CSCI-1680
Transport Layer II
Data over TCP

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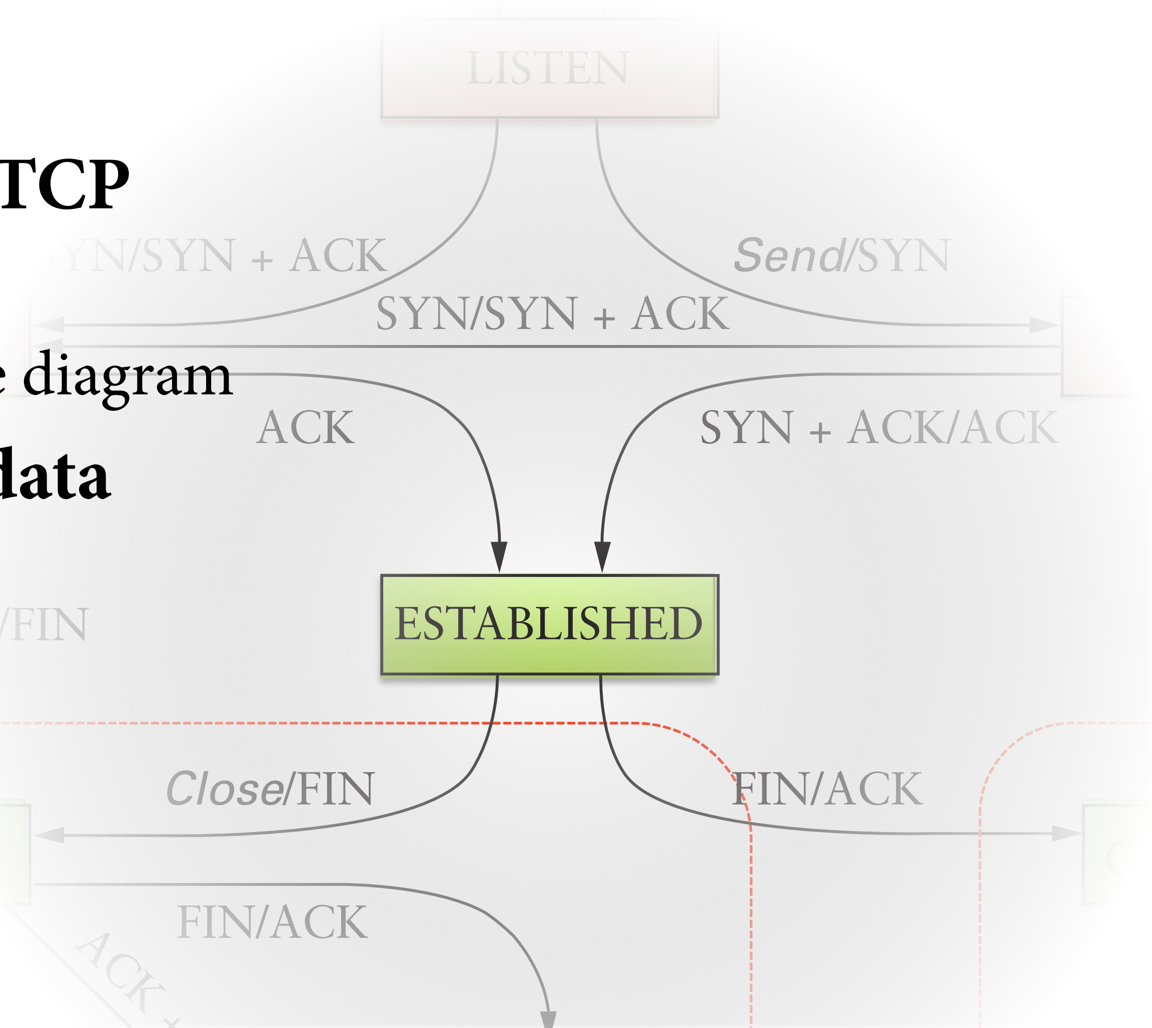
Based partly on lecture notes by David Mazières, Phil Levis, John Jannotti

Last Class

- **Introduction to TCP**
  - Header format
  - Connection state diagram
- **Today: sending data**



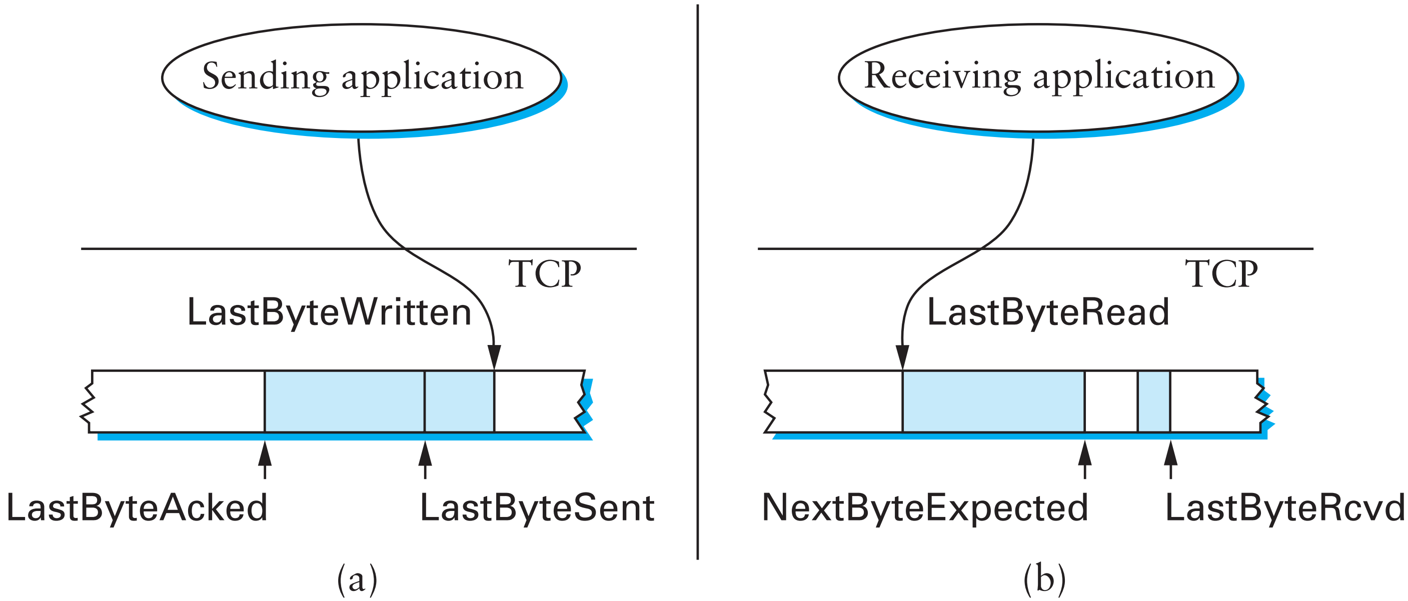
First Goal

- **We should not send more data than the receiver can take: *flow control***
- **When to send data?**
  - Sender can delay sends to get larger segments
- **How much data to send?**
  - Data is sent in MSS-sized segments
    - Chosen to avoid fragmentation

Flow Control

- Part of TCP specification (even before 1988)
- Receiver uses window header field to tell sender how much space it has

Flow Control



- **Receiver:** **AdvertisedWindow**

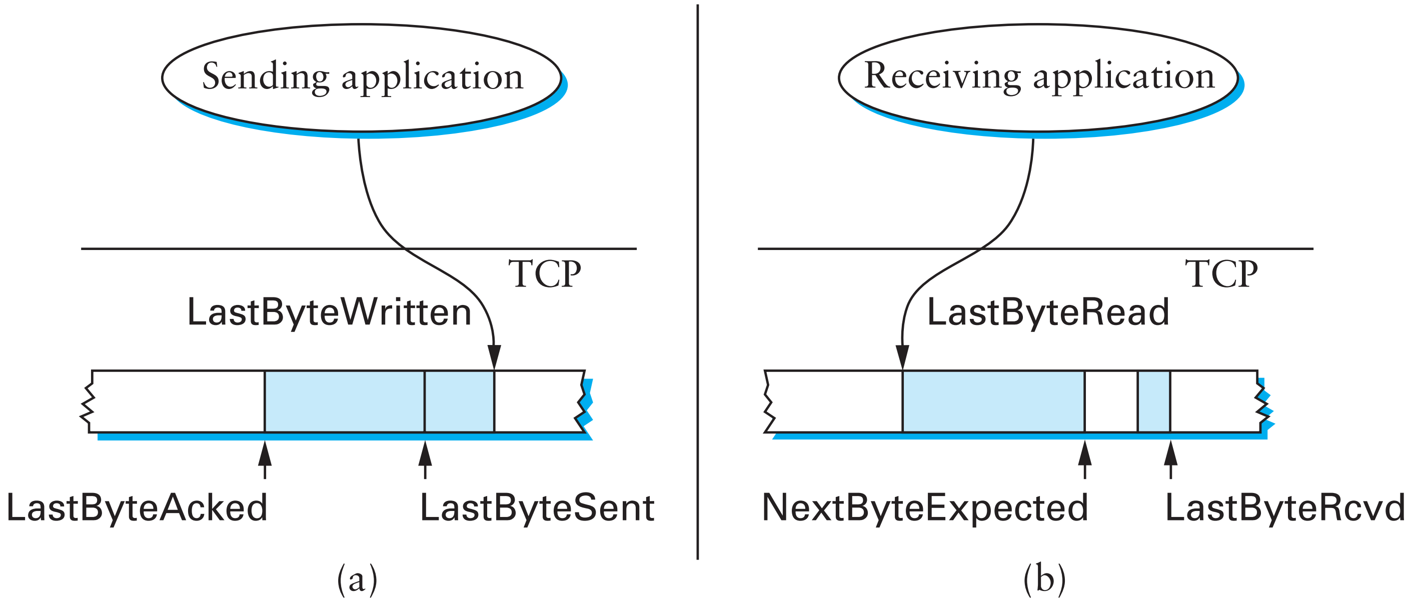
  = MaxRcvBuffer – ((NextByteExpected-1) – LastByteRead)

- **Sender:** LastByteSent – LastByteAcked <= AdvertisedWindow

  EffectiveWindow = AdvertisedWindow – (BytesInFlight)

  LastByteWritten – LastByteAcked <= MaxSendBuffer

Flow Control



- **Advertised window can fall to 0**
  - How?
  - Sender eventually stops sending, blocks application
- **Sender keeps sending 1-byte segments until window comes back > 0**

When to Transmit?

- **Nagle's algorithm**
- **Goal: reduce the overhead of small packets**

```
If available data and window >= MSS
    Send a MSS segment
else
    If there is unAcked data in flight
      buffer the new data until ACK arrives
    else
      send all the new data now
```

- **Receiver should avoid advertising a window <= MSS after advertising a window of 0**

Delayed Acknowledgments

- **Goal: Piggy-back ACKs on data**
  - Delay ACK for 200ms in case application sends data
  - If more data received, immediately ACK second segment
  - Note: never delay duplicate ACKs (if missing a segment)
- **Warning: can interact *very* badly with Nagle**
  - Temporary deadlock
  - Can disable Nagle with TCP_NODELAY
  - Application can also avoid many small writes

Limitations of Flow Control

- Network may be the bottleneck
- Signal from receiver not enough!
- Sending too fast will cause queue overflows, heavy packet loss
- Flow control provides *correctness*
- Need more for performance: congestion control

Second goal

- **We should not send more data than the network can take: *congestion control***

A Short History of TCP

- **1974: 3-way handshake**
- **1978: IP and TCP split**
- **1983: January 1st, ARPAnet switches to TCP/IP**
- **1984: Nagle predicts congestion collapses**
- **1986: Internet begins to suffer congestion collapses**
  - LBL to Berkeley drops from 32Kbps to 40bps
- **1987/8: Van Jacobson fixes TCP, publishes seminal paper*: (TCP Tahoe)**
- **1990: Fast transmit and fast recovery added (TCP Reno)**

* Van Jacobson. Congestion avoidance and control. SIGCOMM '88

Congestion Collapse
Nagle, rfc896, 1984

- **Mid 1980's. Problem with the protocol *implementations*, not the protocol!**
- **What was happening?**
  - Load on the network → buffers at routers fill up → round trip time increases
- **If close to capacity, and, e.g., a large flow arrives suddenly…**
  - RTT estimates become too short
  - Lots of retransmissions → increase in queue size
  - Eventually many drops happen (full queues)
  - Fraction of useful packets (not copies) decreases

TCP Congestion Control

- **3 Key Challenges**
  - Determining the available capacity in the first place
  - Adjusting to changes in the available capacity
  - Sharing capacity between flows
- **Idea**
  - Each source determines network capacity for itself
  - Rate is determined by window size
  - Uses implicit feedback (drops, delay)
  - ACKs pace transmission (self-clocking)

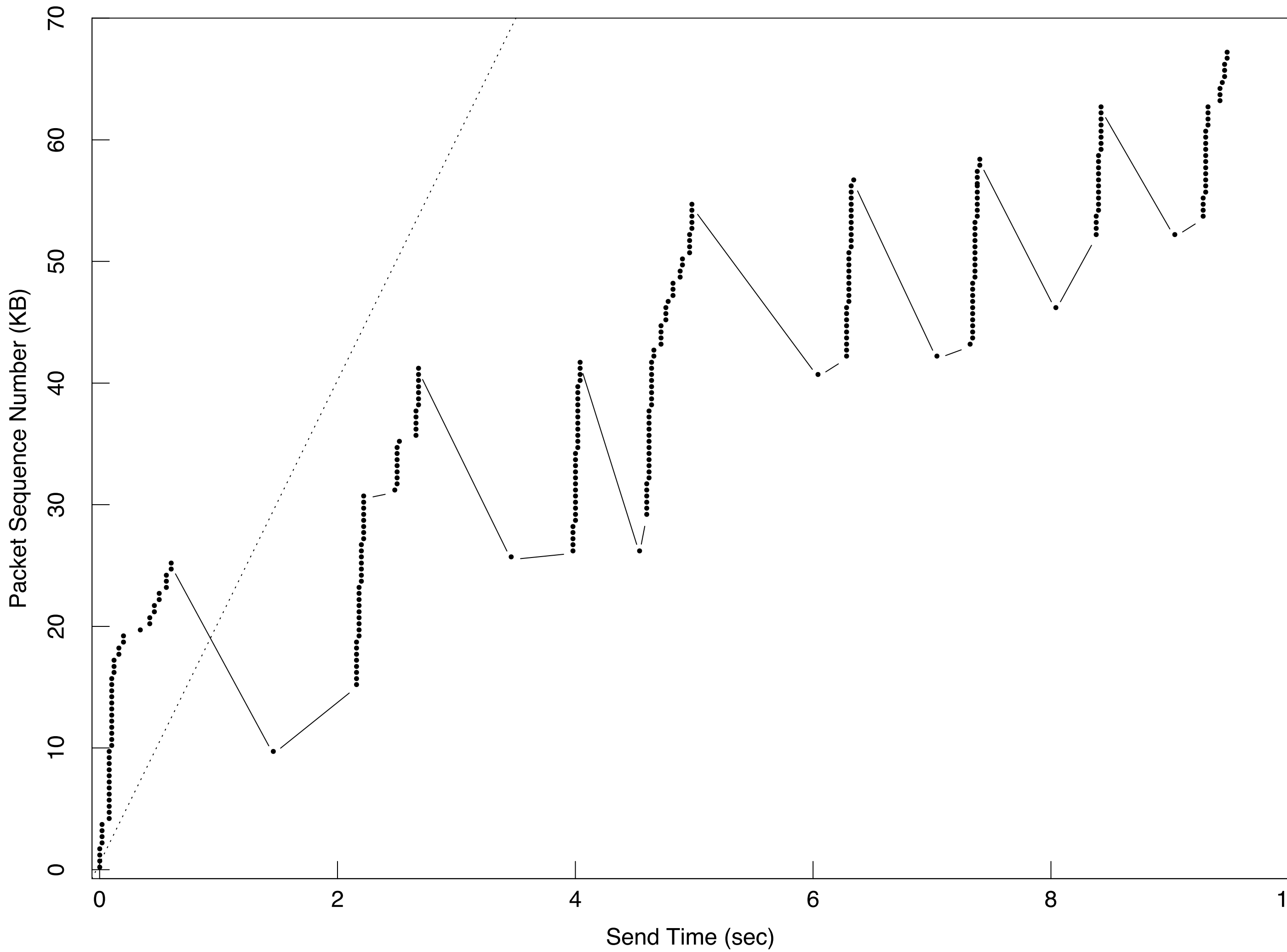
Dealing with Congestion

- **TCP keeps *congestion* and *flow control* windows**
  - Max packets in flight is lesser of two
- **Sending rate: ~Window/RTT**
- **The key here is how to set the congestion window to respond to congestion signals**

Starting Up

- **Before TCP Tahoe**
  – On connection, nodes send full (rcv) window of packets
  – Retransmit packet immediately after its timer expires
- **Result: window-sized bursts of packets in network**

Bursts of Packets

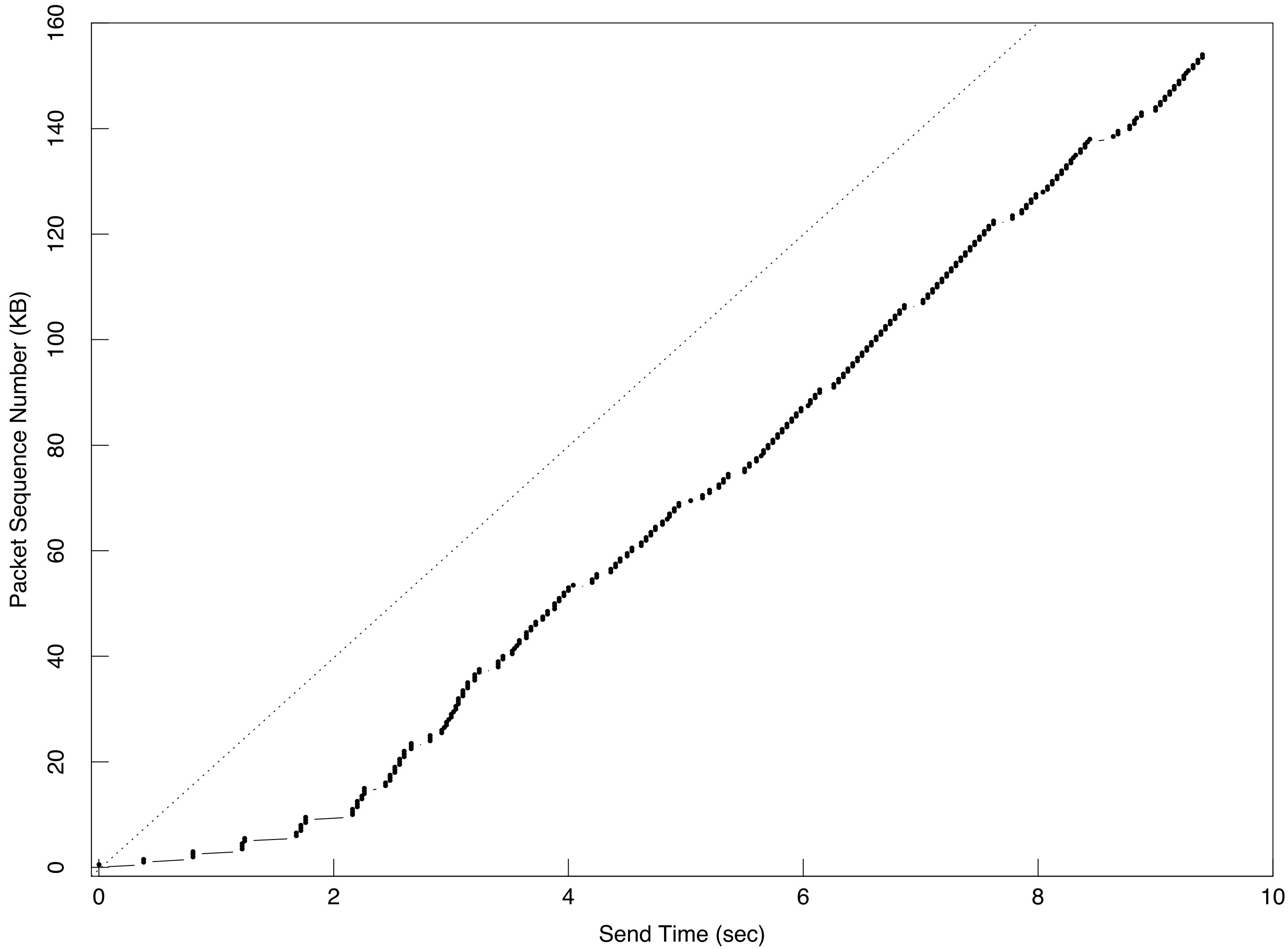


Graph from Van Jacobson and Karels, 1988

Determining Initial Capacity

- **Question: how do we set w initially?**
  - Should start at 1MSS (to avoid overloading the network)
  - Could increase additively until we hit congestion
  - May be too slow on fast network
- **Start by doubling w each RTT**
  - Then will dump at most one extra window into network
  - This is called *slow start*
- ***Slow start*, this sounds quite fast!**
  - In contrast to initial algorithm: sender would dump entire *flow control* window at once

Startup behavior with Slow Start



Slow start implementation

- **Let w be the size of the window in *bytes***
  - We have w/MSS segments per RTT
- **We are doubling w after each RTT**
  - We receive w/MSS ACKs each RTT
  - So we can set w = w + MSS on every ACK
- **At some point we hit the network limit.**
  - Experience loss
  - We are at most one window size above the limit
  - Remember window size (ssthreah) and reduce window

Dealing with Congestion

- Assume losses are due to congestion
- After a loss, reduce congestion window
  - How much to reduce?
- Idea: conservation of packets at equilibrium
  - Want to keep roughly the same number of packets network
  - Analogy with water in fixed-size pipe
  - Put new packet into network when one exits

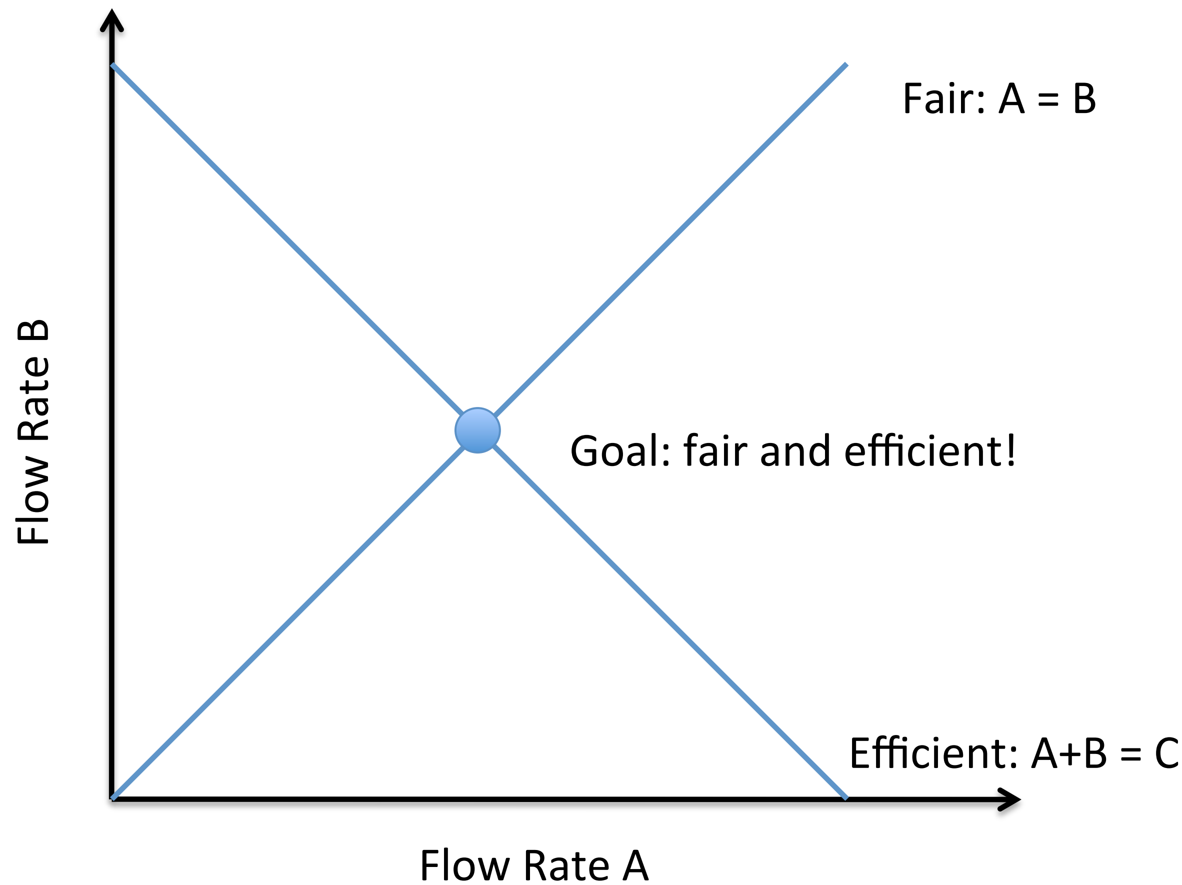
How much to reduce window?

- **Crude model of the network**
  - Let L_i be the load (# pkts) in the network at time I
  - If network uncongested, roughly constant $L_i = N$
- **What happens under congestion?**
  - Some fraction γ of packets can't exit the network
  - Now $L_i = N + \gamma L_{i-1}$, or $L_i \approx g^i L_0$
  - Exponential increase in congestion
- **Sources must decrease offered rate exponentially**
  - i.e, multiplicative decrease in window size
  - TCP chooses to cut window in half

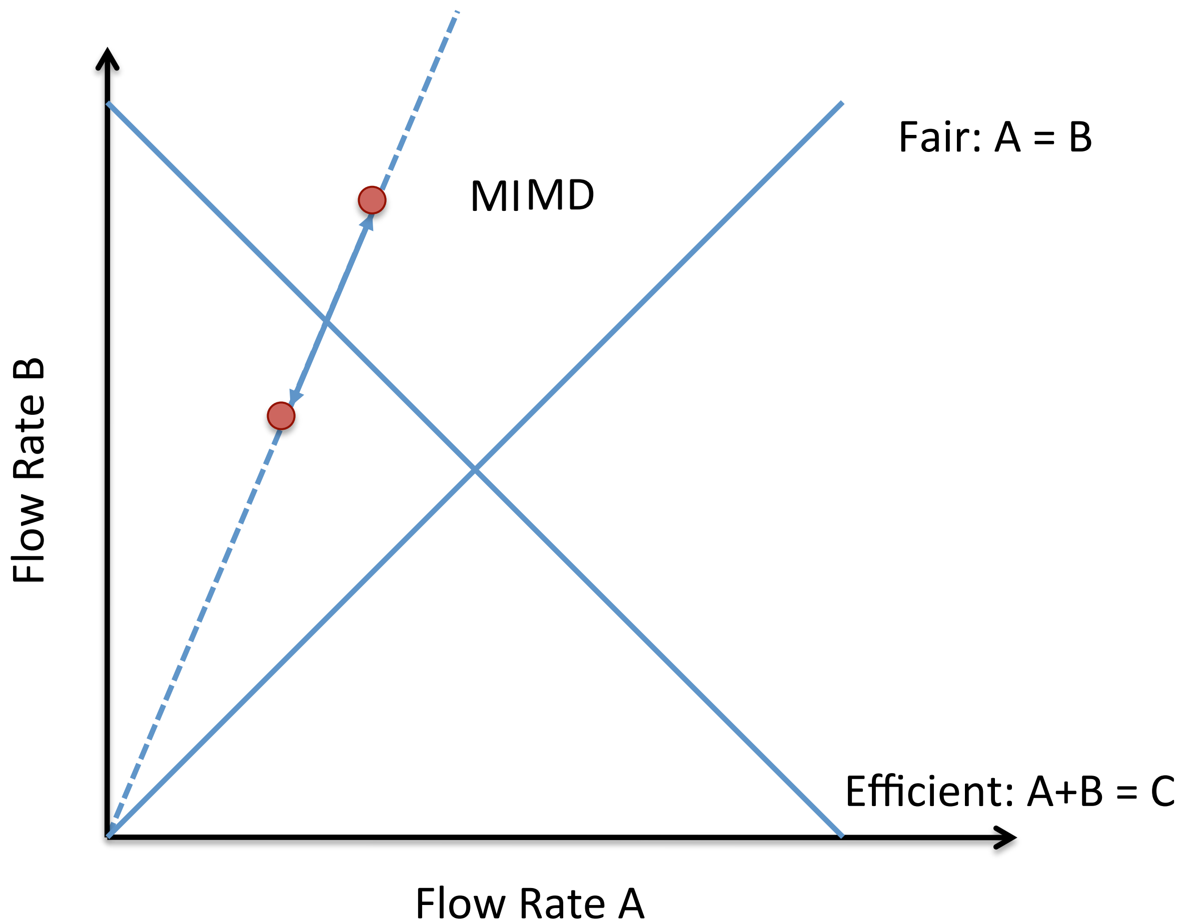
How to use extra capacity?

- **Network signals congestion, but says nothing of underutilization**
  - Senders constantly try to send faster, see if it works
  - So, increase window if no losses… By how much?
- **Multiplicative increase?**
  - Easier to saturate the network than to recover
  - Too fast, will lead to saturation, wild fluctuations
- **Additive increase?**
  - Won't saturate the network
  - Remember fairness (third challenge)?

Chiu Jain Phase Plots



Chiu Jain Phase Plots



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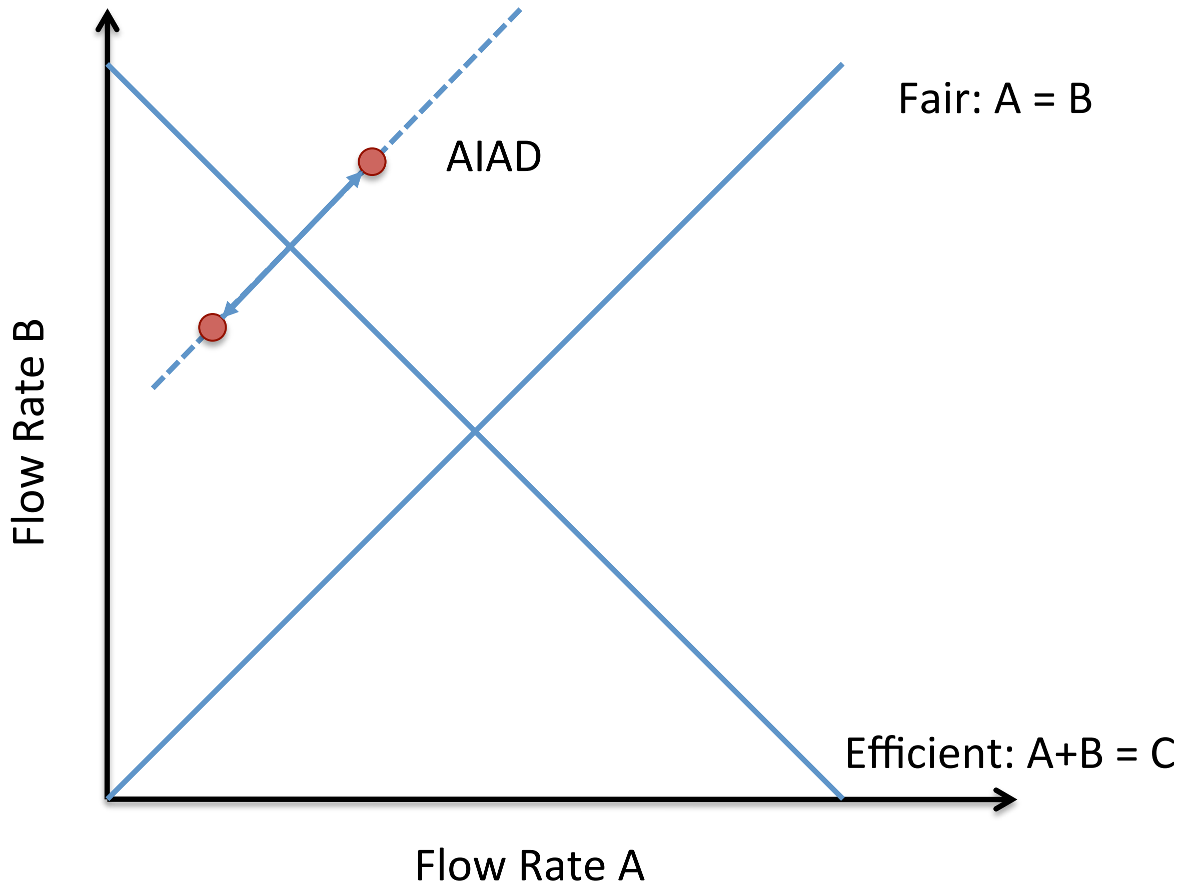
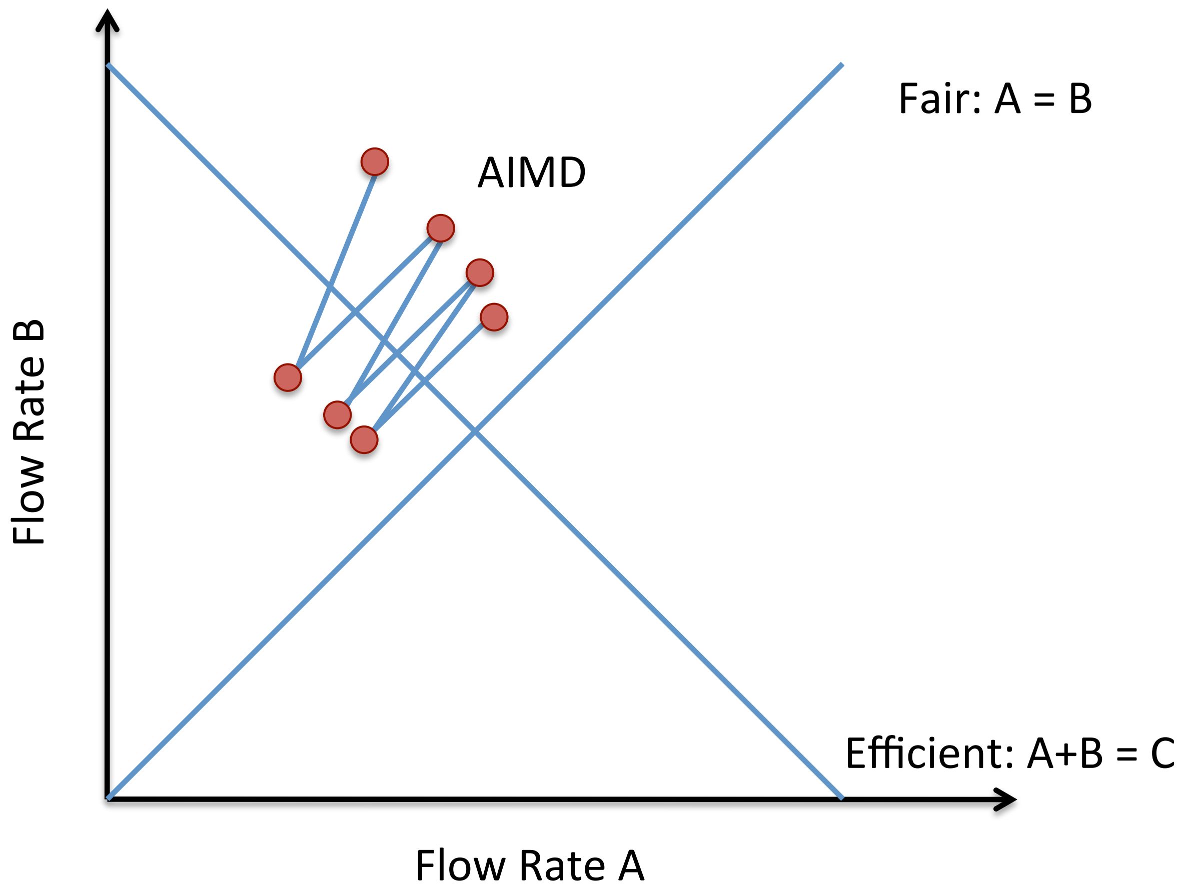


Chiu Jain Phase Plots

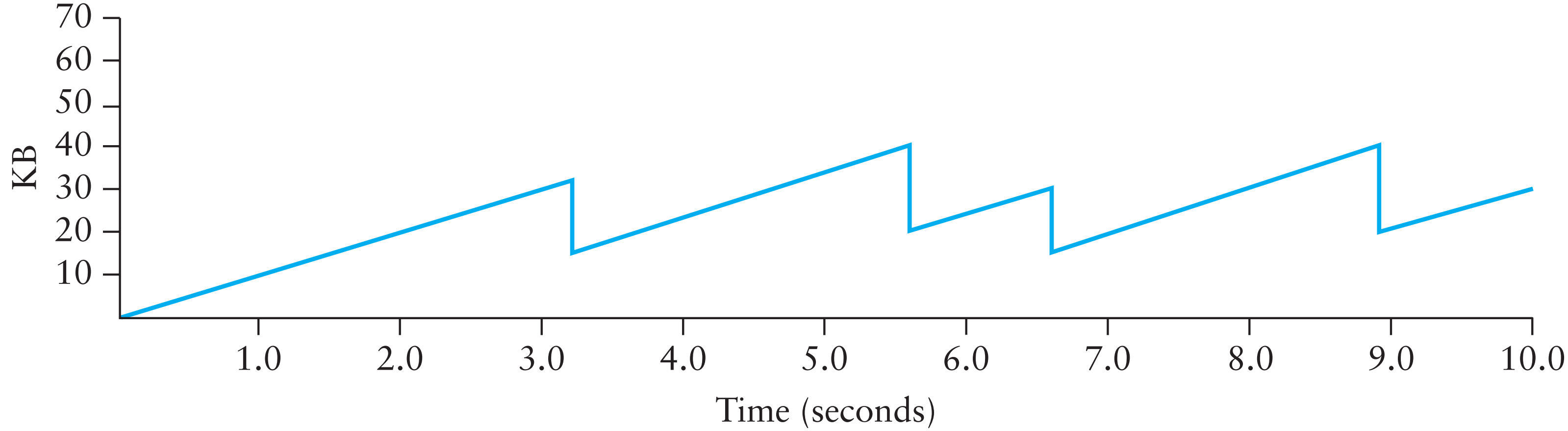


AIMD Implementation

- **In practice, send MSS-sized segments**
  - Let window size in bytes be w (a multiple of MSS)
- **Increase:**
  - After w bytes ACKed, could set w = w + MSS
  - Smoother to increment on each ACK
    - w = w + MSS * MSS/w
    - (receive w/MSS ACKs per RTT, increase by MSS/(w/MSS) for each)
- **Decrease:**
  - After a packet loss, w = w/2
  - But don't want w < MSS
  - So react differently to multiple consecutive losses
  - Back off exponentially (pause with no packets in flight)

AIMD Trace

- **AIMD produces sawtooth pattern of window size**
  - Always probing available bandwidth



Putting it together

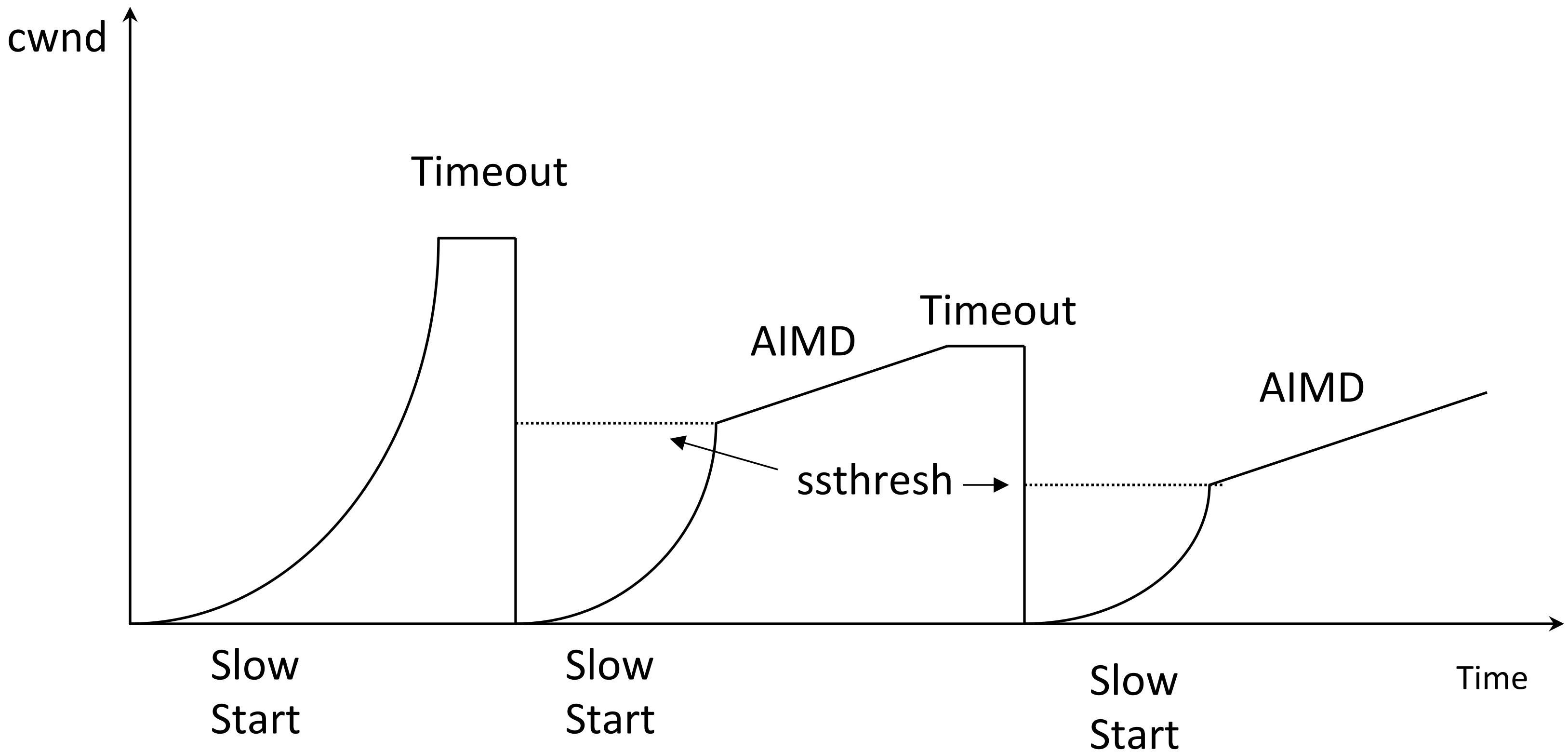
- **TCP has two states: Slow Start (SS) and Congestion Avoidance (CA)**
- **A window size threshold governs the state transition**
  - Window <= threshold: SS
  - Window > threshold: congestion avoidance
- **States differ in how they respond to ACKs**
  - Slow start: $w = w + MSS$
  - Congestion Avoidance: $w = w + MSS^2/w$ (1 MSS per RTT)
- **On loss event: set $w = 1$, slow start**

How to Detect Loss

- **Timeout**
- **Any other way?**
  - Gap in sequence numbers at receiver
  - Receiver uses cumulative ACKs: drops => duplicate ACKs
- **3 Duplicate ACKs considered loss**

- **Which one is worse?**

Putting it all together



RTT

- **We want an estimate of RTT so we can know a packet was likely lost, and not just delayed**
- **Key for correct operation**
- **Challenge: RTT can be highly variable**
  - Both at long and short time scales!
- **Both average and variance increase a lot with load**
- **Solution**
  - Use exponentially weighted moving average (EWMA)
  - Estimate deviation as well as expected value
  - Assume packet is lost when time is well beyond reasonable deviation

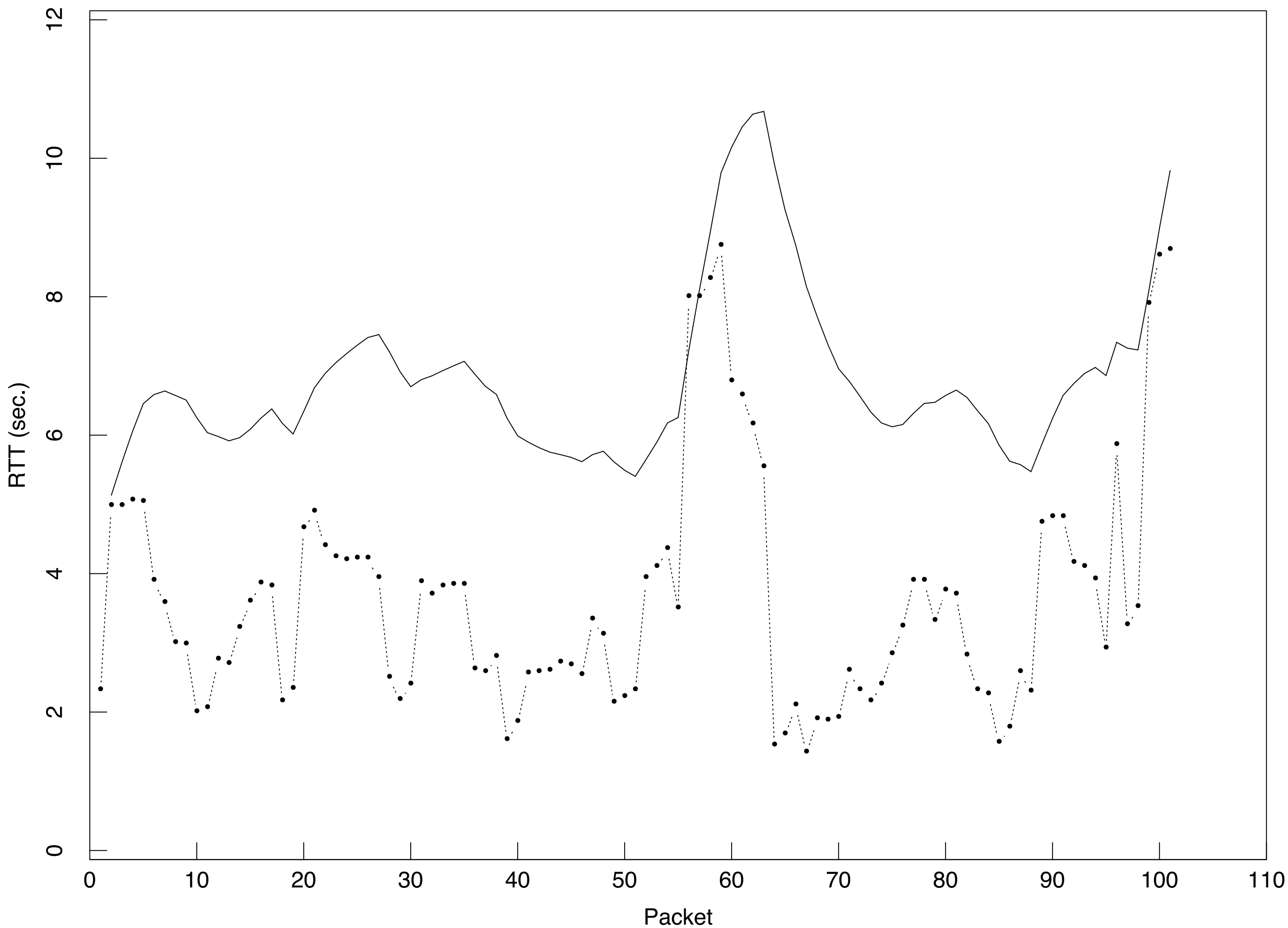
Originally

- **$EstRTT = (1 - \alpha) \times EstRTT + \alpha \times SampleRTT$**
- **$Timeout = 2 \times EstRTT$**
- **Problem 1:**
  - in case of retransmission, ack corresponds to which send?
  - Solution: only sample for segments with no retransmission
- **Problem 2:**
  - does not take variance into account: too aggressive when there is more load!

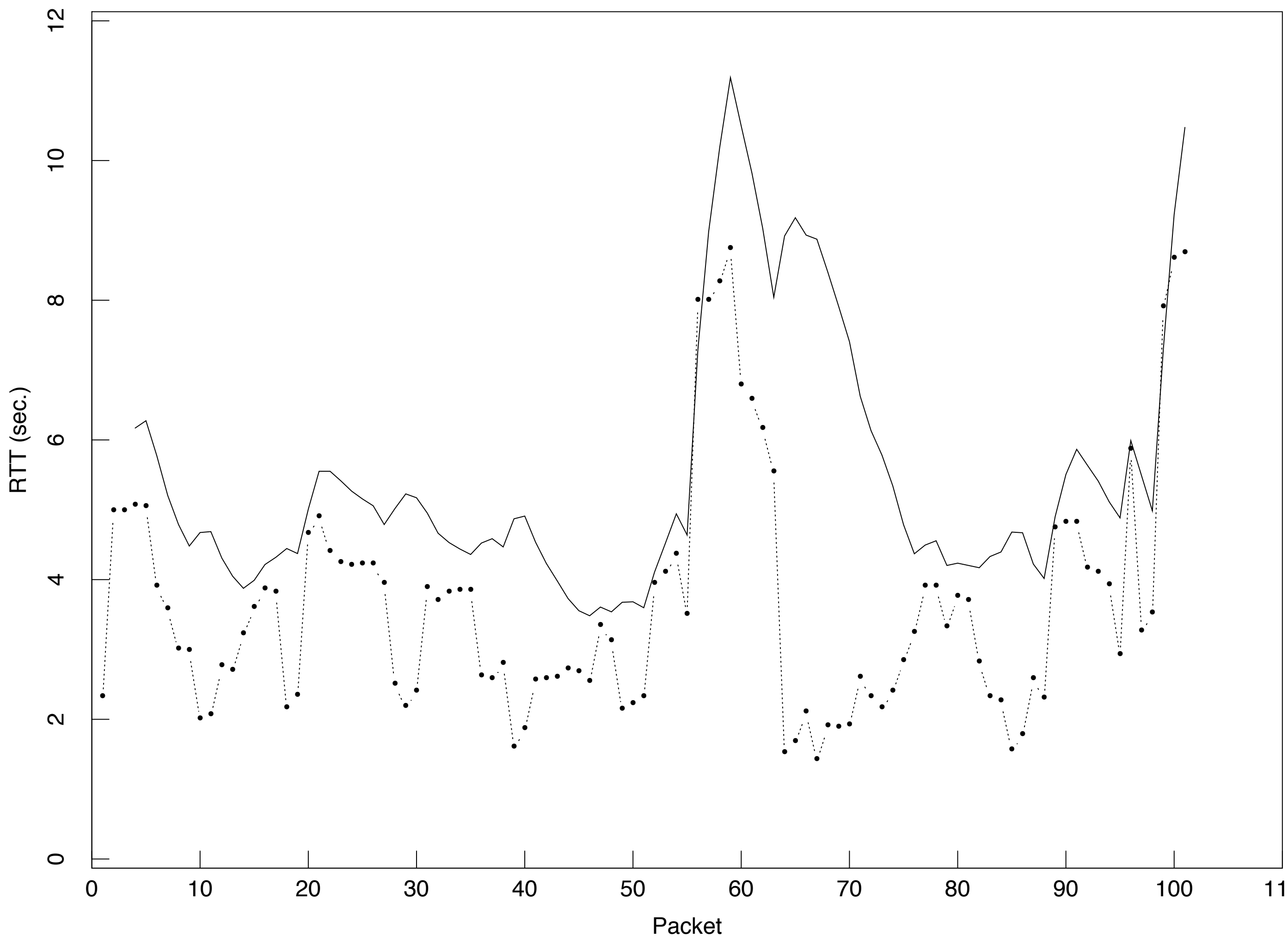
Jacobson/Karels Algorithm (Tahoe)

- **EstRTT = (1 – α) × EstRTT + α × SampleRTT**
  - Recommended α is 0.125
- **DevRTT = (1 – β) × DevRTT + β | SampleRTT – EstRTT |**
  - Recommended β is 0.25
- **Timeout = EstRTT + 4 DevRTT**
- **For successive retransmissions: use exponential backoff**

Old RTT Estimation



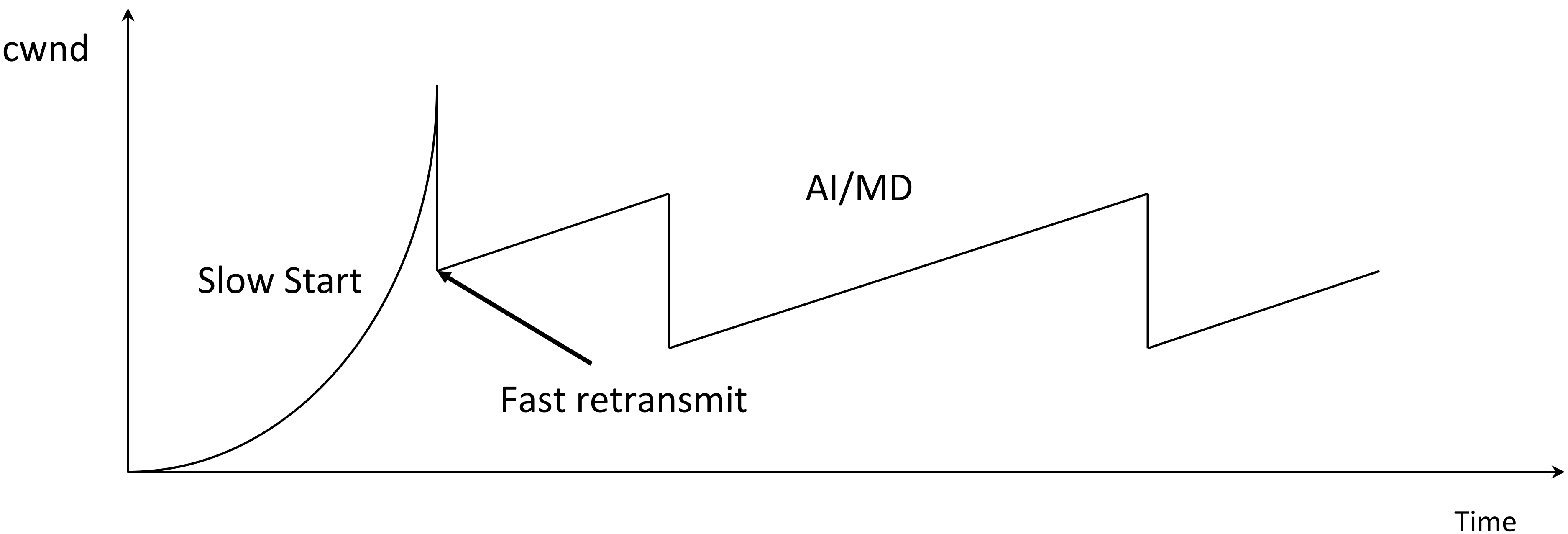
Tahoe RTT Estimation



Slow start every time?!

- **Losses have large effect on throughput**
- **Fast Recovery (TCP Reno)**
  - Same as TCP Tahoe on Timeout: w = 1, slow start
  - On triple duplicate ACKs: w = w/2
  - Retransmit missing segment (fast retransmit)
  - Stay in Congestion Avoidance mode

Fast Recovery and Fast Retransmit



3 Challenges Revisited

- **Determining the available capacity in the first place**
  - Exponential increase in congestion window
- **Adjusting to changes in the available capacity**
  - Slow probing, AIMD
- **Sharing capacity between flows**
  - AIMD
- **Detecting Congestion**
  - Timeout based on RTT
  - Triple duplicate acknowledgments
- **Fast retransmit/Fast recovery**
  - Reduces slow starts, timeouts

Next Class

- More Congestion Control fun
- Cheating on TCP
- TCP on extreme conditions
- TCP Friendliness
- TCP Future