The Delay-Friendliness of TCP

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ABSTRACT

TCP has traditionally been considered unfriendly for real-time applications. Nonetheless, popular applications such as Skype use TCP since UDP packets cannot pass through many NATs and firewalls. Motivated by this observation, we study the delay performance of TCP for real-time media flows. We develop an analytical performance model for the delay of TCP. We use extensive experiments to validate the model and to evaluate the impact of various TCP mechanisms on its delay performance. Based on our results, we derive the working region for VoIP and live video streaming applications and provide guidelines for delay-friendly TCP settings. Our research indicates that simple application-level schemes, such as packet splitting and parallel connections, can reduce the delay of real-time TCP flows by as much as 30% and 90%, respectively.

Categories and Subject Descriptors

C.4 [**Performance of Systems**]: Modeling techniques, Measurement techniques

General Terms

Performance, Measurement

Keywords

Performance modeling, VoIP, live video streaming, TCP congestion control

This material was supported in part by the National Science Foundation under Grants No. CNS-0626795, CCR-0615126 and CNS-0202063, and by FirstHand Technologies. Any opinions, findings, and conclusions or recommendations expressed in this material are those of the authors and do not necessarily reflect the views of the National Science Foundation or FirstHand Technologies.

SIGMETRICS'08, June 2–6, 2008, Annapolis, Maryland, USA.

1. INTRODUCTION

The popularity of real-time applications, such as VoIP and video streaming, has grown rapidly in recent years. The conventional wisdom is that TCP is inappropriate for such applications because its congestion controlled reliable delivery may lead to excessive end-to-end delays that violate the real-time requirements of these applications. This has led to the design of alternative unreliable transport protocols [17,20,30] that favor timely data delivery over reliability while still providing mechanisms for congestion control.

Despite the perceived shortcomings of TCP, it has been reported that more than 50% of the commercial streaming traffic is carried over TCP [16]. Popular media applications such as Skype [7] and Windows Media Services [16] use TCP due to the wide deployment of NATs and firewalls that often block UDP traffic. Further, TCP is by definition TCP-friendly [17] and is a mature and widely-tested protocol whose performance can be fine tuned.

The gap between the perceived shortcomings of TCP and its wide adoption in real-world implementations motivated us to investigate the delay performance of TCP. Our study seeks to address the following questions: (1) Under what conditions can TCP satisfy the delay requirements of real-time applications? (2) Can the performance of these applications be enhanced using simple application-layer techniques? We address these questions in the context of two real-time media applications that are characterized by timely and continuous data delivery: VoIP and live video streaming.

To understand all aspects of the performance of real-time applications, we conduct an extensive performance study using an analytical model and real-world experiments. The analytical model allows us to systematically explore the delay performance over a wide range of parameter settings, a challenging process when relying on experimentation alone. While there exists an extensive literature on TCP modeling and analysis, it is geared towards file transfers [10, 28, 29] and video streaming [19, 33] rather than delay.

We use both test-bed and Internet experiments to validate the model over a wide range of network environments. We analyze how the delay depends on the congestion control and reliable delivery mechanisms of TCP. We further study the impact of recent extensions such as window validation [18] and limited transmit [4]. The results obtained yield guidelines for delay-friendly TCP settings and may further be used to compare the performance of TCP with alternative protocols [17, 20] and experimental real-time enhancements for TCP [15, 22, 25]. We analyze two application-level schemes, namely, packet splitting and parallel connections

which we find to significantly reduce the delay of live video streaming flows.

Our research reveals that real-time application performance over TCP may not be as delay-unfriendly as is commonly believed. One reason is that the congestion control mechanism used by TCP regulates rate as a function of the number of packets sent by the application. Such a packet-based congestion control mechanism results in a significant performance bias *in favor* of flows with small packet sizes, such as VoIP. Second, due to implementation artifacts, the average congestion window size can overestimate the actual load of a rate-limited flow. This overestimation reduces the likelihood of timeouts and consequently the resulting TCP delay.

The main contributions of this paper are:

- To the best of our knowledge, we are the first to present a discrete-time Markov model for the delay distribution of a real-time TCP flow (Section 4).
- We find that under the same network conditions, VoIP flows suffer from lower TCP delays than live video streaming flows. We derive the working region for VoIP and live streaming flows based on our model and experiments (Section 6.1). VoIP operates well when the network loss rate is at most 2% and RTT is at most 100 ms. Live video streaming operates well when the network loss rate is at most 3% and RTT is 100 ms.
- We study the impact of various TCP mechanisms on the TCP delay (Sections 6.2–6.4). We then provide TCP-level guidelines (Section 6.5) and simple application-level heuristics (Section 7) for improving the performance of real-time applications. The most promising heuristic uses parallel connections with shortest-queue first policy and achieves up to 90% delay reduction.

2. APPLICATION SETTING

We study a general real-time media application, with a Constant Bit Rate (CBR) source, that sends data across the network using TCP. CBR is the most basic and dominant encoding for media flows in the Internet [34]. Although our analysis is general, we focus on CBR sources corresponding to VoIP and live video streaming, as detailed in Section 5. The VoIP and live video streaming flows are application-limited, i.e., their sending rate is a function of media encoding and not the underlying network. This is in contrast to greedy flows, such as FTP, which are network-limited.

Throughout the paper, we refer to the transmission unit of TCP as a segment and to the TCP payload (i.e., the application-layer data unit) as a packet. The maximum segment size, MSS, is determined by the maximum transmission unit of the network path [31]. A common characteristic of real-time applications is their sensitivity to end-to-end delay which may vary from application to application. For live video streaming, there is usually minimal interactivity involved, so the application can afford a startup delay in the order of seconds [16]. For VoIP, low delay of up to 400 ms is required in order to maintain acceptable interactivity [15]. To reduce end-to-end delays, VoIP often uses small payloads (e.g., 160-byte packets) that correspond to 20 ms or 30 ms of audio. Thus, in the context of this paper, the difference between VoIP and live video streaming flows is their packet sizes and their tolerance of delay.

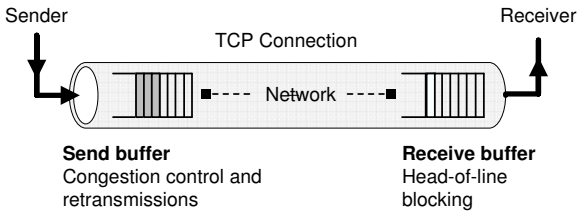

Figure 1: Transport-layer queueing delays

We define TCP delay as the time it takes the application to get a packet from source to destination through a TCP connection. We use the TCP delay distribution to evaluate the performance of real-time applications. From the delay distribution we derive the application-level packet loss rate which is the portion of packets that arrive beyond their playback time. This metric is closely correlated with user-perceived video and audio quality [12,32], and hence is used as an approximate performance measure. The application-level packet loss metric is determined by the α-percentile delay bound, defined as follows. A delay value d of α-percentile corresponds to $1-\alpha$ portion of packets that are delayed more than d time units.

3. TCP DELAY COMPONENTS

Here we examine the various ways in which delay is introduced in a TCP connection with a CBR source. The delay in a TCP connection consists of two main components: (a) network delay, which is the time it takes a segment to get across the network; (b) TCP-level delay, which is an artifact of how TCP reacts to variations in the effective throughput. While throughput variations can occur due to application-level flow control, they are primarily the result of network congestion. To understand TCP-level delays, we briefly describe the transmission behavior of TCP. TCP is a window-based protocol that uses two main mechanisms to regulate its sending rate: Additive-increase-multiplicative-decrease (AIMD) and timeout. These mechanisms may delay data delivery because they require TCP to reduce its sending rate in response to network congestion. In addition, TCP uses packet retransmissions to provide lossless data delivery. This mechanism introduces additional delay for data delivery. A detailed discussion of the mechanisms in TCP can be found in [31].

TCP uses two buffers to provide congestion-controlled reliable data delivery; a send buffer and a receive buffer. The send buffer serves two functions [15]. It absorbs rate mismatches between the application sending rate and the transmission rate of TCP. It also stores a copy of the packets in transit in the network should they be retransmitted. Although these packets are buffered, they do not introduce additional queuing delay for unsent packets. Only the unsent packets held in the send buffer, hereafter referred to as the *backlogged* packets, contribute to the delay of newly admitted packets to the send buffer. The purpose of the receive buffer is to hold out-of-order packets while a loss is being recovered. This buffering results in head-of-line (HOL) blocking delay.

In this paper, we only consider packet backlogging due to network congestion and ignore packet backlogging due to other causes, such as application-level flow-control (e.g., a receiving application that slows down an aggressive sender [31]).

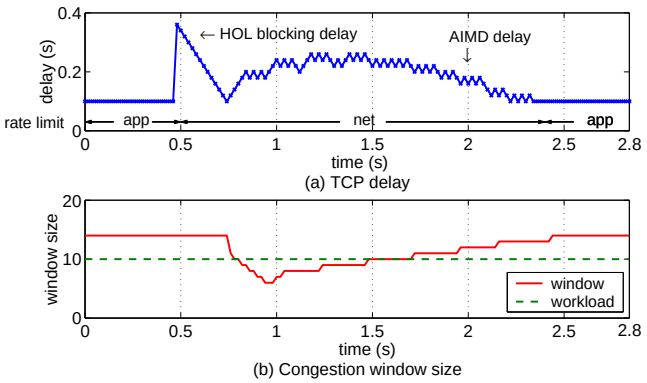

Figure 2: The evolution of the TCP delay and congestion window size for a video-like CBR source.

Applications usually minimize this backlogging by setting a large receive buffer and operating with non-blocking sockets. Packet backlogging can also occur due to Nagle's algorithm [27] that was added to TCP to limit the transmission of small segments. This algorithm ensures that TCP sends data only when there are at least MSS bytes of available data, and hence improves throughput at the expense of increased transmission delay. In practice, many delay-sensitive applications disable this algorithm to reduce transmission delays [37]. We follow this practice in our work. Figure 1 illustrates the TCP-level delay components. The sender-side delay is caused by the congestion control and reliable delivery mechanisms in TCP, whereas the receiver-side delay is caused by the in-order delivery guarantee of TCP.

Figure 2(a) illustrates the delay behavior of a TCP flow driven by a CBR source. The CBR source sends 50 MSS-sized packets per second over a symmetric network with a 200 ms round-trip time (RTT). An application-limited period is seen from 0 s to 0.5 s and from 2.4 s to 2.8 s. We define an application-limited period as a period where the TCP throughput satisfies the source's rate requirement. In this period, the TCP delay is determined by the network delay. A network-limited period is seen from 0.5 s to 2.4 s. In this period, the TCP throughput no longer satisfies the source's rate requirement, resulting in TCP-level delays. TCP moves to a network-limited period when a packet loss occurs. Within the network-limited period there are two subregions: loss recovery, seen from 0.5 s to 0.76 s, and packet backlogging, seen from 0.76 s to 2.4 s. TCP uses retransmission to recover the lost packet, which in turn causes head-of-line blocking delay at the receiver. The receipt of a packet loss indication at time 0.76 s triggers TCP to reduce its congestion window size, resulting in packet backlogging.

Unlike application-limited periods, in network-limited periods TCP probes for additional bandwidth to satisfy the source's rate requirement. In our example, the transmission rate of TCP is governed by the AIMD mechanism and hence is linearly increasing, as seen in Figure 2(b). The mismatch between the input and output rates at the TCP sender results in the quadratic-like delay curve seen in Figure 2(a). TCP moves back to an application-limited period when the rates are matched.

3.1 TCP Interaction with VoIP-like Flows

The performance of real-time applications that use small

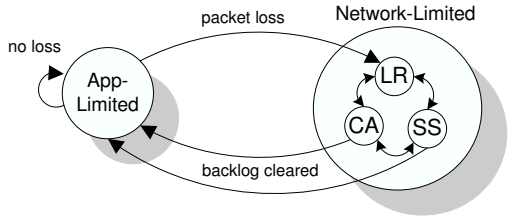

Figure 3: A high-level view of a model for a TCP connection with a CBR source.

packets (e.g., VoIP) is directly affected by whether the congestion control mechanism in TCP is byte or packet-based. According to [6], there are two different issues at work. First, a TCP sender can track the congestion control state in terms of outstanding bytes or outstanding packets. Second, a TCP sender can update the congestion control state based on how many bytes are acknowledged, a mechanism known as byte-counting, or by some constant for each ACK arrival, a mechanism known as ACK-counting. We compare the performance of ACK and byte-counting mechanisms (Section 6.3) and focus on the latter due to its wide deployment [23], as also verified by our measurements.

4. MODELING TCP DELAY

Our model builds upon the detailed TCP model in [35] that predicts the performance of TCP from the viewpoint of throughput. We extend this model in three ways. First, we include the TCP buffer dynamics in order to predict the delay performance of TCP. Second, we model the window behavior during application-limited periods [18] to accurately capture the loss recovery latency of TCP. Third, we capture the effect of window inflation [6] and the limited transmit mechanism [4] to improve the accuracy of the model for small congestion windows. We assume that the sender is using a NewReno TCP implementation, the predominant TCP variant in the Internet [23], and refer the interested reader to [13, 31] for a detailed description of TCP NewReno's mechanisms.

4.1 A TCP Model

We consider a CBR source that sends fixed-size packets at regular intervals across the network using TCP. Throughout the paper, we assume that the average throughput provided by TCP satisfies the rate requirement of the CBR source. However, transient congestion episodes in the network can still lead to TCP throughput fluctuations and hence to TCP-level delays. These episodes cause the TCP connection to alternate between application-limited and network-limited periods, as described in Section 3.

Mimicking the behavior of a real-world TCP flow, our model consists of two main states: application-limited and network-limited. It transitions from an application-limited state to a network-limited state when a loss occurs. TCP-level delays are introduced only during network-limited states. The model transitions back to an application-limited state when the TCP sender matches its input and output rates (e.g., when packet backlog is cleared). While in a network-limited state, the model moves among four states corresponding to TCP's congestion control phases: slow-start (SS), congestion avoidance (CA), fast recovery (FR), and

Notation	Definition
f	load in packets per second
r	load in packets per round-trip time
a	packet size (bytes)
w	congestion window size (in segments)
b	backlog size (bytes)
l	indicates whether loss recovery is required
p	segment loss probability
L	one-way network delay (seconds)
MSS	maximum segment size (bytes)

Table 1: Summary of model notations

retransmission timeouts (TO). A high-level view of a model for a TCP connection with a CBR source is shown in Figure 3; for ease of presentation, we merged the timeout and fast recovery states into a single loss recovery (LR) state.

We make several simplifying assumptions in our model, as follows. First, we assume that TCP increases the congestion window by one *packet* per round-trip time, an assumption motivated by the wide-deployment of ACK-counting TCP implementations [23]. Second, we assume that the TCP implementation does not increase the congestion window when the TCP sender is application-limited, which is the behavior observed for Linux and Windows XP systems (see Section 6.4). Third, we assume that the slow start threshold is statically set to half of the source's sending rate in packets per round-trip time. From our experience, using a static slow-start threshold rather than a dynamic one has a marginal impact on the model's prediction accuracy. Last, we do not model the effect of delayed acknowledgements (ACKs). Nonetheless, our model can be easily extended to support delayed ACKs using a similar approach as in [29].

The CBR source is characterized by two parameters, the data generation rate in packets per second f and the size of a generated packet a. We let r denote the data generation rate in packets per round-trip time. For convenience, we summarized the notations used in this paper in Table 1. We model the behavior of a TCP source by a discrete-time Markov chain with a finite state space $S = \{(w, b, l)\}$ and a probability transition matrix $Q = [q_{s;s'}]$, $s, s' \in S$. Each state is associated with at most three outgoing transitions representing the following events: the receipt of a fast retransmit loss indication, the receipt of a timeout loss indication, and successful delivery of window data. Each transition is associated with a certain number of packet transmissions, and each packet in this transmission is associated with a delay.

In our model, each state is represented by an ordered triple (w, b, l), where w is the current congestion window size in segments, b is the current backlog size in bytes, and l indicates whether a loss has been detected and needs to be recovered from ($l > 0$) or not ($l = 0$). The backlog size value is used to indicate whether the sender is application-limited ($r \leq w, b = 0$) or network-limited. The window size value is used to distinguish between the two loss recovery strategies employed by TCP: fast recovery ($w > 0, l = 1$) and retransmission timeout ($w = 0, l \geq 1$), where l indicates the current exponential back-off stage. Table 2 lists the rules for classifying an arbitrary state $s = (w, b, l)$ according to the congestion control phases of TCP. We use the notation $AL = \{(w, b, l) : r \leq w, b = 0, l = 0\}$ to denote the set of states for which the application-limited condition

Classification	Condition
AL (Application-limited)	$r \leq w, b = 0, l = 0$
NL (Network-limited)	$0 < w, b \neq 0$ or $w < r, b = 0$
CA (Congestion avoidance)	$r/2 < w, l = 0$
SS (Slow start)	$w \leq r/2, l = 0$
FR (Fast recovery)	$0 < w, l = 1$
TO (Timeout)	$w = 0, l = 1, \ldots, 6$

Table 2: State classification

holds. The notations CA, SS, FR, TO are defined in a similar way, as shown in Table 2.

4.2 A Delay Performance Model

In this section, we model the three ways in which TCP introduces delays: congestion control, retransmissions, and head-of-line blocking, as detailed in Section 3. We model the time packets are buffered at the sender (i.e., the congestion control delay) by scaling the backlog size by the sources's rate. This modeling approach can be explained by observing that the (unsent) buffered packets left behind after a packet is transmitted must have been admitted to the send buffer while the transmitted packet was buffered. Since we consider a data source with a constant rate, a transmitted packet that leaves behind a backlog of b bytes must have been buffered for at least $b/(fa)$, the backlog size divided by the source's rate in bytes per second. This approach introduces an error in the order of several packetization intervals. The error arises because the Markov chain captures the backlog size evolution in network-limited states at round-trip time granularity. This error can be reduced by keeping track of the inter-sending packet times. However, this will make the state space of the model prohibitively large and hence will limit its usefulness.

We determine the head-of-line and the retransmission delay by the loss recovery latency (i.e., the time it takes TCP to detect and recover a lost packet). TCP interprets receipt of three duplicate ACKs as an indication of a packet loss. It immediately retransmits the lost packet upon the receipt of the third duplicate ACK. Hence, we take the time required to receive a fast retransmit loss indication to be $RTT + 3/f$; the RTT term is the time needed for the first duplicate ACK feedback and the $3/f$ term is the maximum time to generate three duplicate ACKs, which is attained when the loss occurs in an application-limited state. For sake of simplicity, we assume that fast recovery always takes a single RTT regardless of the number of packets lost in a transmission window, as suggested by [10].

Using the above observations, we express the TCP delay of the i^{th} packet sent in a transition from state s to state s' as:

$$d_{s;s'}^{(i)} = L + \begin{cases} b/(fa) + RTT + (3+i)/f & \text{if } s' \in FR \\ 0 & \text{if } s' \in TO \\ b/(fa) & \text{otherwise} \end{cases} \quad (1)$$

where L is the one-way sender to receiver network delay. For loss-free transitions, the delay added by TCP is determined by the backlogged packets, and hence is modeled as $b/(fa)$, as shown by the third case of (1). For transitions to fast recovery states, an additional delay of $RTT + (3+i)/f$ is introduced by the in-order delivery guarantee of TCP, as shown by the first case of (1). Since the TCP sender is

likely to be idle during timeouts, we assume no packets are sent in transitions to timeout states.

The number of CBR packets sent in a transition from s to s' $n_{s;s'}$ is given by

$$n_{s;s'} = \begin{cases} 1 & \text{if } s \in AL, s' \in AL \\ \lfloor \min(b, wMSS)/a \rfloor & \text{if } s' \in \{CA|SS\} \\ \lfloor \min(b, (w+3)MSS)/a \rfloor & \text{if } s' \in FR \\ 0 & \text{if } s' \in TO \end{cases} \tag{2}$$

Since our model evolves at packet-level granularity while in an application-limited state (see Section 4.1), a single packet is sent in a loss-free transition from an application-limited state, as captured by the first case of (2). The second case models the number of packets sent in a loss-free transition from a network-limited state, which is determined by the number of backlogged packets that fit into the congestion window. The third case accounts for the extra transmissions due to the receipt of the duplicate ACKs needed to trigger a fast recovery, a mechanism known as window inflation [6].

We obtain the stationary distribution of the Markov chain for the TCP source, π_s, using standard steady-state discrete-time Markov analysis; see for example [36]. Let N_t be the number of packets successfully sent in some time interval $[0, t]$ and let $N_t(d)$ be the number of packets out of N_t that experience delay d. Then, the portion of packets sent that experience delay d is given by $N_t(d)/N_t$. Let D be the steady state delay distribution of a TCP connection with a CBR source. Assume D is defined over some finite interval A. Using renewal theory [36], we can now compute the steady-state delay distribution.

$$\begin{aligned} D = d \quad & \text{w.p.} \lim_{t\to\infty} \frac{N_t(d)}{N_t} && \forall d \in A \\ = d \quad & \text{w.p.} \frac{\sum_{s\in S} \pi_s \sum_{s'\in S} q_{s;s'} \sum_{i=1}^{n_{s;s'}} I_{d_{s;s'}=d}}{\sum_{s\in S} \pi_s \sum_{s'\in S} q_{s;s'} n_{s;s'}} && \forall d \in A \end{aligned} \tag{3}$$

where I is the indicator function, π_s is the steady-state distribution of the chain, and $d_{s;s'}$ and $n_{s;s'}$ are given in (1) and (2), respectively. The numerator and denominator correspond, respectively, to the number of packets sent that experience delay d in steady-state and the number of packets sent in steady-state. Equation (3) can be solved numerically to yield the performance statistics of TCP: the delay jitter σ_D and the α-delay percentile $\arg\max_x P\{D \le x\} \le \alpha$, along with other statistics such as the mean delay $E[D]$.

4.3 Backlog and Window Size Evolution

The discrete-time Markov model for the TCP source moves along several states, changing the congestion window size, send buffer size, and congestion control phase based on the packet loss feedback. For example, in the absence of packet loss, the TCP model transitions from state (w, b, l) to state $(w+1, b', l')$ if the sender is in congestion avoidance. A detailed description of the Markov chain is given in [9].

Since TCP is a byte stream protocol, it can assemble a number of small application packets into one TCP segment. An application that uses small packets (e.g., VoIP) yields a TCP flow that dynamically varies its segment size, and hence the packet size on the wire, depending on the congestion in the network. During network-limited periods, the data backlog enables the TCP sender to use the maximum segment size. In application-limited periods, however, there is no backlog at the sender, and TCP matches the segment size to the application payload size. Let M_s be the size of a segment transmitted in a transition from state s. Hence, $M_s = a$ if $s \in AL$ and $M_s = MSS$ otherwise.

The *backlog evolution* (i.e., the TCP send buffer occupancy evolution) for two successive states, $s = (w, b, l)$ and $s' = (w', b', l')$, is modeled by

$$b' = \begin{cases} \max(0, b + aft_{s;s'} - M_s) & \text{if } s \in AL, s' \in AL \\ \max(0, b + aft_{s;s'} - wM_s) & \text{if } s' \in \{CA|SS\} \\ \max(0, b + aft_{s;s'} - (w+3)M_s) & \text{if } s' \in FR \\ \max(0, b + aft_{s;s'}) & \text{if } s' \in TO \end{cases} \tag{4}$$

where $t_{s;s'}$ is the time taken for the transition from s to s', which can be found in [9]. The first term in (4) $b + aft_{s;s'}$ models the increase in backlog size due to newly admitted packets to the send buffer. The second term models the decrease in backlog size due to the transmission of segments, which is obtained by applying similar reasoning to that used to derive (2).

5. MODEL VALIDATION

We evaluated the model using experiments in a controlled network environment and Internet experiments using PlanetLab and residential machines. We use "CBR-TCP" to denote a TCP connection with a CBR source, "FTP" for a TCP connection with bulk data transfer, and "web" for a TCP connection with HTTP traffic.

We wrote a tool that can send and receive bidirectional CBR over TCP flows with different packet sizes and different packetization (inter-sending time) intervals. To validate our model we use CBR sources with packet sizes of 174, 724 and 1448 bytes, and packetization intervals of 20 ms and 30 ms, as these choices approximately reflect typical one-way voice [30], low bit rate interactive video [15] and live video streaming [16]. The size of the packet includes a 12 byte RTP header [30] and two bytes for framing RTP packets over TCP [21]. Hence, excluding the header size, the bit rate of the voice flows is 64 kb/s and 42 kb/s, that of interactive video is 284 kb/s and 187 kb/s, and that of live video streaming is 573 kb/s and 378 kb/s. Unless stated otherwise, we refer to the voice flow with a bit rate of 64 kb/s as 'VoIP' and the live video streaming flow with a bit rate of 573 kb/s as 'video' flow and only present the results for them due to lack of space. Excluded results are available in [9]. We abuse notation and refer to the segment loss rate in the network as the packet loss rate. These rates may be different for VoIP flows because TCP can assemble several small packets into one segment during network-limited periods.

All the experiments, except for those run in the PlanetLab environment, were conducted using Linux (kernel versions 2.6.17.8 and 2.6.9) and Windows XP machines. Both operating systems yielded similar delay performance and hence Windows XP results are not shown. PlanetLab experiments were conduced using Linux machines. The system and session-level TCP settings were determined according to the configuration described in Section 6.5.

5.1 Validation Using Configured Drop Rates

We performed the model validation on a test-bed that emulates a wide range of network settings. The topology

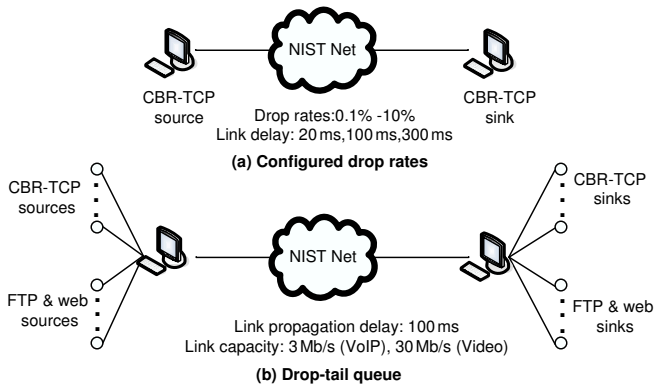

Figure 4: Experiment setup for model verification in a controlled environment.

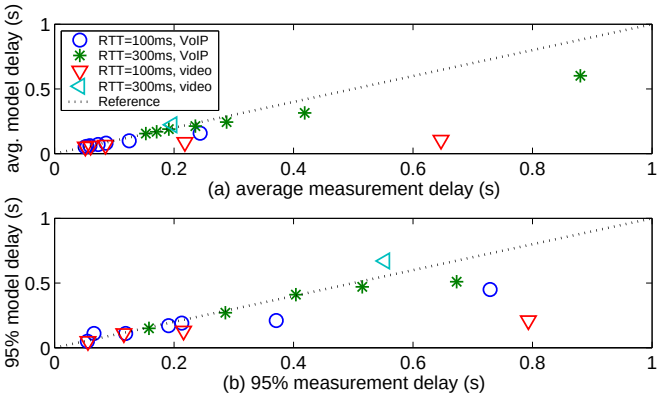

Figure 5: Predicted vs. measured (a) mean (b) 95th percentile TCP delay for VoIP and video flows for various loss rates and RTT of 100 ms and 300 ms.

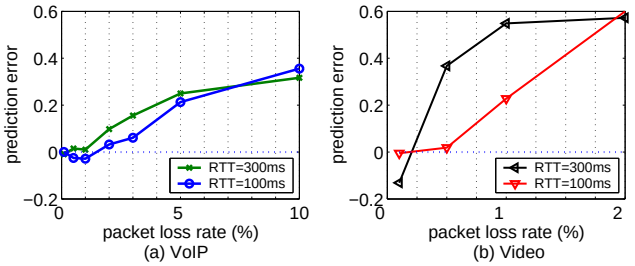

Figure 6: The relative modeling error for VoIP and video flows as a function of loss rate.

of the test-bed is shown in Figure 4(a). We consider a single CBR-TCP flow going through a router running NIST Net [1], a network emulation program which can introduce constant delay and can drop packets according to a configured loss process. We configured NIST Net to drop packets uniformly at random independent of their size.

We varied the network setting, as follows. NIST Net was configured with drop rates of 0.1%, 0.5%, 1%, 2%, 3%, 5% and 10%, and a fixed round-trip propagation delay of 20 ms, 100 ms, and 300 ms. The delay setting choice roughly reflects the delay of sites on the same coast in US, US coast-to-coast delays, and trans-continental delays [15]. Note that in this setting there were no background flows and hence there were no queuing delays on the round-trip. We do not consider loss rates greater than than 10% because the average TCP throughput (i.e., the available network bandwidth) does not satisfy the rate requirement of the CBR-TCP flow for considered RTTs. The average TCP throughput can be computed using Padhye's equation [29]. For each set of parameters, we ran the experiment for five minutes and repeated each experiment ten times. We present the average results of these experiments and compare them to the ones obtained using our model. The model was run with the assumption of random packet losses (see [9]).

Figure 5(a) and Figure 5(b) present the predicted vs. measured mean and 95th percentile TCP delay, respectively, for VoIP and video flows for various network packet loss rates and RTT of 100 ms and 300 ms. As shown, the model provides satisfactory matching for the majority of cases, specifically, when the measured delay is below 0.6 s. To better see the modeling accuracy across various loss rates and RTTs, we plot the relative prediction error of the average TCP delay with respect to the actual measurement for VoIP and video flows in Figure 6. Observe that for VoIP flows, the average error is less than 10% for loss rates up to 2%. For video flows, the relative prediction error is in the order of 20% for loss rates up to 1% and 0.1%, and RTT of 100 ms and 300 ms, respectively. The increase in relative error as compared to VoIP flows is due to high variability in sender packet backlog. Video flows have a higher backlog buildup than VoIP flows since they use higher bit rates.

The modeling mismatches are due to several reasons. First, the simplifying assumptions made by the model such as the recovery of multiple losses in a single transmission window within one round-trip time introduce error. Second, although our model accurately captures the backlog size at round-trip time granularity, it ignores backlog evolution at smaller time scales. A more detailed discussion of these issues can be found in Section 4.1 and 4.2.

Observe from Figure 6 that the prediction error increases with the network loss rate. This is because the size of the state-space of the model is truncated to reduce computational complexity, as explained in [9]. Further, for video flows, the jump in the prediction error at loss rate of 0.5% and 1% for RTT of 100 ms and 300 ms, respectively, occurs because the achievable TCP throughput is close to or below the bit rate of video flow. When the CBR rate is close to the TCP throughput, the TCP connection increasingly suffers from saw-tooth like transmission behavior, resulting in large variability in packet backlog buildup. This variability causes the high modeling error. When the throughput of the TCP connection is below the CBR rate, the CBR-TCP flow can be delayed indefinitely. For TCP throughput of at least twice the bit rate of VoIP and video flows, the modeling error was below 20%.

In general, the modeling error increases as the rate of the CBR source approaches the achievable TCP throughput. The 95th percentile measure pinpoints cases of largest deviation from the measurement, providing a highly conservative measure for the validity of the model; the average measure, demonstrates a better match between the model and the experimental results. Similar results were obtained for the variance and the maximum TCP delay measures.

5.2 Validation Using Internet Experiments

We performed model validation using the PlanetLab en-

vironment and hosts connected to residential DSL and cable modems. We conducted the PlanetLab experiments on machines located in the US (California, New York, Texas), Europe (Germany, Italy, UK), and Asia (China, India, Japan, Taiwan). For each sender and receiver pair, we ran our tool to generate VoIP and video flows for thirty minutes. The DSL experiments were conducted from hosts in the US, Israel, and Pakistan to hosts in New York and California.

For the majority of the PlanetLab and DSL experiments we observed only a handful of losses (<0.5%), whereas in a few cases, the throughput of the TCP connection did not meet the rate requirement of the CBR-TCP flow. We therefore started multiple FTP flows in tandem with the CBR-TCP flows, thereby increasing the congestion on the link and causing the CBR-TCP flows to suffer from higher losses. Figure 7(a) plots the predicted vs. measured 95th percentile delay for VoIP and video flows for a range of sites around the world. The top four settings in the legend of the figure refer to VoIP flows, and the bottom four refer to video flows. The network loss rates p and RTTs experienced by the flows are shown as well. As shown, there is a good match between the model and the measured delay.

5.3 Validation Using Drop-Tail Routers

We consider a scenario where multiple CBR-TCP flows compete with FTP and web flows for a bottleneck router with a drop-tail queueing scheme, as shown in Figure 4(b). We note that the lines in the figure connecting the data sources and sinks to the end hosts are for illustration purposes only and do not represent actual links.

We used the test-bed from Section 5.1 and modified NIST Net to incorporate a drop-tail queue. We devised a multi-flow setting in which five VoIP CBR-TCP flows compete with five long-lived FTP and varying number of web flows. We repeated the experiment for video flows. We used the SRI and ISI traffic generator [2] to generate exponentially distributed web traffic with a mean duration of 50 ms and a constant packet size of 512 bytes. The choice of the number of FTP and web flows, and packet size for web flows was inspired by the configuration used to evaluate the performance of TFRC-small packets [14]. The round-trip propagation delay was set to 100 ms for all experiments. The link capacity was set to 3 Mb/s and 30 Mb/s for voice and video CBR-TCP flows, respectively, so that the ratio of cumulative bit rate of five CBR-TCP flows to link capacity was one to ten. The drop-tail queue was maintained in packets and configured to hold 100 packets. For each configuration, we ran the experiment for five minutes, repeated it five times, and present the average of the results.

Since our goal is to validate our model for a given loss rate and RTT, we measured these parameters in each of the experiments. The network round-trip time consists of a round-trip propagation delay of 100 ms and a queuing delay at the router. For the VoIP and video experiments, the measured RTT was 380 ms and 101 ms, respectively. Hence, the congestion was low for the video experiments. The queuing delays and loss rates are different for these two flows because the link bandwidths in these two settings were different. Figure 7(b) presents the measured and predicted 95th percentile delay for VoIP and video flows. Some of the prediction inaccuracies in the drop-tail router experiments are caused by inaccurate characterization of the loss process. In this environment, as in the PlanetLab case, we run the model

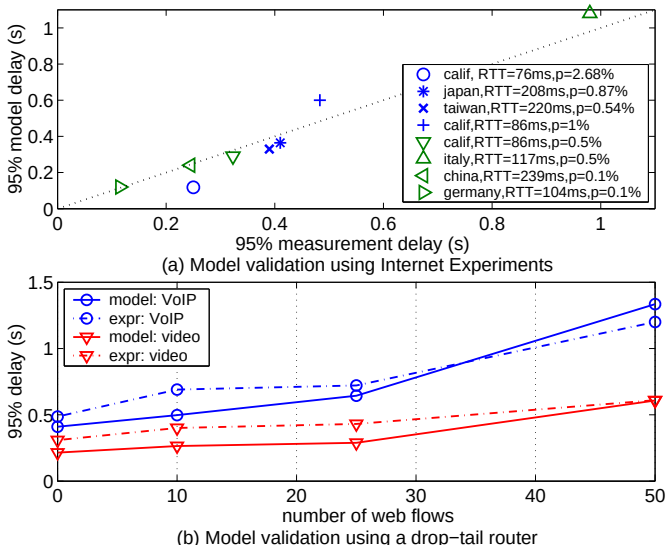

Figure 7: Model validation using (a) PlanetLab experiments (b) drop-tail router experiments for VoIP and video flows.

with the assumption of correlated packet losses (see [9]). However, the actual burstiness of losses experienced by the CBR-TCP flows varies depending on the level of statistical multiplexing at the router.

6. DISCUSSION

In this section, we explore the delay performance of real-time delivery over TCP. We experimentally characterize the working region for VoIP and live video streaming applications with bit-rates of 64 kb/s and 573 kb/s, respectively, and use our model to identify the working region for other bit-rates. Then, we study the impact of various mechanisms in TCP on its delay performance. Finally, we use the insights gained to provide guidelines for configuring TCP for real-time applications.

6.1 Working Region

Here we characterize the working region for VoIP and live video streaming applications, i.e., the conditions under which the performance of these applications is satisfactory. In general, the user perceived media quality is acceptable when the fraction of packets that arrive beyond their playback time is low and the end-to-end delay is low.

For interactive applications, ITU G.114 recommends that the worst-case one-way delay should be 400 ms. Studies show that 200 ms is an acceptable one-way delay limit for VoIP applications [26]. The choice of the delay limit for video is more flexible because people can usually tolerate a few seconds of startup delay. For the analysis we consider a 5 s startup delay, as suggested by [16]. While VoIP can usually tolerate up to 5% of packets that miss their playout deadline without a significant effect on intelligibility [26], video viewing quality drops rapidly at 0.1% [33]. We follow these guidelines and define the working region for VoIP and live video streaming as the range of network loss rates and RTTs where the 95th percentile and maximum TCP delay is at most 200 ms and 5 s, respectively. We explore how the performance varies with the delay limit in [9].

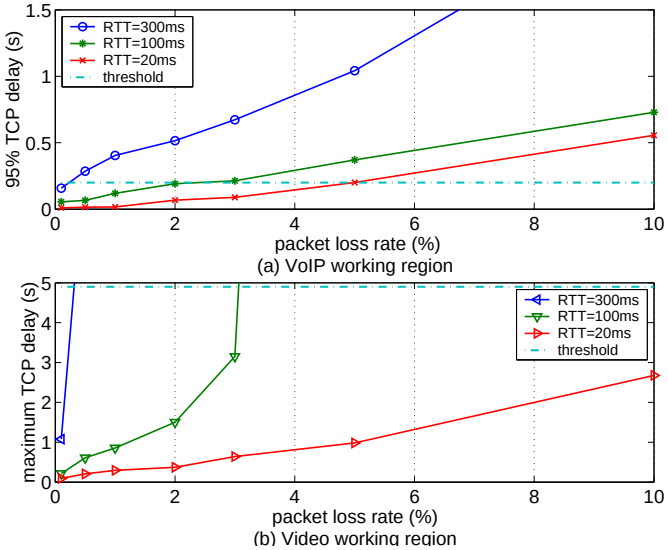

Figure 8: Working region for VoIP and live video streaming as a function of RTT and packet loss rate.

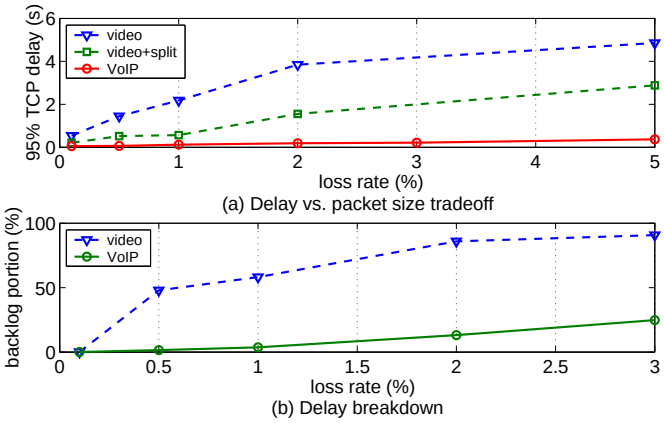

Figure 9: (a) The delay performance of two TCP flows having the same load in kb/s but different load in pps, and a VoIP flow. (b) Delay breakdown: the portion of TCP-level delays caused by the congestion control mechanism.

Figure 8(a) plots the 95th percentile delay for various loss rates from 0.1% to 10% and RTTs of 20 ms, 100 ms, and 300 ms for a VoIP flow with a bit-rate of 64 kb/s. The results shown were obtained empirically using the environment described in Section 5.1. Observe that when the RTT is 100 ms, the delay tolerance for VoIP is satisfied when the network loss rate is at most 2%. However, when the RTT is only 20 ms, the results indicate a tolerance of up to 5%. At the boundary of the working region, the delay added by TCP causes 5% of the packets to miss their playback deadline. Figure 8(b) plots the maximum delay for a live video streaming flow with a bit-rate of 573 kb/s. When the RTT is 100 ms, the streaming threshold is satisfied when the loss rate is at most 3%. For RTT of 300 ms, it is satisfied at a network loss rate of 0.1%. The jump in the maximum delay at a network loss rate of 0.5% and RTT of 300 ms occurs because the 5 s startup delay is no longer sufficient to completely mask TCP delays. This knee of the curve typically occurs when the achievable TCP throughput is close to the bit rate of the video flow, as explained in Section 5.1. The bit rates of 64 kb/s and 573 kb/s are the highest among the bit rates considered in Section 5 for VoIP and video flows and therefore, they give the most conservative estimate of the working region. We used the model to compute the working region for the lower bit rates. While the working region was less constrained due to the lower bit-rates, the results follow similar pattern as in Figure 8(b). Further, the working region can be significantly constrained if the application does not use delay-friendly TCP settings (see Section 6.5).

6.2 The Effect of Packet Size on Performance

Our experiments indicate that under the same network conditions, VoIP flows perform significantly better than video flows. Figure 9(a) plots the 95% delay for VoIP and video flows with the same workload in packet per second (pps) but quite different workload in terms of bits per second (kb/s), i.e., the VoIP flow has a bit rate of 64 kb/s whereas the video flow has a bit rate of 573 kb/s. The figure clearly shows a performance bias towards the VoIP flow. This happens because a video flow has a higher bit rate than a VoIP flow. Hence, during network-limited periods, a TCP sender transmitting a video flow builds up a larger packet backlog and consequently, it requires more time to drain this backlog. For VoIP flows, the TCP sender groups several queued VoIP packets into one transmission packet as permitted by the MSS. This further increases the queue drain rate, thereby reducing the queuing delay at the TCP sender.

An interesting question to ask is that among two flows having the same workload in kb/s, does TCP have a performance bias towards a flow with larger workload in pps? To address this question, we measured the delay performance of two flows having the same workload in kb/s but different workload in pps. The results are shown by the curves labeled video and video+split in Figure 9(a). Specifically, the packet rate of video+split flow is twice of the video flow but the application-level workload rate in bytes is the same. Surprisingly, there is a performance bias towards the flow with twice the packet rate of the other flow.

To illustrate the reason for this performance difference, we plot the TCP delay and congestion window size for two flows with the same application-level workload in kb/s in Figure 10. The flow in Figures 10(a) and (b) corresponds to an application that sends 100 MSS-sized packets per second. The flow in Figures 10(c) and (d) corresponds to an application that sends 200 half MSS-sized packets per second. Both flows operate over a symmetric network with 200 ms RTT and experience two close-by losses. Observe that the flow with half MSS-sized packets experiences lower delay than the other one. This happens because the AIMD mechanism updates the congestion control state as a function of the number of *packets* sent, rather than as a function of the number of *bytes* sent (see Section 3.1). Since TCP adapts its congestion control state and hence its throughput based on the number of packets sent, the magnitude of the throughput fluctuations (in bytes) is smaller for the flow with smaller packet size and higher packet rate, resulting in lower delays. For example, the peak delay of the flow with half MSS-sized packets in Figure 10 is 45% lower than that of the one with MSS-sized packets. As shown, the performance gain of a TCP flow with small packets (e.g., VoIP)

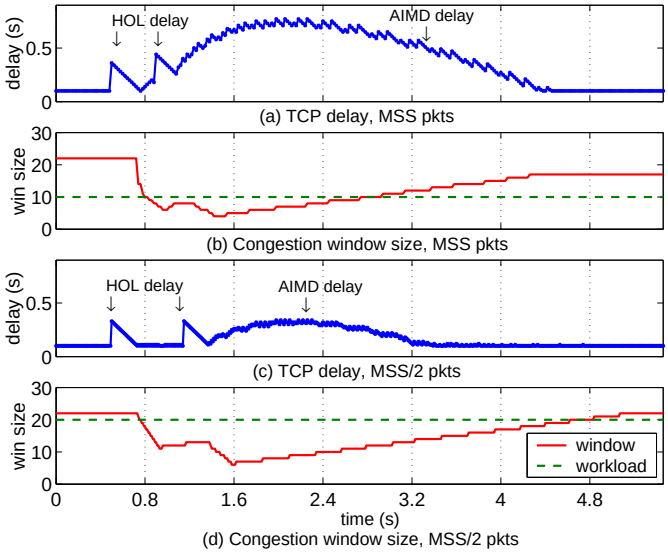

Figure 10: TCP delay and congestion window evolution for two flows with the same workload in kb/s but different packet sizes, MSS and half-MSS.

comes from the reduction in the delays caused by the AIMD mechanism. That is, video flows suffer more from packet backlogging than VoIP flows. However, reducing the packet size has side effects such as increased instances of packet reordering [23].

We analyzed the breakdown of TCP-level delays by computing the time packets are backlogged at the sender (i.e., the congestion control delay component) and the time it takes the TCP sender to get a packet to the receiving application (i.e., the retransmission and head-of-line delay components). Figure 9(b), shows the delay breakdown in terms of these two components for VoIP and video flows. As shown, the delays of a VoIP flow over TCP tend to be dominated by the loss recovery latency, whereas those of a video flow tend to be dominated by the delays caused by the congestion control mechanism. Similar results were obtained for CBR sources with other bit rates.

6.3 Sensitivity to Byte-counting

In order to provide a measured response to ACKs that cover only small amounts of data, [3] proposes to increase the congestion window based on the number of bytes acknowledged by each incoming ACK rather than on the number of ACKs received. This mechanism is known as byte-counting. Byte-counting is configured on a per-system rather than per-connection basis in Linux, and is disabled by default. It is not implemented in Windows XP. A question arises how does the performance of VoIP flows change when TCP increases its congestion window by the number of bytes sent.

To answer this question, we measured the delay of five VoIP flows competing with five long-lived TCP flows and varying number of web flows in a drop-tail queue environment (see Section 5.3). Figure 11 shows 95th percentile and maximum delay for a VoIP flow using ACK and byte-counting. It highlights the gain of a VoIP flow when byte-counting is not used. On average, the use of byte-counting increases the TCP delay by 10-20%. The delay increases because TCP with byte-counting increases its sending rate

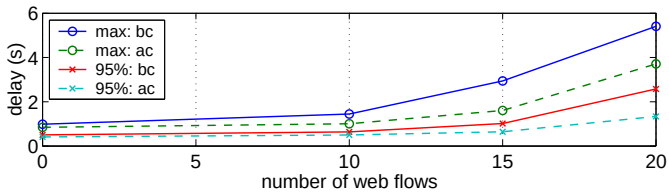

Figure 11: 95% and maximum delay for a VoIP flow using ACK and byte-counting.

in proportion to the number of bytes sent. Hence, a byte-counting TCP can be viewed as more fair than ACK-counting TCP with respect to the congestion control behavior. The support for byte-based congestion control mechanism must come from the underlying operating system. However, since Linux and Windows XP use ACK-counting by default, VoIP flows implicitly benefit from it.

6.4 The Effect of Timeouts on Performance

Since a real-time flow is rate-limited, it has the potential of causing the connection's congestion window to be small. Hence, the chance of sending enough segments for the receiver to generate the three duplicate ACKs becomes small, too. This can harm the delay performance as the sender may need to rely on lengthy retransmission timeouts for loss recovery. Nonetheless, our traces show that the likelihood of timeouts is low.

The likelihood of timeouts is directly effected by the behavior of TCP during application-limited periods. According to [18] there are three possibilities. A TCP sender can reduce the congestion window so that it would reflect the actual amount of data sent, as suggested by the window validation extension [18]. It may increase the congestion window, resulting in an arbitrarily large window value, or it may maintain the same congestion window, resulting in an invalid window value. We focus on the latter case, as it is the one observed in our measurements for Windows XP and Linux systems. The invalid congestion window overestimates the actual amount of data sent, and hence reduces the likelihood of timeouts. This overestimation happens implicitly for CBR-TCP flows, because during application-limited periods, the TCP sender retains memory of an 'inflated' congestion window used to clear the recent data backlog. The congestion window behavior can be observed in Figure 10(b) and (d). The window value overestimates the actual load by 30% and 40%, respectively, for the flow with MSS-byte and MSS/2-byte packets. The other factor that influences the likelihood of retransmission timeouts is the limited transmit mechanism [4], which is enabled by default in Linux 2.6 and Windows XP. This mechanism allows a TCP sender to send a new data segment upon the receipt of each of the first two duplicate ACKs, thereby increasing the chances of the three duplicate ACKs to arrive.

To quantify the impact of these mechanisms (i.e., limited transmit and invalid congestion window) on TCP's loss recovery efficiency, we compared the timeout probability predicted by of our model to that predicted by a recent detailed loss recovery model proposed by Beomjoon *et al.* [8]. The latter model does not consider the impact of the two mechanisms. We show the results in Figure 12 for a window size of three and an overestimation factor of 30%. To validate

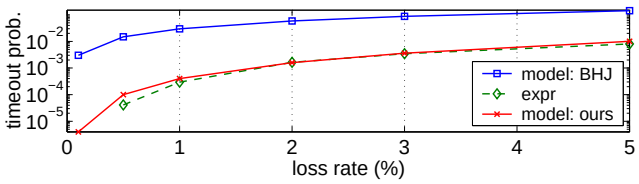

Figure 12: The timeout probability predicted by our model, Beomjoon *et al.* model (BHJ), and the measured probability for various loss rates.

the results we measured the timeout probability of several VoIP flows with an average window size of three in an environment with random packet losses. The figure shows that for small windows the absence of limited transmit and invalid congestion window has a non-negligible impact on the timeout probability.

6.5 TCP and OS Settings

We provide a comprehensive set of guidelines for delay-friendly settings of TCP and OS parameters. While several settings such as disabling Nagle's algorithm and using large receive buffers are common practices in delay-sensitive applications, the impact of others, specifically, window validation, byte counting and limited transmit is less obvious.

As discussed in Section 3, Nagle's algorithm should be disabled as it introduces transmission delays at the TCP sender. CBR-TCP applications should set a large receive buffer and operate with non-blocking sockets so that the TCP transmission is not limited by the flow control mechanism. To increase the loss efficiency of TCP, SACK should be enabled [11] and limited transmit be used. The latter is applicable for a TCP connection with small windows. Congestion window validation during application-limited periods, and byte-counting should be disabled. These settings are disabled by default on Linux and Windows XP systems. The initial window size should be set to four segments as it can remove delays up to three RTTs and a timeout during the initial slow-start period [5].

7. DELAY REDUCTION APPROACHES

Here we discuss application-level heuristics that can improve the performance of real-time media applications without additional help from the network. We analyze whether the delay reduction comes at the expense of other flows, in particular FTP flows. In the following, we first discuss a packet splitting scheme and then consider the use of parallel connections. We show that both schemes are effective for video flows but have only a marginal impact on VoIP flows.

7.1 Packet Splitting

As described in Section 6.2, the congestion control mechanism of TCP results in a performance bias in favor of flows with small packets. A question of interest is whether the delay performance of real-time applications can be improved by masquerading TCP flows with large packets as flows with small packets. The application can split every large packet into a few smaller ones, while maintaining the same workload in bytes per second. We call this scheme *split-N*, where N is the packet split factor. Packet splitting, however, may also backfire: if all CBR-TCP flows started using packet

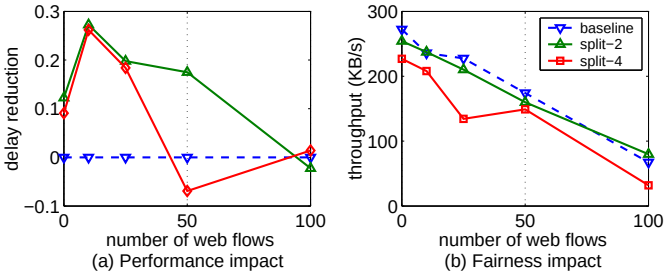

Figure 13: (a) The reduction in the 95th delay percentile of a video flow using *split-N* in a drop-tail queue environment (b) Throughput of background FTP flows in the same environment.

splitting, the network could quickly become congested due to the TCP header overhead. Hence, a wide-scale adoption of such an approach runs the risk of degrading the performance of all flows. Further, reducing the packet size can increase instances of packet reordering [23].

We analyzed the upper bound on the delay reduction of a *split-N* scheme for both video and VoIP flows by applying our model in the configured drop rates environment described in Section 5.1. Though not shown due to lack of space, the *split-2* scheme reduced the 95th delay percentile by 60% on average. This is consistent with the observation made in Section 6.2 that a TCP flow with small packets experiences smaller packet backlogs, and hence smaller delays, than that of a flow with large packets. For VoIP flows, the scheme yielded diminishing gains due to the low backlog levels experienced by these flows.

To understand the performance of *split-N* in a wide-scale deployment, we measured the delay of a video flow (i.e., a 573 kb/s video source) in an environment with a drop-tail queue described in Section 5.3. As shown in Figure 13(a), the *split-2* scheme reduces TCP delay by up to 30% under low and moderate loss rates, whereas schemes with higher split factors yield diminishing gains or perform even worse than a no-split scheme. The performance degradation is partially due to the increase in the burstiness of the flow with packet splitting. This burstiness can be reduced to some extent by evenly spacing split-packets over the packetization interval. However, perfect pacing may be difficult to achieve at the application layer due to the small packetization intervals (e.g., 20 ms) used in practice.

During periods of high congestion (100 web flows), a TCP sender using a *split-N* scheme is heavily backlogged and hence it is unable to obtain a performance bias by using a *split-N* backlogged scheme. We used the drop-tail queue environment to study the fairness implications of this scheme. In particular, we measured the throughput of long-lived TCP flows that share a congested link with video flows employing packet splitting. As shown in Figure 13(b), the *split-N* scheme impacts the throughput of the long-lived TCP flows. For example, the use of *split-4* reduces the throughput of a background TCP flow by 27% on average. From the plot, we observe that the throughput reduction quickly increases with the split factor.

7.2 Parallel Connections

A straightforward approach to improve the delay performance of a CBR-TCP flow is to stripe its load across paral-

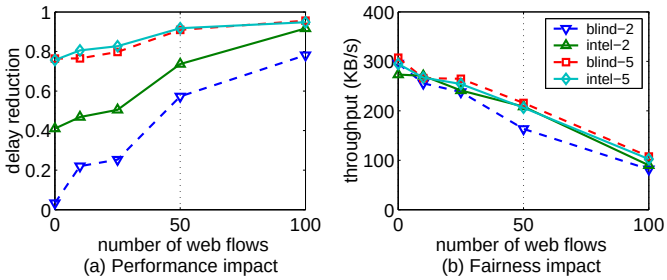

Figure 14: (a) The reduction in the 95th delay percentile for a 'blind' and an 'intelligent' scheme with N connections in a drop-tail queue environment (b) Throughput of background FTP flows in the same environment.

lel TCP connections. The idea is that several TCP streams are more aggressive than one TCP stream with respect to the congestion control behavior [34], which can result in lower TCP delays. The use of parallel TCP connections for streaming and data-intensive applications has mainly focused on enhancing the throughput. However, we focus on reducing the delay. Specifically, we provide insights on the delay performance of parallel connection schemes for real-time applications.

Packet striping can be done in delay-agnostic or delay-aware fashion. The simplest approach is to use a delay-agnostic ('blind') parallel connection scheme that sends packets over parallel TCP connections in a round-robin fashion. Due to limited space, we only show the performance improvement and fairness impact of this scheme for video flows in the drop-tail queue environment described in Section 5.3. Figure 14(a) shows that five parallel connections reduce the 95th delay percentile by 90% on average. The delay reduction stems from lowering the load per connection, which in turn reduces sender backlog buildup and receiver head-of-line blocking per connection, and hence the TCP delay. Though not shown, the performance gain was negligible for VoIP flows. The gain was negligible because the delay reduction is offset by the decrease in TCP's loss recovery efficiency caused by small congestion windows (see Section 6.4). The small congestion windows are due to the low load per connection. We note that the scheme yielded diminishing gains when more than five connections were used.

We propose a delay-aware ('intelligent') scheme which selects a connection for packet transmission that has the smallest TCP send queue and is not in the timeout state and show the results in Figure 14(a). The 'intelligent' scheme outperforms the 'blind' scheme because it dynamically avoids connections with large queues and in timeout states. Further, due to its dynamic nature, this scheme copes better with connections with small congestion windows. Similar to the 'blind' scheme, we observe that using more than five parallel connections results in diminishing gains. We note that the parallelization spectrum ranges from a single flow to having as many flows as the packet rate per RTT. Similar to packet splitting, we study the fairness impact of these schemes on the background traffic using a drop-tail queue environment. We present the results in Figure 14(b). As shown, both 'intelligent' and 'blind' schemes have a negligible impact on the throughput of the background long-lived FTP flows. The impact is negligible because these schemes do not introduce additional traffic besides session setup and teardown. Though parallel TCP streams are more aggressive than a single TCP stream, their aggregated throughput is still limited by the rate of the CBR source.

8. RELATED WORK

There is an extensive literature on analytical and experimental evaluation of TCP. We present only those studies closely related to ours and refer the reader to [28] for a comprehensive survey of TCP modeling. The majority of TCP modeling studies are geared towards file transfers assuming either persistent [29] or short-lived flows [10]. Our work differs from past work in that we consider non-greedy rate-limited flows with real-time delivery constraints. More recently, the performance of TCP-based video streaming has been analytically analyzed by [33]. The receive buffer size requirement for TCP streaming has been determined in [19]. These papers combine TCP throughput and application-layer buffering models to compute the portion of late packets, whereas we directly model the transport-layer delay of TCP. Our work further differs from those above in that we consider applications with tight delay constraints such as VoIP. Wang *et al.* [34] have performed a comprehensive analytical study of the performance of multipath video streaming using TCP. This work explains that the performance of TCP streaming increases as the ratio of the aggregated TCP throughput to video encoding rate increases. However, our contribution is the insights on the TCP delay performance.

Goel *et al.* [15] present an empirical study of kernel-level TCP enhancements to reduce the delays induced by congestion-control for streaming flows. The performance of TCP for real-time flows has also been considered by [22, 25]. However, unlike our study, these papers propose a modification to the TCP stack. Application-layer heuristics for improving the loss recovery latency of TCP have been suggested [24]. These heuristics are geared towards bursty traffic flows and hence may not be effective for real-time flows.

9. CONCLUSION AND FUTURE WORK

We have presented a Markov-chain TCP delay model for CBR-TCP flows. The model captures the behavior of VoIP and streaming flows. We used experiments and the model to derive the working region of these flows. We verified the model in a test-bed and in PlanetLab. We explored the impact of TCP mechanisms and presented guidelines for improving the delay friendliness of CBR-TCP applications. The delay performance of a video flow can be improved using packet splitting or parallel connection heuristics.

This study provides insights on the use of TCP for VoIP and live-video streaming applications. A direct comparison of real-time delivery over TCP versus unreliable protocols is left for future work. We have used delay percentiles to evaluate the performance of CBR-TCP flows. However, Mean Opinion Score (MOS) is considered a better metric for evaluating user-perceived performance. This is another potential topic for future work.

10. ACKNOWLEDGEMENTS

We especially thank Sally Floyd for her detailed and valuable feedback on the initial technical report. We would also like to thank the anonymous reviewers, Oren Laadan, and Jitendra Padhye for their valuable feedback on the paper.

Finally, we sincerely thank our shepherd Martin Arlitt whose guidance helped us refine the final version.

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