EFFECT OF PACKET SIZE ON TCP-RENO PERFORMANCE OVER LOSSY, CONGESTED LINKS

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Abstract— This paper investigates the performance of a single TCP-Reno connection over a lossy, congested link as a function of TCP packet size on the forward link and the ratio α of forward-to-reverse link capacity measured in terms of packets/sec. $\alpha > 1$ constitutes the primary scenario of interest since the reverse link (for the ACKs) is a bottleneck, which directly impacts the forward link throughput. Congestion at the forward buffer is induced by assuming buffer size to be significantly less than the bandwidth-delay product of the round-trip connection. Further, forward link error characteristics are modeled as a continuous-time two-state Markov chain, where packets are either transmitted free of errors (good state) or lost (bad state). By tuning the model parameters, the link error process can be made to vary from independent loss to (highly) correlated loss scenarios. Results show that TCP-Reno performance benefits from larger packet sizes for connections dominated by link errors; throughput is less sensitive to packet size when the connection is dominated by forward path congestion. Additionally, performance sensitivity to variation of forward buffer size and the round-trip propagation delay is discussed and supported by simulation results using ns-2 Network Simulator.

Keywords— TCP-Reno, asymmetry, correlated errors, performance analysis

I. INTRODUCTION

TCP performance over an end-to-end path that includes a wireless/satellite segment is a function of several important parameters: link loss characteristics and tuning of various TCP parameters such as buffer and packet size (which depend on the channel error). Further, in TCP over satellite, applications typically suffer from channel asymmetry, where the forward link capacity (measured in packets/sec) greatly exceeds that of the reverse link, leading to further under-utilization of the forward link. In this work, we focus specifically on the choice of TCP packet size that fundamentally affects TCP dynamics since the sender's current congestion window $W(t)$ is adjusted in units of packets depending on receipt (or non-receipt) of acknowledgements (ACKs). Thus when $W(t)$ is incremented by 1 packet, the equivalent increase in throughput (measured in bits/sec) is greater for con-

nections using larger packet sizes. We investigate the issue of optimal packet sizing in the presence of two types of packet losses - due to i) buffer overflow (congestion loss) and ii) channel loss, respectively. It is useful to characterize links dominated by congestion losses as presenting a “busy connection”, while those dominated by link errors as presenting a “lossy connection”.

II. TCP BACKGROUND

A. Congestion Control Algorithm

A TCP sender transmits data packets¹ in sequence and expects to receive acknowledgements (ACKs) confirming their successful reception. The maximum number of packets allowed to be outstanding in the network at time t is known as the congestion window size $W(t)$. $W(t)$ is typically maintained in units of bytes but for convenience it is expressed in units of packets here, unless otherwise noted. ACKs generated by the standard TCP receivers carry the information about the last *in-order* data packet received; out-of-order packets are stored but not acknowledged until all previously transmitted packets have been received. TCP uses this continuous flow of ACKs as a way of probing the network and detecting signs of congestion. It adapts itself by dynamically adjusting $W(t)$ in one of two distinct phases, known as *slow-start* and *congestion avoidance* phases. TCP switches from slow-start to congestion avoidance when $W(t)$ reaches the slow-start threshold $W_{sssth}(t)$, that is also dynamically adjusted. The slow start phase allows for an exponential increase of $W(t)$ that grows up to $W_{sssth}(t)$ by increments of one (packet) for every “new” ACK² received. On the other hand, congestion avoidance phase allows for a linear window growth with time by incrementing $W(t)$ by 1 for every (current) window's worth of new ACKs received.

¹ Although the term “segment” is widely used in the literature we use “packet” throughout this paper.

² We use “new” ACK to refer to those acknowledging data not previously acknowledged; “repeated” ACKs acknowledge previously ACKed packets.

The sender deduces the congestion state of the network by two different means - in the first, the time elapsed between the transmission of a data packet and the reception of its corresponding ACK, known as the round-trip time (RTT) is compared to an adaptive threshold, known as the retransmission time-out (RTO); if RTT exceeds RTO , the sender assumes the packet has been lost. The second technique requires the sender to monitor the number of consecutive repeated ACKs received (that signifies a break of in-sequence received packets); if it reaches a threshold K (usually $K = 3$), a congestion loss is assumed. The following summarizes the congestion control mechanism actions taken by TCP-Reno in the event of packet loss:

1) Time-Out

- Set $W(t+) = 1$ and $W_{ssth}(t+) = W(t)/2$.
- Retransmit the lost packet and resume slow-start phase.

2) K Repeated ACKs

- Fast-retransmit the lost packet.
- Set $W(t+) = W(t)/2 + K$ and $W_{ssth}(t+) = W(t)/2$
- While a new ACK is not received, increment $W(t+)$ by one for every repeated ACK and transmit new packets if possible (Fast-Recovery Phase).
- Upon receiving a new ACK, set $W(t+) = W(t)/2$ and resume congestion avoidance phase.

While in fast-recovery phase, assuming each repeated ACK indicates a packet has been successfully received allows the sender to avoid dropping the number of outstanding packets in the network below $W(t)/2$. If that happens, the result will be a transmission burst upon reception of the first new ACK.

B. Effects of Asymmetry

Specific problems arise when TCP is used over asymmetric connections, since TCP relies on a regular flow of ACKs on the reverse path to advance its forward congestion window. When $\alpha > 1$ (reverse link is the bottleneck), it results in further under-utilization of forward link capacity. As $W(t)$ grows, the reverse link becomes fully utilized and reverse buffer overflow follows. As $W(t)$ is further increased, the forward link becomes fully utilized at which point only 1 out of α ACKs “survives” the reverse buffer. Finally the forward buffer overflows triggering TCP’s congestion avoidance mechanisms. Previous studies on the effects of link asymmetry on TCP performance highlight the following three observations:

1. *Increased burstiness on forward transmissions* due to ACK losses in the reverse link - the “surviving”

ACKs acknowledge multiple packets and as a result the sender releases multiple new packets into the forward buffer.

2. *Slower congestion window growth* since only a fraction of ACKs generated by the receiver actually reach the sender and the fact that the TCP window evolution algorithm is based on number of ACKs received and *not* the amount of data acknowledged.

3. *Reduced effectiveness of the fast-retransmit algorithm* due to ACK loss, since it depends on the reception of at least 3 ACKs to be triggered. Besides, ACKs for retransmission may be lost resulting in time-outs.

A pioneering analysis of the fundamental issues involving link asymmetry was carried out in [1] where it was shown that there are *three* modes of operation based on the relationship between the normalized asymmetry factor α (defined as the ratio of time to transmit an ACK in the reverse link to that of a data packet in the forward link) and the forward buffer size B_f . When $\alpha > 1$ (the usual scenario of interest), for $B_f < \alpha$, the forward buffer size is insufficient for the bursts of α packets that will occur when the forward link is fully utilized. For $B_f > 3\alpha$, the forward buffer will still not be empty by the time the sender receives 3 repeated ACKs and detects a packet loss due to congestion in the forward link. As a result, the retransmission will find the forward link fully utilized, its corresponding ACK will find the reverse buffer full, and it is likely to be dropped. Thus, “normal” operation takes place for $\alpha < B_f < 3\alpha$, when the above effects are not seen. An initial approach to solving the link asymmetry problem was to reduce α by using smaller ACKs. It was proposed in [2], [3] and [4] to compress TCP/IP headers by exploiting the redundancy between subsequent packets. Another solution early proposed was the use of *delayed-ACKs* - due to the cumulative nature of ACKs in TCP, the receiver may avoid overflowing the reverse buffer by generating less than one ACK for every data packet received. In [5], an optimal delayed-ACK mechanism is described. Its use alleviates congestion in the reverse link but does not address the problems of forward transmission burstiness and slow congestion window growth. In [6] a solution involving modifications only to the TCP stack at the receiver’s terminals is proposed in order to dynamically adjust the number of packets acknowledged per ACK (*ppa*). In [7], the author evaluates the use of some important algorithms but does so for a connection where the reverse link does not represent the bottleneck. A more complete and detailed

analysis of the effects of link and latency asymmetry is presented in [8]; several techniques are suggested to address all three problems that arise with link asymmetry listed above.

C. Effects of Transmission Errors

The control action mechanism in TCP senders was designed to respond to congestion (buffer overflow) losses. Any packet loss in the forward link due to other causes such as link errors are therefore interpreted by the sender as arising due to network congestion, leading to reduction of the transmission rate, thereby needless under-utilizing network resources. We note that transmission errors in the forward link impact TCP connection throughput far more than errors on the reverse channel, since ACKs in TCP are cumulative. Thus whenever an ACK is lost, its information is resident in all subsequent ACKs. Nonetheless, ACK losses still impact forward TCP throughput in a similar manner as reverse buffer overflows do in asymmetric connections.

III. SYSTEM MODEL

We assume a persistent data source (e.g. an ftp application) operating over forward and reverse paths with capacity μ_f and μ_r packets/second and FIFO forward and reverse buffers of sizes B_f and B_r , respectively. The resulting normalized asymmetry factor is $\alpha = \mu_f/\mu_r$ [1], and normalized forward buffer size $\beta = B_f/\mu_f\tau$ where τ is the total (round-trip) propagation delay. The forward link drops data packets due to link-layer errors based on a *continuous-time, two-state* model described next.

The channel alternates between two states - the “good” and the “bad” denoting the extent of packet loss. It remains in the present state for an amount of time defined as an exponential random variable with average $\tau_g(\tau_b)$, before switching to the other state. The probability of a packet being lost due to error in good(bad) state is $p_g(p_b)$. For simplicity, we assume $p_g = 0$ and $p_b = 1$ in our work. The resultant packet loss process is a hidden Markov model with the important feature that it is *not* independent, but correlated in general. The channel state transitions drive TCP’s congestion control algorithm; thus the sequence of packet losses experienced by TCP sender is a result of the interaction between TCP’s own feedback mechanism (congestion control) and the channel state evolution. Consequently it is impossible to translate channel statistics to that of the actual packet loss events and throughput results will be presented in terms of

a channel “unavailability” factor $\rho = \tau_b/(\tau_g + \tau_b)$. It is straightforward to verify that correlation between error events can be increased by increasing τ_b (and τ_g correspondingly) for a fixed ρ .

We make the important distinction between connections whose performance are dominated by congestion losses (“*busy connections*”) and link losses (“*lossy connections*”). For a fixed ρ , increasing the correlation coefficient between errors implies larger (on average) error bursts and consequently fewer *error events*. Hence, for a fixed ρ , sufficiently large (small) values of τ_b indicate “*busy connections*” (“*lossy connections*”). A “*lossy connection*” is extremely difficult to model and will not be attempted; the saturation throughput limit in a “*busy connections*” will now be addressed.

IV. CONNECTIONS WITH $\alpha \leq 1$

Though $\alpha \leq 1$ is not the primary scenario of interest, it is treated first for the result simplicity and preliminary insights it provides for $\alpha > 1$.

A. Saturation Throughput

In “*busy connections*”, reductions of window size $W(t)$ are mainly due to buffer overflows. It is expected that congestion occurs (roughly) every time $W(t)$ reaches some value W_{full} , which for $\alpha \leq 1$ is given by:

$$W_{full} = \mu_f\tau(1 + \beta) \quad (1)$$

Fig.1 illustrates such behavior for two different packet sizes; W_{full} (in bytes), is the same irrespective of the packet size being used. Note that larger packet sizes allow for faster growth of $W(t)$ and as a result cycles have smaller duration. This figure suggests identical

Fig. 1. Typical congestion window evolution for two different packet sizes in a “*busy connection*” and $\alpha \leq 1$.

throughput for both packet sizes since throughput is

“roughly” the area under these curves. However, this estimate must be adjusted to account for time spent in the transition between *cycles*, defined as the time from the last transmission prior to a loss detection and the reception of the second “new” ACK subsequent to retransmission of the lost packet, which are periods of low (almost zero) throughput. The reason for using the second “new” ACK is that in the fast-recovery algorithm, the sender must pause after retransmission while receiving $W_{full}/2-3$ repeated ACKs; it will only then attempt a new transmission which will eventually generate a “new” ACK other than the one for the retransmission. Fig.2 shows a closer look at the tran-

Fig. 2. Closer look at *cycle* transitions for 1000bytes (solid line) and 200bytes (dotted line) packets.

sition between *cycles* for two different packet sizes and their time duration is shown to be the same. Hence we should expect a poorer performance for larger packet sizes as the number of *cycles* in this case is higher.

In [9], expressions to estimate the saturation throughput are derived and details will not be presented here. The approach is to find expressions for both the time duration and the throughput of each one of two cycle stages, namely the *transition* phase, as described above, and the *growth* phase (the remaining portion of the cycle). The throughput of each *cycle* can then be obtained which is also the overall throughput of a “*busy connection*”.

T_1 , T_2 and T_3 are defined as the *RTT* for packets transmitted in three distinct situations: (i) when a retransmission occurs (both links and buffers are fully utilized); (ii) when the sender transmits a new packet after a retransmission (the network load has been reduced by then); (iii) an average *RTT* for packets transmitted in the *growth* phase. The final expression for the saturation throughput as a function of T_1 , T_2 , T_3 and W_{full} , normalized by the forward link ca-

pacity μ_f is given by:

$$\gamma_{sat} = \frac{W_{full} [6T_1 + (3W_{full} + 2) T_3]}{4T_3\mu_f [3T_1 + 2T_2 + W_{full}T_3]} \quad (2)$$

Note that since μ_f is given in packets per second, (2) is also a function of the packet size for a fixed bandwidth.

B. Simulation Results

Fig. 3. Normalized throughput (γ_{norm}) as a function of the normalized average bad state duration (τ_b/τ) for $\rho = 10^{-3}$. The horizontal dotted lines are results from equation (2).

We ran simulations using the Network Simulator (*ns-2*) [10] for various values of τ_b , τ_g , β and packet size. Fig.3 shows the long-run normalized throughput (γ_{norm}) as a function of τ_b/τ for a fixed $\rho = 10^{-3}$ and different packet sizes and values of β . The forward bandwidth is fixed at 1Mbps which results in $\mu_f = 125pkt/s$ and $\mu_f = 625pkt/s$ for 1000 bytes and 200 bytes packets, respectively. Other parameters are $\mu_r = 625pkt/s$ and $\tau = 200ms$. Moving from left to right in the plot, we go from *lossy connections* to *busy connections*. Smaller packet sizes yield slightly better performance for *busy connections* while large packet sizes allow for significantly better performance in *lossy connections*. The choice of B_f is only relevant in *busy connections* since in *lossy connections* the transmission rate is never high enough to build queues in the forward buffer, specially for small packet sizes. The horizontal dotted lines in the plot represent the analytical results using (2). The numeric results are presented in Table I and they show that the analysis is in accordance with the simulation outcome.

V. CONNECTIONS WITH $\alpha > 1$

The more relevant case where the reverse link is a bottleneck ($\alpha > 1$) is investigated in this section.

TABLE I

Throughput Saturation Limits for $\alpha \leq 1$.

β	Pkt Size	Eq. (2)	Simulation
0.1	200	0.801	0.802
	1000	0.766	0.770
0.2	200	0.854	0.859
	1000	0.819	0.826
0.8	200	0.984	0.991
	1000	0.960	0.979

As in the previous section, an expression for the saturation throughput is presented although details are omitted.

A. Saturation Throughput

When $\alpha > 1$, data packets are transmitted in the forward link in less time than it takes to transmit an ACK in the reverse link. By the time $W(t)$ reaches its maximum value W_{full} and a packet is dropped, the reverse buffer will have been overflowing already, allowing at this point for only one “surviving” ACK for every α ACKs generated by the receiver. W_{full} in this case is given by:

$$W_{full} = \mu_f \tau (1 + \beta) + \alpha B_r \quad (3)$$

Referring again to the analysis in [9], an expression for the throughput saturation limit is derived. The results are only valid for the particular case in which only one packet is dropped at the end of a *cycle* and the ACK for the retransmission is dropped at the reverse buffer, causing the sender to time-out. As discussed in section II-B, the authors in [1] find that this is particularly true when $B_f > 3\mu_f$. Fig.4 illustrates a typical window evolution for this case.

The analysis in this case is much more complex and will eventually lead to less reliable results due to necessary approximations. Again three different *RTT* estimates are obtained, T_4 , T_5 and T_6 . They correspond to the following three situations: (i) when a retransmission occurs (both links and buffers are fully utilized); (ii) an average *RTT* for packets transmitted in slow-start phase; (iii) an average *RTT* for packets transmitted in congestion avoidance phase. Also, an expression for α_{avg} , defined as the average number of packets acknowledged by each “surviving” ACK during congestion avoidance, is obtained. The expression for the saturation throughput as a function of T_4 , T_5 , T_6 , α_{avg} and W_{full} , normalized by the forward link

Fig. 4. Typical congestion window evolution for “busy connections” and $\alpha > 1$.

capacity μ_f is given by:

$$\gamma_{sat} = \frac{\sum_{i=1}^{\lfloor \log_2 \frac{W_{full}}{4} \rfloor} 2^i + \sum_{i=1}^{\lfloor \frac{3W_{full}}{4} \rfloor} \alpha_{avg} \left(\frac{W_{full}}{4} + i \right)}{\mu_f \tau_{tot}} \quad (4)$$

where τ_{tot} is the total duration of a *cycle* given by:

$$\tau_{tot} = T_4 + \frac{3}{\mu_r} + RTO + \log_2 \left(\frac{W_{full}}{4} \right) T_5 + \frac{3W_{full}}{4} \alpha_{avg} T_6 \quad (5)$$

RTO is an estimate for the retransmission time-out and μ_r is the reverse link capacity.

B. Simulation Results

Fig. 5. Normalized throughput (γ_{norm}) as a function of the normalized average bad state duration (τ_b/τ) for $\rho = 10^{-3}$. The horizontal dotted lines are results from equation (4).

Simulation traces were obtained using the same parameters as in Section IV-B only now with $\alpha = 3$ fixed (the forward bandwidth is also kept fixed at 1Mbps while μ_f and μ_r vary with the packet size). Results are

shown in Fig.5. Again *lossy connections* are on the left end of the plot while *busy connections* are on the right. The performance sensitivity to packet sizes and buffer sizes remains as described in Section IV-B only now asymmetry is responsible for an overall performance degradation. Among the six scenarios that were simulated, two of them do not satisfy $B_f > 3\alpha$. For 1000 bytes packet and $\beta = 0.1$ ($B_f < \alpha$) throughput is significantly lower than the analytical results, while for 1000 bytes packet and $\beta = 0.2$ ($\alpha < B_f < 3\alpha$) only slightly lower. Table II shows a comparison between analytical and simulation results for $\alpha \geq 1$. Fig.6

TABLE II
Throughput Saturation Limits for $\alpha > 1$.

β	Pkt Size	Eq. (2)	Simulation
0.1	200	0.641	0.620
	1000	0.577	0.274
0.2	200	0.696	0.736
	1000	0.621	0.588
0.8	200	0.876	0.900
	1000	0.787	0.832

illustrates how larger propagation delays impact the results and yield a narrower performance difference between the the two choices of packet sizes in *busy connections*. Only results for $\beta = 0.8$ are shown.

Fig. 6. Normalized throughput (γ_{norm}) as a function of the normalized average bad state duration (τ_b/τ) for different propagation delays and packet sizes ($\rho = 10^{-3}$).

VI. CONCLUSIONS

This paper provides some insights on the effects of choosing packets sizes for a TCP-Reno connection

over a channel with correlated errors. For the scenario where performance is dominated by congestion, analytical expressions were suggested and validated through simulations. Results show that performance sensitivity to the choice of packet size is reduced when the connection is dominated by congestion. In this case, connections using smaller packet sizes are able to achieve slightly higher throughput for both $\alpha \leq 1$ and $\alpha > 1$. When link losses dominate the connection, sensitivity to packet size is higher and larger packets allow for best results.

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