EN.601.414/614 Computer Networks

TCP

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https://github.com/xinjin/course-net

Agenda

- **From reliable data transfer to TCP**
- **TCP connection setup**
- **TCP connection teardown**

Recap: Designing a reliable transport protocol

- **Stop and wait is correct but inefficient**
  - Works packet by packet (of size DATA)
  - Throughput is (DATA/RTT)
- **Sliding window: use pipelining to increase throughput**
  - n packets at a time results in higher throughput
  - MIN(n*DATA/RTT, Link Bandwidth)

Recap: Acknowledgements

- **Cumulative**
  - Acknowledge many packets at a time
- **Selective**
  - Acknowledge individual packets
- **How GBN and SR use these two can be slightly different**

Recap: Sliding window protocols

- **Resending packets: two canonical approaches**
  - Go-Back-N
  - Selective Repeat
- **Many variants that differ in implementation details**

Recap: Go-Back-N (GBN)

- **Sender transmits up to n unacknowledged packets**
- **Receiver only accepts packets in order**
  - Discards out-of-order packets (i.e., packets other than B+1)
- **Receiver uses cumulative acknowledgements**
  - i.e., sequence# in ACK = next expected in-order sequence#
- **Sender sets timer for 1st outstanding ack (A+1)**
- **If timeout, retransmit A+1, … , A+n**

Recap: Selective Repeat (SR)

- **Sender: transmit up to n unacknowledged packets**
- **Assume packet k is lost, k+1 is not**
  - Receiver: indicate packet k+1 correctly received
  - Sender: retransmit only packet k on timeout
- **Efficient in retransmissions but complex book-keeping**
  - Need a timer/flag per packet

Assignment 2: Reliable Transport

- **Reliable transport on top of UDP**
- **Sender**
  - ➢Send packets in the sliding window
  - ➢Reset timer = 500ms
  - ➢If receive expected ACK, move forward the window; otherwise, wait for timer to expire
- **Receiver:** improved GBN
  - ➢Receive packets in the sliding window, send cumulative ACKs, and move forward the window accordingly
- **Optimization:** use selective ACKs

Assignment 2: Reliable Transport

- **Tips**
  - Checksum
    - SignedIntField(“checksum”, 0)
    - binascii.crc32(str(pkt)) & 0xffffffff
  - Testing, debugging: learn to write your own test scripts
    - Idea: write a Python program with UDP sockets acting as a proxy between sender and receiver
    - The Python program can corrupt, drop, reorder, duplicate, and delay packets

TCP: Transmission Control Protocol

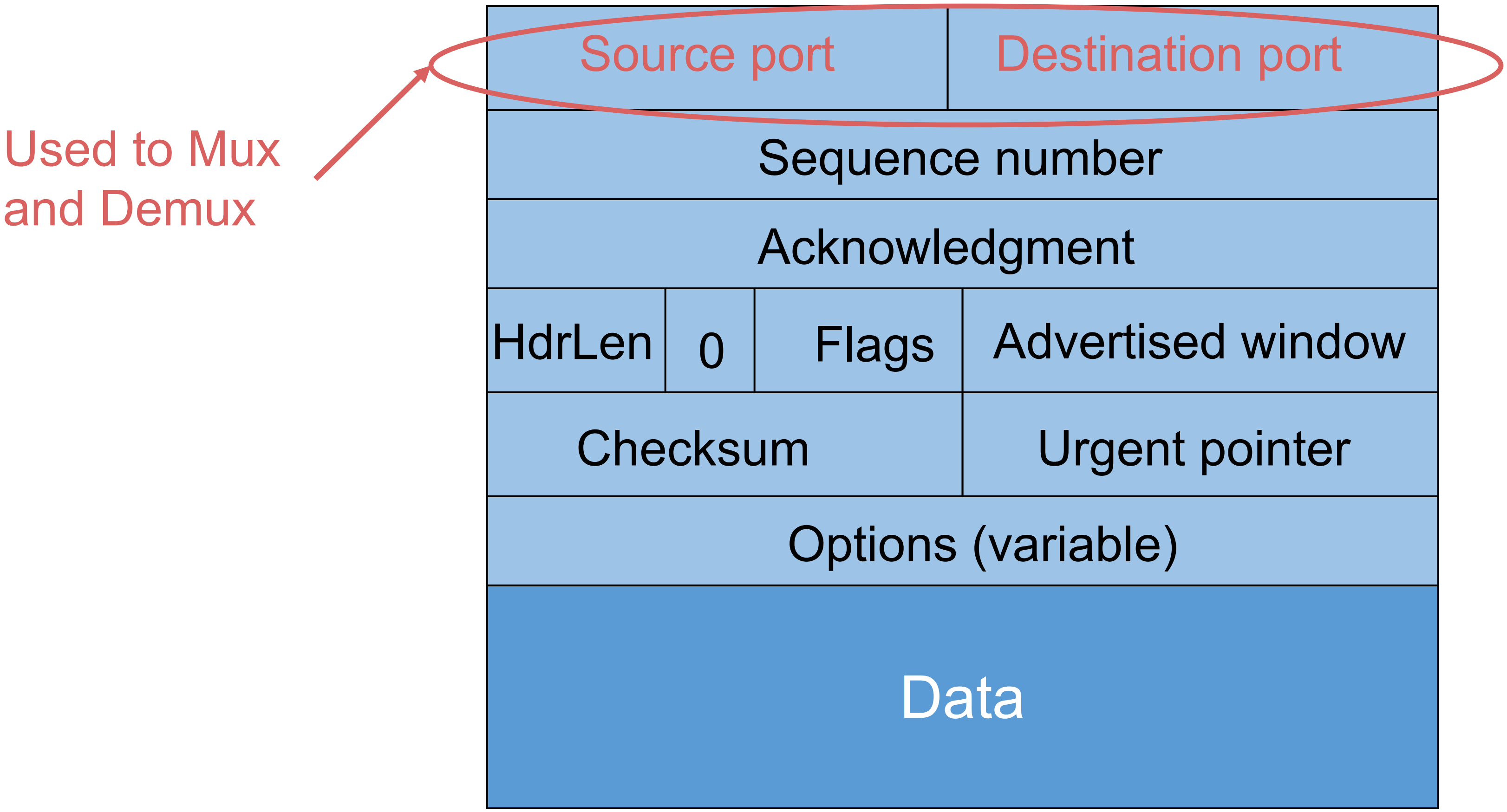
The TCP Abstraction

- **TCP delivers a reliable, in-order, byte stream**
- **Reliable: TCP resends lost packets**
    - ➢Until it gives up and shuts down connection
- **In-order: TCP only hands consecutive chunks of data to application**
- **Byte stream: TCP assumes there is an incoming stream of data, and attempts to deliver it to app**

What does TCP use from what we’ve seen so far?

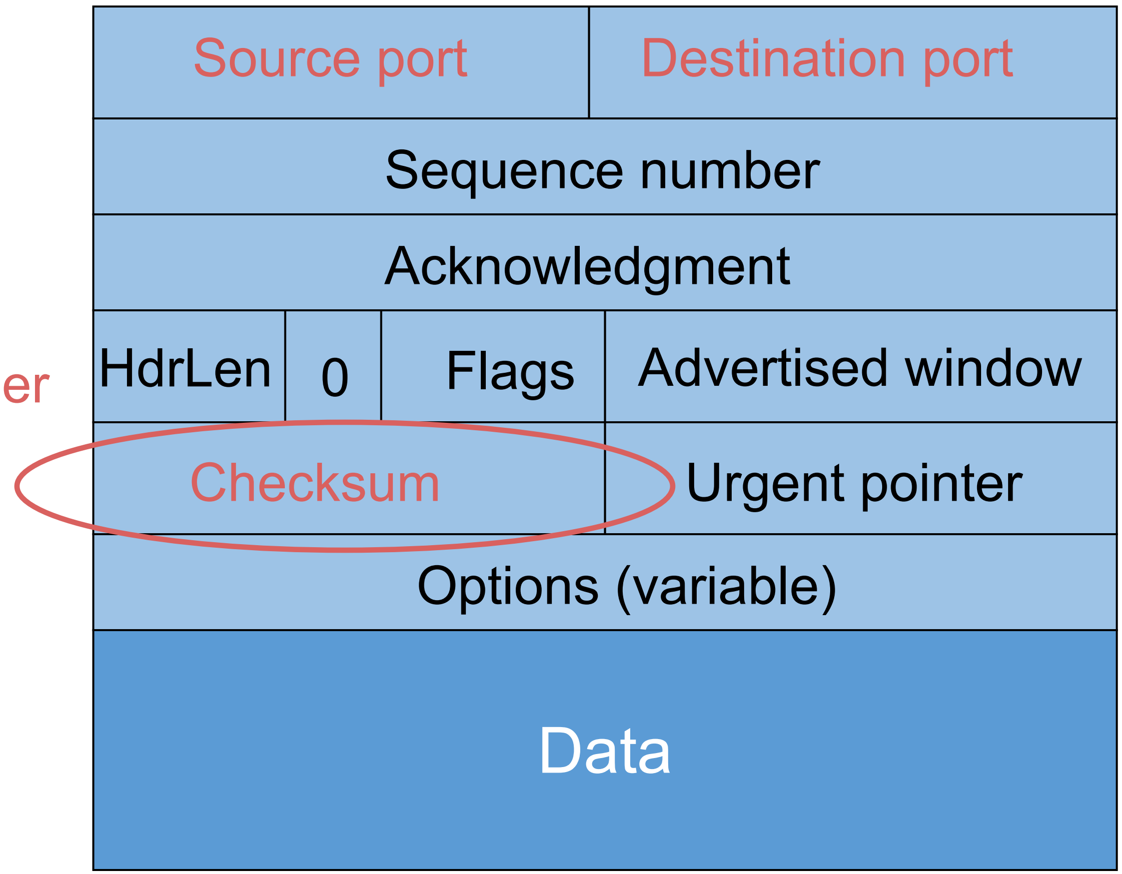
- **Most of what we’ve seen**
  - Checksums
  - Sequence numbers are byte offsets
  - Sender and receiver maintain a sliding window
  - Receiver sends cumulative acknowledgements (like GBN)
    - Sender maintains a single retransmission timer
  - Receivers buffer out-of-sequence packets (like SR)
- **Few more: fast retransmit, timeout estimation algorithms, etc.**

TCP header


Used to Mux
and Demux
Source port
Destination port
Sequence number
Acknowledgment
HdrLen
0
Flags
Advertised window
Checksum
Urgent pointer
Options (variable)
Data

TCP header

Computed
over pseudo-header
and data

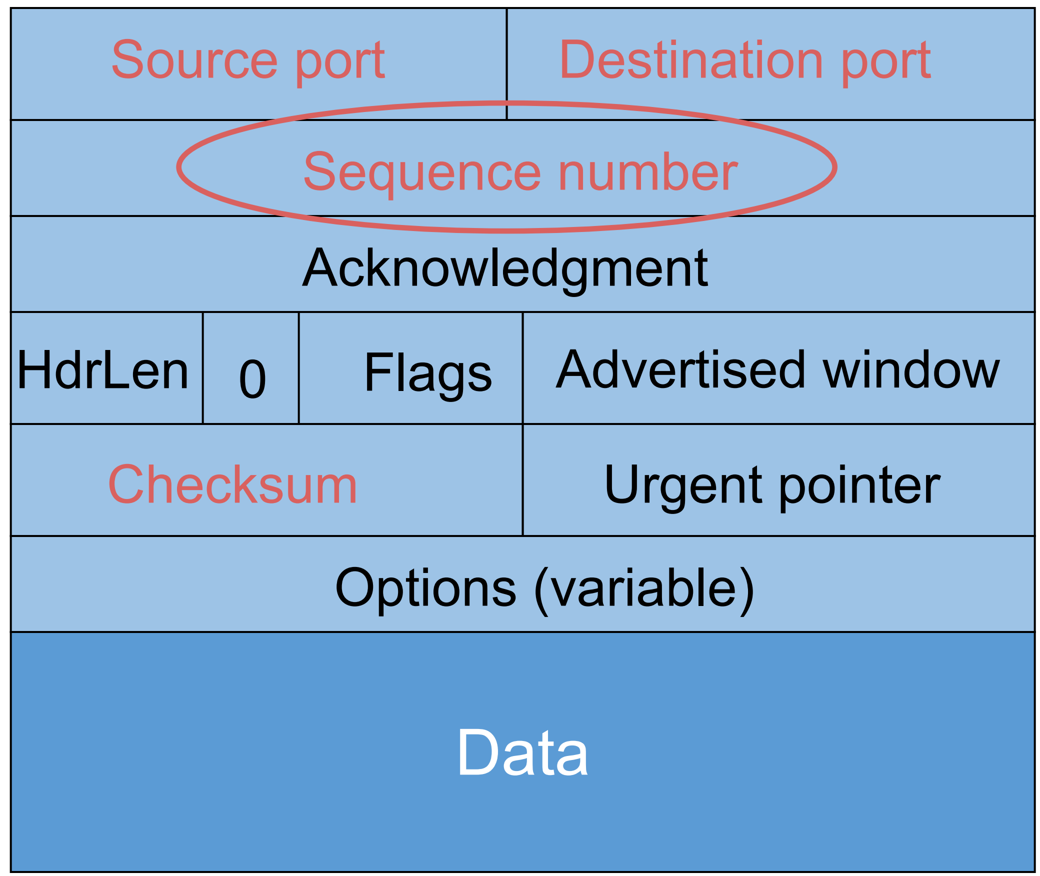


What does TCP do?

- **Most of what we've seen**
  - Checksum
  - Sequence numbers are byte offsets

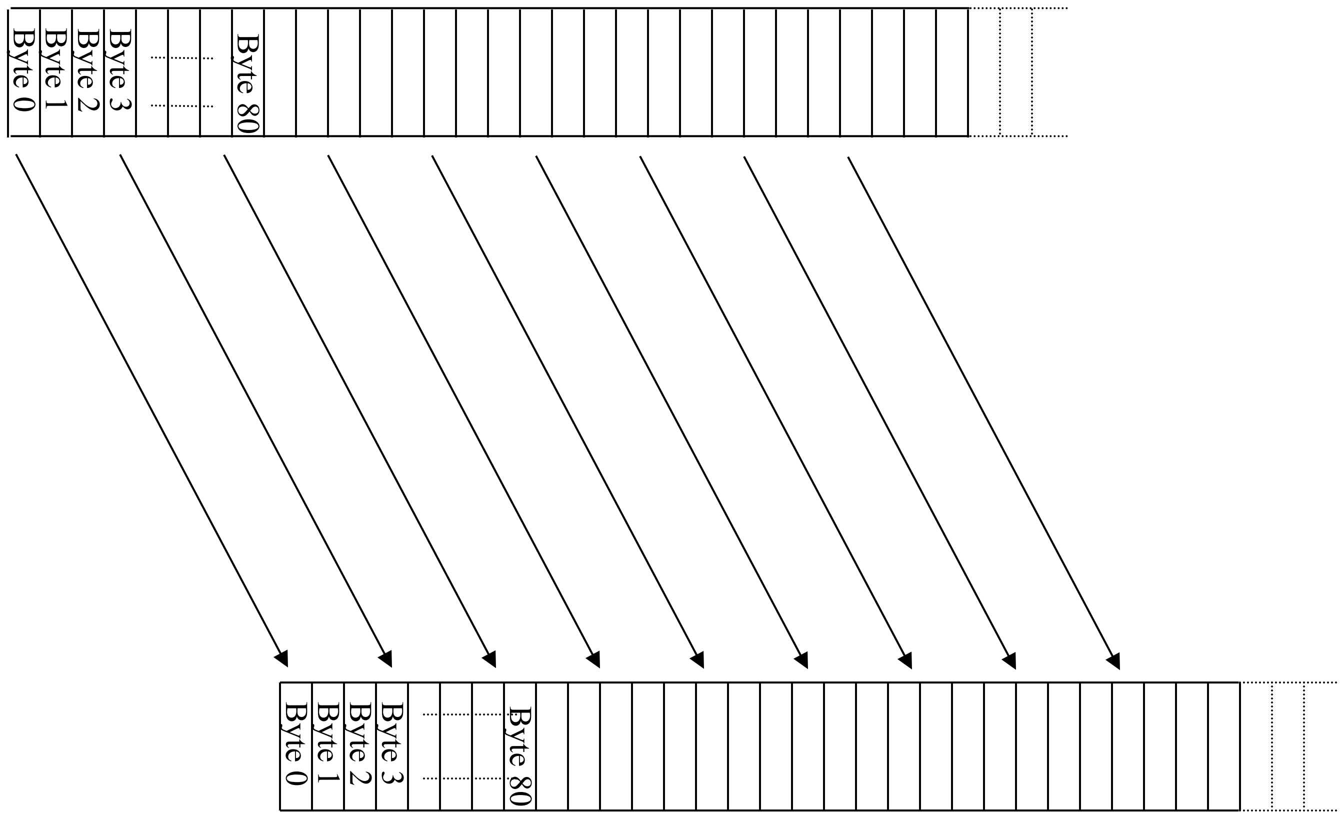
TCP header

Byte offsets (NOT packet id), because TCP is a byte stream



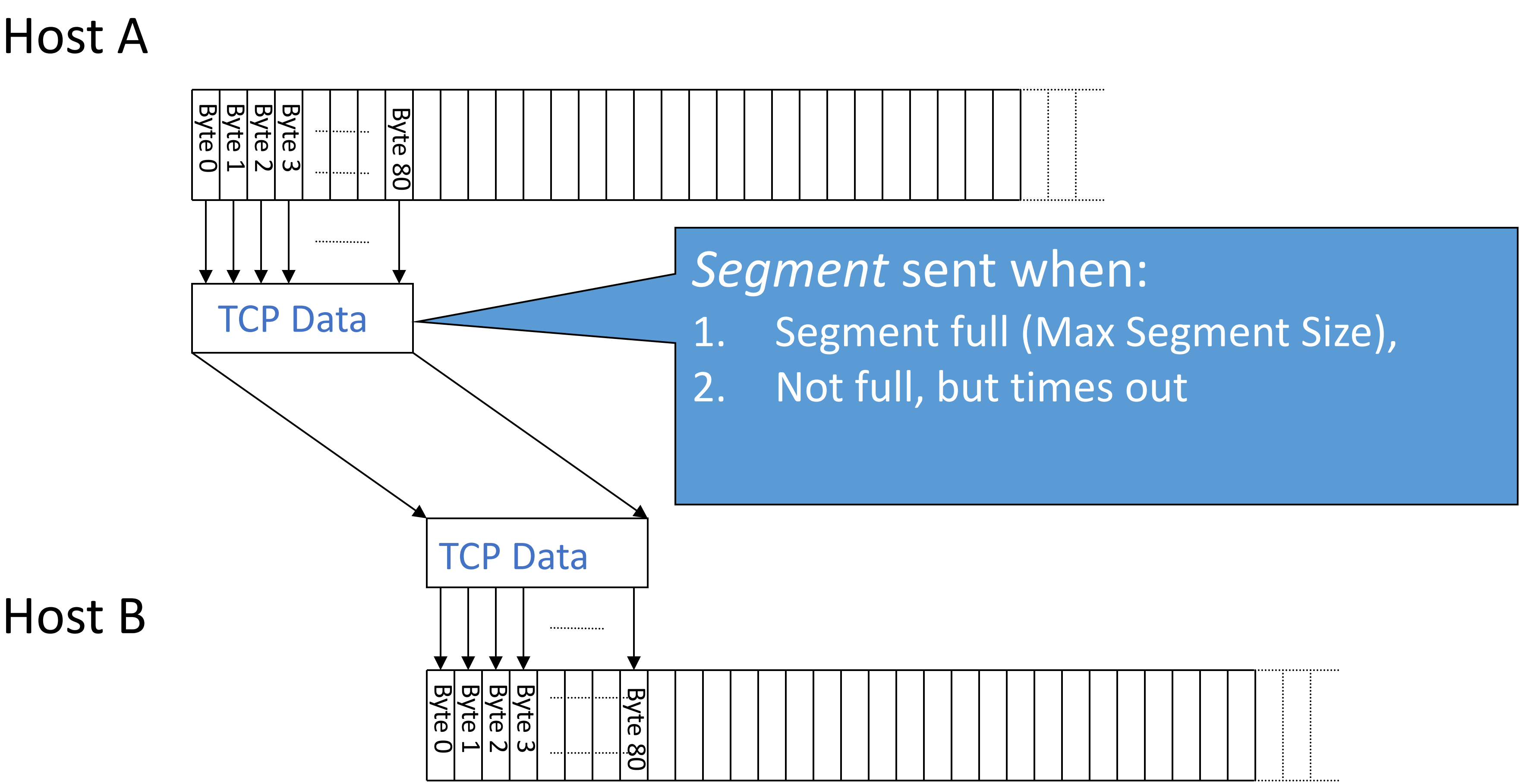
TCP “stream of bytes” service...

Application @ Host A

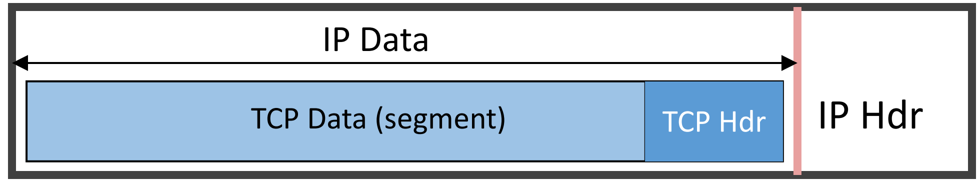


Application @ Host B

… provided using TCP “segments”

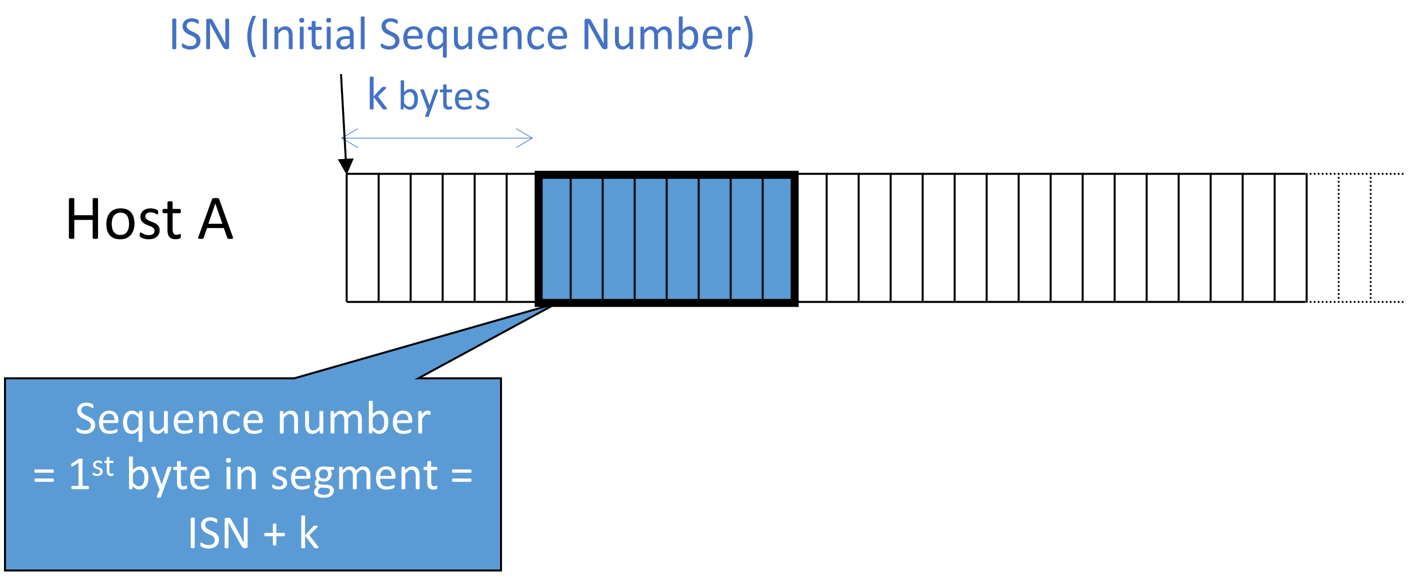


TCP segment

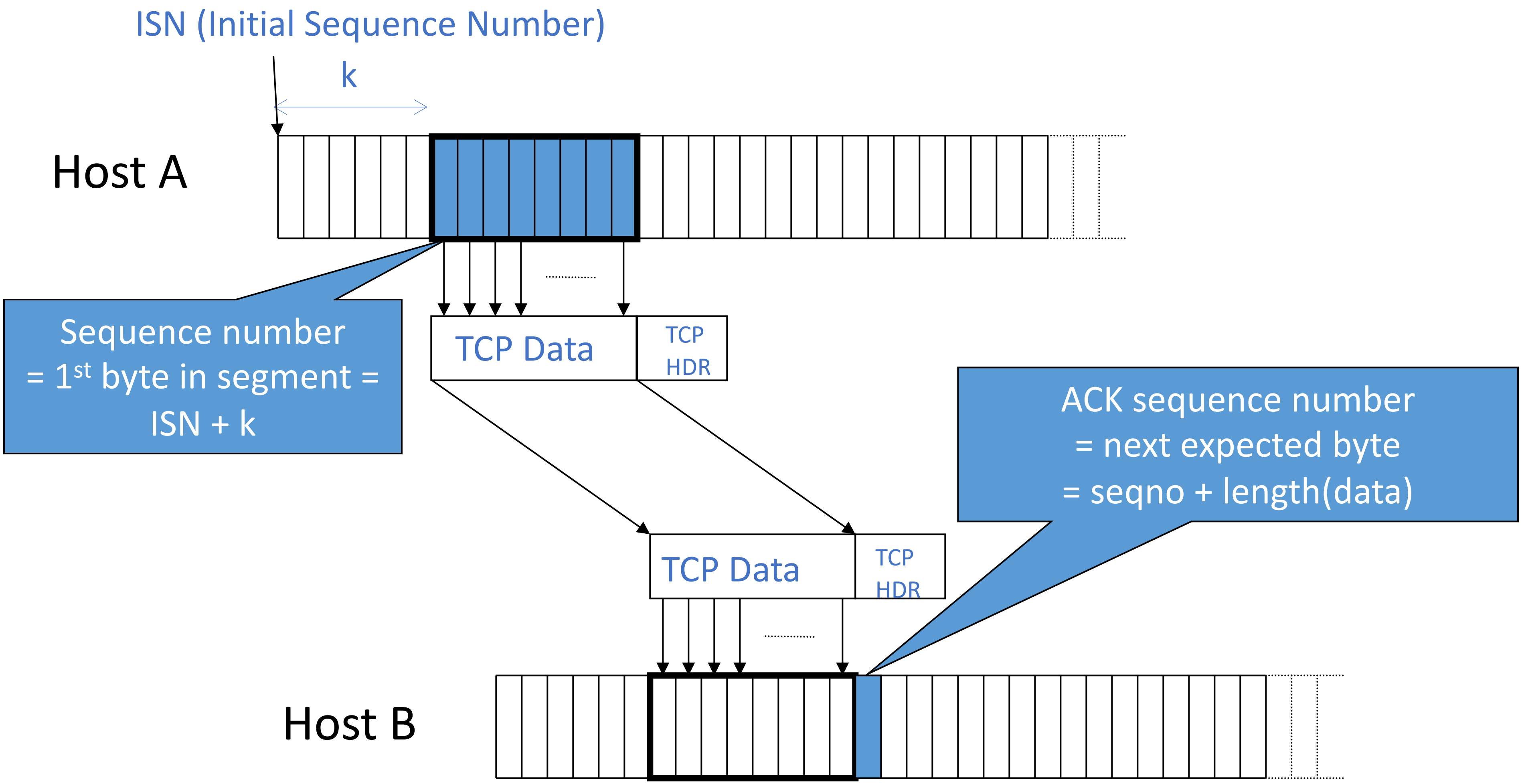


- **IP packet**
  - No bigger than Maximum Transmission Unit (MTU)
  - E.g., up to 1500 bytes with Ethernet
- **TCP packet**
  - IP packet with a TCP header and data inside
  - TCP header $\geq$ 20 bytes long
- **TCP segment**
  - No more than Maximum Segment Size (MSS) bytes
  - E.g., up to 1460 consecutive bytes from the stream
  - MSS = MTU – (IP header) – (TCP header)

Sequence numbers

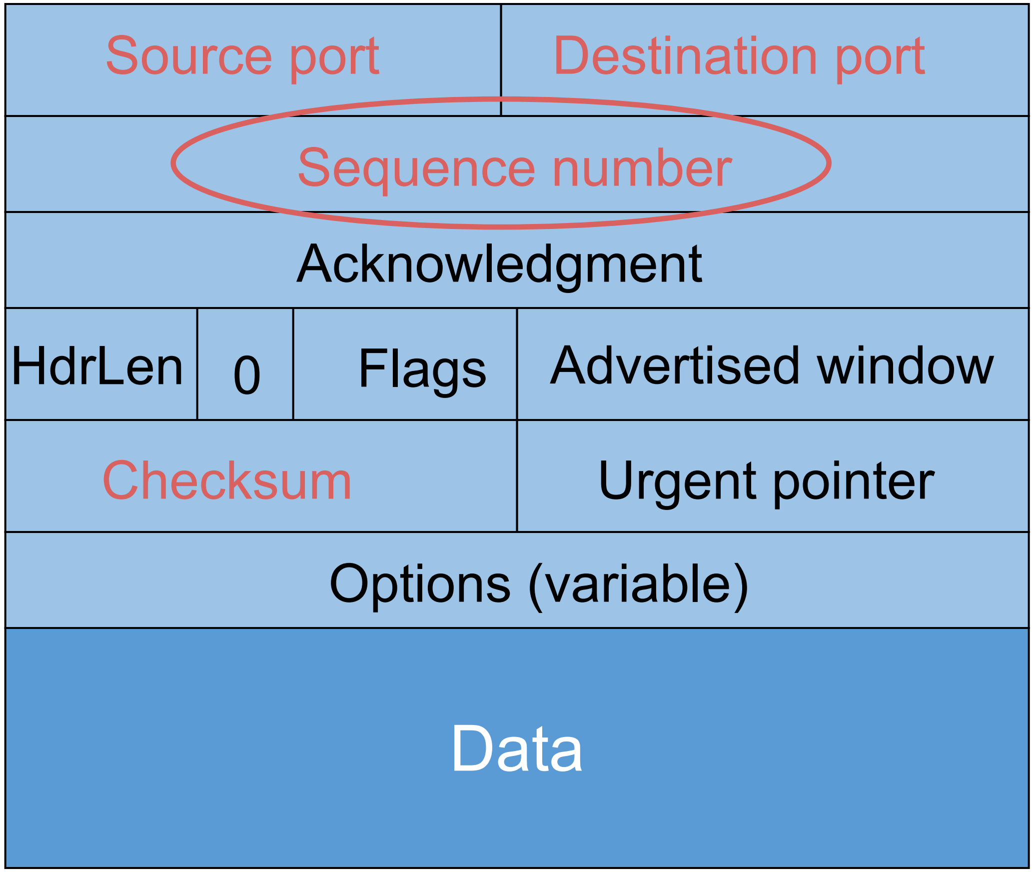


Sequence numbers



TCP header

Starting byte offset of data carried in this segment



What does TCP do?

- **Most of what we've seen**
  - Checksum
  - Sequence numbers are byte offsets
  - Receiver sends cumulative acknowledgements (like GBN)

ACKs and sequence numbers

- **Sender sends packet**
  - Data starts with sequence number X
  - Packet contains B bytes [X, X+1, X+2, ..., X+B-1]
- **Upon receipt of packet, receiver sends an ACK**
  - If all data prior to X already received:
    - ACK acknowledges X+B (because that is next expected byte)
  - If highest in-order byte received is Y s.t. (Y+1) < X
    - ACK acknowledges Y+1
    - Even if this has been ACKed before

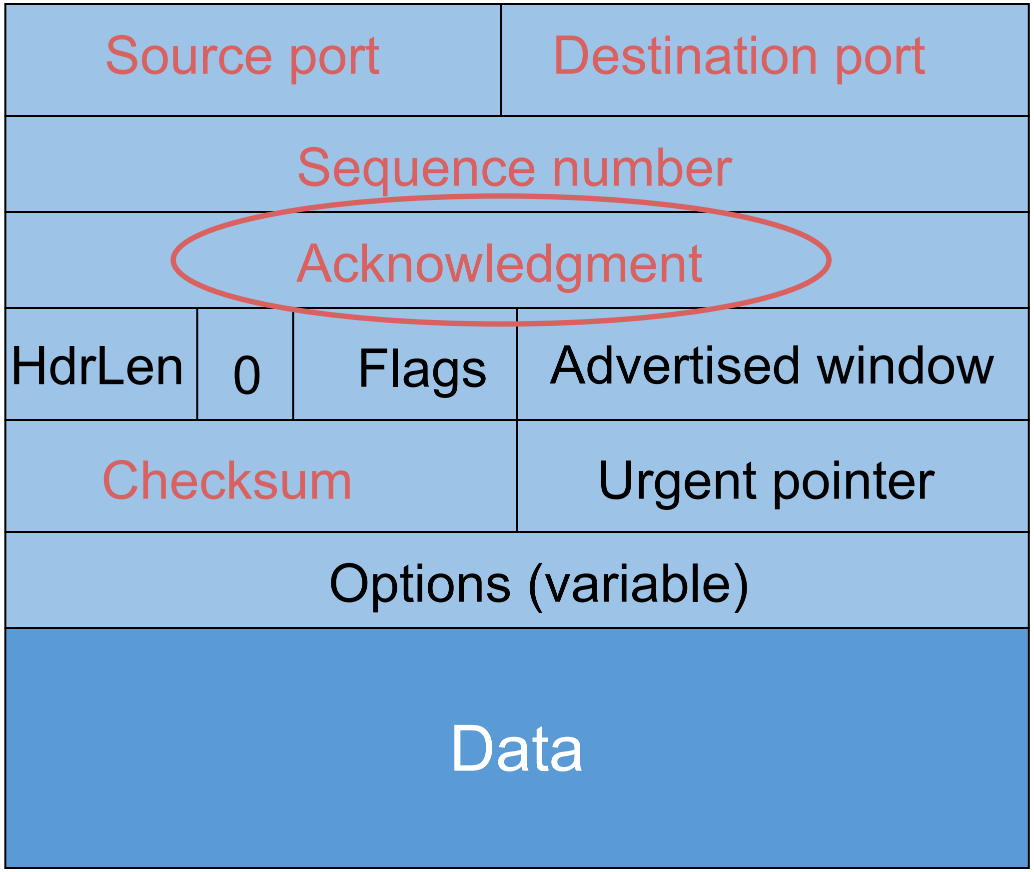
Typical operation

- **Sender: seqno=X, length=B**
- **Receiver: ACK=X+B**
- **Sender: seqno=X+B, length=B**
- **Receiver: ACK=X+2B**
- **Sender: seqno=X+2B, length=B**

- **Seqno of next packet is same as last ACK field**

TCP header

Acknowledgment gives seqno just beyond highest seqno received **in order**



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)

Loss with cumulative ACKs

- **Sender sends packets with 100B and seqnos.:**
  - 100, 200, 300, 400, 500, 600, 700, 800, 900, …
- **Assume the fifth packet (seqno 500) is lost, but no others**
- **Stream of ACKs will be:**
  - 200, 300, 400, 500 (seqno:600), 500 (seqno:700), 500 (seqno:800), 500 (seqno:900),…

What does TCP introduce?

- **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)
- **Introduces fast retransmit: duplicate ACKs trigger early retransmission**

Loss with cumulative ACKs

- **Duplicate ACKs are a sign of an isolated loss**
  - ➢The lack of ACK progress means 500 hasn't been delivered
  - ➢Stream of ACKs means some packets are being delivered
- **Trigger retransmission upon receiving k duplicate ACKs**
    - TCP uses k=3
    - Faster than waiting for timeout

Loss with cumulative ACKs

- **Two choices after resending:**
  - ➢Send missing packet and move sliding window by the number of dup ACKs
    - Speeds up transmission, but might be wrong
  - ➢Send missing packet, and wait for ACK to move sliding window
    - Is slowed down by single dropped packets

- **Which should TCP do?**

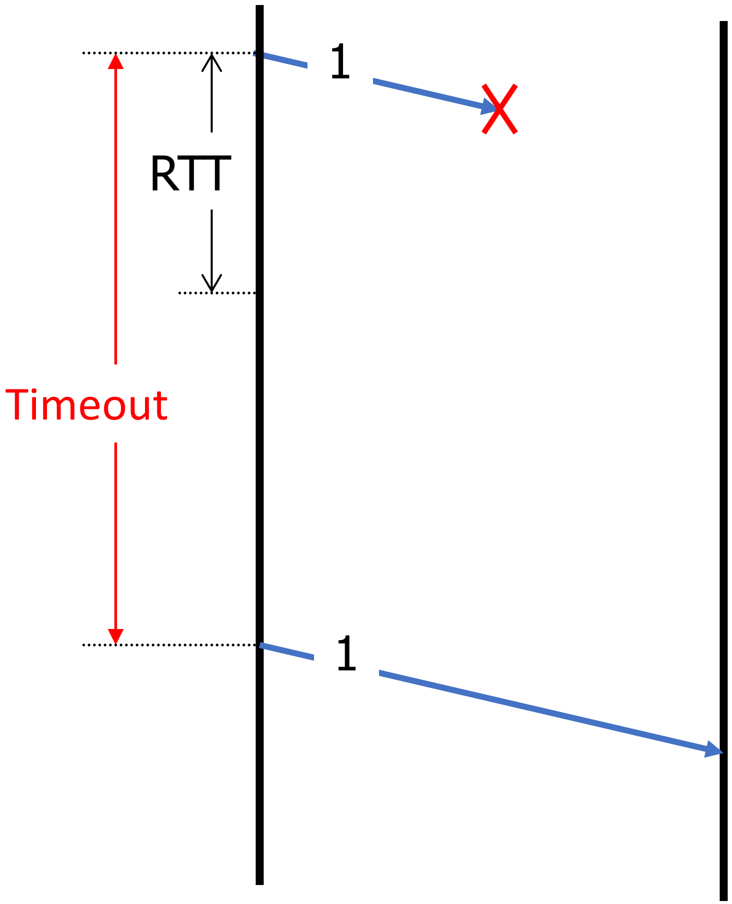
What does TCP introduce?

- **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**

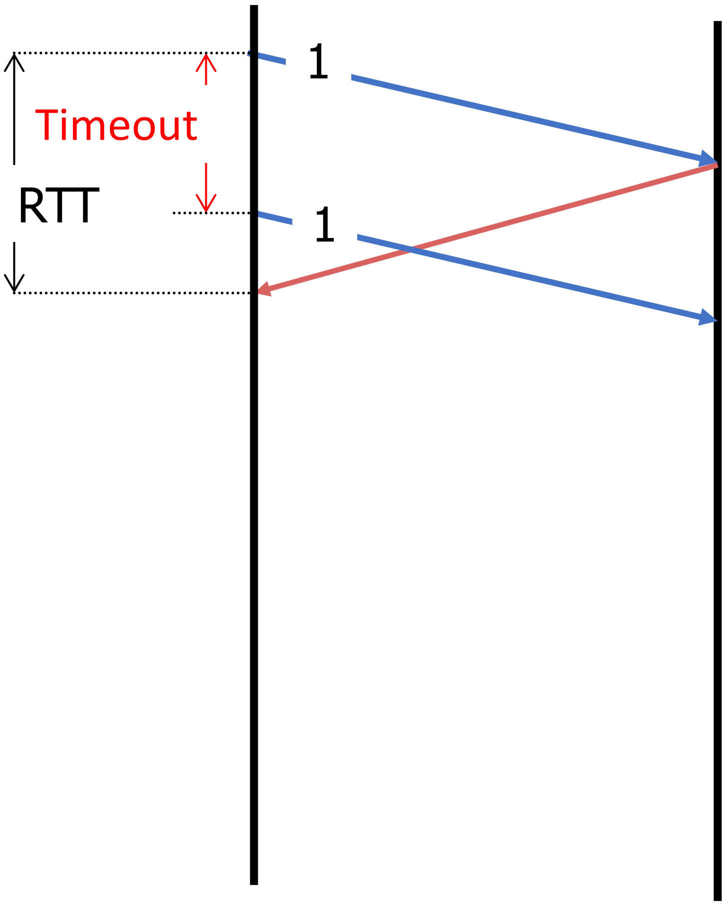
Retransmission timeout

- **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

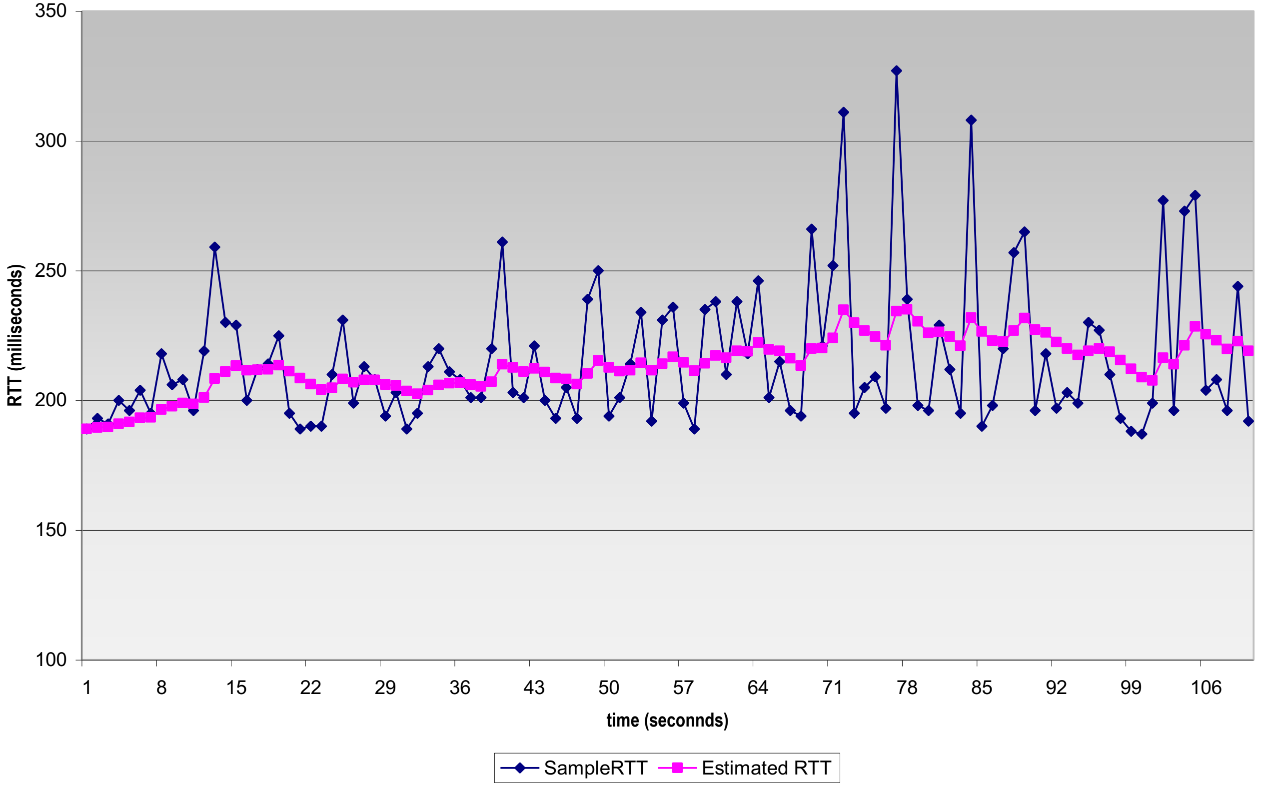
Retransmission timeout

- **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

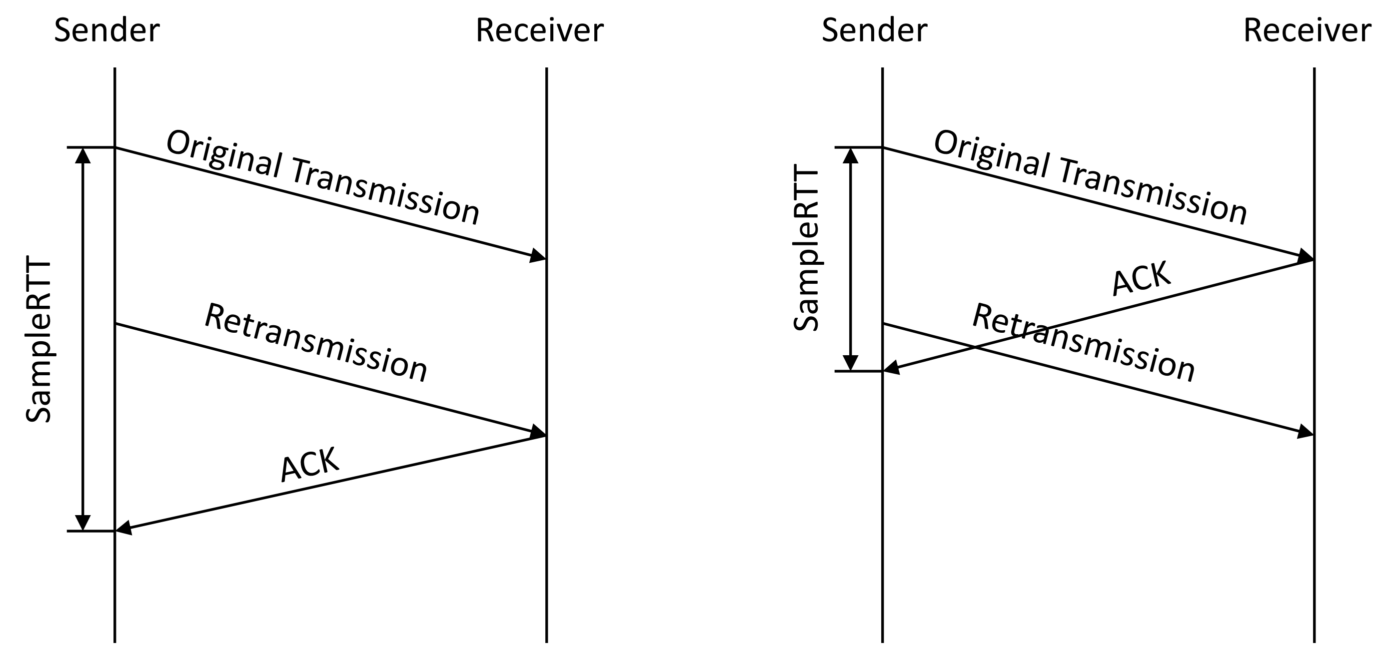
- **Exponential weighted average of RTT samples**

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

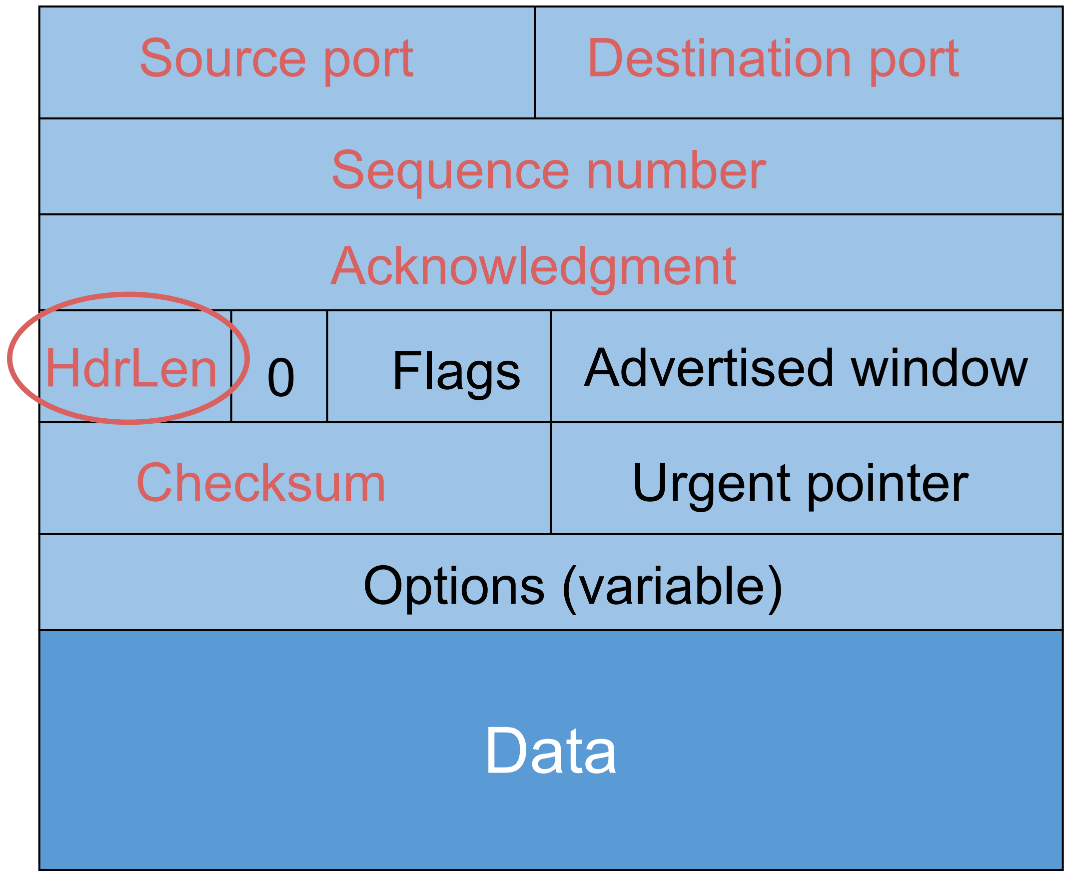
- **Problem: need to better capture variability in RTT**
  - Directly measure deviation

- **Deviation = | SampleRTT – EstimatedRTT |**
- **DevRTT: exponential average of Deviation**

- **RTO = EstimatedRTT + 4 x DevRTT**

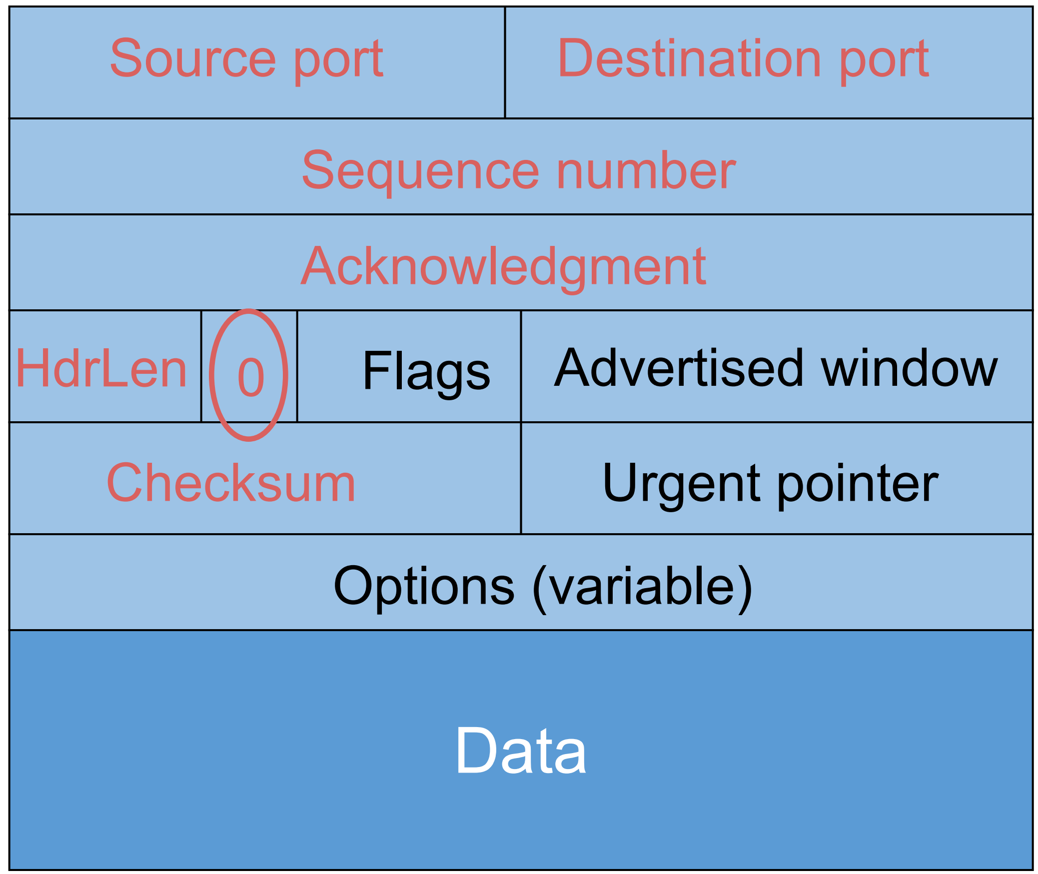
TCP header

Number of 4-byte words in the header



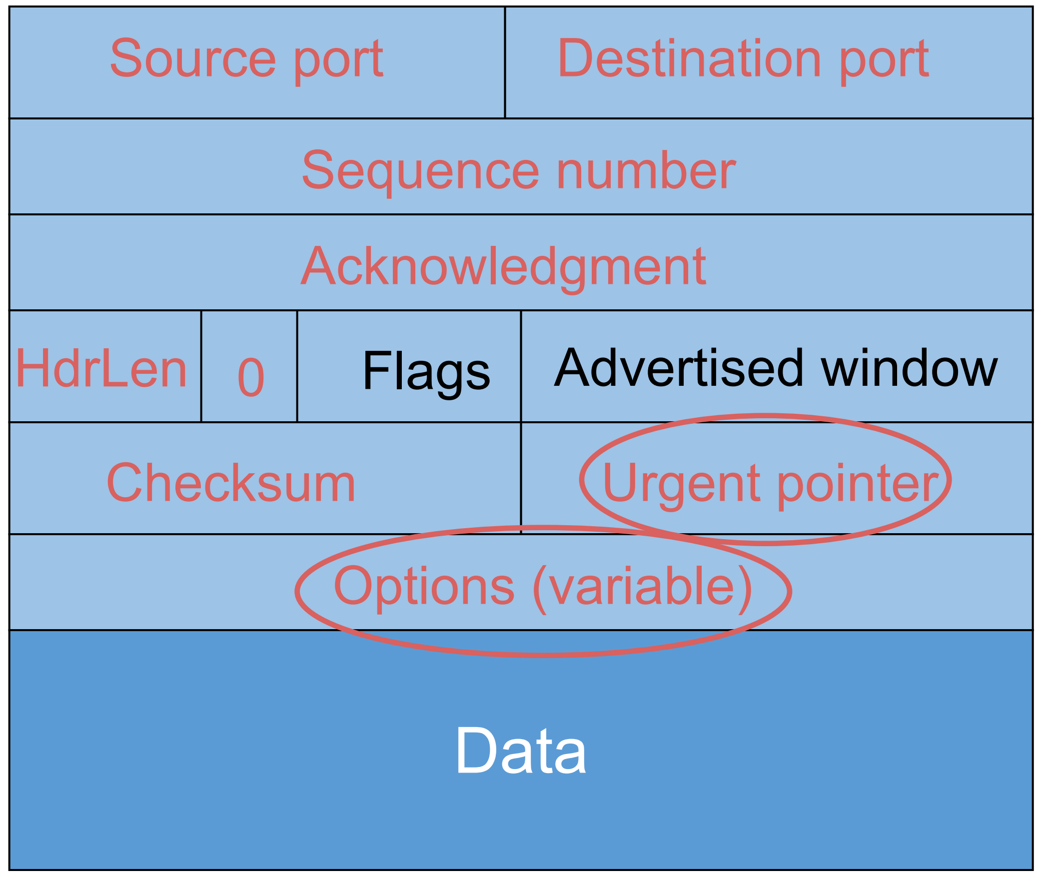
TCP header

Reserved for future use and should be 0



TCP header

Not commonly used



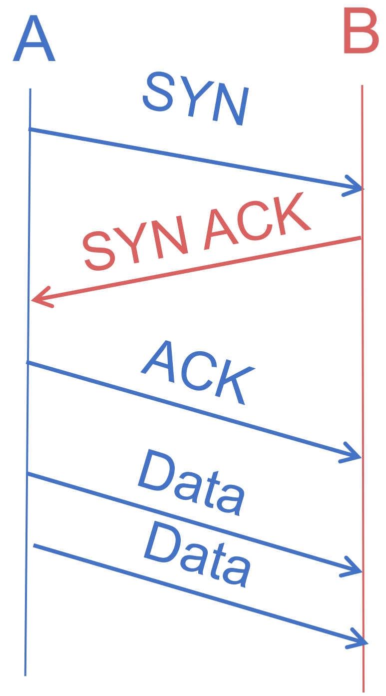
TCP Connection Establishment

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**

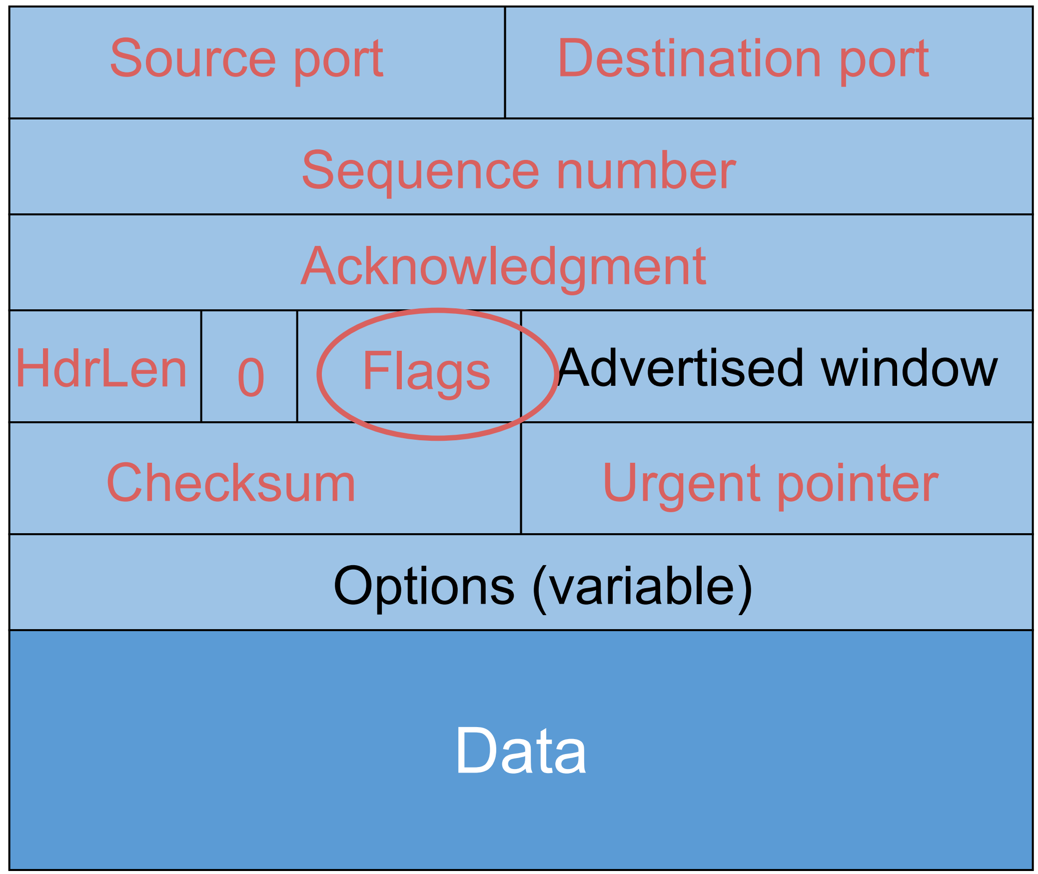
Establishing a TCP connection

- **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



TCP header

Flags:
SYN
ACK
FIN
RST
PSH
URG



Step 1: A's initial SYN packet

A tells B to open a connection

A's port		B's port	
A's Initial Sequence Number			
N/A			
5	0	SYN	Advertised window
Checksum		Urgent pointer	

Step 1: B's SYN-ACK packet

B tells it accepts and is ready to accept next packet

<table>
<tr><td colspan="3">B's port</td><td>A's port</td></tr>
<tr><td colspan="4">B's Initial Sequence Number</td></tr>
<tr><td colspan="4">ACK=A's ISN+1</td></tr>
<tr><td>5</td><td>0</td><td>SYN|ACK</td><td>Advertised window</td></tr>
<tr><td colspan="3">Checksum</td><td>Urgent pointer</td></tr>
</table>

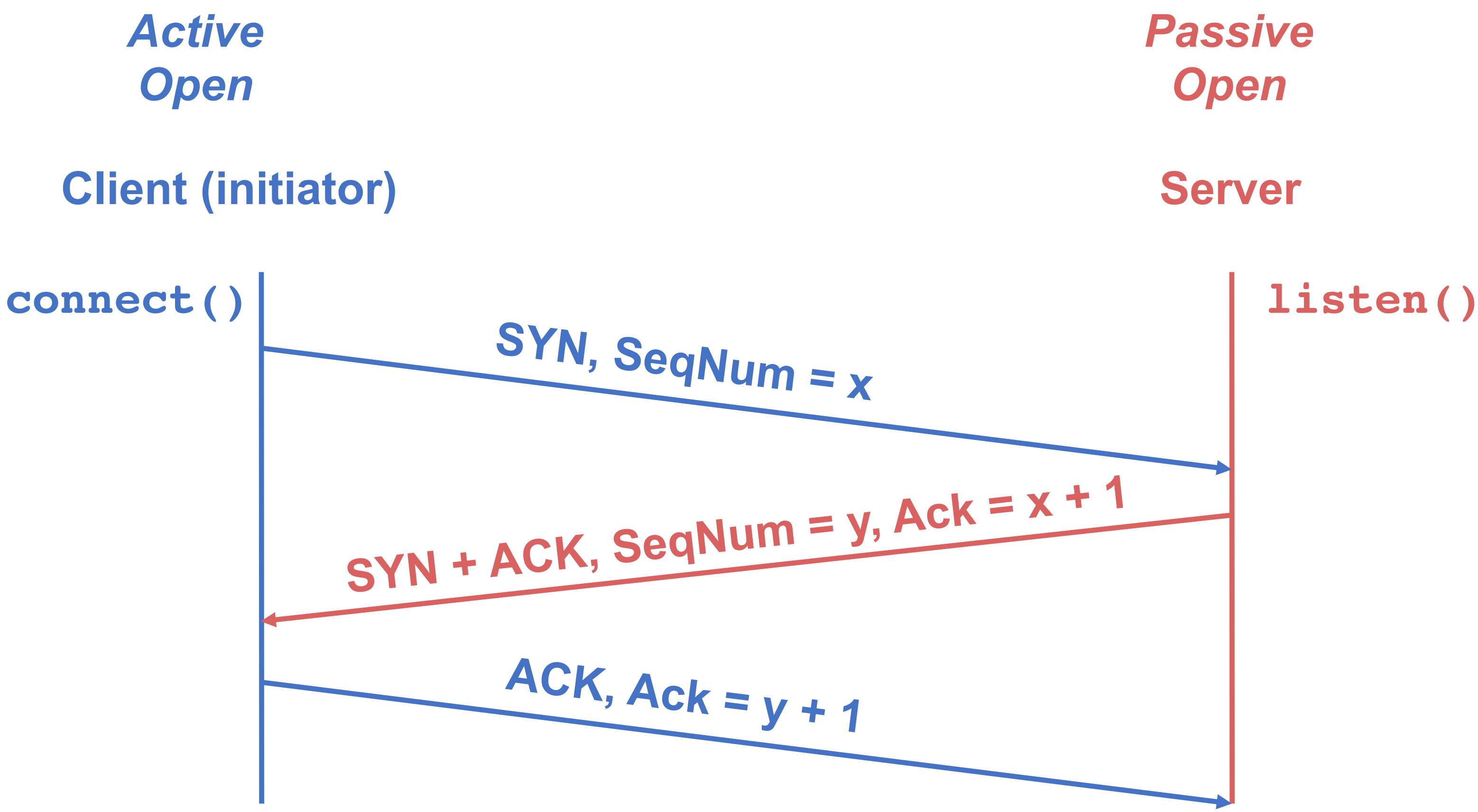
B's Initial Sequence Number = A's Initial Sequence Number?

Step 1: A's ACK to SYN-ACK

A tells B to open a connection

A's port		B's port	
A's Initial Sequence Number			
ACK=B's ISN+1			
5	0	ACK	Advertised window
Checksum		Urgent pointer	

TCP's 3-Way handshaking



What if the SYN Packet Gets Lost?

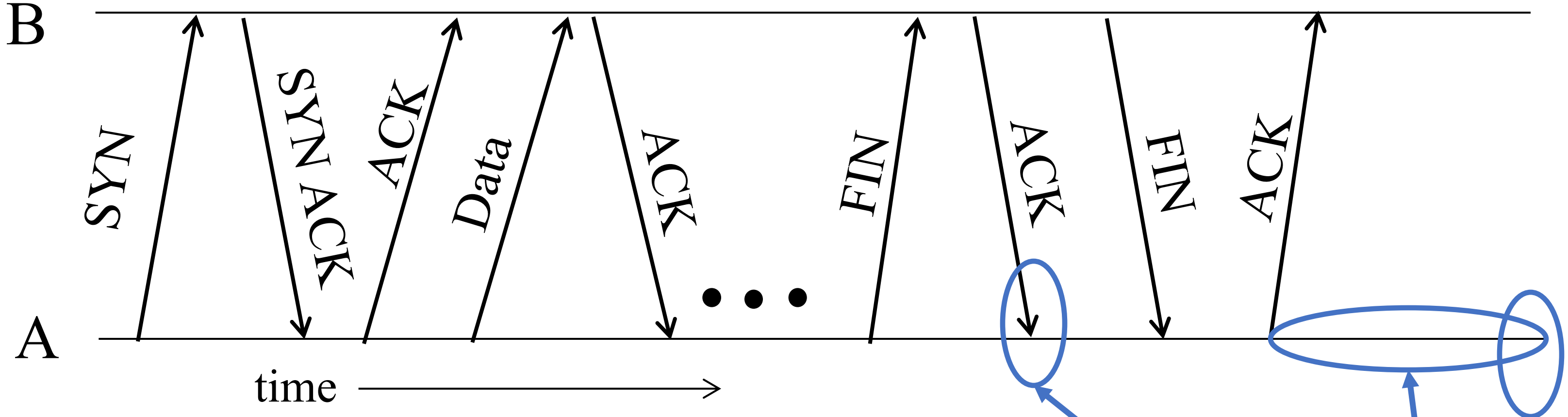
- **Suppose the SYN packet gets lost**
  - ➢Packet dropped by the network or server is busy
- **Eventually, no SYN-ACK arrives**
  - ➢Sender retransmits the SYN on timeout
- **How should the TCP sender set the timer?**
  - ➢Sender has no idea how far away the receiver is
  - ➢Hard to guess a reasonable 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

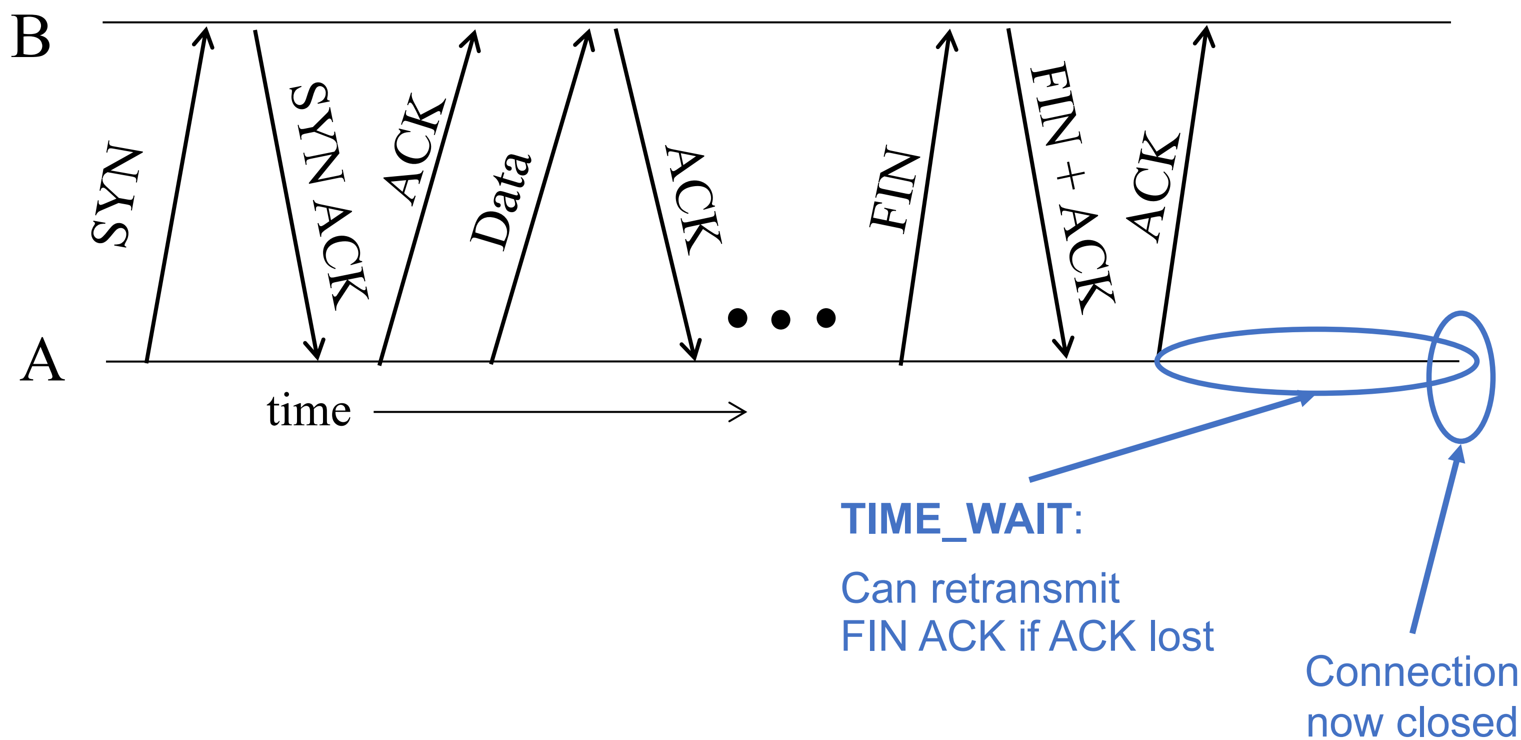
TCP connection teardown

Normal termination, one side at a time



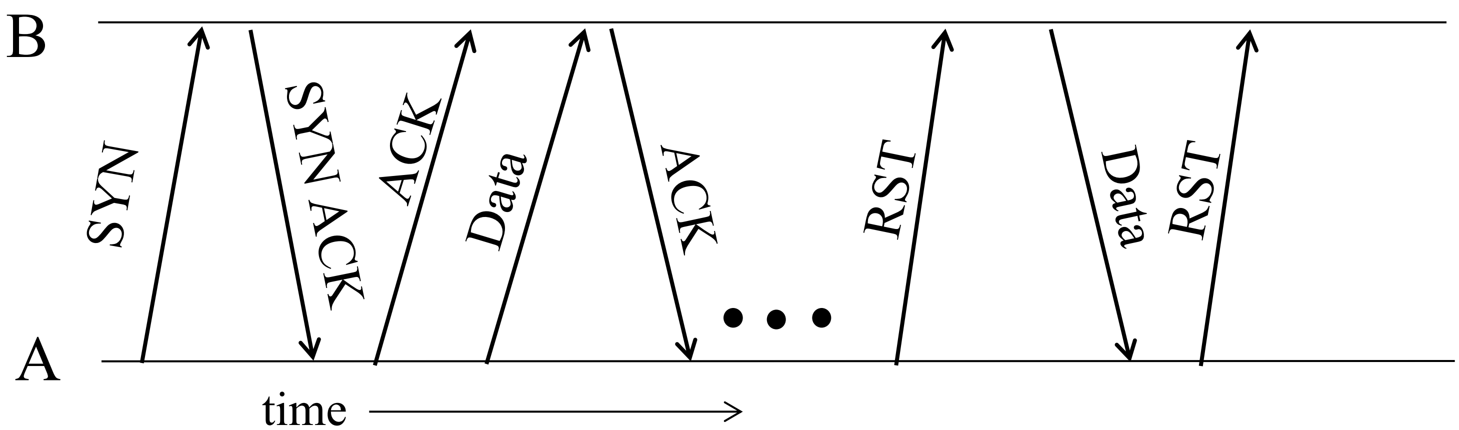
- **Finish (FIN) to close and receive remaining bytes**
  - FIN occupies one byte in the sequence space
- **Other host acks the byte to confirm**
- **Closes A's side of the connection, but not B's**
  - Until B likewise sends a FIN
  - Which A then acks

Normal termination, both together



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

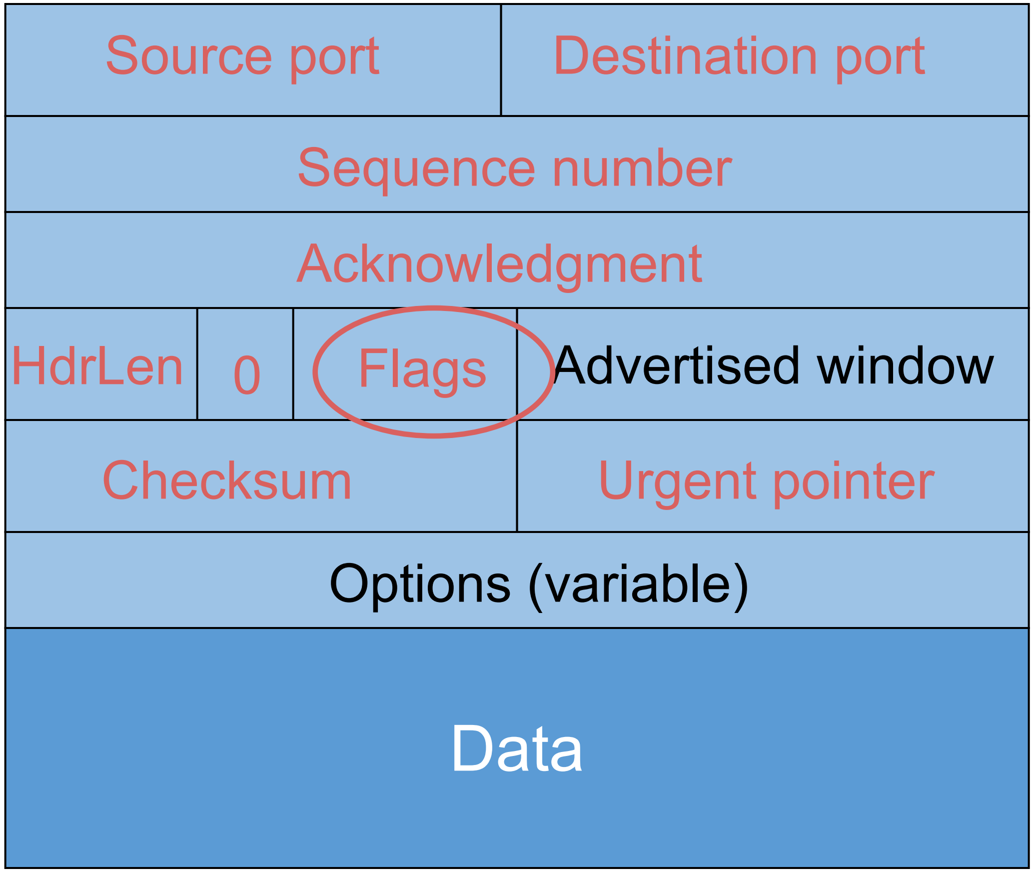
Abrupt termination



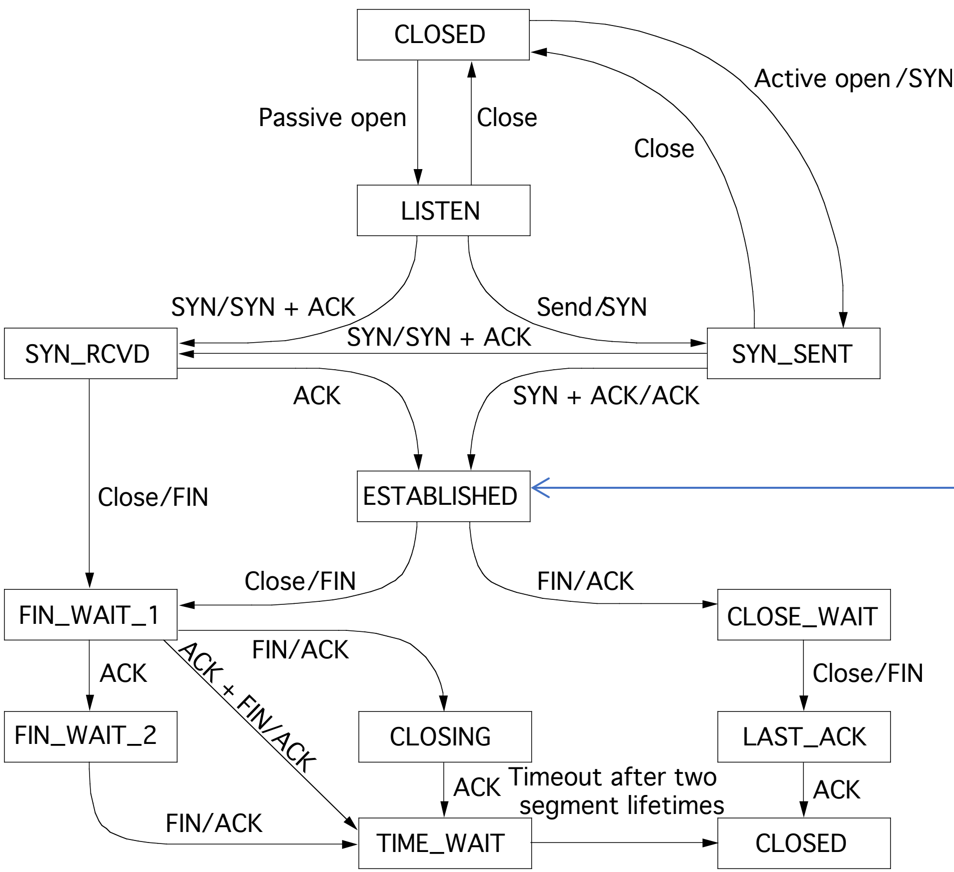
- **A sends a RESET (RST) to B**
  - E.g., because application process on A crashed
- **That's it**
  - B does not ack the RST
  - Thus, RST is not delivered reliably, and any data in flight is lost
  - But: if B sends anything more, will elicit another RST

TCP header

Flags:
SYN
ACK
FIN
RST
PSH
URG



TCP state transitions



Data, ACK exchanges are in here

Group Discussion

- **Topic: TCP header design**
  - Examine each field in TCP header. For each field, is it necessary to have it in the TCP header? What will happen if the TCP header does not have it?

- **Discuss in groups, and each group chooses a leader to summarize the discussion**
  - In your group discussion, please do not dominate the discussion, and give everyone a chance to speak
  - Turn on your video if you can

Summary

- **Reliability is not easy!**

- **Next class**
  - Flow control
  - LOTs of congestion control

Thanks!

Q&A