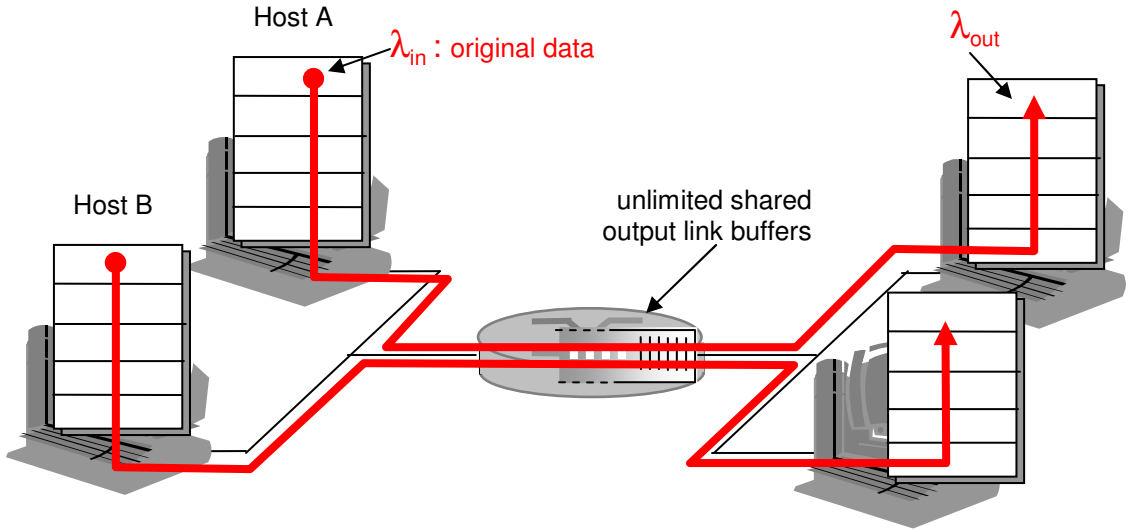
Principles of Congestion Control

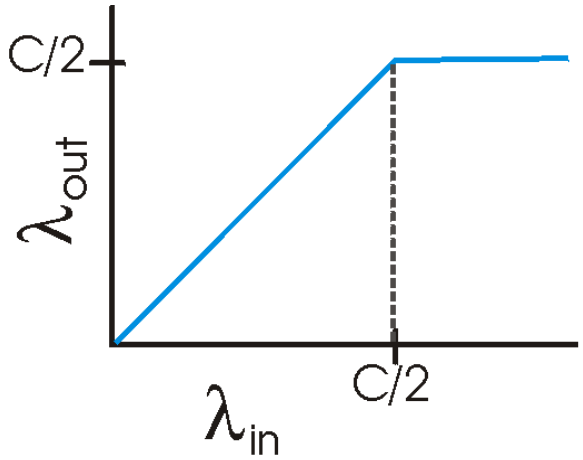
Congestion:

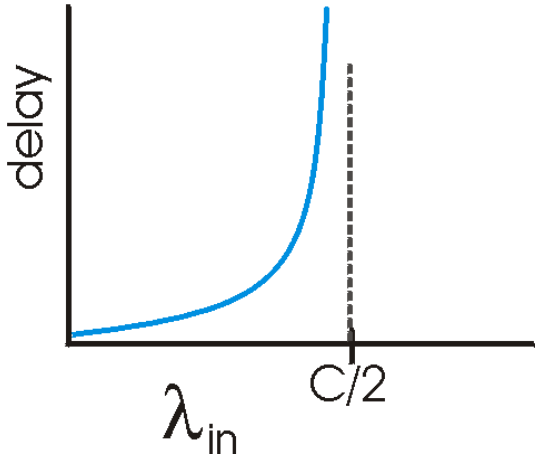
- informally: "too many sources sending too much data too fast for *network* to handle"
- different from flow control!
- manifestations:
  - lost packets (buffer overflow at routers)
  - long delays (queueing in router buffers)
- a top-10 problem!

Causes/costs of congestion: scenario 1

- two senders, two receivers
- one router, infinite buffers
- no retransmission



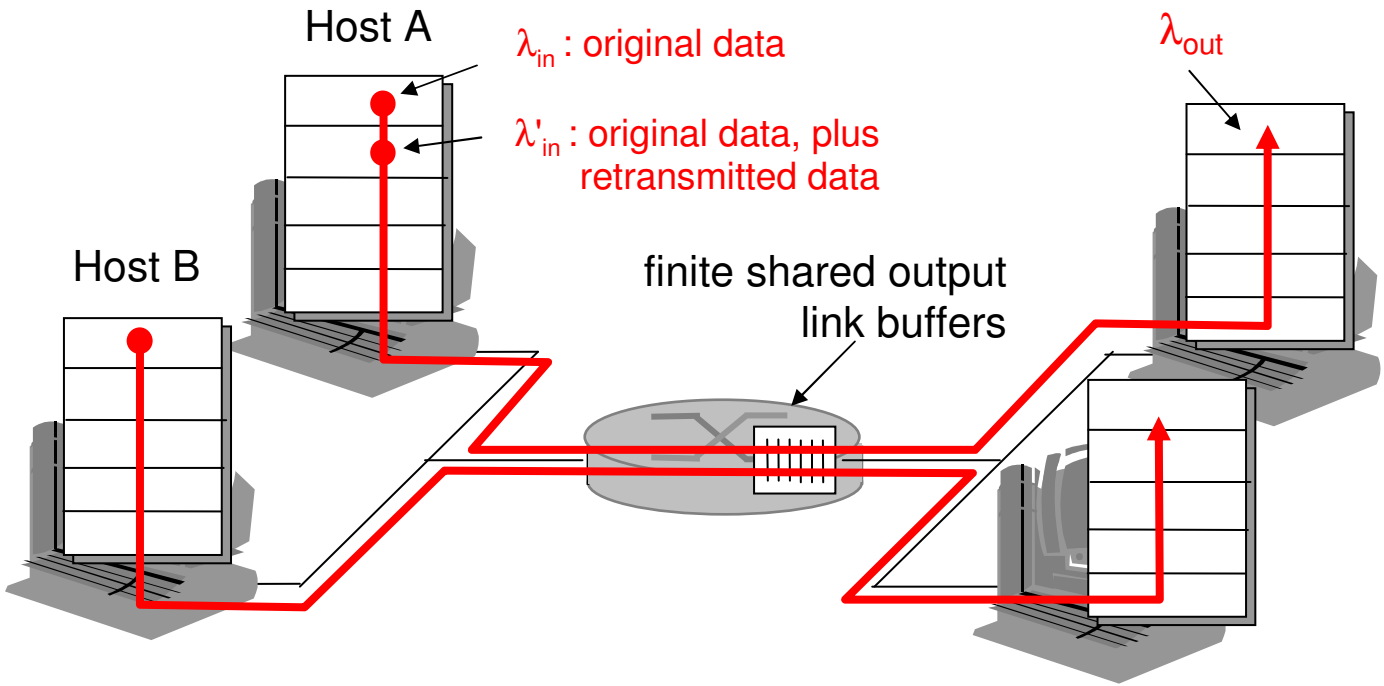




- large delays when congested
- maximum achievable throughput

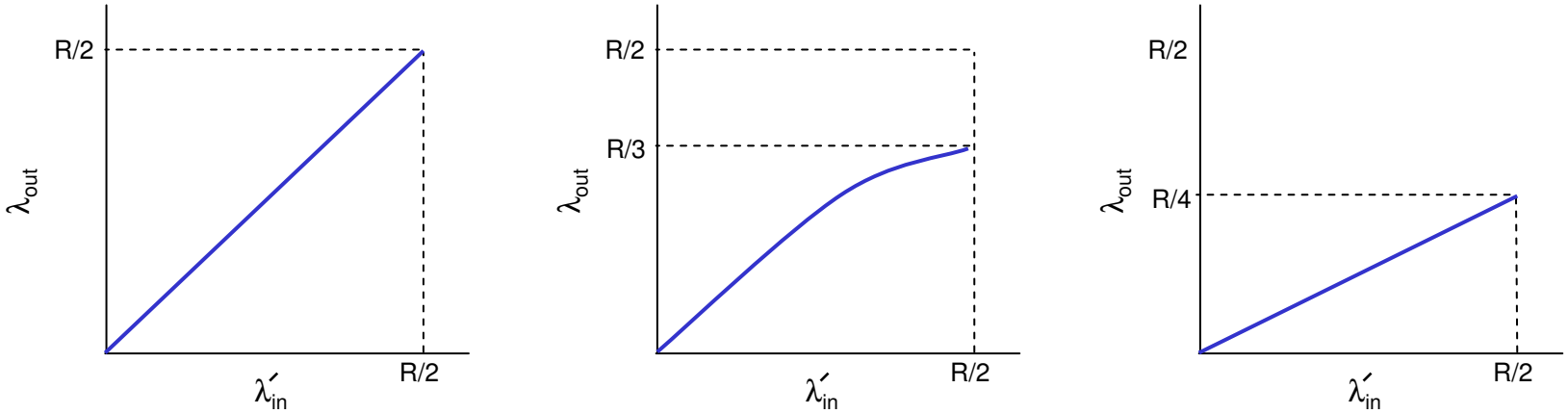
Causes/costs of congestion: scenario 2

- one router, *finite* buffers
- sender retransmission of lost packet



Causes/costs of congestion: scenario 2

- always: $\lambda_{in} = \lambda_{out}$ (goodput)
- "perfect" retransmission only when loss: $\lambda'_{in} < \lambda_{out}$
- retransmission of delayed (not lost) packet makes λ'_{in} larger (than perfect case) for same λ_{out}



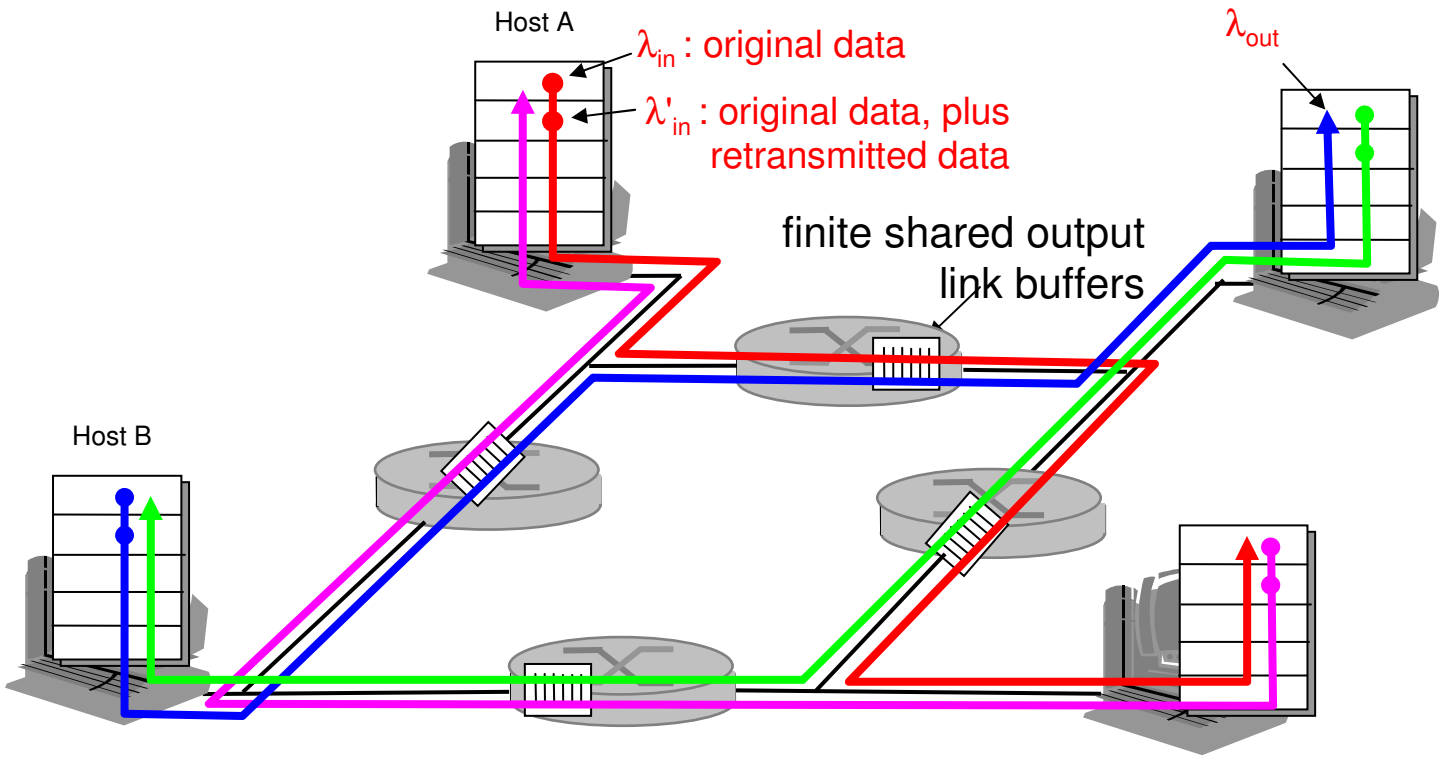
"costs" of congestion:

- more work (retrans) for given "goodput"
- unneeded retransmissions: link carries multiple copies of pkt

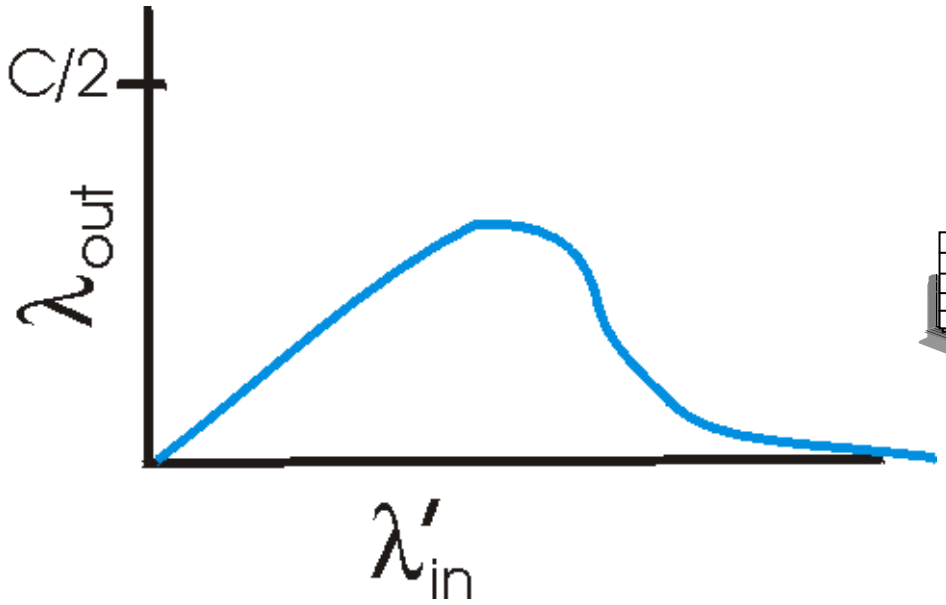
Causes/costs of congestion: scenario 3

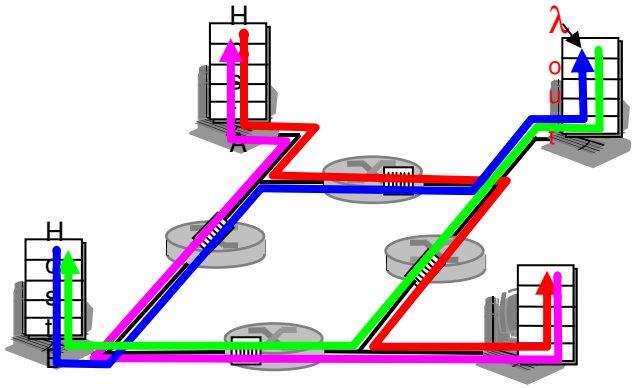
- four senders
- multihop paths
- timeout/retransmit

Q: what happens as λ_{in} and λ'_{in} increase ?



Causes/costs of congestion: scenario 3





another "cost" of congestion:

- when packet dropped, any "upstream transmission capacity used for that packet was wasted!

Approaches towards congestion control

two broad approaches towards congestion control:

end-end congestion control:

- no explicit feedback from network
- congestion inferred from end-system observed loss, delay
- approach taken by TCP

network-assisted congestion control:

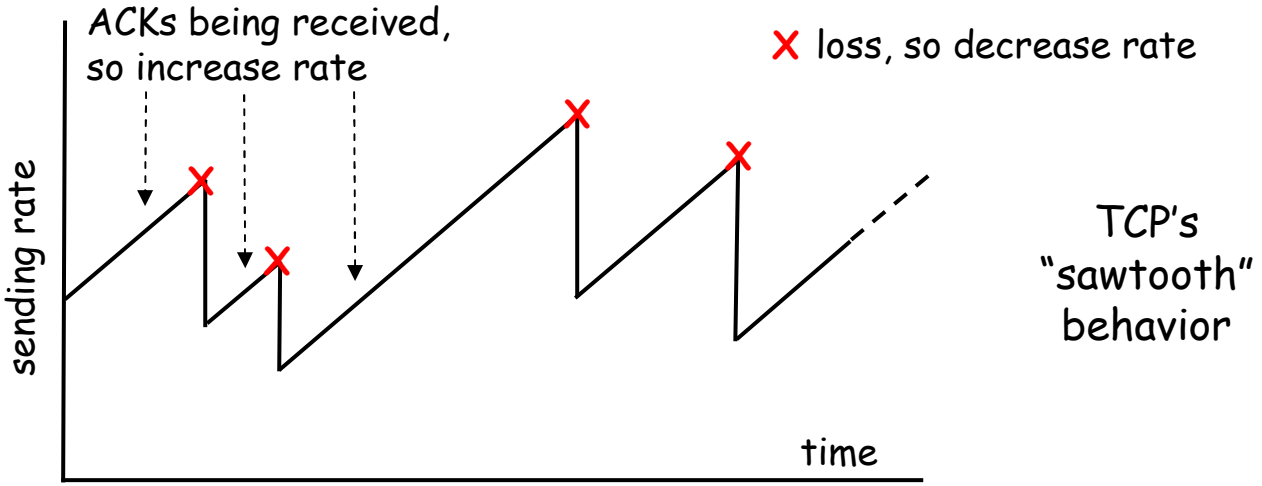
- routers provide feedback to end systems
  - single bit indicating congestion (SNA, DECbit, TCP/IP ECN, ATM)
  - explicit rate sender should send at

TCP congestion control:

- *goal:* TCP sender should transmit as fast as possible, but without congesting network
  - Q: how to find rate *just* below congestion level
- decentralized: each TCP sender sets its own rate, based on *implicit* feedback:
  - *ACK:* segment received (a good thing!), network not congested, so increase sending rate
  - *lost segment:* assume loss due to congested network, so decrease sending rate

TCP congestion control: bandwidth probing

- "probing for bandwidth": increase transmission rate on receipt of ACK, until eventually loss occurs, then decrease transmission rate
  - continue to increase on ACK, decrease on loss (since available bandwidth is changing, depending on other connections in network)



- Q: how fast to increase/decrease?
  - details to follow

TCP Congestion Control: details

- sender limits rate by limiting number of unACKed bytes "in pipeline":

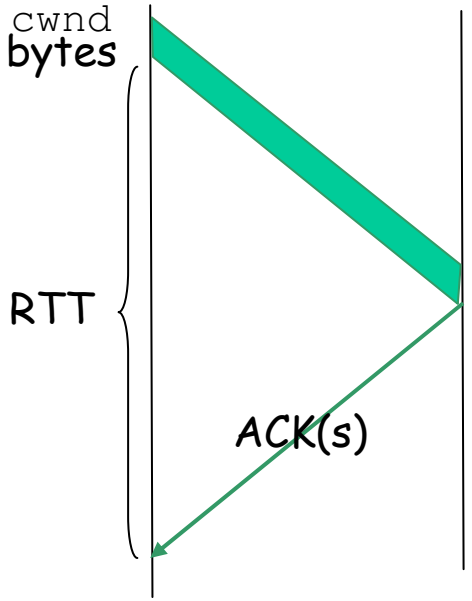
  LastByteSent-LastByteAcked ≤ cwnd

  - **cwnd:** differs from **rwnd** (how, why?)
  - sender limited by **min(cwnd, rwnd)**

- roughly,

$$\text{rate} = \frac{\text{cwnd}}{\text{RTT}} \text{ bytes/sec}$$

- **cwnd** is dynamic, function of perceived network congestion



TCP Congestion Control: more details

segment loss event: reducing **cwnd**

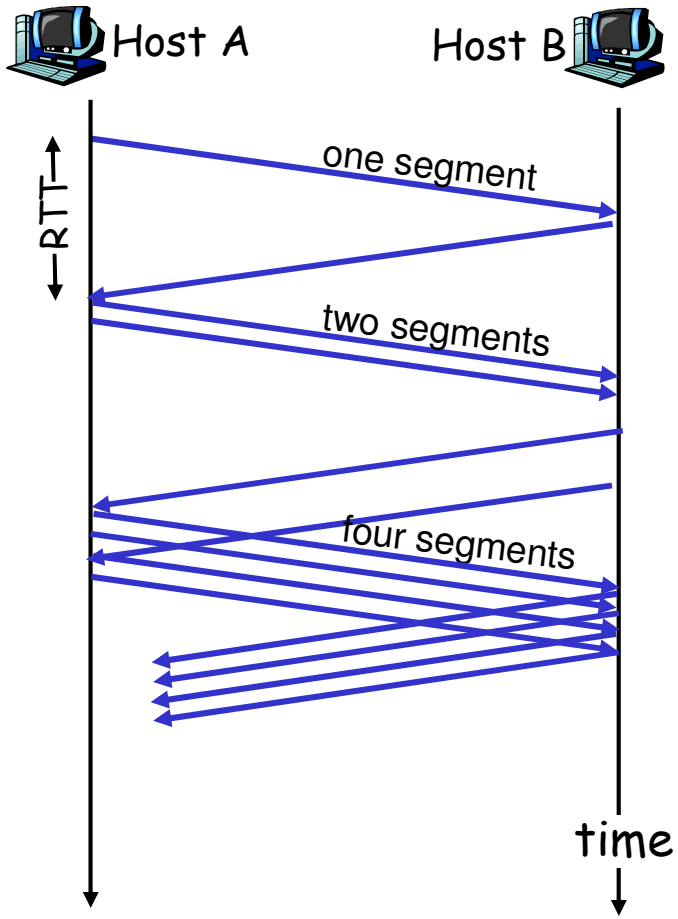
- timeout: no response from receiver
  - cut **cwnd** to 1
- 3 duplicate ACKs: at least some segments getting through (recall fast retransmit)
  - cut **cwnd** in half, less aggressively than on timeout

ACK received: increase **cwnd**

- slowstart phase:
  - increase exponentially fast (despite name) at connection start, or following timeout
- congestion avoidance:
  - increase linearly

TCP Slow Start

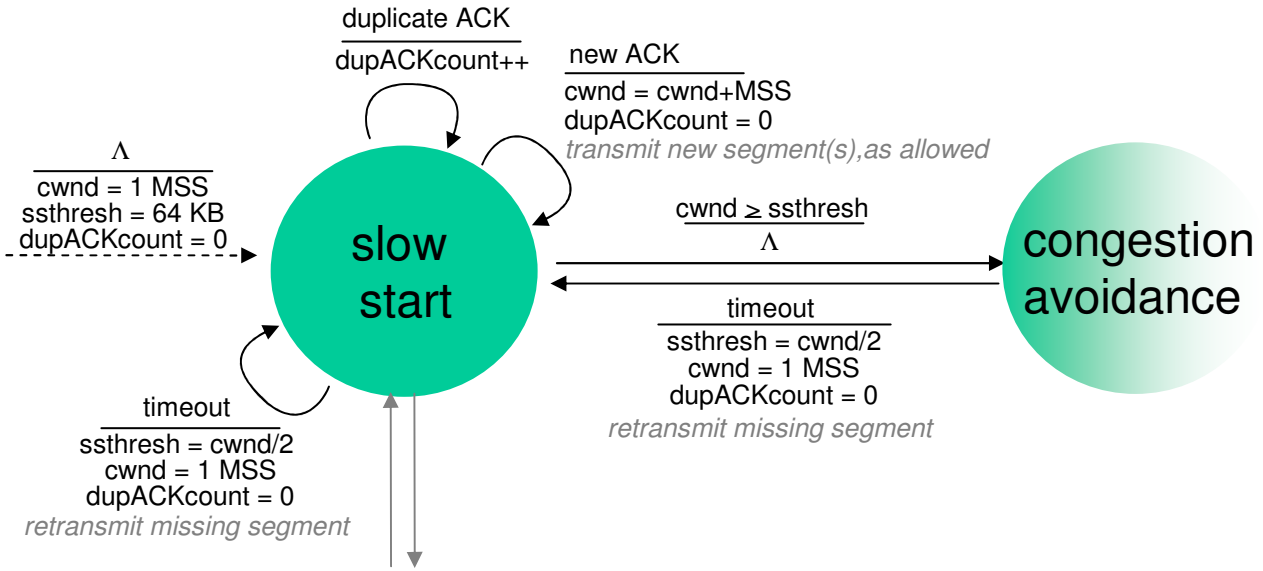
- when connection begins, **cwnd** = 1 MSS
  - example: MSS = 500 bytes & RTT = 200 msec
  - initial rate = 20 kbps
- available bandwidth may be >> MSS/RTT
  - desirable to quickly ramp up to respectable rate
- increase rate exponentially until first loss event or when threshold reached
  - double **cwnd** every RTT
  - done by incrementing **cwnd** by 1 for every ACK received



Transitioning into/out of slowstart

ssthresh: cwnd threshold maintained by TCP

- on loss event: set **ssthresh** to **cwnd/2**
  - remember (half of) TCP rate when congestion last occurred
- when **cwnd >= ssthresh**: transition from slowstart to congestion avoidance phase



TCP: congestion avoidance

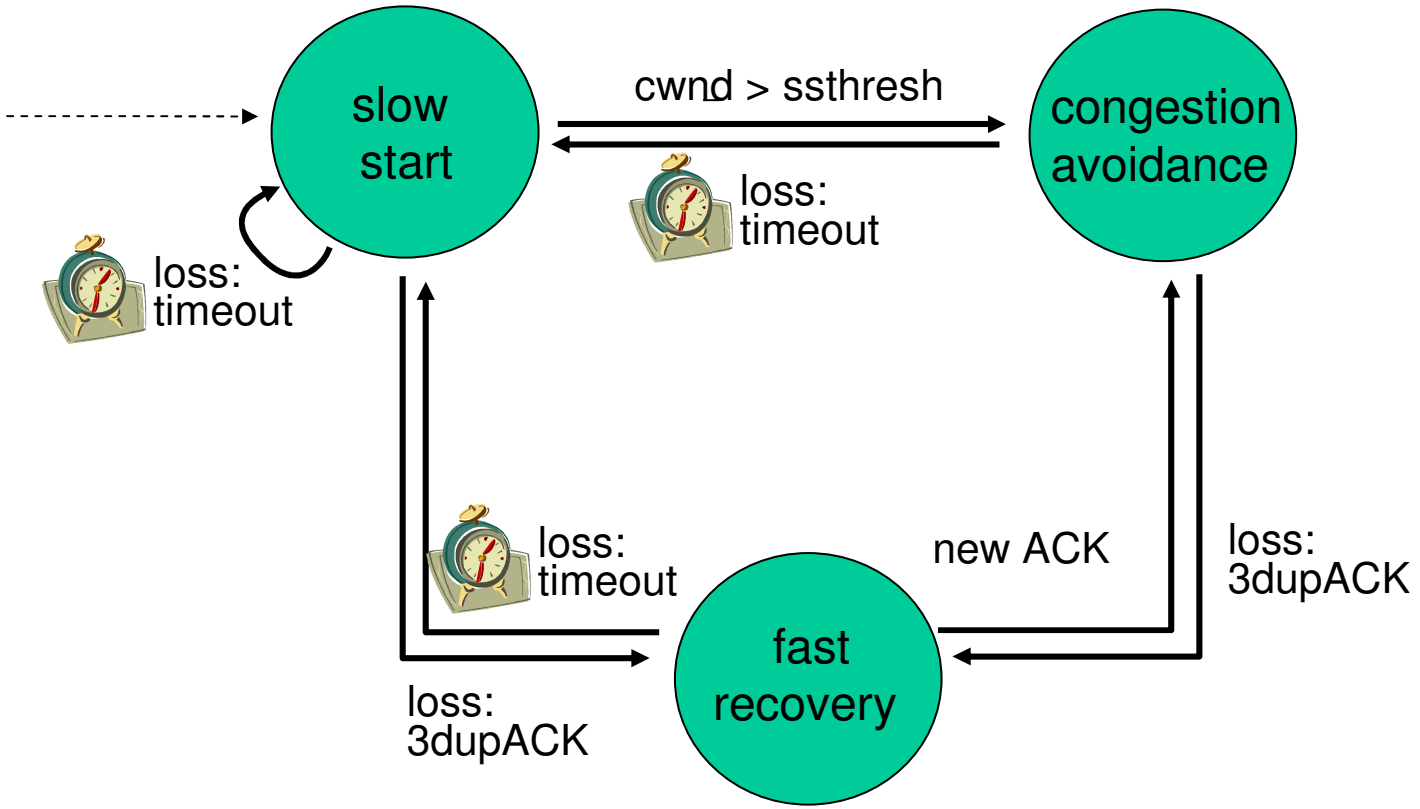
- when **cwnd > ssthresh** grow **cwnd** linearly
  - increase **cwnd** by 1 MSS per RTT
  - approach possible congestion slower than in slowstart
  - implementation: **cwnd = cwnd + MSS/cwnd** for each ACK received

AIMD

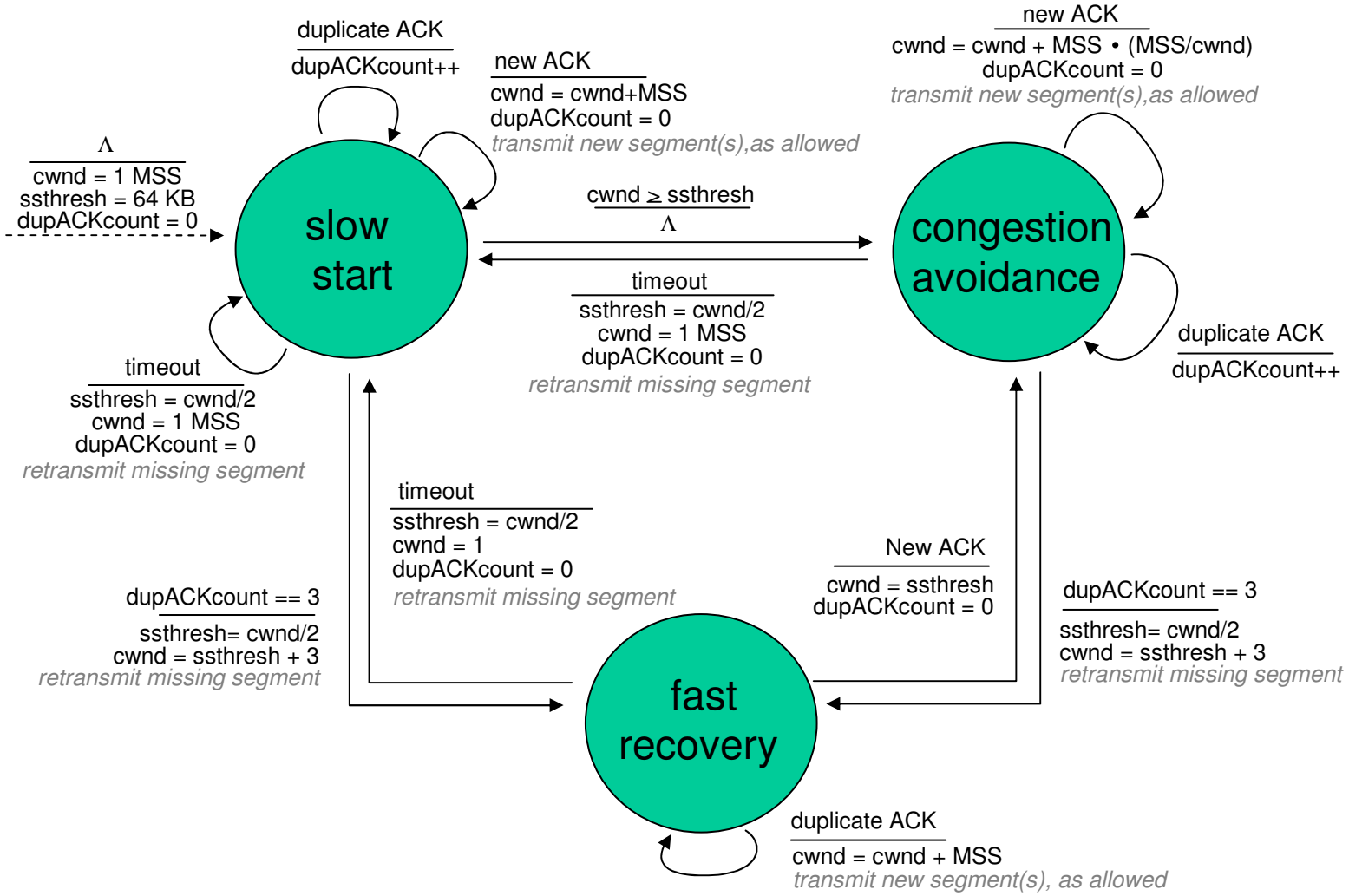
- ACKs: increase **cwnd** by 1 MSS per RTT: additive increase
- loss: cut **cwnd** in half (non-timeout-detected loss): multiplicative decrease

AIMD: Additive Increase Multiplicative Decrease

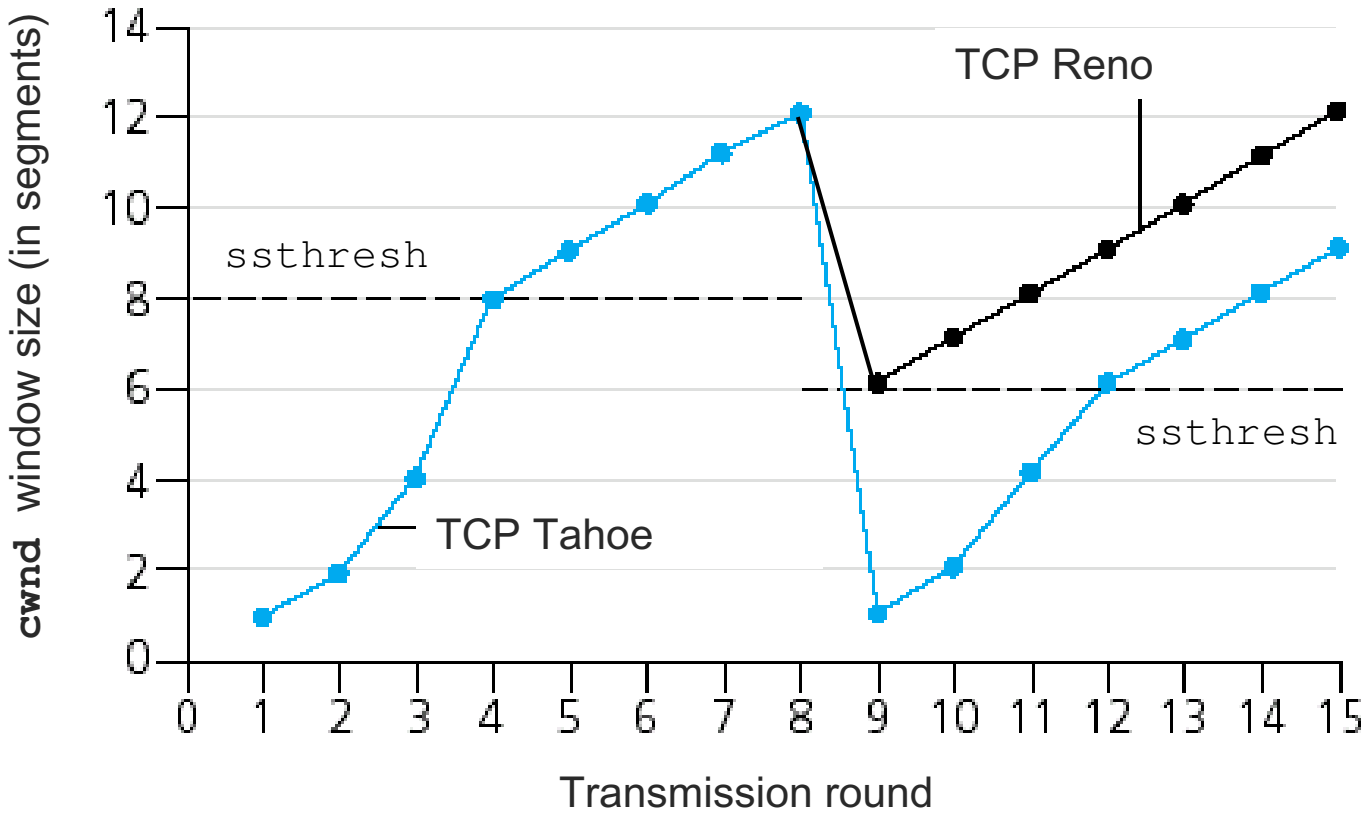
TCP congestion control FSM: overview



TCP congestion control FSM: details



Popular "flavors" of TCP



Summary: TCP Congestion Control

- when **`cwnd < ssthresh`**, sender in slow-start phase, window grows exponentially.
- when **`cwnd >= ssthresh`**, sender is in congestion-avoidance phase, window grows linearly.
- when triple duplicate ACK occurs, **`ssthresh`** set to **`cwnd/2`**, **`cwnd`** set to ~ **`ssthresh`**
- when timeout occurs, **`ssthresh`** set to **`cwnd/2`**, **`cwnd`** set to 1 MSS.

TCP throughput

- *Q:* what's average throughout of TCP as function of window size, RTT?
  - ignoring slow start
- let W be window size when loss occurs.
  - when window is W, throughput is W/RTT
  - just after loss, window drops to W/2, throughput to W/2RTT.
  - average throughout: .75 W/RTT

TCP Futures: TCP over "long, fat pipes"

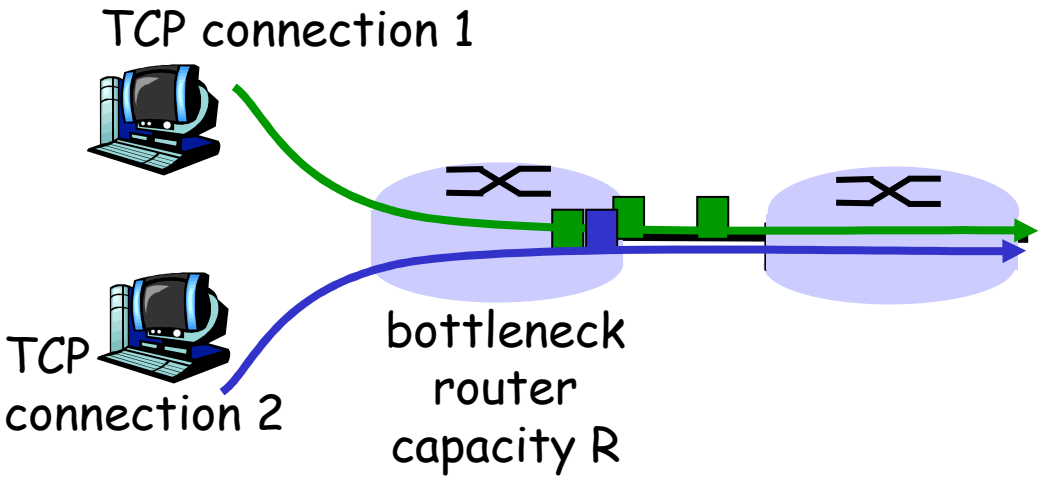
- example: 1500 byte segments, 100ms RTT, want 10 Gbps throughput
- requires window size W = 83,333 in-flight segments
- throughput in terms of loss rate:

$$\frac{1.22 \cdot MSS}{RTT\sqrt{L}}$$

- ➜ $L = 2 \cdot 10^{-10}$ *Wow*
- new versions of TCP for high-speed

TCP Fairness

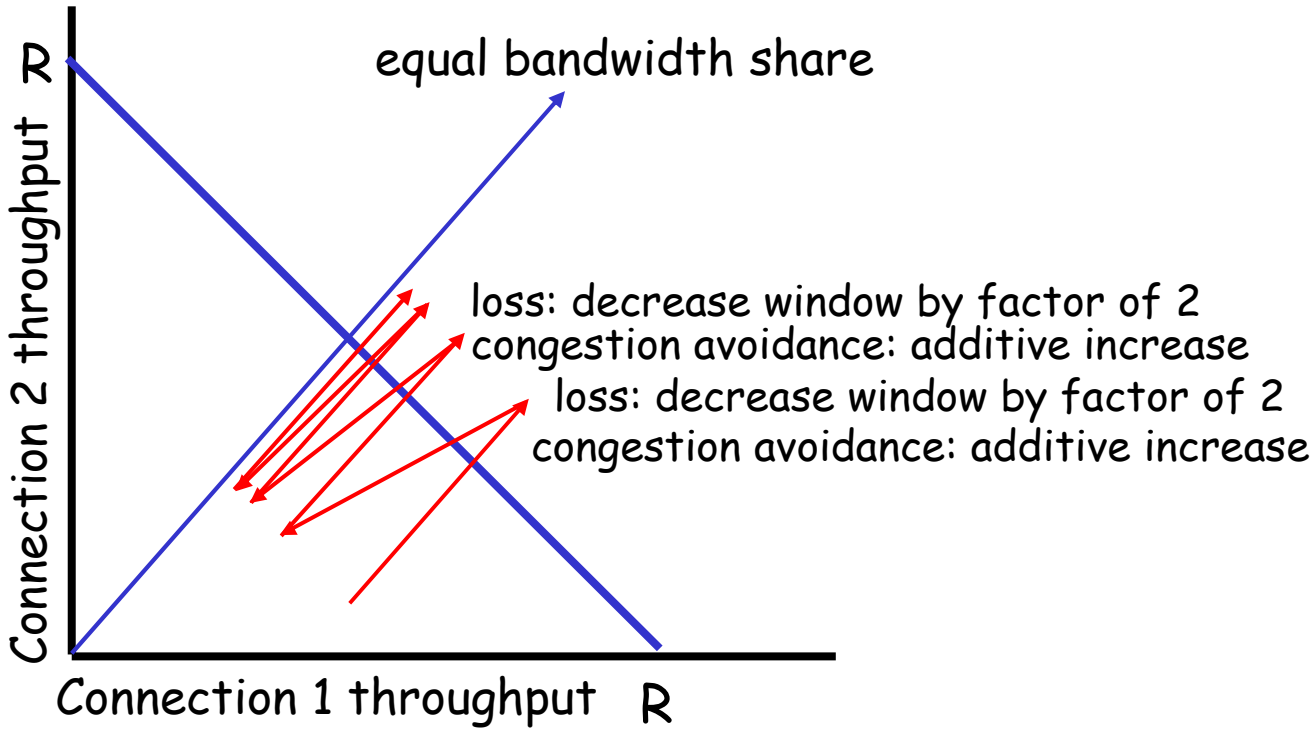
fairness goal: if K TCP sessions share same bottleneck link of bandwidth R, each should have average rate of R/K



Why is TCP fair?

Two competing sessions:

- Additive increase gives slope of 1, as throughput increases
- multiplicative decrease decreases throughput proportionally



Fairness (more)

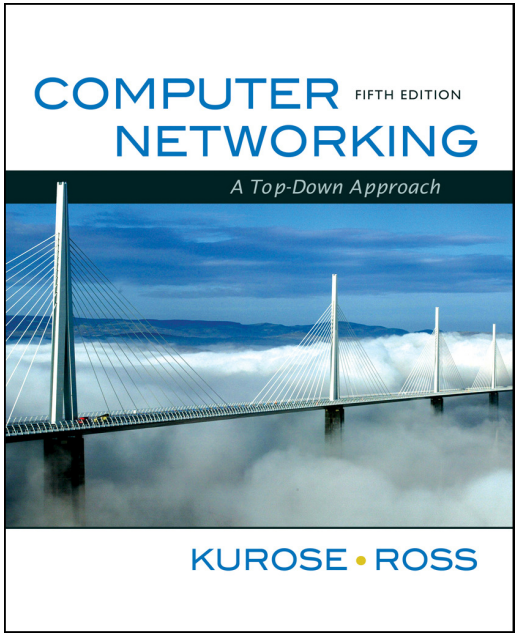
Fairness and UDP

- multimedia apps often do not use TCP
  - do not want rate throttled by congestion control
- instead use UDP:
  - pump audio/video at constant rate, tolerate packet loss

Fairness and parallel TCP connections

- nothing prevents app from opening parallel connections between 2 hosts.
- web browsers do this
- example: link of rate R supporting 9 connections;
  - new app asks for 1 TCP, gets rate R/10
  - new app asks for 11 TCPs, gets R/2 !

Chapter 3
Transport Layer



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Computer Networking: A Top Down Approach 5th edition. Jim Kurose, Keith Ross Addison-Wesley, April 2009.