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Computer Networks

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Material with thanks to James F. Kurose, Mosharaf Chowdhury, and other colleagues.

Outline

- UDP: User Datagram Protocol
- TCP: Transmission Control Protocol
- TCP Connection Setup

User Datagram Protocol (UDP)

UDP: User Datagram Protocol

- Lightweight communication between processes
  - Avoid overhead and delays of order & reliability
- UDP described in RFC 768 – (1980!)
  - Destination IP address and port to support demultiplexing

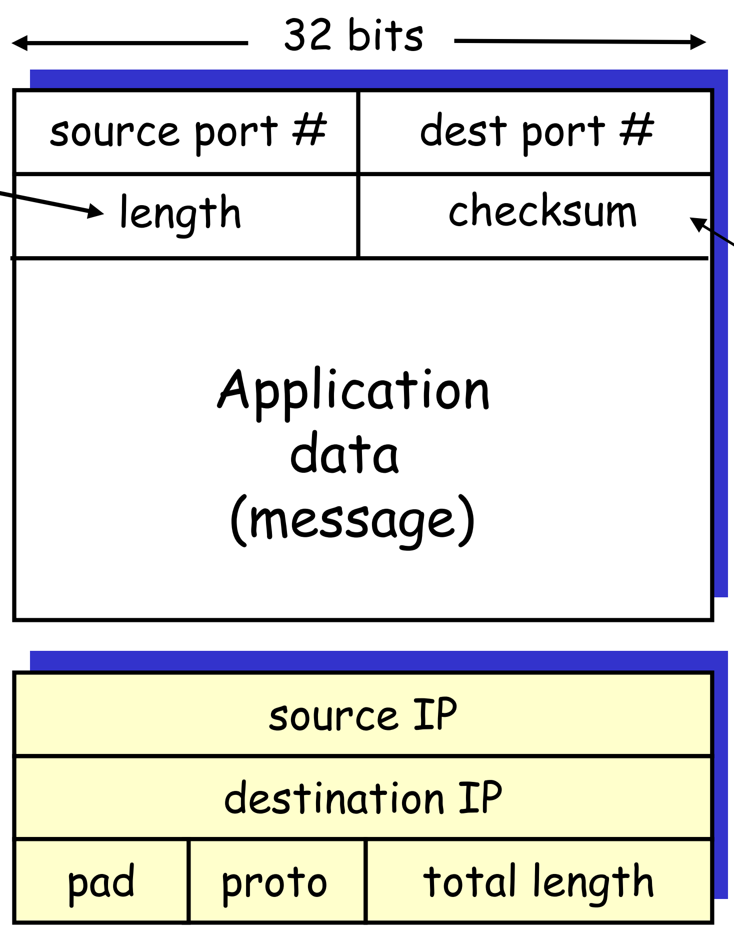
UDP (cont'd)

- "Best effort" service, UDP segments may be:
  - lost
  - delivered out-of-order to app
- Connectionless:
  - no handshaking between UDP sender, receiver
  - each UDP segment handled independently of others
- UDP use:
  - streaming multimedia apps (loss tolerant, rate sensitive)
  - DNS
  - SNMP

Why is there a UDP?

- no connection establishment (which can add delay)
- simple: no connection state at sender, receiver
- small header size
- no congestion control: UDP can blast away as fast as desired

UDP Segment Format



Length in octets, including Header and Data

Header + Data + **Pseudo-header;** **Or set to 0 if no check**

Pseudo-Header for Checksum

The same as UDP length

UDP checksum

Goal: **detect "errors" (e.g., flipped bits) in transmitted segment**

sender:

- treat segment contents, including header fields, as sequence of 16-bit integers
- checksum: addition (one's complement sum) of segment contents
- sender puts checksum value into UDP checksum field

receiver:

- compute checksum of received segment
- check if computed checksum equals checksum field value:
  - NO - error detected
  - YES - no error detected. *But maybe errors nonetheless?* More later ….

Internet checksum: example

example: add two 16-bit integers

		1	1	1	0	0	1	1	0	0	1	1	0	0	1	1	0
		1	1	0	1	0	1	0	1	0	1	0	1	0	1	0	1
wraparound	1	1	0	1	1	1	0	1	1	1	0	1	1	1	0	1	1
sum		1	0	1	1	1	0	1	1	1	0	1	1	1	1	0	0
checksum		0	1	0	0	0	1	0	0	0	1	0	0	0	0	1	1

Note: **when adding numbers, a carryout from the most significant bit needs to be added to the result**

Optional error checking

- Optional error checking on the packet contents
  - (checksum field = 0 means "don't verify checksum")
  - See text on how checksums are calculated
- Source port is also optional
  - Useful to respond back to the sender in some cases

TCP: Transmission Control Protocol

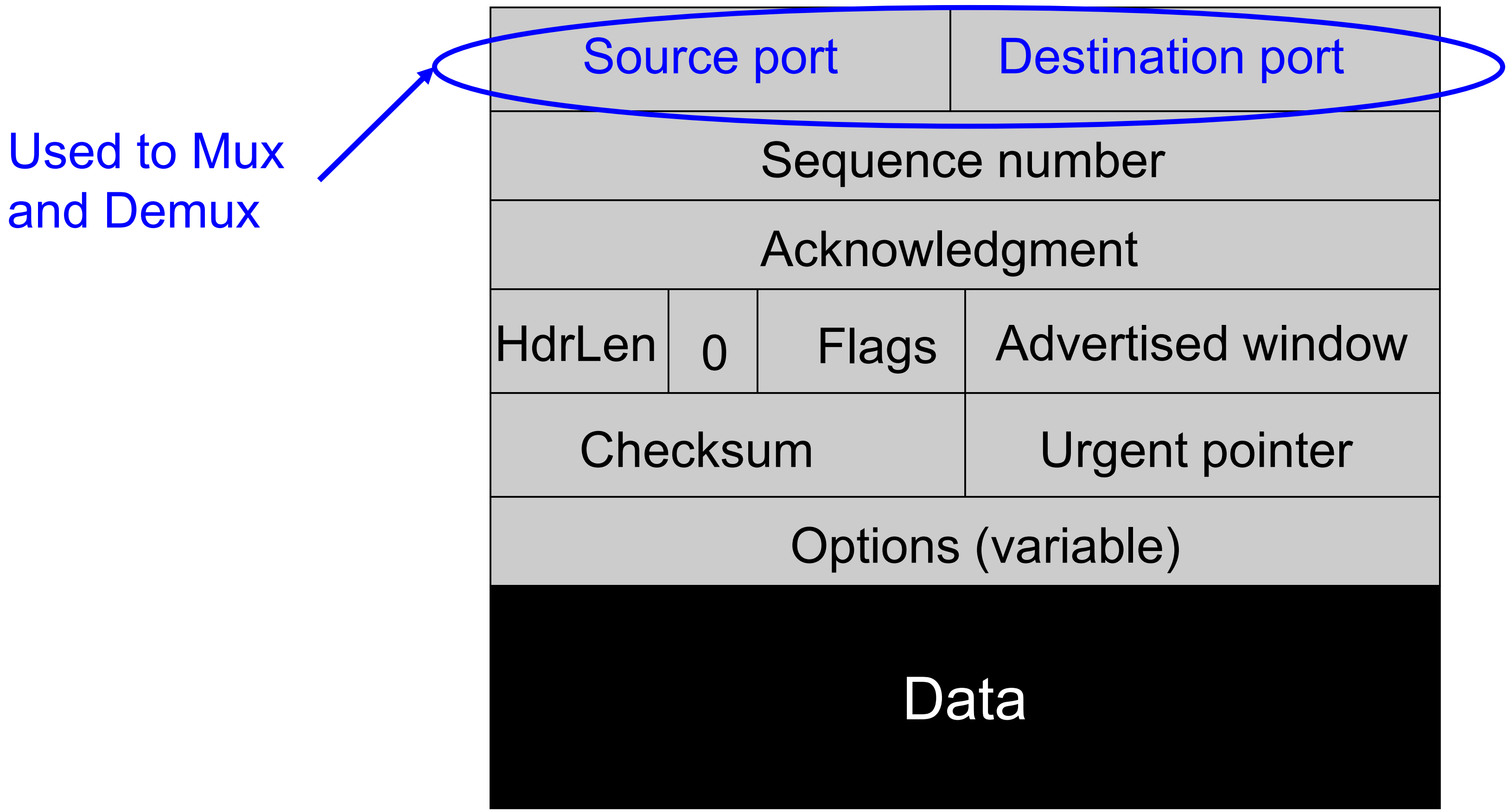
The TCP Abstraction

- 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.

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

Source port			Destination port
Sequence number			
Acknowledgment			
HdrLen	0	Flags	Advertised window
Checksum			Urgent pointer
Options (variable)			
Data			

What does TCP do?

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

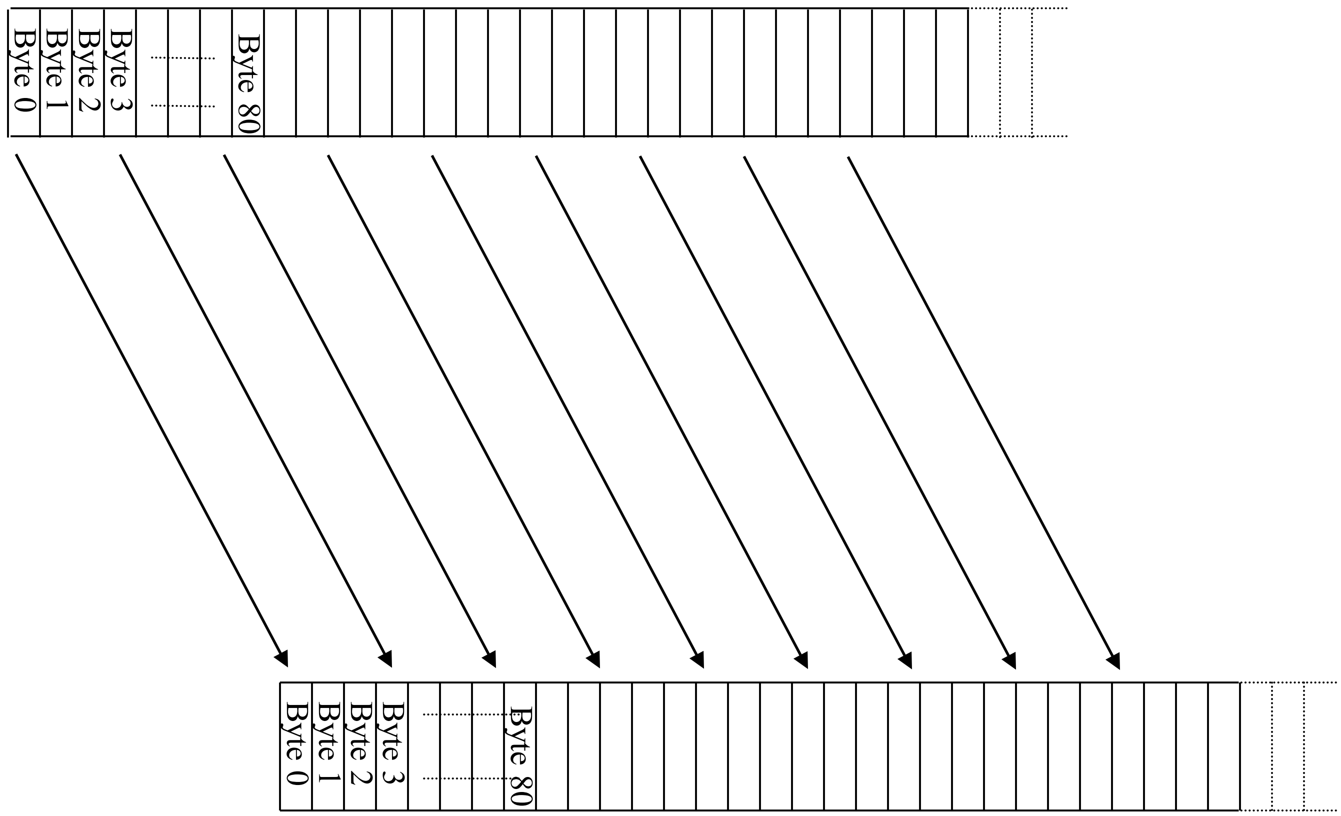
TCP header

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

<table>
<tr><td colspan="3">Source port</td><td>Destination port</td></tr>
<tr><td colspan="4">Sequence number</td></tr>
<tr><td colspan="4">Acknowledgment</td></tr>
<tr><td>HdrLen</td><td>0</td><td>Flags</td><td>Advertised window</td></tr>
<tr><td colspan="3">Checksum</td><td>Urgent pointer</td></tr>
<tr><td colspan="4">Options (variable)</td></tr>
<tr><td colspan="4">Data</td></tr>
</table>

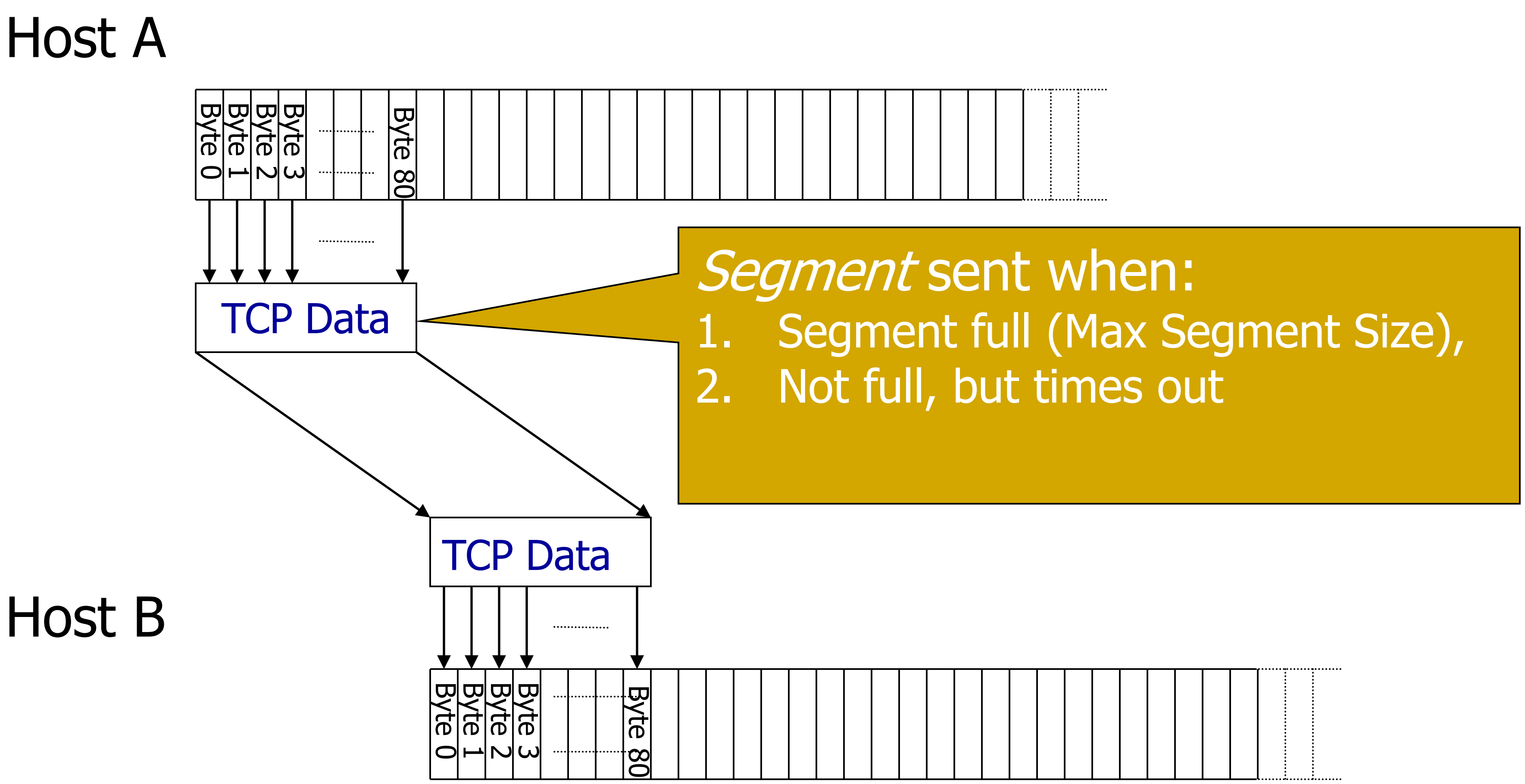
TCP “stream of bytes” service…

Application @ Host A

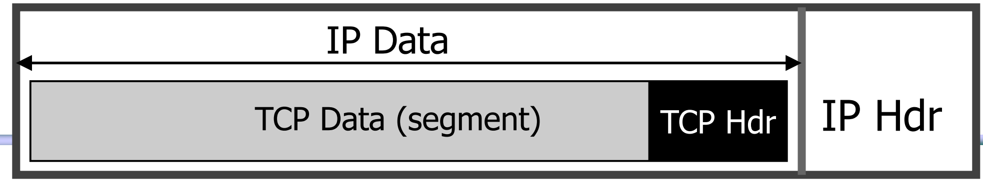


Application @ Host B

… provided using TCP “segments”

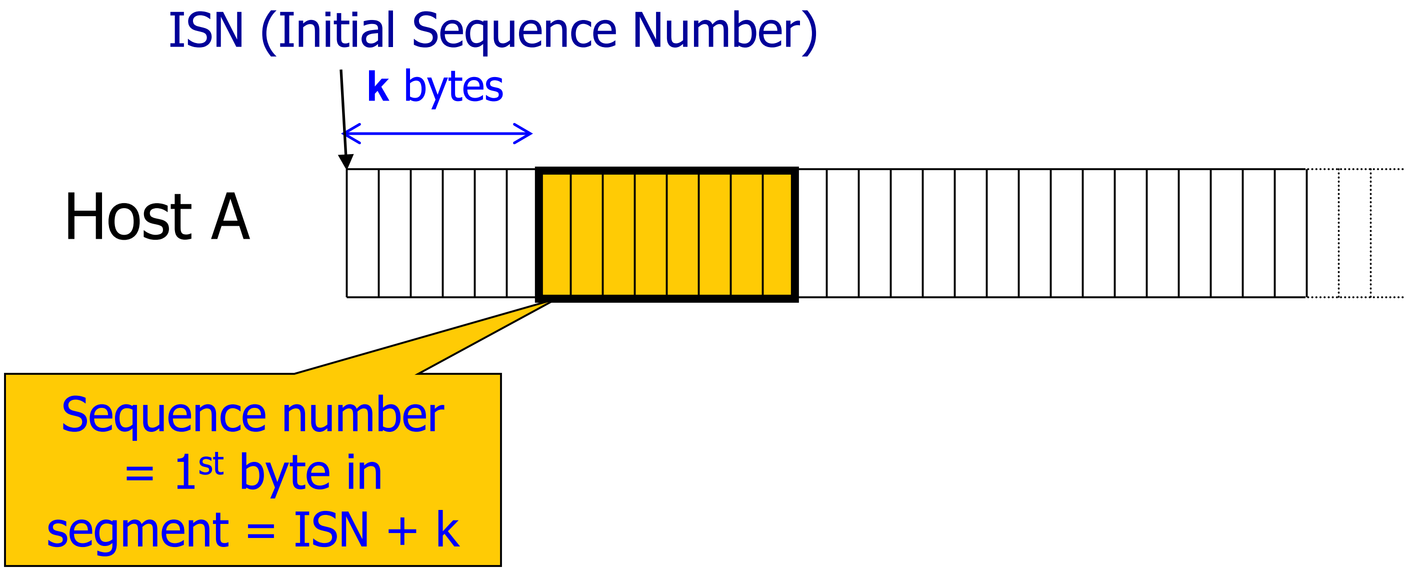


TCP segment

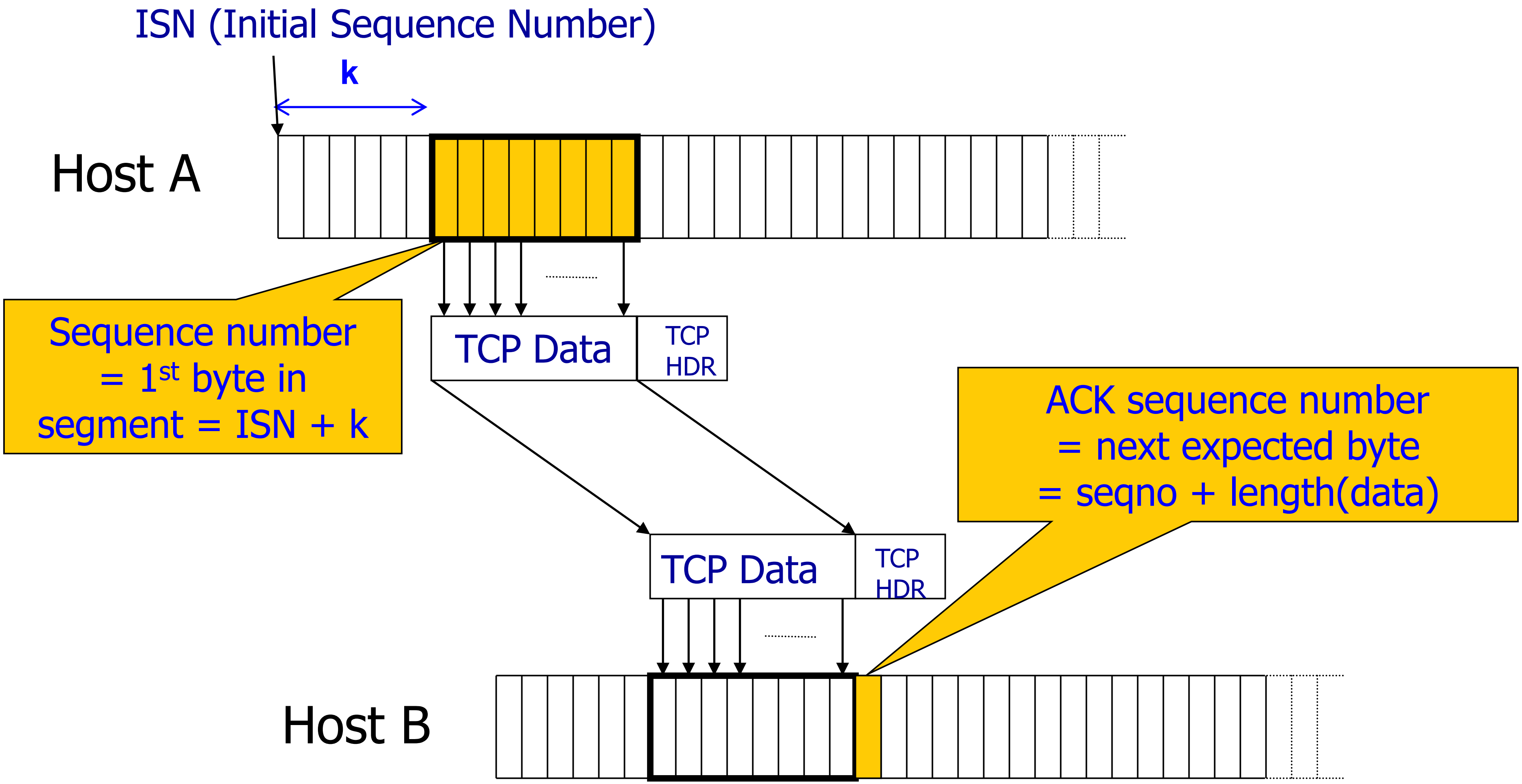


- IP packet
  - No bigger than Maximum Transmission Unit (MTU)
  - E.g., up to 1500 bytes with Ethernet
- TCP packet
  - IP packet with a TCP header and data inside
  - TCP header ≥ 20 bytes long
- TCP segment
  - No more than Maximum Segment Size (MSS) bytes
  - E.g., up to 1460 consecutive bytes from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

Sequence numbers



Sequence numbers



TCP header

Starting byte offset of data carried in this segment

<table>
<tr><td colspan="3">Source port</td><td>Destination port</td></tr>
<tr><td colspan="4">Sequence number</td></tr>
<tr><td colspan="4">Acknowledgment</td></tr>
<tr><td>HdrLen</td><td>0</td><td>Flags</td><td>Advertised window</td></tr>
<tr><td colspan="3">Checksum</td><td>Urgent pointer</td></tr>
<tr><td colspan="4">Options (variable)</td></tr>
<tr><td colspan="4">Data</td></tr>
</table>

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

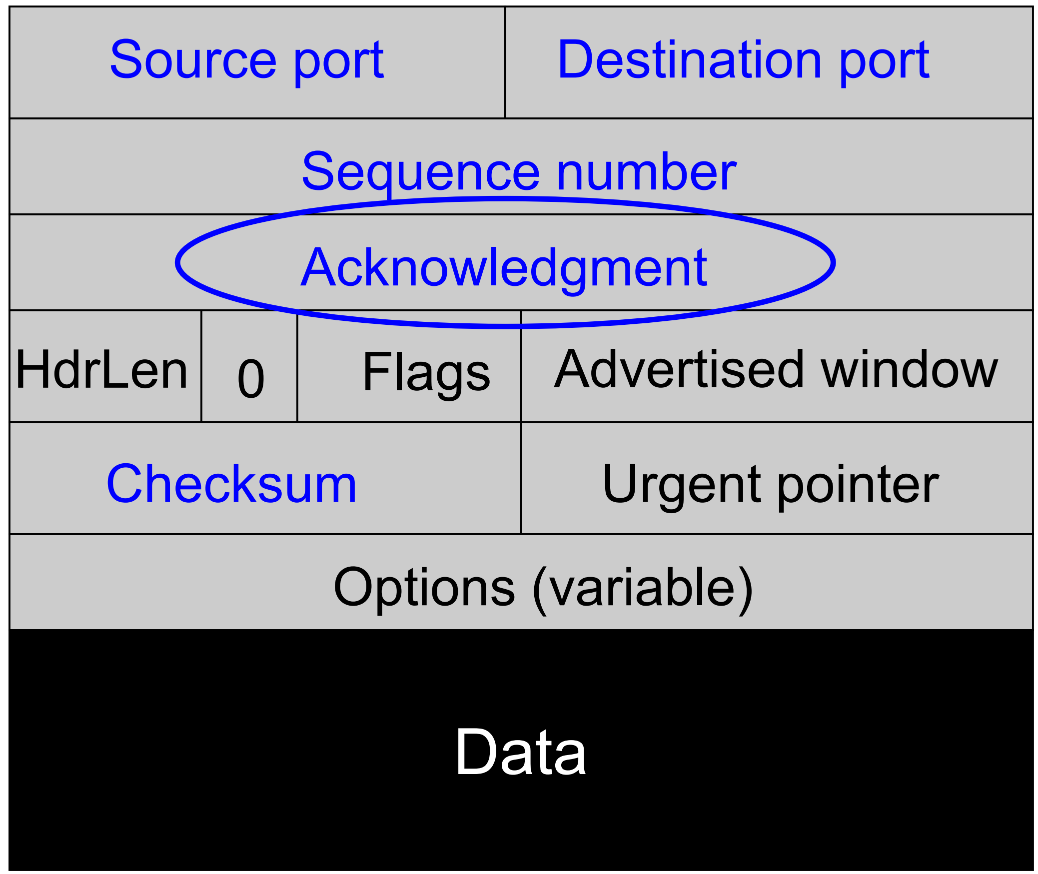
Typical operation

- 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 **<u>in order</u>**



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?

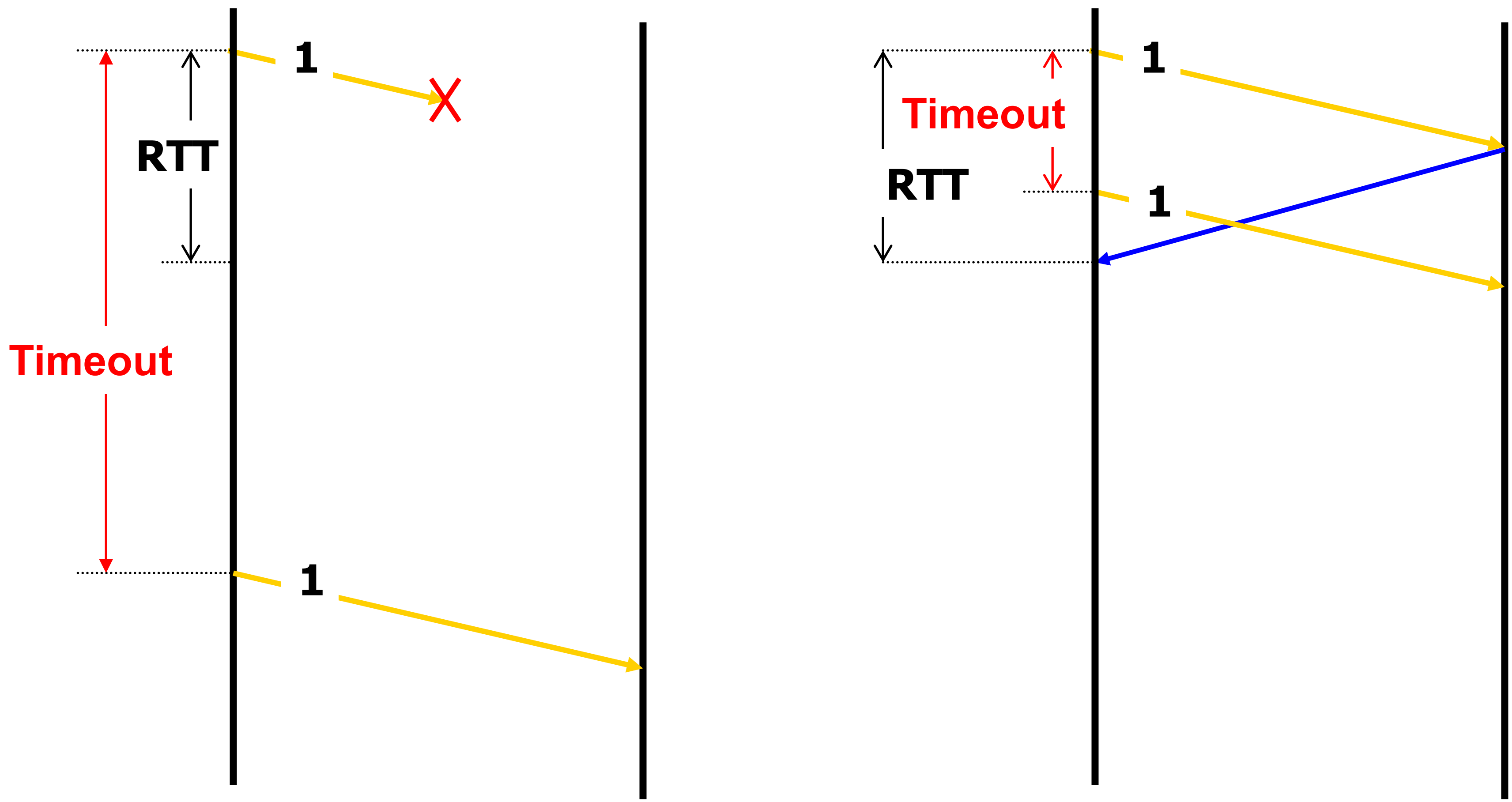
What does TCP introduce?

- 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

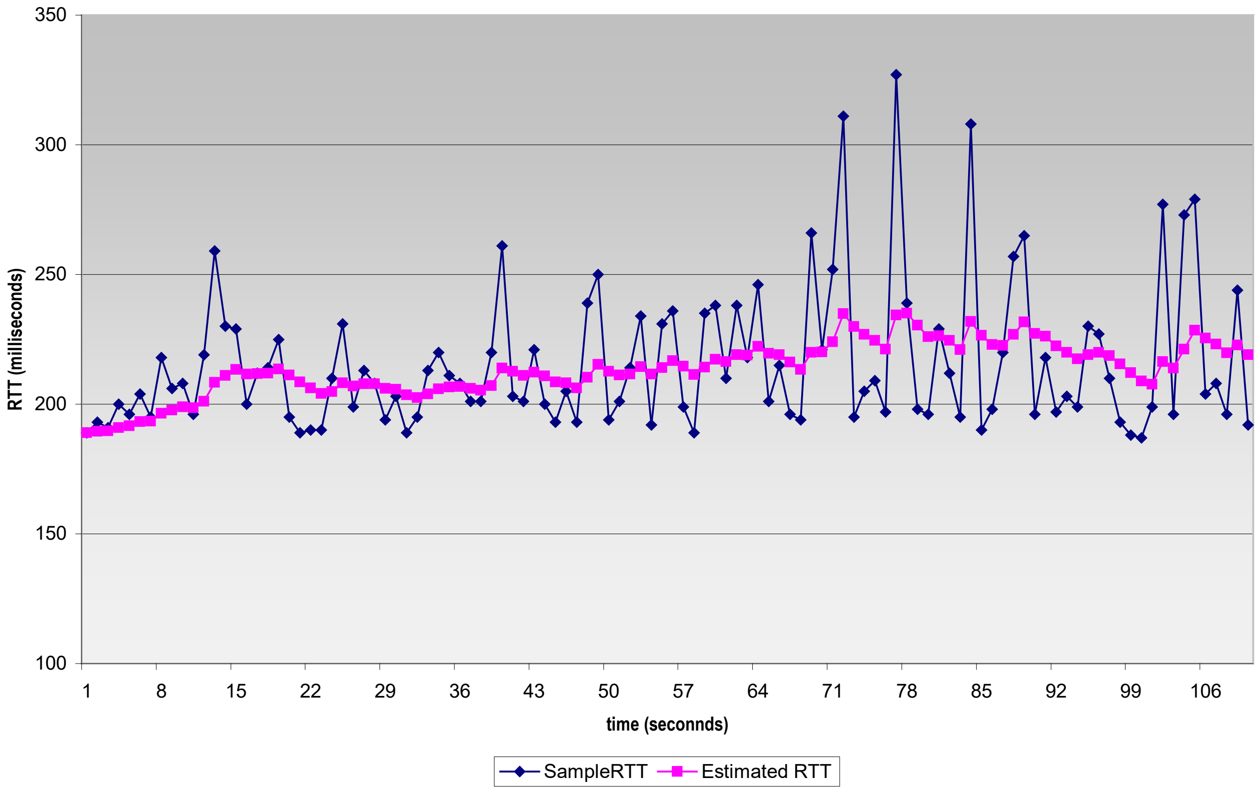
Retransmission timeout

- 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

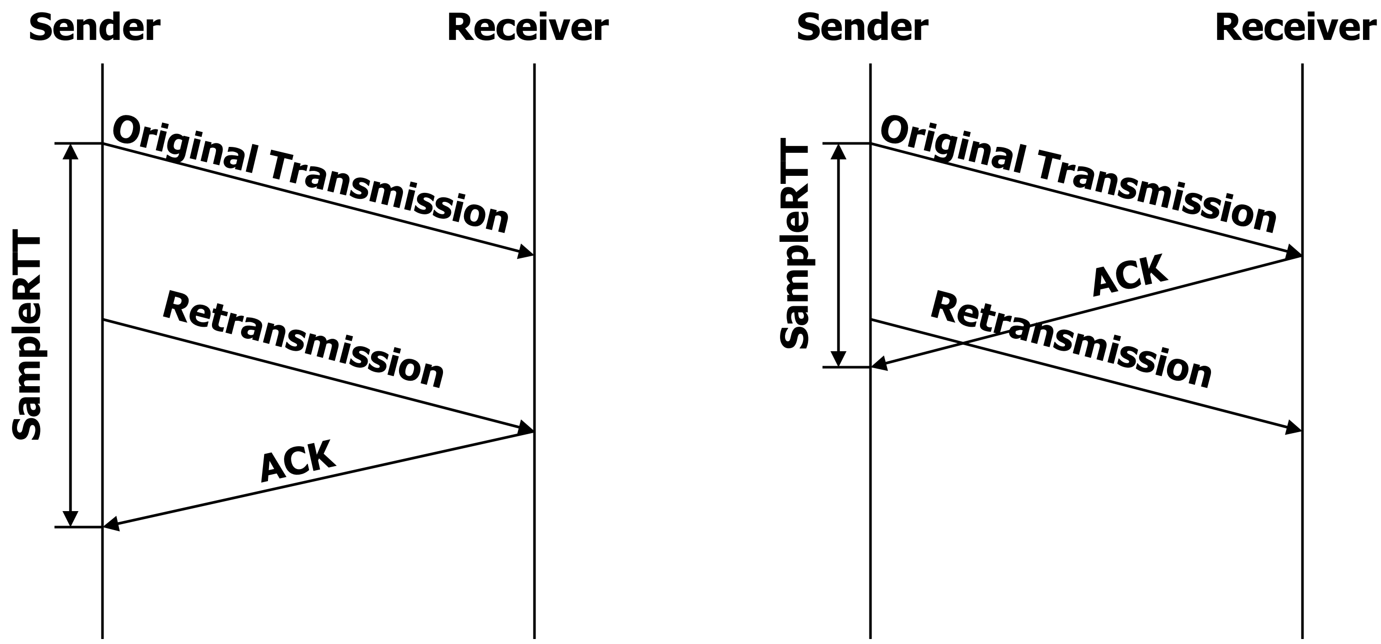
- Exponential weighted average of RTT samples

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 α = 0.125
- Timeout value (RTO) = 2 × EstimatedRTT
  - Employs exponential backoff
    - Every time RTO timer expires, set RTO ← 2·RTO
      - (Up to maximum ≥ 60 sec)
    - Every time new measurement comes in (= successful original transmission), collapse RTO back to 2 × EstimatedRTT
- Insensitive 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

$$SRTT(k+1) = (1-g) \times SRTT(k) + g \times RTT(k+1)$$

$$SERR(k+1) = RTT(k+1) - SRTT(k)$$

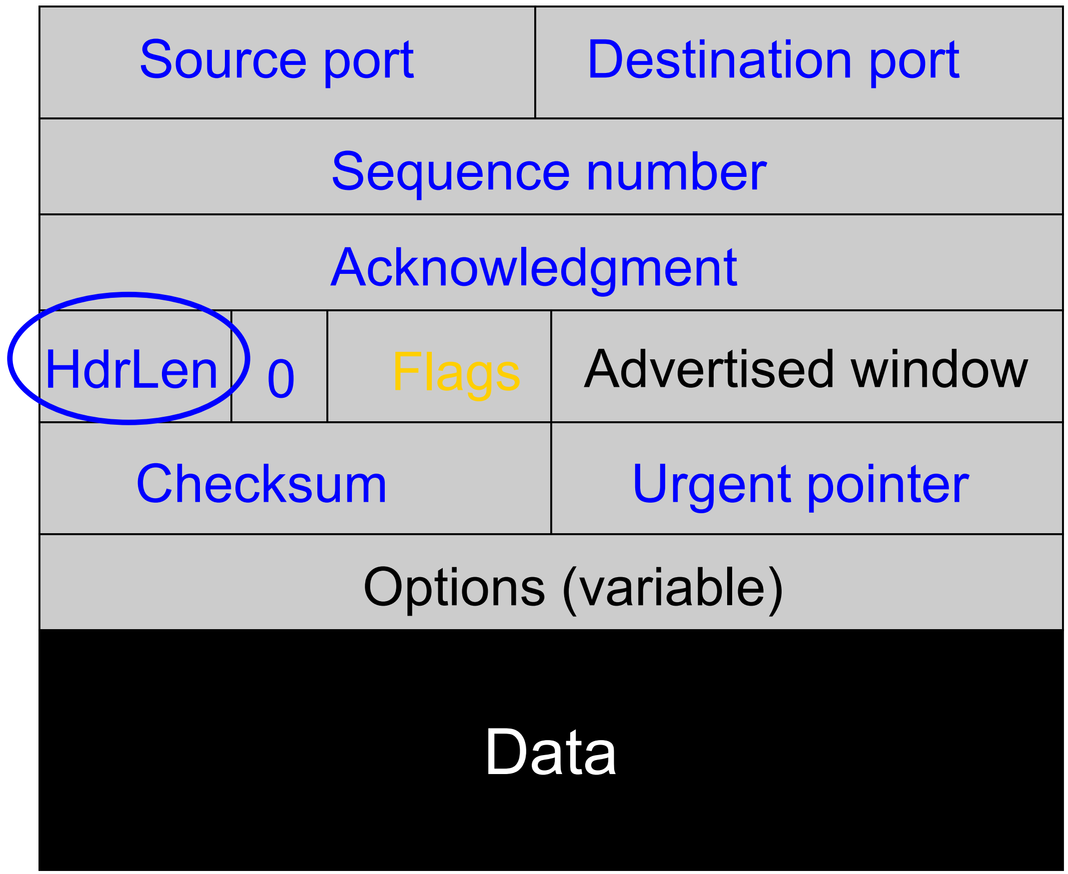
$$SDEV(k+1) = (1-h) \times SDEV(k) + h \times \left| SERR(k+1) \right|$$

$$RTO(k+1) = SRTT(k+1) + f \times SDEV(k+1)$$

$$g = 1/8 = 0.125 \quad h = 1/4 = 0.25 \quad f = 2 \text{ or } 4$$

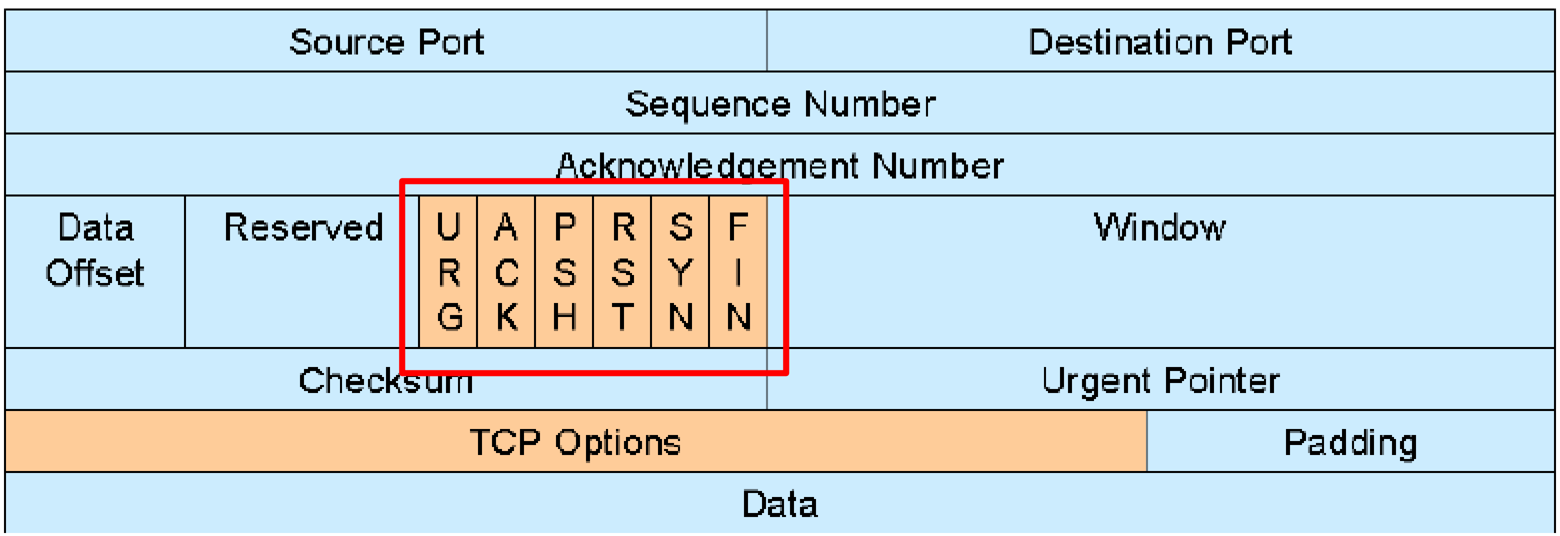
TCP header

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



TCP Connection Establishment

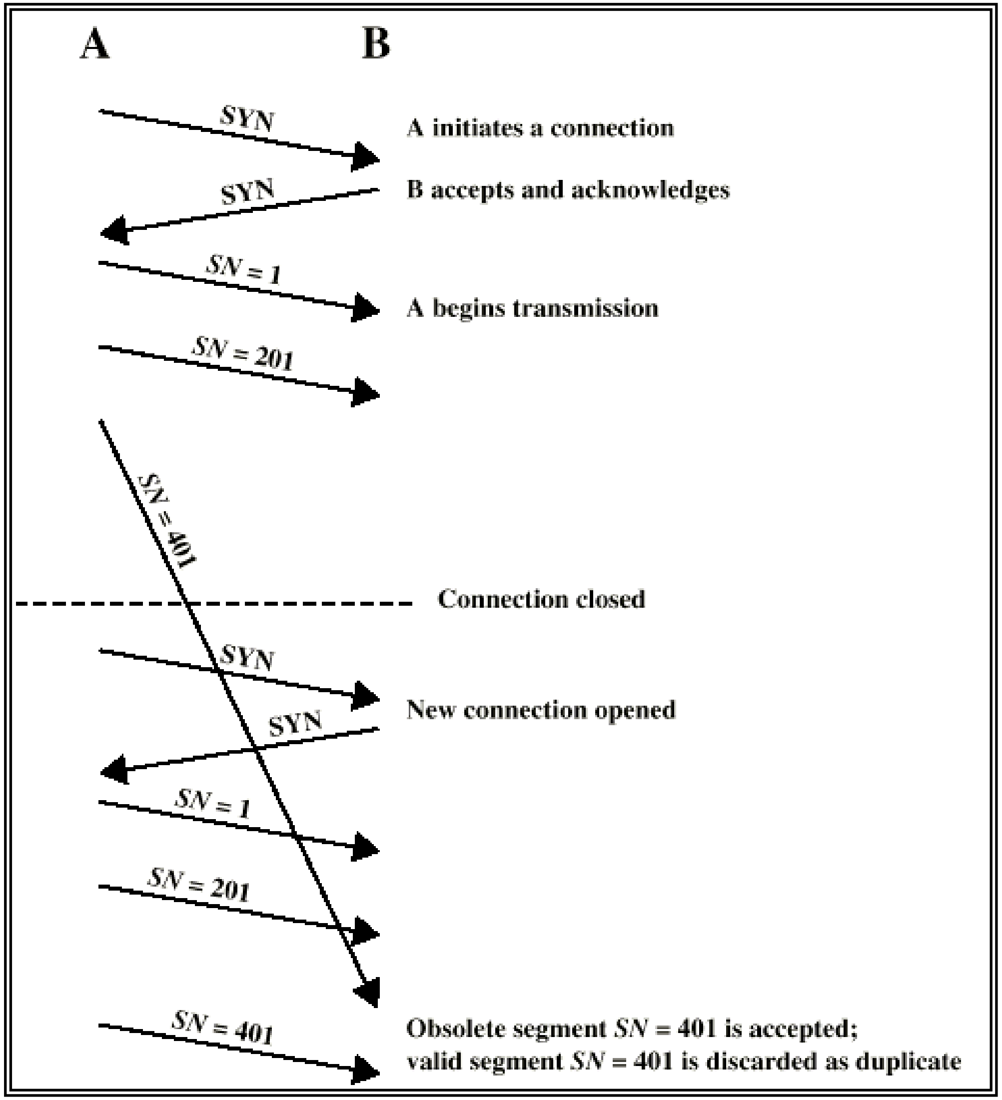
- TCP header field for connection establishment and teardown



Connection Establishment

- 2-way handshake
  - A sends SYN, B replies with SYN
  - Lost SYNs handled by re-transmission
  - Ignore duplicate SYNs once connected
- Problem
  - How to recognize slipped segments from old connection
  - How to recognize duplicated obsolete SYN

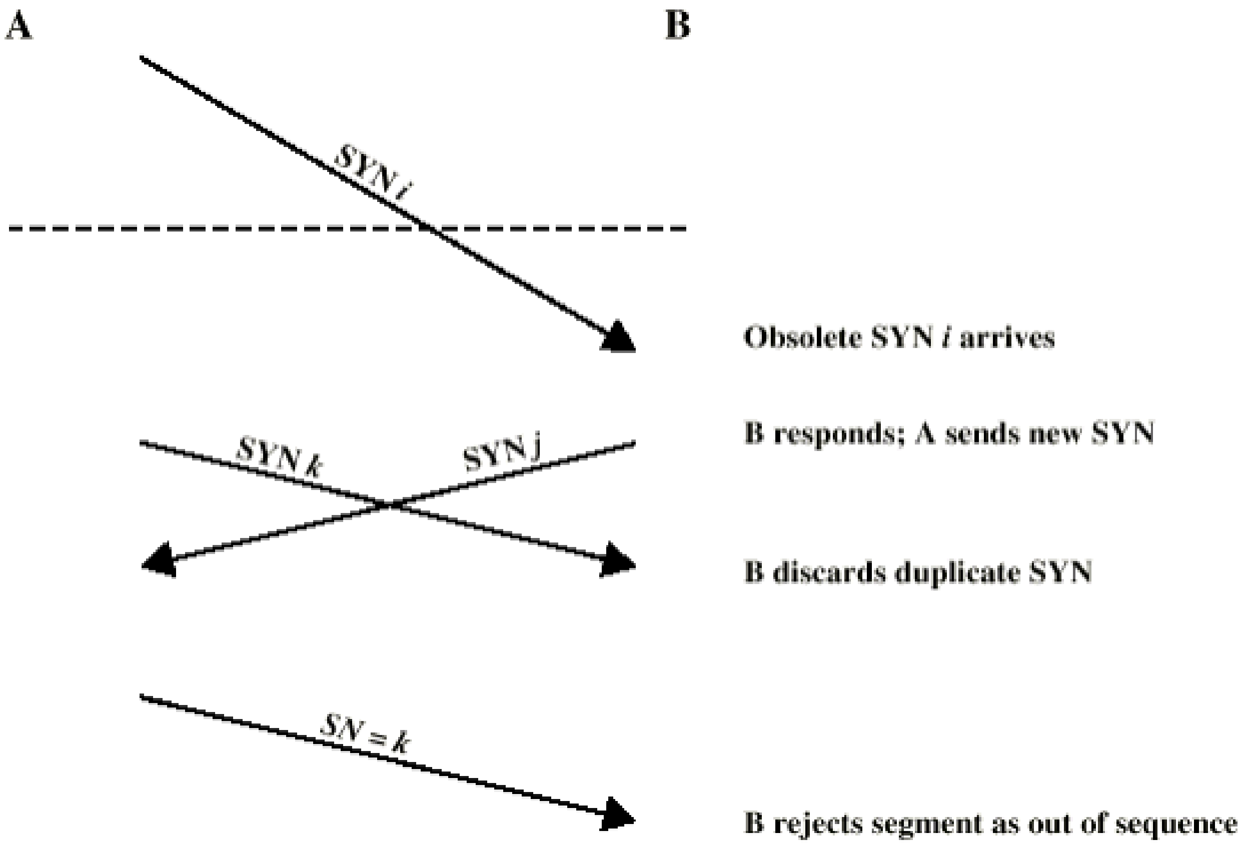
2-Way Handshake: Slipped Data Segment



Initial Sequence Number (ISN)

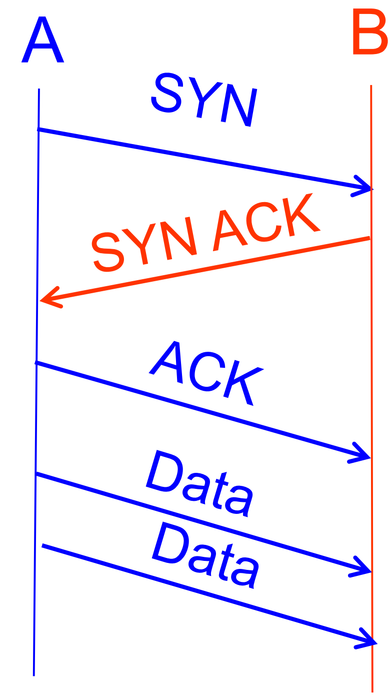
- Handle
  - Start each new connection with a different initial sequence number (ISN) far from previous connection
  - The connection request is of the form SYN i+1, where i is the sequence number of the first data segment that will be sent on this connection.
- However:

2-Way Handshake: Obsolete SYN



Solution: three-way handshake

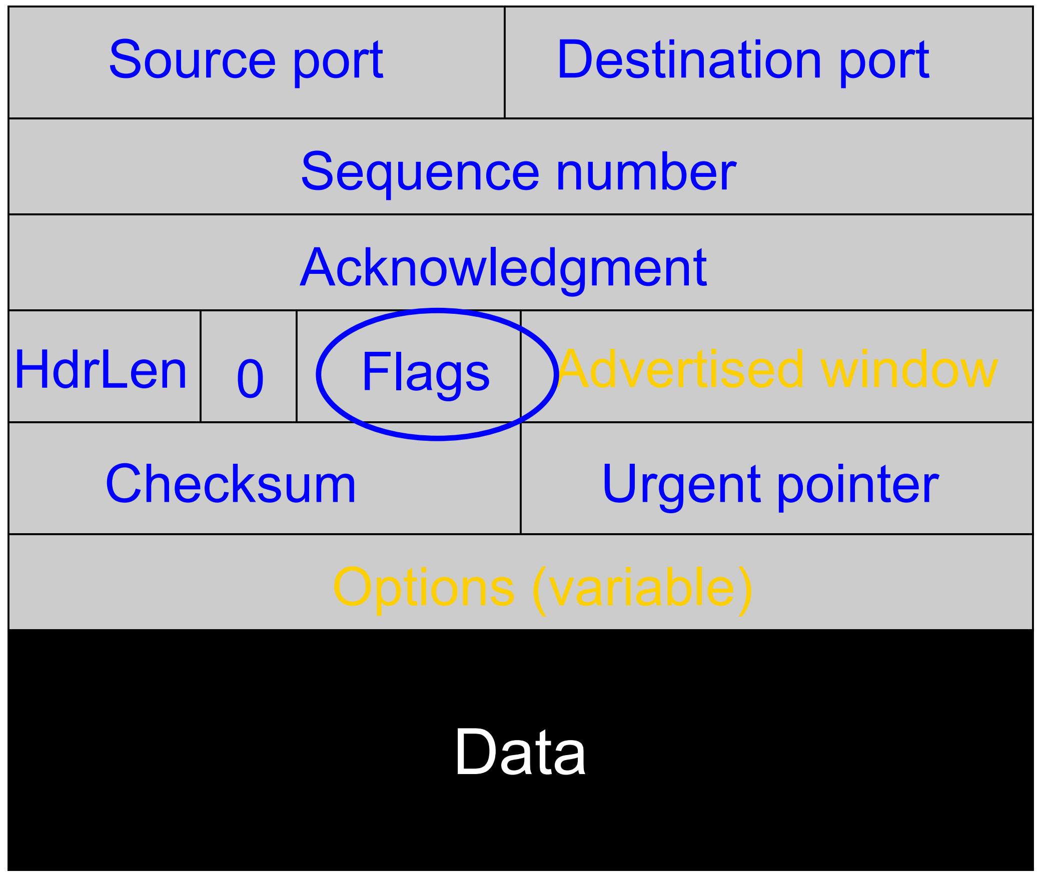
- Three-way handshake to establish connection
  - Host A sends a SYN (open; "synchronize sequence numbers") to host B
  - Host B returns a SYN acknowledgment (SYN ACK)
  - Host A sends an ACK to acknowledge the SYN ACK



三方握手：确认对方的**SYN**和序号

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>Advertised window</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

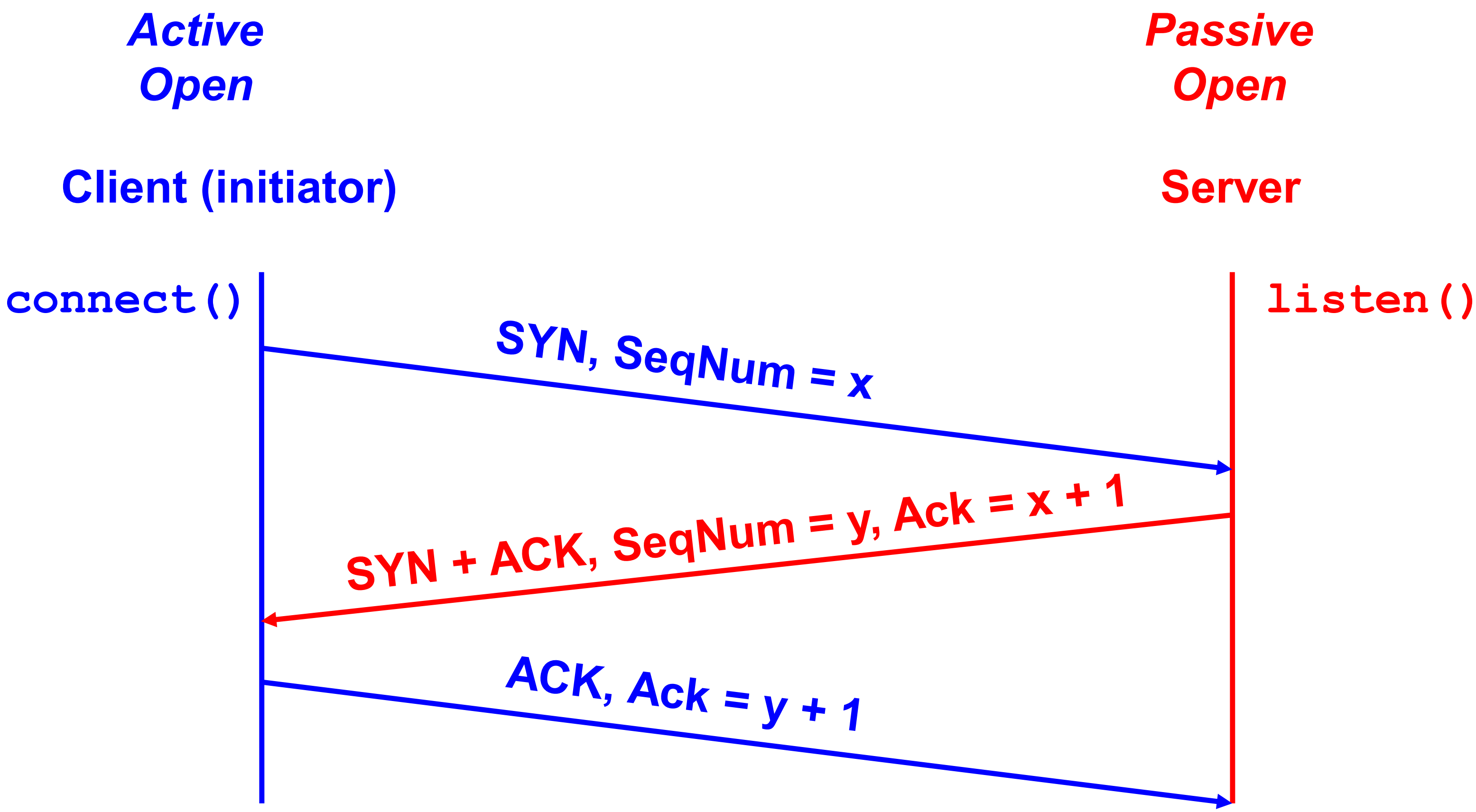
B's port		A's port	
B's Initial Sequence Number			
ACK=A's ISN+1			
5	0	SYN|ACK	Advertised window
Checksum		Urgent pointer	

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



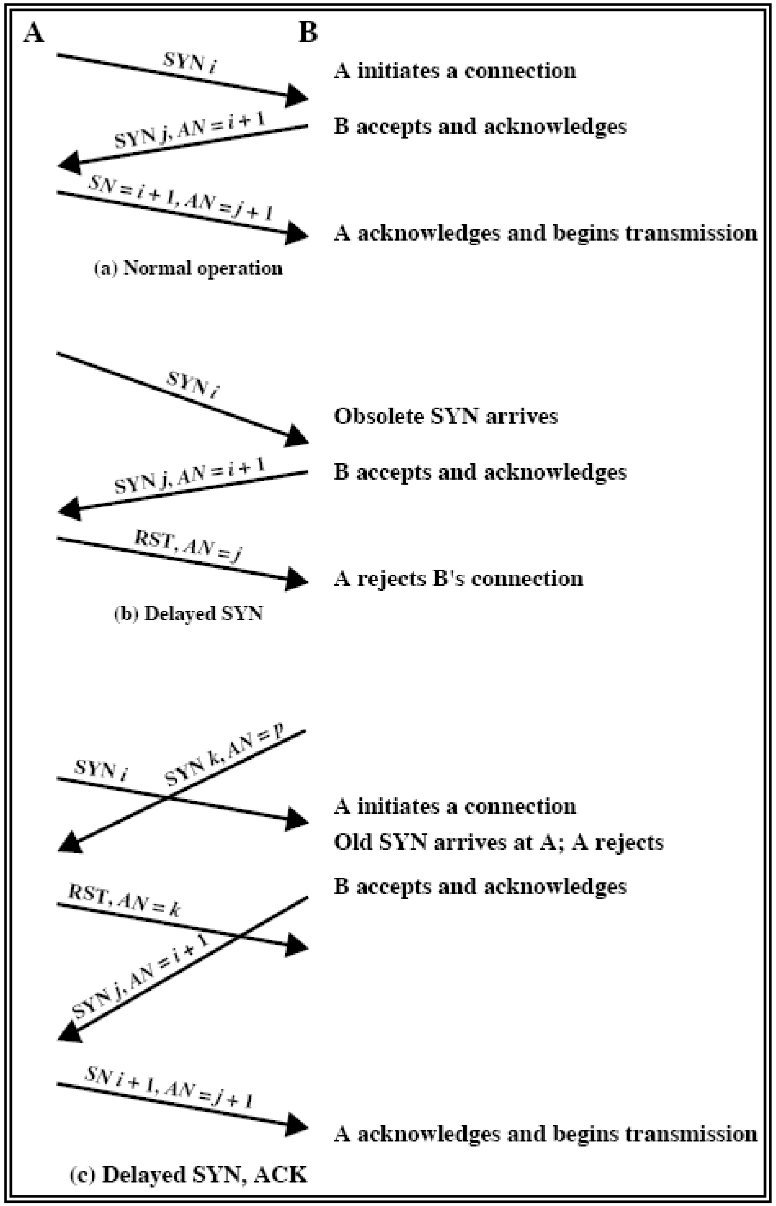
What if the SYN Packet Gets Lost?

- 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

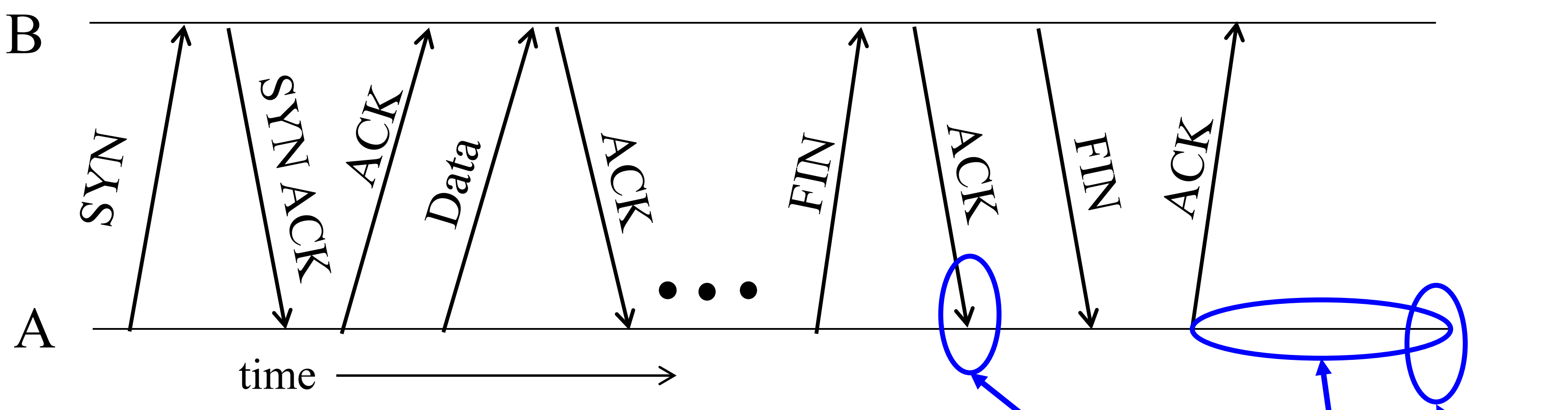
- User clicks on a hypertext link
  - Browser creates a socket and does a "connect"
  - The "connect" triggers the OS to transmit a SYN
- If the SYN is lost…
  - 3-6 seconds of delay: can be very long
  - User may become impatient and can retry
- User triggers an "abort" of the "connect"
  - Browser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

Three-Way Handshake: Examples



TCP connection teardown

Normal termination, one side at a time



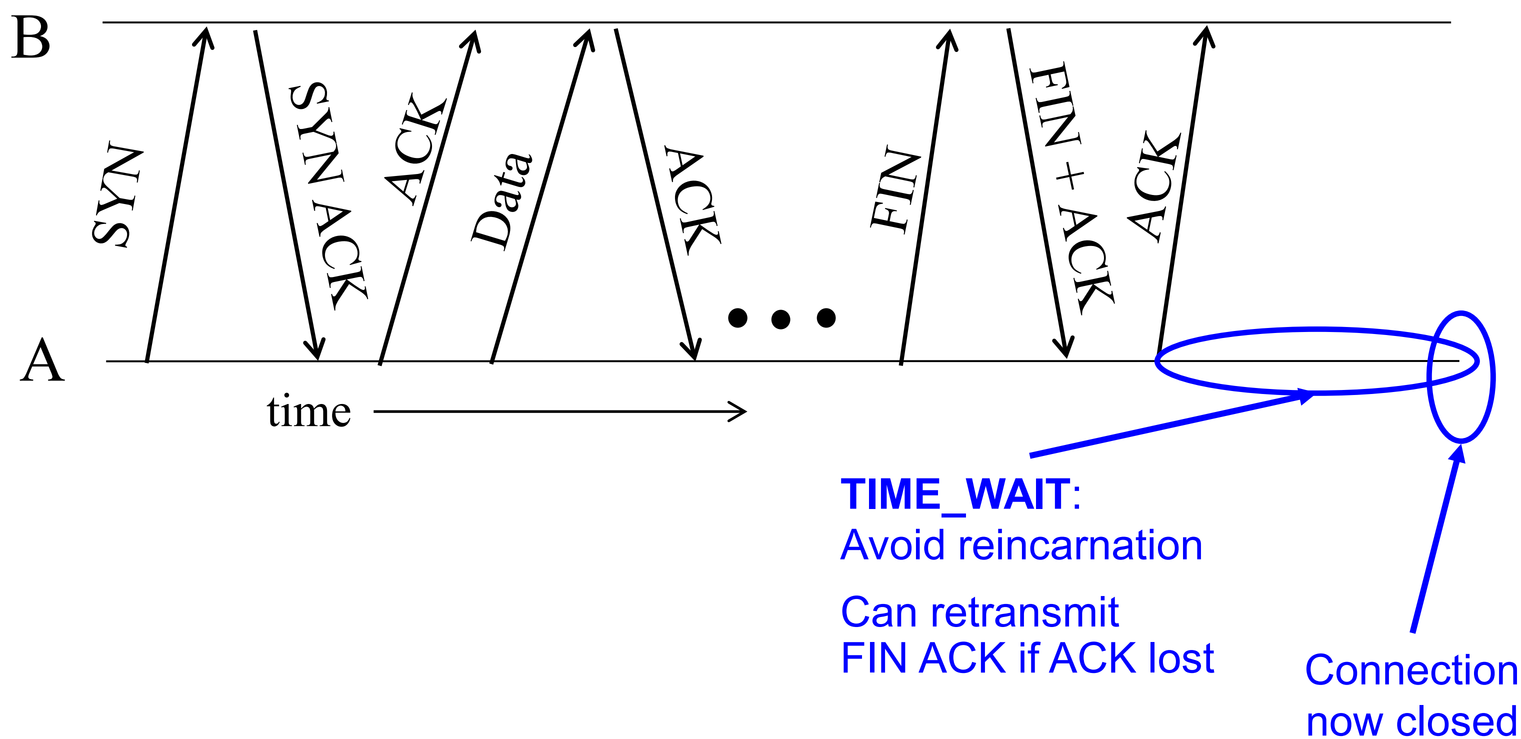
- Finish (FIN) to close and receive remaining bytes
  - FIN occupies one byte in the sequence space
- Other host acks the byte to confirm
- Closes A's side of the connection, but not B's
  - Until B likewise sends a FIN
  - Which A then acks

TIME_WAIT:

Avoid reincarnation

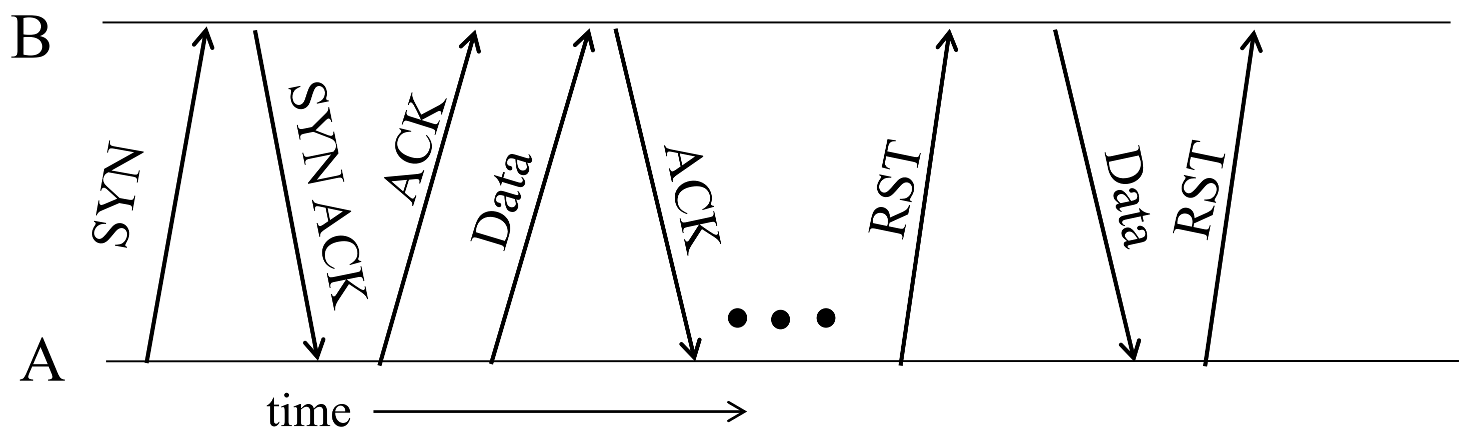
B will retransmit FIN if ACK is lost

Normal termination, both together



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

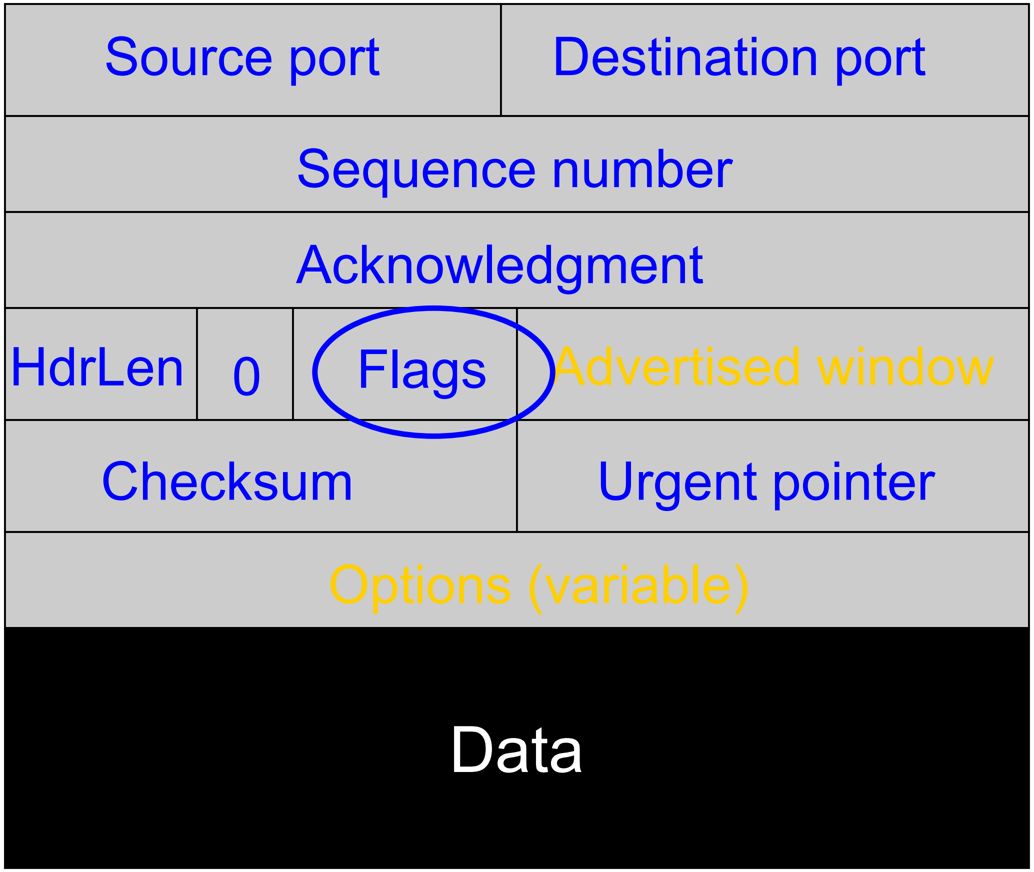
Abrupt termination



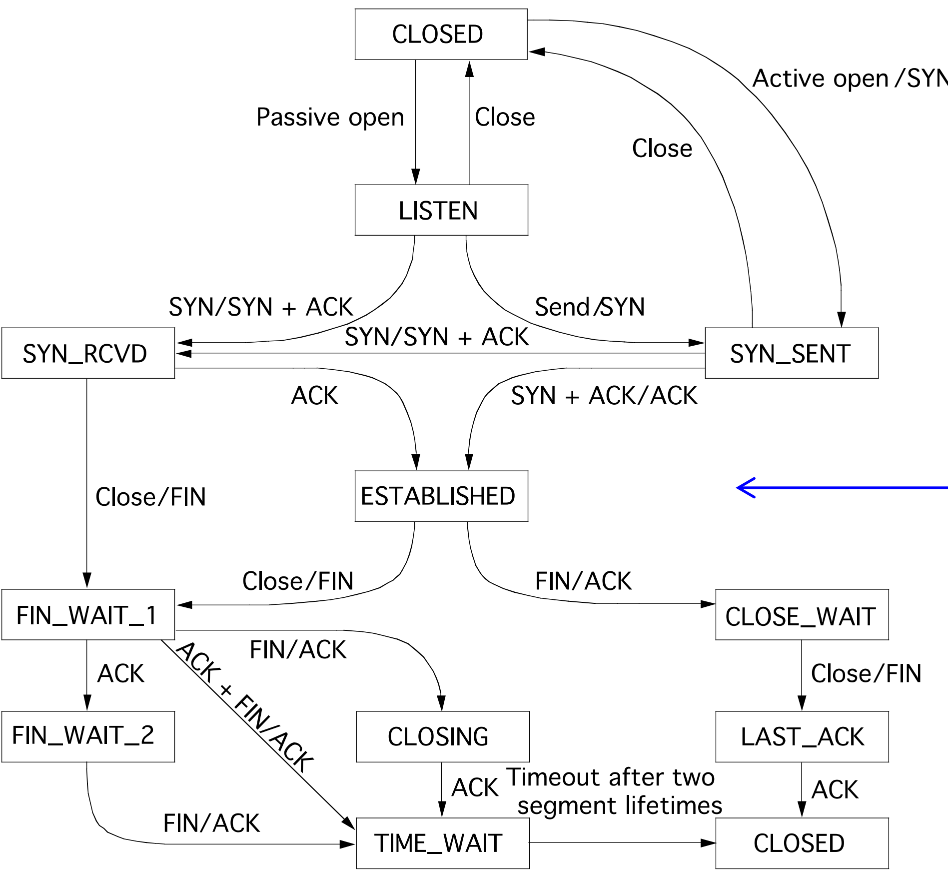
- A sends a RESET (RST) to B
  - E.g., because application process on A crashed
- That’s it
  - B does not ack the RST
  - Thus, RST is not delivered reliably, and any data in flight is lost
  - But: if B sends anything more, will elicit another RST

TCP header

Flags:
SYN
ACK
FIN
RST
PSH
URG

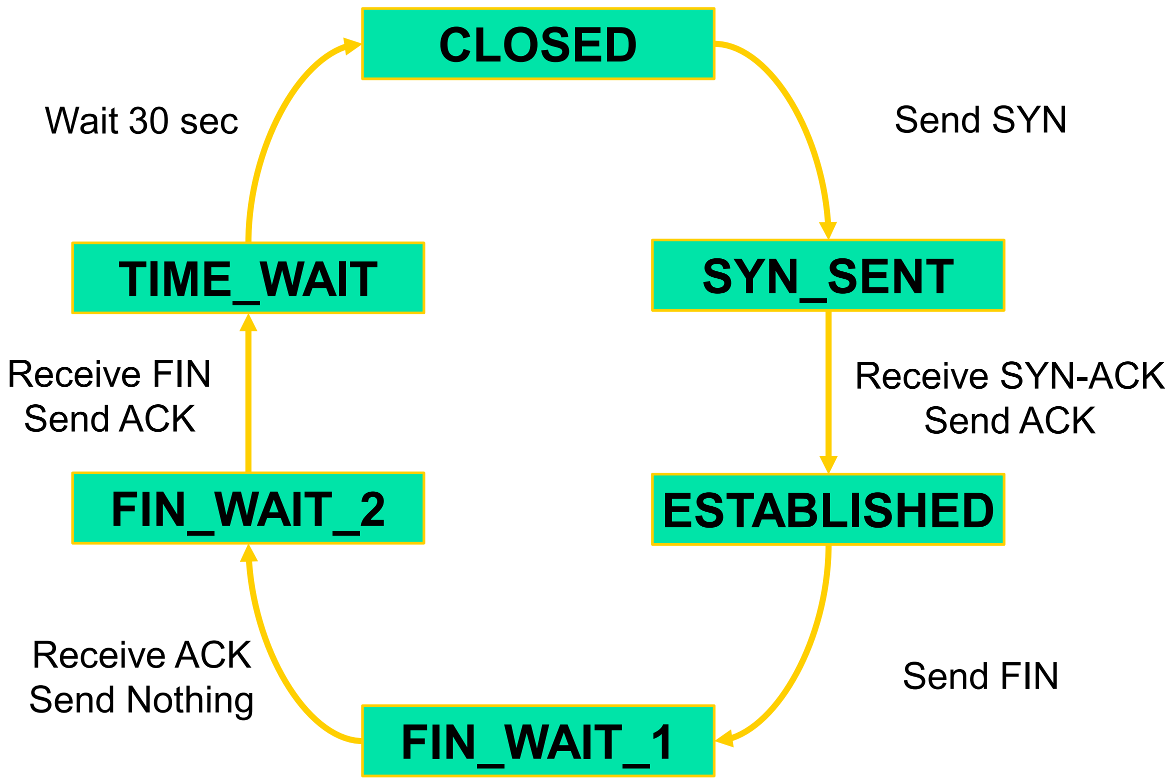


TCP state transitions

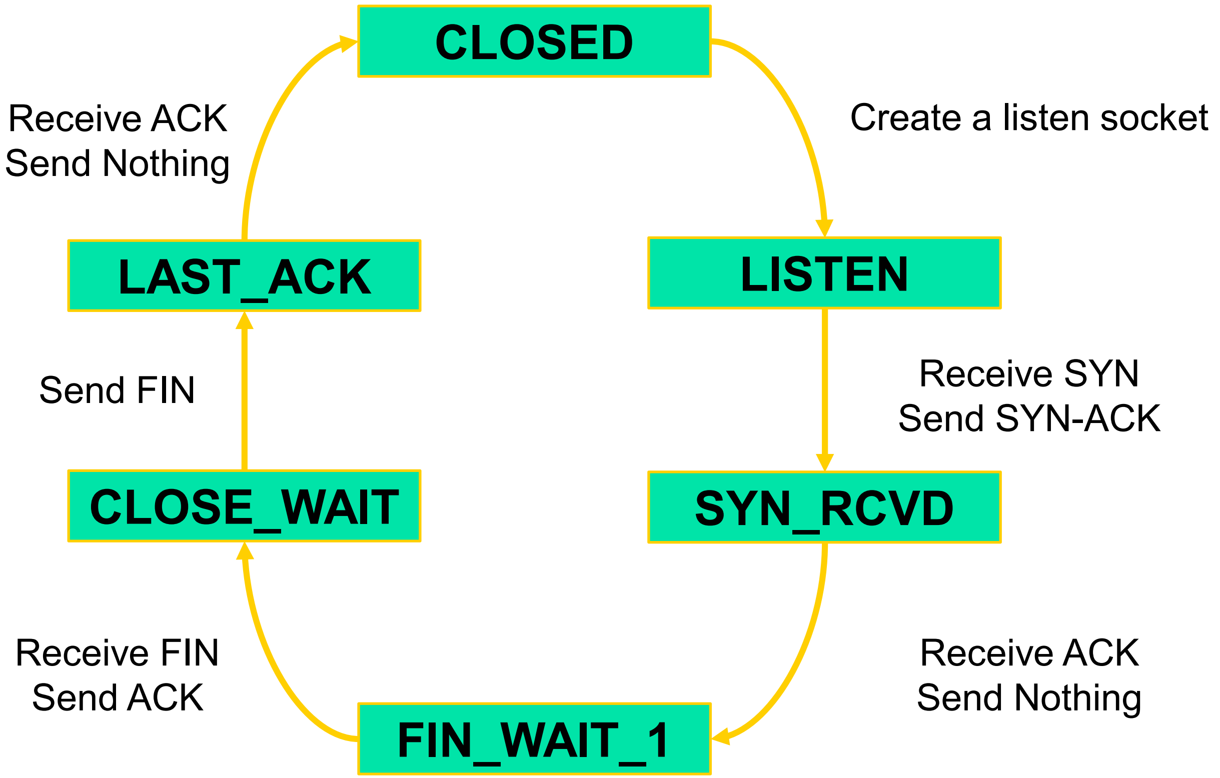


Data, ACK exchanges are in here

TCP client lifecycle



TCP server lifecycle



Summary

- UDP
- TCP
- TCP header fields
- TCP Connection Establishment
  - 2-way and 3-way handshake
- TCP connection teardown