



CSCI 6760 - Computer Networks - Fall 2024

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Transport layer: 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

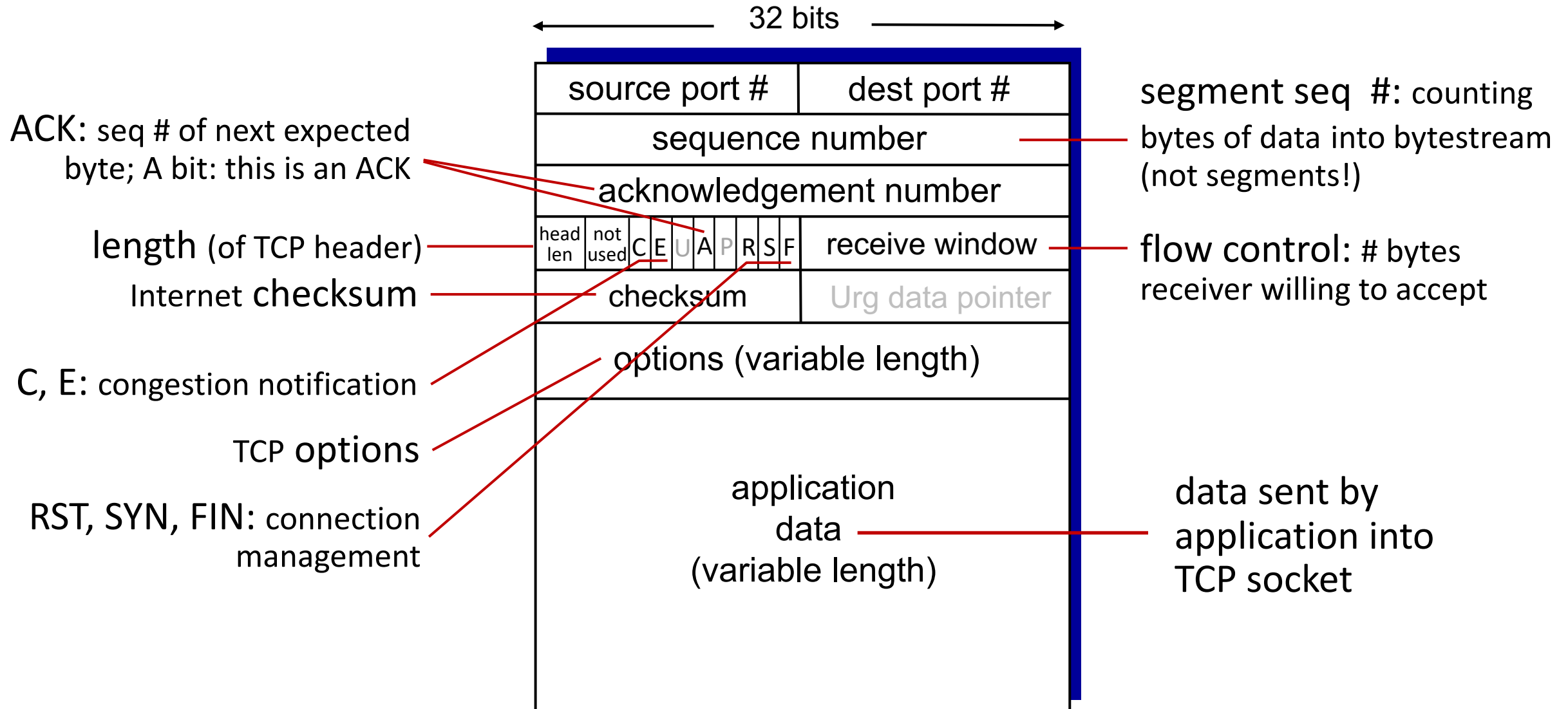


TCP: overview

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:

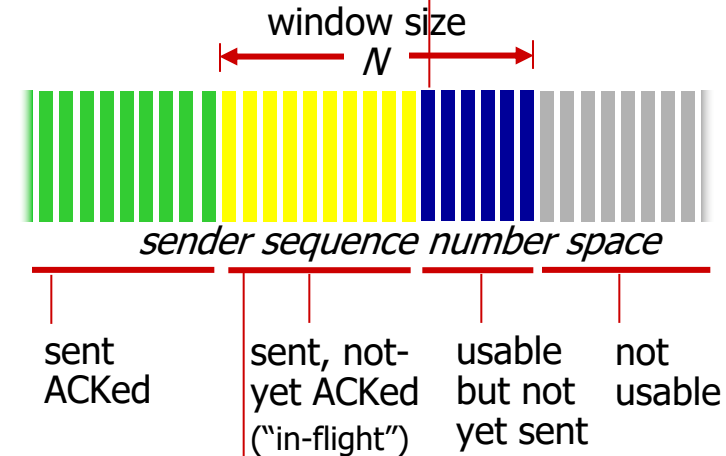
- seq # of next byte expected from other side
- cumulative ACK

Q: how receiver handles out-of-order segments

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

outgoing segment from sender

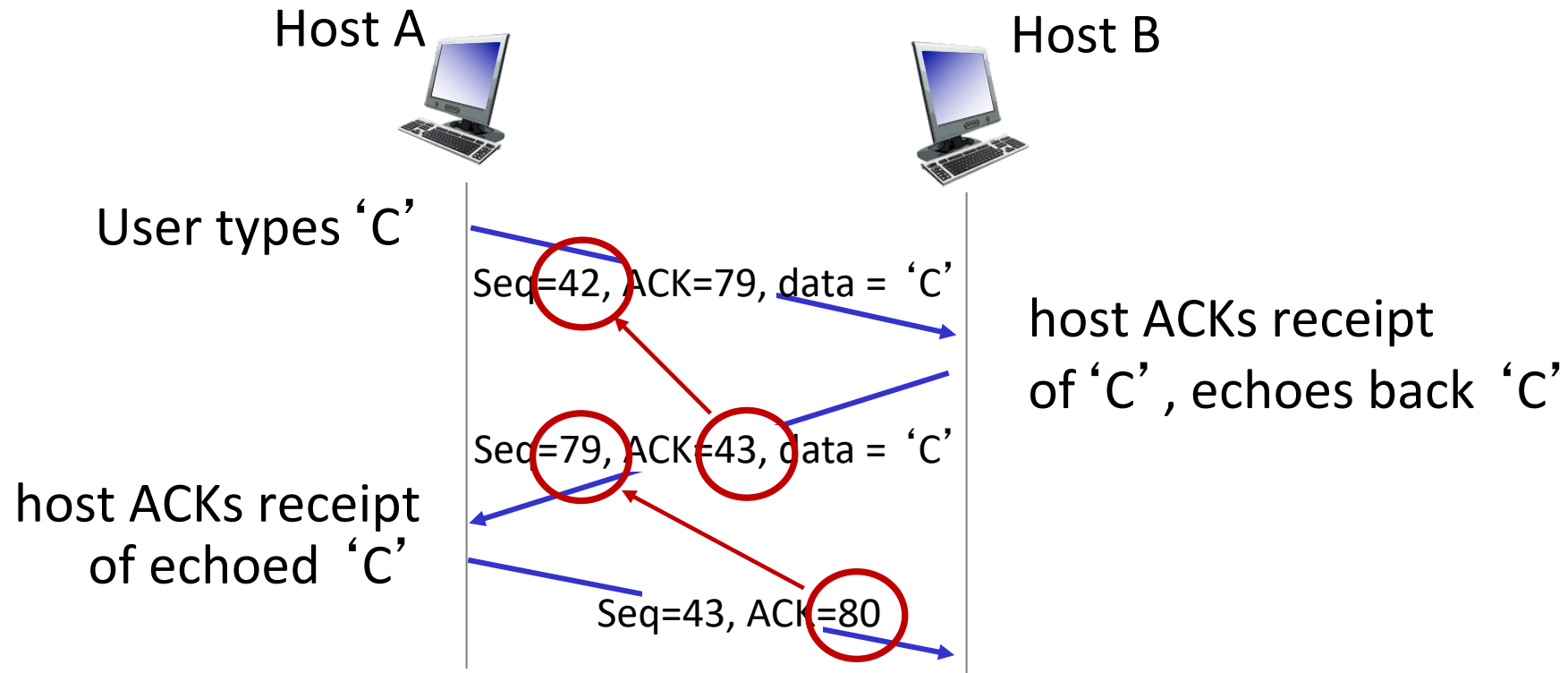
source port #	dest port #
sequence number	
acknowledgement number	
	rwnd
checksum	urg pointer



outgoing segment from receiver

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

TCP sequence numbers, ACKs



simple telnet scenario

TCP 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**

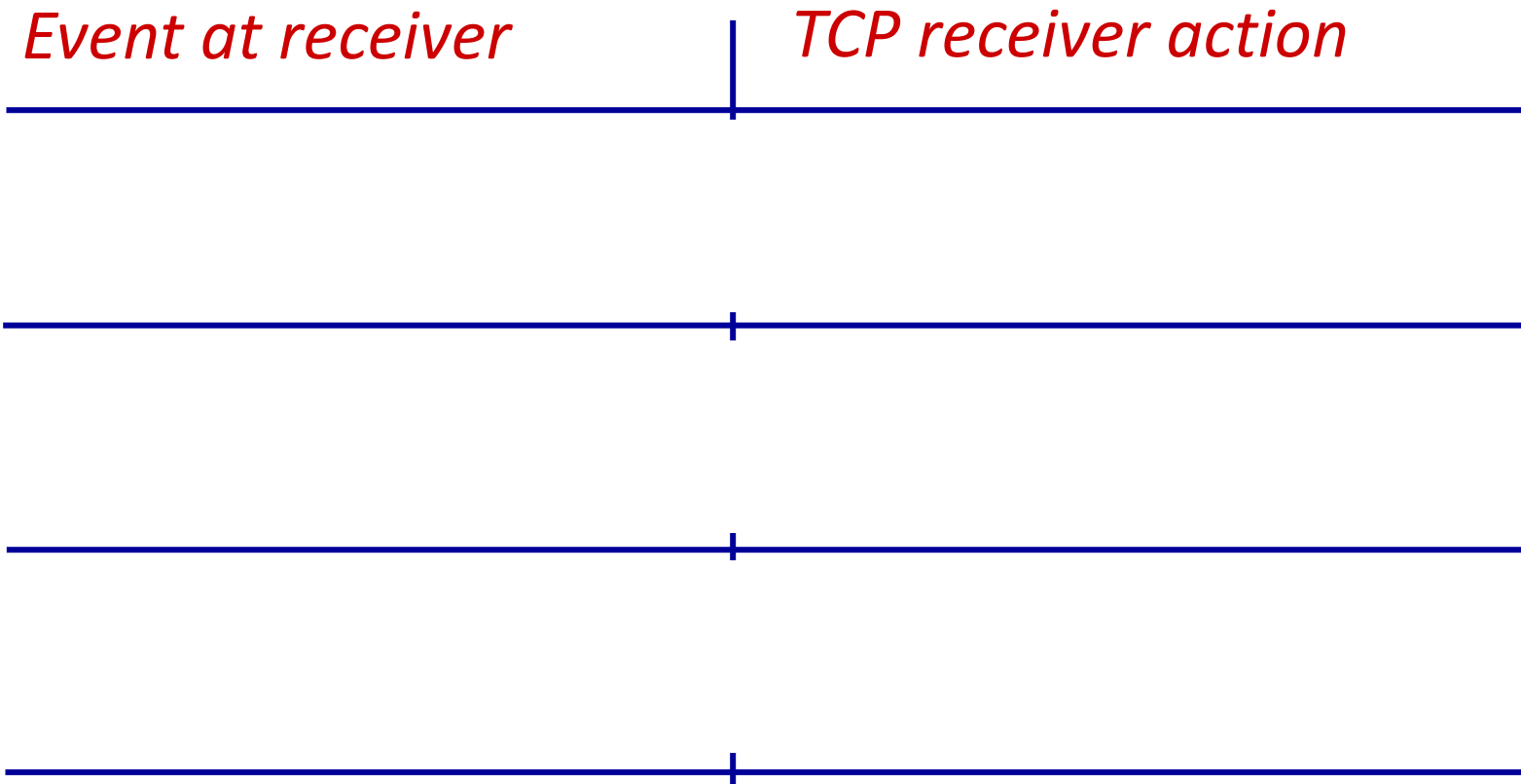
event: timeout

- retransmit segment that caused timeout
- restart timer

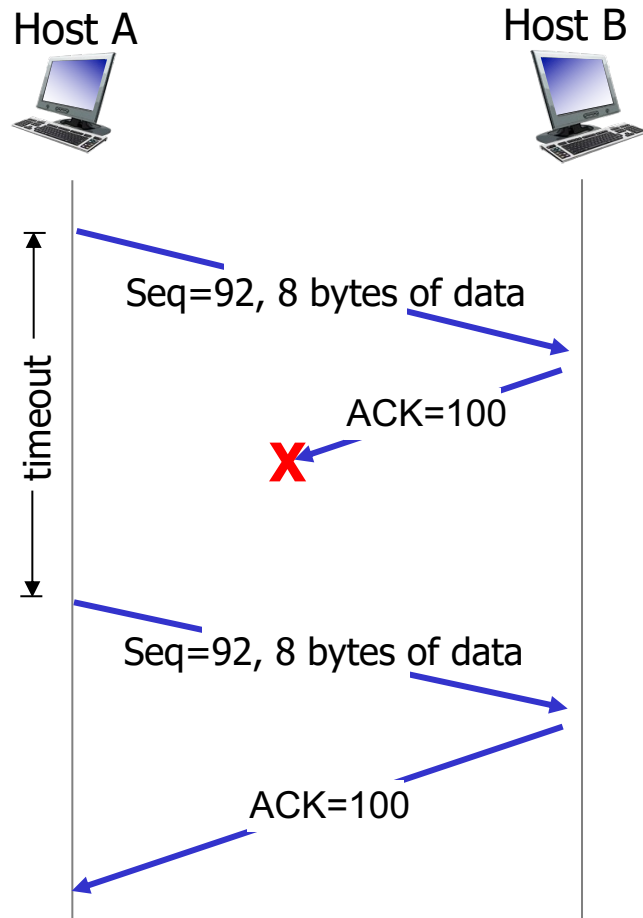
event: ACK received

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

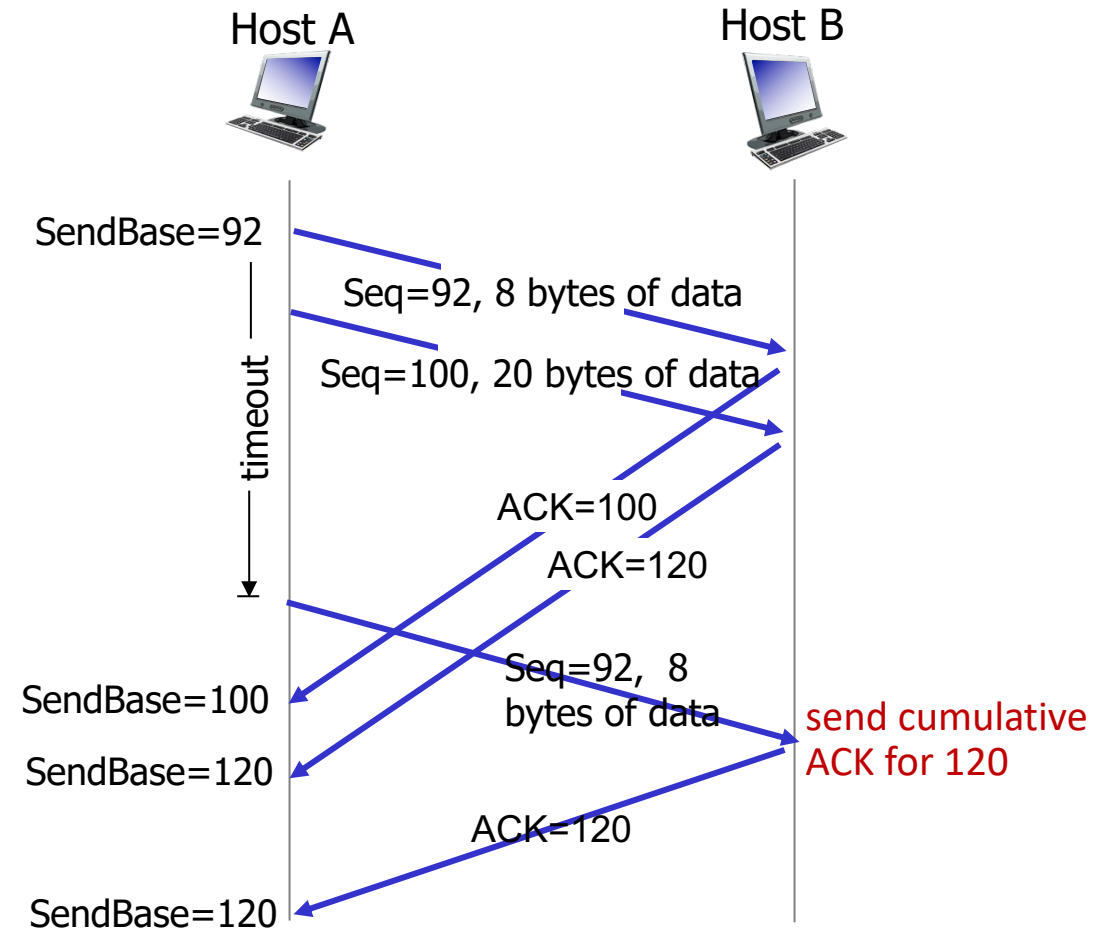
TCP Receiver: ACK generation [RFC 5681]



TCP: retransmission scenarios

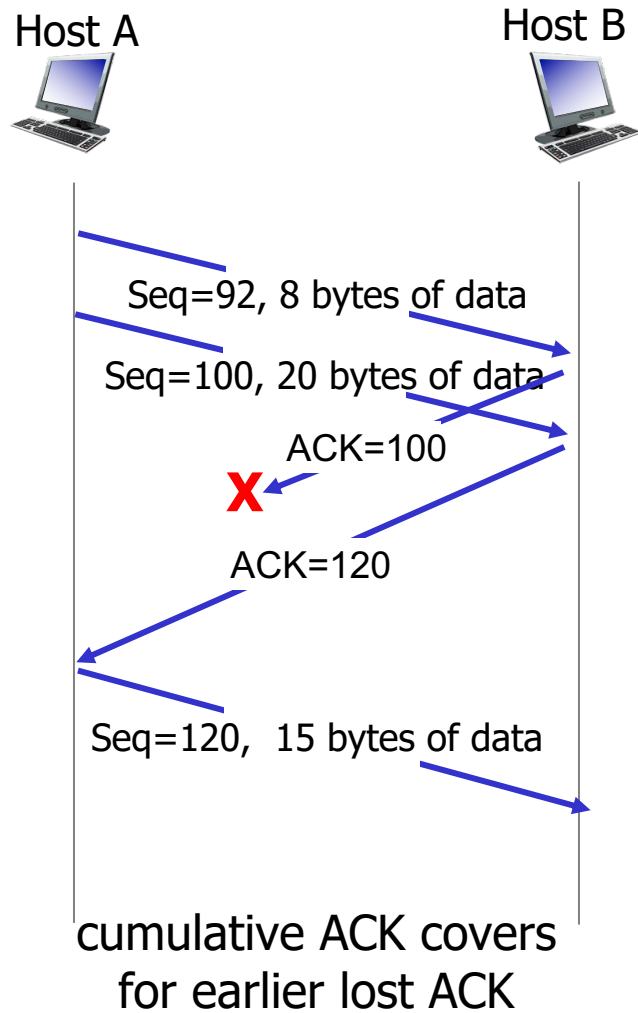


lost ACK scenario

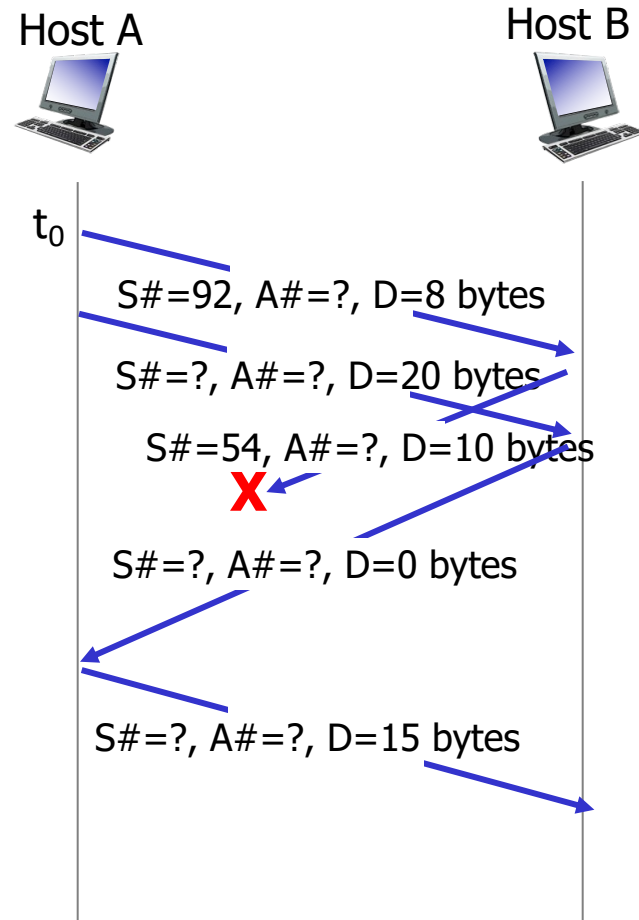


premature timeout

TCP: retransmission scenarios

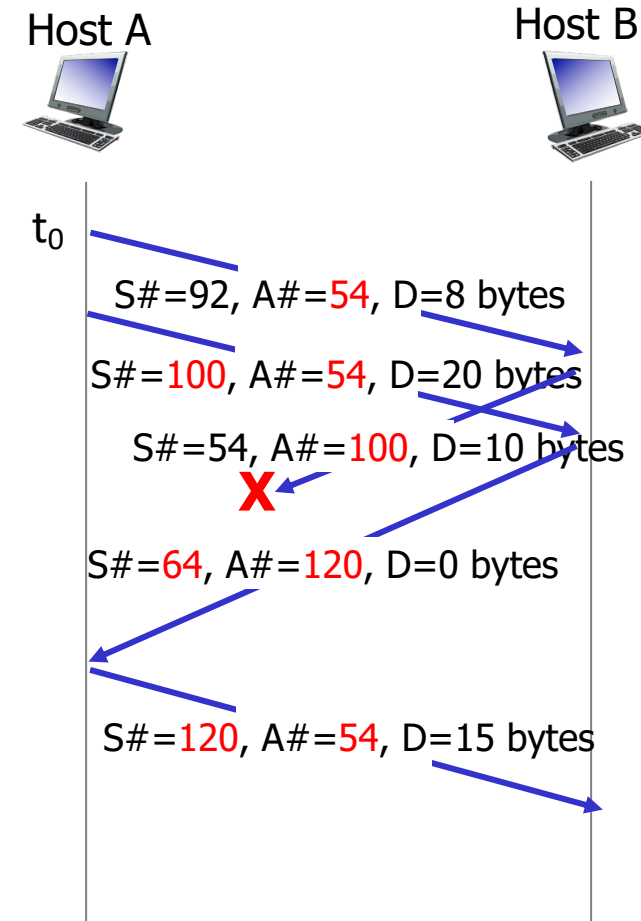
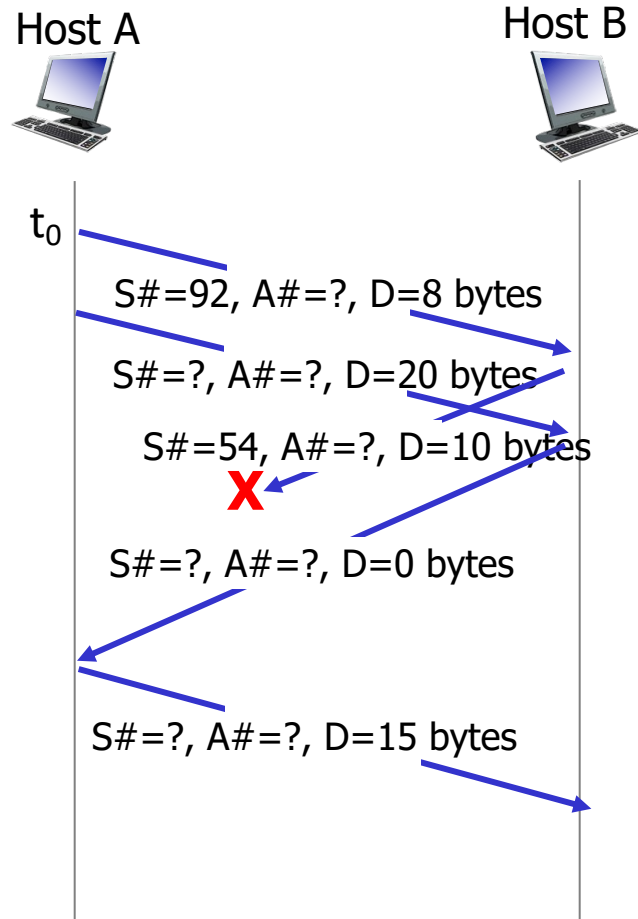


Exercise



Assume that before t_0 all packets have been received and ack'ed correctly on both sides

Exercise



Assume that before t_0 all packets have been received and ack'ed correctly on both sides

TCP fast retransmit

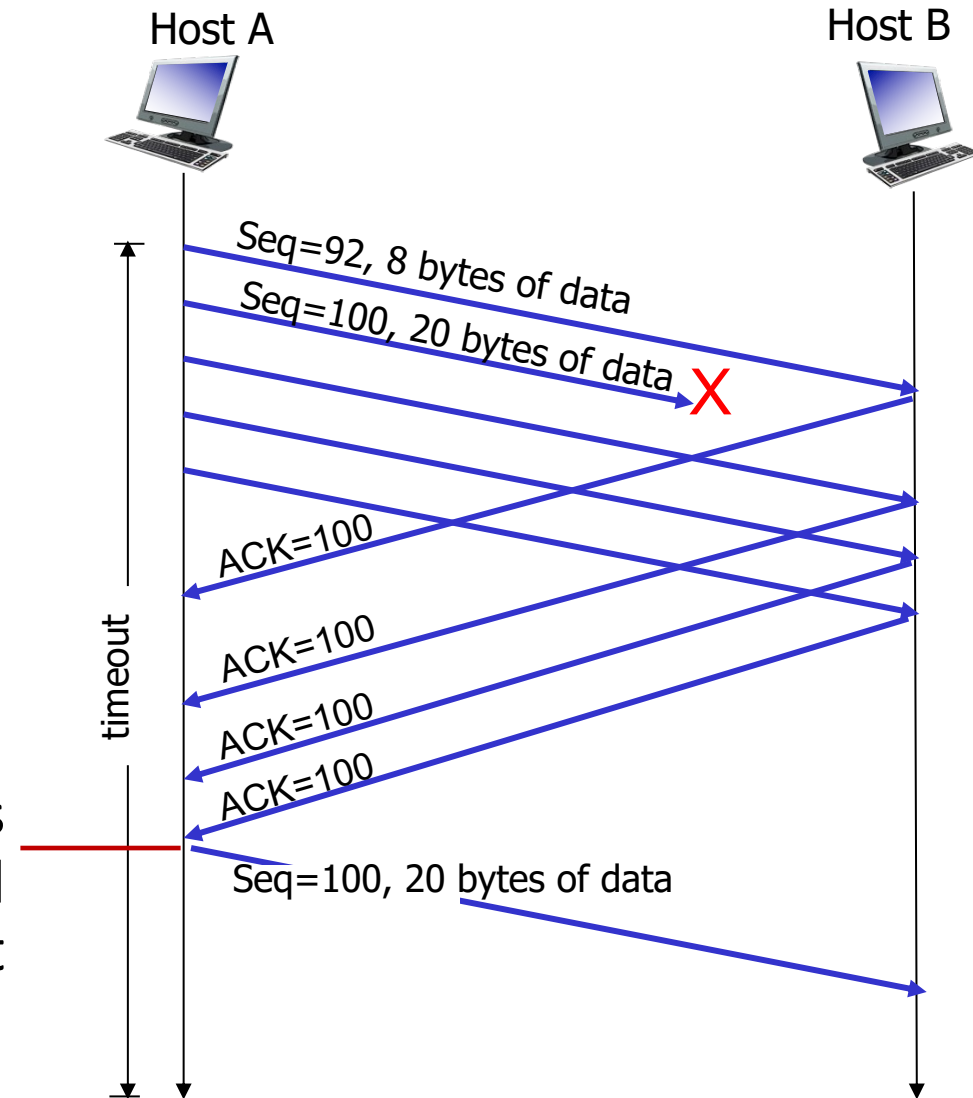
TCP fast retransmit

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!



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

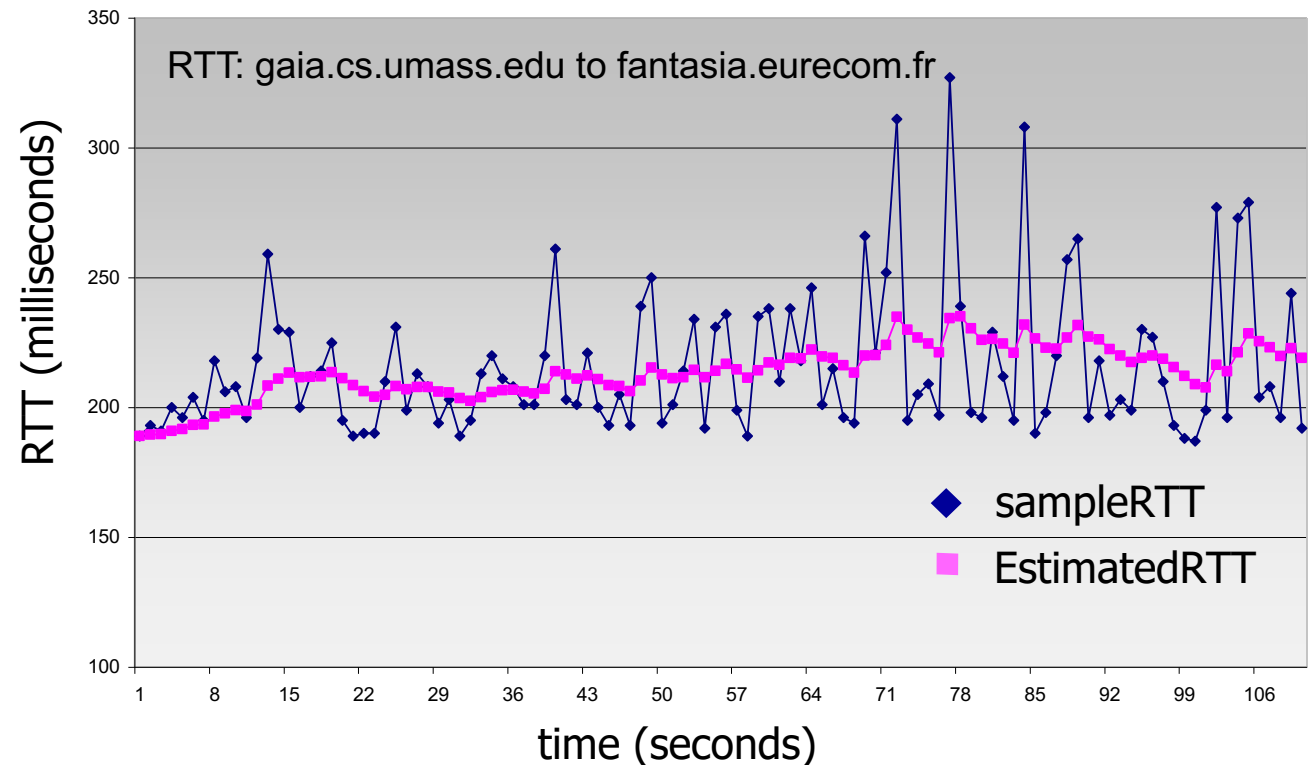
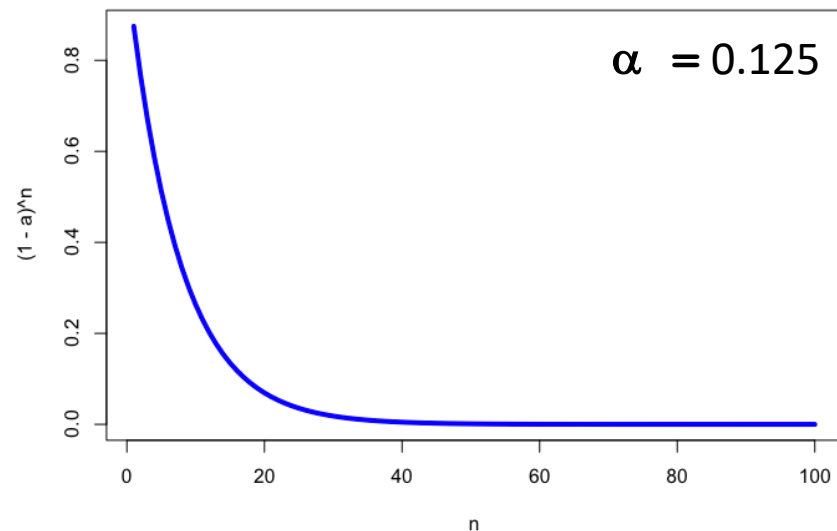
Q: how to estimate RTT?

- *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, $\beta = 0.25$)

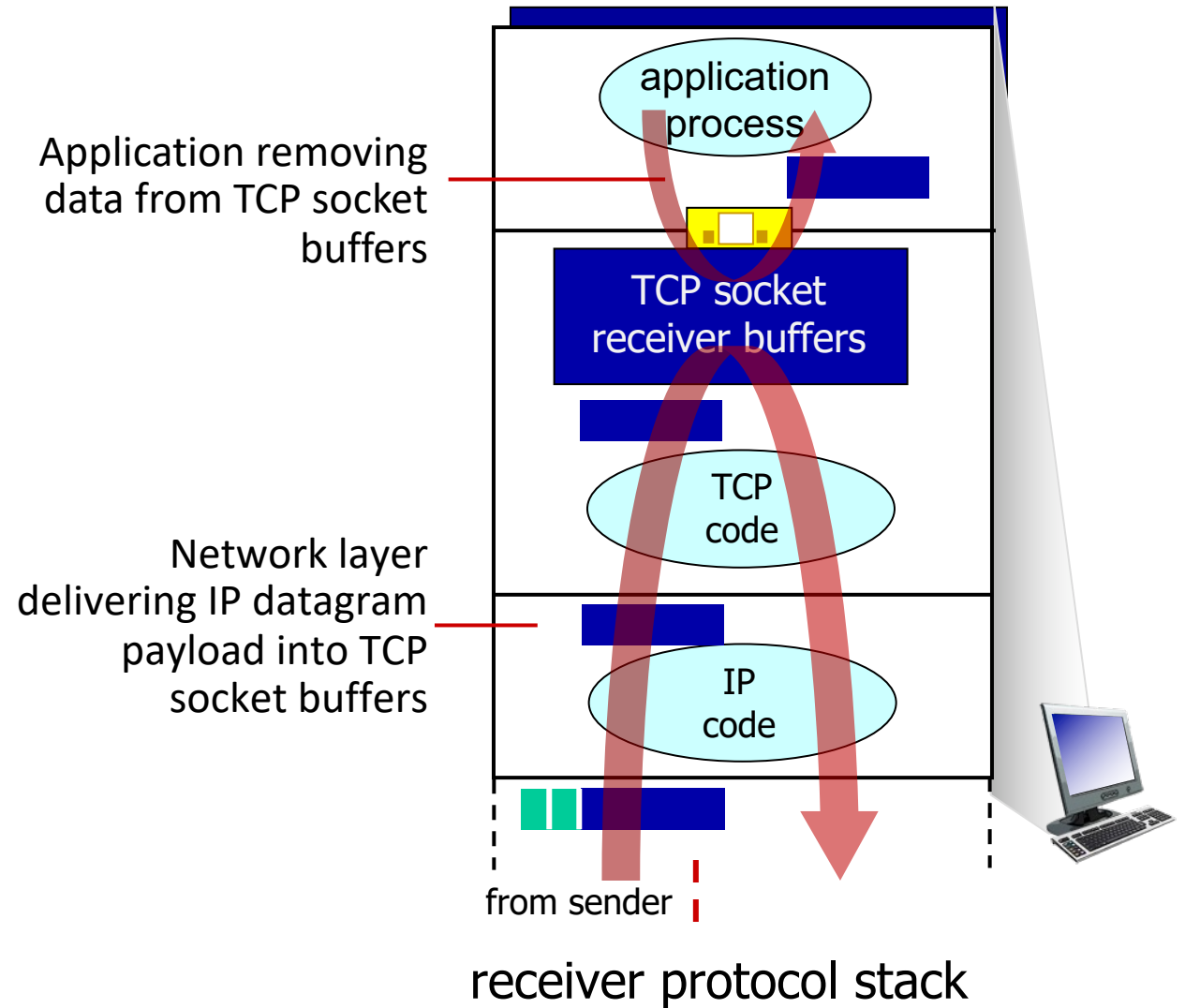
Chapter 3: roadmap

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



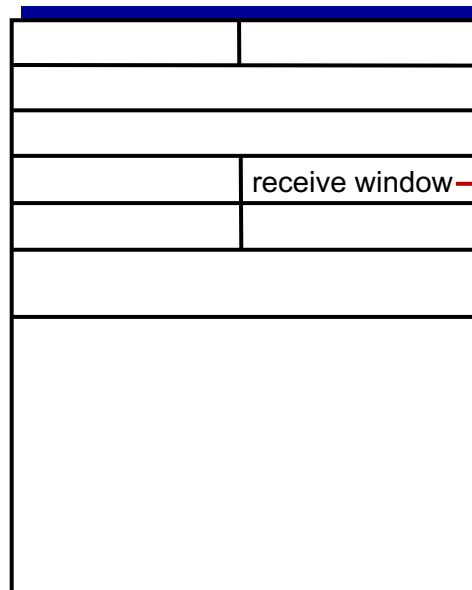
TCP flow control

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



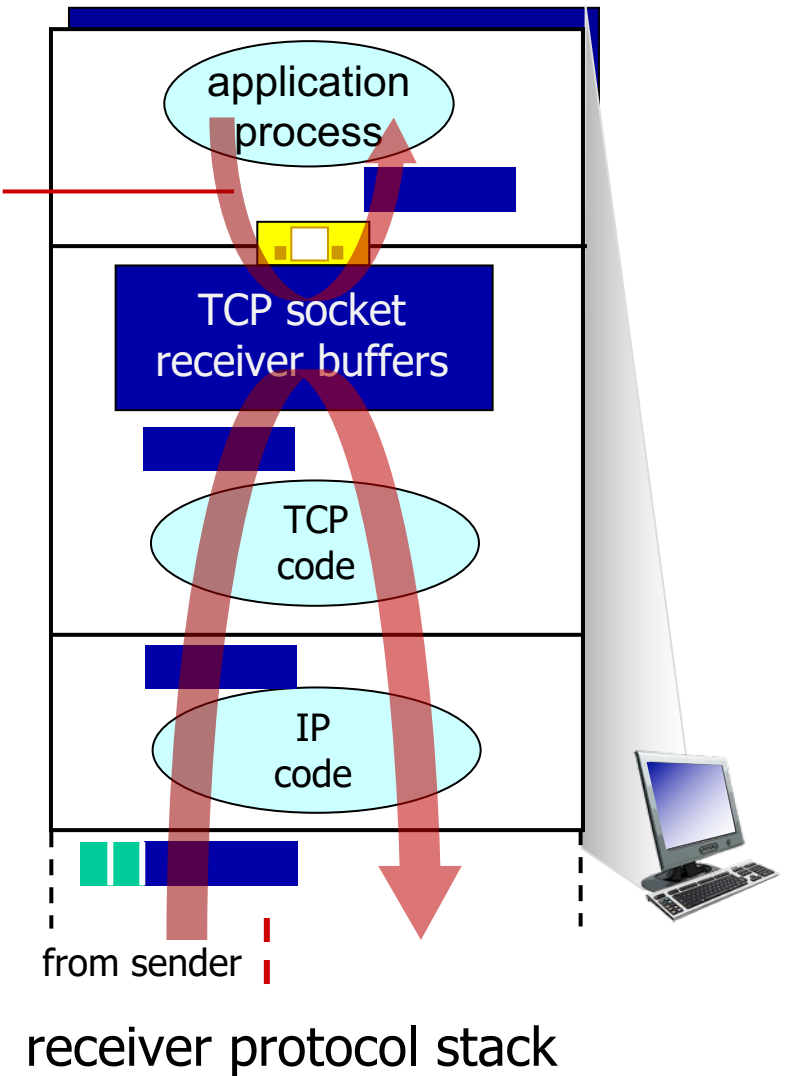
TCP flow control

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

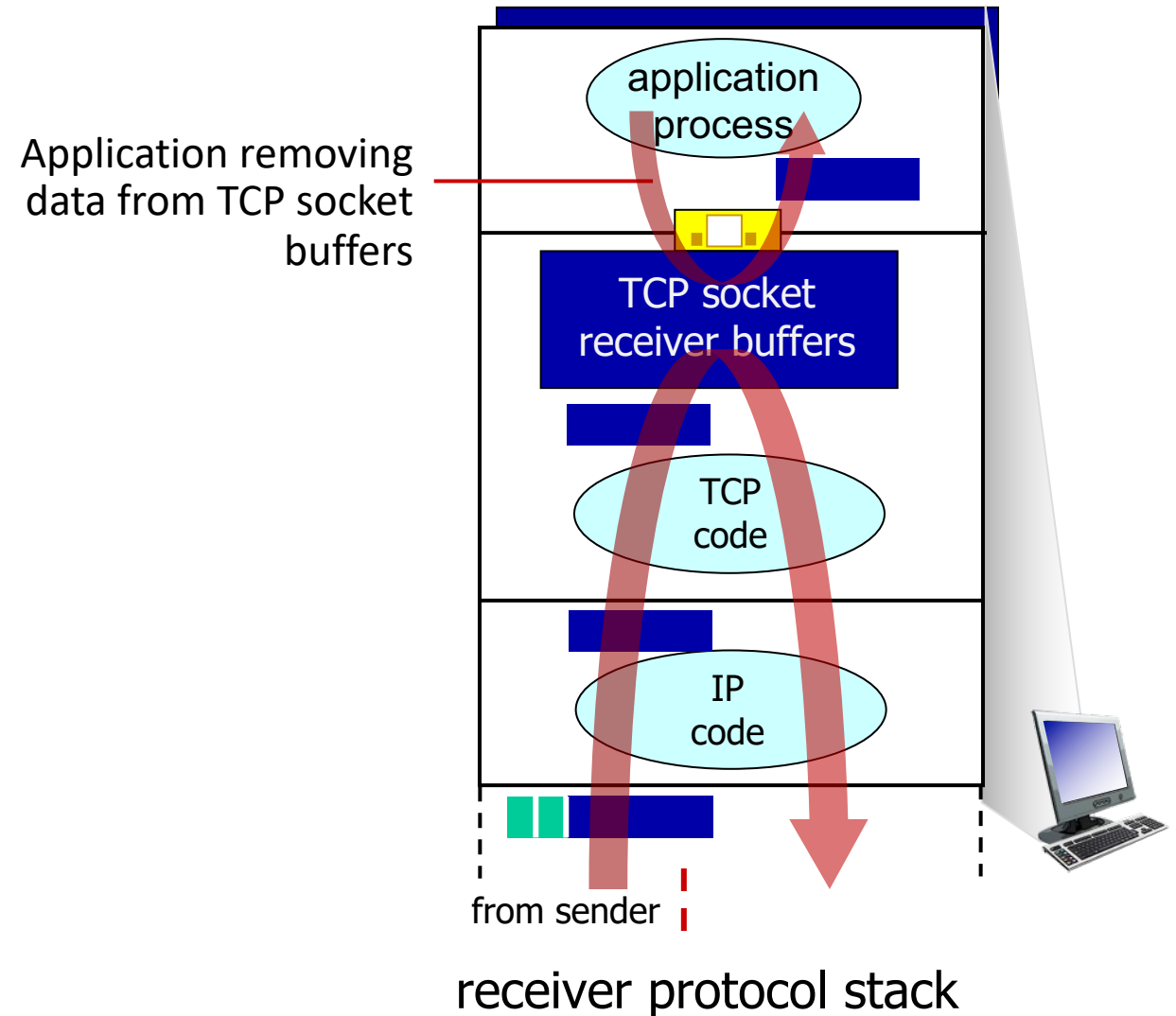


TCP flow control

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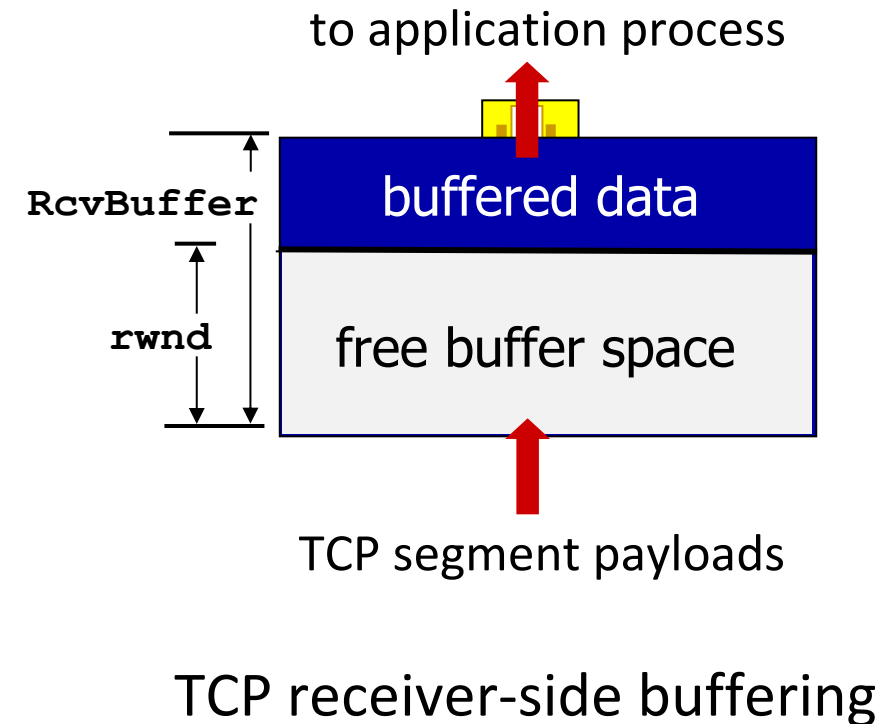
—flow control—

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



TCP flow control

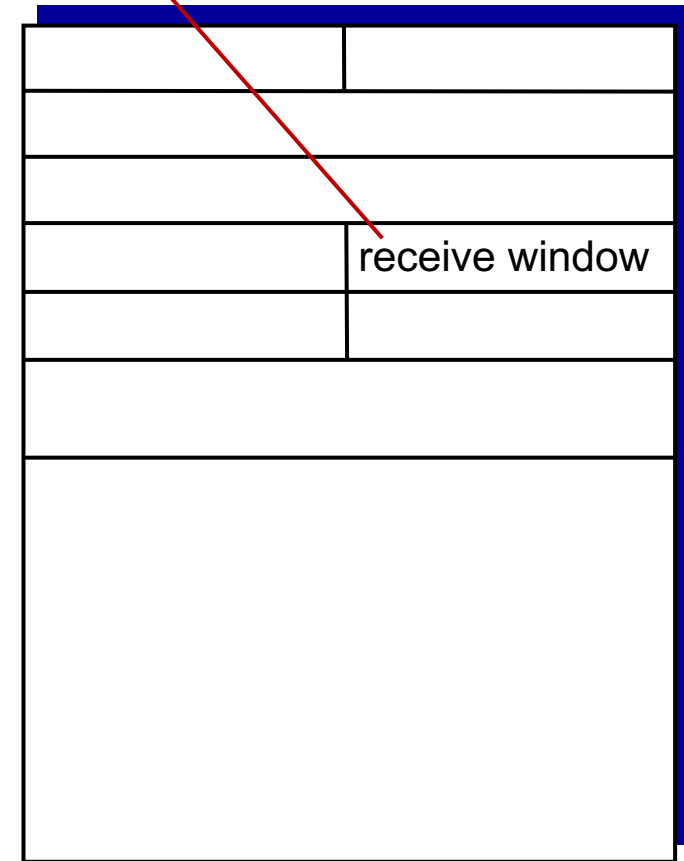
- TCP receiver “advertises” free buffer space in **rwnd** field in TCP header
 - **RcvBuffer** size set via socket options (typical default is 4096 bytes)
 - many operating systems autoadjust **RcvBuffer**
- sender limits amount of unACKed (“in-flight”) data to received **rwnd**
- guarantees receive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



TCP flow control

- TCP receiver “advertises” free buffer space in **rwnd** field in TCP header
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flow control: # bytes receiver willing to accept

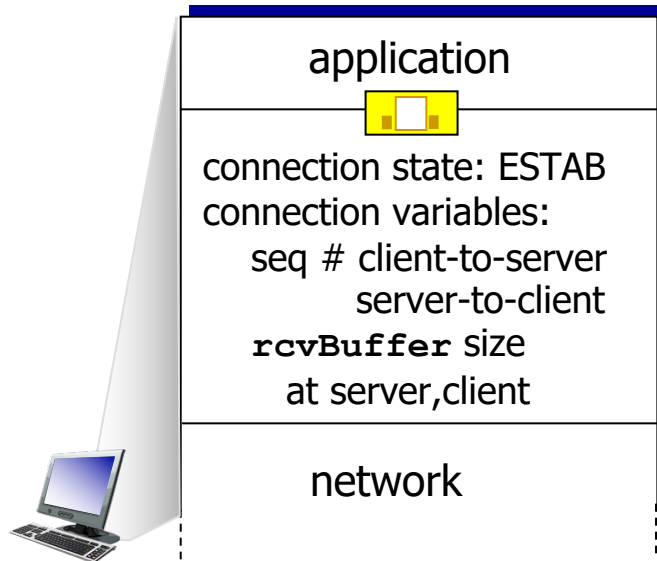


TCP segment format

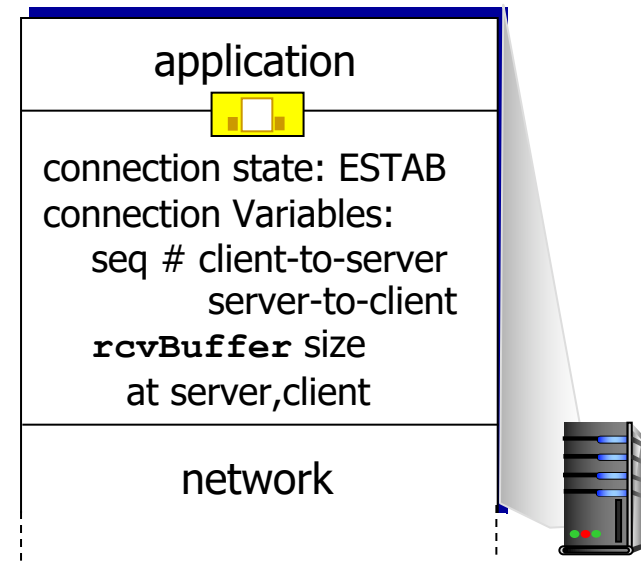
TCP connection management

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();
```

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()
```

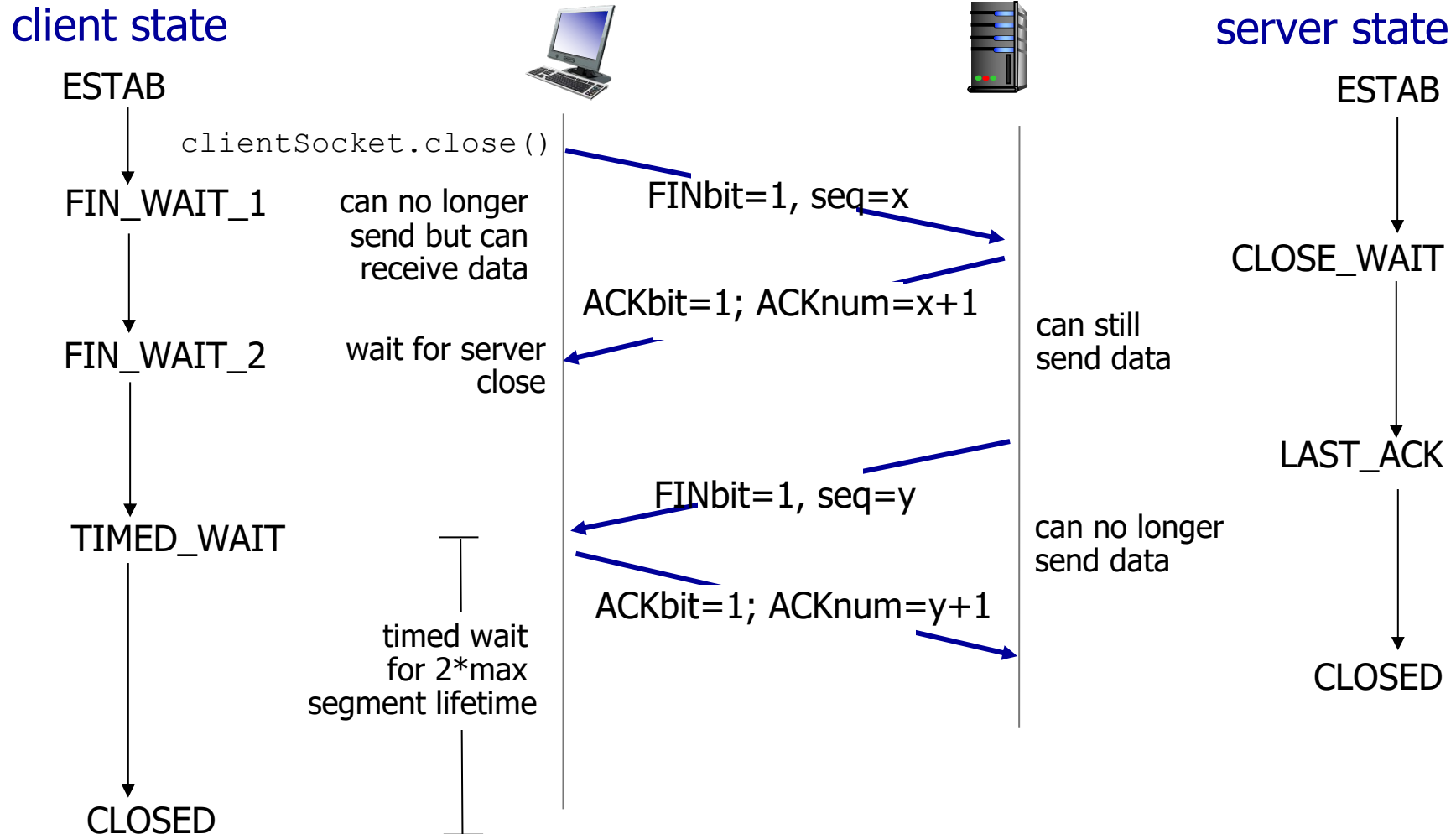
LISTEN

SYN RCVD

ESTAB

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

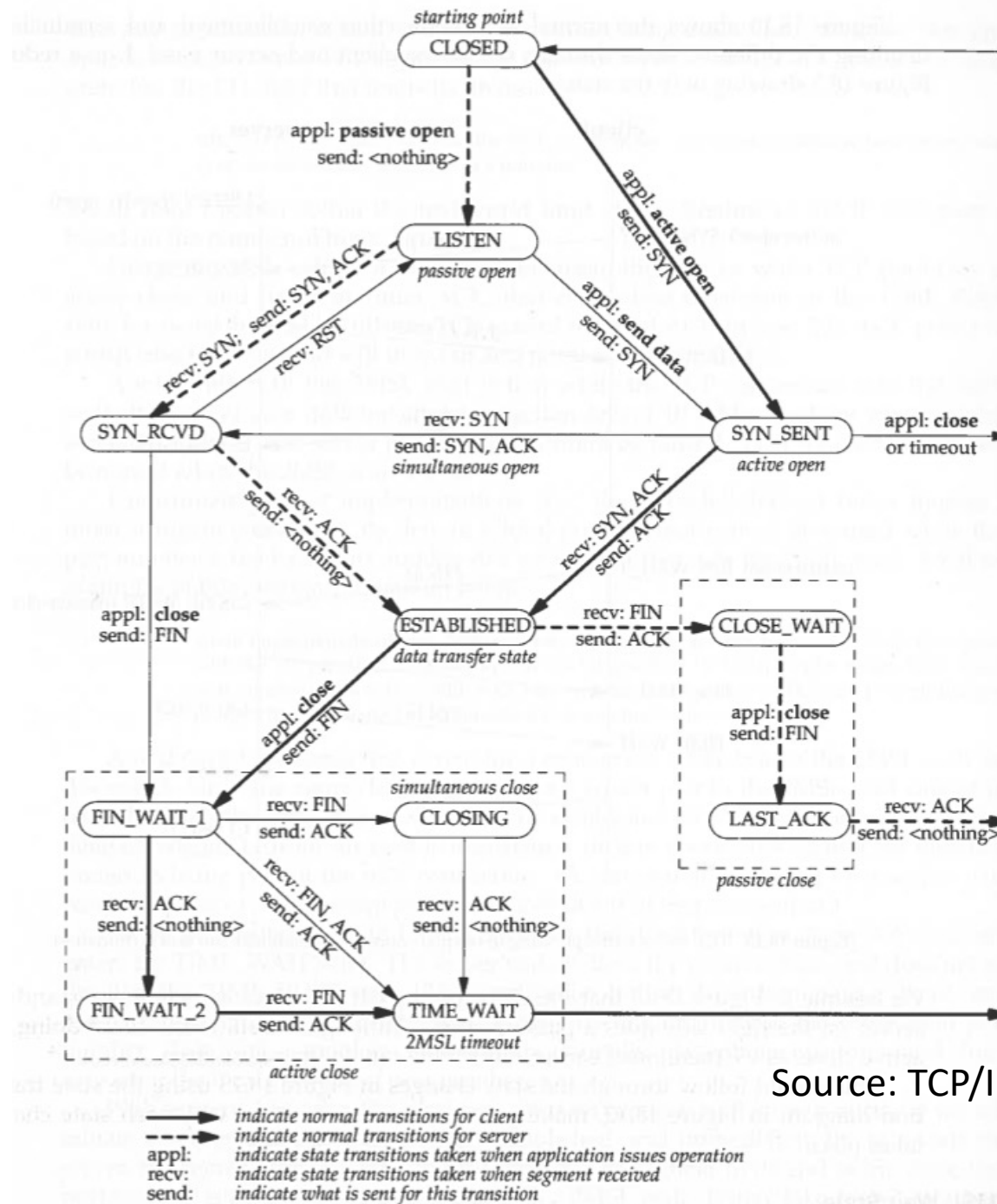
Closing a TCP connection



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

Connection Management



Source: TCP/IP Illustrated, Volume 1

Transport layer: roadmap

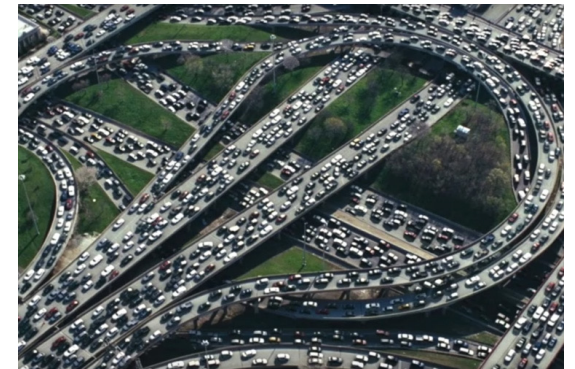
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- TCP congestion control
- Evolution of transport-layer functionality



Principles of congestion control

Congestion:

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



congestion control:

too many senders,
sending too fast

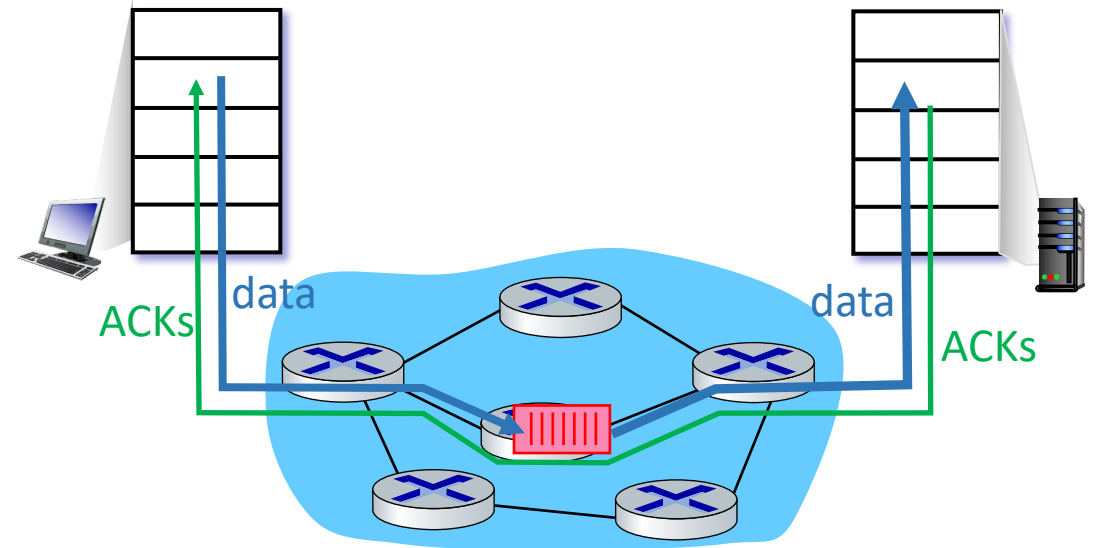


flow control: one sender
too fast for one receiver

Approaches towards congestion control

End-end congestion control:

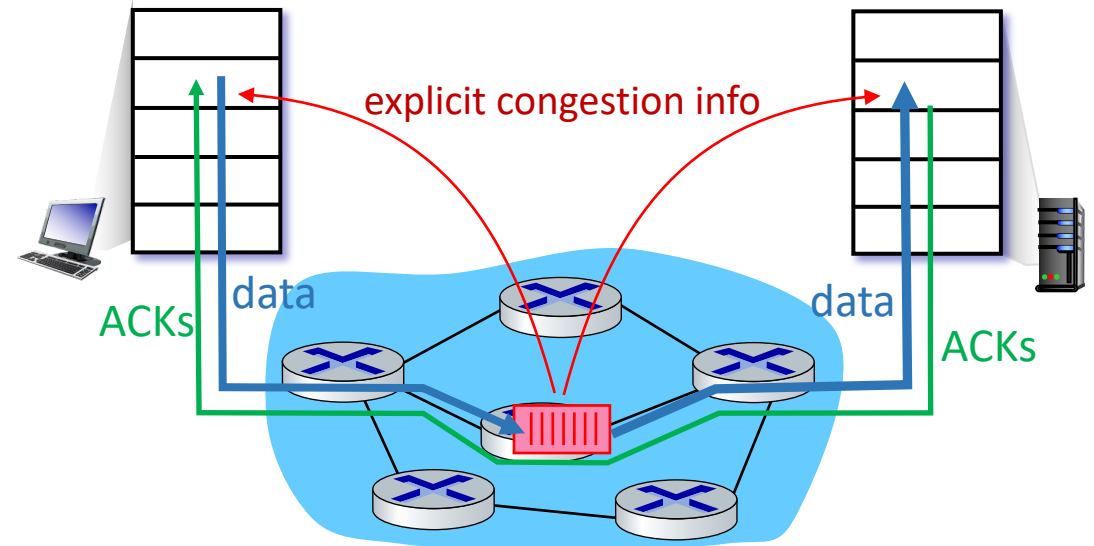
- no explicit feedback from network
- congestion *inferred* from observed loss, delay
- approach taken by TCP



Approaches towards congestion control

Network-assisted congestion control:

- routers provide *direct* feedback to sending/receiving hosts with flows passing through congested router
- may indicate congestion level or explicitly set sending rate
- TCP ECN



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TCP congestion control: AIMD

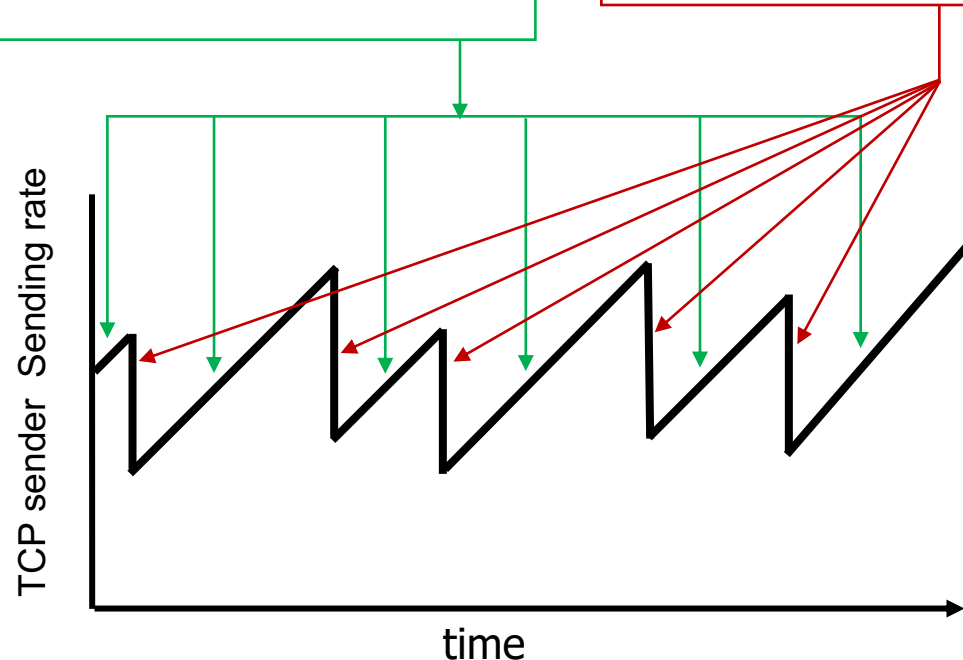
- *approach*: senders can increase sending rate until packet loss (congestion) occurs, then decrease sending rate on loss event

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

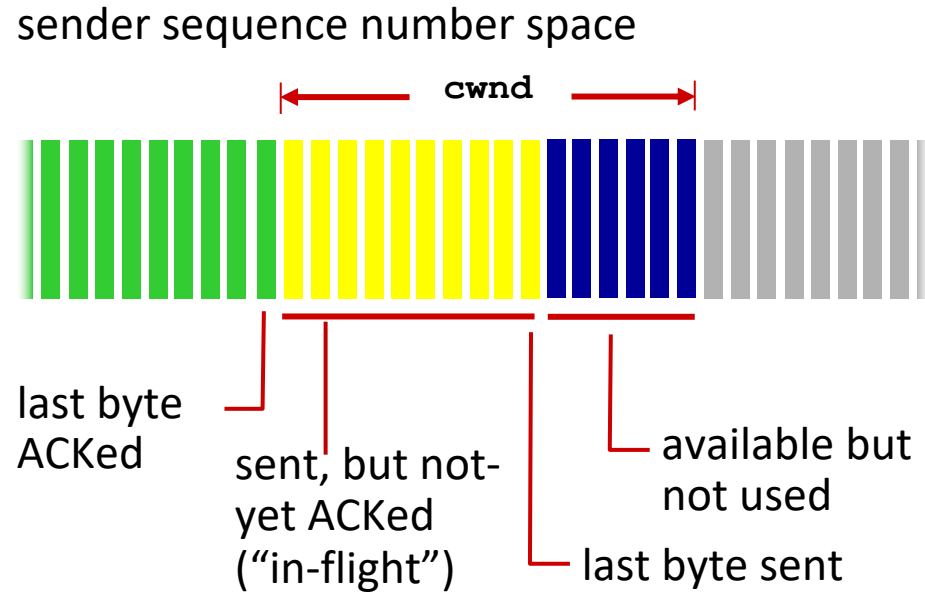
Multiplicative decrease detail: sending rate is

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

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:

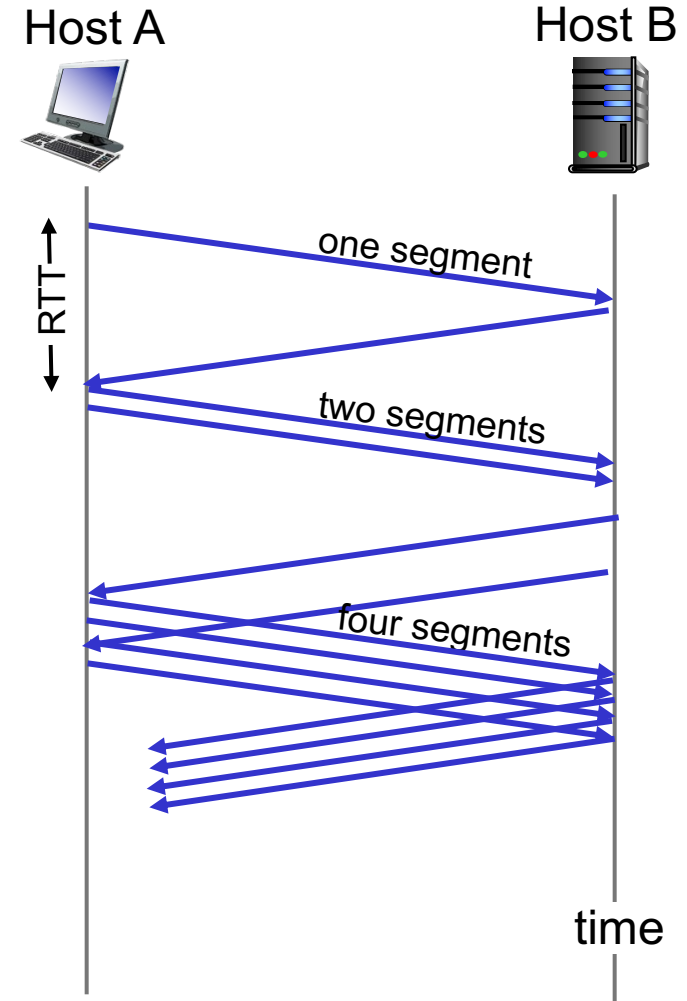
- *roughly*: send `cwnd` bytes, wait RTT for ACKS, then send more bytes

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

- 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



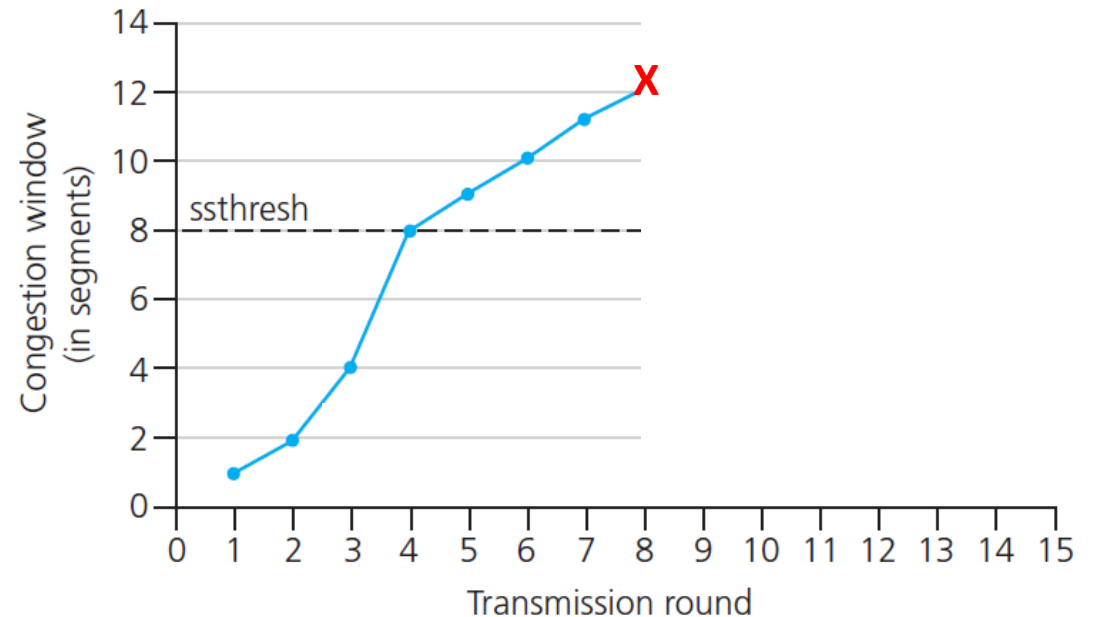
TCP: from slow start to congestion avoidance

Q: when should the exponential increase switch to linear?

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

Implementation:

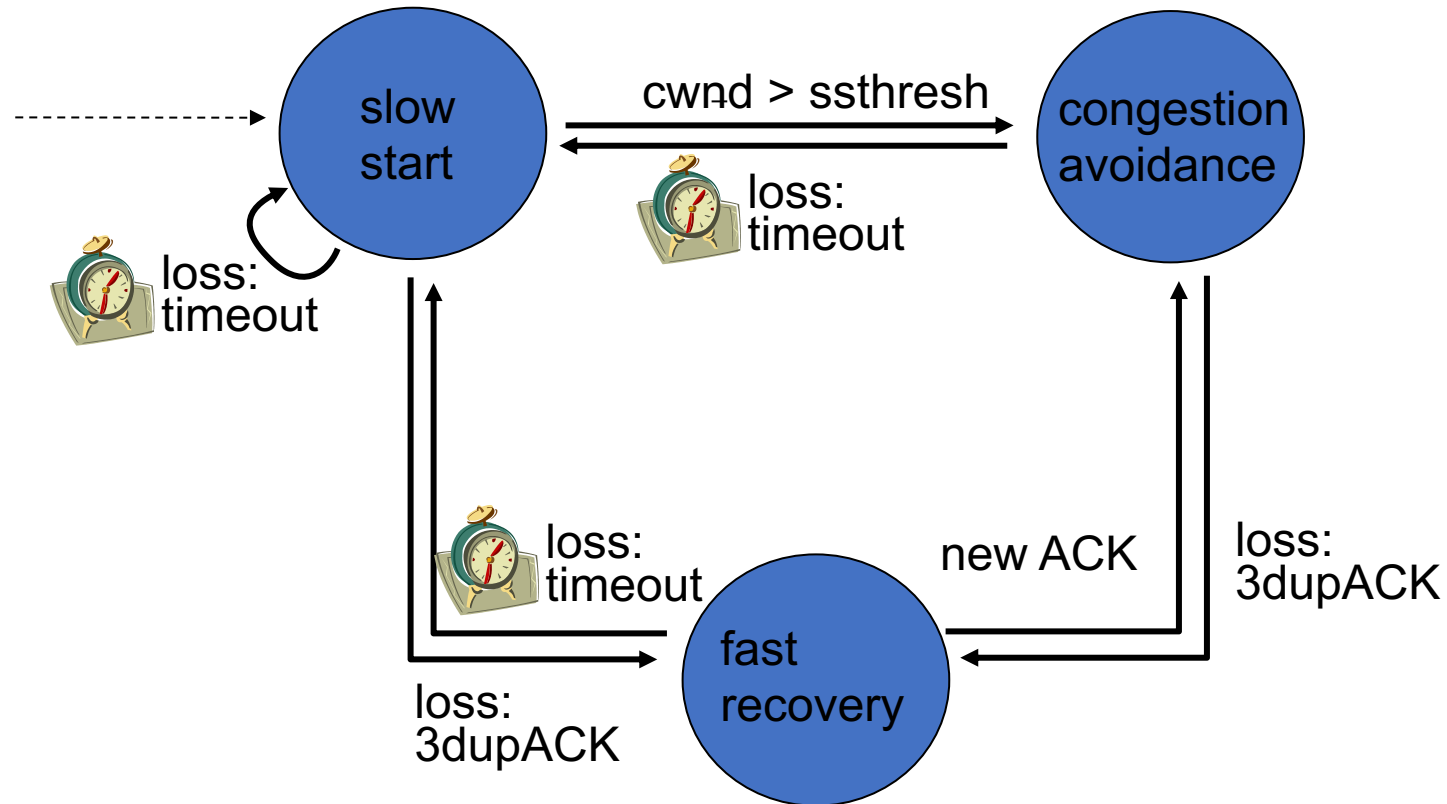
- variable **ssthresh**
- on loss event, **ssthresh** is set to 1/2 of **cwnd** just before loss event



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

Simplified TCP congestion control

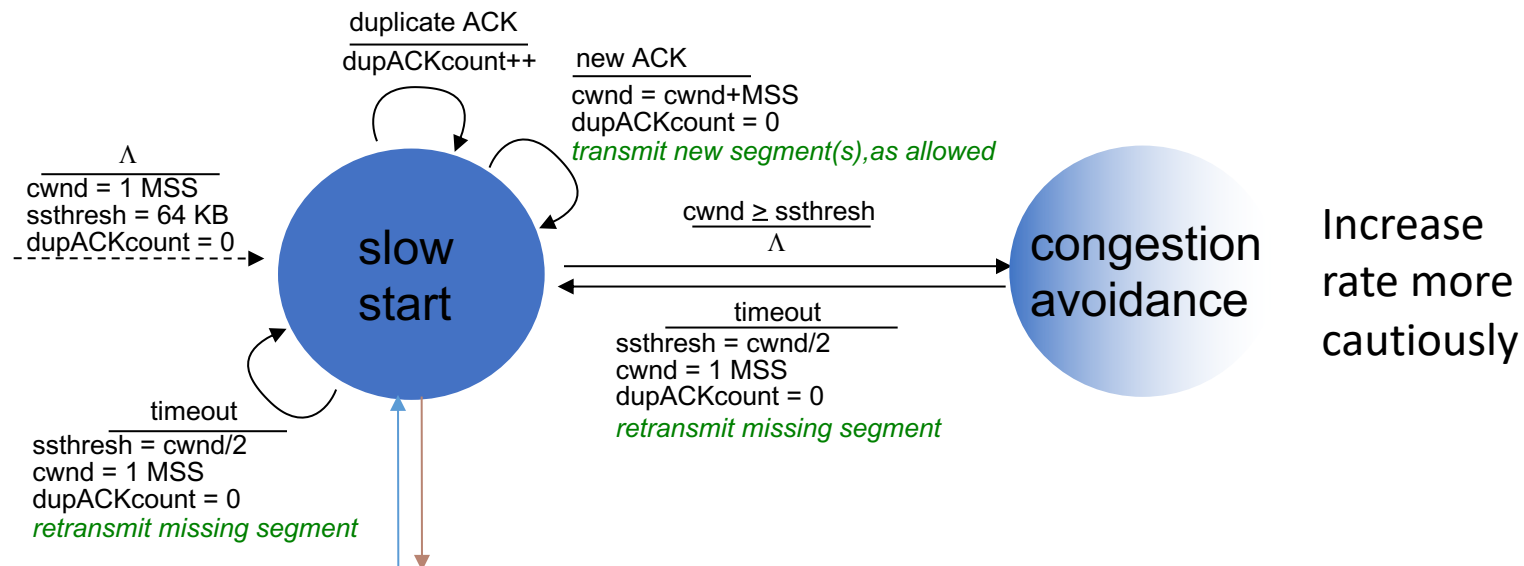
Fast Recovery introduced in 1997 (RFC2001)



TCP Slow Start: more details

ssthresh: **cwnd** threshold maintained by TCP

- on loss event: set **ssthresh** to **cwnd/2**
 - remember (half of) TCP rate when congestion last occurred
- when **cwnd** \geq **ssthresh**: transition from slowstart to congestion avoidance phase



TCP Slow Start: more details

- Things have changed since 1989 or 1997

- 1989: RFC1122 – “Requirements for Internet Hosts -- Communication Layers”
- 1997: RFC2001 – “TCP Slow Start, Congestion Avoidance, Fast Retransmit, and Fast Recovery Algorithms”
- 2009: RFC 5681 – “TCP Congestion Control”

RFC 3390

Increasing TCP's Initial Window

October 2002

The upper bound for the initial window is given more precisely in (1):

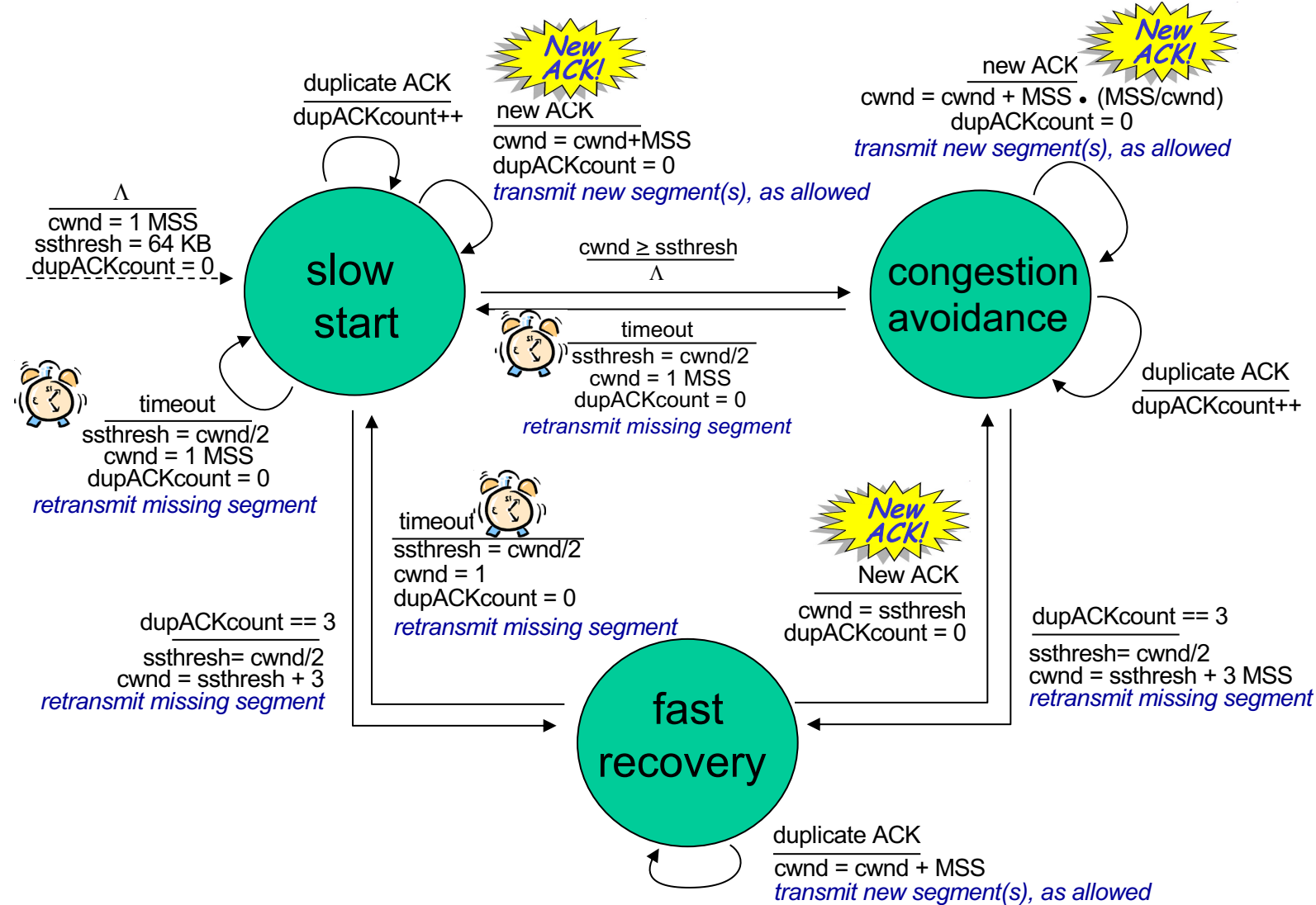
$$\min (4 * \text{MSS}, \max (2 * \text{MSS}, 4380 \text{ bytes})) \quad (1)$$

Note: Sending a 1500 byte packet indicates a maximum segment size (MSS) of 1460 bytes (assuming no IP or TCP options). Therefore, limiting the initial window's MSS to 4380 bytes allows the sender to transmit three segments initially in the common case when using 1500 byte packets.

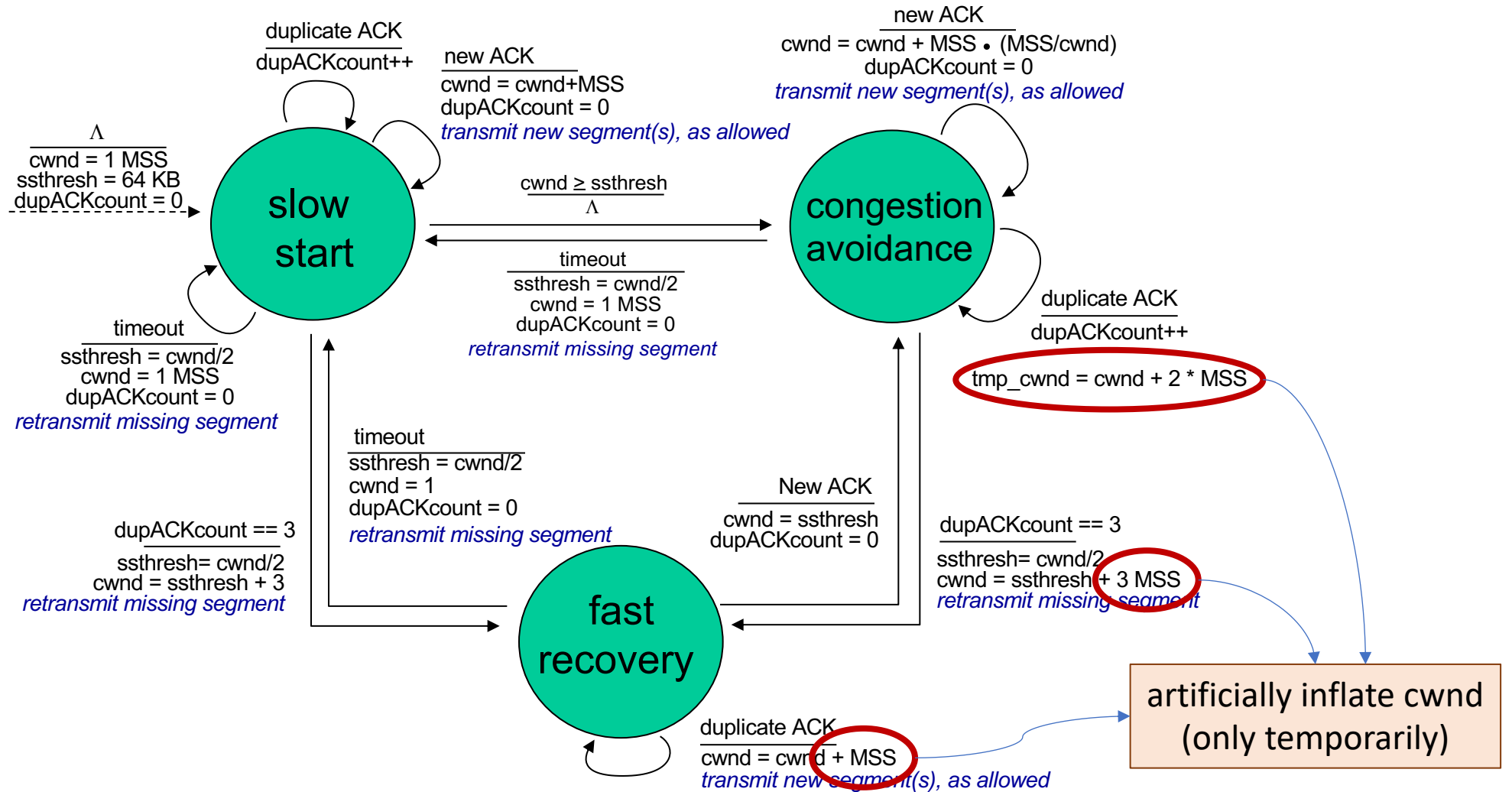
Note: some applications ***cheat*** on the slow start!

<https://web.archive.org/web/20120113055158/http://blog.benstrong.com/2010/11/google-and-microsoft-cheat-on-slow.html>

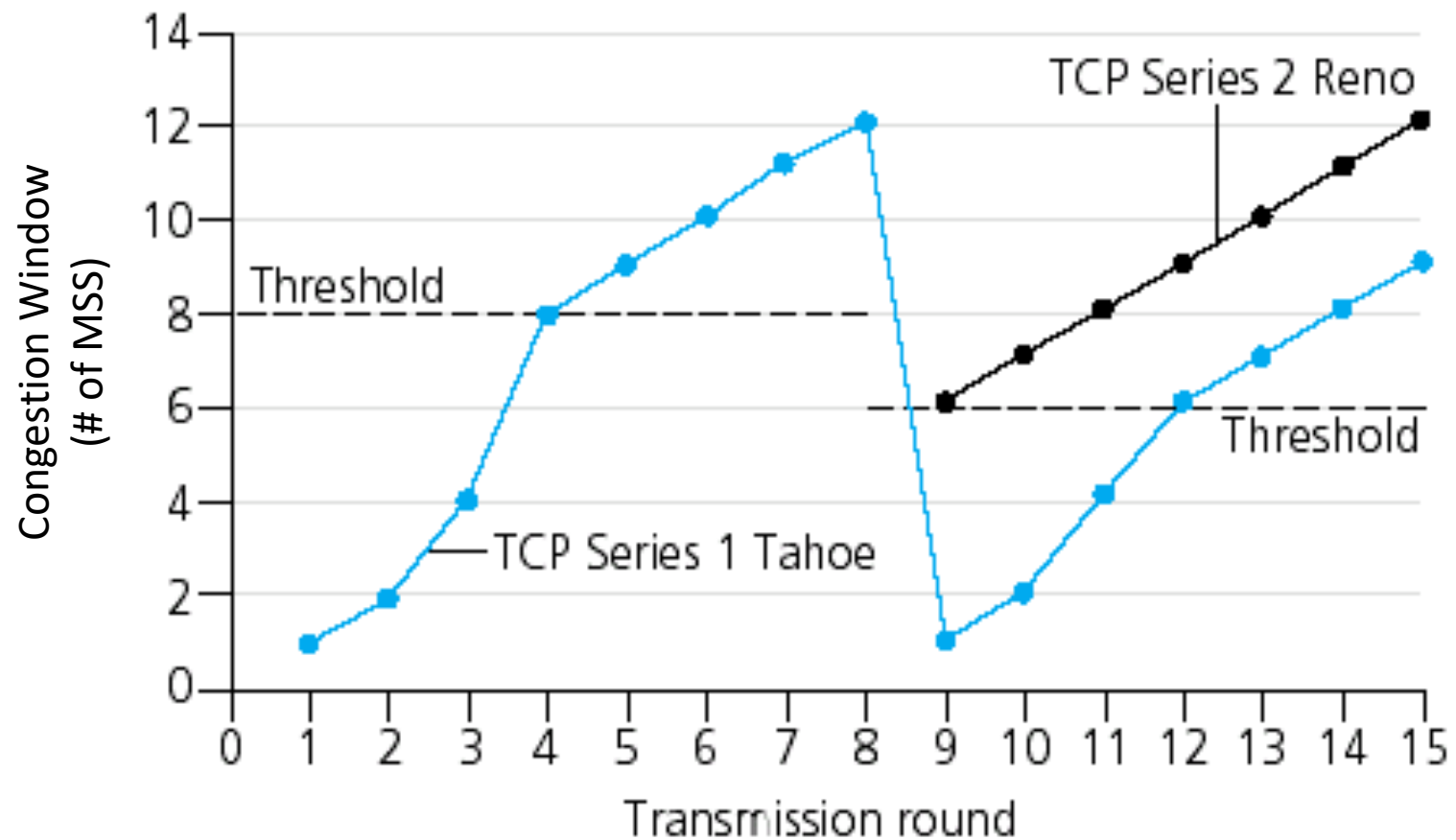
Summary: TCP congestion control



Summary: TCP congestion control



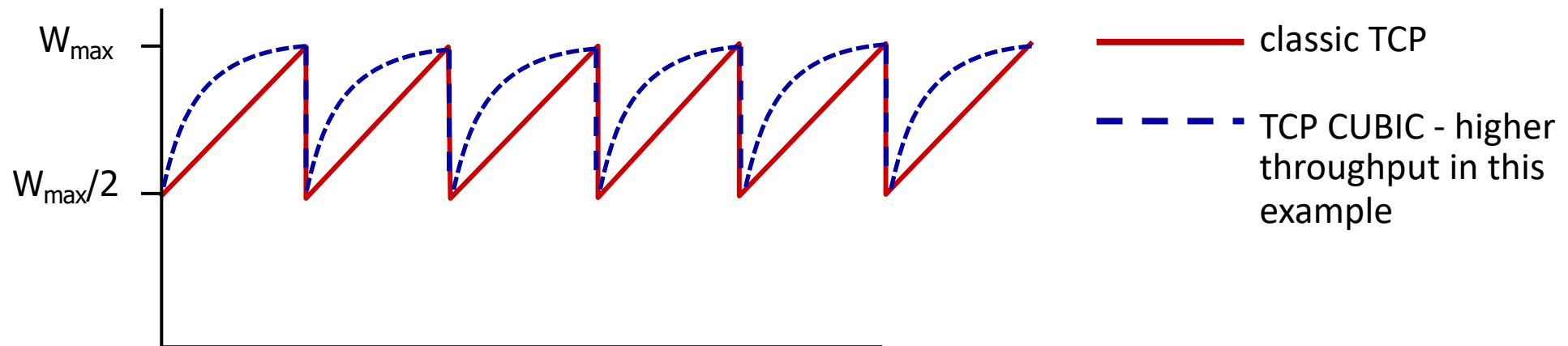
TCP Tahoe vs. Reno



Tahoe did not
use fast recovery

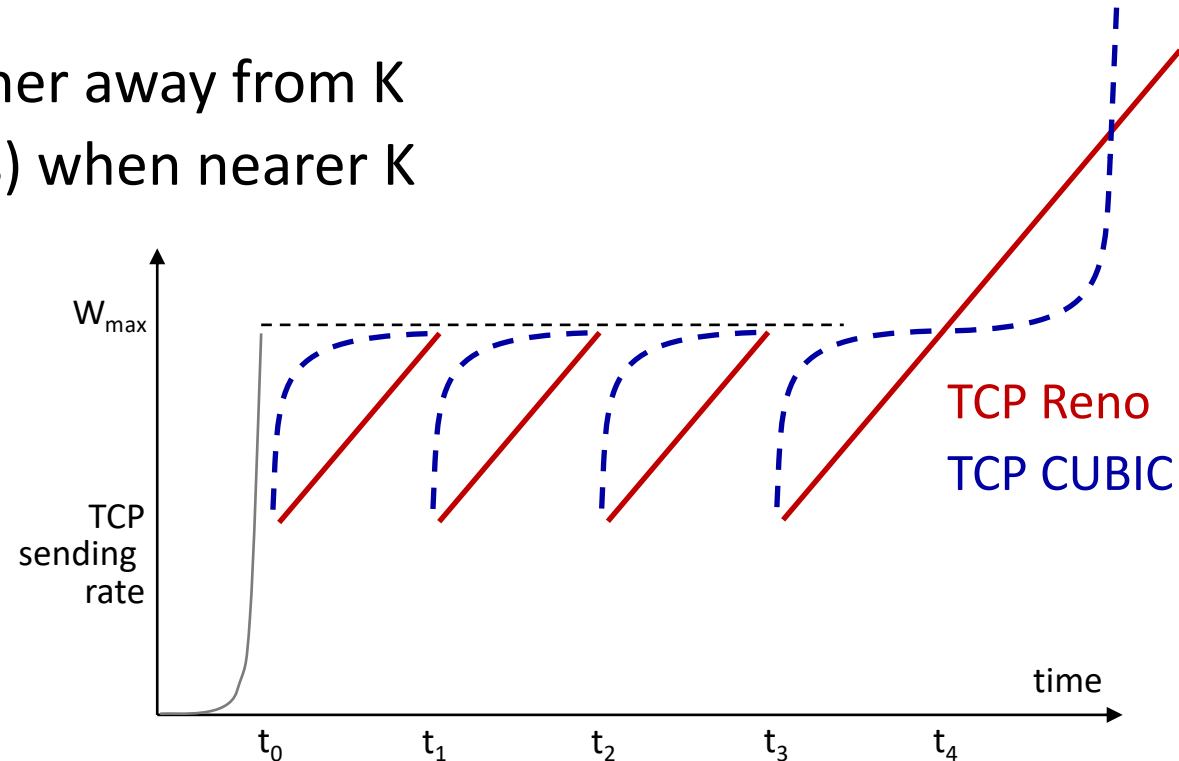
TCP CUBIC

- 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 up to W_{\max} *faster*, but then approach W_{\max} more *slowly*



TCP CUBIC

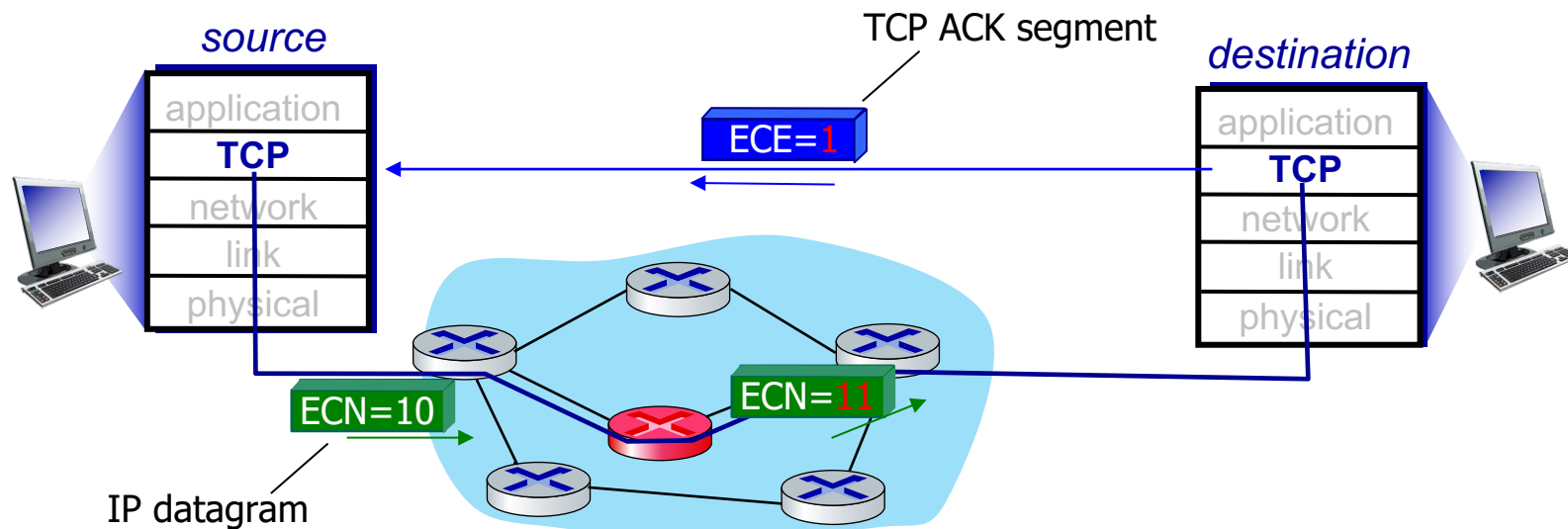
- K: point in time when TCP window size will reach W_{\max}
 - K itself is tunable
- increase W as a function of the *cube* of the distance between current time and K
 - larger increases when further away from K
 - smaller increases (cautious) when nearer K
- TCP CUBIC default in Linux, most popular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



Explicit congestion notification (ECN)

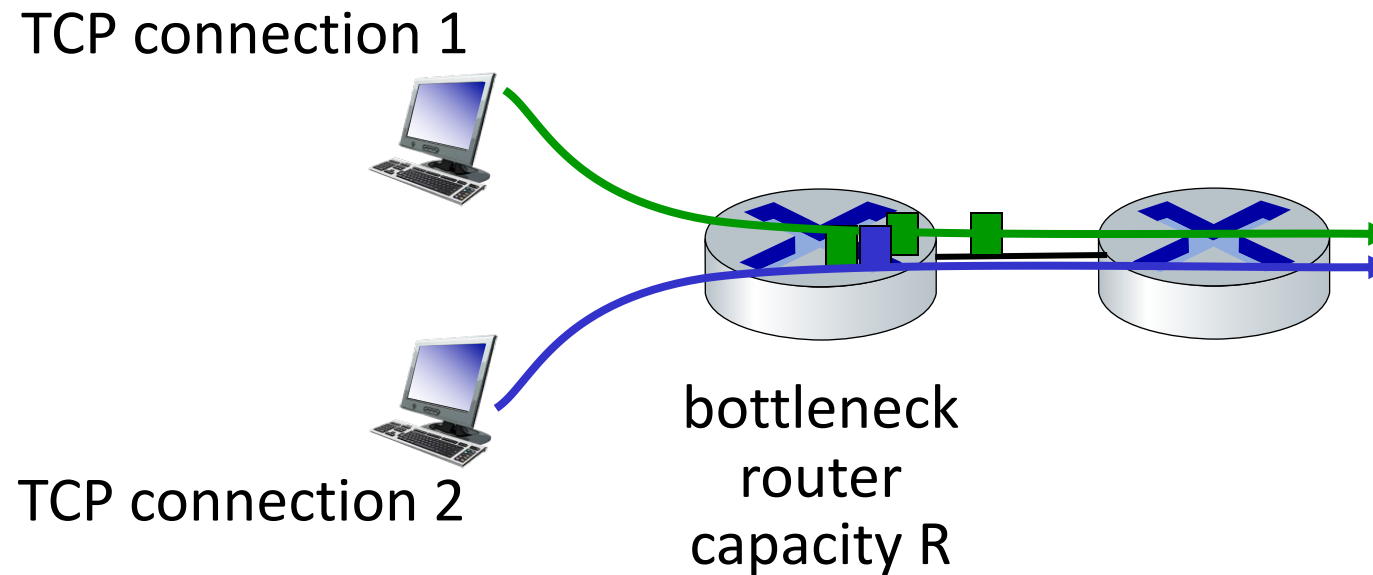
TCP deployments often implement *network-assisted* congestion control:

- 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

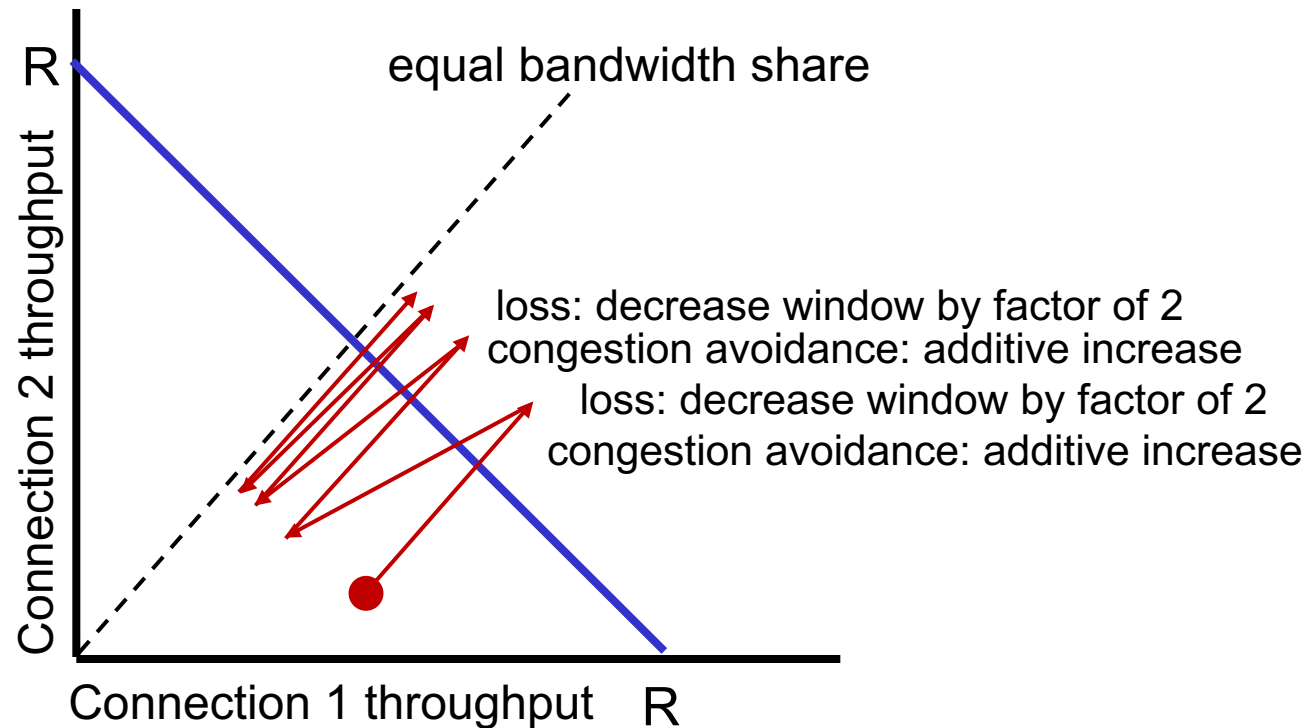
Fairness goal: if K TCP sessions share same bottleneck link of bandwidth R , each should have average rate of R/K



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 $R/2$

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Evolving transport-layer functionality

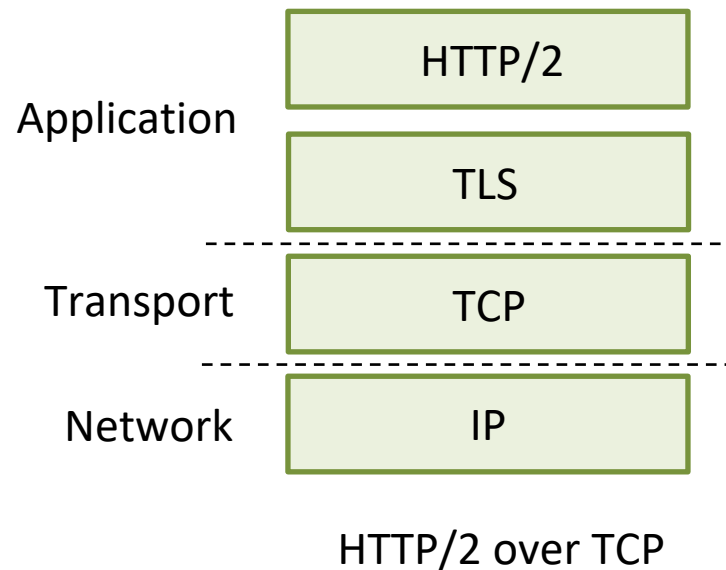
- TCP, UDP: principal transport protocols for 40 years
- different “flavors” of TCP developed, for specific scenarios:

Scenario	Challenges
Long, fat pipes (large data transfers)	Many packets “in flight”; loss shuts down pipeline
Wireless networks	Loss due to noisy wireless links, mobility; TCP treat this as congestion loss
Long-delay links	Extremely long RTTs
Data center networks	Latency sensitive
Background traffic flows	Low priority, “background” TCP flows

- moving transport–layer functions to application layer, on top of UDP
 - HTTP/3: QUIC

QUIC: Quick UDP Internet Connections

- application-layer protocol, on top of UDP
 - increase performance of HTTP
 - deployed on many Google servers, apps (Chrome, mobile YouTube app)

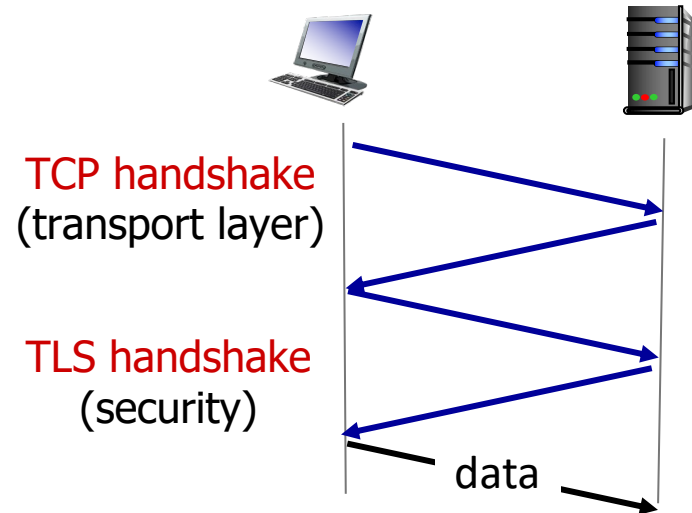


QUIC: Quick UDP Internet Connections

adopts approaches we've studied in this chapter for connection establishment, error control, congestion control

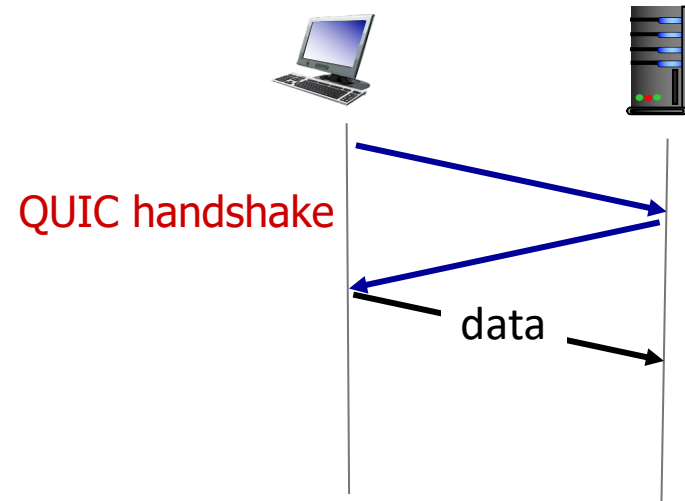
- **error and congestion control:** “Readers familiar with TCP’s loss detection and congestion control will find algorithms here that parallel well-known TCP ones.” [from QUIC specification]
- **connection establishment:** reliability, congestion control, authentication, encryption, state established in one RTT
- multiple application-level “streams” multiplexed over single QUIC connection
 - separate reliable data transfer, security
 - common congestion control

QUIC: Connection establishment



TCP (reliability, congestion control state) + TLS (authentication, crypto state)

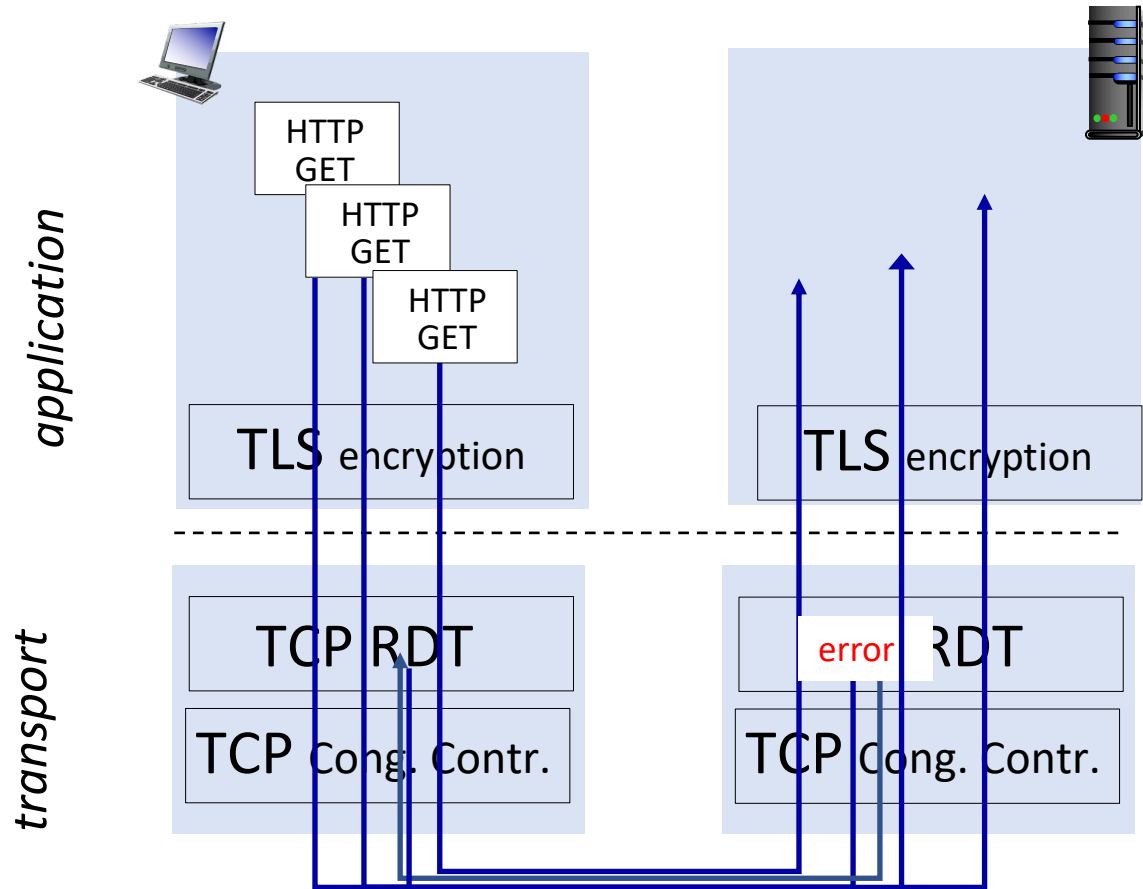
- 2 serial handshakes



QUIC: reliability, congestion control, authentication, crypto state

- 1 handshake

QUIC: streams: parallelism, no HOL blocking



(a) HTTP 1.1

Summary

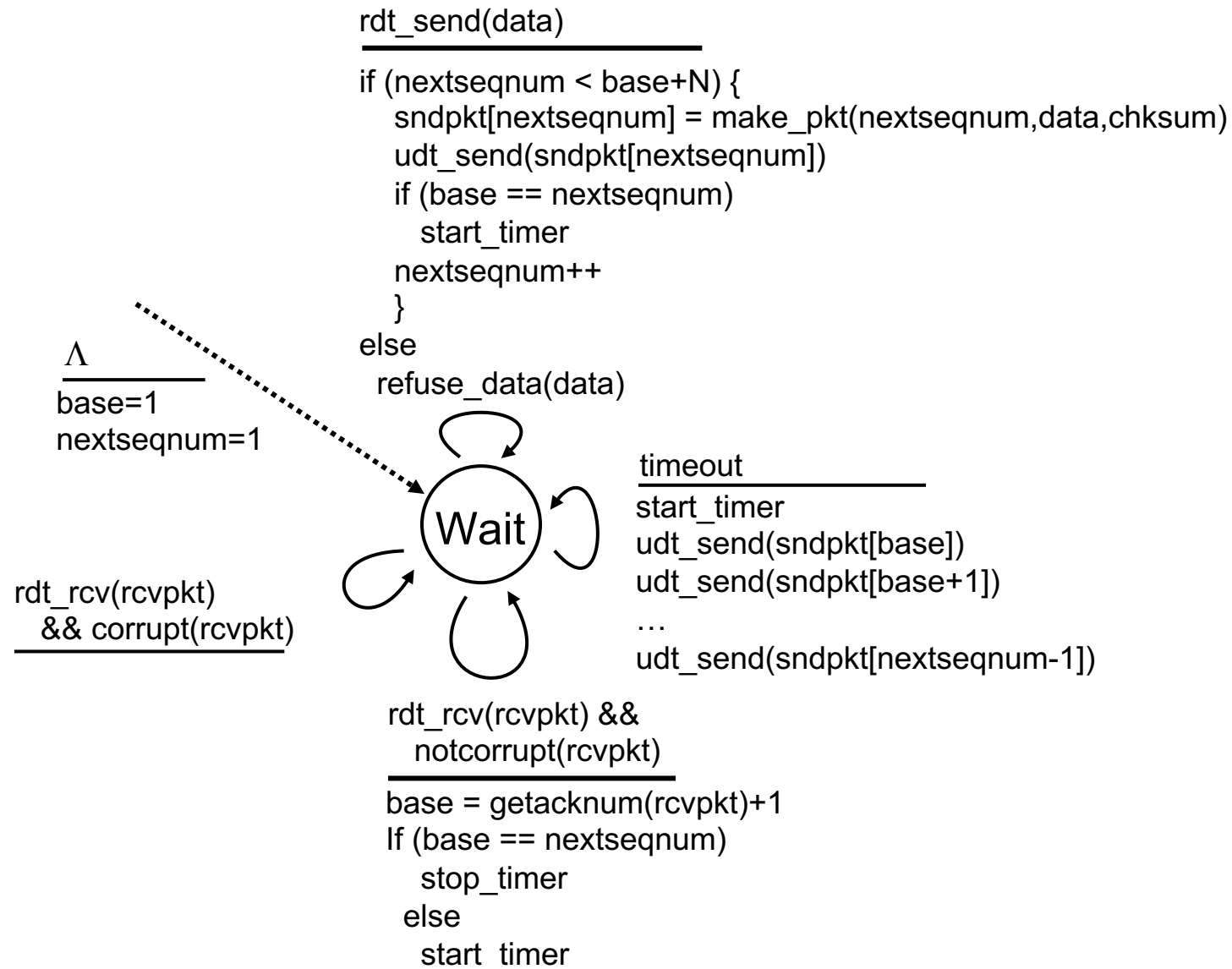
- principles behind transport layer services:
 - multiplexing, demultiplexing
 - reliable data transfer
 - flow control
 - congestion control
- instantiation, implementation in the Internet
 - UDP
 - TCP

Up next:

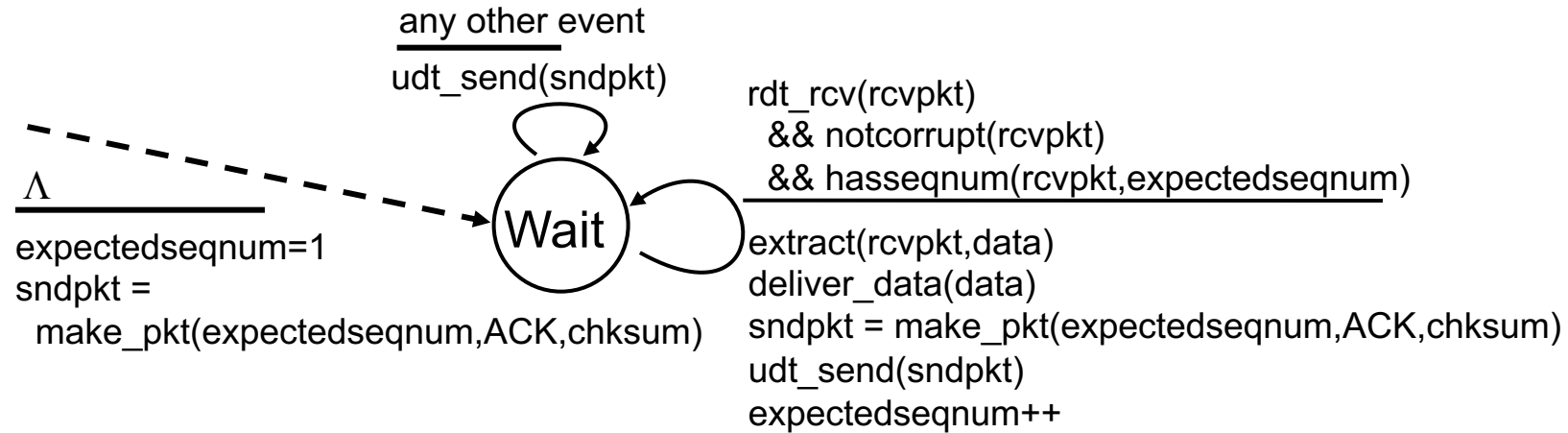
- leaving the network “edge” (application, transport layers)
- into the network “core”
- two network-layer chapters:
 - data plane
 - control plane

Additional Slides

Go-Back-N: sender extended FSM



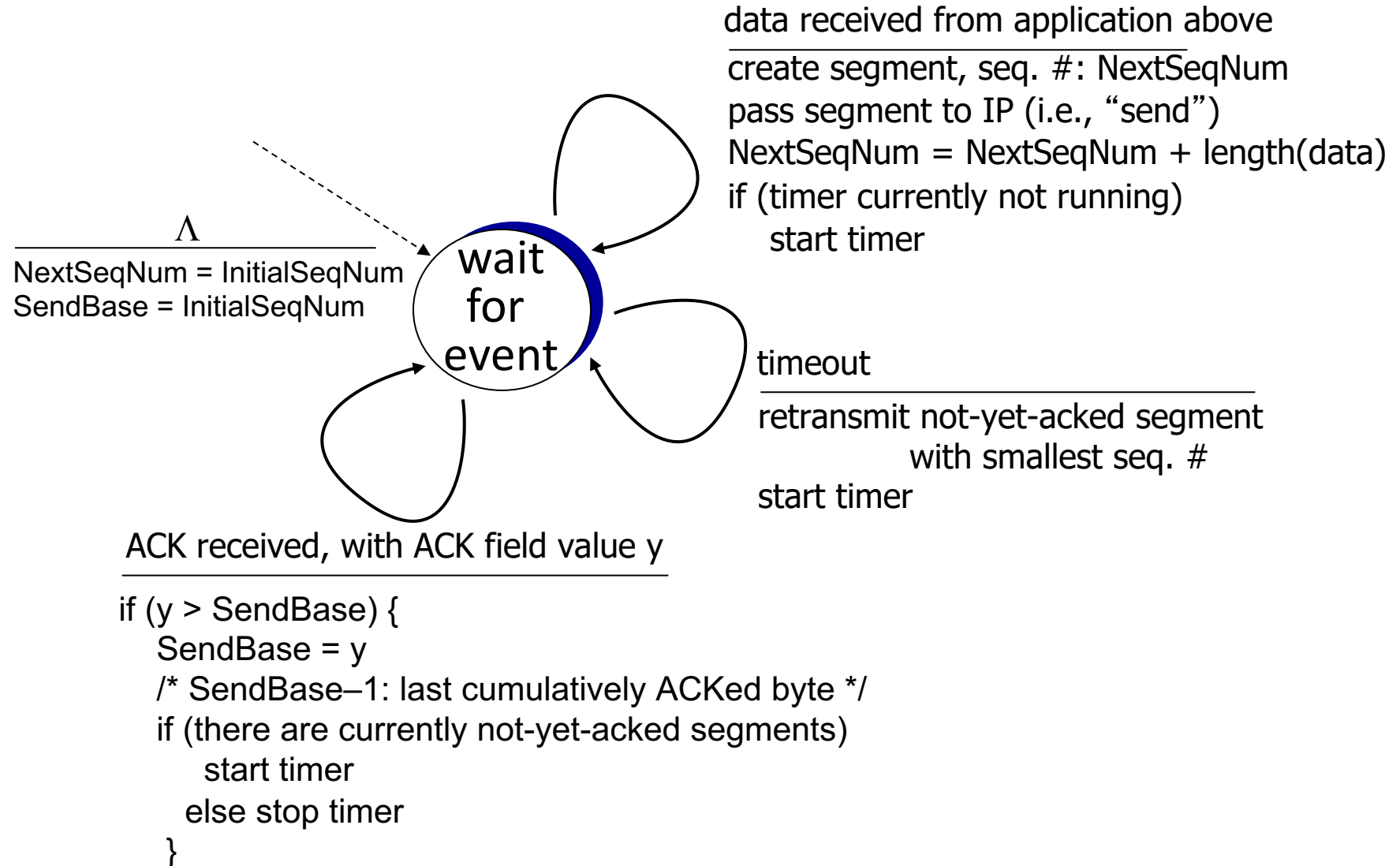
Go-Back-N: receiver extended FSM



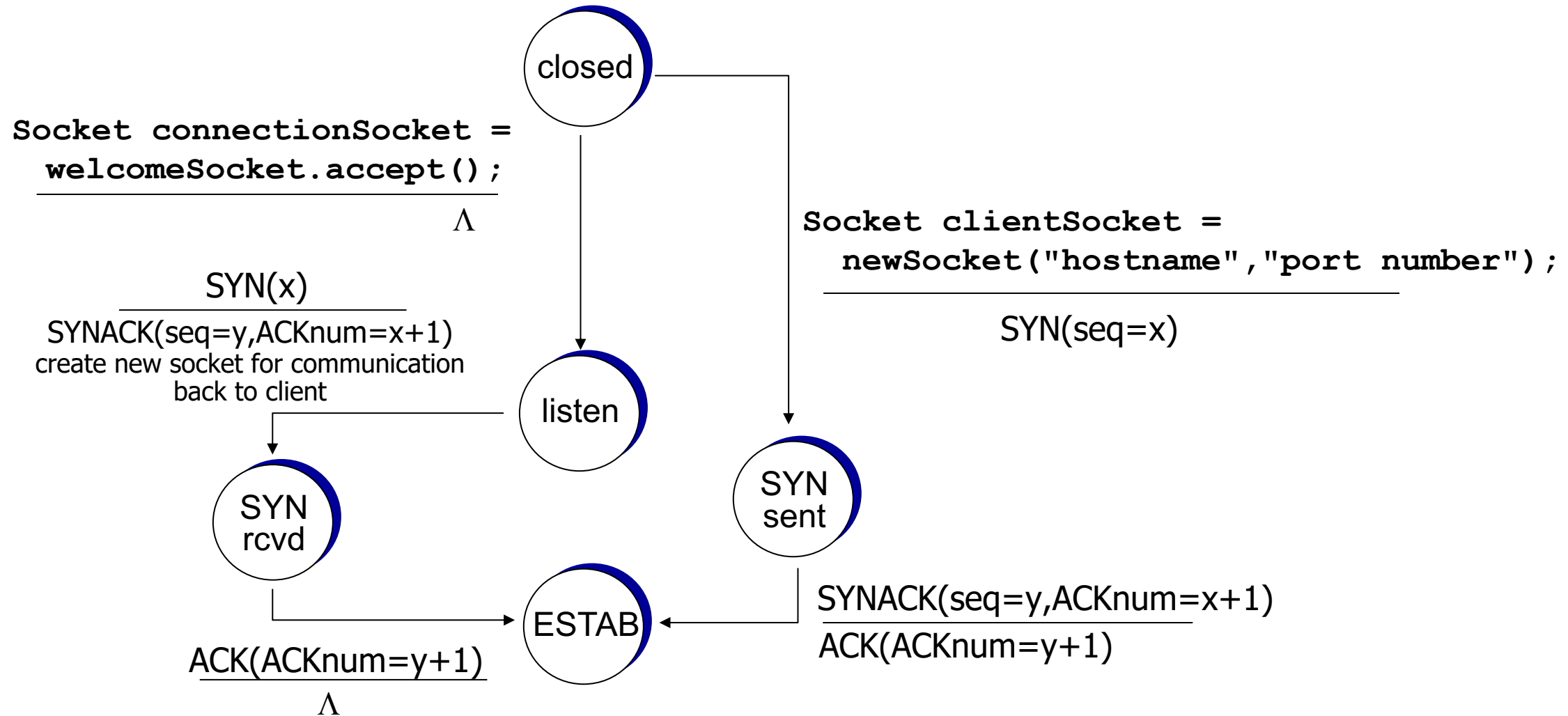
ACK-only: always send ACK for correctly-received packet with highest *in-order* seq #

- may generate duplicate ACKs
 - need only remember **expectedseqnum**
- out-of-order packet:
- discard (don't buffer): *no receiver buffering!*
 - re-ACK pkt with highest in-order seq #

TCP sender (simplified)

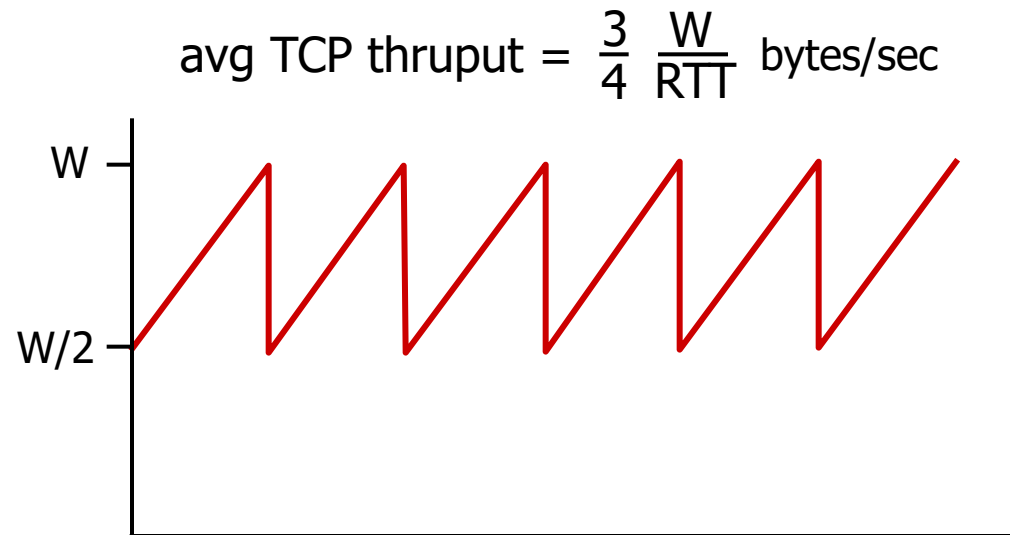


TCP 3-way handshake FSM



TCP throughput

- avg. TCP thruput 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. thruput 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