Chapter 3
Transport Layer

Chapter 3 outline

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

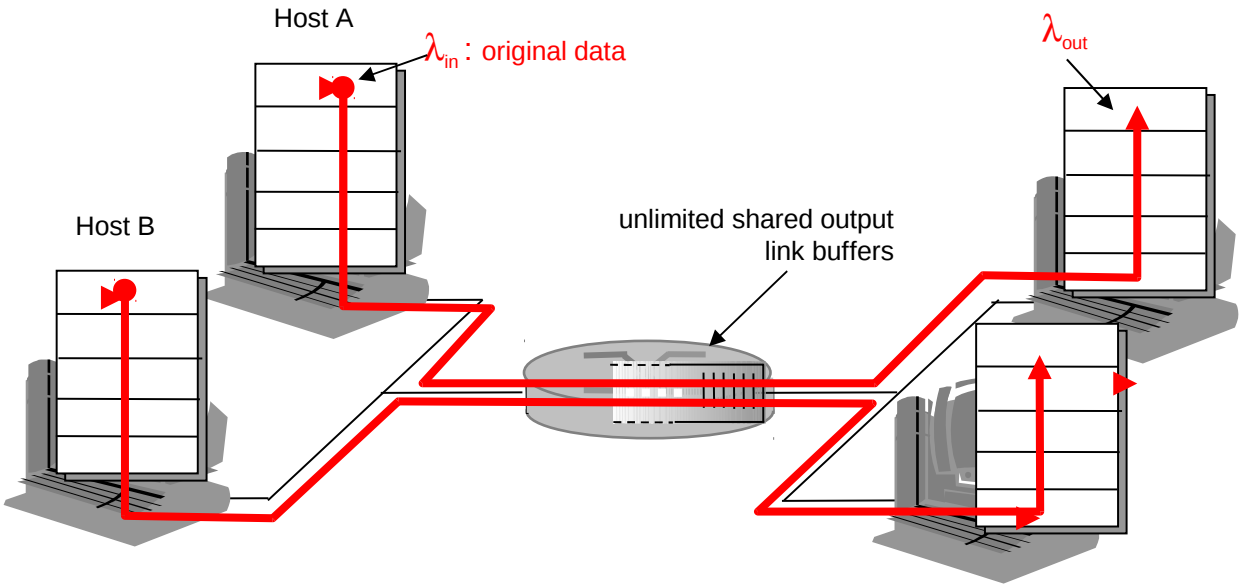
Principles of Congestion Control

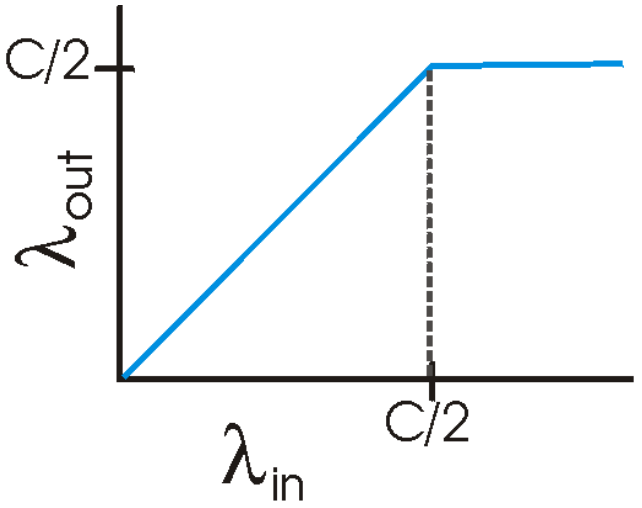
Congestion:

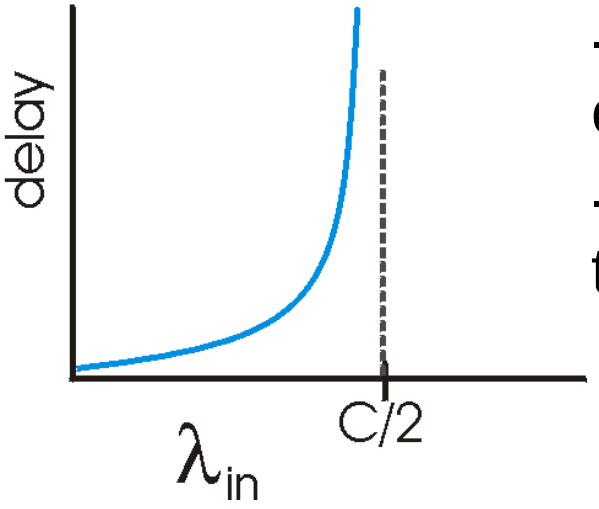
- informally: “too many sources sending too much data too fast for *network* to handle”
- different from flow control!
- manifestations:
  - lost packets (buffer overflow at routers)
  - long delays (queueing in router buffers)
- a top-10 problem!

Causes/costs of congestion: scenario 1

- Two senders, two receivers
- One router, infinite buffers
- no retransmission







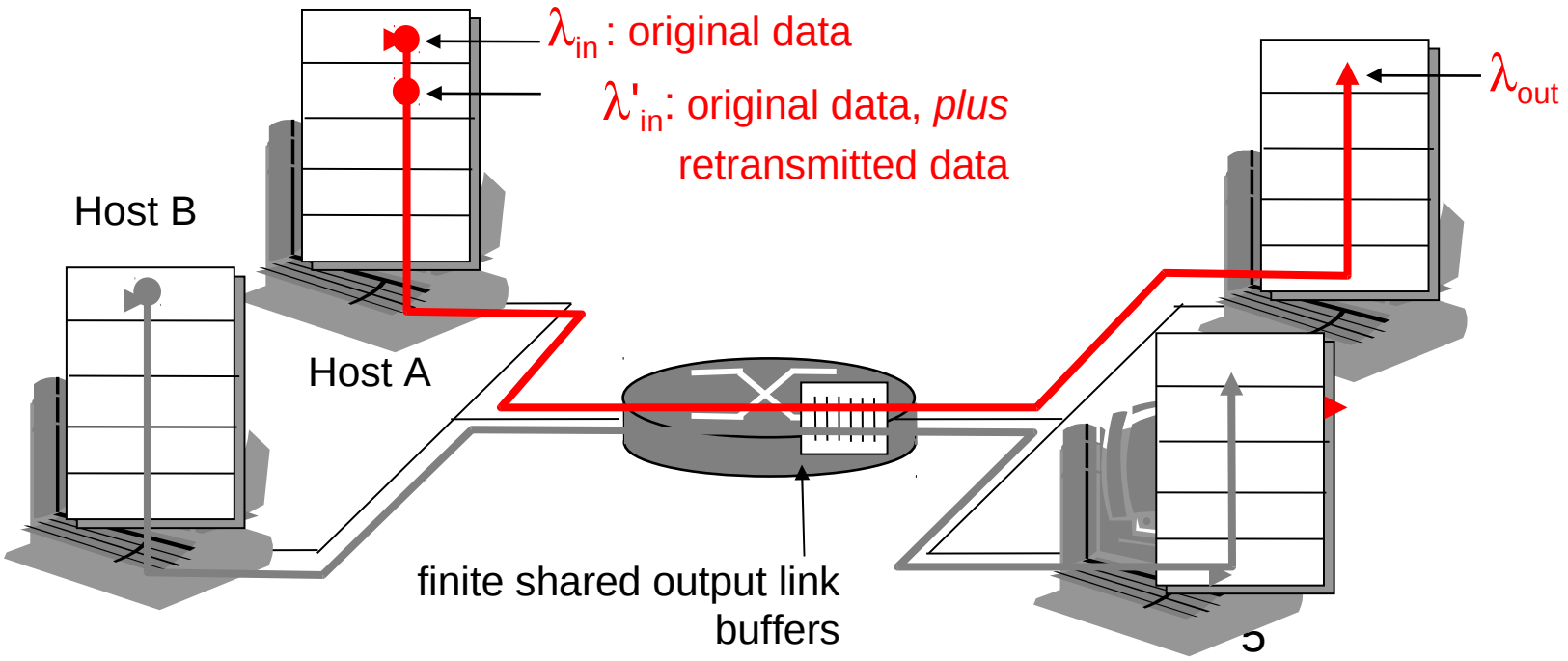
- Large delays when congested
- Maximum achievable throughput

Causes/costs of congestion: scenario 2

- one router, *finite* buffers
- sender retransmission of timed-out packet

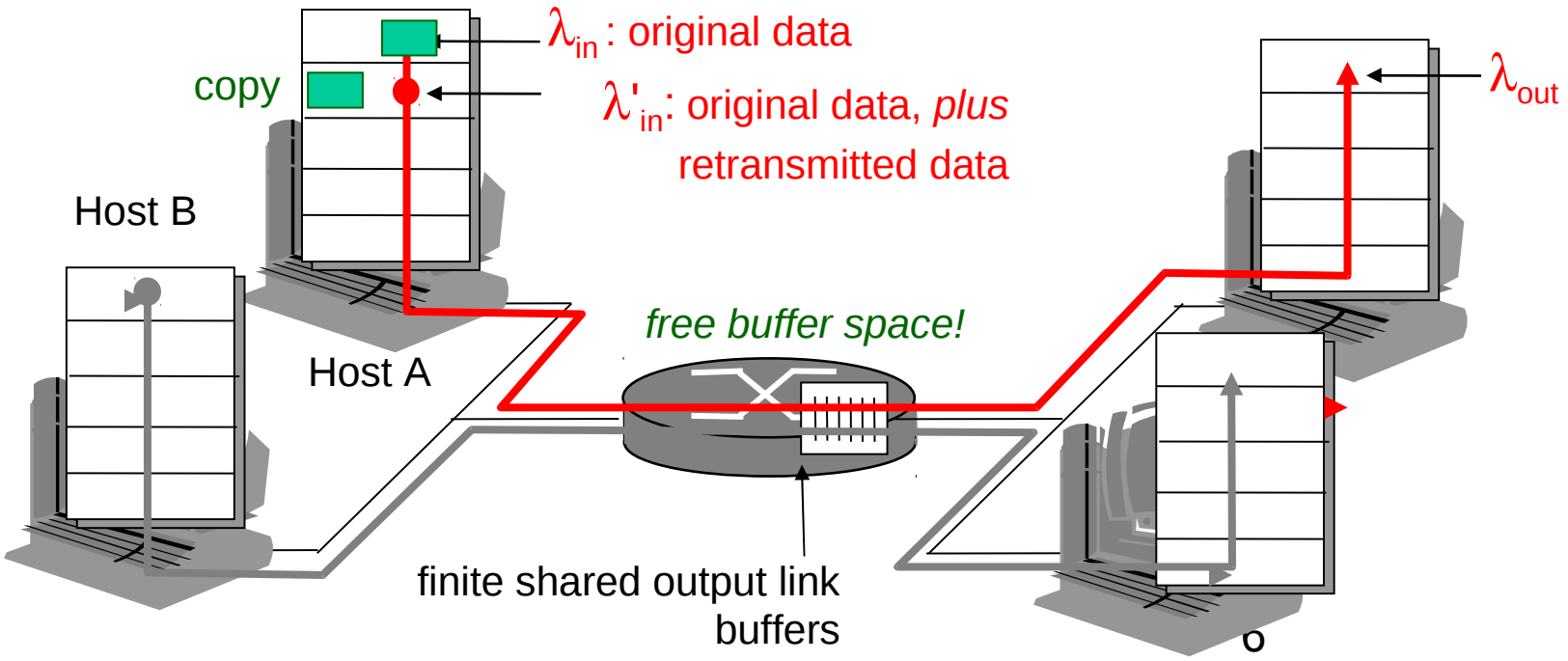
  application-layer input = application-layer output: $\lambda_{in} = \lambda_{out}$

  transport-layer input includes *retransmissions* : $\lambda'_{in} \geq \lambda_{in}$



Congestion scenario 2a: ideal case

- sender sends only when router buffers available



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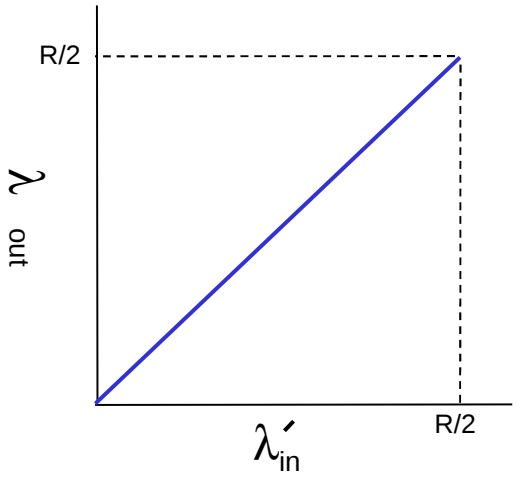
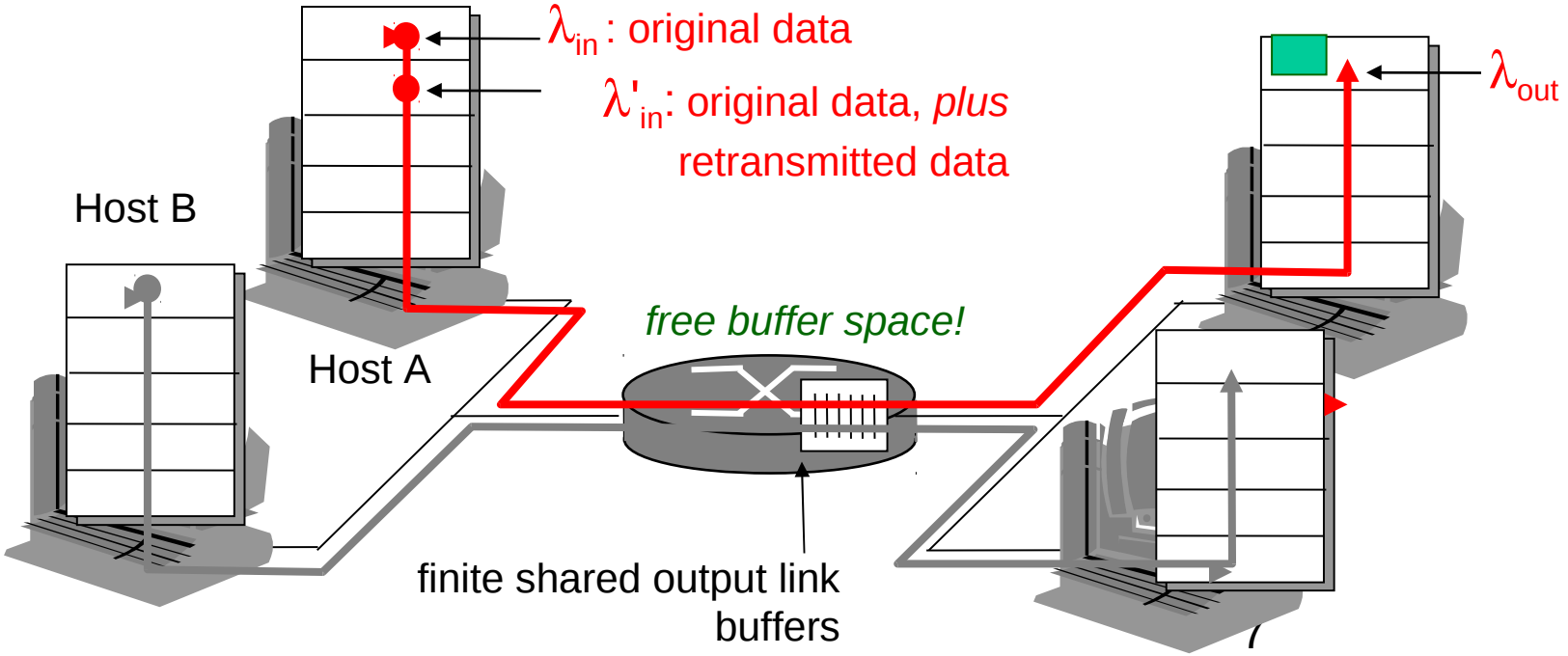
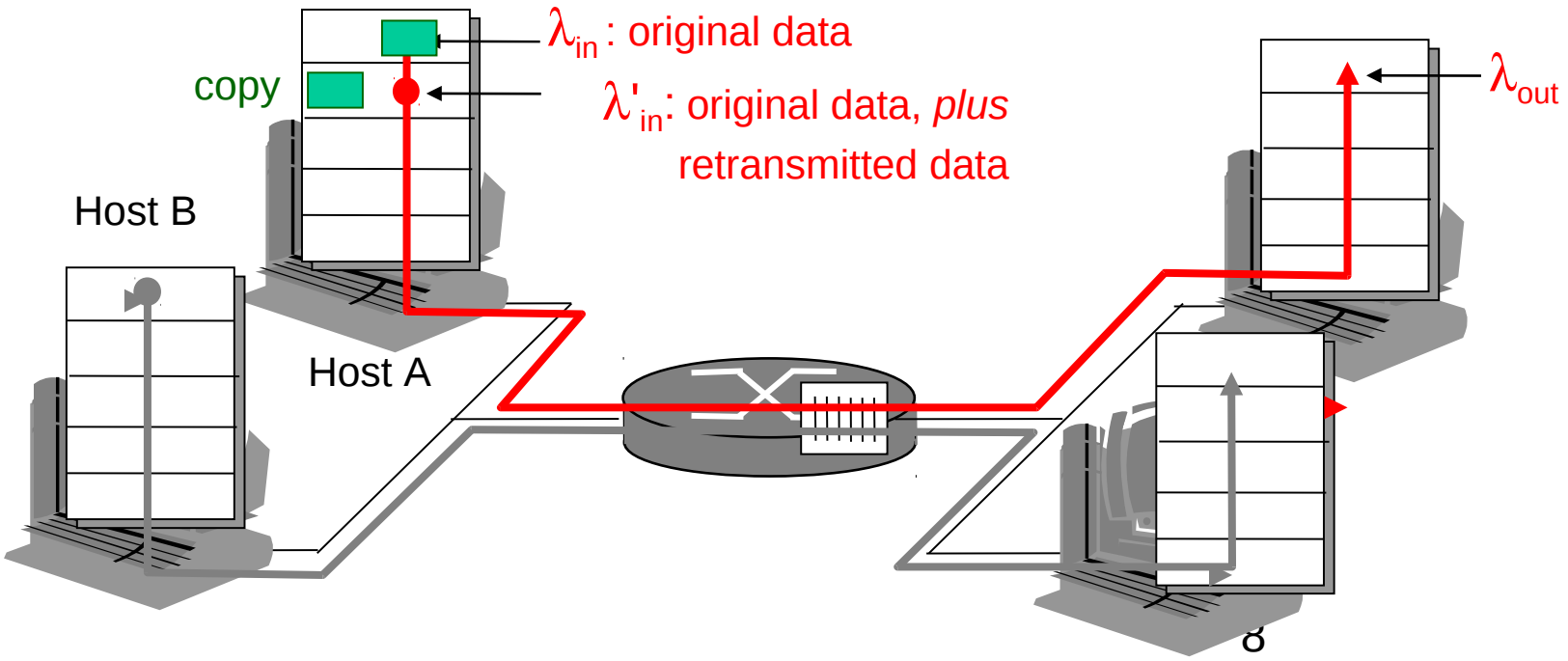




Congestion scenario 2b: *known* loss

- Packets may get dropped at router due to full buffers
  - Sometimes lost
- Sender only resends if packet *known* to be lost (admittedly idealized)



Congestion scenario 2b: *known* loss

packets may get dropped at router due to full buffers

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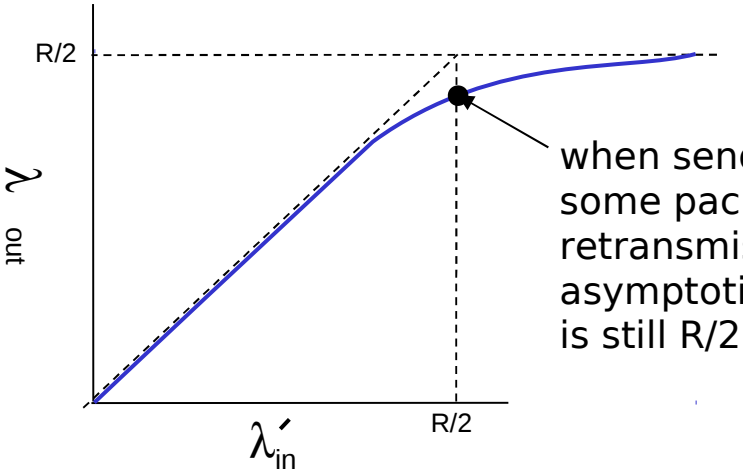
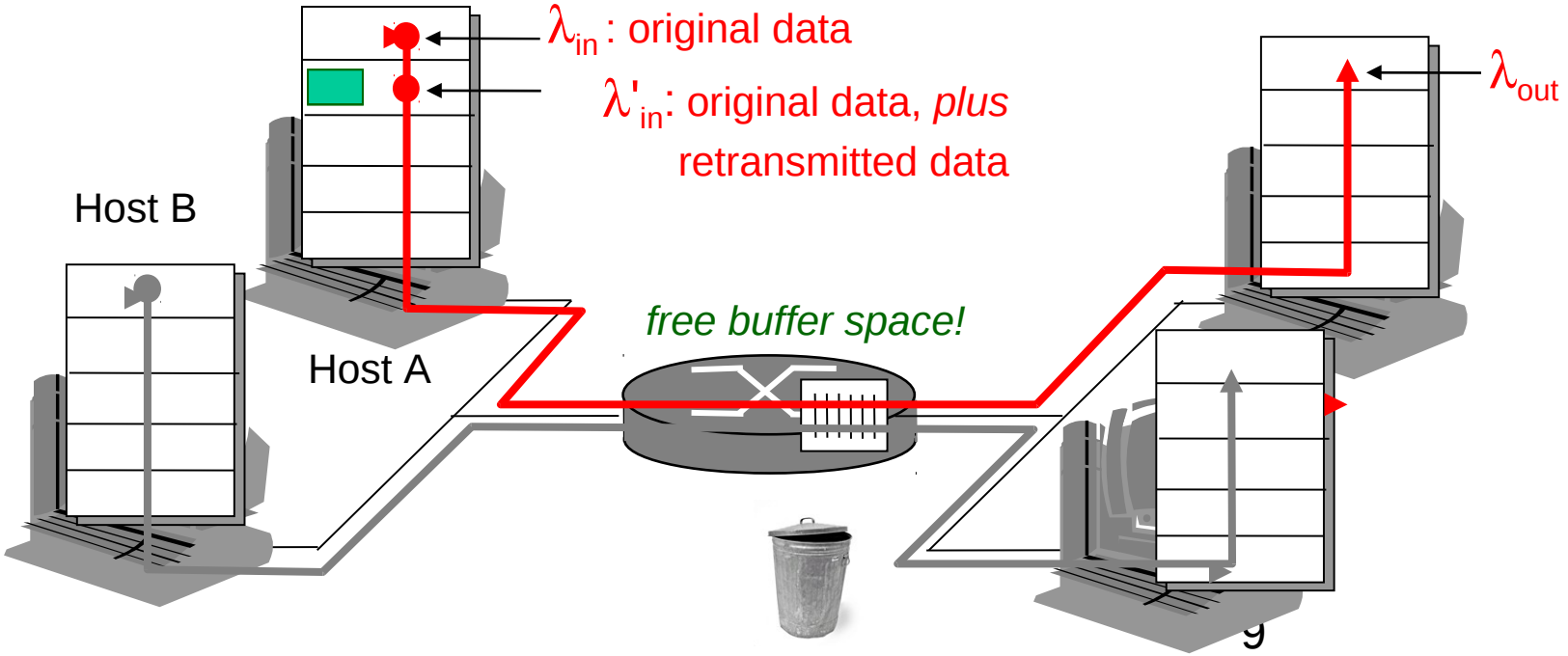
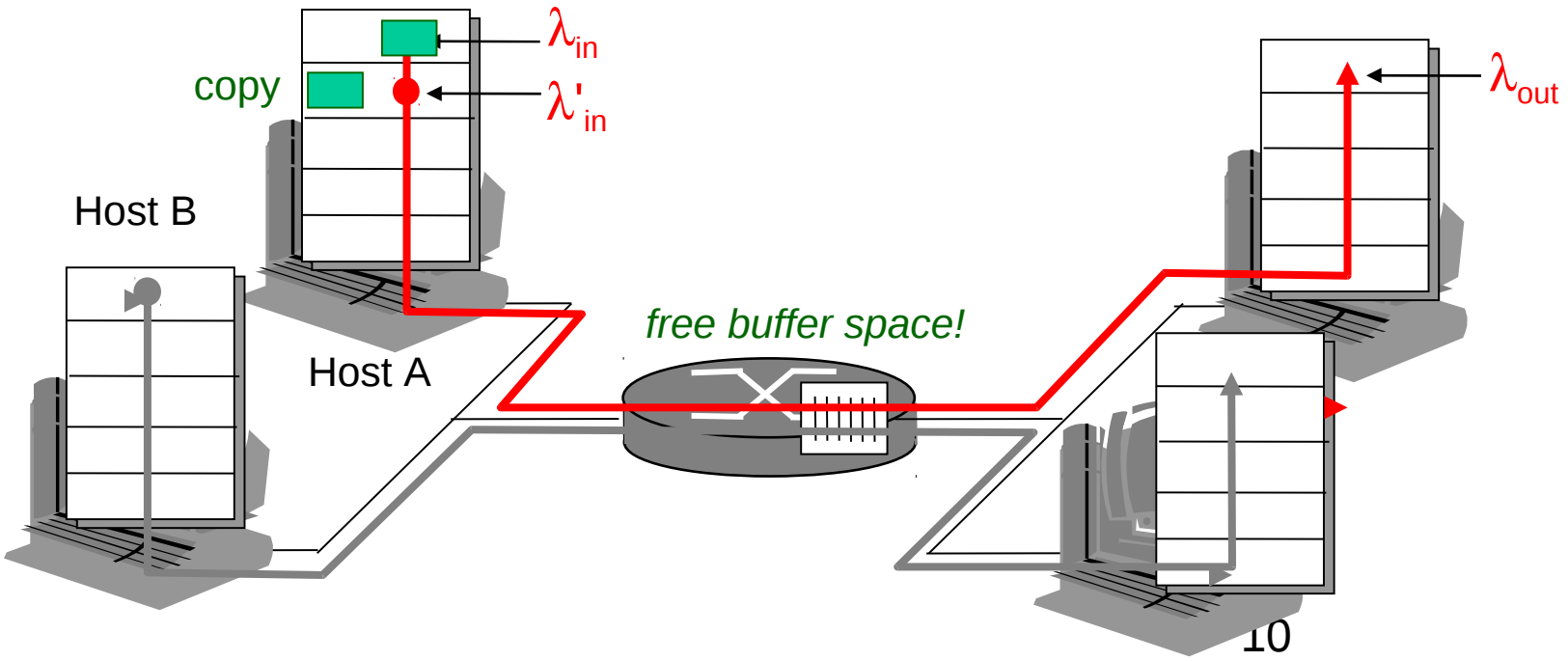


when sending at R/2, some packets are retransmissions but asymptotic goodput is still R/2 (why?)



Congestion scenario 2c: *duplicates*

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



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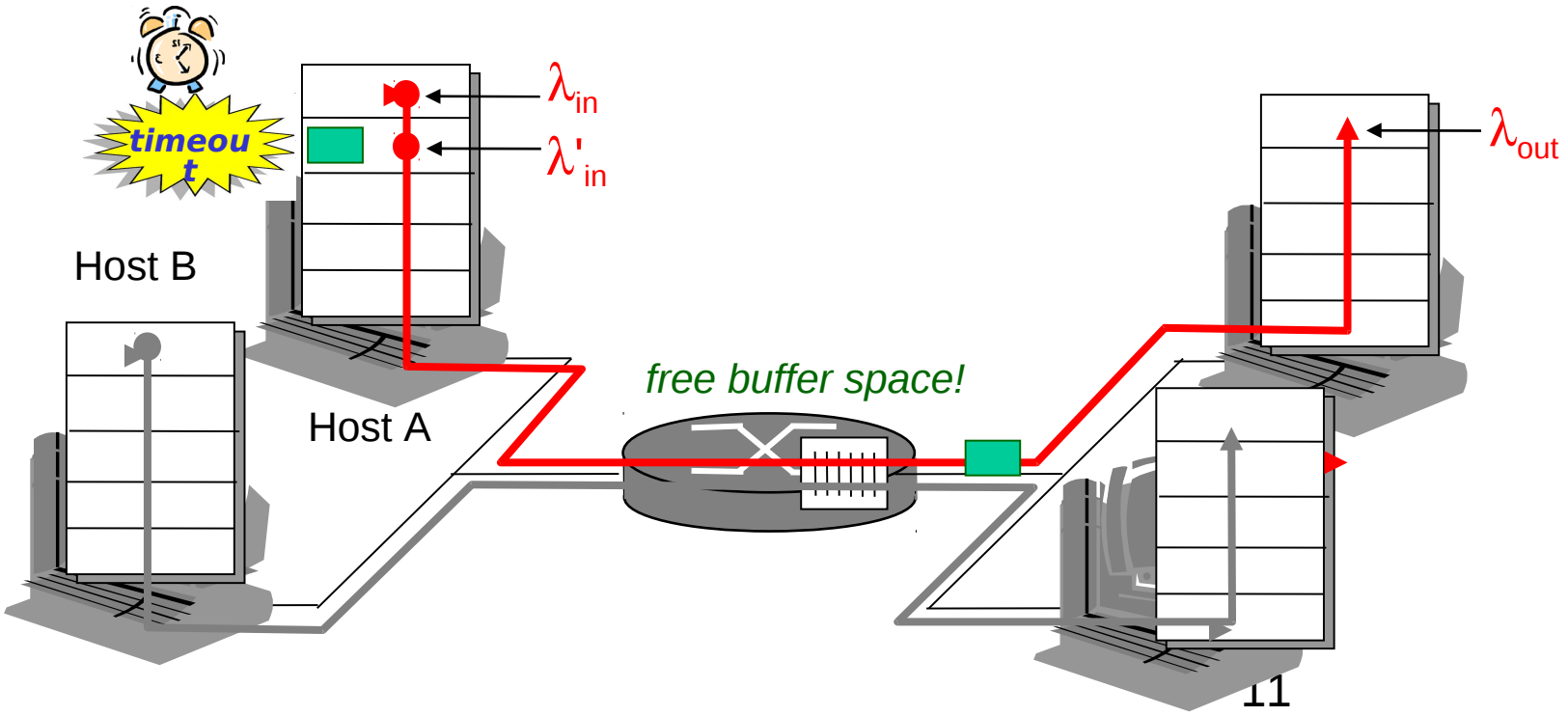

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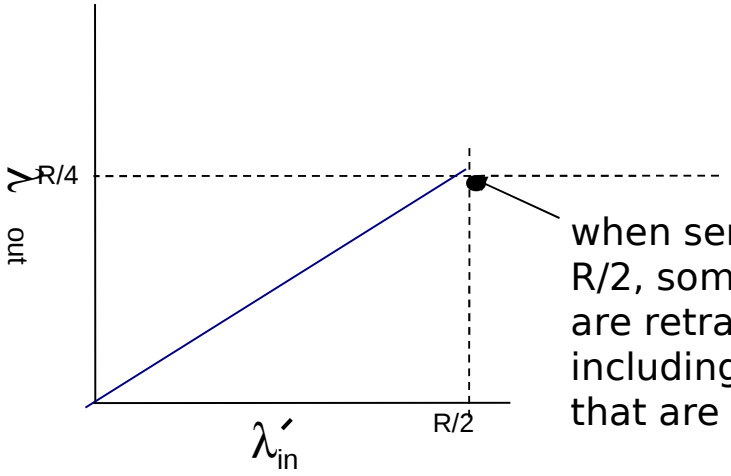


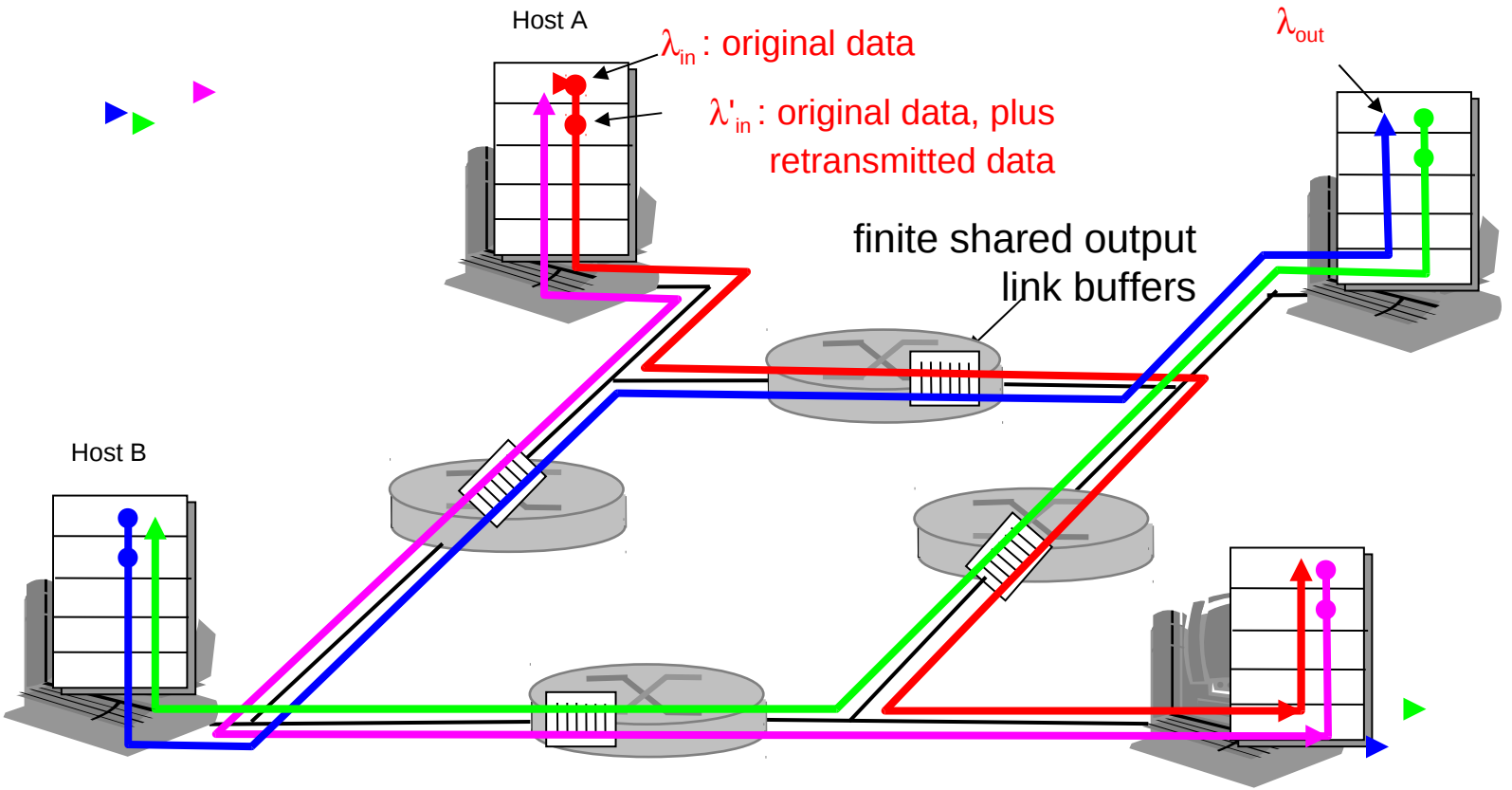
"costs" of congestion:

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

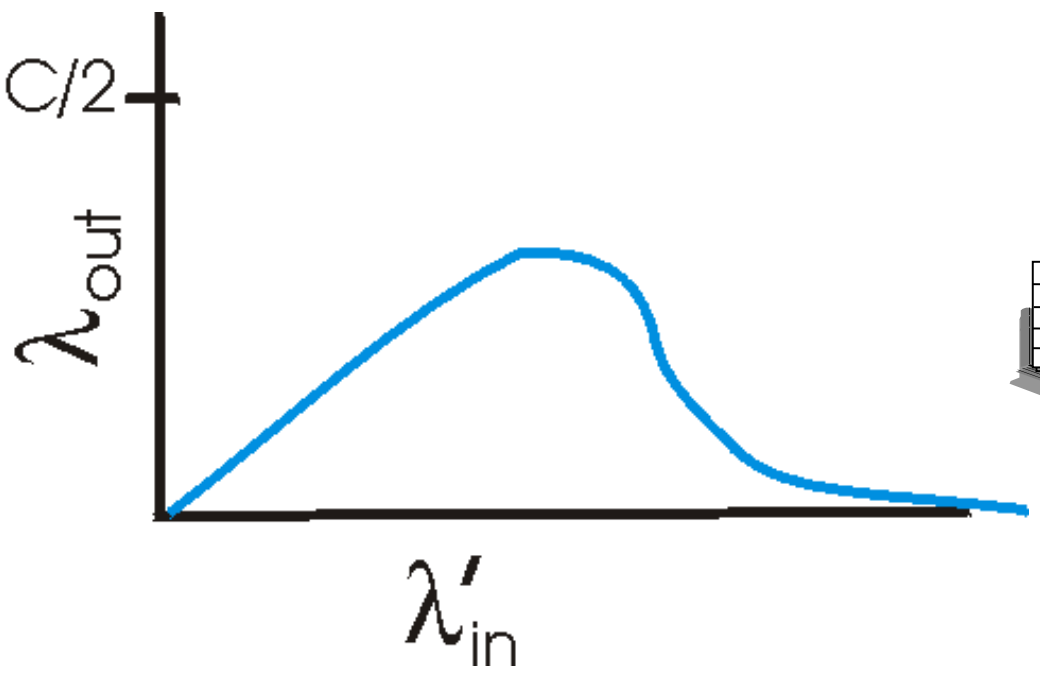
Causes/costs of congestion: scenario 3

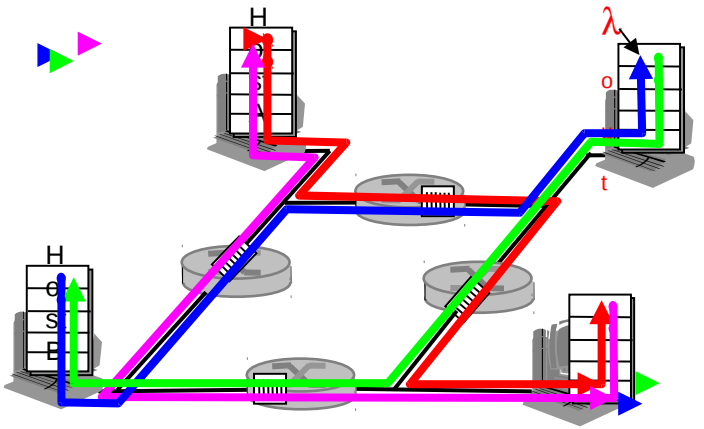
- four senders
- multihop paths
- timeout/retransmit

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



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

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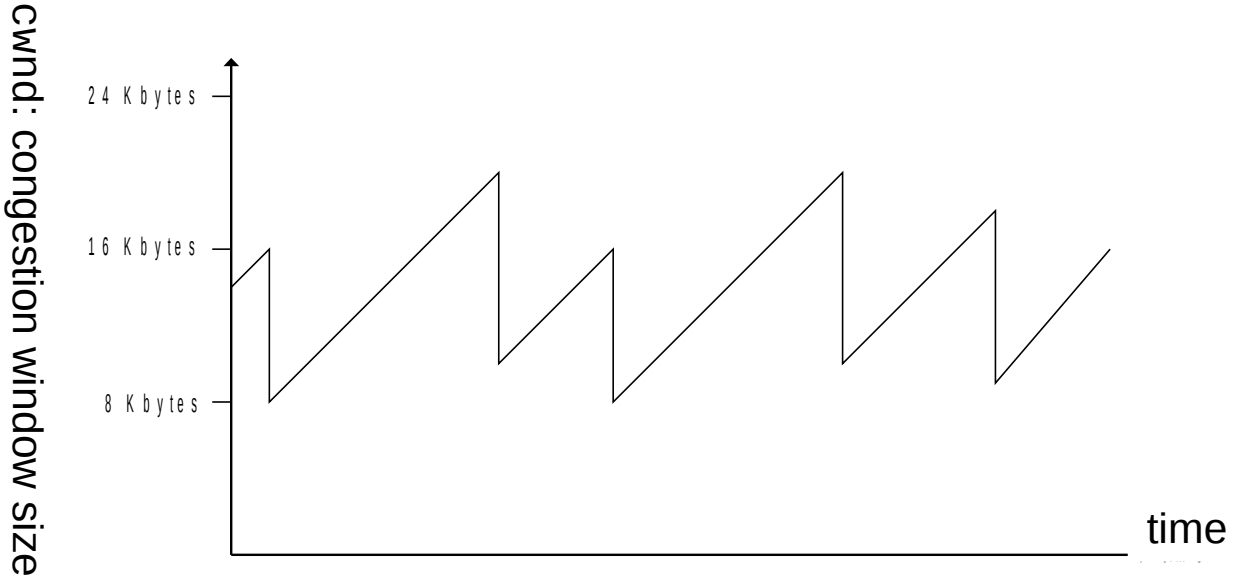
TCP congestion control: additive increase, multiplicative decrease

- *approach:* increase 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

saw tooth behavior: probing for bandwidth



TCP Congestion Control: details

sender limits transmission:

LastByteSent-LastByteAcked

≤ cwnd

roughly,

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

cwnd is dynamic, function of perceived network congestion

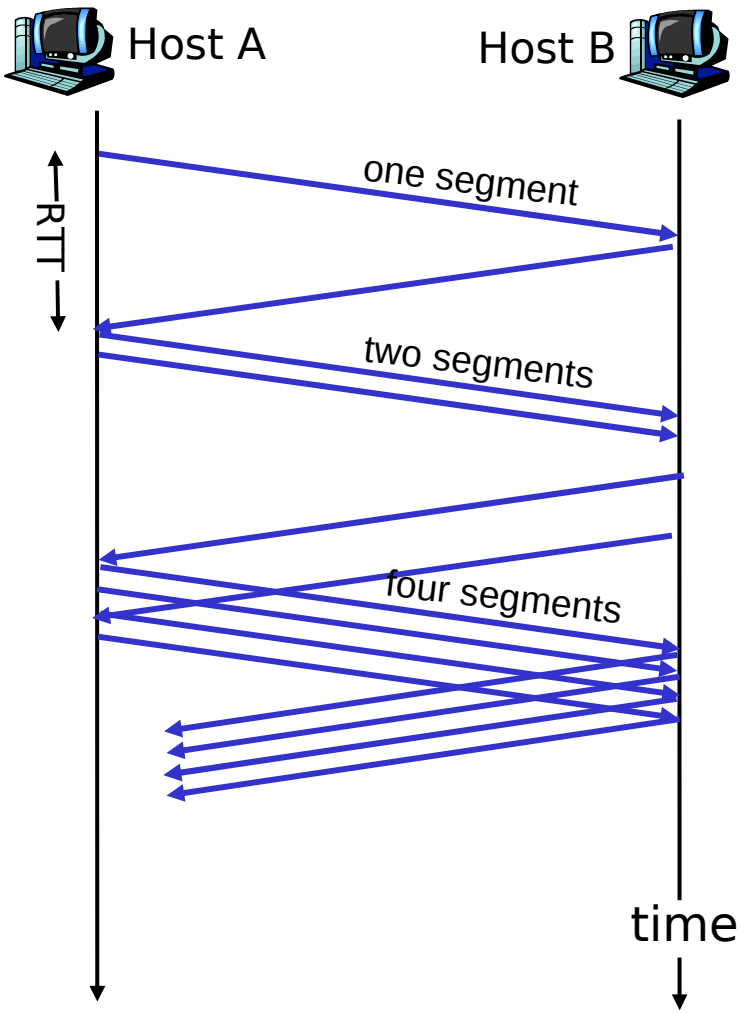
How does sender perceive congestion?

- loss event = timeout *or* 3 duplicate acks
- TCP sender reduces rate (**cwnd**) after loss event

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

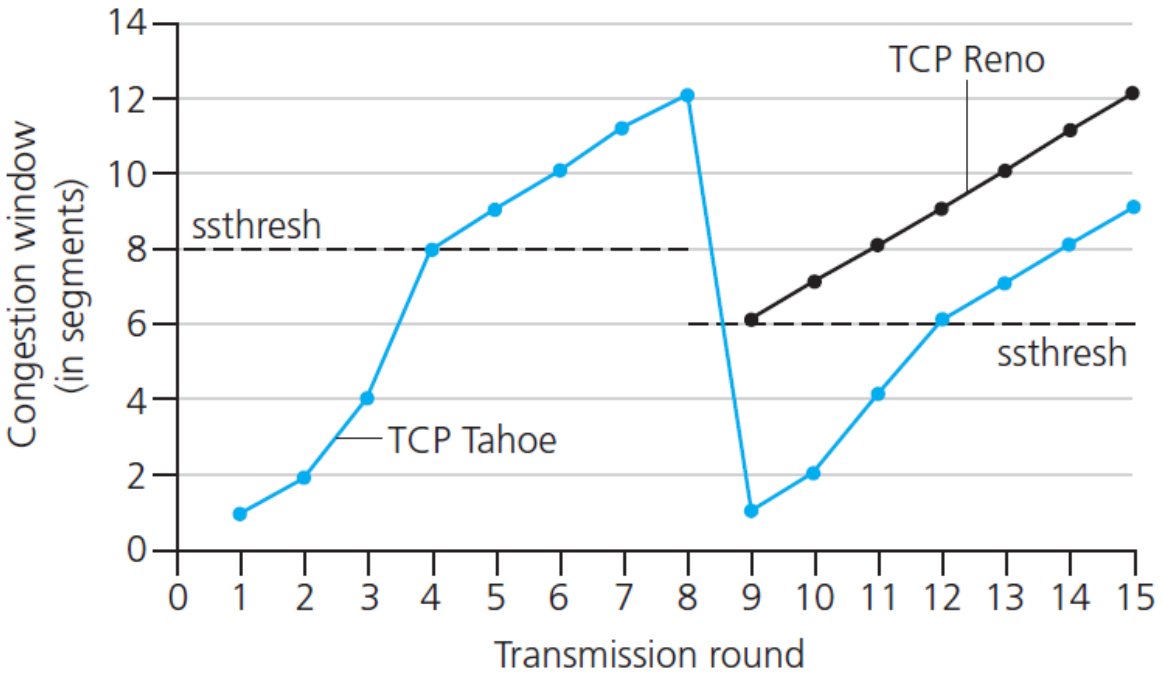
summary: initial rate is slow but ramps up exponentially fast



Refinement

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

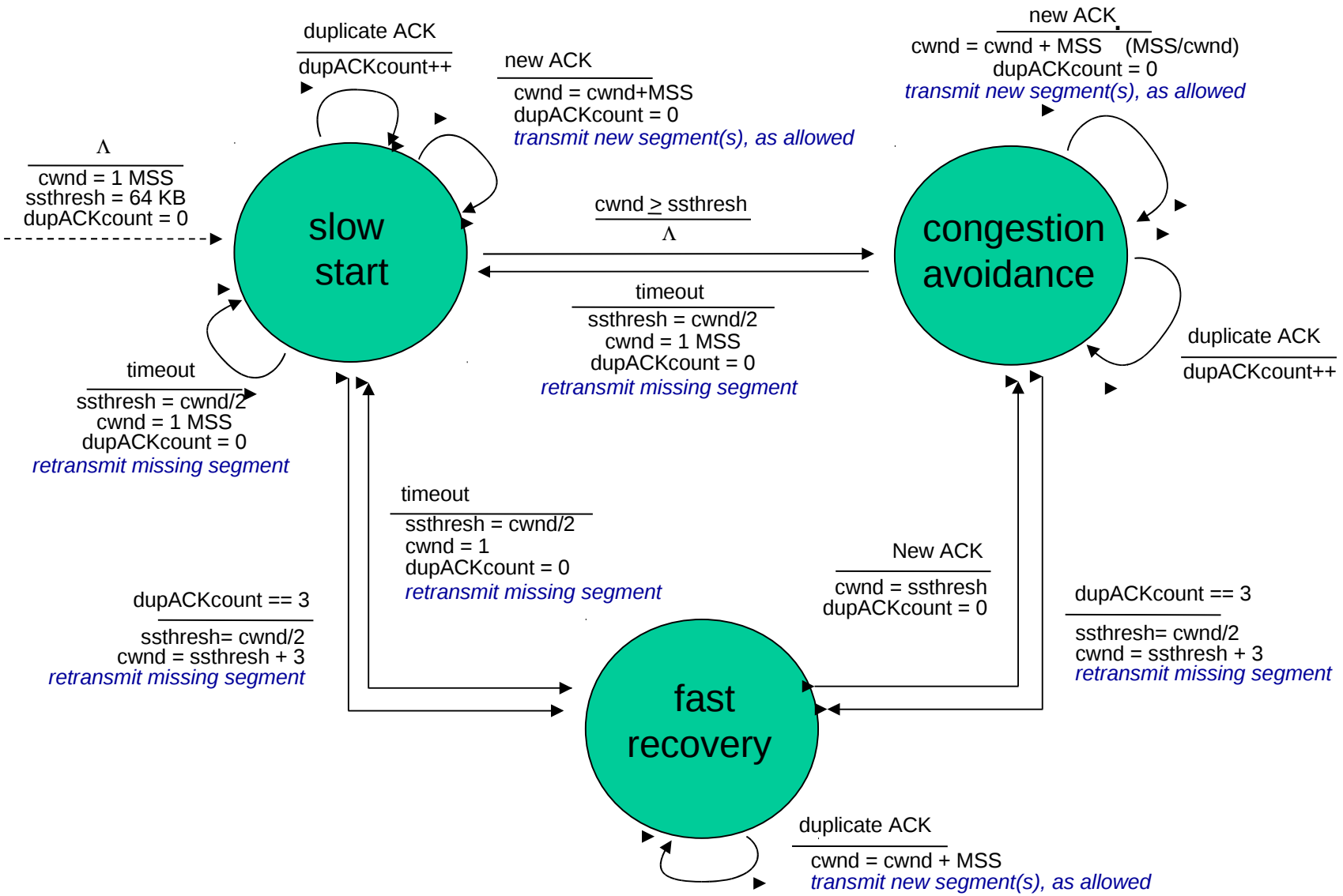
Refinement: inferring loss

- after 3 dup ACKs:
  - **cwnd** is cut in half
  - window then grows linearly
- but after timeout event:
  - **cwnd** instead set to 1 MSS;
  - window then grows exponentially
  - to a threshold, then grows linearly

Philosophy:

- 3 dup ACKs indicates network capable of delivering some segments
- timeout indicates a "more alarming" congestion scenario

Summary: TCP Congestion Control



TCP throughput

- what's the average throughout of TCP as a function of window size and RTT?
  - ignore slow start
- let W be the 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”

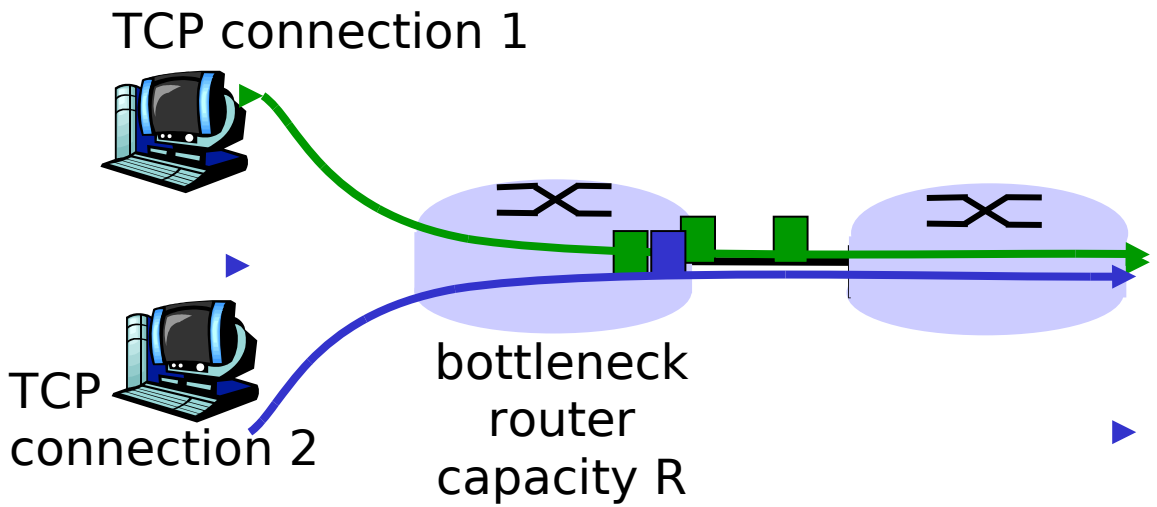
- example: 1500 byte segments, 100ms RTT, want 10 Gbps throughput
- requires window size W = 83,333 in-flight segments
- throughput in terms of loss rate:

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

- → L = $2 \cdot 10^{-10}$ *Wow – a very small loss rate!*
- new versions of TCP for high-speed

TCP Fairness

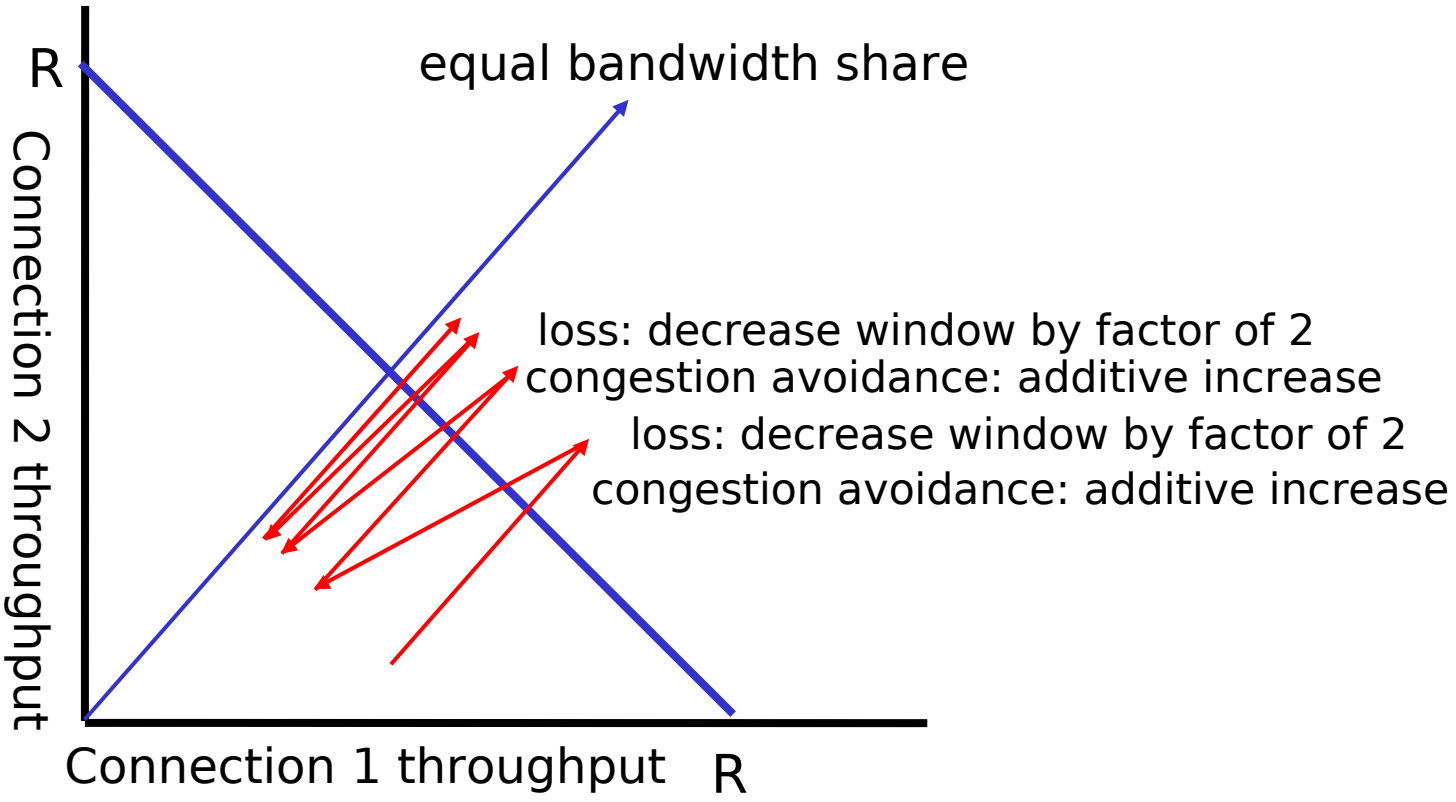
fairness goal: if K TCP sessions share same bottleneck link of bandwidth R, each should have average rate of R/K



Why is TCP fair?

two competing sessions:

- additive increase gives slope of 1, as 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:
  - 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 !