EVALUATION OF PROBABILISTIC EARLY RESPONSE TCP (PERT) FOR VIDEO DELIVERY AND EXTENSION WITH ACK COALESCING

A Thesis

by

BIN QIAN

Submitted to the Office of Graduate Studies of
Texas A&M University
in partial fulfillment of the requirements for the degree of

MASTER OF SCIENCE

August 2011

Major Subject: Computer Engineering

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Approved by:

Chair of Committee,	A.L. Narasimha Reddy
Committee Members,	Srinivas Shakkottai
	Riccardo Bettati
Head of Department,	Costas N. Georghiades

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ABSTRACT

Evaluation of Probabilistic Early Response TCP (PERT) for Video Delivery and Extension with ACK Coalescing. (August 2011)

Bin Qian, B.E., Zhejiang University, Hangzhou, China

Chair of Advisory Committee: Dr. A. L. Narasimha Reddy

This thesis demonstrates the performance of Probabilistic Early Response TCP (PERT), a new TCP congestion control, for video streaming. As a delay based protocol, it measures the delay at the end host and adjusts the congestion window accordingly. Our experiments show that PERT improves video delivery performance by decreasing the fraction of packets delivered late. Furthermore, our Linux live streaming test indicates that PERT is able to reduce the playback glitches, when high resolution video is delivered over a link with non-zero packet loss. In order to operate PERT at higher thoughputs, we design PERT to work with Acknowledgement (ACK) coalescing at the receiver. ACK coalescing makes data transfers burstier and makes it hard to estimate delays accurately. We apply TCP pacing to fix this issue, and validate its effectiveness in the aspects of throughput, packet loss and fairness. Our experiment results also show that PERT with Delayed ACK and Pacing is more friendly, and therefore more suitable when multiple traffic flows are competing for limited bottleneck bandwidth or sharing the same router buffer.

To my parents

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CHAPTER I

INTRODUCTION

A. Motivation

Data transfers over Internet are increasingly dominated by video data transfers. Cisco forecasts that video transfers will account for 60-90% of network traffic by 2014 [1]. Recent studies indicate that a large portion of Internet streaming media in current Internet is delivered over HTTP/TCP. To achieve satisfactory quality of service (QOS), video and audio data are supposed to be delivered before playback or buffered if they arrive earlier. However, current TCP is not suitable for video streaming applications due to its insistence on reliable transmission and inability of real-time data delivery. Moreover, in today's Internet, many other services like web surfing, FTP download, as well as P2P file sharing are also competing for the limited bandwidth. This makes it more difficult for TCP to meet the demands of smooth video streaming.

Nowadays, the demand for ubiquitous connectivity and cloud computing has led to an interest in improving TCP for wireless or high-speed networks, both in terms of commercial exploitation and with regard to research inquiry. Storing massive content in the cloud and streaming to the clients becomes more and more popular today. In the cloud system, the cluster of servers are connected to the high-speed Internet. On the client end, users usually are using mobile devices like laptop or smart phone

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via Wi-Fi or 3G networks. For example, recently Amazon.com Inc. has released its cloud player and instant video to provide on-demand audio and video streaming from their distributed cloud system to their users' personal computers, netbooks and smart phones via TCP connections. Such applications ask for high performance of TCP over complex mixture of high-speed Internet and wireless 3G or Wi-Fi networks. This increasing trend of mobile audio and video streaming services requires further extension and enhancement of TCP to satisfy quality of service.

B. Related Work

Motivated by such demand on multimedia applications over the Internet, protocols for video streaming have been explored by many researchers. TFRC (TCP Friendly Rate Control) [2] and its variant [3] have been proposed to maintain long-term TCP fairness while maintaining smooth transmission rates. In [4], Wang et al. analytically studied the TCP performance for multimedia streaming. They built discrete-time Markov models for both constrained and unconstrained streaming. Smaller than MSS-sized packets have been used in CBR workloads to exploit the TCP ACK counting mechanism, and thereby reducing the TCP transport delay and its impact on congestion window variations in [5]. In [6], the authors compared Linux implementations of NEWRENO, H-TCP and CUBIC and found dynamic latency fluctuations induced by each TCP variant. They noticed that CUBIC induces larger latency than the other two when concurrent TCP flows take place. All of these studies explore the possibility of employing TCP like congestion control even for real-time video delivery.

To improve the on-time data delivery and quality of service, especially in wireless networks, many studies focus on reducing the TCP transmission overhead by minimizing the number of produced ACKs for TCP without compromising reliability. Unlike the TCP DATA packets, ACKs are considered as control traffic since they are used only to confirm the reliable delivery. Therefore, it is desirable to minimize the amount of ACKs traffic so as to make more bandwidth available for the actual data delivery. The delayed ACKs mechanism is first proposed by the paper [7] and has been enabled as a feature of TCP standard. The basic idea relies on the cumulative nature of TCP ACKs. The TCP receiver set-up a timer for ACK (100-500 ms by default [8]) upon receiving the data packet, and inject a single ACK into the pipe when the timer times out. To avoid causing problems to TCP round trip-time (RTT) estimation and ACK-clocking, the maximum timeout for a ACK is limited to no more than a single data packet (usually 500ms), according to RFC [9]. The authors of [10] proposed to enable delay ACKs by default based on its substantial benefit for TCP throughput. It also reports the possibility of raising the TCP delay response so as to reduce the competition of ACK for bottleneck bandwidth with DATA packets. Later on, they implemented the delay ACK in TCP as a default feature, which enables the receiver to wait for a short period of time instead of immediately replying to each data packet [11]. In this case, if the subsequent packet arrives, then the receiver sends ACK to verify both of them. Further, the authors of [12] introduced the possibility of producing delayed ACK for more than two received data packets, and confirmed with extensive experiments that an ACK for four data packets can guarantee good performance when applying delayed ACK in general environments. In wireless networks, the throughput enhancement effect [13] has been fully demonstrated in static and dynamic topologies, with reactive [14] and proactive [15] TCP hosts. But there are still some problems to be solved, before delayed ACK mechanism can be widely deployed on the Internet. The main shortcoming is its impact on

ACK clocking mechanism, which results in the burstiness in the transmission pattern [16]. We refer this delayed ACK mechanism and its data burstiness defects as ACK coalescing. To solve this burstiness problem, the paper [17] initially suggested using pacing to reduce burstiness of TCP traffic caused by ACK compression. The basic idea is to pace out the packets at the intervals of $\frac{RTT}{cwnd}$, so that less data packets will be queued up in the router buffer and the self-induced delay will be reduced. This is later on referred as TCP pacing in literature. Also, many researchers have found that TCP's congestion control mechanism can result in bursty traffic, with a negative influence on network efficiency. According to [18], however authors found TCP pacing is susceptible to synchronized packet losses and delays congestion signals. They further proposed ways to eliminate this impact and validate its effectiveness. Moreover, researchers [19] found that pacing helps to reduce the worst-flow latency and improve the aggregate throughput, which are important for increasingly popular type of distributed application platforms nowadays.

In this thesis, we explore the performance of a new TCP congestion control - PERT and compare it with other TCP variants like RENO and CUBIC for real-time video transmission. Here RENO is short for RENO-SACK. We study if the delay-based PERT mechanism can provide better support for video delivery than RENO and CUBIC. We study this problem through NS-2 based simulations and real live video transmission tests on a testbed. Both our NS-2 simulation and Linux test results show that PERT provides significant improvement on video viewing quality when compared to RENO and CUBIC. Moreover, we made an extension of PERT, making it work with delayed ACK and pacing, in order to achieve better performance. We carefully implemented an adaptive delayed ACK mechanism and verified its benefits of significantly reducing ACK traffic while improving the throughput. We identified that delayed ACK mechanisms can disrupt the delay estimation mechanism of PERT. To be specific, the burst of data sent after the reception of an ACK results in self-induced delay and as a result causes RTT to be incorrectly over-estimated. To overcome this issue, we applied TCP pacing and verified its effectiveness. We further demonstrated that pacing bring more fairness benefits and therefore makes delayed ACK more practical. Finally, we perform video streaming tests for PERT with delayed ACK and pacing. Our results indicated that the more friendly PERT tends to have higher aggregate throughput and lower late packet rate when multiple flows are competing, especially when the Bandwidth-Delay-Product (BDP) is small. This is to say, the new extension makes PERT more useful in wider network settings.

CHAPTER II

A BRIEF DESCRIPTION OF PERT

PERT emulates the behavior of AQM/ECN at the end host [20]. As a delay based protocol, PERT learns about network congestion by measuring delays at end host, and probabilistically reduces the congestion window as delay increases. Fig. 1 shows PERT's response probability curve, where T_{min} and T_{max} are two thresholds, and P_{max} is the probability of response at T_{max}.

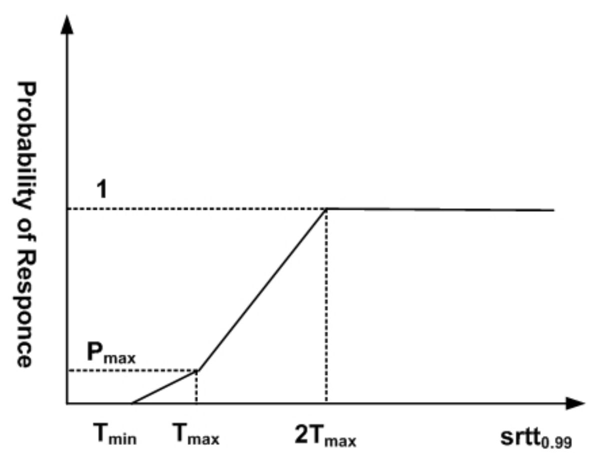


Figure 1: Response probability vs. smoothed RTT

However, delay based protocols lose to loss-based protocols in heterogeneous environments where multiple congestion control algorithms may be employed. In current Internet, any new congestion control algorithm has to be able to coexist with prevailing TCP congestion control algorithms. To address this problem, PERT was redesigned to be adaptive to heterogeneous environments [21]. PERT increases congestion window faster than TCP at low delays to compensate for early response at higher delays, in order to equalize the bandwidth.

PERT basically operates in 3 modes. When the observed delay is very low (or below the minimum threshold), it assumes that it is operating in a "high-speed" mode and increases the window fast to fill the link. In this mode, the window increase factor α in W = W + α, is increased linearly until a maximum value of α_{max} (currently set to 32). When the observed delay is above a TCP-compete threshold (currently set to 0.65*maximum observed queuing delay), PERT assumes it is operating in a heterogeneous environment and increases the window every RTT additively with $\alpha = 1 + p'/p$, where p' is the early response probability and p is the observed packet loss rate. When the observed delay is above the minimum threshold, but below the TCP-compete threshold, PERT assumes it is operating in a "safe" mode and increments window additively with $\alpha = 1$. In addition, PERT reduces the window conservatively in the early response phase, $W = W \times (1 - \beta)$, where $\beta = q'/(q' + q)$, where q' is the estimated queuing delay at early response phase and q is the observed maximum queueing delay. It is observed that this leads to $W = W/2$ upon a packet loss.

Simultions and real-network evalutions have shown that, (a) a single PERT flow can scale to high-speed links of up to 10Gbps, (b) PERT can compete with TCP in heterogeneous environments and (c) still benefit from near-zero packet loss rates and very low queuing delays when operating in homogeneous environments. Details of PERT design can be found in [21, 22].

PERT sends more packets at lower delays and sends fewer packets at higher delays while being fair to TCP. This behavior is shown in Fig. 2. Fig. 2 shows the queuing

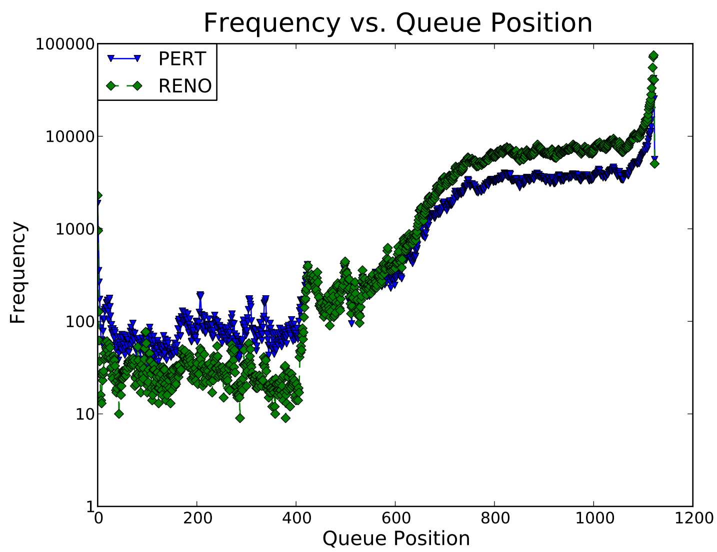


Figure 2: Frequency vs. queue position

frequency at certain queue length when 50 PERT and 50 RENO FTP flows are competing on a 150 Mbps link with 60ms delay. It is observed that PERT enqueues more packets earlier in the queue, and less packets later, and as a result experiences lower losses. We expect this behavior of PERT to be beneficial for video transfers and this thesis investigates this issue through simulations and experiments over a testbed.

Recent work [23] has showed that TCP can be used for transmitting live video as long as the required video stream rate is a fraction of the average bandwidth achievable by a single TCP flow. Both constrained (data is available from a live stream) and unconstrained (data is available from a prerecorded or stored source) streaming were considered. Extensive simulations have shown that TCP can be adequate if the average TCP flow rate is about twice that of the required video stream rate for constrained streaming. A similar study by [24] has shown that TCP can function adequately with a 1.5 higher bandwidth than required stream rate in unconstrained streaming and that Vegas could support unconstrained streaming better than TCP NEWRENO. The question we try to answer in this thesis is if a delay-based protocol such as PERT can support constrained video streaming at a lower available bandwidth than two times of the required video stream rate as required by RENO.

We carry out extensive NS-2 based simulations and live video transmissions on a real network testbed within the lab. We present data from both simulations and the emulations to show that PERT indeed provides better support for live video transmission than RENO and CUBIC.

CHAPTER III

MEASUREMENT FOR VIDEO STREAMING

A. NS-2 Simulation

1. Experiment Setup

To evaluate the performance of PERT and other TCP congestion control variants, we setup a dumbbell topology, as Fig. 3 shows. In such a network environment, multiple TCP streams have sufficient bandwidth over access links separately but compete for the limited bandwidth over the bottleneck link between the two routers.

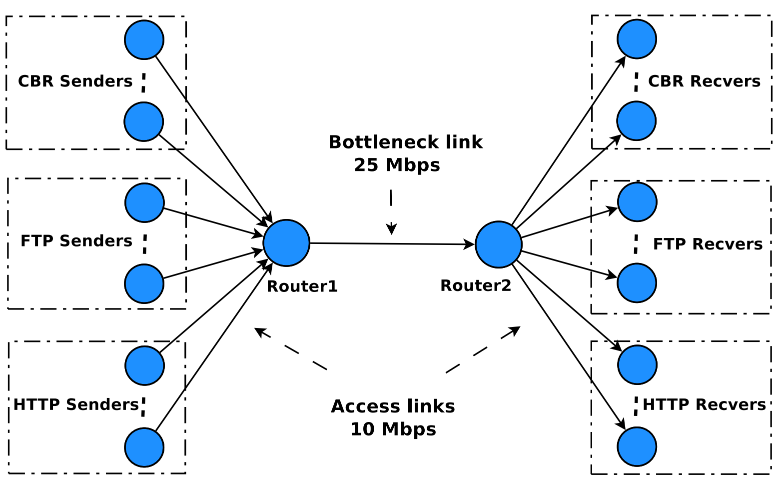


Figure 3: Dumbbell topology

Table. I shows our experiment parameters in NS-2 simulation, we set the access links bandwidth to 10 Mbps and bottleneck link bandwidth to 25 Mbps, and CBR bit rate

Table I.: NS-2 simulation experiment setup

Parameter	Value
CBR Flows #	20 - 35
CBR Senders	PERT/RENO/CUBIC
CBR Recvers	RENO
CBR Rate	300 Kbps
FTP Flows #	20 - 35
FTP Senders	RENO
FTP Recvers	RENO
HTTP Flows #	300
HTTP Senders	RENO
HTTP Recvers	RENO
Video Length	7,000 secs
Packet Size	200/1,000 Bytes
Buffer Size	150 Packets
Access Link Bandwidth	10 Mbps
Access Link Delay	5 - 15 ms
Bottleneck Link Bandwidth	25 Mbps
Bottleneck Link Delay	15 - 45 ms
Round Trip Time	50 - 150 ms
Random Seed	0 - 19

to 300 Kbps. Moreover, TCP packet size is set to 200 or 1,000 bytes, router buffer size to 150 packets, and video length to 7,000 seconds. We keep the above parameters constant to get rid of their impacts on TCP video streaming performance. We take HTTP and FTP flows as the background traffic, RENO and CUBIC as the control group. As a loss based protocol, RENO additively increases the congestion window by one MSS (Maximum Segment Size) every RTT (Round Trip Time), cuts down the congestion window by half on a packet loss and decreases it to one MSS on a timeout event. As for CUBIC, the congestion window growth follows a cubic function in terms of the elapsed time since the last loss event. To emulate a realistic network, we vary link delay (RTT) by altering the access link delay and bottleneck link delay. We also vary the number of CBR streams and the number of FTP streams from 20-35 and keep the number of HTTP streams constant at 300 to achieve different TCP throughputs. Finally, we run the simulation 20 times with seed values of 0-19 to randomize the start time of the TCP streams, in order to statistically reduce its effect on the experiment results.

2. Simulation Results

a. Parameters Exploration

In this section, we explored the experiment parameters to study the performance of PERT under different conditions. As [23] concludes, the performance of TCP generally provides good streaming performance when the T/μ is roughly 2.0, where T is the achievable TCP throughput and μ is the video bit-rate. To demonstrate PERT's performance under different T/μs, we pick sample data with certain CBR stream numbers such that T/μ falls in continual ranges of [1.0 - 1.2], [1.2 - 1.4], [1.4 - 1.6], [1.6 - 1.8], [1.8 - 2.0], as Fig. 4 shows. Under such T/μ distribution, we also plot the bandwidth allocation among CBR, FTP and HTTP streams. As Fig. 5 shows, as the number of CBR streams increases, the total bandwidth of CBR streams increases proportionally, and the rest of bandwidth is taken by FTP streams and HTTP streams.

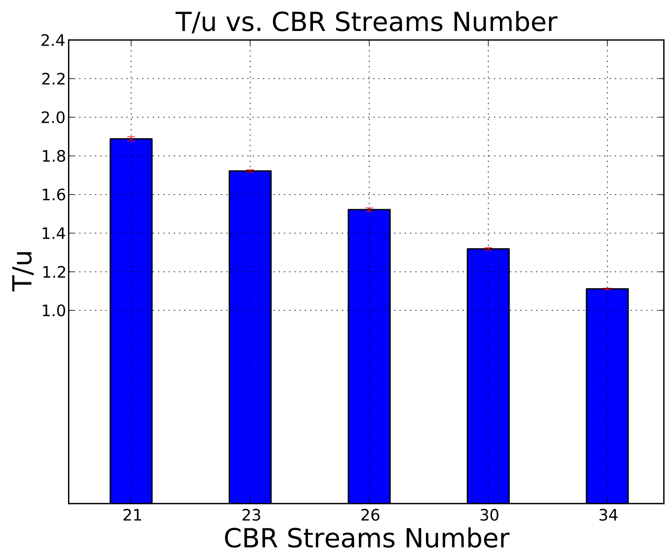


Figure 4: T/μ distribution vs. CBR number

As Fig. 6 shows, the fraction of late packets for PERT CBR streams becomes smaller as T/μ increases from 1.0 to 2.0. It is clear that the late packets can be reduced by giving CBR traffic more bandwidth. Fig. 6 also indicates that the fraction of late packets with PERT drops sharply when T/μ is increased from 1.0 to 1.4, and stays almost the same as T/μ ranges from 1.4 to 2.0. Moreover, as the RTT increases from 50ms to 150ms, the fraction of late packets goes up. This agrees with our intuition that it is more difficult to achieve satisfactory performance for video streaming in

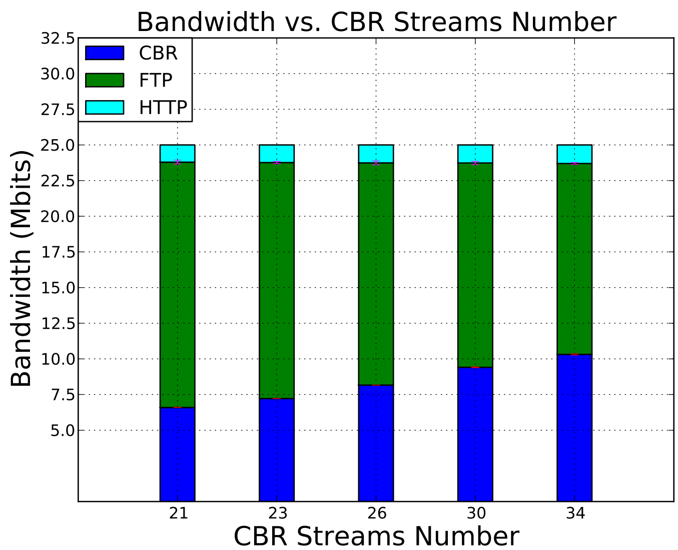


Figure 5: CBR, FTP and HTTP bandwidth vs. CBR number

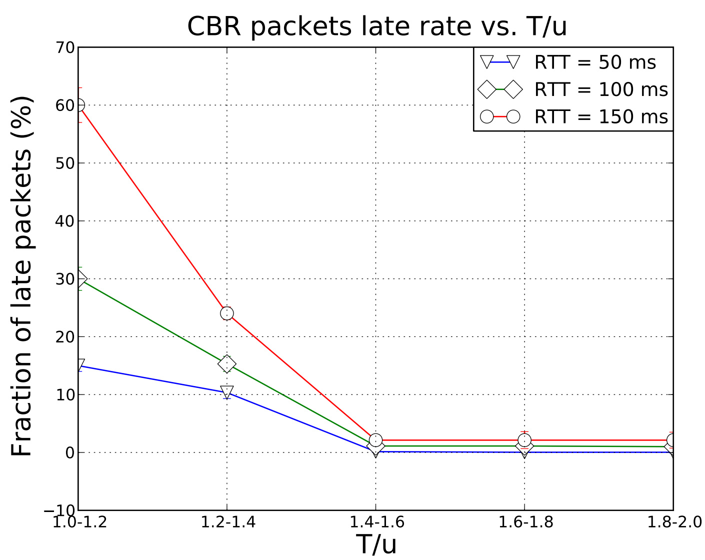


Figure 6: Fraction of late packets vs. T/μ with PERT

higher delay networks (retransmissions may not arrive in time, for example).

Studies show that the video viewing quality is closely related to the fraction of late packets. We define that a CBR stream is successful as long as the fraction of late packets is below 10^{-4}, with only a few seconds of startup delay as the paper [23] did. Under such an evaluation metric, we validate PERT's performance and compare with others in the aspect of delivered video quality.

According to Fig. 7, as the start-up delay increases from 1 to 30 seconds, the fraction of successful streams goes up. This is quite intuitive that more packets can meet their playback deadline if we allow larger start-up delay for initial buffering. It is also noticeable that when T/μ is in the range of [1.6 -2.0] and start-up delay is greater than 11 seconds, the fraction of successful CBR streams gets to nearly 100%. In other words, video streaming by PERT works well when T/μ is above 1.6, i.e. 20% lower than what is required, 2.0 for RENO. In the next part, we will show that RENO does not perform well under the same conditions.

As Fig. 8 shows, as packet loss rate gets higher, the fraction of successful streams gets lower. At low loss rates (0.00 - 0.02), by using PERT, the fraction of successful CBR streams can be above 90%, across all T/μ ranges considered, if more than 11 seconds of start-up delay is allowed. Even when the transmission suffers from severe packets loss (0.04 - 0.06), 60% of PERT streams can successfully deliver video. This is because when the loss event occurs, PERT does not drastically cut down the congestion window and lower the flow rate as RENO does. In the next part, we will show that almost of all RENO streams fail in similar situations.

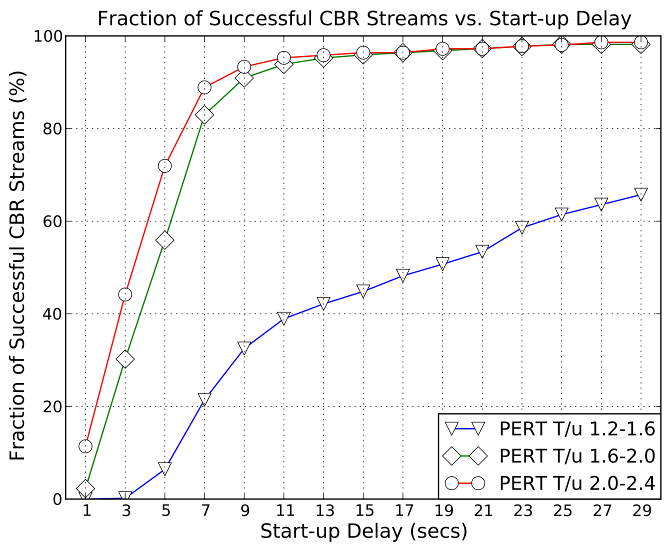


Figure 7: Fraction of successful CBR streams over different T/μ

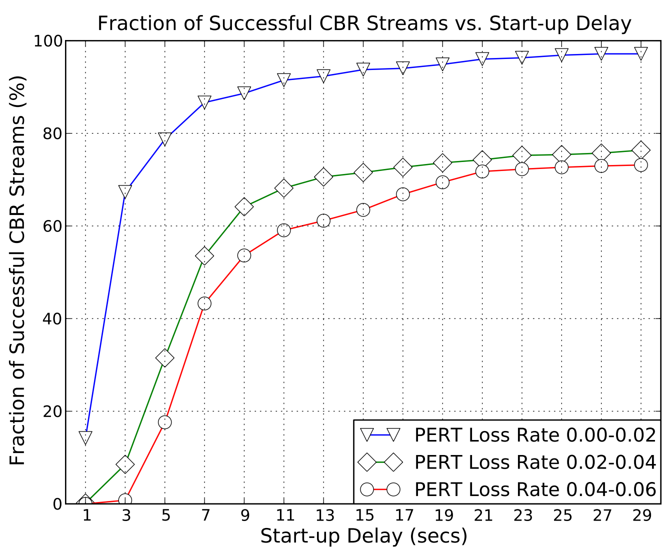


Figure 8: Fraction of successful CBR streams over different packet loss rate

b. Performance Comparison

In this part, we compare the performance of PERT, RENO and CUBIC for video streaming in the same parameter space that we described in the last section. As our simulation results indicate, PERT outperforms RENO over all T/μ, loss rates and start-up delays, and CUBIC over low T/μ, high loss rates and strict start-up delay constraints.

First, we demonstrate the performance in terms of fraction of successful CBR streams that we defined before. Fig. 9 indicates that as T/μ increases, the fraction of successful CBR streams goes up. In the high T/μ range [1.4 - 2.0], the percentage of successful CBR streams is high and changes slightly as T/μ increases. While in the low T/μ range [1.0 - 1.4], the performance drops drastically as T/μ decreases. In comparison, PERT and CUBIC perform better than RENO. When packet size is moderate - 1000 bytes, PERT performs better than CUBIC in the low T/μ range but has similar performance as CUBIC in the high T/μ range. Fig. 9 also shows the impact of packet size, we consider two packet sizes of 200 and 1000 bytes. It is observed that the fraction of successful streams is higher with smaller packet sizes, in all T/μ ranges. Moreover, smaller packet sizes of 200 bytes helps to boost the performance, and PERT has better or at least the same performance when comparing to CUBIC.

Fig. 10 show that the percentage of successful CBR streams vs. start-up delays in different T/μ ranges when three different TCP congestion controls are employed in video transmission. As T/μ increases, the CBR streams can achieve higher throughput and lower packet loss rate. And as the start-up delay goes up and the constraint becomes loose, more CBR streams successfully are able to meet the streaming quality

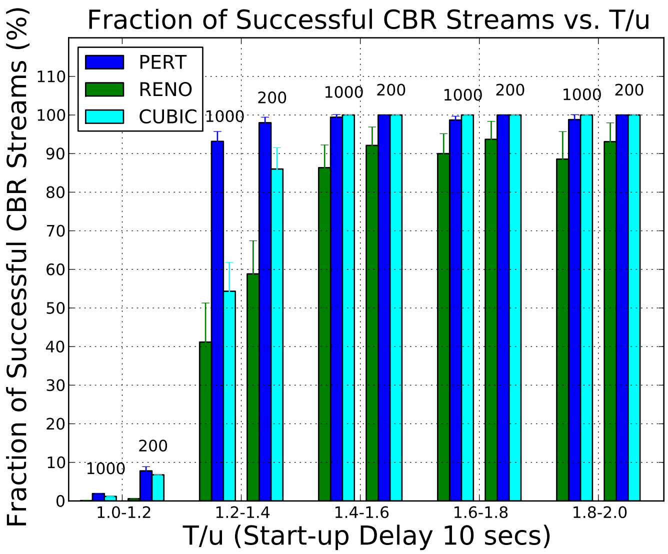


Figure 9: Fraction of successful CBR streams vs. T/μ

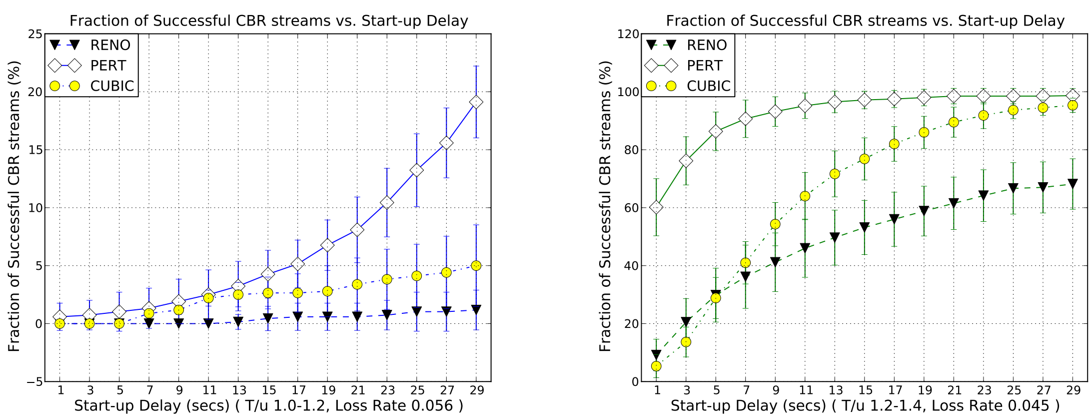


Figure 10: Fraction of successful CBR streams over T/μ 1.0-1.4

requirement. These observations are consistent with the earlier study [23].

In comparison, when the T/μ is in the low range [1.0 - 1.4], the loss rate is relatively high, PERT achieves the best performance, CUBIC is in the middle, and RENO is the worst.

As Fig. 11 shows, in the high T/μ range [1.4 - 1.8], PERT and CUBIC are both superior to RENO, and especially when the start-up delay constraint is tight (from 1 to 11 seconds). In addition, when the start-up delay is greater than 11 seconds, PERT and CUBIC achieve almost 100% success rate. In other words, the T/μ constraint for satisfactory streaming is improved from roughly 2.0 to approximately 1.4, by using PERT or CUBIC.

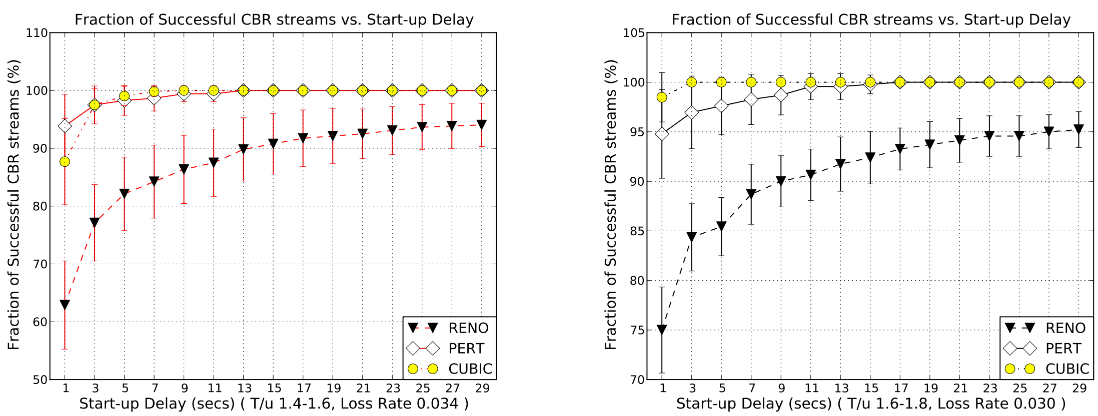


Figure 11: Fraction of successful CBR streams over T/μ 1.4-1.8

Fig. 12 shows that PERT generally performs better than RENO and CUBIC, in the loss rate range of [0.04 - 0.06]. This is because PERT reduces the congestion window by small amounts ahead of packet loss and experiences fewer packet losses [21], which brings more stable throughput and therefore smoother video streaming.

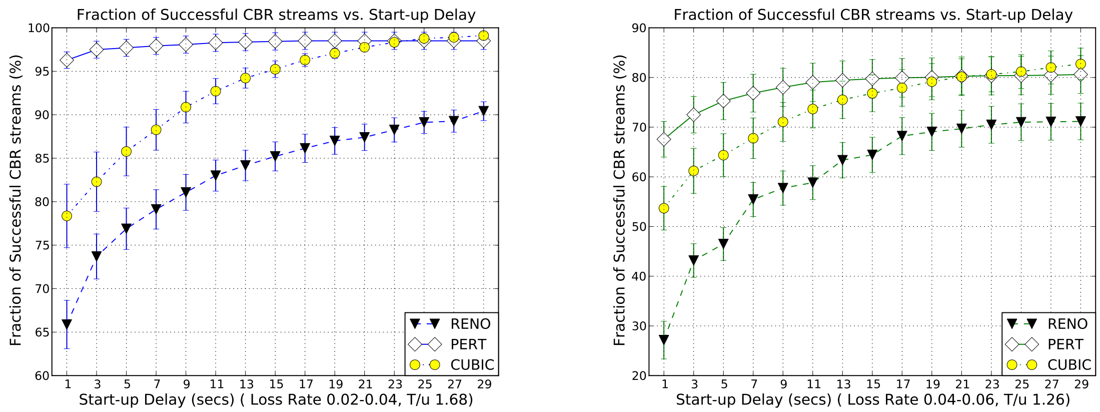


Figure 12: Fraction of successful CBR streams over loss rate 0.02-0.06

B. Linux Video Streaming Test

In order to test our Linux implementation of PERT, we configured a testbed environment with the help of a network emulator. Fig. 13 displays our testbed setup. Two PCs with the same hardware and operating system, are connected to a PC configured as a network switch, through a 10/100 Mbs Ethernet link. The VLC [25] application is installed to stream a video from the sender to the receiver. DummyNet [26] is employed as the network emulator to create a more realistic testing condition. It works on the switch to emulate buffering, queuing delays and bottleneck link bandwidth.

Based on the testbed described above, we performed the video streaming test with PERT, RENO and CUBIC. We set the link bandwidth to 15 Mbps, the link delay to 45 ms, and the queue length to 500 kbytes on the bridge. Moreover, a 1080p version of

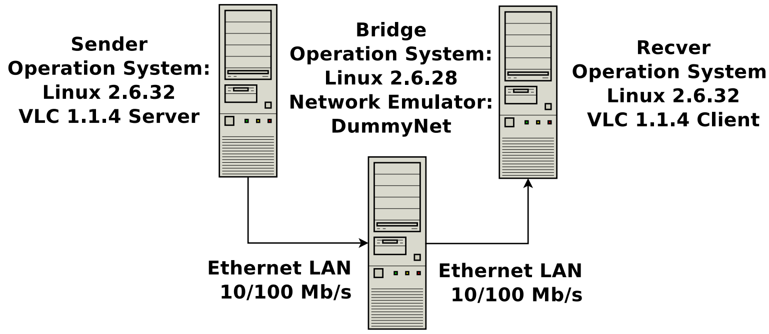


Figure 13: Experiment computer platform

the Avatar movie trailer is chosen as the sample video, with file size of 286.5 Mb and playback duration of 3 minutes and 30 seconds. VLC 1.1.4 works on both the server and the client end as the video streaming tool. The codec of the video and audio are H-264 and MPEG 4 Audio (AAC) respectively. As a high resolution movie, the video is played at a frame rate of 24 fps and the audio at a sampling rate of $48,000$ Hz. This asks for video peak bit rate of 25 mbps and audio bit rate of 192 kbps. Besides, HTTP is chosen as the application layer protocol, which takes advantage of TCP on the transport layer. We employ different versions of TCP during our experiments to measure their effectiveness at streaming.

1. Test Results

To compare the effects of PERT, RENO and CUBIC on the perspective of video viewing quality, we analysed VLC's playback logs, which record when and how if any glitch happens during the video transmission. We play the same video with the same setting 20 times and count how many times the events of late picture skipping and audio output starving occur. When the video or audio frames are played but not found in the client's buffer, one of these events will occur. They lead to playback glitches, VLC client buffering, user waiting and therefore impair the viewing quality. As Table. II shows, the playback experienced smaller number of late picture skipping and audio output starving events when using PERT instead of RENO or CUBIC as the TCP congestion control.

Table II.: Video streaming performance comparison

TCP Congestion Control	PERT	RENO	CUBIC
Late Picture Skipping # (per playback)	5.5	33.5	30.5
Audio Output Starving # (per playback)	3.0	11.0	7.5

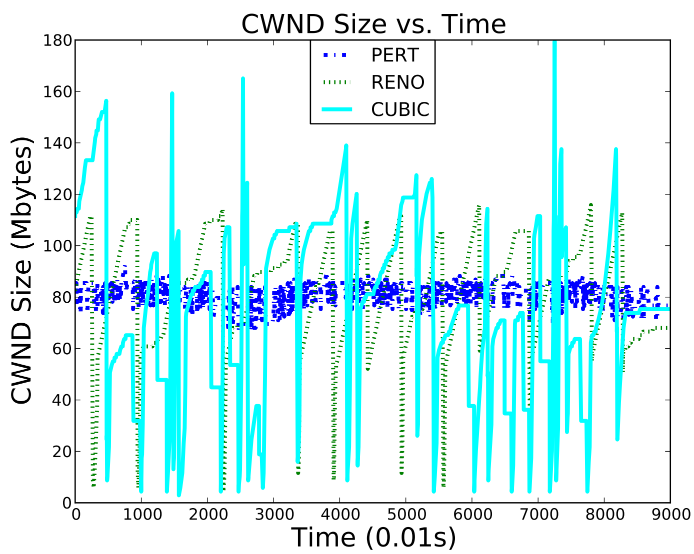


Figure 14: Congestion window in Linux Test

To confirm our observation, we plot the TCP congestion window size with web100 [27] at the sending end during playback in Fig. 14. We choose a period of 90 seconds of playback time at the end of the movie, since the streaming is relatively stable during that time. And we track the congestion window size every 0.01 second. As Fig. 14 shows, the congestion window size of PERT has fewer fluctuations than that of RENO or CUBIC do. CUBIC increases the congestion window fast and achieves sightly better performance than RENO does, but still incurs large fluctuations. Therefore, the throughput of PERT's video streams are more steady and their data frames are more likely to arrive before playback deadline. This can explain the smaller number of late picture skipping and audio output starving events we observed, when PERT is employed as the TCP congestion control.

CHAPTER IV

ENHANCEMENT WITH DELAY ACKS AND PACING

A. Delayed ACK

In this section, we discuss our extension of PERT with delayed ACK and pacing. We first describe our implementation of adaptive delayed ACK mechanism and observation that delayed ACK mechanisms can disrupt the delay estimation mechanism of PERT. To overcome the RTT over-estimation issue, we introduce TCP pacing and its effectiveness of reducing data bustiness. We further demonstrate that pacing bring more fairness benefits and therefore makes delayed ACK more practical. Finally, we show our video streaming tests results for PERT with delayed ACK and pacing.

1. Implementation

Combining the ideas of paper [11] and [12], we proposed an adaptive delayed ACK algorithm. In our algorithm, not only the number of accumulated ACKs (1 - 4 in our implementation), but also The ACKs sending time (0 - 2 $\times$ data packet arrival interval in out implementation), are adaptive, according to current transmission status. The specific mechanism will be fully described and its effectiveness will be validated in this section.

As the Fig. 15 shows. Our proposed algorithm takes advantage of the cumulative

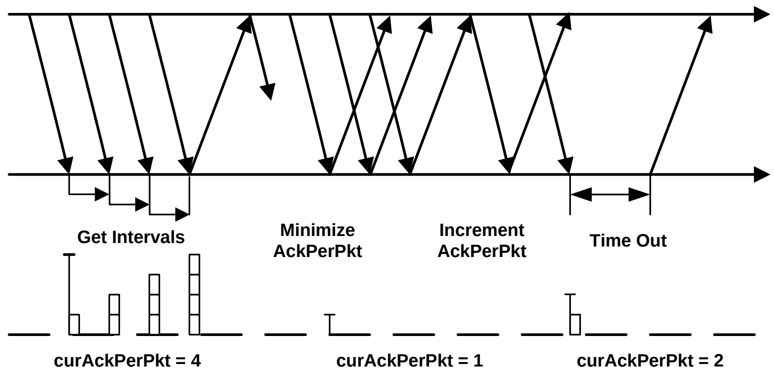


Figure 15: The mechanism of adaptive delayed ACK

property of ACKs and simply added two kinds of adaptiveness: 1. The number of accumulated ACKs can be varied, when the data packet arrives in order and in time, we increment the maximum value of the ACKs number we can accumulate, which means we can save more ACK traffic. In our implementation, the number of accumulated ACKs is initialed to the minimum value 1, and can be incremented to the maximum value 4, if packets are delivered in order without failures. 2. The time when ACKs are sent out is varied. For each in-order delivered packet, we record its time and calculate the interval between the last one, and calculate the expected data packet arrival time with exponential averaging ($new\_estimate = \alpha \times old\_estimate + (1 - \alpha) \times new\_sample$). We set β ($\beta > 1$) times expected data packet arrival time as the estimated time out threshold for each data packet. If timer for the expect packet is time out, we simply send out ACK with the last data packet sequence number.

Moreover, when out of order packet comes or packet loss happens, we just return back to the normal one packet one ACK routine, to guarantee the reliability of the transmission. First, if the channel experiences some bad condition or severe congestion, the expected data packet comes after a large delay, then the ACK should be sent to keep the sender updated about the transmission status. And this time-out mechanism is also designed to avoid 'deadlock', which can happen when the receiver will still accumulate (does not send) ACKs but actually the sender has no more packets to send. In such a case, the sender expects the receiver to send the acknowledgement for the last data packet, but, the receiver cannot send out the ACK since it try to accumulate more ACKs.

Our implementation pseudo code is displayed as Alg. 1, it achieves the first adaptiveness by using two thresholds: minPktPerAck and maxPktPerAck, which denote the minimum and maximum values of how many ACKs can be accumulated. And curPktPerAck is maintained to keep a record of how many ACKs currently we can accumulate, while counter records the number of ACKs we have already accumulated. This implementation makes sure that we reduce the ACK traffic when the channel condition is good, which is indicated by whether the data packets arrive in order or not. We can also quickly response to the link error or congestion, by immediately switching to normal TCP ACK behavior by reducing curPktPerAck to minPktPerAck. In our implementation, we set minPktPerAck to one. This is because TCP sender starts from the congestion window size of one packet and cuts down the congestion window size to one when sender waits for ACK and times out. The reason why the curPktPerAck cannot exceed the congestion windows size is obvious. If it is so, the 'deadlock' described above will happen.

In addition, it basically calls two sub-routines to realize the second adaptiveness as follows:

1. getPktArrInterval() (as Alg. 2 shows)- which basically calculates the expected

Algorithm 1: Adaptive Delay ACK

Input: $minPktPerAck = 1$, $maxPktPerAck = 4$, $\alpha = 0.9$, $\beta = 2.0$.

begin
- $interval \longleftarrow 0$;
- $counter \longleftarrow 0$;
- $curPktPerAck \longleftarrow minPktPerAck$;
- **if** $dataPktSeqNo == ackPktSeqNo$ **then**
  - getPktArrInterval(); (Algorithm 2) ;
  - $counter \longleftarrow counter + 1$;
  - **if** $curPktPerAck == curPktPerAck$ **then**
    - send ACK with ackPktSeqNo;
    - $counter \longleftarrow 0$;
  - **else**
    - setupPktTimer(); (Algorithm 3) ;
  - **end**
  - **if** $curPktPerAck < maxPktPerAck$ **then**
    - $curPktPerAck \longleftarrow curPktPerAck + 1$;
  - **end**
- **else**
  - send ACK with ackPktSeqNo;
  - $curPktPerAck \longleftarrow 0$;
  - $curPktPerAck \longleftarrow minPktPerAck$;
- **end**

end

packet inter-arrival time. As it shows, it only updates the value when the packet is delivered in order, and smooths the measured value to get the expected one. Let $\hat{t_i}$ be the last expected value, $\hat{t_{i+1}}$ is the next expected value, then $\hat{t_{i+1}} = \alpha \times \hat{t_i} + (1-\alpha) \times t_i$, where α is the smoothed factor (in our implementation α is set to 0.9) and t_i is our actual sample packet arrival interval.

Algorithm 2: getPktArrInterval

Input: $curPktPerAck$, $interval$, $lastTime$.

begin

    if $curPktPerAck > 0$ **then**

        $interval \longleftarrow \alpha * (interval) + (1 - \alpha) * (curTime - lastTime)$;

        $lastTime \longleftarrow curTime$;

    end

end

2. setupPktTimer() (as Alg. 3 shows) simply set-up a timer for each accumulated packet based on the expected packet arriving interval estimated by Alg. 2. To be specific, $T_i = \beta * \hat{t_i}$, where β is the timeout tolerance factor (in our implementation β is set to 2.0) and T_i is the timeout threshold. It is also worth mentioning that if any data packet arrives after T_i since the last one, the ACK will be immediately sent and ACK accumulating mechanism will be aborted and restarted. This is to say, our algorithm cannot differentiate packet loss or ACK timeout. We conservatively re-start the whole mechanism to avoid potential deadlocks. Our experiment results confirm our expectation that this conservative mechanism achieves better performance in terms of throughput and latency, especially when the transmission experiences some timeouts.

Algorithm 3: setupPktTimer()

Input: $curPktPerAck$, $interval$, $lastTime$.

begin

```
// set-up a timer for each accumulated packet,
// in case of packet loss or no more packets.
```

$timeout \longleftarrow \beta * interval$;

```
// if timeout occurs, send ACK with ackPktSeqNo,
```

// $curPktPerAck \longleftarrow minPktPerAck$,

// $counter \longleftarrow 0$.

end

About the implementation overhead, our algorithm only requires the modification of the receiving end and does not require the cooperation of sender. There is some processing overhead associated with this method, such as the tracking of data packet arriving interval, setting up the timer, as well as condition judgement. But the trade-off is beneficial in comparison with the general increase in useful throughput and better utilization of channel, that we will show later.

2. Validation

To verify the correctness of our delay ACK implementation, we first performed a simple NS2 simulation with one sender streaming a FTP flow to one receiver. We plotted the packet sequence number vs. time as Fig. 16 shows. We observed that upon receiving every four data packets one ACK packet was sent, which is exactly as we expected.

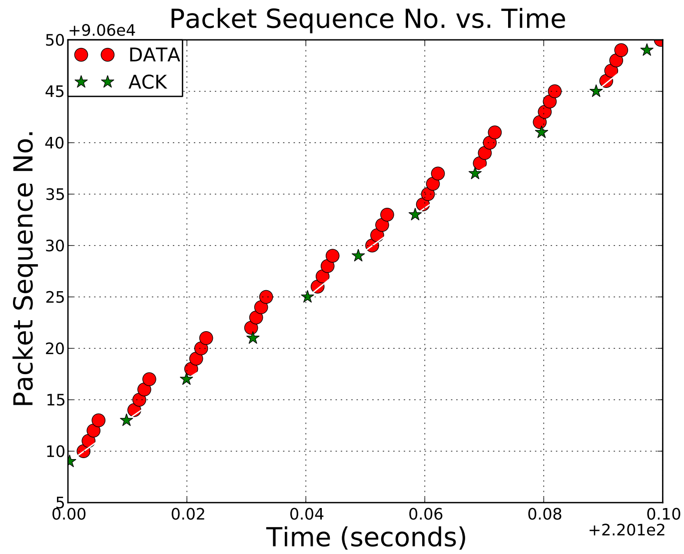
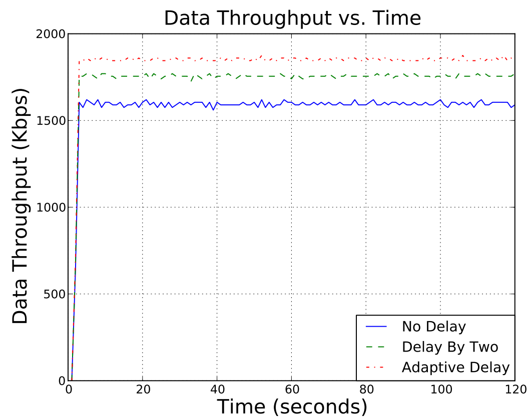
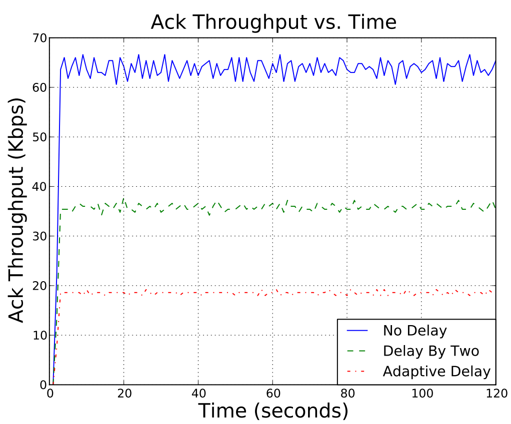


Figure 16: Packet sequence number vs. time after applying delayed ACK

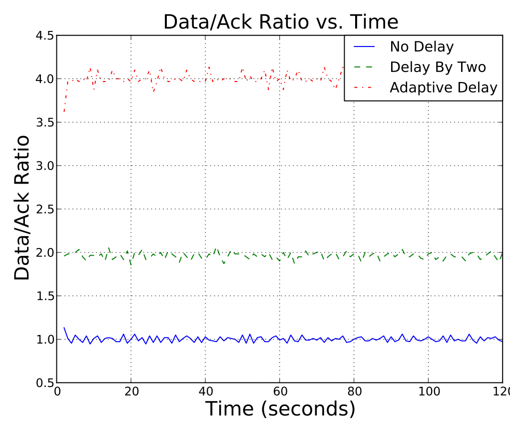
Furthermore, we validate the effectiveness of our implementation of Delayed ACK. We performed experiment with three kinds of TCP sinks: No Delay, Delay by two, Adaptive Delay. The No Delay means TCP sends ACKs back without any delay. The Delay By Two is the default implementation of accumulating ACK by two in NS2 code. Adaptive Delay is our implementation of adaptive ACK coalescing of up to 4 data packets. As Fig. 17a shows, Adaptive Delay has the highest data throughput, No Delay has the lowest, and Delay By two is in the middle. Meanwhile, Fig. 17b shows the amount of ACK traffic of the three mechanisms separately. And Adaptive Delay has significantly reduces the ACK traffic, in comparison with Delay By Two and No Delay. These two figures confirm our expectation of reducing ACK traffic while improving Data traffic effectiveness of ACK coalescing. Finally, we plotted out the Data/ACK ratio vs. time in Fig. 17c, and found the Data/ACK ratio for Adaptive Delay is nearly 4.0, for Delay By Two is 2.0, and for No Delay is 1.0. This



(a) data throughput



(b) ack throughput



(c) data per ack ratio

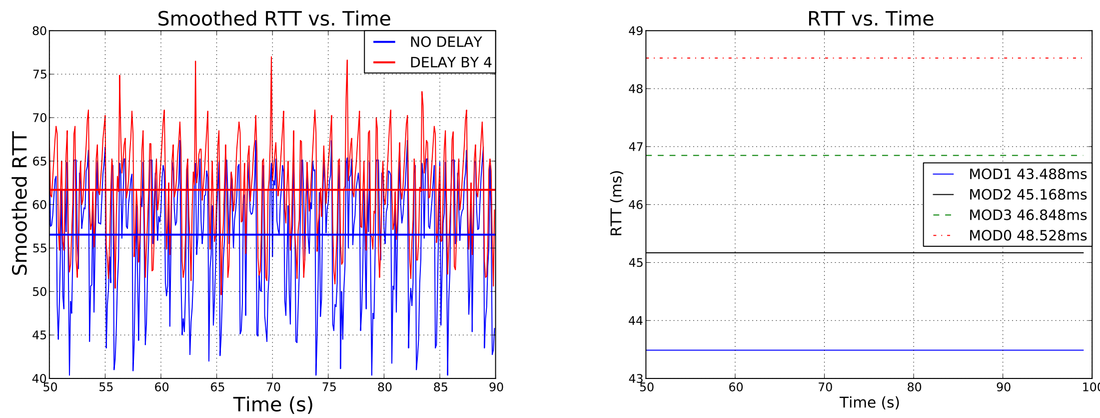
Figure 17: Delayed ACK's effectiveness

exactly matches our design objective.

B. RTT Measurement Issue

While ACK coalescing at the receiver is beneficial for increasing throughput and reducing the interrupts at the sender, ACK coalescing can cause problems for RTT estimation at the sender. ACK coalescing causes bursts of packets to be sent out by the sender. This increased burstiness results in self-induced congestion which causes sender's RTT estimation to become incorrect. That is the estimated RTT of Adaptive Delay is always higher than the actual value. As Fig. 18a shows, the SRTT value of Adaptive Delay mechanism (DELAY BY 4) always have about 5ms more than that of No Delay. This mistaken estimation of RTT is potential harmful to PERT, since PERT measures smoothed RTT to detect network congestion. The higher SRTT can be misleading so that PERT may generate the false alarm signal of 'network congestion' and reduce the congestion window. It is obvious this may lead to inferior transmission performance.

To get rid of this misguiding effects of RTT measurement error, we need to know the source of the error. Looking the Fig. 16 again, we observed burst of data packets in four, with an ACK for every fourth packet. The problem is the fourth packet may 'see' the queuing delay of the first three packets. In other words, the sampled RTT values for each packet in the packet train is different. The later sent packet may observe more queuing delay because of the previous sent packets (this is self-induced delay). To confirm this reasoning, we plot the RTT of packets with different groups of sequence numbers in Fig. 18b. We simply divide RTT values of packets with different



(a) RTT estimation error

(b) RTT difference

Figure 18: Problem of RTT measurement

sequence number into four groups. We mod their packet sequence number with four and therefore have four groups: 0, 1, 2, 3. And as we can see in the Fig. 18b, different groups of packets have different measured RTT values. And the difference between values of different groups is the same. It is clear that if we only sample the RTT with the packets of group # 0, we get higher RTT estimation.

Based on the observation and analysis of this RTT measurement issue. As showed in Fig. 19, we understand that $measured\_rtt = actual\_rtt + self\_induced\_delay + cross\_induced\_delay$. Considering the fundamental of PERT's congestion detecting mechanism, we simply expect to get rid of this self-induced delay and leave the cross traffic delay alone. In this case, PERT can still detect and responses early to the congestion that results from cross traffic, but won't be affected by the self induced delay, which is being caused by burstiness resulting from ACK coalescing. There are two possible solutions to correct this problem. The first approach is to estimate the

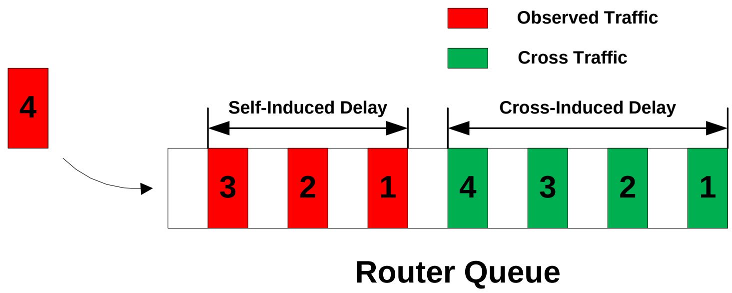


Figure 19: Analysis on bursty data and RTT error

delay from self-induced congestion and appropriately correct the RTT measurements. The second approach is to possibly eliminate the self-induced congestion which causes the mis-estimation of RTT . We have explored both two ways and will describe them in the following two sections.

C. TCP Pacing

Our implementation of TCP pacing in NS2 is based on the ideas in [18]. Our implementation keeps track of congestion window in terms of number of packets. We calculate the packet sending interval, which equals to RTT divided by the congestion window size. We set-up a timer for each data packet with the timeout value that equals to the packet sending interval calcaulated before.

We ran simulation to verify the correctness of our implementation of TCP pacing. Fig. 20 shows the packet sequence number increasing curve of three kinds of TCP Sink: No Delay, Delay ACK, Delay Pacing. As we can observe, the data packets

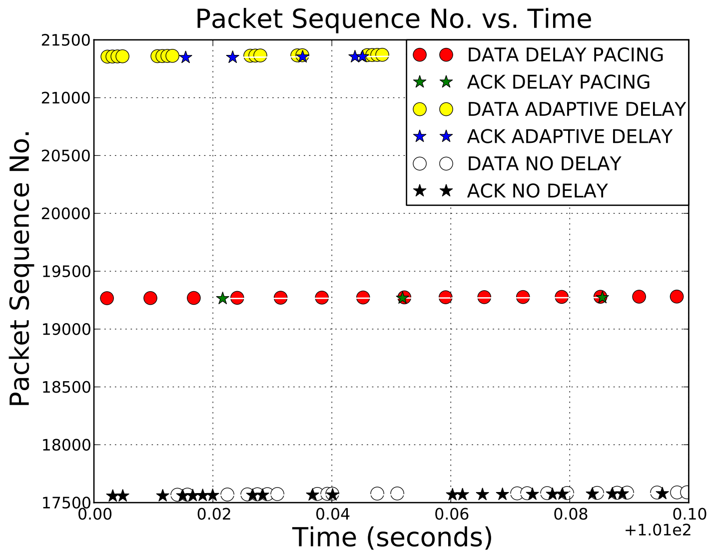


Figure 20: Packet sequence number of TCP Pacing

along with ACKs distributed almost uniformly as the transmission time passes when pacing is applied. In comparison, Delay ACK experiences severe data burstiness as expected. It is also worth noticing that No Delay's sending pattern is also slightly bursty. This is consistent with the observation that TCP congestion control will also incur some burstiness, described in paper [18]. Hence, according to our experiment result, TCP pacing successfully help in reducing the burstiness of data.

Fig. 21 shows the results of RTT estimation in various scenarios. We considered several experiments with different TCP sinks, with and without pacing on the sender side. It is observed that in all cases, the RTT estimation is accurate when TCP pacing is employed by the sender. In particular, it is observed that RTT overestimation is no longer a problem since TCP pacing avoids the self-induced congestion with ACK coalescing.

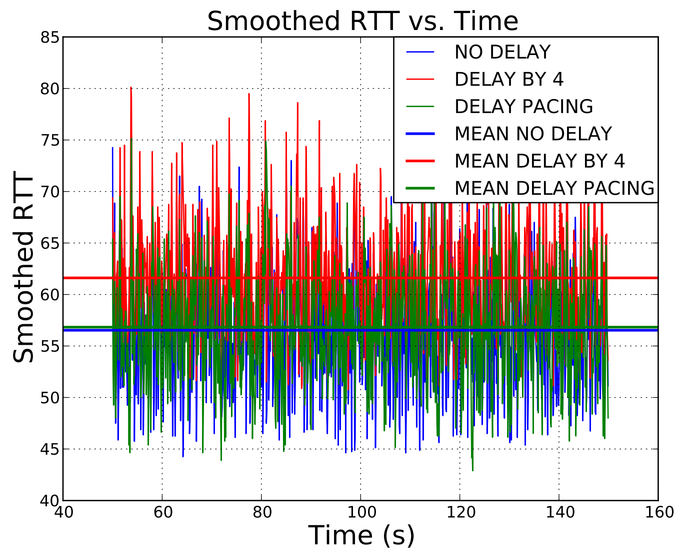
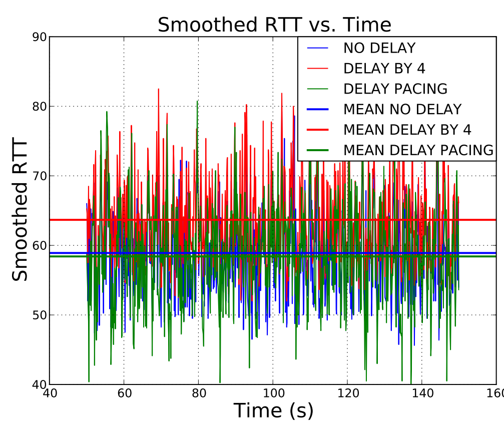
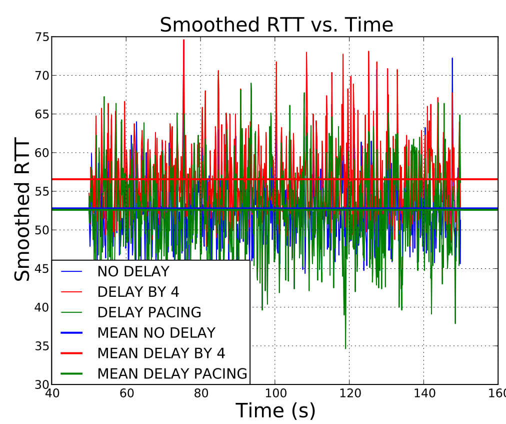


Figure 21: TCP Pacing correctly estimates RTT with ACK Coalescing

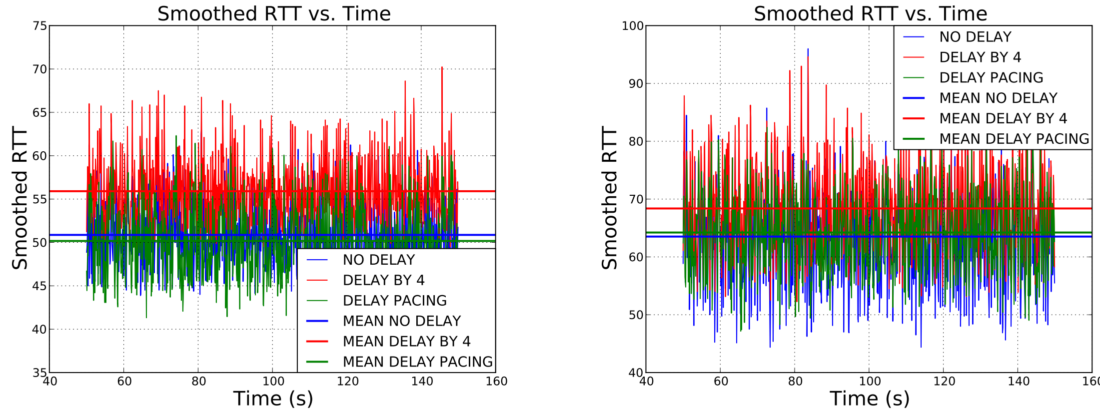


(a) 5 cross streams



(b) 10 cross streams

Figure 22: RTT estimation with pacing when cross traffic number varies



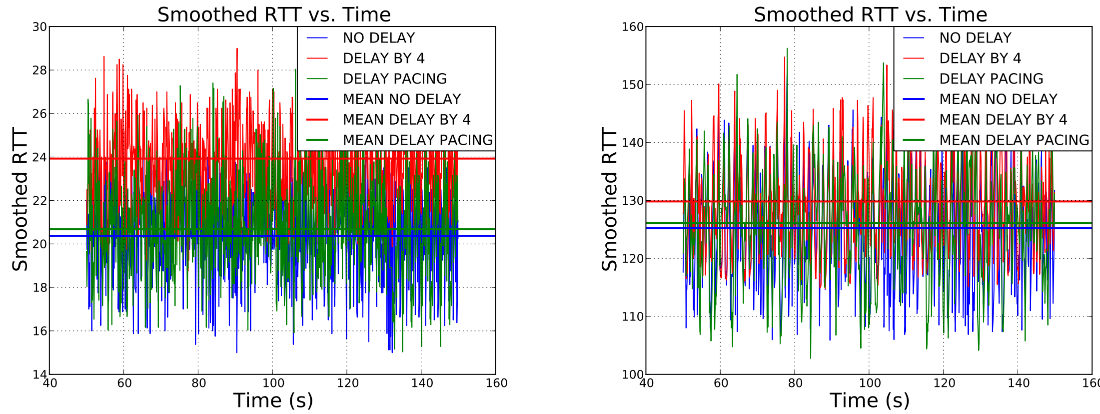
(a) packet size 500 bytes

(b) packet size 1500 bytes

Figure 23: RTT estimation with pacing when cross traffic packet size varies

To confirm that TCP Pacing works in various network settings, we also perform experiments with varied cross traffic streams numbers (from 1 to 10), cross traffic packet sizes (from 500 - 1500 bytes), as well as end-to-end delay (from 5 - 50ms). As Figs. 22, 23, 24 show, the RTT estimation error disappears after applying TCP Pacing for most of scenarios.

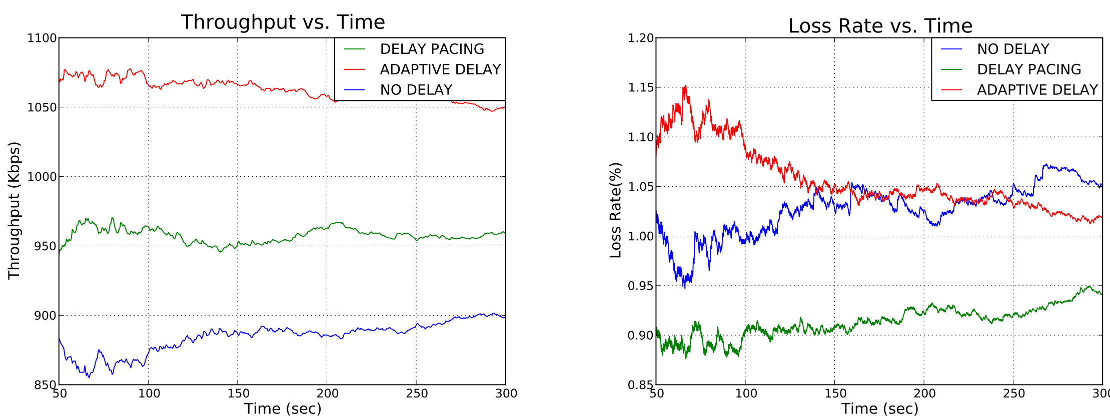
These results show that RTT can be correctly estimated when pacing is employed at the sender when ack coalescing is employed at the receiver. We combined pacing and ack coalescing in PERT to study its performance. In these experiments we study the performance of three different flavors of PERT. The first version employed both pacing and ack coalescing, the second one employed ack coalescing without pacing and third flavor employed neither mechanism. The results are displayed as Figs. 25. Fig. 25a shows that the three different flavors of PERT nearly achieve the same bandwidth. This shows that ack coalescing and pacing keep PERT fair to other flavors of PERT.



(a) RTT 10 ms

(b) RTT 100 ms

Figure 24: RTT estimation with pacing when end-to-end delay varies

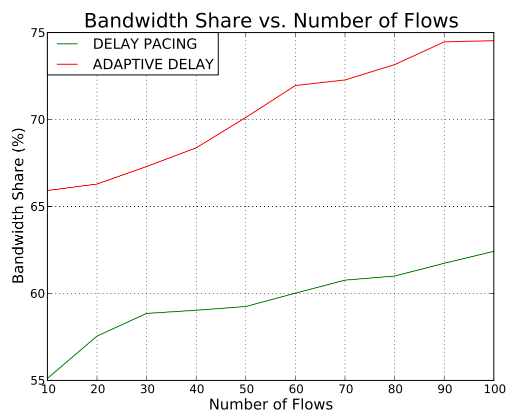


(a) bandwidth share

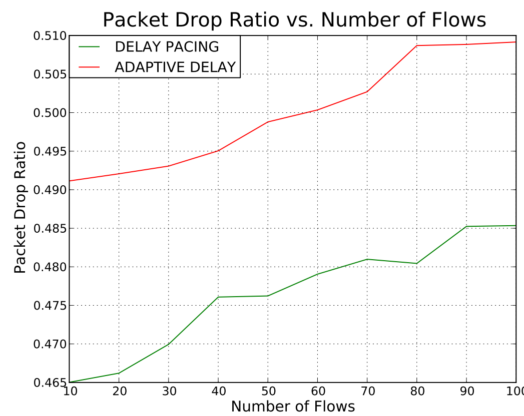
(b) packet loss

Figure 25: Benefit of delayed ACK and pacing

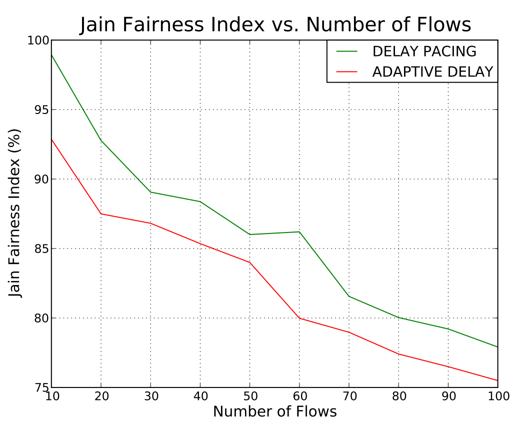
Fig. 25b shows that PERT with pacing and ACK coalescing has lower packet loss rate than the other two flavors of PERT. Pacing helps the data sending pattern to become less bursty and this allows packets to experience a smaller loss rate at the buffers. These results show that PERT with pacing and ACK coalescing can attain fair bandwidth at a lower loss rate. We explore the performance of PERT with these two mechanisms further in different network settings.



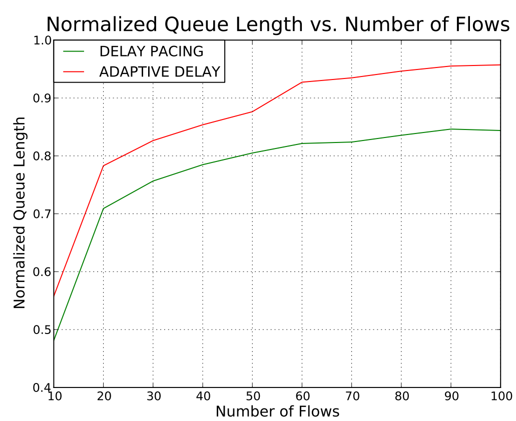
(a) bandwidth share



(b) packet drop ratio



(c) Jain fairness index

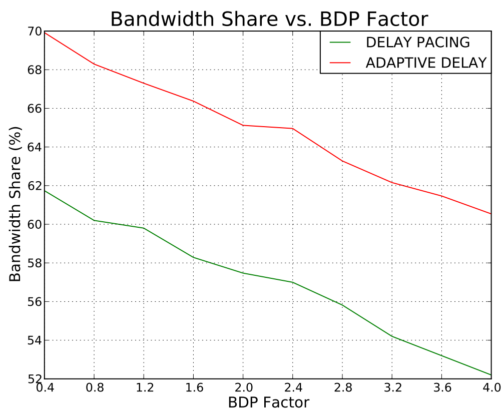


(d) normalized queue length

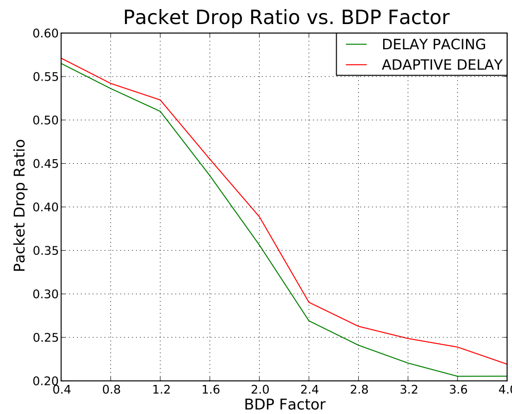
Figure 26: Fairness of delayed ACK and pacing when number of both pacing and non-pacing (50%-50%) flows increases

Next, we further explore the fairness and potential benefits brought by Delay ACK. First we change the number of total flows from 10 to 100 with Delay Pacing and Delay ACK in proportion of 50% and 50%, and plot four kinds of important fairness metrics in Fig. 26. Our results show that as the flow number increases, the fairness of both Delay Pacing and Delay ACK decreases in the aspects of bandwidth share, packets drop ratio, Jain fairness index as well as normalized queue length. Delay Pacing always show better fairness than Delay ACK (without TCP pacing). TCP pacing helps to improve the fairness by avoiding congestion on the router.

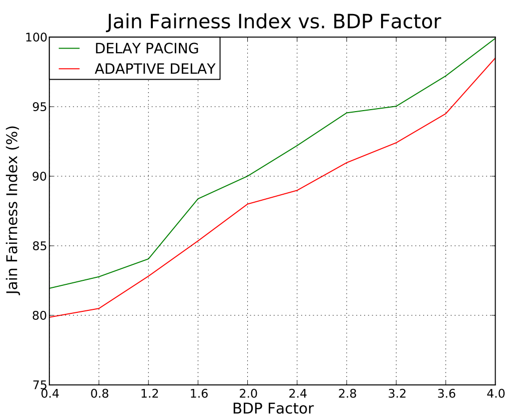
To confirm our observation on the improved fairness of Delay Pacing, we change the percentage of Delay ACK flows (from 10% to 100%), BDP (Bandwidth Delay Product) Factor (from 0.4 to 4.0), Round Trip Time (from 10 to 100ms), as well as number of cross traffic (HTTP) flows (from 100 to 1000), as Figs. 27, 28 show. Then we are safe to claim in almost cases, Delay Pacing has better fairness than Delay ACK (without pacing). This is to say, in most cases Delay Pacing has better performance when multiple network flows are competing for limited bottleneck bandwidth.



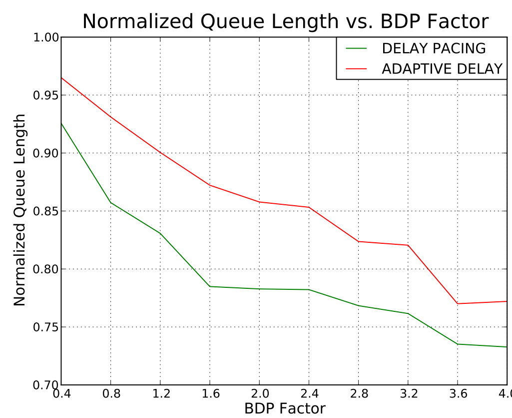
(a) bandwidth share



(b) packet drop ratio

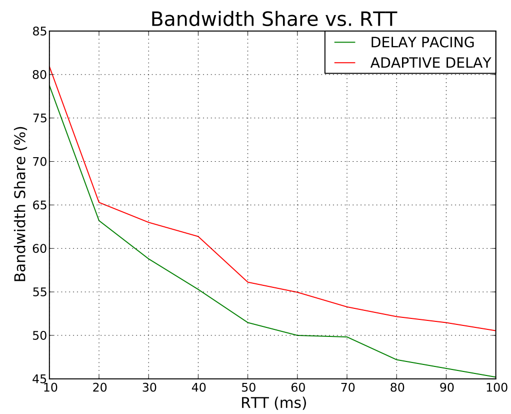


(c) Jain fairness index

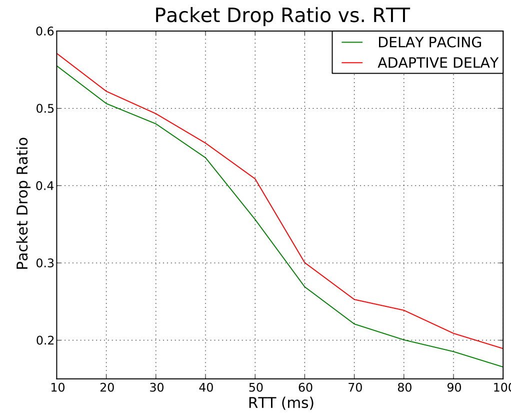


(d) normalized queue length

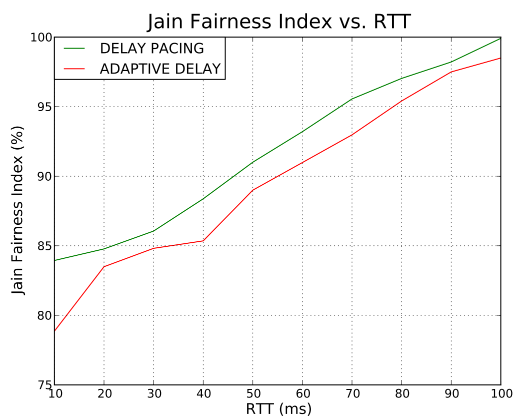
Figure 27: Fairness of delayed ACK and pacing when Bandwidth Delay Product (BDP) factor increases



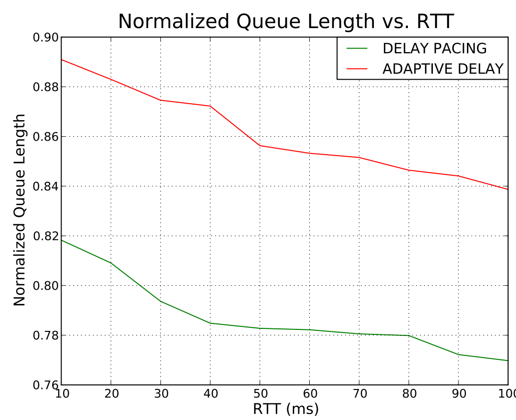
(a) bandwidth share



(b) packet drop ratio



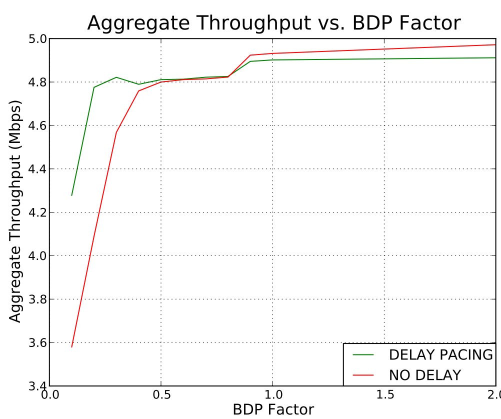
(c) Jain fairness index



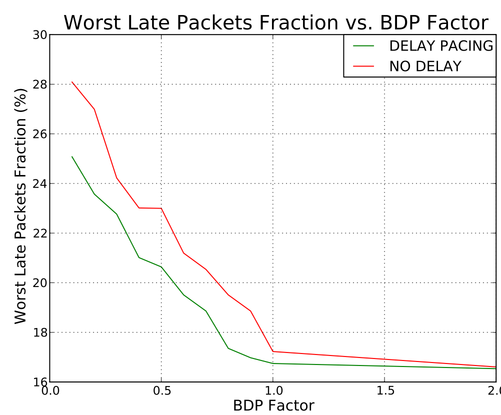
(d) normalized queue length

Figure 28: Fairness of delayed ACK and pacing when the Round Trip Time (RTT) increases

Finally, we look into the performance of PERT with Delay Pacing for video streaming, in multiple flows environment. We take two performance metrics similar as described in paper [19]: aggregate throughput - the total throughput achieved by all flows when they work together, and late packet arrival rate of the worst flow. From Fig. 29a, we oberve that Delay Pacing achieve higher aggregate throughput when BDP is smaller than 1.0 in comparison with No Delay. And from Fig. 29b, we notice that Delay Pacing has worst-case late packet arrival rate than No Delay. In other words, Delay Pacing helps to better the overall performance for video streaming when multiple flows are competing for bandwidth at a network link. And along with better QOS, PERT flows achieved better fairness when pacing and ACK coalescing are employed.



(a) aggregate throughput



(b) packet late rate of worst flow

Figure 29: Video streaming performance of delayed ACK and pacing

CHAPTER V

ESTIMATING THE RTT ERROR DUE TO SELF-INDUCED CONGESTION

A. Approach

We consider a second solution for RTT estimation error brought by burstiness of data. [1]. The basic idea is to estimate the self-induced delay and subtract it from the observed RTT value, in order to obtain a correct RTT estimate.

Before we can describe describe our approach, we introduce some notation. First we model the burst of data as a packet train, and define the average time to finish a packet transmission in a packet train as average packet dispersion. Let D denote the average packet dispersion and k denote the number of packets in a packet train. We draw the timeline of the transmission of a packet train with packet sequence number from n to $n+k-1$, as Fig. 30 shows. We define T_s as the sending time of data packets, and T_r as the receiving time of ACK packets. Then we can see that T_{s_n} is the sending time of (more precisely the first packet in) our packet train, and $T_{r_{n+k-1}}$ is the receiving time of the ACK of our packet train. Hence we have the total delivery time of our packet train:

$$R = T_{r_{n+k-1}} - T_{s_n} \tag{5.1}$$

Let Δ_2 denote the difference between the receiving time of ACK for current and last

[1]joint work with Ankit Singh

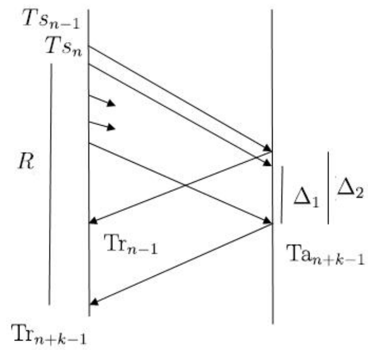


Figure 30: RTT measurement model

packet train. And let Δ_1 denote the self-induced delay that we try to estimate. We have the following equations:

$$\begin{cases} E[\Delta_2] & = E[T_{r_{n+k-1}} - T_{r_{n-1}}] \\ & = T_{s_n} - T_{s_{n-1}} + \Delta_1 + D \\ E[\Delta_1] & = (k-1)D \end{cases} \tag{5.2}$$

Here the estimated value of Δ_2 equals to the difference between sending time of first packet of current packet train T_{s_n} and that of last packet of last packet train $T_{s_{n-1}}$ plus self-induced delay Δ_1 and average packet dispersion D. Moreover, the estimated value of Δ_1 equals to $k-1$ multiplied by average dispersion D. Here T_{s_n}, $T_{s_{n-1}}$, k and Δ_2 can be measured, while D is unknown. So we combine the equations and get the estimated self-induced delay Δ_1 as follows:

$$E[\Delta_1] = \frac{k-1}{k}[\Delta_2 - (T_{s_n} - T_{s_{n-1}})] \tag{5.3}$$

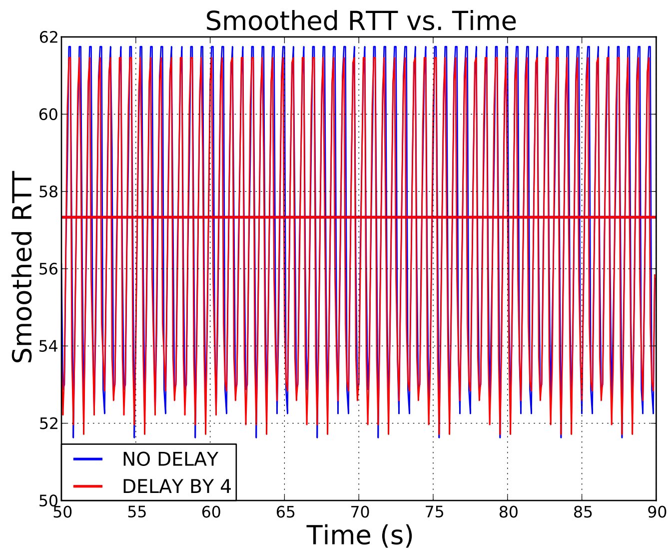
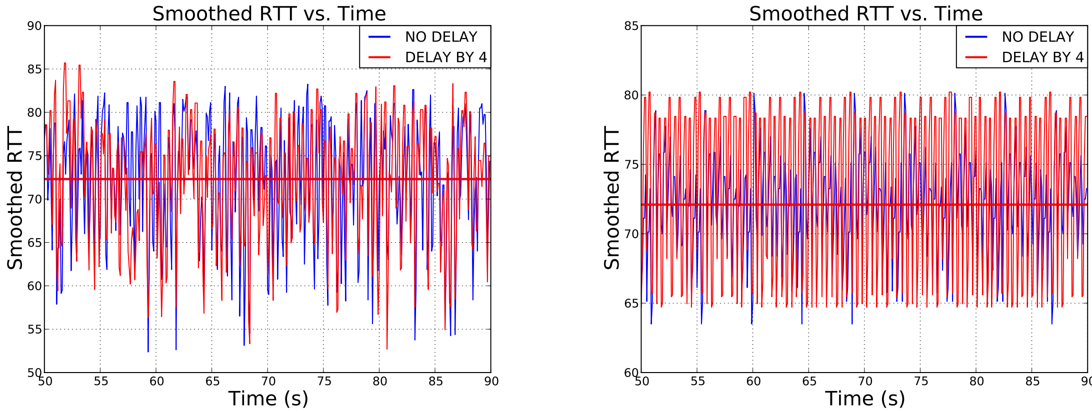


Figure 31: Fixed RTT measurement

After we get the estimated value of the self-induced delay Δ_1, we subtracted it from the total delivery time R calculated in Eq. (5.1), to get the estimated value of round-trip time RTT, as the following equation shows:

$$E[RTT] = R - E[\Delta_1] \tag{5.4}$$

Our approach relies on this analysis and tries to correct the error in RTT estimation as shown in equation (5.4).



(a) 5 cross streams

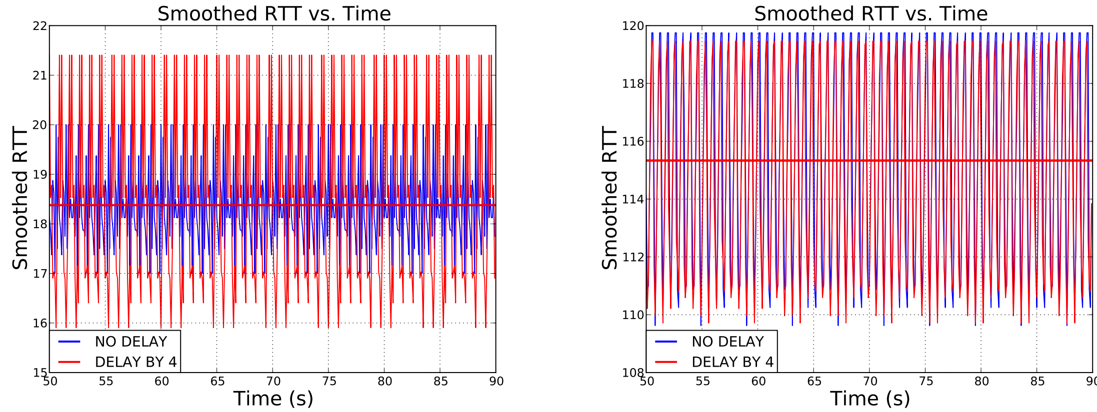
(b) 10 cross streams

Figure 32: Fixed RTT measurement when cross traffic streams number varies

B. Results

We modified the sender side PERT based on the analysis above to correctly estimate the RTT, we implement it to fix the RTT measurement for Delay ACK mechanism on the sending end. We first simply validate the correctness of the implementation with two flows of PERT. with Delay ACK and the other flow with No Delay. The end-to-end delay is set to 20ms, the bottleneck link bandwidth is set to 3 Mbps, and the packet size is 1000 bytes. Fig. 31 shows the smoothed RTT values for both Delay ACK and No Delay. We can see both of them have same mean value, and they have very small difference. This is to say, our model and method of estimating and deducing self-induced delay is correct.

To confirm that this method works in other scenarios, we change bottleneck bandwidth (from 1 to 10 Mbps), end-to-end delay (from 5ms - 50ms), number of cross

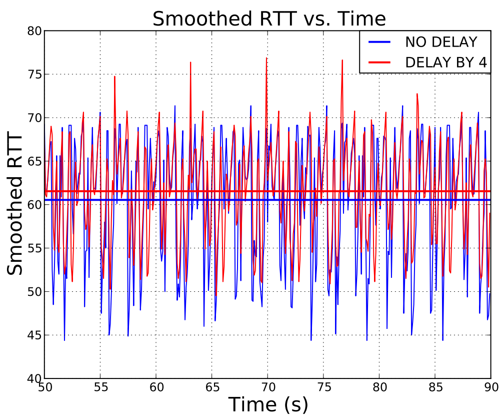


(a) RTT 10 ms

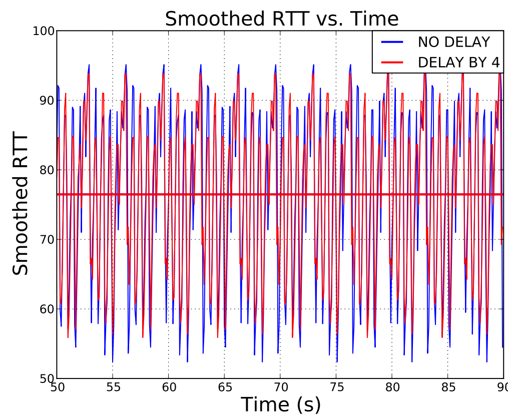
(b) RTT 100 ms

Figure 33: Fixed RTT measurement when end-to-end delay varies

traffic flows (from 5 to 10) and cross traffic packet size (from 500 to 1500). In Figs. 32, 33, 34, we observe that the mean value of Delay ACK and that of No Delay are nearly the same in all these experiments. These results show that the proposed method for correcting the error from self-induced delay works in various network settings.



(a) packet size 500 bytes



(b) packet size 1500 bytes

Figure 34: Fixed RTT measurement when cross traffic packet size varies

CHAPTER VI

CONCLUSION AND FUTURE WORK

In this thesis, we have demonstrated the performance of PERT for on-demand video streaming. Our NS-2 simulation experiments are performed in a heterogeneous environment, where the background traffic is delivered by RENO. PERT outperforms RENO and CUBIC in successfully delivering video in constrained streaming scenarios we considered here. Moreover, the real-life video streaming test confirms PERT's ability to improve video playback quality when comparing to RENO and CUBIC.

Also, we extended PERT with to work with ACK coalescing to improve PERT's performance in high-speed and wireless networks. We identified that ACK coalescing causes data to be sent in bursts or packet trains and these bursts result in overestimating RTT due to self-induced congestion. We proposed two solutions to correctly estimate RTT in the presence of ACK coalescing. The first method employed pacing at the sender side and the second method estimated the error caused by the self-induced congestion and appropriately corrected RTT estimations. We showed through experiments that both these techniques correctly estimate RTT and allowed PERT to function well with ACK coalescing. In addition, pacing is shown to improve the observed loss rate and hence delivered video performance. These enhancements allow PERT's suitability to a wider set of network scenarios.

In the future, we will implement PERT with Delay ACK and Pacing in Linux kernel and measure its processing overhead. The processing overhead is especially important in high speed networks. ACK coalescing is expected to relieve the sender by reducing the number of interrupts processed per Mbyte. However pacing requires timers which results in higher overheads. We will study if the resulting overall overhead is lower than normal PERT that doesn't employ both these mechanisms. Further, our efforts will be devoted to carry out more evaluations on PERT with Delay ACK and Pacing, in comparison against other protocols, especially for video streaming.

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VITA

Bin Qian received his B.E degree in automation from the Zhejiang University (ZJU), Hangzhou, China in 2007 and his M.S. in computer enginnering from Texas A&M University, College Station, Texas, in 2011. His research interests are in the areas of networking congestion protocols and multimeida networking. He has been a recipient of the Scholarship for Merits Students from the ZJU, Hangzhou, China during 2003-2006 and Industry Huawei Scholarship sponsored by Huawei Inc., China in 2008. Prior to arriving at Texas A&M University, he studied in the postgradaute program in the University of Science and Technology of China during 2007-2009. He can be contacted at the following address : Department of Electrical and Computer Engineering, Texas A&M University, 331B WERC, College Station, TX 77843-3118.

The typist for this thesis was Bin Qian.