Lecture 4: Congestion Control

Overview

- **Internet is a network of networks**
- **Narrow waist of IP: unreliable, best-effort datagram delivery**
- **Packet forwarding: input port to output port**
- **Routing protocols: computing port mappings**
- **Transport: end-to-end, reliability, flow control, and congestion control**

Transport

- **Provides end-to-end communication between applications**
- **UDP: unreliable datagram delivery (thin layer on top of IP)**
- **TCP: reliable stream delivery**

TCP

- Connection establishment
- Connection teardown
- Retransmission timeout (RTT and variance)
- Flow control (receiver window)
- Congestion control (transmit window)

TCP

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- **Flow control (receiver window)**
- **Congestion control (transmit window)**

A Bit of History

- **1974: 3-way handshake**
- **1978: TCP and IP split into TCP/IP**
- **1983: January 1, ARPAnet switches to TCP/IP**
- **1986: Internet begins to suffer congestion collapses**
- **1987-8: Van Jacobson fixes TCP, publishes seminal paper (Tahoe)**
- **1990: Fast recovery and fast retransmit added (Reno)**

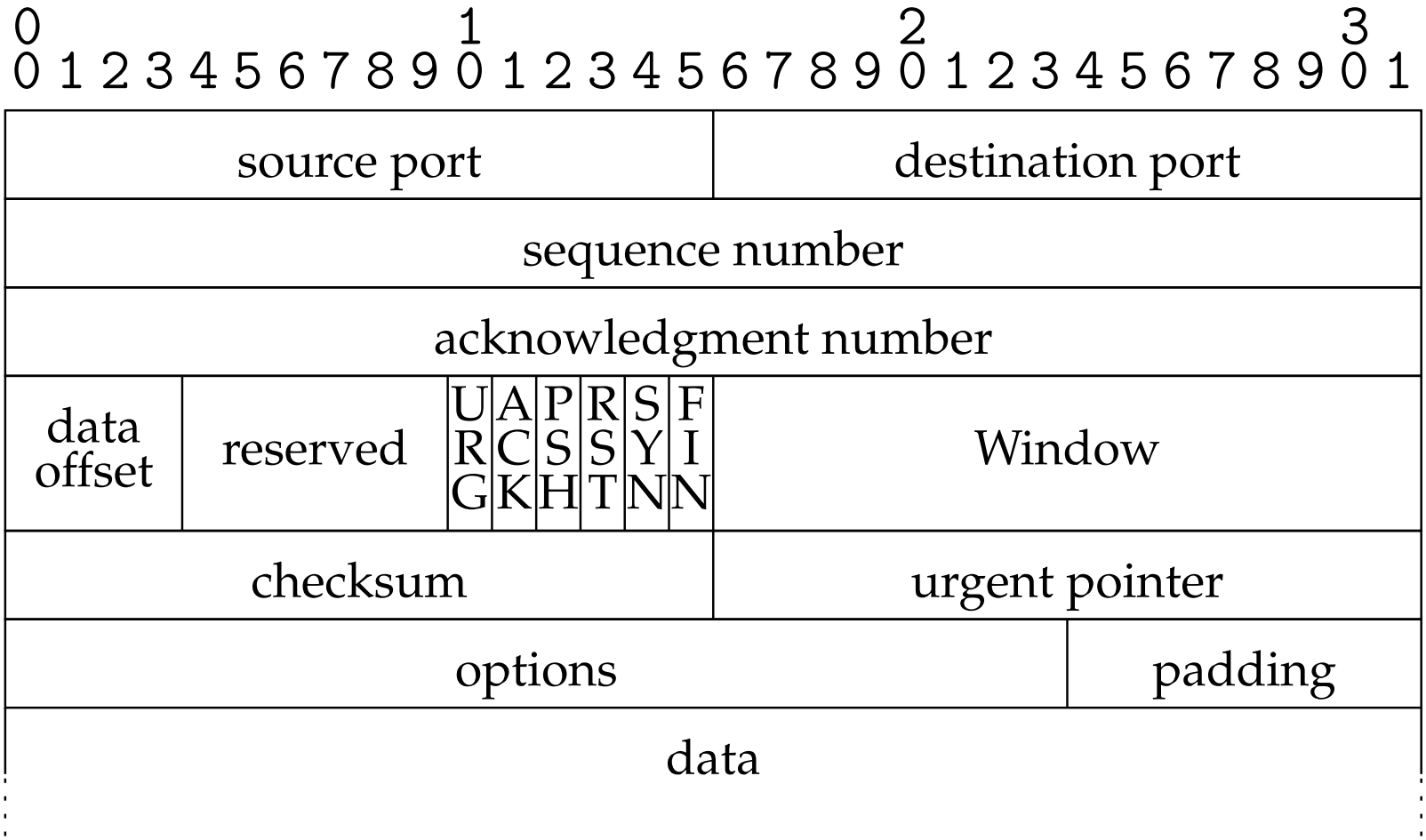
Three questions

- **Goal: maintain TCP goodput at equilibrium**
- **When does TCP retransmit packets?**
- **When does TCP transmit packets?**
- **When does TCP ack packets?**

Flow Control

- **Part of TCP specification (pre-1988)**
- **Want to make sure we don't send more than what the receiver can handle**
- **Sliding window protocol as described in lectures 2 and 3**
- **Use `window` header field to tell other side how much space you have**
- **Rule: Sent and unacknowledged bytes $\leq$ `window`**

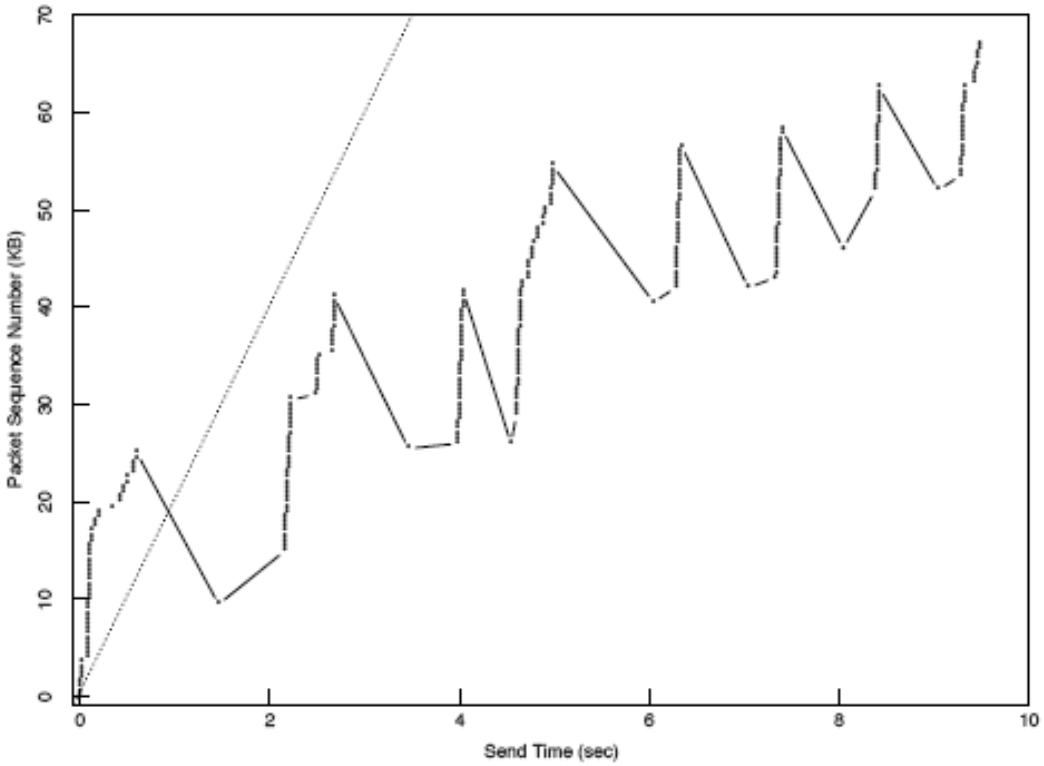
TCP segment


0 1 2 3
0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1
source port
destination port
sequence number
acknowledgment number
data offset
reserved
URG
ACK
PSH
RST
SYN
FIN
Window
checksum
urgent pointer
options
padding
data

Send Timing

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

Bursts of packets



Retransmission

- TCP dynamically estimates round trip time
- If segment goes unacknowledged, must retransmit
- Use exponential backoff (in case loss from congestion)
- After $\sim$10 minutes, give up and reset connection

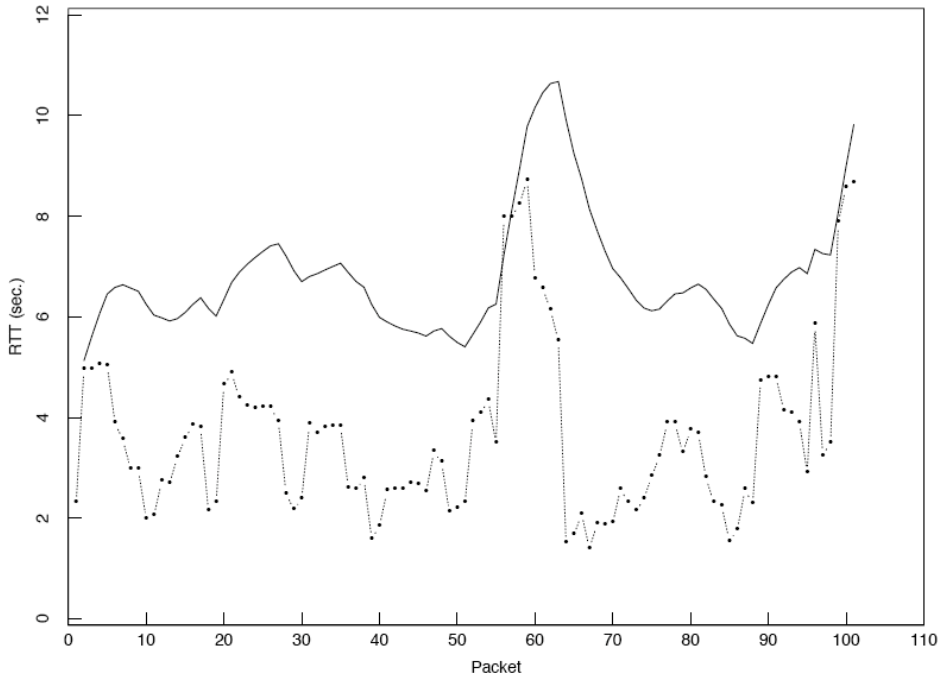
Round Trip Time (RTT)

- **We want to estimate RTT so we can know a packet was likely lost, not just delayed**
- **The challenge is that RTTs for individual packets can be highly variable, both on long-term and short-term time scales**
- **Can increase significantly 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

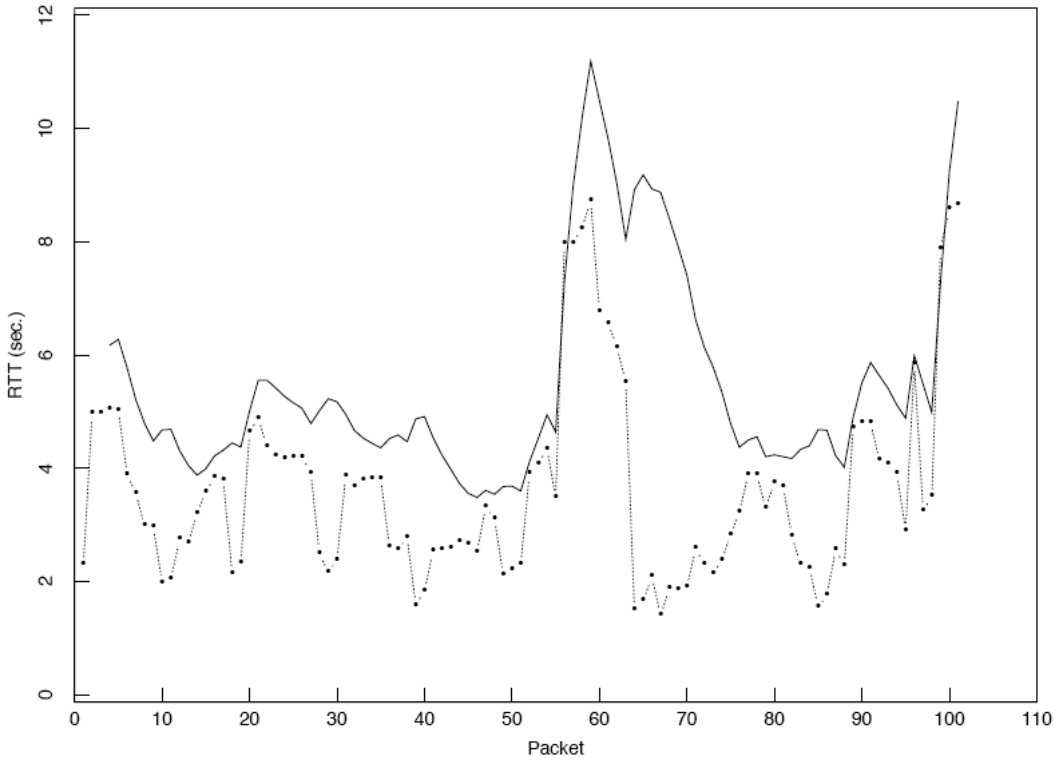
RTT with EWMA

- $Est_{RTT} = (1 - \alpha) \cdot Est_{RTT} + \alpha \cdot Sample_{RTT}$
  - Recommended α is 0.125
- $Dev_{RTT} = (1 - \beta) \cdot Dev_{RTT} + \beta \cdot |Sample_{RTT} - Est_{RTT}|$
  - Recommended β is 0.25
- **Timeout is** $Est_{RTT} + 4 \cdot Dev_{RTT}$

Old RTT estimation

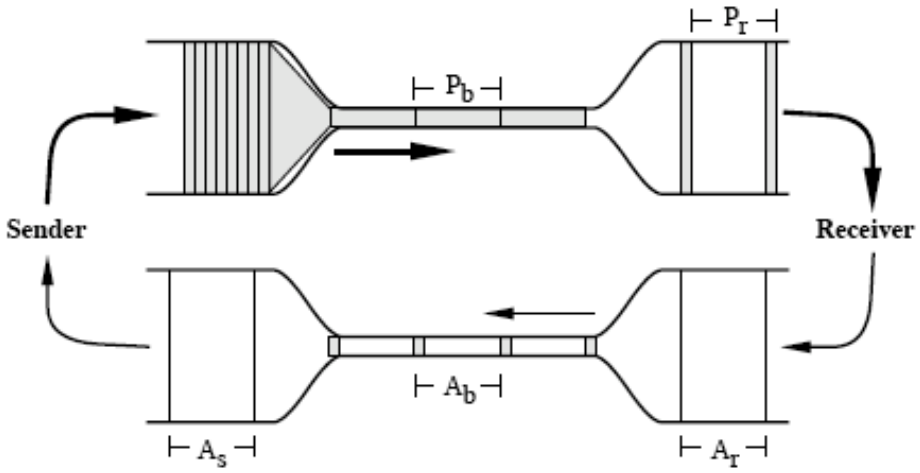


Tahoe RTT estimation



Self-Clocking

- **Goal is *conservation of packets* in steady state**
  - A new packet isn't put into the network until an old one leaves
  - Very effective in avoiding and controlling congestion
- **Solution: send a data packet for each ack**



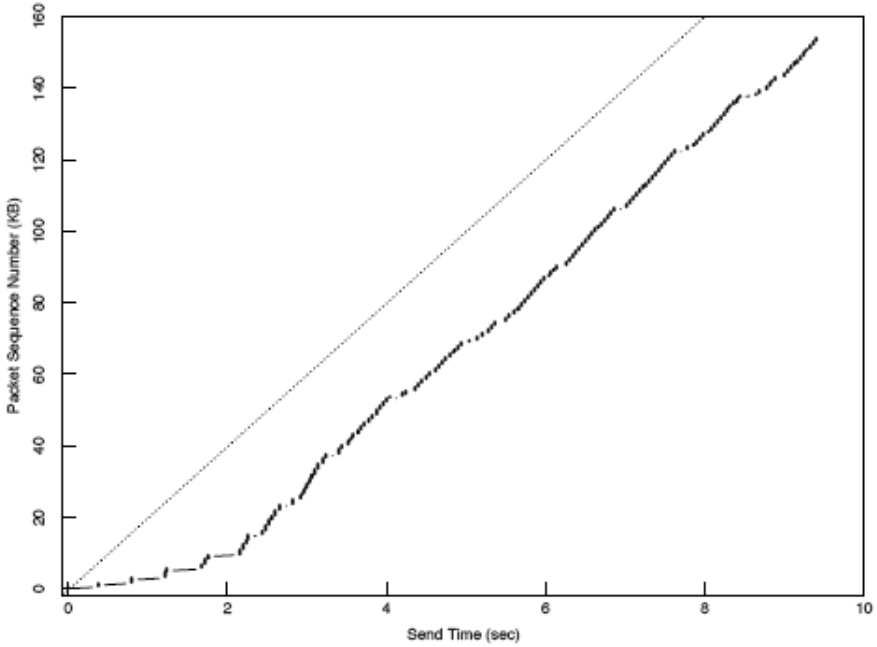
Self-Clocking, Continued

- **Assuming good RTT estimation, self-clocking prevents congestion by making `window` limit the number of packets in the network**
- **Very simple to implement**
- **But how big should the window be?**

Slow start

- **But how do you start?**
- **Introduce a *congestion window* to connection state,** `cwnd`
- **Rule: Sent and unacknowledged bytes $\leq$** `cwnd`
- **On start or loss, set** `cwnd` **to one packet (or... more on this later)**
- **On each data ack, increase** `cwnd` **by one packet**
- **Prevents bursts of packets**

With Slow Start



- Slow start allows connection to quickly reach equilibrium
- How do we maintain it?

Two States

- **TCP has two states: Slow Start (SS) and Congestion Avoidance (CA)**
- **A window size threshold governs the state transition**
  - Window $\leq$ threshold: slow start
  - Window $>$ threshold: collision avoidance
- **States differ in how they respond to new acks**
  - Slow start: cwnd += MSS
  - Congestion avoidance: cwnd += $\frac{MSS^2}{cwnd}$ (MSS every RTT)

Congestion Control

- **Two conflicting goals**
  - Provider: high utilization
  - User: fairness among users
- **Want to converge to a state where everyone gets $\frac{1}{N}$**
- **Avoid congestion collapse**

Taming Congestion

- **The optimal congestion window is the bandwidth/delay product**
- **Sender learns this by adapting window size**
- **Senders must slow down when there is congestion**
- **Absence of ACKs (timeout) implies serious congestion**
- **Duplicate ACKs imply some congestion**

Responding to Loss (TCP Tahoe)

- **On triple duplicate ACK or timeout**
  - Implies lost packet
  - Set threshold to $\frac{cwnd}{2}$
  - Set cwnd to 1
  - Enter slow start state

TCP Reno [RFC 2581]

- **Same as Tahoe on timeout**
- **On triple duplicate ACK**
  - Set threshold to $\frac{cwnd}{2}$
  - Set cwnd to $\frac{cwnd}{2}$ (fast recovery)
  - Retransmit missing segment (fast retransmit)
  - Stays in congestion avoidance state

TCP NewReno [RFC 3782]

- **Same as Tahoe/Reno on timeout**
- **During fast recovery**
  - Keep track of last unacknowledged packet on entering fast recovery
  - On every duplicate ACK, inflate congestion window by MSS
  - When last packet is acknowledged, return to congestion avoidance, set cwnd back to original value

Walk Through Three Examples

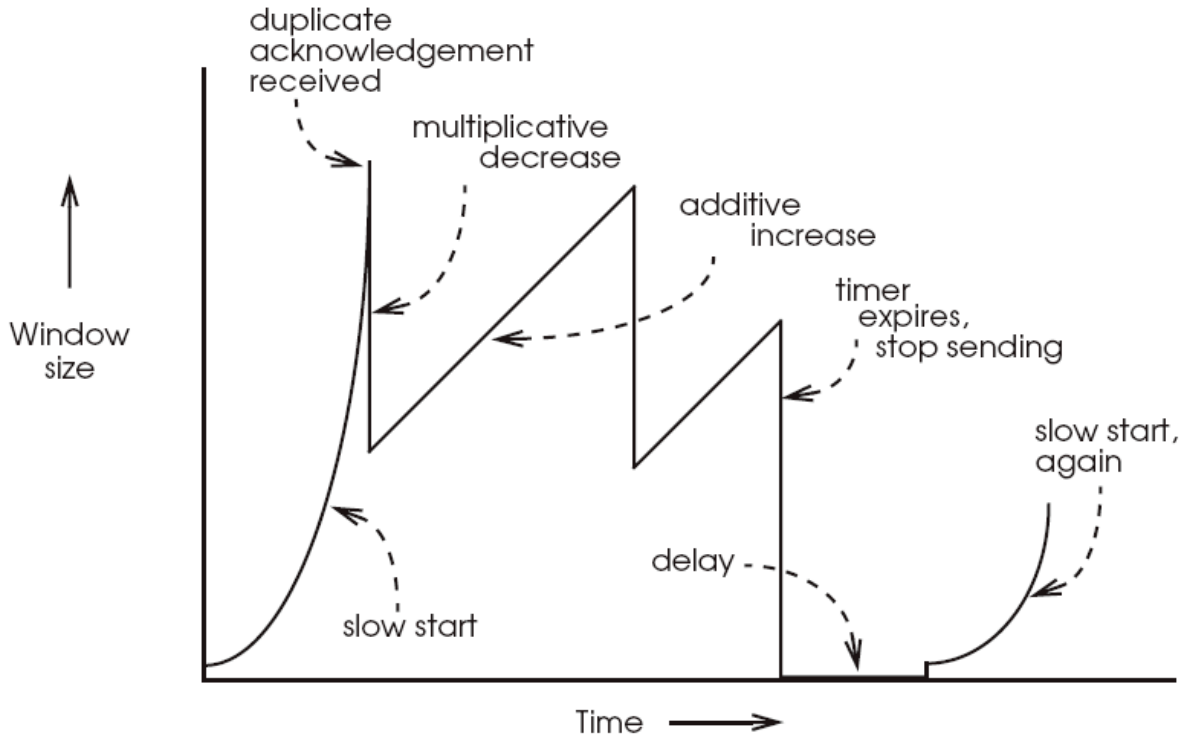
- **TCP Tahoe (RTT estimation, congestion window)**
- **TCP Reno (Fast retransmit, fast recovery)**
- **TCP New Reno (cwnd inflation in fast recovery state)**

2-minute stretch

AIMD

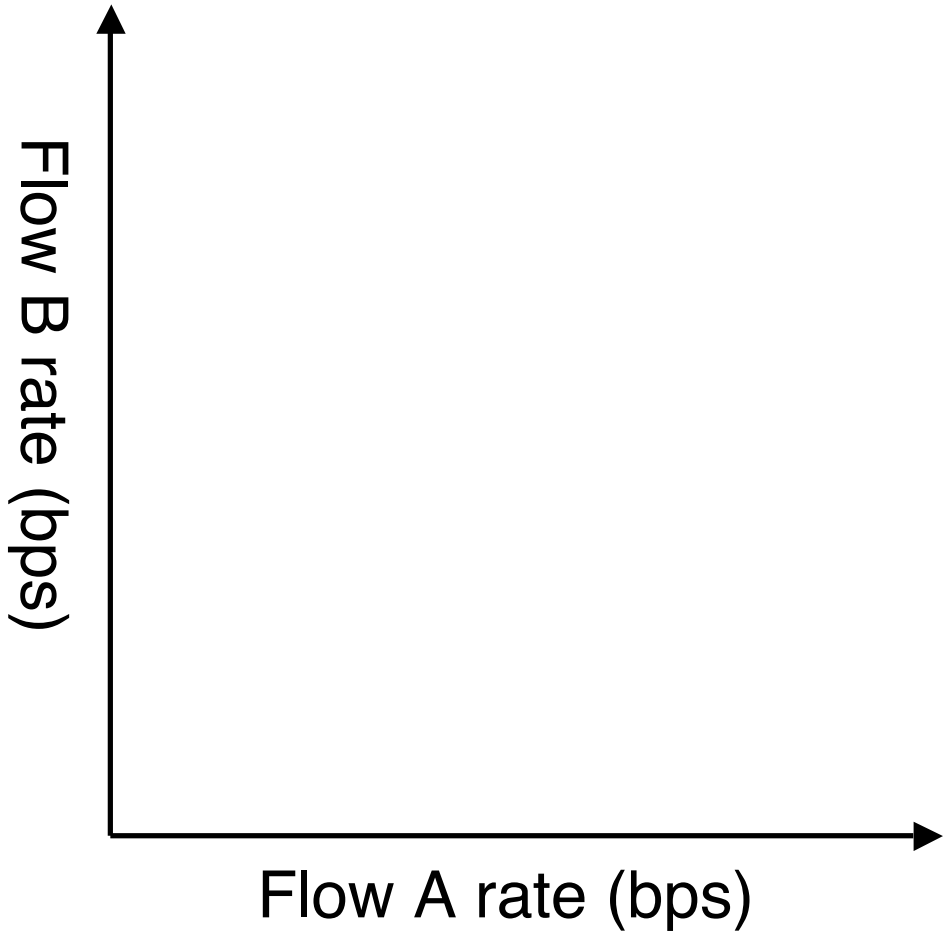
- **Additive increase, multiplicative decrease**
- **Behavior in congestion avoidance mode**
- **Why AIMD? Remember goals:**
  - Link utilization
  - Fairness

AIMD in action



- **Figure thanks to Brad Karp**

Chiu Jain Phase Plots



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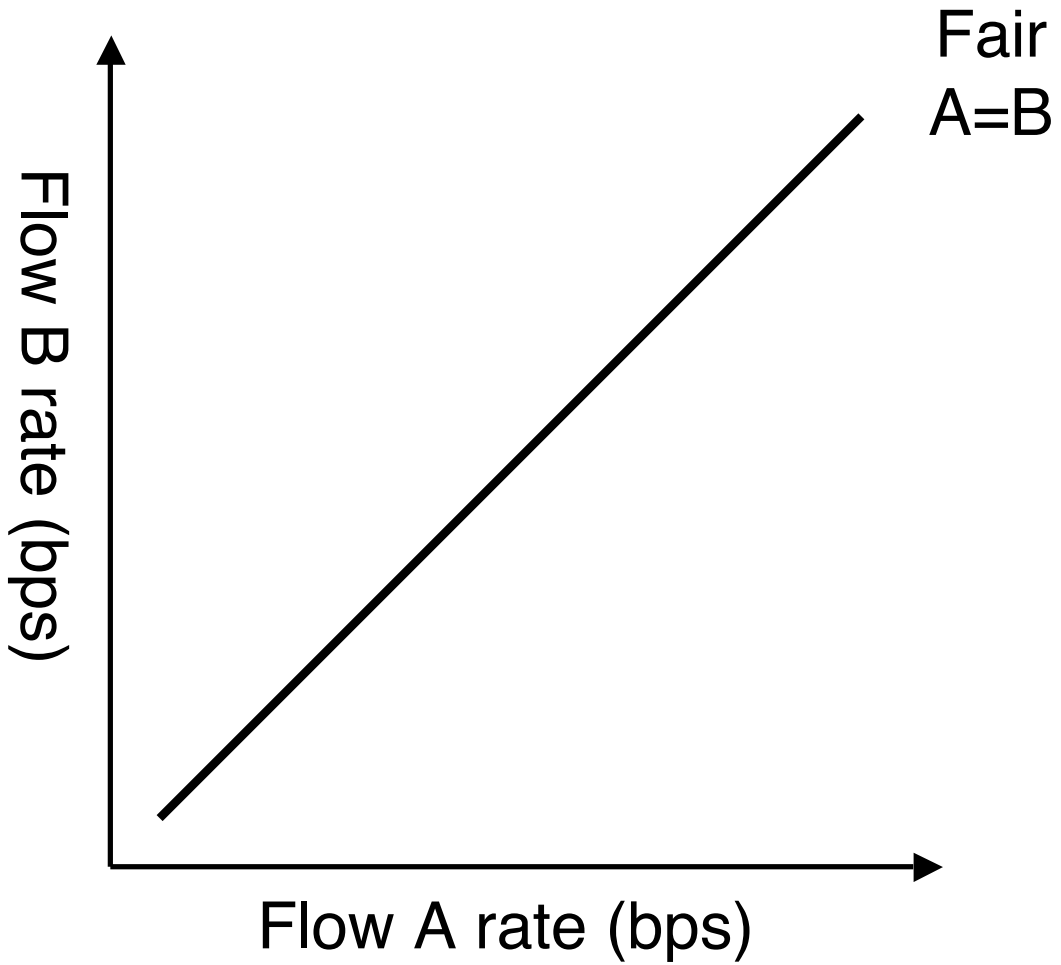
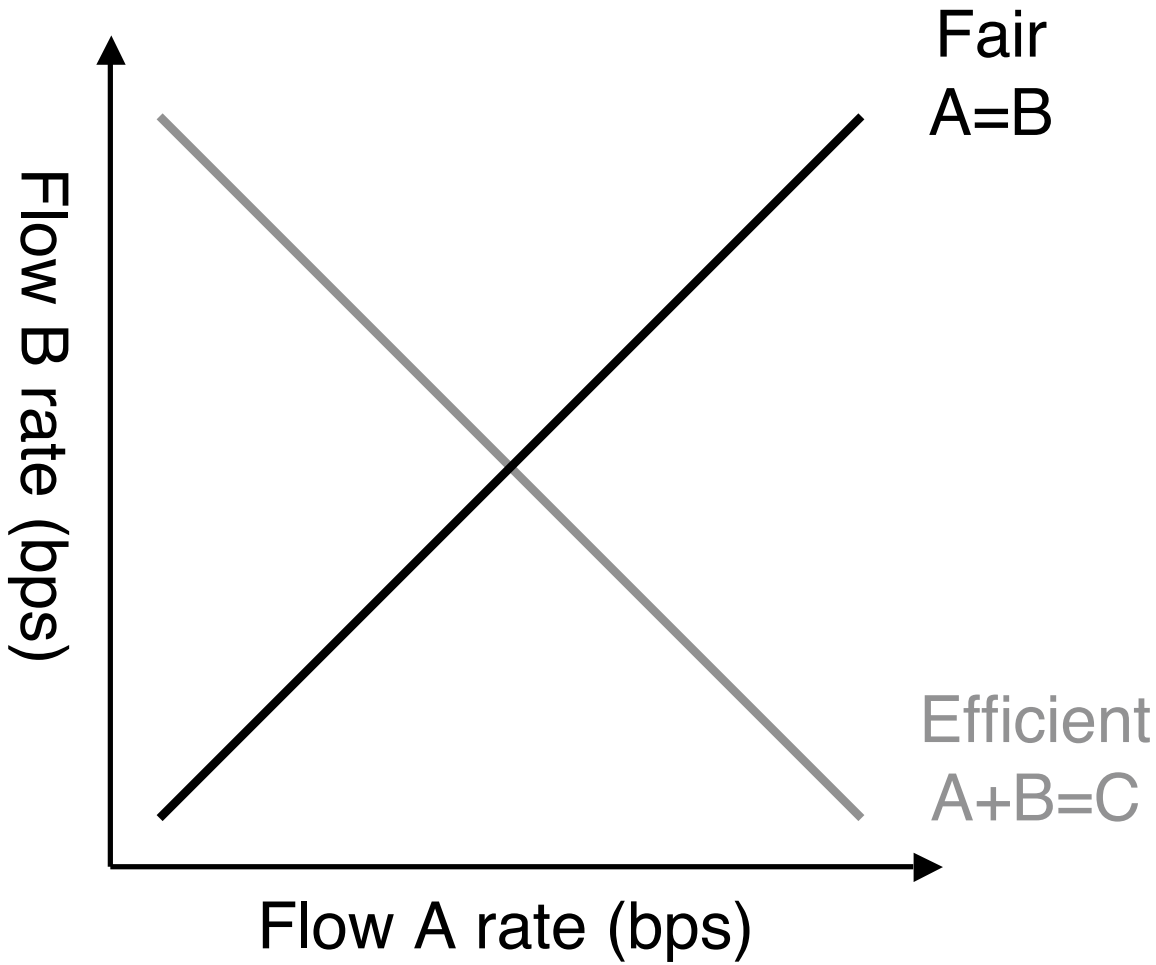


Chiu Jain Phase Plots



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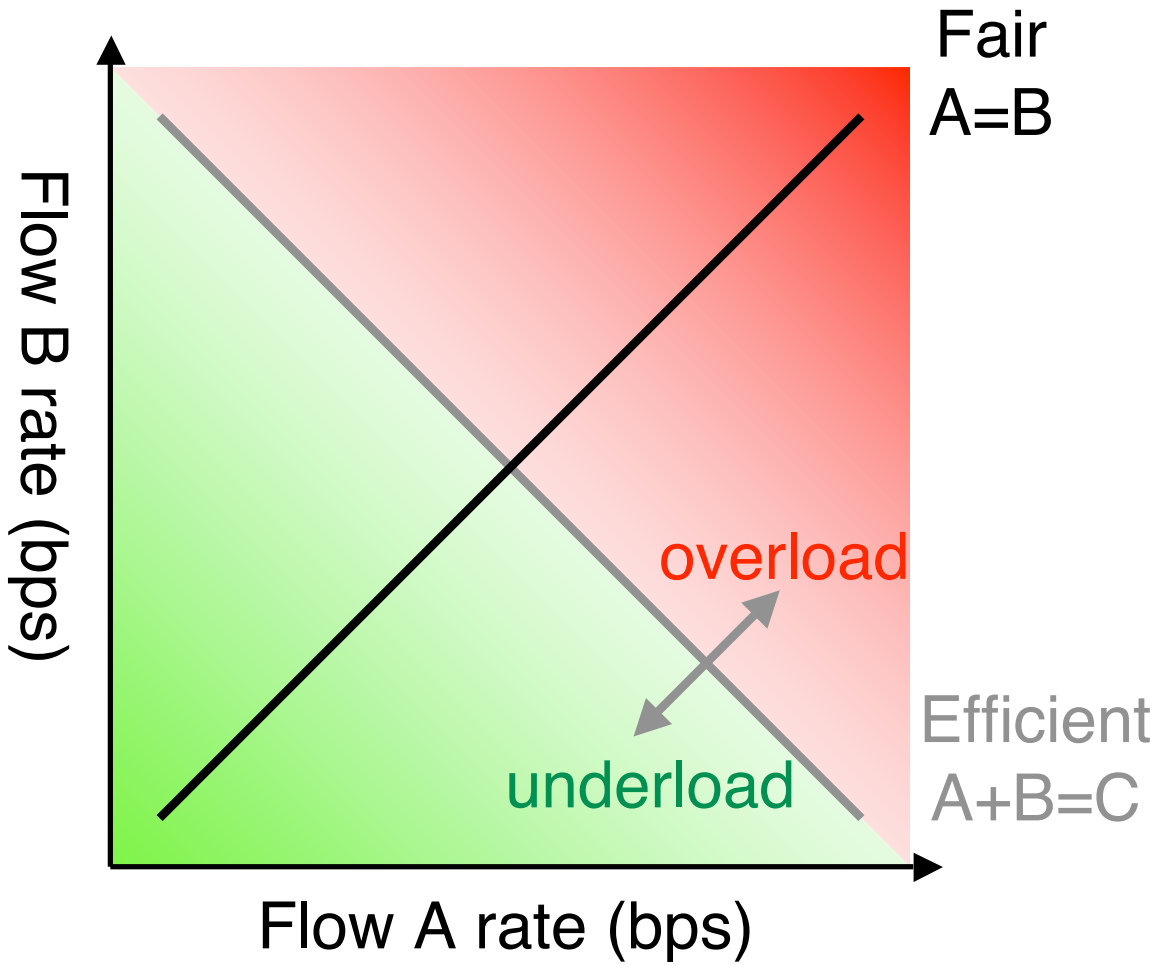

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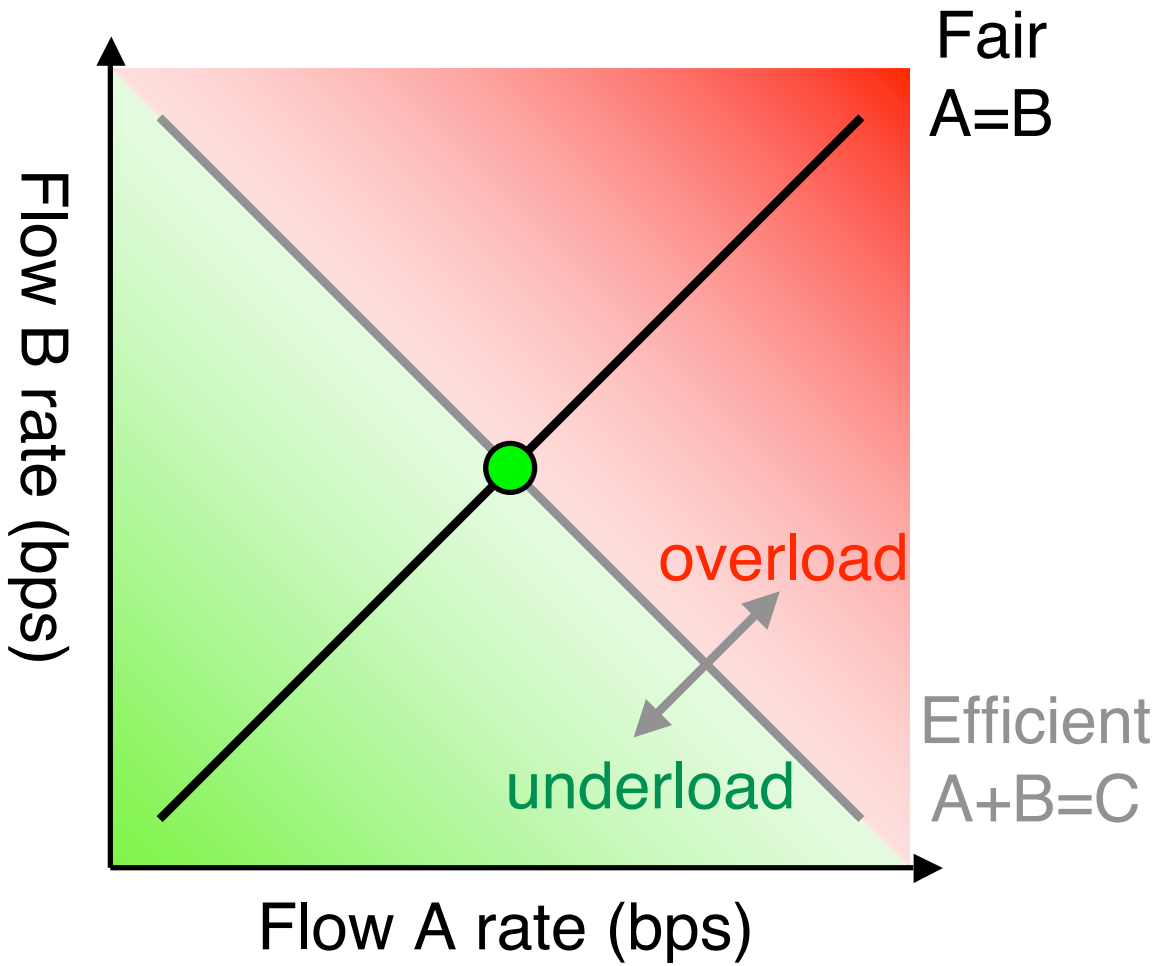

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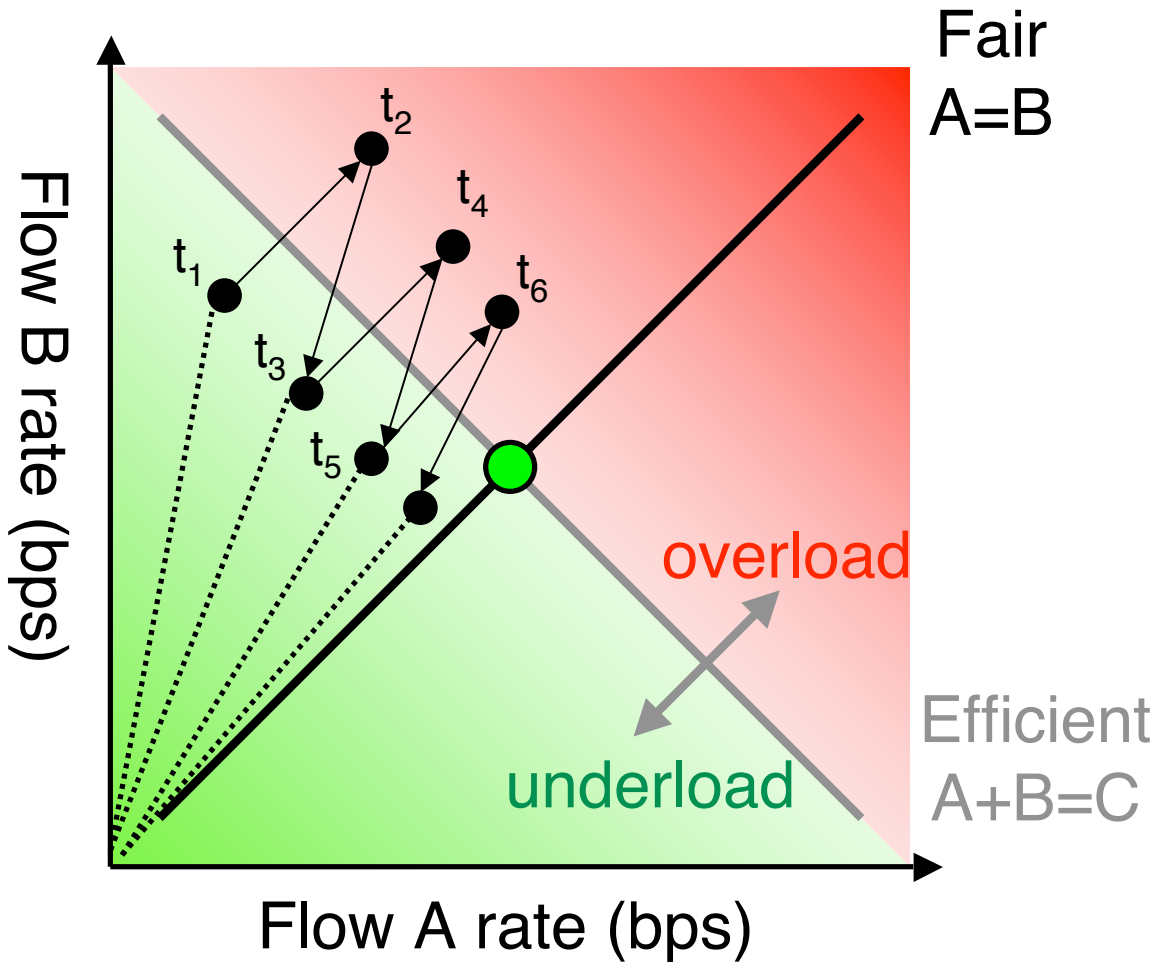
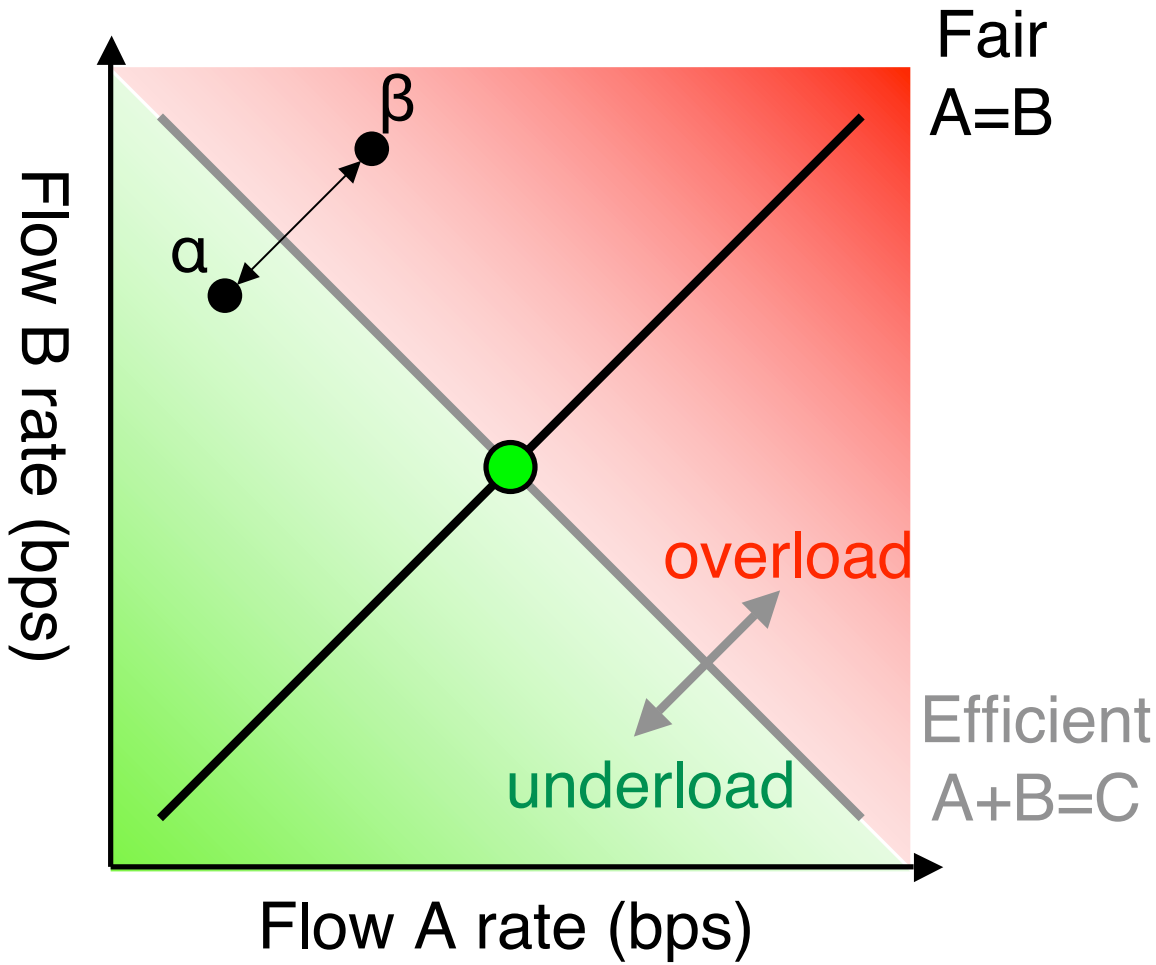


Chiu Jain Phase Plots



Three questions, revisited

- Goal: maintain TCP goodput at equilibrium
- When does TCP retransmit packets?
- When does TCP transmit packets?
- When does TCP ack packets?

Sending acknowledgements

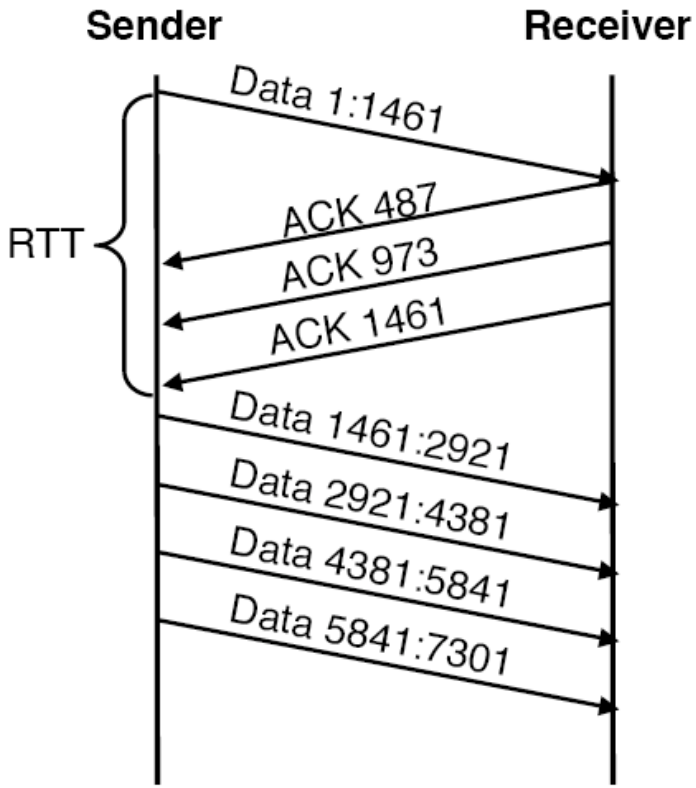
- **An ACK must be sent for every other segment (RFC 2581)**
- **An ACK must be sent within 500ms (RFC 2581)**
- **Send duplicate acks aggressively**

Acknowledgement Issues

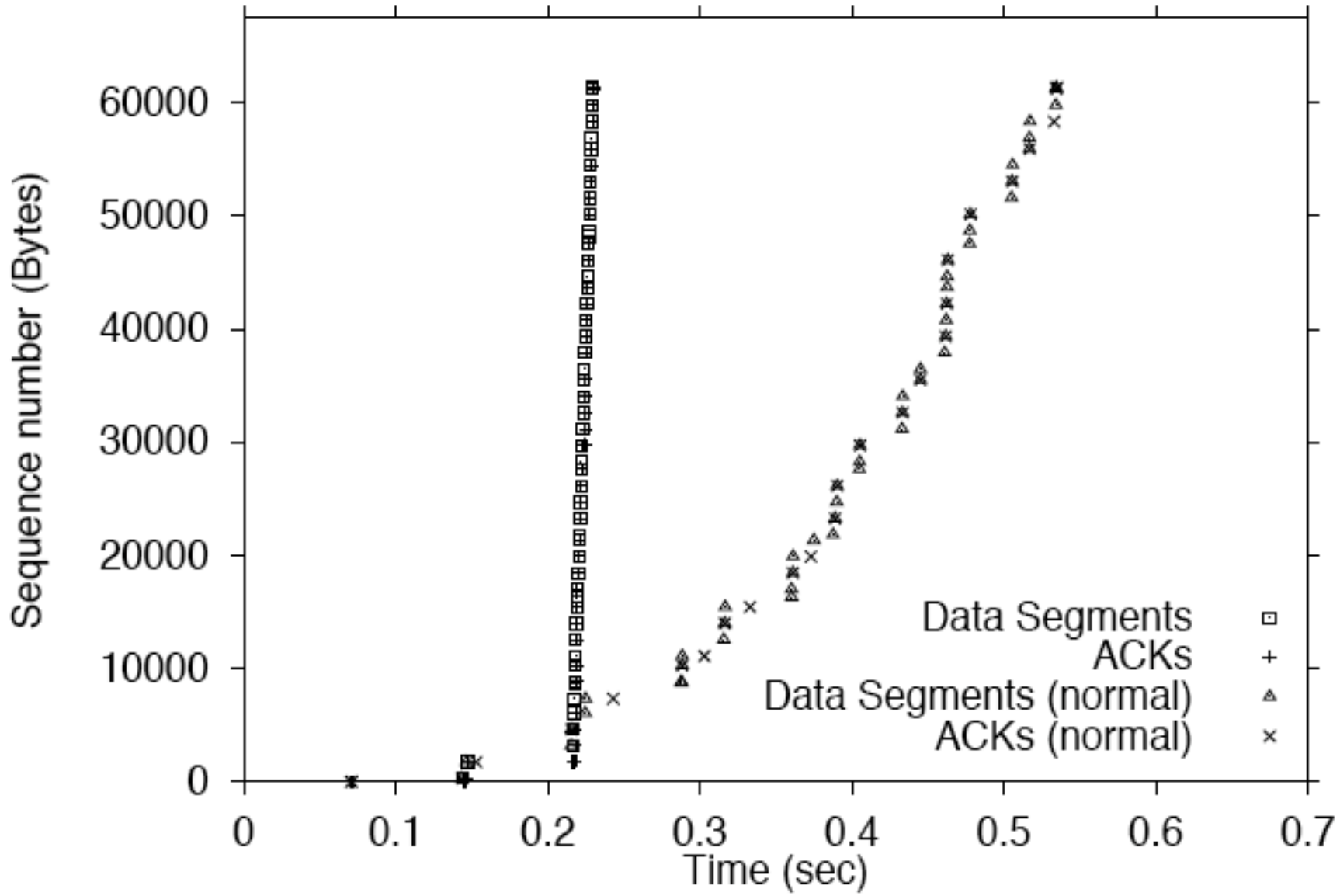
- Savage et al., CCR 1999
- ACKs are in terms of bytes
- Congestion control is in terms of segments
- What if you send multiple ACKs per segment?

Multiplicative increase

- **Receiver can force multiplicative increase, for $M \leq$ segment length**
- **Can lead to 4GB `cwnd` in 4 RTTs!**



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)

TCP Daytona Lessons

- **TCP implementations now symmetrically use bytes**
- **Protocol specification is difficult!**
- **Very subtle and simple design decisions can lead to undesired behavior**
- **IETF requires at least two interoperating implementations before moving to standards track**
- **Protocols evolve**

Future of TCP

- **BitTorrent can lead to other apps performing poorly**
  - BitTorrent (and other P2P applications) can open 10-100 connections
  - TCP provides per-flow, not per-application, fairness
- **Low Extra Delay Background Transport (ledbat)**
  - ”LEDBAT is a transport-area WG that will focus on broadly applicable techniques that allow large amounts of data to be consistently transmitted without substantially affecting the delays experienced by other users and applications.”

Future of TCP, Continued

- **High speed links/high bandwidth-delay product**
  - Single packet loss cuts cwnd in half
  - Can take a long time to grow
  - Fast TCP, High-Speed TCP
- **Wireless**
  - Wireless exhibits many losses due to poor links, limits throughput
  - High-throughput TCP in wireless meshes does not yet exist
  - More in lecture 11