

# Chapter 3

## Transport Layer

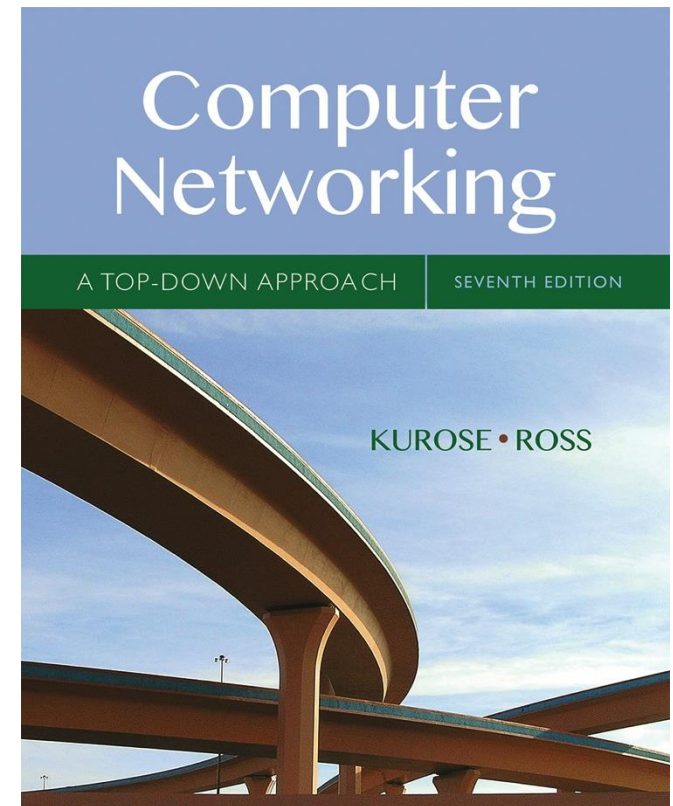
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## *Computer Networking: A Top Down Approach*

7<sup>th</sup> edition

Jim Kurose, Keith Ross

Pearson/Addison Wesley

April 2016

# Chapter 3: Transport Layer

## our goals:

- understand principles behind transport layer services:
  - multiplexing, demultiplexing
  - reliable data transfer
  - flow control
  - congestion control
- learn about Internet transport layer protocols:
  - UDP: connectionless transport
  - TCP: connection-oriented reliable transport
  - TCP congestion control

# 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

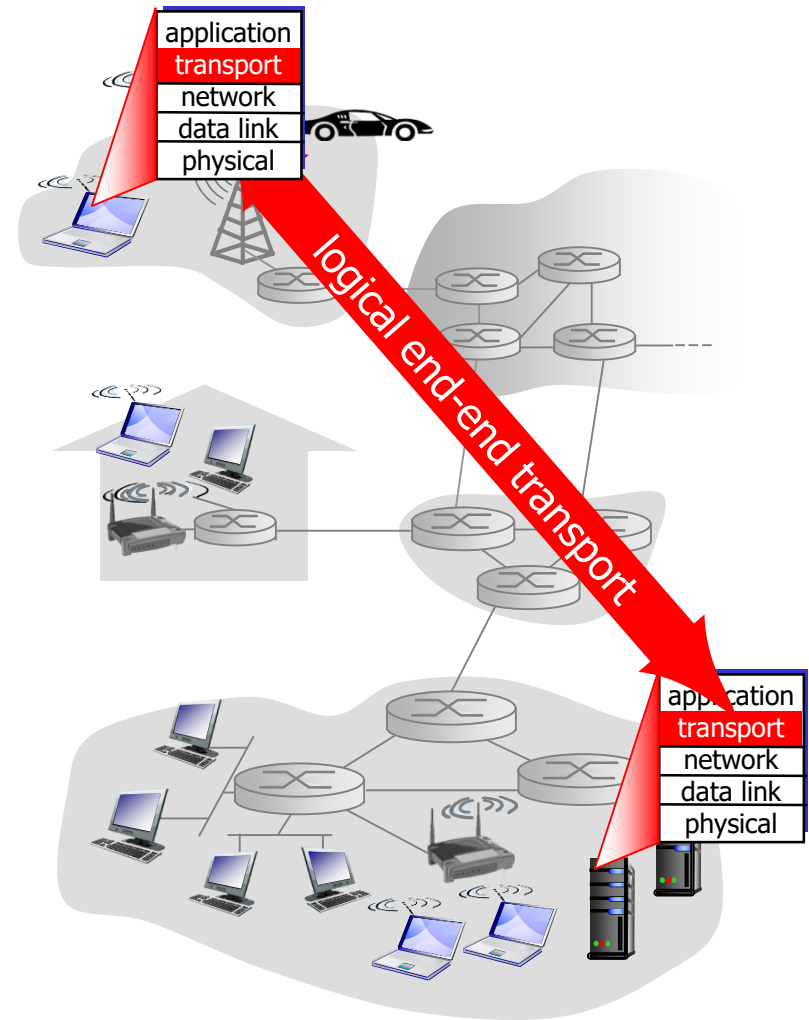
- segment structure
- reliable data transfer
- flow control
- connection management

## 3.6 principles of congestion control

## 3.7 TCP congestion control

# Transport services and protocols

- provide *logical communication* between app processes running on different hosts
- transport protocols run in end systems
  - send side: breaks app messages into *segments*, passes to network layer
  - rcv side: reassembles segments into messages, passes to app layer
- more than one transport protocol available to apps
  - Internet: TCP and UDP



# Transport vs. network layer

- *network layer*: logical communication between hosts
- *transport layer*: logical communication between processes
  - relies on, enhances, network layer services

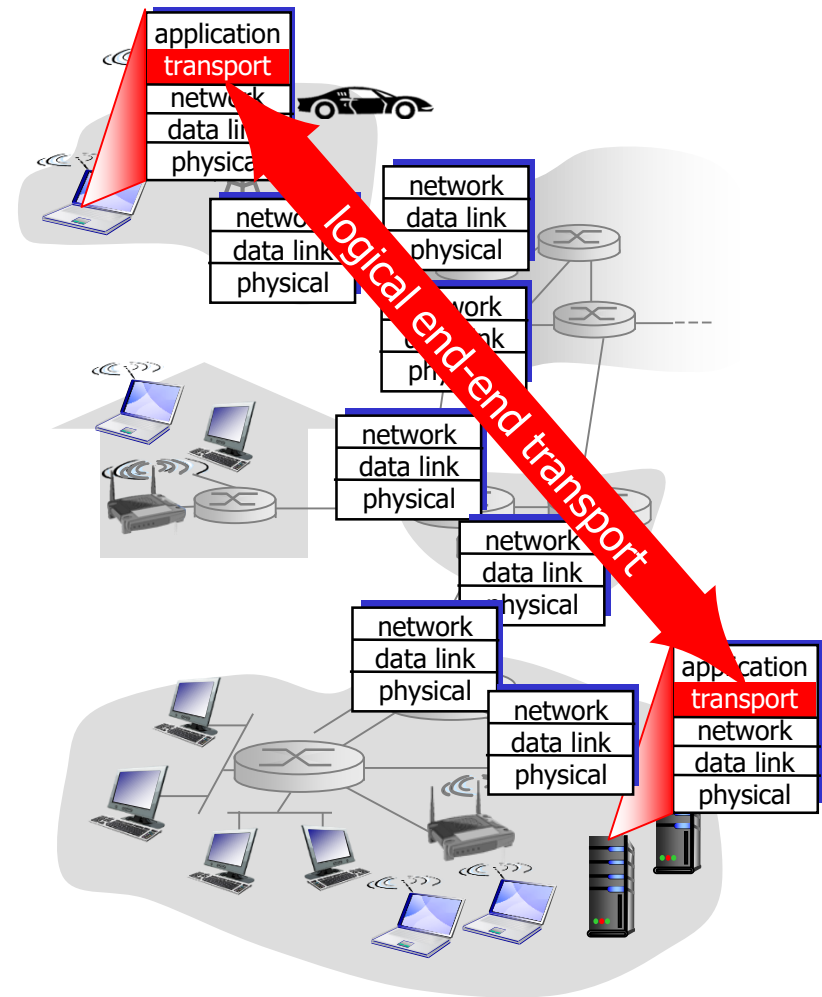
## *household analogy:*

*12 kids in Ann's house sending letters to 12 kids in Bill's house:*

- hosts = houses
- processes = kids
- app messages = letters in envelopes
- transport protocol = Ann and Bill who demux to in-house siblings
- network-layer protocol = postal service

# Internet transport-layer protocols

- reliable, in-order delivery (TCP)
  - congestion control
  - flow control
  - connection setup
- unreliable, unordered delivery: UDP
  - no-frills extension of “best-effort” IP
- services not available:
  - delay guarantees
  - bandwidth guarantees



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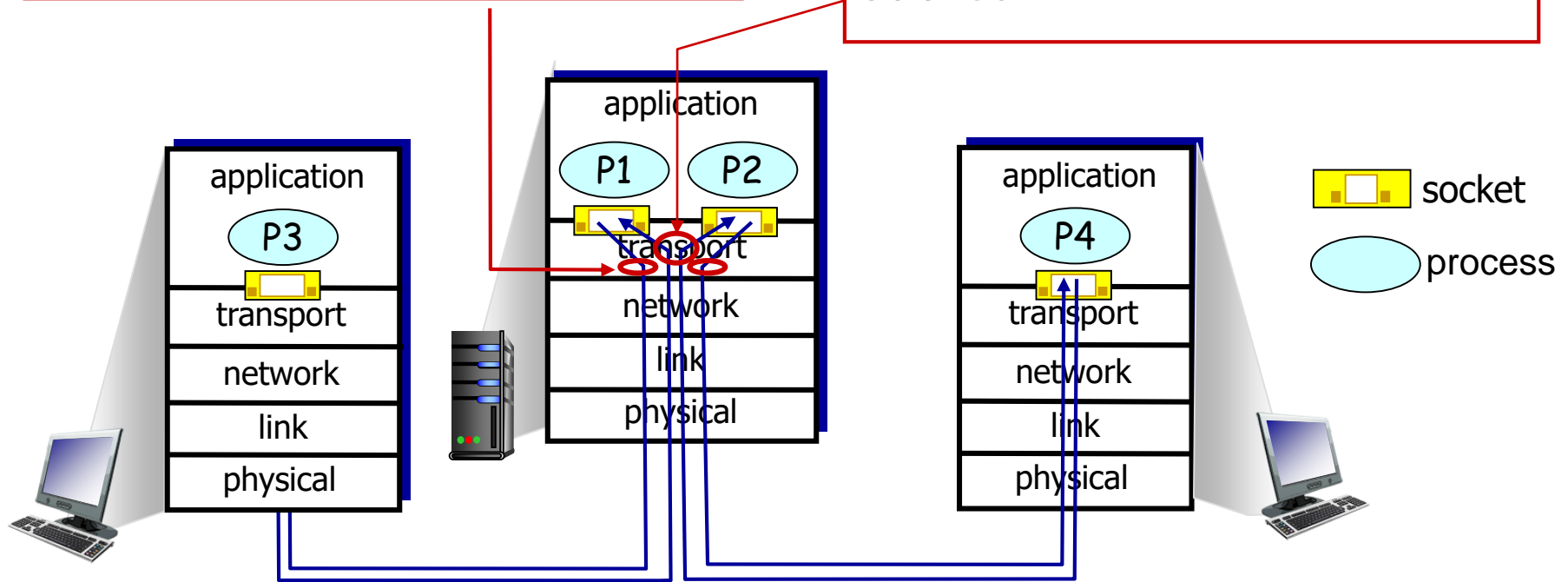
# Multiplexing/demultiplexing

*multiplexing at sender:*

handle data from multiple sockets, add transport header (later used for demultiplexing)

*demultiplexing at receiver:*

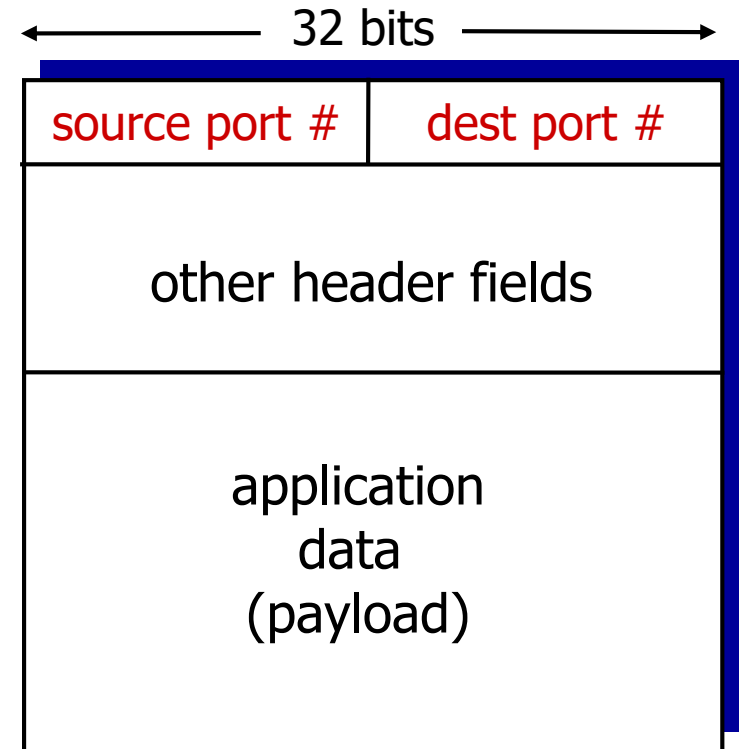
use header info to deliver received segments to correct socket





# How demultiplexing works

- host receives IP datagrams
  - each datagram has source IP address, destination IP address
  - each datagram carries one transport-layer segment
  - each segment has source, destination port number
- host uses *IP addresses & port numbers* to direct segment to appropriate socket



TCP/UDP segment format

# Connectionless demultiplexing

- *recall*: created socket has host local port #:  

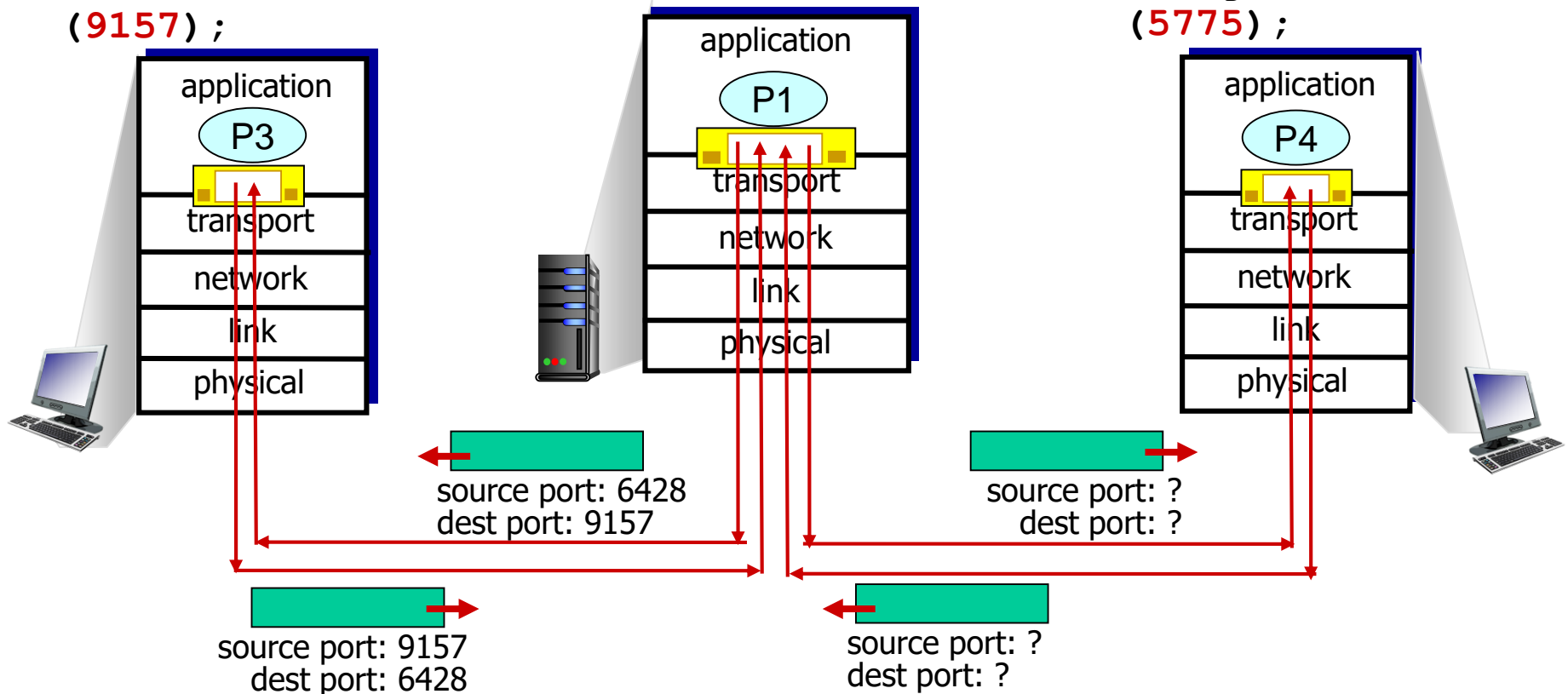
```
DatagramSocket mySocket1  
= new DatagramSocket(12534);
```
  - *recall*: when creating datagram to send into UDP socket, must specify
    - destination IP address
    - destination port #
  - when host receives UDP segment:
    - checks destination port # in segment
    - directs UDP segment to socket with that port #
- 
- IP datagrams with *same dest. port #*, but different source IP addresses and/or source port numbers will be directed to *same socket* at dest

# Connectionless demux: example

```
DatagramSocket  
mySocket2 = new  
DatagramSocket  
(9157) ;
```

```
DatagramSocket  
serverSocket = new  
DatagramSocket  
(6428) ;
```

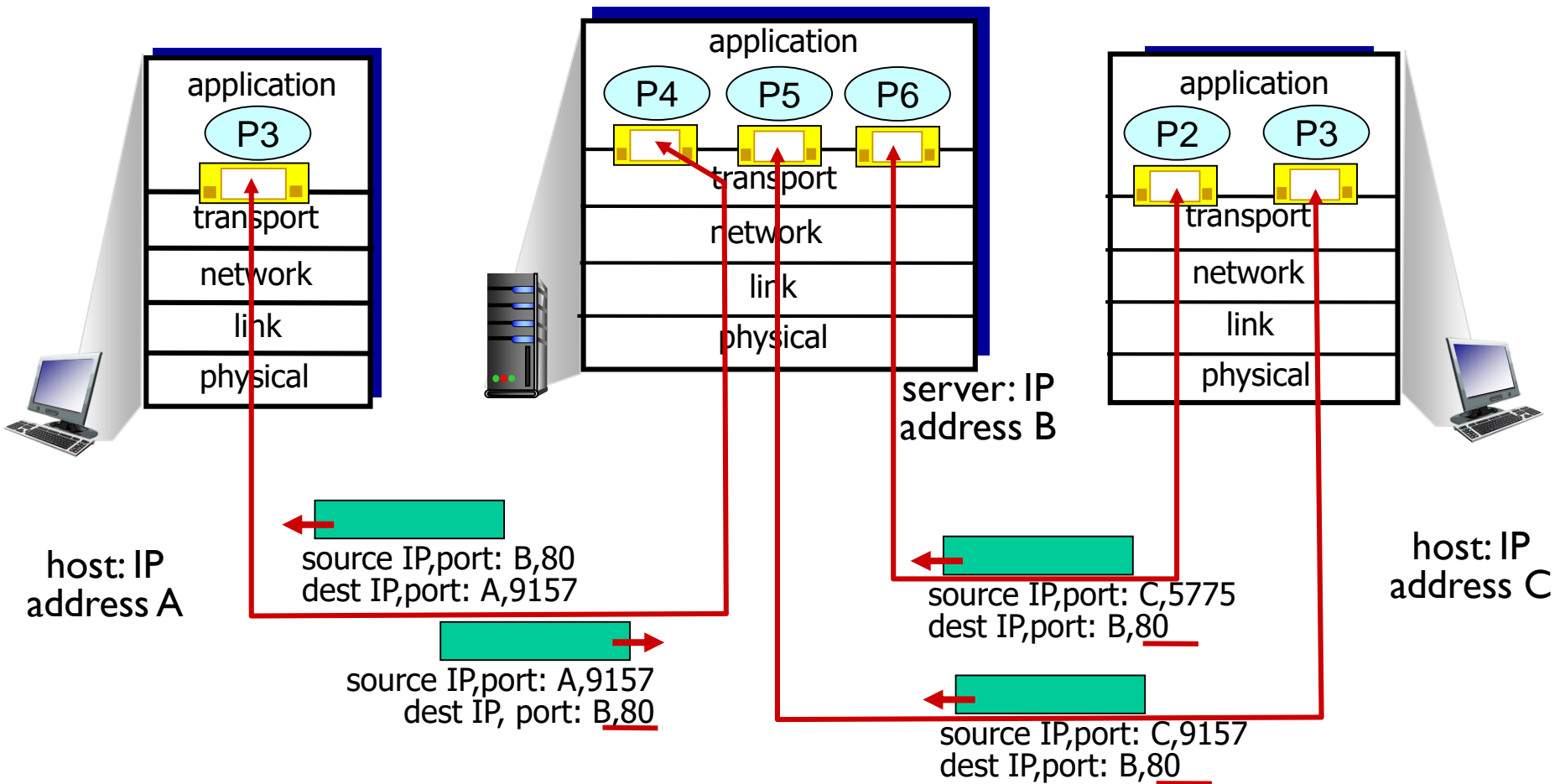
```
DatagramSocket  
mySocket1 = new  
DatagramSocket  
(5775) ;
```



# Connection-oriented demux

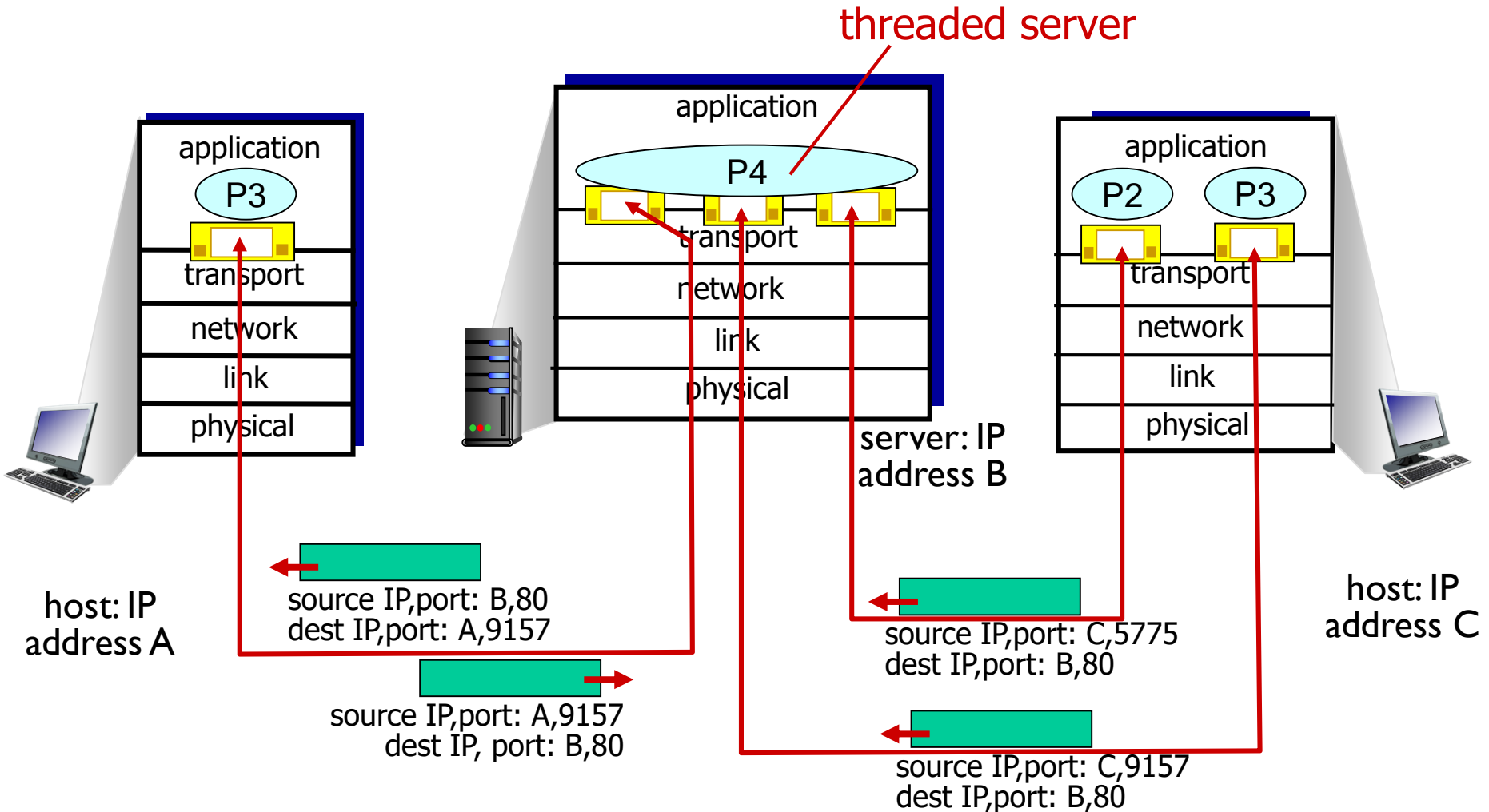
- TCP socket identified by 4-tuple:
  - source IP address
  - source port number
  - dest IP address
  - dest port number
- demux: receiver uses all four values to direct segment to appropriate socket
- server host may support many simultaneous TCP sockets:
  - each socket identified by its own 4-tuple
- web servers have different sockets for each connecting client
  - non-persistent HTTP will have different socket for each request

# Connection-oriented demux: example



three segments, all destined to IP address: B,  
dest port: 80 are demultiplexed to *different* sockets

# Connection-oriented demux: example



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- segment structure
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- flow control
- connection management

3.6 principles of congestion control

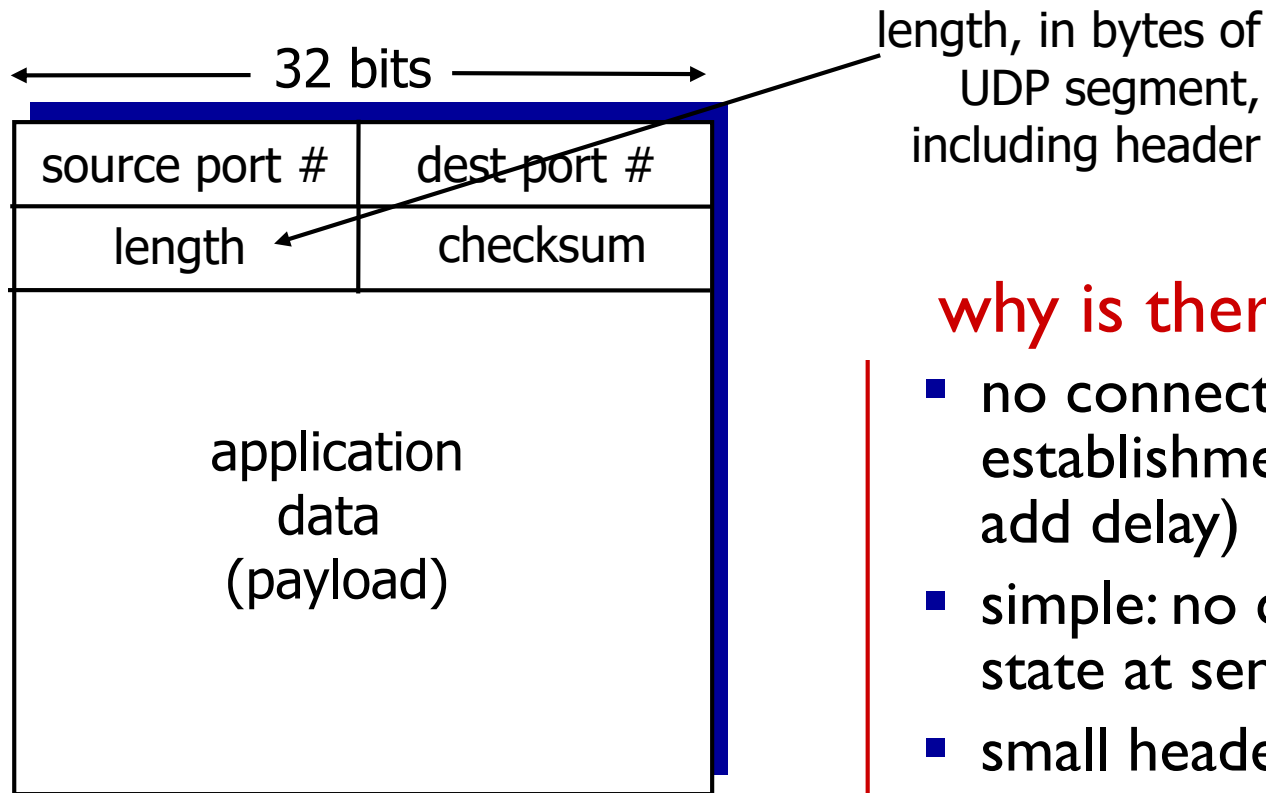
3.7 TCP congestion control

# UDP: User Datagram Protocol [RFC 768]

- “no frills,” “bare bones”  
Internet transport protocol
- “best effort” service, UDP segments may be:
  - lost
  - delivered out-of-order to app
- *connectionless*:
  - no handshaking between UDP sender, receiver
  - each UDP segment handled independently of others
- UDP use:
  - streaming multimedia apps (loss tolerant, rate sensitive)
  - DNS
  - SNMP
- reliable transfer over UDP:
  - add reliability at application layer
  - application-specific error recovery!



# UDP: segment header



UDP segment format

## why is there a UDP?

- no connection establishment (which can add delay)
- simple: no connection state at sender, receiver
- small header size
- no congestion control: UDP can blast away as fast as desired

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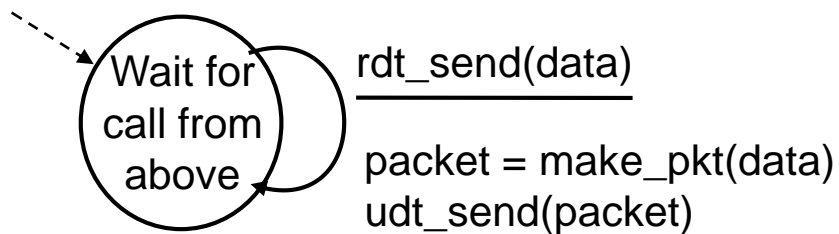
- segment structure
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3.6 principles of congestion control

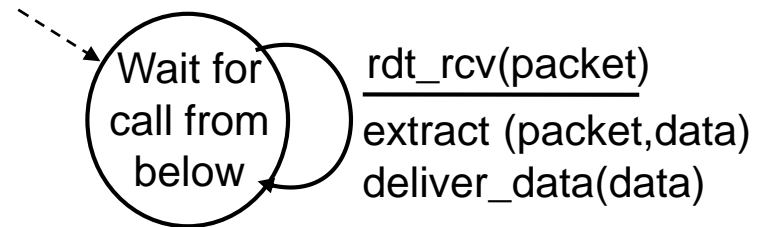
3.7 TCP congestion control

# rdt1.0: reliable transfer over a reliable channel

- underlying channel perfectly reliable
  - no bit errors
  - no loss of packets
- separate FSMs for sender, receiver:
  - sender sends data into underlying channel
  - receiver reads data from underlying channel



sender



receiver

# rdt2.0: channel with bit errors

- underlying channel may flip bits in packet
  - checksum to detect bit errors
- *the question*: how to recover from errors:

*How do humans recover from “errors”  
during conversation?*

# rdt3.0: channels with errors *and* loss

## new assumption:

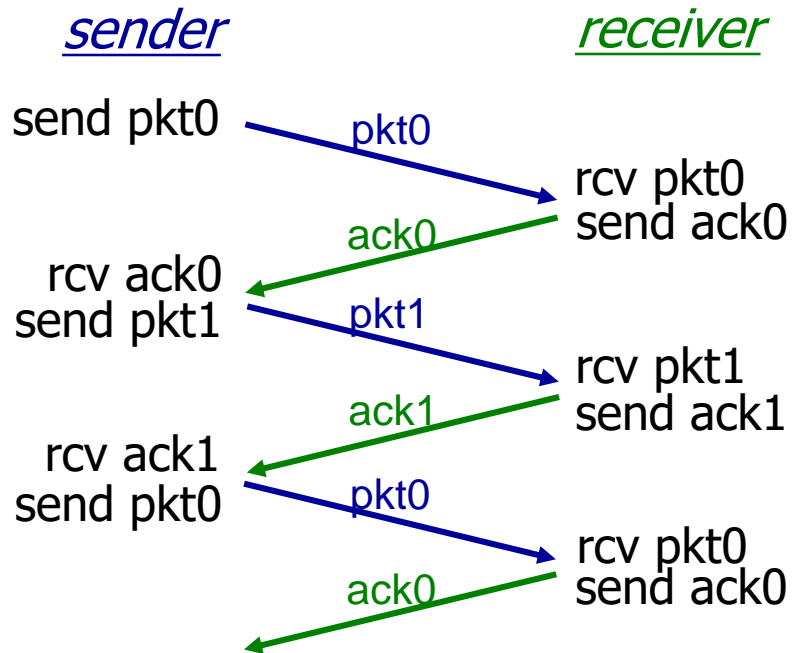
underlying channel can also lose packets (data, ACKs)

- checksum, seq. #, ACKs, retransmissions will be of help ... but not enough

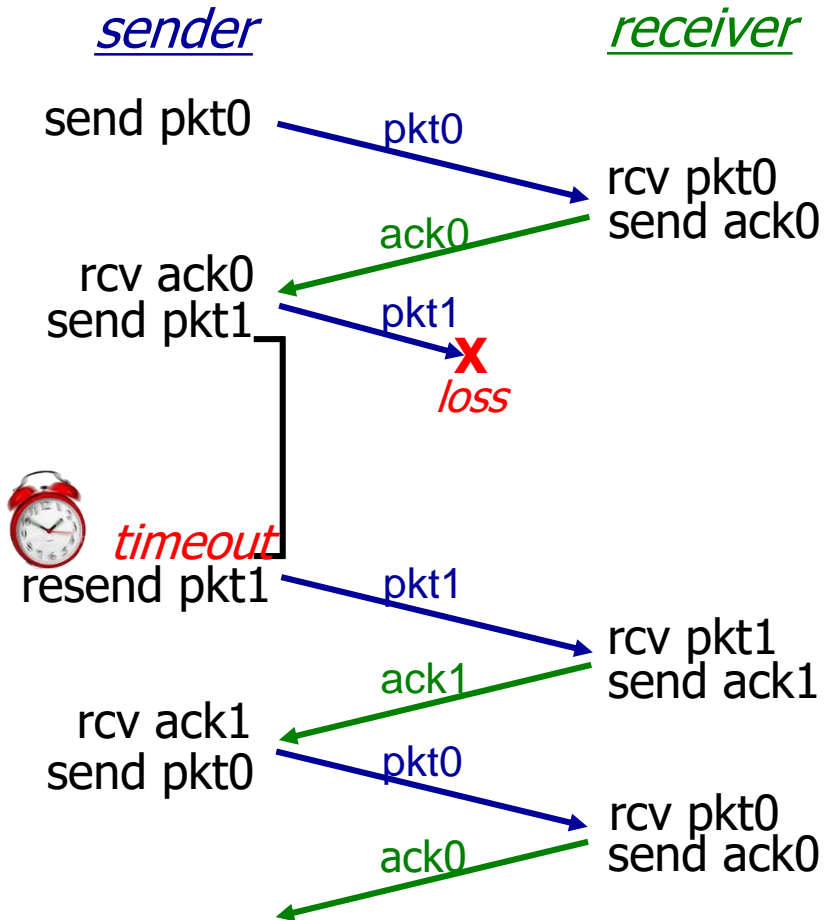
approach: sender waits “reasonable” amount of time for ACK

- retransmits if no ACK received in this time
- if pkt (or ACK) just delayed (not lost):
  - retransmission will be duplicate, but seq. #'s already handles this
  - receiver must specify seq # of pkt being ACKed
- requires countdown timer

# rdt3.0 in action

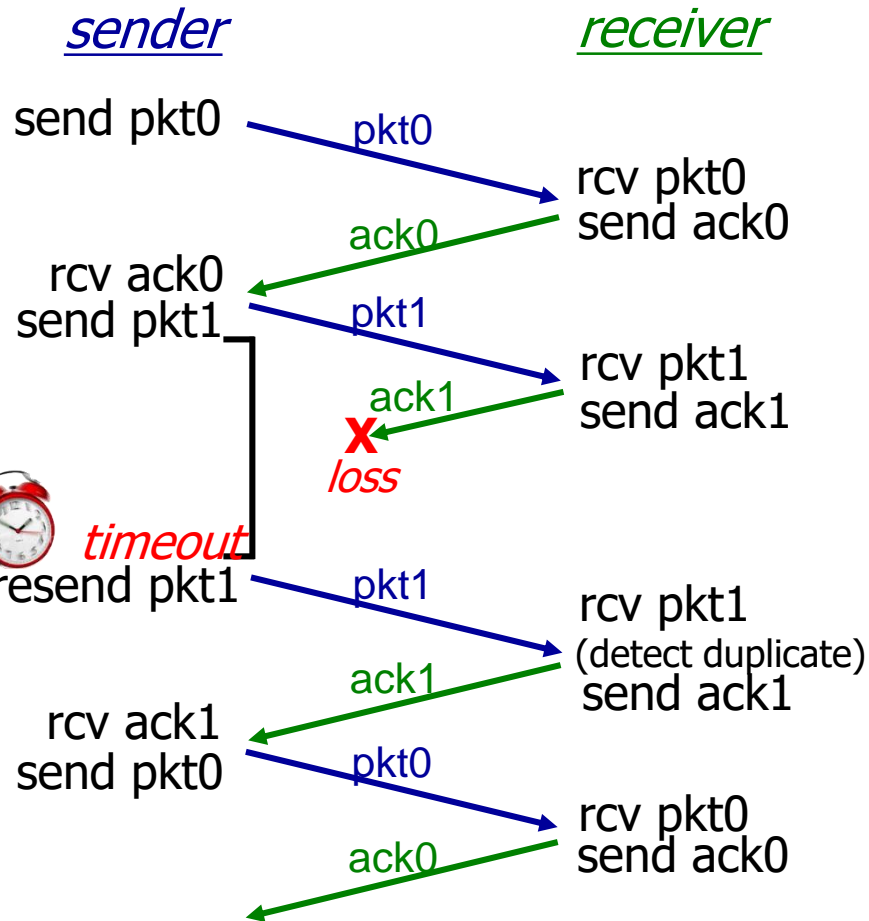


(a) no loss

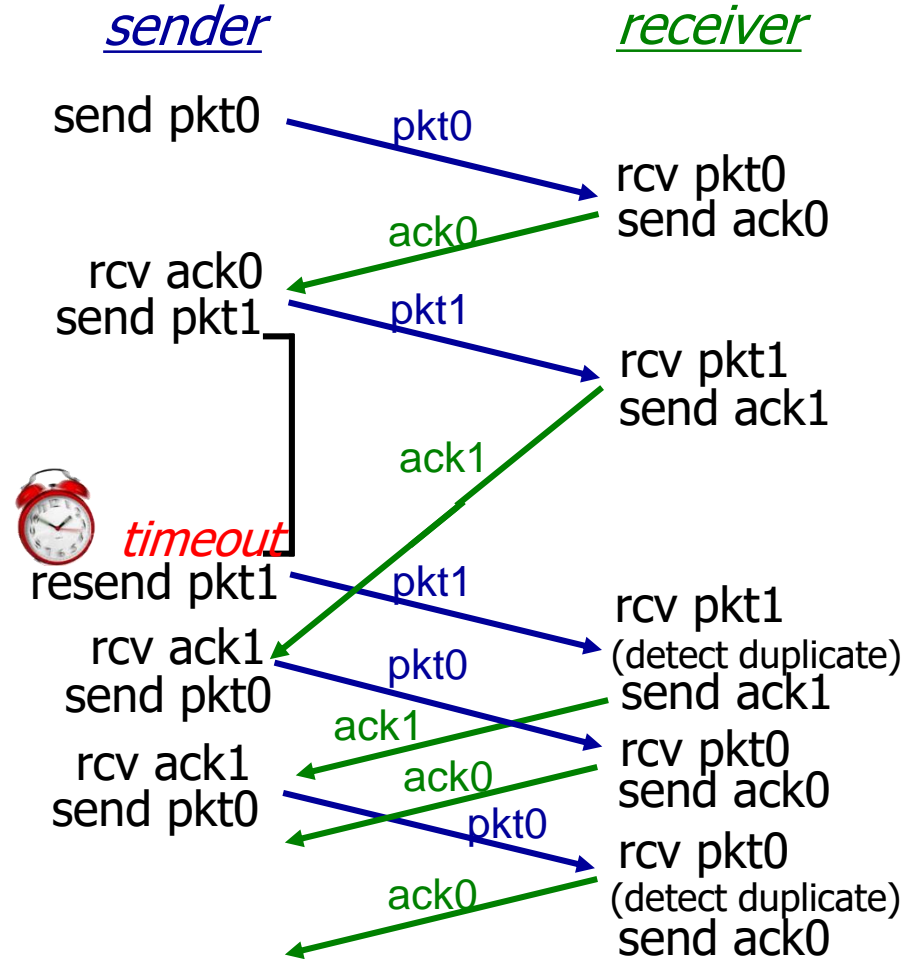


(b) packet loss

# rdt3.0 in action



(c) ACK loss



(d) premature timeout/ delayed ACK

# Performance of rdt3.0

- rdt3.0 is correct, but performance stinks
- e.g.: 1 Gbps link, 15 ms prop. delay, 8000 bit packet:

$$D_{trans} = \frac{L}{R} = \frac{8000 \text{ bits}}{10^9 \text{ bits/sec}} = 8 \text{ microseconds}$$

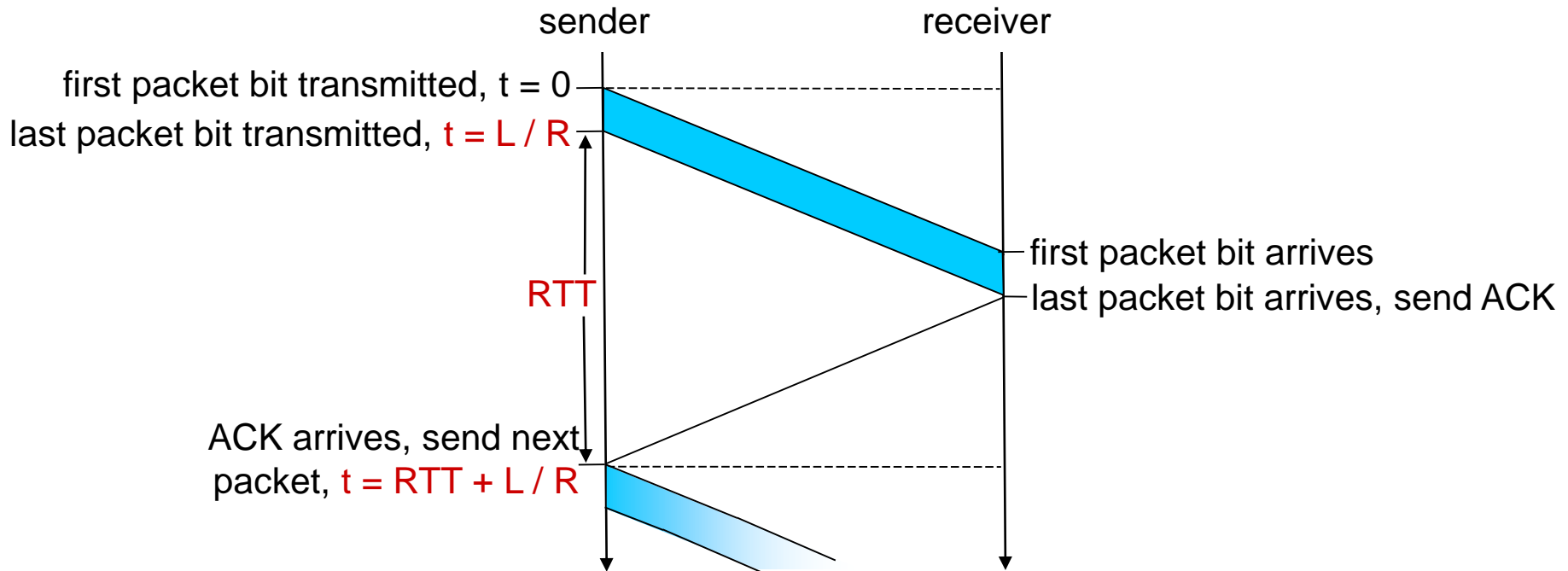
- $U_{\text{sender}}$ : **utilization** – fraction of time sender busy sending

$$U_{\text{sender}} = \frac{L / R}{RTT + L / R} = \frac{.008}{30.008} = 0.00027$$

- if RTT=30 msec, 1KB pkt every 30 msec: 33kB/sec thruput over 1 Gbps link
- network protocol limits use of physical resources!



# rdt3.0: stop-and-wait operation

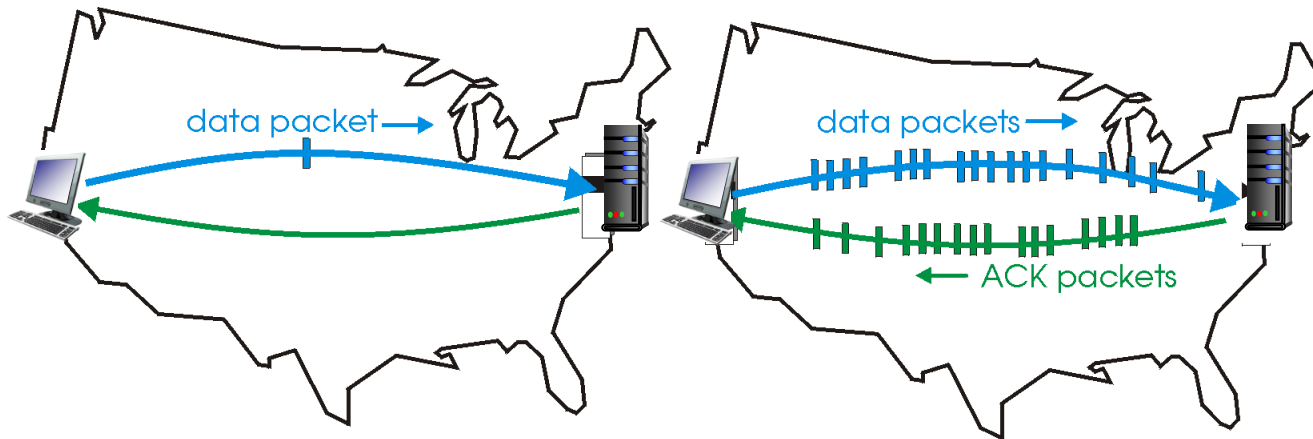


$$U_{\text{sender}} = \frac{L/R}{RTT + L/R} = \frac{.008}{30.008} = 0.00027$$

# Pipelined protocols

**pipelining:** sender allows multiple, “in-flight”, yet-to-be-acknowledged pkts

- range of sequence numbers must be increased
- buffering at sender and/or receiver

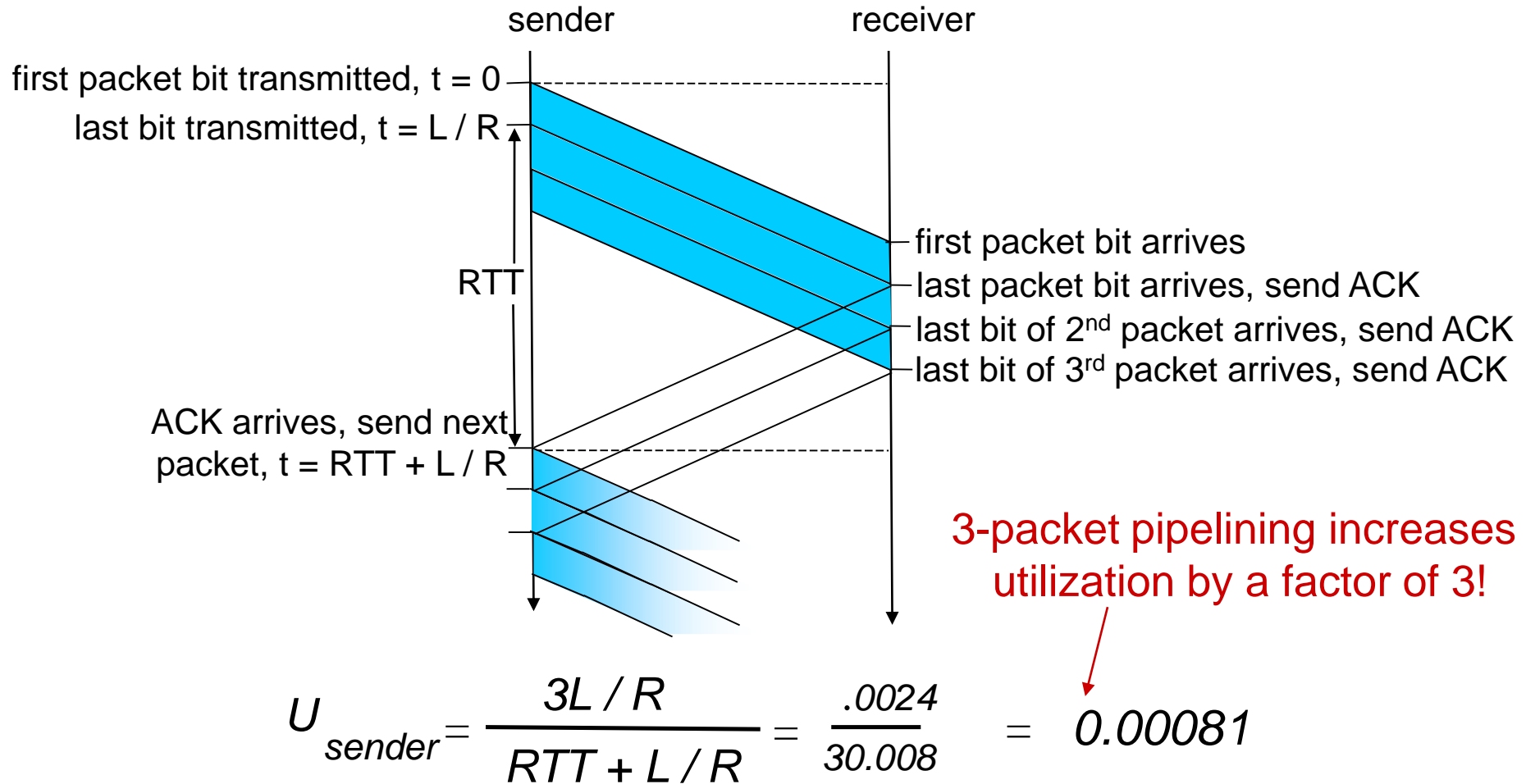


(a) a stop-and-wait protocol in operation

(b) a pipelined protocol in operation

- two generic forms of pipelined protocols: *go-Back-N*, *selective repeat*

# Pipelining: increased utilization



# Pipelined protocols: overview

## Go-back-N:

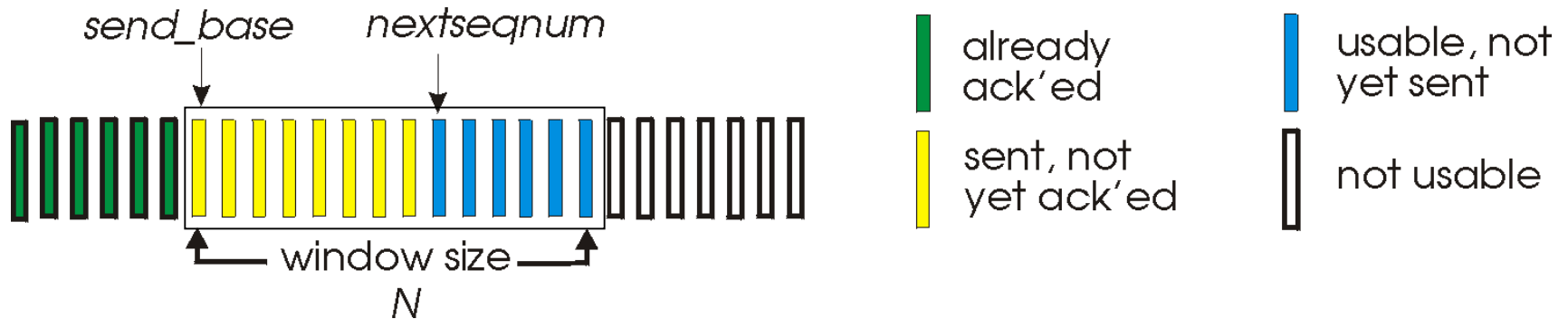
- sender can have up to N unacked packets in pipeline
- receiver only sends *cumulative ack*
  - doesn't ack packet if there's a gap
- sender has timer for oldest unacked packet
  - when timer expires, retransmit *all* unacked packets

## Selective Repeat:

- sender can have up to N unack'ed packets in pipeline
- rcvr sends *individual ack* for each packet
- sender maintains timer for each unacked packet
  - when timer expires, retransmit only that unacked packet

# Go-Back-N: sender

- k-bit seq # in pkt header
- “window” of up to N, consecutive unack’ed pkts allowed



- ACK(n): ACKs all pkts up to, including seq # n - “*cumulative ACK*”
  - may receive duplicate ACKs (see receiver)
- timer for oldest in-flight pkt
- *timeout(n)*: retransmit packet n and all higher seq # pkts in window

# GBN in action

sender window (N=4)

0 1 2 3 4 5 6 7 8  
0 1 2 3 4 5 6 7 8  
0 1 2 3 4 5 6 7 8  
0 1 2 3 4 5 6 7 8

0 1 2 3 4 5 6 7 8  
0 1 2 3 4 5 6 7 8

0 1 2 3 4 5 6 7 8  
0 1 2 3 4 5 6 7 8  
0 1 2 3 4 5 6 7 8  
0 1 2 3 4 5 6 7 8

sender

send pkt0  
send pkt1  
send pkt2  
send pkt3  
(wait)

rcv ack0, send pkt4  
rcv ack1, send pkt5

ignore duplicate ACK



*pkt 2 timeout*

send pkt2  
send pkt3  
send pkt4  
send pkt5

receiver

receive pkt0, send ack0  
receive pkt1, send ack1

receive pkt3, discard,  
(re)send ack1

receive pkt4, discard,  
(re)send ack1

receive pkt5, discard,  
(re)send ack1

rcv pkt2, deliver, send ack2

rcv pkt3, deliver, send ack3

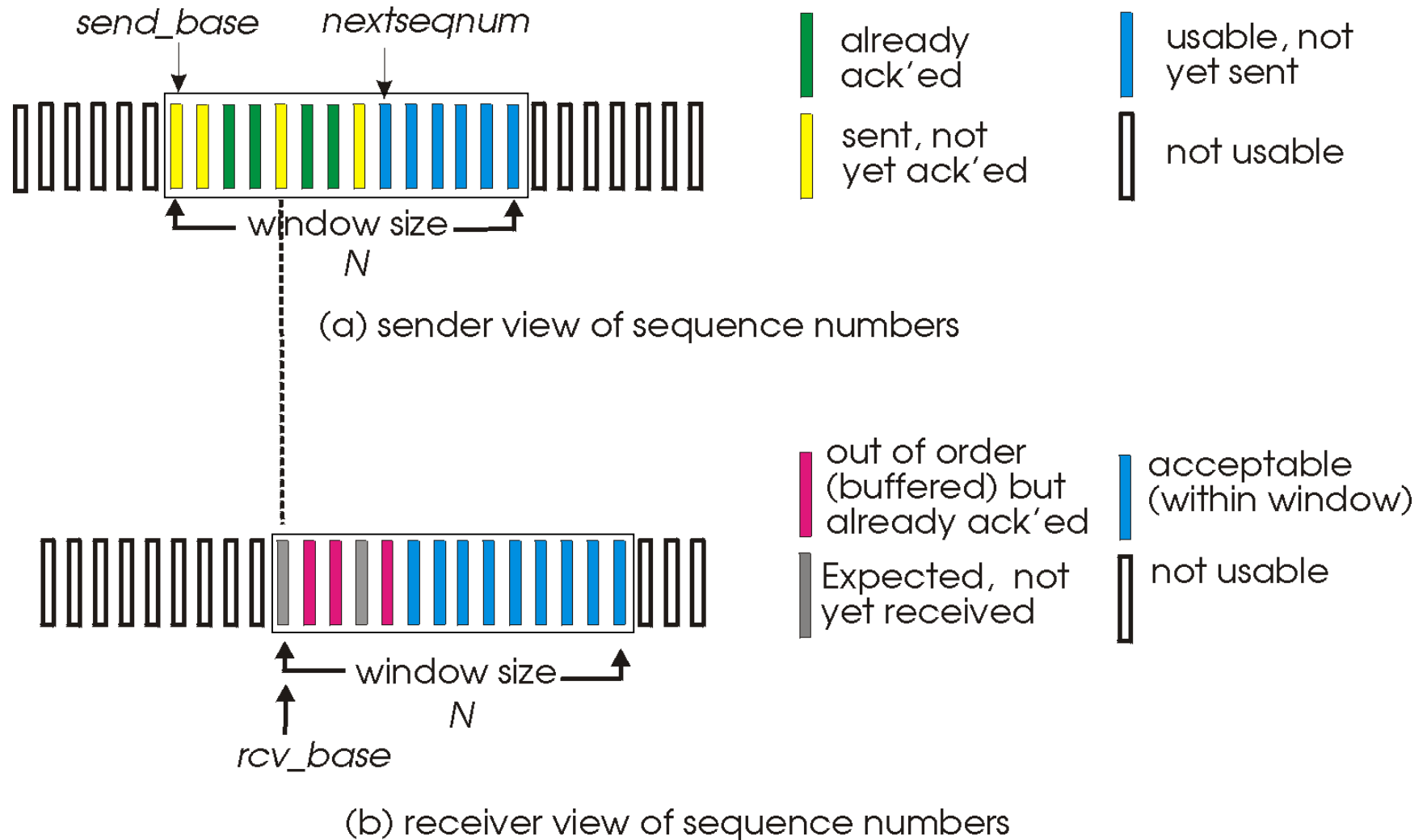
rcv pkt4, deliver, send ack4

rcv pkt5, deliver, send ack5

# Selective repeat

- receiver *individually* acknowledges all correctly received pkts
  - buffers pkts, as needed, for eventual in-order delivery to upper layer
- sender only resends pkts for which ACK not received
  - sender timer for each unACKed pkt
- sender window
  - $N$  consecutive seq #'s
  - limits seq #s of sent, unACKed pkts

# Selective repeat: sender, receiver windows





# Selective repeat

## sender

### data from above:

- if next available seq # in window, send pkt

### timeout(n):

- resend pkt n, restart timer

### ACK(n) in [sendbase, sendbase+N]:

- mark pkt n as received
- if n smallest unACKed pkt, advance window base to next unACKed seq #

## receiver

### pkt n in [rcvbase, rcvbase+N-1]

- send ACK(n)
- out-of-order: buffer
- in-order: deliver (also deliver buffered, in-order pkts), advance window to next not-yet-received pkt

### pkt n in [rcvbase-N, rcvbase-1]

- ACK(n)

### otherwise:

- ignore

# Selective repeat in action

sender window (N=4)

0 1 2 3 4 5 6 7 8

0 1 2 3 4 5 6 7 8

0 1 2 3 4 5 6 7 8

0 1 2 3 4 5 6 7 8

0 1 2 3 4 5 6 7 8

0 1 2 3 4 5 6 7 8

0 1 2 3 4 5 6 7 8

0 1 2 3 4 5 6 7 8

0 1 2 3 4 5 6 7 8

0 1 2 3 4 5 6 7 8

sender

send pkt0

send pkt1

send pkt2

send pkt3

(wait)

rcv ack0, send pkt4

rcv ack1, send pkt5

record ack3 arrived



*pkt 2 timeout*

send pkt2

record ack4 arrived

record ack5 arrived

receiver

receive pkt0, send ack0

receive pkt1, send ack1

receive pkt3, buffer,  
send ack3

receive pkt4, buffer,  
send ack4

receive pkt5, buffer,  
send ack5

rcv pkt2; deliver pkt2,  
pkt3, pkt4, pkt5; send ack2

*Q: what happens when ack2 arrives?*

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3.6 principles of congestion control

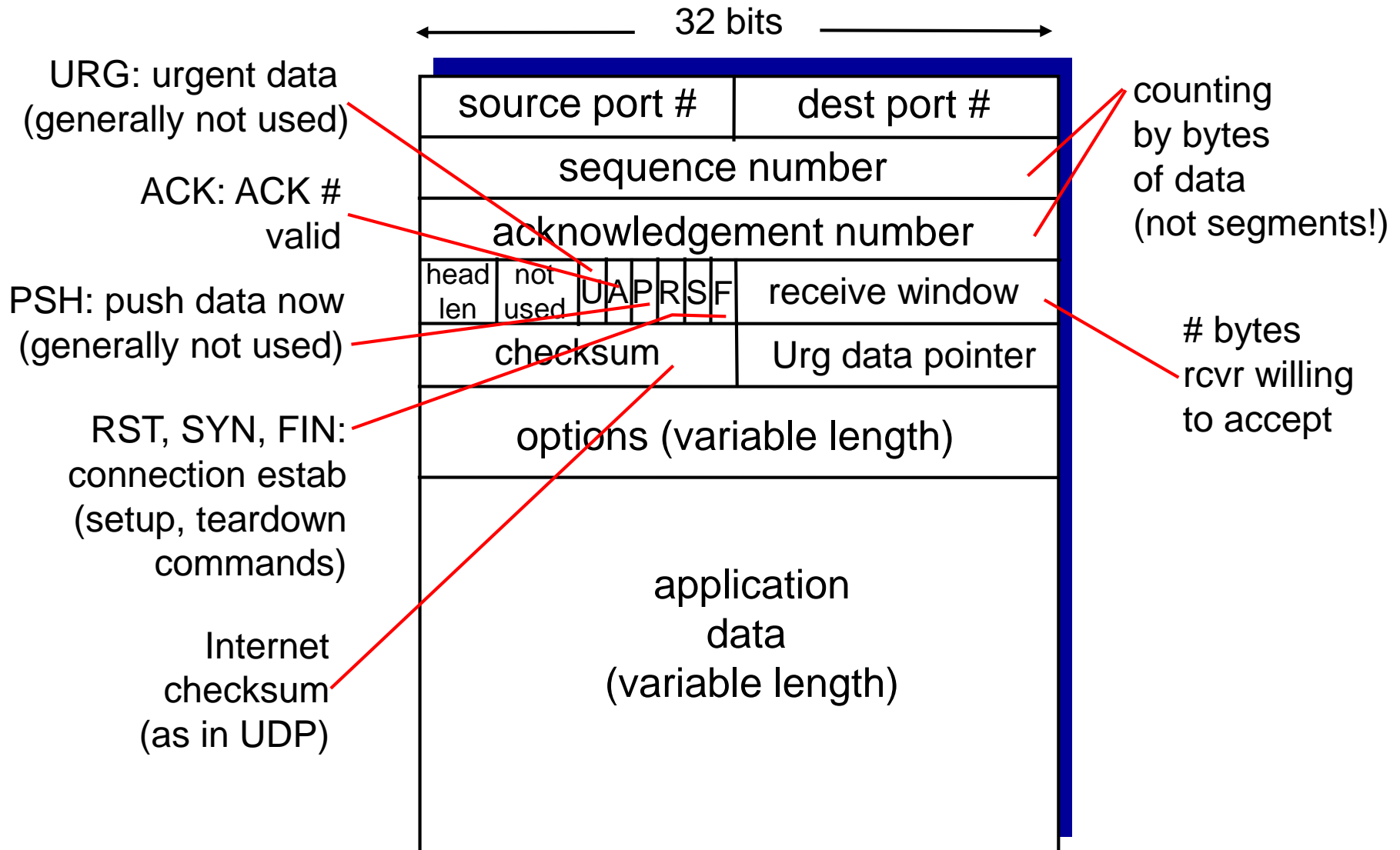
3.7 TCP congestion control

# TCP: Overview

RFCs: 793, 1122, 1323, 2018, 2581

- **point-to-point:**
  - one sender, one receiver
- **reliable, in-order *byte stream*:**
  - 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) initializes sender, receiver state before data exchange
- **flow controlled:**
  - sender will not overwhelm receiver

# TCP segment structure



# TCP seq. numbers, ACKs

## sequence numbers:

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

## acknowledgements:

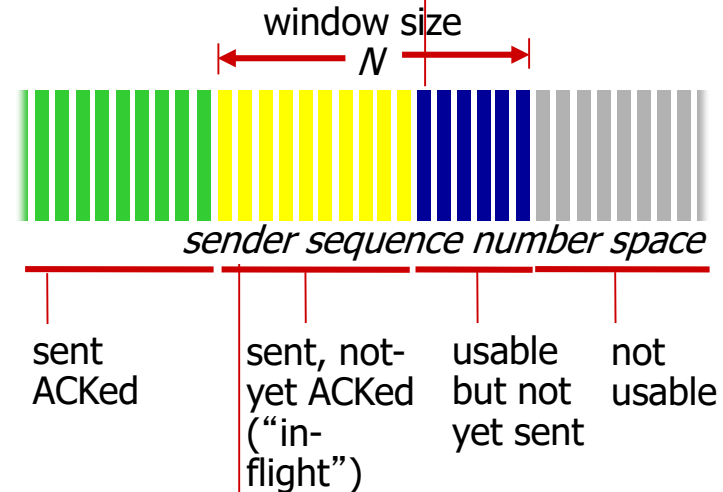
- seq # of next byte expected from other side
- cumulative ACK

**Q:** how receiver handles out-of-order segments

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

outgoing segment from sender

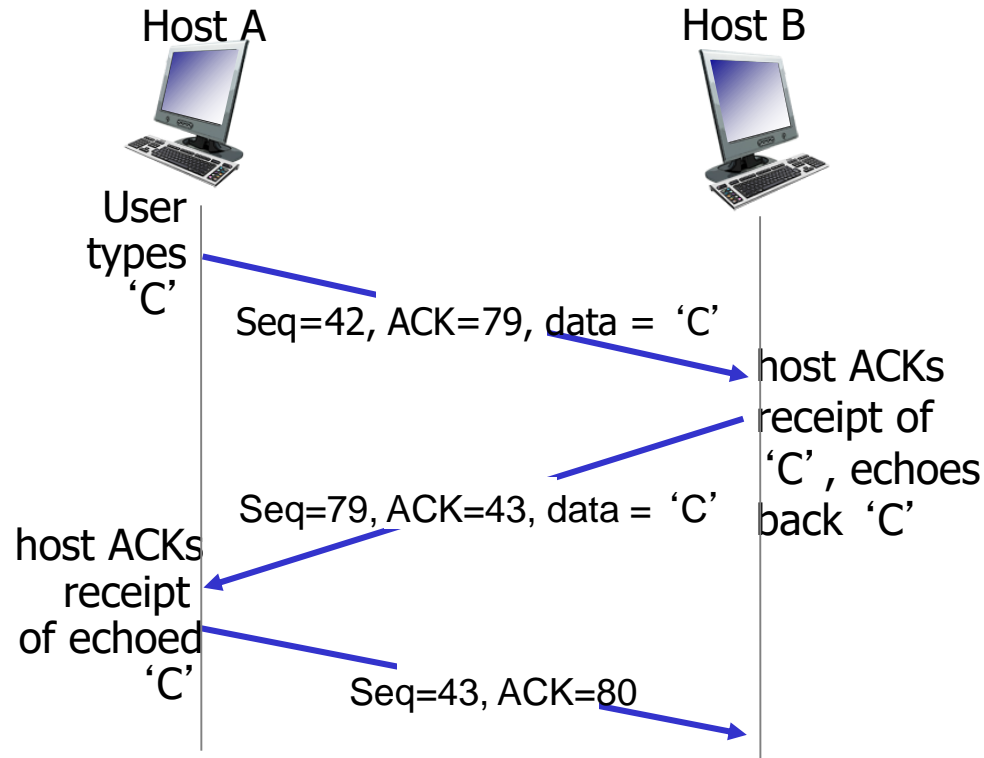
source port #	dest port #
sequence number	
acknowledgement number	
	rwnd
checksum	urg pointer



incoming segment to sender

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

# TCP seq. 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?

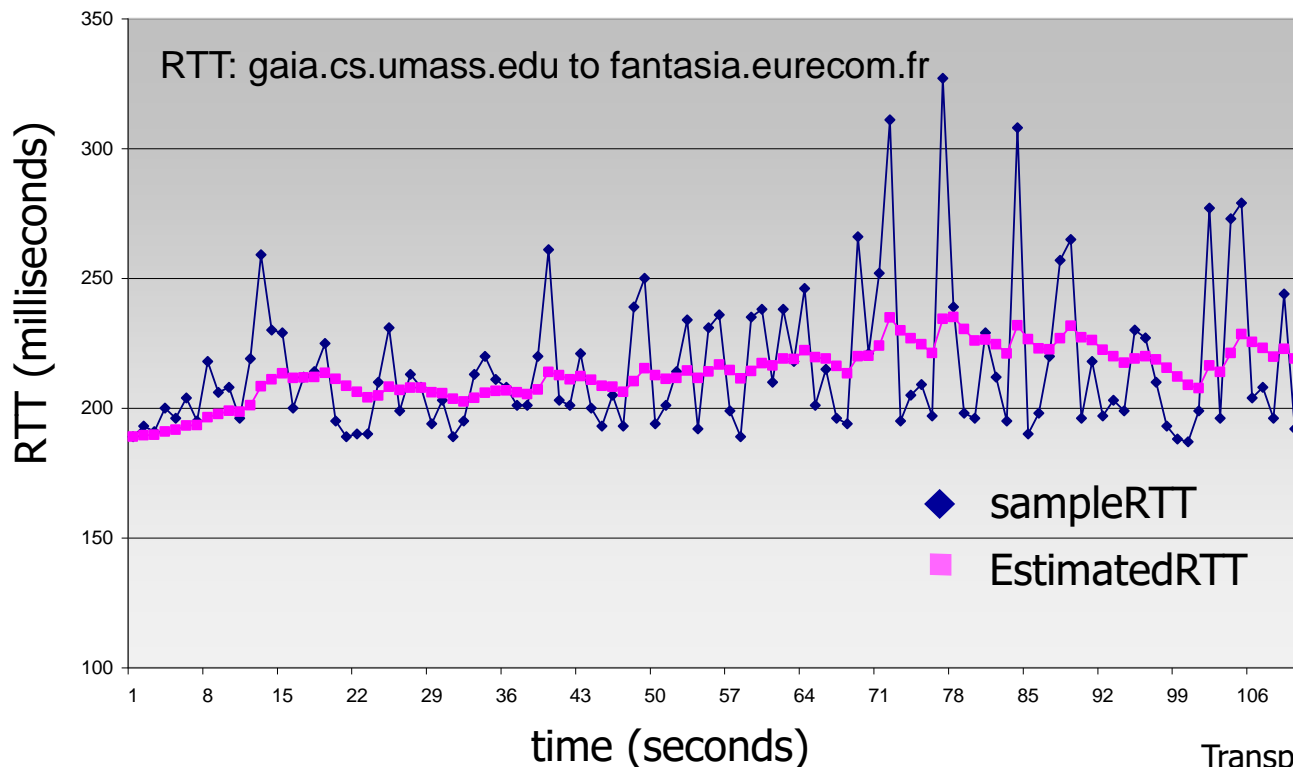
- **SampleRTT**: measured time from segment transmission until ACK receipt
  - ignore retransmissions
- **SampleRTT** will vary, want estimated RTT “smoother”
  - average several *recent* measurements, not just current **SampleRTT**



# TCP round trip time, timeout

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

- exponential weighted moving average
- 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:
$$\text{DevRTT} = (1-\beta) * \text{DevRTT} + \beta * |\text{SampleRTT} - \text{EstimatedRTT}|$$
(typically,  $\beta = 0.25$ )

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



↑  
estimated RTT

↑  
“safety margin”

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3.7 TCP congestion control

# 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`

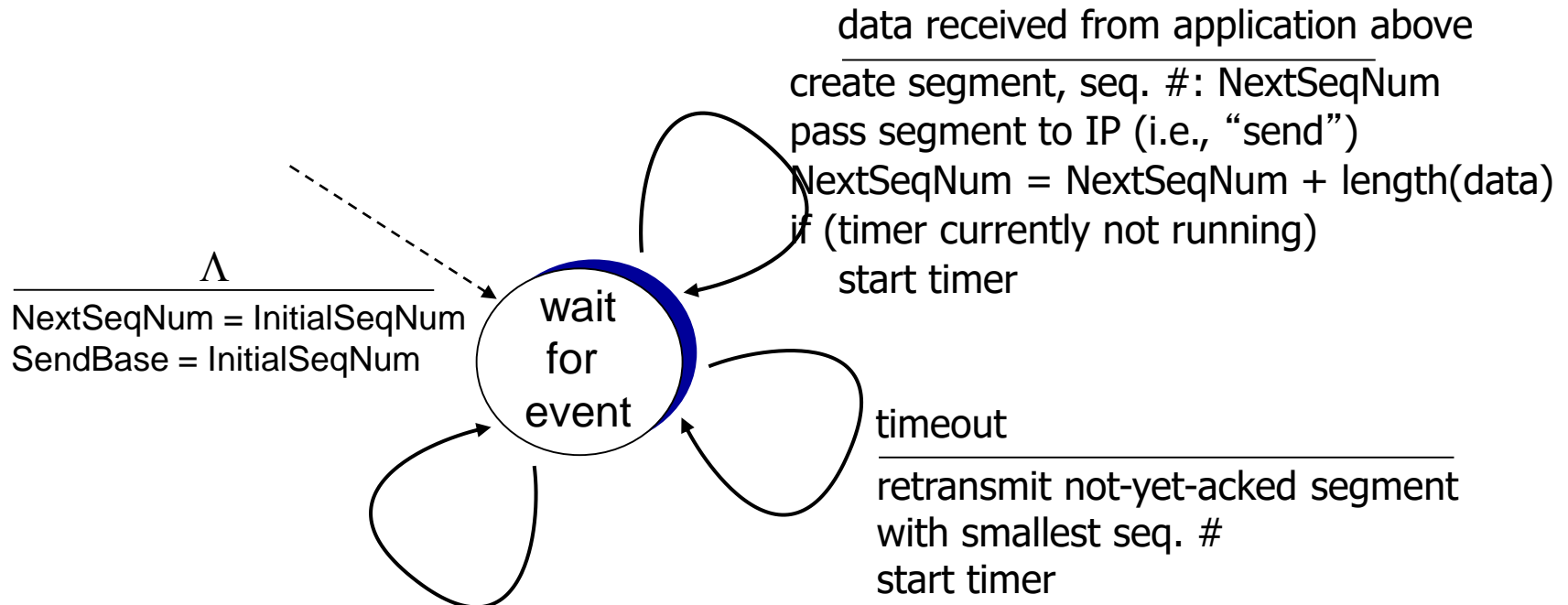
## *timeout:*

- 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

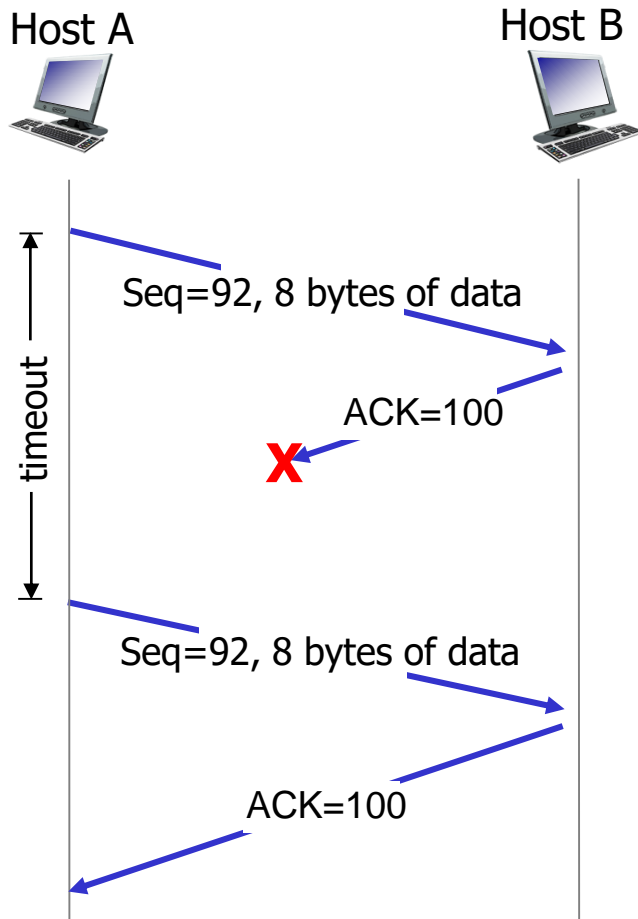
# TCP sender (simplified)



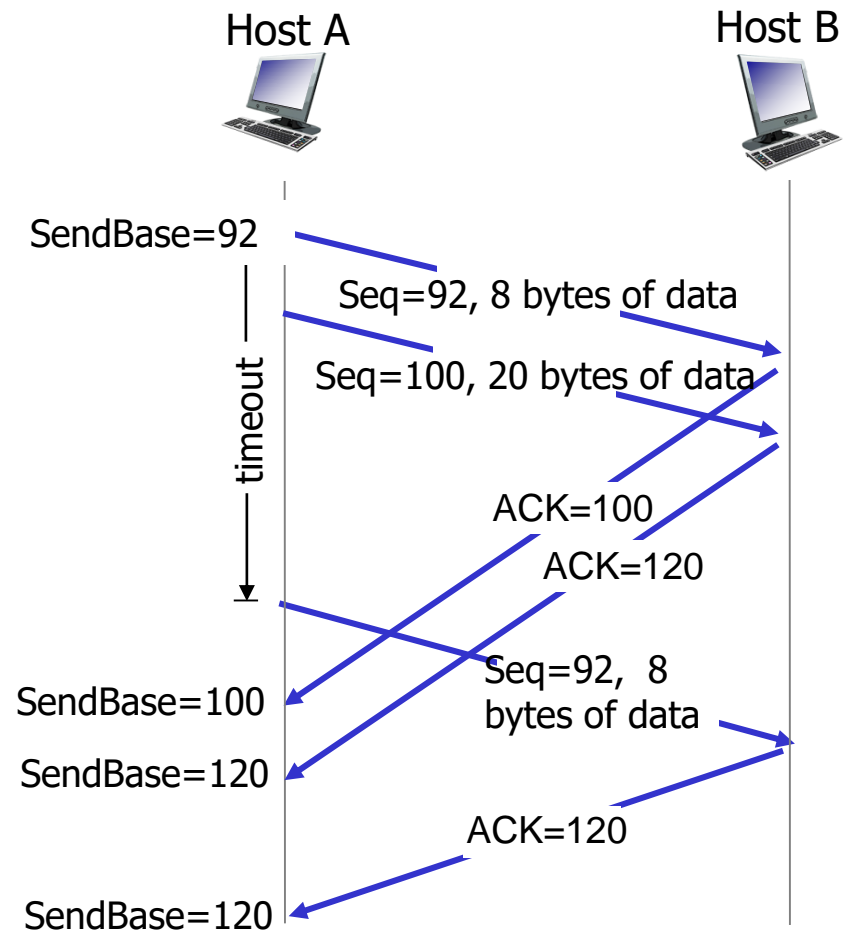
ACK received, with ACK field value y

```
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

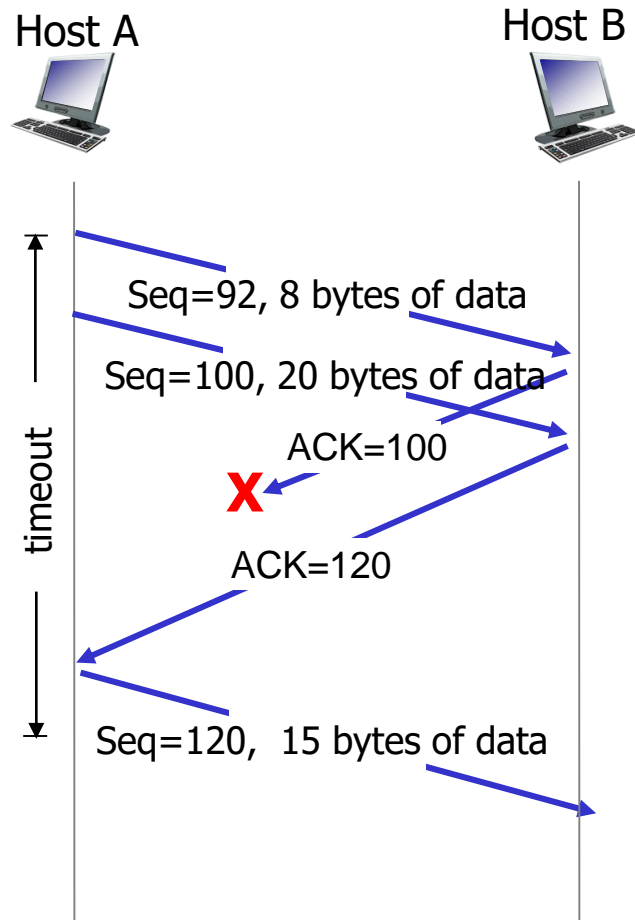


lost ACK scenario



premature timeout

# TCP: retransmission scenarios



cumulative ACK



# TCP fast retransmit

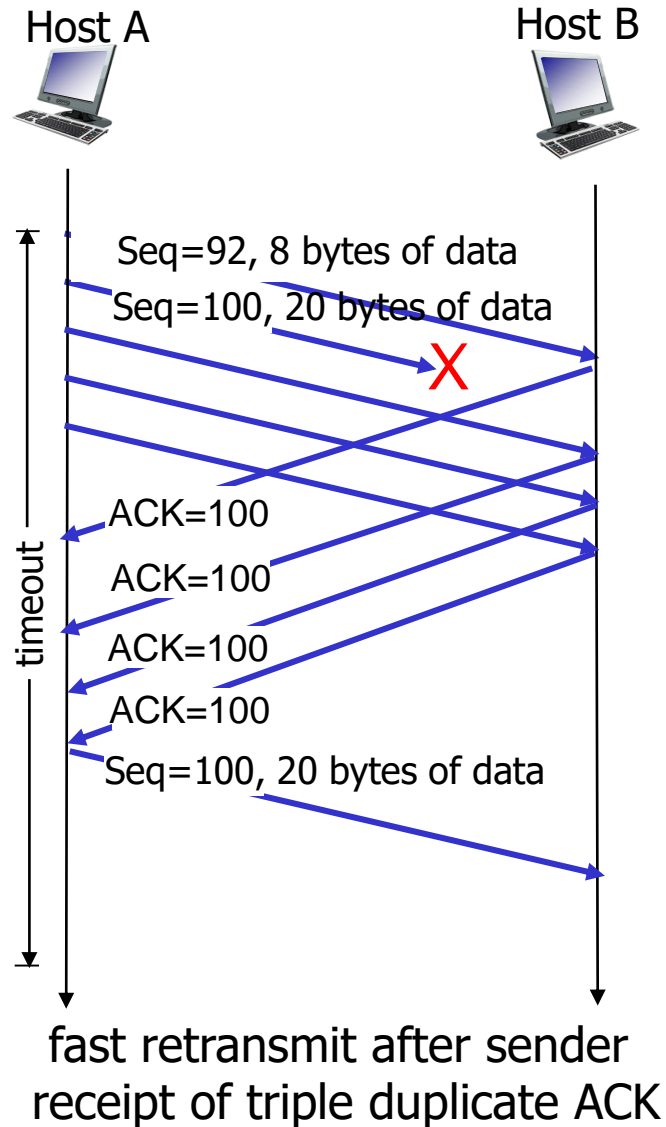
- time-out period often relatively long:
  - long delay before resending lost packet
- detect lost segments via duplicate ACKs.
  - sender often sends many segments back-to-back
  - if segment is lost, there will likely be many duplicate ACKs.

## *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



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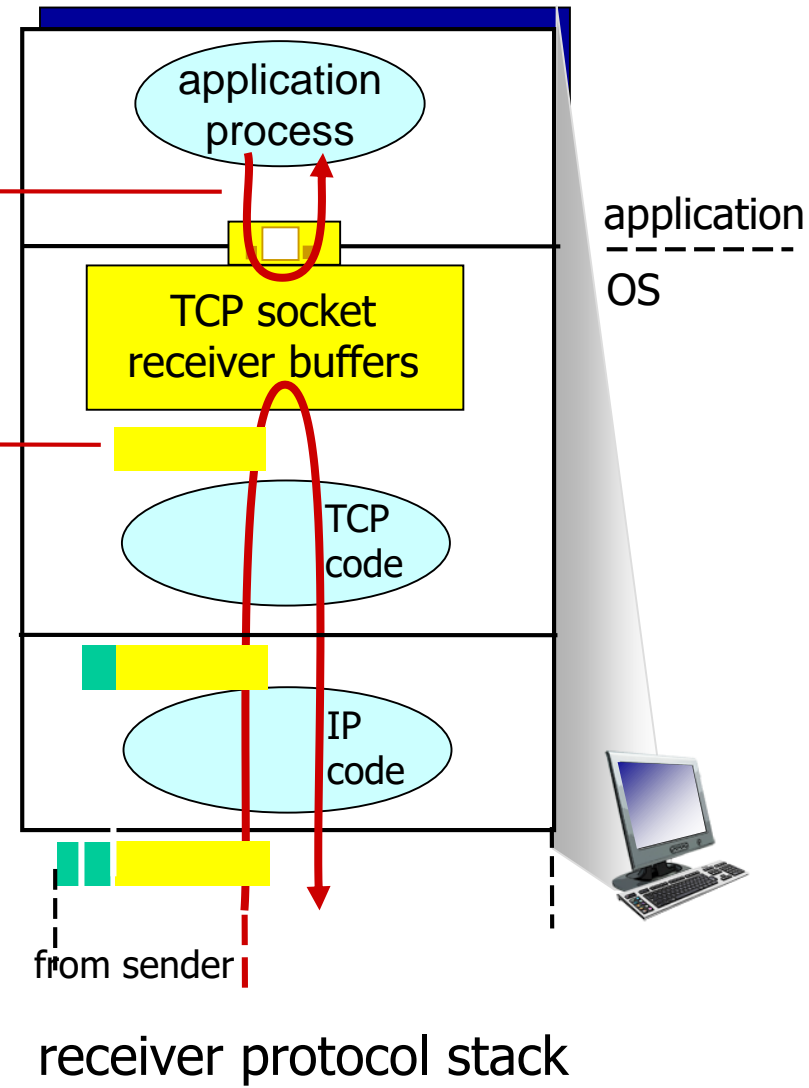
3.7 TCP congestion control

# TCP flow control

application may  
remove data from  
TCP socket buffers ....

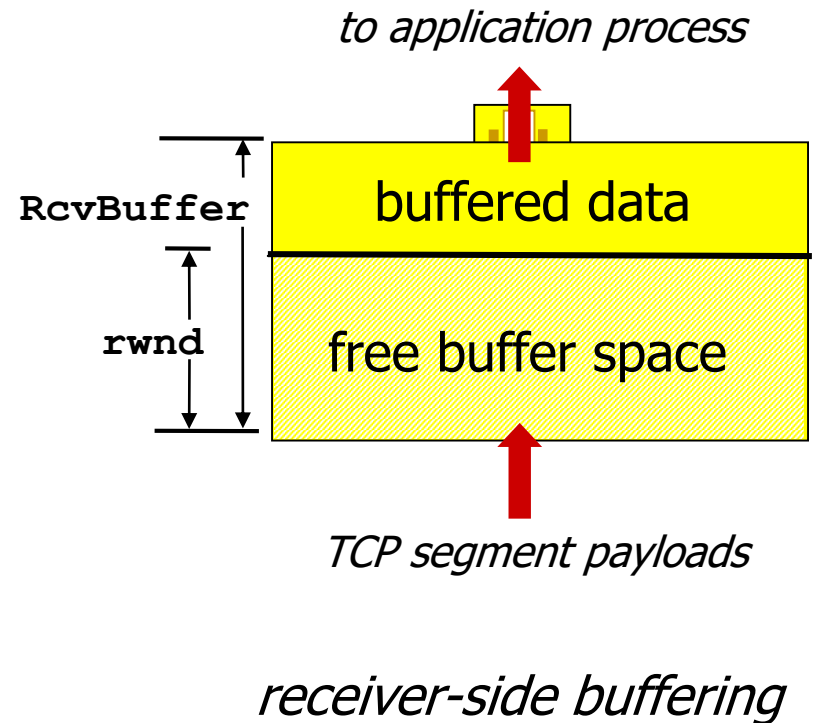
... slower than TCP  
receiver is delivering  
(sender is sending)

***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



# 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

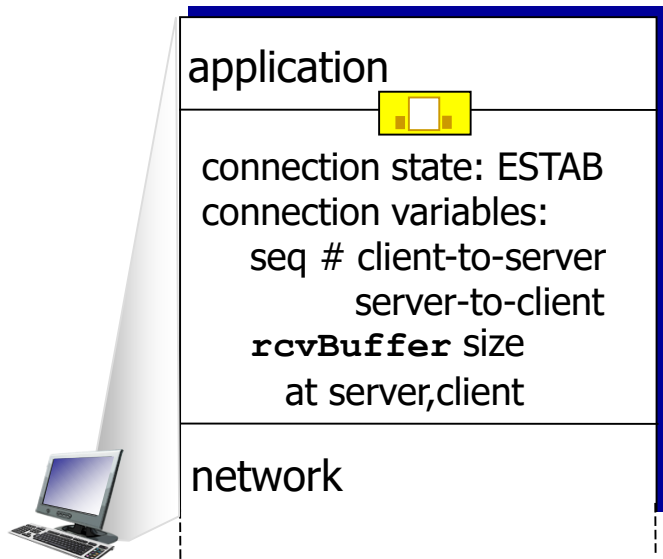
3.6 principles of congestion control

3.7 TCP congestion control

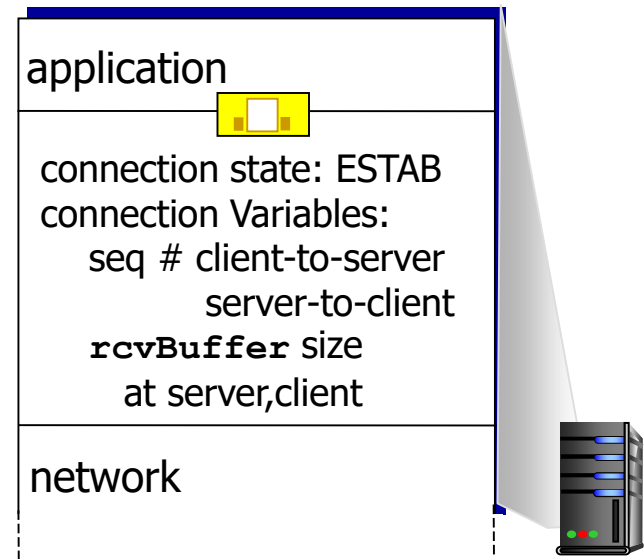
# Connection Management

before exchanging data, sender/receiver “handshake”:

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



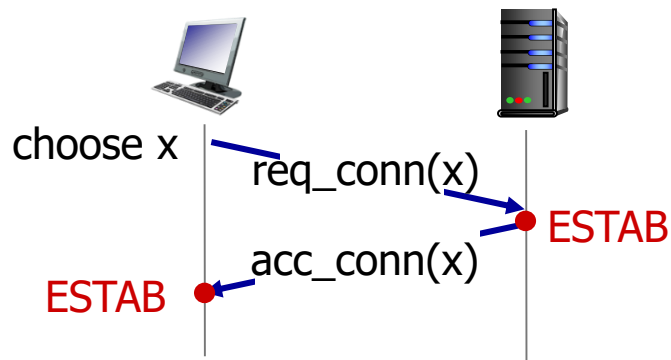
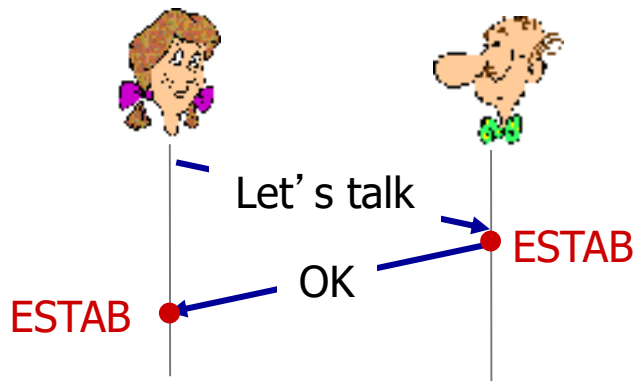
```
Socket clientSocket =  
    newSocket("hostname", "port  
    number");
```



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

# Agreeing to establish a connection

2-way handshake:



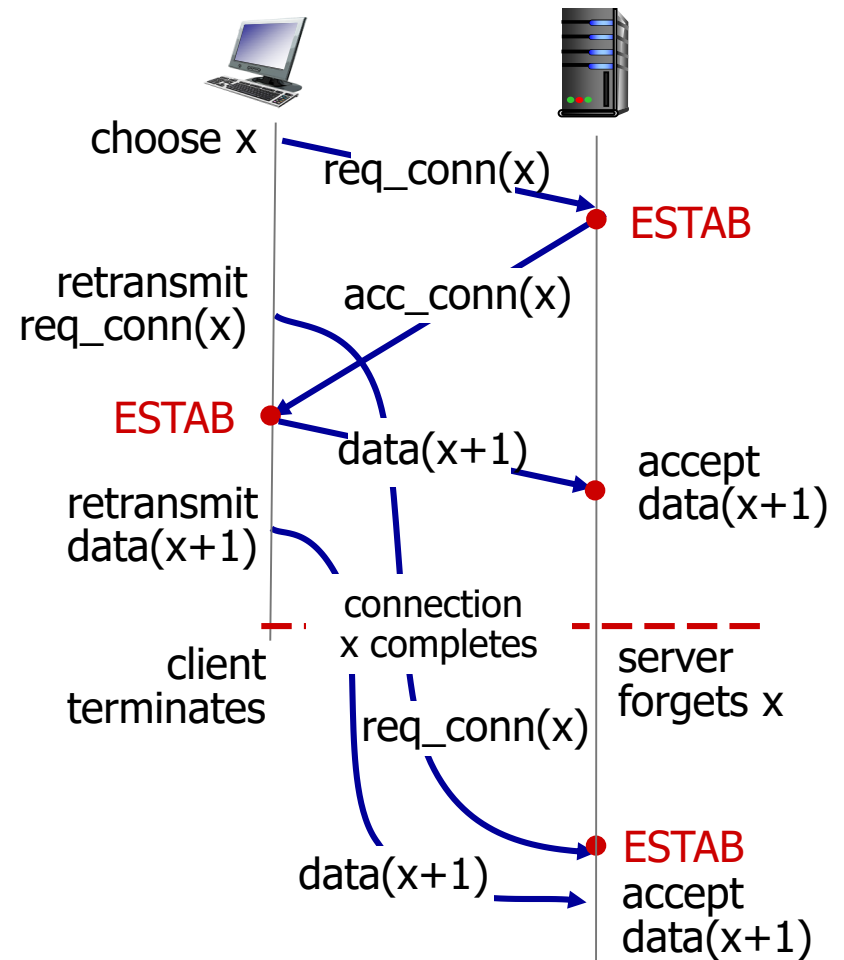
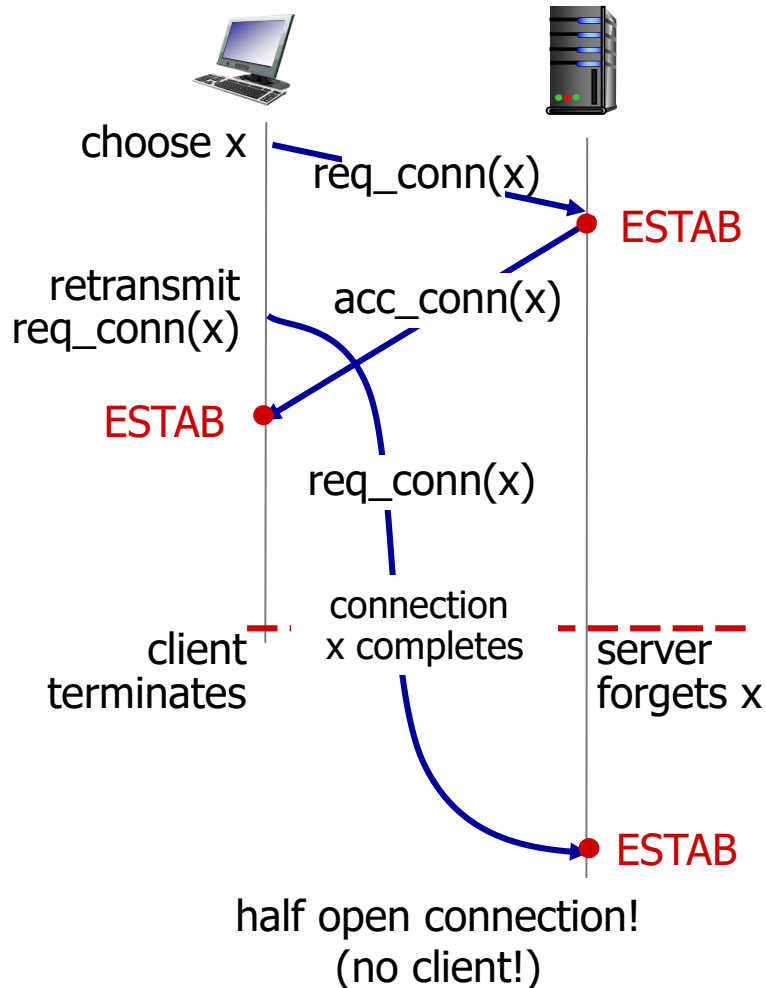
Q: will 2-way handshake always work in network?

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

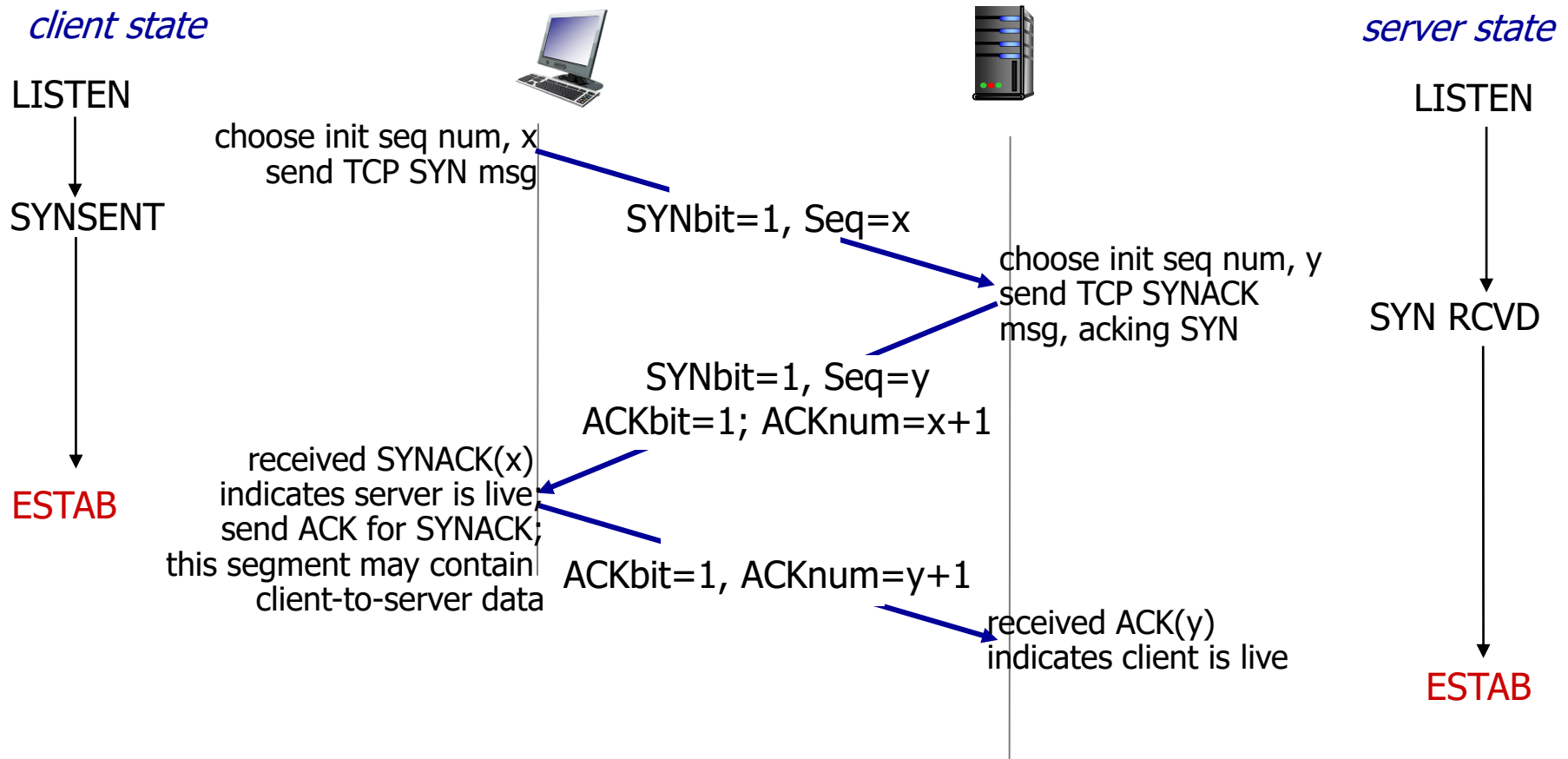


# Agreeing to establish a connection

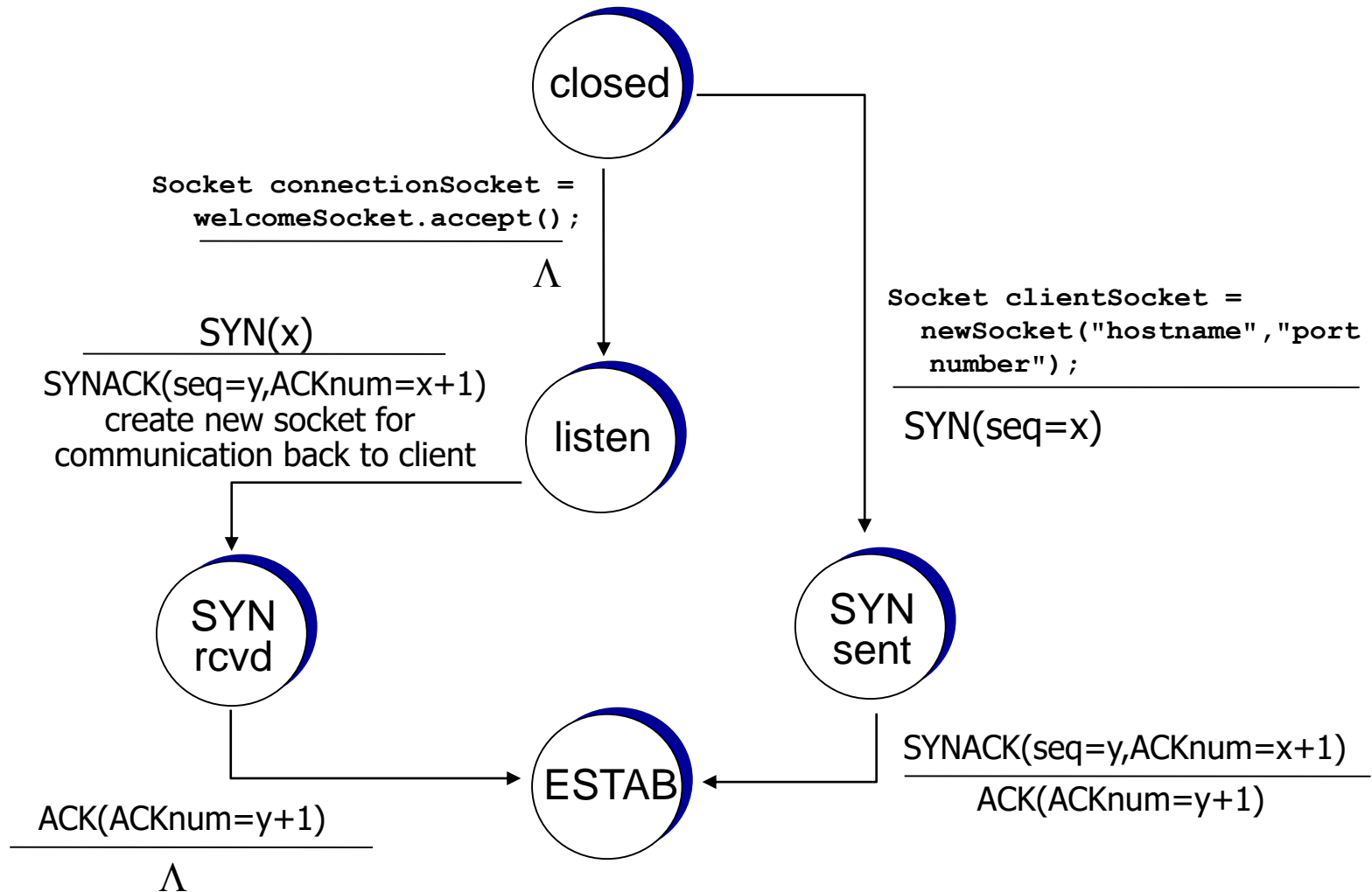
2-way handshake failure scenarios:



# TCP 3-way handshake



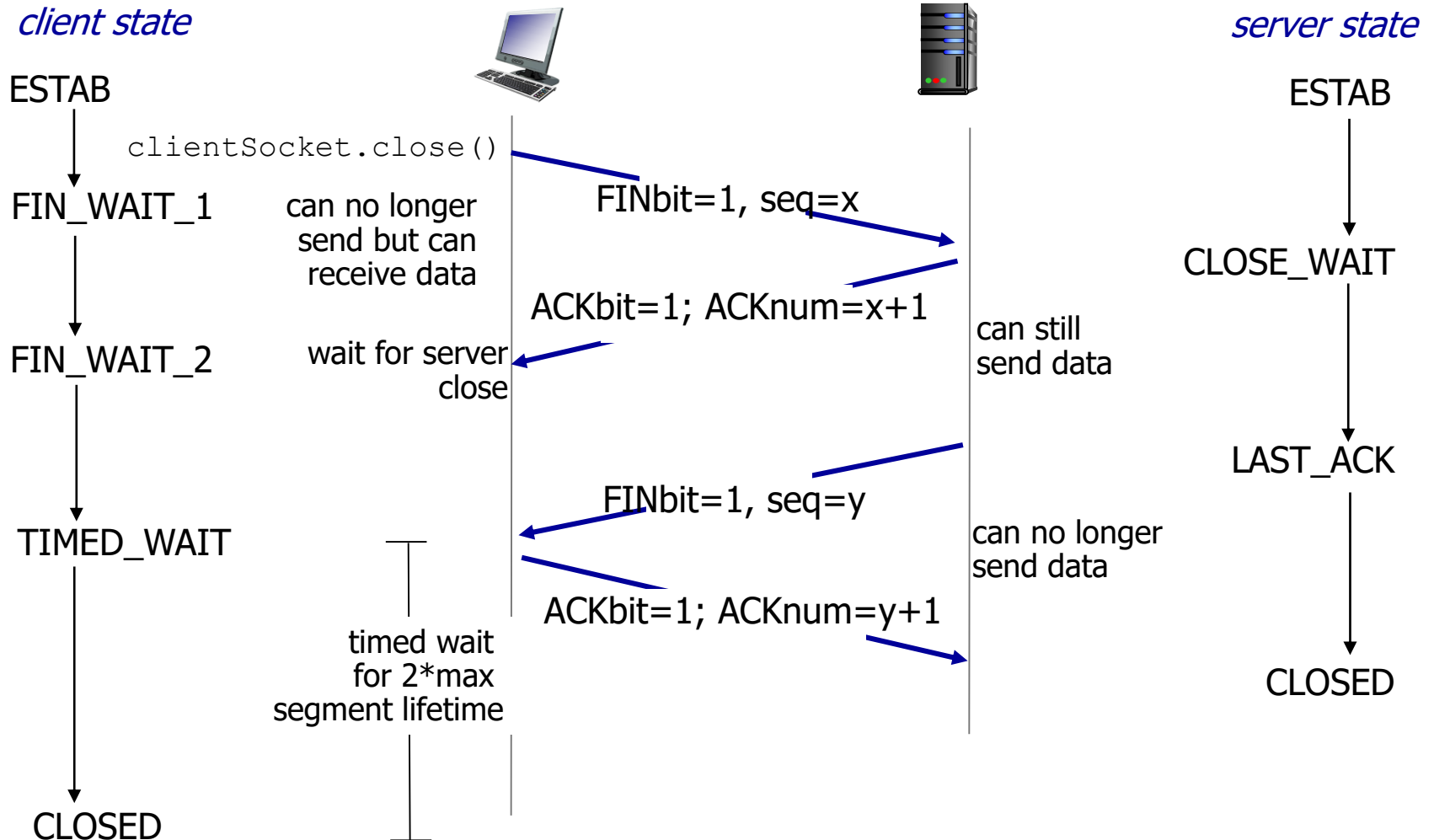
# TCP 3-way handshake: FSM



# 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



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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!

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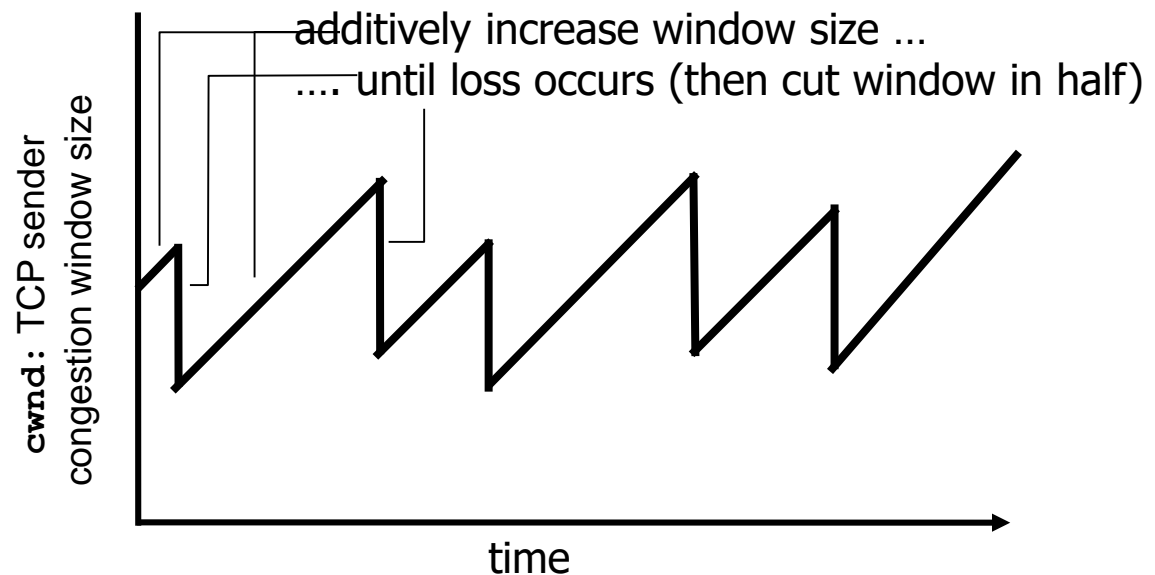
3.7 TCP congestion control



# TCP congestion control: additive increase multiplicative decrease

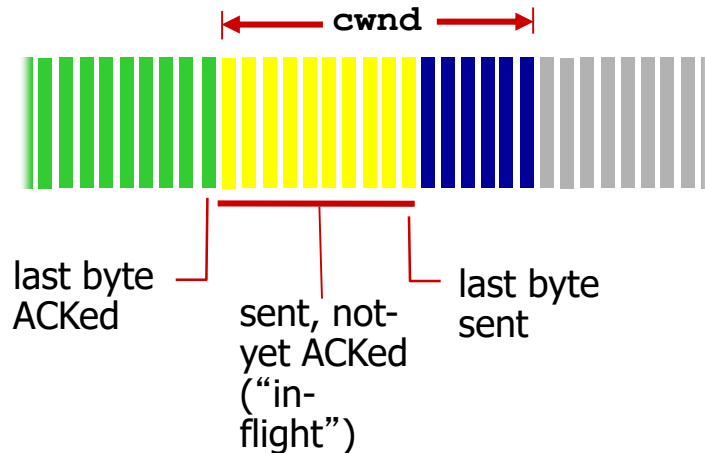
- *approach*: sender increases transmission rate (window size), probing for usable bandwidth, until loss occurs
  - *additive increase*: increase **cwnd** by 1 MSS every RTT until loss detected
  - *multiplicative decrease*: cut **cwnd** in half after loss

AIMD saw tooth  
behavior: probing  
for bandwidth



# TCP Congestion Control: details

sender sequence number space



- sender limits transmission:

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

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

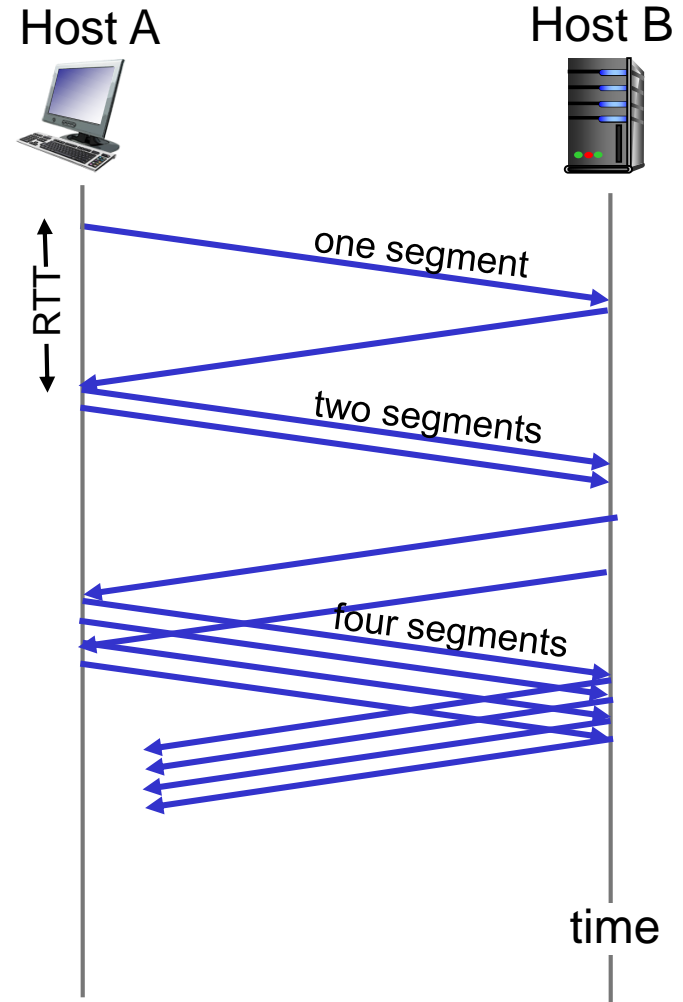
*TCP sending rate:*

- *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:
  - `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)

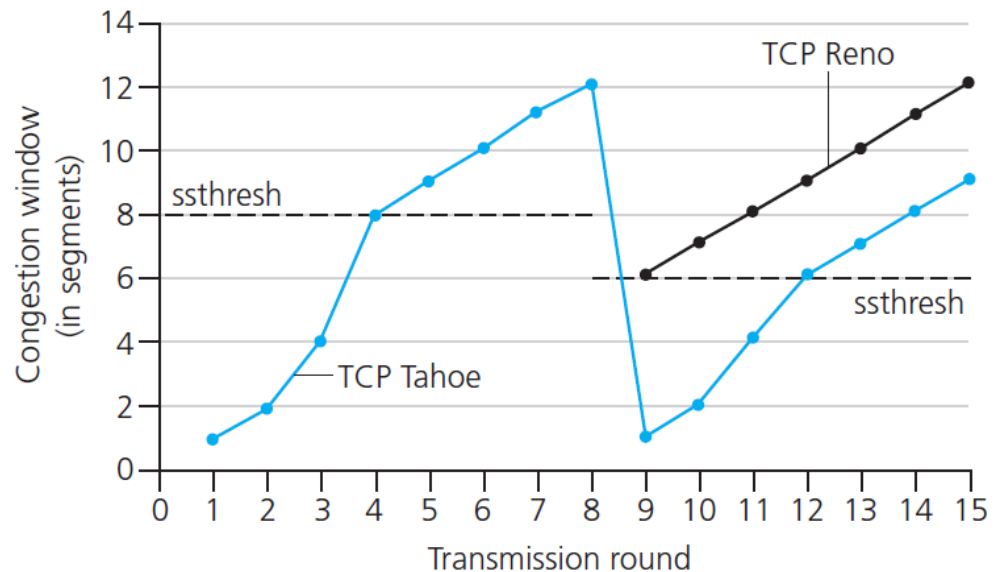
# TCP: switching from slow start to CA

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

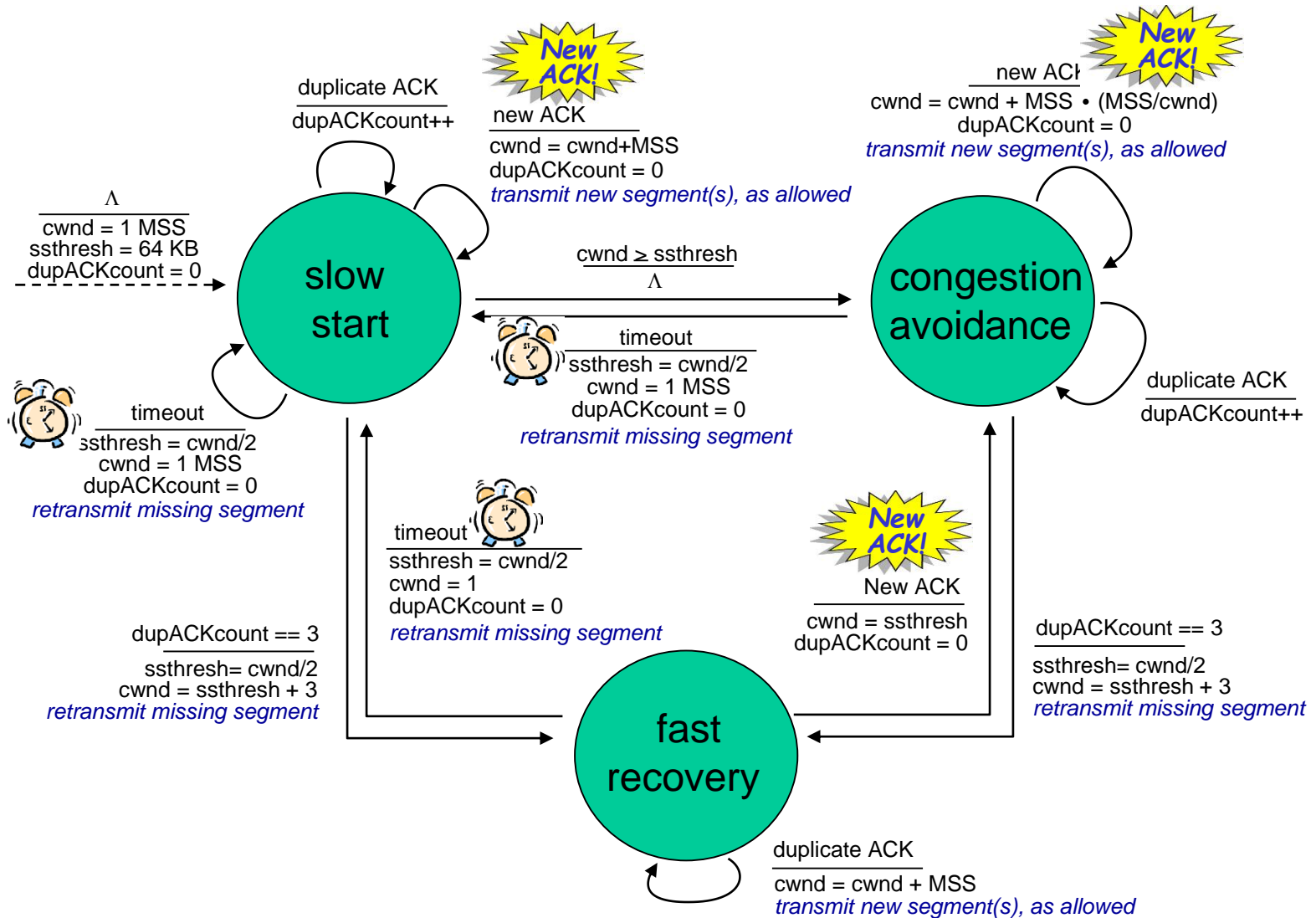
**A:** when **cwnd** gets to 1/2 of its value before timeout.

## Implementation:

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



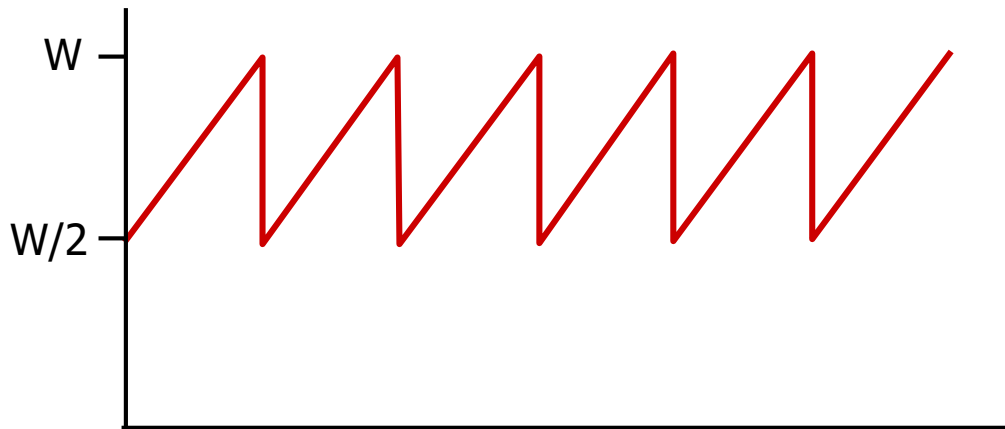
# Summary: TCP Congestion Control



# 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  $\frac{3}{4} W$
  - avg. thruput is  $\frac{3}{4} W$  per RTT

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



# 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]:

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

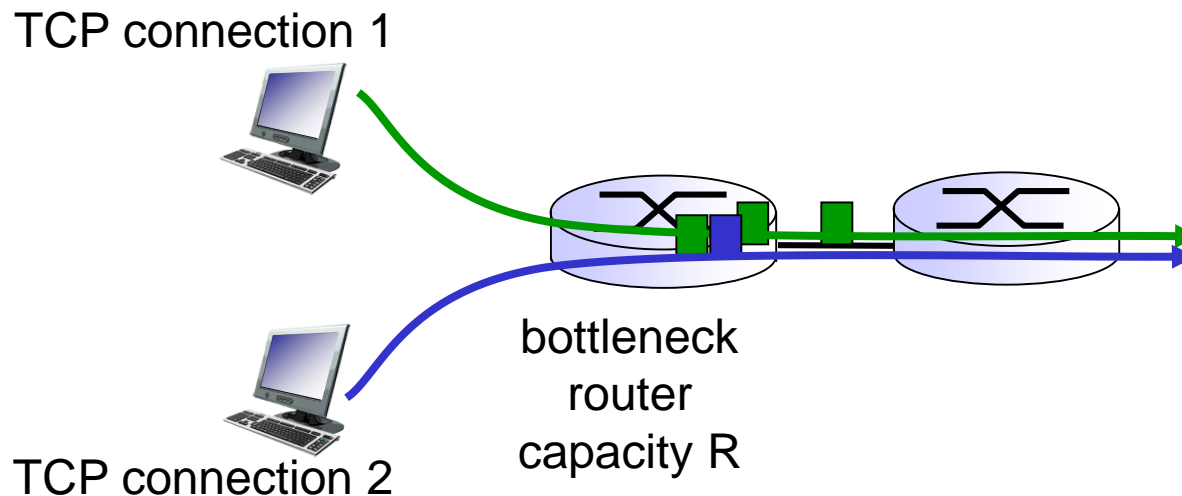
→ to achieve 10 Gbps throughput, need a loss rate of  $L = 2 \cdot 10^{-10}$  — *a very small loss rate!*

- 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  $R/K$



# 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