TCP Westwood: Efficient Transport for High-speed wired/wireless Networks

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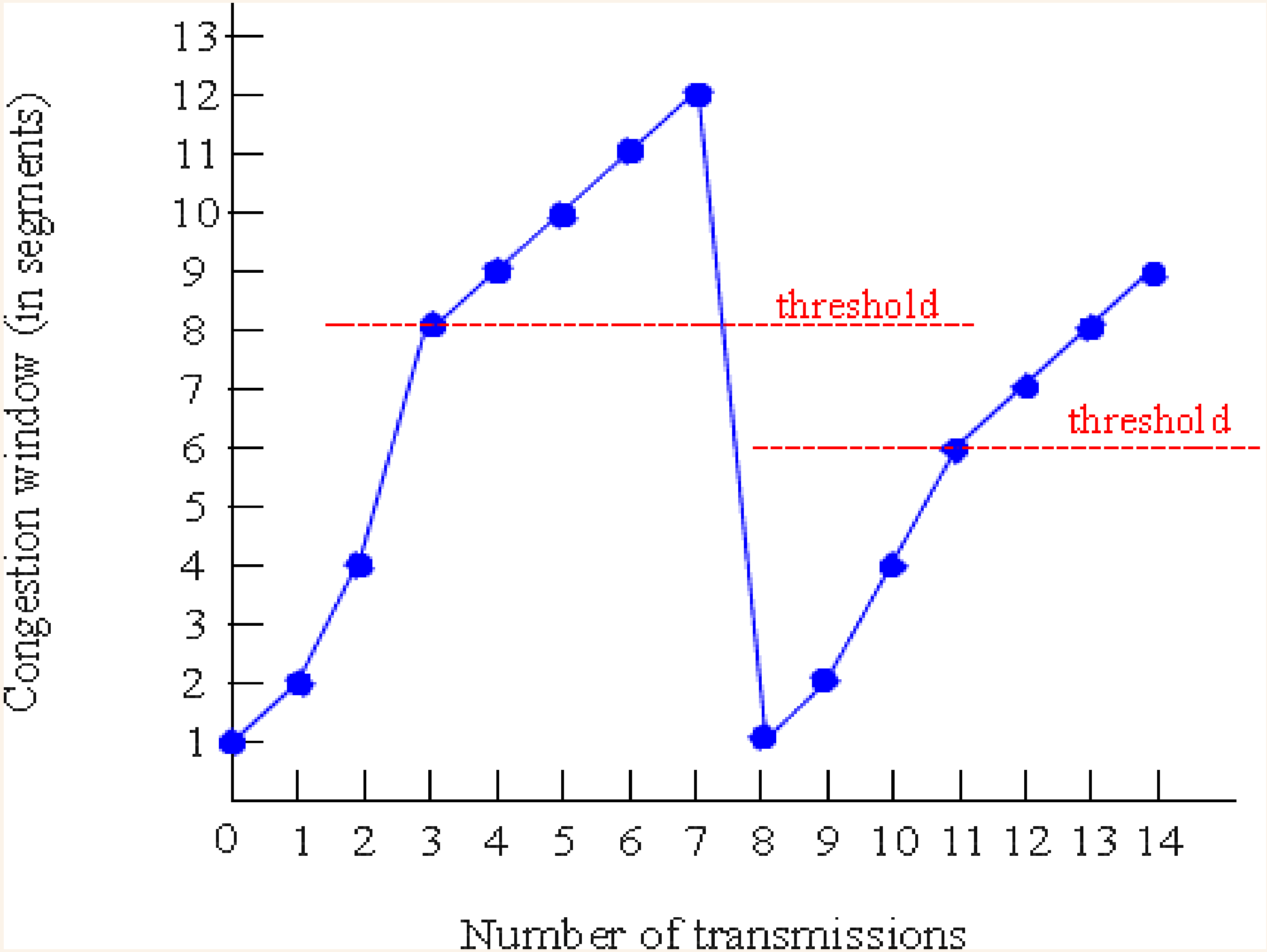
Outline

1. TCP Overview
2. Bandwidth Estimation and TCP Westwood
3. Bandwidth Estimation vs. Rate Estimation
4. Performance Evaluation

TCP Congestion Control

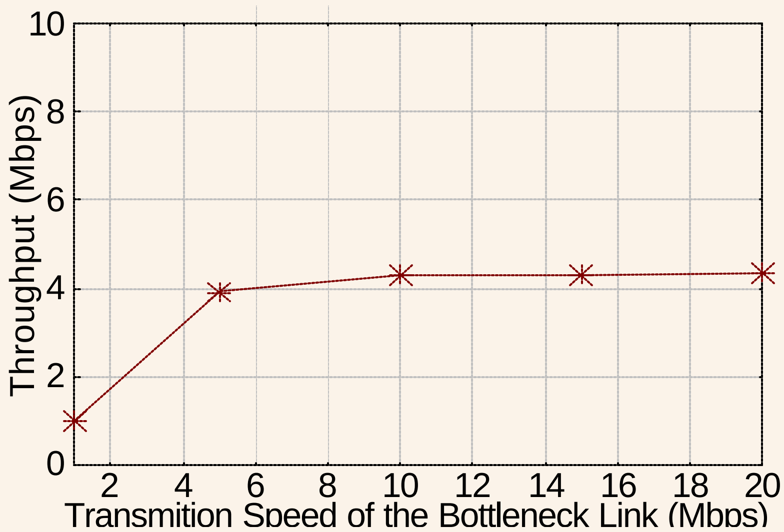


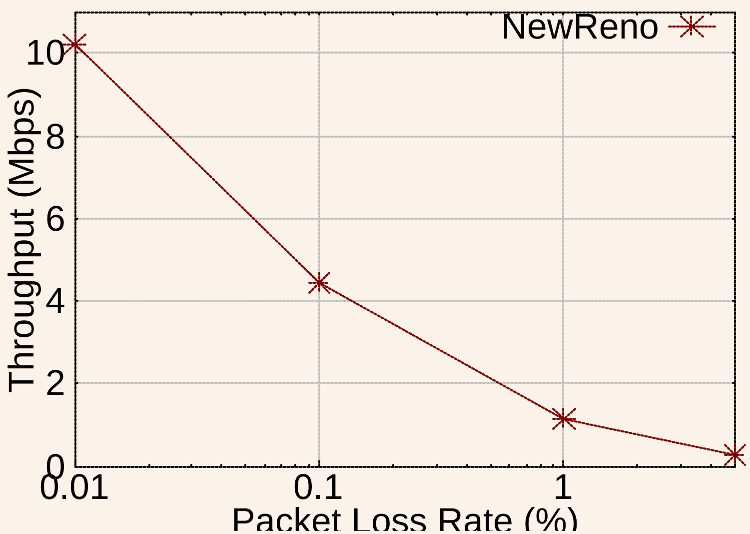
TCP Congestion Control Overview

- Evolved over the years: *Tahoe*, *Reno*, *NewReno*, *SACK*
- *Window* based, window size => offered traffic rate
- *Probing*: a connection “probes” for available bandwidth, when perceives packet loss, backs down to slower rate
- Two *phases* with differing probing behavior
- Initial design assumes packet losses are almost all due to *buffer overflow*
- Network layer assistance (*RED*, *ECN*, *XCP*, *BA-TCP*) for better efficiency, fairness and stability
- Link layer assistance has been suggested for hybrid networks where packets loss can be caused by both (1) *random error*, and (2) *buffer overflow*

TCP Reno Limitations (Ren, please clarify…)

- In wireless (lossy) networks random packet loss causes unnecessary window reduction and thus *inefficiency*
- In High speed networks blind halving of cwnd also results in *inefficiency* (why?)





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TCP Westwood (2000)

Key Idea:

- Enhance congestion control via the Rate Estimate (*RE*)
  - Estimate is computed at the sender by *sampling* and *exponential filtering*
  - Samples are determined from ACK *inter-arrival times* and info in ACKs regarding amounts of *bytes delivered*
- RE is used by sender to properly set *cwnd* and ssthresh after packet loss (indicated by 3 DUPACKs, or Timeout)

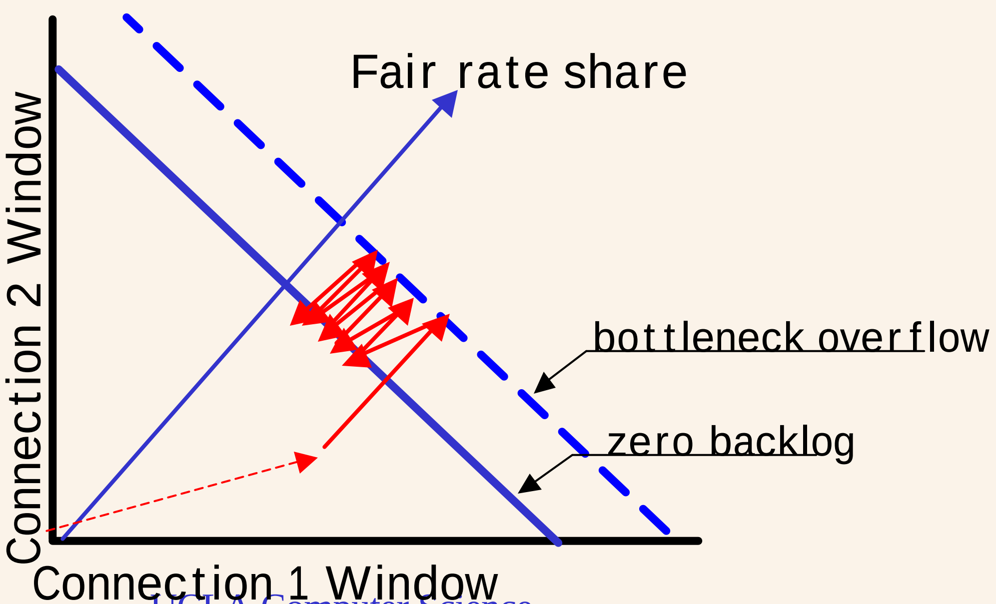
TCP Westwood: the control algorithm

- TCPW Algorithm Outline:
  - When three duplicate ACKs are detected:
    - set `ssthresh=RE*RTTmin` (instead of `ssthresh=cwin/2` as in Reno)
    - if `(cwin > ssthresh)` set `cwin=ssthresh`
  - When a TIMEOUT expires:
    - set `ssthresh=RE*RTTmin` (instead of `ssthresh=cwnd/2` as in Reno) and `cwin=1`

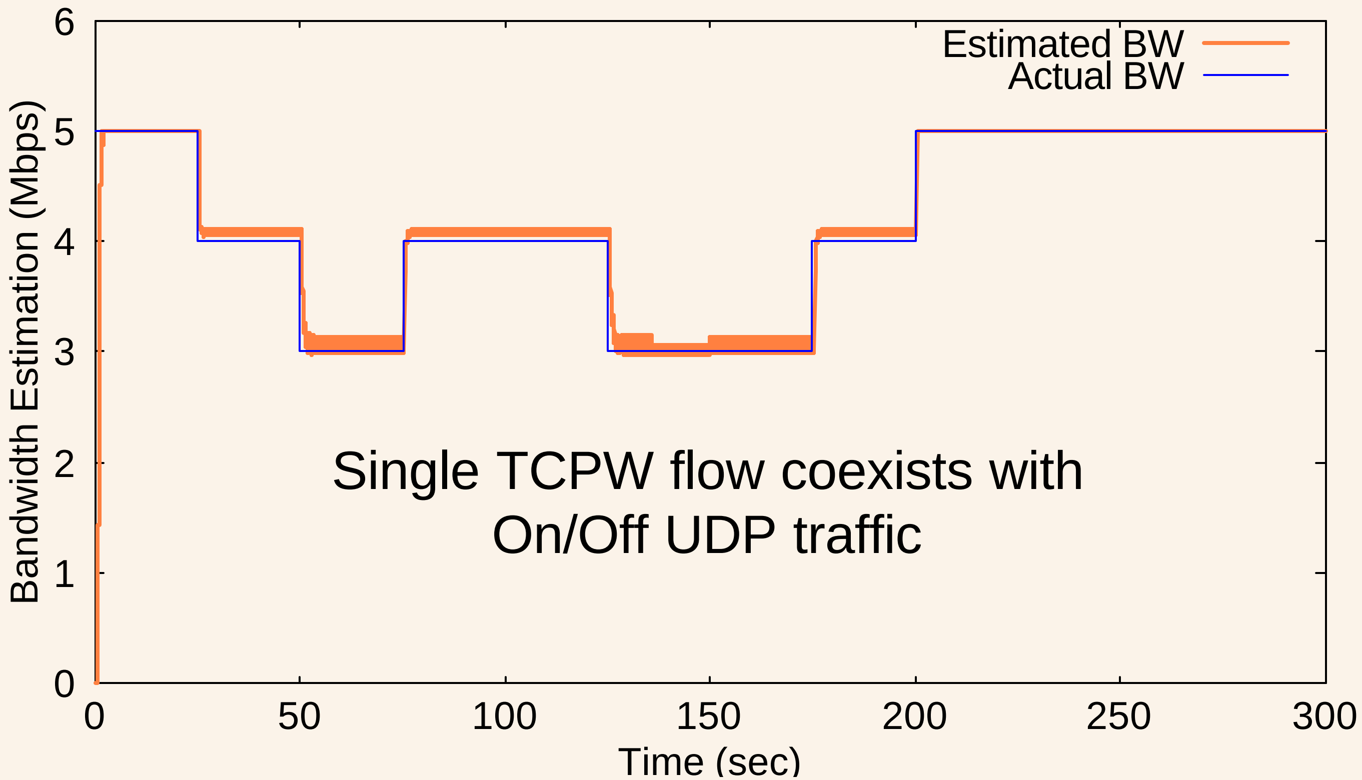
Note: RTTmin = min round trip delay experienced by the connection

At equilibrium, RE -> Fair RE

- Initially, two connections have different Wi and Ri.
- In the increase phase windows grow at the same rate
- Just before overflow : **Wi = Ri (Buf/Cap + RTTm) for i = 1,2**
- At overflow, RE estimate reduces windows back to "zero backlog" line, ie: **Wi = RE RTTm = Ri RTTm**



Fair RE = "Residual Bandwidth" Estimate?



Single TCPW flow at equilibrium does estimate "residual bottleneck bandwidth"

Related TCP + Bdw estimation work

- Note: the concept of using the bandwidth estimate to control the TCP flow is not new
- **TCP Vegas** monitors Bdw and RTT to infer the *bottleneck backlog*; then, from backlog it derives feedback to congestion window
- **Keshav's Packet Pair** scheme also monitors bandwidth to estimate the *bottleneck backlog* and compare to common target; it adjusts source rate

Related Works (cont)

- TCP Vegas
  - Sender watches for some sign that router's queue is building up and congestion will happen; e.g.,
    - RTT grows
    - sending rate flattens
  - Sender adjust sending rate to avoid filling the buffer
  - Fairness problem has been reported
- Packet Pair Flow Control
  - Using Packet Pair method to estimate bottleneck service rate to a connection
  - Adjusts the **transmitting rate** to maintain the TCP connection bottleneck queue equal to a target called **setpoint** (B)
  - Under round-robin, packet pair measures fair share; otherwise measure is inaccurate, and can overestimate fair share, up to link capacity

TCPW Benefits

What do we gain by using **RE** “feedback” in addition to **packet loss**)?

(a) better performance with **random loss** (ie, loss caused by random errors as opposed to overflow)

(b) ability to distinguish random loss from buffer loss

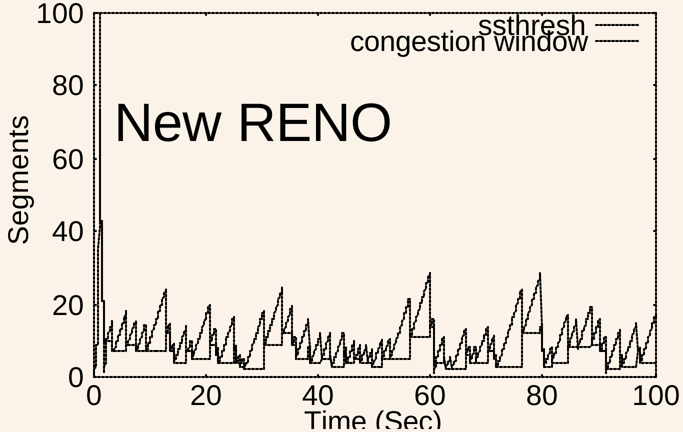
(c) using RE to **estimate** bottleneck bdw during slow start

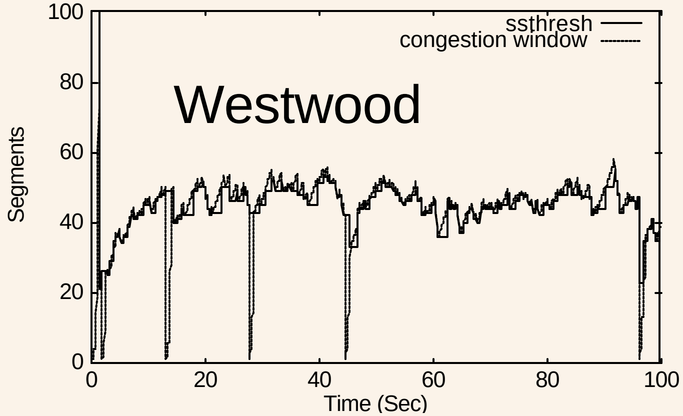
TCPW and random loss

- Reno overreacts to random loss (**cwin** cut by half)
- TCPW less sensitive to random loss
- a small fraction of "randomly" lost packets minimally impacts the rate estimate RE
- Thus, **cwin** = RE x RTT remains unchanged
- As a result, TCPW throughput is higher than Reno and SACK

TCPW And Random Loss

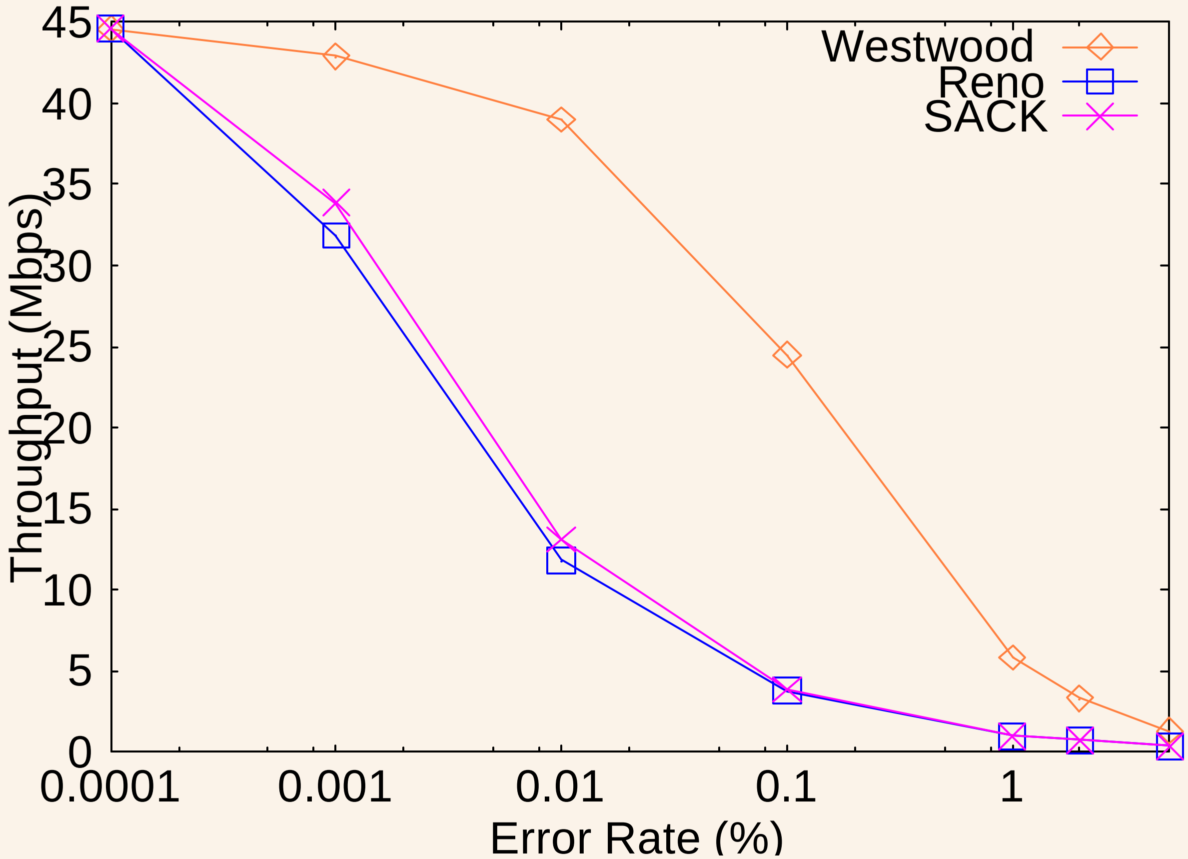
Cwnd and ssthresh of TCPW and NewReno under random losses:





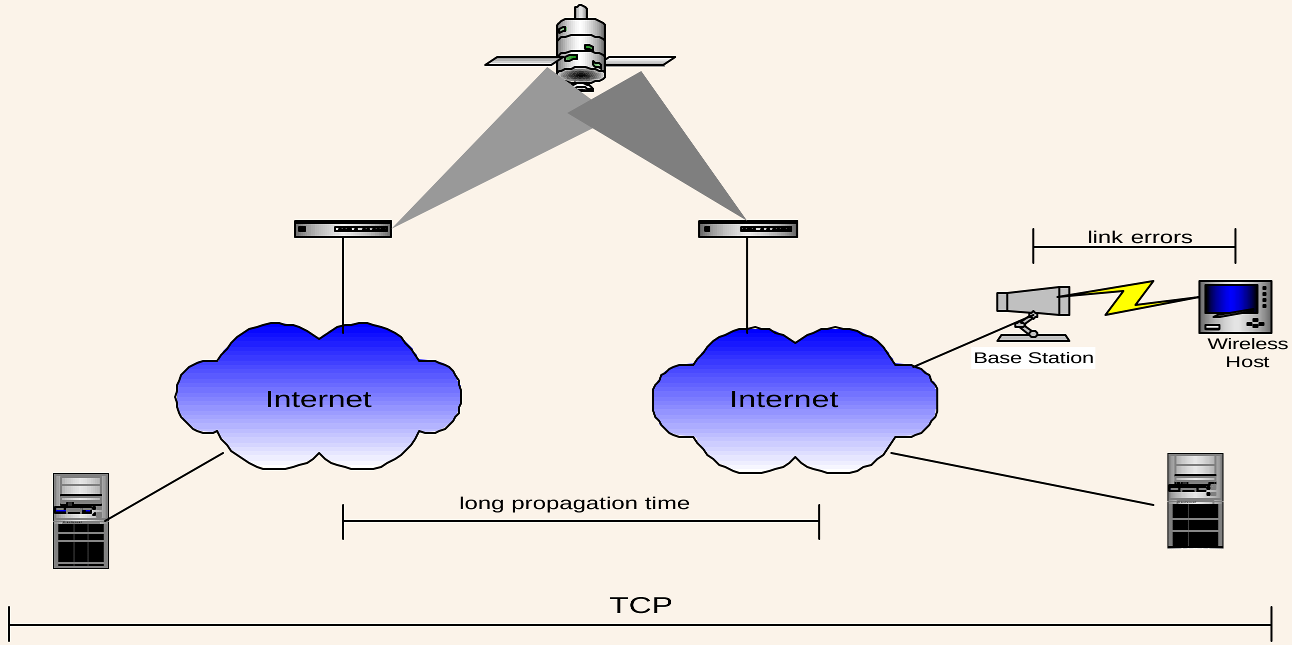
- NewReno overreacts to random loss (cwin cut by half)
- A small fraction of isolated "randomly" lost packets does not impact the RE estimate
- Thus, *cwnd = RE * RTTmin* remains unchanged
- As a result, TCPW efficiency is higher than NewReno and SACK

TCPW in “lossy” environment



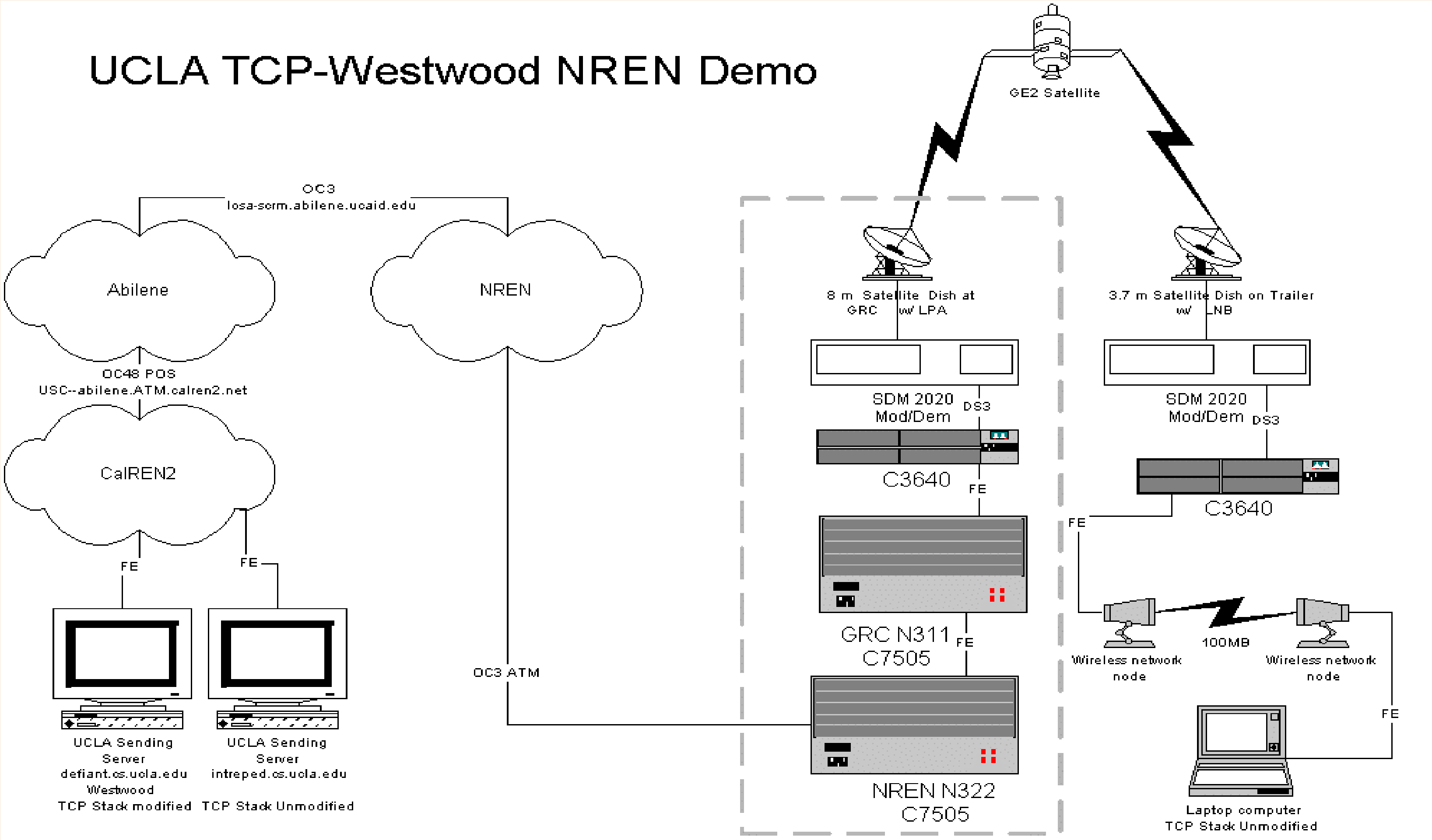
TCPW in a wireless lossy environment

- *Efficiency*: Improvement significant on high (Bdw x Length) paths

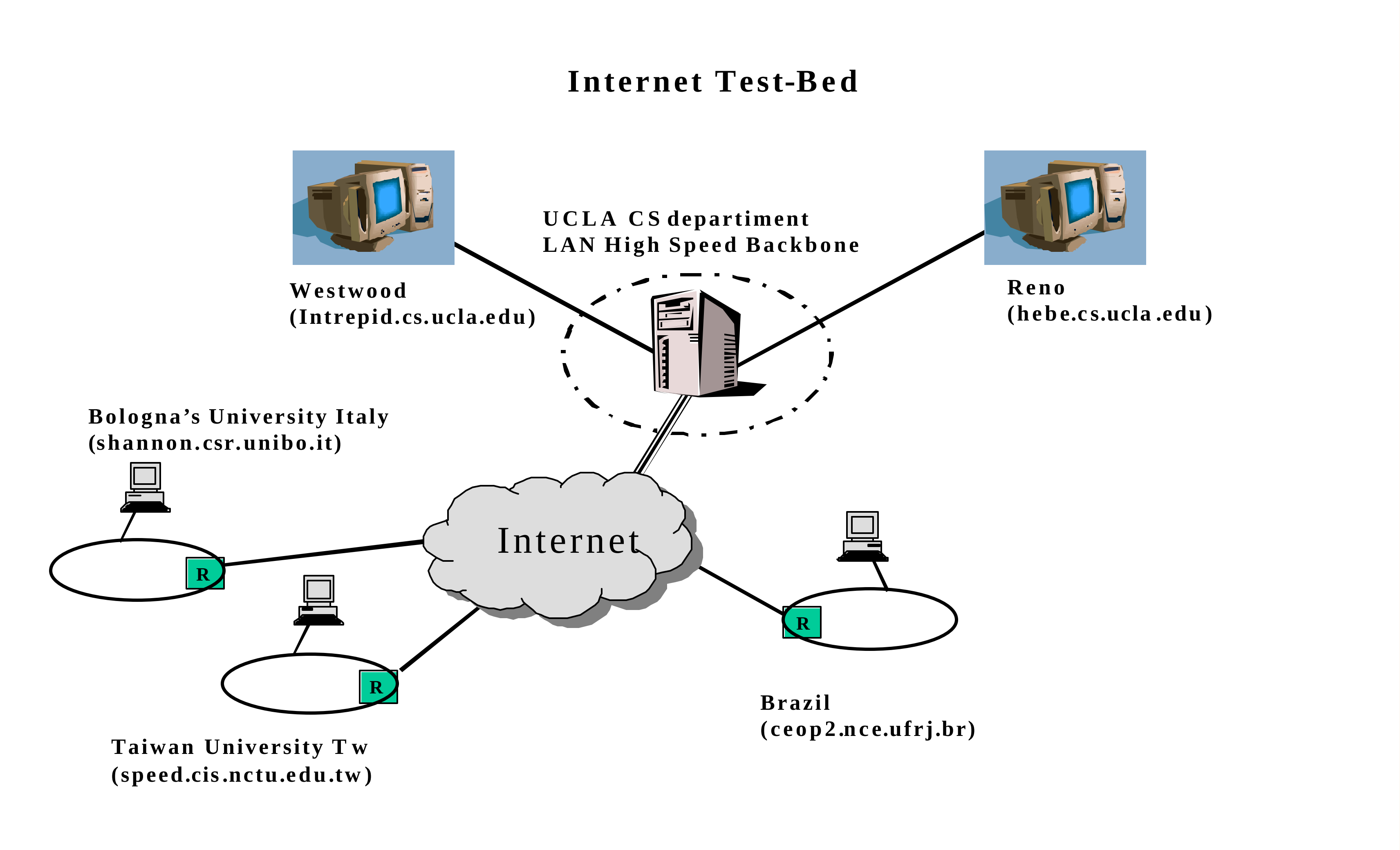


- *Fairness*: better fairness than RENO under varying RTT
- *Friendliness*: TCPW is friendly to TCP Reno

NASA Workshop Demo
(From Steve Schultz, NASA)

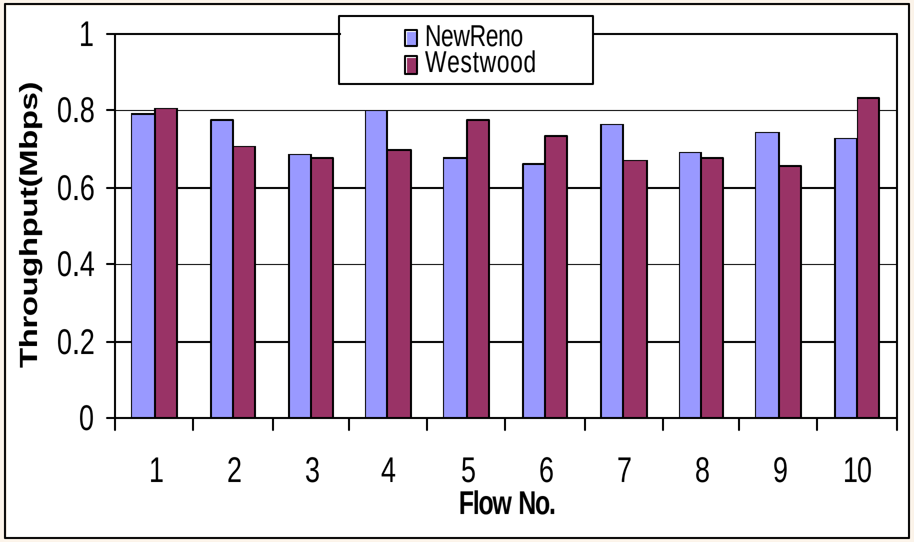


Internet Measurements Testbed



TCPW Fairness

- **Fairness**: how equitably is bandwidth shared among same flavor TCP flows?
  - Internet experiment with 10 TCPW and 10 TCP NR
  - Jain's index for this experiment is ???

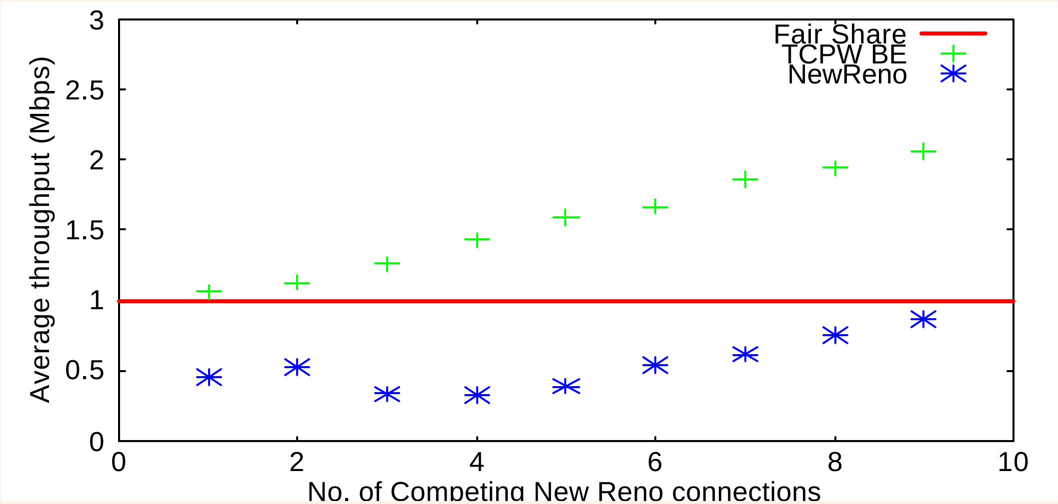


TCPW Friendliness

- **Friendliness**: fairness across different TCP flavors

  "Friendly share" principle: *TCPW is allowed to recover the bandwidth wasted by NewReno because of "blind" window reduction*

- TCPW **original RE filter** has Friendliness Problem….
  - 10 connections total (TCPW + RENO) ; No random errors
  - Average throughput per connection is shown below:



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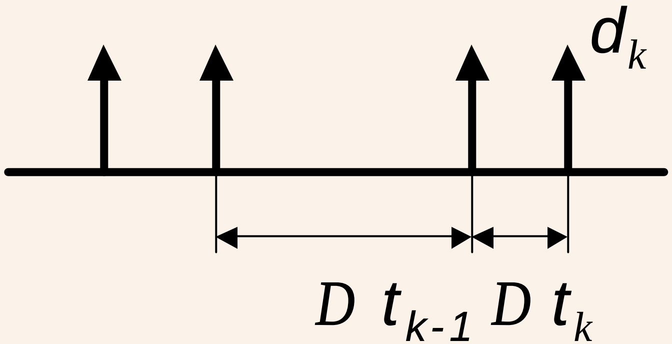
TCPW original estimation (BE)



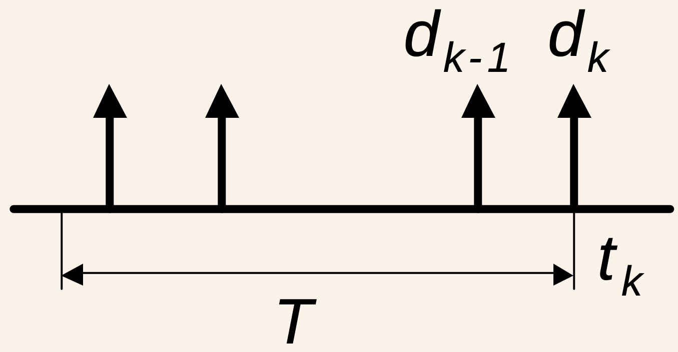
- First TCPW version (referred to as: TCPW BE) used a “bandwidth like” estimator (BE) given by:

$$b_k = d_k / (t_k - t_{k-1})$$ sample

$$BE_k = a_k BE_{k-1} + (1 - a_k)\left(\frac{b_k + b_{k-1}}{2}\right)$$ exponential filter

$$a_k = \frac{2t - \Delta t_k}{2t + \Delta t_k}$$ filter gain

TCPW Rate Estimation (TCP RE)



T is the sample interval

- Rate estimate (RE) is obtained by aggregating the data ACKed during the interval T (typically = RTT):

$$b_k = \frac{\sum_{t_j > t_k - T} d_j}{T}$$ sample

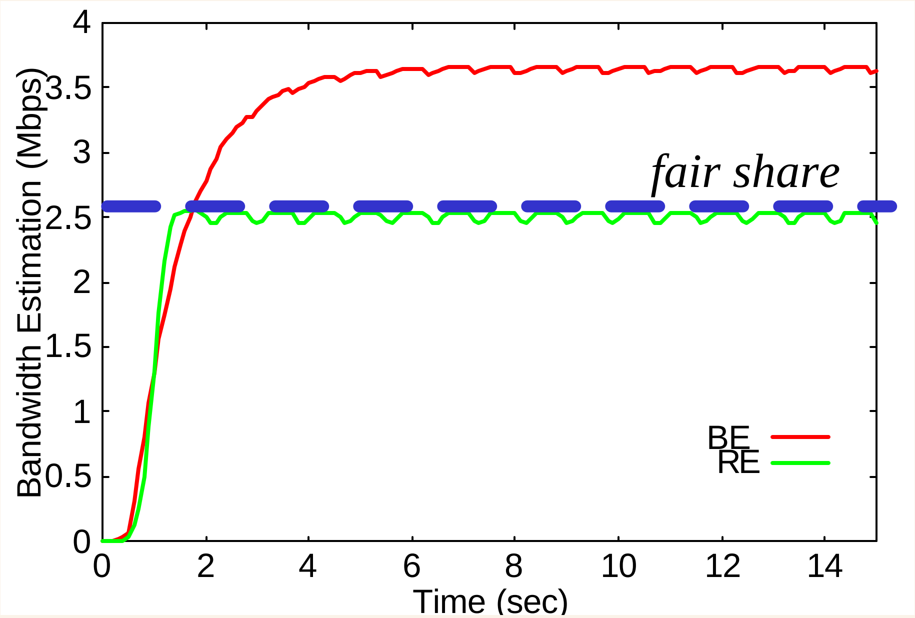
$$RE_k = a_k RE_{k-1} + (1 - a_k)\left(\frac{b_k + b_{k-1}}{2}\right)$$ exponential filter

$$a_k = \frac{2t - \Delta t_k}{2t + \Delta t_k}$$ filter gain

TCPW RE/BE interaction with RENO

One TCPW RE/BE and one Reno share a 5Mbps bottleneck

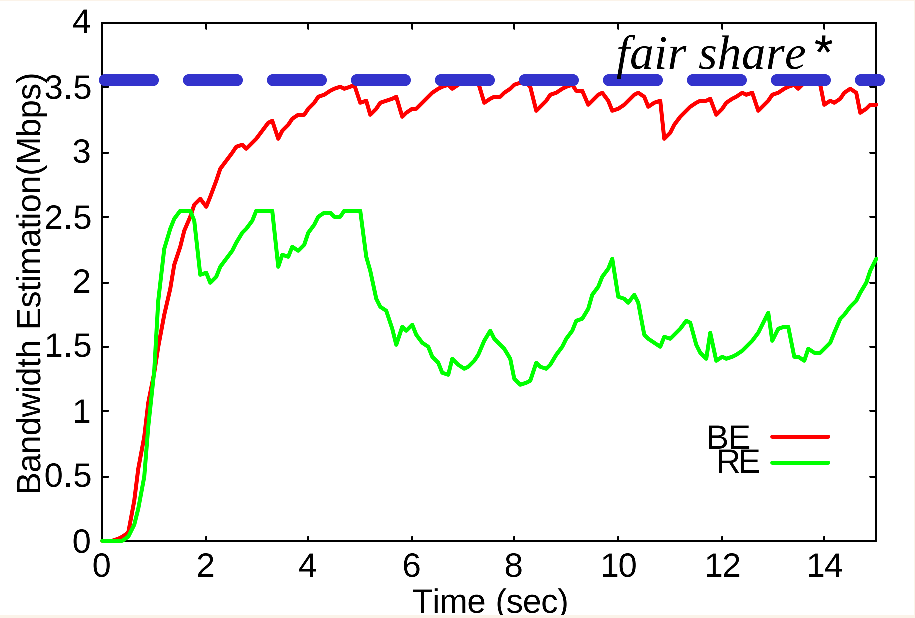
No errors (bottleneck gets saturated)



BE overestimates fair rate (= 2.5 Mbps)

TCPW BE Not friendly to NewReno!

Errors (0.5%), no congestion



RE underestimates fair rate (=3.6 Mbps)

TCPW RE does not improve thruput!

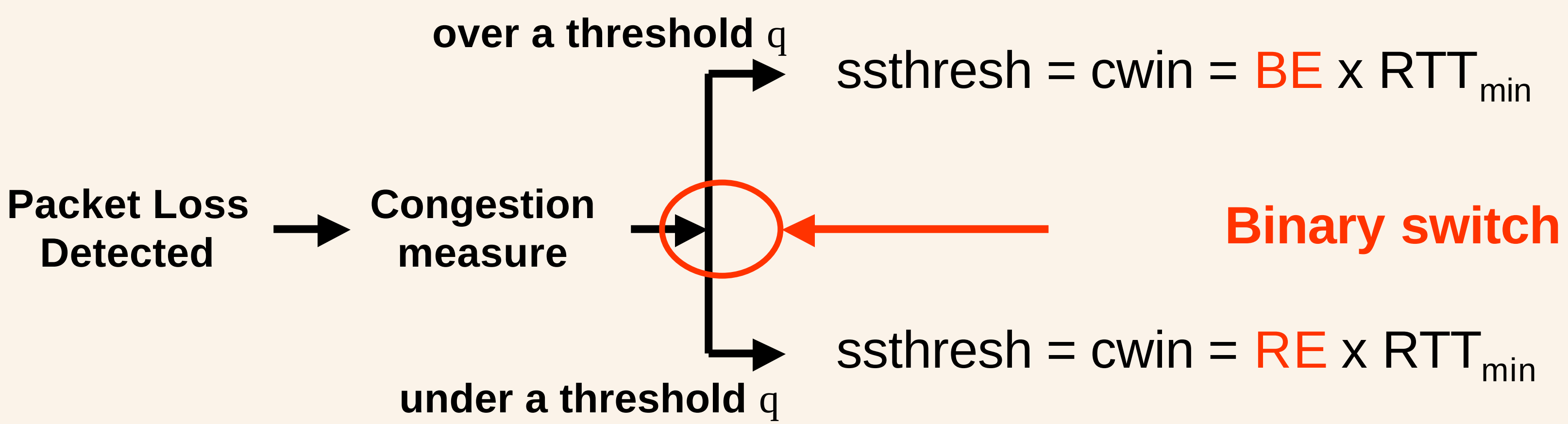
(*) TCPW fair share > 50% because NewReno is incapable of getting 50%

TCPW Adaptation

- Neither RE or BE estimator are optimal for all situations
  - BE is more effective in random loss
  - RE is more appropriate in congestion loss (ie, buffer overflow)
- KEY IDEA: dynamically select the aggressive estimate (BE) or the conservative estimate (RE) depending on current channel status (congestion or random loss?)
- NEEDED: a "congestion measure" that gives us an idea of the most probable cause of packet loss (congestion or random)

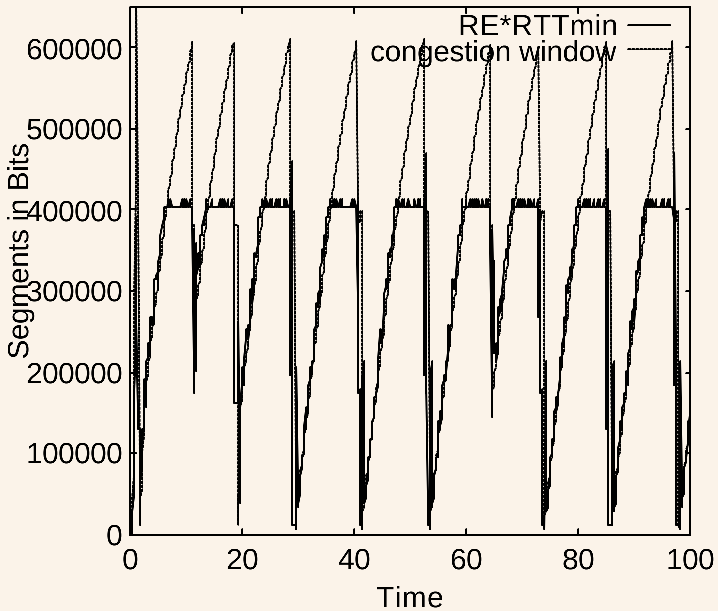
Combining Rate and Bandwidth estimations: TCPW CRB

- **TCPW CRB** chooses between RE or BE upon packet loss to set the ssthresh

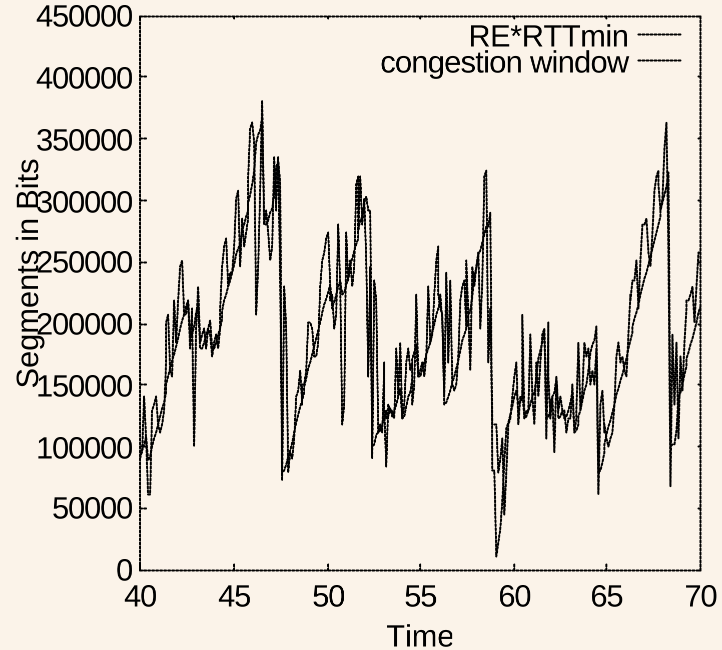


Congestion Measure

- IF cwnd/ RTTmin (ie, **max** achievable rate) is larger than vs. RE (**currently** achieved rate) the channel is congested;
- if **max** is equal to **current** rate, the loss is random loss



Congested Channel



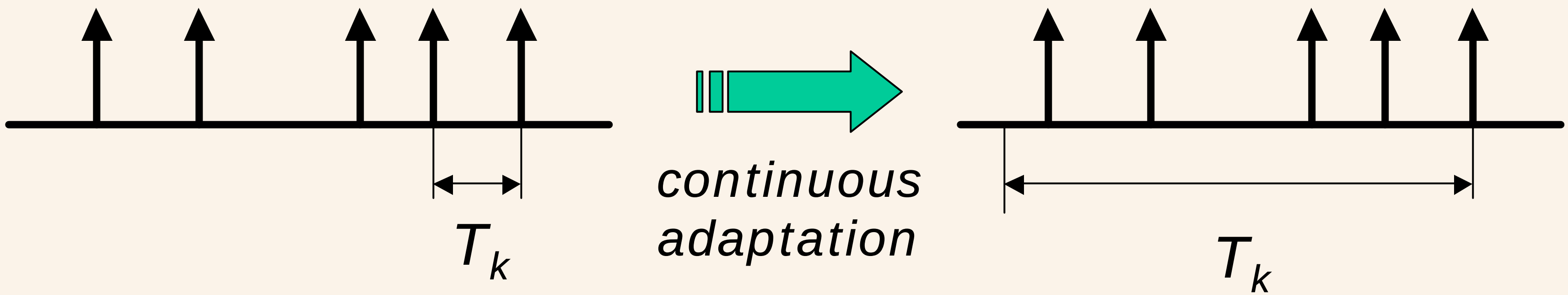
Random loss channel

TCPW with continuous filter adaptation

- Next step is to have a *continuous* instead of *switched* filter adaptation
- IDEA:
  - adapt continuously the sample size according to congestion level
  - adapt continuously the filter agility according to network instability
- In TCPW AF (Adaptive Filtering) we adapt the sample interval T_k according to current measured congestion level
  - T_k ranges from T_k = *inter ACK interval* to T_k = *RTT*
- Filter agility (more or less weight on history) must be limited so that it does not overreact to network jitter

TCPW AF: Sampling

- Adapting the size of sampling intervals to congestion level measure



No Congestion: T_k *= inter ACK*   *Congestion:* T_k *grows*

$$s_k = \frac{\sum_{t_j > t_k - T_k} d_j}{T_k}$$

Rate sample

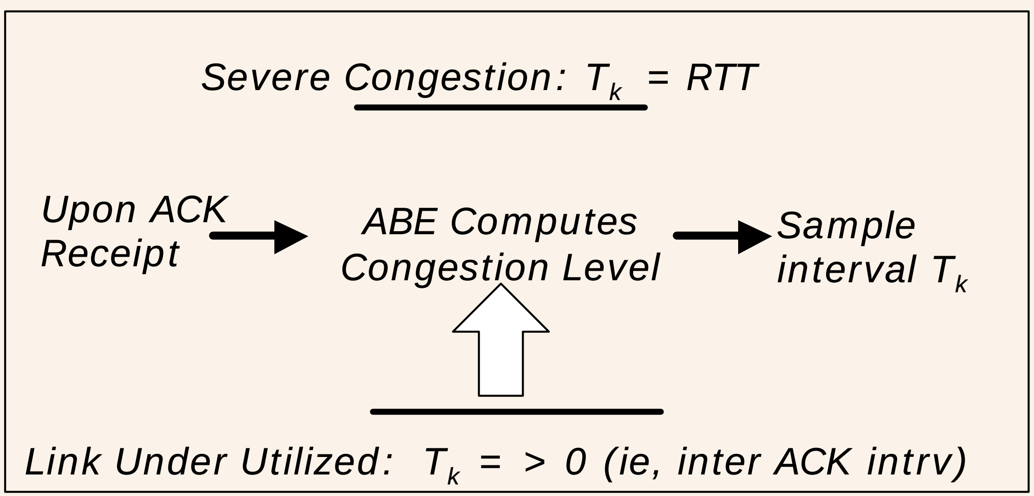
TCPW AF: Sampling (cont)

- The sample size T_k is continuously adjusted according to current congestion measure:

$$T_k = RTT * \left(\frac{cwin}{RTT_{min}} - \hat{Th}_k\right) / \frac{cwin}{RTT_{min}}$$

Max throughput assuming there is no congestion in the network

actual achieved throughput



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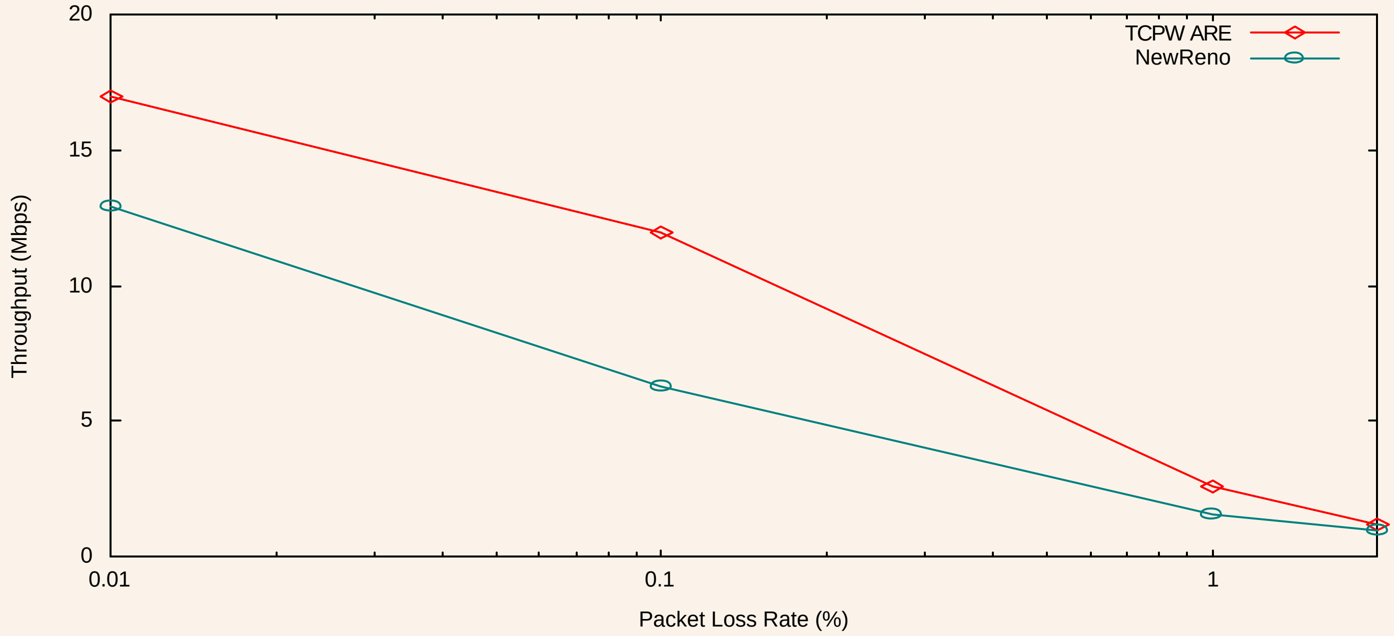
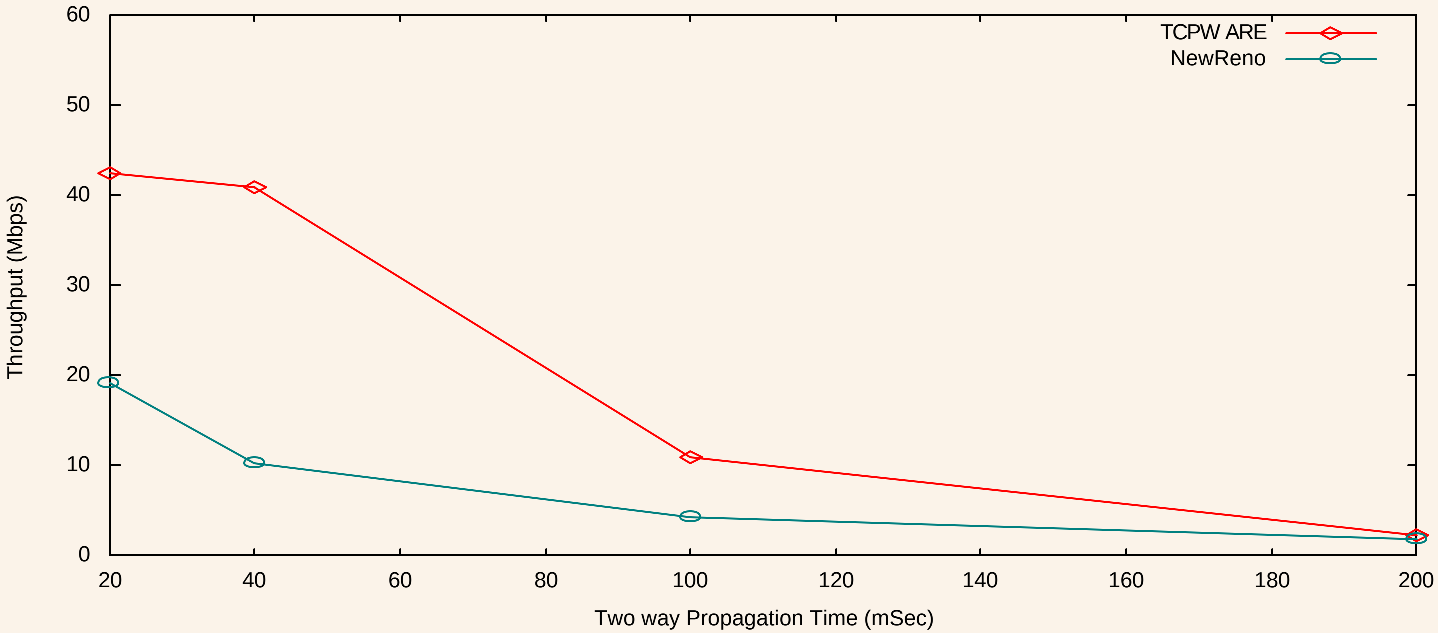
TCPW AF Simulation Results (1)



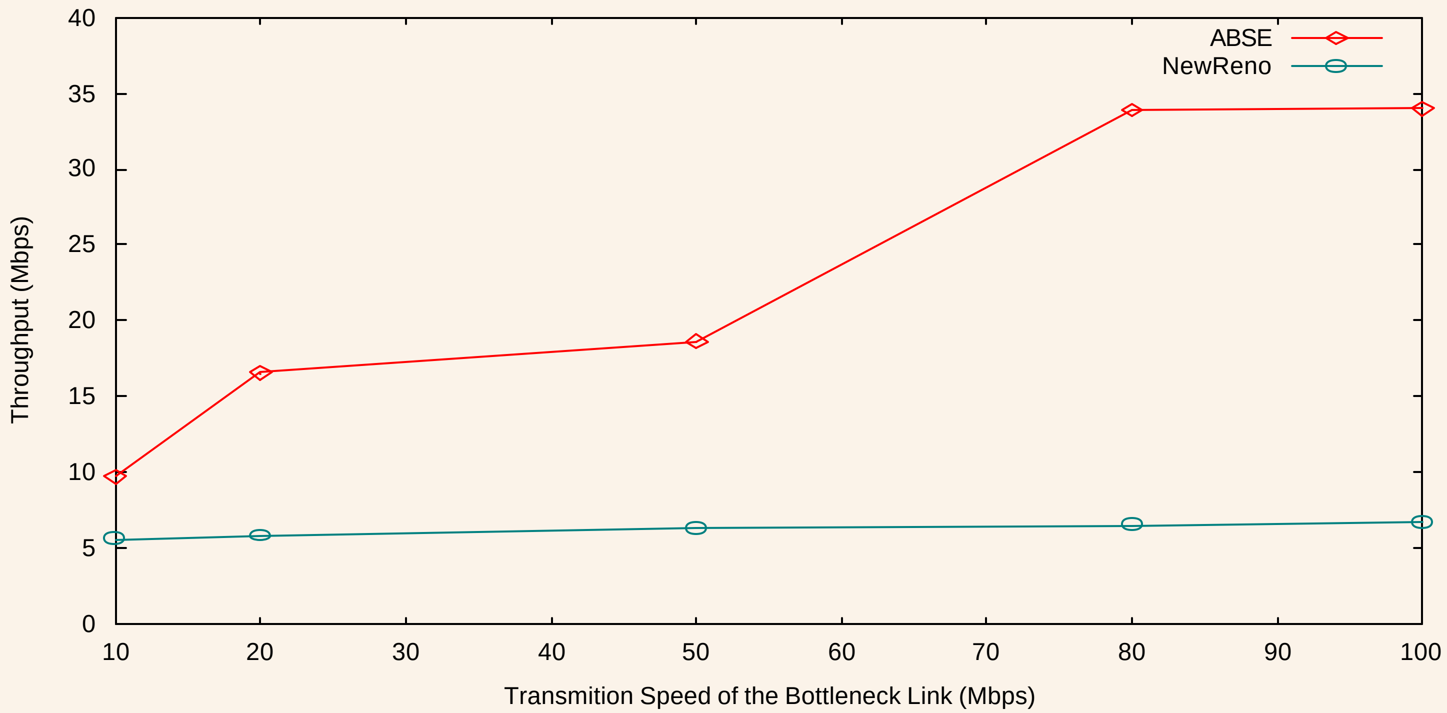
Throughput vs. packet loss rate

TCPW AF Simulation Results (2)



Throughput vs. Two-way Propagation Time
Is there loss?

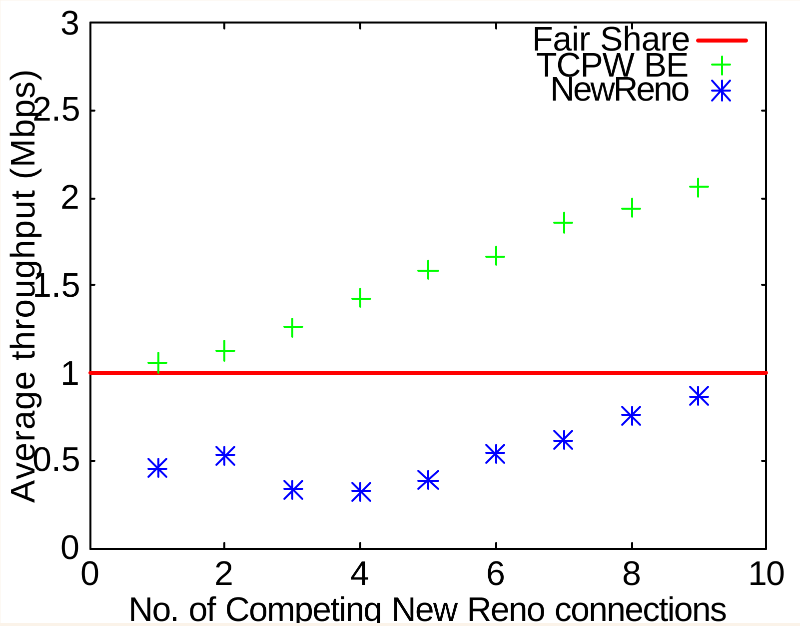
TCPW AF Simulation Results (3)

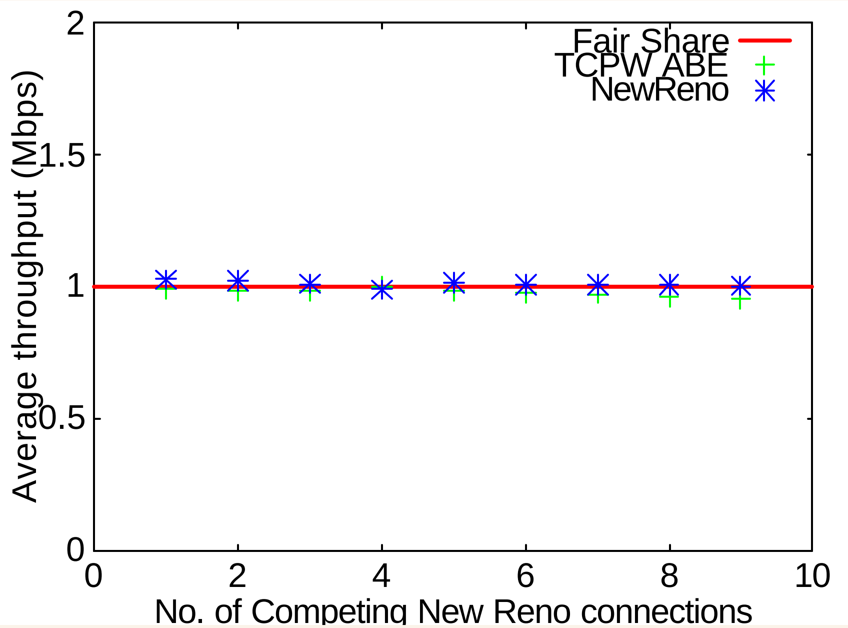


Throughput vs. Bottleneck Capacity

Is there loss?

TCPW AF Simulation results (4)





TCPW AF Is Friendly Towards NewReno !

Summary

- Introduced the concept of Rate Estimation and related work
- Reviewed end-to-end estimation based congestion control methods
- Presented TCP Westwood, and the evolution of "fair rate" estimate to improve the performance; showed simulation results to evaluate the method
- Compared TCPW with other methods

References

- The papers about TCP Westwood, TCP Westwood CRB and ABE can be found in the papers section of the TCP Westwood Web Page: http://www.cs.ucla.edu/NRL/hpi/tcpw/
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