CSC358 *Intro. to Computer Networks*

Lecture 7: *TCP, flow and congestion control*

Amir H. Chinaei, Winter 2016

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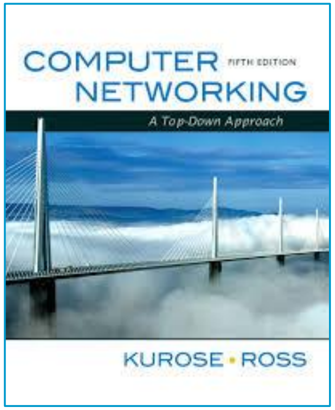

Many slides are (inspired/adapted) from the above source

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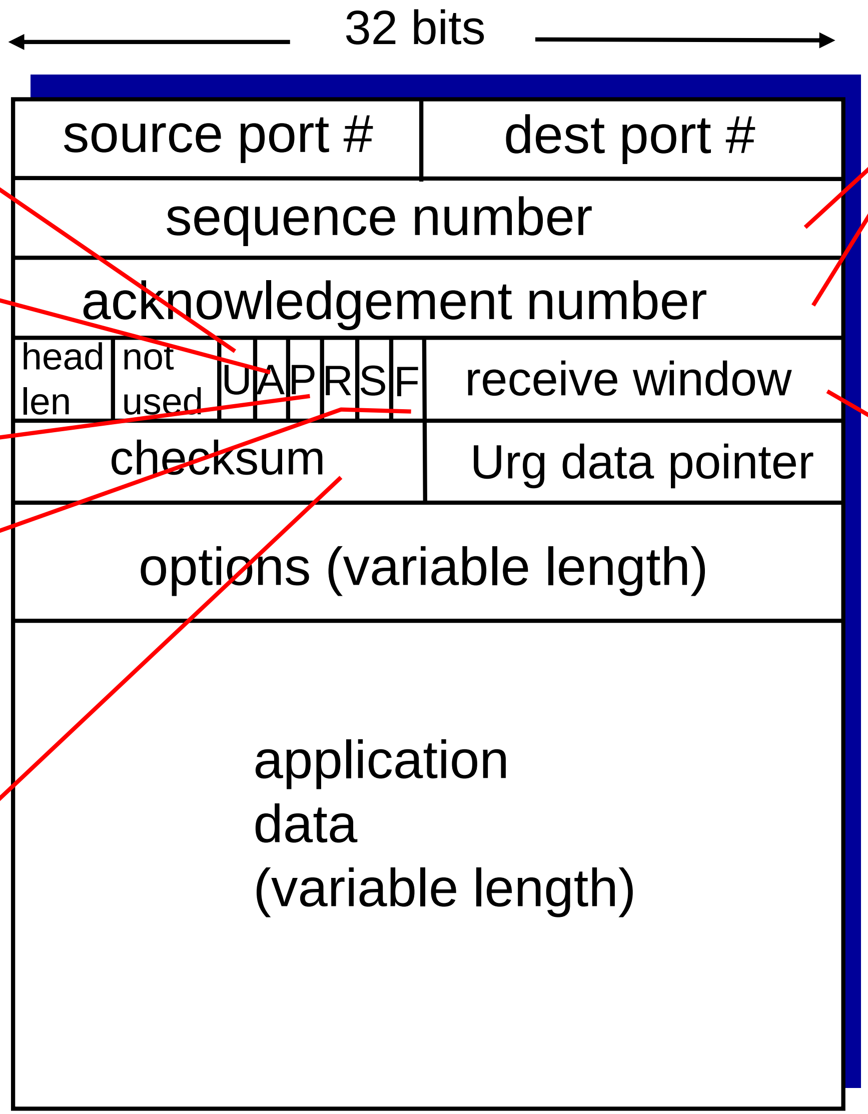
TCP: Overview RFCs: 793,1122,1323, 2018, 2581

- point-to-point:
  - one sender, one receiver
- reliable, in-order *byte steam:*
  - no “message boundaries”
- pipelined:
  - TCP congestion and flow control set window size
- full duplex data:
  - bi-directional data flow in same connection
  - MSS: maximum segment size
- connection-oriented:
  - handshaking (exchange of control msgs) inits sender, receiver state before data exchange
- flow controlled:
  - sender will not overwhelm receiver

TCP segment structure



URG: urgent data (generally not used)

ACK: ACK # valid

PSH: push data now (generally not used)

RST, SYN, FIN: connection estab (setup, teardown commands)

Internet checksum (as in UDP)

counting by bytes of data (not segments!)

bytes rcvr willing to accept

TCP seq. numbers, ACKs

sequence numbers:

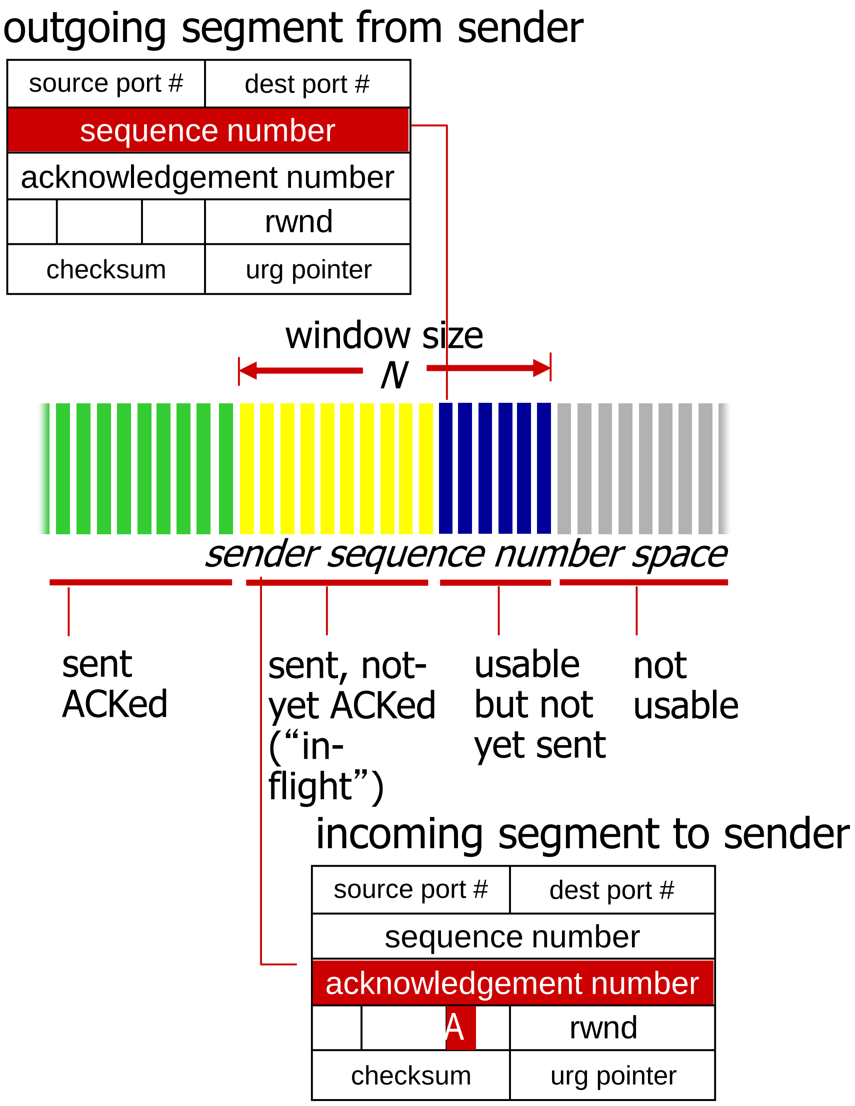
- byte stream "number" of first byte in segment's data

acknowledgements:

- seq # of next byte expected from other side
- cumulative ACK

Q: how receiver handles out-of-order segments

- A: TCP spec doesn't say, - up to implementor



outgoing segment from sender

source port #	dest port #
sequence number	
acknowledgement number	
	rwnd
checksum	urg pointer

window size

N

sender sequence number space

sent ACKed

sent, not-yet ACKed ("in-flight")

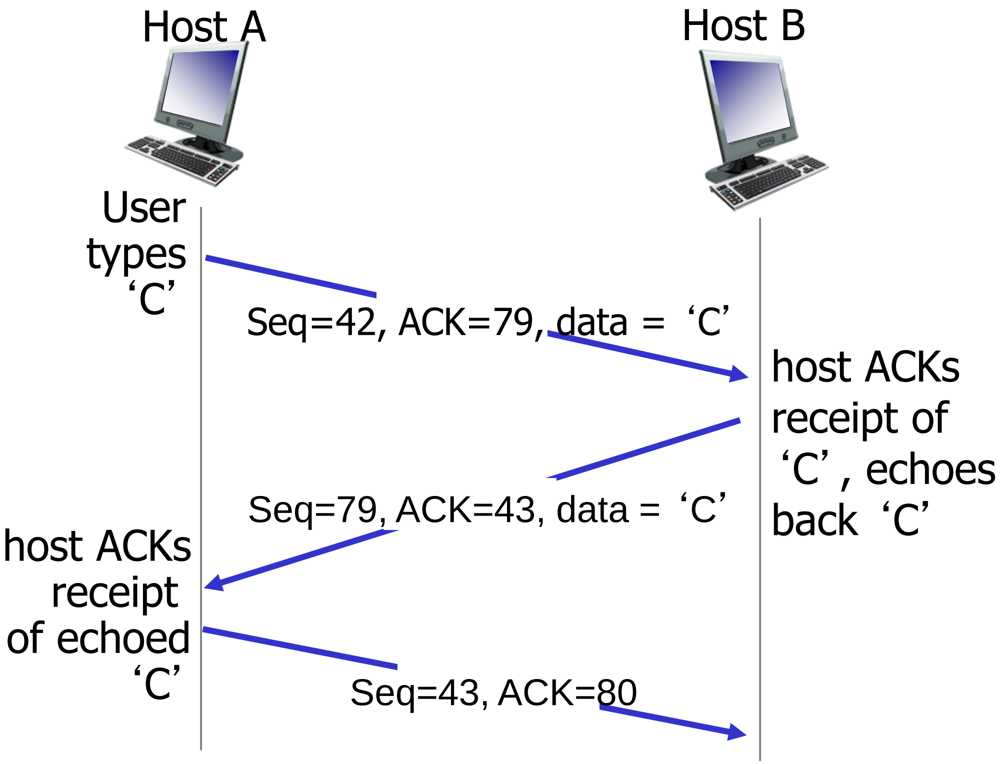
usable but not yet sent

not usable

incoming segment to sender

source port #	dest port #
sequence number	
acknowledgement number	
A	rwnd
checksum	urg pointer

TCP seq. numbers, ACKs



Host A

Host B

User
types
'C'

Seq=42, ACK=79, data = 'C'

host ACKs
receipt of
'C', echoes
back 'C'

Seq=79, ACK=43, data = 'C'

host ACKs
receipt
of echoed
'C'

Seq=43, ACK=80

simple telnet scenario

TCP round trip time, timeout

Q: how to set TCP timeout value?

- longer than RTT
  - but RTT varies
- *too short:* premature timeout, unnecessary retransmissions
- *too long:* slow reaction to segment loss

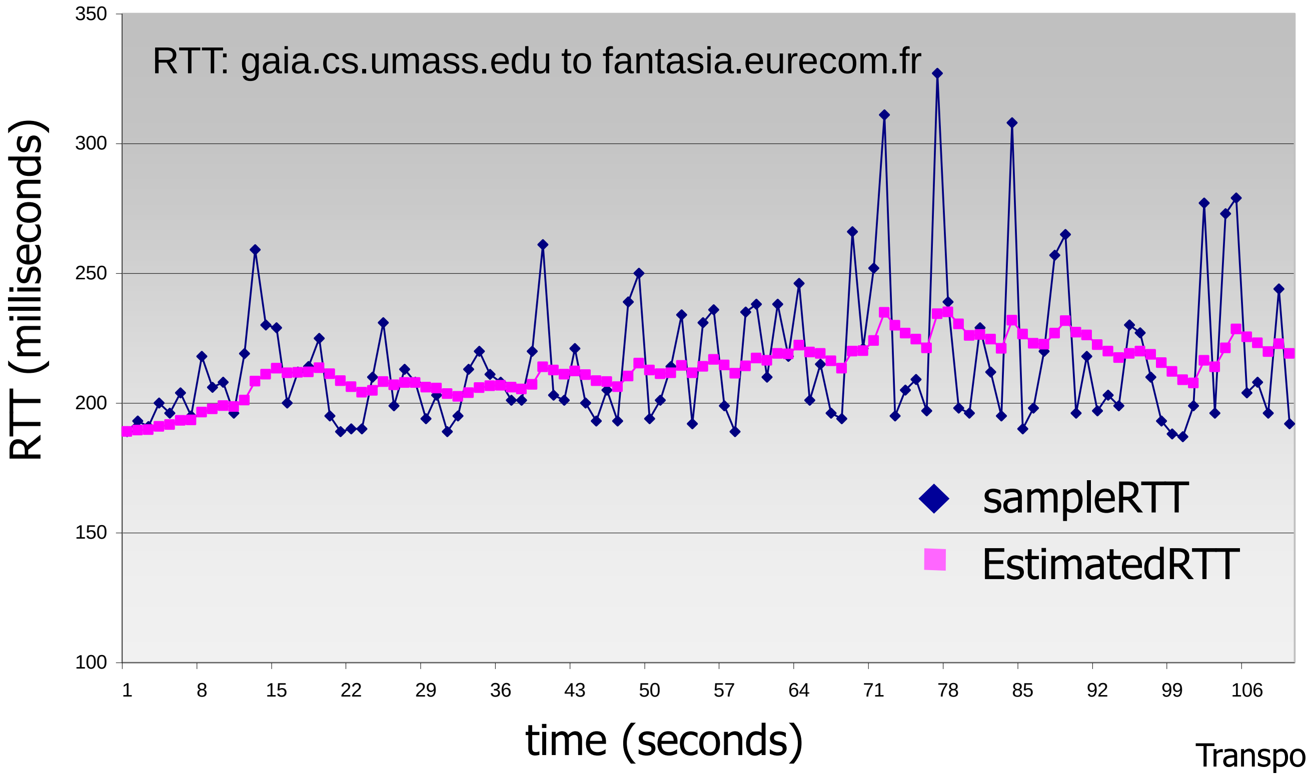
Q: how to estimate RTT?

- **`SampleRTT`**: measured time from segment transmission until ACK receipt
  - ignore retransmissions
- **`SampleRTT`** will vary, want estimated RTT "smoother"
  - average several *recent* measurements, not just current **`SampleRTT`**

TCP round trip time, timeout

`EstimatedRTT = (1- α)*EstimatedRTT + α*SampleRTT`

- exponential weighted moving average
- influence of past sample decreases exponentially fast
- typical value: $\alpha = 0.125$



TCP round trip time, timeout

- timeout interval: **`EstimatedRTT`** plus "safety margin"
  - large variation in **`EstimatedRTT`** **->** larger safety margin
- estimate SampleRTT deviation from EstimatedRTT:

```
DevRTT = (1-β)*DevRTT +
                 β*|SampleRTT-EstimatedRTT|

                (typically, β = 0.25)
```

```
TimeoutInterval = EstimatedRTT + 4*DevRTT
```

↑ estimated RTT

↑ "safety margin"

TCP reliable data transfer

- TCP creates rdt service on top of IP's unreliable service
  - pipelined segments
  - cumulative acks
  - single retransmission timer
- retransmissions triggered by:
  - timeout events
  - duplicate acks

let's initially consider simplified TCP sender:

- ignore duplicate acks
- ignore flow control, congestion control

TCP sender events:

data rcvd from app:

- create segment with seq #
- seq # is byte-stream number of first data byte in segment
- start timer if not already running
  - think of timer as for oldest unacked segment
  - expiration interval: **`TimeOutInterval`**

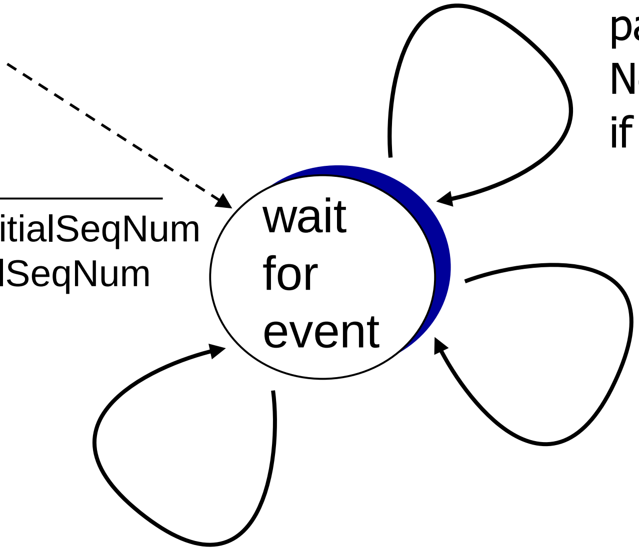
timeout:

- retransmit segment that caused timeout
- restart timer

ack rcvd:

- if ack acknowledges previously unacked segments
  - update what is known to be ACKed
  - start timer if there are still unacked segments

TCP sender (simplified)



Λ

```
NextSeqNum = InitialSeqNum
SendBase = InitialSeqNum
```

data received from application above

```
create segment, seq. #: NextSeqNum
pass segment to IP (i.e., “send”)
NextSeqNum = NextSeqNum + length(data)
if (timer currently not running)
    start timer
```

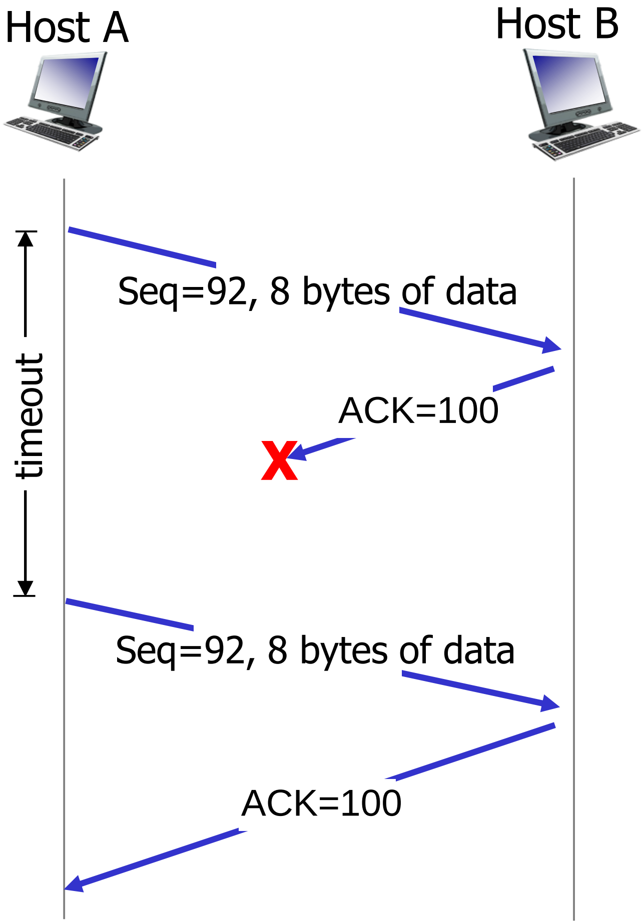
timeout

```
retransmit not-yet-acked segment
        with smallest seq. #
start timer
```

ACK received, with ACK field value y

```
if (y > SendBase) {
   SendBase = y
   /* SendBase–1: last cumulatively ACKed byte */
   if (there are currently not-yet-acked segments)
       start timer
   else stop timer
 }
```

TCP: retransmission scenarios



Host A

Host B

timeout

Seq=92, 8 bytes of data

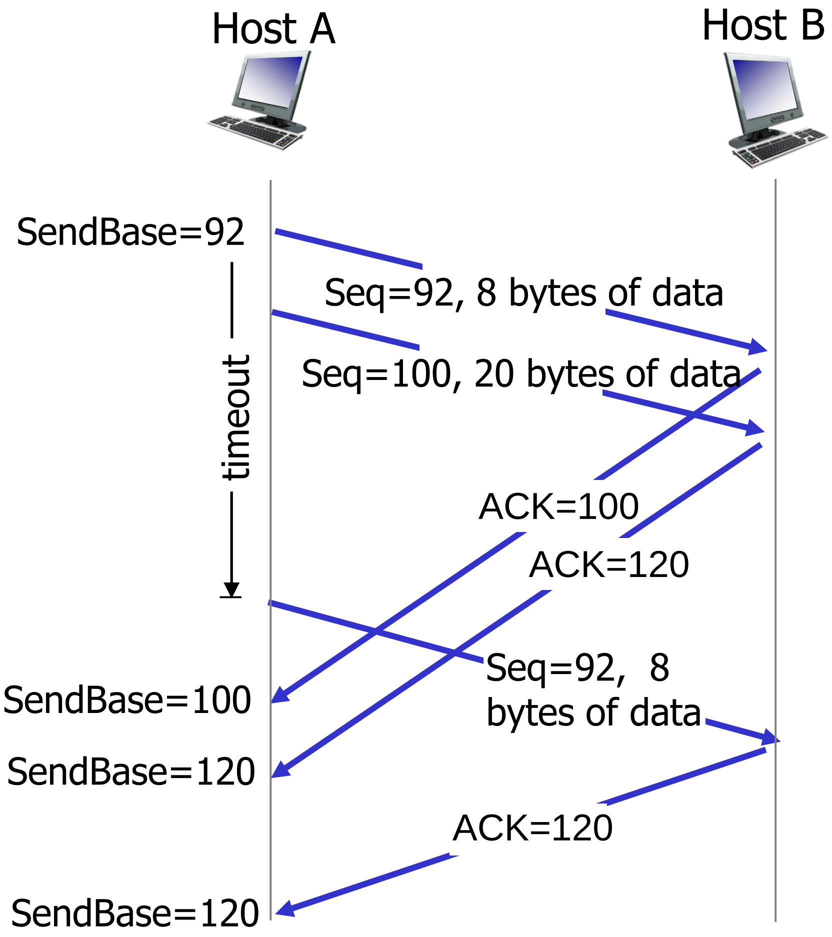
ACK=100

X

Seq=92, 8 bytes of data

ACK=100

lost ACK scenario



Host A

Host B

SendBase=92

timeout

Seq=92, 8 bytes of data

Seq=100, 20 bytes of data

ACK=100

ACK=120

Seq=92, 8 bytes of data

SendBase=100

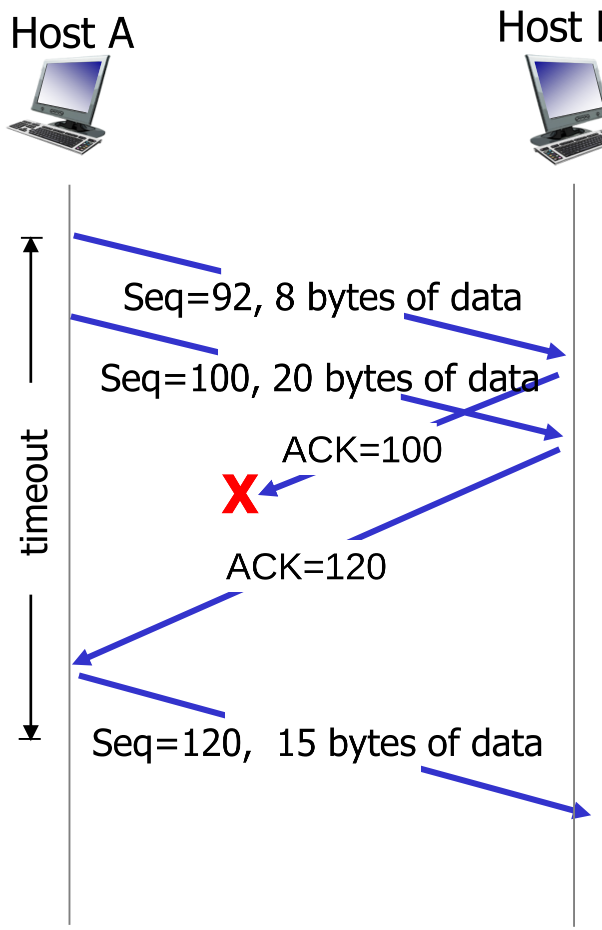
SendBase=120

ACK=120

SendBase=120

premature timeout

TCP: retransmission scenarios



Host A

Host B

timeout

Seq=92, 8 bytes of data

Seq=100, 20 bytes of data

ACK=100

X

ACK=120

Seq=120, 15 bytes of data

cumulative ACK

TCP ACK generation [RFC 1122, RFC 2581]

event at receiver	*TCP receiver action*
arrival of in-order segment with expected seq #. All data up to expected seq # already ACKed	delayed ACK. Wait up to 500ms for next segment. If no next segment, send ACK
arrival of in-order segment with expected seq #. One other segment has ACK pending	immediately send single cumulative ACK, ACKing both in-order segments
arrival of out-of-order segment higher-than-expect seq. # . Gap detected	immediately send *duplicate ACK,* indicating seq. # of next expected byte
arrival of segment that partially or completely fills gap	immediate send ACK, provided that segment starts at lower end of gap

TCP fast retransmit

- time-out period often relatively long:
  - long delay before resending lost packet
- detect lost segments via duplicate ACKs.
  - sender often sends many segments back-to-back
  - if segment is lost, there will likely be many duplicate ACKs.

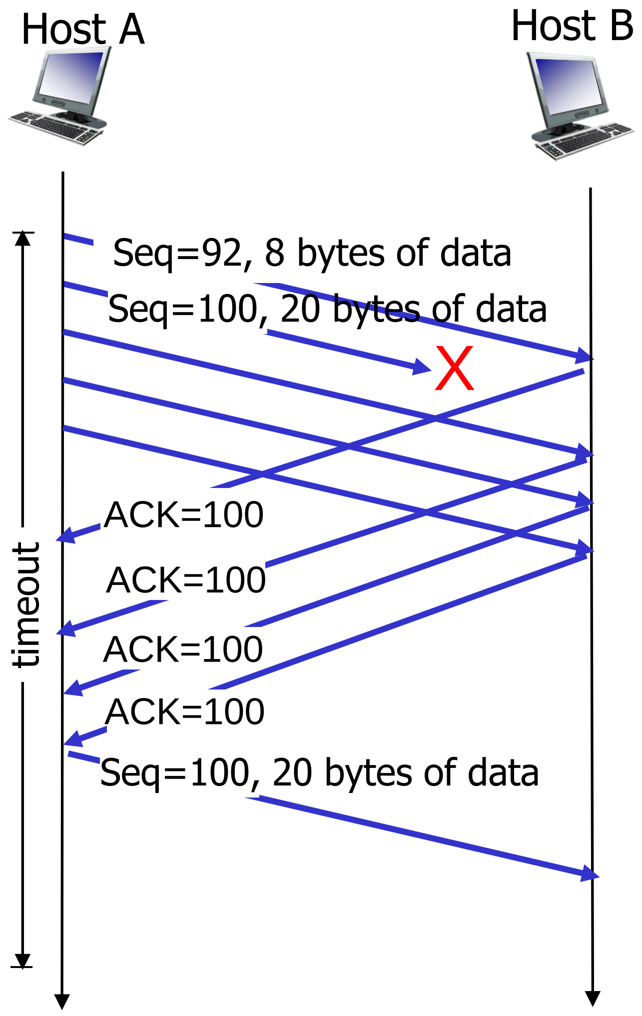
TCP fast retransmit

if sender receives 3 ACKs for same data

("triple duplicate ACKs"), resend unacked segment with smallest seq #

- likely that unacked segment lost, so don't wait for timeout

TCP fast retransmit

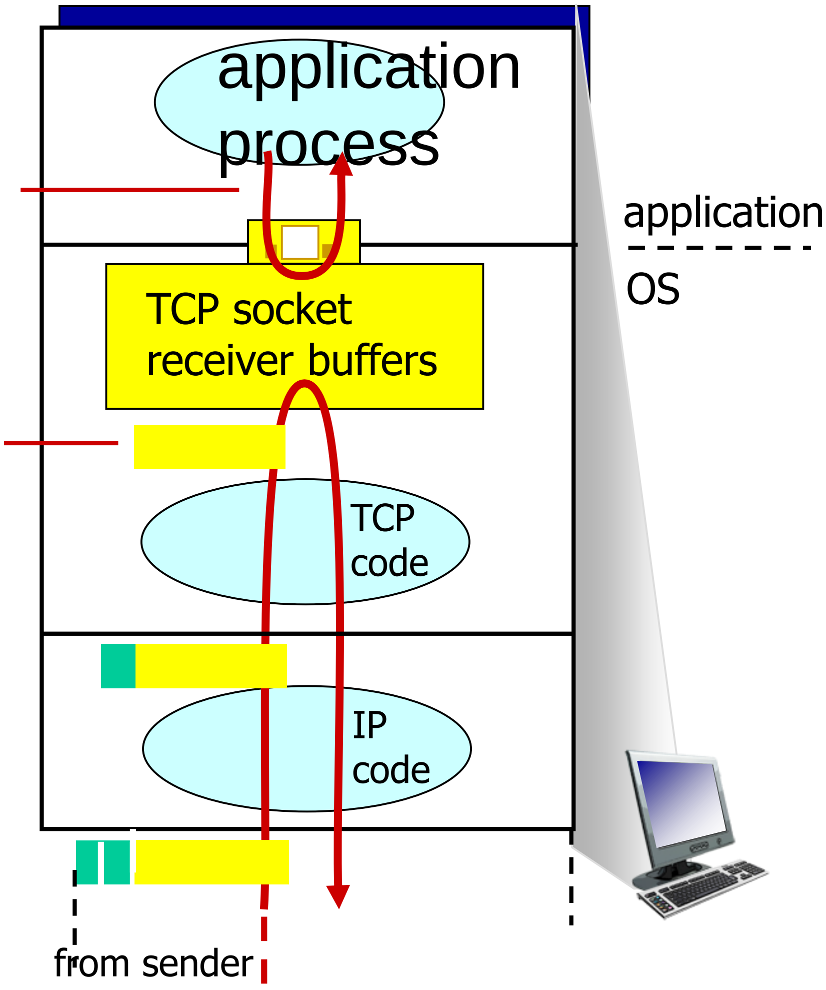


fast retransmit after sender receipt of triple duplicate ACK

TCP flow control

application may remove data from TCP socket buffers ….

… slower than TCP receiver is delivering (sender is sending)



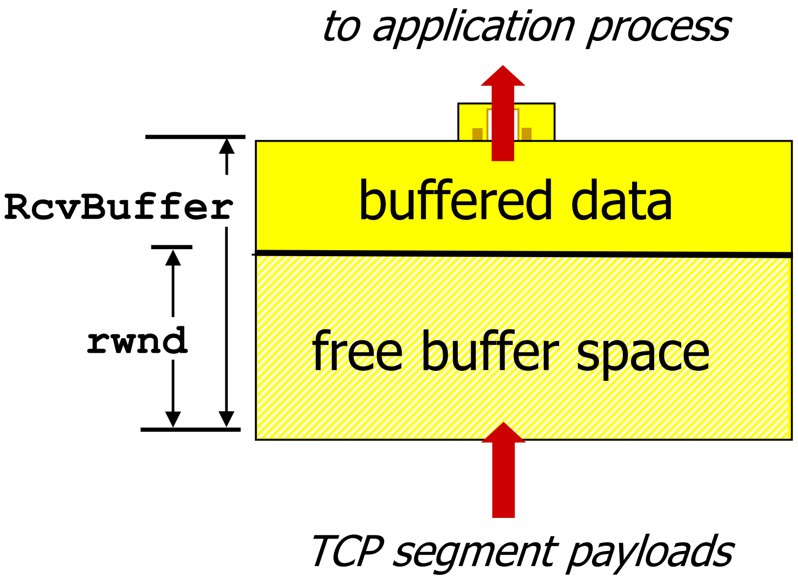
receiver protocol stack

flow control

receiver controls sender, so sender won't overflow receiver's buffer by transmitting too much, too fast

TCP flow control

- receiver "advertises" free buffer space by including **`rwnd`** value in TCP header of receiver-to-sender segments
  - **`RcvBuffer`** size set via socket options (typical default is 4096 bytes)
  - many operating systems autoadjust **`RcvBuffer`**
- sender limits amount of unacked ("in-flight") data to receiver's **`rwnd`** value
- guarantees receive buffer will not overflow

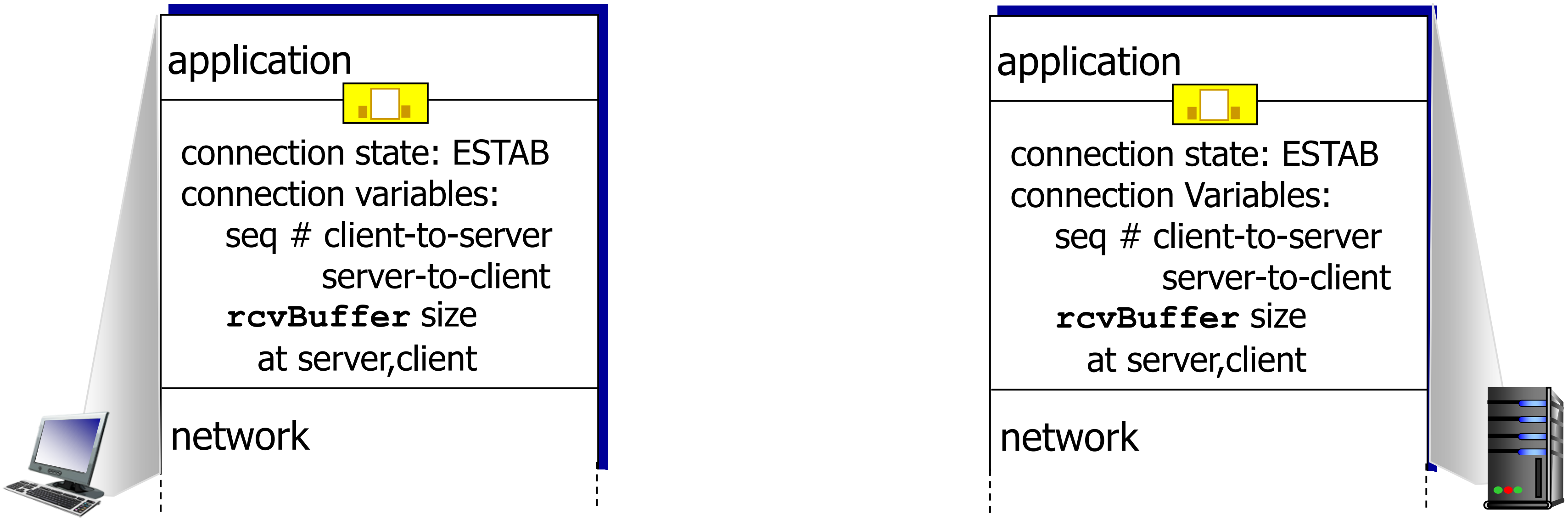


receiver-side buffering

Connection Management

before exchanging data, sender/receiver "handshake":

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



application

connection state: ESTAB
connection variables:
seq # client-to-server
server-to-client
rcvBuffer size
at server,client

network

application

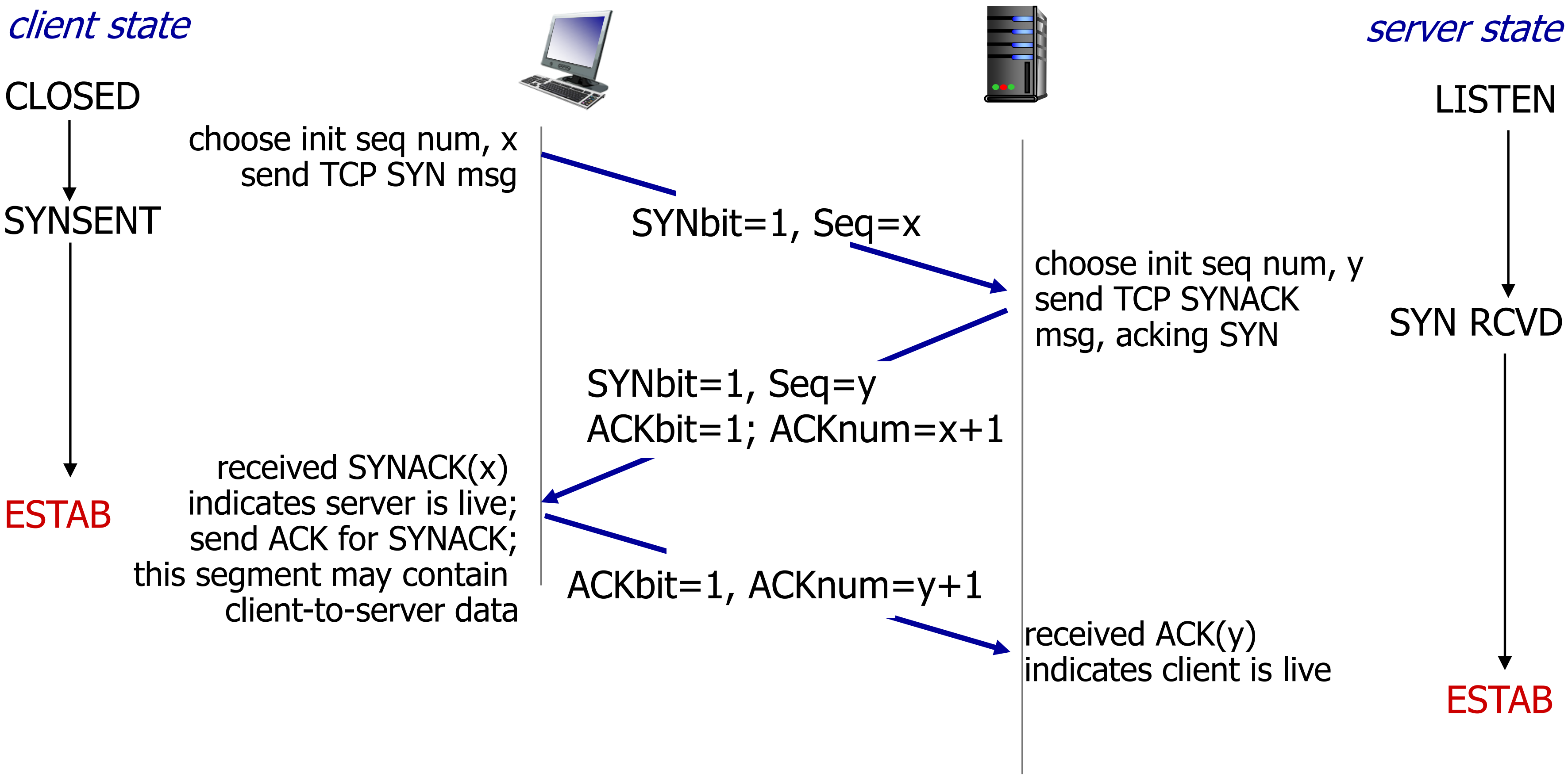
connection state: ESTAB
connection Variables:
seq # client-to-server
server-to-client
rcvBuffer size
at server,client

network

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

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

TCP 3-way handshake



client state

CLOSED

SYNSENT

ESTAB

choose init seq num, x
send TCP SYN msg

SYNbit=1, Seq=x

server state

LISTEN

choose init seq num, y
send TCP SYNACK
msg, acking SYN

SYN RCVD

SYNbit=1, Seq=y
ACKbit=1; ACKnum=x+1

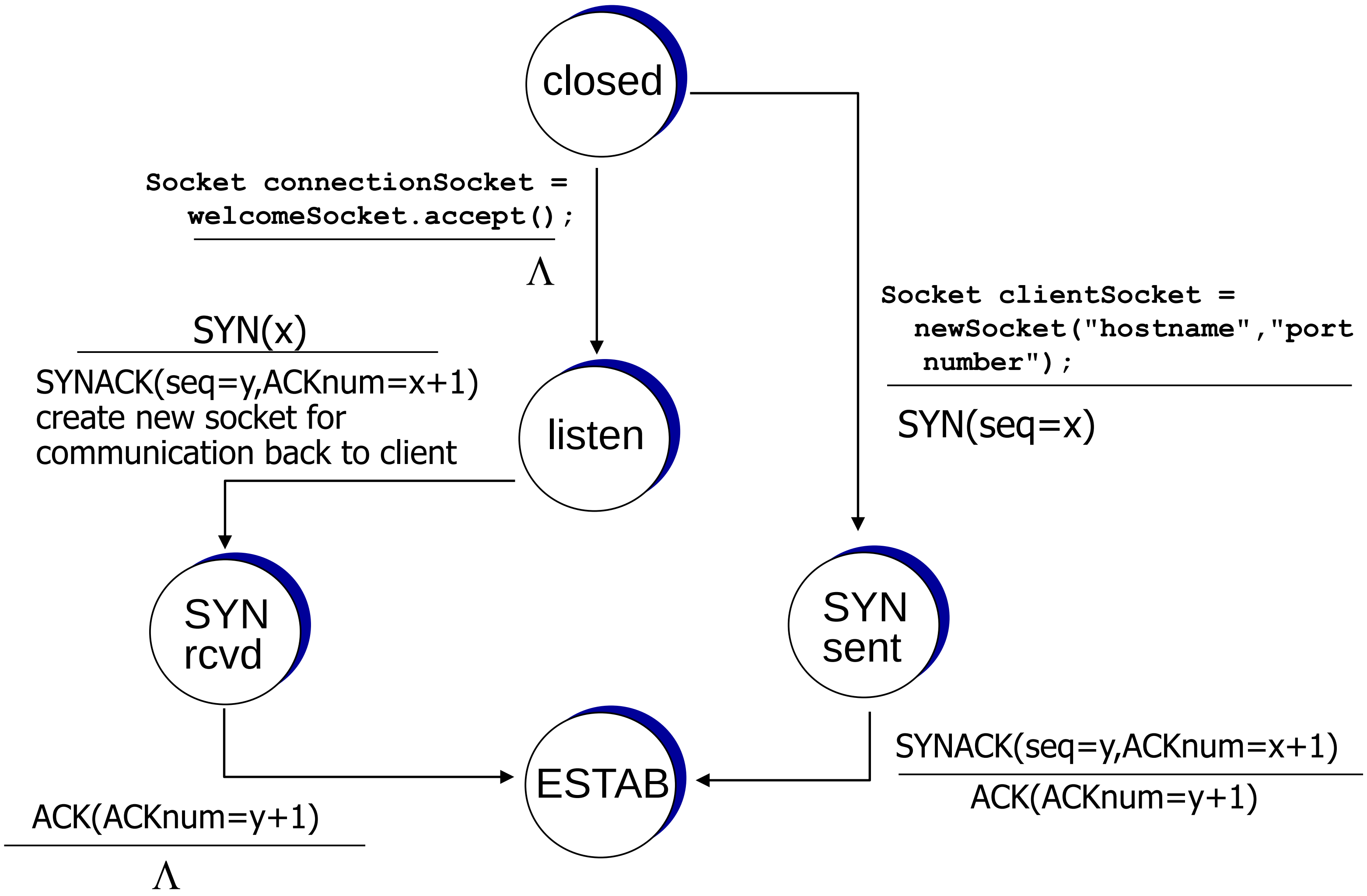
received SYNACK(x)
indicates server is live;
send ACK for SYNACK;
this segment may contain
client-to-server data

ACKbit=1, ACKnum=y+1

received ACK(y)
indicates client is live

ESTAB

TCP 3-way handshake: FSM



closed

`Socket connectionSocket = welcomeSocket.accept();`

Λ

listen

SYN(x)

SYNACK(seq=y,ACKnum=x+1)
create new socket for
communication back to client

SYN rcvd

ACK(ACKnum=y+1)

Λ

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

SYN(seq=x)

SYN sent

SYNACK(seq=y,ACKnum=x+1)

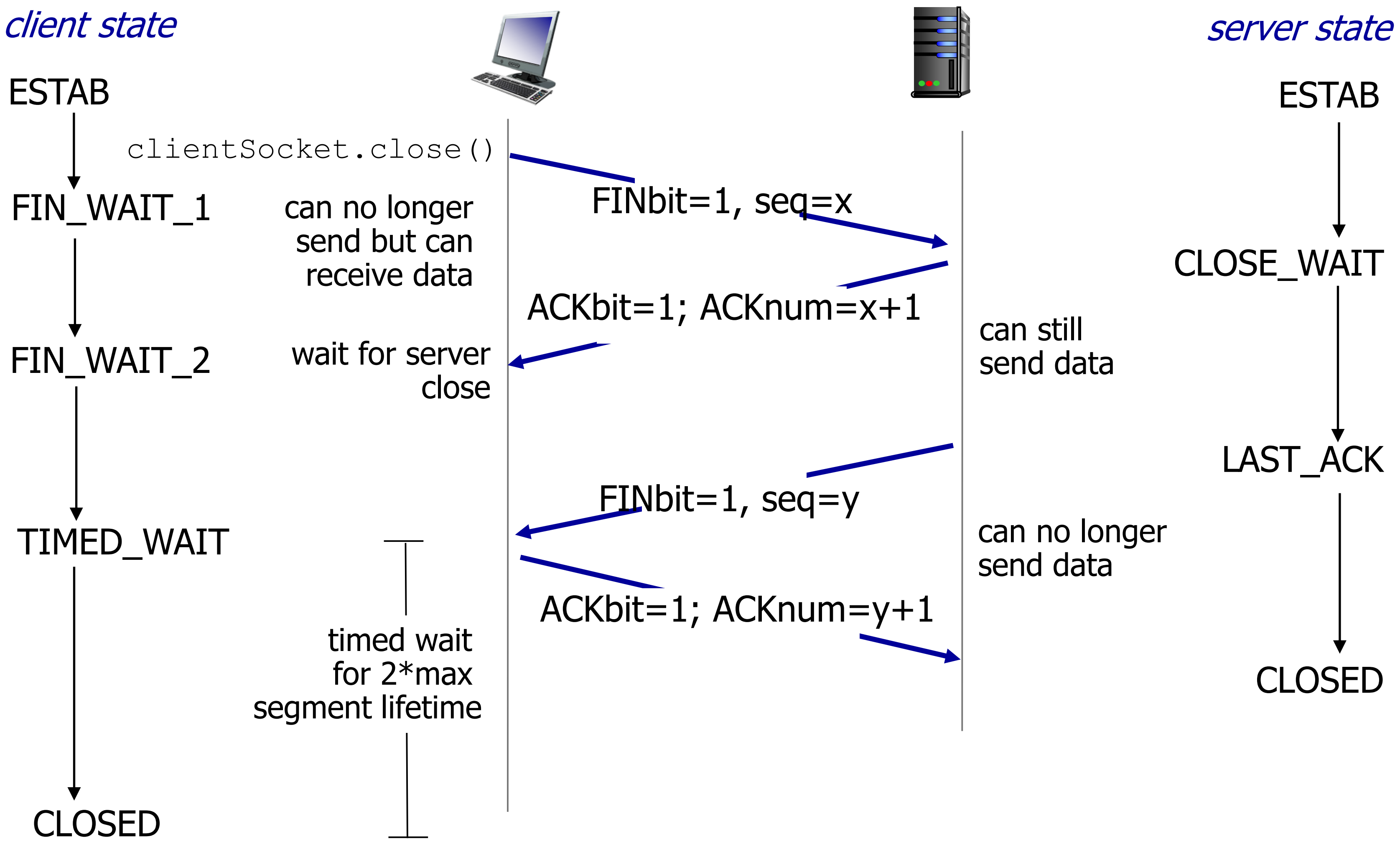
ACK(ACKnum=y+1)

ESTAB

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*max segment lifetime

CLOSED

FINbit=1, seq=x

ACKbit=1; ACKnum=x+1

FINbit=1, seq=y

ACKbit=1; ACKnum=y+1

server state

ESTAB

CLOSE_WAIT

can still send data

LAST_ACK

can no longer send data

CLOSED

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

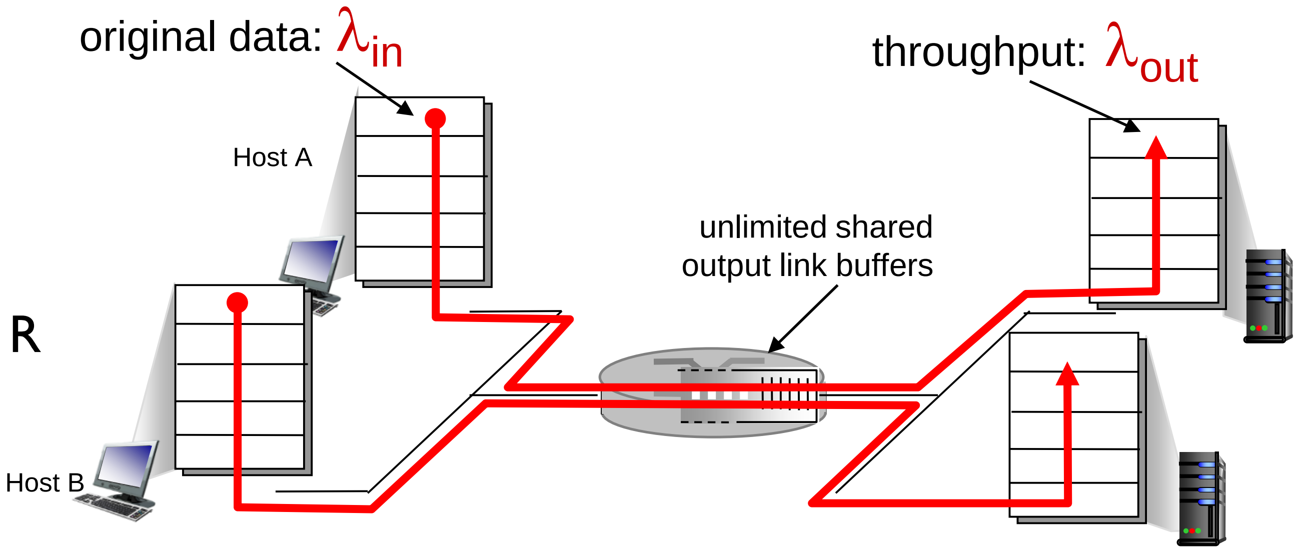
Principles of congestion control

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
- output link capacity: R
- no retransmission



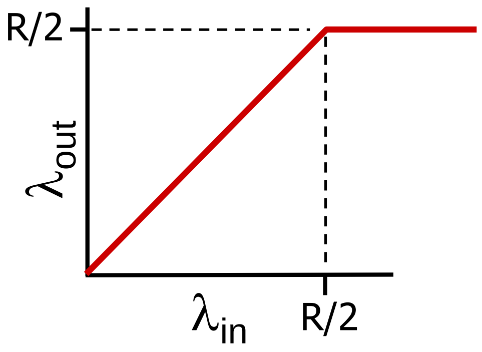
original data: λ_{in}

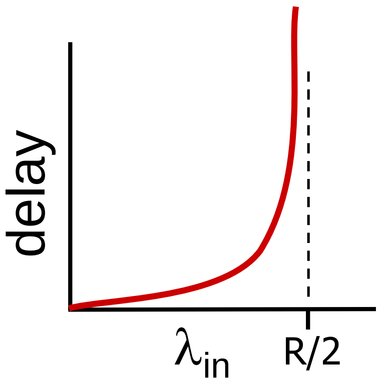
throughput: λ_{out}

Host A

Host B

unlimited shared output link buffers

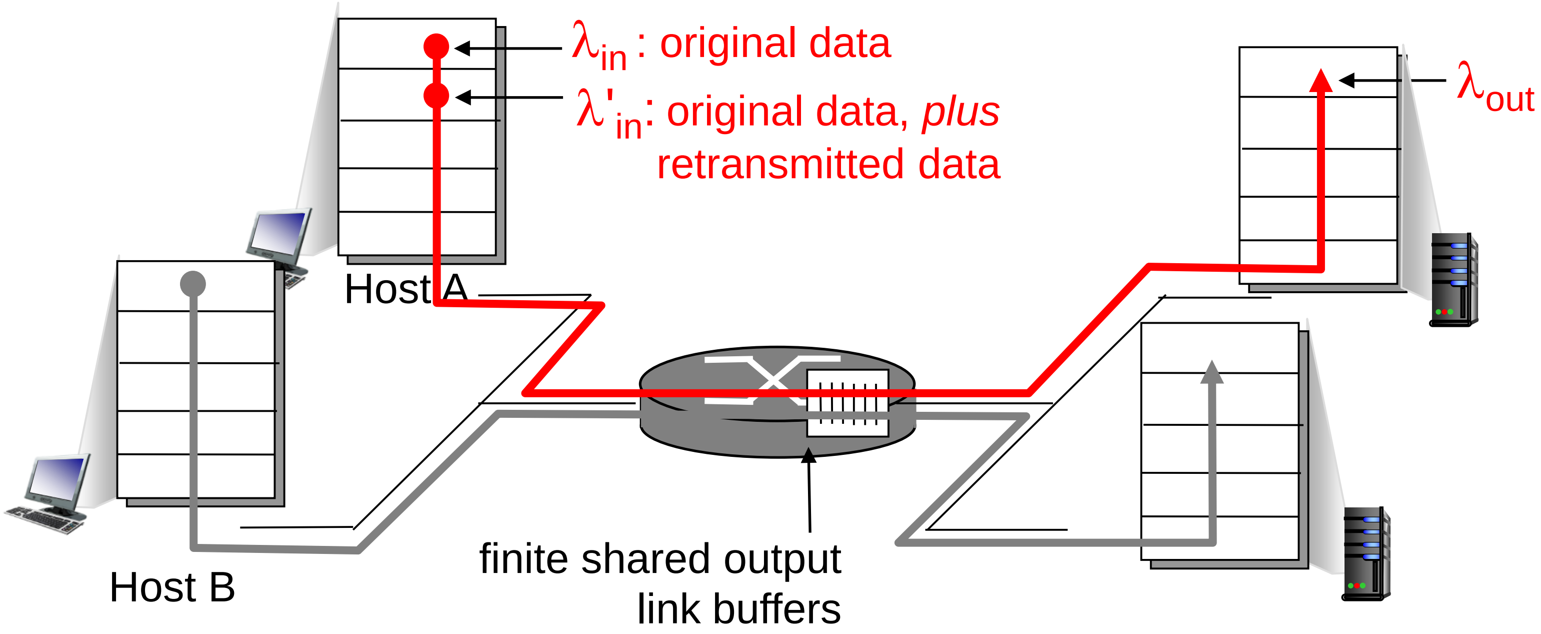




- maximum per-connection throughput: R/2
- large delays as arrival rate, λ_{in}, approaches capacity

Causes/costs of congestion: scenario 2

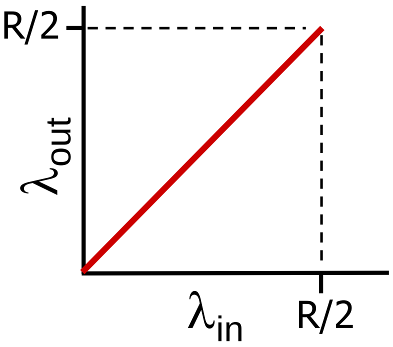
- one router, *finite* buffers
- sender retransmission of timed-out packet
  - application-layer input = application-layer output: $\lambda_{in} = \lambda_{out}$
  - transport-layer input includes *retransmissions* : $\lambda'_{in} \geq \lambda_{in}$

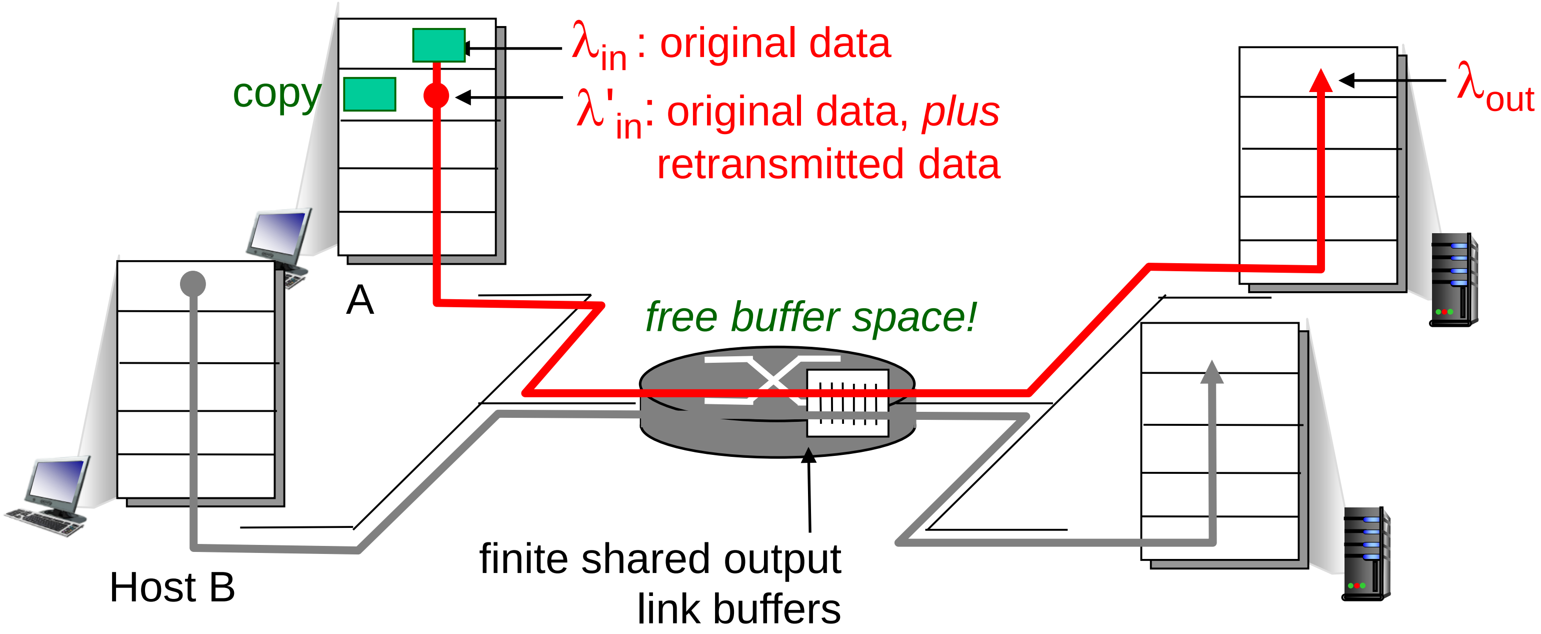


Causes/costs of congestion: scenario 2

idealization: perfect knowledge

- sender sends only when router buffers available



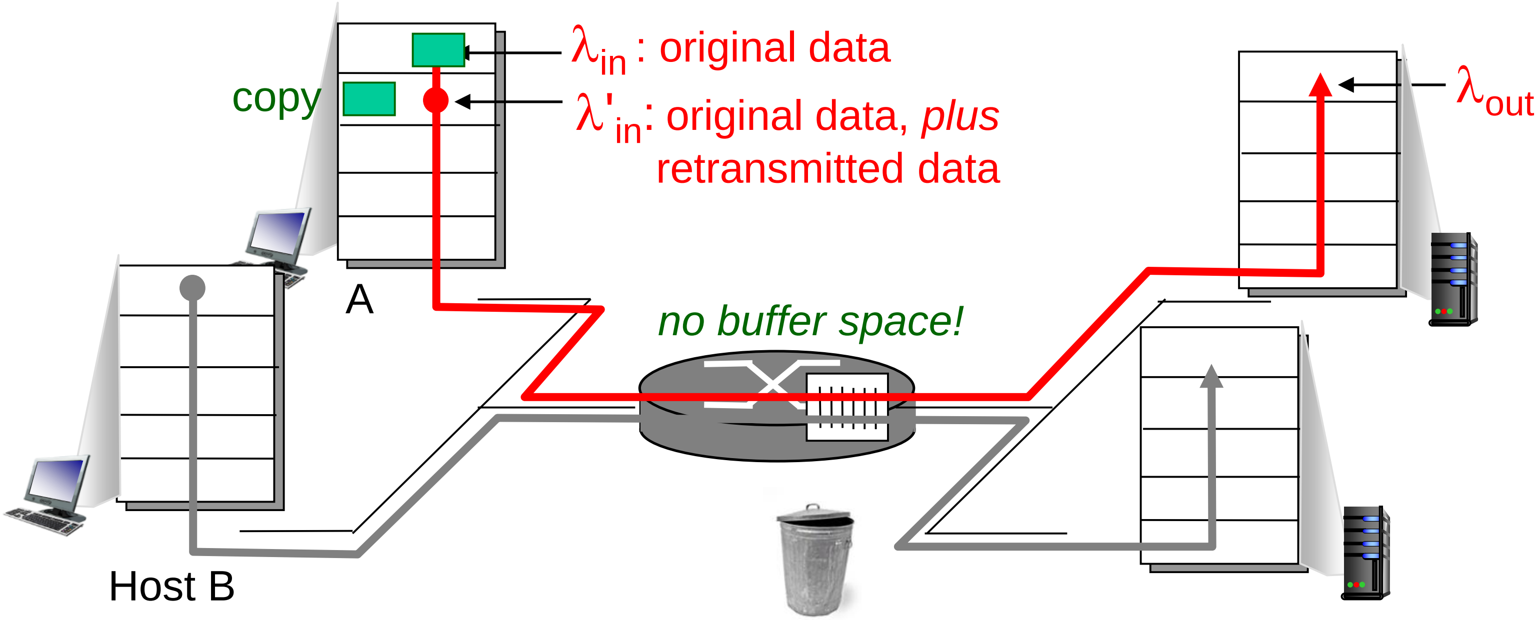


Causes/costs of congestion: scenario 2

Idealization: known loss

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

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



λ_{in} : original data

λ'_{in}: original data, *plus* retransmitted data

λ_{out}

Causes/costs of congestion: scenario 2

Idealization: known loss

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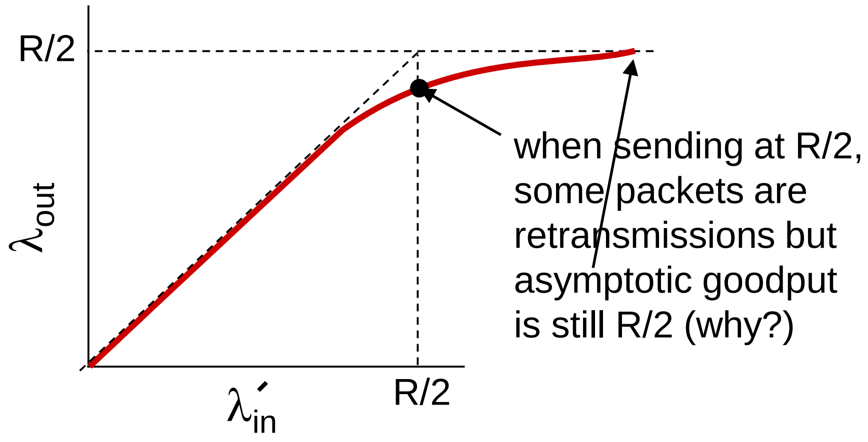
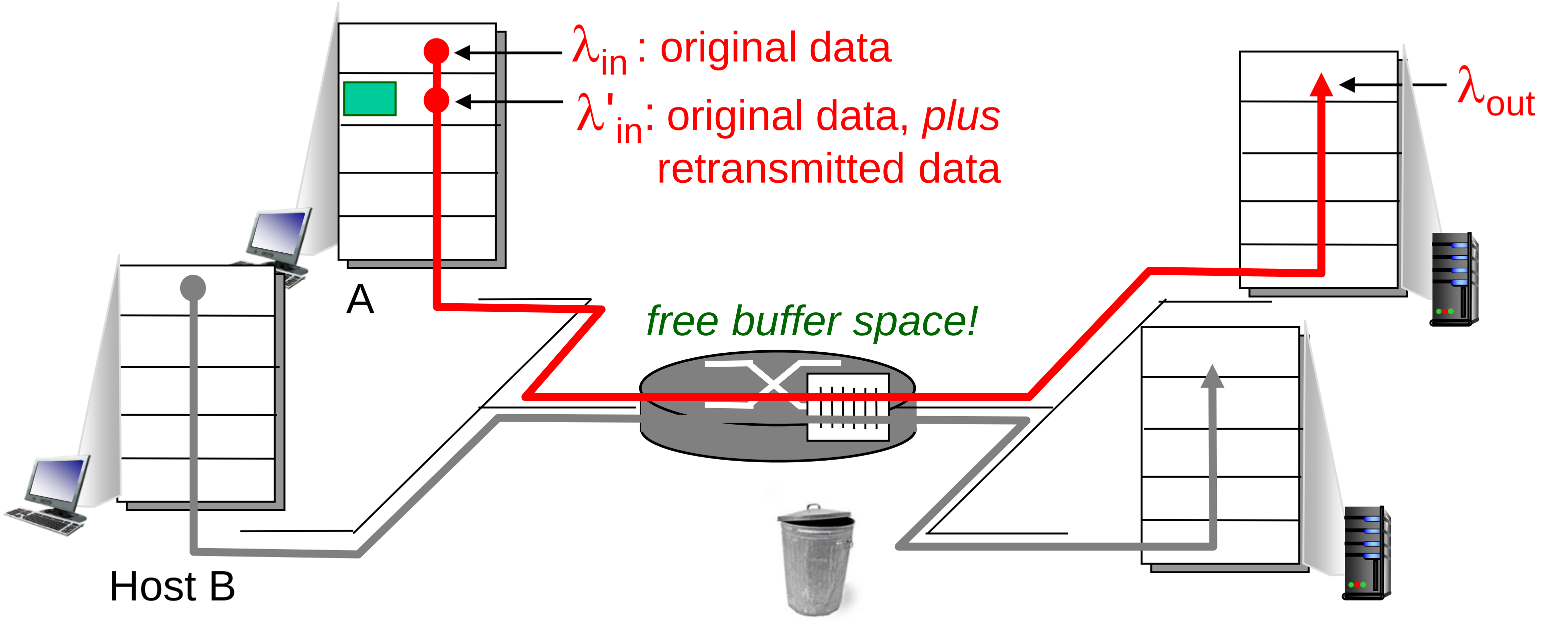


when sending at R/2, some packets are retransmissions but asymptotic goodput is still R/2 (why?)



λ_{in} : original data

λ'_{in}: original data, *plus* retransmitted data

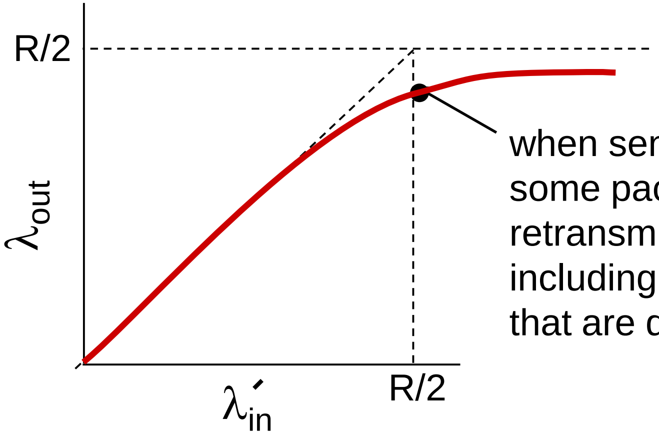
λ_{out}

free buffer space!

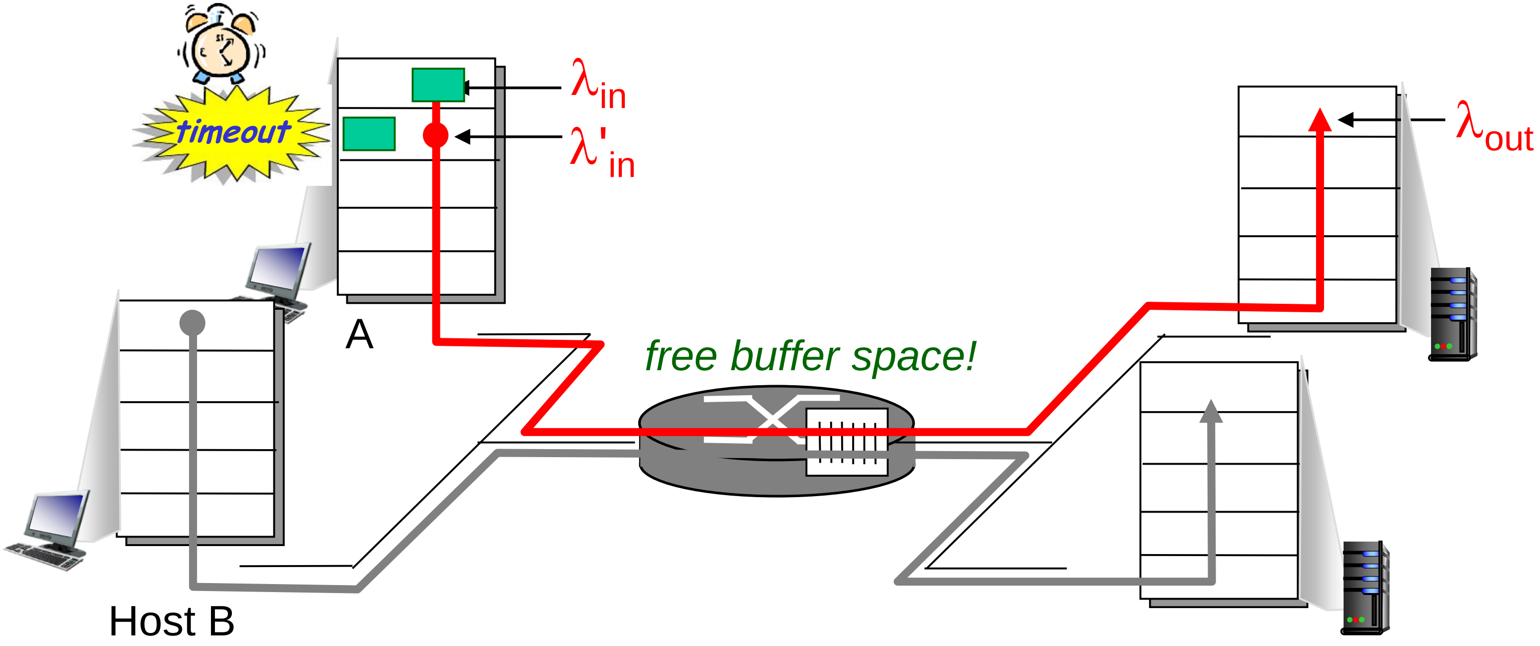
Causes/costs of congestion: scenario 2

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



when sending at R/2, some packets are retransmissions including duplicated that are delivered!



Causes/costs of congestion: scenario 2

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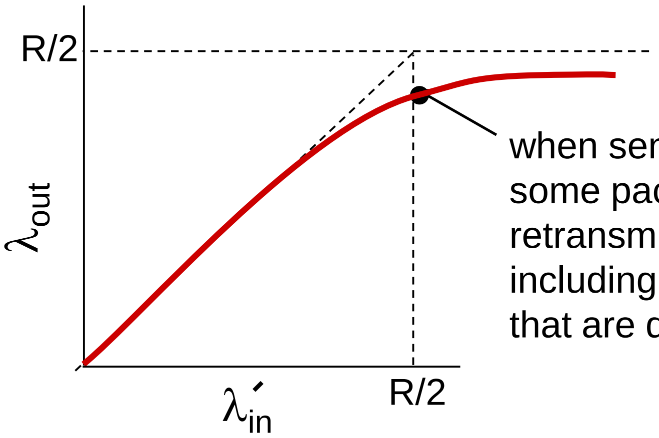


when sending at R/2, some packets are retransmissions including duplicated that are delivered!

"costs" of congestion:

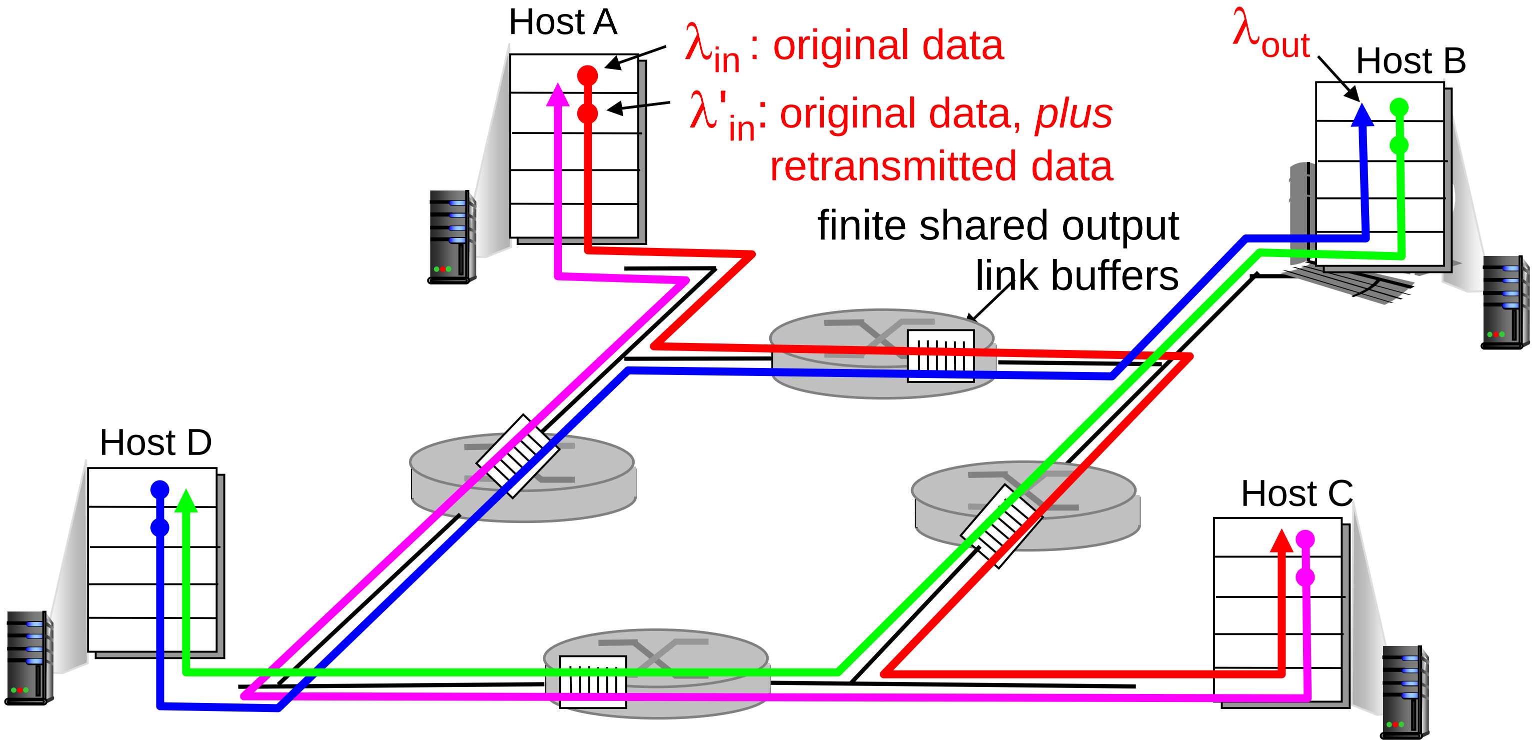
- more work (retrans) for given "goodput"
- unneeded retransmissions: link carries multiple copies of pkt
  - decreasing goodput

Causes/costs of congestion: scenario 3

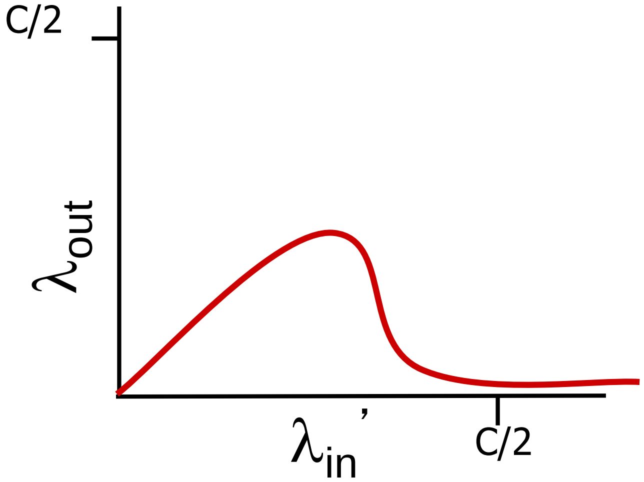
- four senders
- multihop paths
- timeout/retransmit

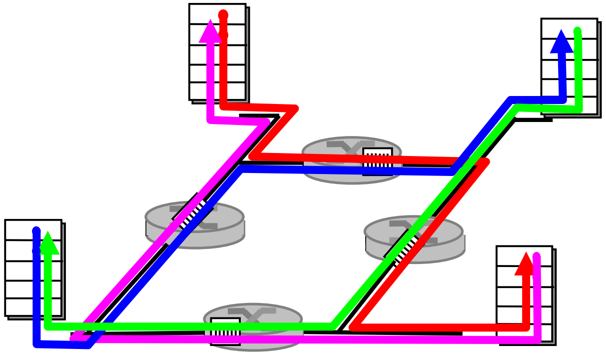
Q: what happens as λ_{in} and λ_{in}' increase ?

A: as red λ_{in}' increases, all arriving blue pkts at upper queue are dropped, blue throughput → 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 (SNA, DECbit, TCP/IP ECN, ATM)
  - explicit rate for sender to send at

Case study: ATM ABR congestion control

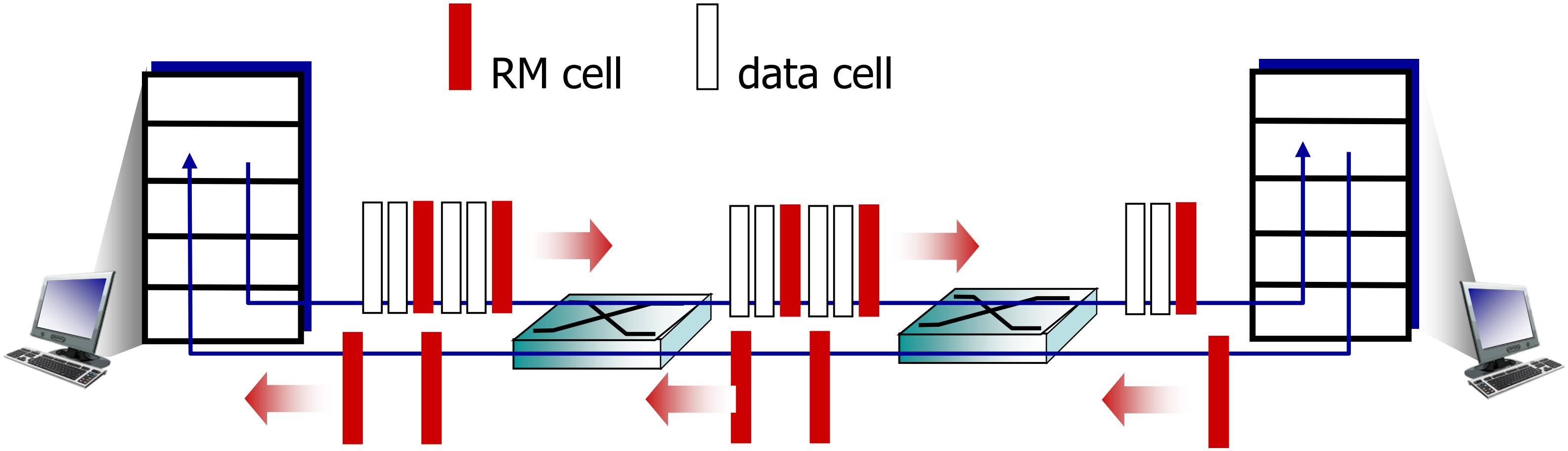
ABR: available bit rate:

- "elastic service"
- if sender's path "underloaded":
  - sender should use available bandwidth
- if sender's path congested:
  - sender throttled to minimum guaranteed rate

RM (resource management) cells:

- sent by sender, interspersed with data cells
- bits in RM cell set by switches ("*network-assisted*")
  - *NI bit:* no increase in rate (mild congestion)
  - *CI bit:* congestion indication
- RM cells returned to sender by receiver, with bits intact

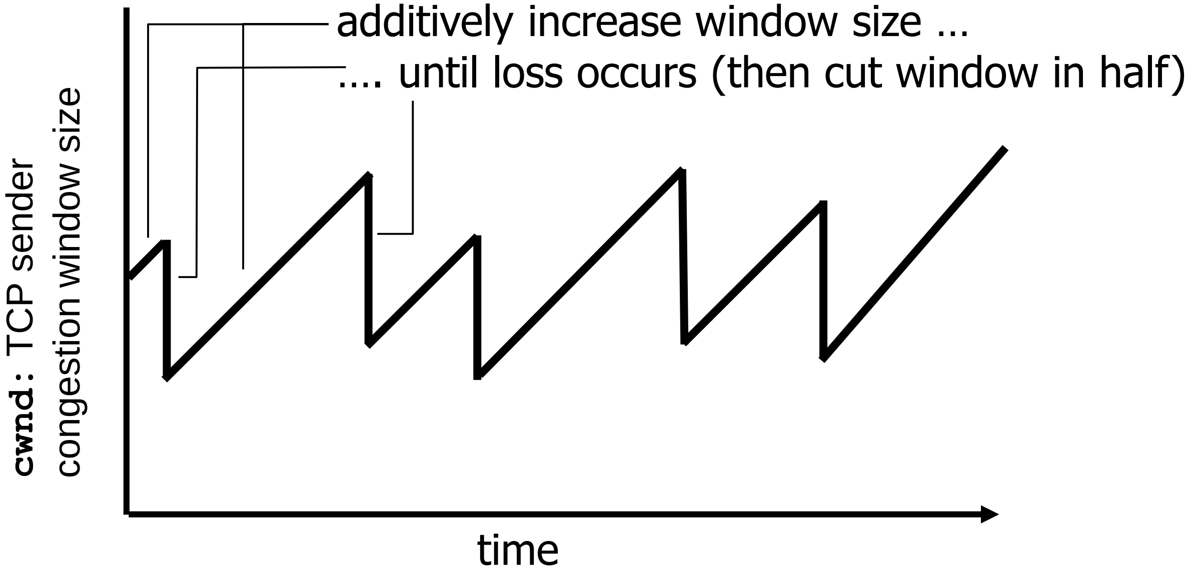
Case study: ATM ABR congestion control



- two-byte ER (explicit rate) field in RM cell
  - congested switch may lower ER value in cell
  - senders' send rate thus max supportable rate on path
- EFCI bit in data cells: set to 1 in congested switch
  - if data cell preceding RM cell has EFCI set, receiver sets CI bit in returned RM cell

TCP congestion control: additive increase multiplicative decrease

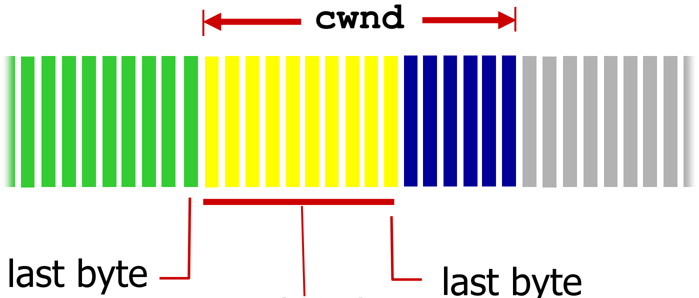
- *approach:* sender increases transmission rate (window size), probing for usable bandwidth, until loss occurs
  - *additive increase:* increase **`cwnd`** by 1 MSS every RTT until loss detected
  - *multiplicative decrease:* cut **`cwnd`** in half after loss



AIMD saw tooth behavior: probing for bandwidth

TCP Congestion Control: details

sender sequence number space



last byte ACKed

sent, not-yet ACKed ("in-flight")

last byte sent

- sender limits transmission:

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

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

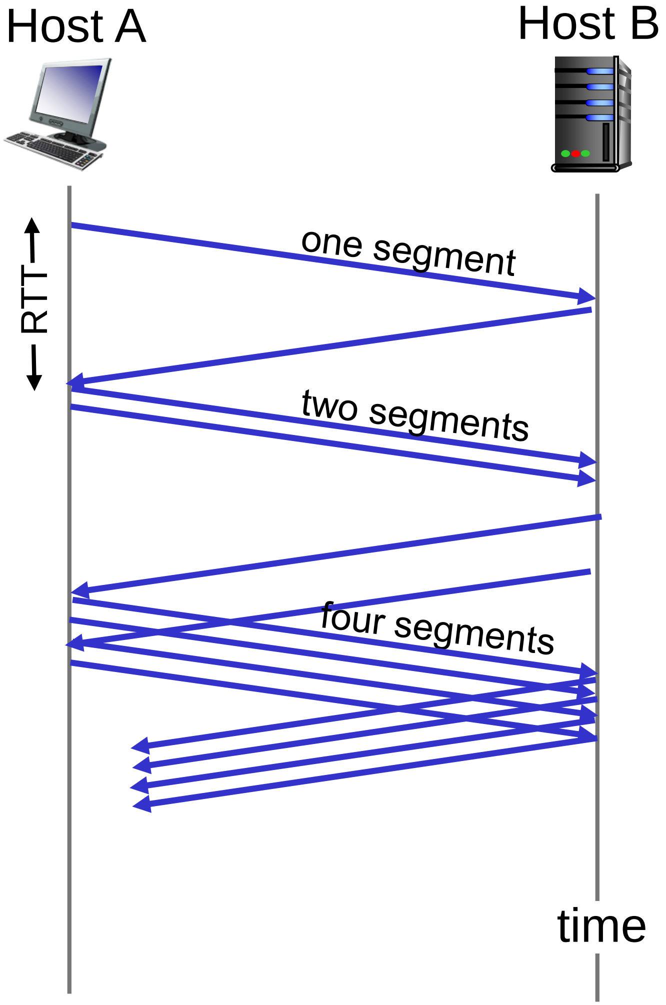
TCP sending rate:

- *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



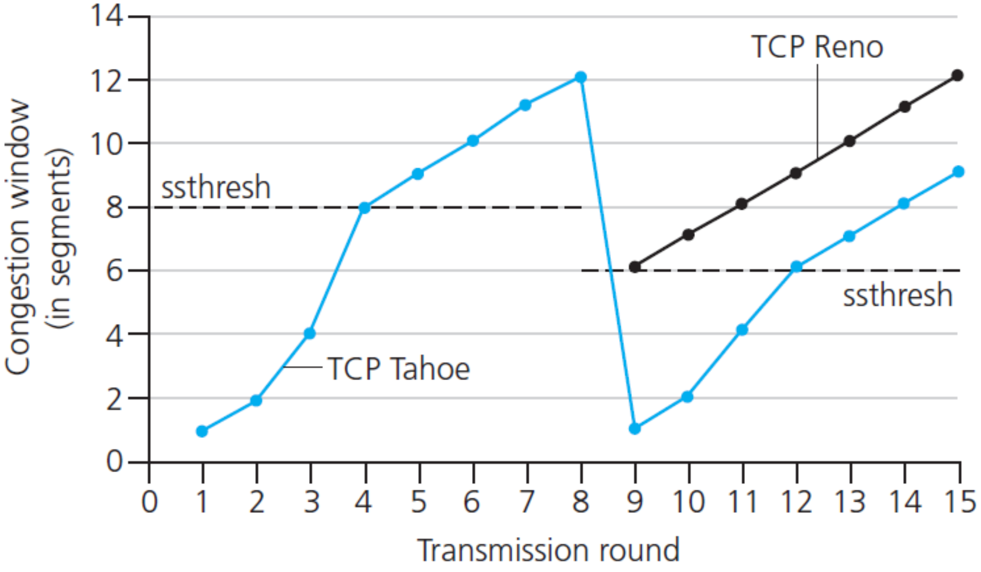
TCP: detecting, reacting to loss

- 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?

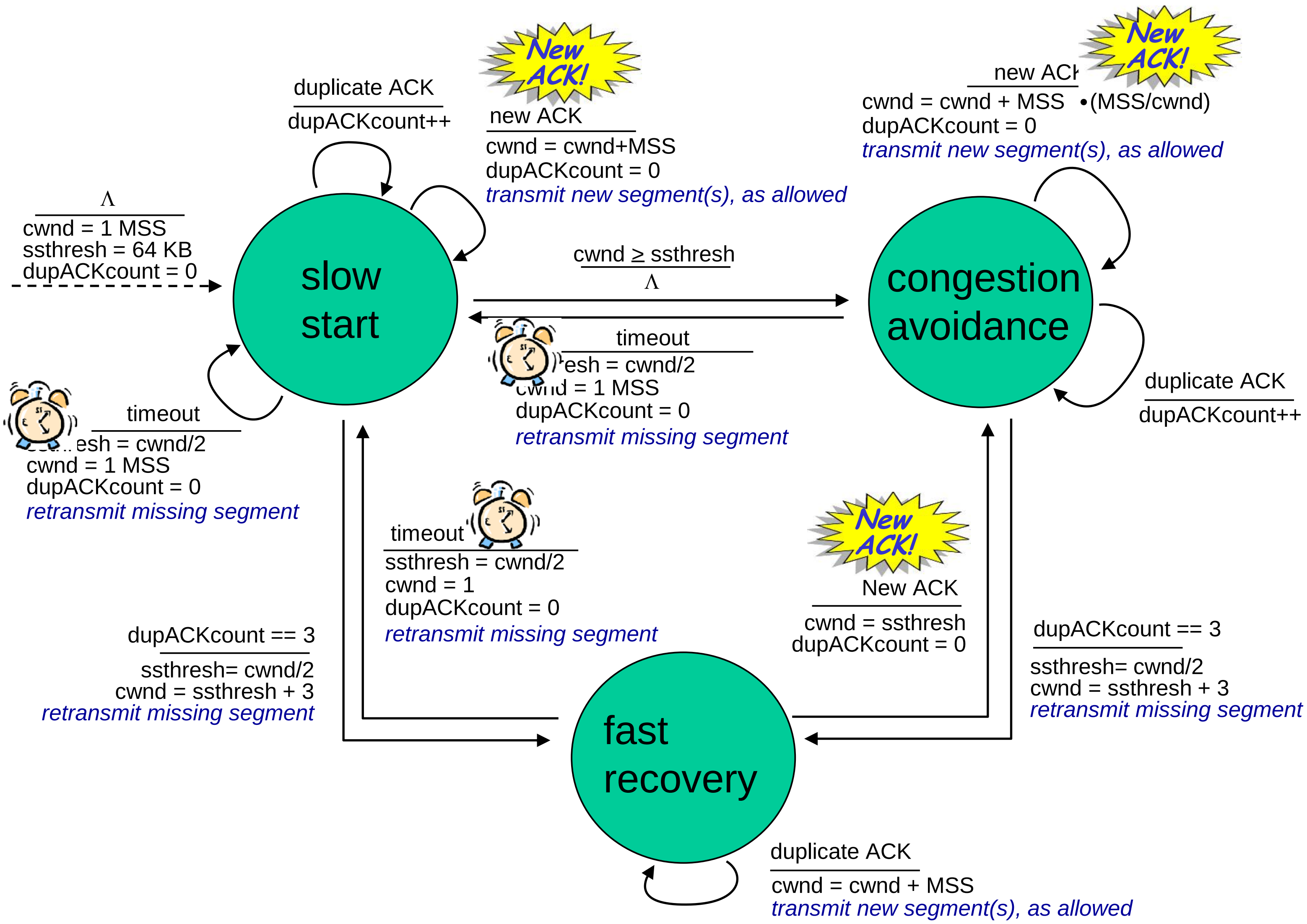
A: when **`cwnd`** gets to 1/2 of its value before timeout.



Implementation:

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

Summary: TCP Congestion Control



slow start

congestion avoidance

fast recovery

Λ
cwnd = 1 MSS
ssthresh = 64 KB
dupACKcount = 0

duplicate ACK
dupACKcount++

New ACK!

new ACK
cwnd = cwnd+MSS
dupACKcount = 0
transmit new segment(s), as allowed

cwnd ≥ ssthresh
Λ

timeout
esh = cwnd/2
cwnd = 1 MSS
dupACKcount = 0
retransmit missing segment

timeout
esh = cwnd/2
cwnd = 1 MSS
dupACKcount = 0
retransmit missing segment

New ACK!

new ACK
cwnd = cwnd + MSS •(MSS/cwnd)
dupACKcount = 0
transmit new segment(s), as allowed

duplicate ACK
dupACKcount++

timeout
ssthresh = cwnd/2
cwnd = 1
dupACKcount = 0
retransmit missing segment

dupACKcount == 3
ssthresh= cwnd/2
cwnd = ssthresh + 3
retransmit missing segment

New ACK!

New ACK
cwnd = ssthresh
dupACKcount = 0

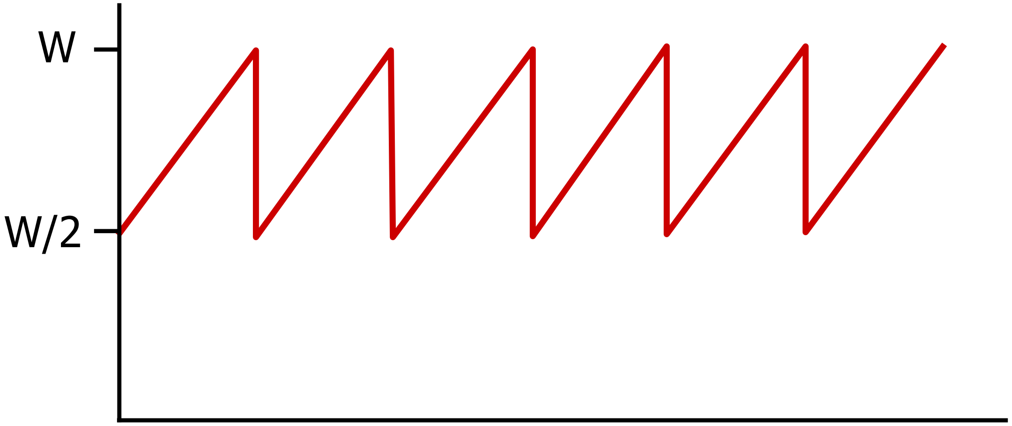
dupACKcount == 3
ssthresh= cwnd/2
cwnd = ssthresh + 3
retransmit missing segment

duplicate ACK
cwnd = cwnd + MSS
transmit new segment(s), as allowed

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 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]:

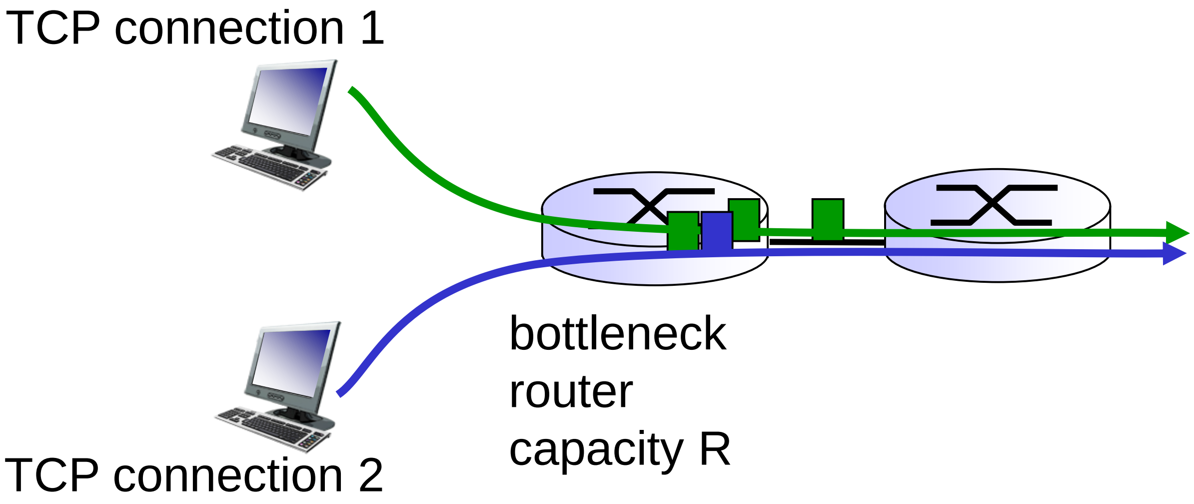
$$\text{TCP throughput} = \frac{1.22 \cdot \text{MSS}}{\text{RTT}\sqrt{\text{L}}}$$

➜ 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

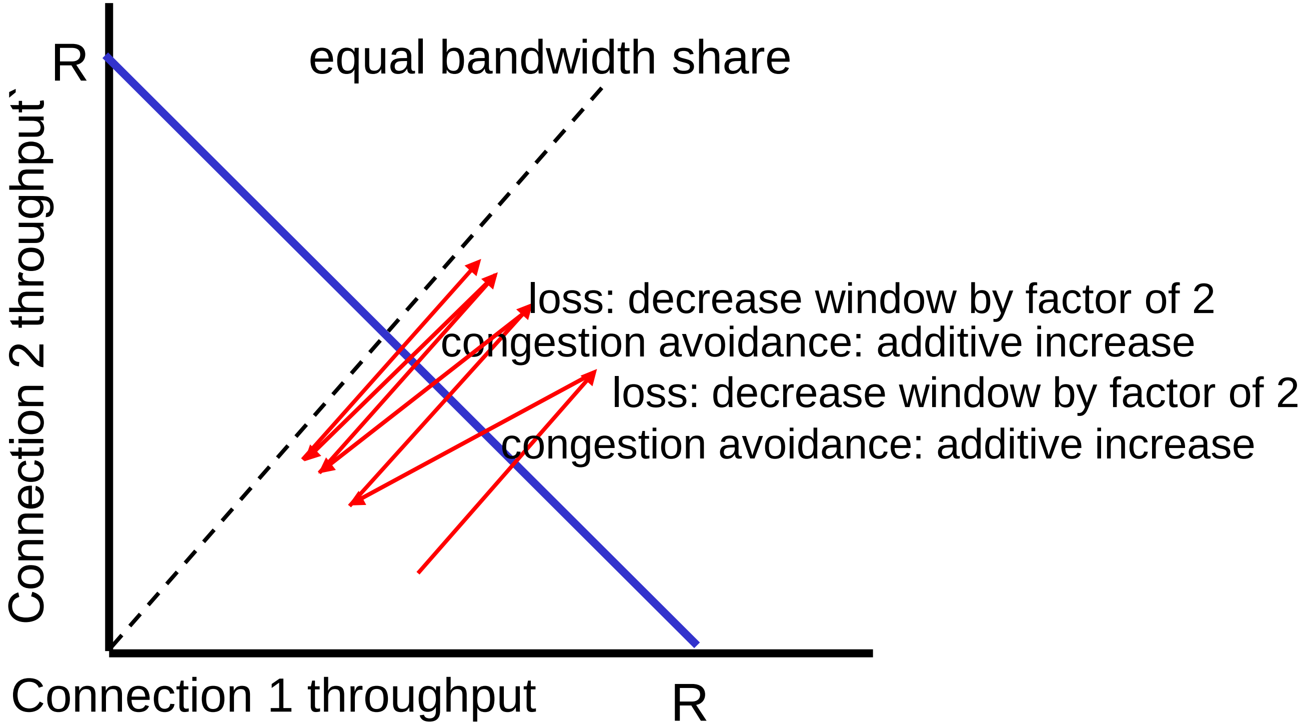
fairness goal: if K TCP sessions share same bottleneck link of bandwidth R, each should have average rate of R/K



Why is TCP fair?

two competing sessions:

- additive increase gives slope of 1, as 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