Computer Networks and Internet Technology

2021W703033 VO Rechnernetze und Internettechnik
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Jan Beutel

universität innsbruck

With materials and inspiration from Scott Shenker, Jennifer Rexford, Phil Levis, Adrian Perrig and Laurent Vanbever

Communication Networks and Internet Technology

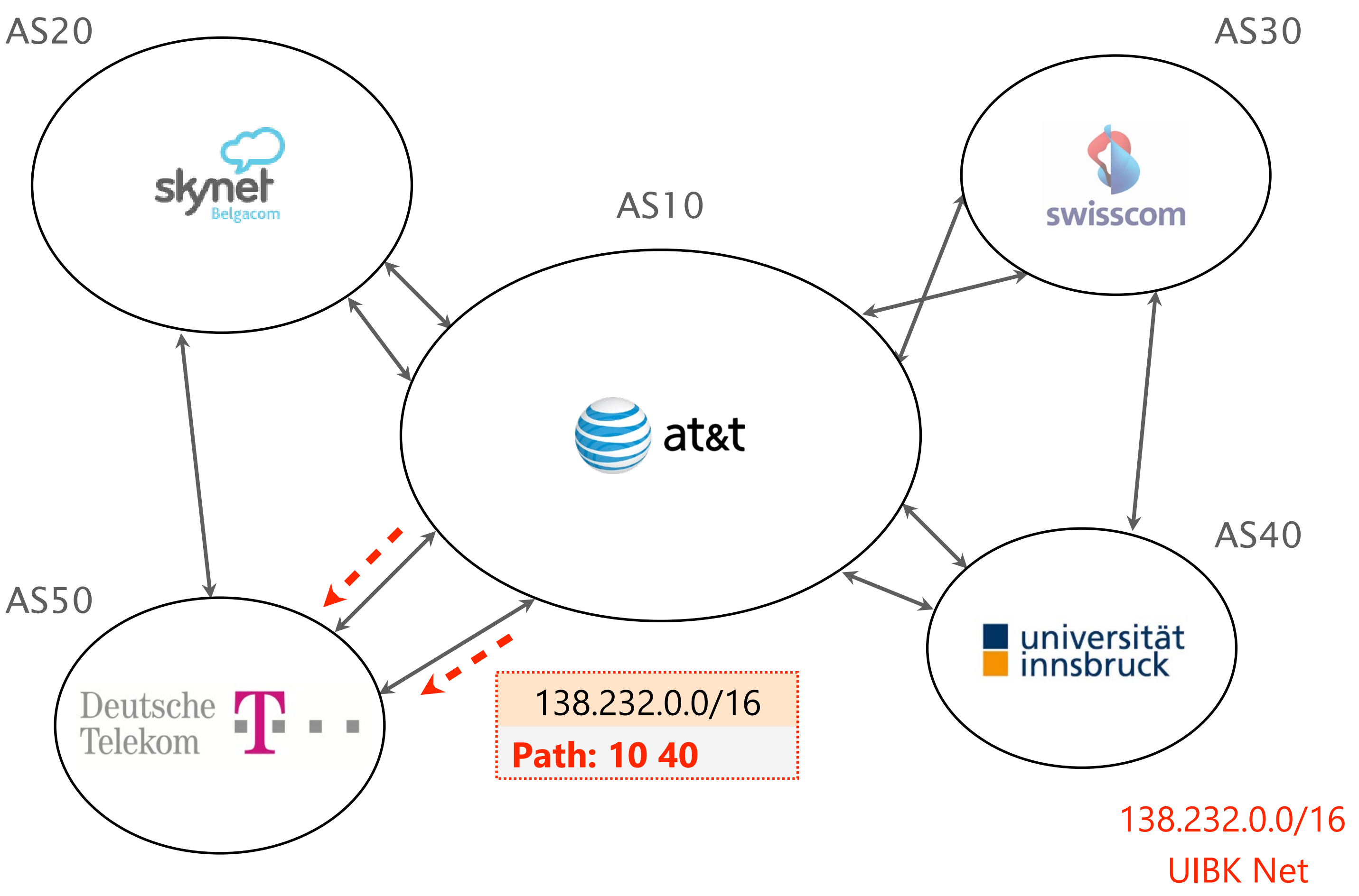
Recap of last weeks lecture

BGP relies on path-vector routing to support flexible routing policies and avoid count-to-infinity

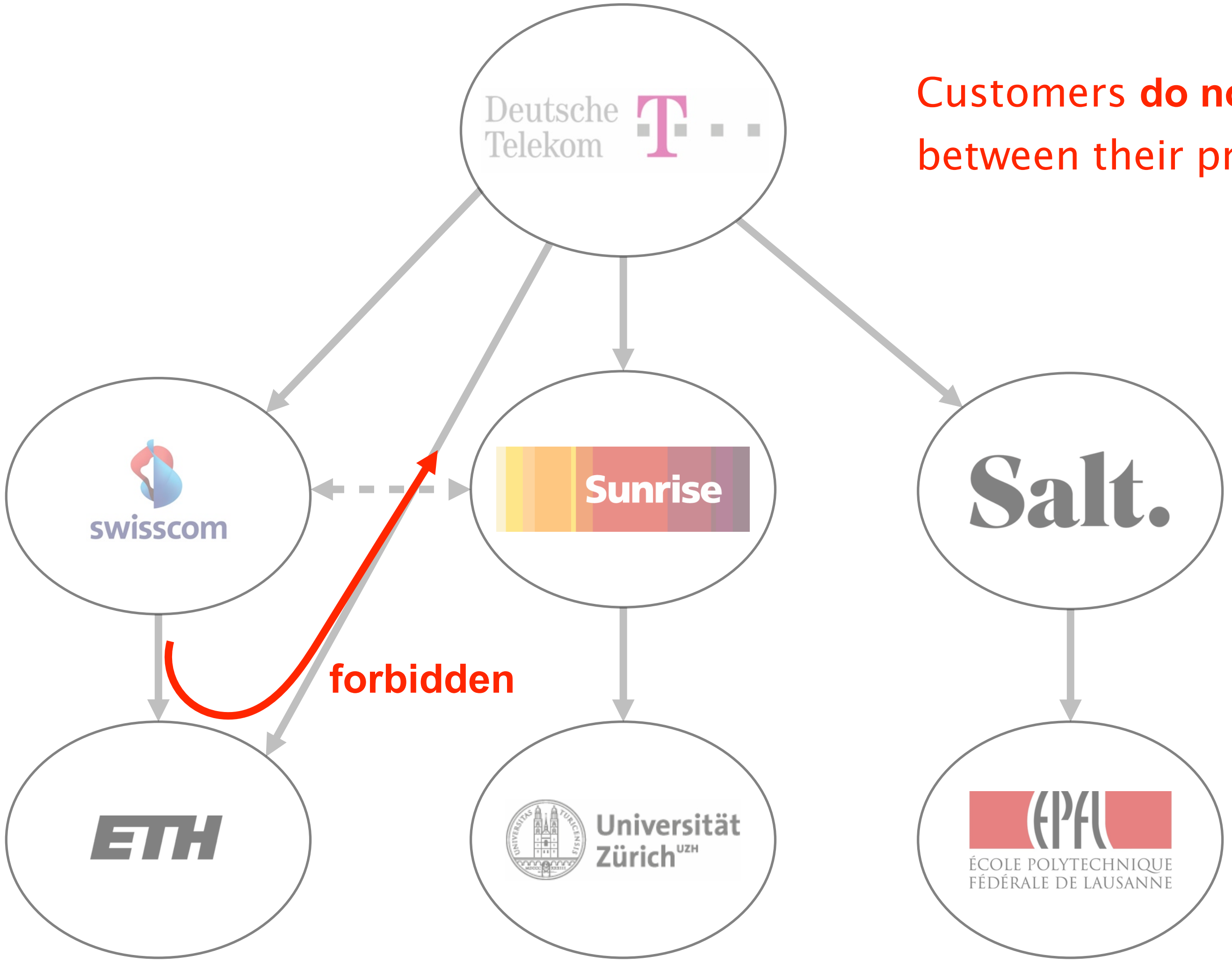
key idea

advertise the entire path instead of distances

Each AS appends itself to the path when it propagates announcements

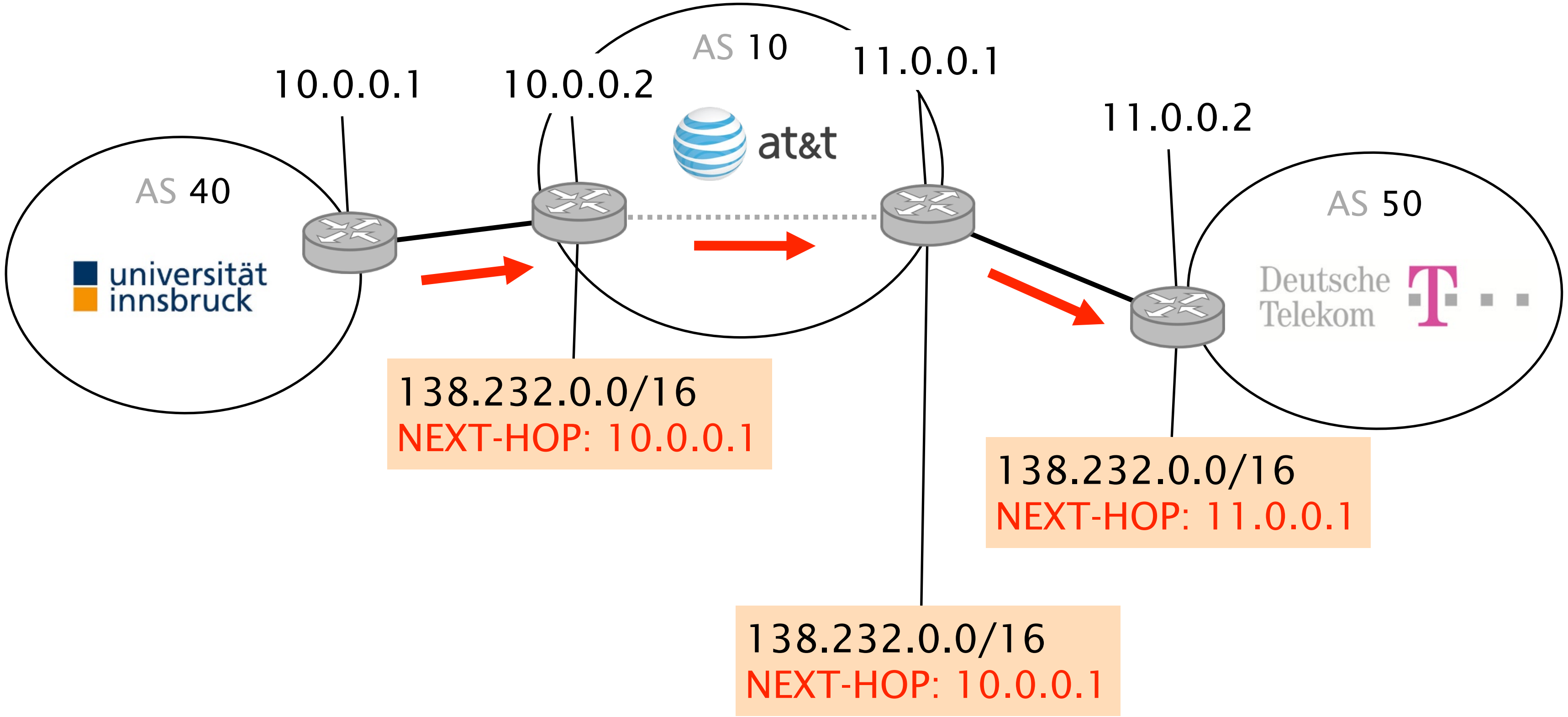


Customers **do not** transit traffic between their providers



The NEXT-HOP is set when the route enters an AS,

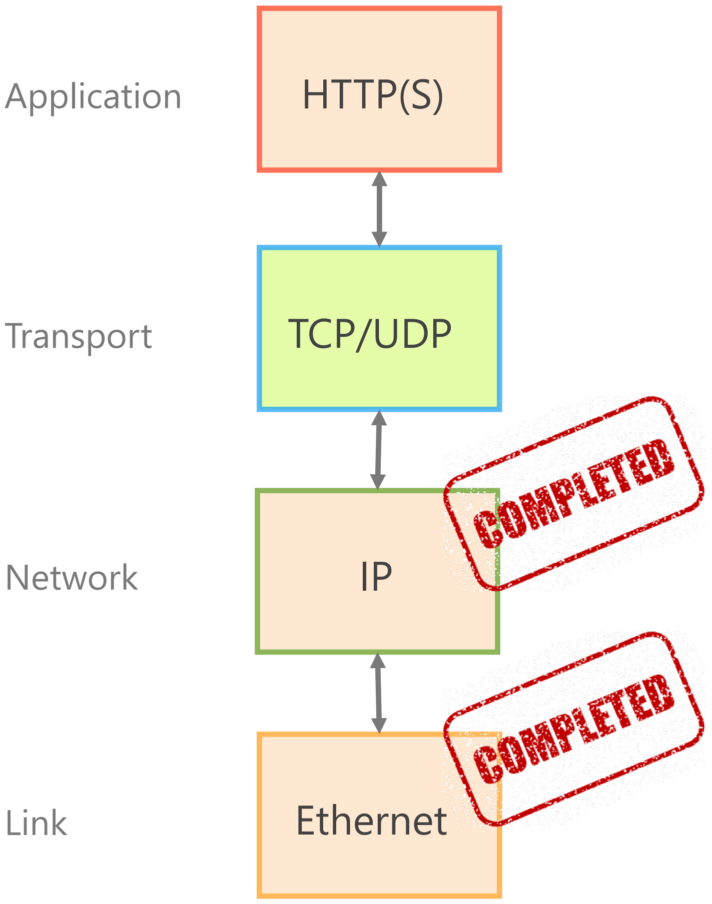
it does not change within the AS



Communication Networks and Internet Technology

This weeks lecture

We're continuing our journey up the layers,
now looking at the transport layer



UDP / TCP

Congestion
Control

Un/reliable Transport

UDP / TCP

Congestion
Control

Un/reliable Transport

What do we need in the Transport layer?

- Functionality implemented in network
  - Keep minimal (easy to build, broadly applicable)

- Functionality implemented in the application
  - Keep minimal (easy to write)
  - Restricted to application-specific functionality

- Functionality implemented in the “network stack”
  - The shared networking code on the host
  - This relieves burden from both app and network
  - The transport layer is a key component here

What do we need in the Transport layer?

- **Application layer**
  - Communication for specific applications
  - *e.g.*, HyperText Transfer Protocol (HTTP), File Transfer Protocol (FTP)
- **Network layer**
  - Global communication between hosts
  - Hides details of the link technology
  - *e.g.*, Internet Protocol (IP)

What Problems Should Be Solved Here?

- Data delivering, to the *correct* application
  - IP just points towards next protocol
  - *Transport needs to demultiplex incoming data (ports)*
- Files or bytestreams abstractions for the applications
  - Network deals with packets
  - *Transport layer needs to translate between them*
- Reliable transfer (if needed)
- Not overloading the receiver
- Not overloading the network

What Is Needed to Address These?

- *Demultiplexing*: identifier for application process
  - Going from host-to-host (IP) to process-to-process
- *Translating between bytestreams and packets:*
  - Do segmentation and reassembly
- *Reliability*: ACKs and all that stuff
- *Corruption*: Checksum
- *Not overloading receiver*: “Flow Control”
  - Limit data in receiver’s buffer
- *Not overloading network*: “Congestion Control”

UDP: Datagram messaging service

UDP provides a connectionless, unreliable transport service

- No-frills extension of “best-effort” IP
- UDP provides only two services to the App layer
  - Multiplexing/Demultiplexing among processes
  - Discarding corrupted packets (optional)

TCP: Reliable, in-order delivery

- TCP provides a connection-oriented, reliable, bytestream transport service

- ***What UDP provides, plus:***
  - Retransmission of lost and corrupted packets
  - Flow control (to not overflow receiver)
  - Congestion control (to not overload network)
  - “Connection” set-up & tear-down

Connections (or sessions)

- Reliability requires keeping state
  - Sender: packets sent but not ACKed, and related timers
  - Receiver: noncontiguous packets

- Each bytestream is called a connection or session
  - Each with their own connection state
  - State is in hosts, not network!

What transport protocols do **not** provide

- Delay and/or bandwidth guarantees
  - This cannot be offered by transport
  - Requires support at IP level (*and let's not go there*)

- Sessions that survive change-of-IP-address
  - This is an artifact of current implementations
  - As we shall see….

Important Context: Sockets and Ports

- **Sockets**: an operating system abstraction

- **Ports**: a networking abstraction
  - This is not a port on a switch (which is an interface)
  - Think of it as a *logical interface* on a host

Sockets

- A socket is a software abstraction by which an application process exchanges network messages with the (transport layer in the) operating system
  - socketID = socket(…, socket.TYPE)
  - socketID.sendto(message, …)
  - socketID.recvfrom(…)

- Two important types of sockets
  - UDP socket: TYPE is SOCK_DGRAM
  - TCP socket: TYPE is SOCK_STREAM

Ports

- **Problem**: which app (socket) gets which packets
- **Solution**: port as transport layer identifier (16 bits)
  - Packet carries source/destination port numbers in transport header
- OS stores mapping between sockets and ports
  - Port: in packets
  - Socket: in OS

More on Ports

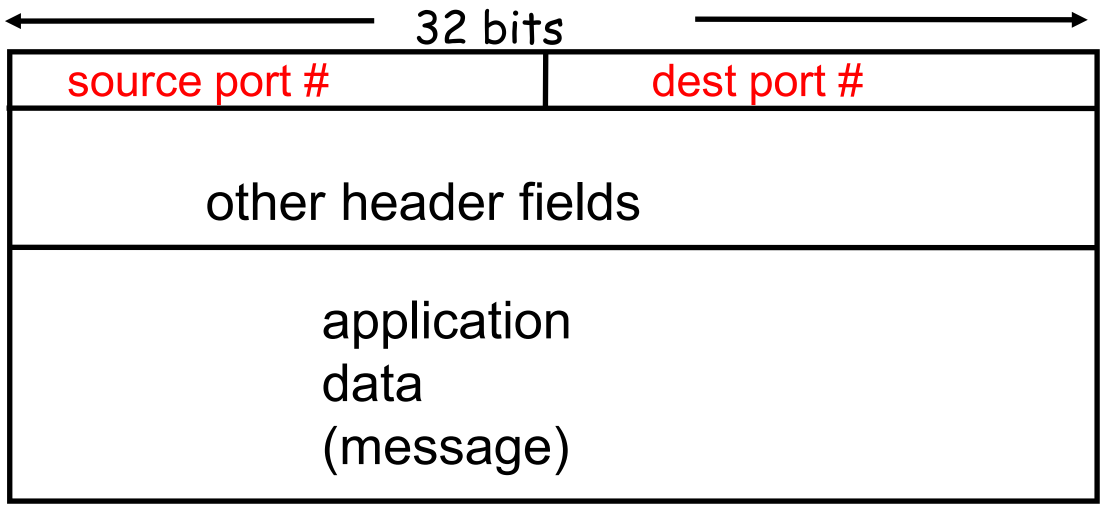
- Separate 16-bit port address space for UDP, TCP

- "Well known" ports (0-1023)
  - Agreement on which services run on these ports
  - *e.g.*, ssh:22, http:80
  - Client (app) knows appropriate port on server
  - Services can listen on well-known port

- Ephemeral ports (most 1024-65535):
  - Given to clients (at random)

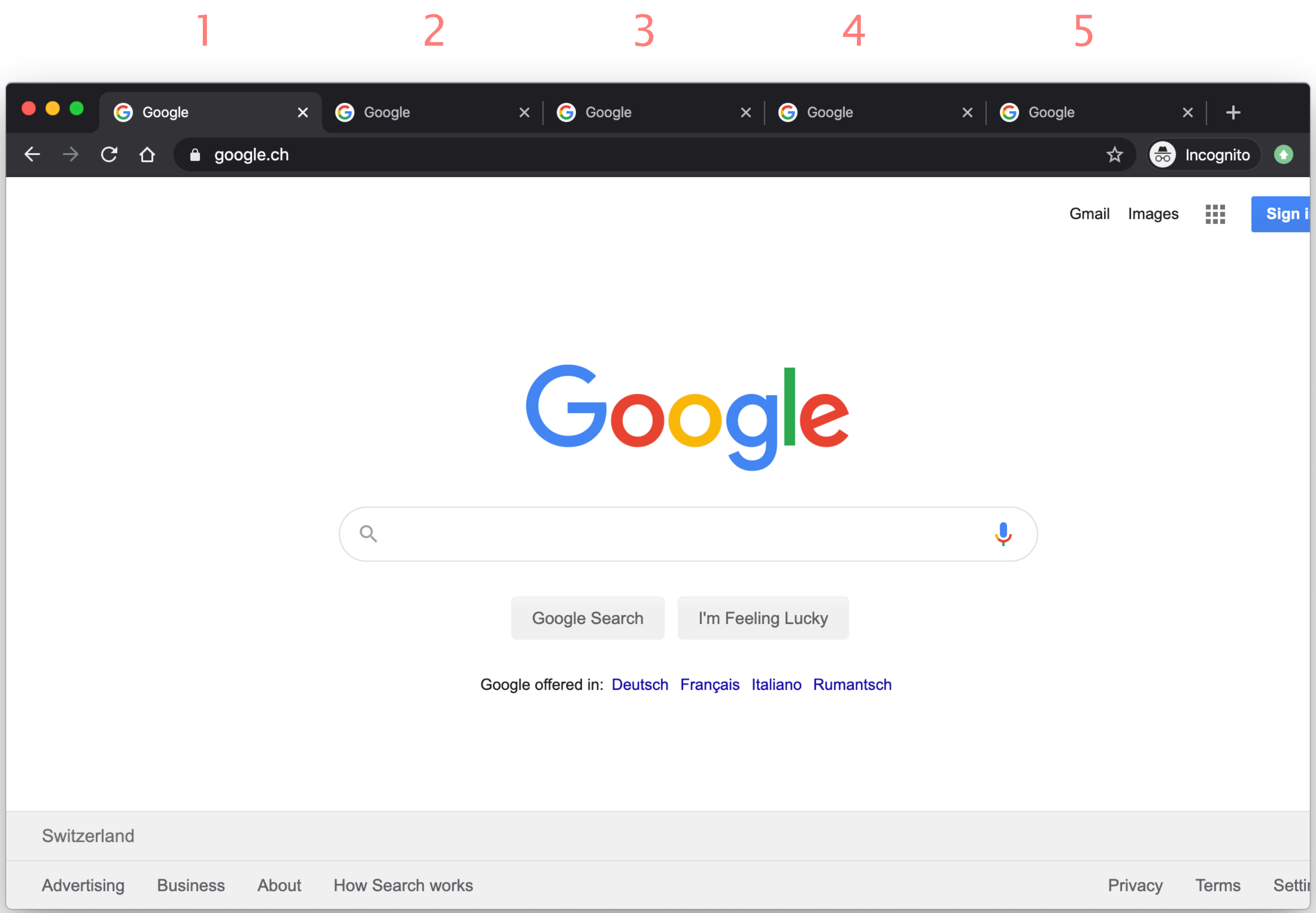
Multiplexing and Demultiplexing

- Host receives IP datagrams
  - Each datagram has source and destination IP address,
  - Each segment has source and destination port number
- Host uses IP addresses and port numbers to direct the segment to appropriate socket



A TCP/UDP socket is identified by a 4-tuple:

(src IP, src port, dst IP, dest port)



Let's say you open 5 tabs to google.ch

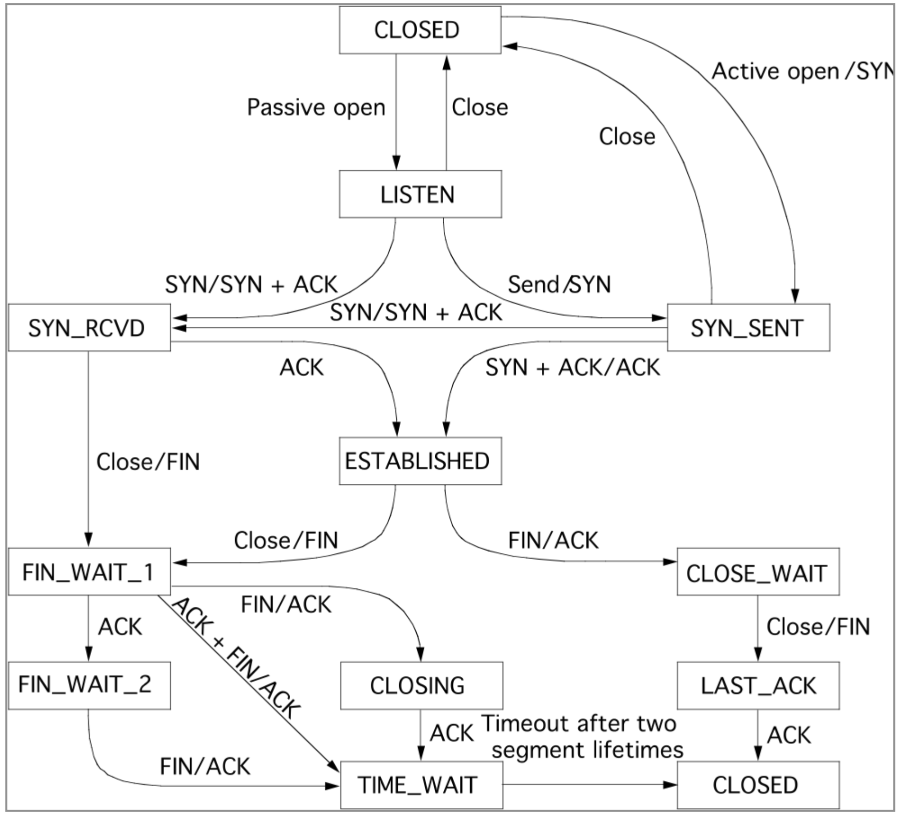
Your IP: 129.132.19.1

Google's IP: 172.217.168.3

Client OS		src IP	src port	dest IP	dest port
socket	1	129.132.19.1	54001	172.217.168.3	443
	2	129.132.19.1	55240	172.217.168.3	443
	3	129.132.19.1	48472	172.217.168.3	443
	4	129.132.19.1	35456	172.217.168.3	443
	5	129.132.19.1	42001	172.217.168.3	443

Server OS		src IP	src port	dest IP	dest port
socket	1	172.217.168.3	443	129.132.19.1	54001
	2	172.217.168.3	443	129.132.19.1	55240
	3	172.217.168.3	443	129.132.19.1	48472
	4	172.217.168.3	443	129.132.19.1	35456
	5	172.217.168.3	443	129.132.19.1	42001

The life of a TCP connection is a sequence of states, described with a Finite State Machine



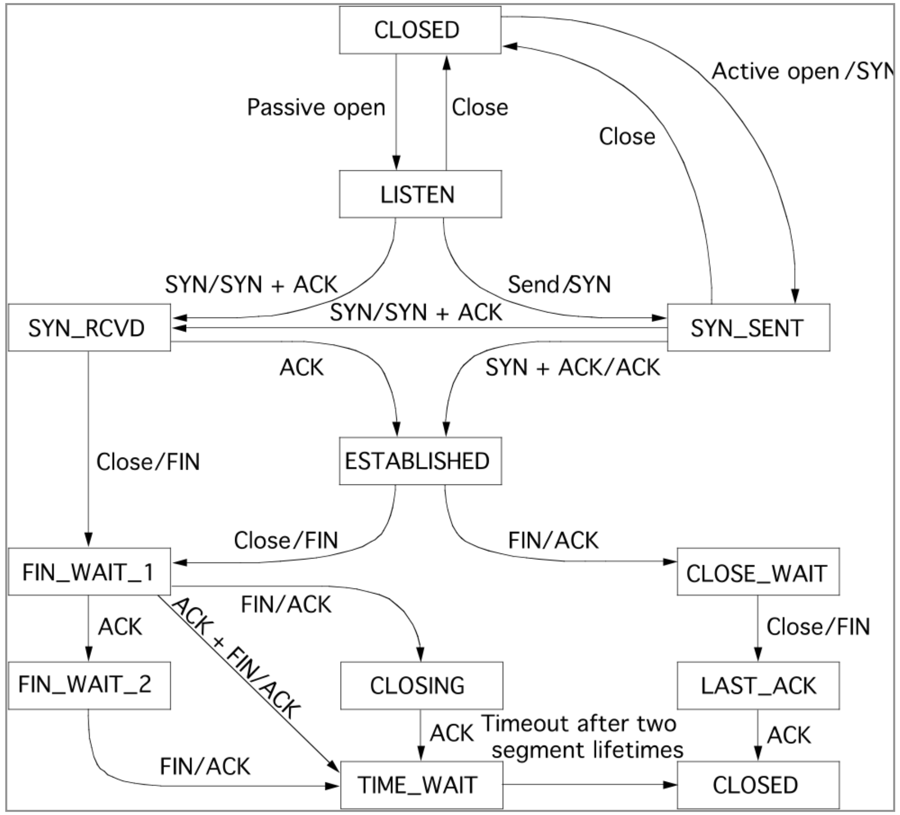
TCP connections start/end in the CLOSED state

Most of states relate to

- the connection establishment (three-way handshake)
- the connection termination (ensuring reliability)

Data is exchanged in the ESTABLISHED state

The TCP connection moves from one state to another in response of events (timeouts, "flagged" segments, …)



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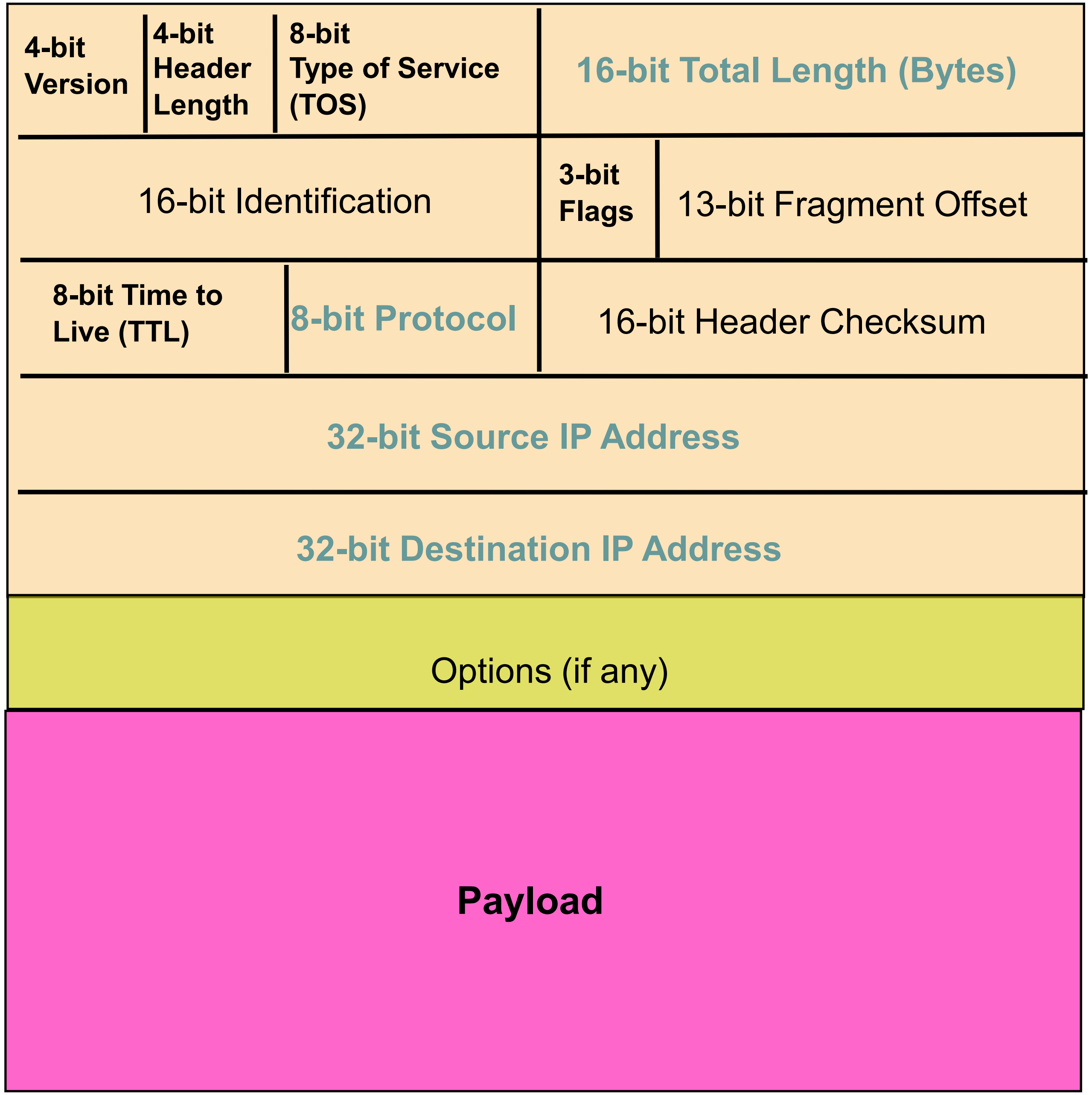
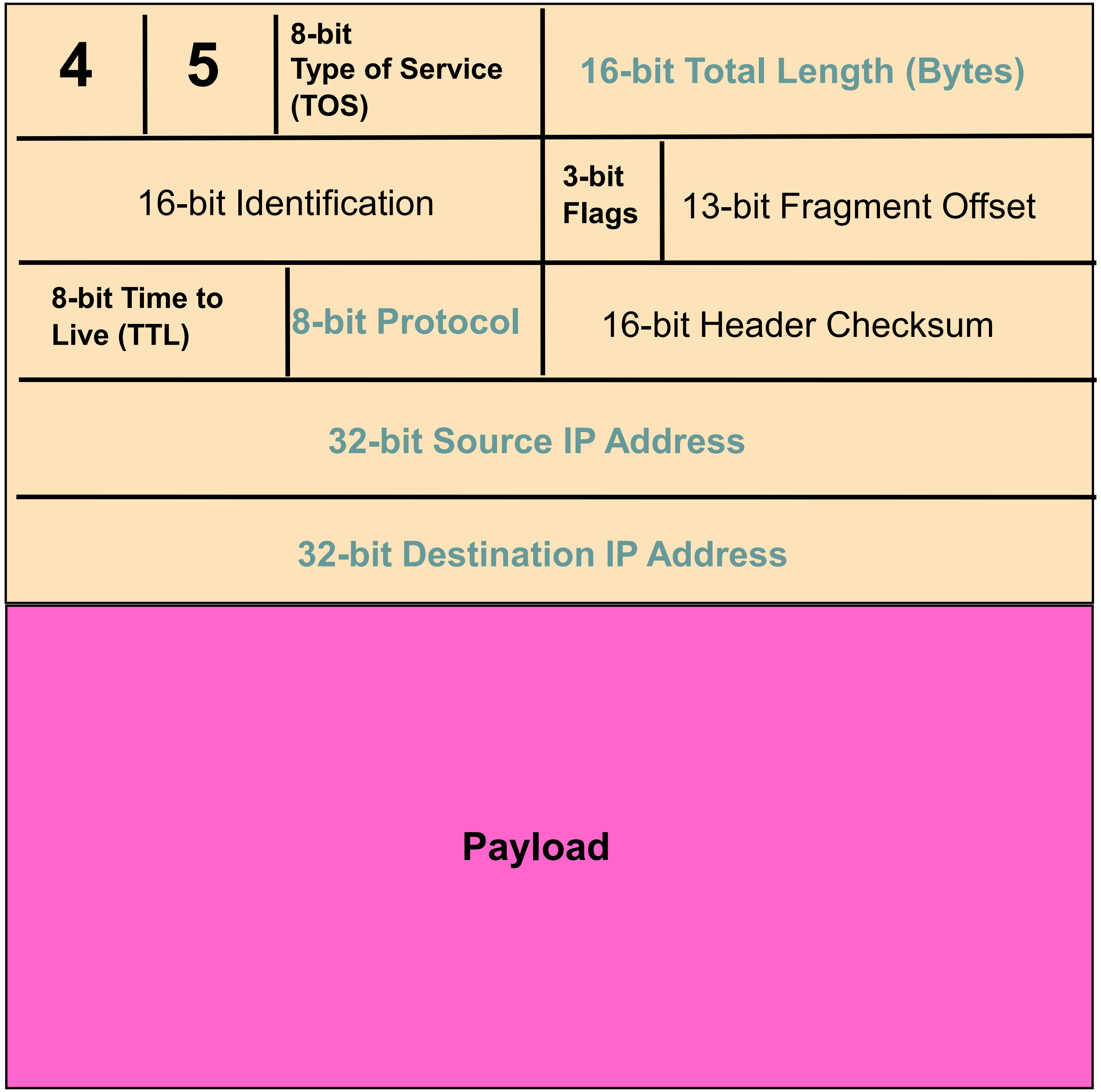

4-bit Version
4-bit Header Length
8-bit Type of Service (TOS)
16-bit Total Length (Bytes)
16-bit Identification
3-bit Flags
13-bit Fragment Offset
8-bit Time to Live (TTL)
8-bit Protocol
16-bit Header Checksum
32-bit Source IP Address
32-bit Destination IP Address
Options (if any)
Payload


4
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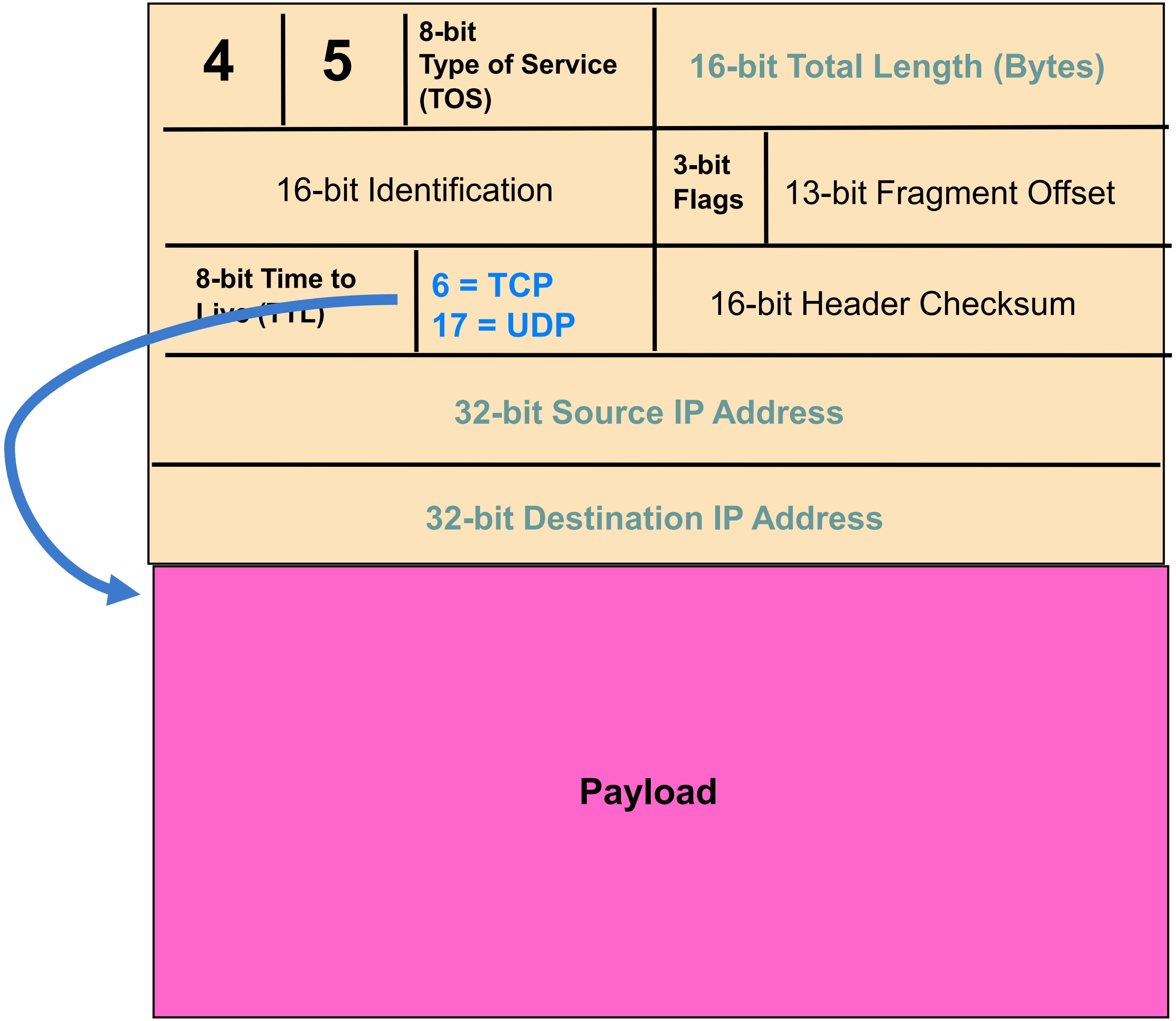

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8-bit Type of Service (TOS)
16-bit Total Length (Bytes)
16-bit Identification
3-bit Flags
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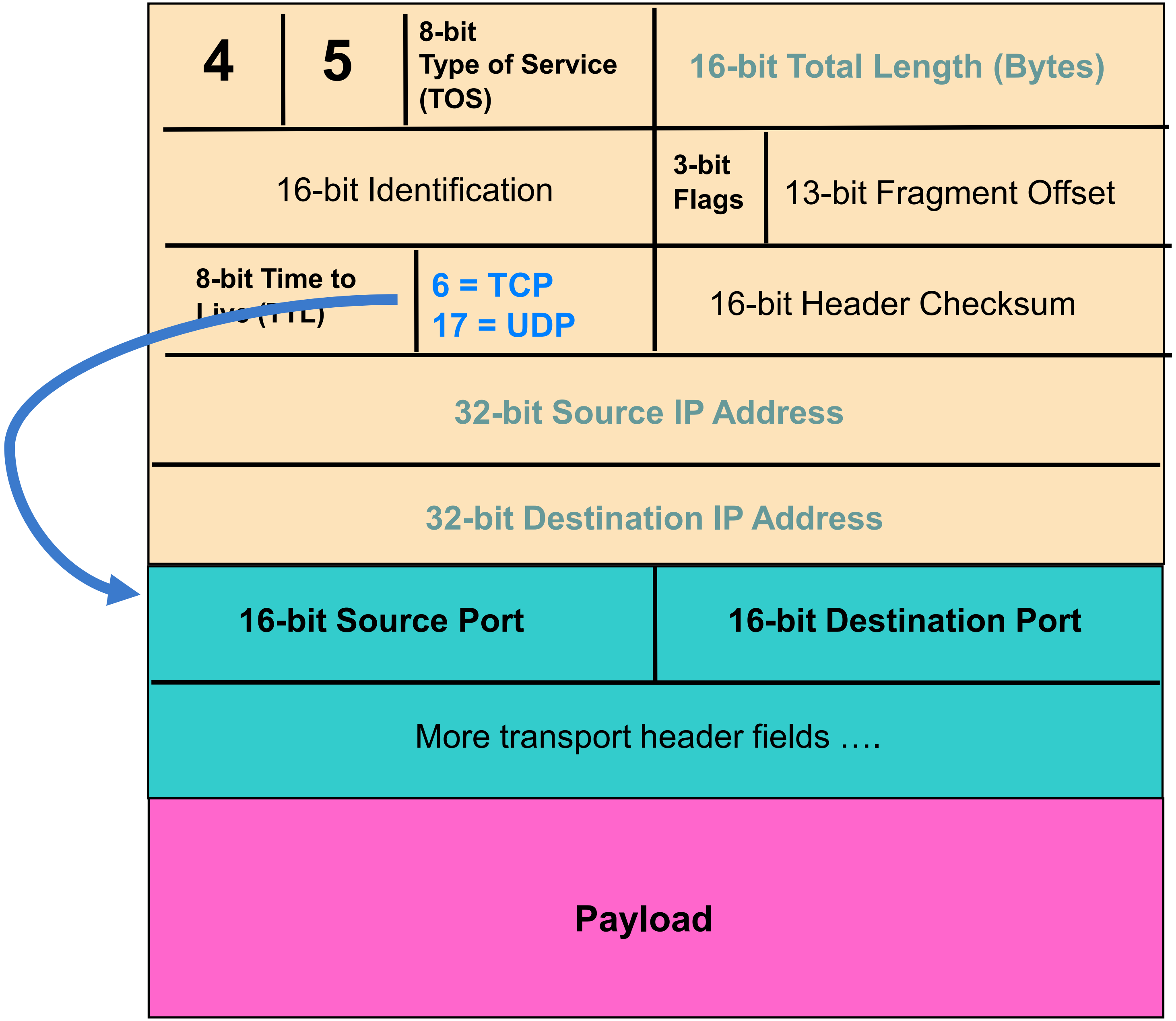
6 = TCP
17 = UDP
16-bit Header Checksum
32-bit Source IP Address
32-bit Destination IP Address
Payload


4
5
8-bit Type of Service (TOS)
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16-bit Identification
3-bit Flags
13-bit Fragment Offset
8-bit Time to Live (TTL)
6 = TCP
17 = UDP
16-bit Header Checksum
32-bit Source IP Address
32-bit Destination IP Address
16-bit Source Port
16-bit Destination Port
More transport header fields ….
Payload

UDP

UDP: User Datagram Protocol

- Lightweight communication between processes
  - Avoid overhead and delays of ordered, reliable delivery
  - Send messages to and receive them from a socket

- UDP described in RFC 768 – (1980!)
  - IP plus port numbers to support (de)multiplexing
  - Optional error checking on the packet contents
    - (checksum field = 0 means “don’t verify checksum”)

SRC port	DST port
checksum	length
DATA	

Why Would Anyone Use UDP?

- Finer control over what data is sent and when
  - As soon as an application process writes into the socket
  - … UDP will package the data and send the packet

- No delay for connection establishment
  - UDP just blasts away without any formal preliminaries
  - … which avoids introducing any unnecessary delays

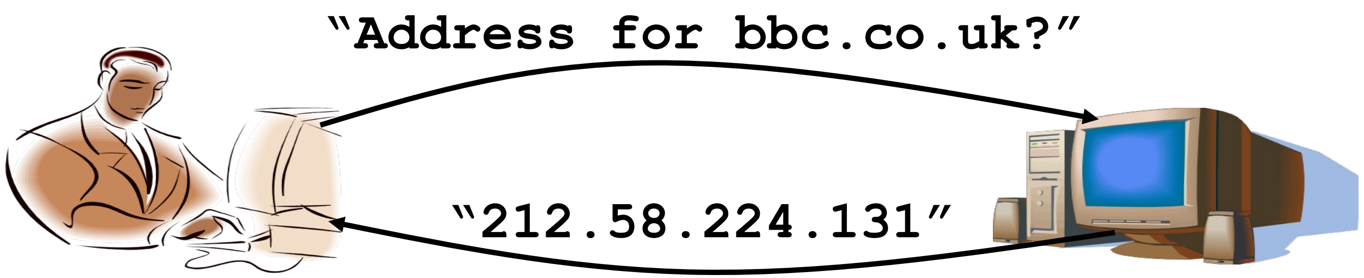
- No connection state
  - No allocation of buffers, sequence #s, timers …
  - … making it easier to handle many active clients at once

- Small packet header overhead
  - UDP header is only 8 bytes

Popular Applications That Use UDP

- Some **interactive streaming** apps
  - Retransmitting lost/corrupted packets often pointless: by the time the packet is retransmitted, it's too late
    - telephone calls, video conferencing, gaming…
    - ***Modern streaming protocols using TCP (and HTTP)***

- Simple query protocols like Domain Name System (DNS)
  - Connection establishment overhead would double cost
  - Easier to have **application** retransmit if needed



TCP

Transmission Control Protocol (TCP)

- Reliable, in-order delivery *(previously, but quick review)*
  - Ensures byte stream (eventually) arrives intact
    - In the presence of **corruption** and **loss**

- Connection oriented *(today)*
  - Explicit set-up and tear-down of TCP session

- Full duplex stream-of-bytes service *(today)*
  - Sends and receives a stream of bytes, not messages

- Flow control *(previously, but quick review)*
  - Ensures that sender doesn't overwhelm receiver

- Congestion control *(next week)*
  - Dynamic adaptation to network path's capacity

Basic Components of Reliability

- ACKs
  - Can’t be reliable without knowing whether data has arrived
  - ***TCP uses byte sequence numbers to identify payloads***

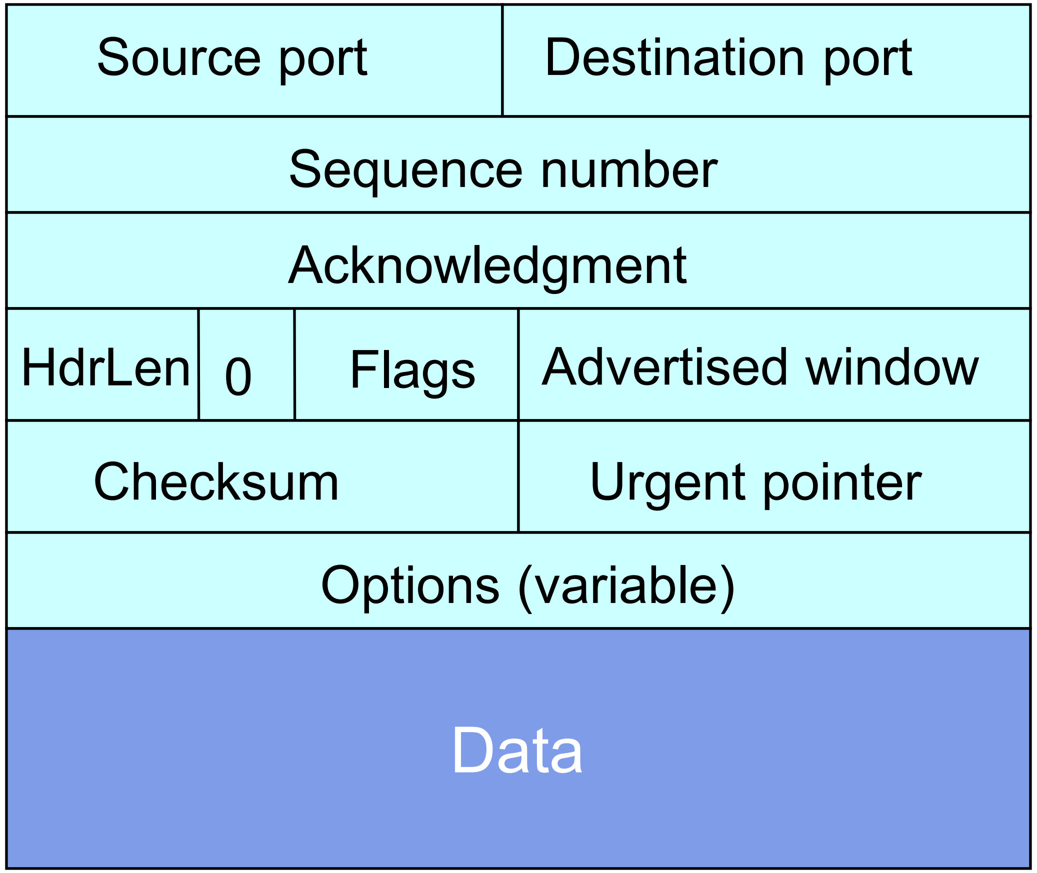
- Checksums
  - Can’t be reliable without knowing whether data is corrupted
  - ***TCP does checksum over TCP and pseudoheader***

- Timeouts and retransmissions
  - Can’t be reliable without retransmitting lost/corrupted data
  - ***TCP retransmits based on timeouts and duplicate ACKs***
  - *Timeout based on estimate of RTT*

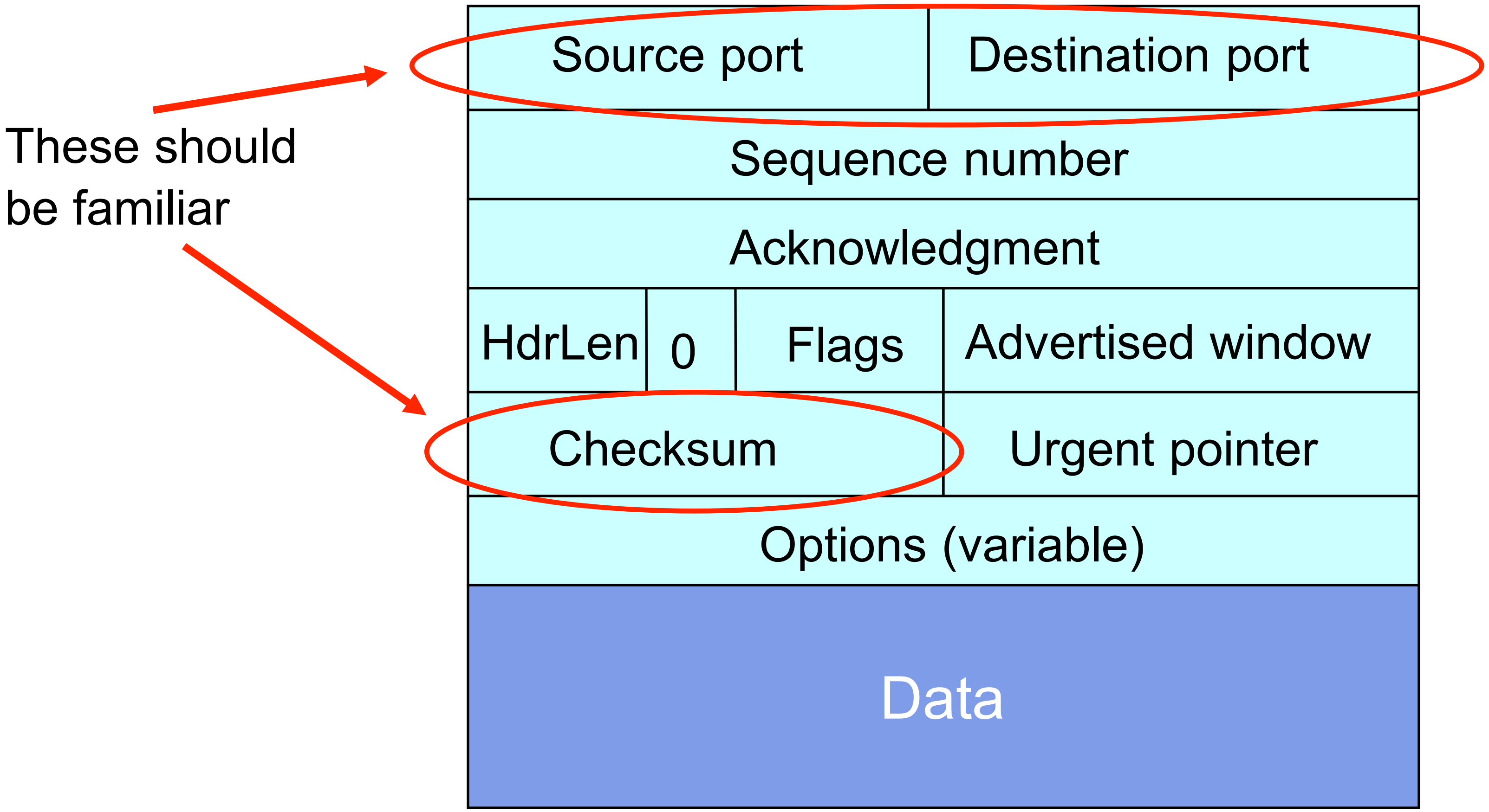
Other TCP Design Decisions

- Sliding window flow control
  - Allow W contiguous bytes to be in flight
- Cumulative acknowledgements
  - Selective ACKs (full information) also supported (ignore)
- Single timer set after each payload is ACKed
  - Timer is effectively for the “next expected payload”
  - When timer goes off, resend that payload and wait
  - And double timeout period
- Various tricks related to “fast retransmit”
  - Using duplicate ACKs to trigger retransmission

TCP Header

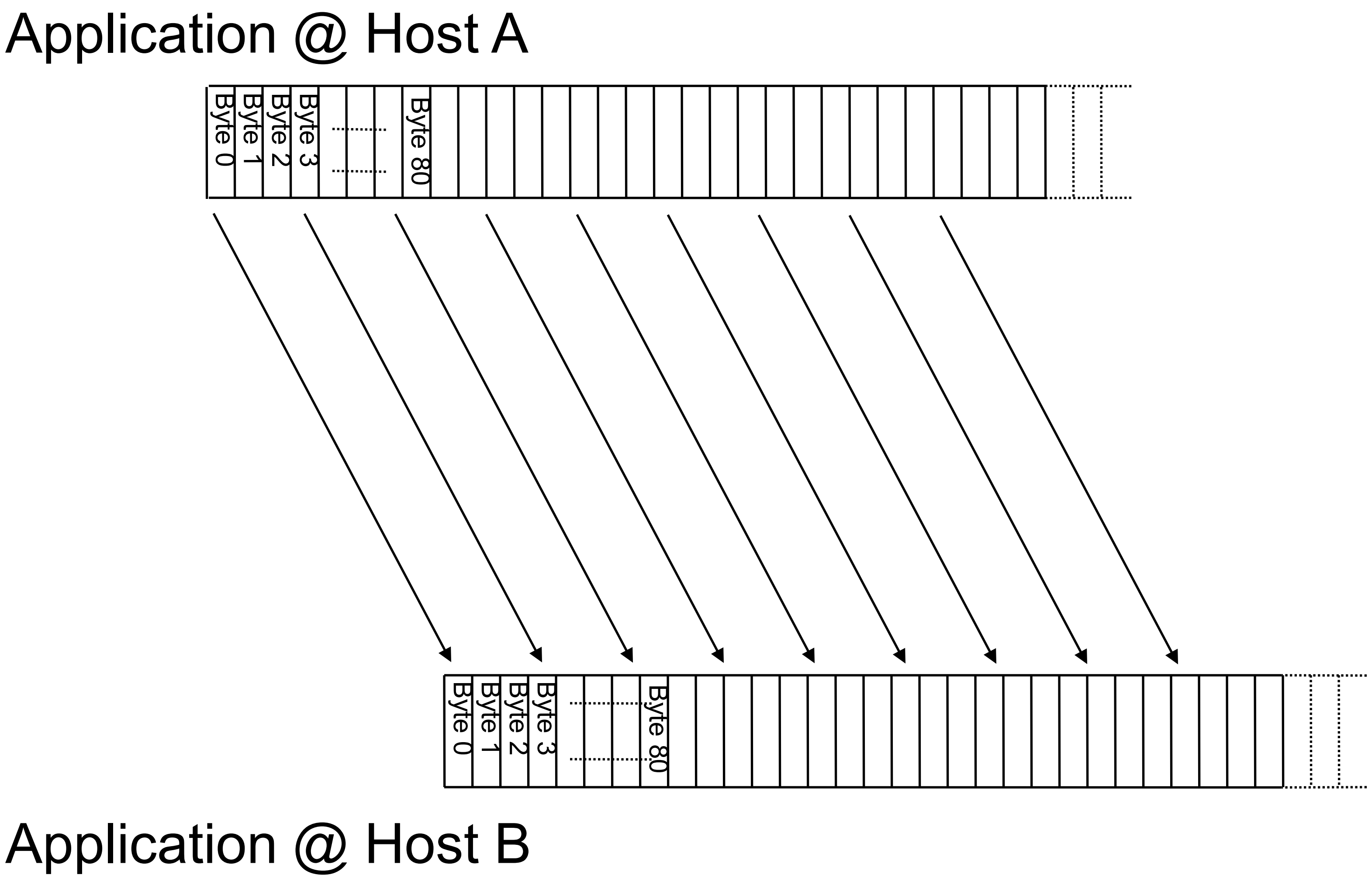

Source port
Destination port
Sequence number
Acknowledgment
HdrLen
0
Flags
Advertised window
Checksum
Urgent pointer
Options (variable)
Data

TCP Header

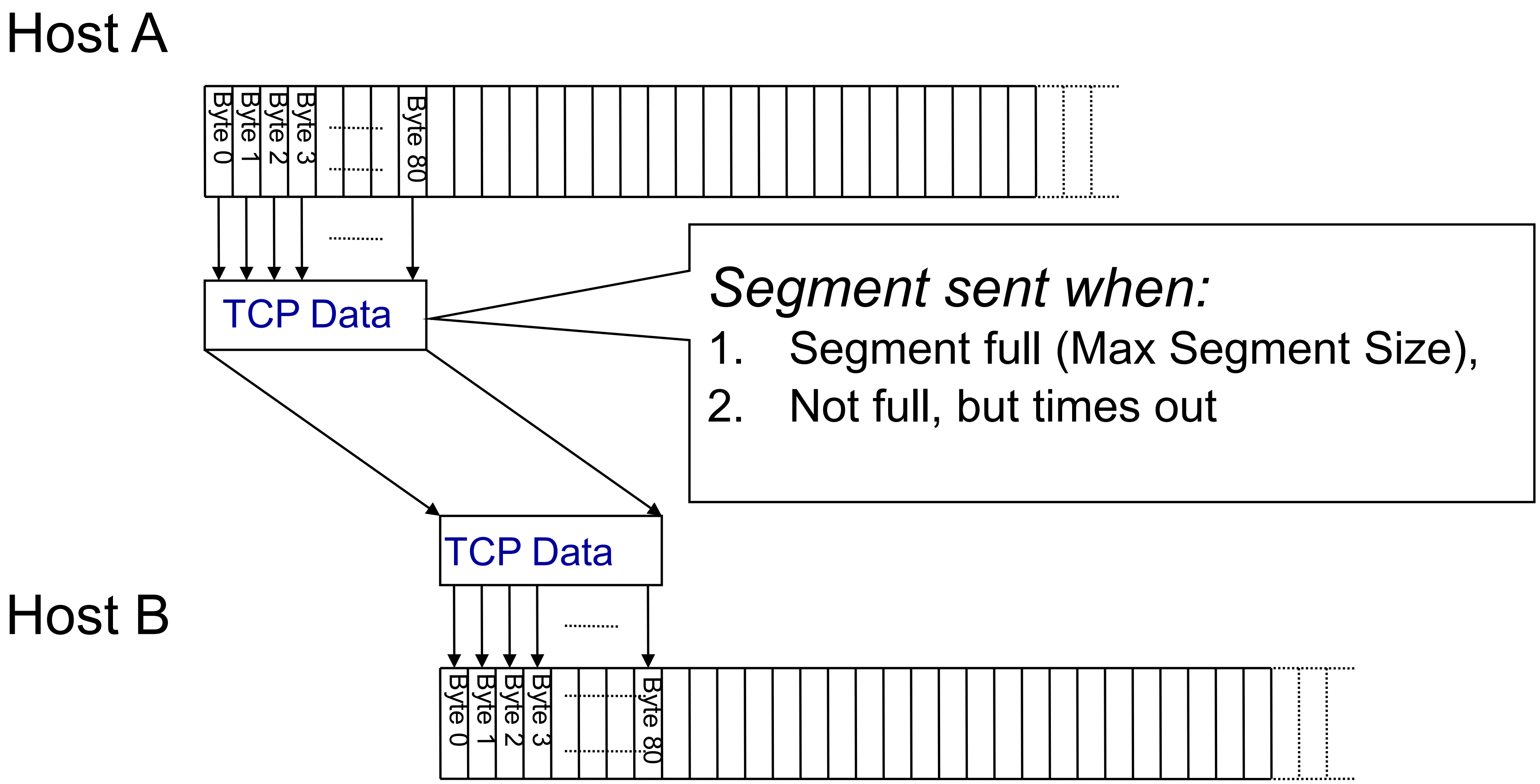

These should be familiar
Source port
Destination port
Sequence number
Acknowledgment
HdrLen
0
Flags
Advertised window
Checksum
Urgent pointer
Options (variable)
Data

Segments and Sequence Numbers

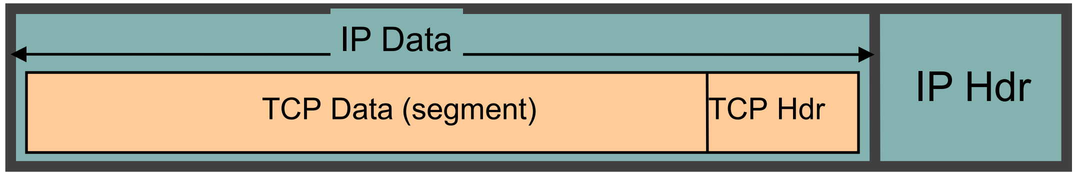
TCP "Stream of Bytes" Service…



… Provided Using TCP "Segments"

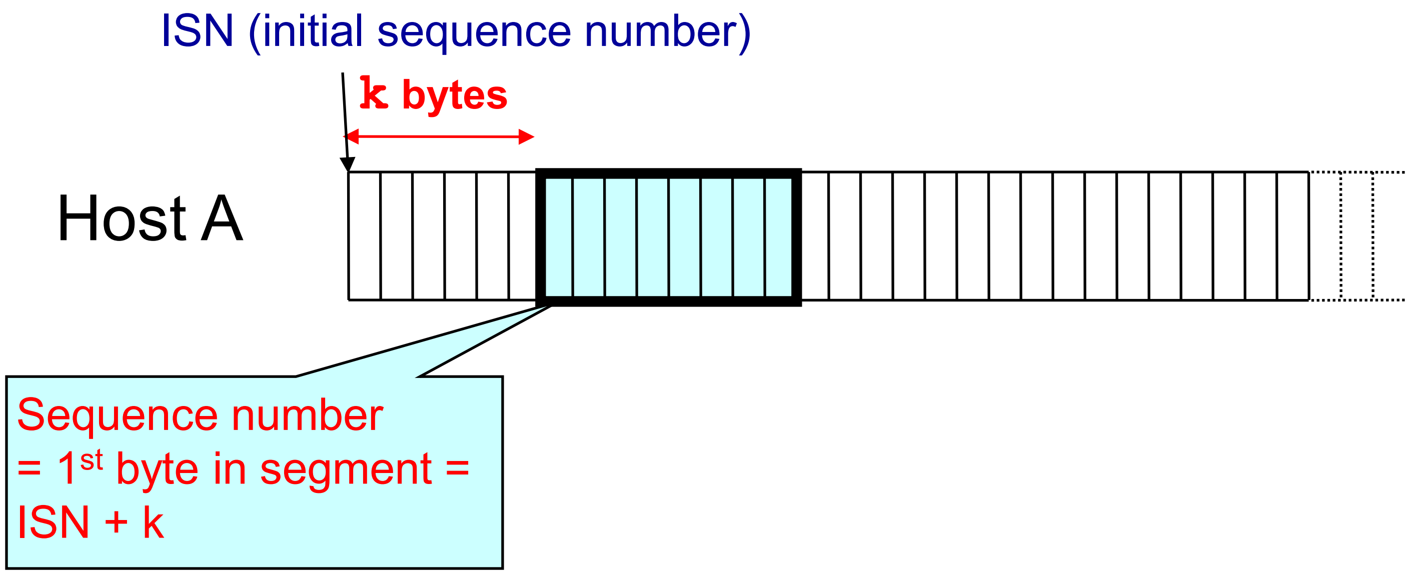


TCP Segment

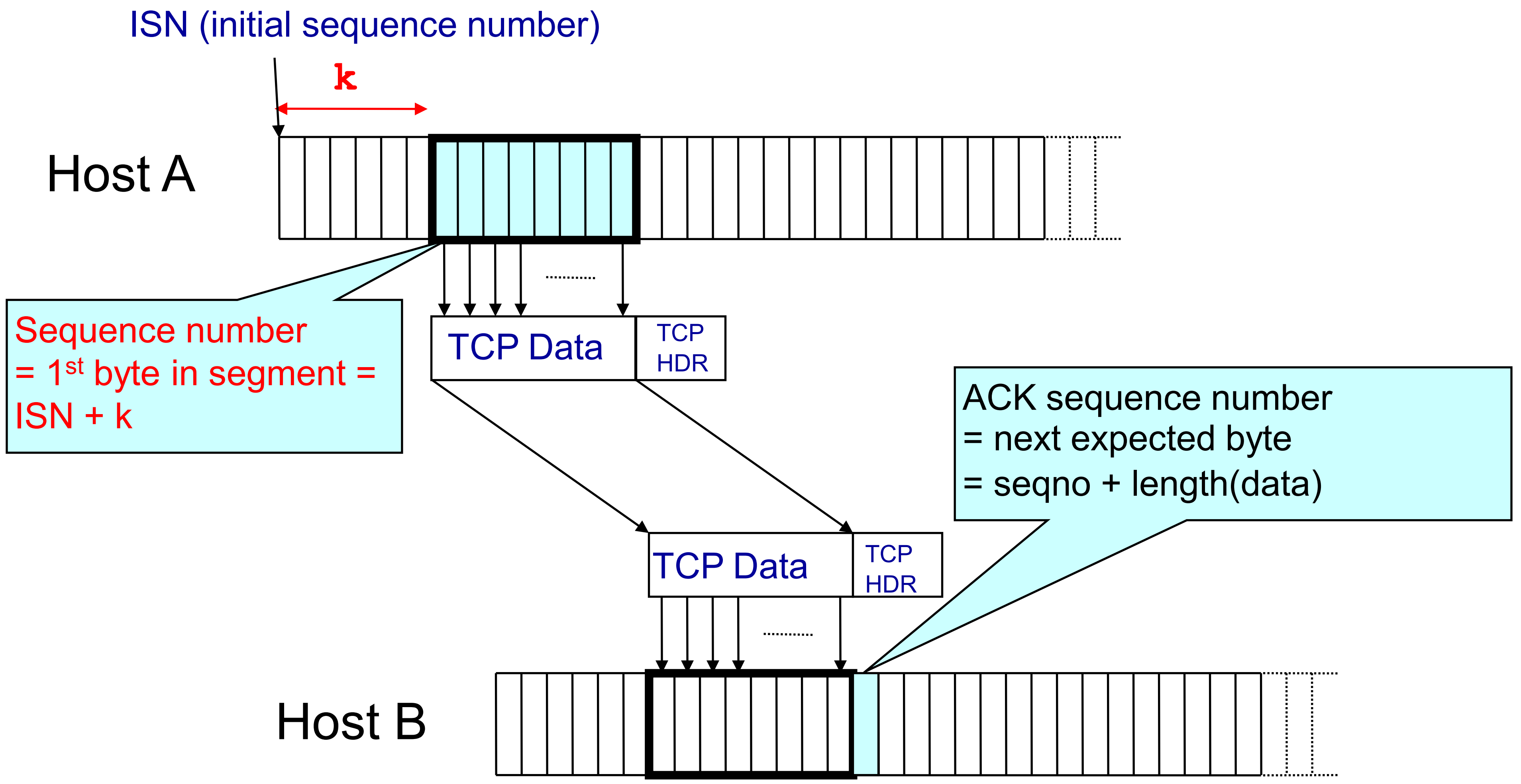


- IP **packet**
  - No bigger than Maximum Transmission Unit (MTU)
  - E.g., up to 1500 bytes with Ethernet
- TCP **packet**
  - IP packet with a TCP header and data inside
  - TCP header ≥ 20 bytes long
- TCP **segment**
  - No more than Maximum Segment Size (MSS) bytes
  - E.g., up to 1460 consecutive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

Sequence Numbers



Sequence Numbers



ACKing 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 contiguous byte received is smaller value Y
    - ACK acknowledges Y+1
    - Even if this has been ACKed before

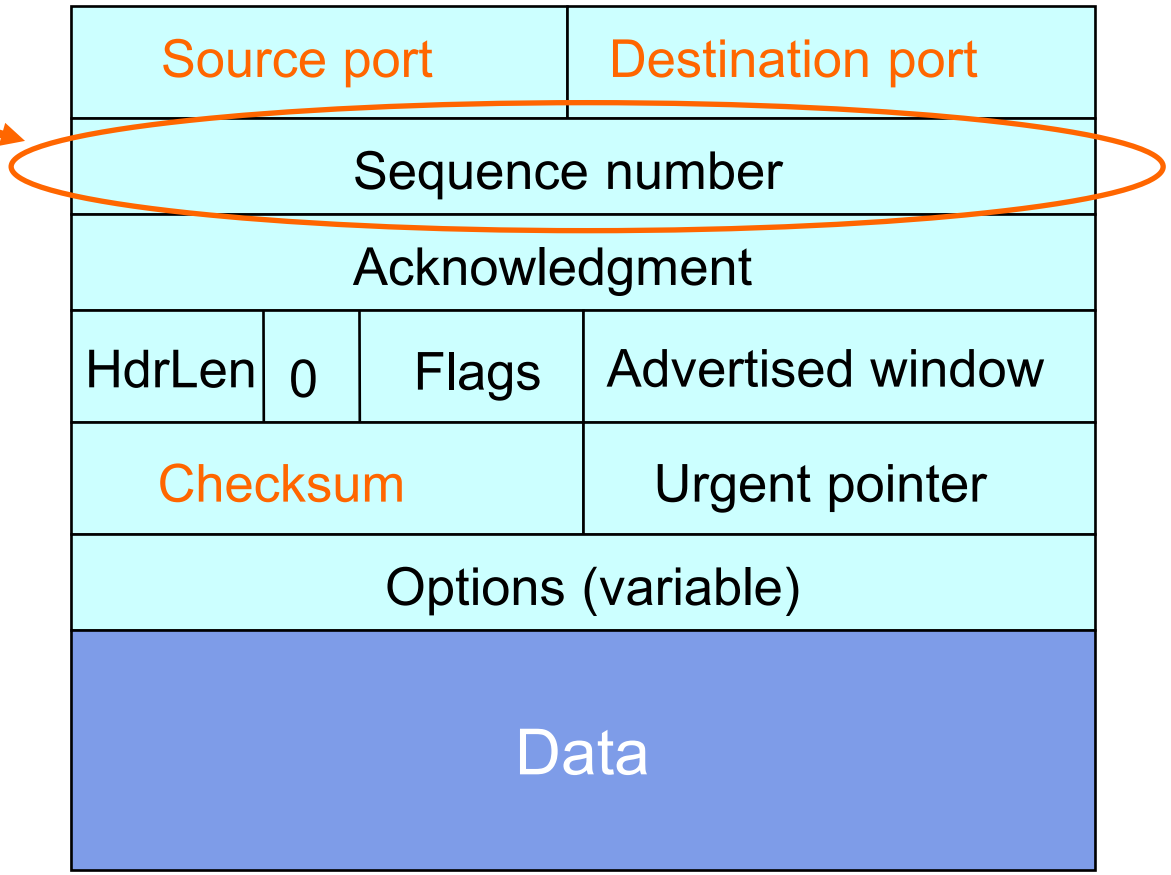
Normal Pattern

- 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

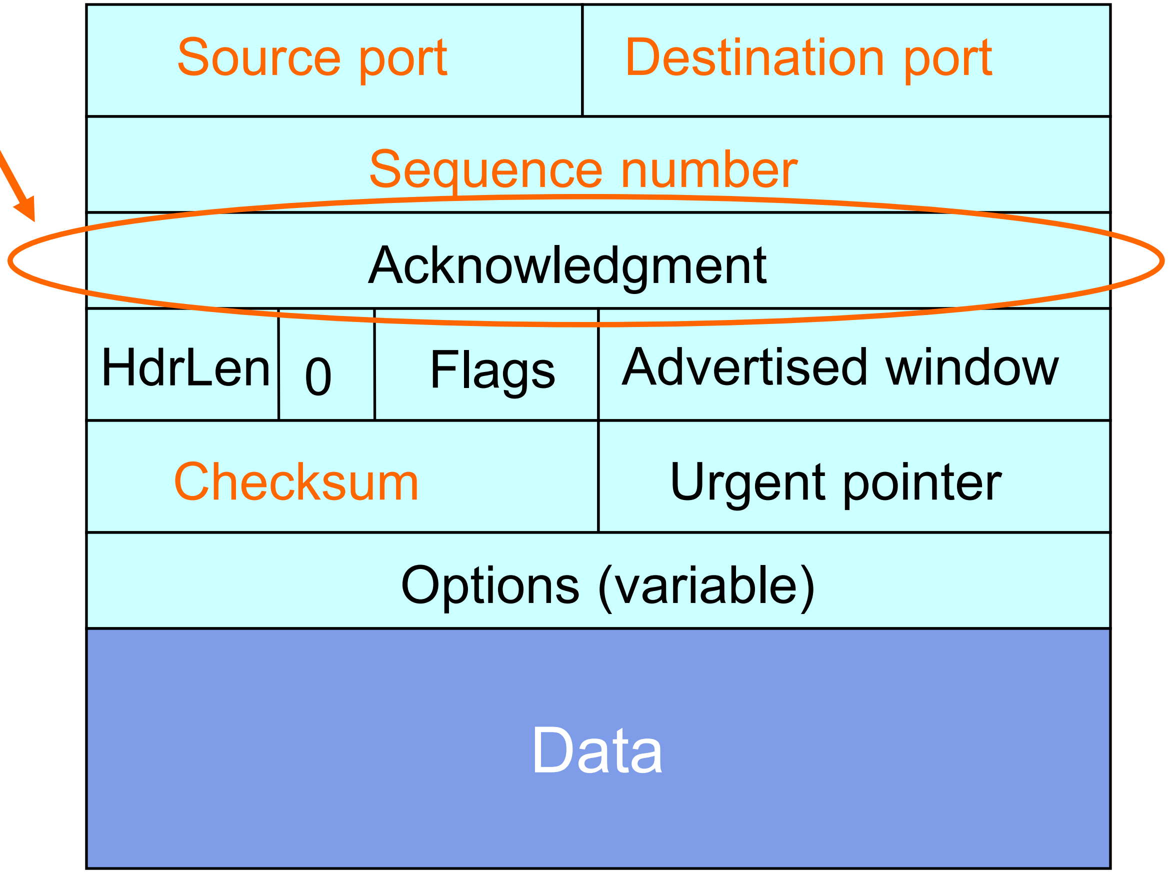
Starting byte offset of data carried in this segment



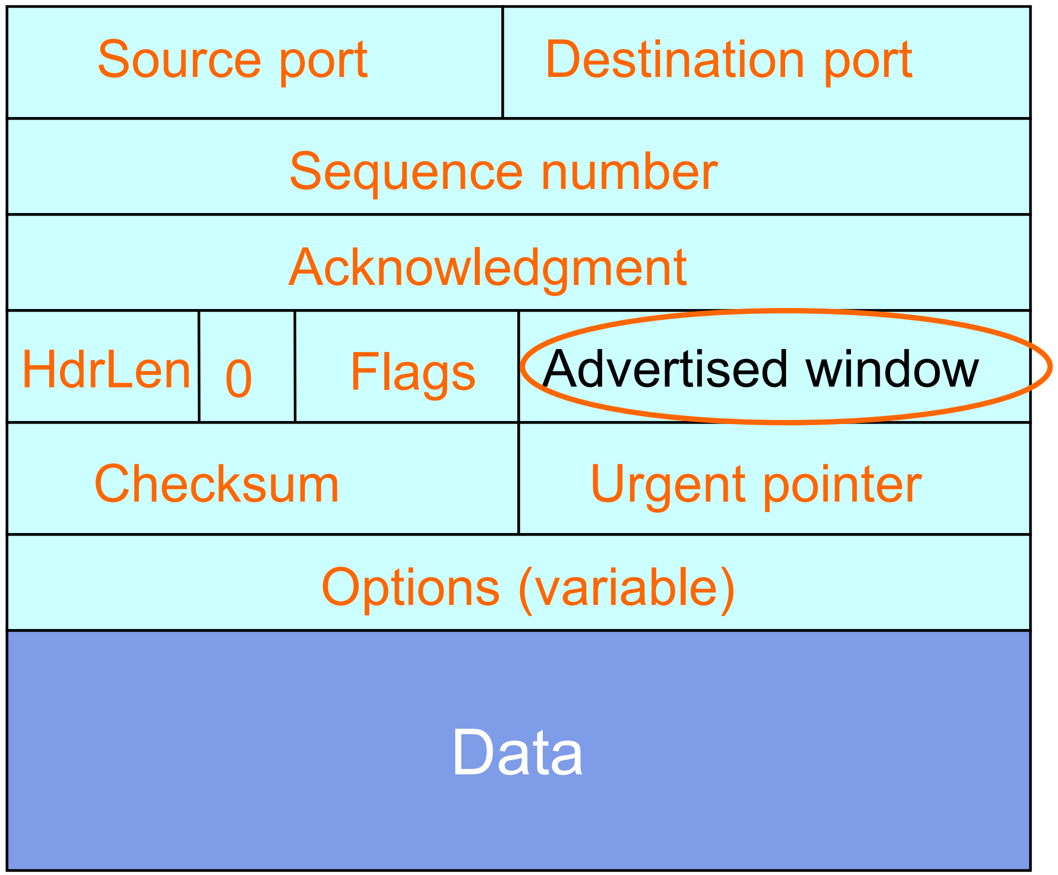
TCP Header

Acknowledgment gives seqno just beyond highest seqno received in order

"What Byte is Next"



TCP Header


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

Sliding Window Flow Control

- Advertised Window: W
  - Can send W bytes beyond the next expected byte
- Receiver uses W to prevent sender from overflowing buffer
- Limits number of bytes sender can have in flight

Implementing Sliding Window

- Both sender & receiver maintain a **window**
  - Sender: not yet ACK'ed
  - Receiver: not yet delivered to application

- Left edge of window:
  - Sender: beginning of unacknowledged data
  - Receiver: beginning of undelivered data

- For the sender:
  - Window size = maximum amount of data in flight

- For the receiver:
  - Window size = maximum amount of undelivered data

Sliding Window Summary

- Sender: window advances when new data ack'd
- Receiver: window advances as receiving process consumes data
- Receiver advertises to the sender where the receiver window currently ends ("righthand edge")
    - Sender agrees not to exceed this amount
    - It makes sure by setting its own window size to a value that can't send beyond the receiver's righthand edge

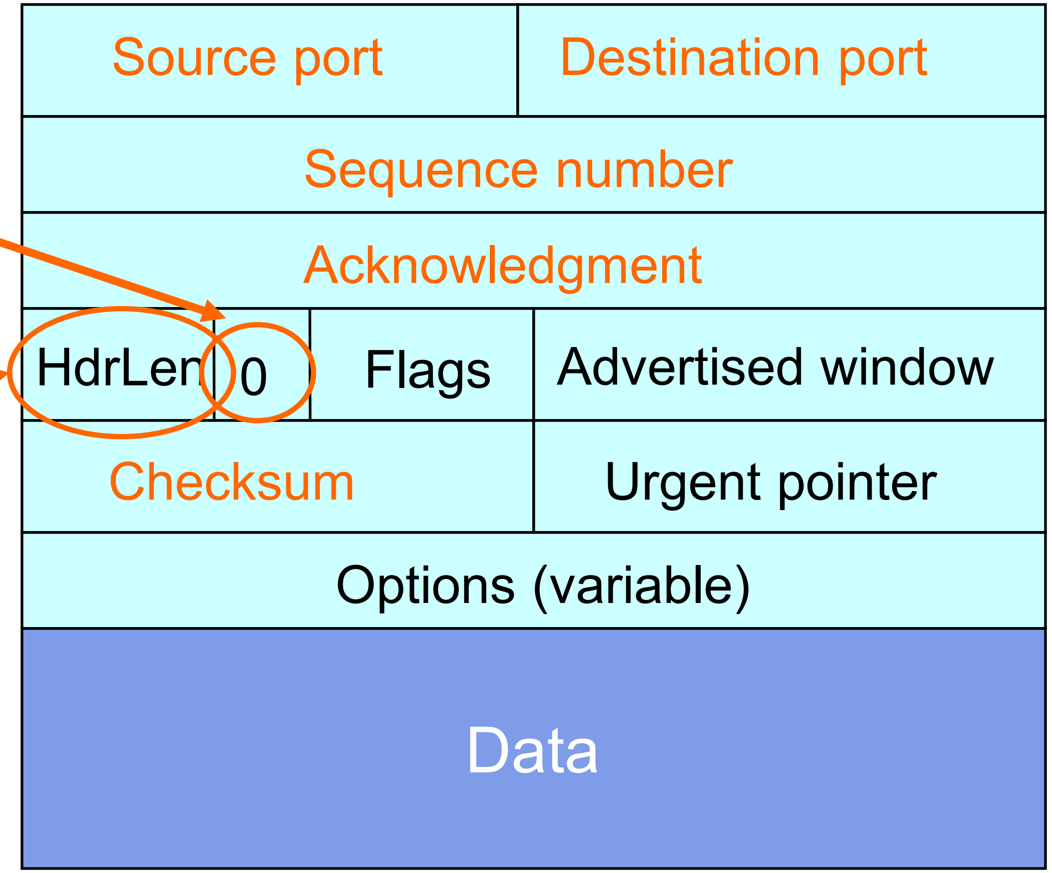
Advertised Window Limits Rate

- Sender can send no faster than W/RTT bytes/sec
- Receiver only advertises more space when it has consumed old arriving data
- In original TCP design, that was the **sole** protocol mechanism controlling sender's rate
- What's missing?

TCP Header: What's left?

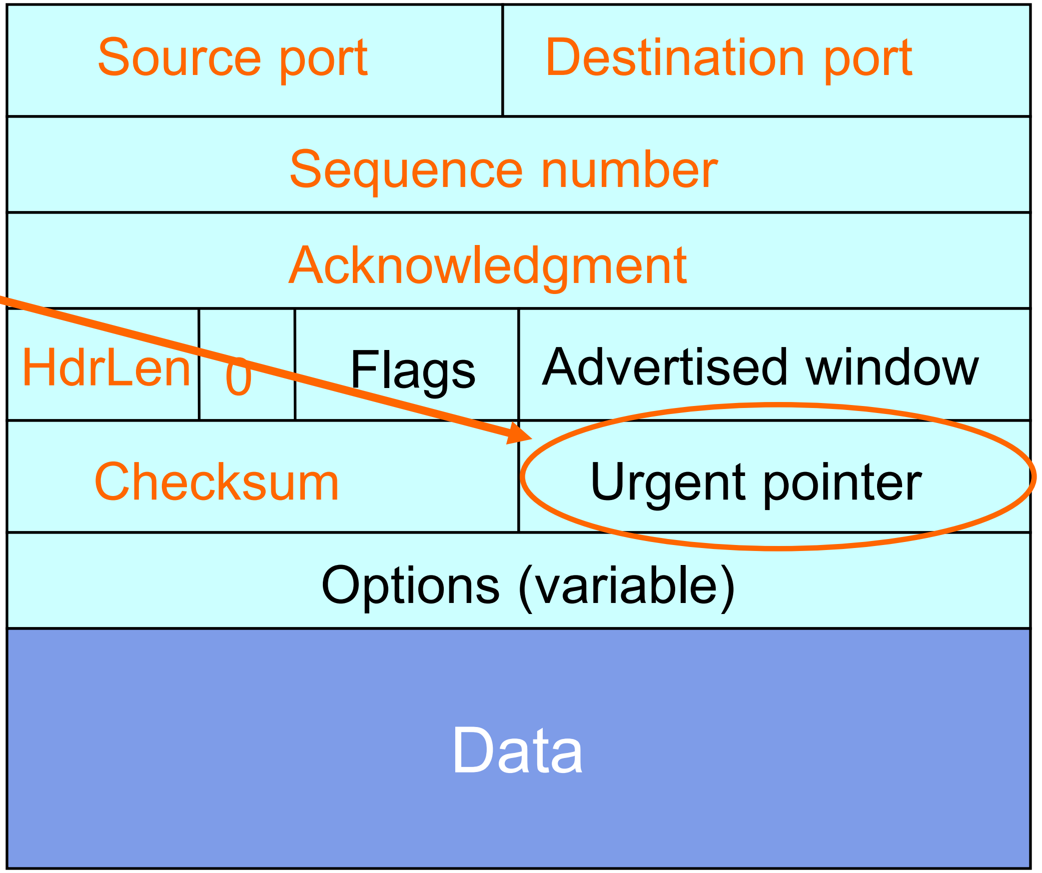
"Must Be Zero"
6 bits reserved

Number of 4-byte words in TCP header;
5 = no options

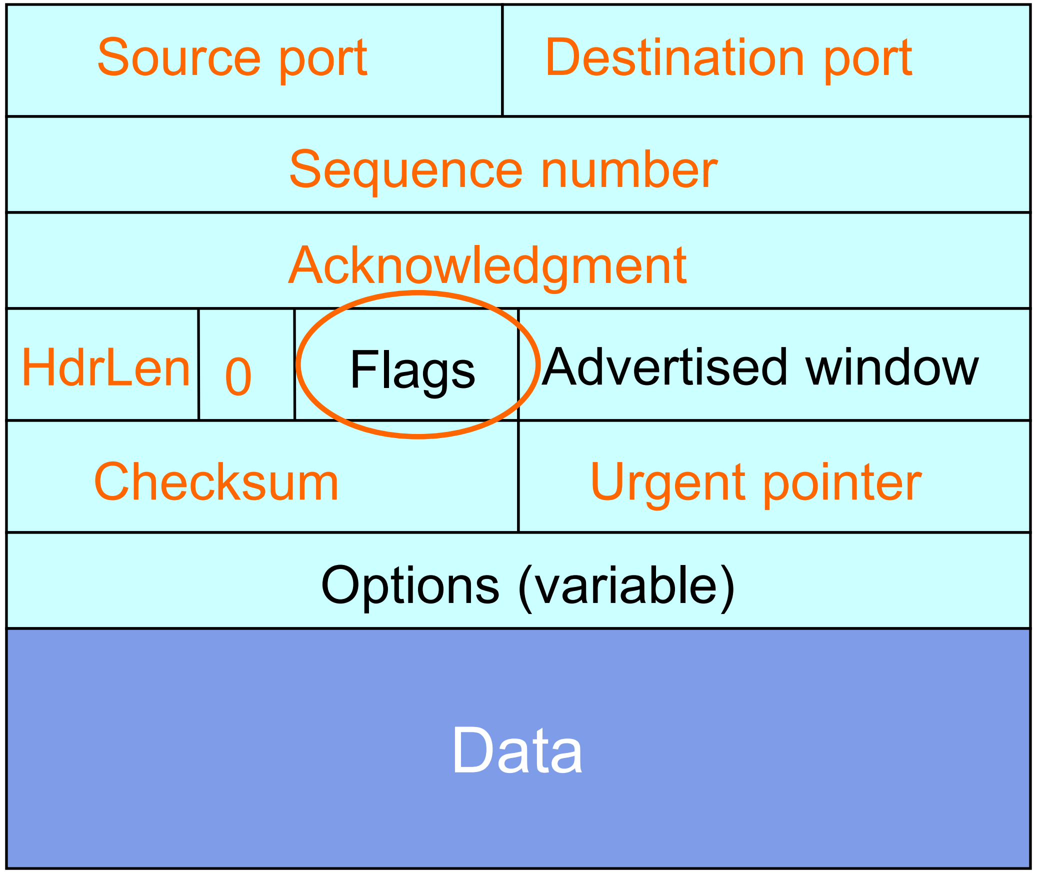


TCP Header: What's left?

Used with **URG** flag to indicate urgent data (not discussed further)



TCP Header: What's left?



TCP Connection Establishment and Initial Sequence Numbers

Initial Sequence Number (ISN)

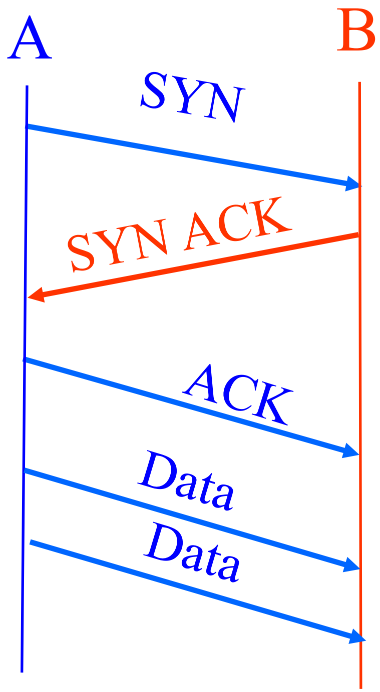
- Sequence number for the very first byte
  - E.g., Why not just use ISN = 0?

- 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

- TCP therefore requires changing ISN
  - initially set from 32-bit clock that ticks every 4 microseconds
  - now drawn from a pseudo random number generator (security)

- To establish a connection, hosts exchange ISNs
  - **How does this help?**

Establishing a TCP Connection

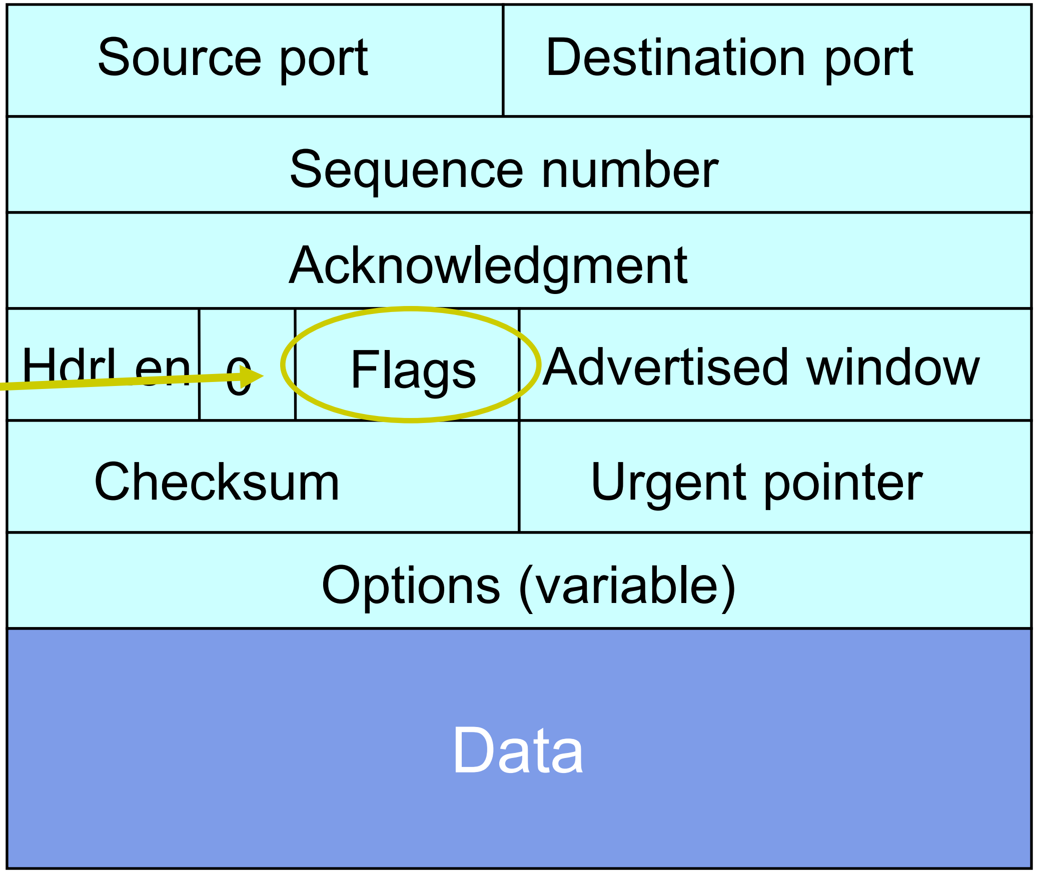


Each host tells its ISN to the other host.

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

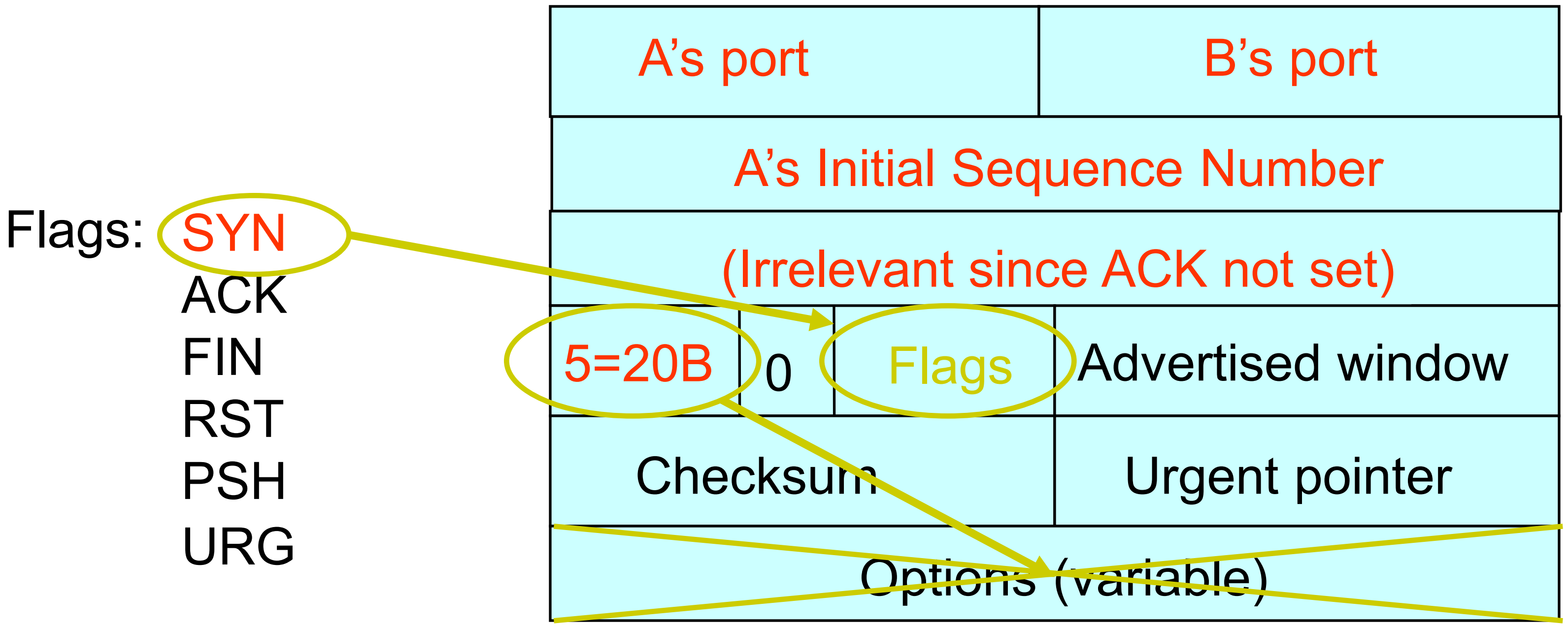
TCP Header

Flags: **SYN**
ACK
FIN
RST
PSH
URG



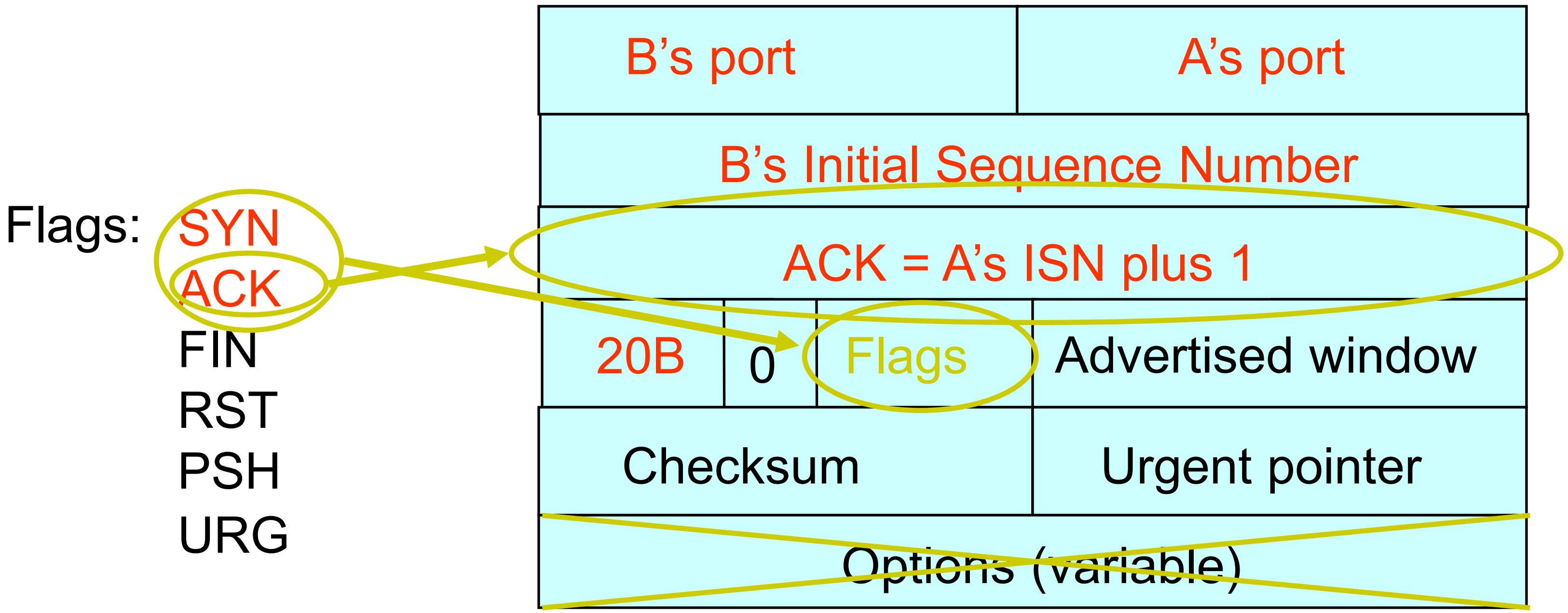
See /usr/include/netinet/tcp.h on Unix Systems

Step 1: A's Initial SYN Packet



A tells B it wants to open a connection…

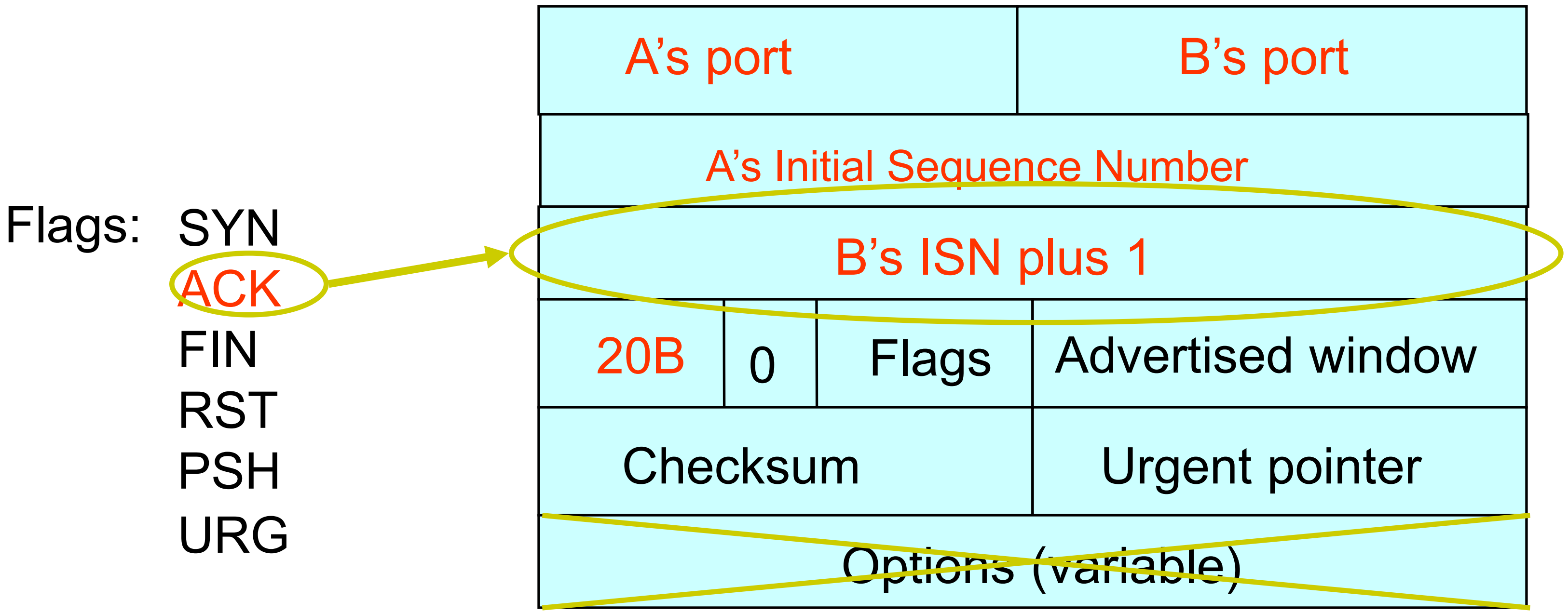
Step 2: B's SYN-ACK Packet



B tells A it accepts, and is ready to hear the next byte…

… upon receiving this packet, A can start sending data

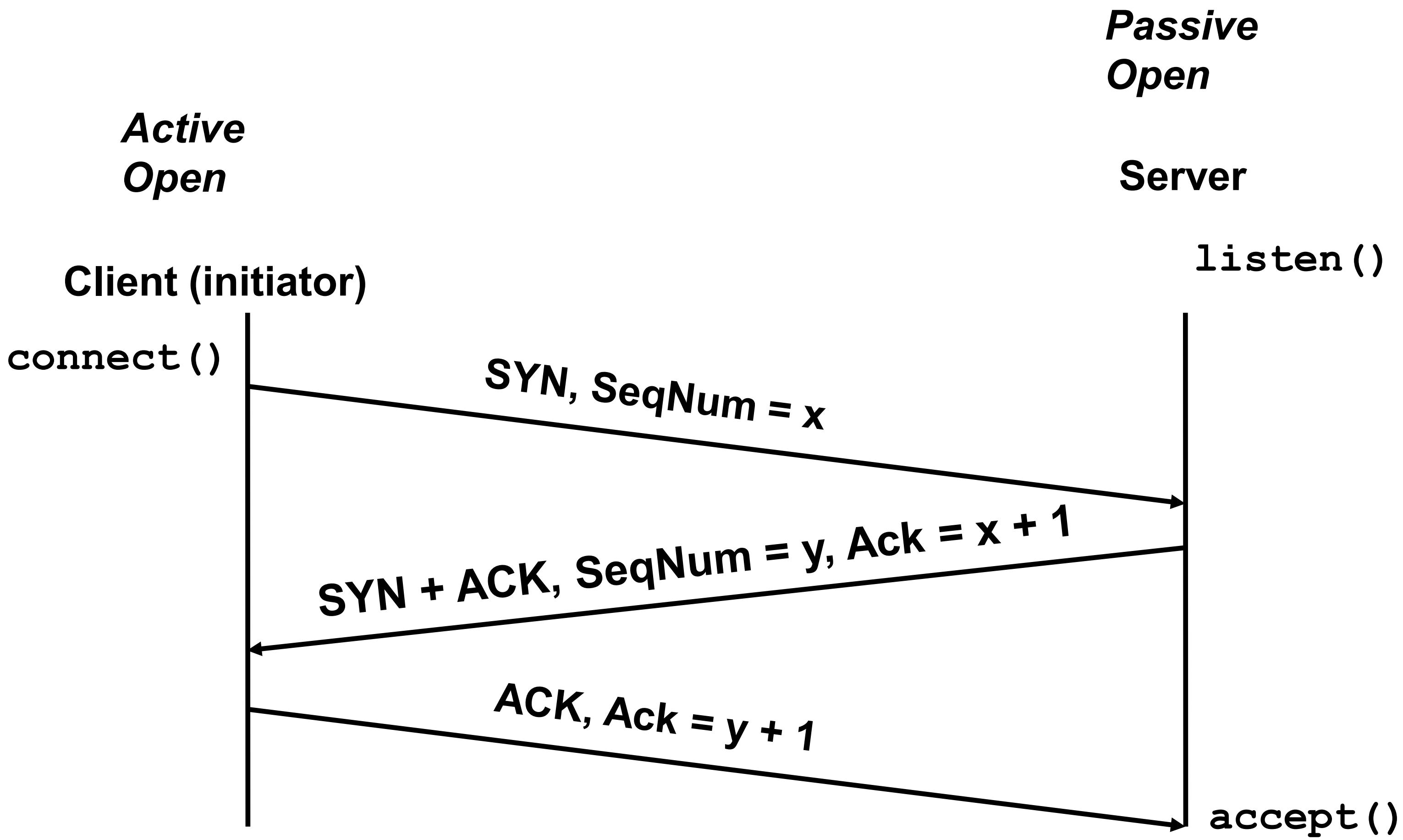
Step 3: A’s ACK of the SYN-ACK



A tells B it’s likewise okay to start sending

… upon receiving this packet, B can start sending data

Timing Diagram: 3-Way Handshaking



What if the SYN Packet Gets Lost?

- Suppose the SYN packet gets lost
  - Packet is lost inside the network, or:
  - Server discards the packet (e.g., listen queue is full)

- Eventually, no SYN-ACK arrives
  - Sender sets a timer and waits for the SYN-ACK
  - … and retransmits the SYN if needed

- 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
    - Other implementations instead use 6 seconds

SYN Loss and Web Downloads

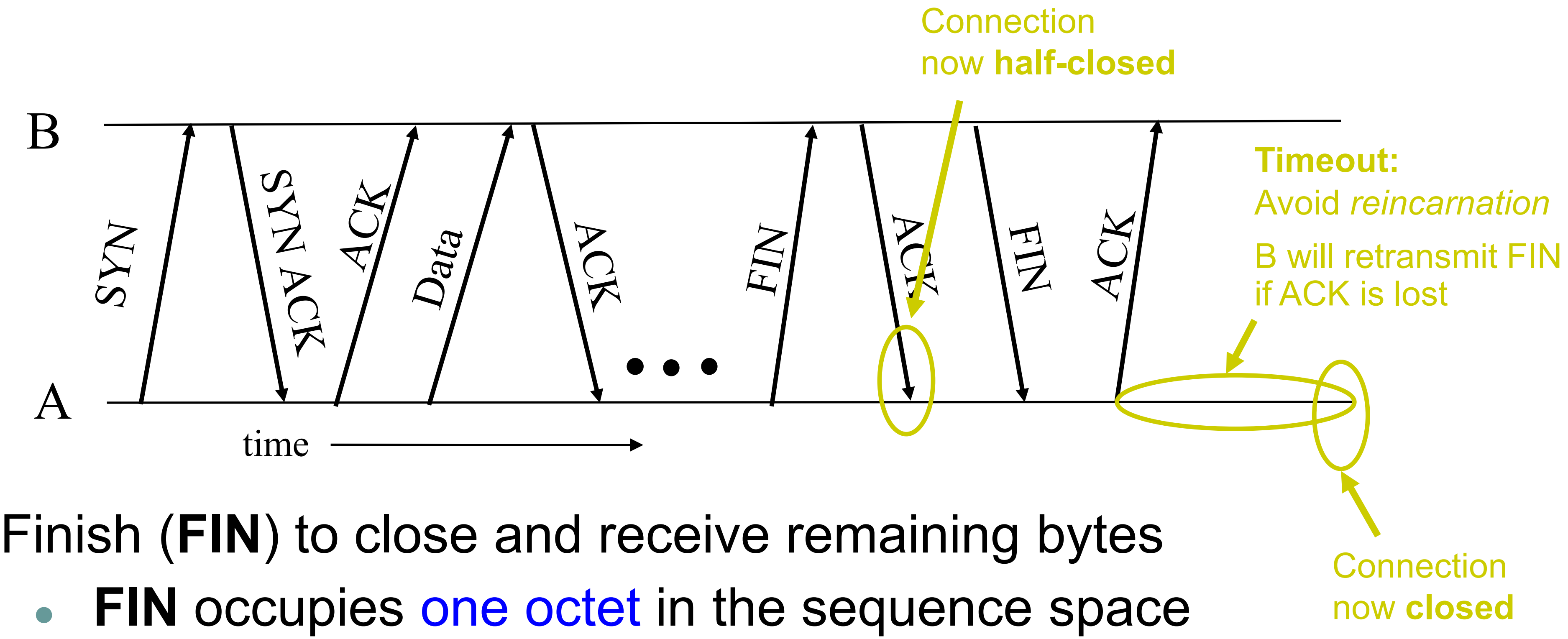
- User clicks on a hypertext link
  - Browser creates a socket and does a "connect"
  - The "connect" triggers the OS to transmit a SYN

- If the SYN is lost…
  - 3-6 seconds of delay: can be very long
  - User may become impatient
  - … and click the hyperlink again, or click "reload"

- User triggers an "abort" of the "connect"
  - Browser creates a new socket and another "connect"
  - Essentially, forces a faster send of a new SYN packet!
  - Sometimes very effective, and the page comes quickly

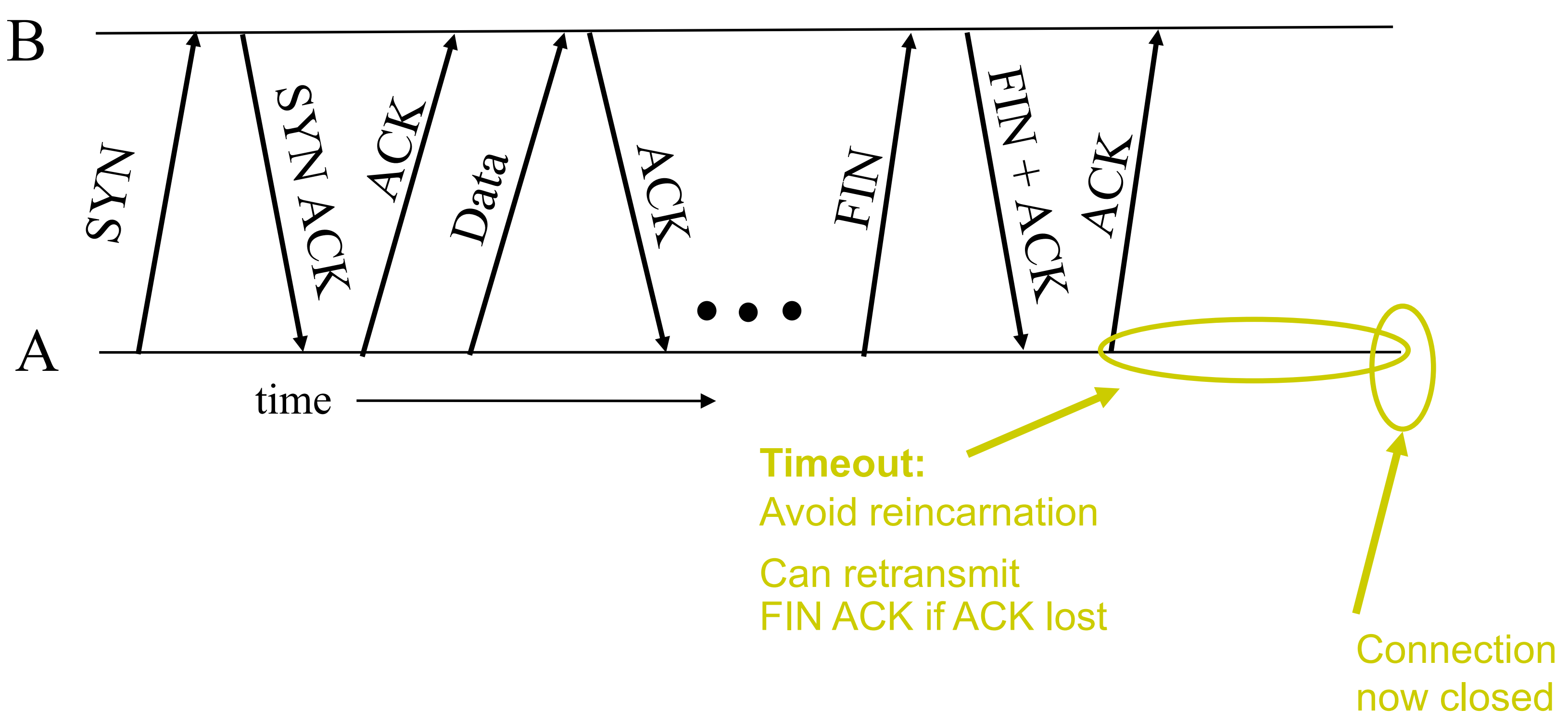
Tearing Down the Connection

Normal Termination, One Side At A Time



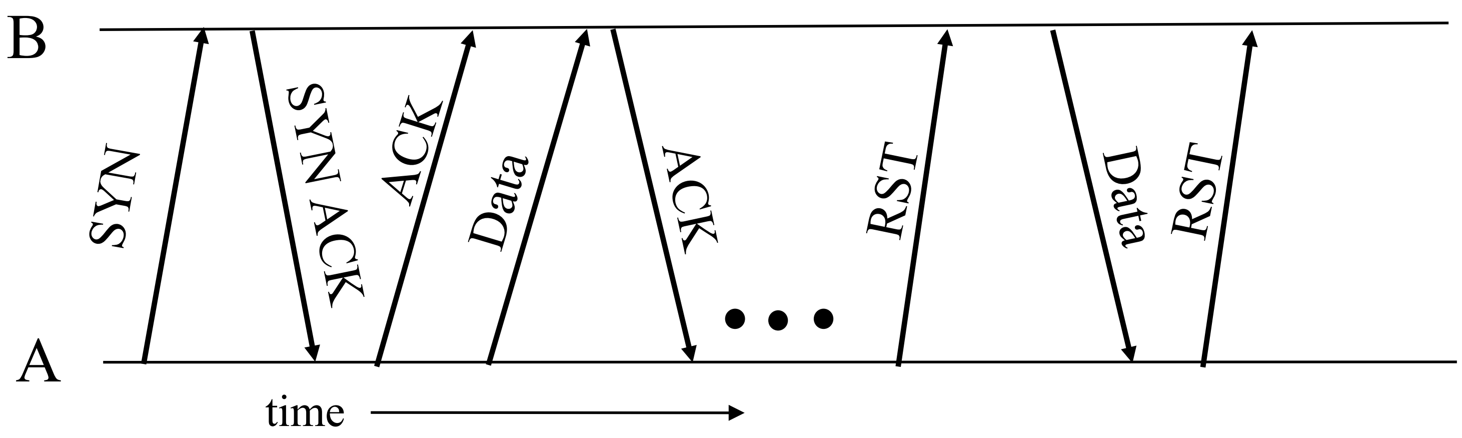
- Finish (**FIN**) to close and receive remaining bytes
  - **FIN** occupies one octet in the sequence space
- Other host ack's the octet to confirm
- Closes A's side of the connection, but not B's
  - Until B likewise sends a **FIN**
  - Which A then acks

Normal Termination, Both Together



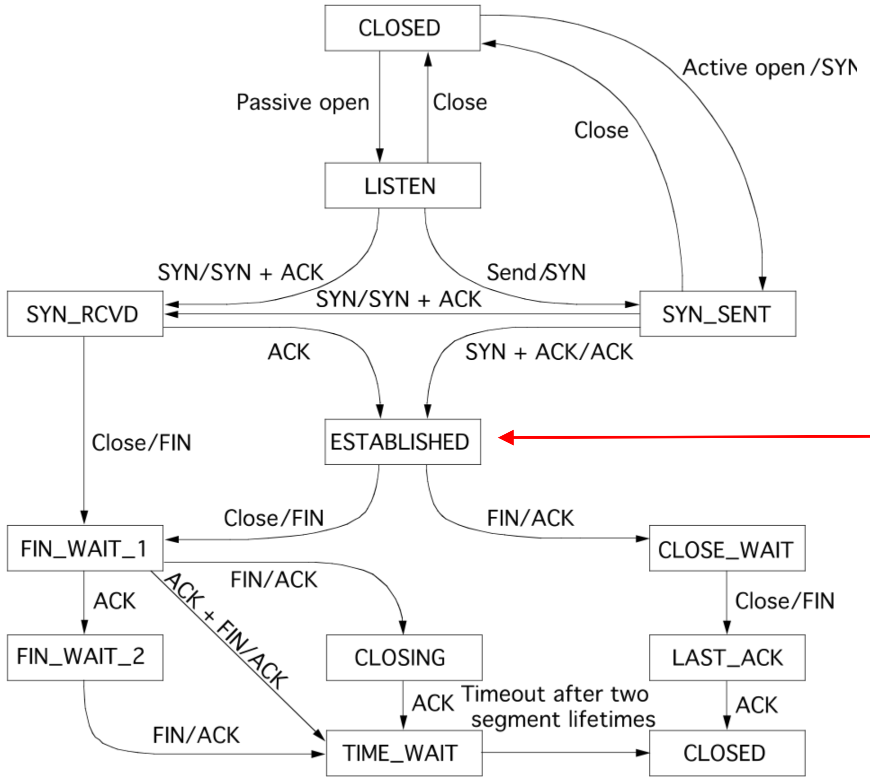
- Same as before, but B sets **FIN** with their ack of A's **FIN**

Abrupt Termination



- A sends a RESET (**RST**) to B
  - E.g., because app. 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 Transitions



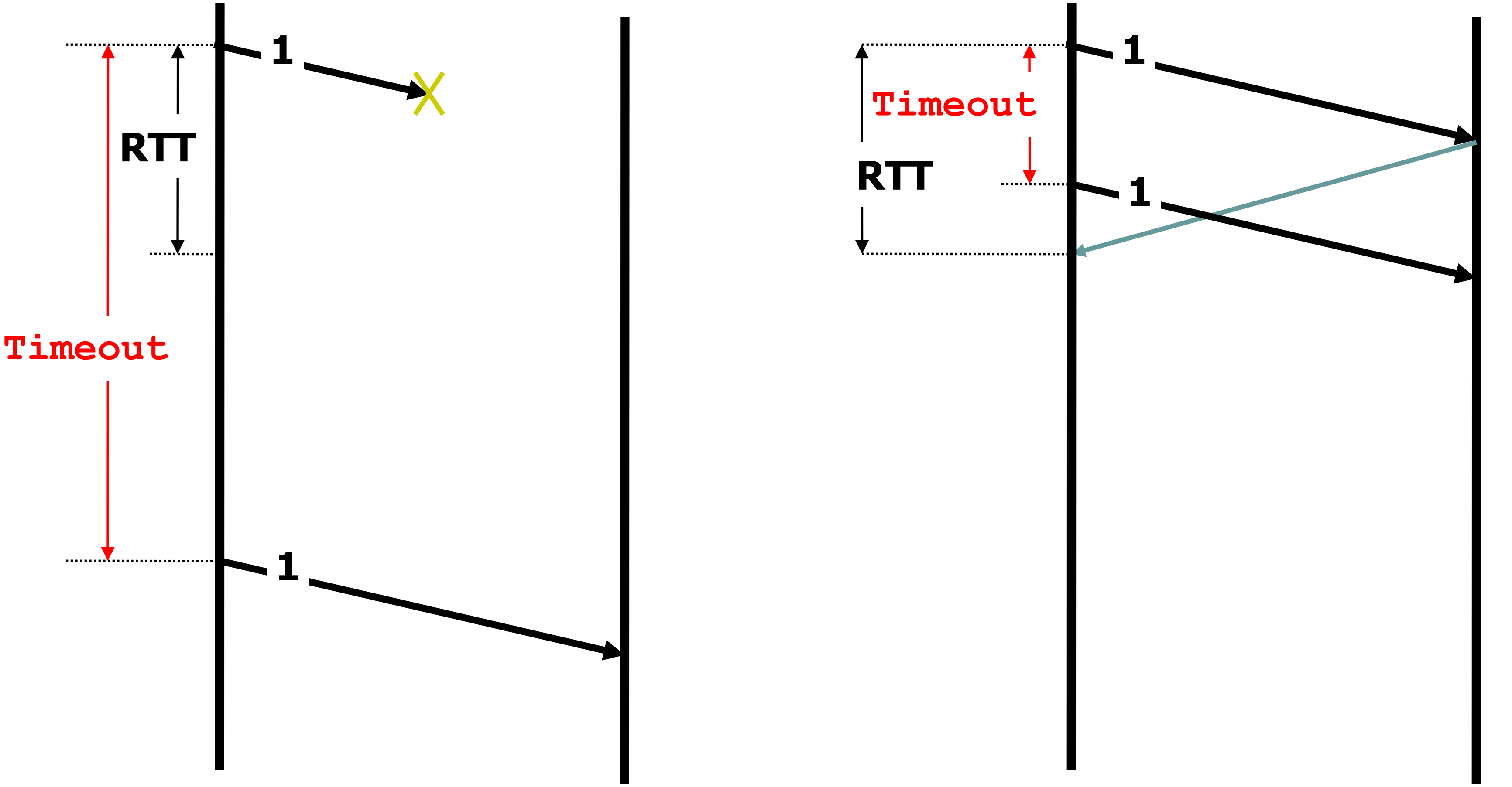
Data, ACK exchanges are in here

Reliability: TCP Retransmission

Timeouts and Retransmissions

- Reliability requires retransmitting lost data
- Involves setting timer and retransmitting on timeout
- TCP resets timer whenever new data is ACKed
    - Retx of packet containing “next byte” when timer goes off

Setting the Timeout Value



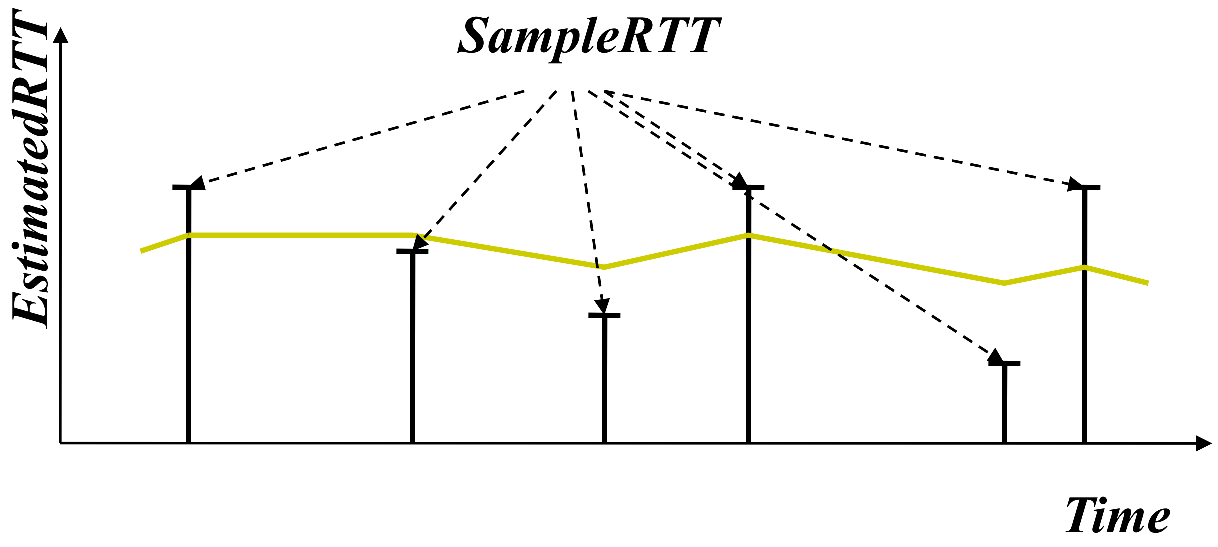
Timeout too long → inefficient

Timeout too short → duplicate packets

RTT Estimation

- Use exponential averaging of RTT samples

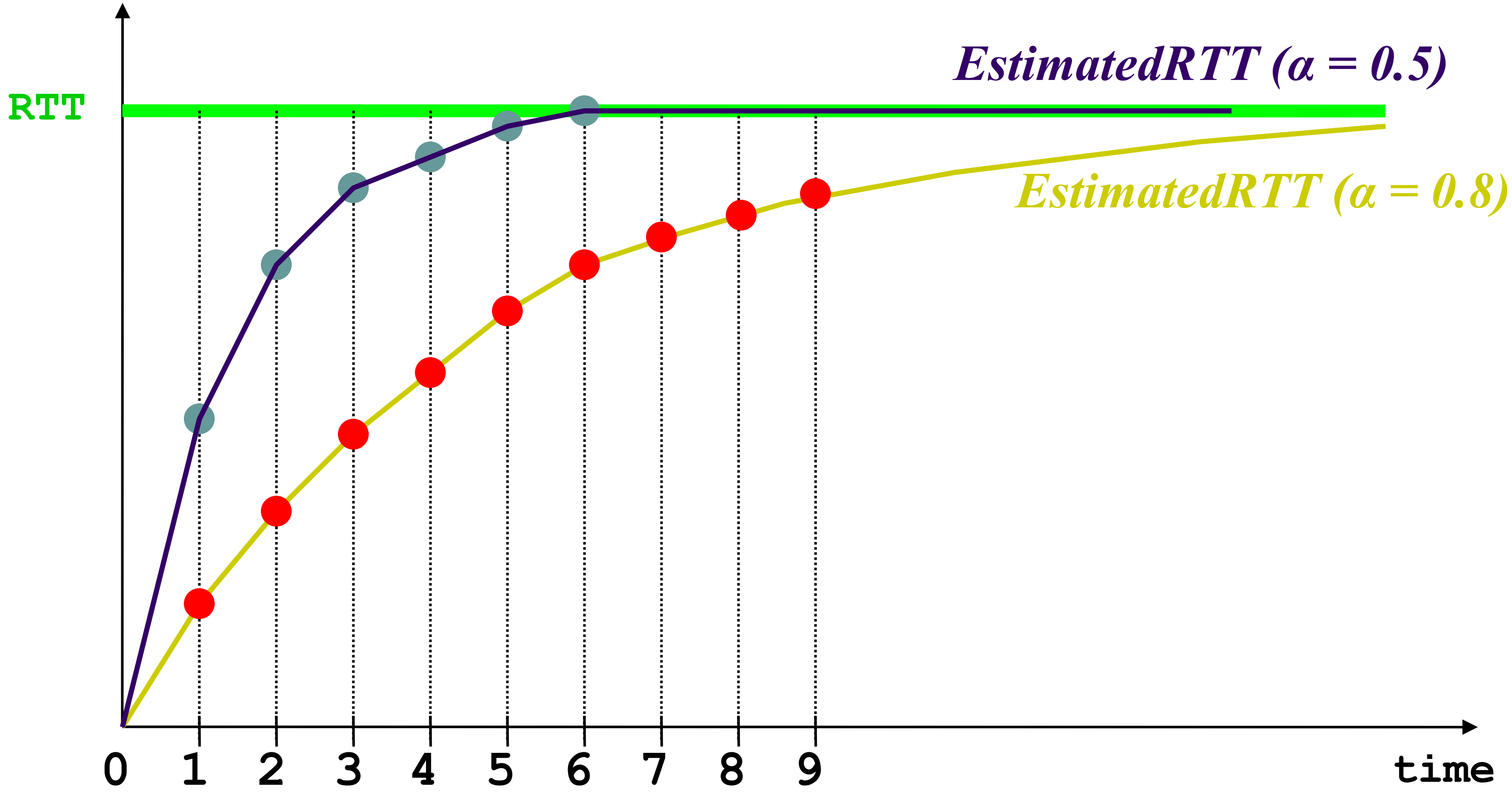
$$SampleRTT = AckRcvdTime - SendPacketTime$$
$$EstimatedRTT = \alpha \times EstimatedRTT + (1-\alpha) \times SampleRTT$$
$$0 < \alpha \le 1$$



Exponential Averaging Example

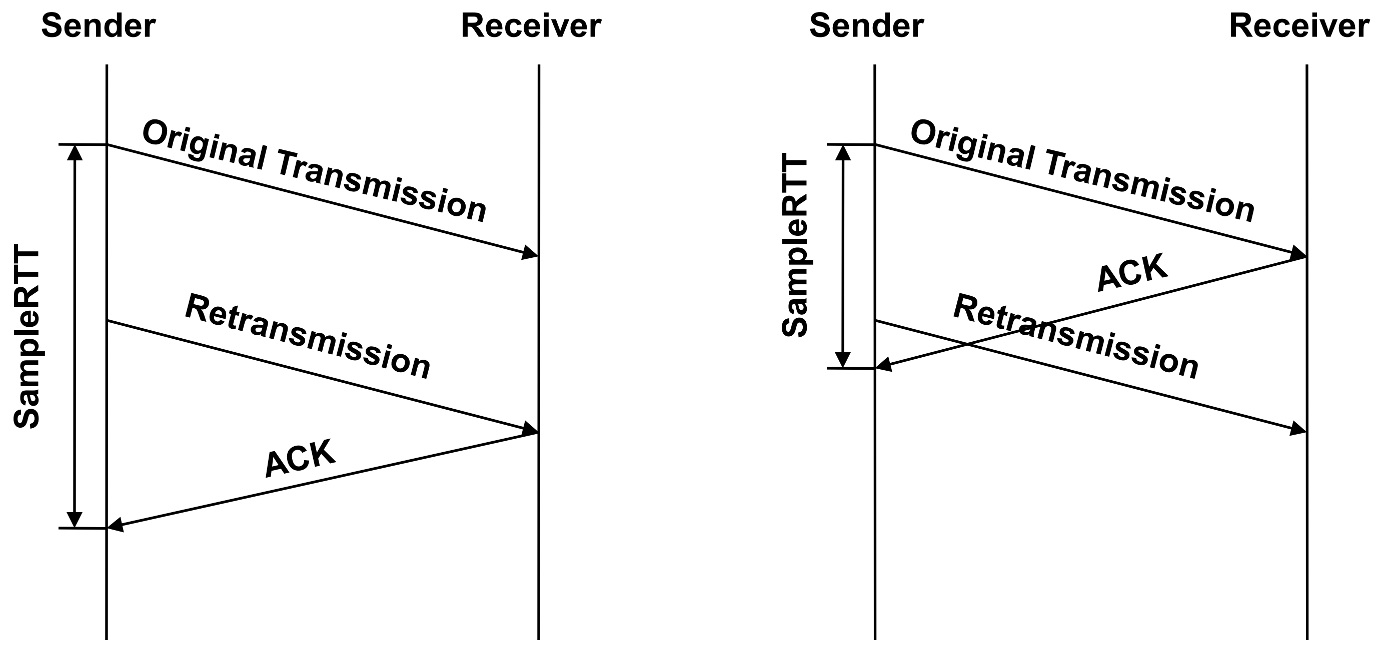
EstimatedRTT = α**EstimatedRTT* + (1 – α)**SampleRTT*

Assume RTT is constant → *SampleRTT* = RTT



Problem: Ambiguous Measurements

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



Karn/Partridge Algorithm

- Measure *SampleRTT* only for original transmissions
  - Once a segment has been retransmitted, do not use it for any further measurements
  - Computes *EstimatedRTT* using $\alpha = 0.875$

- Timeout value (RTO) = 2 × *EstimatedRTT*

- Use exponential backoff for repeated retransmissions
  - 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

This is all very interesting, but…..

- Implementations often use a coarse-grained timer
  - 500 msec is typical

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

- So we rely on duplicate ACKs

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

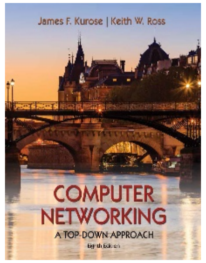
- Stream of ACKs will be:
  - 200, 300, 400, 500, 500, 500, 500,…

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
- Therefore, could trigger resend upon receiving k duplicate ACKs
  - TCP uses k=3
- We will revisit this in congestion control

Reading: Book Kurose & Ross

Class textbook:
Computer Networking: A Top-Down Approach (8th ed.)
J.F. Kurose, K.W. Ross
Pearson, 2020
http://gaia.cs.umass.edu/kurose_ross



- Week 09
  - 3.5 (Connection-Oriented Transport: TCP)
  - 3.6 (Principles of Congestion Control) and 3.7 (TCP Congestion Control)

Check Your Knowledge

PROBLEM SOLVING HOME   TRY A RANDOM PROBLEM

INTERACTIVE END-OF-CHAPTER EXERCISES

Supplement to Computer Networking: A Top Down Approach 8th Edition

"Tell me and I forget. Show me and I remember. Involve me and I understand." Chinese proverb

CHAPTER 3: TRANSPORT LAYER

- Internet checksum (similar to Chapter 3, P3 and P4)
- Reliable data transfer: rdt22
- TCP sequence and ACK numbers, with segment loss (similar to Chapter 3, P27)
- TCP RTT and timeout (similar to Chapter 3, P31)
- TCP congestion window evolution (similar to Chapter 3, P40)
- TCP retransmissions (reliable data transmission with ACK loss)
- UDP Mux and Demux
- TCP Mux and Demux

http://gaia.cs.umass.edu/kurose_ross/interactive/