

CSC 458/2209 – Computer Networks

Handout # 9: Transport Protocols



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Announcements

- Programming assignment 1
 - Due: Friday, October 14th (5PM)
- This week
 - Chapter 5 of the textbook
 - Tutorial: programming assignment 1
- Next week: no tutorial
- Sample midterm and solutions on class web site

Midterm Exam

- Next week
 - Section L0101: Thu. Oct. 17th, 1-3 PM
 - Section L5101: Tue. Oct. 22nd, 6-8 PM
 - Section L0201: Tue. Oct. 22nd, 1-3 PM
 - Same room and time as the lecture
 - For undergraduate and graduate students
- Everything covered up to the end of today's lecture
 - Emphasis on the slides, problem set, and sample midterm provided.
- I won't have office hours next week.

Role of Transport Layer

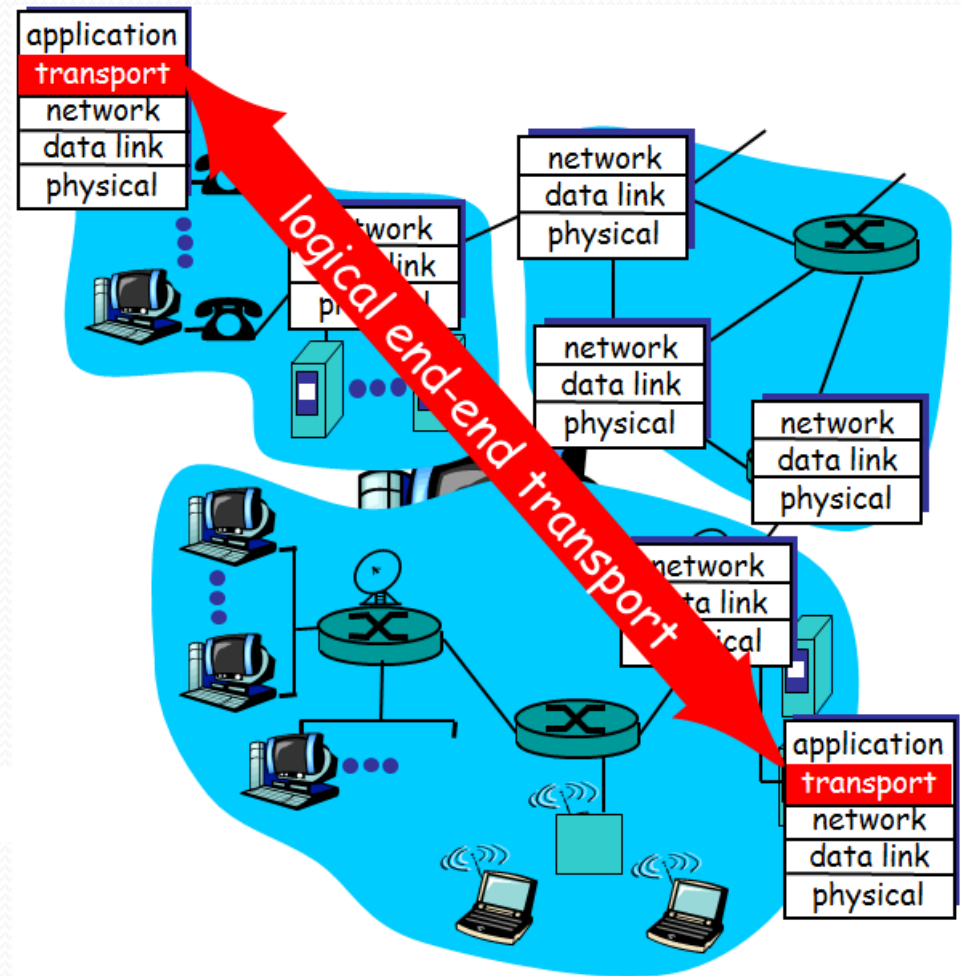
- Link layer
 - Transfer bit frames between neighboring nodes
 - E.g., Ethernet
- Network layer
 - Logical communication between nodes
 - Hides details of the link technology
 - E.g., IP
- Transport layer
 - Communication between processes (e.g., socket)
 - Relies on network layer and serves the application 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 protocols available to applications
 - Internet: TCP and UDP

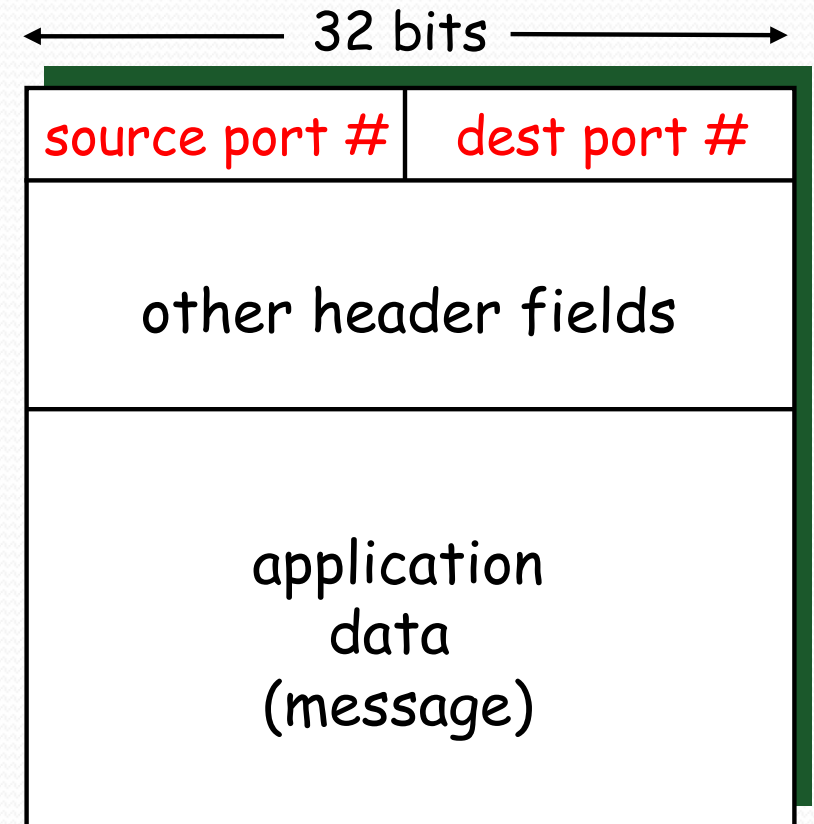


Internet Transport Protocols

- 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
 - Congestion control (next lecture)
- Other services not available
 - Delay guarantees
 - Bandwidth guarantees

Multiplexing and Demultiplexing

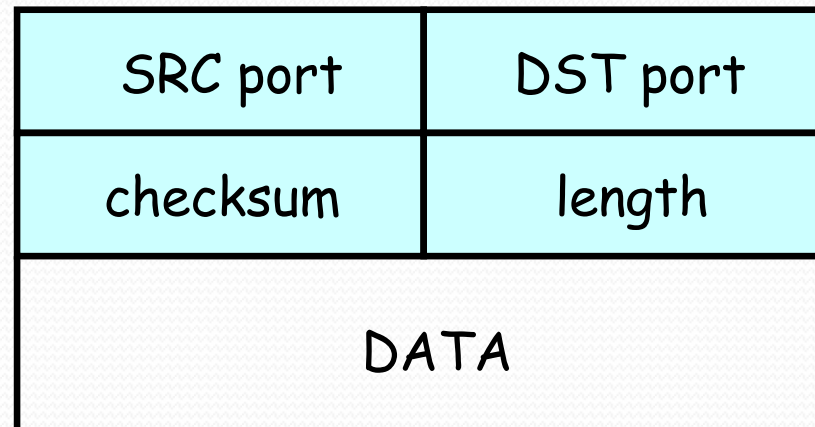
- Host receives IP datagrams
 - Each datagram has source and destination IP address,
 - Each datagram carries one transport-layer segment
 - Each segment has source and destination port number
- Host uses IP addresses and port numbers to direct the segment to appropriate socket



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

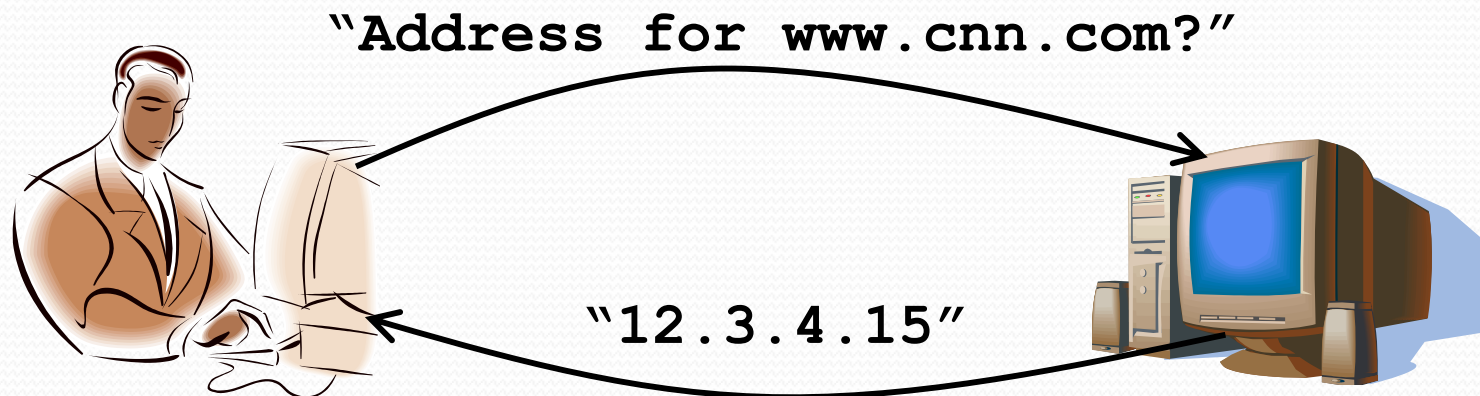


Why Would Anyone Use UDP?

- 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

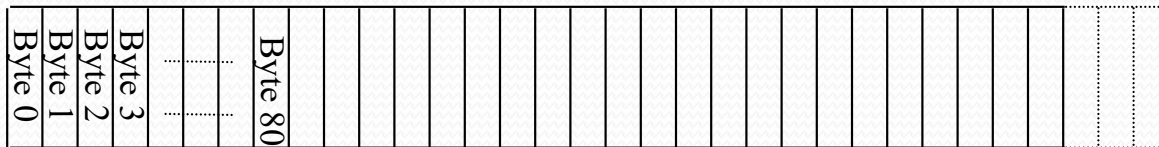
TCP Support for Reliable Delivery

- 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

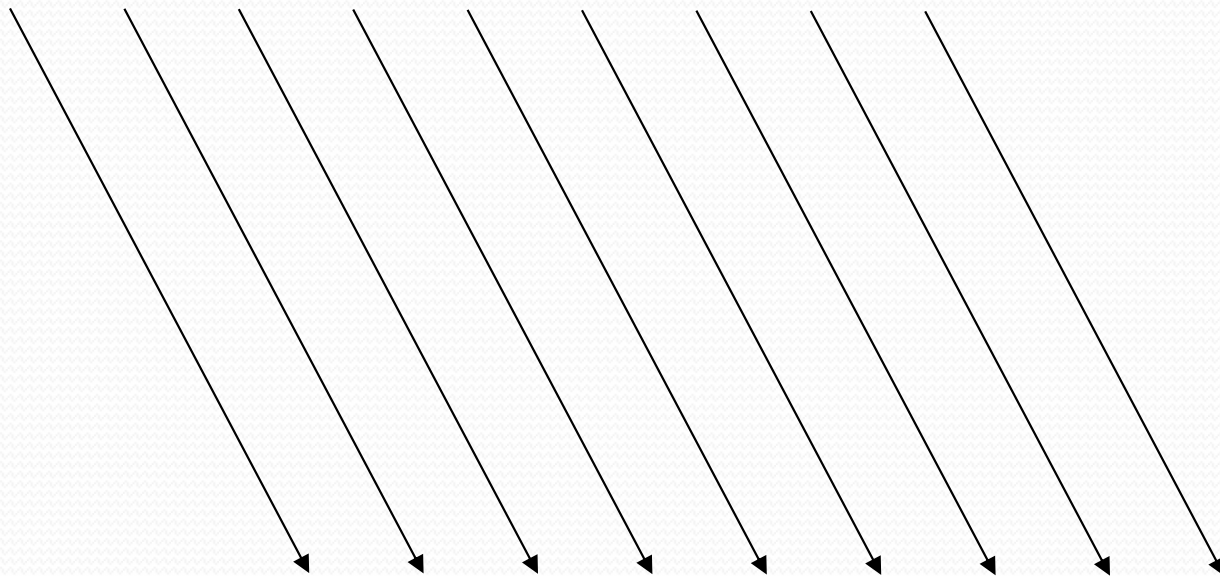
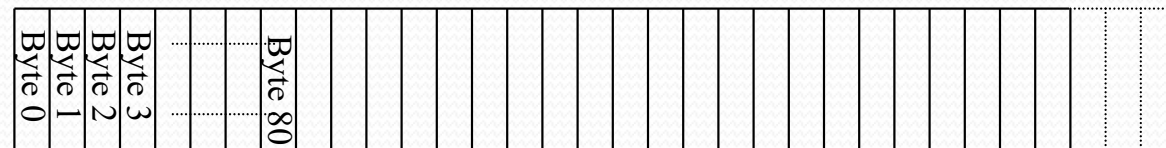
TCP Segments

TCP “Stream of Bytes” Service

Host A

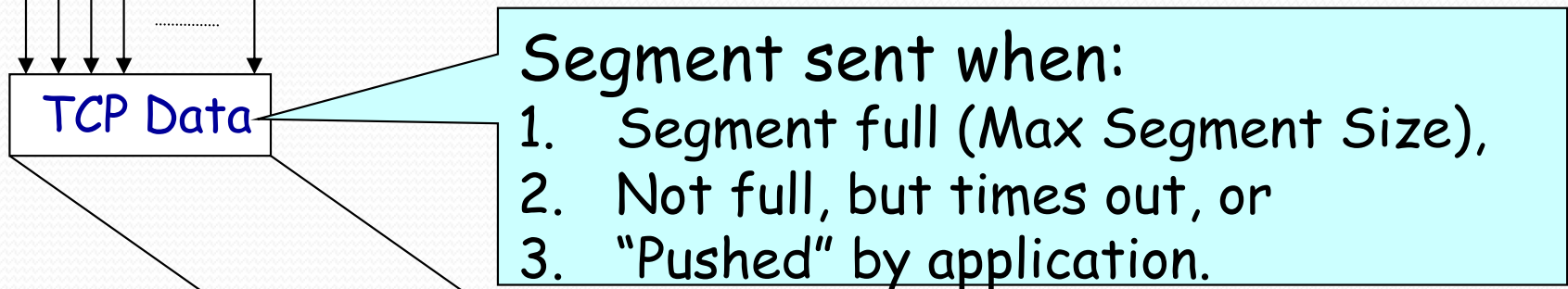
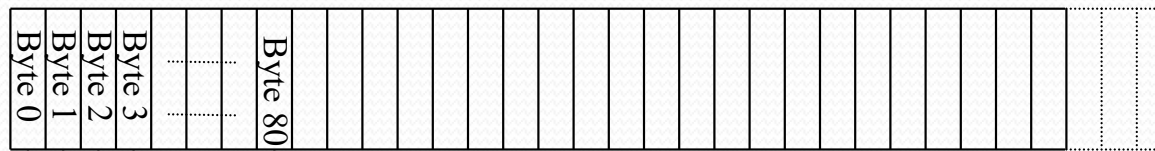


Host B

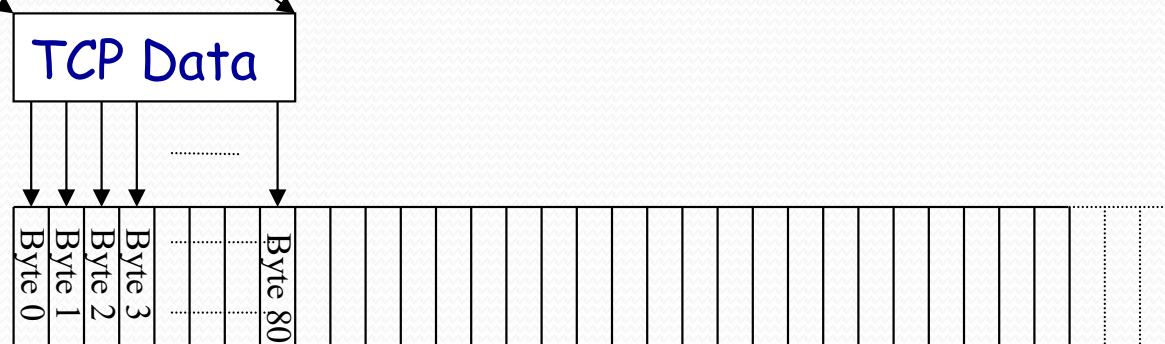


...Emulated Using TCP "Segments"

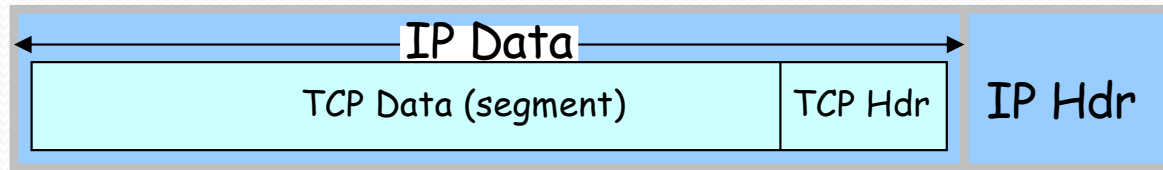
Host A



Host B



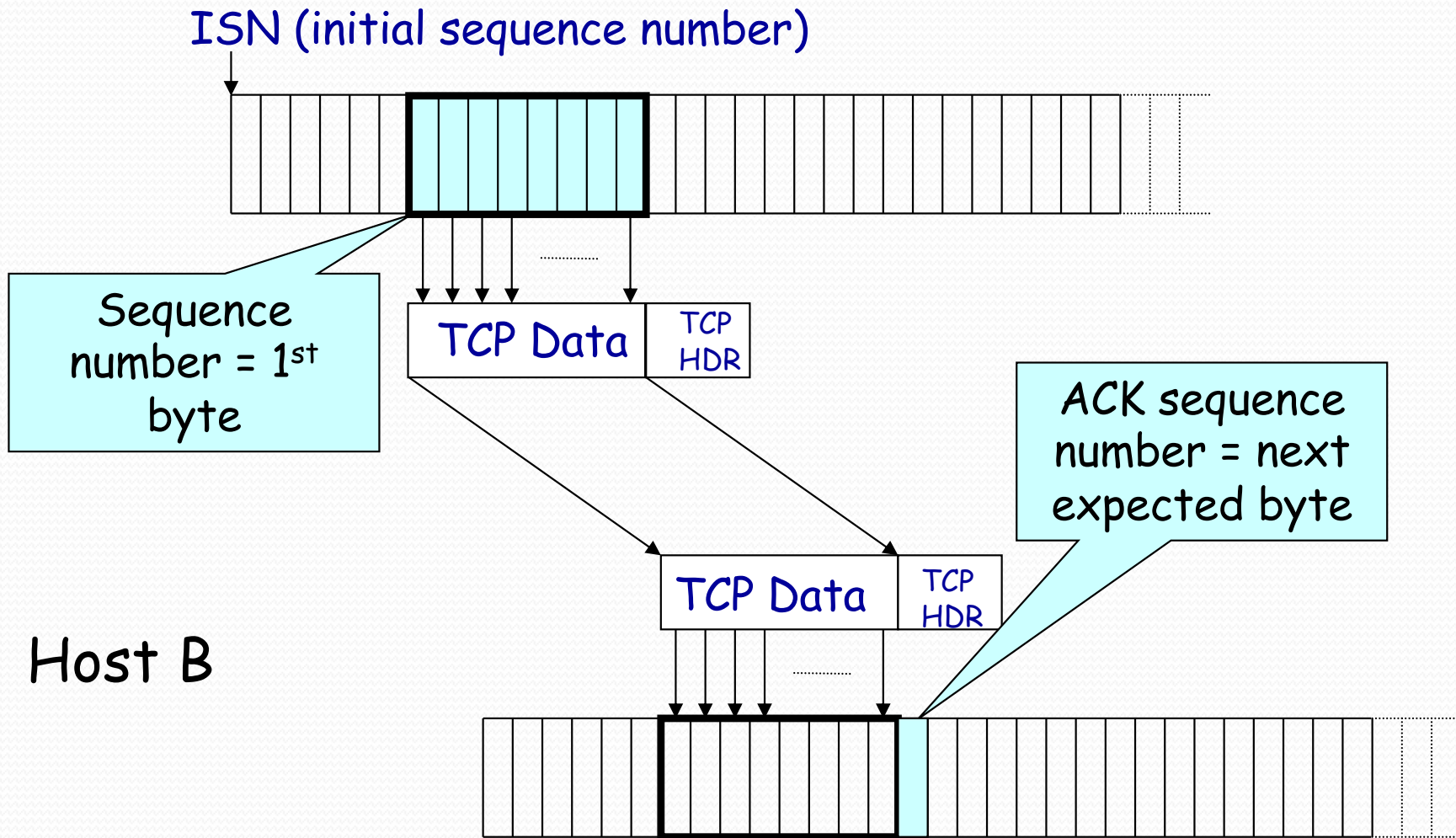
TCP Segment



- 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

Host A

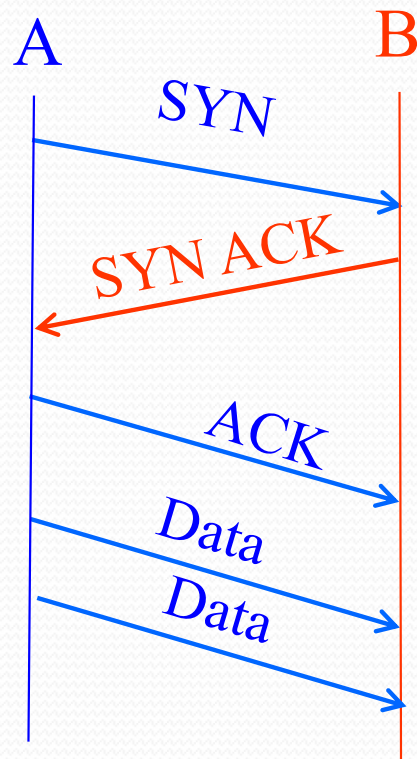


Initial Sequence Number (ISN)

- 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

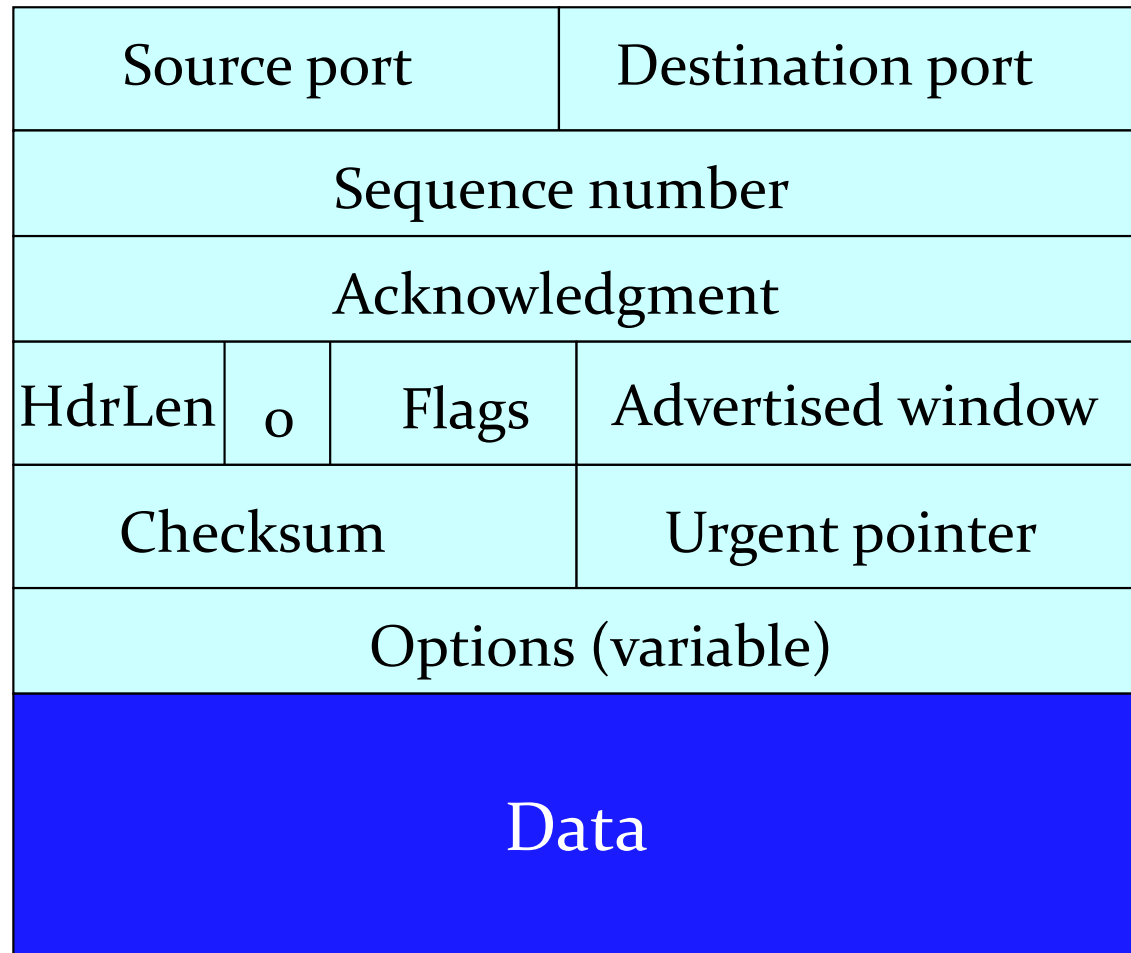


Each host tells its ISN to the other host.

- 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

B's port		A's port	
B's Initial Sequence Number			
A's ISN plus 1			
20	0	Flags	Advertised window
Checksum		Urgent pointer	
Options (variable)			

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's port		B's port	
Sequence number			
B's ISN plus 1			
20	0	Flags	Advertised window
Checksum		Urgent pointer	
Options (variable)			

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

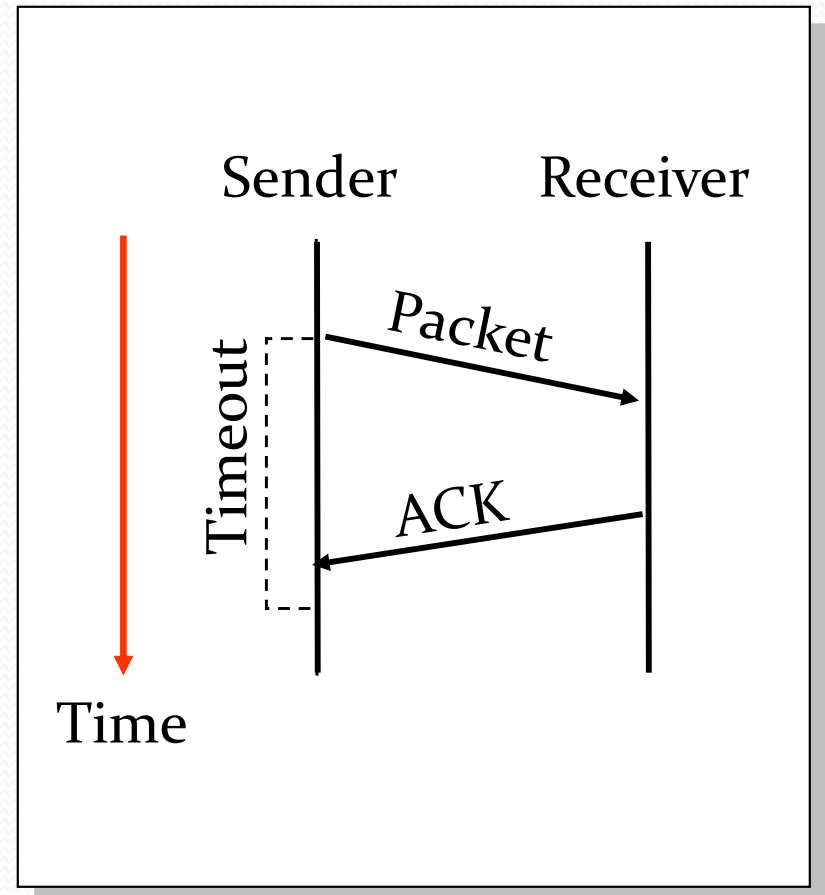
SYN Loss and Web Downloads

- 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

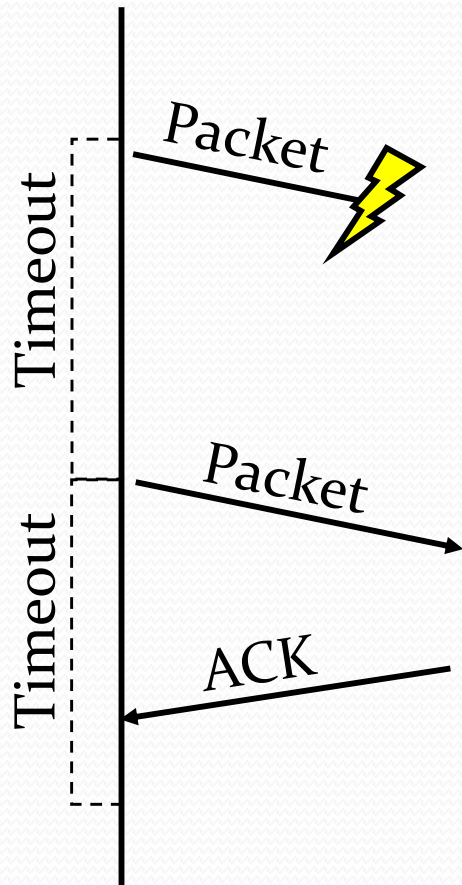
TCP Retransmissions

Automatic Repeat reQuest (ARQ)

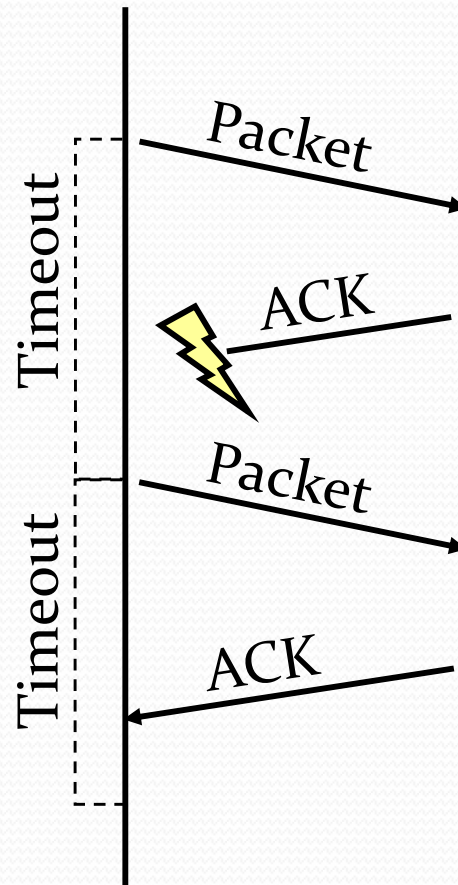
- Automatic Repeat reQuest
 - Receiver sends acknowledgment (ACK) when it receives packet
 - Sender waits for ACK and timeouts if it does not arrive within some time period
- Simplest ARQ protocol
 - Stop and wait
 - Send a packet, stop and wait until ACK arrives



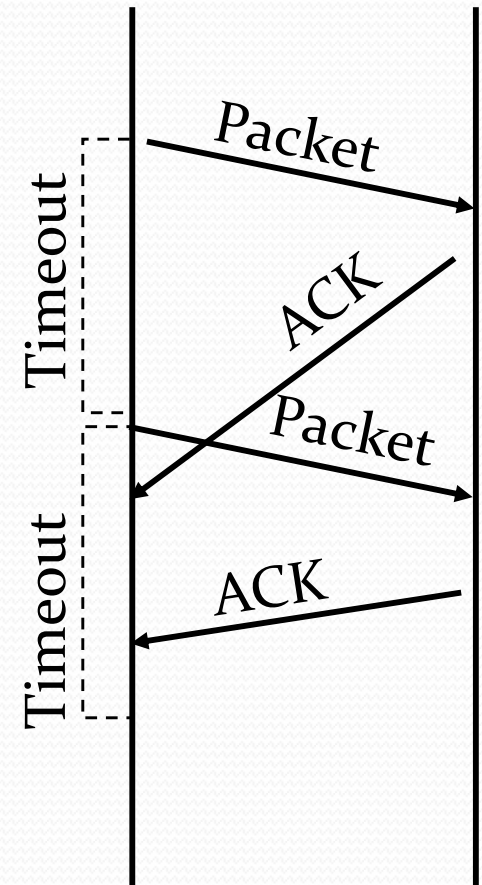
Reasons for Retransmission



Packet lost



**ACK lost
DUPLICATE
PACKET**



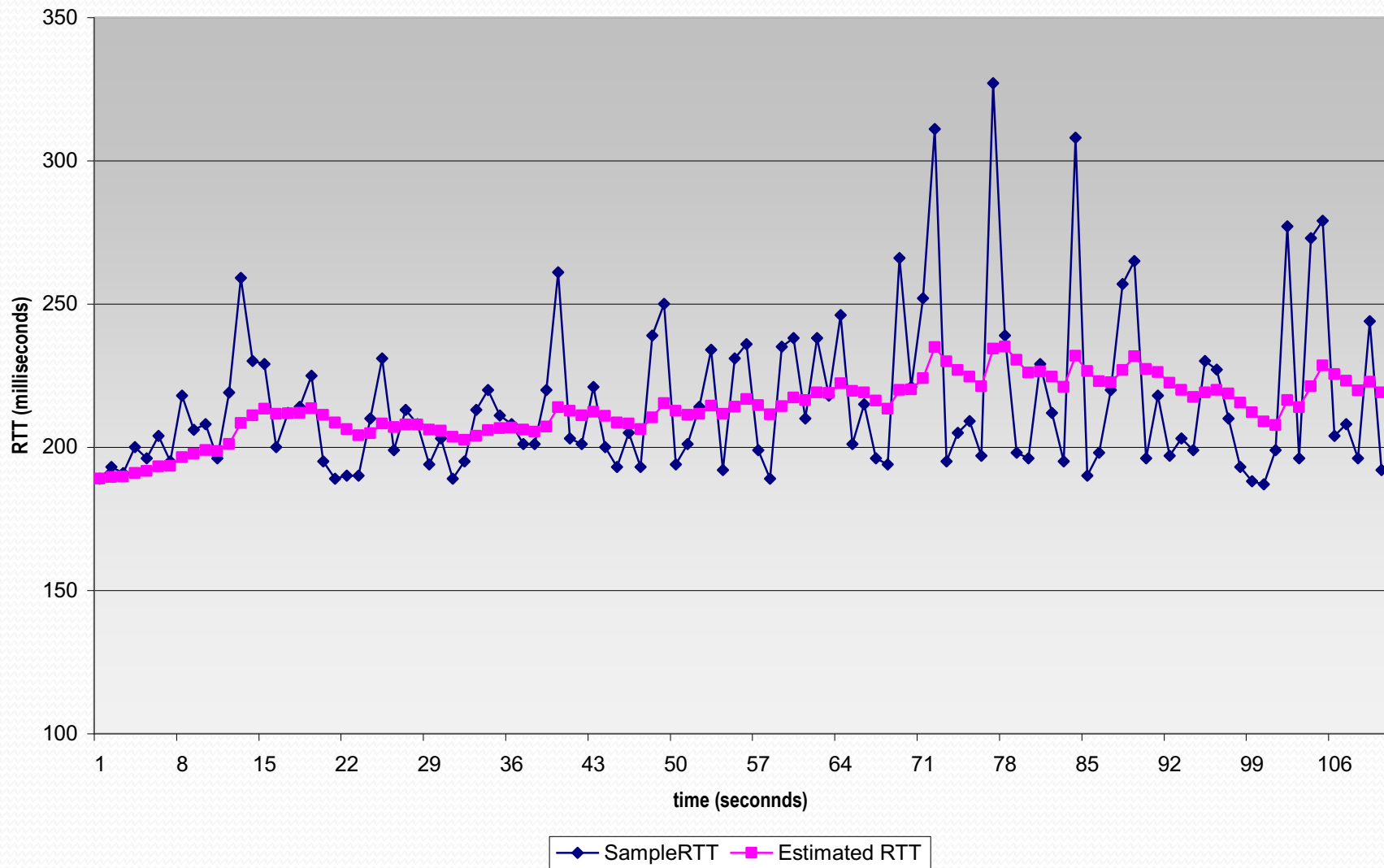
**Early timeout
DUPLICATE
PACKETS**

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
 - $\text{EstimatedRTT} = a * \text{EstimatedRTT} + (1 - a) * \text{SampleRTT}$
 - Compute timeout: $\text{TimeOut} = 2 * \text{EstimatedRTT}$

Example RTT Estimation

RTT: gaia.cs.umass.edu to fantasia.eurecom.fr



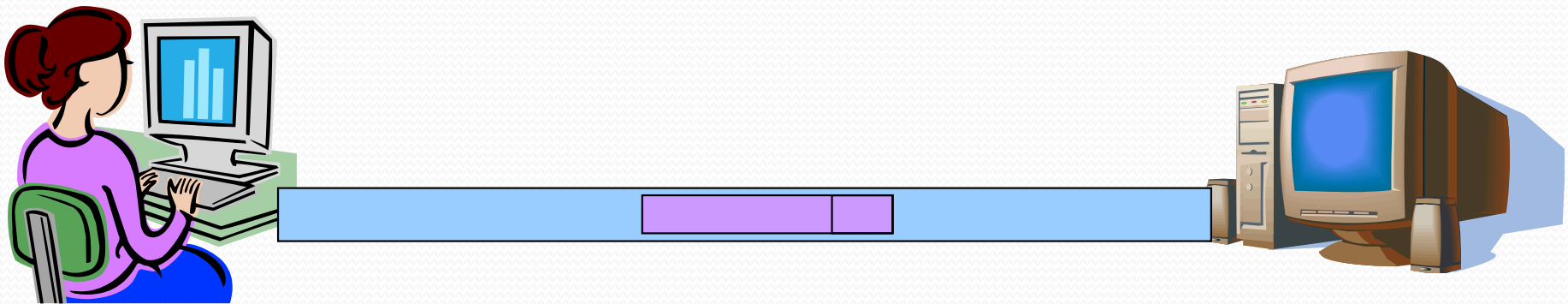
A Flaw in This Approach

- 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

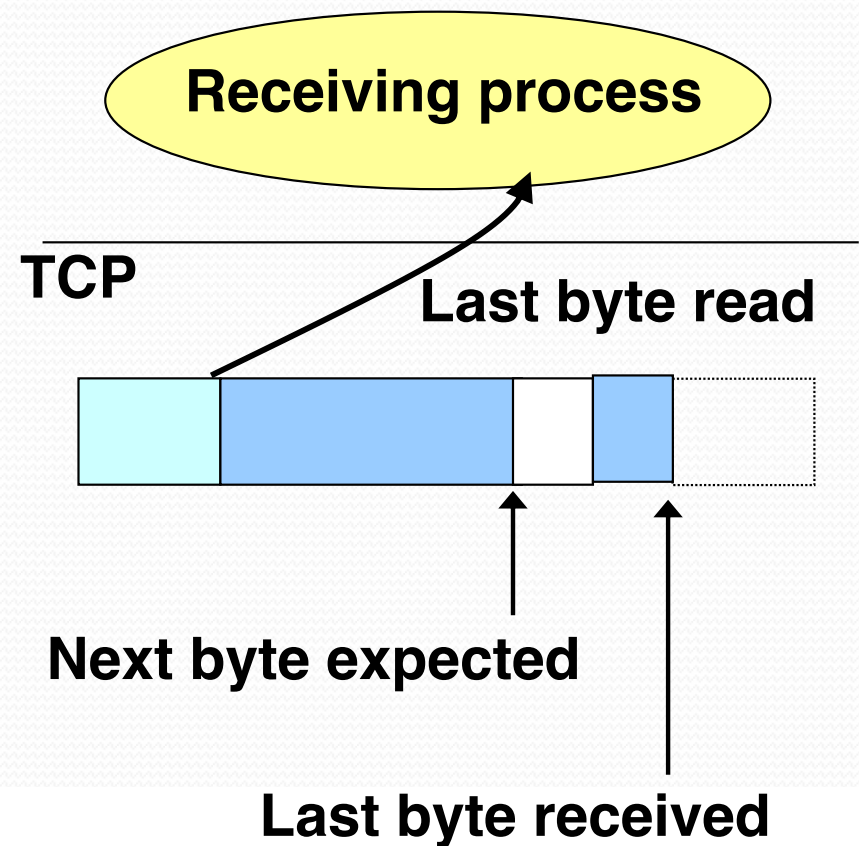
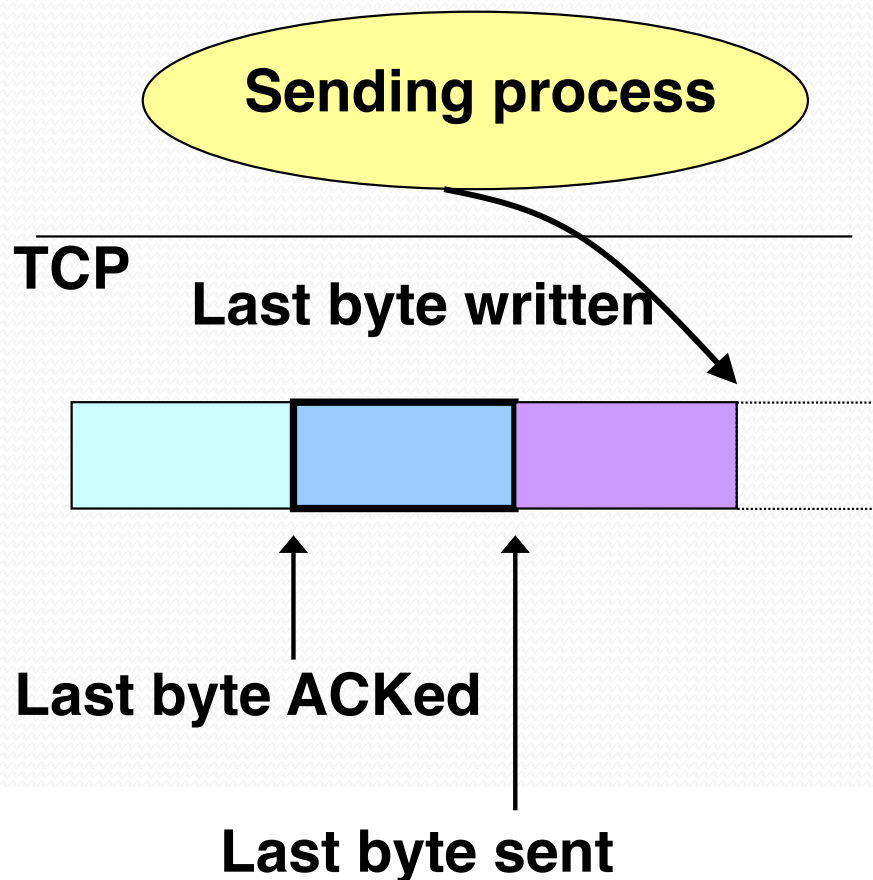
Motivation for Sliding Window

- 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



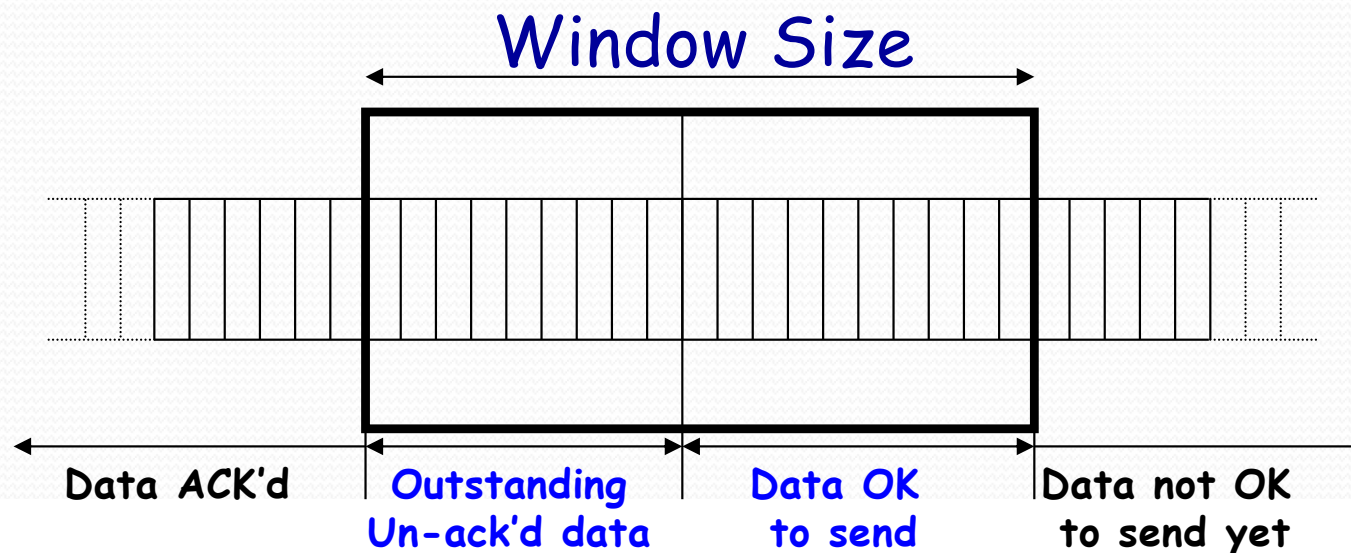
Sliding Window

- 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

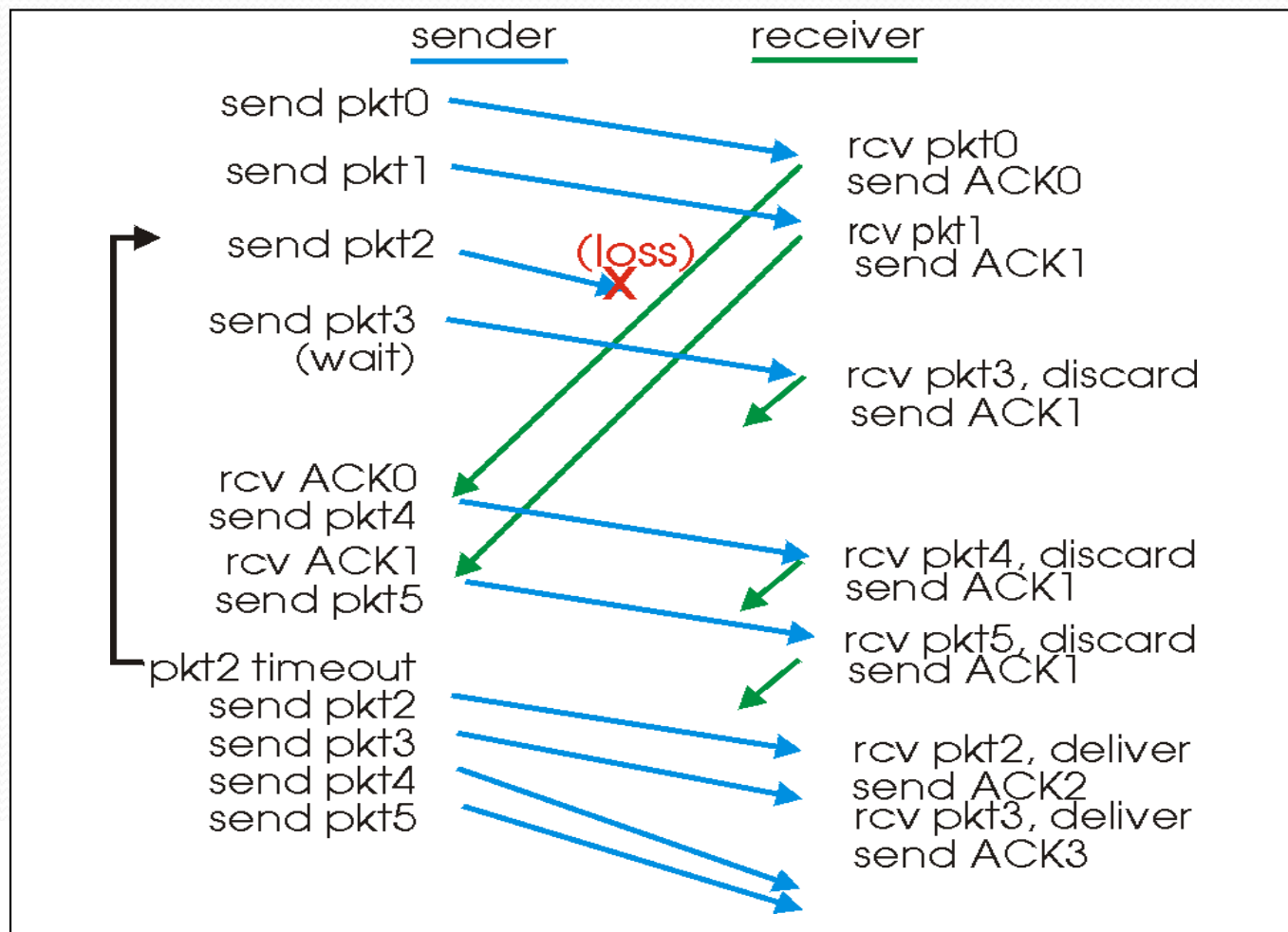
Flags: SYN
FIN
RST
PSH
URG
ACK

Source port		Destination port	
Sequence number			
Acknowledgment			
HdrLen	o	Flags	Advertised window
Checksum		Urgent pointer	
Options (variable)			
Data			

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

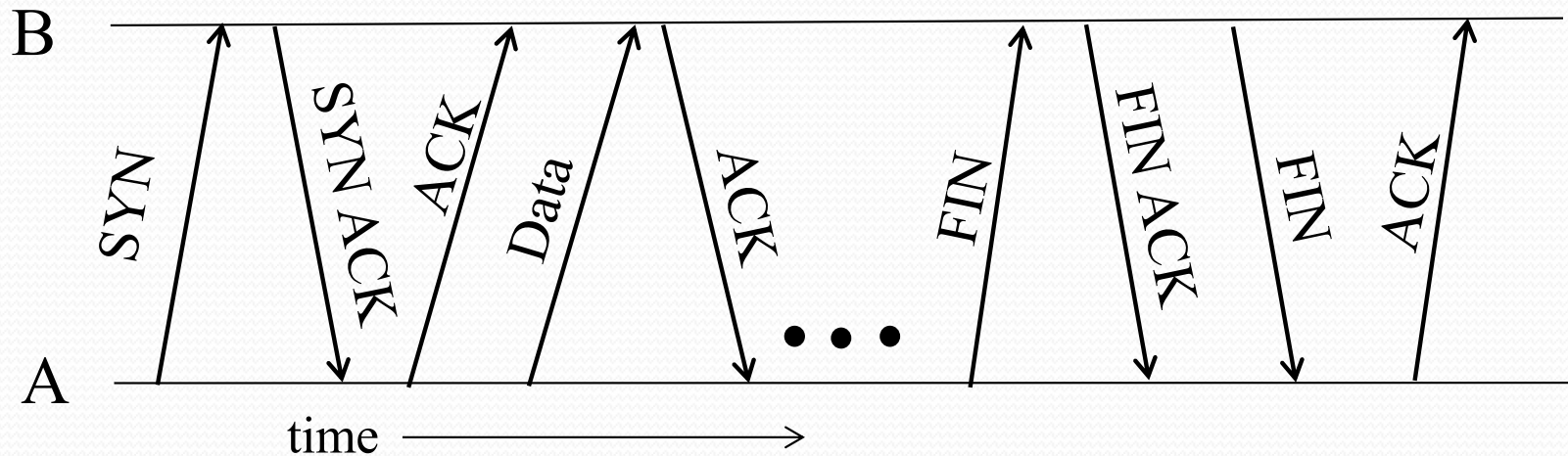
- Better solution possible under sliding window
 - Although packet n might have been lost
 - ... packets $n+1$, $n+2$, and so on might get through
- Idea: have the receiver send ACK packets
 - ACK says that receiver is still awaiting n th packet
 - And repeated ACKs suggest later packets have arrived
 - Sender can view the “duplicate ACKs” as an early hint
 - ... that the n th 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 midterm)
 - Congestion control