EECS 489
Computer Networks

Winter 2025

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Material with thanks to Aditya Akella, Sugih Jamin, Philip Levis, Sylvia Ratnasamy, Peter Steenkiste, and many other colleagues.

Agenda

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

Recap: Designing a reliable transport protocol

- Stop and Wait vs Sliding Window

- Sliding Window
  - Acknowledgements: Cumulative vs Selective
  - Resending packets: Go-Back-N vs Selective Repeat

TCP: TRANSMISSION CONTROL PROTOCOL

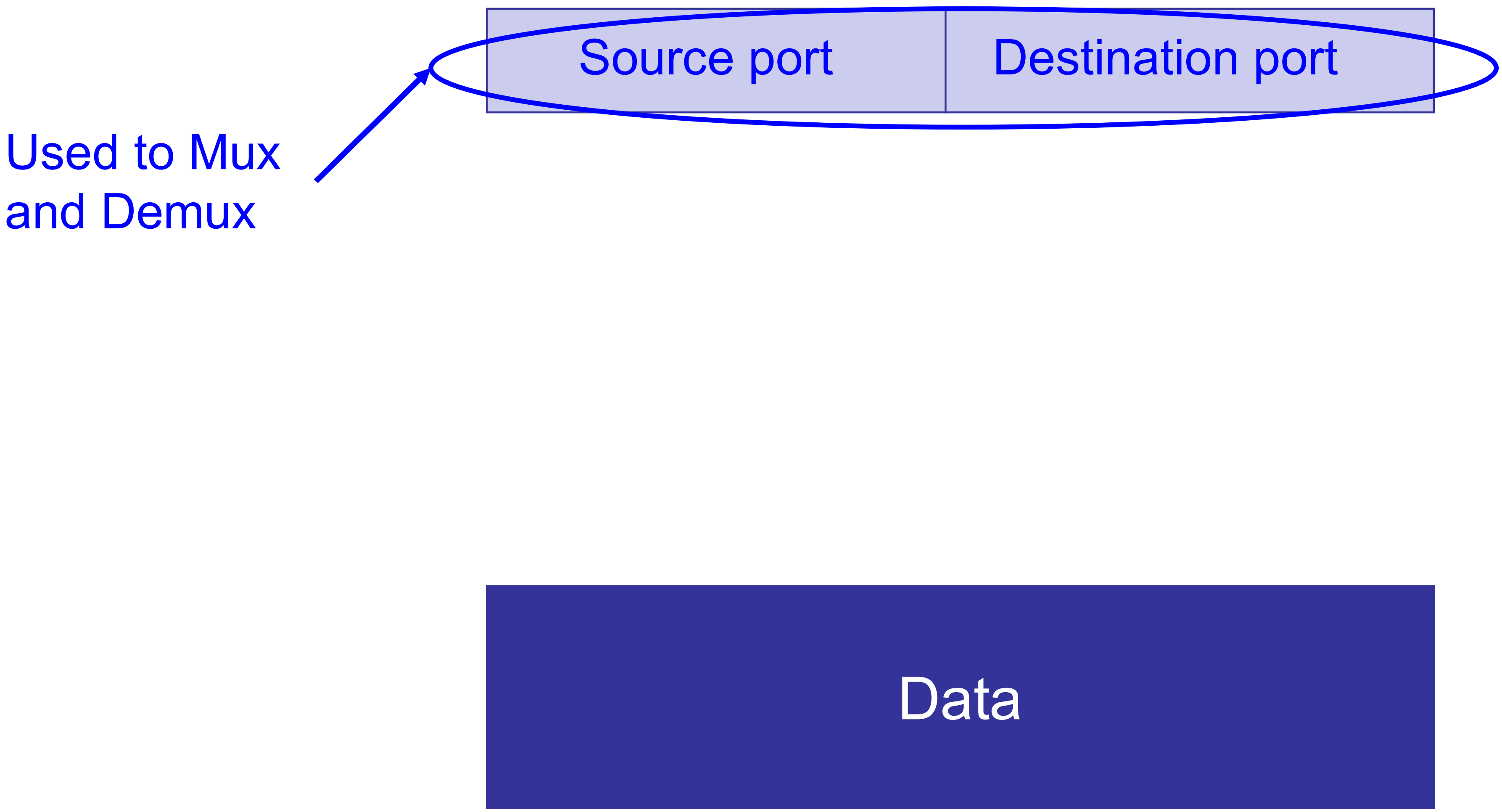
The TCP Abstraction

- TCP delivers a reliable, in-order, byte stream
- Reliable: TCP resends lost packets (recursively)
  - 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?

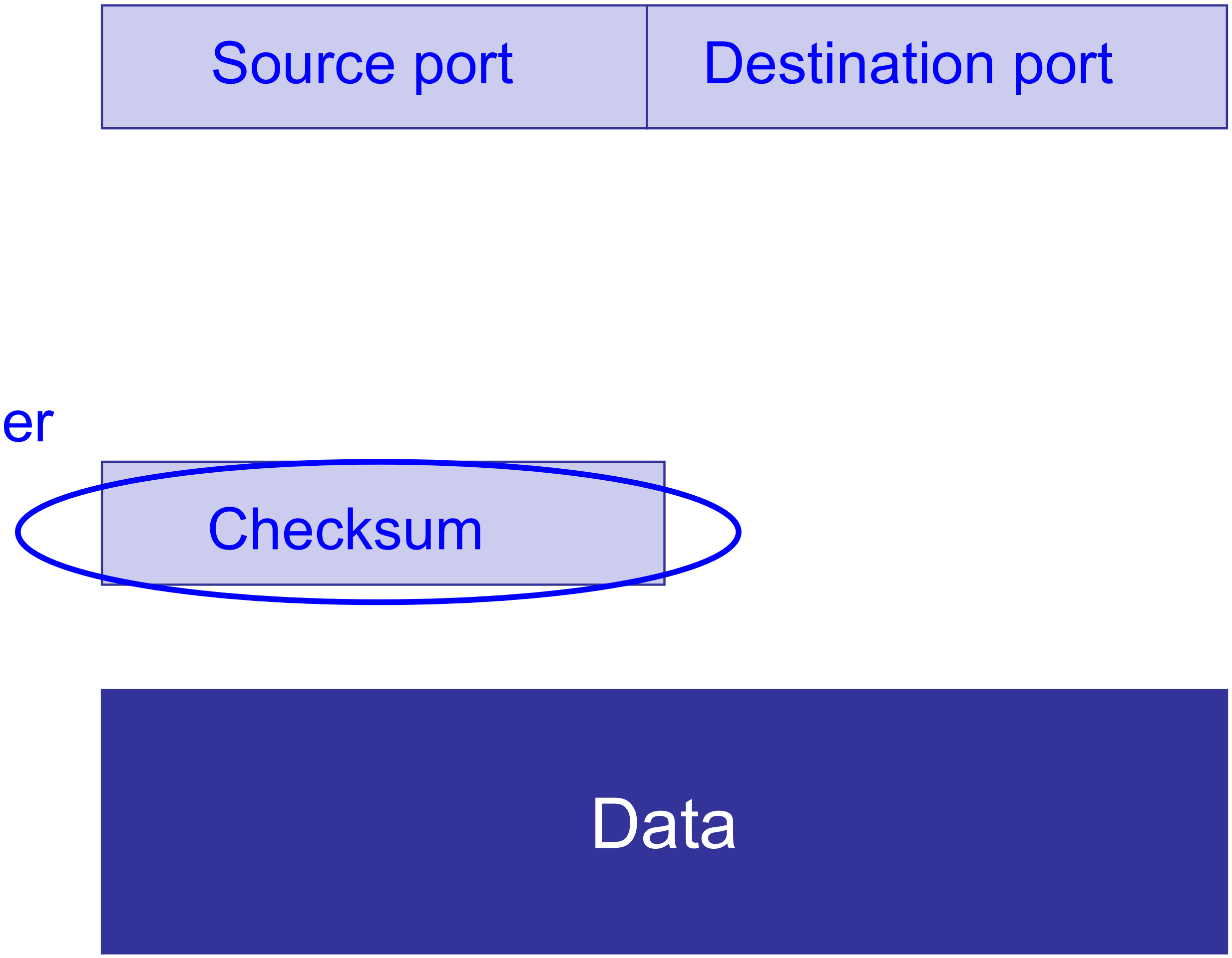
- Most of what we've seen
  - Checksums
  - Sequence numbers are byte offsets
  - Sender and receiver maintain a sliding window
  - Receiver sends cumulative acknowledgements (like GBN)
    - Sender maintains a single retransmission timer
  - Receivers buffer out-of-sequence packets (like SR)
- Few more: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

Build the TCP header



Build the TCP header

Computed
over pseudo-header
and data

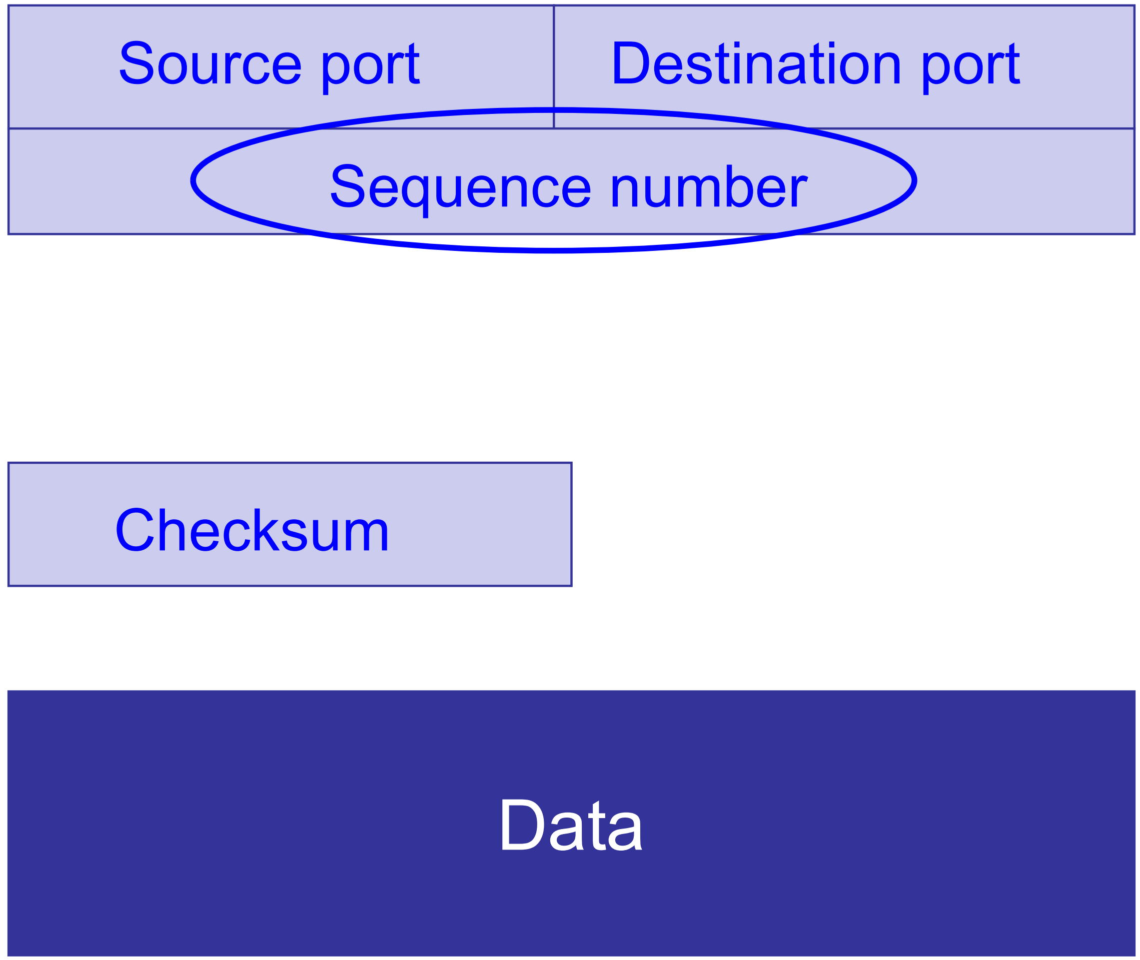


What does TCP do?

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

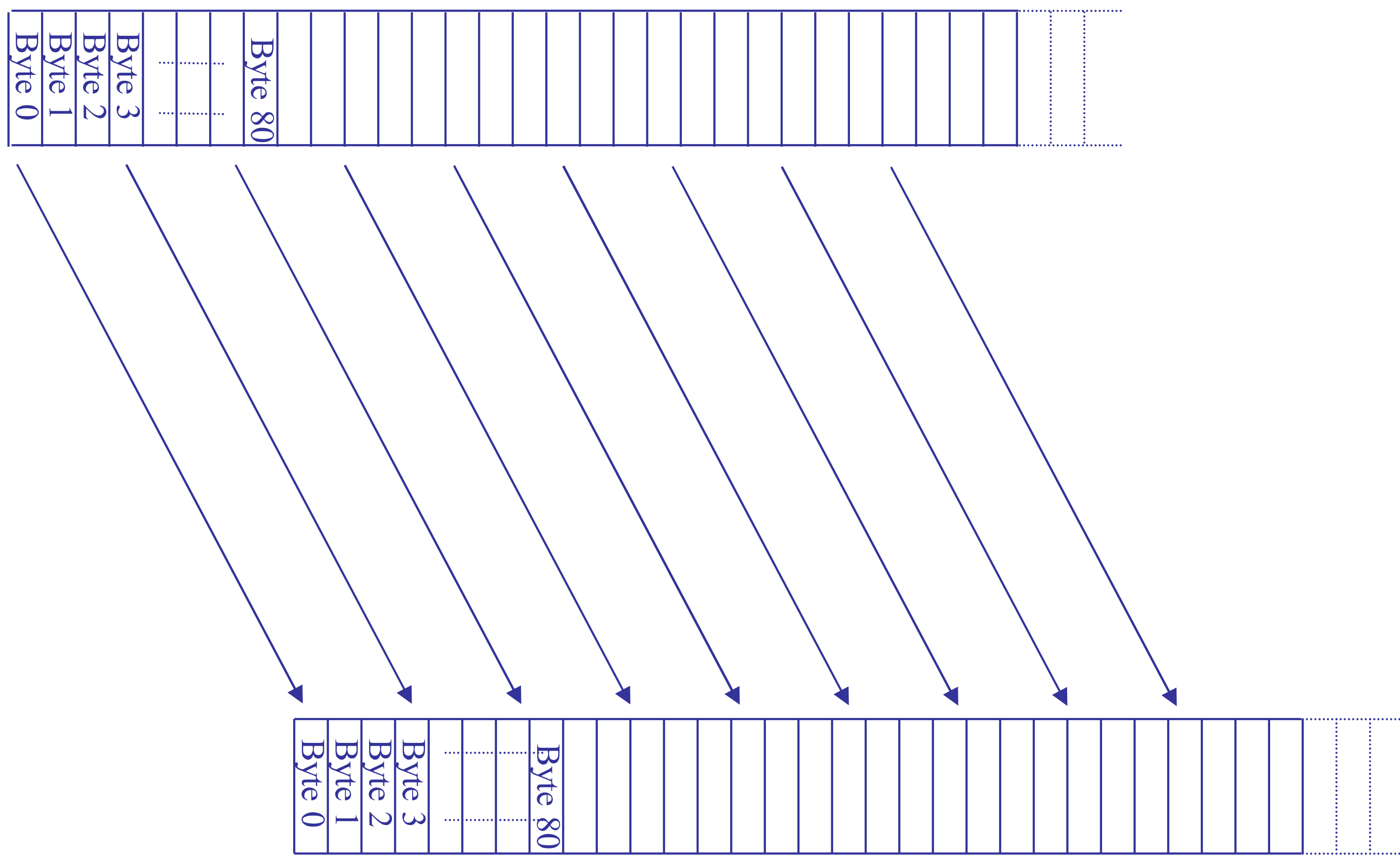
Build the TCP header

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



TCP "stream of bytes" service...

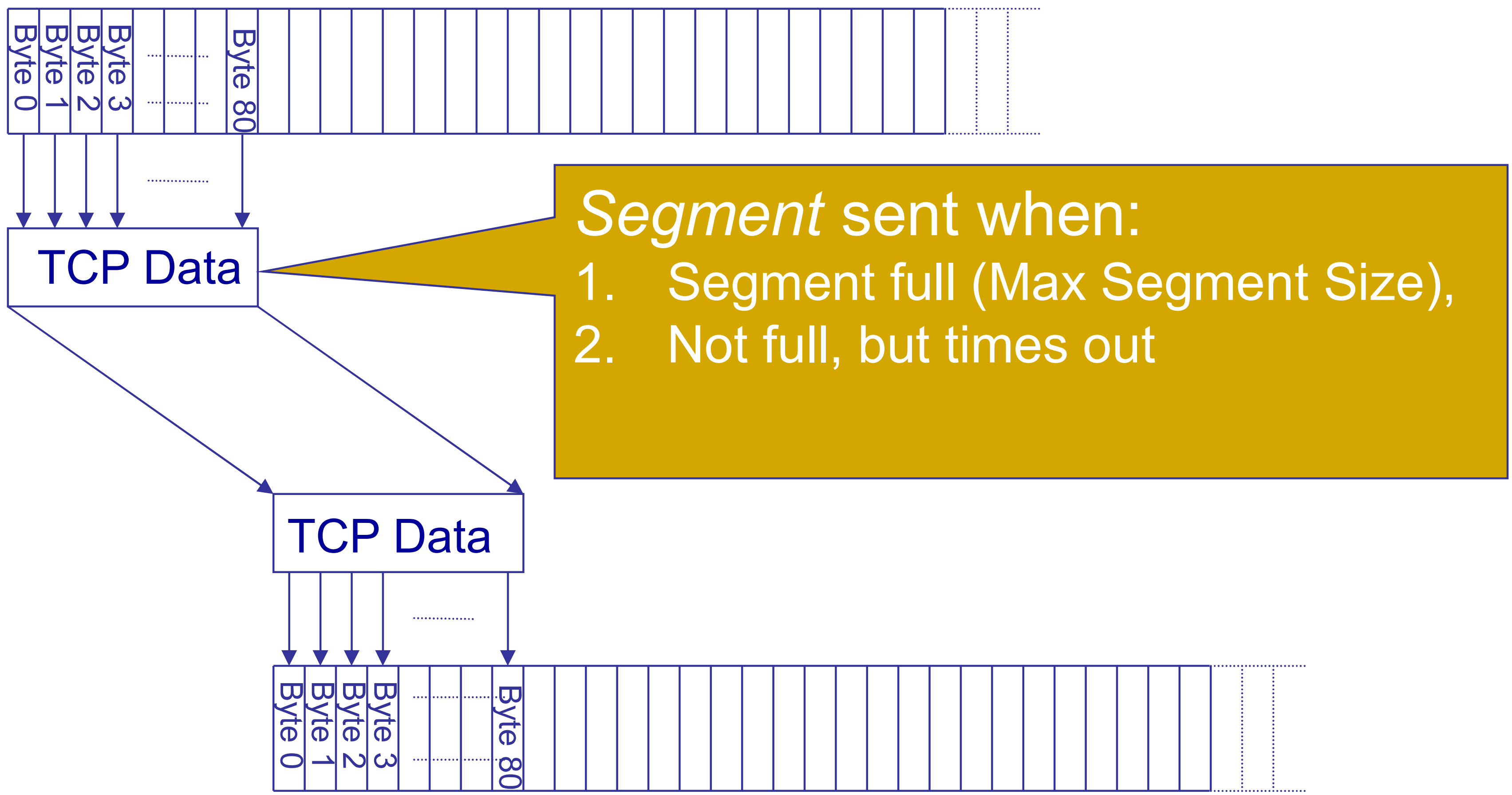
Application @ Host A



Application @ Host B

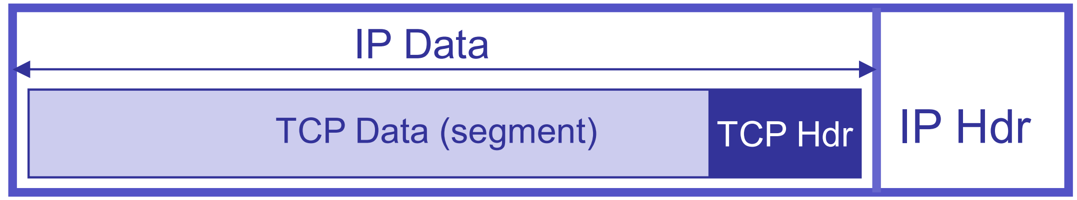
… provided using TCP "segments"

Host A



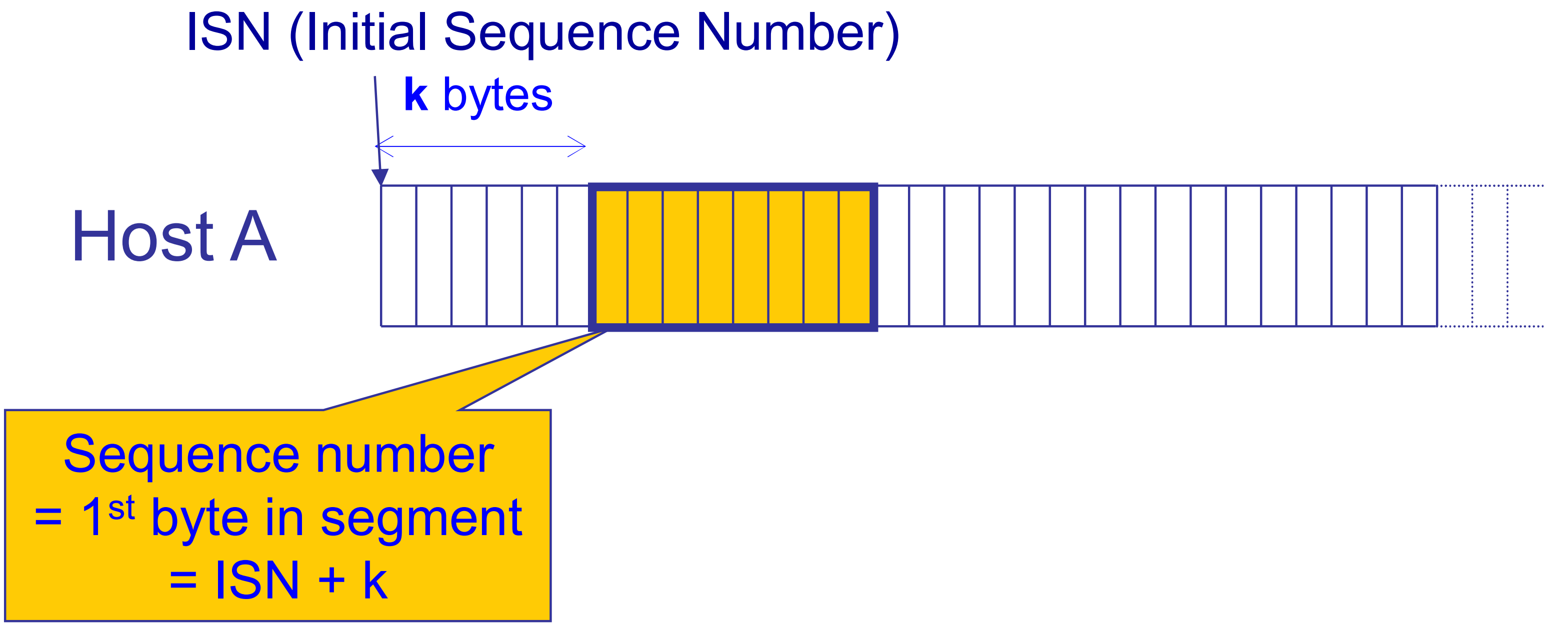
Host B

TCP segment

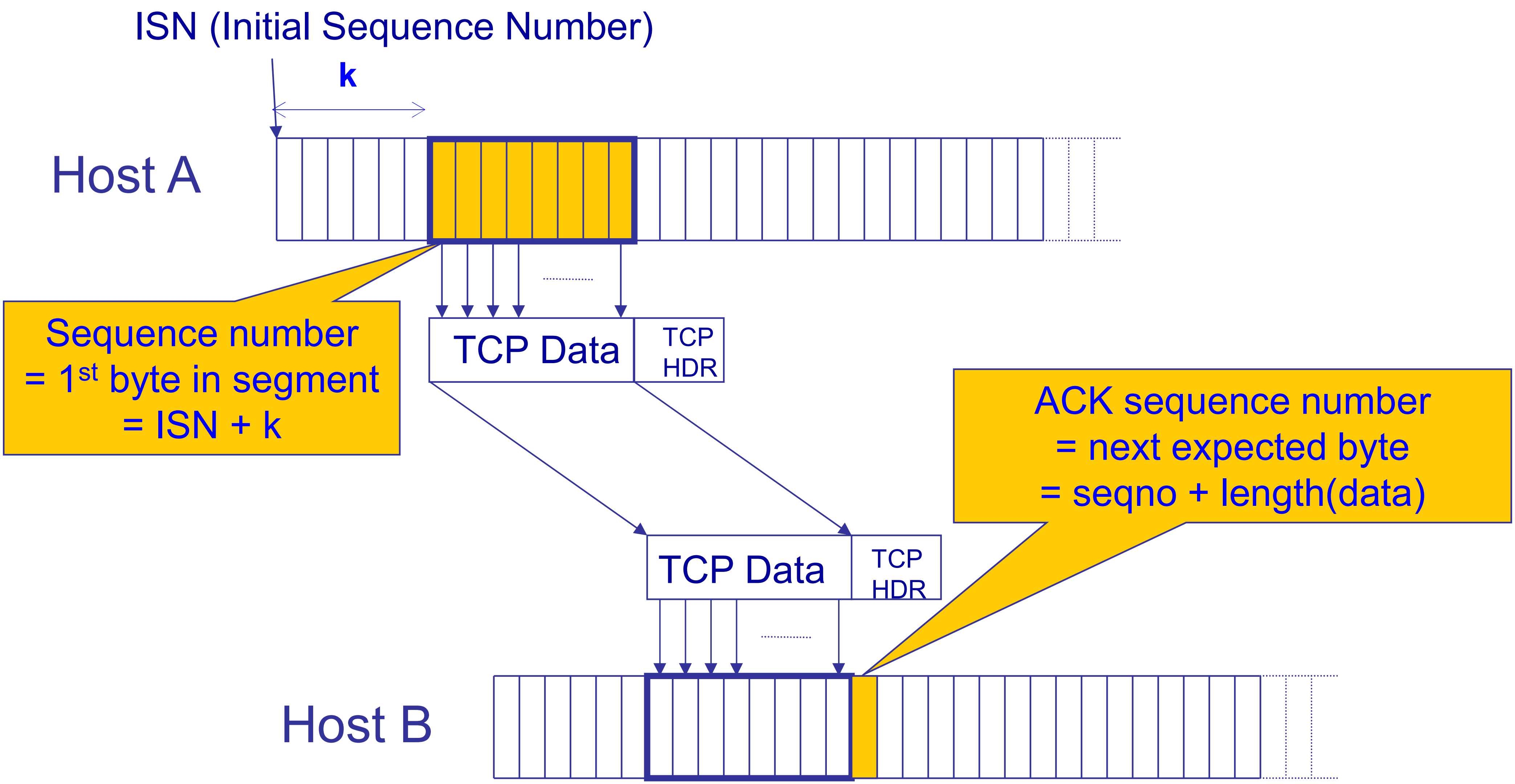


- IP packet
  - No bigger than Maximum Transmission Unit (MTU)
  - E.g., up to 1500 bytes with Ethernet
- TCP packet
  - IP packet with a TCP header and data inside
  - TCP header ≥ 20 bytes long
- TCP segment
  - No more than Maximum Segment Size (MSS) bytes
  - E.g., up to 1460 consecutive bytes from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

Sequence numbers

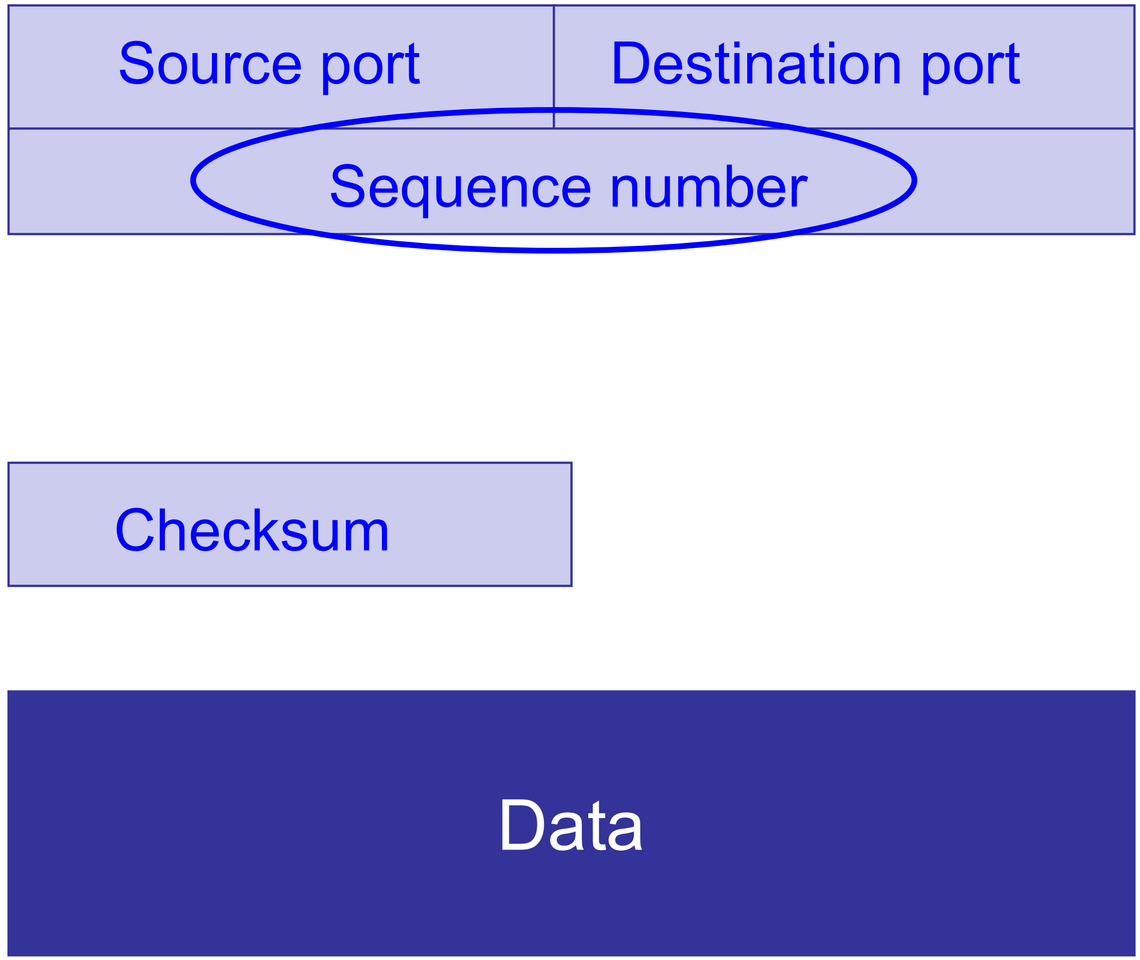


Sequence numbers



Build the 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

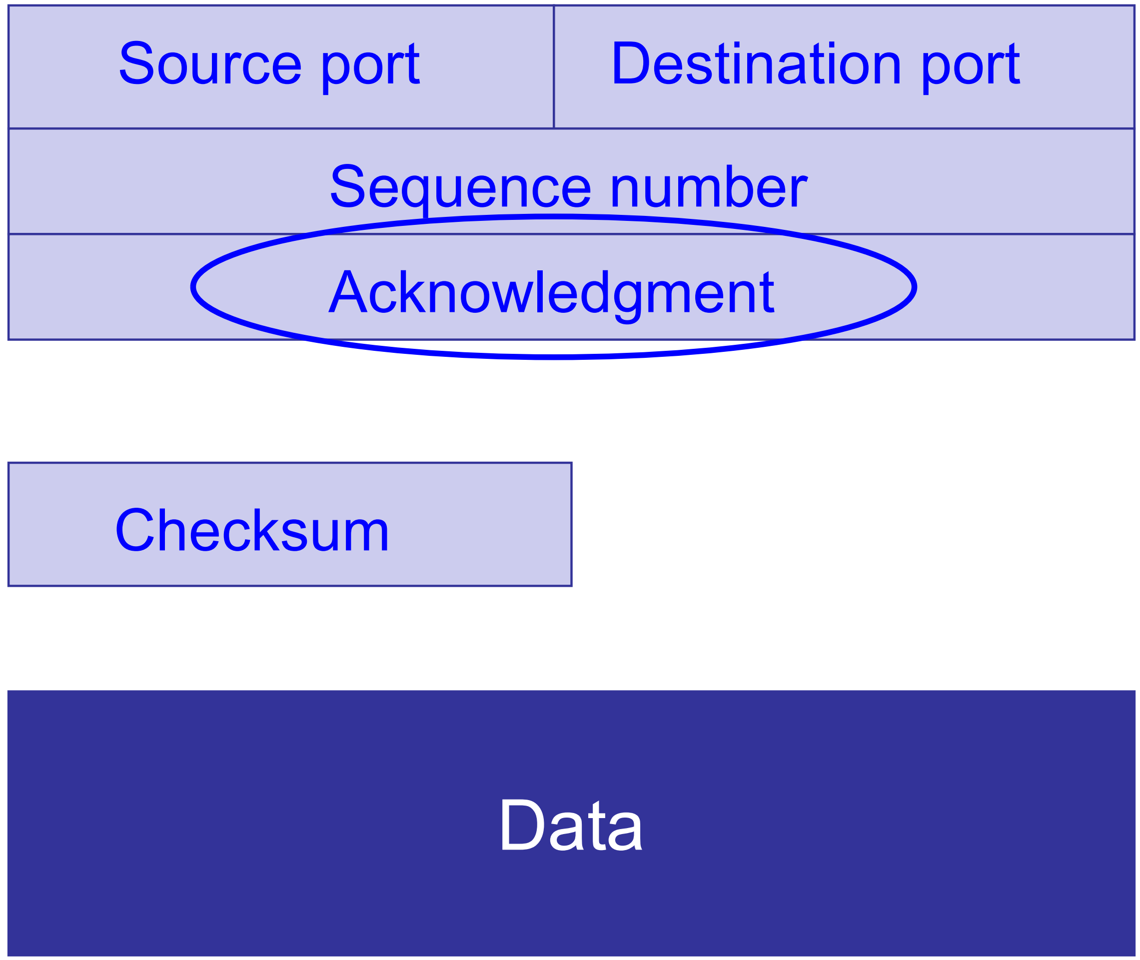
Typical operation

- 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

Build the 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, …
- Assume the fifth packet (seqno 500) is lost, but no others
- Stream of ACKs will be:
  - 200, 300, 400, 500, 500, 500, 500,…

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?
  - Choose correctness

5-MINUTE BREAK!

Announcements

- A2 is out!
  - Due on Feb 28

What does TCP introduce?

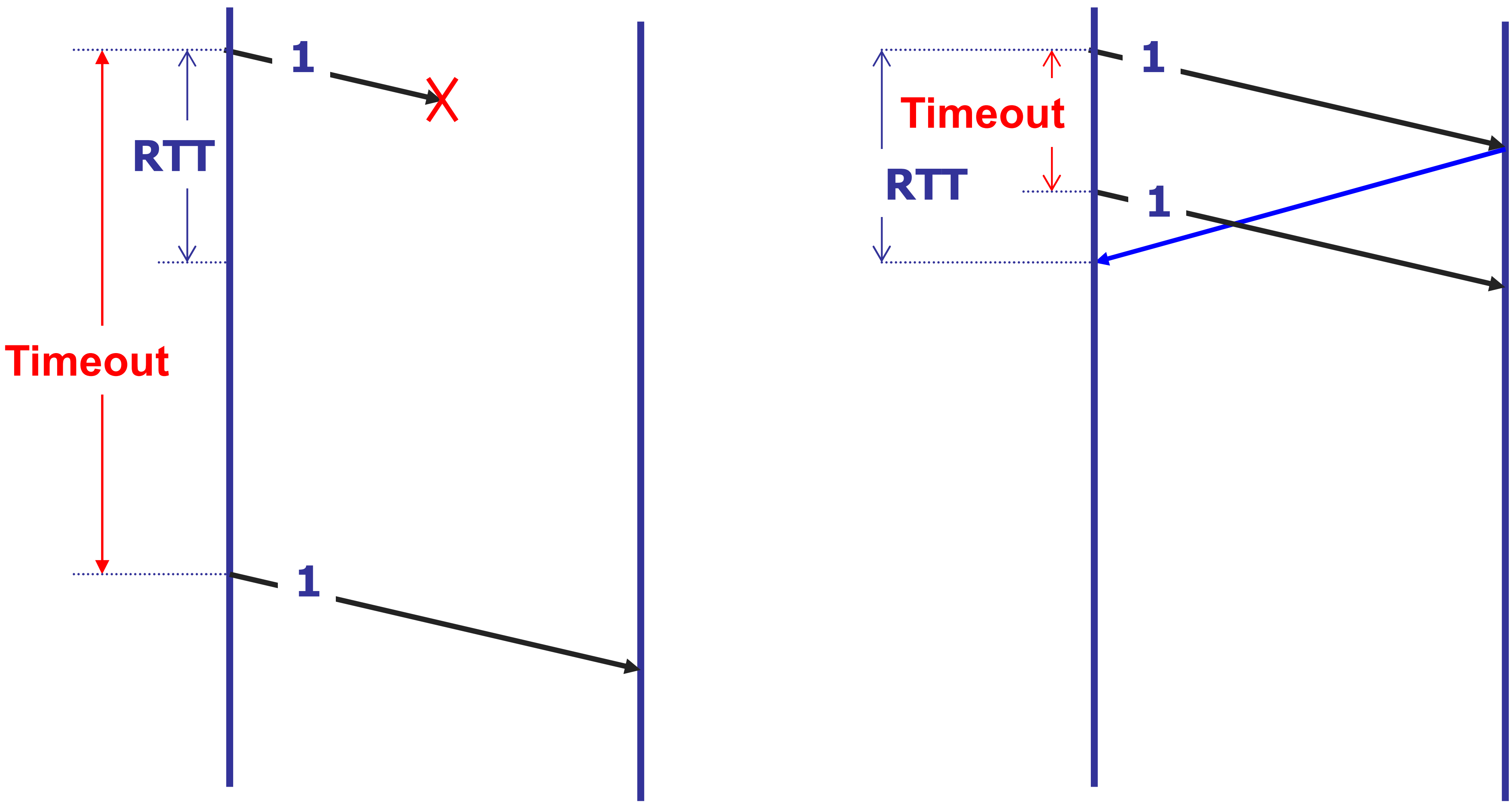
- Most of what we've seen
  - Checksum
  - Sequence numbers are byte offsets
  - Receiver sends cumulative acknowledgements (like GBN)
  - Receivers buffer out-of-sequence packets (like SR)
- Introduces fast retransmit: duplicate ACKs trigger early retransmission
- Sender maintains a single retransmission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

- Challenge: How do we pick a timeout value?

Timing illustration



Timeout too long → inefficient

Timeout too short → duplicate packets

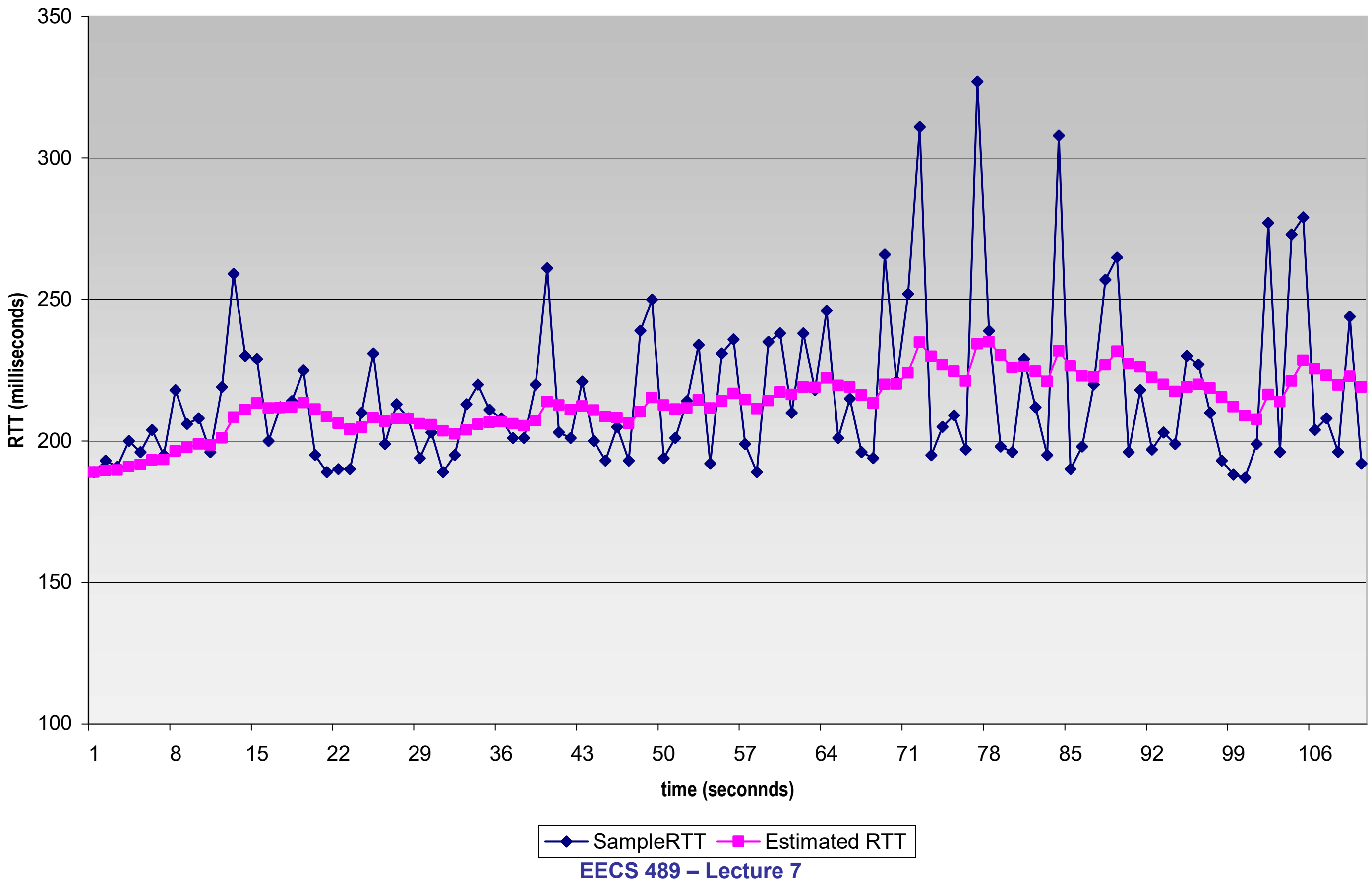
Retransmission timeout

- 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

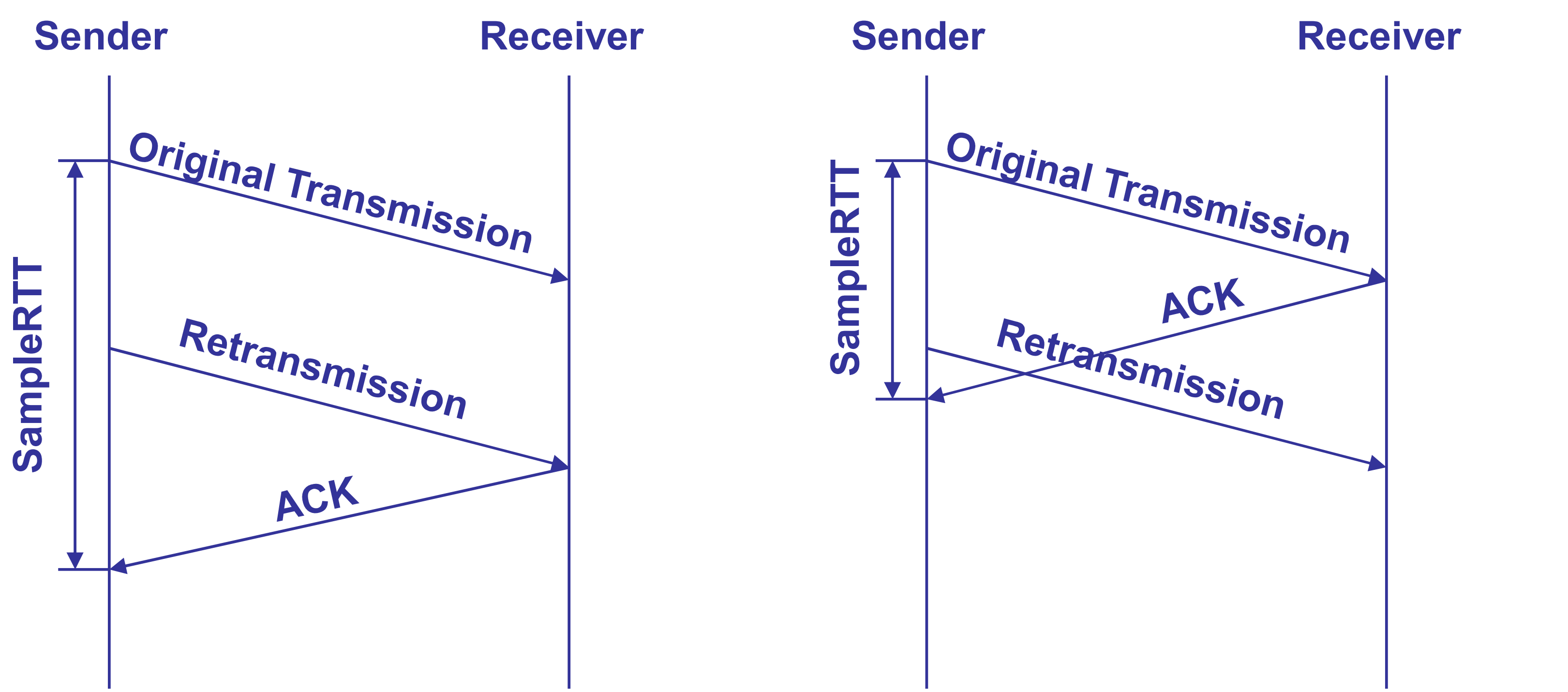
- Exponential weighted average of RTT samples

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



Problem: Ambiguous measurements

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



Solution: Ambiguous measurements

- Don’t use SampleRTT from retransmissions
  - Once retransmitted, ignore that segment in the future

Karn/Partridge algorithm

- Computes EstimatedRTT using $\alpha = 0.125$
- Timeout value (RTO) $= 2 \times$ EstimatedRTT
  - Every time RTO timer expires, set RTO $\leftarrow 2 \cdot$RTO
    - Employs exponential backoff (Up to $\geq$ 60 sec)
  - Every time new measurement comes in (i.e., successful original transmission), reset RTO back to $2 \times$ EstimatedRTT

- Sensitive to RTT variations

Jacobson/Karels algorithm

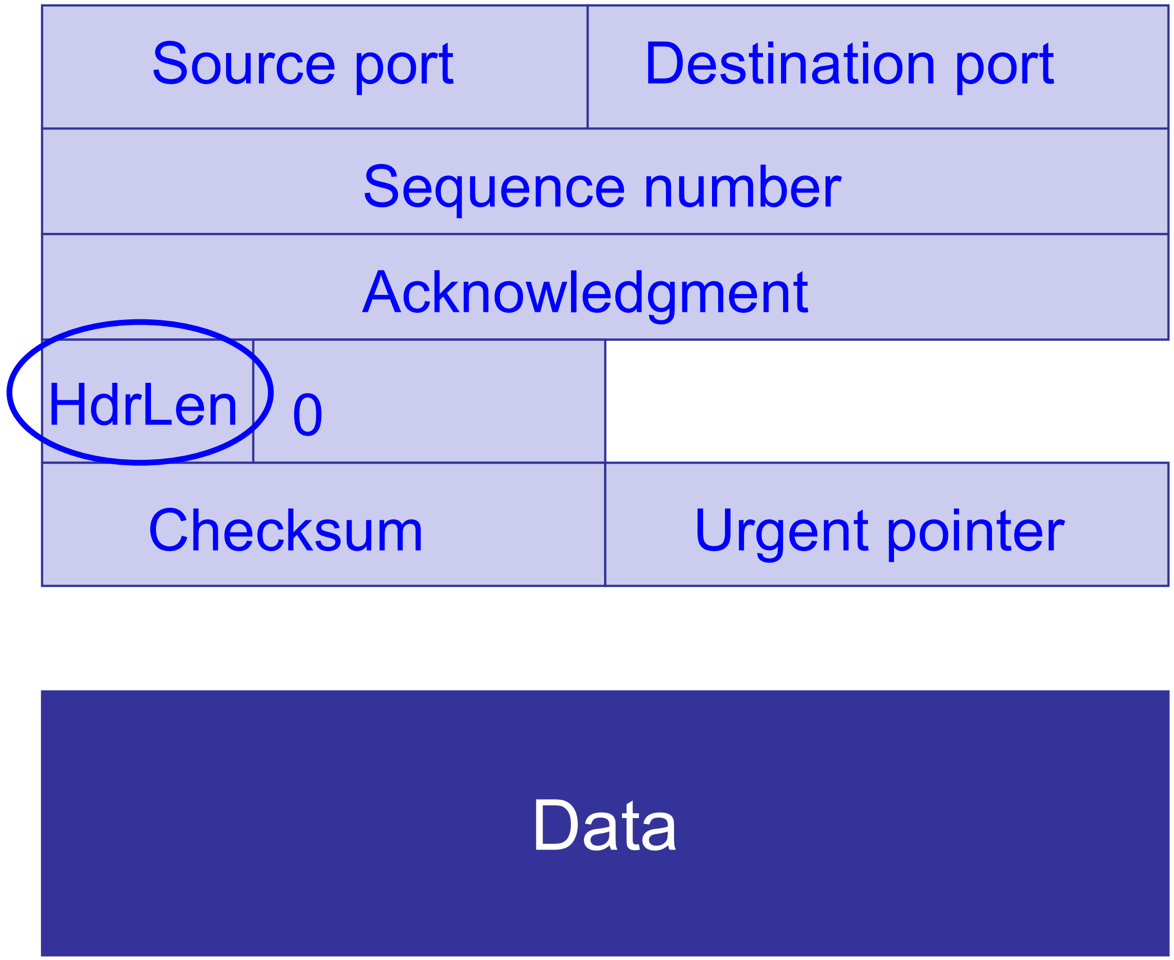
- Problem: need to better capture variability in RTT
  - Solution: Directly measure deviation

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

- RTO = EstimatedRTT + 4 x DevRTT

Build the TCP header

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



TCP CONNECTION ESTABLISHMENT

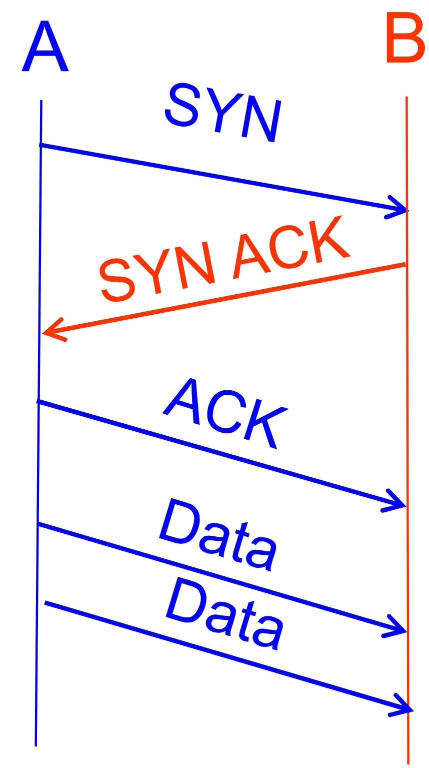
Initial Sequence Number (ISN)

- Why not just use ISN = 0?
  - IP addresses and port #s uniquely identify a connection; however, these port #s get used again
    - Small chance an old packet is still in flight
  - Predictable; makes spoofing connection easier

- Hosts exchange ISNs when establishing connection

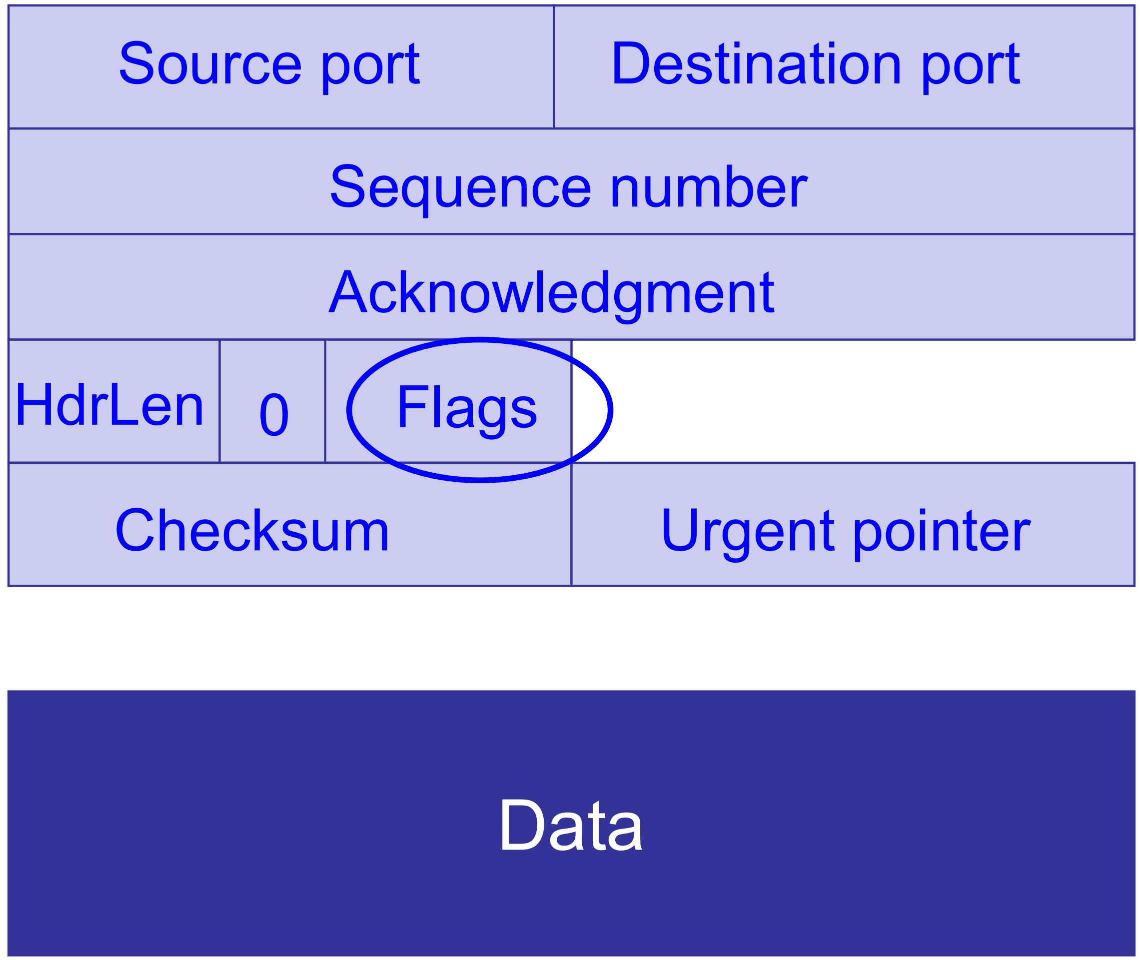
Establishing a TCP connection

- 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

<table>
<tr><td colspan="3">A's port</td><td>B's port</td></tr>
<tr><td colspan="4">A's Initial Sequence Number</td></tr>
<tr><td colspan="4">N/A</td></tr>
<tr><td>5</td><td>0</td><td>SYN</td><td></td></tr>
<tr><td colspan="3">Checksum</td><td>Urgent pointer</td></tr>
</table>

Step 1: B’s SYN-ACK packet

B tells it accepts and is ready to accept next packet

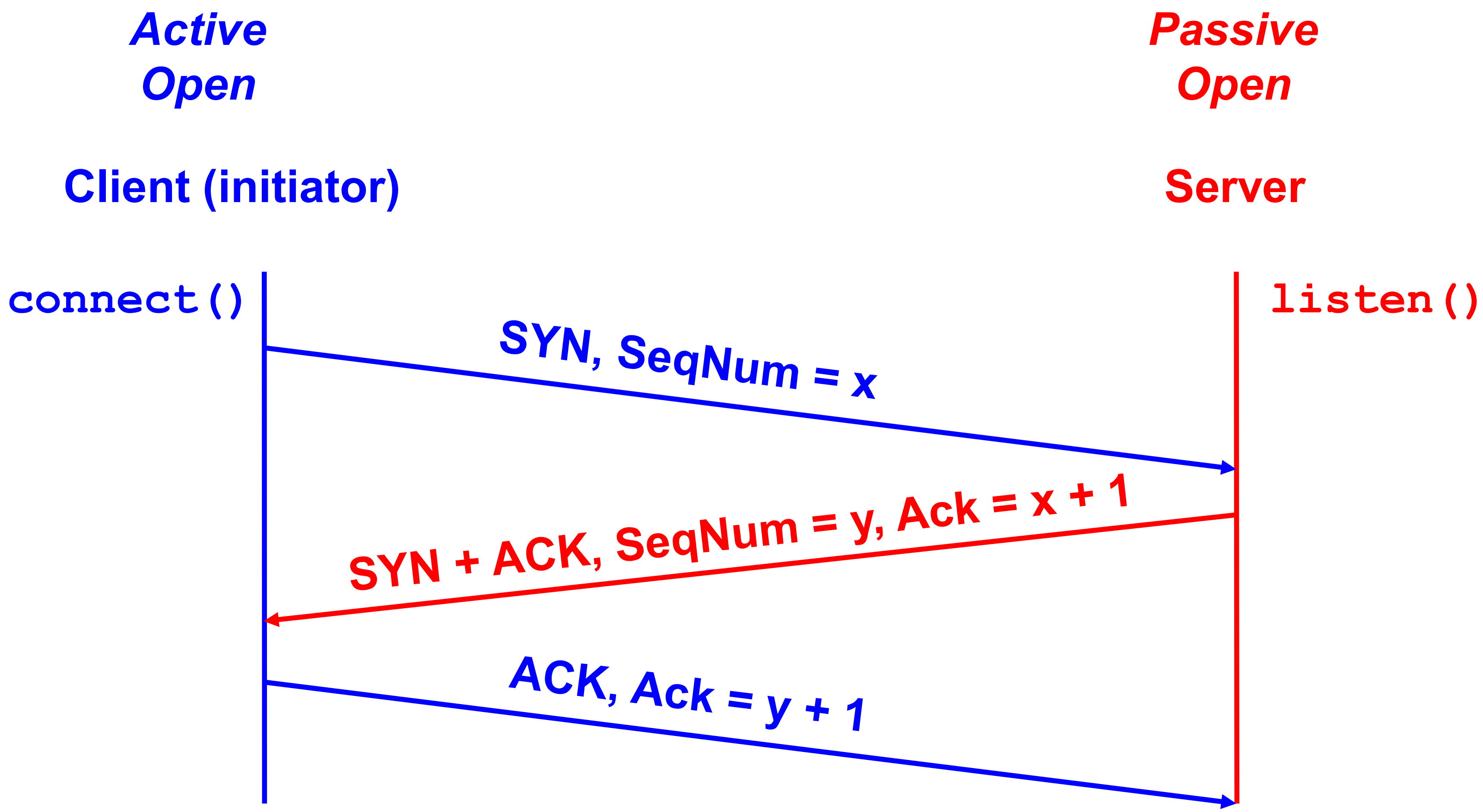
<table>
<tr><td colspan="3">B’s port</td><td colspan="2">A’s port</td></tr>
<tr><td colspan="5">B’s Initial Sequence Number</td></tr>
<tr><td colspan="5">ACK=A’s ISN+1</td></tr>
<tr><td>5</td><td>0</td><td colspan="2">SYN|ACK</td><td></td></tr>
<tr><td colspan="4">Checksum</td><td>Urgent pointer</td></tr>
</table>

Step 1: A's ACK to SYN-ACK

A tells B to open a connection

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

TCP's 3-Way handshaking



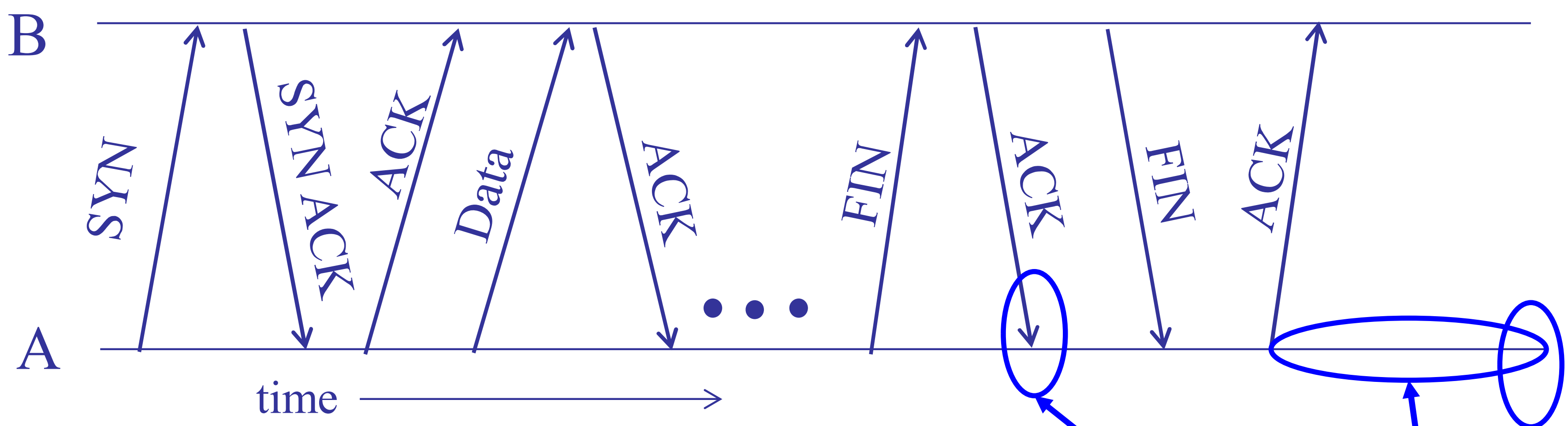
What if the SYN Packet Gets Lost?

- 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
  - Default is 3 seconds

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

Connection now **closed**

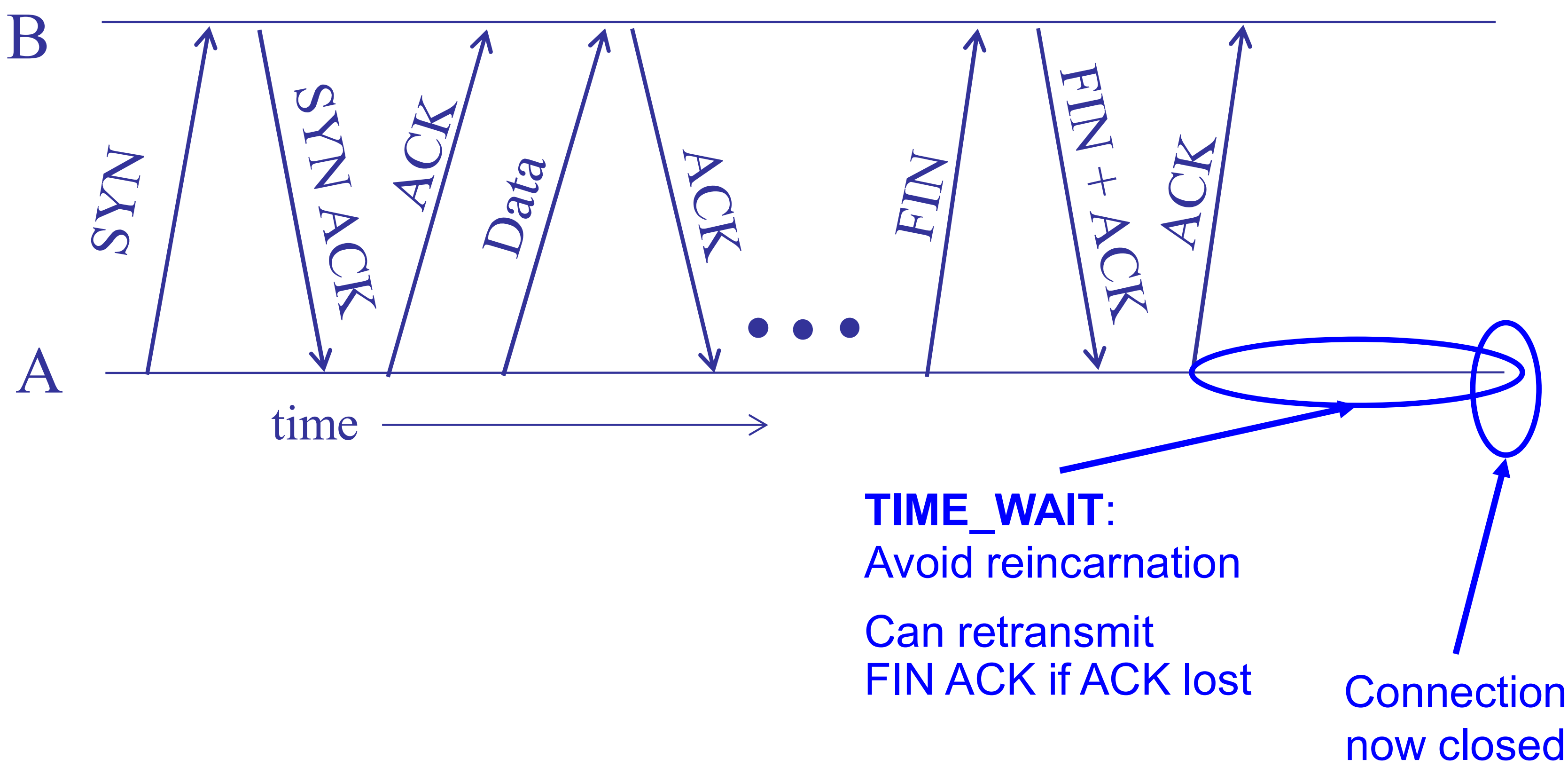
Connection now **half-closed**

TIME_WAIT:

Avoid reincarnation

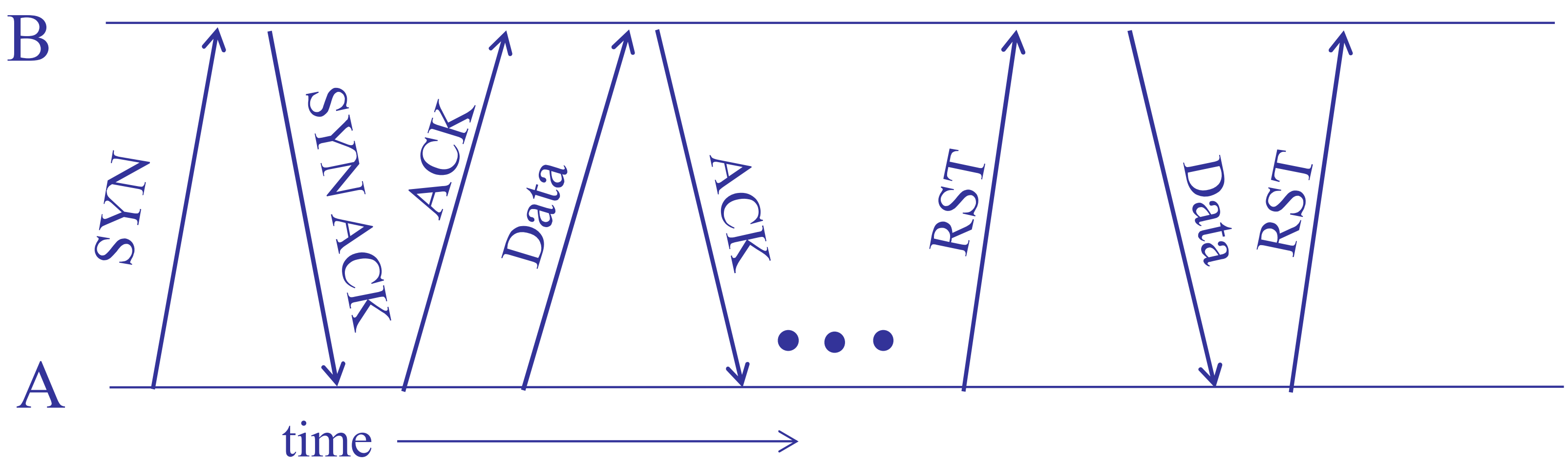
B will retransmit FIN if ACK is lost

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

Summary

- Reliability is not easy!

- Next
  - Flow control
  - LOTs of congestion control