

Chapter 3

Transport Layer

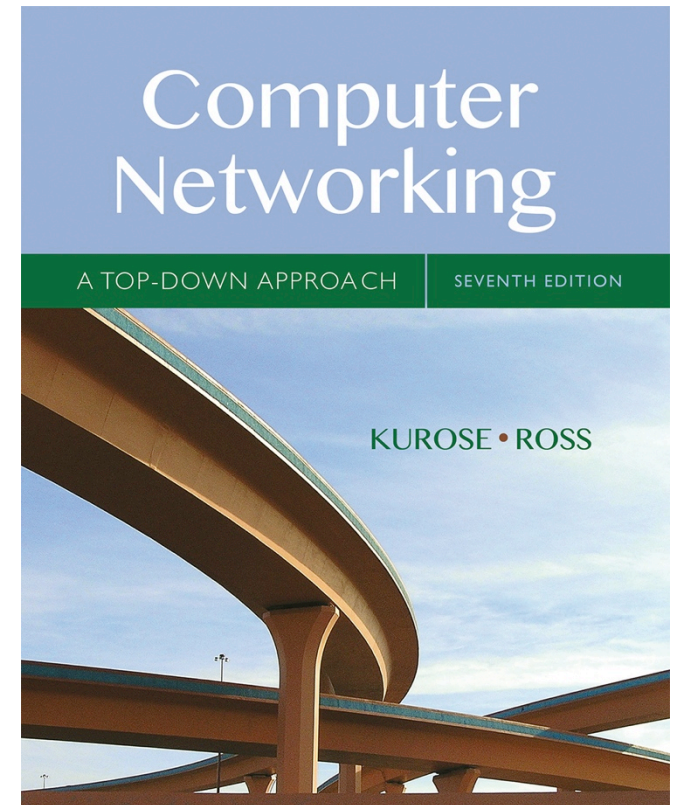
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Computer Networking: A Top Down Approach

7th edition

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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

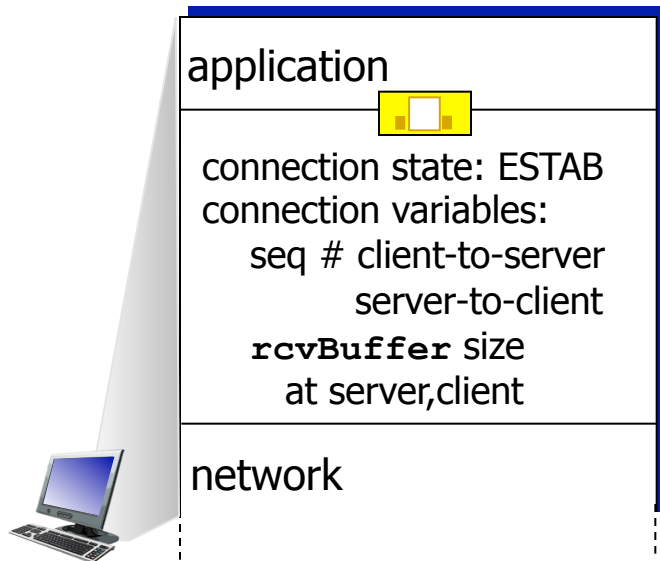
3.6 principles of congestion control

3.7 TCP congestion control

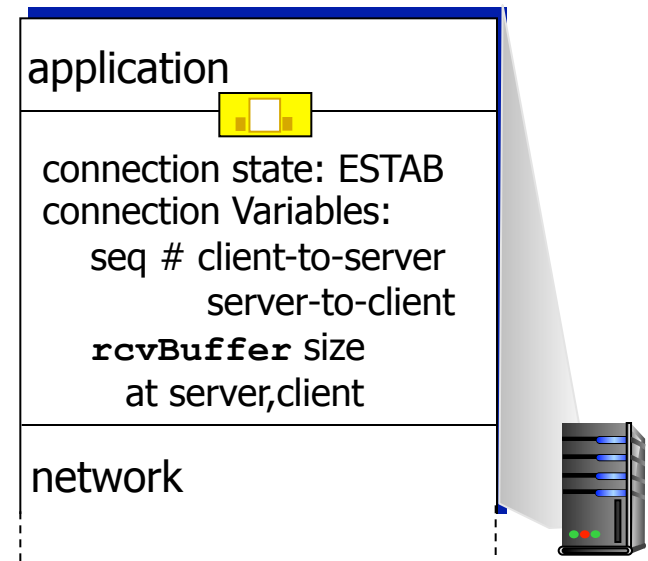
Connection Management

before exchanging data, sender/receiver “handshake”:

- agree to establish connection (each knowing the other willing to establish connection)
- agree on connection parameters



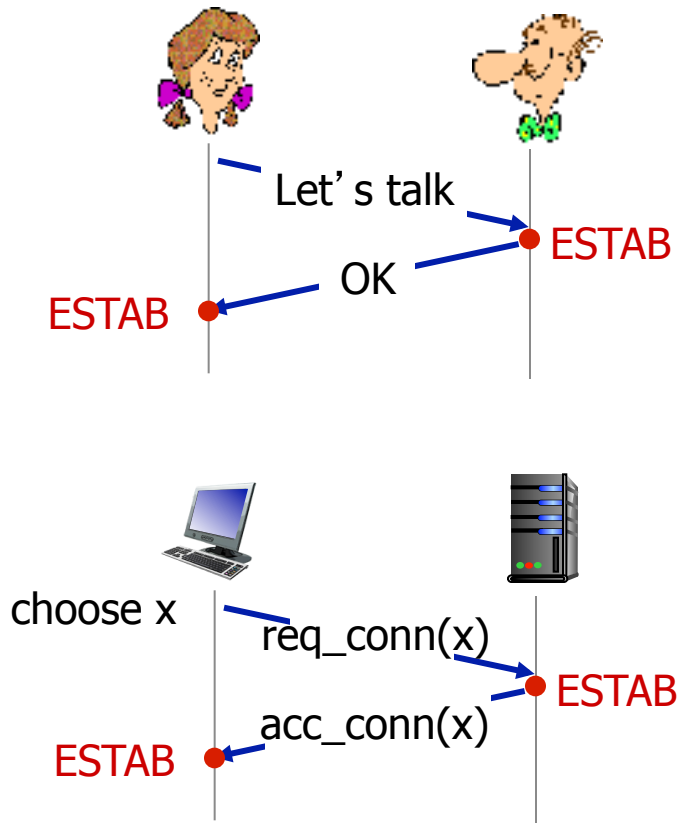
```
Socket clientSocket =  
newSocket("hostname", "port  
number");
```



```
Socket connectionSocket =  
welcomeSocket.accept();
```

Agreeing to establish a connection

2-way handshake:

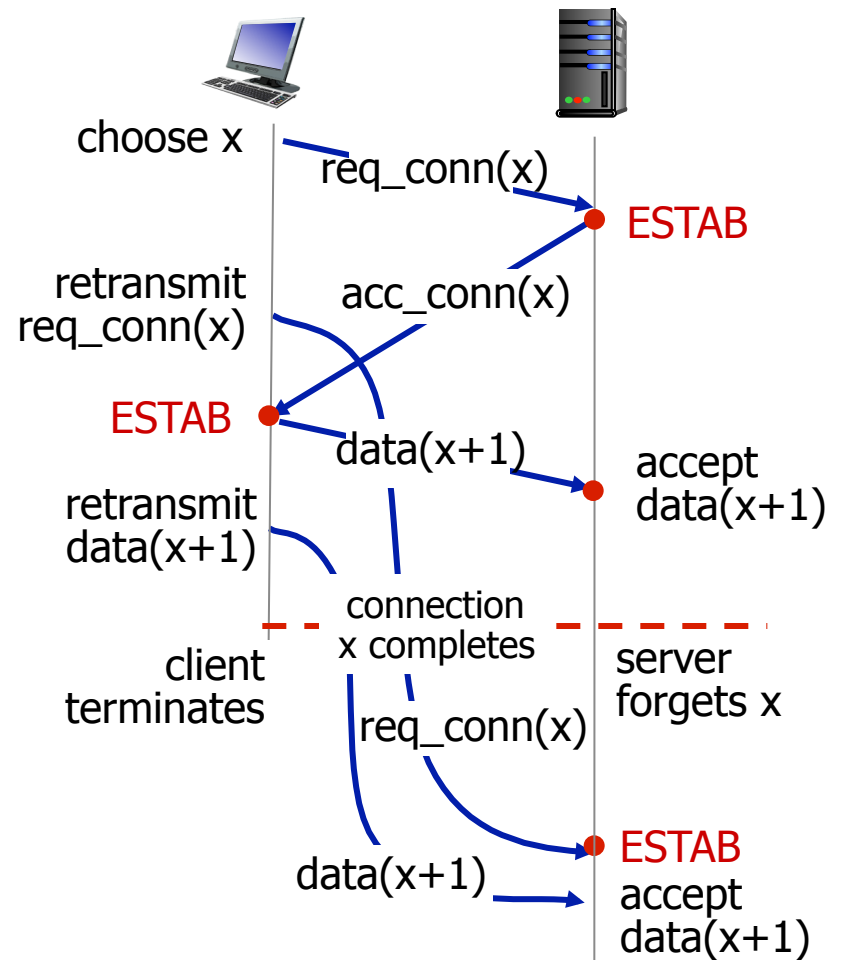
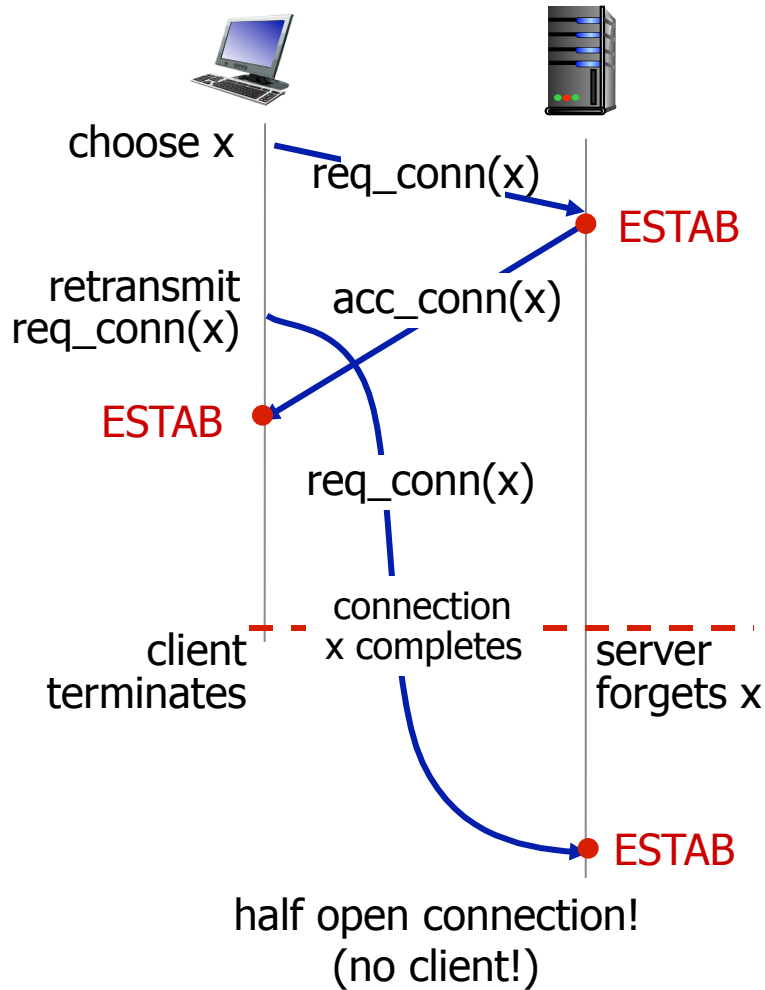


Q: will 2-way handshake always work in network?

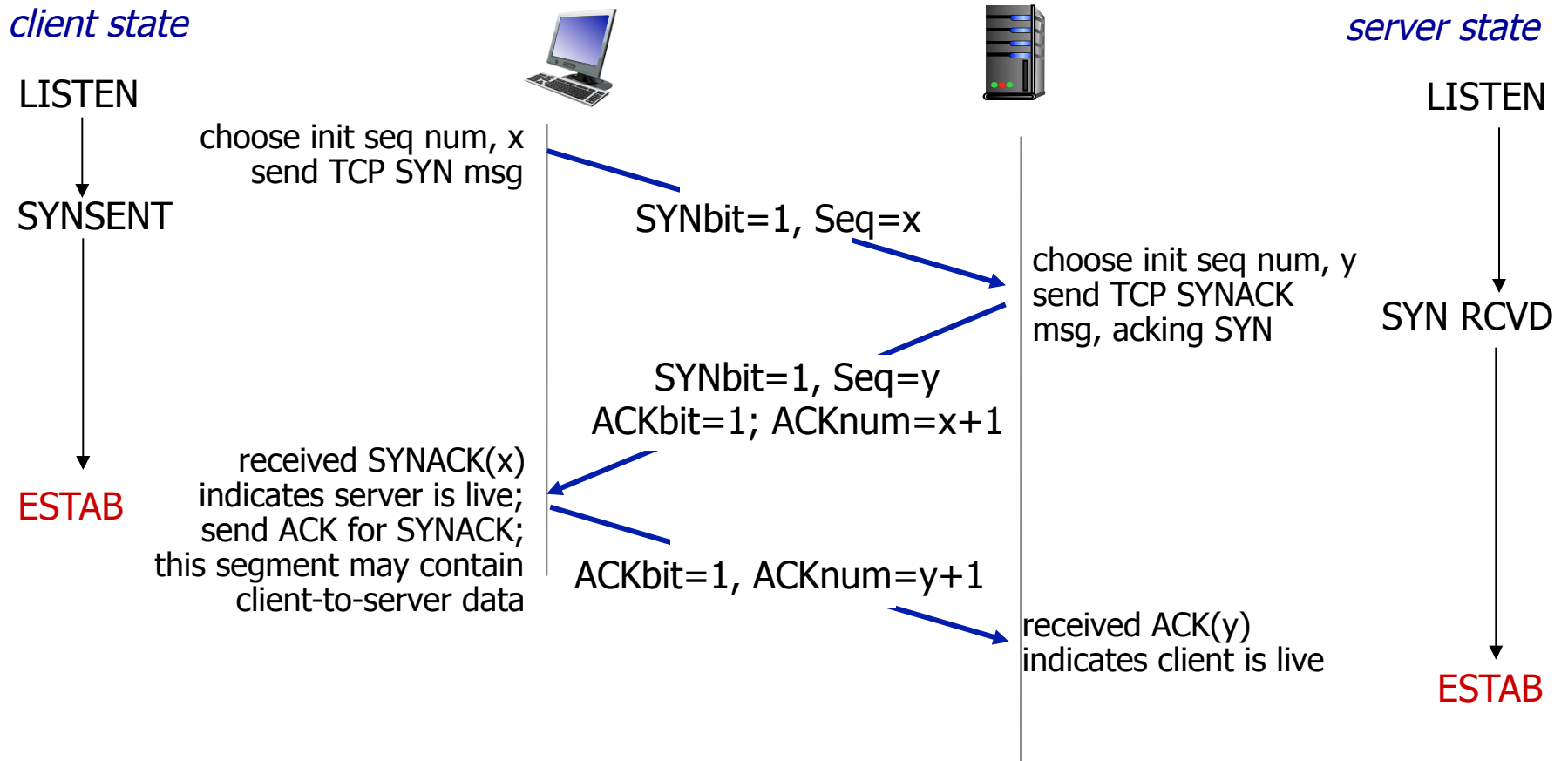
- variable delays
- retransmitted messages (e.g. req_conn(x)) due to message loss
- message reordering
- can't "see" other side

Agreeing to establish a connection

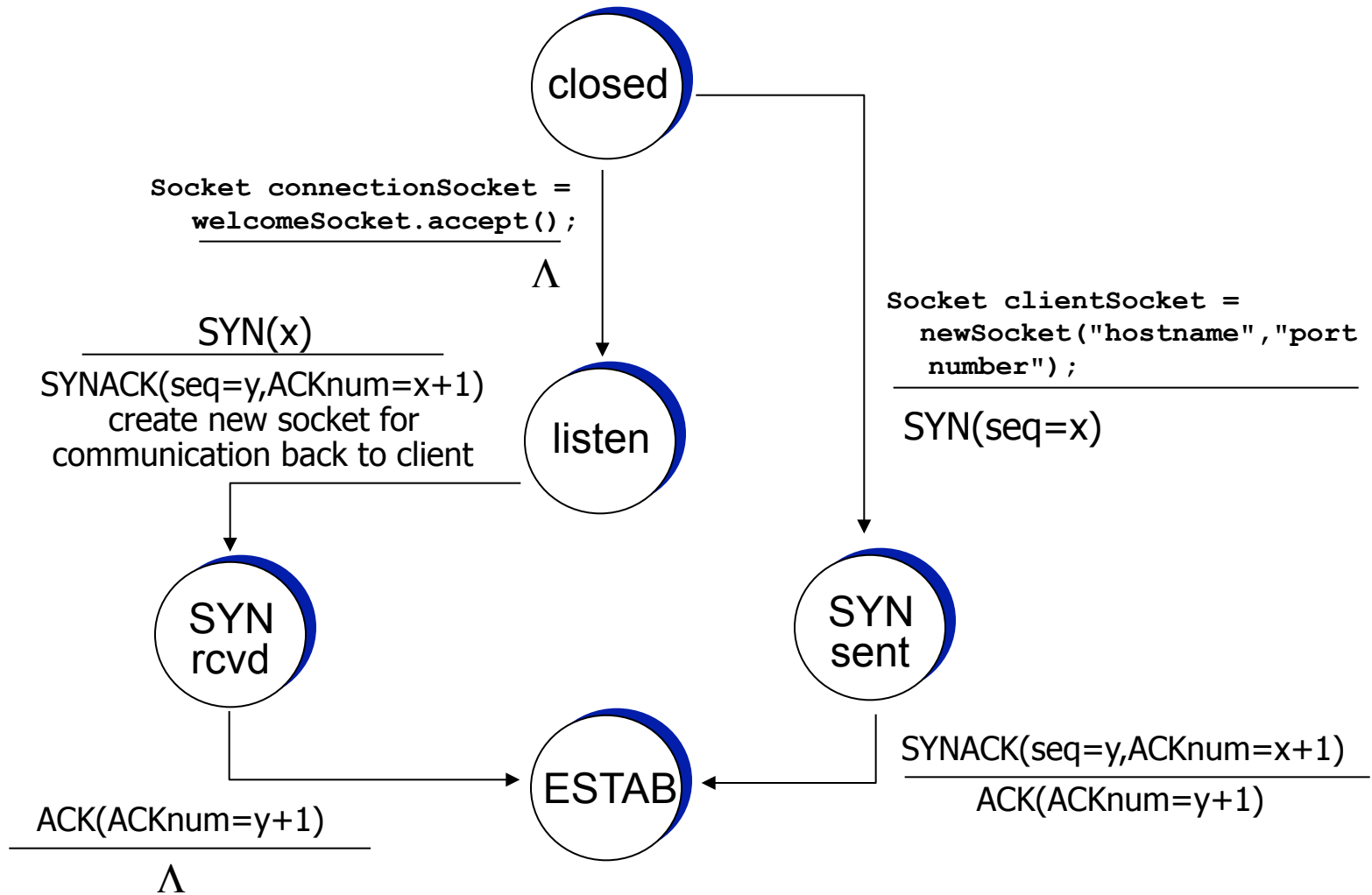
2-way handshake failure scenarios:



TCP 3-way handshake



TCP 3-way handshake: FSM



TCP: closing a connection

- client, server each close their side of connection
 - send TCP segment with FIN bit = 1
- respond to received FIN with ACK
 - on receiving FIN, ACK can be combined with own FIN
- simultaneous FIN exchanges can be handled

TCP: closing a connection

client state

ESTAB

`clientSocket.close()`

FIN_WAIT_1

can no longer
send but can
receive data

FIN_WAIT_2

wait for server
close

TIMED_WAIT

timed wait
for $2 * \text{max}$
segment lifetime

CLOSED



FINbit=1, seq=x

ACKbit=1; ACKnum=x+1

FINbit=1, seq=y

ACKbit=1; ACKnum=y+1

can still
send data

can no longer
send data

server state

ESTAB

CLOSE_WAIT

LAST_ACK

CLOSED

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3.7 TCP congestion control

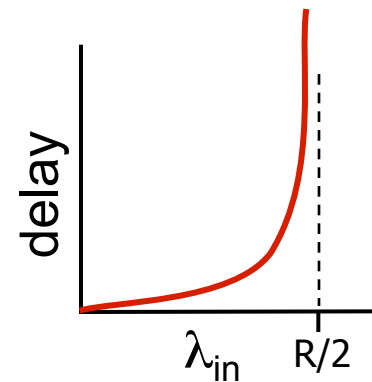
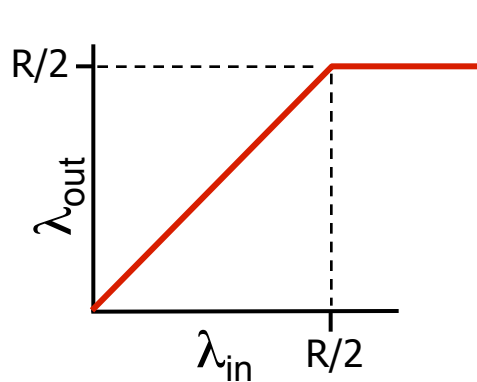
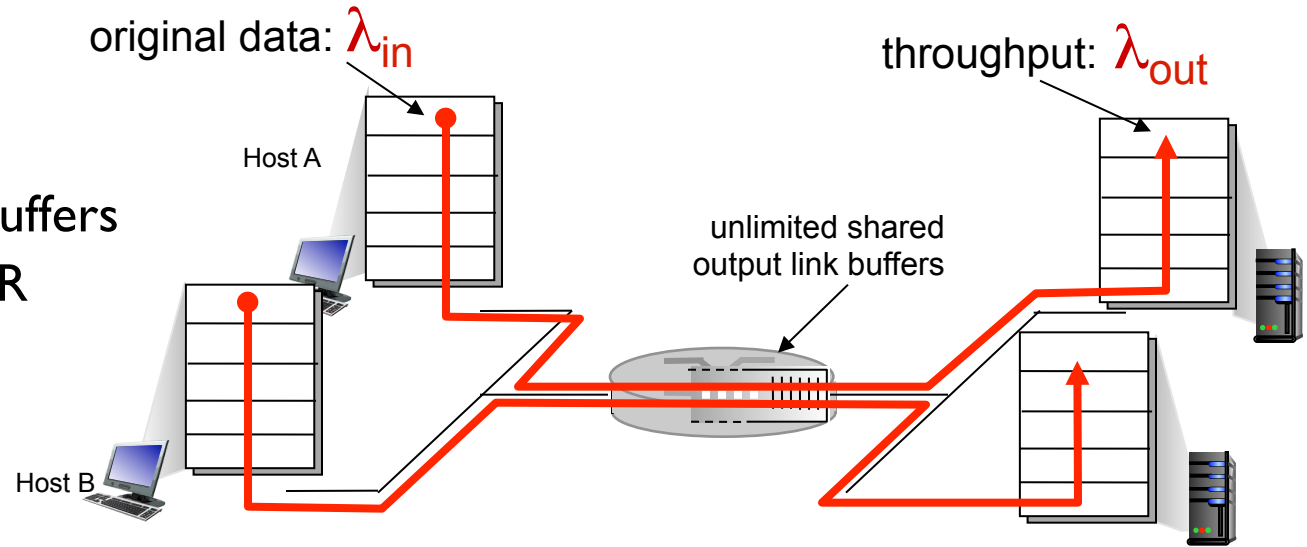
Principles of congestion control

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 I

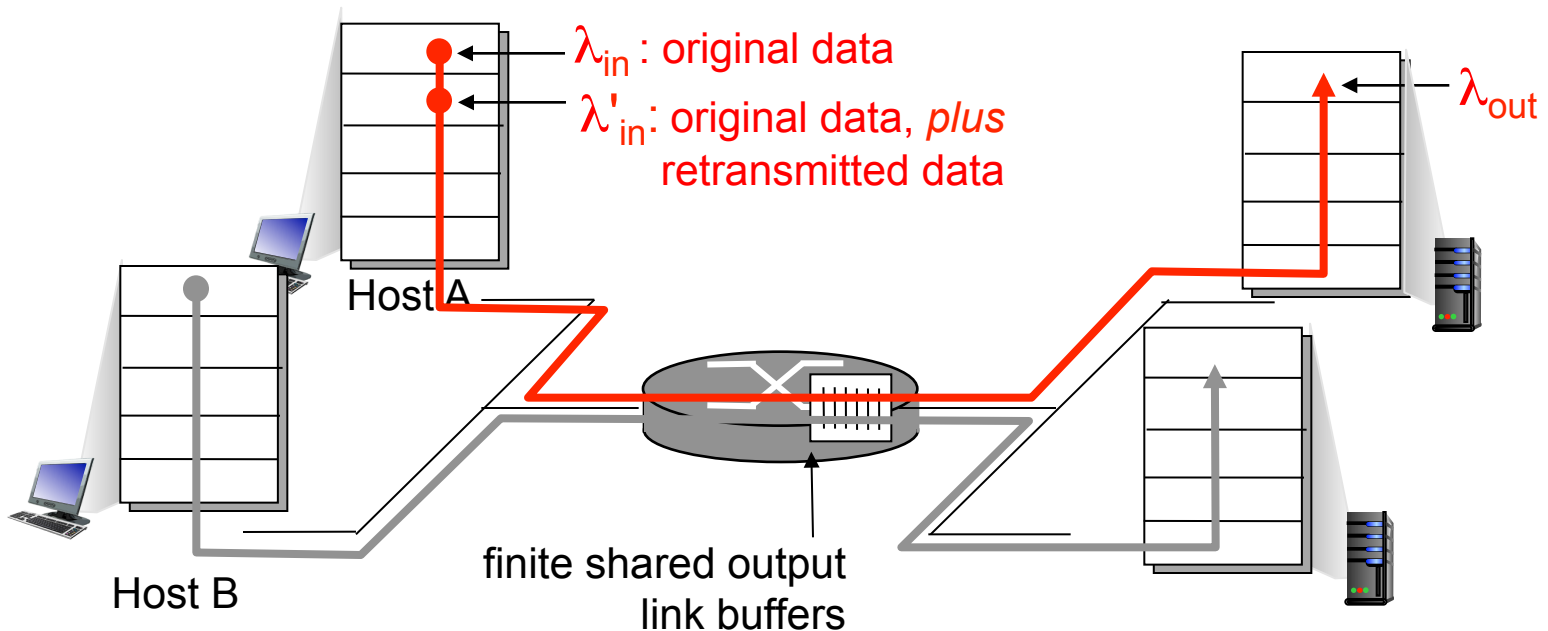
- two senders, two receivers
- one router, infinite buffers
- output link capacity: R
- no retransmission



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

Causes/costs of congestion: scenario 2

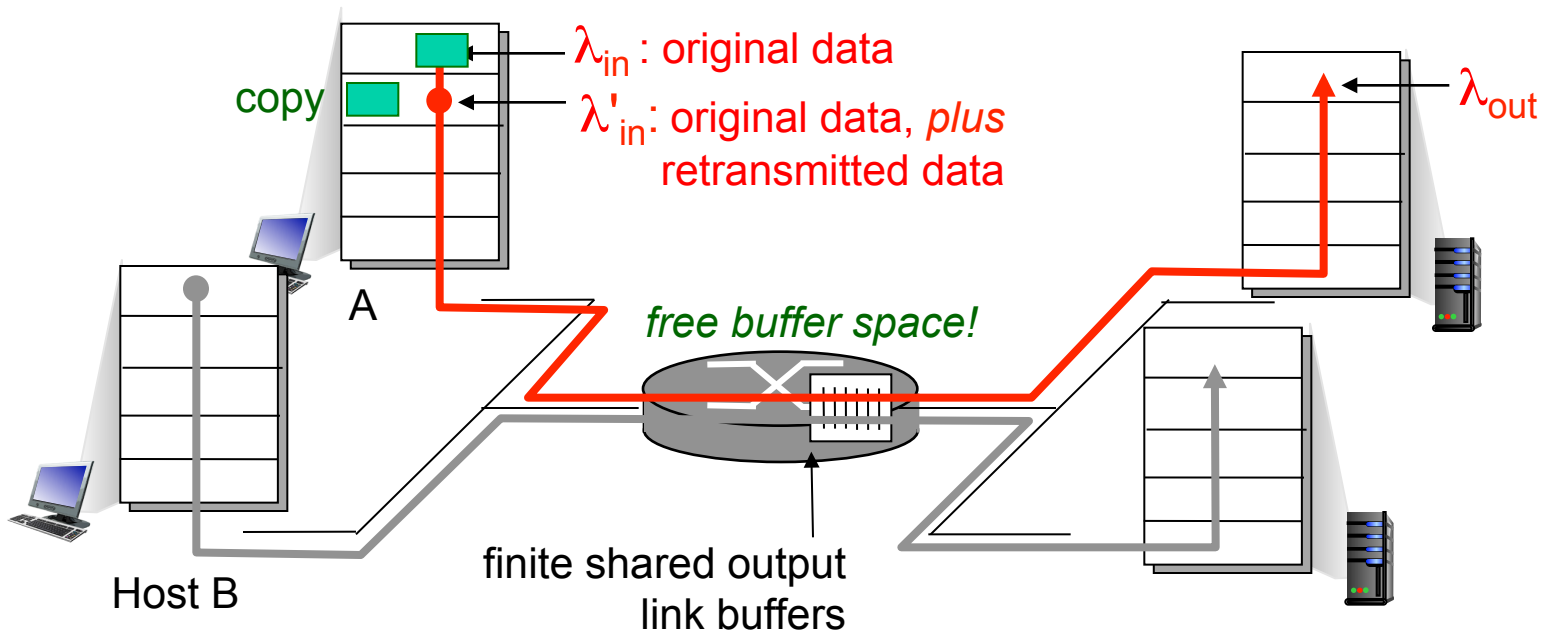
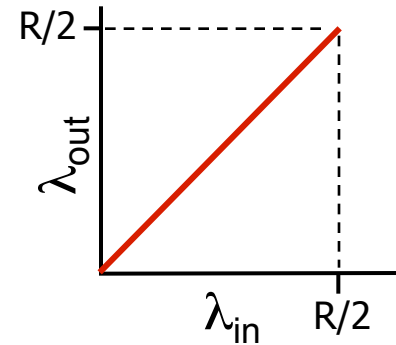
- one router, *finite* buffers
- sender retransmission of timed-out packet
 - application-layer input = application-layer output: $\lambda_{in} = \lambda_{out}$
 - transport-layer input includes *retransmissions* : $\lambda'_{in} \geq \lambda_{in}$



Causes/costs of congestion: scenario 2

idealization: perfect knowledge

- sender sends only when router buffers available

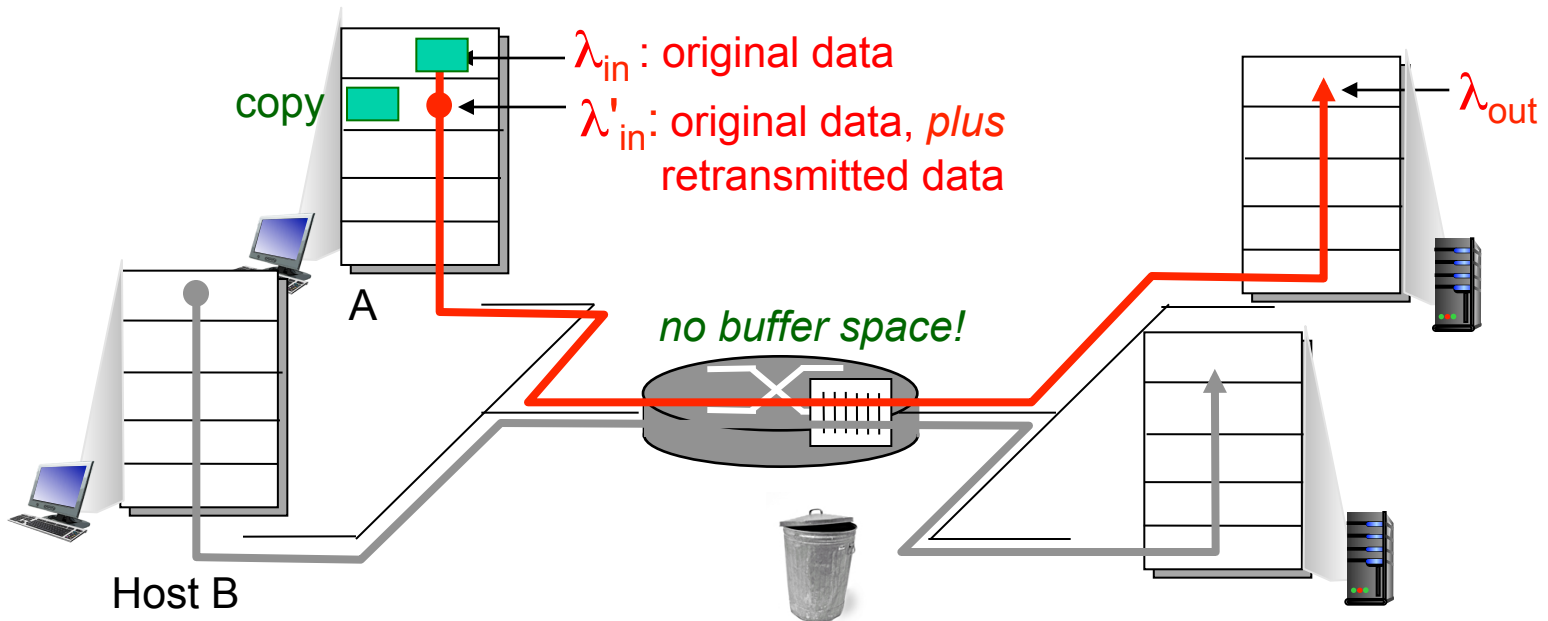


Causes/costs of congestion: scenario 2

Idealization: known loss

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

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

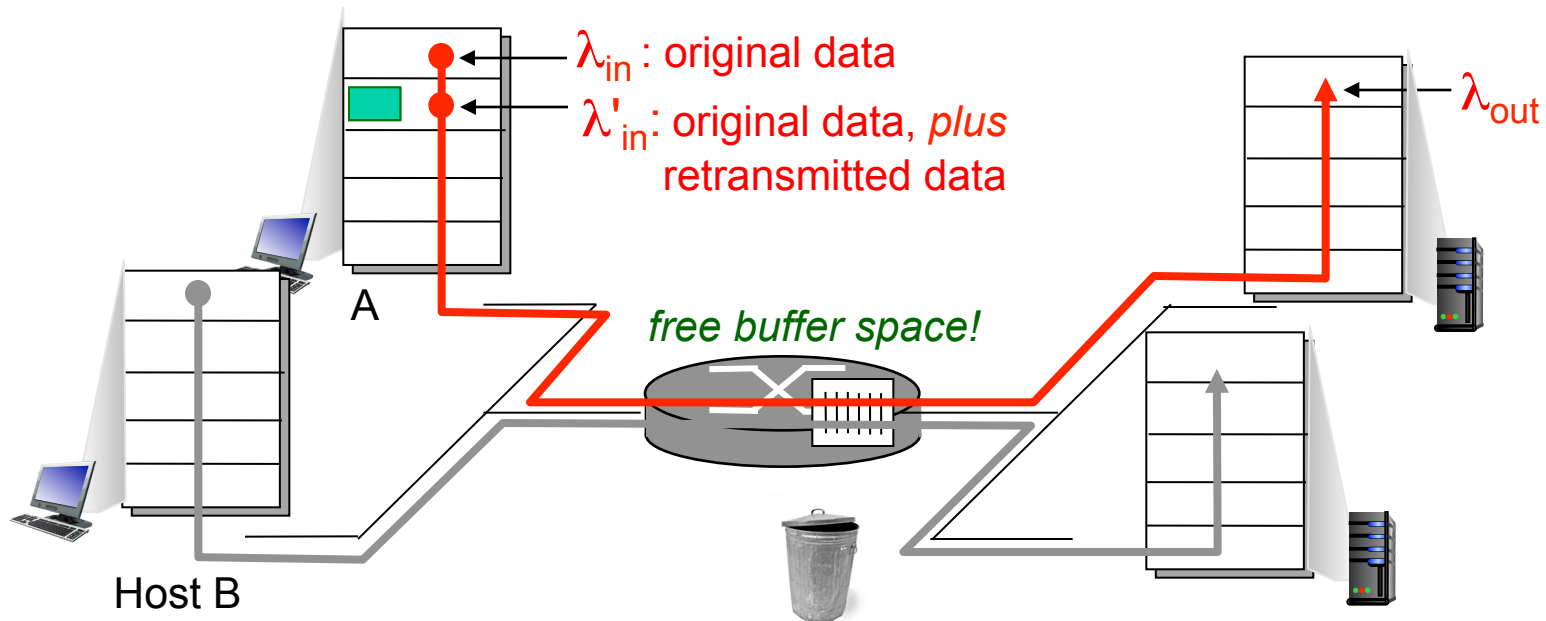
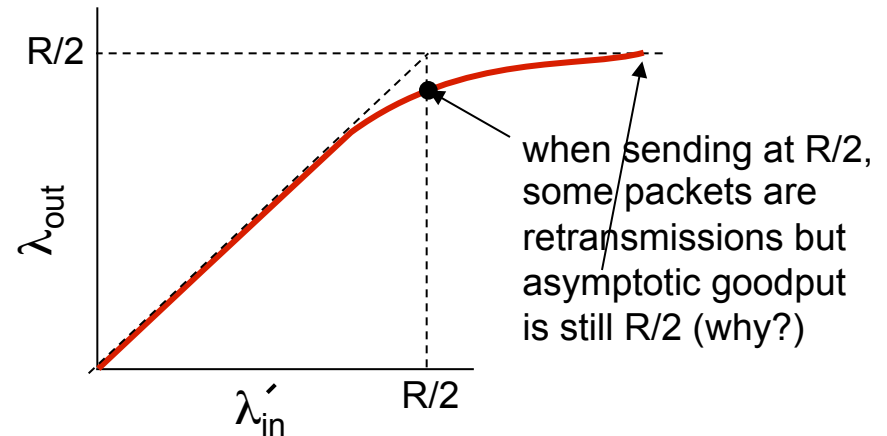


Causes/costs of congestion: scenario 2

Idealization: known loss

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

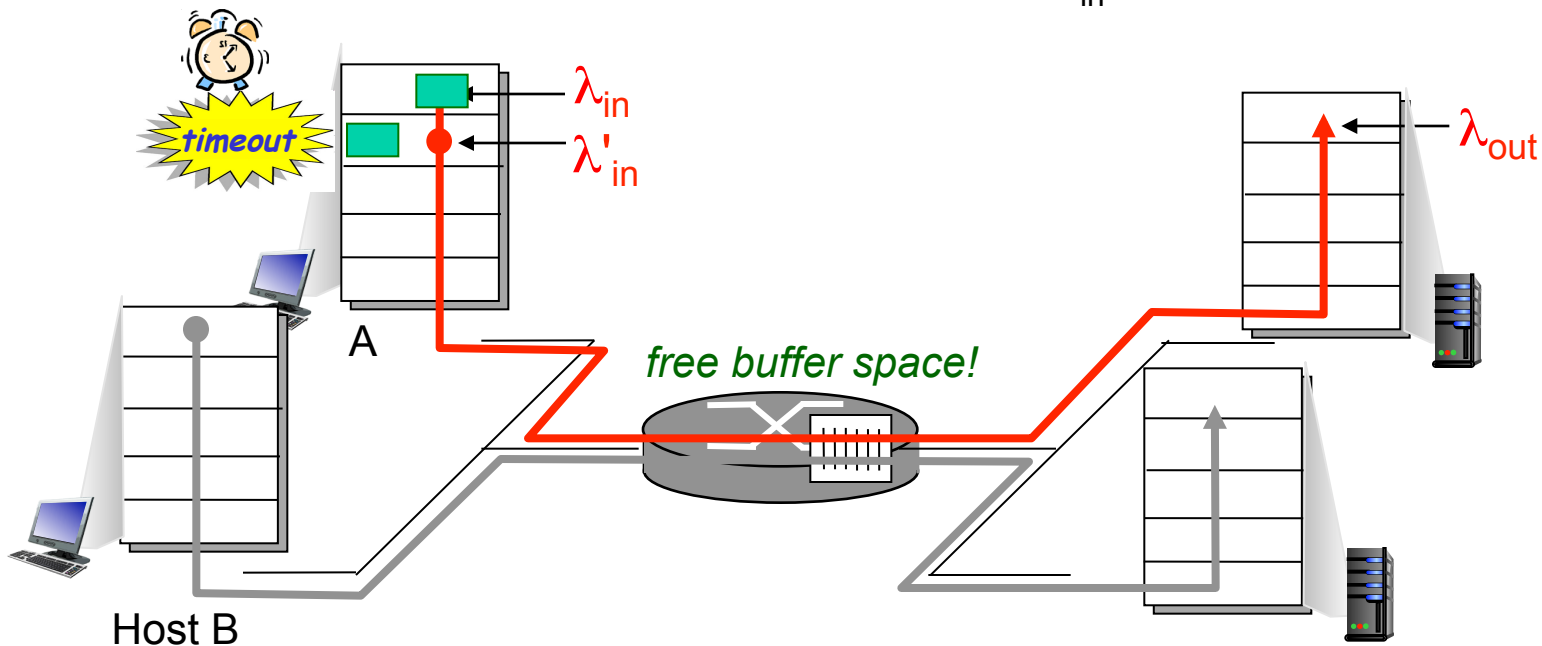
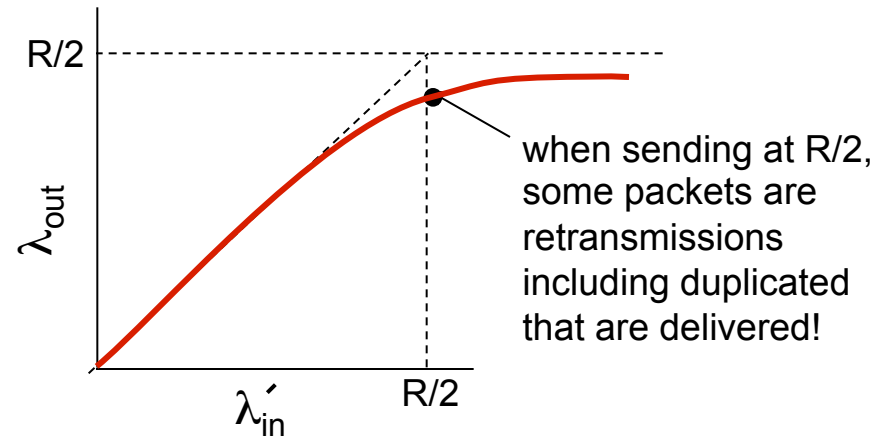
- sender only resends if packet *known* to be lost



Causes/costs of congestion: scenario 2

Realistic: *duplicates*

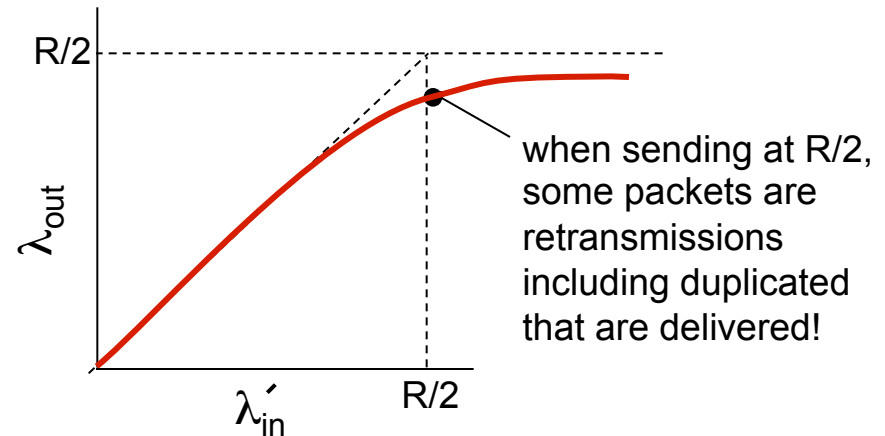
- packets can be lost, dropped at router due to full buffers
- sender times out prematurely, sending *two* copies, both of which are delivered



Causes/costs of congestion: scenario 2

Realistic: *duplicates*

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

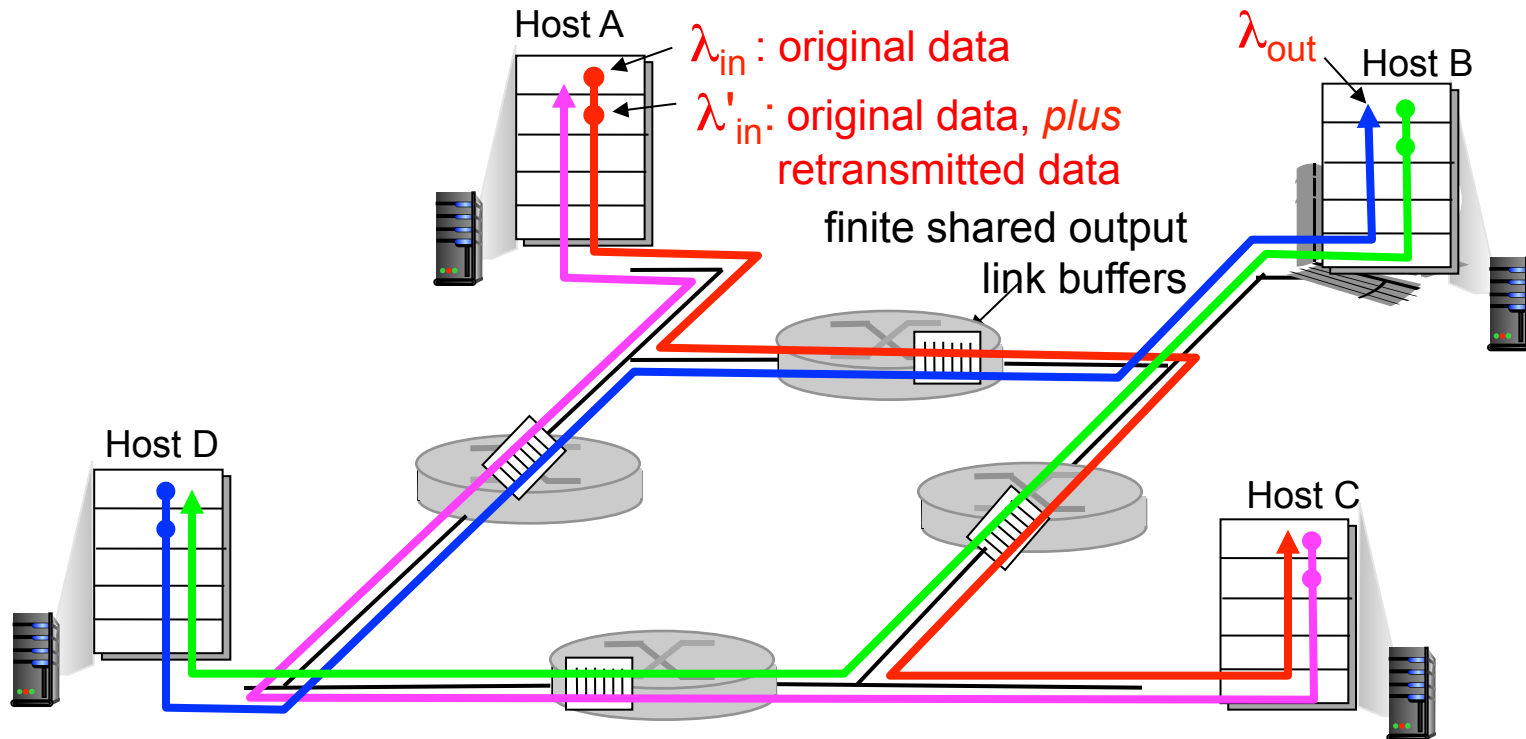
- more work (retrans) for given “goodput”
- unneeded retransmissions: link carries multiple copies of pkt
 - decreasing goodput

Causes/costs of congestion: scenario 3

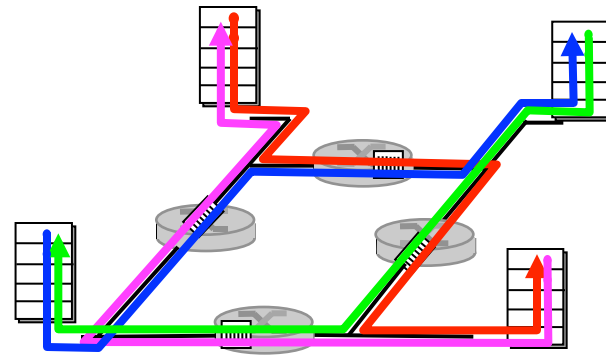
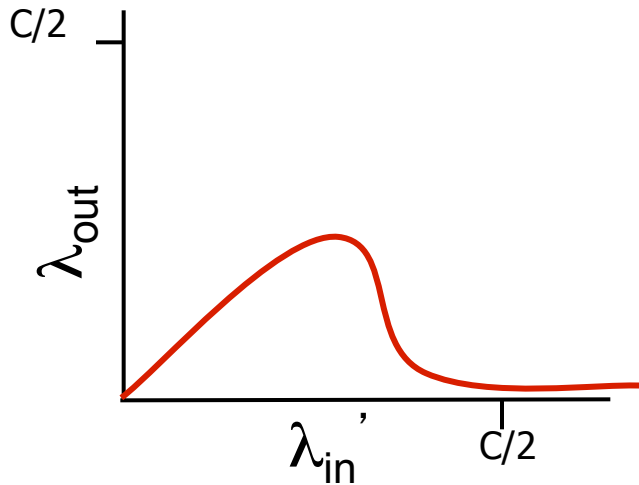
- four senders
- multihop paths
- timeout/retransmit

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!

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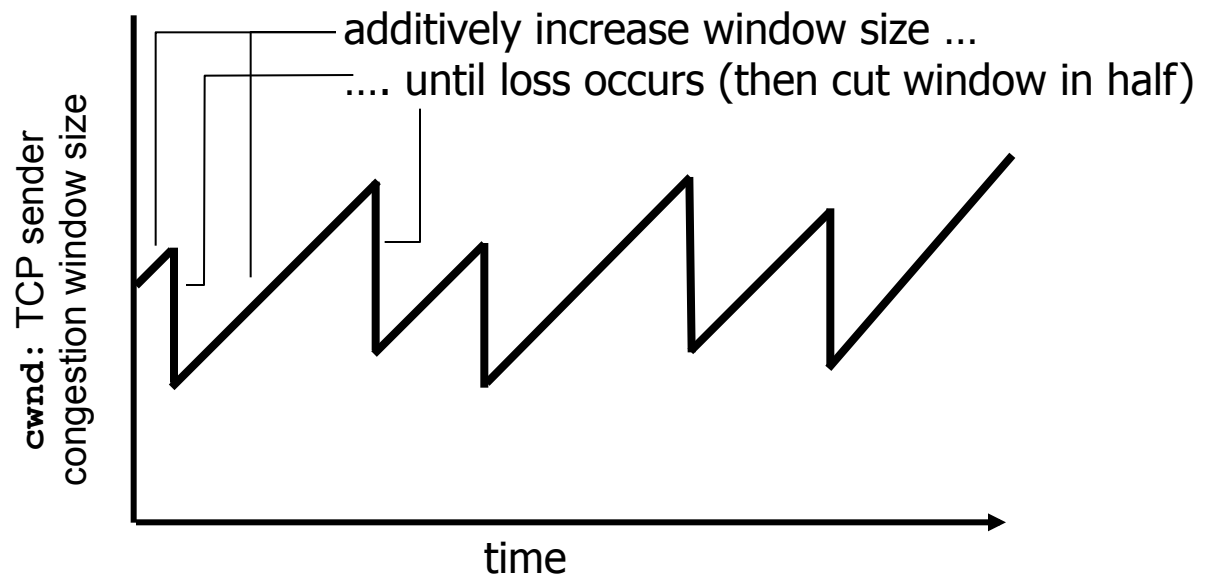
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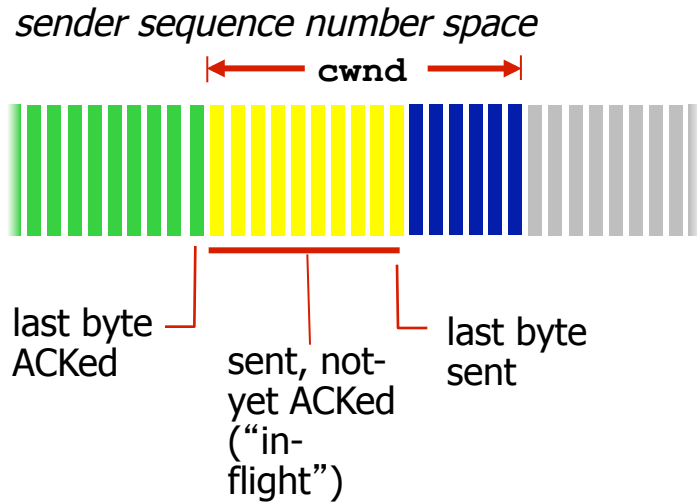
TCP congestion control: additive increase multiplicative decrease

- *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:

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

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

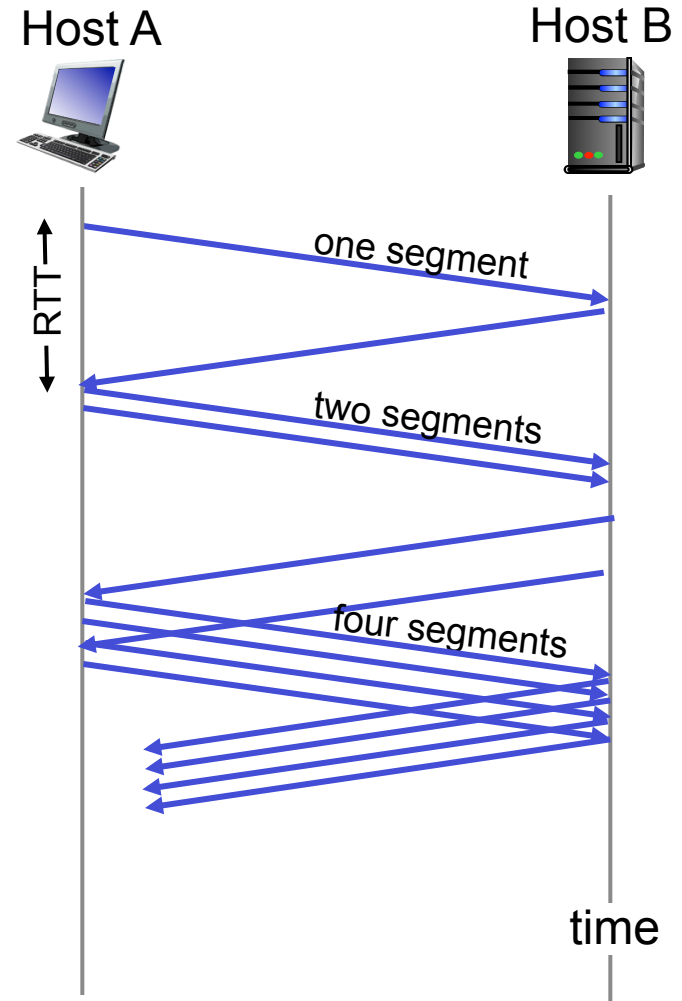
TCP sending rate:

- 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:
 - `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)

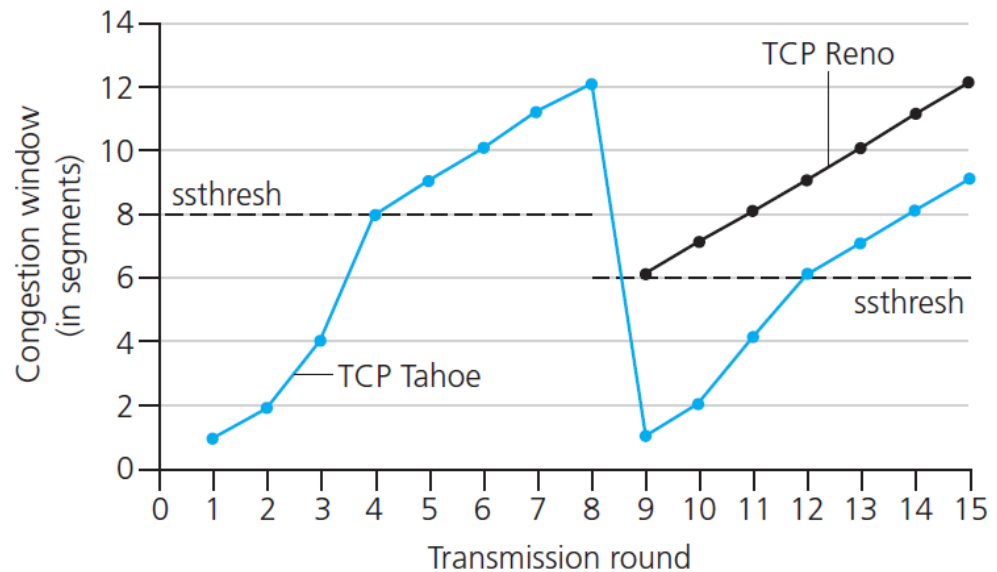
TCP: switching from slow start to CA

Q: when should the exponential increase switch to linear?

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

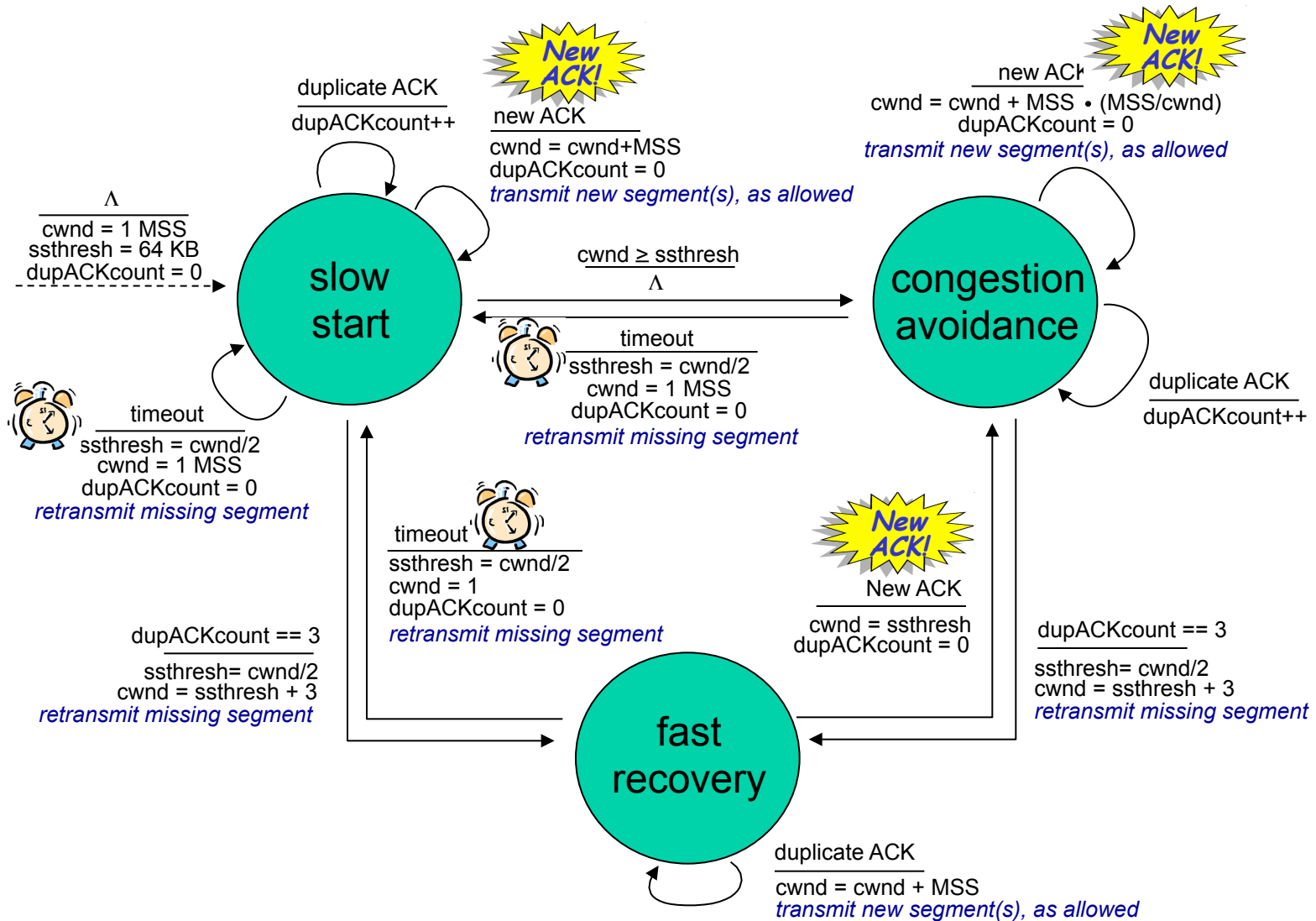
Implementation:

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



* Check out the online interactive exercises for more examples: http://gaia.cs.umass.edu/kurose_ross/interactive/

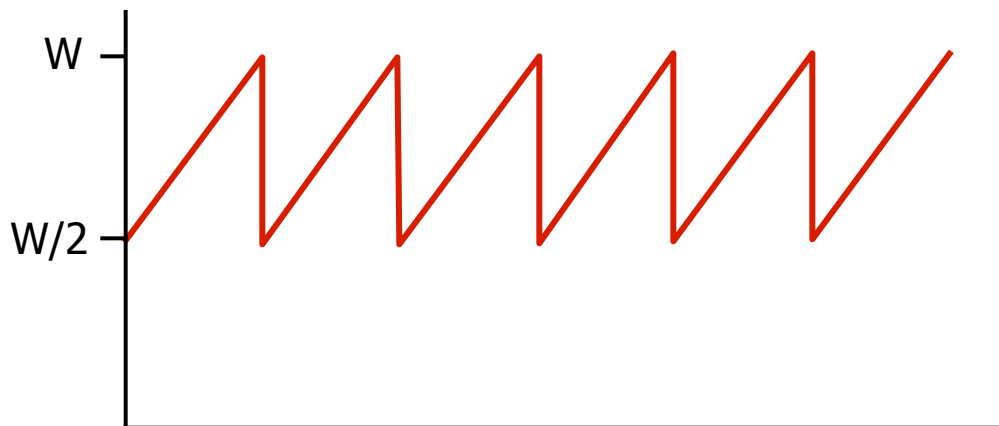
Summary: TCP Congestion Control



TCP throughput

- avg. TCP thrupt 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 $\frac{3}{4} W$
 - avg. thrupt is $\frac{3}{4}W$ per RTT

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



TCP Futures: TCP over “long, fat pipes”

- example: 1500 byte segments, 100ms RTT, want 10 Gbps throughput
- requires $W = 83,333$ in-flight segments
- throughput in terms of segment loss probability, L
[Mathis 1997]:

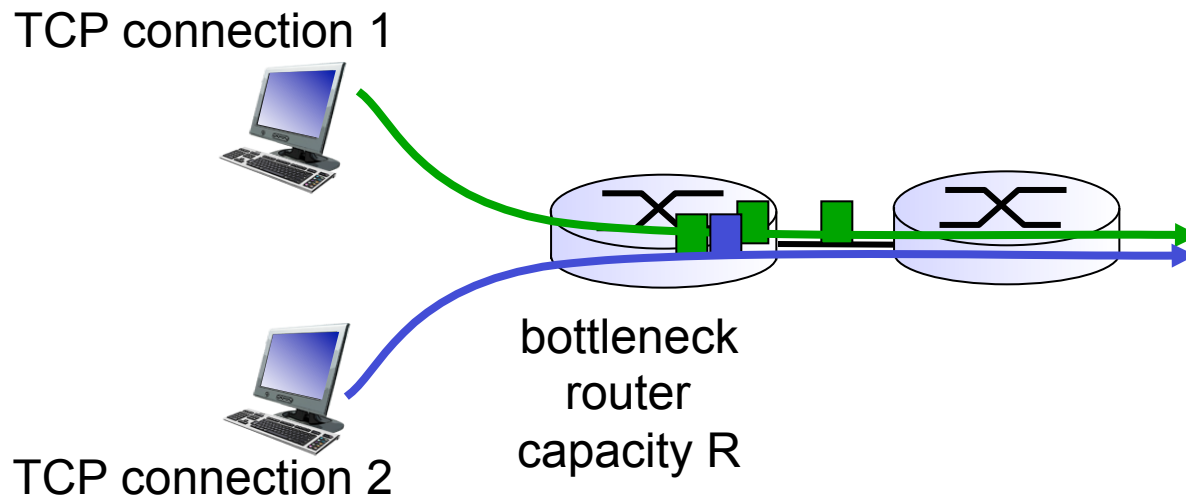
$$\text{TCP throughput} = \frac{1.22 \cdot \text{MSS}}{\text{RTT} \sqrt{L}}$$

→ to achieve 10 Gbps throughput, need a loss rate of $L = 2 \cdot 10^{-10}$ – *a very small loss rate!*

- new versions of TCP for high-speed

TCP Fairness

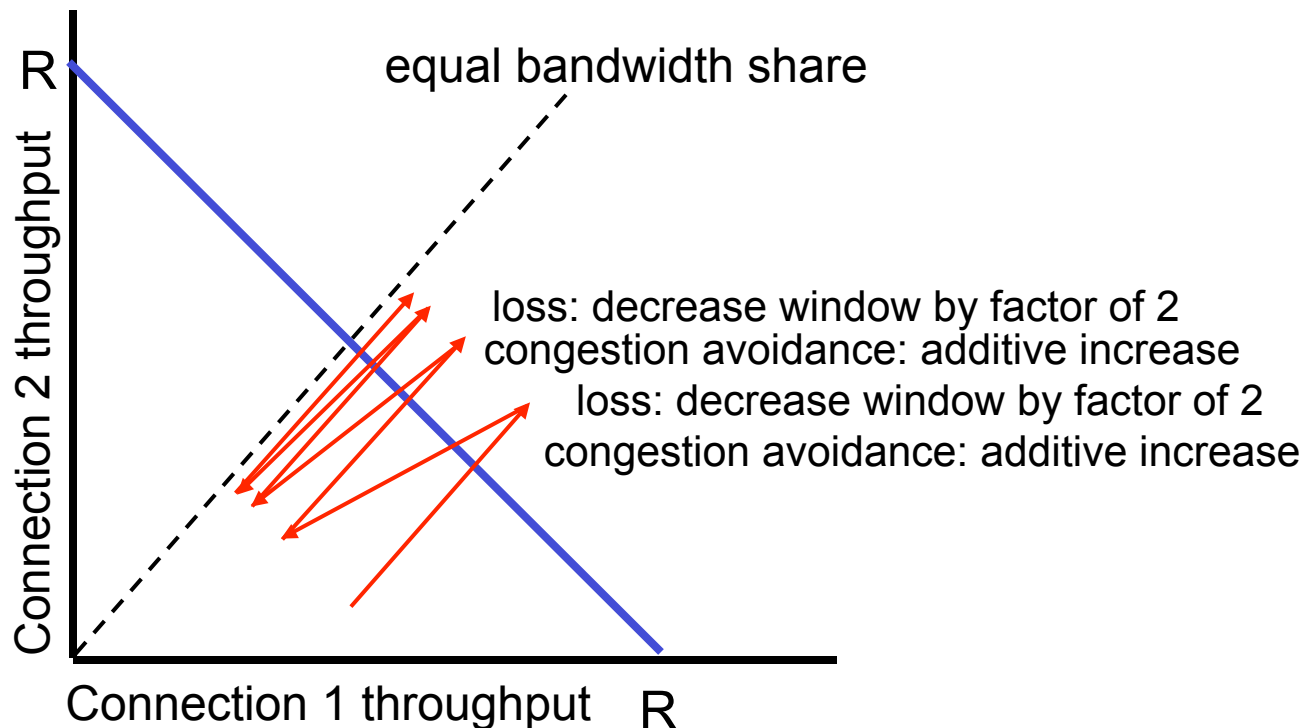
fairness goal: if K TCP sessions share same bottleneck link of bandwidth R , each should have average rate of R/K



Why is TCP fair?

two competing sessions:

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

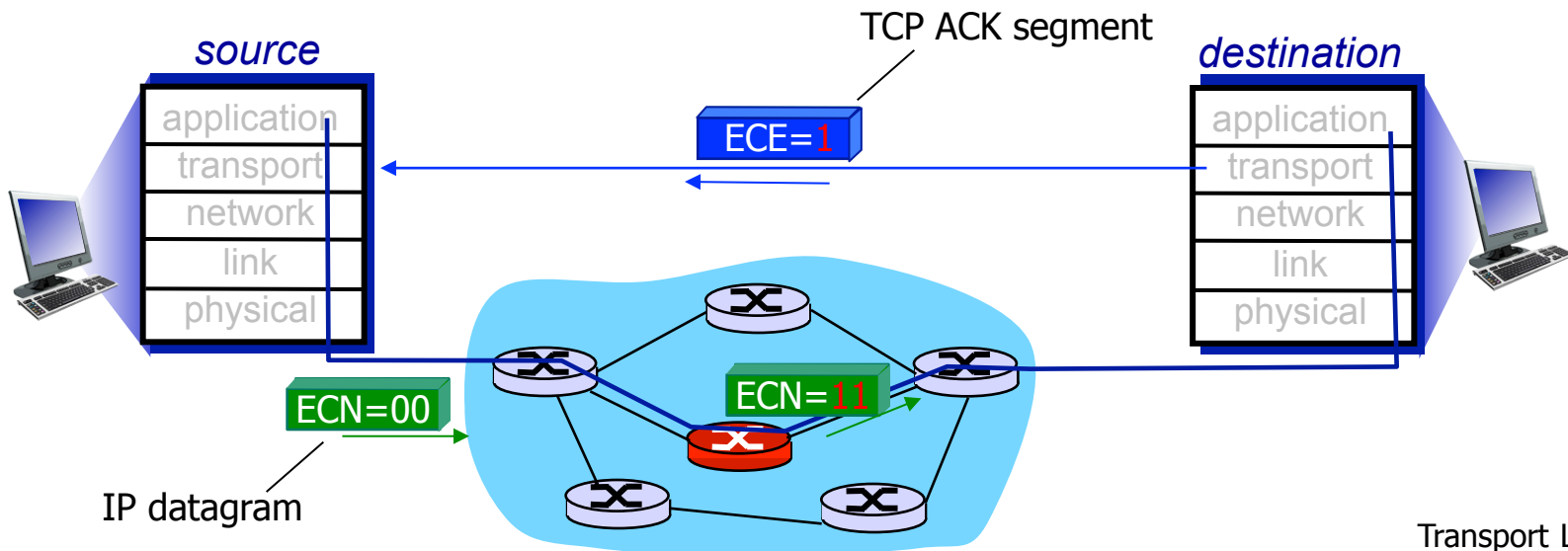
Fairness, parallel TCP connections

- 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



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