Reliable Delivery, TCP

CS168 - Fall 2024

Agenda

- Logistics
- TCP Review
- TCP Handshakes — Connection Establishment
- TCP Handshakes — Teardown
- Worksheet

Logistics

- Homework 3 due next monday, October 28th
- Project 3 will be **officially** released soon!
- How’d the midterm go?
    - STP redemption question! Check out Edstem for more details

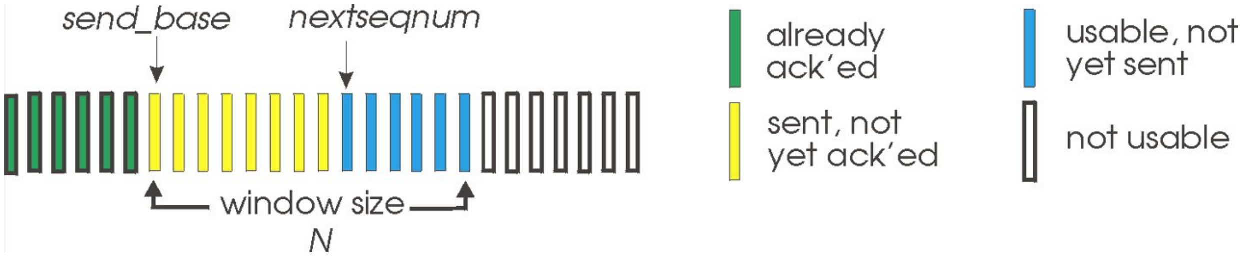
Reliability

- Best-effort network
  - Need to handle packet loss, corruption, reordering, delays, duplications, etc.
- Building blocks
  - Checksums: detect corruption
  - Feedback: positive/negative feedback from receiver
  - Retransmissions: sender re-sends packets
  - Timeouts: when to resend a packet
  - Sequence numbers: indicate which packets have been received
- Design considerations
  - Window size, nature of feedback, detection of loss, response to loss

Go-Back-N

Not what TCP does, but useful as a starting point to figure out something better.

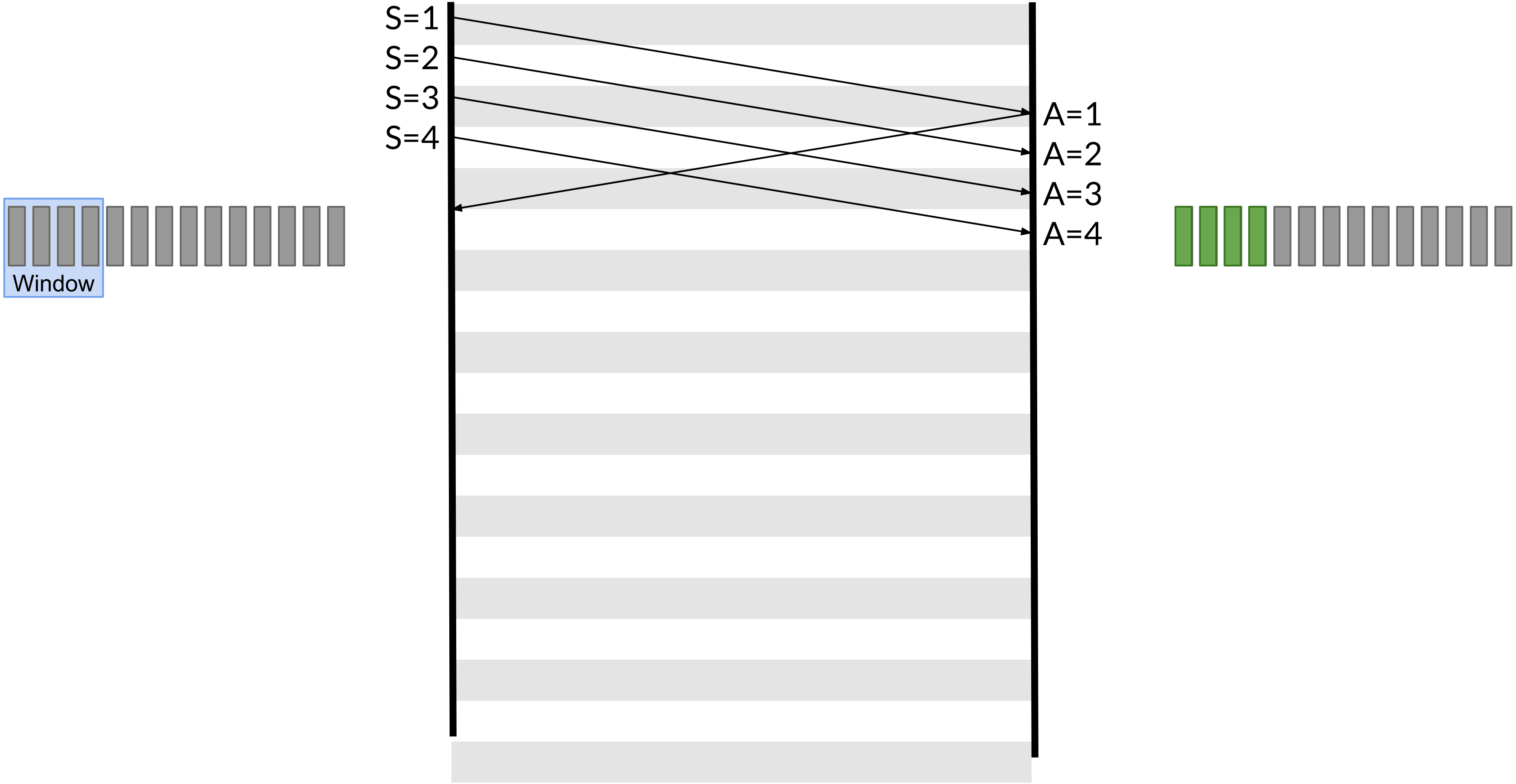
- Simple (though not advisable algorithm)
- “Sliding window” protocol: sender keeps a window of **up to W transmitted but unACKed** packets



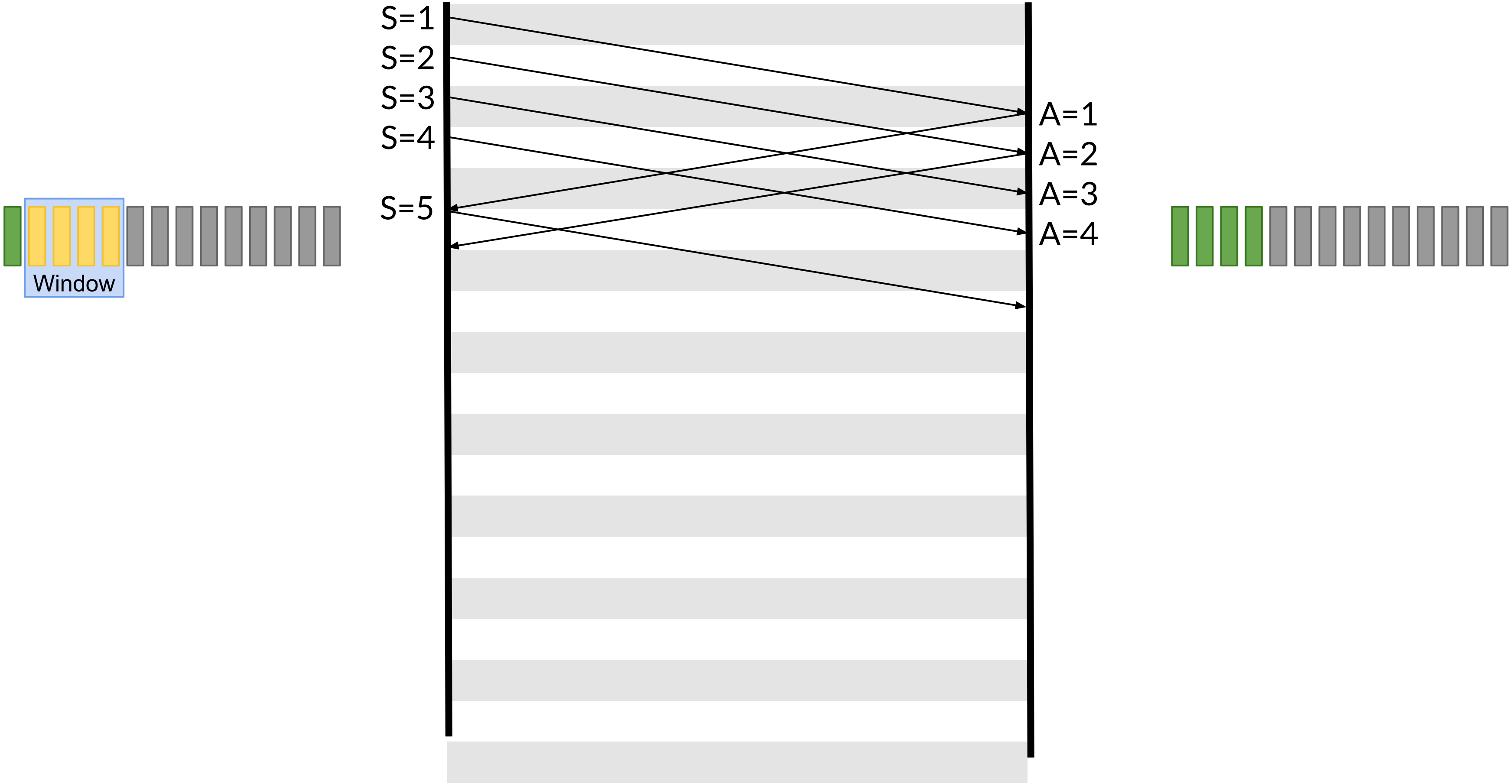
- Timer for oldest in-flight packet
- On timeout, resend all W packets (starting with the lost one)
- Receiver discards out-of-order packets

Kurose Ross, *Computer Networking: A Top-down Approach,* Chapter 3

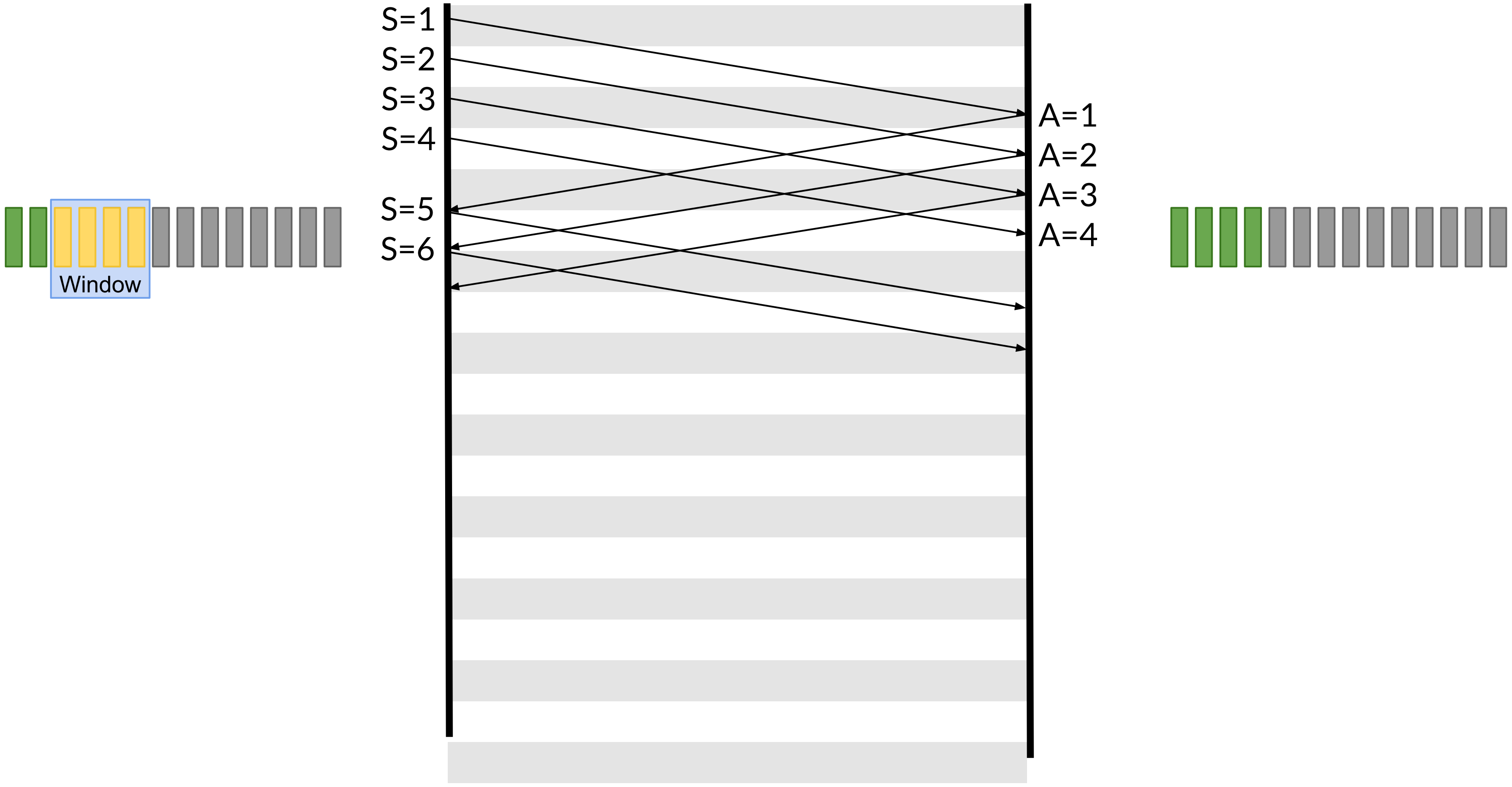
Go-Back-N w/o Errors



Go-Back-N w/o Errors



Go-Back-N w/o Errors



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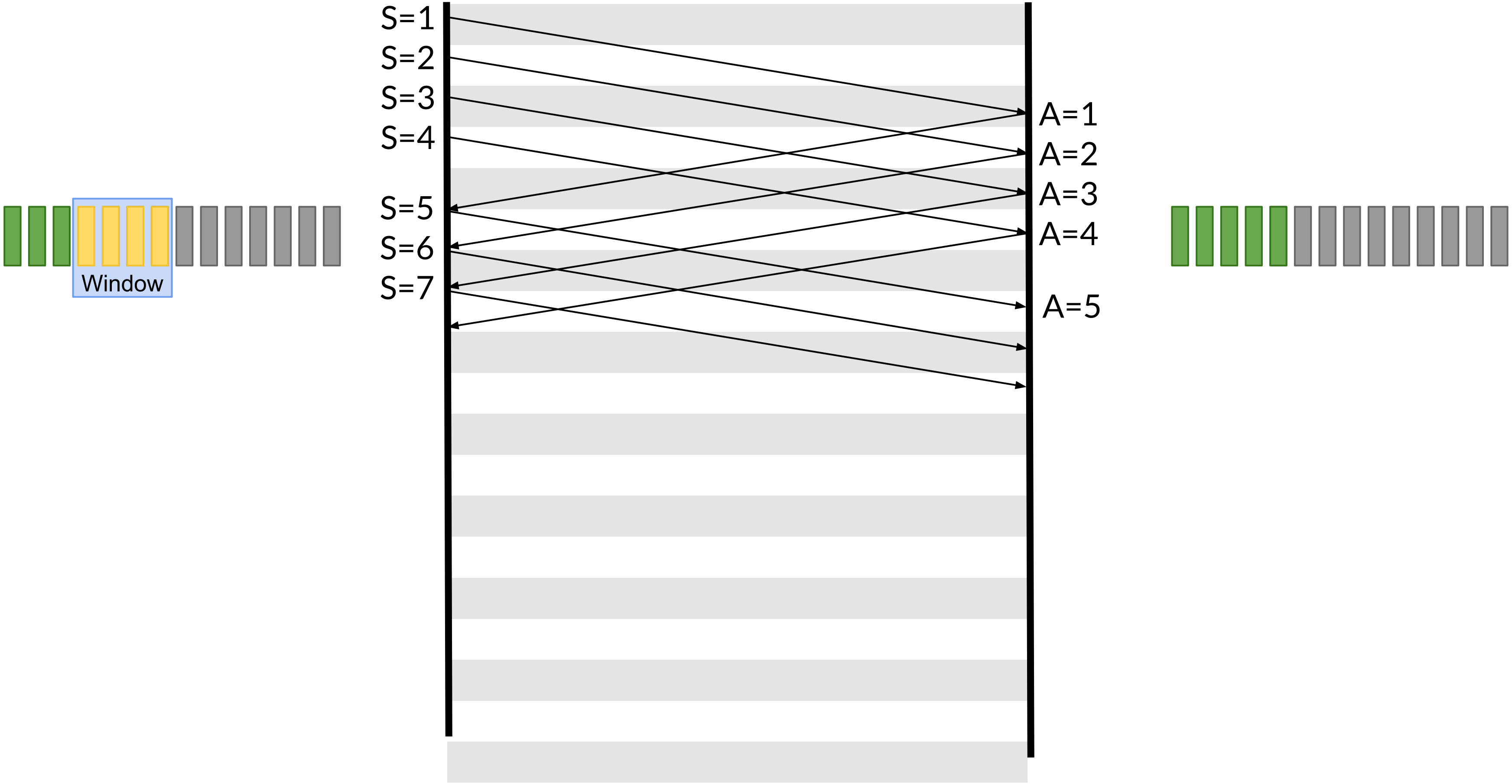
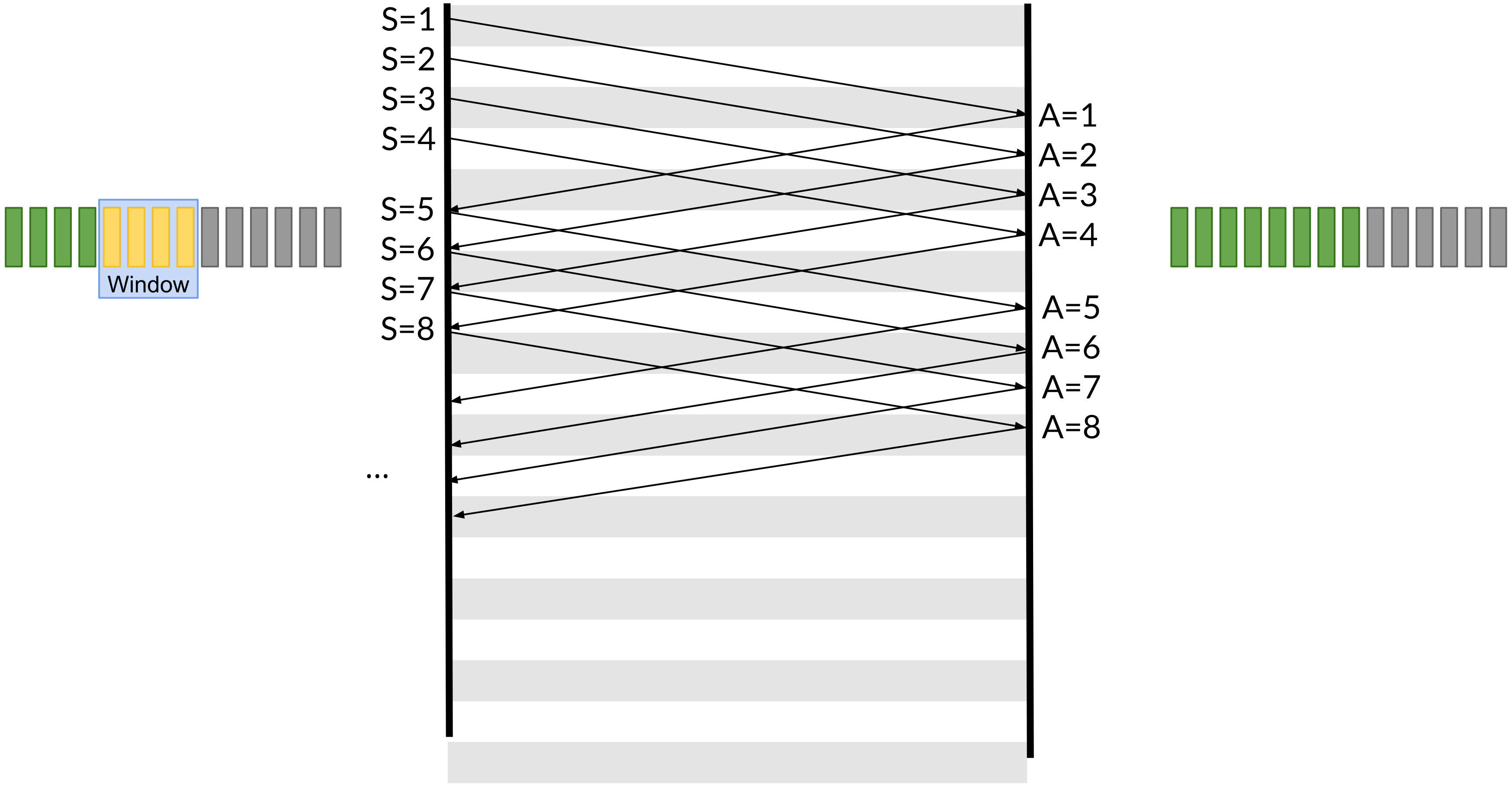
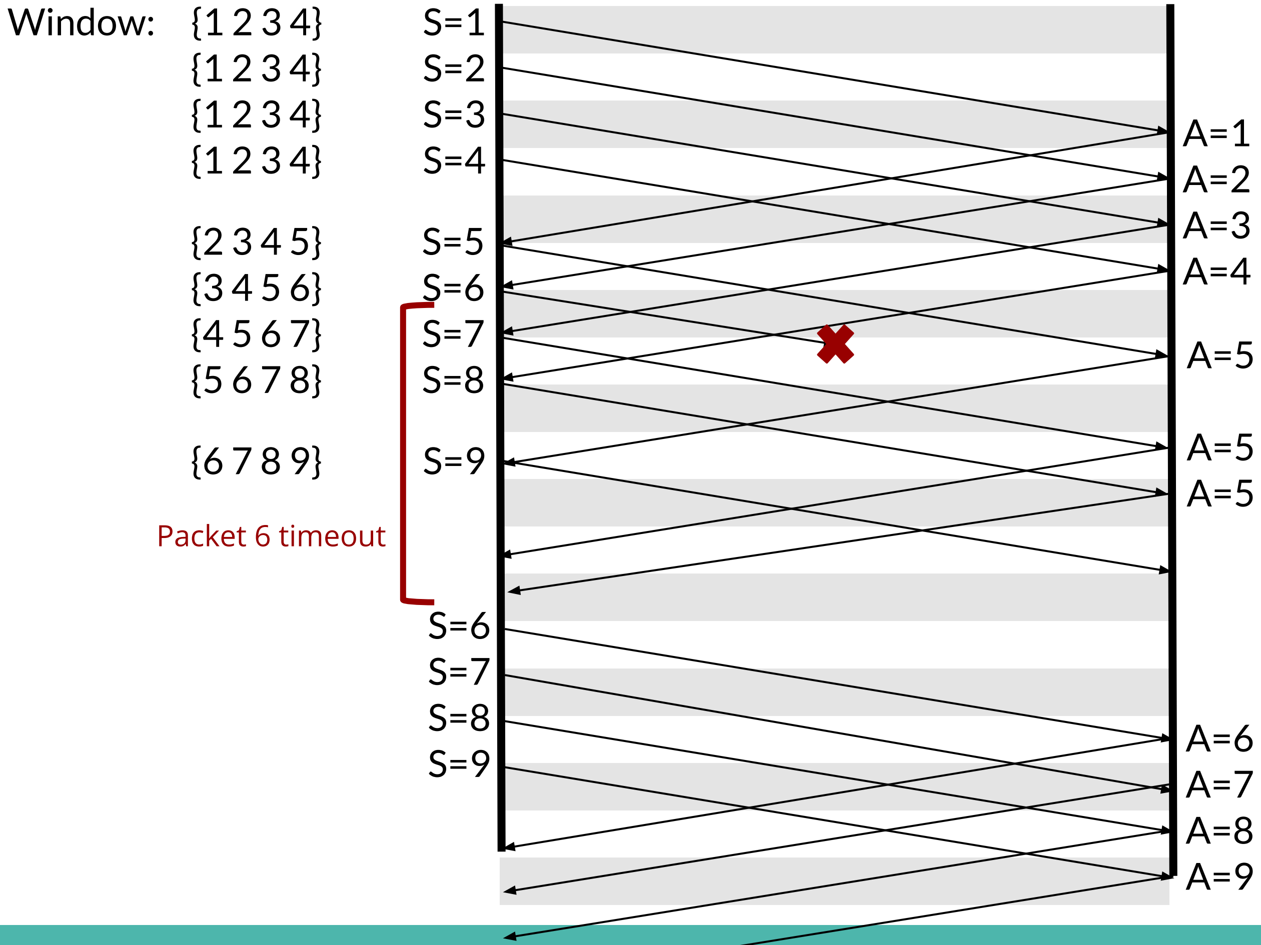


Go-Back-N w/o Errors



Go-Back-N w/ Errors



Window: {1 2 3 4} S=1

{1 2 3 4} S=2

{1 2 3 4} S=3

{1 2 3 4} S=4

{2 3 4 5} S=5

{3 4 5 6} S=6

{4 5 6 7} S=7

{5 6 7 8} S=8

{6 7 8 9} S=9

Packet 6 timeout

S=6

S=7

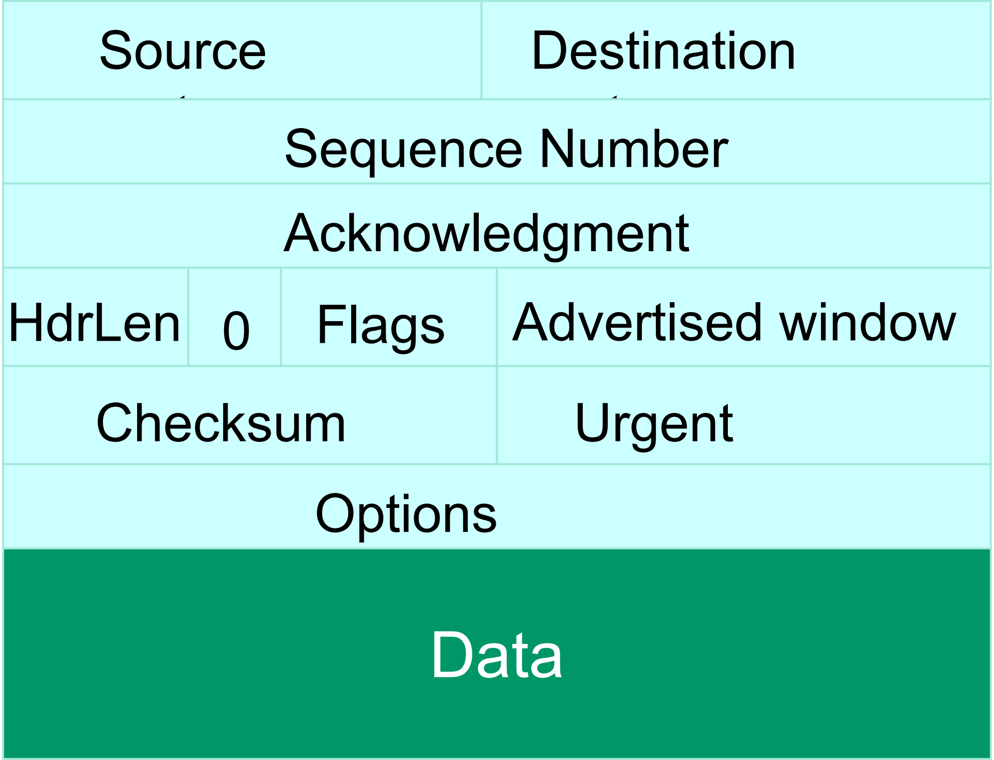
S=8

S=9

Review

TCP

- L4 Protocol (Transport)
- Byte Stream
- Bi-directional
- Reliable
- In-order



Source | Destination

Sequence Number

Acknowledgment

HdrLen | 0 | Flags | Advertised window

Checksum | Urgent

Options

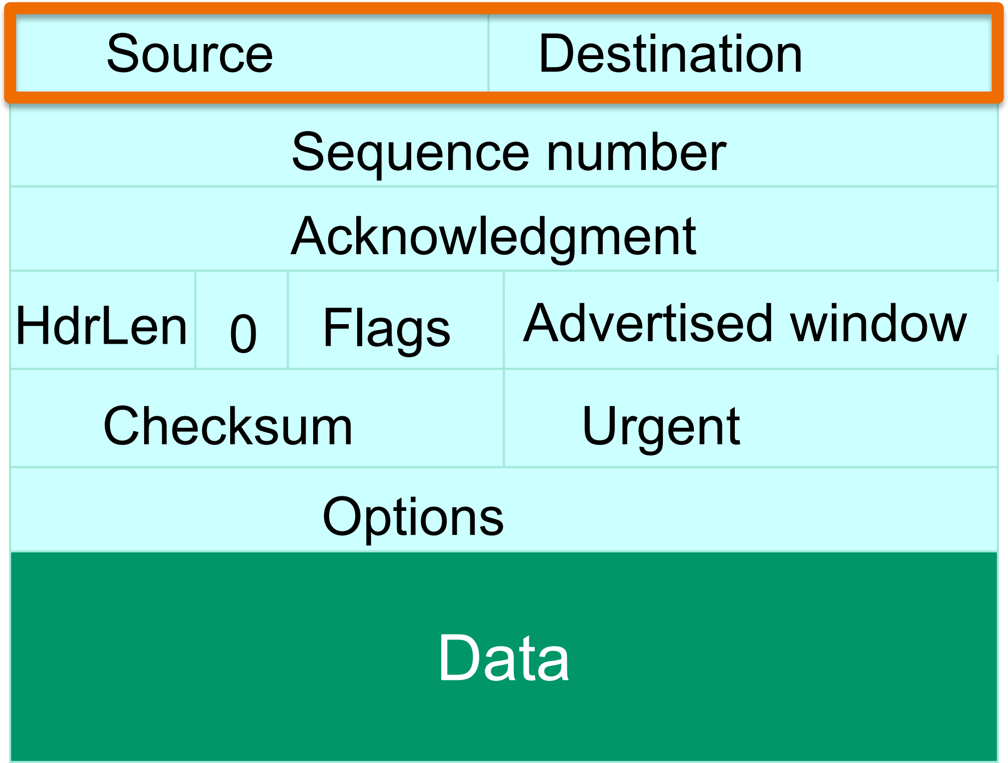
Data

TCP Header

- Host port numbers
  - Multiplexing and demultiplexing
  - 16 bits

But wait - why no addr?!

- IP header has addr.



Source | Destination

Sequence number

Acknowledgment

HdrLen | 0 | Flags | Advertised window

Checksum | Urgent

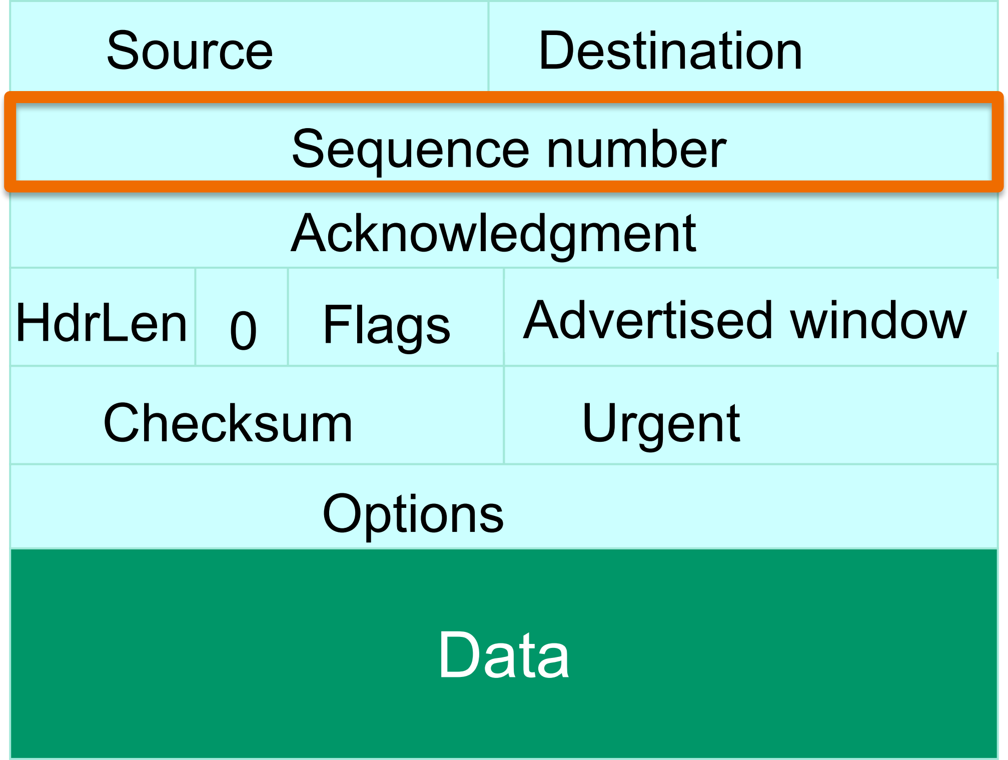
Options

Data

TCP Header

- Byte offset
  - Of first payload byte
  - Initialized randomly

- **Byte Stream Protocol**
  - Seq # refers to **bytes**



Source | Destination

Sequence number

Acknowledgment

HdrLen | 0 | Flags | Advertised window

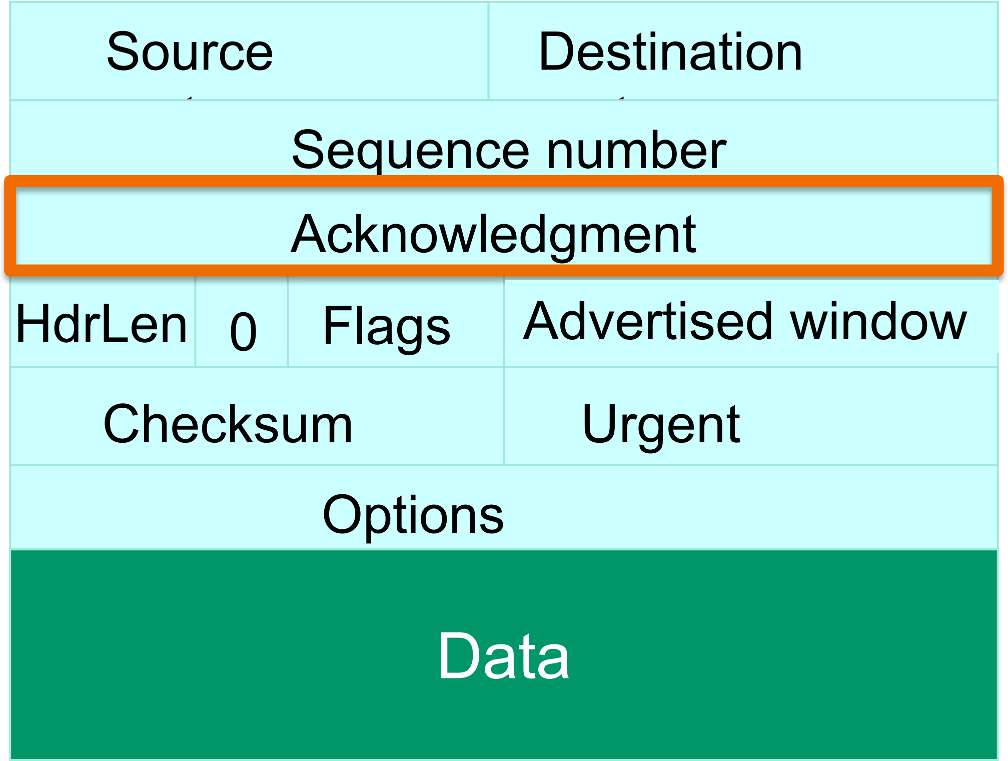
Checksum | Urgent

Options

Data

TCP Header

- Cumulative ACKs
- Seq # of **next byte**
- Data packets carry ACKs



Source

Destination

Sequence number

Acknowledgment

HdrLen

0

Flags

Advertised window

Checksum

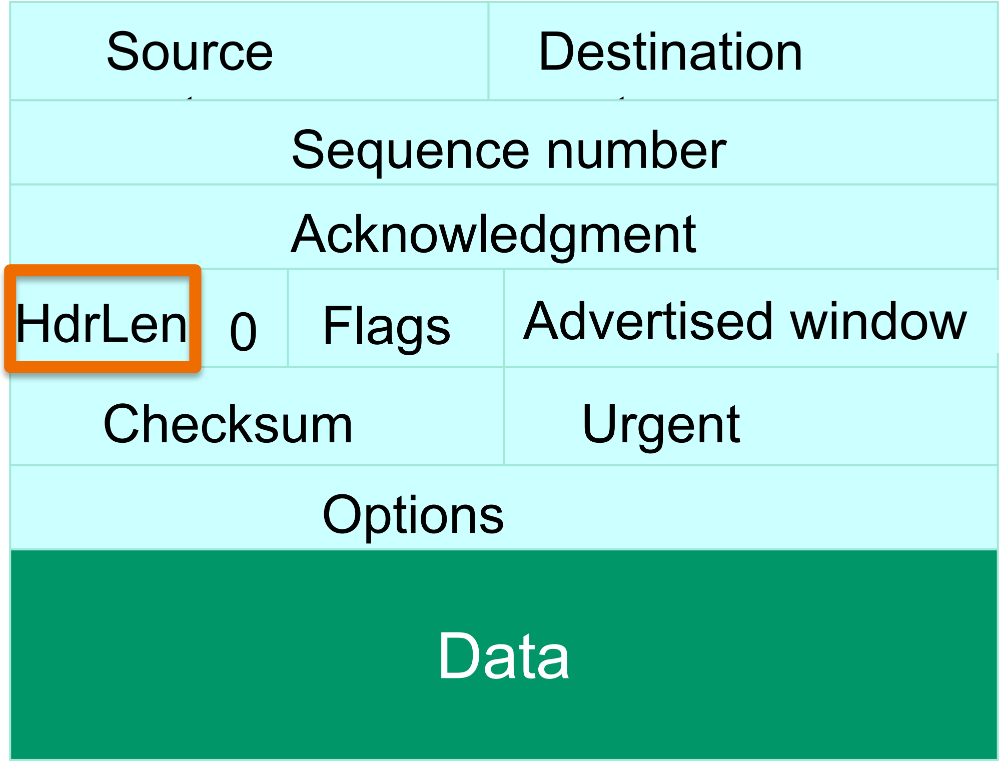
Urgent

Options

Data

TCP Header

- Header length
  - 4 bits
  - In 4-byte words
- Minimum 5 words (20B)
- Maximum 15 words
  - Why?
  - Only 4 bits, so max is 2^4 – 1 = 15 words



Source

Destination

Sequence number

Acknowledgment

HdrLen

0

Flags

Advertised window

Checksum

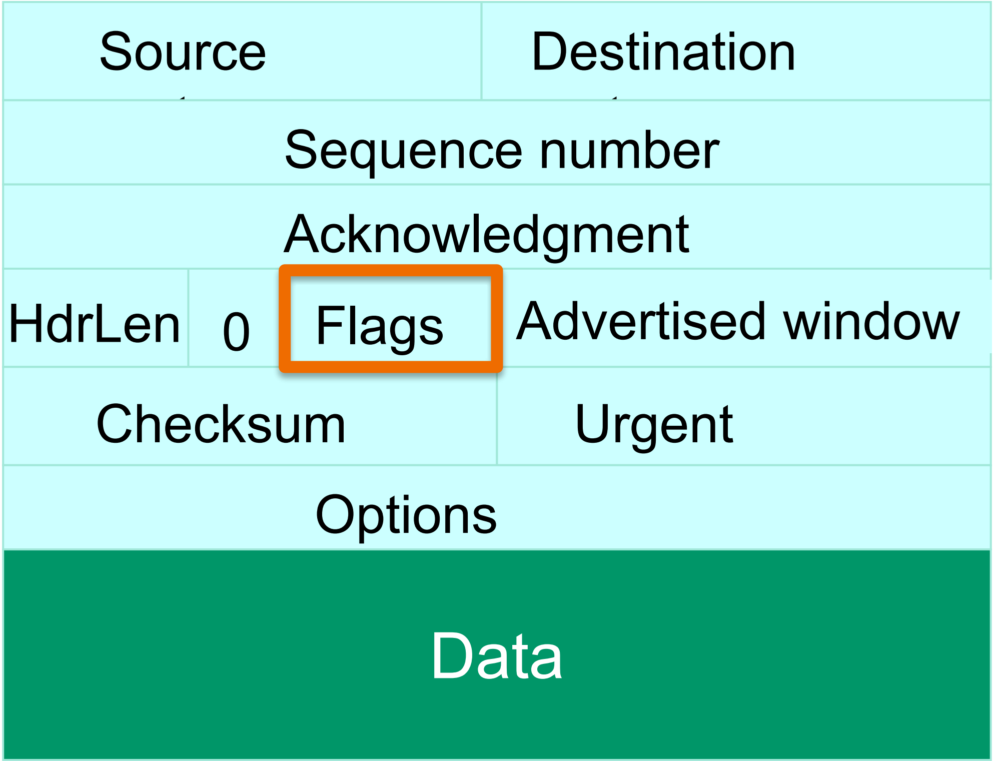
Urgent

Options

Data

TCP Header

- SYN
  - SYNchronize initial state
- ACK
  - ACKnowledgement
- FIN
  - No more data
- RST
  - Connection ReSeT



Source

Destination

Sequence number

Acknowledgment

HdrLen

0

Flags

Advertised window

Checksum

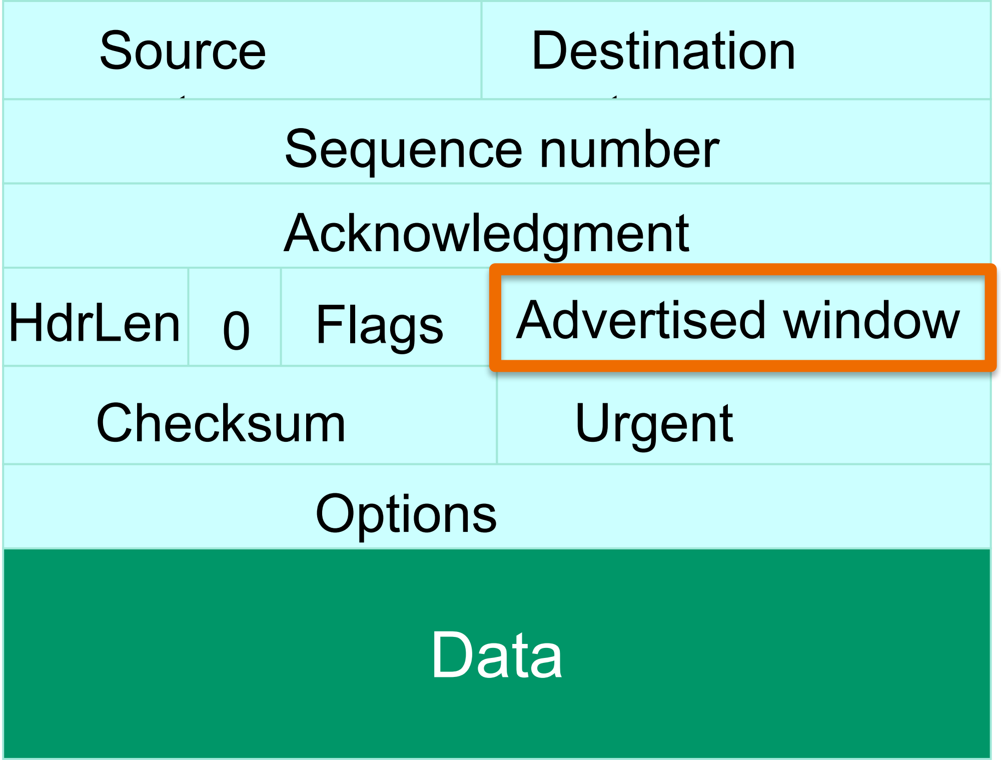
Urgent

Options

Data

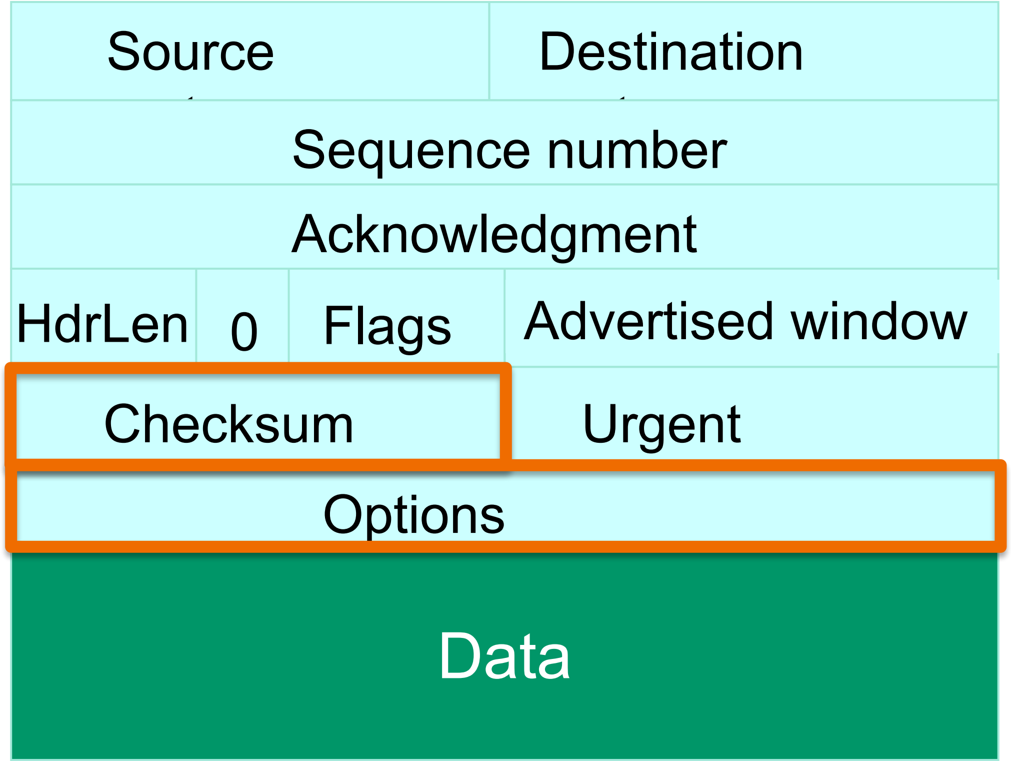
TCP Header

- Receive Window Size
  - Maximum receiver buffer
- Limits sending rate
  - Don’t send faster than receiver can process



TCP Header

- Checksum
  - Includes header and payload
- Options common
  - Unlike IP
  - Not covered
  - Assume no options unless specified

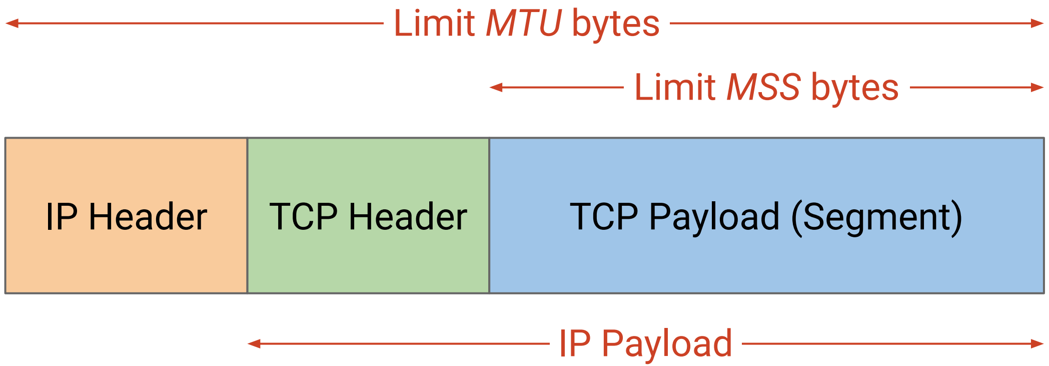


TCP Segments

TCP/IP packet: IP packet with TCP header and TCP data inside.

Size limits:

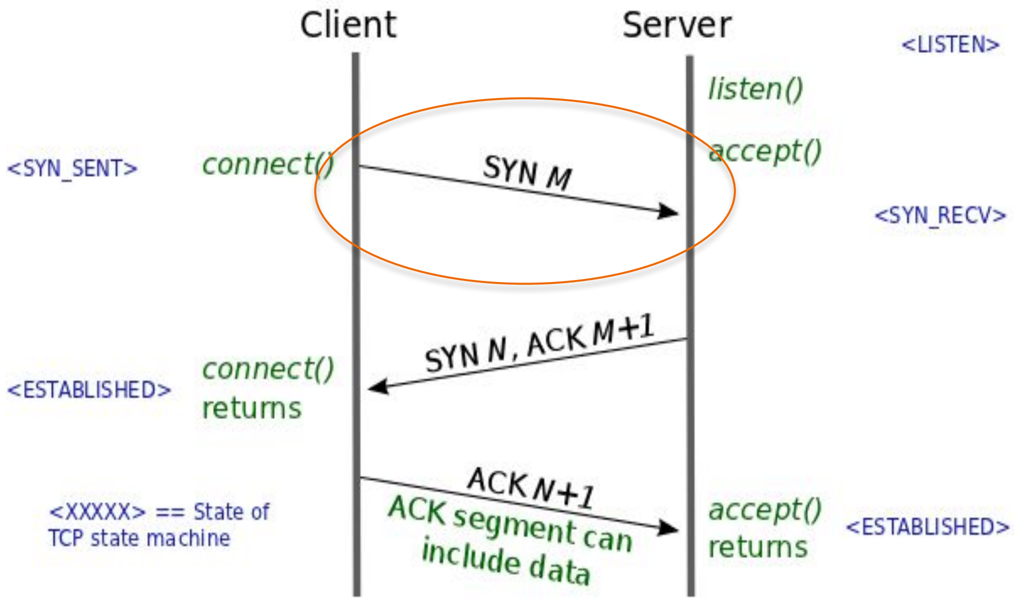
- IP packet: Maximum transmission unit (MTU).
- TCP segment: Maximum segment size (MSS).
- MSS = MTU – (IP header) – (TCP header).



Connection Establishment

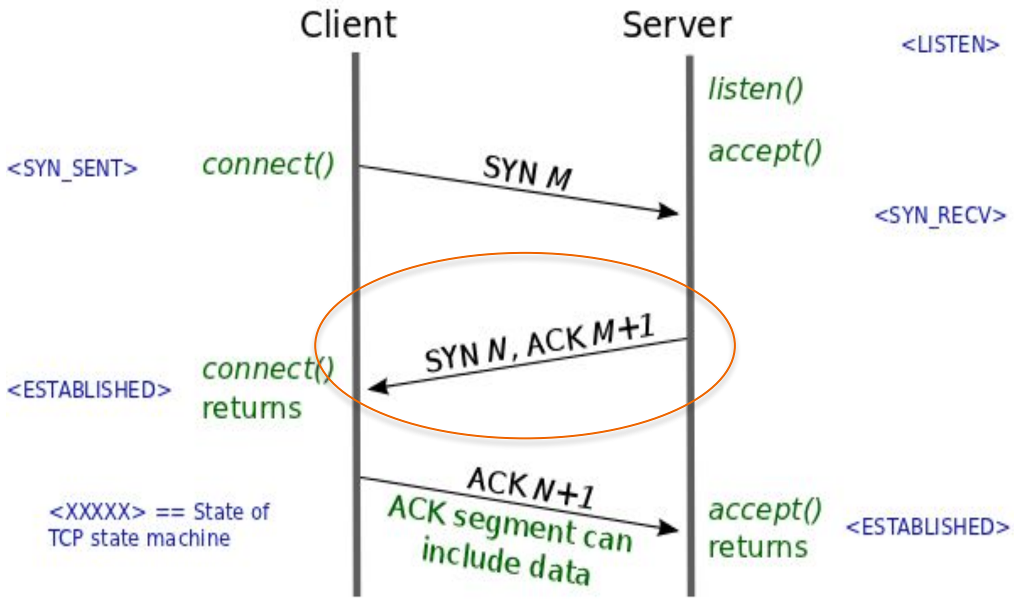
Three Way Handshake

- Client sends server a SYN
  - A TCP packet with the SYN flag set
- SYN packet carries initial state
  - Receive window
  - Source port number
  - Initial sequence number



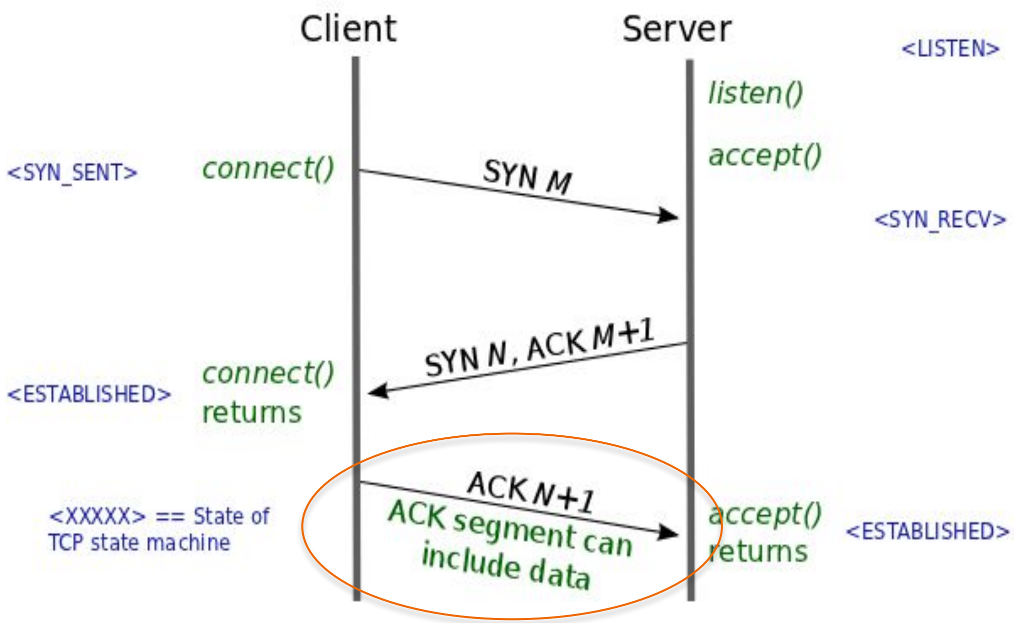
Three Way Handshake

- Server responds with a SYN-ACK
- Acknowledge the client's SYN
- Respond with the server's own SYN
- Carries server's initial state
  - Receive window
  - Source port number
  - Initial sequence number



Three Way Handshake

- Client responds with an ACK
- Now the connection is established!
  - Client and server can freely communicate



Connection Establishment

- Sequence number and acknowledgement number don’t start from zero
- Instead client and server choose a **random** initial seq #
  - Why?
- Must be communicated between client and server

Connection Establishment

- Client/Server exchange state
  - Initial Sequence numbers
  - Port number
  - TCP Options

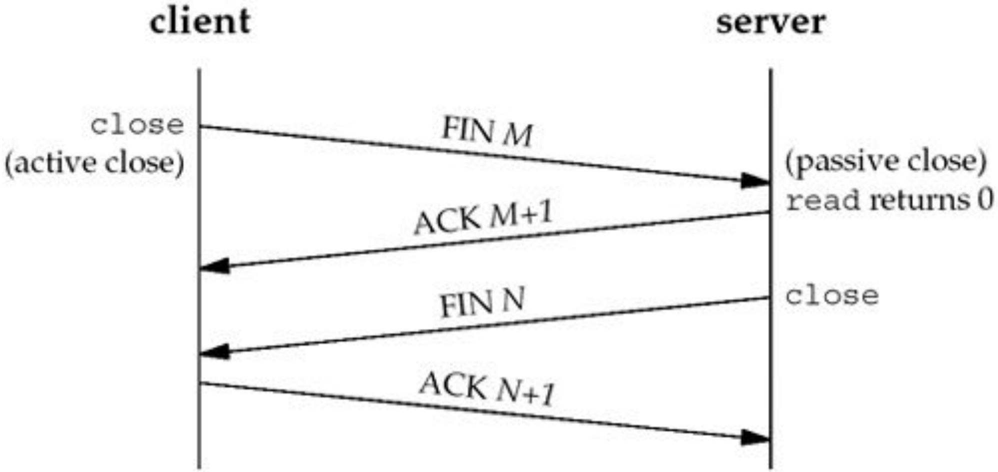
Teardown

Two Teardown Methods

- Four way handshake
  - Graceful/normal teardown
- Reset
  - Exceptional Teardown

Normal Teardown

- One side closes their connection
  - Indicates they will send no more data
  - Sends a FIN
  - Receives an ACK
- Other side may continue transmitting
- Eventually it closes as well
  - Sends a FIN
  - Receives an ACK
- Connection is closed



TCP Reset

- Reset the connection
  - Most commonly, attempt to SYN on a closed port
- Send RST packet(s) only (no ACK)

Reliability

Reliability

- Cumulative ACKs
  - Allow detection of dropped packets
- How do we know a packet was lost?
  - Timeout
  - Duplicate ACKs

Timeout

- Retransmission Time Out (RTO)
  - Timeout after which packets are retransmitted
  - Based on a constantly updated RTT estimate (and variance)
- Single timer (not per-packet)
  - Each received ACK of new data resets RTO
  - If RTO times out
    - Retransmit packet containing “next byte”

Timeout

- What if RTO is too large?
- Packet is dropped . . .
  - Wait
  - Keep waiting
  - … Keep waiting
  - Timer goes off
    - Finally retransmit
- Can we do better?

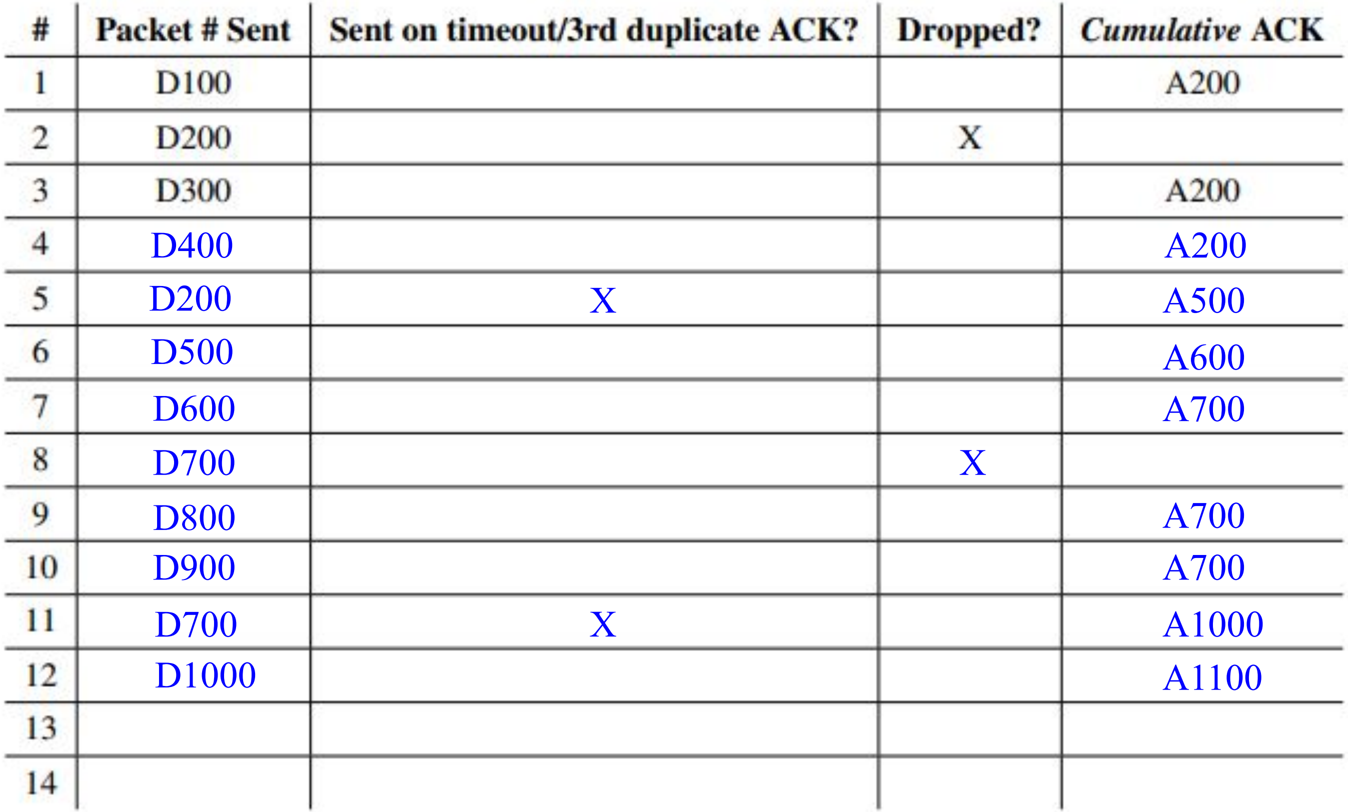
Duplicate ACKs

- Transmit
  - Seq 1000
  - Seq 2000
  - ***Seq 3000 Dropped in flight***
  - Seq 4000
  - Seq 5000
  - Seq 6000
- What ACKs do we receive?
  - ACK 2000
  - ACK 3000
  - ACK 3000
  - ACK 3000
  - ACK 3000
- Duplicate ACKs indicate packet loss/reorder

Feedback Form:
https://tinyurl.com/cs168-disc-fa24

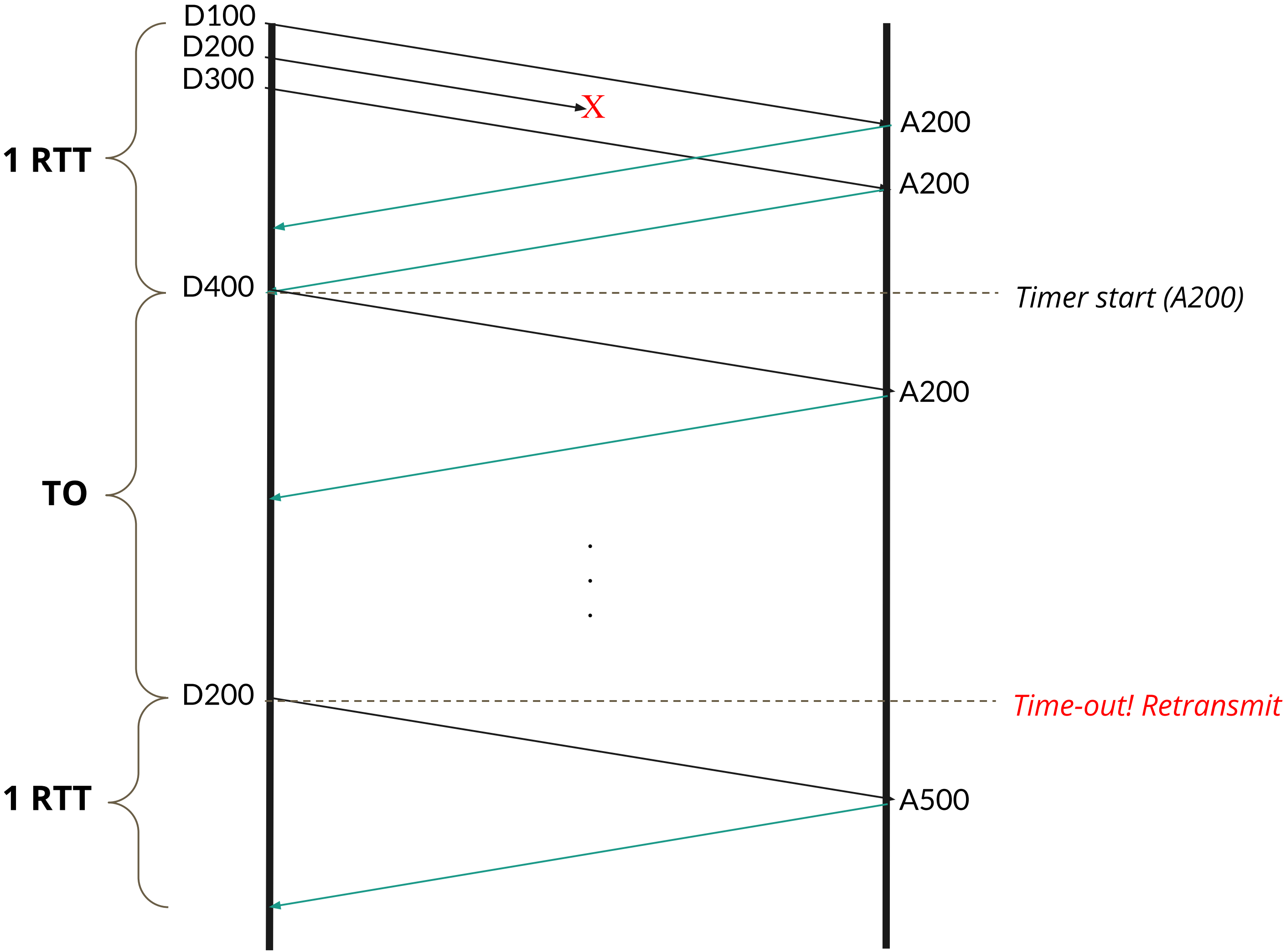
Worksheet

Q1.1

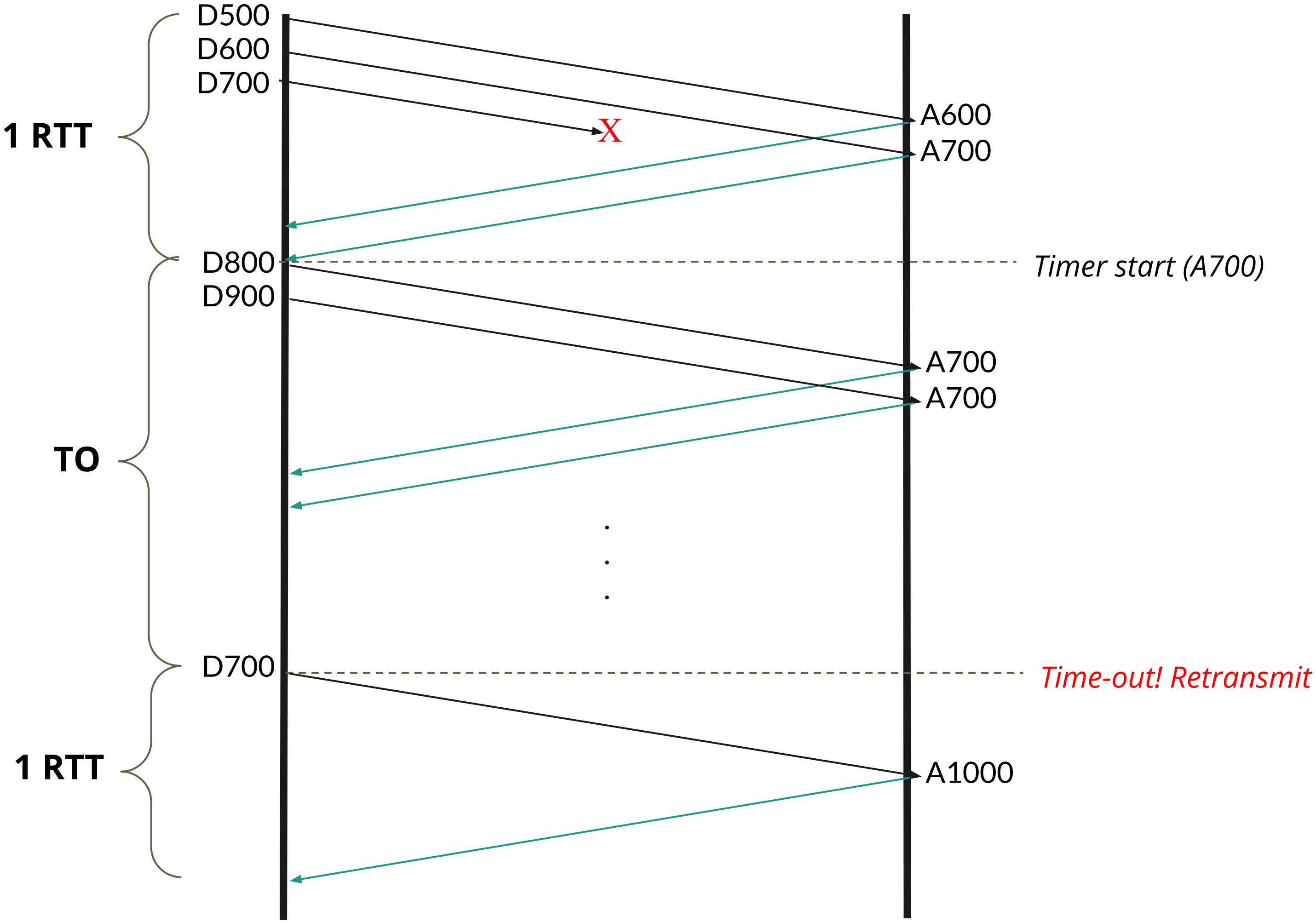


#	Packet # Sent	Sent on timeout/3rd duplicate ACK?	Dropped?	*Cumulative* ACK
1	D100			A200
2	D200		X	
3	D300			A200
4	D400			A200
5	D200	X		A500
6	D500			A600
7	D600			A700
8	D700		X	
9	D800			A700
10	D900			A700
11	D700	X		A1000
12	D1000			A1100
13				
14				

Q1.2

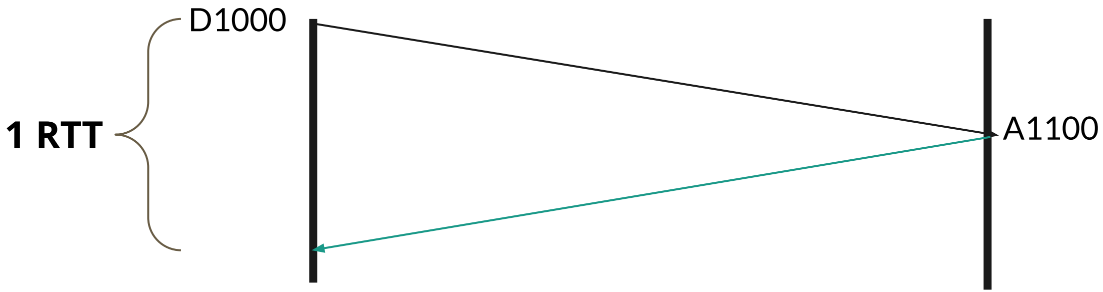

D100
D200
D300
X
A200
A200
1 RTT
D400
Timer start (A200)
A200
TO
D200
Time-out! Retransmit
A500
1 RTT

Q1.2



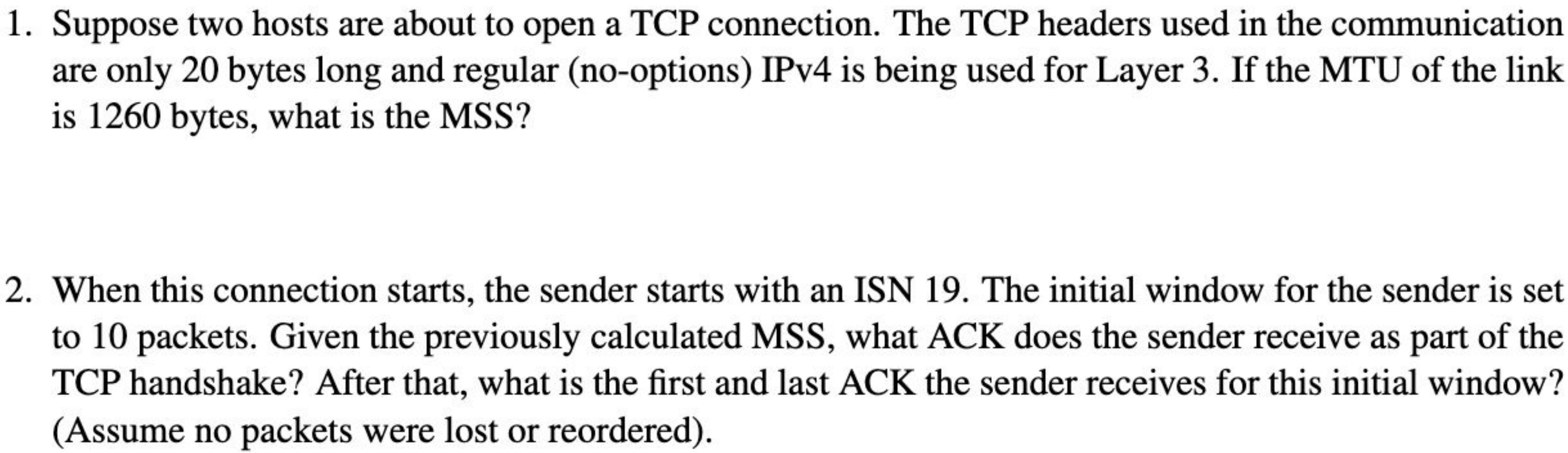
D500
D600
D700
X
A600
A700
1 RTT
D800
D900
Timer start (A700)
A700
A700
TO
D700
Time-out! Retransmit
A1000
1 RTT

Q1.2



Total Time = RTT + TO + RTT + RTT + TO + RTT + RTT
= 5 * RTT + 2 * TO
= 6.05 seconds

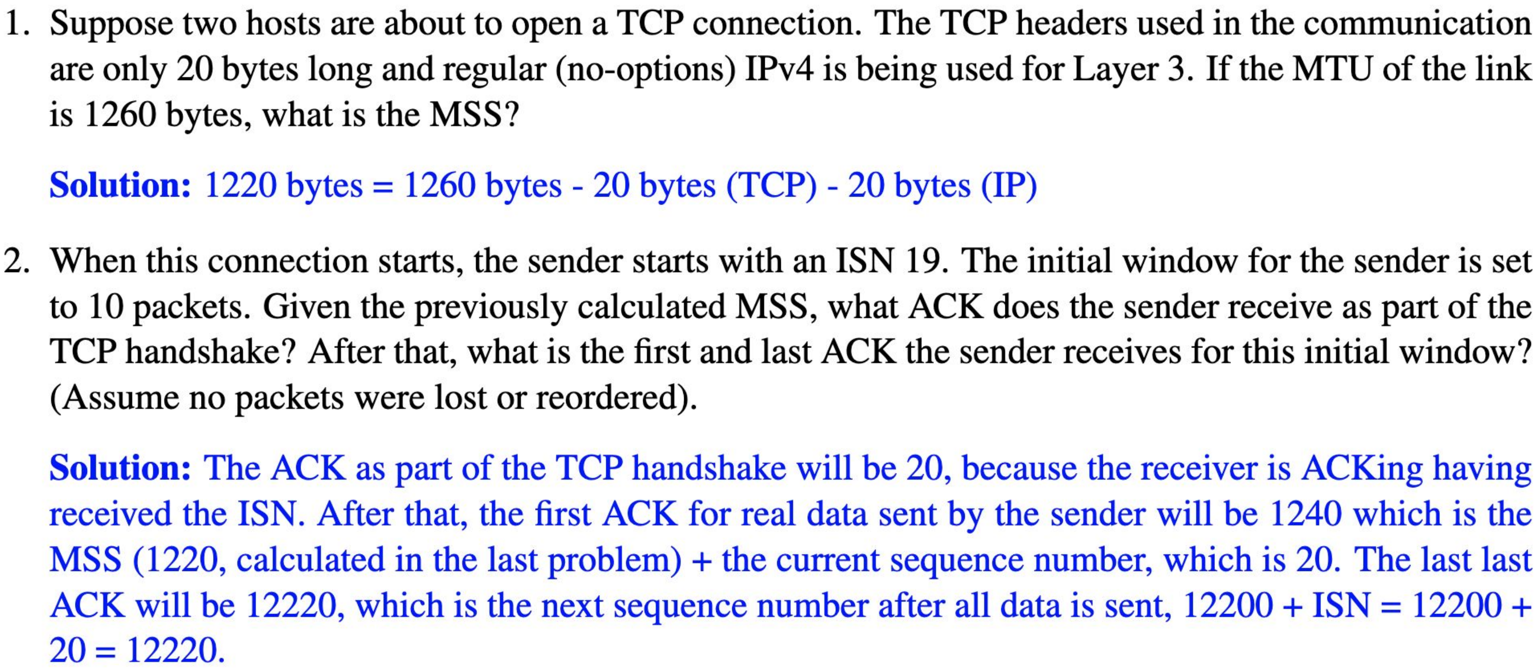
Q2.1 & 2.2



1. Suppose two hosts are about to open a TCP connection. The TCP headers used in the communication are only 20 bytes long and regular (no-options) IPv4 is being used for Layer 3. If the MTU of the link is 1260 bytes, what is the MSS?

2. When this connection starts, the sender starts with an ISN 19. The initial window for the sender is set to 10 packets. Given the previously calculated MSS, what ACK does the sender receive as part of the TCP handshake? After that, what is the first and last ACK the sender receives for this initial window? (Assume no packets were lost or reordered).

Q2.1 & 2.2



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   Solution: 1220 bytes = 1260 bytes - 20 bytes (TCP) - 20 bytes (IP)

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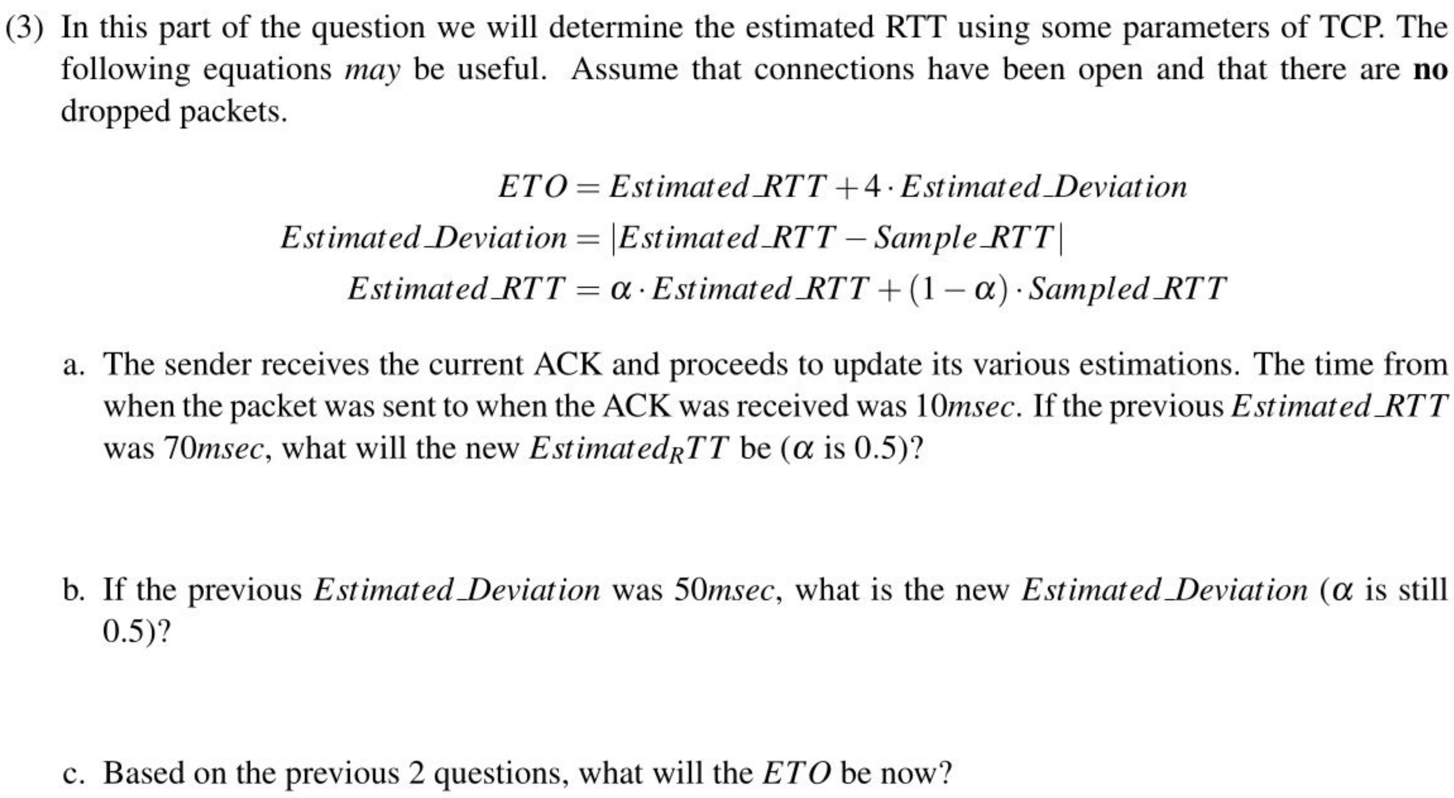
   Solution: The ACK as part of the TCP handshake will be 20, because the receiver is ACKing having received the ISN. After that, the first ACK for real data sent by the sender will be 1240 which is the MSS (1220, calculated in the last problem) + the current sequence number, which is 20. The last last ACK will be 12220, which is the next sequence number after all data is sent, 12200 + ISN = 12200 + 20 = 12220.

Q2.3



(3) In this part of the question we will determine the estimated RTT using some parameters of TCP. The following equations *may* be useful. Assume that connections have been open and that there are **no** dropped packets.

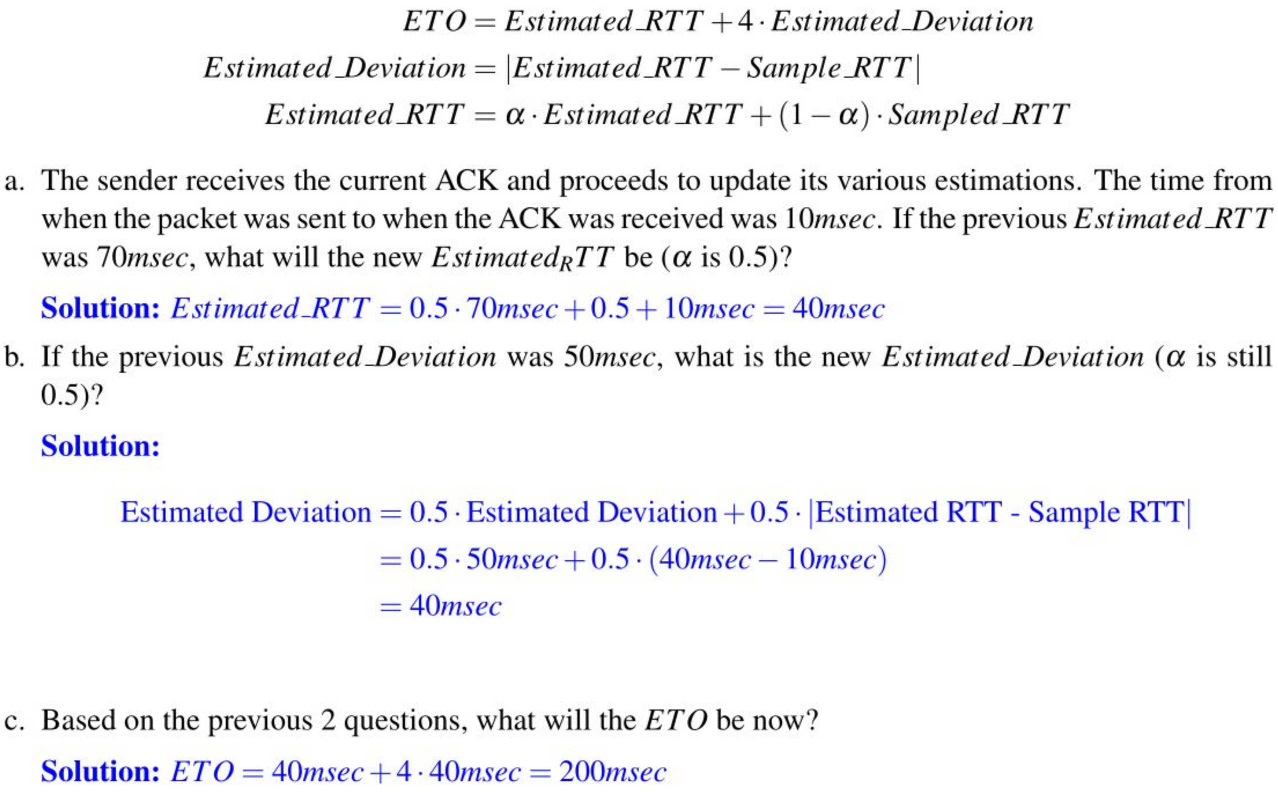
$$ETO = Estimated\_RTT + 4 \cdot Estimated\_Deviation$$
$$Estimated\_Deviation = |Estimated\_RTT - Sample\_RTT|$$
$$Estimated\_RTT = \alpha \cdot Estimated\_RTT + (1 - \alpha) \cdot Sampled\_RTT$$

a. The sender receives the current ACK and proceeds to update its various estimations. The time from when the packet was sent to when the ACK was received was $10msec$. If the previous $Estimated\_RTT$ was $70msec$, what will the new $Estimated_RTT$ be (α is 0.5)?

b. If the previous $Estimated\_Deviation$ was $50msec$, what is the new $Estimated\_Deviation$ (α is still 0.5)?

c. Based on the previous 2 questions, what will the ETO be now?

Q2.3



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Solution: $Estimated\_RTT = 0.5 \cdot 70msec + 0.5 + 10msec = 40msec$

b. If the previous $Estimated\_Deviation$ was $50msec$, what is the new $Estimated\_Deviation$ (α is still 0.5)?

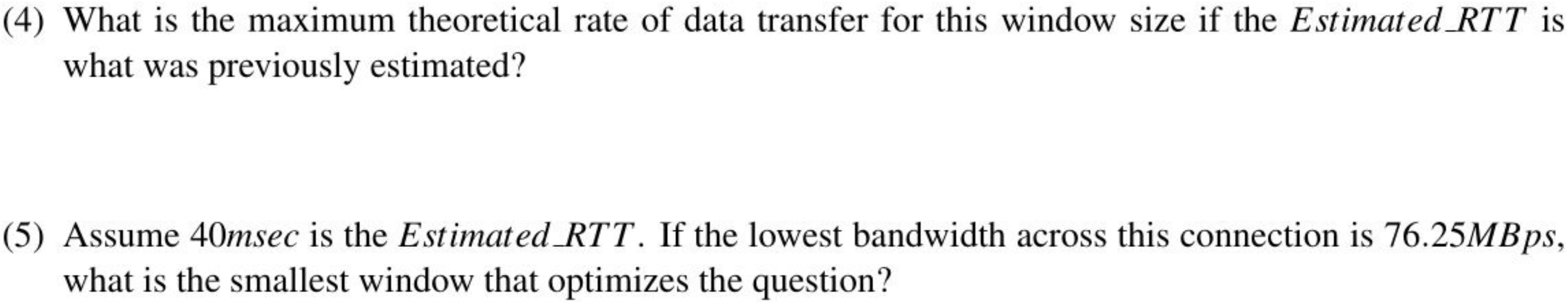
Solution:

$$\begin{aligned} \text{Estimated Deviation} &= 0.5 \cdot \text{Estimated Deviation} + 0.5 \cdot |\text{Estimated RTT - Sample RTT}| \\ &= 0.5 \cdot 50msec + 0.5 \cdot (40msec - 10msec) \\ &= 40msec \end{aligned}$$

c. Based on the previous 2 questions, what will the ETO be now?

Solution: $ETO = 40msec + 4 \cdot 40msec = 200msec$

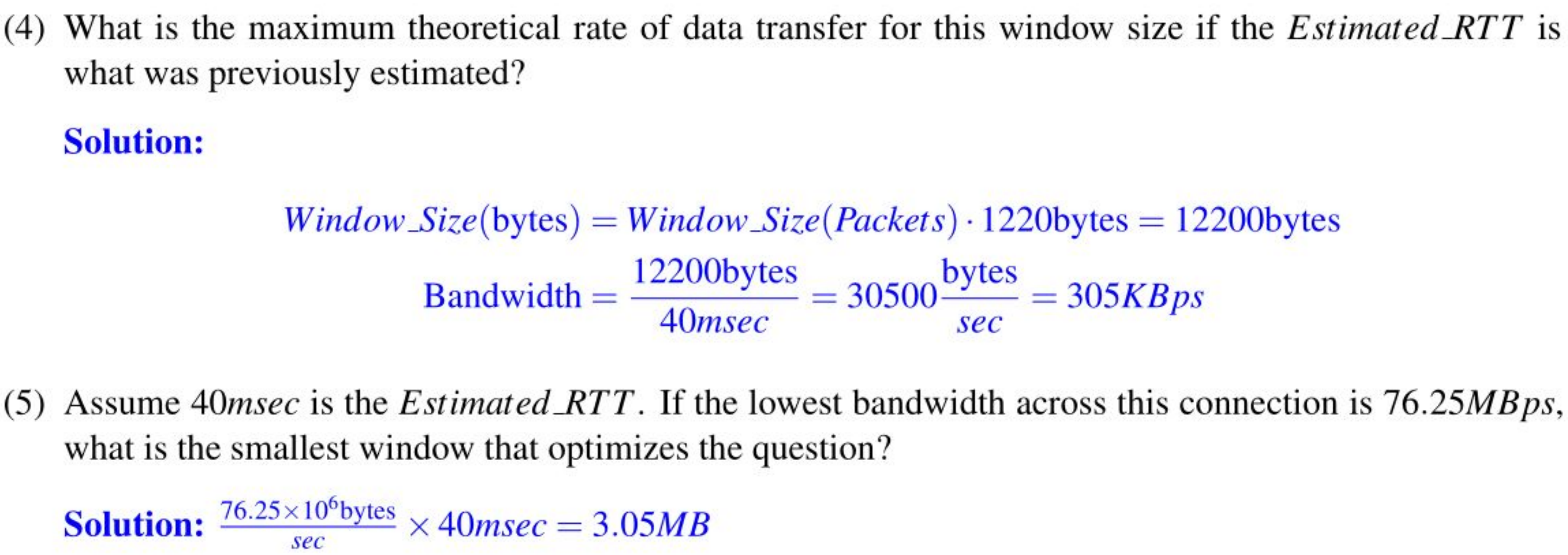
Q2.4 & 2.5



(4) What is the maximum theoretical rate of data transfer for this window size if the $Estimated\_RTT$ is what was previously estimated?

(5) Assume $40msec$ is the $Estimated\_RTT$. If the lowest bandwidth across this connection is $76.25MBps$, what is the smallest window that optimizes the question?

Q2.4 & 2.5



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Solution:

$$Window\_Size(\text{bytes}) = Window\_Size(Packets) \cdot 1220\text{bytes} = 12200\text{bytes}$$

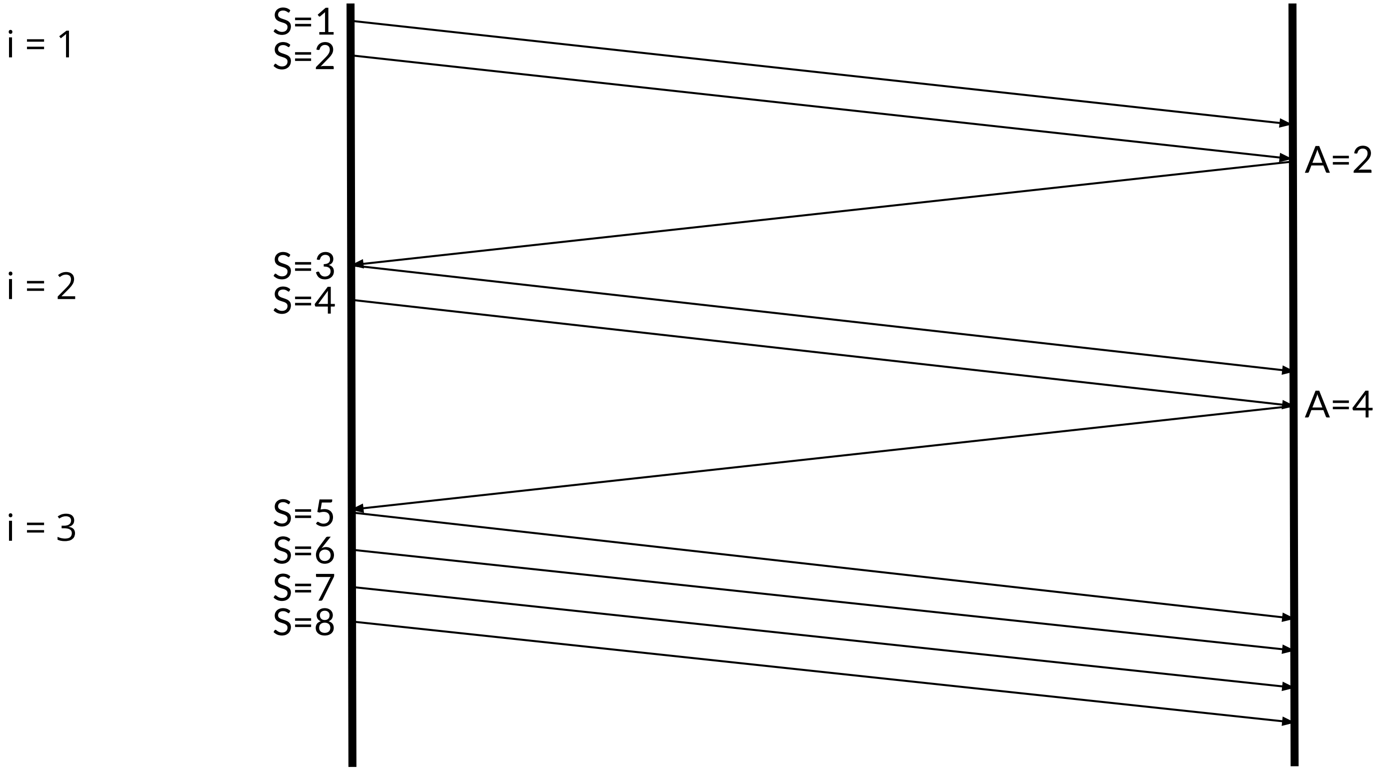
$$\text{Bandwidth} = \frac{12200\text{bytes}}{40msec} = 30500\frac{\text{bytes}}{sec} = 305KBps$$

(5) Assume $40msec$ is the $Estimated\_RTT$. If the lowest bandwidth across this connection is $76.25MBps$, what is the smallest window that optimizes the question?

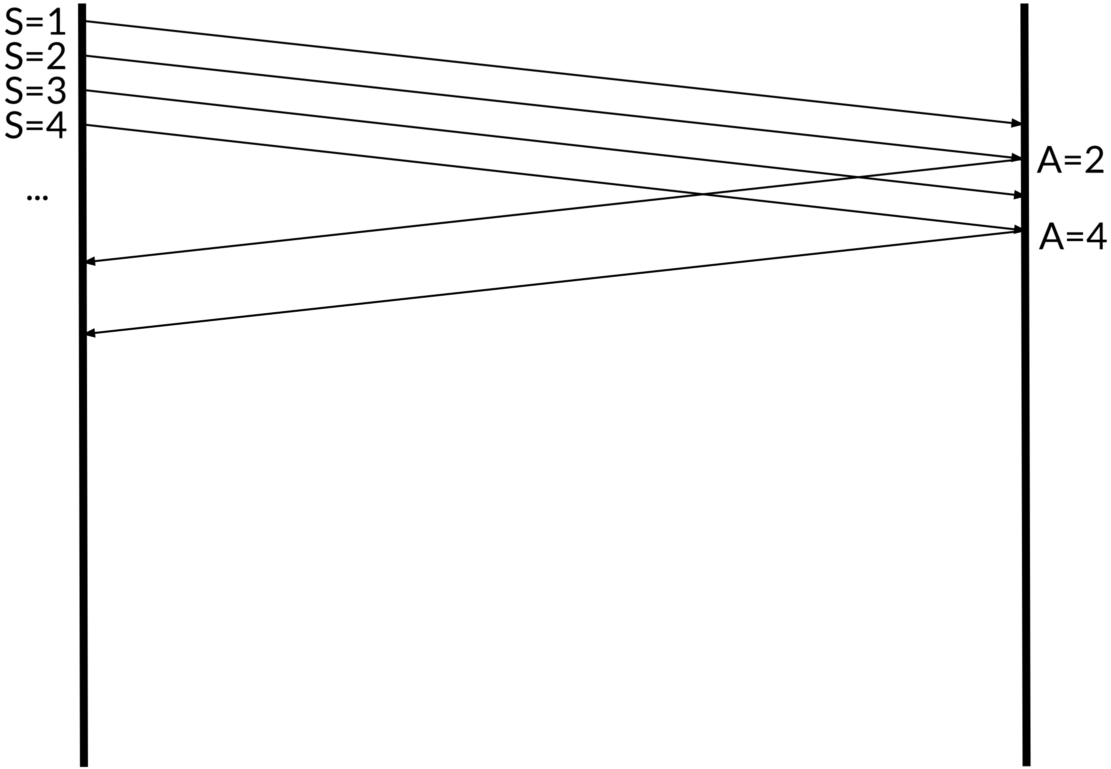
Solution: $\frac{76.25\times10^6\text{bytes}}{sec} \times 40msec = 3.05MB$

Question 3: Reliability ?

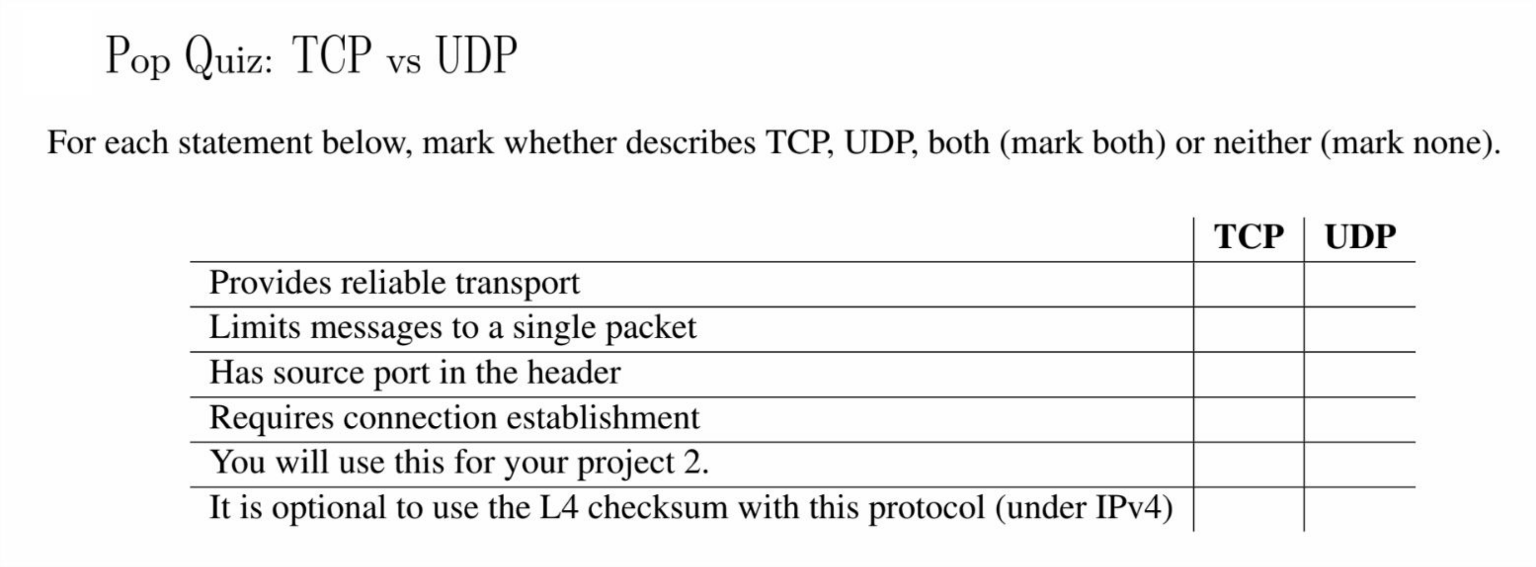
Q3: Bob's Idea



Q3: Alice’s Idea



Q4

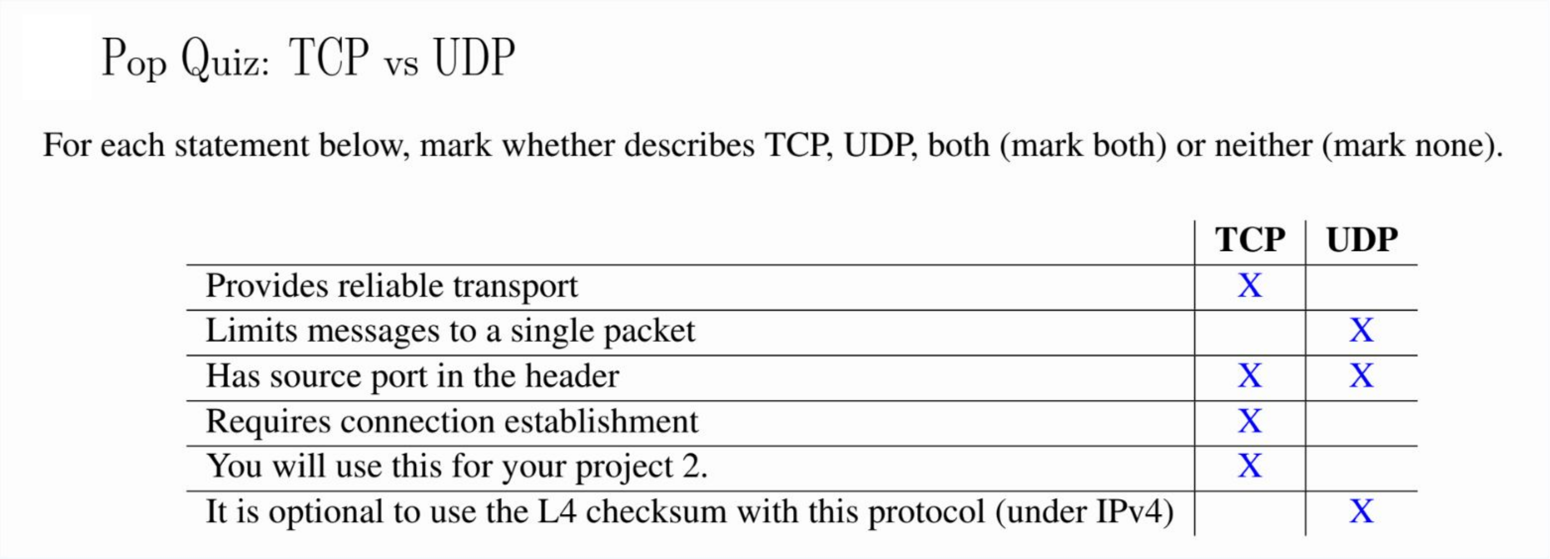


Pop Quiz: TCP vs UDP

For each statement below, mark whether describes TCP, UDP, both (mark both) or neither (mark none).

	TCP	**UDP**
Provides reliable transport		
Limits messages to a single packet		
Has source port in the header		
Requires connection establishment		
You will use this for your project 2.		
It is optional to use the L4 checksum with this protocol (under IPv4)		

Q4



Pop Quiz: TCP vs UDP

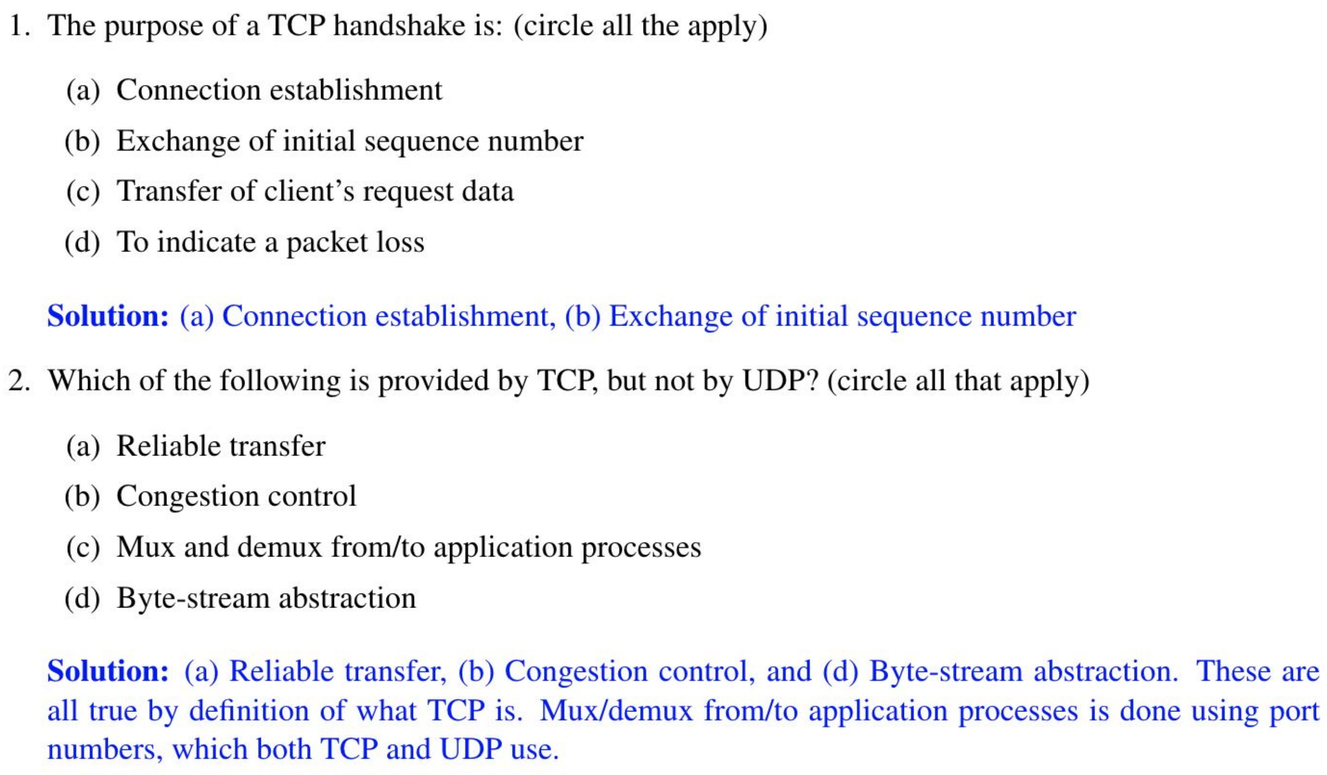
For each statement below, mark whether describes TCP, UDP, both (mark both) or neither (mark none).

	TCP	**UDP**
Provides reliable transport	X	
Limits messages to a single packet		X
Has source port in the header	X	X
Requires connection establishment	X	
You will use this for your project 2.	X	
It is optional to use the L4 checksum with this protocol (under IPv4)		X

Q5. TCP Short Questions

Discuss with a person next to you.

Q5. TCP Short Questions



1. The purpose of a TCP handshake is: (circle all the apply)

   (a) Connection establishment

   (b) Exchange of initial sequence number

   (c) Transfer of client's request data

   (d) To indicate a packet loss

   Solution: (a) Connection establishment, (b) Exchange of initial sequence number

2. Which of the following is provided by TCP, but not by UDP? (circle all that apply)

   (a) Reliable transfer

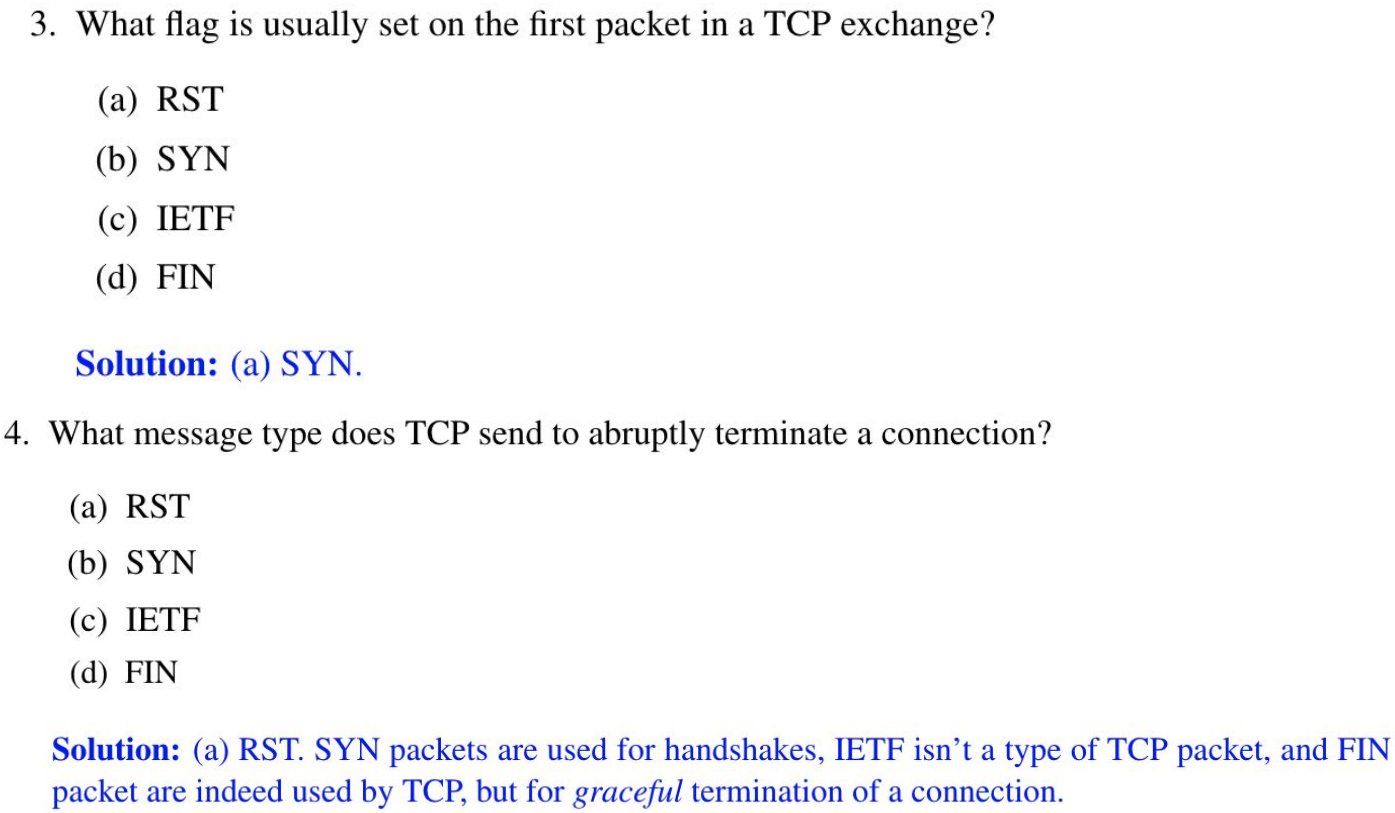
   (b) Congestion control

   (c) Mux and demux from/to application processes

   (d) Byte-stream abstraction

   Solution: (a) Reliable transfer, (b) Congestion control, and (d) Byte-stream abstraction. These are all true by definition of what TCP is. Mux/demux from/to application processes is done using port numbers, which both TCP and UDP use.

Q5. TCP Short Questions



3. What flag is usually set on the first packet in a TCP exchange?

   (a) RST

   (b) SYN

   (c) IETF

   (d) FIN

   Solution: (a) SYN.

4. What message type does TCP send to abruptly terminate a connection?

   (a) RST

   (b) SYN

   (c) IETF

   (d) FIN

   Solution: (a) RST. SYN packets are used for handshakes, IETF isn't a type of TCP packet, and FIN packet are indeed used by TCP, but for *graceful* termination of a connection.

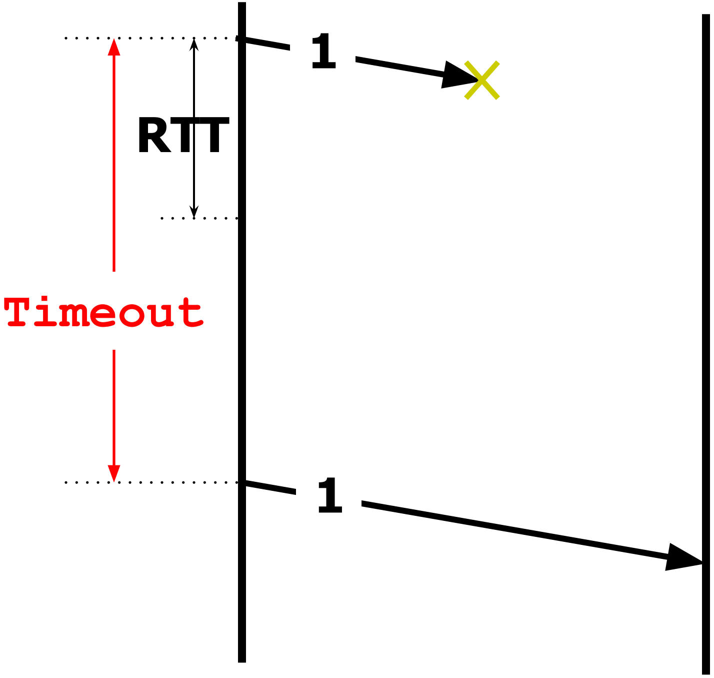
Extra Slides

Estimating RTO

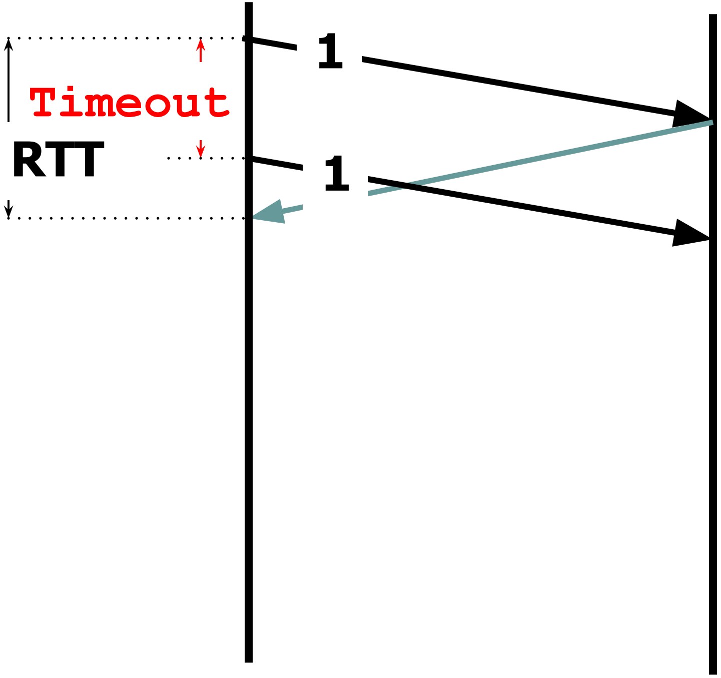
Timeouts and Retransmissions

- Reliability requires retransmitting lost data
- Involves setting timers and retransmitting on timeouts
- TCP only has a single timer
- TCP resets timer whenever new data is ACKed
- Retx packet containing “next byte” when timer expires
- RTO (Retransmit Time Out) is basic timeout value

Setting the Timeout Value (RTO)



Timeout too long → inefficient



Timeout too short → duplicate packets

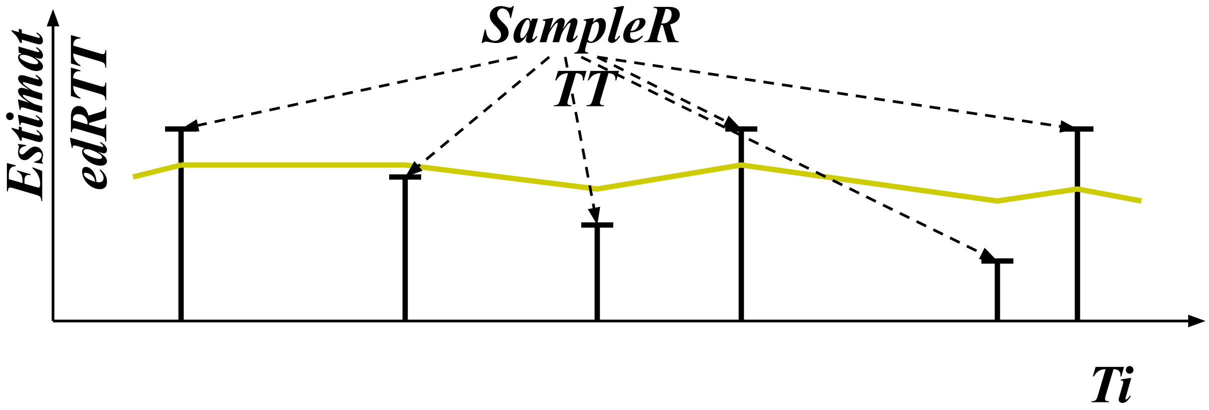
Could Base RTO on RTT Estimation

- Use exponential averaging of RTT samples

$$SampleRTT = AckRcvdTime - SendPacketTime$$

$$EstimatedRTT = \alpha \times EstimatedRTT + (1-\alpha) \times SampleRTT$$

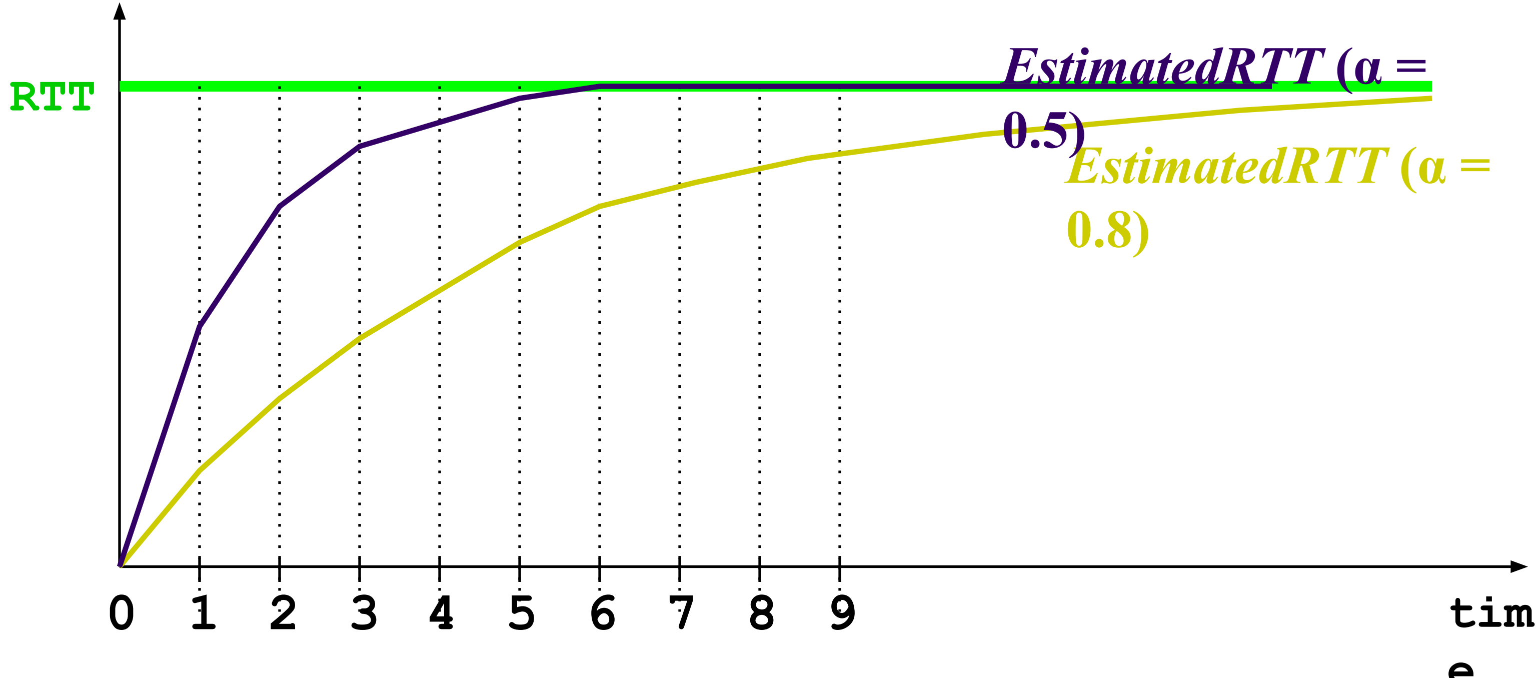
$$0 < \alpha \leq 1$$



Exponential Averaging Example

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

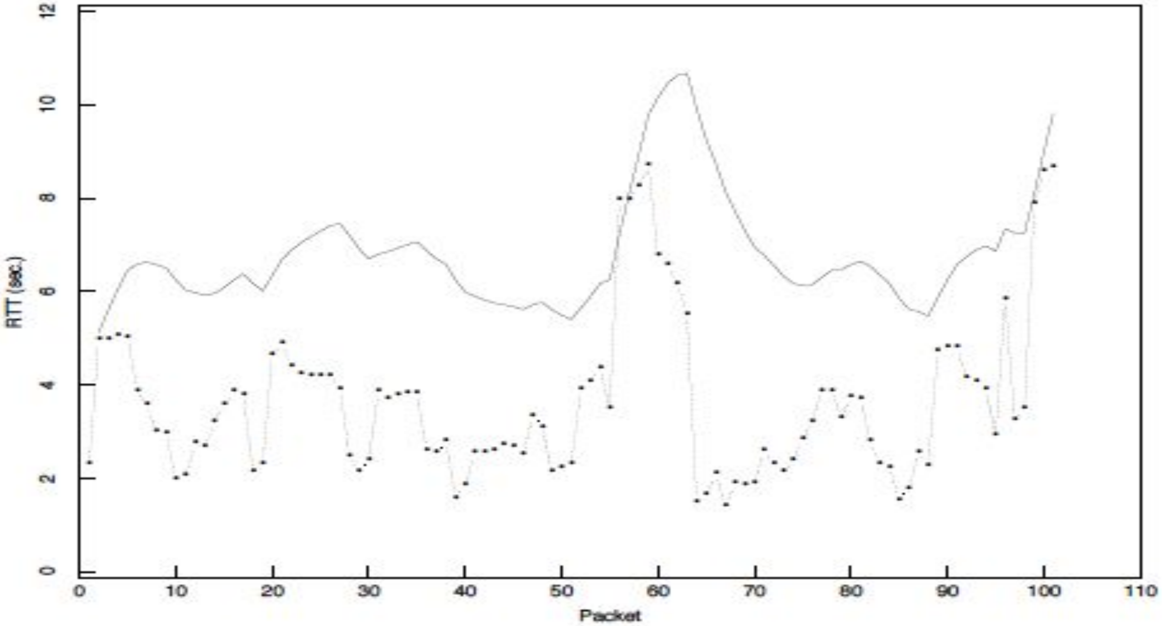
Assume RTT is constant → *SampleRTT* = RTT



Exponential Averaging in Action

Set Timeout estimate (ETO) = 2 × *EstimatedRTT*

Figure 5: Performance of an RFC793 retransmit timer



from Jacobson and Karels, SIGCOMM 1988

Jacobson/Karels Algorithm

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

- Deviation = | SampleRTT – EstimatedRTT |

- EstimatedDeviation: exponential average of Deviation

- ETO = EstimatedRTT + 4 x EstimatedDeviation

With Jacobson/Karels

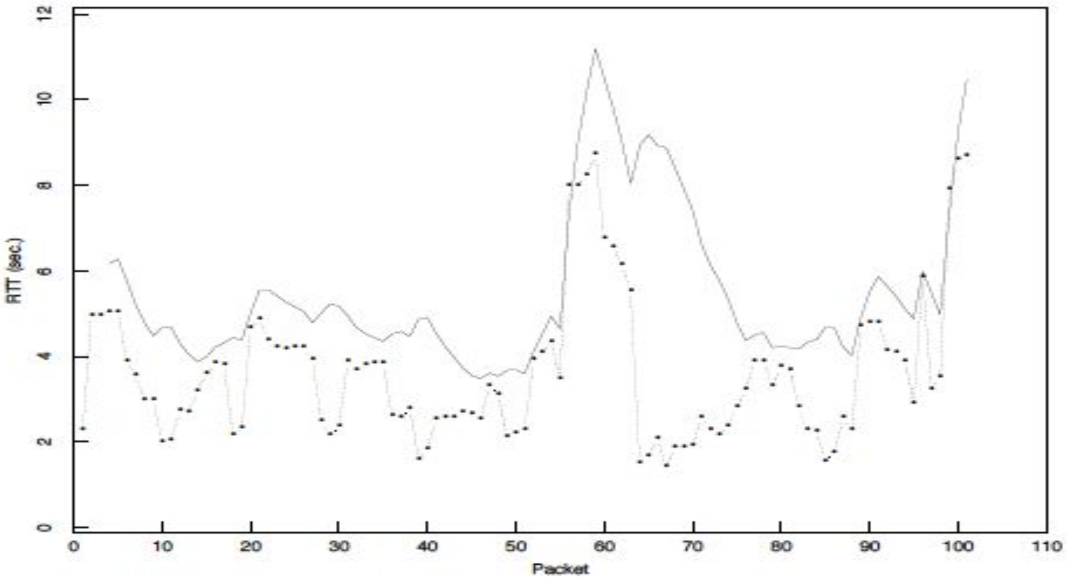
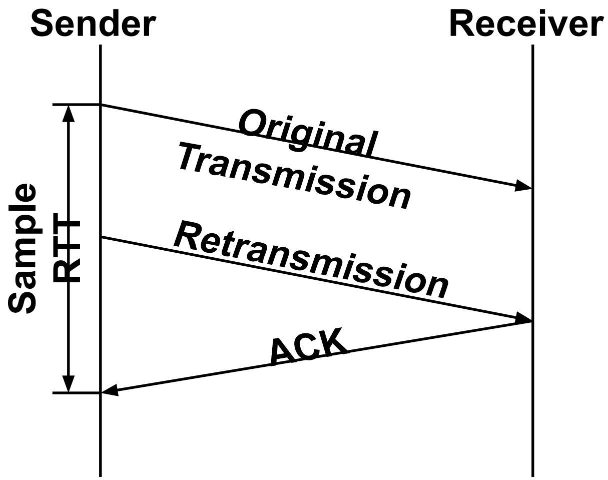
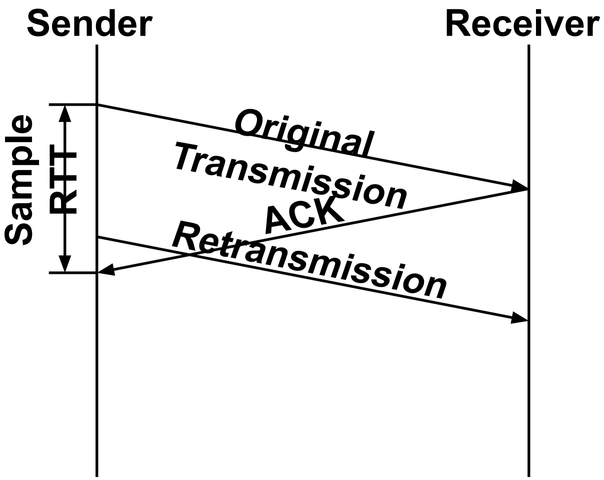


Figure 6: Performance of a Mean+Variance retransmit timer

Problem: Ambiguous Measurements

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





The Following TCP Rules….

- Do not describe current implementations….
- ...but will be considered the “Truth” in this class

TCP Timers

- Two important quantities:
  - RTO: value you set timer to for timeouts
  - ETO: current estimate of appropriate “raw” timeout

- Use exponential averaging to estimate:
  - RTT
  - Deviation = | EstimatedRTT – SampleRTT|

- ETO = EstimatedRTT + 4 x EstimatedDeviation

Use Only “Clean” Samples for ETO

- Only update ETO when you get a clean sample

- Where clean means ACK includes no retransmitted segments

Example

- Send 100, 200, 300
  - 100 means packet whose first byte is 100, last byte is 199
- Receive A200:
  - A200 means bytes up to 199 rec'd, expecting 200 next.
  - **Clean sample**
- 200 times out, resend 200, receive A300
  - No clean samples
- Send 400, 500, receive A600
  - **Clean samples**

Setting RTO

- Every time RTO timer expires, set RTO ← 2·RTO
    - (Up to maximum ≥ 60 sec)

- Every time clean sample arrives set RTO to ETO

Example

- First arriving ACK expects 100 (adv. window=500)
  - Initialize ETO; RTO = ETO
  - Restart timer for RTO seconds (new data ACKed)
    - Remember, TCP only has one timer, not timer per packet
  - Send packets 100, 200, 300, 400, 500
- Arriving ACK expects 300 (A300)
  - Update ETO; RTO = ETO
  - Restart timer for RTO seconds (new data ACKed)
  - Send packets 600, 700
- Arriving ACK expects 300 (A300)

Example (Cont'd)

- Timer goes off
  - RTO = 2*RTO (back off the timer)
  - Restart timer for RTO seconds (it had expired!)
  - Resend packet 300
- Arriving ACK expects 800
  - Don't update ETO (ACK includes a retransmission)
  - Restart timer for RTO seconds (new data ACKed)
  - Send packets 800, 900, 1000, 1100, 1200

Example (Cont’d)

- Arriving ACK expects 1000
  - Update ETO; RTO = ETO
  - Restart timer for RTO seconds (new data ACKed)
  - Send packets 1300, 1400

- … Connection continues …

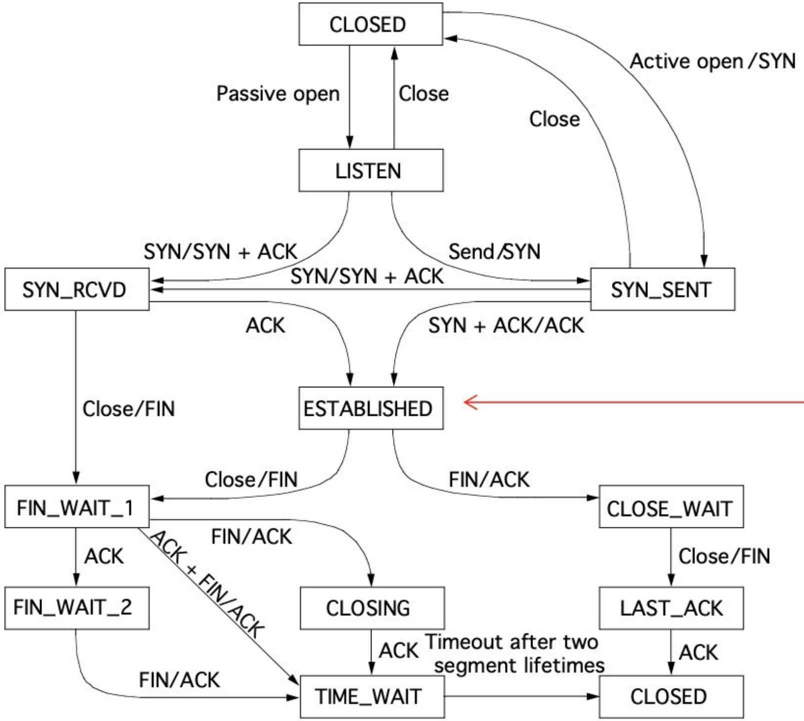
This is all very interesting, but…..

- Implementations often use a coarse-grained timer
  - 200 msec is typical (depends on implementation)

- So what?
  - Above algorithms are largely irrelevant
  - **Incurring a timeout is expensive**

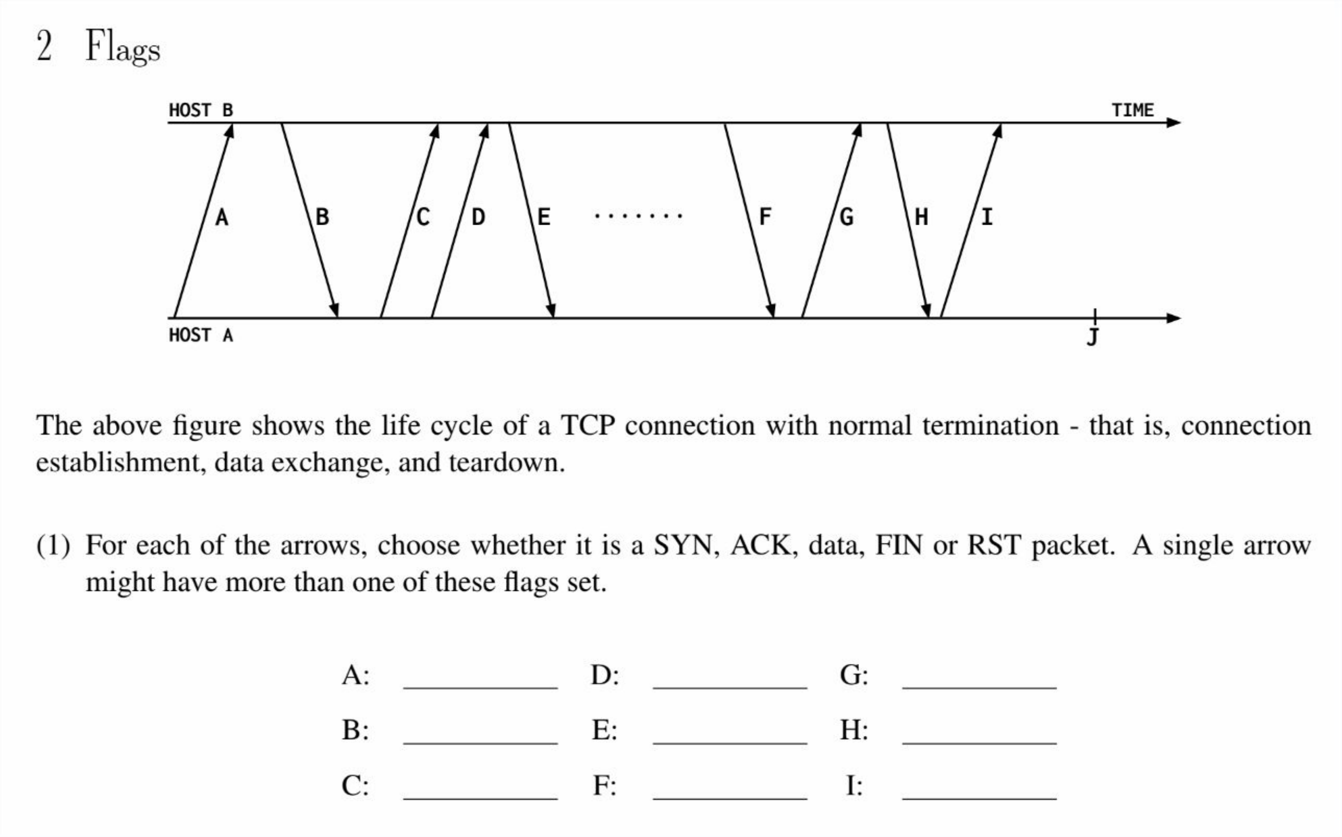
- So we rely on duplicate ACKs

TCP State Transitions



Data, ACK exchanges are in here

Bonus Question: TCP Connection Life Cycle



2 Flags

HOST B
TIME
A
B
C
D
E
·······
F
G
H
I
HOST A
J

The above figure shows the life cycle of a TCP connection with normal termination - that is, connection establishment, data exchange, and teardown.

(1) For each of the arrows, choose whether it is a SYN, ACK, data, FIN or RST packet. A single arrow might have more than one of these flags set.

A: ____________

B: ____________

C: ____________

D: ____________

E: ____________

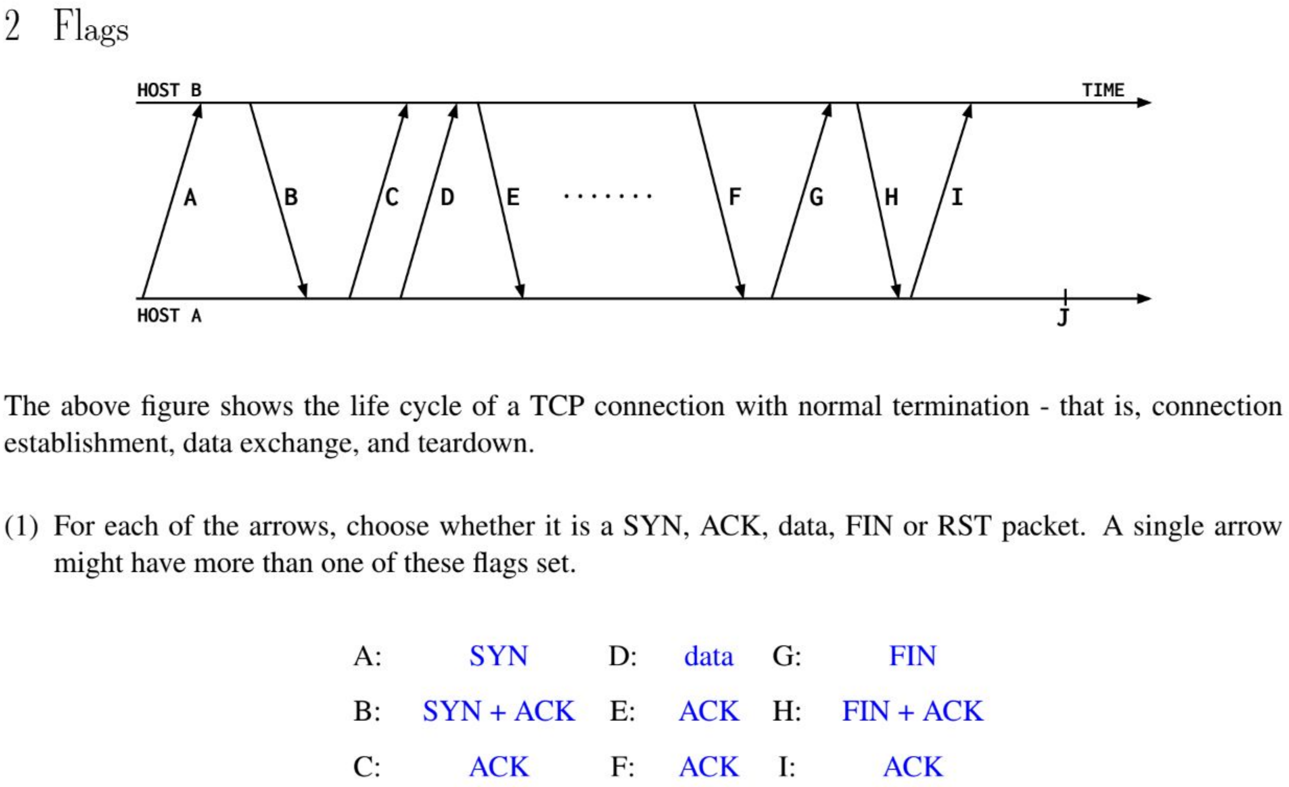
F: ____________

G: ____________

H: ____________

I: ____________

Bonus Question: TCP Connection Life Cycle



2 Flags

HOST B
TIME
A
B
C
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E
·······
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A:	SYN	D:	data	G:	FIN
B:	SYN + ACK	E:	ACK	H:	FIN + ACK
C:	ACK	F:	ACK	I:	ACK