CSCI-1680
Transport Layer III
Congestion Control Strikes Back

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Based partly on lecture notes by David Mazières, Phil Levis, John Jannotti, Ion Stoica

Last Time

- Flow Control
- Congestion Control

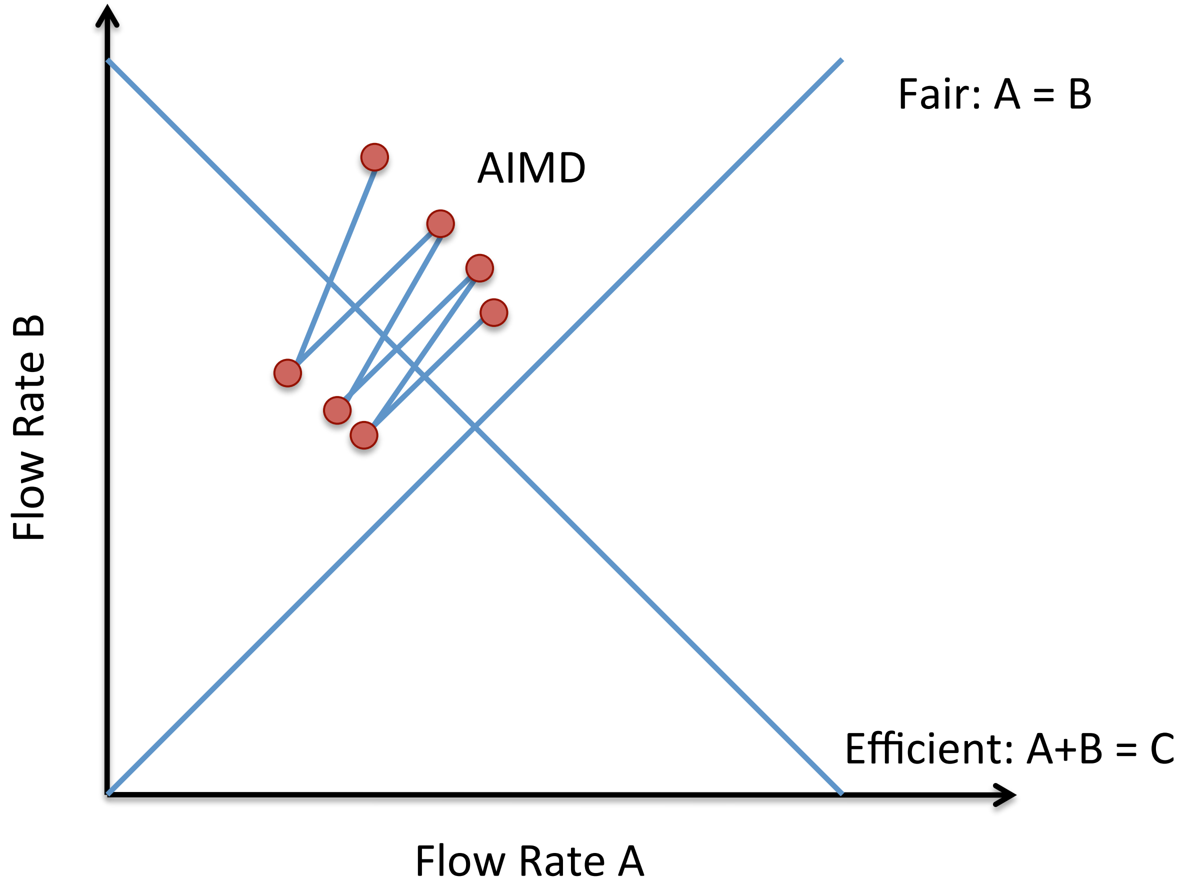
Today

- **More TCP Fun!**
- **Congestion Control Continued**
  - Quick Review
  - RTT Estimation
- **TCP Friendliness**
  - Equation Based Rate Control
- **TCP on Lossy Links**
- **Congestion Control versus Avoidance**
  - Getting help from the network
- **Cheating TCP**

Quick Review

- **Flow Control:**
  - Receiver sets Advertised Window
- **Congestion Control**
  - Two states: Slow Start (SS) and Congestion Avoidance (CA)
  - A window size threshold governs the state transition
    - Window <= ssthresh: SS
    - Window > ssthresh: Congestion Avoidance
  - States differ in how they respond to ACKs
    - Slow start: +1 w per RTT (Exponential increase)
    - Congestion Avoidance: +1 MSS per RTT (Additive increase)
  - On loss event: set ssthresh = w/2, w = 1, slow start

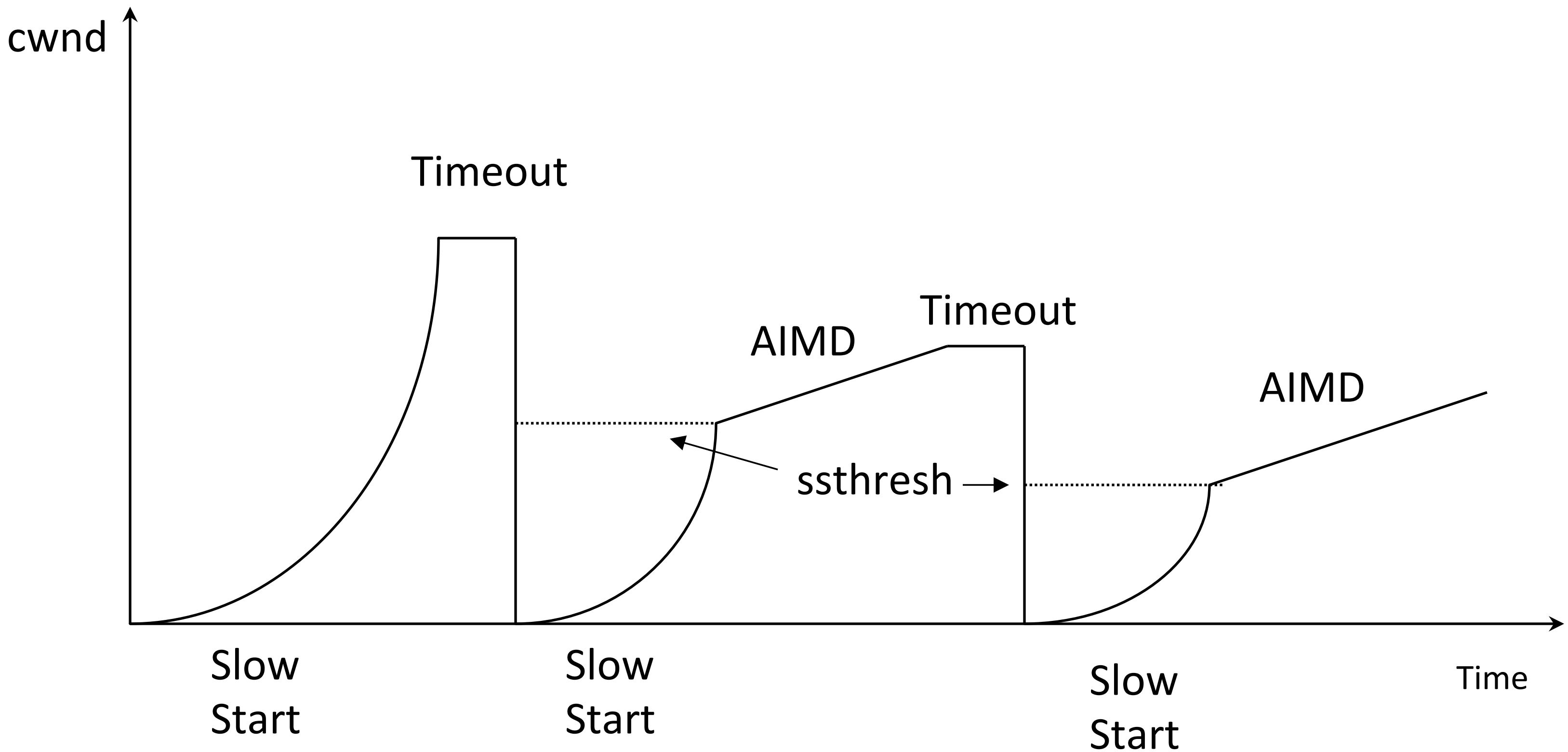
AIMD



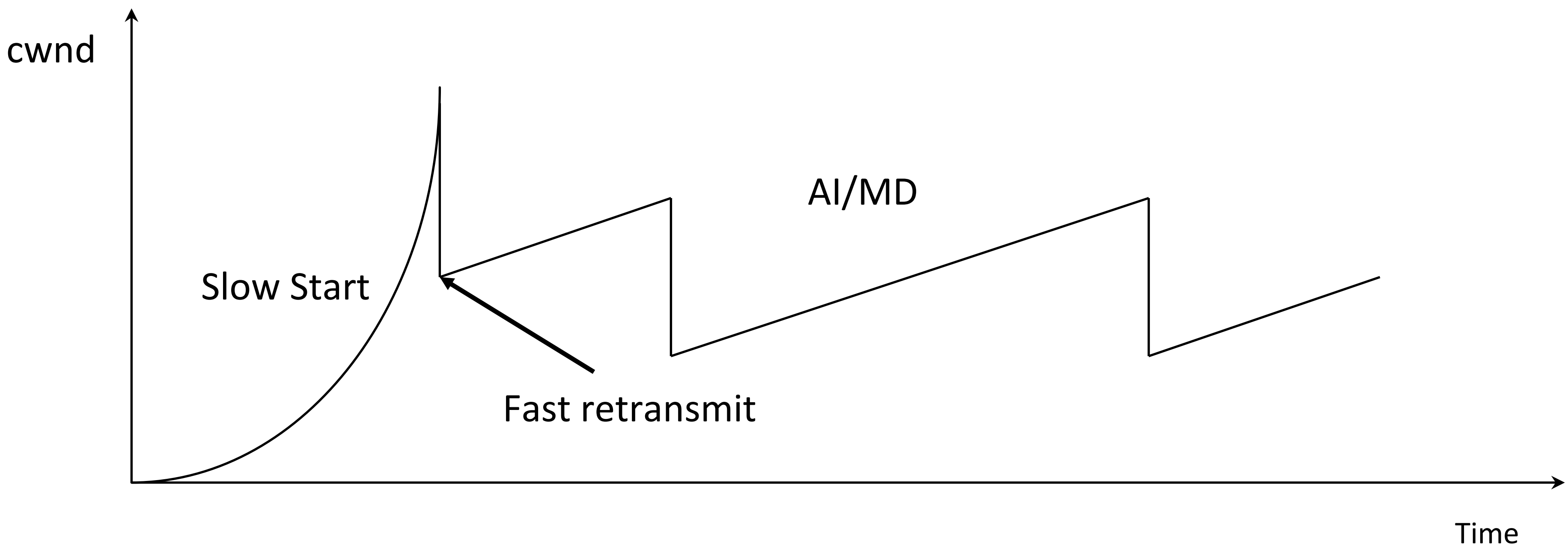
States differ in how they respond to acks

- **Slow start: double w in one RTT**
  - There are w/MSS segments (and acks) per RTT
  - Increase w per RTT → how much to increase per ack?
    - w / (w/MSS) = MSS
- **AIMD: Add 1 MSS per RTT**
  - $MSS/(w/MSS) = MSS^2/w$ per received ACK

Putting it all together

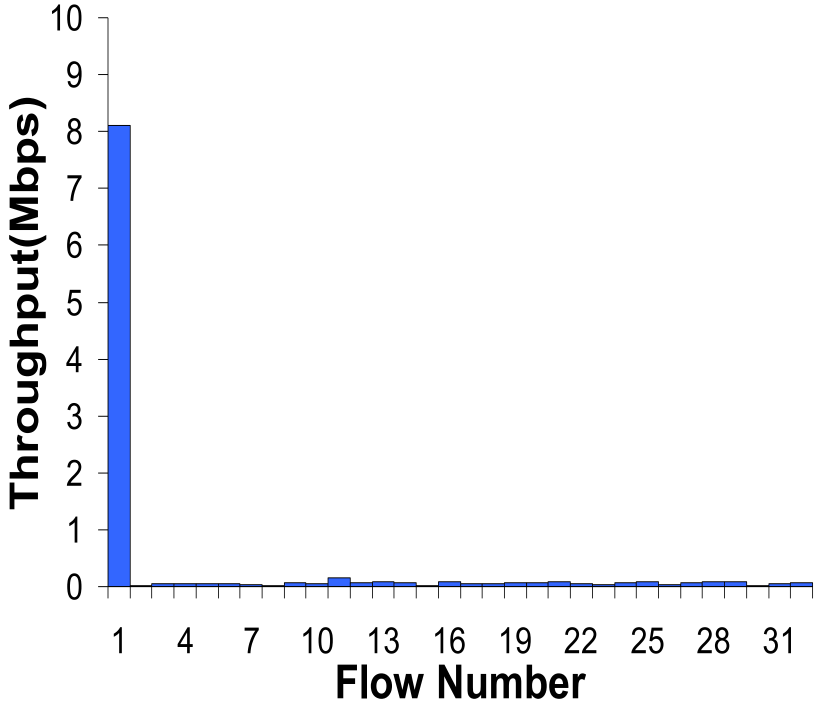


Fast Recovery and Fast Retransmit



TCP Friendliness

- **Can other protocols co-exist with TCP?**
  - E.g., if you want to write a video streaming app using UDP, how to do congestion control?



1 UDP Flow at 10MBps
31 TCP Flows
Sharing a 10MBps link

TCP Friendliness

- **Can other protocols co-exist with TCP?**
  - E.g., if you want to write a video streaming app using UDP, how to do congestion control?
- **Equation-based Congestion Control**
  - Instead of implementing TCP's CC, estimate the rate at which TCP would send. Function of what?
  - RTT, MSS, Loss
- **Measure RTT, Loss, send at that rate!**

TCP Throughput

- **Assume a TCP congestion of window W (segments), round-trip time of RTT, segment size MSS**
  - Sending Rate $S = W \text{ x } MSS / RTT$ (1)
- **Drop: W = W/2**
  - grows by MSS for $W/2$ $RTTs$, until another drop at $W \approx W$
- **Average window then 0.75x*S***
  - From (1), $S = 0.75\ W\ MSS / RTT$ (2)
- **Loss rate is 1 in number of packets between losses:**
  - $\text{Loss} = 1 / (1 + (W/2 + W/2+1 + W/2 + 2 + \ldots + W)$
    $= 1 / (3/8\ W^2)$ (3)

TCP Throughput (cont)

– Loss = 8/(3W²) $\Rightarrow W = \sqrt{\dfrac{8}{3 \cdot Loss}}$ (4)

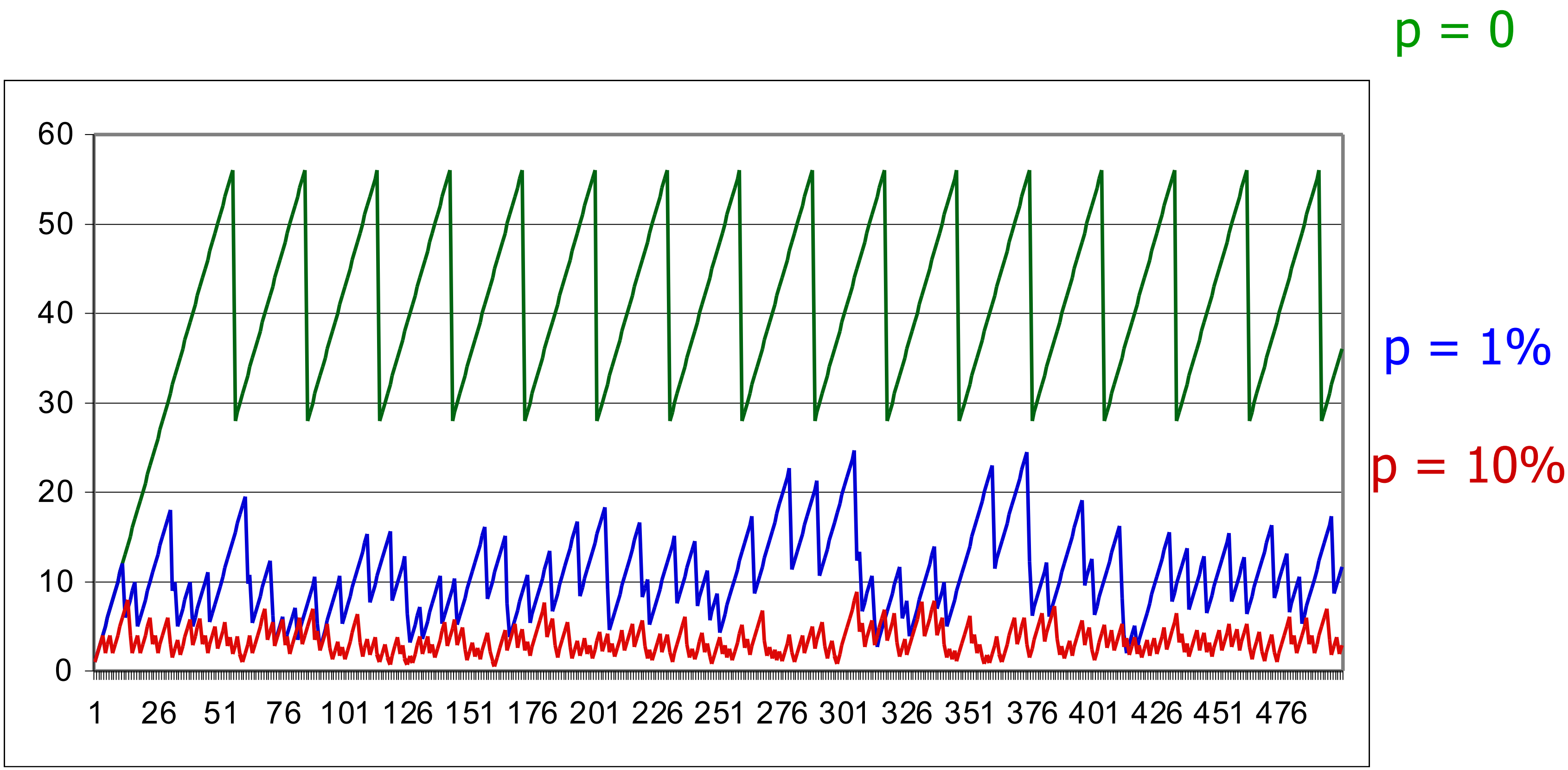
– Substituting (4) in (2), $S = 0.75\ W\ MSS\ /\ RTT$,

$$\text{Throughput} \approx 1.22 \times \frac{MSS}{RTT \cdot \sqrt{Loss}}$$

- Equation-based rate control can be TCP friendly and have better properties, e.g., small jitter, fast ramp-up…

What Happens When Link is Lossy?

- Throughput ≈ 1 / sqrt(Loss)



What can we do about it?

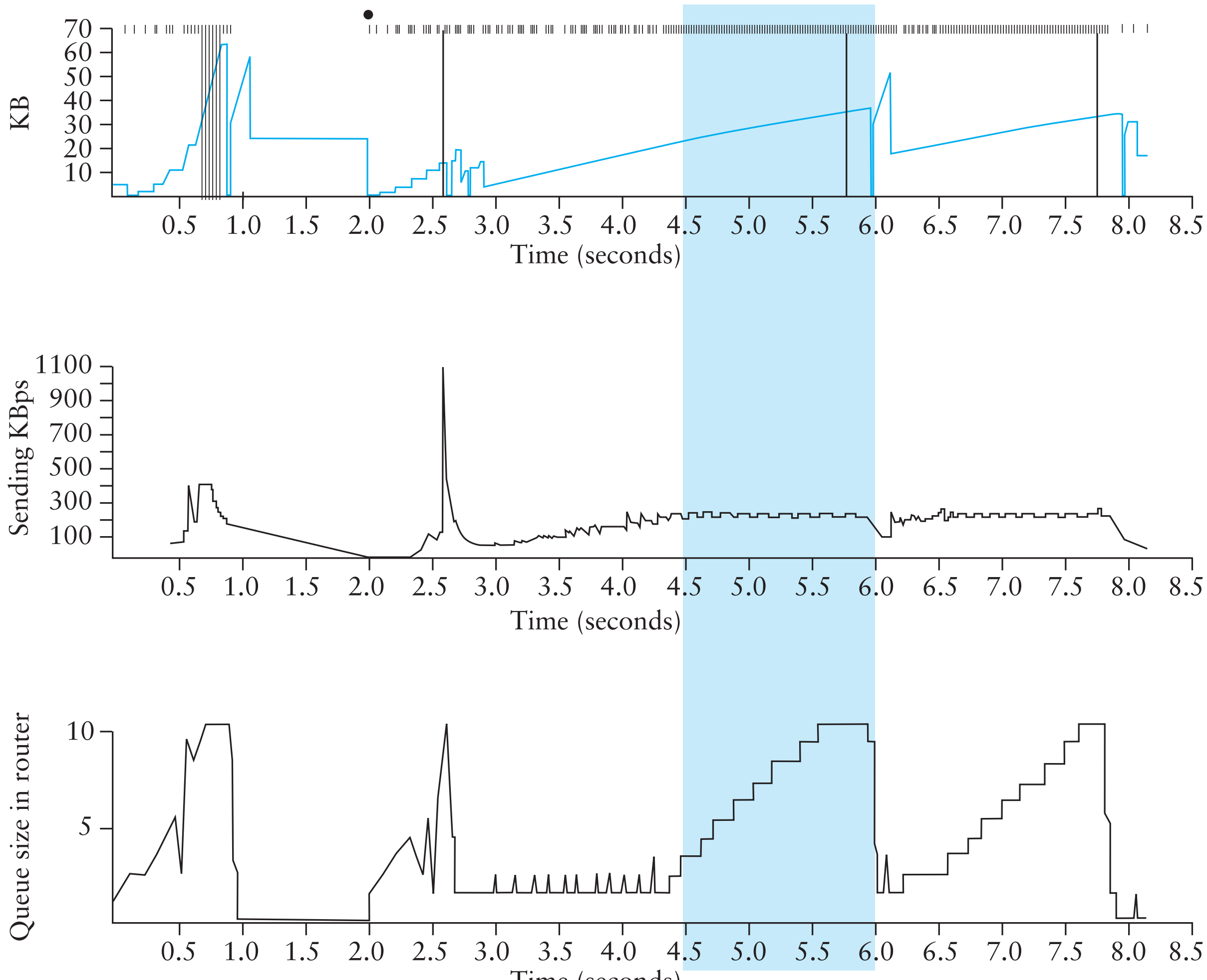
- **Two types of losses: congestion and corruption**
- **One option: mask corruption losses from TCP**
  - Retransmissions at the link layer
  - E.g. Snoop TCP: intercept duplicate acknowledgments, retransmit locally, filter them from the sender
- **Another option:**
  - Tell the sender about the cause for the drop
  - Requires modification to the TCP endpoints

Congestion Avoidance

- **TCP creates congestion to then back off**
  - Queues at bottleneck link are often full: increased delay
  - Sawtooth pattern: jitter
- **Alternative strategy**
  - Predict when congestion is about to happen
  - Reduce rate early
- **Two approaches**
  - Host centric: TCP Vegas (won't cover)
  - Router-centric: RED, ECN, DECBit, DCTCP

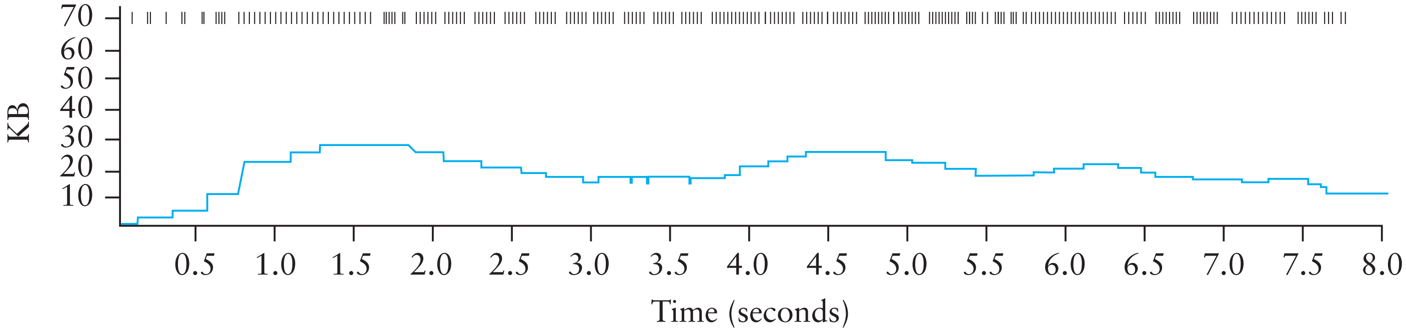
TCP Vegas

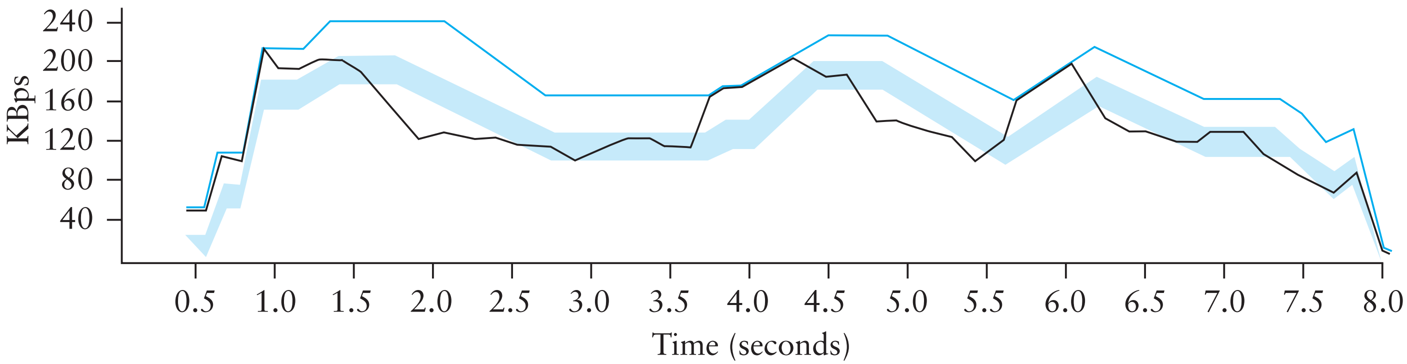
- **Idea: source watches for sign that router's queue is building up (e.g., sending rate flattens)**



TCP Vegas

- **Compare Actual Rate (A) with Expected Rate (E)**
  - If E-A > β, decrease cwnd linearly : A isn't responding
  - If E-A < α, increase cwnd linearly : Room for A to grow





Vegas

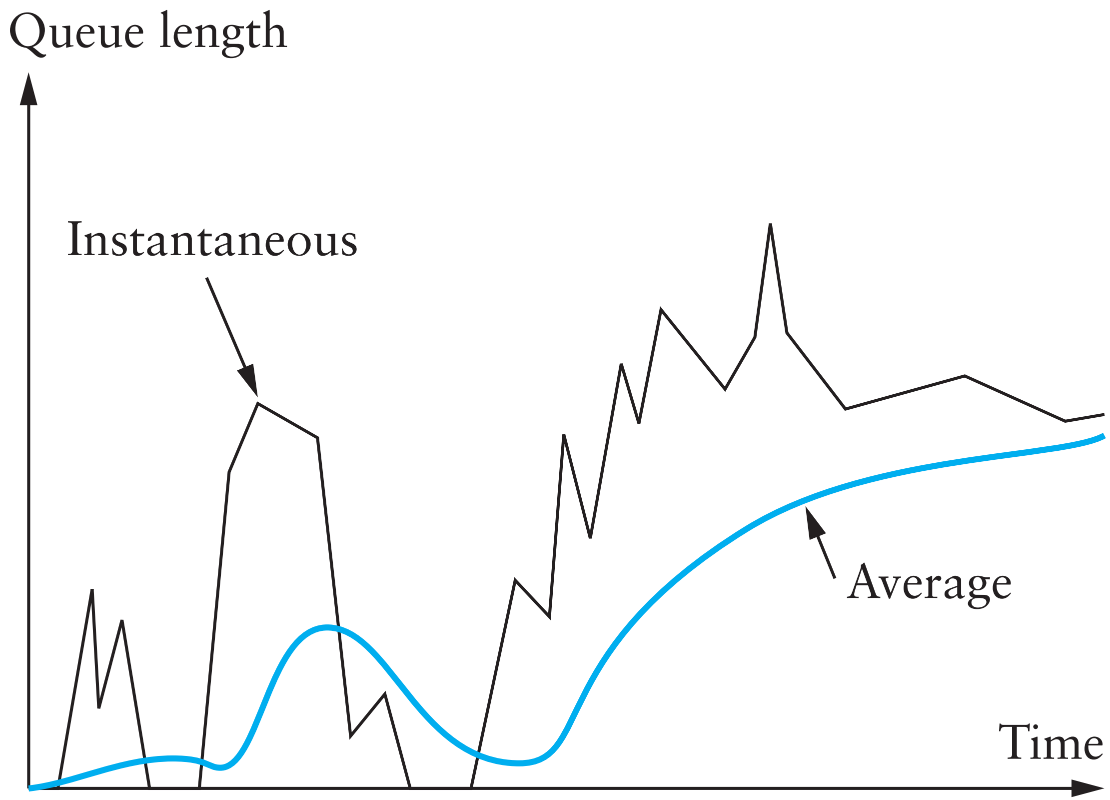
- **Shorter router queues**
- **Lower jitter**
- **Problem:**
  - Doesn't compete well with Reno. Why?
  - Reacts earlier, Reno is more aggressive, ends up with higher bandwidth…

Help from the network

- **What if routers could *tell* TCP that congestion is happening?**
  - Congestion causes queues to grow: rate mismatch
- **TCP responds to drops**
- **Idea: Random Early Drop (RED)**
  - Rather than wait for queue to become full, drop packet with some probability that increases with queue length
  - TCP will react by reducing cwnd
  - Could also mark instead of dropping: ECN

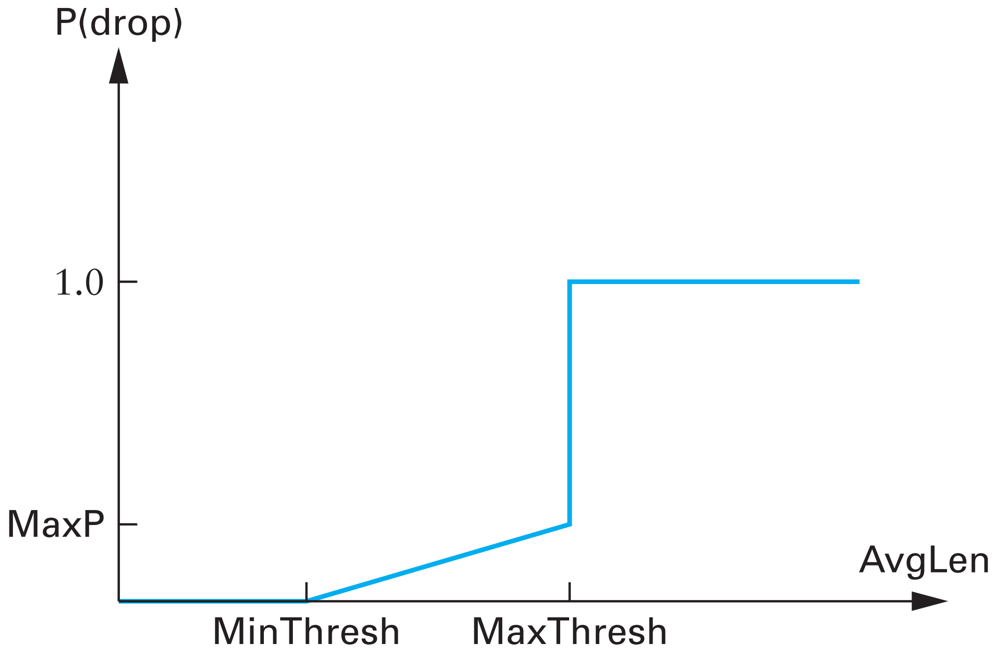
RED Details

- **Compute average queue length (EWMA)**
  - Don't want to react to very quick fluctuations



RED Drop Probability

- Define two thresholds: MinThresh, MaxThresh
- Drop probability:



- Improvements to spread drops (see book)

RED Advantages

- **Probability of dropping a packet of a particular flow is roughly proportional to the share of the bandwidth that flow is currently getting**
- **Higher network utilization with low delays**
- **Average queue length small, but can absorb bursts**
- **ECN**
  - Similar to RED, but router sets bit in the packet
  - Must be supported by both ends
  - Avoids retransmissions optionally dropped packets

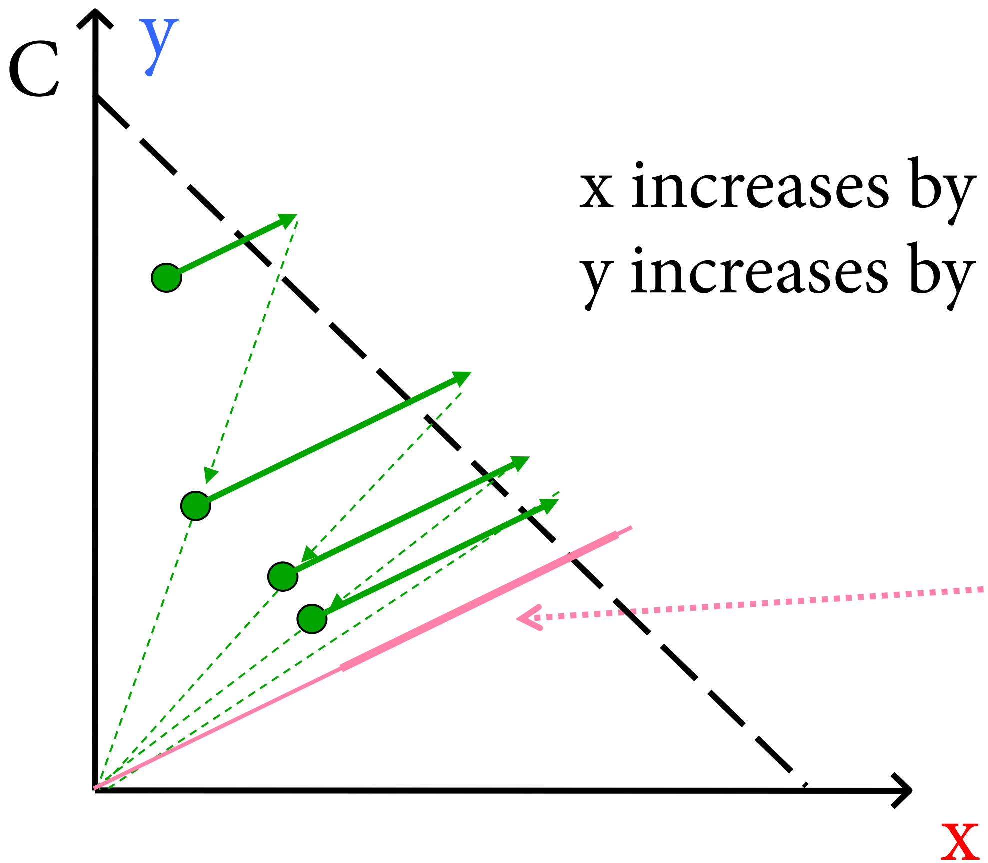
What happens if not everyone cooperates?

- **TCP works extremely well when its assumptions are valid**
  - All flows correctly implement congestion control
  - Losses are due to congestion

Cheating TCP

- **Possible ways to cheat**
  - Increasing cwnd faster
  - Large initial cwnd
  - Opening many connections
  - Ack Division Attack

Increasing cwnd Faster



x increases by 2 per RTT
y increases by 1 per RTT

Limit rates:
$x = 2y$

Figure from Walrand, Berkeley EECS 122, 2003

Larger Initial Window

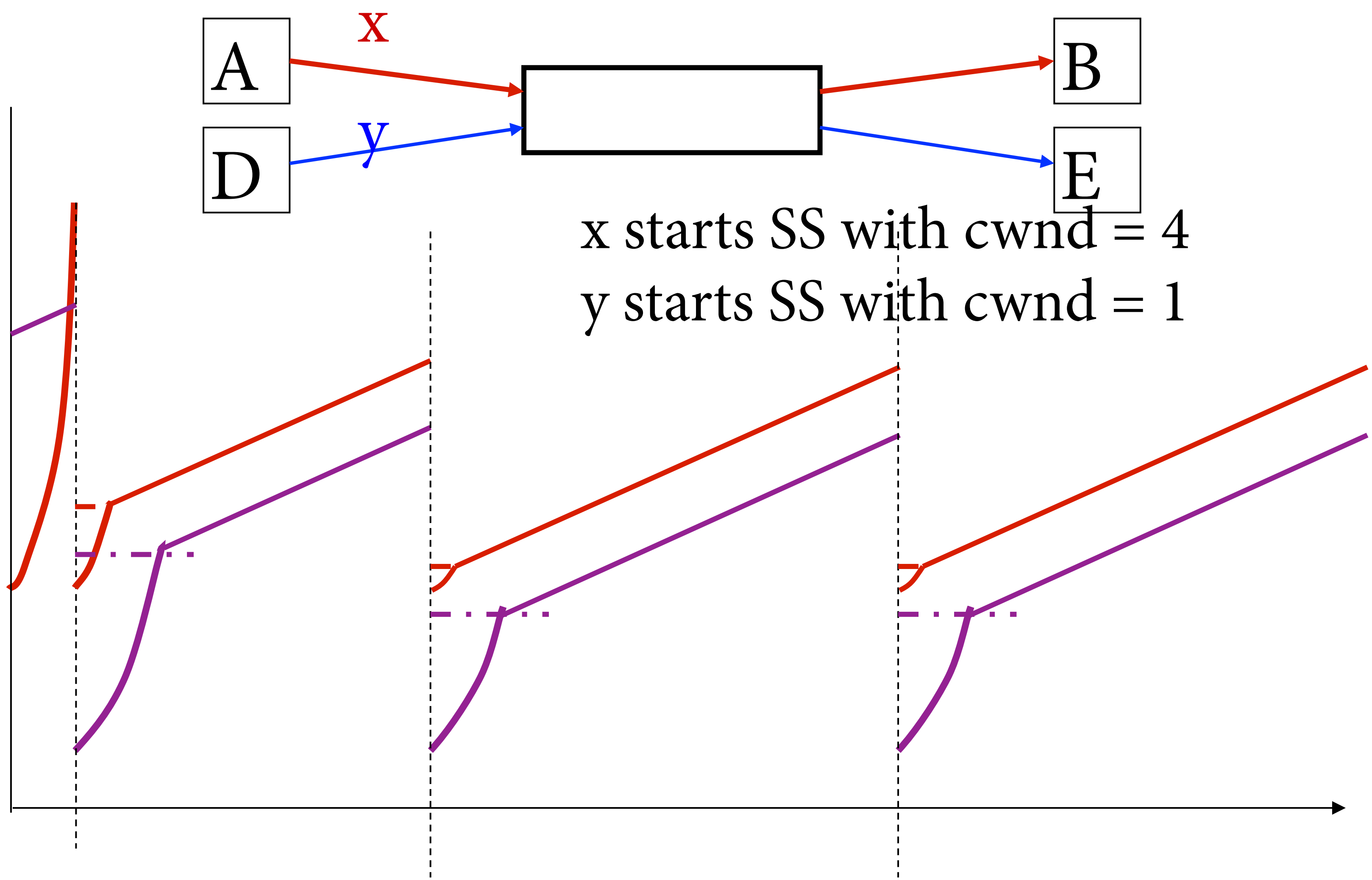
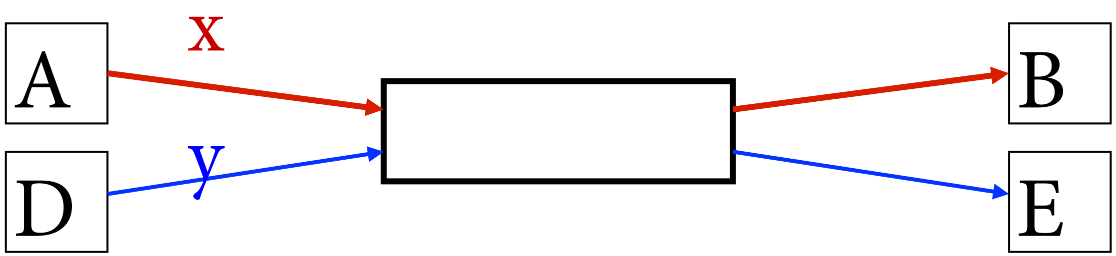


Figure from Walrand, Berkeley EECS 122, 2003

Open Many Connections

- **Web Browser: has to download k objects for a page**
  - Open many connections or download sequentially?



- **Assume:**
  - A opens 10 connections to B
  - B opens 1 connection to E
- **TCP is fair among connections**
  - A gets 10 times more bandwidth than B

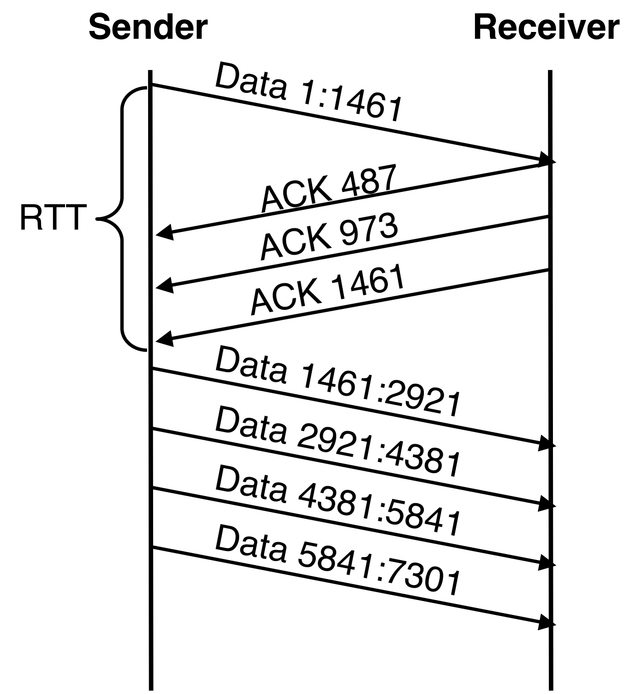
Figure from Walrand, Berkeley EECS 122, 2003

Exploiting Implicit Assumptions

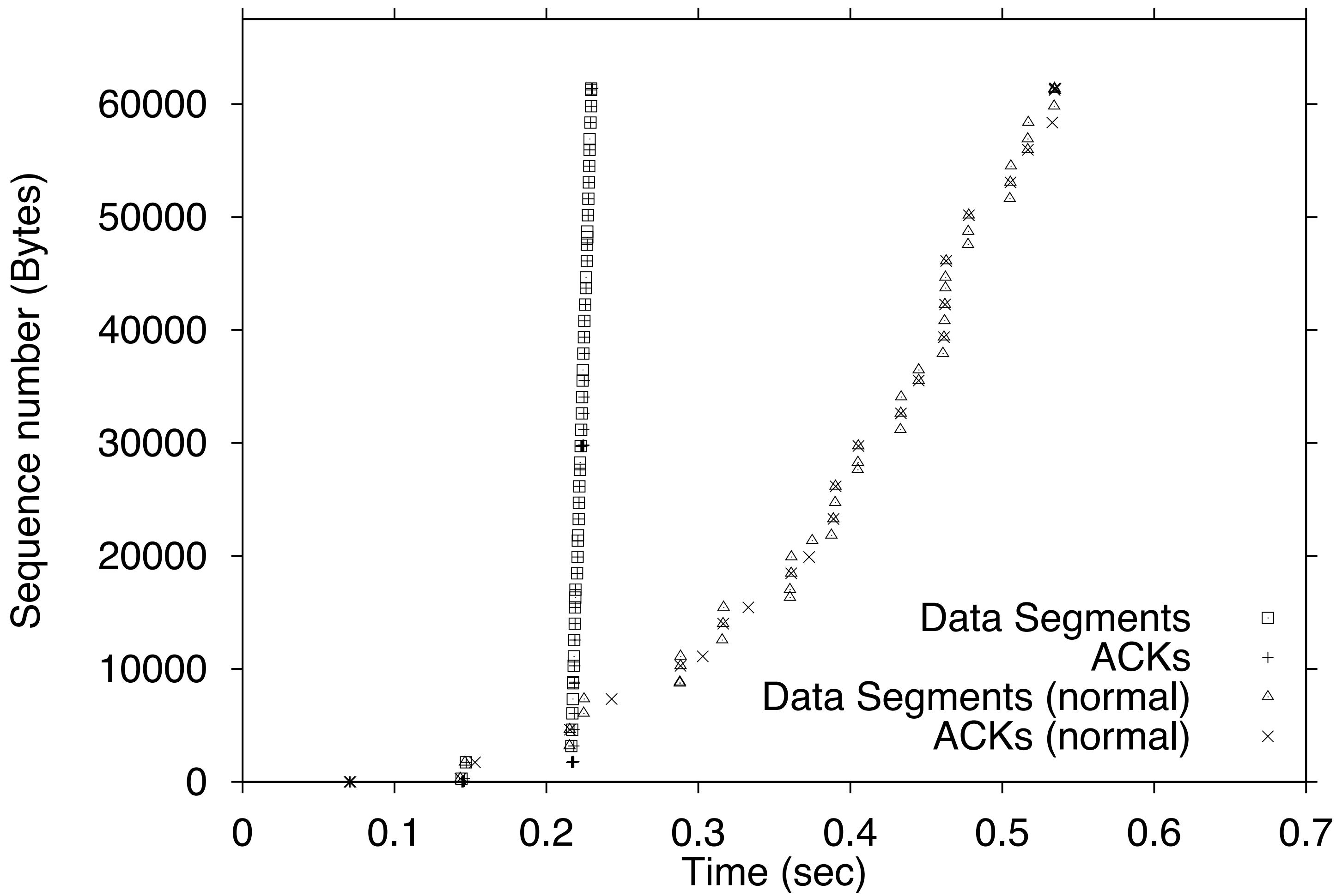
- **Savage, et al., CCR 1999:**
  - "TCP Congestion Control with a Misbehaving Receiver"
- **Exploits ambiguity in meaning of ACK**
  - ACKs can specify any byte range for error control
  - Congestion control assumes ACKs cover entire sent segments
- **What if you send multiple ACKs per segment?**

ACK Division Attack

- **Receiver: "***upon receiving a segment with N bytes, divide the bytes in M groups and acknowledge each group separately*"
- **Sender will grow window M times faster**
- **Could cause growth to 4GB in 4 RTTs!**
  - M = N = 1460



TCP Daytona!



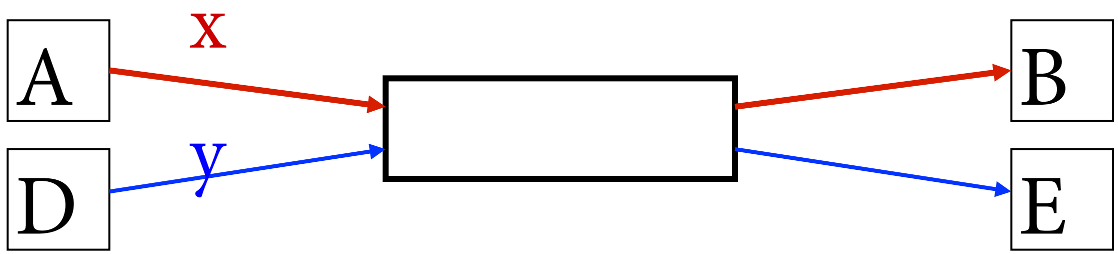
Sequence number (Bytes)
60000
50000
40000
30000
20000
10000
0
0
0.1
0.2
0.3
0.4
0.5
0.6
0.7
Time (sec)
Data Segments
ACKs
Data Segments (normal)
ACKs (normal)

Defense

- **Appropriate Byte Counting**
  - [RFC3465 (2003), RFC 5681 (2009)]
  - In slow start, cwnd += min (N, MSS)

  where N is the number of newly acknowledged bytes in the received ACK

Cheating TCP and Game Theory



A ↓	D → Increases by 1	Increases by 5
Increases by 1	22, 22	10, 35
Increases by 5	35, 10	15, 15

(x, y)

Too aggressive
→Losses
→Throughput falls

Individual incentives: cheating pays
Social incentives: better off without cheating

Classic PD: resolution depends on accountability

An alternative for reliability

- **Erasure coding**
  - Assume you can detect errors
  - Code is designed to tolerate entire missing packets
    - Collisions, noise, drops because of bit errors
  - Forward error correction
- **Examples: Reed-Solomon codes, LT Codes, Raptor Codes**
- **Property:**
  - From K source frames, produce B > K encoded frames
  - Receiver can reconstruct source with *any* K' frames, with K' *slightly* larger than K
  - Some codes can make B as large as needed, on the fly

LT Codes

- **Luby Transform Codes**
  - Michael Luby, circa 1998
- **Encoder: repeat B times**
  1. Pick a degree *d (*)*
  2. Randomly select *d* source blocks. Encoded block t_n= XOR or selected blocks

* The degree is picked from a distribution, *robust soliton distribution*, that guarantees that the decoding process will succeed with high probability

LT Decoder

- **Find an encoded block t_n with d=1**
- **Set $s_n = t_n$**
- **For all other blocks $t_{n'}$ that include s_n ,**

  set $t_{n'} = t_{n'}$ XOR s_n
- **Delete s_n from all encoding lists**
- **Finish if**
  1. You decode all source blocks, or
  2. You run out out blocks of degree 1

Next Time

- Move into the application layer
- DNS, Web, Security, and more…

Backup slides

- We didn't cover these in lecture: won't be in the exam, but you might be interested ☺

More help from the network

- **Problem: still vulnerable to malicious flows!**
  - RED will drop packets from large flows preferentially, but they don't have to respond appropriately
- **Idea: Multiple Queues (one per flow)**
  - Serve queues in Round-Robin
  - Nagle (1987)
  - Good: protects against misbehaving flows
  - Disadvantage?
  - Flows with larger packets get higher bandwidth

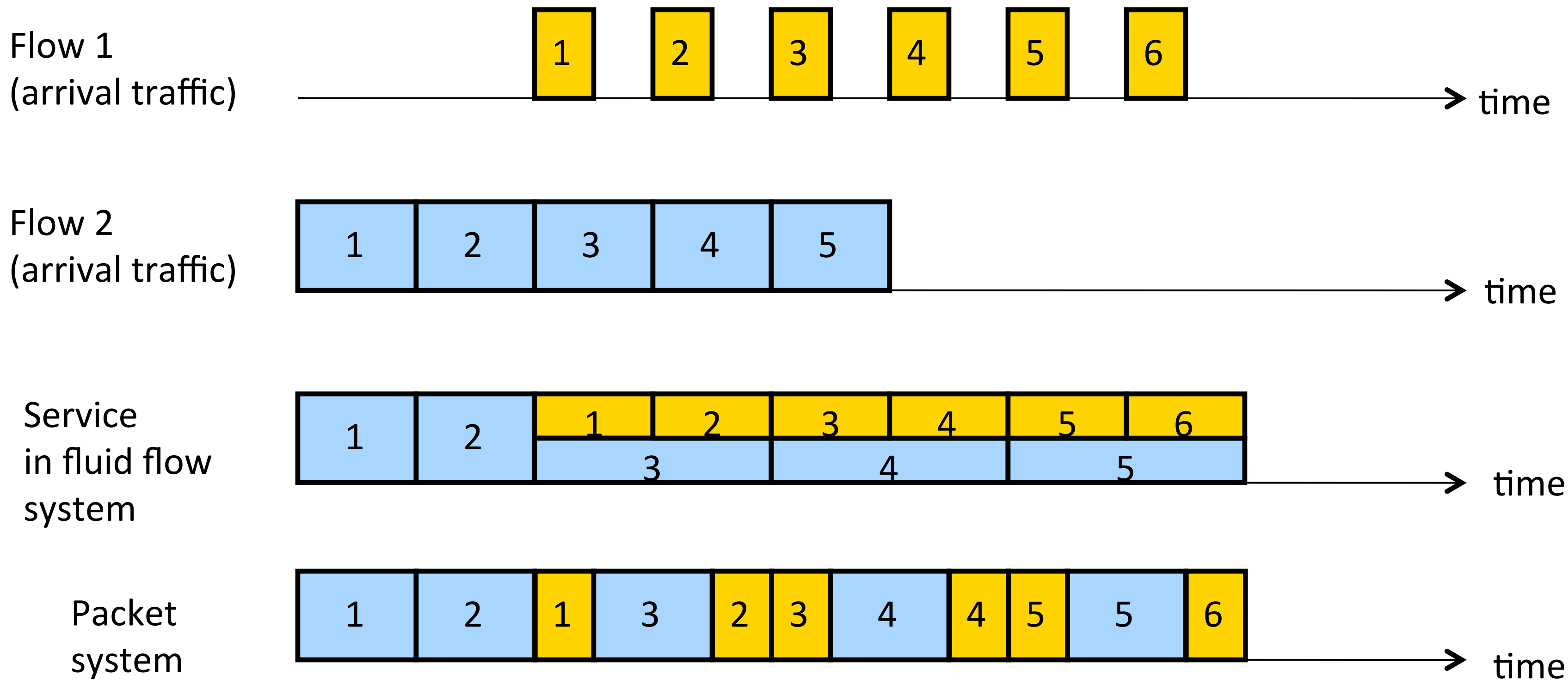
Solution

- **Bit-by-bit round robing**
- **Can we do this?**
  - No, packets cannot be preempted!
- **We can only approximate it…**

Fair Queueing

- **Define a *fluid flow* system as one where flows are served bit-by-bit**
- **Simulate *ff*, and serve packets in the order in which they would finish in the *ff* system**
- **Each flow will receive exactly its fair share**

Example



Flow 1
(arrival traffic)
1 2 3 4 5 6
time
Flow 2
(arrival traffic)
1 2 3 4 5
time
Service
in fluid flow
system
1 2
1 2 3 4 5 6
3 4 5
time
Packet
system
1 2 1 3 2 3 4 4 5 5 6
time

Implementing FQ

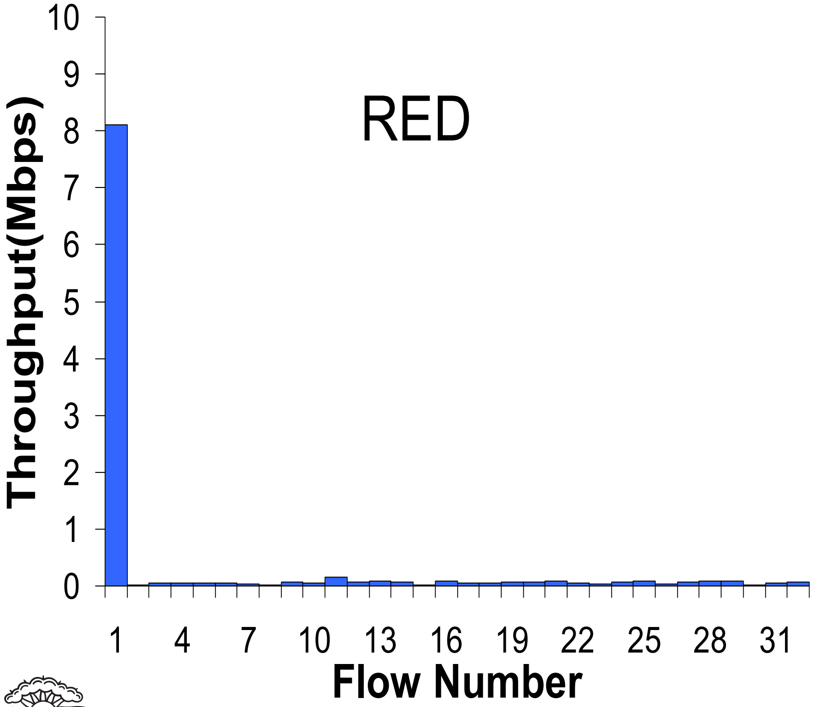
- **Suppose clock ticks with each bit transmitted**
  - (RR, among all active flows)
- **P_i is the length of the packet**
- **S_i is packet i's start of transmission time**
- **F_i is packet i's end of transmission time**
- **$F_i = S_i + P_i$**
- **When does router start transmitting packet i?**
  - If arrived before F_{i-1}, $S_i = F_{i-1}$
  - If no current packet for this flow, start when packet arrives (call this A_i): $S_i = A_i$
- **Thus, $F_i = \max(F_{i-1}, A_i) + P_i$**

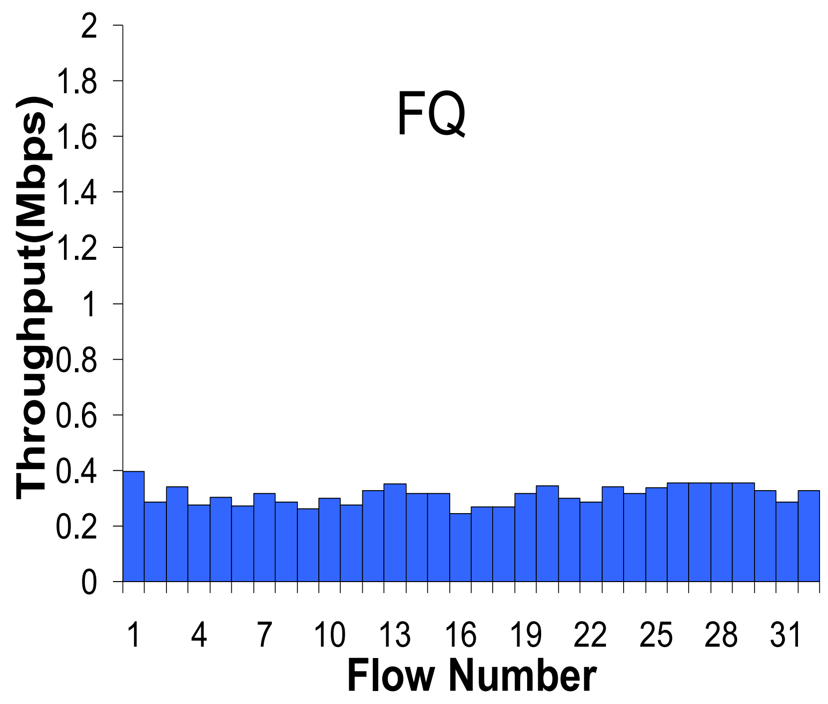
Fair Queueing

- **Across all flows**
  - Calculate F_i for each packet that arrives on each flow
  - Next packet to transmit is that with the lowest F_i
  - Clock rate depends on the number of flows
- **Advantages**
  - Achieves max-min fairness, independent of sources
  - Work conserving
- **Disadvantages**
  - Requires non-trivial support from routers
  - Requires reliable identification of flows
  - Not perfect: can't preempt packets

Fair Queueing Example

- **10Mbps link, 1 10Mbps UDP, 31 TCPs**





Big Picture

- **Fair Queuing doesn't eliminate congestion: just manages it**
- **You need both, ideally:**
  - End-host congestion control to adapt
  - Router congestion control to provide isolation