Information Network

Lecture 6 : Transport Layer Introduction

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Today's lecture

1. video streaming and content distribution networks (CDNs)
2. Summary of the application layer
3. Introduction to the transport layer

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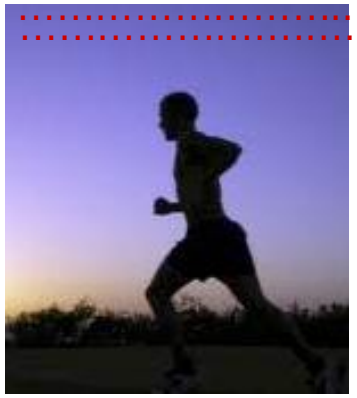
Multimedia: video

- video: sequence of images displayed at constant rate
  - e.g., 24 images/sec
- digital image: array of pixels
  - each pixel represented by bits
- coding: use redundancy *within* and *between* images to decrease number of bits used to encode image
  - spatial (within image)
  - temporal (from one image to next)

spatial coding example: instead of sending *N* values of same color (all purple), send only two values: color value (*purple*) *and number of repeated values* (N)



frame *i*

frame *i+1*

temporal coding example: instead of sending complete frame at i+1, send only differences from frame i

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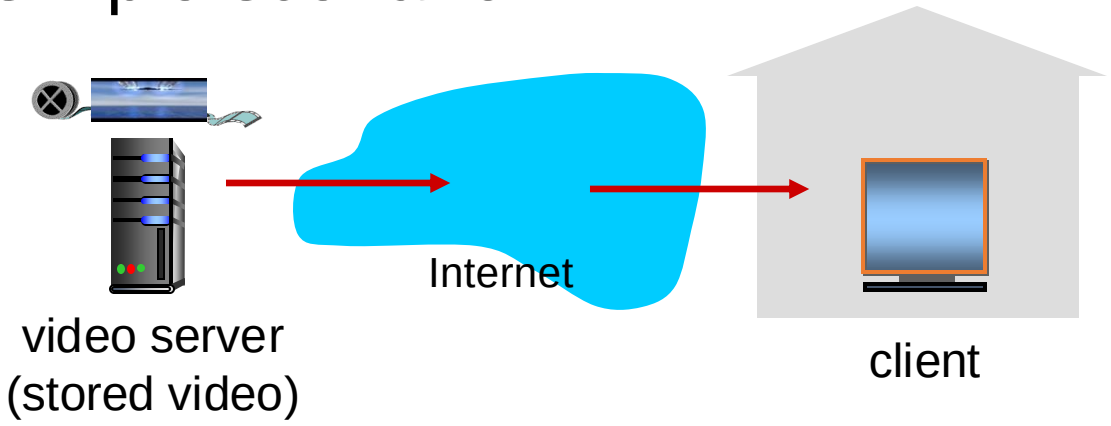
Streaming stored video

simple scenario:



Main challenges:

- server-to-client bandwidth will *vary* over time, with changing network congestion levels
- packet loss, delay due to congestion will delay playout, or result in poor video quality

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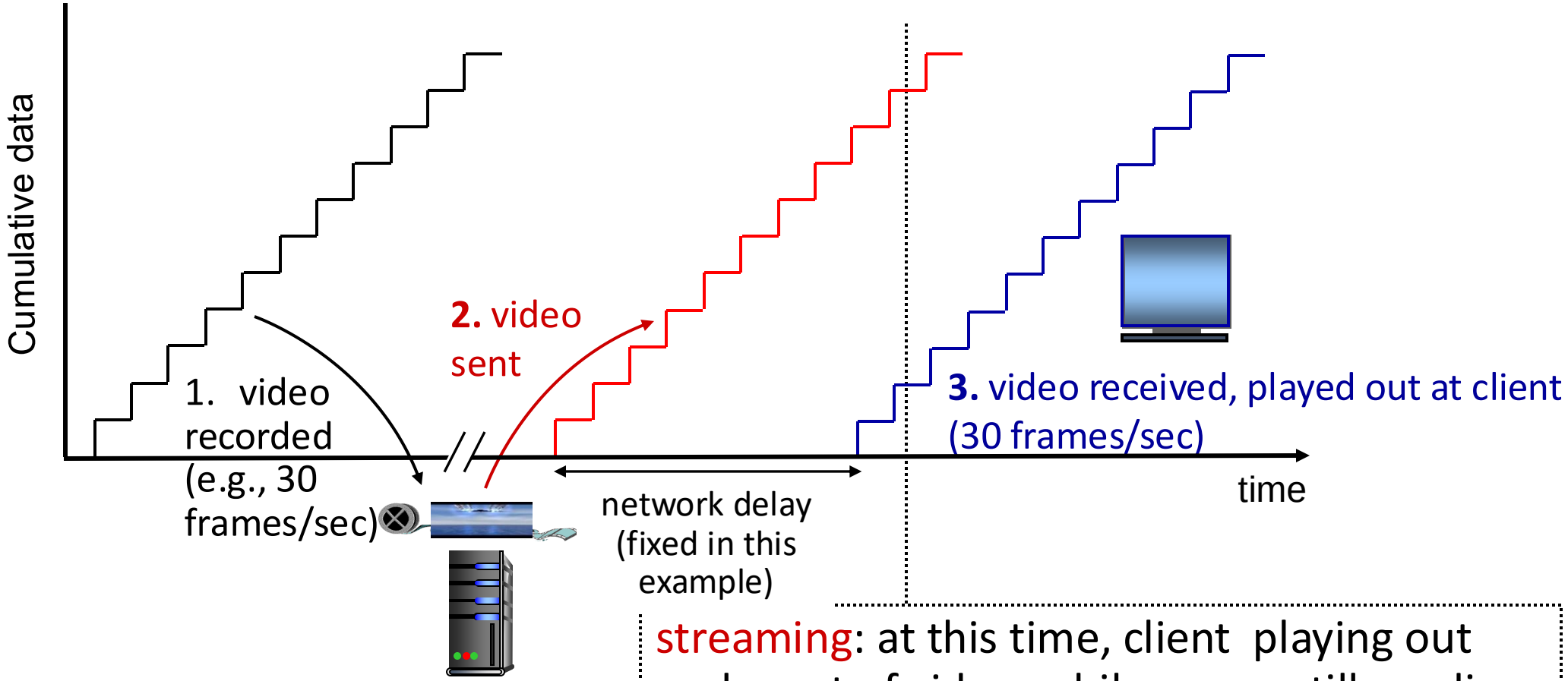
Streaming stored video



streaming: at this time, client playing out early part of video, while server still sending later part of video

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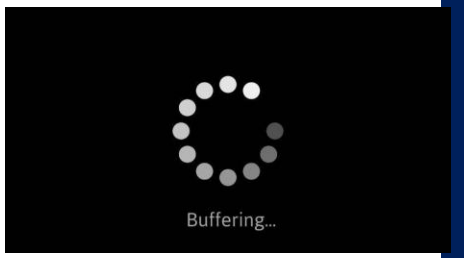
Streaming stored video: challenges

- continuous playout constraint: during client video playout, playout timing must match original timing
  - … but network delays are variable, so will need client-side buffer to match continuous playout constraint
- other challenges:
  - client interactivity: pause, fast-forward, rewind, jump through video
  - video packets may be lost, retransmitted (additional delay)



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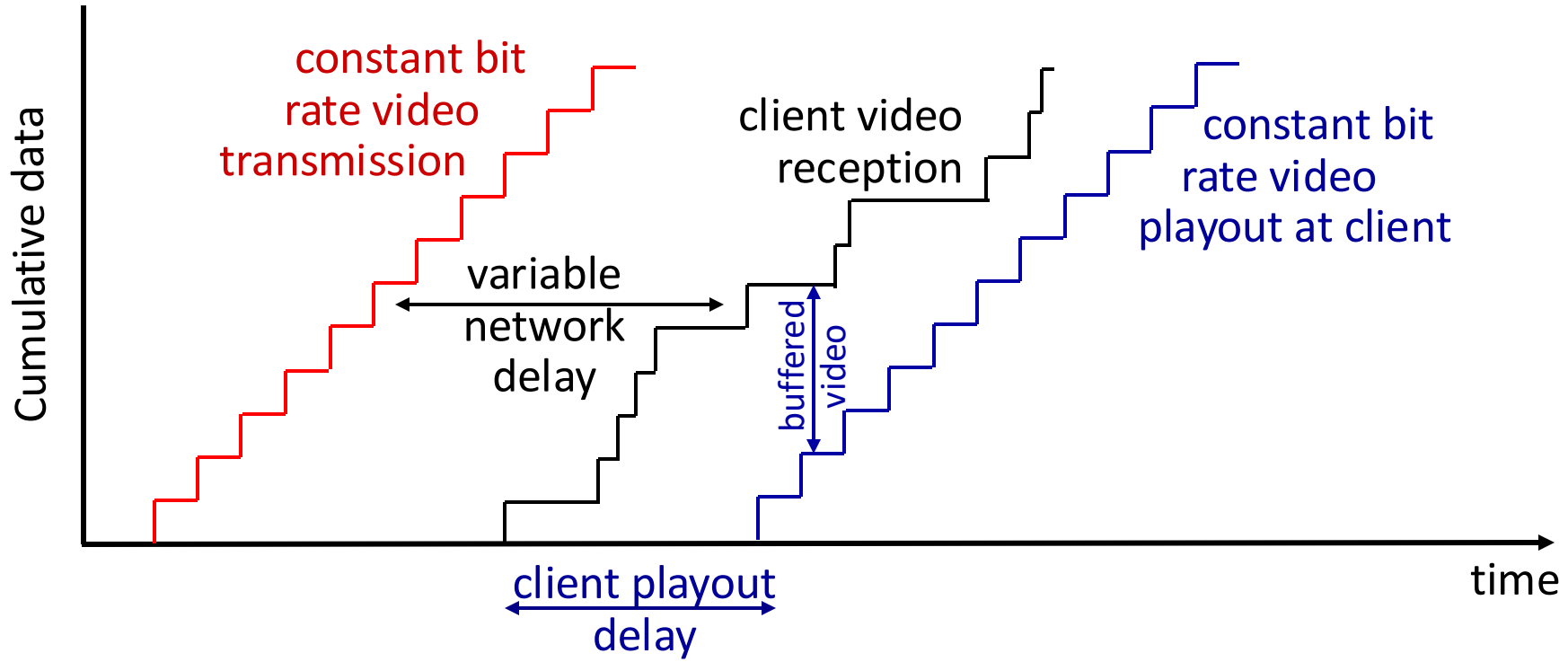
Streaming stored video: playout buffering



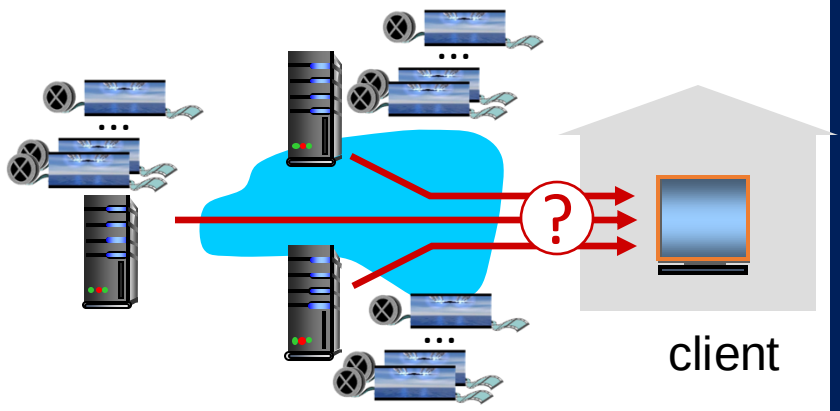
- *client-side buffering and playout delay:* compensate for network-added delay, delay jitter

Streaming multimedia: DASH

*D*ynamic, *A*daptive
*S*treaming over *H*TTP

server:

- divides video file into multiple chunks
- each chunk encoded at multiple different rates
- different rate encodings stored in different files
- files replicated in various locations
- *manifest file:* provides URLs for different chunks



client:

- periodically estimates server-to-client bandwidth
- consulting manifest, requests one chunk at a time
  - chooses maximum coding rate sustainable given current bandwidth
  - can choose different coding rates at different points in time (depending on available bandwidth at time), and from different servers

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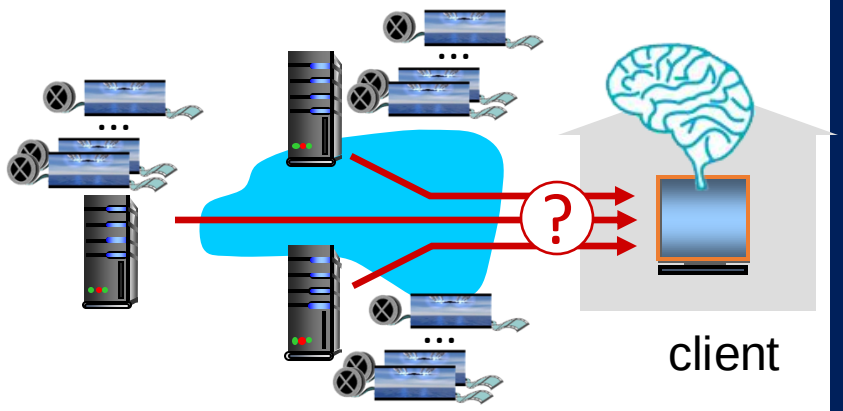
Streaming multimedia: DASH

- *"intelligence"* at client: client determines
  - *when* to request chunk (so that buffer starvation, or overflow does not occur)
  - *what encoding rate* to request (higher quality when more bandwidth available)
  - *where* to request chunk (can request from URL server that is "close" to client or has high available bandwidth)



Streaming video = encoding + DASH + playout buffering

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Content distribution networks

- *challenge:* how to stream content (selected from millions of videos) to hundreds of thousands of *simultaneous* users?

- *option 1:* single, large "mega-server"
  - single point of failure
  - point of network congestion
  - long path to distant clients

This solution *doesn't scale*

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Content distribution networks

- *challenge:* how to stream content (selected from millions of videos) to hundreds of thousands of simultaneous users?

- *option 2:* store/serve multiple copies of videos at multiple geographically distributed sites *(CDN)*

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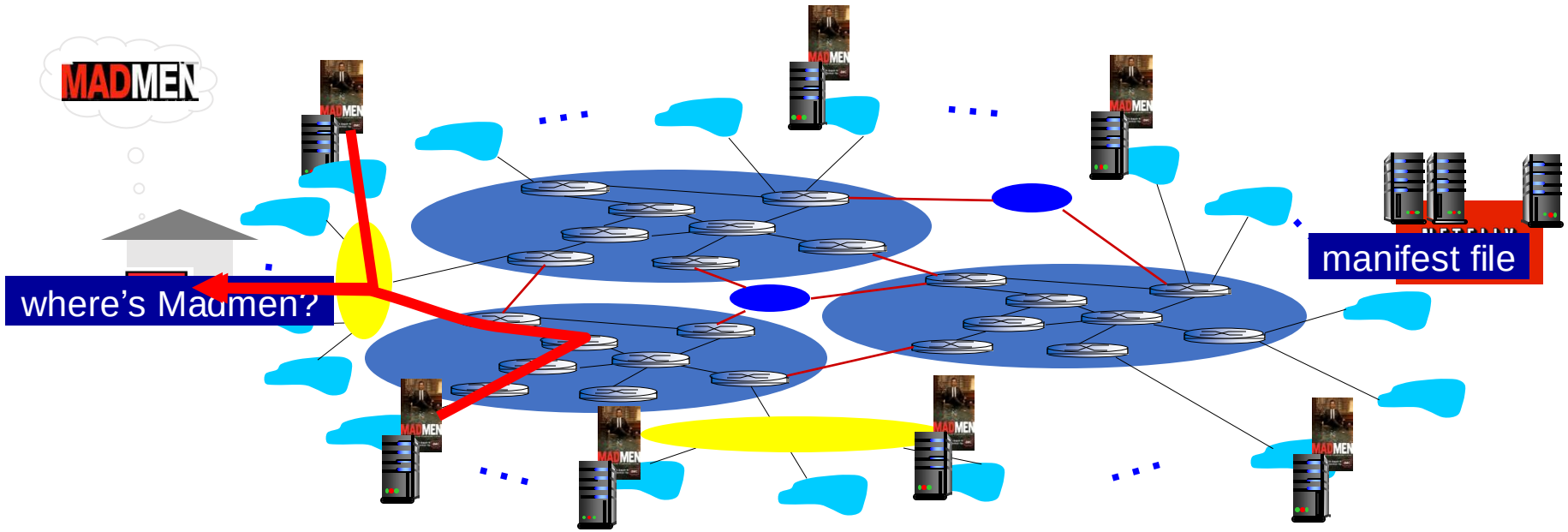
Content Distribution Networks (CDNs)

- CDN: stores copies of content at CDN nodes
  - e.g. Netflix stores copies of MadMen
- subscriber requests content from CDN
  - directed to nearby copy, retrieves content
  - may choose different copy if network path congested



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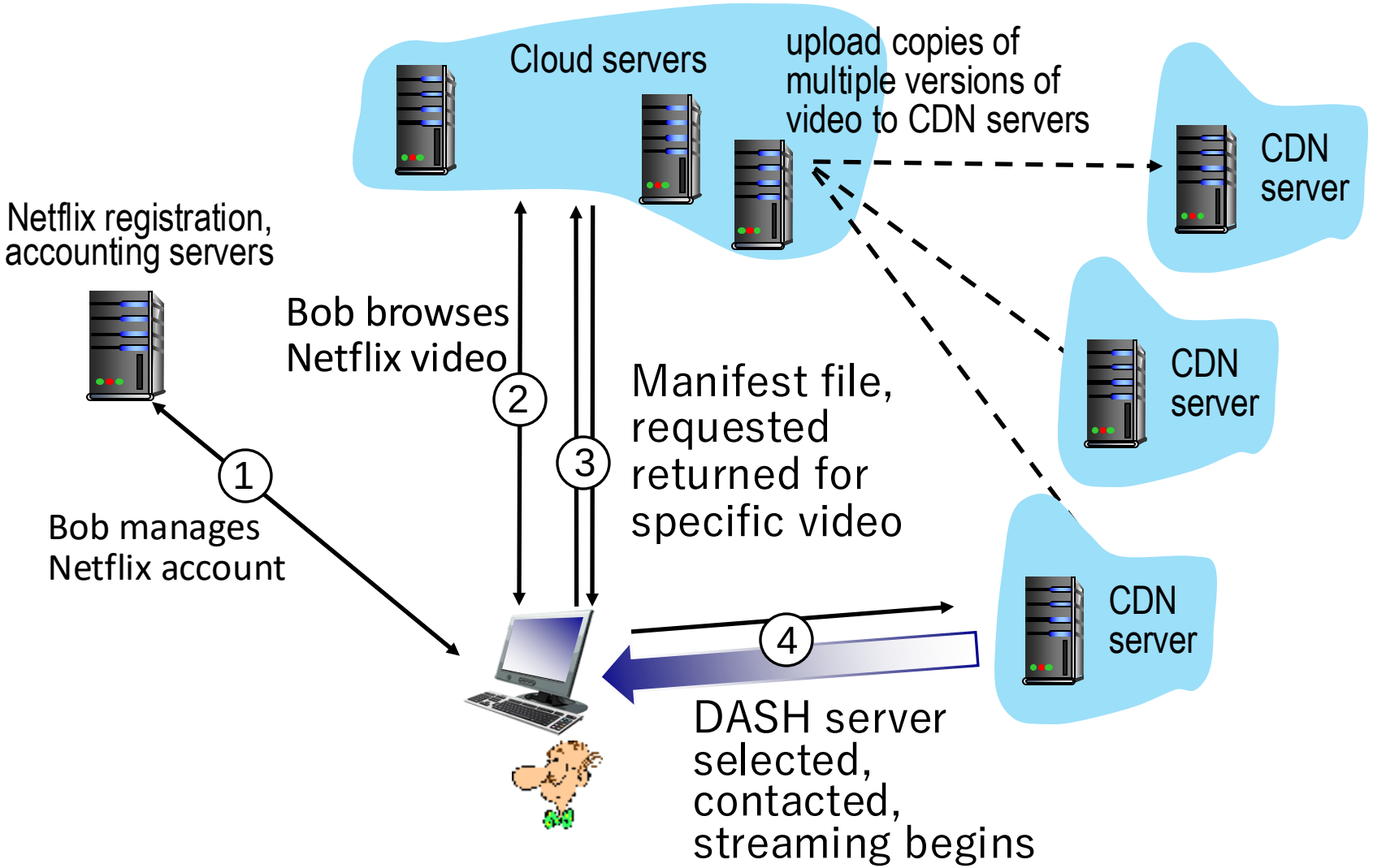
Case study: Netflix



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Application Layer Summary

our study of network application layer is now complete!

- application architectures
  - client-server
  - P2P
- application service requirements:
  - reliability, bandwidth, delay
- Internet transport service model
  - connection-oriented, reliable: TCP
  - unreliable, datagrams: UDP

- specific protocols:
  - HTTP
  - SMTP, POP3, IMAP
  - DNS

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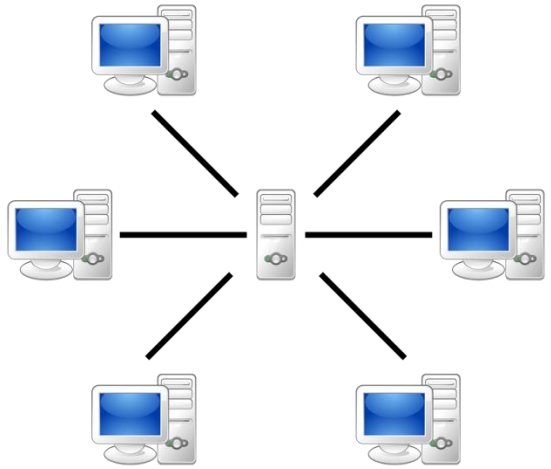
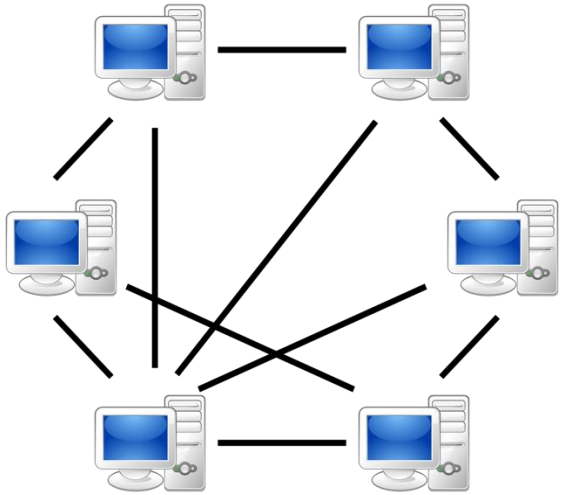
Application architectures

Two main architectures for the application layer have been developed:



Client-server architecture



Peer-to-Peer (P2P) architecture

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What transport service does an app need?

reliable data transfer

- some apps (e.g., file transfer, web transactions) require 100% reliable data transfer
- other apps (e.g., audio) can tolerate some loss

timing

- some apps (e.g., Internet telephony, interactive games) require low delay to be "effective"

throughput

- some apps (e.g., multimedia) require minimum amount of throughput to be "effective"
- other apps ("elastic apps") make use of whatever throughput they get

security

- encryption, data integrity, …

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Internet transport protocols services

TCP (Transmission Control Protocol):

- *reliable transport* between sending and receiving process
- *flow control:* sender won't overwhelm receiver
- *congestion control:* throttle sender when network overloaded
- *connection-oriented:* setup required between client and server processes
- *does not provide:* timing, minimum throughput guarantee, security

UDP (User Datagram Protocol):

- *unreliable data transfer* between sending and receiving process
- *does not provide:* reliability, flow control, congestion control, timing, throughput guarantee, security, or connection setup

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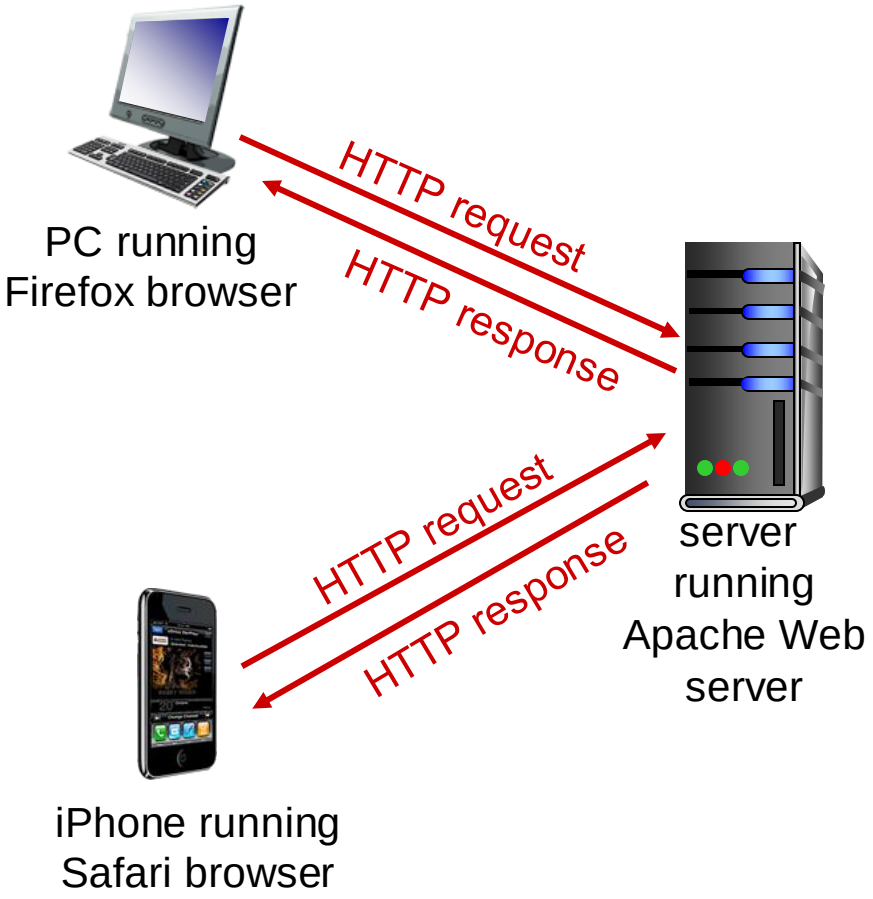
HTTP overview

HTTP: hypertext transfer protocol

- Web's application layer protocol
- client/server model
  - *client:* browser that requests, receives, (using HTTP protocol) and "displays" Web objects
  - *server:* Web server sends (using HTTP protocol) objects in response to requests



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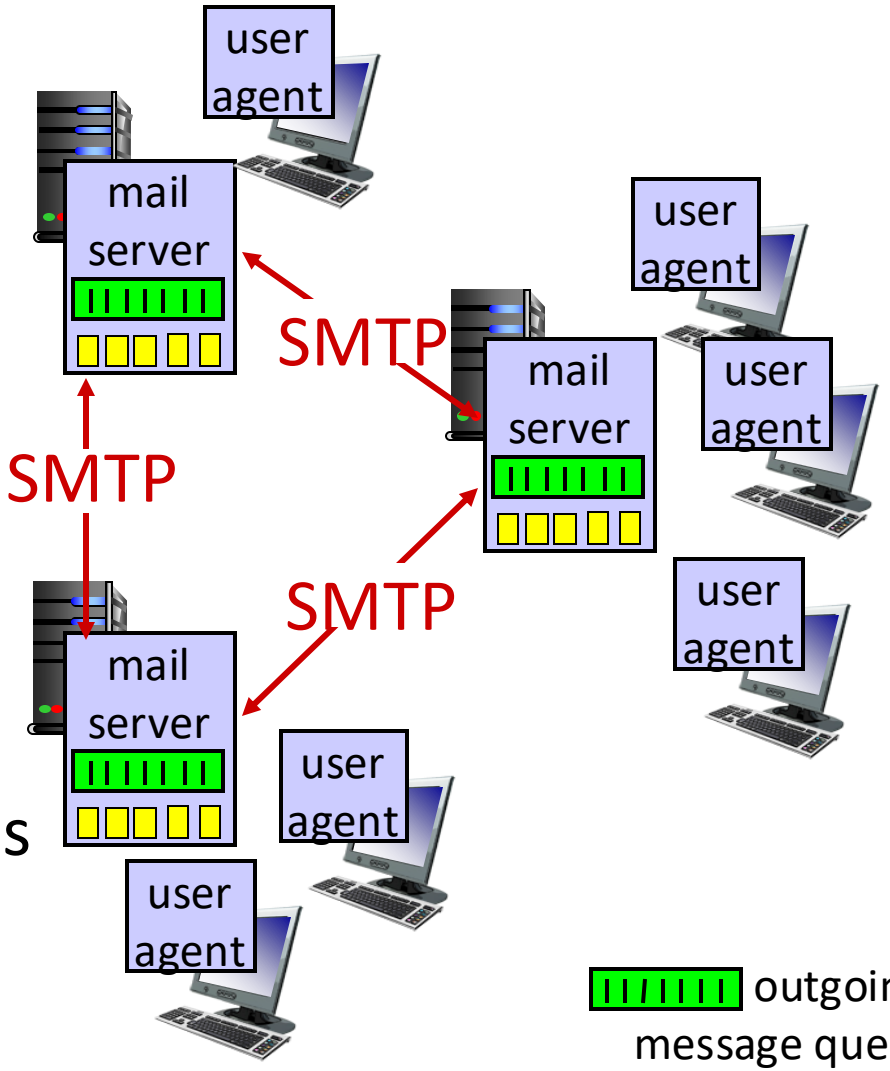
E-mail

Three major components:

- user agents
- mail servers
- simple mail transfer protocol: SMTP

User Agent

- a.k.a. “mail reader”
- composing, editing, reading mail messages
- e.g., Outlook, iPhone mail client
- outgoing, incoming messages stored on server



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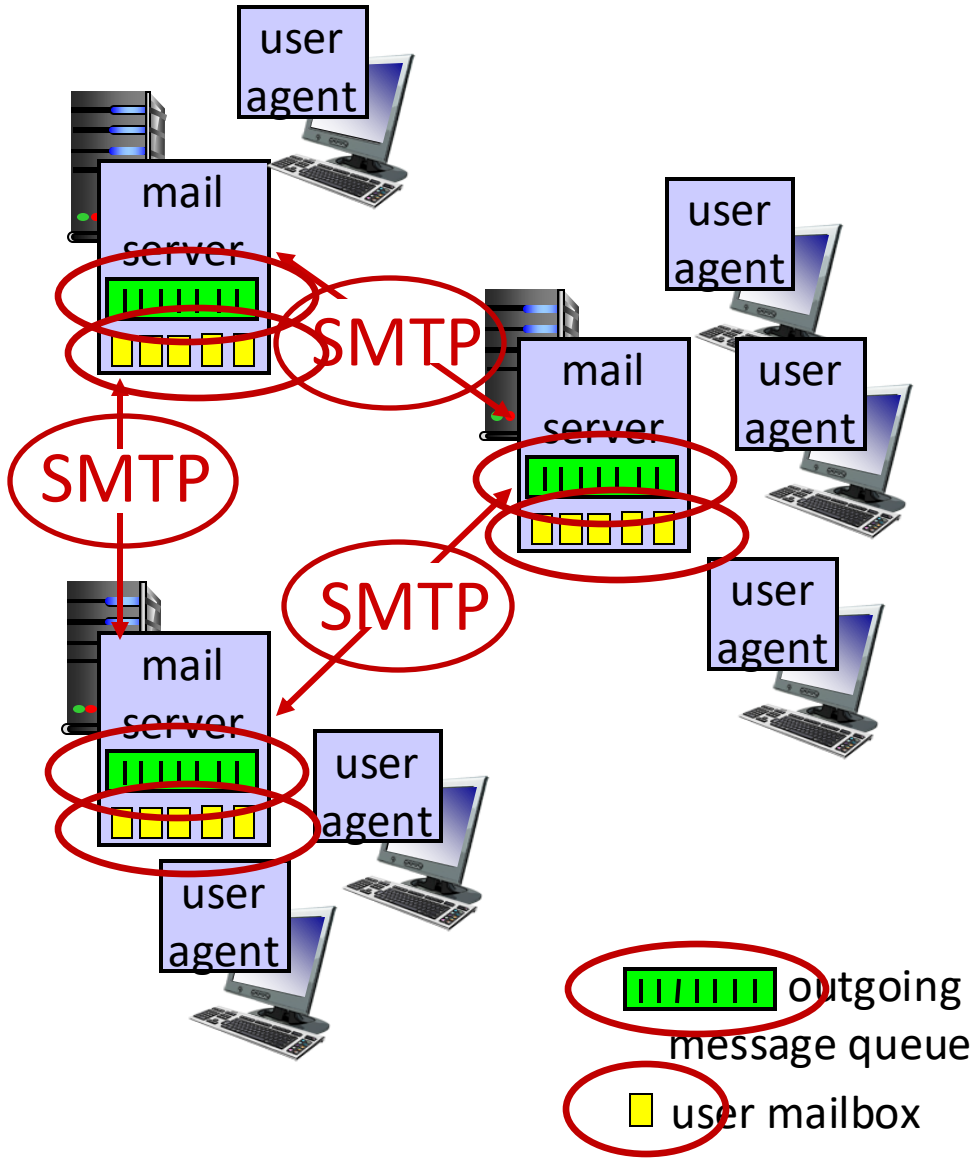
E-mail: mail servers

mail servers:

- *mailbox* contains incoming messages for user
- *message queue* of outgoing (to be sent) mail messages

SMTP protocol between mail servers to send email messages

- client: sending mail server
- "server": receiving mail server



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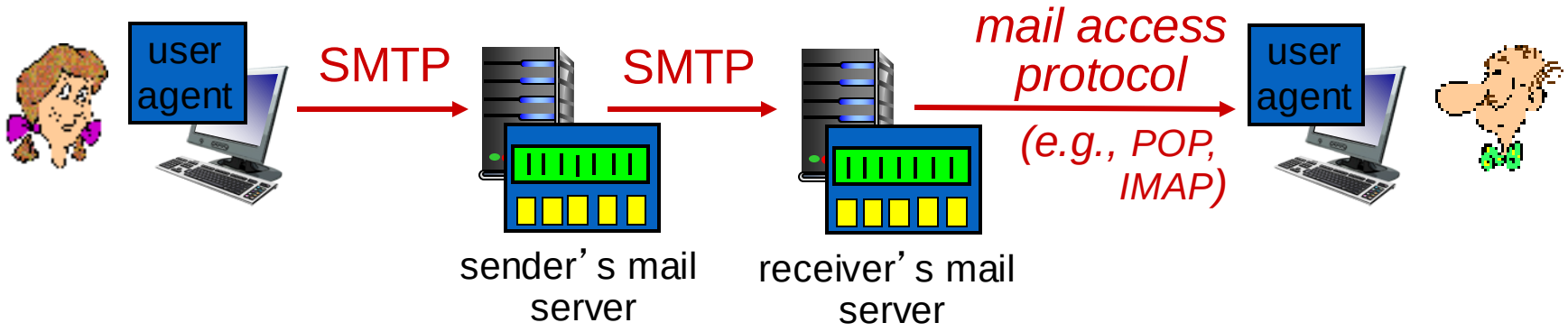
Mail access protocols



- SMTP: delivery/storage to receiver's server
- mail access protocol: retrieval from server
  - POP: Post Office Protocol: authorization, download
  - IMAP: Internet Mail Access Protocol: more features, including manipulation of stored messages on server
  - HTTP: gmail, Hotmail, Yahoo! Mail, etc.

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DNS: domain name system

people: many identifiers:

- Name, passport #

Internet hosts, routers:

- IP address (32 bit) - used for addressing by routers
- IP address looks like 127.7.106.83
- Hierarchical: Going from left to right gives more and more specific information about the location
- "name", e.g., www.yahoo.com - used by humans

Q: how to map between IP address and name, and vice versa ?

Domain Name System:

Service that translates hostnames to IP addresses

- *distributed database* implemented in hierarchy of many *name servers*
- *application-layer protocol:* hosts, name servers communicate to *resolve* names (address/name translation)

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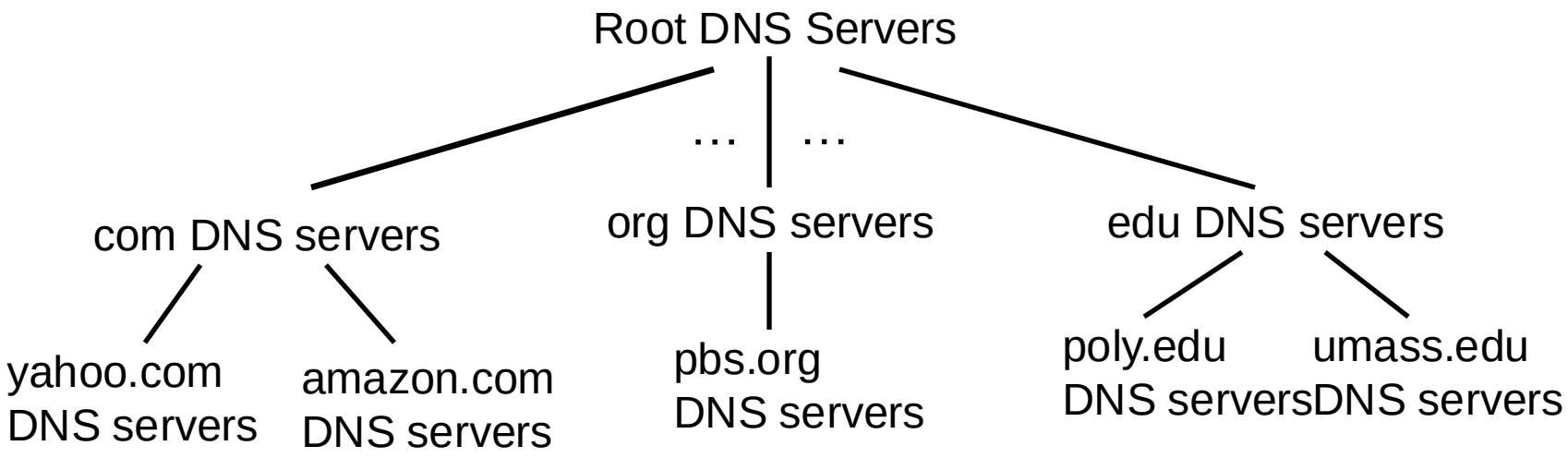
DNS: a distributed, hierarchical database



client wants IP for www.amazon.com; 1st approximation:

- client queries root server to find com DNS server
- client queries .com DNS server to get amazon.com DNS server
- client queries amazon.com DNS server to get IP address for www.amazon.com

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Today's lecture

1. video streaming and content distribution networks (CDNs)
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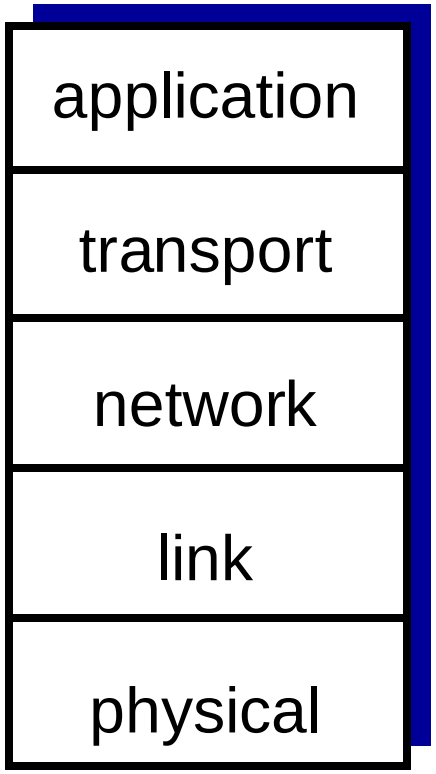
Internet protocol stack

- *application:* supporting network applications
  - FTP, SMTP, HTTP
- *transport:* process-process data transfer
  - TCP, UDP
- *network:* routing of datagrams from source to destination
  - IP, routing protocols
- *link:* data transfer between neighboring network elements
  - Ethernet, 802.111 (WiFi), PPP
- *physical:* bits "on the wire"



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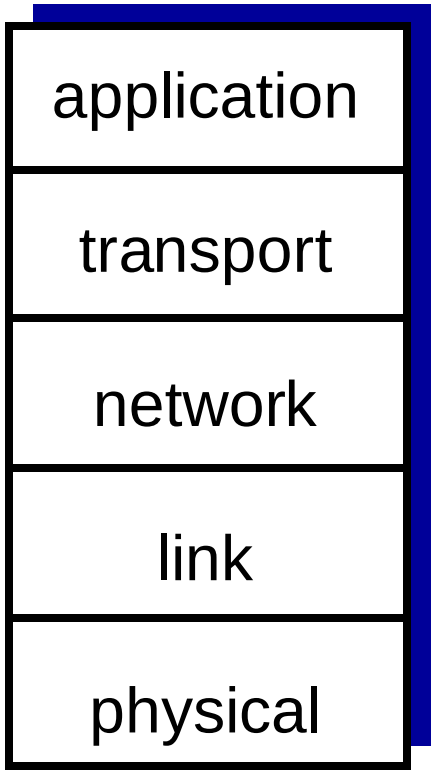
Packets, Segments, Datagrams, etc.

- The application layer gives data (of any size) to the underlying layers.
- To send the data it needs to be broken into smaller chunks of data (packets)
- Each layer adds additional header information
- We therefore use different terms for packets at different layers:
  - **Packet** is the general term for a chunk of data transmitted through the network
  - **Segment** is used for the data together with the header data from the transport layer
  - **Datagram** is used for the segment with additional information from the network layer.
  - **Frame** is used on the link layer
  - **Bits** are transferred on the physical layer



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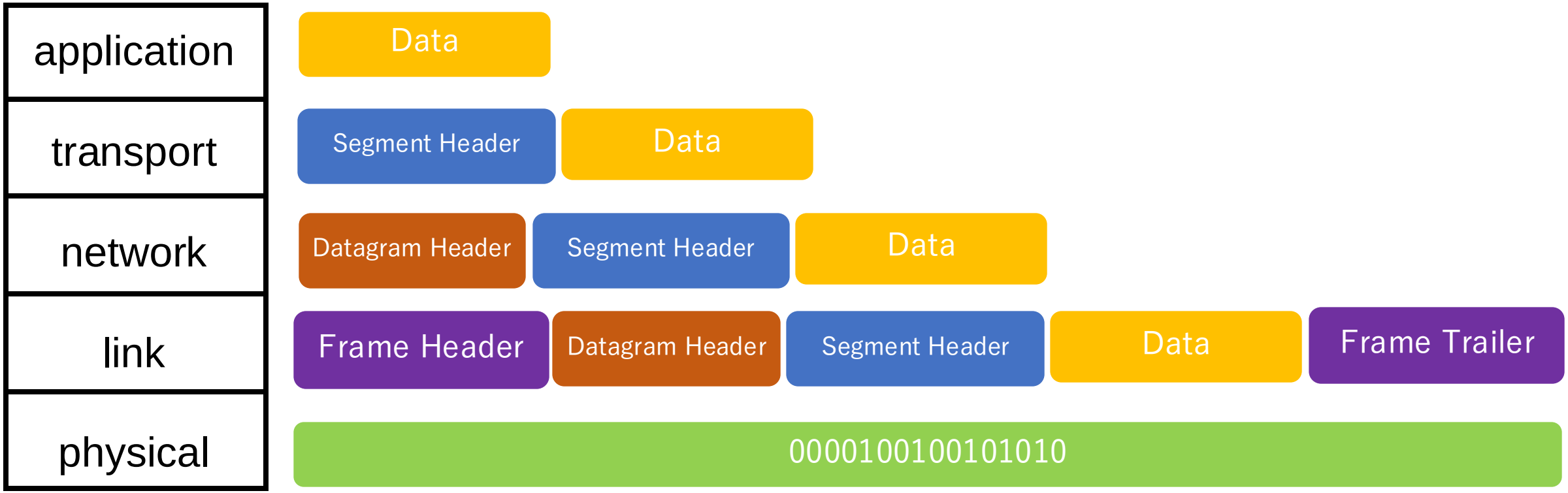
Packets, Segments, Datagrams, etc.



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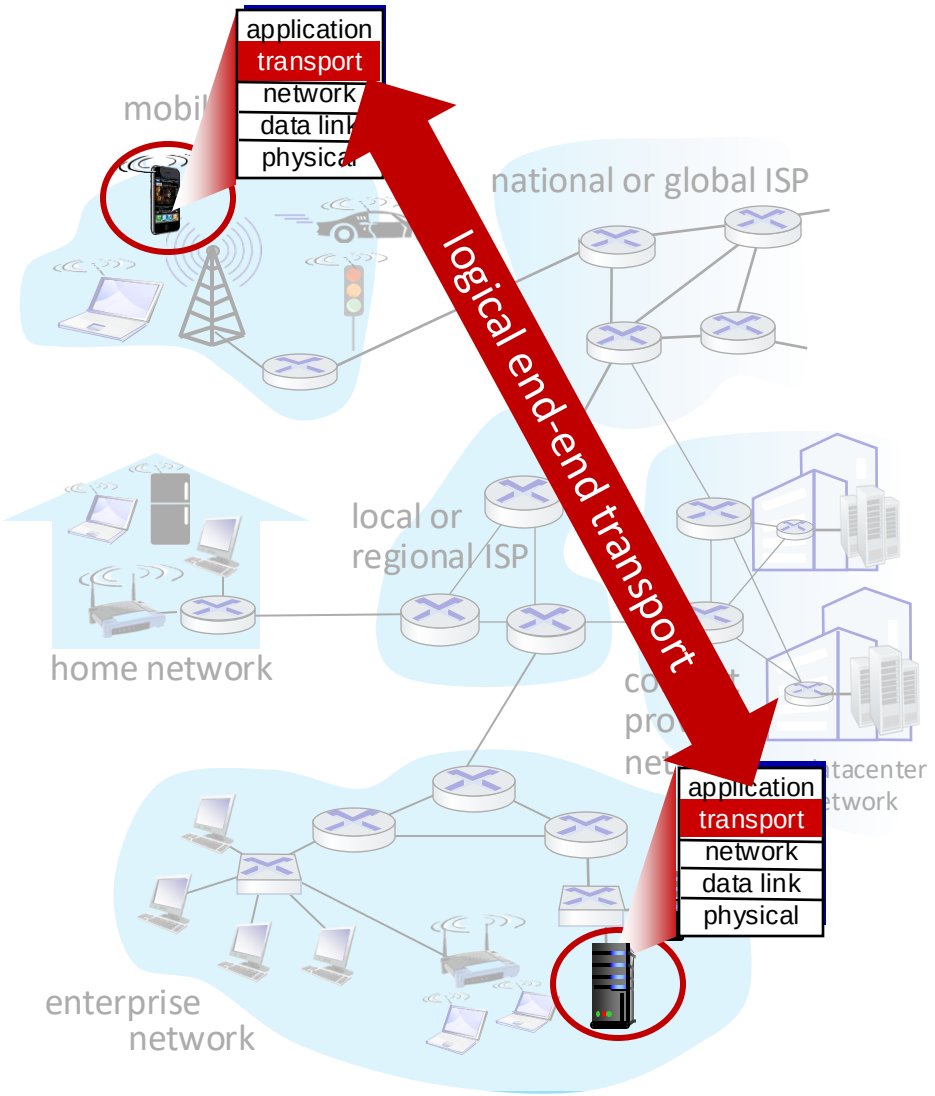
Two principal Internet transport protocols

- **TCP:** Transmission Control Protocol
  - reliable, in-order delivery
  - congestion control
  - flow control
  - connection setup
- **UDP:** User Datagram Protocol
  - unreliable, unordered delivery
  - extension of “best-effort” IP
- services not available:
  - delay guarantees
  - bandwidth guarantees



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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

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Overview

3.1 transport-layer services

3.2 multiplexing and demultiplexing

3.3 connectionless transport: UDP

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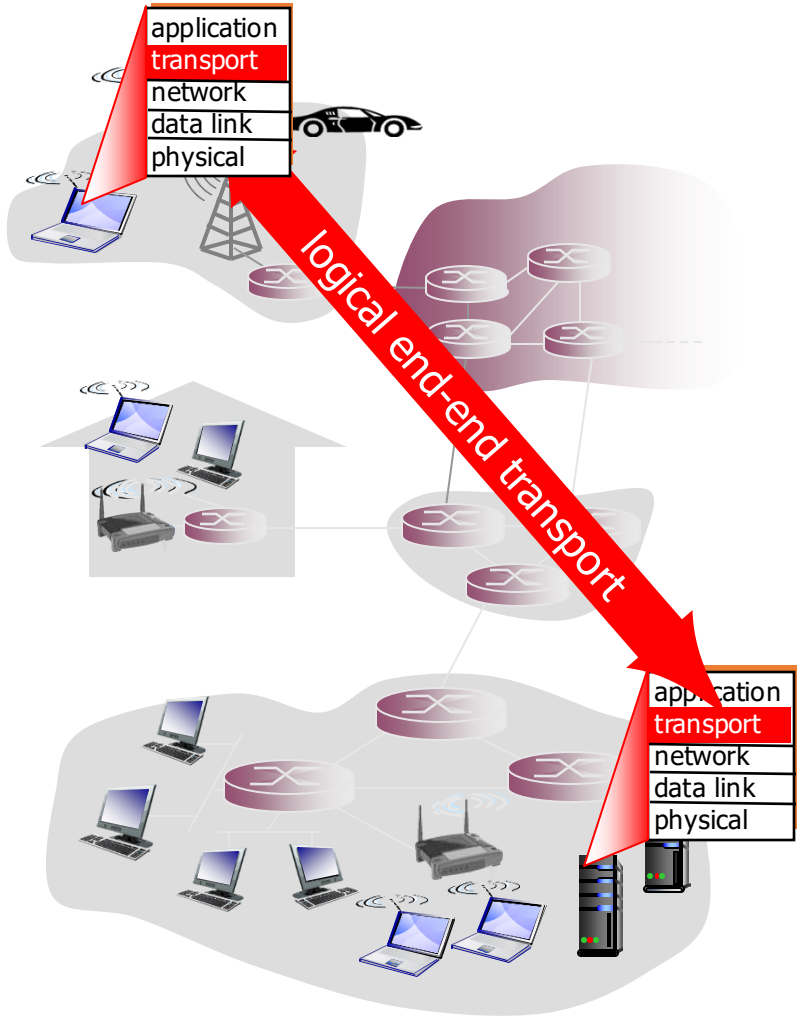
Transport services and protocols

- 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
  - receive side: reassembles segments into messages, passes to app layer
- more than one transport protocol available to apps
  - Internet: TCP and UDP



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Transport vs. network layer

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

household analogy:

12 kids in house A sending letters to 12 kids in house B:

- hosts = houses
- processes = kids
- app messages = letters in envelopes
- transport protocol = Kid in house A and kid in house B responsible for collecting and distributing letters
- network-layer protocol = postal service

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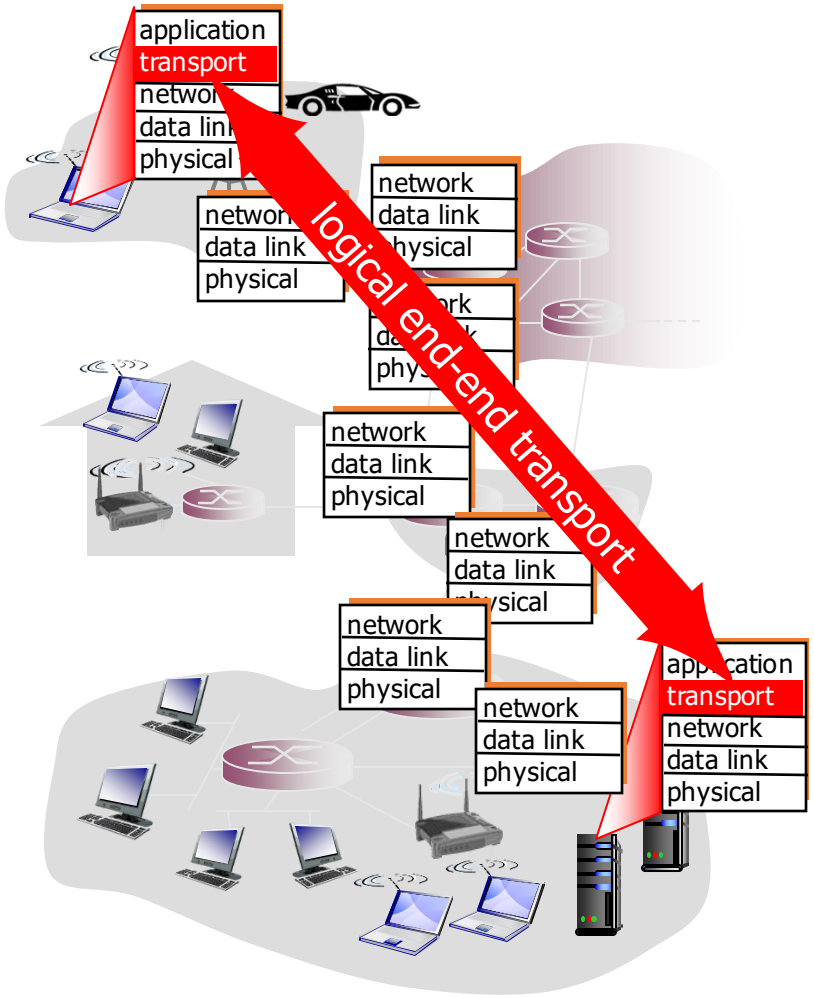
Internet transport-layer protocols

- reliable, in-order delivery (TCP)
  - congestion control
  - flow control
  - connection setup
- unreliable, unordered delivery: UDP
- services not available in both:
  - delay guarantees
  - bandwidth guarantees
  - These services are difficult to realize in packet switched networks



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Chapter 3 outline

3.1 transport-layer services

3.2 multiplexing and demultiplexing

3.3 connectionless transport: UDP

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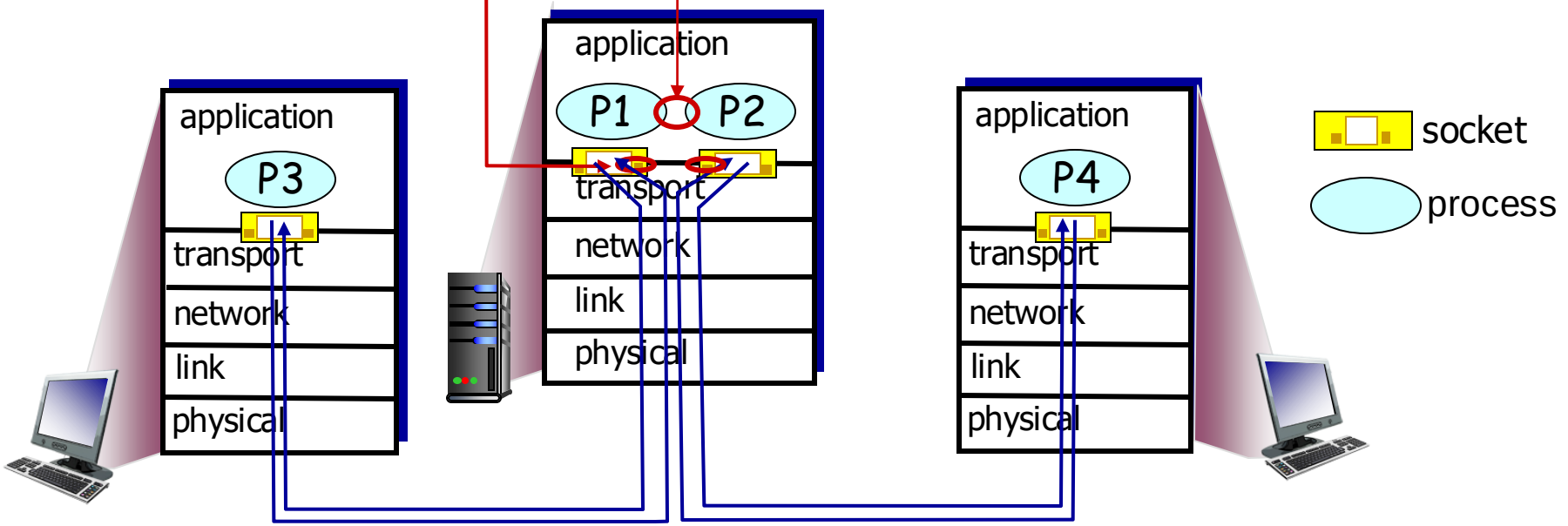
Multiplexing/demultiplexing

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



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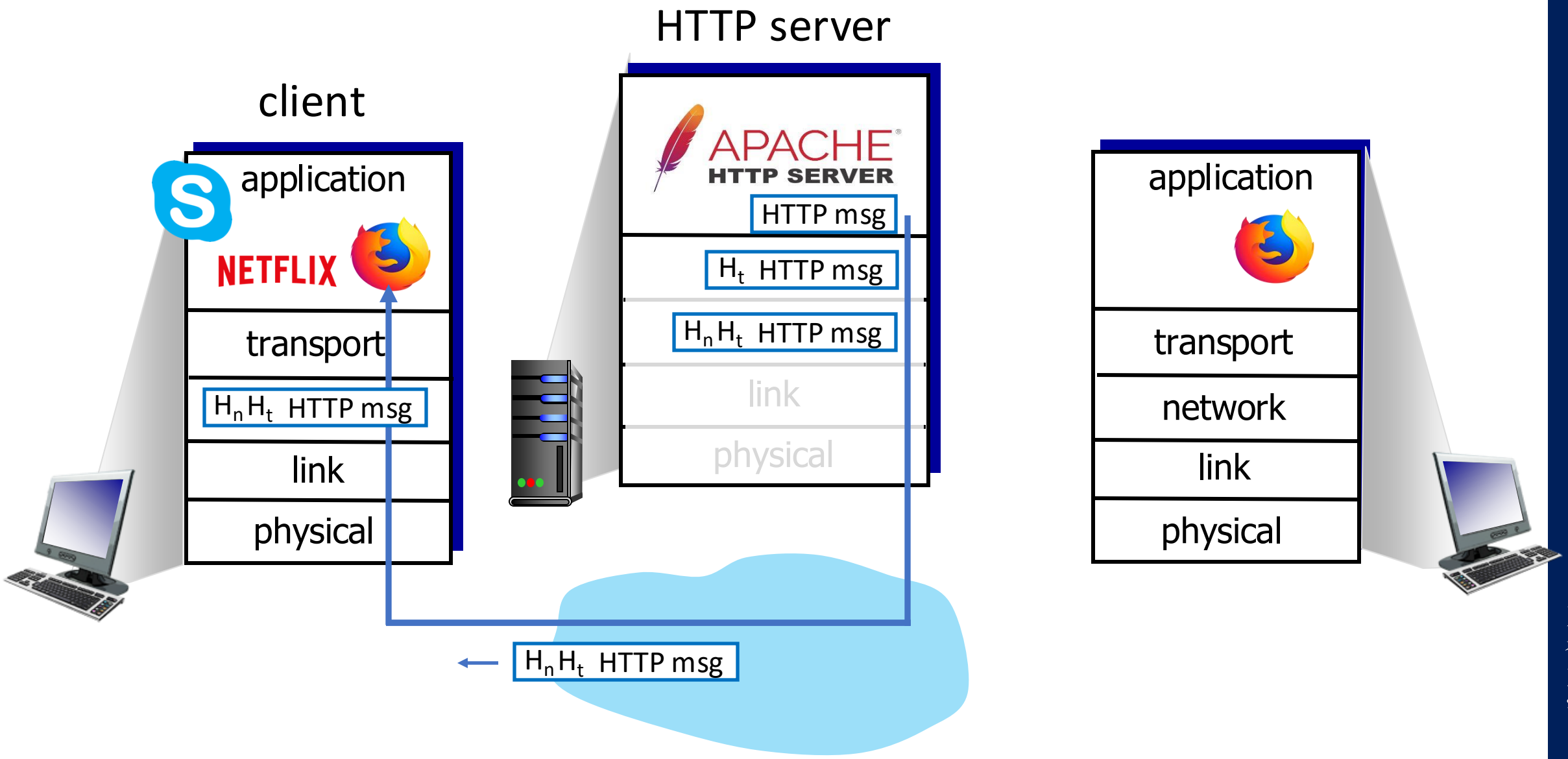
HTTP server
client
application
NETFLIX
transport
$H_n H_t$ HTTP msg
link
physical
APACHE
HTTP SERVER
HTTP msg
H_t HTTP msg
$H_n H_t$ HTTP msg
link
physical
application
transport
network
link
physical
$H_n H_t$ HTTP msg

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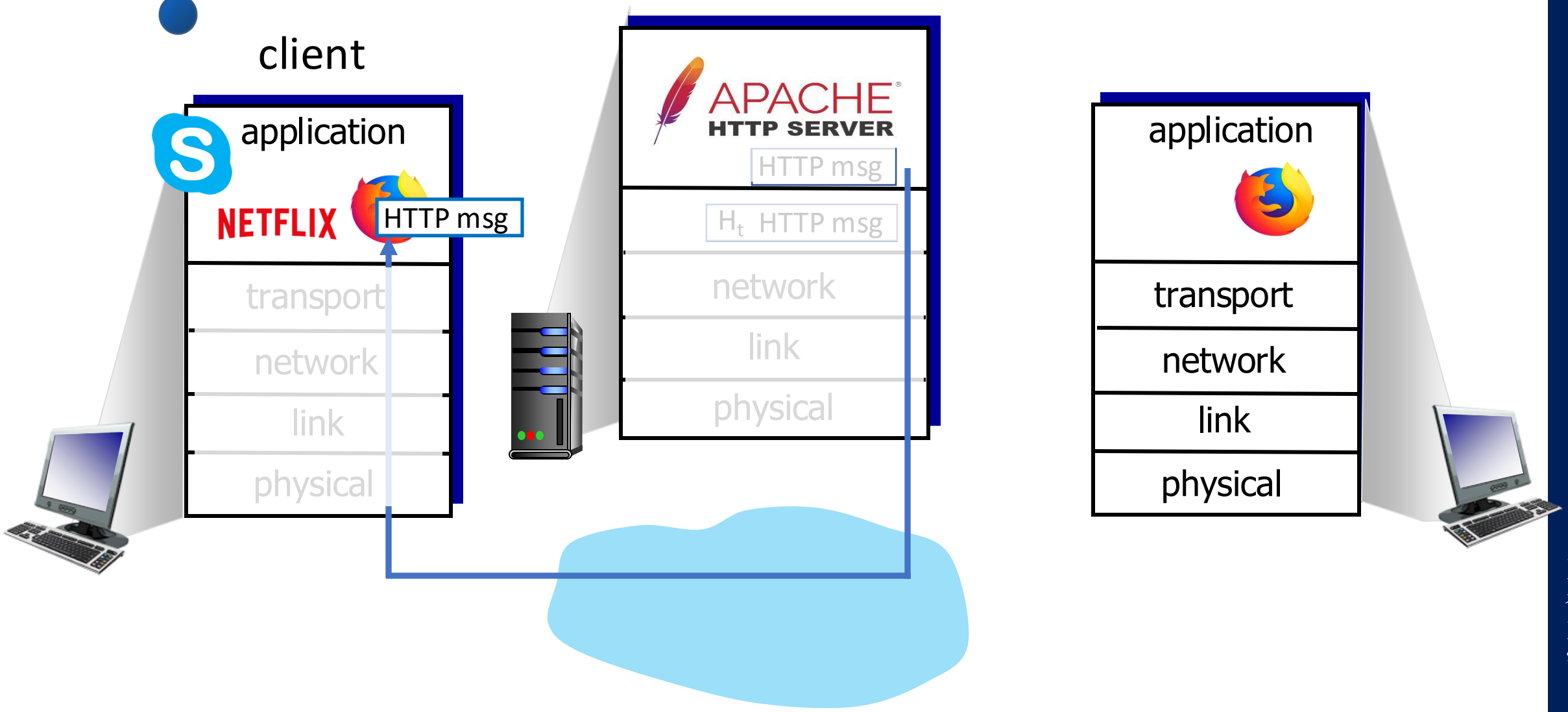
Q: how did transport layer know to deliver message to Firefox browser process rather then Netflix process or Skype process?



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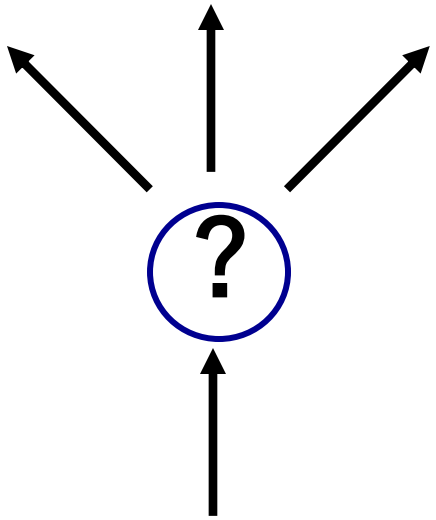

de-multiplexing

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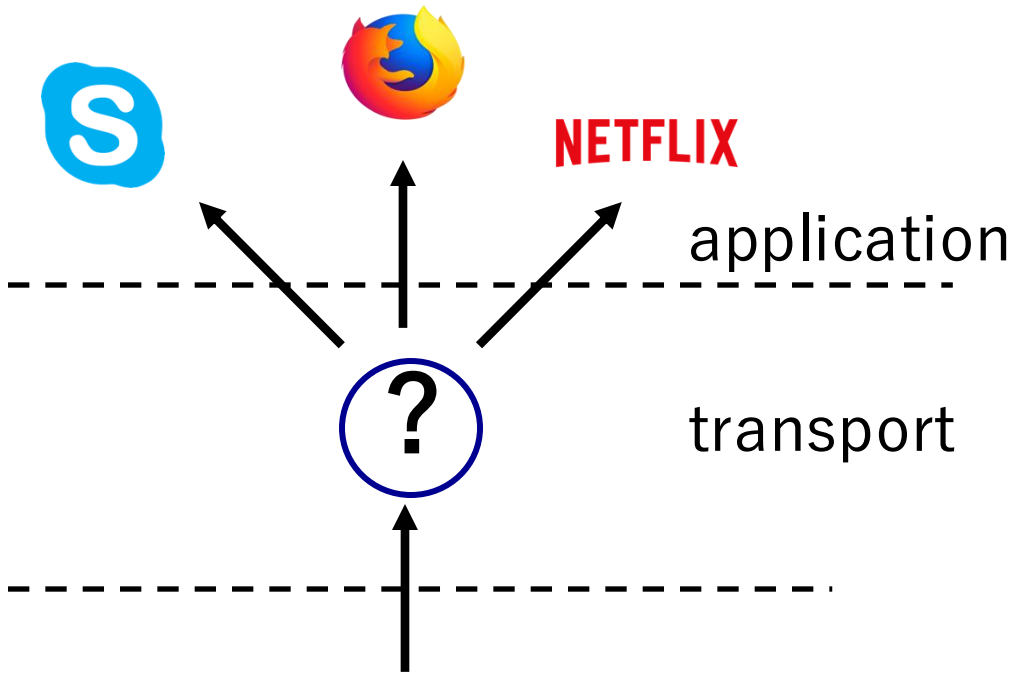

de-multiplexing

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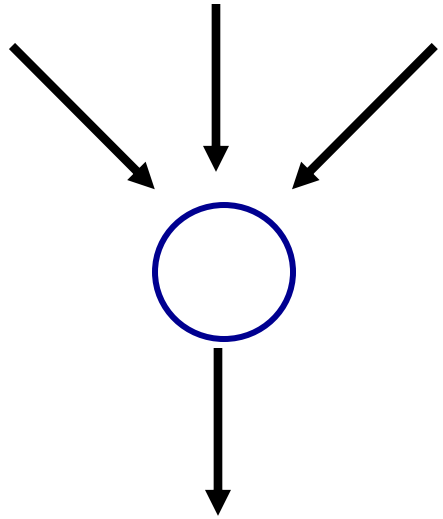

multiplexing

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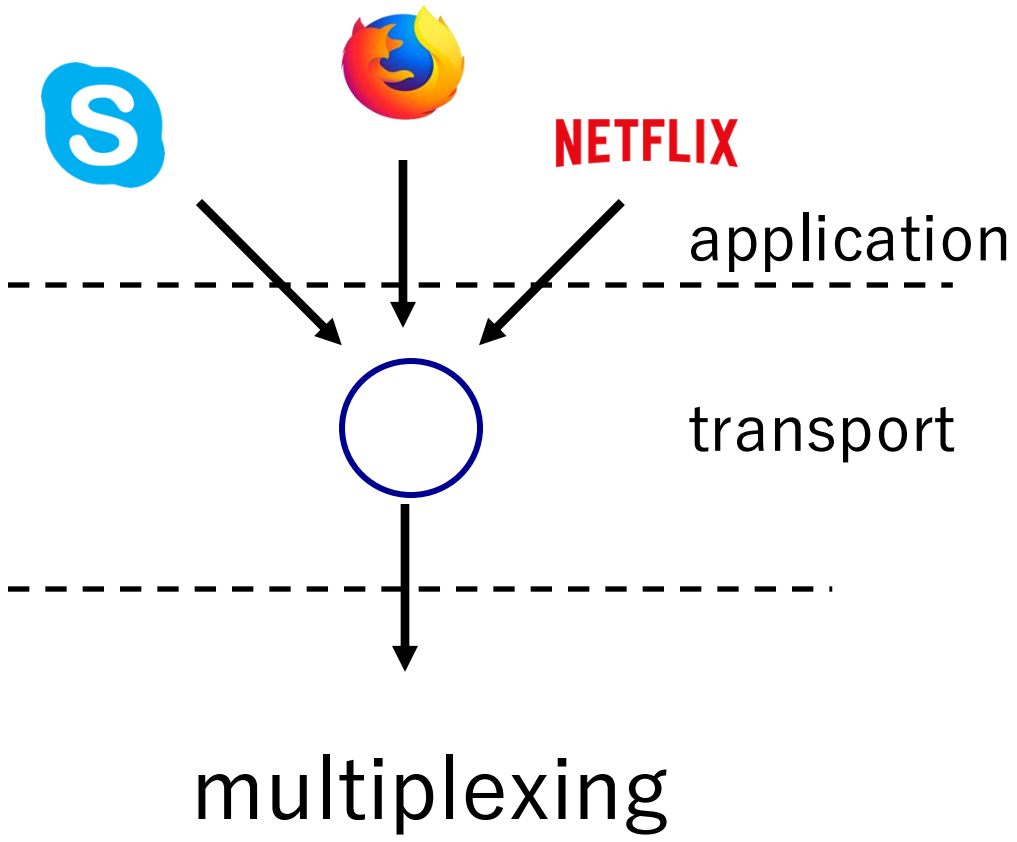

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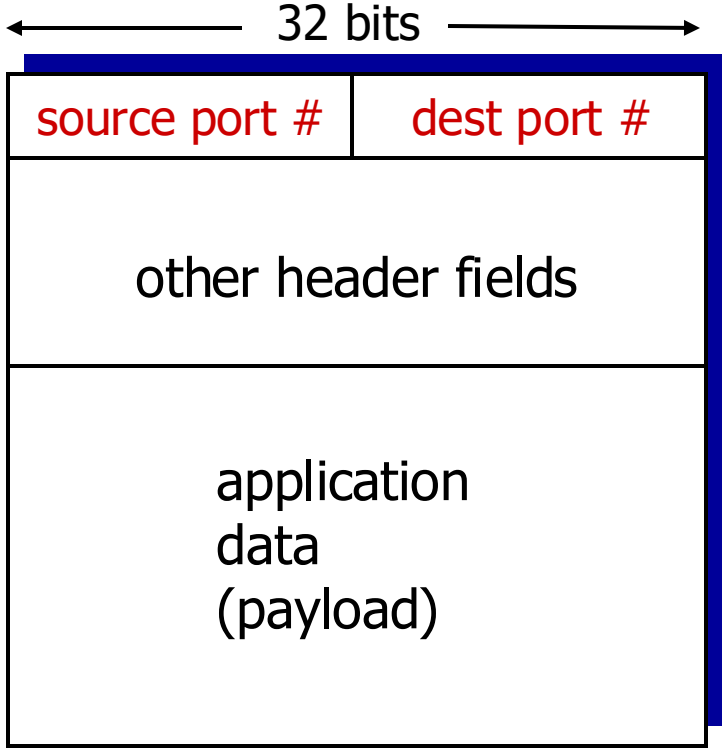
How demultiplexing works

- 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

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Connectionless demultiplexing

- created socket has host-local port number
- *recall:* when creating datagram to send into UDP socket, must specify
  - destination IP address
  - destination port number

- when host receives UDP segment:
  - checks destination port number in segment
  - directs UDP segment to socket with that port number

IP datagrams with *same dest. port number,* but different source IP addresses and/or source port numbers will be directed to *same socket* at destination

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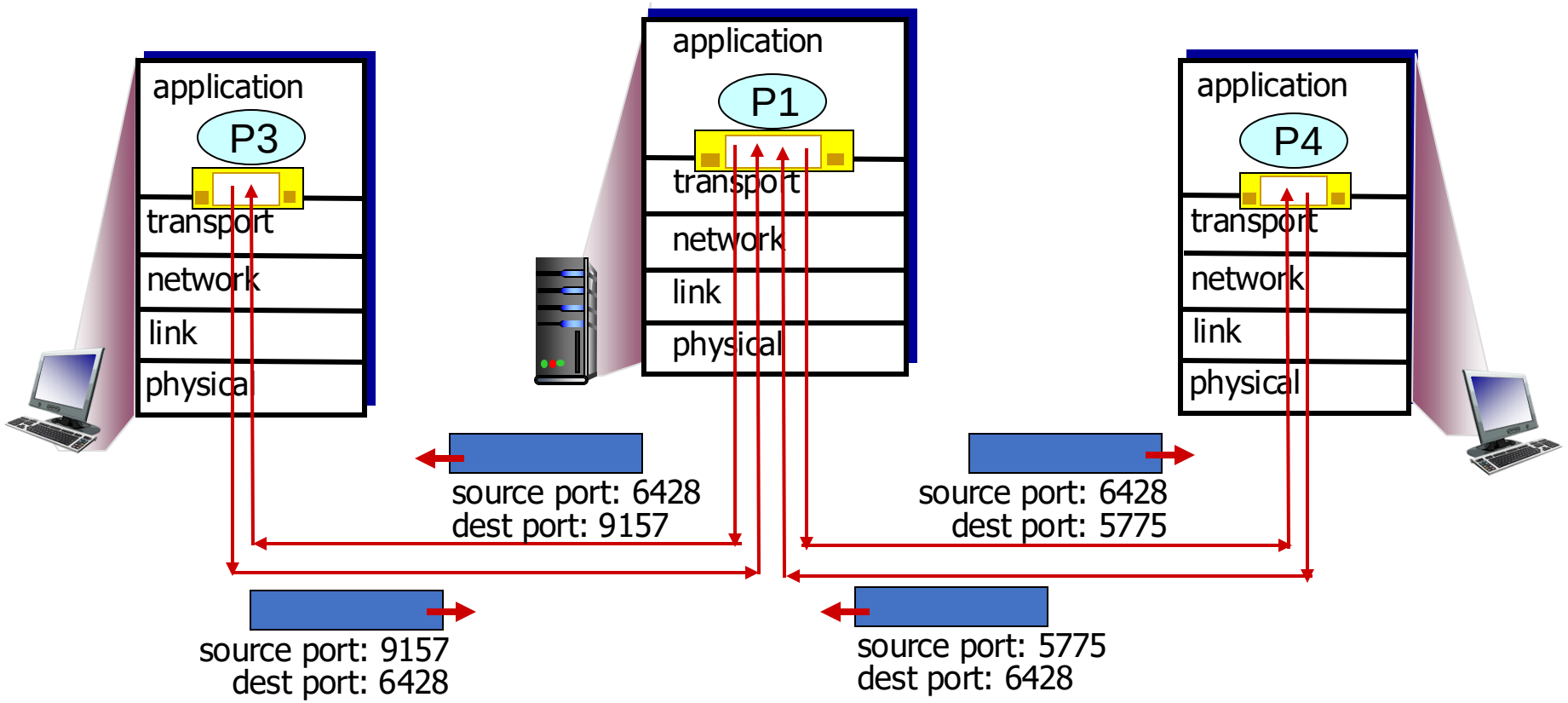
Connectionless demux: example



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Connection-oriented demux

- TCP socket identified by 4-tuple:
  - source IP address
  - source port number
  - dest IP address
  - dest port number
- demux: receiver uses all four values to direct segment to appropriate socket
- server host may support many simultaneous TCP sockets:
  - each socket identified by its own 4-tuple
- web servers have different sockets for each connecting client
  - non-persistent HTTP will have different socket for each request

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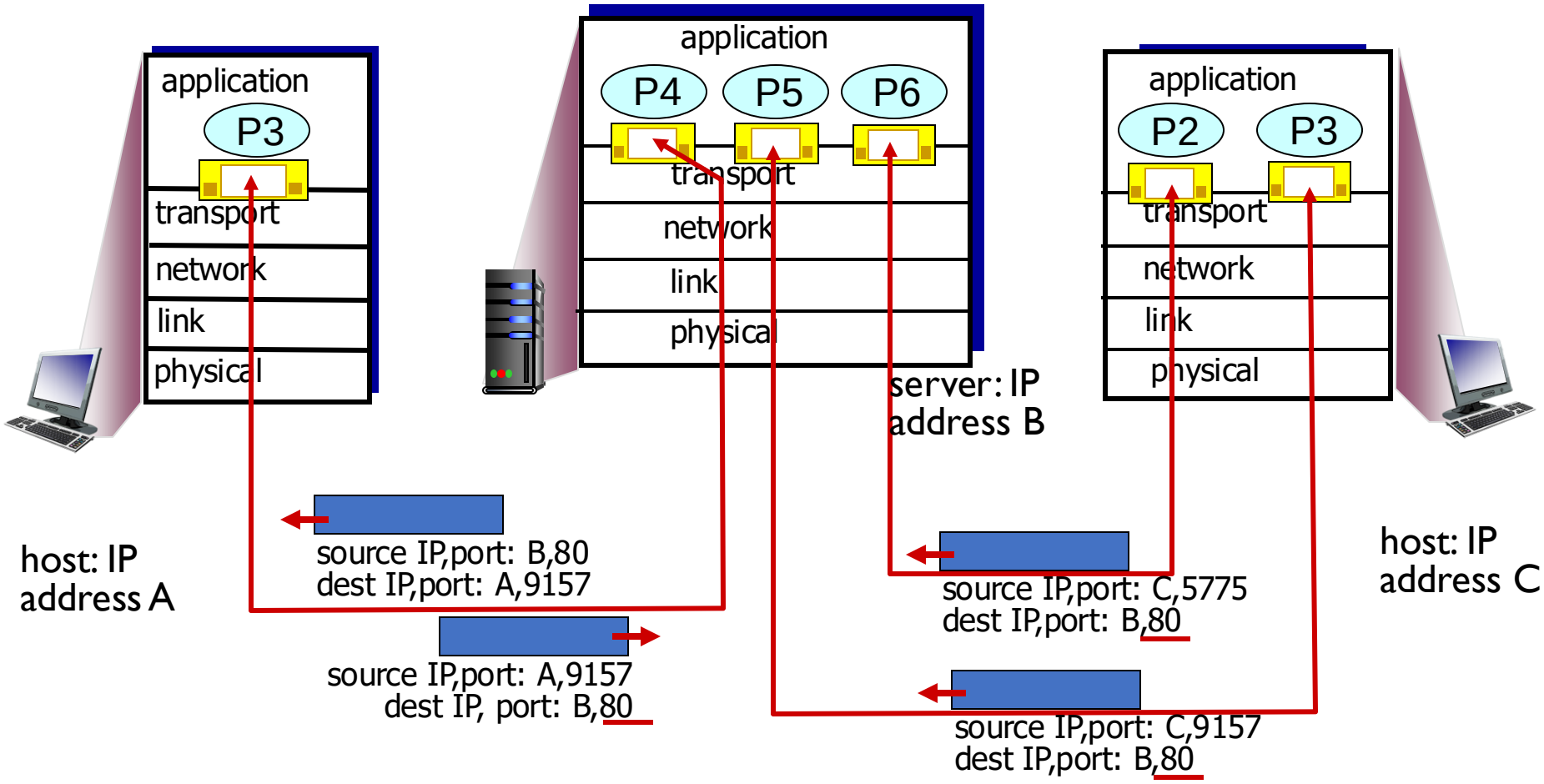
Connection-oriented demux: example



three segments, all destined to IP address: B,
dest port: 80 are demultiplexed to *different* sockets

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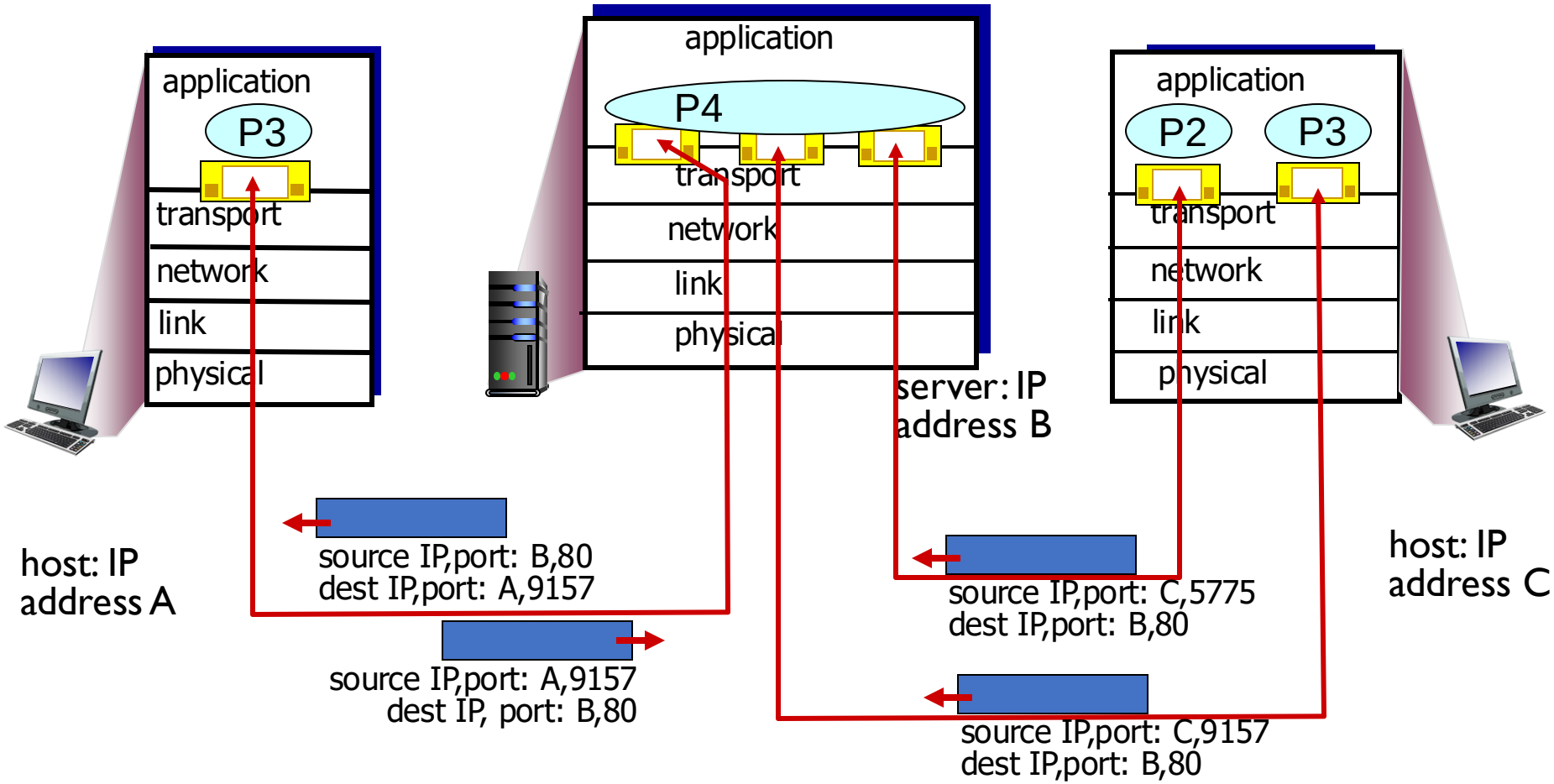
Connection-oriented demux: example



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Summary

- Multiplexing, demultiplexing: based on segment, datagram header field values
- **UDP:** demultiplexing using destination port number (only)
- **TCP:** demultiplexing using 4-tuple: source and destination IP addresses, and port numbers
- Multiplexing/demultiplexing happen at *all* layers

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Today's lecture

3.1 transport-layer services

3.2 multiplexing and demultiplexing

3.3 connectionless transport: UDP

3.4 principles of reliable data transfer

3.5 connection-oriented transport: TCP

- segment structure
- reliable data transfer
- flow control
- connection management

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UDP: User Datagram Protocol

- Very simple Internet transport protocol
- "best effort" service, UDP segments may be:
  - lost
  - delivered out-of-order to app
- *connectionless:*
  - no handshaking between UDP sender, receiver
  - each UDP segment handled independently of others

- UDP use:
  - streaming multimedia apps (loss tolerant, rate sensitive)
  - DNS
  - SNMP (used to manage network printers etc.)
- reliable transfer over UDP:
  - add reliability at application layer
  - application-specific error recovery!

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Advantages of UDP over TCP

- Finer application-level control over what data is sent and when.
  - Packet will be immediately passed to network layer
  - Real-time applications often require a minimum sending rate.
  - TCP not well-suited for these applications.
- No connection establishment
  - No delay to establish connection.
- No connection state
  - A server can support many more active clients when the application runs over UDP rather than TCP.
- Small packet overhead
  - TCP segment has 20 bytes of header overhead, UDP only 8 bytes.

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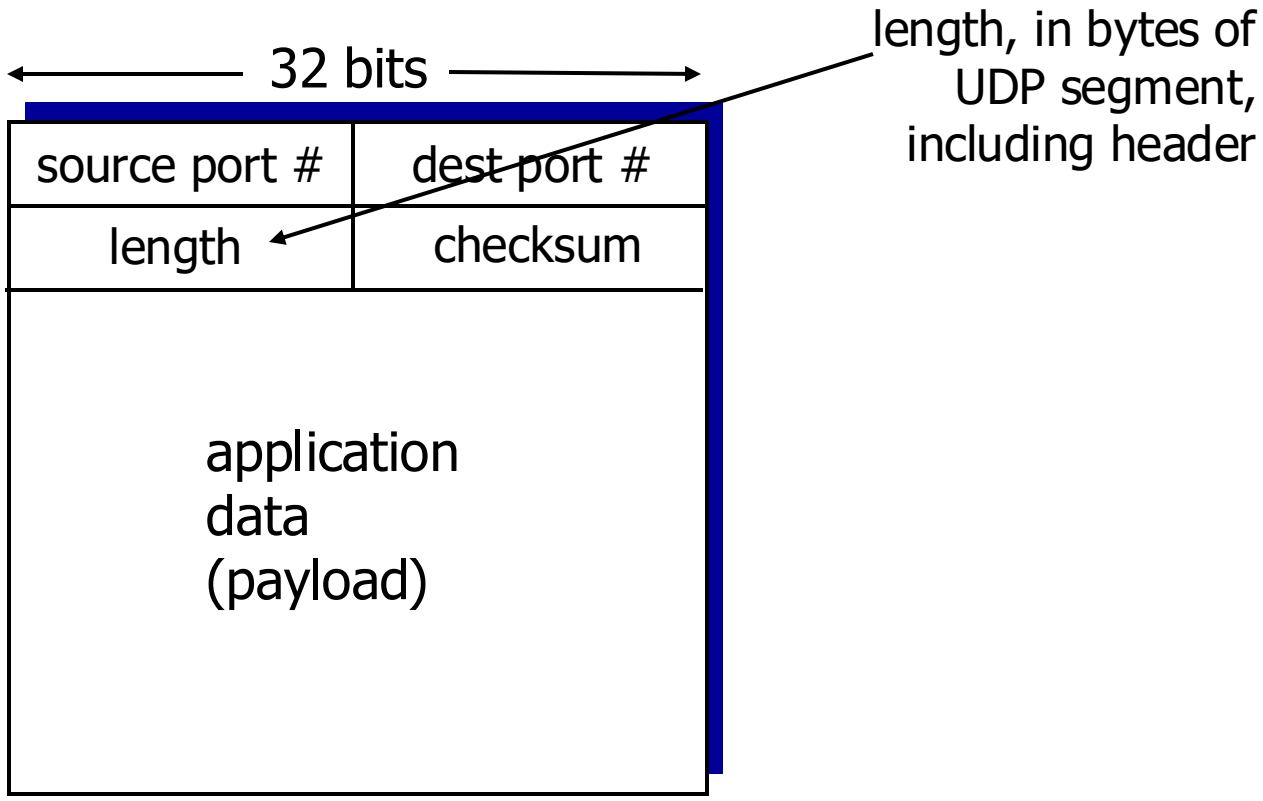
UDP: segment header



UDP segment format

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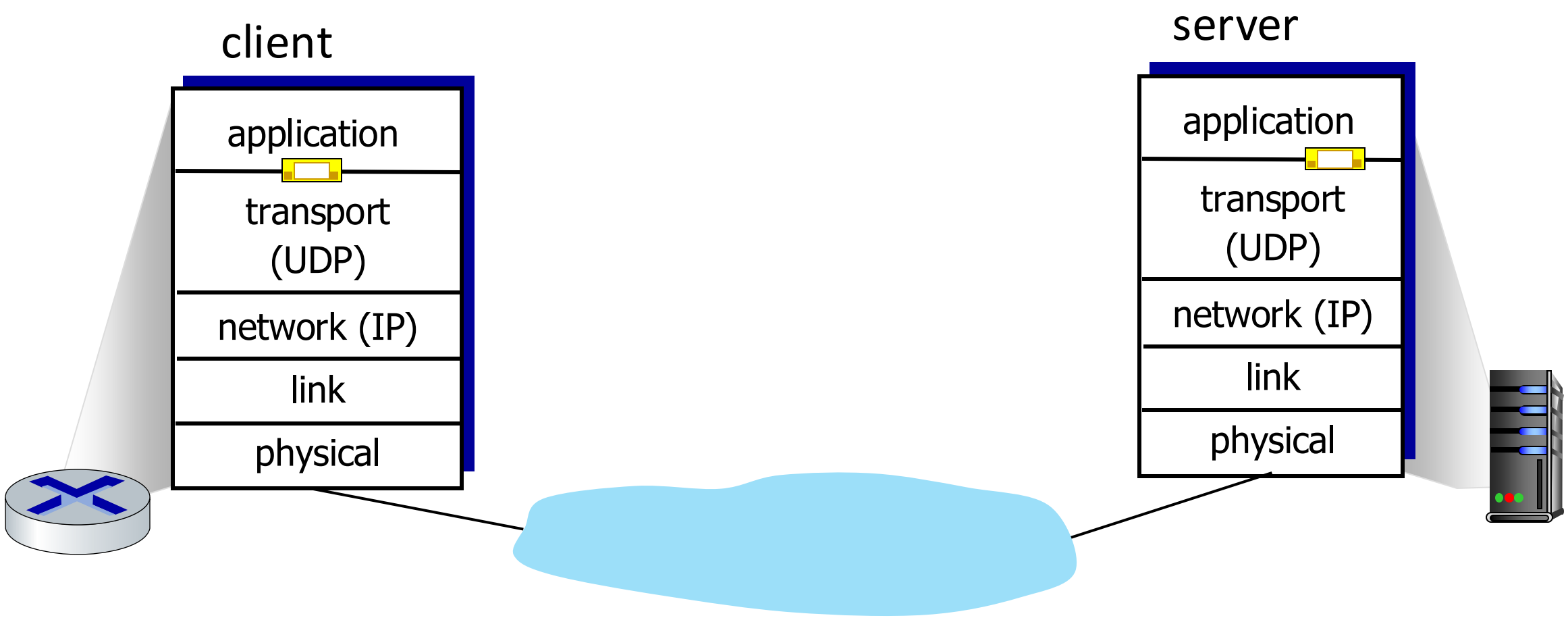
UDP: Transport Layer Actions



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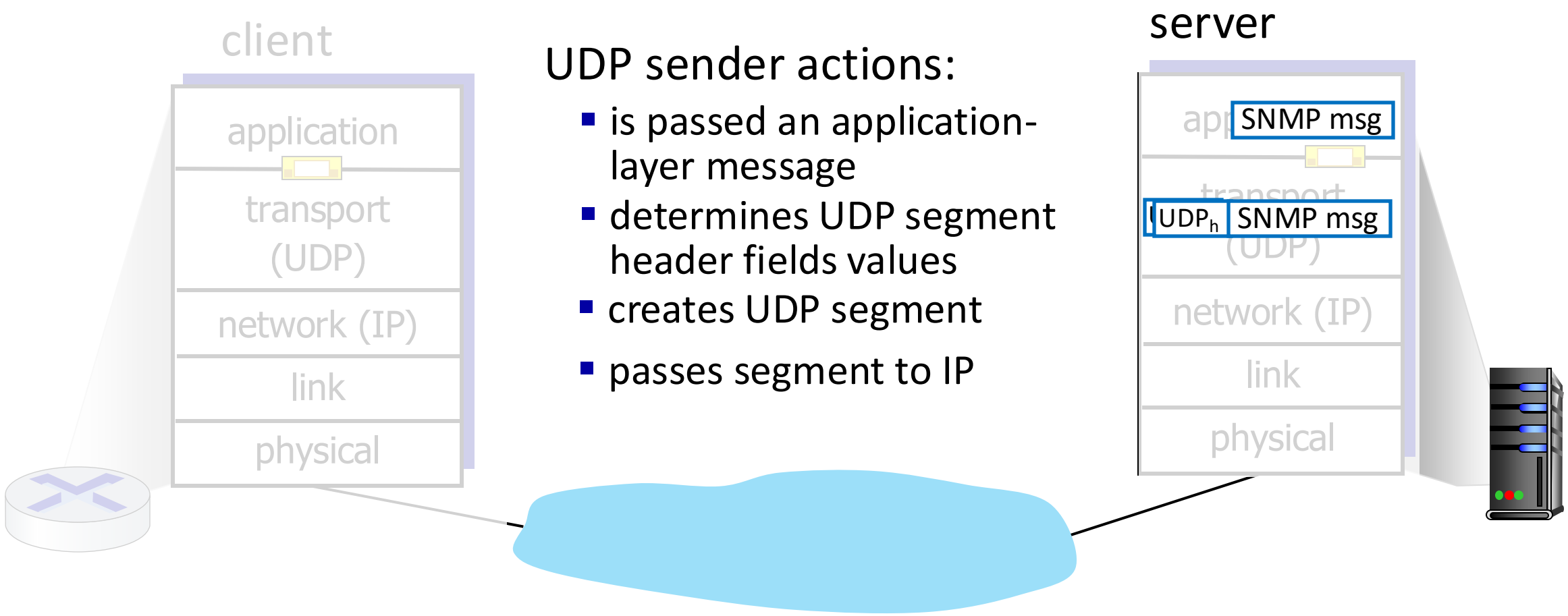
UDP: Transport Layer Actions



UDP sender actions:

- is passed an application-layer message
- determines UDP segment header fields values
- creates UDP segment
- passes segment to IP

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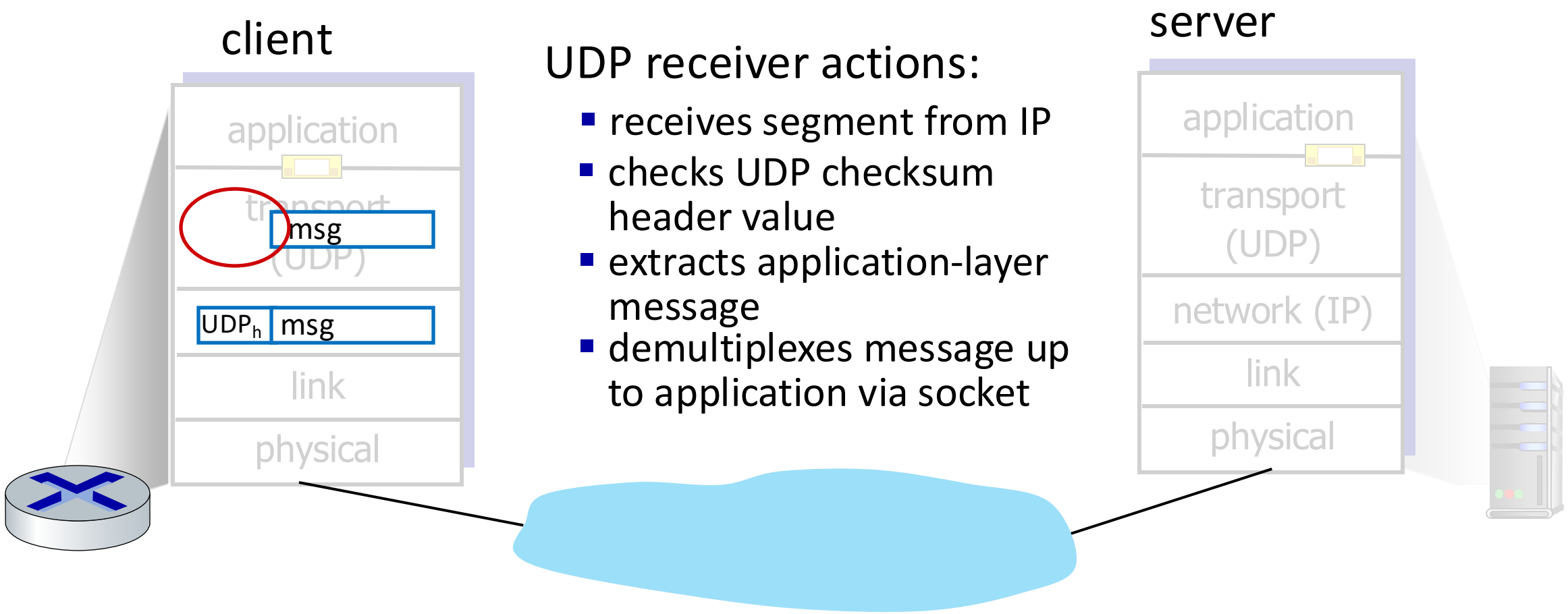
UDP: Transport Layer Actions



UDP receiver actions:

- receives segment from IP
- checks UDP checksum header value
- extracts application-layer message
- demultiplexes message up to application via socket

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UDP checksum

Goal: detect "errors" (e.g., flipped bits) in transmitted segment

sender:

- treat segment contents, including header fields, as sequence of 16-bit integers
- checksum: addition (one's complement sum) of segment contents
- sender puts checksum value into UDP checksum field

receiver:

- compute checksum of received segment
- check if computed checksum equals checksum field value:
  - NO - error detected
  - YES - no error detected. *But maybe errors nonetheless?*

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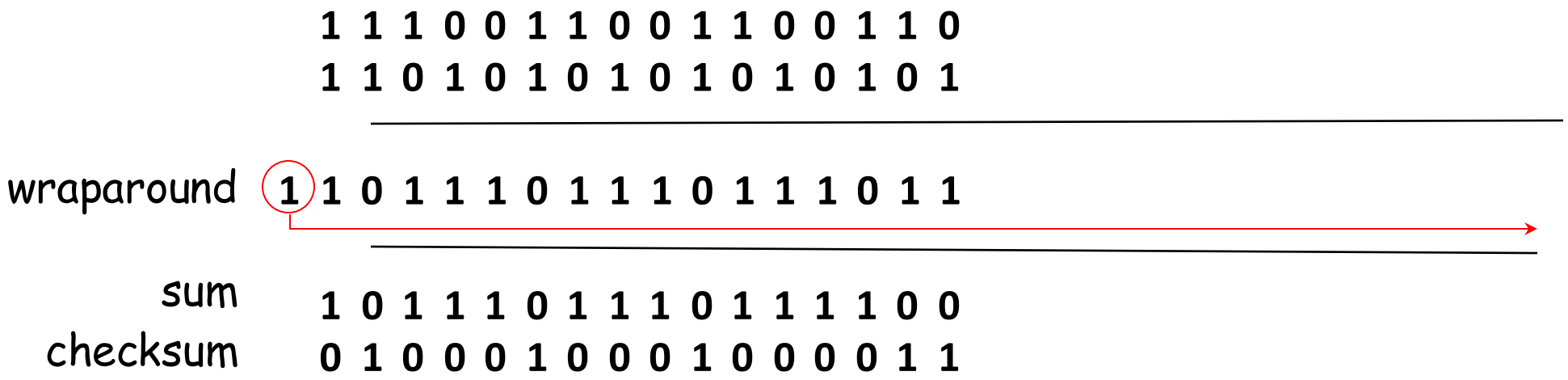
Internet checksum: example

example: add two 16-bit integers



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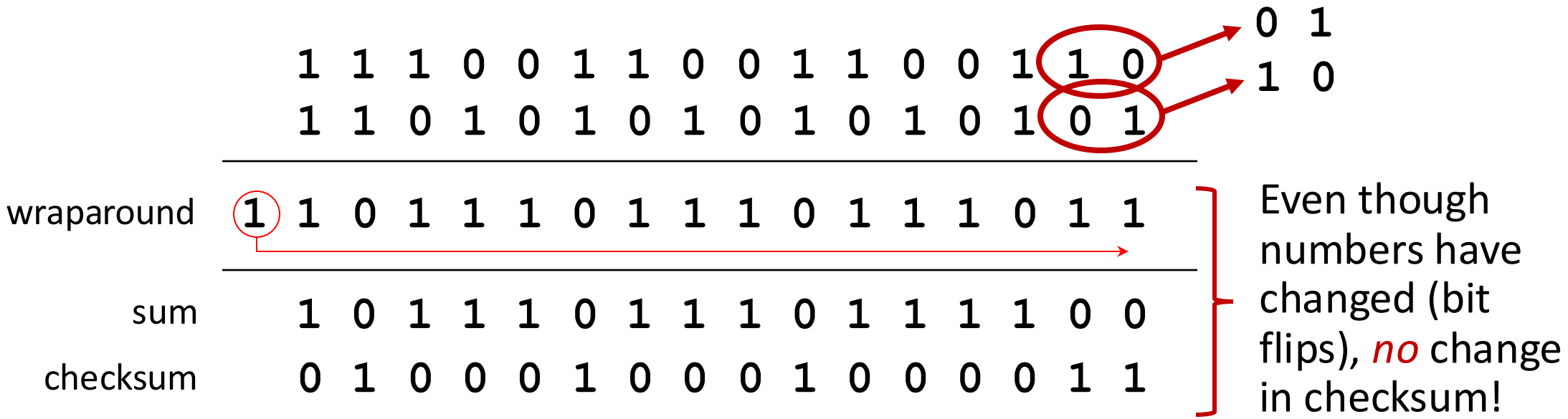
Internet checksum: weak protection!

example: add two 16-bit integers



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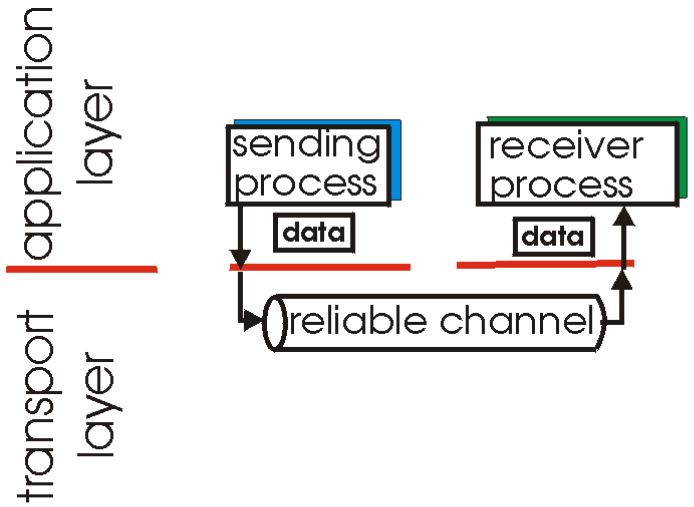
Principles of reliable data transfer

- important in application, transport, link layers



(a) provided service

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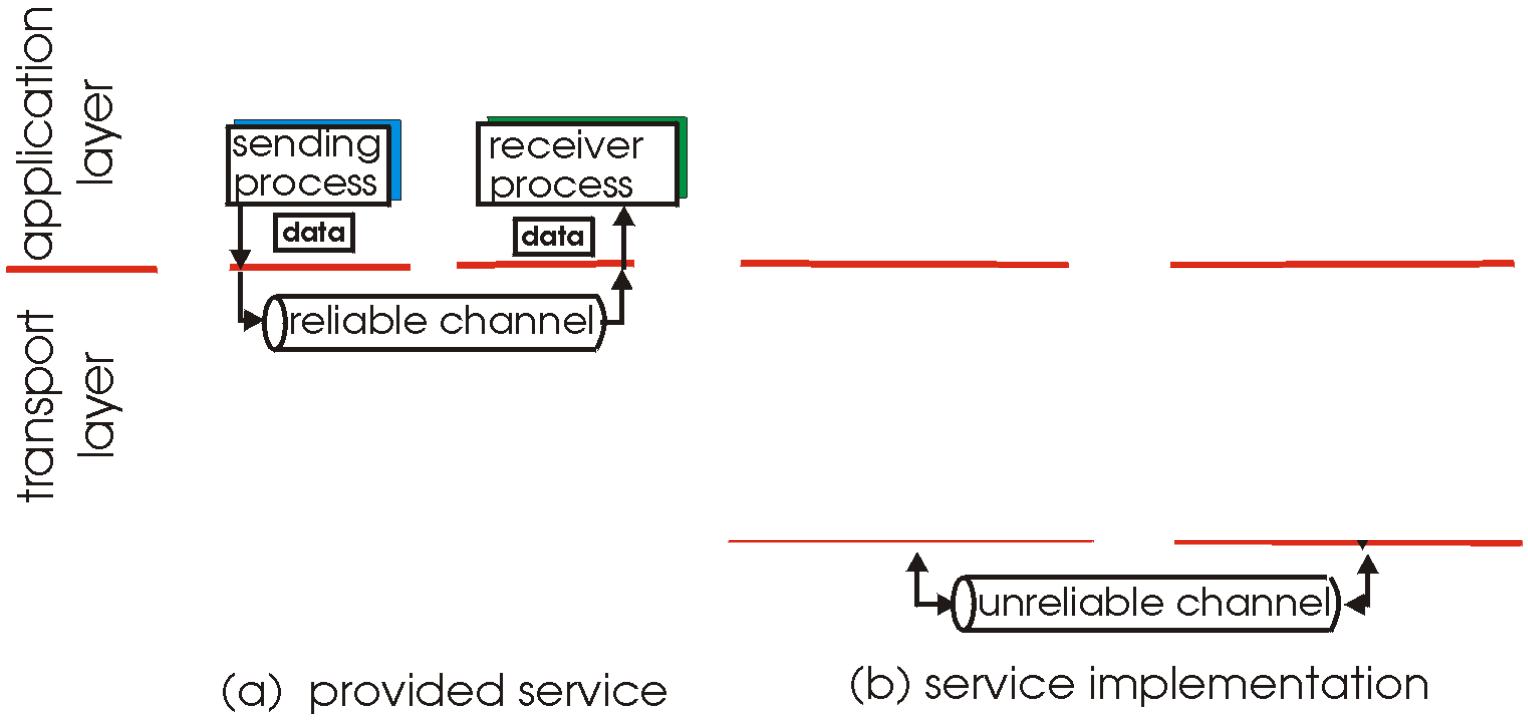
Principles of reliable data transfer

- important in application, transport, link layers
  - top-10 list of important networking topics!



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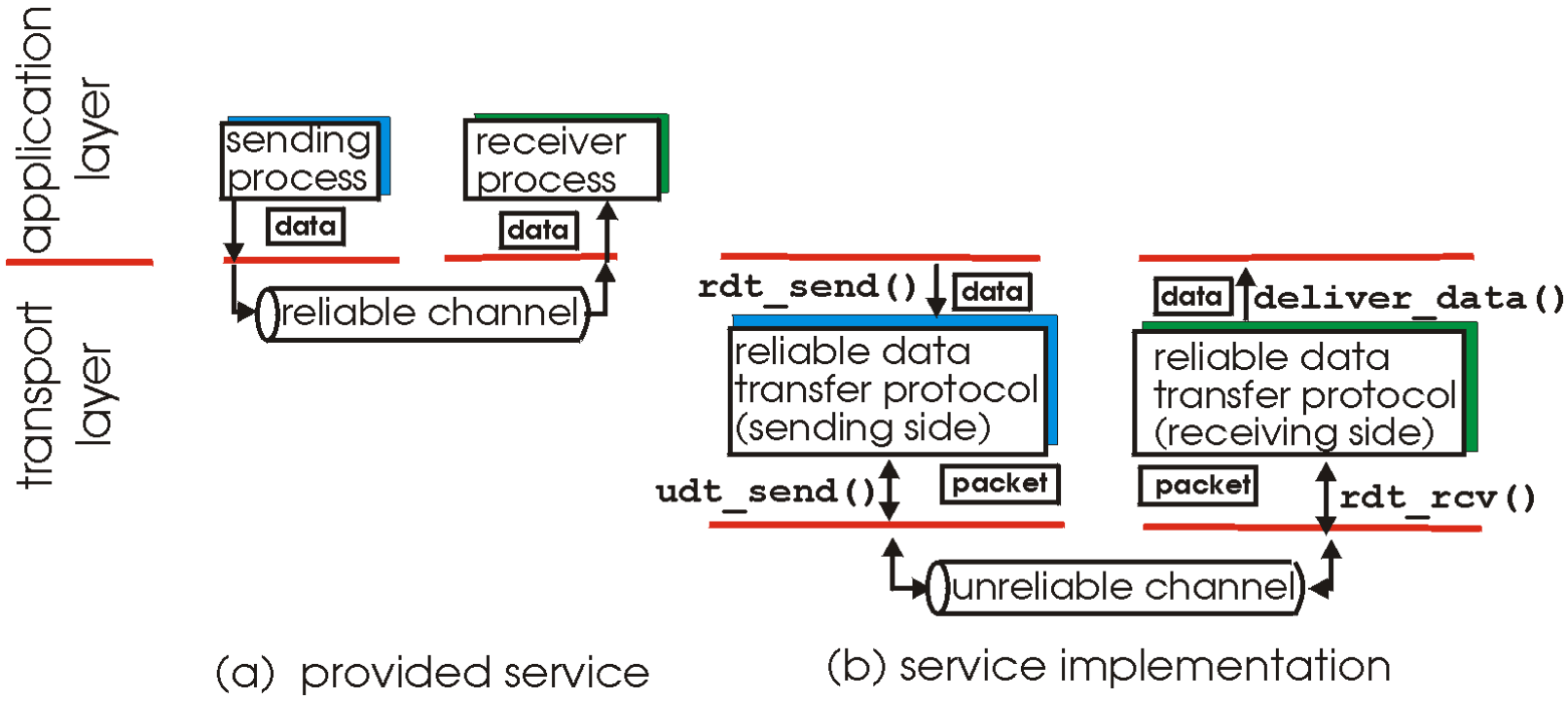
Principles of reliable data transfer

- important in application, transport, link layers
  - top-10 list of important networking topics!



- characteristics of unreliable channel will determine complexity of reliable data transfer protocol
- We assume that unreliable channel can lose packets and that packets can have errors.

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Acknowledgment message

- Packets are sent one by one and are numbered (sequence numbers).
- When a packet is sent, its sequence number is contained in the header information.
- When the receiver receives a packet and it does not contain any errors an acknowledgment message (ACK) is sent for this packet and the receiver remembers the sequence number.
- The sender waits for the acknowledgment packet until it sends the next packet.
- If the sender does not receive an acknowledgment for some time, the same packet is sent again.
- If the receiver receives a packet with a sequence number other than expected or the packet has errors, the packet is discarded, and an acknowledgment message for the last valid packet is send.

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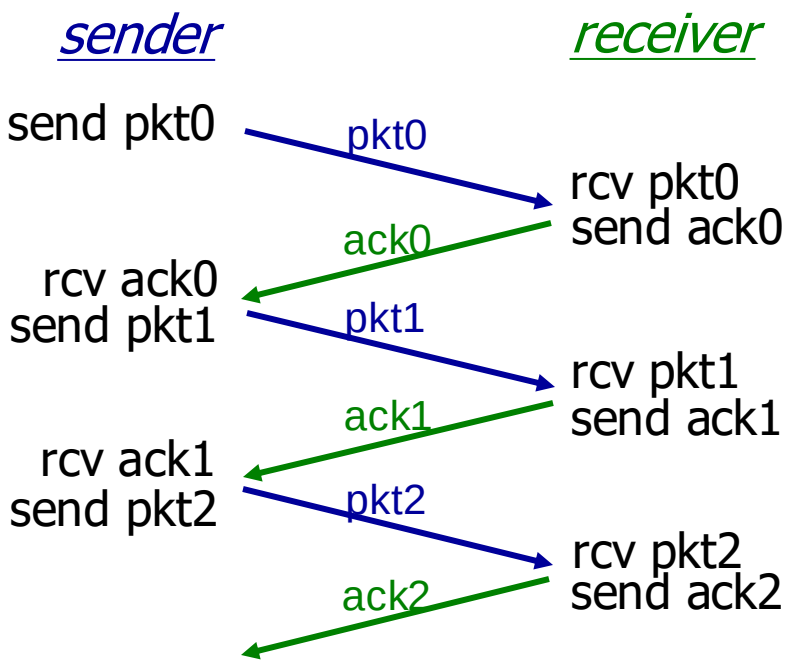
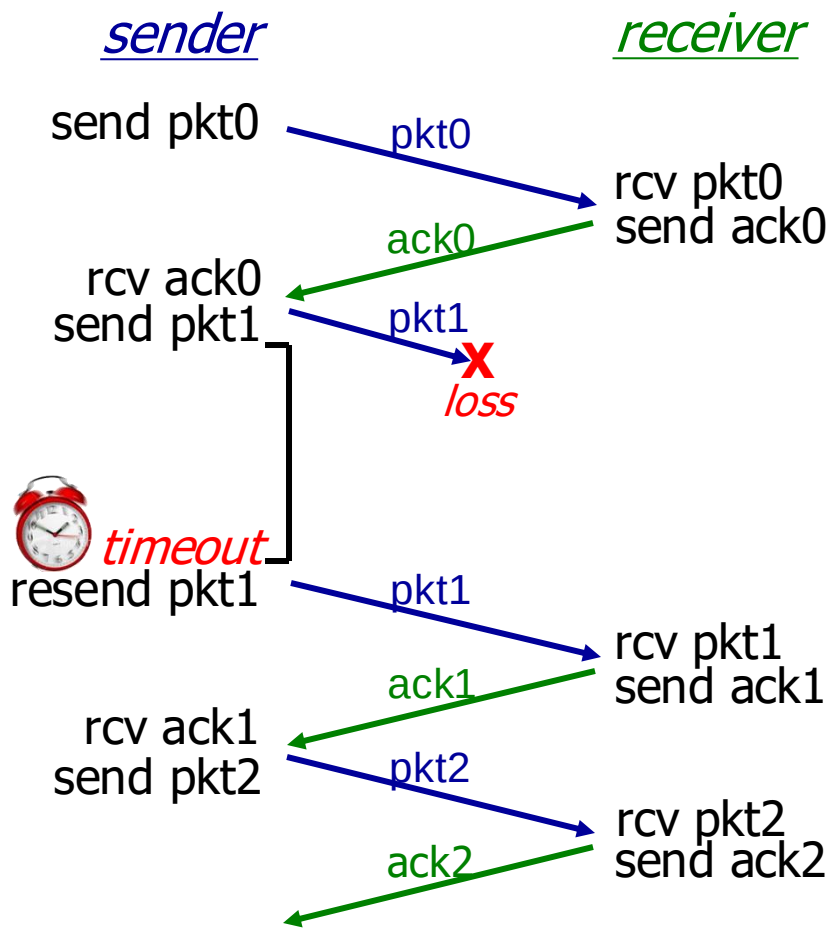
Reliable data transfer using ack messages



(a) no loss



(b) packet loss

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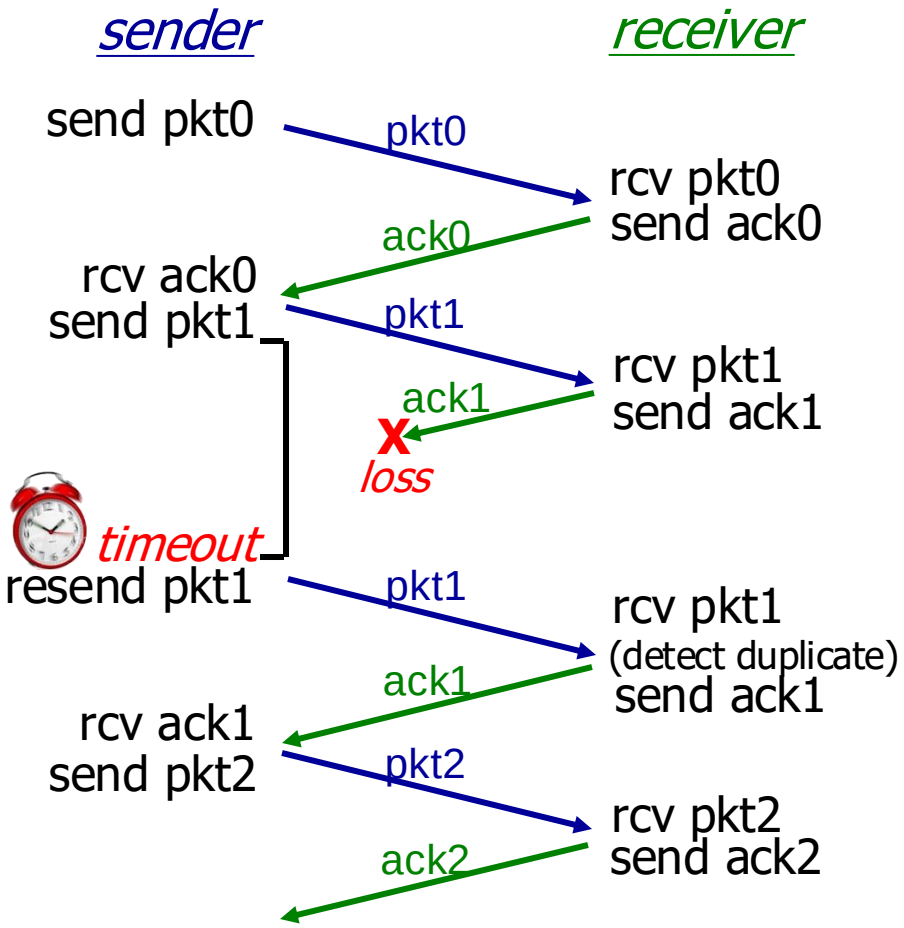
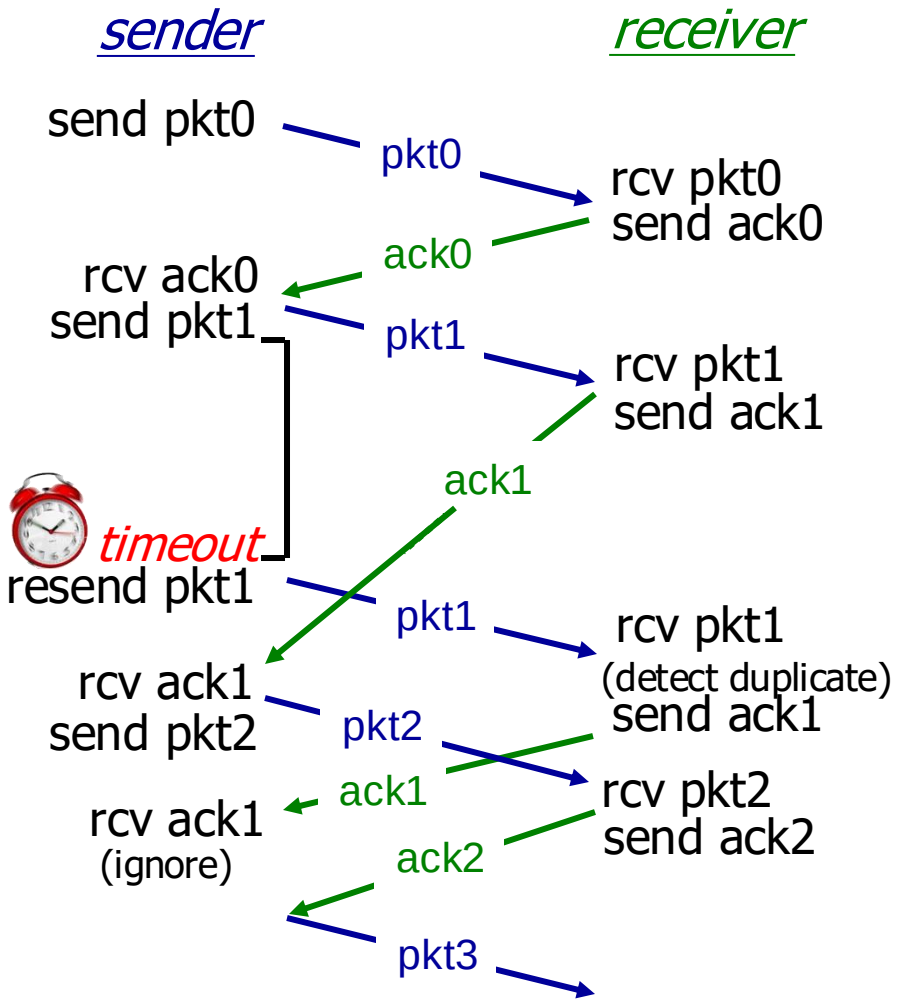
Reliable data transfer using ack messages



(c) ACK loss



(d) premature timeout/ delayed ACK

Note: 1 bit sequence number (0 or 1) suffices

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Performance

- Protocol is correct, but performance is not good
- We consider the utilization of the sender, i.e., the fraction of time the sender is busy sending data
- e.g.: 1 Gbps link, 15 ms prop. delay, 8000 bit packet
- Time to transmit data into channel:

$$D_{trans} = \frac{L}{R} = \frac{8000\ bits}{10^9\ bits/sec} = 8\ microsecs$$

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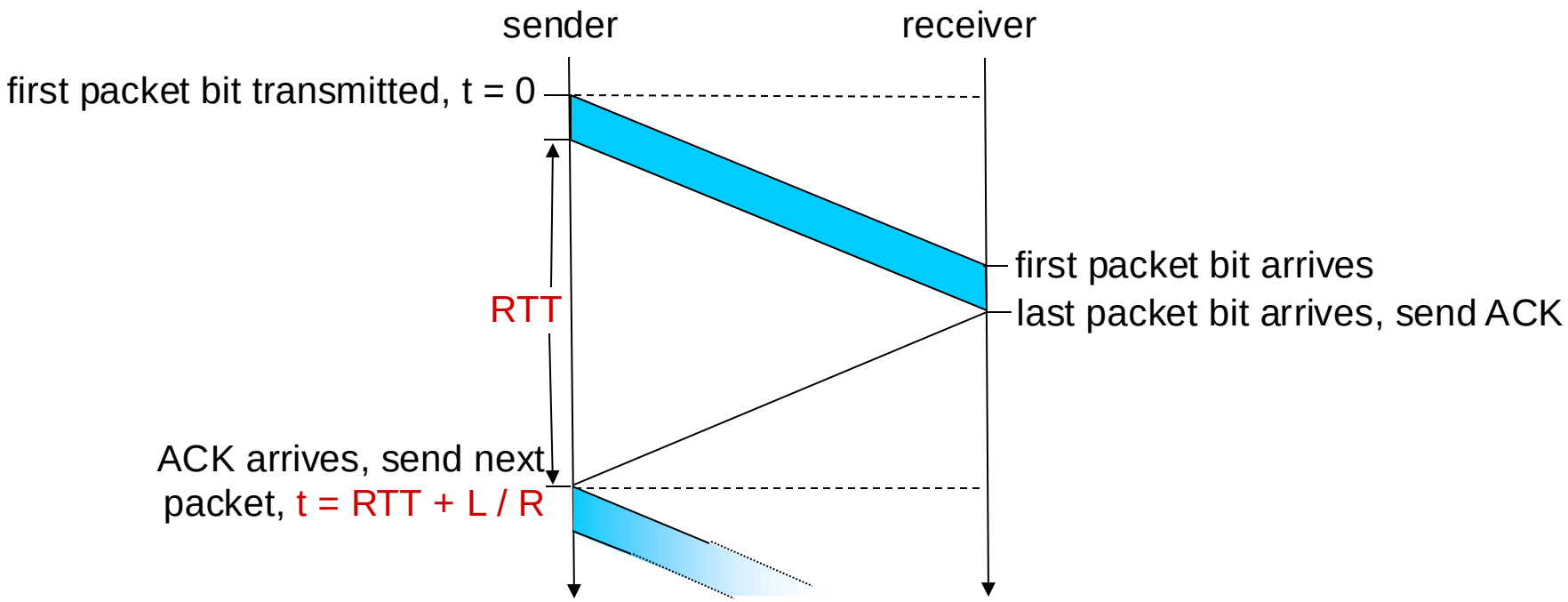
Stop-and-wait operation



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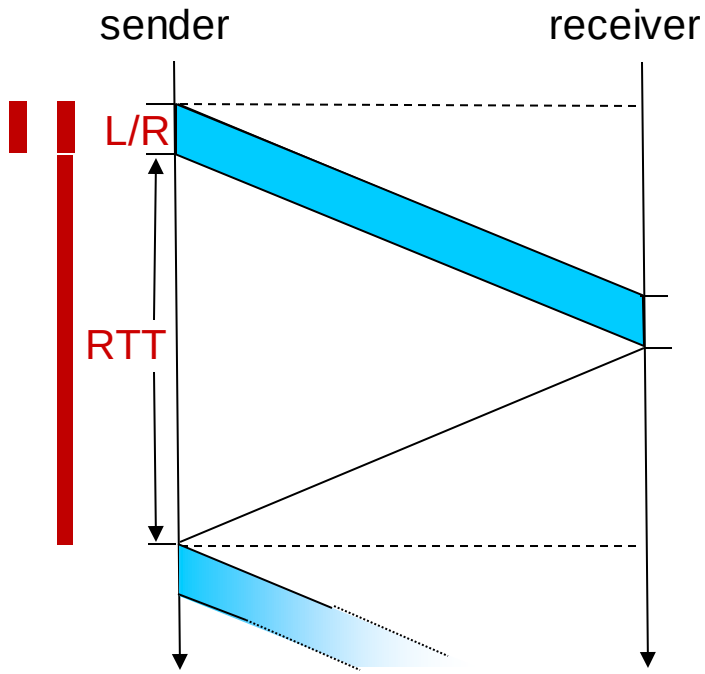
Stop-and-wait operation

$$U_{sender} = \frac{L / R}{RTT + L / R}$$

$$= \frac{.008}{30.008}$$

$$= 0.00027$$



- if RTT=30 msec, 1KB packet every 30 msec: 33kB/sec throughput over 1 Gbps link
- Protocol limits performance of underlying infrastructure (channel)

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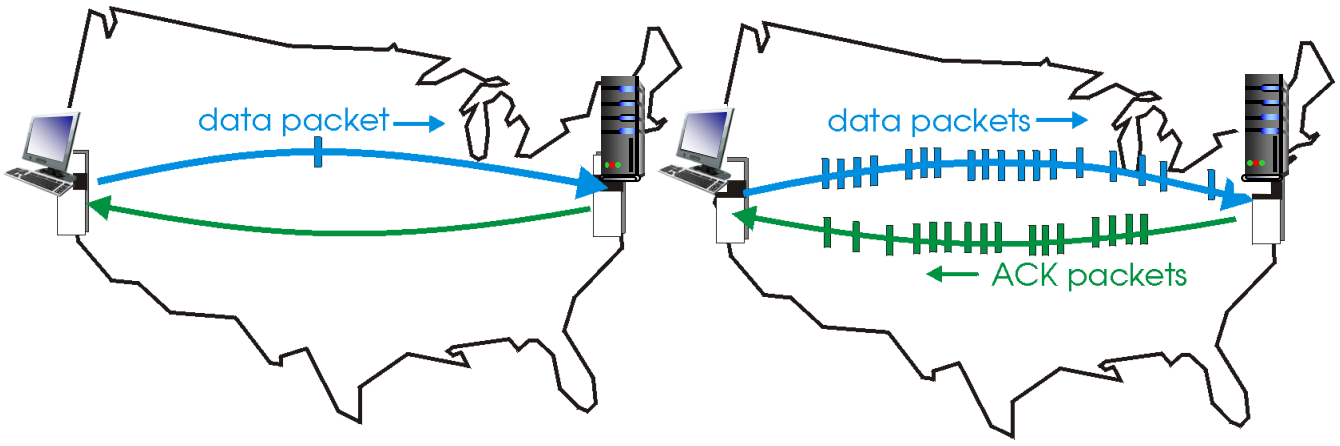
Pipelined protocols

pipelining: sender allows multiple, “in-flight”, yet-to-be-acknowledged packets



(a) a stop-and-wait protocol in operation

(b) a pipelined protocol in operation

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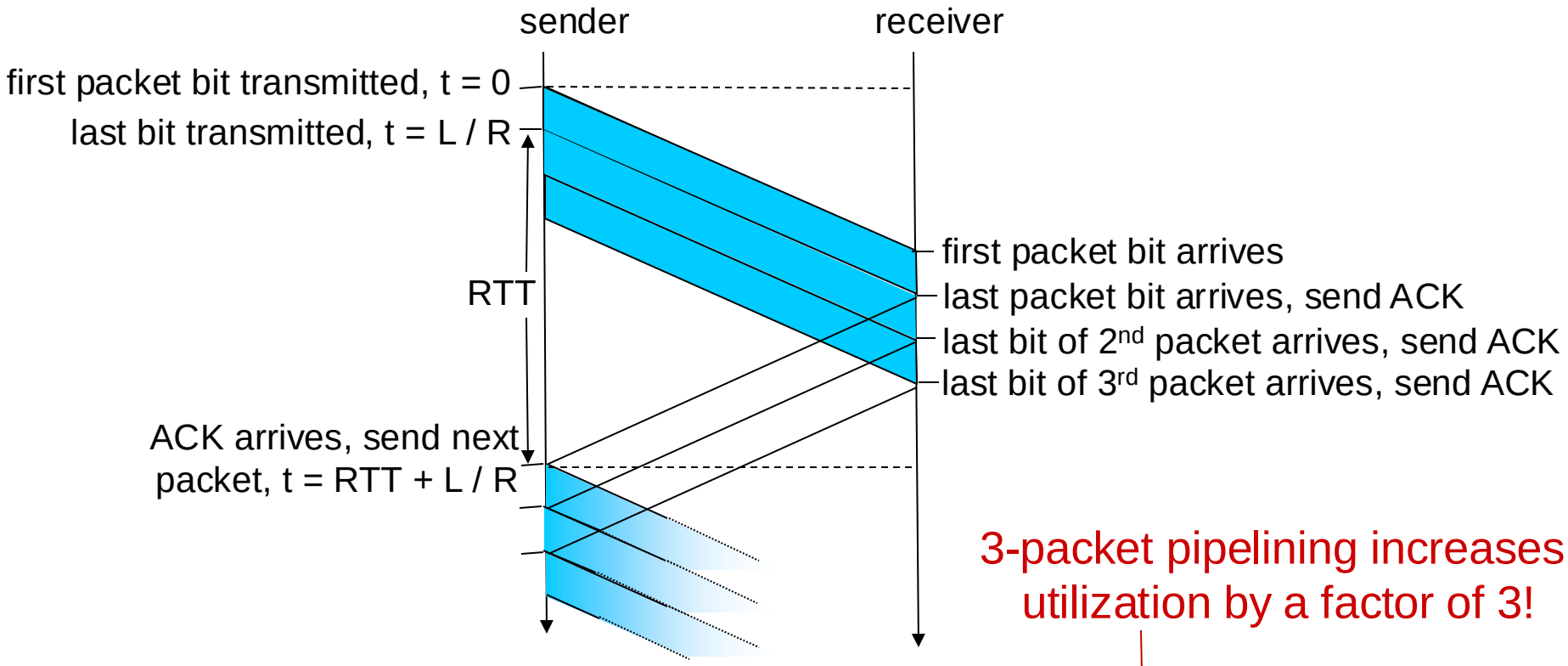
Pipelining: increased utilization



3-packet pipelining increases utilization by a factor of 3!

$$U_{sender} = \frac{3L / R}{RTT + L / R} = \frac{.0024}{30.008} = 0.00081$$

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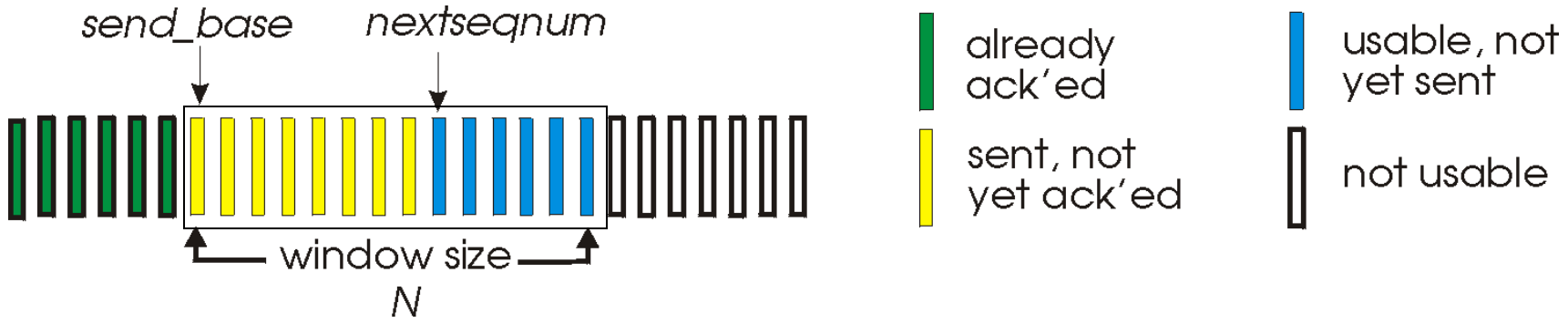
Go-Back-N: sender

- k-bit sequence number in packet header
- "window" of up to N, consecutive unacknowledged packets allowed



- ACK(n): ACKs all pkts up to, including sequence number n - *"cumulative ACK"*
  - may receive duplicate ACKs (see receiver)
- timer for oldest in-flight pkt
- *timeout(n):* retransmit packet n and all higher sequence number packets in window

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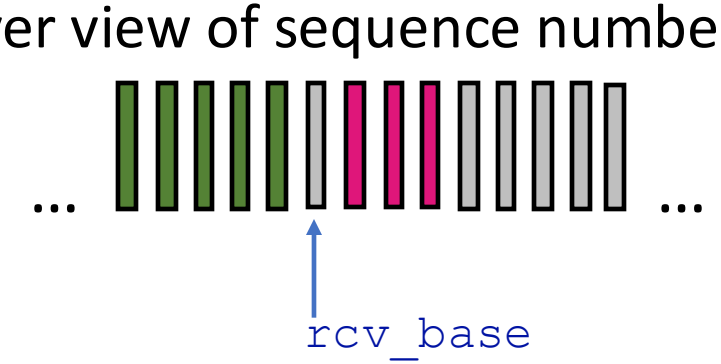
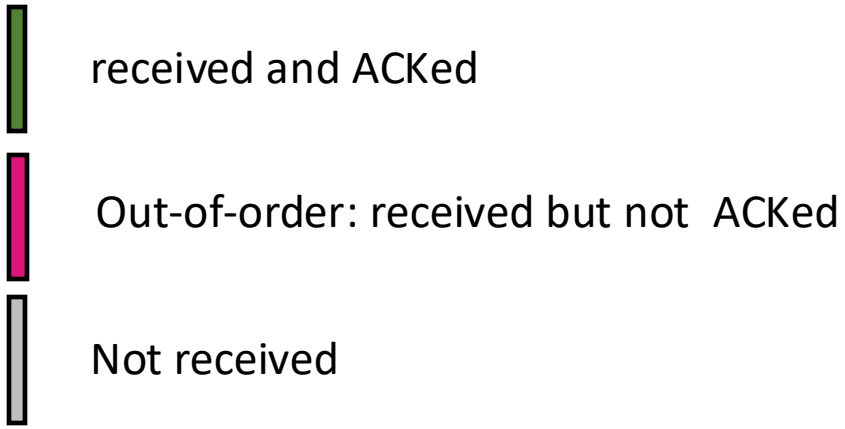
Go-Back-N: receiver

- ACK-only: always send ACK for correctly-received packet so far, with highest *in-order* seq number
  - may generate duplicate ACKs
  - need only remember `rcv_base`
- on receipt of out-of-order packet:
  - can discard (don't buffer) or buffer: an implementation decision
  - re-ACK packet with highest in-order sequence number

Receiver view of sequence number space:





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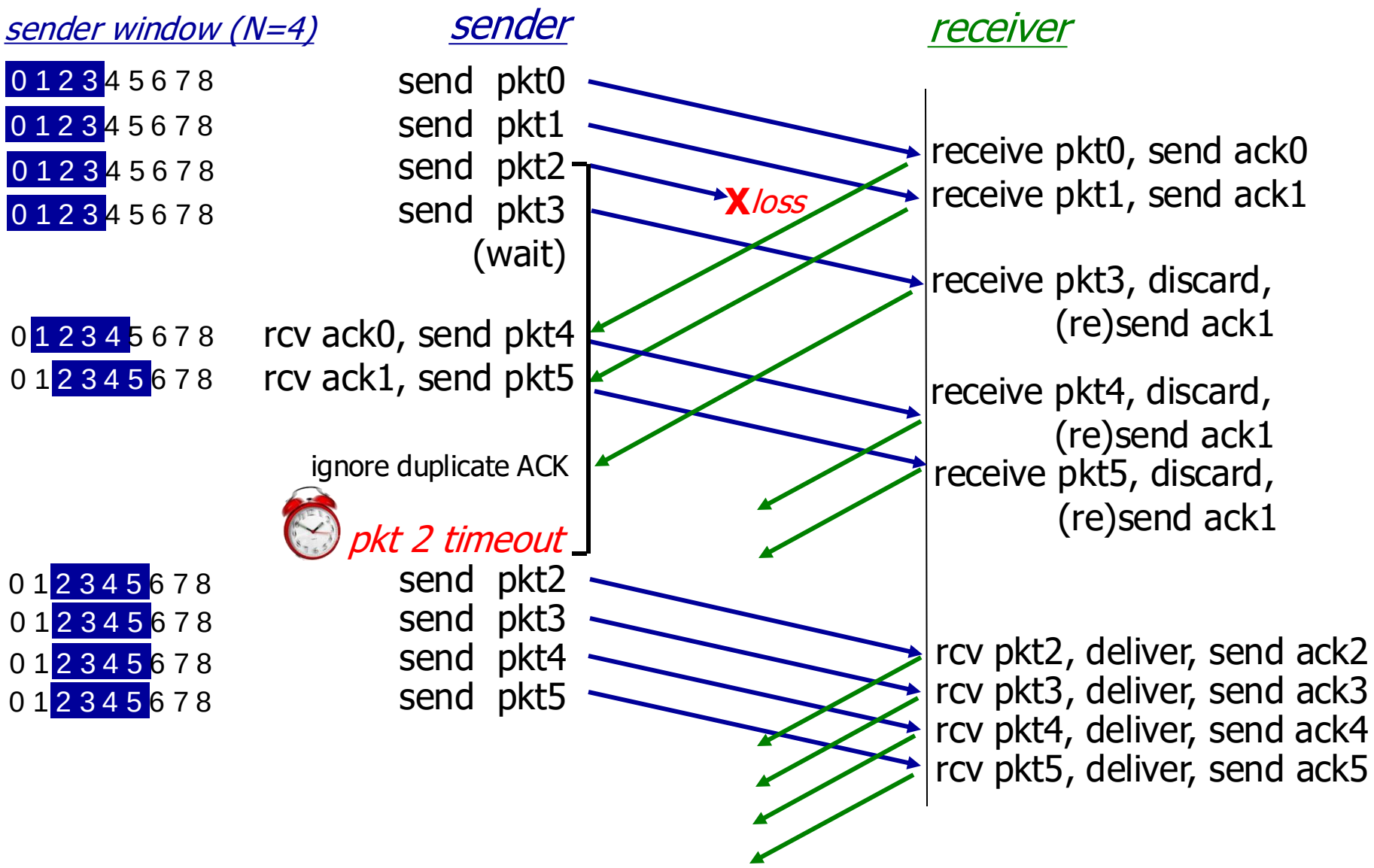
GBN in action



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3.1 transport-layer services

3.2 multiplexing and demultiplexing

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- connection management

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TCP: Overview

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

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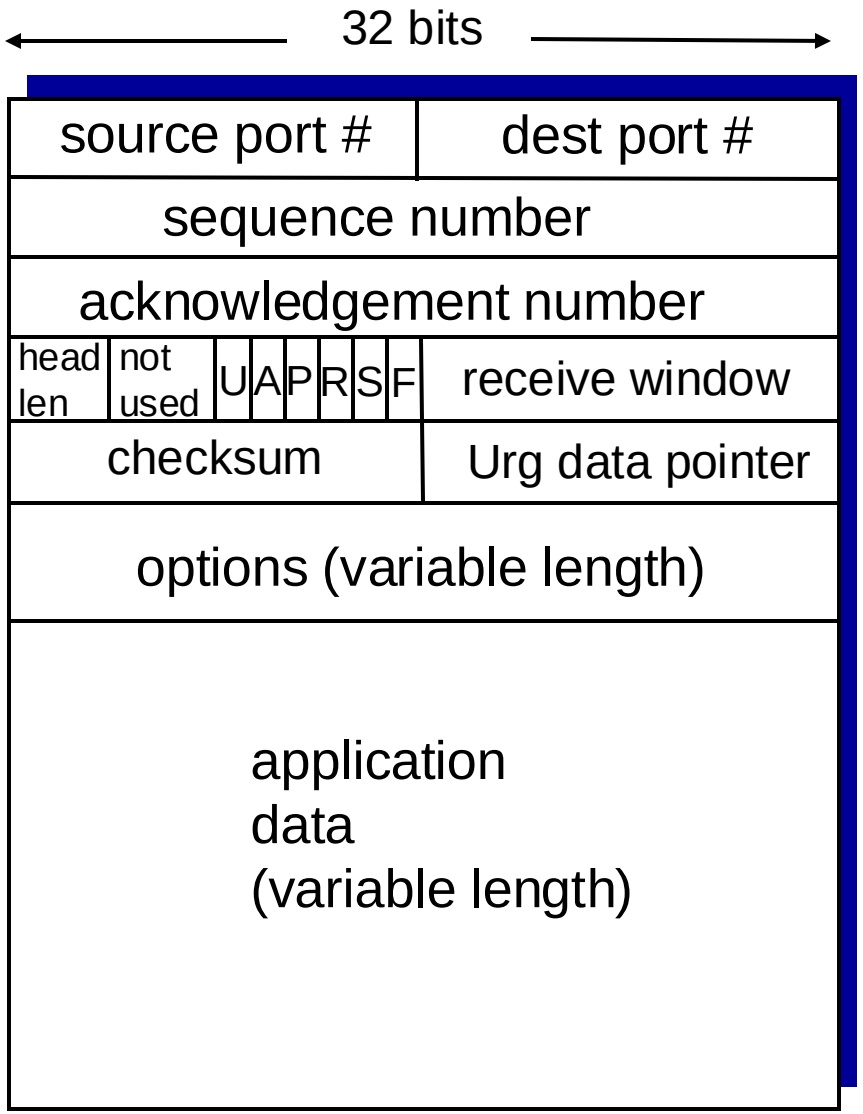
TCP segment structure



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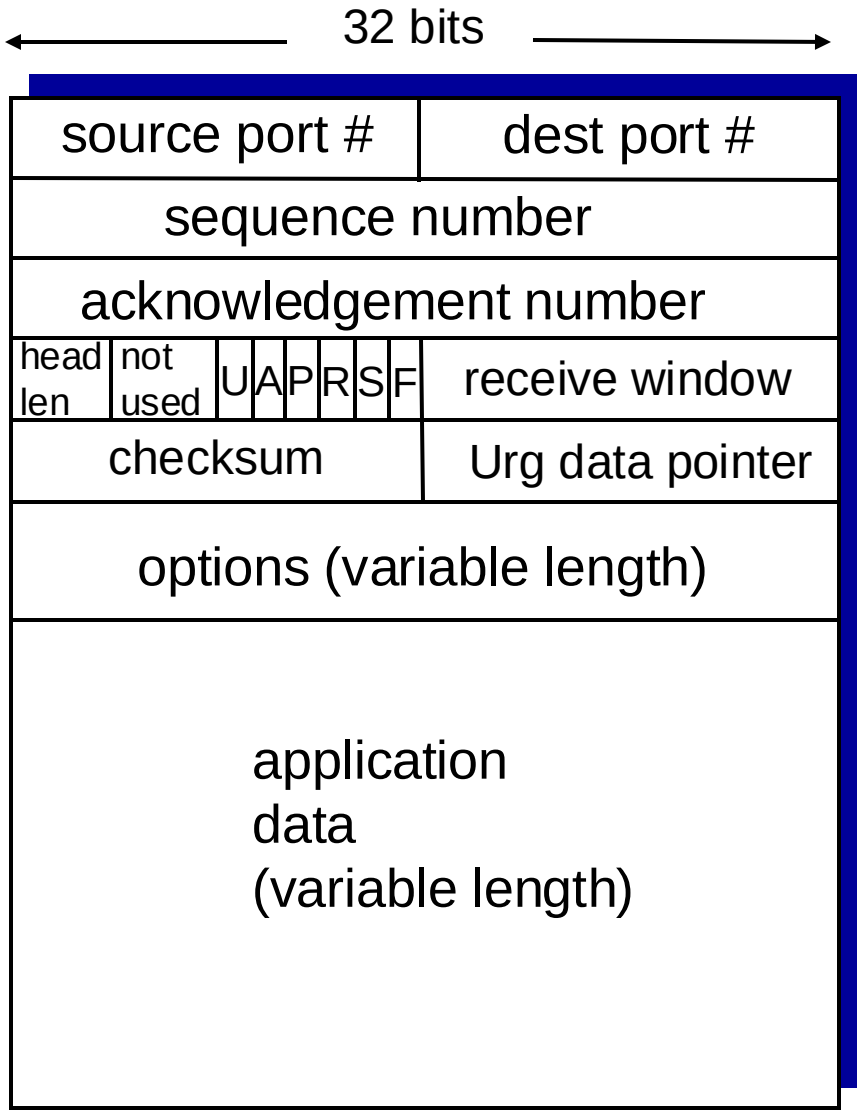
TCP Segment Header

- Source Port, Destination Port
- Sequence Number
  - At the transport layer application data is split into several smaller segments.
  - Sequence Number is used to keep track of the position of the current segment in the sequence.
- Acknowledgment Number: Number of the next expected segment
  - Used for reliable data transfer.



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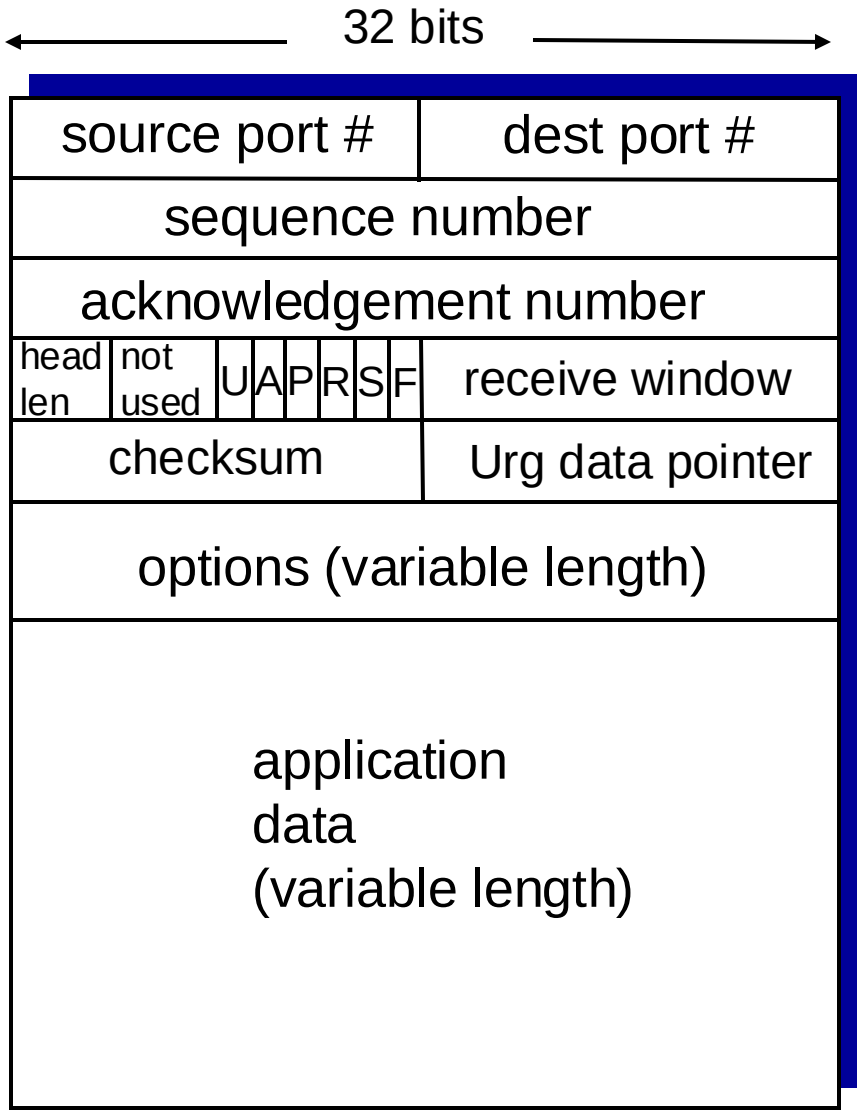
TCP Segment Header

- head len: Length of Header
  - For the receiver to know where the header ends and the application data begins
- 6 TCP control flags
- receive window
  - Number of bytes the receiver is willing to accept until an acknowledgment will be sent
- Checksum
  - Same as in UDP
- Urg data pointer
  - Point out segments that are urgent
  - Rarely used
- Options
  - Additional options such as more complicated flow control
  - rarely used



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TCP seq. numbers, ACKs

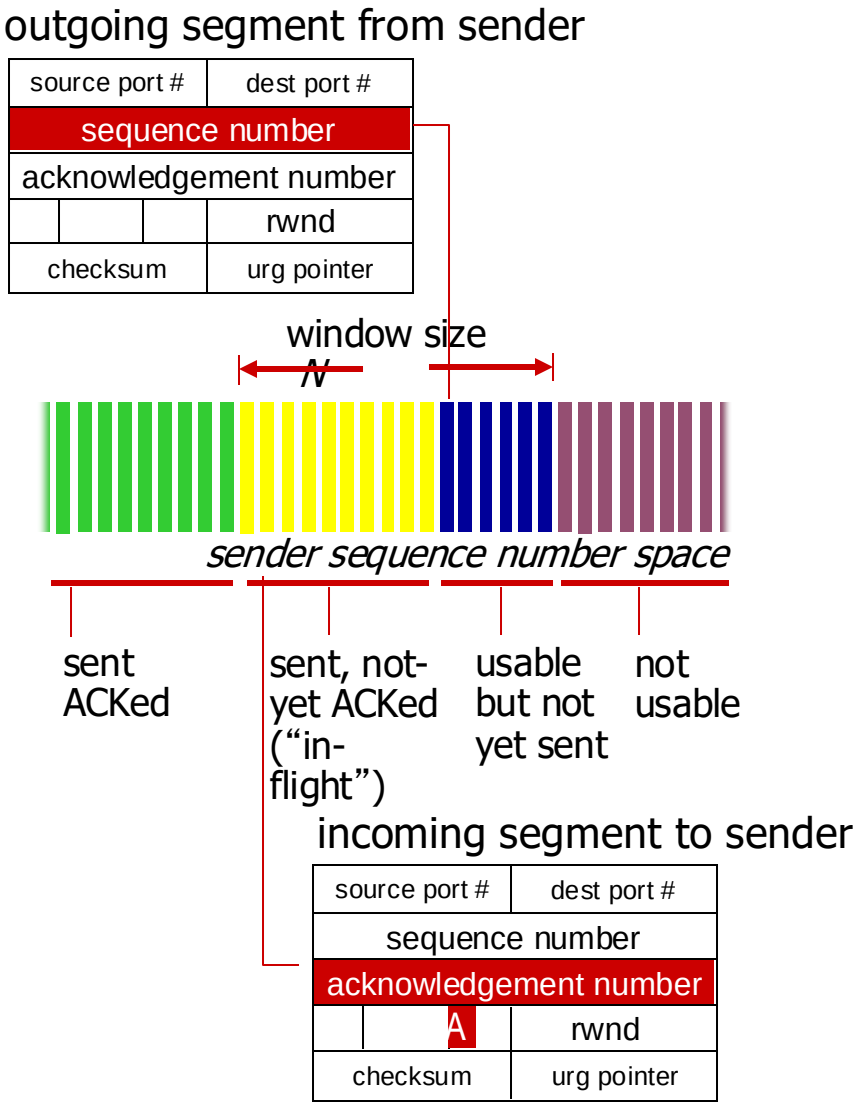
sequence numbers:

- byte stream "number" of first byte in segment's data

acknowledgements:

- Sequence number of next byte expected from other side
- cumulative ACK

- TCP does not specify how the receiver handles out-of-order segments, it is up to implementor



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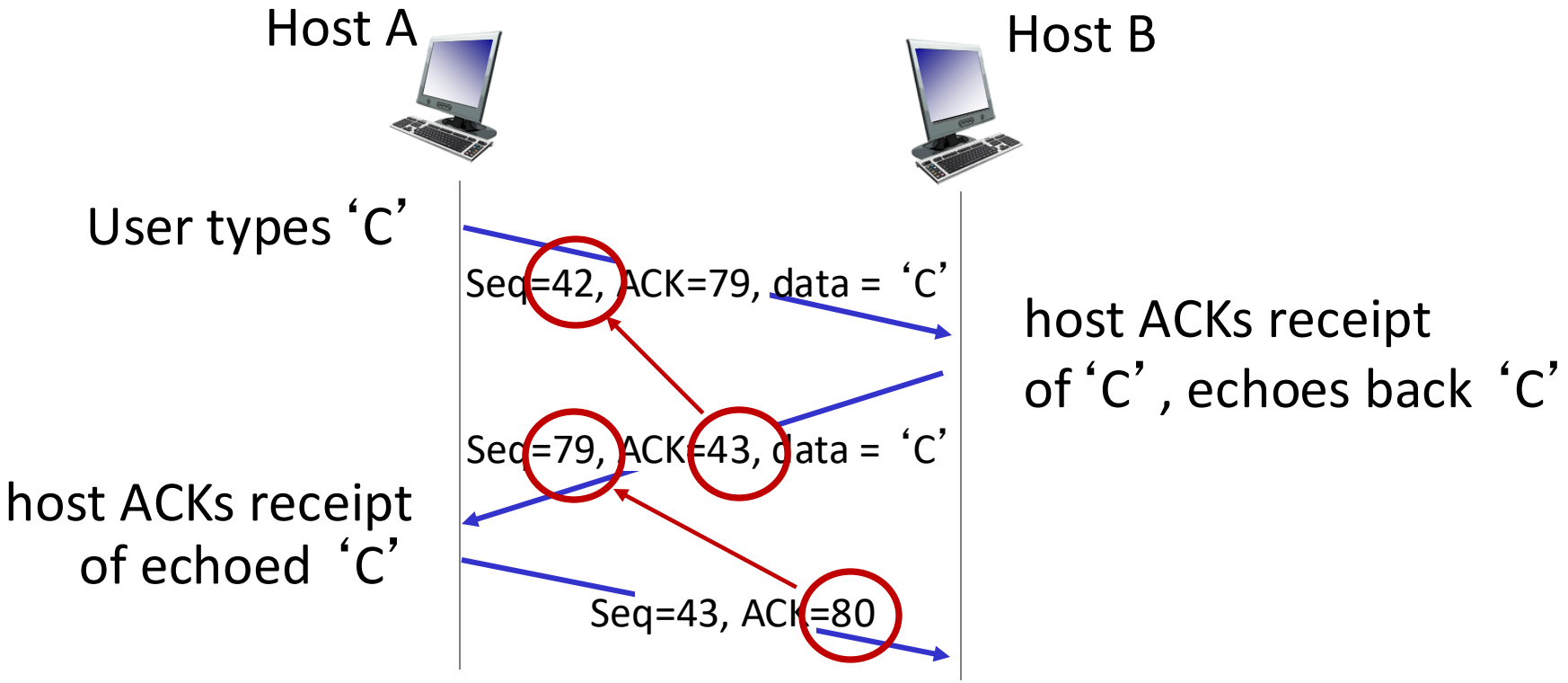
TCP sequence numbers, ACKs



simple telnet scenario

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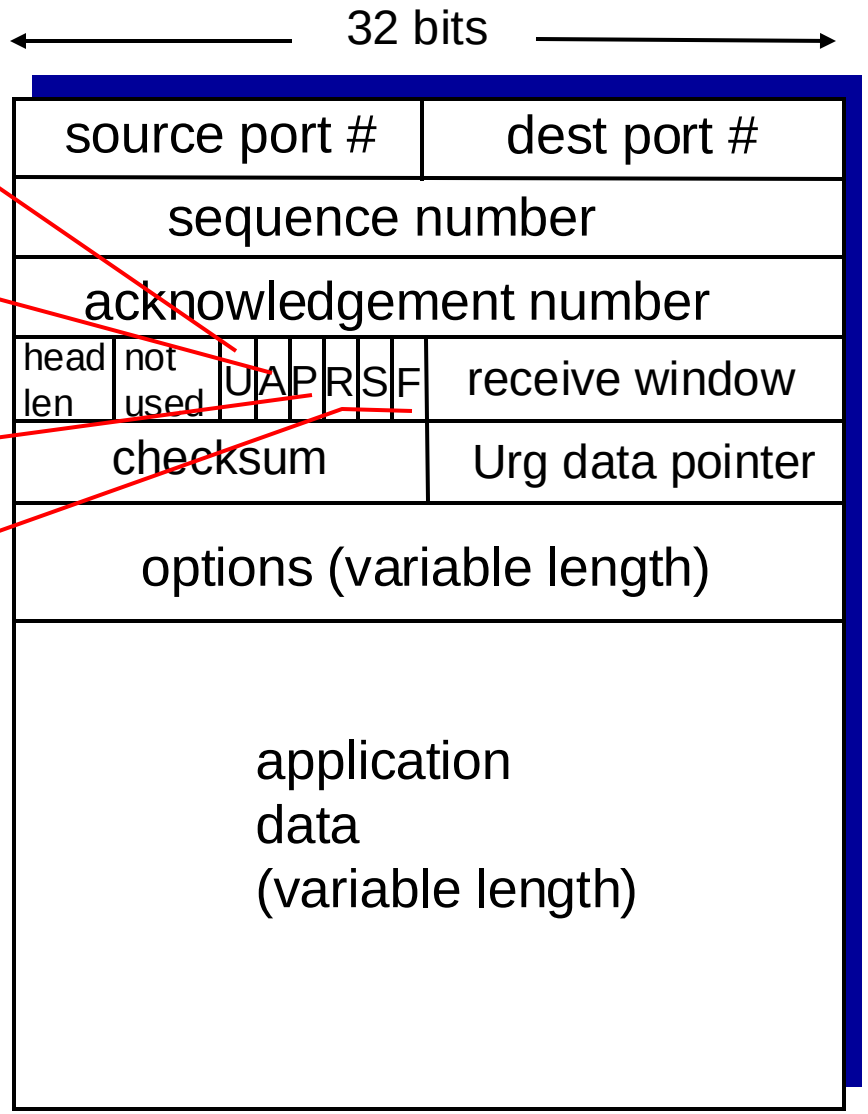
TCP control flags

URG: urgent data
(generally not used)

ACK: ACK #
valid

PSH: push data now
(generally not used)

RST, SYN, FIN:
connection estab
(setup, teardown
commands)



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3.6 principles of congestion control

3.7 TCP congestion control

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TCP reliable data transfer

- TCP creates reliable data transfer service on top of IP's unreliable service
  - pipelined segments
  - cumulative acks
  - single retransmission timer
- retransmissions triggered by:
  - timeout events
  - duplicate acks

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TCP Sender (simplified)

event: data received from application

- create segment with seq number
- seq number is byte-stream number of first data byte in segment
- start timer if not already running
  - think of timer as for oldest unACKed segment
  - expiration interval: **`TimeOutInterval`**

event: timeout

- retransmit segment that caused timeout
- restart timer

event: ACK received

- if ACK acknowledges previously unACKed segments
  - update what is known to be ACKed
  - start timer if there are still unACKed segments

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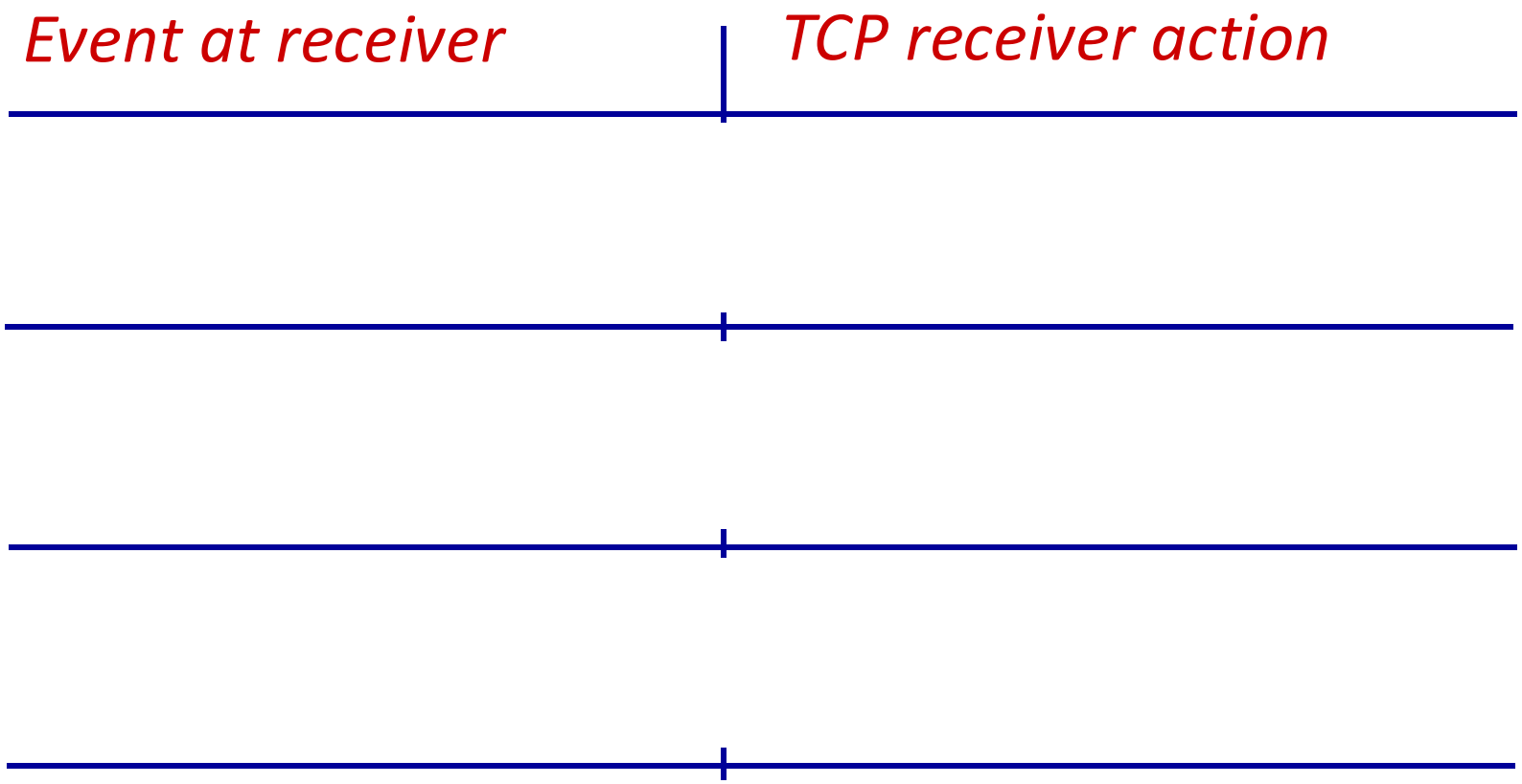
TCP Receiver: ACK generation [RFC 5681]



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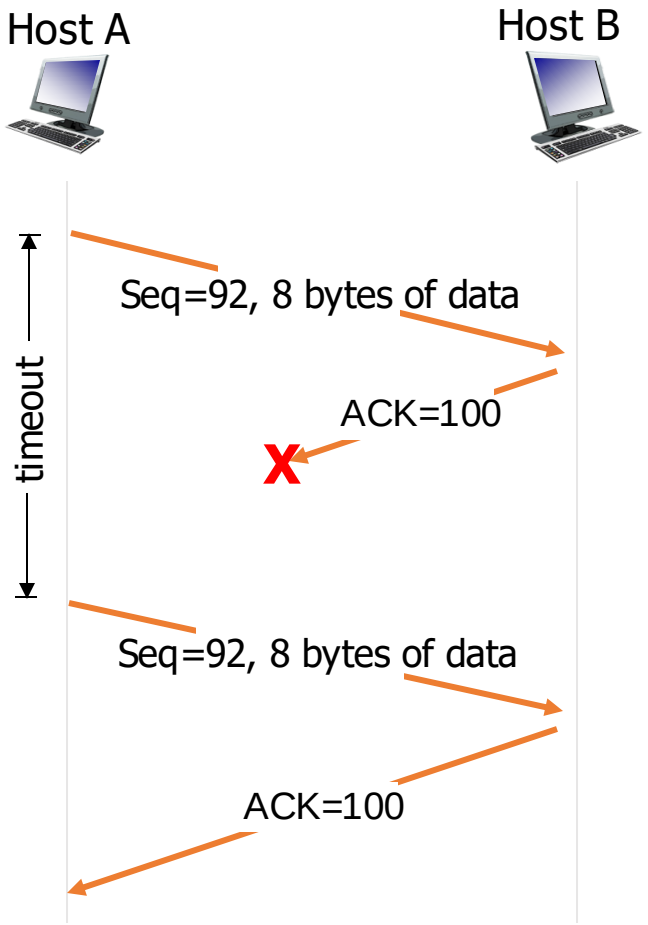
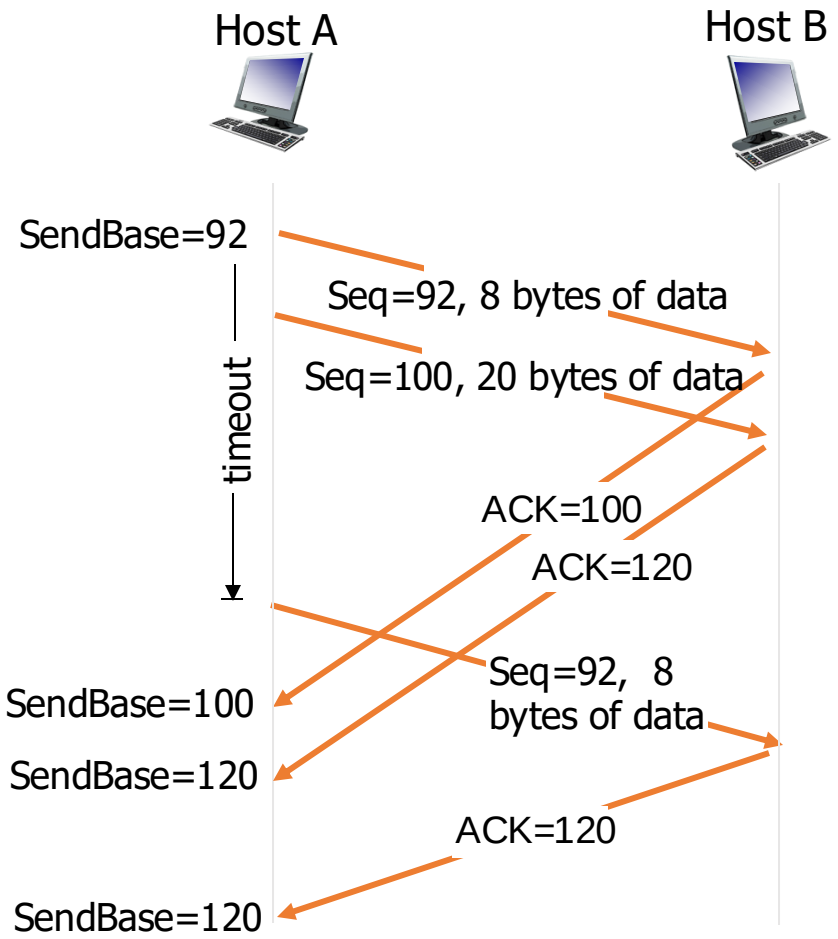
TCP: retransmission scenarios



lost ACK scenario



premature timeout

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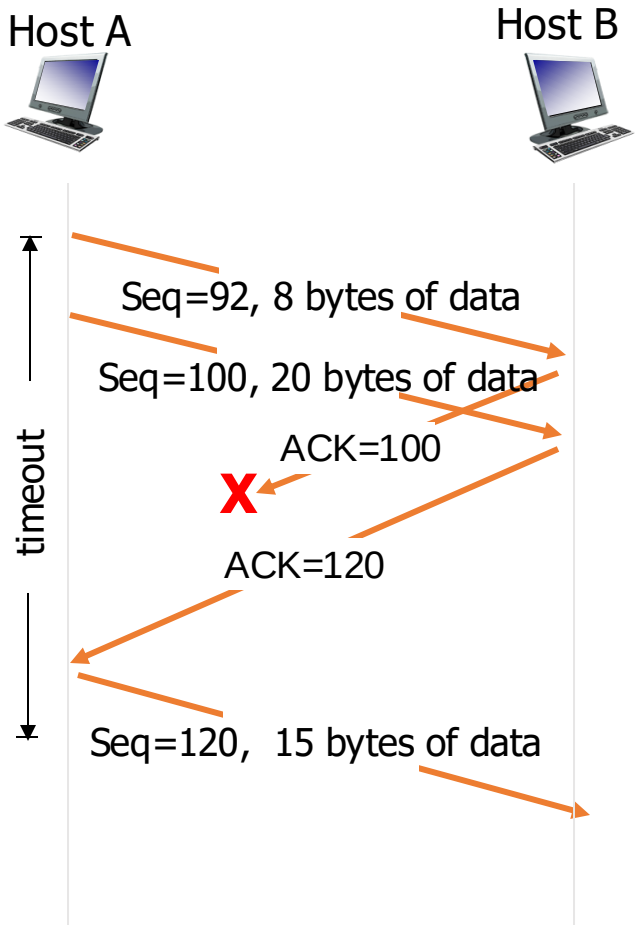
TCP: retransmission scenarios



cumulative ACK

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TCP fast retransmit

- time-out period often relatively long:
  - long delay before resending lost packet
- detect lost segments via duplicate ACKs.
  - sender often sends many segments back-to-back
  - if segment is lost, there will likely be many duplicate ACKs.

TCP fast retransmit

if sender receives 3 ACKs for same data ("triple duplicate ACKs"), resend unacked segment with smallest seq number

- likely that unacked segment lost, so don't wait for timeout

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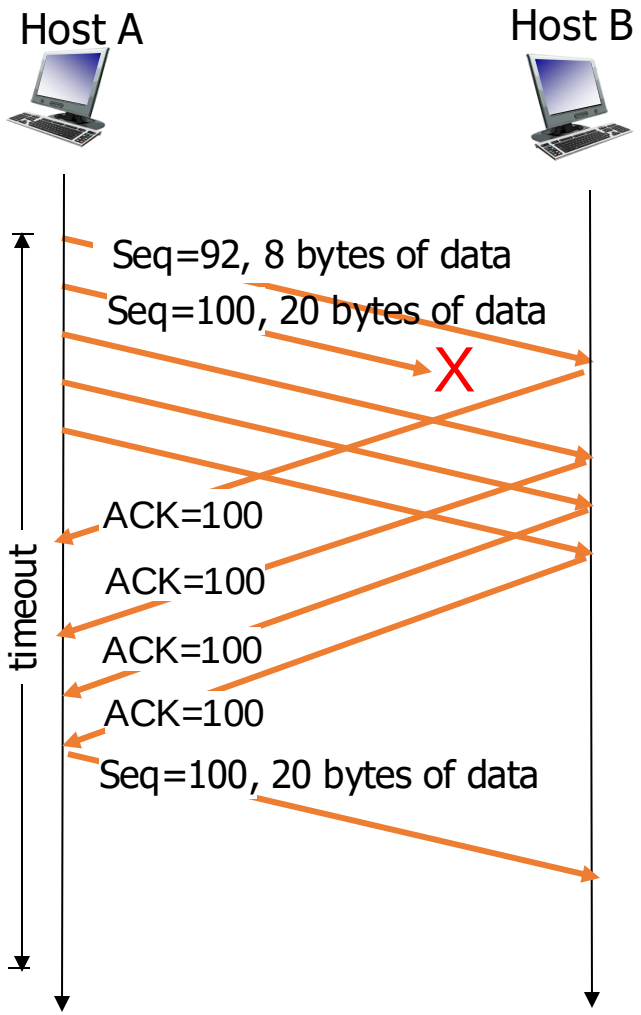
TCP fast retransmit



fast retransmit after sender receipt of triple duplicate ACK

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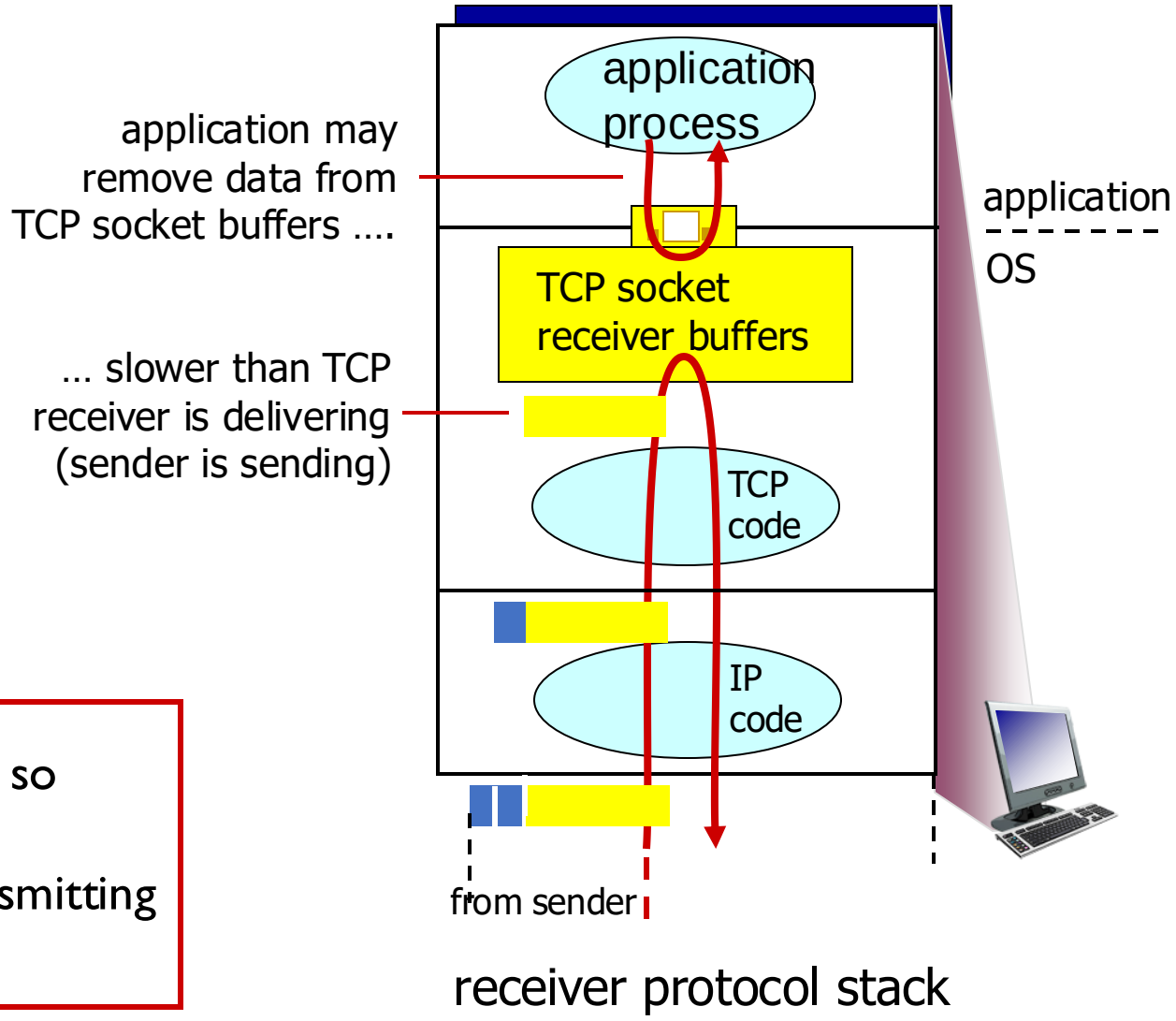
TCP flow control



flow control

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

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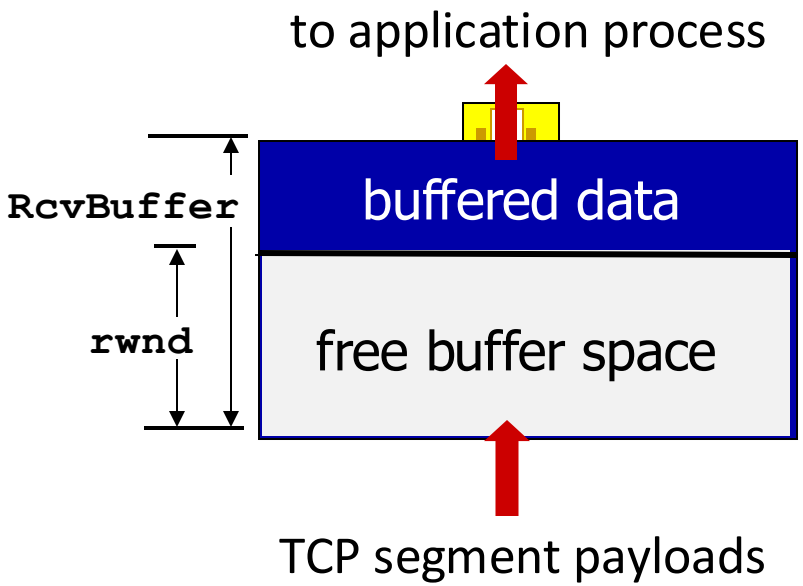
TCP flow control

- TCP receiver "advertises" free buffer space in **`rwnd`** field in TCP header
  - **`RcvBuffer`** size set via socket options (typical default is 4096 bytes)
  - many operating systems autoadjust **`RcvBuffer`**
- sender limits amount of unACKed ("in-flight") data to received **`rwnd`**
- guarantees receive buffer will not overflow



TCP receiver-side buffering

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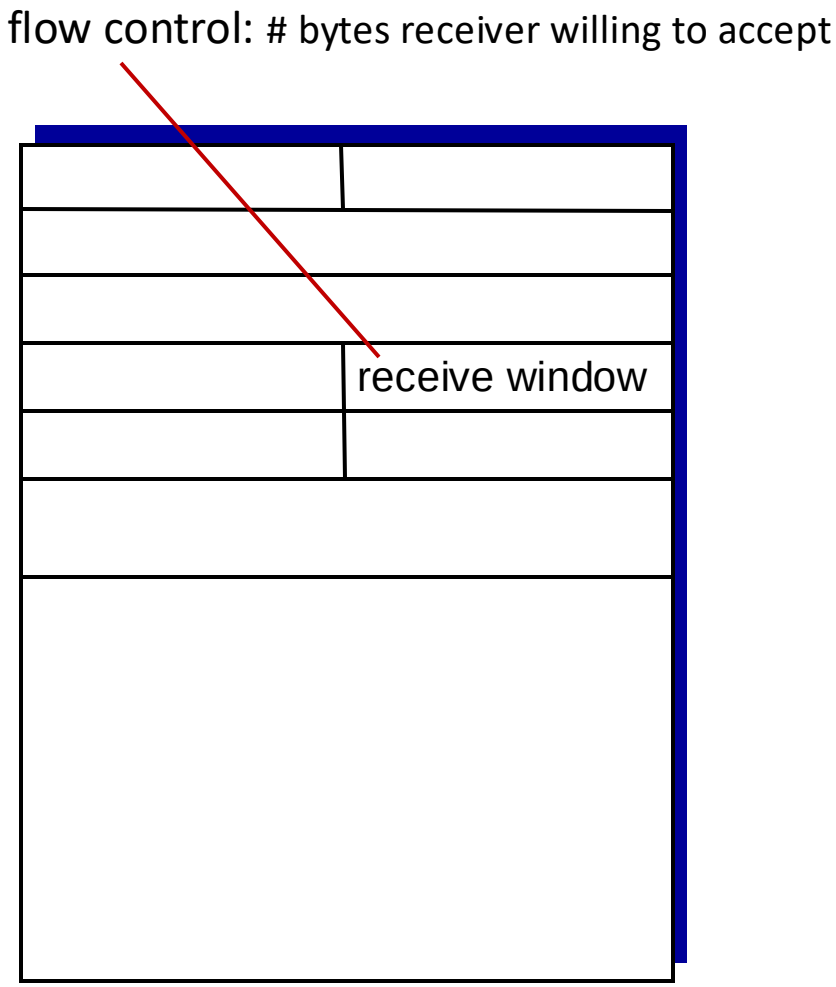
TCP flow control

- TCP receiver “advertises” free buffer space in **`rwnd`** field in TCP header
  - **`RcvBuffer`** size set via socket options (typical default is 4096 bytes)
  - many operating systems autoadjust **`RcvBuffer`**
- sender limits amount of unACKed (“in-flight”) data to received **`rwnd`**
- guarantees receive buffer will not overflow



TCP segment format

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Chapter 3 outline

3.1 transport-layer services

3.2 multiplexing and demultiplexing

3.3 connectionless transport: UDP

3.4 principles of reliable data transfer

3.5 connection-oriented transport: TCP

- segment structure
- reliable data transfer
- flow control
- connection management

3.6 principles of congestion control

3.7 TCP congestion control

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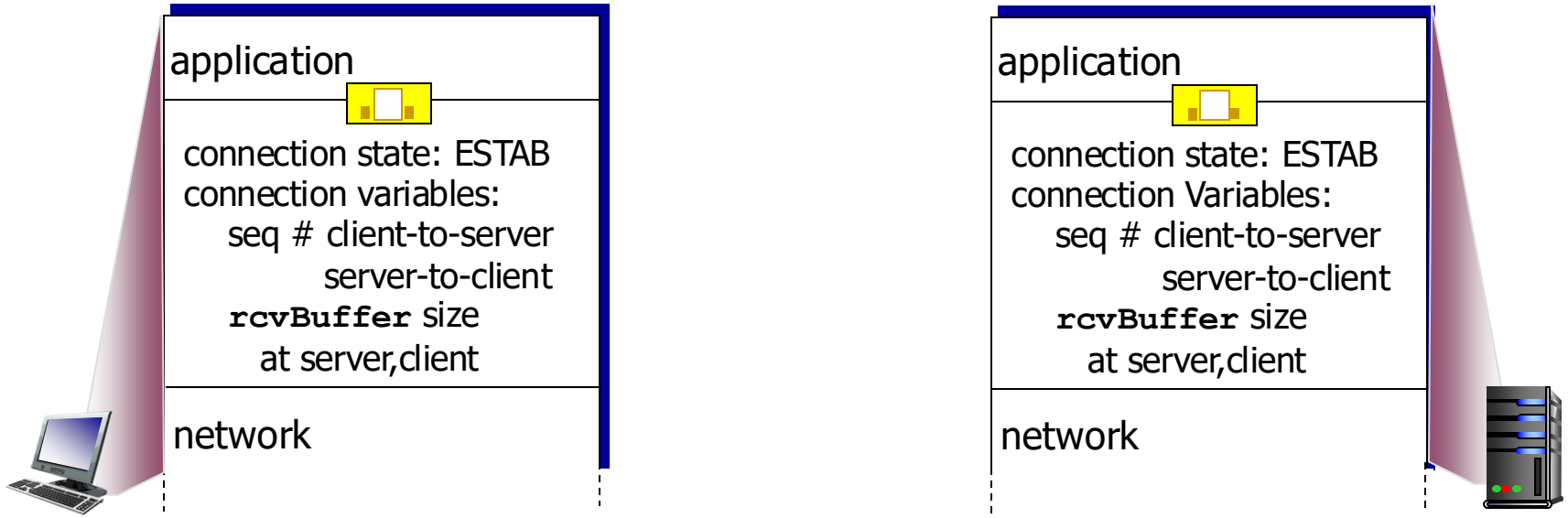
Connection Management

before exchanging data, sender/receiver “handshake”:

- agree to establish connection (each knowing the other willing to establish connection)
- agree on connection parameters



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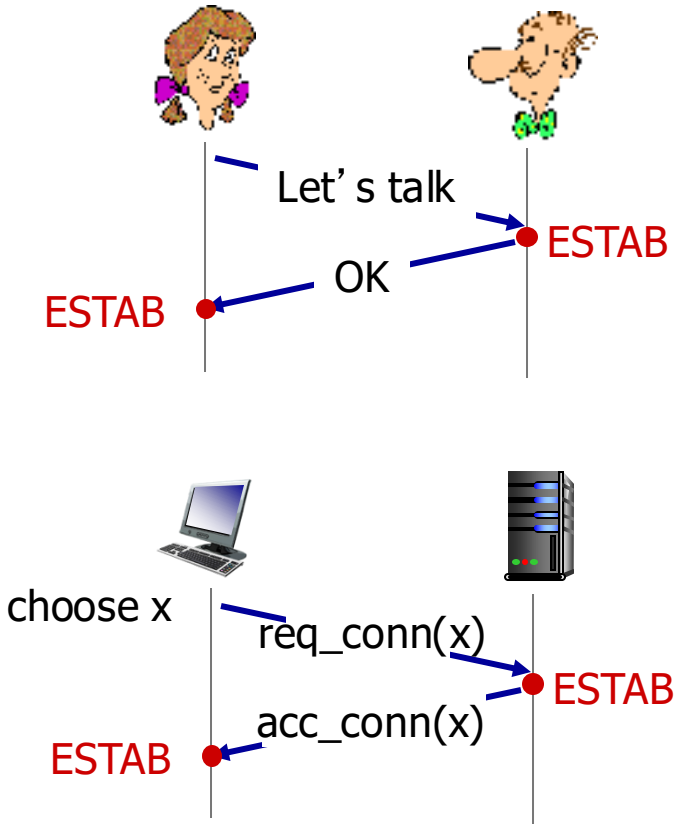
Agreeing to establish a connection

2-way handshake:



Q: will 2-way handshake always work in network?

- variable delays
- retransmitted messages (e.g. req_conn(x)) due to message loss
- message reordering
- can't "see" other side

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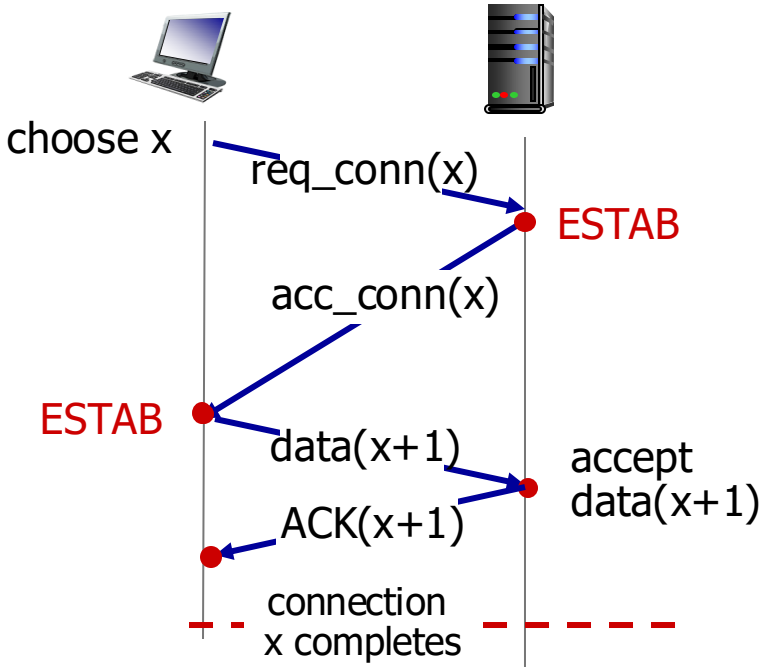
2-way handshake scenarios



No problem!

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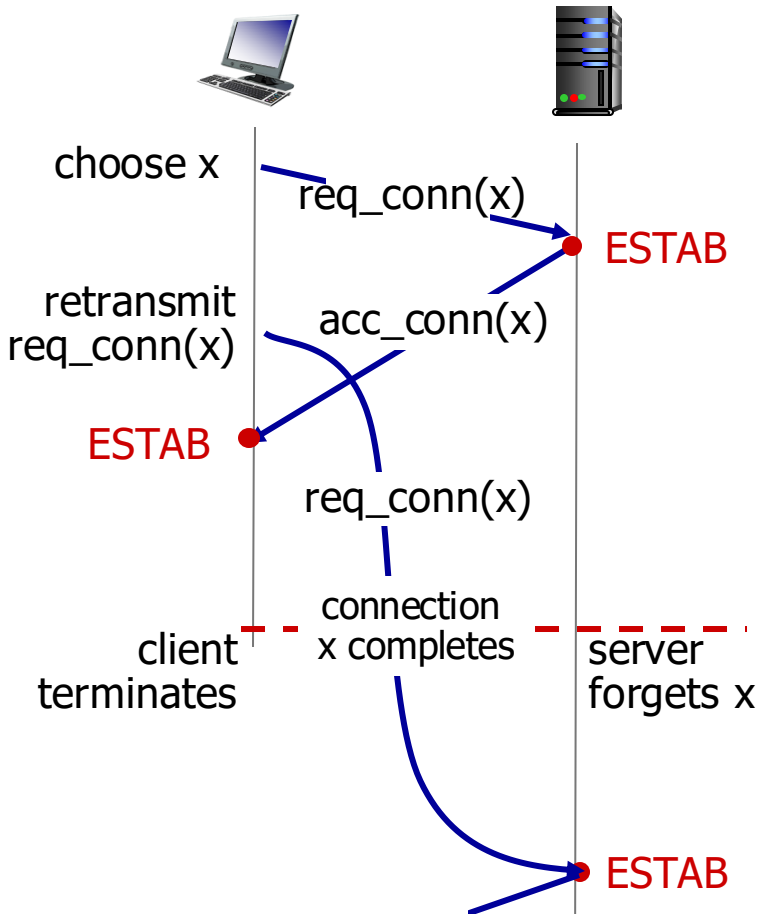
2-way handshake scenarios



Problem: half open connection! (no client)

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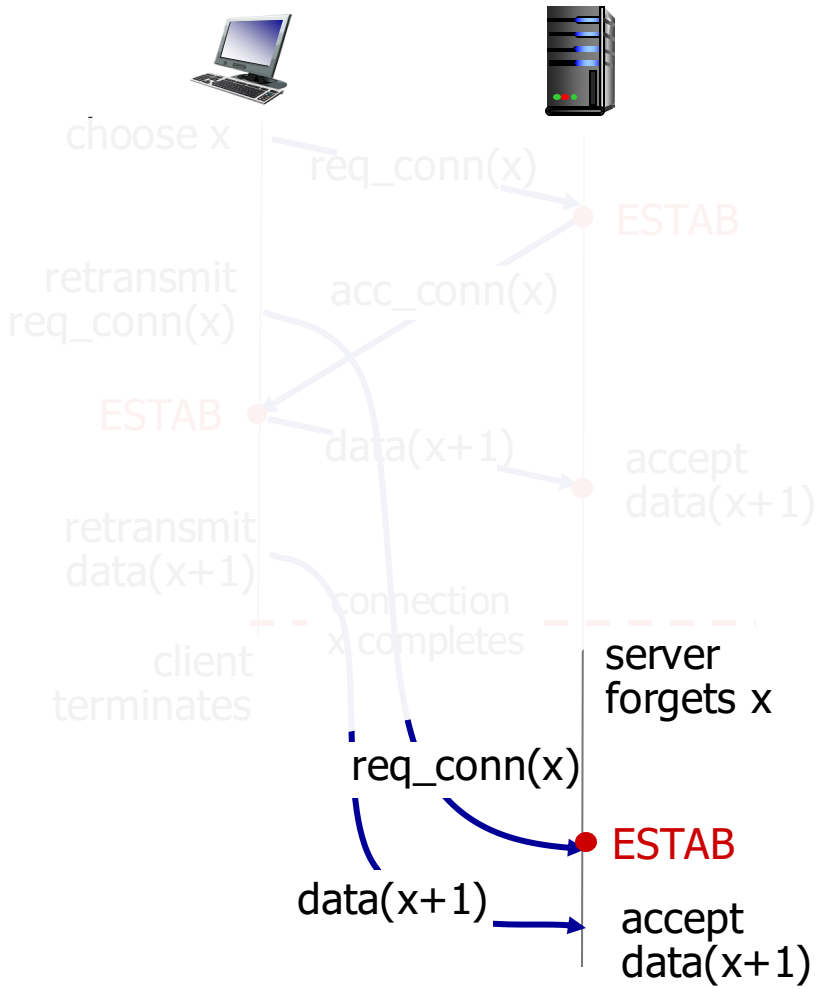
2-way handshake scenarios



Problem: duplicate data accepted!

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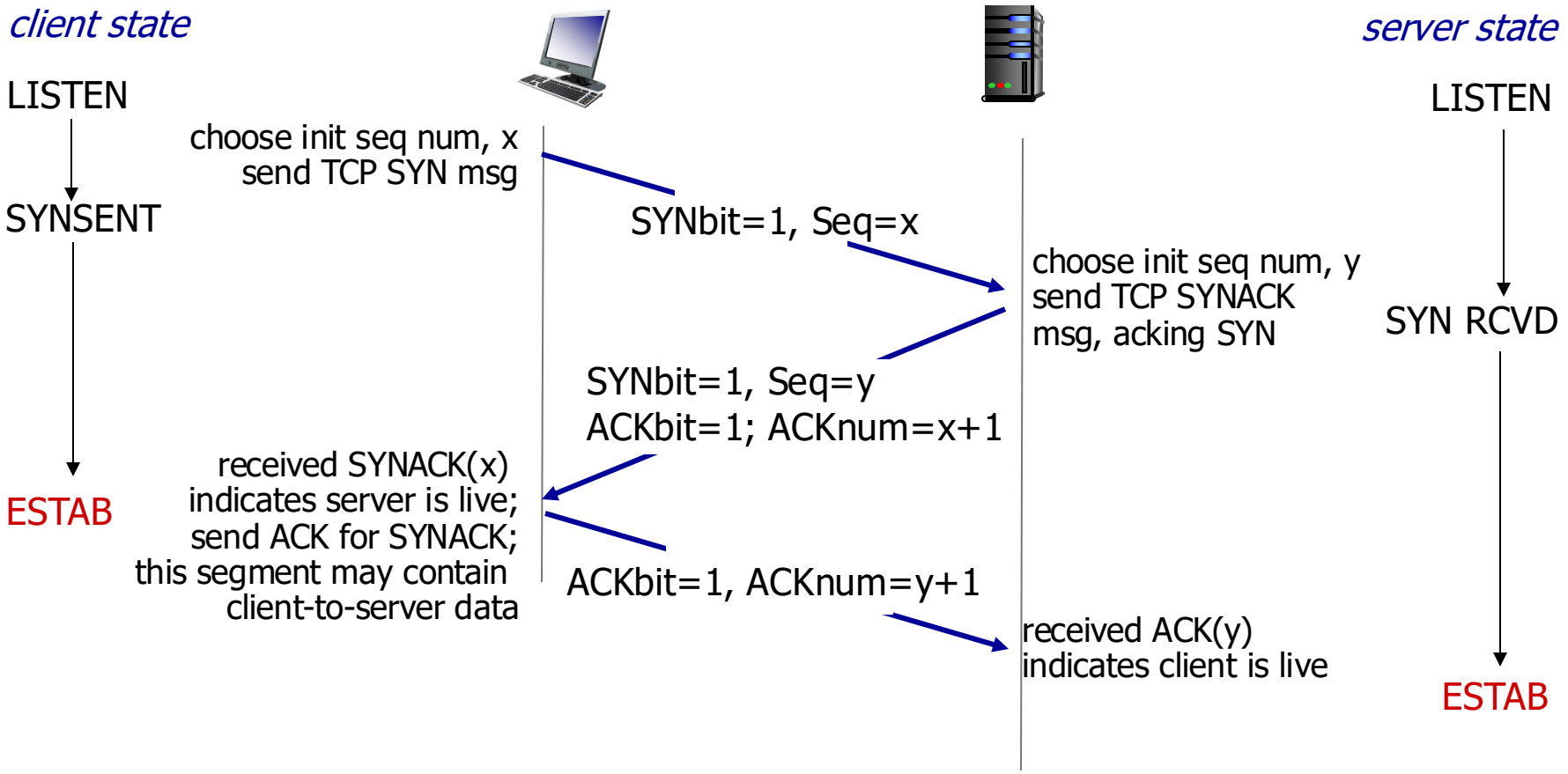
TCP 3-way handshake



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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

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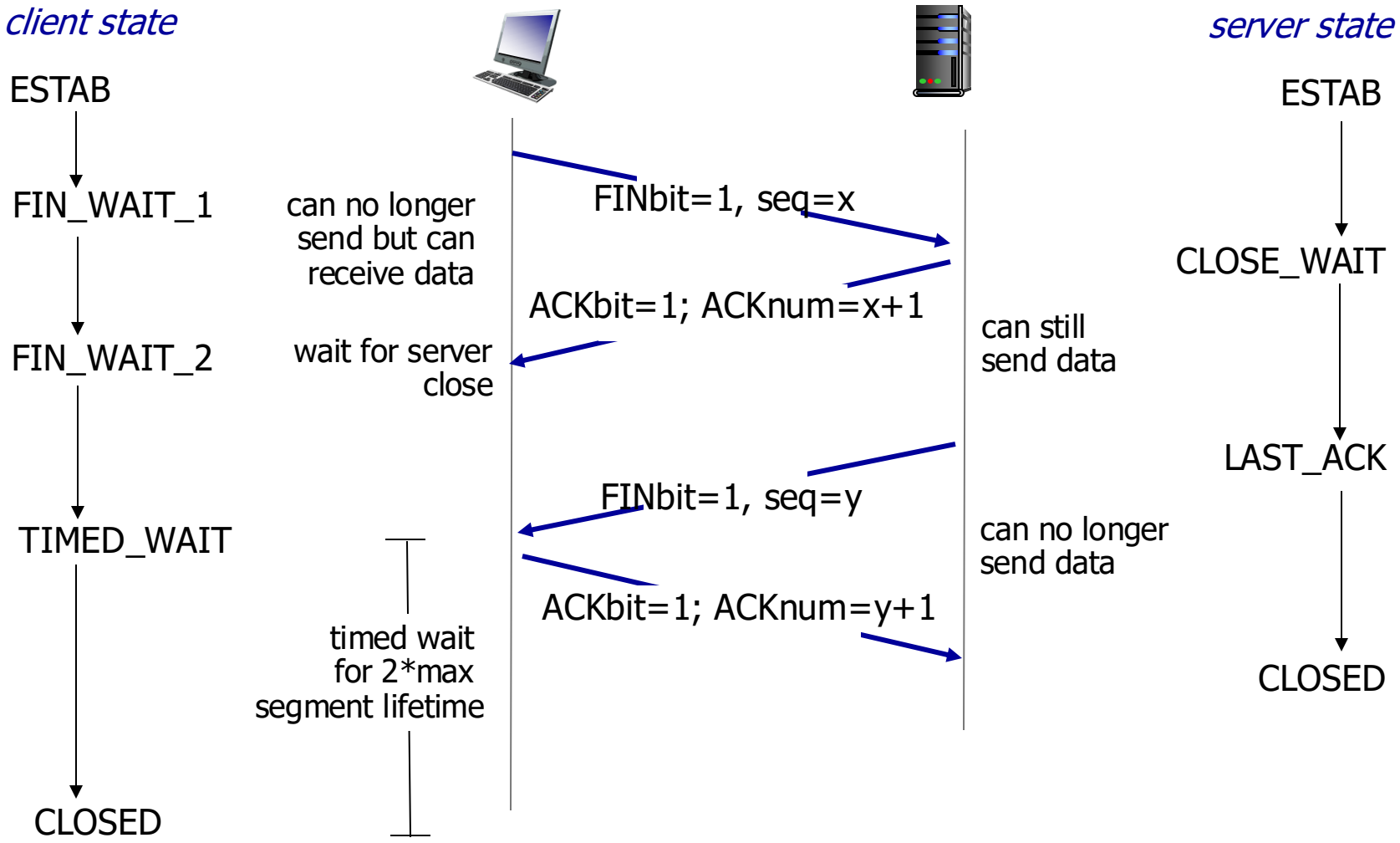
TCP: closing a connection



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