Reliable Transport II: TCP and Congestion Control

Brad Karp

UCL Computer Science

CS 3035/GZ01

24th November 2011

Outline

- RTT estimator
- AIMD Congestion control
- Throughput, loss, and RTT equation
- Connection teardown
- Protocol state machine

TCP: Retransmit Timeouts

- Sender sets timer for each sent packet
  - when ACK returns, timer canceled
  - if timer expires before ACK returns, packet resent
- Expected time for ACK to return: RTT
- TCP estimates round-trip time using EWMA
  - measurements m_i from timed packet/ACK pairs
  - $RTT_i = ((1-\alpha) \times RTT_{i-1} + \alpha \times m_i)$
  - Retransmit timeout: $RTO_i = \beta \times RTT_i$
  - original TCP: $\beta = 2$
- Is this accurate enough?
  - Recall dangers of too-short and too-long RTT estimates from previous lecture

Mean and Variance: Jacobson's RTT Estimator

- Above link load of 30% at router, $\beta \times RTT_i$ will retransmit too early!
- Response to increasing load: waste bandwidth on duplicate packets
- Result: **congestion collapse!**
- [Jacobson 88]: estimate v_i, mean deviation (EWMA of $|m_i - RTT_i|$), stand-in for variance

$$v_i = v_{i-1} \times (1-\gamma) + \gamma \times |m_i - RTT_i|$$

- Use $RTO_i = RTT_i + 4v_i$

Mean and Variance: Jacobson's RTT Estimator

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Mean and Variance RTT estimator used by all modern TCPs

for variance

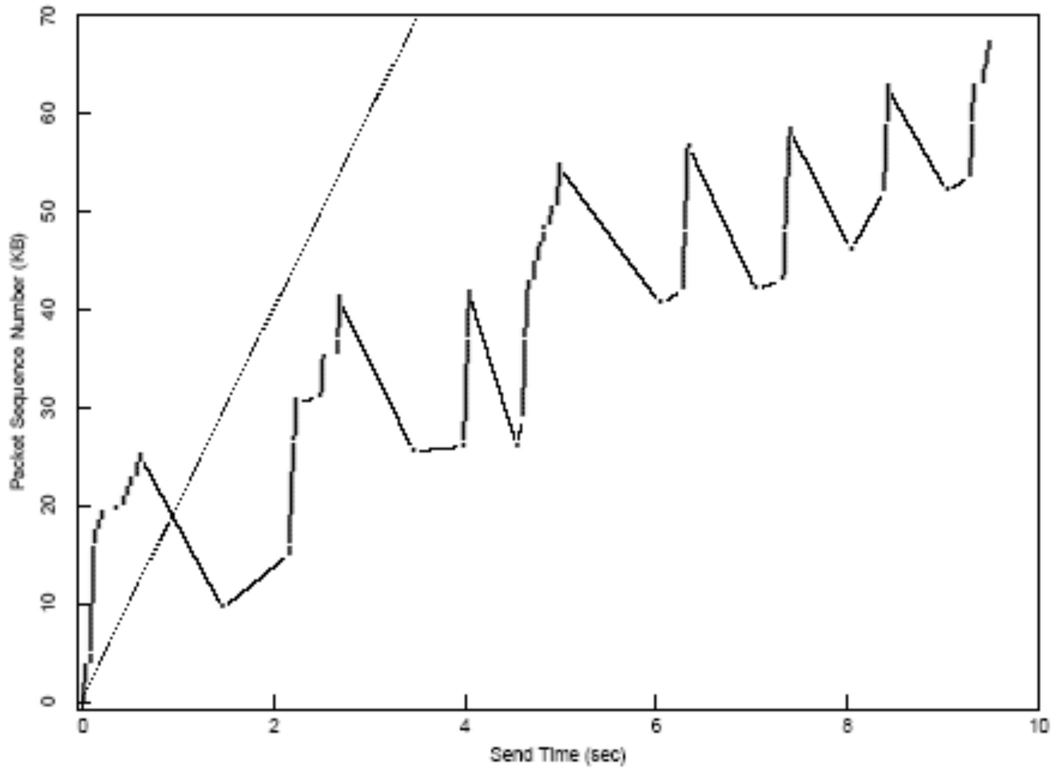
$$v_i = v_{i-1} \times (1-\gamma) + \gamma \times |m_i - RTT_i|$$

- Use $RTO_i = RTT_i + 4v_i$

Retransmit Behavior

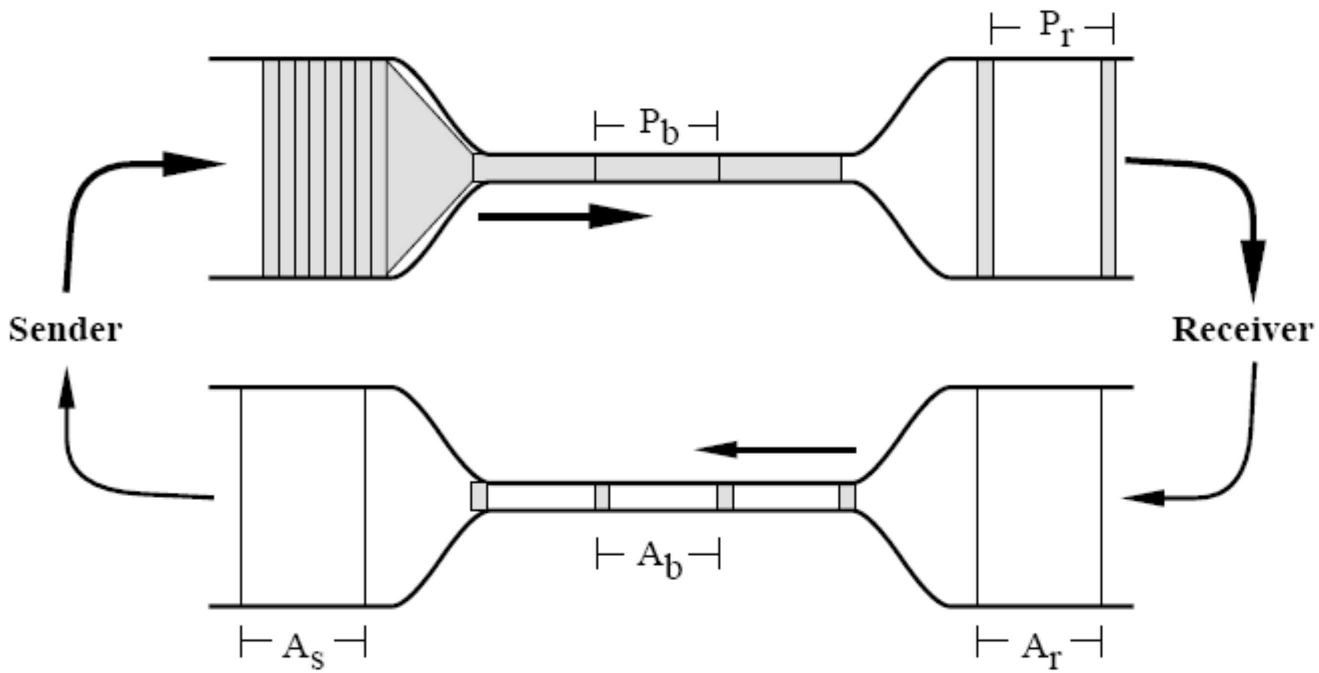
- Original TCP, before [Jacobson 88]:
  - at start of connection, send full window of packets
  - retransmit each packet immediately after its timer expires
- Result: window-sized bursts of packets sent into network

Pre-Jacobson TCP (Obsolete!)



- Time-sequence plot taken at sender
- Bursts of packets: vertical lines
- Spurious retransmits: repeats at same y value
- Dashed line: available 20 Kbps capacity

Self-Clocking: Conservation of Packets

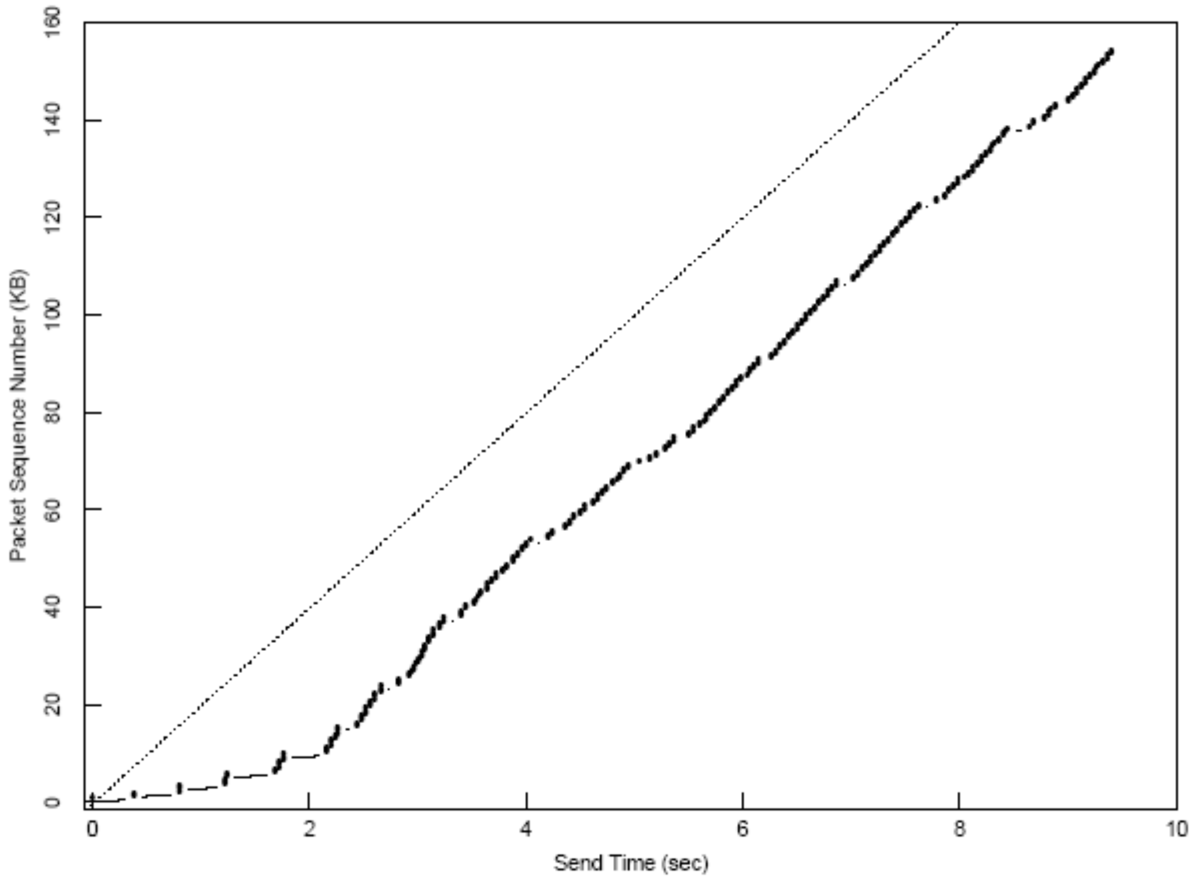


- Goal: self-clocking transmission
  - each ACK returns, one data packet sent
  - spacing of returning ACKs: matches spacing of packets in time at slowest link on path

Reaching Equilibrium: Slow Start

- At connection start, sender sets congestion window size, cwnd, to pktSize (one packet's worth of bytes), not whole window
- Sender sends up to minimum of receiver's advertised window and cwnd
- Upon return of each ACK until receiver's advertised window size reached, increase cwnd by pktSize bytes
- "Slow" means exponential window increase!
- Takes $\log_2 W$ RTTs to reach receiver's advertised window size W

Post-Jacobson TCP: Slow Start and Mean+Variance RTT Estimator



- Time-sequence plot at sender
- "Slower" start
- No spurious retransmits

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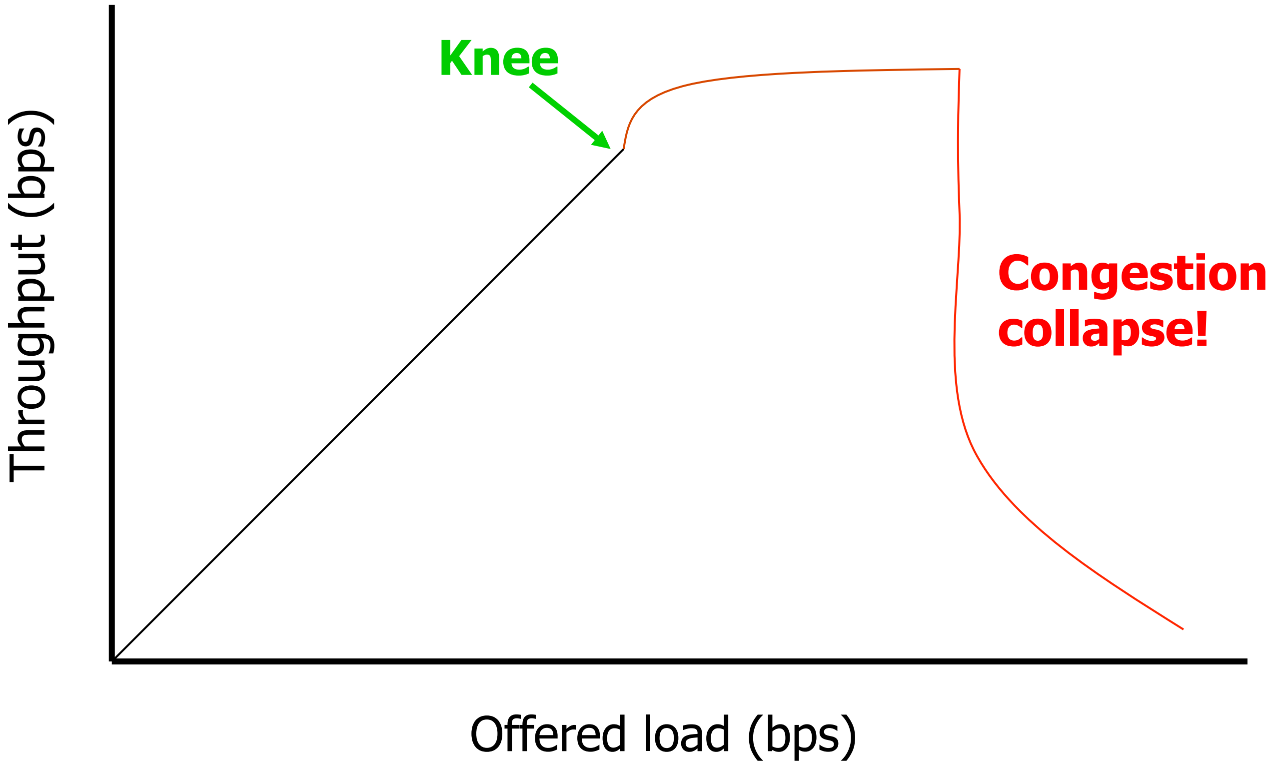
Goals in Congestion Control

- Achieve high utilization on links; don’t waste capacity!
- Divide bottleneck link capacity fairly among users
- Be stable: converge to a steady allocation among users
- Avoid congestion collapse

Congestion Collapse



- Cliff behavior observed in [Jacobson 88]

Congestion Requires Slowing Senders

- Recall: bigger buffers cannot prevent congestion
- Senders must slow to alleviate congestion
- Absence of ACKs implicitly indicates congestion
- TCP sender's window size determines sending rate
- Recall: correct window size is bottleneck bandwidth-delay product
- How can sender learn this value?
  - **Search** for it, by adapting window size
  - **Feedback** from network: ACKs return (window OK) or do not return (window too big)

Avoiding Congestion: Multiplicative Decrease

- Recall that sender uses sending window of size min(cwnd, rwnd), where rwnd is receiver's advertised window
- Upon timeout for sent packet, sender presumes packet lost to congestion, and:
  - sets ssthresh = cwnd / 2
  - sets cwnd = pktSize
  - uses slow start to grow cwnd up to ssthresh
- End result: cwnd = cwnd / 2, via slow start
- Sender sends one window per RTT; halving cwnd halves transmit rate

Avoiding Congestion: Additive Increase

- Drops indicate TCP sending more than its fair share of bottleneck
- No feedback to indicate TCP using **less** than its fair share of bottleneck
- Solution: speculatively increase window size as ACKs return
- Additive increase: for each returning ACK,

  cwnd = cwnd + (pktSize × pktSize)/cwnd

  - Increases cwnd by ~pktSize bytes per RTT

Avoiding Congestion: Additive Increase

- Drops indicate TCP sending more than its fair share of bottleneck
- No feedback to indicate TCP using **less** than its fair share of bottleneck

Combined algorithm:
Additive Increase, Multiplicative Decrease (AIMD)

cwnd = cwnd + (pktSize × pktSize)/cwnd

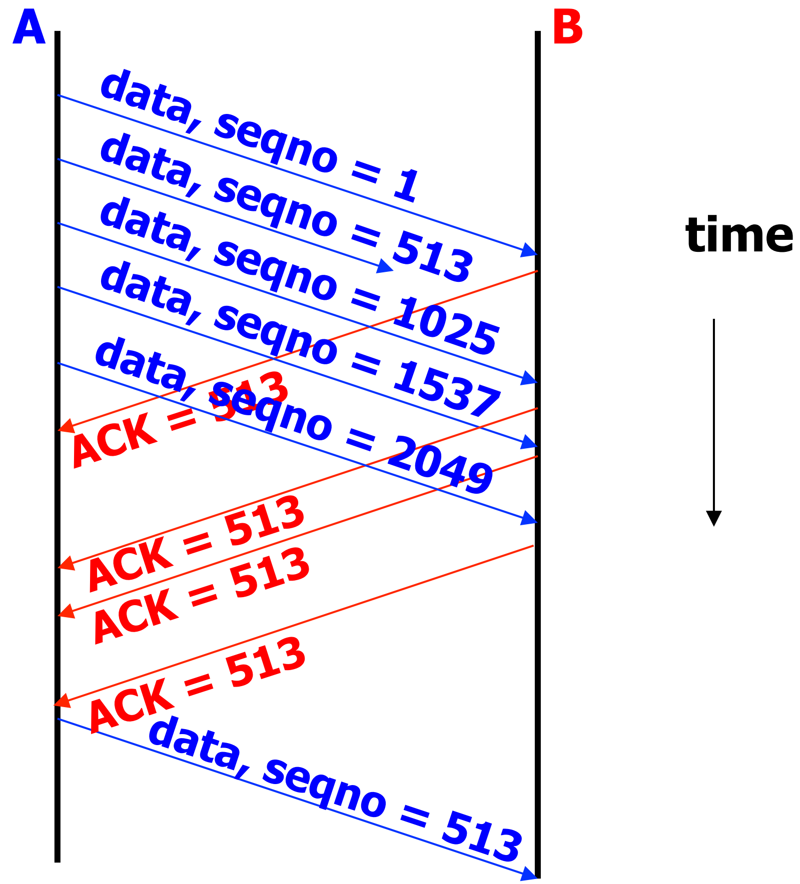
– Increases cwnd by ~pktSize bytes per RTT

Refinement: Fast Retransmit (I)

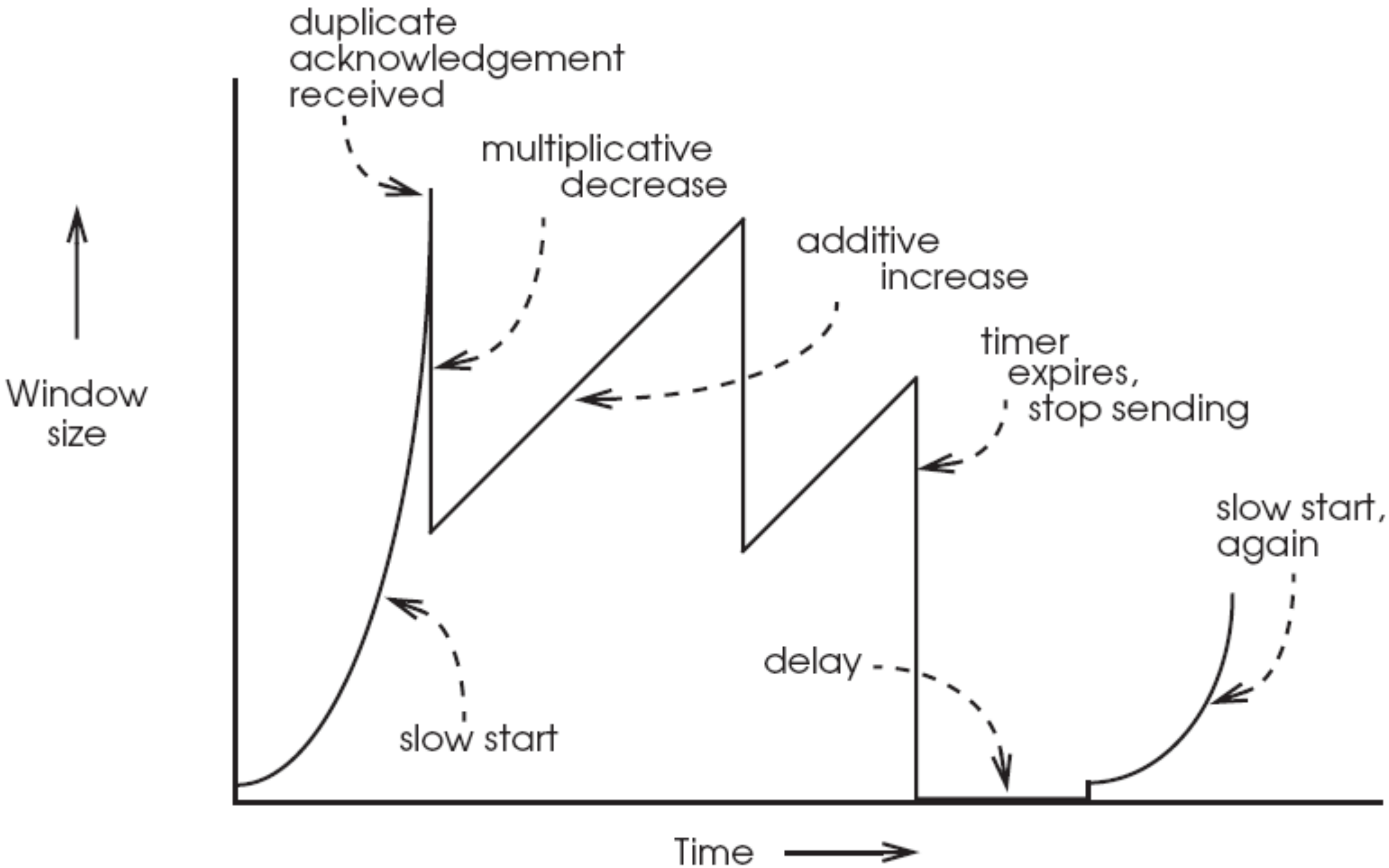
- Sender must wait well over RTT for timer to expire before loss detected
- TCP’s minimum retransmit timeout: 1 second
- Another loss indication: duplicate ACKs
  - Suppose sender sends 1, 2, 3, 4, 5, but 2 lost
  - Receiver receives 1, 3, 4, 5
  - Receiver sends cumulative ACKs 2, 2, 2, 2
  - Loss causes duplicate ACKs!

Fast Retransmit (II)

- Upon arrival of 3 duplicate ACKs, sender:
  - sets cwnd = cwnd/2
  - retransmits "missing" packet
  - no slow start
- Not only loss causes dup ACKs
  - Reordering, too



AIMD in Action



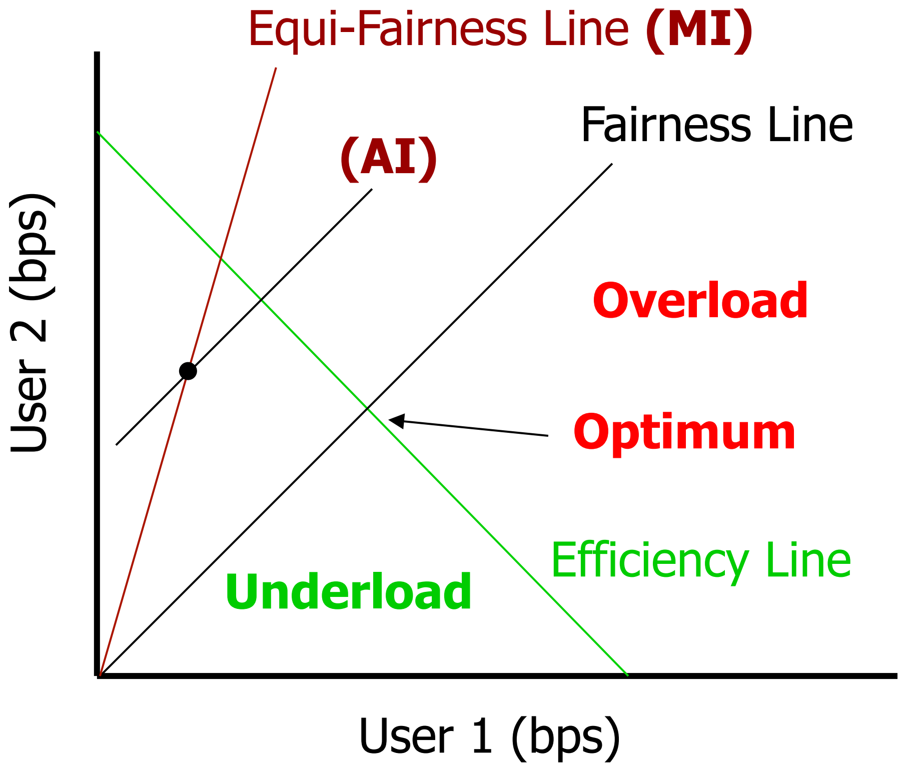
- Sender **searches** for correct window size

Why AIMD?

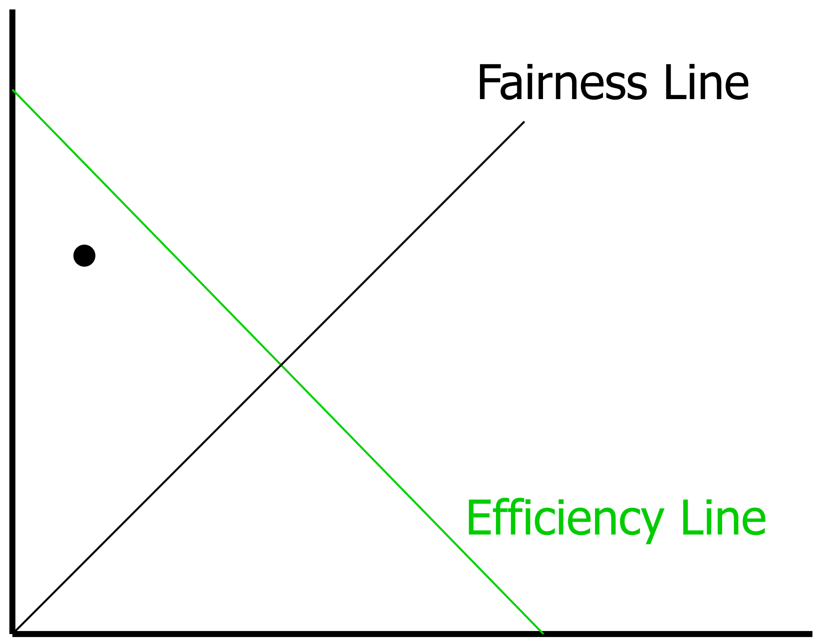
- Other control rules possible
  - E.g., MIMD, AIAD, …
- Recall goals:
  - Links fully utilized (efficient)
  - Users share resources fairly
- TCP adapts all flows' window sizes independently
- Must choose a control that will always converge to an efficient and fair allocation of windows

Chiu-Jain Phase Plots

- Consider two users sharing a bottleneck link
- Plot bandwidths allocated to each
- Efficiency: sum of two users' rates fixed
- Fairness: two users' rates equal
- Equi-Fairness: ratio of two users' rates fixed

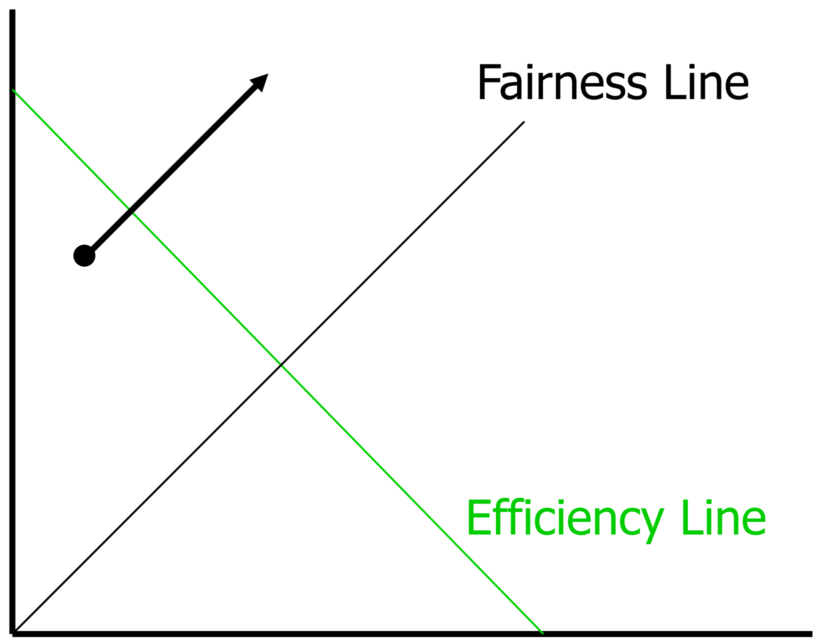


Chiu Jain: AIMD



- AIMD converges to optimum efficiency and fairness

Chiu Jain: AIMD



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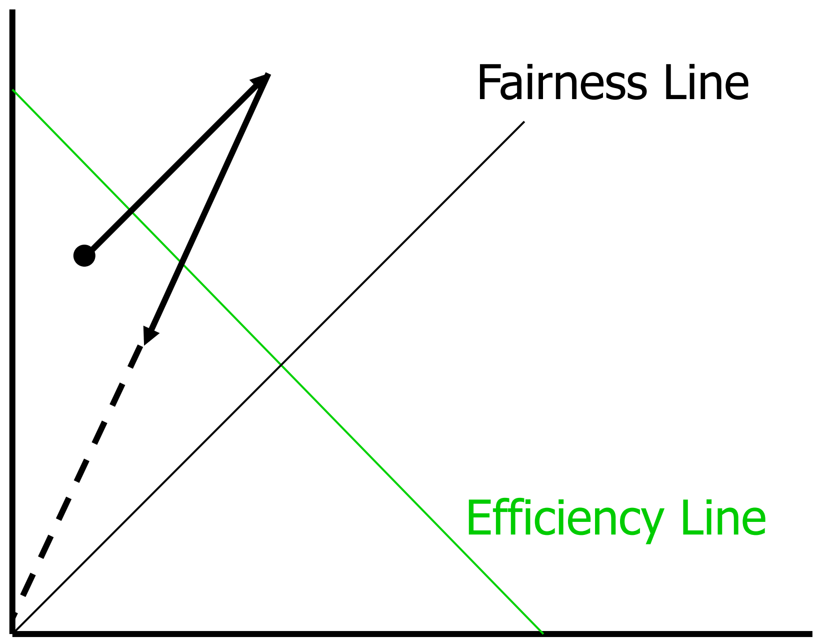

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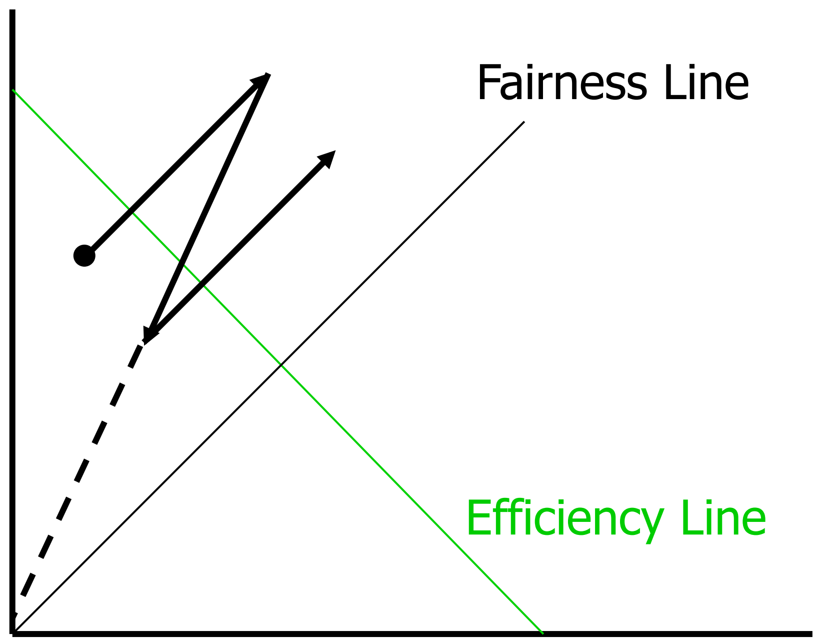

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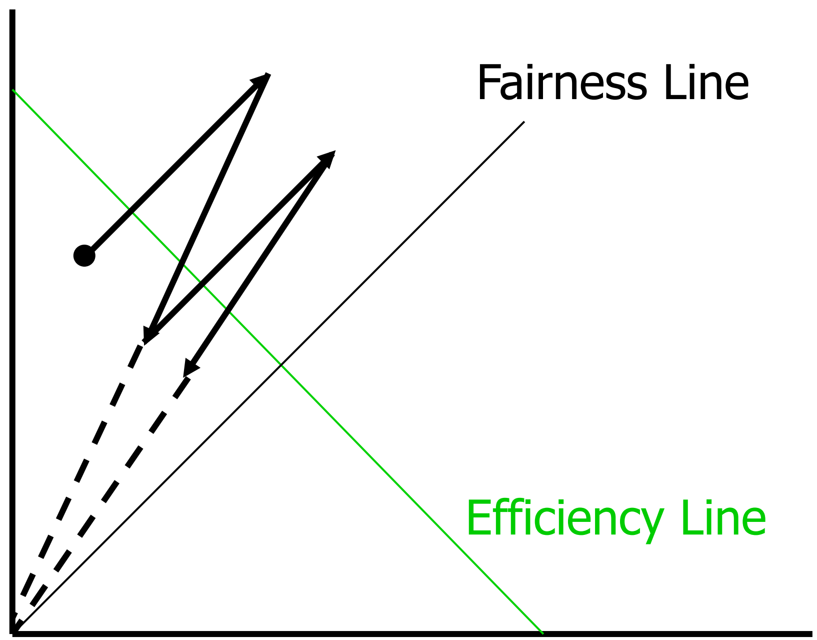

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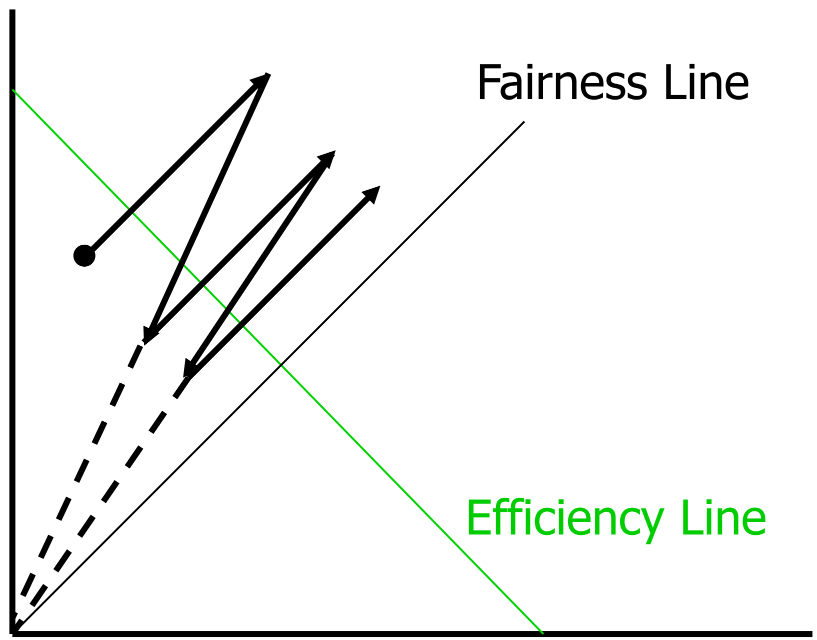

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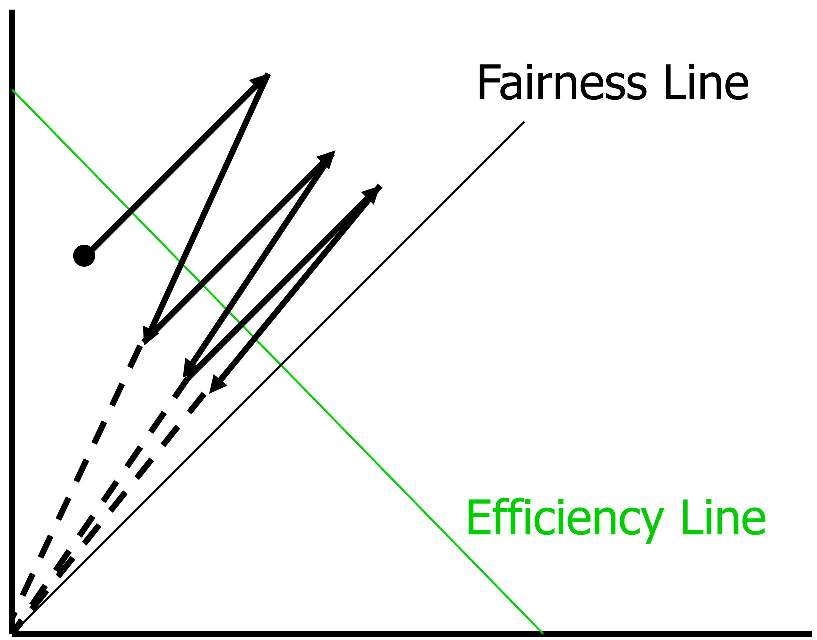

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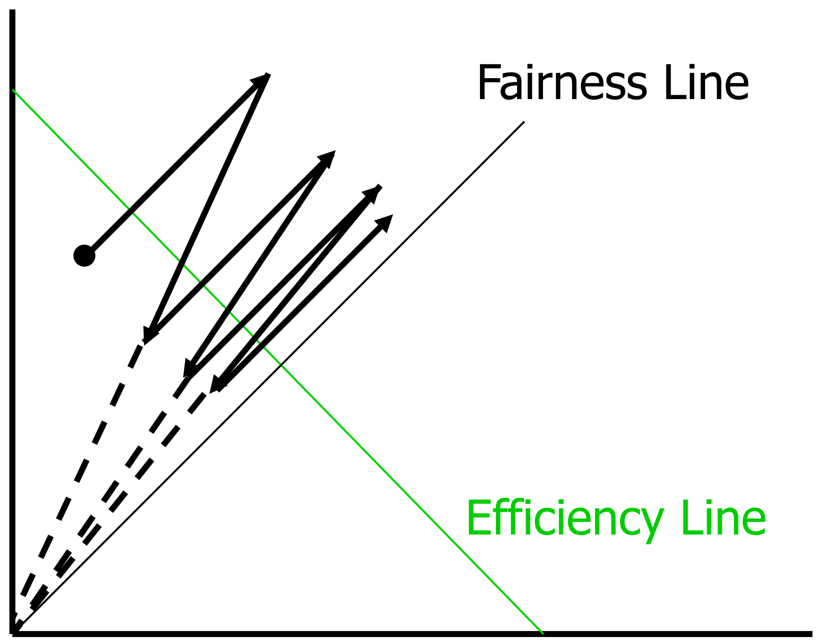
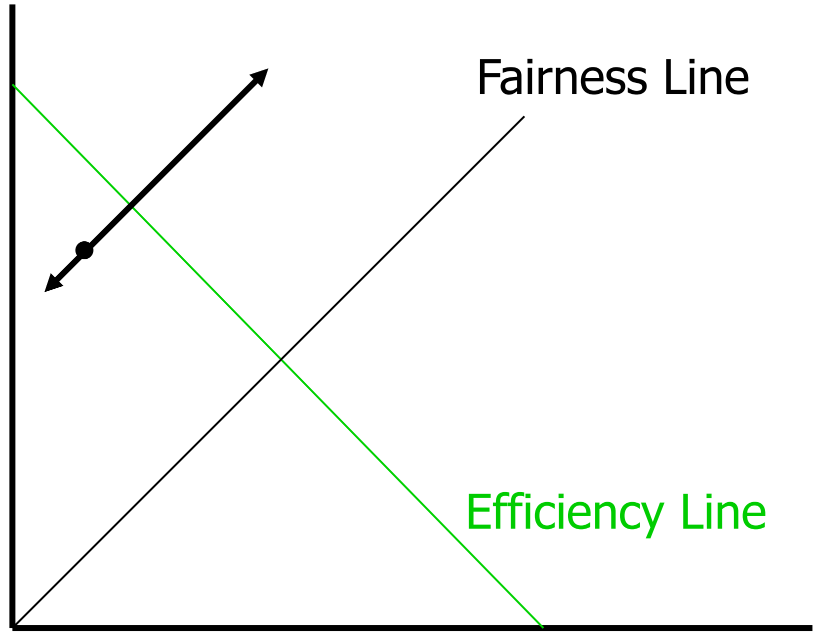


- AIMD converges to optimum efficiency and fairness

Chiu Jain: AIAD



- AIAD doesn't converge to optimum point!
- Similar oscillations for MIMD

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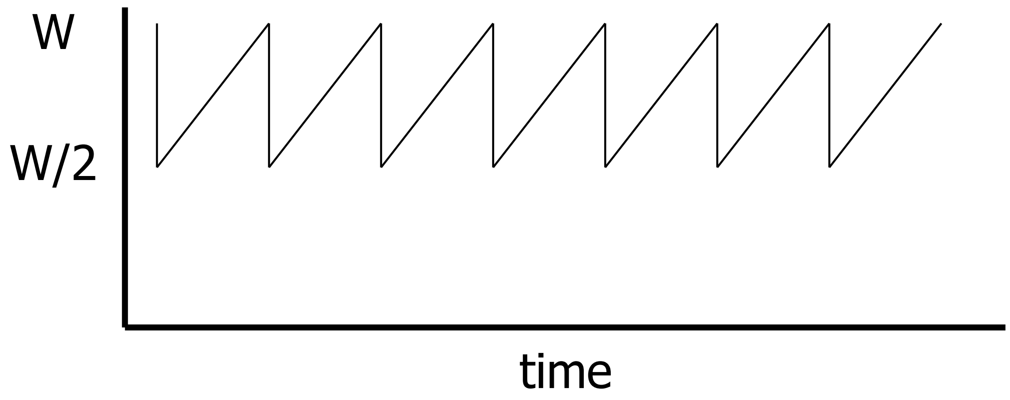
Modeling Throughput, Loss, and RTT

- How do packet loss rate and RTT affect throughput TCP achieves?
- Assume:
  - only fast retransmits
  - no timeouts (so no slow starts in steady-state)

Evolution of Window Over Time



- Average window size: 3W/4
- One window sent per RTT
- Bandwidth:
  - 3W/4 packets per RTT
  - (3W/4 x packet size) / RTT bytes per second
  - W depends on loss rate…

Loss and Window Size

- Assume no delayed ACKs, fixed RTT
- cwnd grows by one packet per RTT
- So it takes W/2 RTTs to go from window size W/2 to window size W; this period is one cycle
- How many packets sent in total?
  - ((3W/4) / RTT) x (W/2 x RTT) = $3W^2/8$
- One loss per cycle (as window reaches W)
  - loss rate: $p = 8/3W^2$
  - W = sqrt(8/3p)

Throughput, Loss, and RTT Model

- W = sqrt(8/3p) = (4/3) x sqrt(3/2p)
- Recall:
  - Bandwidth: B = (3W/4 x packet size) / RTT
- B = packet size / (RTT x sqrt(2p/3))
- Consequences:
  - Increased loss quickly reduces throughput
  - At same bottleneck, flow with longer RTT achieves less throughput than flow with shorter RTT!

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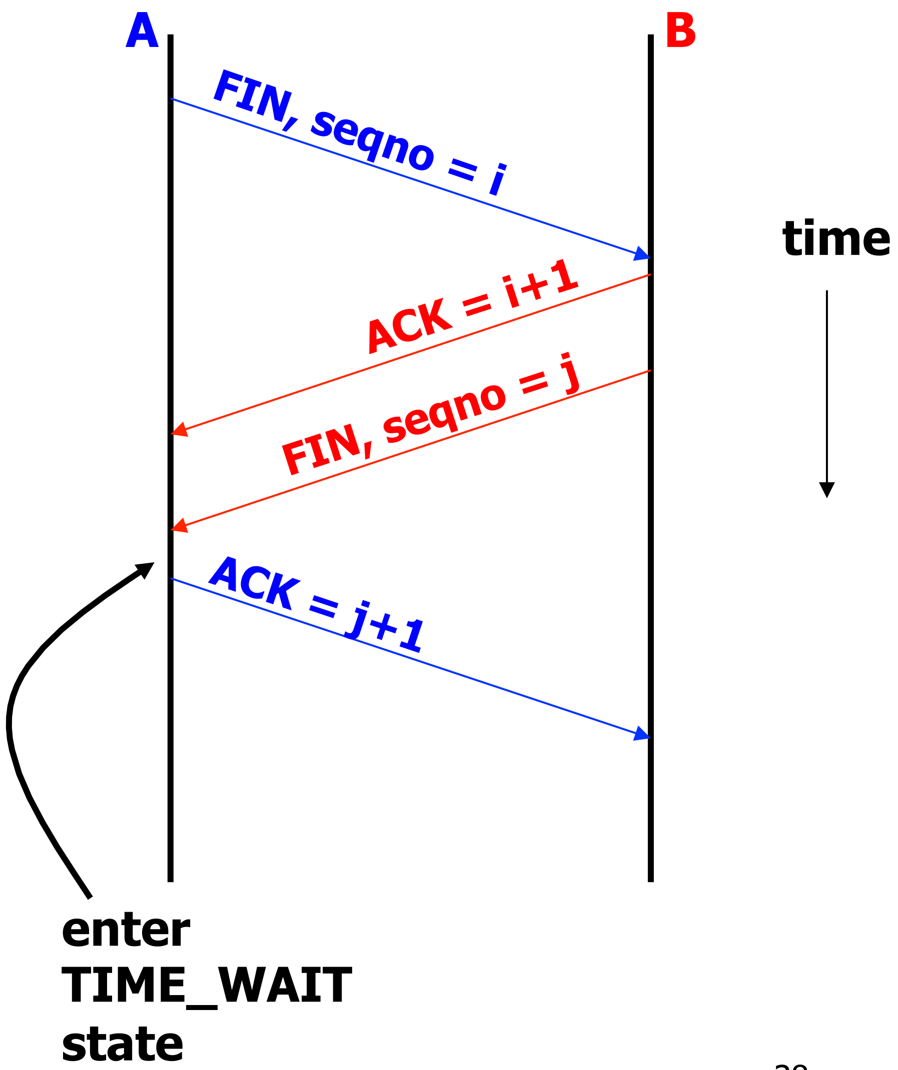
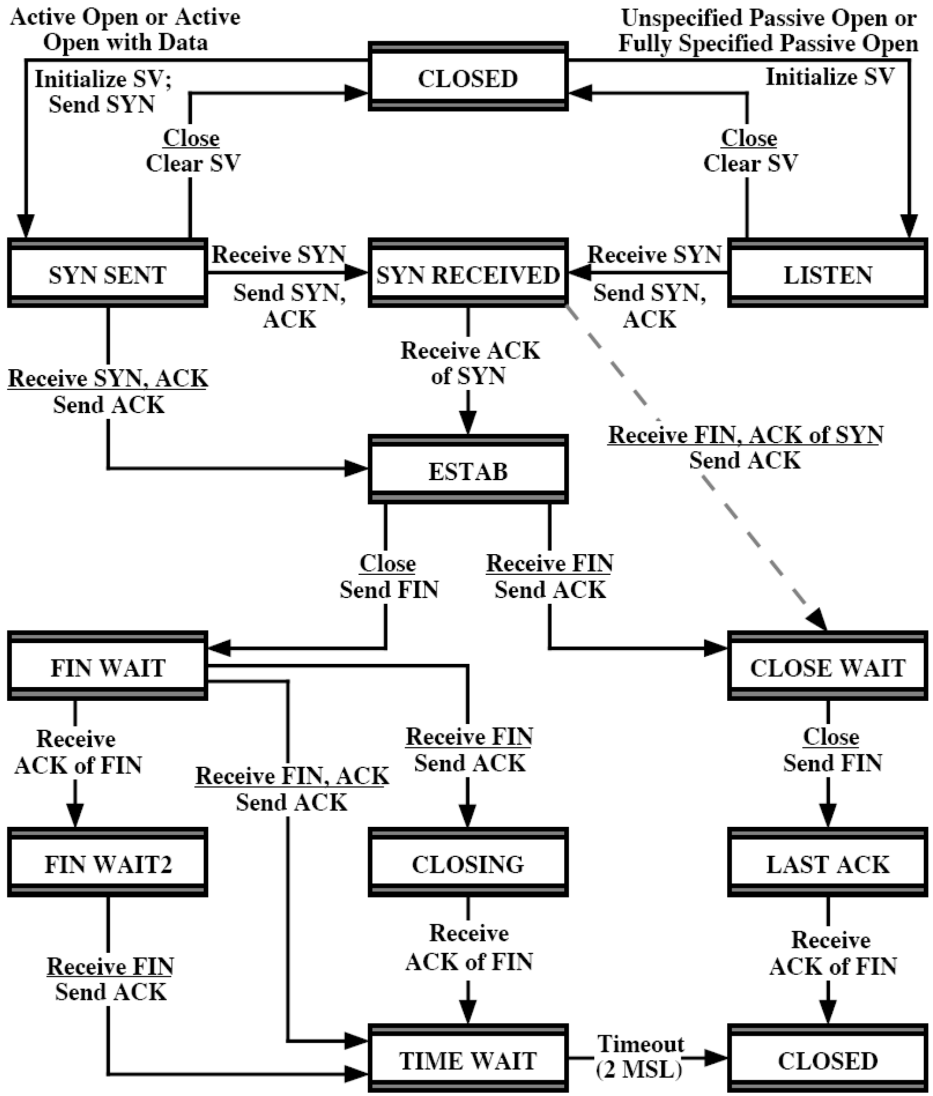
TCP: Connection Teardown

- Data may flow bidirectionally
- Each side independently decides when to close connection
- In each direction, FIN answered by ACK
- Must reliably terminate connection for both sides
  - During TIME_WAIT state at first side to send FIN, ACK valid FINs that arrive
- Must avoid mixing data from old connection with new one
  - During TIME_WAIT state, disallow all new connections for 2 x max segment lifetime



TCP: Protocol State Machine



Summary: TCP and Congestion Control

- Connection establishment and teardown
  - Robustness against delayed packets crucial
- Round-trip time estimation
  - EWMAs estimate both RTT mean and deviation
- Congestion detection at sender
  - Timeout: retransmit timer expires, half window, slow start from one packet
  - Fast Retransmit: three duplicate ACKs, half window, no slow start
- Search for optimal sending window size
  - Additive increase, multiplicative decrease (AIMD)
  - AIMD converges to high utilization, fair sharing