Equation-Based Congestion Control for Unicast Applications

Sally Floyd, Mark Handley
AT&T Center for Internet Research (ACIRI)

Jitendra Padhye
Umass Amherst

Jorg Widmer
International Computer Science Institute (ICSI)

Proceedings of ACM SIGCOMM, 2000

Outline

- Intro
- Foundations
- TFRC
- Experimental Evaluation
- Related Work
- Conclusions

Introduction

- TCP
  - Dominant on Internet
  - Needed for stability
  - AIMD
  - Window-based
- "Bulk-data" applications fine with TCP
  - But real-time find window fluctuations annoying
- Equation-based congestion control to the rescue!
  - Smooth the rate
  - (Note, class-based isolation beyond this paper)

But don't we need TCP?

- Practical
  - Primary threat are from unresponsive flows
    - Choose UDP over TCP
  - Give others protocol so they have something!
- Theoretical
  - Internet does not require reduction by ½
    - Other rates have been 7/8 (DECbit)
  - Even 'fairness' to TCP doesn't require this
  - Needs some control to avoid high sending rate during congestion

Guiding Basics for Equation-Based Protocol

- Determine maximum acceptable sending rate
  - Function of loss event rate
  - Round-trip time
- If competing with TCP (like Internet) should use TCP response equation during steady state
- There has been related work (see later sections) but still far away from deployable protocol
- This work presents one such protocol
  - TFRC

TFRC Goals

- Want reliable and as quick as possible?
  - Use TCP
- Slowly changing rate?
  - Use TFRC (ms. vs. s.)
- Tackle tough issues in equation-based
  - Responsiveness to persistent congestion
  - Avoiding unnecessary oscillations
  - Avoiding unnecessary noise
  - Robustness over wide-range of time scales
  - Loss-event rate is a key component!
- Multicast
  - If all receivers change rates a lot, never can scale

Foundations of Equation-Based Congestion Control

- *TCP-Friendly* Flow
  - In steady-state, uses no more bandwidth than conformant TCP running under same conditions
- One formulation:

$$T = \frac{s}{R\sqrt{\frac{2p}{3}} + t_{RTO}(3\sqrt{\frac{3p}{8}})p(1+32p^2)}$$

- s – packet size R – Round Trip Time
- p – loss event rate t_{RTO} – TCP timeout
- (Results from analytic model of TCP)

Outline

- Intro
- Foundations
- ***TFRC***
- Experimental Evaluation
- Related Work
- Conclusions

TFRC Basics

- Maintain steady sending rate, but still respond to congestion
- Refrain from aggressively seeking out bandwidth
  - Increase rate slowly
- Do not respond as rapidly
  - Slow response to one loss event
  - Halve rate when multiple loss events
- Receiver reports to sender once per RTT
  - If it has received packet
- If no report for awhile, sender reduces rate

Protocol Overview

- Compute p (at receiver)
- Compute R (at sender)
- RTO and s are easy (like TCP and fixed)
- Computations could be split up many ways
  - Multicast would favor 'fat' receivers
- TFRC has receiver only compute p and send it to sender
- Next:
  - Sender functionality
  - Receiver functionality

Sender Functionality

- Computing RTT
  - Sender time-stamps data packets
  - Smooth with exponentially weighted avg
  - Echoed back by receiver
- Computing RTO
  - From TCP: $RTO = RTT + 4 * RTT_{var}$
  - But only matters when loss rate very high
  - So, use: $RTO = 4 * R$
- When receive p, calculate new rate T
  - Adjust application rate, as appropriate

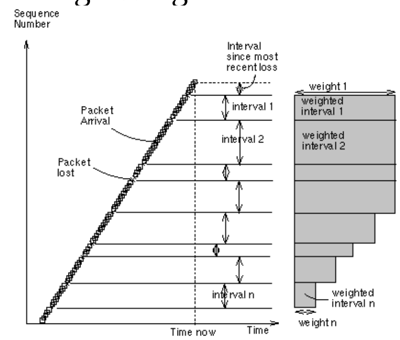
Receiver Functionality

- Compute loss event rate, p
  - Longer means subject to less 'noise'
  - Shorter means respond to congestion
- After "much testing":
  - *Loss event rate* instead of *packet loss rate*
    - + Multiple packets may be one event
  - Should track smoothly when steady loss rate
  - Should respond strongly when multiple loss events
- Different methods:
  - *Dynamic History Window, EWMA Loss Interval, Average Loss Interval*

Computing Loss Event Rate

- Dynamic History Window
  - Window of packets
  - Even at 'steady state' as packets arrive and leave window, added 'noise' could change rate
- Exponentially Weighted Moving Average
  - Count packets between loss events
  - Hard to adjust weights correctly
- Average Loss Interval
  - Weighted average of packets between loss events over last *n* intervals
  - The winner! (Comparison not in paper here)

Average Weighted Loss Intervals

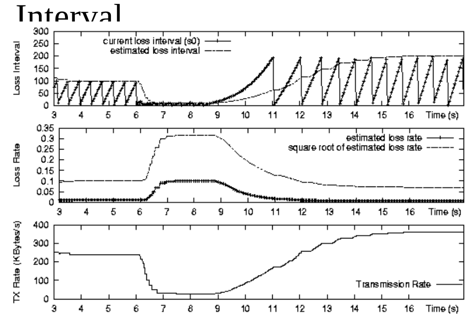


Loss Interval Computation

$$\hat{s}_{(1,n)} = \frac{\sum_{i=1}^{n} w_i s_i}{\sum_{i=1}^{n} w_i},$$

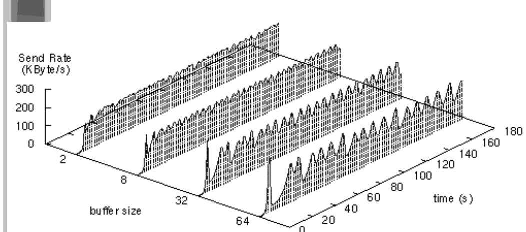
- w_i = 1 for 1 <= I <= n/2
- w_i = 1 – (I – *n*/2) / (*n*/2 + 1)
- 1, 1, 1, 1, 0.8, 0.6, 0.4, 0.2
- Rate depends upon *n*
  - *n* = 8 works well during increase in congestion (Later section validates)
  - Have not investigated relative weights
- History discounting for sudden decreases in congestion
  - Interval s_0 is a lot larger
  - Can speed up
- Loss event rate, *p*, is inverse of loss interval

Illustration of Average Loss Interval



Instability from RTT Variance

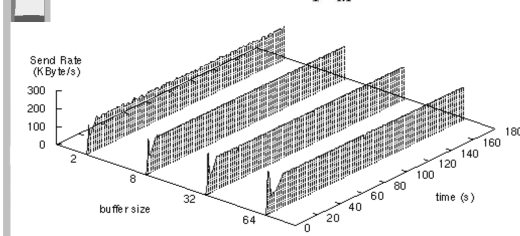
- Inter-packet time varies with RTT
  - Fluctuations when RTT changes



Improving Stability

- Take square root of current RTT (M is sqrt of average)

$$t_{inter-packet} = \frac{s}{T}\frac{\sqrt{R_0}}{M}$$



Slowstart

- TCP slowstart can no more than double congestion bottleneck
  - 2 packets for each ack
- Rate-based could more than double
  - Actual RTTs getting larger as congestion but measured RTTs too slow
- Have receiver send arrival rate
  - $T_{i+1} = \min(2T_i, 2T_{recv})$
  - Will limit it to double cong bwidth
- Loss occurs, terminate “slowstart”
  - Loss intervals? Set to ½ of rate for all
  - Fill in normally as progress

Outline

- Intro
- Foundations
- *TFRC*
  - Mechanics (done)
  - Discussion of features
- Experimental Evaluation
- Related Work
- Conclusions

Loss Fraction vs. Loss Event Fraction

- Obvious is packets lost/packets received
  - But different TCP’s respond to multiple losses in one window differently
    - Tahoe, Reno, Sack all halve window
    - New Reno reduces it twice
- Use loss event fraction to ignore multiple drops within one RTT
- Previous work shows two rates are within 10% for steady state queues
  - But DropTail queues are bursty

Increasing the Transmission Rate

- What if T_{new} is a lot bigger than T_{old}?
  - May want to dampen the increase amount
- Typically, only increase 0.14 packets / RTT
  - History discounting provides 0.22 packets / RTT
- Theoretical limit on increase
  - A is number of packets in interval, w is weight

$$\delta_T = 1.2\left(\sqrt{A + w1.2\sqrt{A}} - \sqrt{A}\right)$$

  - So … no need to dampen more

Response to Persistent Congestion

- To be smooth, TFRC does not respond as fast as does TCP to congestion
  - TFRC requires 4-8 RTTs to reduce by ½
- Balanced by milder increase in sending rate
  - 0.14 packets per RTT rather than 1
- Does respond, so will avoid congestion collapse
- (Me, but about response to bursty traffic?)

Response to Quiescent Senders

- Assume sender sending at maximum rate
  - Like TCP
- But if sender stops, and later has data to send
  - the previous estimated rate, *T*, may be too high
- Solution:
  - if sender stops, receiver stops feedback
- Sender ½ rate every 2 RTTs
- (Me, what about just a reduced rate that is significantly less than *T*?
  - May happen for coarse level MM apps)

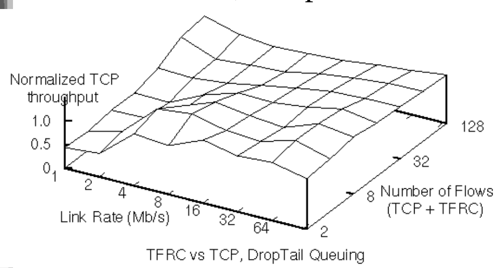
Outline

- Intro
- Foundations
- TFRC
- ***Experimental Evaluation***
  - Simulation
  - Implementation
    - + Internet
    - + Dummynet
- Related Work
- Conclusions

Simulation Results (NS)

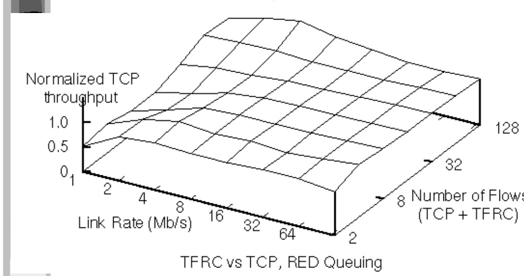
- TFRC co-exist with many kinds of TCP traffic
  - SACK, Reno, NewReno …
  - Lots of flows
- TFRC works well in isolation
  - Or few flows
- Many network conditions

TFRC vs. TCP, DropTail



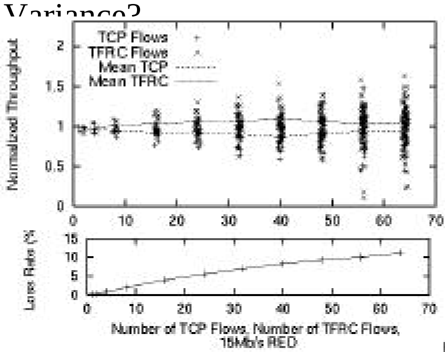
- Mean TCP throughput (want 1.0)
- Fair (?)

TFRC vs. TCP, RED



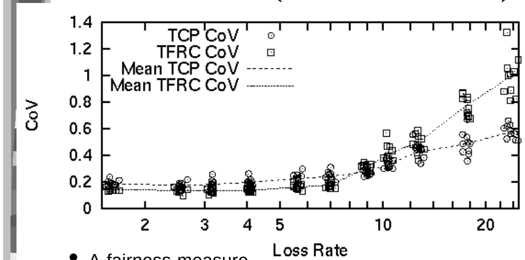
- Even more fair
- Not fair for small windows
- (Me … *bursty* traffic with *many* flows?)

Fair Overall, but what about Variance?



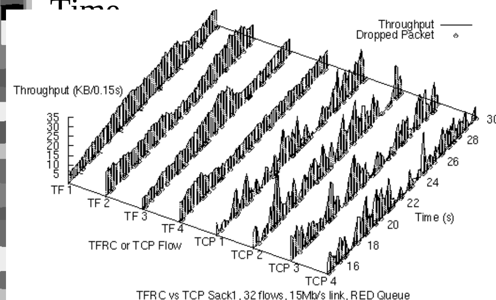
- Variance increases with loss rate, flows

CoV of Flows (Std Dev / Mean)



- A fairness measure
- Average of 10 runs
- TFRC less fair for high loss rates (above typical)
- Same w/Tahoe and Reno, SACK does better
  - timer granularity is better with SACK

Individual Throughputs over Time



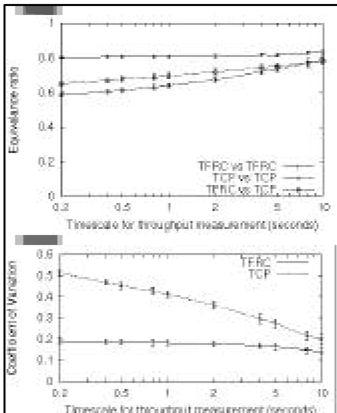
• .15 second interval (about multimedia sensitivity)

•Smoother rate from TFRC

Equivalence at Different Timescale

- Compare two flows
- Number between 0 and 1 (equation (4))
- Cases
  - Long duration flows in background
  - On-Off flows in background

Equivalence for Long Duration



•Single bottleneck
•32 flows
•15 Mbps link
•Monitor 1 flow
•95% confidence interval

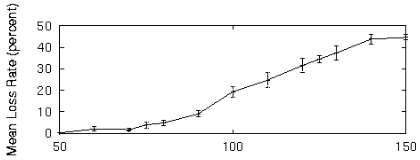
• Results hold over Broad range of timescales

Outline

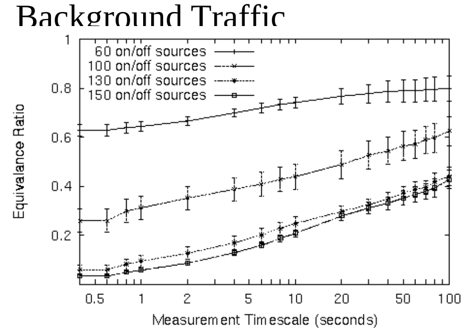
- Intro
- Foundations
- TFRC
- *Experimental Evaluation*
  - Simulation
    - + Fairness and Smoothness (CoV) (done)
    - + Long Duration (done)
    - + On-Off flows
  - Implementation
- Related Work
- Conclusions

Performance with On-Off Flows

- 50 – 150 On/Off UDP flows
  - On 1 second, off 2 seconds (mean)
  - Send at 500 kbps rate
- Monitor TCP, Monitor TFRC



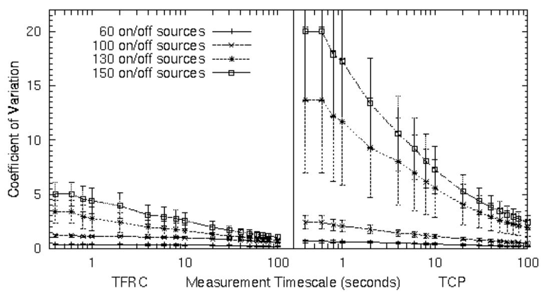
Equivalence with TCP with Background Traffic



•At high loss rates, less equivalent (40% more, less)

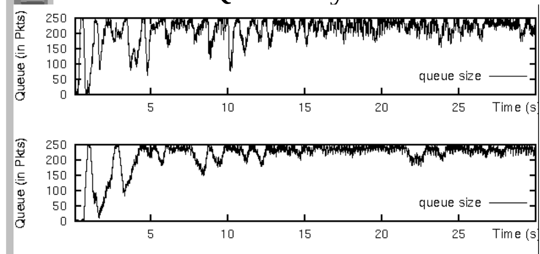
•(Me, room for improvement)

CoV with Background Traffic



•TFRC rate has less variance, especially at high loss rates

Effect on Queue Dynamics



•40 flows, staggered start times
•TCP (top) has 4.9% loss and TFRC (bottom) has 3.5% loss
•99% utilization for all (Bursty?)
•Basically, look the same
•Extensive tests, w/RED and background look the same

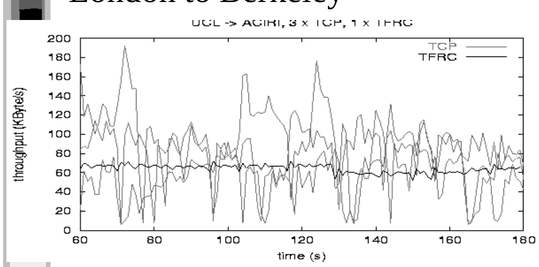
Outline

- Intro
- Foundations
- TFRC
- ***Experimental Evaluation***
  - Simulation (done)
  - Implementation
    - + Internet
- Related Work
- Conclusions

Implementation Results

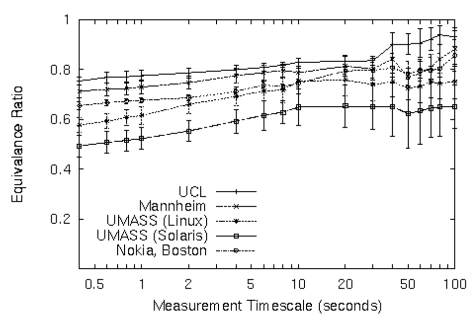
- TFRC on Internet
  - Microwave
  - T1
  - OC3
  - Cable modem
  - Dialup modem
- Generally fair
- (See tech report for details)

London to Berkeley

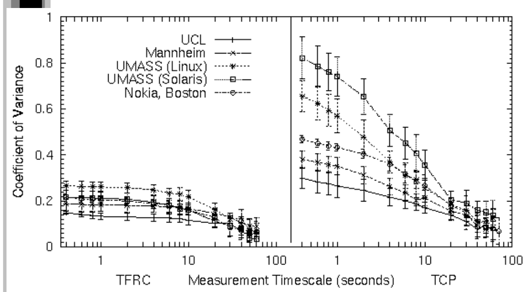


• 3 TCP flows, 1 TFRC flow
• TFRC slightly lower bandwidth but smoother
• Typical loss rates .1% to 5%

TCP Equivalence over Internet



CoV over Internet

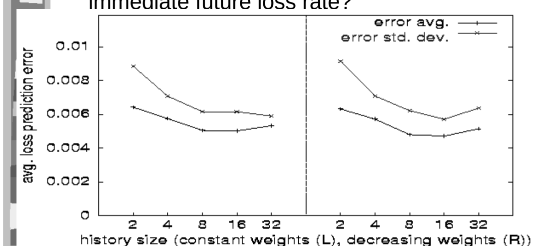


TFRC unfair to TCP when …

- When flows have one packet per RTT
  - TFRC can get far more than its fair share
  - Due to 'conservative' clock (500ms) in FreeBSD?
- Some TCP variants are 'buggy'
  - Linux vs. Solaris
  - (Me, a neat project)
- Real-world "Phase Effect" (?)

Testing the Loss Predictor

- How effective do *X* intervals predict immediate future loss rate?



- But not just great prediction but reaction, too

Related Work

- TCP Emulation At Receiver (TEAR)
  - Compute window at receiver, convert to rate
- Rate Adaptation Protocol (RAP)
  - AIMD approach
  - No slow start, no timeout
- Other equation based
  - One ties with MPEG (application)
  - One TFRCP direct comparison

Issues for Multicast Congestion Control

- Still feedback every RTT
  - Must change to aggregate or hierarchical
  - Or lowest transmission rate
- Slowstart especially problematic as needs very timely feedback
- Synchronized clocks needed so receivers can determine RTT in scalable manner

Conclusions

- TFRC gives TCP-fair allocation of bandwidth over wide range of environments
- TFRC smoother than TCP
- Evaluated over wide range of network conditions

Future Work?

Future Work

- What is some retransmission?
  - How to divide up T
- What if some extra repair information?
  - How to divide up T?
- Duplex TFRC?
- ECN and TFRC?