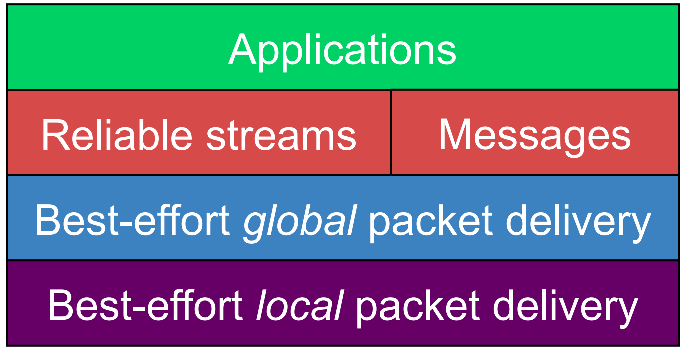


Transport Layer

Kyle Jamieson

COS 461: Computer Networks

www.cs.princeton.edu/courses/archive/fall21/cos461

IP Protocol Stack: Key Abstractions

- Transport layer is where we:
  - Provide applications with good abstractions
    - Without support or feedback from the network

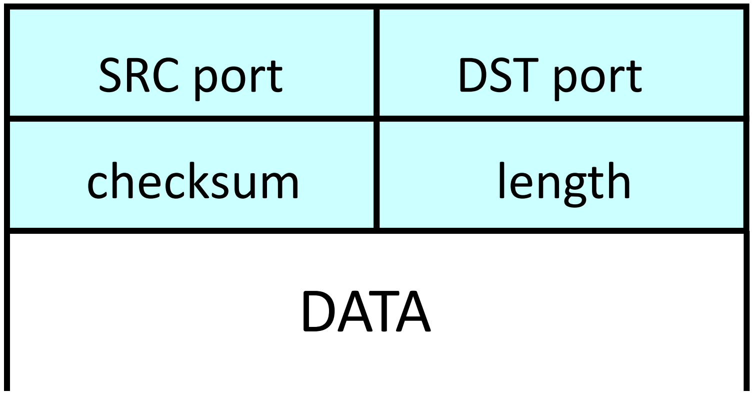
Transport Protocols

- Logical communication between processes
  - Sender divides a message into segments
  - Receiver reassembles segments into message
- Transport services
  - (De)multiplexing packets
  - Detecting corrupted data
  - Optionally: reliable delivery, flow control, …

User Datagram Protocol (UDP)

- Lightweight communication between processes
  - Send and receive messages
  - Avoid overhead of ordered, reliable delivery
    - No connection setup delay, no in-kernel connection state
- Used by popular apps
  - Query/response for DNS
  - Real-time data in VoIP

8 byte header

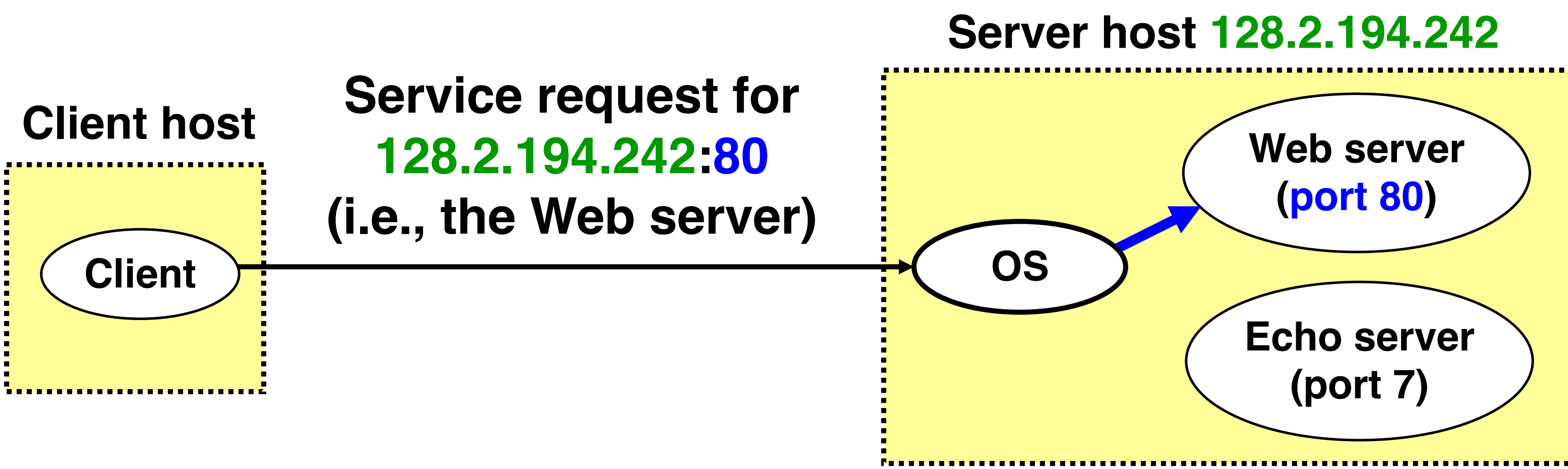


Advantages of UDP

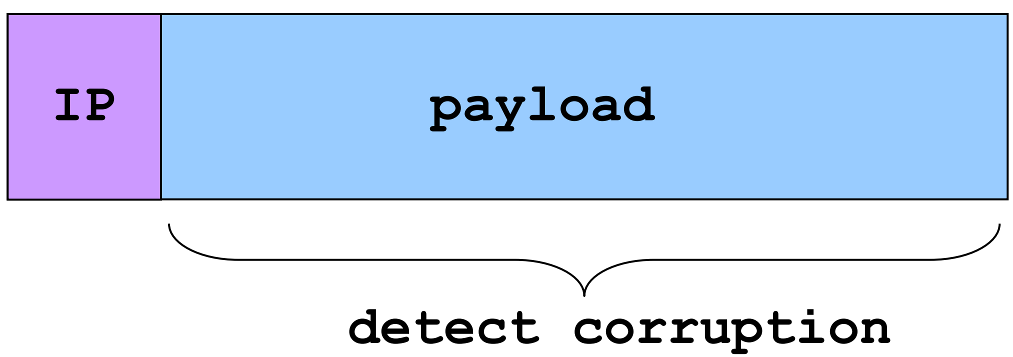
- Fine-grain control
  - UDP sends as soon as the application writes
- No connection set-up delay
  - UDP sends without establishing a connection
- No connection state in host OS
  - No buffers, parameters, sequence #s, etc.
- Small header overhead
  - UDP header is only eight-bytes long

Two Basic Transport Features

- **Demultiplexing:** port numbers



- **Error detection:** checksums

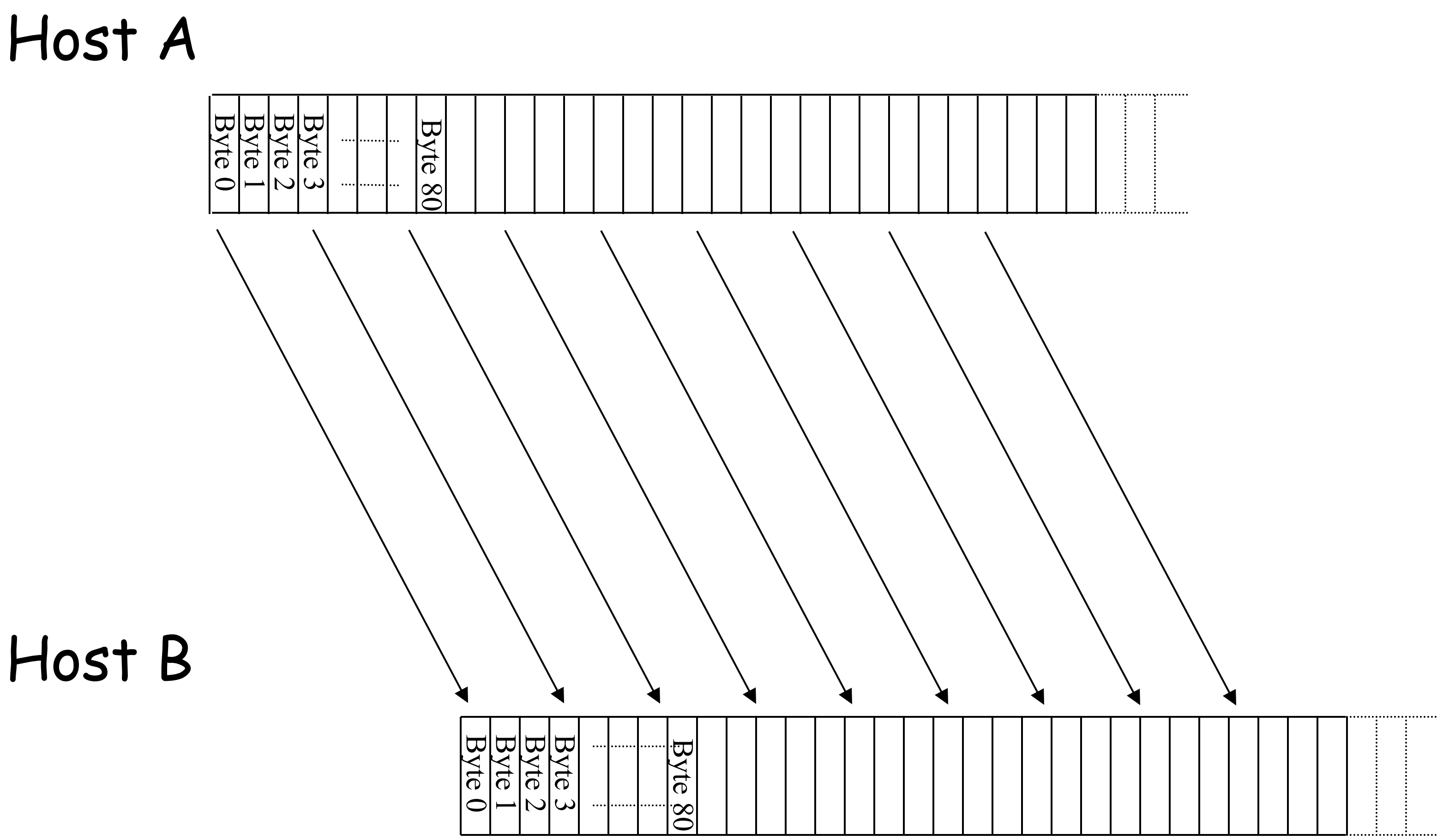


Transmission Control Protocol (TCP)

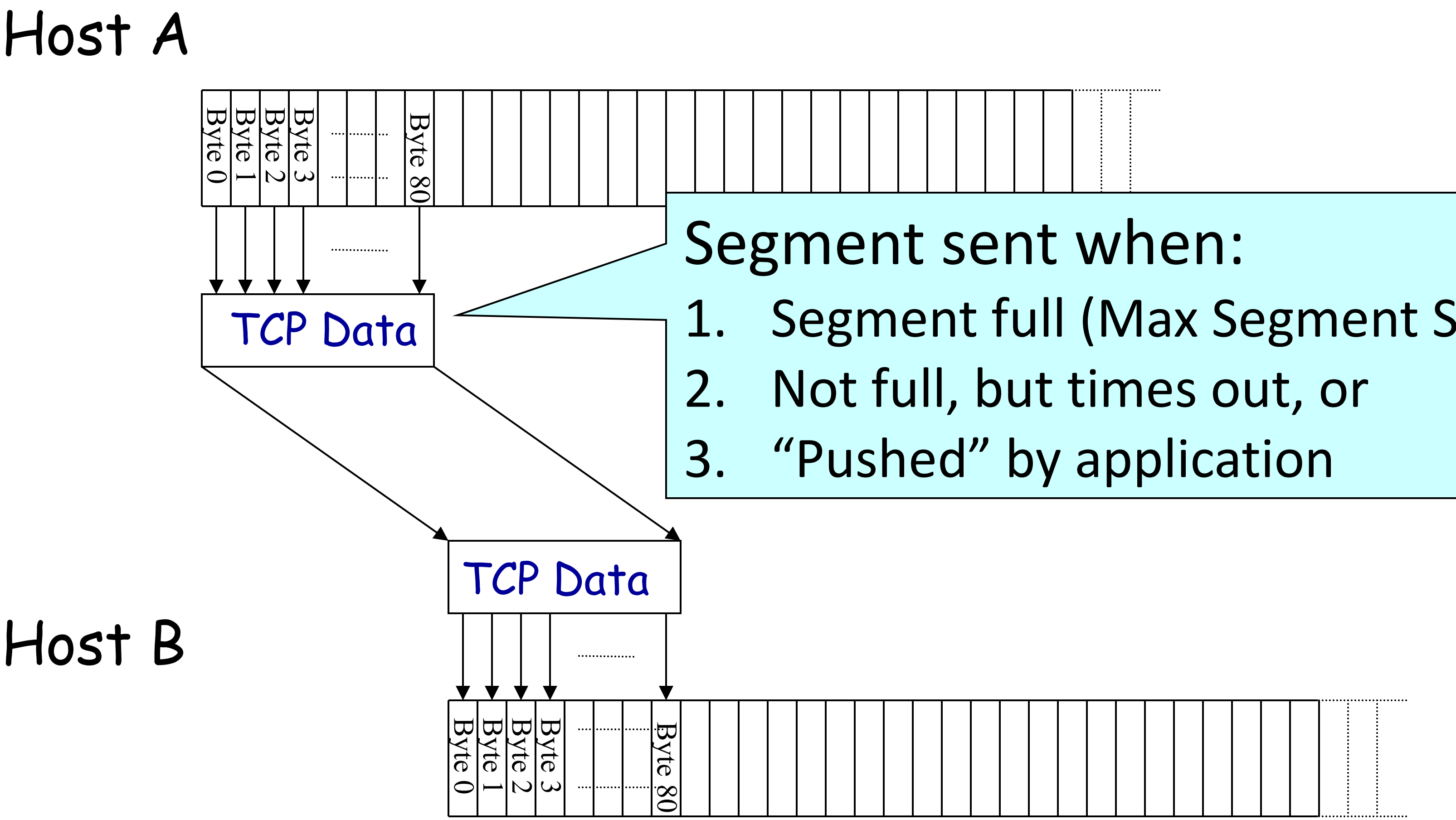
- Stream-of-bytes service
  - Sends and receives a stream of bytes
- Reliable, in-order delivery
  - Corruption: checksums
  - Detect loss/reordering: sequence numbers
  - Reliable delivery: acknowledgments and retransmissions
- Connection oriented
  - Explicit set-up and tear-down of TCP connection
- Flow control
  - Prevent overflow of the receiver's buffer space
- Congestion control
  - Adapt to network congestion for the greater good

Breaking a Stream of Bytes into TCP Segments

TCP "Stream of Bytes" Service



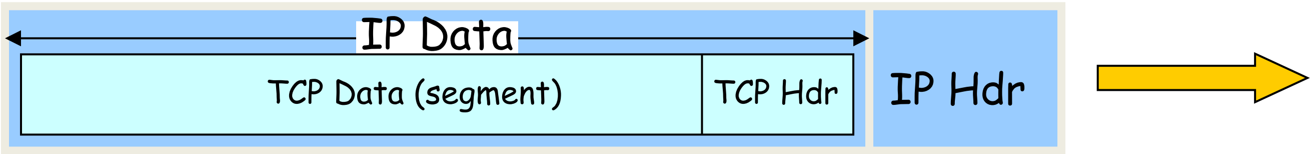
…Emulated Using TCP "Segments"



Segment sent when:

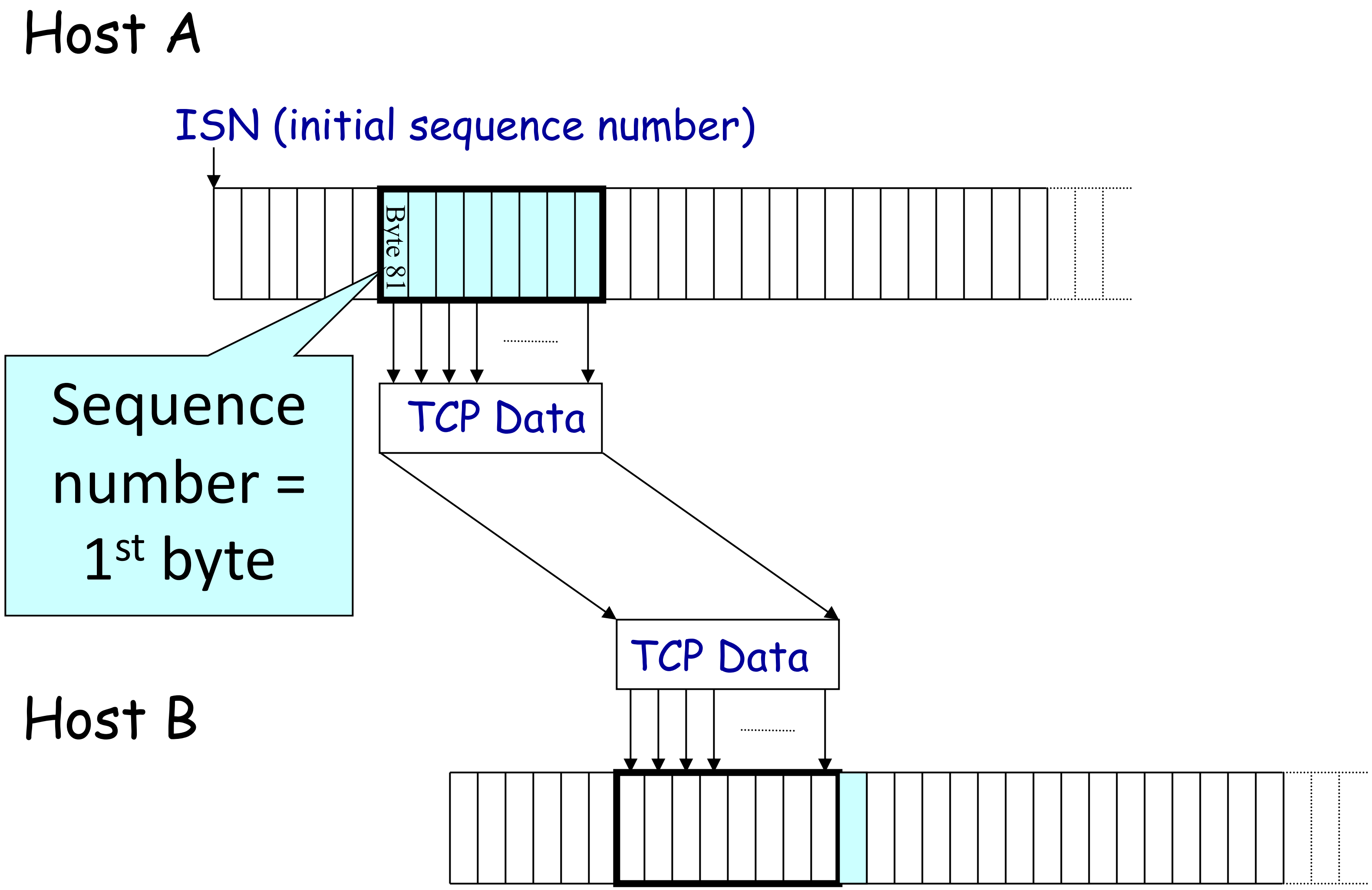
1. Segment full (Max Segment Size),
2. Not full, but times out, or
3. "Pushed" by application

TCP Segment



- IP packet
  - No bigger than Maximum Transmission Unit (MTU)
  - E.g., up to 1500 bytes on an Ethernet link
- 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: MTU (1500) - IP header (20) - TCP header (20)

Sequence Number



Reliable Delivery on a Lossy Channel With Bit Errors

Challenges of Reliable Data Transfer

- Over a perfectly reliable channel: Done
- Over a channel with bit errors
  - Receiver detects errors and requests retransmission
- Over a lossy channel with bit errors
  - Some data missing, others corrupted
  - Receiver cannot easily detect loss
- Over a channel that may reorder packets
  - Receiver cannot easily distinguish loss vs. out-of-order

An Analogy

- 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? Has she 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?"

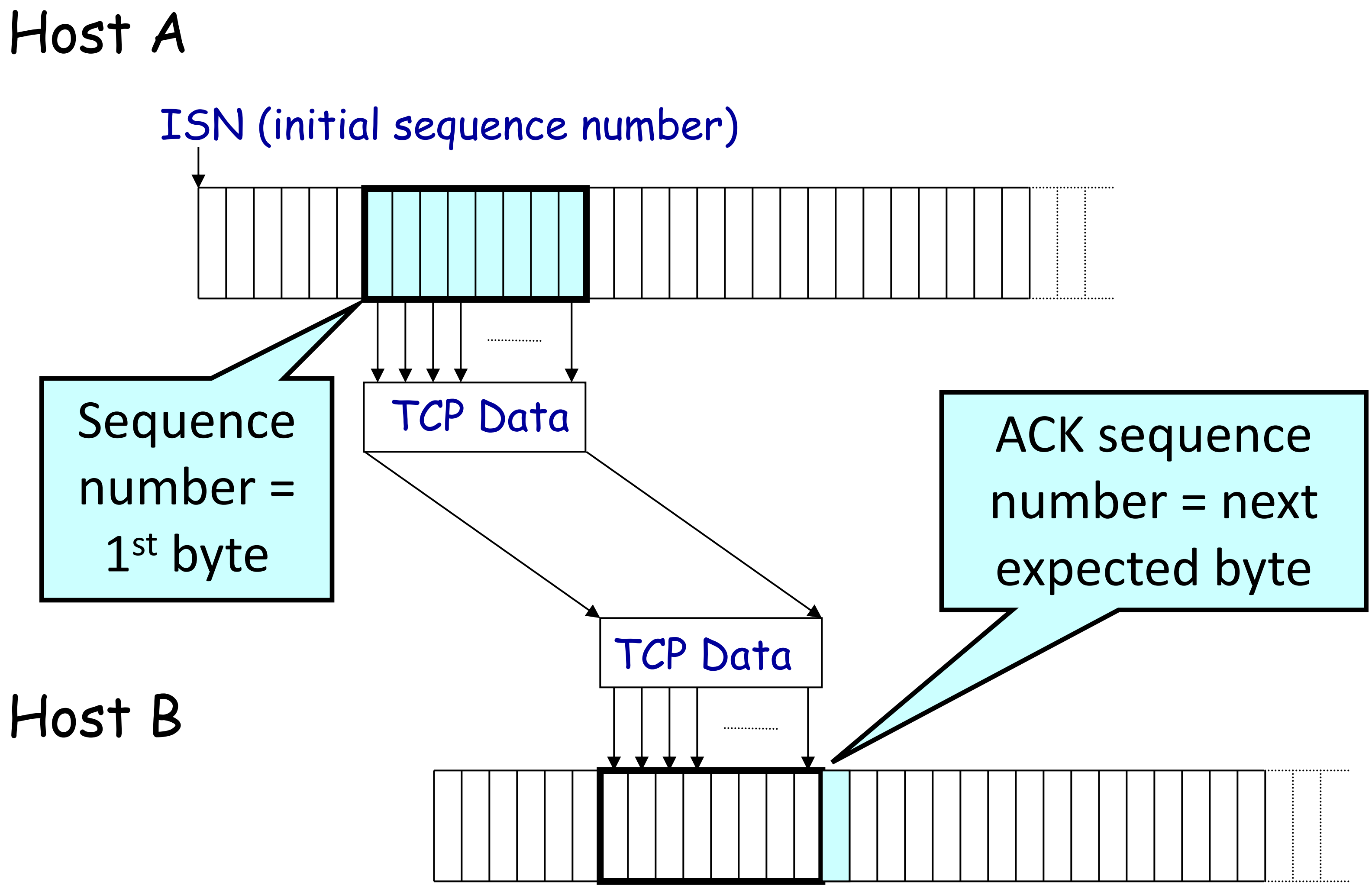
Take-Aways from the Example

- Acknowledgments from receiver
  - Positive: “okay” or “uh huh” or “ACK”
  - Negative: “please repeat that” or “NACK”
- Retransmission by the sender
  - After *not* receiving an “ACK”
  - After receiving a “NACK
  - You can use both (as TCP does implicitly)
- Timeout by the sender (“stop and wait”)
  - Don’t wait forever without some acknowledgment

TCP Support for Reliable Delivery

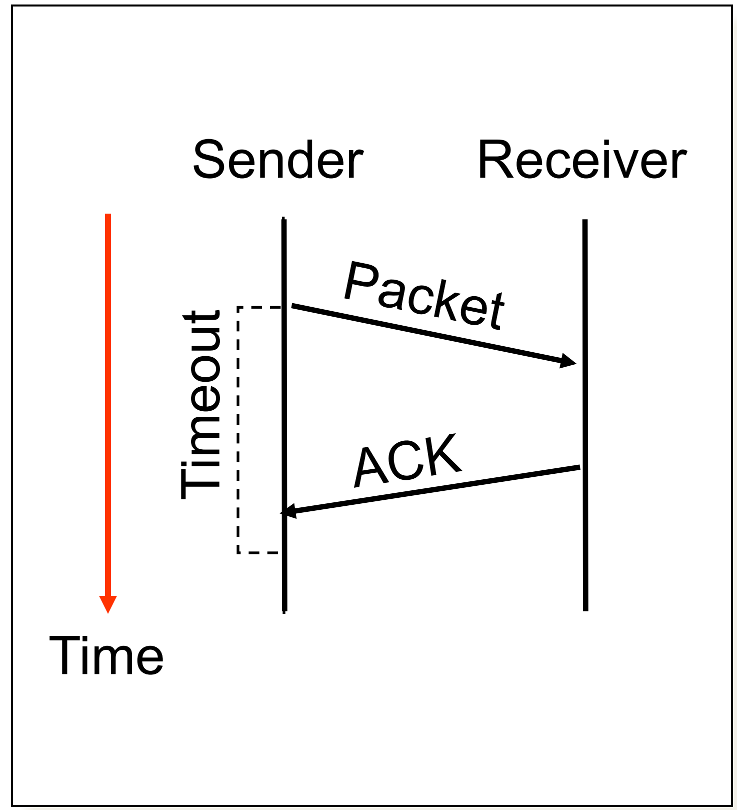
- **Detect bit errors:** checksum
  - Used to detect corrupted data at the receiver
  - …leading the receiver to drop the packet
- **Detect missing data:** sequence number
  - Used to detect a gap in the stream of bytes
  - ... and for putting the data back in order
- **Recover from lost data:** retransmission
  - Sender retransmits lost or corrupted data
  - Two main ways to detect lost packets

TCP Acknowledgments



Automatic Repeat reQuest (ARQ)

- ACK and timeouts
  - Receiver sends ACK when it receives packet
  - Sender waits for ACK and times out
- Simplest ARQ protocol
  - Stop and wait
  - Send a packet, stop and wait until ACK arrives



Initial Sequence Number (ISN)

- Sequence number for the very first byte
  - E.g., Why not a de facto ISN of 0?
- Practical issue: reuse of port numbers
  - Port numbers must (eventually) get used again
  - … and an old packet may still be in flight
  - … and associated with the new connection
- So, TCP must change the ISN over time
  - Set from a 32-bit clock that ticks every 4 microsec
  - … which wraps around once every 4.55 hours!

Quick TCP Math

- Initial Seq No = 501. Sender sends 4500 bytes successfully acknowledged. Next sequence number to send is:

    (Y) 5000   (M) 5001   (C) 5002

- Next 1000 byte TCP segment received. Receiver acknowledges with ACK number:

    (Y) 5001   (M) 6000   (C) 6001

Quick TCP Math

- Initial Seq No = 501. Sender sends 4500 bytes successfully acknowledged. Next sequence number to send is:

  (Y) 5000 $\boxed{\text{(M) 5001}}$ (C) 5002

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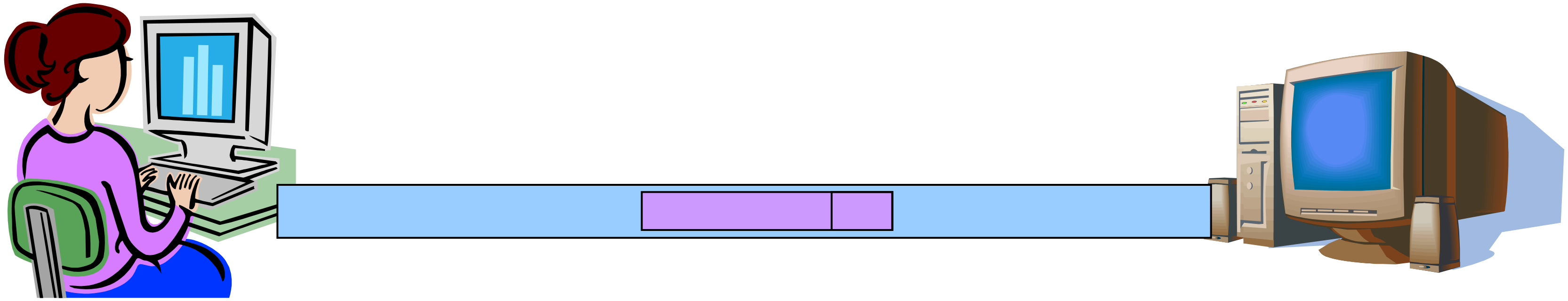
  (Y) 5001 (M) 6000 $\boxed{\text{(C) 6001}}$

Flow Control:
TCP Sliding Window

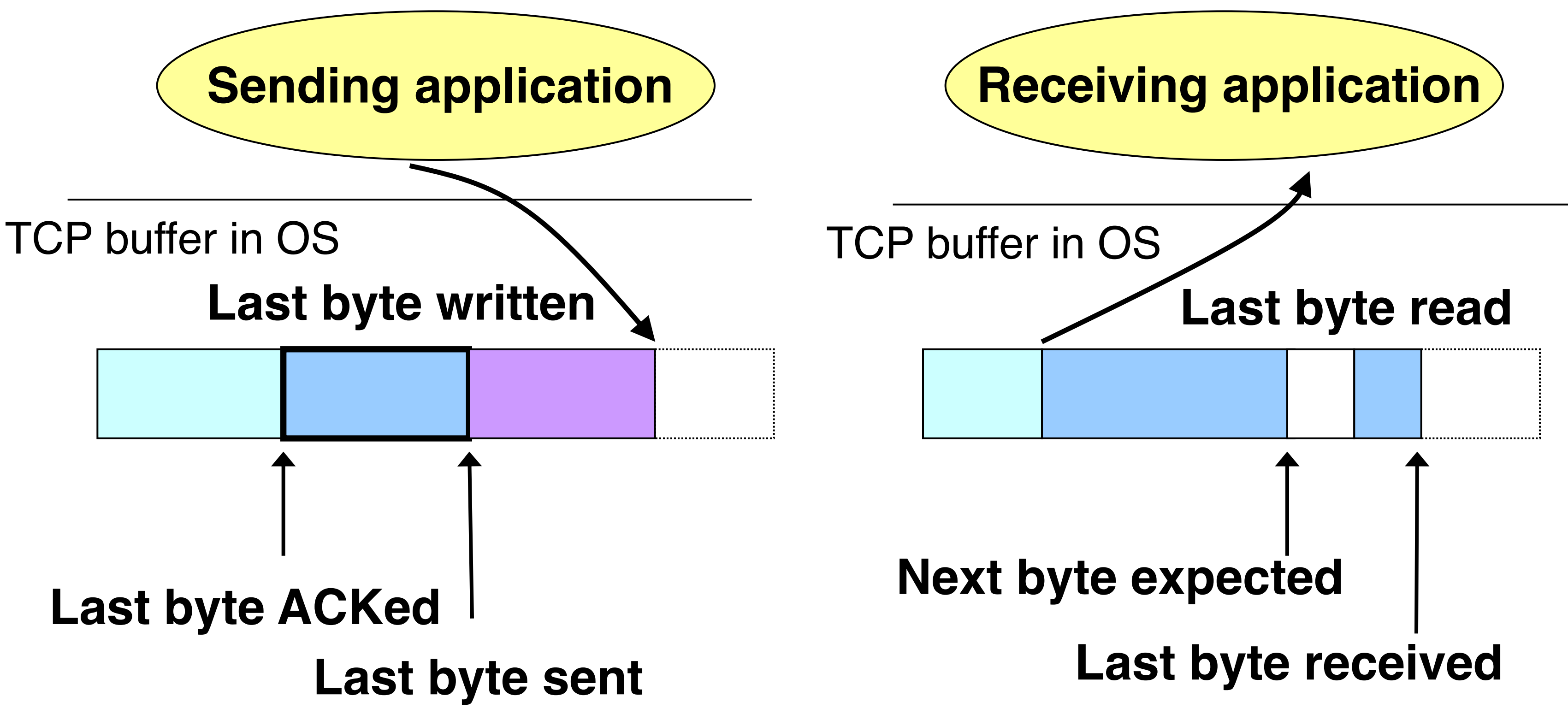
Motivation for Sliding Window

- Stop-and-wait is inefficient
  - Only one TCP segment is “in flight” at a time
- Consider: 1.5 Mbps link with 50 ms round-trip-time (RTT)
  - Assume TCP segment size of 1 KB (8 Kbits)
  - 8 Kbits/segment at 50 msec/segment → 160 Kbps
  - That’s 11% of the capacity of 1.5 Mbps link



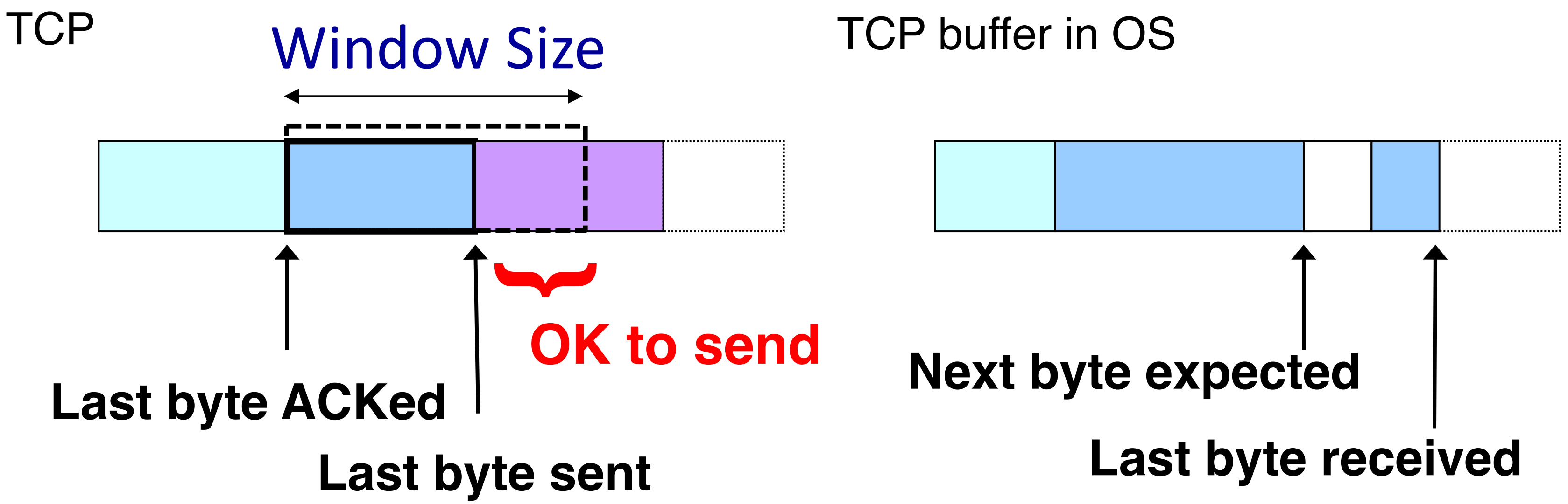
Sliding Window

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



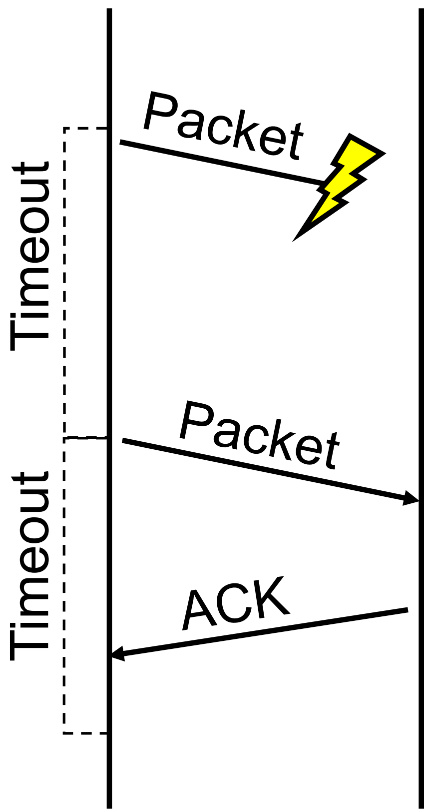
Sliding Window

- Receive window size
  - Amount that can be sent without acknowledgment
  - Receiver must be able to store this amount of data
- Receiver tells the sender the window
  - Tells the sender the amount of free space left

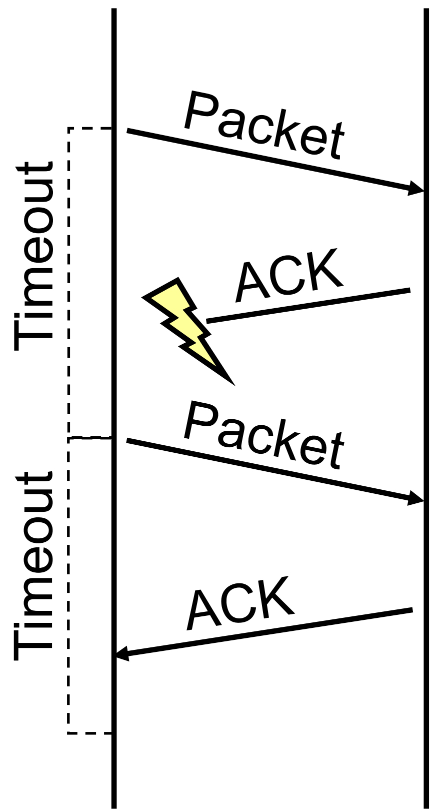


Optimizing Retransmissions

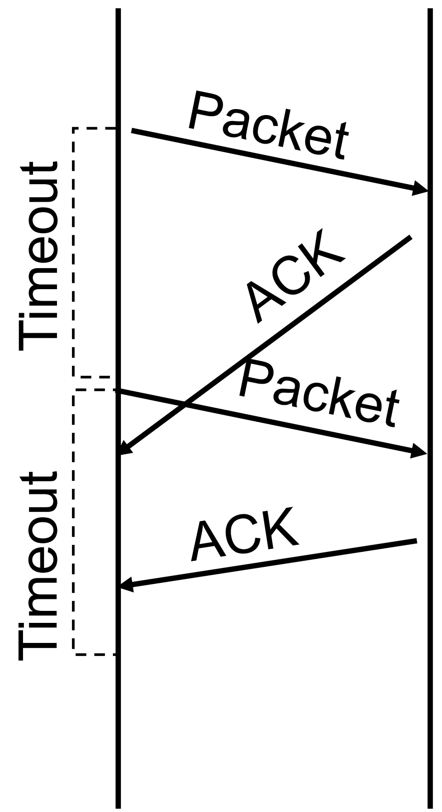
Reasons for Retransmission



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 “round-trip time”
  - … plus a fudge factor to account for queuing
- But, how does the sender know the RTT?
  - Running average of delay to receive an ACK

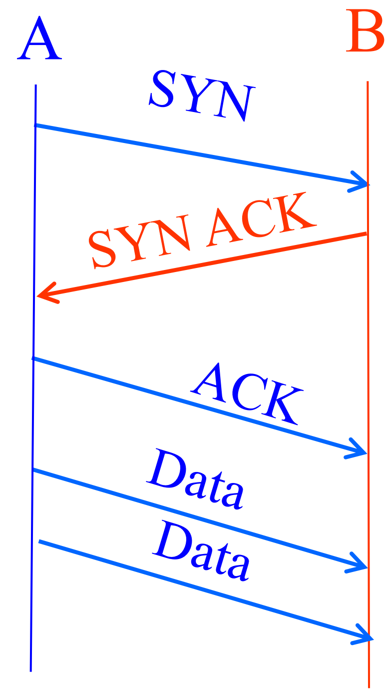
Still, timeouts are slow (≈RTT)

- When packet n is lost…
  - … packets n+1, n+2, and so on may get through
- Exploit the ACKs of these packets
  - ACK says receiver is still awaiting nth packet
  - Duplicate ACKs suggest later packets arrived
  - Sender uses “duplicate ACKs” as a hint
- Fast retransmission
  - Retransmit after “triple duplicate ACK”

Effectiveness of Fast Retransmit

- When does Fast Retransmit work best?
  - High likelihood of many packets in flight
  - Long data transfers, large window size, …
- Implications for Web traffic
  - Many Web transfers are short (e.g., 10 packets)
    - So, often there aren't many packets in flight
  - Making fast retransmit is less likely to "kick in"
    - Forcing users to click "reload" more often…

Establishing a TCP Connection



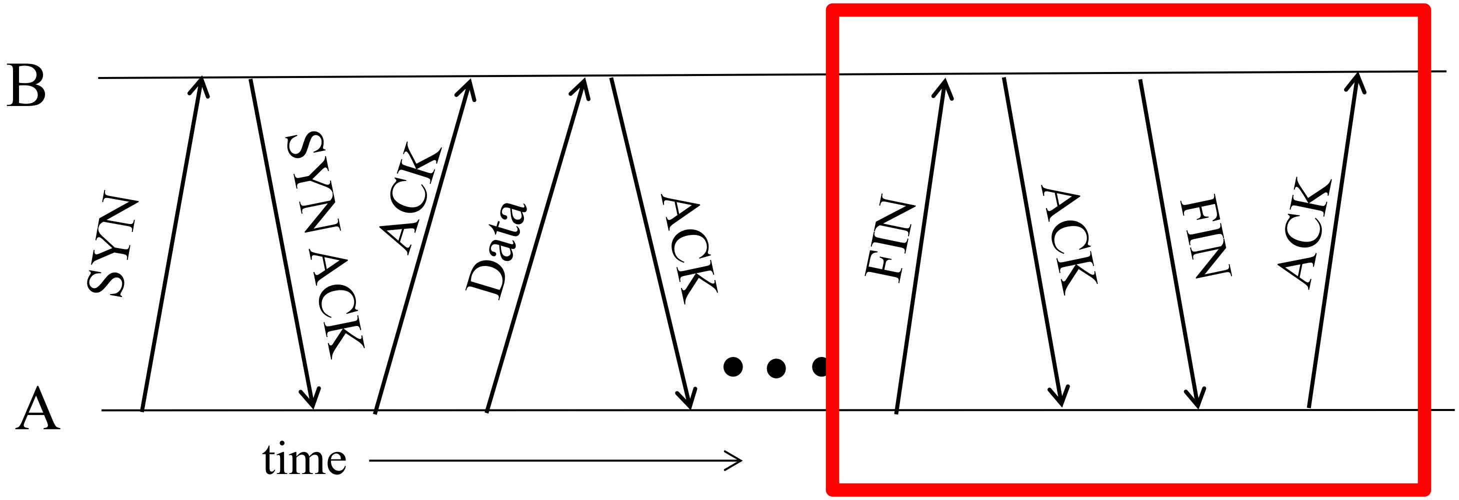
Each host tells its ISN to the other host.

- 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

SYN Loss and Web Downloads

- Upon sending SYN, sender sets a timer
  - If SYN lost, timer expires before SYN-ACK received
  - Sender retransmits SYN
- How should the TCP sender set the timer?
  - No idea how far away the receiver is
  - Some TCPs use default of 3 or 6 seconds
- Implications for web download
  - User gets impatient and hits reload
  - … Users aborts connection, initiates new socket
  - Essentially, forces a fast send of a new SYN!

Tearing Down the Connection



- Closing (each end of) 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 socket
  - Process invokes "close()"
  - Once TCP has sent all the outstanding bytes…
  - … then TCP sends a FIN
- Receiving a FIN: EOF
  - Process is reading data from socket
  - Eventually, read call returns an EOF