



UNIVERSITY OF
WATERLOO

CS 456/656

Computer Networks

Lecture 7: Transport Layer – Part 3

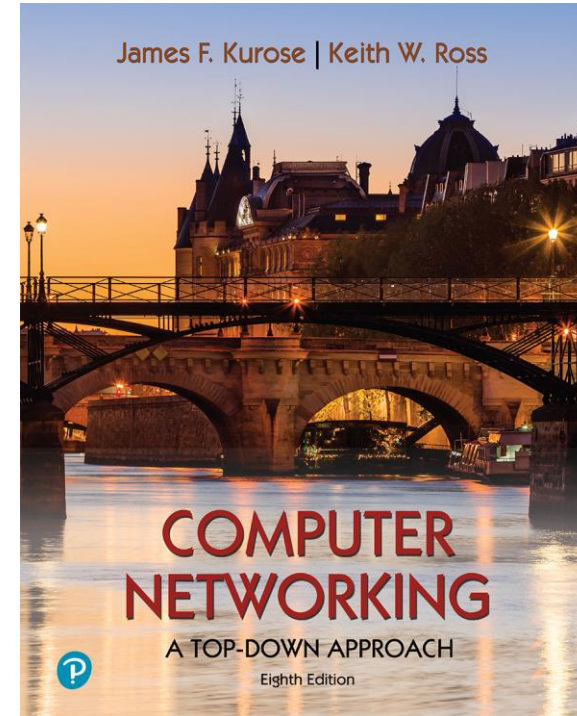
Mina Tahmasbi Arashloo and Bo Sun

Fall 2024

A note on the slides

Adapted from the slides that accompany this book.

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Computer Networking: A Top-Down Approach

8th edition

Jim Kurose, Keith Ross

Pearson, 2020

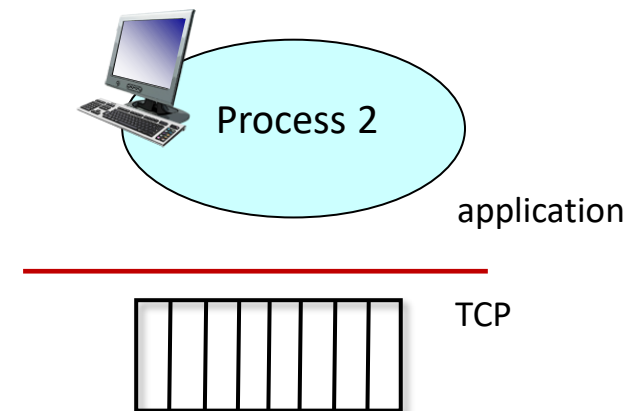
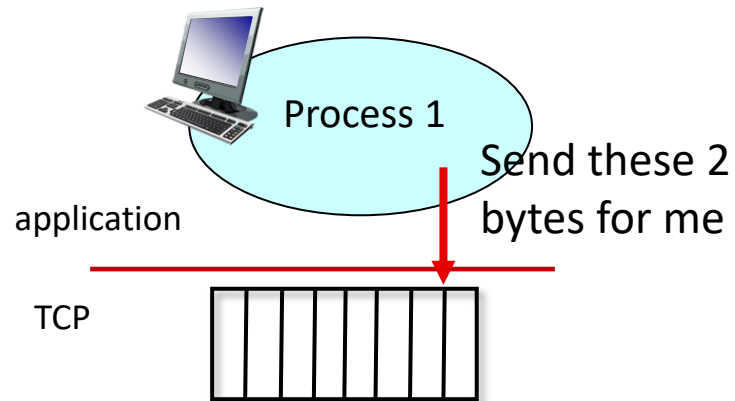
Transport layer: roadmap

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



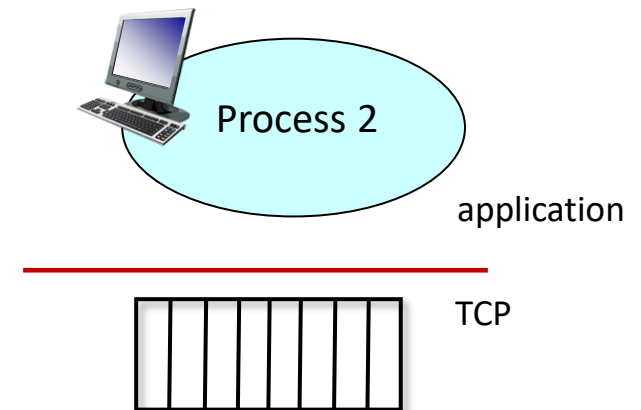
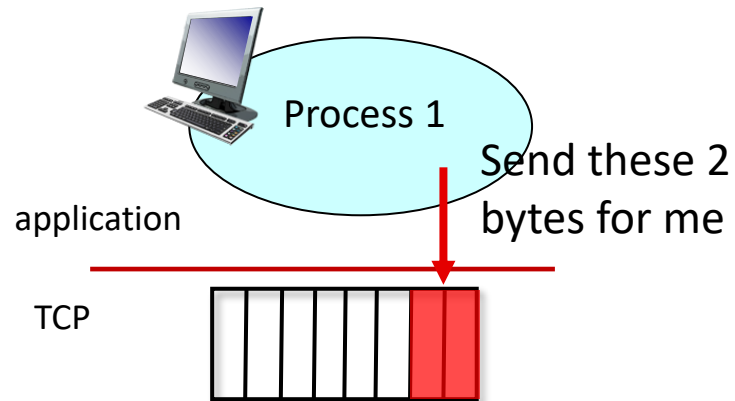
TCP: a widely-used reliable transport protocol

- Guarantees **reliable, in-order *byte stream***:
 - no “message boundaries”



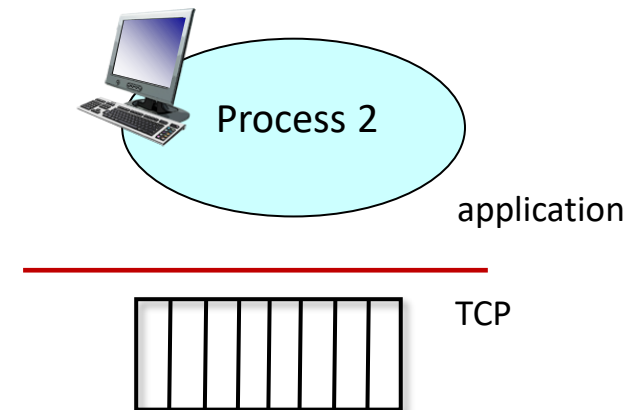
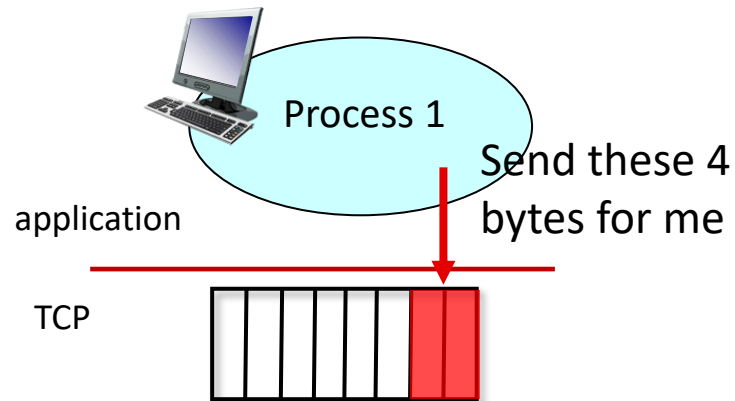
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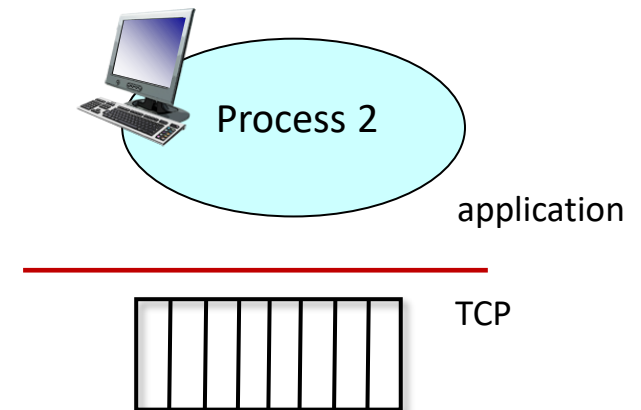
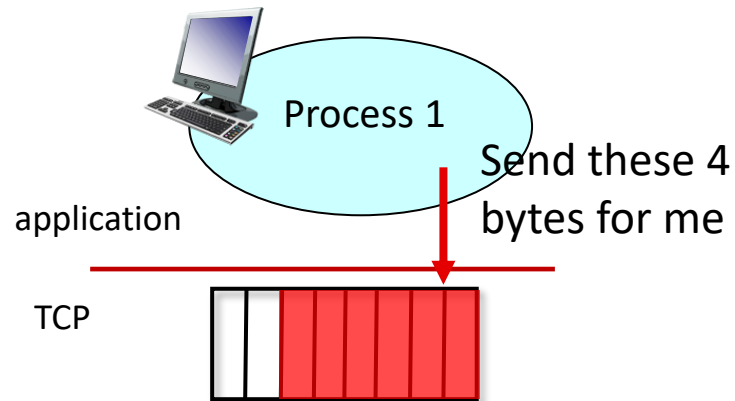
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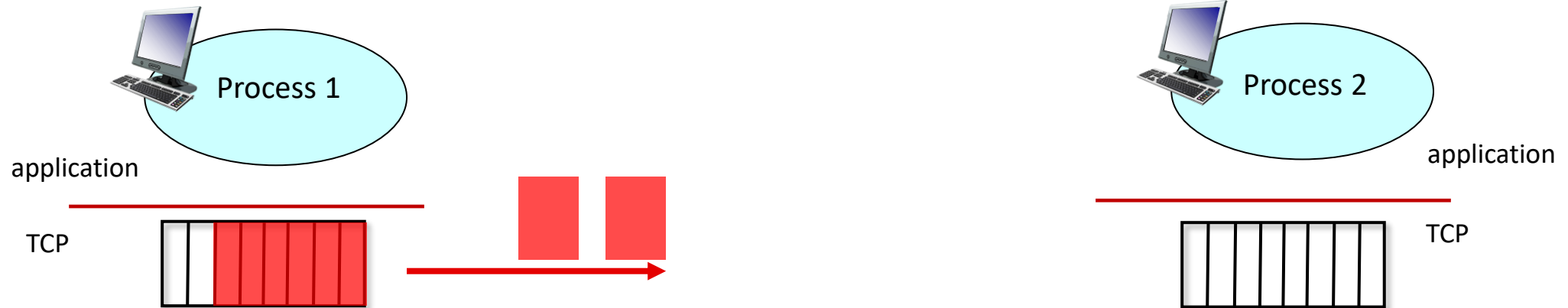
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TCP: a widely-used reliable transport protocol

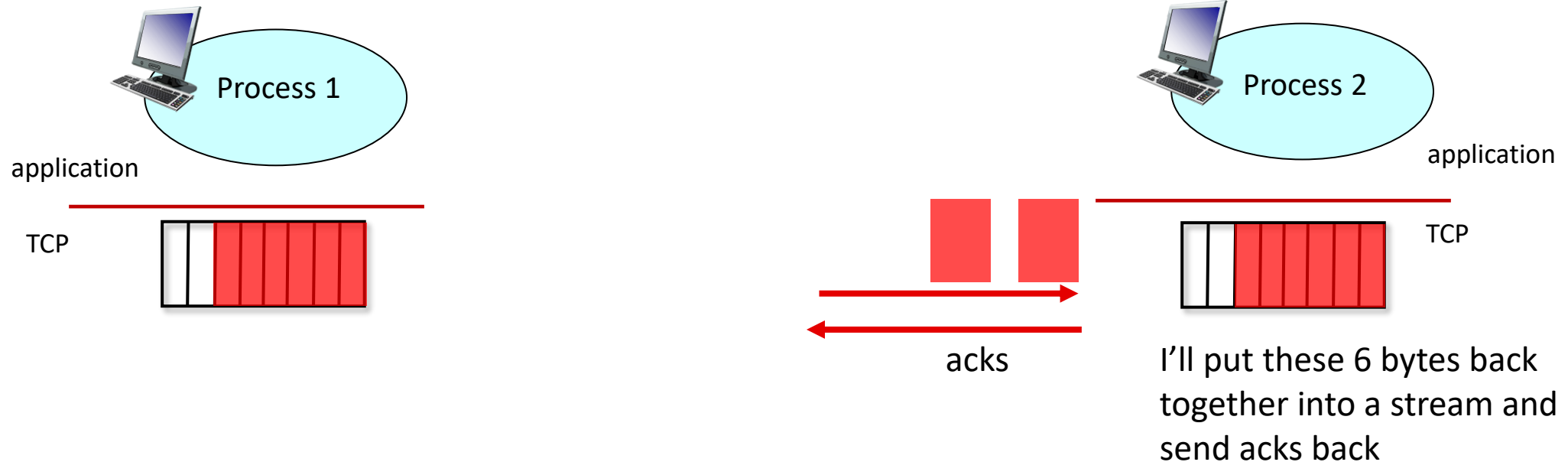
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 - no “message boundaries”



I'll fit them into 2 packets,
each with 3 bytes of the data

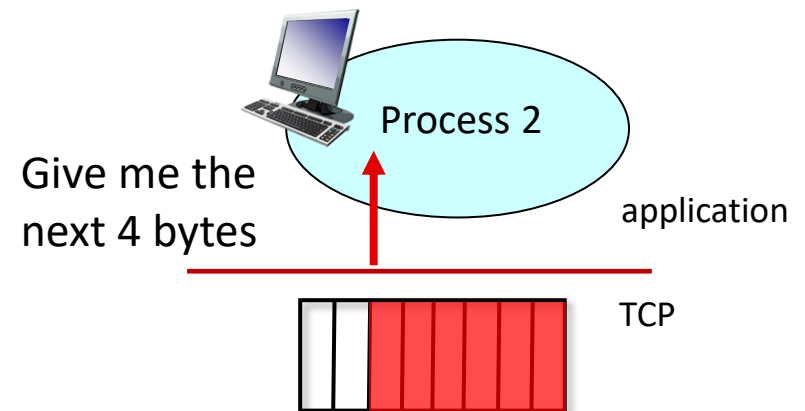
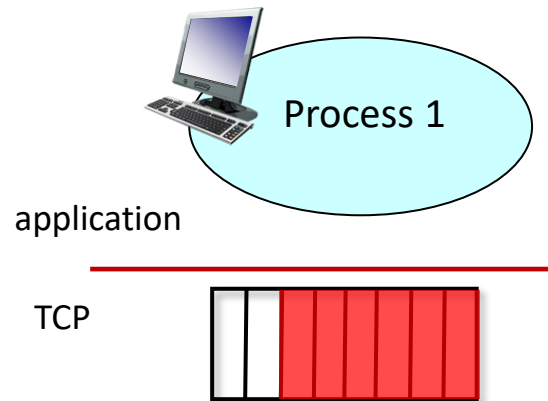
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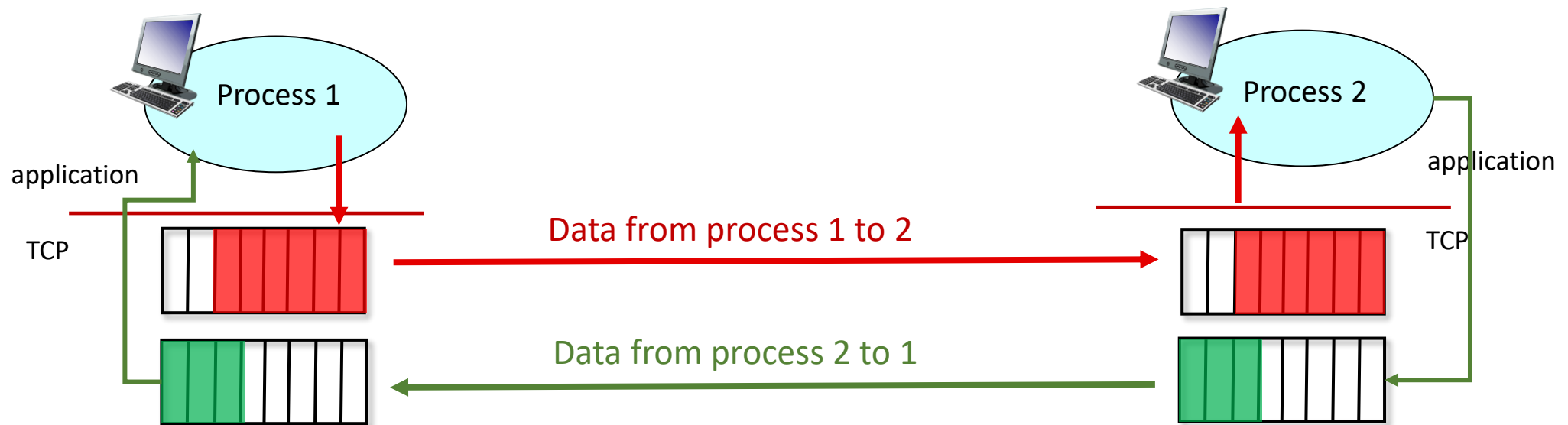
TCP: a widely-used reliable transport protocol

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 - no “message boundaries”



TCP: a widely-used reliable transport protocol

- Guarantees **reliable, in-order *byte stream***:
 - no “message boundaries”
- **full duplex data**: Possible to send data both ways once the two processes establish a connection



TCP: a widely-used reliable transport protocol

- Guarantees **reliable, in-order *byte stream***:
 - no “message boundaries”
- **full duplex data**:
 - Possible to send data both ways once the two processes establish a connection
- Uses the **pipelining approach to reliable data transfer**
 - A combination of techniques from Go-Back-N (cumulative acks) and Selective Repeat (only retransmitting presumably lost segment)
 - Performance optimizations like fast retransmit and delayed acks.

TCP: a widely-used reliable transport protocol

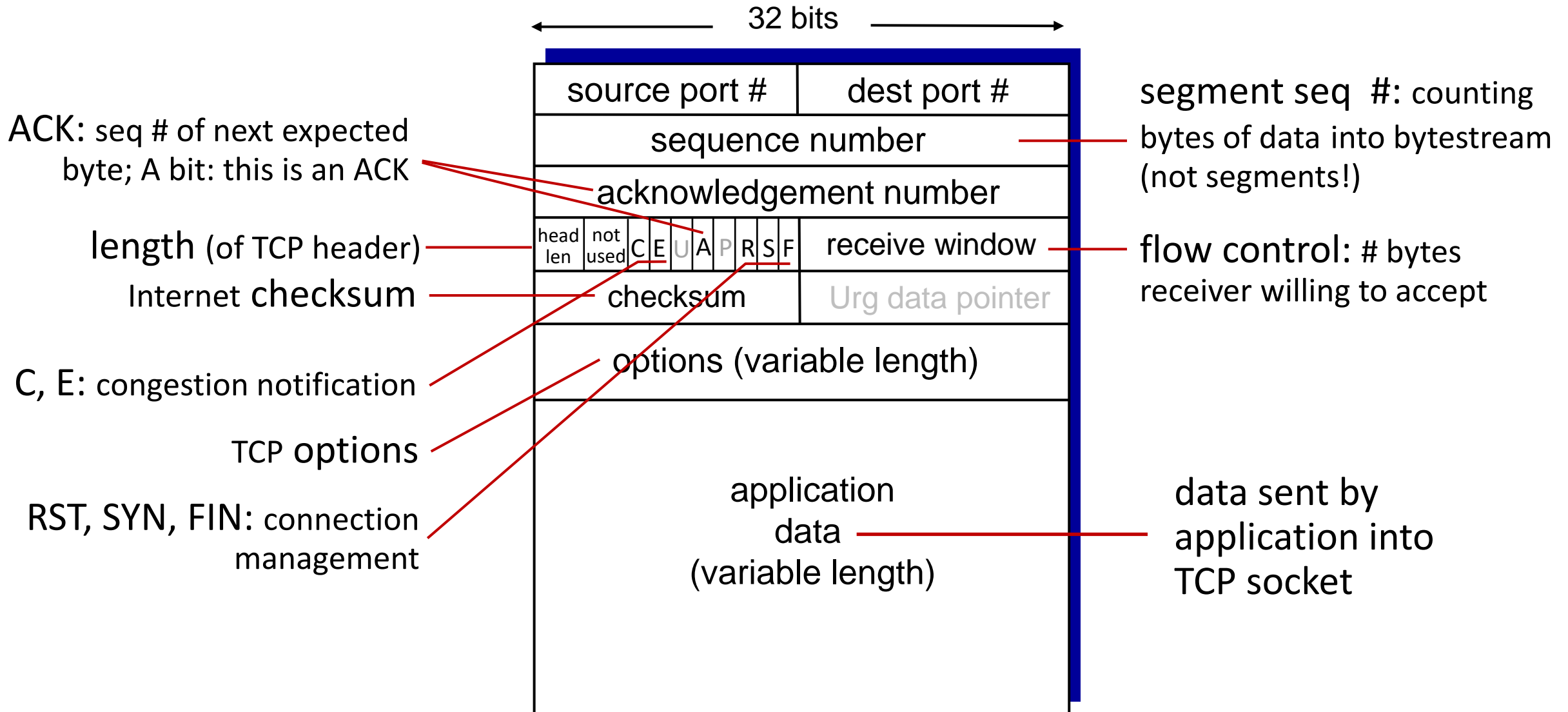
- **Connection-oriented**
 - Connection establishment: Control messages prior to data exchange to initialize the proper state in the communication endpoints
 - Connection tear-down: Control messages after data exchange to end connection
- **Flow controlled**
 - sender will not overwhelm receiver

Transport layer: roadmap

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TCP segment structure



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TCP reliable data transfer

TCP uses all the reliable data transfer tools we have discussed!

- Checksum
- Sequence number
- Receiver feedback (ACK)
- Timer
- Sliding window/pipelining

TCP sequence numbers – one for every byte

- The interface between a sending process and TCP is a byte stream.
- TCP assigns a sequence number to every byte
 - As opposed to every segment, as we discussed in the last lecture
- It keeps track of the “status” of every byte
 - Is it sent yet? Is it acknowledged yet?

First byte has sequence number `init_seq` (initial sequence number)

Next byte has sequence number `init_seq + 1`



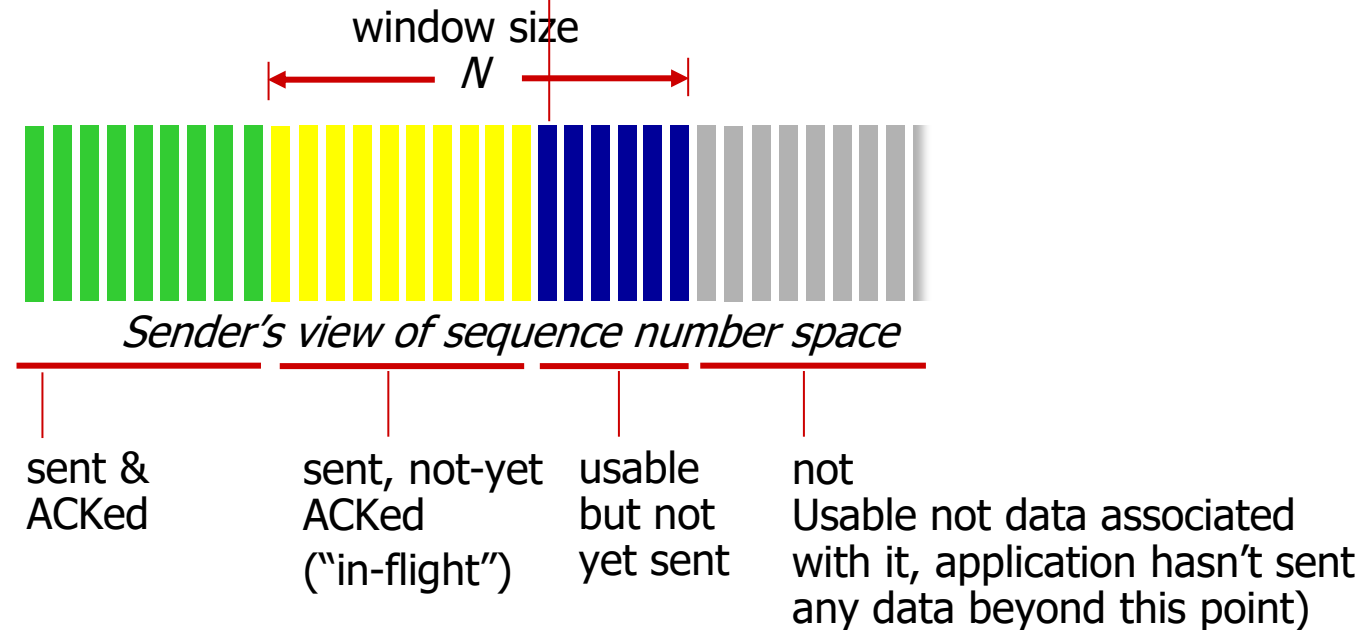
(Colors represent segment status -- see next slide)

The Nth byte has sequence number `init_seq + N - 1`

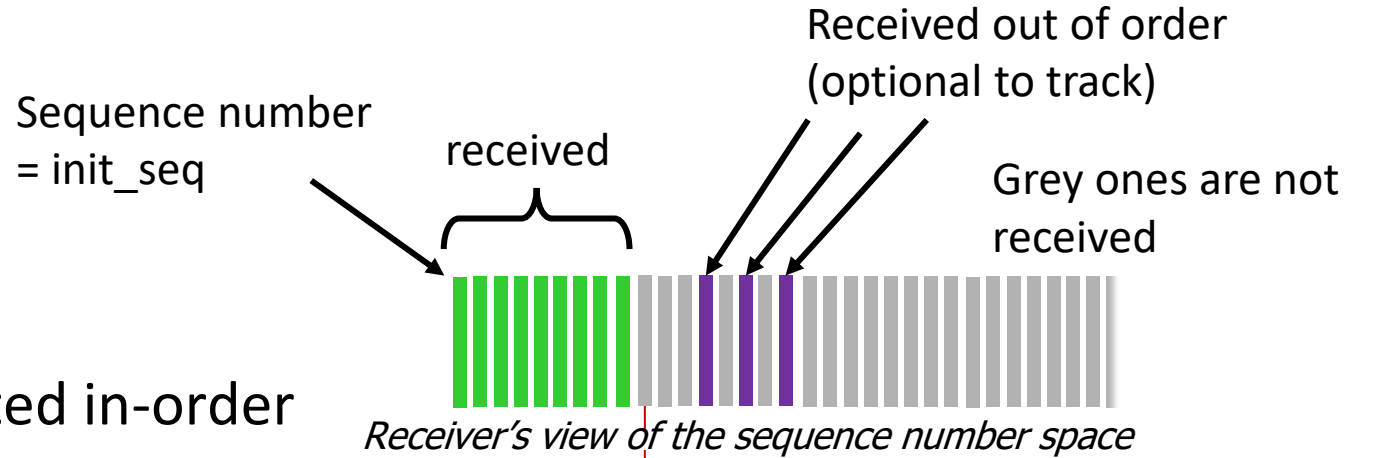
TCP sequence numbers

outgoing segment from sender

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



TCP ACKs



- Cumulative ACK
 - Has seq number of next expected in-order byte
- ACK(n) means:
 - All bytes in [init_seq, n - 1] are received.
 - The receiver is expecting byte n next
- Note the difference from Go-Back-N ack

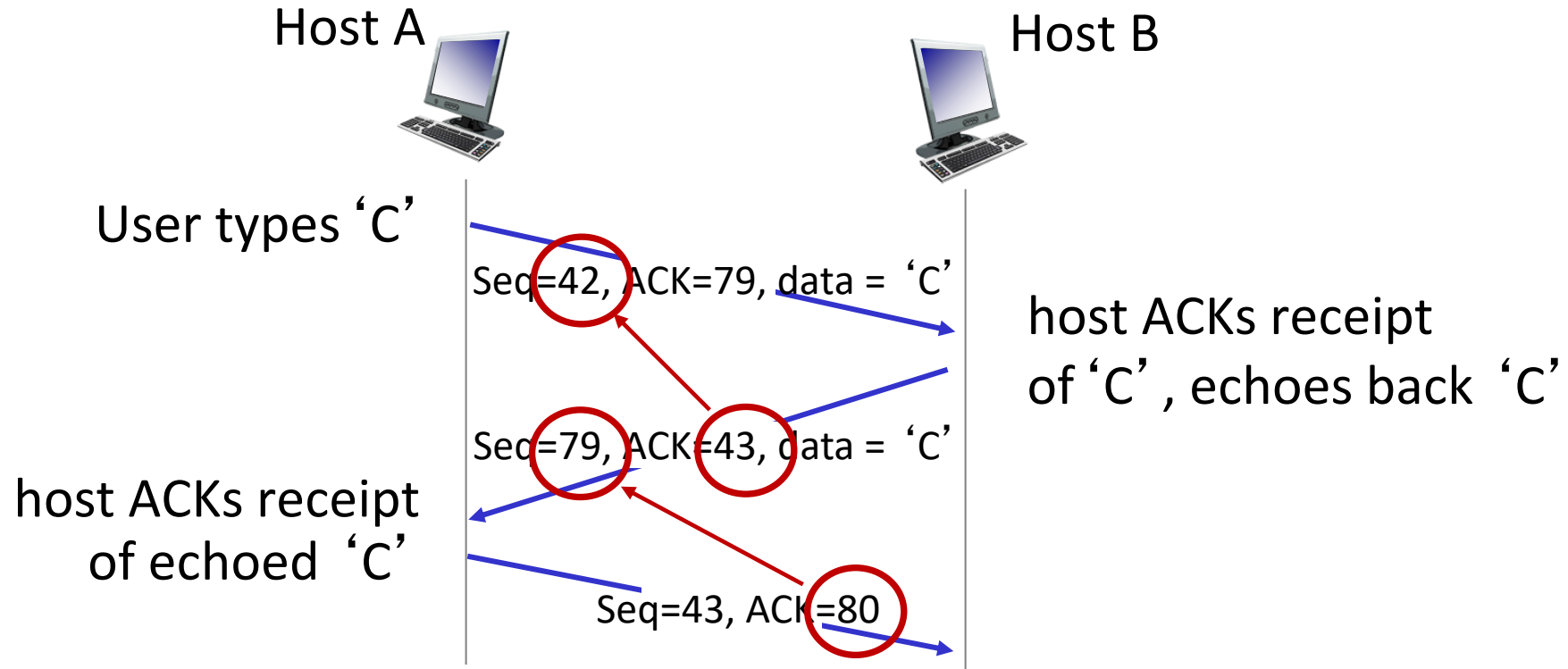
Q: What about out-of-order segments?

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

outgoing segment from receiver

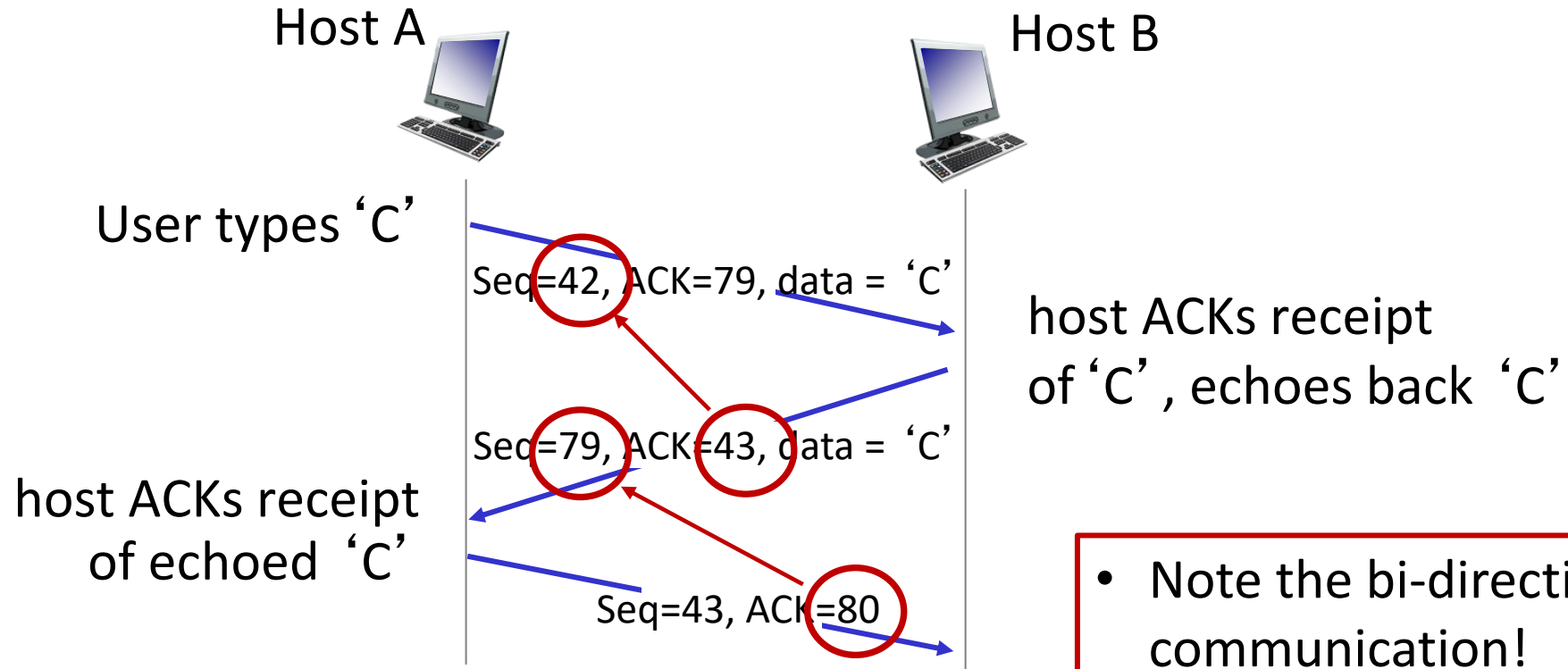
source port #	dest port #
sequence number	
acknowledgement number	
	A
checksum	urg pointer

TCP sequence numbers, ACKs



simple telnet scenario

TCP sequence numbers, ACKs



simple telnet scenario

- Note the bi-directional communication!
- There are two data streams:
 - one in each direction
 - each with its own sequence number space

TCP Sender (simplified)

event: data received from application

- create segment with seq #
- seq # is byte-stream offset 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

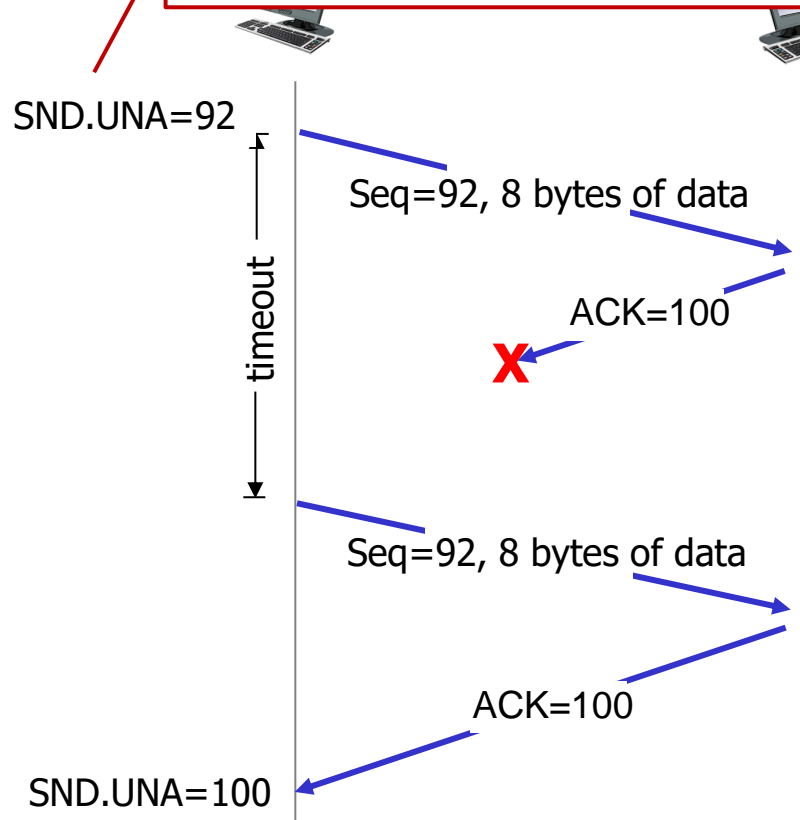
- retransmit segment that caused timeout
- restart timer

event: ACK received

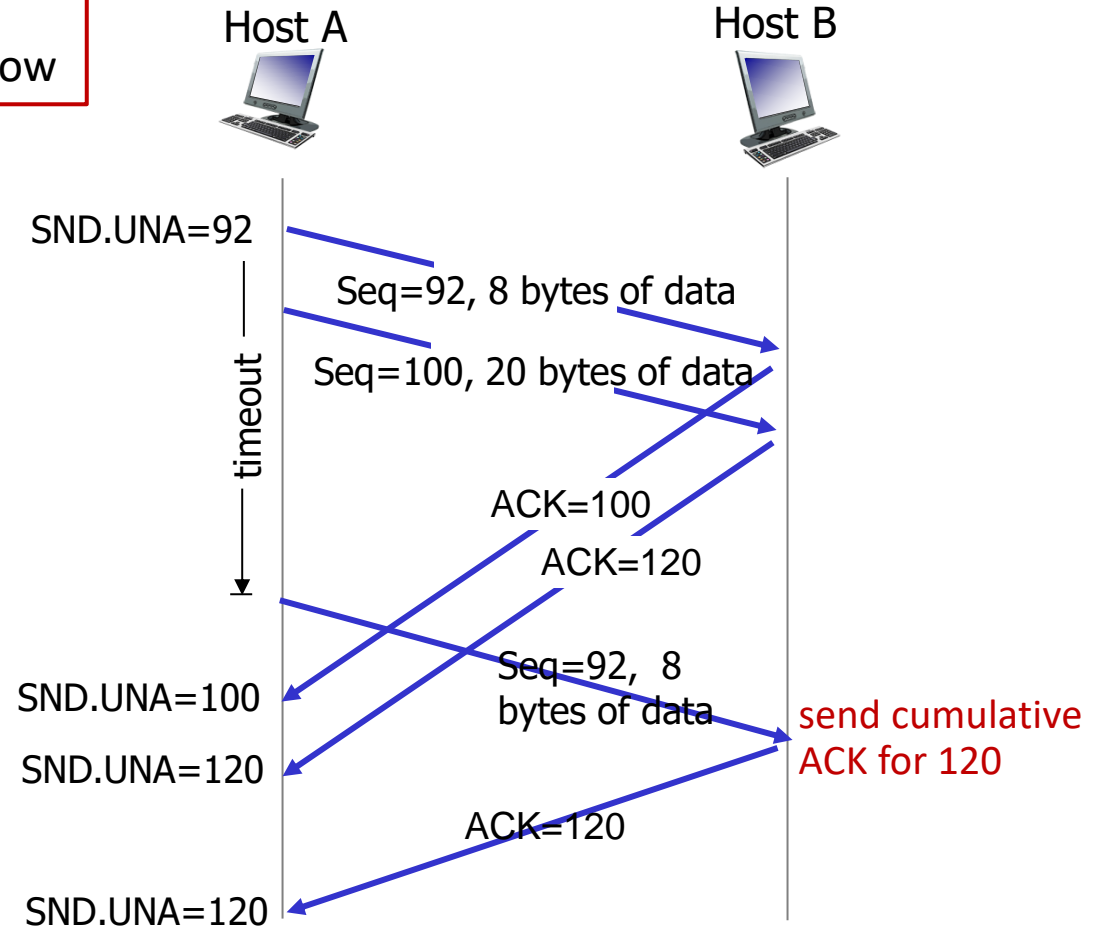
- if ACK acknowledges previously unACKed segments
 - update what is known to be ACKed
 - restart timer if there are still unACKed segments

TCP: retransmission scenarios

First Sent but unacknowledged sequence number
The sequence number at the beginning of the window

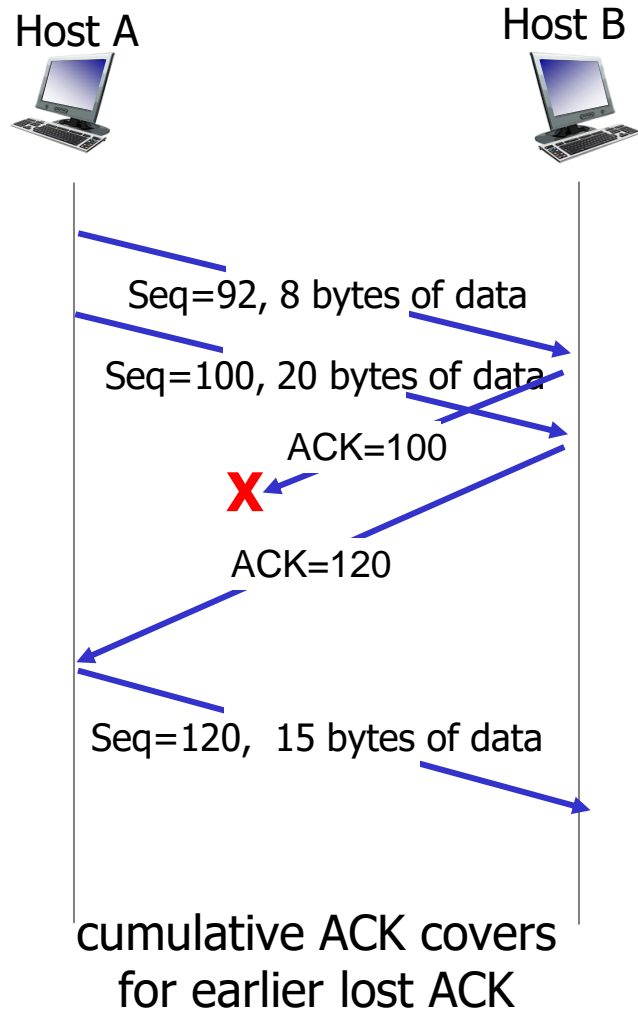


lost ACK scenario

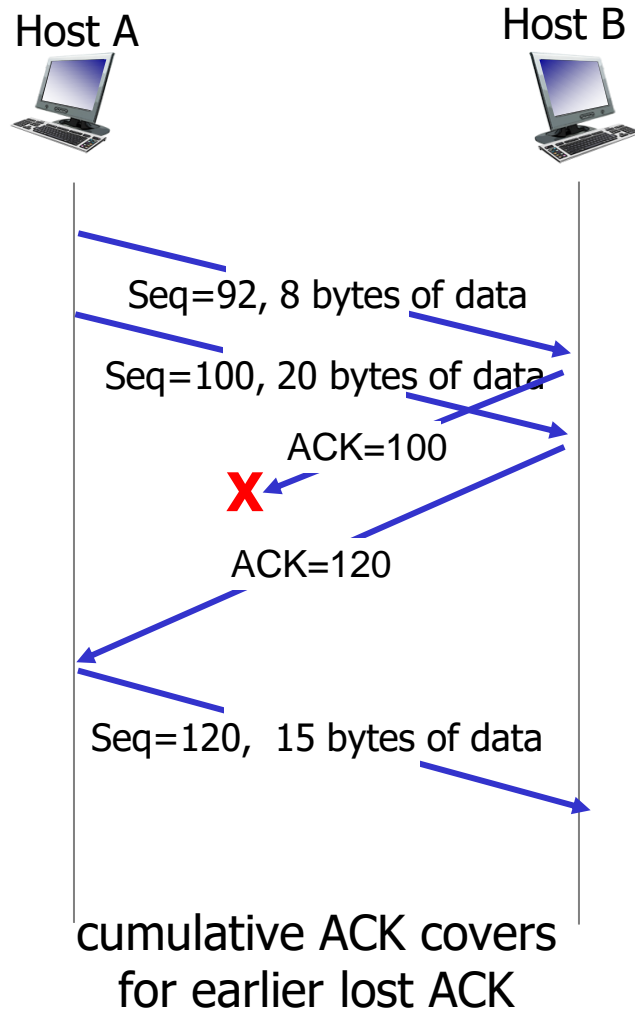


premature timeout

TCP: retransmission scenarios

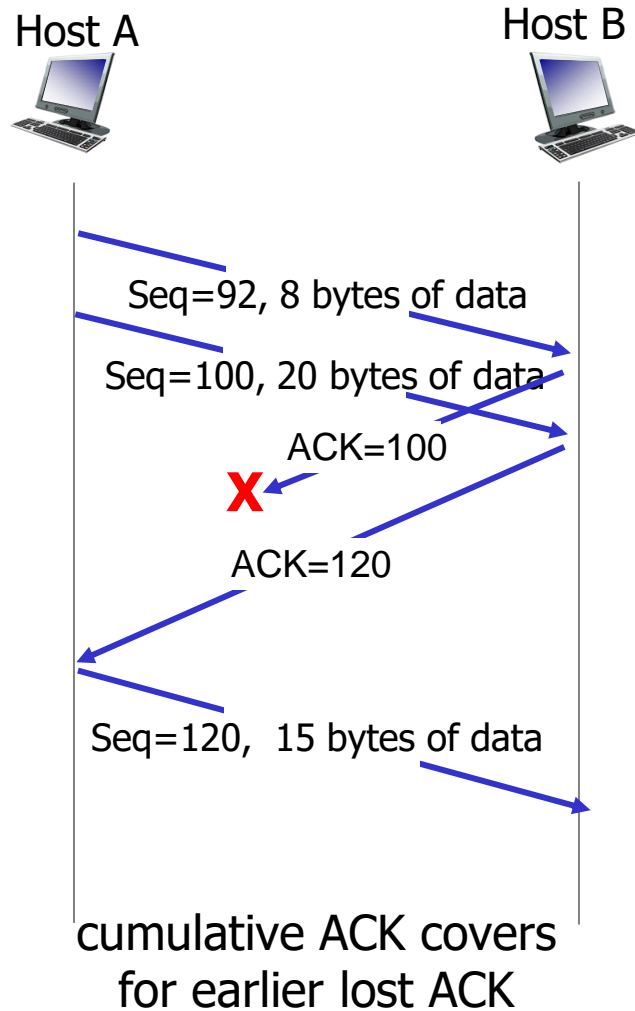


TCP: retransmission scenarios



- (short) in class exercise:
 - What is the value of `SND.UNA` after sending and receiving each packet?

TCP: retransmission scenarios



- Q: How is TCP similar to Go-Back-N? How is it different? How about Selective Repeat?

Knowledge Check

- Make sure you understand and can complete a TCP send and receive timeline.
- This includes, but is not limited to
 - sequence and acknowledgement numbers on packets going back and forth
 - how the sender and receiver view of the sequence number space changes as a result of packets being sent and received (e.g., status of the bytes, position of the sliding window, etc.)

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

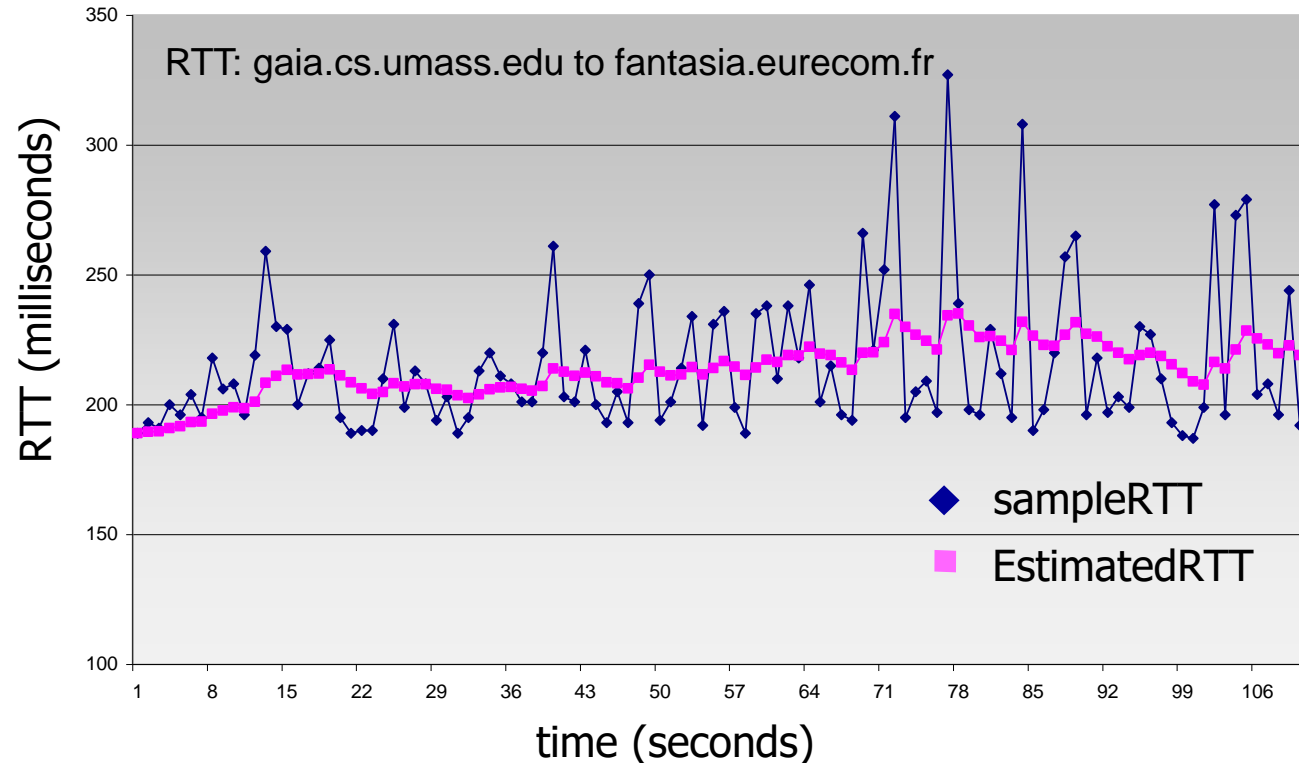
Q: how to estimate RTT?

- `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, $\beta = 0.25$)

Performance optimizations for TCP

- So far, we have covered “the basics” of TCP’s rdt
 - Sequence number
 - Cumulative ACKs
 - Pipelined segments
 - Retransmission timer
- Next, we will discuss some optimizations

Optimization 1: Fast Retransmit

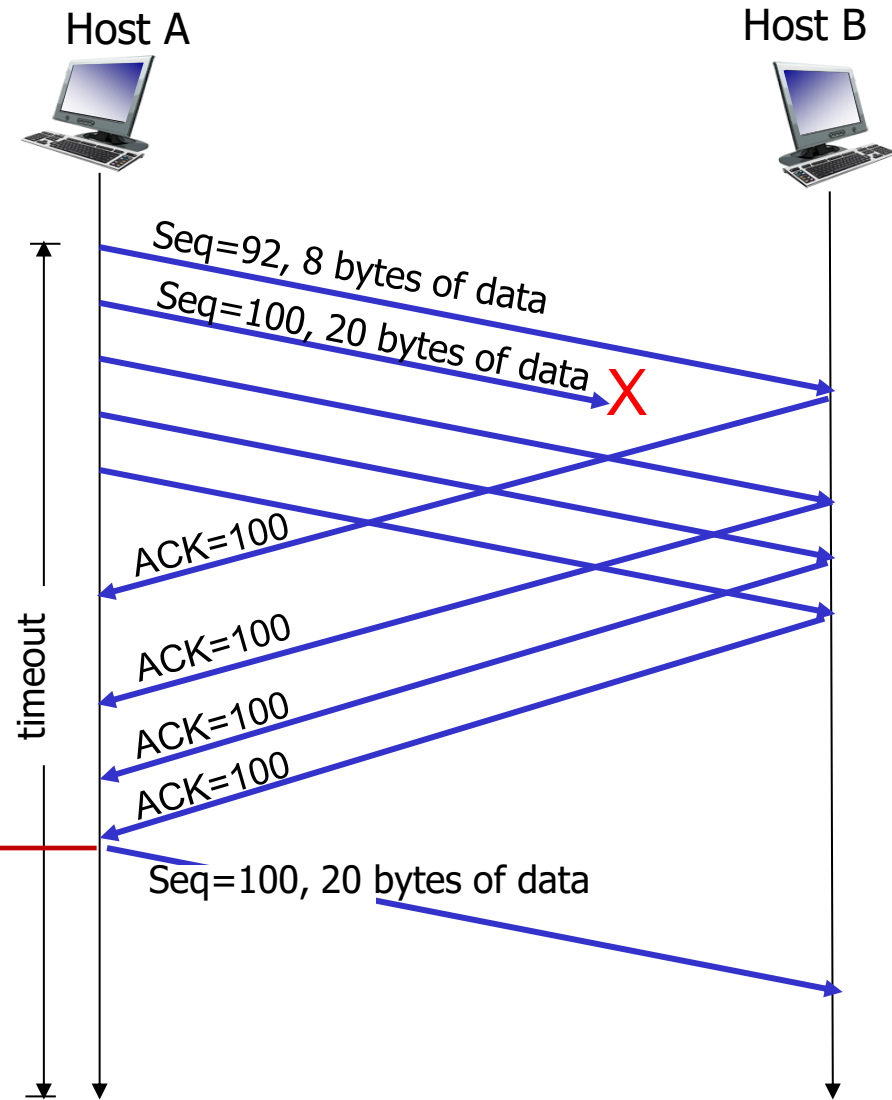
TCP fast retransmit

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!



Optimization 2: Delayed ACKs

- Instead of generating an ACK in response to every segment the moment it arrives
 - Wait for some time to see if there is another segment right afterwards
 - Create one ACK for both.
- Benefits?
 - Saves bandwidth
- Disadvantages?
 - Increases delay in responding to the sender.

Optimizations 2: Delays ACKs (cont.)

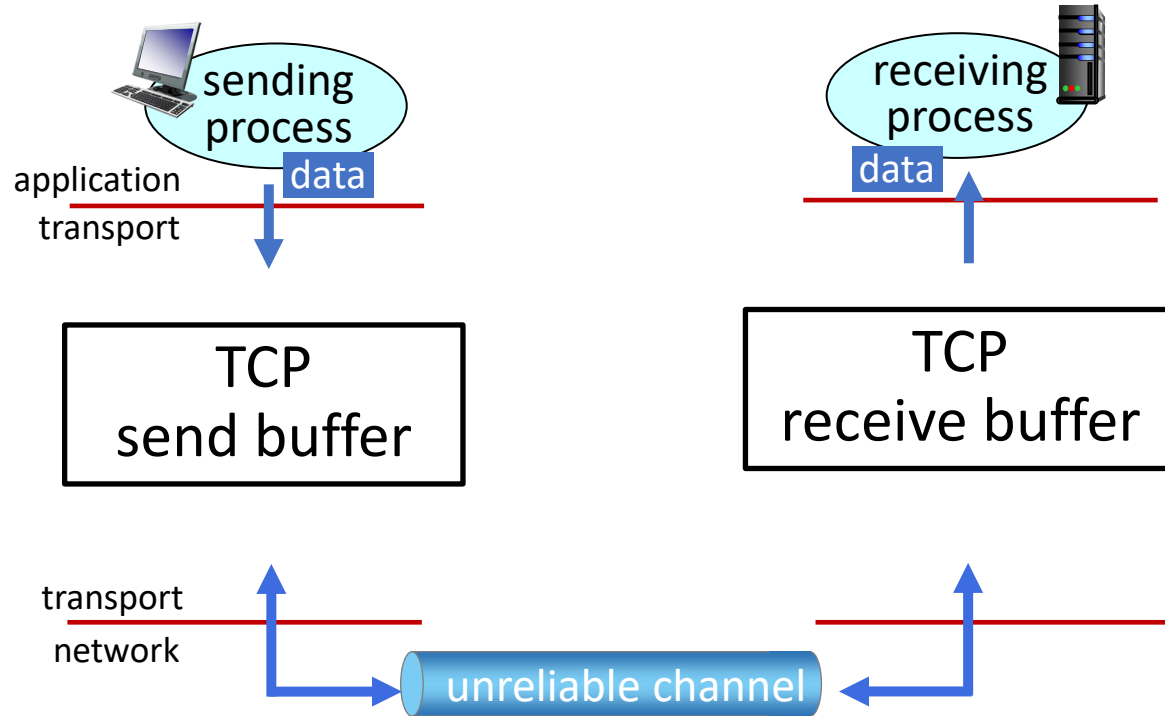
<i>Event at receiver</i>	<i>TCP receiver action</i>
arrival of in-order segment with expected seq #. All data up to expected seq # already ACKed	delayed ACK. Wait up to 500ms for next segment. If no next segment, send ACK
arrival of in-order segment with expected seq #. One other segment has ACK pending	immediately send single cumulative ACK, ACKing both in-order segments
arrival of out-of-order segment higher-than-expect seq. # . Gap detected	immediately send <i>duplicate ACK</i> , indicating seq. # of next expected byte

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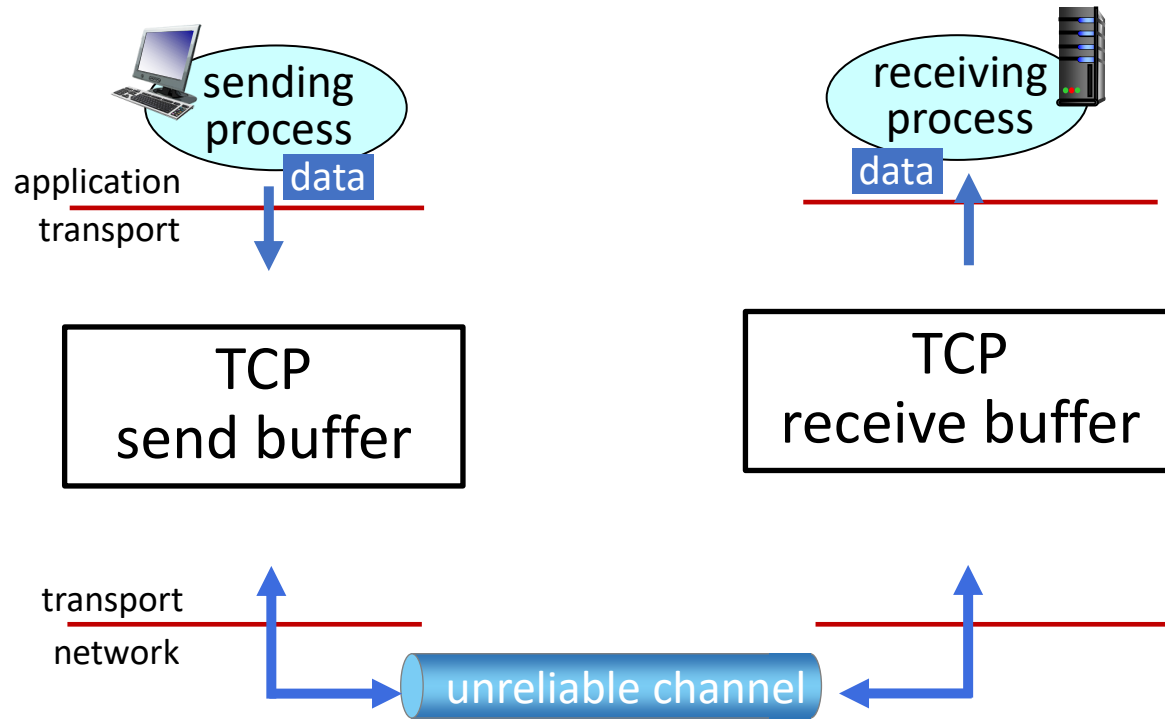


TCP flow control

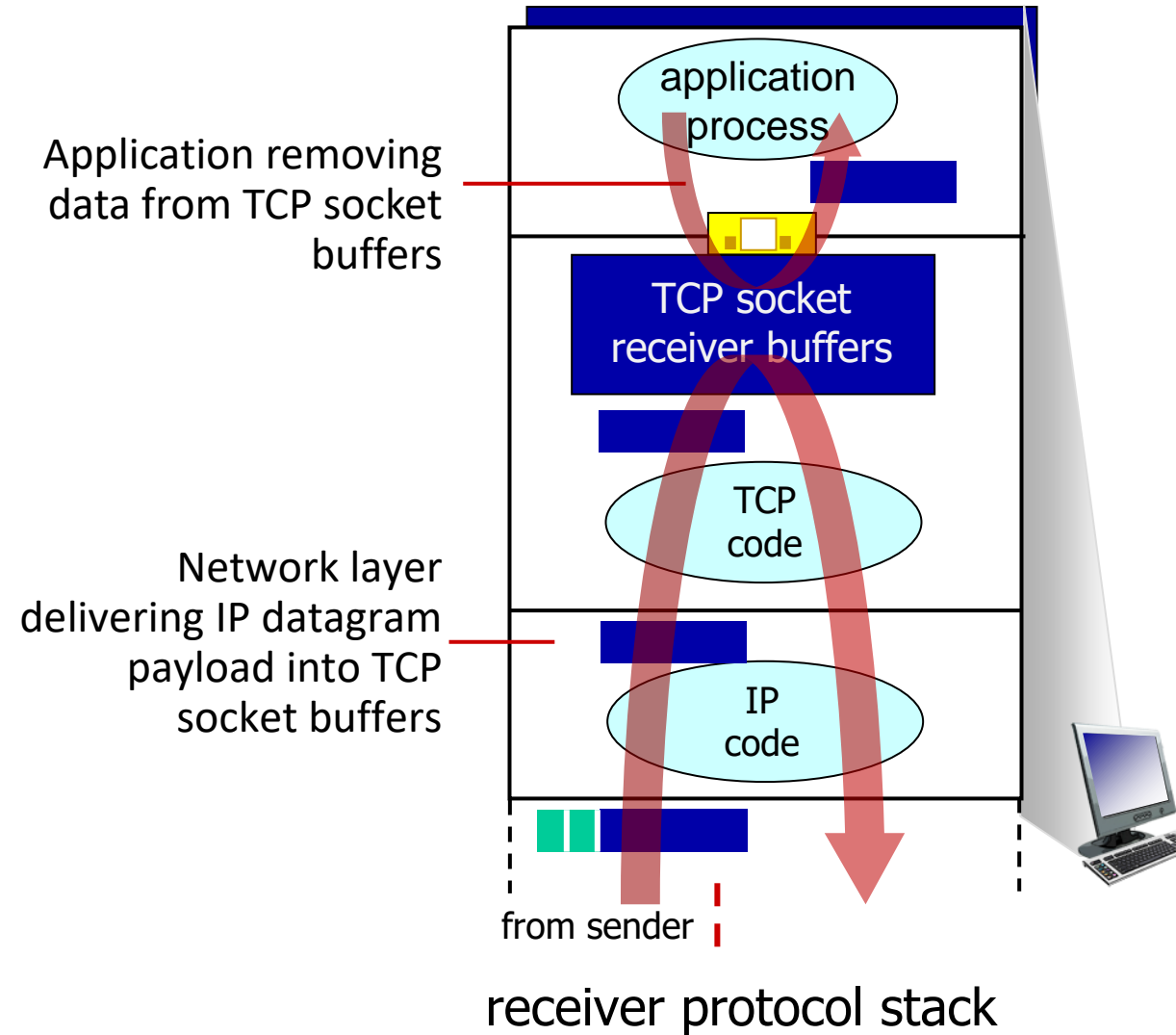


- The send buffer holds the data the application sends to TCP until it is delivered
- The receive buffer holds the data TCP receives from the network until it is delivered to the application

TCP flow control

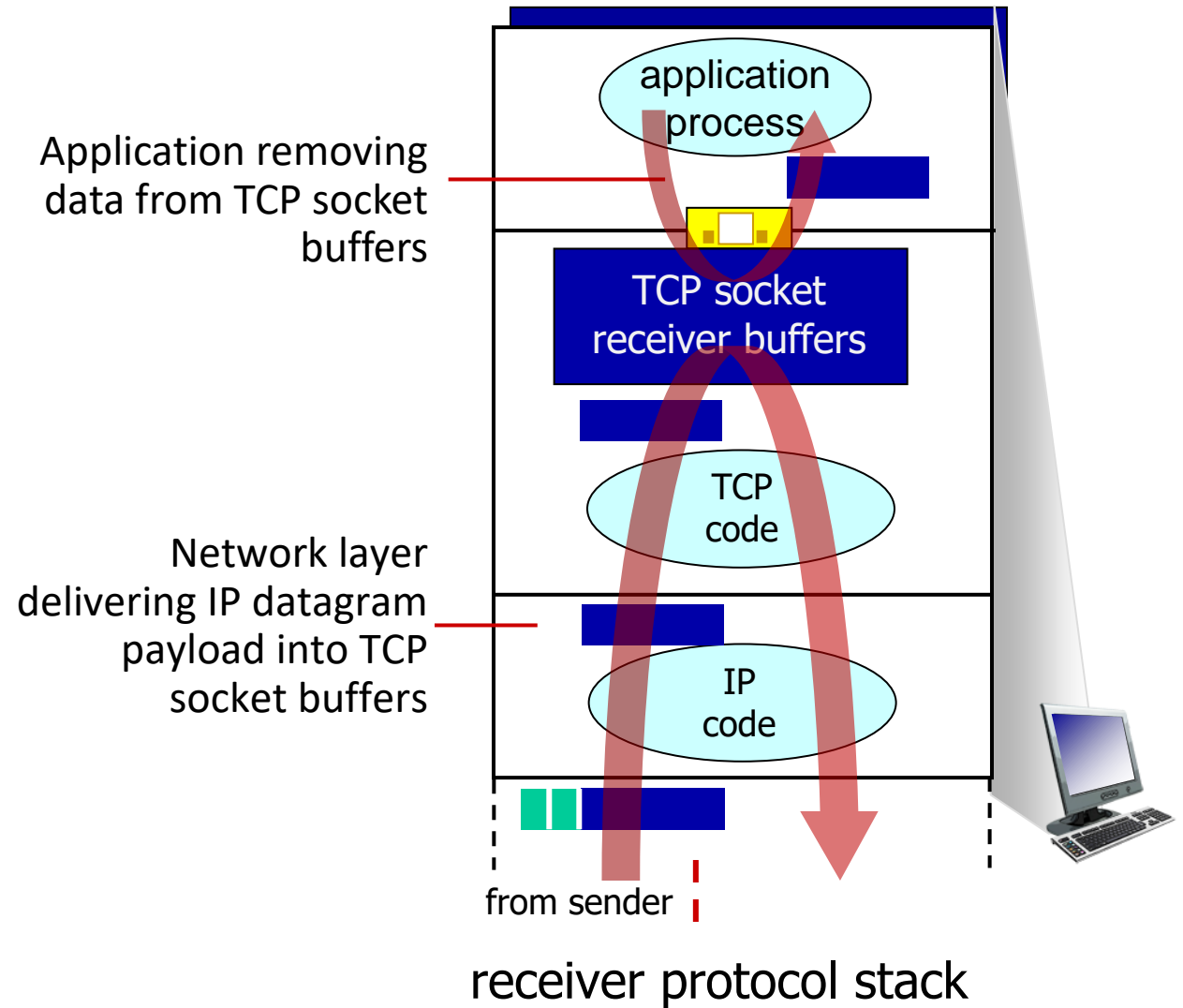


Q: What happens if network layer delivers data faster than application layer removes data from socket buffers?



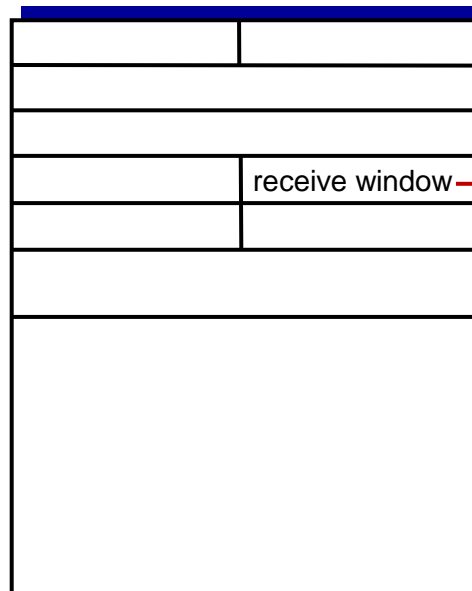
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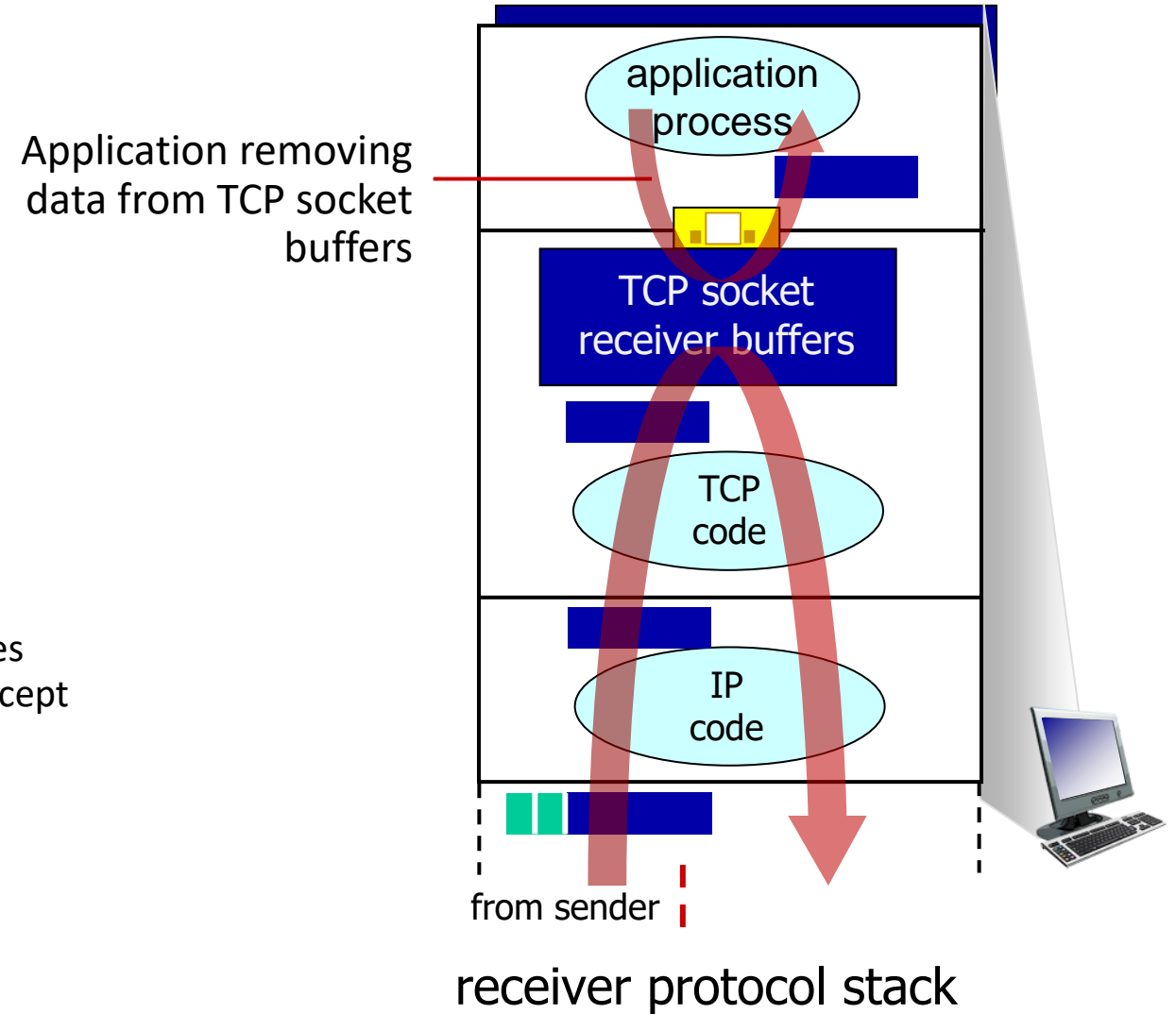


TCP flow control

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flow control: # bytes receiver willing to accept

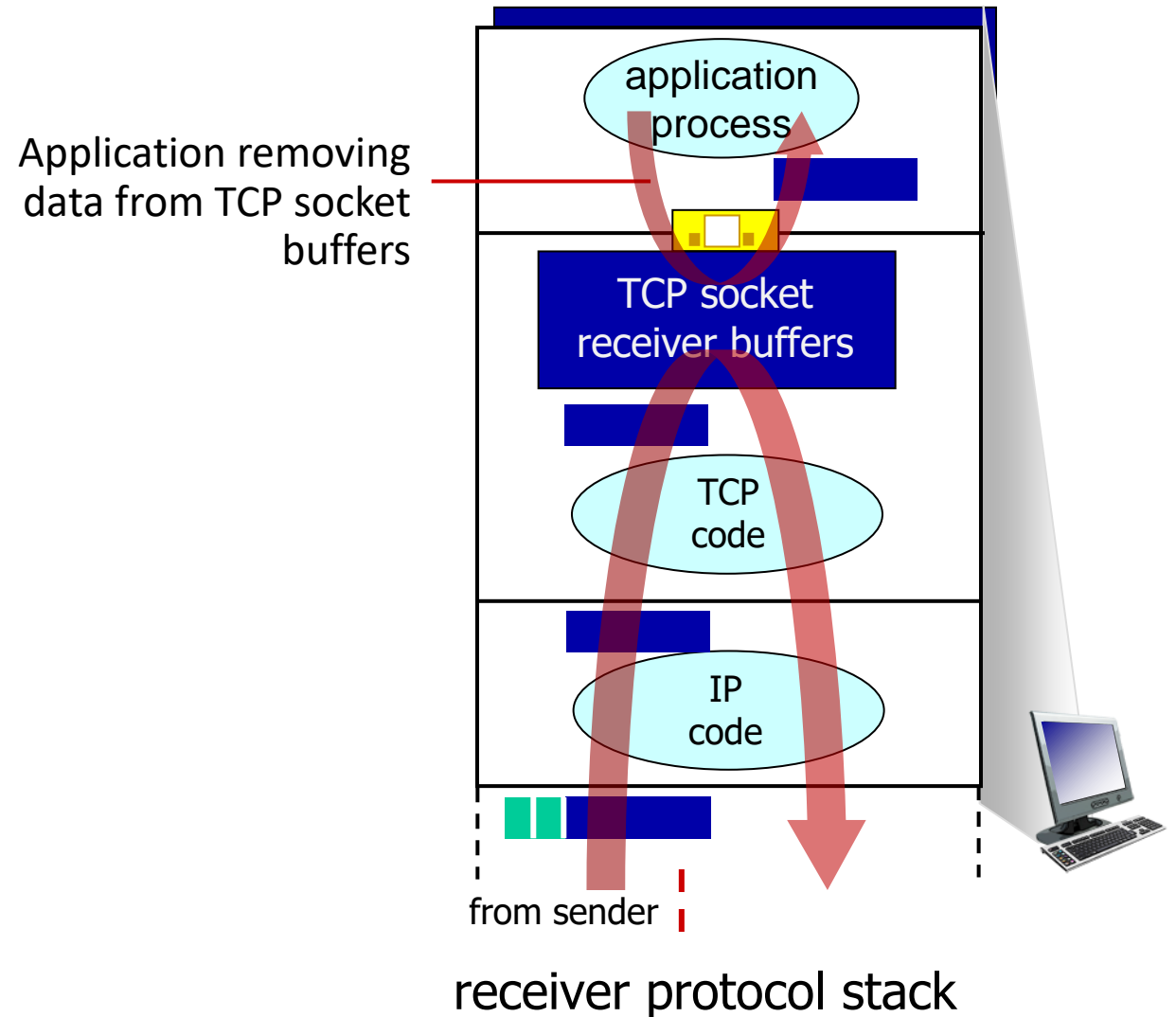


TCP flow control

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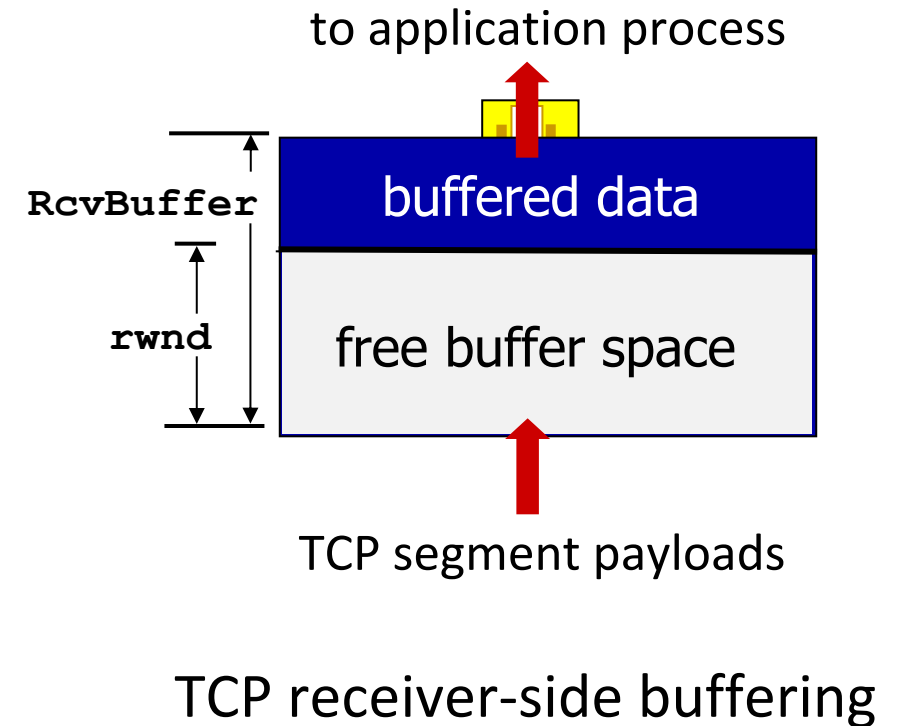
flow control

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



TCP flow control

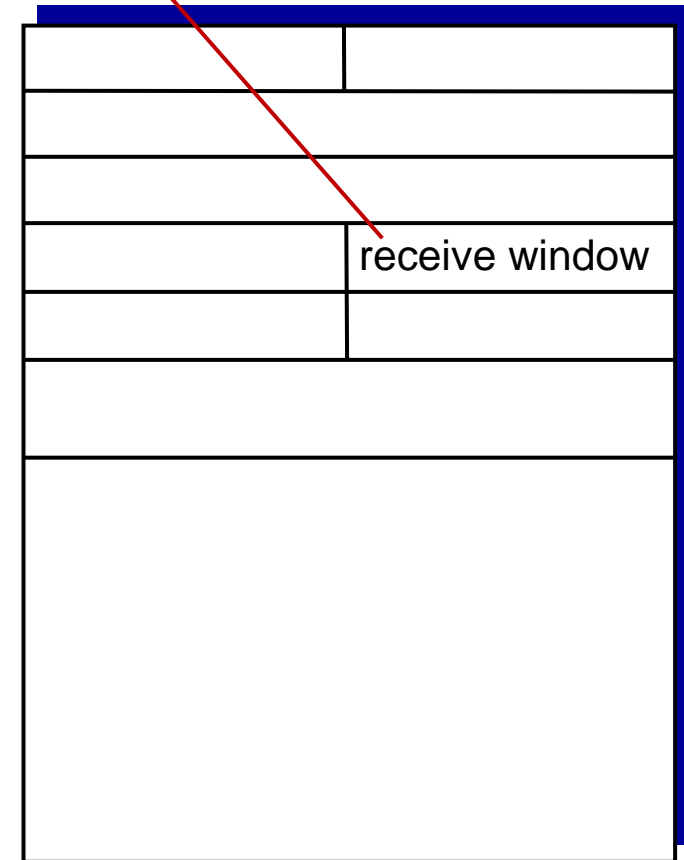
- TCP receiver “advertises” free buffer space in **rwnd** field in TCP header
 - **RcvBuffer** size set via socket options
 - many operating systems auto-adjust **RcvBuffer**
- sender limits amount of unACKed (“in-flight”) data to received **rwnd**
- guarantees receive buffer will not overflow



TCP flow control

- TCP receiver “advertises” free buffer space in **rwnd** field in TCP header
 - **RcvBuffer** size set via socket options
 - many operating systems auto-adjust **RcvBuffer**
- sender limits amount of unACKed (“in-flight”) data to received **rwnd**
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flow control: # bytes receiver willing to accept



TCP segment format

Transport layer: roadmap

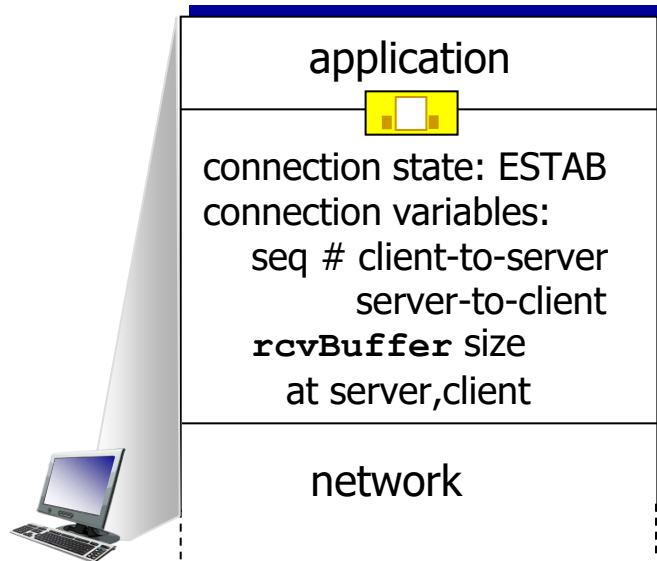
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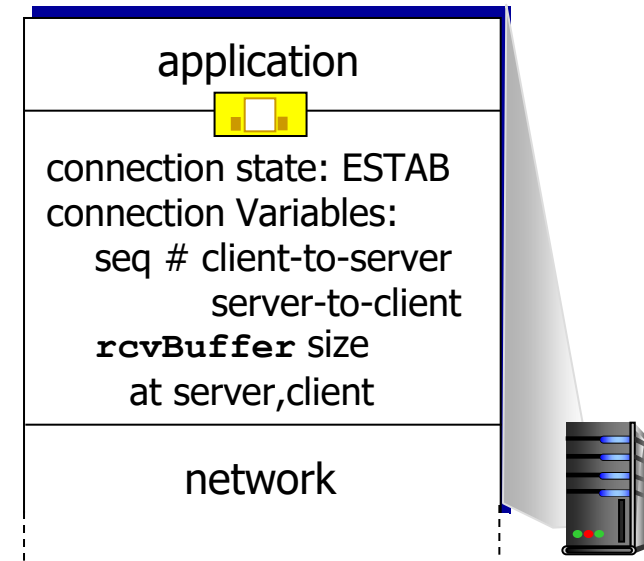
TCP connection management

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

Pay close attention to sequence and ack numbers during 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



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

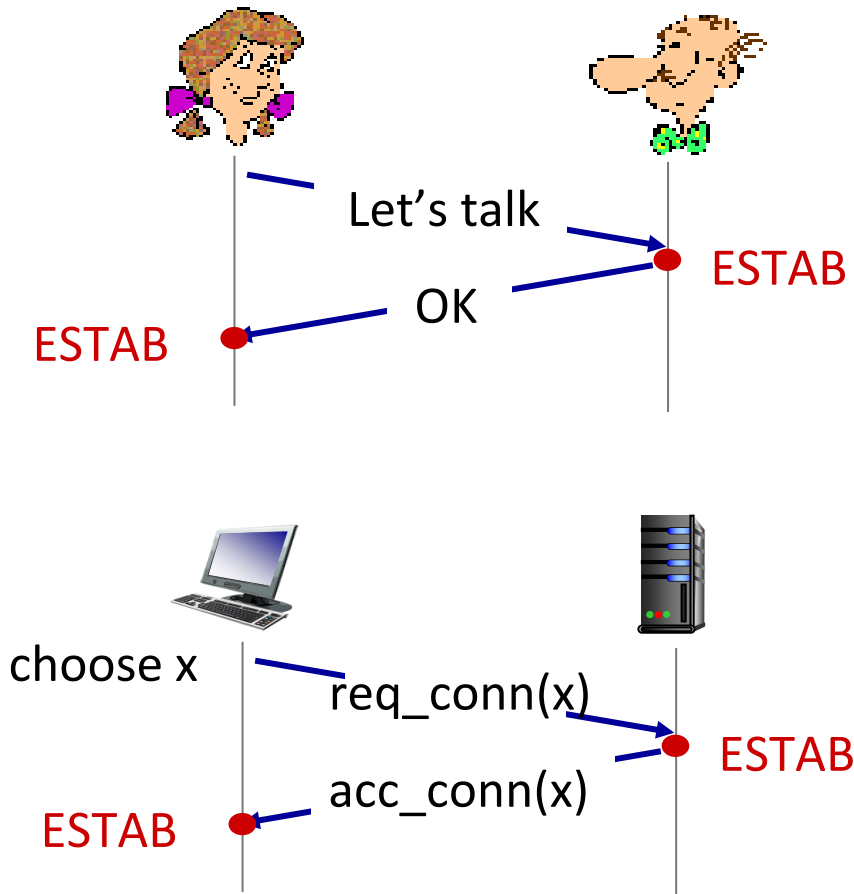
Knowledge Check

- Make sure you understand and can complete a TCP connection timeline
 - From connection establishment, through reliable data transfer (with optimizations and flow control), to connection tear-down
- This includes, but is not limited to
 - sequence and acknowledgement numbers on packets going back and forth
 - how the sender and receiver view of the sequence number space changes as a result of packets being sent and received (e.g., status of the bytes, position of the sliding window, etc.)

Additional Slides

Agreeing to establish a connection

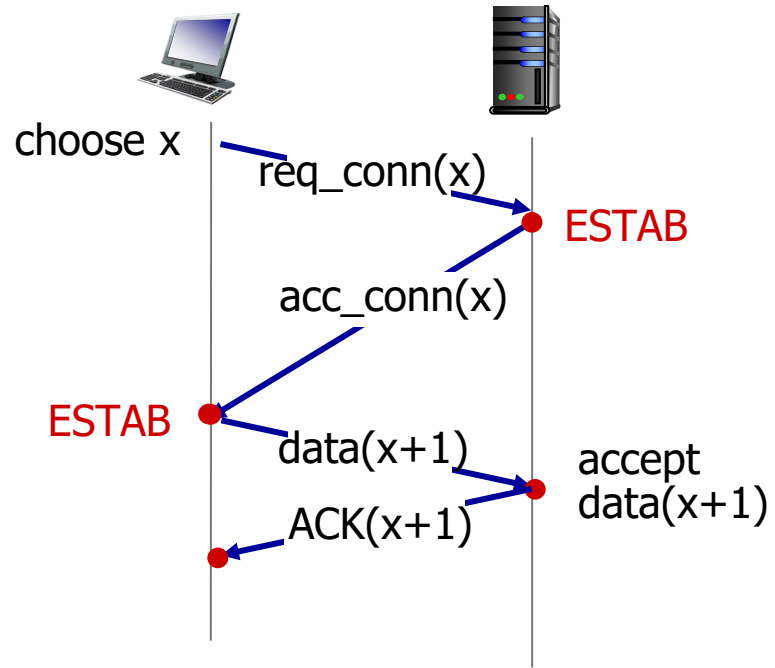
2-way handshake:



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

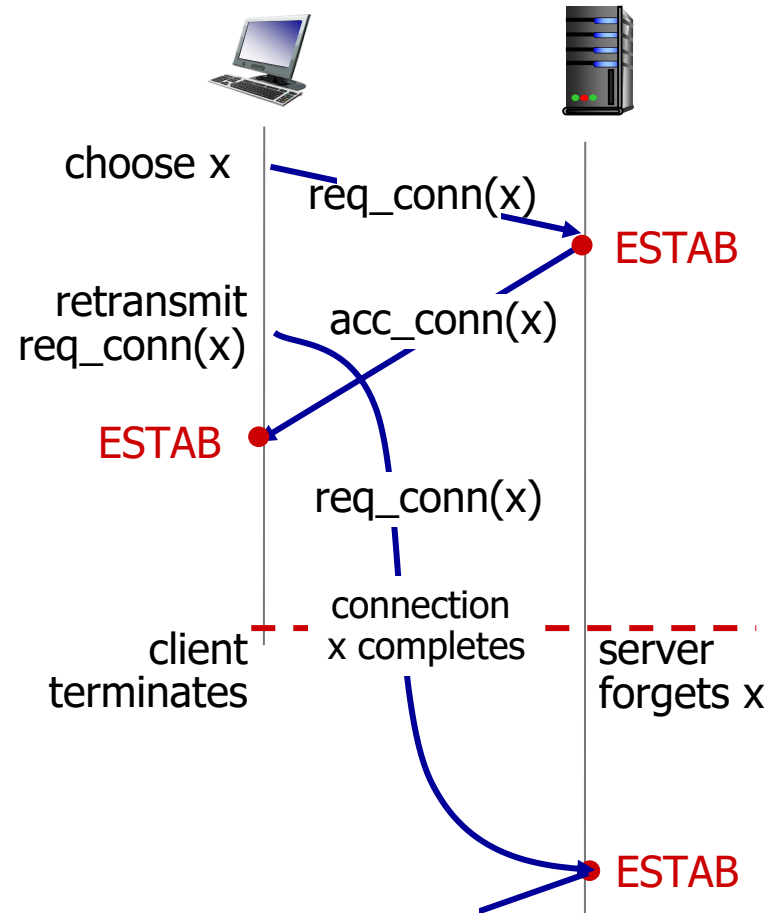
2-way handshake scenarios




No problem!

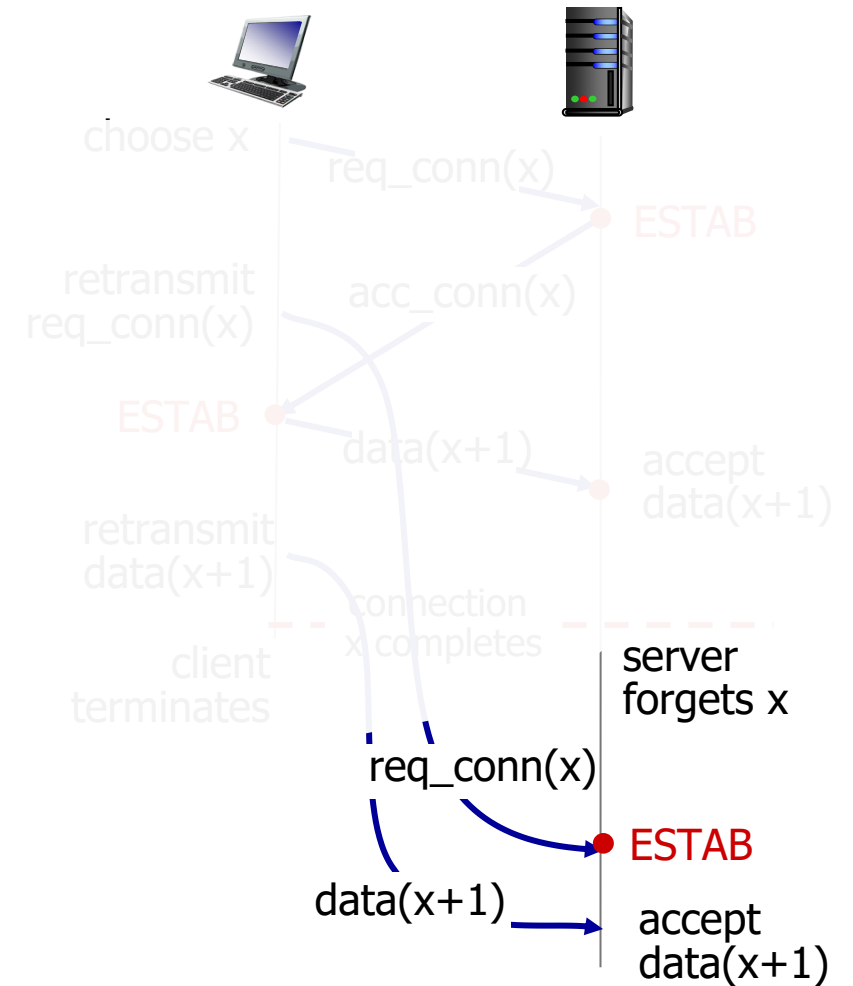


2-way handshake scenarios



 Problem: half open connection! (no client)

2-way handshake scenarios



✘ Problem: dup data accepted!