CS 168
Transport and TCP

Fall 2024

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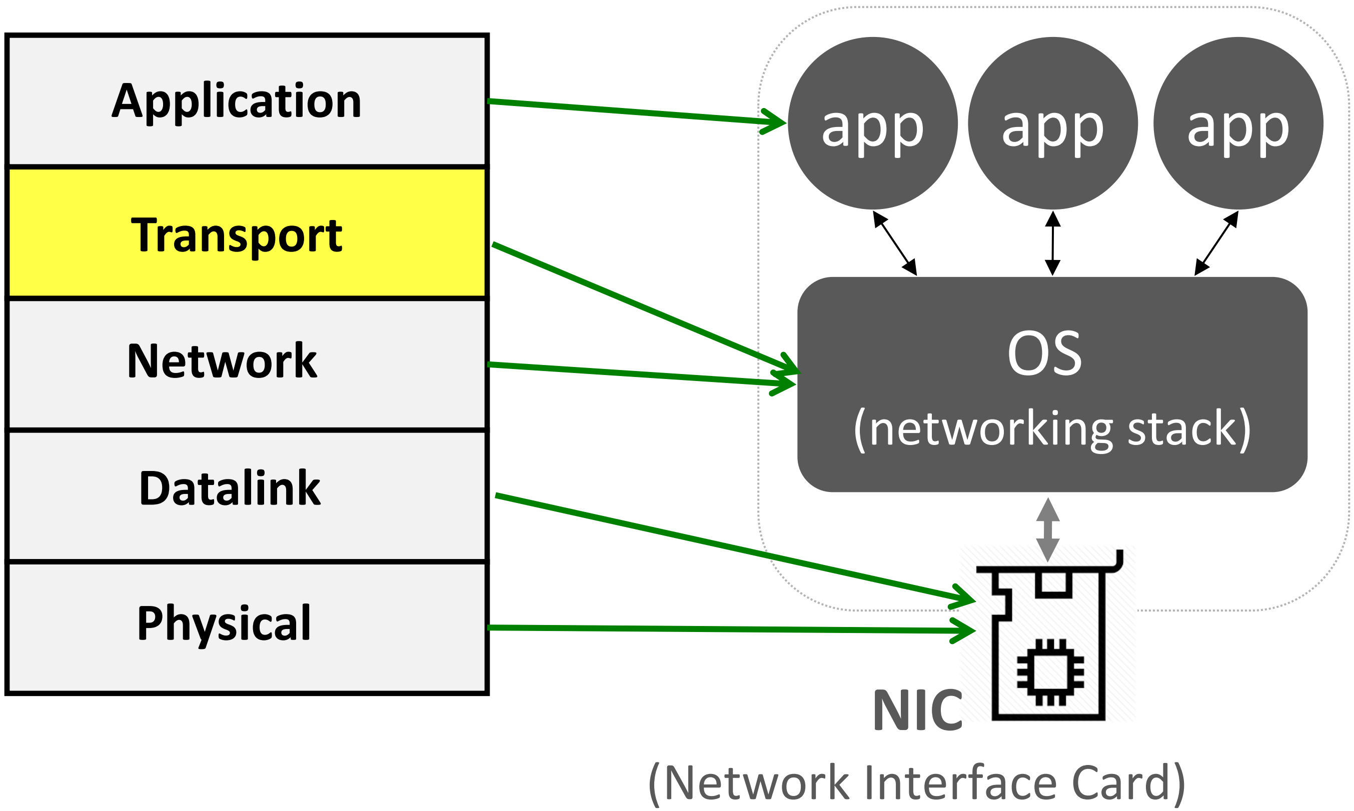
CS168.io

Taking Stock

- Last time: started on the transport layer (L4)
- Developed the techniques for *reliable* data delivery
- Today
  - A more comprehensive look at the transport layer
  - The design of TCP

Transport Layer

Transport in our layered architecture



Role of Transport Layer

- Bridging the gap between
  - **The abstractions application designers want**
  - **The abstractions networks can easily support**

- Having a common implementation simplifies app development

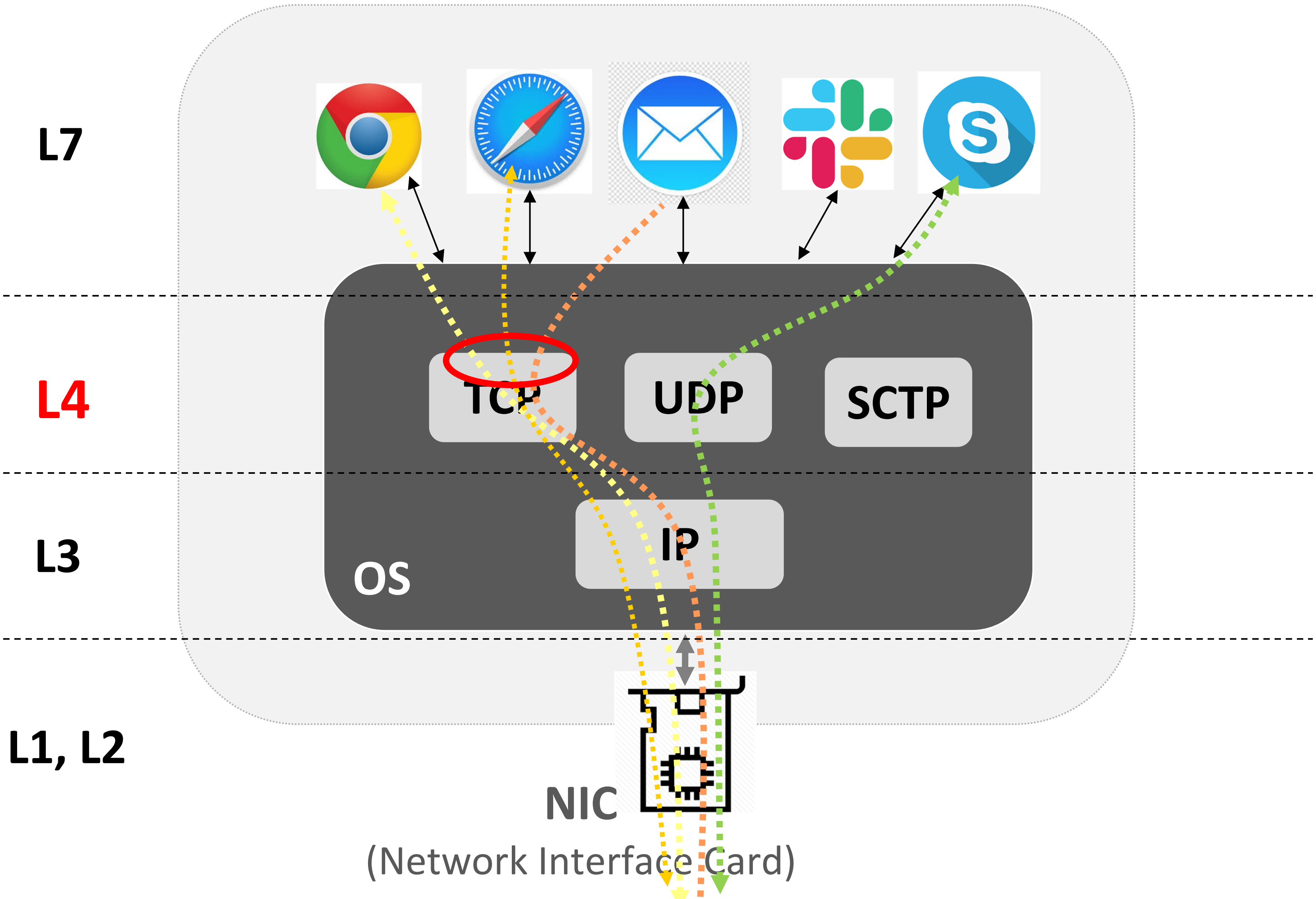
What functions does the transport layer implement?

- **Demultiplexing** between processes/apps (lecture#3)
- **Reliability** (last lecture)
- **Translate** from packets to app-level abstractions (today)
- **Flow control**: avoid overloading the receiver (today)
- **Congestion control**: avoid overloading the network (next week)

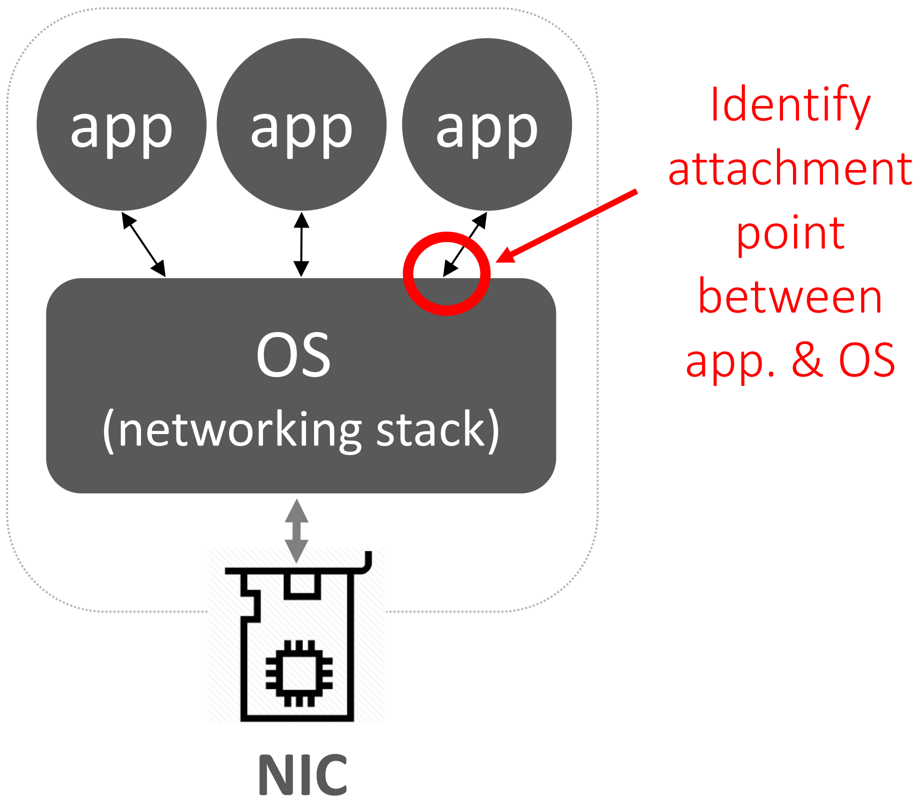
Let’s first talk about these issues in general

...and then how TCP implements them

Demultiplexing?



Recall: logical ports



Place where app connects to the OS network stack

Hence, demultiplexing

- Achieved by defining a field (“port”) that identifies the application
- Field is carried in a packet’s **L4 protocol header**

Reliable Delivery

- Last lecture

  - We've identified our design building blocks
    - Checksums
    - ACK/NACKs
    - Timeouts
    - Retransmissions
    - Sequence numbers
    - Windows
  - And discussed tradeoffs in how to apply them
    - Individual vs. Full vs. Cumulative ACKs
    - Timeout vs. ACK-driven loss detection

Application-layer abstractions

- Ideally, app doesn't see the gory details of the network
  - packets, ACKs, duplicates, reordering, corruption, ...
- Want a higher-level abstraction that meets app needs

Application Abstractions

- **Reliable in-order bytestream** delivery (TCP)
  - Logical "pipe" between sender and receiver
  - Bytes inserted into pipe by sender-side app
  - They emerge, in order, at the receiving app

- **Individual message** delivery (UDP)
  - Unreliable (application responsible for resending)
  - Messages limited to single packet

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- **Flow control**: avoid overloading the receiver (today)
- **Congestion control**: avoid overloading the network (next week)

How big should the window be?

- **Last lecture:** Pick window size W to balance three goals
  - Take advantage of network capacity (“fill the pipe”)
  - But don’t overload the receiver (flow control)
  - And don’t overload links (congestion control)
- **Last lecture:** For the first goal: W x packet-size ~ RTT x B
  - RTT is round-trip time and B is the bottleneck BW
  - This is **an upper bound** on the desired size of W
- Now consider the other two goals...

Don’t overload the receiver

- Consider the transport layer at the receiver side
- May receive packets out-of-order but can only deliver them to the application in order
- Hence, the receiver must buffer incoming packets that are out of order
  - Must continue to do so until all “missing” packets arrive!
- Must ensure the receiver doesn’t run out of buffers

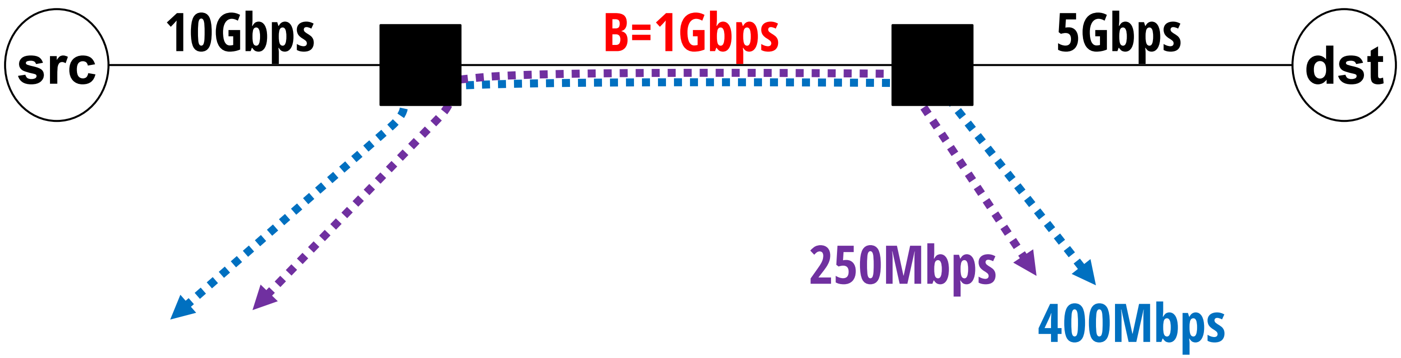
Hence: Flow Control

The basic idea is very simple...

- Receiver tells sender how much space it has left
  - TCP calls this the **"advertised" window**
- Advertisement is carried in ACKs
- Sender adjusts its window accordingly
  - Packets in flight cannot exceed the receiver's advertised window

Don’t overload the network

- Previously: sender sets W to fully consume the bottleneck link bandwidth
  - I.e., sender is sending data at the rate of B
- In practice, bottleneck is shared with other flows
- Hence, sender should only consume *its share* of B
- But what is this share?



Congestion Control

- The transport layer at the sender implements a congestion control algorithm that dynamically computes the sender's share of the bottleneck link BW
- TCP calls this the sender's **congestion window** (cwnd)
- Computed to balance multiple goals
  - Maximize my performance
  - Without overloading any link (avoid dropped packets)
  - While sharing bandwidth ”fairly” with other senders
- Topic for (multiple) future lectures

So, how big should the window be?

- Pick window size **W** to balance three goals
  - Take advantage of network capacity ("fill the pipe")
  - But don't overload the receiver (flow control)
  - And don't overload links (congestion control)

- First goal: W ~ RTT x B
- Second: W ~ receiver's advertised window
- Third: W ~ sender's congestion window (cwnd)

- Window size is set to the minimum of the above

In practice

- A sender's cwnd should be <= RTT x B
- And it's difficult for the sender to discover B
- Hence, window size is the minimum of:
  - The congestion window computed at the sender
  - The receiver's advertised window

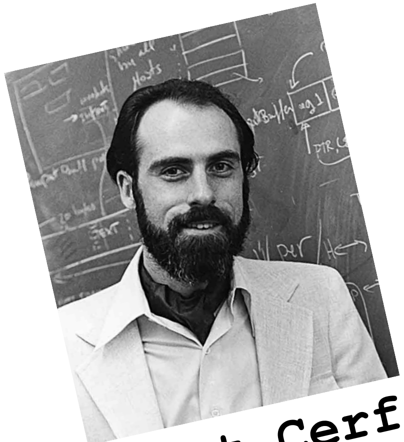
Recap: what the transport layer tackles

- Demultiplexing
  - logical ports
- Reliability
  - acks, timeouts, windows, etc.
- Translation between abstractions
  - between packets and bytestreams (coming up)
- Avoid overloading the receiver
  - receiver's "advertised window"
- Avoid overloading the network
  - sender computes a "congestion window"

What if your app doesn't want all these features?

- E.g., an application that doesn't need reliability
- E.g., an app that exchanges very short messages

- UDP: User Datagram Protocol
  - A no-frills, minimalist protocol
  - Only implements mux/demux


Vint Cerf

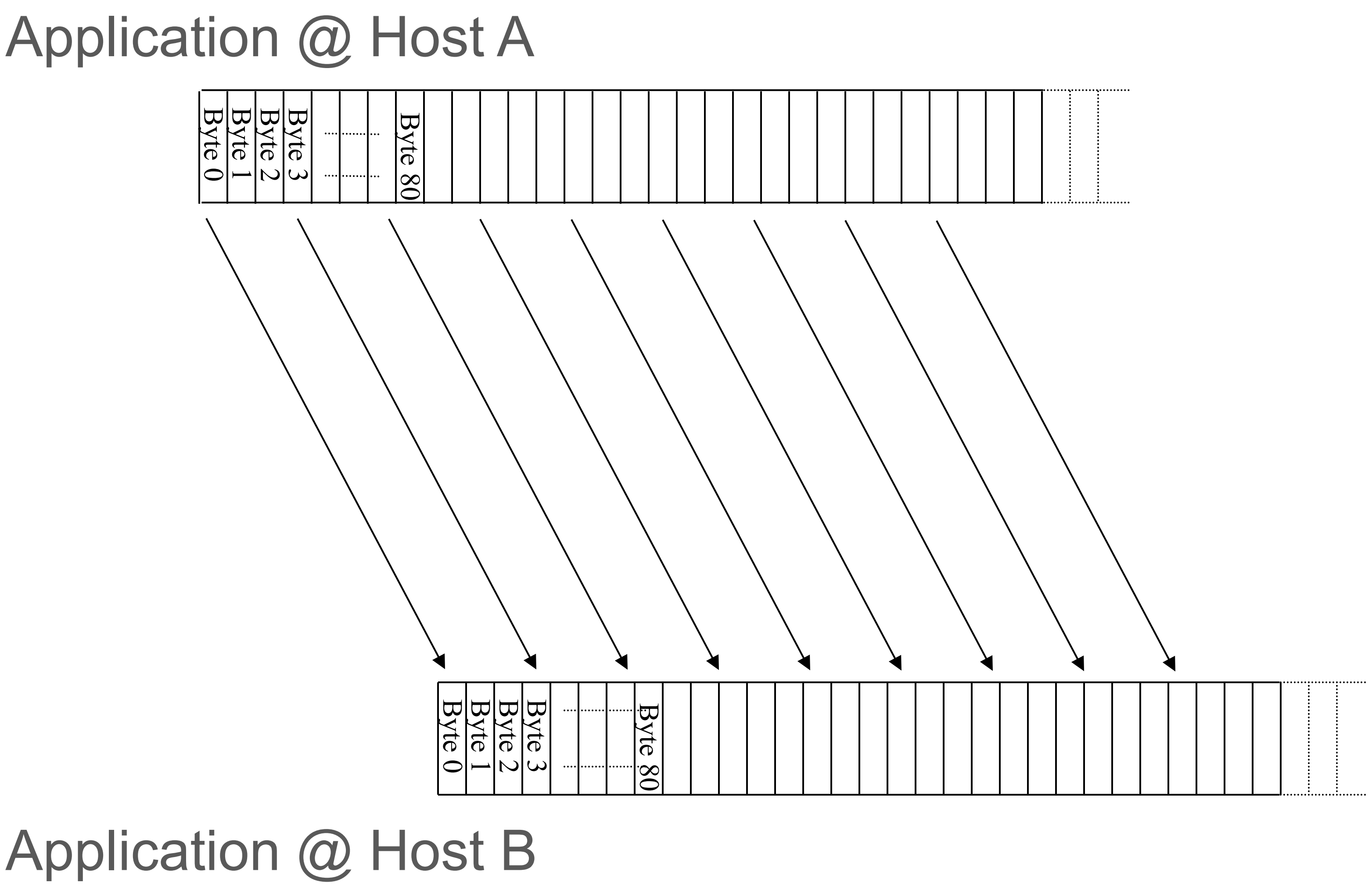

Bob Kahn

TCP

The TCP Abstraction

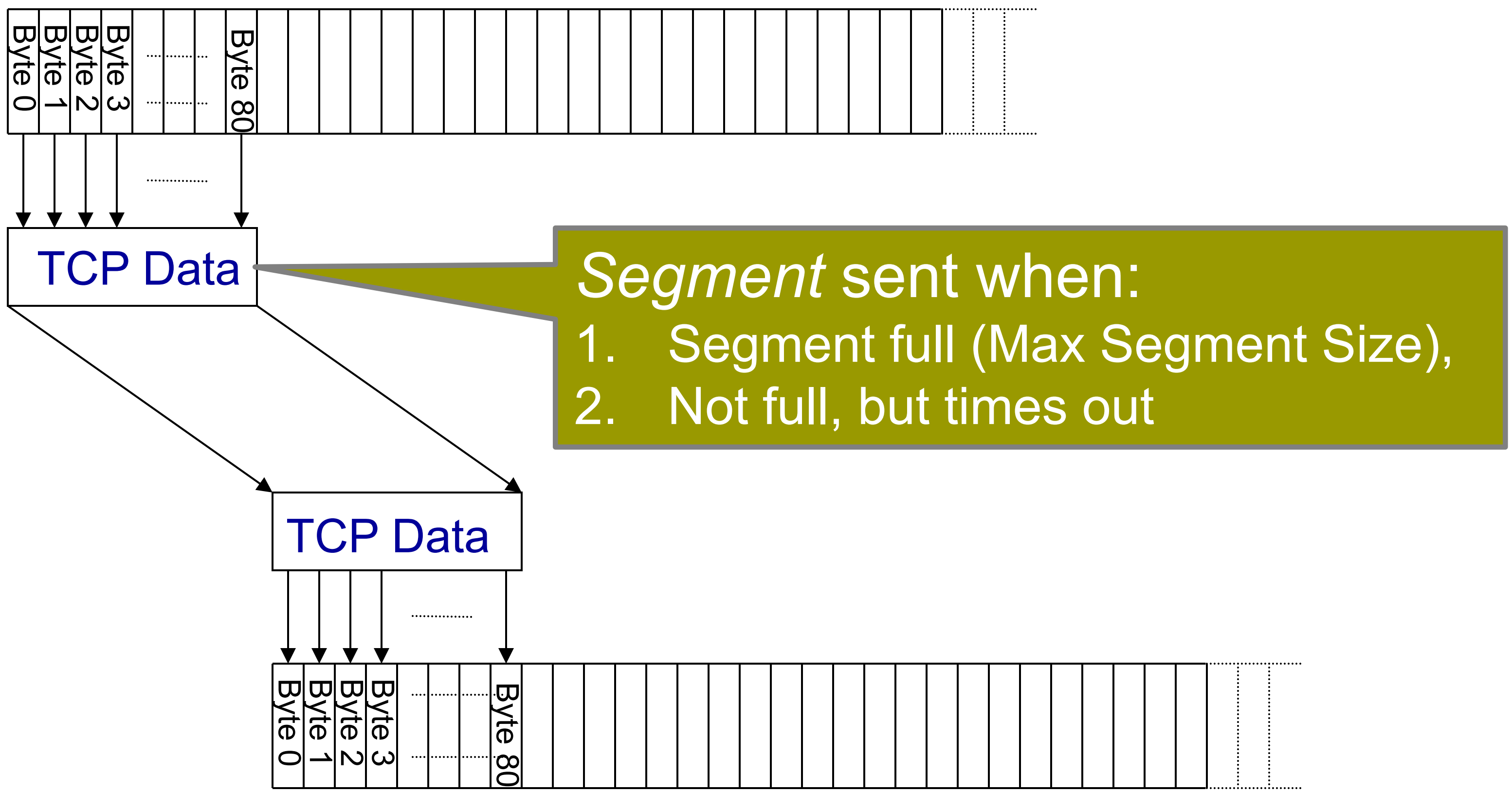
- TCP delivers a **reliable, in-order, bytestream**

TCP “Stream of Bytes” Service…



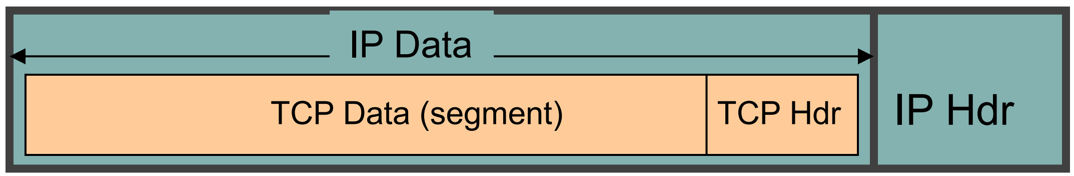
… Implemented Using TCP “Segments”

Application @ Host A



Application @ Host B

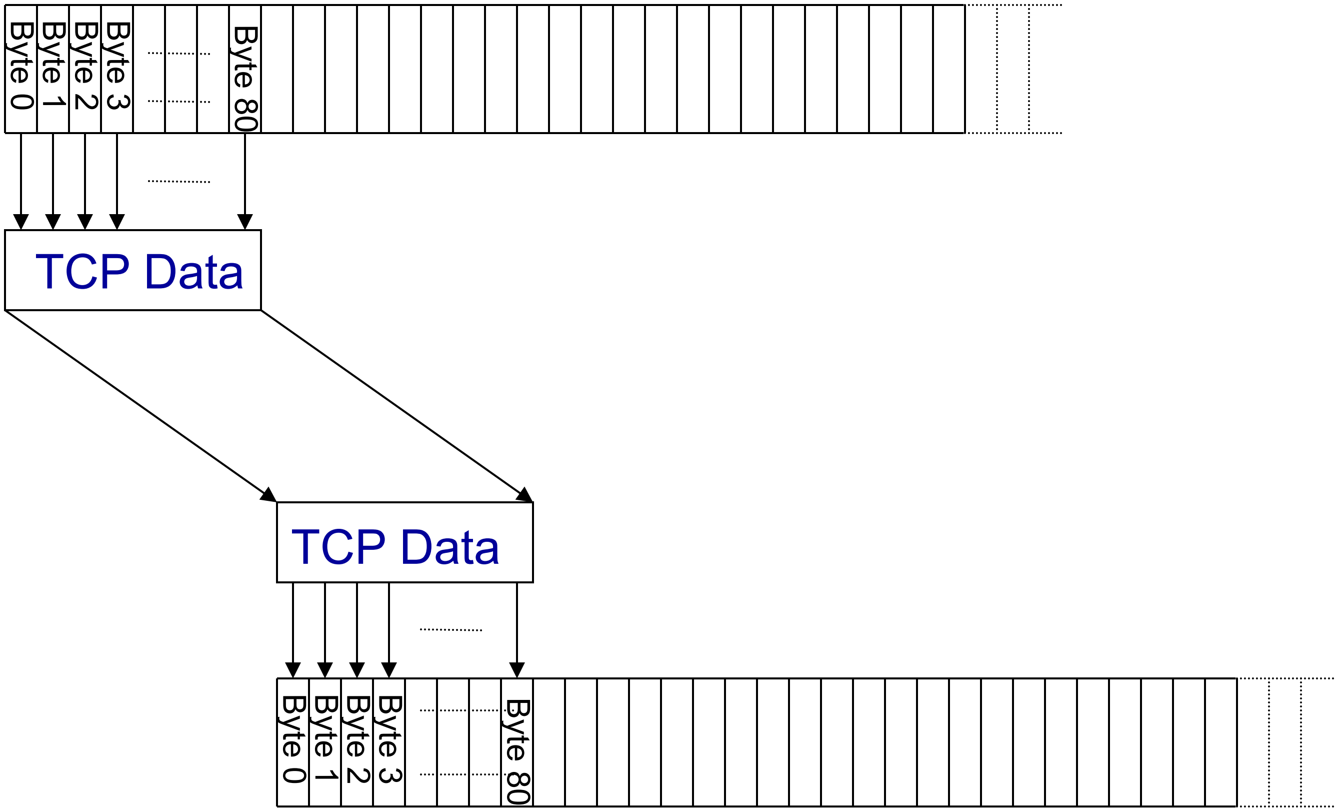
TCP Segment



- TCP/IP packet
  - IP packet with a TCP header and data inside
- IP packet
  - No bigger than Maximum Transmission Unit (MTU)
- TCP **segment**
  - No more than Maximum Segment Size (MSS) bytes
  - MSS = MTU – (IP header) – (TCP header)

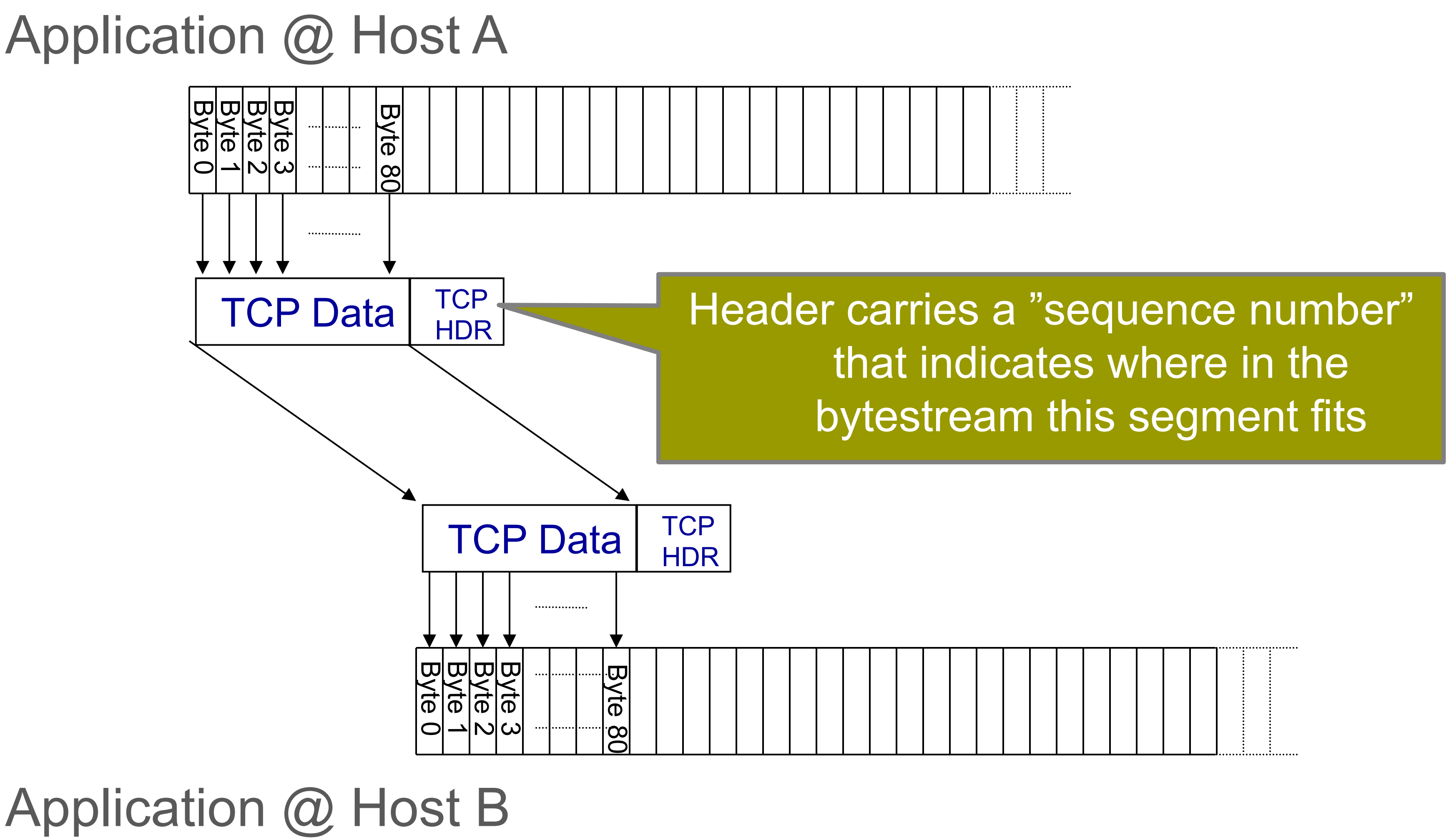
… Implemented Using TCP “Segments”

Application @ Host A



Application @ Host B

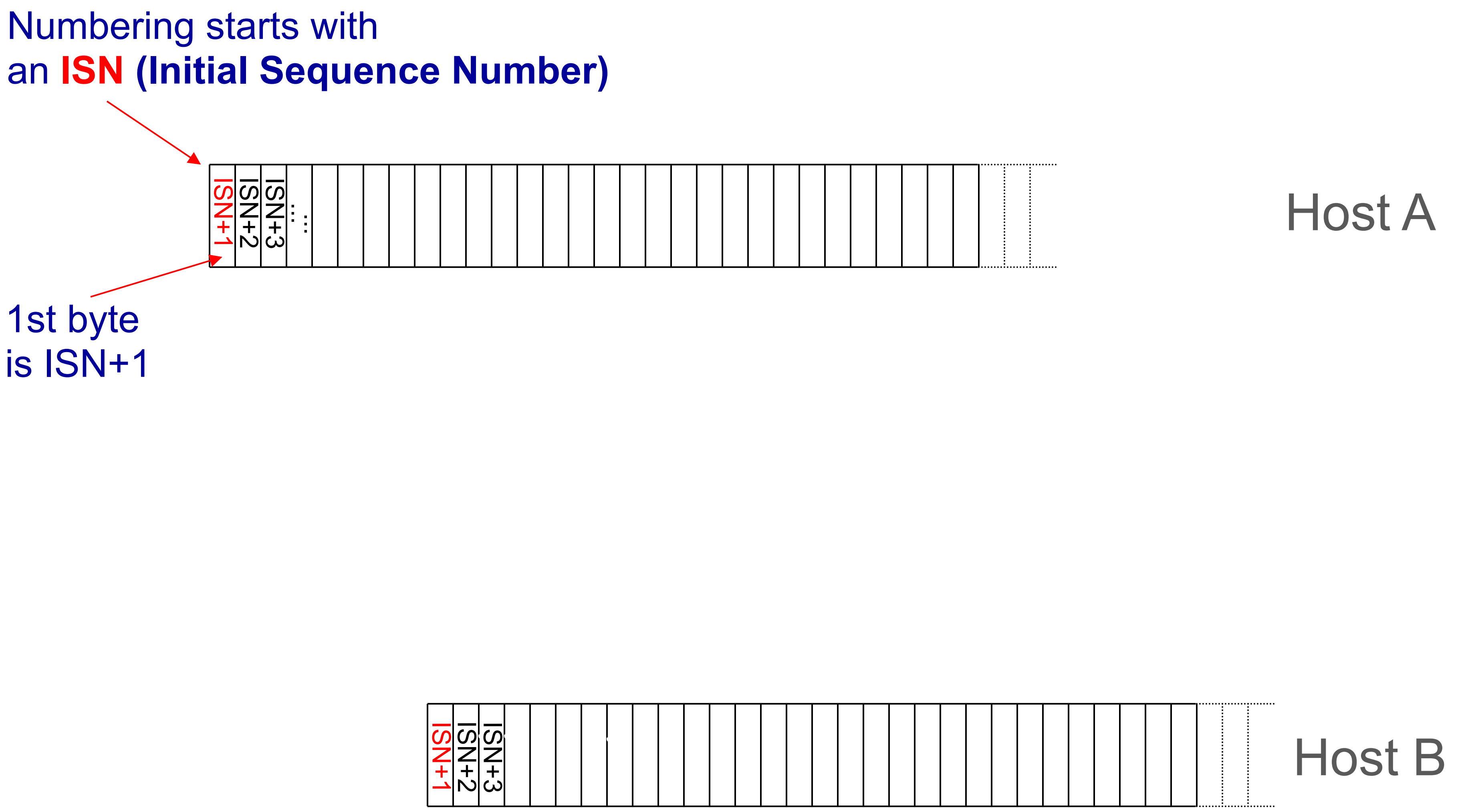
… Described by TCP headers



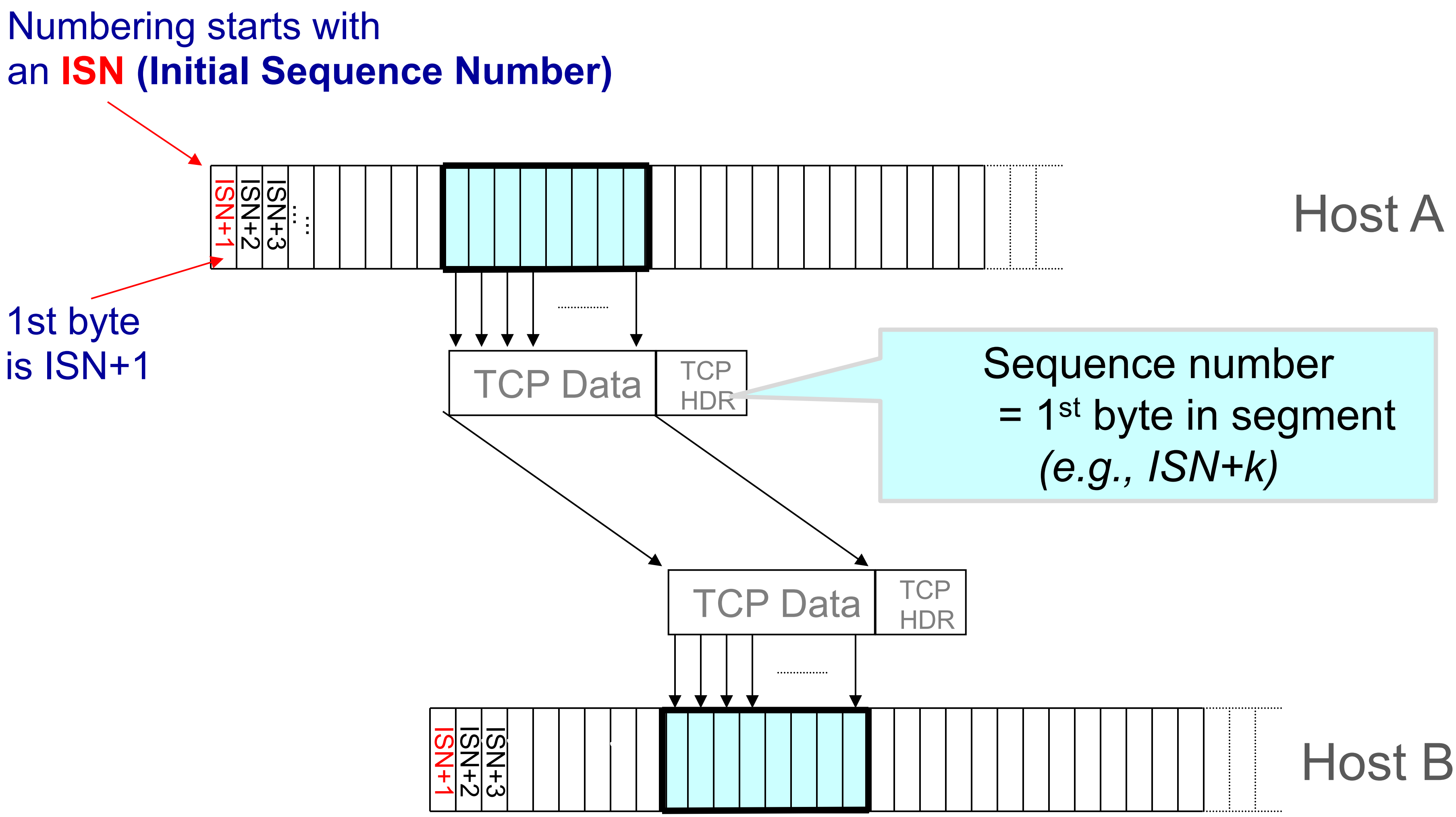
Major Notation Change

- Previously we focused on packets:
  - Packets had numbers
  - ACKs referred to those numbers
  - Window sizes expressed in terms of # of packets
- TCP focuses on bytes. Thus,
  - Packets identified by the bytes they carry
  - ACKs refer to the bytes received
  - Window size expressed in terms of # of bytes
- You should be prepared to reason in terms of either

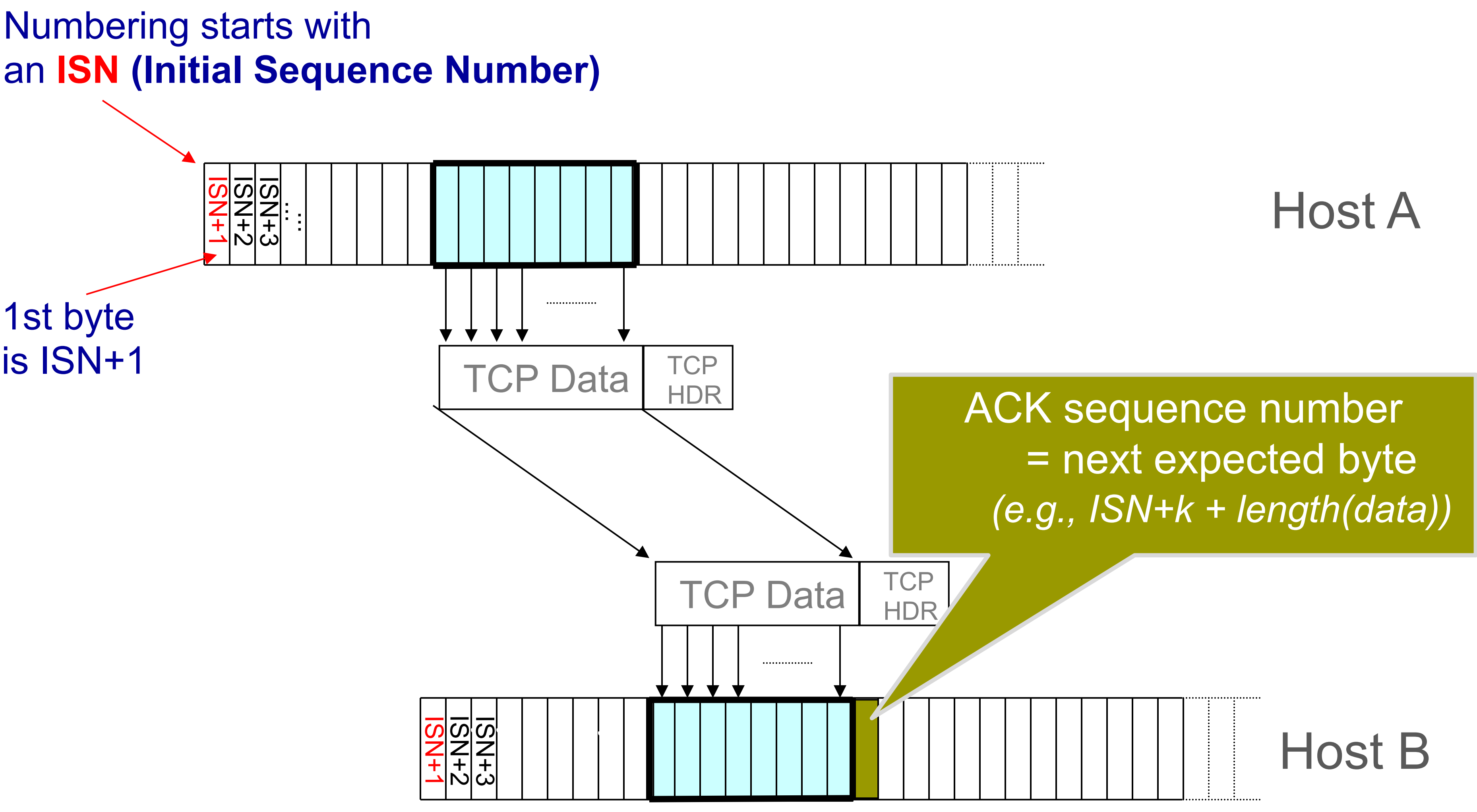
TCP Sequence Numbers



TCP Sequence Numbers



TCP Sequence Numbers



The TCP Abstraction

- TCP delivers a **reliable,** **in-order**, **bytestream**
- Reliability requires keeping state
  - Sender: packets sent but not ACKed, related timers
  - Receiver: out-of-order packets
- Each bytestream is called a connection or session
  - Each with their own connection state
  - State is in hosts, not network!

Note#1: TCP is “connection oriented”

- TCP includes a connection setup and tear-down step
  - Used to initialize connection state at both endpoints
  - Details coming up ...

#2: TCP connections are full-duplex

- So far, we've talked about a connection as having a sender side and a receiver side

- But connections in TCP are full-duplex
  - Each side of the connection can be sender *and* receiver
  - I.e., A can send data to B, while B sends data to A
  - Simultaneously, over the same connection
  - Packets carry both data and ACK info

- We can usually ignore this point (for this class)
- Except when it comes to connection establishment
- Will return to this later ...

The TCP Abstraction

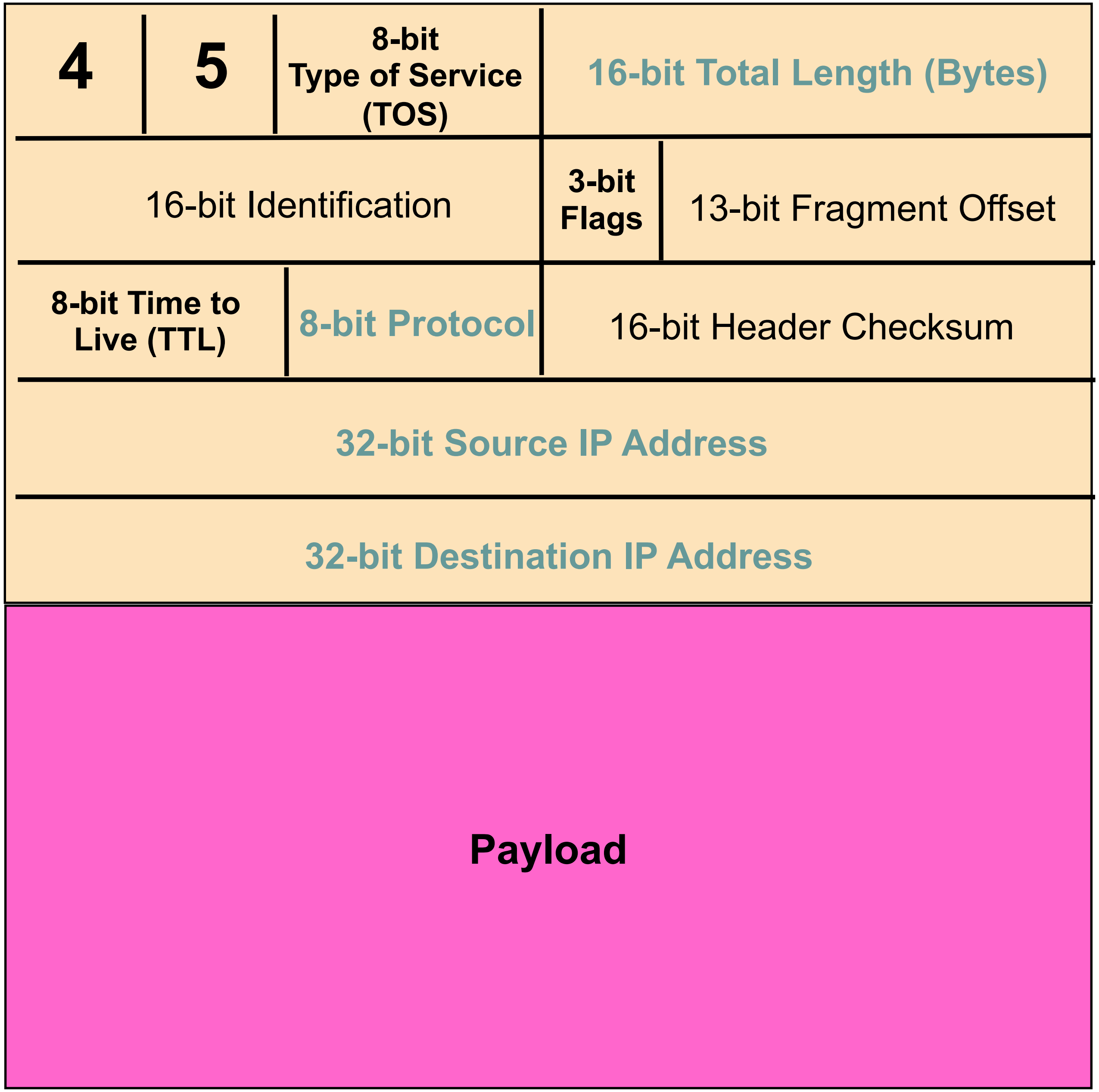
- TCP delivers a **reliable, in-order, bytestream**
- TCP is connection-oriented
  - Per-connection state is maintained at sender & receiver

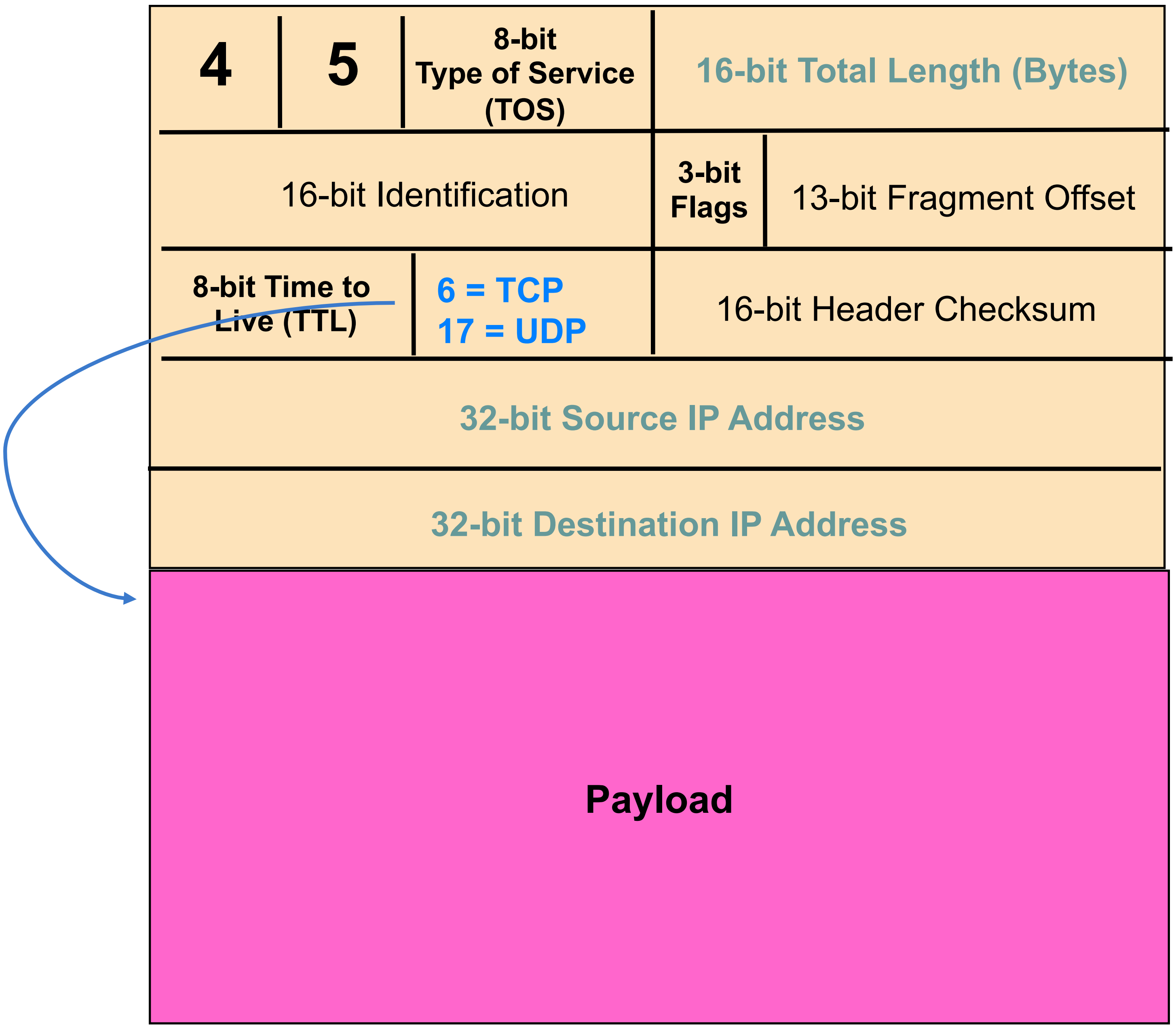
Functionality

- Mux/demux among processes
- Reliability
- Flow control (to not overflow receiver)
- Congestion control (to not overload network)
- "Connection" set-up & tear-down

Ports

- 16-bit port address space for TCP and UDP
- Some ports are “well known” (0-1023)
  - e.g., ssh:22, http:80
  - Services can listen on well-known port
  - Client (app) knows appropriate port on server
- Other ports are “ephemeral” (most 1024-65535):
  - Given to clients (at random)


4
5
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
Payload



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17 = UDP
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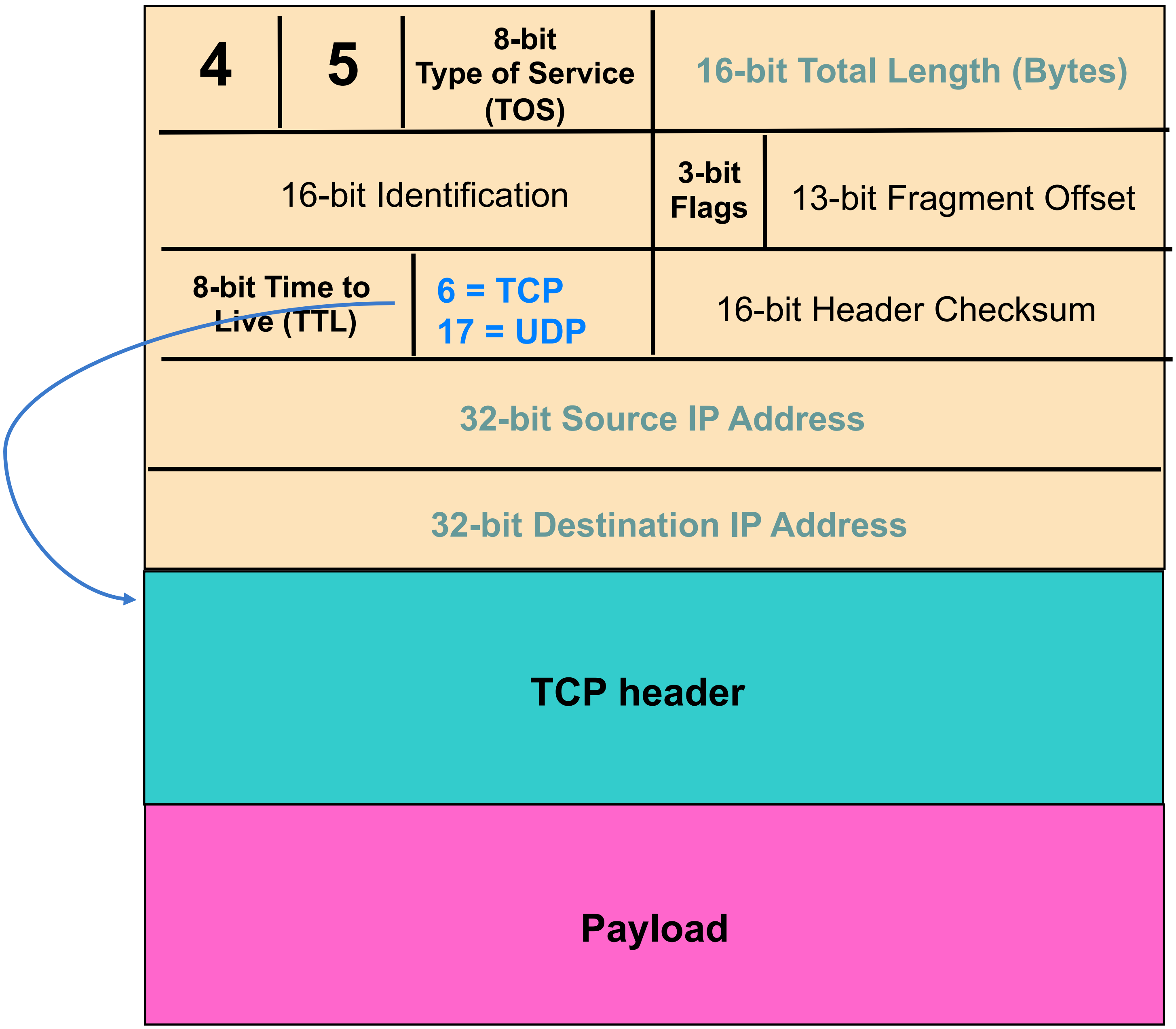
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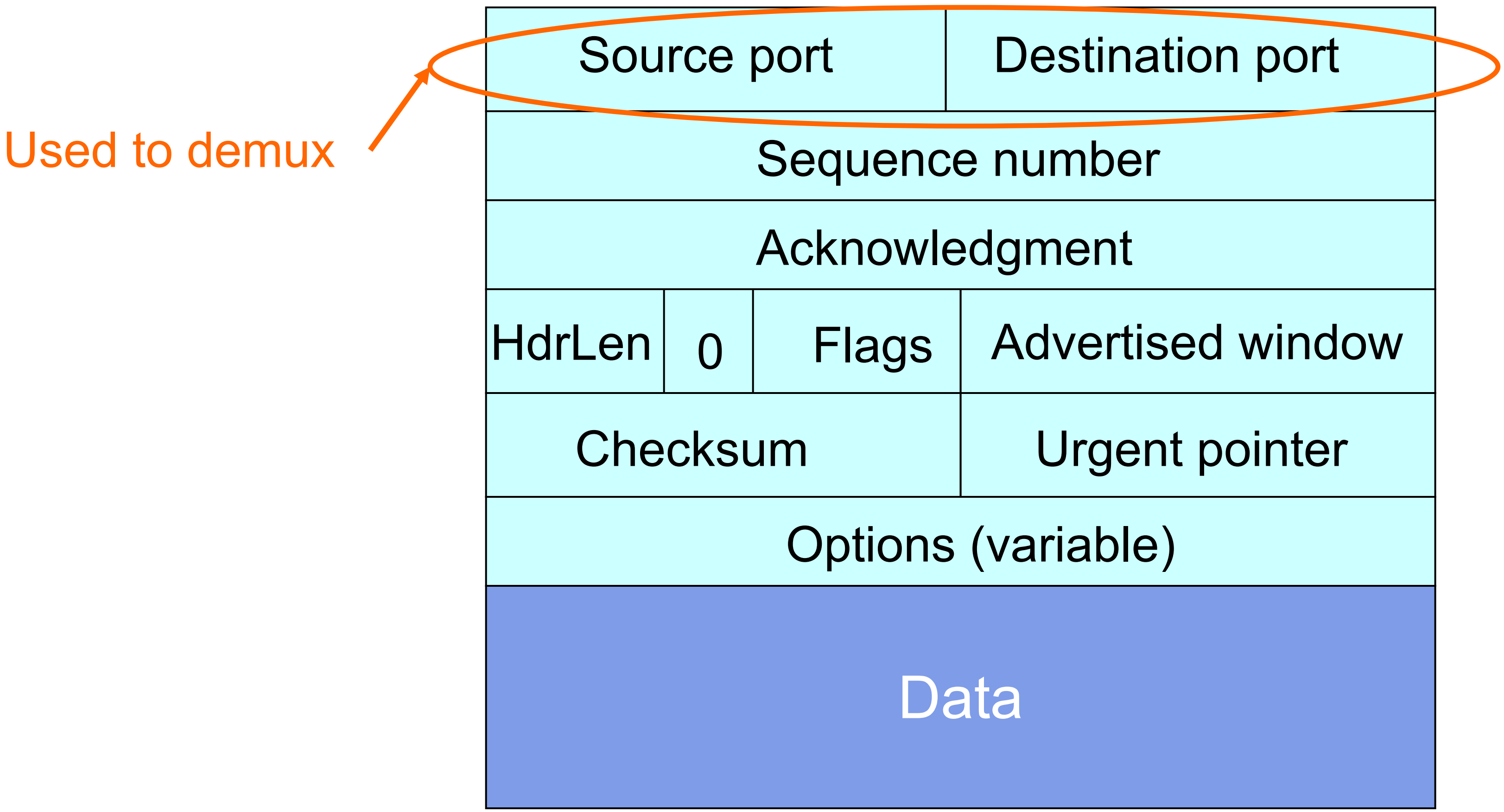
TCP header
Payload

TCP Header



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

Functionality

- Mux/demux among processes
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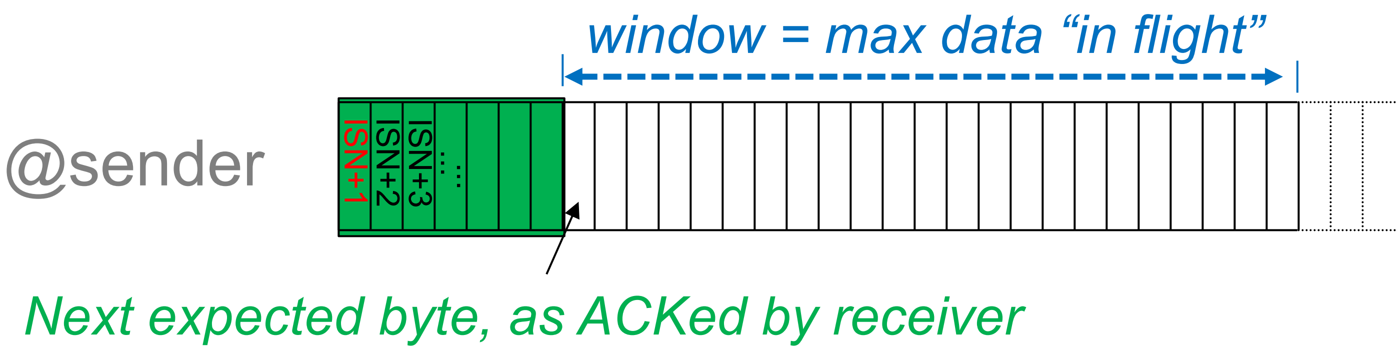
How does TCP handle reliability?

Many of our previous ideas, with some key differences

- Sequence numbers are byte offsets
- Uses cumulative ACKs; with "next expected byte" semantics
- Uses sliding window: up to W <u>contiguous</u> bytes in flight



How does TCP handle reliability?

Many of our previous ideas, with some key differences

- Sequence numbers are byte offsets
- Uses cumulative ACKs; with “next expected byte” semantics
- Uses sliding window: up to W <u>contiguous</u> bytes in flight
- Retransmissions triggered by timeouts and duplicate ACKs
- Single timer, for left hand side (1st byte) of the window
- Window size is a function of cwnd and advertised window
  - With special accounting for duplicate ACKs (future lecture)
- Timeouts are computed from RTT measurements
  - Covered in section

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 a smaller value Y
    - ACK acknowledges Y+1 (because TCP uses cumulative ACKs)

Pattern (w/ only one packet in flight)

- Sender: seq number =X, length=B
- Receiver: ACK=X+B
- Sender: seq number =X+B, length=B
- Receiver: ACK=X+2B
- Sender: seq number =X+2B, length=B

- Seq number of next packet is same as last ACK

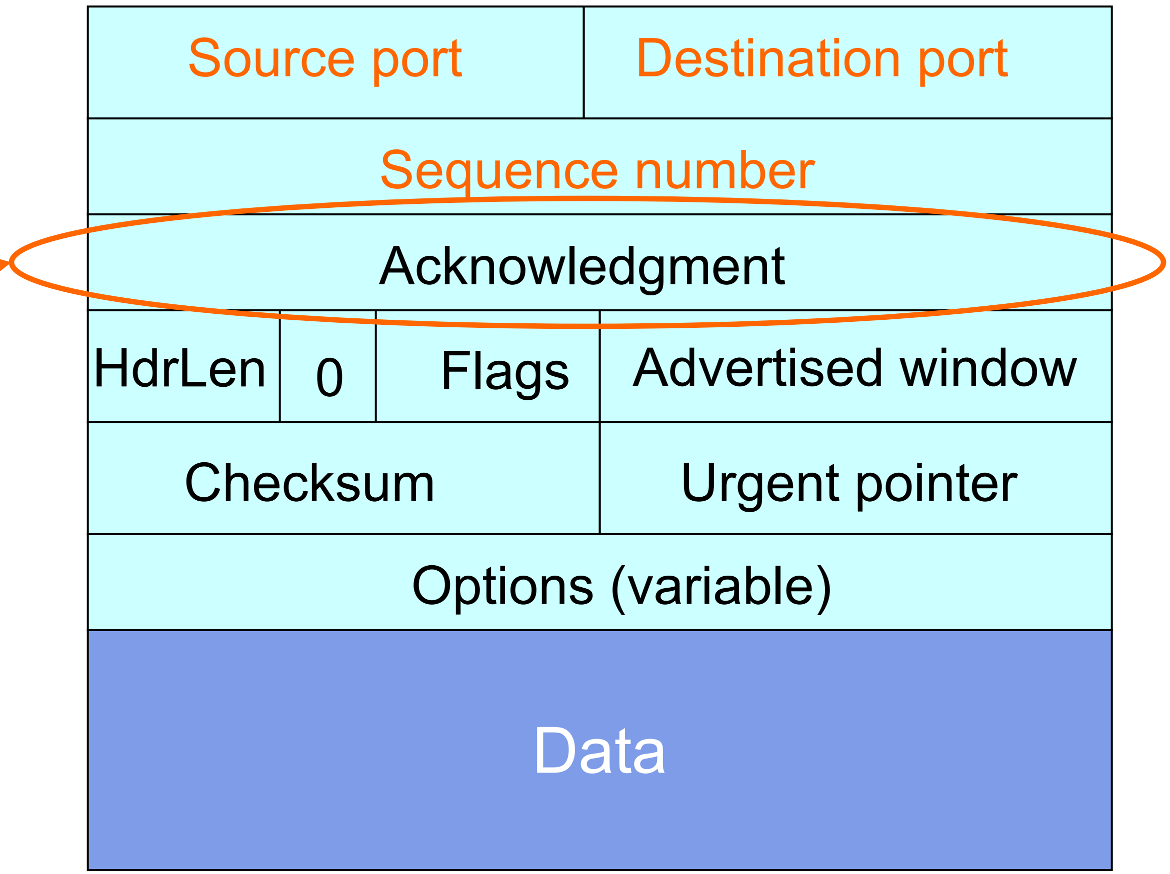
TCP Header

Starting byte offset of data carried in this segment

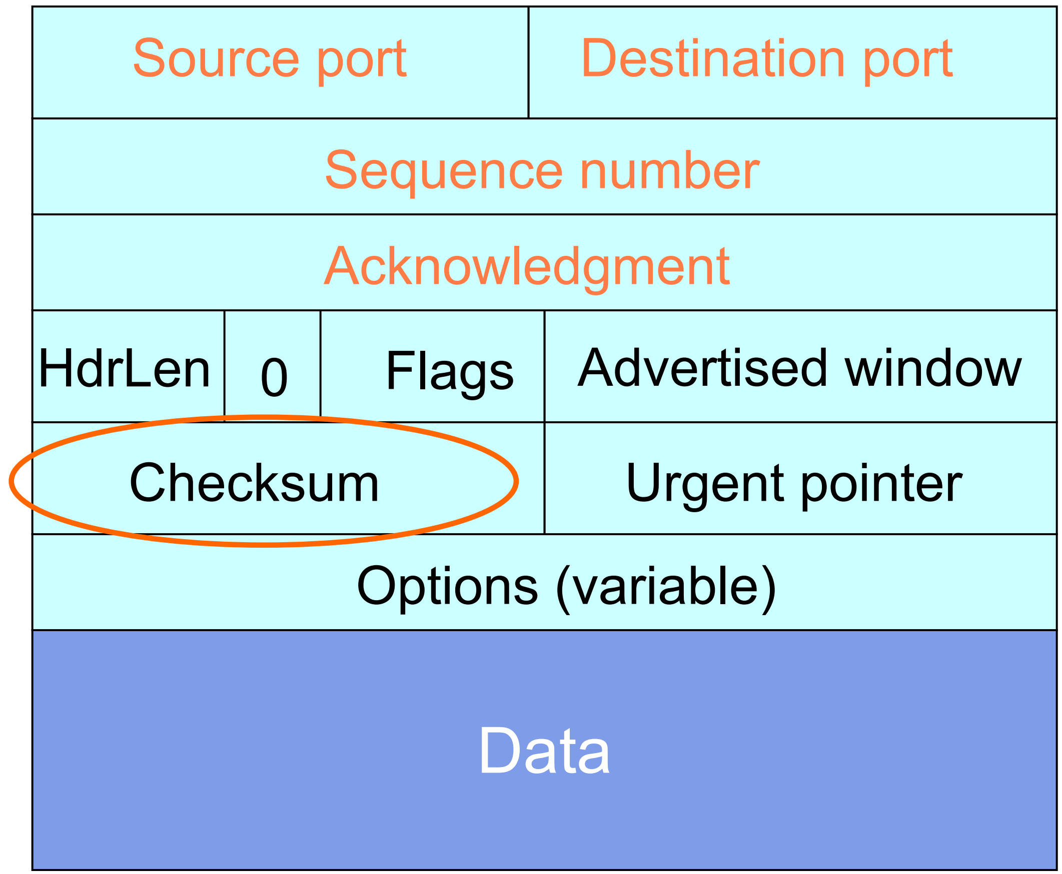
<table>
<tr><td colspan="3">Source port</td><td>Destination port</td></tr>
<tr><td colspan="4">Sequence number</td></tr>
<tr><td colspan="4">Acknowledgment</td></tr>
<tr><td>HdrLen</td><td>0</td><td>Flags</td><td>Advertised window</td></tr>
<tr><td colspan="3">Checksum</td><td>Urgent pointer</td></tr>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

TCP Header

Acknowledgment gives sequence number just beyond the highest sequence number received **in order** (*i.e.*, next expected byte)



TCP Header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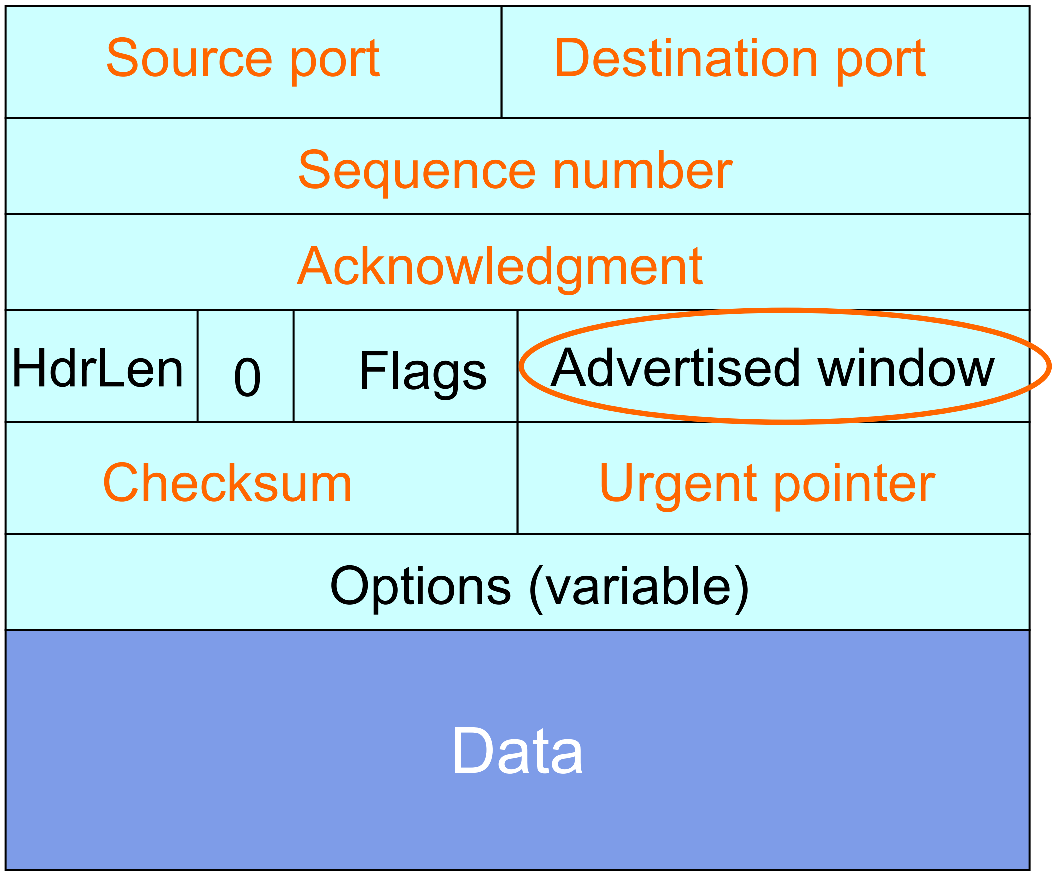
TCP Header



Source port
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Sequence number
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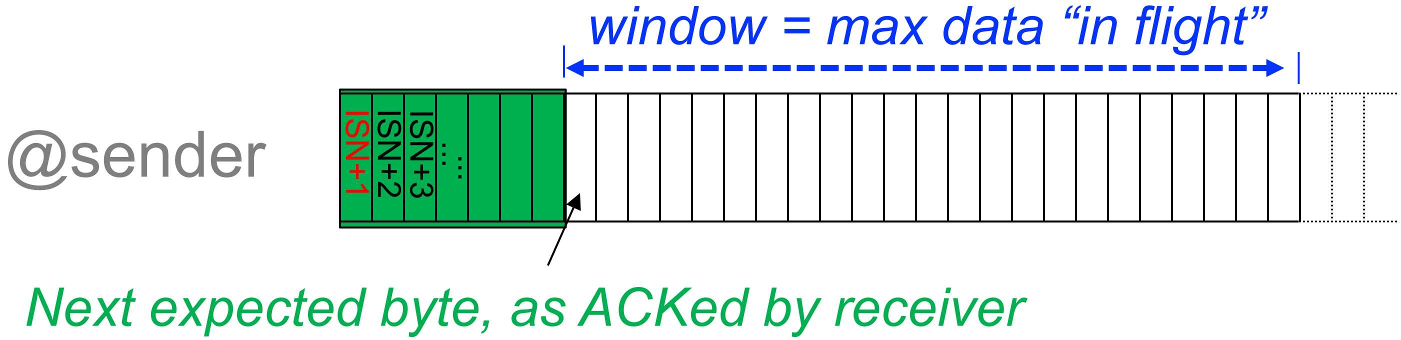
Implementing Sliding Window

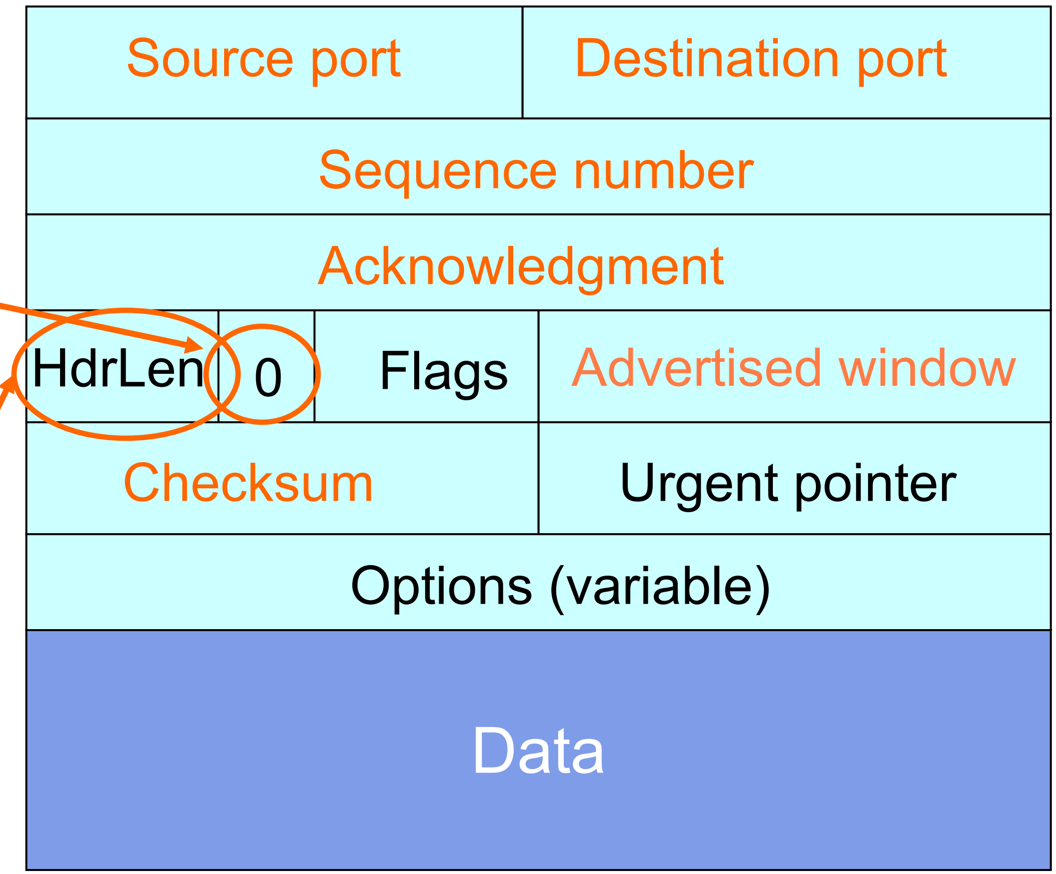
- Sender maintains a window
  - Data that has been sent but not yet ACK'ed
  - Window size = maximum amount of data in flight
- Left edge of window:
  - Beginning of unacknowledged data
- Right edge of window (ignoring congestion control)
  - Depends on the window advertised by receiver
  - Which depends on receiver's buffer space



TCP Header: What's left?

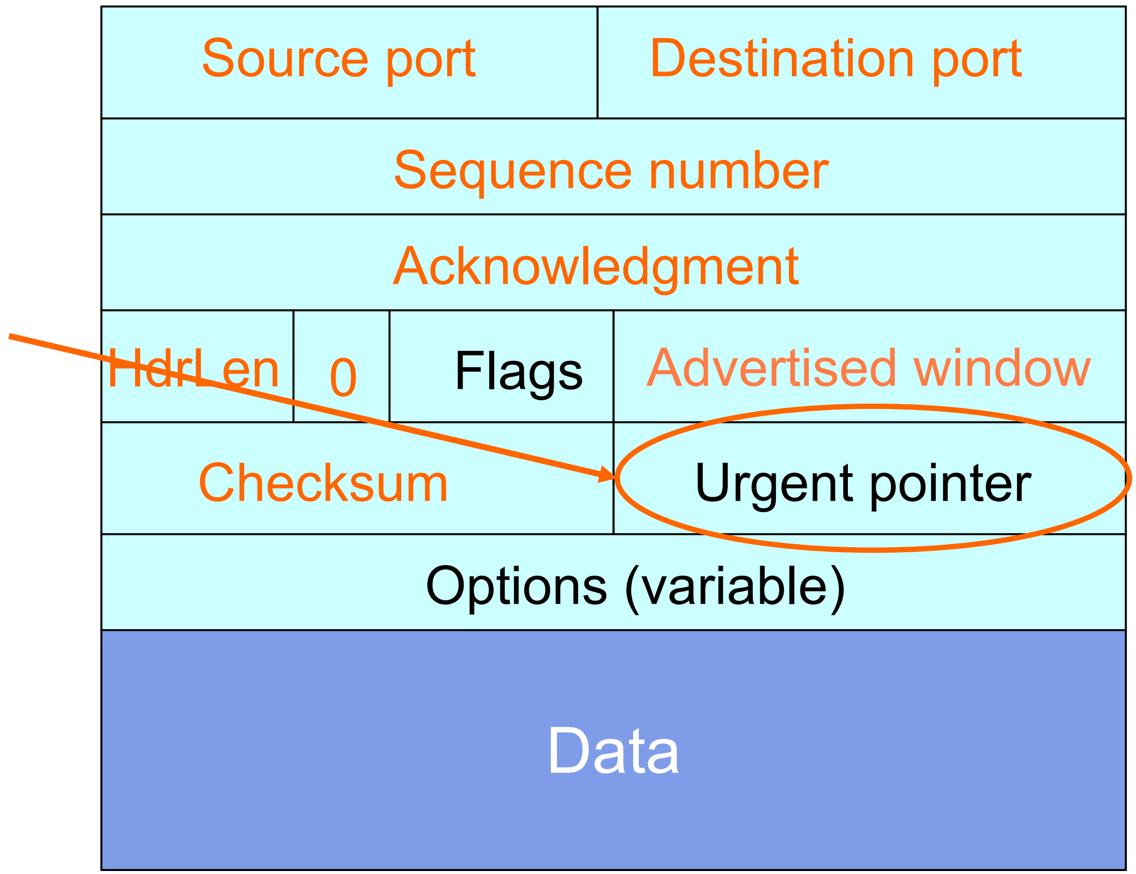
"Must Be Zero"
6 bits reserved

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



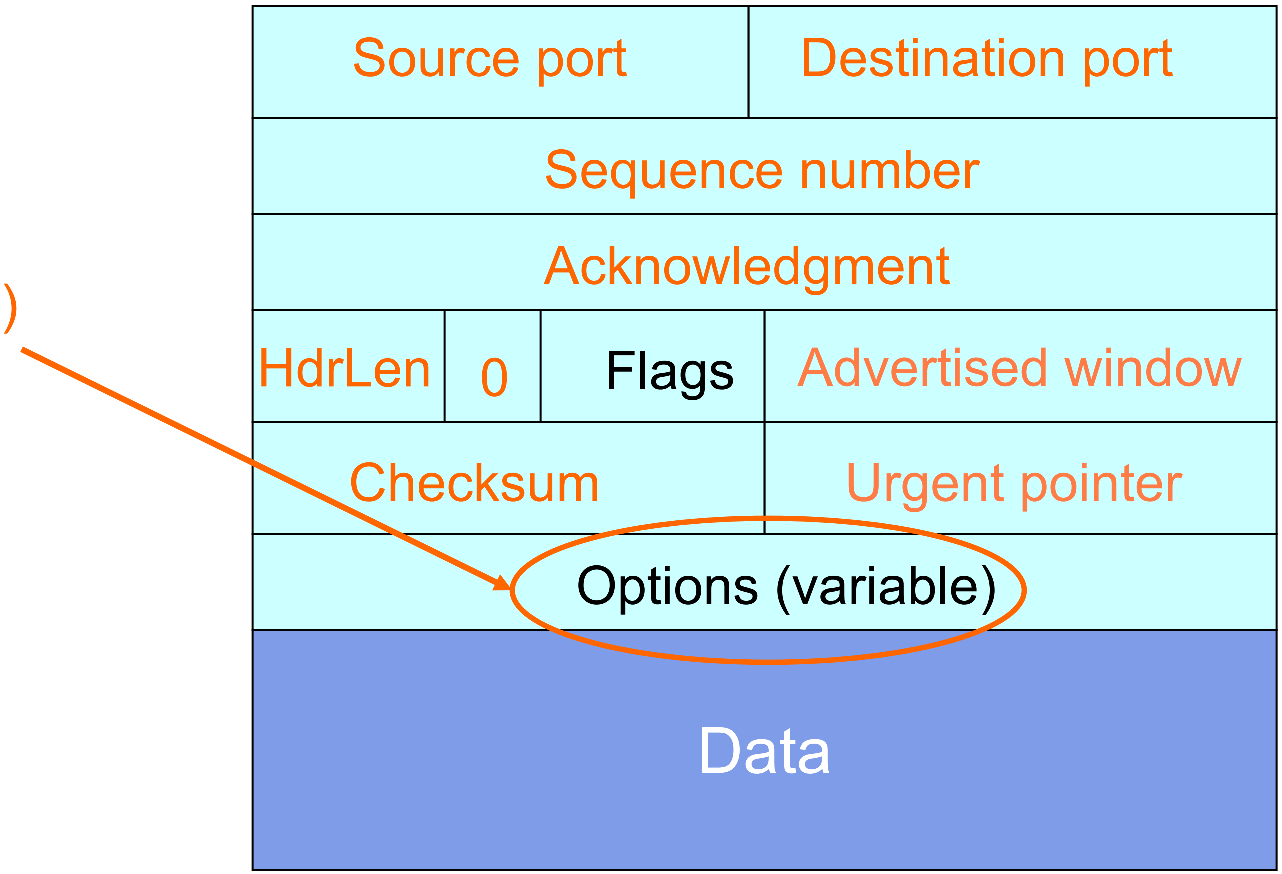
TCP Header: What's left?

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



TCP Header: What's left?

Options
(we'll ignore)



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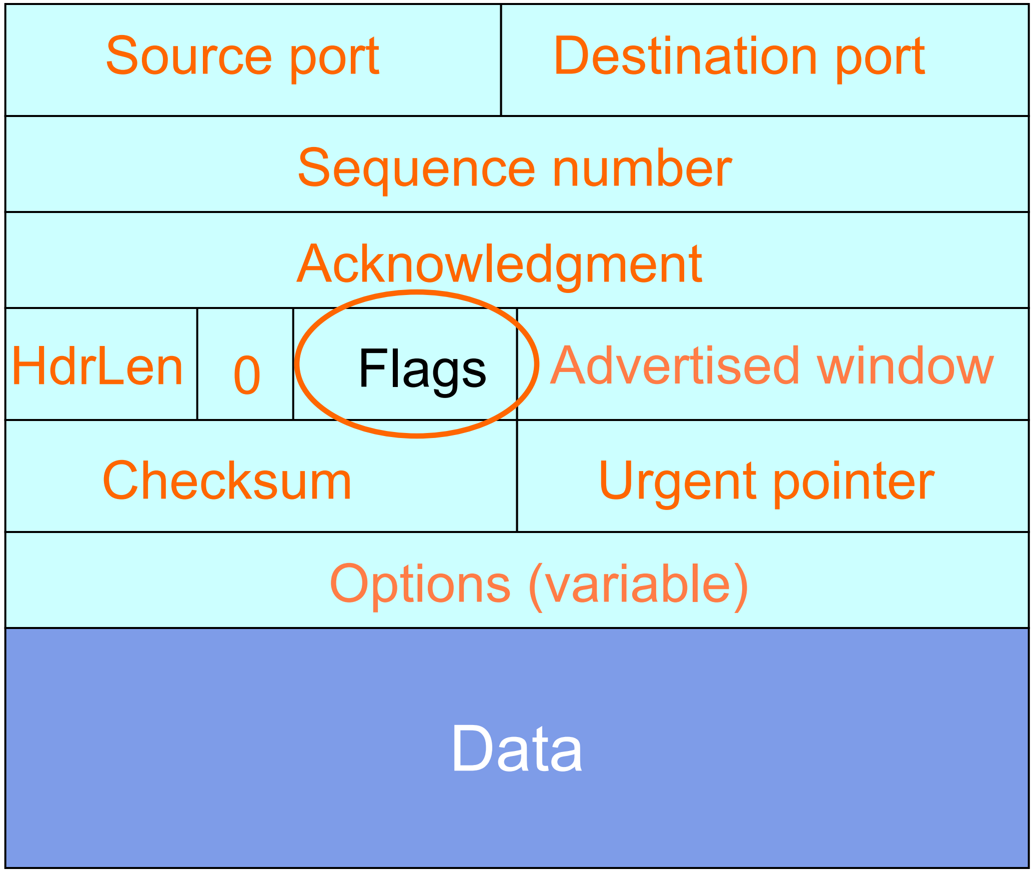

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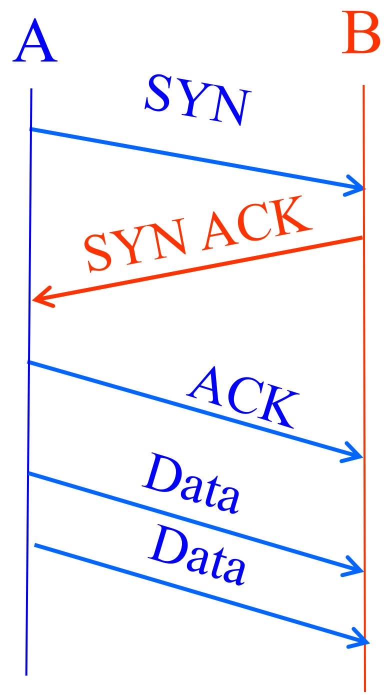
TCP Connection Establishment and Initial Sequence Numbers

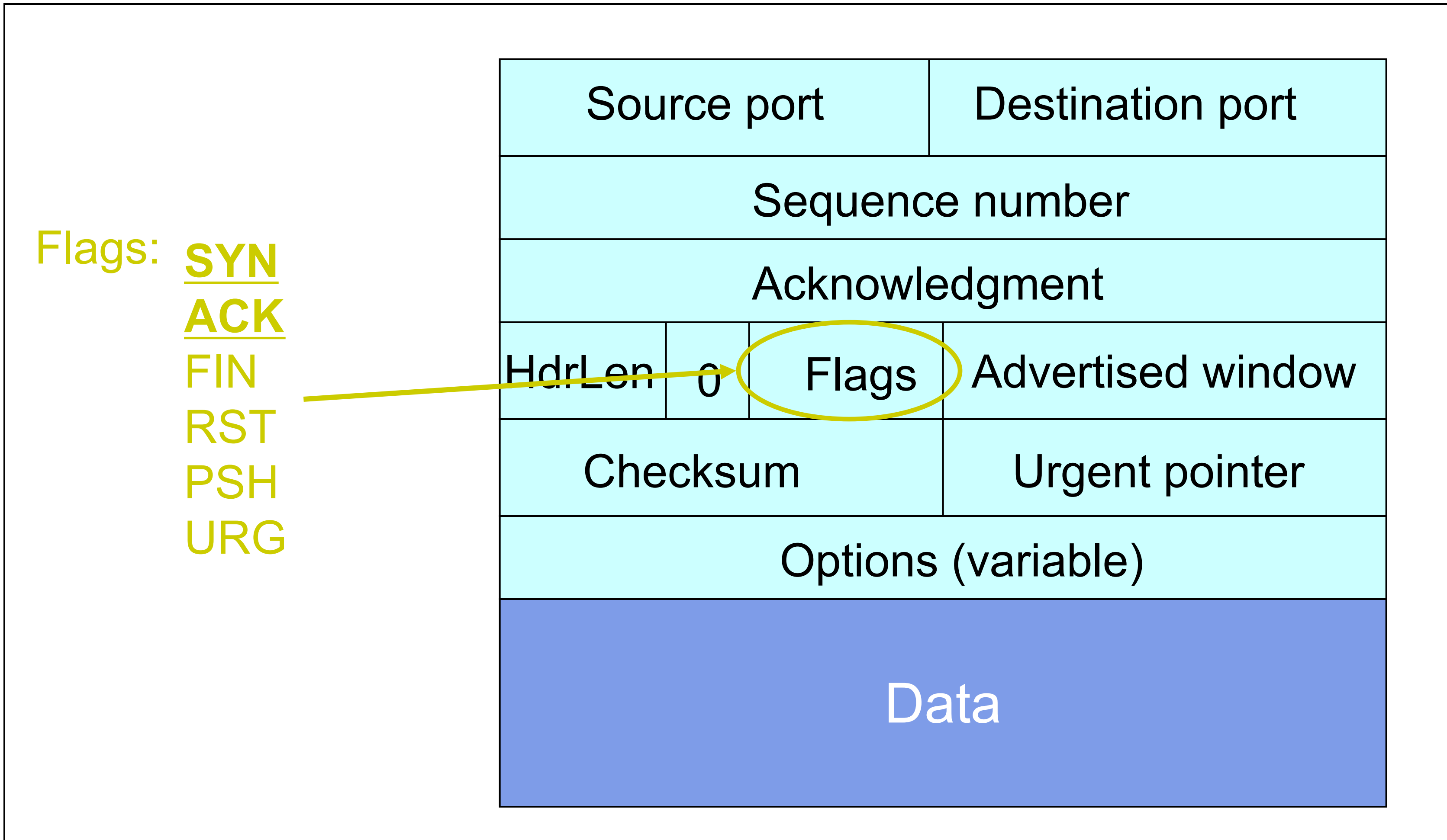
Establishing a TCP Connection



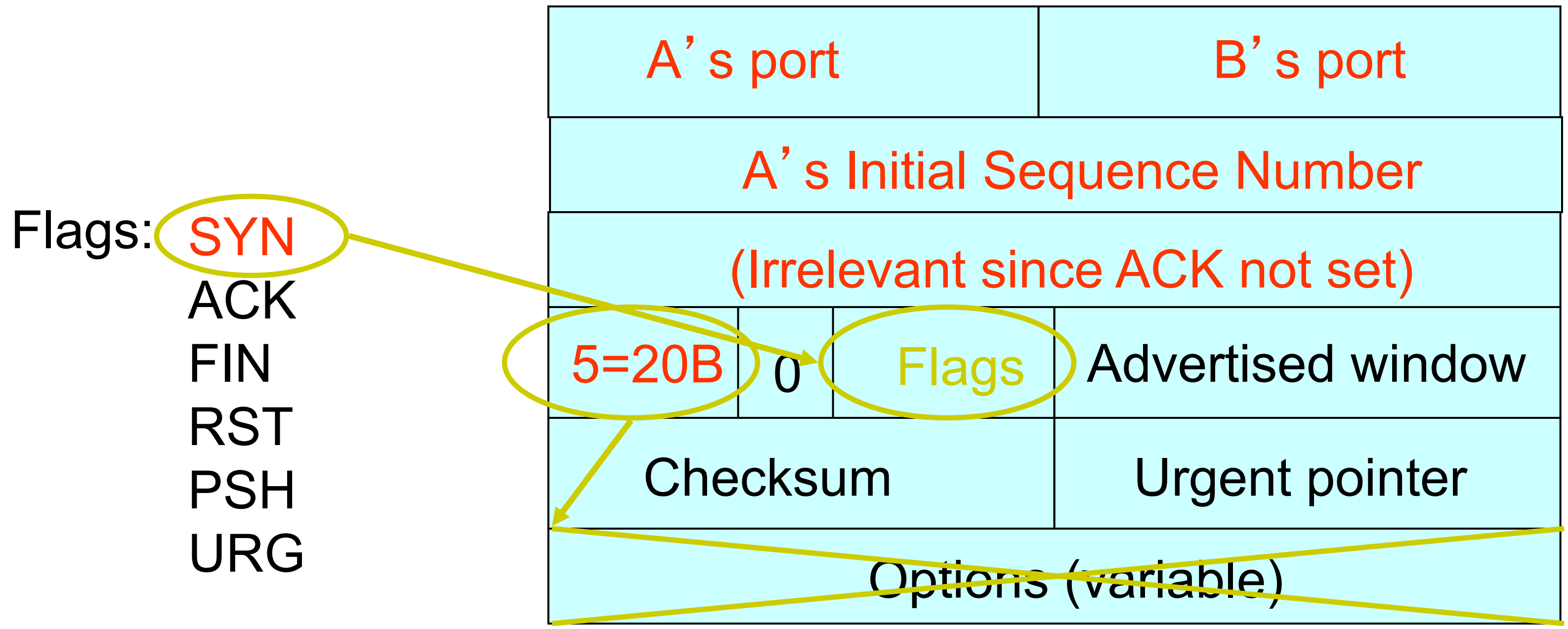
Each host tells its ISN to the other host.

- Three-way handshake to establish connection
  - Host A sends a **SYN** to host B
  - Host B returns a SYN acknowledgment (**SYN ACK**)
  - Host A sends an **ACK** to acknowledge the SYN ACK

TCP Header

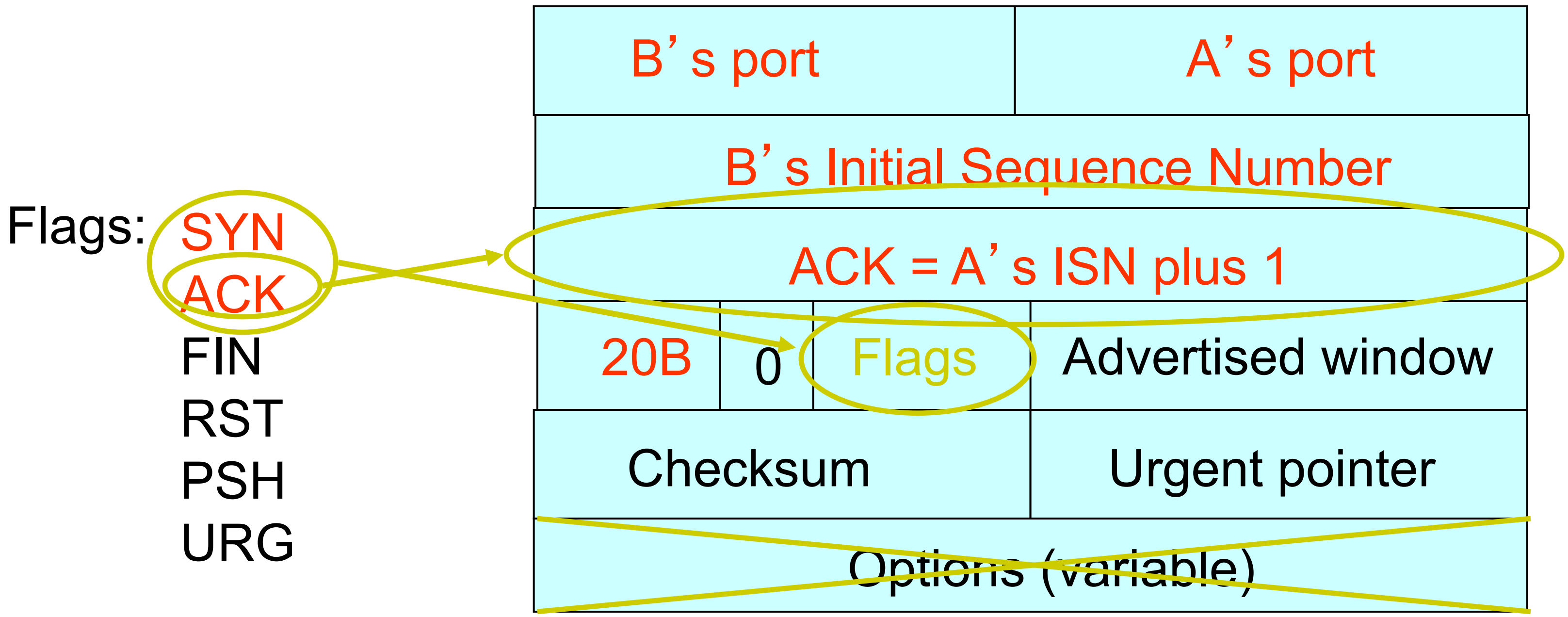

Flags:
SYN
ACK
FIN
RST
PSH
URG
Source port
Destination port
Sequence number
Acknowledgment
HdrLen
0
Flags
Advertised window
Checksum
Urgent pointer
Options (variable)
Data

Step 1: A's Initial SYN Packet



A tells B it wants to open a connection…

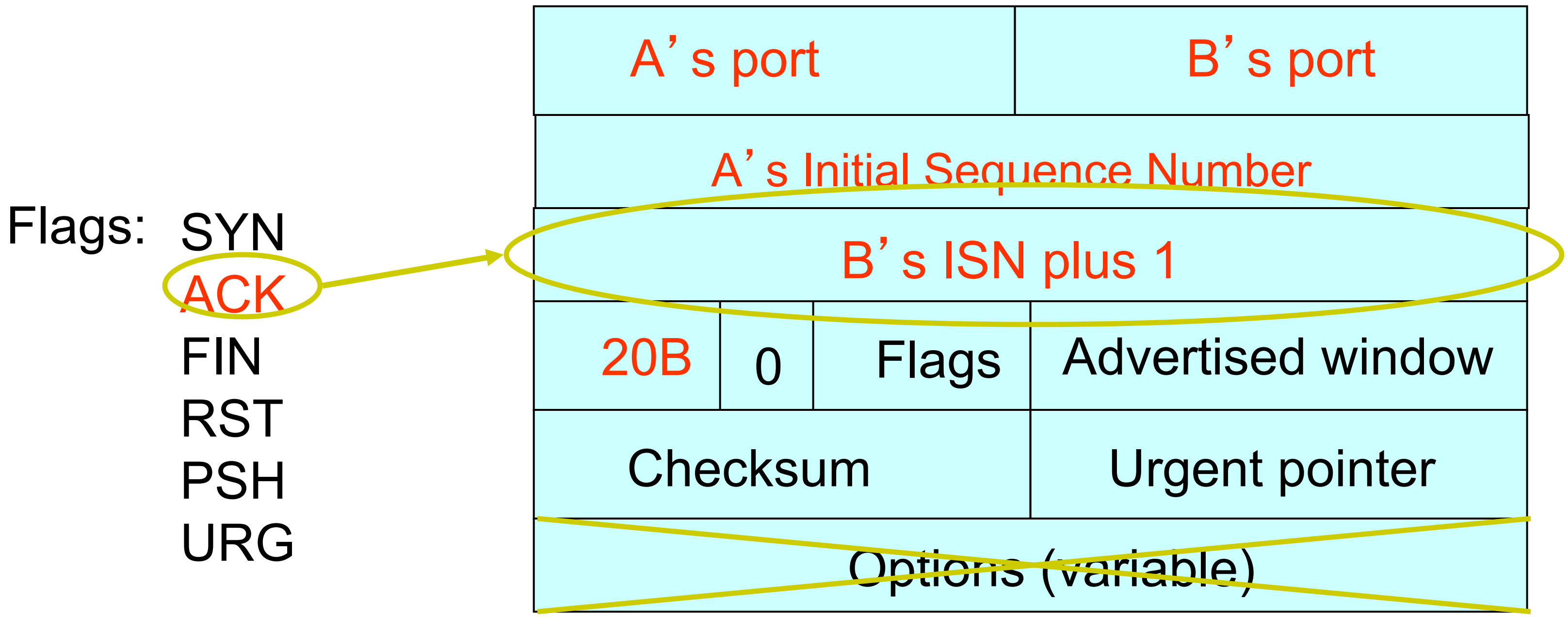
Step 2: B's SYN-ACK Packet



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

… upon receiving this packet, A can start sending data

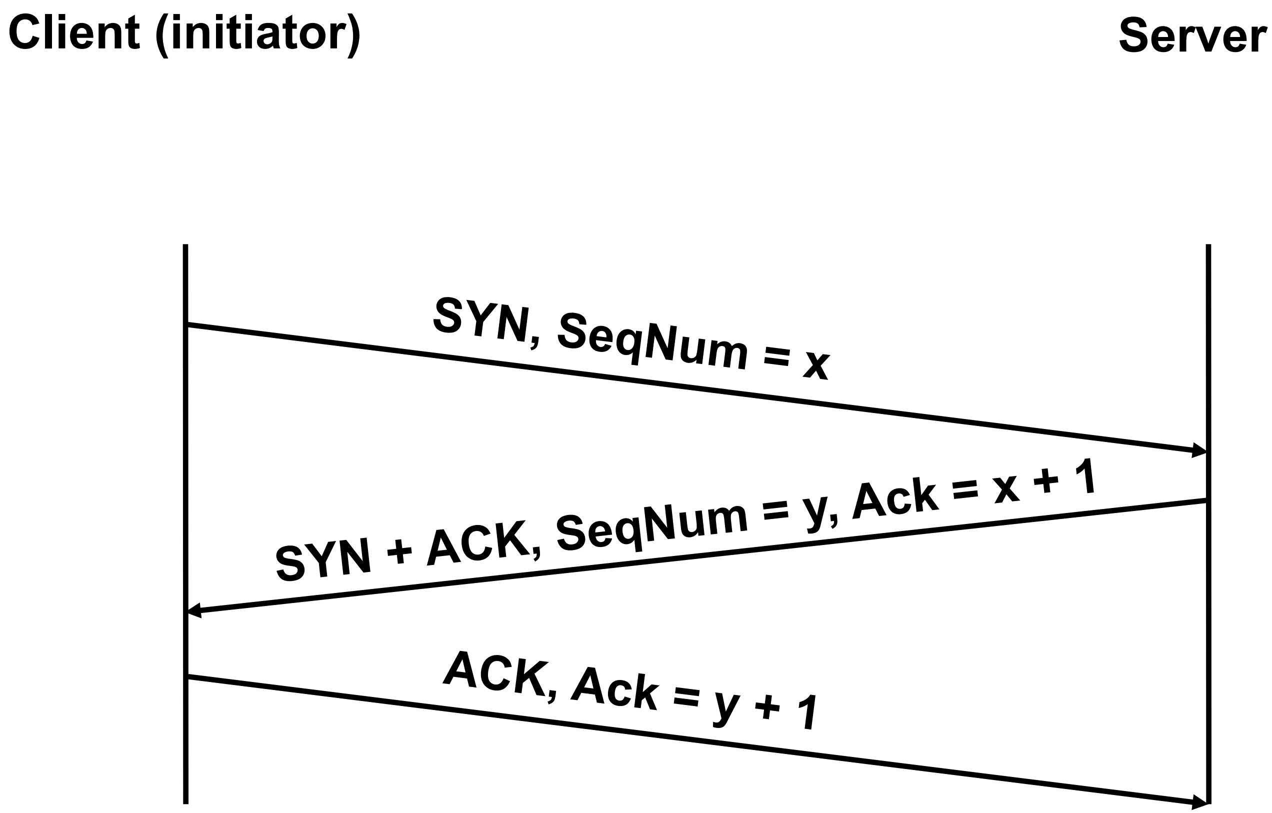
Step 3: A’s ACK of the SYN-ACK



A tells B it’s likewise okay to start sending

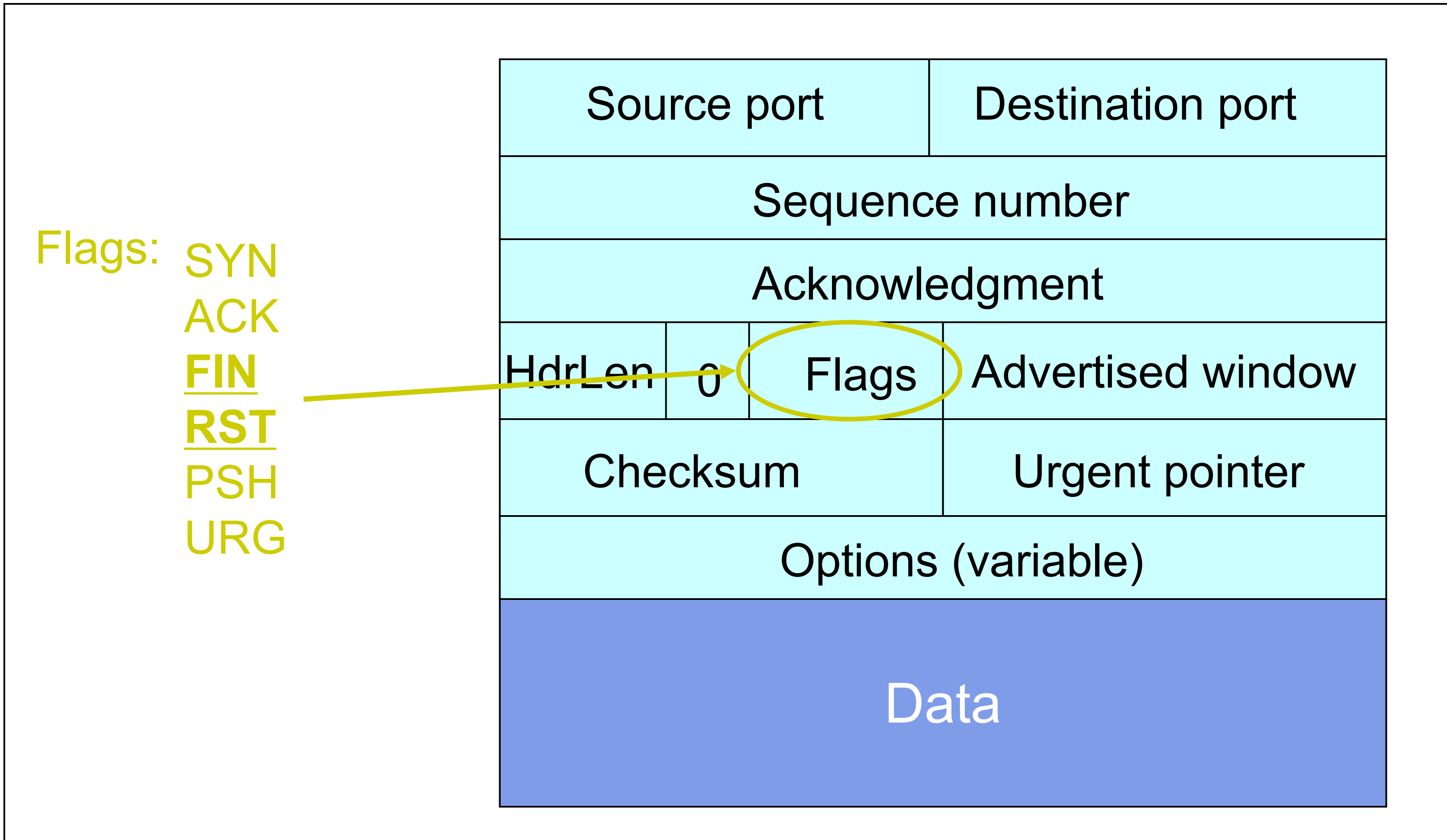
… upon receiving this packet, B can start sending data

Timing Diagram: 3-Way Handshaking



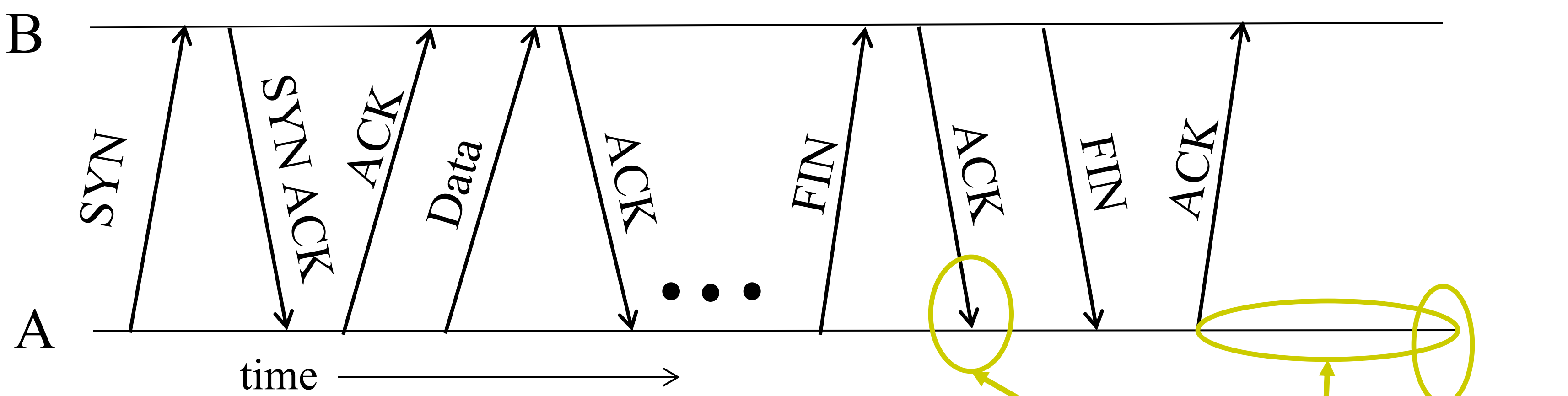
Tearing Down the Connection

TCP Header



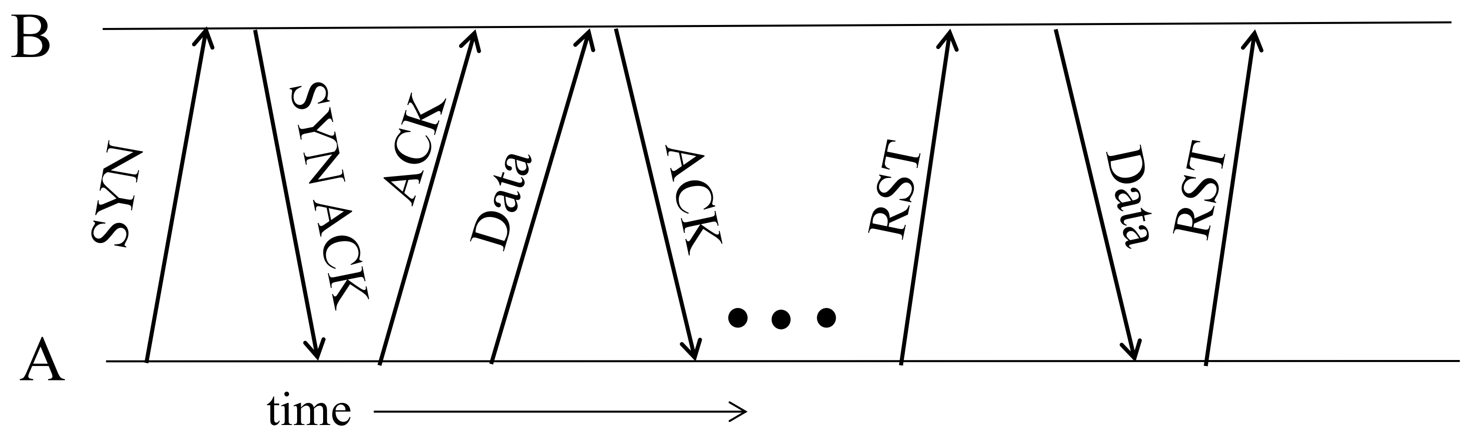
See /usr/include/netinet/tcp.h on Unix Systems

Normal Termination, One Side At A Time



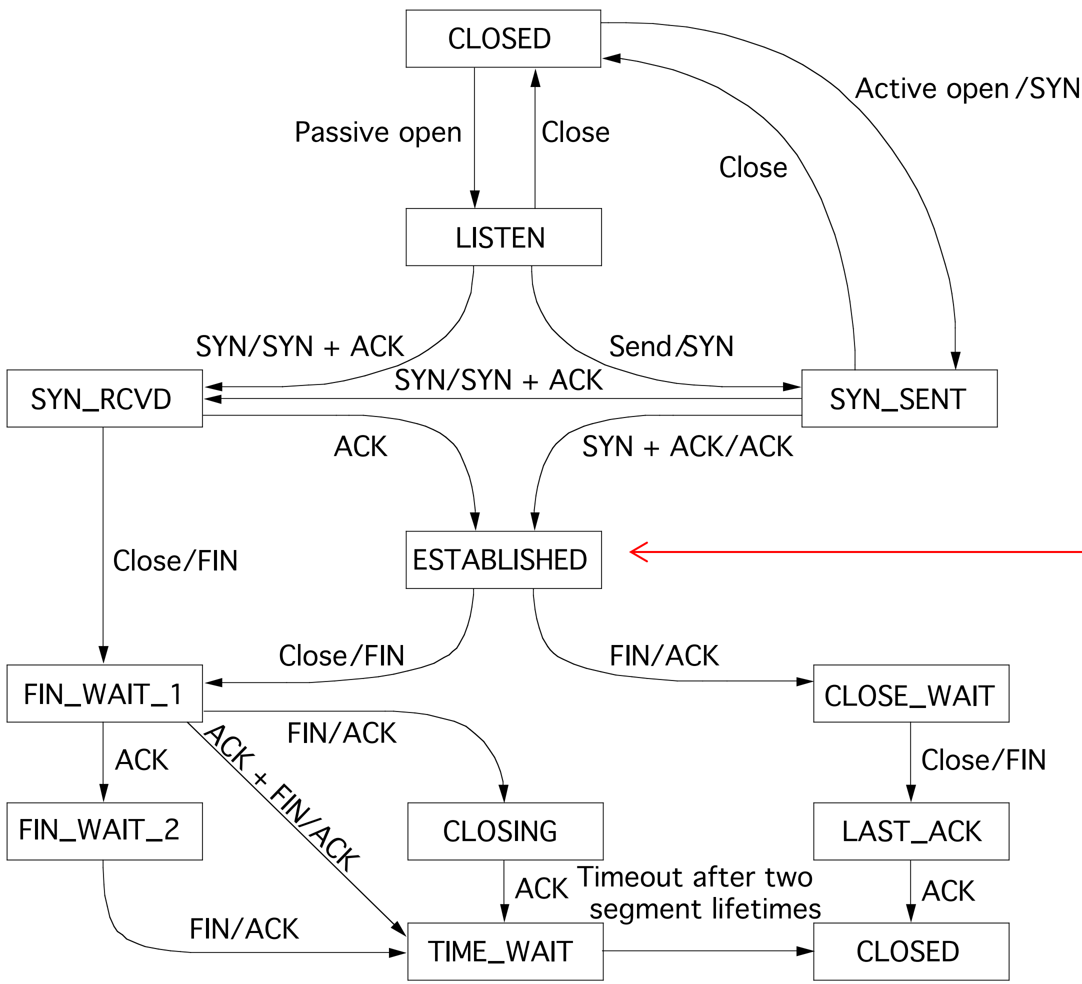
- Finish (**FIN**) to close connections
- Other host ack's
- Closes A's side of the connection, but not B's
  - Until B likewise sends a **FIN**
  - Which A then acks

Abrupt Termination



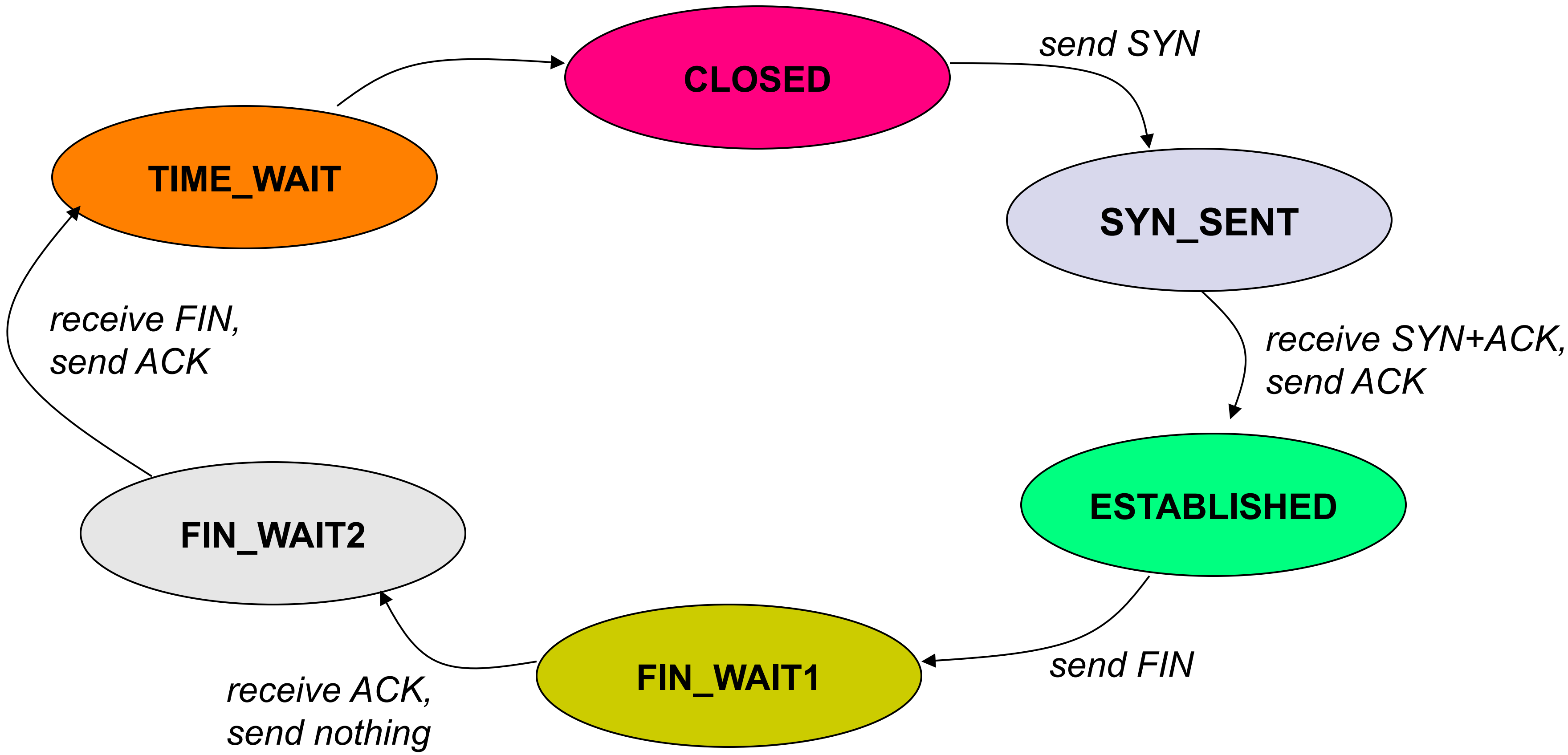
- A sends a RESET (**RST**) to B
  - E.g., because A restarted
- That's it
  - B does not ack the **RST**
  - Thus, **RST** is not delivered reliably
  - And: any data in flight is lost
  - If B sends anything more, will elicit another **RST**

TCP State Transitions



Data, ACK exchanges are in here

An Simpler View of the Client Side



In Summary

- **TCP**
  - An elegant (though not perfect) piece of engineering that has stood the test of time
    - Thought experiment: will TCP continue to be a good solution?
  - Plenty of evolution in individual pieces
    - **Congestion control**
    - Better acknowledgements, ISN selection, timer estimation, *etc.*
  - But core architectural decisions/abstractions remain
    - Bytestreams, connection oriented, windows etc.

- Next time: start on congestion control!

Any Questions?