CSC4200/5200 – COMPUTER NETWORKING

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TRANSPORT LAYER PROTOCOLS

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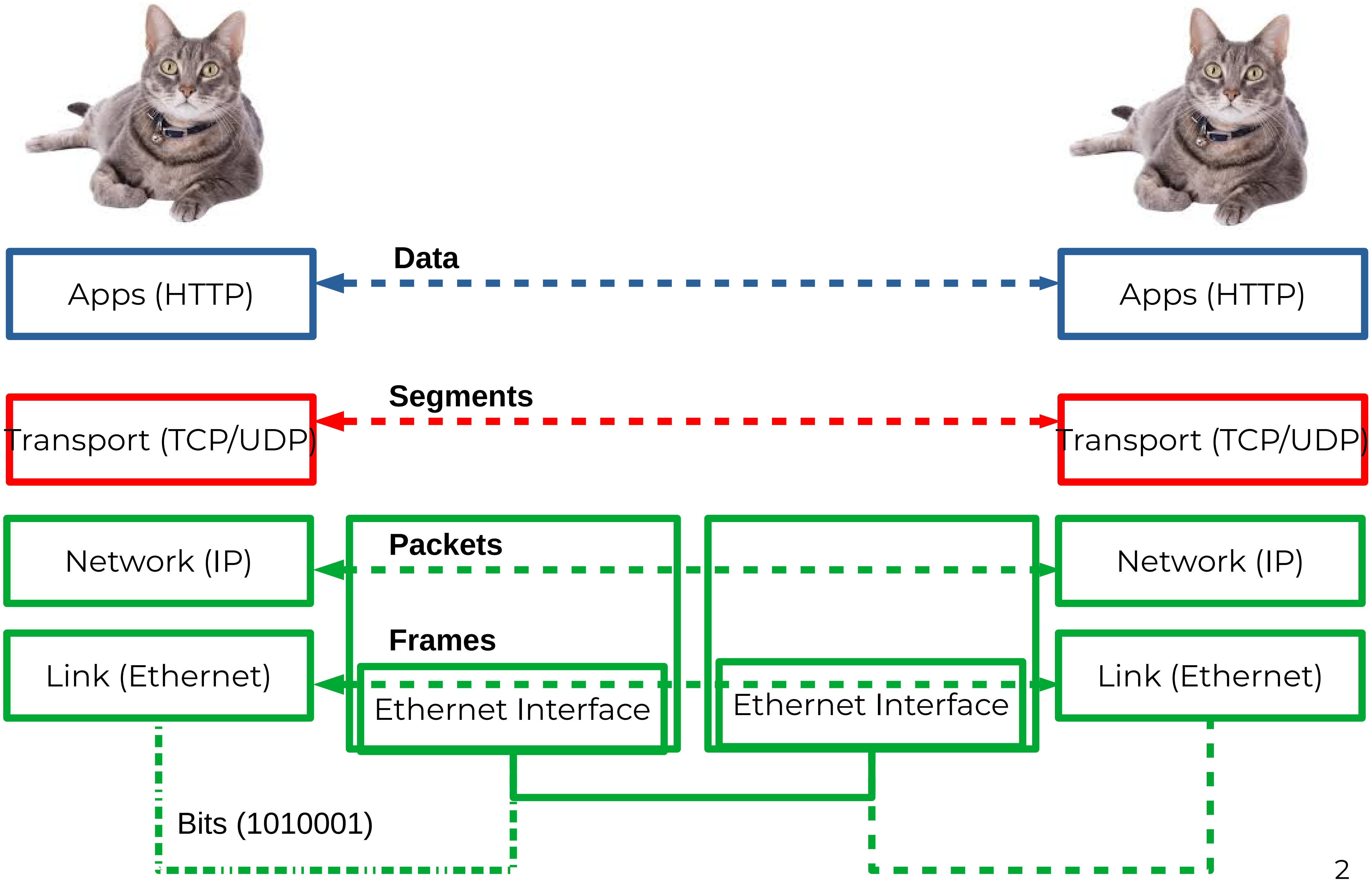
Tennessee
TECH

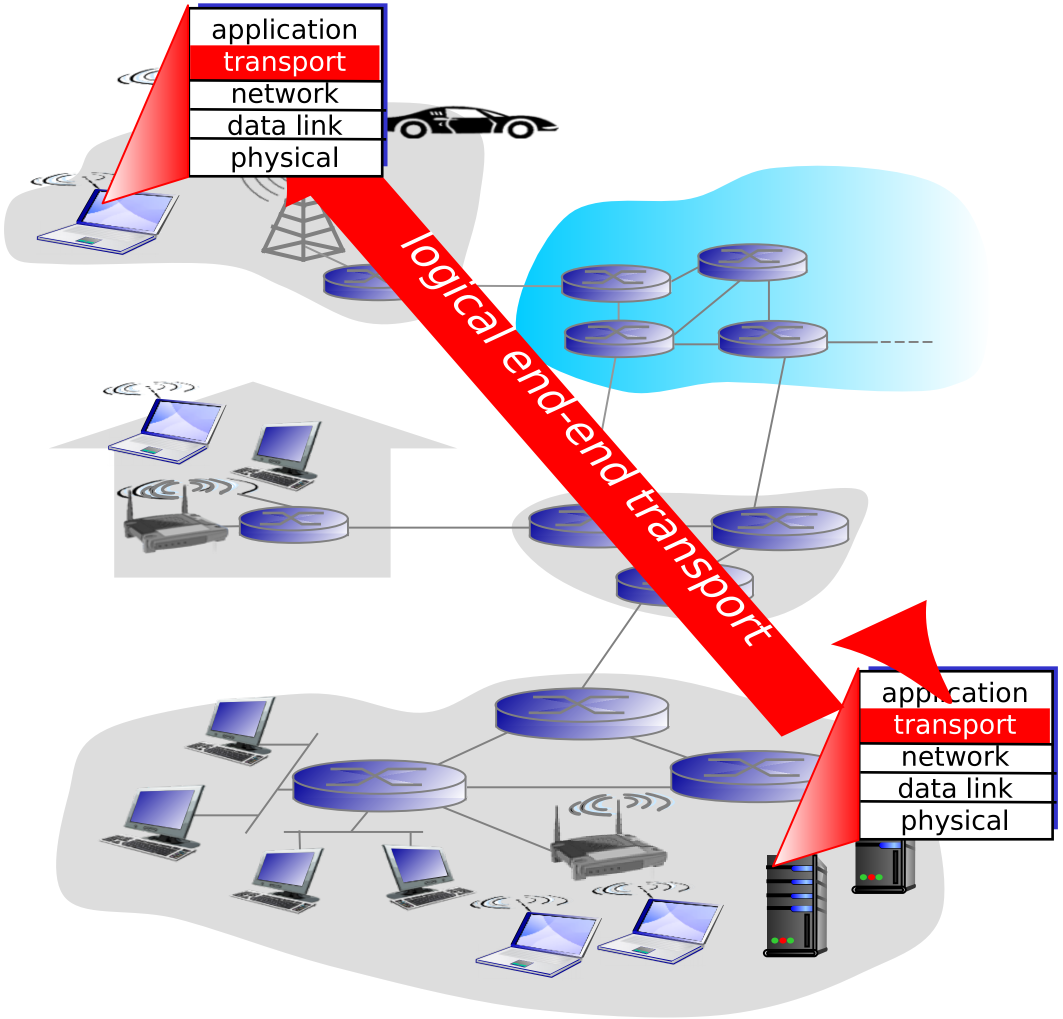

Data
Apps (HTTP)
Apps (HTTP)
Segments
Transport (TCP/UDP)
Transport (TCP/UDP)
Packets
Network (IP)
Network (IP)
Frames
Link (Ethernet)
Ethernet Interface
Ethernet Interface
Link (Ethernet)
Bits (1010001)

What is transport layer?

- Problem: How to turn this host-to-host packet delivery service into a process-to-process communication channel?

Transport services and protocols

- provide *logical communication* between app processes running on different hosts
- transport protocols run in end systems
  - send side: breaks app messages into *segments*, passes to network layer
  - rcv side: reassembles segments into messages, passes to app layer
- more than one transport protocol available to apps
  - Internet: TCP and UDP



Transport Layer

Our goals:

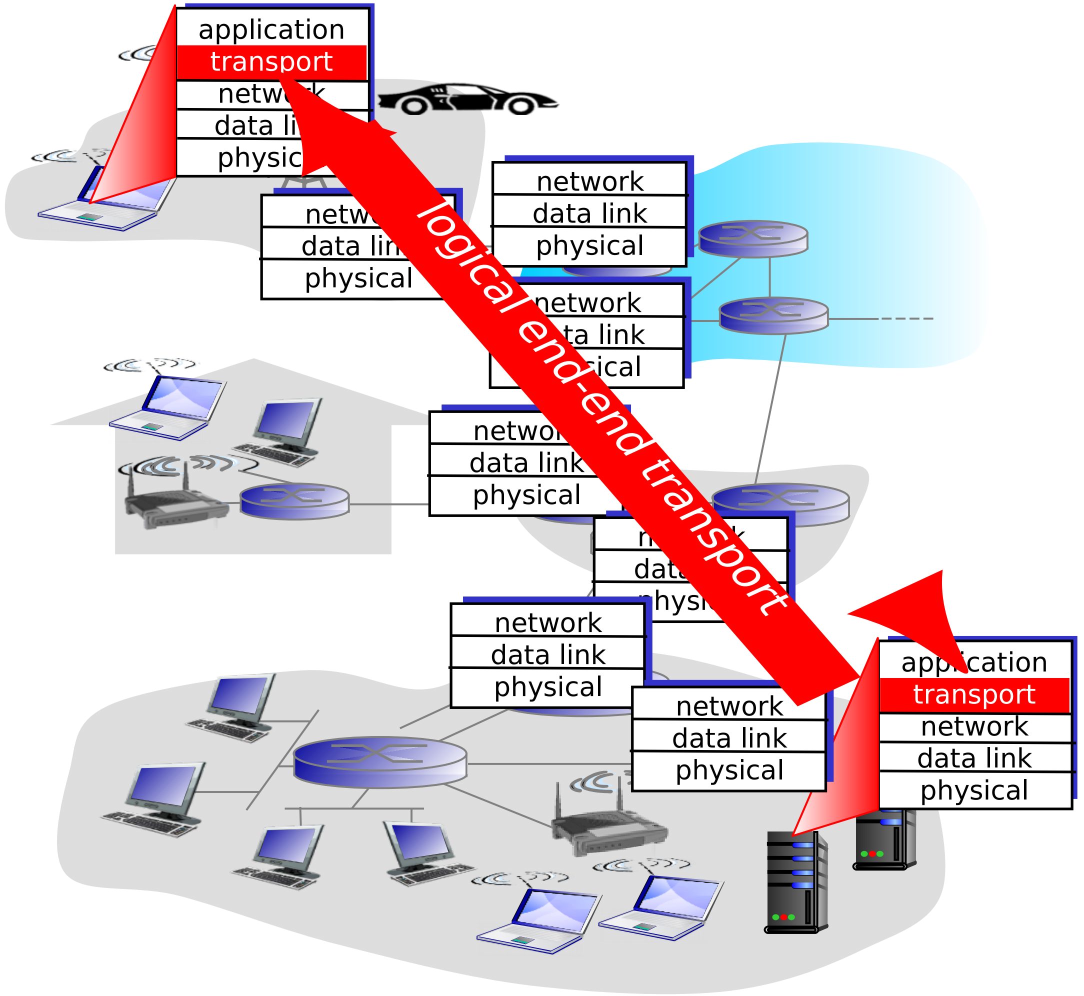
- understand principles behind transport layer services:
  - multiplexing, demultiplexing
  - reliable data transfer
  - flow control
  - congestion control
- learn about Internet transport layer protocols:
  - UDP: connectionless transport
  - TCP: connection-oriented reliable transport
  - TCP congestion control

Transport vs. network layer

- network layer: logical communication between hosts
- transport layer: logical communication between processes
  - relies on, enhances, network layer services

Internet transport-layer protocols

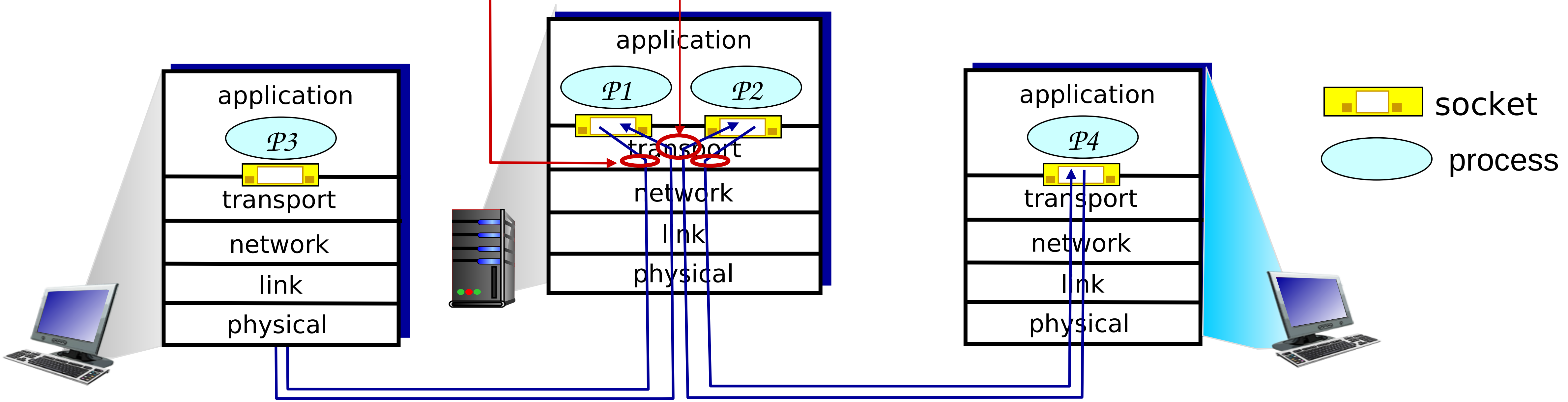
- Reliable, in-order delivery (TCP)
  - congestion control
  - flow control
  - connection setup
- unreliable, unordered delivery: UDP
  - no-frills extension of "best-effort" IP
- services not available:
  - delay guarantees
  - bandwidth guarantees



Multiplexing/demultiplexing

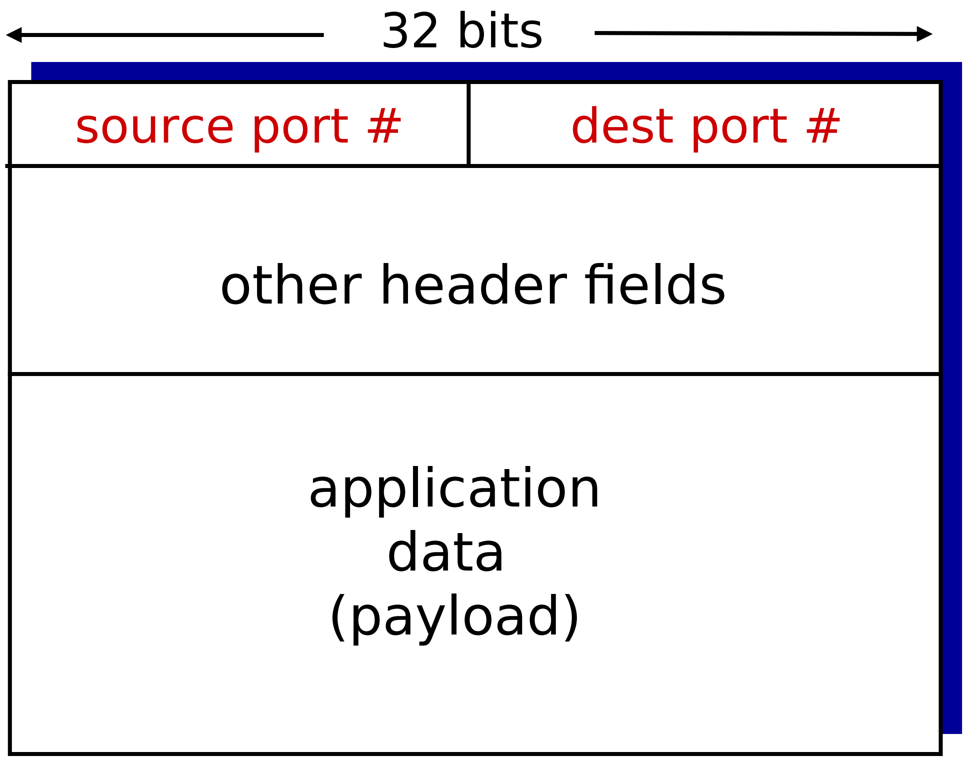
multiplexing at sender:
handle data from multiple sockets, add transport header (later used for demultiplexing)

demultiplexing at receiver:
use header info to deliver received segments to correct socket



How demultiplexing works

- host receives IP datagrams
  - each datagram has source IP address, destination IP address
  - each datagram carries one transport-layer segment
  - each segment has source, destination port number
- host uses *IP addresses & port numbers* to direct segment to appropriate socket



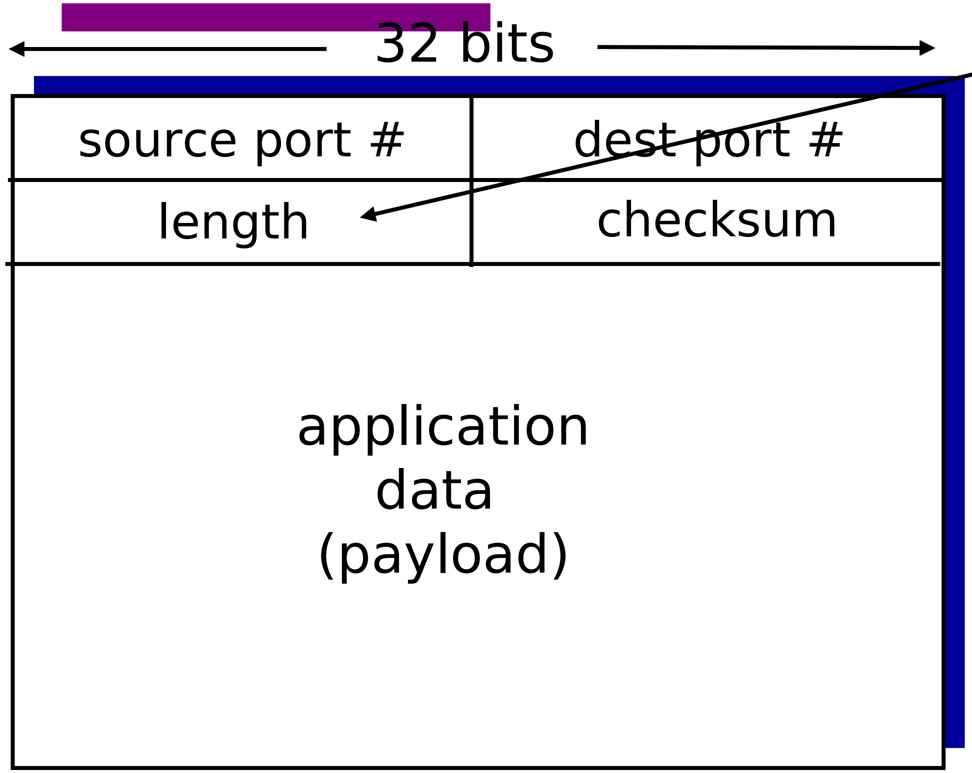
TCP/UDP segment format

UDP: User Datagram Protocol

- Lightweight communication
  - Avoid overhead and delays of ordered delivery
  - Send messages to and receive them from a socket

- *connectionless:*
  - no handshaking between UDP sender, receiver
  - each UDP segment handled independently of others

UDP: segment header

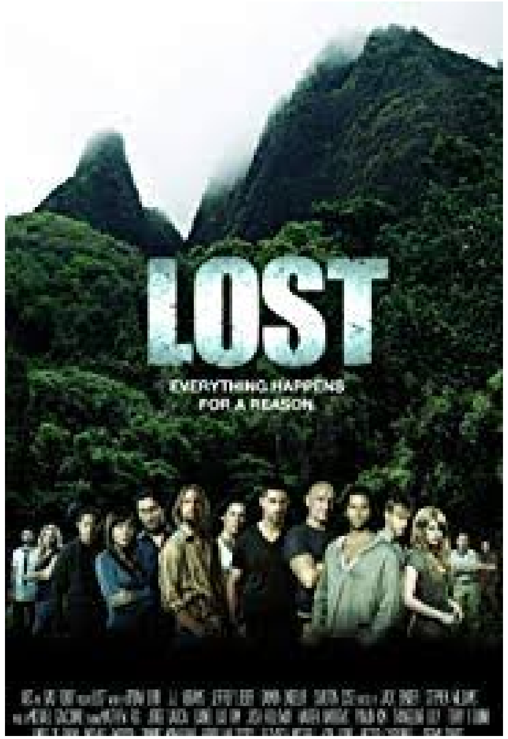


UDP segment format

Why would anyone use UDP?

- no connection establishment (which can add delay)
- simple: no connection state at sender, receiver
- small header size
- no congestion control: UDP can blast away as fast as desired

Who uses UDP?



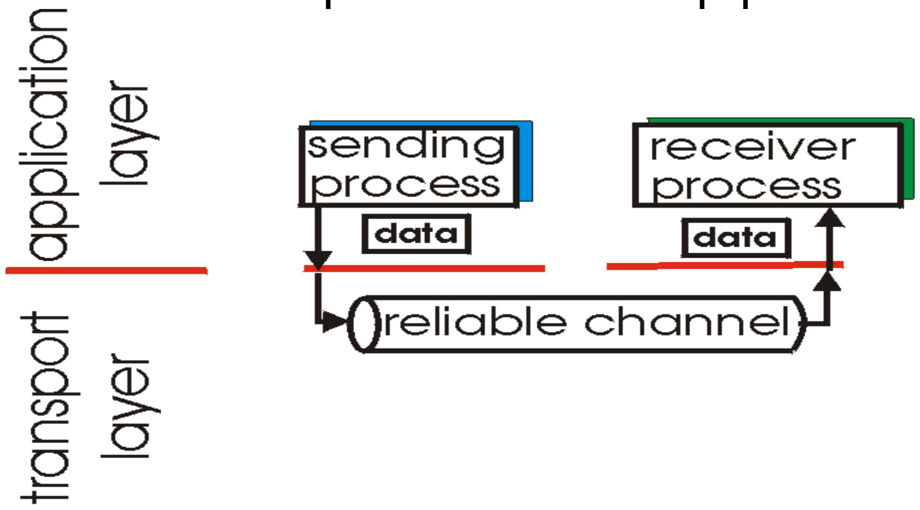
- Multimedia applications

  - Sending a lost frame is not worth it
  - By the time the packet is retransmitted, it's too late

- DNS
  - Small query
  - Connection establishment might be an overkill

Principles of reliable data transfer

- important in application, transport, link layers

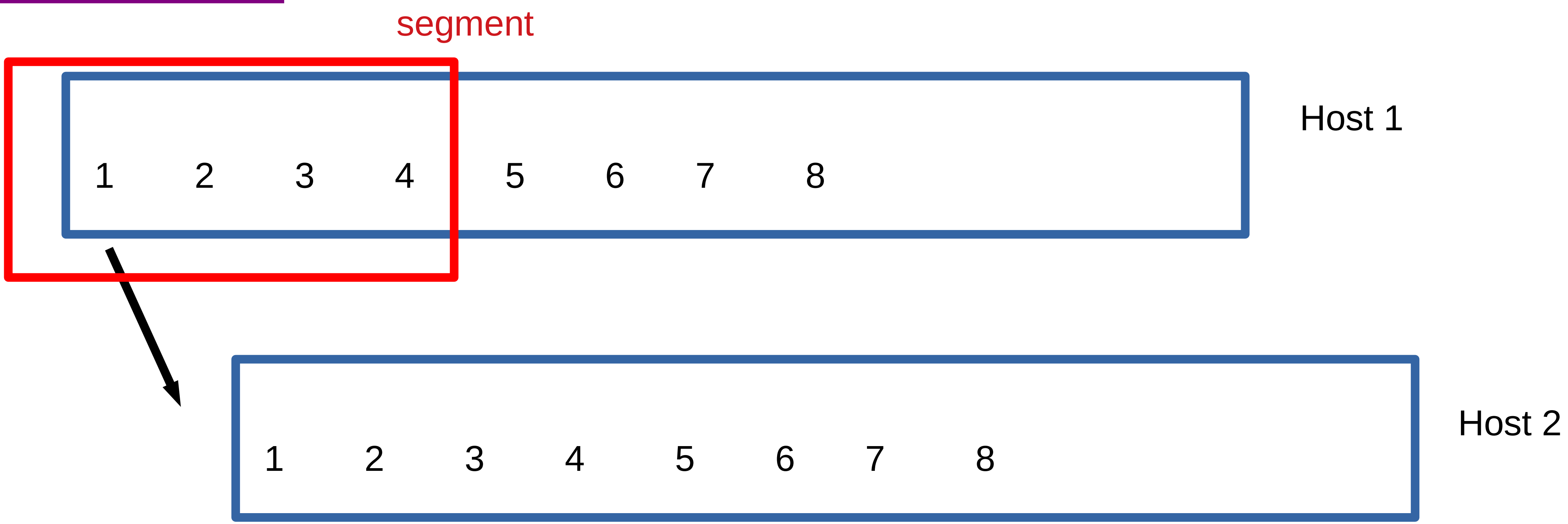


(a) provided service

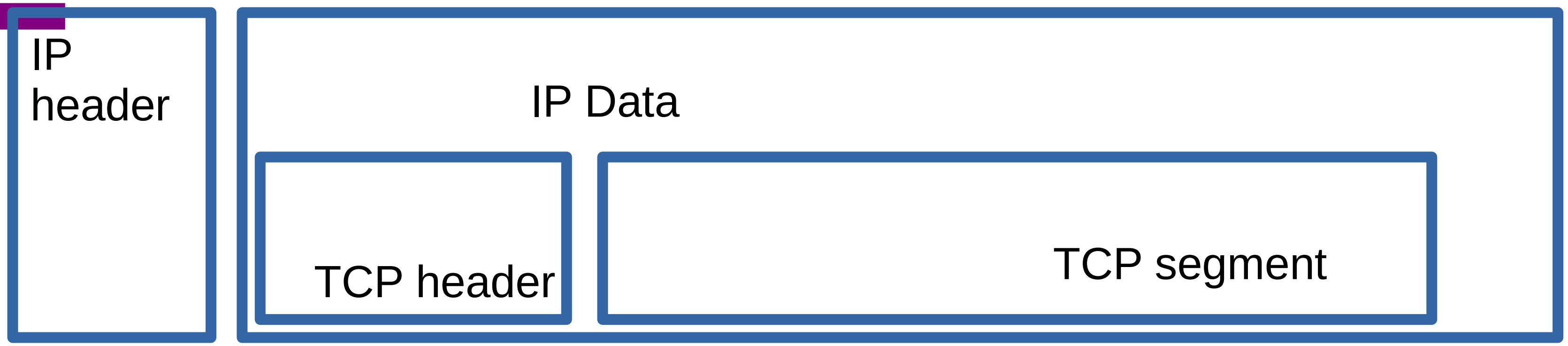
TCP – Transmission Control Protocol

- point-to-point:
  - one sender, one receiver
- reliable, in-order *byte steam:*
  - no “message boundaries”
- pipelined:
  - TCP congestion and flow control set window size
- full duplex data:
  - bi-directional data flow in same connection
  - MSS: maximum segment size
- connection-oriented:
  - handshaking (exchange of control msgs) inits sender, receiver state before data exchange
- flow controlled:
  - sender will not overwhelm receiver

TCP – Transmission Control Protocol



TCP Segment



IP → No more than MTU (1500 Bytes)

TCP header → 20 bytes

TCP segment → 1460 bytes

Why?

TCP Header

SYN

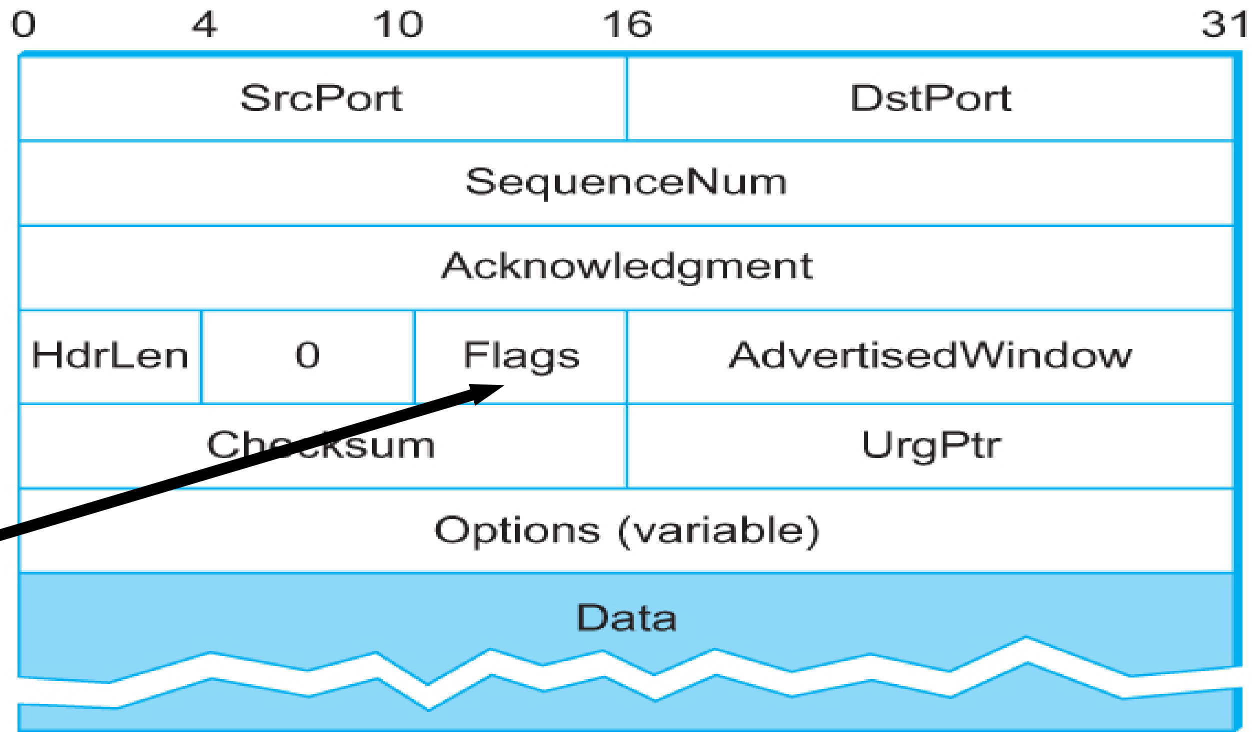
FIN

RST

PSH

URG

ACK



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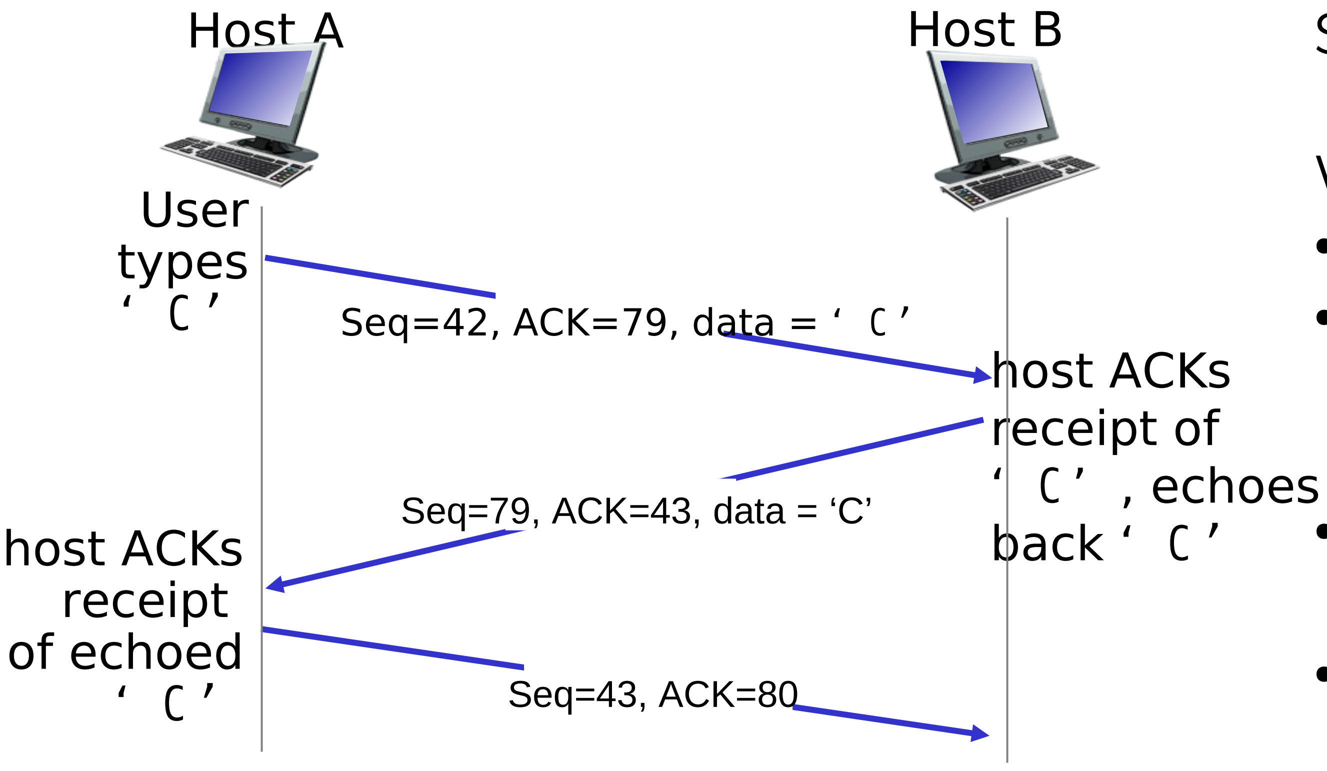
TCP seq. numbers, ISNs



simple telnet scenario

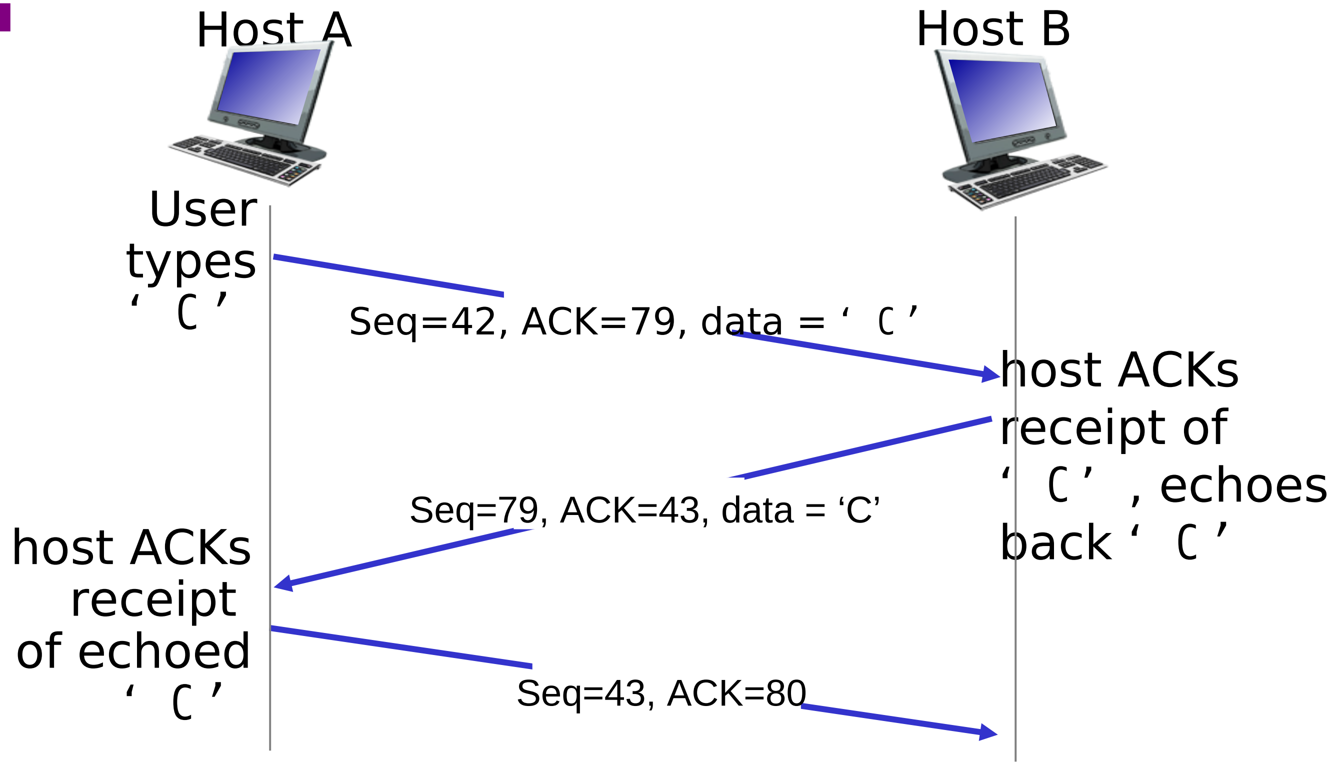
Sequence number for the first byte

Why not use 0 all the time?

- Security
- Port are reused, you might end up using someone else’s previous connection
- Phone number analogy

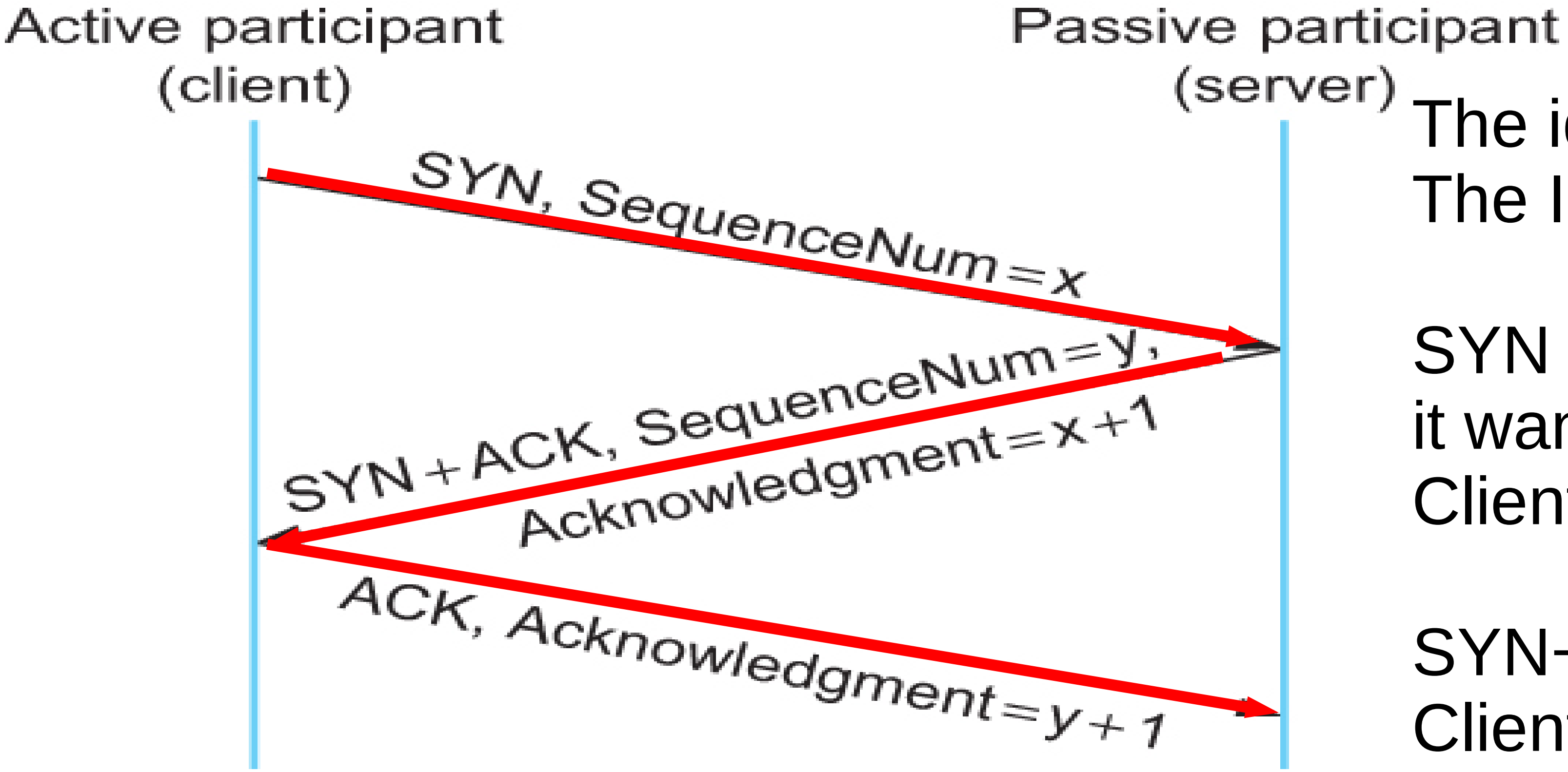
- TCP ISNs are clock based
  - 32 bits, increments in 4 microseconds
  - 4.55 hours wrap around time

TCP seq. numbers, ACKs



simple telnet scenario

TCP Three-way Handshake



Timeline for three-way handshake algorithm

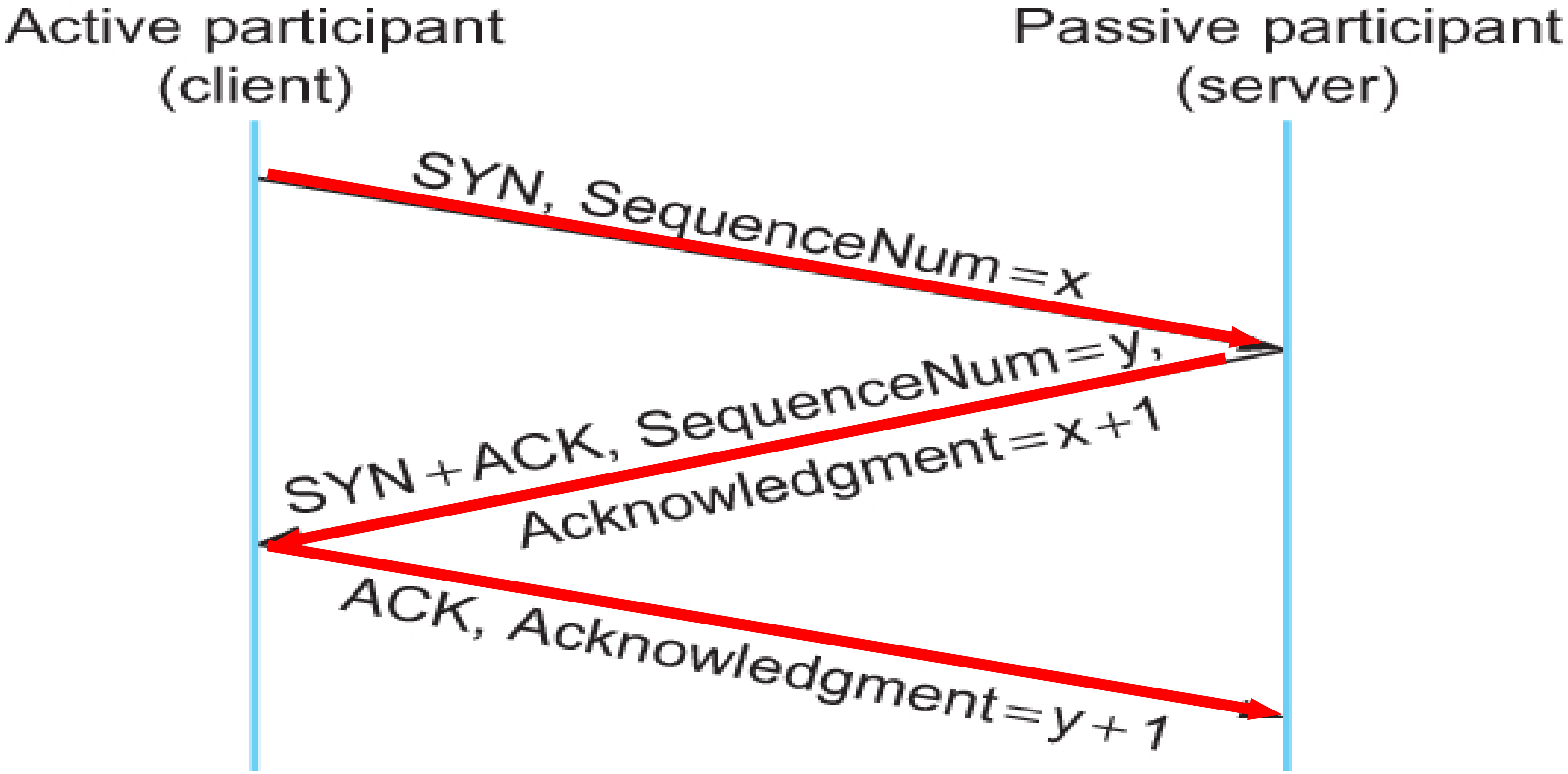
The idea is to tell each other
The ISNs

SYN → Client tells server that it wants to open a connection, Client's ISN = x

SYN+ ACK → Server tells Client → Okay → Server's ISN = y, ACK = CLSeq + 1

Why increment by 1?

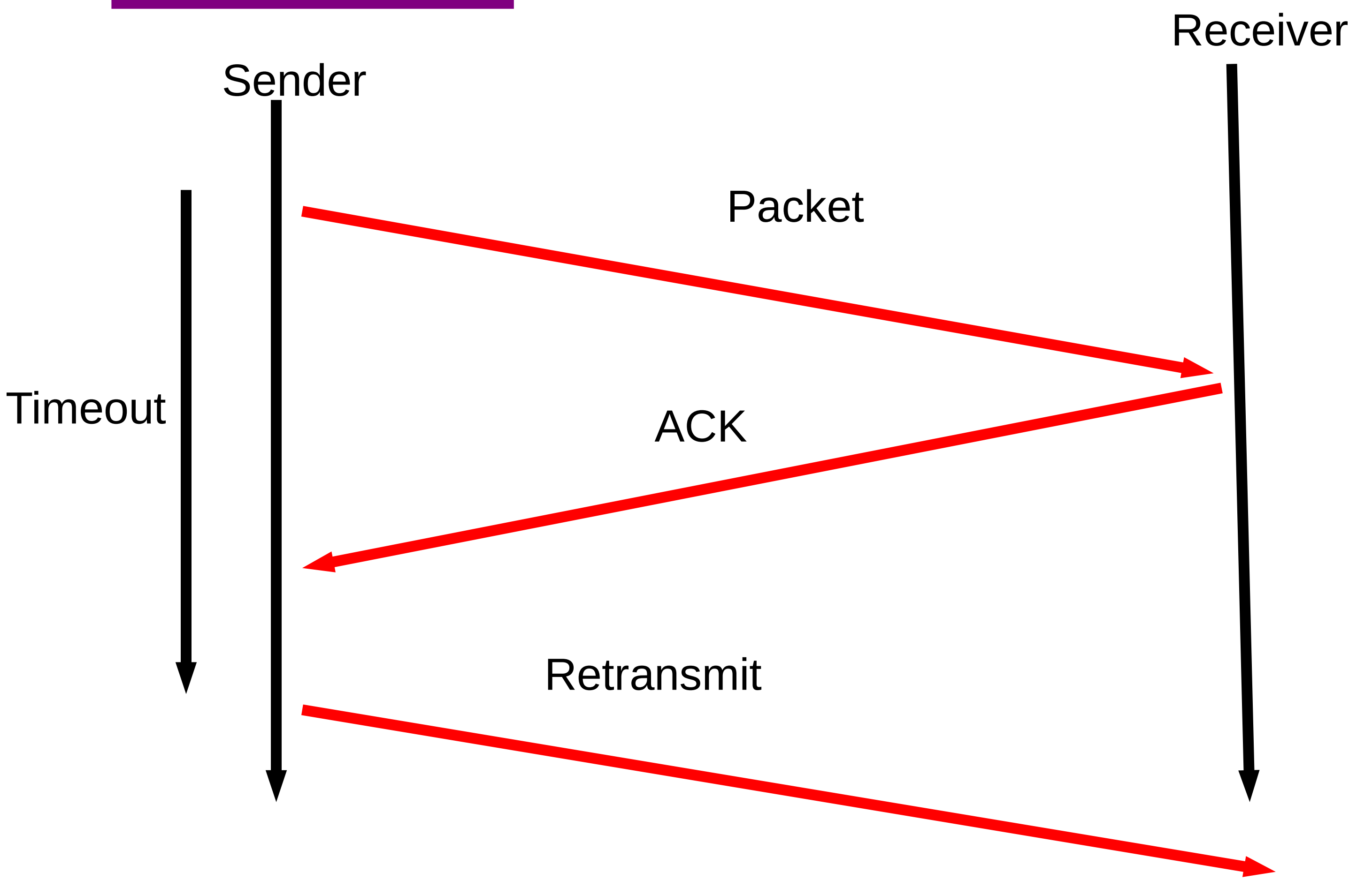
What if the SYN is lost?



Timeline for three-way handshake algorithm

Start Timer and resend

TCP Retransmission - ARQ

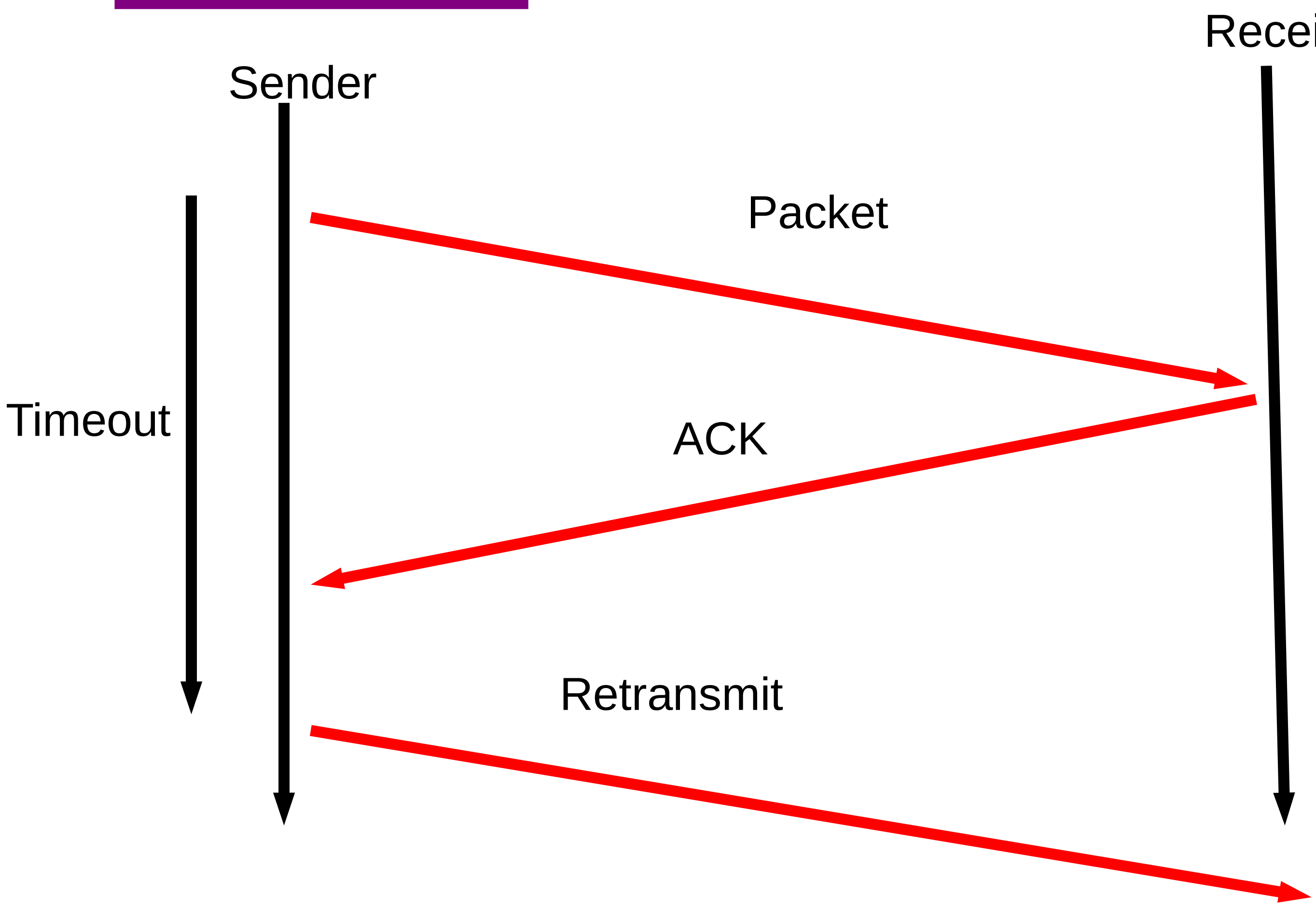


Each packet is “ACK”ed by the receiver

If ACK isn’t received by timeout, resend

Example, Stop-n-wait

How long should the sender wait?



For TCP, timeout is a function of RTT

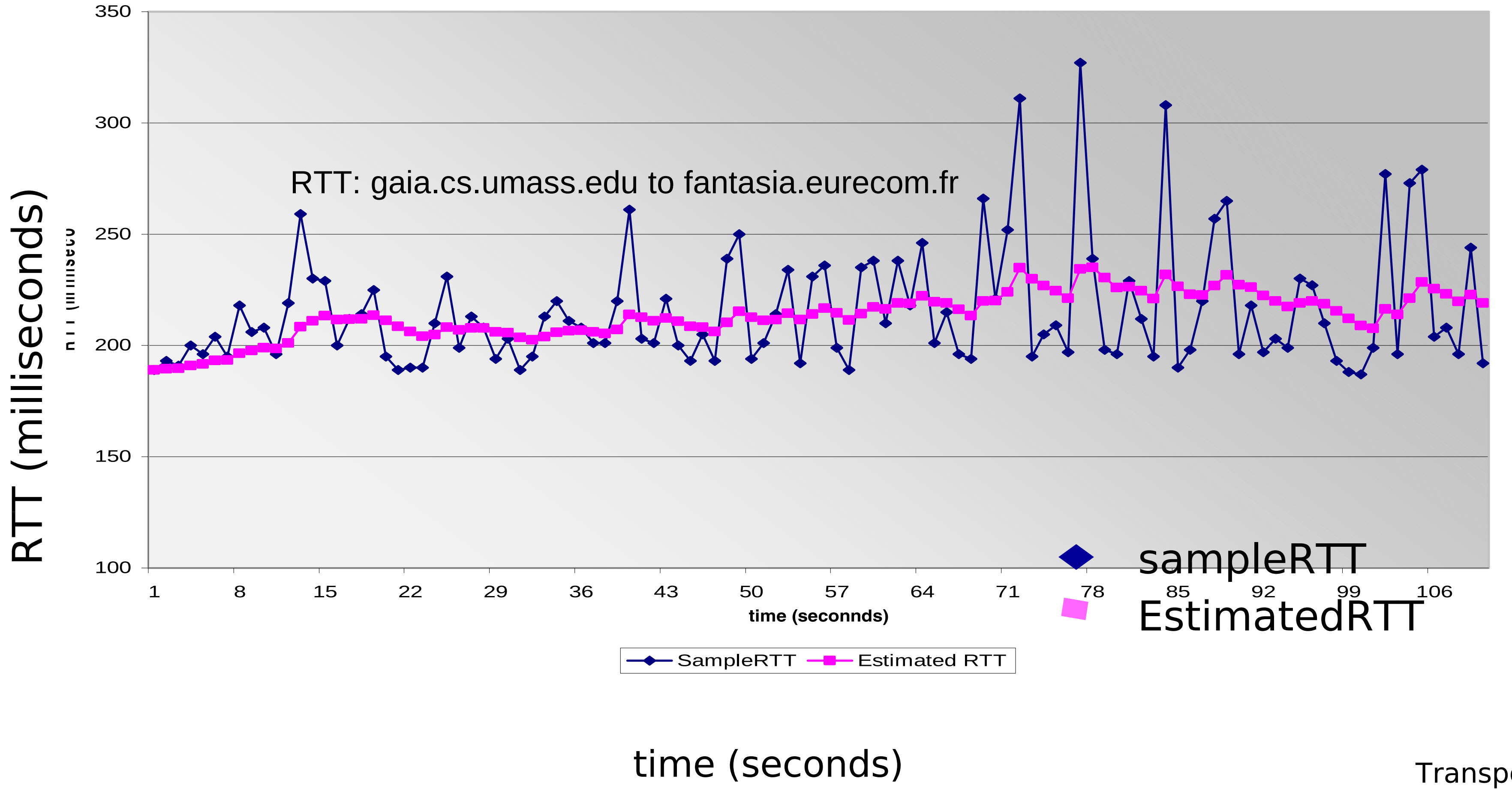
Keep a running estimate of RTTs by watching the ACKs

EstimatedRTT = (1 – α) • EstimatedRTT
+ α • SampleRTT

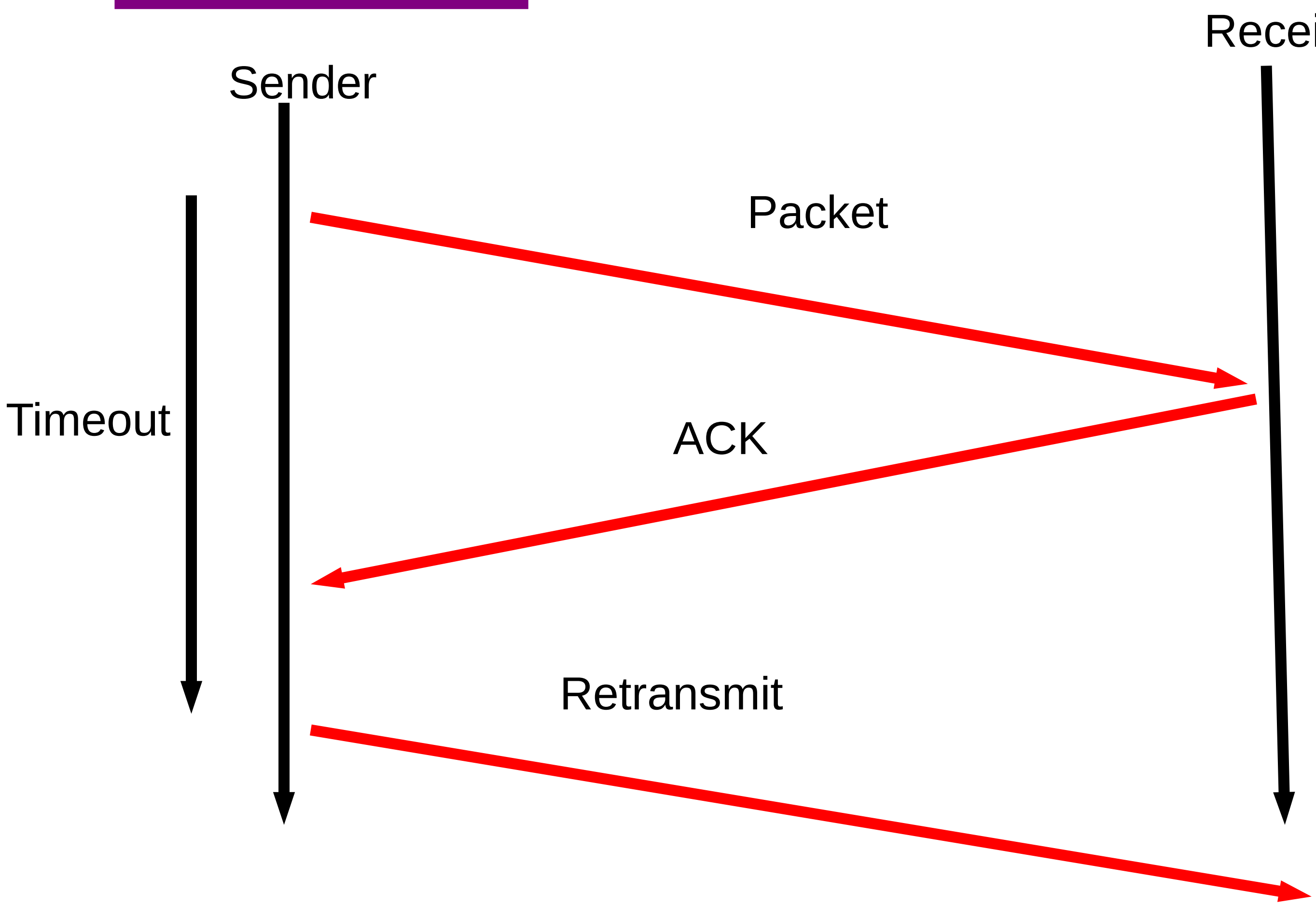
Alpha is generally 0.125

Timeout = 2* EstimatedRTT

TCP round trip time, timeout



But stop and wait is inefficient



Only one segment in-flight

Especially bad if delay is high!

Solution – sliding window

Sliding Window Revisited

Sending Side

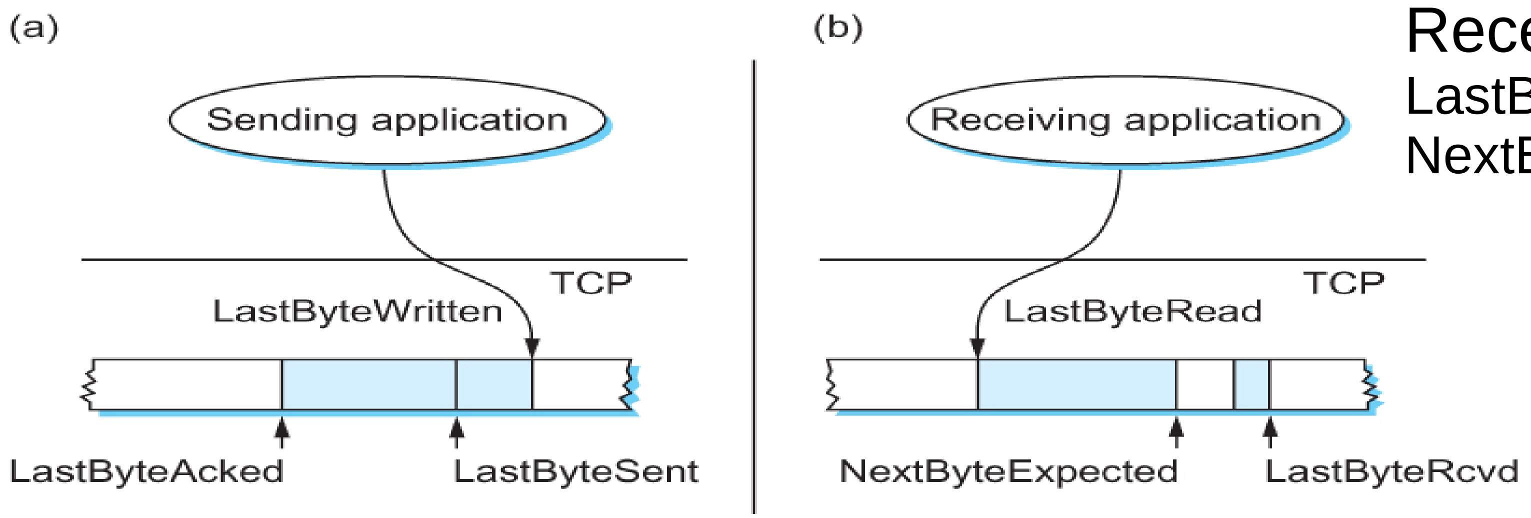
LastByteAcked ≤ LastByteSent

LastByteSent ≤ LastByteWritten

Receiving Side

LastByteRead < NextByteExpected

NextByteExpected ≤ LastByteRcvd + 1

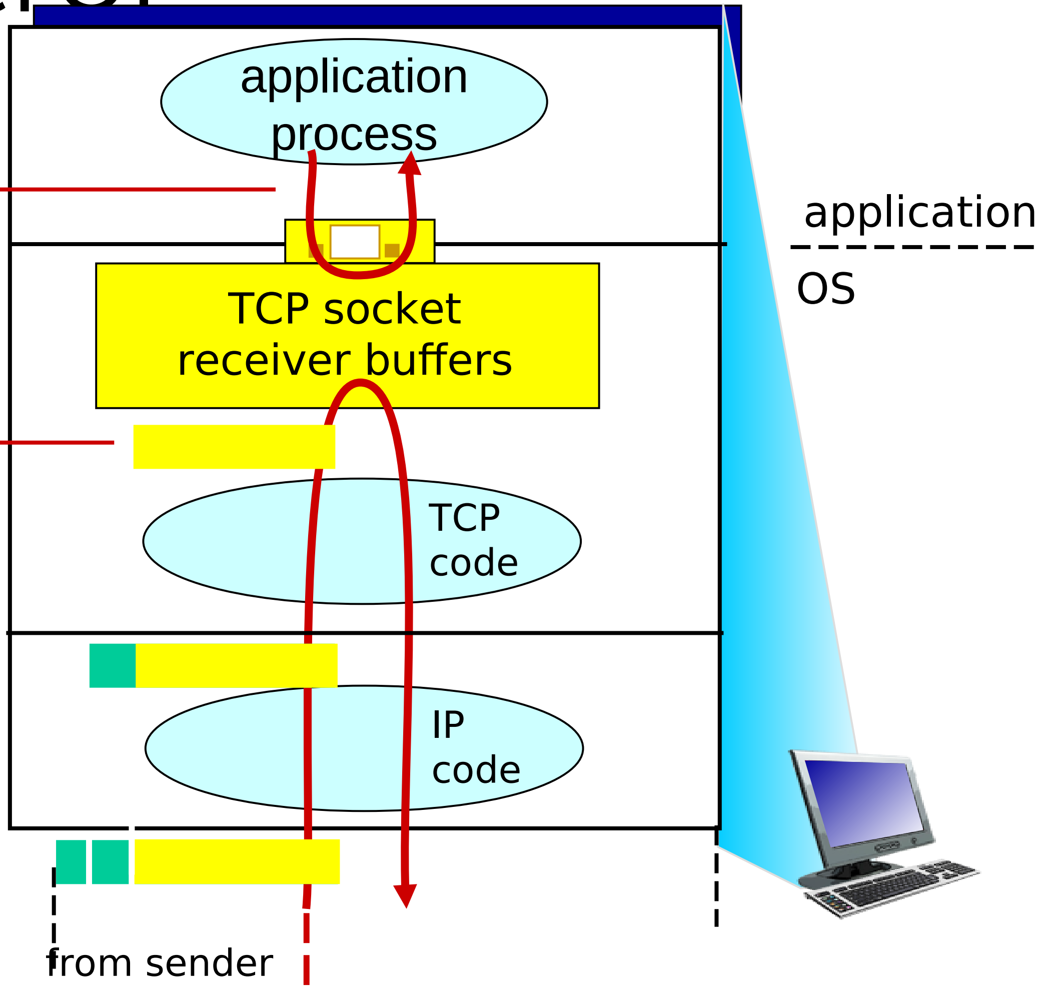


Relationship between TCP send buffer (a) and receive buffer (b).

Used for TCP flow control

application may remove data from TCP socket buffers ….

… slower than TCP receiver is delivering (sender is sending)



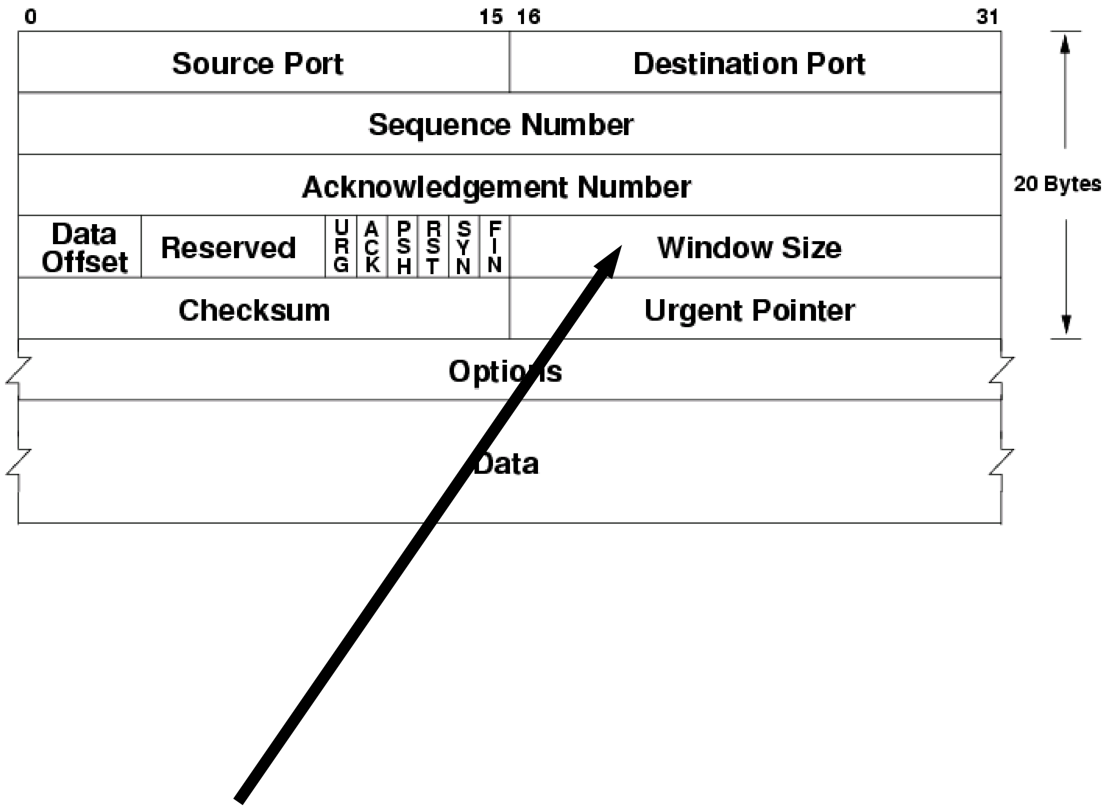
receiver protocol stack

flow control

receiver controls sender, so sender won't overflow receiver's buffer by transmitting too much, too fast

TCP flow control

- receiver "advertises" free buffer space in the header
- sender limits amount of unacked ("in-flight") data to receiver's **`rwnd`** value
- guarantees receive buffer will not overflow

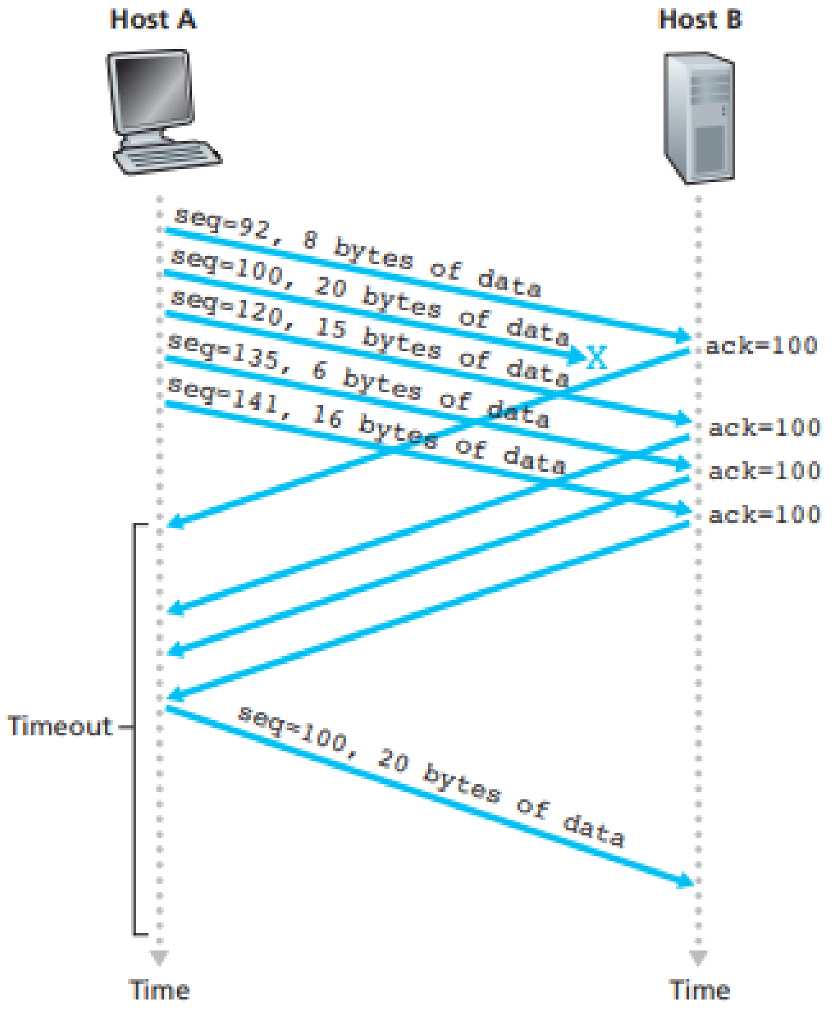


TCP Fast Retransmission

Timeouts are wasteful

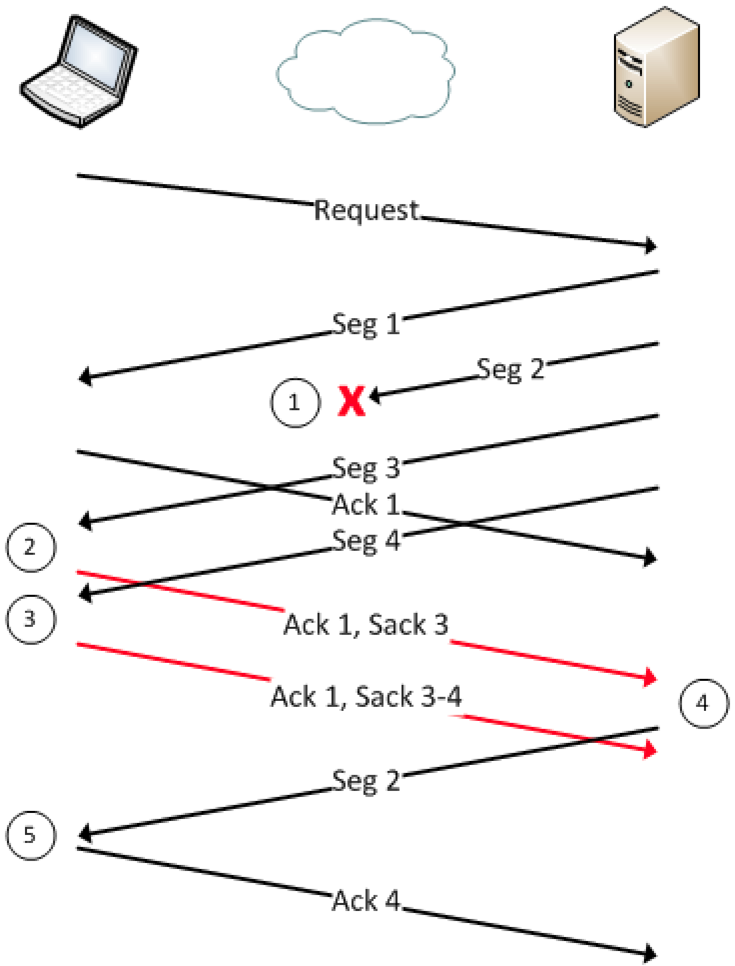
Triple duplicate ACKs

Retransmits before timeout



TCP Fast Retransmission - SACK

What if multiple segments are lost?

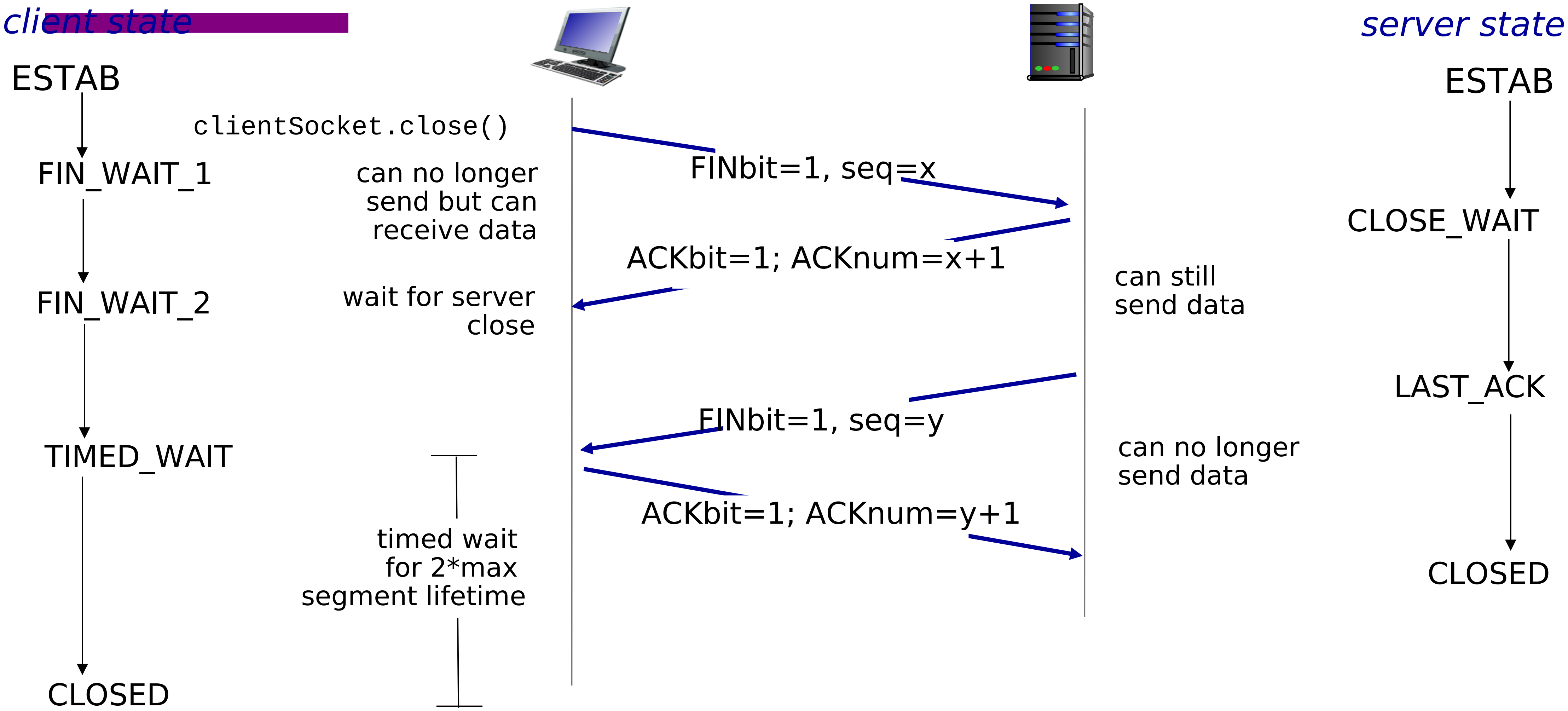


Very good explanation:
https://packetlife.net/blog/2010/jun/17/tcp-selective-acknowledgments-sack/

TCP: closing a connection

- client, server each close their side of connection
  - send TCP segment with FIN bit = 1
- respond to received FIN with ACK
  - on receiving FIN, ACK can be combined with own FIN
- simultaneous FIN exchanges can be handled

TCP: closing a connection



Next steps

TCP Congestion control

Programming assignment 3 posted