CS 348

Computer Networks

Lec 17
TCP

Spring 2020 IIT Goa

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Note: These slides are adapted from "Computer Networking: A Top-down Approach" by Kurose & Ross, 7th ed

Guiding Questions

- Now that we understand how reliable data transfer can be achieved over an unreliable channel (using checksums, ACKs, timeouts etc.), how does TCP work?
- Which of these protocols does TCP employ? Go-Back-N? Selective-Repeat? How does TCP overcome some of the performance issues in these protocols?
- What does the TCP segment header contain?
- Why is TCP said to be "connection-oriented"?
- Why is a 3-way handshake necessary for setting up a TCP connection? How is a connection breakup performed?
- What is Maximum Segment Size (MSS) and Maximum Transmission Unit (MTU)? How do they differ?
- How is the timeout interval chosen in TCP for retransmission of lost packets?

Features of TCP (RFCs: 793,1122, 2018, 5681, 7323)

- **Point-to-point** (A TCP connection is always between a pair of nodes/processes. No multicasting).
- **Full-duplex** (each end-point of the connection can send as well as receive data)
- **ACKs can be piggy-backed** within a data packet. Example:
  - Say there is a connection between A and B, then the packet flowing from B to A can have data sent by B to A as well as ACK sent by B for some other data packet that arrived from A.

Features of TCP (RFCs: 793,1122, 2018, 5681, 7323)

- **ACKs can be piggy-backed** within a data packet...because of these

  fields in the TCP header:
  - 1-bit Flags (Such as ACK, RST, FIN)
  - sequence number (for DATA being sent)
  - acknowledgement number (for ACK being sent for received data). Only valid if ACK flag==1. Ignored if ACK flag==0.

TCP Header and Segment structure

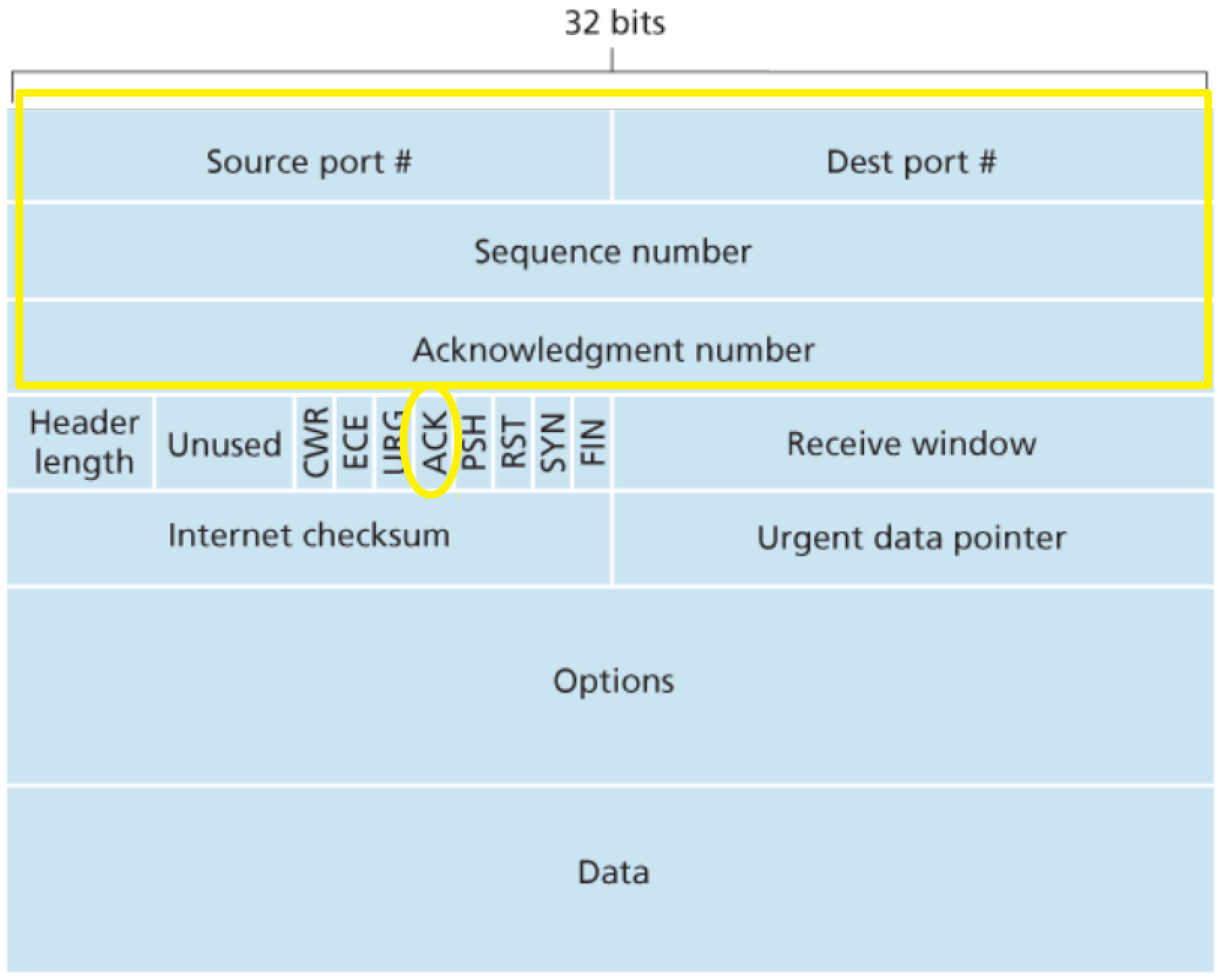


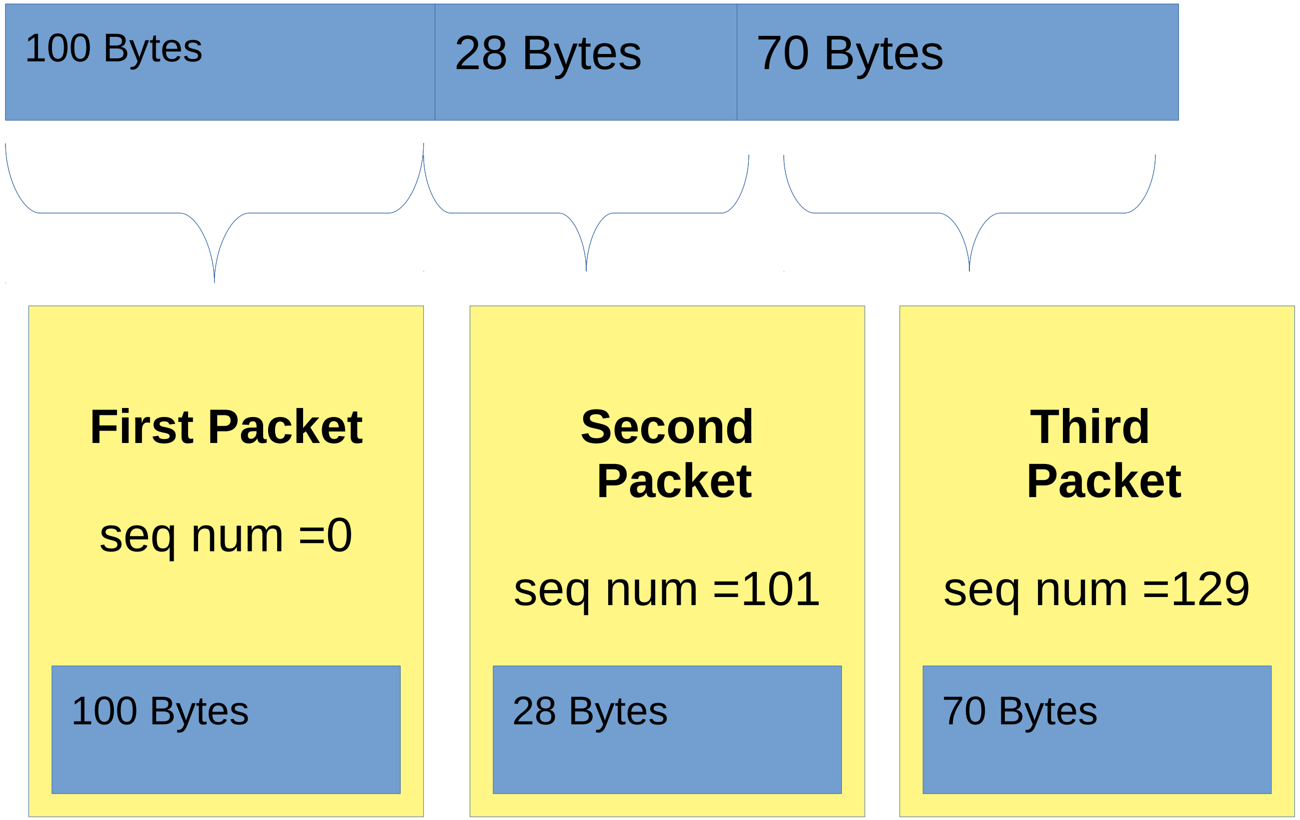
Figure 3.29 TCP segment structure

Sequence and ACK numbers

- **Do not** correspond to the number of packets sent/received!
- **Sequence number**: correponds to the number of the first Byte in the packet in a stream of bytes being sent
- **Ack number:** number of the next Byte expected in the next packet.

Sequence and ACK numbers

Stream of Bytes being sent from **A to B**



Sequence and ACK numbers

Stream of Bytes being sent from **A to B**

100 Bytes	28 Bytes	70 Bytes

Similarly, stream of Bytes being sent from **B to A**

10 Bytes	8 Bytes	71 Bytes

Sequence and ACK numbers

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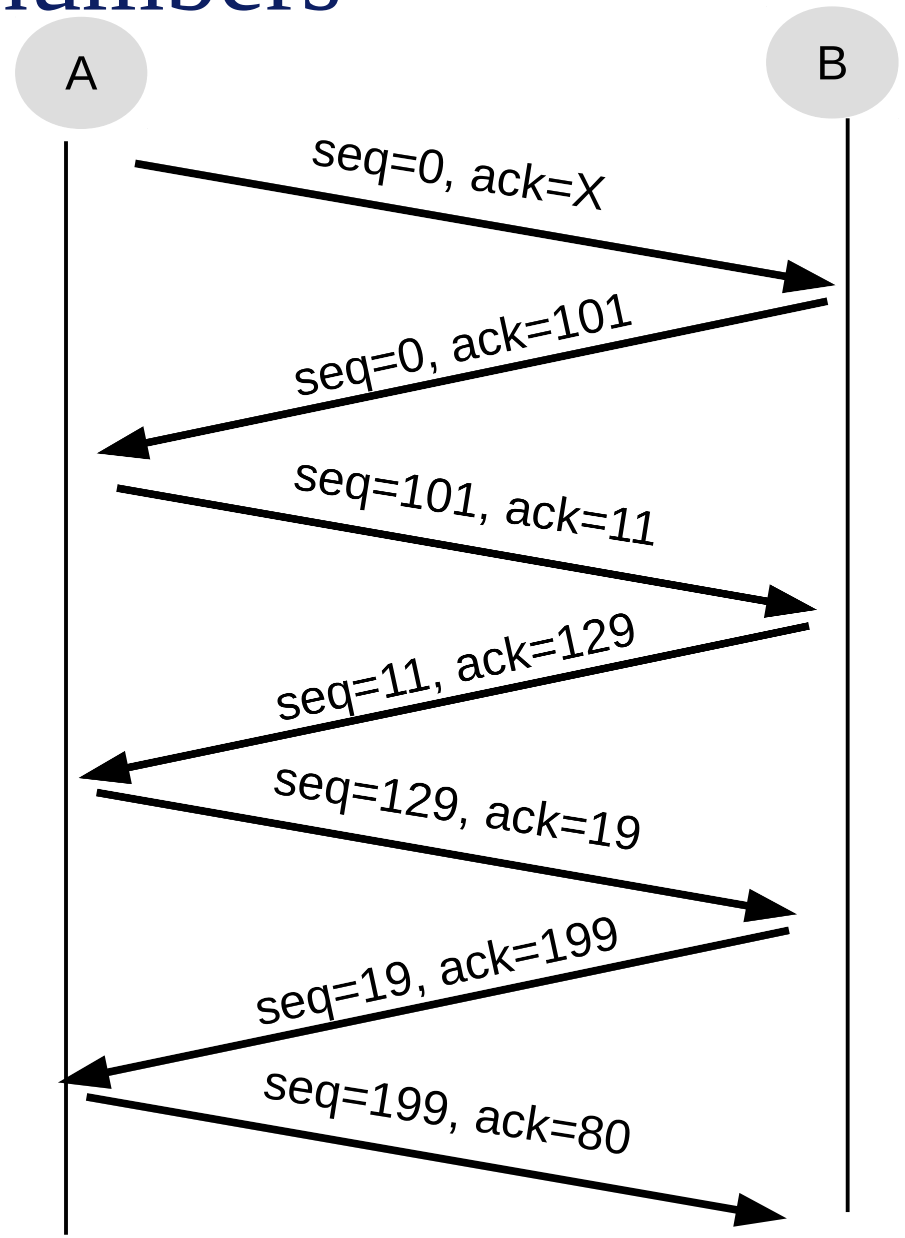
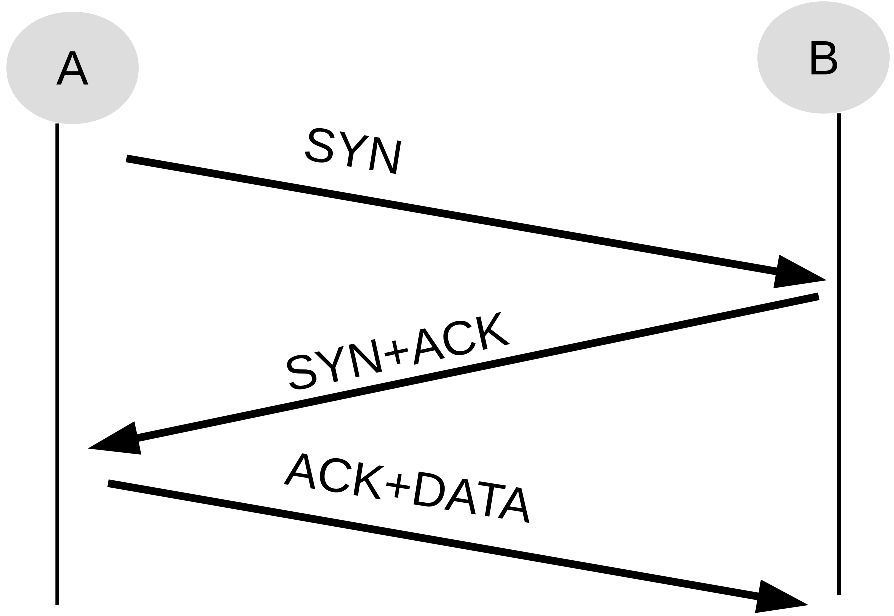


Sequence and ACK numbers

- **However,** the Initial Sequence Number is not 0, but a randomly chosen 32-bit number.
  - **Why is it not 0?**
- At connection establishment stage, each side conveys their chosen ISN to the other side.---->This is one reason why a 3-way handshake is required)



Features of TCP

- **Point-to-point** (A TCP connection is always between a pair of nodes/processes. No multicasting).
- **Full-duplex** (each end-point of the connection can send as well as receive data)
- **ACKs can be piggy-backed** within a data packet. Seq and ack numbers correspond to the Byte number in the data stream.
- Seq and Ack numbers **do not start from 0**, but a randomly chosen Initial Sequence Number.
- A connection is established using a 3 -way handshake consisting of SYN, SYN+ACK and ACK messages
  - Why is a connection establishment phase necessary at all?
  - Why is a 3-way handshake required? Why can't it be a 2-way handshake?

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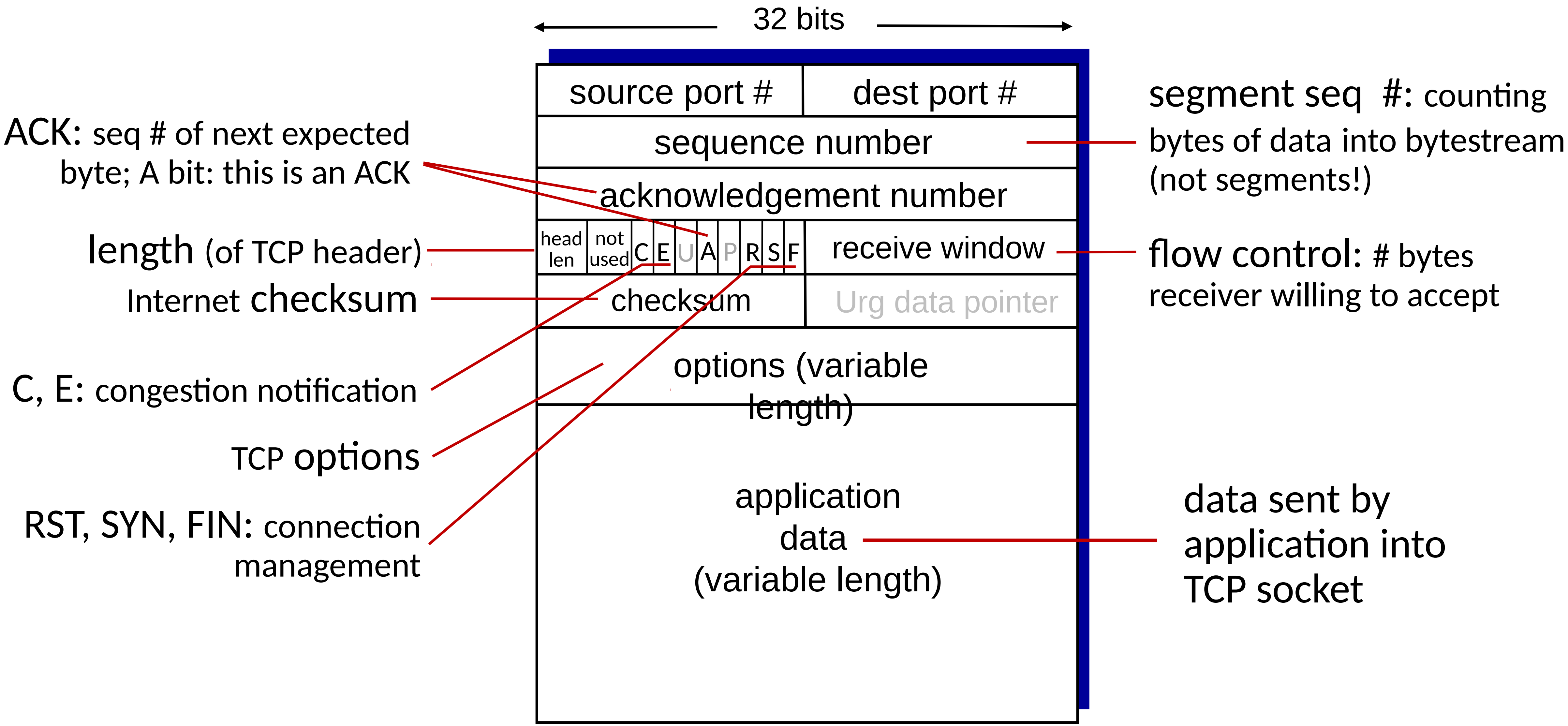
TCP: overview RFCs: 793,1122, 2018, 5681, 7323

- 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 segment structure

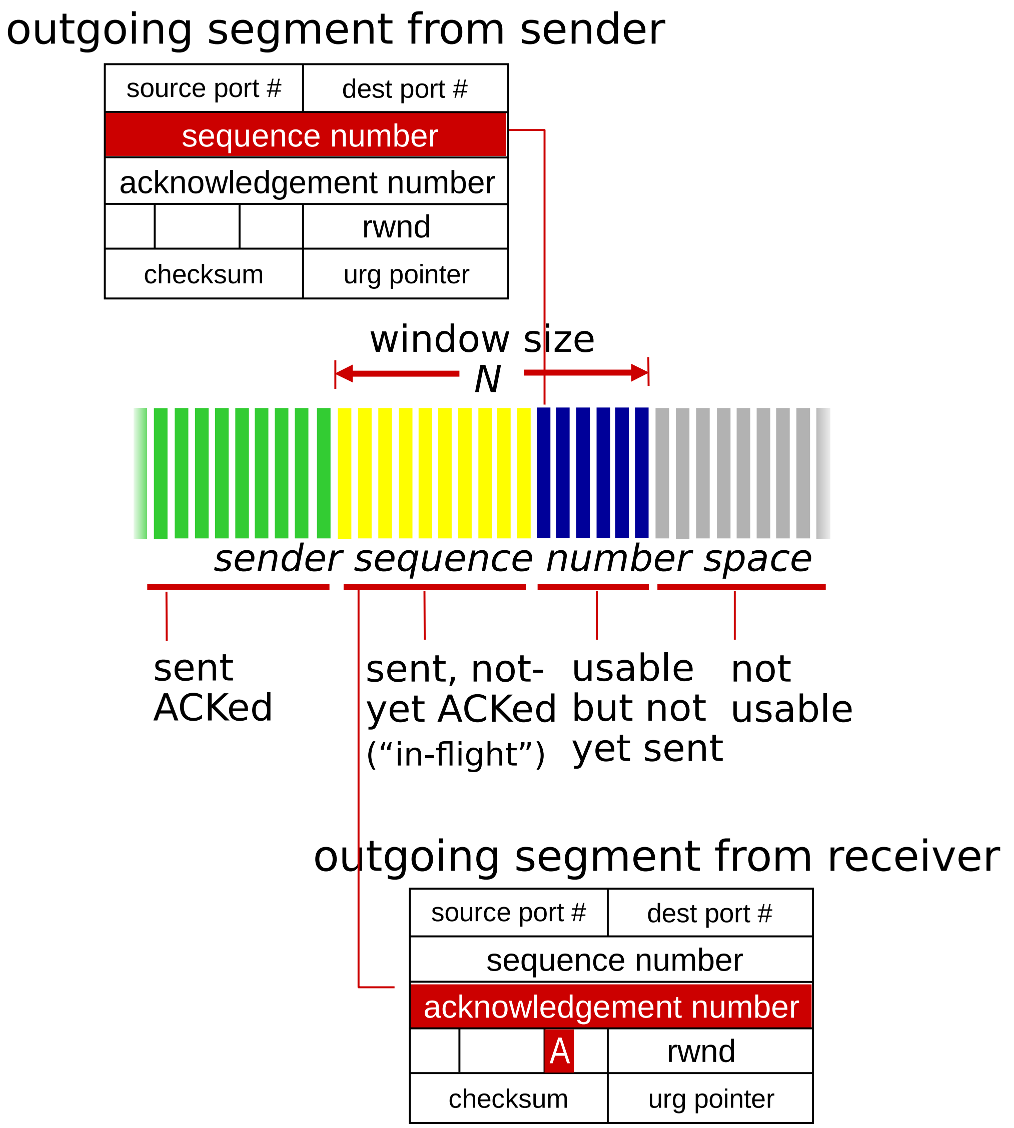


TCP sequence numbers, ACKs

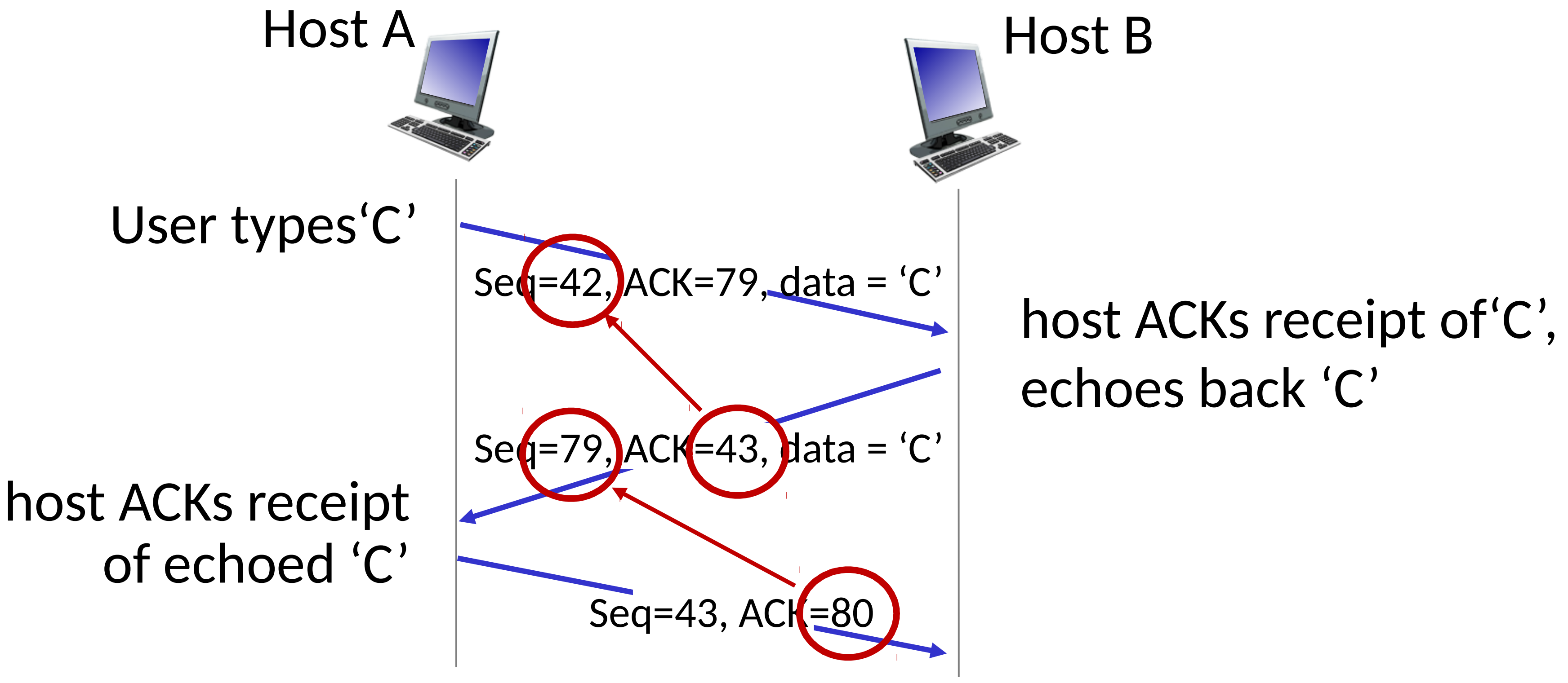
Sequence numbers:
- byte stream "number" of first byte in segment's data

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



TCP sequence numbers, ACKs



simple telnet scenario

TCP round trip time, timeout

Q: how to set TCP timeout value?

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

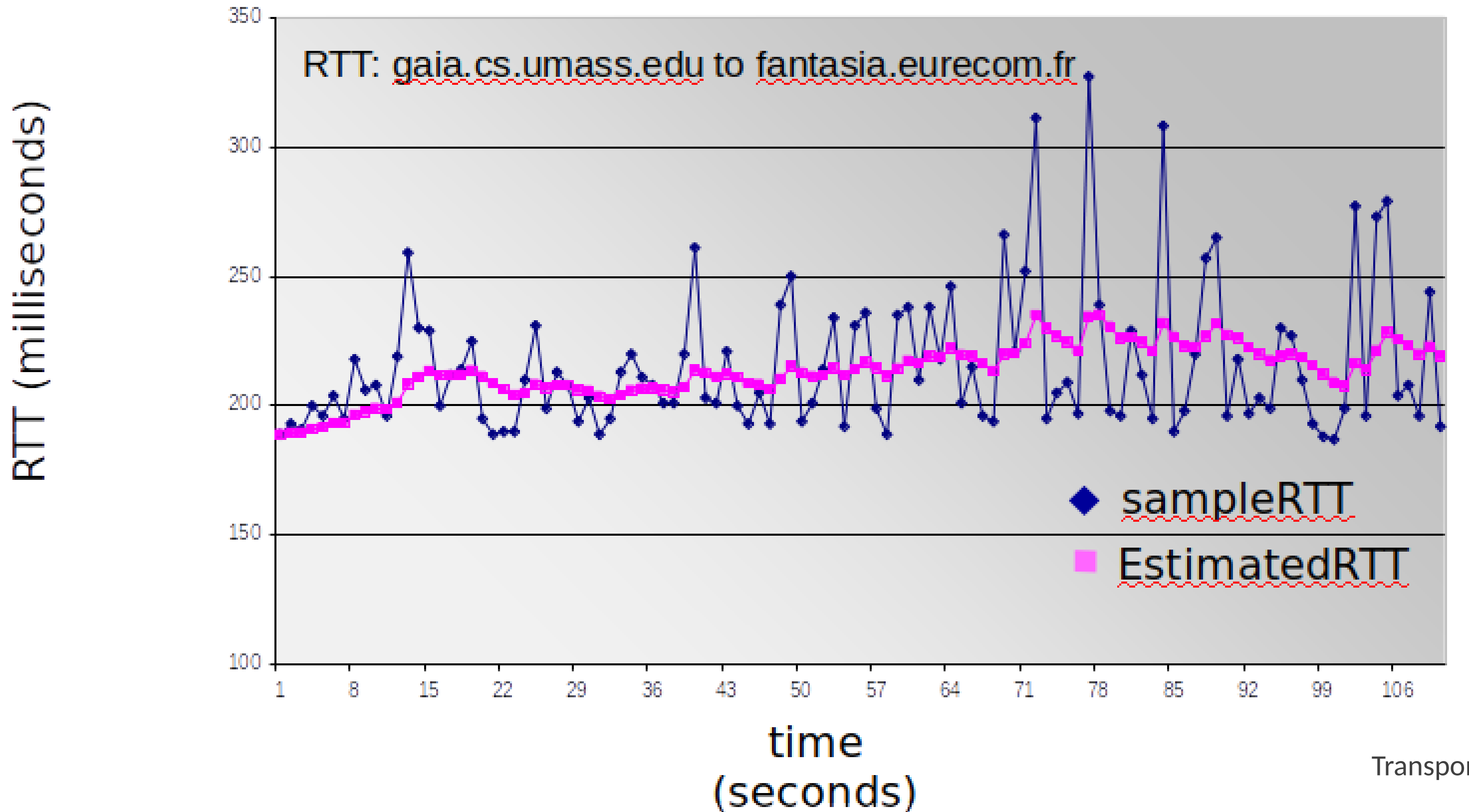
Q: how to estimate RTT?

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

TCP round trip time, timeout

```
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

```
TimeoutInterval = EstimatedRTT + 4*DevRTT
```

estimated RTT "safety margin"

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

```
DevRTT = (1-β)*DevRTT + β*|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

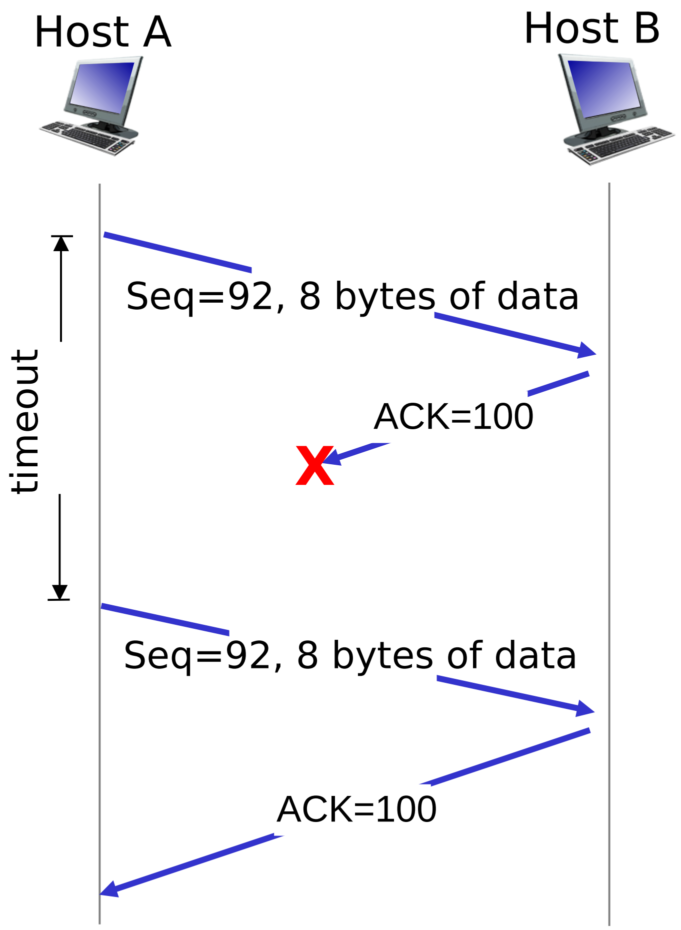
event: ACK received

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

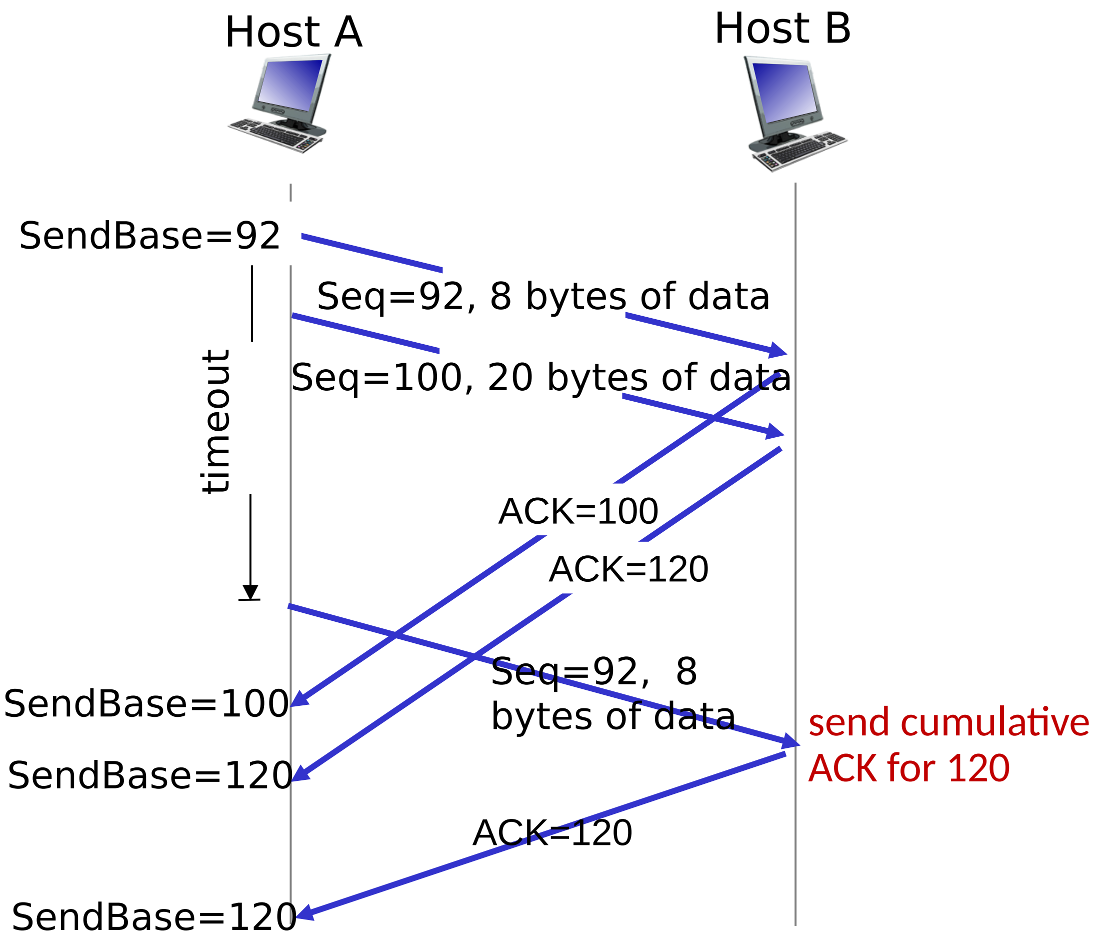
TCP Receiver: ACK generation [RFC 5681]

Event at receiver	*TCP receiver action*

TCP: retransmission scenarios

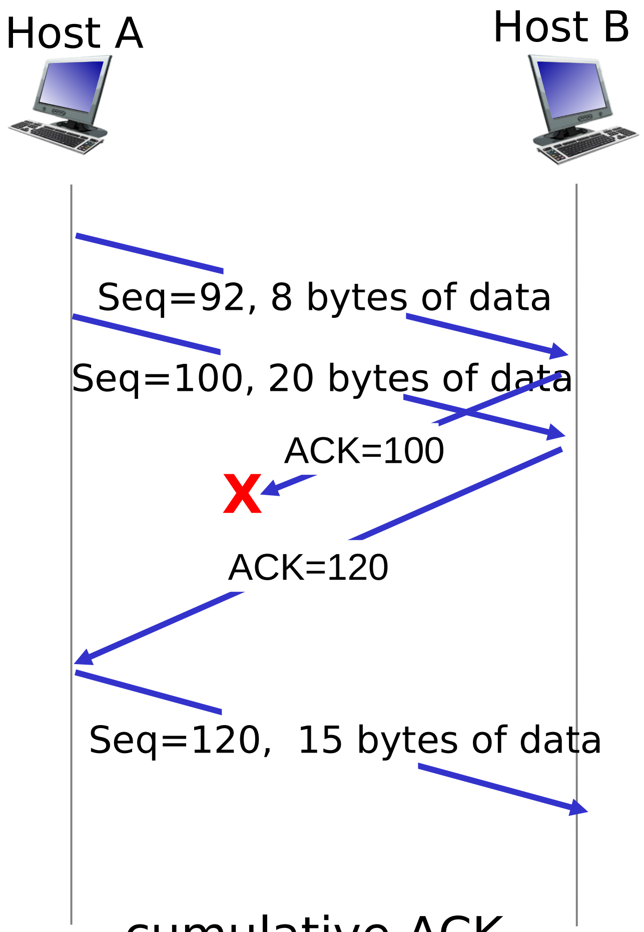


lost ACK scenario



premature timeout

TCP: retransmission scenarios



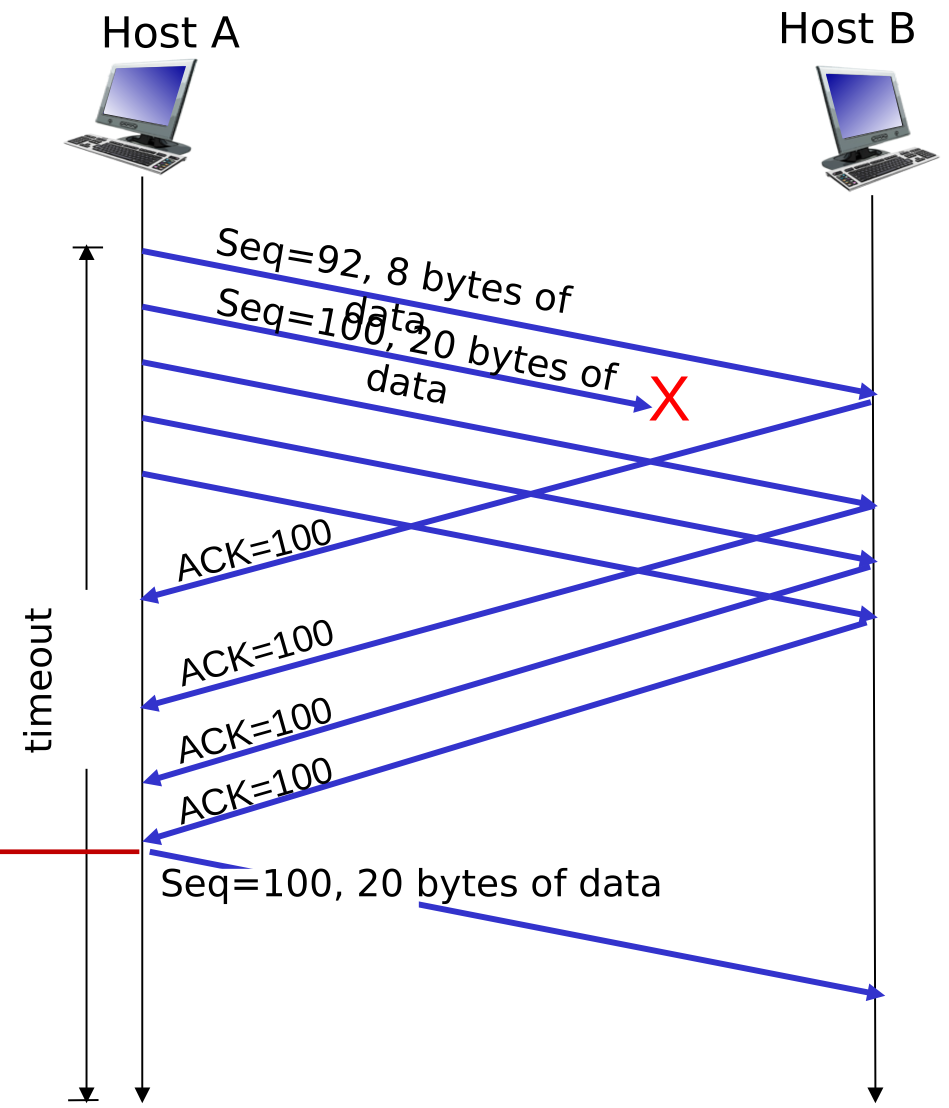
cumulative ACK covers for earlier lost ACK

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



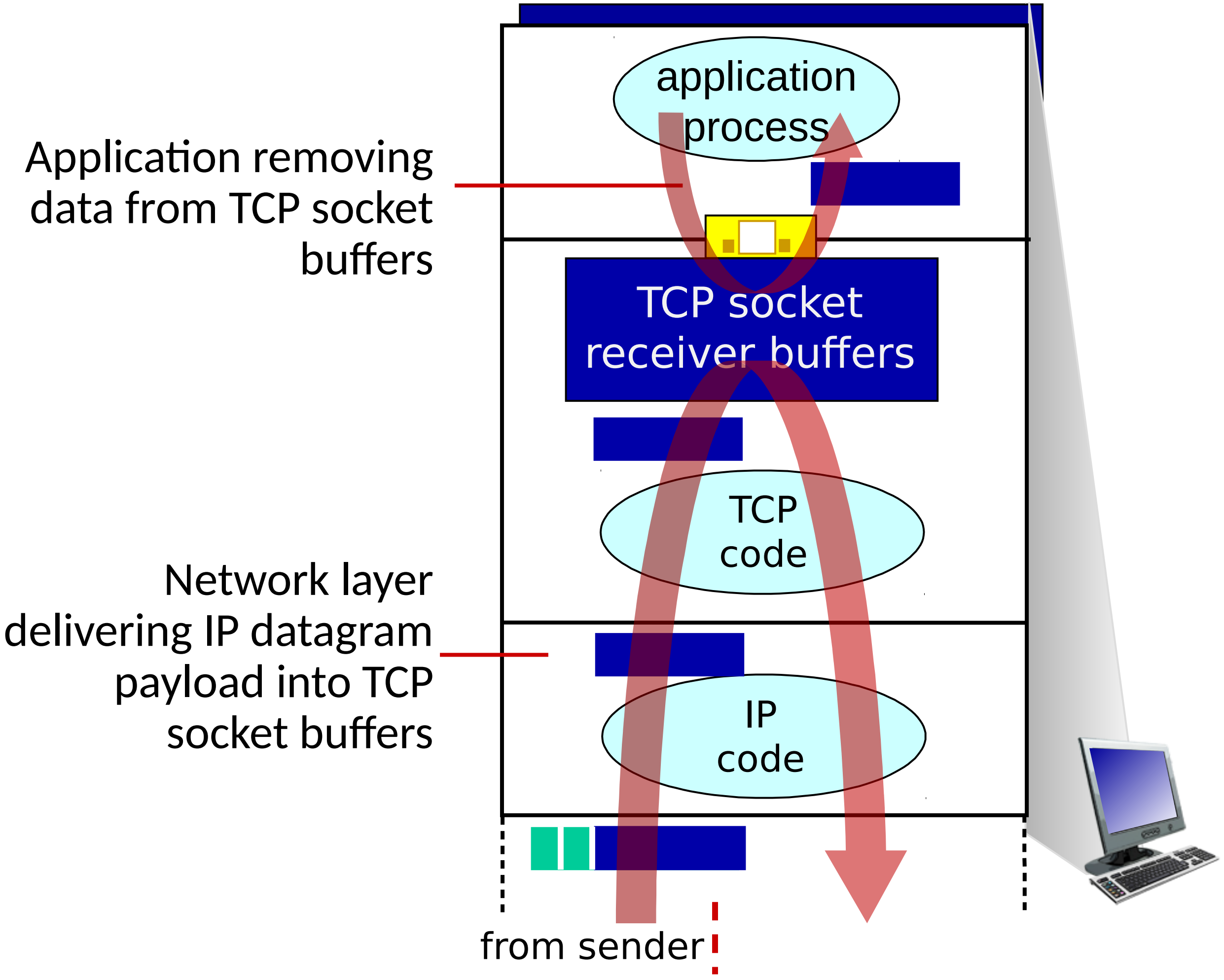
Receipt of three duplicate ACKs indicates 3 segments received after a missing segment – lost segment is likely. So retransmit!

Chapter 3: roadmap

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

TCP flow control

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



receiver protocol stack

TCP flow control

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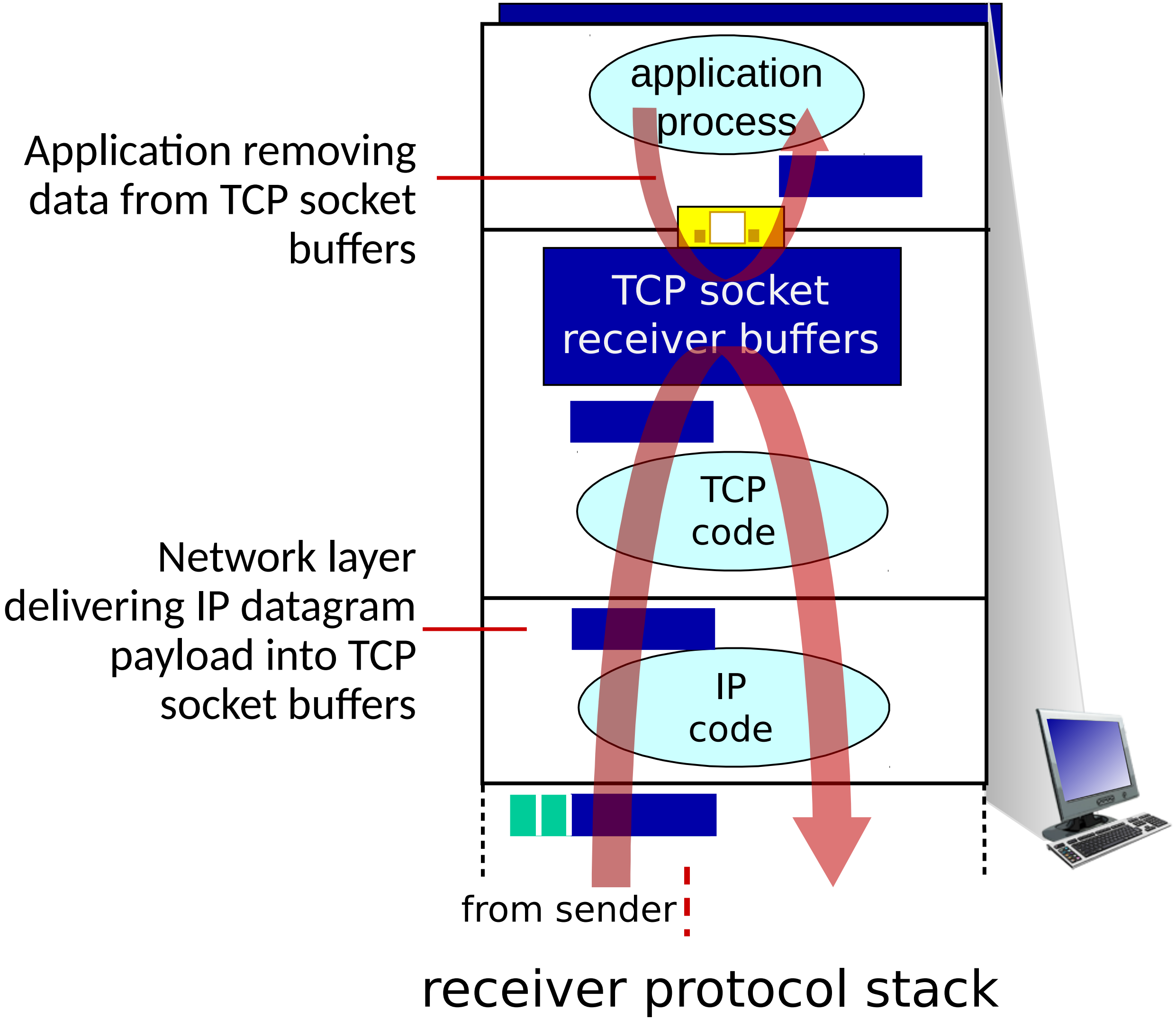

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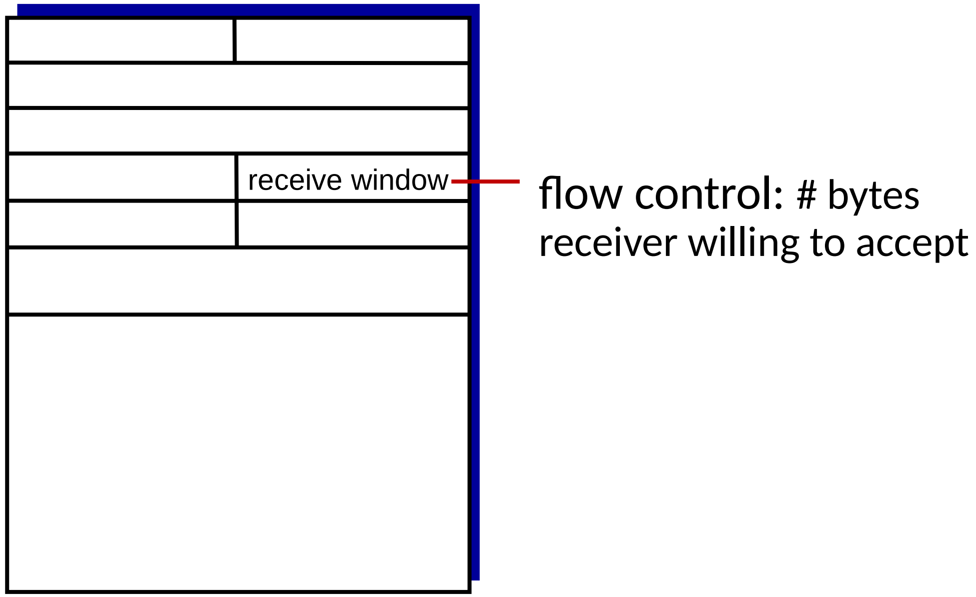

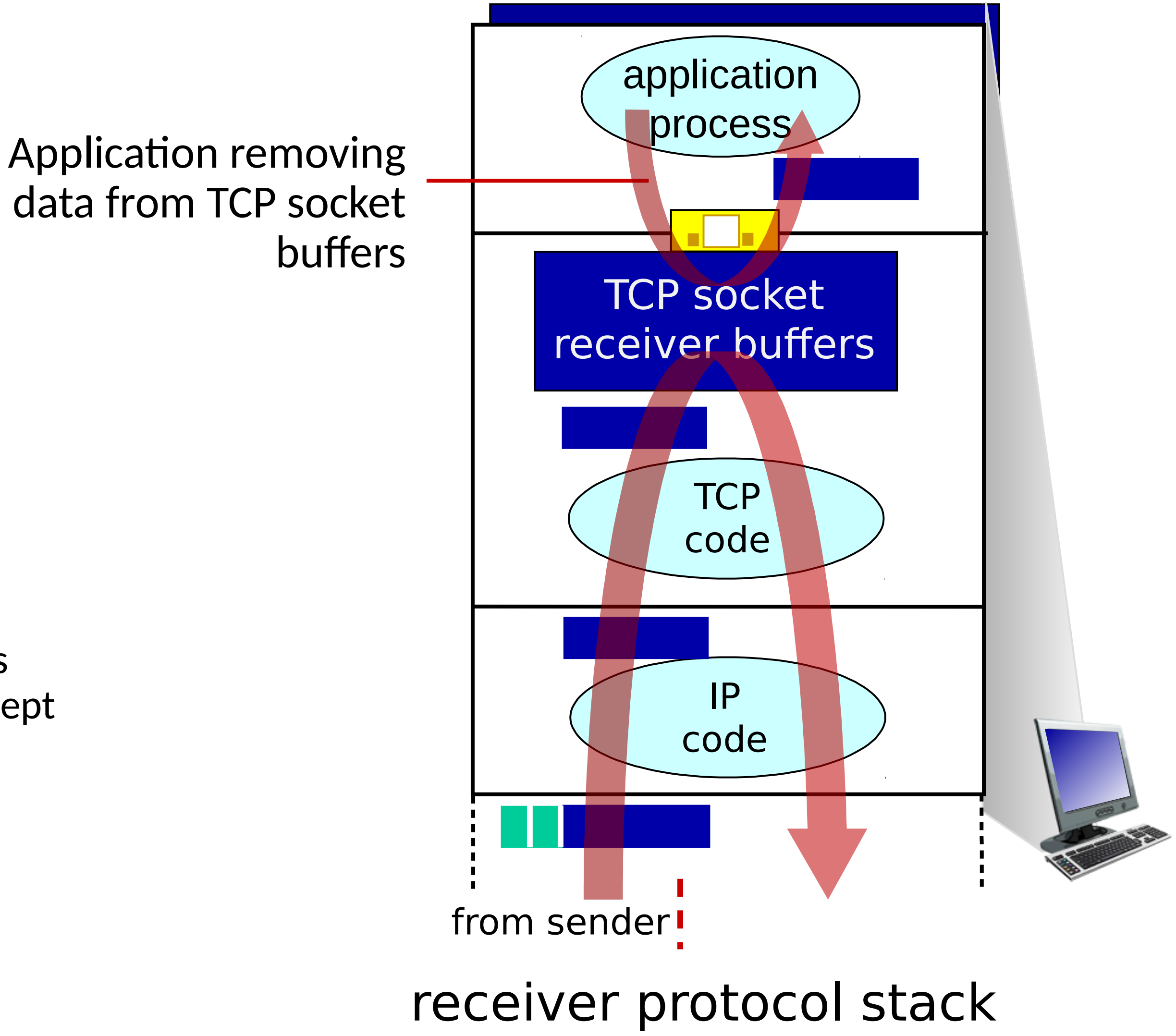

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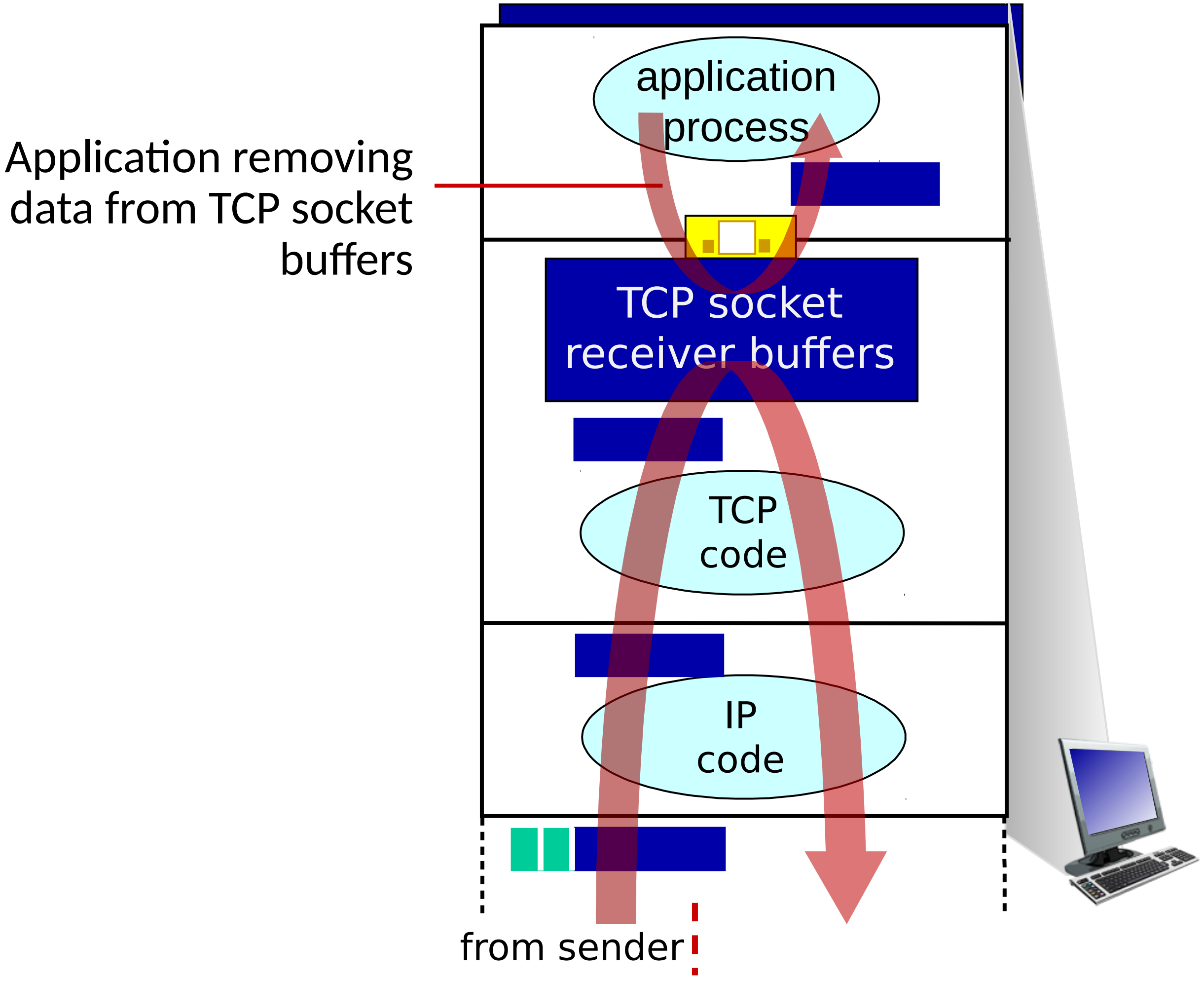


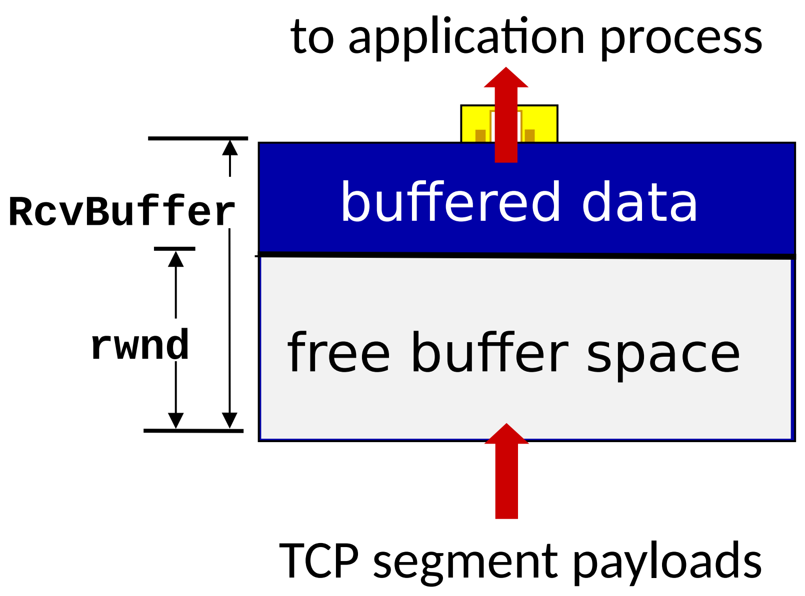
receiver protocol stack

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

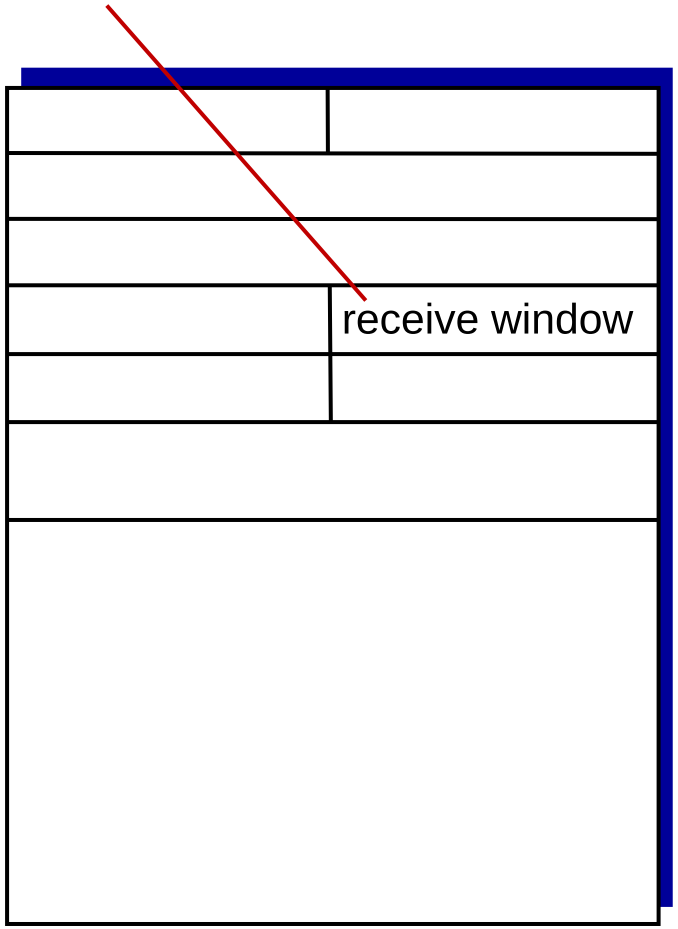


TCP receiver-side buffering

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

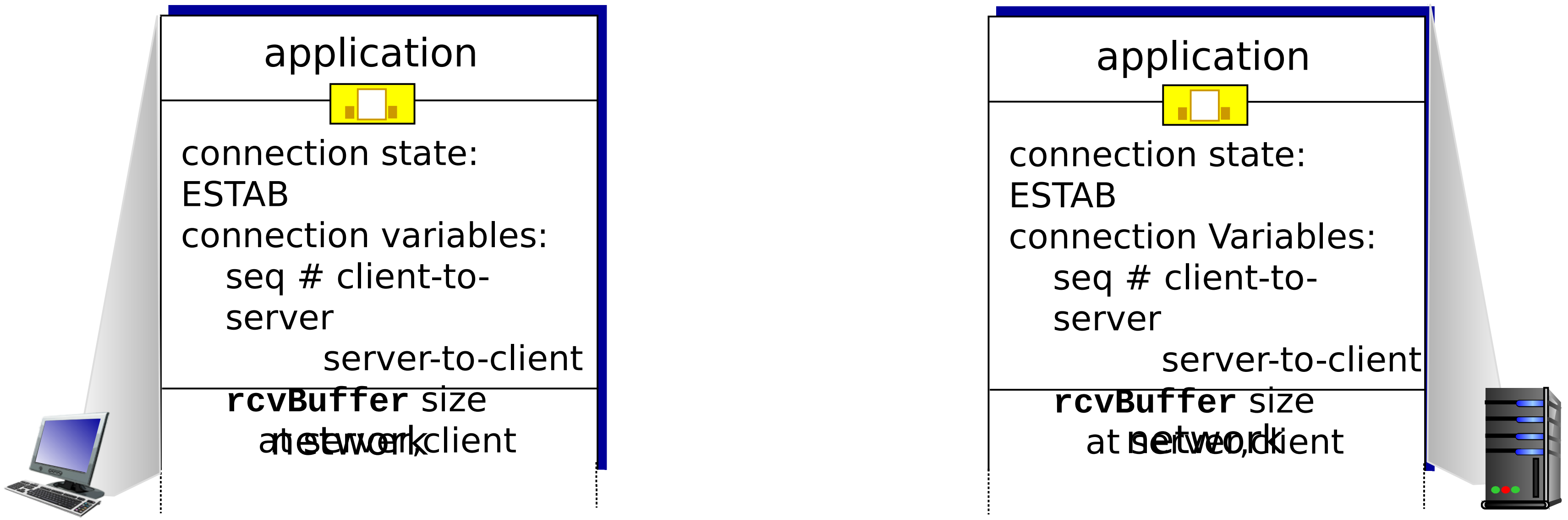


TCP segment format

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)

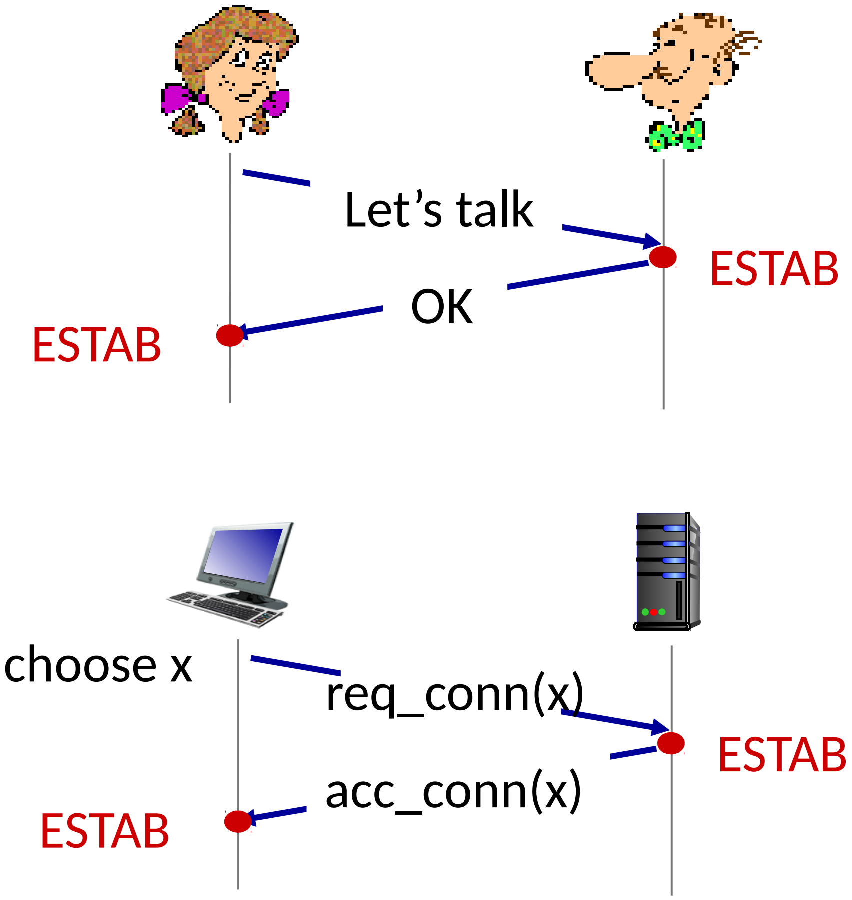



```
Socket clientSocket =
  newSocket("hostname","port number");
```

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

Agreeing to establish a connection

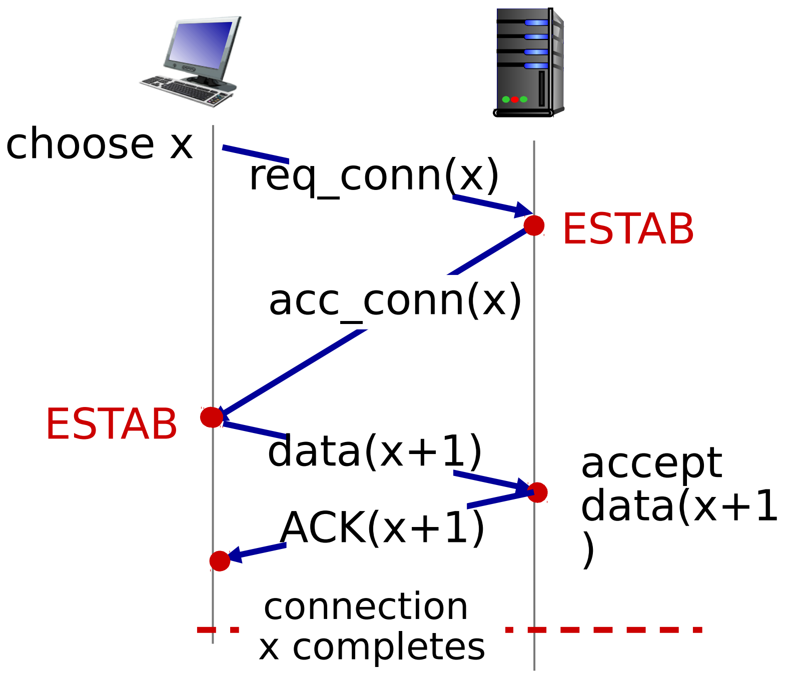
2-way handshake:



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

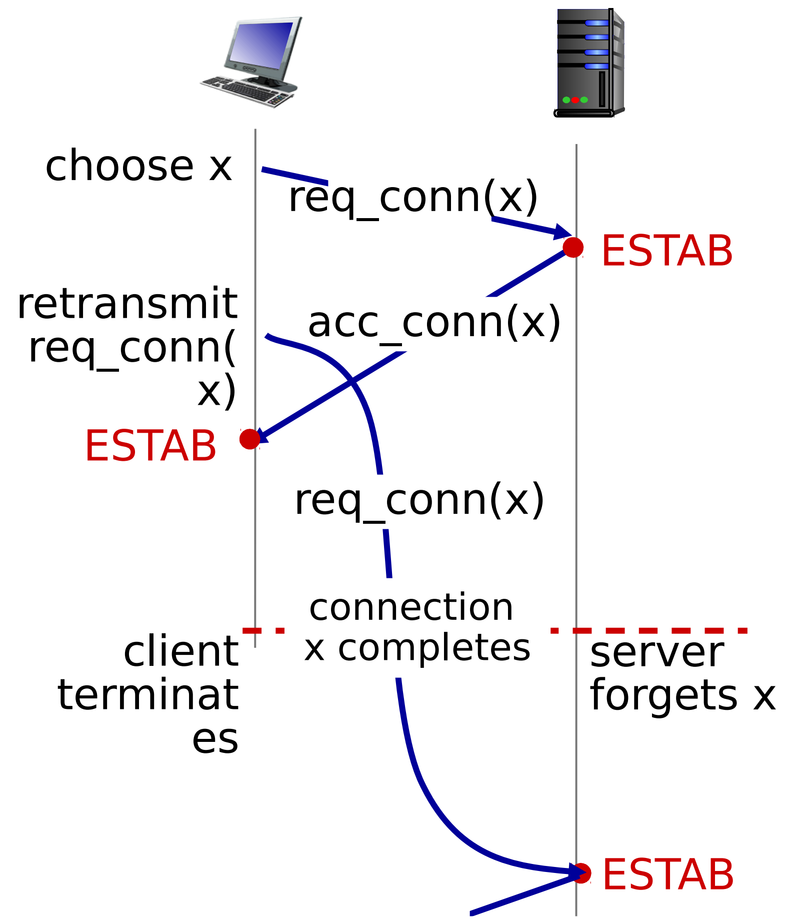
- variable delays
- retransmitted messages (e.g. req_conn(x)) due to message loss
- message reordering
- can't "see" other side

2-way handshake scenarios



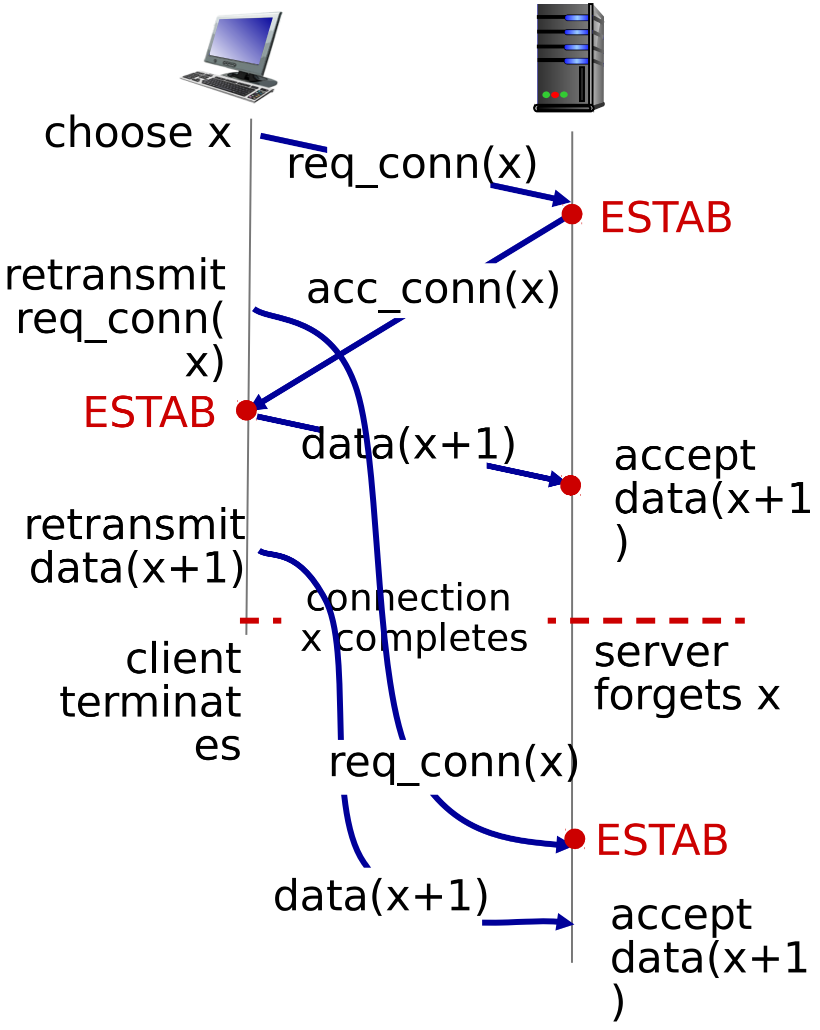
No problem!

2-way handshake scenarios



Problem: half open connection! (no client)

2-way handshake scenarios



Problem: dup data accepted!

TCP 3-way handshake

Client state

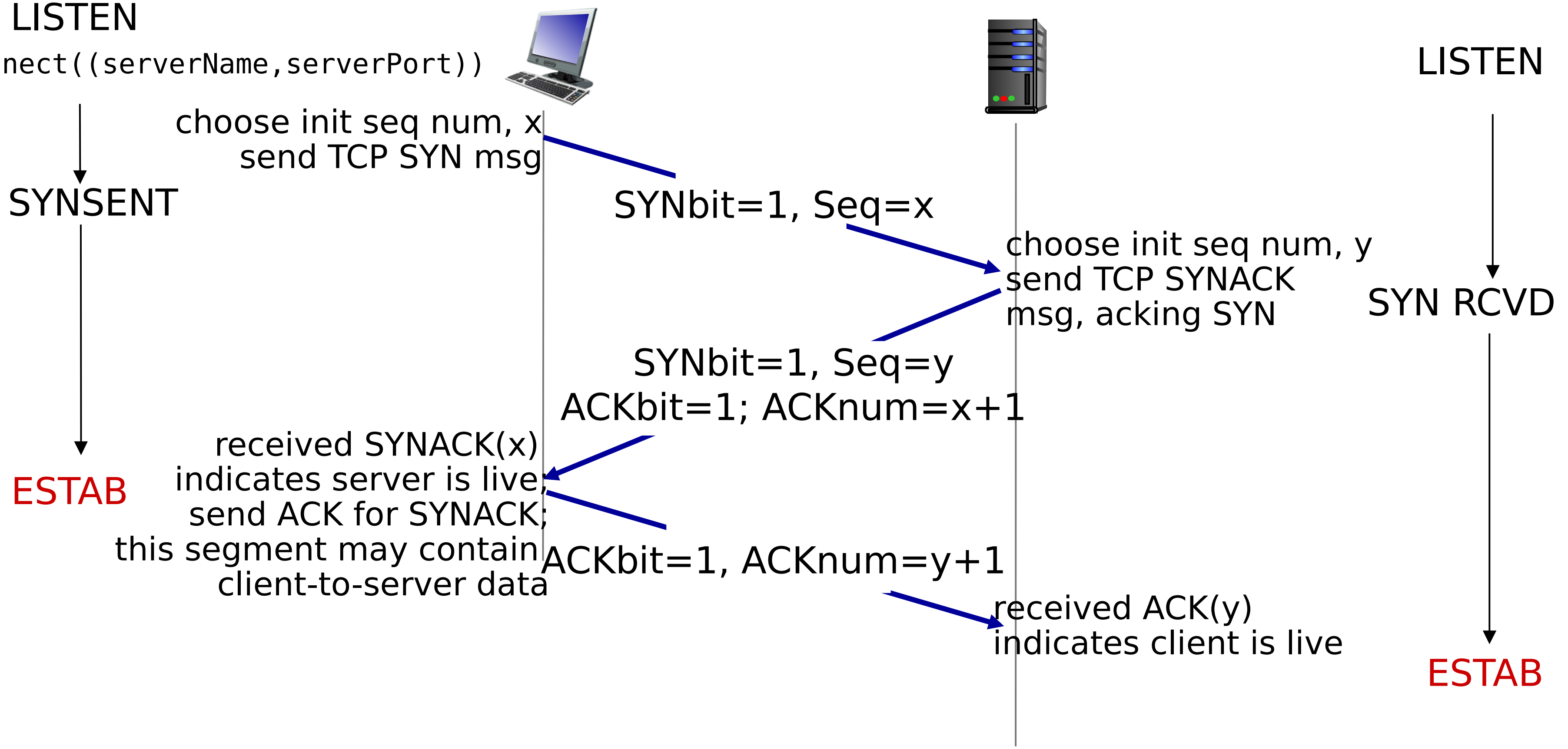
```
clientSocket = socket(AF_INET, SOCK_STREAM)
```

LISTEN

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

Server state

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



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

References and Reading Assignment

- **Kurose and Ross 6th ed or 7th ed:** Section 3.5

So far...

- Structure and Physical components of the Internet
- Design of the Internet: Layering and Encapsulation
- The Applications Layer:
  - Sockets Interface
  - The Web and HTTP
  - DNS
- **The Transport Layer: how it works**
  - **Basic services, UDP**
  - **Principles of Reliable Data Transfer (rdt 3.0 etc)**
  - **Pipelined data transfer (Sliding window protocols)**
  - **TCP details**
  - **Congestion and Flow control**