



Week 6

# Transmission Control Protocol (TCP)

CSCM603154 – **Computer Networks**

Faculty of Computer Science Universitas Indonesia

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# Chapter 3: roadmap

- Transport-layer services
- Multiplexing and demultiplexing
- Connectionless transport: UDP
- **Connection-oriented transport: TCP**
  - segment structure
  - reliable data transfer
  - flow control
  - connection management
- TCP congestion control

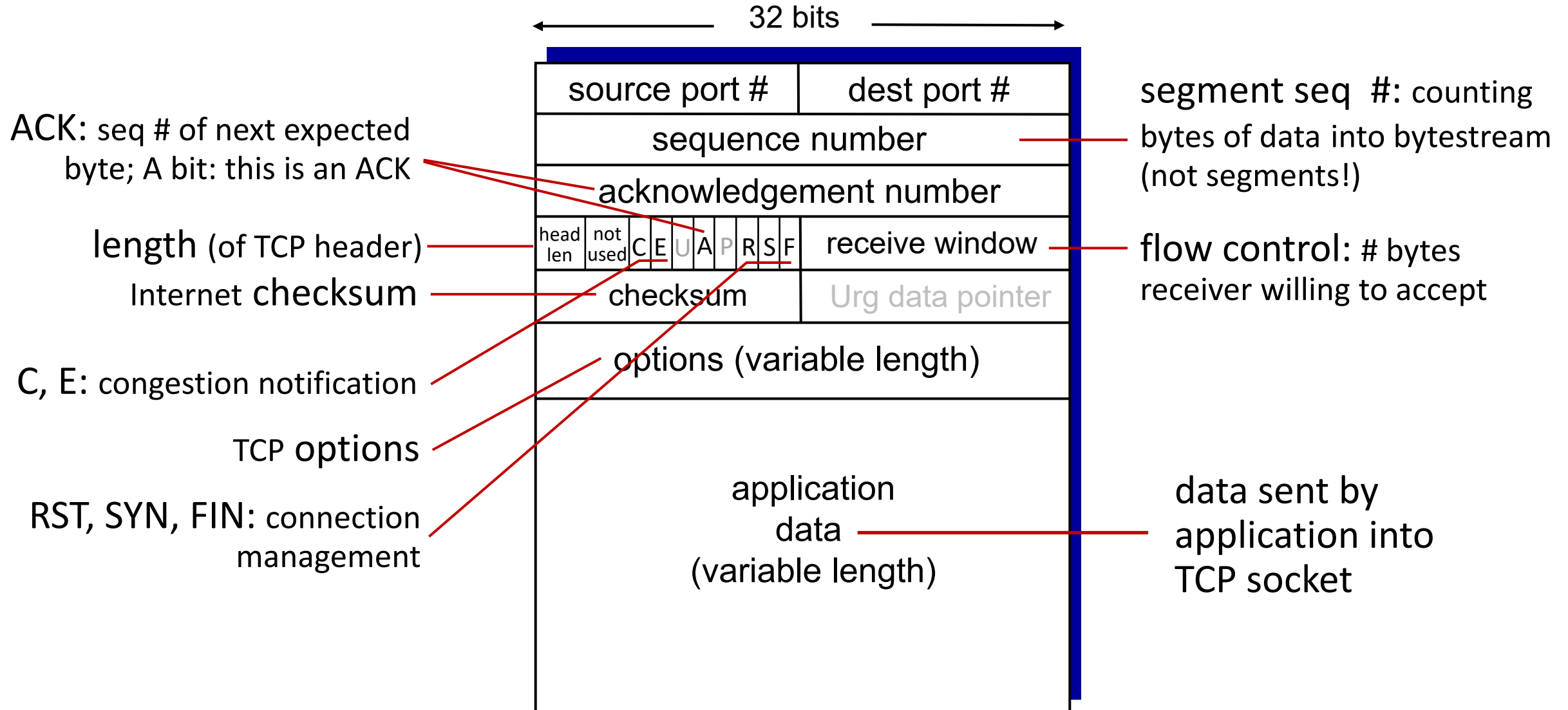


# TCP: overview

RFCs: 793, 1122, 2018, 5681, 7323

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

# TCP segment structure



# TCP sequence numbers, ACKs

## Sequence numbers:

- byte stream “number” of first byte in segment’s data

## Acknowledgements:

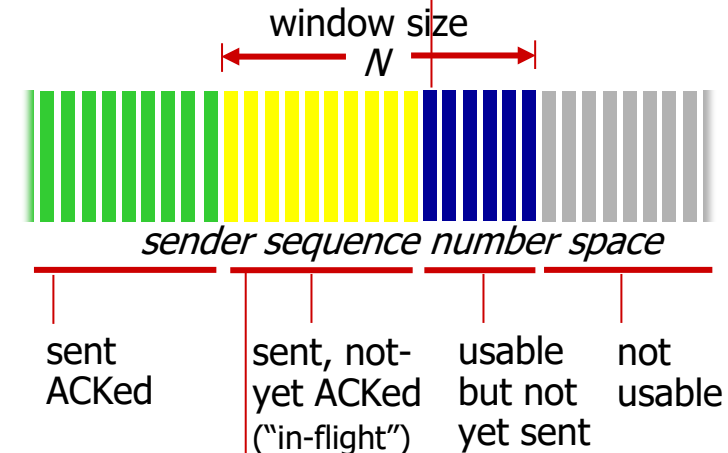
- seq # of next byte expected from other side
- cumulative ACK

Q: how receiver handles out-of-order segments

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

outgoing segment from sender

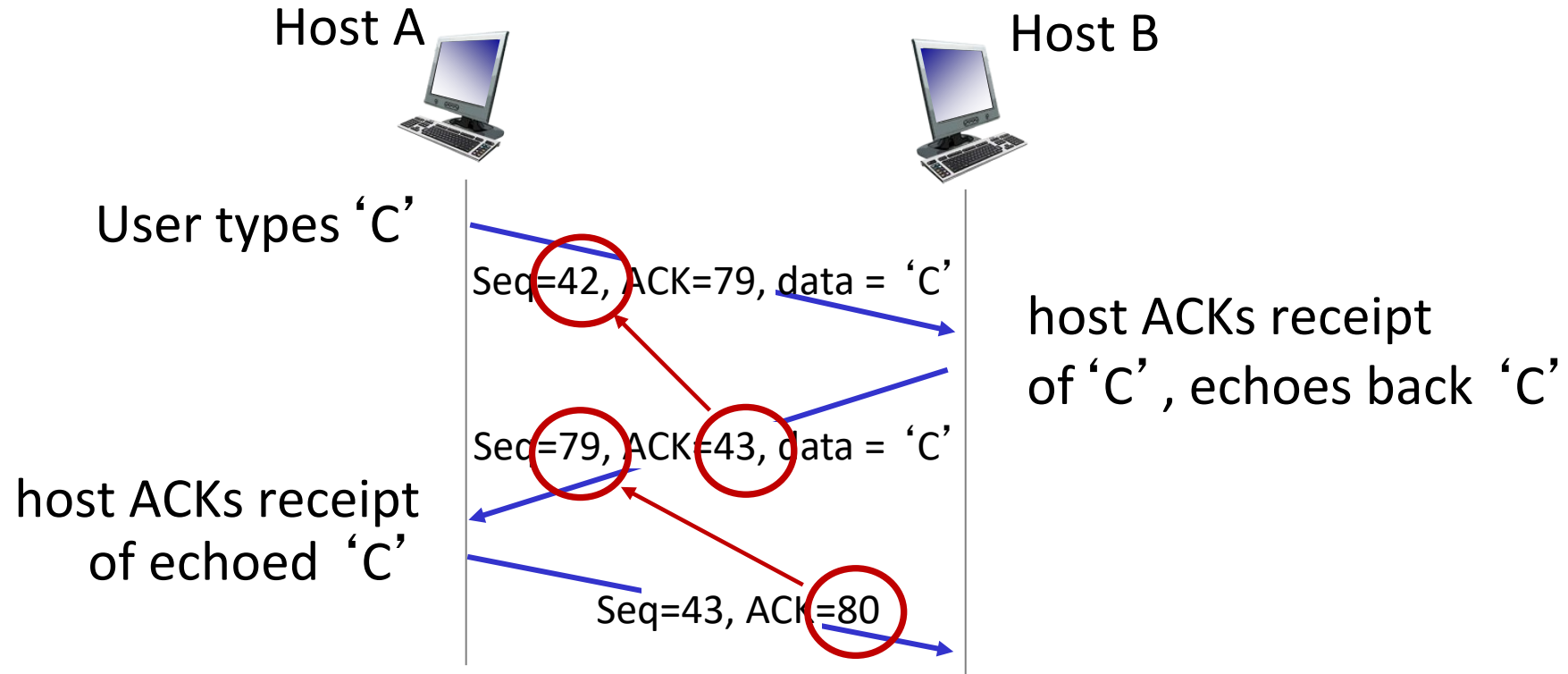
source port #	dest port #
sequence number	
acknowledgement number	
	rwnd
checksum	urg pointer



outgoing segment from receiver

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

# TCP sequence numbers, ACKs



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

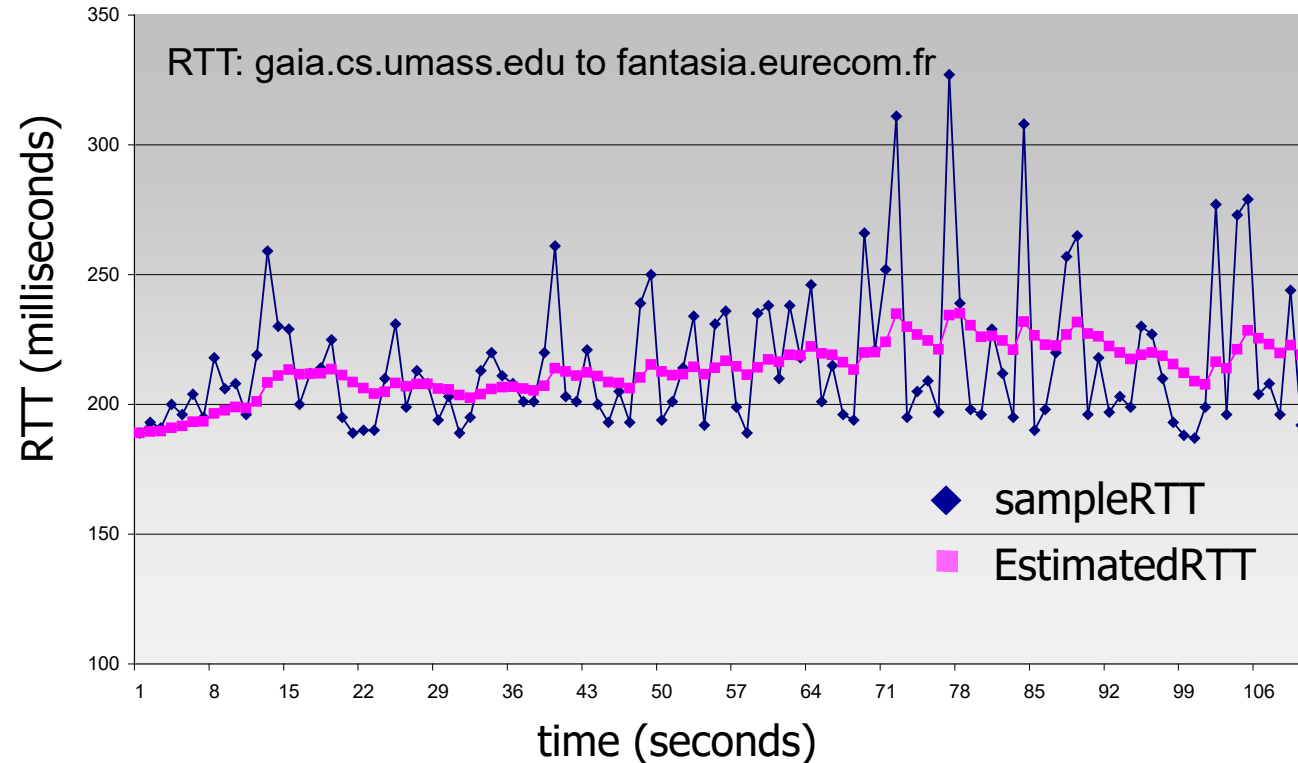
Q: how to estimate RTT?

- *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

$$\text{EstimatedRTT} = (1 - \alpha) * \text{EstimatedRTT} + \alpha * \text{SampleRTT}$$

- exponential weighted moving average (EWMA)
- 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**: want a larger safety margin

$$\text{TimeoutInterval} = \text{EstimatedRTT} + 4 * \text{DevRTT}$$



↑  
estimated RTT

↑  
“safety margin”

- **DevRTT**: EWMA of **SampleRTT** deviation from **EstimatedRTT**:

$$\text{DevRTT} = (1 - \beta) * \text{DevRTT} + \beta * |\text{SampleRTT} - \text{EstimatedRTT}|$$

(typically,  $\beta = 0.25$ )

# TCP Sender (simplified)

event: data received from application

- 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**

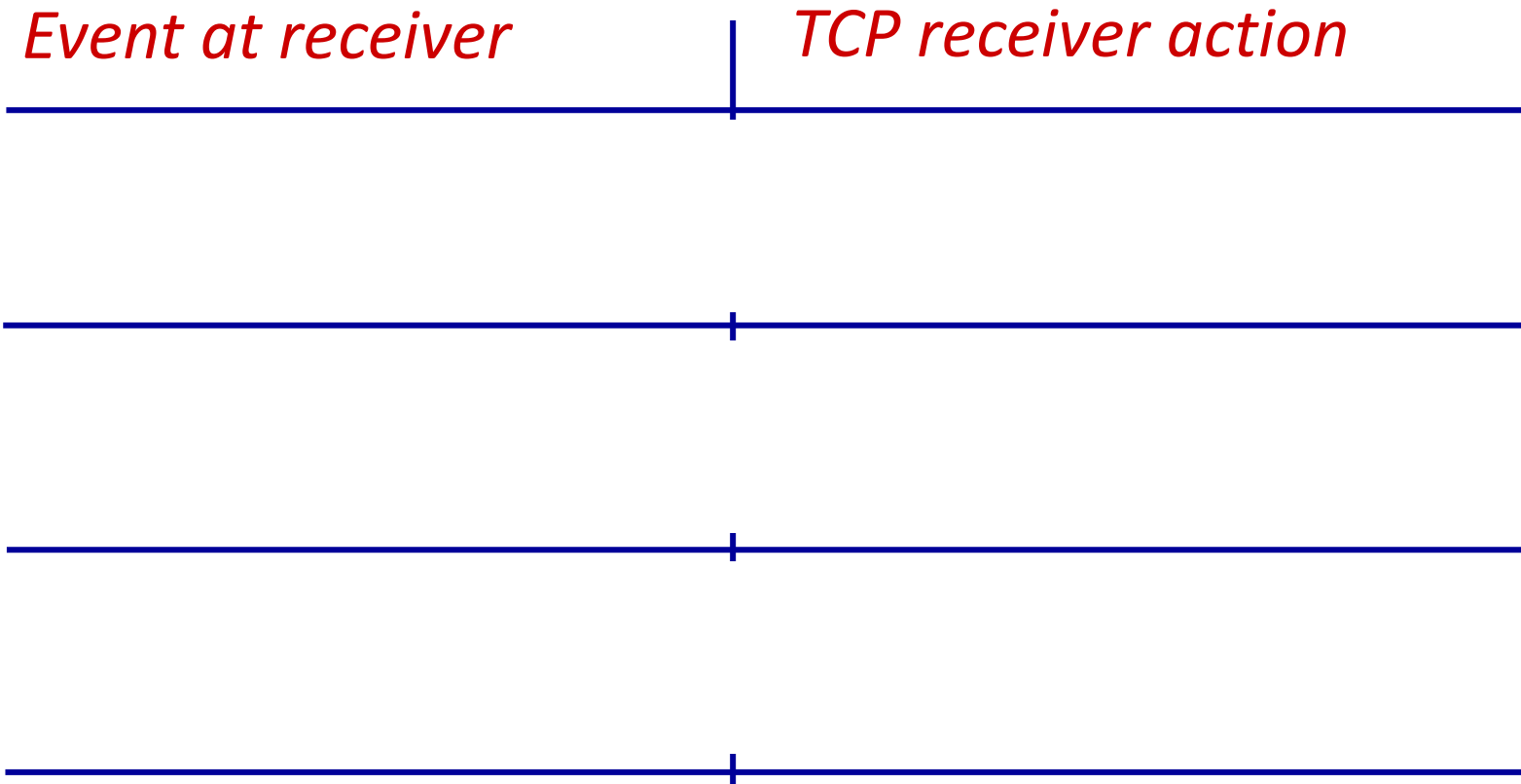
event: *timeout*

- retransmit segment that caused timeout
- restart timer

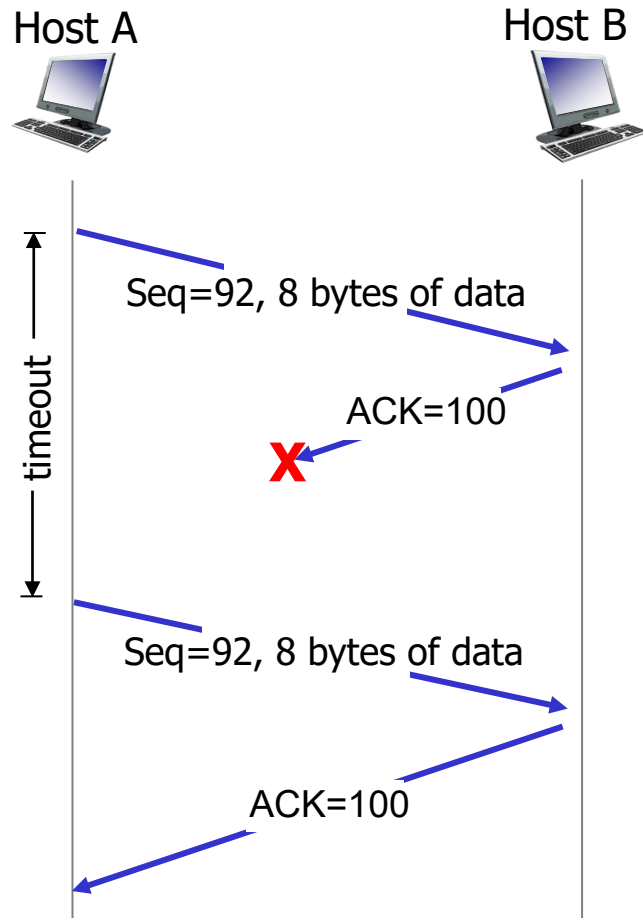
event: *ACK received*

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

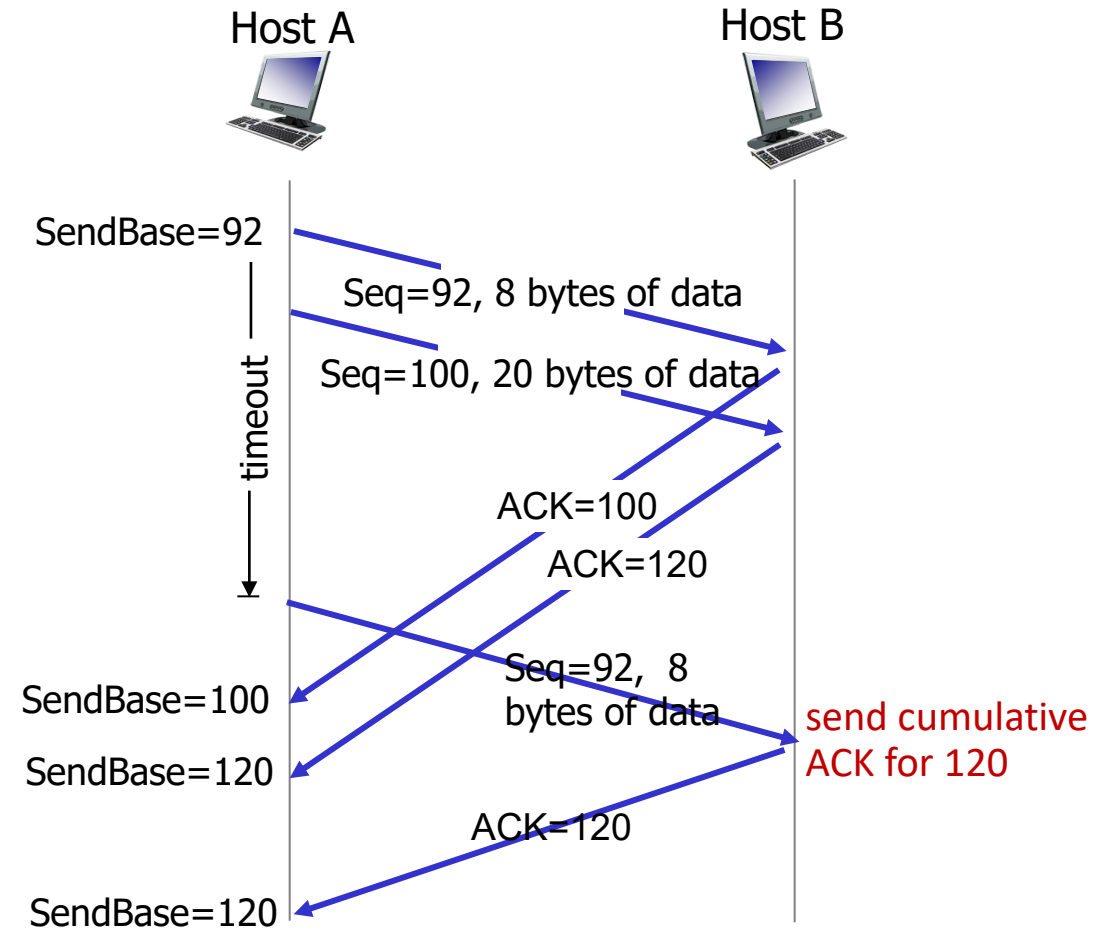
# TCP Receiver: ACK generation [RFC 5681]



# TCP: retransmission scenarios

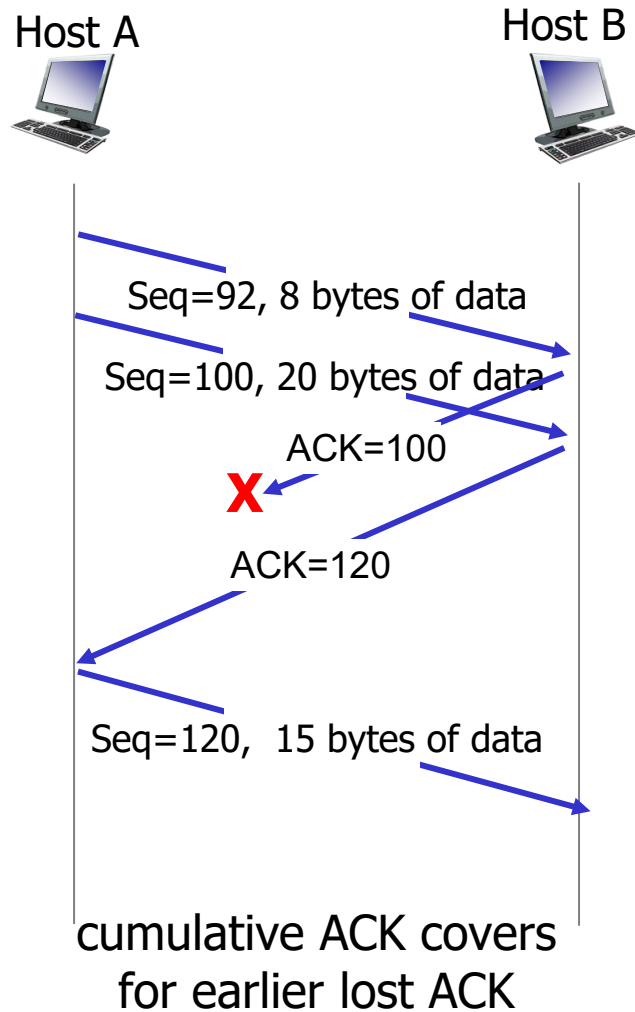


lost ACK scenario



premature timeout

# TCP: retransmission scenarios



# TCP fast retransmit

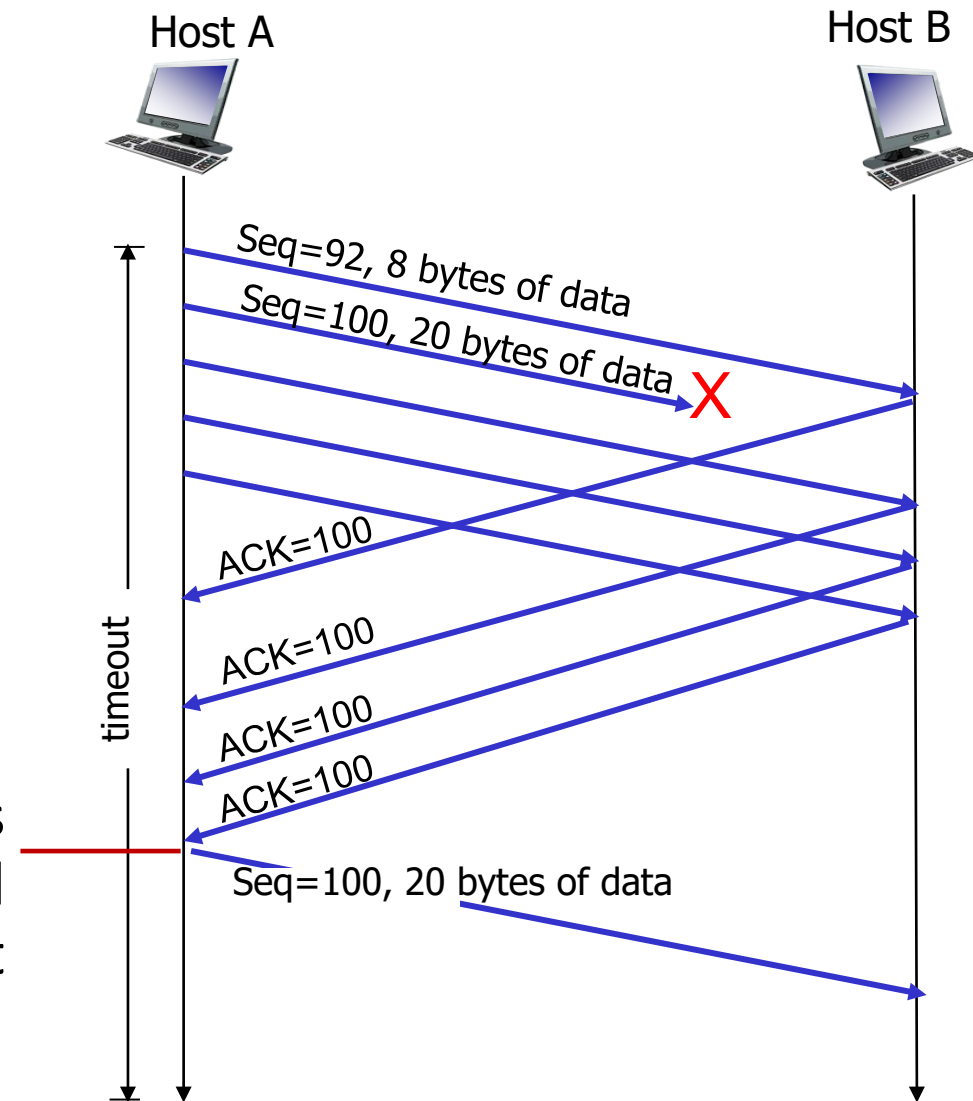
## *TCP fast retransmit*

if sender receives 3 additional ACKs for same data (“triple duplicate ACKs”), resend unACKed segment with smallest seq #

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



Receipt of three duplicate ACKs indicates 3 segments received after a missing segment – lost segment is likely. So retransmit!



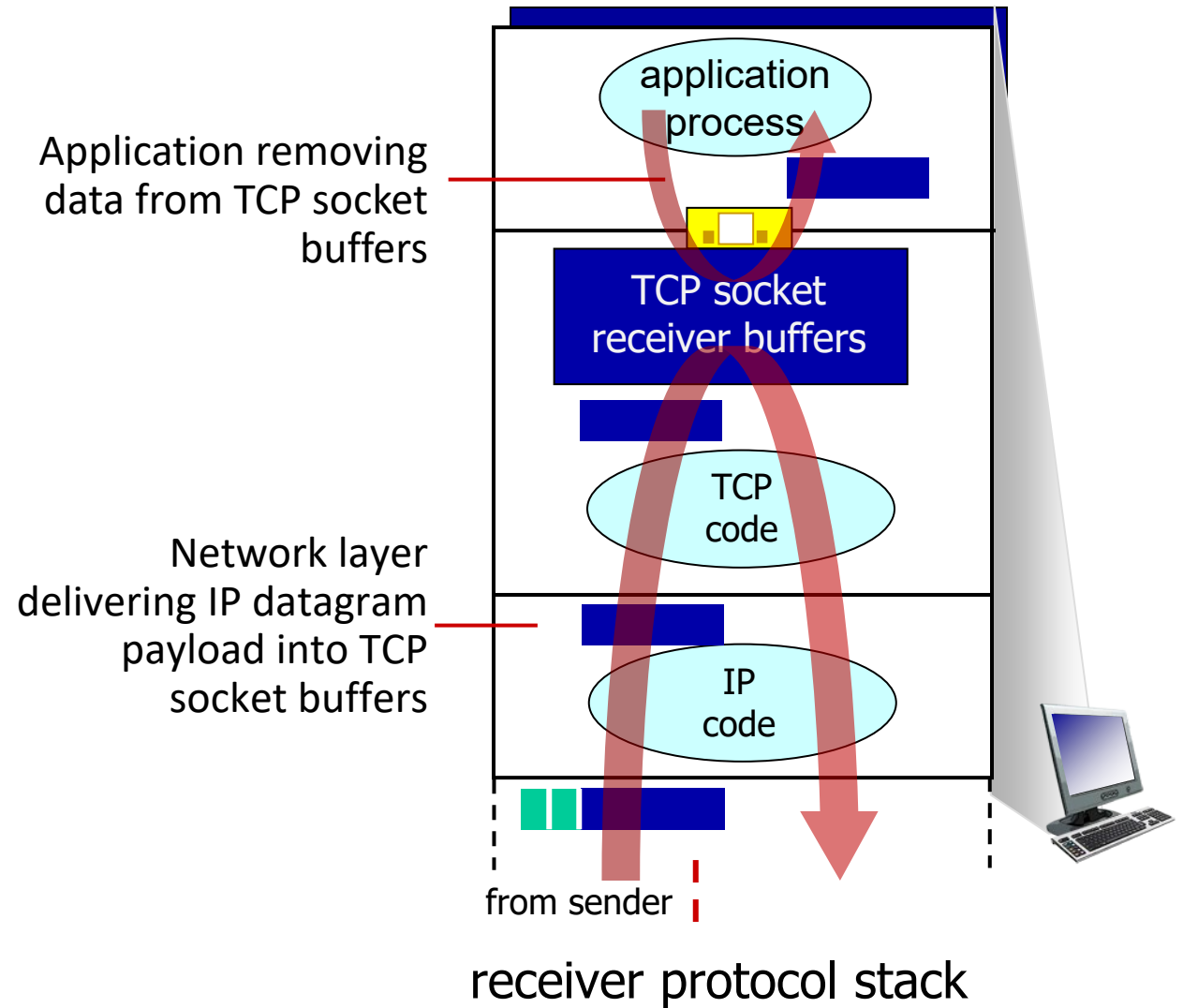
# Chapter 3: roadmap

- Transport-layer services
- Multiplexing and demultiplexing
- Connectionless transport: UDP
- Principles of reliable data transfer
- **Connection-oriented transport: TCP**
  - segment structure
  - reliable data transfer
  - flow control
  - connection management
- Principles of congestion control
- TCP congestion control



# TCP flow control

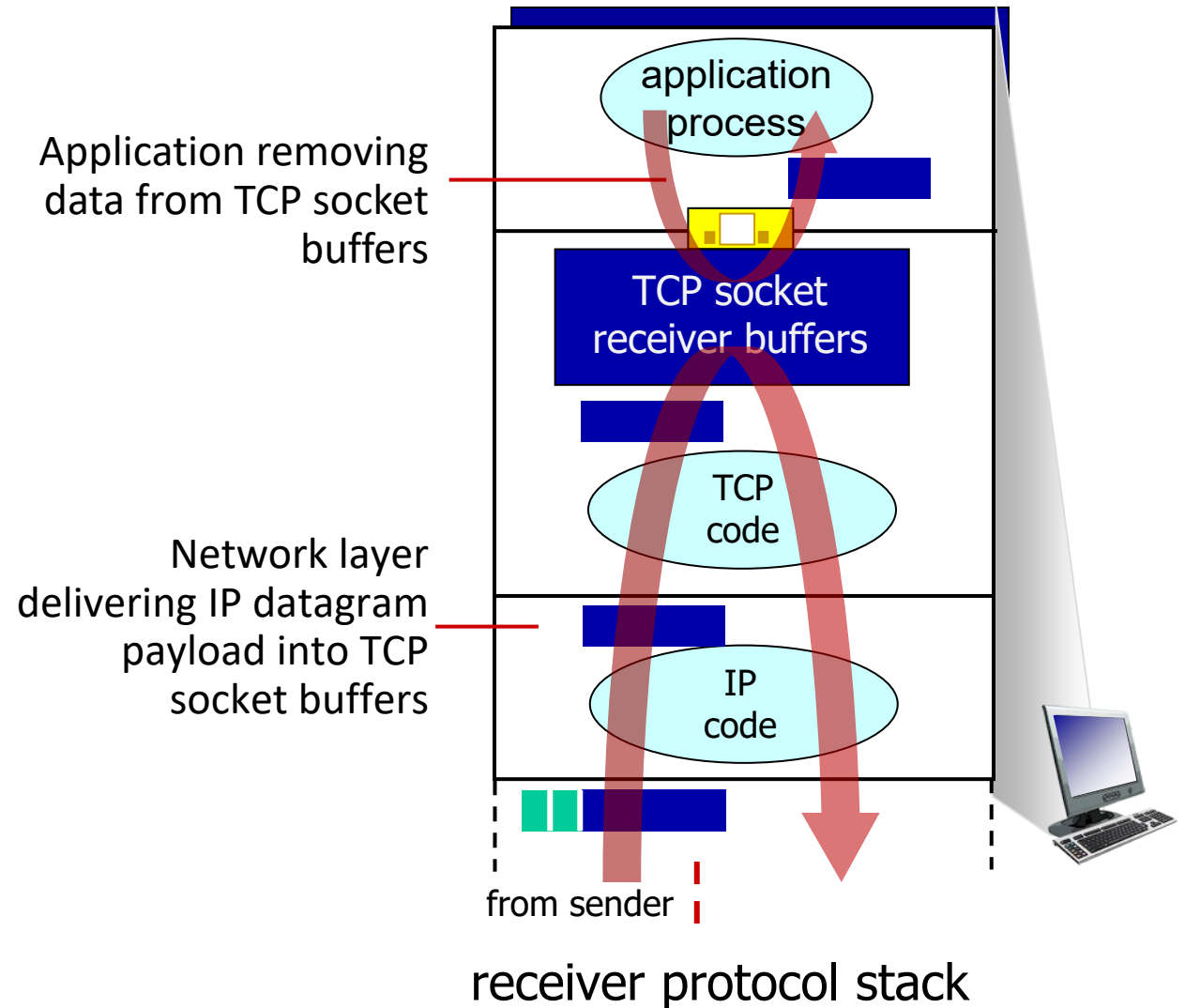
Q: What happens if network layer delivers data faster than application layer removes data from socket buffers?





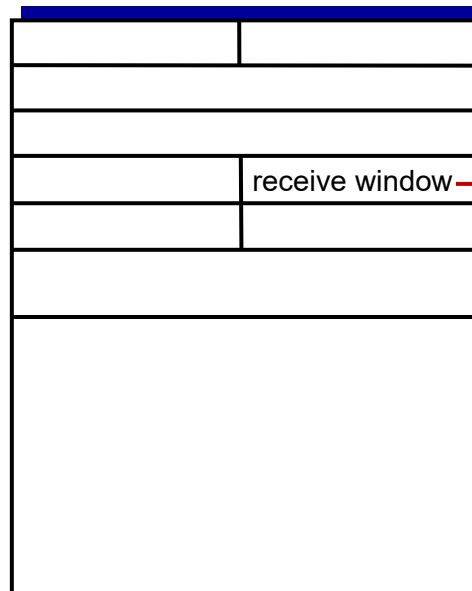
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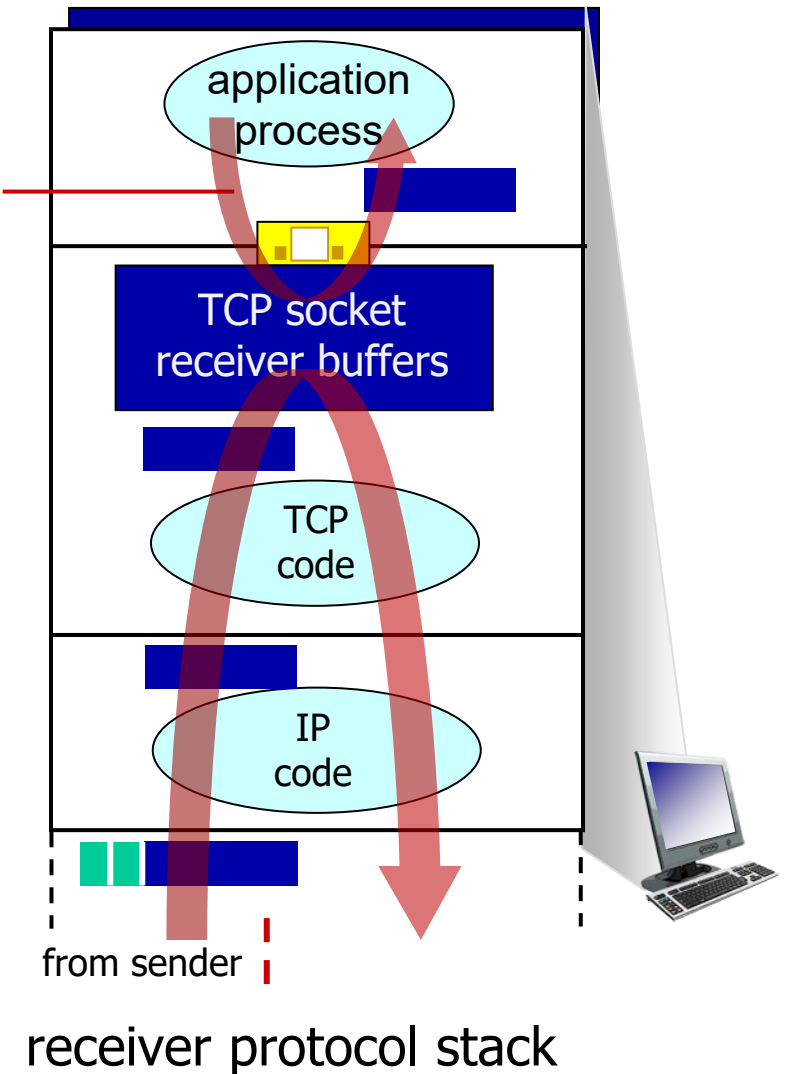
# TCP flow control

Q: What happens if network layer delivers data faster than application layer removes data from socket buffers?



flow control: # bytes  
receiver willing to accept

Application removing  
data from TCP socket  
buffers

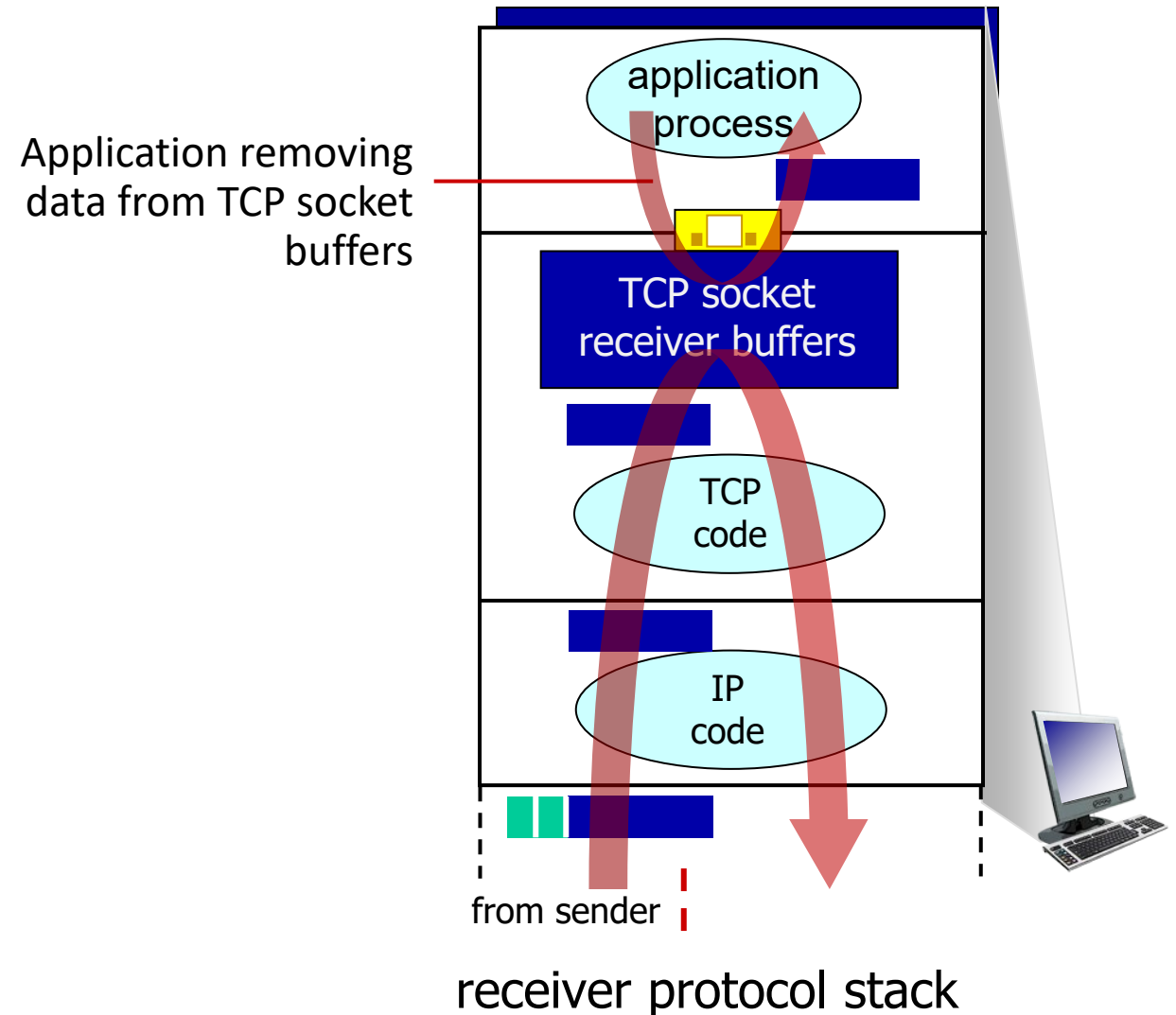


# TCP flow control

Q: What happens if network layer delivers data faster than application layer removes data from socket buffers?

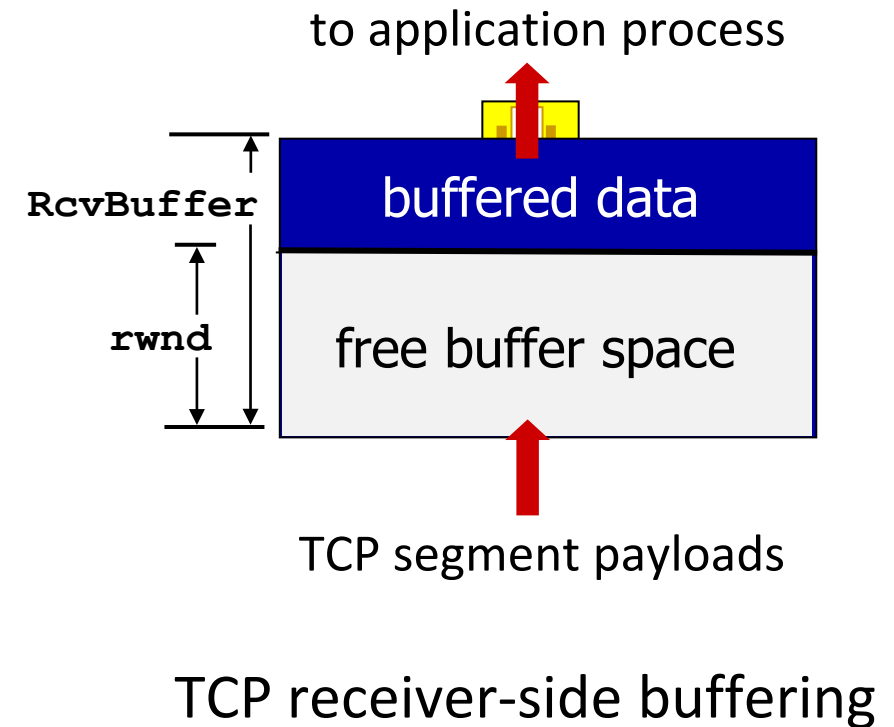
## —flow control—

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



# TCP flow control

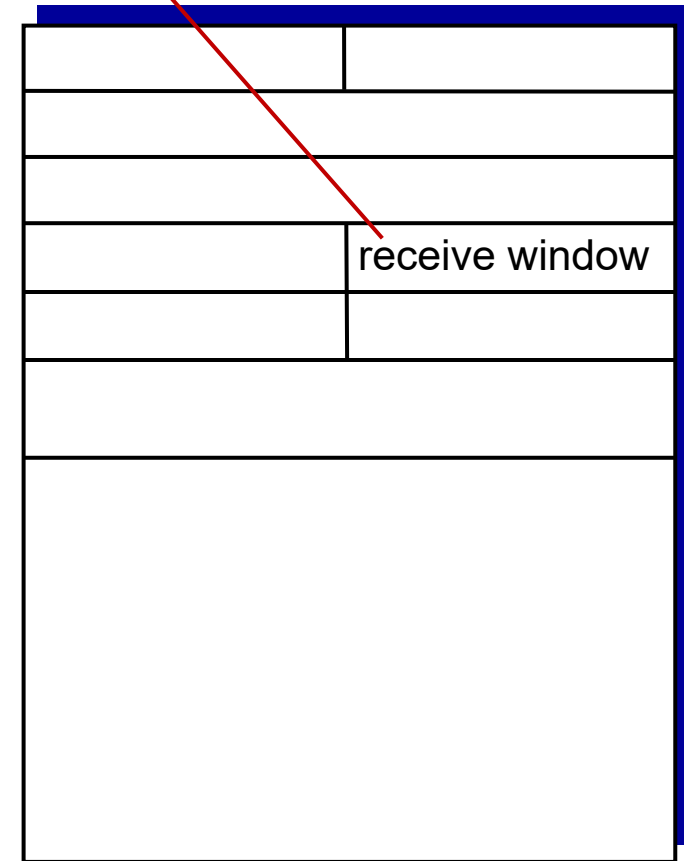
- TCP receiver “advertises” free buffer space in **rwnd** field in TCP header
  - **RcvBuffer** size set via socket options (typical default is 4096 bytes)
  - many operating systems autoadjust **RcvBuffer**
- sender limits amount of unACKed (“in-flight”) data to received **rwnd**
- guarantees receive buffer will not overflow



# TCP flow control

- TCP receiver “advertises” free buffer space in **rwnd** field in TCP header
  - **RcvBuffer** size set via socket options (typical default is 4096 bytes)
  - many operating systems autoadjust **RcvBuffer**
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- guarantees receive buffer will not overflow

flow control: # bytes receiver willing to accept

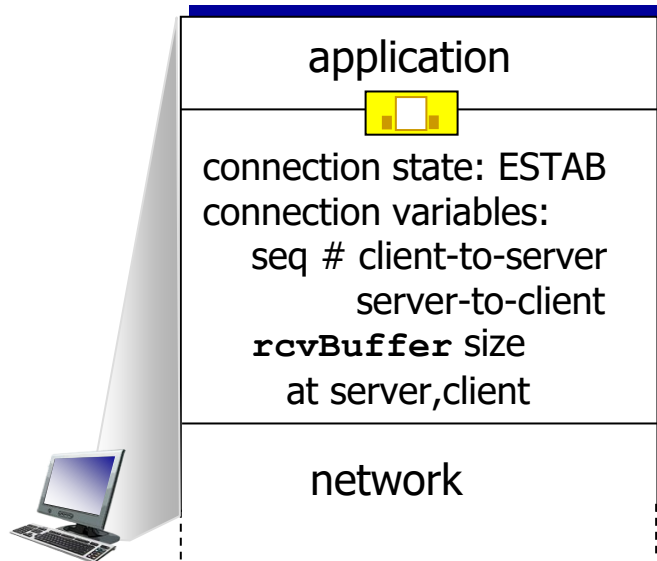


TCP segment format

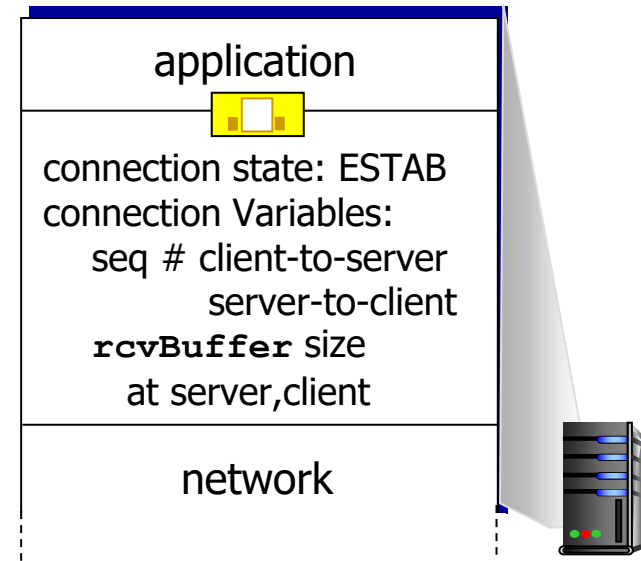
# TCP connection management

before exchanging data, sender/receiver “handshake”:

- agree to establish connection (each knowing the other willing to establish connection)
- agree on connection parameters (e.g., starting seq #s)



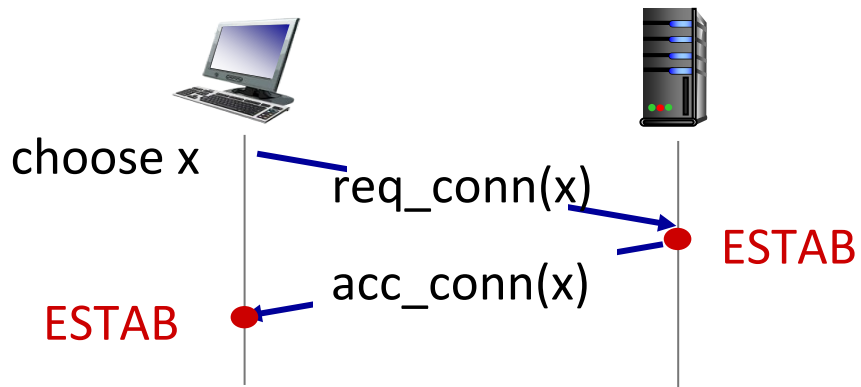
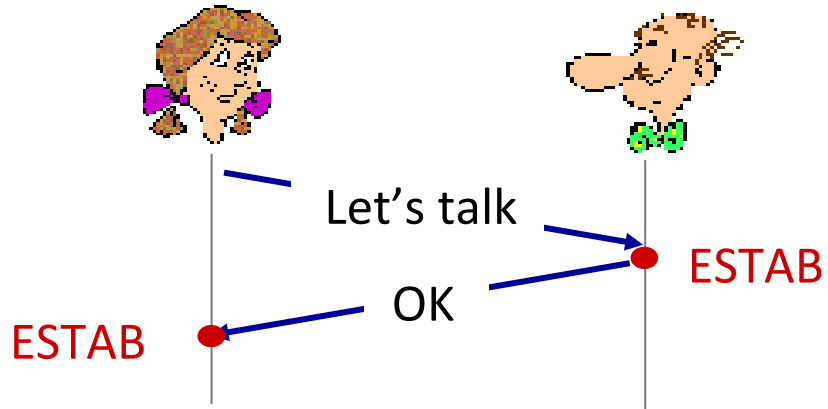
```
Socket clientSocket =  
    newSocket("hostname", "port number");
```



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

# Agreeing to establish a connection

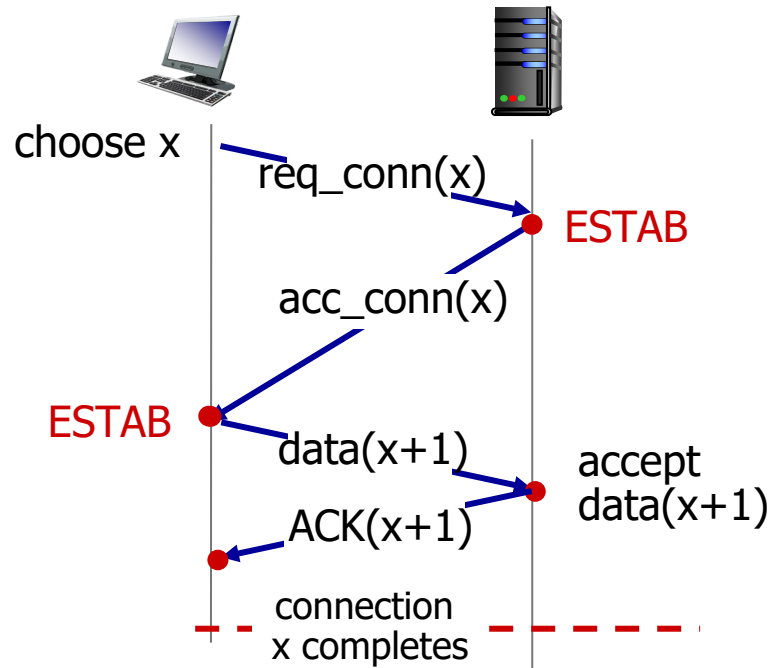
2-way handshake:



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

# 2-way handshake scenarios

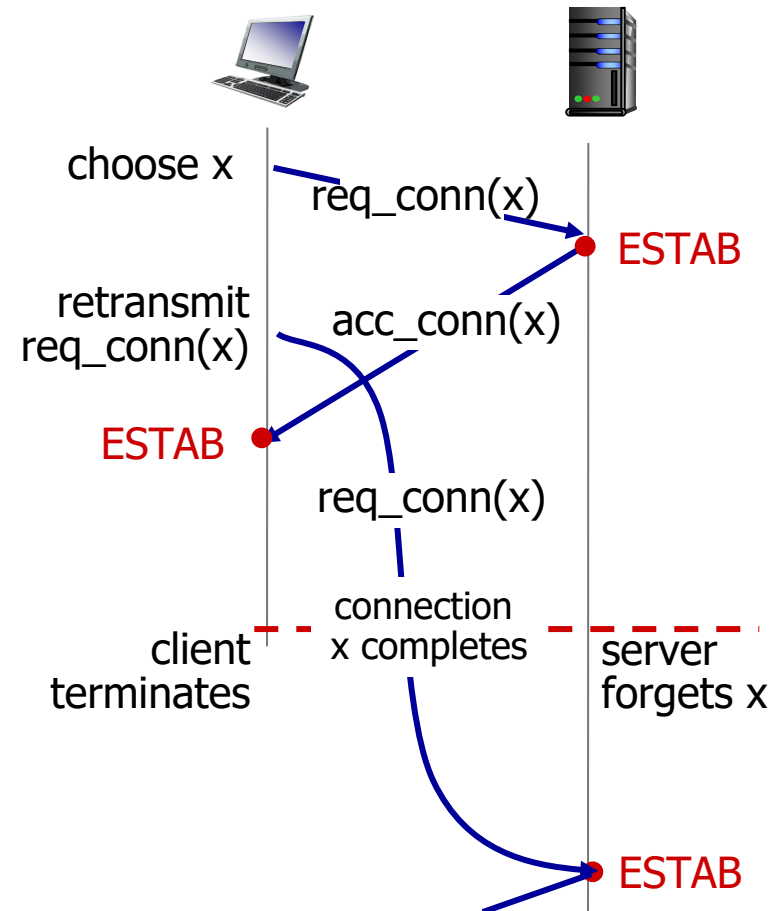



No problem!



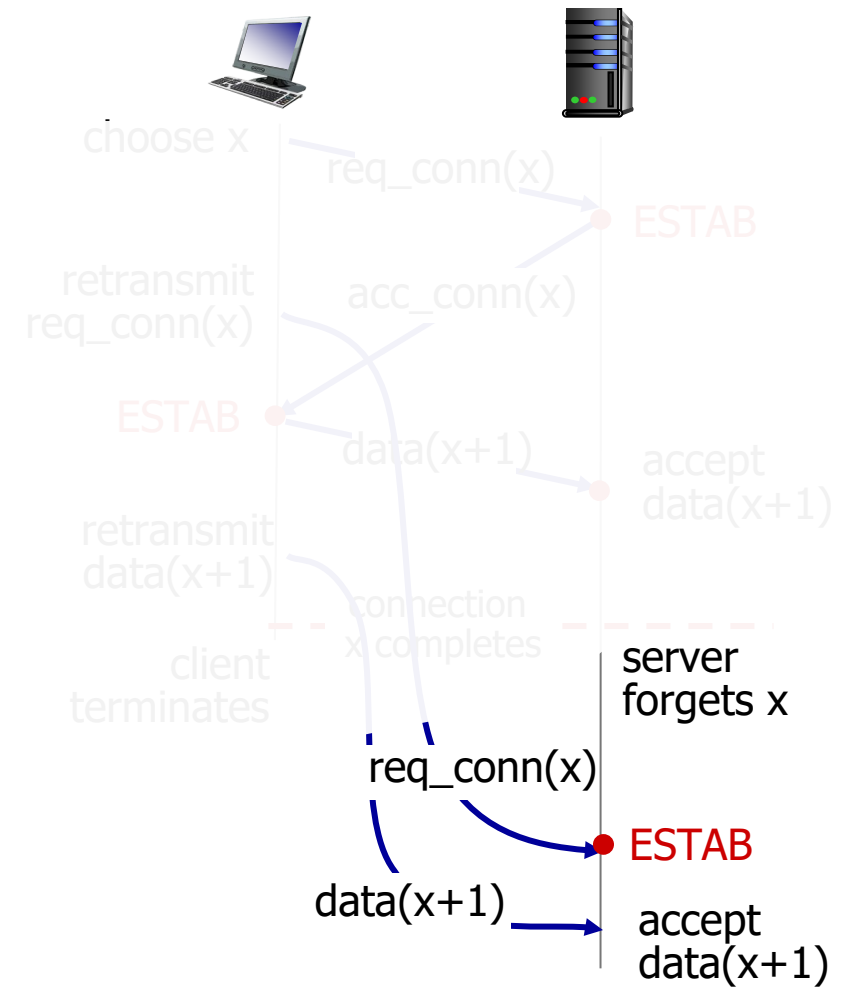



# 2-way handshake scenarios



 Problem: half open connection! (no client)

# 2-way handshake scenarios



 Problem: dup data accepted!

# TCP 3-way handshake

## Client state

```
clientSocket = socket(AF_INET, SOCK_STREAM)
```

LISTEN

```
clientSocket.connect((serverName, serverPort))
```

SYNSENT

ESTAB

choose init seq num, x  
send TCP SYN msg

SYNbit=1, Seq=x

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

## Server state

```
serverSocket = socket(AF_INET, SOCK_STREAM)  
serverSocket.bind(('', serverPort))  
serverSocket.listen(1)  
connectionSocket, addr = serverSocket.accept()
```

LISTEN

SYN RCVD

ESTAB

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

# A human 3-way handshake protocol



# Closing a TCP 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

# Chapter 3: roadmap

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- TCP congestion control



# TCP congestion control: AIMD

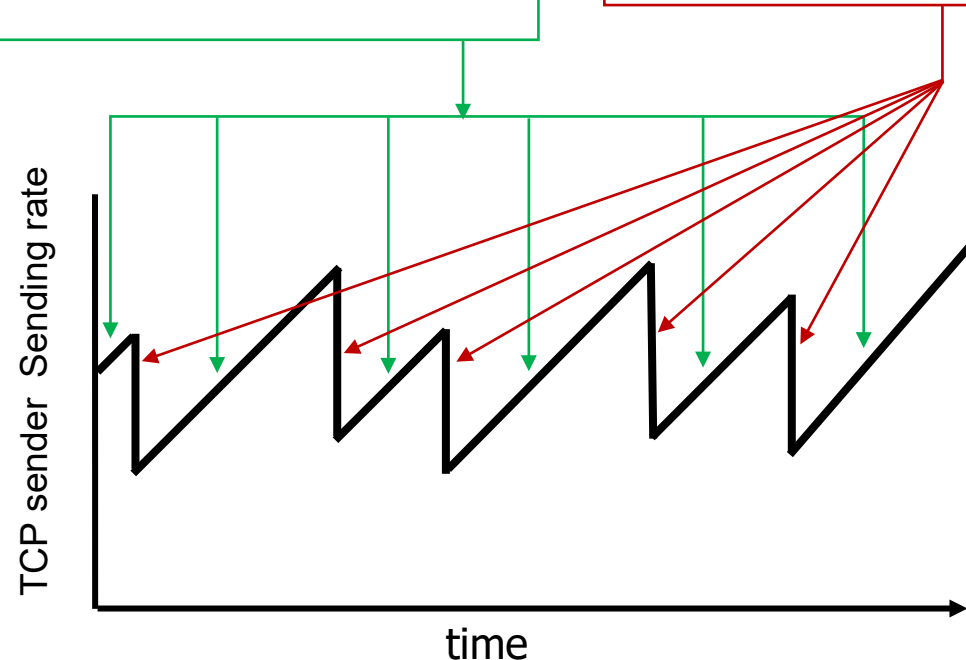
- *approach*: senders can increase sending rate until packet loss (congestion) occurs, then decrease sending rate on loss event

## Additive Increase

increase sending rate by 1 maximum segment size every RTT until loss detected

## Multiplicative Decrease

cut sending rate in half at each loss event



**AIMD** sawtooth behavior: *probing* for bandwidth



# TCP AIMD: more

*Multiplicative decrease* detail: sending rate is

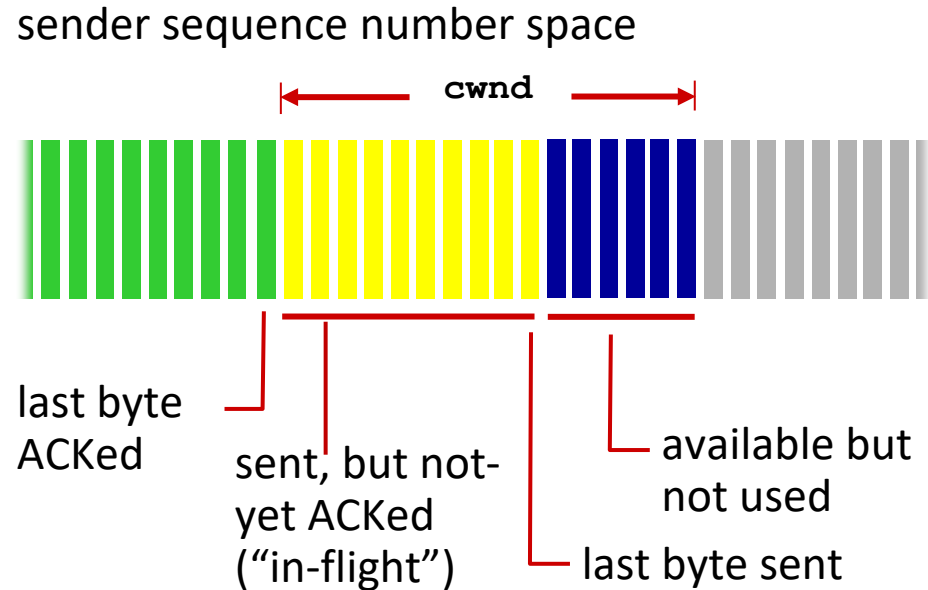
- Cut in half on loss detected by triple duplicate ACK (TCP Reno)
- Cut to 1 MSS (maximum segment size) when loss detected by timeout (TCP Tahoe)

Why AIMD?

- AIMD – a distributed, asynchronous algorithm – has been shown to:
  - optimize congested flow rates network wide!
  - have desirable stability properties



# TCP congestion control: details



TCP sending behavior:

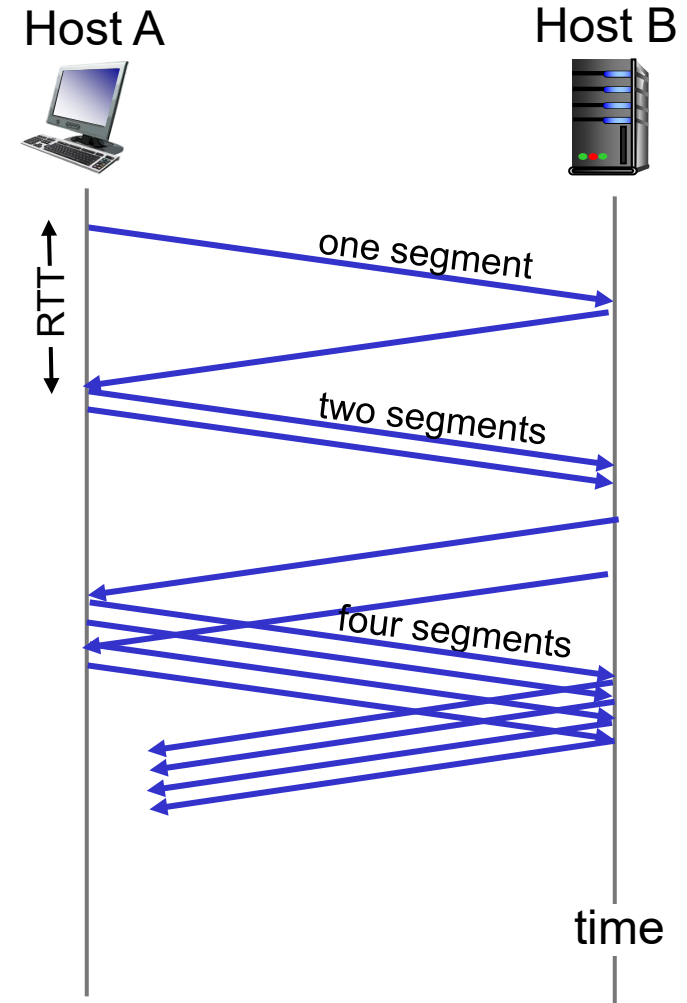
- *roughly*: send `cwnd` bytes, wait RTT for ACKS, then send more bytes

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

- TCP sender limits transmission:  $\text{LastByteSent} - \text{LastByteAcked} \leq \text{cwnd}$
- `cwnd` is dynamically adjusted in response to observed network congestion (implementing TCP congestion control)

# 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



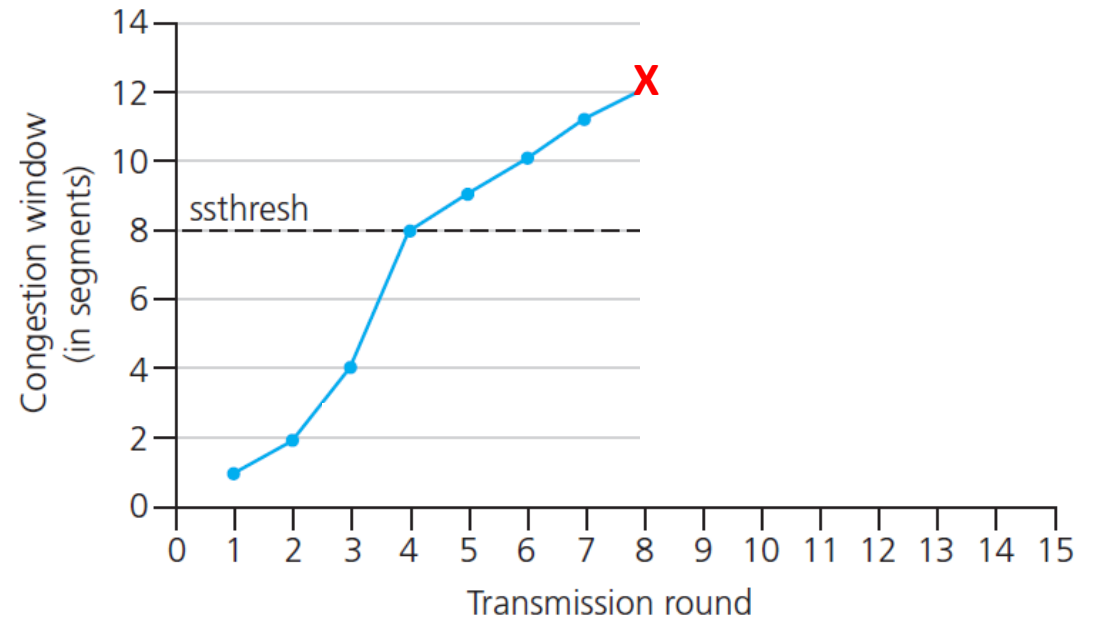
# TCP: from slow start to congestion avoidance

**Q:** when should the exponential increase switch to linear?

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

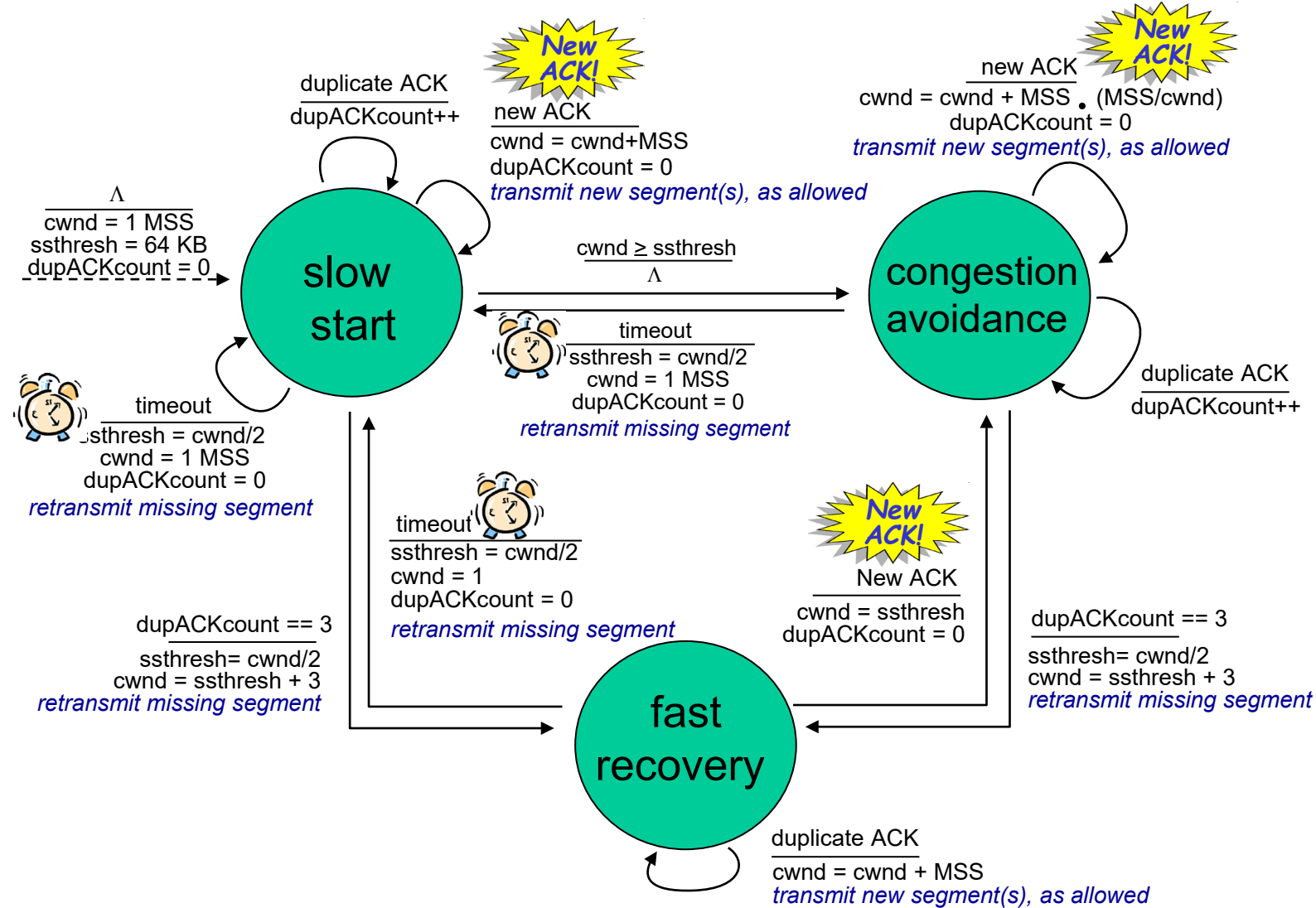
## Implementation:

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



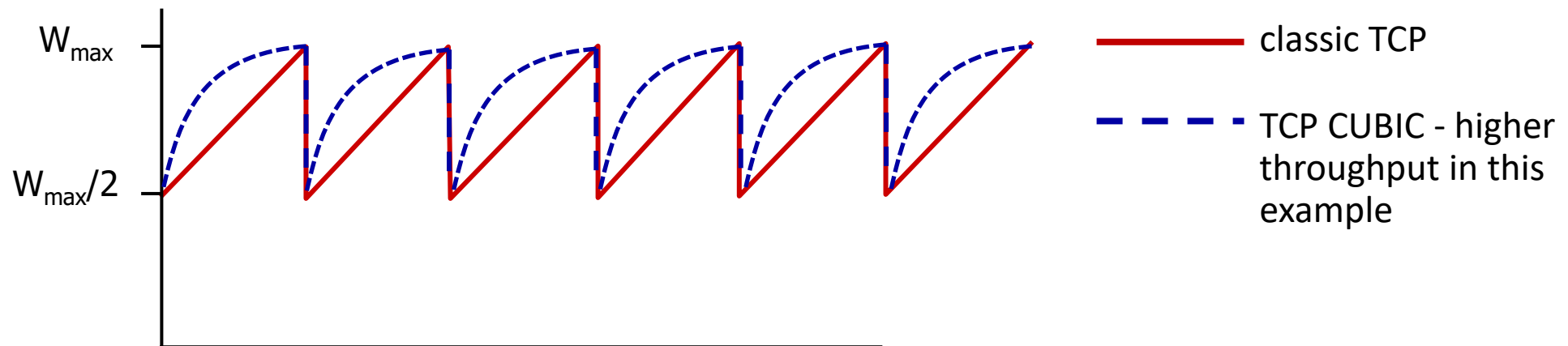
\* Check out the online interactive exercises for more examples: [http://gaia.cs.umass.edu/kurose\\_ross/interactive/](http://gaia.cs.umass.edu/kurose_ross/interactive/)

# Summary: TCP congestion control



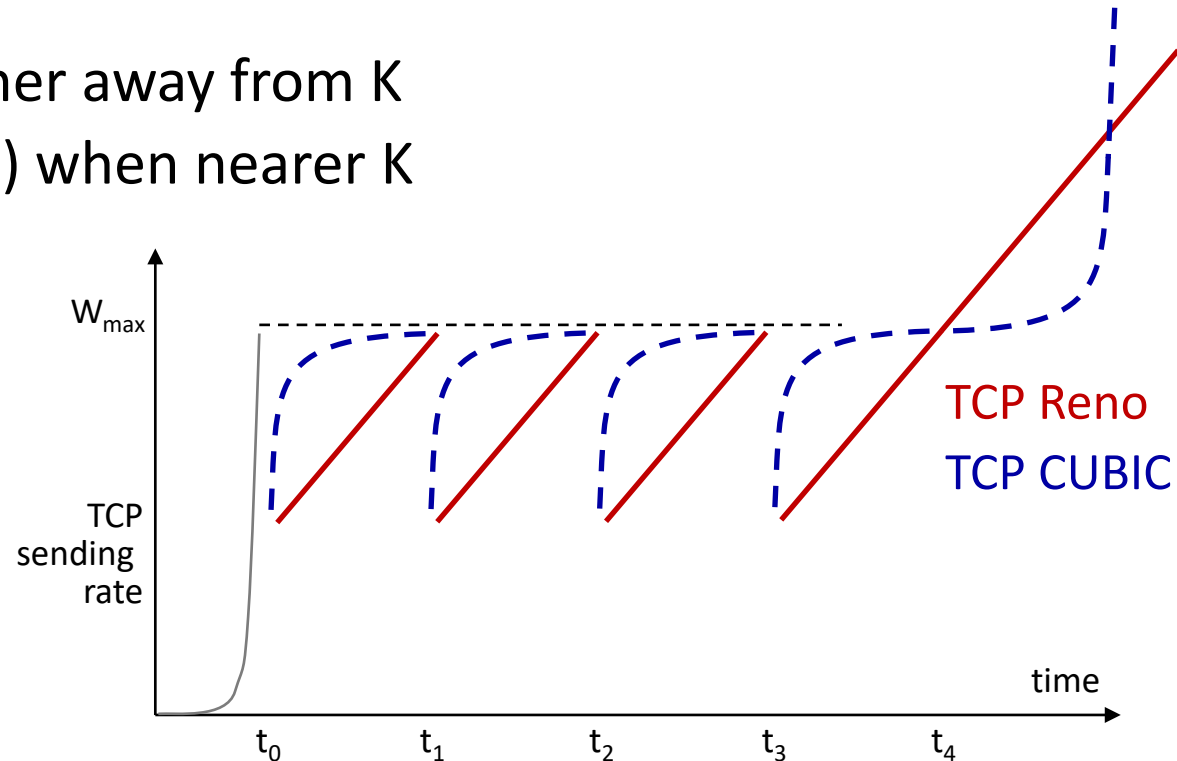
# TCP CUBIC

- Is there a better way than AIMD to “probe” for usable bandwidth?
- Insight/intuition:
  - $W_{\max}$ : sending rate at which congestion loss was detected
  - congestion state of bottleneck link probably (?) hasn't changed much
  - after cutting rate/window in half on loss, initially ramp to to  $W_{\max}$  *faster*, but then approach  $W_{\max}$  more *slowly*



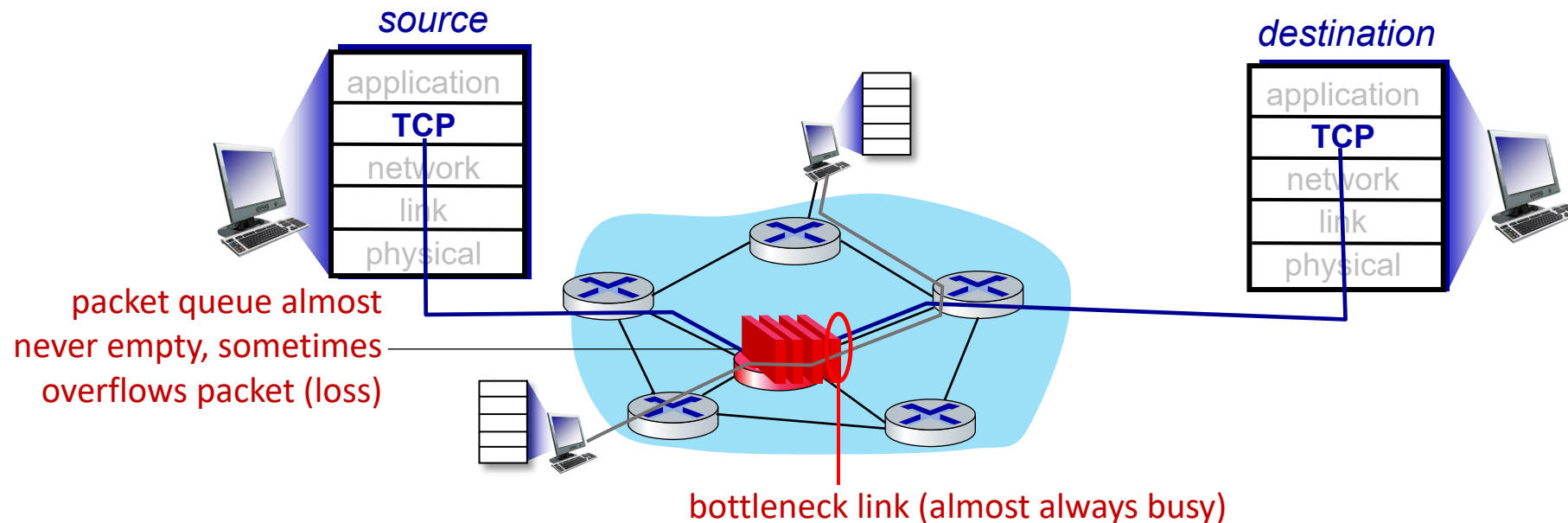
# TCP CUBIC

- K: point in time when TCP window size will reach  $W_{\max}$ 
  - K itself is tuneable
- increase W as a function of the *cube* of the distance between current time and K
  - larger increases when further away from K
  - smaller increases (cautious) when nearer K
- TCP CUBIC default in Linux, most popular TCP for popular Web servers



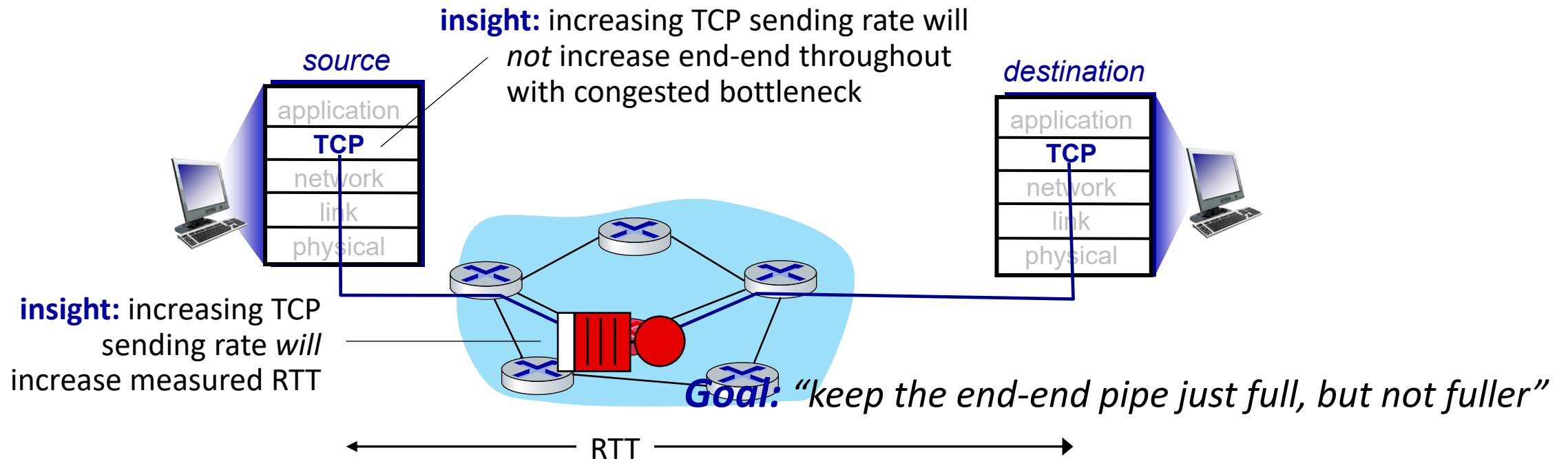
# TCP and the congested “bottleneck link”

- TCP (classic, CUBIC) increase TCP’s sending rate until packet loss occurs at some router’s output: the *bottleneck link*



# TCP and the congested “bottleneck link”

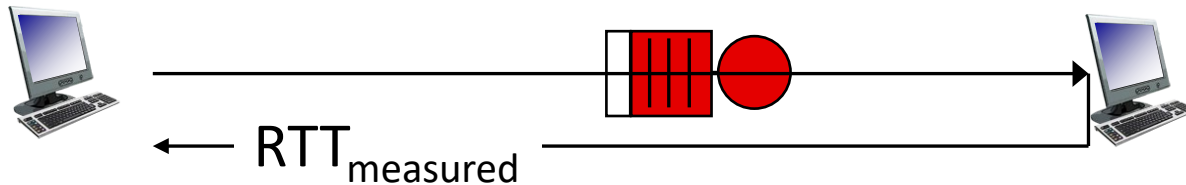
- TCP (classic, CUBIC) increase TCP’s sending rate until packet loss occurs at some router’s output: the *bottleneck link*
- understanding congestion: useful to focus on congested bottleneck link





# Delay-based TCP congestion control

Keeping sender-to-receiver pipe “just full enough, but no fuller”: keep bottleneck link busy transmitting, but avoid high delays/buffering



$$\text{measured throughput} = \frac{\text{\# bytes sent in last RTT interval}}{\text{RTT}_{\text{measured}}}$$

## Delay-based approach:

- $\text{RTT}_{\min}$  - minimum observed RTT (uncongested path)
- uncongested throughput with congestion window  $\text{cwnd}$  is  $\text{cwnd}/\text{RTT}_{\min}$

if measured throughput “very close” to uncongested throughput  
    increase  $\text{cwnd}$  linearly                      /\* since path not congested \*/  
else if measured throughput “far below” uncongested throughput  
    decrease  $\text{cwnd}$  linearly                      /\* since path is congested \*/

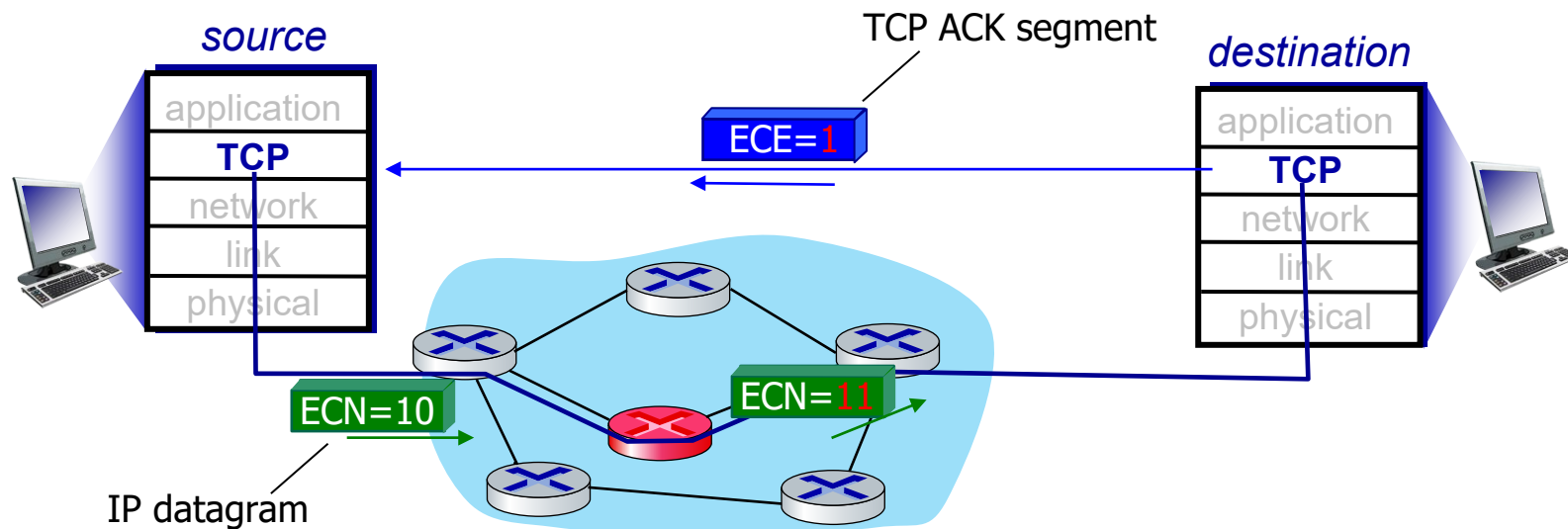
# Delay-based TCP congestion control

- congestion control without inducing/forcing loss
- maximizing throughput (“keeping the just pipe full...”) while keeping delay low (“...but not fuller”)
- a number of deployed TCPs take a delay-based approach
  - BBR deployed on Google’s (internal) backbone network

# Explicit congestion notification (ECN)

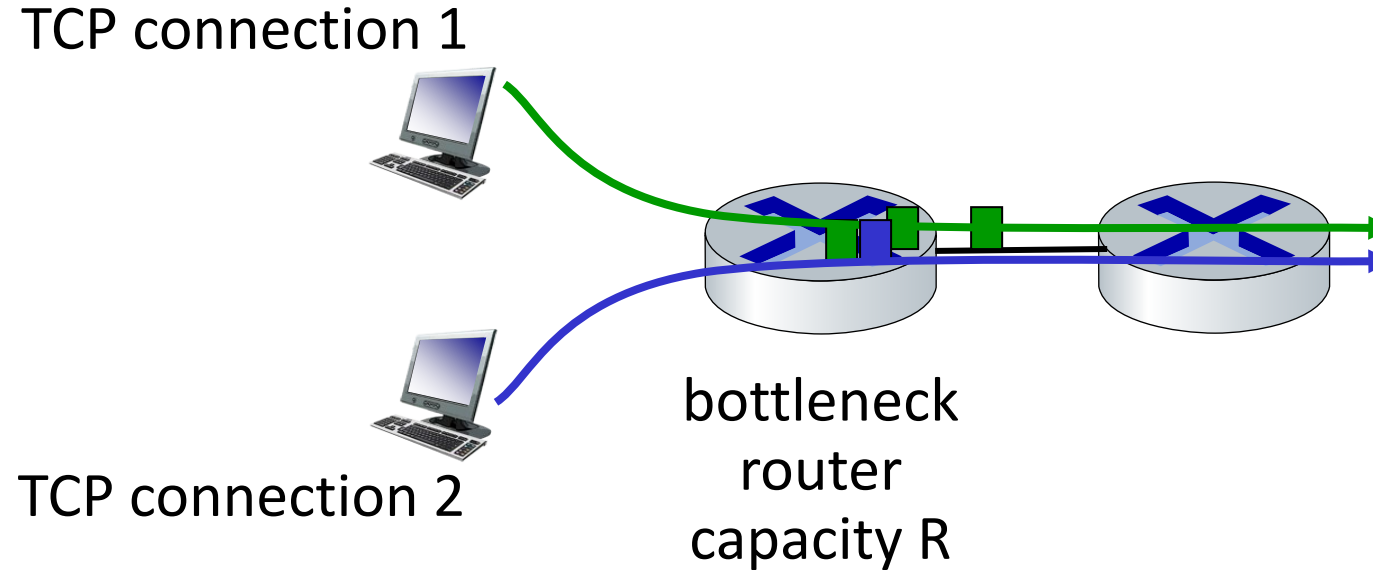
TCP deployments often implement *network-assisted* congestion control:

- two bits in IP header (ToS field) marked *by network router* to indicate congestion
  - *policy* to determine marking chosen by network operator
- congestion indication carried to destination
- destination sets ECE bit on ACK segment to notify sender of congestion
- involves both IP (IP header ECN bit marking) and TCP (TCP header C,E bit marking)



# TCP fairness

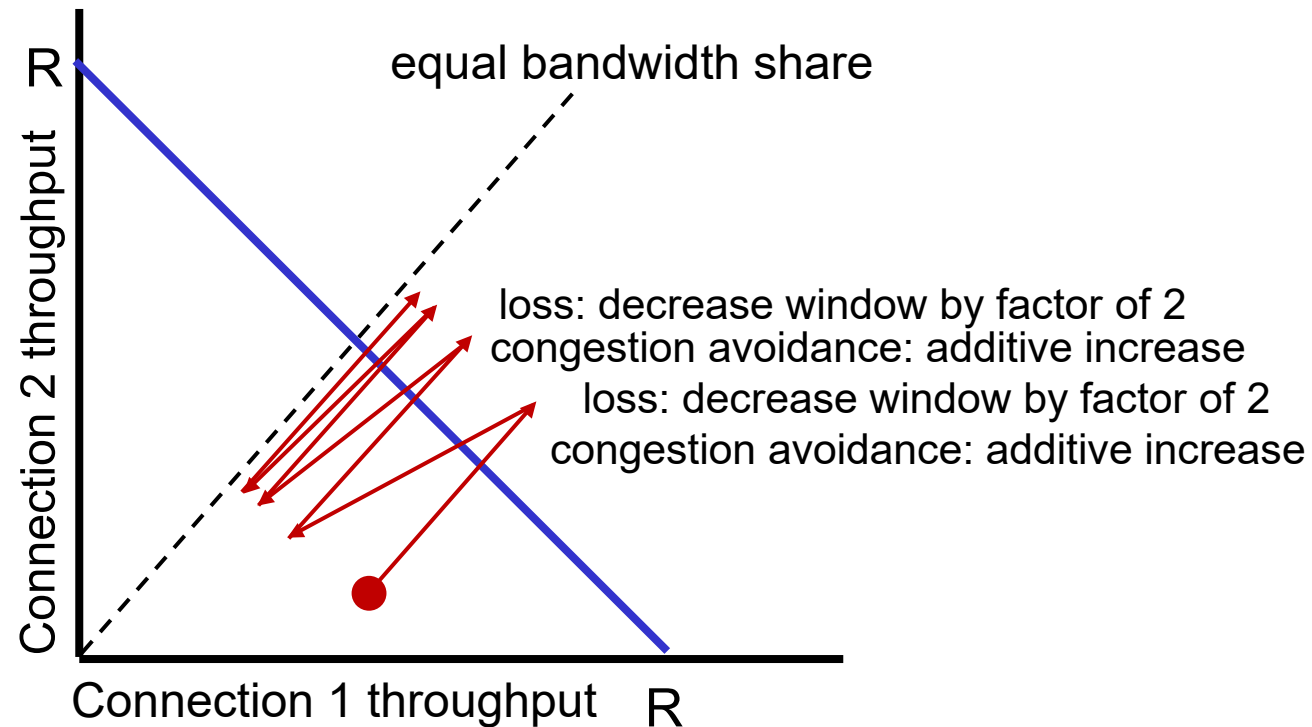
**Fairness goal:** if  $K$  TCP sessions share same bottleneck link of bandwidth  $R$ , each should have average rate of  $R/K$



# Q: is TCP Fair?

Example: two competing TCP sessions:

- additive increase gives slope of 1, as throughput increases
- multiplicative decrease decreases throughput proportionally



*Is TCP fair?*

**A:** Yes, under idealized assumptions:

- same RTT
- fixed number of sessions only in congestion avoidance

# Fairness: must all network apps be “fair”?

## 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
- there is no “Internet police” policing use of congestion control

## 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  $R/2$

# Chapter 3: summary

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

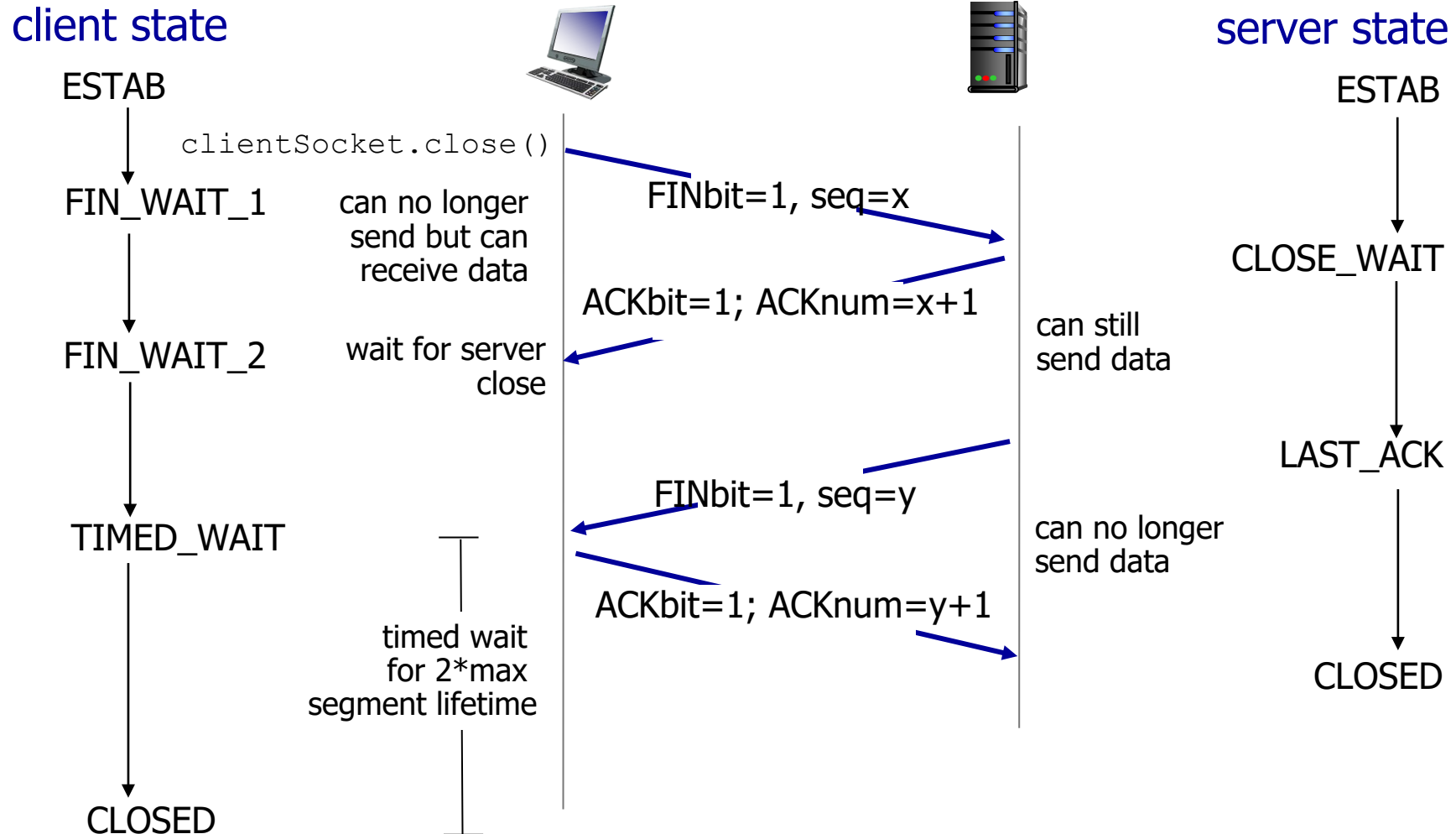
## Up next:

- leaving the network “edge” (application, transport layers)
- into the network “core”
- two network-layer chapters:
  - data plane
  - control plane

# Additional Chapter 3 slides

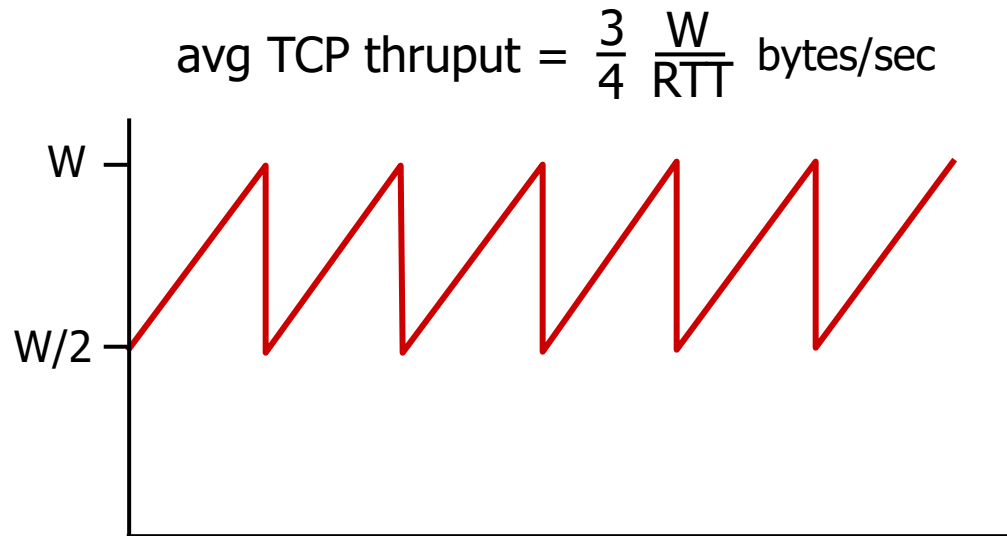


# Closing a TCP connection



# TCP throughput

- avg. TCP throughput as function of window size, RTT?
  - ignore slow start, assume there is 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. throughput is  $3/4W$  per RTT



# 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!*

- versions of TCP for long, high-speed scenarios