ACS-3911-050 Computer Network

Chapter 3
Transport Layer

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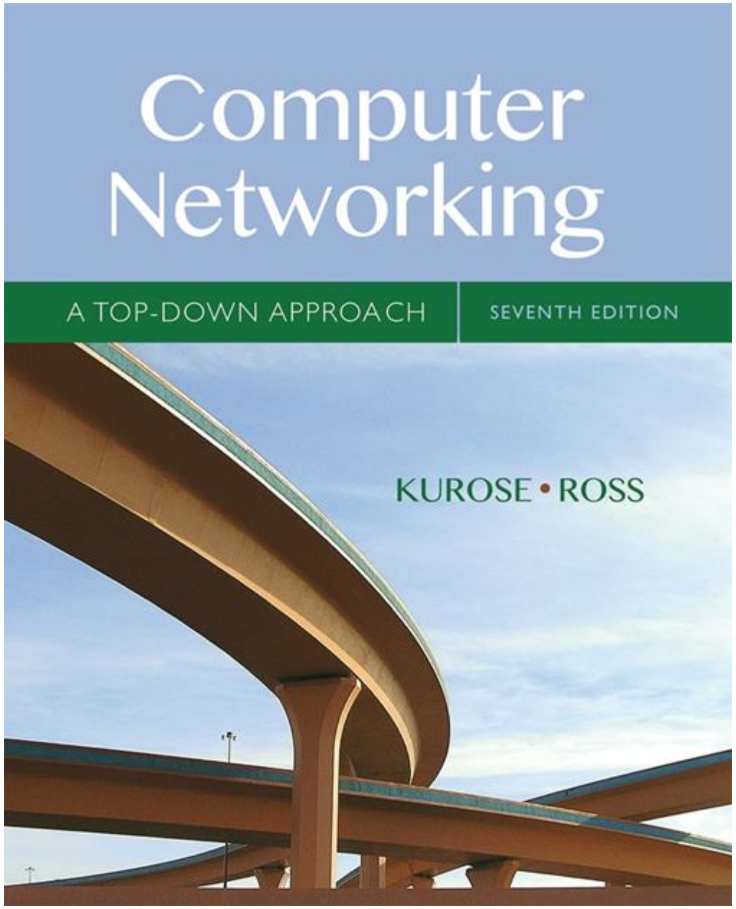

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Thanks and enjoy! JFK/KWR

Roadmap

3.1 transport-layer services

3.2 multiplexing and demultiplexing

3.3 connectionless transport: UDP

3.4 principles of reliable data transfer

3.5 connection-oriented transport: TCP

- segment structure
- reliable data transfer
- flow control
- connection management

3.6 principles of congestion control

3.7 TCP congestion control

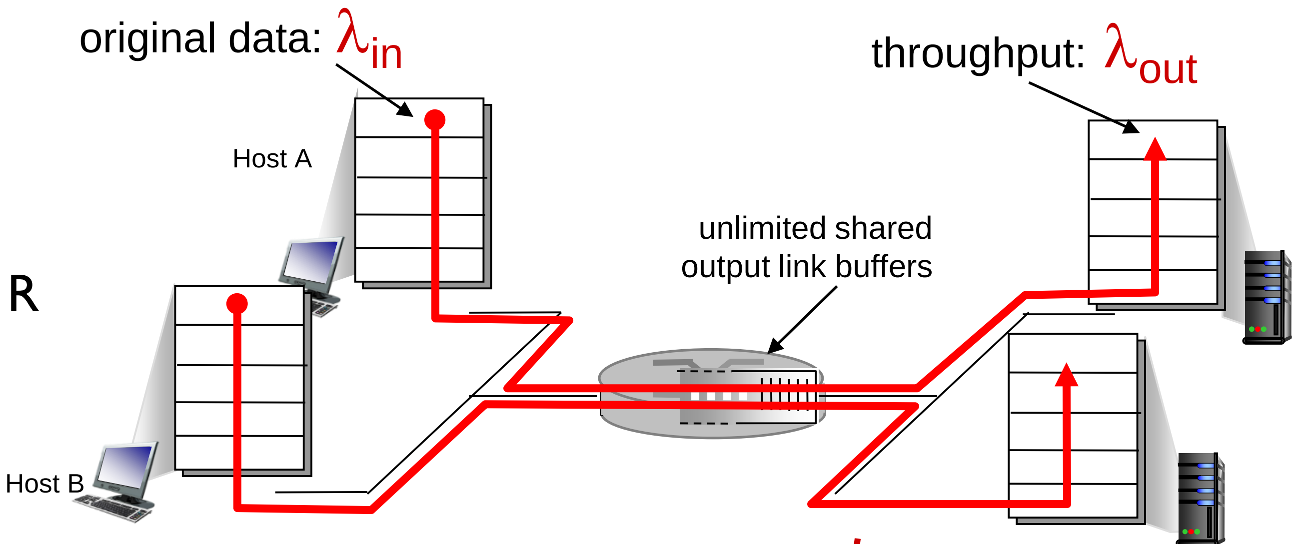
Principles of Congestion Control

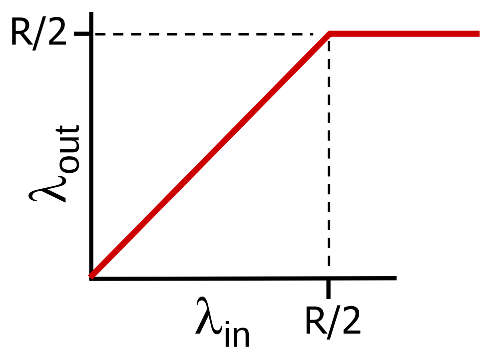
congestion:

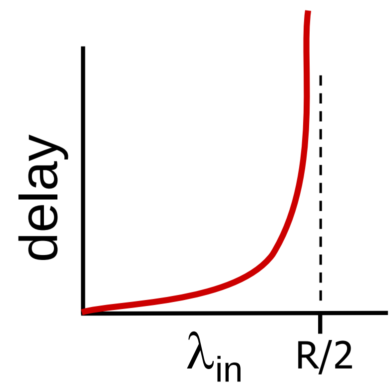
- informally: "too many sources sending too much data too fast for *network* to handle"
- different from flow control!
- manifestations:
  - lost packets (buffer overflow at routers)
  - long delays (queueing in router buffers)
- a top-10 problem!

Causes/Costs of Congestion: Scenario 1

- two senders, two receivers
- one router, infinite buffers
- output link capacity: R
- no retransmission



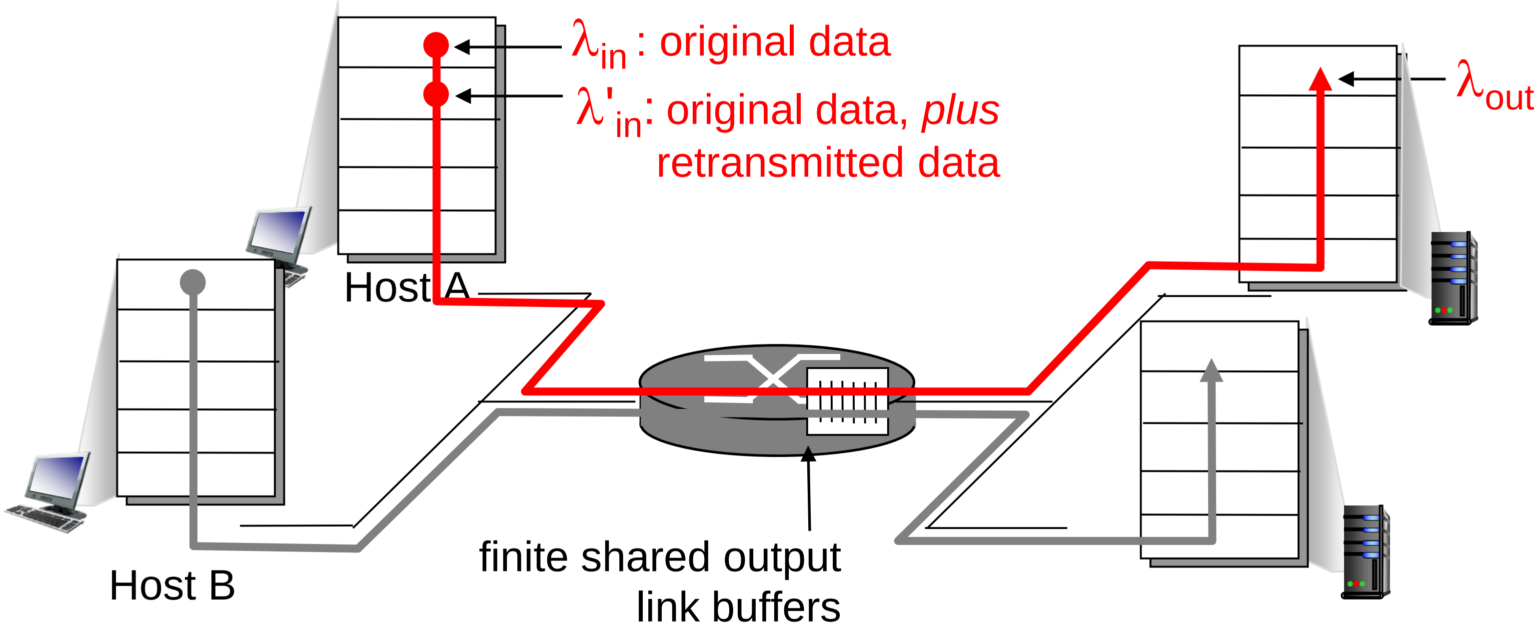




- maximum per-connection throughput: R/2
- large delays as arrival rate, λ_{in}, approaches capacity

Causes/Costs of Congestion: Scenario 2

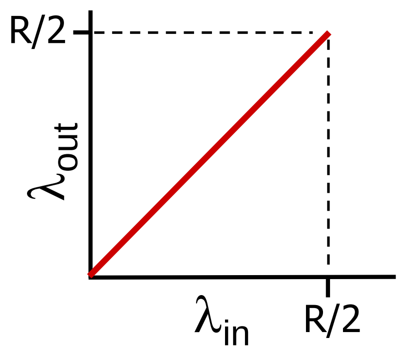
- one router, *finite* buffers
- sender retransmission of timed-out packet
  - application-layer input = application-layer output: $\lambda_{in} = \lambda_{out}$
  - transport-layer input includes *retransmissions* : $\lambda_{in}' \geq \lambda_{in}$

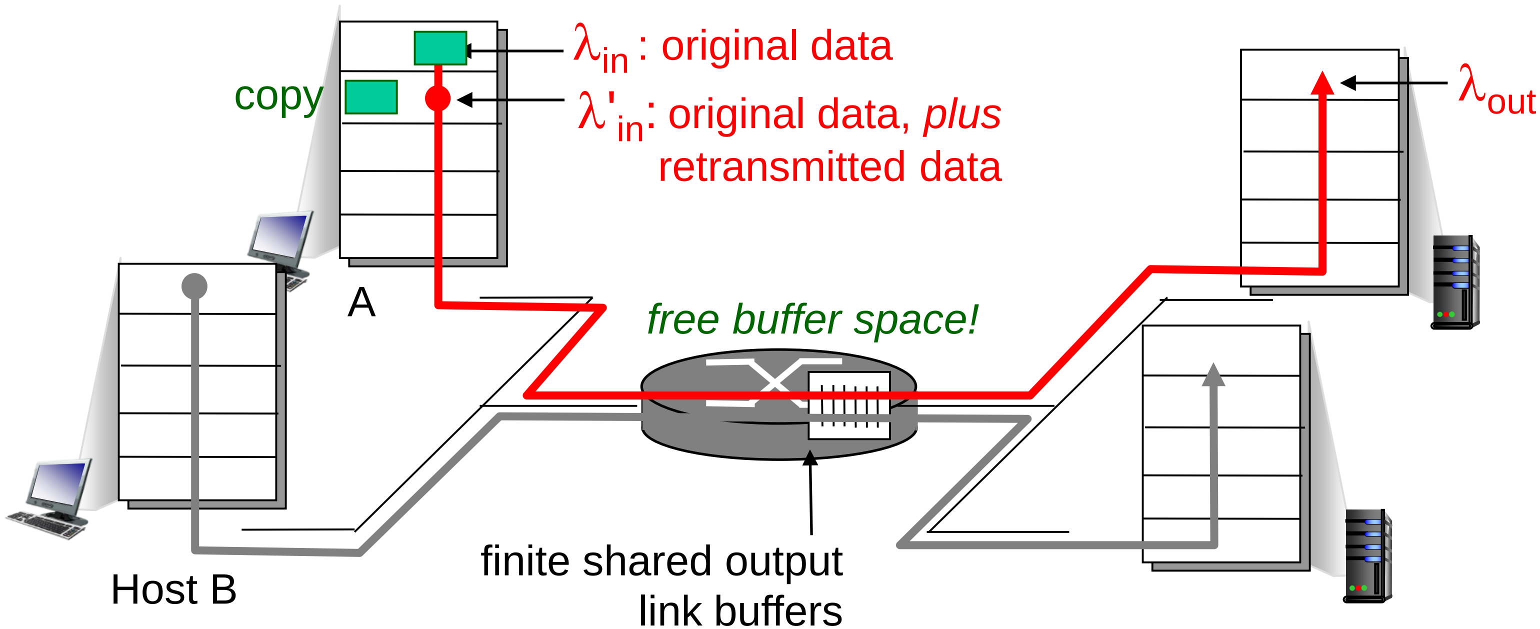


Causes/Costs of Congestion: Scenario 2

idealization: perfect knowledge

- sender sends only when router buffers available

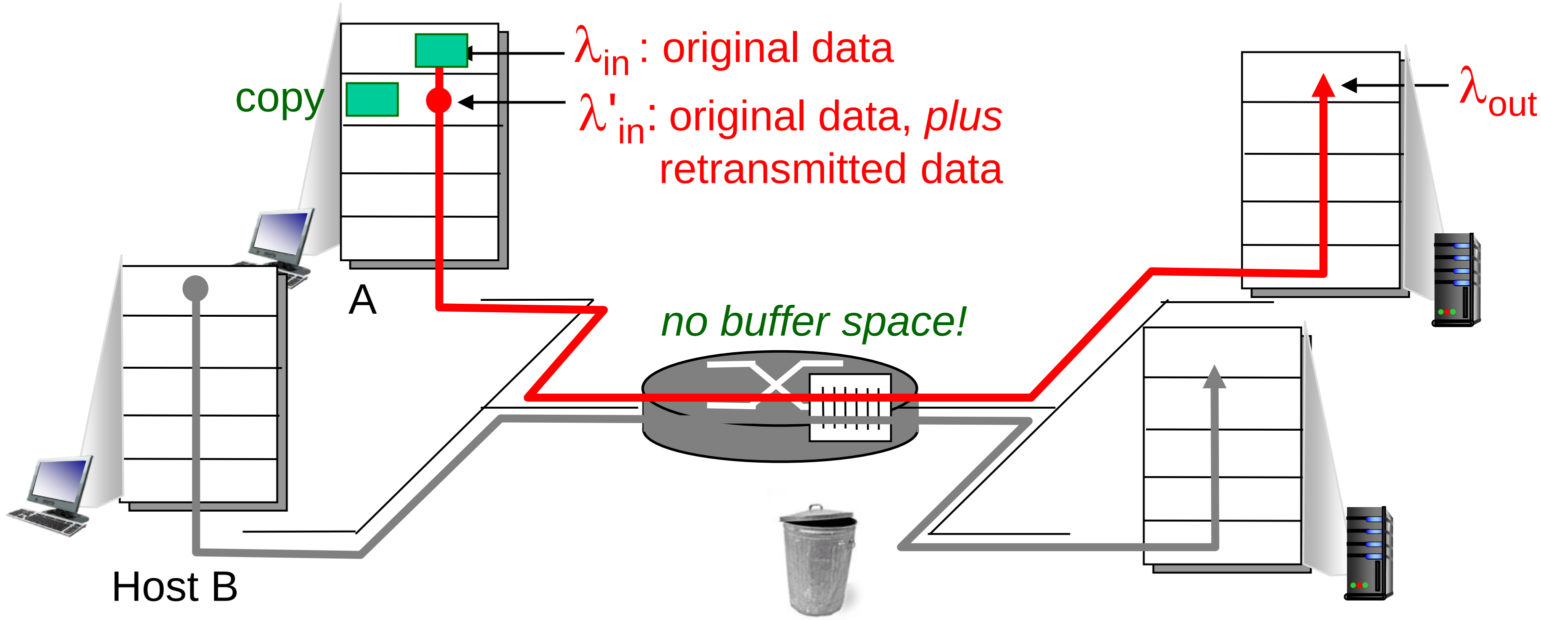




Causes/Costs of Congestion: Scenario 2

Idealization: known loss packets can be lost, dropped at router due to full buffers

- sender only resends if packet *known* to be lost



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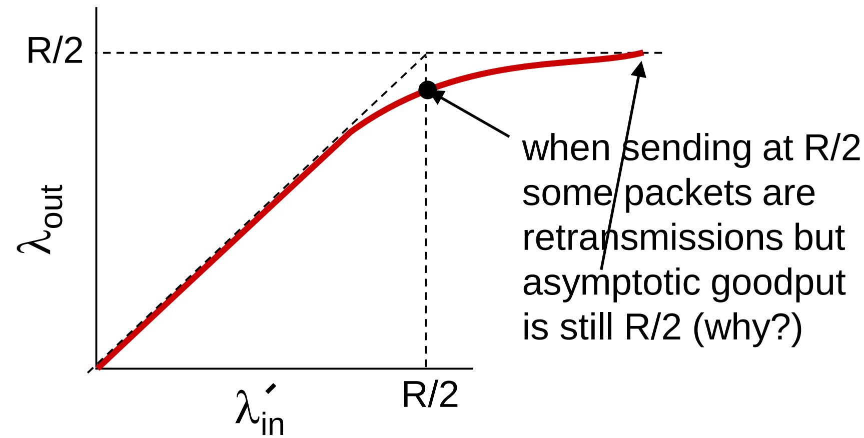
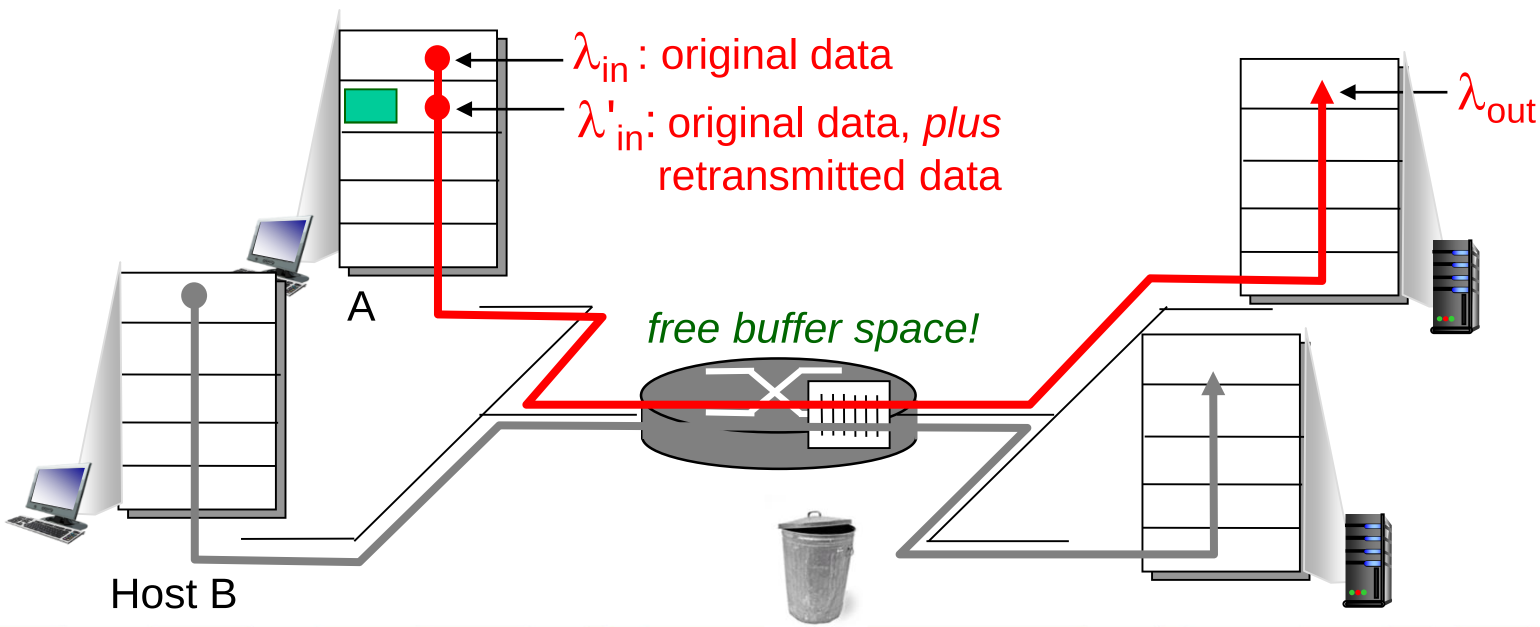
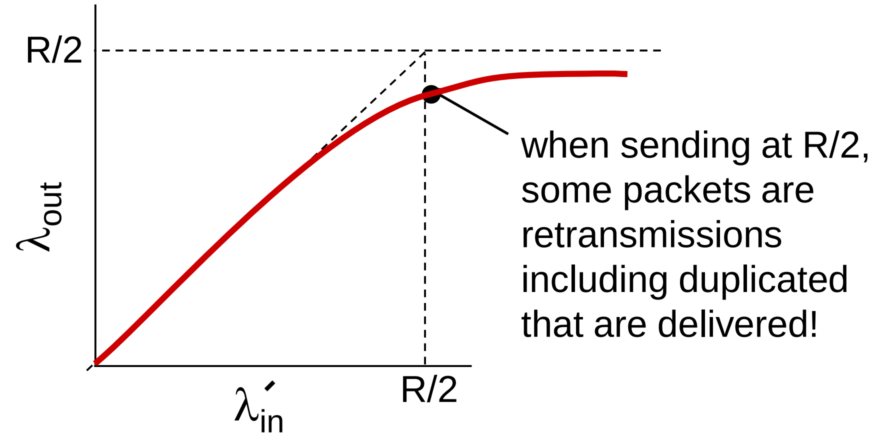


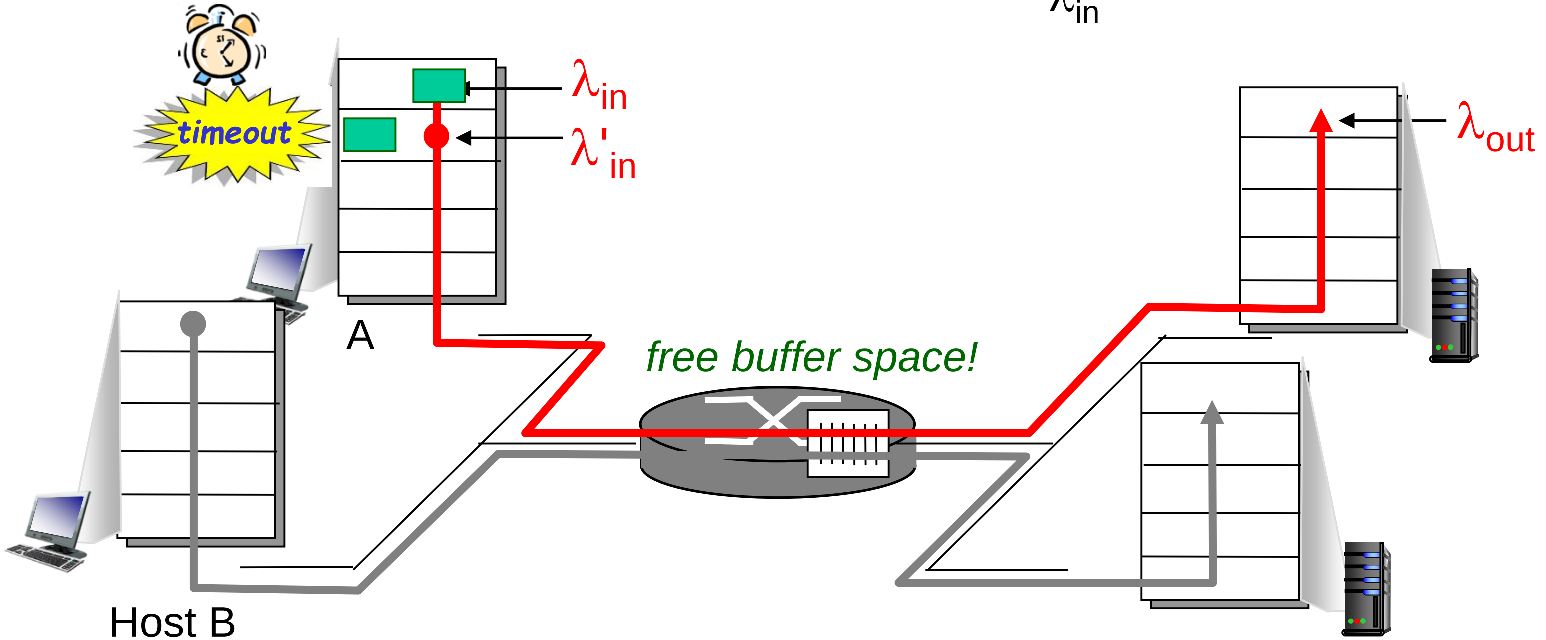


Causes/Costs of Congestion: Scenario 2

Realistic: *duplicates*

- packets can be lost, dropped at router due to full buffers
- sender times out prematurely, sending *two* copies, both of which are delivered





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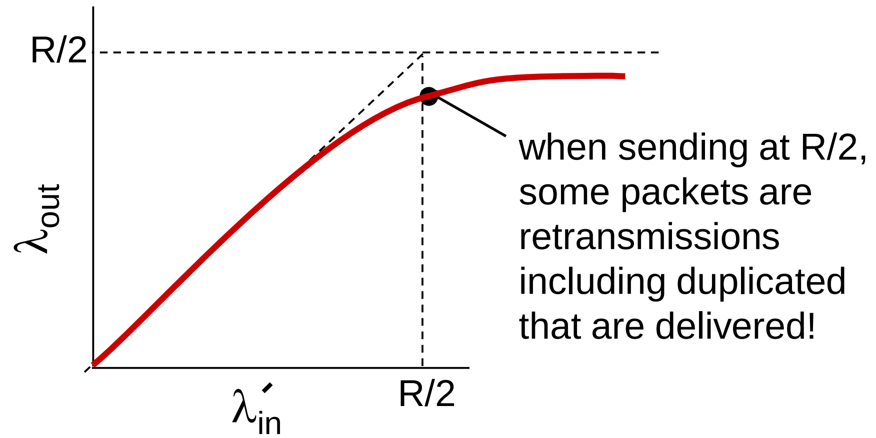


"costs" of congestion:

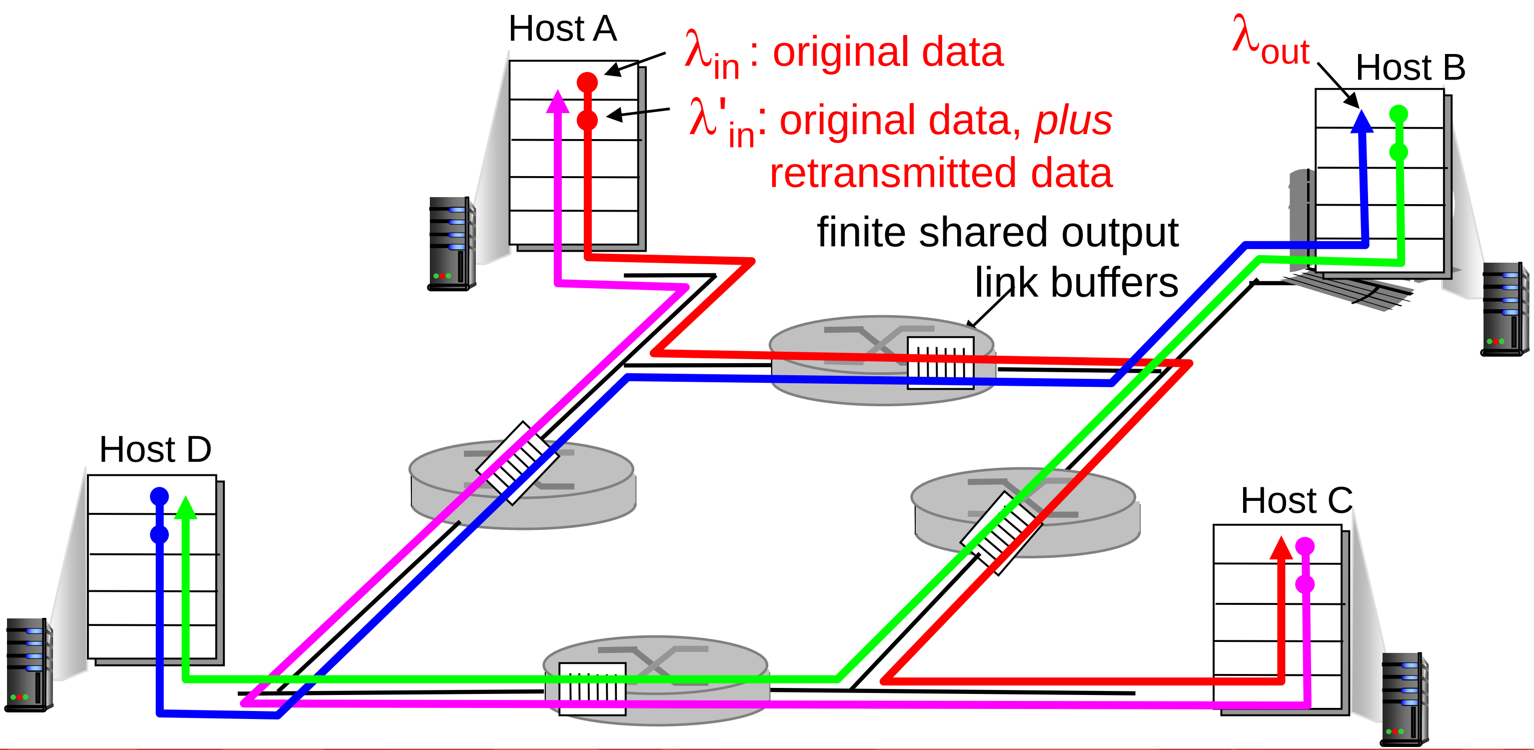
- more work (retrans) for given "goodput"
- unneeded retransmissions: link carries multiple copies of pkt
  - decreasing goodput

Causes/Costs of Congestion: Scenario 3

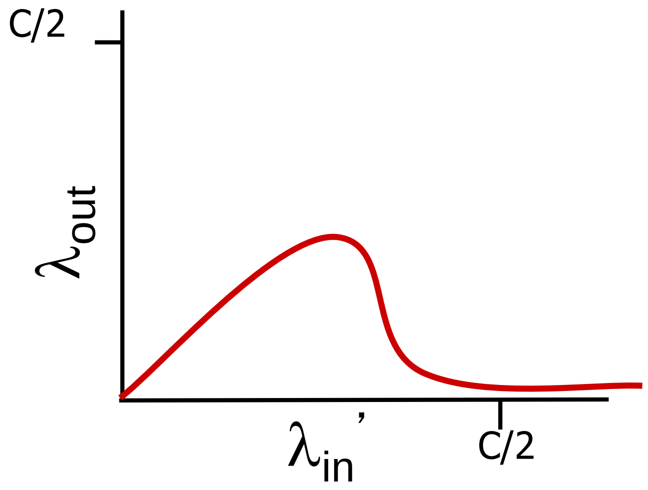
- four senders
- multihop paths
- timeout/retransmit

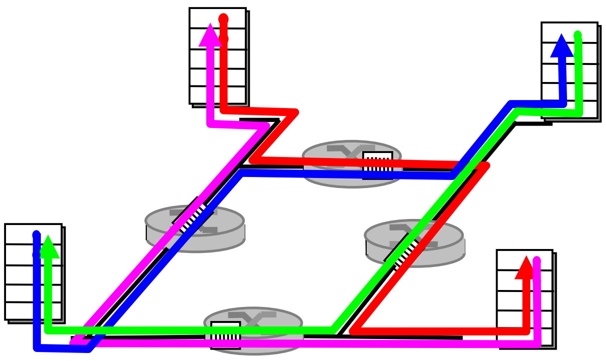
Q: what happens as λ_{in} and λ_{in}' increase ?

A: as red λ_{in}' increases, all arriving blue pkts at upper queue are dropped, blue throughput $\rightarrow$ 0



Causes/Costs of Congestion: Scenario 3





another "cost" of congestion:

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

Roadmap

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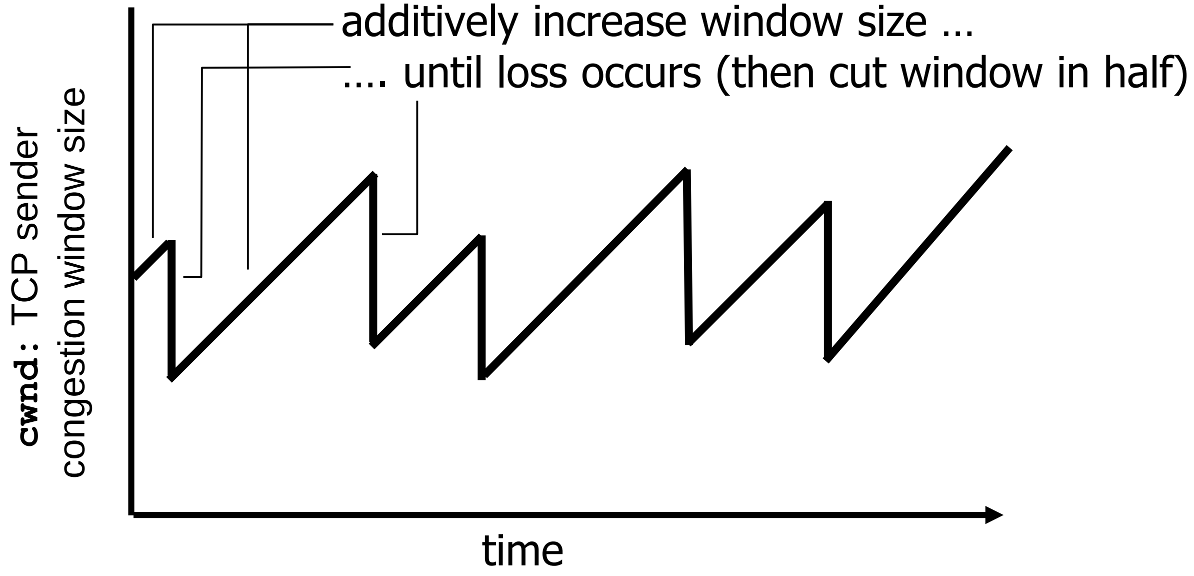
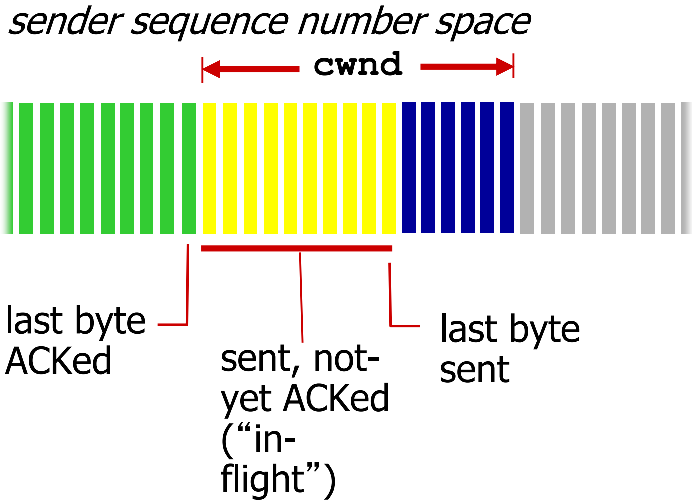
TCP Congestion Control: Additive Increase/ Multiplicative Decrease

- *approach:* sender increases transmission rate (window size), probing for usable bandwidth, until loss occurs
  - *additive increase:* increase **cwnd** by 1 MSS every RTT until loss detected
  - *multiplicative decrease:* cut **cwnd** in half after loss

AIMD saw tooth behavior: probing for bandwidth



TCP Congestion Control: Details



- sender limits transmission:

$$\texttt{LastByteSent-LastByteAcked} \leq \texttt{cwnd}$$

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

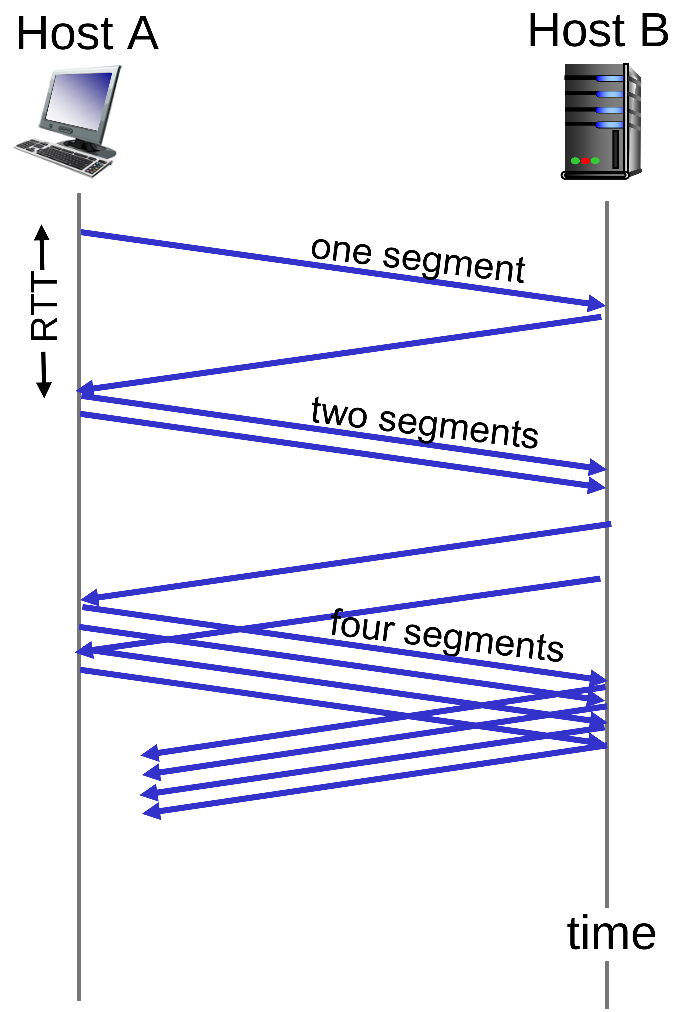
TCP sending rate:

- *roughly:* send cwnd bytes, wait RTT for ACKS, then send more bytes

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

TCP Slow Start

- when connection begins, increase rate exponentially until first loss event:
  - initially **cwnd** = 1 MSS
  - double **cwnd** every RTT
  - done by incrementing **cwnd** for every ACK received
- *summary:* initial rate is slow but ramps up exponentially fast



DISCOVER · ACHIEVE · BELONG

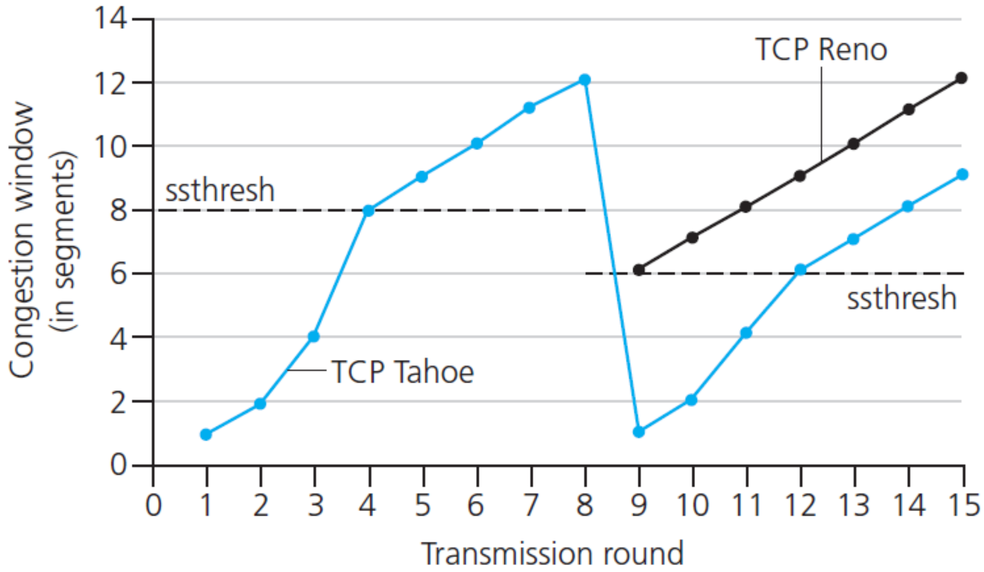
TCP: Detecting, Reacting To Loss

- loss indicated by timeout:
  - **`cwnd`** set to 1 MSS;
  - window then grows exponentially (as in slow start) to threshold, then grows linearly
- loss indicated by 3 duplicate ACKs: TCP RENO
  - dup ACKs indicate network capable of delivering some segments
  - **`cwnd`** is cut in half window then grows linearly
- TCP Tahoe always sets **`cwnd`** to 1 (timeout or 3 duplicate acks)

TCP: Switching From Slow Start To CA

Q: when should the exponential increase switch to linear?

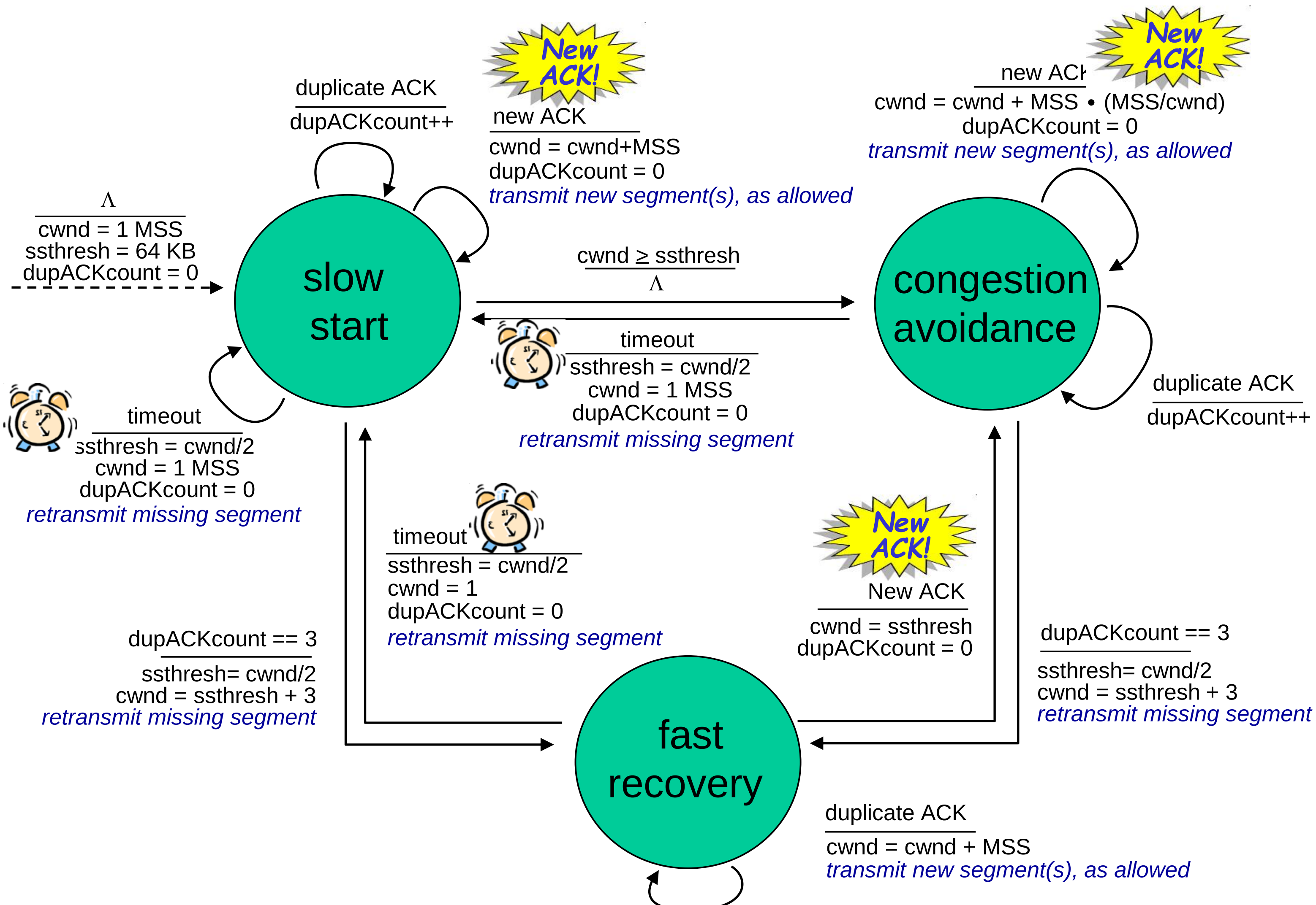
A: when `cwnd` gets to 1/2 of its value before timeout.



Implementation:

- variable `ssthresh`
- on loss event, `ssthresh` is set to 1/2 of `cwnd` just before loss event

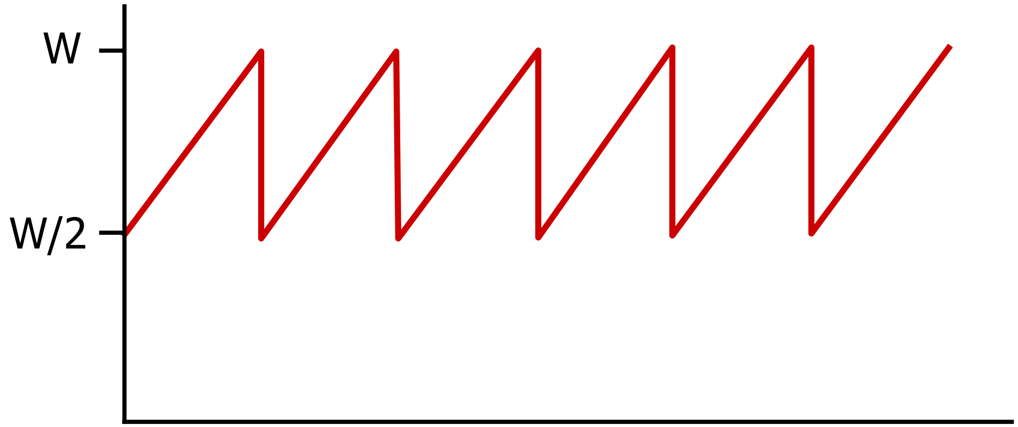
Summary: TCP Congestion Control



TCP Throughput

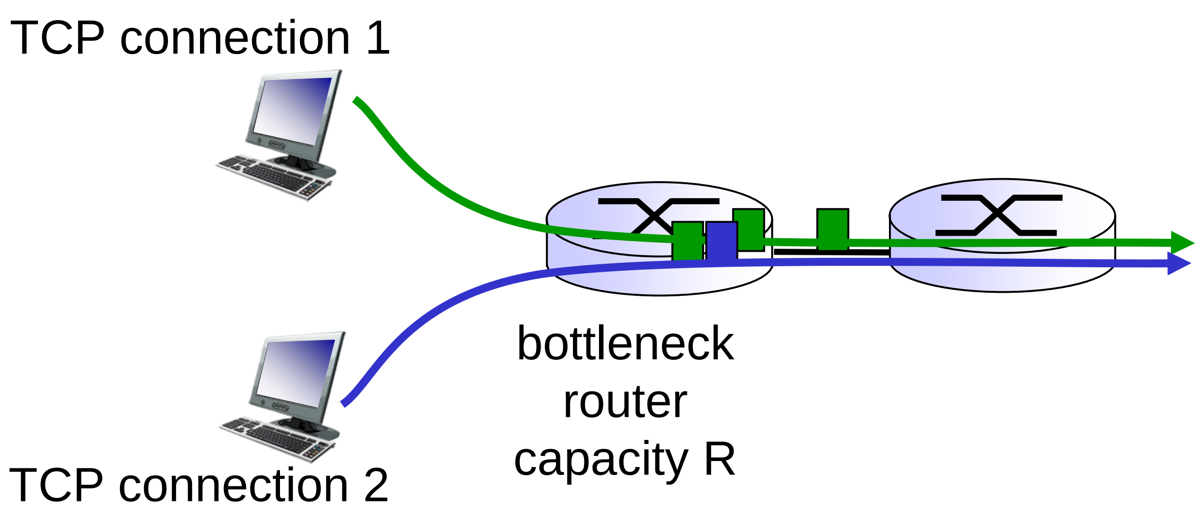
- avg. TCP thruput as function of window size, RTT?
  - ignore slow start, assume always data to send
- W: window size (measured in bytes) where loss occurs
  - avg. window size (# in-flight bytes) is ¾ W
  - avg. thruput is 3/4W per RTT

$$\text{avg TCP thruput} = \frac{3}{4}\frac{W}{RTT} \text{ bytes/sec}$$



TCP Fairness

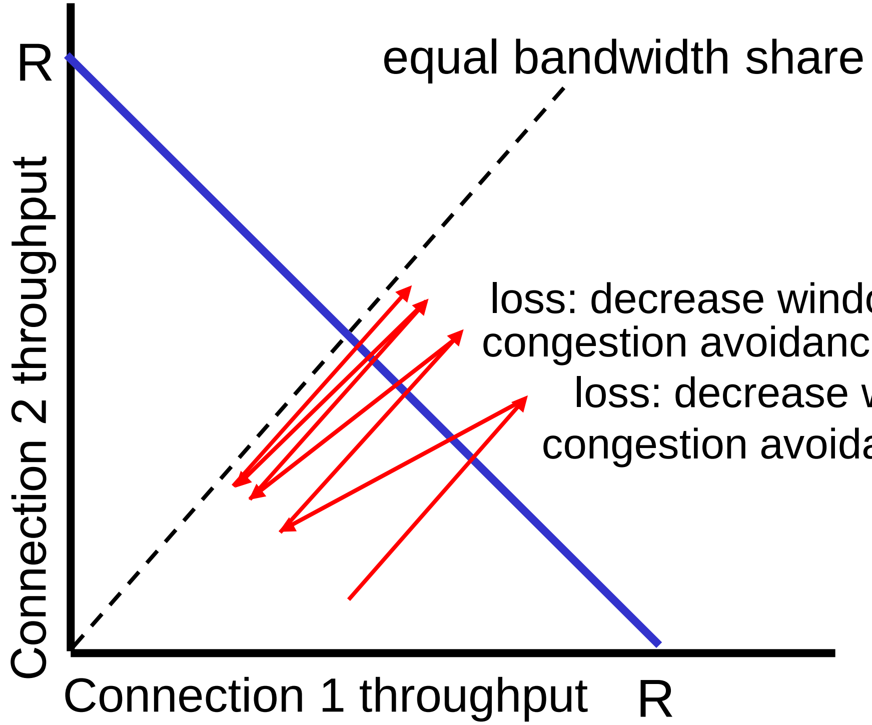
fairness goal: if K TCP sessions share same bottleneck link of bandwidth R, each should have average rate of R/K



Why Is TCP Fair?

two competing sessions:

- additive increase gives slope of 1, as throughout 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:
  - send audio/video at constant rate, tolerate packet loss

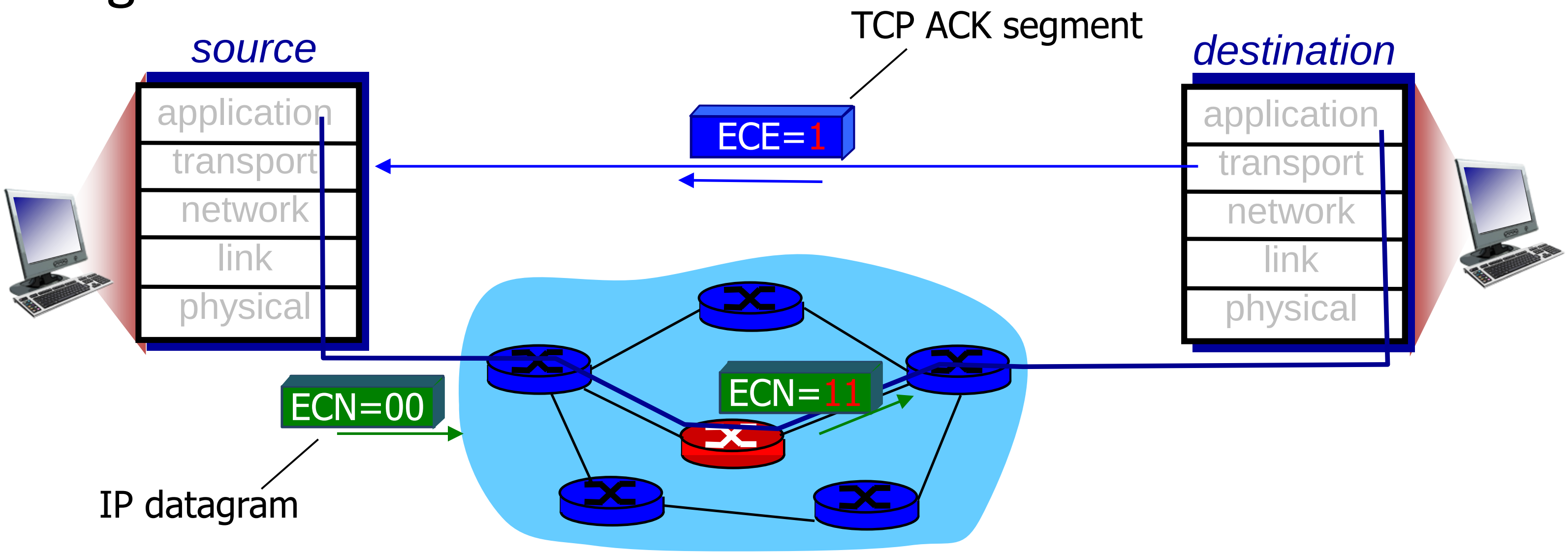
Fairness, parallel TCP connections

- application can open multiple parallel connections between two hosts
- web browsers do this
- e.g., link of rate R with 9 existing connections:
  - new app asks for 1 TCP, gets rate R/10
  - new app asks for 11 TCPs, gets R/2

Explicit Congestion Notification (ECN)

network-assisted congestion control:

- two bits in IP header (ToS field) marked *by network router* to indicate congestion
- congestion indication carried to receiving host
- receiver (seeing congestion indication in IP datagram)) sets ECE bit on receiver-to-sender ACK segment to notify sender of congestion



Summary

- Principles behind transport layer services:
  - multiplexing, demultiplexing
  - reliable data transfer
  - flow control
  - congestion control
- Instantiation, implementation in the Internet
  - UDP
  - TCP

next:

- Leaving the network "edge" (application, transport layers)
- Into the network "core"

Questions?