CS4450

Computer Networks:
Architecture and Protocols

Lecture 20
Reliable Transport and TCP

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Goal of Today's Lecture

- Continue our understanding of reliable transport **conceptually**
- Understanding TCP will become infinitely easier
  - TCP involves lots of detailed mechanisms
  - **Knowing WHY TCP uses these mechanisms is most important**

Lets start with recapping last lecture

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**
- …

How can you possible make anything work with such a service model?

Recap: Four Goals for Reliable Transfer

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

Recap: Complete Correctness Condition

A transport mechanism is "reliable" if and only if

(a) It resends all dropped or corrupted packets

(b) It attempts to make progress

Recap: Solution v1

- **Send every packet as often and fast as possible…**

- Not correct
    - **if** condition **not** satisfied: Transport must **attempt to make progress**
    - No way to check whether the packet was dropped or corrupted
        - So, must continue sending the same packet
    - Showed **why we need receiver feedback**

Recap: Solution v2

- Resend packet until you get an ACK
  - And receiver sends per-packet ACKs until data finally stops
- Correct
- Fair
- Good but suboptimal performance
- Suboptimal utilization
  - **A specific kind of under-utilization:**
    - **The source is unnecessarily sending the same packet**
  - Showed **why we must wait for an ACK after sending a packet**
    - **But how long shall we wait for an ACK?**
    - **Indeed, the ACK may be lost as well**

Recap: Solution v3

- Send packet
  - But now, **set a timer**
- receiver sends per-packet ACKs
- If sender receives ACK, done
- If no ACK when timer expires, resend

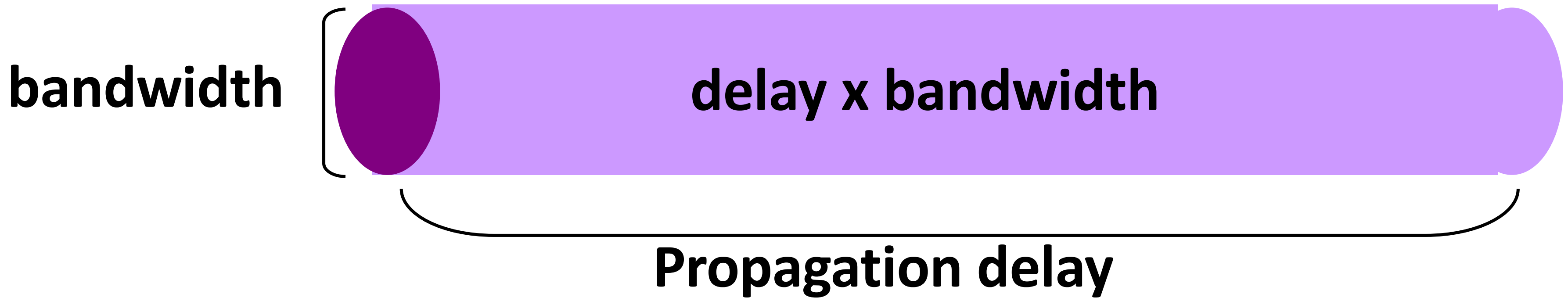
- Correct
- Fair
- Good but suboptimal performance
- Suboptimal utilization
  - **A different kind of under-utilization**
    - **source is not "work conserving": could send, but is not**
  - **What to do while waiting?**
    - **Send more packets**
    - **How many?**

Window-based Algorithms

- Very simple concept
  - Send W packets
  - When one gets ACK'ed send the next packet in line
- **We want to set W such that:**
  - if I am sending at rate = link bandwidth, then
  - the ACK of the first packet arrives
  - exactly when I just finish sending the last of my W packets
  - **(assuming same transmission time for data and ACK packets)**
- **Lets me send as fast as the path can deliver…**

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…**

Where Are We?

- **Figured out correctness condition:**
  - Always resend lost/corrupted packets
  - Always try to make progress (but can give up entirely)
- **Figured out single packet case:**
  - Send packet, set timer, resend if no ACK when timer expires
- **Some progress towards multiple packet case:**
  - Allow many packets (W) in flight at once
  - And know what the ideal window size is
    - RTT x B / Packet size
- What's left to design?

Three Design Considerations

- Nature of feedback
  - What should ACKs tell us when we have many packets in flight
- Detection of loss
- Response to loss

ACK Individual Packets

The receiver sends ACK for each individual packet that it receives

Example:

- Assume that packet 5 is lost, but no others
- Stream of ACKs will be
    - 1
    - 2
    - 3
    - 4
    - **6**
    - **7**
    - **8**
    - …

ACK Individual Packets

- **Nature of feedback**: simple - the receiver ACKs each packet
- **Loss detection:** simple - ACKs tell the fate of each packet to the source
- **Response to loss: moderate**:
  - + Retransmit the packet for which ACK not received
  - + Reordering not a problem
  - + Simple window algorithm
    - W independent single packet algorithms
    - When one finishes grab next packet
  - - **Loss of ACK packet requires a retransmission**

Full Information Feedback

- **List all packets that have been received**
  - Give highest cumulative ACK plus any additional packets

Same Example (suppose packet 5 gets lost):

- Same story, except that the “hole” is explicit in each ACK
- Stream of ACKs will be
  - Up to 1
  - Up to 2
  - Up to 3
  - Up to 4
  - Up to 4, plus 6
  - Up to 4, plus 6,7
  - Up to 4, plus 6,7,8
  - …

Full Information Feedback

- **Nature of feedback**: **complex** - feedback may have high overheads
    - If packets 1, 5, 6, …., 100 received: ACK(1, 5, 6, …,100)
- **Loss detection:** simple - the source still knows fate of each packet
- **Response to loss:** simple:
    - + Retransmit the packet for which ACK not received
    - + Reordering not a problem
    - + Simple window algorithm
    - - **Loss of ACK does not necessarily requires a retransmission**
        - **The next ACK will tell that the packet was indeed received**
        - Resilient form of individual ACKs

Cumulative ACK

- **Individual ACKs** can get lost, and require **unnecessary retransmission**
- **Full information feedback** can handle lost ACKs but has **high overheads**
- **Cumulative ACKs: a sweet spot between the two**
- Just the first part of full information feedback
- ACK the highest sequence number for all previously received packets

Cumulative ACKs (same example; say packet 5 lost)

Full information feedback:

- Stream of ACKs will be
    - Up to 1
    - Up to 2
    - Up to 3
    - Up to 4
    - Up to 4, plus 6
    - Up to 4, plus 6,7
    - Up to 4, plus 6,7,8
    - …

Tells **"which" packet arrived**, and **which packet did not**

Cumulative ACKs:

- Stream of ACKs will be
    - Up to 1
    - Up to 2
    - Up to 3
    - Up to 4
    - Up to 4
    - Up to 4
    - Up to 4
    - …

Tells **"some" packet arrived,** and **which packet did not**

Loss With Cumulative ACKs (cont'd)

- Duplicate ACKs are a sign of loss
  - The lack of ACK progress means 5 hasn't been delivered
  - Stream of duplicate ACKs means some packets are being delivered (one for each subsequent packet)
- Response to loss is trickier… When shall the source retransmit packet 5?
  - Packet may be delayed (so, source should wait)
  - Packet may be reordered (so, source should wait)
  - Or, packet may be dropped (source should immediately retransmit)
  - Impossible to know which one is the case
    - Life lesson: **be optimistic**!
    - Until optimism starts hurting
  - **Solution: retransmit after k duplicate ACKs**
    - **for some value of k, depending on how optimistic you feel!**

Cumulative ACKs (how is reordering handled; large k)

Receiver events:

- Packet 1 received
- Packet 2 received
- Packet 3 received
- Packet 4 received
- **Packet 6 received**
- Packet 7 received
- **Packet 5 received**
- **Packet 8 received**
- …

Cumulative ACKs:

- Up to 1
- Up to 2
- Up to 3
- Up to 4
- **Up to 4**
- Up to 4
- **Up to 7**
- **Up to 8**
- …

Cumulative ACKs **naturally handle packet reordering**

(Packet delays are similar to reordering)

Cumulative ACKs (confusion with duplication)

- Produce duplicate ACKs
  - Could be confused for loss with cumulative ACKs
  - But duplication is rare…

Source events:	Receiver events:	Cumulative ACKs:
• Packet 1 sent	• Packet 1 received	• Up to 1
• Packet 2 sent	• Packet 2 received	• Up to 2
• Packet 3 sent	• **Packet 4 received**	• **Up to 2**
• Packet 4 sent	• Packet 5 received	• **Up to 2**
• Packet 5 sent	• Packet 6 received	• **Up to 2**
• Packet 6 sent	• **Packet 3 received**	• Up to 6
• **Packet 3 resent**	• **Packet 3 received**	• **Up to 6**
• Packet 7 sent	• Packet 7 received	• **Up to 7**
• …	• …	• …

Possible Design For Reliable Transport

- Cumulative ACKs
- Window based, with retransmissions after
  - Timeout
  - k subsequent ACKs
- This is correct, high-performant and high-utilization
  - At least as much as we can efficiently
- How about fairness?

Fairness? (Come back to later)

- The question of fairness comes up when:
  - Senders want to send data at rate higher than bandwidth
  - There will be packet loss!
- Adjust W based on losses…
- In a way that flows receive same shares
- Short version:
  - Loss: cut W by 2
  - Successful receipt of window: W increased by 1

Overview of Reliable Transport

- Window based self control separate concerns
  - Size of W
  - Nature of feedback
  - Response to loss
- Can design each aspect relatively independently
- Can be correct, fair, high-performant and high-utilization
- All of these are important concerns
  - **But correctness is most fundamental**
- Design **must** start with correctness
  - Can then "engineer" its performance with various hacks
  - These hacks can be "fun", but don't let them distract you

What Have We Done so far?

- 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)

- … to understanding **why cumulative ACKs may be a good idea**
    - Very close to modern TCP

- **You are now ready to learn TCP**

Lets learn TCP

Transport layer

- Transport layer offer a “pipe” abstraction to applications
- Data goes in one end of the pipe and emerges from other
- **Pipes are between processes, not hosts**
- There are two basic pipe abstractions

Two Pipe Abstractions

- **Unreliable packet** delivery (UDP)
  - Unreliable (application responsible for resending)
  - Messages limited to single packet
- **Reliable byte stream** delivery
  - Bytes inserted into pipe by sender
  - They emerge, in order at receiver (to the app)
- What features must transport protocol implement to support these abstractions?

UDP (Datagram Messaging Service)

- Sources send packets
- **Destinations do nothing**, but receive packets
- If packets delayed/reordered/lost:
    - Meh!
    - Let application handle packet loss (or be oblivious to drops)
    - If application needs reliable delivery, it must use reliable transport
- Discarding corrupted packets (optional)
- Nothing else!
- A minimal extension of IP

TCP (Reliable, In Order Delivery)

- Source send **segments**
- Destinations send ACKs
- Source retransmits lost and/or corrupted **segments**
- Sources perform **Flow control** (to not overflow receiver)
- Sources perform **Congestion control** (to not overload network)
- Source and destination participate in “Connection” set-up and tear-down

Connections (Or Sessions)

- Reliability requires keeping state
  - Sender: packets sent but not yet ACKed, and related timers
  - Receiver: packets that arrived out-of-order
- Each byte stream is called a **connection** or **session**
  - Each with their own connection state
  - State is in hosts, not network

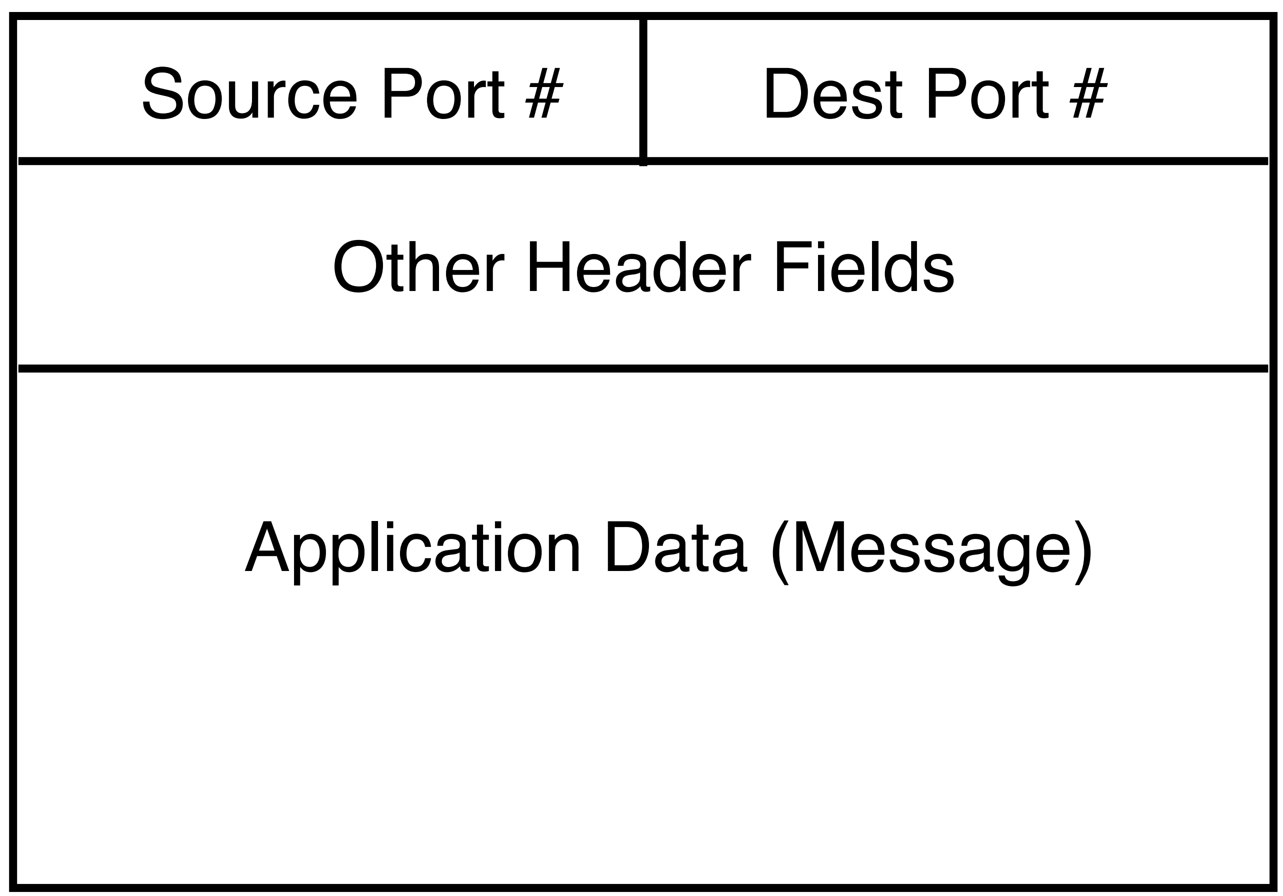
Ports

- 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

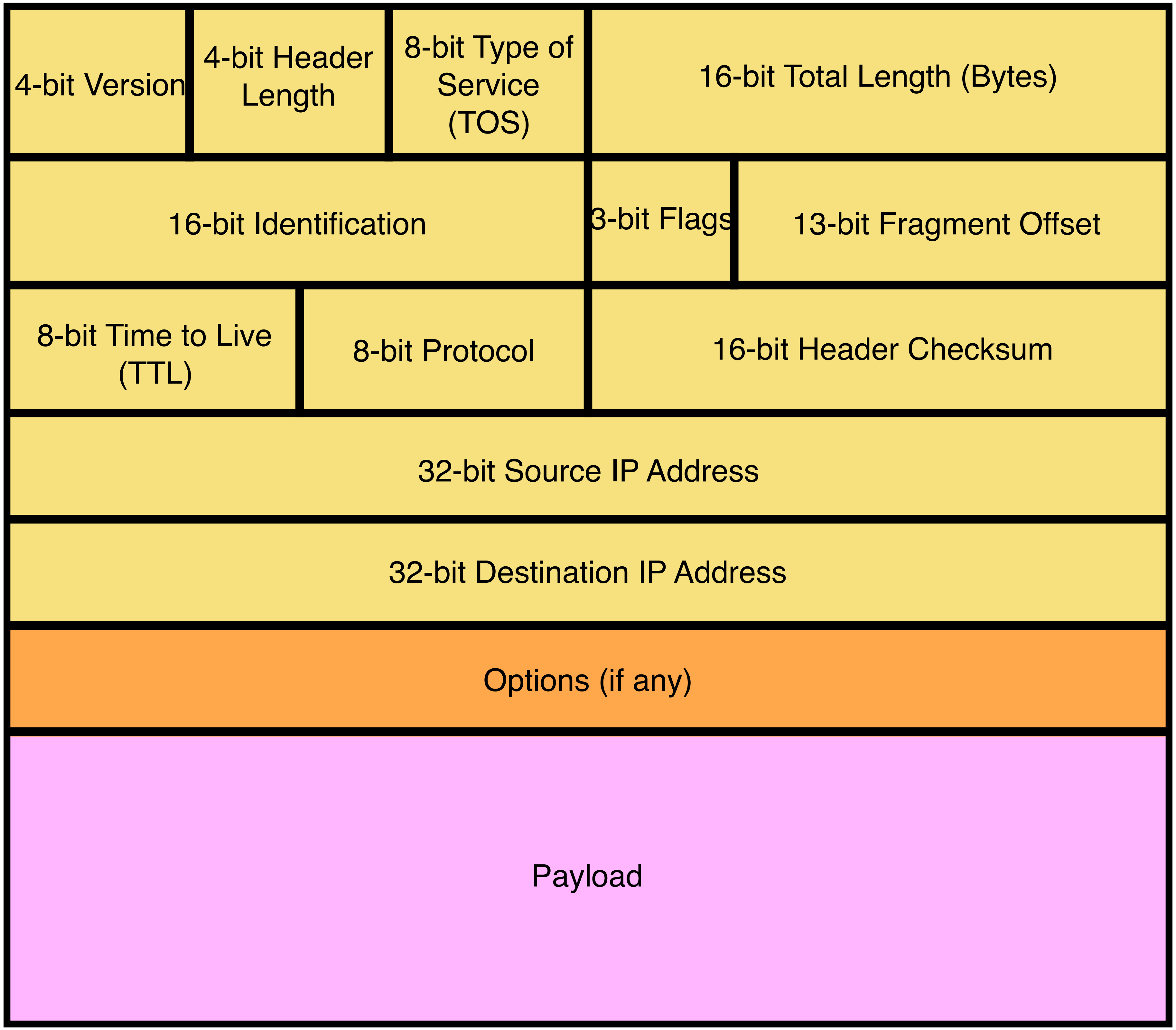
Multiplexing and Demultiplexing

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

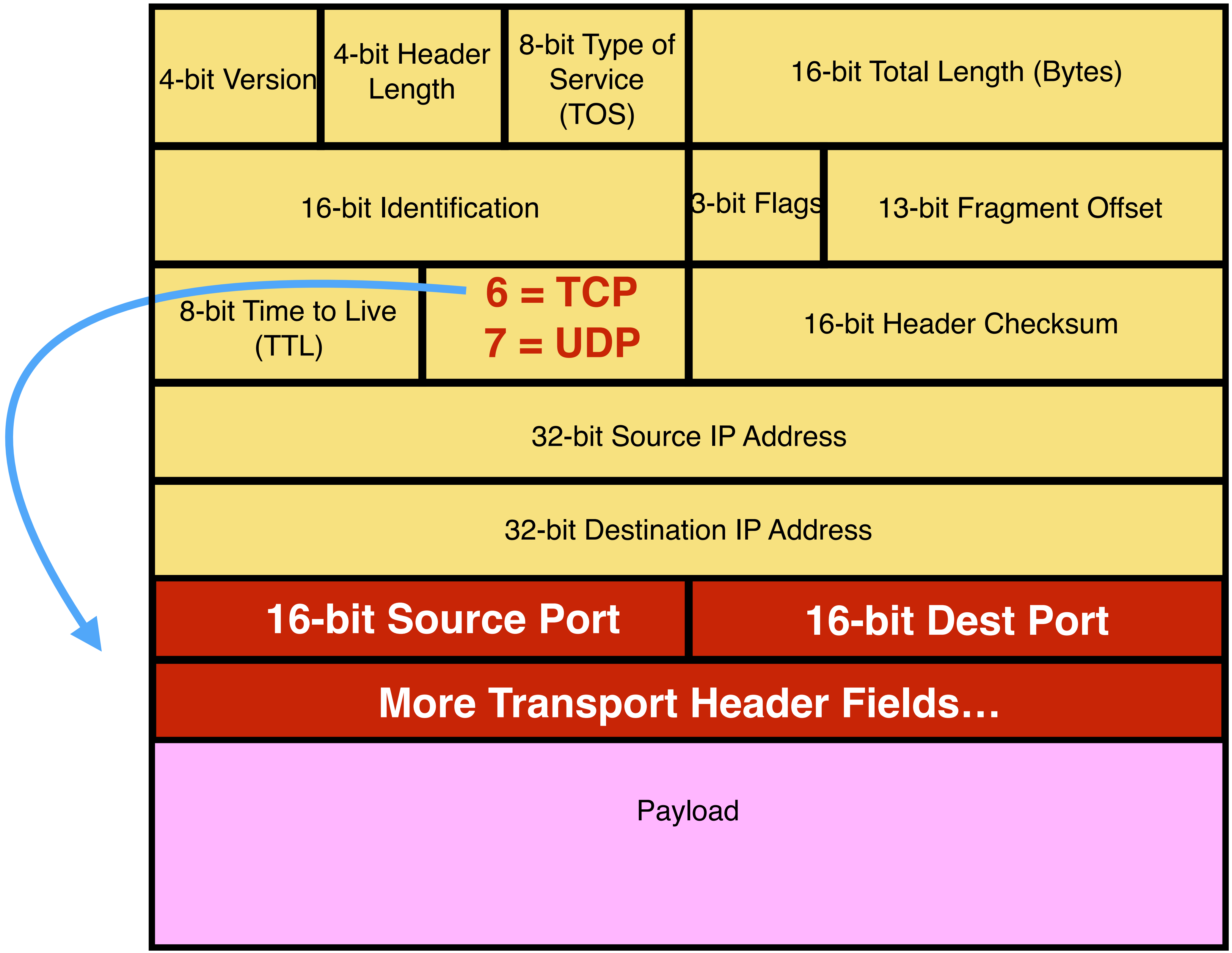
- Host uses IP address and port numbers to direct the segment to appropriate socket



IP Packet Structure



IP Packet Structure



Any Questions?

Transmission Control Protocol (TCP)

- Reliable, in-order delivery
    - Ensures byte stream (eventually) arrives intact
    - In the presence of corruption, delays, reordering, loss
- Connection oriented
    - Explicit set-up and tear-down of TCP session
- Full duplex stream of **byte service**
    - **Sends and receives stream of bytes, not messages**
- **Flow control**
    - Ensures the sender does not overwhelm the receiver
- **Congestion control**
    - Dynamic adaptation to network path's capacity

From design to implementation: 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

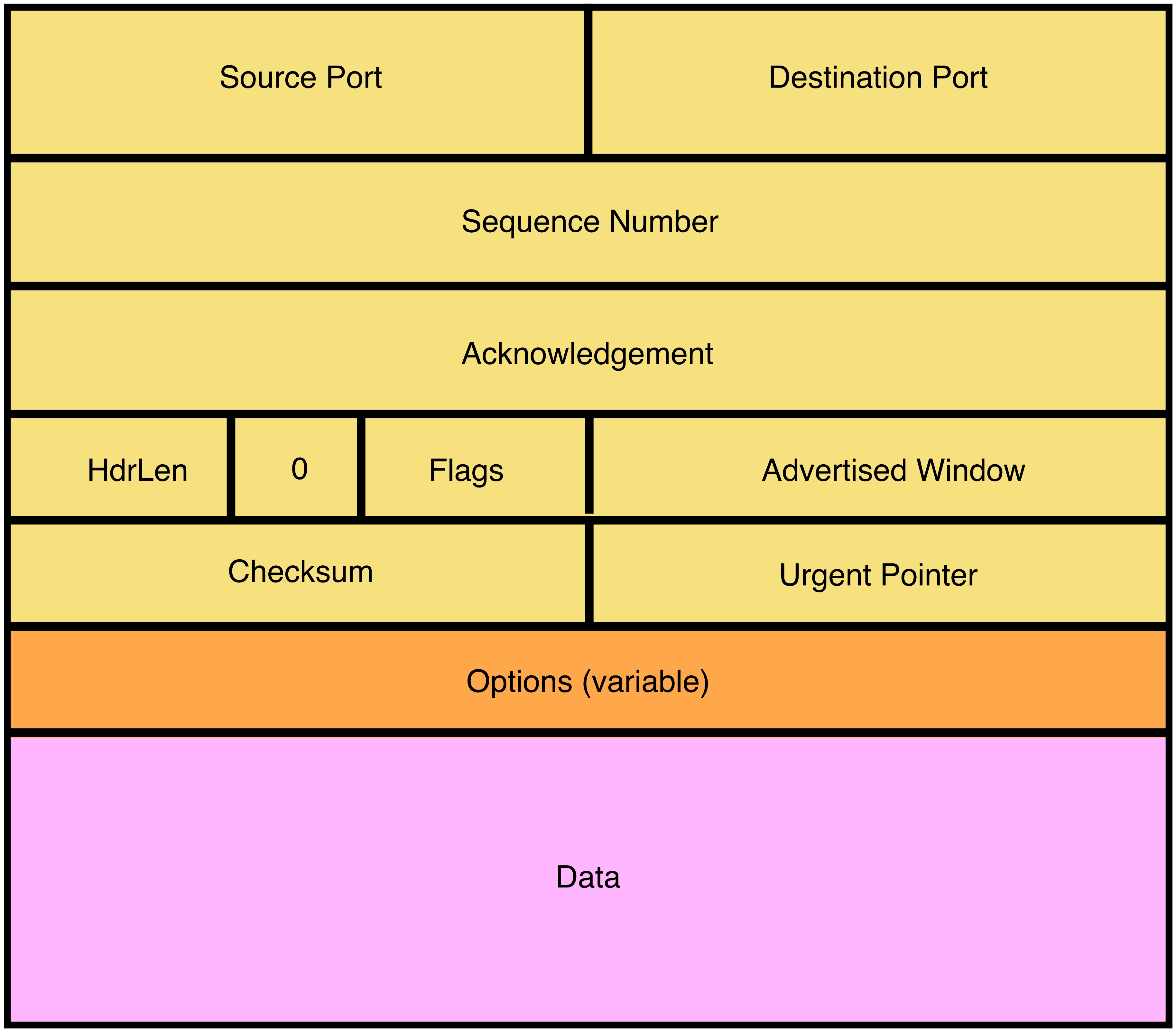
Basic Components of Reliability

- ACKs
  - TCP uses byte sequence numbers to identify payloads
  - ACKs referred to those numbers
- Timeouts and retransmissions
  - Can’t be reliable without retransmitting lost/corrupted data
  - **TCP retransmits based on timeouts and duplicate ACKs**
  - **Timeouts based on estimate of RTT**

Other TCP Design Decisions

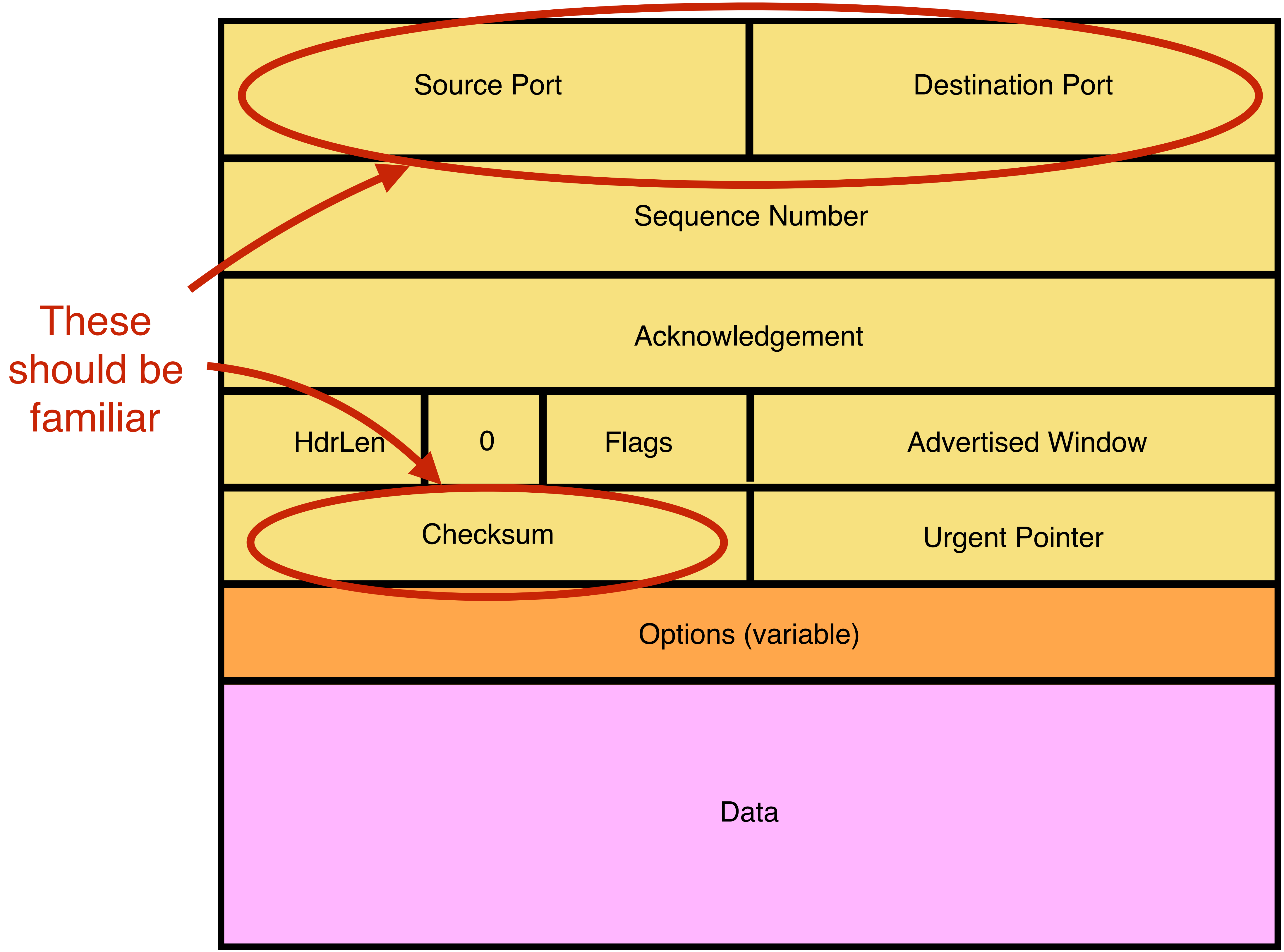
- Sliding window flow control
  - Allow W contiguous bytes to be in flight
- Cumulative Acknowledgements
  - Selective ACKs (full information) also supported (ignore)
- Set timer after each payload is ACK'ed
  - Timer is effectively for the "next expected payload"
  - When the timer goes off, resend that payload and wait
    - And double timeout period
- Various tricks related to "fast retransmit"

TCP Header



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

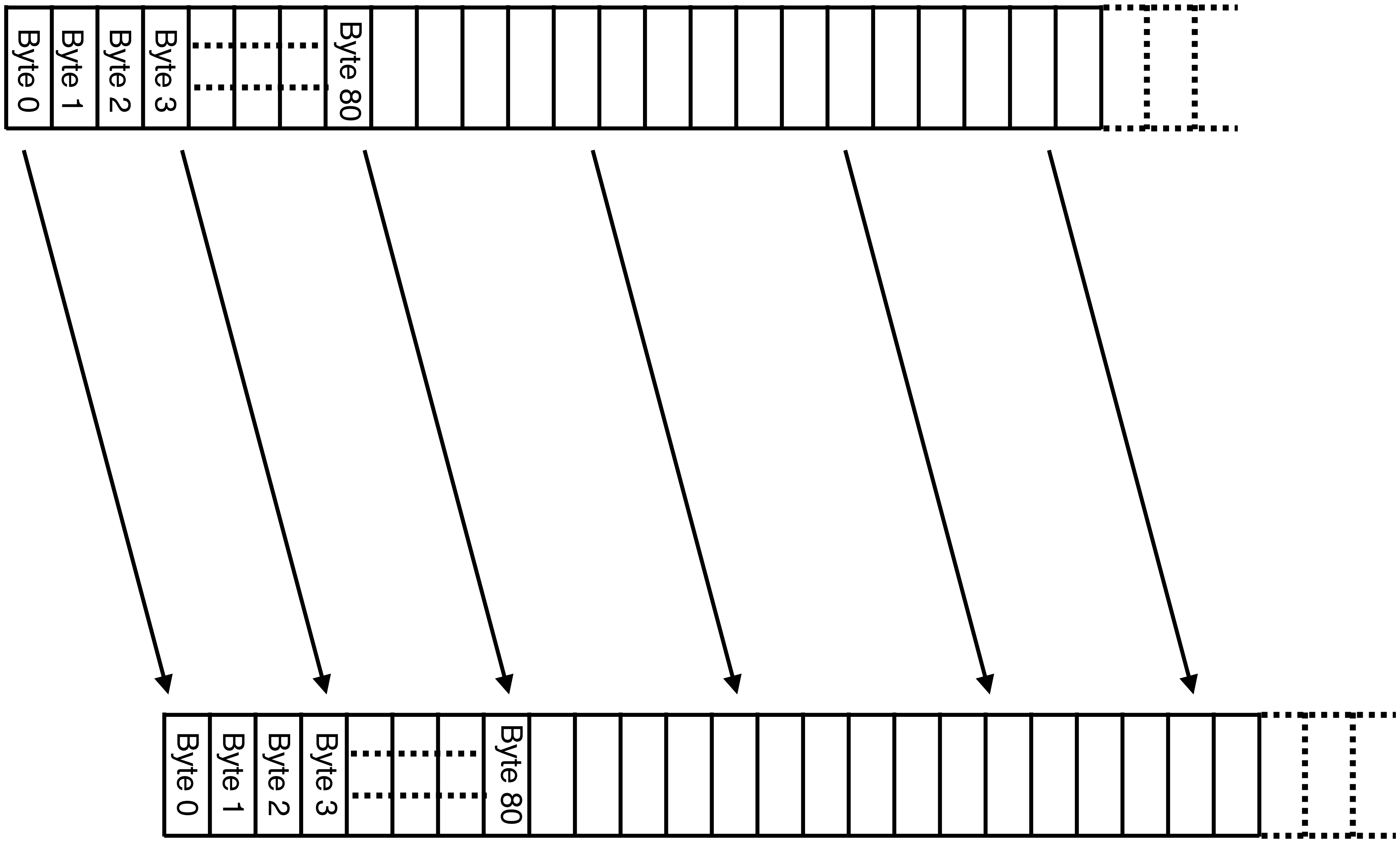
TCP Header


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

Segments and Sequence Numbers

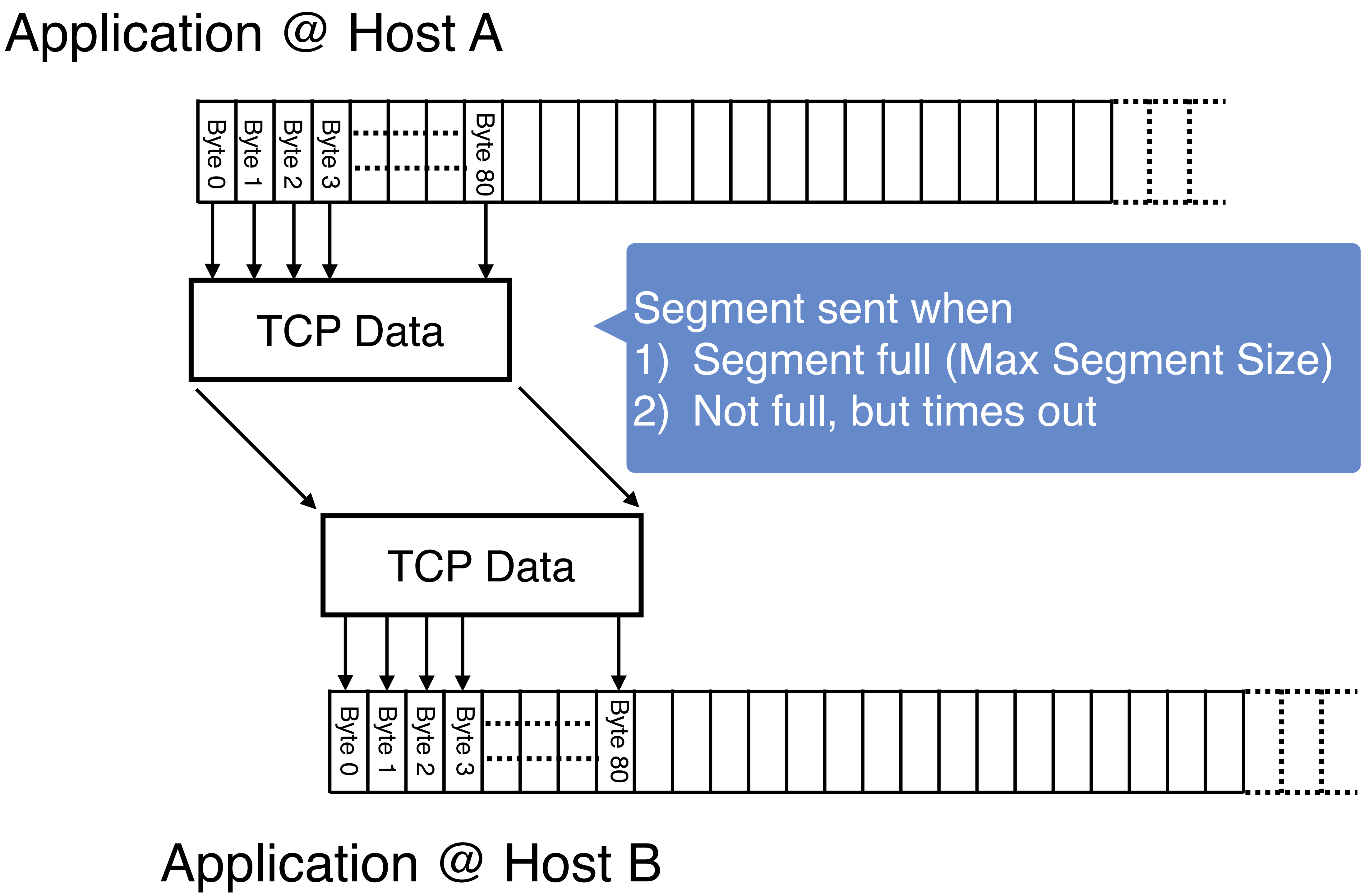
TCP “Stream of Bytes” Service

Application @ Host A

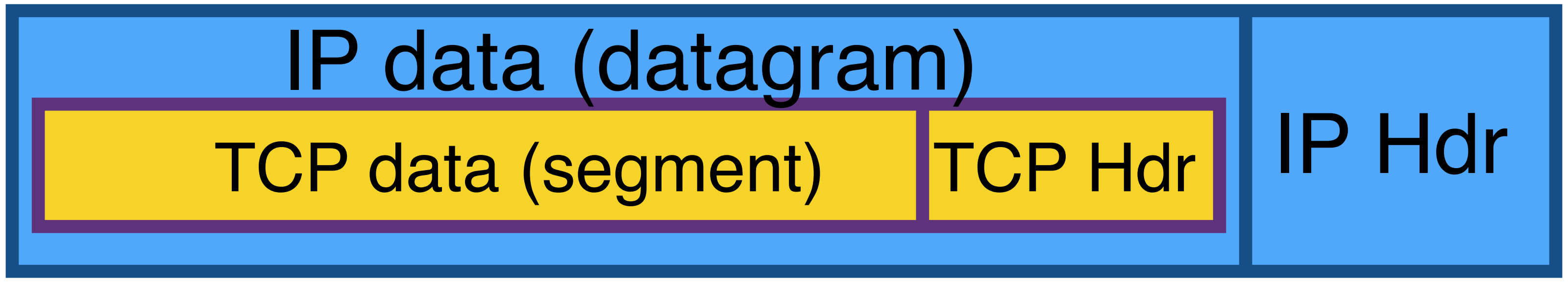


Application @ Host B

TCP “Stream of Bytes” Service

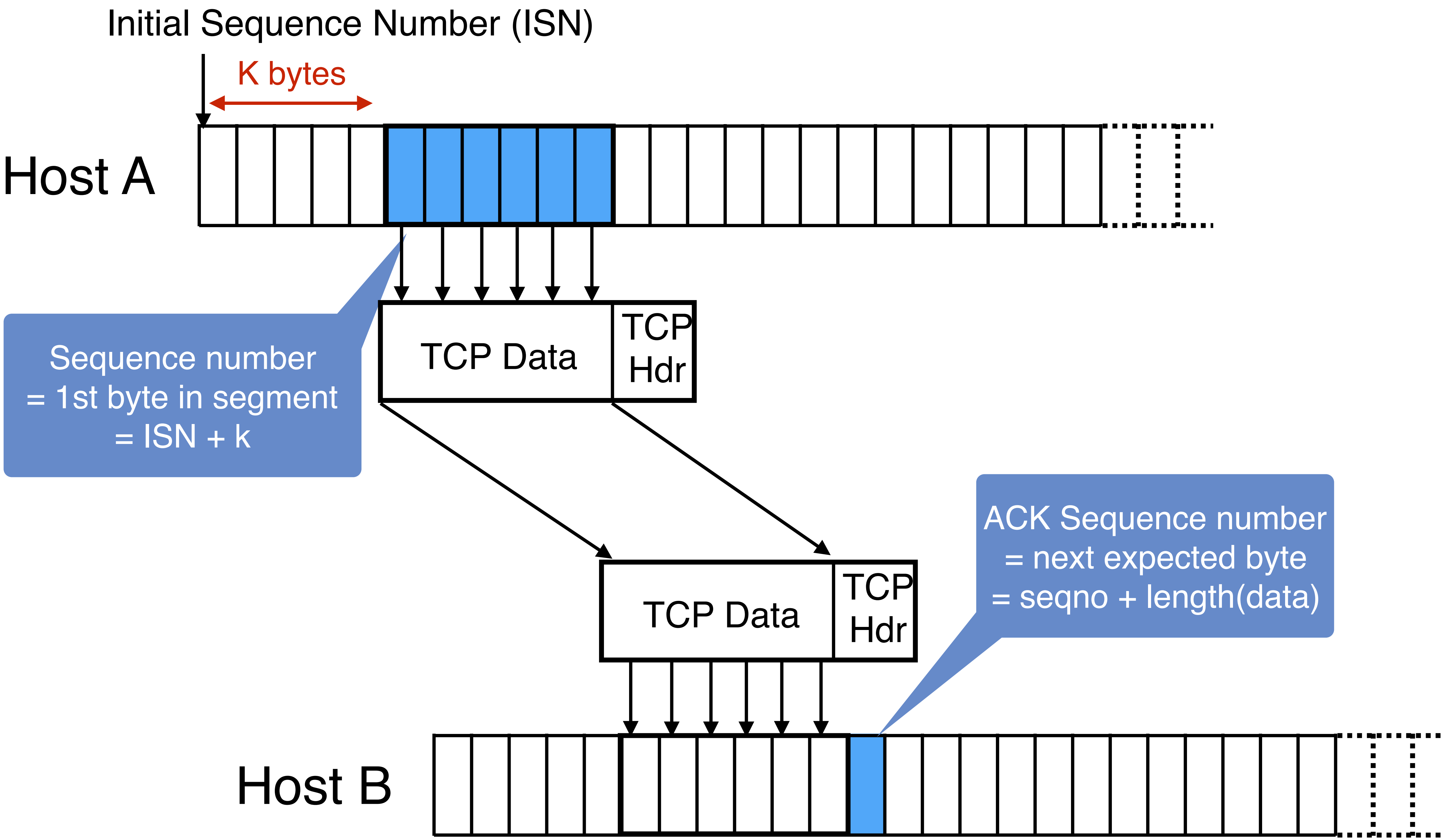


TCP Segment



- 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

Sequence Numbers



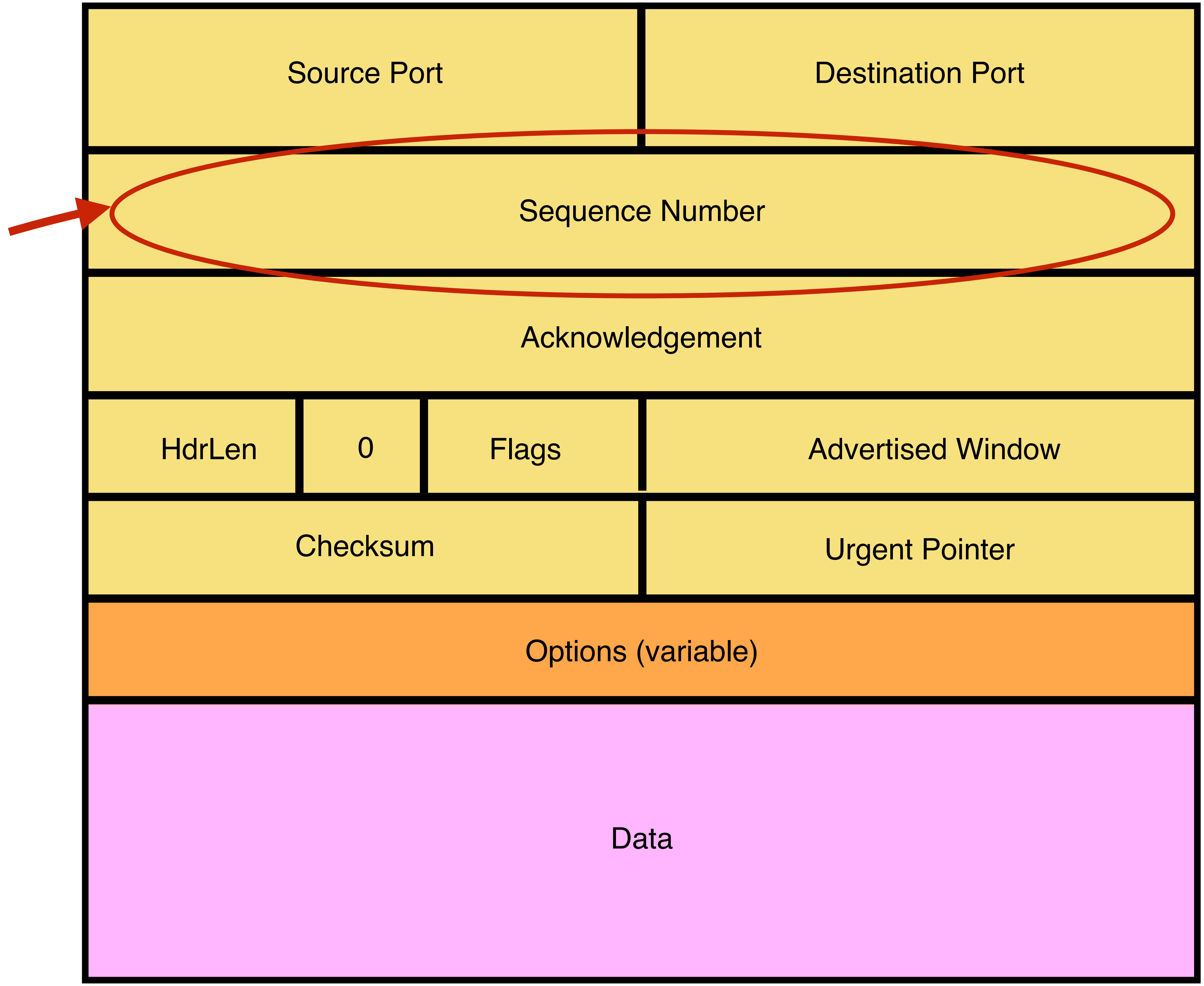
ACKing and Sequence Numbers

- Sender sends segments (byte stream)
    - Data starts with sequence number X
    - Packet contains B bytes
        - X, X+1, X+2, …, X+B-1

- Upon receipt of a segment, 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

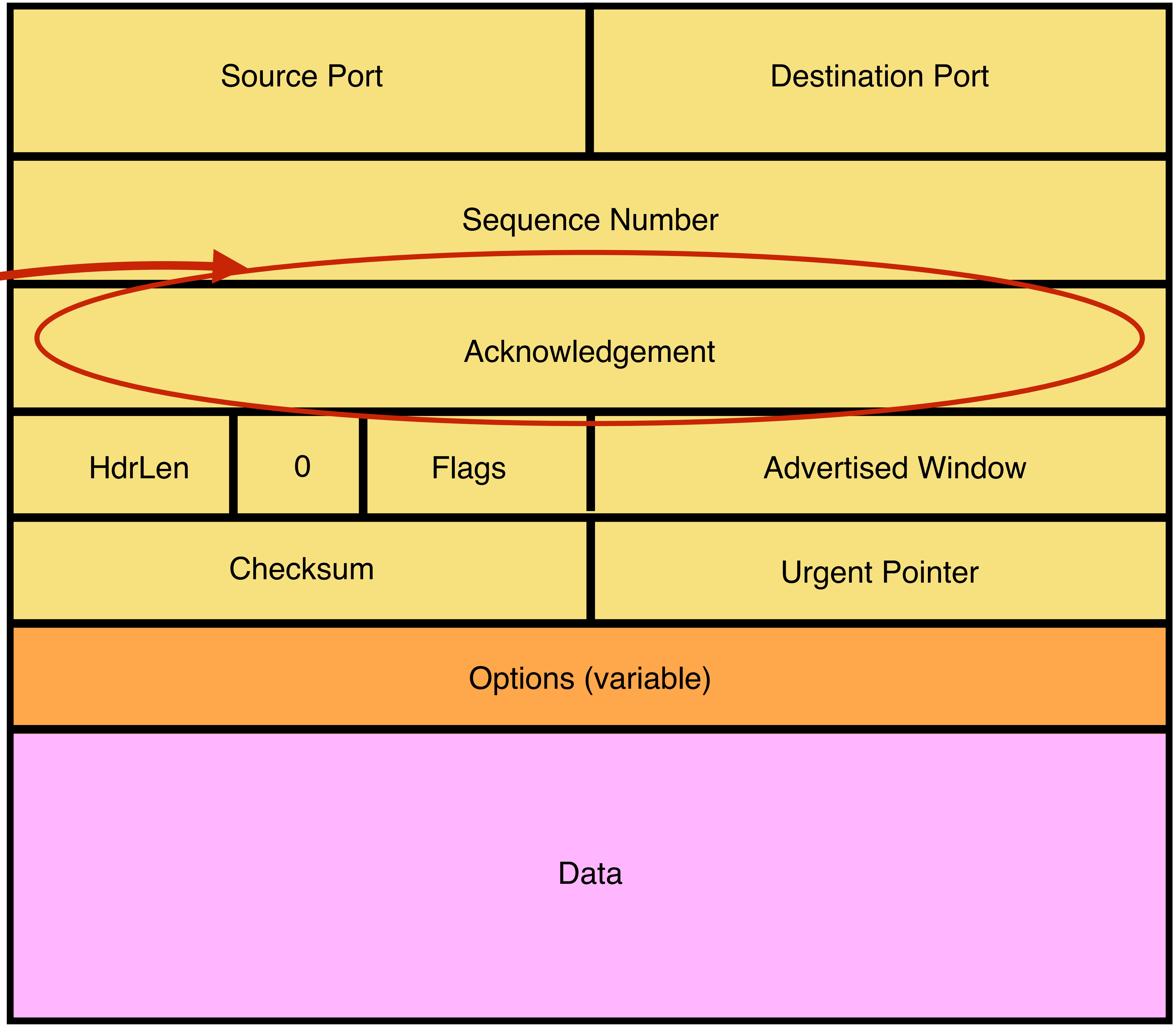
TCP Header

Starting byte offset of data carried in this segment

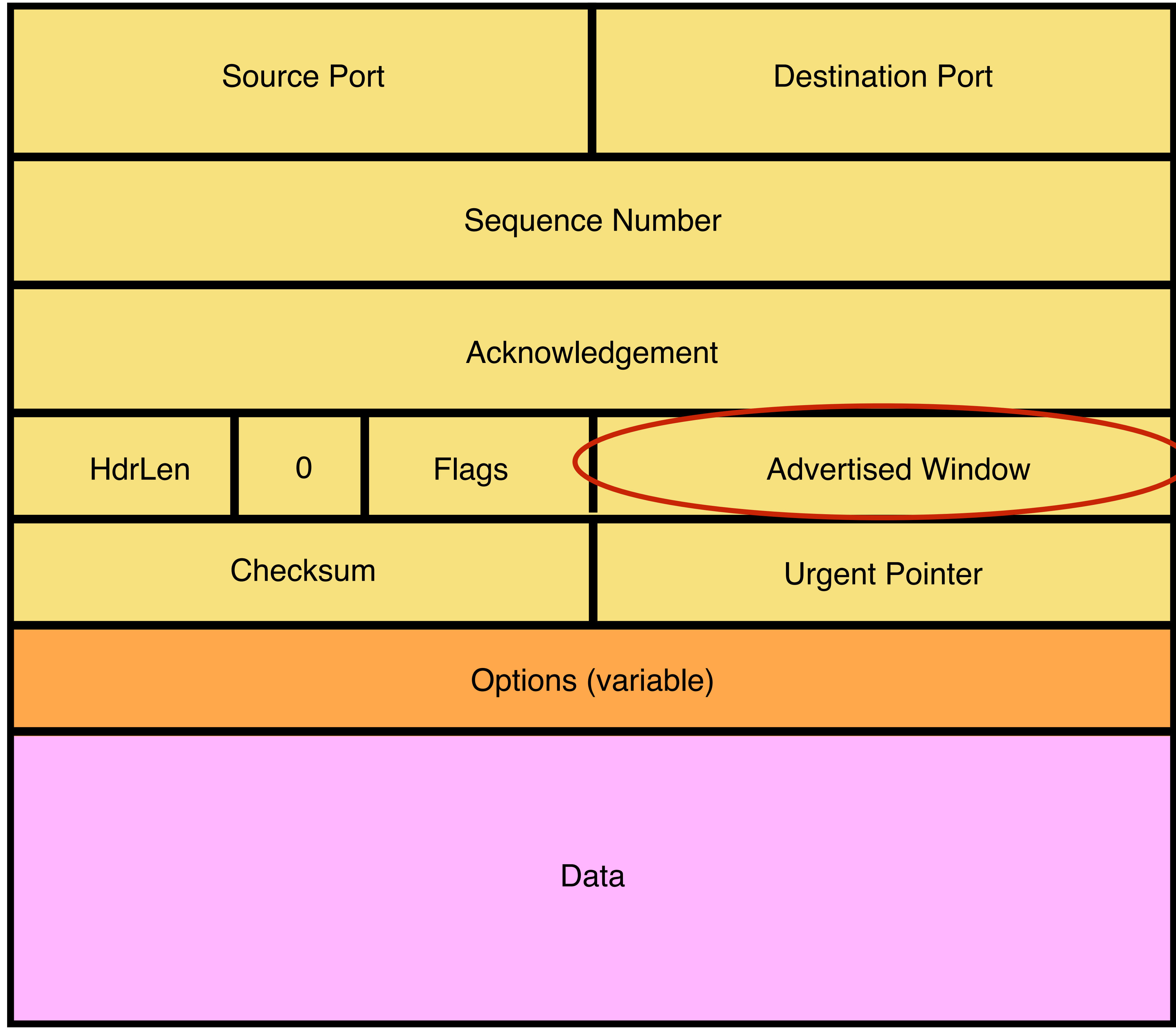


TCP Header

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

Flow Control (Sliding Window)

- 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

Filling the Pipe

- Simple example:
    - W (in bytes), which we assume is constant
    - RTT (in sec), which we assume is constant
    - B (in **bytes**/sec)
- How fast will data be transferred?
- If W/RTT < B, the transfer has speed W/RTT
- If W/RTT > B, the transfer has speed B

Advertised Window Limits Rate

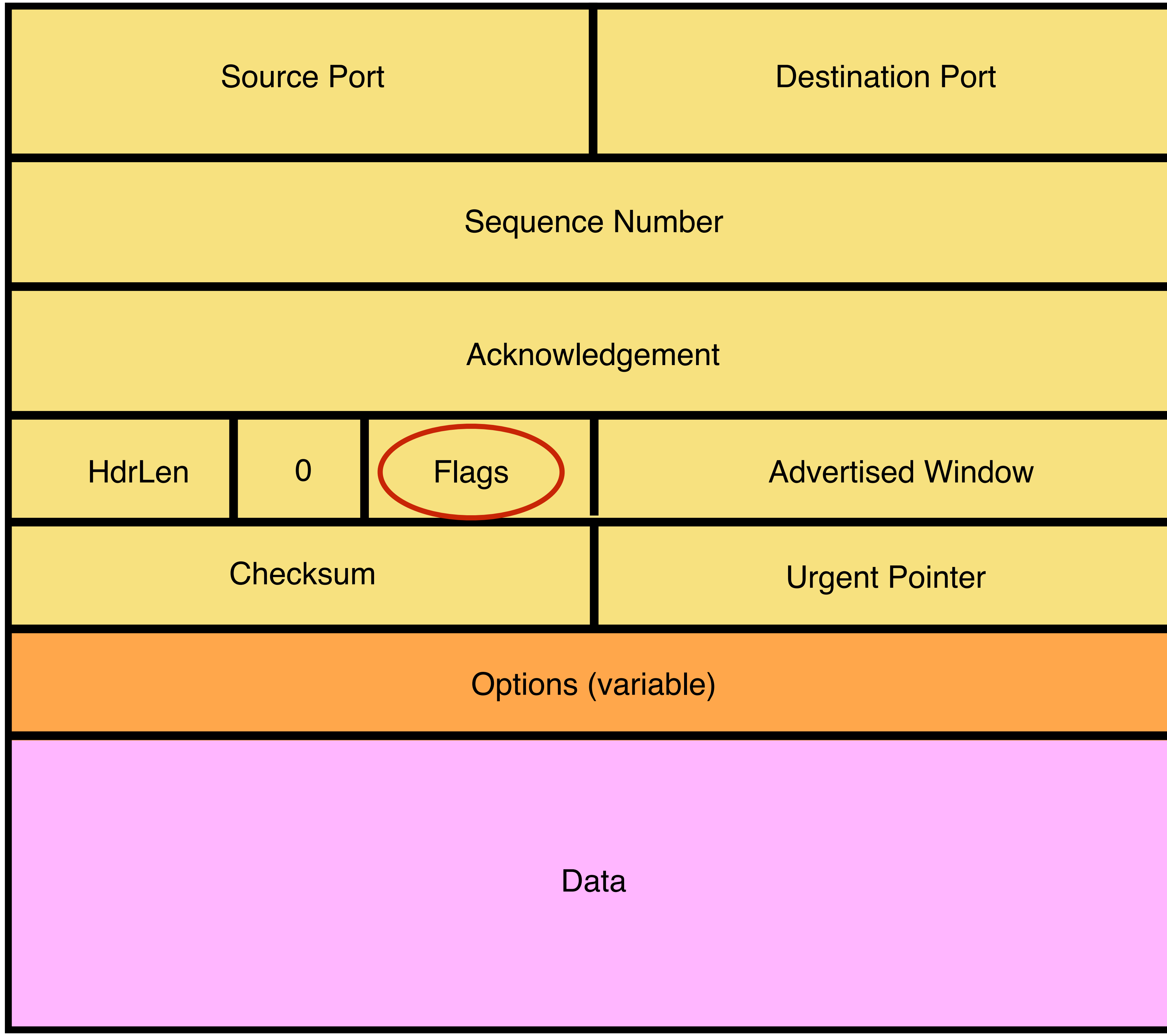
- Sender can send no faster than W/RTT bytes/sec
- In original TCP, that was the sole protocol mechanism controlling sender's rate
- What's missing?
- **Congestion control** about how to adjust W to avoid network congestion (next lecture)

Any Questions?

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?

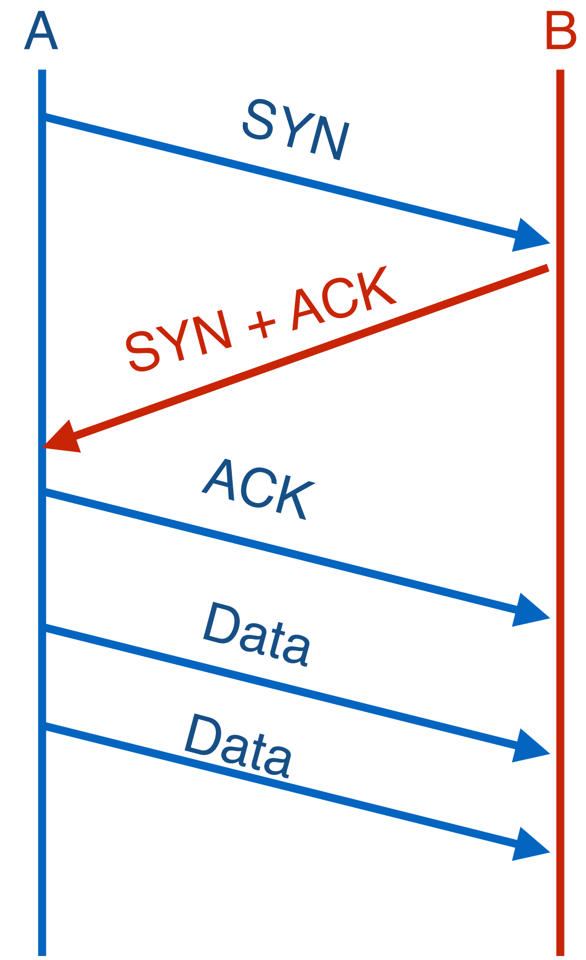


TCP Connection Establishment and Initial Sequence Numbers

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
  - Set from 32-bit clock that ticks every 4 microseconds
  - … only wraps around once every 4.55 hours
- To establish a connection, hosts exchange ISNs
  - How does this help?

Establishing a TCP Connection

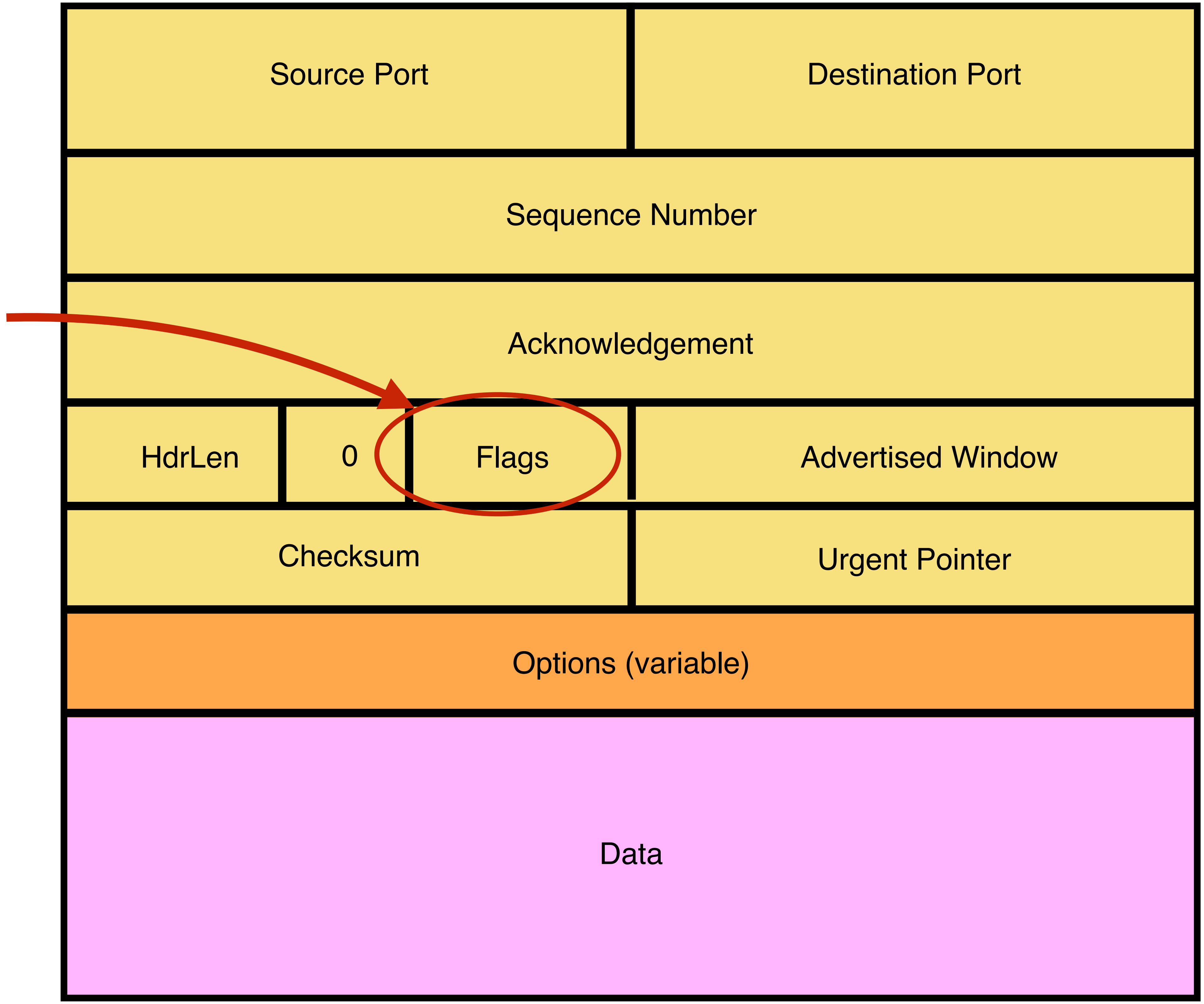


Each host tells its ISN to the other host.

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

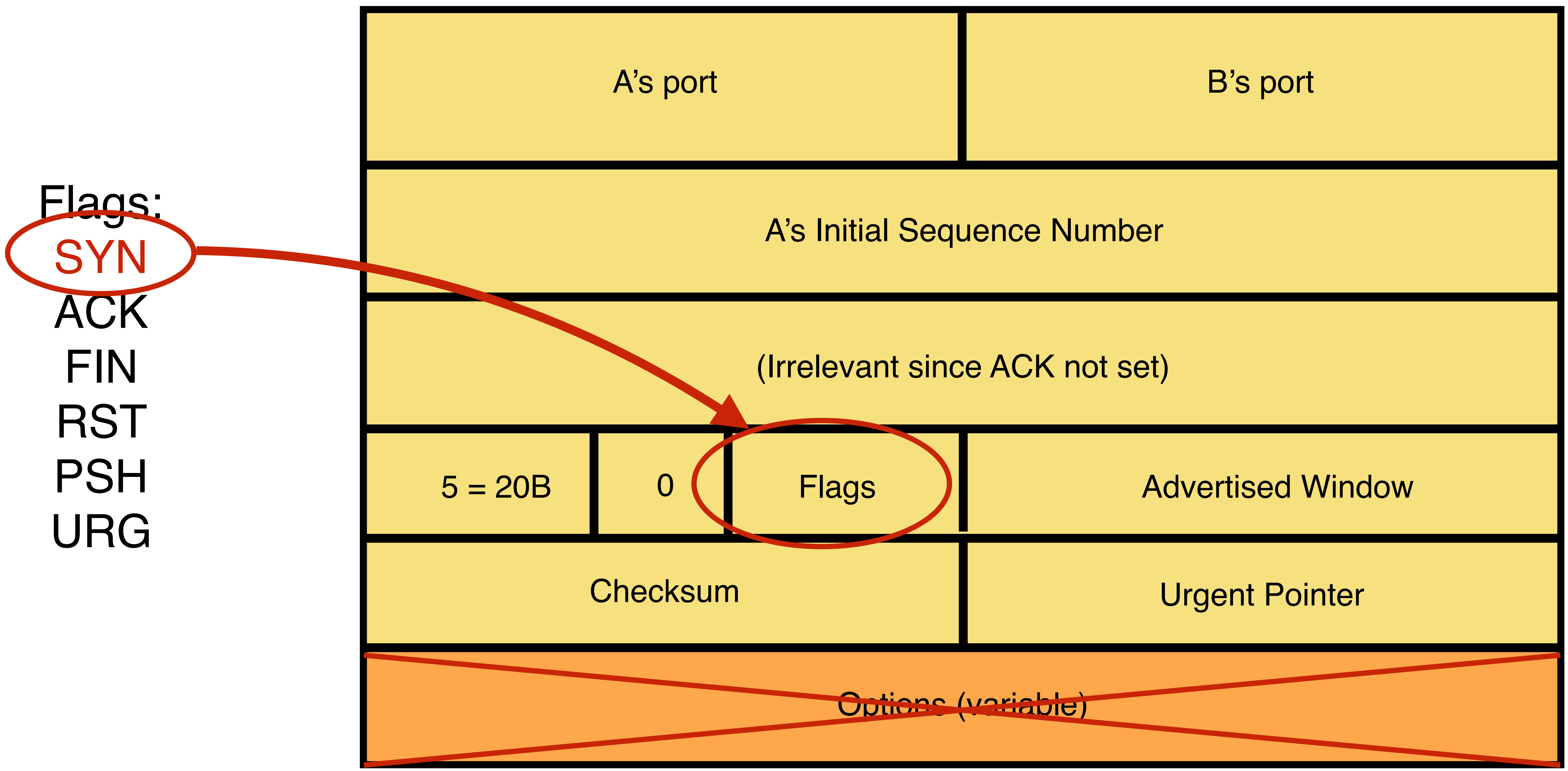
TCP Header

Flags:
SYN
ACK
FIN
RST
PSH
URG



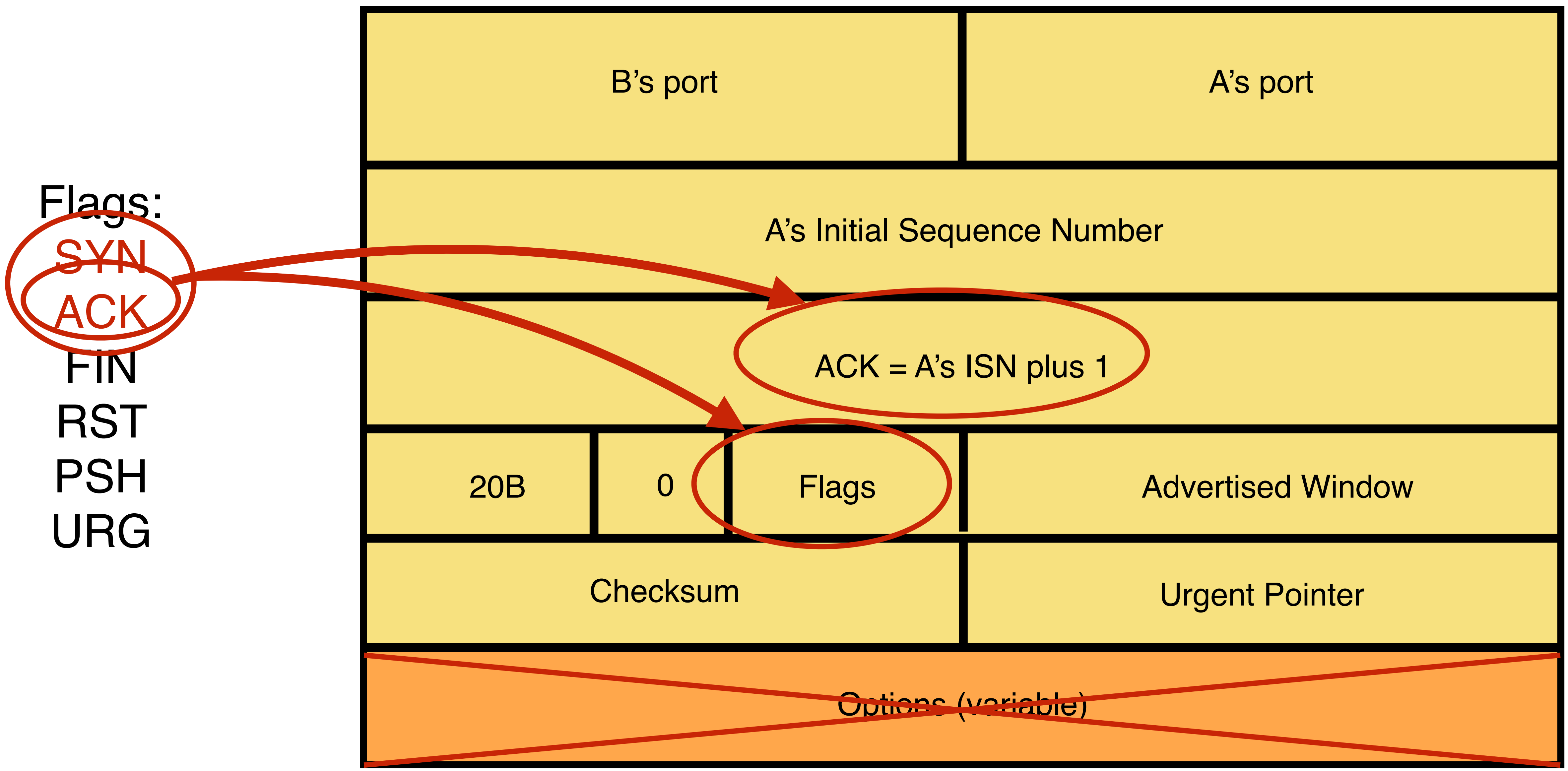
See /usr/include/netinet/tcp.h on Unix Systems

Step 1: A's Initial SYN Packet



A tells B it wants to open a connection…

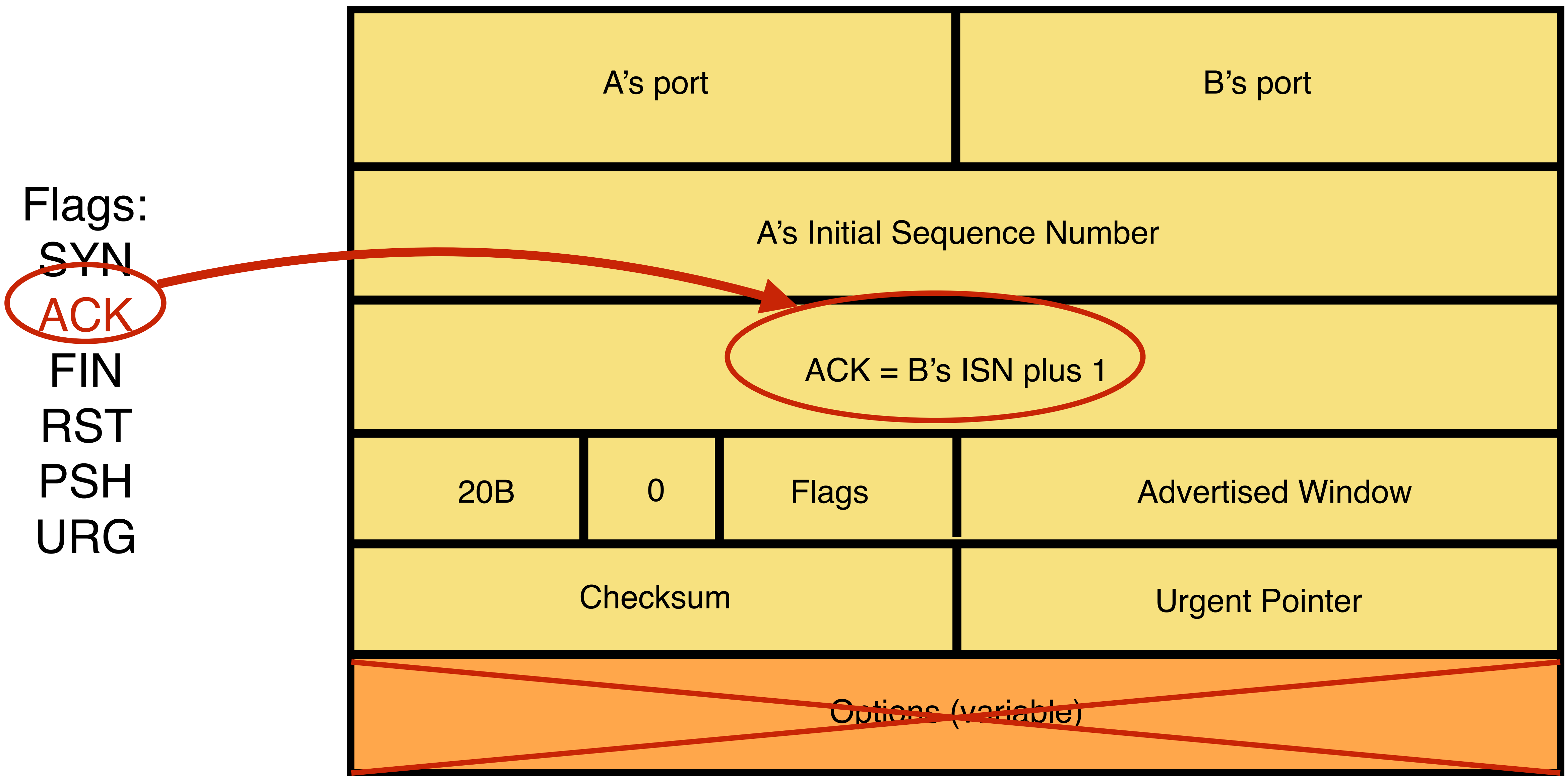
Step 2: B's SYN-ACK Packet



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

… upon receiving this packet, A can start sending data

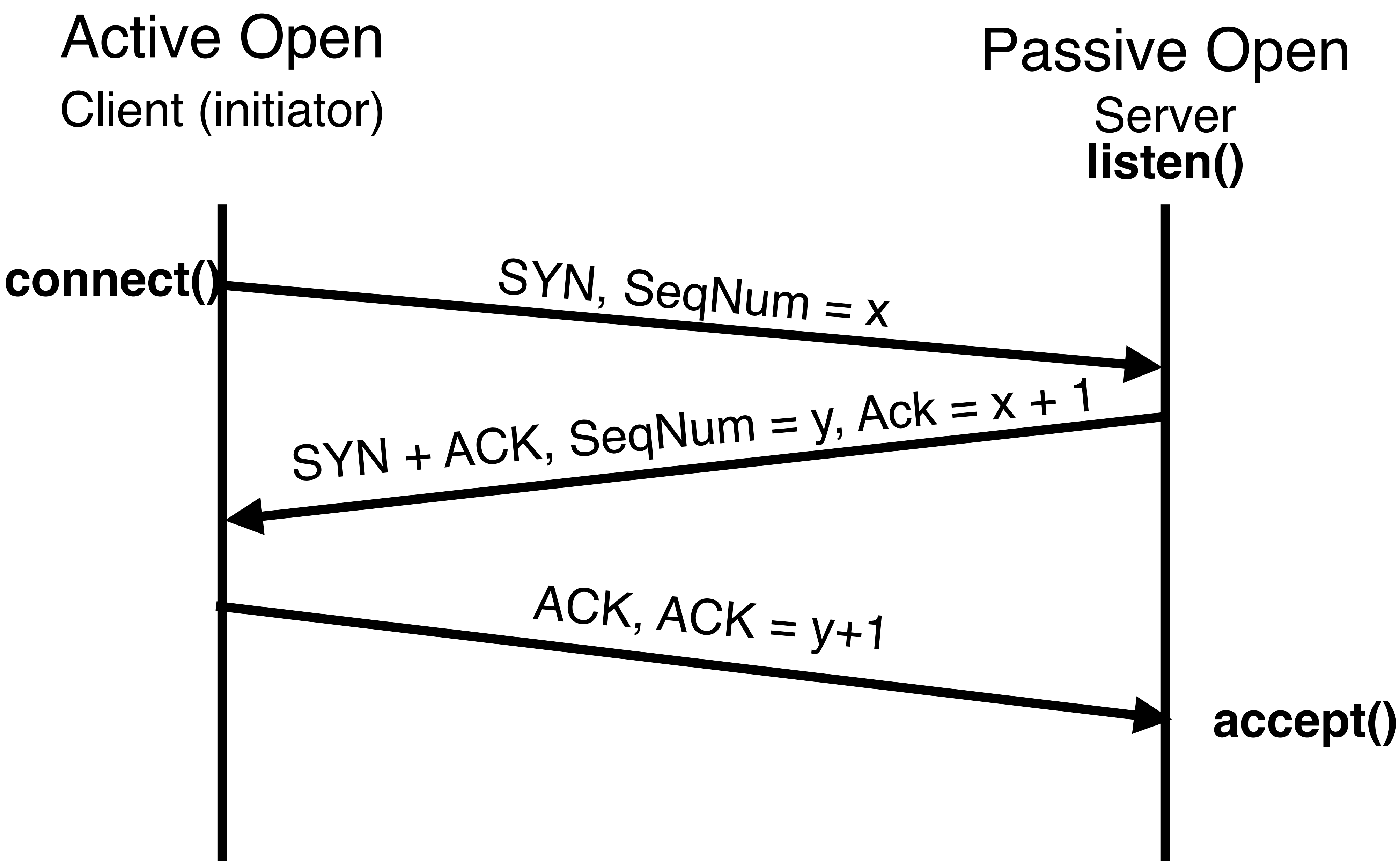
Step 3: A's ACK of the SYN-ACK



A tells B it's likewise okay to start sending

… upon receiving this packet, B can start sending data

Timing Diagram: 3-Way Handshaking



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 …)

Any Questions?

Done for today

Next lecture: Congestion control

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

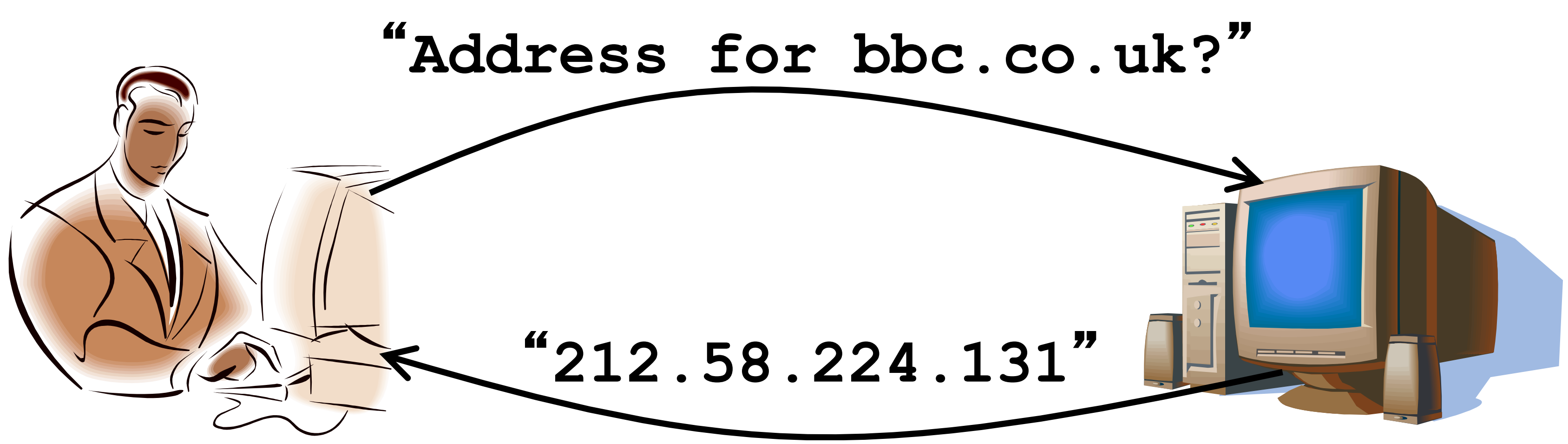
Source Port #	Dest Port #
Checksum	Length
Application Data (Message)	

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)

What if the SYN Packet Gets Lost?

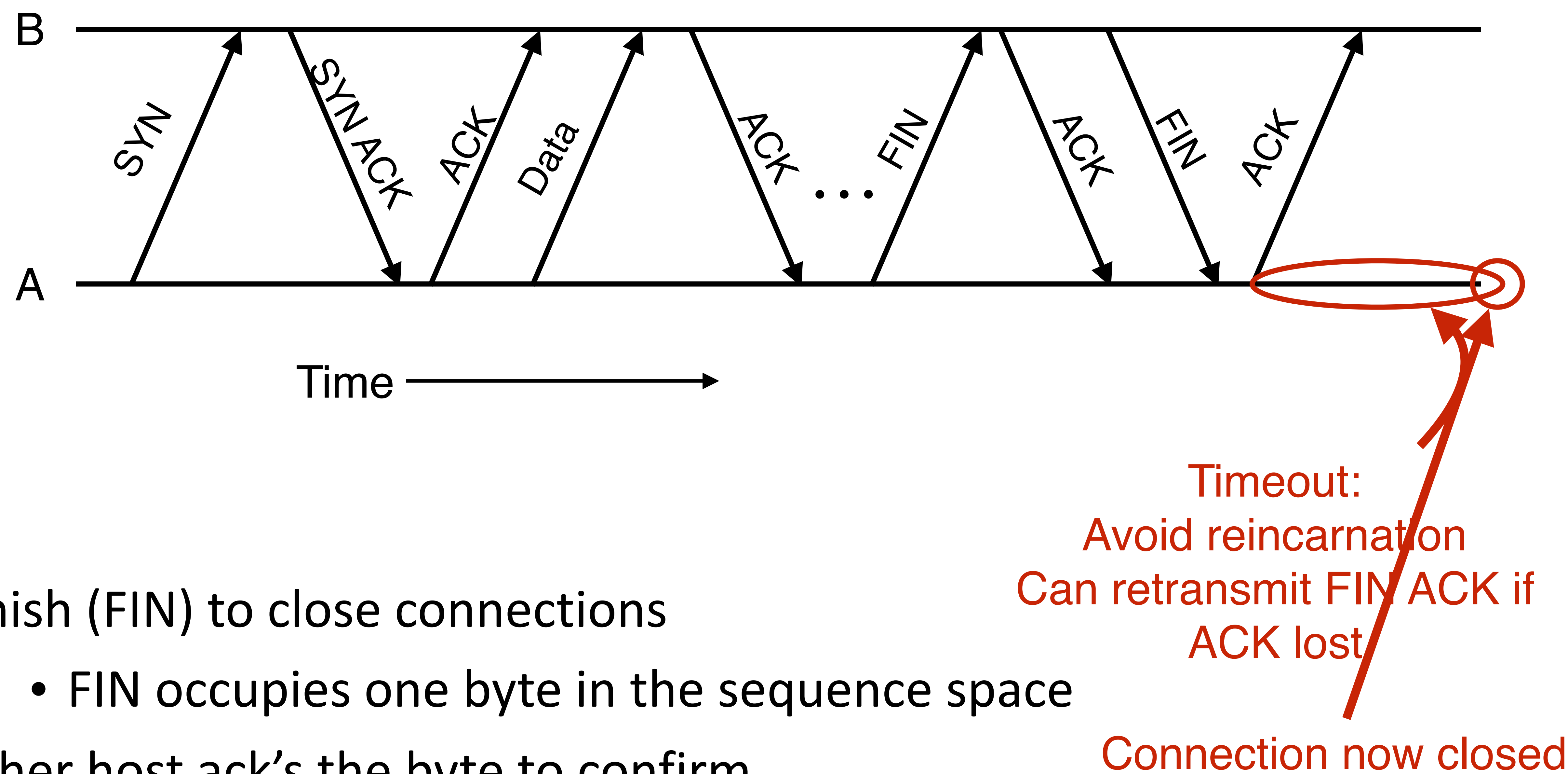
- Suppose the SYN packet gets lost
    - Packet is lost inside the network, or
    - Server **discards** the packet (e.g., listen queue is full)
- Eventually, no SYN-ACK arrives
    - Sender sets a **timer** and **waits** for the SYN-ACK
    - … and retransmits the SYN if needed
- How should the TCP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

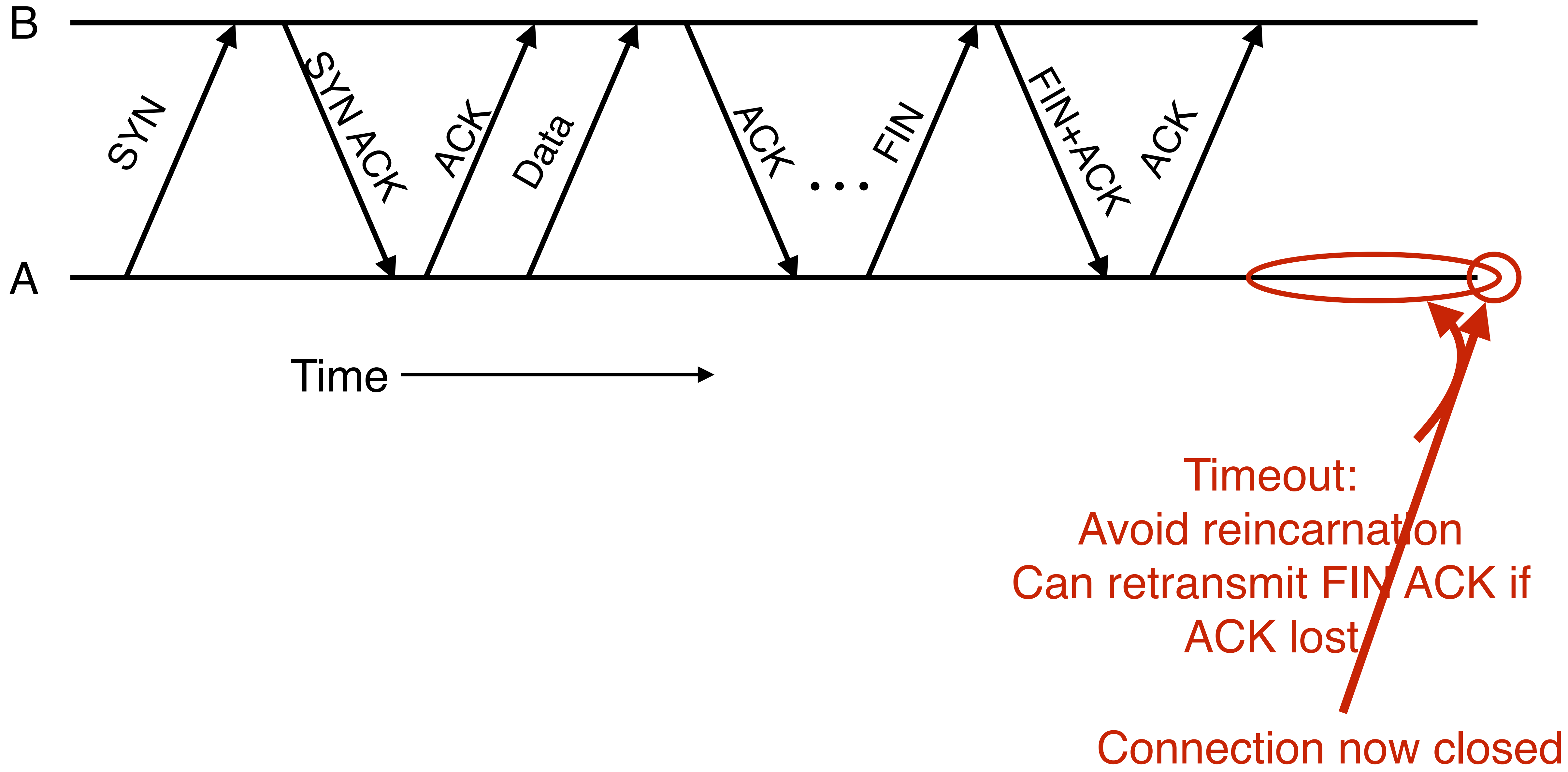
Normal Termination



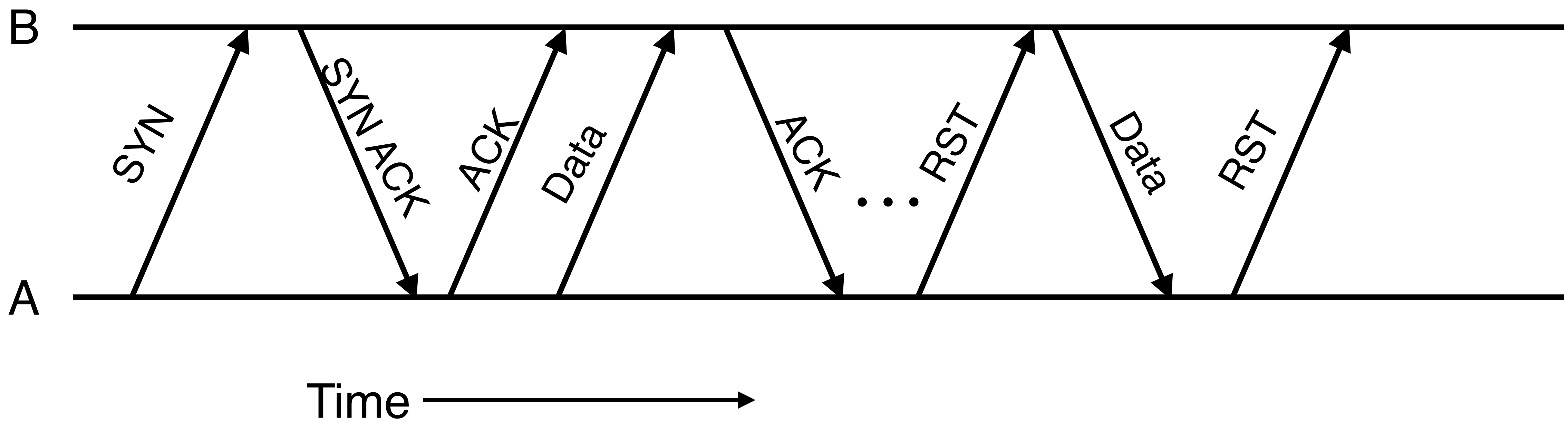
- Finish (FIN) to close connections
  - FIN occupies one byte in the sequence space
- Other host ack's the byte to confirm
- Closes A's side of connection, but not B's
  - Until B likewise sends a FIN
  - Which A then acks

Normal Termination, Both Together

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

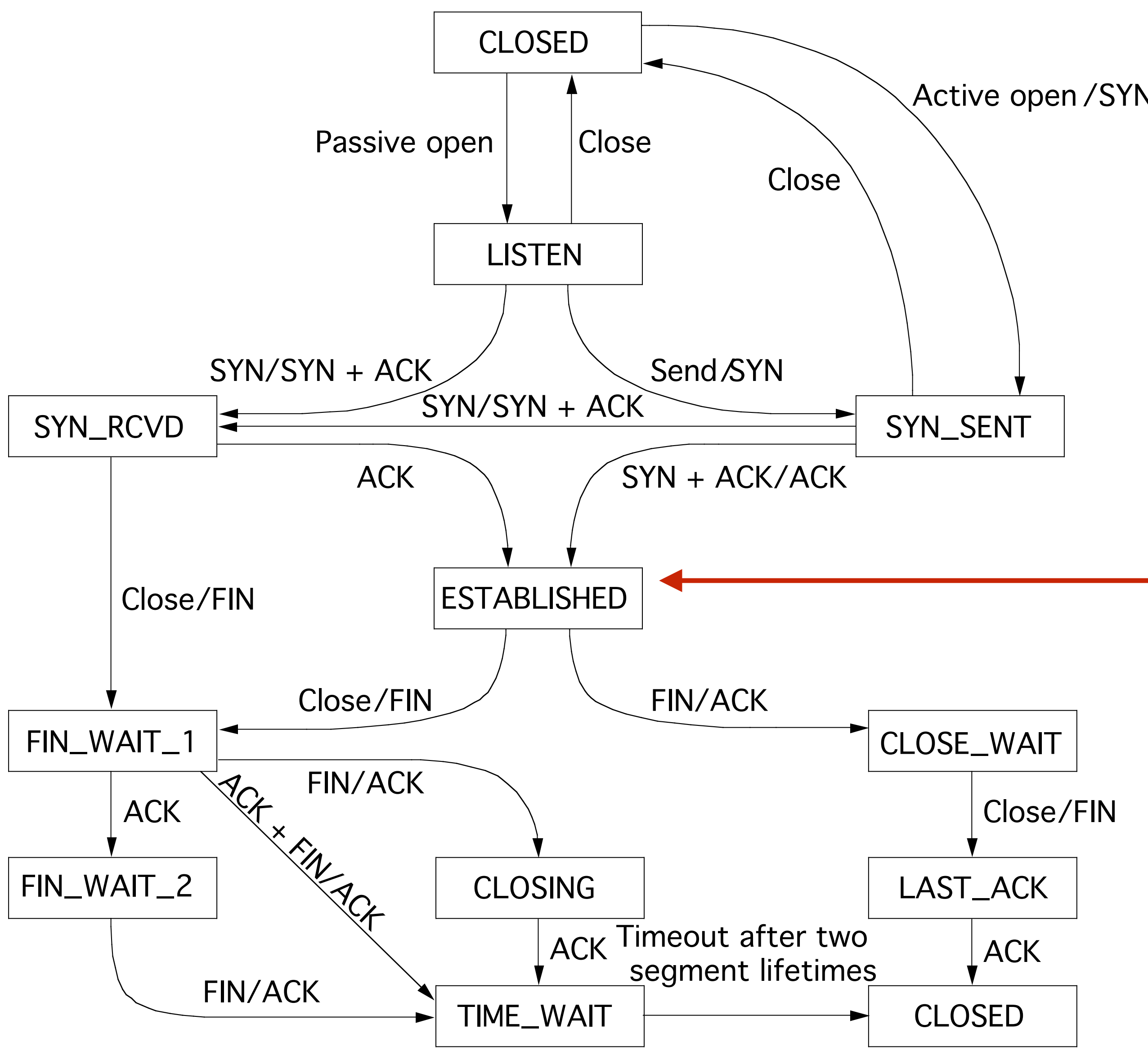


Abrupt Termination



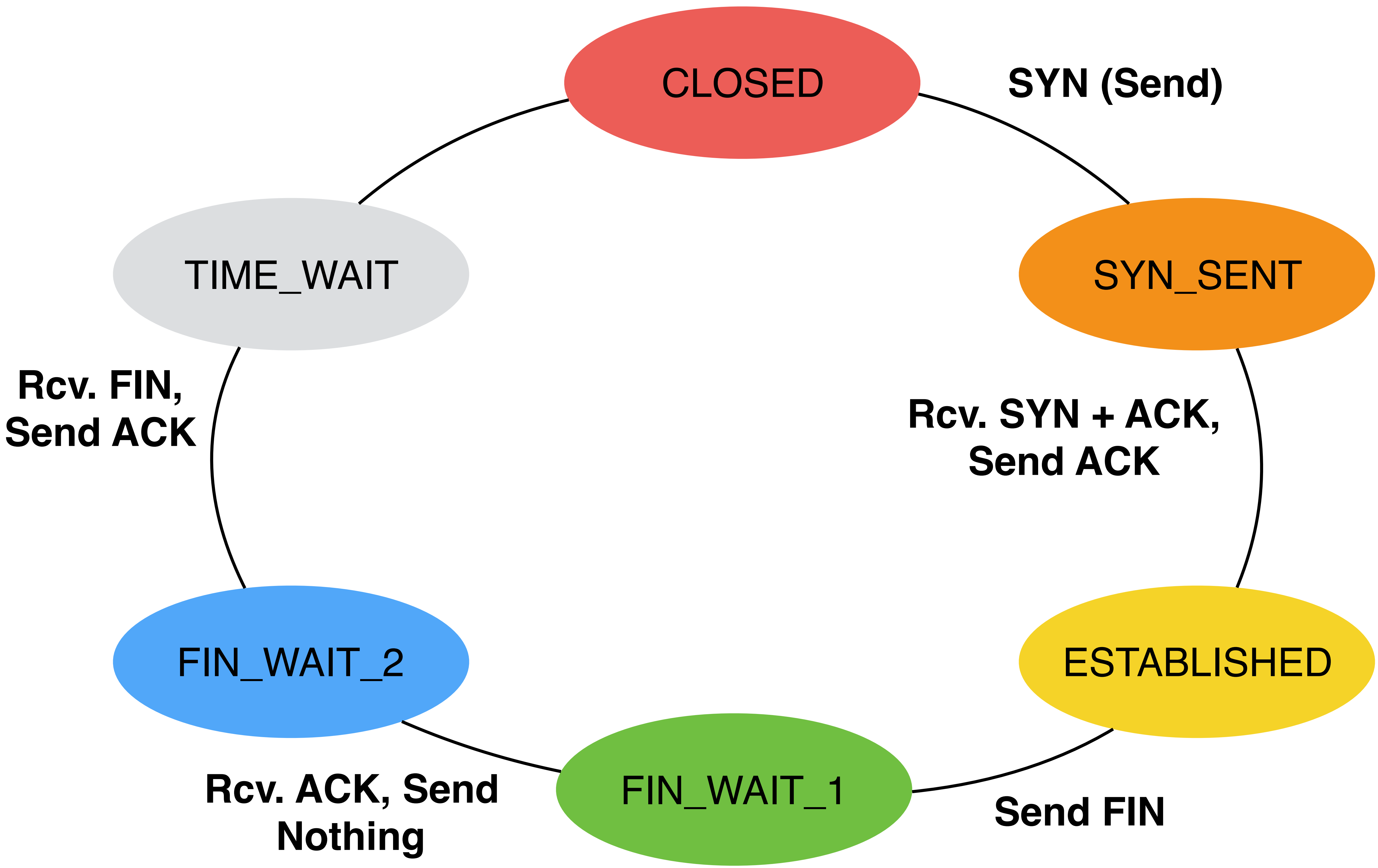
- A sends a RESET (RST) to B
  - E.g., because app. Process on A crashed
- That's it
  - B does not ack the RST
  - This, RST is not delivered reliably
  - And, any data in flight is lost
  - But, if B sends anything more, will elicit another RST

TCP State Transitions

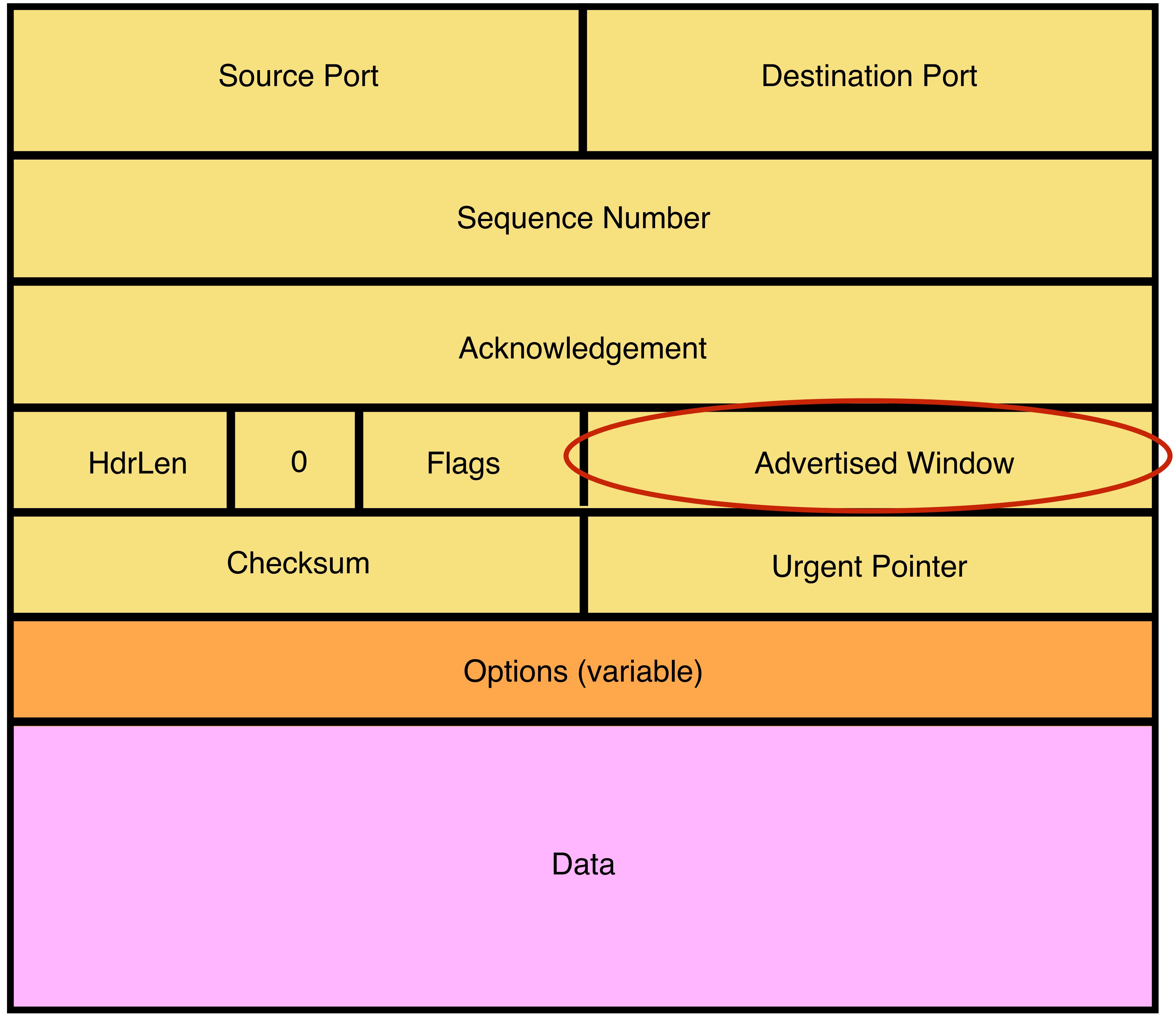


Data, ACK exchanges are in here

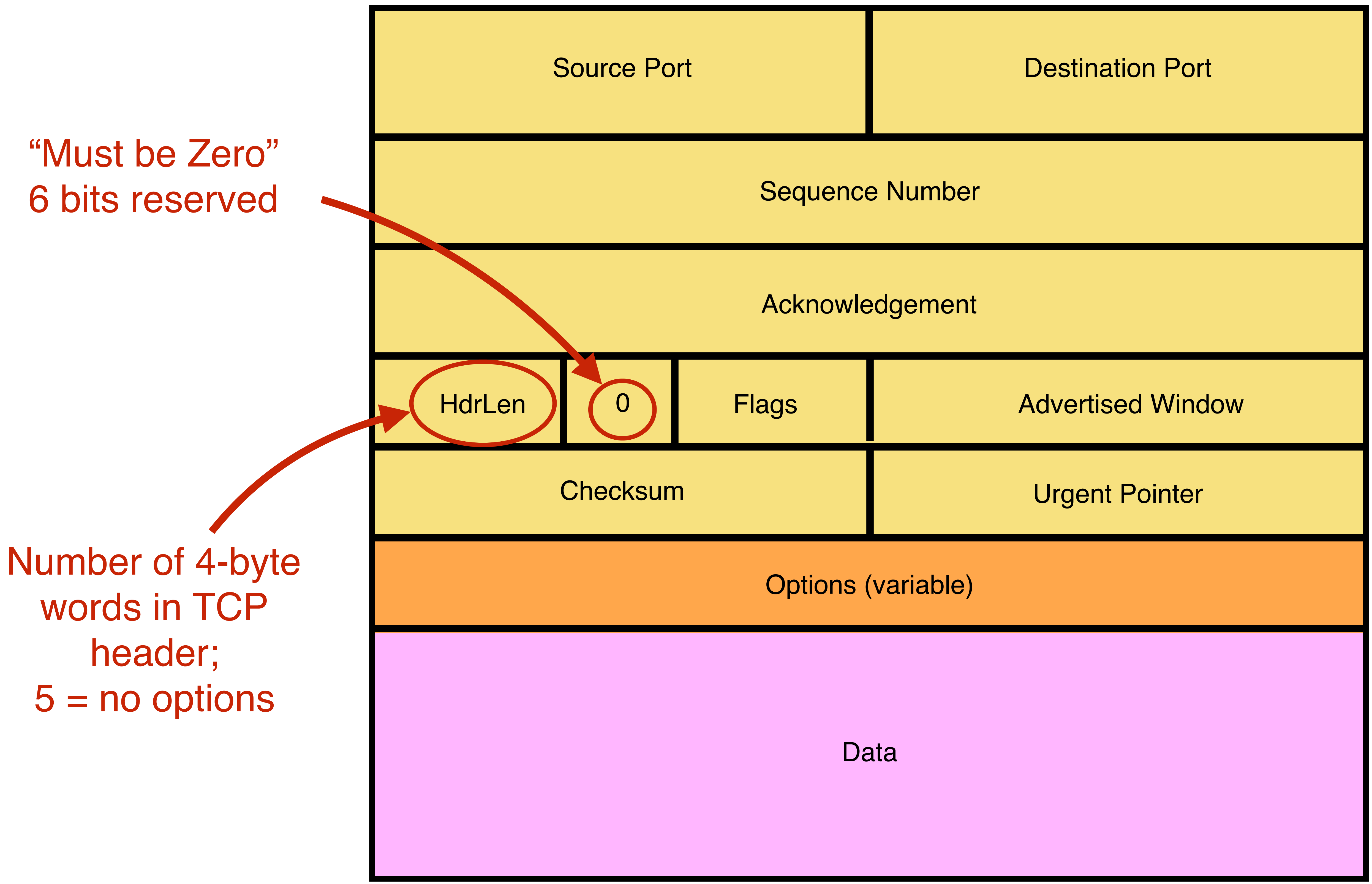
A Simpler View of the Client Side



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?



TCP Header: What's left?

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