

### Queueing theory, control theory, & buffer sizing

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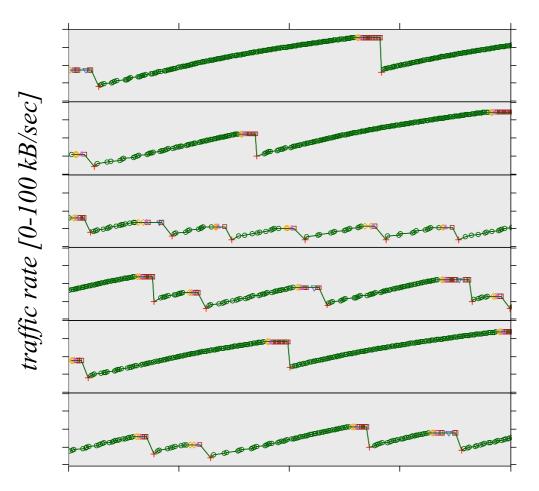
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# My interest

- Internet routers have buffers,
  - to accomodate bursts in traffic
  - to keep the link fully utilized
- How big do the buffers need to be, to accommodate TCP traffic?
  - -3 GByte? Rule of thumb says buffer = bandwidth×delay
  - 300 MByte? buffer = bandwidth×delay/√#flows [Appenzeller, Keslassy, McKeown, 2004]
  - 30 kByte? constant buffer size, independent of line rate [Kelly, Key, etc.]
- What is the role of probabilistic queueing theory? *Is it all just fluid models & differential equations?*

# TCP



*time* [0-8 sec]

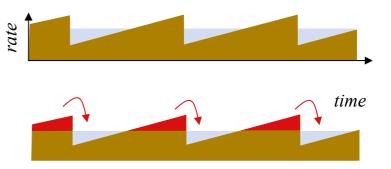
if (seqno > \_last\_acked) { if (!\_in\_fast\_recovery) { \_last\_acked = seqno; dupacks = 0;inflate window(); send packets(now); last sent time = now; return; if (seqno < recover) { uint32\_t new\_data = seqno - \_last\_acked; last acked = seqno; if (new data < cwnd) cwnd -= new data; else cwnd=0;  $\_cwnd += \_mss;$ retransmit packet(now); send packets(now); return; uint32 t flightsize = highest sent - sequo; \_cwnd = min(\_ssthresh, flightsize + \_mss); \_last\_acked = seqno; dupacks = 0;\_in\_fast\_recovery = false; send packets(now); return; if ( in fast recovery) { \_cwnd += \_mss; send\_packets(now); return; dupacks++; if (\_dupacks!=3) { send\_packets(now); return;  $ssthresh = max(\_cwnd/2, (uint32\_t)(2 * \_mss));$ retransmit\_packet(now); cwnd = ssthresh + 3 \* mss;\_in\_fast\_recovery = true; recover = highest sent;

# TCP sawtooth & buffer size

- The traffic rate produced by a TCP flow follows a 'sawtooth'
- To prevent the link's going idle, the buffer must be big enough to smooth out a sawtooth
  - buffer = bandwidth×delay
- When there are many TCP flows, the sawteeth should average out, so a smaller buffer is sufficient
  - buffer = bandwidth×delay/\#flows
     [Appenzeller, Keslassy, McKeown, 2004]
- If we could keep the traffic rate just a little bit lower, virtually no buffer would be needed...
- ...unless the flows are synchronized



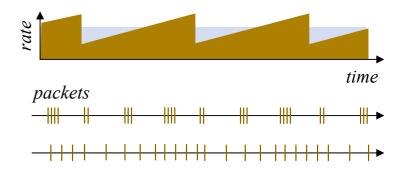




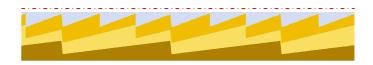


# TCP packets & buffer size

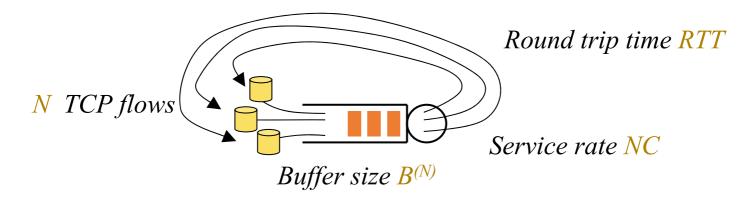
- TCP traffic is made up of packets
  - there may be packet clumps, if the access network is fast
  - or the packets may be spaced out



• Even if we manage to keep total data rate < service rate, chance alignment of packets will still lead to some queueing & loss, so we can't dispense with buffers entirely



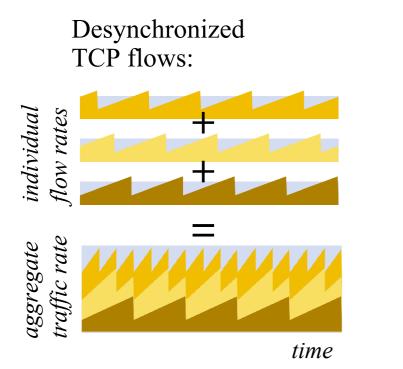
#### Formulate a maths question



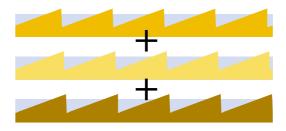
- What is the limiting queue-length process, as  $N \rightarrow \infty$ , in the three regimes
  - large buffers  $B^{(N)}=BN$
  - intermediate buffers  $B^{(N)}=B\sqrt{N}$
  - small buffers  $B^{(N)}=B$
- Why are these limiting regimes interesting? What are the alternatives? [Bain, 2003]

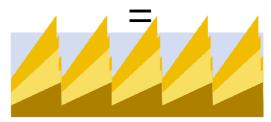
# TCP traffic model

• A single TCP flow follows a characteristic 'sawtooth'



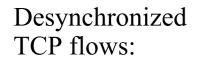
Synchronized TCP flows:



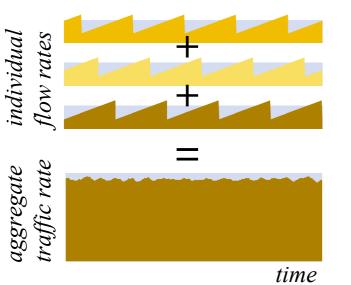


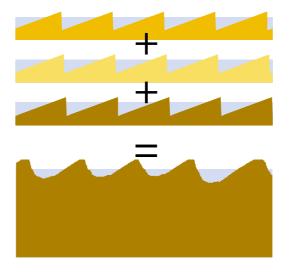
# TCP traffic model

- A single TCP flow follows a characteristic 'sawtooth'
- Many TCP flows added together are smoother



Synchronized TCP flows:





# TCP traffic model

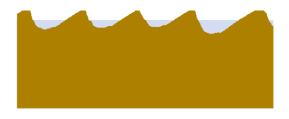
• When there are many TCP flows, the average traffic rate  $x_t$  varies smoothly, according to a delay differential equation [Misra, Gong, Towsley, 2000]

$$\frac{dx_t}{dt} = \frac{1}{RIT^2} - p_{t-RIT}x_{t-RIT}\frac{x_t}{2}$$

- The equation involves
  - $-p_t$ , the packet loss probability at time t
  - *RTT*, the average round trip time

ıggregate raffic rate





## Queue model

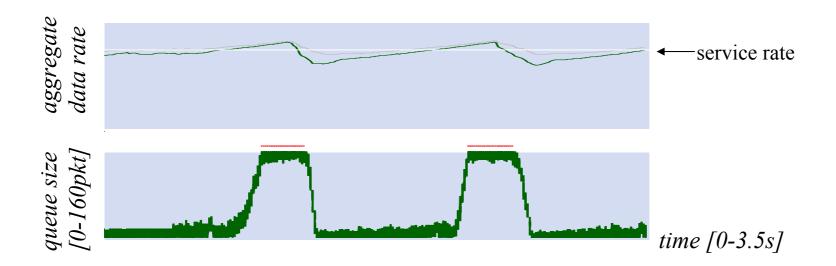
- How does packet loss probability  $p_t$  depend on buffer size?
- The answer depends on the buffer size
  - large buffers  $B^{(N)}=BN$
  - intermediate buffers  $B^{(N)}=B\sqrt{N}$
  - small buffers  $B^{(N)}=B$



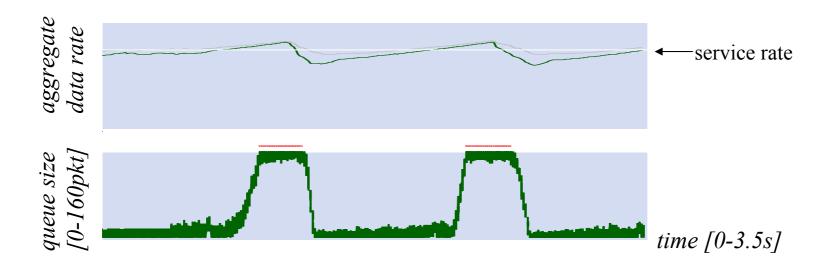
- When the aggregate data rate is less than the service rate, the queue stays small
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- No packet drops, so TCPs increase their data rate
- Eventually the aggregate data rate exceeds the service rate, and a queue starts to build up
- When the queue is full, packets start to get dropped
- One round trip time later, TCPs respond and cut back They may overreact, leading to synchronization *i.e. periodic fluctuations*



- Queue size & arrival rate vary on the same timescale
- The total queue size  $Nq_t$  satisfies the fluid model

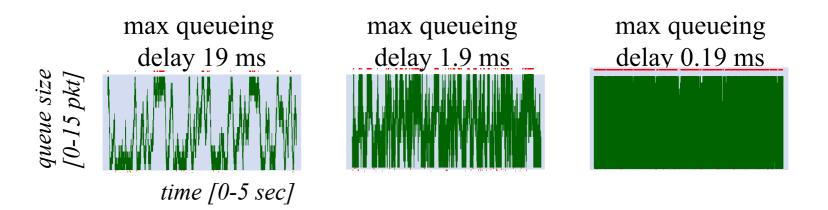
$$\frac{dq_t}{dt} = x_t(1-p_t) - C$$

• When the queue is near full, Little's Law gives the drop probability

$$p_t = 1_{\{q_t = B\}}(x_t - C)/x_t$$

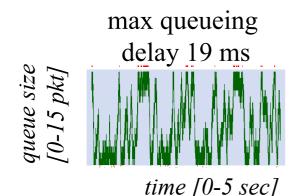
[cf McDonald, Reynier, 2003]

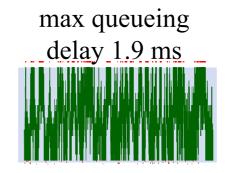
## Small buffer B(N)=B



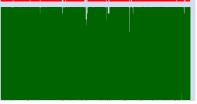
- As the number of flows N and the capacity NC increase, we observe
  - queues aise because of chance alignments of packets
  - queue size fluctuates more and more rapidly, much more rapidly than variations in arrival rate
  - queue size distribution does not change
  - like an  $M_{Nx}/M_{NC}/1/B$  queue

# Small buffer B(N)=B





max queueing delay 0.19 ms

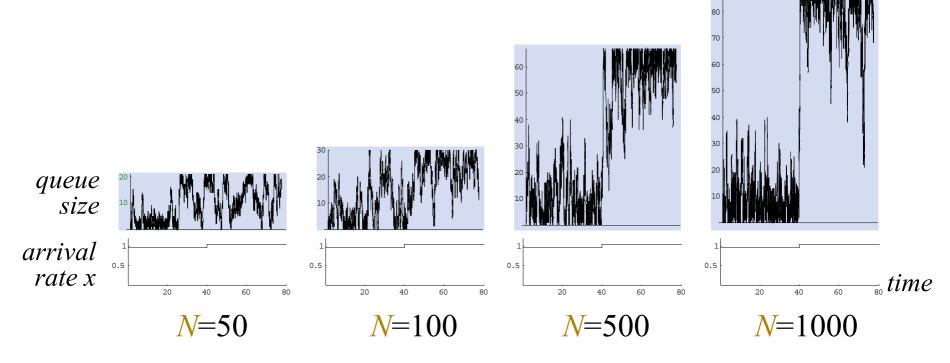


- We conjecture
  - a typical busy cycle lasts O(1/N)
  - packet arrivals over timescale O(1/N) look like a Poisson process with constant arrival rate  $x_t \approx x_{t+O(1/N)}$
  - drop probability converges to that for an M/D/1/B queue:  $p_t \approx (x_t/C)^B$
- Evidence
  - In a queue with a small buffer, fed by arbitrary exogenous traffic, a typical busy cycle lasts O(1/N), and queue size matches that in an M/D/1/B queue [Cao, Ramanan, 2002]
  - Over short timescales (<1ms), TCP traffic is approximately Poisson ["Internet traffic tends toward Poisson and independent as the load increases", Cao, Cleveland, Lin, Sun, 2002]

### Intermediate buffers $B^{(N)}=B\sqrt{N}$

Consider a queue

- fed by *N* flows, each of rate *x* pkts/sec (*x*=0.95 then 1.05 pkts/sec)
- served at rate NC (C=1 pkt/sec)
- with buffer size  $B\sqrt{N}$  (B=3 pkts)



## System summary

- $x_t$  = average traffic rate at time t  $p_t$  = packet loss probability at time tC = capacity/flow B = buffer size RTT = round trip time N=# flows
- TCP traffic model

 $\frac{dx_t}{dt} = \frac{1}{RIT^2} - p_{t-RIT}x_{t-RIT}\frac{x_t}{2}$ 

• Small buffer queueing model (though this is sensitive to traffic statistics)

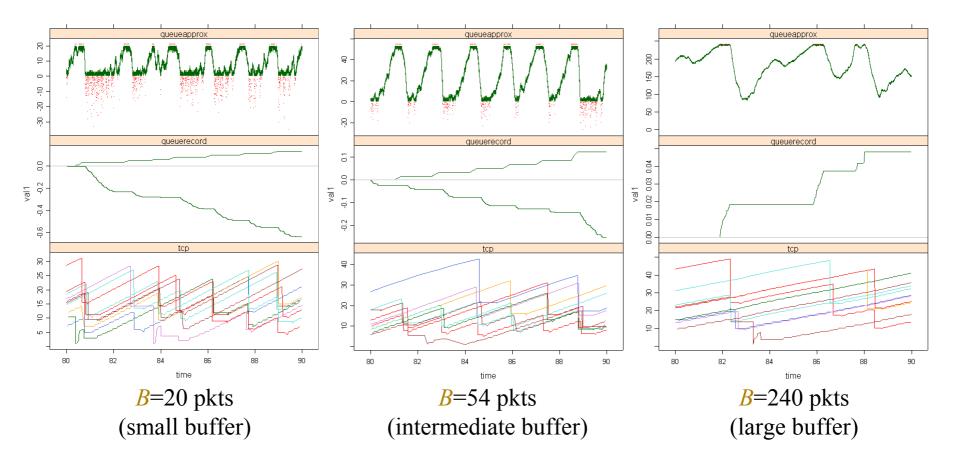
 $p_t \approx (x_t/C)^B$ 

• Large buffer queueing model

$$p_t = \mathbf{1}_{\{q_t = B\}}(x_t - C)/x_t$$
$$\frac{dq_t}{dt} = x_t(1 - p_t) - C$$

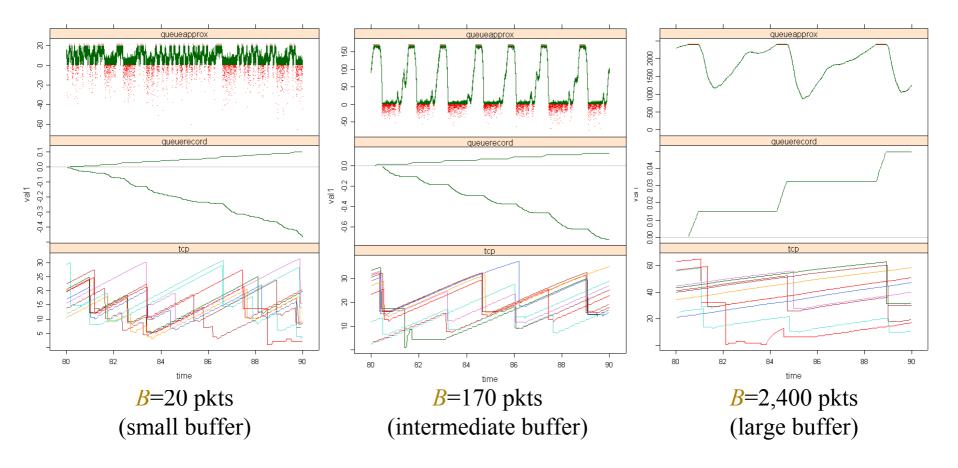
#### Illustration 20 flows

# Standard TCP, single bottleneck link, no AQM service *C*=1.2 kpkt/sec, *RTT*=200 ms, #flows *N*=20



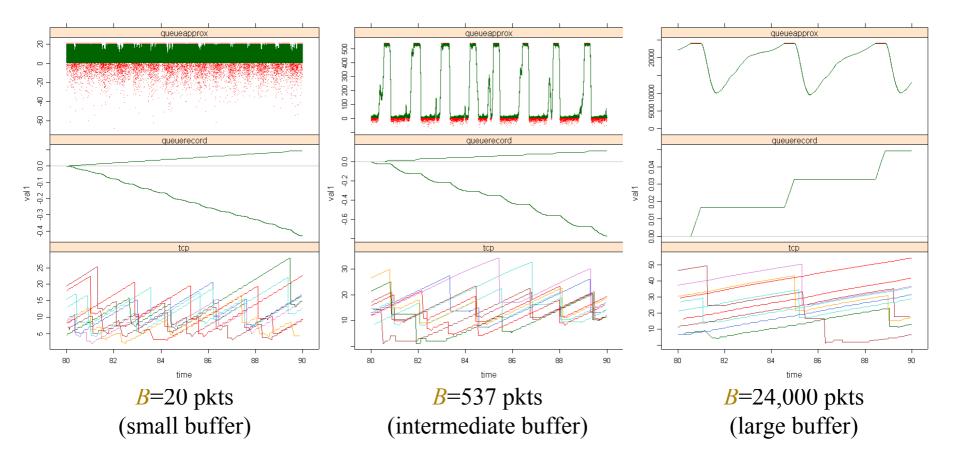
#### Illustration 200 flows

# Standard TCP, single bottleneck link, no AQM service *C*=12 kpkt/sec, *RTT*=200 ms, #flows *N*=200

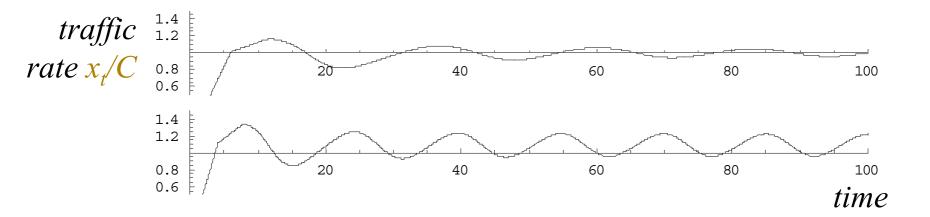


#### Illustration 2000 flows

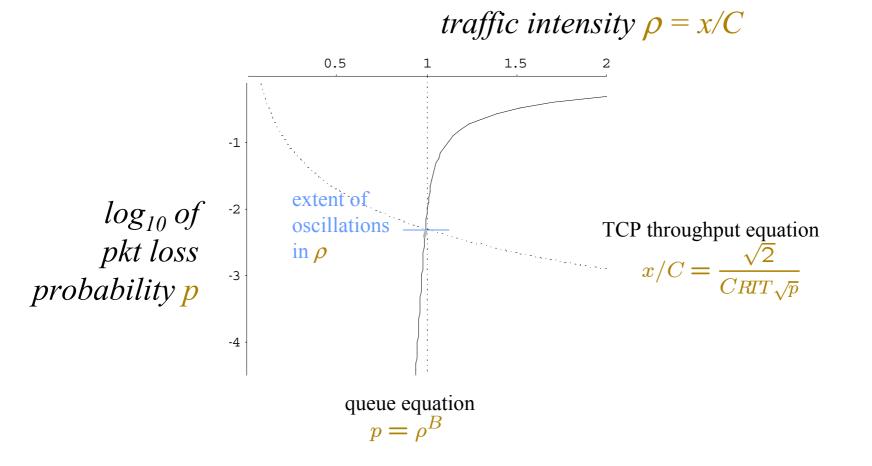
# Standard TCP, single bottleneck link, no AQM service *C*=120 kpkt/sec, *RTT*=200 ms, #flows *N*=2000

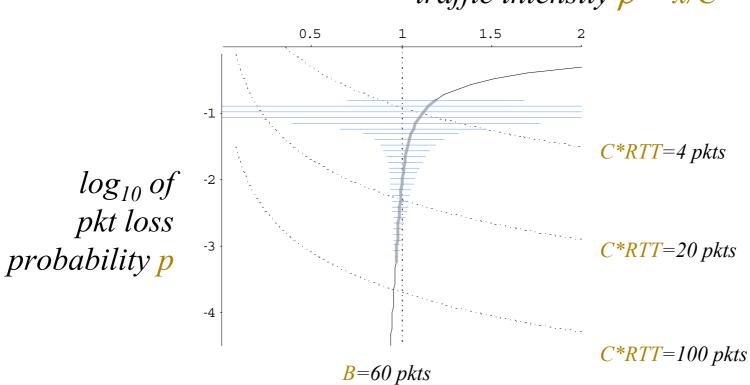


## Stability/instability analysis

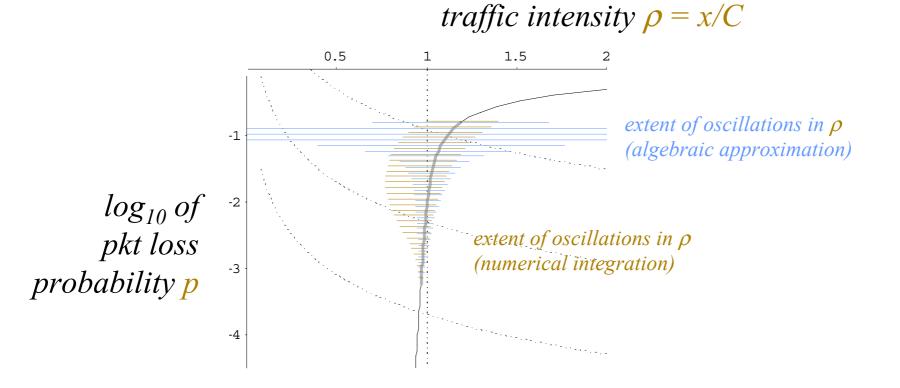


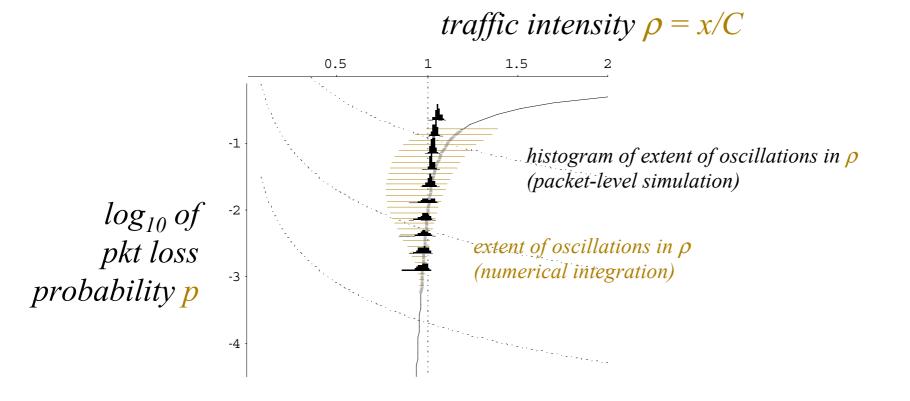
- For some values of  $C^*RTT$ , the dynamical system is stable
  - we calculate the steady-state traffic rate, loss probability etc.
- For others it is unstable and there are oscillations (i.e. the flows are partially synchronized)
  - we calculate the amplitude of the oscillations
    [Gaurav Raina, PhD thesis, 2005]

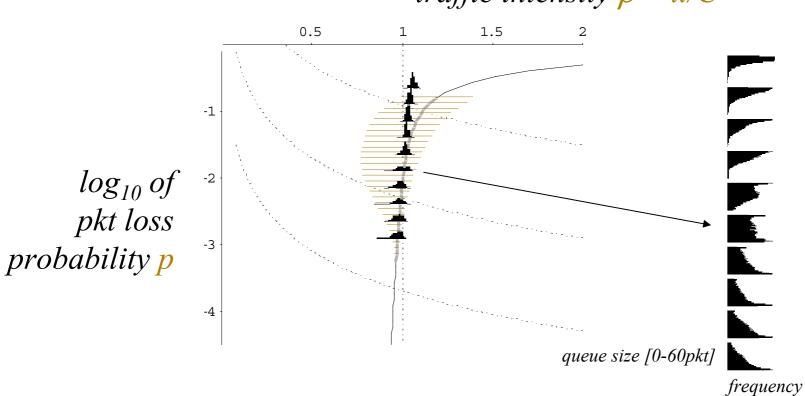




traffic intensity  $\rho = x/C$ 



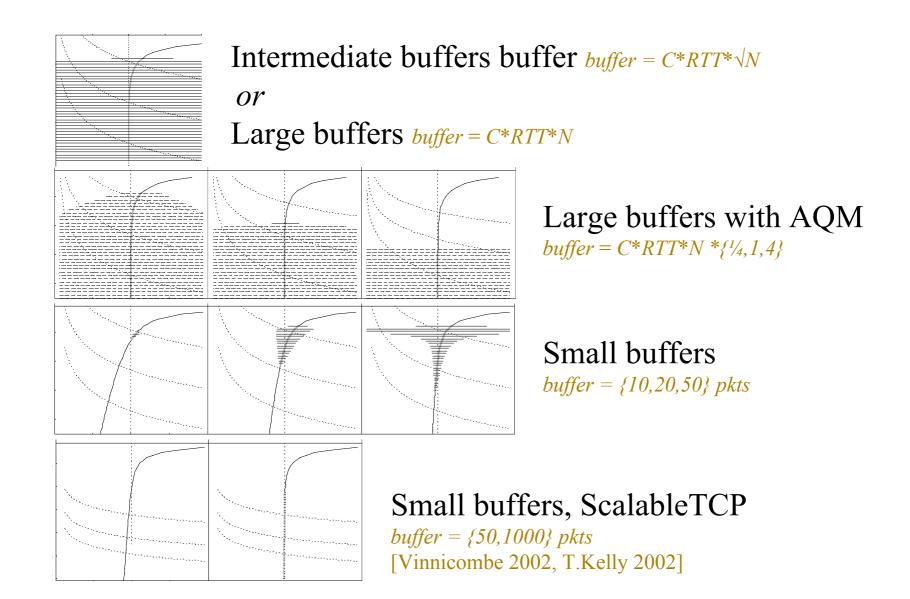




traffic intensity  $\rho = x/C$ 

histogram of queue size (packet-level simulation)

## Alternative buffer-sizing rules



# Limitations/concerns

- Surely bottlenecks are at the access network, not the core network?
  - Unwise to rely on this!
  - If the core is underutilized, it definitely doesn't need big buffers
  - The small-buffer theory works fine for as few as 20 flows
- The Poisson model sometimes breaks down
  - because of short-timescale packet clumps
  - need more measurement of short-timescale Internet traffic statistics

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• Limited validation so far [McKeown et al. at Stanford, Level3, Internet2]

- Proper validation needs
  - goodly amount of traffic
  - full measurement kit
  - ability to control buffer size

# Conclusion

- Buffer sizes can be very small
  - a buffer of 25pkt gives link utilization > 90%
  - small buffers mean that TCP flows get more regular feedback, so they can better judge how much capacity is available
  - use Poisson traffic models for the router, differential equation models for aggregate traffic

- TCP can be improved with simple changes
  - e.g. space out the packets
  - e.g. modify the window increase/decrease rules
    [ScalableTCP: Vinnicombe 2004, Kelly 2004; XCP: Katabi, Handley, Rohrs 2000]
  - any future transport protocol should be designed along these lines
  - improved TCP may find its way into Linux/Windows within 5 years