Application layer: overview

- Principles of network applications
- Web and HTTP
- E-mail, SMTP, IMAP
- The Domain Name System DNS
- P2P applications
- video streaming and content distribution networks (CDN)
- socket programming with UDP and TCP

Video Streaming and CDNs: context

- stream video traffic: major consumer of Internet bandwidth
  - Netflix, YouTube, Amazon Prime: 80% of residential ISP traffic (2020)
- challenge: scale - how to reach ~1B users?
  - single mega-video server won't work (why?)
- challenge: heterogeneity
  - different users have different capabilities (e.g., wired versus mobile; bandwidth rich versus bandwidth poor)
- *solution: distributed, application-level infrastructure*

Multimedia: video

- video: sequence of images displayed at constant rate
  - e.g., 24 images/sec or frames / sec (FPS)
- digital image: array of pixels
  - each pixel represented by bits
- coding: use redundancy *within* and *between* images to decrease # 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*

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

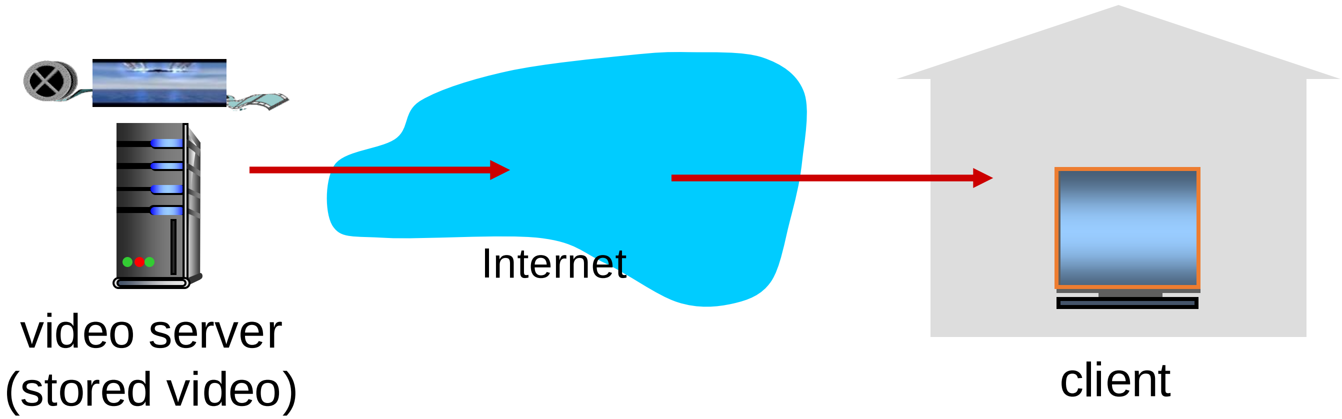
frame *i+1*

Multimedia: video

- CBR: (constant bit rate): video encoding rate fixed
- VBR: (variable bit rate): video encoding rate changes as amount of spatial, temporal coding changes
- examples:
  - MPEG 1 (CD-ROM) 1.5 Mbps
  - MPEG2 (DVD) 3-6 Mbps
  - MPEG4 (often used in Internet, 64Kbps – 12 Mbps)

Streaming stored video

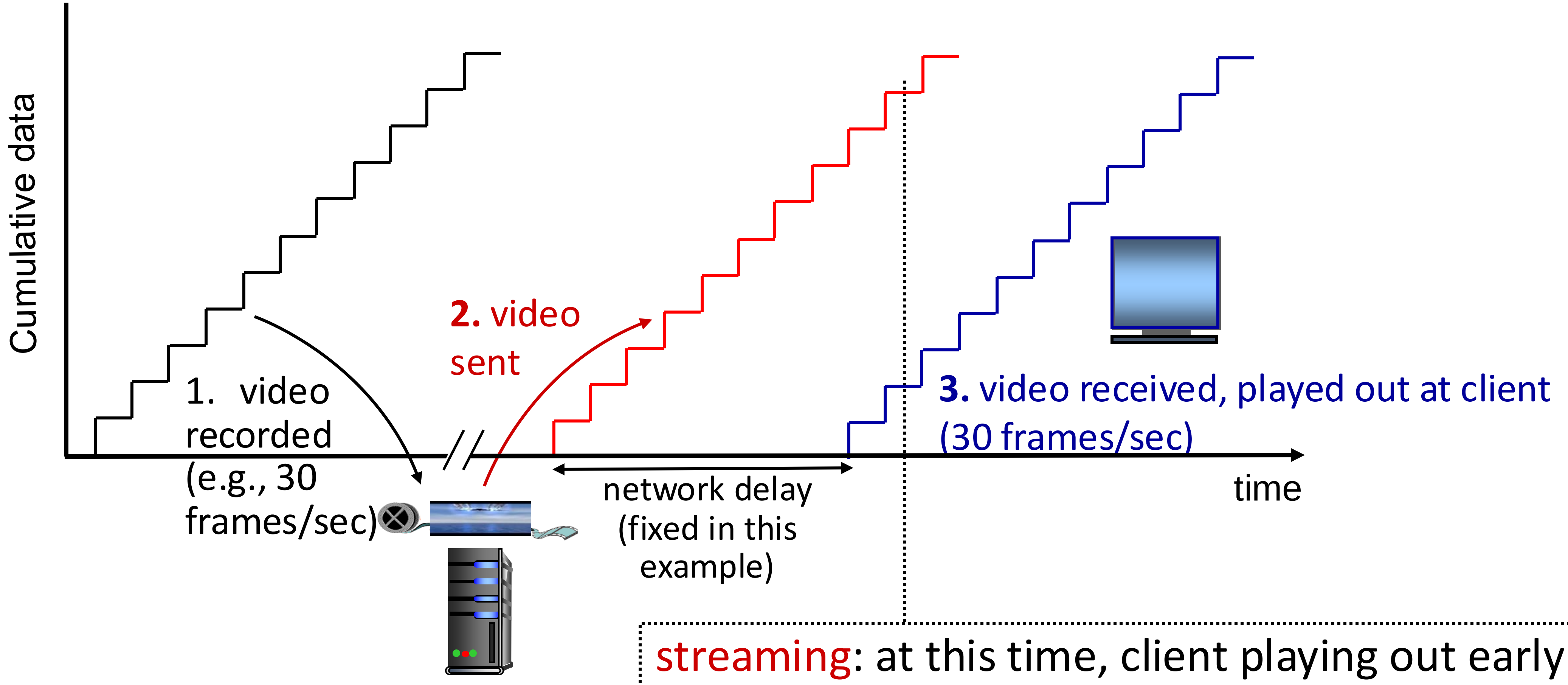
simple scenario:



Main challenges:

- server-to-client bandwidth will *vary* over time, with changing network congestion levels (in house, in access network, in network core, at video server)
- packet loss and delay due to congestion will delay playout, or result in poor video quality

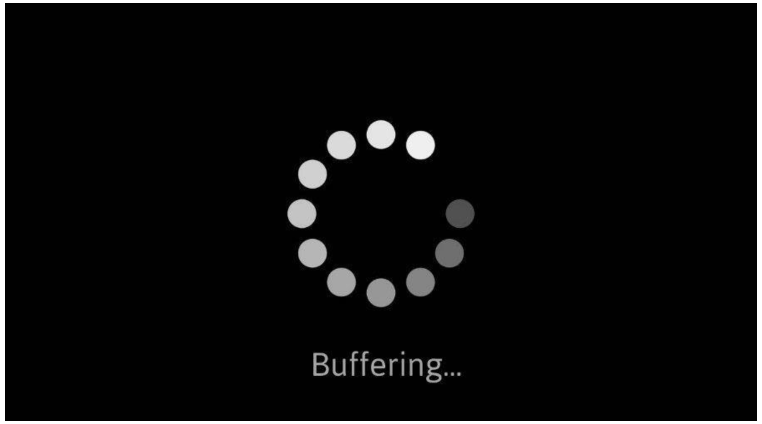
Streaming stored video



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

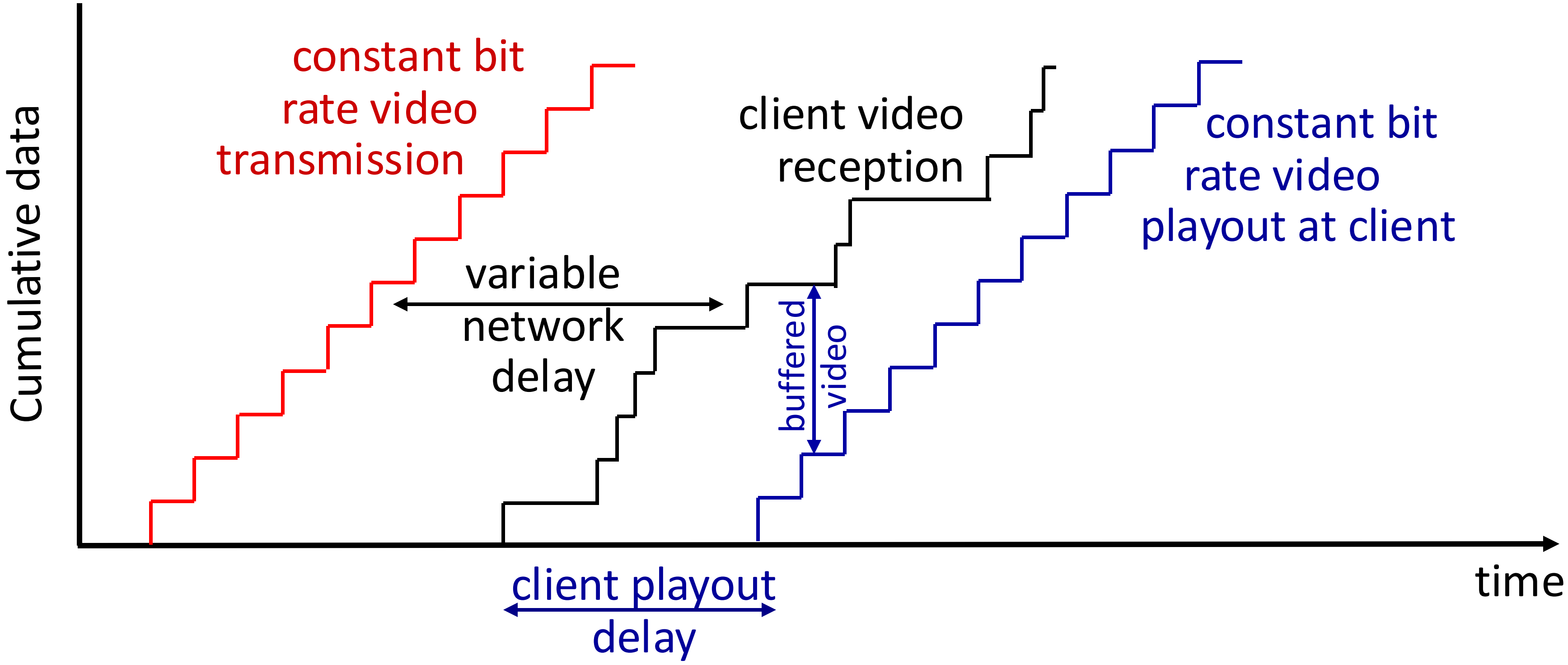
Streaming stored video: challenges

- continuous playout constraint: once client playout begins, playback must match original timing
  - … but network delays are variable (jitter), so will need client-side buffer to match playout requirements



- other challenges:
  - client interactivity: pause, fast-forward, rewind, jump through video
  - video packets may be lost, retransmitted

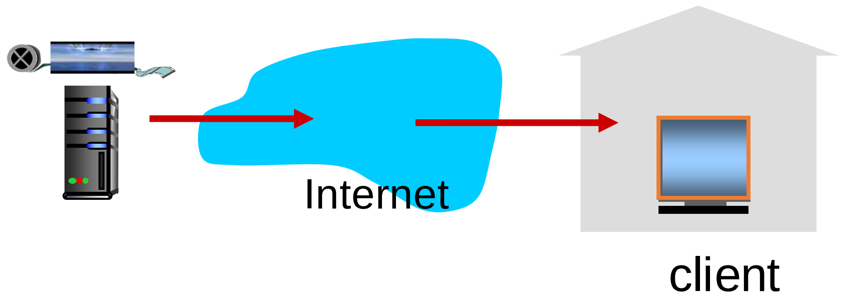
Streaming stored video: playout buffering



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

Streaming multimedia: DASH

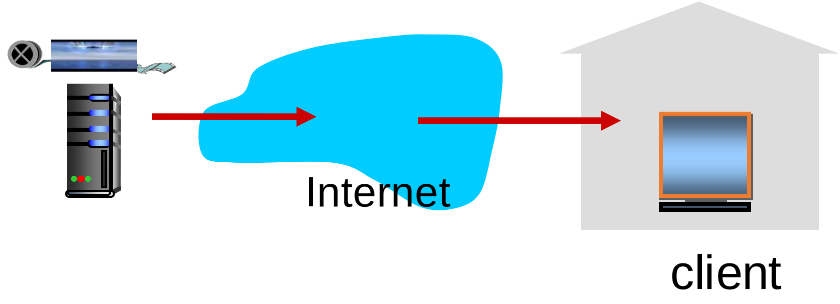
- *DASH: D*ynamic, *A*daptive *S*treaming over *H*TTP
- *server:*
  - divides video file into multiple chunks
  - each chunk stored, encoded at different rates
  - *manifest file:* provides URLs for different chunks



- *client:*
  - periodically measures 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)

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

Content distribution networks (CDNs)

- *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
  - multiple copies of video sent over outgoing link

....quite simply: this solution *doesn't scale*

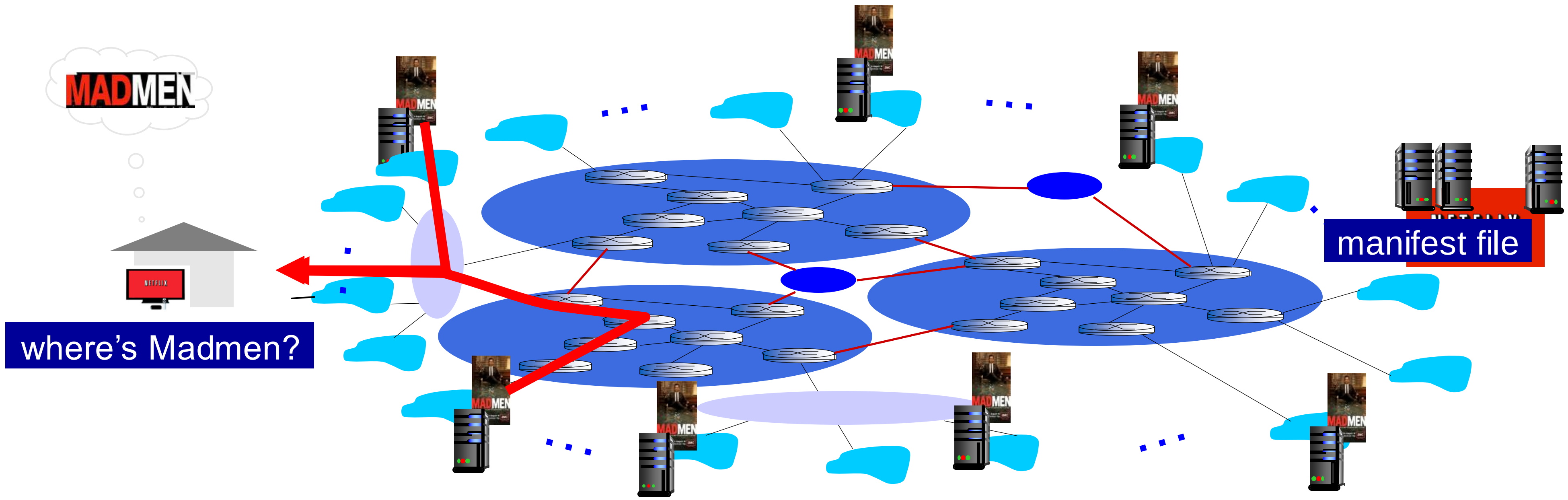
Content distribution networks (CDNs)

- *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)*

- *enter deep:* push CDN servers deep into many access networks
  - close to users
  - Akamai: 240,000 servers deployed in more than 120 countries (2015)
- *bring home:* smaller number (10's) of larger clusters in POPs near (but not within) access networks
  - used by Limelight

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



Content distribution networks (CDNs)

Internet host-host communication as a service

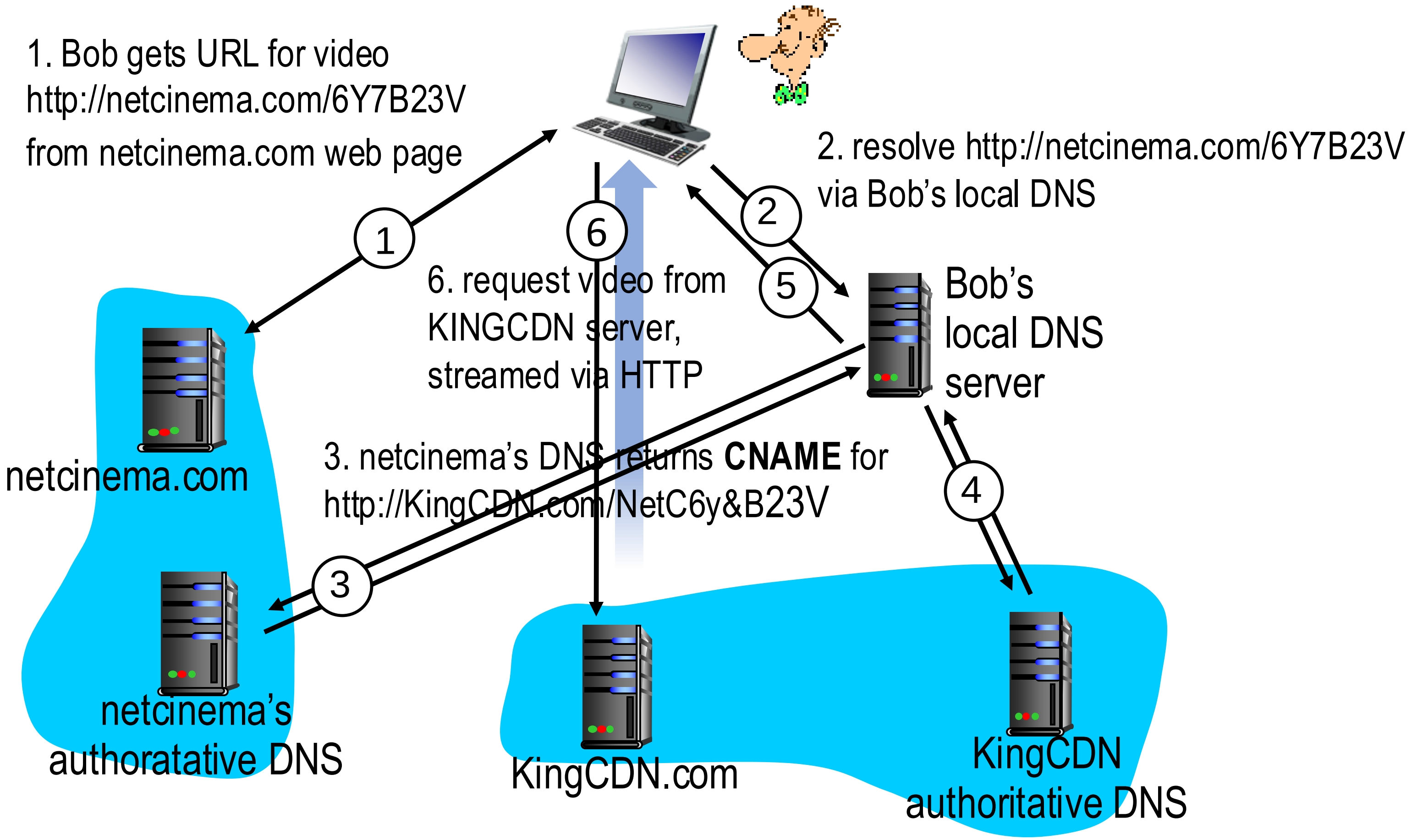
OTT challenges: coping with a congested Internet

- from which CDN node to retrieve content?
- viewer behavior in presence of congestion?
- what content to place in which CDN node?

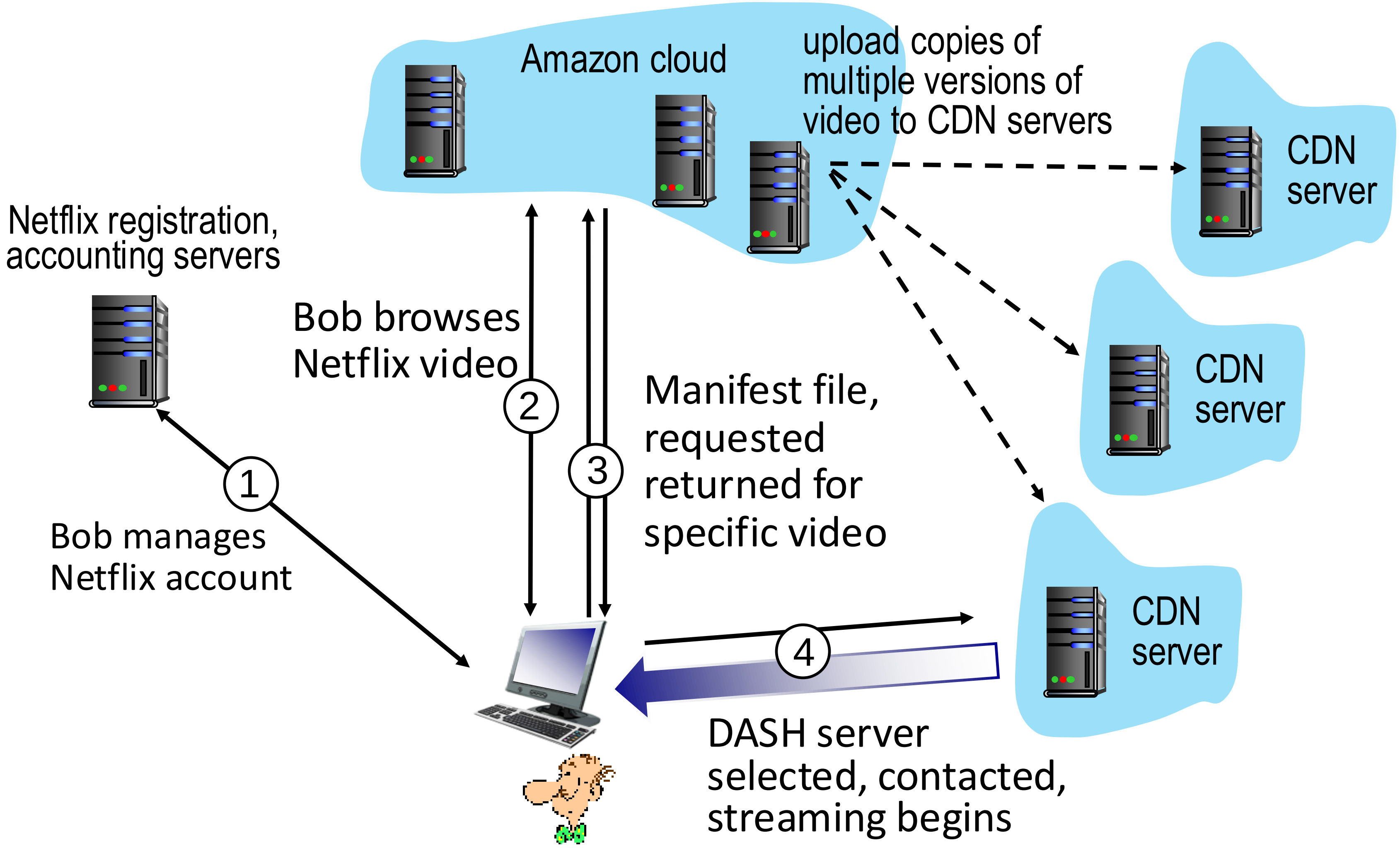
CDN content access: a closer look

Bob (client) requests video http://netcinema.com/6Y7B23V

- video stored in CDN at http://KingCDN.com/NetC6y&B23V



Case study: Netflix



Application Layer: Overview

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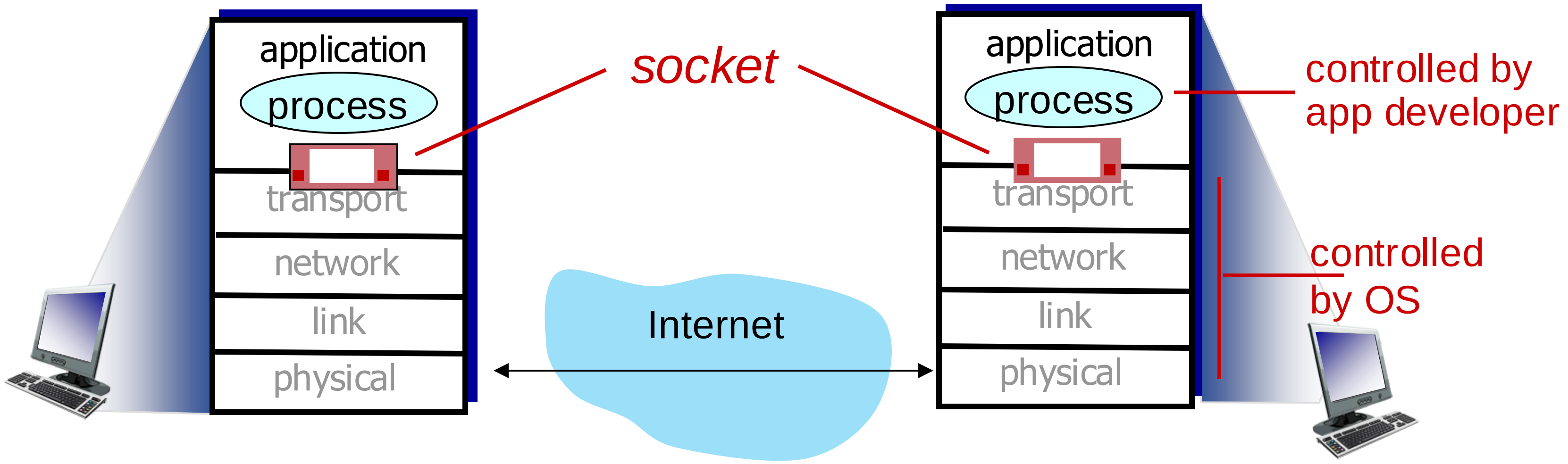
Socket programming

goal: learn how to build client/server applications that communicate using sockets

socket: door between application process and end-end-transport protocol



Socket programming

Two socket types for two transport services:

- *UDP:* unreliable datagram
- *TCP:* reliable, byte stream-oriented

Application Example (lower case -> upper case conversion):

1. client reads a line of characters (data) from its keyboard and sends data to server
2. server receives the data and converts characters to uppercase
3. server sends modified data to client
4. client receives modified data and displays line on its screen

Socket programming with UDP

UDP: no "connection" between client & server

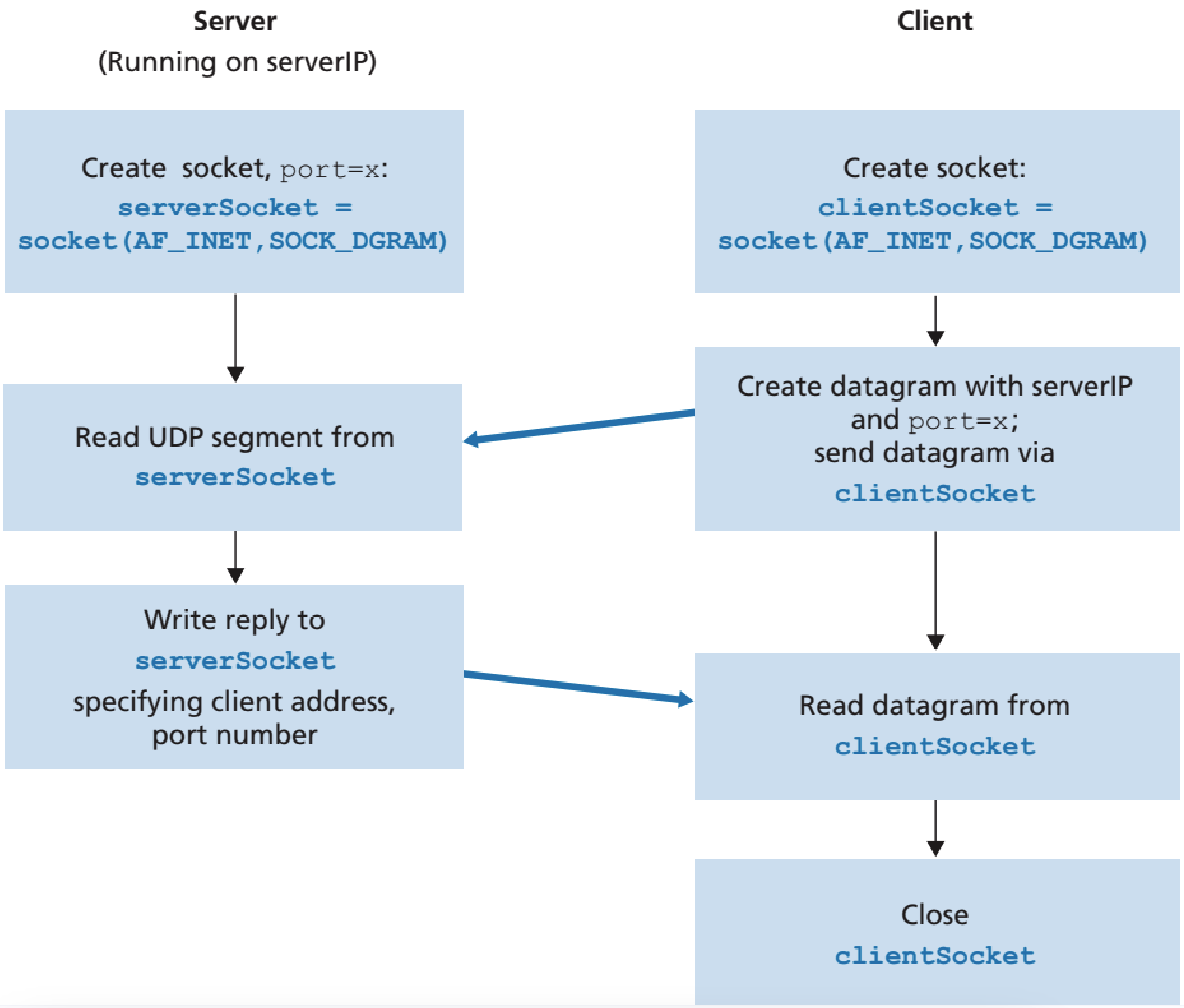
- no handshaking before sending data
- sender explicitly attaches IP destination address and port # to each packet
- receiver extracts sender IP address and port# from received packet

UDP: transmitted data may be lost or received out-of-order

Application viewpoint:

- UDP provides *unreliable* transfer of groups of bytes ("datagrams") between client and server

Client/server socket interaction: UDP



Example app: UDP client

Python UDPClient

include Python's socket library →
create UDP socket for server →
get user keyboard input →
attach server name, port to message; send into socket →
read reply characters from socket into string →
print out received string and close socket →

```
from socket import *
serverName = ‘hostname’
serverPort = 12000
clientSocket = socket(AF_INET,
                      SOCK_DGRAM)
message = raw_input(’Input lowercase sentence:’)
clientSocket.sendto(message.encode(),
                      (serverName, serverPort))
modifiedMessage, serverAddress =
                      clientSocket.recvfrom(2048)
print modifiedMessage.decode()
clientSocket.close()
```

Example app: UDP server

Python UDPServer

```
from socket import *
serverPort = 12000
serverSocket = socket(AF_INET, SOCK_DGRAM)          # create UDP socket
serverSocket.bind(('', serverPort))                 # bind socket to local port number 12000
print ("The server is ready to receive")
while True:                                         # loop forever
    message, clientAddress = serverSocket.recvfrom(2048)   # Read from UDP socket into message, getting client's address (client IP and port)
    modifiedMessage = message.decode().upper()
    serverSocket.sendto(modifiedMessage.encode(),          # send upper case string back to this client
                        clientAddress)
```

Socket programming with TCP

Client must contact server

- server process must first be running
- server must have created socket (door) that welcomes client's contact

Client contacts server by:

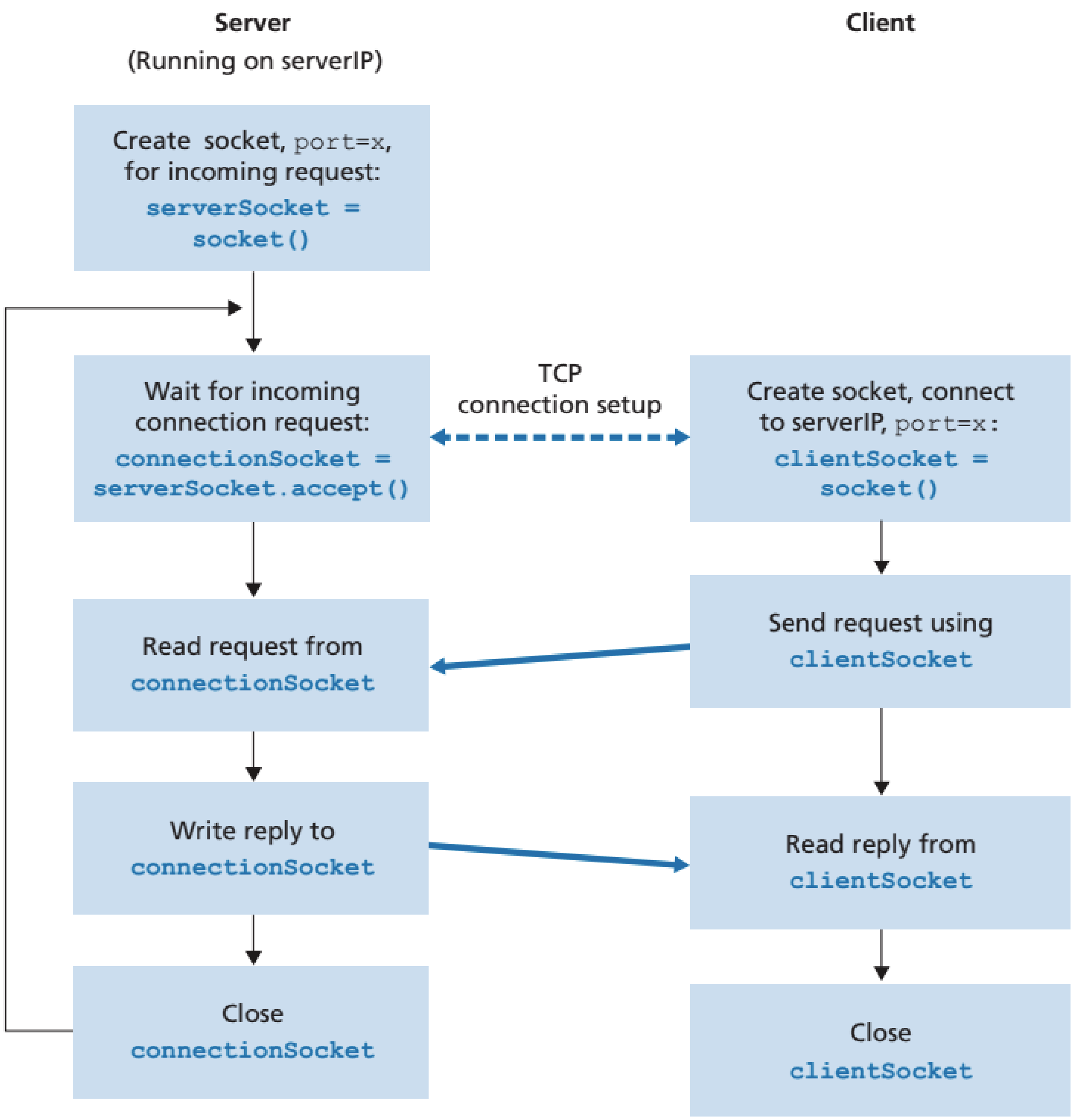
- Creating TCP socket, specifying IP address, port number of server process
- *when client creates socket:* client TCP establishes connection to server TCP

- when contacted by client, *server TCP creates new socket* for server process to communicate with that particular client
  - allows server to talk with multiple clients
  - **source port numbers** used to distinguish clients (more in Chap 3)

Application viewpoint

TCP provides reliable, in-order byte-stream transfer ("pipe") between client and server

Client/server socket interaction: TCP



Example app: TCP client

Python TCPClient

create TCP socket for server, remote port 12000

No need to attach server name, port

```
from socket import *
serverName = ’servername’
serverPort = 12000
clientSocket = socket(AF_INET, SOCK_STREAM)
clientSocket.connect((serverName,serverPort))
sentence = raw_input(‘Input lowercase sentence:’)
clientSocket.send(sentence.encode())
modifiedSentence = clientSocket.recv(1024)
print (‘From Server:’, modifiedSentence.decode())
clientSocket.close()
```

Example app: TCP server

Python TCPServer

create TCP welcoming socket → `serverSocket = socket(AF_INET,SOCK_STREAM)`

server begins listening for incoming TCP requests → `serverSocket.listen(1)`

loop forever → `while True:`

server waits on accept() for incoming requests, new socket created on return → `connectionSocket, addr = serverSocket.accept()`

read bytes from socket (but not address as in UDP) → `sentence = connectionSocket.recv(1024).decode()`

close connection to this client (but *not* welcoming socket) → `connectionSocket.close()`

```
 from socket import *
serverPort = 12000
serverSocket = socket(AF_INET,SOCK_STREAM)
serverSocket.bind((‘’,serverPort))
serverSocket.listen(1)
print ‘The server is ready to receive’
while True:
     connectionSocket, addr = serverSocket.accept()

     sentence = connectionSocket.recv(1024).decode()
     capitalizedSentence = sentence.upper()
     connectionSocket.send(capitalizedSentence.
                                                encode())
     connectionSocket.close()
```

Chapter 2: 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, IMAP
  - DNS
  - P2P: BitTorrent
- video streaming, CDNs
- socket programming: TCP, UDP sockets

Chapter 2: Summary

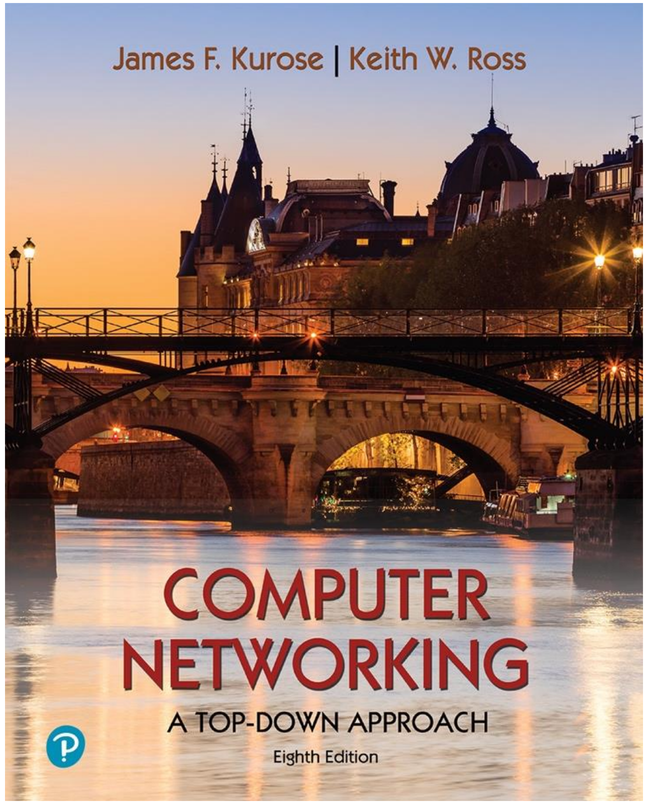
Most importantly: learned about *protocols*!

- typical request/reply message exchange:
  - client requests info or service
  - server responds with data, status code
- message formats:
  - *headers*: fields giving info about data
  - *data:* info(payload) being communicated

important themes:

- centralized vs. decentralized
- stateless vs. stateful
- scalability
- reliable vs. unreliable message transfer
- “complexity at network edge”

Chapter 3
Transport Layer



Computer Networking: A Top-Down Approach
8th edition
Jim Kurose, Keith Ross
Pearson, 2020

Transport layer: overview

Our goal:

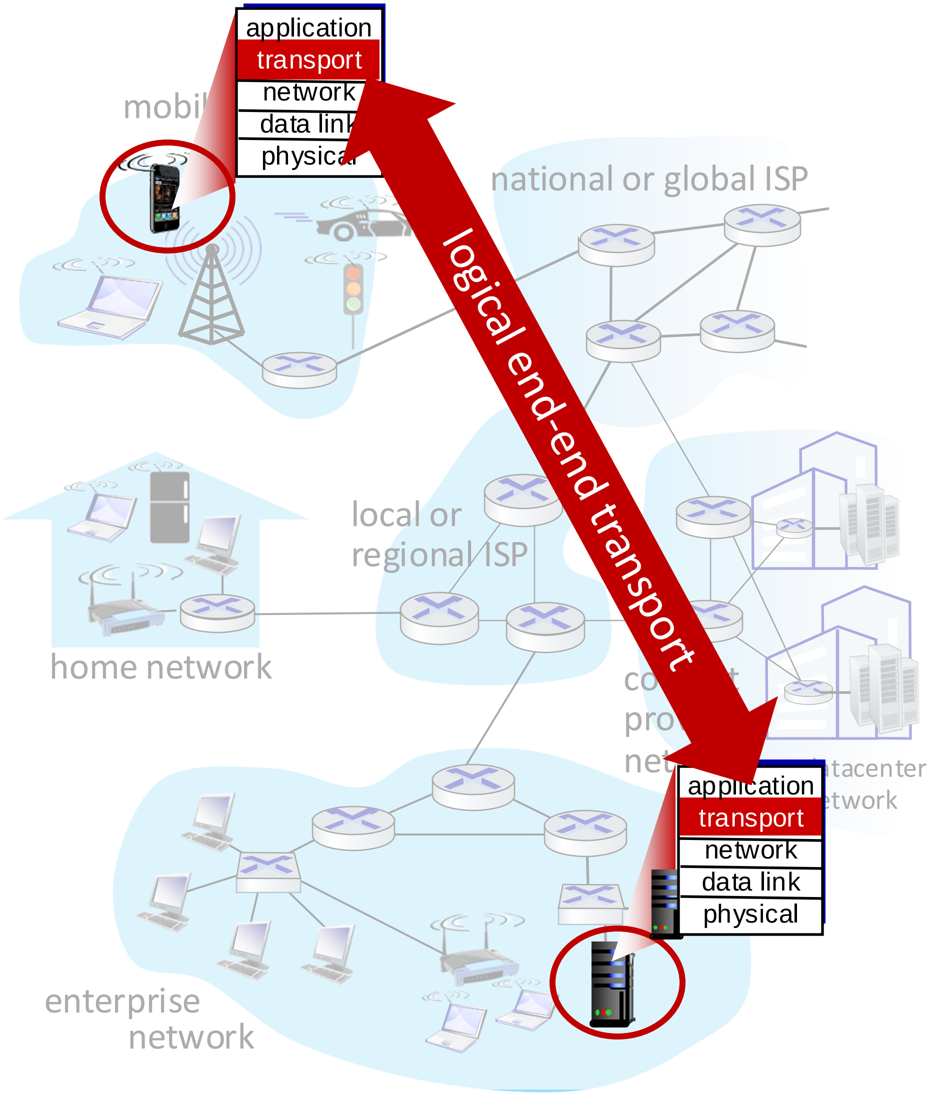
- understand principles behind transport layer services:
  - multiplexing, demultiplexing
  - reliable data transfer
  - flow control
  - congestion control
- learn about Internet transport layer protocols:
  - UDP: connectionless transport
  - TCP: connection-oriented reliable transport
  - TCP congestion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

Transport layer: roadmap

- Transport-layer services
- Multiplexing and demultiplexing
- Connectionless transport: UDP
- Principles of reliable data transfer
- Connection-oriented transport: TCP
- Principles of congestion control
- TCP congestion control
- Evolution of transport-layer functionality

Transport services and protocols

- provide *logical communication* between application processes running on different hosts
- transport protocols actions in end systems:
  - sender: breaks application messages into *segments*, passes to network layer
  - receiver: reassembles segments into messages, passes to application layer
- two transport protocols available to Internet applications
  - TCP, UDP



Transport vs. network layer services and protocols

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

household analogy:

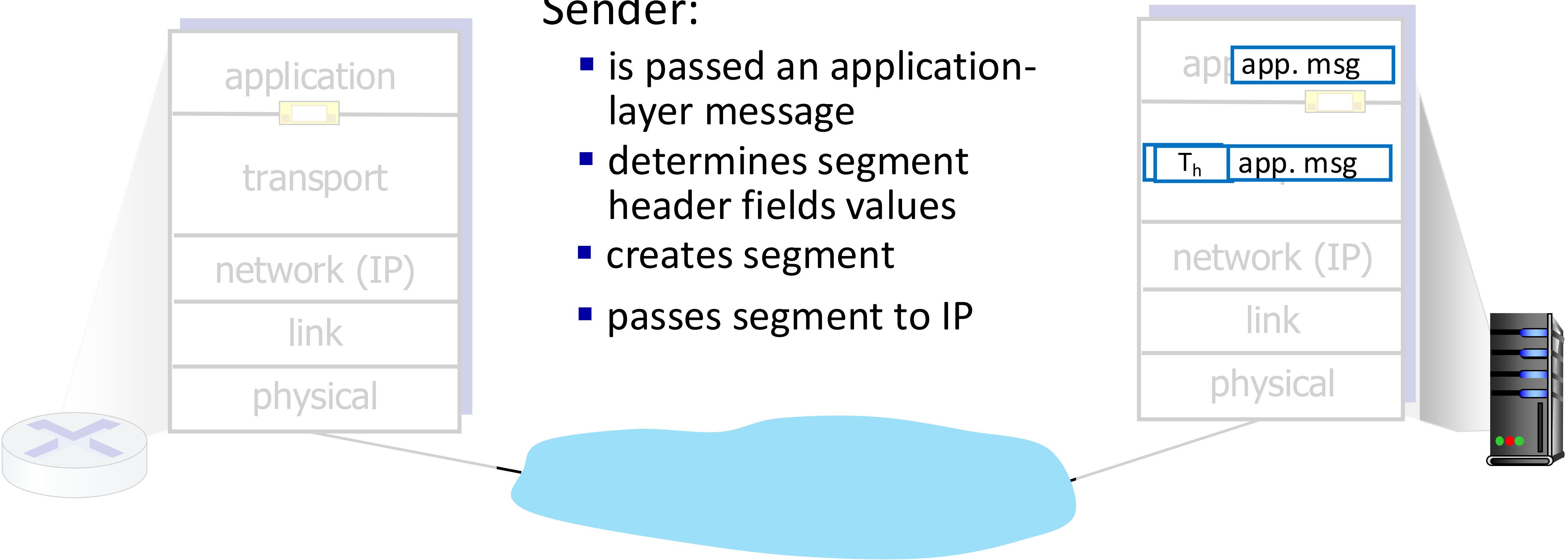
12 kids in Ann's house sending letters to 12 kids in Bill's house:

- hosts = houses
- processes = kids
- app messages = letters in envelopes
- transport protocol = Ann and Bill who demux to in-house siblings
- network-layer protocol = postal service

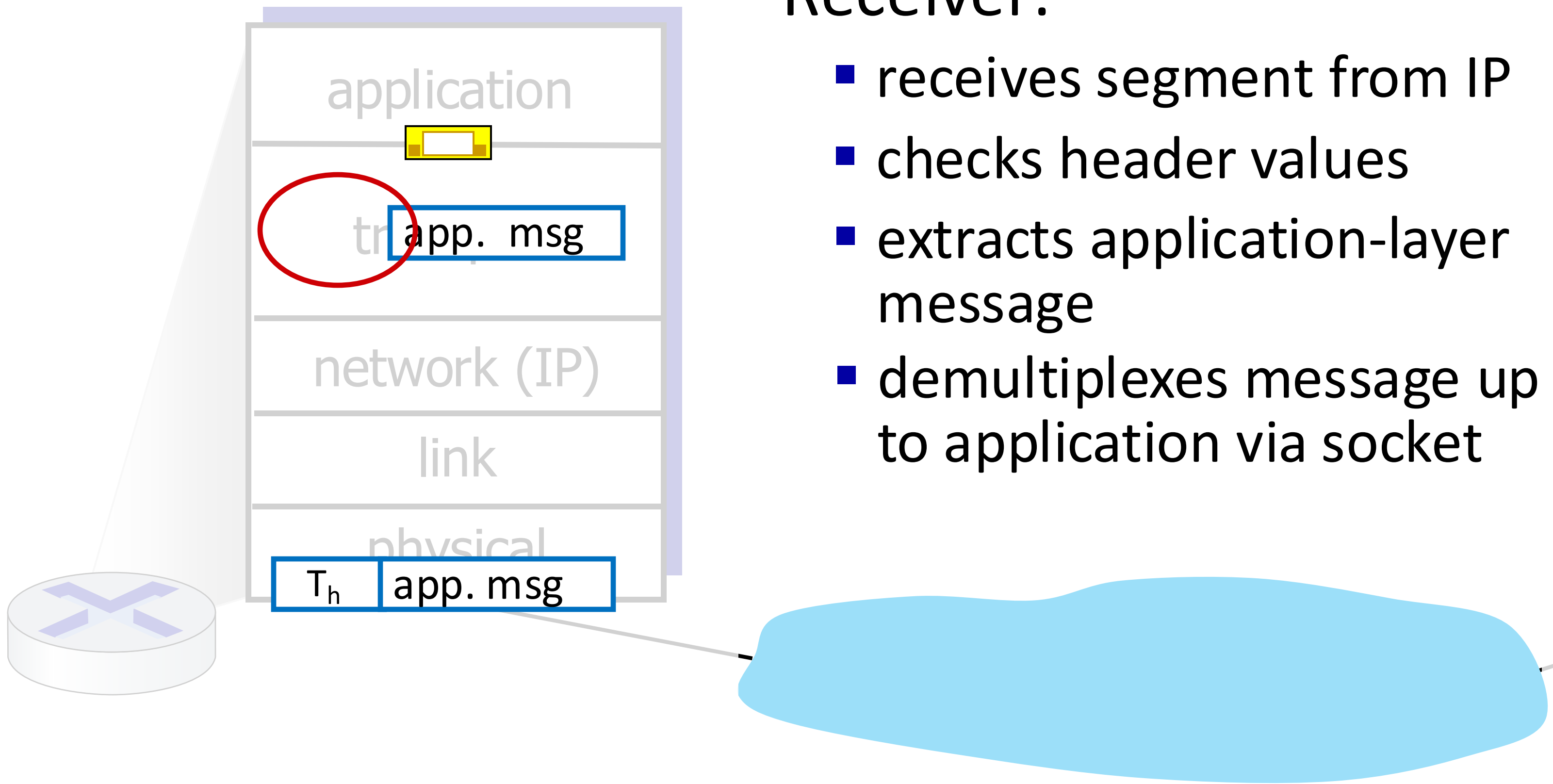
Transport Layer Actions

Sender:

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

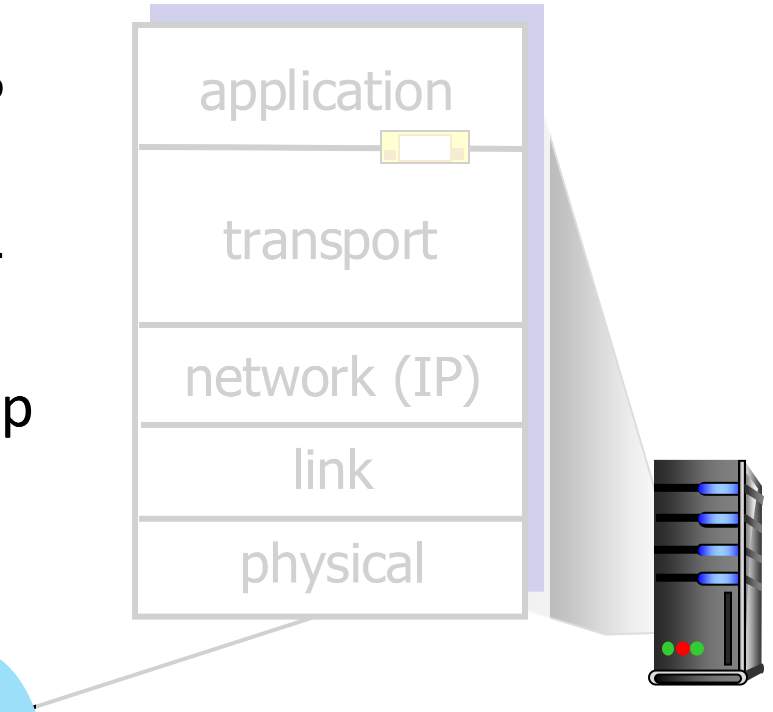


Transport Layer Actions



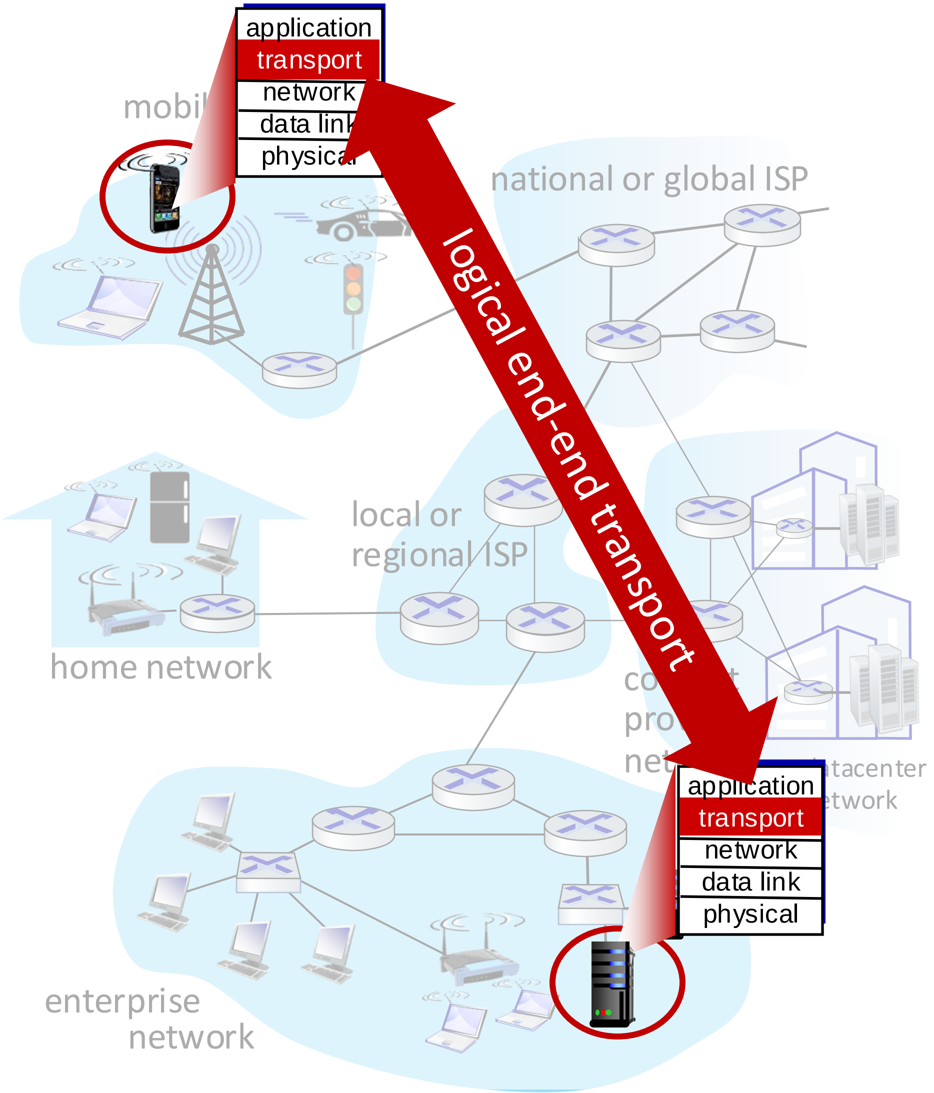
Receiver:

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



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
  - no-frills extension of "best-effort" IP
- services not available:
  - delay guarantees
  - bandwidth guarantees



Chapter 3: roadmap

- Transport-layer services
- Multiplexing and demultiplexing
- Connectionless transport: UDP
- Principles of reliable data transfer
- Connection-oriented transport: TCP
- Principles of congestion control
- TCP congestion control
- Evolution of transport-layer functionality

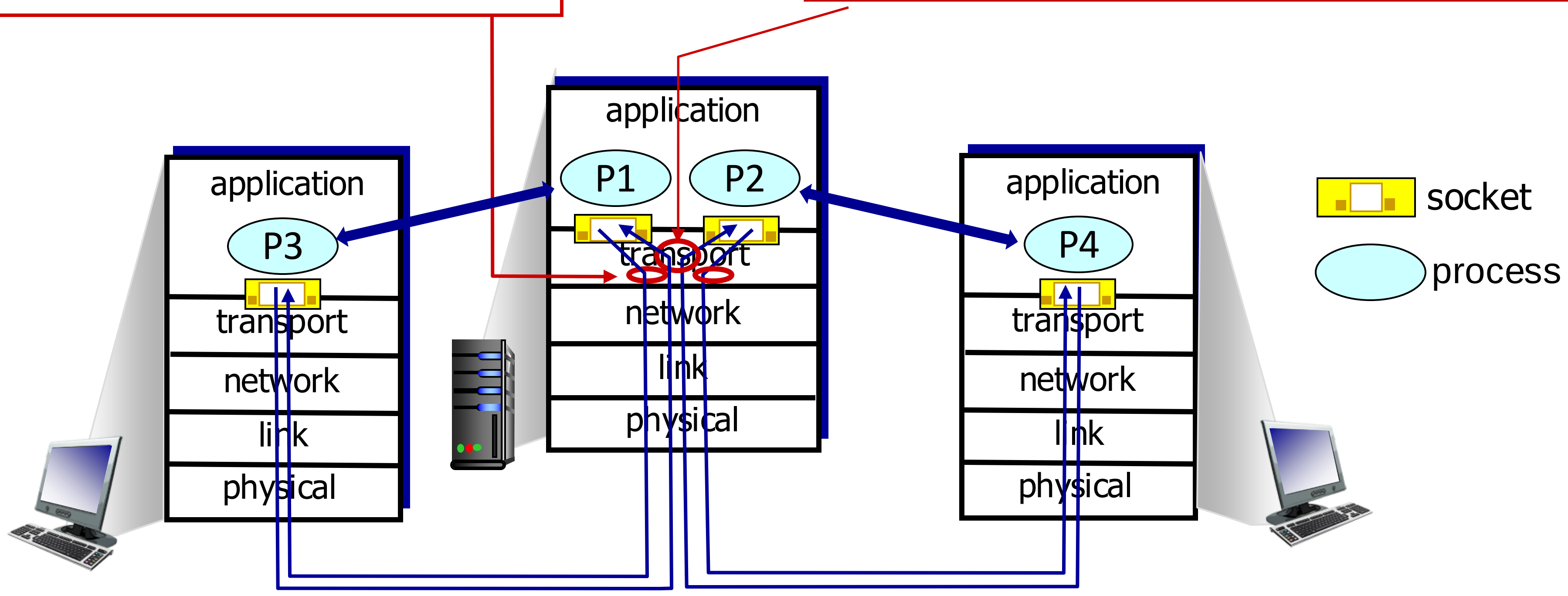
Multiplexing/demultiplexing

multiplexing at sender:

handle data from multiple sockets, add transport header (later used for demultiplexing)

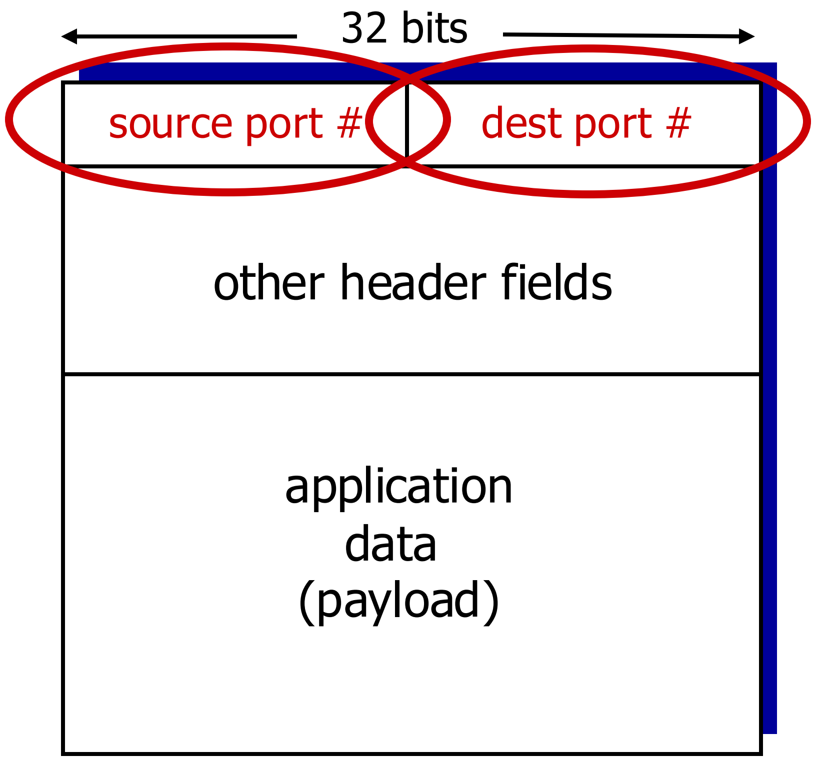
demultiplexing at receiver:

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

Connectionless demultiplexing

Recall:

- when creating socket, must specify *host-local* port #:

```
DatagramSocket mySocket1
  = new DatagramSocket(12534);
```

- when creating datagram to send into UDP socket, must specify
  - destination IP address
  - destination port #

when receiving host receives *UDP* segment:

- checks destination port # in segment
- directs UDP segment to socket with that port #

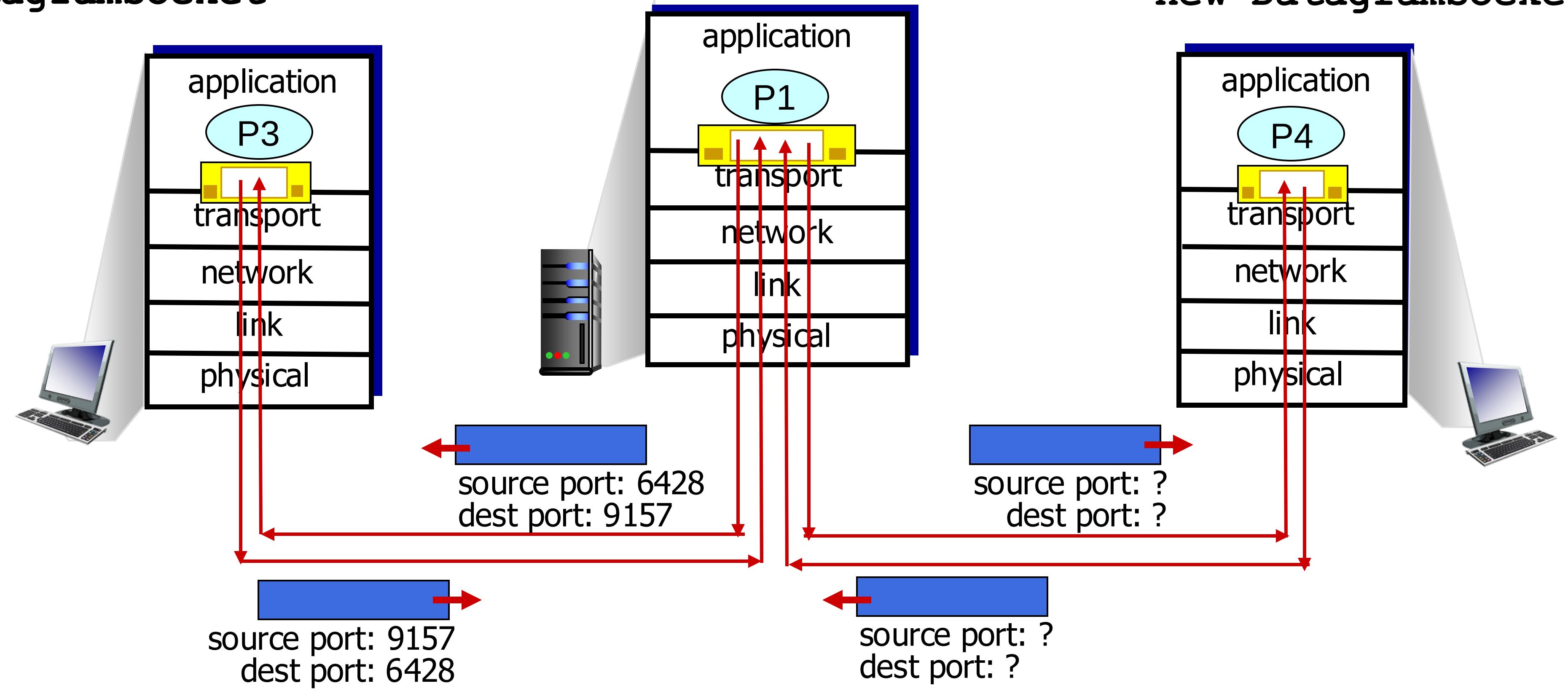
IP/UDP datagrams with *same dest. port #,* but different source IP addresses and/or source port numbers will be directed to *same socket* at receiving host

Connectionless demultiplexing: an example

```
DatagramSocket mySocket1 =
 new DatagramSocket
 (9157);
```

```
DatagramSocket
 serverSocket = new
 DatagramSocket
 (6428);
```

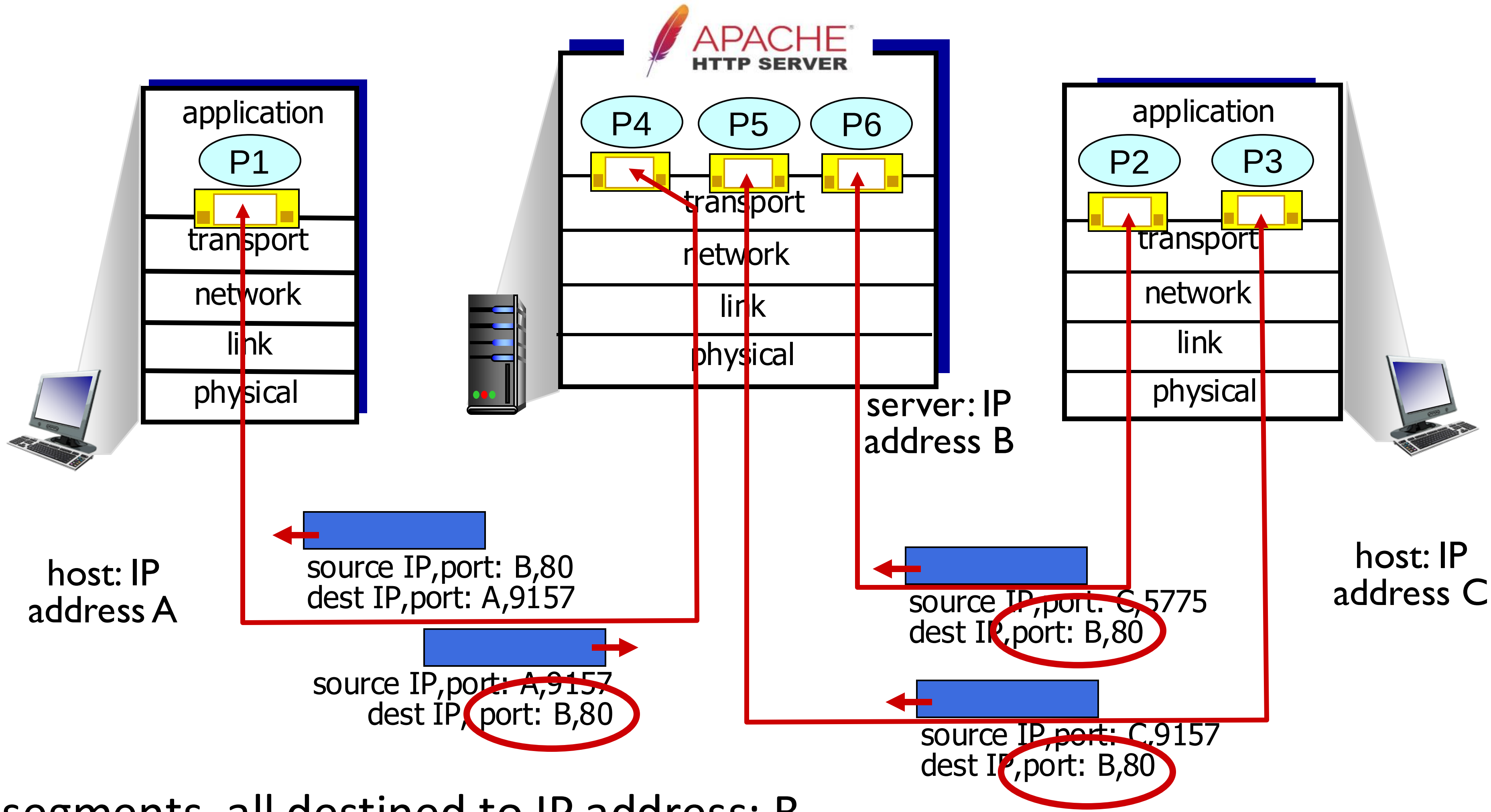
```
DatagramSocket mySocket2 =
 new DatagramSocket (5775);
```



Connection-oriented demultiplexing

- TCP socket identified by 4-tuple:
  - source IP address
  - source port number
  - dest IP address
  - dest port number
- demux: receiver uses *all four values (4-tuple)* to direct segment to appropriate socket
- server may support many simultaneous TCP sockets:
  - each socket identified by its own 4-tuple
  - each socket associated with a different connecting client

Connection-oriented demultiplexing: example



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

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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UDP: User Datagram Protocol

- "no frills," "bare bones" 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

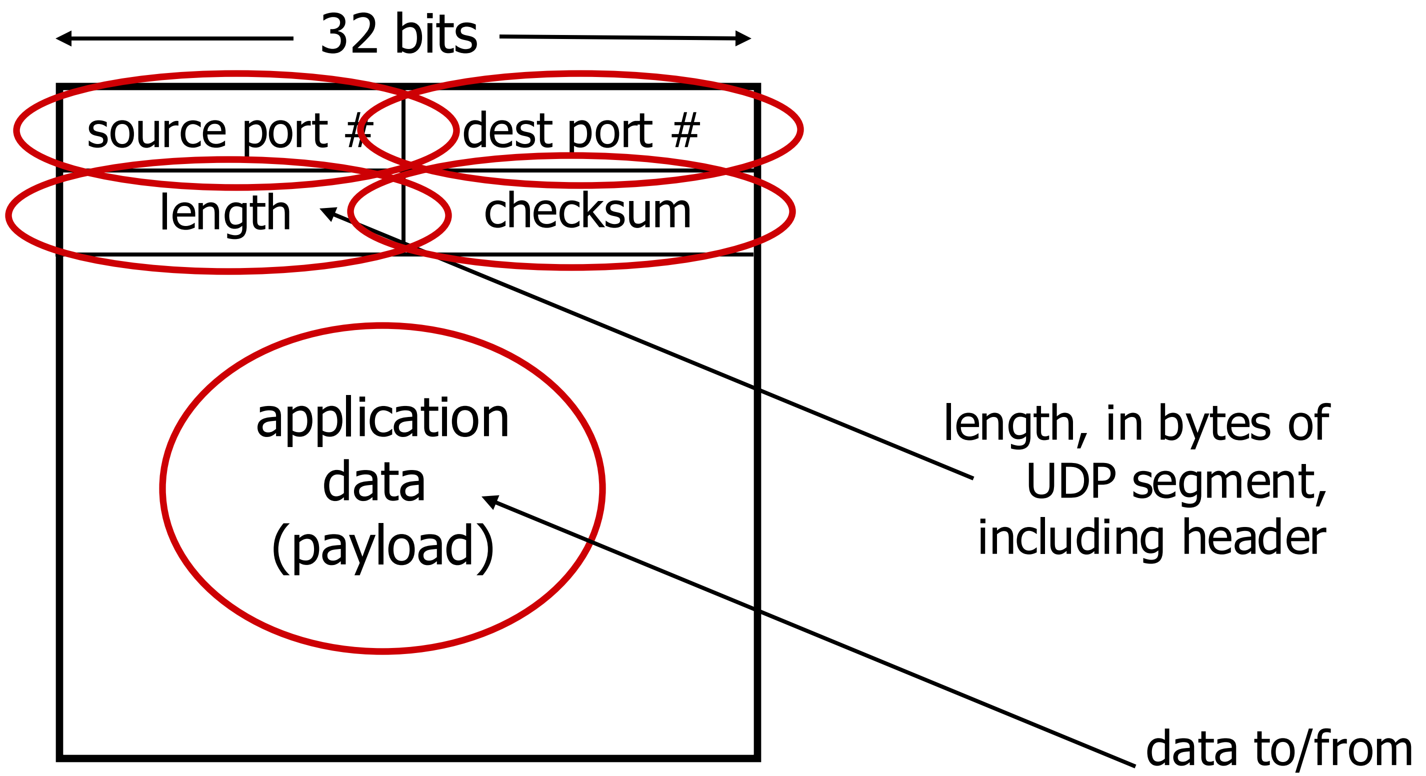
Why is there a UDP?

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

UDP: User Datagram Protocol

- UDP use:
  - streaming multimedia apps (loss tolerant, rate sensitive)
  - DNS (quick lookups)
    - DNS transactions are small. Lost packet -> retransmission request (initiated at the application layer)
  - SNMP
  - HTTP/3
- if reliable transfer needed over UDP (e.g., HTTP/3):
  - add needed reliability at application layer
  - add congestion control at application layer

UDP segment header



UDP segment format

Internet checksum

Goal: detect errors (*i.e.,* flipped bits) in transmitted segment

sender:

- treat contents of UDP segment (including UDP header fields and IP addresses) as sequence of 16-bit integers
- checksum: addition (one's complement sum) of segment content
- checksum value put into UDP checksum field

receiver:

- compute checksum of received segment
- check if computed checksum equals checksum field value:
  - not equal - error detected
  - equal - no error detected. *But maybe errors nonetheless?* More later ….

Internet checksum: an example

example: add two 16-bit integers

		1	1	1	0	0	1	1	0	0	1	1	0	0	1	1	0
		1	1	0	1	0	1	0	1	0	1	0	1	0	1	0	1
wraparound	1	1	0	1	1	1	0	1	1	1	0	1	1	1	0	1	1
sum		1	0	1	1	1	0	1	1	1	0	1	1	1	1	0	0
checksum		0	1	0	0	0	1	0	0	0	1	0	0	0	0	1	1

one's complement: inverting all the bits

Note: when adding numbers, a carryout from the most significant bit needs to be added to the result

Internet checksum: weak protection!

example: add two 16-bit integers

		1	1	1	0	0	1	1	0	0	1	1	0	0	1	1	0	0 1
		1	1	0	1	0	1	0	1	0	1	0	1	0	1	0	1	1 0
wraparound	1	1	0	1	1	1	0	1	1	1	0	1	1	1	0	1	1	
sum		1	0	1	1	1	0	1	1	1	0	1	1	1	1	0	0	
checksum		0	1	0	0	0	1	0	0	0	1	0	0	0	0	1	1	

Even though numbers have changed (bit flips), *no* change in checksum!

Summary: UDP

- "no frills" protocol:
  - segments may be lost, delivered out of order
  - best effort service: "send and hope for the best"
- UDP has its plusses:
  - no setup/handshaking needed (no RTT incurred)
  - can function when network service is compromised
  - helps with reliability (checksum)
- build additional functionality on top of UDP in application layer (e.g., HTTP/3)