CSC 458/2209 – Computer Networking Systems

Handout # 12: Transport Protocols

Professor Yashar Ganjali
Department of Computer Science
University of Toronto

ganjali7@cs.toronto.edu
http://www.cs.toronto.edu/~yganjali

Announcements

- Programming Assignment 1
  - Due Friday February 14th at 5pm.
  - Submission instructions on course web page.

- Problem Set 1
  - Solutions will be posted on Friday

- This week's tutorial:
  - Programming Assignment 1 Q&A

- Reading for this week:
  - Chapter 5 of the textbook

Announcements – Cont'd

- Midterm exam
  - L0101: Monday February 24th
  - L0201: Tuesday February 25th
  - In class: same room and time as the lecture
  - For undergraduate and graduate students

- Sample midterm and solutions on class web page.

- Everything covered up to the end of today's lecture
  - Emphasis on the slides, problem set, and sample midterm provided.
  - Textbook: up to Chapter 5

Role of Transport Layer

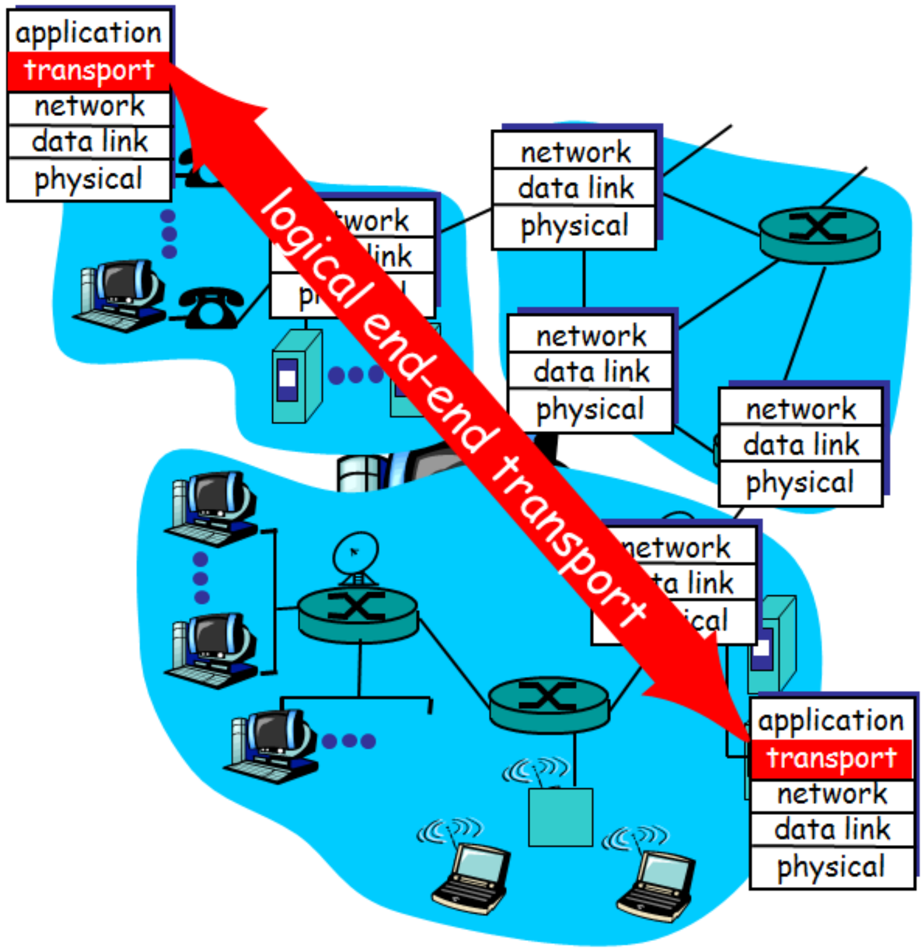
- Link layer
  - Transfer bit frames between neighboring nodes
  - E.g., Ethernet
- Network layer
  - Logical communication between nodes
  - Hides details of the link technology
  - E.g., IP
- Transport layer
  - Communication between processes (e.g., socket)
  - Relies on network layer and serves the application layer
  - E.g., TCP and UDP
- Application layer
  - Communication for specific applications
  - E.g., HyperText Transfer Protocol (HTTP), File Transfer Protocol (FTP), Network News Transfer Protocol (NNTP)

Today's Lecture

- Principles underlying transport-layer services
  - (De)multiplexing
  - Detecting corruption
  - Reliable delivery
  - Flow control
- Transport-layer protocols in the Internet
  - User Datagram Protocol (UDP)
  - Transmission Control Protocol (TCP)

Transport Protocols

- Provide logical communication between application processes running on different hosts
- Run on end hosts
  - Sender: breaks application messages into segments, and passes to network layer
  - Receiver: reassembles segments into messages, passes to application layer
- Multiple transport protocol available to applications
  - Internet: TCP and UDP

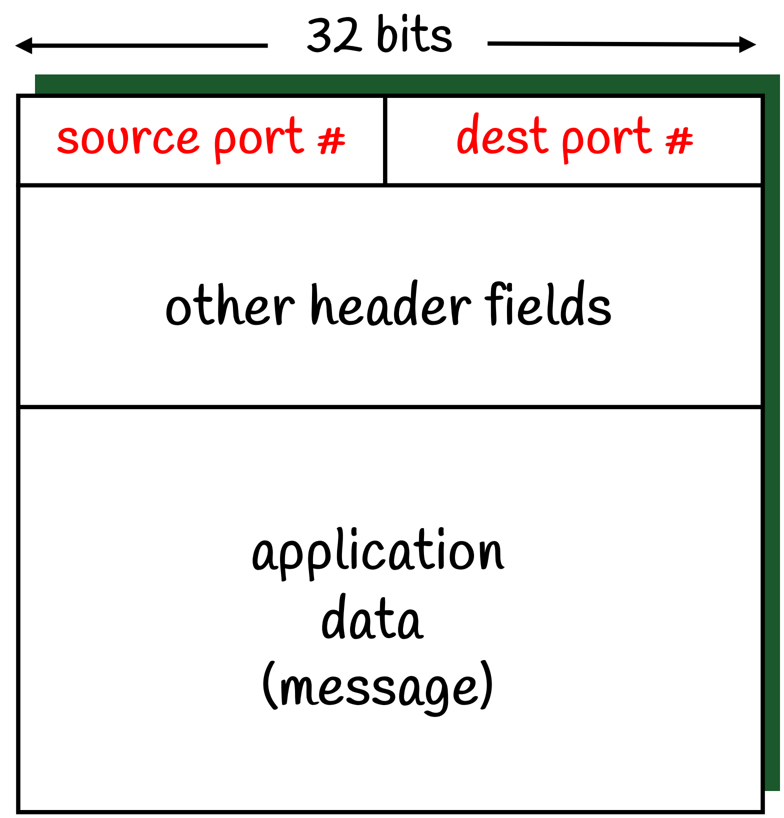


Internet Transport Protocols

- Datagram messaging service (UDP)
  - No-frills extension of "best-effort" IP
- Reliable, in-order delivery (TCP)
  - Connection set-up
  - Discarding of corrupted packets
  - Retransmission of lost packets
  - Flow control → Do not overload the receiver
  - Congestion control (next lecture) → Do not overload the network
- Other services not available
  - Delay guarantees
  - Bandwidth guarantees

Multiplexing and Demultiplexing

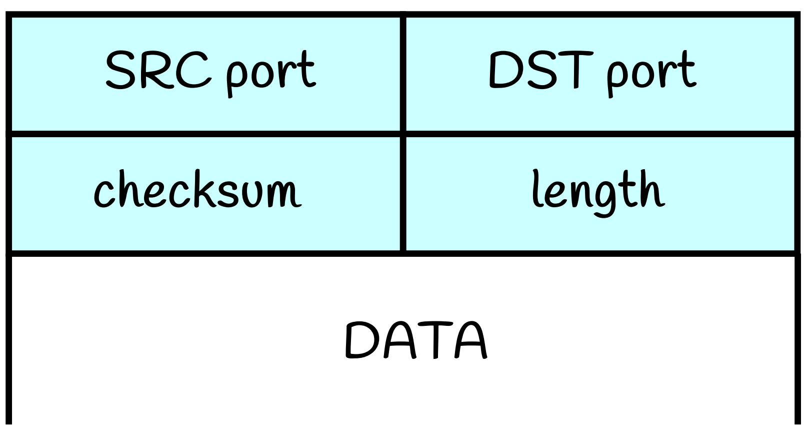
- Host receives IP datagrams
  - Each datagram has source and destination IP address,
  - Each datagram carries one transport-layer segment
  - Each segment has source and destination port number
- Host uses IP addresses and port numbers to direct the segment to appropriate socket



TCP/UDP segment format

Unreliable Message Delivery Service

- Lightweight communication between processes
  - Avoid overhead and delays of ordered, reliable delivery
  - Send messages to and receive them from a socket
- User Datagram Protocol (UDP)
  - IP plus port numbers to support (de)multiplexing
  - Optional error checking on the packet contents



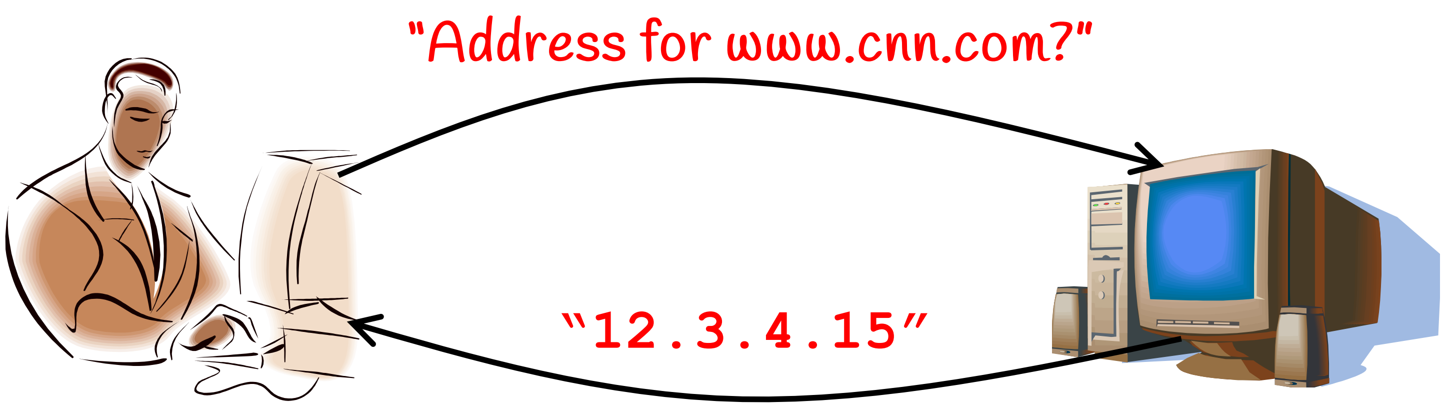
Why Would Anyone Use UDP?

- Finer control over what data is sent and when
  - As soon as an application process writes into the socket
  - … UDP will package the data and send the packet
- No delay for connection establishment
  - UDP just blasts away without any formal preliminaries
  - … which avoids introducing any unnecessary delays
- No connection state
  - No allocation of buffers, parameters, sequence #s, etc.
  - … making it easier to handle many active clients at once
- Small packet header overhead
  - UDP header is only eight-bytes long

Popular Applications That Use UDP

- Multimedia streaming
  - Retransmitting lost/corrupted packets is not worthwhile
  - By the time the packet is retransmitted, it's too late
  - E.g., telephone calls, video conferencing, gaming

- Simple query protocols like Domain Name System
  - Overhead of connection establishment is overkill
  - Easier to have application retransmit if needed



Transmission Control Protocol (TCP)

- Connection oriented
  - Explicit set-up and tear-down of TCP session
- Stream-of-bytes service
  - Sends and receives a stream of bytes, not messages
- Reliable, in-order delivery
  - Checksums to detect corrupted data
  - Acknowledgments & retransmissions for reliable delivery
  - Sequence numbers to detect losses and reorder data
- Flow control
  - Prevent overflow of the receiver's buffer space
- Congestion control
  - Adapt to network congestion for the greater good

An Analogy: Talking on a Cell Phone

- Alice and Bob on their cell phones
  - Both Alice and Bob are talking
- What if Bob couldn't understand Alice?
  - Bob asks Alice to repeat what she said
- What if Bob hasn't heard Alice for a while?
  - Is Alice just being quiet?
  - Or, have Bob and Alice lost reception?
  - How long should Bob just keep on talking?
  - Maybe Alice should periodically say "uh huh"
  - … or Bob should ask "Can you hear me now?" ☺

Some Take-Aways from the Example

- Acknowledgments from receiver
  - Positive: “okay” or “ACK”
  - Negative: “please repeat that” or “NACK”
- Timeout by the sender (“stop and wait”)
  - Don’t wait indefinitely without receiving some response
  - … whether a positive or a negative acknowledgment
- Retransmission by the sender
  - After receiving a “NACK” from the receiver
  - After receiving no feedback from the receiver

Challenges of Reliable Data Transfer

- Over a perfectly reliable channel
  - All of the data arrives in order, just as it was sent
  - Simple: sender sends data, and receiver receives data
- Over a channel with bit errors
  - All of the data arrives in order, but some bits corrupted
  - Receiver detects errors and says “please repeat that”
  - Sender retransmits the data that were corrupted
- Over a lossy channel with bit errors
  - Some data are missing, and some bits are corrupted
  - Receiver detects errors but cannot always detect loss
  - Sender must wait for acknowledgment (“ACK” or “OK”)
  - … and retransmit data after some time if no ACK arrives

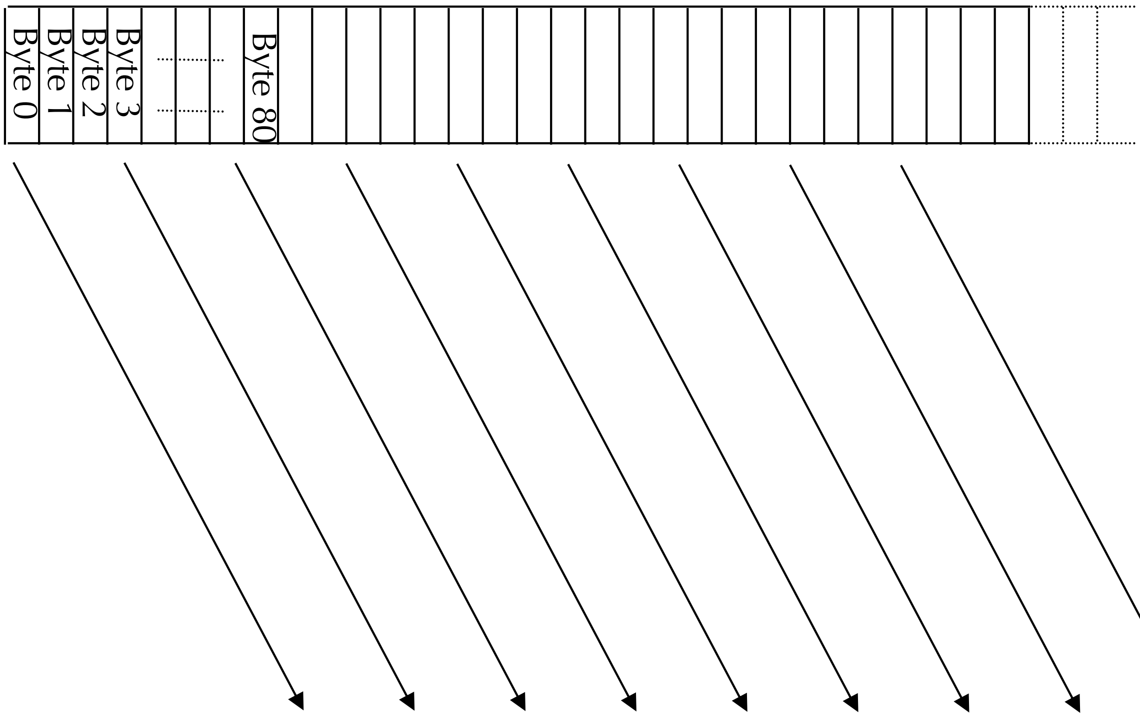
TCP Support for Reliable Delivery

- Checksum
  - Used to detect corrupted data at the receiver
  - …leading the receiver to drop the packet
- Sequence numbers
  - Used to detect missing data
  - ... and for putting the data back in order
- Retransmission
  - Sender retransmits lost or corrupted data
  - Timeout based on estimates of round-trip time
  - Fast retransmit algorithm for rapid retransmission

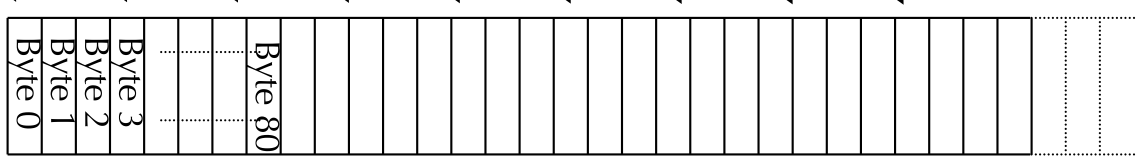
TCP Segments

TCP "Stream of Bytes" Service

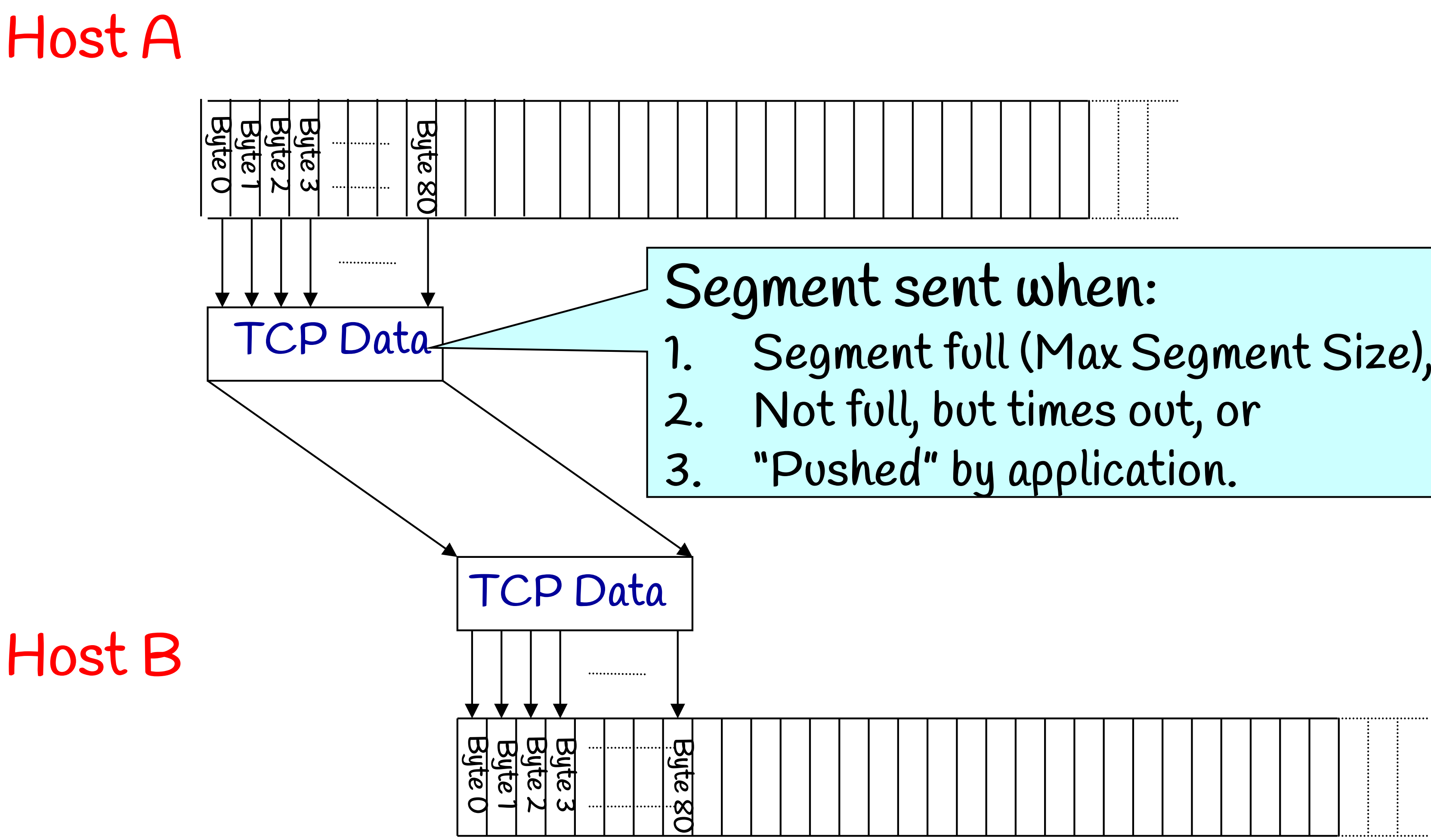
Host A



Host B



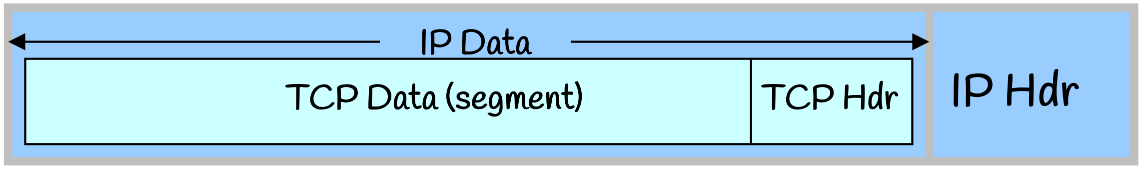
…Emulated Using TCP "Segments"



Segment sent when:

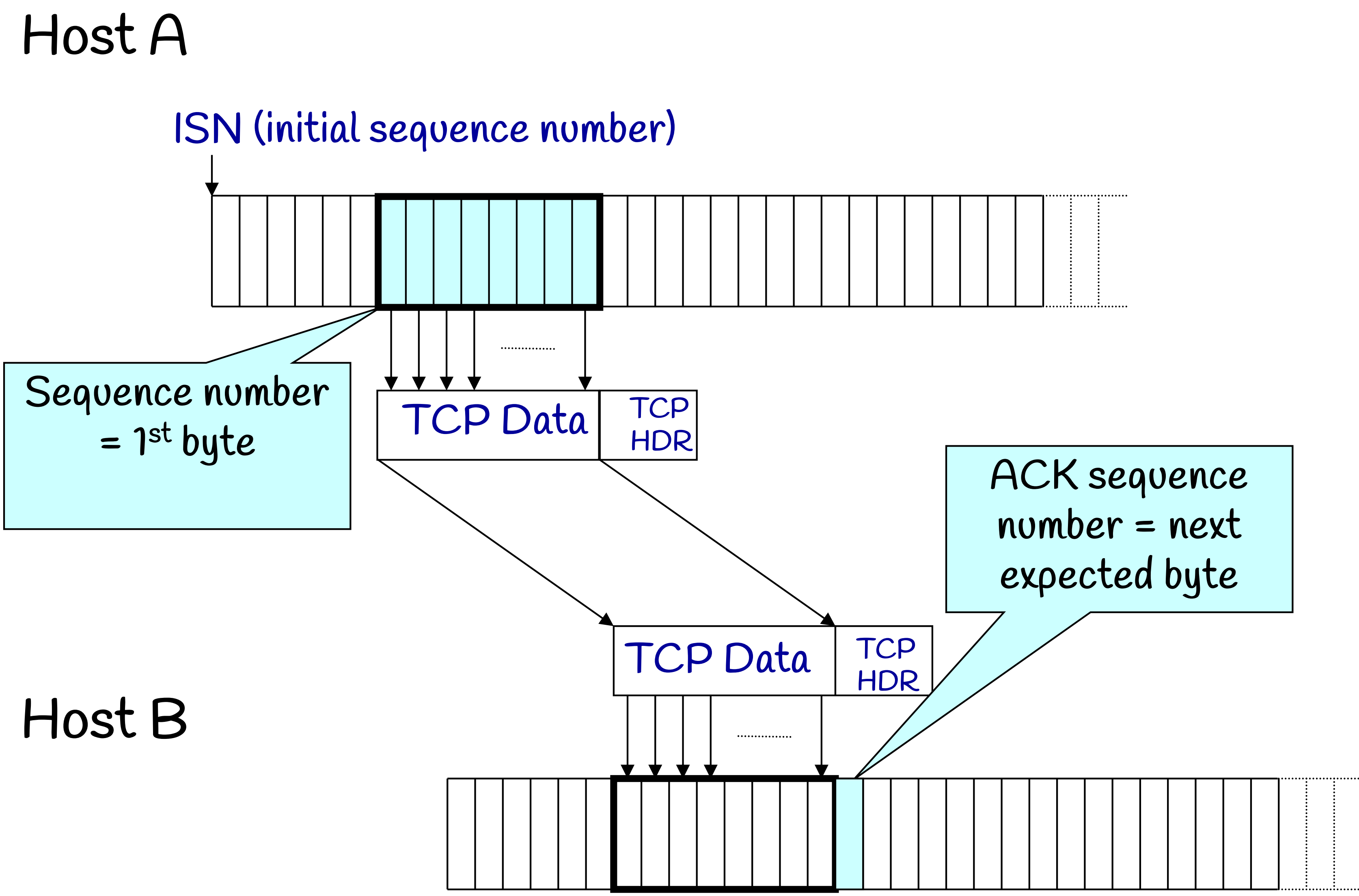
1. Segment full (Max Segment Size),
2. Not full, but times out, or
3. "Pushed" by application.

TCP Segment



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

Sequence Numbers

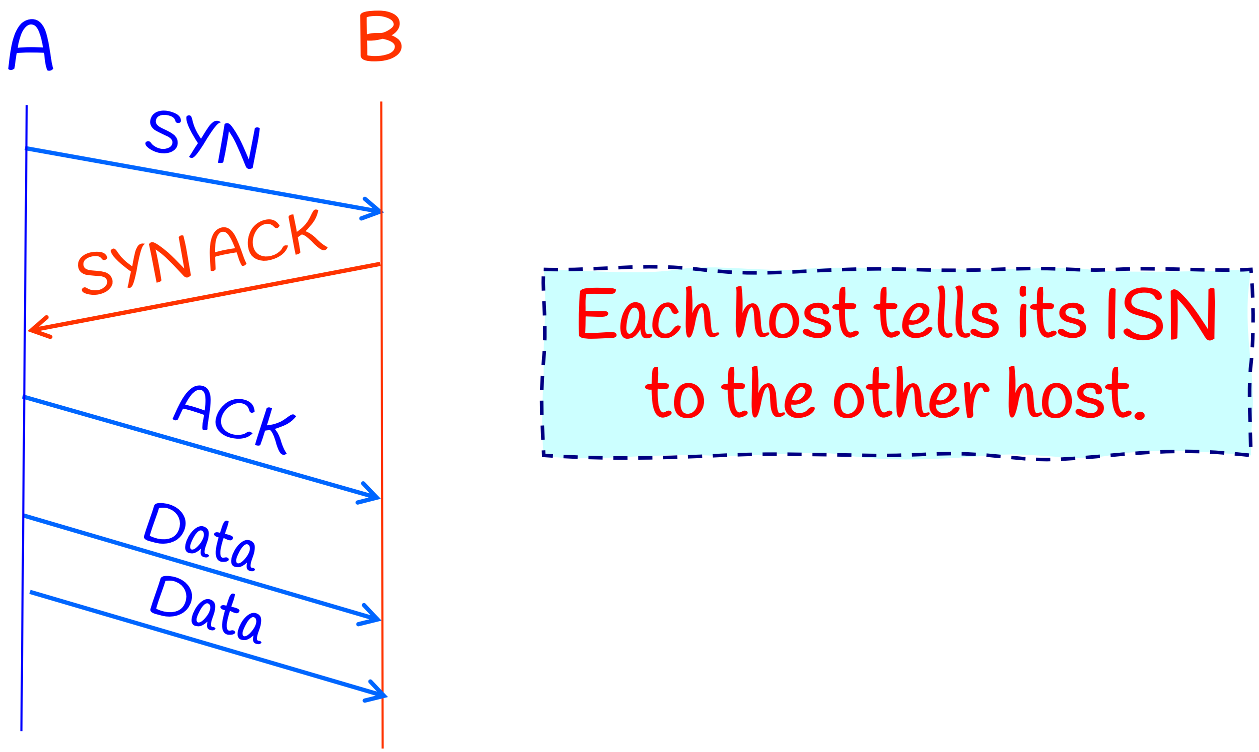


Initial Sequence Number (ISN)

- Sequence number for the very first byte
  - E.g., Why not a de facto ISN of 0?
- Practical issue
  - IP addresses and port #s uniquely identify a connection
  - Eventually, though, these port #s do get used again
  - … and there is a chance an old packet is still in flight
  - … and might be associated with the new connection
- So, TCP requires changing the ISN over time
  - Set from a 32-bit clock that ticks every 4 microseconds
  - … which only wraps around once every 4.55 hours!
- But, this means the hosts need to exchange ISNs

TCP Three-Way Handshake

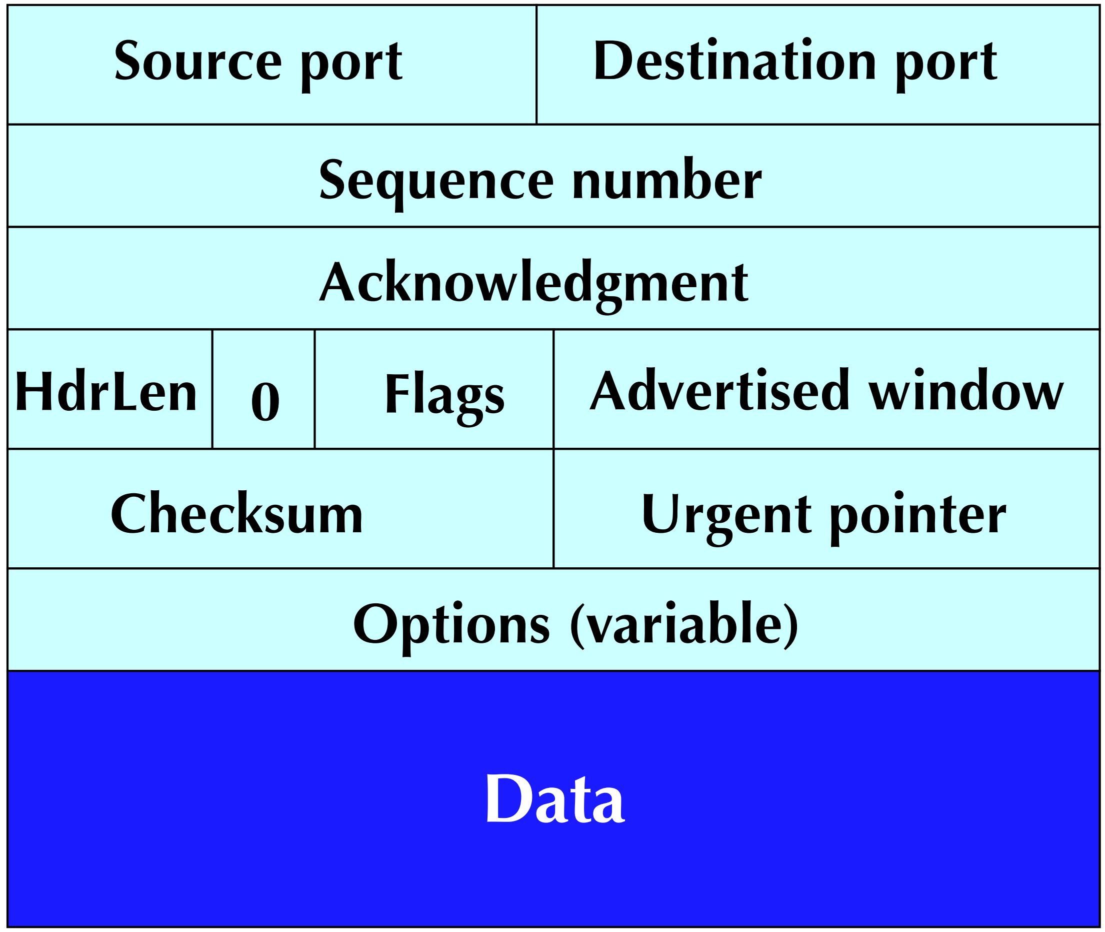
Establishing a TCP Connection



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

TCP Header

Flags: SYN
FIN
RST
PSH
URG
ACK



Step 1: A's Initial SYN Packet

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

A's port	B's port
A's Initial Sequence Number	
Acknowledgment	
20 | 0 | Flags	Advertised window
Checksum	Urgent pointer
Options (variable)	

A tells B it wants to open a connection…

Step 2: B's SYN-ACK Packet

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

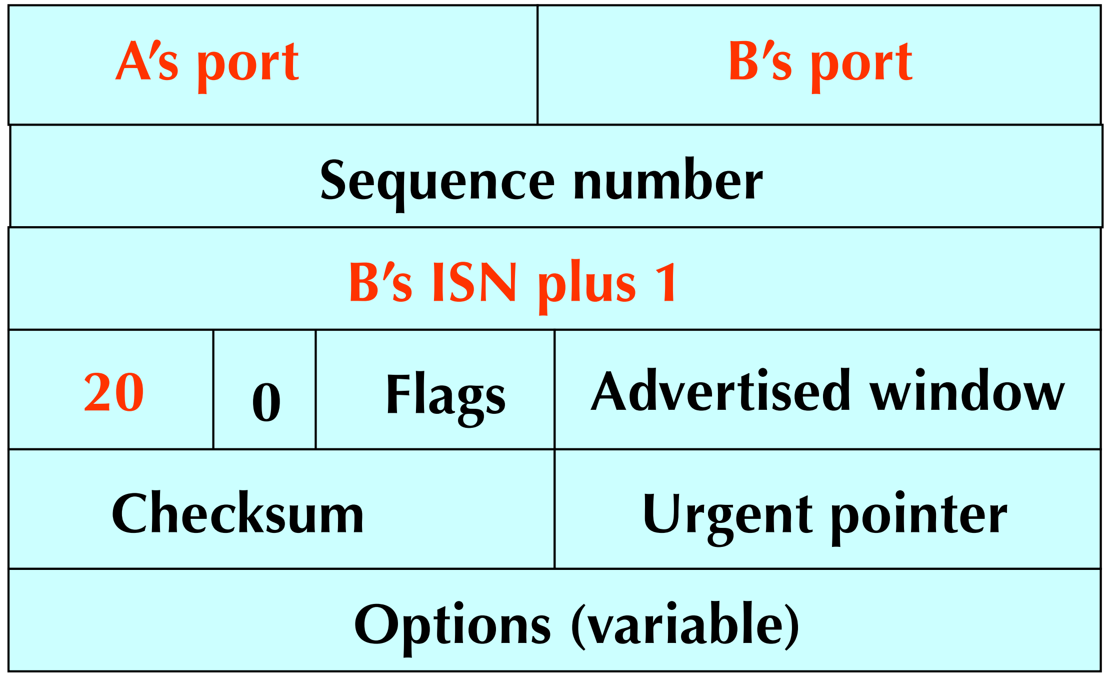
<table>
<tr><td colspan="3">B's port</td><td>A's port</td></tr>
<tr><td colspan="4">B's Initial Sequence Number</td></tr>
<tr><td colspan="4">A's ISN plus 1</td></tr>
<tr><td>20</td><td>0</td><td>Flags</td><td>Advertised window</td></tr>
<tr><td colspan="3">Checksum</td><td>Urgent pointer</td></tr>
<tr><td colspan="4">Options (variable)</td></tr>
</table>

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

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



A tells B it is okay to start sending

... upon receiving this packet, B can start sending data

What if the SYN Packet Gets Lost?

- Suppose the SYN packet gets lost
  - Packet is lost inside the network, or
  - Server rejects the packet (e.g., listen queue is full)
- Eventually, no SYN-ACK arrives
  - Sender sets a timer and wait 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
  - Some TCPs use a default of 3 or 6 seconds

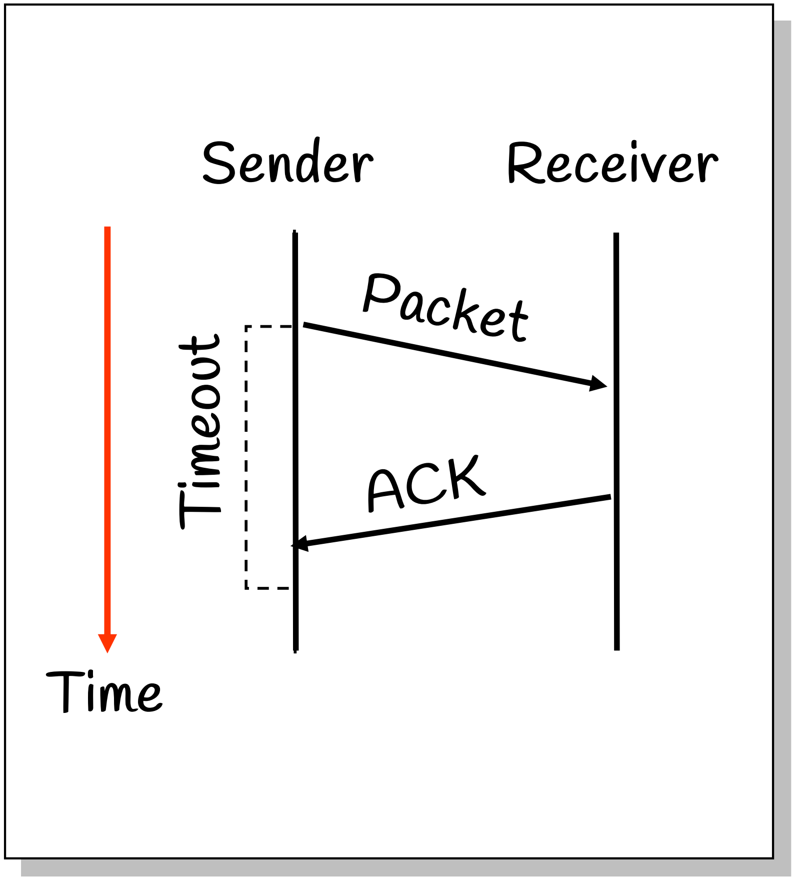
SYN Loss and Web Downloads

- 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…
  - The 3-6 seconds of delay may be very long
  - The user may get impatient
  - … and click the hyperlink again, or click "reload"
- User triggers an "abort" of the "connect"
  - Browser creates a new socket and does a "connect"
  - Essentially, forces a faster send of a new SYN packet!
  - Sometimes very effective, and the page comes fast

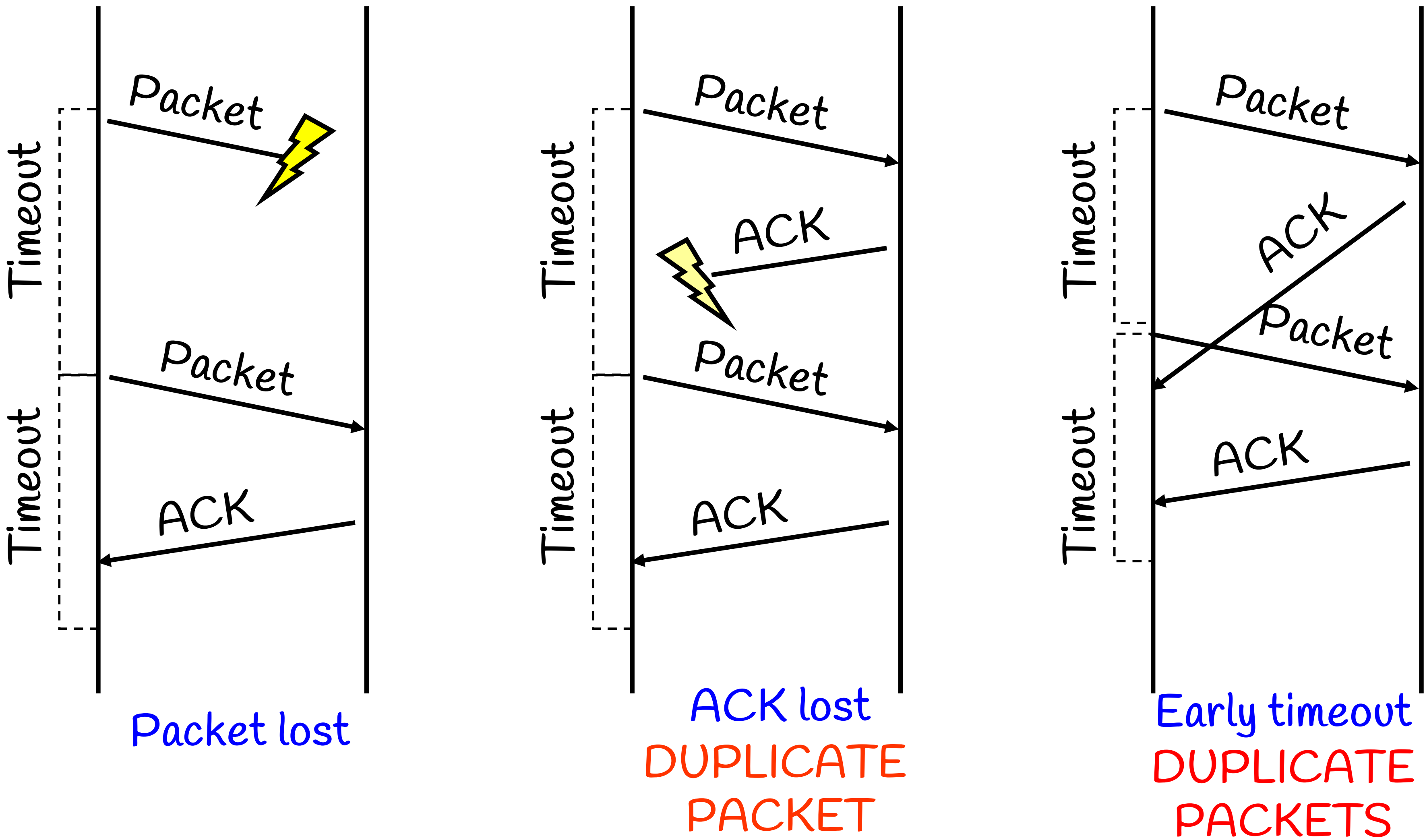
TCP Retransmissions

Automatic Repeat reQuest (ARQ)

- Automatic Repeat reQuest
  - Receiver sends acknowledgment (ACK) when it receives packet
  - Sender waits for ACK and timeouts if it does not arrive within some time period
- Simplest ARQ protocol
  - Stop and wait
  - Send a packet, stop and wait until ACK arrives



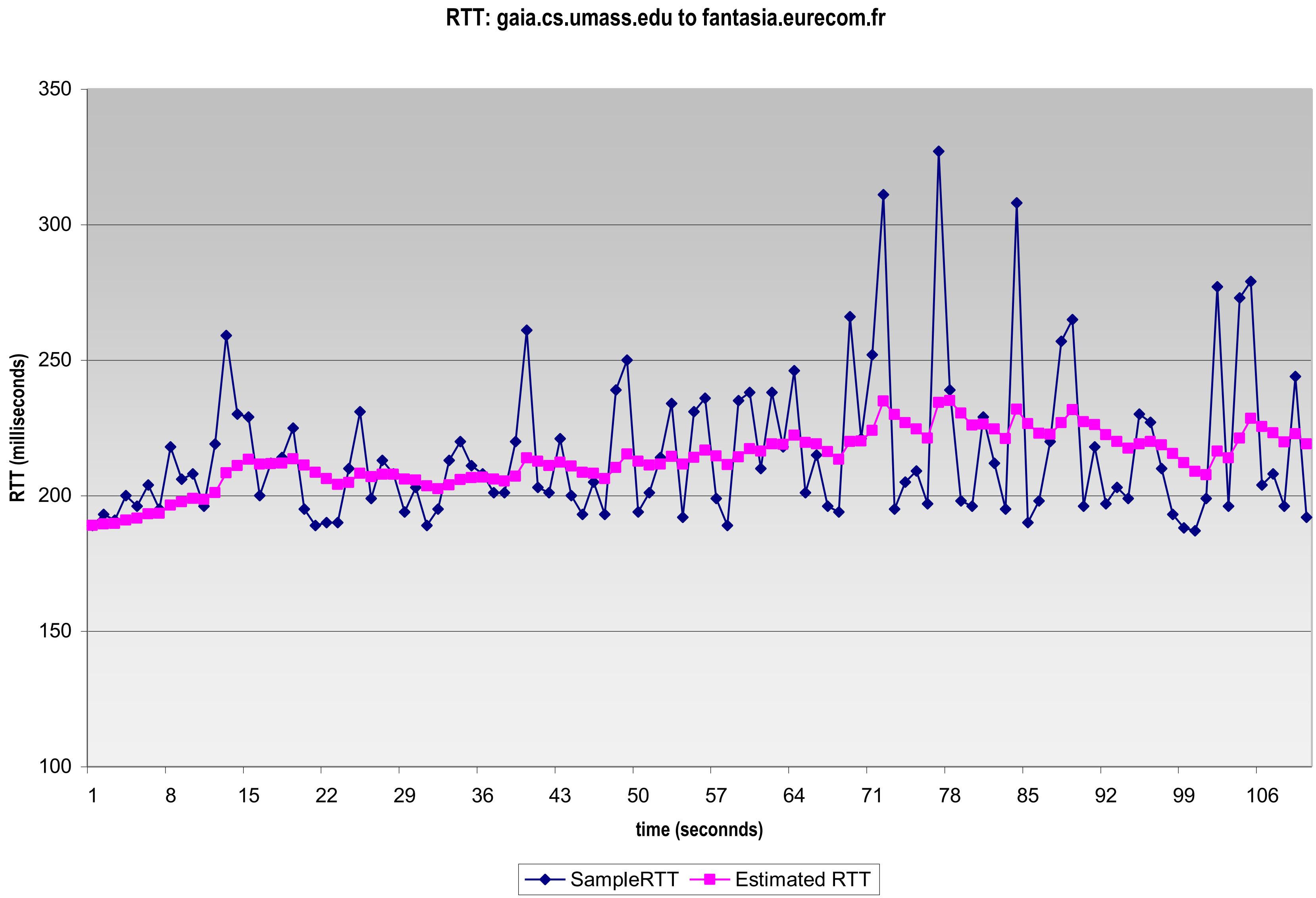
Reasons for Retransmission



How Long Should Sender Wait?

- Sender sets a timeout to wait for an ACK
  - Too short: wasted retransmissions
  - Too long: excessive delays when packet lost
- TCP sets timeout as a function of the RTT
  - Expect ACK to arrive after an RTT
  - … plus a fudge factor to account for queuing
- But, how does the sender know the RTT?
  - Can estimate the RTT by watching the ACKs
  - Smooth estimate: keep a running average of the RTT
    - EstimatedRTT = a * EstimatedRTT + (1 –a) * SampleRTT
  - Compute timeout: TimeOut = 2 * EstimatedRTT

Example RTT Estimation



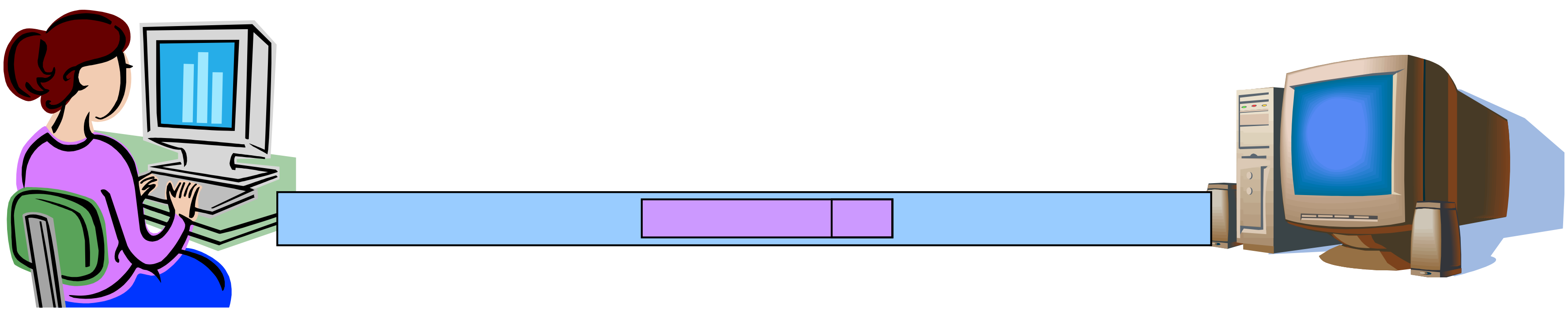
A Flaw in This Approach

- An ACK doesn't really acknowledge a transmission
  - Rather, it acknowledges receipt of the data
- Consider a retransmission of a lost packet
  - If you assume the ACK goes with the 1st transmission
  - … the SampleRTT comes out way too large
- Consider a duplicate packet
  - If you assume the ACK goes with the 2nd transmission
  - … the Sample RTT comes out way too small
- Simple solution in the Karn/Partridge algorithm
  - Only collect samples for segments sent one single time

TCP Sliding Window

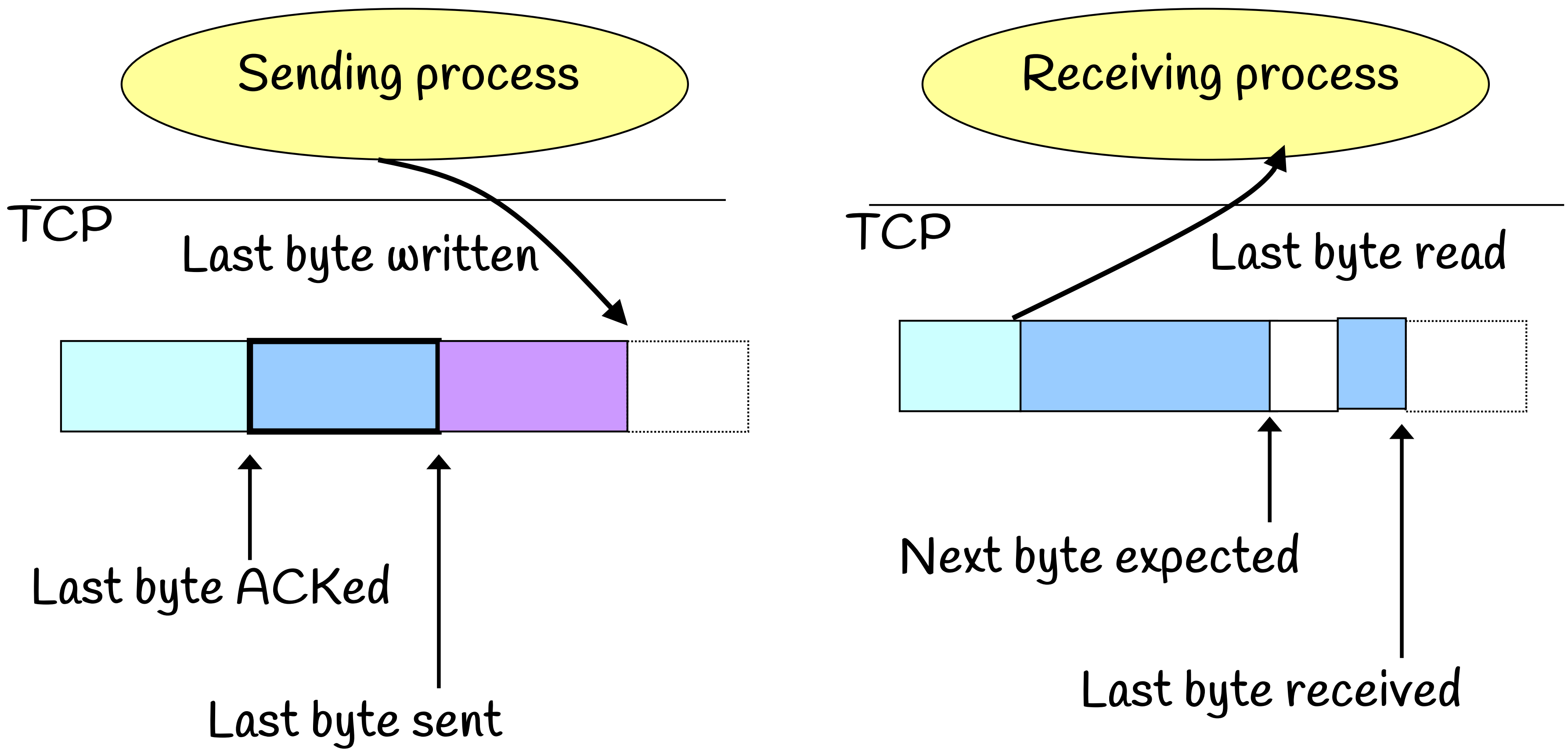
Motivation for Sliding Window

- Stop-and-wait is inefficient
  - Only one TCP segment is "in flight" at a time
  - Especially bad when delay-bandwidth product is high
- Numerical example
  - 1.5 Mbps link with a 45 msec round-trip time (RTT)
    - Delay-bandwidth product is 67.5 Kbits (or 8 KBytes)
  - But, sender can send at most one packet per RTT
    - Assuming a segment size of 1 KB (8 Kbits)
    - … leads to 8 Kbits/segment / 45 msec/segment ➔ 182 Kbps
    - That's just one-eighth of the 1.5 Mbps link capacity



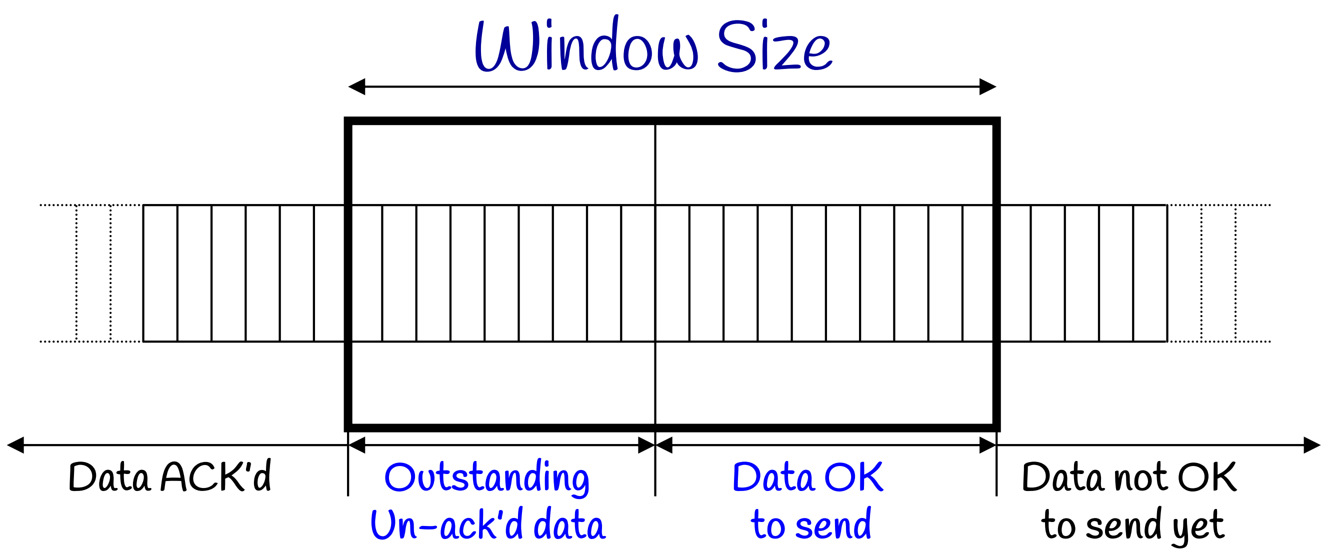
Sliding Window

- Allow a larger amount of data "in flight"
  - Allow sender to get ahead of the receiver
  - … though not too far ahead



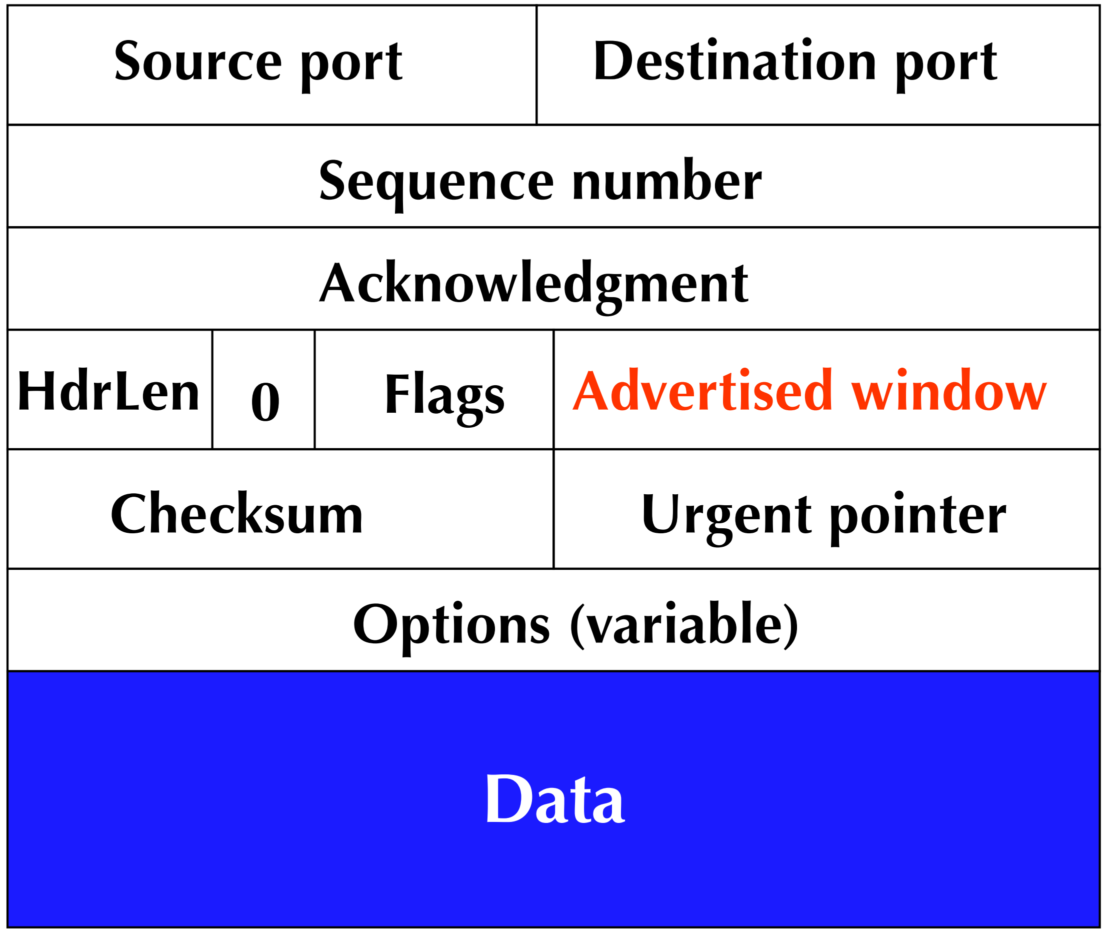
Receiver Buffering

- Window size
  - Amount that can be sent without acknowledgment
  - Receiver needs to be able to store this amount of data
- Receiver advertises the window to the sender
  - Tells the sender the amount of free space left
  - … and the sender agrees not to exceed this amount



TCP Header for Receiver Buffering

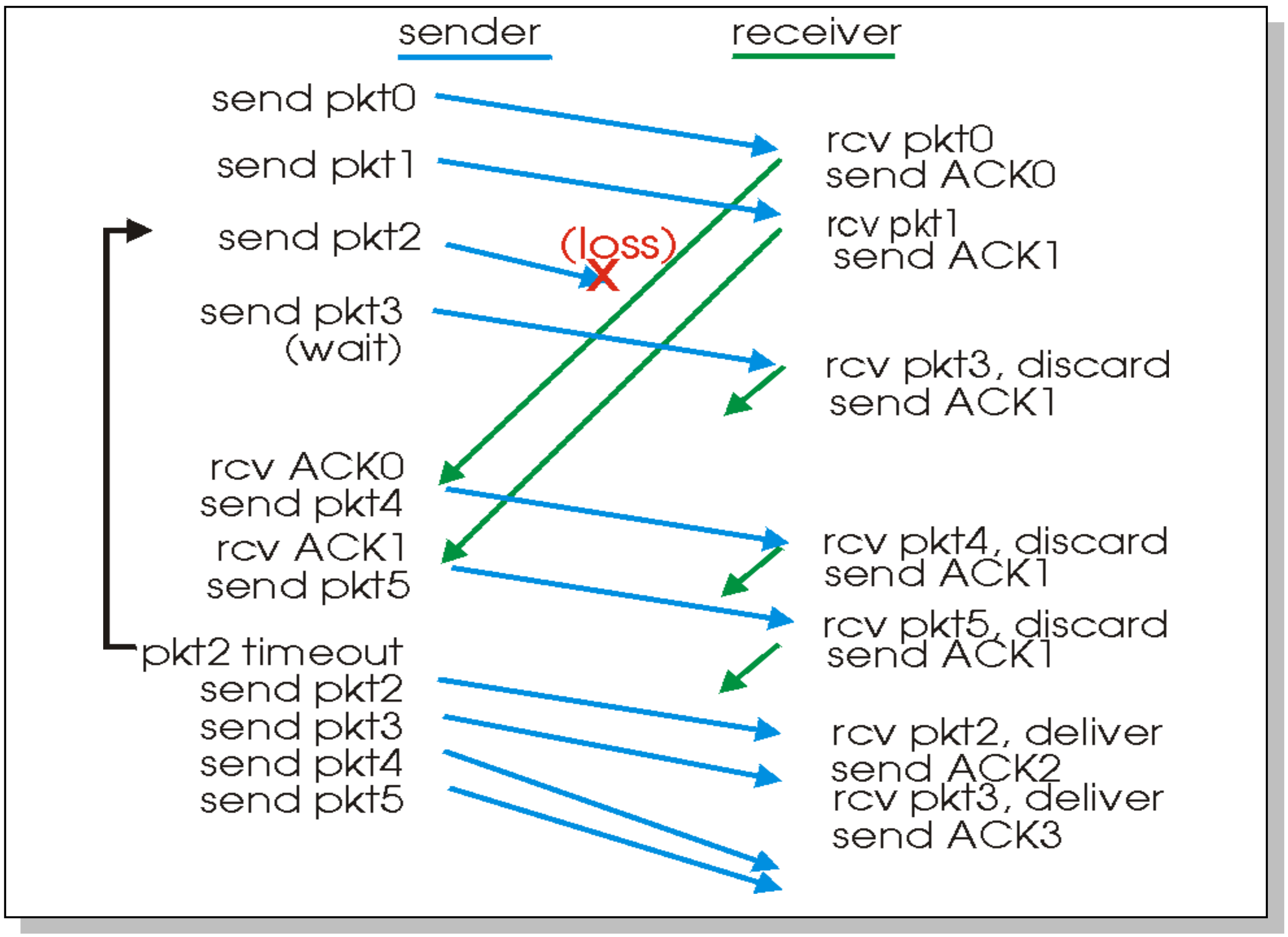
Flags: SYN
FIN
RST
PSH
URG
ACK



Fast Retransmission

Timeout is Inefficient

- Timeout-based retransmission
  - Sender transmits a packet and waits until timer expires
  - … and then retransmits from the lost packet onward



Fast Retransmission

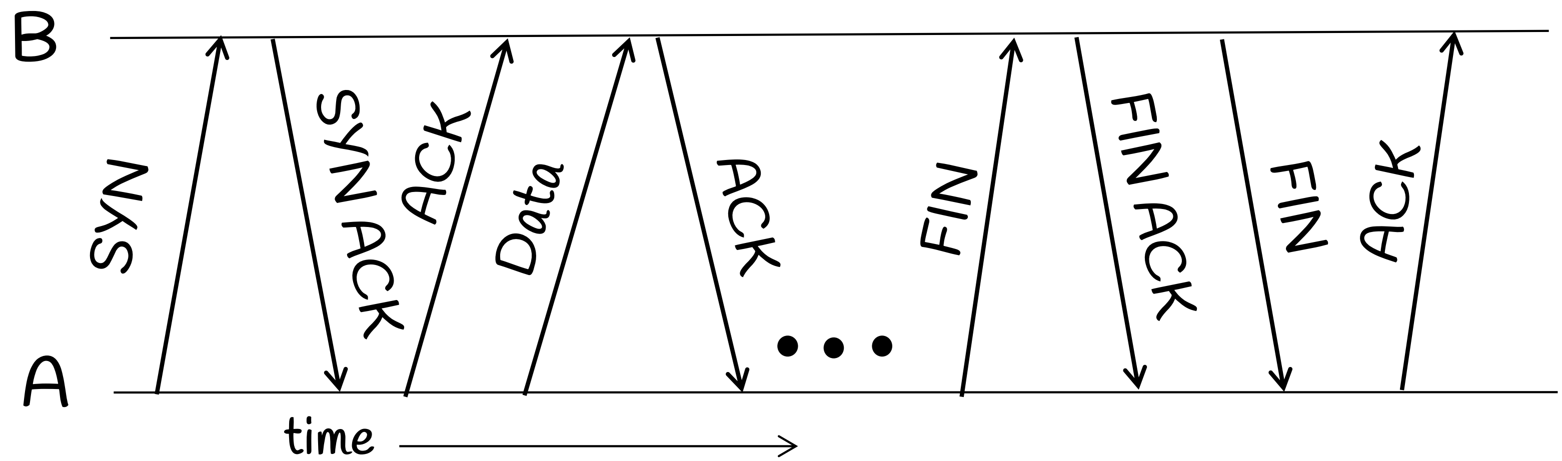
- Better solution possible under sliding window
  - Although packet n might have been lost
  - … packets n+1, n+2, and so on might get through
- Idea: have the receiver send ACK packets
  - ACK says that receiver is still awaiting nth packet
    - And repeated ACKs suggest later packets have arrived
  - Sender can view the “duplicate ACKs” as an early hint
    - … that the nth packet must have been lost
    - … and perform the retransmission early
- Fast retransmission
  - Sender retransmits data after the triple duplicate ACK

Effectiveness of Fast Retransmit

- When does Fast Retransmit work best?
  - Long data transfers
    - High likelihood of many packets in flight
  - High window size
    - High likelihood of many packets in flight
  - Low burstiness in packet losses
    - Higher likelihood that later packets arrive successfully
- Implications for Web traffic
  - Most Web transfers are short (e.g., 10 packets)
    - Short HTML files or small images
  - So, often there aren't many packets in flight
  - … making fast retransmit less likely to "kick in"
  - Forcing users to like "reload" more often… ☺

Tearing Down the Connection

Tearing Down the Connection



- Closing the connection
  - Finish (FIN) to close and receive remaining bytes
  - And other host sends a FIN ACK to acknowledge
  - Reset (RST) to close and not receive remaining bytes

Sending/Receiving the FIN Packet

- Sending a FIN: close()
  - Process is done sending data via the socket
  - Process invokes “close()” to close the socket
  - Once TCP has sent all of the outstanding bytes…
  - … then TCP sends a FIN
- Receiving a FIN: EOF
  - Process is reading data from the socket
  - Eventually, the attempt to read returns an EOF

Conclusions

- Transport protocols
  - Multiplexing and demultiplexing
  - Sequence numbers
  - Window-based flow control
  - Timer-based retransmission
  - Checksum-based error detection

- Next lecture (after reading week and midterm)
  - Congestion control