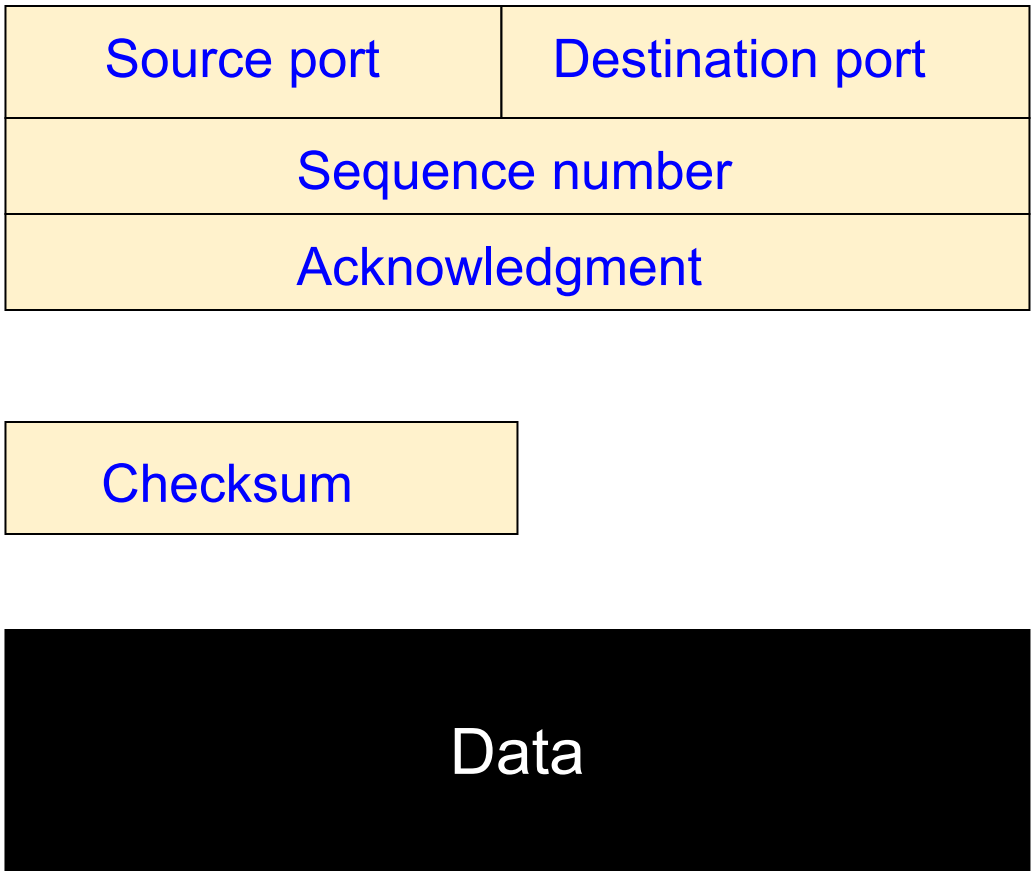
EECS 489
Computer Networks

Transport Control Protocol - TCP

TCP: Transmission Control Protocol

Build the TCP header



What does TCP do?

- Most of what we've seen
  - Checksum
  - Sequence numbers are byte offsets
  - Receiver sends cumulative acknowledgements (like GBN)
  - Receivers can buffer out-of-sequence packets (like SR)

What does TCP introduce?

- Most of what we've seen
  - Checksum
  - Sequence numbers are byte offsets
  - Receiver sends cumulative acknowledgements (like GBN)
  - Receivers buffer out-of-sequence packets (like SR)
- Introduces fast retransmit: duplicate ACKs trigger early retransmission
- Sender maintains a single retransmission timer (like GBN) and retransmits on timeout

Review from Discussion 1:

- Send message

```
do {
    ssize_t n = send(sockfd, message + sent, message_len - sent, 0);
    sent += n;
  } while (sent < message_len);
```

- Receive message

```
do {
    // Receive as many additional bytes as we can in one call to recv()
    // (while not exceeding MAX_MESSAGE_SIZE bytes in total).
    rval = recv(connectionfd, msg + recvd, MAX_MESSAGE_SIZE - recvd, 0);
    recvd += rval;
  } while (rval > 0); // recv() returns 0 when client closes
```

HTTP exchange example

- Client connects to a webserver and generates a GET request

GET / HTTP/1.1

Accept: text/html,application/xhtml+xml,application/xml;q=0.9,image/avif,image/webp,image/apng,*/*;q=0.8,application/signed-exchange;v=b3;q=0.7

Accept-Encoding: gzip, deflate

Accept-Language: en-US,en;q=0.9

Cache-Control: no-cache

Connection: keep-alive

Host: www.eecs489.org

Pragma: no-cache

Upgrade-Insecure-Requests: 1

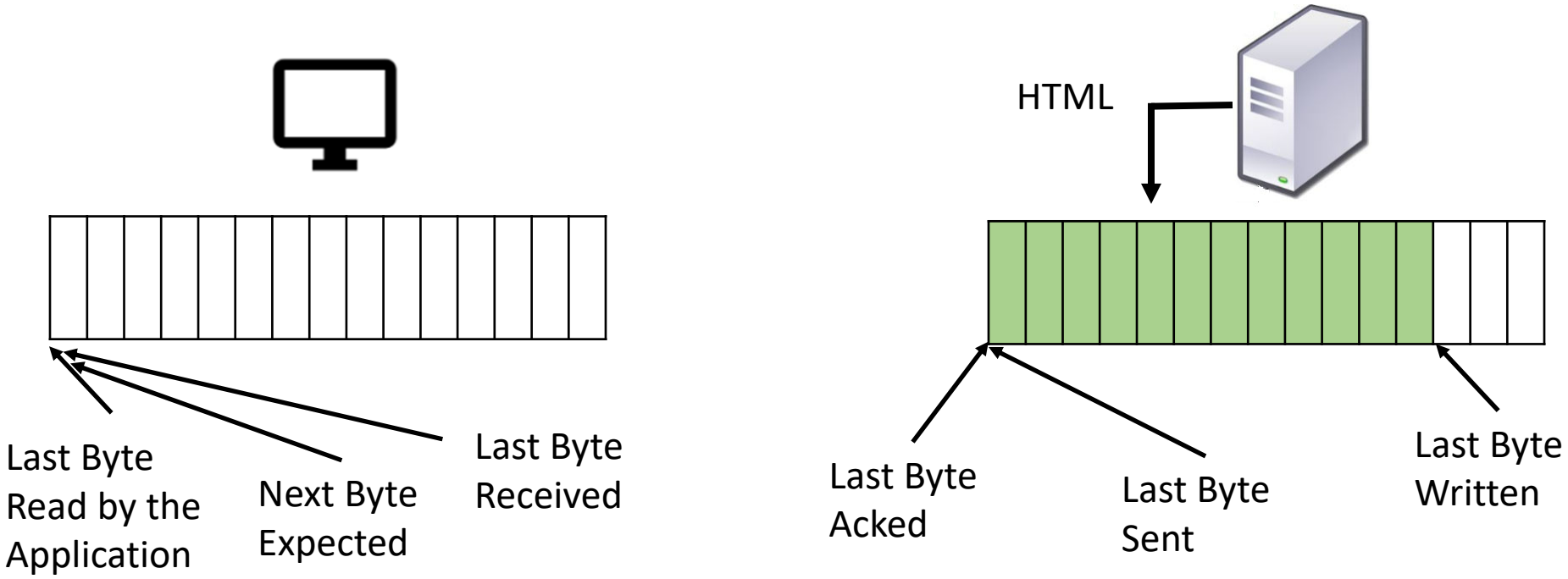
User-Agent: Mozilla/5.0 (Windows NT 10.0; Win64; x64) AppleWebKit/537.36 (KHTML, like Gecko) Chrome/116.0.0.0 Safari/537.36

HTTP exchange example

- Server sends a response

HTTP/1.1 200 OK
Connection: keep-alive
Content-Length: 5000
Server: GitHub.com
Content-Type: text/html; charset=utf-8
Last-Modified: Wed, 20 Sep 2024 17:31:52 GMT
… (Total of 5,000 bytes)

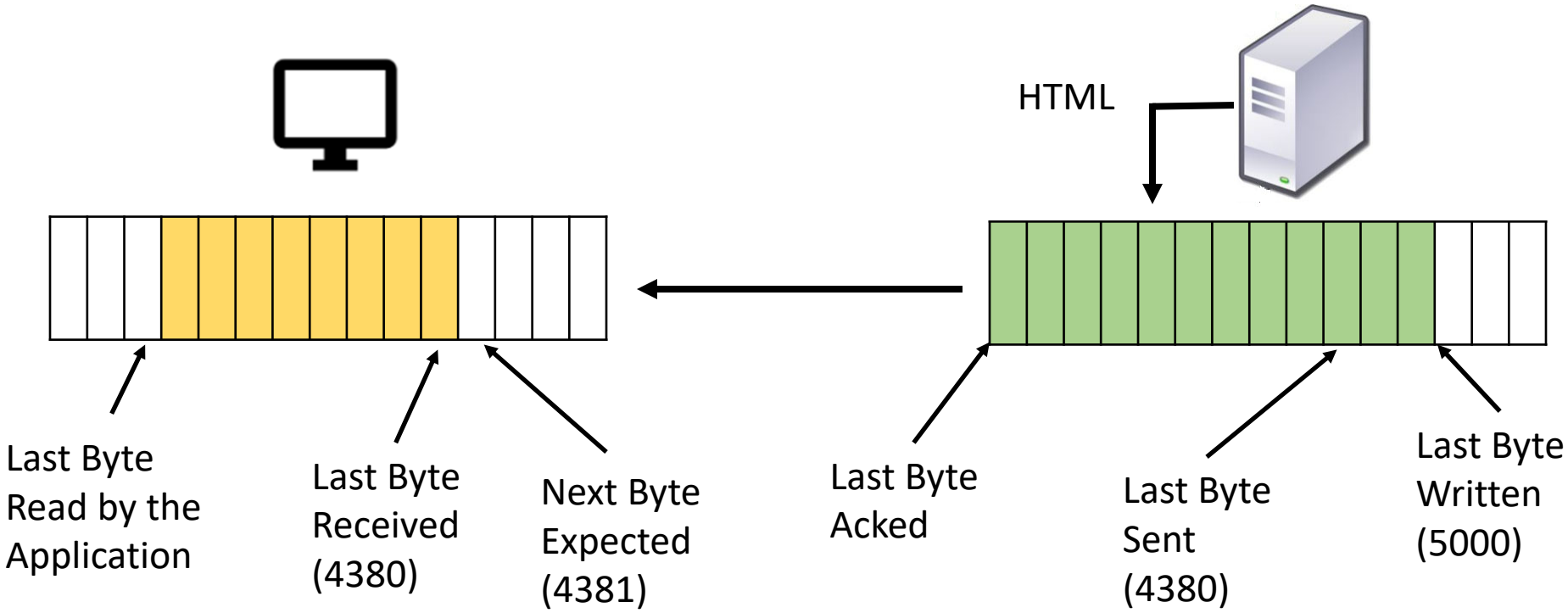
Now the server is going to send the webpage



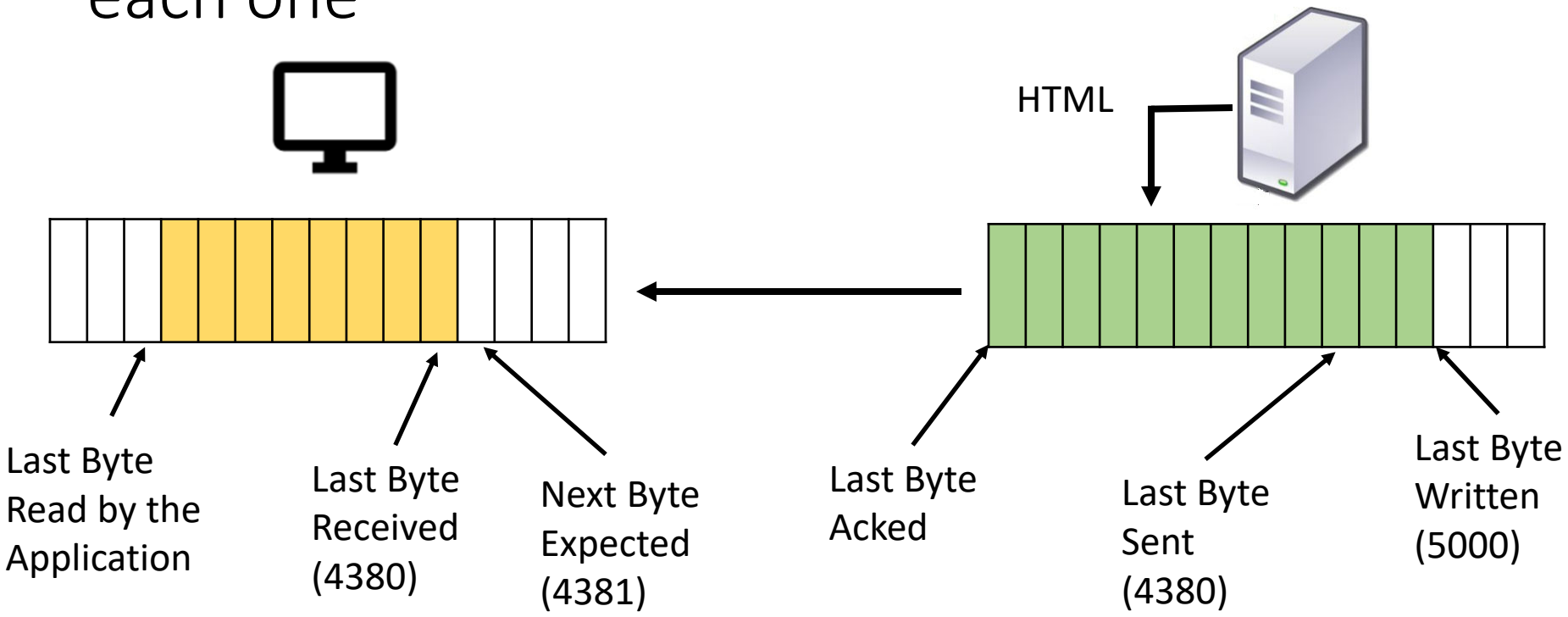
How much data can the server send

- Let's say for each MTU (Maximum Transmission Unit) is 1500 Bytes
- IP Header and TCP headers are 20 bytes each.
- Each TCP Packet can hold 1500 - 20 - 20 = 1460 Bytes
- So TCP can transmit 1460 bytes of data in each packet
- Server's Sliding window is 5000 Bytes

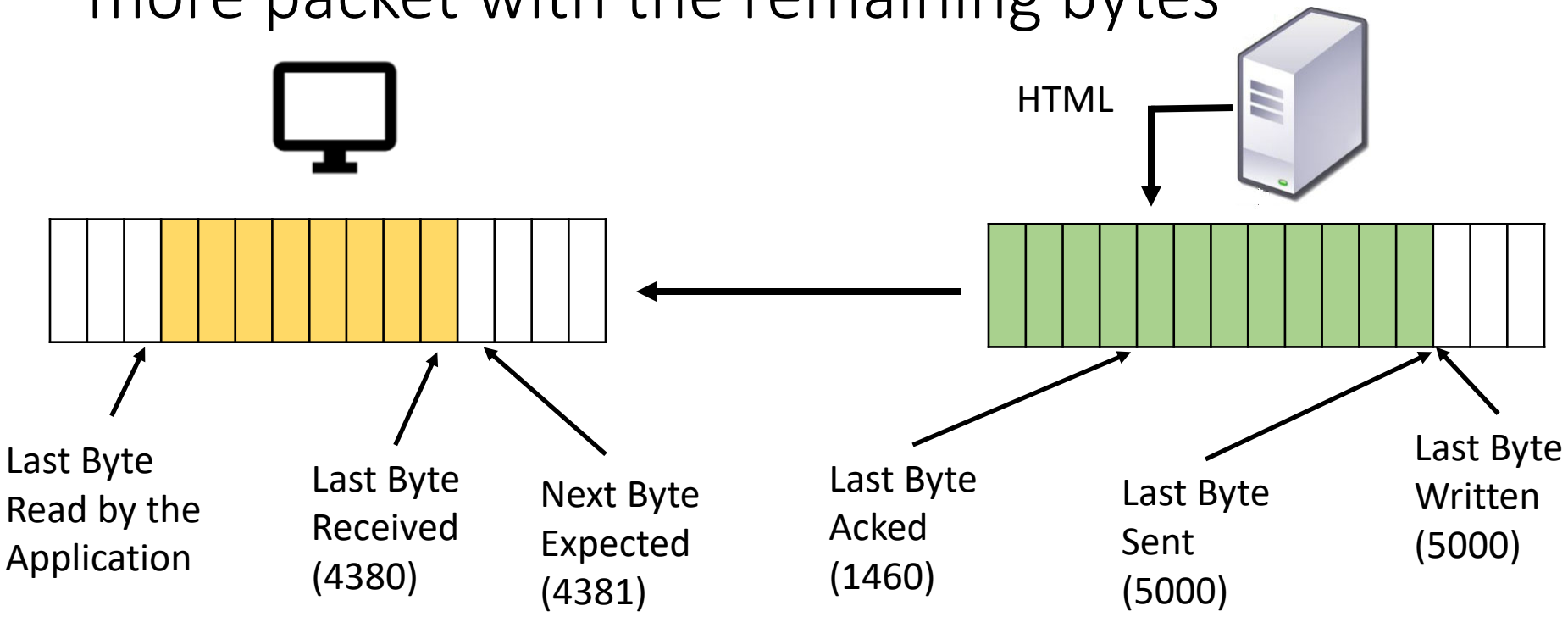
Server sends 3 packets, each with 1460 bytes



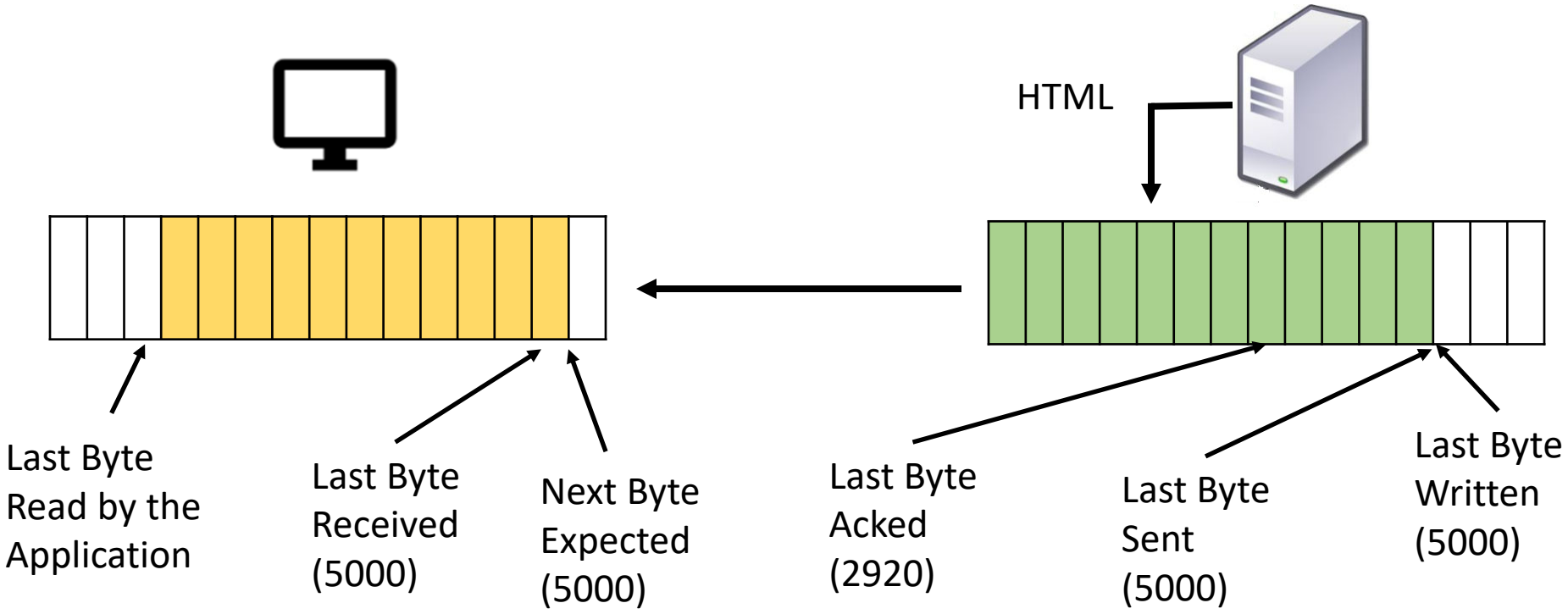
Host receives the three packets and ACKS each one



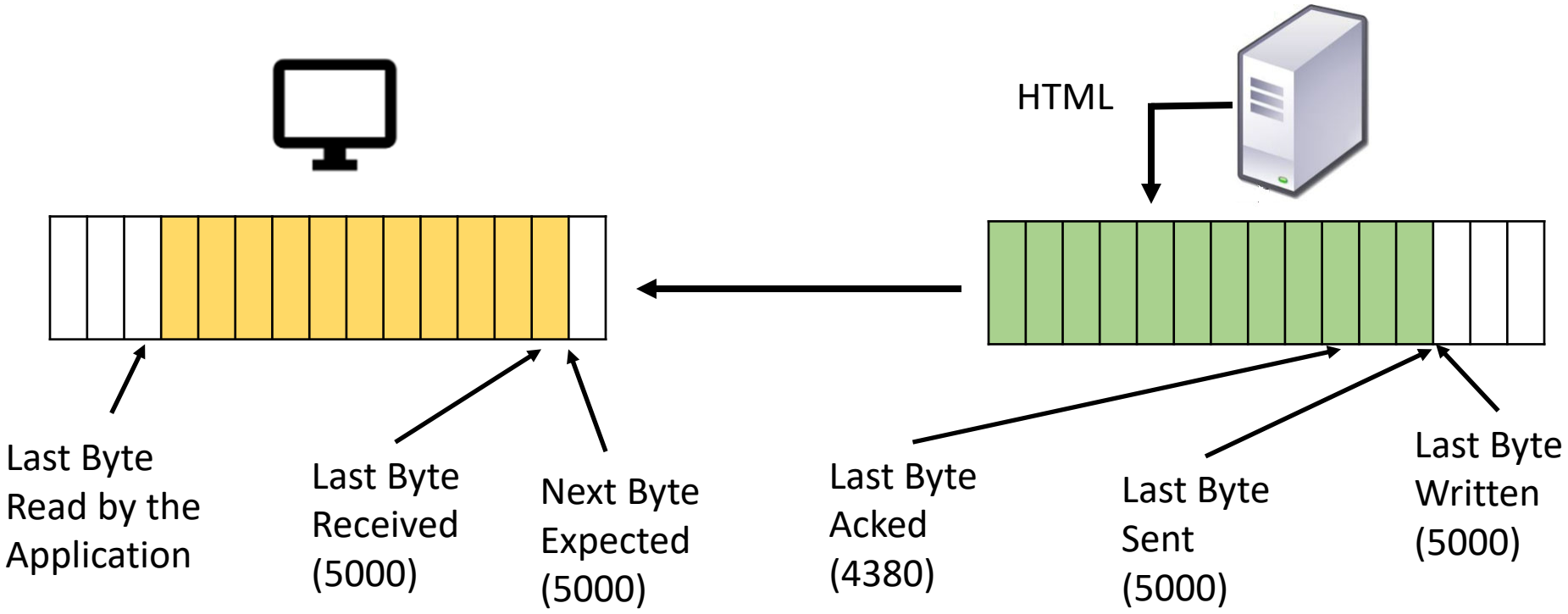
Server after receiving first ACK – send one more packet with the remaining bytes



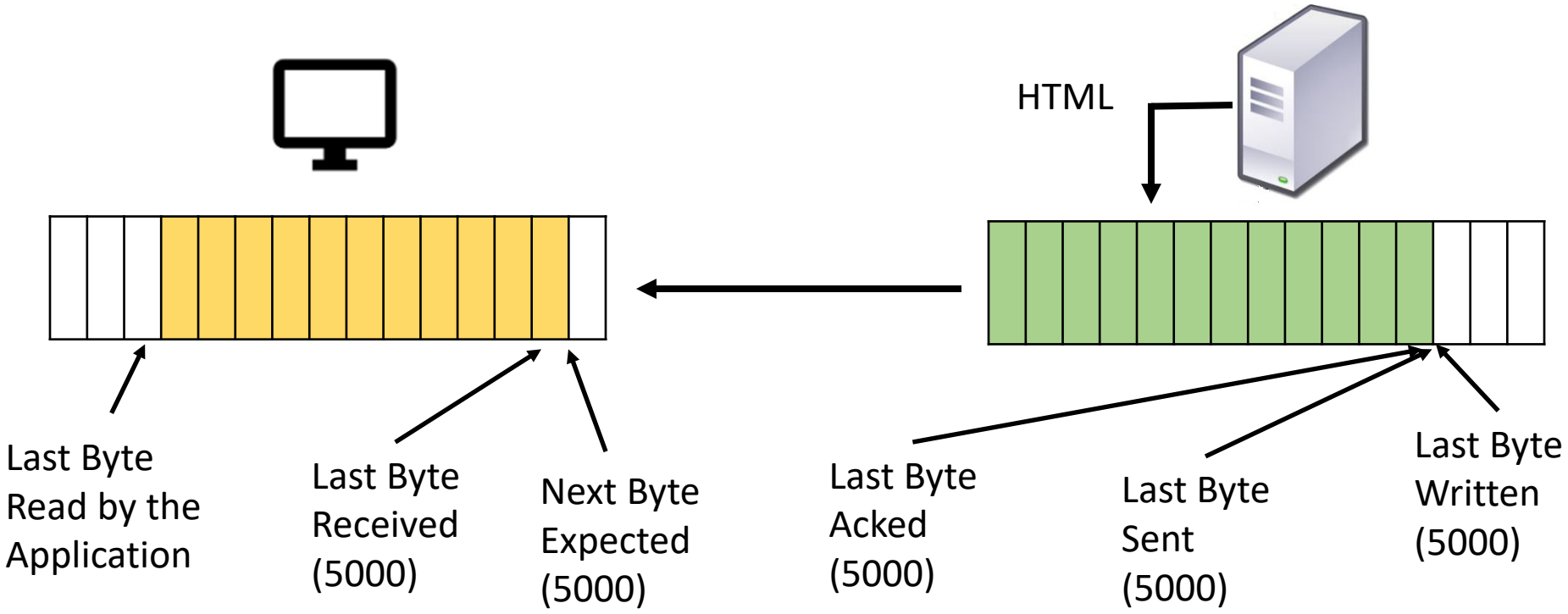
Server after receiving second ACK



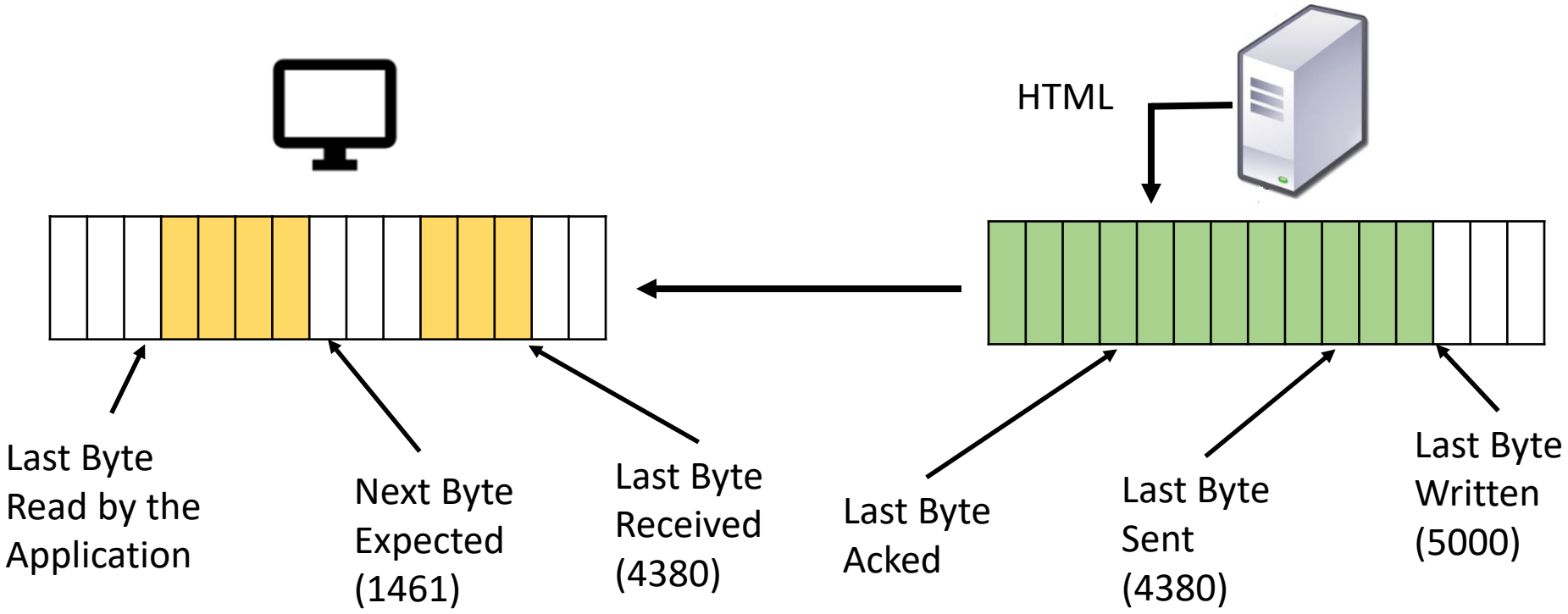
Server after receiving third ACK



Server after receiving fourth ACK



Out of Order Delivery/Missing packet

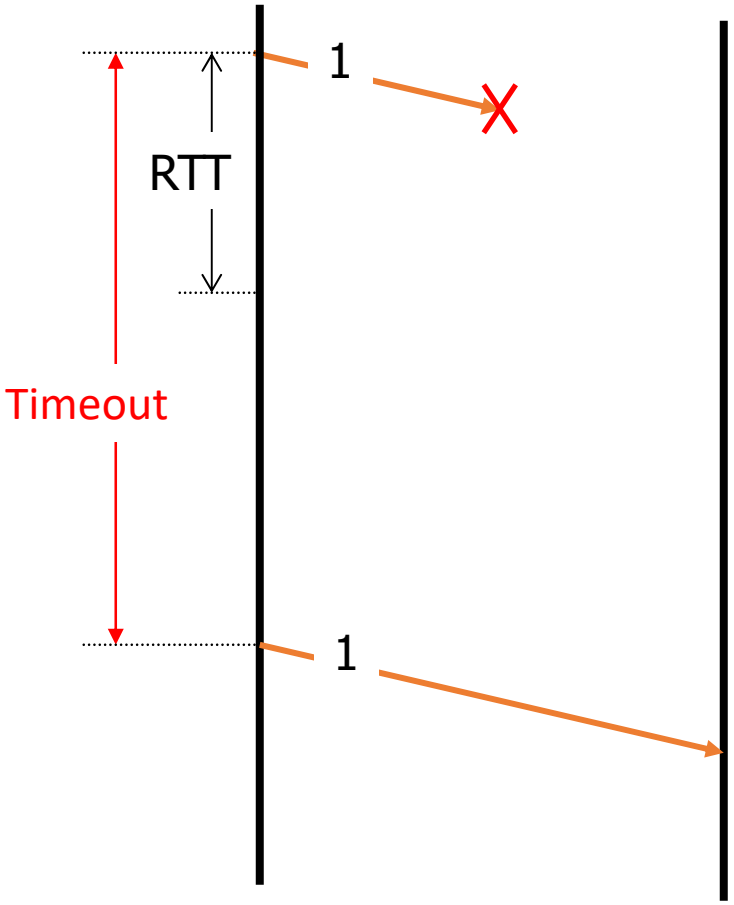


- Host Acks with 1461 on Receiving Bytes 2920 – 4380
- Server retransmits Bytes 1461 – 2919 on receiving the above ACK or on a timeout

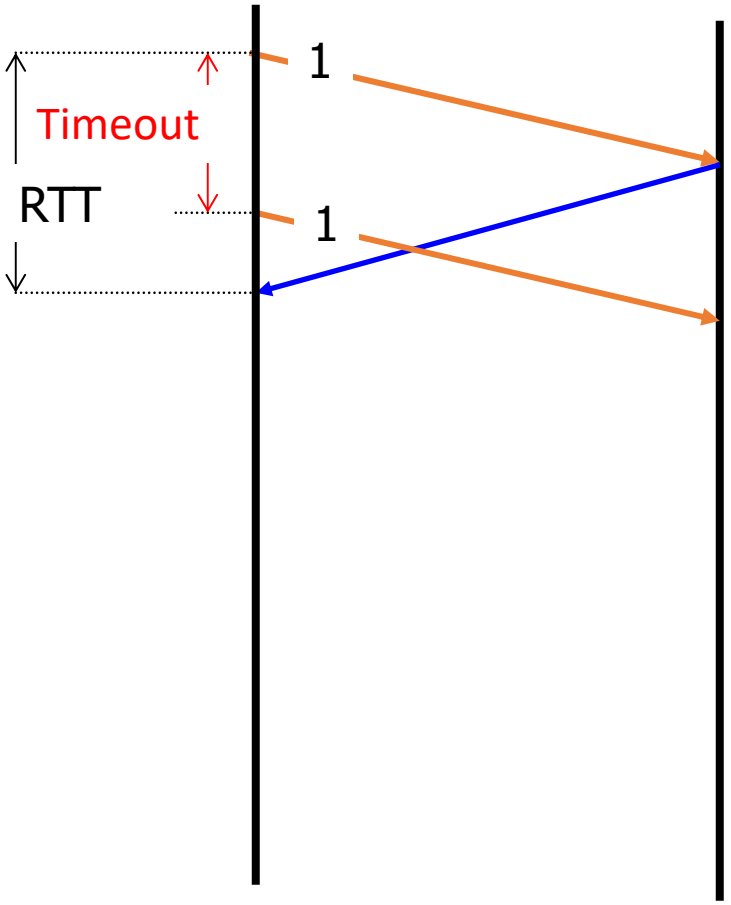
Retransmission timeout

- If the sender hasn't received an ACK by timeout, retransmit the first packet in the window
- How do we pick a timeout value?

Timing illustration



Timeout too long → inefficient



Timeout too short → duplicate packets

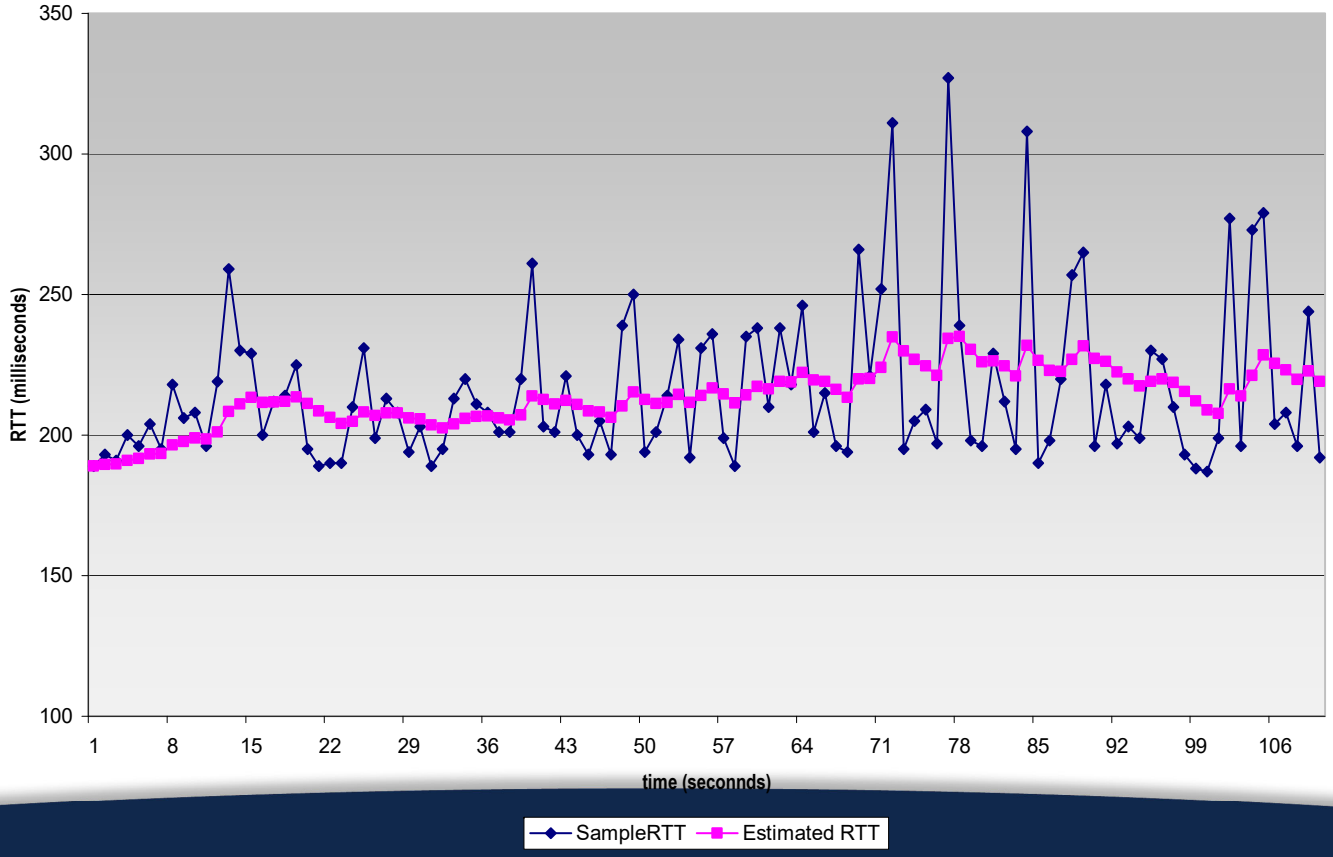
Retransmission timeout

- If the sender hasn't received an ACK by timeout, retransmit the first packet in the window
- How to set timeout?
  - Too long: connection has low throughput
  - Too short: retransmit packet that was just delayed
- Solution: make timeout proportional to RTT
  - But how do we measure RTT?

RTT estimation

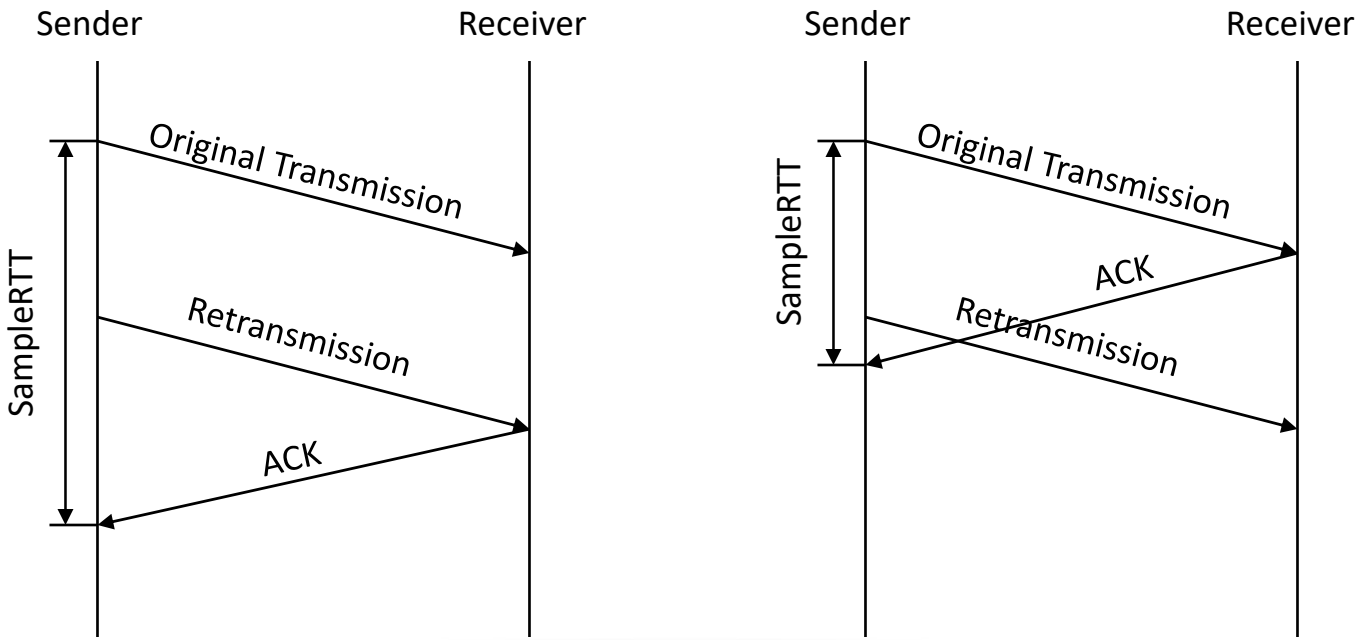
- Exponential weighted average of RTT samples

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



Problem: Ambiguous measurements

- How do we differentiate between the real ACK, and ACK of the retransmitted packet?



Karn/Partridge algorithm

- Don't use SampleRTT from retransmissions
  - Once retransmitted, ignore that segment in the future
- Computes EstimatedRTT using $\alpha = 0.125$
- Timeout value (RTO) = 2 × EstimatedRTT
  - Employs exponential backoff
    - Every time RTO timer expires, set RTO $\leftarrow$ 2·RTO

      (Up to maximum $\geq$ 60 sec)
    - Every time new measurement comes in (= successful original transmission), collapse RTO back to 2 × EstimatedRTT
- Sensitive to RTT variations

Jacobson/Karels algorithm

- Problem: need to better capture variability in RTT
  - Directly measure deviation

$$\text{Difference} = \text{SampleRTT} - \text{EstimatedRTT}$$

$$\text{EstimatedRTT} = \text{EstimatedRTT} + (\delta \times \text{Difference})$$

$$\text{Deviation} = \text{Deviation} + \delta(|\text{Difference}| - \text{Deviation})$$

$$\text{TimeOut} = \mu \times \text{EstimatedRTT} + \phi \times \text{Deviation}$$

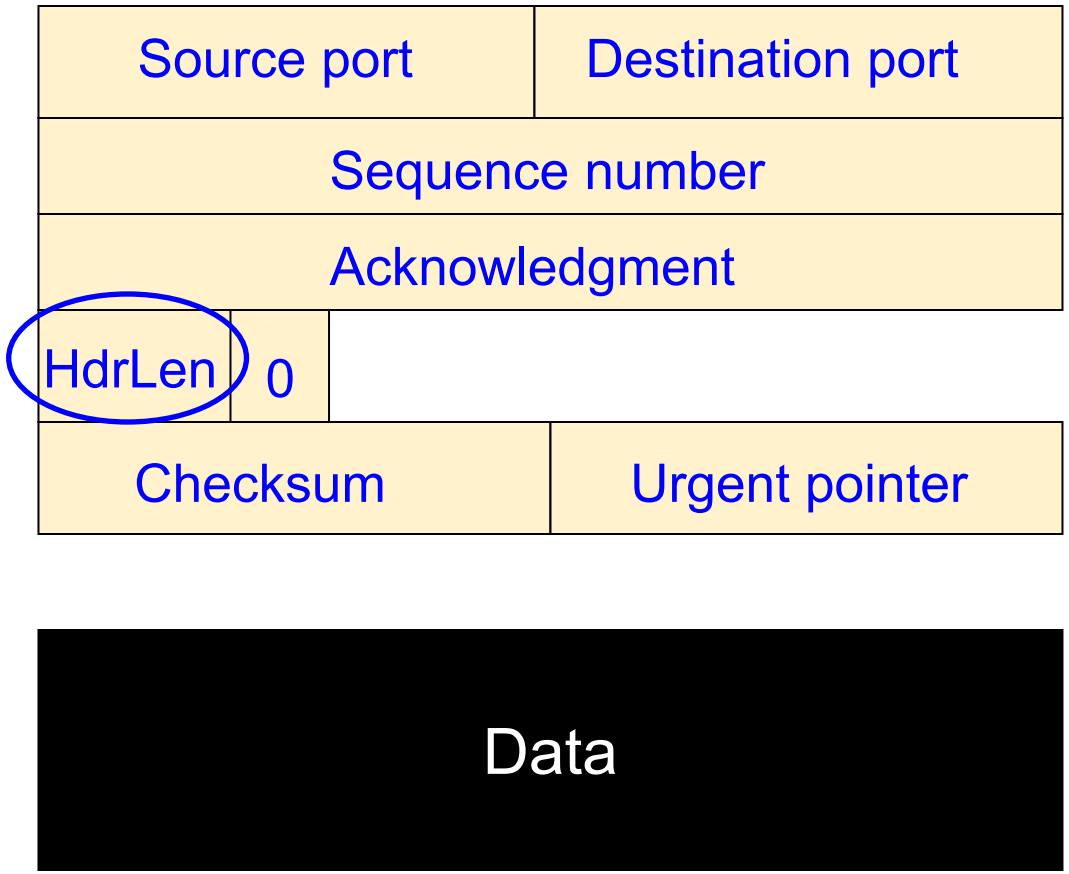
μ is typically set to 1 and ϕ is set to 4*

- RTO = EstimatedRTT + 4 x DeviationRTT

*Peterson and Davie

Build the TCP header

Number of 4-byte words in the header; 5: No options



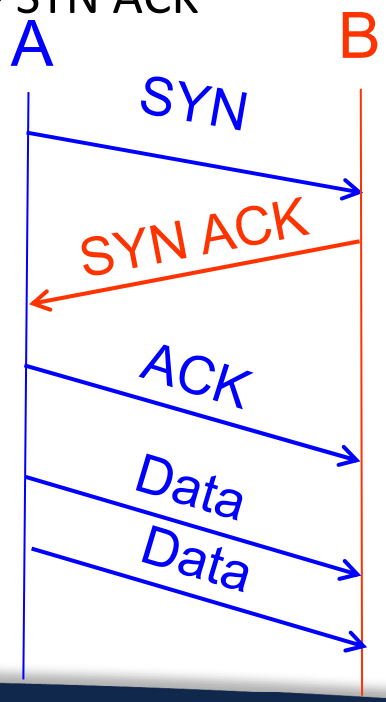
TCP Connection Establishment

Initial Sequence Number (ISN)

- Sequence number for the very first byte
- Why not just use ISN = 0?
  - Practical issue
    - IP addresses and port #s uniquely identify a connection
    - Eventually, though, these port #s do get used again; small chance an old packet is still in flight
    - Also, others might try to spoof your connection
  - Why does using ISN help?
- Hosts exchange ISNs when establishing connection

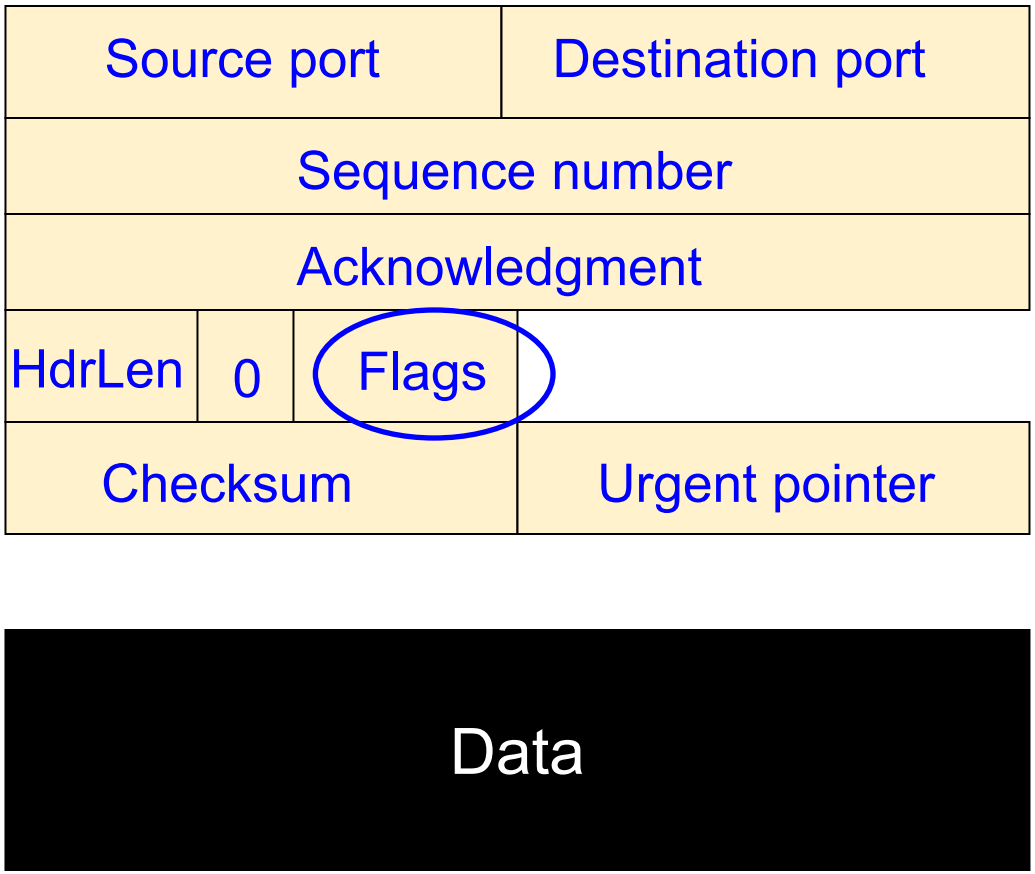
Establishing a TCP connection

- Three-way handshake to establish connection
  - Host A sends a SYN (open; "synchronize sequence numbers") to host B
  - Host B returns a SYN acknowledgment (SYN ACK)
  - Host A sends an ACK to acknowledge the SYN ACK



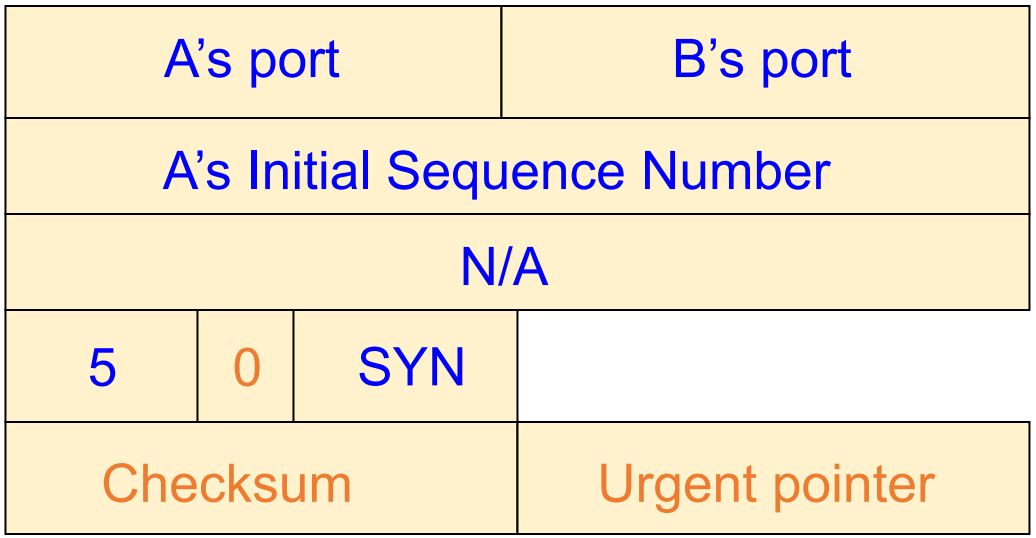
Build the TCP header

Flags:
SYN
ACK
FIN
RST
PSH
URG



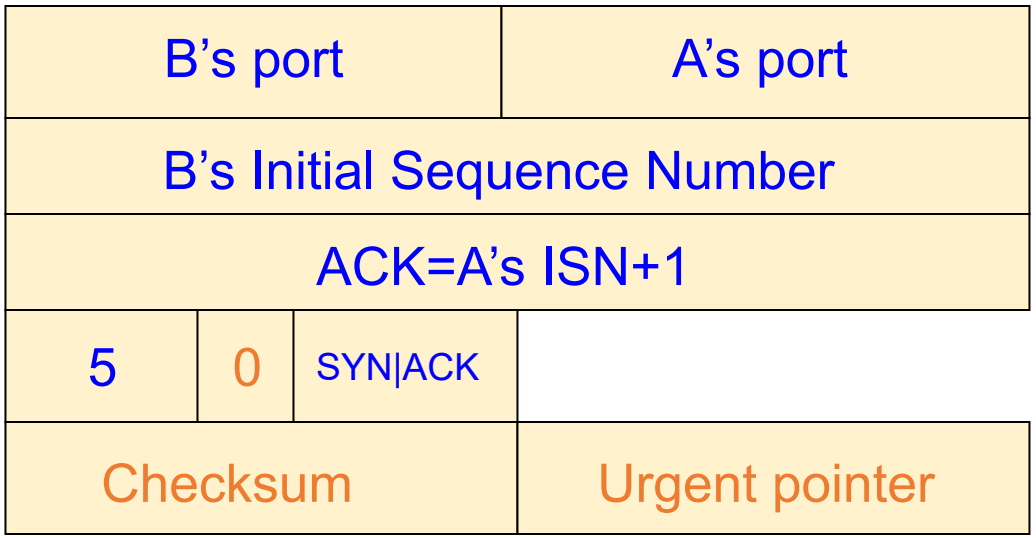
Step 1: A's initial SYN packet

A tells B to open a connection



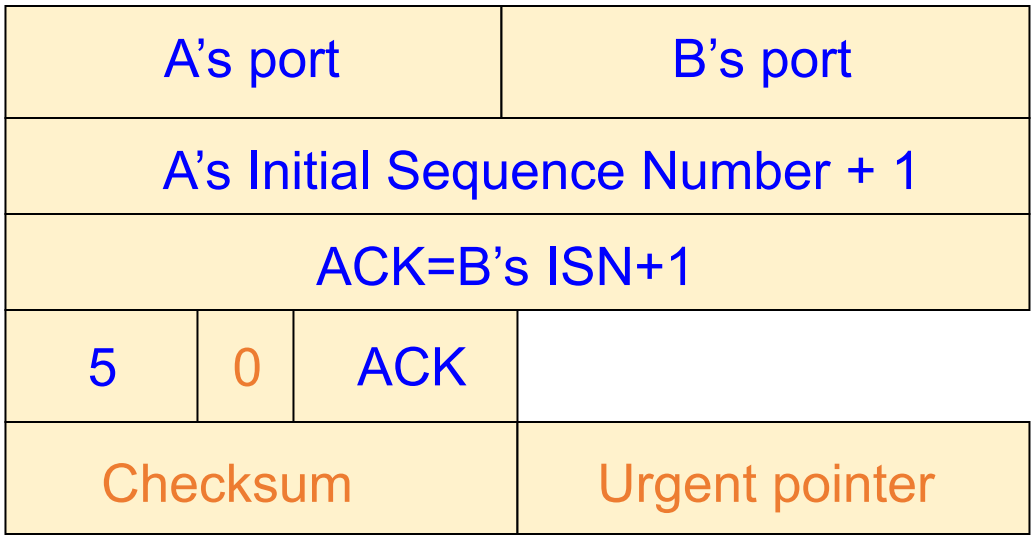
Step 1: B's SYN-ACK packet

B tells it accepts and is ready to accept next packet

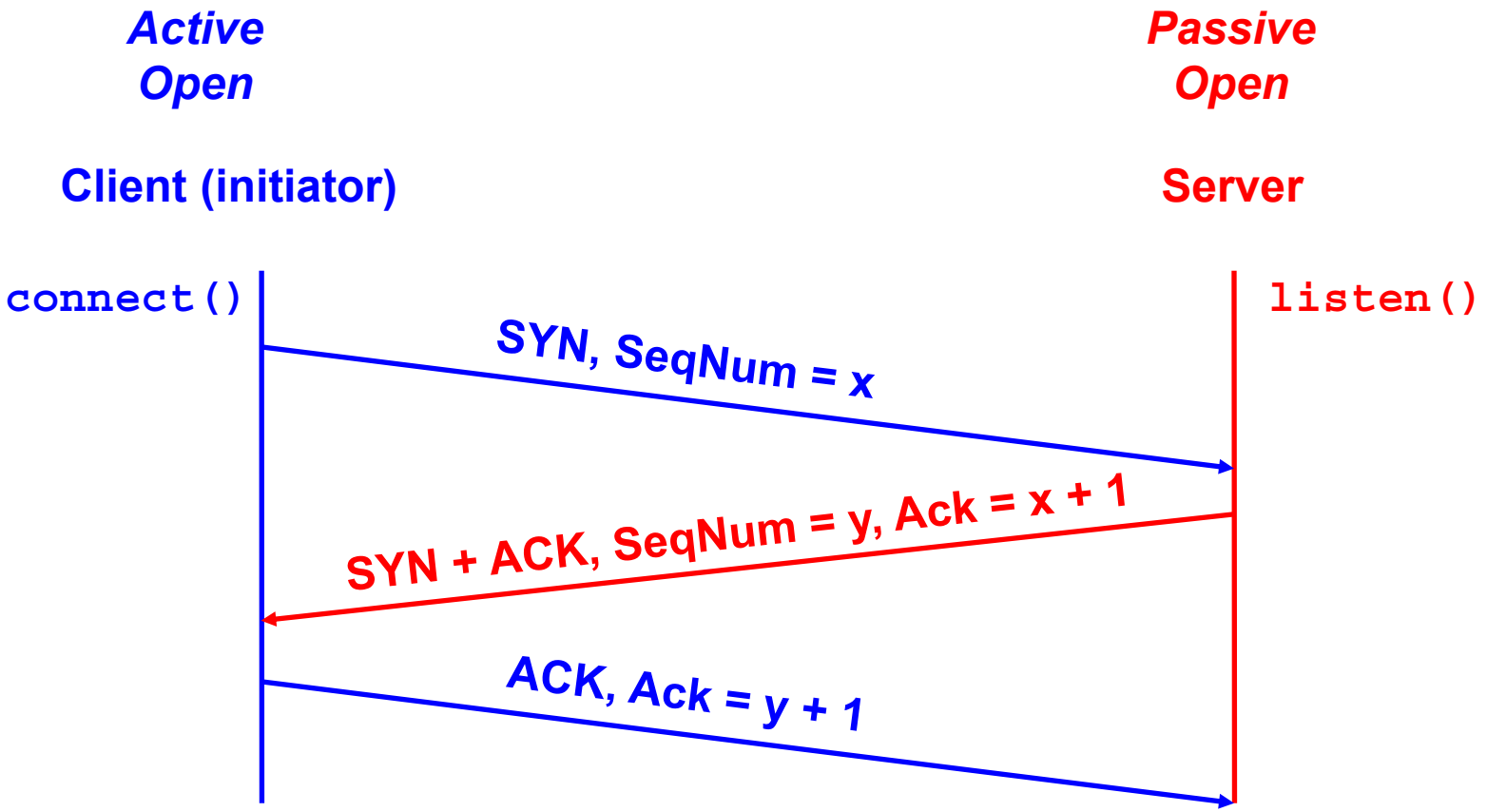


Step 1: A's ACK to SYN-ACK

A tells B to open a connection



TCP's 3-Way handshaking



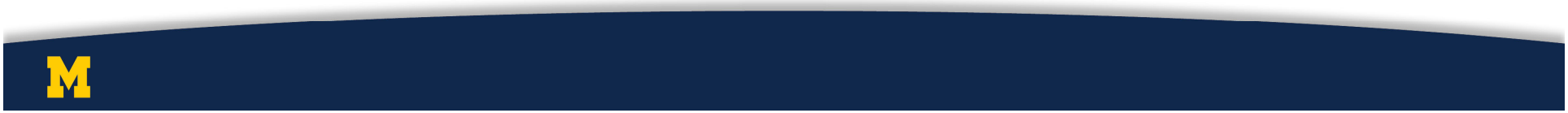
What if the SYN Packet Gets Lost?

- Suppose the SYN packet gets lost
  - Packet dropped by the network or server is busy
- Eventually, no SYN-ACK arrives
  - Sender retransmits the SYN on timeout
- How should the TCP sender set the timer?
  - Sender has no idea how far away the receiver is
  - Hard to guess a reasonable length of time to wait
  - SHOULD (RFCs 1122 & 2988) use default of 3 seconds
    - Some implementations instead use 6 seconds

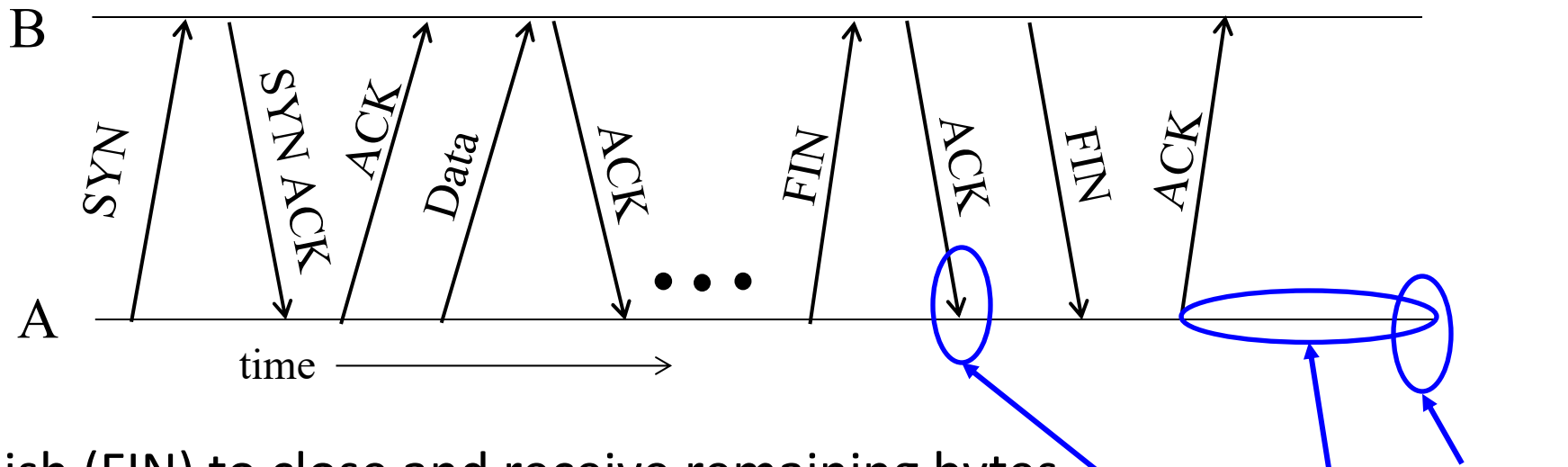
SYN loss and web downloads

- User clicks on a hypertext link
  - Browser creates a socket and does a "connect"
  - The "connect" triggers the OS to transmit a SYN
- If the SYN is lost…
  - 3-6 seconds of delay: can be very long
  - User may become impatient and can retry
- User triggers an "abort" of the "connect"
  - Browser creates a new socket and another "connect"
  - Can be effective in some cases

TCP connection teardown

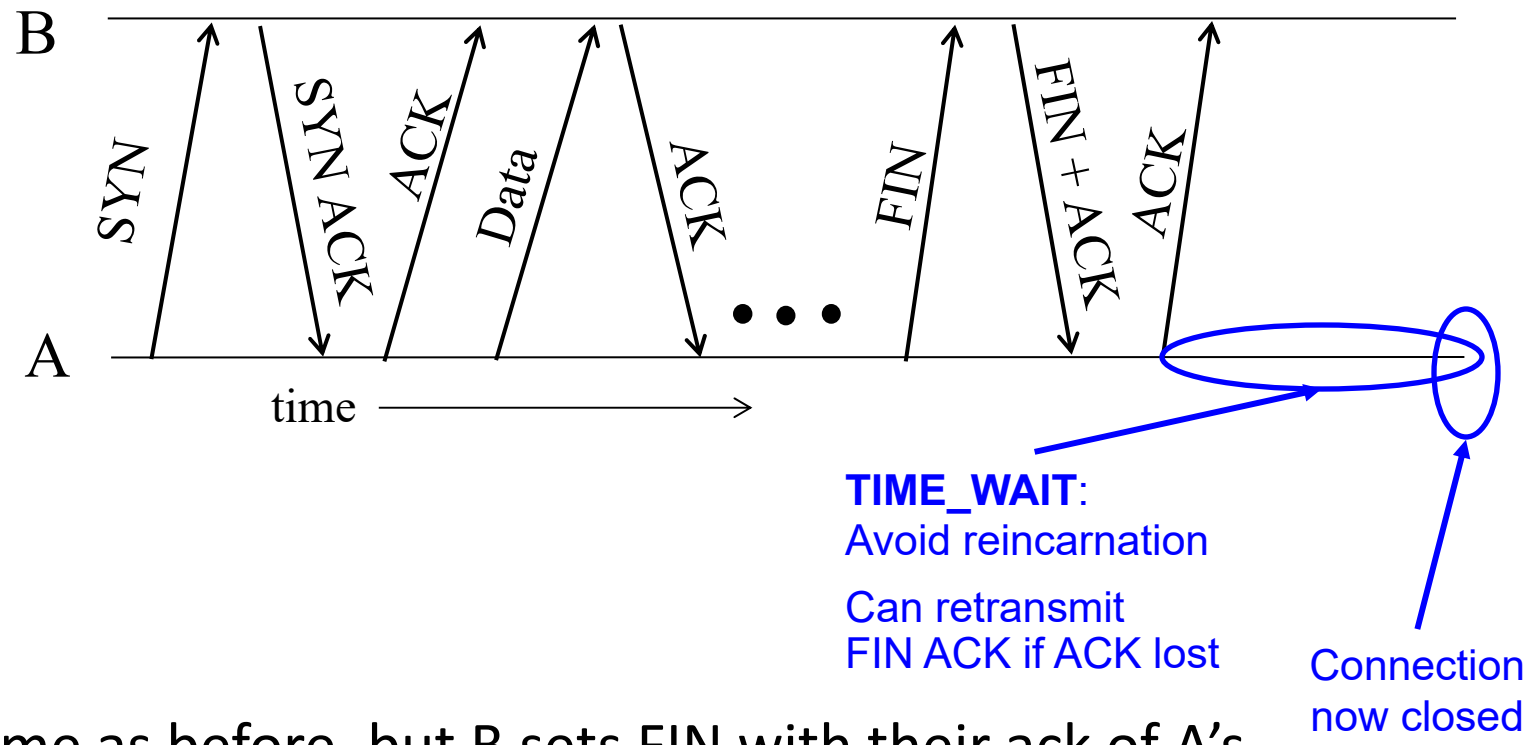


Normal termination, one side at a time



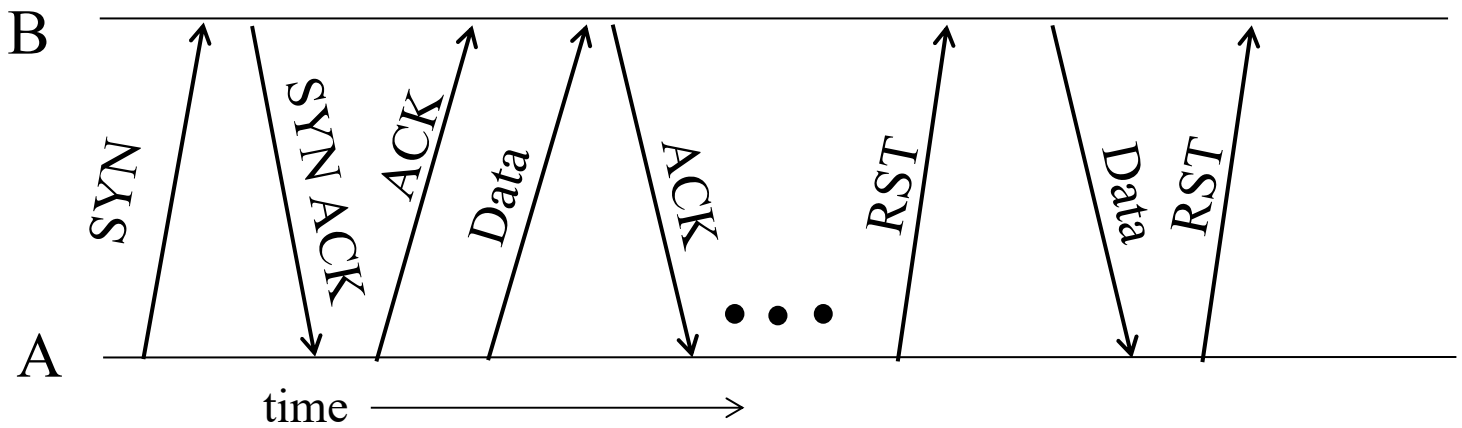
- Finish (FIN) to close and receive remaining bytes
  - FIN occupies one byte in the sequence space
- Other host acks the byte to confirm
- Closes A's side of the connection, but not B's
  - Until B likewise sends a FIN
  - Which A then acks

Normal termination, both together



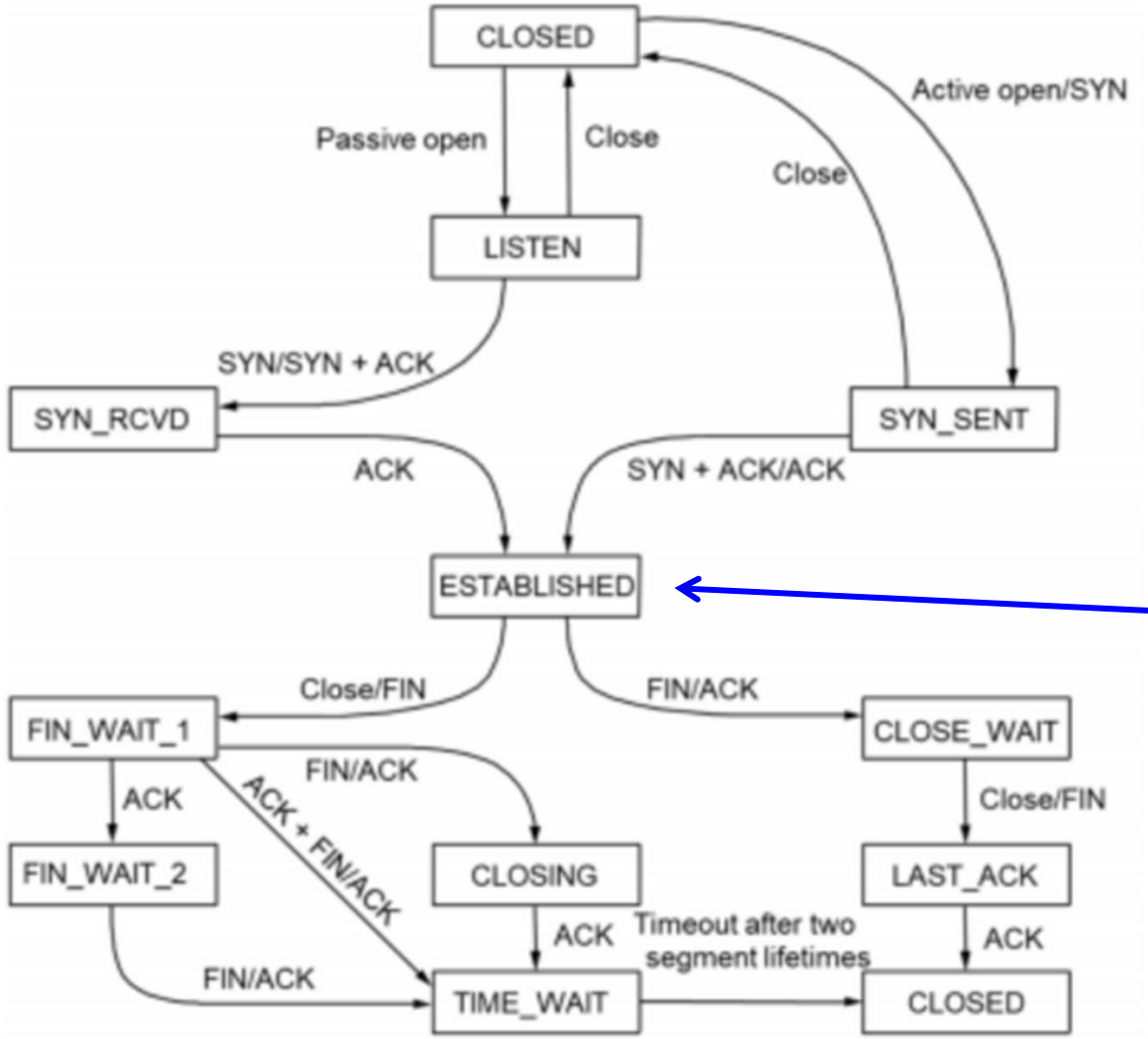
- Same as before, but B sets FIN with their ack of A's FIN

Abrupt termination



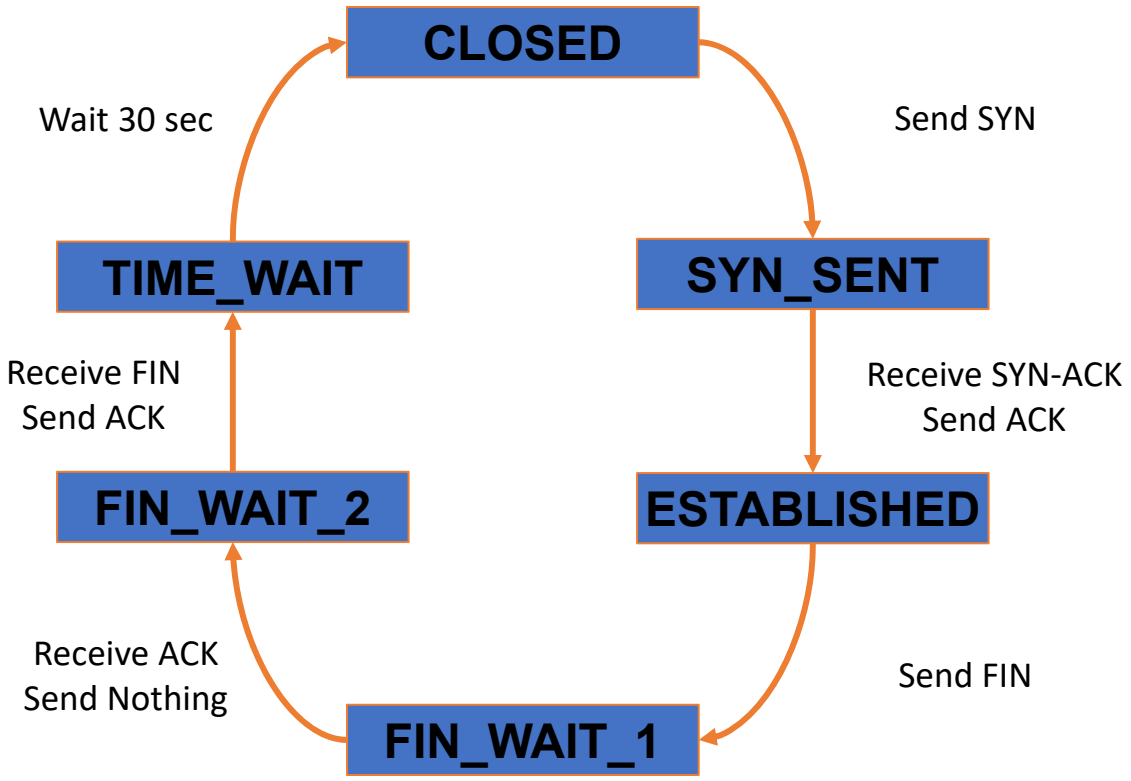
- A sends a RESET (RST) to B
  - E.g., because application process on A crashed
- That's it
  - B does not ack the RST
  - Thus, RST is not delivered reliably, and any data in flight is lost
  - But: if B sends anything more, will elicit another RST

TCP State Transition



Data, ACK exchanges are in here

TCP client lifecycle



HTTP exchange example

- Client connects to a webserver and generates a GET request

GET / HTTP/1.1

Accept: text/html,application/xhtml+xml,application/xml;q=0.9,image/avif,image/webp,image/apng,*/*;q=0.8,application/signed-exchange;v=b3;q=0.7

Accept-Encoding: gzip, deflate

Accept-Language: en-US,en;q=0.9

Cache-Control: no-cache

Connection: keep-alive

Host: www.eecs489.org

Pragma: no-cache

Upgrade-Insecure-Requests: 1

User-Agent: Mozilla/5.0 (Windows NT 10.0; Win64; x64) AppleWebKit/537.36 (KHTML, like Gecko) Chrome/116.0.0.0 Safari/537.36

HTTP exchange example

- Server sends a response

HTTP/1.1 200 OK
Connection: keep-alive
Content-Length: 10257
Server: GitHub.com
Content-Type: text/html; charset=utf-8
Last-Modified: Wed, 20 Sep 2023 17:31:52 GMT
… (Total of 53,568 bytes)

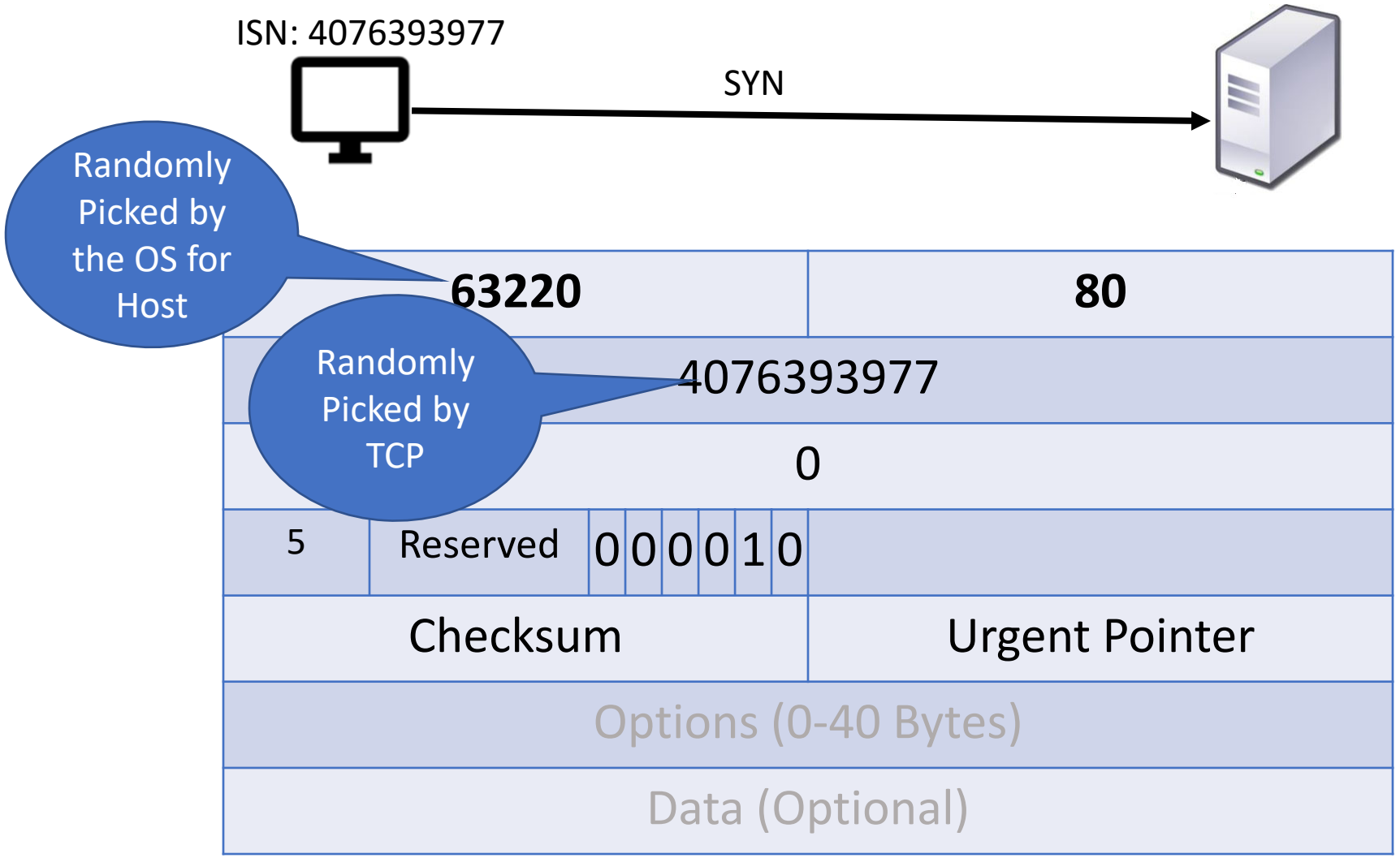
TCP Message Header

U	A	P	R	S	F

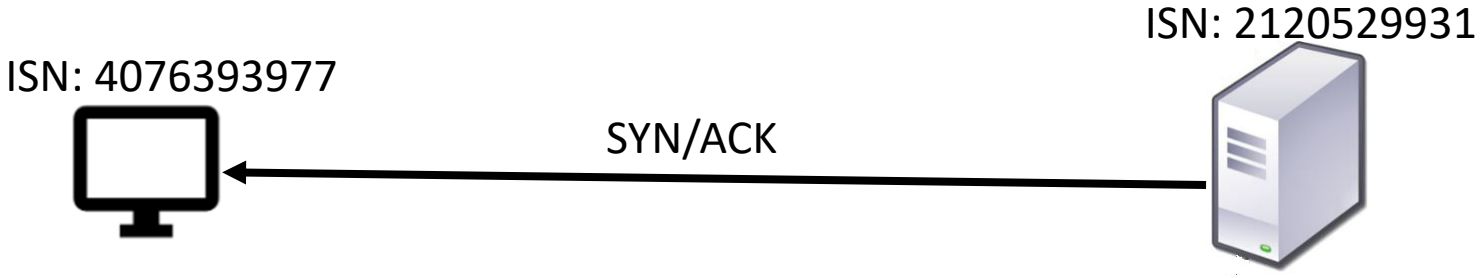
Source Port								Destination Port
Sequence Number								
Acknowledgement								
Length	Reserved	U	A	P	R	S	F	
Checksum								Urgent Pointer
Options (0-40 Bytes)								
Data (Optional)								

Application says connect to server

UAPRSF



Application says connect to server



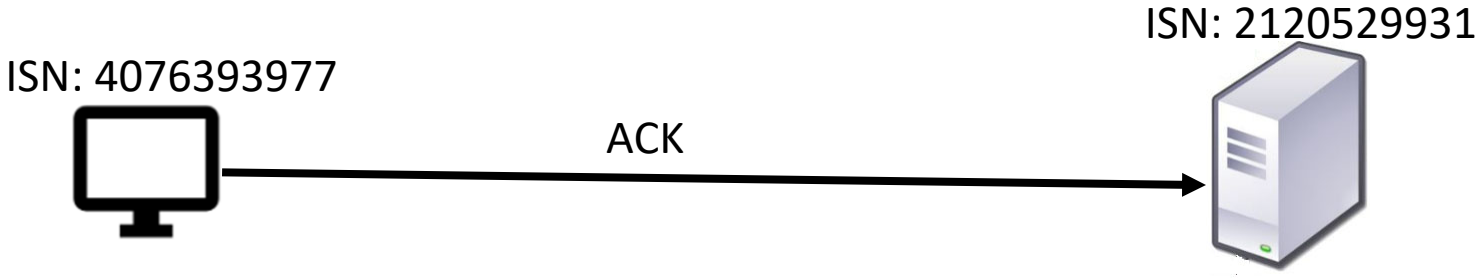
80	63220
2120529931	
4076393978	
ed 0 1 0 0 1 0	
ecksum	Urgent Pointer
Options (0-40 Bytes)	
Data (Optional)	

I have received all packets before this Seq Num

TCP on the server picks an ISN

Application says connect to server

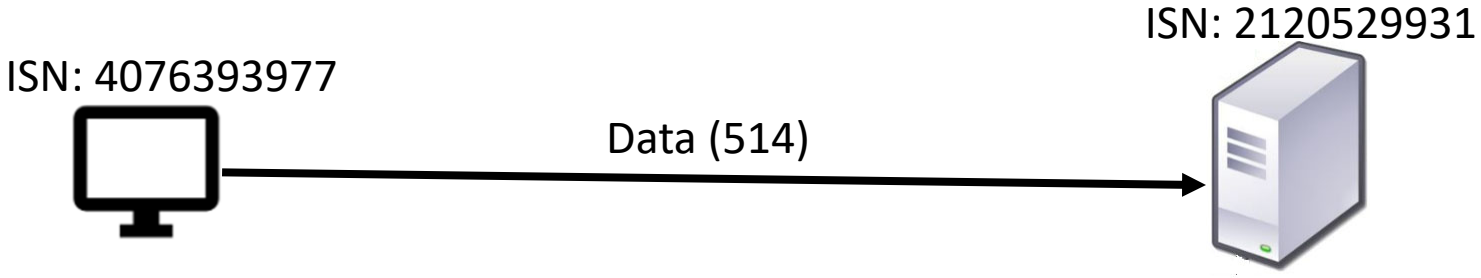
UAPRSF



63220									**80**
4076393978									
2120529932									
5	Reserved	0	1	0	0	0	0		
Checksum									Urgent Pointer
Options (0-40 Bytes)									
Data (Optional)									

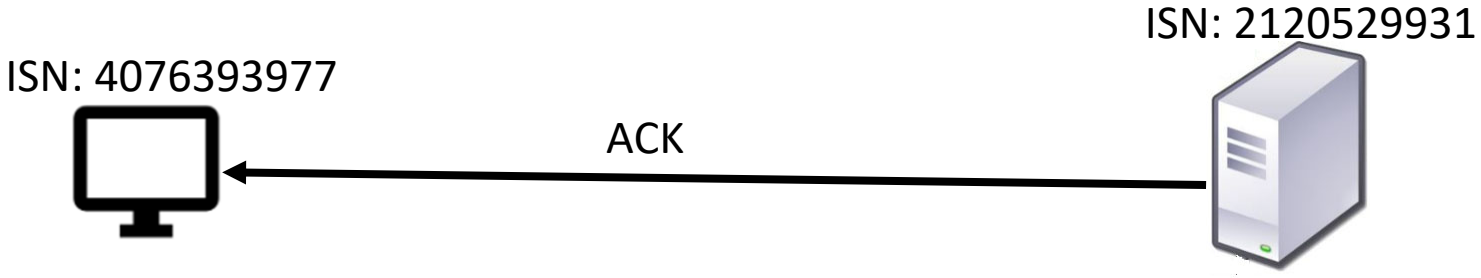
Application Sends data (451 Bytes)

U	A	P	R	S	F



62330								80
4076393979								
2120529932								
5	Reserved	0	0	0	0	0	0	
Checksum								Urgent Pointer
Options (0-40 Bytes)								
G	E	T		i	n	d	e	x	.	h	t	m	l		H	T	T	P	/	1	.	1		A	C	C	E	P	T	.	.

Application says connect to server



80														**6230**	
2120529932															
4076394493															
5		Reserved		0	1	0	0	0	0						
Checksum														Urgent Pointer	
Options (0-40 Bytes)															
Data (Optional)															

4076393979
+ 514

Bonus Quiz 7

- https://forms.gle/iroh9cEyjUxi1L238