Lecture 15: Congestion Control

CSE 123: Computer Networks

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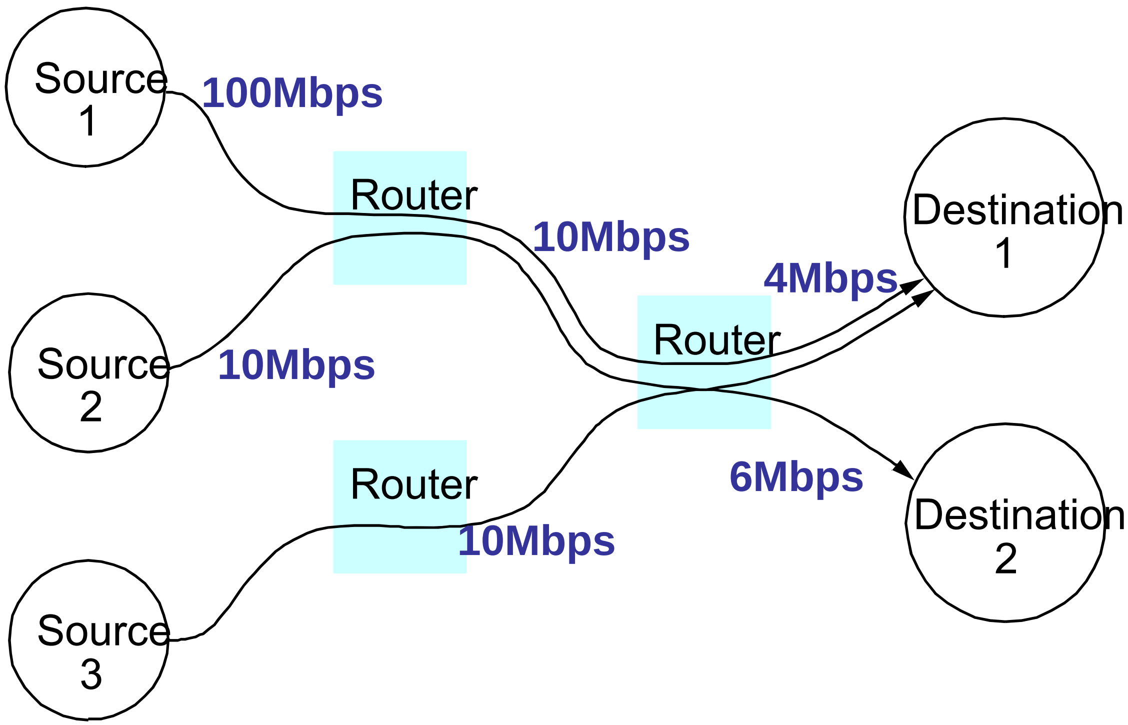
Overview

- Yesterday:
  - TCP & UDP overview
  - Connection setup
  - Flow control: resource exhaustion at end node
- Today: Congestion control
  - Resource exhaustion within the network

Today

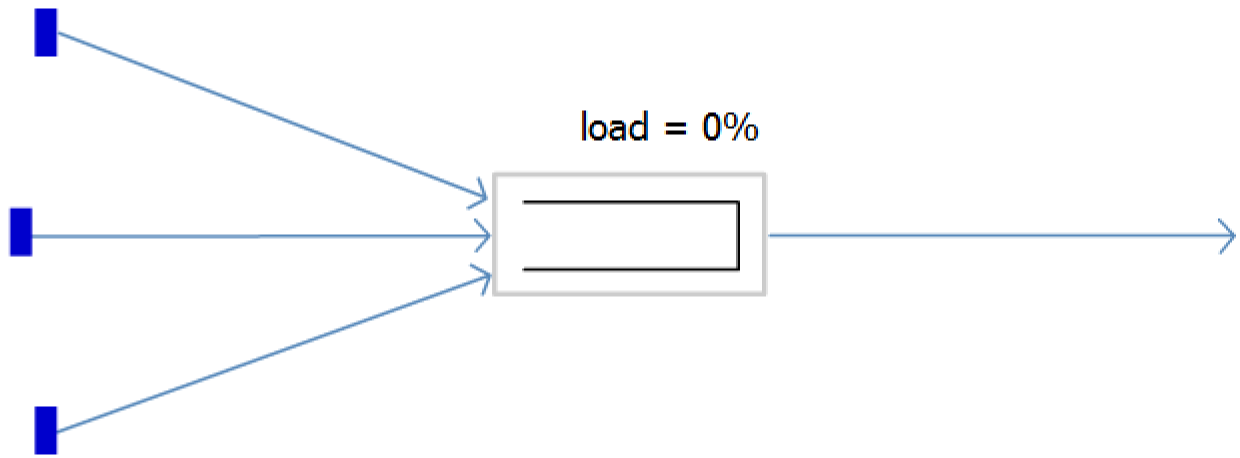
- How fast should a sending host transmit data?
  - Not too fast, not too slow, just right…
- Should not be faster than the receiver can process
  - Flow control (last class)
- Should not be faster than the sender’s share
  - **Bandwidth allocation**
- Should not be faster than the network can process
  - **Congestion control**
- **Congestion control & bandwidth allocation are separate ideas, but frequently combined**

Resource Allocation



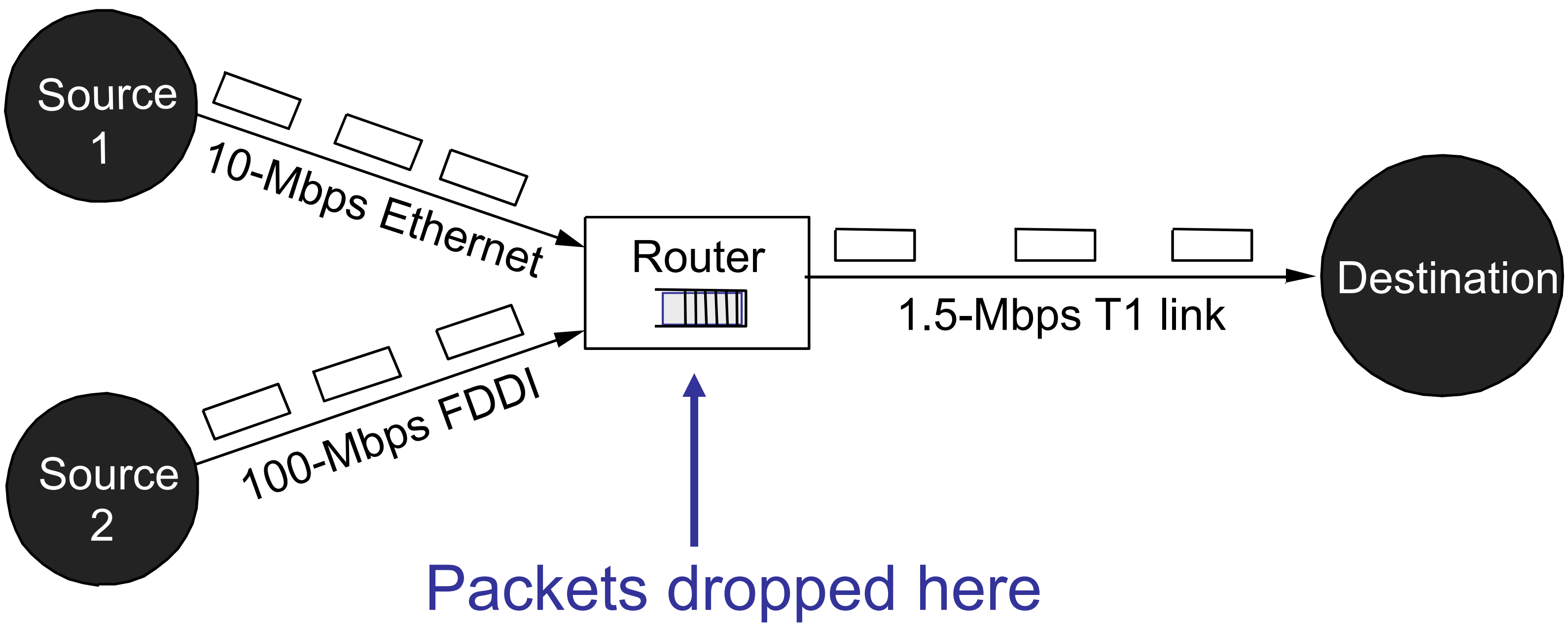
- How much bandwidth should each <u>flow</u> from a source to a destination receive when they compete for resources?
- What is a “fair” allocation?

Statistical Multiplexing



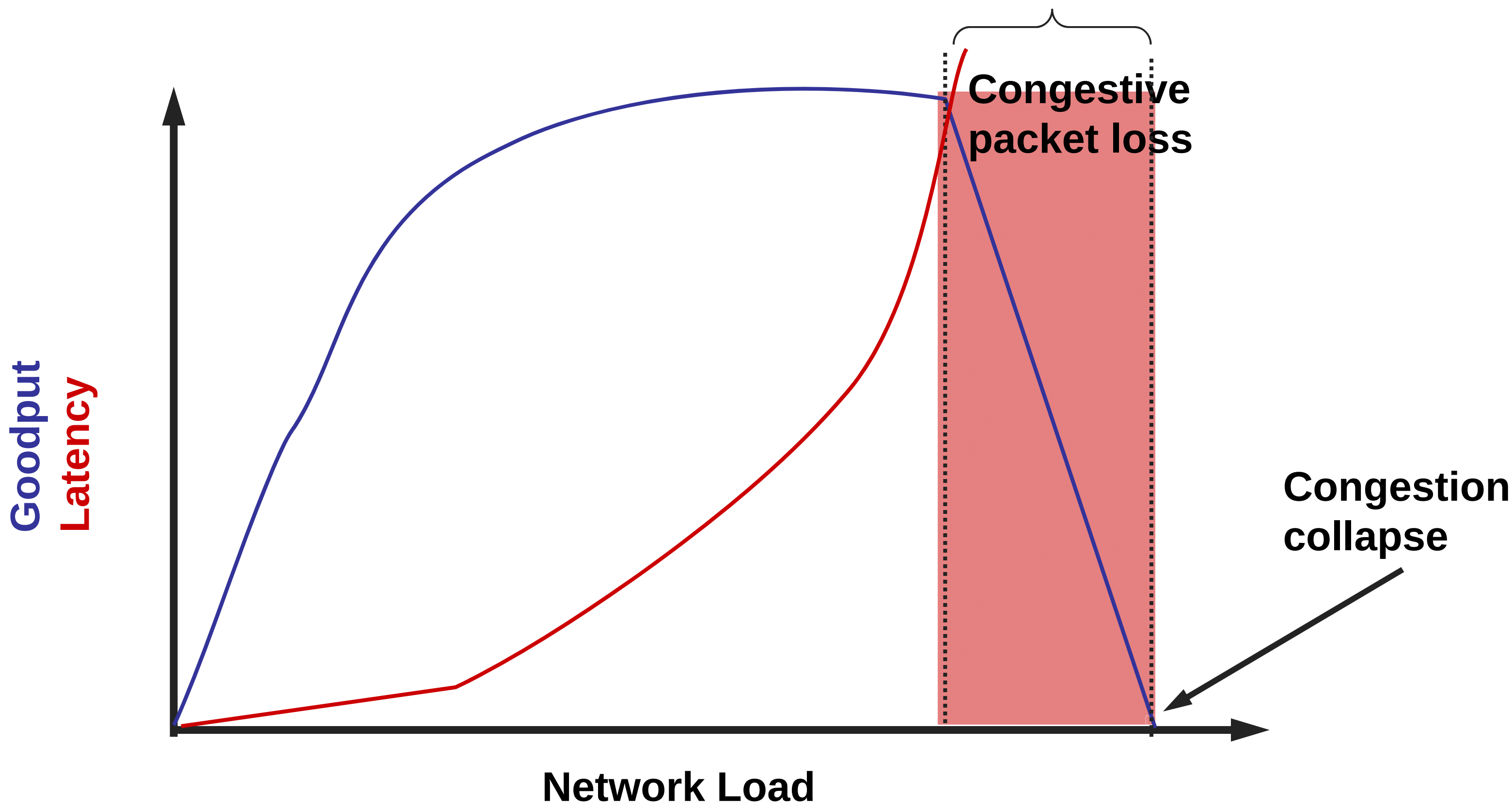
1. The bigger the buffer, the lower the packet loss.
2. If the buffer never goes empty, the outgoing line is busy 100% of the time.

Congestion



- Buffer intended to absorb **bursts** when input rate > output
- But if sending rate is *persistently* > drain rate, queue builds
- Dropped packets represent wasted work; *goodput* < throughput (goodput = useful throughput)

Drop-Tail Queuing



Congestion Collapse

- Rough definition: “When an increase in network load produces a decrease in useful work”

- Why does it happen?
  - Sender sends faster than **bottleneck link** speed
    - Whats a bottleneck link?
  - Packets queue until dropped
  - In response to packets being dropped, sender retransmits
    - Retransmissions further congest link
  - All hosts repeat in steady state…

Mitigation Options

- **Increase network resources**
  - More buffers for queuing
  - Increase link speed
  - Pros/Cons of these approaches?

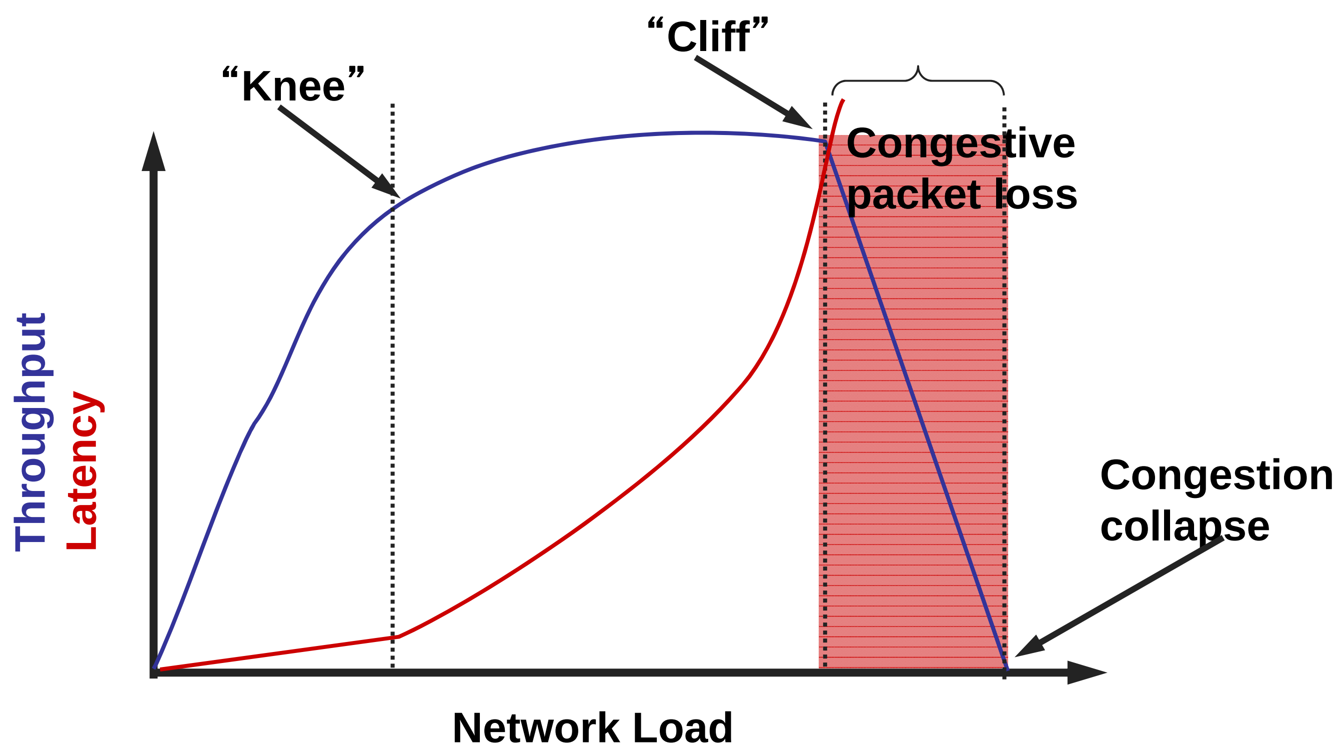
- **Reduce network load**
  - Send data more slowly
  - How much more slowly?
  - When to slow down?

Designing a Control

- Open loop
  - Explicitly reserve bandwidth in the network in advance of sending (next class, a bit)
- Closed loop
  - Respond to feedback and adjust bandwidth allocation
- Network-based
  - Network implements and enforces bandwidth allocation (next class)
- Host-based
  - Hosts are responsible for controlling their sending rate to be no more than their share
- What is typically used on the Internet? Why?

Proactive *vs.* Reactive

- **Congestion avoidance**: try to stay to the left of the “knee”
- **Congestion control**: try to stay to the left of the “cliff”



Challenges to Address

- How to detect congestion?
- How to limit sending data rate?
- How fast to send?

Detecting Congestion

- **Explicit congestion signaling**
  - Source Quench: ICMP message from router to host
    - » Problems?
  - DECBit / Explicit Congestion Notification (ECN):
    - » Router *marks* packet based on queue occupancy (i.e. indication that packet encountered congestion along the way)
    - » Receiver tells sender if queues are getting too full (typically in ACK)
- **Implicit congestion signaling**
  - Packet loss
    - » Assume congestion is primary source of packet loss
    - » Lost packets (timeout, NAK) indicate congestion
  - Packet delay
    - » Round-trip time increases as packets queue
    - » Packet inter-arrival time is a function of bottleneck link

Throttling Options

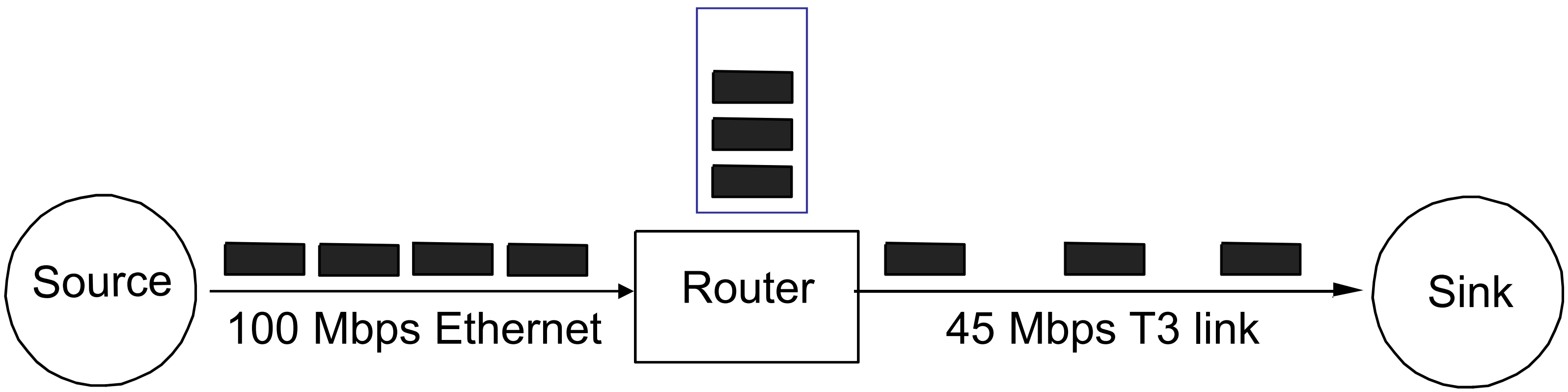
- Window-based (**TCP**)
  - Artificially constrain number of outstanding packets allowed in network
  - Increase window to send faster; decrease to send slower
  - Pro: Cheap to implement, good failure properties
  - Con: Creates traffic *bursts* (requires bigger buffers)
- Rate-based *(many streaming media protocols)*
  - Two parameters (period, packets)
  - Allow sending of x packets in period y
  - Pro: smooth traffic
  - Con: fine-grained per-connection timers,
    what if receiver fails?

Choosing a Send Rate

- Ideally: Keep equilibrium at “knee” of power curve
  - Find “knee” somehow
  - Keep number of packets “in flight” the same
    - E.g., don’t send a new packet into the network until you know one has left (i.e. by receiving an ACK)
  - What if you guess wrong, or if bandwidth availability changes?

- Compromise: adaptive approximation
  - If congestion signaled, reduce sending rate by x
  - If data delivered successfully, increase sending rate by y
  - How to relate x and y? Most choices don’t converge…

TCP's Probing Approach



- Each source independently probes the network to determine how much bandwidth is available
  - Changes over time, since everyone does this
- Assume that packet loss implies congestion
  - Since errors are rare; also, requires no support from routers

Basic TCP Algorithm

- Window-based congestion control
  - Allows congestion control and flow control mechanisms to be unified
  - ***rwin***: *advertised* flow control window from receiver
  - ***cwnd***: congestion control window *estimated* at sender
    - » Estimate of how much outstanding data network can deliver in a round-trip time
  - Sender can only send MIN(*rwin,cwnd*) at any time
- Idea: decrease *cwnd* when congestion is encountered; increase *cwnd* otherwise
- Question: how much to adjust?

Congestion Avoidance

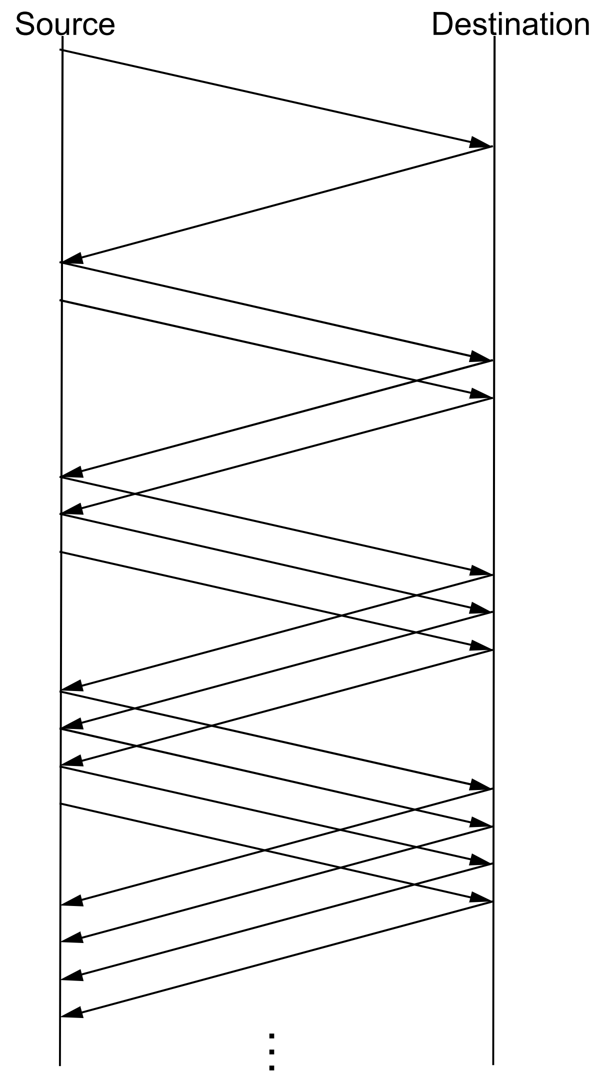
- Goal: Adapt to changes in available bandwidth

- Additive increase, Multiplicative Decrease (AIMD)
  - Increase sending rate by a constant (e.g. by 1500 bytes)
  - Decrease sending rate by a linear factor (e.g. divide by 2)

- Rough intuition for why this works
  - Let L_i be queue length at time i
  - In steady state: $L_i = N$, where N is a constant
  - During congestion, $L_i = N + yL_{i-1}$, where $y > 0$
  - Consequence: queue size increases multiplicatively
    - » Must reduce sending rate multiplicatively as well

AIMD

- Increase slowly while we believe there is bandwidth
  - Additive increase per RTT
  - Cwnd += 1 full packet / RTT
- Decrease quickly when there is loss (went too far!)
  - Multiplicative decrease
  - Cwnd /= 2

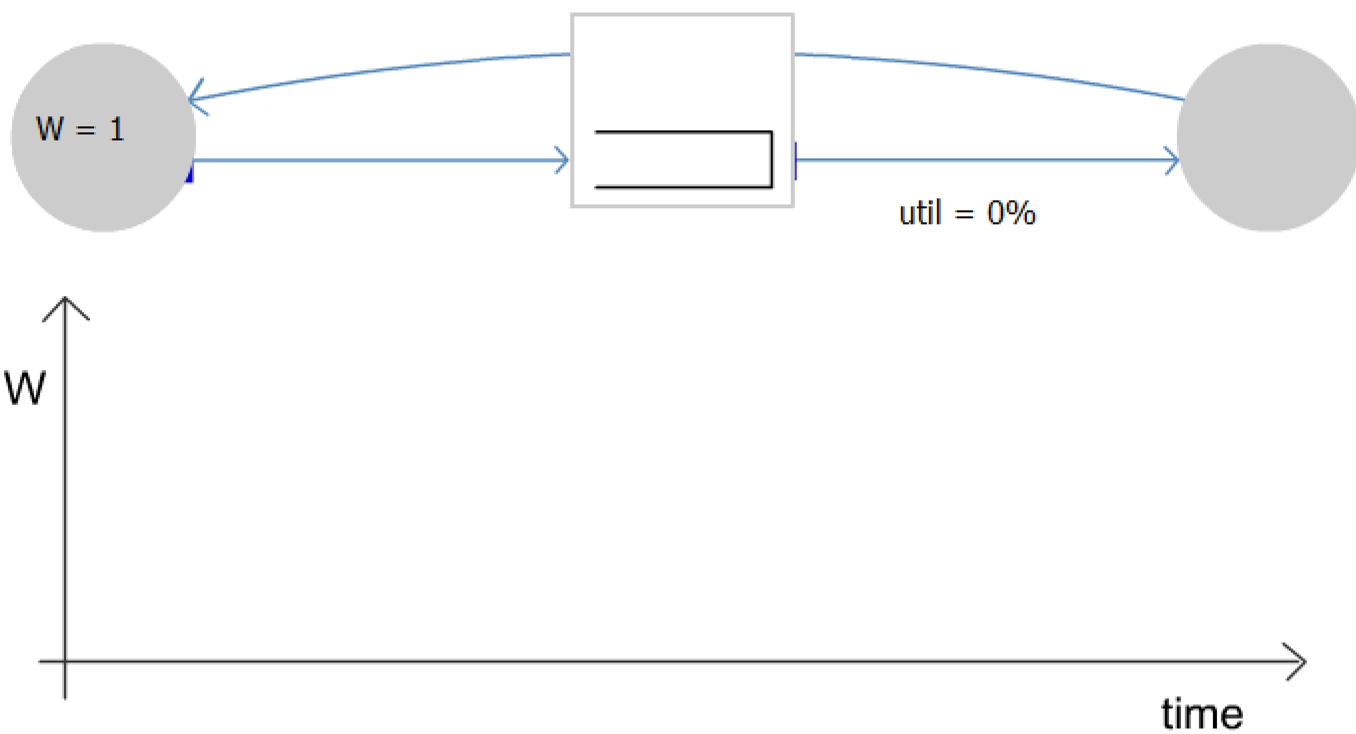


TCP Congestion Control

Rule for adjusting congestion window (W)

- If an ACK is received: W ← W+1/W
- If a packet is lost: W ← W/2

Only W packets
may be outstanding



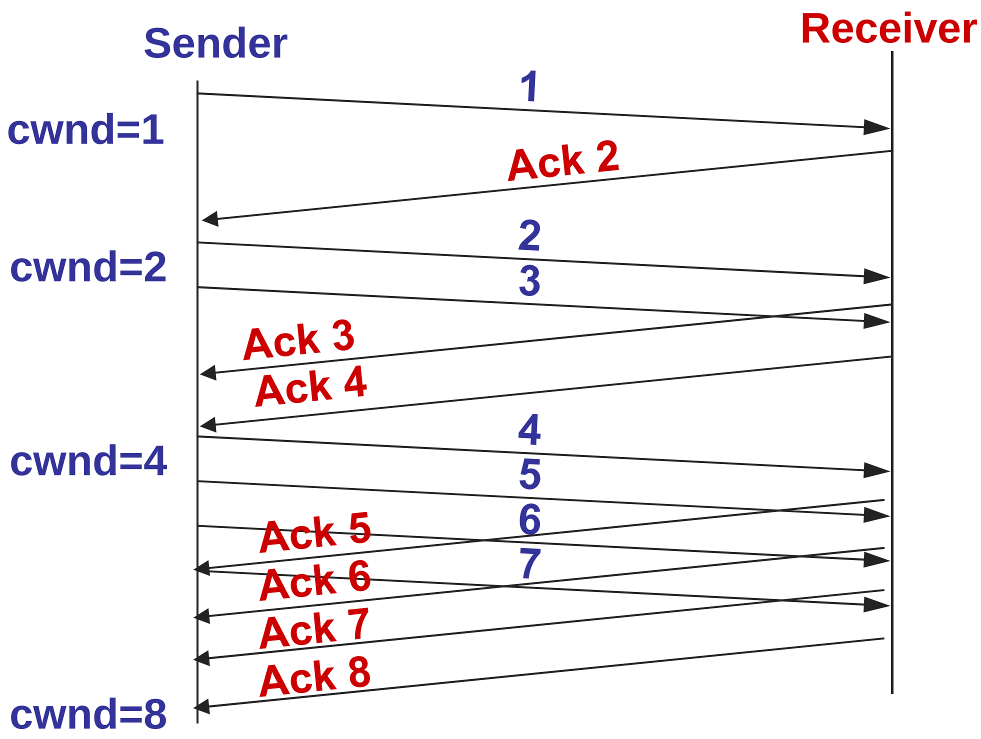
Slow Start

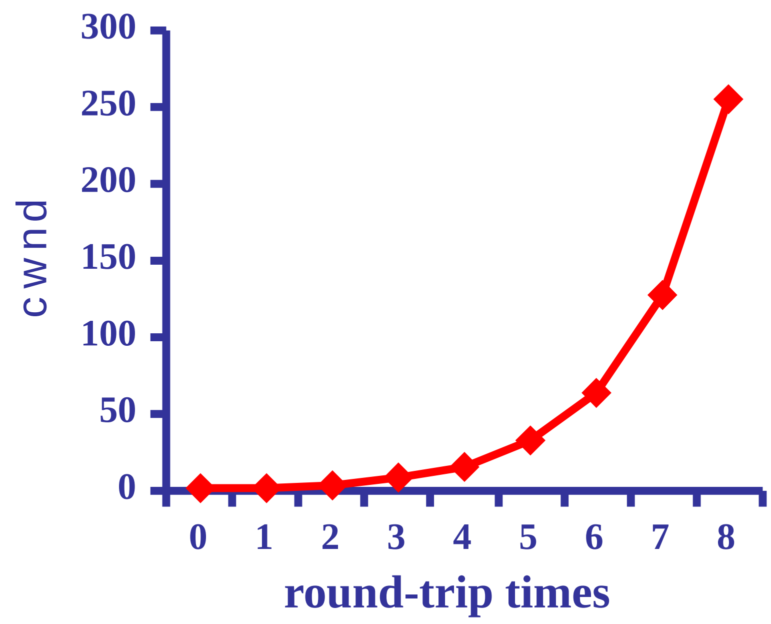
- Goal: **quickly** find the equilibrium sending rate

- Quickly increase sending rate until congestion detected
- Remember last rate that worked and don’t overshoot it
- Algorithm:
  - On new connection, or after timeout, set *cwnd*=1 full pkt
  - For each segment acknowledged, increment *cwnd* by 1 pkt
  - If timeout then divide *cwnd* by 2, and set *ssthresh* = *cwnd*
  - If *cwnd* >= *ssthresh* then exit slow start

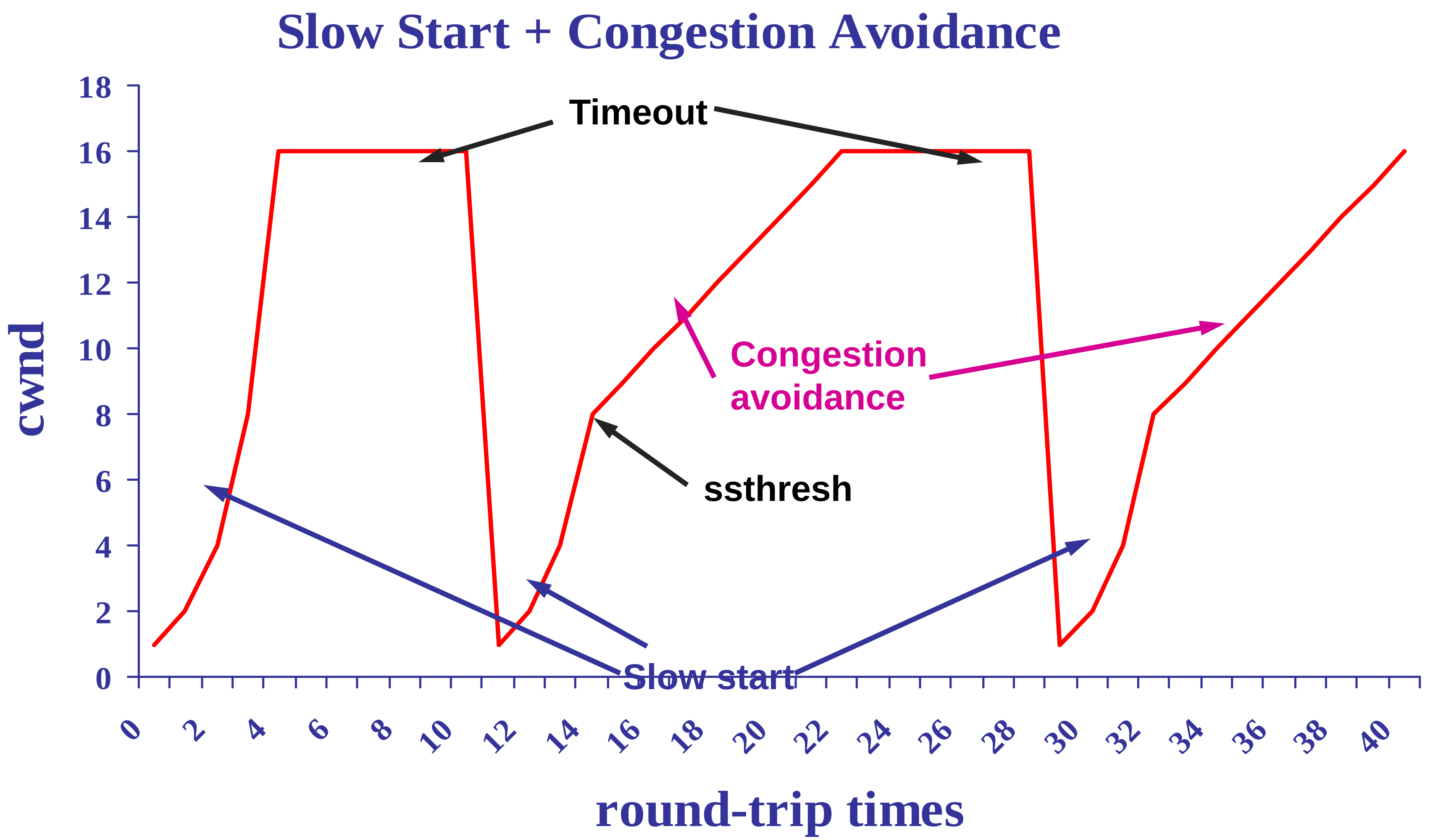
- Why called slow? Its exponential after all…

Slow Start Example





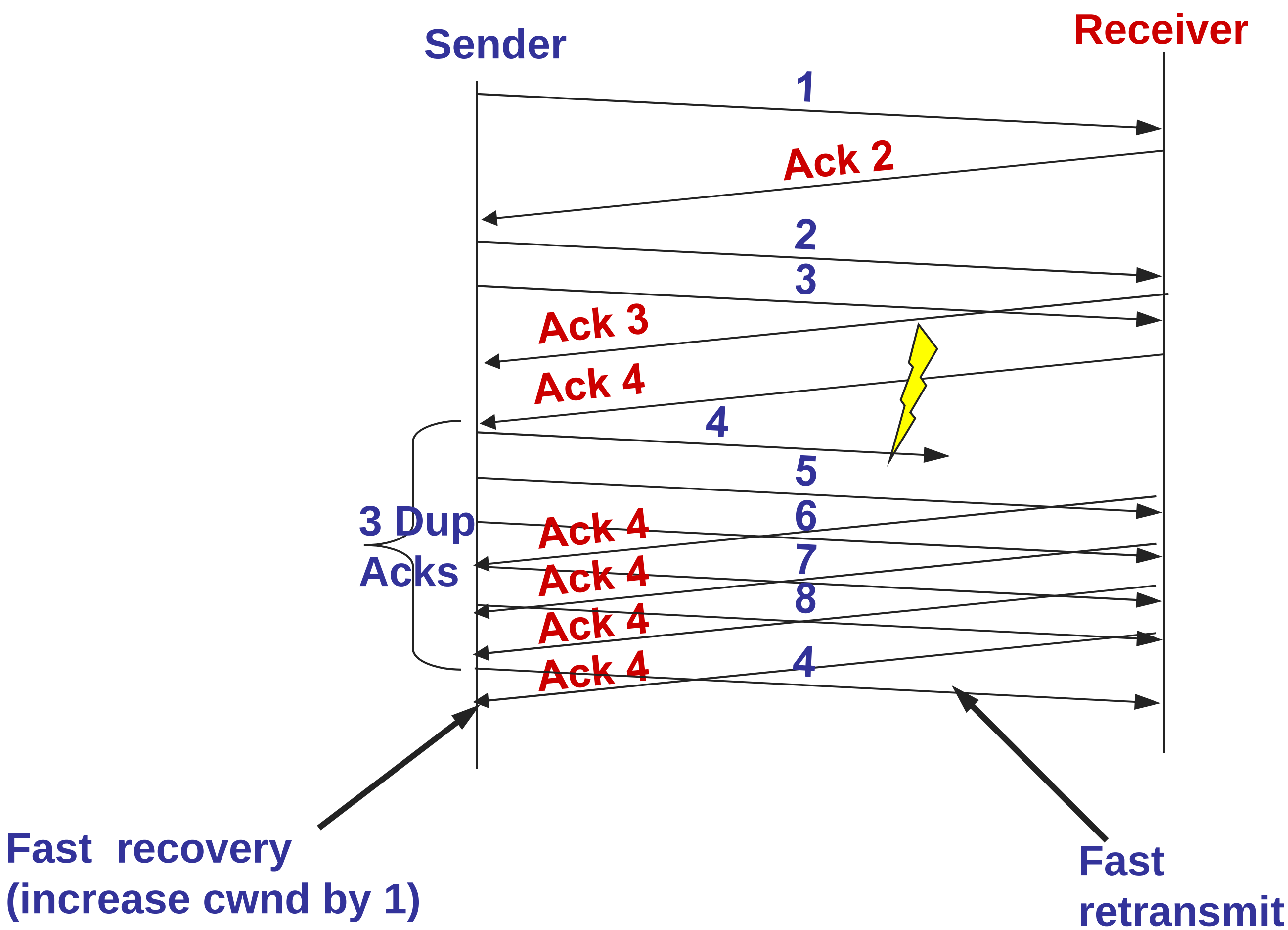
Basic Mechanisms



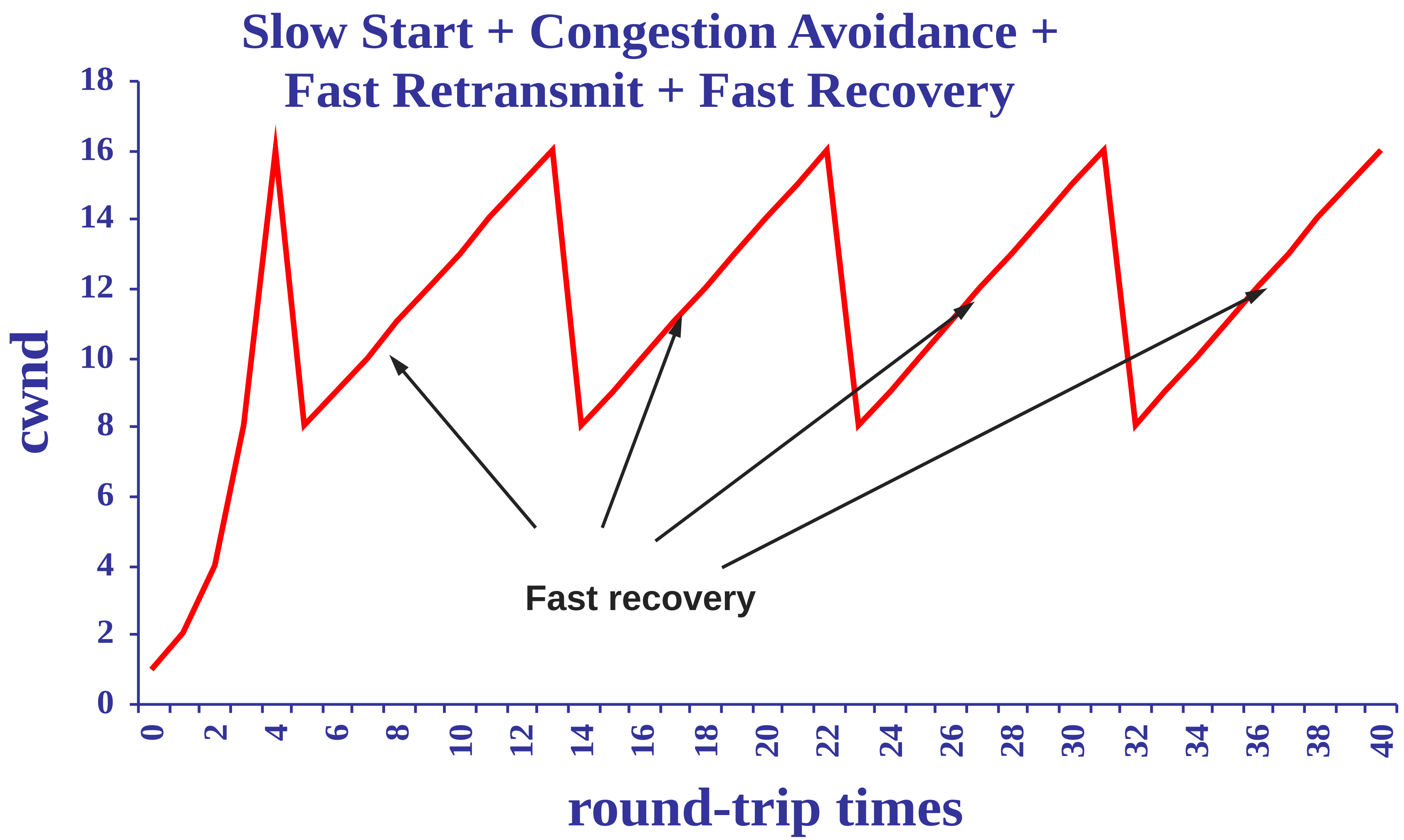
Fast Retransmit & Recovery

- Fast retransmit
  - Timeouts are slow (1 second is fastest timeout on many TCPs)
  - When packet is lost, receiver still ACKs last **in-order** packet
  - Use 3 duplicate ACKs to indicate a loss; detect losses quickly
    - » Why 3? When wouldn't this work?
- Fast recovery
  - Goal: avoid stalling after loss
  - If there are still ACKs coming in, then no **need** for slow start
  - If a packet has made it through -> we can send another one
  - Divide *cwnd* by 2 after fast retransmit
  - Increment cwnd by 1 full pkt for each additional duplicate ACK

Fast Retransmit Example



More Sophistication



Delayed ACKs

- In request/response programs, want to combine an ACK to a request with a response in the same packet
  - Delayed ACK algorithm:
    - Wait 200ms before ACKing
    - Must ACK every other packet (or packet burst)
  - Impact on slow start?

- Must not delay **duplicate ACKs**
  - Why? What is the interaction with the congestion control algorithms?

Short Connections

- Short connection: only contains a handful of pkts
- How do short connections and Slow-Start interact?
  - What happens when a packet is lost during Slow-Start?
  - Will short connections be able to use full bandwidth?

- Do you think most flows are short or long?
- Do you think most traffic is in short flows or long flows?

Open Issues

- TCP is designed around the premise of cooperation
  - What happens to TCP if it competes with a UDP flow?
  - What if we divide cwnd by 3 instead of 2 after a loss?
- There are a bunch of magic numbers
  - Decrease by 2x, increase by 1/cwnd, 3 duplicate acks, initial timeout = 3 seconds, etc
- But overall it works quite well!

Summary

- TCP actively probes the network for bandwidth, assuming that a packet loss signals congestion
- The congestion window is managed with an additive increase/multiplicative decrease policy
  - It took fast retransmit and fast recovery to get there
- Slow start is used to avoid lengthy initial delays
  - Ramp up to near target rate, then switch to AIMD
- Fast recovery is used to keep network “full” while recovering from a loss

For next time…

- Last class: Resource management in the network (i.e., Quality of Service)

- P&D 6.1-6.2, 6.5.1