Computer Networks
CS3001
(Section BDS-7A)
Lecture 15

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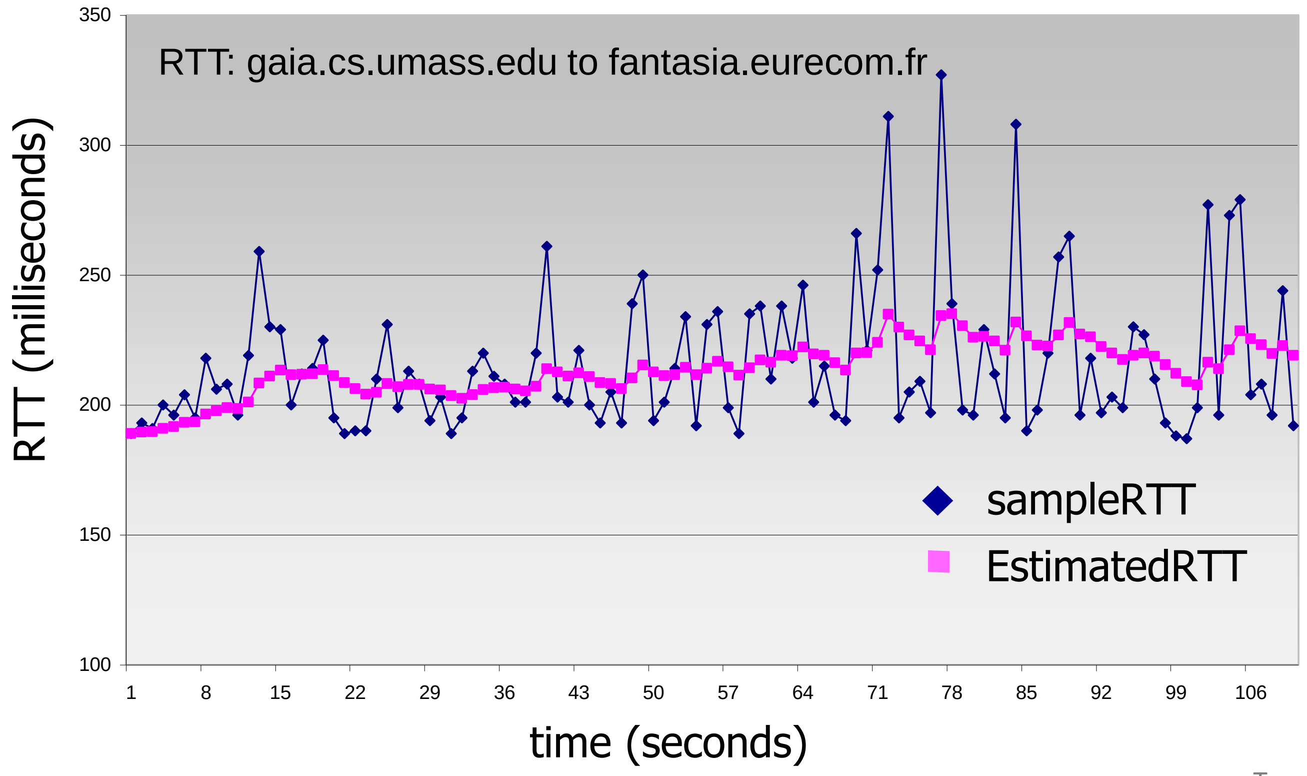
TCP round trip time, timeout

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 (EstimatedRTT) "safety margin" (4*DevRTT)

- **`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 round trip time, timeout

Suppose that the five measured SampleRTT values are 106 ms, 120 ms, 140 ms, 90 ms, and 115 ms.
Compute the EstimatedRTT after each of these SampleRTT values is obtained, using a value of $\alpha = 0.125$ and assuming that the value of EstimatedRTT was 100 ms just before the first of these five samples were obtained.

Compute also the DevRTT after each sample is obtained, assuming a value of $\beta = 0.25$ and assuming the value of DevRTT was 5 ms just before the first of these five samples was obtained.
Last, compute the TCP TimeoutInterval after each of these samples is obtained.

TCP round trip time, timeout

Calculate the EstimatedRTT after obtaining the first sample RTT=106*ms*,

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

EstimatedRTT = 0.125 * 106 + (1-0.125) * 100

=0.125* 106 + 0.875 * 100

=13.25 + 87.5

=100.75*ms*

DevRTT = β * | *SampleRTT- EstimatedRTT*|+(1- β)* *DevRTT*

=0.25 * |106-100.75| + (1-0.25) *5

=0.25 *5.25 + 0.75 * 5

=1.3125 + 3.75

=5.0625*ms*

TimeoutInterval = *EstimatedRTT* +4* *DevRTT*

= 100.75 + 4 *5.0625

=121*ms*

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

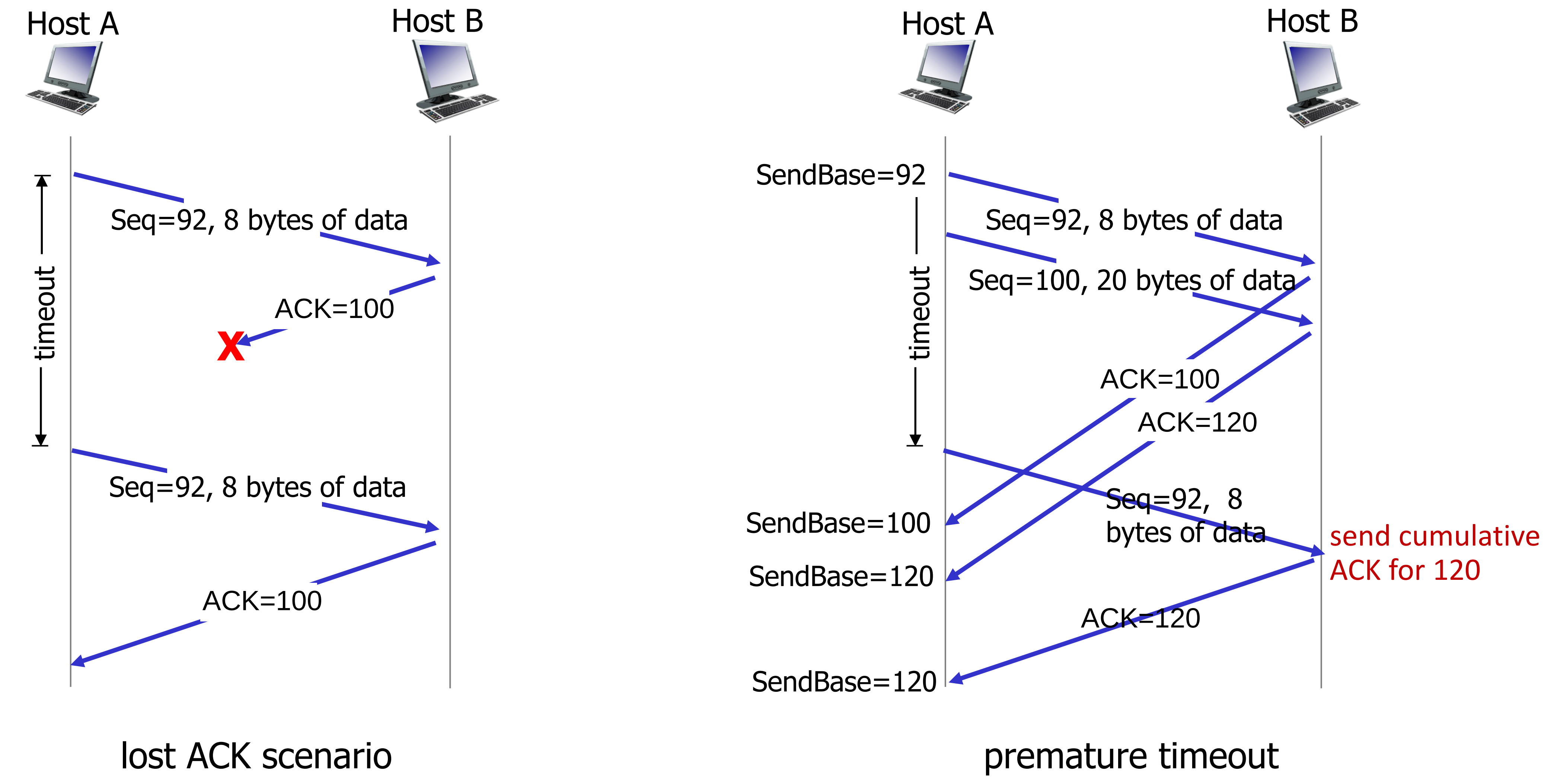
event: ACK received

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

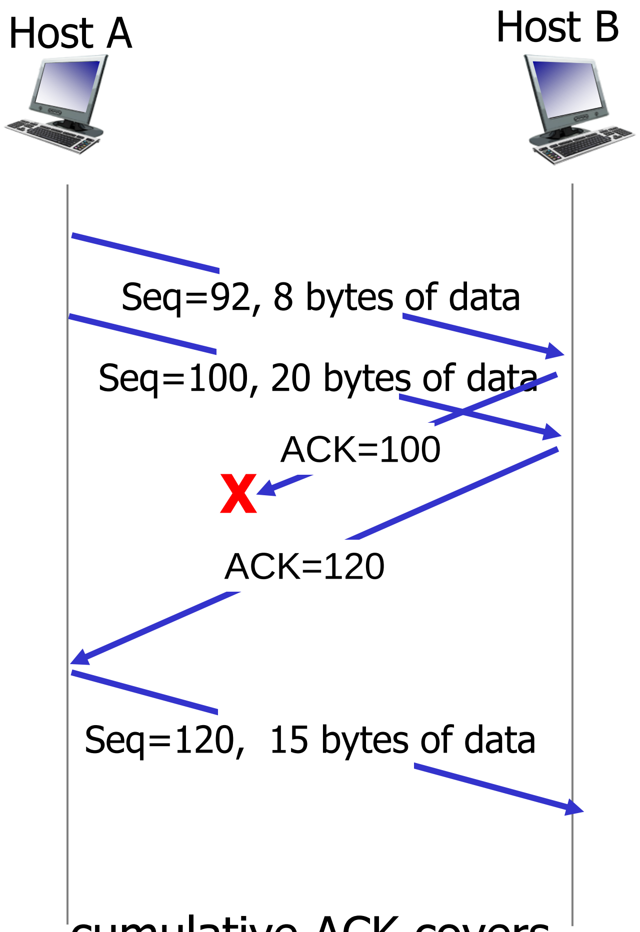
TCP Receiver: ACK generation [RFC 5681]

Event at receiver	*TCP receiver action*

TCP: retransmission scenarios



TCP: retransmission scenarios



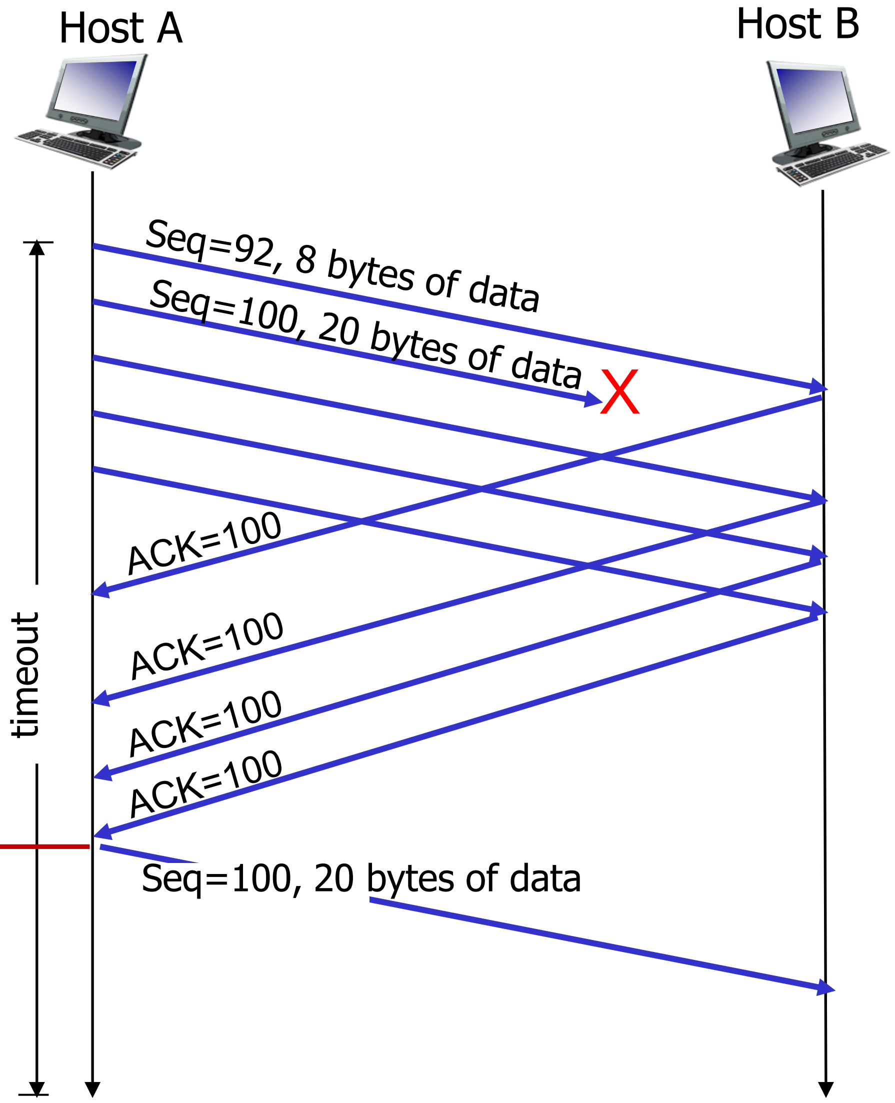
cumulative ACK covers for earlier lost ACK

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



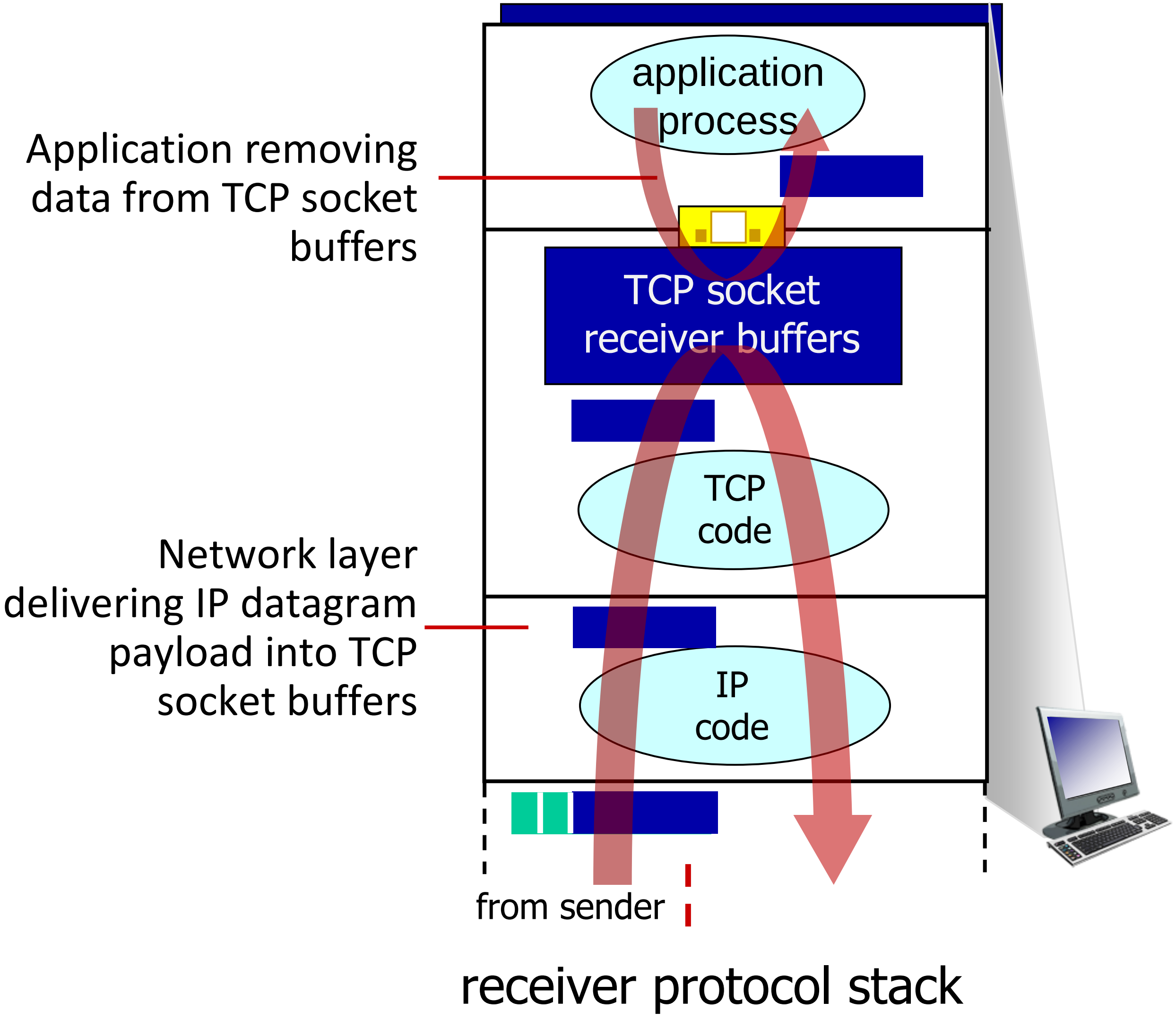
Receipt of three duplicate ACKs indicates 3 segments received after a missing segment – lost segment is likely. So retransmit!

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

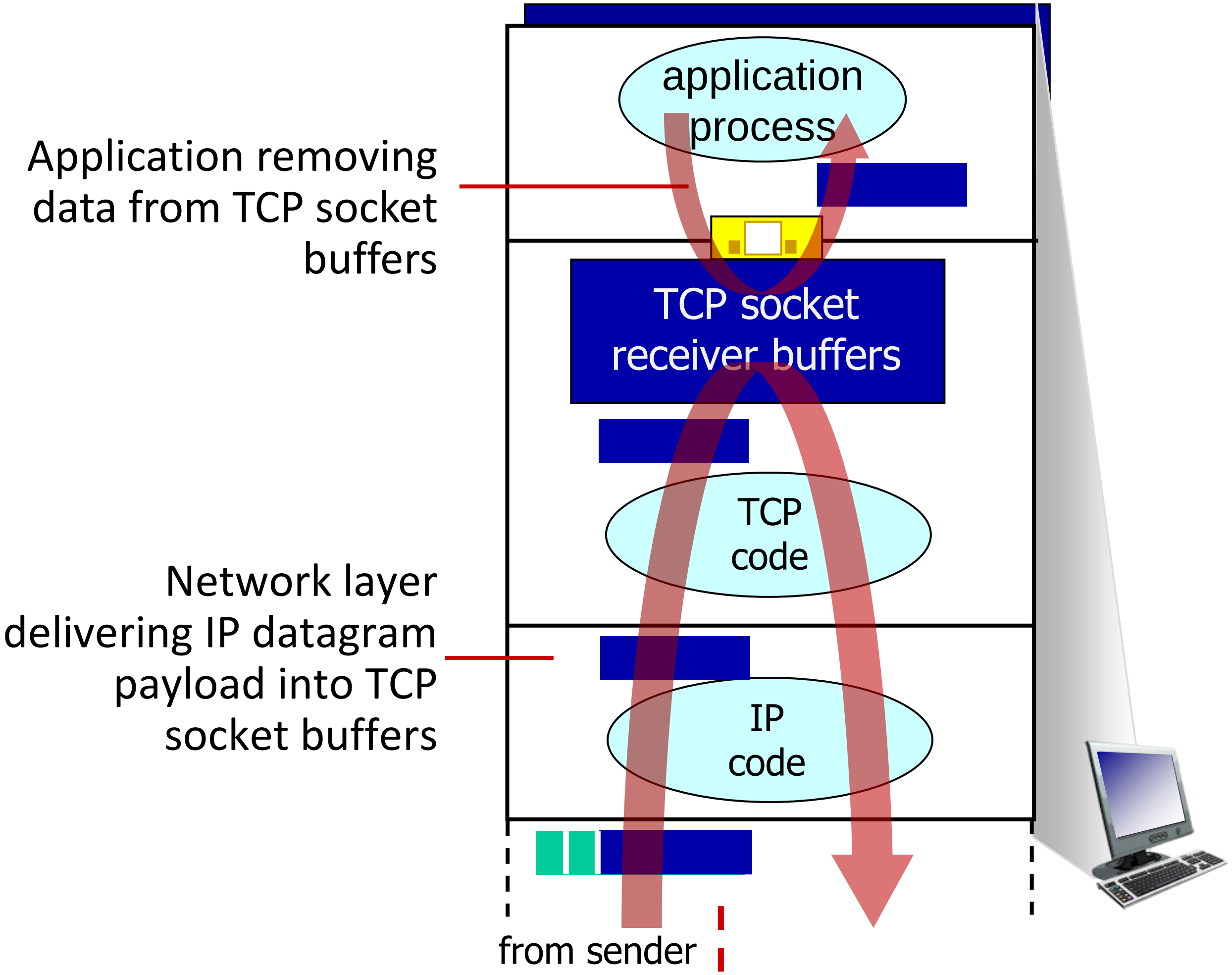
TCP flow control

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



TCP flow control

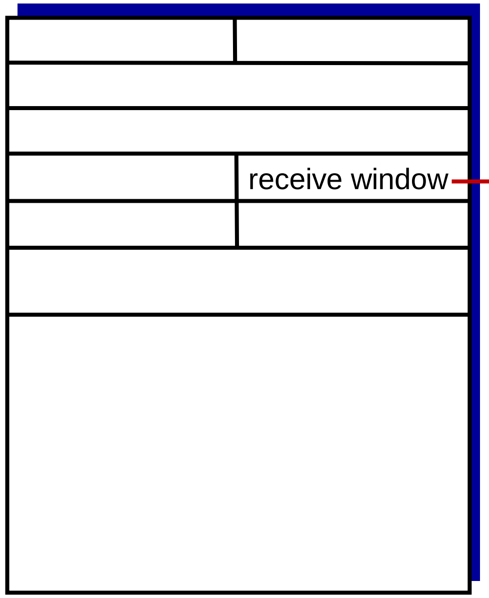
Q: What happens if network layer delivers data faster than application layer removes data from socket buffers?



receiver protocol stack

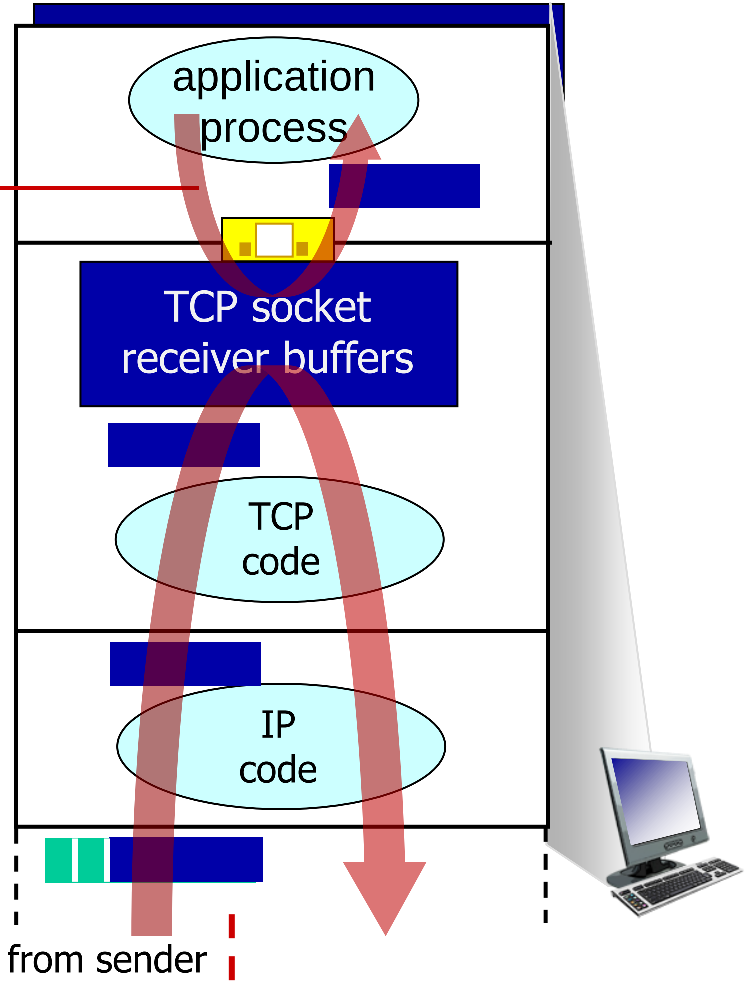
TCP flow control

<u>*Q:*</u> 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



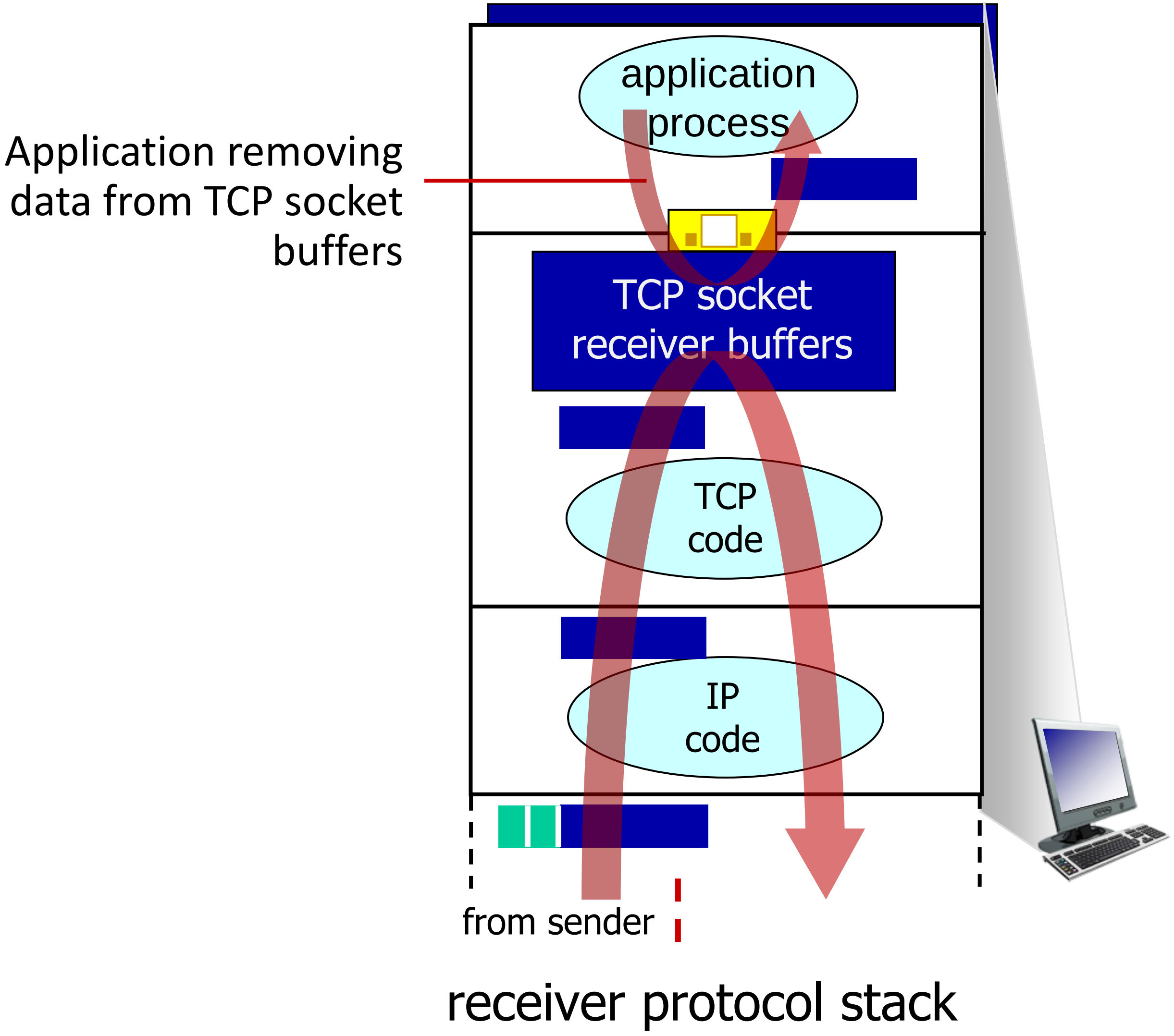
receiver protocol stack

TCP flow control

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

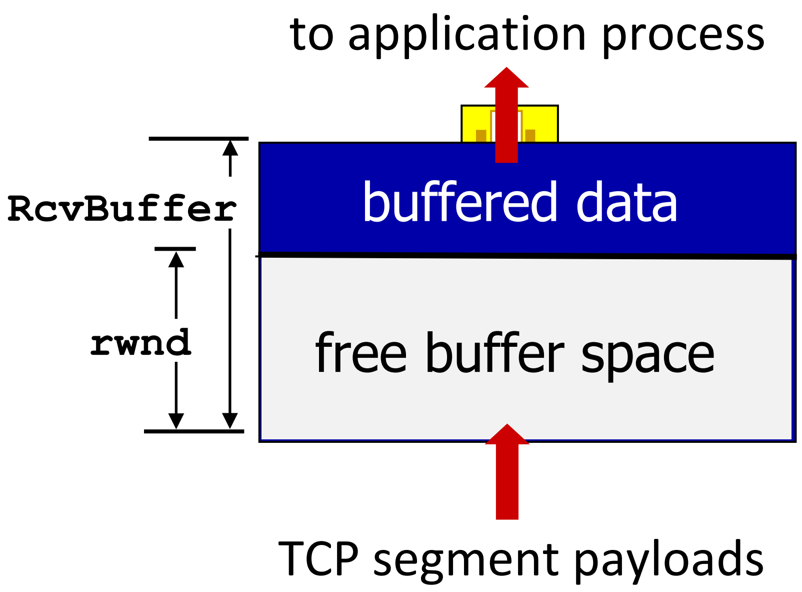
flow control

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



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 auto-adjust **`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 auto-adjust **`RcvBuffer`**
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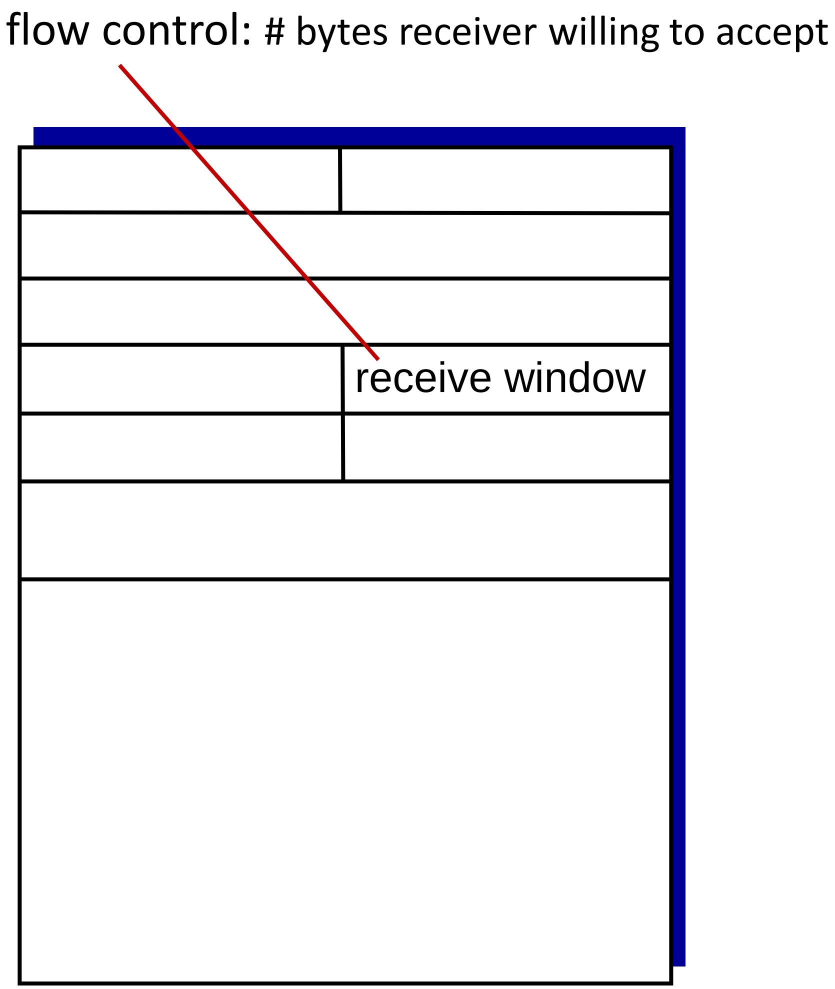
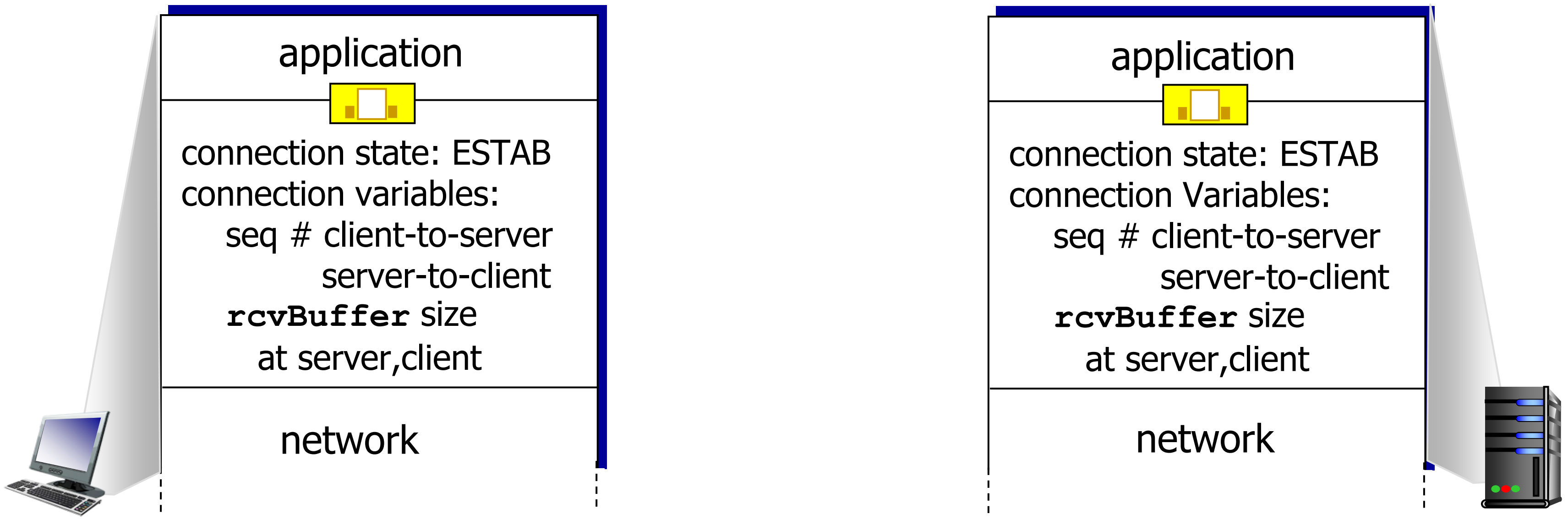


TCP segment format

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)

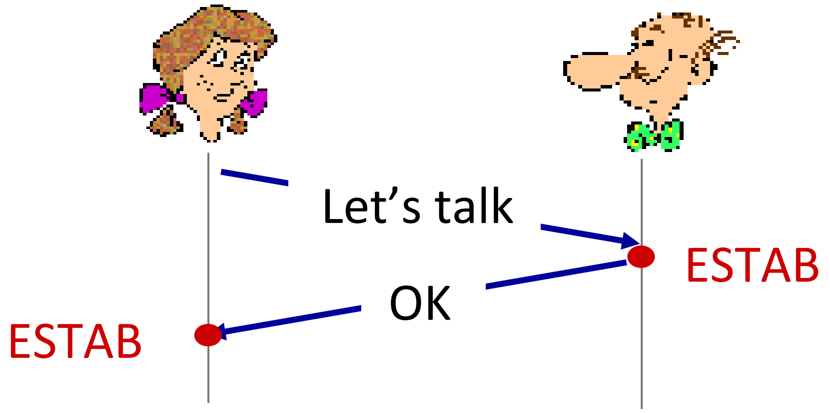


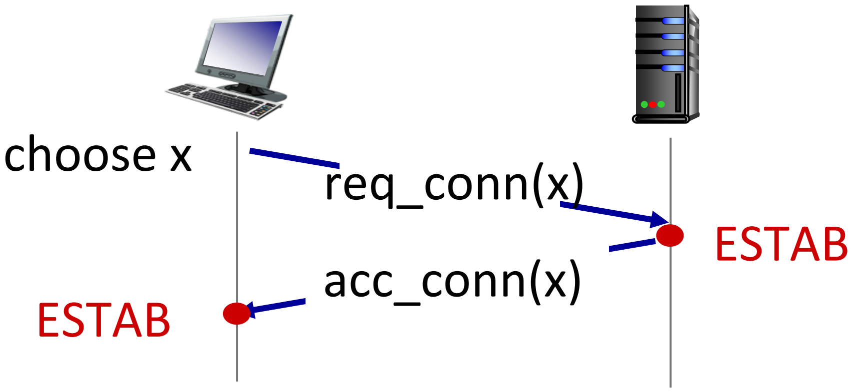

```
Socket clientSocket =
  newSocket("hostname","port number");
```

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

Agreeing to establish a connection

2-way handshake:

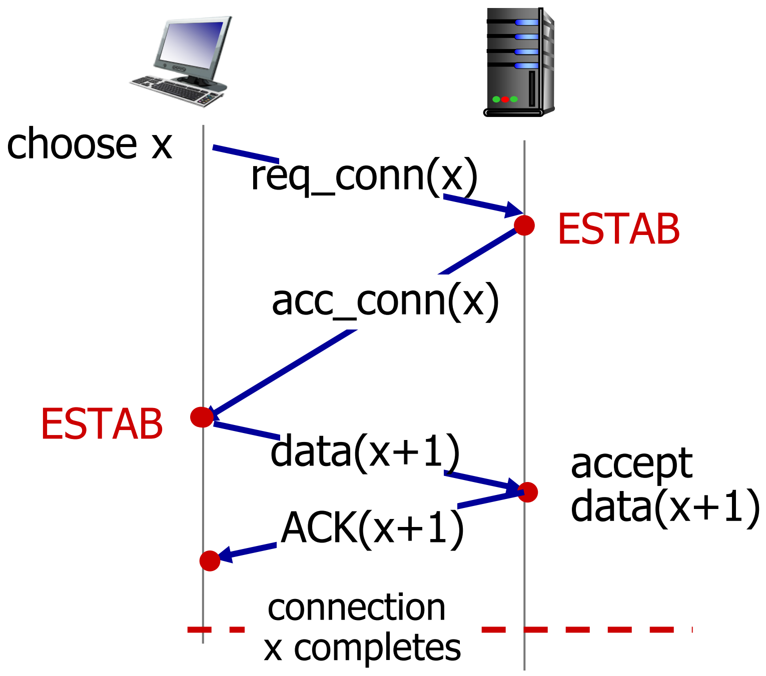




Q: will 2-way handshake always work in network?

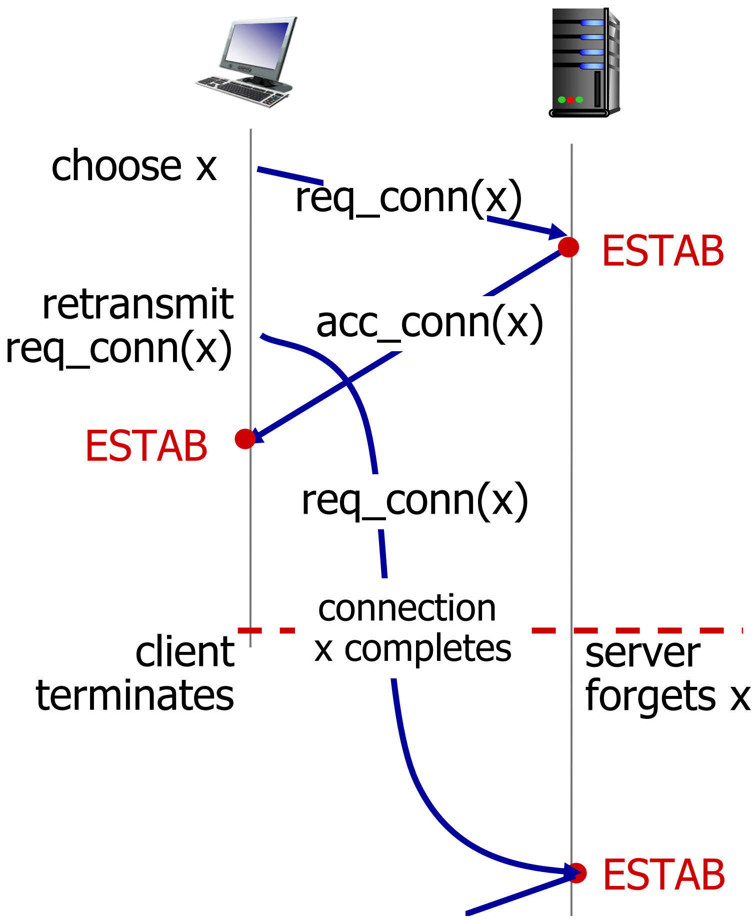
- variable delays
- retransmitted messages (e.g. req_conn(x)) due to message loss
- message reordering
- can't "see" other side

2-way handshake scenarios



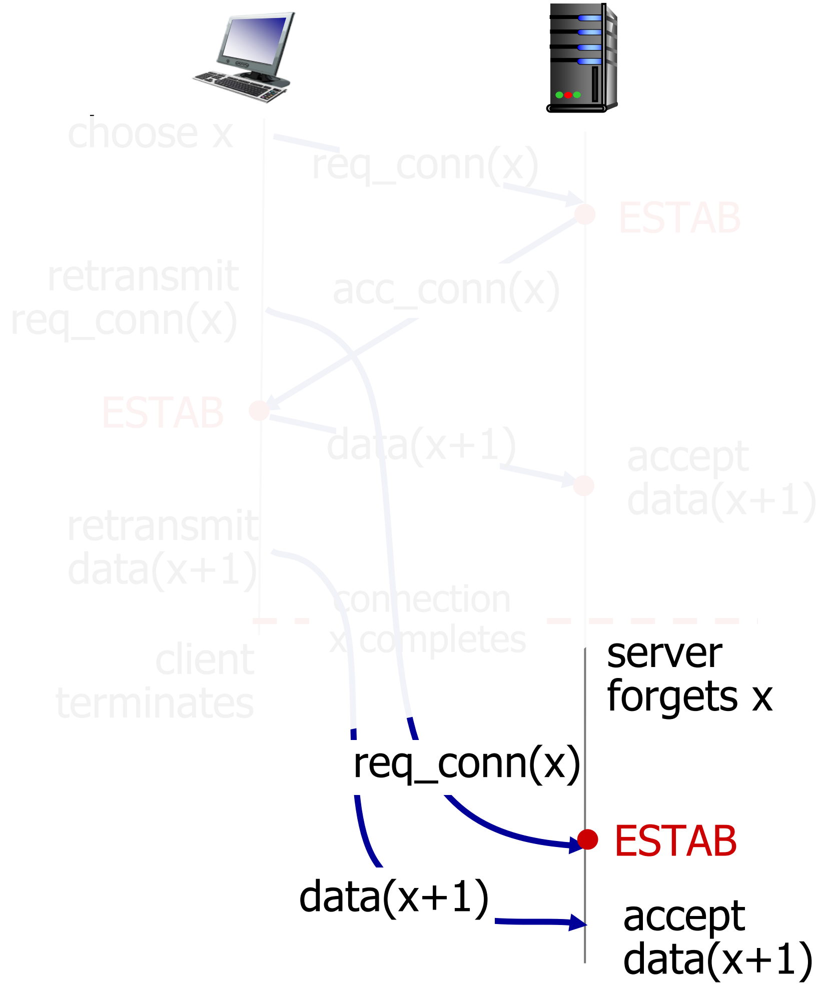
No problem!

2-way handshake scenarios



Problem: half open connection! (no client)

2-way handshake scenarios



Problem: dup data accepted!