Chapter 3: roadmap

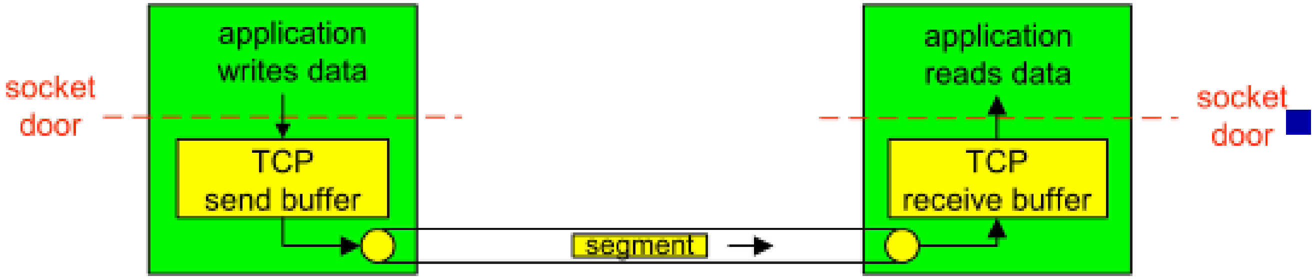
- Transport-layer services
- Multiplexing and demultiplexing
- Connectionless transport: UDP
- Principles of reliable data transfer
- Connection-oriented transport: TCP
  - segment structure
  - reliable data transfer
  - flow control
  - connection management
- Principles of congestion control
- TCP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

TCP: overview RFCs: 793,1122, 2018, 5681, 7323

- point-to-point:
  - one sender, one receiver
- reliable, in-order *byte steam:*
  - 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: overview RFCs: 793,1122, 2018, 5681, 7323



- 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:
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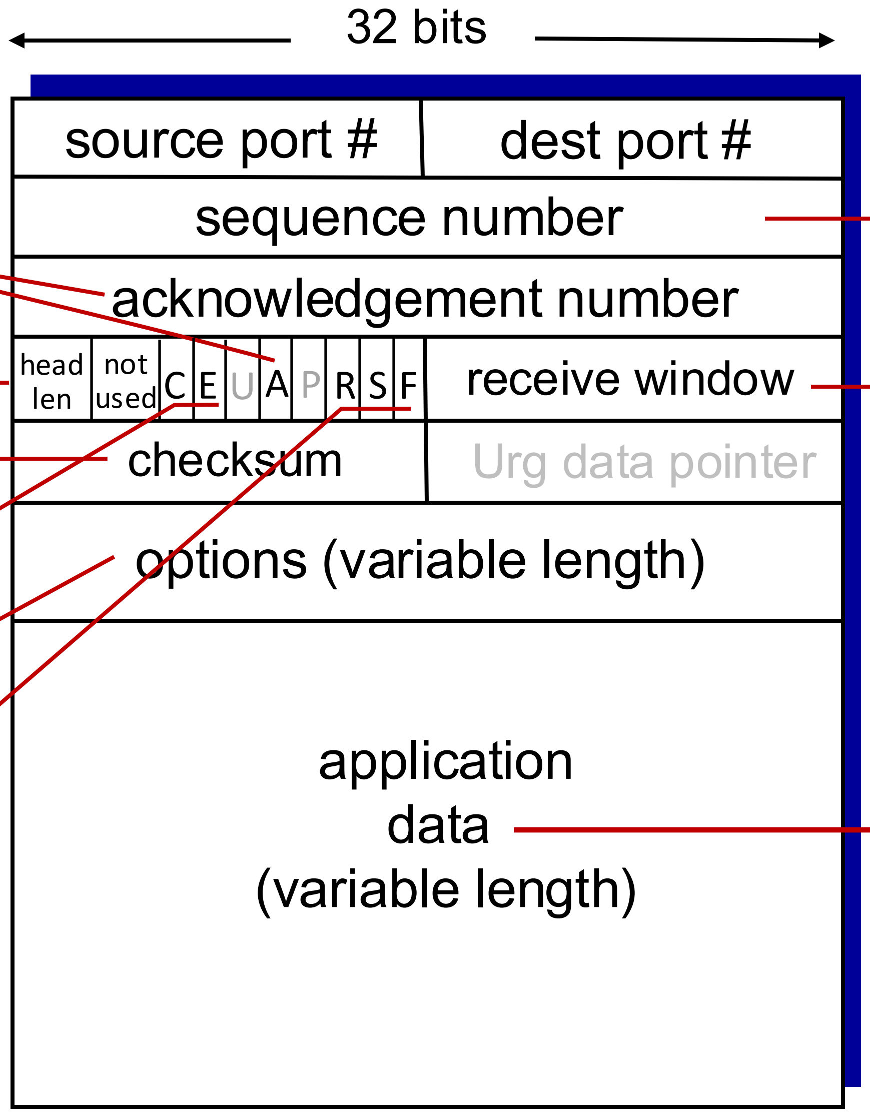
Question

- Given what we have discussed regarding protocols for a variety of lower level guarantees.
- And given that the network layer guarantees…nothing
  - Packets can be dropped
  - Packets can be duplicated
  - Packets can arrive out of order
  - Packets can arrive corrupted

- What do we need in the TCP header?

TCP segment structure



32 bits

source port # | dest port #

sequence number

acknowledgement number

head len | not used | C | E | U | A | P | R | S | F | receive window

checksum | Urg data pointer

options (variable length)

application data (variable length)

ACK: seq # of next expected byte; A bit: this is an ACK

length (of TCP header)

Internet checksum

C, E: congestion notification

TCP options

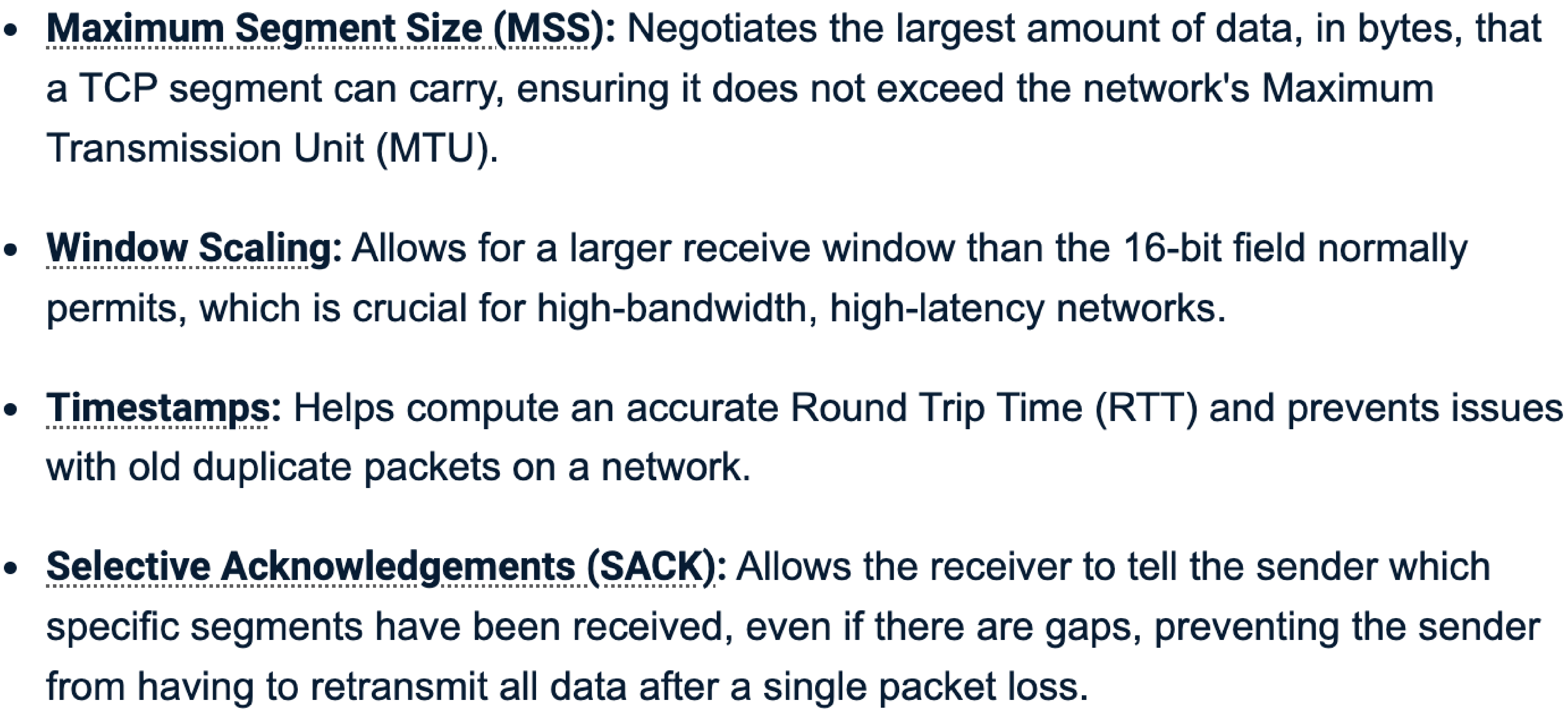
RST, SYN, FIN: connection management

segment seq #: counting bytes of data into bytestream (not segments!)

flow control: # bytes receiver willing to accept

data sent by application into TCP socket

TCP Options Examples



- **Maximum Segment Size (MSS):** Negotiates the largest amount of data, in bytes, that a TCP segment can carry, ensuring it does not exceed the network's Maximum Transmission Unit (MTU).

- **Window Scaling:** Allows for a larger receive window than the 16-bit field normally permits, which is crucial for high-bandwidth, high-latency networks.

- **Timestamps:** Helps compute an accurate Round Trip Time (RTT) and prevents issues with old duplicate packets on a network.

- **Selective Acknowledgements (SACK):** Allows the receiver to tell the sender which specific segments have been received, even if there are gaps, preventing the sender from having to retransmit all data after a single packet loss.

TCP sequence numbers, ACKs

Sequence numbers:

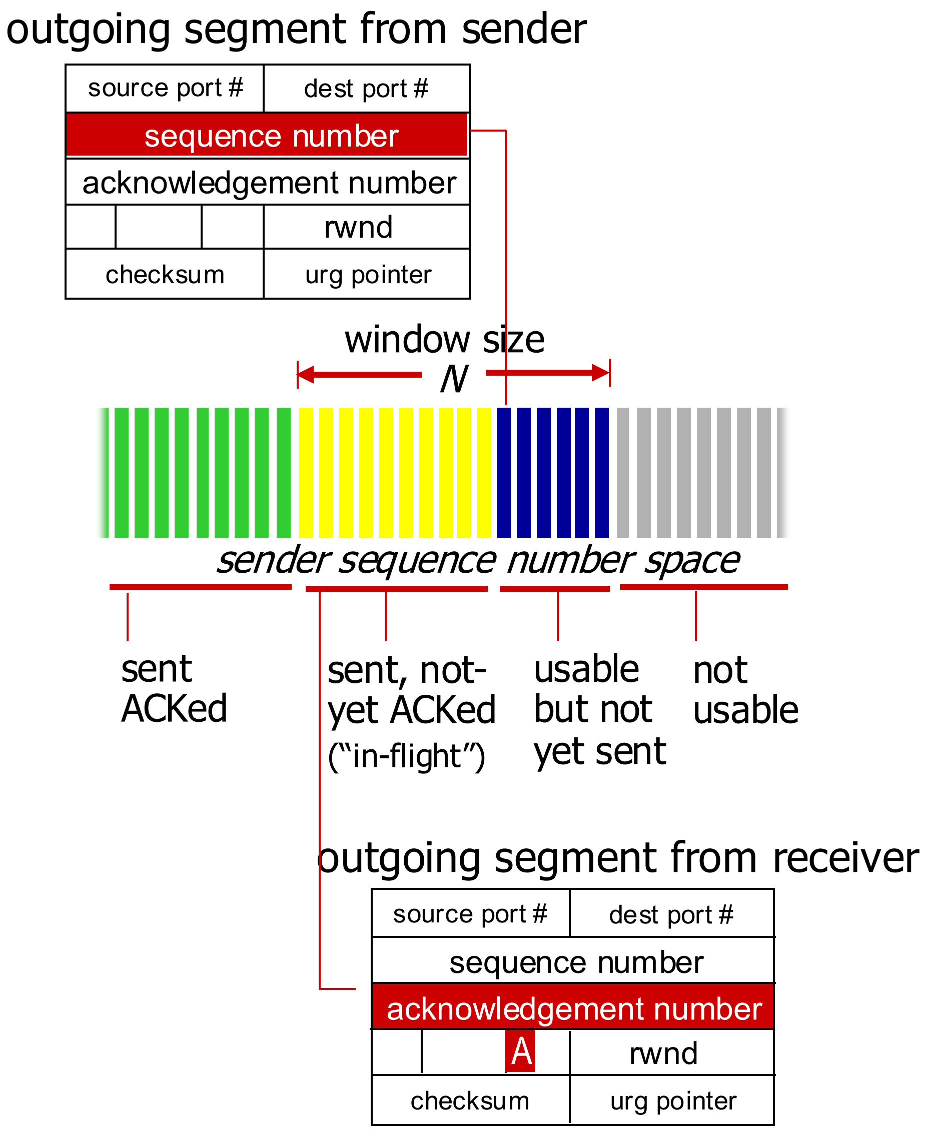
- byte stream "number" of first byte in segment's data

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

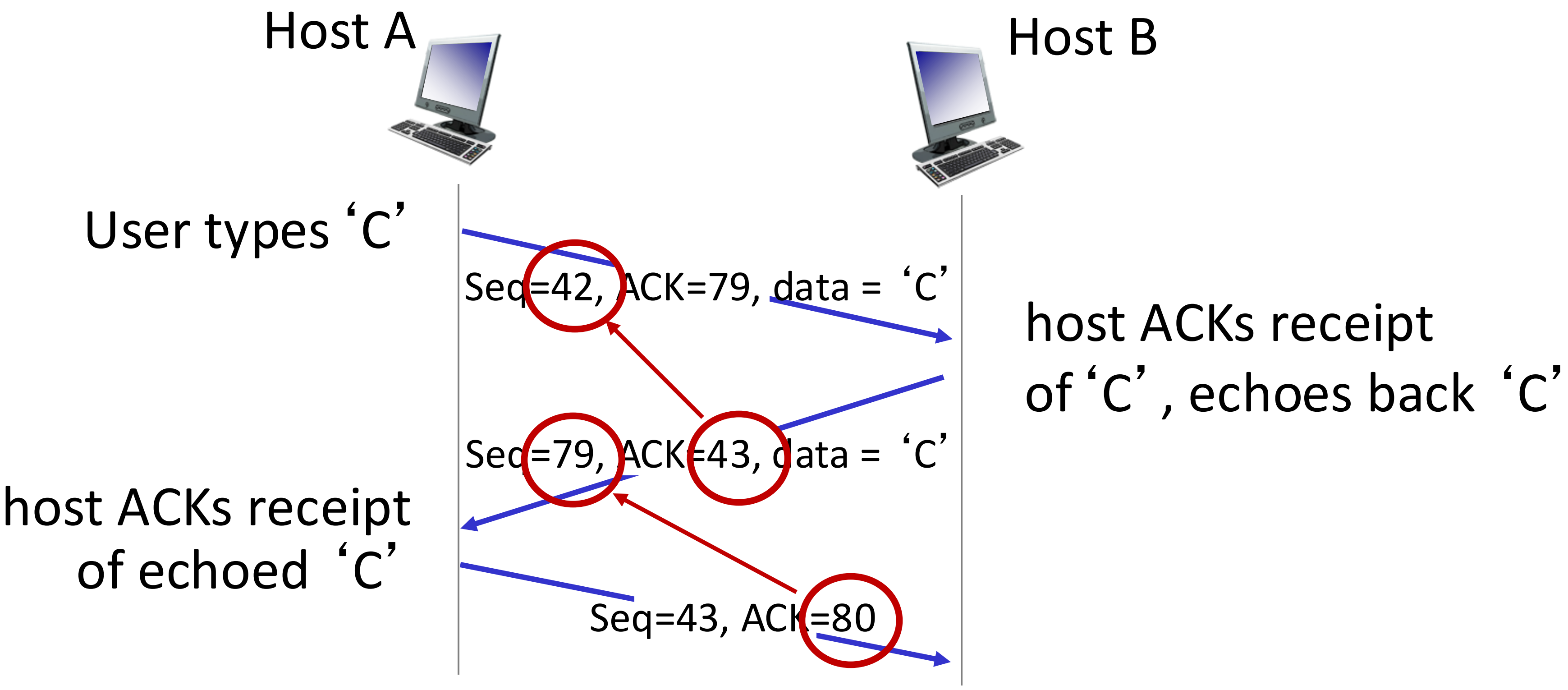
source port #	dest port #
sequence number	
acknowledgement number	
	rwnd
checksum	urg pointer

window size
N
sender sequence number space
sent ACKed
sent, not-yet ACKed ("in-flight")
usable but not yet sent
not usable

outgoing segment from receiver

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

TCP sequence numbers, ACKs


Host A
Host B
User types ‘C’
Seq=42, ACK=79, data = ‘C’
host ACKs receipt of ‘C’, echoes back ‘C’
Seq=79, ACK=43, data = ‘C’
host ACKs receipt of echoed ‘C’
Seq=43, ACK=80

simple telnet scenario

Another Question

- We know that when packets can be dropped, we need to set a timer.
- The question is, to what value?
- Think about that a bit.
  - And about what difficulties we might encounter
  - And think about the consequences if we get this wrong

TCP round trip time, timeout

Q: how to set TCP timeout value?

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

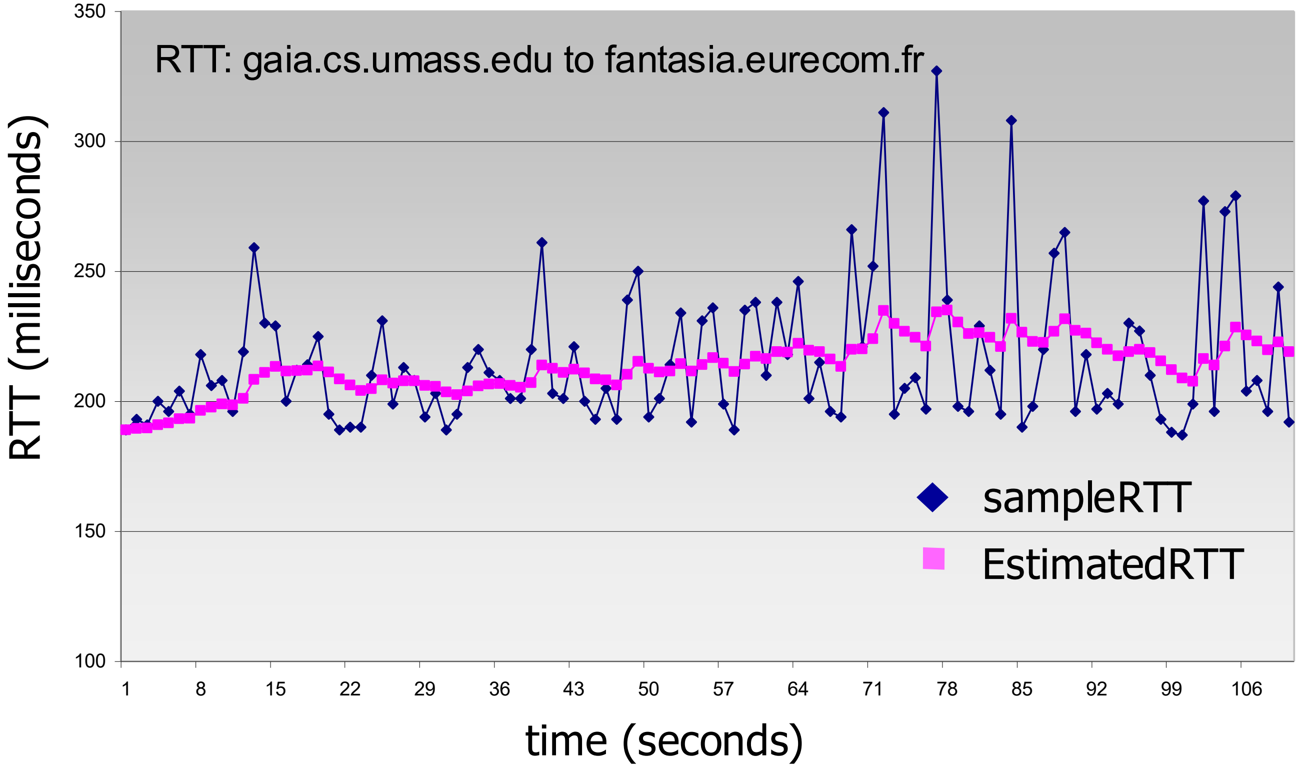
Q: how to estimate RTT?

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

TCP round trip time, timeout

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

- exponential weighted moving average (EWMA)
- influence of past sample decreases exponentially fast
- typical value: α = 0.125



TCP round trip time, timeout

- timeout interval: **`EstimatedRTT`** plus “safety margin”
  - large variation in **`EstimatedRTT`**: want a larger safety margin

$$\texttt{TimeoutInterval = } \underset{\text{estimated RTT}}{\underset{\uparrow}{\texttt{EstimatedRTT}}} \texttt{ + 4*}\underset{\text{“safety margin”}}{\underset{\uparrow}{\texttt{DevRTT}}}$$

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

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

(typically, β = 0.25)

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

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

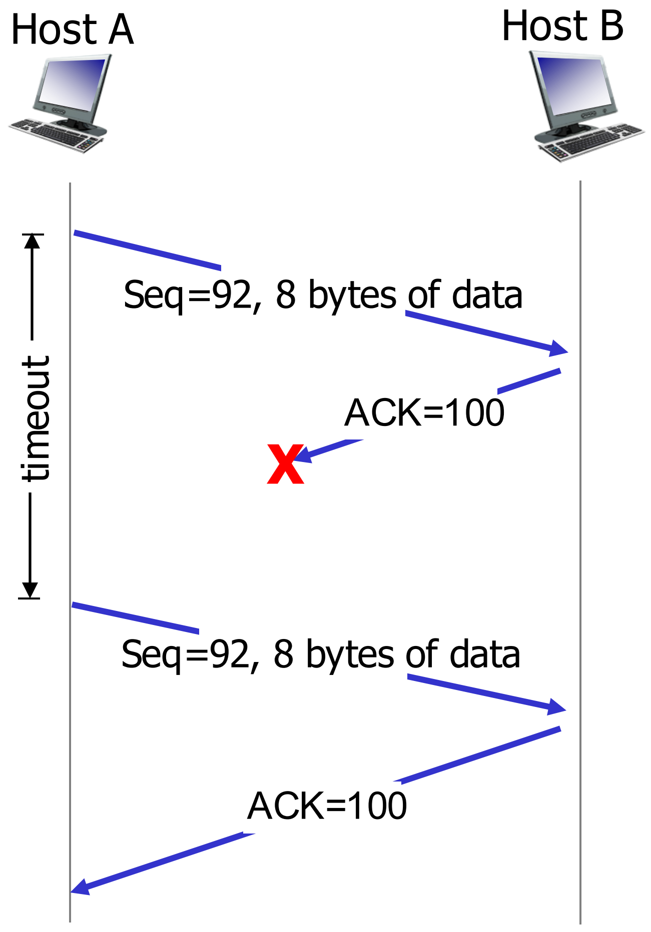
event: ACK received

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

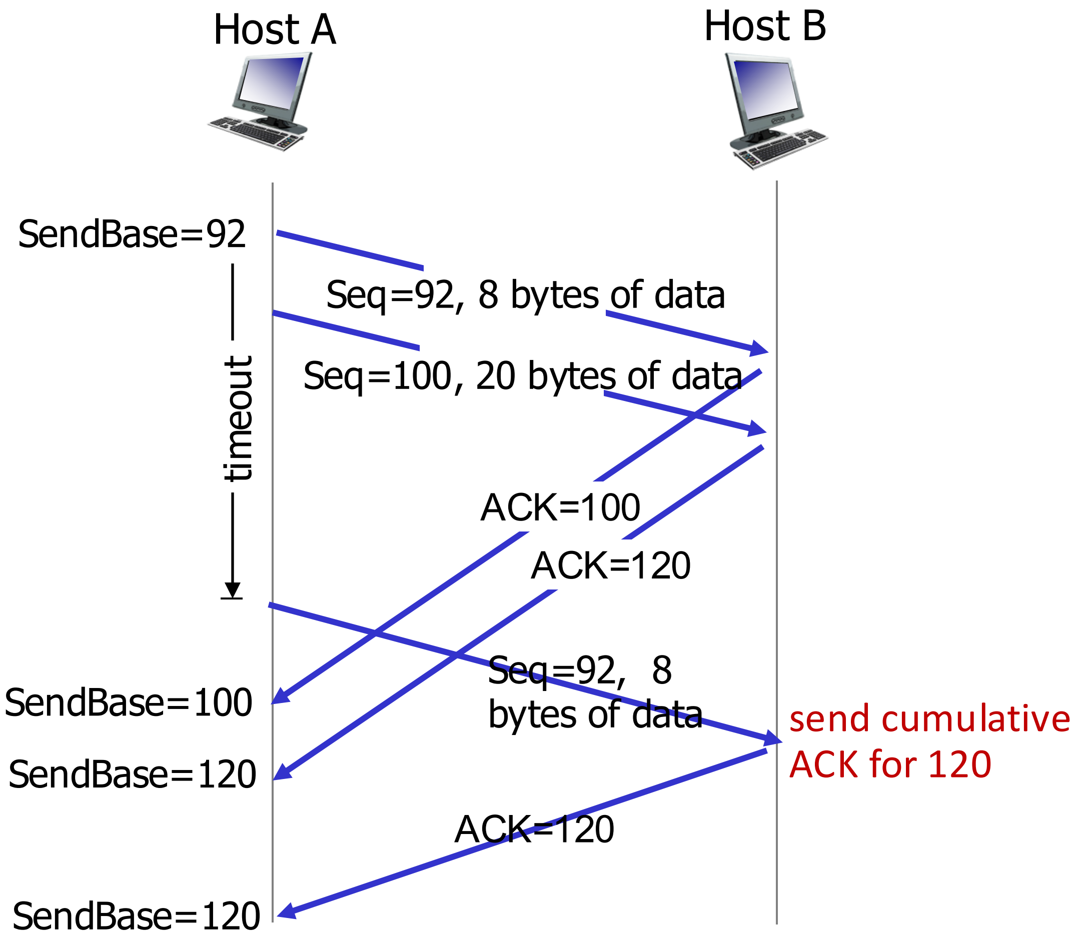
TCP Receiver: ACK generation [RFC 5681]

Event at receiver	*TCP receiver action*

TCP: retransmission scenarios



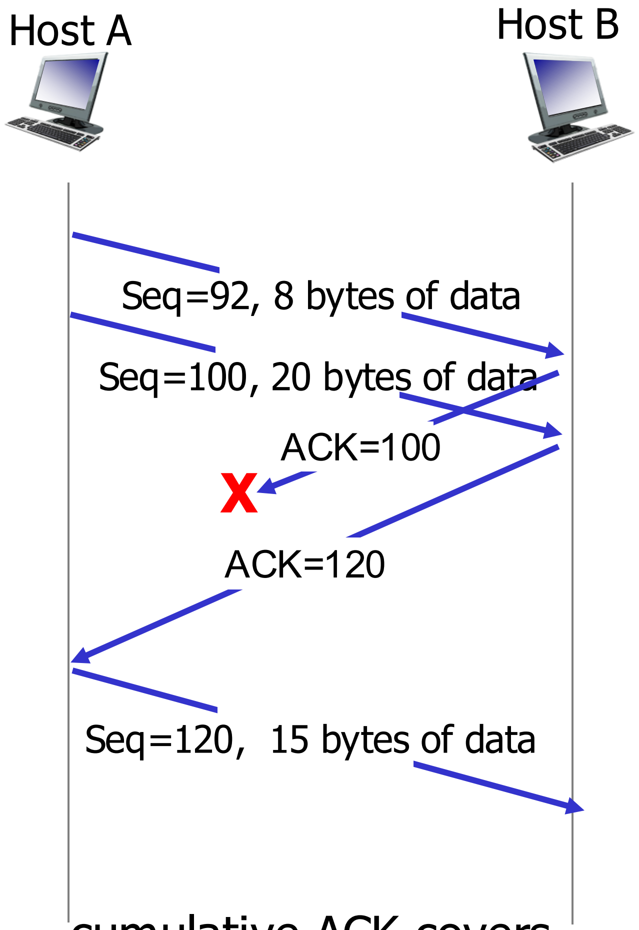
lost ACK scenario



send cumulative ACK for 120

premature timeout

TCP: retransmission scenarios



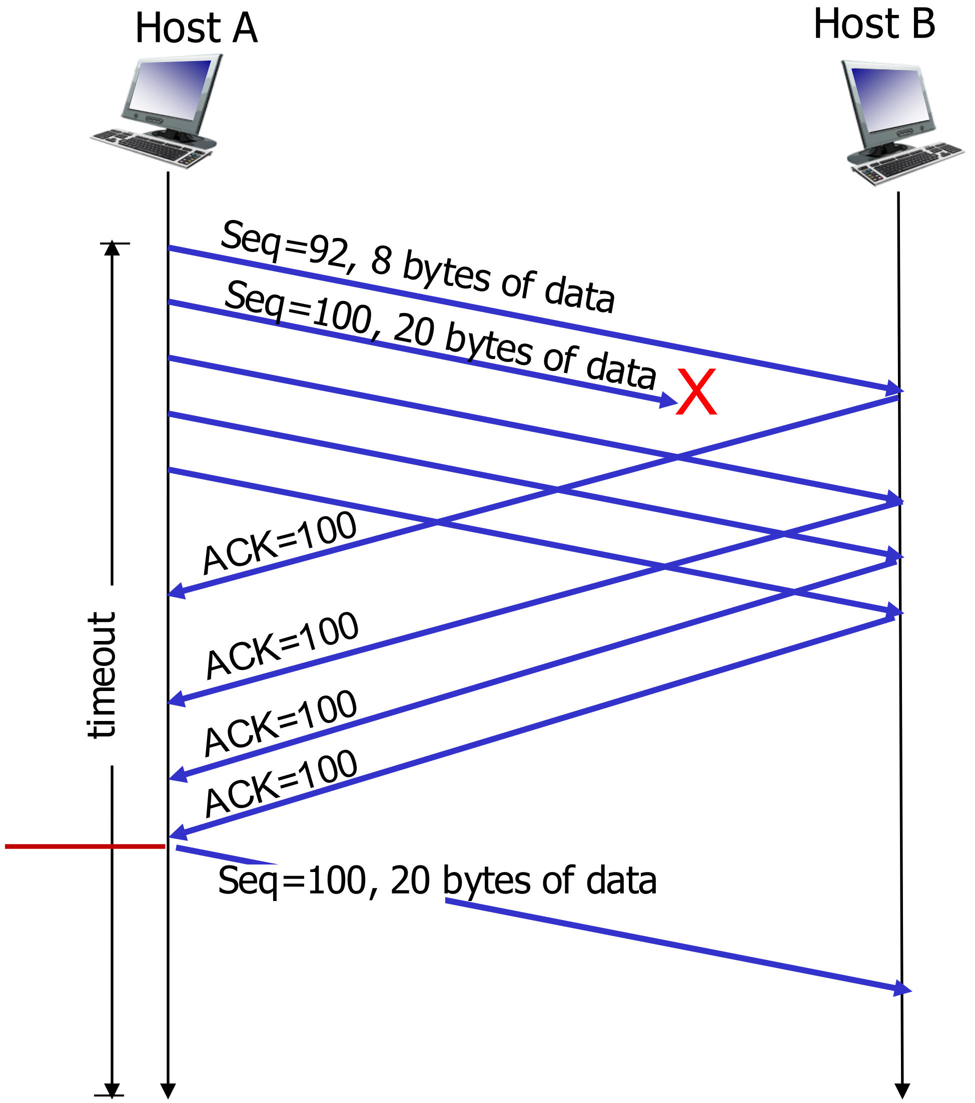
cumulative ACK covers
for earlier lost ACK

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!

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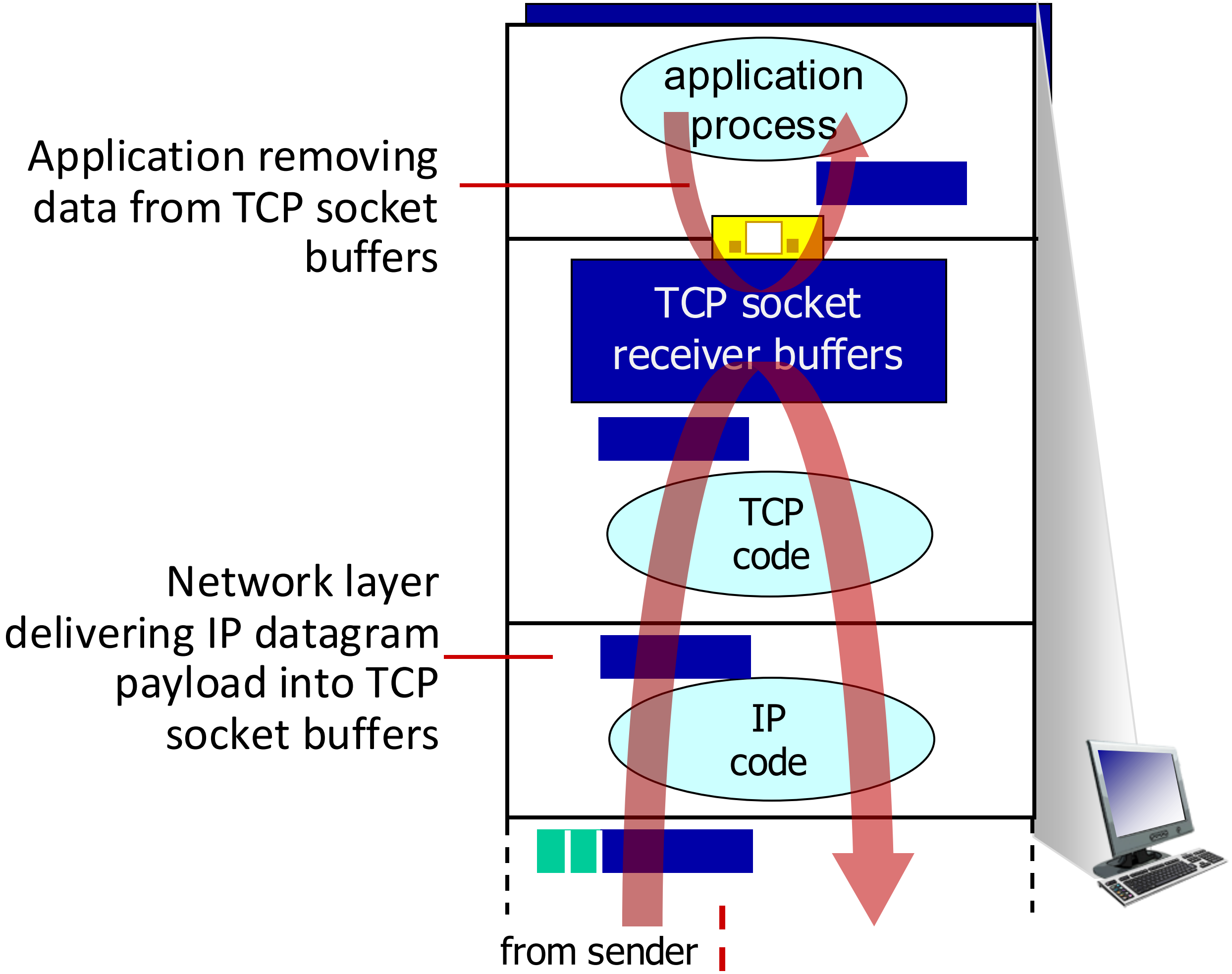
TCP flow control

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

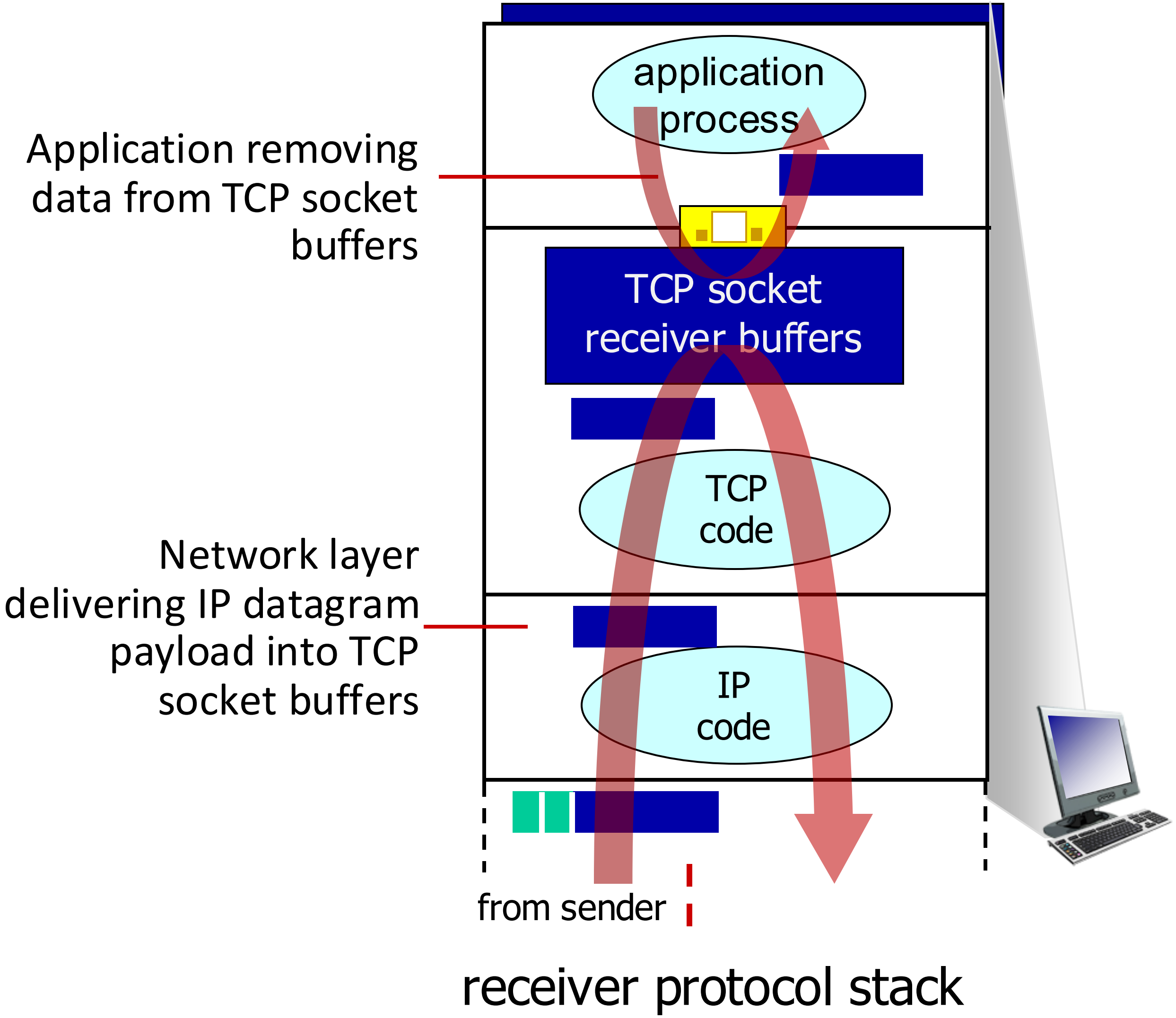


application process
Application removing data from TCP socket buffers
TCP socket receiver buffers
TCP code
Network layer delivering IP datagram payload into TCP socket buffers
IP code
from sender

receiver protocol stack

TCP flow control

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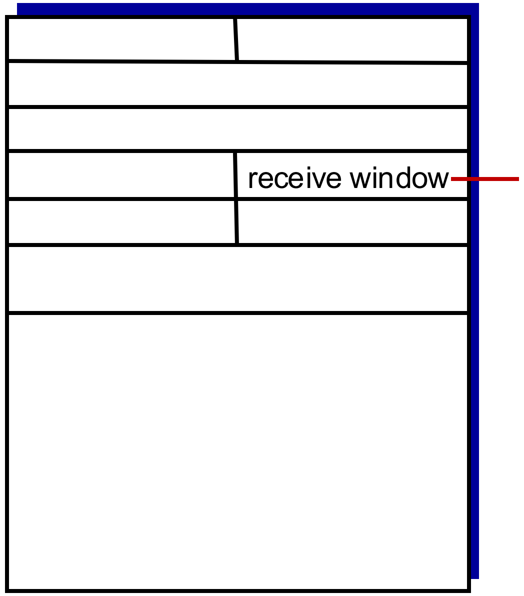
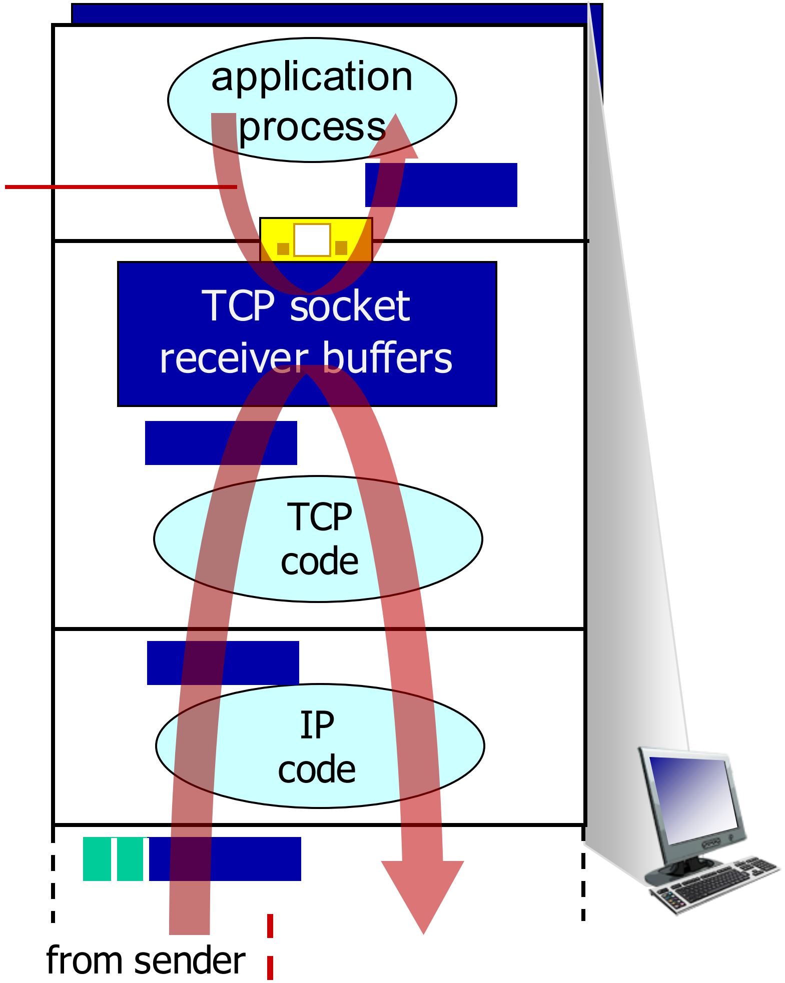


flow control: # bytes receiver willing to accept

Application removing data from TCP socket buffers



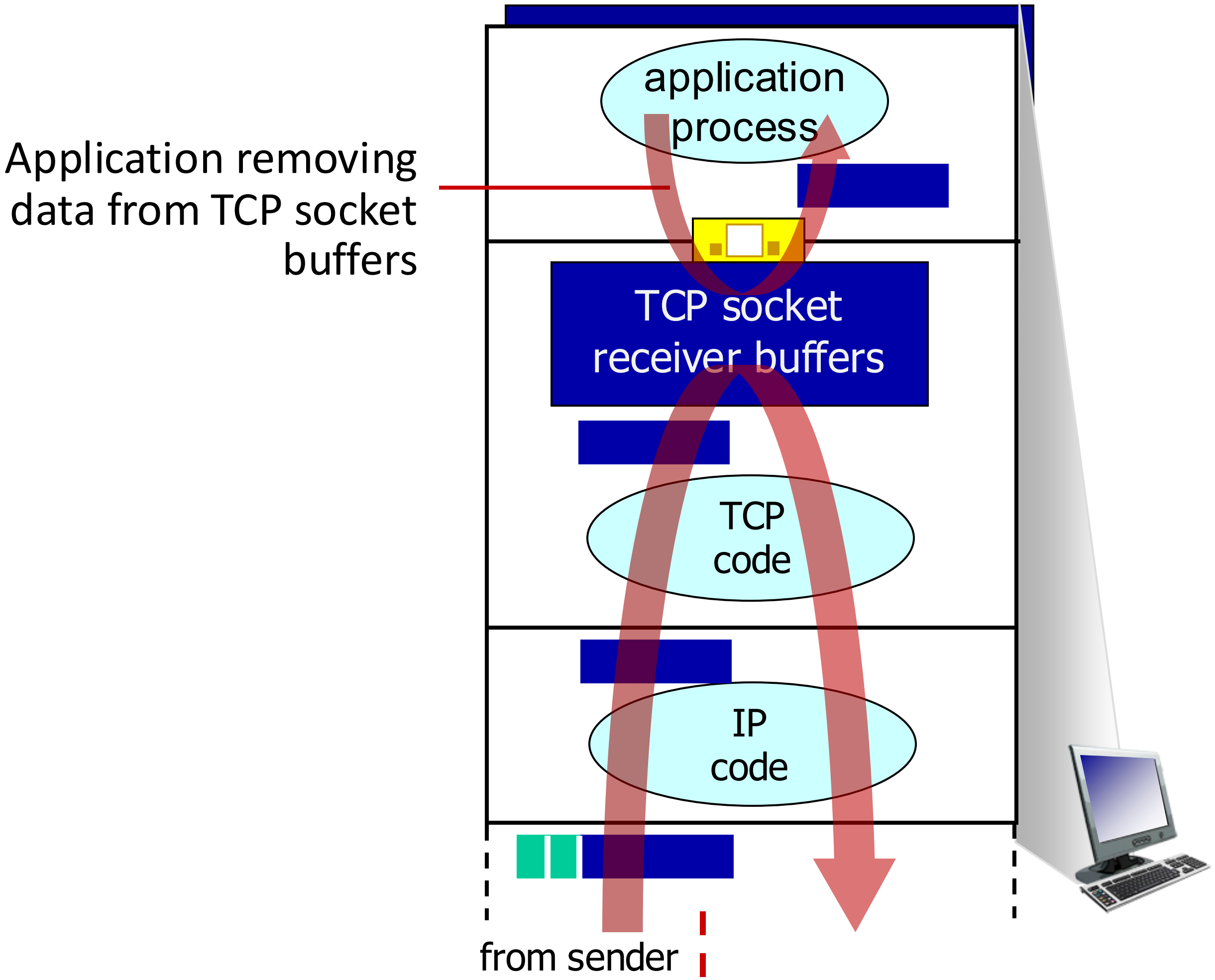
receiver protocol stack

TCP flow control

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

flow control

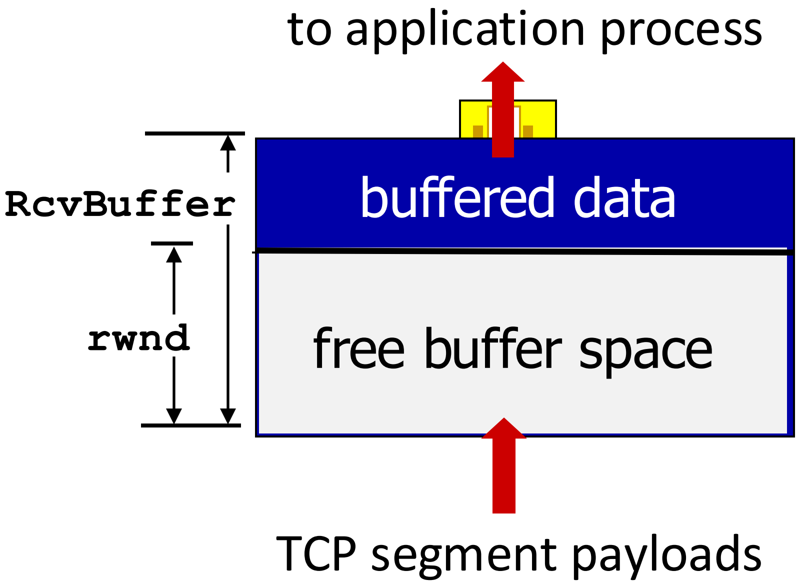
receiver controls sender, so sender won't overflow receiver's buffer by transmitting too much, too fast


Application removing data from TCP socket buffers
application process
TCP socket receiver buffers
TCP code
IP code
from sender

receiver protocol stack

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

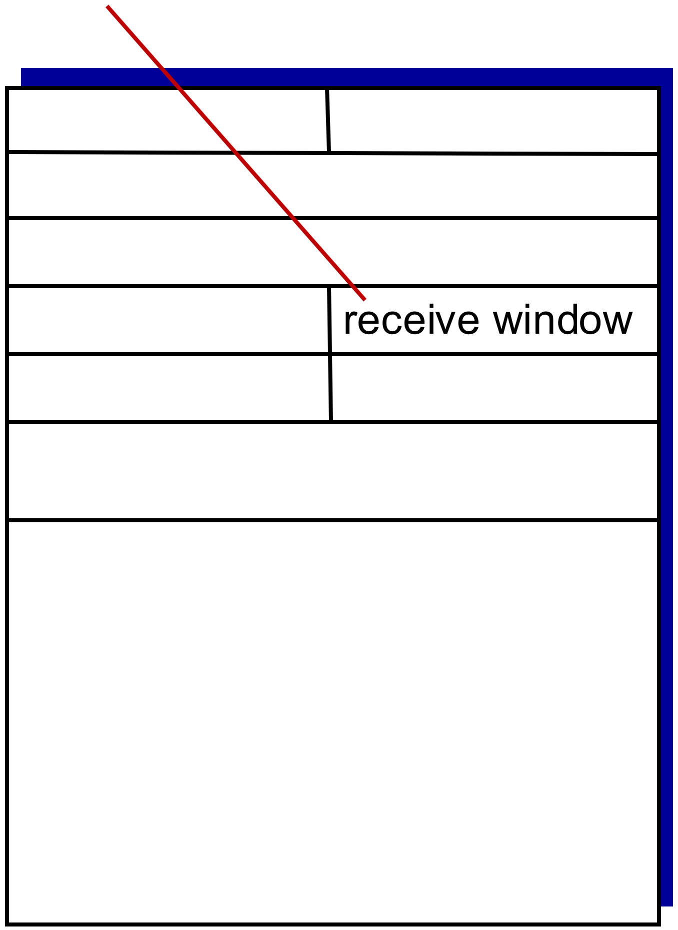


TCP receiver-side buffering

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

flow control: # bytes receiver willing to accept

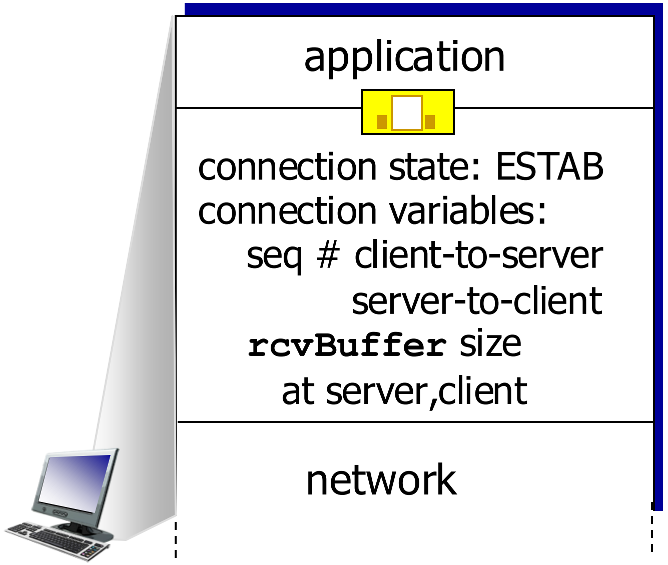


TCP segment format

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)

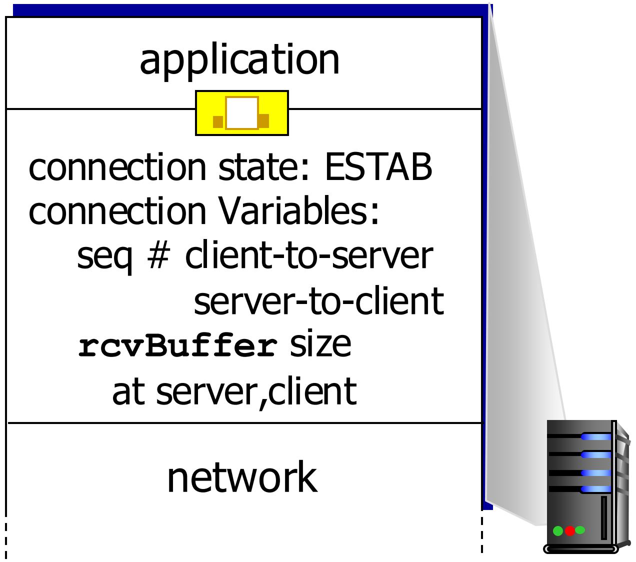


application

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

network

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



application

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

network

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

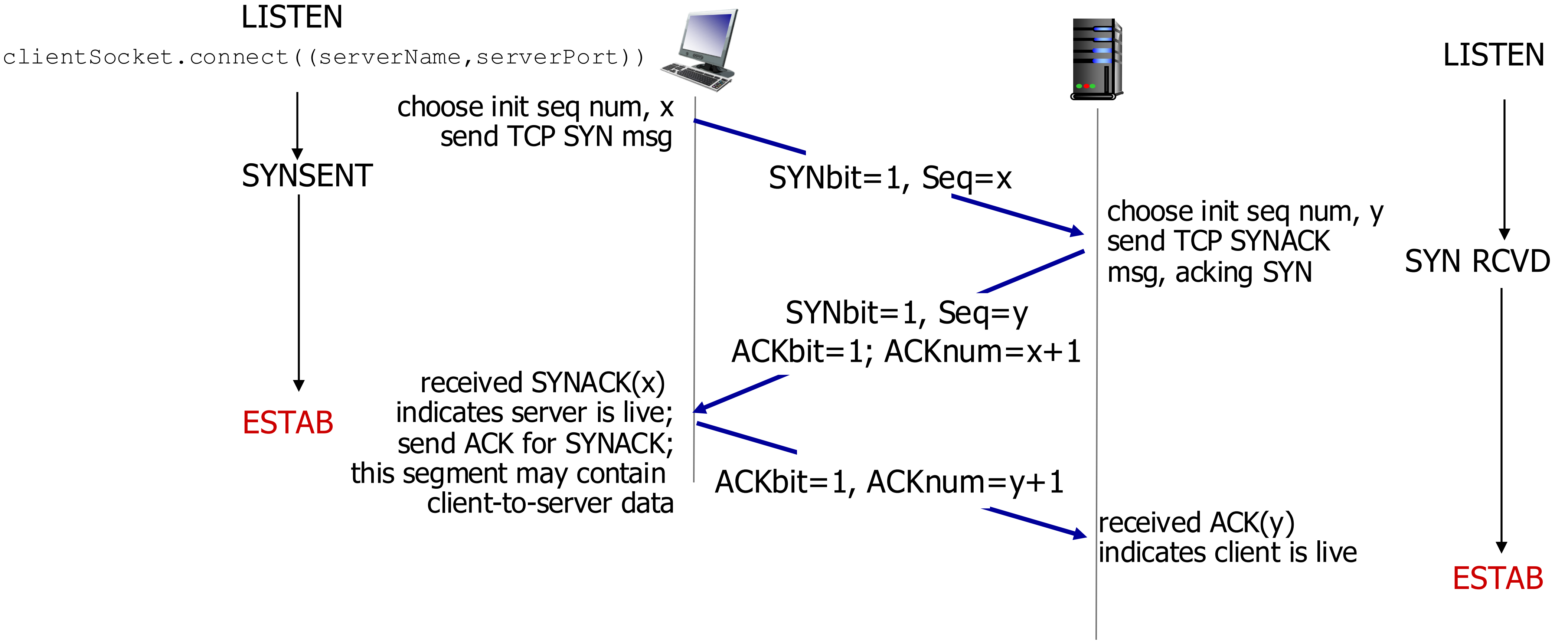
TCP 3-way handshake

Client state

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

Server state

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



LISTEN

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

choose init seq num, x
send TCP SYN msg

SYNSENT

SYNbit=1, Seq=x

LISTEN

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

SYN RCVD

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

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

ESTAB

ACKbit=1, ACKnum=y+1

received ACK(y)
indicates client is live

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