CS4450

Computer Networks:
Architecture and Protocols

Lecture 21
TCP congestion control

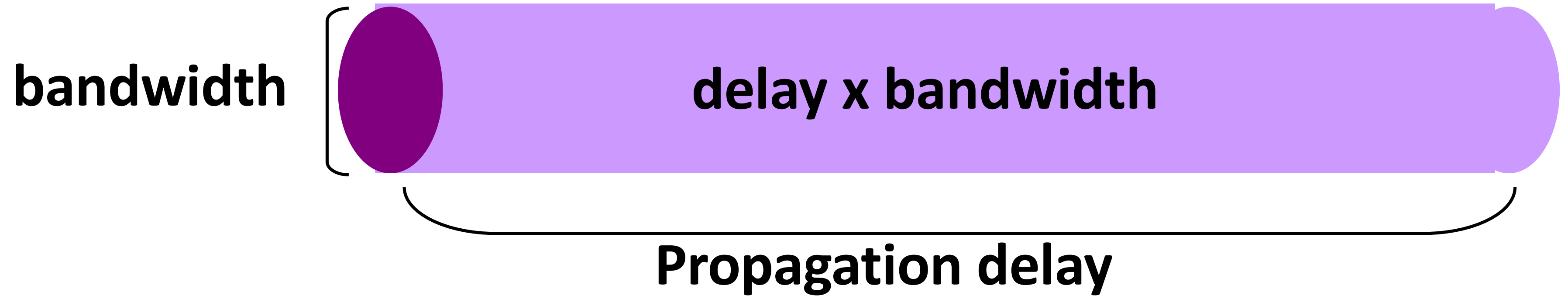
Rachit Agarwal

Recap: Reliable transport

- Started from first principles
    - Correctness condition for reliable transport
- … to understanding **why feedback from receiver is necessary** (sol-v1)
- … to understanding **why timers may be needed** (sol-v2)
- … to understanding **why window-based design may be needed** (sol-v3)
    - Allow many packets (W) in flight at once
    - And know what the ideal window size is
        - RTT x B / Packet size
- … to understanding **why cumulative ACKs may be a good idea**
- Very close to modern TCP

Recap: Ideal window size: RTT x B ~ W x Packet Size

- Recall that **Bandwidth Delay Product**
  - BDP = bandwidth x propagation delay



- **B x RTT is merely 2x BDP**

- Window sizing rule:
  - Total bits in flight is roughly the amount of data that fits into forward and reverse "pipes"
    - Here pipe is complete path, not single link…
    - **This is not "detail", this is a fundamental concept…**

Recap: Basic Components of TCP

- **Connections:** Explicit set-up and tear-down of TCP sessions/connections
- **Segments, Sequence numbers, ACKs**
  - TCP uses byte sequence numbers to identify payloads
  - ACKs referred to sequence numbers
  - Window sizes expressed in terms of # of bytes
- **Retransmissions**
  - Can’t be correct without retransmitting lost/corrupted data
  - TCP retransmits based on timeouts and duplicate ACKs
    - Timeouts based on estimate of RTT
- **Flow Control:** Ensures the sender does not overwhelm the receiver
- **Congestion Control:** Dynamic adaptation to network path’s capacity

TCP Congestion Control

TCP congestion control: high-level idea

- End hosts adjust sending rate
- Based on implicit feedback from the network
    - Implicit: router drops packets because its buffer overflows, not because it’s trying to send message
- Hosts probe network to test level of congestion
    - Speed up when no congestion (i.e., no packet drops)
    - Slow down when when congestion (i.e., packet drops)
- How to do this efficiently?
    - Extend TCP’s existing window-based protocol…
    - Adapt the window size based in response to congestion

All These Windows…

- **Flow control window**: Advertised Window (RWND)
  - How many bytes can be sent without overflowing receivers buffers
  - Determined by the receiver and reported to the sender
- **Congestion Window** (CWND)
  - How many bytes can be sent without overflowing routers
  - Computed by the sender using congestion control algorithm
- **Sender-side window** = minimum{CWND,RWND}
  - Assume for this lecture that RWND >> CWND

Note

- This lecture will talk about CWND in units of MSS
    - Recall MSS: Maximum Segment Size, the amount of payload data in a TCP packet
    - This is only for pedagogical purposes
- Keep in mind that real implementations maintain CWND in bytes

Basics of TCP Congestion

- Congestion Window (CWND)
    - Maximum # of unacknowledged bytes to have in flight
    - Rate ~CWND/RTT
- Adapting the congestion window
    - Increase upon lack of congestion: optimistic exploration
    - Decrease upon detecting congestion
- But how do you detect congestion?

Not All Losses the Same

- **Duplicate ACKs: isolated loss**
  - Still getting ACKs
- **Timeout: possible disaster**
  - Not enough duplicate ACKs
  - Must have suffered several losses

How to Adjust CWND?

- Consequences of over-sized window much worse than having an under-sized window
    - Over-sized window: packets dropped and retransmitted
    - Under-sized window: somewhat lower throughput
- Approach
    - Gentle increase when un-congested (exploration)
    - Rapid decrease when congested

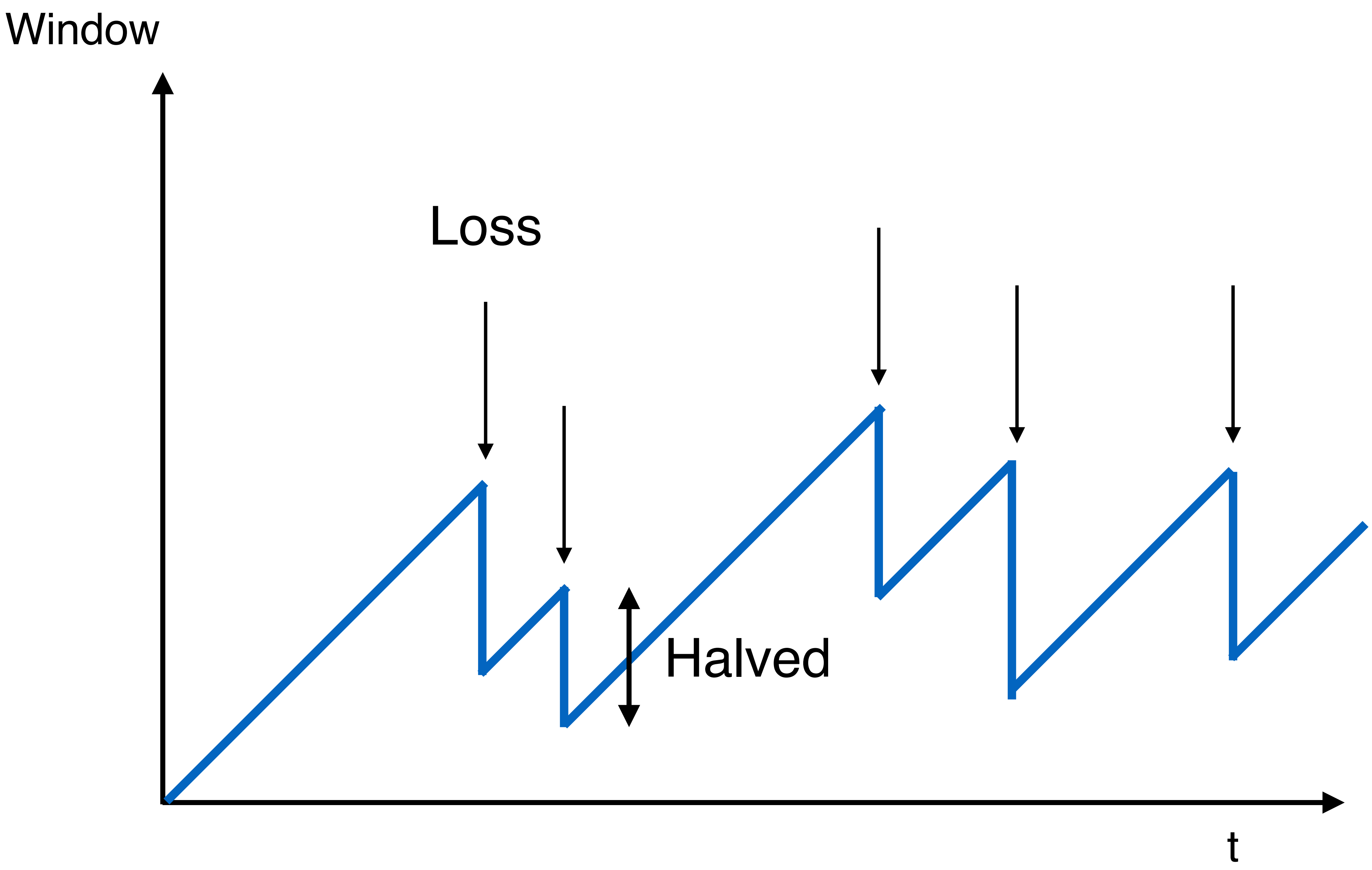
Additive Increase, Multiplicative Decrease (AIMD)

- Additive increase
  - On success of last window of data, increase by one MSS
  - If W packets in a row have been ACKed, increase W by one
  - i.e., +1/W per ACK
- Multiplicative decrease
  - On loss of packets by DupACKs, divide congestion window by half
  - Special case: when timeout, reduce congestion window to one MSS

AIMD

- ACK: CWND -> CWND + 1/CWND
  - When CWND is measured in MSS
  - Note: after a full window, CWND increase by 1 MSS
  - Thus, **CWND increases by 1 MSS per RTT**
- 3rd DupACK: CWND -> CWND/2
- Special case of timeout: CWND -> 1 MSS

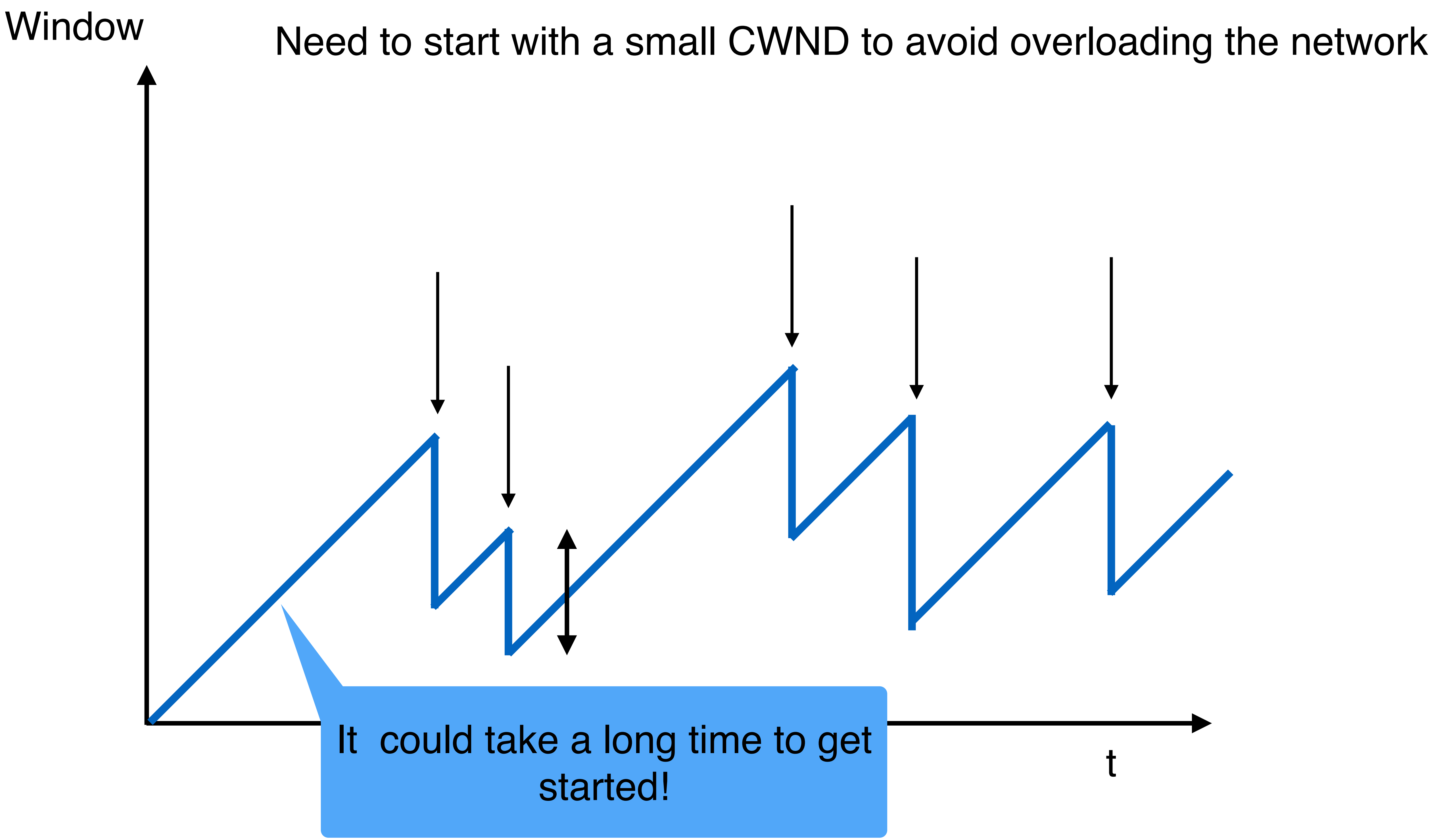
Leads to the TCP Sawtooth



Questions?

Slow Start

AIMD Starts Too Slowly



Bandwidth Discovery with Slow Start

- Goal: estimate available bandwidth
    - Start slow (for safety)
    - But ramp up quickly (for efficiency)

- Consider
    - RTT = 100ms, MSS=1000bytes
    - Window size to fill 1Mbps of BW = 12.5 MSS
    - Window size to fill 1 Gbps = 12,500 MSS
        - With just AIMD, it takes about 12500 RTTs to get to this window size!
        - ~21 mins

“Slow Start” Phase

- Start with a small congestion window
  - Initially, CWND is 1 MSS
  - So, initial sending rate is MSS/RTT

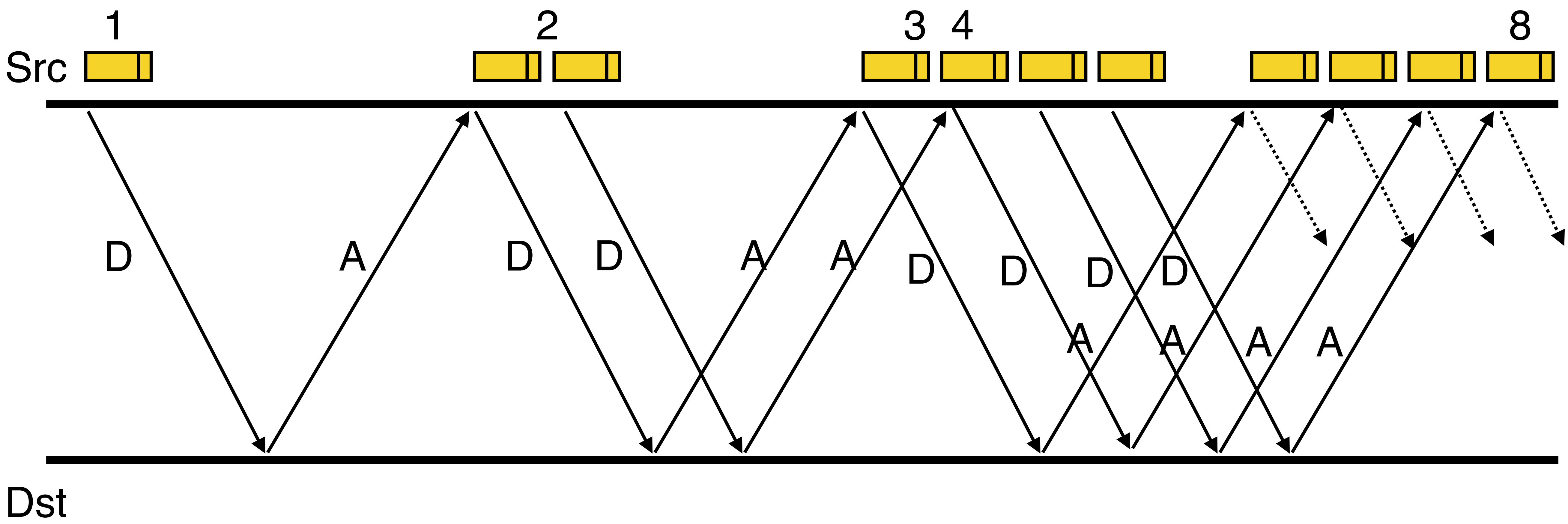
- That could be pretty wasteful
  - Might be much less than the actual bandwidth
  - Linear increase takes a long time to accelerate

- Slow-start phase (**actually “fast start”**)
  - Sender starts at a slow rate (hence the name)
  - … but increases exponentially until first loss

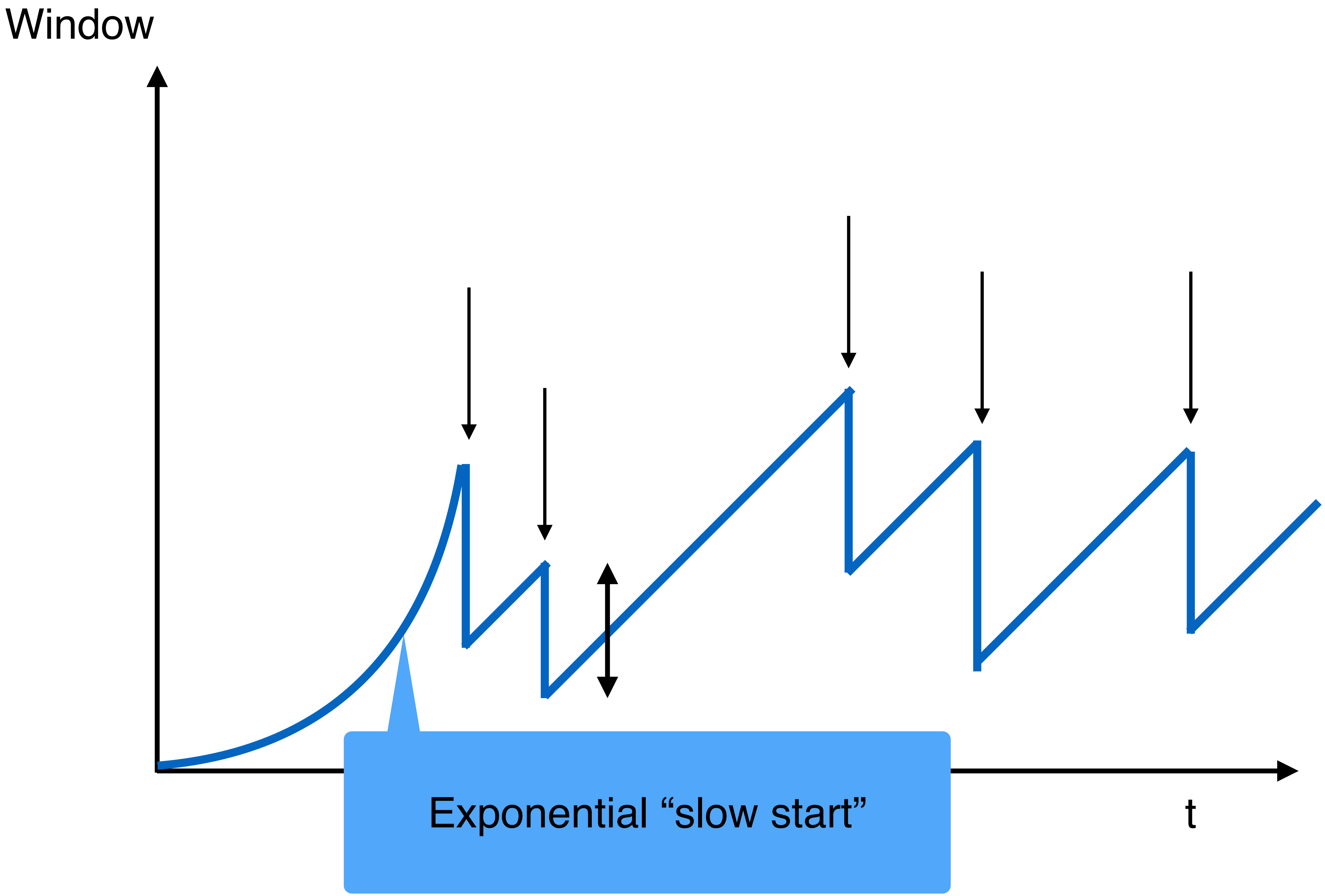
Slow Start in Action

Double CWND per round-trip time

Simple implementation: on each ACK, CWND += MSS



Slow Start and the TCP Sawtooth



Why is it called slow-start? Because TCP originally had no congestion control mechanism. The source would just start by sending a whole window's worth of data.

Slow-Start vs AIMD

- When does a sender stop Slow-Start and start Additive Increase?
- Introduce a “slow start threshold” (ssthresh)
  - Initialized to a large value
  - On timeout, ssthresh = CWND/2
- When CWND > ssthresh, sender switches from slow-start to AIMD-style increase

Timeouts

Loss Detected by Timeout

- Sender starts a timer that runs for RTO seconds
- **Restart timer whenever ACK for new data arrives**
- If timer expires
  - Set SSHTHRESH <- CWND/2 (“Slow Start Threshold”)
  - Set CWND <- 1 (MSS)
  - Retransmit **first** lost packet
  - Execute Slow Start until CWND > SSTHRESH
  - After which switch to Additive Increase

Summary of Increase

- "Slow start": increase CWND by 1 (MSS) for each ACK
    - A factor of 2 per RTT
- Leave slow-start regime when either:
    - CWND > SSTHRESH
    - Packet drop detected by dupacks
- Enter AIMD regime
    - Increase by 1 (MSS) for each window's worth of ACKed data

Summary of Decrease

- Cut CWND half on loss detected by dupacks
    - **Fast retransmit to avoid overreacting**
- Cut CWND all the way to 1 (MSS) on **timeout**
    - Set ssthresh to CWND/2
- Never drop CWND below 1 (MSS)
    - Our correctness condition: always try to make progress

TCP Congestion Control Details

Implementation

- State at sender
    - CWND (initialized to a small constant)
    - ssthresh (initialized to a large constant)
    - dupACKcount
    - Timer, as before
- Events at sender
    - ACK (new data)
    - dupACK (duplicate ACK for old data)
    - Timeout
- What about receiver? Just send ACKs upon arrival

Event: ACK (new data)

- If in slow start
  - CWND += 1

CWND packets per RTT
Hence after one RTT with no drops:
CWND = 2 x CWND

Event: ACK (new data)

- If CWND <= ssthresh
  - CWND += 1
- Else
  - CWND = CWND + 1/CWND

Slow Start Phase

Congestion Avoidance Phase

(additive increase)

CWND packets per RTT
Hence after one RTT with no drops:
CWND = CWND + 1

Event: Timeout

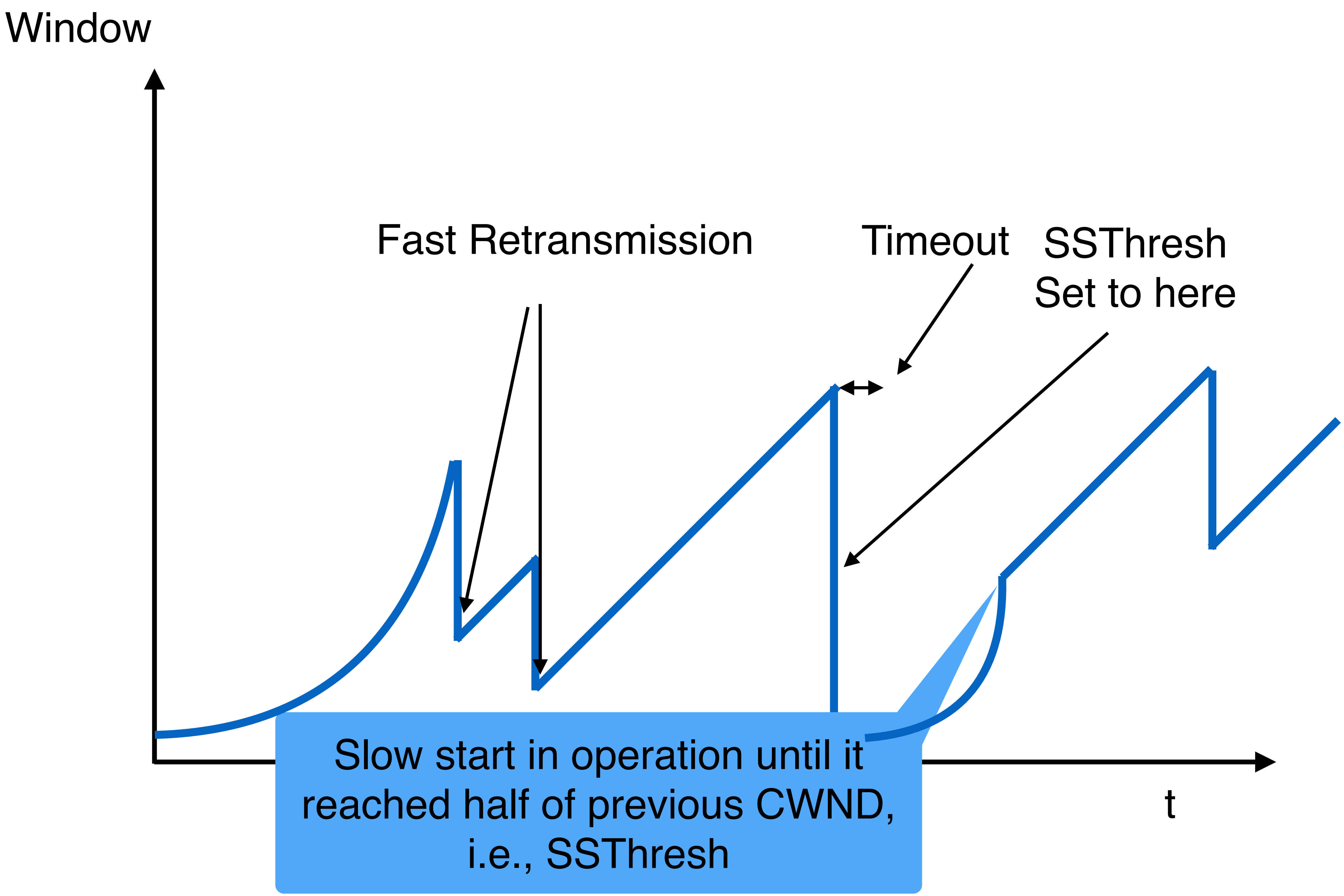
- On Timeout
  - ssthresh <- CWND/2
  - CWND <- 1

Event: dupACK

- dupACKcount++
- If dupACKcount = 3 /* fast retransmit */
    - ssthresh <- CWND/2
    - CWND <- CWND/2

Remains in congestion avoidance after fast retransmission

Time Diagram



Slow-start restart: Go back to CWND of 1 MSS, but take advantage of knowing the previous value of CWND.

TCP Flavors

- TCP Tahoe
  - CWND = 1 on triple dupACK
- TCP Reno
  - CWND = 1 on timeout
  - CWND = CWND/2 on triple dupACK
- TCP-newReno
  - TCP-Reno + improved fast recovery
- TCP-SACK
  - Incorporates selective acknowledgements

Our default assumption

TCP and fairness guarantees

Consider A Simple Model

- Flows **ask** for an amount of bandwidth r_i
    - In reality, this request is implicit (the amount they send)
- The link gives them an amount a_i
    - Again, this is implicit (by how much is forwarded)
    - $a_i <= r_i$
- There is some total capacity C
    - Sum $a_i <= C$

Fairness

- When all flows want the same rate, fair is easy
  - Fair share = C/N
  - C = capacity of link
  - N = number of flows
- Note:
  - This is fair share per link. This is not a global fair share
- When not all flows have the same demand?
  - What happens here?

Example 1

- Requests: r_i Allocations: a_i
- $C = 20$
  - Requests: $r_1 = 6$, $r_2 = 5$, $r_3 = 4$
- Solution
  - $a_1 = 6$, $a_2 = 5$, $a_3 = 4$
- When bandwidth is plentiful, everyone gets their request
- This is the easy case

Example 2

- Requests: r_i Allocations: a_i
- $C = 12$
  - Requests: $r_1 = 6$, $r_2 = 5$, $r_3 = 4$
- One solution
  - $a_1 = 4$, $a_2 = 4$, $a_3 = 4$
  - Everyone gets the same
- Why not proportional to their demands?
  - $a_i = (12/15)\ r_i$
- Asking for more gets you more!
  - Not incentive compatible (i.e., cheating works!)
  - You can't have that and invite innovation!

Example 3

- Requests: r_i Allocations: a_i
- $C = 14$
  - Requests: $r_1 = 6$, $r_2 = 5$, $r_3 = 4$
- $a_3 = 4$ (can't give more than a flow wants)
- Remaining bandwidth is 10, with demands 6 and 5
  - From previous example, if both want more than their share, they both get half
  - $a_1 = a_2 = 5$

Max-Min Fairness

- Given a set of bandwidth demands r_i and total bandwidth C, max-min bandwidth allocations are $a_i = \min(f, r_i)$
  - Where f is the unique value such that $\mathrm{Sum}(a_i) = C$ or set f to be infinite if no such value exists
- **This is what round-robin service gives**
  - If all packets are MTU
- Property:
  - If you don't get full demand, no one gets more than you

Computing Max-Min Fairness

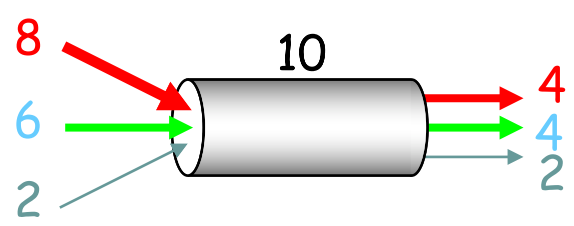
- Assume demands are in increasing order…
- If $C/N <= r_1$, then $a_i = C/N$ for all i
- Else, $a_1 = r1$, set $C = C - a_1$ and $N = N-1$
- Repeat
- Intuition: all flows requesting less than fair share get their request. Remaining flows divide equally

Example

- Assume link speed C is 10Mbps
- Have three flows:
    - Flow 1 is sending at a rate 8 Mbps
    - Flow 2 is sending at a rate 6 Mbps
    - Flow 3 is sending at a rate 2 Mbps
- How much bandwidth should each get?
    - According to max-min fairness?
- Work this out, talk to your neighbors

Example

- Requests: r_i Allocations: a_i
- Requests: $r_1 = 8$, $r_2 = 6$, $r_3 = 2$
- $C = 10$, $N = 3$, $C/N = 3.33$
  - Can serve all for r_3
  - Remove r_3 from the accounting: $C = C - r_3 = 8$, $N = 2$
- $C/2 = 4$
  - Can't service all for r_1 or r_2
  - So hold them to the remaining fair share: $f = 4$

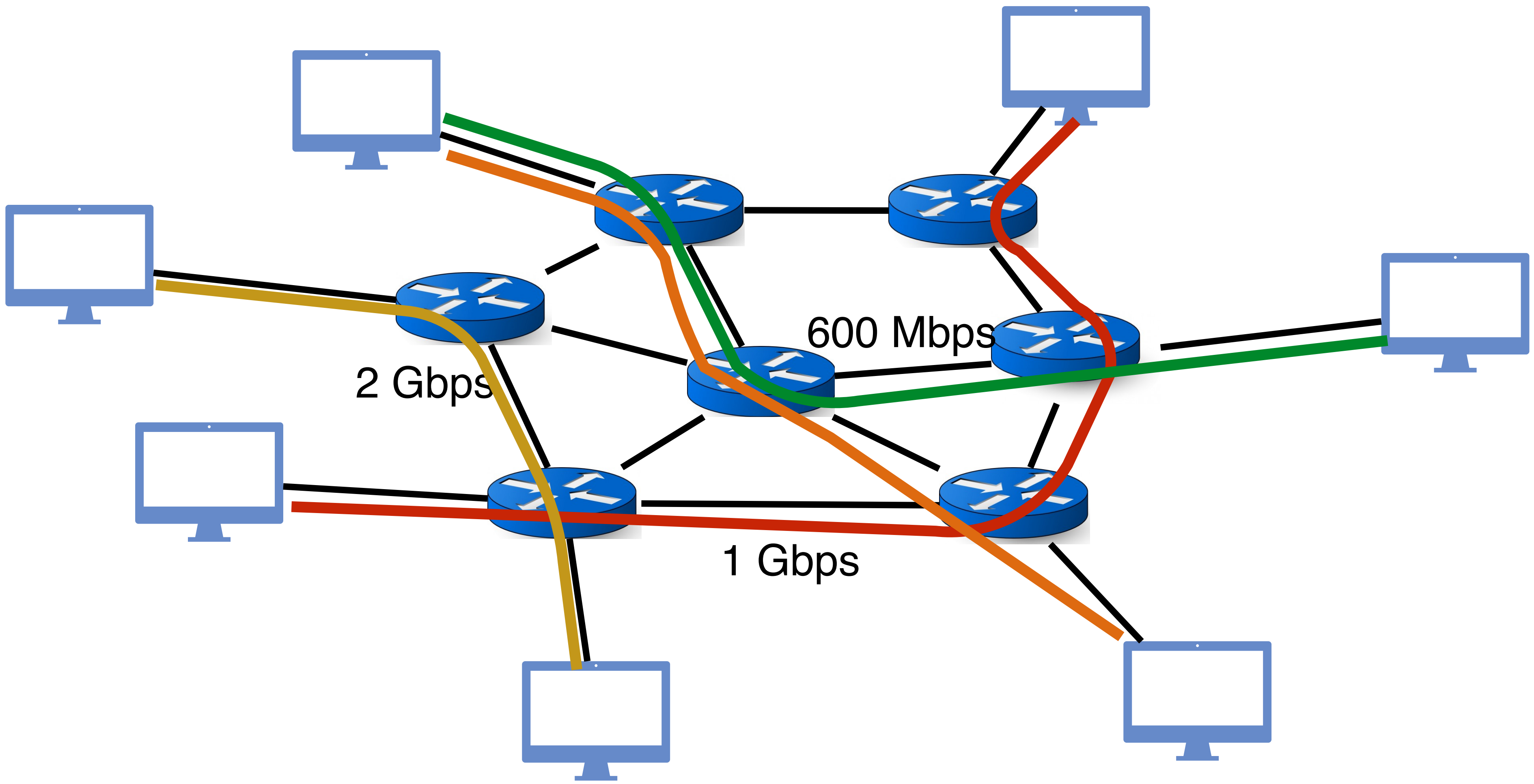


$f = 4$:
min(8, 4) = 4
min(6, 4) = 4
min(2, 4) = 2

Max-Min Fairness

- Max-min fairness the natural per-link fairness
- Only one that is
  - Symmetric
  - Incentive compatible (asking for more doesn’t help)

Reality of Congestion Control



Congestion control is a resource allocation problem involving many flows, many links and complicated global dynamics

Classical result:

In a stable state

(no dynamics; all flows are infinitely long; no failures; etc.)

TCP guarantees max-min fairness

Any Questions?

Back up slides on UDP

(not needed for exams)

UDP: User Datagram Protocol

- Lightweight communication between processes
    - Avoid overhead and delays of ordered, reliable delivery
    - Send messages to and receive 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”
    - (local port, local IP, remote port, remote IP) <—> socket

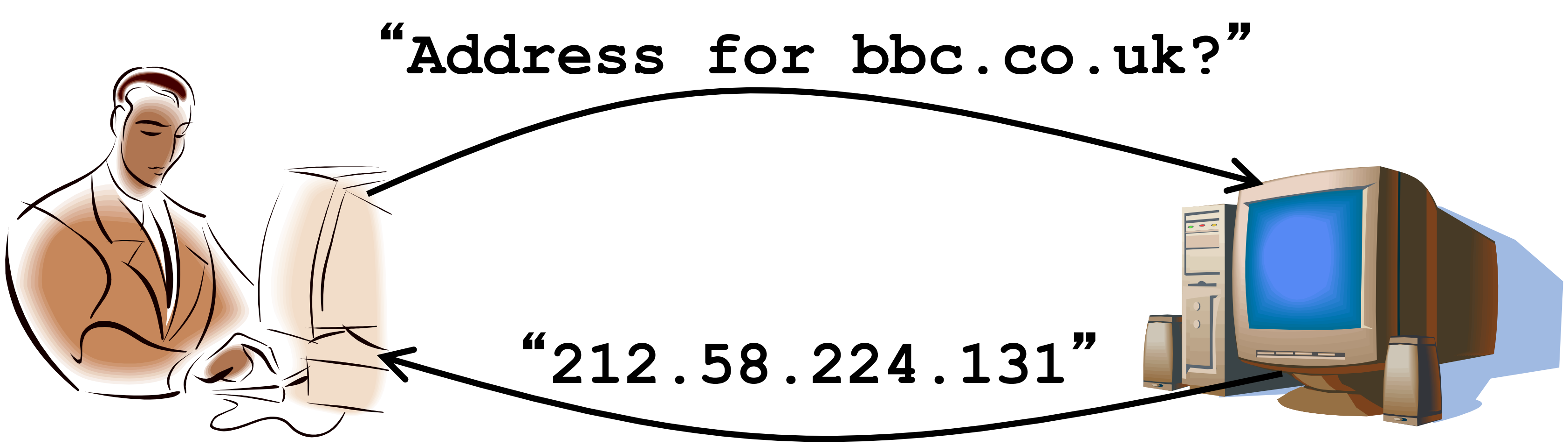
Source Port #	Dest Port #
Checksum	Length
Application Data (Message)	

Question

- Why do UDP packets carry sender’s port?

Popular Applications That Use UDP

- Some interactive streaming apps
  - Retransmitting lost/corrupted packets is often pointless — by the time the packet is transmitted, it's too late
  - E.g., telephone calls, video conferencing, gaming
  - Modern streaming protocols using TCP (and HTTP)
- Simple query protocols like Domain Name System
  - Connection establishment overhead would double cost
  - Easier to have application retransmit if needed



Back up slides on TCP

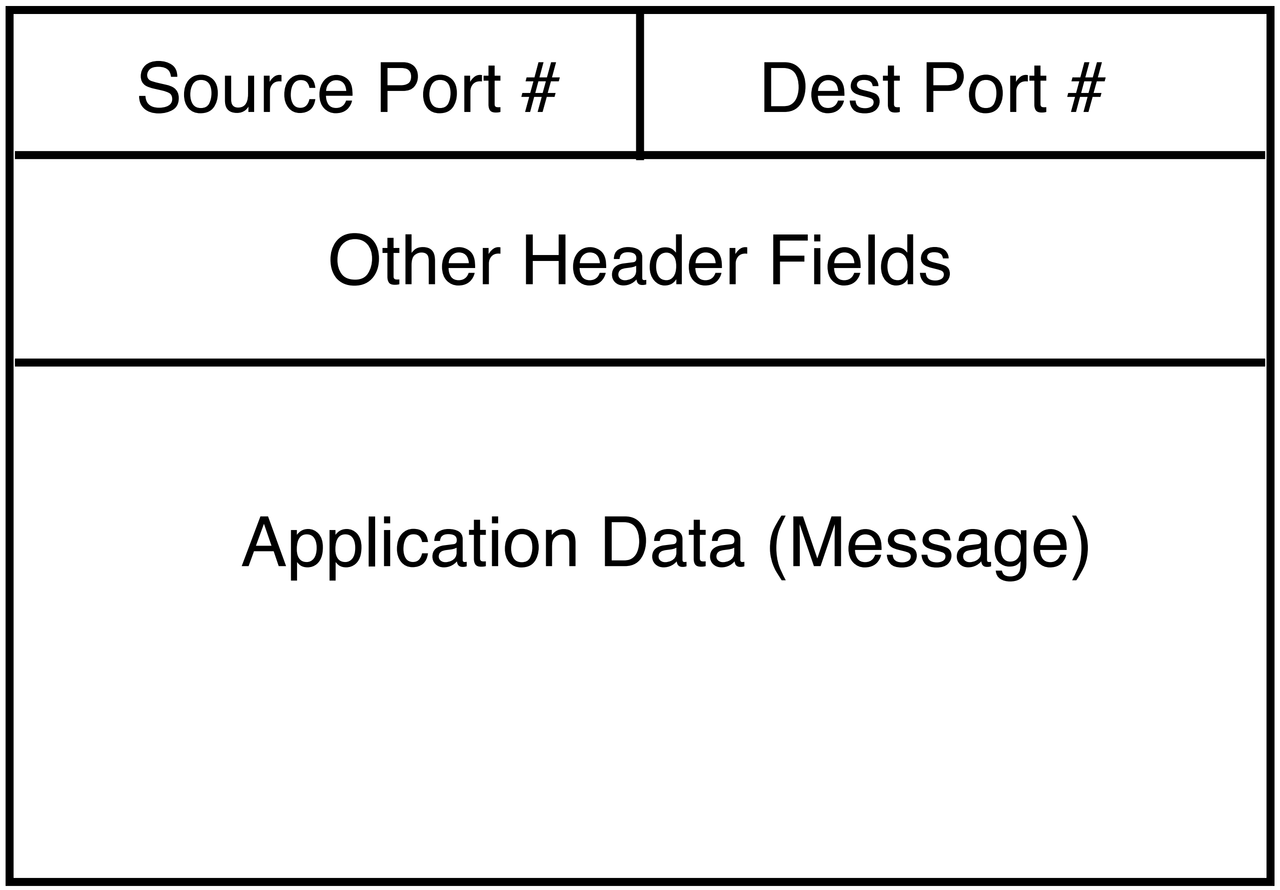
(not needed for exams)

Ports

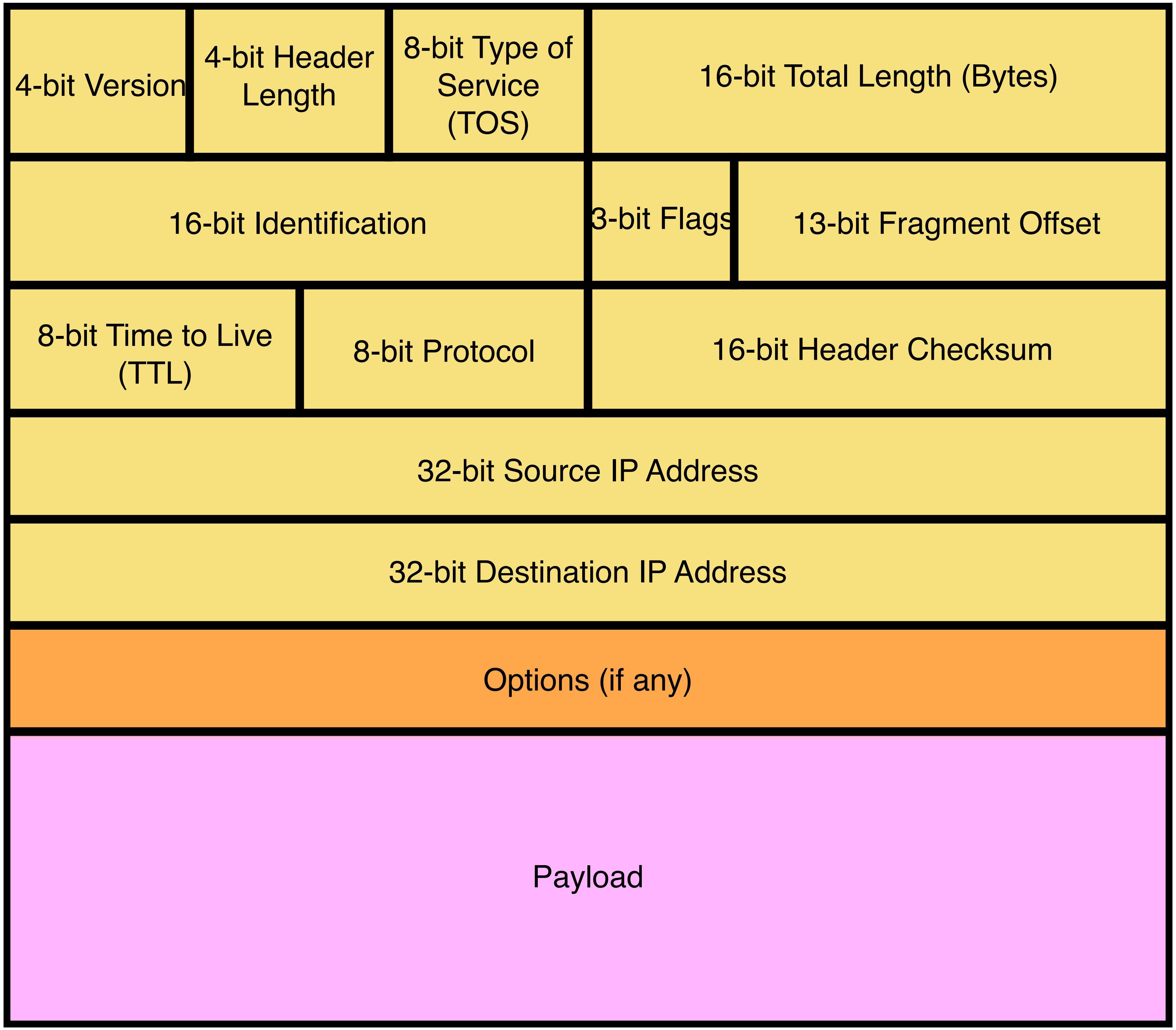
- Separate 16-bit port address space for UDP, TCP
- “Well known” ports (0-1023)
  - Agreement on which services run on these ports
  - e.g., ssh:22, http:80
  - Client (app) knows appropriate port on sender
  - Services can listen on well-known ports

Multiplexing and Demultiplexing

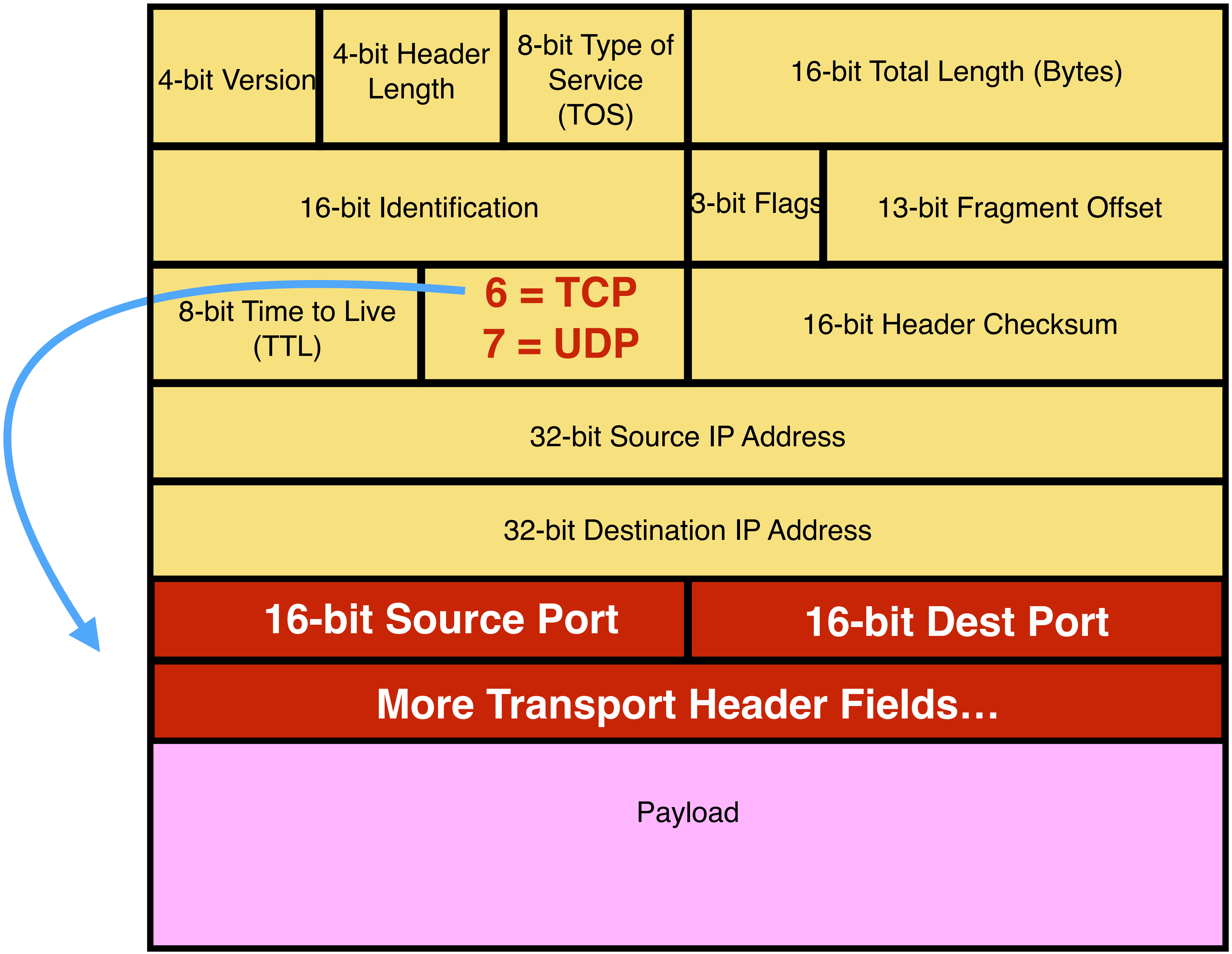
- Host receives IP datagrams
  - Each datagram has source and destination IP address
  - Each segment has source and destination port number
- Host uses IP address and port numbers to direct the segment to appropriate socket



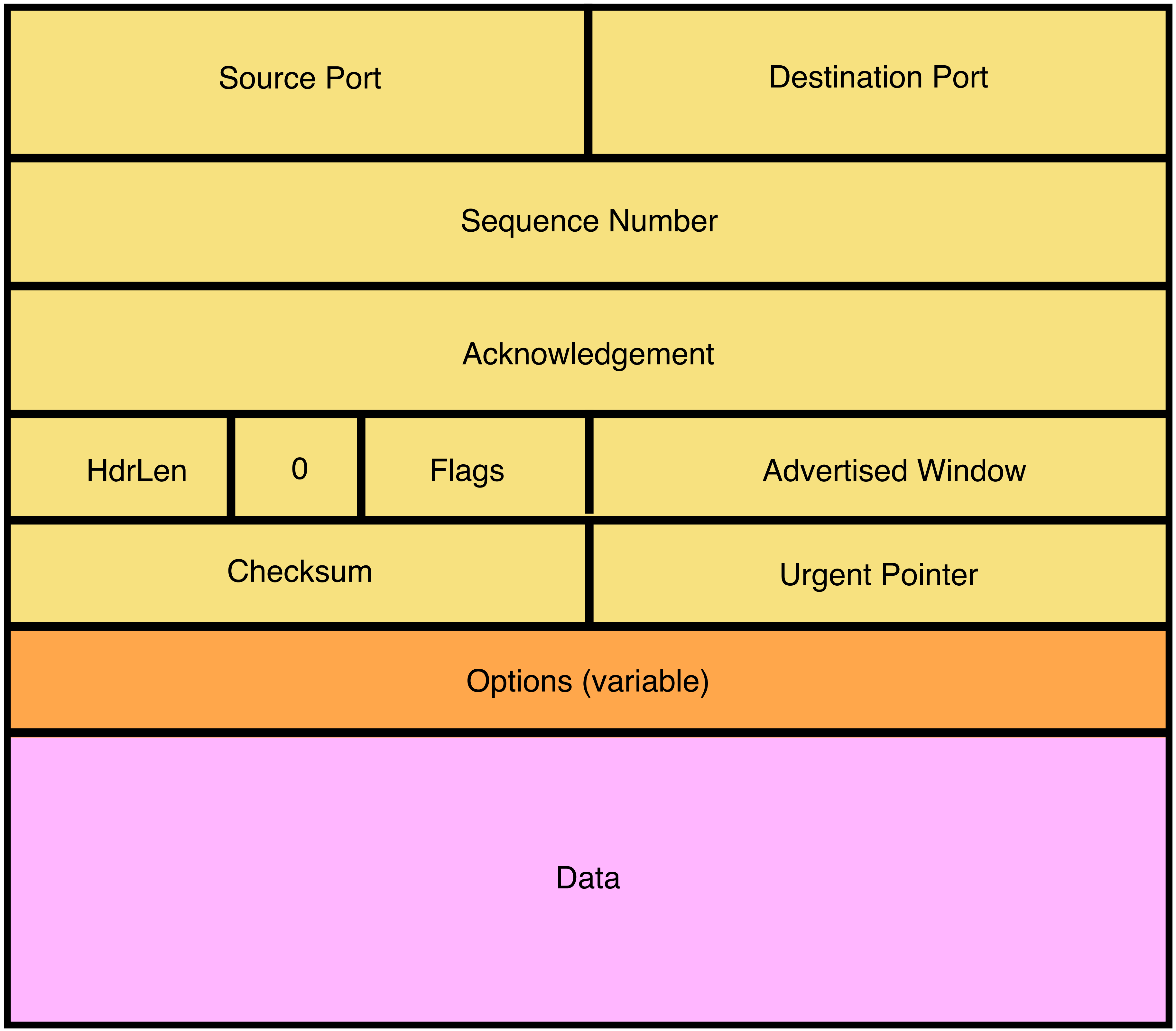
IP Packet Structure



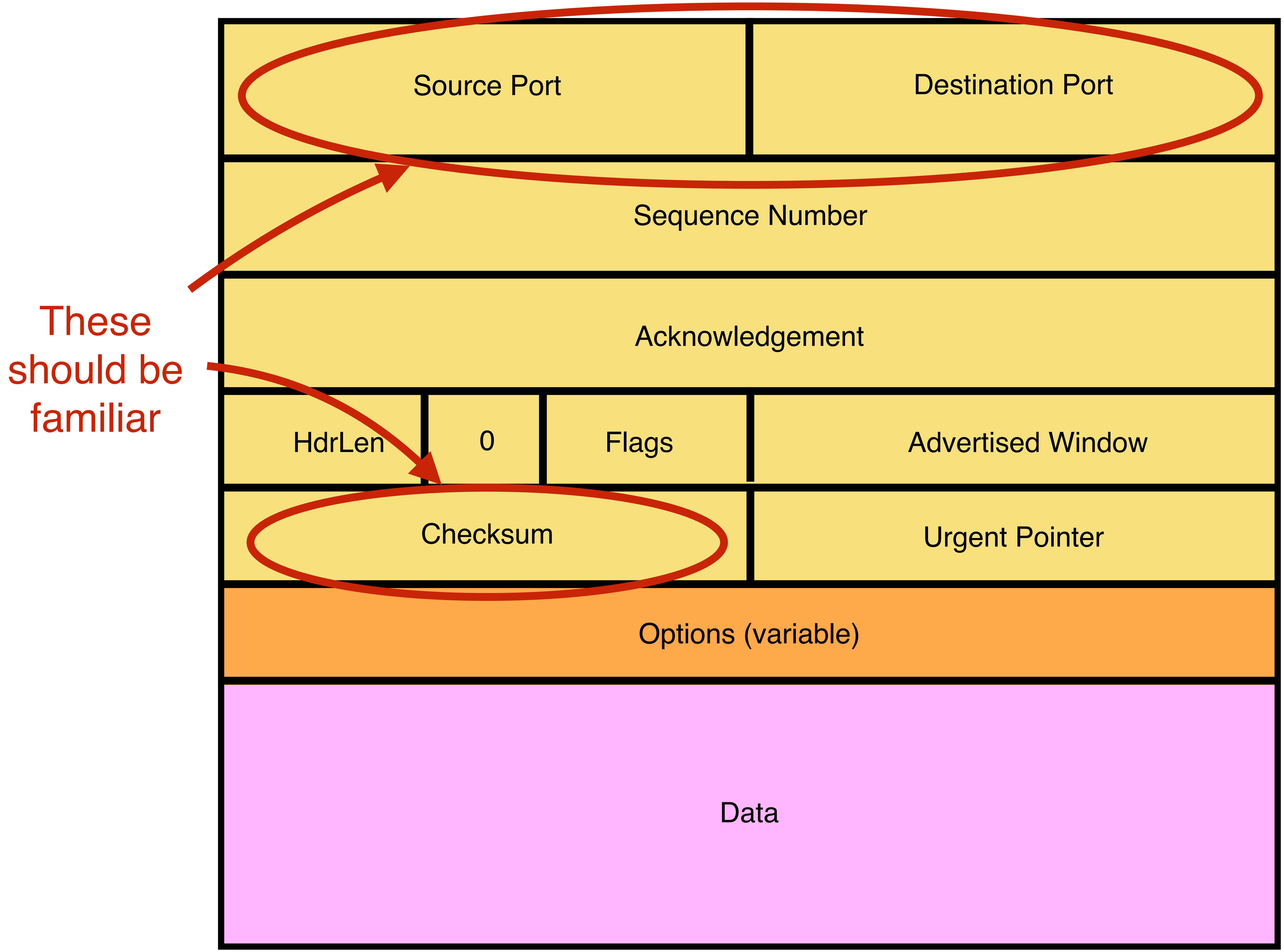
IP Packet Structure



TCP Header

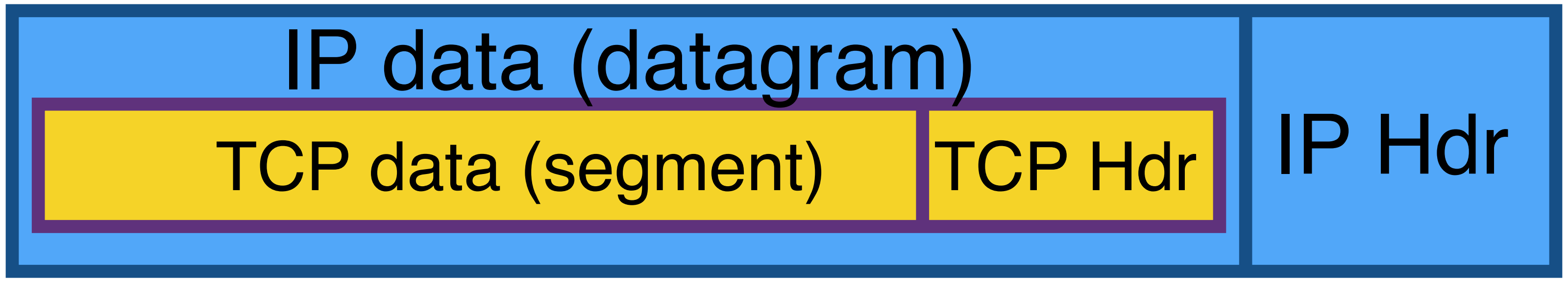

Source Port
Destination Port
Sequence Number
Acknowledgement
HdrLen
0
Flags
Advertised Window
Checksum
Urgent Pointer
Options (variable)
Data

TCP Header



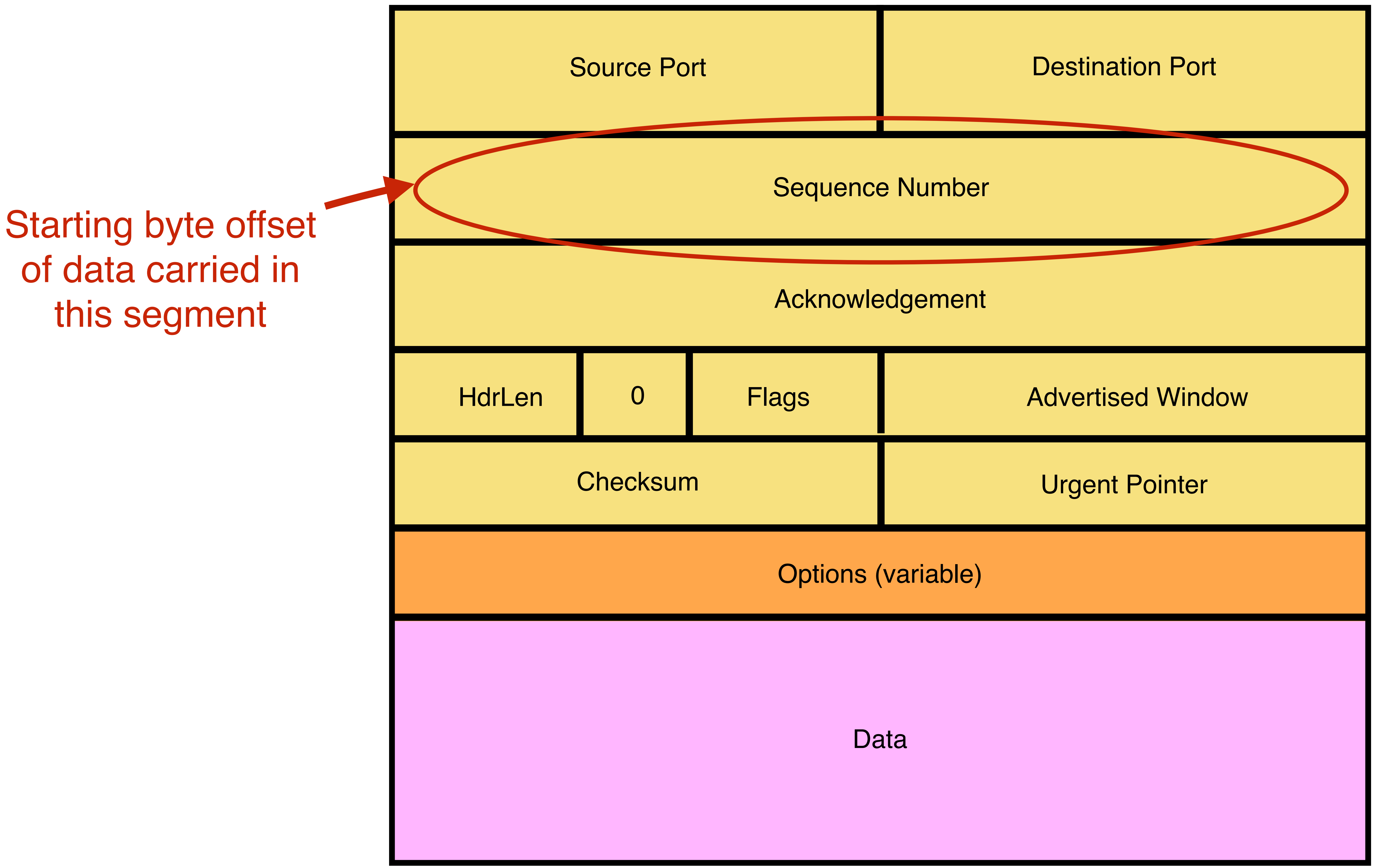
Source Port
Destination Port
Sequence Number
Acknowledgement
HdrLen
0
Flags
Advertised Window
Checksum
Urgent Pointer
Options (variable)
Data
These should be familiar

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 MSS (Maximum Segment Size) bytes
  - E.g., upto 1460 consecutive bytes from the stream
  - MSS = MTU - IP header - TCP header

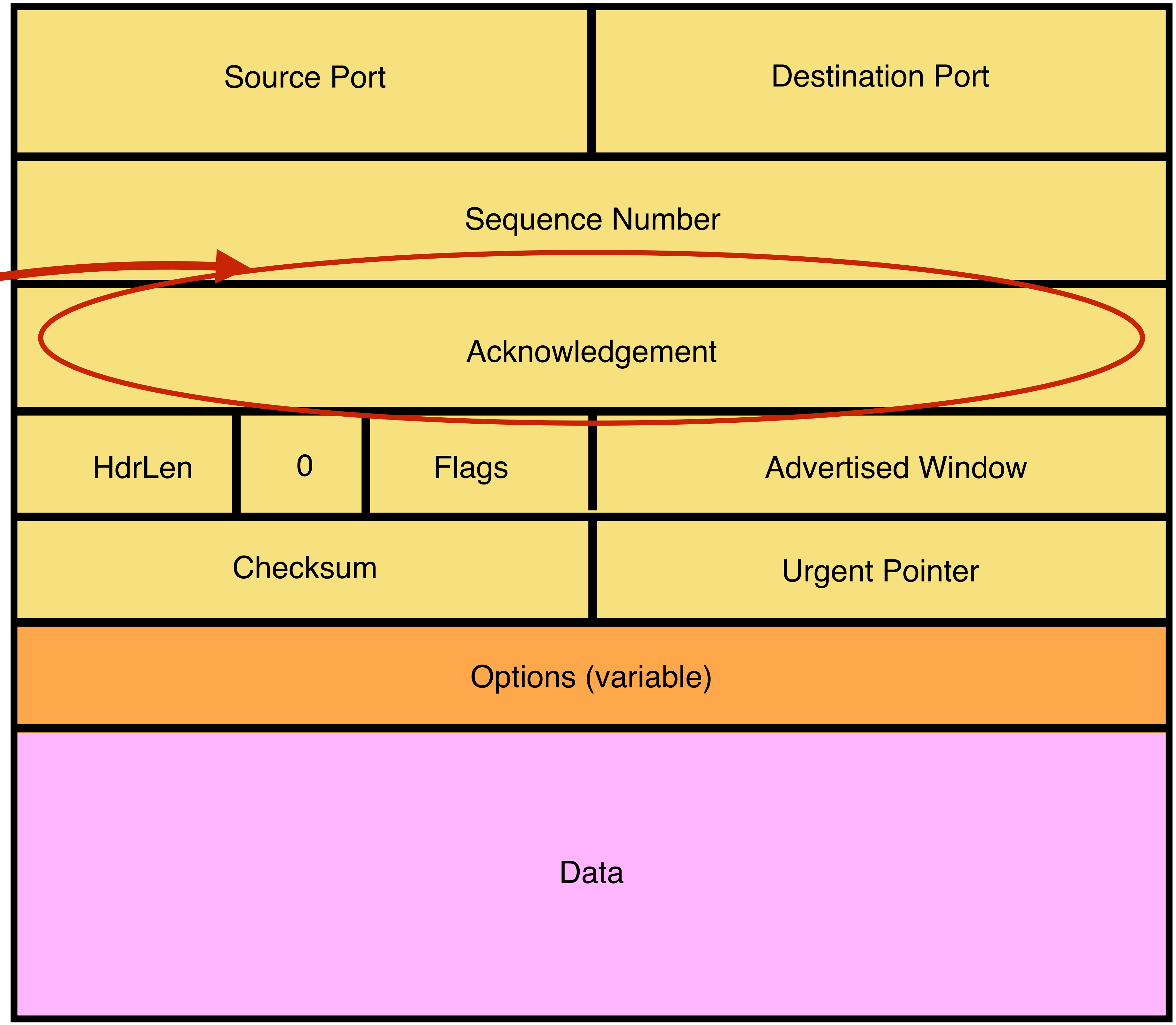
TCP Header



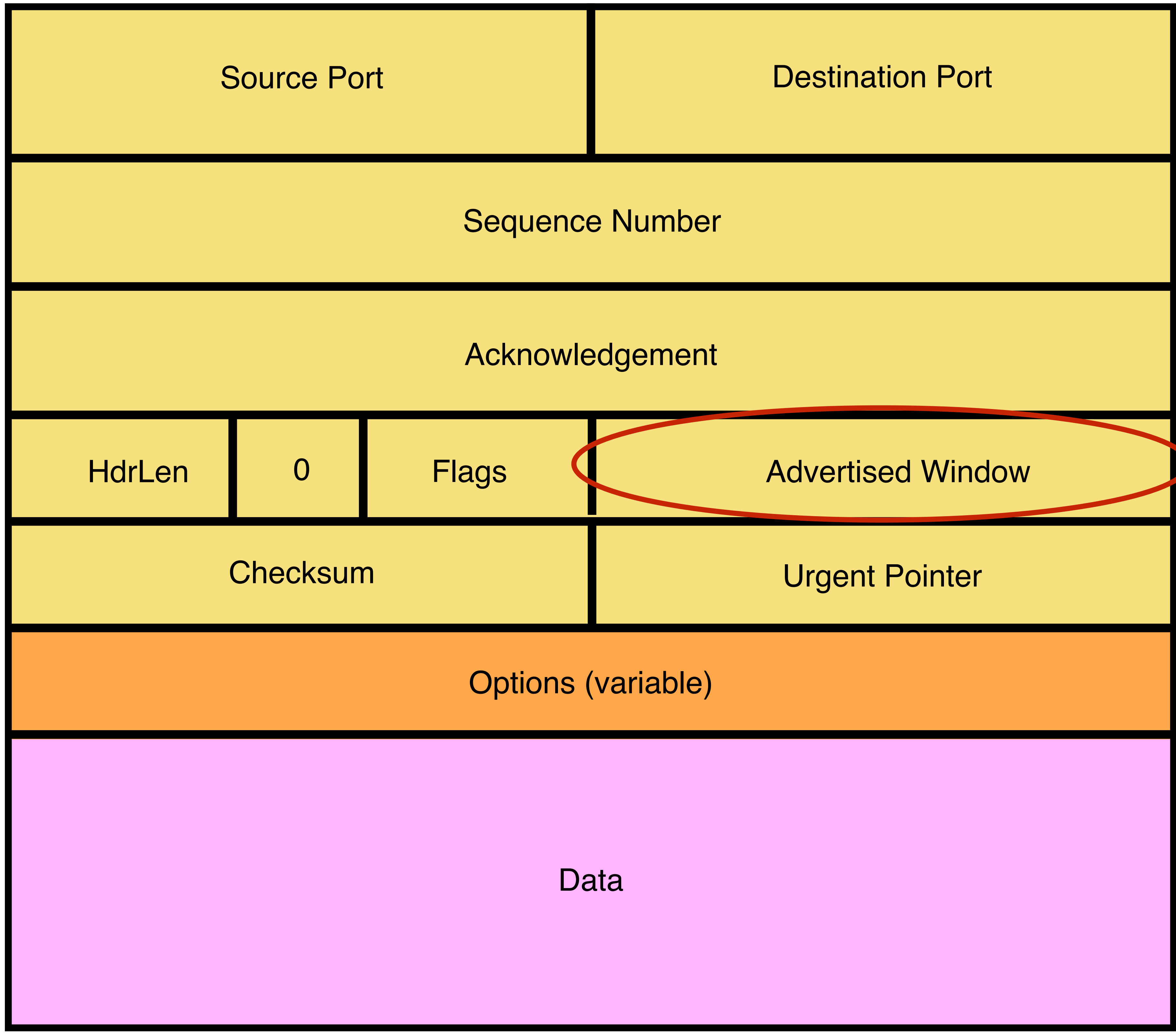
Starting byte offset of data carried in this segment
Source Port
Destination Port
Sequence Number
Acknowledgement
HdrLen
0
Flags
Advertised Window
Checksum
Urgent Pointer
Options (variable)
Data

TCP Header

Acknowledgement gives sequence number just beyond highest sequence number received in order ("What byte is next")



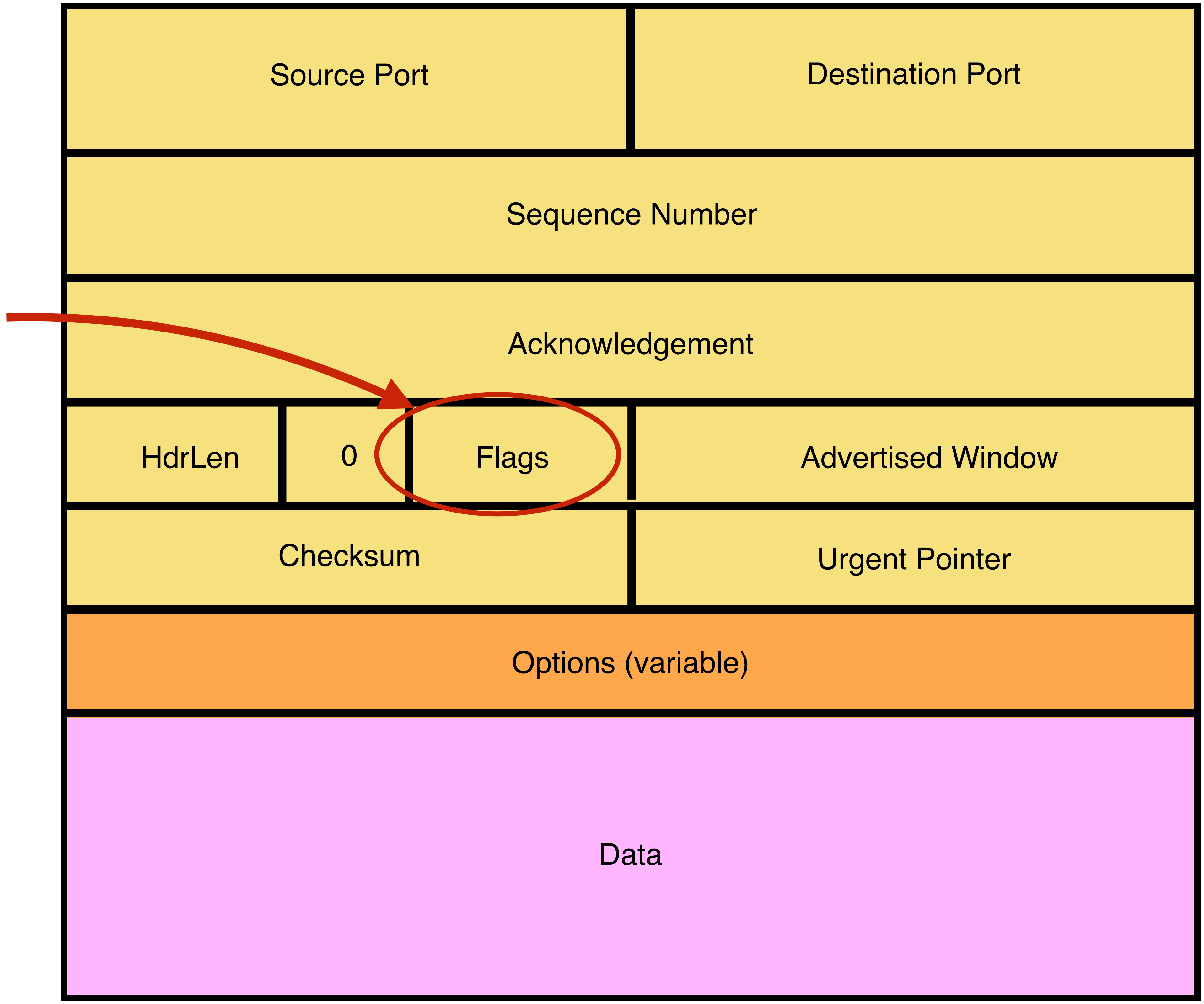
TCP Header



Source Port
Destination Port
Sequence Number
Acknowledgement
HdrLen
0
Flags
Advertised Window
Checksum
Urgent Pointer
Options (variable)
Data

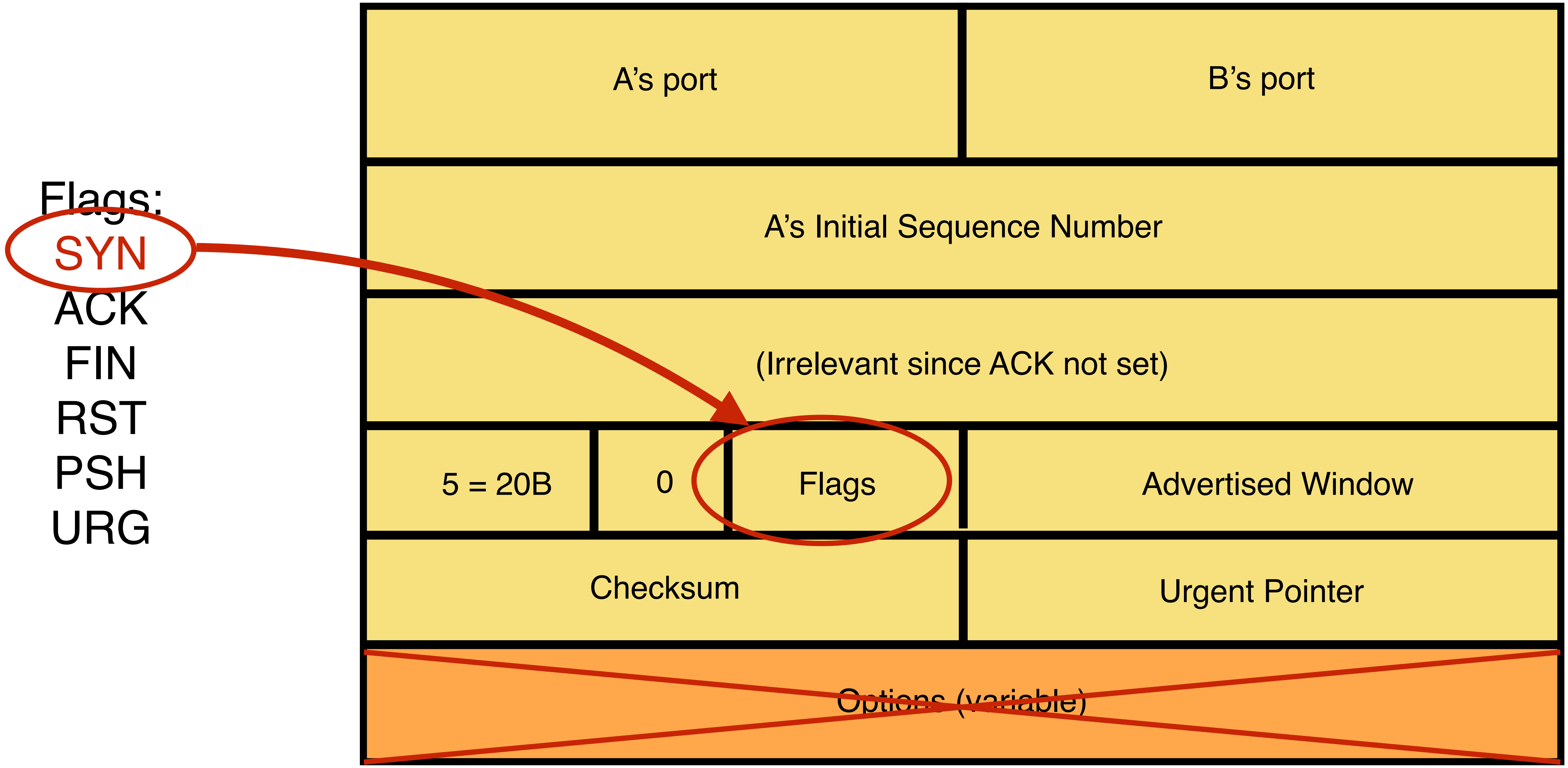
TCP Header

Flags:
SYN
ACK
FIN
RST
PSH
URG



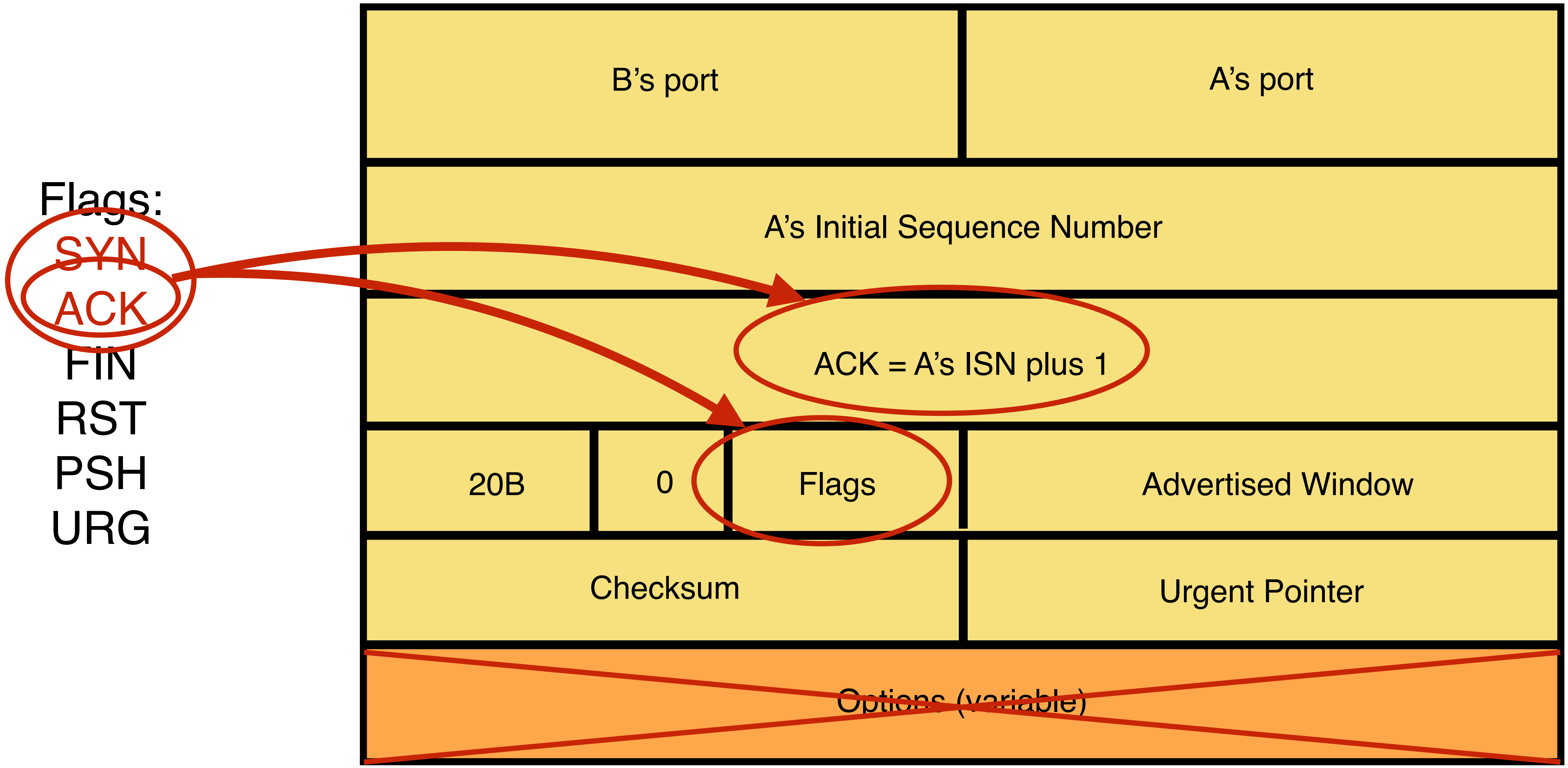
See /usr/include/netinet/tcp.h on Unix Systems

Step 1: A's Initial SYN Packet



A tells B it wants to open a connection…

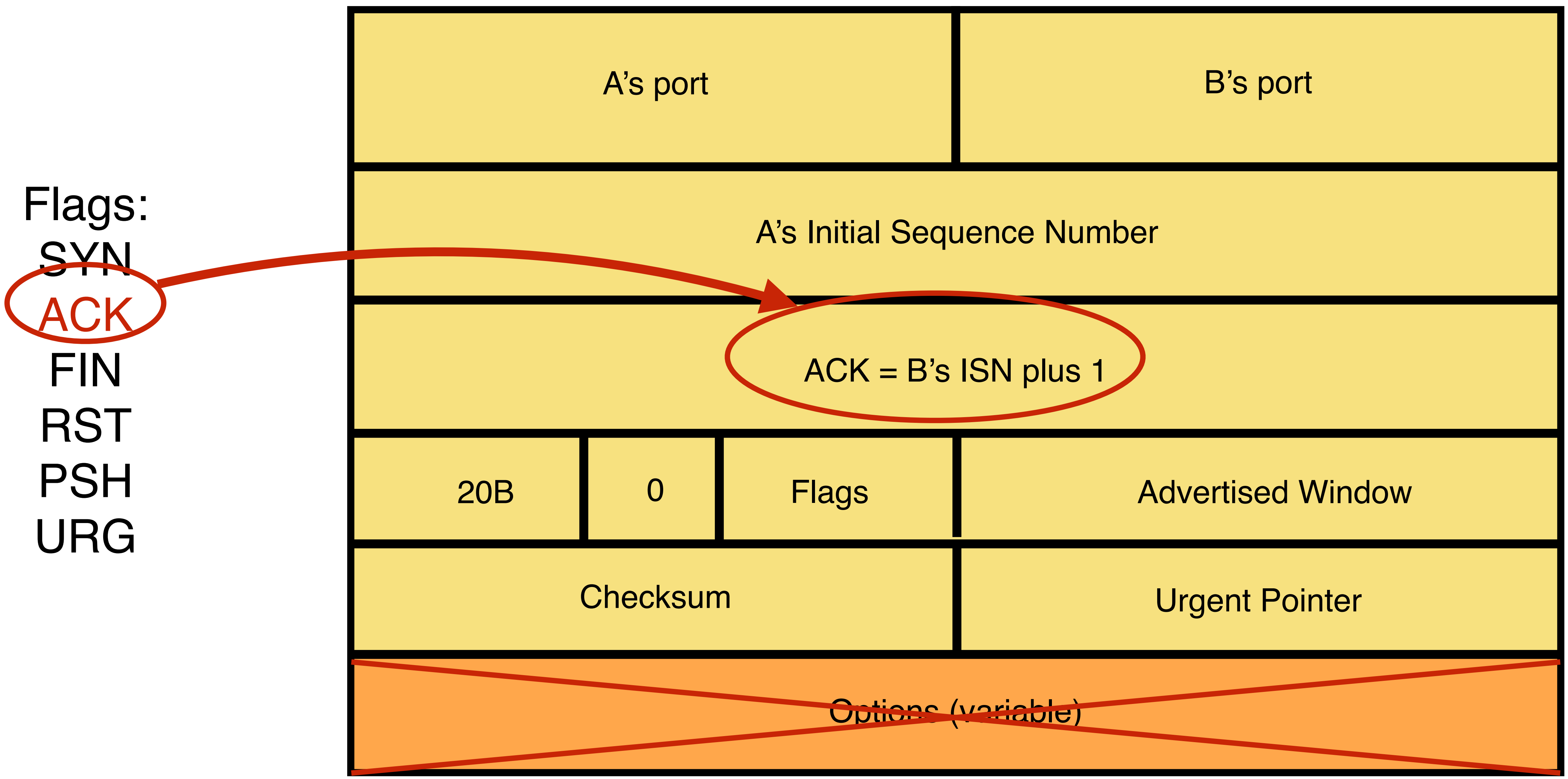
Step 2: B's SYN-ACK Packet



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

… upon receiving this packet, A can start sending data

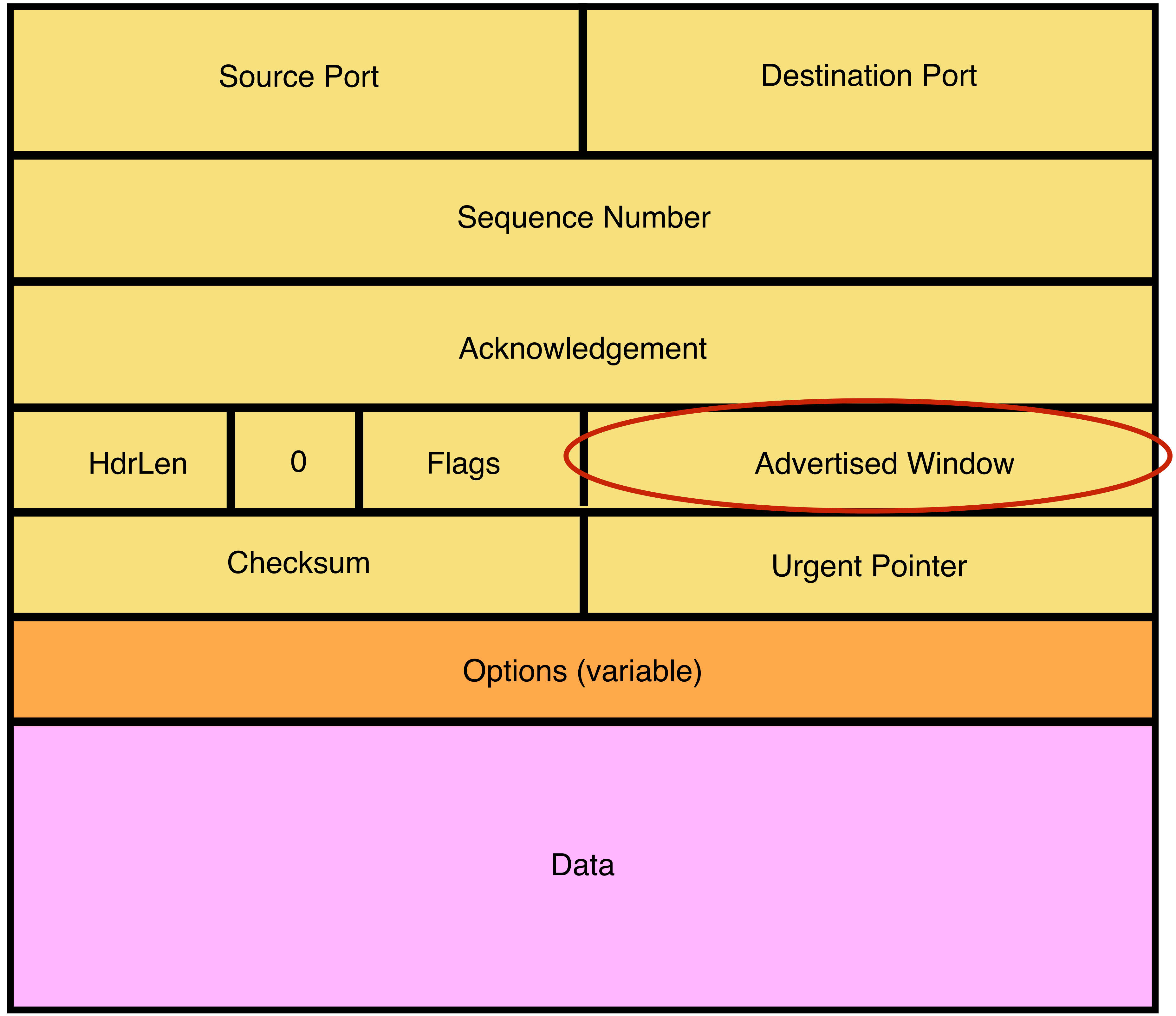
Step 3: A's ACK of the SYN-ACK



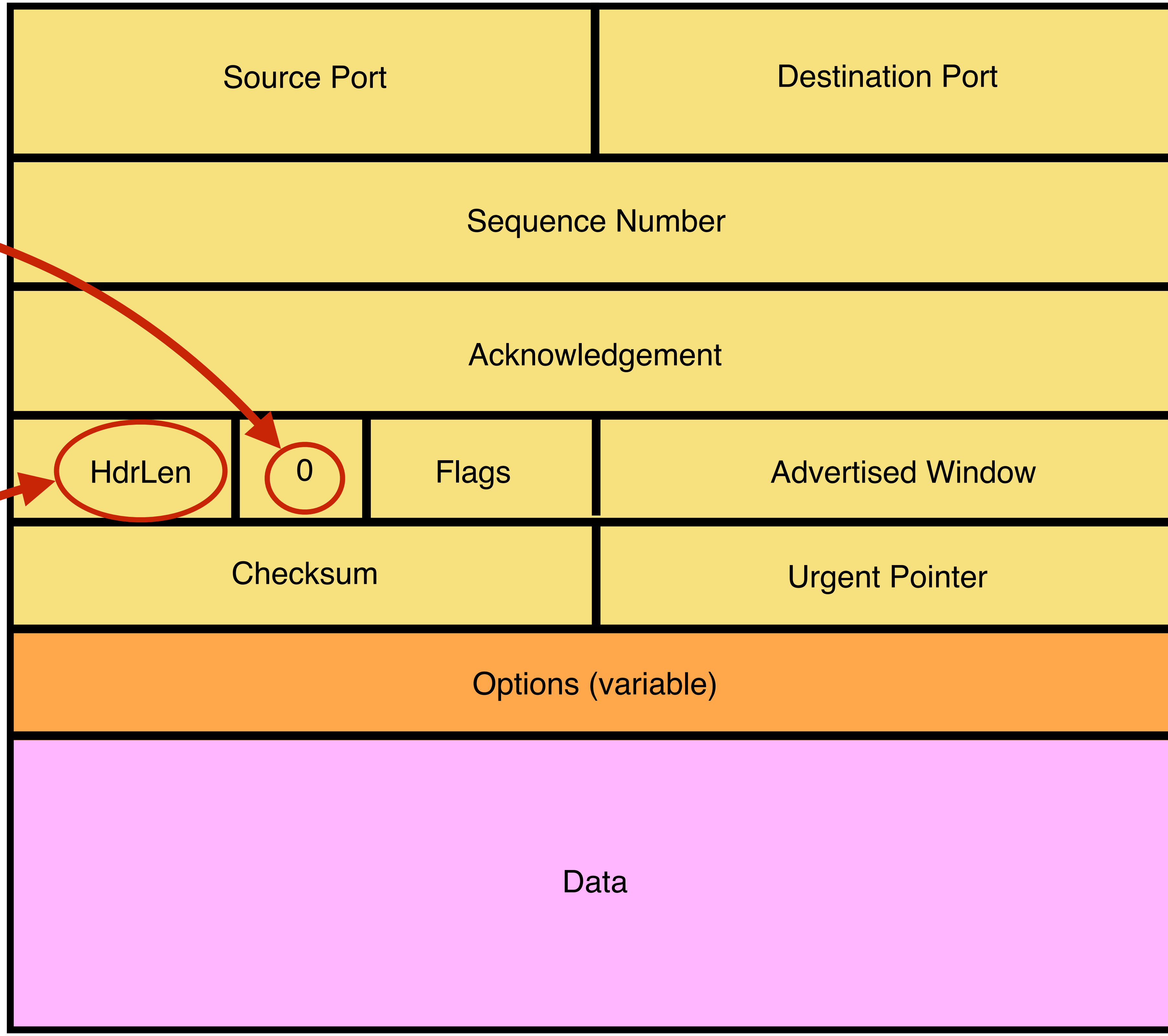
A tells B it's likewise okay to start sending

… upon receiving this packet, B can start sending data

TCP Header


Source Port
Destination Port
Sequence Number
Acknowledgement
HdrLen
0
Flags
Advertised Window
Checksum
Urgent Pointer
Options (variable)
Data

TCP Header: What's left?

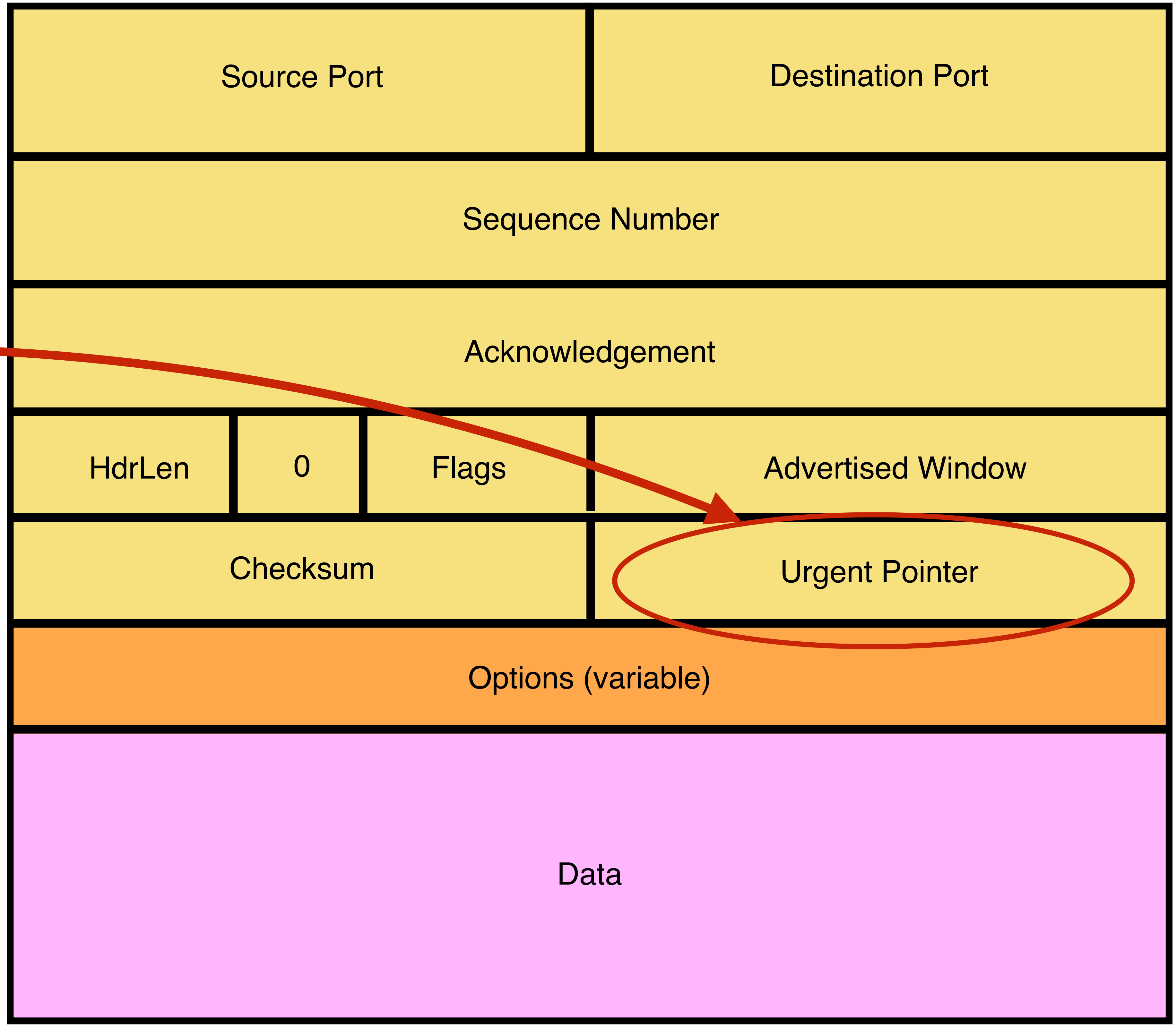


"Must be Zero"
6 bits reserved

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

TCP Header: What's left?

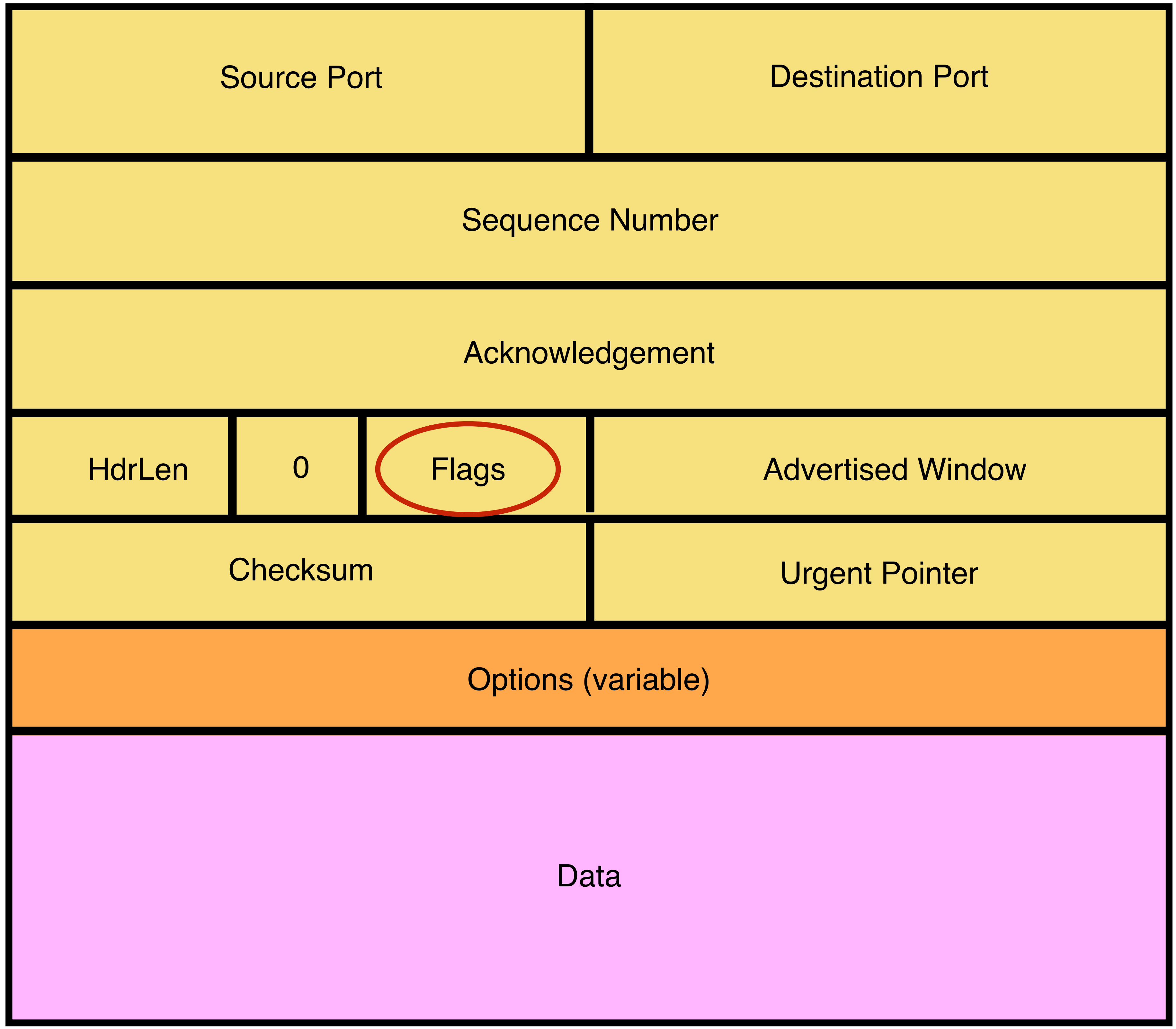
Used with URG flag to indicate urgent data (not discussed further)



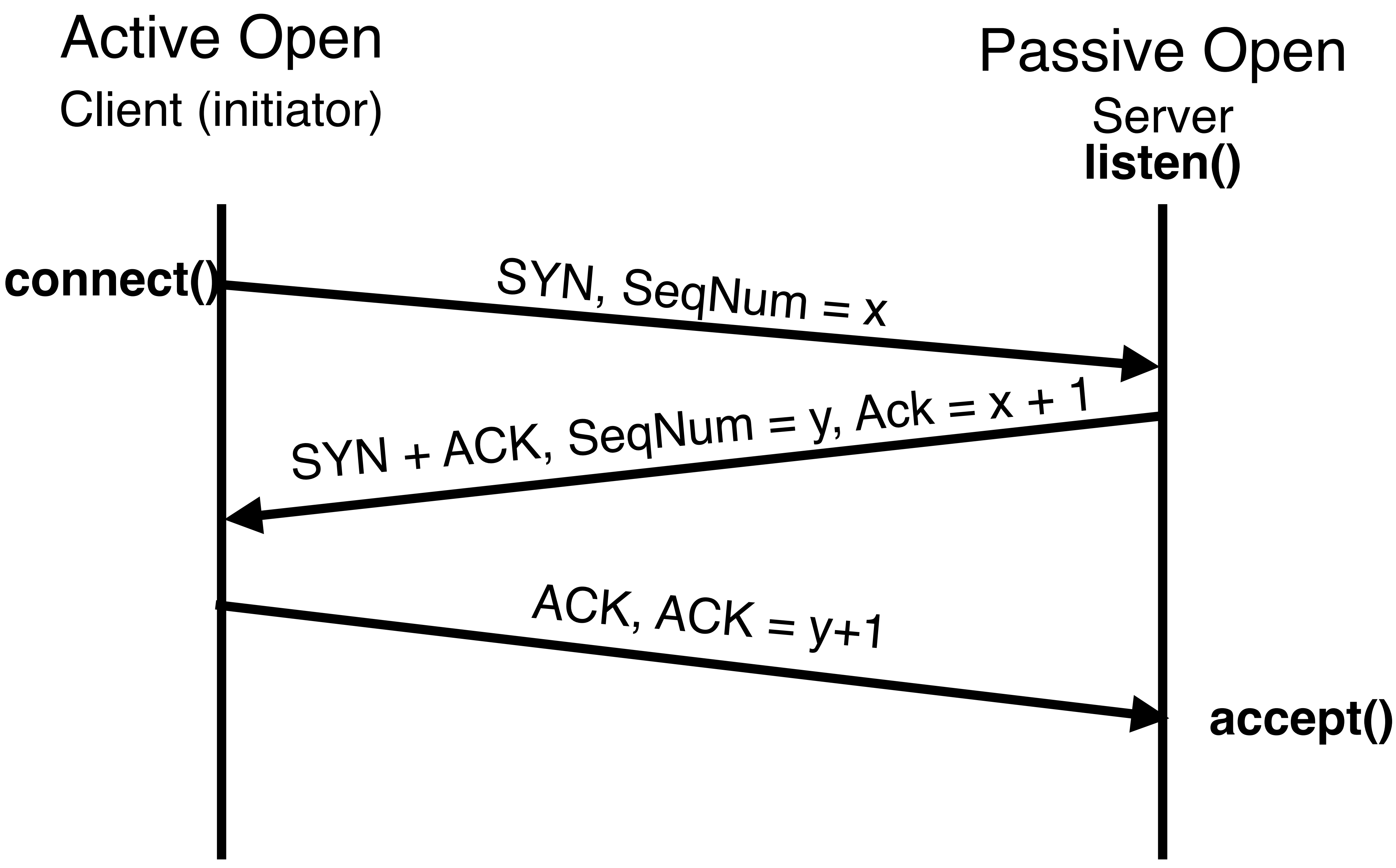
Implementing Sliding Window

- Sender maintains a window
  - Data that has been sent out but not yet ACK’ed
- Left edge of window:
  - Beginning of unacknowledged data
  - Moves when data is ACKed
- Window size = maximum amount of data in flight
- Receiver sets this amount, based on its available buffer space
  - If it has not yet sent data up to the app, this might be small

TCP Header: What's left?



Timing Diagram: 3-Way Handshaking



Note: TCP is Duplex

- A TCP connection between A and B can carry data in both directions
- Packets can both carry data and ACK data
- If the ACK flag is set, then it is ACKing data
- (details to follow …)

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 and 2988) use default of 3 seconds
        - Other implementations instead use 6 seconds

SYN Loss and Web Downloads

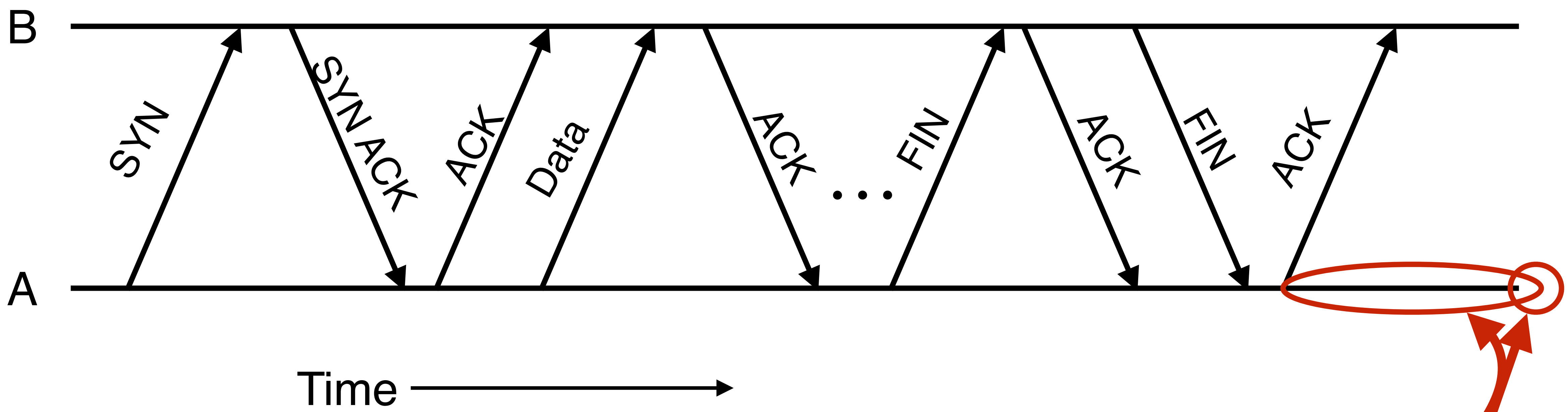
- User clicks on a hypertext link
  - Browser creates a socket and does a “connect”
  - The “connect” triggers the OS to transmit a SYN

- If the SYN is lost…
  - 3-4 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



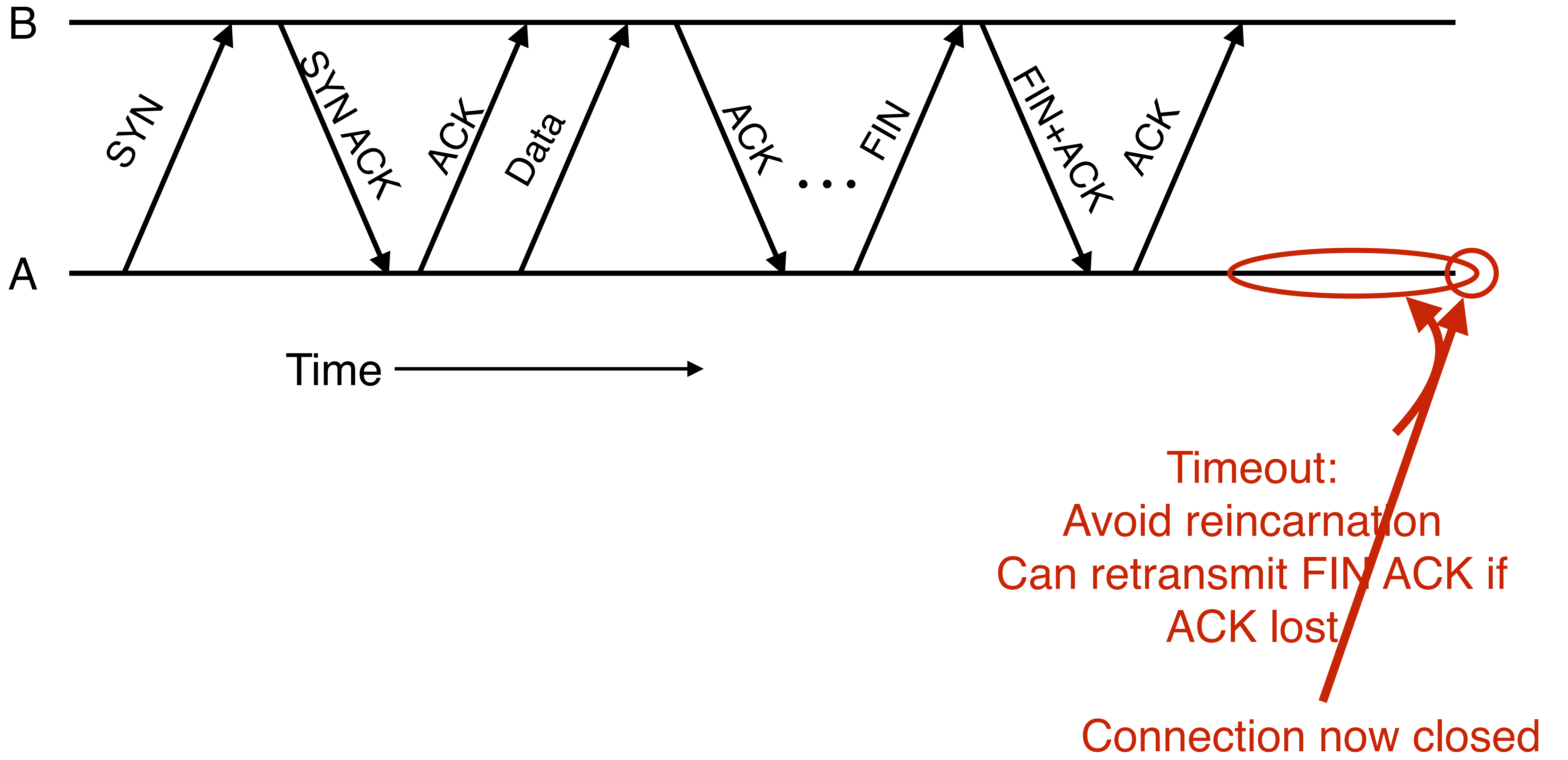
Timeout:
Avoid reincarnation
Can retransmit FIN ACK if ACK lost

Connection now closed

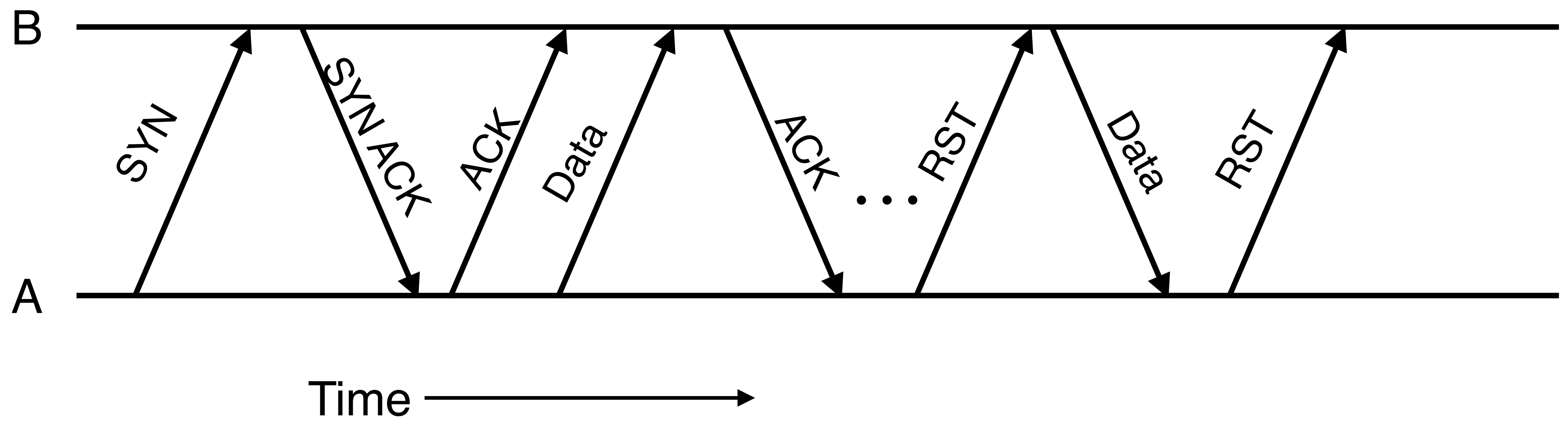
- Finish (FIN) to close connections
    - FIN occupies one byte in the sequence space
- Other host ack's the byte to confirm
- Closes A's side of connection, but not B's
    - Until B likewise sends a FIN
    - Which A then acks

Normal Termination, Both Together

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

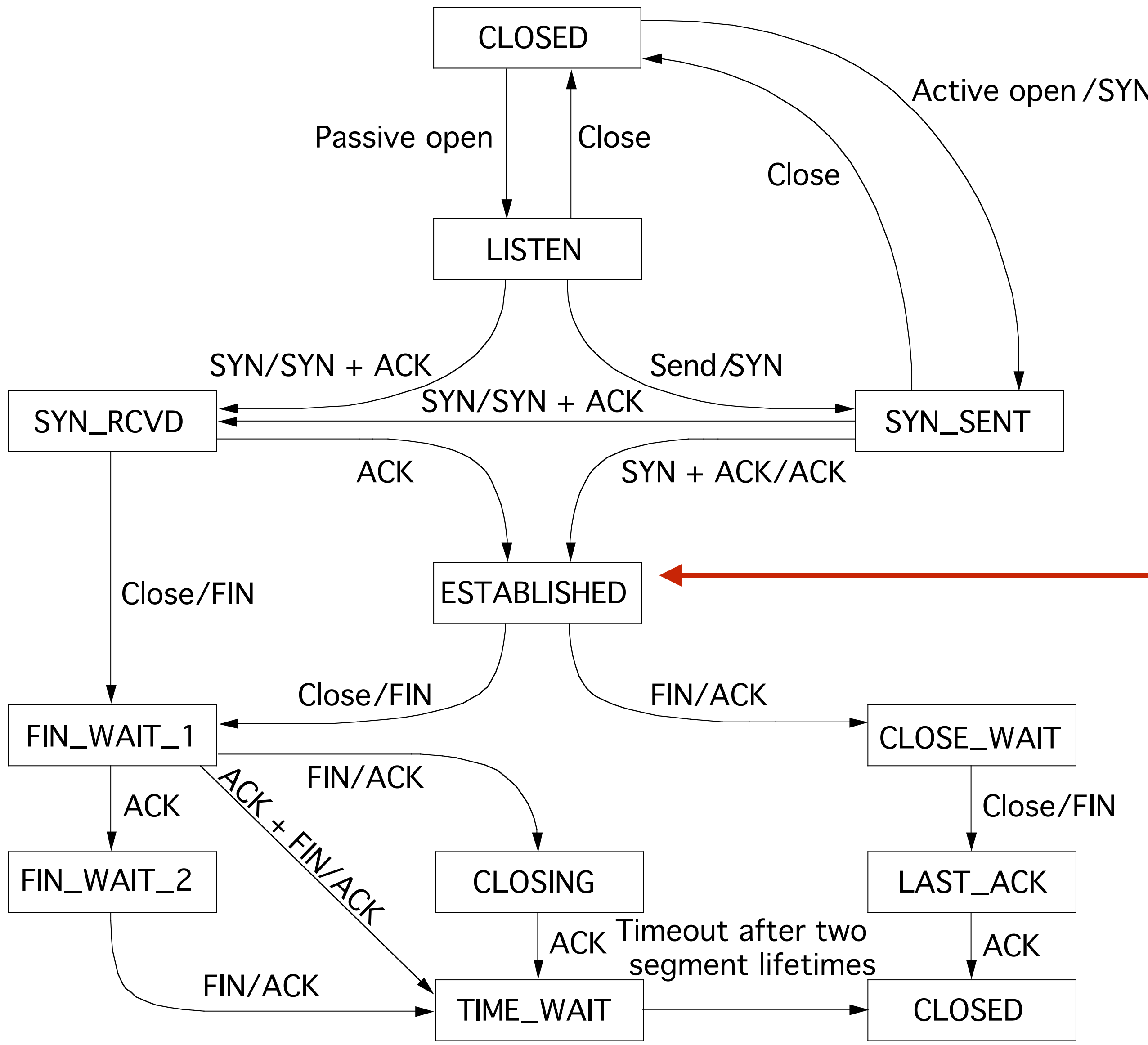


Abrupt Termination



- A sends a RESET (RST) to B
  - E.g., because app. Process on A crashed
- That's it
  - B does not ack the RST
  - This, 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

A Simpler View of the Client Side