

Chapter 3

Transport Layer

Part 5/5

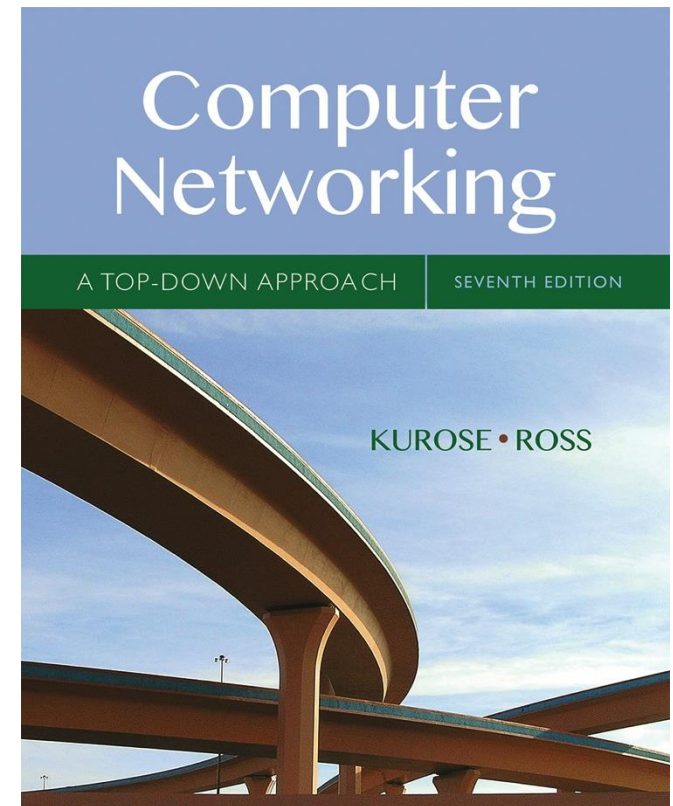
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Computer Networking: A Top Down Approach

7th edition

Jim Kurose, Keith Ross

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Chapter 3 outline

3.1 transport-layer services

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

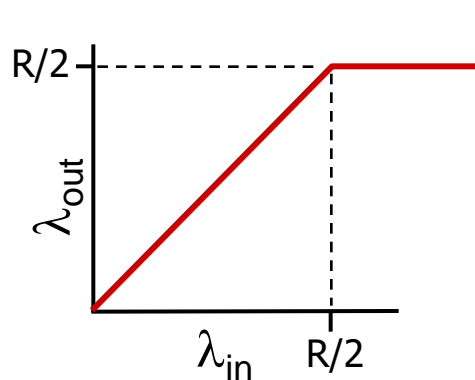
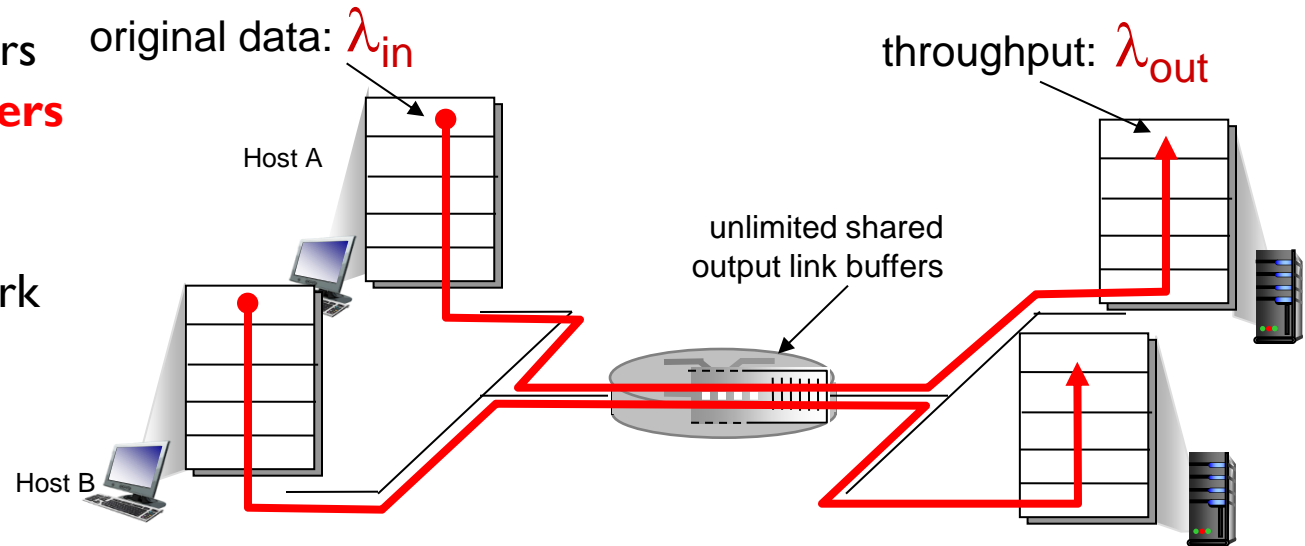
Principles of congestion control

congestion:

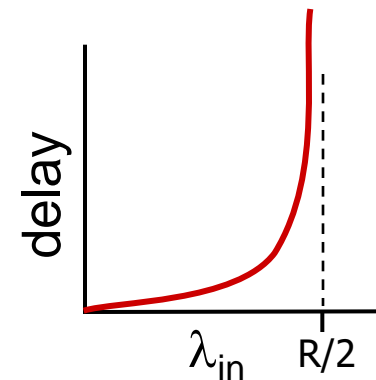
- informally: “too many sources sending too much data too fast for *network* to handle”
- different from flow control!
- Signs of Congestion:
 - lost packets (buffer overflow at routers)
 - long delays (queueing in router buffers)
- a top-10 problem!

Causes/costs of congestion: scenario (ideal 1)

- two senders, two receivers
- one router, **infinite buffers**
- output link capacity: R
- no retransmission
- Cost of congested network
 - Large queuing delay



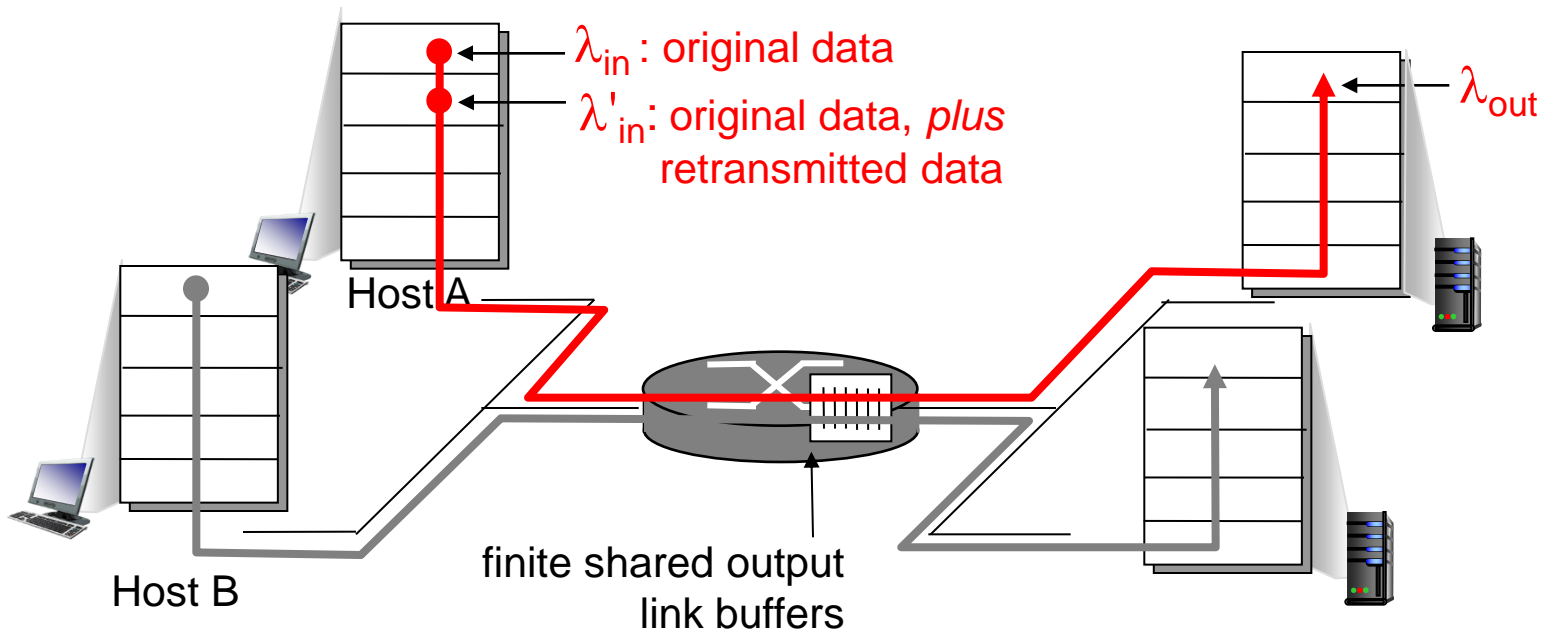
- maximum per-connection throughput: $R/2$



- ❖ large delays as arrival rate, λ_{in} , approaches capacity

Causes/costs of congestion: scenario 2

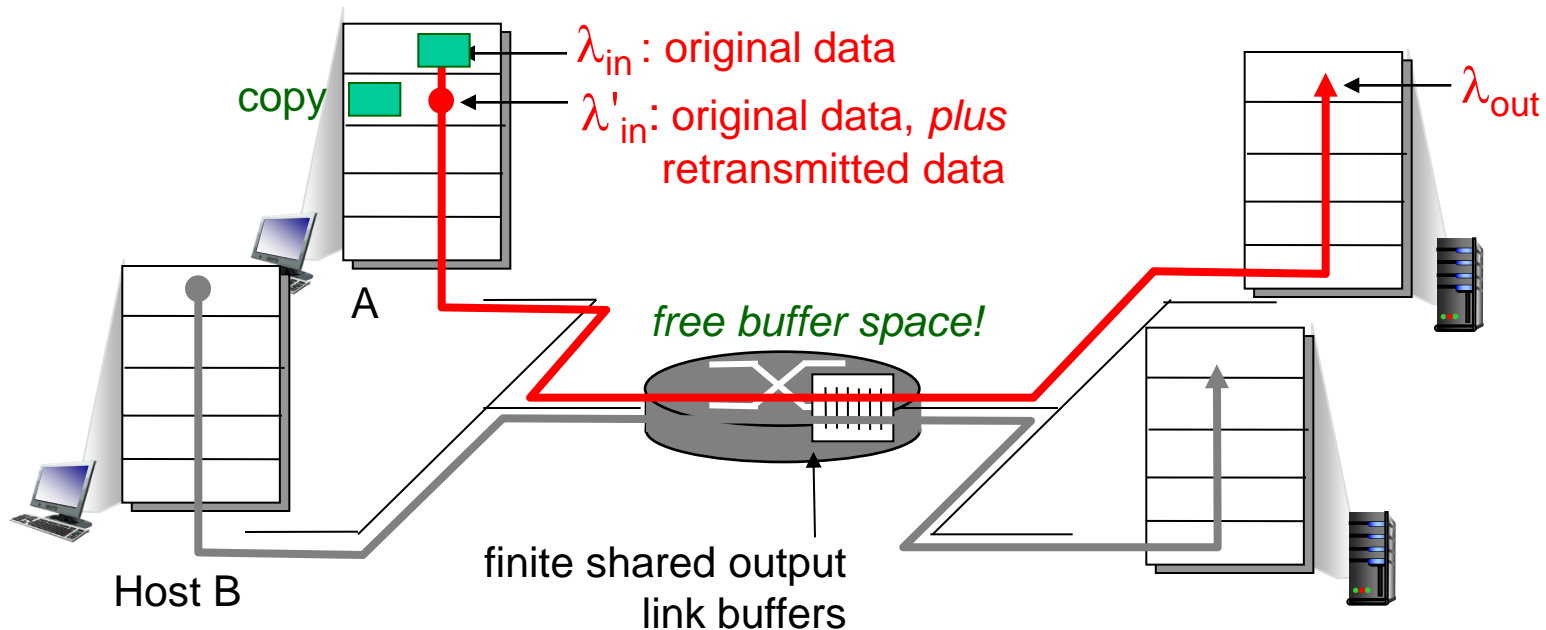
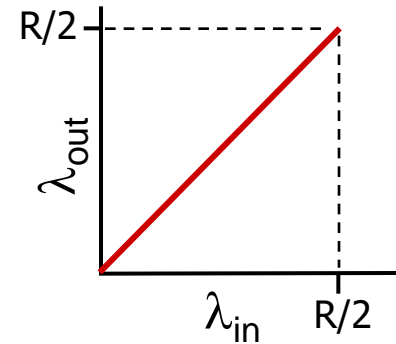
- Two senders, one router, *finite* buffers
- sender retransmission of timed-out packet
 - application-layer input = application-layer output: $\lambda_{in} = \lambda_{out}$
 - transport-layer input includes *retransmissions* : $\lambda'_{in} \geq \lambda_{in}$



Causes/costs of congestion: scenario 2(a)

idealization: perfect knowledge

- sender sends only when router buffers available
- No retransmission
- Packet loss is assumed never to occur

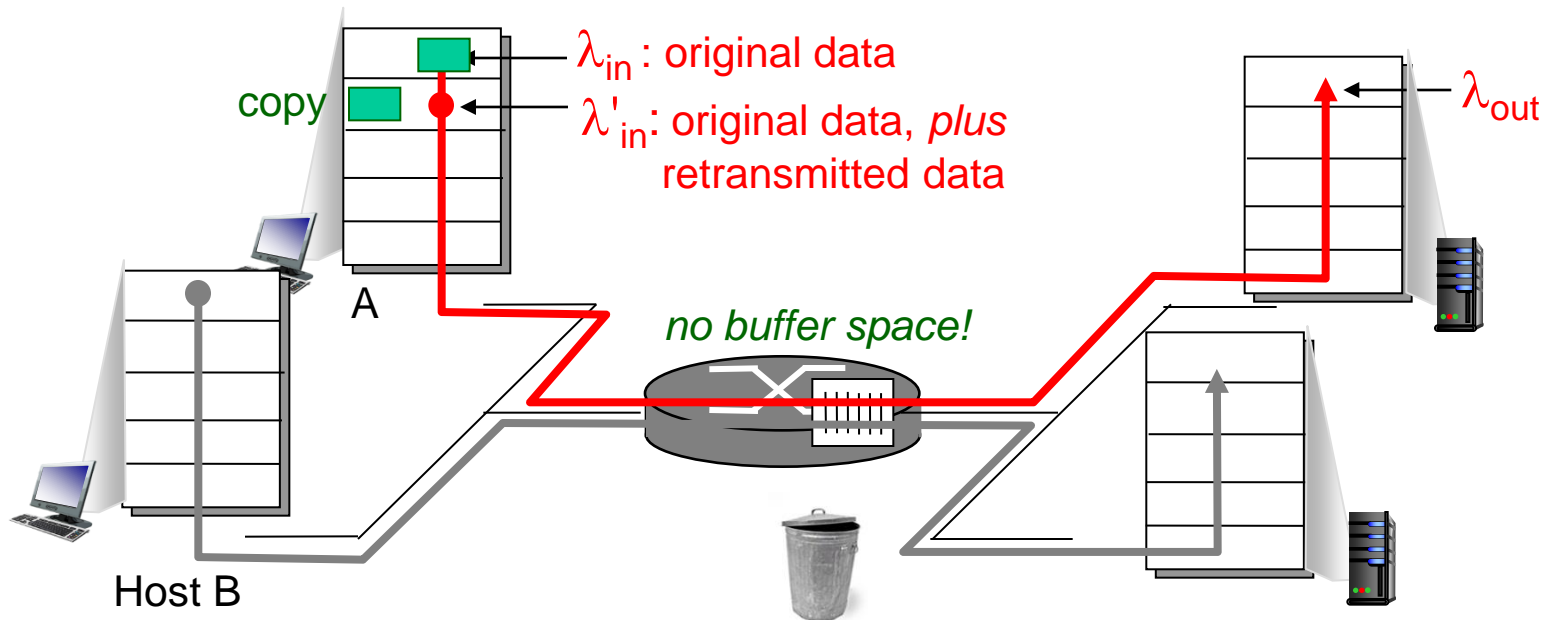


Causes/costs of congestion: scenario 2(b)

Idealization: known loss

packets can be lost, dropped at router due to full buffers

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

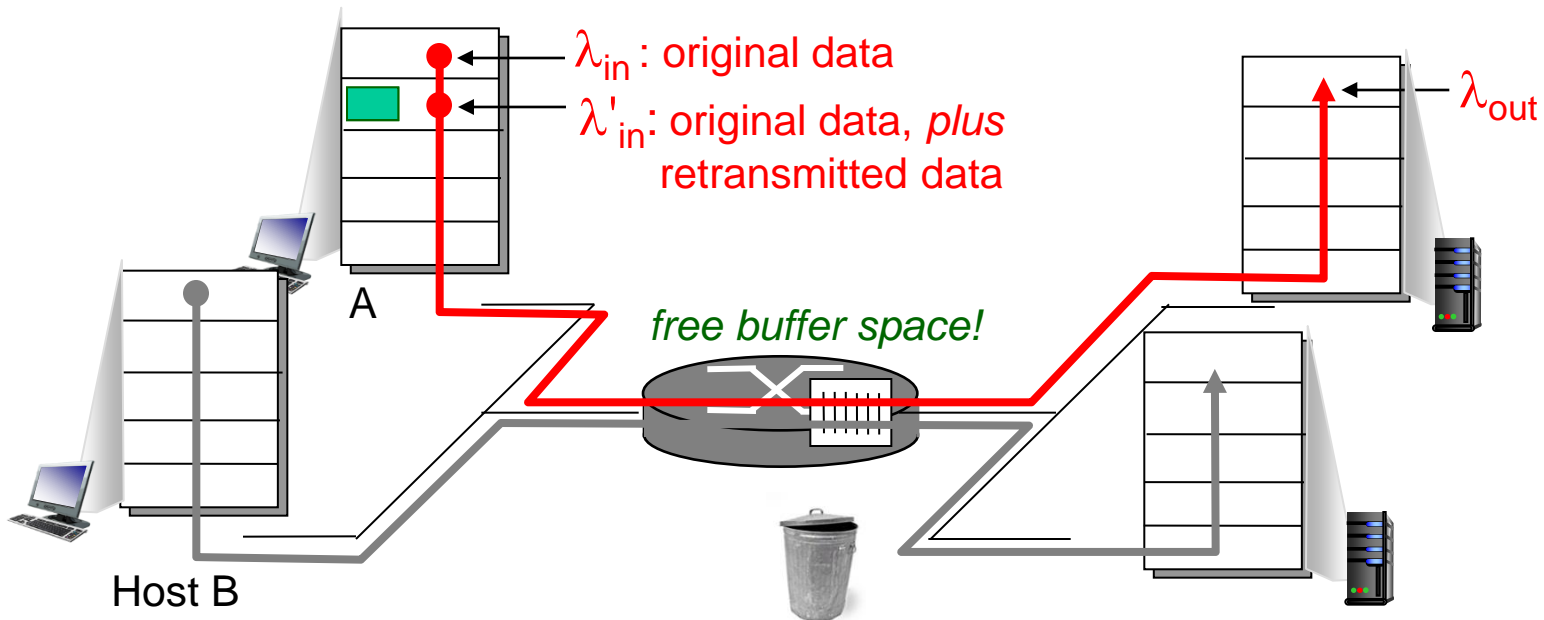
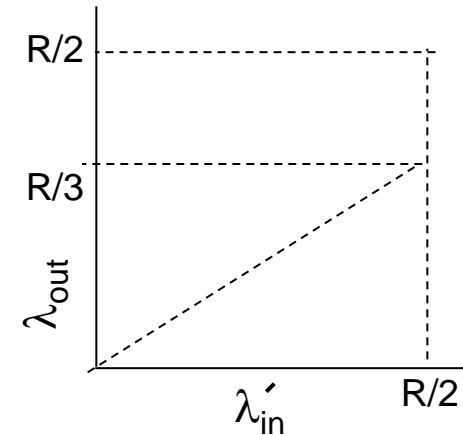


Causes/costs of congestion: scenario 2(b)

Idealization: known loss

packets can be lost, dropped at router due to full buffers

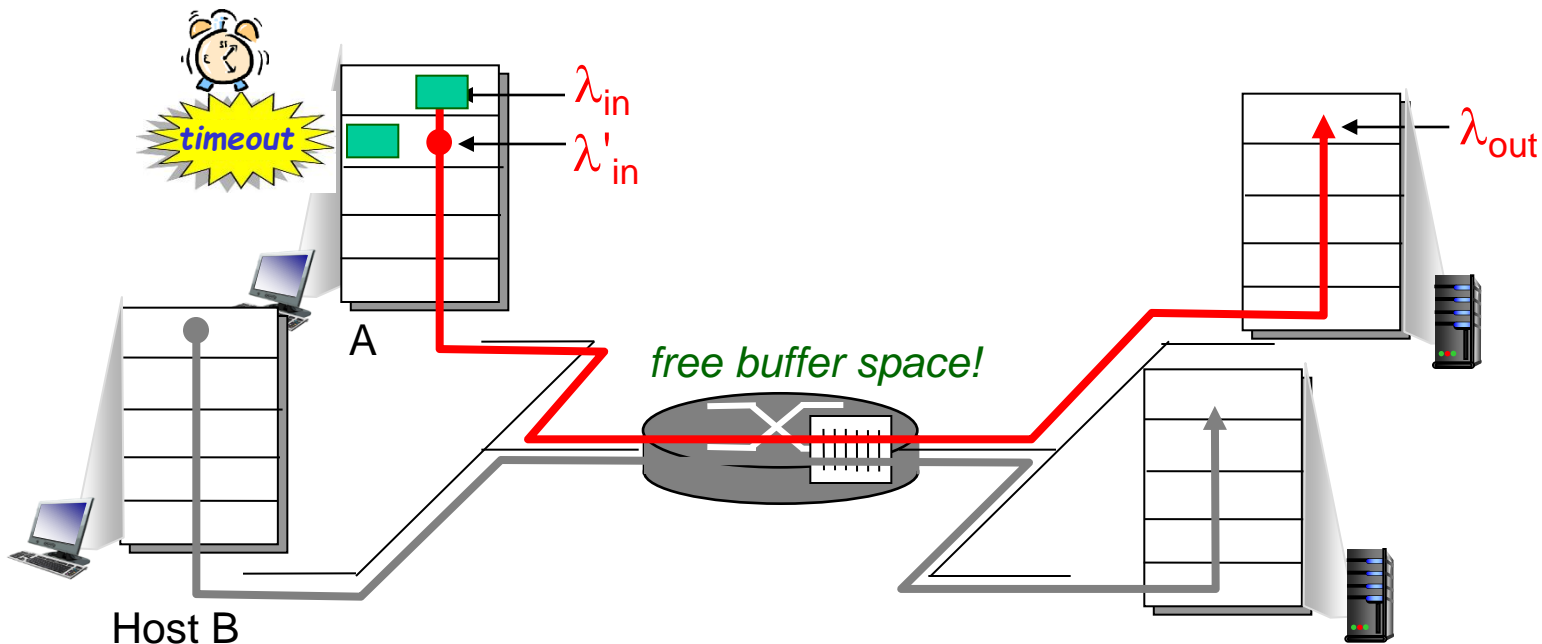
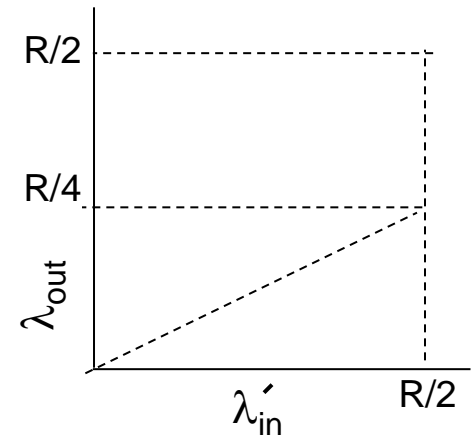
- sender only resends if packet *known* to be lost
- Cost of congestion:
 - Retransmission to compensate lost pkts



Causes/costs of congestion: scenario 2(c)

Realistic: *duplicates*

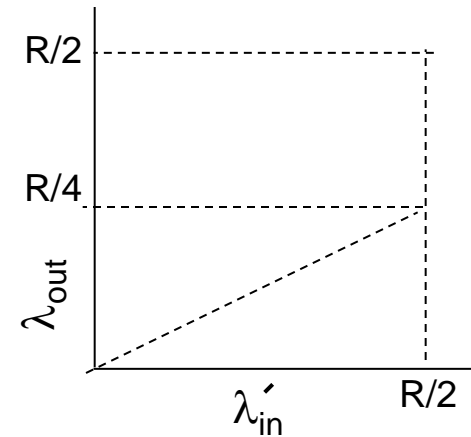
- packets can be lost, dropped at router due to full buffers
- sender times out prematurely, sending *two* copies, both of which are delivered



Causes/costs of congestion: scenario 2(c)

Realistic: duplicates

- packets can be lost, dropped at router due to full buffers
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“costs” of congestion:

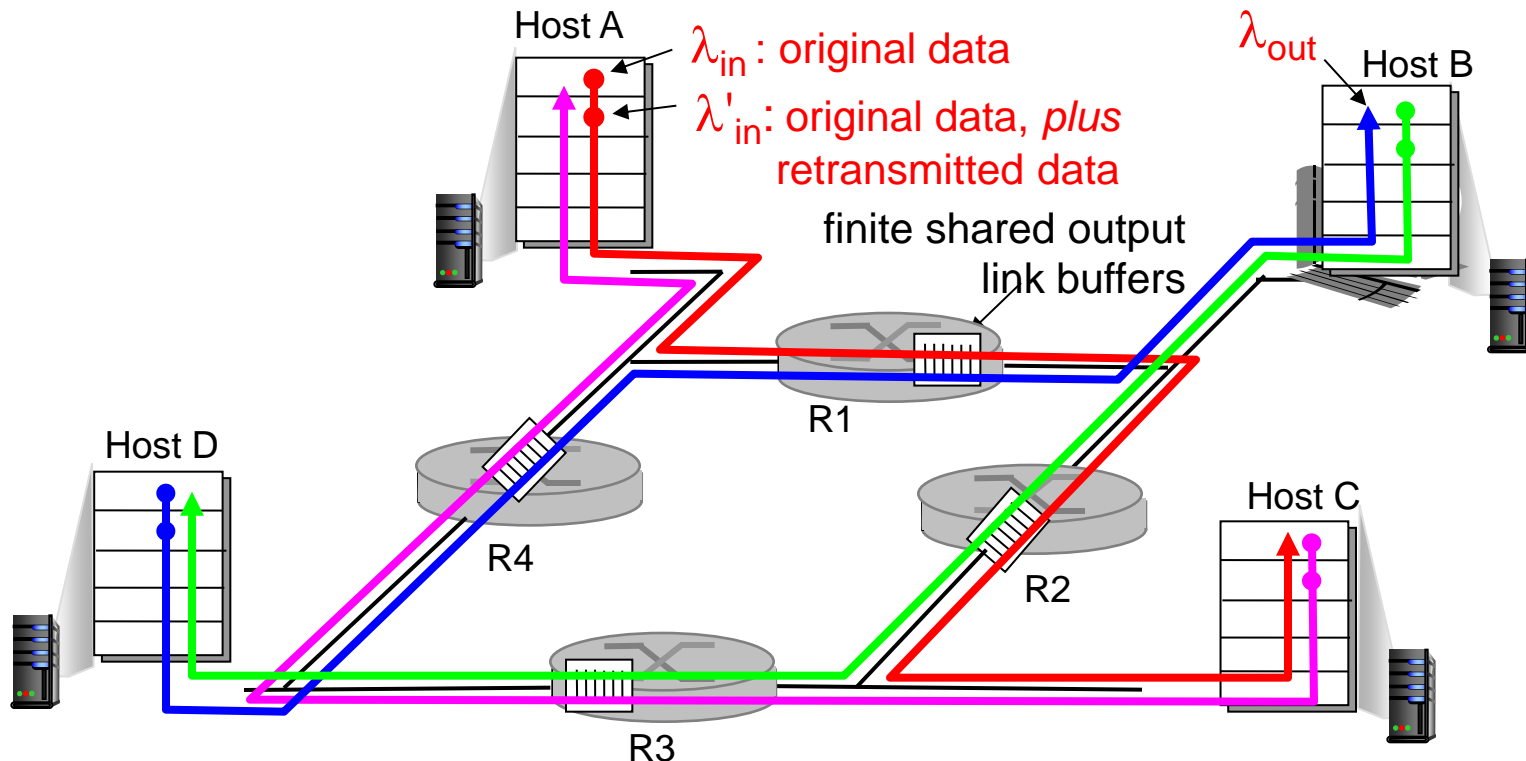
- more work (retransmissions) for given “goodput”
- unneeded retransmissions: link carries multiple copies of pkt
 - decreasing goodput

Causes/costs of congestion: scenario 3

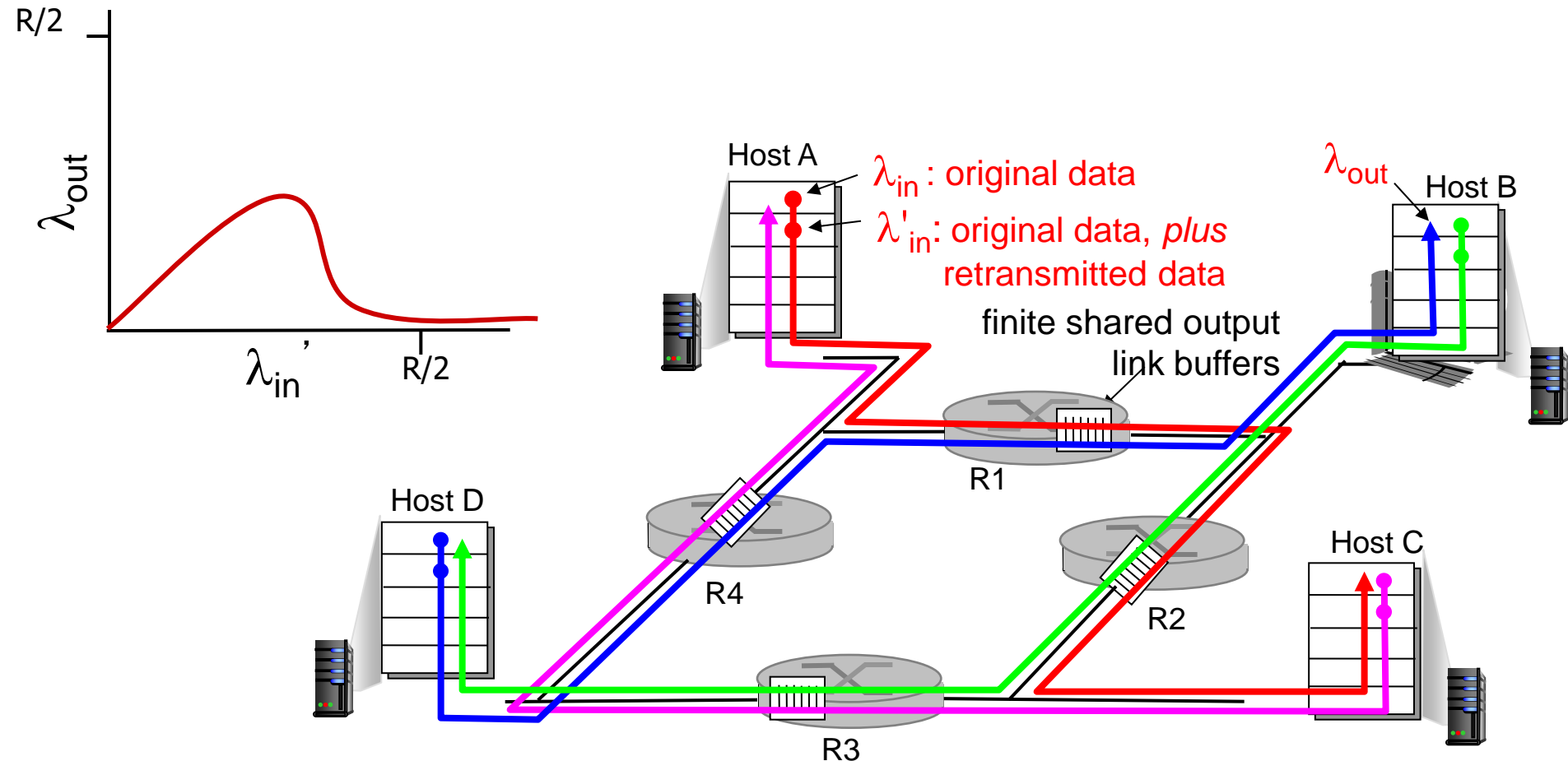
- four senders, four routers,
- Finite buffers
- multihop paths
- Shared path between **A-C** and **B-D**
- timeout/retransmit

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

A: as B-D (green) λ'_{in} increases, all arriving A-C pkts (red) at R2 are dropped, red 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!

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
 - explicit rate sender should send at

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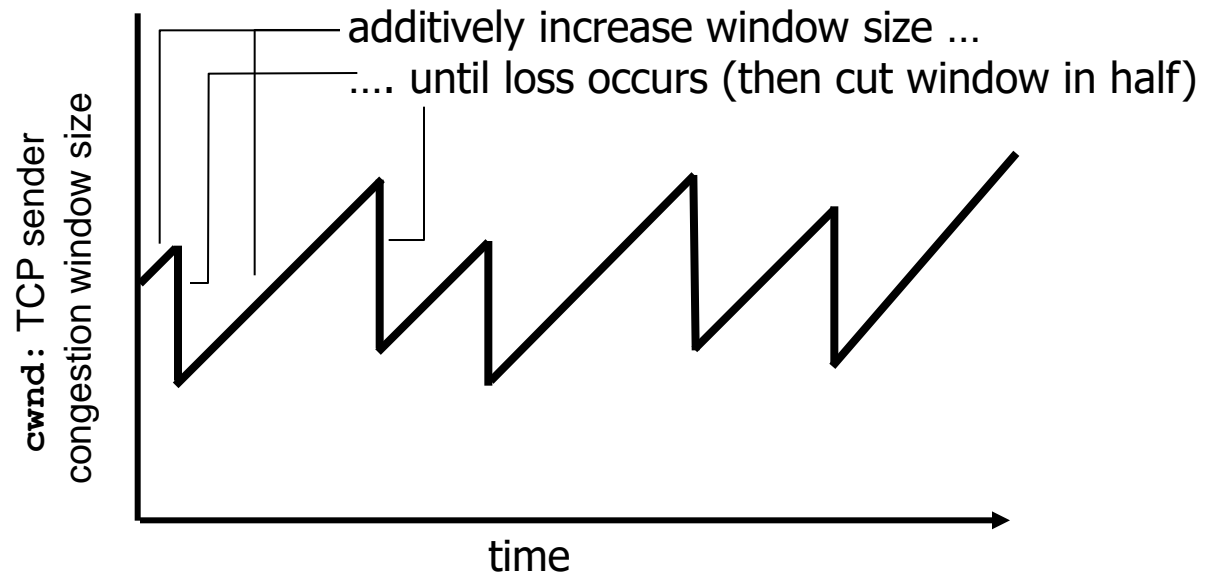
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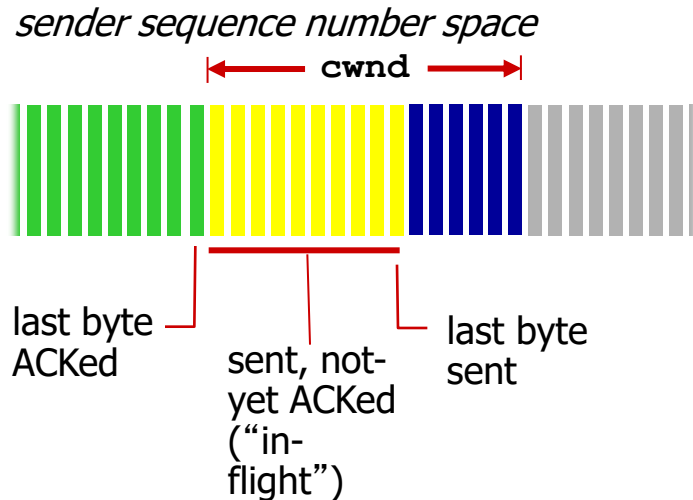
TCP congestion control: additive increase multiplicative decrease

- *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:

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

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

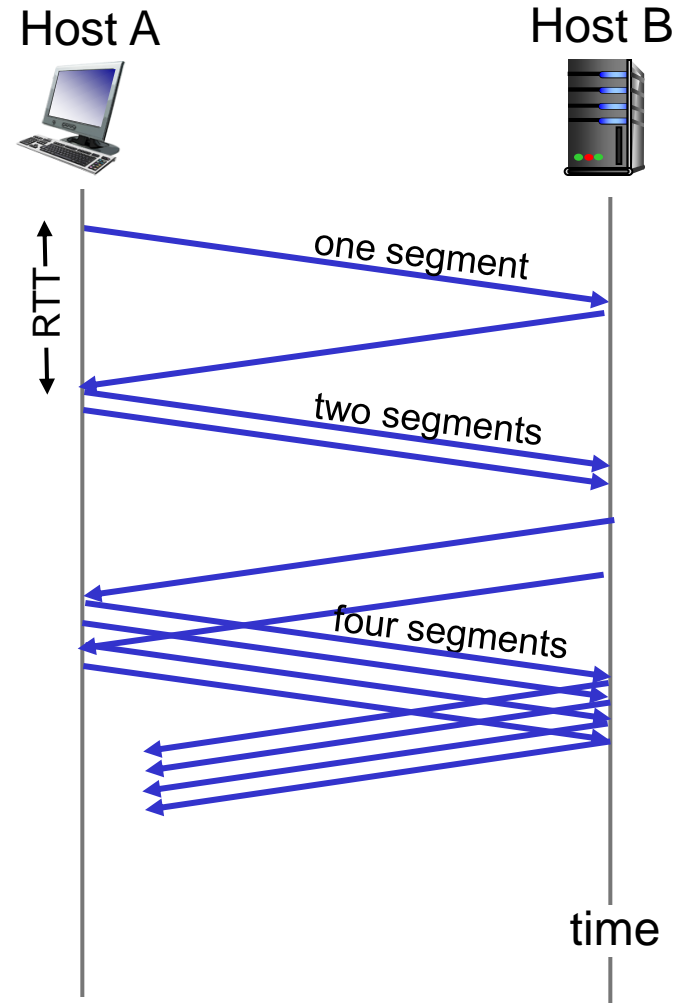
TCP sending rate:

- *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



TCP: detecting, reacting to loss

- loss indicated by timeout: **TCP RENO**
 - **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)
 - window then grows exponentially (as in slow start) to threshold, then grows linearly

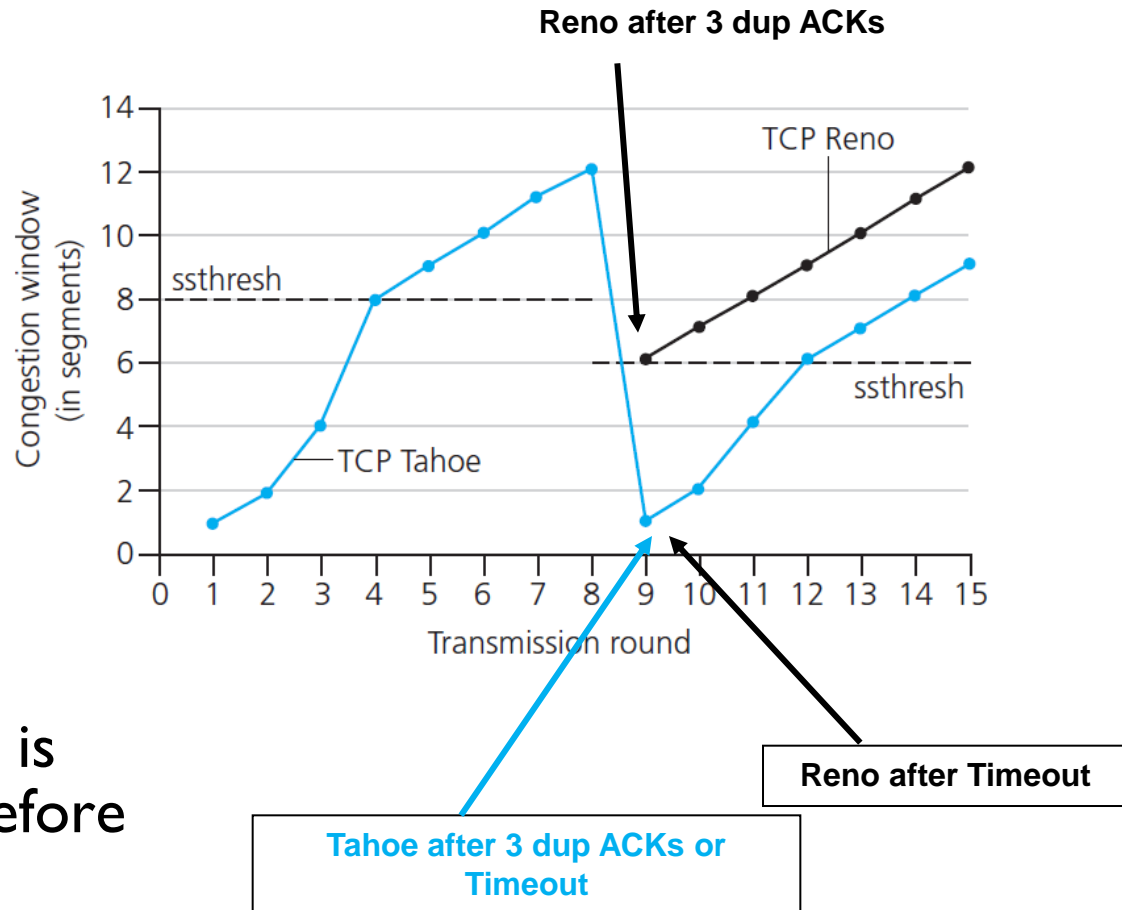
TCP: switching from slow start to CA (Congestion Avoidance)

Q: when should the exponential increase switch to linear?

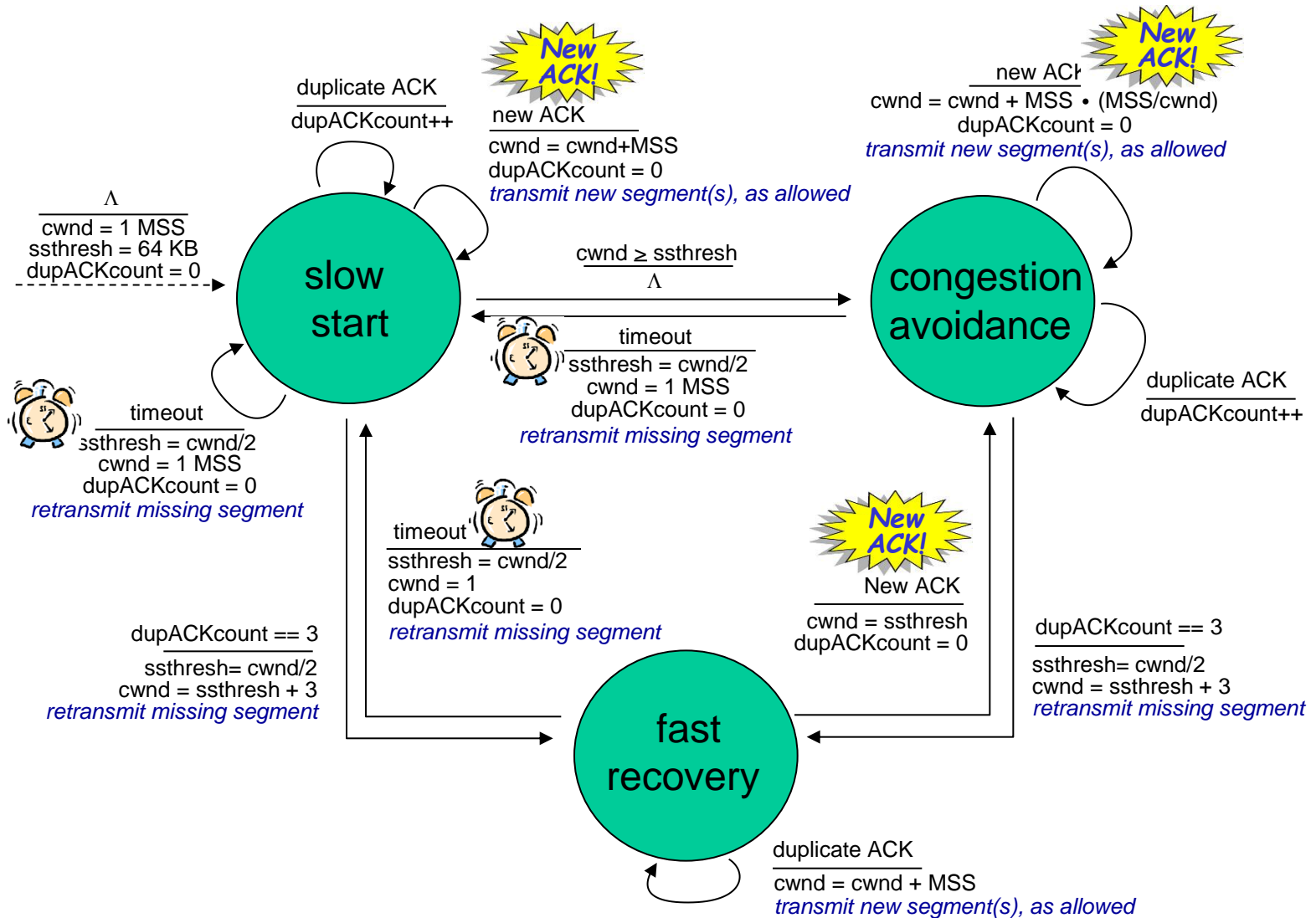
A: when **cwnd** gets to 1/2 of its value before timeout.

Implementation:

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

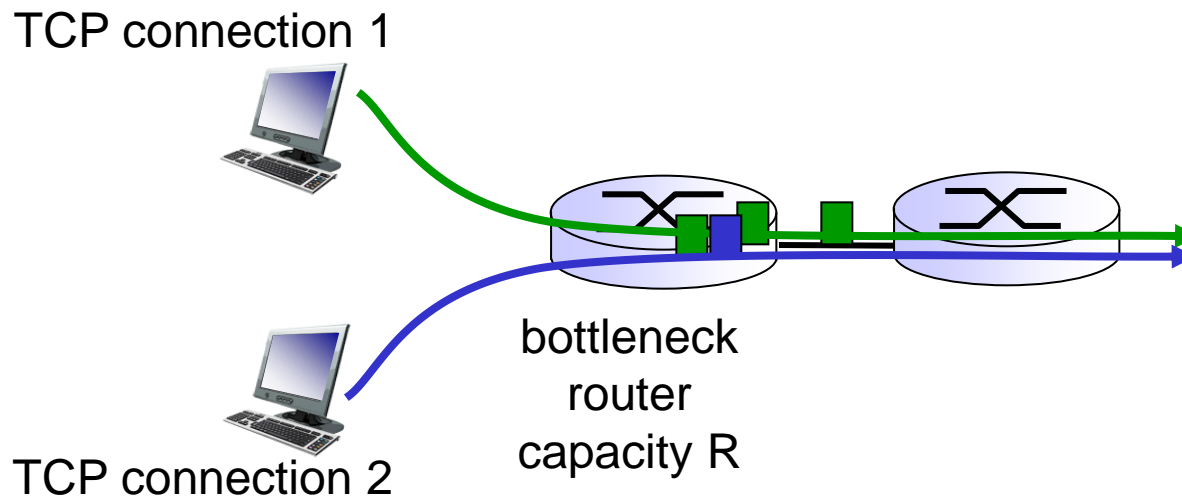


Summary: TCP Congestion Control



TCP Fairness

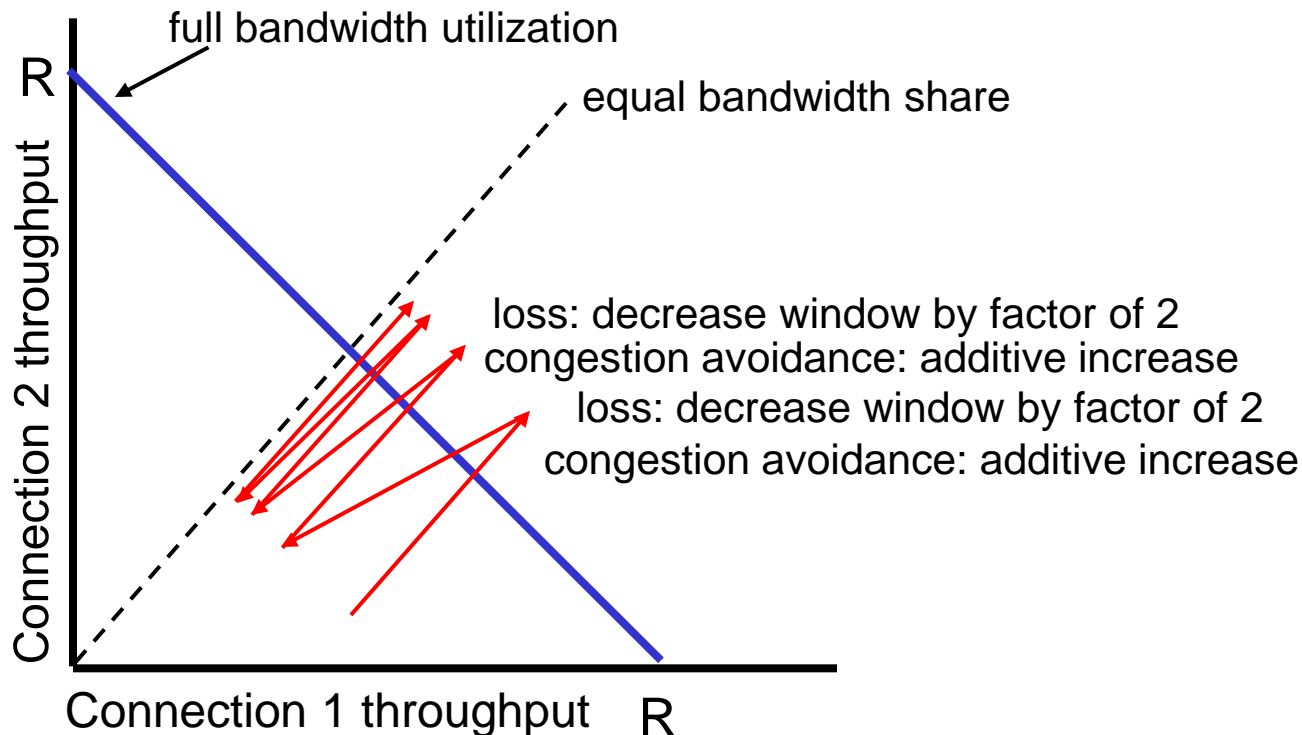
fairness goal: if K TCP sessions share same bottleneck link of bandwidth R , each should have average rate of R/K



Why is TCP fair?

two competing sessions:

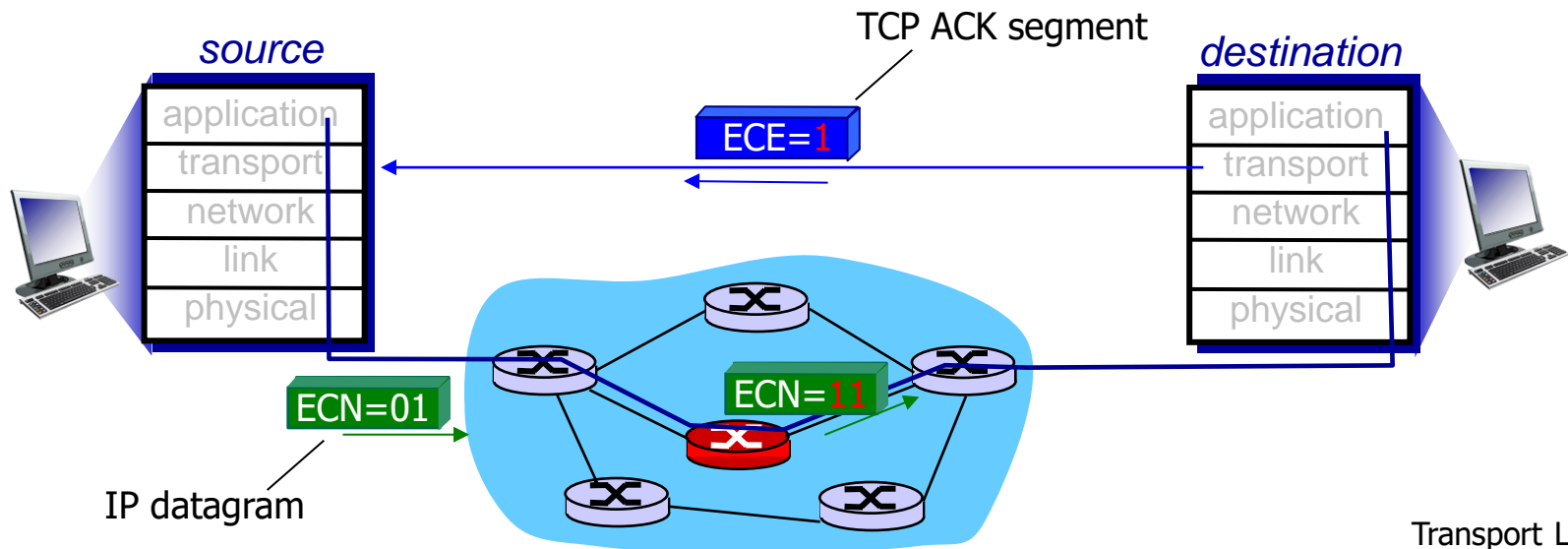
- **additive increase** throughout increases proportionally
- **multiplicative decrease** decreases throughput proportionally
- Both connections have equal RTT and MSS



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 (ECN Echo) bit on receiver-to-sender ACK segment to notify sender of congestion



Chapter 3: 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”
- two network layer chapters:
 - data plane
 - control plane