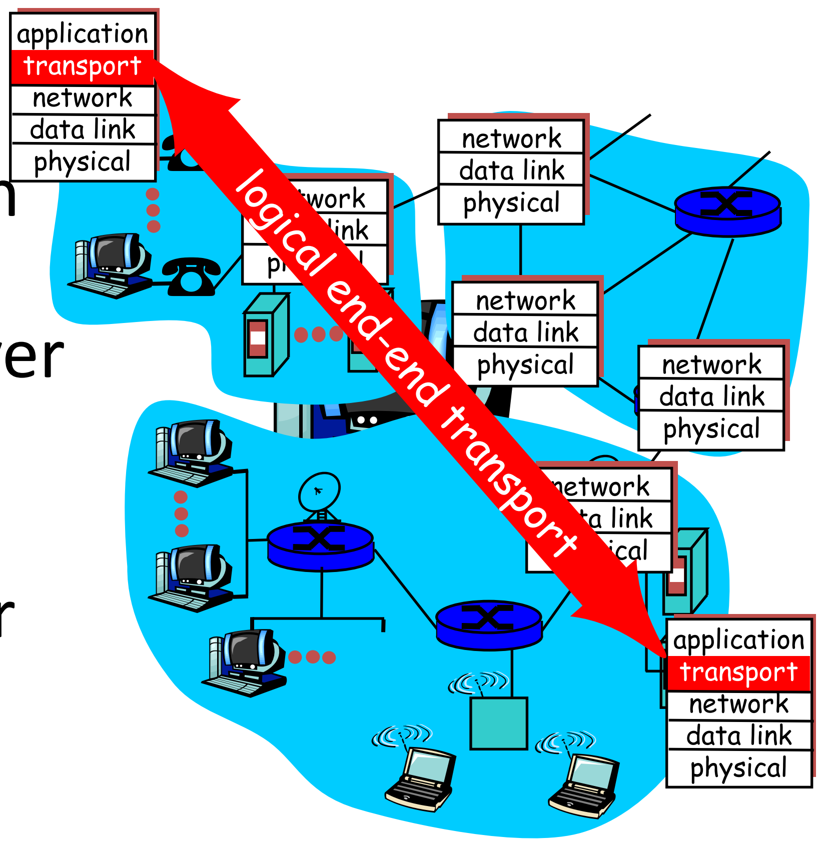
Transport Protocols

Role of Transport Layer

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

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

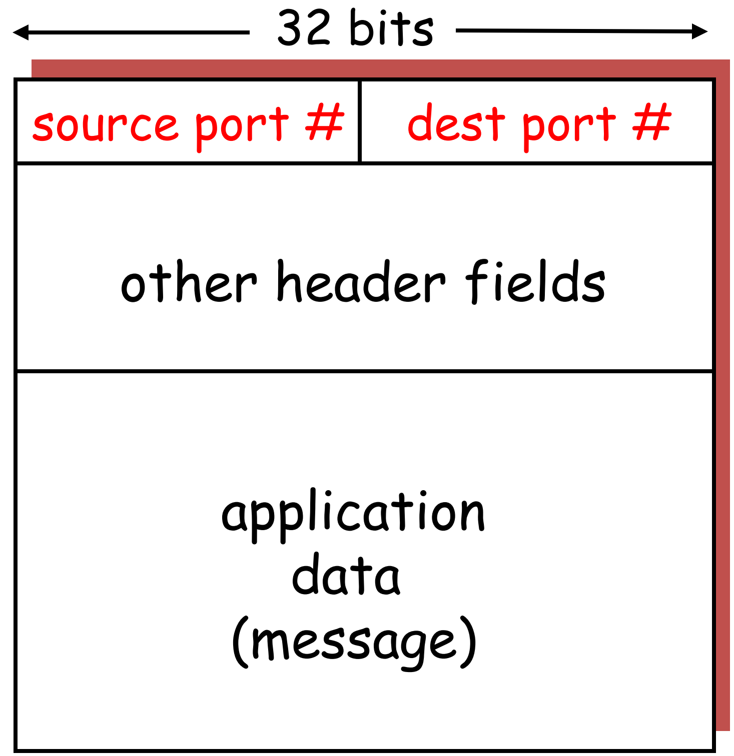


Internet Transport Protocols

- 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
  - Congestion control
- Other services not available
  - Delay guarantees
  - Bandwidth guarantees

Multiplexing and Demultiplexing

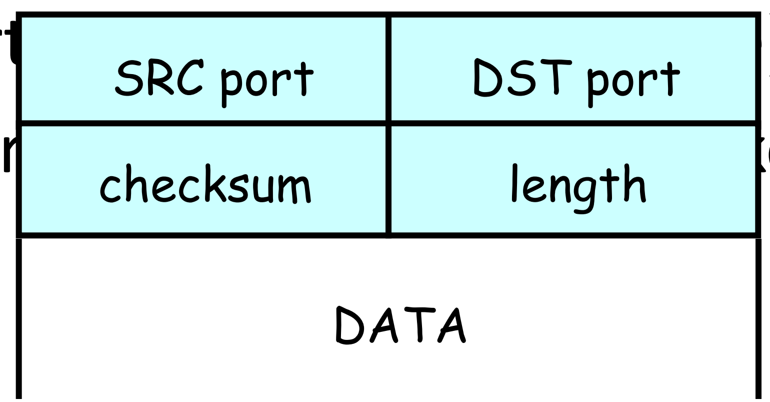
- Host receives IP datagrams
  - Each datagram has source and destination IP address,
  - Each datagram carries one transport-layer segment
  - Each segment has source and destination port number
- Host uses IP addresses and port numbers to direct the segment to appropriate socket



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)multiplexing
  - Optional er ket contents



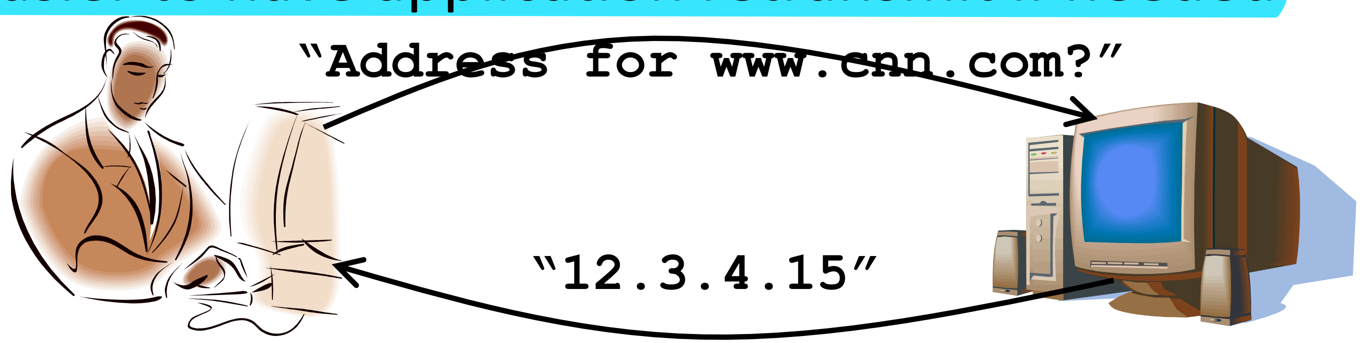
Why Would Anyone Use UDP?

- 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 Alice couldn’t understand Bob?
  - 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

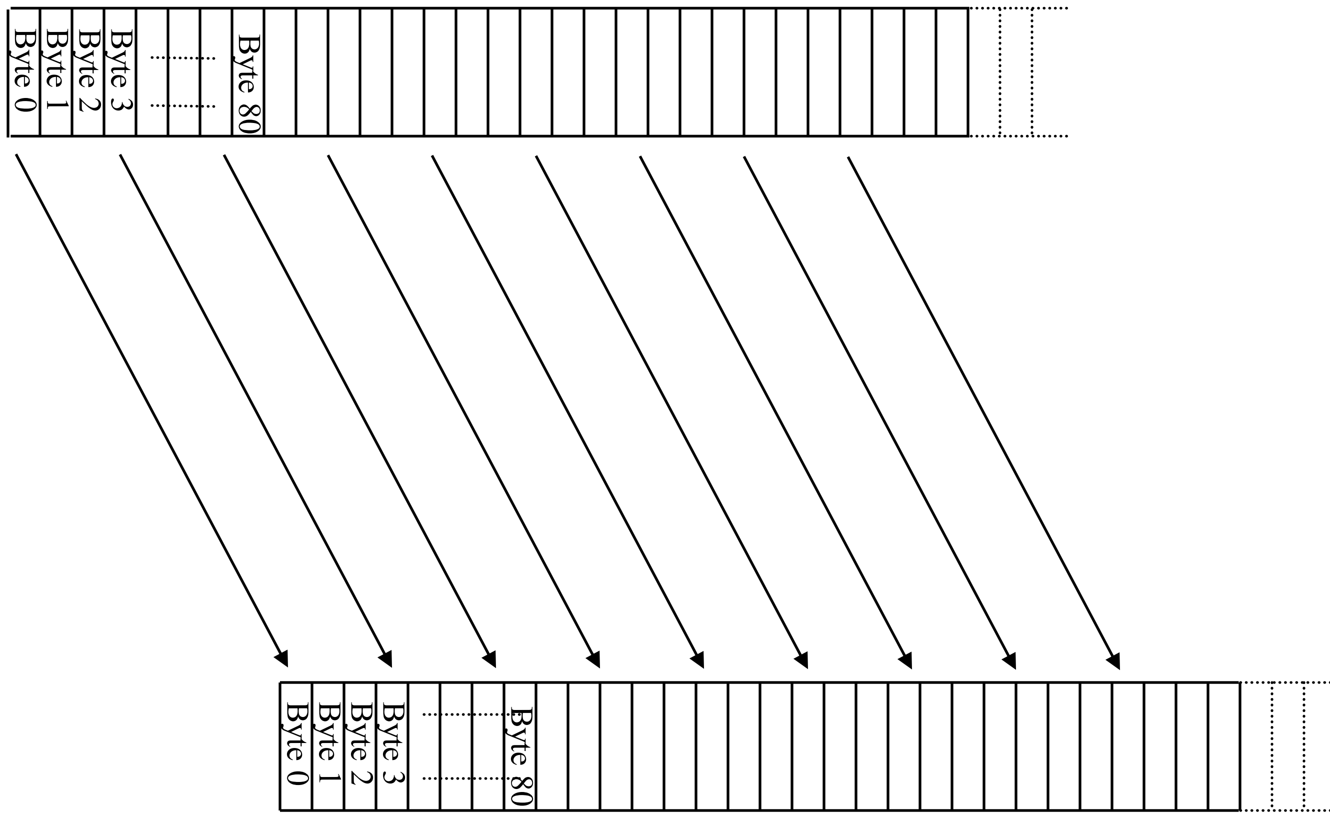
TCP Support for Reliable Delivery

- 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

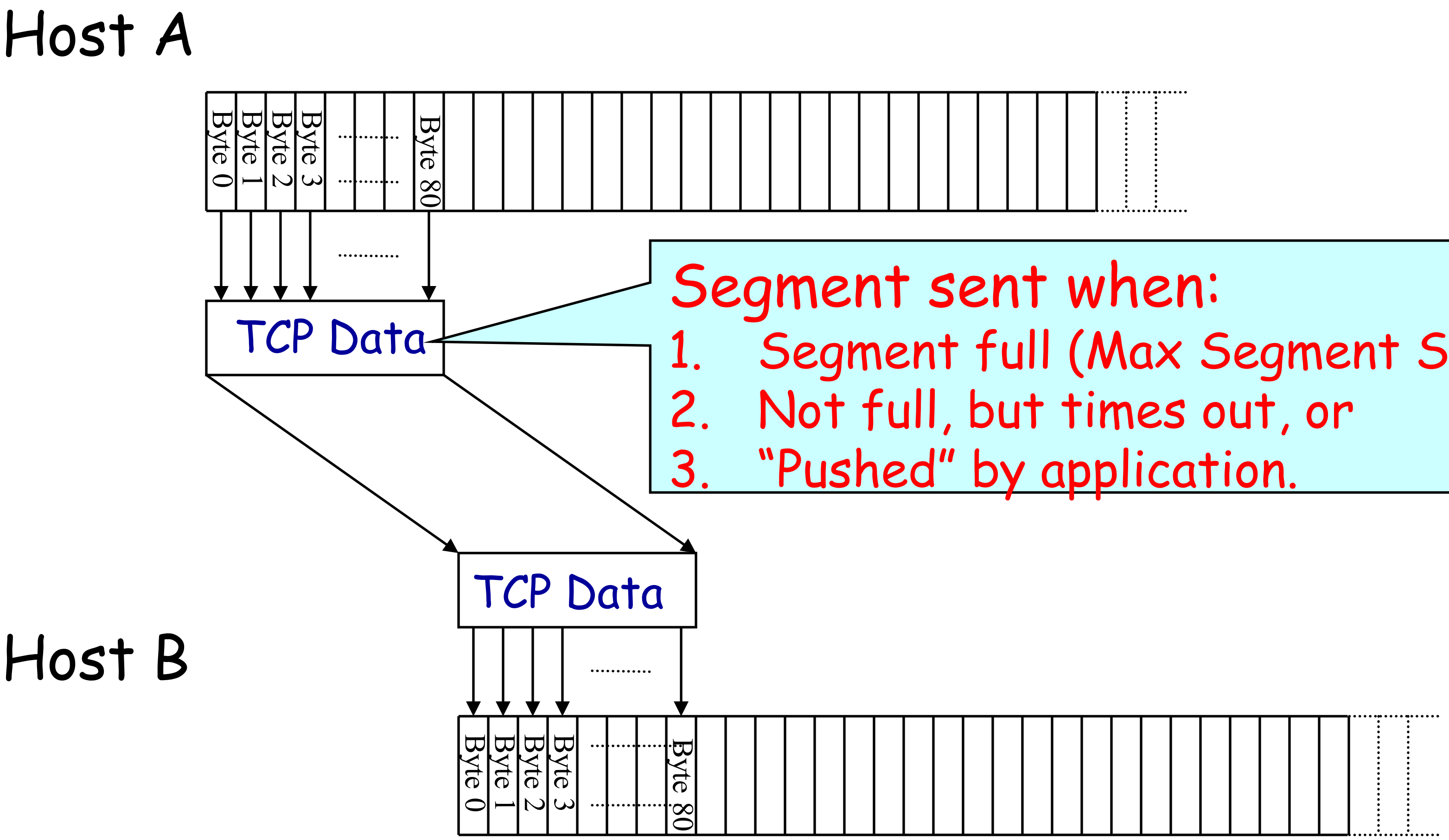
TCP "Stream of Bytes" Service

Host A



Host B

…Emulated Using TCP "Segments"



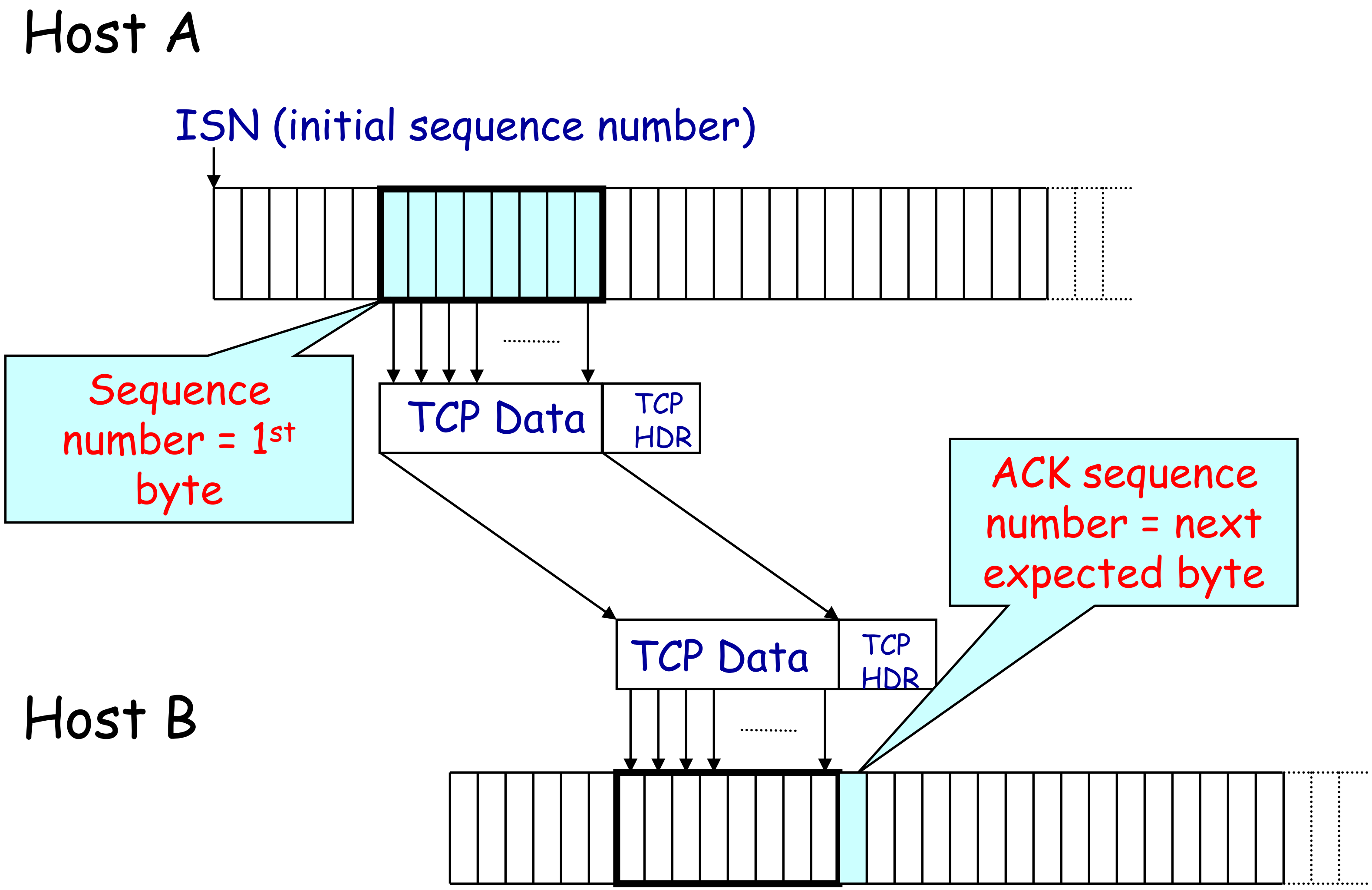
Segment sent when:

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

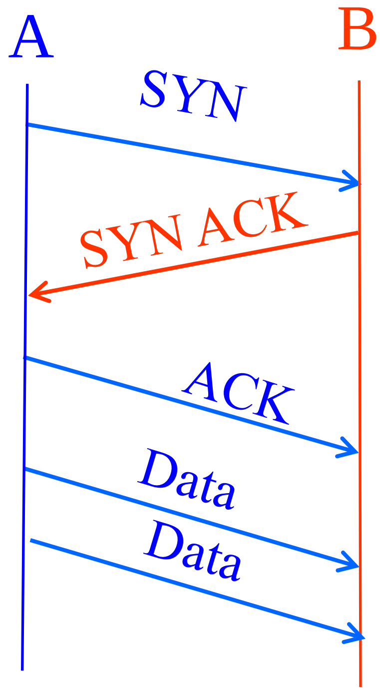


Initial Sequence Number (ISN)

- 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

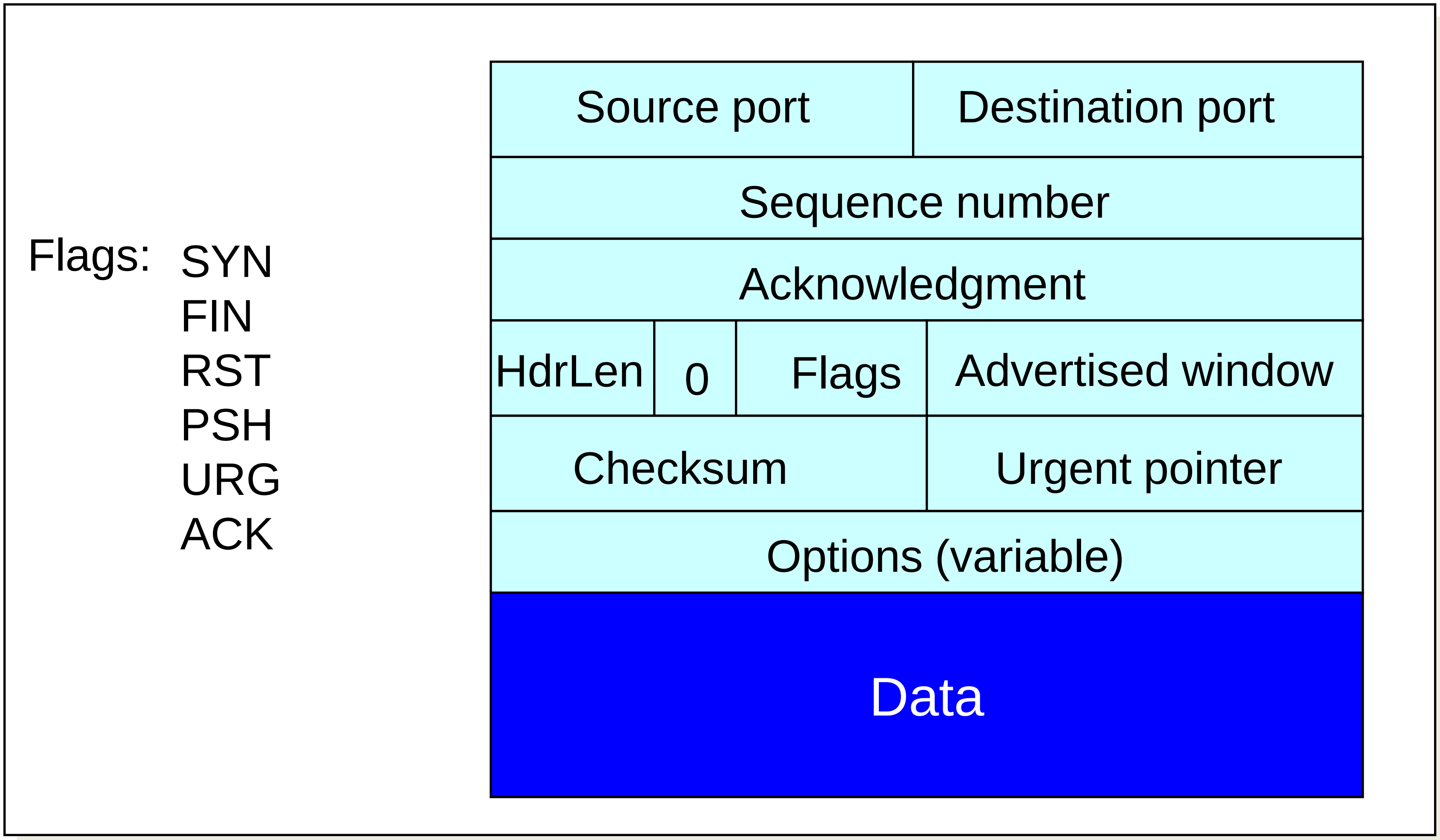
Establishing a TCP Connection



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
Source port
Destination port
Sequence number
Acknowledgment
HdrLen
0
Flags
Advertised window
Checksum
Urgent pointer
Options (variable)
Data

Step 1: A's Initial SYN Packet

Flags: SYN
FIN
RST
PSH
URG
ACK

<table>
<tr><td colspan="3">A's port</td><td>B's port</td></tr>
<tr><td colspan="4">A's Initial Sequence Number</td></tr>
<tr><td colspan="4">Acknowledgment</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>

A tells B it wants to open a connection…

Step 2: B's SYN-ACK Packet

Flags: SYN
FIN
RST
PSH
URG
ACK

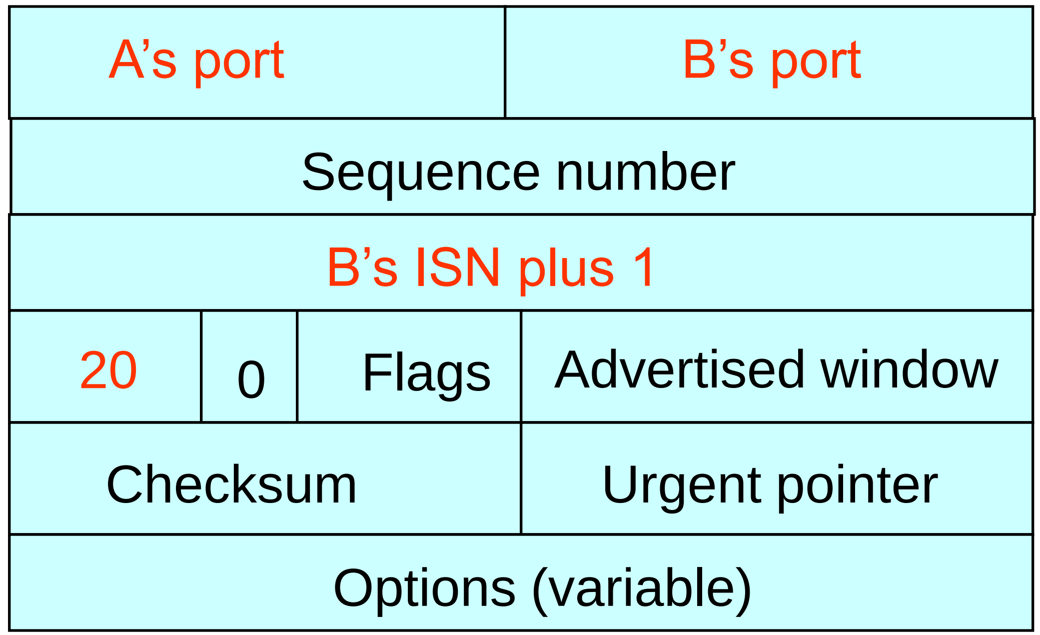
B's port | A's port
B's Initial Sequence Number
A's ISN plus 1
20 | 0 | Flags | Advertised window
Checksum | Urgent pointer
Options (variable)

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 tells B it wants 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-ACK 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

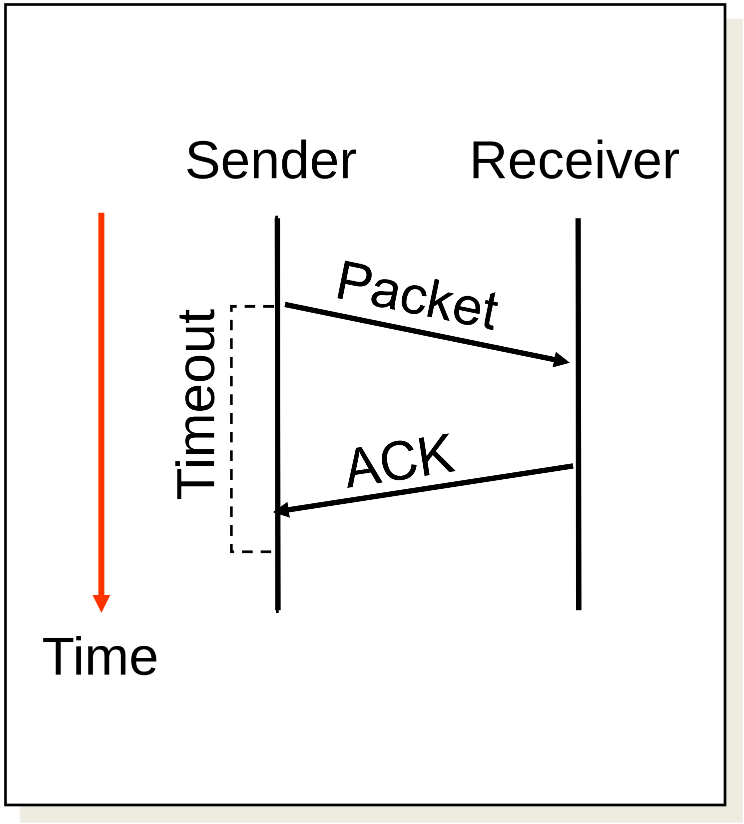
SYN Loss and Web Downloads

- 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

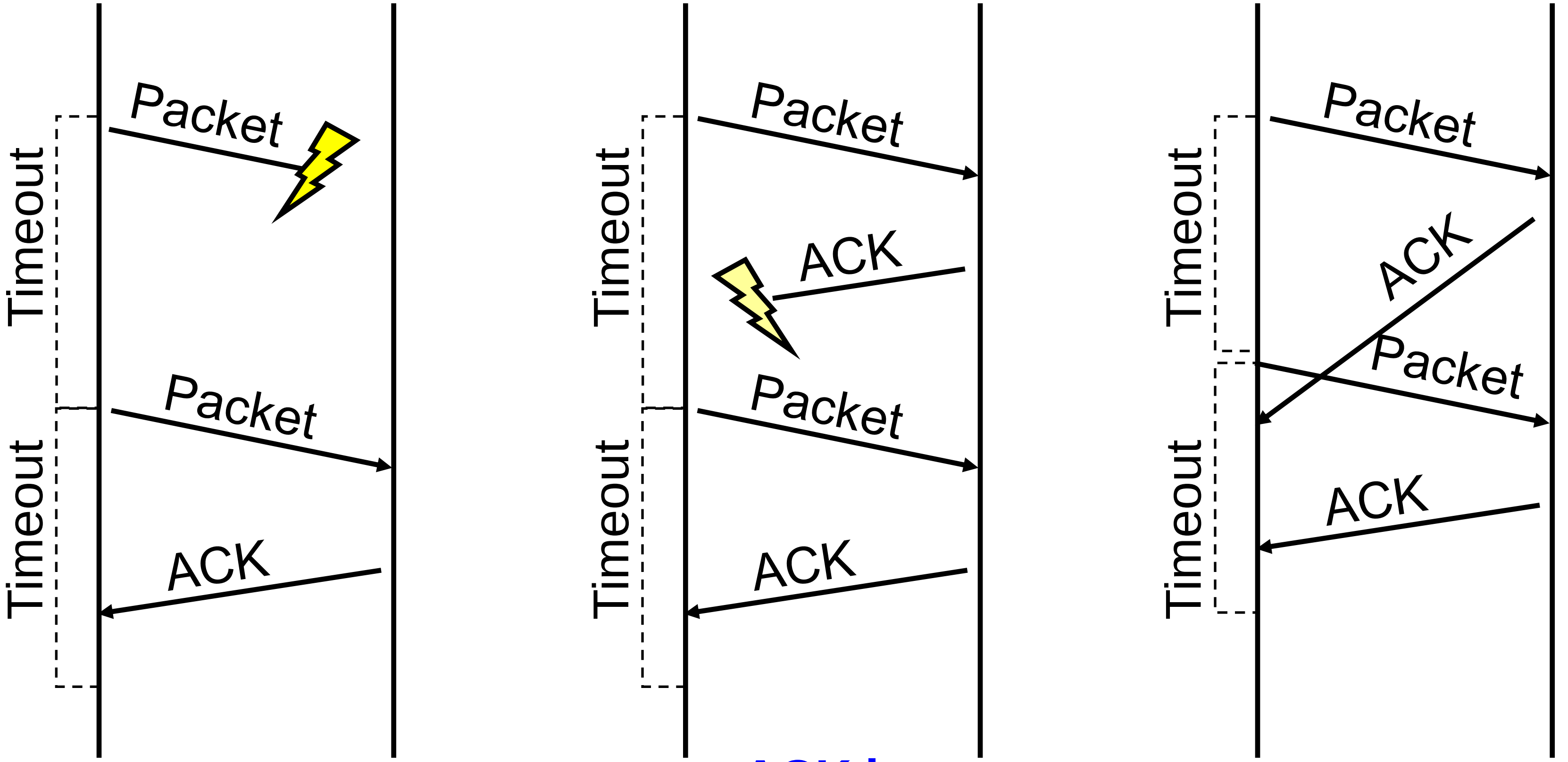
TCP Retransmissions

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

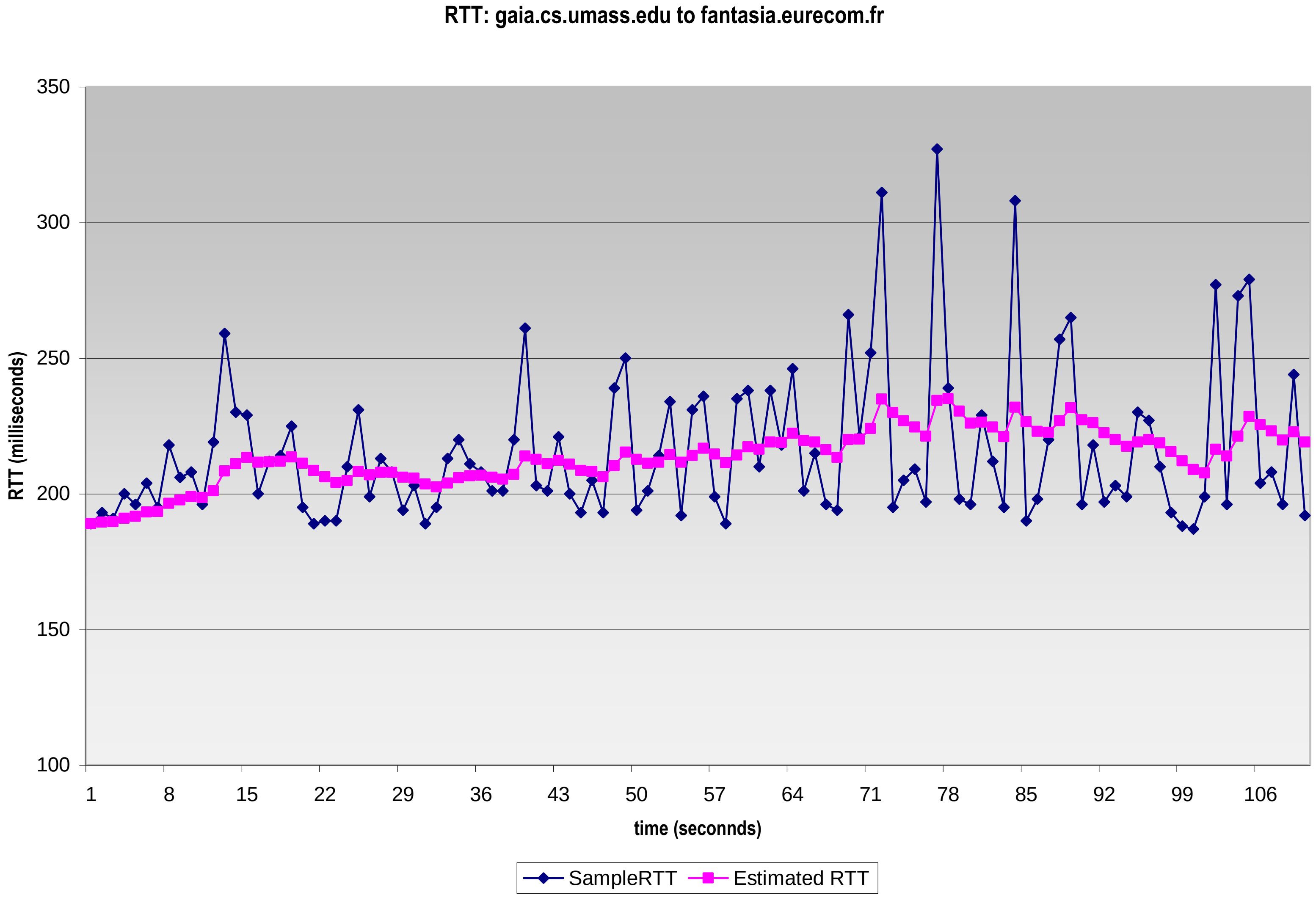
ACK lost
DUPLICATE PACKET

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
    - 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

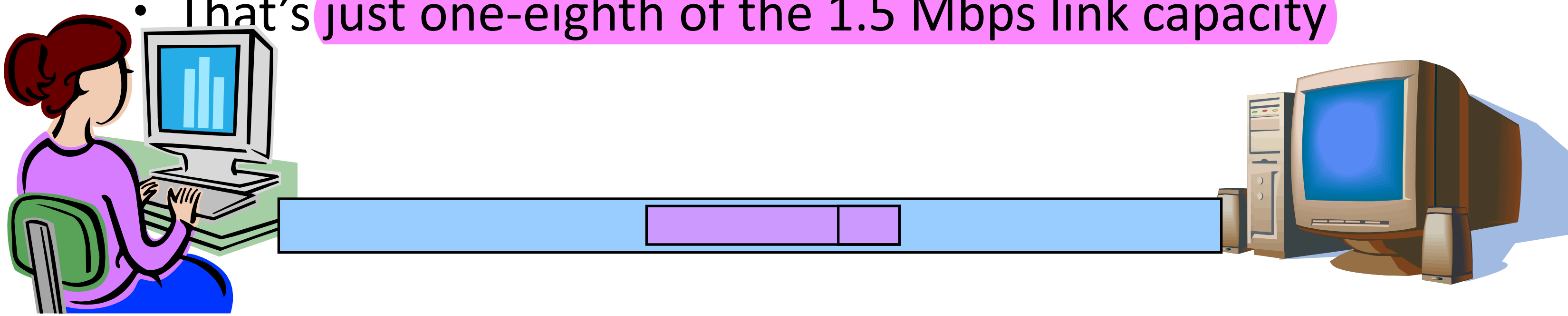
Yet Another Limitation…

- Doesn't consider variance in the RTT
  - If variance is small, the EstimatedRTT is pretty accurate
  - … but, if variance is large, the estimate isn't all that good
- Better to directly consider the variance
  - Consider difference: SampleRTT – EstimatedRTT
  - Boost the estimate based on the difference

TCP Sliding Window

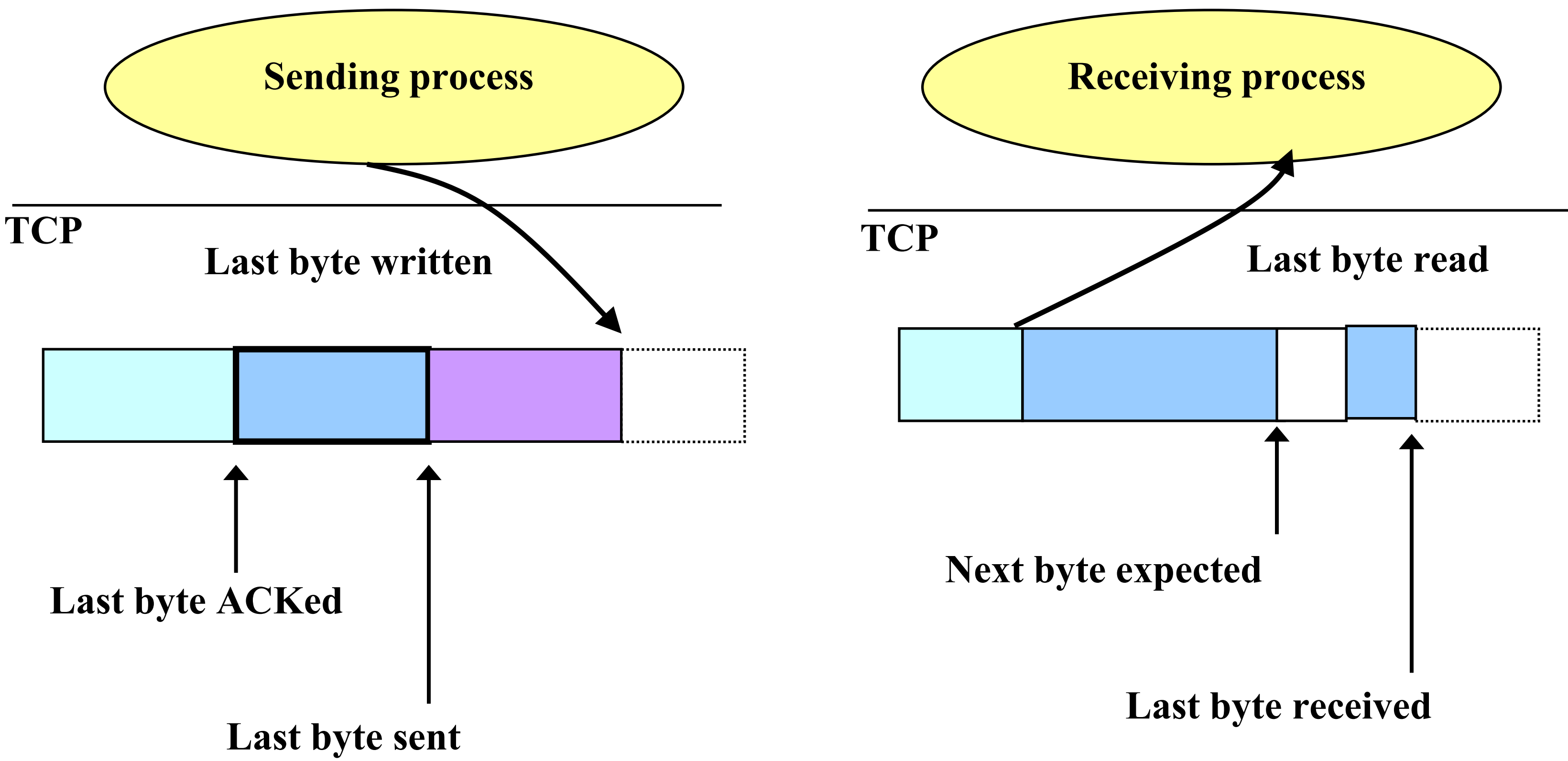
Motivation for Sliding Window

- 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



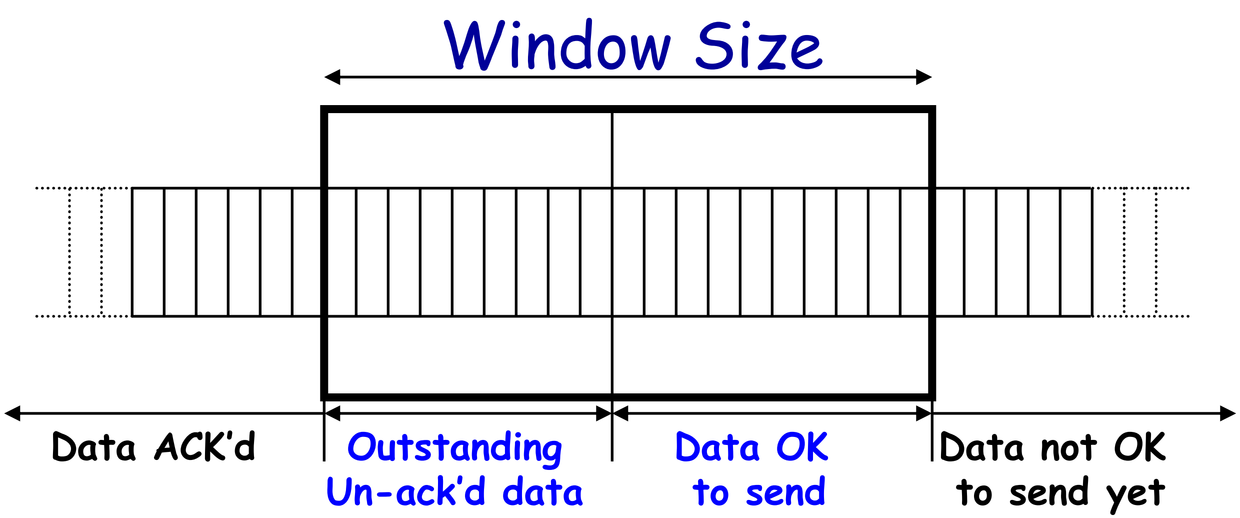
Sliding Window

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

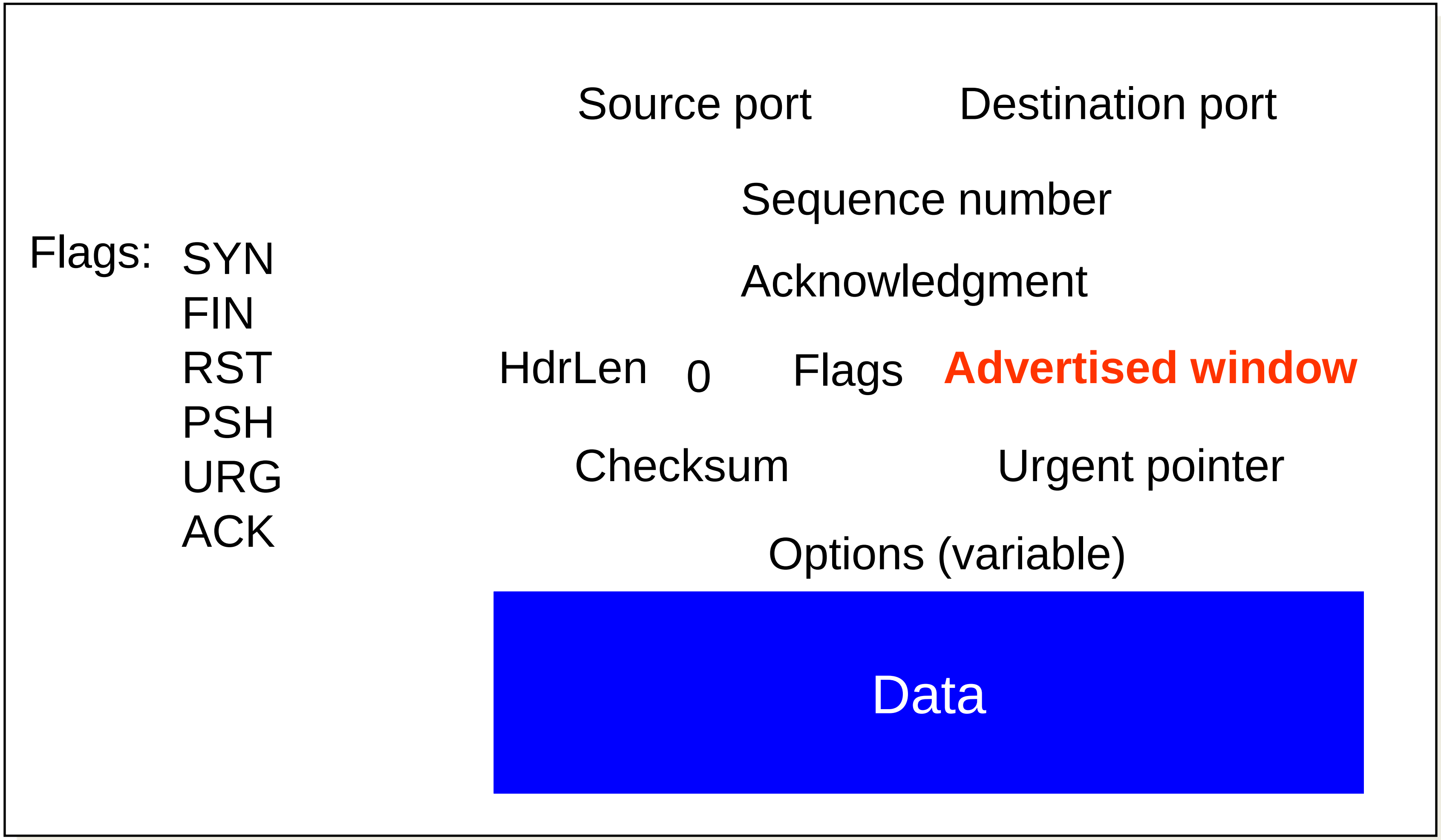


Receiver Buffering

- 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 receiver
  - Tells the receiver the amount of free space left
  - … and the sender agrees not to exceed this amount



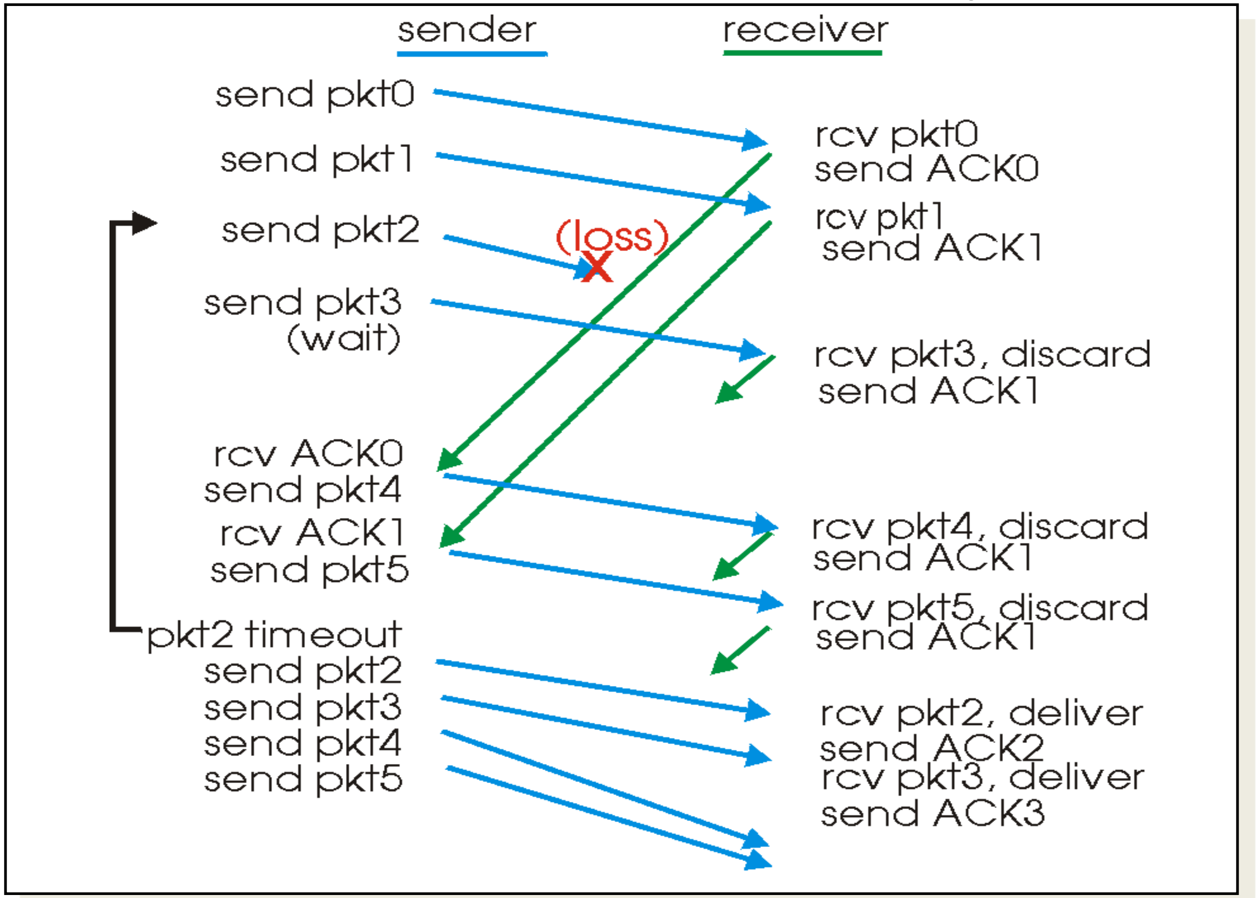
TCP Header for Receiver Buffering



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

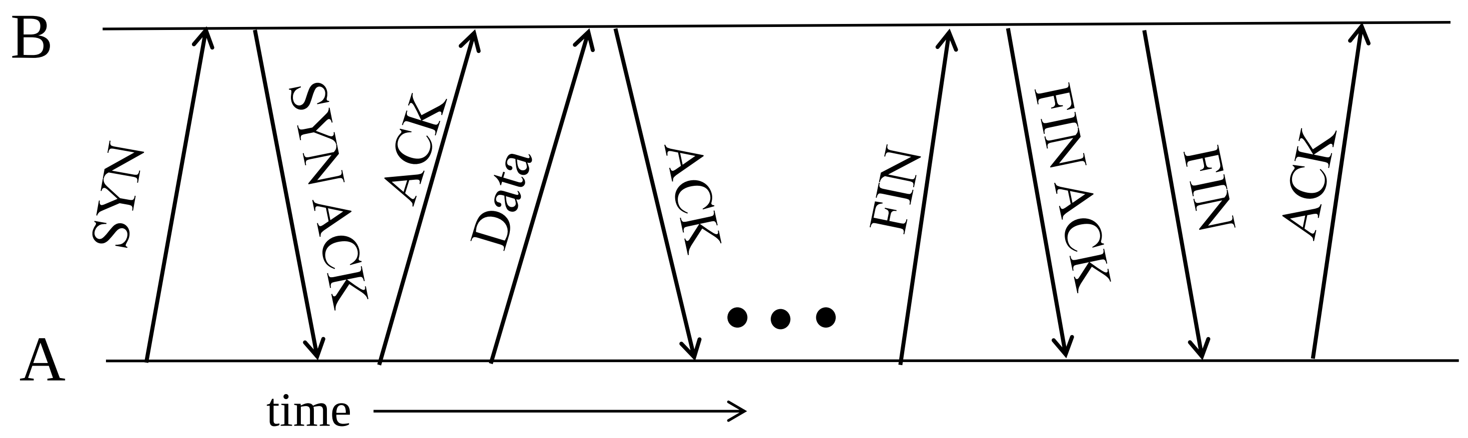
- Better solution possible under sliding window
  - Although packet n might have been lost
  - … packets n+1, n+2, and so on might get through
- Idea: have the receiver send ACK packets
  - ACK says that receiver is still awaiting n^{th} packet
    - And *repeated* ACKs suggest later packets have arrived
  - 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

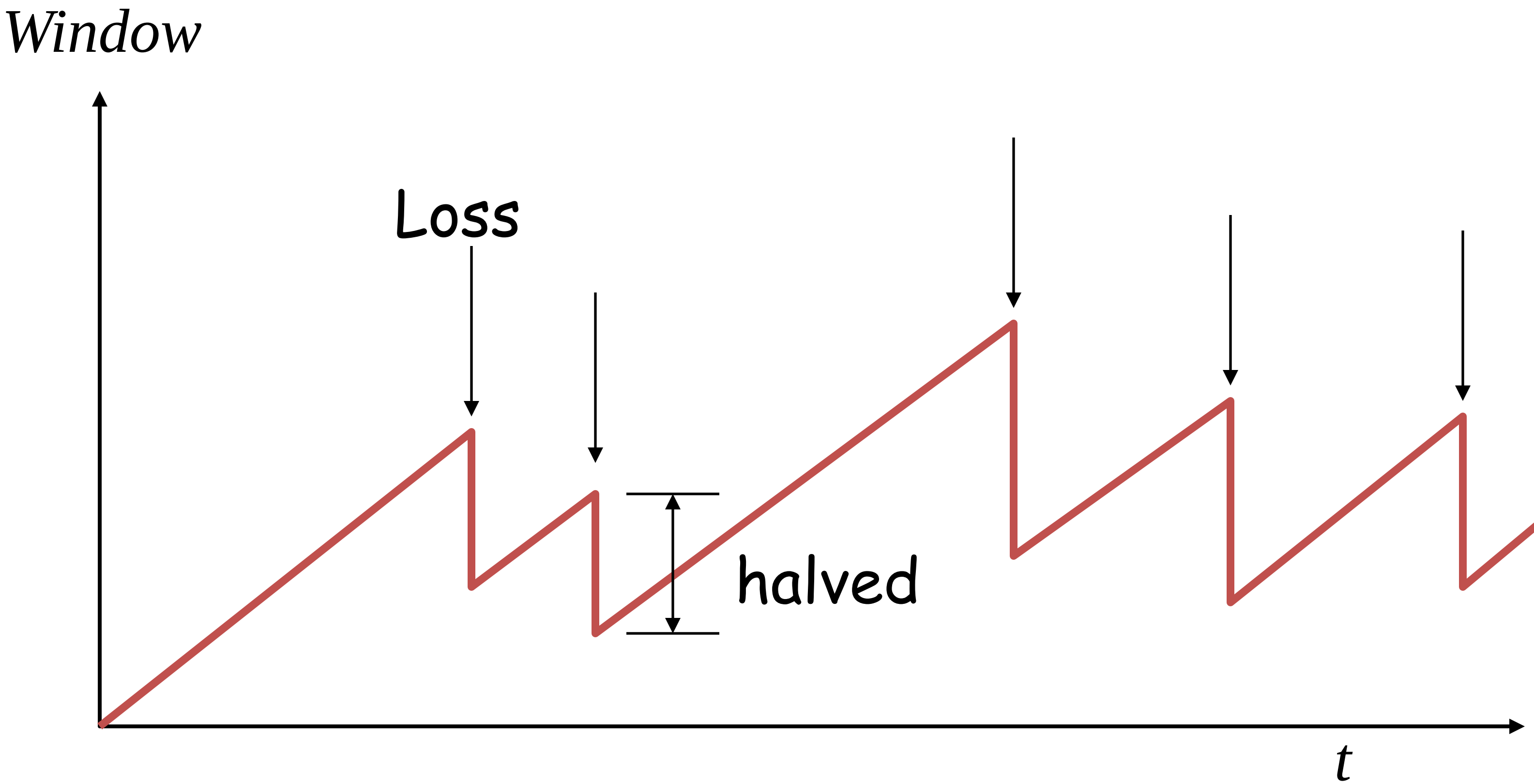
Idea of TCP Congestion Control

- Each source determines the available capacity
  - … so it knows how many packets to have in transit
- Congestion window
  - Maximum # of unacknowledged bytes to have in transit
  - The congestion-control equivalent of receiver window
  - MaxWindow = min{congestion window, receiver window}
  - Send at the rate of the slowest component
- Adapting the congestion window
  - Decrease upon losing a packet: backing off
  - Increase upon success: optimistically exploring

Additive Increase, Multiplicative Decrease

- How much to increase and decrease?
  - Increase linearly, decrease multiplicatively
  - A necessary condition for stability of TCP
  - Consequences of over-sized window are much worse than having an under-sized window
    - Over-sized window: packets dropped and retransmitted
    - Under-sized window: somewhat lower throughput
- Multiplicative decrease
  - On loss of packet, divide congestion window in half
- Additive increase
  - On success for last window of data, increase linearly

Leads to the TCP "Sawtooth"



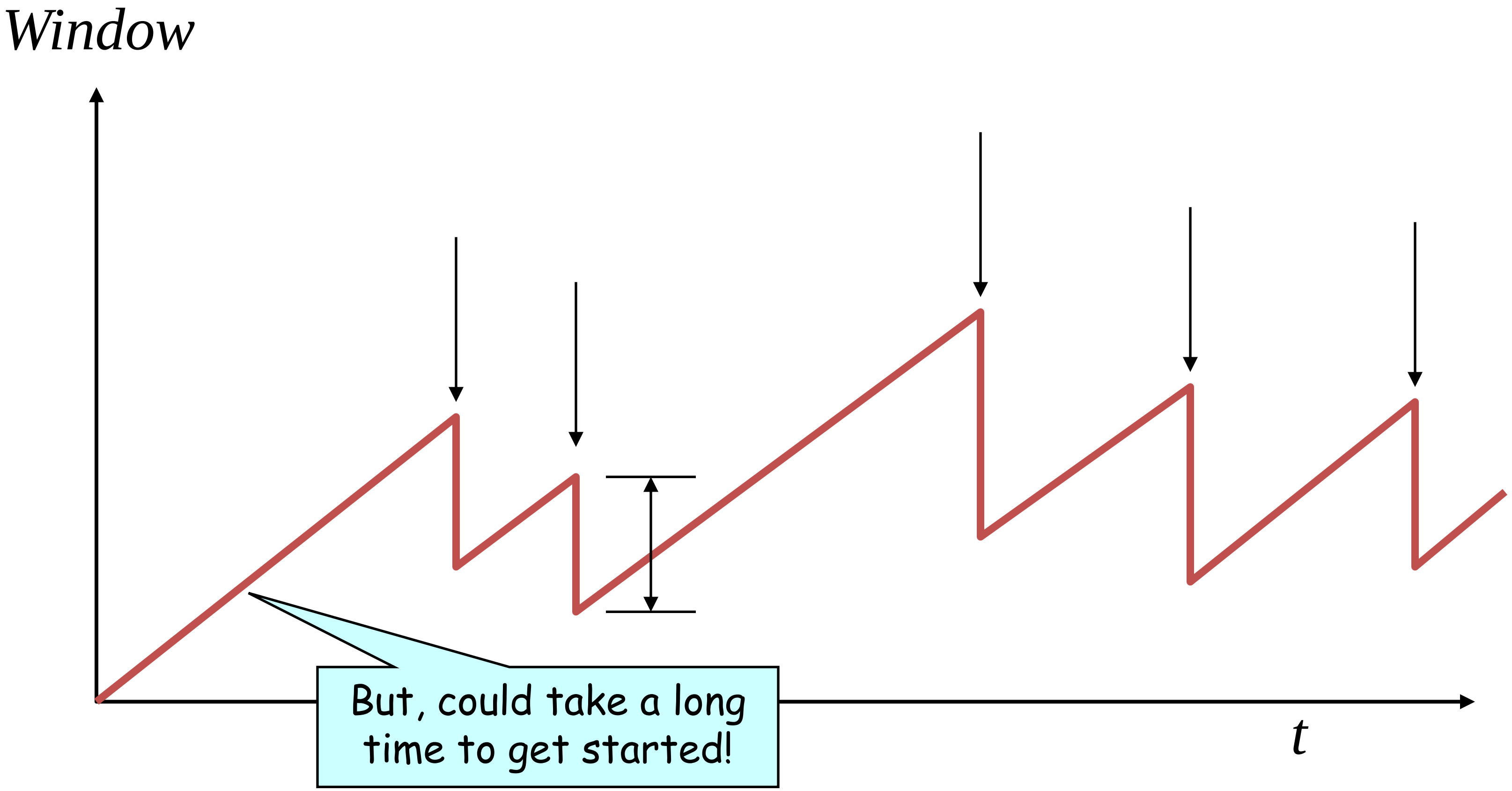
Practical Details

- Congestion window
  - Represented in bytes, not in packets (Why?)
  - Packets have MSS (Maximum Segment Size) bytes
- Increasing the congestion window
  - Increase by MSS on success for last window of data
  - In practice, increase a fraction of MSS per received ACK
    - # packets per window: CWND / MSS
    - Increment per ACK: MSS * (MSS / CWND)

- Decreasing the congestion window

Getting Started

Need to start with a small CWND to avoid overloading the network.

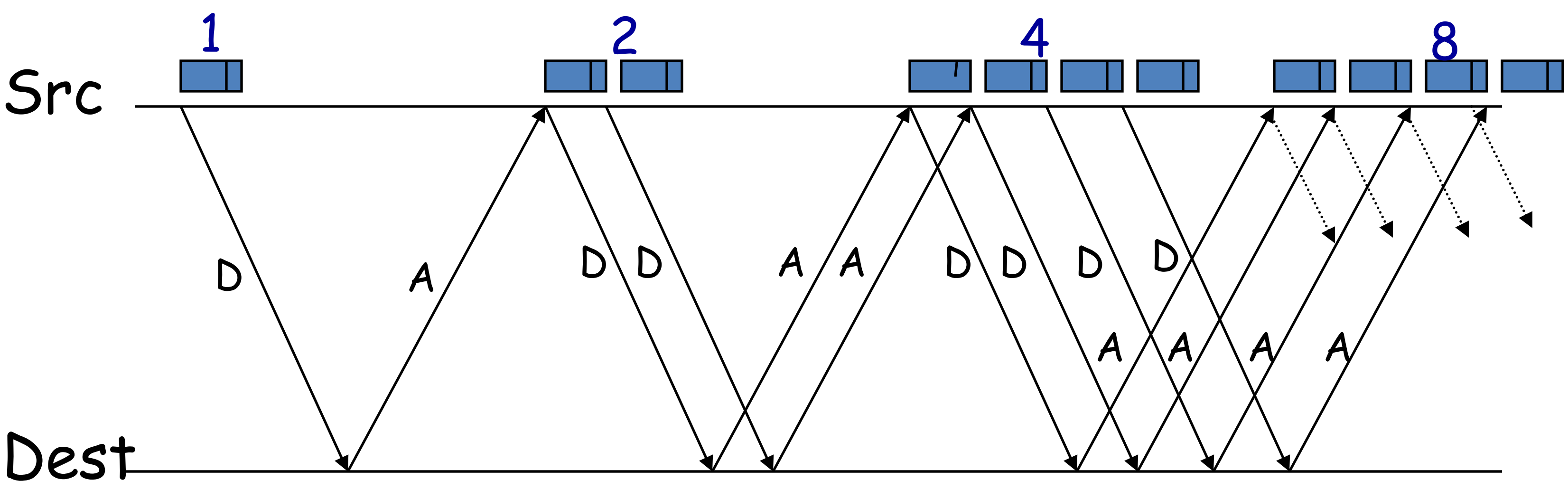


“Slow Start” Phase

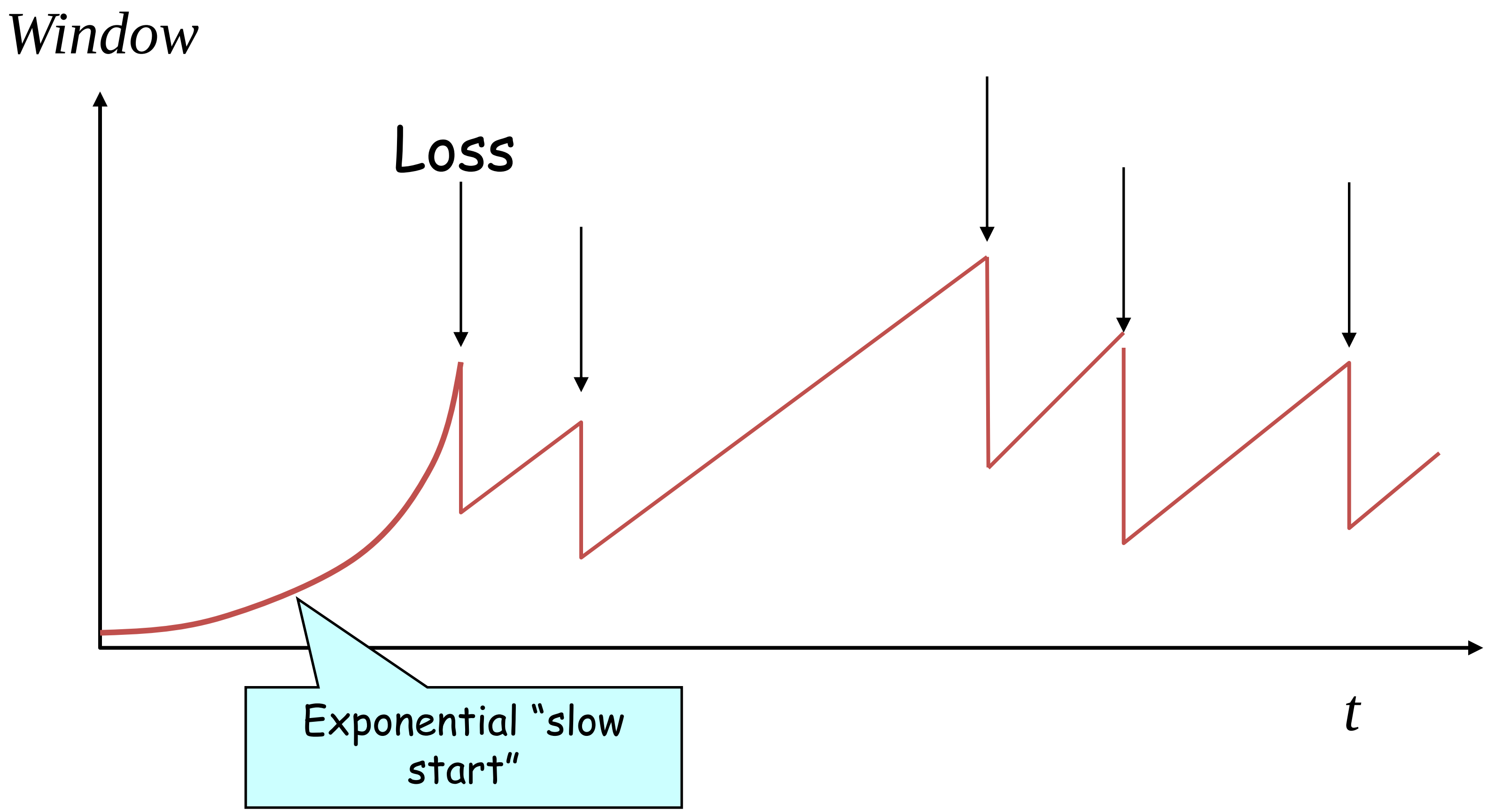
- Start with a small congestion window
  - Initially, CWND is 1 MSS
  - So, initial sending rate is MSS/RTT
- That could be pretty wasteful
  - Might be much less than the actual bandwidth
  - Linear increase takes a long time to accelerate
- Slow-start phase (really “fast start”)
  - Sender starts at a slow rate (hence the name)
  - … but increases the rate exponentially
  - … until the first loss event

Slow Start in Action

Double CWND per round-trip time



Slow Start and the TCP Sawtooth

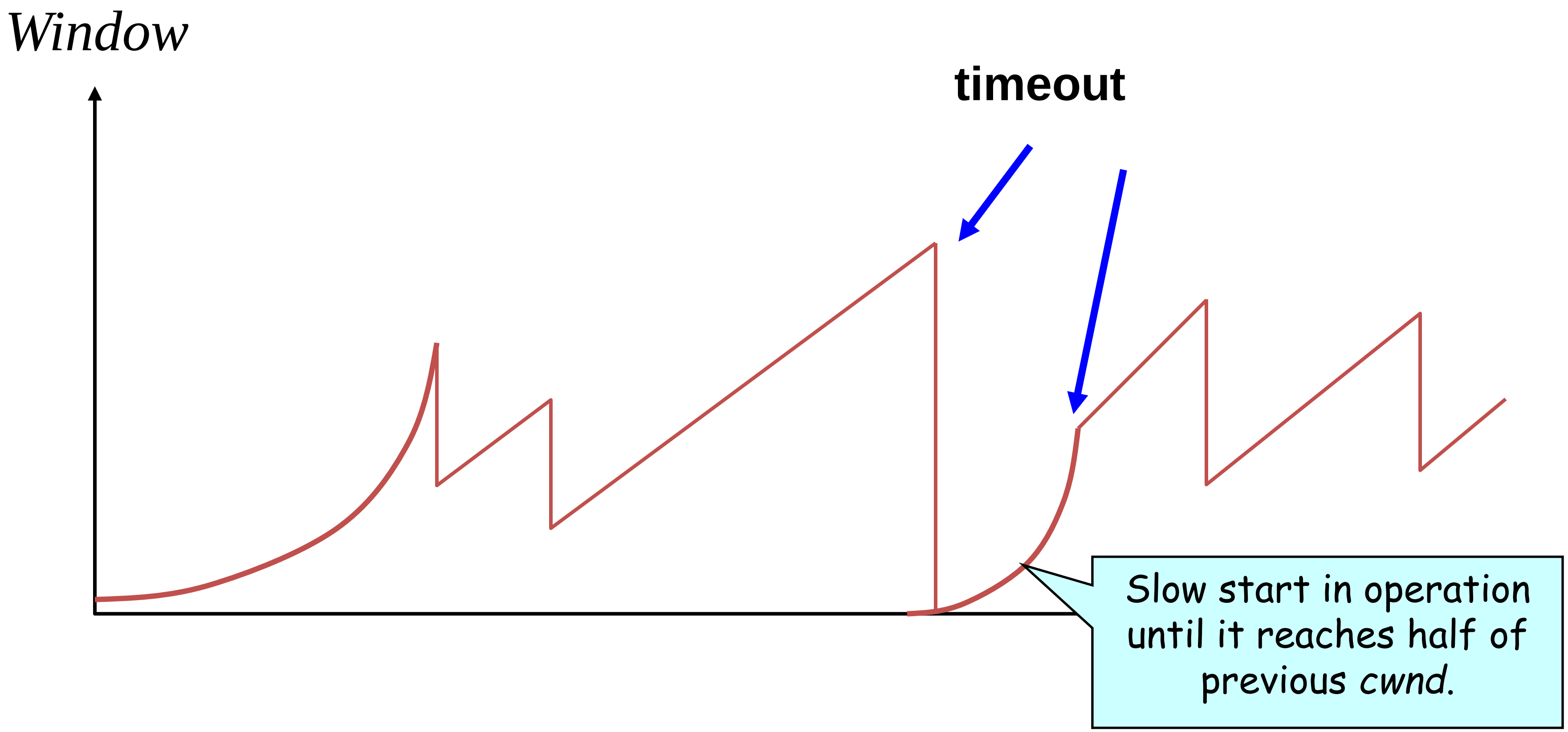


Why is it called slow-start? Because TCP originally had no congestion control mechanism. The source would just start by sending a whole window's worth of data.

Two Kinds of Loss in TCP

- Triple duplicate ACK
  - Packet n is lost, but packets n+1, n+2, etc. arrive
  - Receiver sends duplicate acknowledgments
  - … and the sender retransmits packet n quickly
  - Do a multiplicative decrease and keep going
- Timeout
  - Packet n is lost and detected via a timeout
  - E.g., because all packets in flight were lost
  - After the timeout, blasting away for the entire CWND

Repeating Slow Start After Timeout



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

Repeating Slow Start After Idle Period

- Suppose a TCP connection goes idle for a while
  - E.g., Telnet session where you don't type for an hour
- Eventually, the network conditions change
  - Maybe many more flows are traversing the link
  - E.g., maybe everybody has come back from lunch!
- Dangerous to start transmitting at the old rate
  - Previously-idle TCP sender might blast the network
  - … causing excessive congestion and packet loss
- So, some TCP implementations repeat slow start
  - Slow-start restart after an idle period