out an email on 26 March from the Royal Signals and Radar Establishment (RSRE) in Malvern.
UUCP (Unix-to-Unix CoPy) developed at AT&T Bell Labs and distributed with UNIX one year later.

1977: March, TCP v.2 was documented.
August, TCP v.2 should be split into two protocols. According to Jon Postel, IEN 2

1978: TCP v.3, The process of dividing TCP into two portions began in version 3 of TCP and it was written.

1980: User Datagram Protocol (UDP) created by David P. Reed.

1980: TCP v.4, first formal standard for the versions of IP and TCP used in modern networks.
January, RFC 760 761 outline new specifications for the two protocols IP and TCP. {source: www.cs.utexas.edu}
February, RFC TCP/IP becomes the preferred military protocol. {source: www.cs.utexas.edu}
TCP/IP Internet was born.

1981: September, IETF RFC 793 (TCP), RFC 791 (IP). RFC-791 replaced RFC-760.

1981: Jerome H. Saltzer, David P. Reed, and David D. Clark, s.l. End-to-End Arguments in System Design. IEEE Computer Society, 1981, Proceedings of the 2nd International Conference on Distributed Computing Systems, Paris, 1981, pp. 509-512.

Jerome H. Saltzer, David P. Reed, and David D. Clark. End-to-end arguments in system design. ACM Transactions on Computer Systems 2, 4 (November 1984) pages 277-288. An earlier version appeared in the Second International Conference on Distributed Computing Systems (April, 1981) pages 509-512.
{source: wikipedia, End-to-end principle}

1982: The term 'Internet' was used for the first time. {source: Gromov's Timeline, inthistory file}

1983: 4.2BSD, Many of the implementations of TCP/IP at the time (1973-1980) were pulled together to create the first widely available (and used) version of TCP/IP. {source: node8 file}.

1984: Domain Name System (DNS) introduced.

-Number of hosts breaks 1,000.

-JUNET (Japan Unix Network) established using UUCP.
-JANET (Joint Academic Network) established in the UK using the Coloured Book protocols; previously SERCnet.

1986: In October of '86, the Internet had the first of what became a series of 'congestion collapses'. During this period, the data throughput from LBL (Lawrence Berkeley Laboratories) to UC Berkeley (sites separated by 400 yards and two IMP (Infrastructure Message Processor, i.e router) hops) dropped from 32 Kbps to 40bps. This sudden factor-of-thousand drop in bandwidth embarked on an investigation of why things had gotten so bad. In particular, we wondered if the 4.3bsd (Berkeley UNIX) TCP was mis-behaving or if it could be tuned to work better under abysmal network conditions. The answer to both of these questions was "yes".

1987: Karn's Algorithm: P. Karn and C. Partridge. Improving Round-Trip Time Estimates in Reliable Transport Protocols. Computer Communication Review, 17(5), August 1987.

1988: 4.3BSD TAHOE TCP

The Tahoe implementation of TCP (1988) introduced significant improvements for working over a shared network (leading in 1989 to the Net/1 release). An algorithm (Slow Start (congestion control) and multiplicative decrease (congestion avoidance)) was introduce to control transmission following any detected congestion. Under this algorithm, a TCP transmitter is allowed to transmit a number of bytes determined by the smallest value of the window advertised by the receiver and a congestion window (cwnd). The cwnd is initially assigned a value of 1 segment, and is doubled following receipt of each ACK, normally resulting in exponential window growth.

The algorithm also uses a variable to keep the threshold value of the send window (ssthresh), which is initialised to the receiver's advertised window size. Following a retransmission time out, the algorithm assigns half of the current window to ssthresh (this is the multiplicative decrease part of the algorithm), the cwnd is set to one packet and the slow start phase begins. The cwnd is increased by one segment whenever an acknowledgement is received, until it reaches the ssthresh. At this point, the algorithm switches to the congestion avoidance phase and the window size is only increased by a fraction of the segment (equivalent to an increment of one segment per round trip delay).

The Fast Retransmission {RFC 2001} algorithm was also implemented into avoid waiting for the retransmission timer to expire following every packet loss. In this

algorithm, a receiver sends a duplicate ACK immediately on reception of each out of sequence packet. The transmitter interprets reception of 3 duplicate ACKs (sufficient to avoid spurious retransmissions due to reordering of segments) as a congestion packet loss and triggers the Slow Start algorithm.

1989: NET/1 TCP

1989: XTP

Xpress Transport Protocol (XTP) is a transport layer protocol for high-speed networks promoted by the XTP Forum developed to replace TCP. XTP provides protocol options for error control, flow control, and rate control. Instead of separate protocols for each type of communication, XTP controls packet exchange patterns to produce different models, e.g. reliable datagrams, transactions, unreliable streams, and reliable multicast connections. XTP does not employ congestion avoidance algorithms. XTP is a real-time option at Layer 4 for the US Navy SAFENET LAN Profile.

"XTP Protocol Definition Revision 3.4", Protocol Engines, Incorporated, 1900 State Street, Suite D, Santa Barbara, California 93101, 1989.

{source:http://en.wikipedia.org/wiki/Xpress_Transport_Protocol + http://www.cs.virginia.edu/papers/p67-sanders.pdf}

1990: 4.4BSD RENO TCP

Reno TCP (1990, followed by Net/2 in 1991) introduced a further optimisation (Fast Recovery) to improve performance following retransmission. When the third duplicate ACK is received, the Reno TCP transmitter sets ssthresh to one half of the current congestion window (cwnd) and retransmits the missing segment. The cwnd is then set to ssthresh plus three segments (one segment per each duplicate ACK that has already received). cwnd is then increased by one segment on reception of each duplicate ACK which continues to arrive after fast-retransmission. This allows the transmitter to send new data when cwnd is increased beyond the value of the cwnd before the fast-retransmission. When an ACK arrives which acknowledges all outstanding data sent before the duplicate ACKs were received, the cwnd is set to ssthresh so that the transmitter slows down the transmission rate and enters the linear increase phase.

Much had also been learned through research into alternative protocols to TCP (e.g. XTP, NetBLT). This experience was transferred to TCP through the addition of "header prediction", providing a very significant performance improvement.

1991: Tri-S, a congestion control algorithm.

1992: T/TCP

TCP for Transactions,

RFC 1379: Braden, R. T., Extending TCP for Transactions-Concepts, 38 pages, Nov. 1992.
RFC 1644: Braden, R. T., T/TCP-TCP Extensions for Transactions, Functional Specification, 38 pages, July 1994.

{source: http://www.kohala.com/start/ttcp.html}.

1993: 4.4BSD NET/3 TCP

The 4.4 BSD release (1993) lead to Net/3 TCP, which continues to be one of the reference implementations used by developers. This added capability for multicast, long fat network extensions and various other refinements.

1993: Floyd, S., and Jacobson, V., Random Early Detection gateways for Congestion Avoidance V.1 N.4, August 1993, p. 397-413. This is the basic paper that describes RED queue management.

1994: VEGAS TCP

An experimental modification to TCP was proposed by Lawrence Brakmo at University of Arizona. This primarily added rate control to avoid congestion (rather than react after detection of congestion) it emphasises packet delay, rather than packet loss, as a signal to help determine the rate at which to send packets. Vegas has not been widely implemented and is not universally accepted by the Internet community and is still a subject of much controversy.
{source: wikipedia + see SIGCOMM94 paper}.

1994: T-TCP

T/TCP (Transactional TCP) is a variant of the TCP protocol. It is an experimental TCP extension for efficient transaction-oriented (request/response) service. It was developed to fill the gap between TCP and UDP, by Bob Braden in 1994. Its definition can be found in RFC 1644.

This protocol is faster than TCP and delivery reliability is comparable to that of TCP. Unfortunately, T/TCP suffers from a major spoofing problem pointed out by Vasim Valejev in 1998 in a posting to Bugtraq, and has not gained widespread popularity.

{source: http://en.wikipedia.org/wiki/Category:Transport_layer_protocols}

1995: I-TCP

I-TCP: indirect TCP for mobile hosts

Abstract: IP based solutions to accommodate mobile hosts within existing internetworks do not address the distinctive features of wireless mobile computing. IP-based transport protocols thus suffer from poor performance when a mobile host communicates with a host on the fixed network. This is caused by frequent disruptions in network layer connectivity due to - i) mobility and ii) unreliable nature of the wireless link. We describe I-TCP, which is an indirect transport layer protocol for mobile hosts. I-TCP utilizes the resources of Mobility Support Routers (MSRs) to provide transport layer communication between mobile hosts and hosts on the fixed network. With I-TCP, the problems related to mobility and unreliability of wireless link are handled entirely within the wireless link; the TCP/IP software on the fixed hosts is not modified. Using I-TCP on our testbed, the throughput between a fixed host and a mobile host improved substantially in comparison to regular TCP.
{source: http://citeseer.ist.psu.edu/bakre95itcp.html}

1995-1996: NEW-RENO TCP

Janey Hoe (of MIT) proposed a modification to Reno TCP usually called New-Reno, which addressed two problems in TCP, these ideas are gradually finding acceptance within the IETF. Hoe noted that a smaller value for ssthresh causes premature termination of the slow start phase and subsequent slow increase of the cwnd (i.e. the linear increase phase). A larger value causes the sender to over-feed packets to the network (i.e. transmit too long a burst of data packets) causing congestion. Since most TCP sessions last only for a short period of time, the initial slow start period is significant for the over all performance.

Hoe proposed a method to estimate an optimum ssthresh value by calculating the byte equivalent of bandwidth delay product of the network, when a new connection is made. The bandwidth is calculated using the Packet-Pair algorithm (measuring the arrival time of closely spaced ACKs at the sender).

If two or more segments have been lost from the transmitted data (window), the Fast Retransmission and Fast Recovery algorithms will not be able to recover the losses without waiting for retransmission time out. New-Reno overcomes this problem by introducing the concept of a Fast Retransmission Phase, which starts on detection of a packet loss (receiving 3 duplicate ACKs) and ends when the receiver acknowledges reception of all data transmitted at the start of the Fast

Retransmission phase. If more than one packet is missing within the same window, a retransmission only recovers the first lost packet from the window. The receiver then ACKs reception of the retransmitted segment and all following segments up to the next lost segment. This ACK is called a "partial ACK", because it has not ACKed all the packets which were transmitted prior to the start of the current Fast Retransmission Phase.

The transmitter assumes reception of a partial ACK during the Fast Retransmission phase as an indication that another packet has been lost within the window and retransmits that packet immediately to prevent expiry of the retransmission timer. New Reno sets the cwnd to one segment on reception of 3 duplicate ACKs (i.e. when entering the Fast Retransmission Phase) and unacknowledged data are retransmitted using the Slow Start algorithm. The transmitter is also allowed to transmit a new data packet on receiving 2 duplicate ACKs. While the transmitter is in the Fast Retransmission Phase, it continues to retransmit packets using Slow Start until all packets have been recovered (without starting a new retransmission phase for partial ACKs). Although this modification may cause unnecessary retransmissions, it reduces transmitter time outs and efficiently recovers multiple packet loss using partial ACKs.
{source: tcp-evol file + RFC 2582}

1995-1996: Snoop TCP

The Snoop protocol is a TCP-aware link layer protocol designed to improve the performance of TCP over networks of wired and single-hop wireless links. (Note: The TCP split approach was before this work - mid 1990s-, see the background part of [10], section 2.5.3 Split connection protocols).
{source: http://nms.lcs.mit.edu/ hari/papers/snoop.html}

1996: RTP

The Real-time Transport Protocol (or RTP) defines a standardised packet format for delivering audio and video over the Internet. It was developed by the Audio-Video Transport Working Group of the IETF and first published in 1996 as RFC 1889 which was made obsolete in 2003 by RFC 3550. {source: wikipedia}

Note: Call setup and tear-down for VoIP applications is usually performed by either SIP (Session Initiation Protocol an application-layer control) or H.323 (an ITU Telecommunication Standardisation Sector (ITU-T) standard) protocols, SIP is designed to be independent of the underlying transport layer; it can run on TCP, UDP, or SCTP.

1996-1998: SELECTIVE ACKNOWLEDGEMENT OPTION (SACK)

The probability of multiple packet loss in a window is much greater for a long fast network, where more packets are in flight. Although TCP is able to recovering multiple packet losses without waiting for expiry of the retransmission timer, frequent packet loss may still not be efficiently recovered. The SACK extension (1996-98) improves TCP performance over such a network, and has been included in some recent TCP implementations.

The SACK option is triggered when the receiver buffer holds in-sequence data segments following a packet loss. The receiver then sends duplicate ACKs bearing the SACK option to inform the transmitter which segments have been correctly received. When the third duplicate ACKis received, a SACK TCP transmitter retransmits only the missing packets starting with the sequence number acknowledged by the duplicate ACKs, and followed by any subsequent unacknowledged segments. The Fast Retransmission and Recovery algorithms are also modified to avoid retransmitting already SACKed segments. The explicit information carried by SACKs enables the transmitter to also accurately estimate the number of transmitted data packets that have left the network (this procedure is known as Forward Acknowledgement (FACK)), allowing transmission of new data to continue during retransmission. The SACK option is able to sustain high throughput over a network subject to high packet loss and is therefore desirable for bulk transfers over a DVB network. Optimisation of the algorithms which govern use of the SACK information are still the subject of research, however the basic algorithms are now widely implemented.

N Samaweera & G Fairhurst have proposed a refinement to SACK to solve the reliance of SACK on protocol timers when a retransmission is unsuccessful. {source: tcp-evol file}

1997: RSVP

The Resource ReSerVation Protocol (RSVP), version 1 is described in RFC 2205, is a Transport layer protocol designed to reserve resources across a network for an integrated services Internet. "RSVP does not transport application data but is rather an Internet control protocol, like ICMP, IGMP, or routing protocols" - RFC 2205.
{source wikipedia + http://www.isi.edu/div7/rsvp/rsvp.html}

1997: M-TCP

http://web.cecs.pdx.edu/ singh/mtcp.html
http://www.cs.pdx.edu/ singh/ftp/mtcp.ps.gz
Kevin Brown and Suresh Singh, M-TCP: TCP for Mobile Cellular Networks, ACM

CCR Vol. 27(5), 1997

1998: IPv6 RFC 2460.

1999: WTCP

WTCP (Wireless Transmission Control Protocol) is a proxy based modification of TCP that preserves the end-to-end semantics of TCP. As its name suggests, it is used in wireless networks to improve the performance of TCP.
{source: http://en.wikipedia.org/wiki/WTCP +
http://timely.crhc.uiuc.edu/Projects/wtcp/wtcp.html}

2000: SCTP

Stream Control Transmission Protocol standard draft document (RFC2960) in October 2000 (This RFC Updated by RFC 3309 and obsoleted by RFC 4960).
{source: wikipedia + http://tdrwww.exp-math.uni-essen.de/inhalt/
forschung/sctp_fb/sctp_intro.html}

2000: RFC 2914 Congestion Control Principles.

2000: TCP Westwood

TCP Westwood: Handling Dynamic Large Leaky Pipes

TCP Westwood (TCPW), is a sender-side-only modification to TCP NewReno that is intended to better handle large bandwidth-delay product paths (large pipes), with potential packet loss due to transmission or other errors (leaky pipes), and with dynamic load (dynamic pipes).

TCPW relies on mining the ACK stream for information to help it better set the congestion control parameters: Slow Start Threshold (ssthresh), and Congestion Window (cwin). In TCPW, an "Eligible Rate" is estimated and used by the sender to update ssthresh and cwin upon loss indication, or during its "Agile Probing" phase, a proposed modification to the well-known Slow Start phase. In addition, a scheme called Persistent Non Congestion Detection (PNCD) has been devised to detect persistent lack of congestion and induce an Agile Probing phase to expeditiously utilise large dynamic bandwidth.

The resultant performance gains in efficiency, without undue sacrifice of fairness, friendliness, and stability have been reported in numerous papers that can be found on this web site. Significant efficiency gains can be obtained for large leaky dynamic pipes, while maintaining fairness. Under a more appropriate criterion for friendliness, i.e. "opportunistic friendliness", TCPW is shown to have good, and

controllable, friendliness.
{source: Westwood Homepage, http://www.cs.ucla.edu/NRL/hpi/tcpw/ TCP }

2001: TCP Westwood+[2]

TCP Westwood+ is a sender-side only modification of the TCP Reno/NewReno classic congestion control protocol stack that optimises the performance of TCP congestion control especially over wireless networks. TCPW is based on end-to-end bandwidth estimation to set congestion window and slow start threshold after a congestion episode, that is, after three duplicate acknowledgements or a timeout. The bandwidth is estimated by properly low-pass filtering the rate of returning acknowledgement packets. The rationale of this strategy is simple: in contrast with TCP Reno, which blindly halves the congestion window after three duplicate ACKs, TCP Westwood+ adaptively sets a slow start threshold and a congestion window which takes into account the bandwidth used at the time congestion is experienced. TCP Westwood significantly increases fairness with respect to TCP (New) Reno in wired networks and throughput over wireless links.
{source: Westwood Homepage, http://www.cs.ucla.edu/NRL/hpi/tcpw/ TCP }

2001: ATCP

ATCP: TCP for mobile ad hoc networks Liu, J.; Singh, S. Selected Areas in Communications, IEEE Journal on Volume 19, Issue 7, Jul 2001 Page(s):1300 - 1315 Digital Object Identifier 10.1109/49.932698 Summary:Transport connections set up in wireless ad hoc networks are plagued by problems such as high bit error rates, frequent route changes, and partitions. If we run the transmission control protocol (TCP) over such connections, the throughput of the connection is observed to be extremely poor because TCP treats lost or delayed acknowledgments as congestion. We present an approach where we implement a thin layer between Internet protocol and standard TCP that corrects these problems and maintains high end-to-end TCP throughput. We have implemented our protocol in FreeBSD, and we present results from extensive experimentation done in an ad hoc network. We show that our solution improves the TCP's throughput by a factor of 2-3

2001: TCP HACK

R.K.Balan, B.P.Lee and K.R.R.Kumar " TCP HACK: TCP Header checksum option to Improve performance over Lossy Links ", in proc. INFOCOM '2001, Anchorage, AK, pp.309-318.

[2]Also begining of Veno

2001: TCP-Real

TCP-real: improving real-time capabilities of TCP over heterogeneous networks.

Source:International Workshop on Network and Operating System Support for Digital Audio and Video archive Proceedings of the 11th international workshop on Network and operating systems support for digital audio and video table of contents Port Jefferson, New York, United States Pages: 189 - 198 Year of Publication: 2001 ISBN:1-58113-370-7

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Abstract
We present a TCP-compatible and -friendly protocol which abolishes three major shortfalls of TCP for reliable multimedia applications over heterogeneous networks: (i) ineffective bandwidth utilisation, (ii) unnecessary congestion-oriented responses to wireless link errors (e.g., fading channels) and operations (e.g. handoffs), and (iii) wasteful window adjustments over asymmetric, low-bandwidth reverse paths. We propose TCP-Real, a high-throughput transport protocol that minimises transmission-rate gaps, thereby enabling better performance and reasonable playback timers. In TCP-Real, the receiver decides with better accuracy about the appropriate size of the congestion window. Slow Start and timeout adjustments are used whenever congestion avoidance fails; however, rate and timeout adjustments are cancelled whenever the receiving rate indicates sufficient availability of bandwidth. We detail the protocol design and we report significant improvement on the performance of the protocol with time-constrained traffic, wireless link errors and asymmetric paths.
{source: http://portal.acm.org/citation.cfm?id=378371}

2001-2002: HighSpeed TCP (HSTCP) is a new congestion control algorithm protocol defined in RFC 3649 for TCP (Sally Floyd).

{source: wikipedia + http://www.icir.org/floyd/hstcp.html}

2002: XCP The eXplicit Control Protocol

SIGCOMM02: http://www.acm.org/sigcomm/sigcomm2002/papers/xcp.pdf

{source: http://www.isi.edu/isi-xcp/}

2002: TCP Nice

TCP Nice, Arun Venkataramani, Ravi Kokku, and Mike Dahlin, 2002.

"makes good use of the spare bandwidth without affecting the traffic already present."
{source: http://www.cs.utexas.edu/users/arun/nice/ and link (2) at the end }

2003: TCP Veno

TCP Veno is a novel end-to-end congestion control scheme which can improve TCP performance quite significantly over heterogeneous networks, particularly when wireless links form part of such networks. The key innovation in Veno is the enhancement of Reno congestion control algorithm by using the estimated state of a connection based on Vegas. This scheme significantly reduces "blind" reduction of TCP window regardless of the cause of packet loss. The salient feature of TCP Veno is that it only needs simple modification at sender side of Reno protocol stack. Considering practical issues deployability and compatibility (conformance with legacy connections), Veno TCP may be quickly deployed in "hot" Mobile Internet industry.
{source: www.ntu.edu.sg/home/ascpfu/veno/index.html}

2003: TCP Hybla

TCP Hybla is a experimental TCP enhancement conceived with the primary aim of counteracting the performance deterioration caused by the long RTTs typical of satellite connections. It consists of a set of procedures which includes, among others:

- an enhancement of the standard congestion control algorithm (to grant long RTT connections the same instantaneous segment transmission rate of a comparatively fast reference connection).
- the mandatory adoption of the SACK policy.
- the use of timestamps.
- the adoption of Hoes channel bandwidth estimate.
- the implementation of packet spacing techniques (also known as "pacing").

TCP Hybla involves only sender-side modification of TCP. As that, it is fully compatible with standard receivers.
{source: TCP Hybla Homepage, http://hybla.deis.unibo.it/}

2003: TCP-LP TCP Low Priority

Service prioritisation among different traffic classes is an important goal for the future Internet. Conventional approaches to solving this problem consider the existing best-effort class as the low-priority class, and attempt to develop mechanisms that provide "better-than-best-effort" service. We explore the opposite approach, and devise a new distributed algorithm to realise a low-priority service

(as compared to the existing best effort) from the network endpoints. To this end, we develop TCP Low Priority (TCP-LP), a distributed algorithm whose goal is to utilise only the excess network bandwidth as compared to the "fair share" of bandwidth as targeted by TCP. The key mechanisms unique to TCP-LP congestion control are the use of one-way packet delays for congestion indications and a TCP-transparent congestion avoidance policy. On the other hand, HSTCP-LP (High-Speed TCP Low Priority) is an advanced TCP version targeted towards fast long-distance networks (i.e., networks operating at 622 Mb/s, 2.5 Gb/s, or 10 Gb/s and spanning several countries or states). HSTCP-LP is a TCP-LP version with aggressive window increase policy. More details on HSTCP-LP could be found below. {source: http://www.ece.rice.edu/networks/TCP-LP/}

2003: FAST TCP

FAST TCP is a new TCP congestion avoidance algorithm especially targeted at high-speed, long-distance links, developed at the Netlab, California Institute of Technology and now being commercialised by Fastsoft. It is compatible with existing TCP algorithms, requiring modification only to the computer which is sending data.
{source: wikipedia + http://netlab.caltech.edu/FAST/}

2003: Scalable TCP improved performance in highspeed networks

Scalable TCP is a simple change to the traditional TCP congestion control algorithm (RFC2581) which dramatically improves TCP performance in highspeed wide area networks. This page provides information on various aspects of Scalable TCP. {source: http://www.deneholme.net/tom/scalable/}

2003: TFRC

TFRC is a TCP-Friendly, rate-based congestion control protocol, which intends to compete fairly for bandwidth with TCP flows.
RFC 3448: TCP Friendly Rate Control (TFRC): Protocol Specification.
Handley, M., Floyd, S., Pahdye, J., and Widmer, J.
RFC 3448, Proposed Standard, January 2003.
{source: http://www.icir.org/tfrc/}

2003: TCP-DCR

Delayed Congestion Response TCP protocol (TCP-DCR).
TCP-DCR delays responding to a packet loss indication by a small period of time (one RTT) to allow channel errors to be recovered by link level retransmission... An interested by-product of using TCP-DCR is the inherent robustness it provides against... degradation due to packet re-ordering in the network.

{source: http://dropzone.tamu.edu/techpubs/2003/TAMU-ECE-2003-01.pdf and link (2) at the end }

2003: TCP-PR

TCP-PR: TCP for Persistent Packet Reordering
The key feature of TCP-PR is that duplicate ACKs are not used as an indication of packet loss. Rather, TCP-PR relies exclusively on timeout.{source: http://www-rcf.usc.edu/ junsool/tcp-pr/tcp-pr.html and link (2) at the end }

2004: H-TCP

H-TCP is another implementation of TCP with an optimised congestion control algorithm for high speed networks with high latency (LFN: Long Fat Networks). It was created by researchers at the Hamilton Institute in Ireland.
H-TCP is an optional module in recent Linux 2.6 kernels.
{source: wikipedia + http://www.hamilton.ie/net/htcp/}

2004: TCP-Jersey

TCP-Jersey is a new TCP scheme that focuses on the capability of the transport mechanism to distinguish the wireless from congestion packet losses.
TCP-Jersey for wireless IP communications Kai Xu Ye Tian Ansari, N. Dept. of Electr. & Comput. Eng., New Jersey Inst. of Technol., Newark, NJ, USA;

This paper appears in: Selected Areas in Communications, IEEE Journal on Publication Date: May 2004 Volume: 22, Issue: 4 On page(s): 747- 756 ISSN: 0733-8716
INSPEC Accession Number: 7956371 Digital Object Identifier: 10.1109/JSAC.2004.825989
Posted online: 2004-05-04 13:48:11.0

Abstract
Improving the performance of the transmission control protocol (TCP) in wireless Internet protocol (IP) communications has been an active research area. The performance degradation of TCP in wireless and wired-wireless hybrid networks is mainly due to its lack of the ability to differentiate the packet losses caused by network congestions from the losses caused by wireless link errors. In this paper, we propose a new TCP scheme, called TCP-Jersey, which is capable of distinguishing the wireless packet losses from the congestion packet losses, and reacting accordingly. TCP-Jersey consists of two key components, the available bandwidth estimation (ABE) algorithm and the congestion warning (CW) router configuration. ABE is a TCP sender side addition that continuously estimates the bandwidth available to the connection and guides the sender to adjust its transmission rate when the network becomes congested. CW is a configuration of network routers such that routers alert end stations by marking all packets when there is

a sign of an incipient congestion. The marking of packets by the CW configured routers helps the sender of the TCP connection to effectively differentiate packet losses caused by network congestion from those caused by wireless link errors. This paper describes the design of TCP-Jersey, and presents results from experiments using the NS-2 network simulator. Results from simulations show that in a congestion free network with 1% of random wireless packet loss rate, TCP-Jersey achieves 17% and 85% improvements in goodput over TCP-Westwood and TCP-Reno, respectively; in a congested network where TCP flow competes with VoIP flows, with 1% of random wireless packet loss rate, TCP-Jersey achieves 9% and 76% improvements in goodput over TCP-Westwood and TCP-Reno, respectively. Our experiments of multiple TCP flows show that TCP-Jersey maintains the fair and friendly behavior with respect to other TCP flows.

2004: UDP Lite RFC 3828 , July 2004.

UDP Lite is a connectionless protocol, very similar to UDP. Unlike UDP, where either all or none of a packet is protected by a checksum, UDP Lite allows for partial checksums that only cover part of a datagram, and will therefore deliver packets that have been partially corrupted. It is particularly useful for multimedia protocols, such as voice over IP, in which receiving a packet with a partly damaged payload is better than receiving no packet at all.
{source: http://en.wikipedia.org/wiki/UDP_Lite}

2004: A TCP congestion control algorithm for wireless Internet connections Yayche, H. Pierre, S. Quintero, A.Ecole Polytech. de Montreal, Que., Canada;

This paper appears in: Electrical and Computer Engineering, 2004. Canadian Conference on Publication Date: 2-5 May 2004 Volume: 3, On page(s): 1797- 1800 Vol.3 ISSN: 0840-7789 ISBN: 0-7803-8253-6 INSPEC Accession Number: 8088655 Posted online: 2004-11-01 11:50:34.0

Abstract This letter proposes a new congestion control algorithm for TCP in a wireless Internet. TCP performs poorly in such an environment because of the high frame error rate and the assumption that packet loss is always a sign of congestion. Our TCP congestion control algorithm is distributed and is based on a numerical algorithm that solves this optimization problem using a gradient-based method. Simulation results show that, according to the proposed algorithm, TCP converges to the proportional fair allocation if some precautions are taken to limit the delays experienced by packets and the corresponding acknowledgements.

2004: BIC (Binary Increase Congestion control) TCP
2005: and CUBIC (cubic function) TCP

The demands for fast transfer of large volumes of data, and the deployment of the network infrastructures to support the demand are ever increasing. However, the dominant network transport protocol of today, TCP, does not meet this demand. The slow response of TCP in fast long distance networks leaves sizeable unused bandwidth in such networks. BIC TCP and CUBIC are congestion control protocols designed to remedy this problem. Our goal is to design a protocol that can scale its performance up to several tens of gigabits per second over high-speed long distance networks while maintaining strong fairness, stability and TCP friendliness.
{source: http://netsrv.csc.ncsu.edu/twiki/bin/view/Main/BIC}

BIC is the abbreviation for Binary Increase Congestion control. BIC uses a unique window growth function. In case of packet loss, the window is reduced by a multiplicative factor. The window size just before and after the reduction is then used as parameters for a binary search for the new window size. BIC was used as standard algorithm in the Linux kernel. CUBIC is a less aggressive variant of BIC (meaning, it doesn't steal as much throughput from competing TCP flows as does BIC).

BIC TCP is implemented and used by default in Linux kernels 2.6.8 and above. The default implementation was again changed to CUBIC TCP in the 2.6.19 version.
{source: wikipedia}

2005: Compound-TCP

Compound TCP (CTCP) is a Microsoft algorithm that is part of the Windows Vista and Window Server 2008 TCP stack. It is designed to aggressively adjust the sender's congestion window to optimise TCP for connections with large bandwidth-delay products while trying not to harm fairness (as can occur with HSTCP).
{source: wikipedia + http://research.microsoft.com/research/pubs/view.aspx?type=Technical%20Report&id=940}

2005: TCP Africa

R. King, R. Baraniuk, R. Riedi, TCP Africa: An Adaptive and Fair Rapid Increase Rule for Scalable TCP, in proc. of IEEE INFOCOM 2005, Miami, USA, Mar.

2005: TCP-Casablanca

"De-Randomizing" congestion losses to improve TCP performance over wired-wireless networks Biaz, S.; Vaidya, N.H. Networking, IEEE/ACM Transactions on Volume 13, Issue 3, June 2005 Page(s): 596 - 608 Digital Object Identifier 10.1109/TNET.2005.850205 Summary: Currently, a TCP sender considers all losses as congestion signals and reacts to them by throttling its sending rate. With Internet becoming more heterogeneous with more and more wireless error-prone links, a TCP connection may unduly throttle its sending rate and experience poor performance over paths experiencing random losses unrelated to congestion. The problem of distinguishing congestion losses from random losses is particularly hard when congestion is light: congestion losses themselves appear to be random. The key idea is to "de-randomize" congestion losses. This paper proposes a simple biased queue management scheme that "de-randomizes" congestion losses and enables a TCP receiver to diagnose accurately the cause of a loss and inform the TCP sender to react appropriately. Bounds on the accuracy of distinguishing wireless losses and congestion losses are analytically established and validated through simulations. Congestion losses are identified with an accuracy higher than 95% while wireless losses are identified with an accuracy higher than 75%. A closed form is derived for the achievable improvement by TCP endowed with a discriminator with a given accuracy. Simulations confirm this closed form. TCP-Casablanca, a TCP-Newreno endowed with the proposed discriminator at the receiver, yields through simulations an improvement of more than 100% on paths with low levels of congestion and about 1% random wireless packet loss rates. TCP-Ifrane, a sender-based TCP-Casablanca yields encouraging performance improvement.

2006: DCCP

Created on: 2006/04/03 RFC4340 March 2006. The Datagram Congestion Control Protocol (DCCP) is a transport protocol that provides bidirectional unicast connections of congestion-controlled unreliable datagrams. DCCP is suitable for applications that transfer fairly large amounts of data and that can benefit from control over the tradeoff between timeliness and reliability.
{source: http://www.read.cs.ucla.edu/dccp/}

2006: TCP-Illinois

TCP-Illinois is a congestion control algorithm for high speed networks. It inherits the good features of TCP and modifies the features of TCP which are not suitable for high speed networks. It satisfies all the requirements for a high speed variants and has great performance. TCP-Illinois is invented by Shao Liu, under the direction of his co-advisors, Professor Tamer Basar and Professor R. Srikant. This

work is done in University of Illinois, and this new protocol is named after this university.
{source: http://www.ews.uiuc.edu/%7Eshaoliu/tcpillinois/}

2007: YeAH-TCP

YeAH-TCP is a sender-side high-speed enabled TCP congestion control algorithm, which uses a mixed loss/delay approach to compute the congestion window. It's design goals target high efficiency, internal, RTT and Reno fairness, resilience to link loss while keeping network elements load as low as possible.

For further details look here:
http://wil.cs.caltech.edu/pfldnet2007/paper/YeAH_TCP.pdf
This is the YeAH-TCP implementation of the algorithm presented to PFLDnet2007 (http://wil.cs.caltech.edu/pfldnet2007/).
{source: Internet}

Freeze-TCP, TCP-Door, JTCP, TCP ADA, TCP Peach Peach+, TCP Probing.

2010: TCP-Gentle

TCP-Gentle is variant of YeAH-TCP developed by Talal A.Edwan at Loughborough University. The algorithm uses different increase rules depending on network conditions. It uses an adaptive additive increase rule instead of multiplicative increase. It is belived to be more gentle to the network and has better fairness properties than YeAH-TCP, it has a nearly flat rate in steady-state while at the same time maintains high responsiveness, high link utilisation and friendliness to TCP-NewReno.

For further details, see the paper or contact: <t.edwan@lboro.ac.uk>.

Note: For a Taxonomy of congestion control see:
{source: http://en.wikipedia.org/wiki/Taxonomy_of_congestion_control}

More Information:
1-http://www.icir.org/floyd/
2-http://www.icir.org/floyd/tcp_small.html

Appendix B

High-Speed TCP Equations

B.1 My Derivation of TCP-BIC Equations

In this I show the work out of derivation of the equations mentioned in [102], mainly a closed form for the response function and RTT faireness. This is a bit pedantic, but I would like to document it for the benifit for people interested in the subject.

Binary search increase

BI-TCP switches from additive increase to binary search increase when the distance from the current window size to the target window size is less than S_{max}.

During binary search increase, the algorithm treats the problem as a search problem, the target window size is initially computed as the midway between a large window (maximum window) and the current window (minimum window at which the flow does not see any packet loss). Upon a packet loss event, the target window is computed as the half way between the window size before loss and that after loss (after multiplicative decrease), if the target is reached with no loss, the current window becomes the new minimum and a new target is calculated and so on, until the difference between the maximum and the minimum falls below a preset threshold S_{min}. Figure B.2 shows a hypothetical window growth for binary search increase. $A = \beta W_{max} - N_1 S_{max}$, where, W_{max} is the maximum window, β is the multiplicative decrease factor, S_{max} is the maximum increment and N_1 is the number of RTT in additive increase period. The number of RTT in binary search period N_2 can be found from S_{min} expression:

$$S_{min} = \frac{1}{2^{N_2}}(\beta W_{max} - N_1 S_{max})$$

Thus,

$$N_2 = \log_2 \left(\frac{W_{max}\beta - N_1 S_{max}}{S_{min}} \right) + 2$$

And the total number of packets Y_2 (the area under the curve Figure B.2) is:

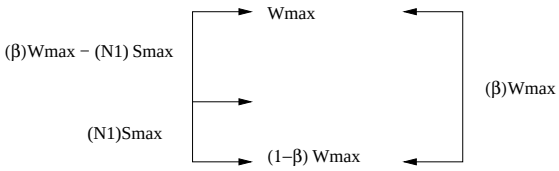


Figure B.1

$$\begin{aligned} Y_2 &= Total\ Area\ -\ Top\ Area \\ &= W_{max}N_2 - (\beta W_{max} - N_1 S_{max}) \sum_{k=0}^{N_2-2} (\tfrac{1}{2})^k \end{aligned}$$

Substituting for $N_2 - 2$:

$$\begin{aligned} Top\ Area &= (\beta W_{max} - N_1 S_{max}) \left(\tfrac{1-(1/2)^{N_2-1}}{1-(1/2)} \right) \\ &= (\beta W_{max} - N_1 S_{max}) \left(\tfrac{2(W_{max}\beta - N_1 S_{max}) - S_{min}}{W_{max}\beta - N_1 S_{max}} \right) \end{aligned}$$

$$Y_2 = W_{max}N_2 - 2(W_{max} - N_1 S_{max}) + S_{min}$$

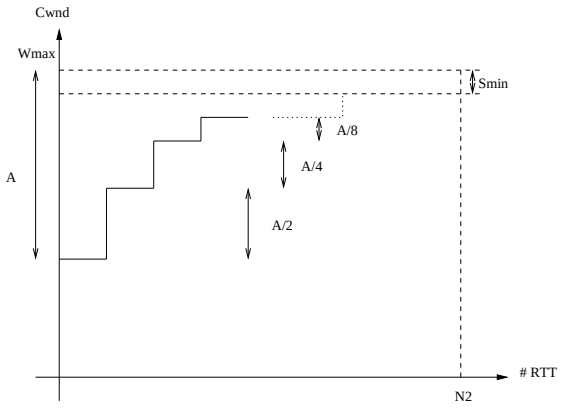


Figure B.2: Congestion window growth in binary search increase

Additive Increase:

In order to maintain faster converngence time and RTT fairness the BI-TCP algorithm uses additive increase. Additive increase works if the distance from the current window to the target is larger than S_{max}, which implies that the distance from the minimum window to the maximum window is larger than $2S_{max}$. The number of RTT N_1 can be found from: $N_1 Smax + 2S_{max} - \beta W_{max} = 0$, then:

$$N_1 = \max(\left\lceil \frac{\beta W_{max}}{S_{max}} - 2 \right\rceil, 0)$$

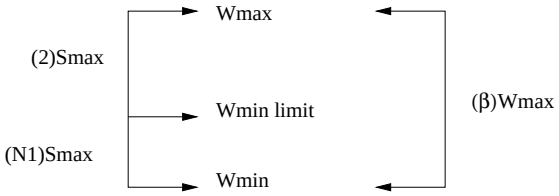


Figure B.3

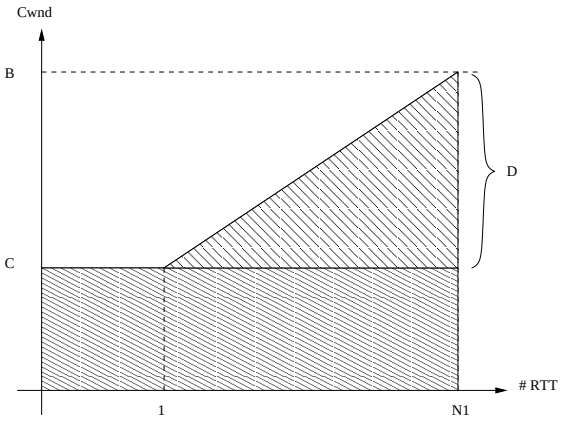


Figure B.4: Congestion window growth in additive increase

Now N_2 can be rewritten as:

$$N_2 = \log_2 \left(\frac{W_{max}\beta - W_{max}\beta + 2S_{max}}{S_{min}} \right) + 2 = \log_2(\frac{S_{max}}{S_{min}}) + 3$$

Similarly, Figure B.4 shows a hypothetical window growth for the additive increase period. $B = W_{max}(1-\beta) + (N_1 - 1)S_{max}$, $C = (1-\beta)W_{max}$, $D = (N_1 - 1)S_{max}$, N_1 is the number of RTT during the additive increase period. The area under the curve is:

$$Y_1 = \frac{1}{2}(N_1 - 1)S_{max}N_1 + W_{max}(1-\beta)N_1$$

Average sending rate for Binary Increase:

The total number of packets can be expressed as: $Y = Y_1 + Y_2$ and $Y = 1/p$,

$$\begin{aligned} Y &= W_{max}(1-\beta)(\tfrac{W_{max}\beta}{S_{max}} - 2) + \tfrac{1}{2}S_{max}(\tfrac{W_{max}\beta}{S_{max}} - 3)(\tfrac{W_{max}\beta}{S_{max}} - 2) + \\ &\quad W_{max}\log_2(S_{max}/S_{min}) + 3W_{max} - 2W_{max}\beta + 2S_{max}(\tfrac{W_{max}\beta}{S_{max}} - 2) + S_{min} \\ &= \tfrac{\beta(1-\beta)}{S_{max}}W_{max}^2 - 2(1-\beta)W_{max} + \tfrac{\beta^2}{2S_{max}}W_{max}^2 - \beta W_{max} - \tfrac{3}{2}W_{max}\beta + 3S_{max} + \\ &\quad W_{max}\log_2(S_{max}/S_{min}) + 3W_{max} - 2W_{max}\beta + 2W_{max}\beta - 4S_{max} + S_{min} \\ &= \left(\tfrac{\beta(1-\beta)}{S_{max}} + \tfrac{\beta^2}{2S_{max}}\right)W_{max}^2 + \left(1 - \tfrac{\beta}{2} + \log_2(S_{max}/S_{min})\right)W_{max} + S_{min} - S_{max} \end{aligned}$$

$$\begin{aligned} Y &= aW_{max}^2 + bW_{max} - C \\ 0 &= aW_{max}^2 + bW_{max} - (C+Y) \\ W_{max} &= \frac{-b+\sqrt{b^2+4a(1/p)}}{2a} \end{aligned}$$

Where, $a = \beta(2-\beta)/(2S_{max}),\ b = \log_2(S_{max}/S_{min}) + (2-\beta)/2,\ c = S_{max} - S_{min}$.

The average sending rate is expressed as, $X = Y/N$:

$$N = N_1 + N_2 = \frac{W_{max}\beta}{S_{max}} - 2 + \log_2(S_{max}/S_{min}) + 3 = \frac{W_{max}\beta}{S_{max}} + \log_2(S_{max}/S_{min}) + 1$$

$$\begin{aligned} X = Y/N &= \frac{1/p}{\frac{W_{max}\beta}{S_{max}} + \log_2(S_{max}/S_{min}) + 1} \\ &= \frac{(2-\beta)/p}{W_{max}\frac{\beta(2-\beta)}{S_{max}} + (2-\beta)\log_2(S_{max}/S_{min}) + (2-\beta)} \\ &= \frac{(2-\beta)/p}{-b+\sqrt{b^2+4a(1/p)} + 2\log_2(S_{max}/S_{min}) + (2-\beta) - \beta\log_2(S_{max}/S_{min})} \\ &= \frac{(2-\beta)/p}{-b+\sqrt{b^2+4a(1/p)} + 2(\log_2(S_{max}/S_{min}) + \frac{(2-\beta)}{2}) - \beta\log_2(S_{max}/S_{min})} \\ &= \frac{(2-\beta)/p}{-b+\sqrt{b^2+4a(1/p)} + 2b - \beta\log_2(S_{max}/S_{min})} \\ &= \frac{(2-\beta)/p}{\sqrt{b^2+4a(1/p)} + b - \beta(b - \frac{2-\beta}{2})} \\ &= \frac{(2-\beta)/p}{\sqrt{b^2+4a(1/p)} + b - \beta b + \beta\left(\frac{2-\beta}{2}\right)} \\ &= \frac{(2-\beta)/p}{\sqrt{b^2+4a(1/p)} + (1-\beta)b + \beta\left(\frac{2-\beta}{2}\right)} \end{aligned}$$

Average rate for Additive Increase:

$W_{max} \gg 2S_{max}$,$N_1 \gg N_2$, fixed S_{min} and small values of p

$$\begin{aligned} X &\approx \frac{(2-\beta)/p}{\sqrt{\left(\frac{2-\beta}{2}\right)^2 + \frac{4\beta(2-\beta)}{2S_{max}}(1/p)}} \\ &= \frac{(2-\beta)/p}{\sqrt{\left(\frac{2-\beta}{2}\right)^2 + \frac{2\beta(2-\beta)}{S_{max}}(1/p)}} \\ &= \sqrt{\frac{(2-\beta)^2/p}{\left(\frac{2-\beta}{2}\right)^2 + \frac{2\beta(2-\beta)}{S_{max}}}} \\ &= \sqrt{\frac{(2-\beta)/p}{\frac{2-\beta}{2} + \frac{2\beta}{S_{max}}}} \\ &= \sqrt{\frac{S_{max}(2-\beta)}{2\beta p}} \end{aligned}$$

Note that S_{min} is not appearing in the final result.

Average rate for Binary Search Increase:

$W_{max} \leq 2S_{max}$,$N_1 = 0$ and $(1/p) \gg S_{min}$

In this case:

$$\begin{aligned} N_2 &= \log_2\left(\tfrac{W_{max}\beta}{S_{min}}\right)+2 \\ Y_1 &= 0 \\ Y_2 &= W_{max}N_2 - 2(W_{max}\beta) + S_{min} \\ Y = Y_1 + Y_2 &= W_{max}N_2 - 2(W_{max}\beta) + S_{min} \end{aligned}$$

$$\begin{aligned} Y &= W_{max}\log_2(W_{max}\beta/S_{min}) - 2W_{max} - 2W_{max}\beta - S_{min} \\ &= W_{max}\left[\log_2(W_{max}\beta/S_{min}) + 2(1-\beta)\right] + S_{min} \\ W_{max} &= \tfrac{(1/p)-S_{min}}{\log_2(W_{max}\beta/S_{min})+2(1-\beta)} \\ &\approx \tfrac{1/p}{\log_2(W_{max}\beta/S_{min})+2(1-\beta)} \\ \log_2(W_{max}\beta/S_{min}) &= (1/pW_{max}) - 2(1-\beta) \\ \ln(W_{max}\beta/S_{min}) &= (\ln(2)/pW_{max}) - 2a\ln(2)(1-\beta) \\ \tfrac{e^{\ln(2)/pW_{max}}}{W_{max}} &= \tfrac{\beta e^{2\ln(2)}.e^{-2\ln(2)\beta}}{S_{min}} \end{aligned}$$

$$\frac{\ln(2)e^{\ln(2)/pW_{max}}}{pW_{max}} = \frac{\ln(2)\beta e^{2\ln(2)}.e^{-2\ln(2)\beta}}{pS_{min}}$$

And, using:

$$\begin{aligned} y = e^x &\rightarrow x = \ln y \\ y = xe^x &\rightarrow x = LambertW(y) \end{aligned}$$

$$\frac{\ln(2)}{pW_{max}} = LambertW\left(\frac{4\ln(2)\beta}{pS_{min}}e^{-2\ln(2)\beta}\right)$$

$$W_{max} = \frac{\ln(2)}{p.LambertW\left(\frac{4\ln(2)\beta}{pS_{min}}e^{-2\ln(2)\beta}\right)}$$

$$X = Y/N = \frac{1/p}{\log_2\left(\frac{W_{max}\beta}{S_{min}}\right)+2} \approx W_{max}\left(1 - \frac{2\beta}{\log_2\left(\frac{W_{max}\beta}{S_{min}}\right)+2}\right)$$

Note that S_{max} is not appearing the final result, and for $2\beta \ll \log_2(W_{max}\beta/S_{min})+2$, $R \approx W_{max}$.

RTT fairness of Binary Increase[1]:

Let RTT_i be the RTT of flow $i(i = 1, 2)$ and W_i denote W_{max} of flow i, n_i denote the number of RTT in a congetsion epoch of flow i, t denote the length of an epoch during steady state. We already expect that the additive increase will exhibit a linear RTT unfairness, in fact this is one of the reasons for including additive increase in the algorithm. However the situation for binary search increase is different. As can be seen below:

Additive Increase: For $W_{max}\beta > 2S_{max}$ we have: $N_1 = (W_{max}\beta/S_{max}) - 2$,$N_2 = \log_2(S_{max}/S_{min}) + 3$,$N = N_1 + N_2$. Now we will find the window ratio:

$$n = N = \frac{W_{max}}{S_{max}} + \log_2(S_{max}/S_{min}) + 1$$

$$n = t/RTT,\ \frac{t}{RTT} = \frac{W_{max}\beta}{S_{max}} + \log_2(S_{max}/S_{min}) + 1$$

$$\begin{aligned} W_{max} &= \left(\tfrac{t}{RTT} - \log_2(S_{max}/S_{min}) - 1\right)\tfrac{S_{max}}{\beta} \\ &= \left(\tfrac{1}{RTT} - \tfrac{1}{t}\log_2(S_{max}/S_{min}) - \tfrac{1}{t}\right)\tfrac{S_{max}.t}{\beta} \end{aligned}$$

The window ratio is:

$$\frac{W_1}{W_2} = \frac{\left(\frac{1}{RTT_1} - \frac{1}{t}\log_2(S_{max}/S_{min}) - \frac{1}{t}\right)\frac{S_{max}.t}{\beta}}{\left(\frac{1}{RTT_2} - \frac{1}{t}\log_2(S_{max}/S_{min}) - \frac{1}{t}\right)\frac{S_{max}.t}{\beta}}$$

$$\approx \frac{RTT_2}{RTT_1}\ \ , for\ large\ t$$

Binary Search Increase: For $W_{max}\beta \leq 2S_{max}$ we have: $N_1 = 0$, $N_2 = \log_2(\beta W_{max}/S_{min}) + 2$,$n = N_2 = t/RTT$.

$$\ln(2)(N_2 - 2) = \ln(\frac{\beta W_{max}}{S_{min}})$$

$$\frac{S_{min}}{\beta}e^{N_2\ln(2)}.e^{-2\ln(2)} = W_{max}$$

$$W_{max} = \frac{S_{min}}{\beta}e^{-2\ln(2)}.e^{\ln(2)\frac{t}{RTT}}$$

The window ratio is:

$$\frac{W_1}{W_2} = \frac{\frac{S_{min}}{\beta} e^{-2\ln(2)} . e^{\ln(2)\frac{t}{RTT_1}}}{\frac{S_{min}}{\beta} e^{-2\ln(2)} . e^{\ln(2)\frac{t}{RTT_2}}} = e^{(\frac{1}{RTT_1} - \frac{1}{RTT_2})t\ln(2)}$$

B.2 TCP-HS RTT Fairness

Here I also provide the work out of RTT faireness the final result is mentioned in [102]. Consider two flows

$$\begin{cases} w_1 &= w_1(1-\beta_1) + \alpha_1 \frac{t}{RTT_1} \\ w_2 &= w_2(1-\beta_2) + \alpha_2 \frac{t}{RTT_2} \end{cases}$$

Where: $\beta_1 = \beta(w_1)$ and $\alpha_1 = \alpha(w_1)$. The additive incease is given by: $\alpha(w) = \frac{2w^2\beta p}{2-\beta}$, where the window is: $w = \frac{0.15}{p^{0.82}}, p = p(w)$

$$\begin{aligned} w &= w(1-\beta) + \frac{2w^2\beta p}{2-\beta}\frac{t}{RTT} \\ 1 &= (1-\beta) + \frac{2w\beta p}{2-\beta}\frac{t}{RTT} \\ \beta &= \frac{2w\beta p}{2-\beta}\frac{t}{RTT} \\ w &= \frac{(2-\beta)RTT}{2pt} \end{aligned}$$

The RTT faireness is often defined in terms of throughput ratio which can written as two muliplied ratios: the window ratio and the round trip time ratio. So: $\frac{Th_2}{Th_1} = \frac{w_2}{w_1}\frac{RTT_1}{RTT_2}$

Let us have look at the window ratio:

$$\frac{w_1}{w_2} = \frac{\frac{(2-\beta_1)RTT_1}{p_1}}{\frac{(2-\beta_2)RTT_2}{p_2}}$$

$$\begin{aligned} &= \frac{2-\beta_1}{2-\beta_2}\frac{RTT_1}{RTT_2}\frac{p_2}{p_1} \\ &= \frac{2-\beta_1}{2-\beta_2}\frac{RTT_1}{RTT_2}\frac{\sqrt[0.82]{0.15}}{\sqrt[0.82]{w_2}}\frac{\sqrt[0.82]{w_1}}{\sqrt[0.82]{0.15}} \end{aligned}$$

$$\frac{\frac{w_1}{w_2}}{\sqrt[0.82]{\frac{w_1}{w_2}}} = \frac{2-\beta_1}{2-\beta_2}\frac{RTT_1}{RTT_2}$$

$$\left(\frac{w_1}{w_2}\right)^{(1-\frac{1}{0.82})} = \frac{2-\beta_1}{2-\beta_2}\frac{RTT_1}{RTT_2}$$

$$\begin{aligned}\frac{w_1}{w_2} &= \left(\frac{2-\beta_1}{2-\beta_2}\frac{RTT_1}{RTT_2}\right)^{-4.56} \\ &= \left(\frac{2-\beta_2}{2-\beta_1}\frac{RTT_2}{RTT_1}\right)^{4.56}\end{aligned}$$

since: $\frac{Th_1}{Th_2} \propto \frac{RTT_2}{RTT_1}$ to keep a linear RTT unfairness, the term $\left(\frac{RTT_2}{RTT_1}\right)^{\frac{d}{1-d}}$ where $d = 0.82$ in this case, should be kept minimum. In other words as d increases, the RTT unfaireness increases

Appendix C

Philosophical Thoughts

Philosophical Point of View:

Is noise always a bad signal?
In order to illustrate the idea, let us take two analogous cases, one from electrical engineering field and the other from real life:
If things are expected to arrive in a stream in normal operation, then any disturbance to this stream indicates a problem. An excellent example can be seen in the principle of water leak detector, the stream of water flow in a pipe is the normal operation (packets arriving at the bottleneck speed), any leak problem in the pipe disturbs this flow and produces a "noise", special high sensitivity microphones are then used to locate the faulty part.
Electromagnetic signals are detected using special equipment (antennas, then low noise amplifiers, etc), obviously the whole system tries to eliminate any noise picked up by the main signal, the interest is in the original transmitted signal which contains the information, anything else is noise. If we switch our interest to noise, certainly customers have nothing to do with this information they just want to have a clear voice and/or video, but for analytical purposes, the amount of noise (signal to noise ratio), can give information about the environment and the path traversed by the signal (the nature of the environment good or bad for transmission, the amount of interference, security threats signal jamming etc).
Another example inspired from real life is the difference between two horses (signals), one coming from a far distance and the other just walked from a near place. Each have different characteristics (noise) The first one could be breathing fast, sweating,... (more noise) the second one might look more relaxed.

Appendix D

TCP-Gentle Experiments

Testing TCP In High-Speed Long-Delay Environment

Objectives

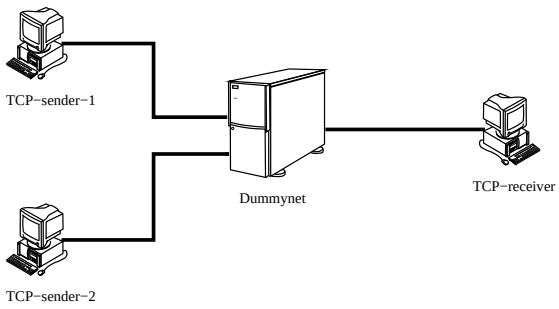
Single flow testing:

1. Plot the congestion window evolution of single TCP-Gentle flow in a high-speed long-delay environment;
2. Measure TCP-Gentle response function.

Double flow testing:

1. Measure Intra-protocol fairness/RTT-unfairness;
2. Measure Inter-protocol fairness (NewReno).

*E*xperimental Set-up



Step-1: 4 Linux boxes and 1 FreeBSD box as a router running `dummynet`. This should be configured to give two pipes with different RTT i.e. different propagation delay. Switch ports should be operating in full-duplex mode. There should be a third pipe (outgoing link) configured as the bottleneck with fixed BW=C, buffer size is set to BDP which can be caclulated as: RTT$\times$ C and a fixed propagation delay. Minimum of both RTTs can be used in the calculation of buffer size.

Step-2: 3 PCs running `Iperf` (TCP sender and receiver) or using other tool like `wget` to download a file, the first option is usually used. The PCs should also have GNU/Linux-2.6.33.3 with TCP-Gentle enabled.

Step-3: Check that 2 PCs (TCP sender) have `tcpprobe` enabled in kernel.

Step-4: When running TCP experiments over high BDP we need to change the maximum availalbe TCP window size, this is done via `sysctl` variables documented in:
`/linux-2.6.33.3/Documentation/networking/ip-sysctl.txt` , at minimum we should increase `tcp_rmem` for the receiver and `tcp_wmem` for the sender to $2 \times BDP$ [3]. Thus we check that the buffer sizes (the aformentioned variables) are big enough to allow the window to grow. The following lines should be added in `/etc/sysctl.conf` for the sender and receiver:

```
# increase Linux TCP buffer limits
net.core.rmem_max = 33554432
net.core.wmem_max = 33554432
net.core.default = 65536
net.core.default = 65536
net.ipso.tcp_rmem = 4096 87380 33554432
net.ipso.tcp_wmem = 4096 65536 33554432
net.ipso.tcp_mem = 33554432 33554432 33554432
```

Other alternative values:

```
# increase TCP max buffer size set-able using setsockopt()
net.core.rmem_max = 16777216
net.core.wmem_max = 16777216

# increase Linux auto-tuning TCP buffer limits
# min, default, and max number of bytes to use
# set max to at least 4MB, or higher if you use very high BDP paths
net.ipv4.tcp_rmem = 4096 87380 16777216
net.ipv4.tcp_wmem = 4096 65536 16777216
```

Step-5: Verify that the following are all set to the default value of 1:

```
sysctl net.ipv4.tcp_window_scaling
sysctl net.ipv4.tcp_timestamps
sysctl net.ipv4.tcp_sack
```

Step-6: Flush old route information, the following lines should be added in `/etc/sysctl.conf` for the sender and receiver:

```
# flush old route information
net.ipv4.route.flush = 1
```

If the test involves repeated connections, you should also turn off the route metrics:

```
sysctl -w net.ipv4.tcp_no_metrics_save=1
```

Because GNU/Linux remember the last `ssthresh`, this causes errors in second tests, [3], `ssthresh` for the route should not be remembered.

Procedure-1: congestion window evolution

Dummynet configuration

Step-1: Configure the path bottleneck BW = 100 Mbps, propagation delay Td = 50 ms (RTT ≈ 100 ms), queue size of BDP where, BDP = (BW × RTT) / (8 × pkt-size) in pkts.

```
ipfw add 100 pipe 1 out src-ip <tcp-tx> // IP address or subnet.
ipfw pipe 1 config bw 100Mbit/s delay 50ms queue 1000
# for bw values are expressed as Mbit/s Kbit/s,
# it has been reported that fractions cause errors
#e.g. dummynet fail to transmit data.
```

Other nodes configuration

Step-1: To plot the congestion window evolution for 20 minutes we use `Iperf` and only one TCP sender, first we start `Iperf` server at the receiver:

```
iperf -s -w <window_size_in_bytes> -p<port_no>
#or for default values just
iperf -s
```

Step-2: Load TCP-Gentle as the congestion control algorithm on the sender and receiver:

```
echo "gentle" > /proc/sys/net/ipv4/tcp_congestion_control
#or
sysctl -w net.ipv4.tcp_congestion_control=gentle
#and check
cat /proc/sys/net/ipv4/tcp_congestion_control
#or
sysctl net.ipv4.tcp_congestion_control=gentle
```

Step-3: Run `tcpprobe` on the sender and place it in background:

```
cat /proc/net/tcpprobe | awk '{print $1 " " $7}' > /tmp/cwnd.dat &
TCPCAP=$!
```

The field description for `tcpprobe` is shown below, we modified the code to probe the values of: AI, estimated queue size and the `do_reno` flag so we can have a microscopic look.

Step-4: Run `Iperf` on the sender for 20 minutes


```
iperf -c <server_ip_addr> -w <window_size_in_bytes> -p <port_no>
      -t <time_in_seconds> -i <interval_in_seconds> -M <mss> -m -r
#or
iperf -i 2 -t 1200 -c <server_ip_addr>
```

Step-5: Kill `tcpprobe` process:

```
kill $TCPCAP
```

Step-6: Use `gnuplot` or any other plotting tool to plot the captured data in `cwnd.dat`.

Procedure-2: Response function

Dummynet configuration

Step-1: Configure the path bottleneck BW = 100 Mbps, propagation delay Td = 50 ms (RTT $\approx$ 100 ms), queue size of BDP where, BDP = (BW $\times$ RTT) / (8 $\times$ pkt-size) in pkts.

```
ipfw add 100 pipe 1 out src-ip <tcp-tx> // IP address or subnet.
ipfw pipe 1 config bw 100Mbit/s delay 50ms queue 1000
```

Step-2: Use the uniform packet drop distribution and set dropping probabilities, Pr, in the range from 10^{-8} to 10^{-1}, first start with 10^{-8}.

```
ipfw add 101 prob Pr deny src-ip <tcp-tx> // IP address or subnet.
#or use it in the bottleneck pipe, so in the above use the following
#line for pipe 1
ipfw pipe 1 config bw 100Mbit/s delay 50ms queue 1000 plr Pr
```

Other nodes configuration

Step-1: Load TCP-Gentle in the same way in Procedure-1.

Step-2: Run `Iperf` in the same way in Procedure-1.

Step-3: Report the average throughput measured by `Iperf`.

Step-4: Go to Step-2 of Dummynet configuration in Procedure-2 and change Pr.

Step-5: Repeat Step-2 to Step-4.

Step-6: Use `gnuplot` or any other plotting tool to plot average throughput versus Pr.

Procedure-3: Intra-protocol fairness

Dummynet configuration

Step-1: Use 2 pipes (for TCP-senders), the delay for the second pipe is variable z (used for RTT-unfairness) and is set to: 1ms, 50ms, 100ms, 150ms, 200ms, 250ms. First start with 1ms.

```
ipfw add 100 pipe 1 in src-ip <tcp-tx> //IP address for first TCP-sender.
ipfw add 101 pipe 2 in src-ip <tcp-tx> //IP address for second TCP-sender.
ipfw pipe 1 config bw 1000Mbit/s delay 1ms
ipfw pipe 2 config bw 1000Mbit/s delay zms
```

Step-2: Configure the path bottleneck BW = 100 Mbps, propagation delay Td = 50 ms (RTT $\approx$ 100 ms), queue size of BDP where, BDP = (BW $\times$ RTT) / (8 $\times$ pkt-size) in pkts.

```
ipfw add 102 pipe 3 out dst-ip <tcp-rx> // IP address or subnet.
ipfw pipe 3 config bw 100Mbit/s delay 50ms queue 1000
```

Other nodes configuration

Step-1: Load TCP-Gentle in the same way in Procedure-1 for the two TCP senders.

Step-2: Run `Iperf` in the same way in Procedure-1 for the two TCP senders.

Step-3: Report the average throughput measured by `Iperf` at each sender side.

Step-4: Calculate Jain's fairness index.

Step-5: Go to Step-1 of Dummynet configuration in Procedure-3 and change z.

Step-6: Repeat Step-2 to Step-5.

Step-7: Use `gnuplot` or any other plotting tool to plot Jain's index versus RTT.

Procedure-4: Inter-protocol fairness (NewReno)

Dummeynet configuration

Step-1: Configure the path bottleneck BW = 100 Mbps, propagation delay Td = 10 ms, 20 ms, 30 ms, 40 ms, 50 ms. Queue size of BDP where, $BDP_{Td=10ms}$ = (BW $\times$ $RTT_{Td=10ms}$) / (8 $\times$ pkt-size) in pkts.

```
ipfw add 102 pipe 3 out dst-ip <tcp-rx> // IP address or subnet.
ipfw pipe 3 config bw 100Mbit/s delay 10ms queue 200
```

Other nodes configuration

Step-1: Load TCP-Gentle in the same way in Procedure-1 for the first TCP-sender.

Step-2: Load TCP-NewReno in the same way in Procedure-1 for the second TCP-sender. The following command can be used to see the available congestion control modules.

```
sysctl net.ipv4.tcp_available_congestion_control
#or if they are loaded
ls /lib/modules/‘uname -r‘/build/net/ipv4
```

Step-3: Run `Iperf` in the same way in Procedure-1 for the two TCP senders.

Step-4: Report the average throughput measured by `Iperf` at each sender side.

Step-5: Calculate the asymmetry index.

Step-6: Repeat Step-3 to Step-5.

Step-7: Use `gnuplot` or any other plotting tool to plot asymmetry index versus RTT.

Repeat Procedure-3 and Procedure-4 for TCP-YeAH[1] instead of TCP-Gentle.

[1]The name is YeAH TCP, but we reversed it for consistency since we used TCP-X to refer to TCP variants.