Traffic Rate Control for Real-Time Applications

Motivations

- Real-time applications need constant-rate delivery but are not stringent in reliability.
  - Teleconference, multimedia playback
- TCP Problems
  - Bursty delivery
  - exponential backoff interrupts regular delivery
- UDP Problem: no congestion control
  - unfair to TCP applications
  - the network may collapse (if no one backs off in congestion, no one's packets get thru)

TCP-Friendly Congestion Control

- In this talk, we discuss congestion control algorithms that are TCP-conformant in terms of bandwidth usages.
  - real-time applications must react to congestion in the same way as TCP so as not to disadvantage existing TCP applications.
- With such alogrithms, one can design realtime transport potocols for use by multimedia applications

Realtime Transport Protocols

- Rate-regulated traffic (no burst traffic)
- Relaxed reliability model
- Adjust traffic rate according to network congestion conditions (congestion control)
- Ideally, a real-time application adjusts its "quality" according to the current transmission rate.
  - Change frame rate/size, color depth, ...
  - Change audio bit rate

Performance Criteria for Rate-Control Algorithms

- ❑ Smoothness
  - – Difficult for multimedia applications to adapt if traffic rate go up and down too much
- ❑ Responsiveness
  - – How fast do we respond to developing congestion situations ?
  - – How fast do we take advantages of newly available resources ?

Two Approaches

- ❑ Additive increase, multiplicative decrease (**AIMD**)
  - – Follow the basic AIMD design of TCP
  - – Changes are made to produce smoother changes in traffic rates while maintaining TCP conformance in the long term.
- ❑ Equation-base rate control
  - – Estimate the packet loss rate *p.*
  - – Compute TCP sending rate according to *p*.

AIMD in TCP

- No packet loss, increase `cwnd` by 1 (or 1/*b*) per RTT.
- For each loss, decrease `cwnd` by half.
- The pure AIMD model is similar to TCP with only triple duplicate events.
- The pure AIMD model outperforms TCP for the `cwnd` is never set to 1.

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Generalized AIMD

- No packet loss, increase `cwnd` by α (or α/b) per RTT, $\alpha > 0$.
- For each loss, multiply `cwnd` by β, $0 < \beta < 1$.
- In TCP, $\alpha = 1$ and $\beta = 0.5$.
- **Question:** Are there other α-β combinations that produce the throughput of TCP ?
- Why ask the question ?
  - A larger-than-0.5 β value smooths out the fluctuation of caused by `cwnd` reduction.

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Analysis

- TCP performance formula can be generalized to

$$B_{\alpha,\beta}(p,RTT,T_0,b)=\frac{1}{\underbrace{RTT\sqrt{\frac{2b(1-\beta)p}{\alpha(1+\beta)}}}_{TD_{\alpha,\beta}(p,RTT,b)}+\underbrace{T_0\min\left(1,3\sqrt{\frac{b(1-\beta^2)p}{2\alpha}}\right)p(1+32p^2)}_{TO_{\alpha,\beta}(p,T_0,b)}}$$

- You can verify that

$$B_{1,0.5}(p,RTT,T_0,b)=\frac{1}{RTT\sqrt{\frac{2bp}{3}}+T_0\min\left(1,3\sqrt{\frac{3bp}{8}}\right)p(1+32p^2)}$$

is the performance of TCP with both TD and TO

Solving GAIMD Parameters

- We would like to find α and β values such that, for fixed *RTT*, T_0, and *b*, and for "most" *p* values:

$$B_{\alpha,\beta}(p,RTT,T_0,b)\approx B_{1,0.5}(p,RTT,T_0,b)$$

- Focusing on TD:

$$TD_{\alpha,\beta}(p,RTT,b)=TD_{1,0.5}(p,RTT,b)$$

$$\Rightarrow\quad RTT\sqrt{\frac{2b(1-\beta)p}{\alpha(1+\beta)}}=RTT\sqrt{\frac{2b(1-0.5)p}{1\times(1+0.5)}}$$

$$\Rightarrow\quad \frac{1-\beta}{\alpha(1+\beta)}=\frac{1-0.5}{\alpha(1+0.5)}\quad\Rightarrow\quad \alpha=\frac{3(1-\beta)}{1+\beta}$$

- Focusing on TO:

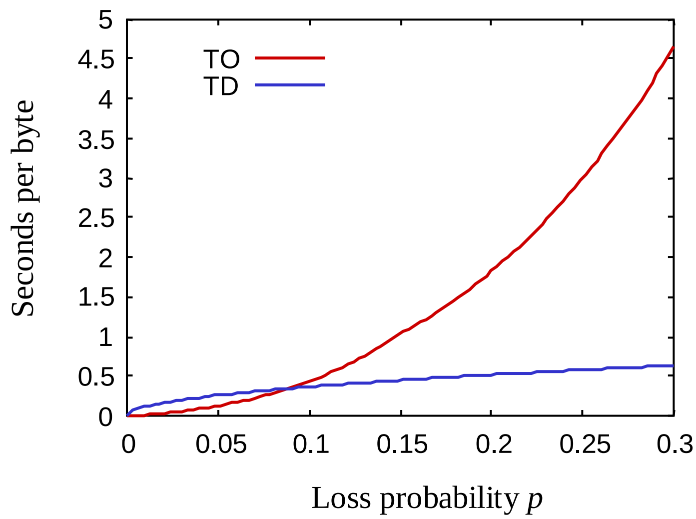
$$TO_{\alpha,\beta}(p,T_0,b) = TO_{1,0.5}(p,T_0,b)$$

$$\Rightarrow\ T_0 \min\left(1, 3\sqrt{\frac{b(1-\beta^2)p}{2\alpha}}\right) p(1+32p^2) = T_0 \min\left(1, 3\sqrt{\frac{b(1-0.5^2)p}{2}}\right) p(1+32p^2)$$

$$\Rightarrow\ \sqrt{\frac{1-\beta^2}{\alpha}} = \sqrt{\frac{1-0.5^2}{1}} \quad \Rightarrow\ \alpha = \frac{4(1-\beta^2)}{3}$$

- Example: if $\beta = 0.875$, then $\alpha = 0.3125$.

TD vs TO (RTT=1, T_0=4, b=2)



Discussion

- We use α and β values such that

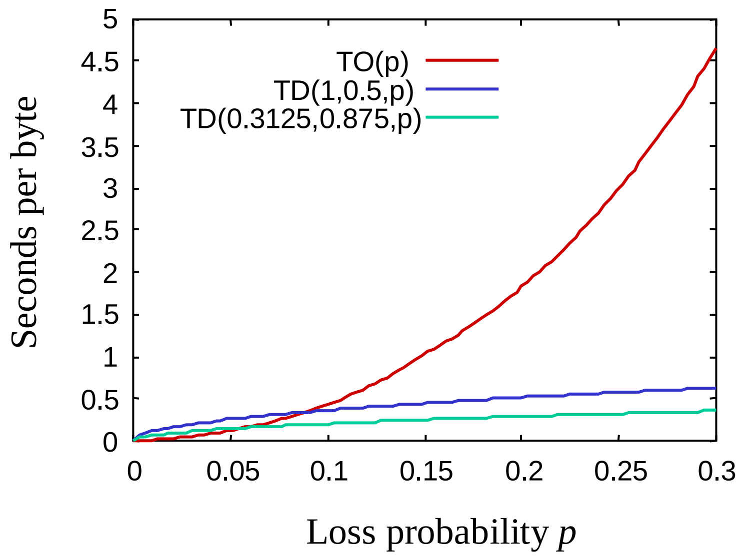
$$TO_{\alpha,\beta}(p,T_0,b) = TO_{1,0.5}(p,T_0,b)$$

- These α and β values will *not* satisfy

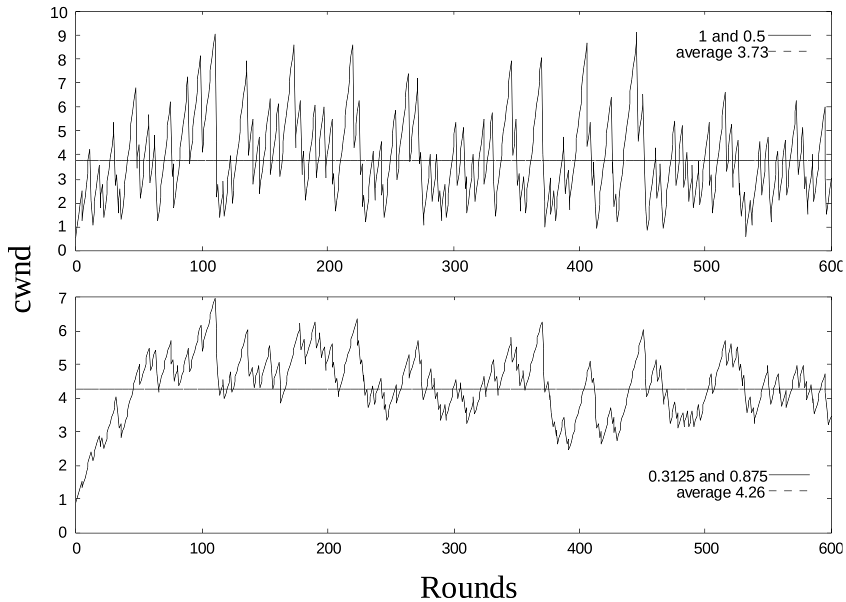
$$TD_{\alpha,\beta}(p,RTT,b) = TD_{1,0.5}(p,RTT,b)$$

- However, since TD are not important for most p values, the discrepancy is insignificant in the combined result $B_{\alpha,\beta}(p)$.
- For very small p, our choice of α and β tends to slightly over estimate throughput.

$\alpha = 0.3125$ and $\beta = 0.875$



Results (p=0.05, b=2)



Discussion

- We've discussed a method to produce TCP conformant traffic with smoother reactions to packet losses.
- This is important to multimedia applications to seamlessly adjust quality.
- However, as seen in the previous page, GIMD changes `cwnd` (thus traffic rate and in turn application quality) without changes in packet loss probabilities (that is, network condition is stable).

Equation-Based Rate Control

- Estimate p and RTT and computes $B(p,RTT,T_0,b)$, where T_0 and b are implementation parameters.
  - Of course, if p does not change, B is stable.
- Packets are transmitted at a rate determined by $B(p,RTT,T_0,b)$.
- We will discuss an equation-based approach proposed by Sally Floyd, called TCP-Friendly Rate Control (**TFRC**).

TFRC Protocol Overview

- Sender adds sequence numbers to packets.
- Receiver detects packet losses by gaps in sequence numbers.
- Receiver computes packet loss probability p.
- Receiver piggybacks p to ACK.
- Using ACK, sender computes RTT and T_0 in the same way as TCP.
- Using the packet loss probability p from the receiver, sender computes $B(p)$, and delivers packets at the rate set by $B(p)$.

- Packet transmissions are *not* ACK-clocked.
  - You don’t have to wait for an ACK to “release” the next packet.
- Rather packets are transmitted at fixed intervals according to $B(p)$.
- Packet losses are detected through ACK and retransmissions attempted after timeouts.
  - Retransmissions shall not violate $B(p)$.
  - A retransmission occupies a “slot” like regular transmissions.

Average Loss Rate

- We measure loss event rate, not packet loss rate.
  - What is the difference ?
- Define a loss interval be the number of packets between successive loss events.
- Computing loss event rate is equivalent to computing the average loss interval.
  - Can you see why ?

- Moving exponential average is "considered" too slow to reflect recent conditions.
- We need a way to give recent rounds high weights while still taking into account the past.

- Let s_i be the i-th sample of loss intervals with s_1 being the most recent sample and s_0 being the present, unfinished sample.
- We compute the average loss interval from the past n samples by

$$s(0,n-1)=\frac{\sum_{i=0}^{n-1} w_i s_i}{\sum_{i=0}^{n-1} w_i}, \quad w_i=\begin{cases} 1 & 0 \le i \le n/2-1 \\ 1-\frac{i+1-n/2}{n/2+1} & n/2 \le i < n \end{cases}$$

- Example: n = 8 gives the weights, from recent to past: 1, 1, 1, 1, 0.8, 0.6, 0.4, 0.2
- Finally, set loss probability $p = 1/s(0,n\text{-}1)$.

Properties of *s*(0,*n*–1)

- It is obvious that the sum of the first half of the weights is

$$\sum_{i=0}^{n/2-1} w_i = \frac{n}{2}$$

- The sum of the second half is

$$\sum_{i=n/2}^{n-1} w_i = \frac{n}{2} - \sum_{i=n/2}^{n-1} \frac{i+1-n/2}{n/2+1}$$

$$= \frac{n}{2} - \frac{1+2+\cdots(n/2)}{n/2+1}$$

$$= \frac{n}{2} - \frac{\frac{(n/2)(n/2+1)}{2}}{n/2+1} = \frac{n}{4}$$

- Thus, all the weights add up to (¾)×*n*.
- The effective weight w' of the present, unfinished sample s_0 is

$$\frac{1}{\frac{3}{4}\times n}$$

- For $n = 8$, we have $w' = 1/6$.

Rate Increase Per RTT

- For simplicity, we compute traffic rate by the equation bellow and set *b* to 1.

$$B(p) = \frac{1}{RTT}\sqrt{\frac{3}{2bp}}$$

- Let A=1/p be the average loss interval (in packets).
- $B(p)$ dedicates the transmission rate of

$$\frac{\sqrt{1.5}}{\sqrt{p}} \approx \frac{1.2}{\sqrt{p}} = 1.2\sqrt{A}$$

packets per second

- After a RTT of no loss, unfinished sample s_0 is increased by $1.2\sqrt{A}$
- Thus A is increased to $A + w'1.2\sqrt{A}$
- In turn, the transmission rate is increased to

$$1.2\sqrt{A + w'1.2\sqrt{A}} = 1.2\sqrt{A} + \Delta_B$$

- The increase in transmission rate per RTT without packet losses is

$$\Delta_B = 1.2\left(\sqrt{A + w'1.2\sqrt{A}} - \sqrt{A}\right)$$

- With w = 1/6, the increase in transmission rate is approximately 0.12 packets/RTT.

QoS Concerns

- Do the solutions discussed here address the Quality of Service (QoS) requirements of realtime applications ?