# TCP


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# Outline

- TCP flow control
- Persistent source models
  - Square root p law
  - Fixed points
- Short-lived sources

# Why Flow Control?

- October 1986, Internet had its first congestion collapse
- Link LBL to UC Berkeley
  - 400 yards, 3 hops, 32 Kbps
  - throughput dropped to 40 bps
  - factor of ~1000 drop!
- 1988, Van Jacobson proposed TCP flow control

# Congestion Control

- TCP seeks to
  - Achieve high utilization
  - Avoid congestion
  - Share bandwidth
- Window flow control
  - Source rate = $\frac{W}{RTT}$ packets/sec
  - Adapt W to network (and conditions)

    W = BW x RTT

# Network Flow Control

- Source calculates cwnd from indication of network congestion
- Congestion indications
  - **Losses**
  - Delay
  - Marks
- Algorithms to calculate cwnd
  - Tahoe, Reno, Vegas, RED, REM …

# TCP Congestion Control

- Has four main parts
  - Slow Start (SS)
  - Congestion Avoidance (CA)
  - Fast Retransmit
  - Fast Recovery

Tahoe

Reno

- ssthresh: slow start threshold determines whether to use SS or CA
- Assume packet losses are caused by congestion

# Slow Start

- Start with cwnd = 1 (slow start)
- On each successful ACK increment cwnd

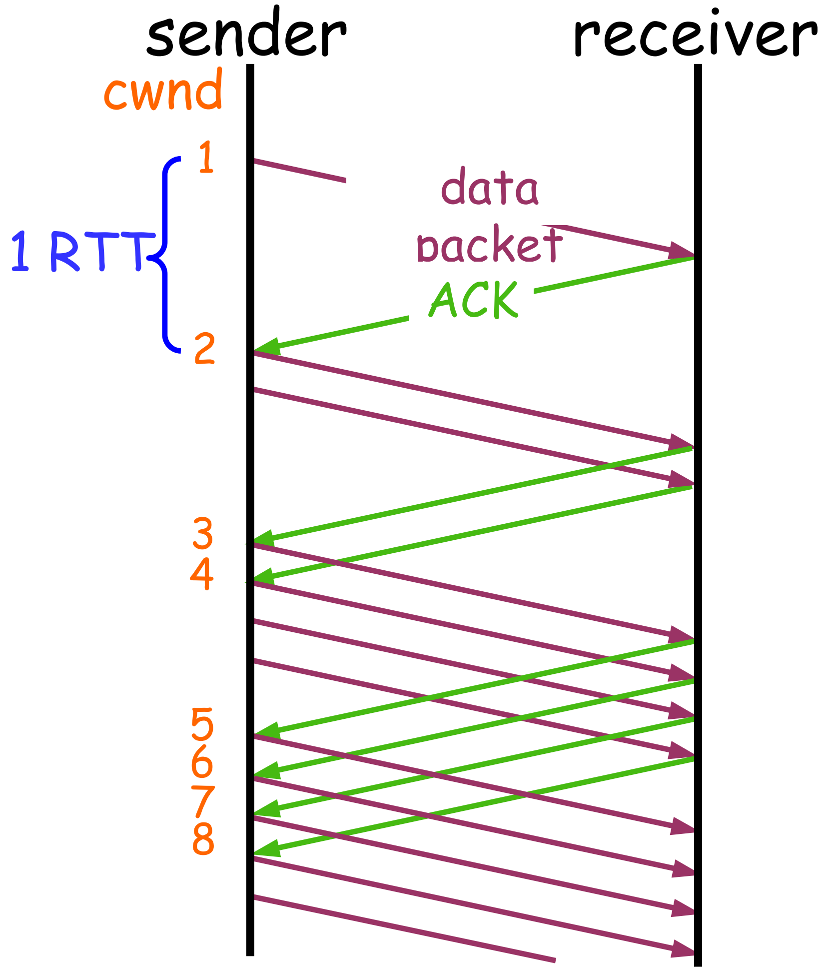
  $$cwnd \leftarrow cnwd + 1$$

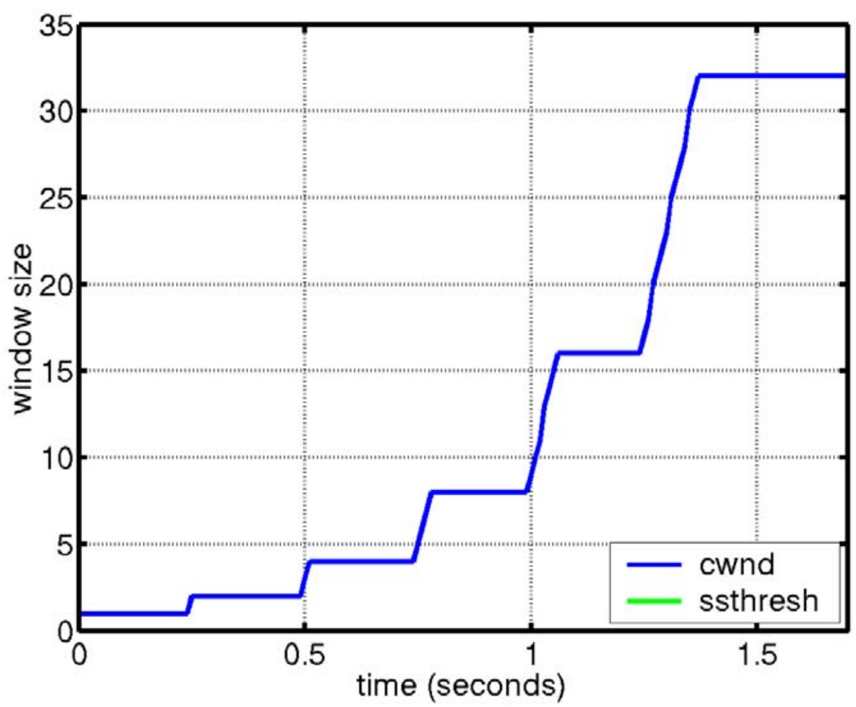
- Exponential growth of cwnd

  each RTT: $cwnd \leftarrow 2 \times cwnd$

- Enter CA when cwnd >= ssthresh

# Slow Start




cwnd ← cwnd + 1 (for each ACK)

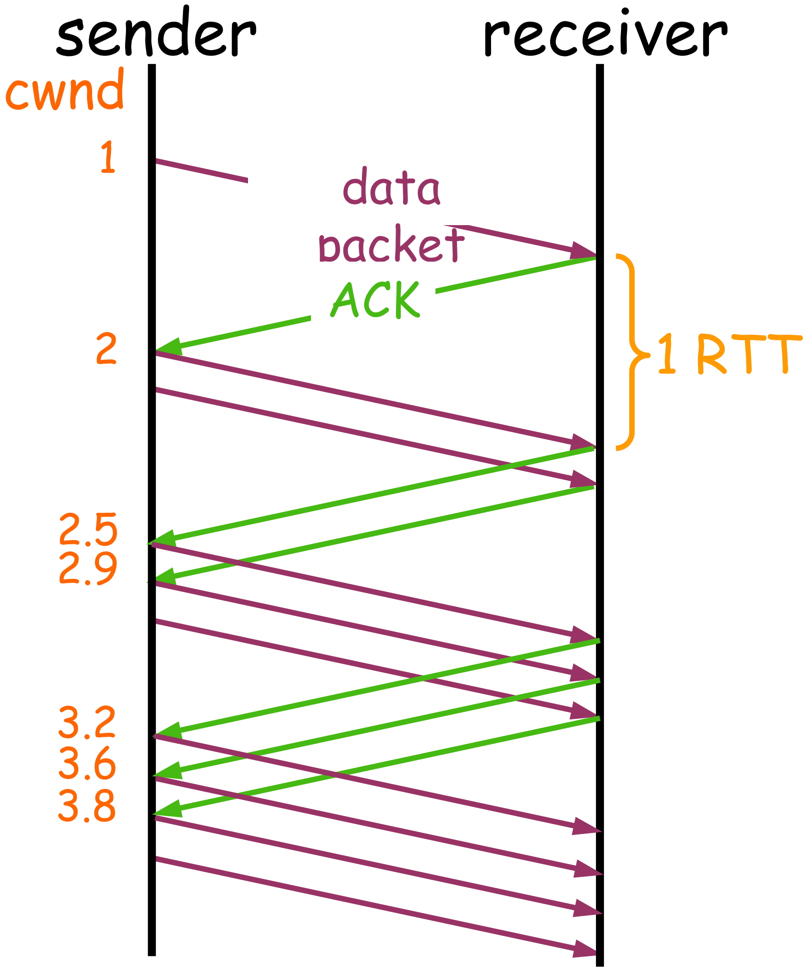
# Congestion Avoidance

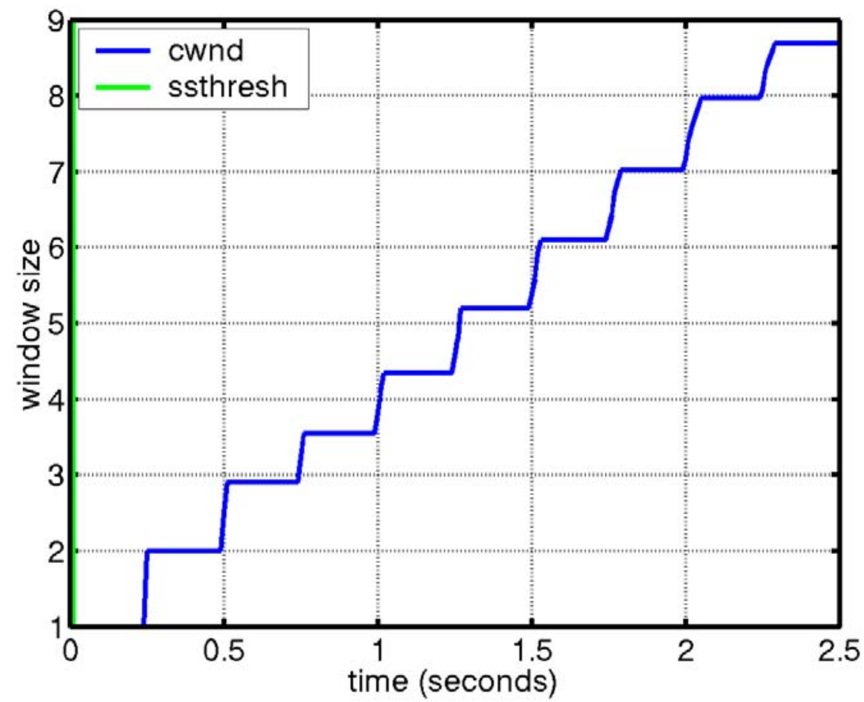
- Starts when $cwnd \geq ssthresh$
- On each successful ACK:

  $cwnd \leftarrow cwnd + 1/cwnd$
- <u>Linear growth</u> of cwnd

  each RTT: $cwnd \leftarrow cwnd + 1$

# Congestion Avoidance

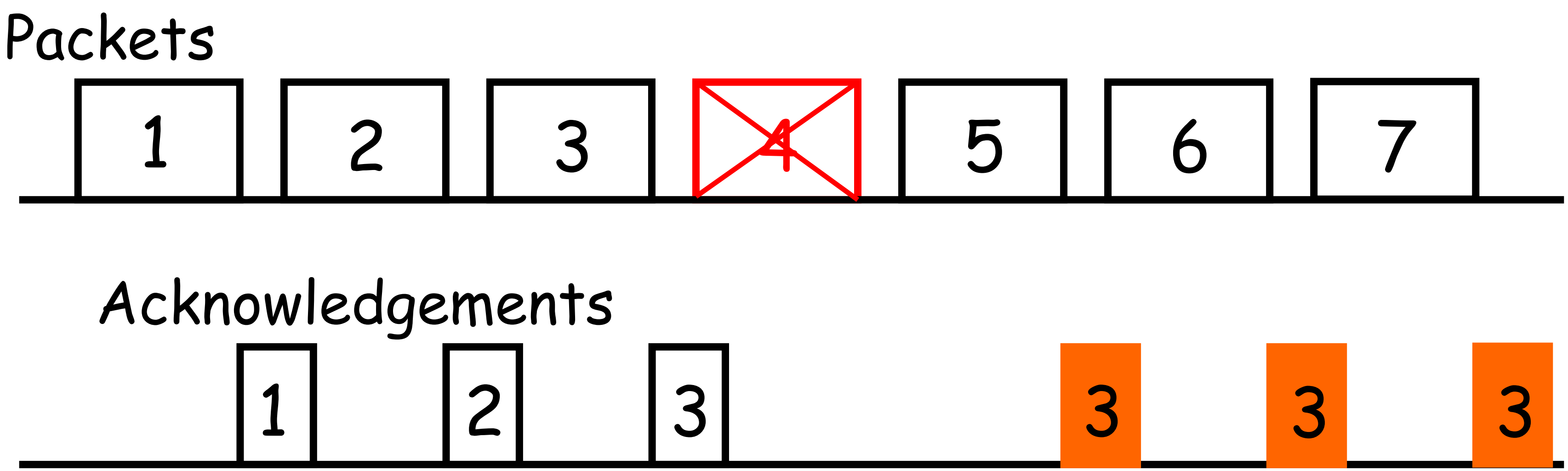



cwnd ← cwnd + 1/cwnd (for each ACK)

# Packet Loss

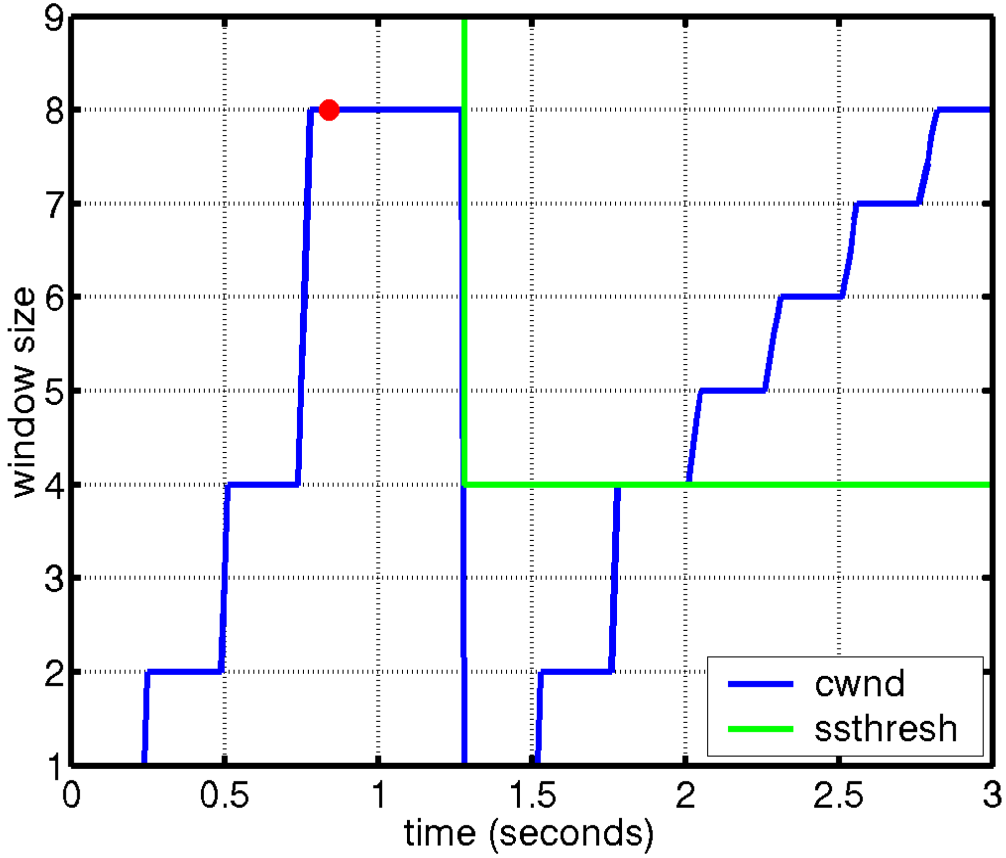
- Assumption: loss indicates congestion
- Packet loss detected by
  - Retransmission TimeOuts (RTO timer)
  - Duplicate ACKs (at least 3)



# Timeout

ssthresh ← cwnd/2

cwnd = 1



# Fast Retransmit

- Wait for a timeout is quite long
- Immediately retransmits after 3 dupACKs without waiting for timeout
- Adjusts ssthresh

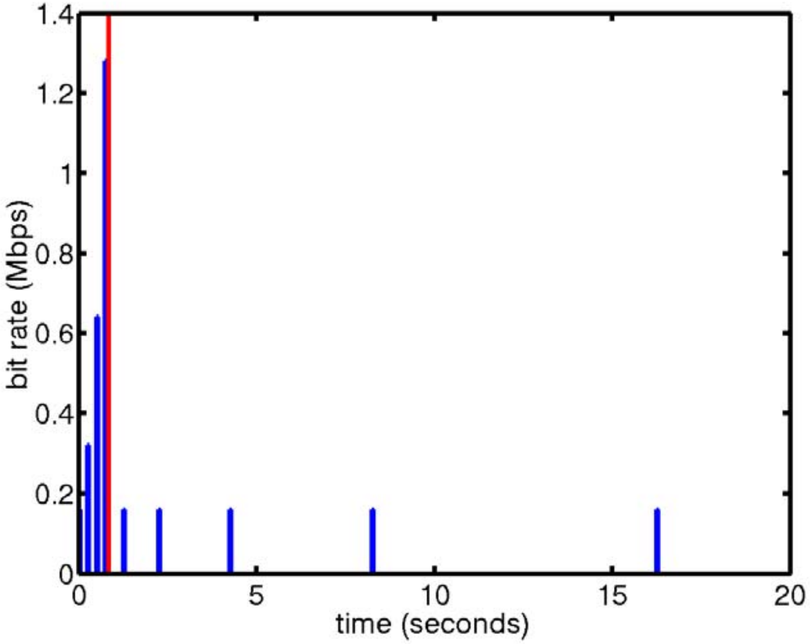
  flightsize = min(awnd, cwnd)

  ssthresh $\leftarrow$ max(flightsize/2, 2)
- Enter Slow Start (cwnd = 1)

# Successive Timeouts

- When there is a timeout, double the RTO
- Keep doing so for each lost retransmission
  - Exponential back-off
  - Max 64 seconds[1]
  - Max 12 restransmits[1]

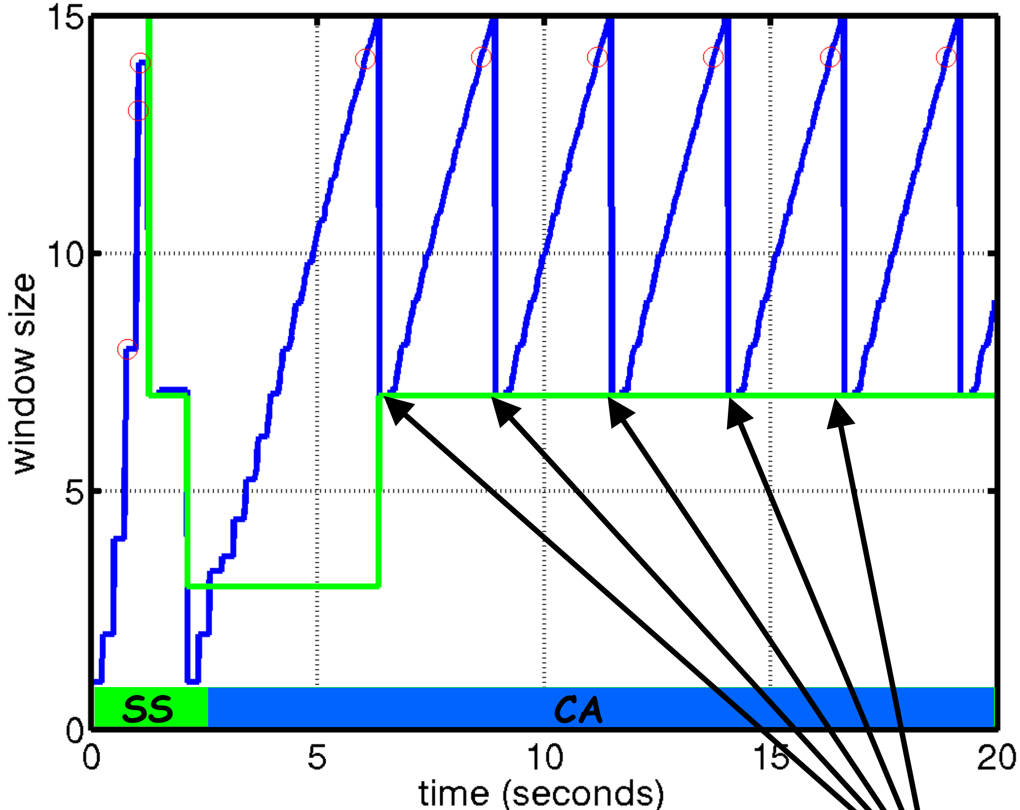
1 - Net/3 BSD



# Fast recovery

- Motivation: prevent `pipe' from emptying after fast retransmit
- Idea: each dupACK represents a packet having left the pipe (successfully received)
- Enter FR/FR after 3 dupACKs
  - Set ssthresh ← max(flightsize/2, 2)
  - Retransmit lost packet
  - Set cwnd ← ssthresh + ndup (window inflation)
  - Wait till W=min(awnd, cwnd) is large enough; transmit new packet(s)
  - On non-dup ACK (1 RTT later), set cwnd ← ssthresh (window deflation)
- Enter CA

# TCP Reno




Fast retransmission/fast recovery

# RTO Calculation

- An accurate RTT measure is required to judge timeouts
- We can measure RTT by measuring the time to receive a packets ACK
- Use a smoothed RTT, $S_{RTT}$ and the smoothed mean deviation $D_{RTT}$

$$RTO = S_{RTT} + 4\, D_{RTT}$$

- Initial RTT should be > 3 seconds
  - Avoid spurious retransmission

# Round Trip Time Estimation

- RTT is not known
  - From <1 ms up to >1 second
- Need to know RTT to calculate RTO
- The measurement of RTT

$$S_{RTT} = S_{RTT} + g\,(M_{RTT} - S_{RTT})$$
$$D_{RTT} = D_{RTT} + h\,(\,|M_{RTT} - S_{RTT}| - D_{RTT})$$

- Need to minimize processing requirements
  - Only 1 counter (regardless of how many packets are extant)
  - Counter granularity is typically 500 ms
- Measurement equations have gain

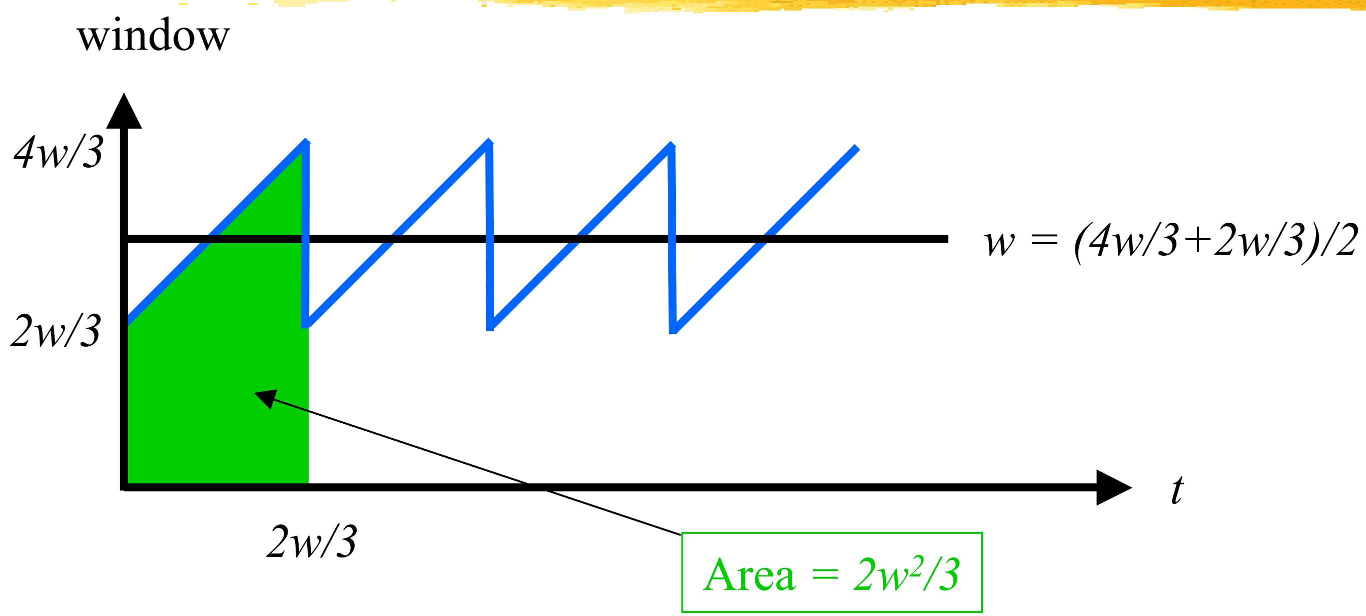
# $1/\sqrt{p}$ Law

- Equilibrium window size $w_s = \dfrac{a}{\sqrt{p}}$
- Equilibrium rate $x_s = \dfrac{a}{D_s\sqrt{p}}$
- Empirically constant $a \sim 1$
- Verified extensively through simulations and on Internet
- References
  - T.J.Ott, J.H.B. Kemperman and M.Mathis (1996)
  - M.Mathis, J.Semke, J.Mahdavi, T.Ott (1997)
  - T.V.Lakshman and U.Mahdow (1997)
  - J.Padhye, V.Firoin, D.Towsley, J.Kurose (1998)
  - J.Padhye, V.Firoin, D.Towsley (1999)

# Implications

- Applicability
  - Additive increase multiplicative decrease (Reno)
  - Congestion avoidance dominates
  - No timeouts, e.g., SACK+RH
  - Small losses
  - Persistent, greedy sources
  - Receiver not bottleneck
- Implications
  - Reno equalizes window
  - Reno discriminates against long connections

# Derivation (I)

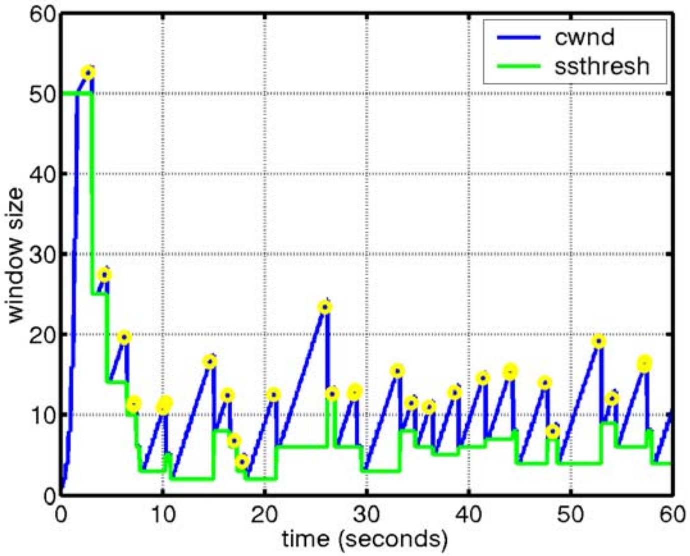



- Each cycle delivers $2w^2/3$ packets
- Assume: each cycle delivers $1/p$ packets
  - Delivers 1/p packets followed by a drop
  - Loss probability = $p/(1+p) \sim p$ if $p$ is small.
- Hence $w = \sqrt{3/2p}$

# Derivation (II)

- *Assume*: loss occurs as Bernoulli process rate p
- *Assume*: spend most time in CA
- *Assume*: p is small
- $w_n$ is the window size after

$$w_{n+1} = \begin{cases} w_n/2, & \text{if a packet is lost (prob. } pw_n) \\ w_n + 1, & \text{if no packet is lost (prob. } (1 - pw_n)) \end{cases}$$




$$\begin{aligned} \overline{w} &= \frac{\overline{w}}{2} p\overline{w} + (\overline{w}+1)(1 - p\overline{w}) \\ \overline{w}^2 &\approx 2/p \\ \overline{w} &\approx \sqrt{2/p} \end{aligned}$$

# Refinement (Padhye, Firoin, Towsley & Kurose 1998)

- Renewal model including
  - FR/FR with Delayed ACKs (b packets per ACK)
  - Timeouts
  - Receiver awnd limitation
- Source rate

$$x_s = \min\left( \frac{W_r}{D_s}, \frac{1}{D_s\sqrt{\frac{2bp}{3}} + T_o \min\left(1, 3\sqrt{\frac{3bp}{8}}\right) p(1+32p^2)} \right)$$

- When $p$ is small and $W_r$ is large, reduces to

$$x_s = \frac{a}{D_s\sqrt{p}}$$

# Calculating Performance

- Single link, capacity C, buffer B
  - Window size: $w = f(p)$
  - Loss rate: $p = g(w; C,B)$
  - Find w*: $w^* = f(g(w^*; C,B))$
- Example:
  - Window size: $w = 1/\sqrt{p}$
  - Loss rate approx. $p = \frac{[w-C]^+}{w}$

$$w^* = \frac{C + \sqrt{C^2 + 4}}{2}$$

# Fixed Point Models

- Mean field theory
  - Solve for a particular source given the mean field
  - Use single source to approximate the mean field
- Generalize previous example
  - Multiple sources
  - Network
    - various routes, RTTs, capacities, …
  - Arbitrary functions $f$, and $g$
- Solve using
  - Repeated substitution
  - Newton-Raphson

# Network Formulation

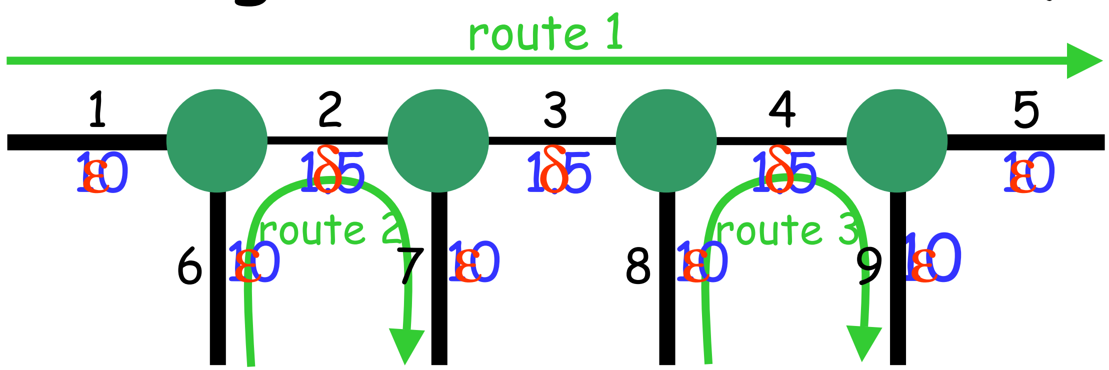
- N links, R routes
- Capacity $\mathbf{c} = \{c_j\}$ $j=1,..,N$
- Propagation time $\mathbf{t} = \{t_j\}$ $j=1,..,N$
- Routing matrix $\mathbf{A} = \{a_{ij}\}$ $j=1,..,N,\ i=1,..,R$

  $a_{ij} = 1$, if link j is in route i

  $a_{ij} = 0$, if link j isn't in route j
- Sources per route $\mathbf{n} = \{n_i\}$ $i=1,..,R$
- MSS per route $\mathbf{m} = \{m_j\}$ $i=1,..,R$
- Route send rate $\mathbf{s} = \{s_j\}$ $i=1,..,R$
- Link loss rate $\mathbf{q} = \{d_i\}$ $j=1,..,N$
- Route loss rate $\mathbf{p} = \{p_i\}$ $i=1,..,R$

# Example Network

- N congested bottlenecks (e.g. 2)




$$\mathbf{t} = (\varepsilon,\ \delta,\ \delta,\ \delta,\ \varepsilon,\ \varepsilon,\ \varepsilon,\ \varepsilon,\ \varepsilon)^\dagger$$

$$\mathbf{c} = (10, 1.5, 1.5, 1.5, 10, 10, 10, 10, 10)^\dagger$$

$$A = \begin{pmatrix} 1 & 1 & 1 & 1 & 1 & 0 & 0 & 0 & 0 \\ 0 & 1 & 0 & 0 & 0 & 1 & 1 & 0 & 0 \\ 0 & 0 & 0 & 1 & 0 & 0 & 0 & 1 & 1 \end{pmatrix}$$

# Solution

- Estimate RTT delay from propagation time

$$d = 2At$$ (can use queueing delays)

- Route send rates

$$x(w) = (w \;.\!*\; n \;.\!*\; m) \;./\; d$$

- Link rates

$$b(w) = A^{t}x$$

- Link loss rate

$$q(w;c) = [b - c]^{+}./b$$ (can use queueing losses)

- Route loss rate

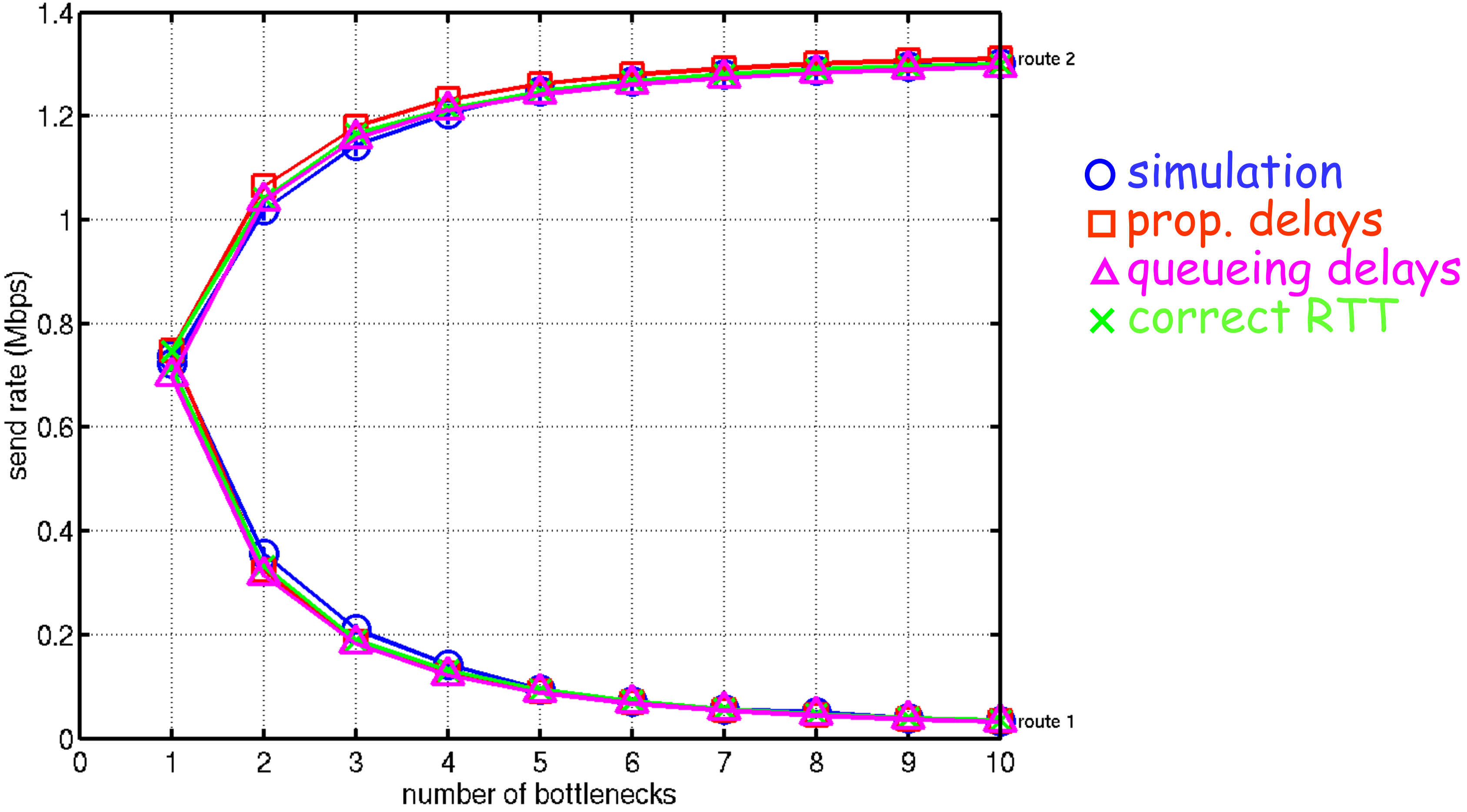
$$p(w;c) = 1 - e^{A\ln(1-q(w;c))}$$

- Window size

$$W^2\, p(w;c) - a = 0$$ (could use refined model, or a transient model)

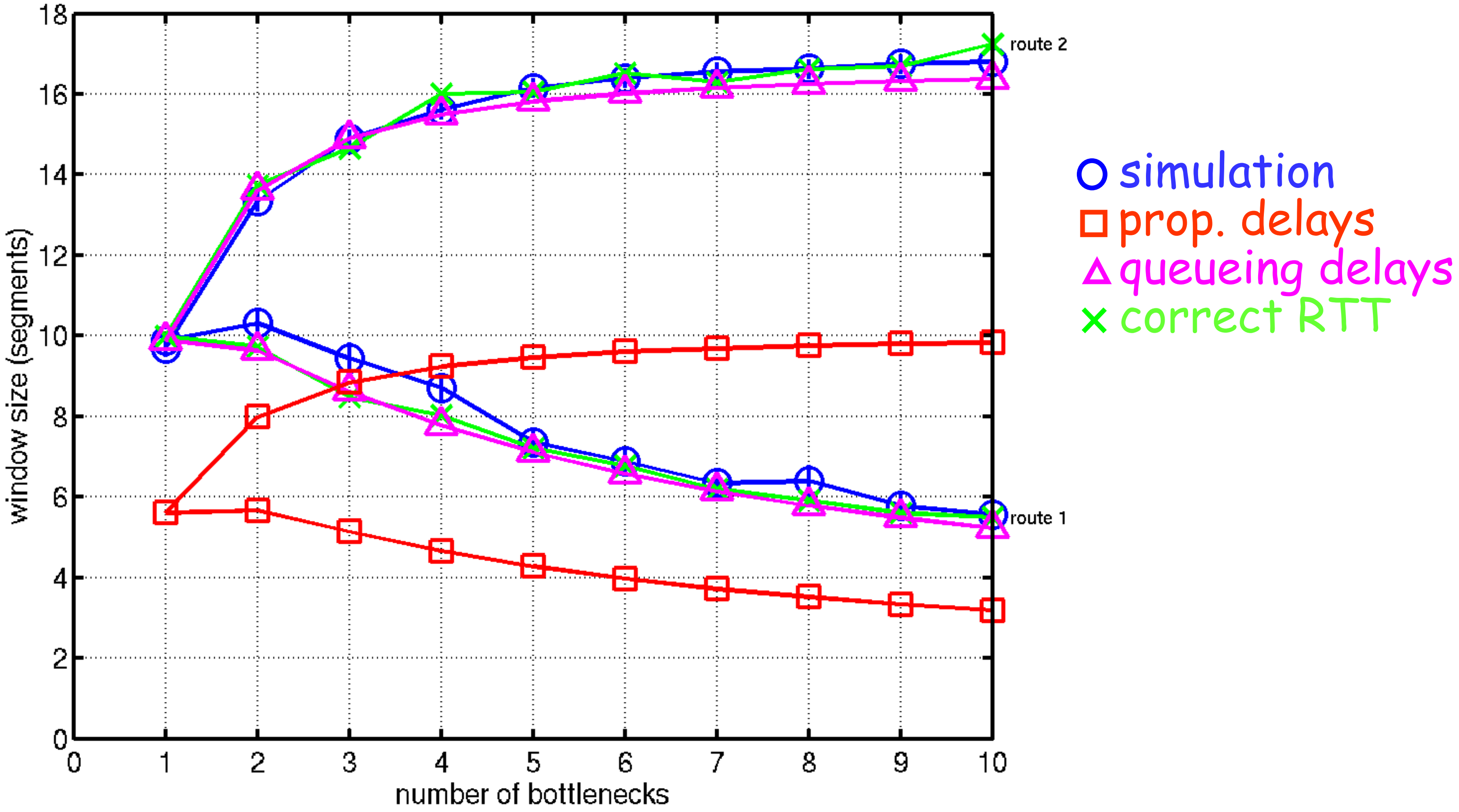
# Numerical Example

- Send rates



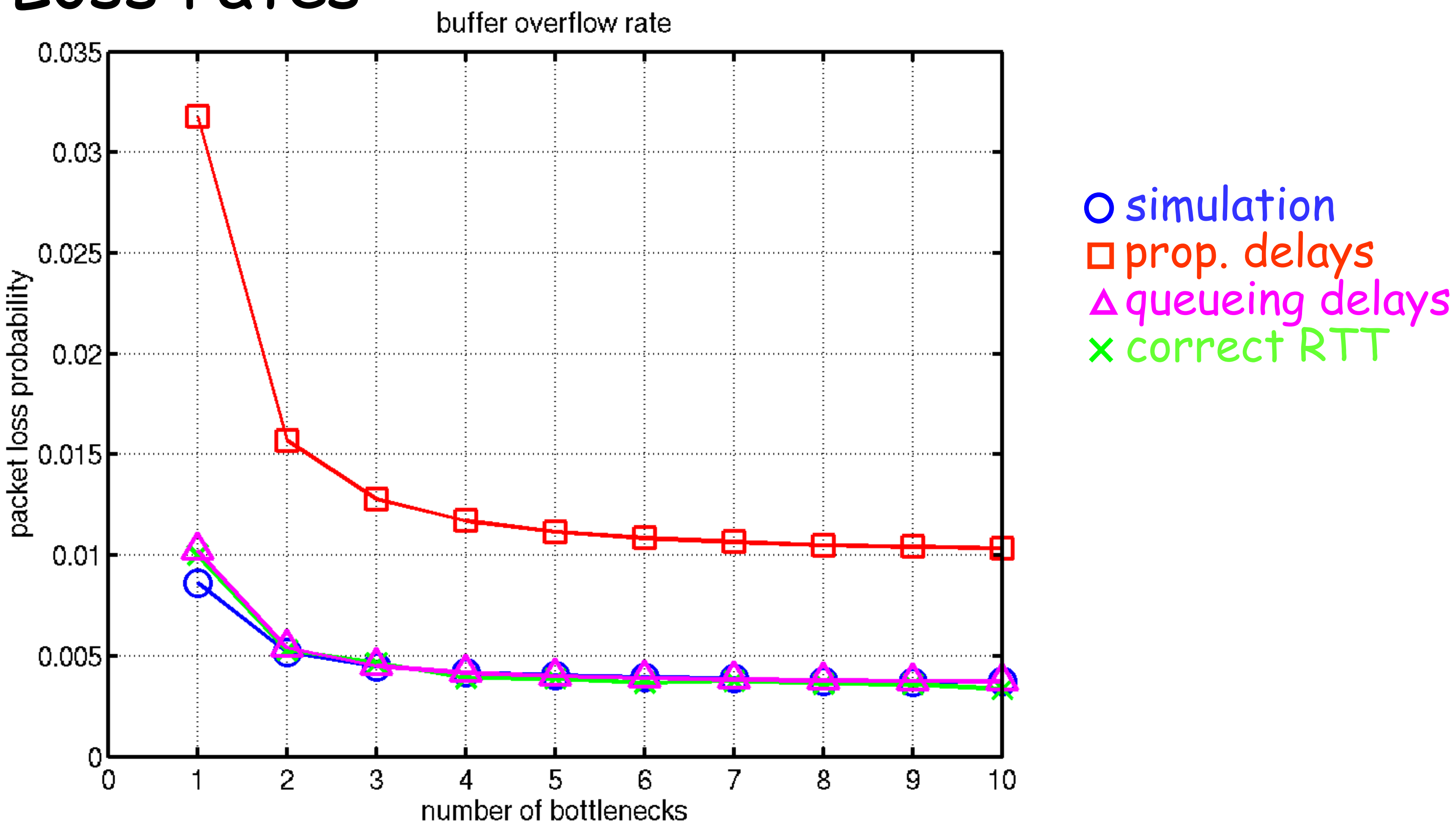
# Numerical Example

- Window sizes



# Numerical Example

- Loss rates



# Short-lived sources

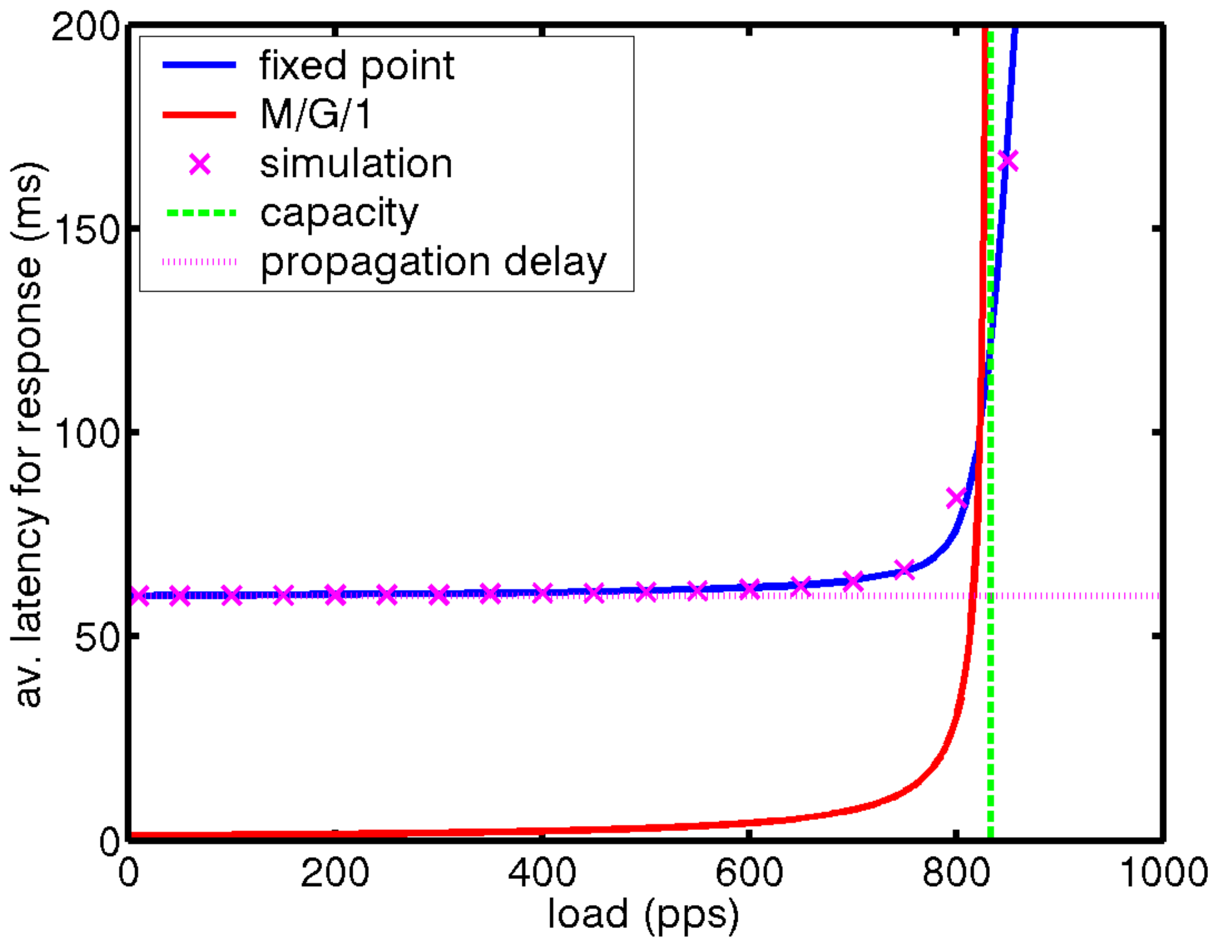
- Heavy-tailed distribution of flow sizes
  - Some really big files — elephants
  - Many small files — mice
- Persistent model only good for elephants
  - Concentrates on Congestion Avoidance
  - Short lived sources always in Slow Start
  - M/G/1 processor sharing suggested
  - Really we need a new model, e.g.
    - Cardwell, Savage and Anderson, Infocom 2000
    - Sikdar, Kalyanaraman and Vastola, IPCCC 2001
    - Mellia, Stoica and Zhang, IEEE Communications Let. 2002

# New approach

- Use the loss rate to estimate transfer latency (e.g. from Cardwell *et al*)
- Use transfer latency to compute the number of sessions in progress
  - M/G/1 processor sharing queue (for number of sources)
- Use the number of sessions in progress (and their duration) to estimate the load and thence the loss rate
  - M/G/1/K FIFO model (for packets in each buffer)

# Simple example

- Poisson arrivals of single packet transfers



# Results

- Processor sharing
  - Doesn't get latency right for low load (can't get RTT)
  - Asymptote at capacity
- Even so result is not responsive to congestion!
- Can get a good measure from fixed point approach

# Conclusion

- Can use fixed point methods to estimate performance for TCP flow controls
  - Persistant case (based on CA)
  - Short-lived case (based on SS)
- Nice because they generalize to networks
- Need to understand limitations of SS models for TCP window flow controls
  - RTT estimation used in RTO computation
  - In BSD simple because of 500ms timer