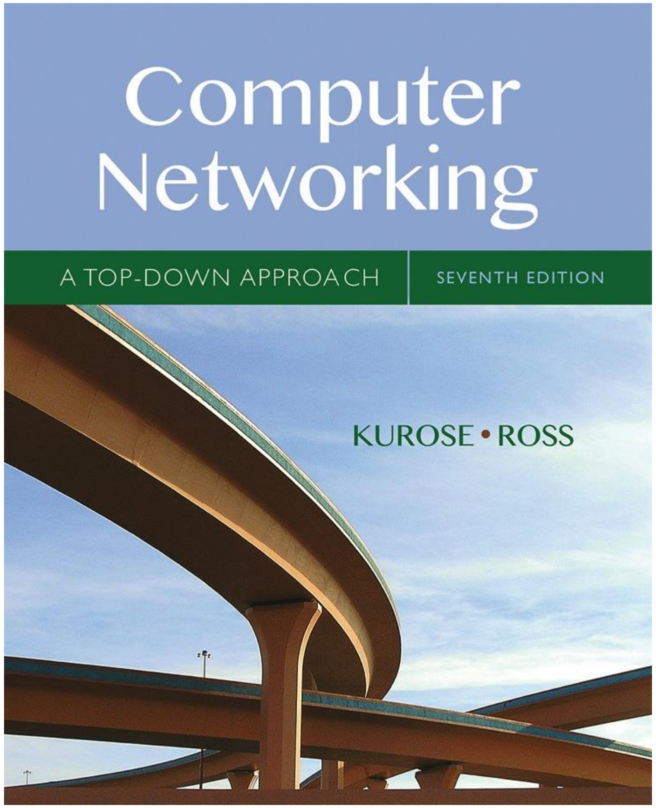
Chapter 2
Application Layer



Computer Networking: A Top Down Approach

7th edition
Jim Kurose, Keith Ross
Pearson/Addison Wesley
April 2016

Slides adopted from original ones provided by the textbook authors.

Chapter 2: outline

2.1 principles of network applications

2.2 Web and HTTP

2.3 electronic mail

- SMTP, POP3, IMAP

2.4 DNS

2.5 P2P applications

2.6 video streaming and content distribution networks

2.7 socket programming with UDP and TCP

Application architectures

- Client-server
  - Always-on server, intermittently connected client.
  - Servers are bottlenecks.
- Peer-to-peer (P2P)
  - Peers intermittently connected.
  - Highly scalable but difficult to manage.

App-layer protocol defines

- types of messages exchanged,
  - e.g., request, response
- message syntax:
  - what fields in messages & how fields are delineated
- message semantics
  - meaning of information in fields
- rules for when and how processes send & respond to messages

open protocols:

- defined in RFCs
- allows for interoperability
- e.g., HTTP, SMTP

proprietary protocols:

- e.g., Skype

Sockets

- Process sends/receives messages to/from its socket.
  - Processes are identified by IP addresses and TCP/UDP port numbers.
- OS provides APIs for creating sockets.

Internet transport protocols services

TCP service:

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

UDP service:

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

Chapter 2: outline

2.1 principles of network applications

2.2 Web and HTTP

2.3 electronic mail
- SMTP, POP3, IMAP

2.4 DNS

2.5 P2P applications

2.6 video streaming and content distribution networks

2.7 socket programming with UDP and TCP

HTTP overview

uses TCP:

- client initiates TCP connection (creates socket) to server, port 80
- server accepts TCP connection from client
- HTTP messages (application-layer protocol messages) exchanged between browser (HTTP client) and Web server (HTTP server)
- TCP connection closed

HTTP is "stateless"

- server maintains no information about past client requests

HTTP connections

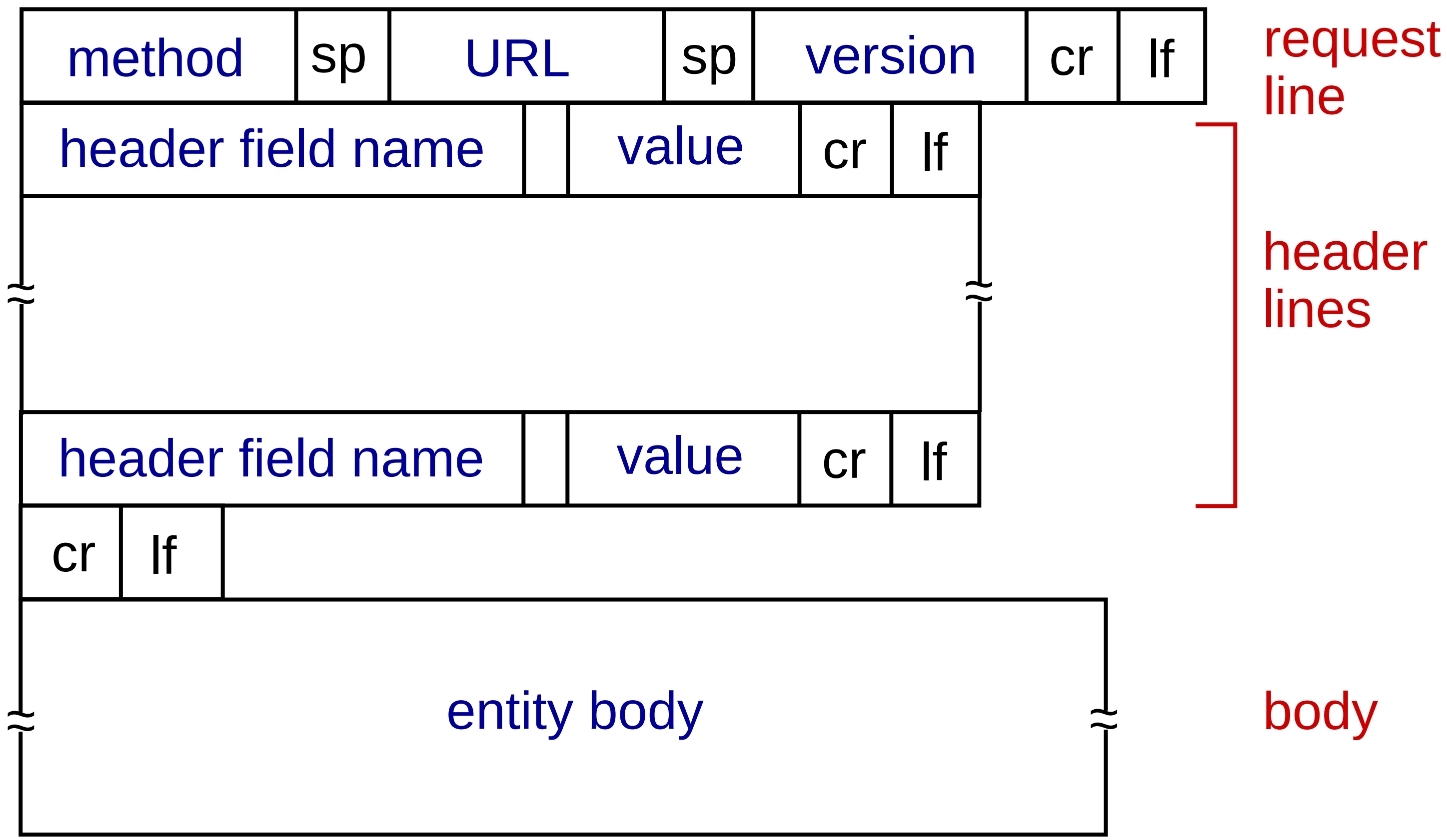
non-persistent HTTP

- at most one object sent over TCP connection
  - connection then closed
- downloading multiple objects required multiple connections
  - use parallel TCP connections to accelerate

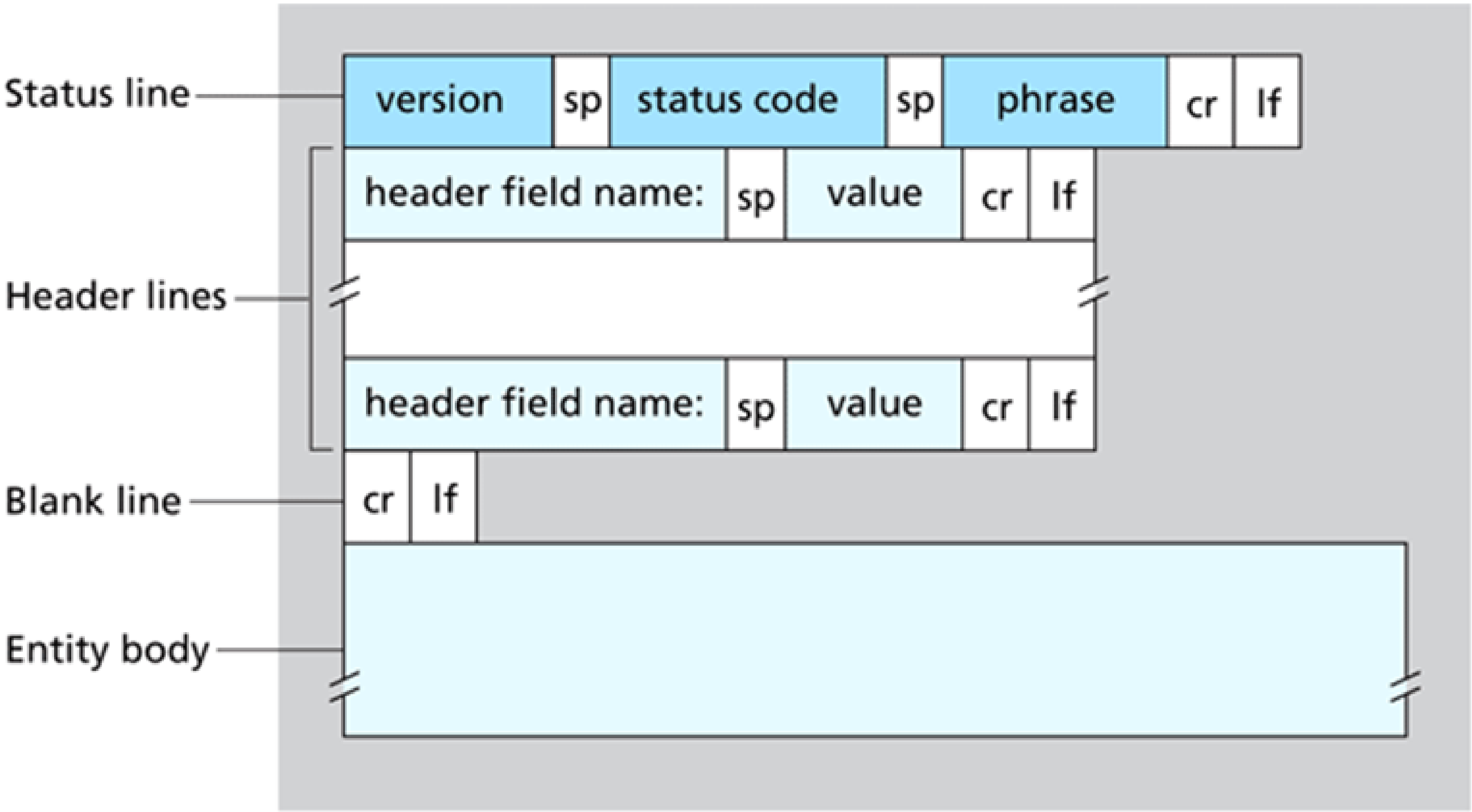
persistent HTTP

- multiple objects can be sent over single TCP connection between client, server

HTTP request message: general format



HTTP response message: general format



User-server state: cookies

- Cookies help web sites remember use states.
- Four components:
  1) set-cookie header line in HTTP *response* message
  2) cookie header line in HTTP *request* message
  3) cookie file kept on user's host, managed by user's browser
  4) back-end database at Web site

Web caches

- Goal: satisfy client request without involving origin server
  - reduce response time and traffic
- user sets browser: Web accesses via cache, browser sends all HTTP requests to cache
  - object in cache: cache returns object
  - else cache requests object from origin server, then returns object to client
- Conditional GET: don't send object if cache has up-to-date cached version

Chapter 2: outline

2.1 principles of network applications

2.2 Web and HTTP

2.3 electronic mail

- SMTP, POP3, IMAP

2.4 DNS

2.5 P2P applications

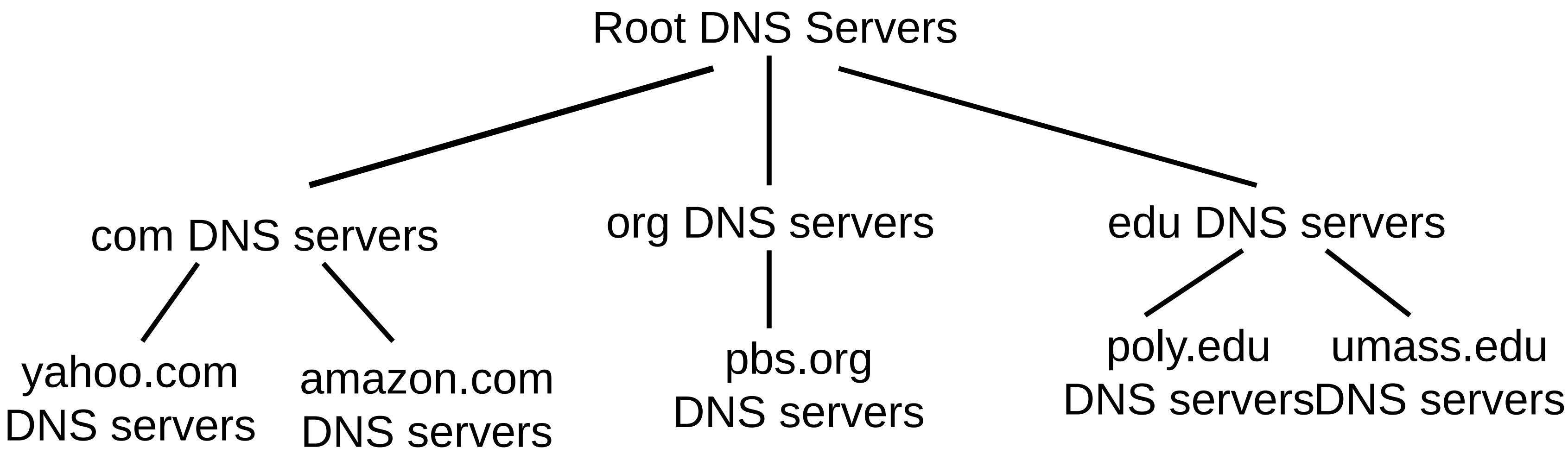
2.6 video streaming and content distribution networks

2.7 socket programming with UDP and TCP

DNS: Domain Name System

- hierarchical distributed database for name-IP translation
- DNS services
  - hostname to IP address translation
  - host aliasing: canonical, alias names
  - mail server aliasing
  - load distribution

Distributed, Hierarchical Database



- Root name servers
- Top-level domain (TLD) servers
- Authoritative DNS servers

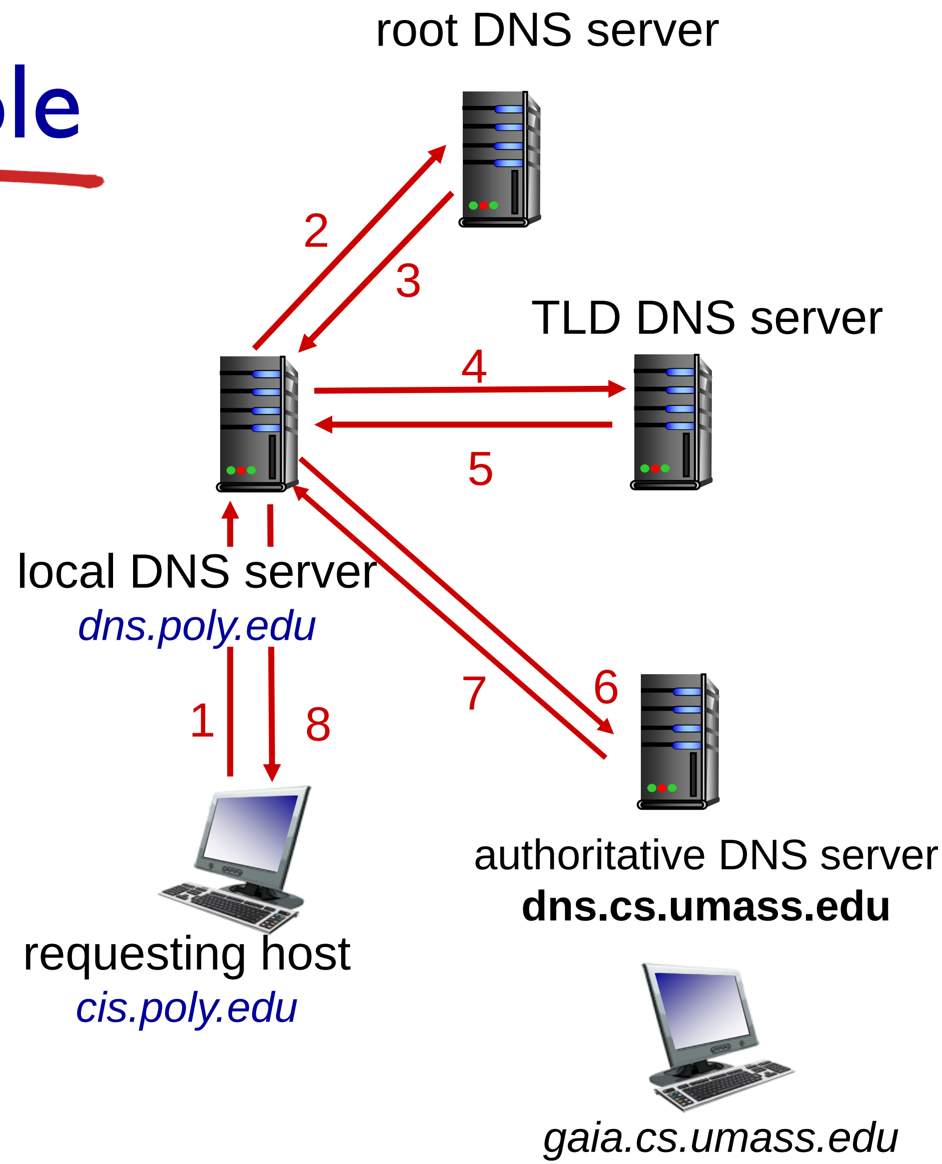
- Local name server

DNS name resolution example

- host at cis.poly.edu wants IP address for gaia.cs.umass.edu

iterated query:

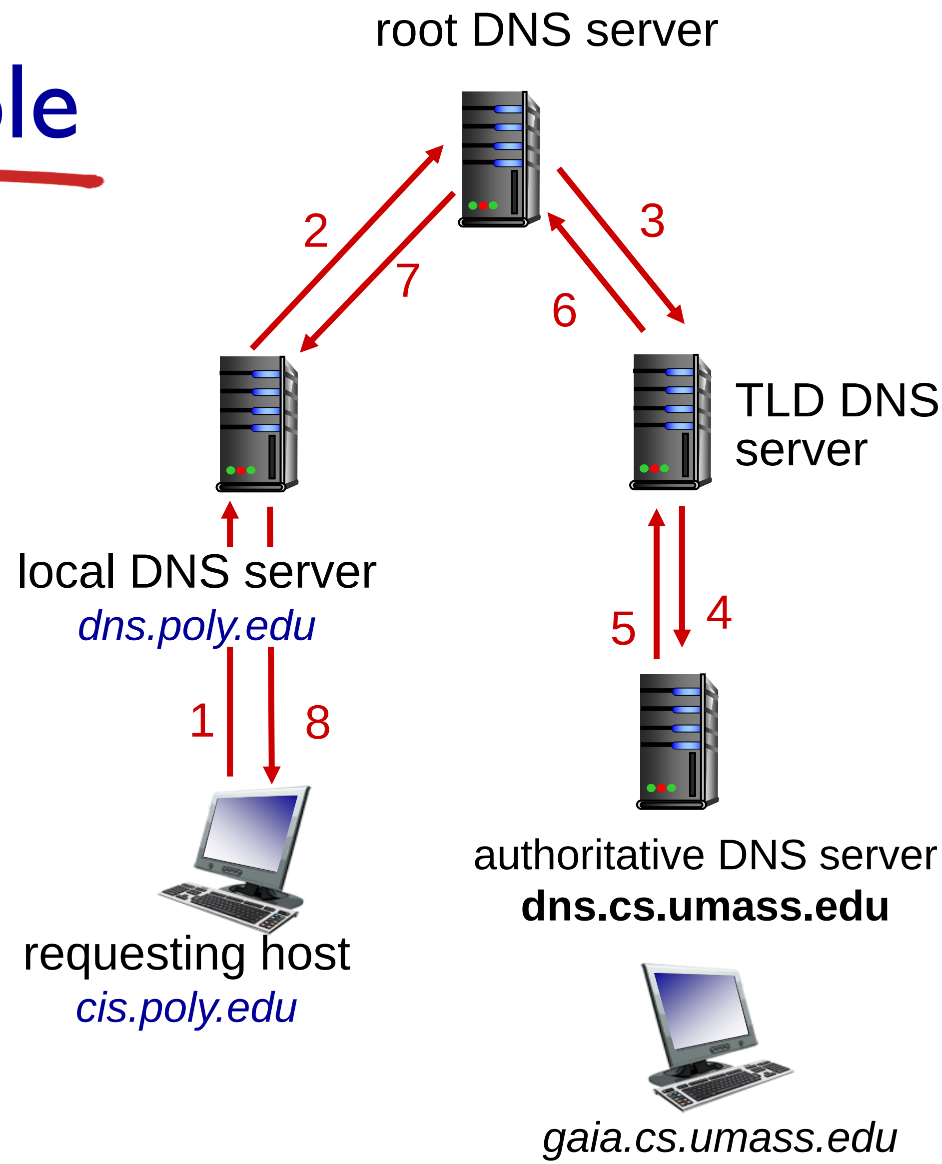
- contacted server replies with name of server to contact
- "I don't know this name, but ask this server"



DNS name resolution example

recursive query:

- puts burden of name resolution on contacted name server
- heavy load at upper levels of hierarchy



DNS: caching, updating records

- once (any) name server learns mapping, it *caches* mapping
  - cache entries timeout (disappear) after some time (TTL)
  - TLD servers typically cached in local name servers

- cached entries may be *out-of-date* (best effort name-to-address translation!)
  - if name host changes IP address, may not be known Internet-wide until all TTLs expire

Chapter 2: outline

2.1 principles of network applications

2.2 Web and HTTP

2.3 electronic mail

- SMTP, POP3, IMAP

2.4 DNS

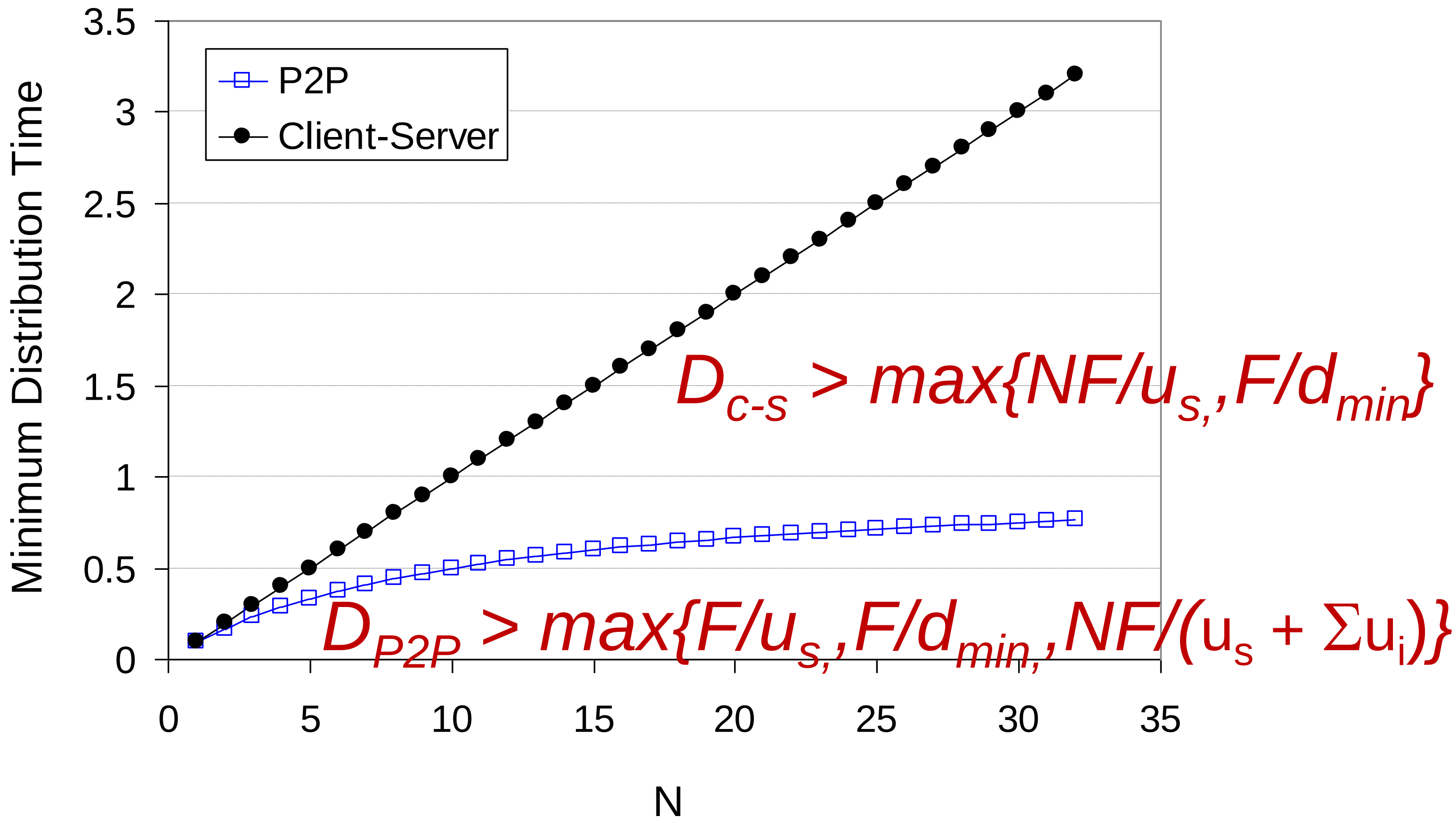
2.5 P2P applications

2.6 video streaming and content distribution networks

2.7 socket programming with UDP and TCP

Client-server vs. P2P: example

client upload rate = u, F/u = 1 hour, $u_s = 10u$, $d_{min} \geq u_s$

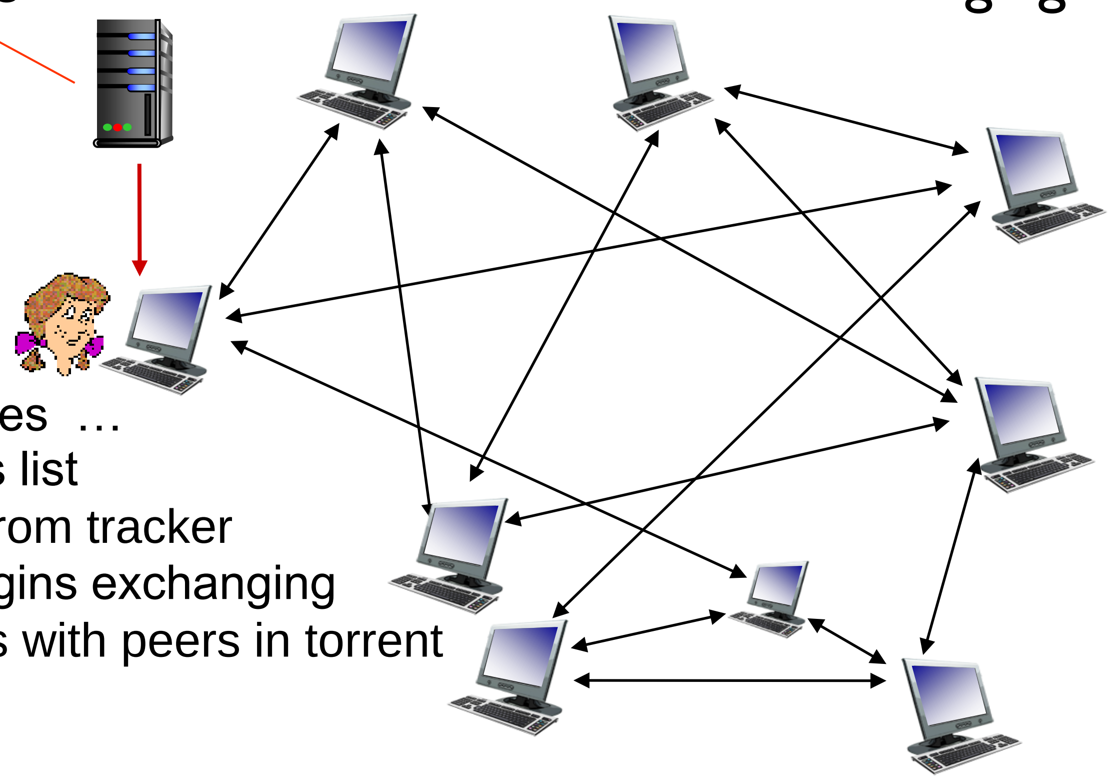


P2P file distribution: BitTorrent

- file divided into 256Kb chunks
- peers in torrent send/receive file chunks

tracker: tracks peers participating in torrent

torrent: group of peers exchanging chunks of a file



Alice arrives …
… obtains list
of peers from tracker
… and begins exchanging
file chunks with peers in torrent

BitTorrent: requesting, sending file chunks

requesting chunks:

- at any given time, different peers have different subsets of file chunks
- periodically, Alice asks each peer for list of chunks that they have
- Alice requests missing chunks from peers, rarest first

sending chunks: tit-for-tat

- Alice sends chunks to those four peers currently sending her chunks *at highest rate*
  - other peers are choked by Alice (do not receive chunks from her)
  - re-evaluate top 4 every10 secs
- every 30 secs: randomly select another peer, starts sending chunks
  - "optimistically unchoke" this peer
  - newly chosen peer may join top 4

Chapter 2: outline

2.1 principles of network applications

2.2 Web and HTTP

2.3 electronic mail

- SMTP, POP3, IMAP

2.4 DNS

2.5 P2P applications

2.6 video streaming and content distribution networks (CDNs)

2.7 socket programming with UDP and TCP

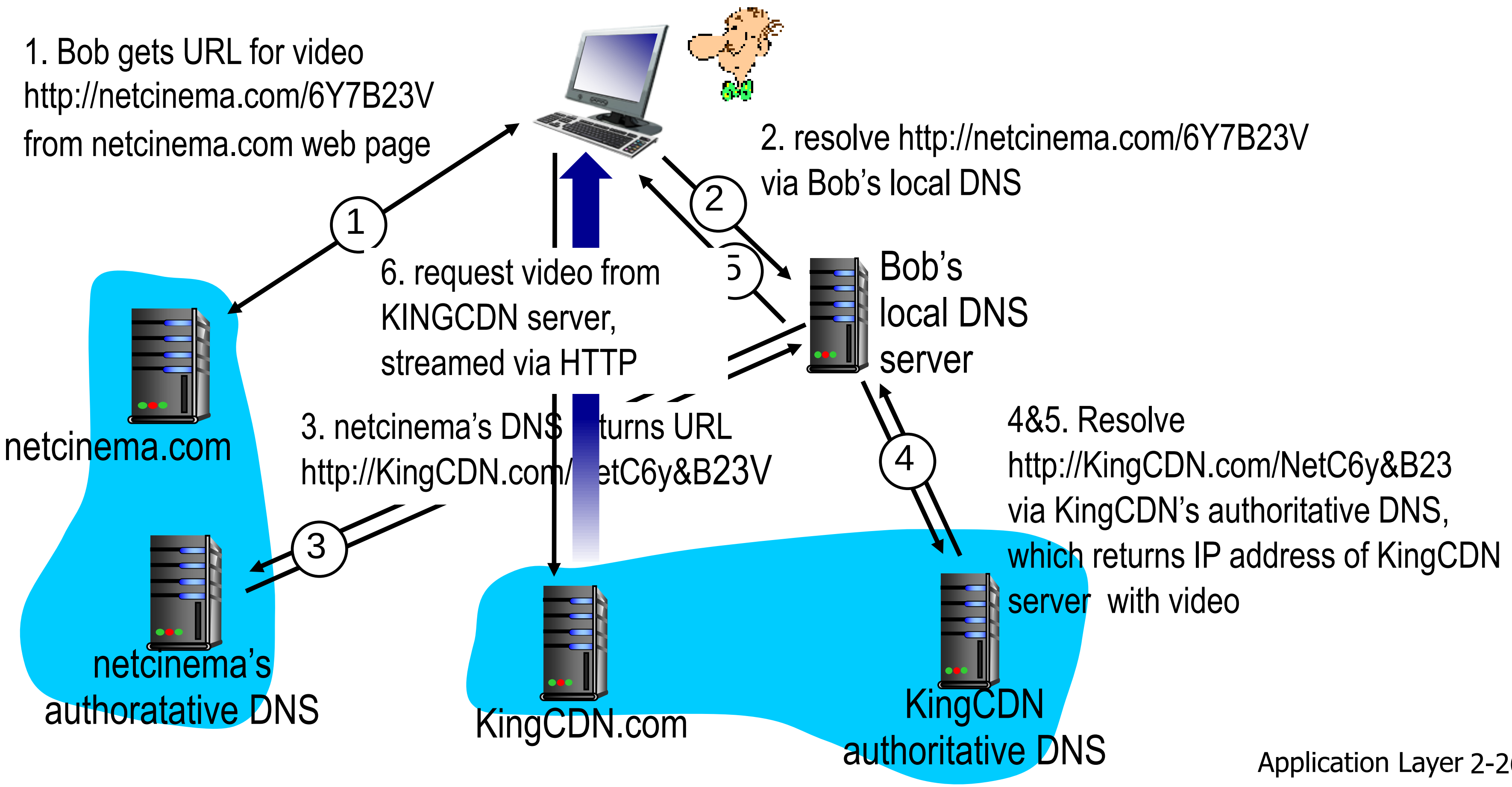
Streaming multimedia: DASH

- *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)

CDN content access: a closer look

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

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



Chapter 2: outline

2.1 principles of network applications

2.2 Web and HTTP

2.3 electronic mail
- SMTP, POP3, IMAP

2.4 DNS

2.5 P2P applications

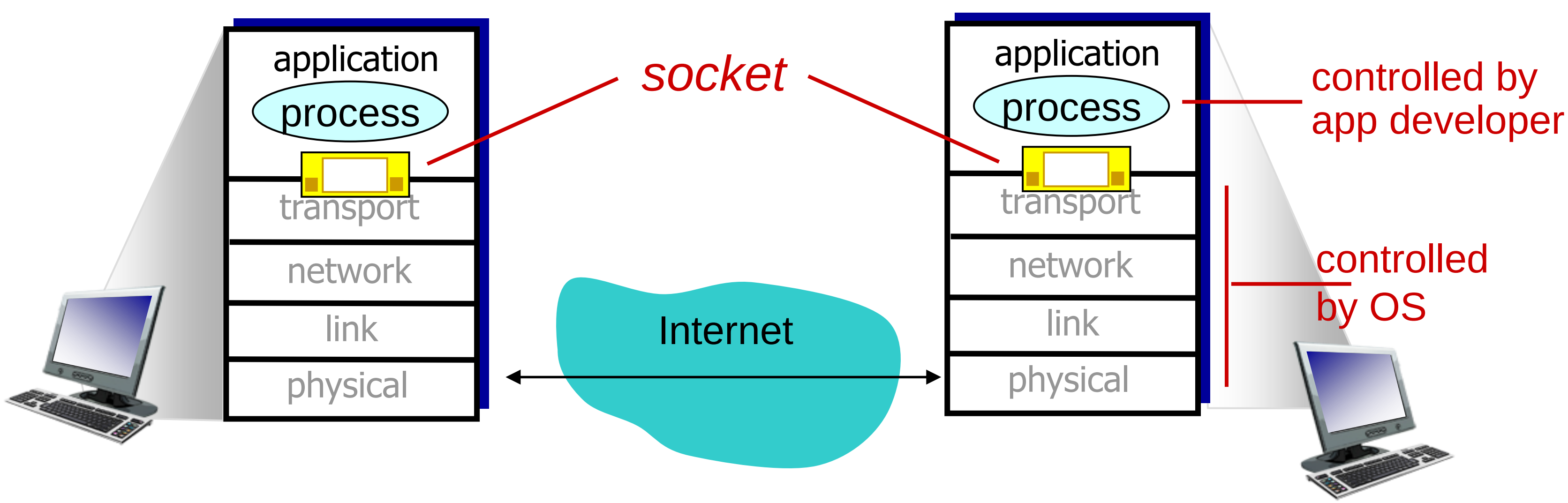
2.6 video streaming and content distribution networks

2.7 socket programming with UDP and TCP

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 datagrams
- *TCP:* reliable, stream-oriented bytes

Application Example:

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

server (running on serverIP)

create socket, port= x:
serverSocket =
socket(AF_INET,SOCK_DGRAM)

read datagram from
serverSocket

write reply to
serverSocket
specifying
client address,
port number

client

create socket:
clientSocket =
socket(AF_INET,SOCK_DGRAM)

Create datagram with server IP and
port=x; send datagram via
clientSocket

read datagram from
clientSocket

close
clientSocket

Example app: UDP client

Python UDPClient

include Python's socket library → `from socket import *`

```
from socket import *
serverName = 'hostname'
serverPort = 12000
clientSocket = socket(AF_INET,
                      SOCK_DGRAM)
message = raw_input('Input lowercase sentence:')
clientSocket.sendto(message,(serverName, serverPort))
modifiedMessage, serverAddress =
                      clientSocket.recvfrom(1024)
print modifiedMessage
clientSocket.close()
```

- include Python's socket library → `from socket import *`
- create UDP socket for server → `clientSocket = socket(AF_INET, SOCK_DGRAM)`
- get user keyboard input → `message = raw_input('Input lowercase sentence:')`
- Attach server name, port to message; send into socket → `clientSocket.sendto(message,(serverName, serverPort))`
- read reply characters from socket into string → `modifiedMessage, serverAddress = clientSocket.recvfrom(1024)`
- print out received string and close socket → `print modifiedMessage`, `clientSocket.close()`

Example app: UDP server

Python UDPServer

create UDP socket → `serverSocket = socket(AF_INET, SOCK_DGRAM)`

bind socket to local port number 12000 → `serverSocket.bind(('', serverPort))`

loop forever → `while 1:`

Read from UDP socket into message, getting client's address (client IP and port) → `message, clientAddress = serverSocket.recvfrom(1024)`

send upper case string back to this client → `serverSocket.sendto(modifiedMessage, clientAddress)`

```
from socket import *
serverPort = 12000
serverSocket = socket(AF_INET, SOCK_DGRAM)
serverSocket.bind(('', serverPort))
print 'The server is ready to receive'
while 1:
    message, clientAddress = serverSocket.recvfrom(1024)
    modifiedMessage = message.upper()
    serverSocket.sendto(modifiedMessage, 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:

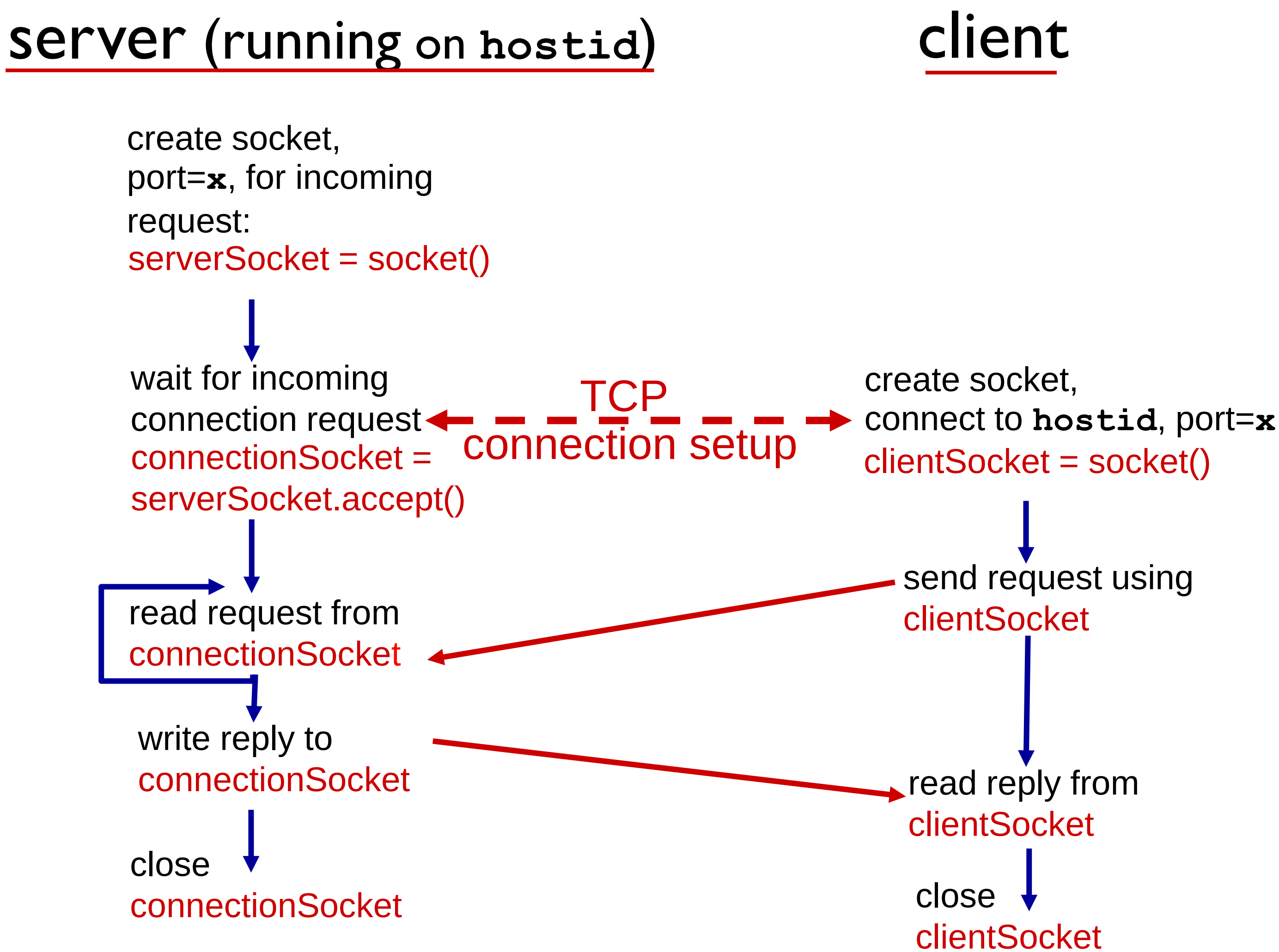
- creating TCP socket, specifying IP address, port number of server process
- *when client creates socket:* client TCP establishes connection to server TCP

- 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 IPs/ports 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)
modifiedSentence = clientSocket.recv(1024)
print 'From Server:', modifiedSentence
clientSocket.close()
```

Example app: TCP server

Python TCPServer

create TCP welcoming socket

server begins listening for incoming TCP requests

loop forever

server waits on accept() for incoming requests, new socket created on return

read bytes from socket (but not address as in UDP)

close connection to this client (but *not* welcoming socket)

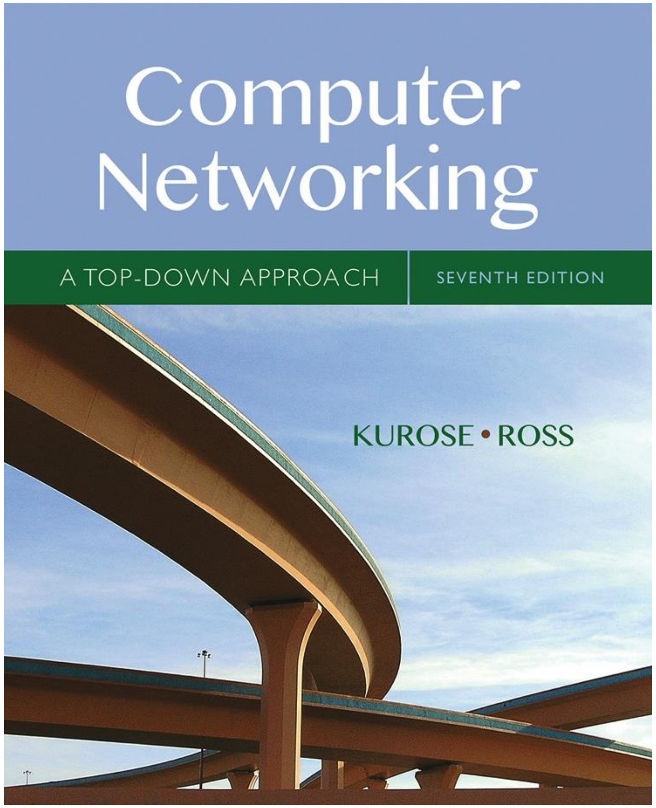
```
 from socket import *
serverPort = 12000
serverSocket = socket(AF_INET,SOCK_STREAM)
serverSocket.bind(('',serverPort))
serverSocket.listen(1)
print 'The server is ready to receive'
while 1:
    connectionSocket, addr = serverSocket.accept()

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

Chapter 2: summary

our study of network apps 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, POP, IMAP
  - DNS
  - P2P: BitTorrent
  - Video streaming
- socket programming: TCP, UDP sockets

Chapter 3
Transport Layer



Computer Networking: A Top Down Approach

7th edition
Jim Kurose, Keith Ross
Pearson/Addison Wesley
April 2016

Slides adopted from original ones provided by the textbook authors.

Chapter 3: 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

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

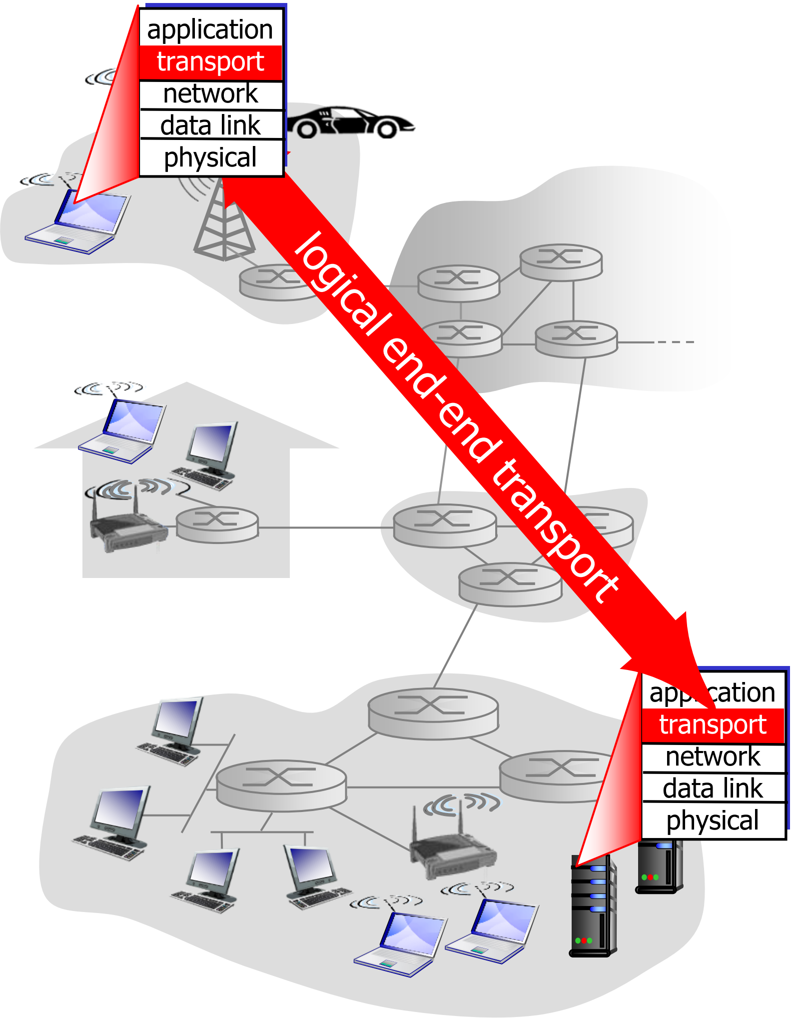
3.5 connection-oriented transport: TCP
- segment structure
- reliable data transfer
- flow control
- connection management

3.6 principles of congestion control

3.7 TCP congestion control

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 vs. network layer

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

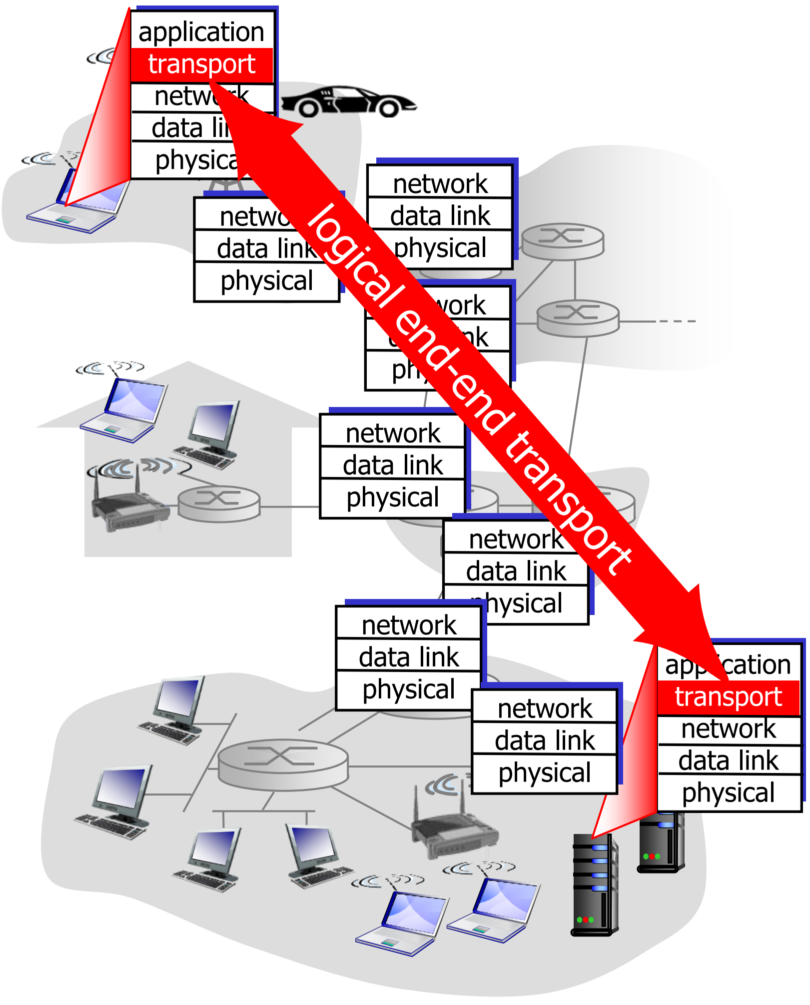
household analogy:

12 kids in 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

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



Example

- Is it possible for an application to enjoy reliable data transfer even when the application runs over UDP? If so, how?

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

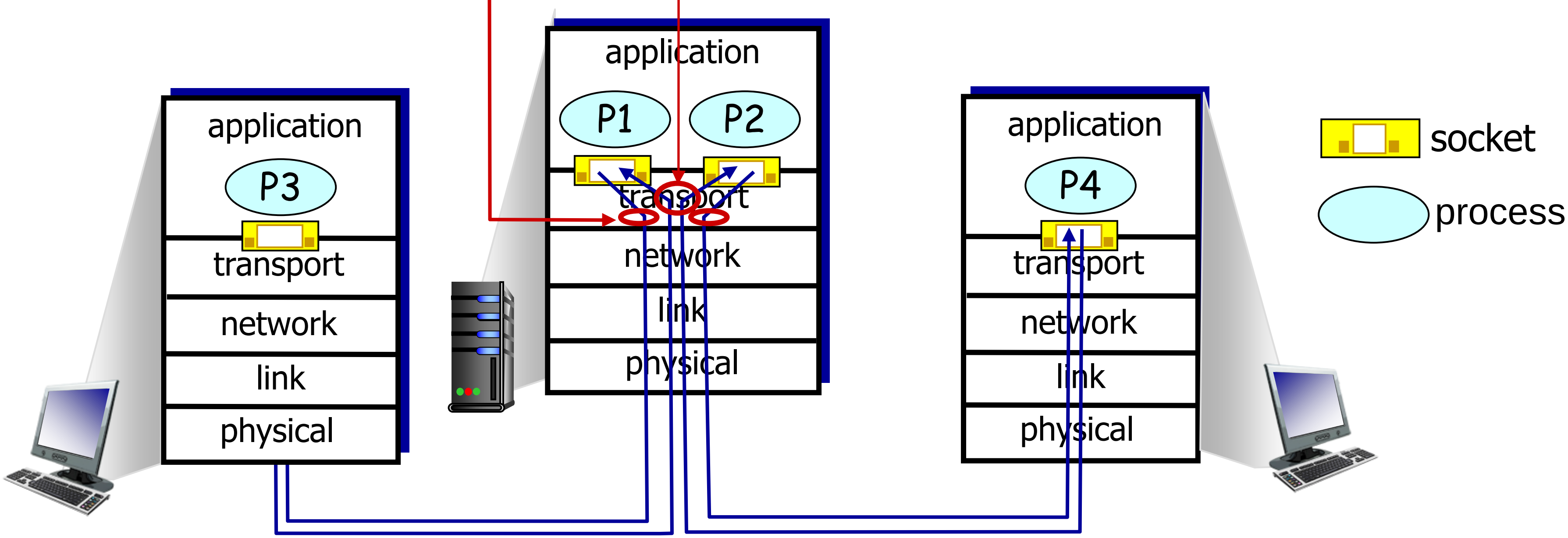
3.6 principles of congestion control

3.7 TCP congestion control

Multiplexing/demultiplexing

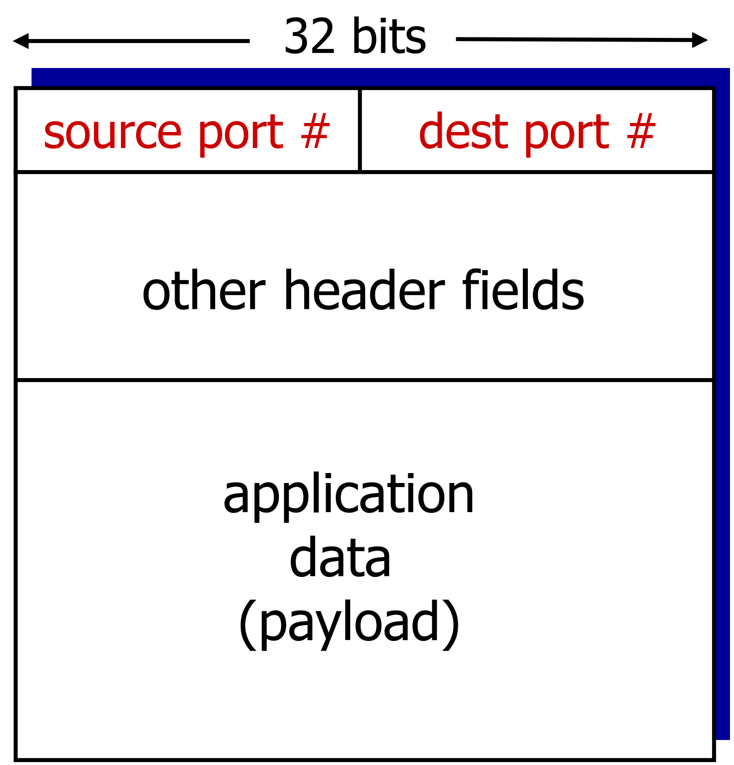
multiplexing at sender:
handle data from multiple sockets, add transport header (later used for demultiplexing)

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 numbers
- host uses *IP addresses & port numbers* to direct segment to appropriate socket



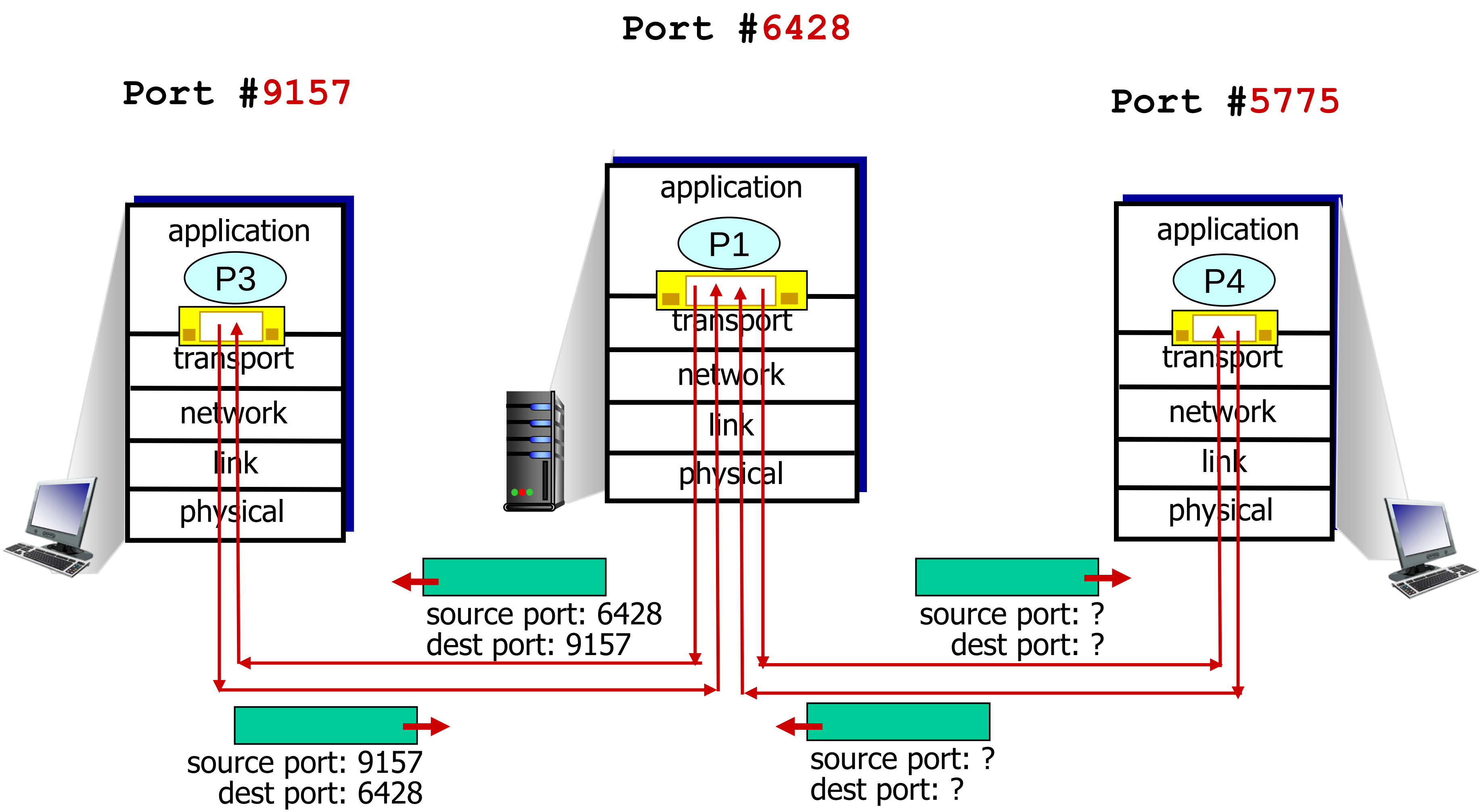
TCP/UDP segment format

Connectionless demultiplexing

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

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

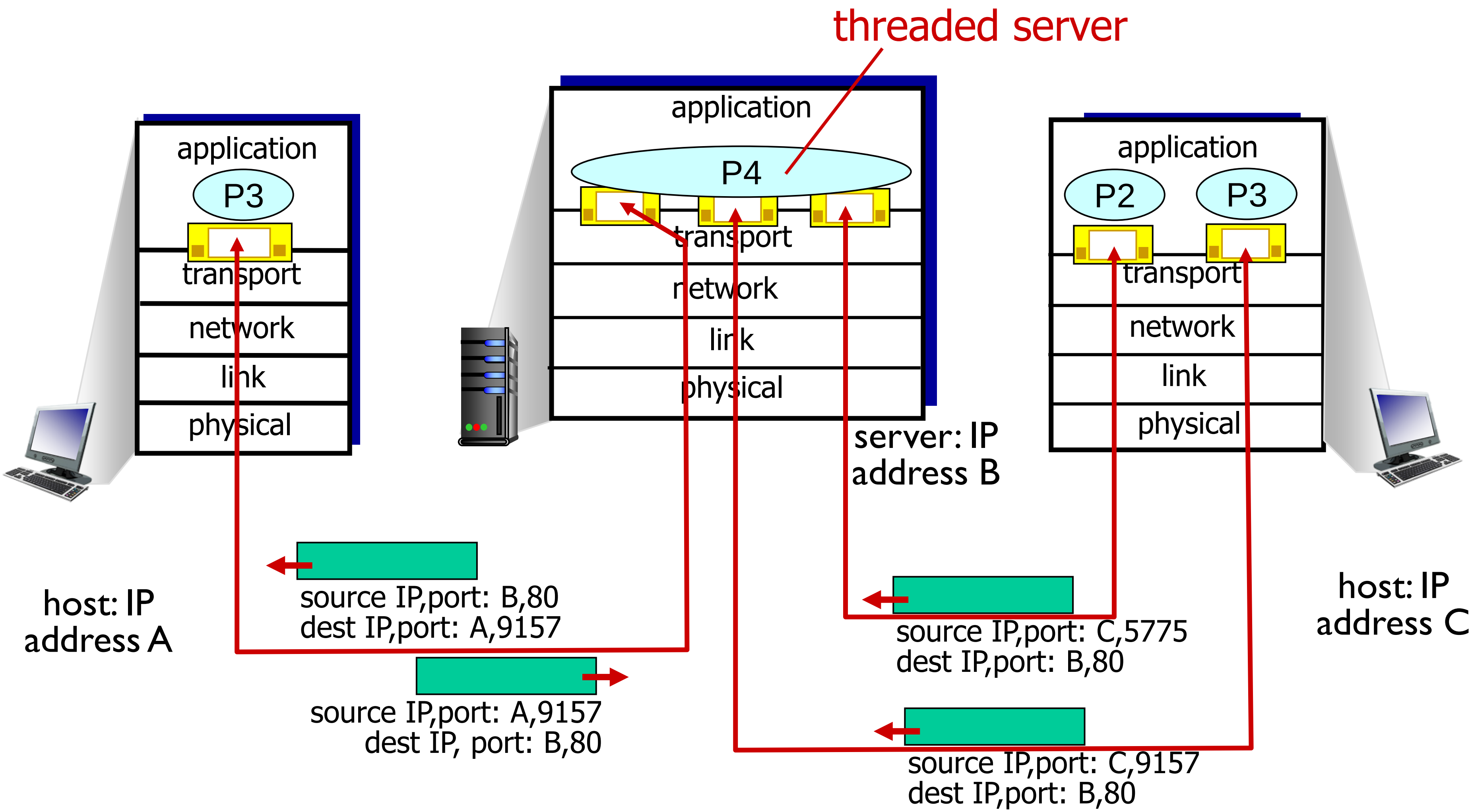
Connectionless demux: example



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

Connection-oriented demux: example



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

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

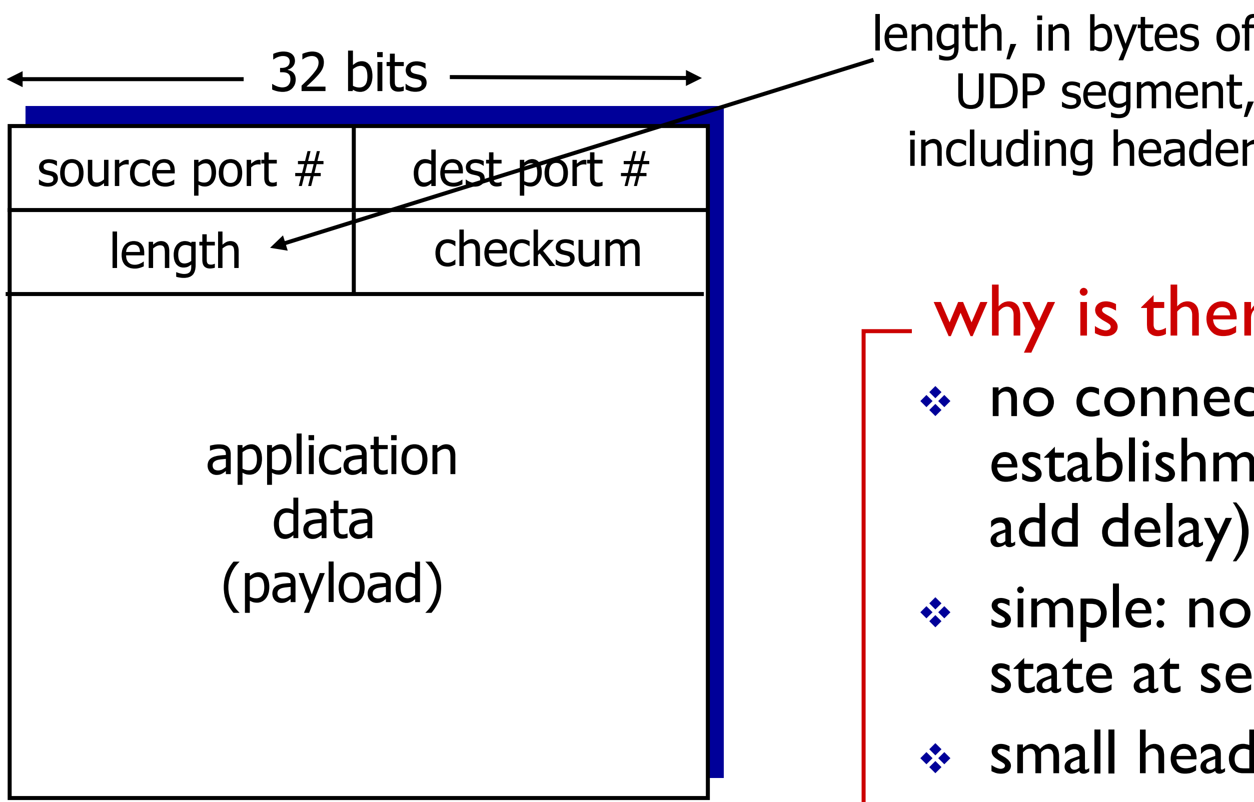
3.7 TCP congestion control

UDP: User Datagram Protocol [RFC 768]

- "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

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

UDP: segment header



UDP segment format

why is there a 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

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?* More later ….

Internet checksum: 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

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

Example

UDP and TCP use 1s complement for their checksums. Suppose you have the following three 8-bit bytes: 01010011, 01010100, 01110100.

a. What is the 1s complement of the sum of these 8-bit bytes?

b. With the 1s complement scheme, how does the receiver detect errors?

c. Is it possible that a 1-bit error will go undetected? How about a 2-bit error?

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

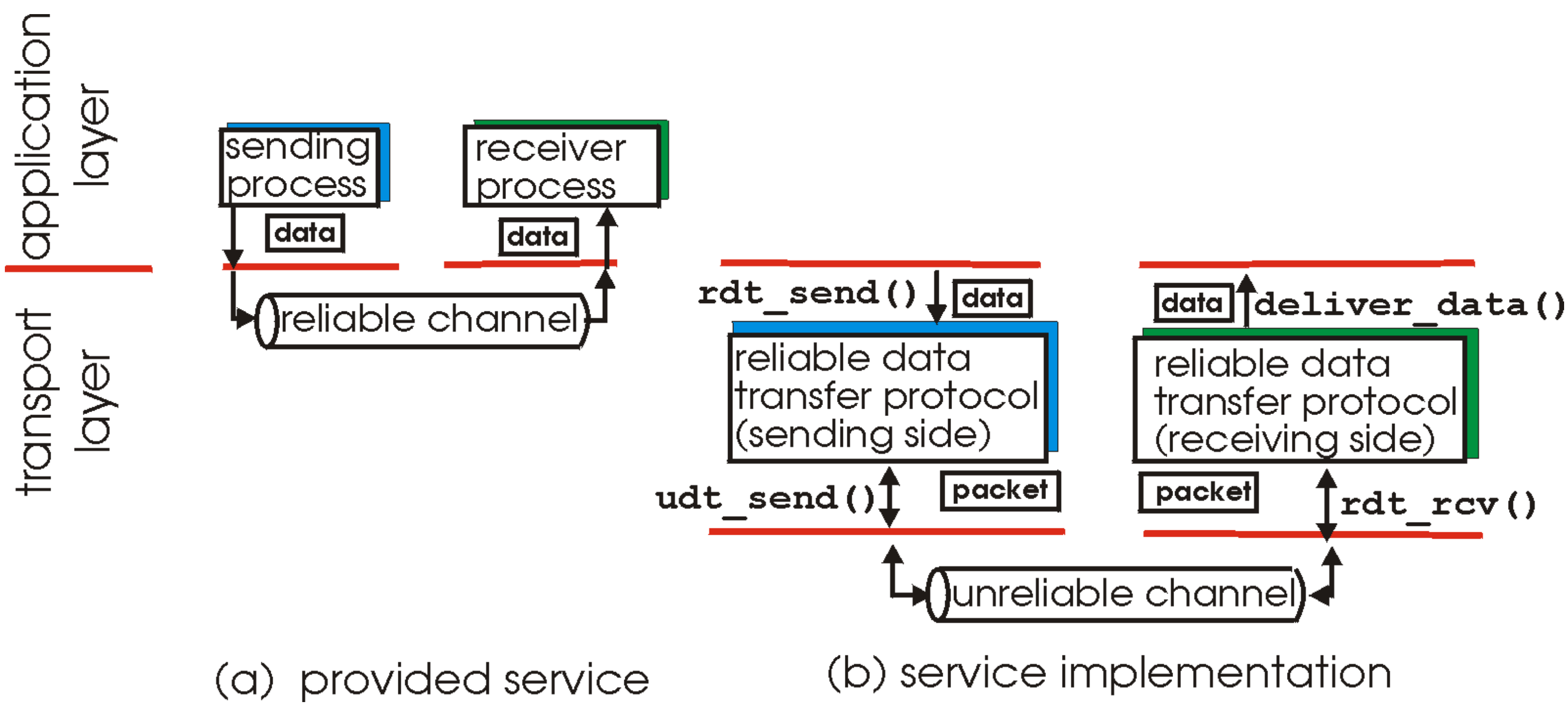
3.5 connection-oriented transport: TCP
- segment structure
- reliable data transfer
- flow control
- connection management

3.6 principles of congestion control

3.7 TCP congestion control

Principles of reliable data transfer

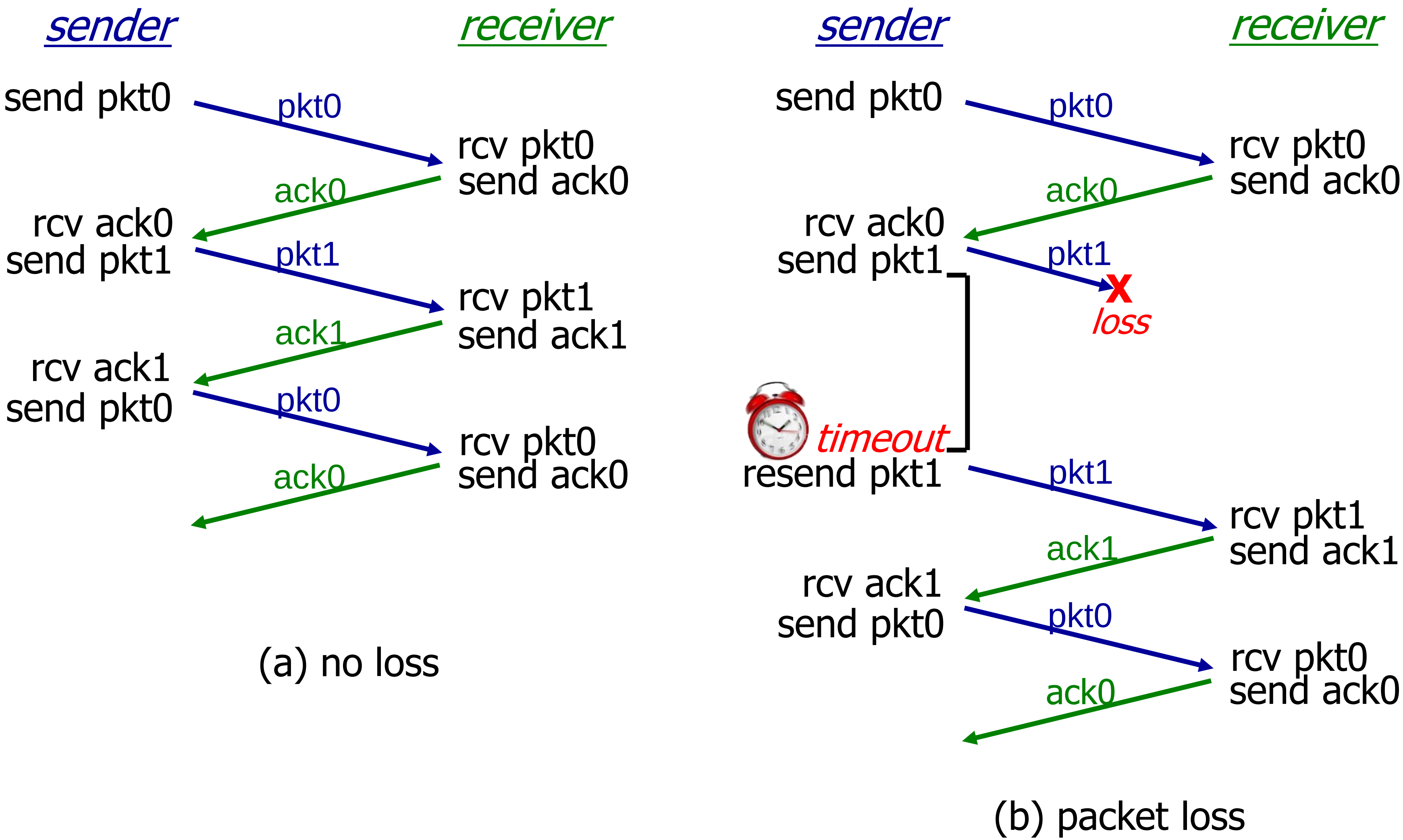
- Challenge: TCP requires reliable data transfer, but the underlying protocol IP is not reliable.



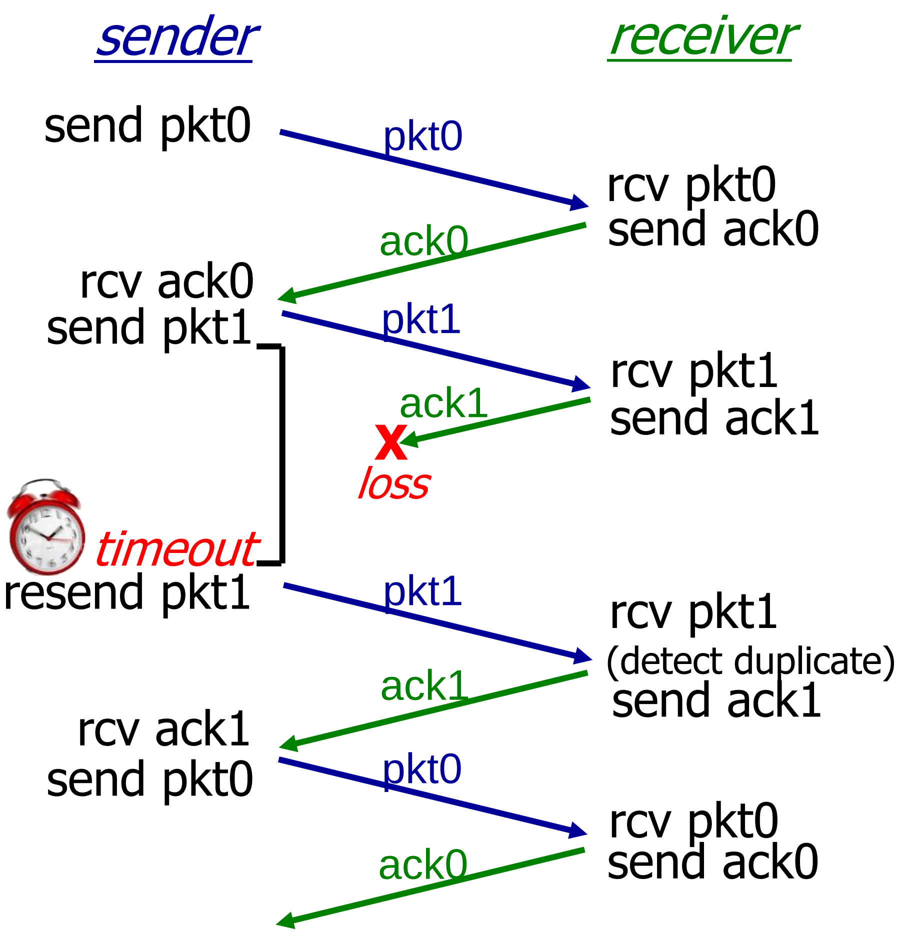
Possible errors

- Transmission errors in data:
  - Checksum to detect
  - Send ACK or NAK (negative ack)
  - Retransmit
- Transmission errors in ACK/NAKs
  - Retransmit
  - Sequence # to differentiate new and retransmission
  - Remove NAKs via ACK retransmission
- Packet loss
  - Timer

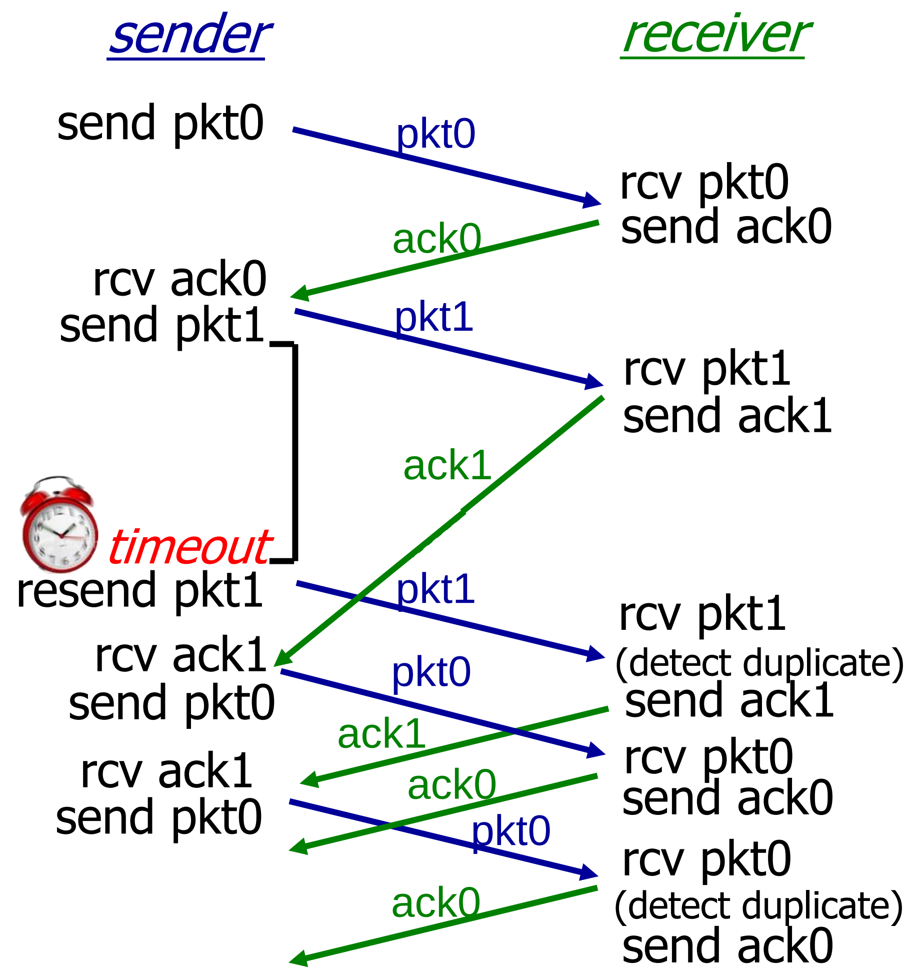
rdt in action



rdt3.0 in action



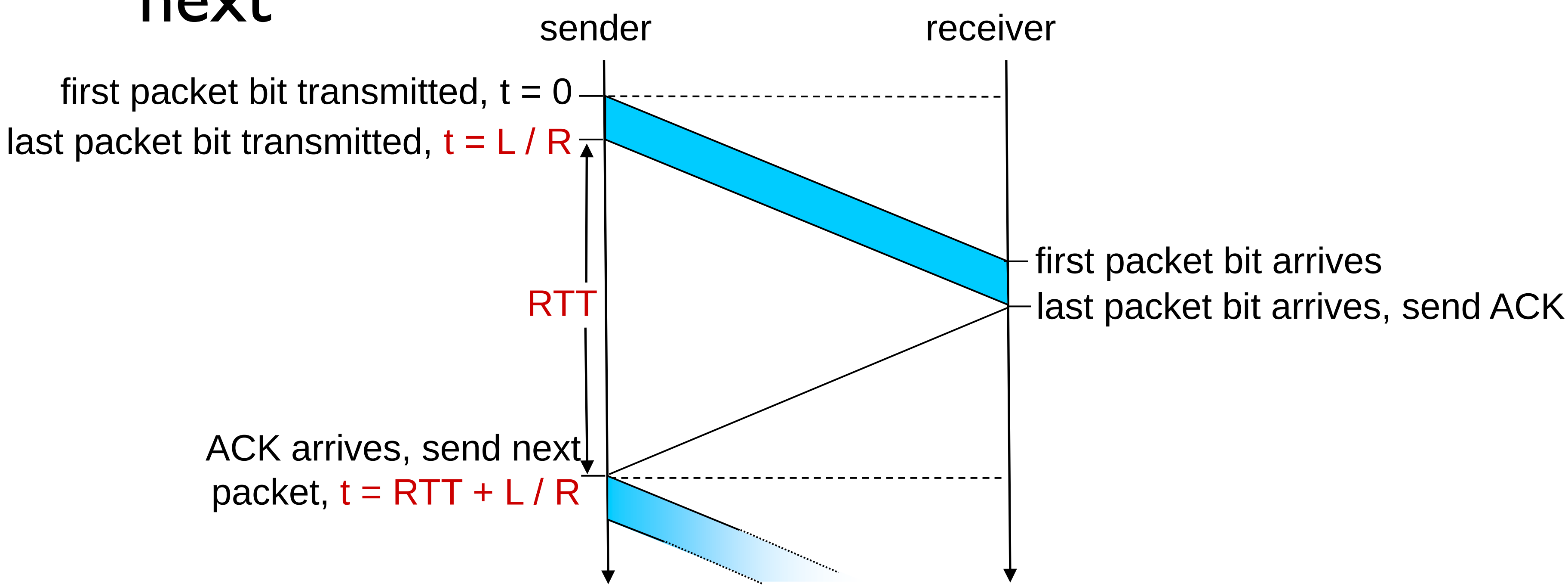
(c) ACK loss



(d) premature timeout/ delayed ACK

stop-and-wait operation

- Send packet, stop and wait for ack, send next

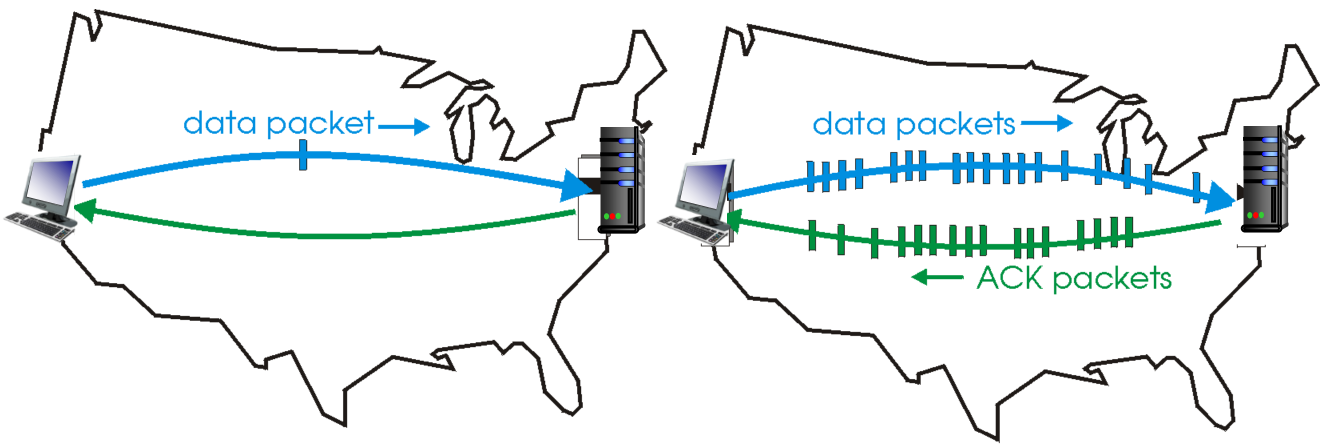


$$U_{sender} = \frac{L / R}{RTT + L / R} = \frac{.008}{30.008} = 0.00027$$

Pipelined protocols

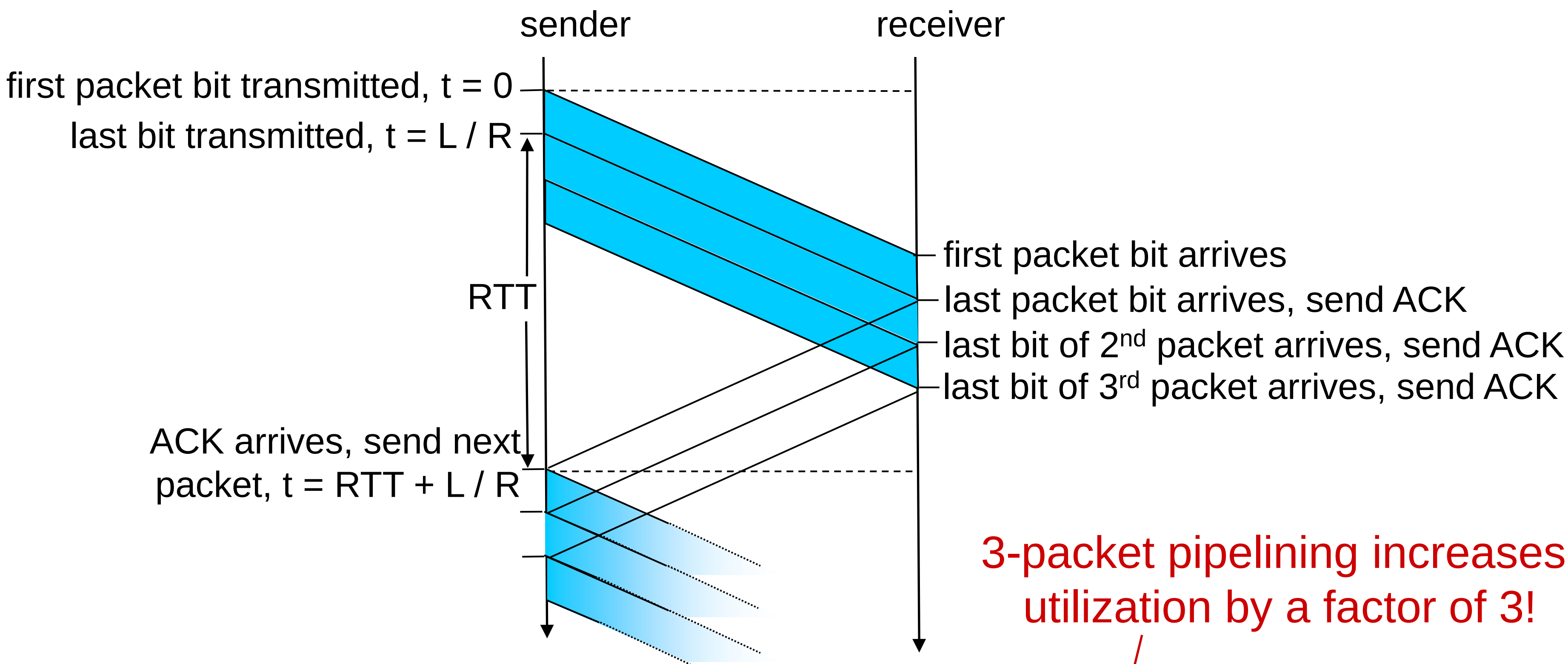
pipelining: sender allows multiple, "in-flight", yet-to-be-acknowledged pkts

- range of sequence numbers must be increased
- buffering at sender and/or receiver



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

Pipelining: increased utilization



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

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

Example

- Consider the cross-country example with bandwidth R=1 Gbps and round trip time RTT=30 s. How big would the windows size have to be for the channel utilization to be greater than 95 percent? Suppose that the size of a packet is 1,500 bytes, including both header fields and data.