CS4450

Computer Networks:
Architecture and Protocols

Lecture 22
Reliable Transport and TCP

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Goal of Today's Lecture

- Wrap up **reliable transport**
- TCP congestion control mechanisms
  - And the properties that they provide
  - **And when they fail to be "good enough"**

Lets start with recapping where we are in reliable transport

Recap: Best Effort Service (L3)

- Packets can be **lost**
- Packets can be **corrupted**
- Packets can be **reordered**
- Packets can be **delayed**
- Packets can be **duplicated**
- …

Transport layer:
Enabling reliability over such a best-effort service model

Recap: Complete Correctness Condition for reliability

A transport mechanism is “reliable” if and only if

(a) It resends all dropped or corrupted packets

(b) It attempts to make progress

Recap: Four Goals for Reliable Transfer

- **Correctness**
  - As defined in the last slide
- **“Fairness”**
  - Every flow must get a fair share of network resources
- **Flow Performance (Latency-related)**
  - Latency, jitter, etc.
- **Utilization (Throughput-related)**
  - Would like to maximize bandwidth utilization
  - If network has bandwidth available, flows should be able to use it!

Recap: Reliable transport

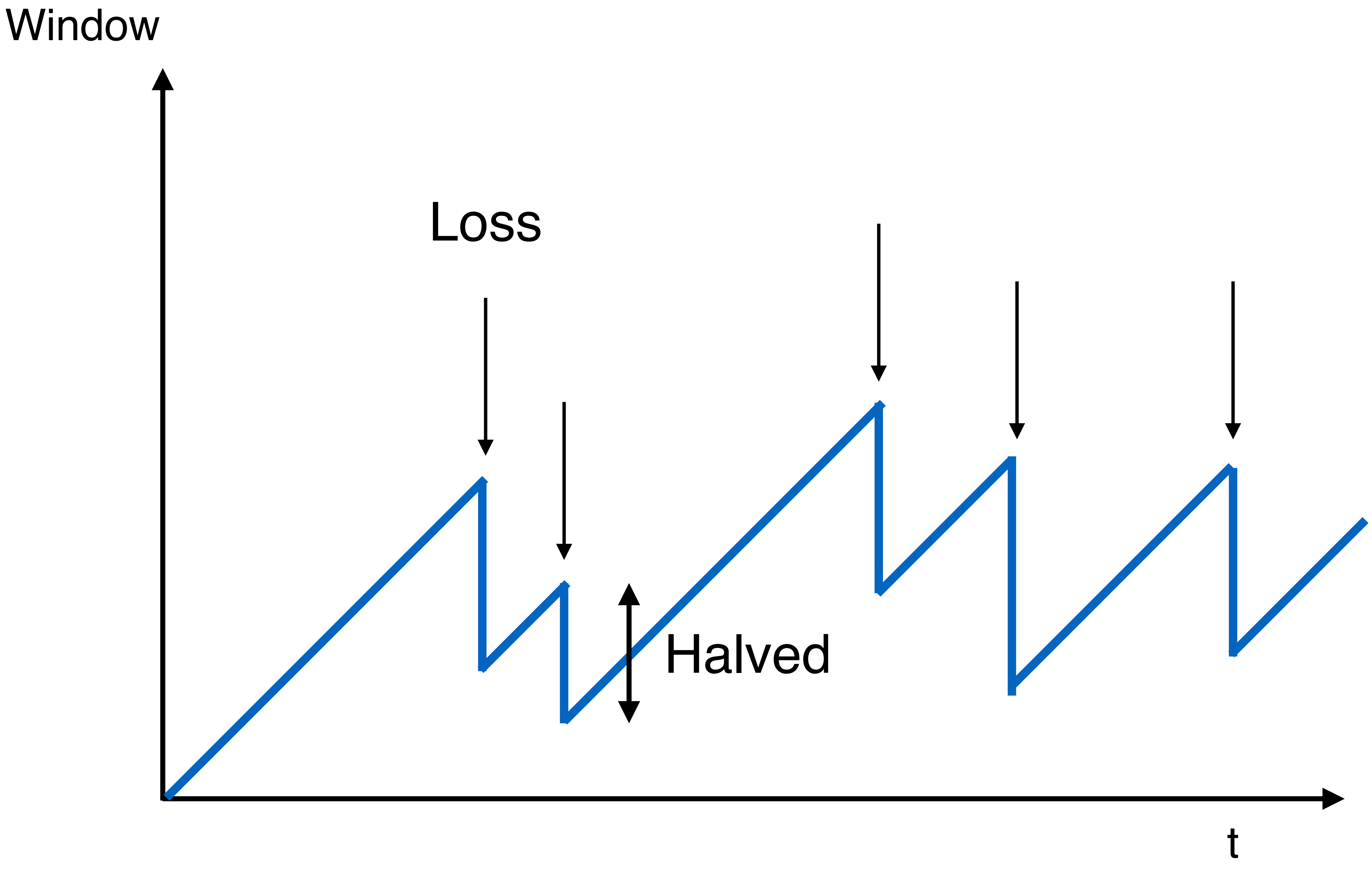
- Started from first principles
    - Correctness condition for reliable transport
- … to understanding **why feedback from receiver is necessary** (sol-v1)
- … to understanding **why timers may be needed** (sol-v2)
    - **Larger timeout:** potentially lower utilization
    - **Smaller timeout:** potentially lower latency, but also more retransmissions
- … 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

TCP Congestion Control

Recap: 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

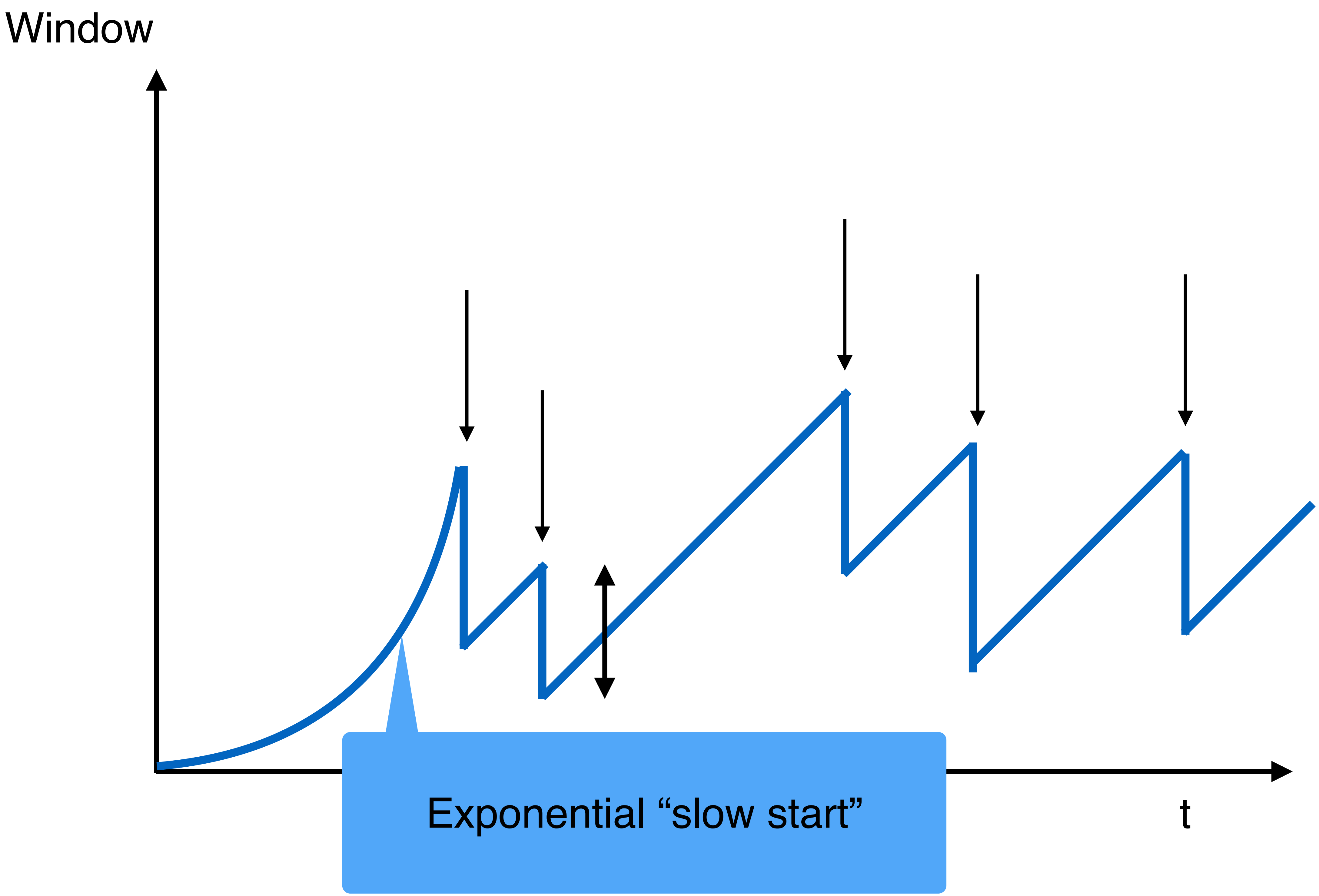
Recap: Leads to the TCP Sawtooth



Recap: “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

Recap: 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.

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 $\text{Sum}(a_i) = C$ 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

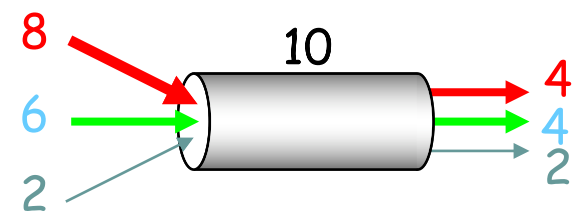
- Assume demands are in increasing order…
- If $C/N <= r_1$, then $a_i = C/N$ for all i
- Else, $a_1 = r1$, set $C = C - a_1$ and $N = N-1$
- Repeat
- Intuition: all flows requesting less than fair share get their request. Remaining flows divide equally

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: $f = 4$

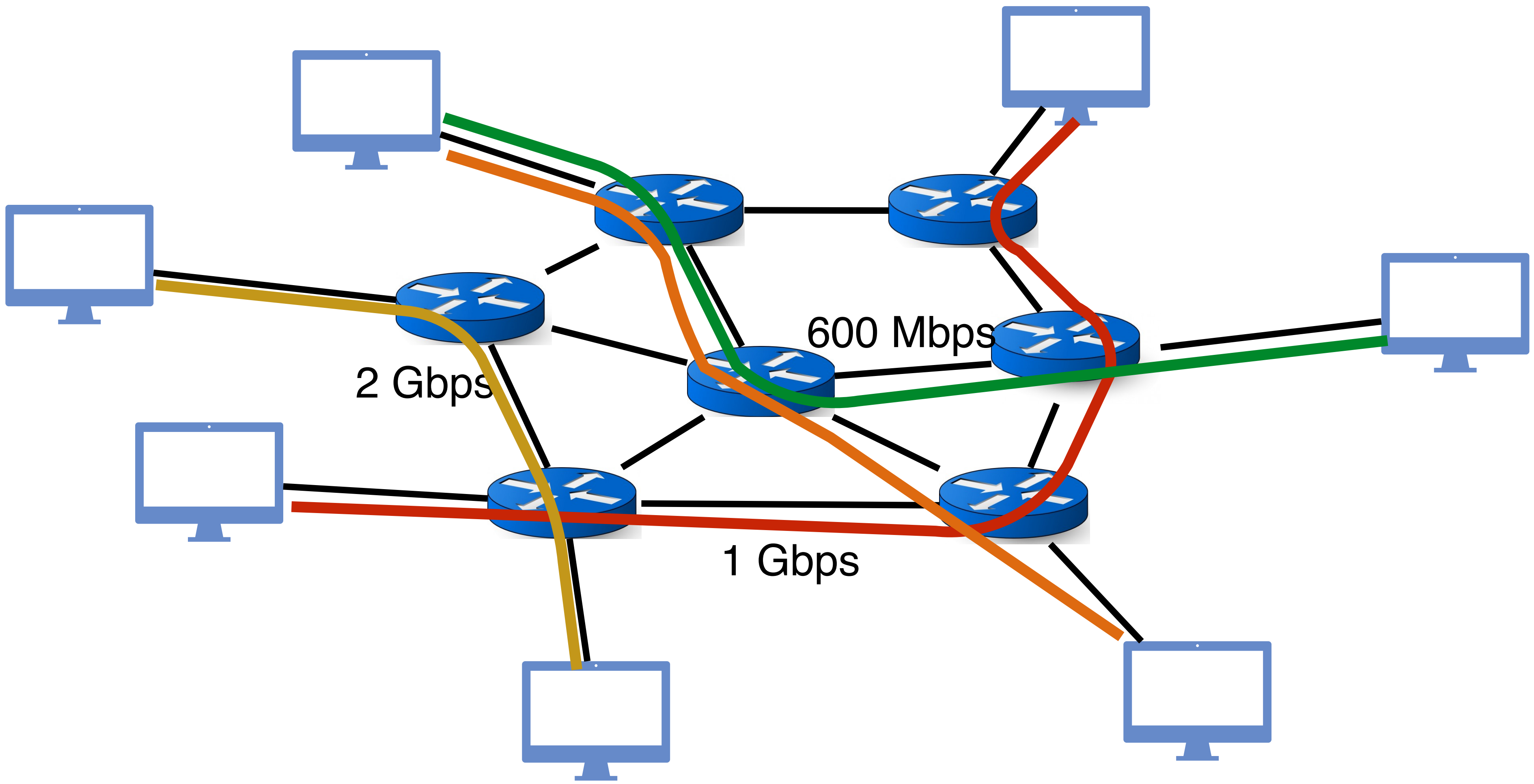


$f = 4$:
min(8, 4) = 4
min(6, 4) = 4
min(2, 4) = 2

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

Questions?

The Many Failings of TCP Congestion Control

1. Fills up queues (large queueing delays)
2. Every segment not ACKed is a loss (non-congestion related losses)
3. Produces irregular saw-tooth behavior
4. Biased against long RTTs (unfair)
5. Not designed for short flows
6. Easy to cheat

(1) TCP Fills Up Queues

- TCP only slows down when queues fill up
  - High queueing delays
- Means that it is not optimized for latency
  - What is it optimized for then?
    - **Answer: Fairness (discussion in next few slides)**
- And many packets are dropped when buffer fills
- Alternative 1: Use small buffers
  - Is this a good idea?
  - Answer: No, bursty traffic will lead to reduced utilization
- Alternative: **Random Early Drop (RED)**
  - Drop packets on purpose **before** queue is full
  - A very clever idea

Random Early Drop (or Detection)

- Measure average queue size A with exponential weighting
  - Average: Allows for short bursts of packets without over-reacting
- Drop probability is a function of A
  - No drops if A is very small
  - Low drop rate for moderate A's
  - Drop everything if A is too big
- Drop probability applied to incoming packets
- Intuition: link is fully utilized well before buffer is full

Advantages of RED

- Keeps queues smaller, while allowing bursts
  - Just using small buffers in routers can’t do the latter
- Reduces synchronization between flows
  - Not all flows are dropping packets at once
  - Increases/decreases are more gentle
- Problem
  - Turns out that RED does not guarantee fairness

(2) Non-Congestion-Related Losses?

- For instance, RED drops packets intentionally
    - TCP would think the network is congested
- Can use **Explicit Congestion Notification (ECN)**
- Bit in IP packet header (actually two)
    - TCP receiver returns this bit in ACK
- When RED router would drop, it sets bit instead
    - Congestion semantics of bit exactly like that of drop
- Advantages
    - Doesn’t confuse corruption with congestion

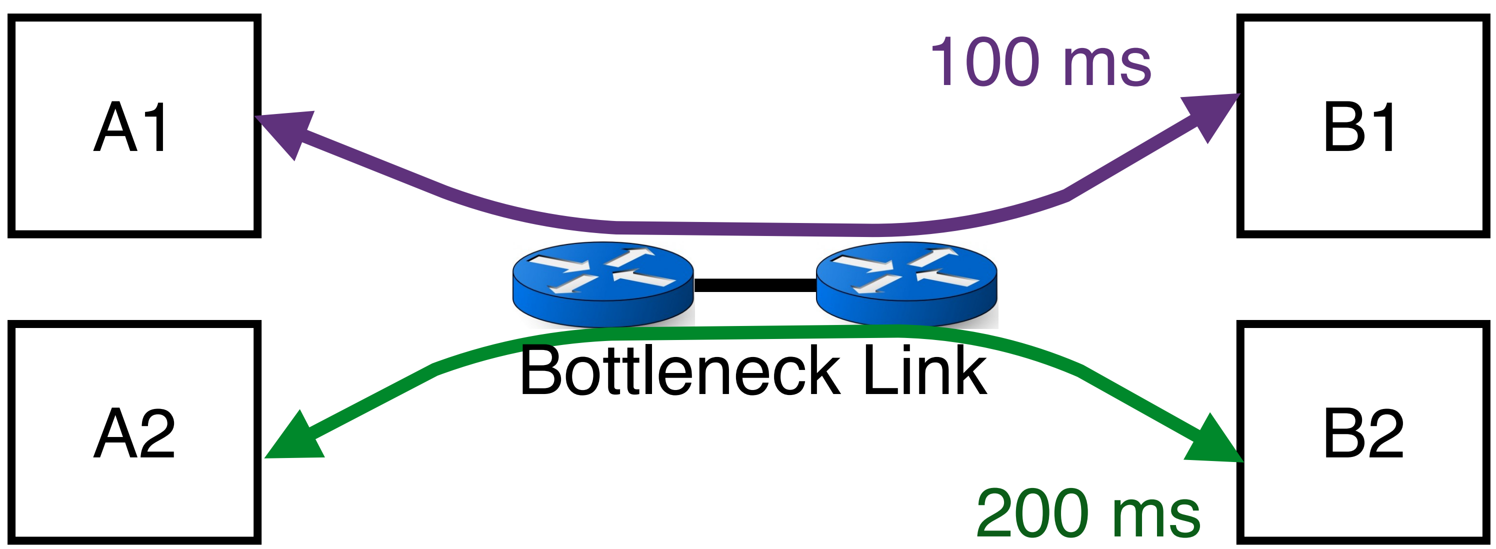
(3) Sawtooth Behavior Uneven

- TCP throughput is “choppy"
  - Repeated swings between W/2 to W
- Some apps would prefer sending at a steady rate
  - E.g., streaming apps
- A solution: “Equation-based congestion control”
  - Ditch TCP’s increase/decrease rules and just follow the equation:
  - **[Matthew Mathis, 1997] TCP Throughput = MSS/RTT sqrt(3/2p)**
    - **Where p is drop rate**
  - Measure drop percentage p and set rate accordingly
- Following the TCP equation ensures we’re TCP friendly
  - I.e., use no more than TCP does in similar setting

Any Questions?

(4) Bias Against Long RTTs

- Flows get throughput inversely proportional to RTT
- **TCP unfair in the face of heterogeneous RTTs!**
- [Matthew Mathis, 1997] TCP Throughput = MSS/RTT sqrt(3/2p)
    - Where p is drop rate
- Flows with long RTT will achieve lower throughput



Possible Solutions

- Make additive constant proportional to RTT
- But people don’t really care about this…

(5) How Short Flows Fare?

- Internet traffic:
    - Elephant and mice flows
    - Elephant flows carry most bytes (>95%), but are very few (<5%)
    - Mice flows carry very few bytes, but most flows are mice
        - 50% of flows have < 1500B to send (1 MTU);
        - 80% of flows have < 100KB to send

- Problem with TCP?
    - Mice flows do not have enough packets for duplicate ACKs!!
    - Drop ~=~ Timeout (unnecessary high latency)
    - These are precisely the flows for which latency matters!!!

- Another problem:
    - Starting with small window size leads to high latency

Possible Solutions?

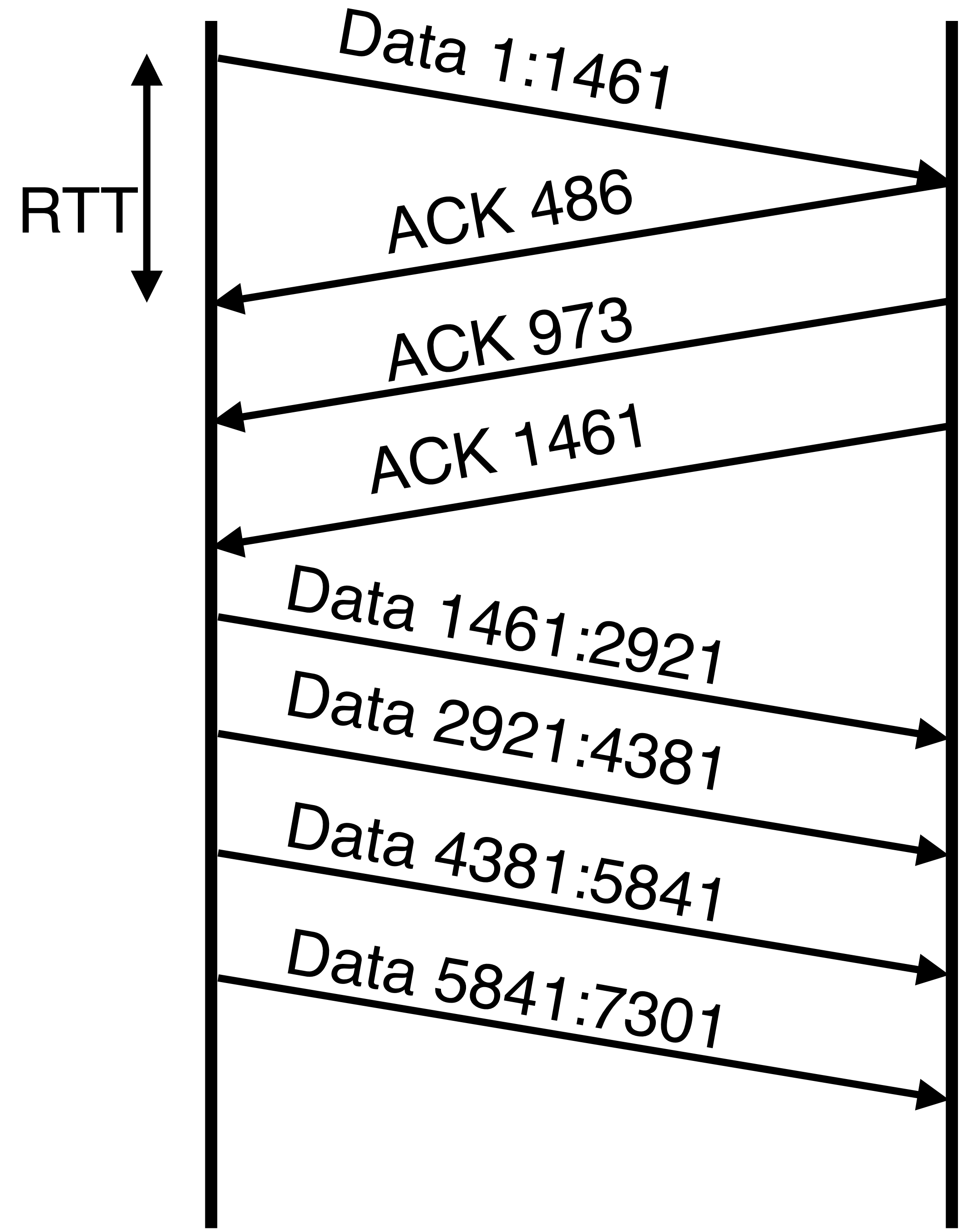
- Larger initial window?
  - Google proposed moving from ~4KB to ~15KB
  - Covers ~90% of HTTP Web
  - Decreases delay by 5%
- Many recent research papers on the timeout problem
  - Require network support

(6) Cheating

- TCP was designed assuming a cooperative world
- No attempt was made to prevent cheating
- Many ways to cheat, will present three

Cheating #1: ACK-splitting (receiver)

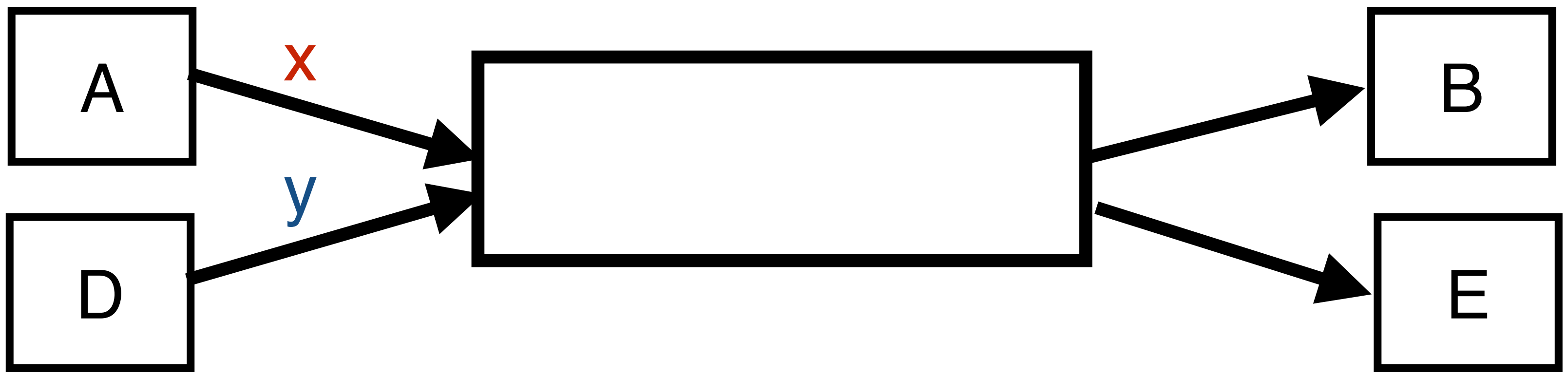
- TCP Rule: grow window by one MSS for each valid ACK received
- Send **M** (distinct) ACKs for one MSS
- Growth factor proportional to **M**



Cheating #2: Increasing CWND Faster (source)

- TCP Rule: increase window by one MSS for each valid ACK received
- Increase window by **M** per ACK
- Growth factor proportional to **M**

Cheating #3: Open Many Connections (source/receiver)



- Assume
    - A start 10 connections to B
    - D starts 1 connection to E
    - Each connection gets about the same throughput
- Then A gets 10 times more throughput than D

Cheating

- Either sender or receiver can independently cheat!
- **Why hasn't Internet suffered congestion collapse yet?**
  - Individuals don't hack TCP (not worth it)
  - Companies need to avoid TCP wars
- How can we prevent cheating
  - Verify TCP implementations
  - Controlling end points is hopeless
- Nobody cares, really

Questions?

How Do You Solve These Problems?

- Bias against long RTTs
- Slow to ramp up (for short-flows)
- Cheating
- Need for uniformity

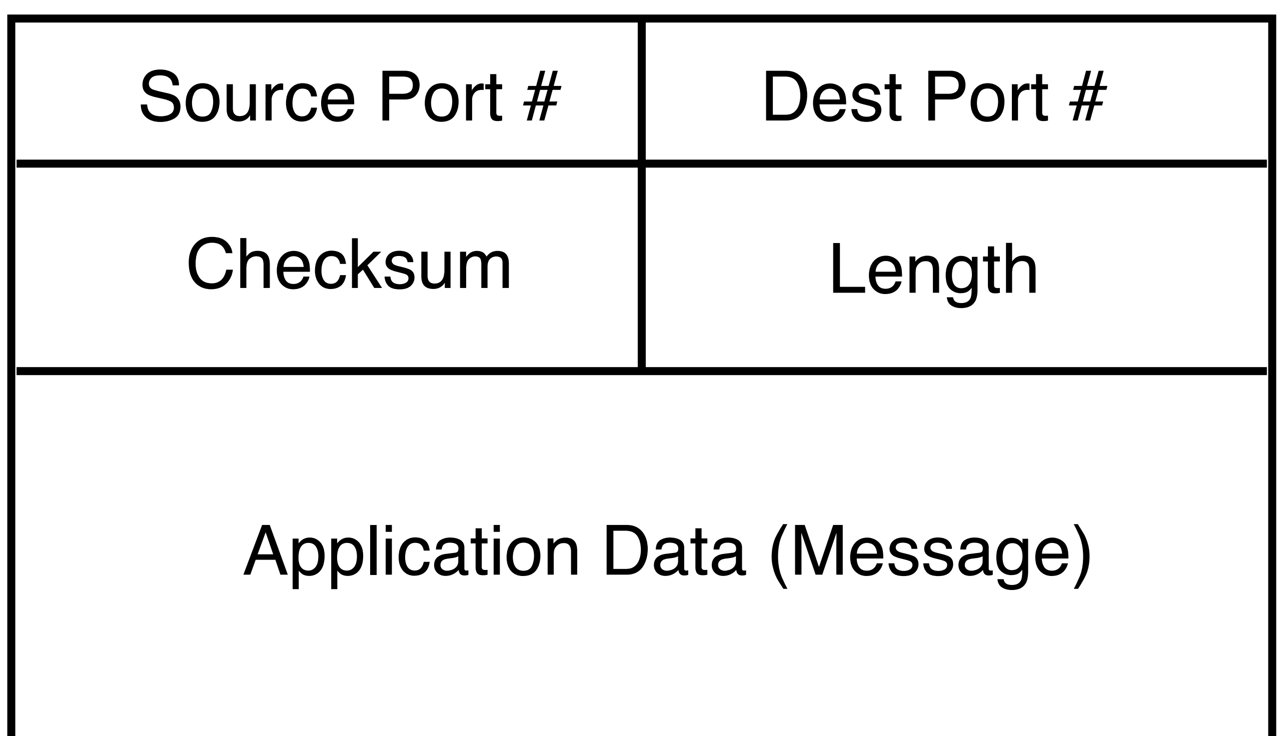
Back up slides on UDP

(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

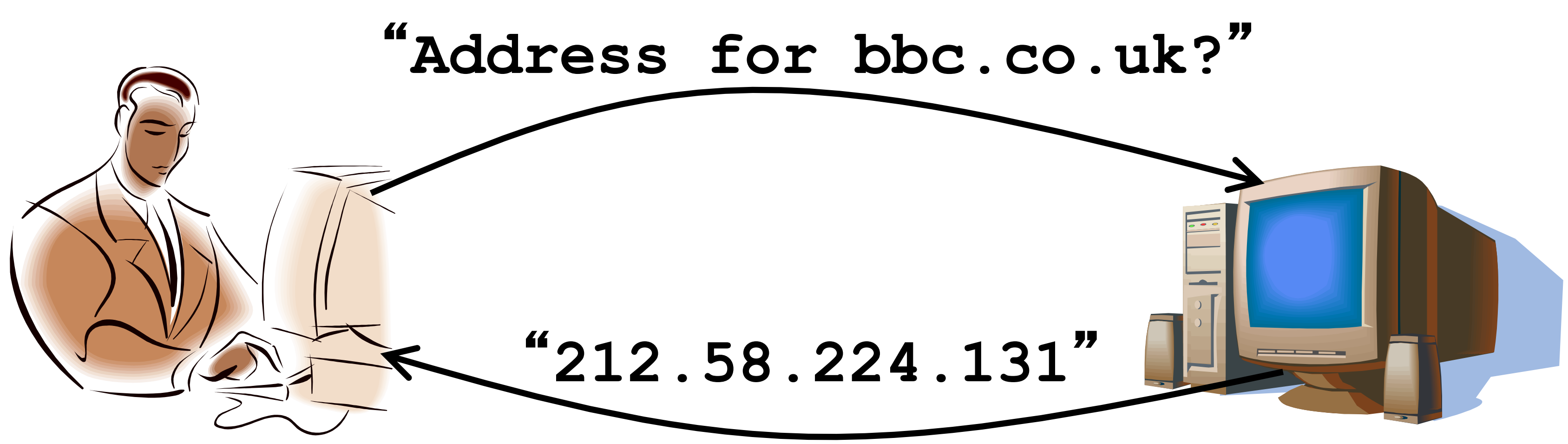


Question

- 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)

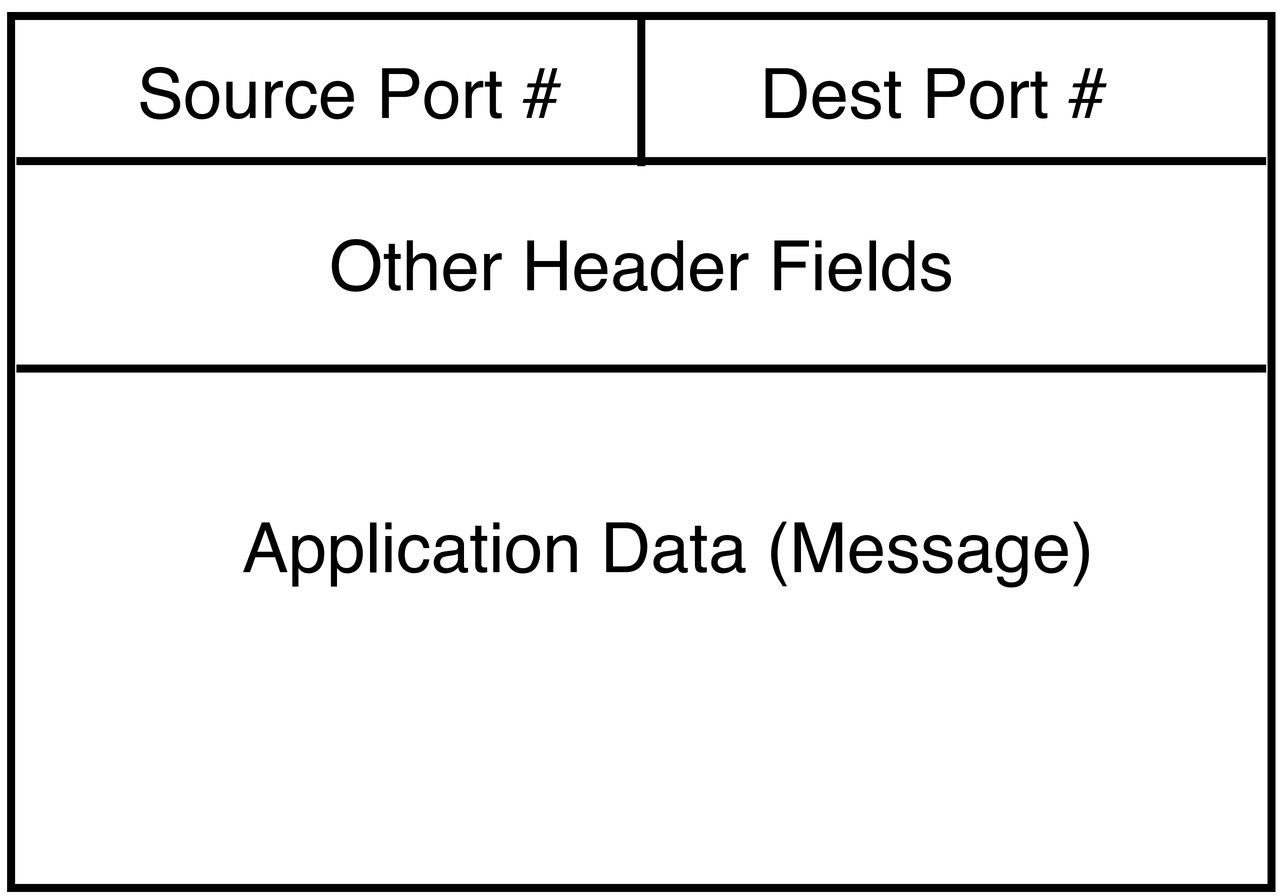
Ports

- 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

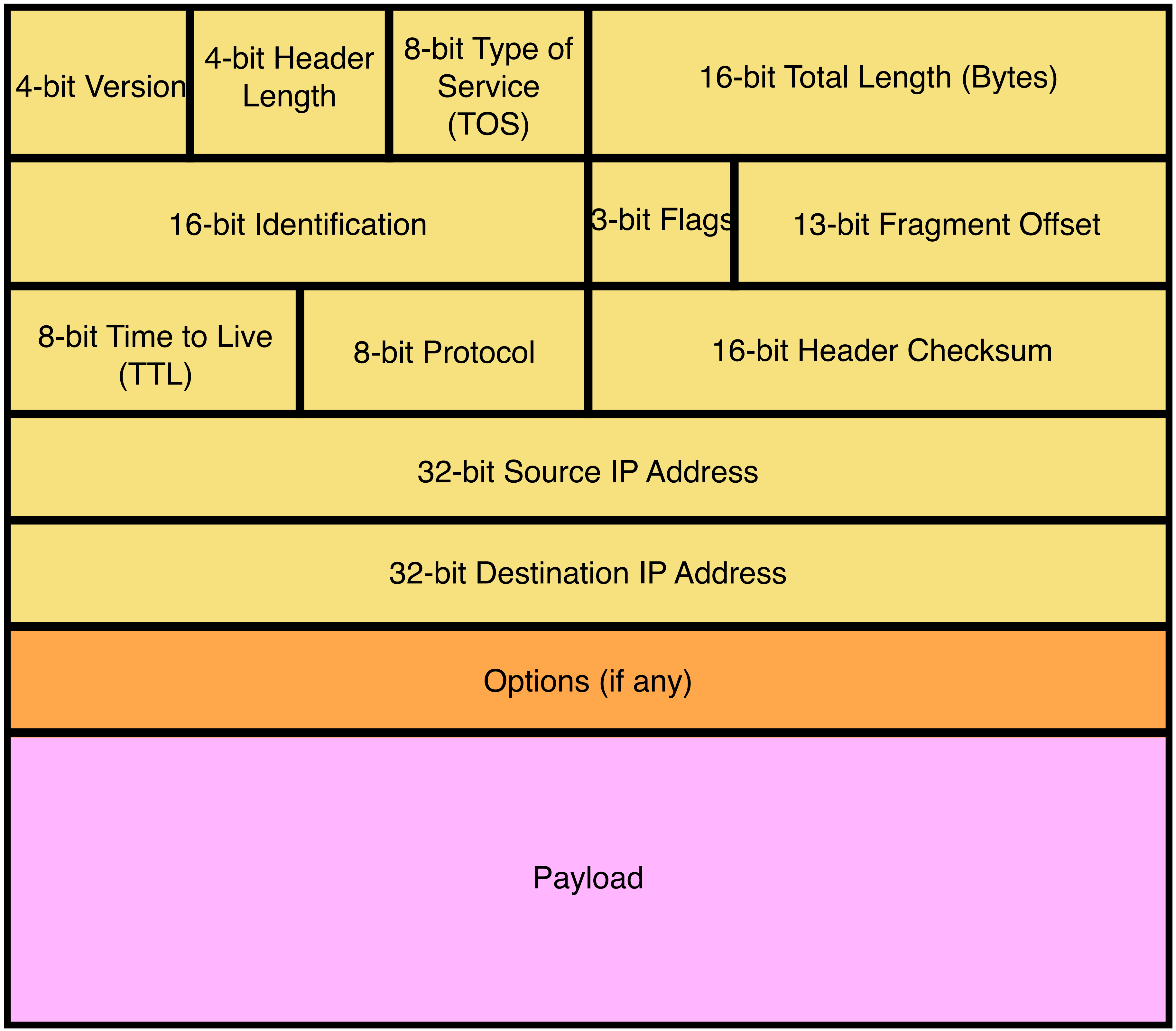
Multiplexing and Demultiplexing

- Host receives IP datagrams
  - Each datagram has source and destination IP address
  - Each segment has source and destination port number

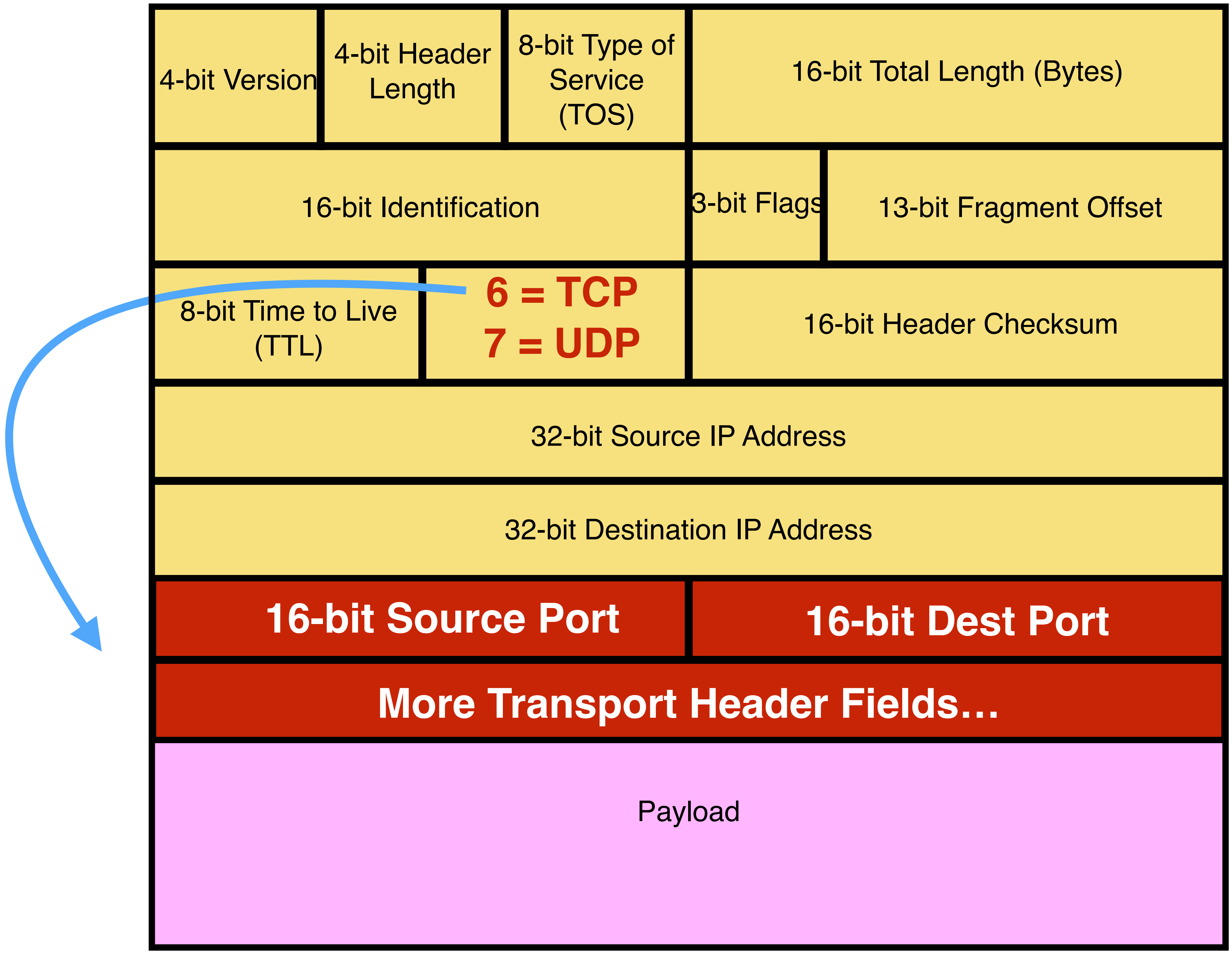
- Host uses IP address and port numbers to direct the segment to appropriate socket



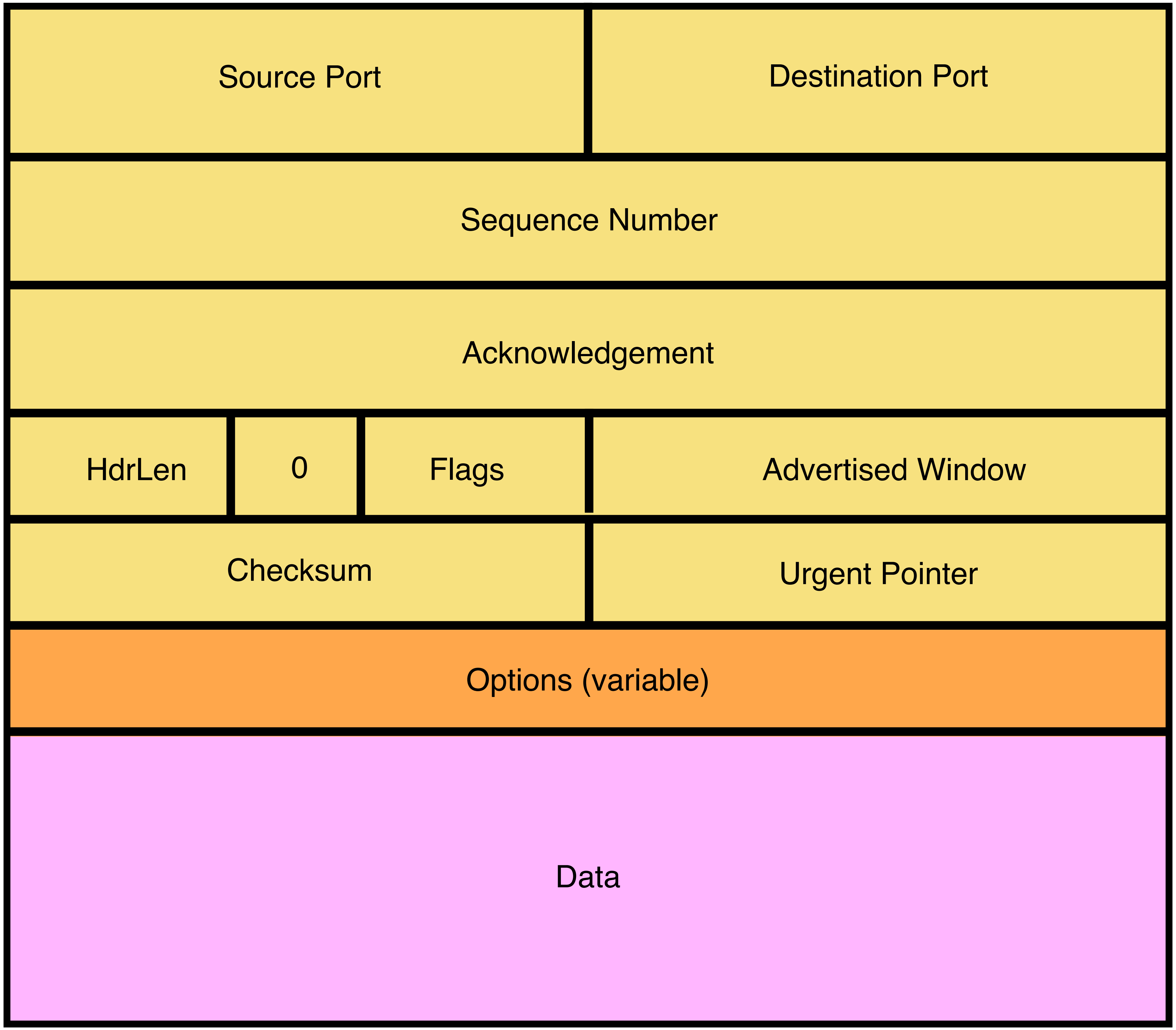
IP Packet Structure



IP Packet Structure

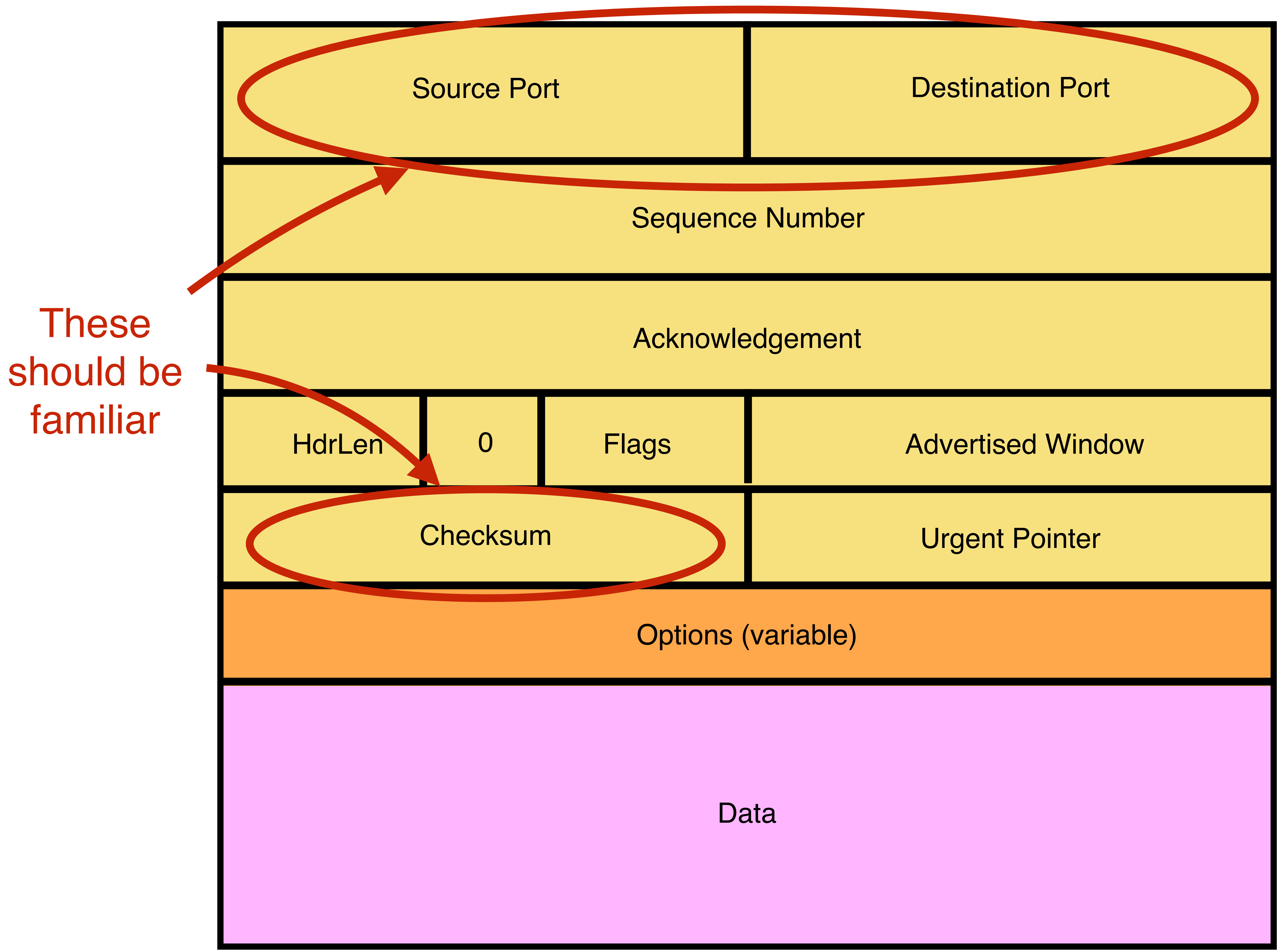


TCP Header



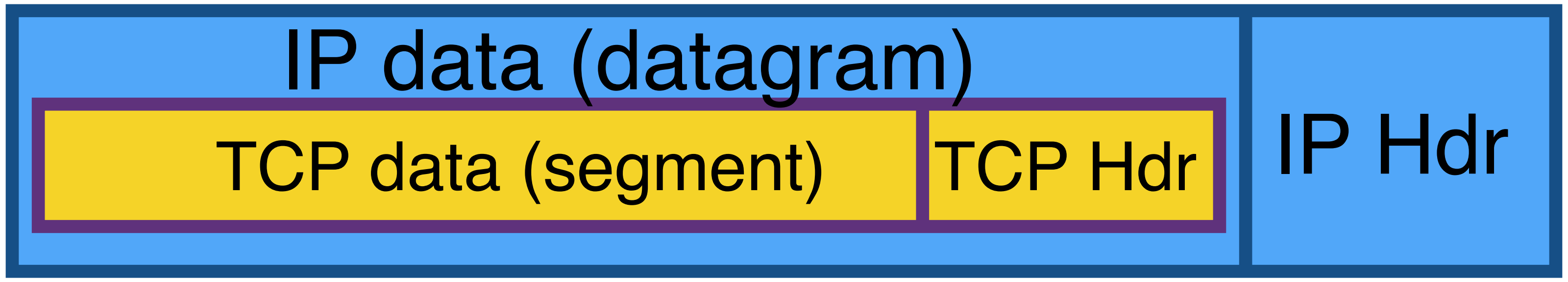
Source Port
Destination Port
Sequence Number
Acknowledgement
HdrLen
0
Flags
Advertised Window
Checksum
Urgent Pointer
Options (variable)
Data

TCP Header



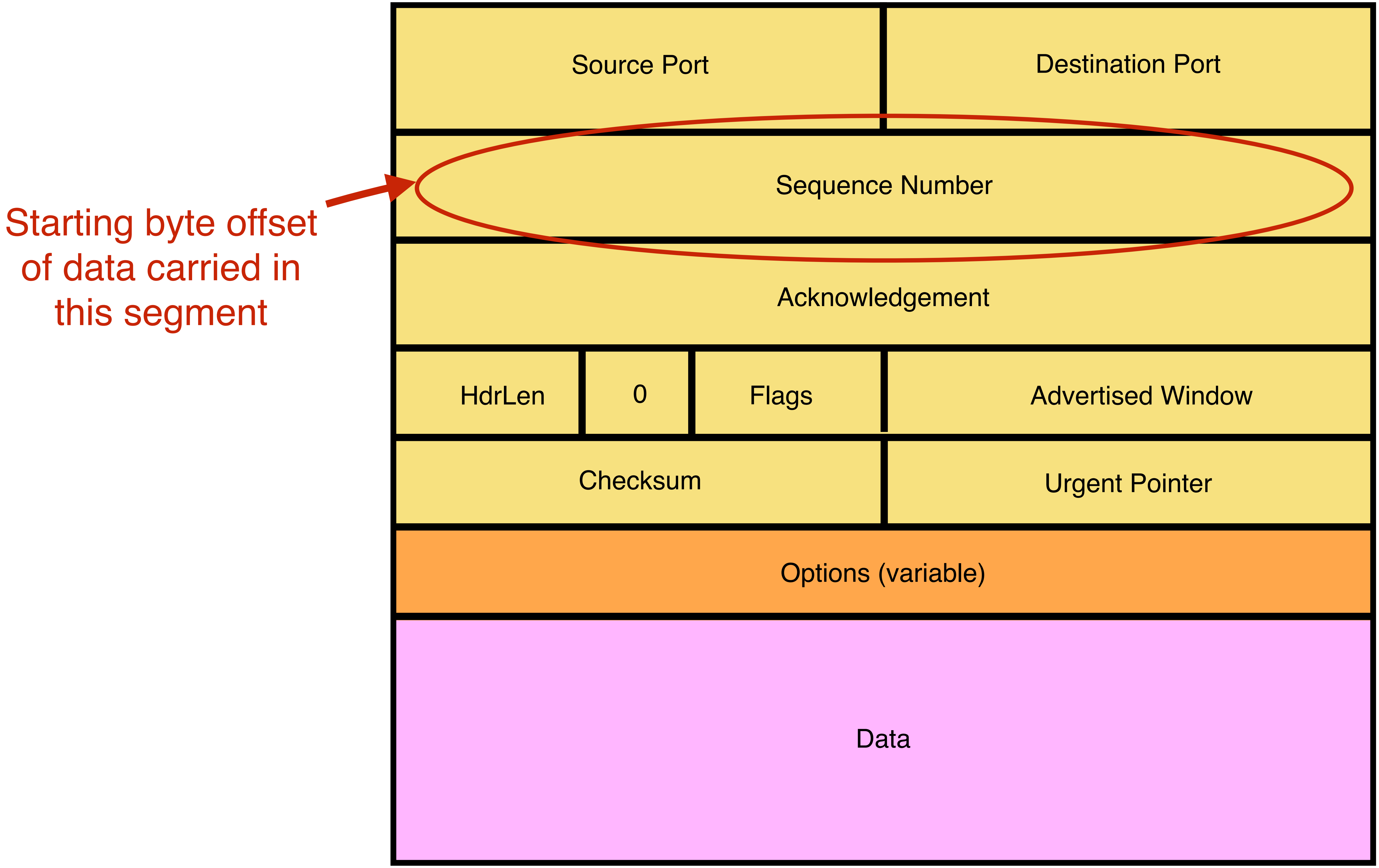
Source Port
Destination Port
Sequence Number
Acknowledgement
HdrLen
0
Flags
Advertised Window
Checksum
Urgent Pointer
Options (variable)
Data
These should be familiar

TCP Segment



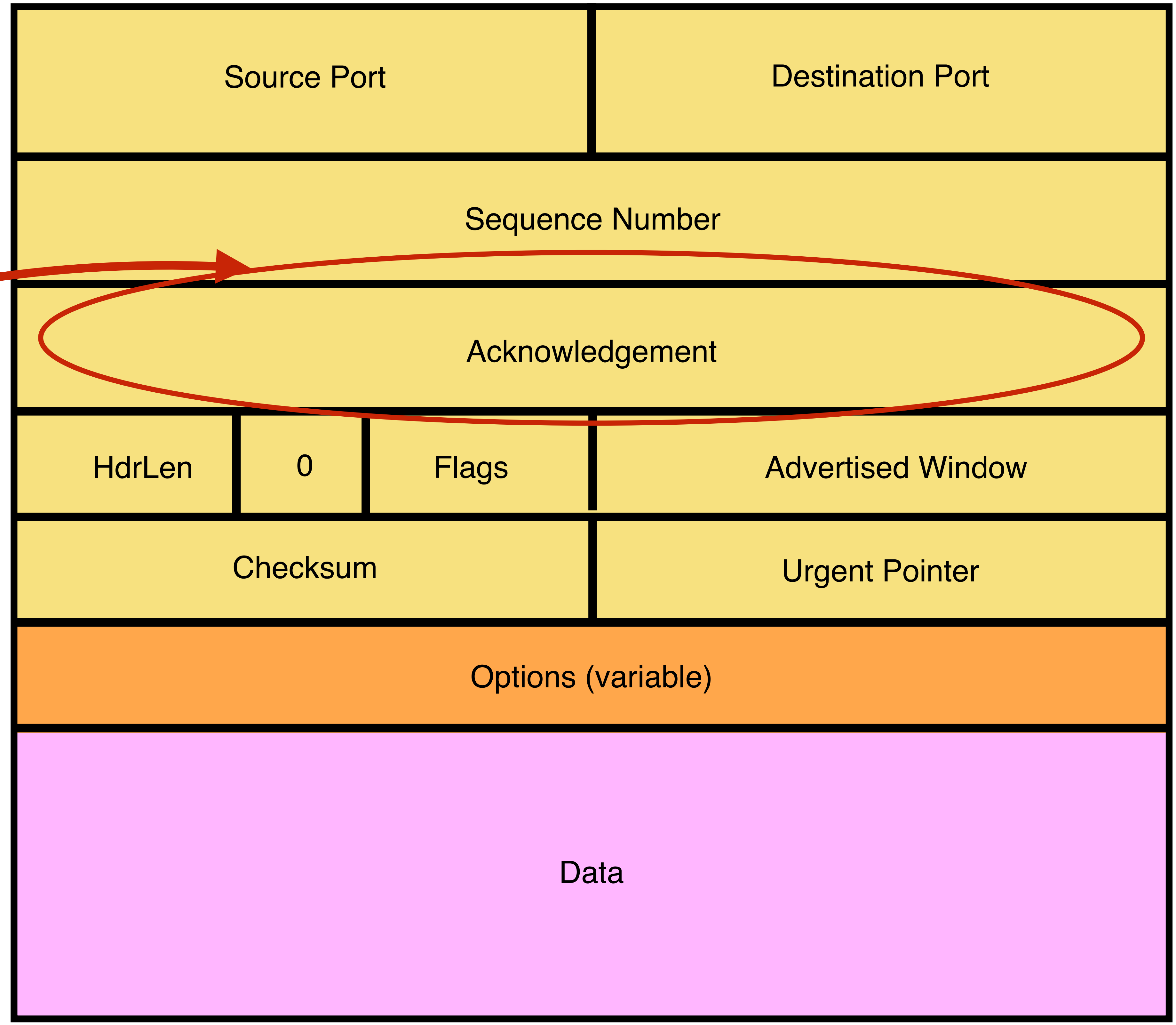
- IP Packet
  - No bigger than Maximum Transmission Unit (MTU)
  - E.g., up to 1500 bytes with Ethernet
- TCP Packet
  - IP packet with a TCP header and data inside
  - TCP header >= 20 bytes long
- TCP Segment
  - No more than MSS (Maximum Segment Size) bytes
  - E.g., upto 1460 consecutive bytes from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

TCP Header

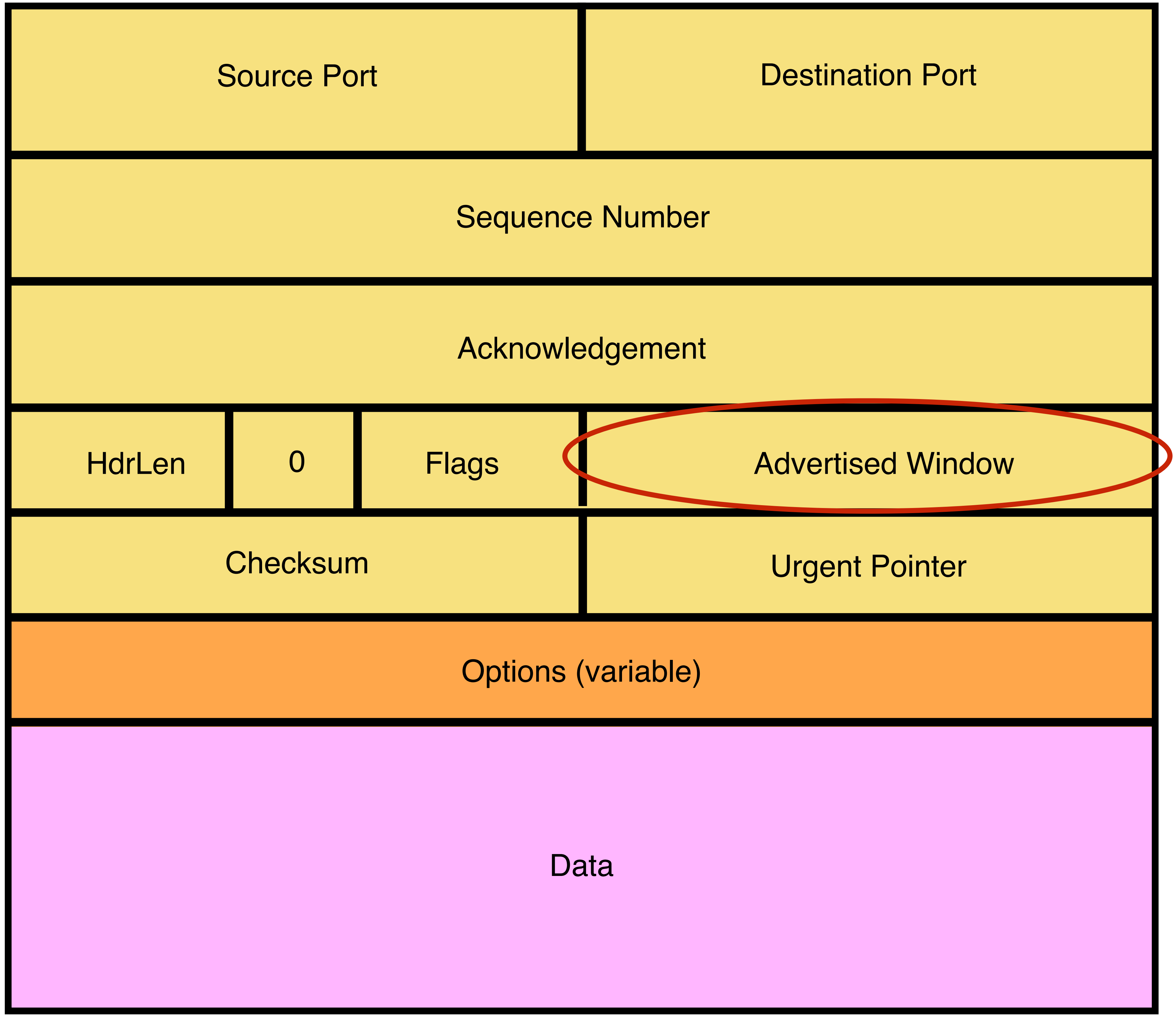

Source Port
Destination Port
Sequence Number
Acknowledgement
HdrLen
0
Flags
Advertised Window
Checksum
Urgent Pointer
Options (variable)
Data
Starting byte offset of data carried in this segment

TCP Header

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

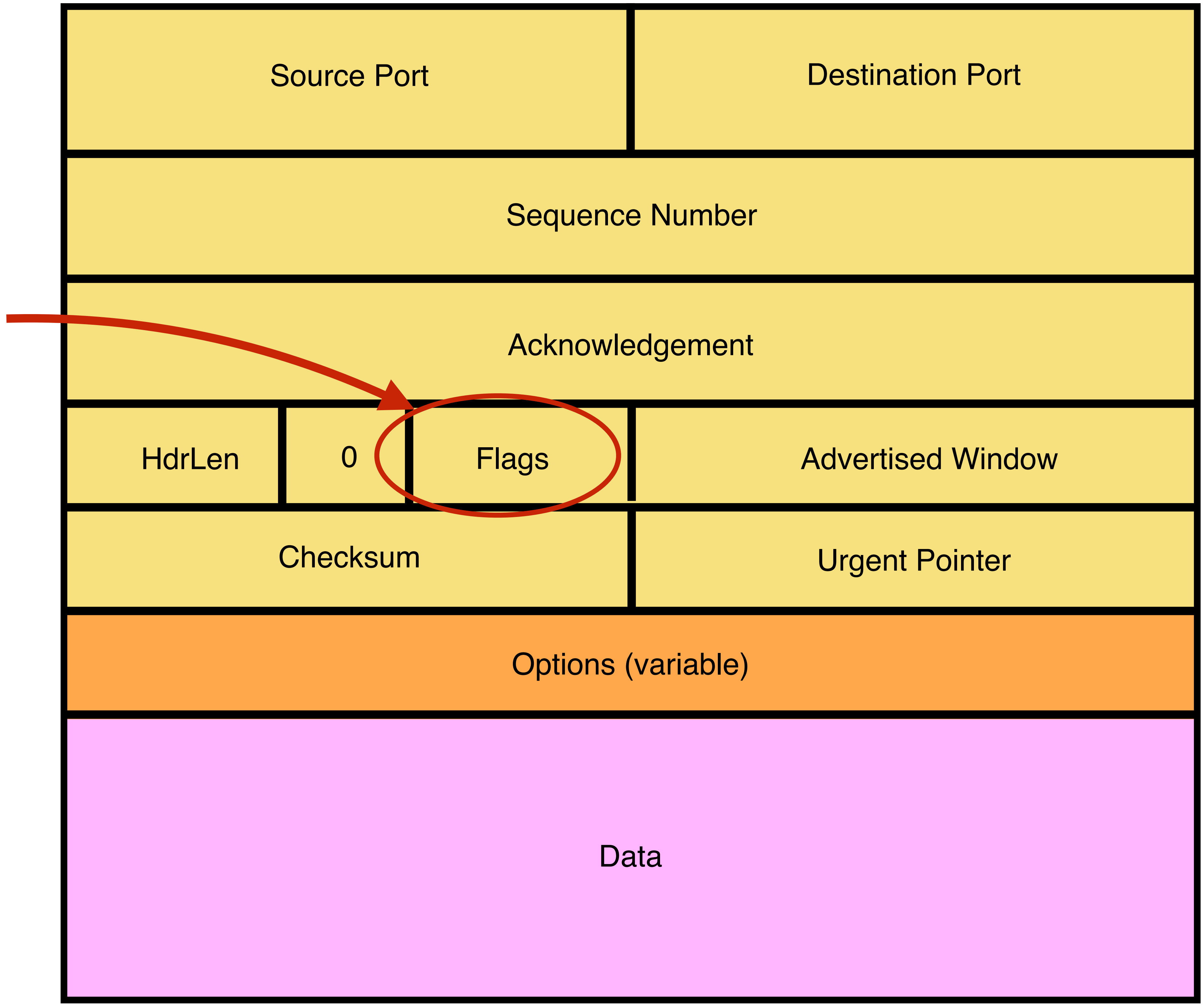


TCP Header


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

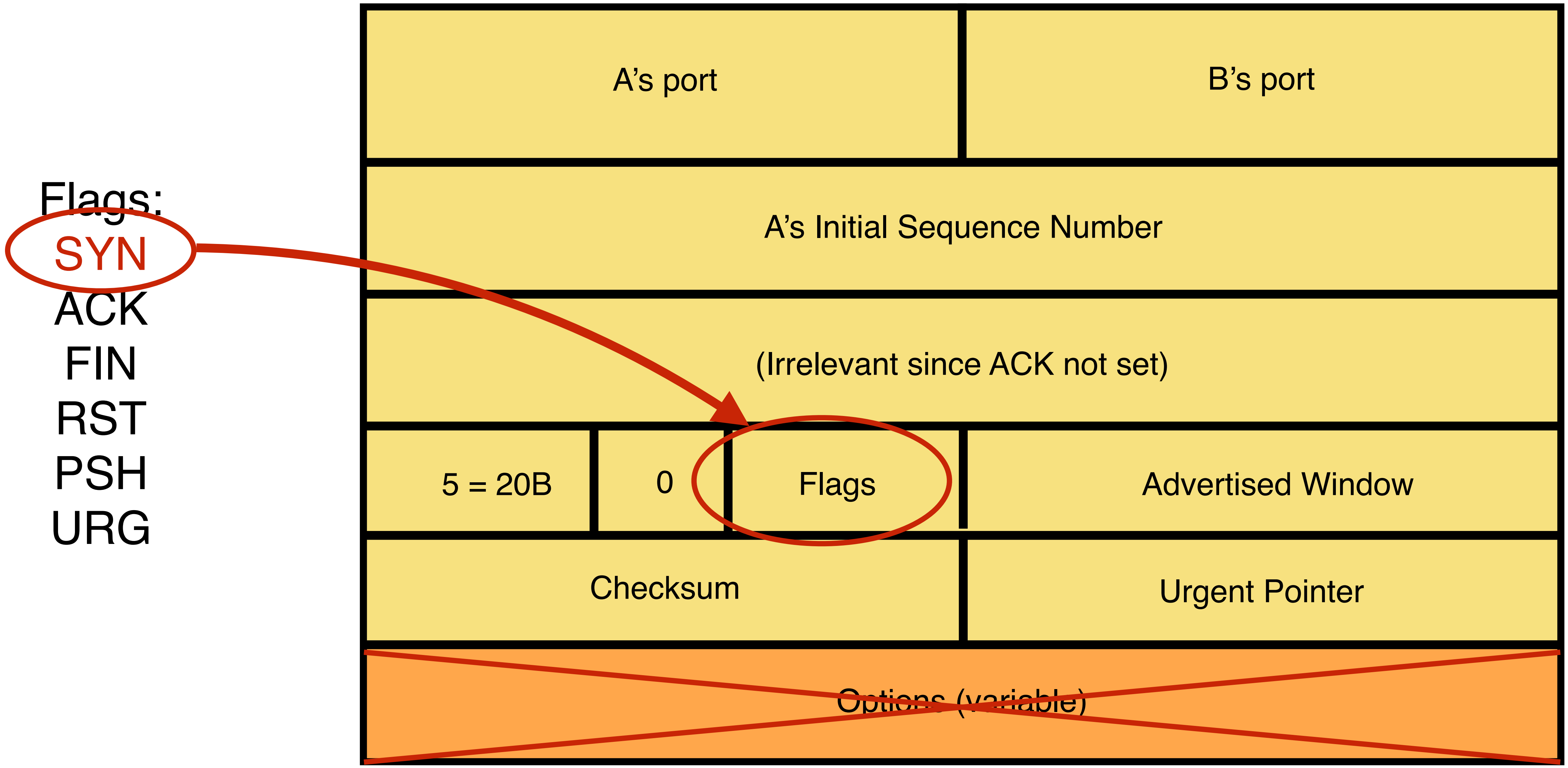
TCP Header

Flags:
<u>**SYN**</u>
<u>**ACK**</u>
FIN
RST
PSH
URG



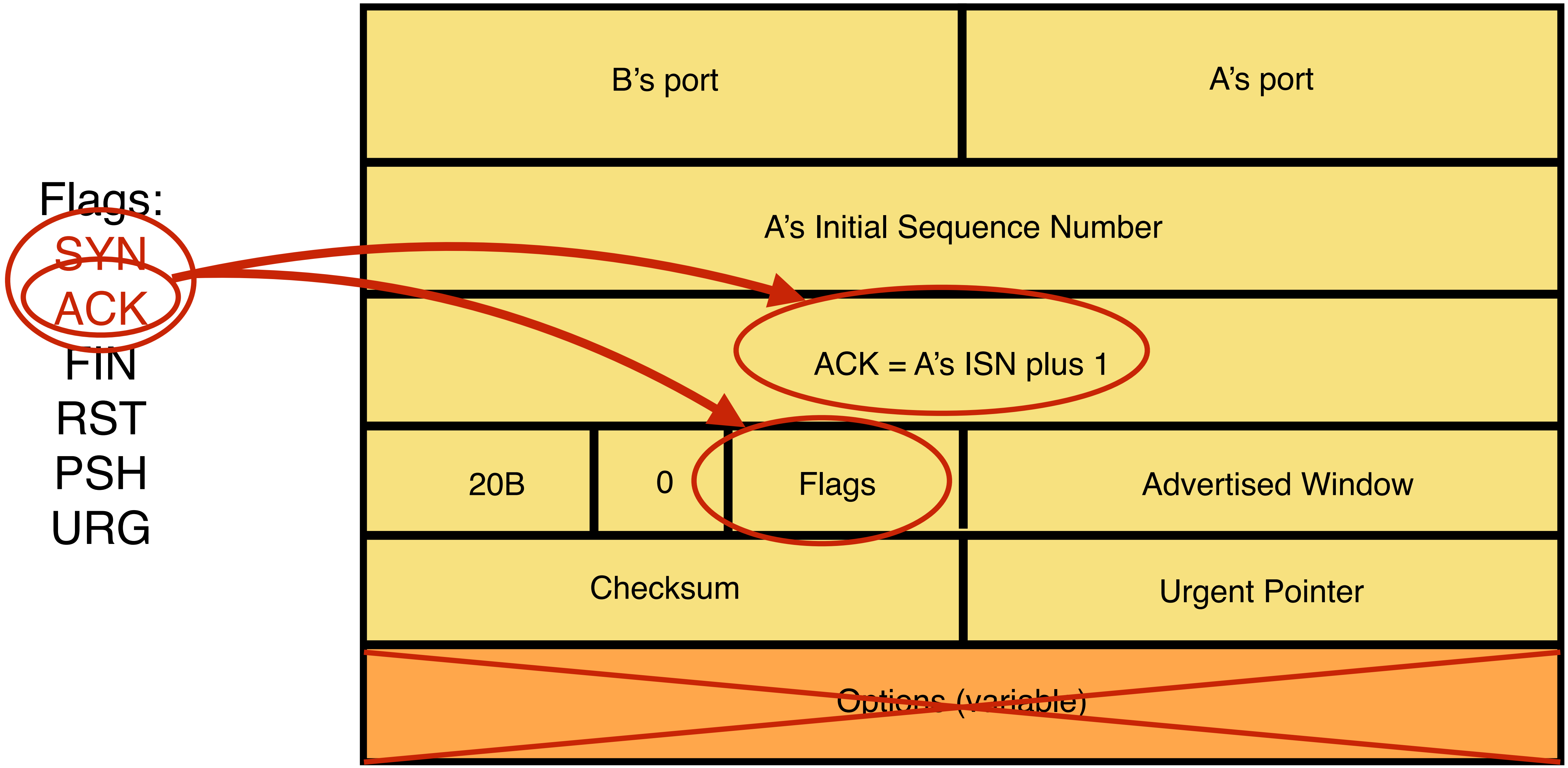
See /usr/include/netinet/tcp.h on Unix Systems

Step 1: A's Initial SYN Packet



A tells B it wants to open a connection…

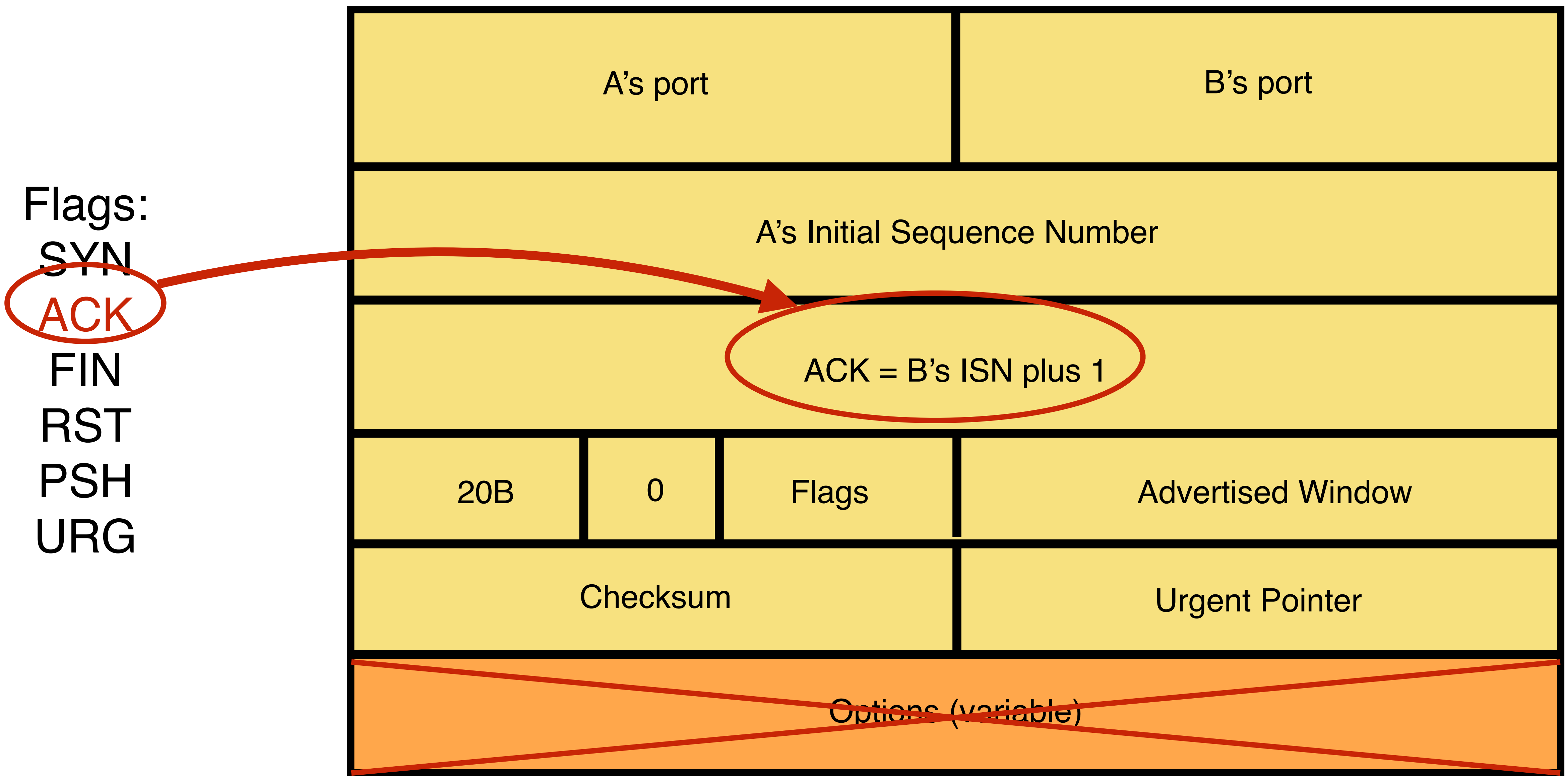
Step 2: B's SYN-ACK Packet



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

… upon receiving this packet, A can start sending data

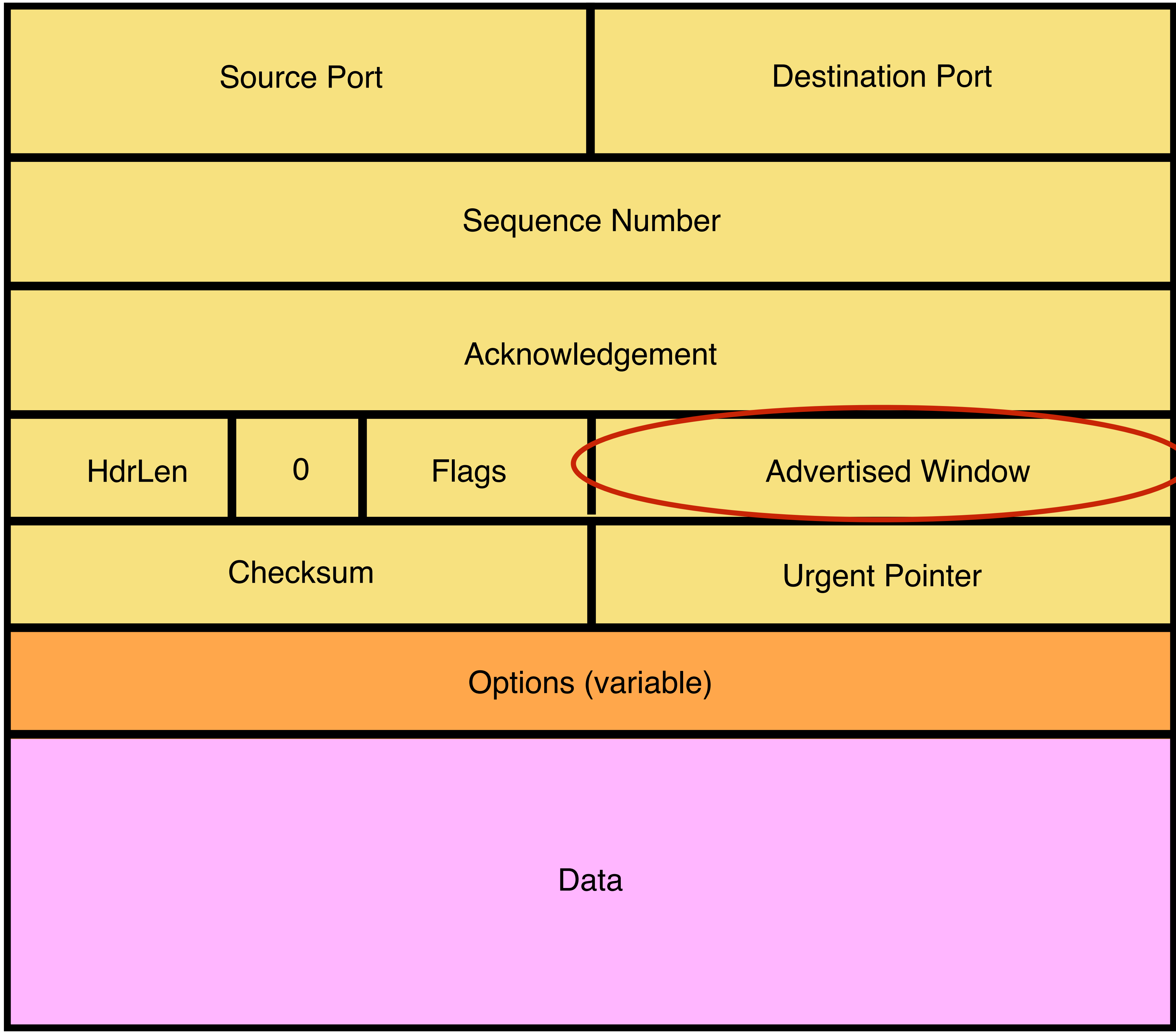
Step 3: A's ACK of the SYN-ACK



A tells B it's likewise okay to start sending

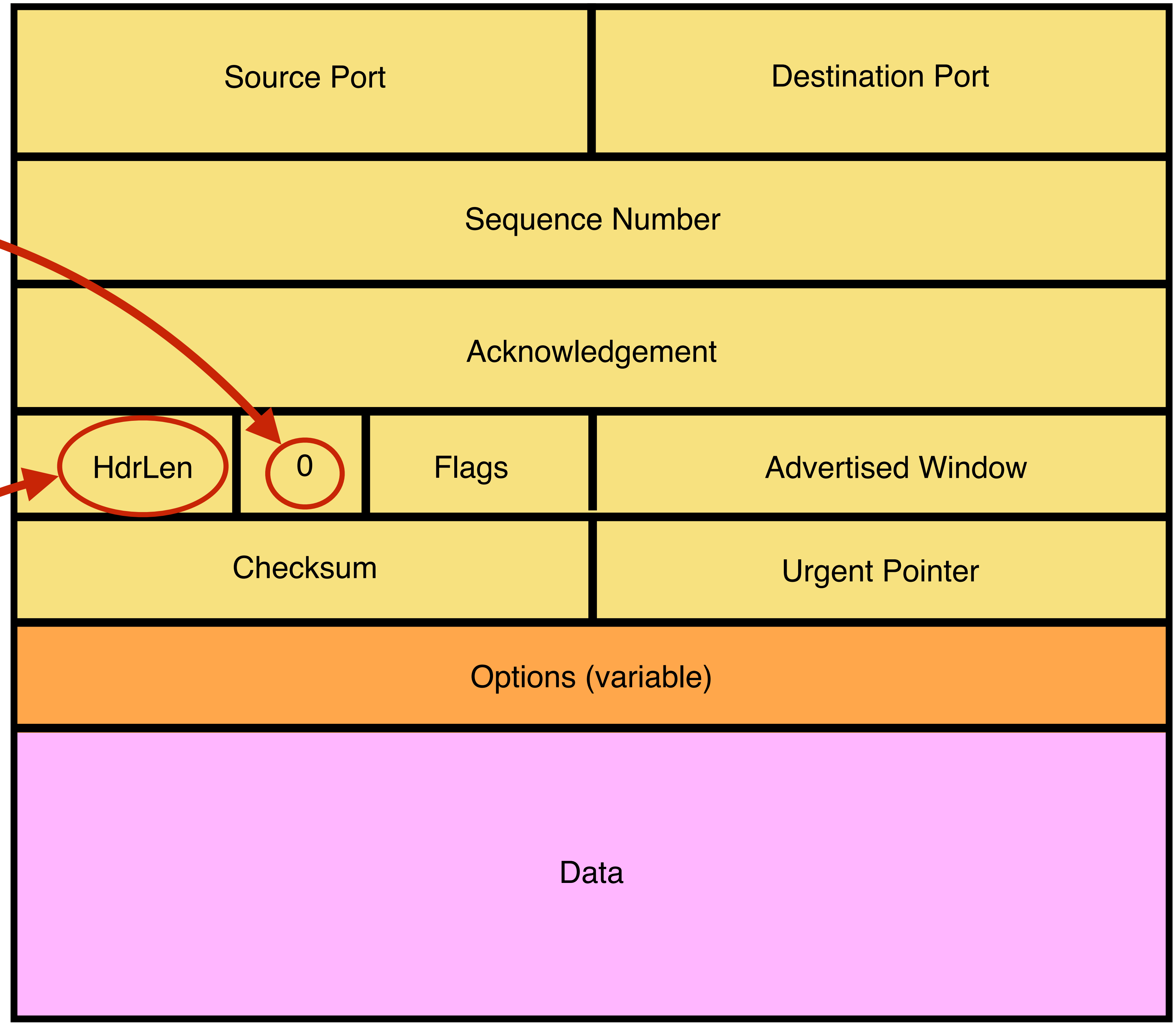
… upon receiving this packet, B can start sending data

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?

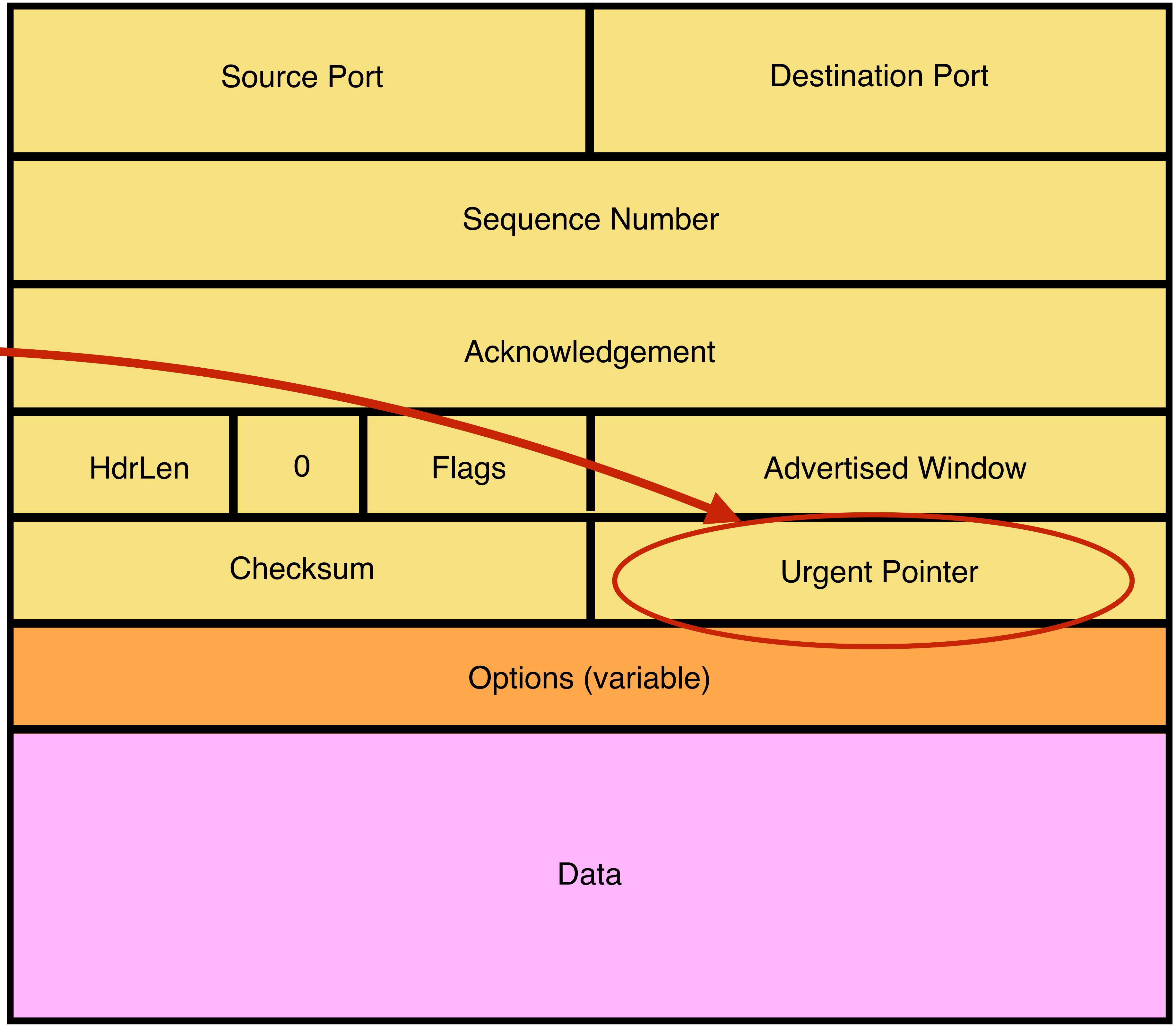
“Must be Zero”
6 bits reserved



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

TCP Header: What's left?

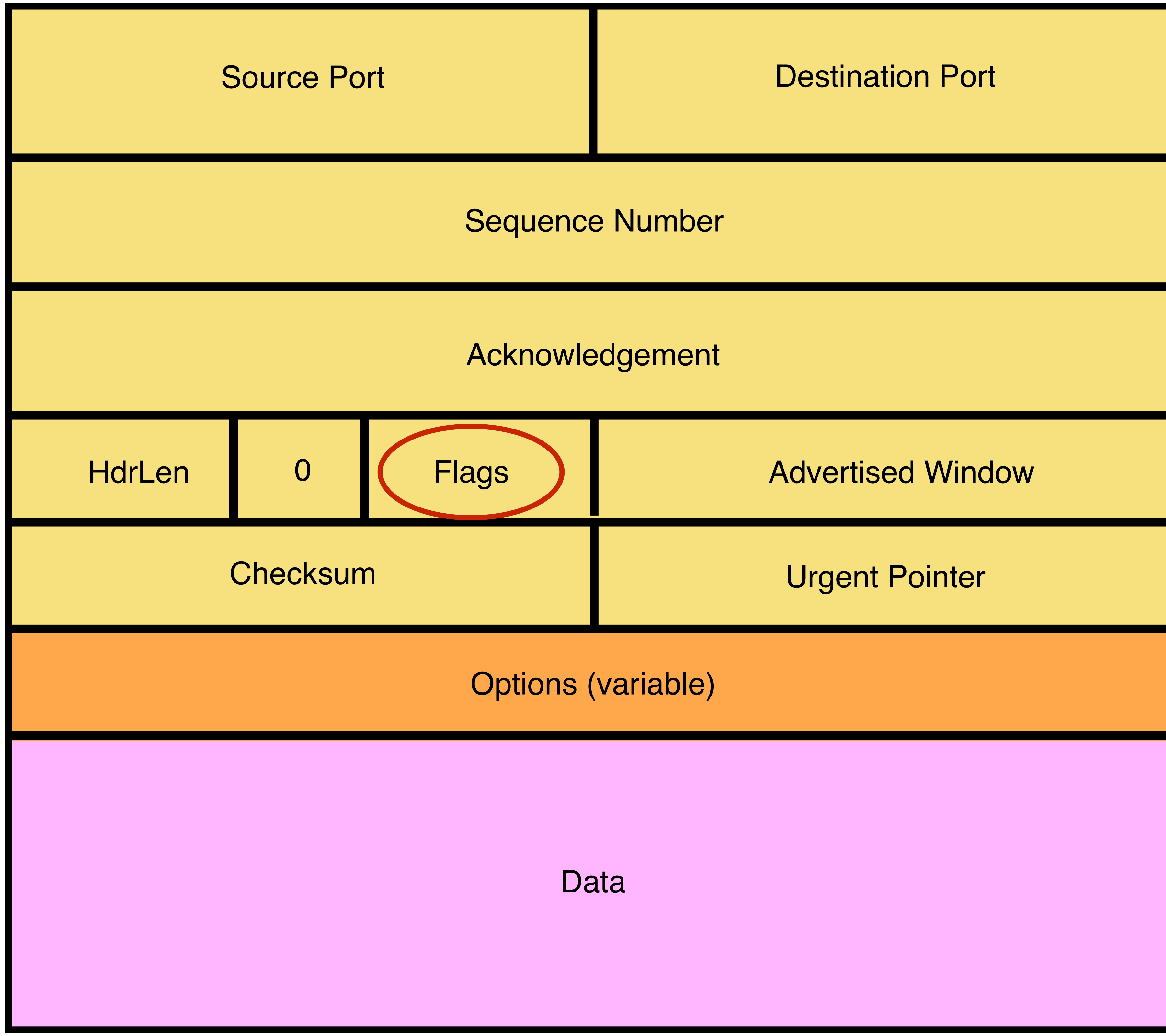
Used with URG flag to indicate urgent data (not discussed further)



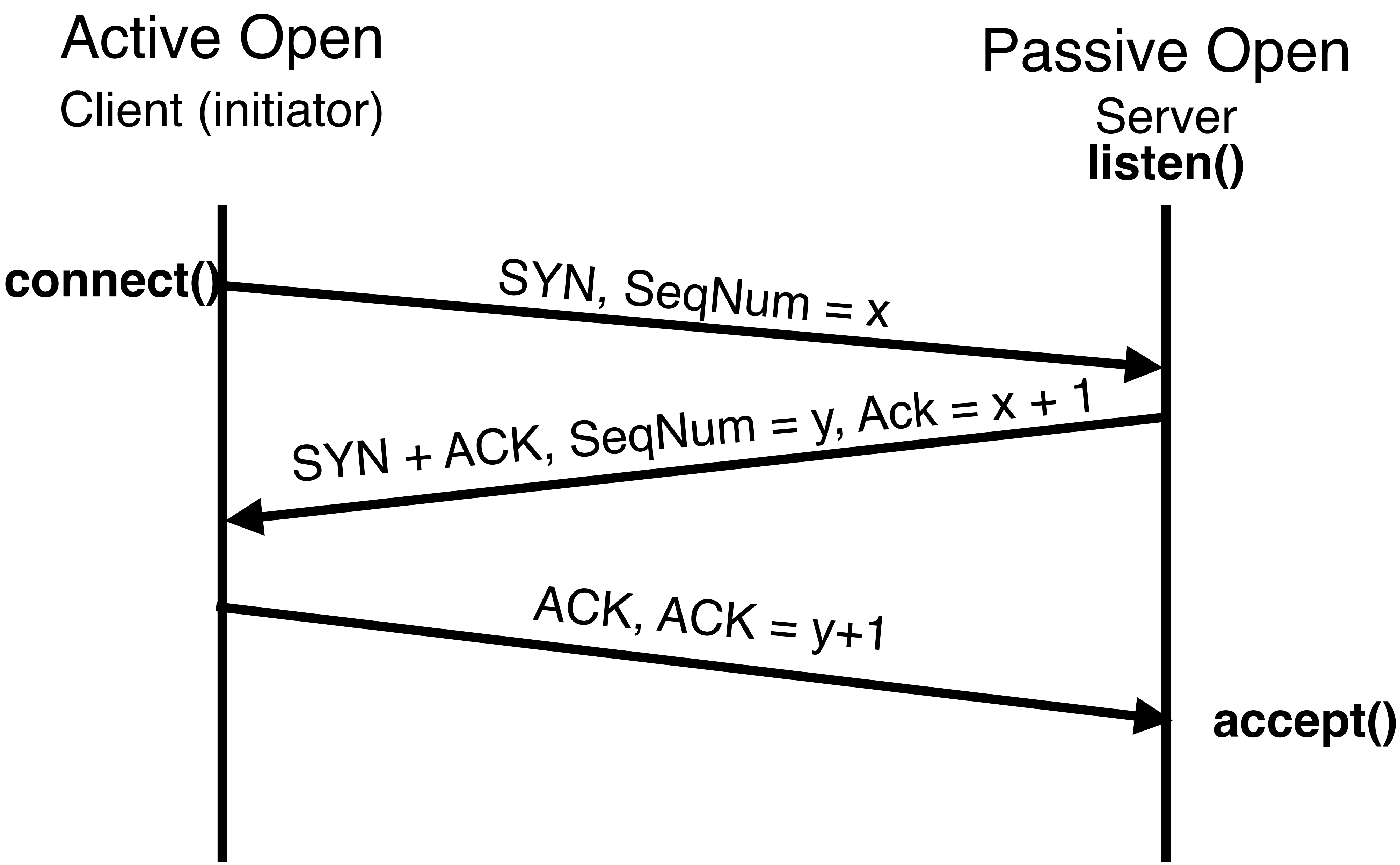
Implementing Sliding Window

- 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?

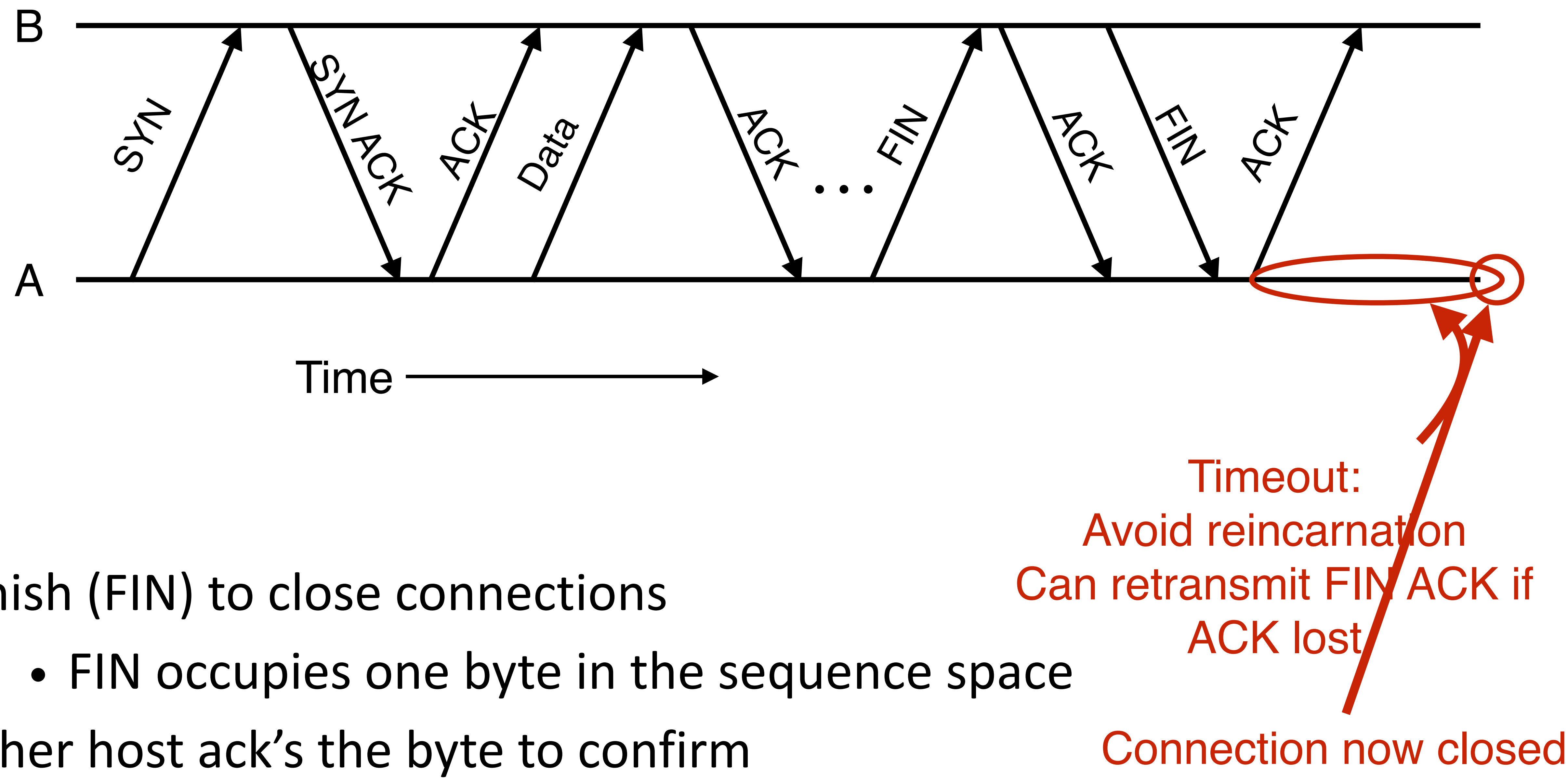
- Suppose the SYN packet gets lost
  - Packet is lost inside the network, or
  - Server **discards** the packet (e.g., listen queue is full)
- Eventually, no SYN-ACK arrives
  - Sender sets a **timer** and **waits** for the SYN-ACK
  - … and retransmits the SYN if needed
- How should the TCP sender set 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

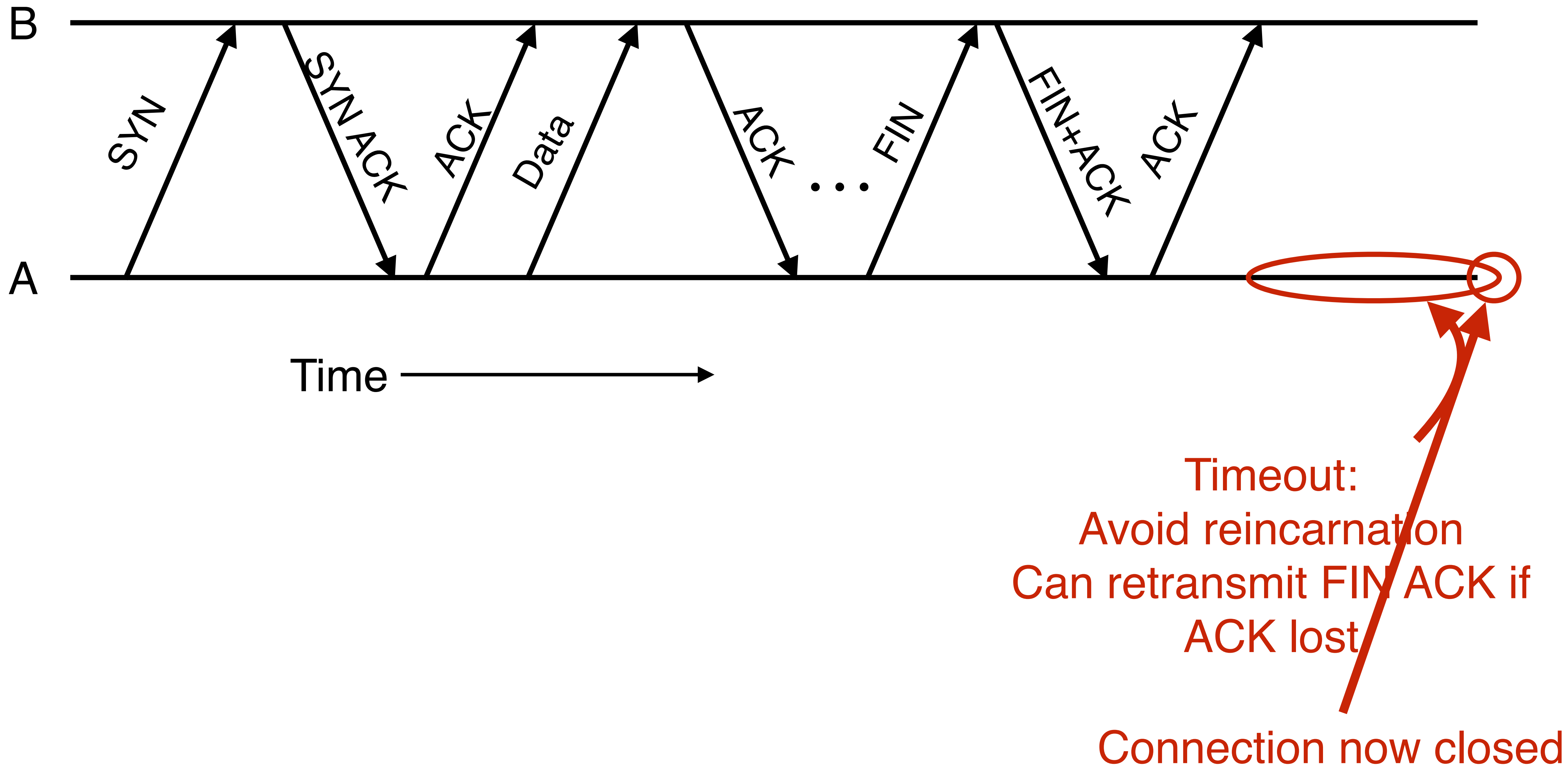
Normal Termination



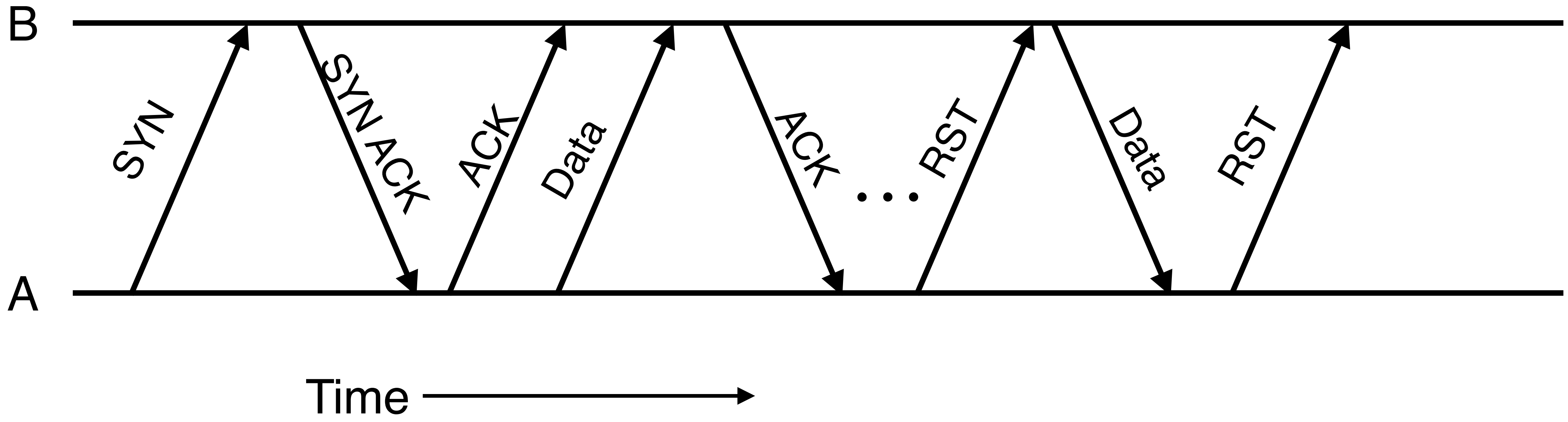
- Finish (FIN) to close connections
    - FIN occupies one byte in the sequence space
- Other host ack's the byte to confirm
- Closes A's side of connection, but not B's
    - Until B likewise sends a FIN
    - Which A then acks

Normal Termination, Both Together

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

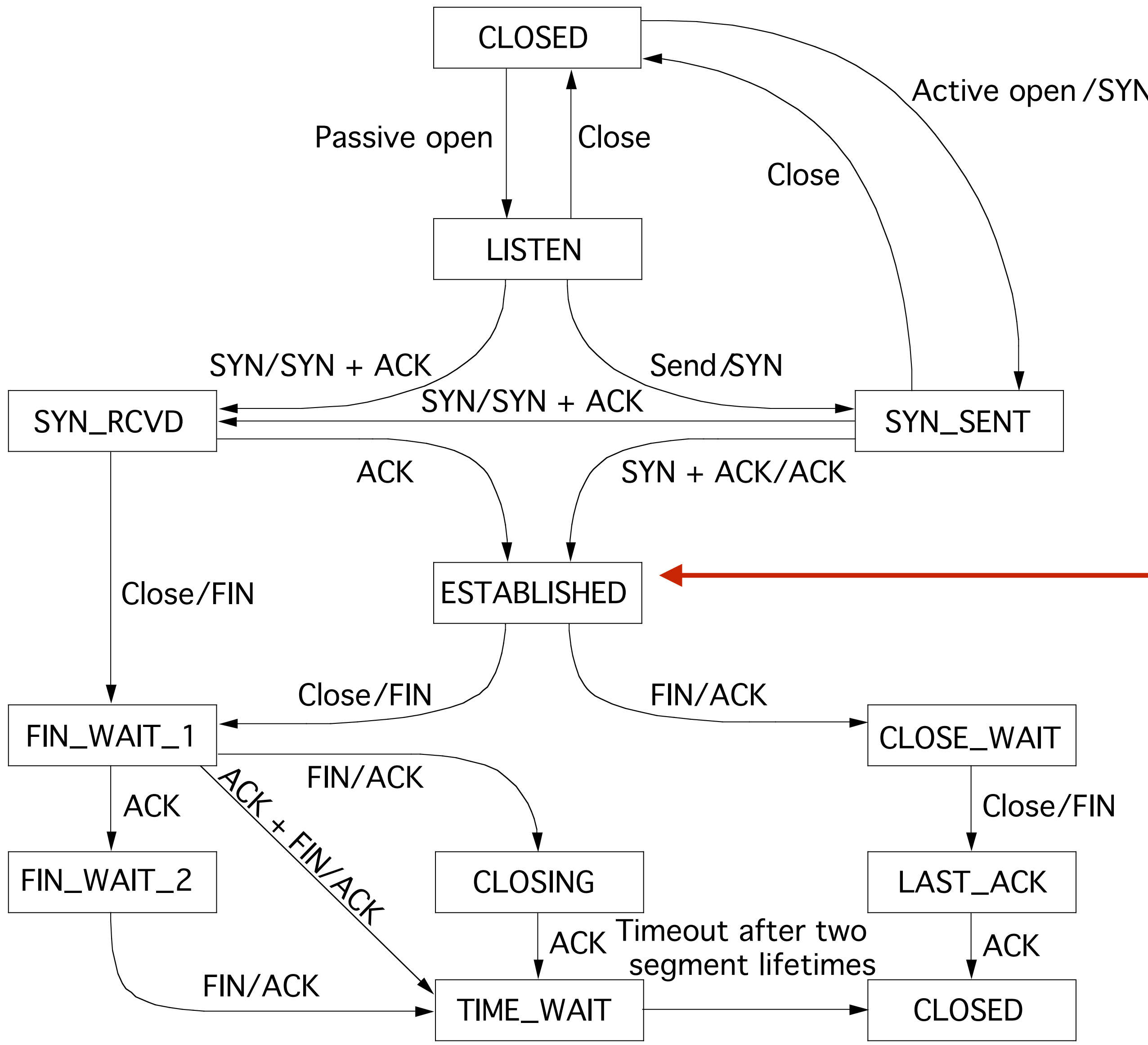


Abrupt Termination



- 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