

Chapter 3

Transport Layer

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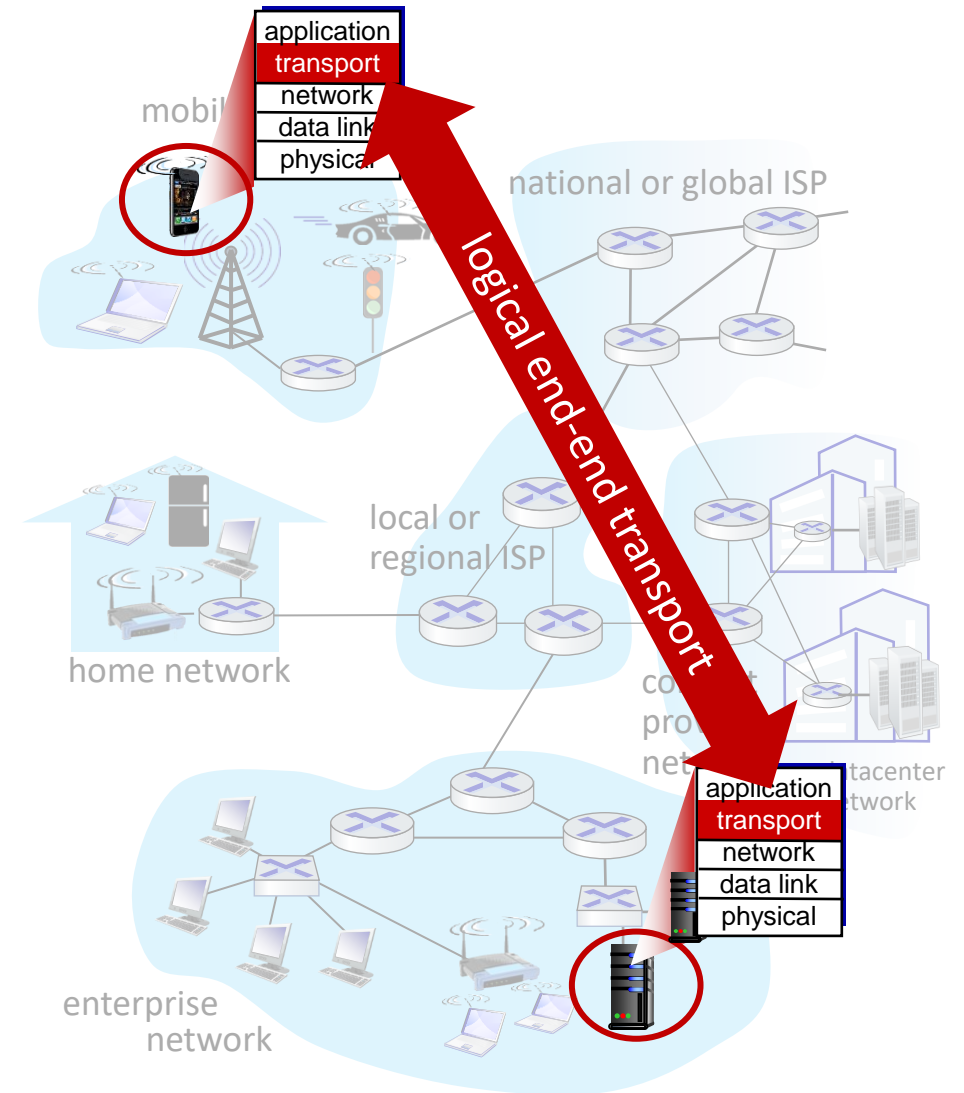
Computer Networking: A Top-Down Approach

8th edition

Jim Kurose, Keith Ross
Pearson, 2020

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:

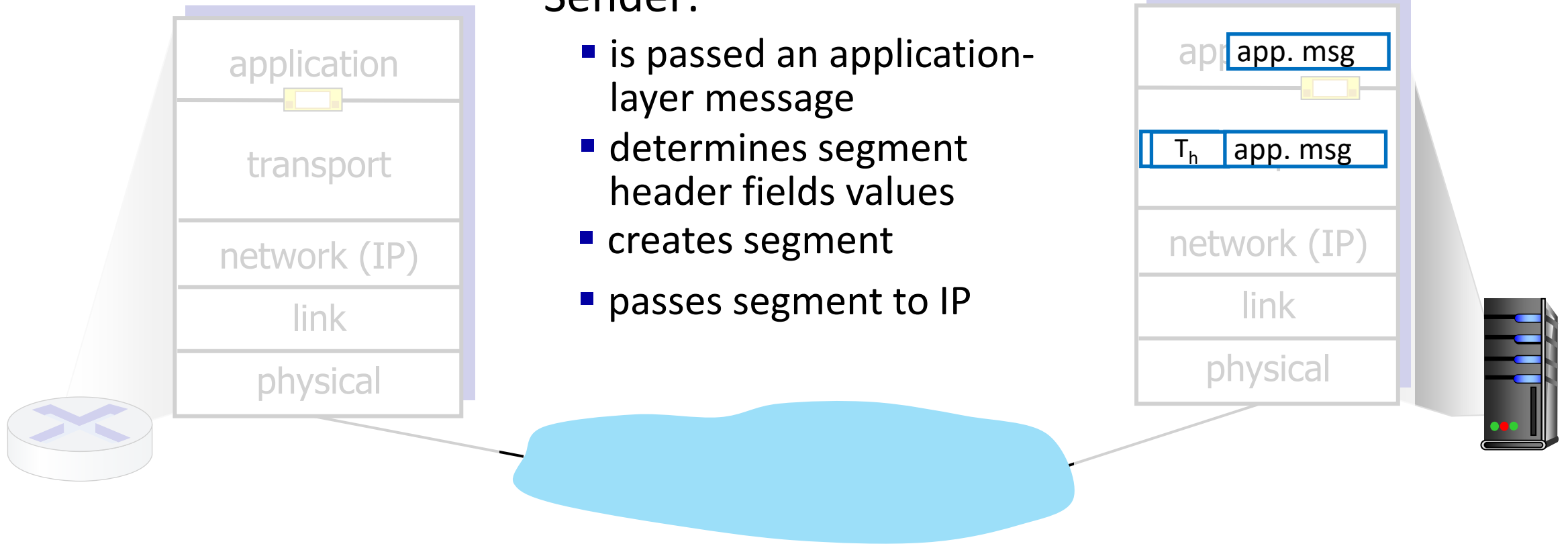
12 kids in Ann's house sending letters to 12 kids in Bill's house:

- hosts = houses
- processes = kids
- app messages = letters in envelopes

Transport Layer Actions

Sender:

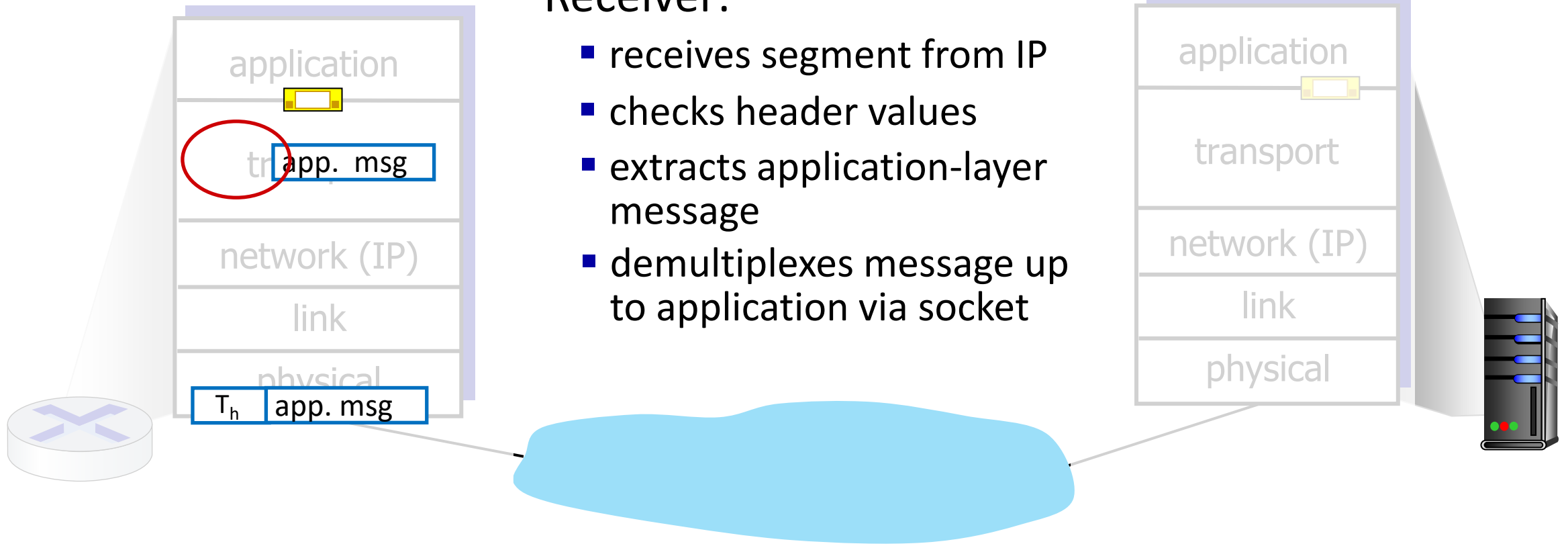
- is passed an application-layer message
- determines segment header fields values
- creates segment
- passes segment to IP



Transport Layer Actions

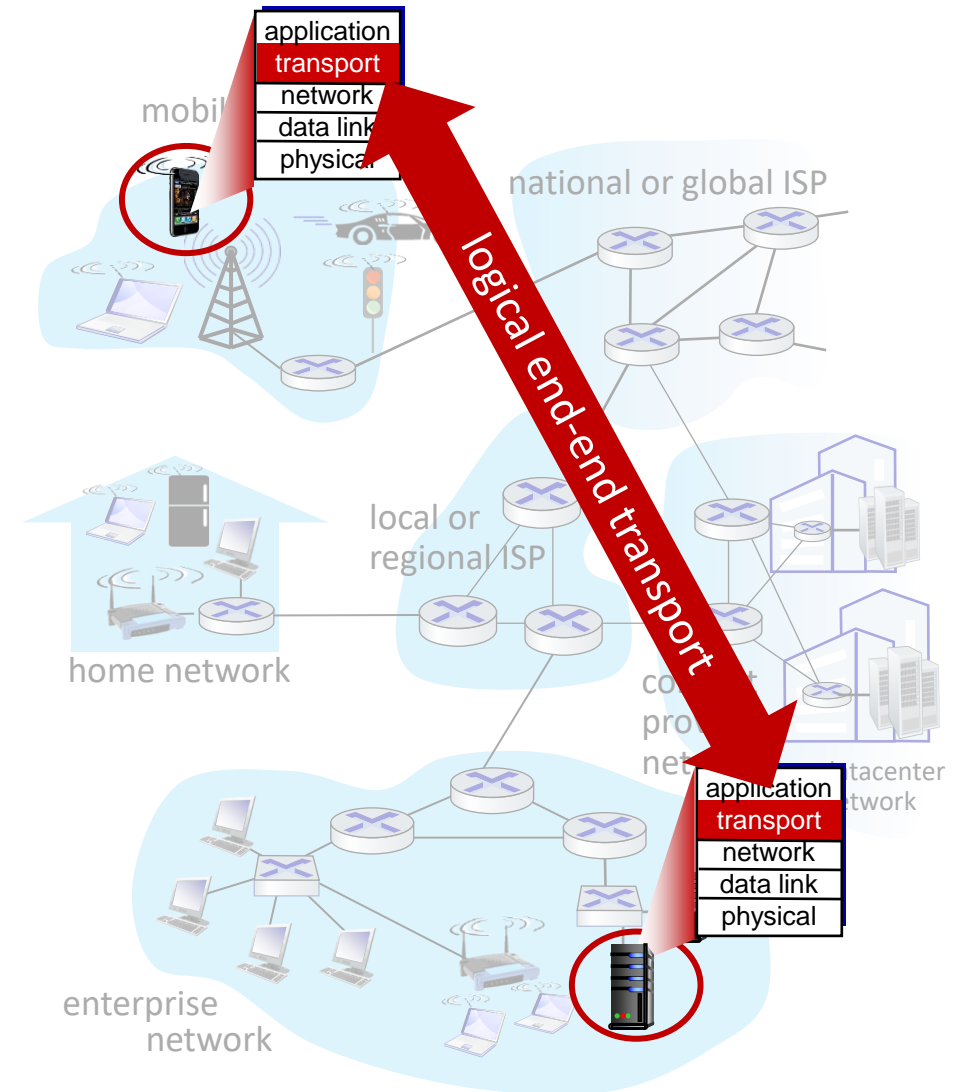
Receiver:

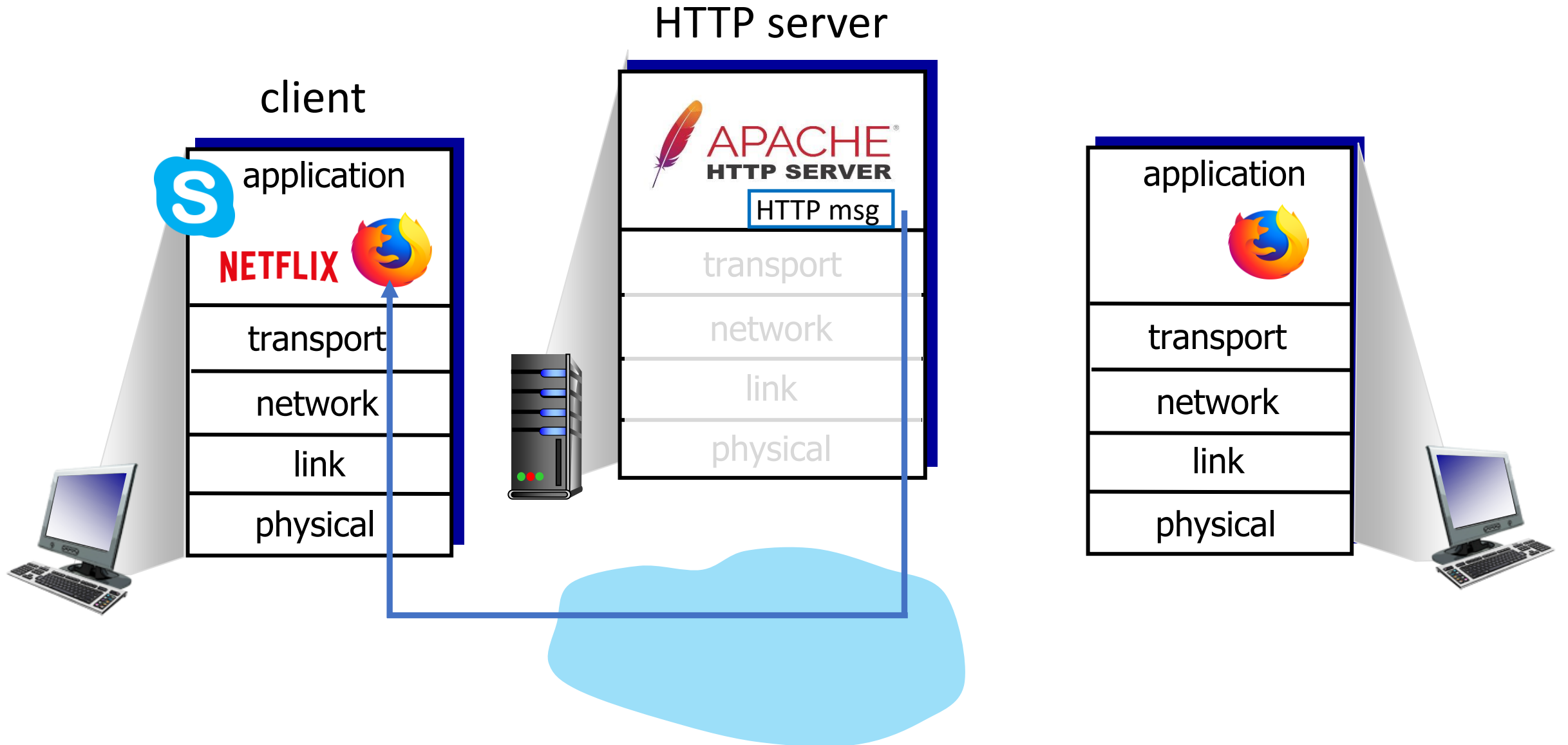
- receives segment from IP
- checks header values
- extracts application-layer message
- demultiplexes message up to application via socket

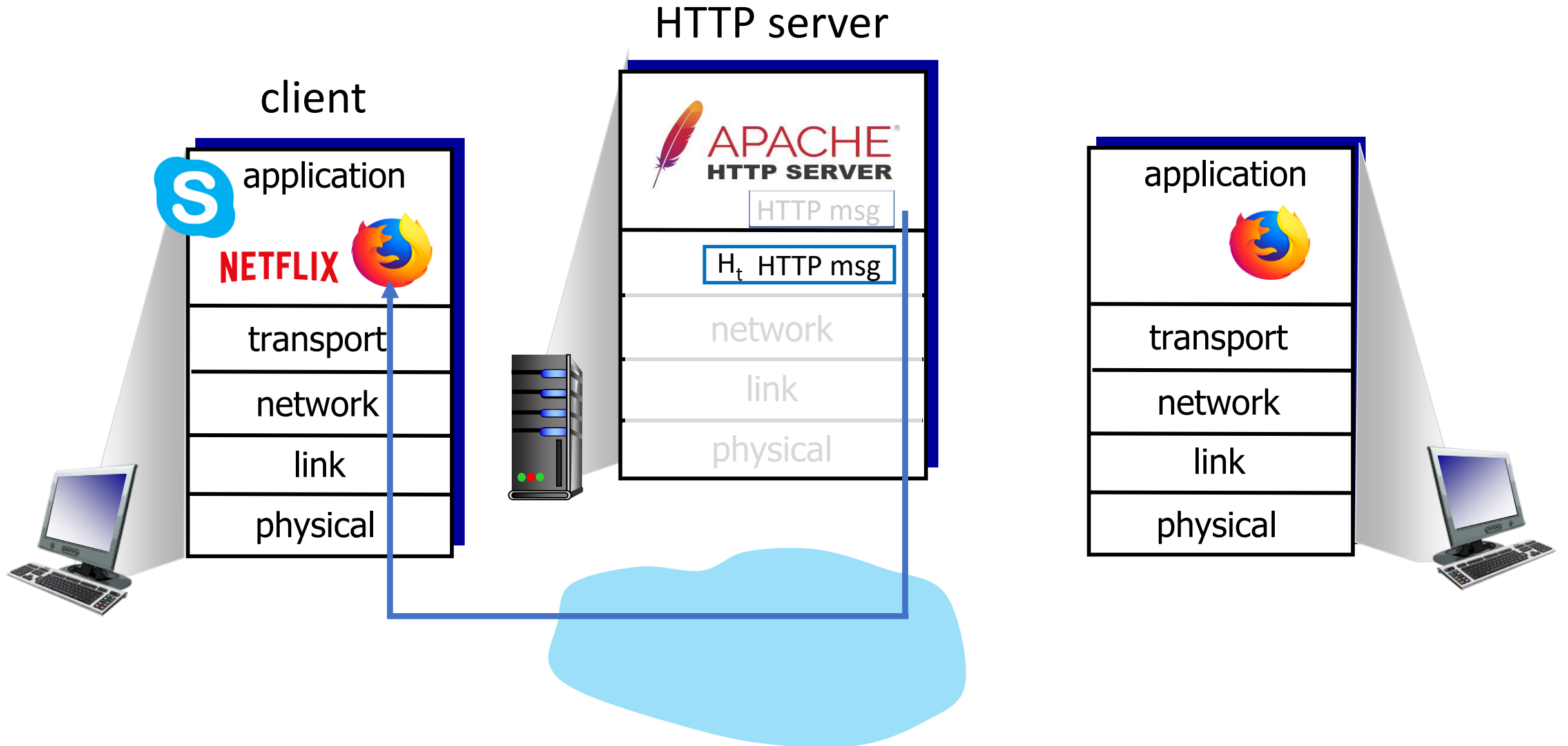


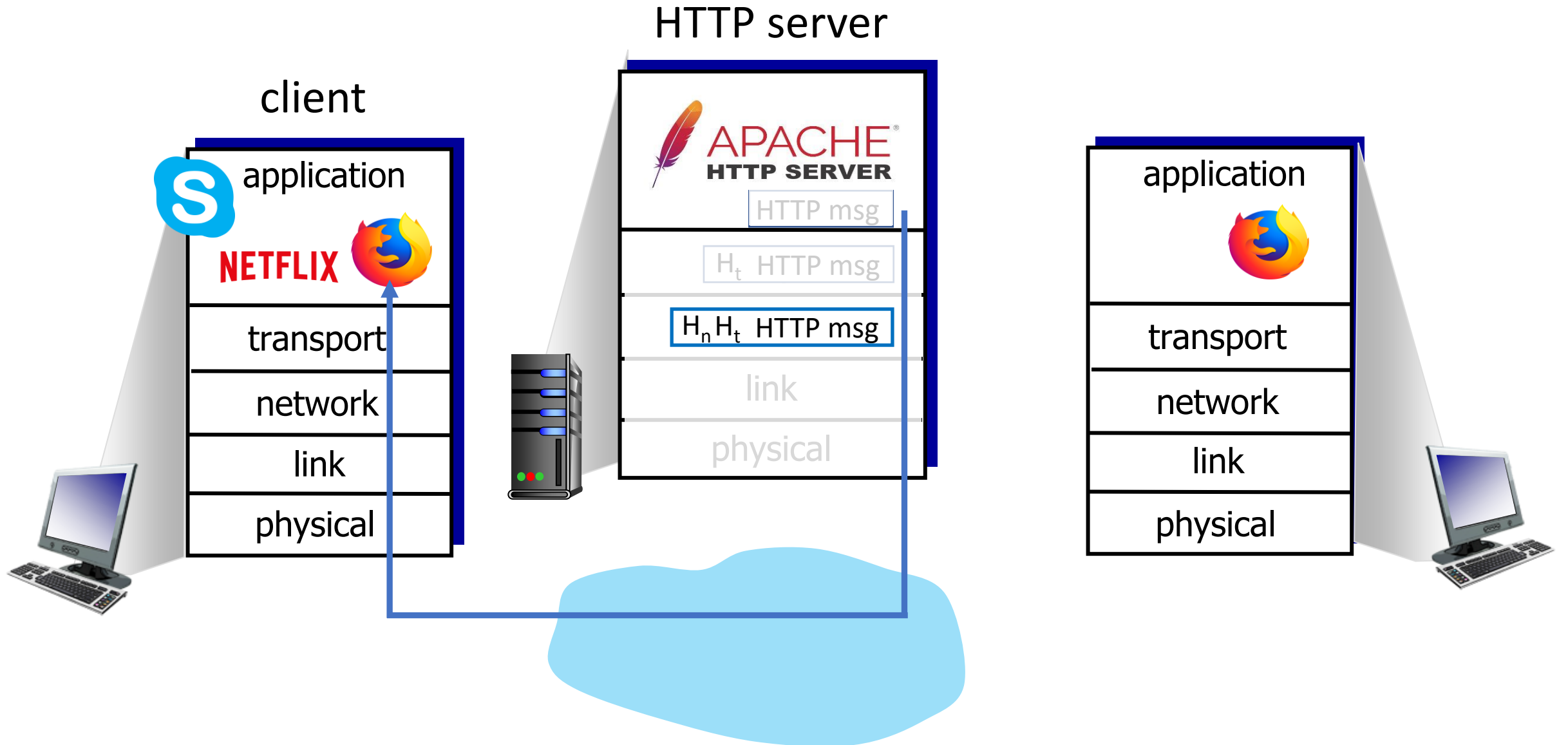
Two principal Internet transport protocols

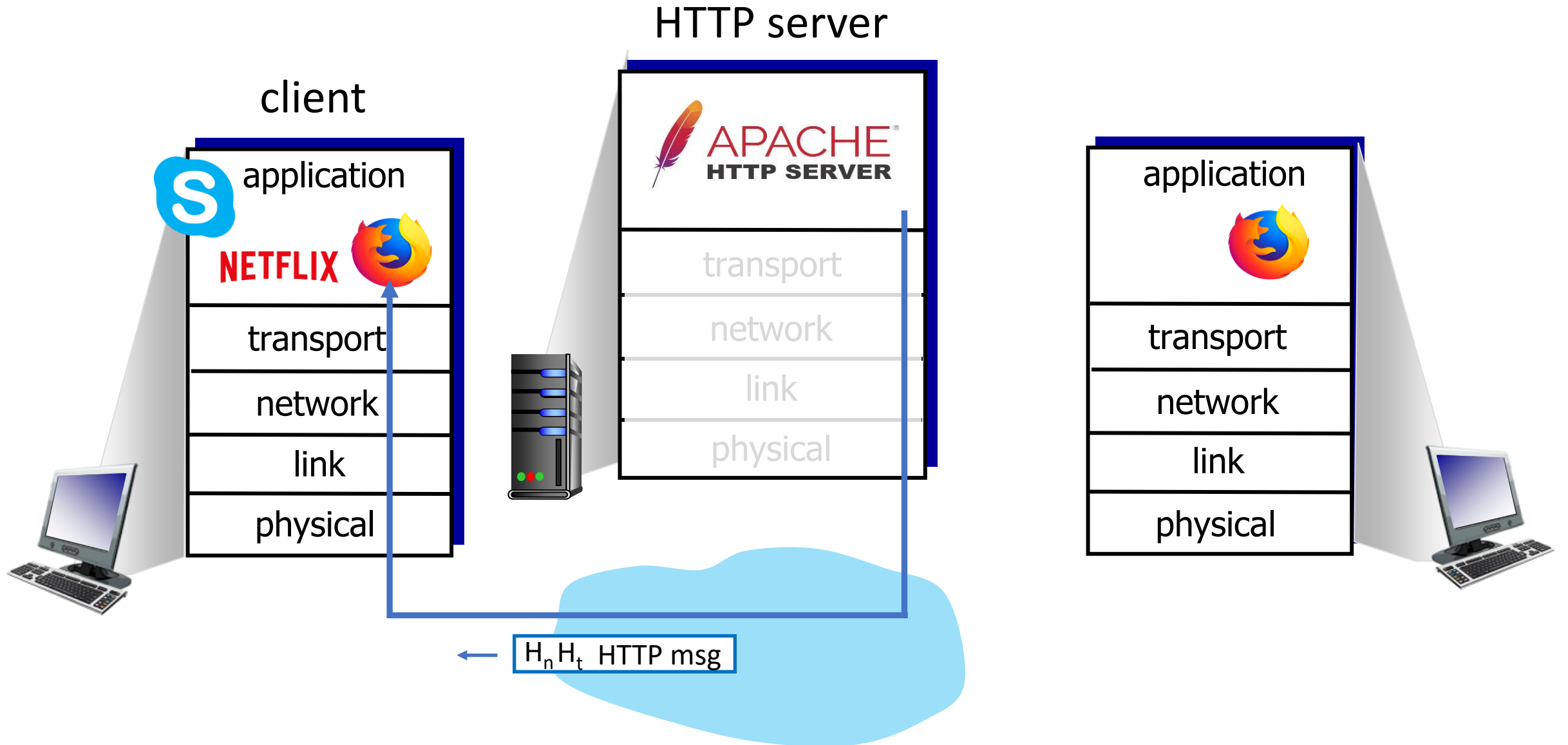
- **TCP:** Transmission Control Protocol
 - reliable, in-order delivery
 - congestion control
 - flow control
 - connection setup
- **UDP:** User Datagram Protocol
 - unreliable, unordered delivery
 - no-frills extension of “best-effort” IP
- services not available:
 - delay guarantees
 - bandwidth guarantees

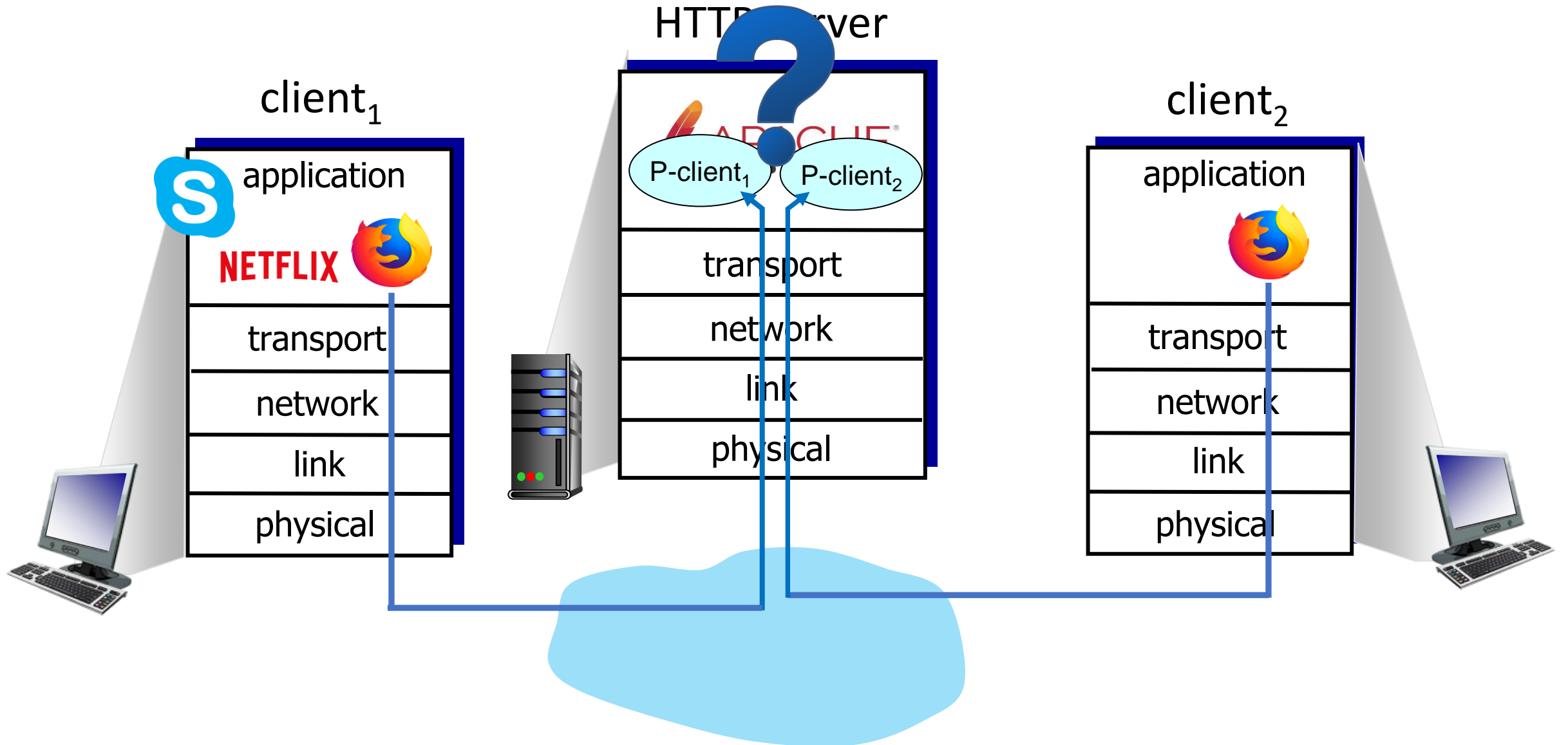












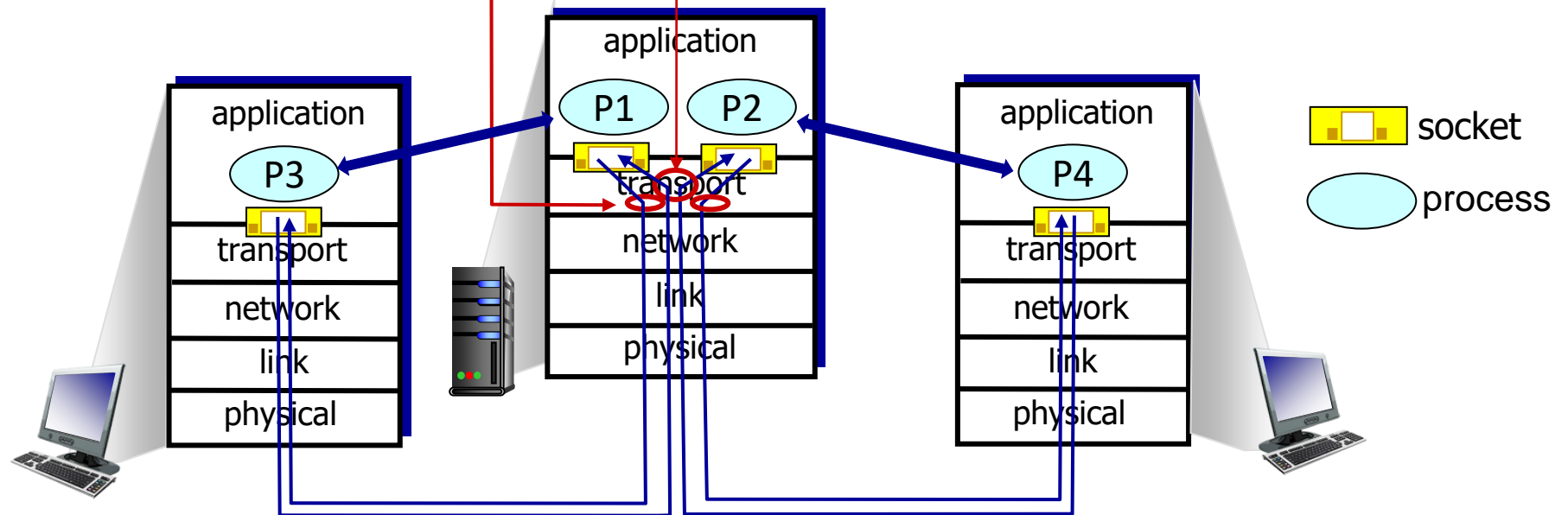
Multiplexing/demultiplexing

multiplexing at sender:

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

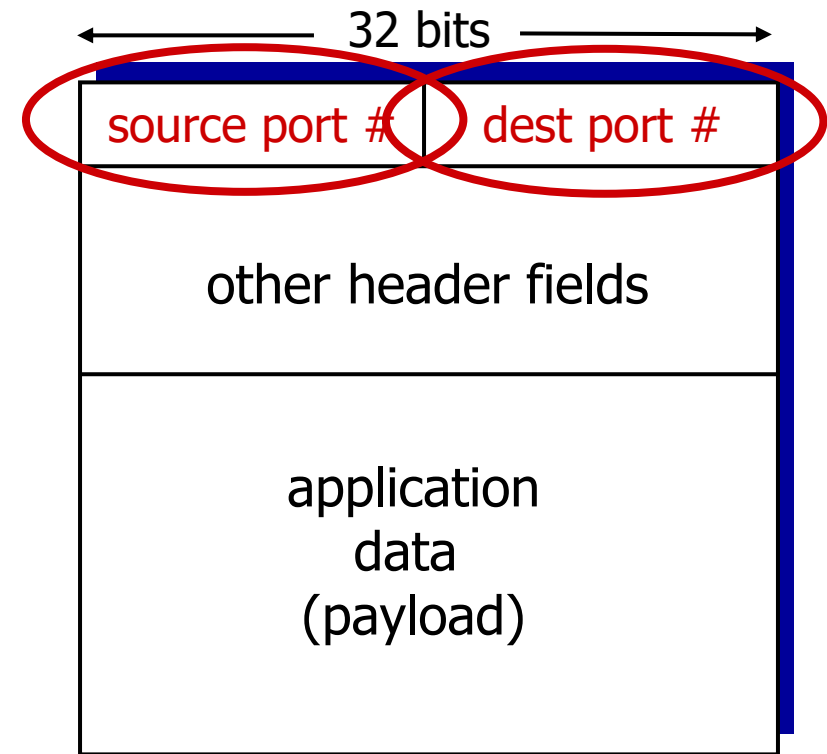
demultiplexing at receiver:

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 #



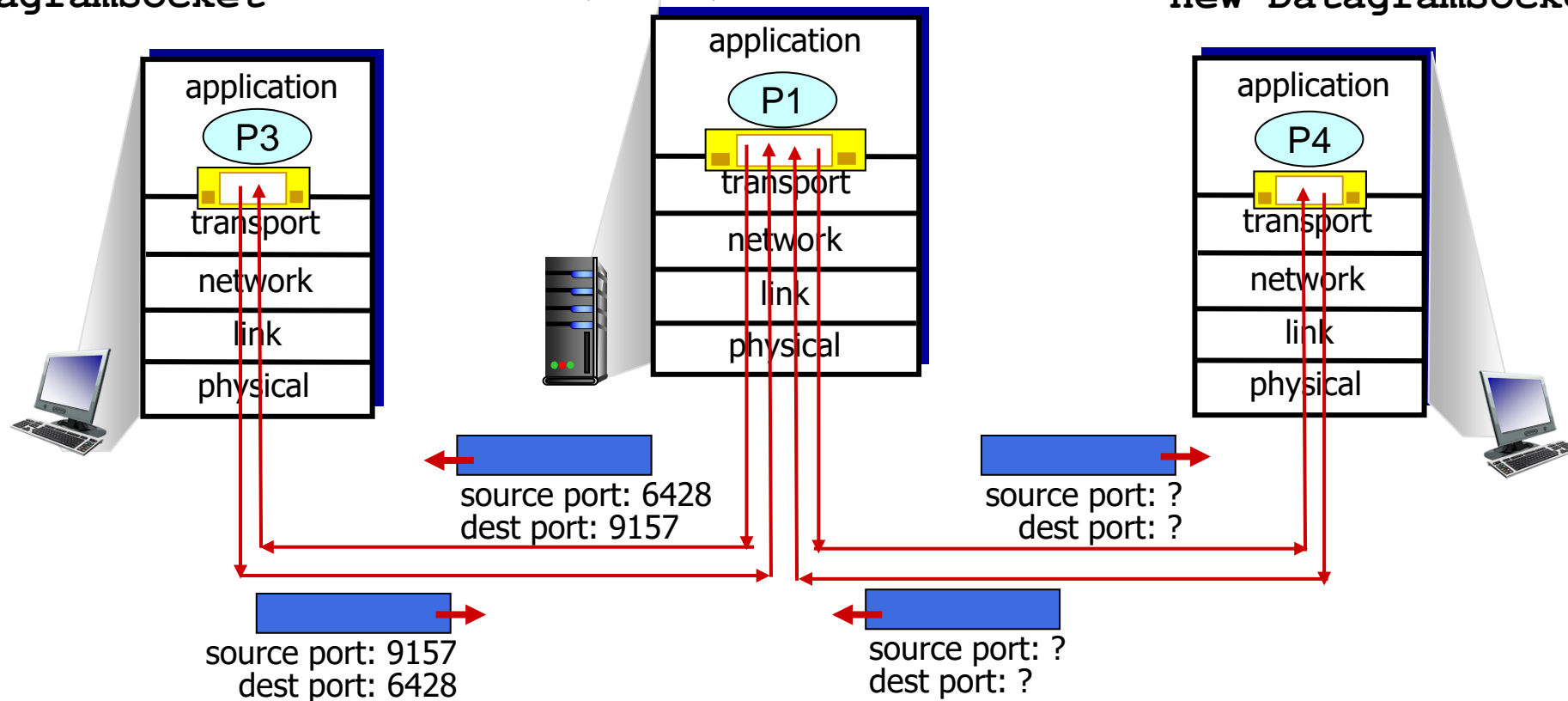
IP/UDP datagrams with *same dest. port #*, but different source IP addresses and/or source port numbers will be directed to *same socket* at receiving host

Connectionless demultiplexing: an example

```
DatagramSocket mySocket2 =  
new DatagramSocket  
(9157) ;
```

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

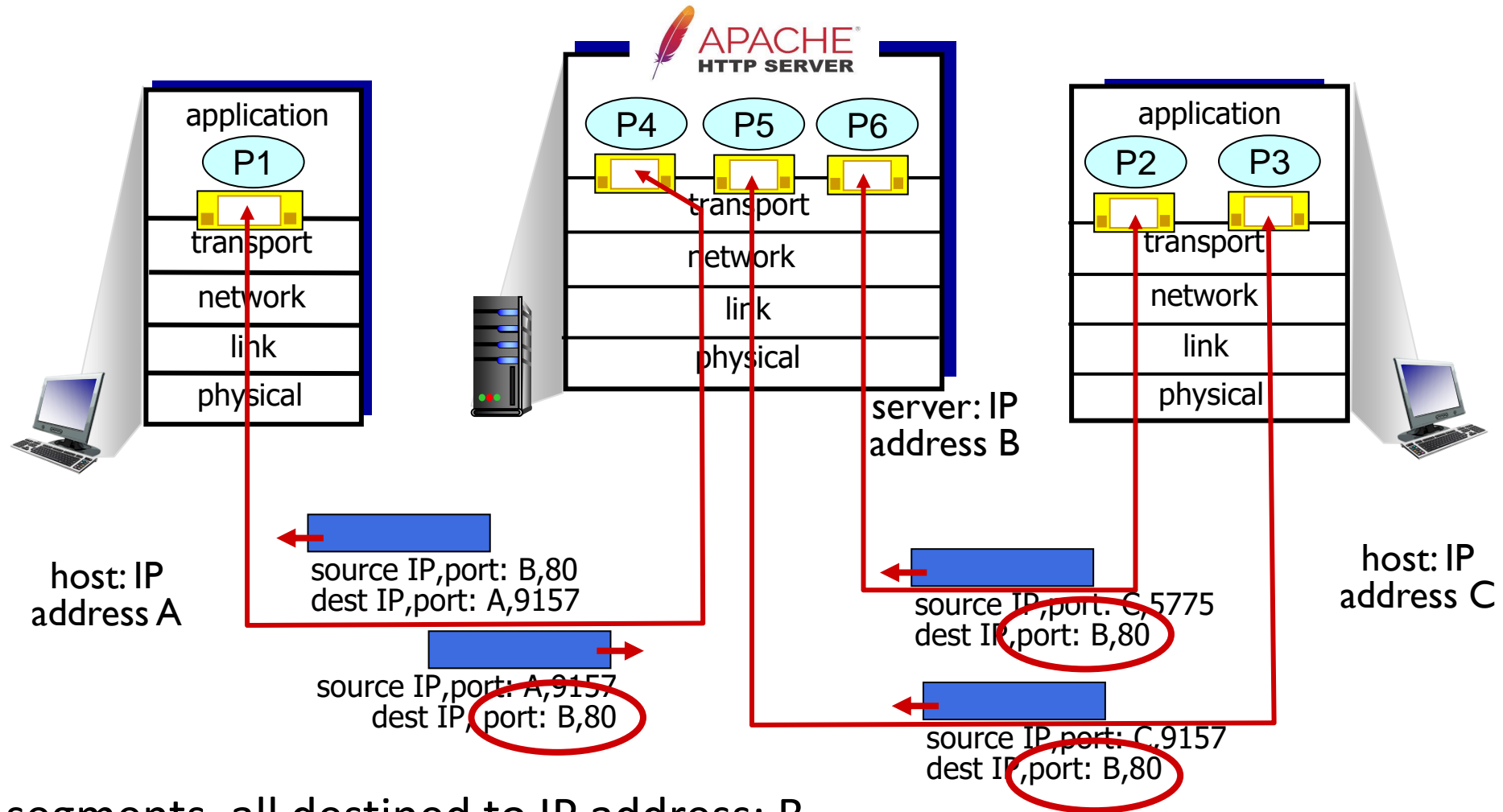
```
DatagramSocket mySocket1 =  
new DatagramSocket (5775) ;
```



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

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
 - 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: User Datagram Protocol [RFC 768]

INTERNET STANDARD

RFC 768

J. Postel

ISI

28 August 1980

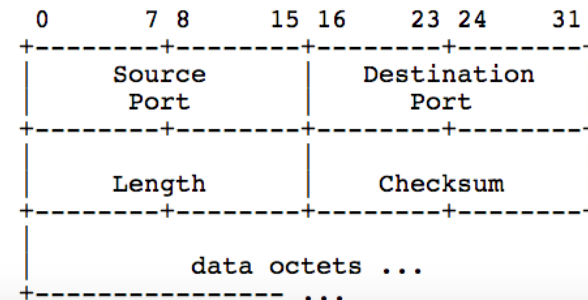
User Datagram Protocol

Introduction

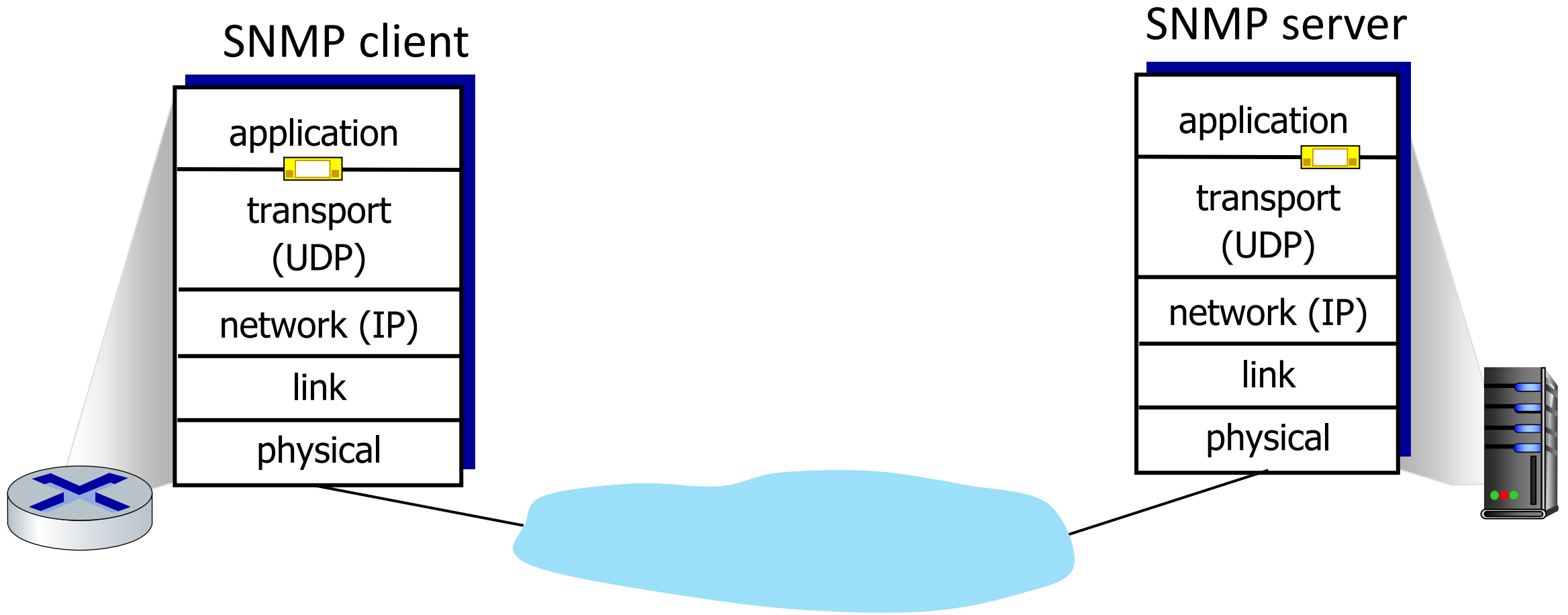
This User Datagram Protocol (UDP) is defined to make available a datagram mode of packet-switched computer communication in the environment of an interconnected set of computer networks. This protocol assumes that the Internet Protocol (IP) [1] is used as the underlying protocol.

This protocol provides a procedure for application programs to send messages to other programs with a minimum of protocol mechanism. The protocol is transaction oriented, and delivery and duplicate protection are not guaranteed. Applications requiring ordered reliable delivery of streams of data should use the Transmission Control Protocol (TCP) [2].

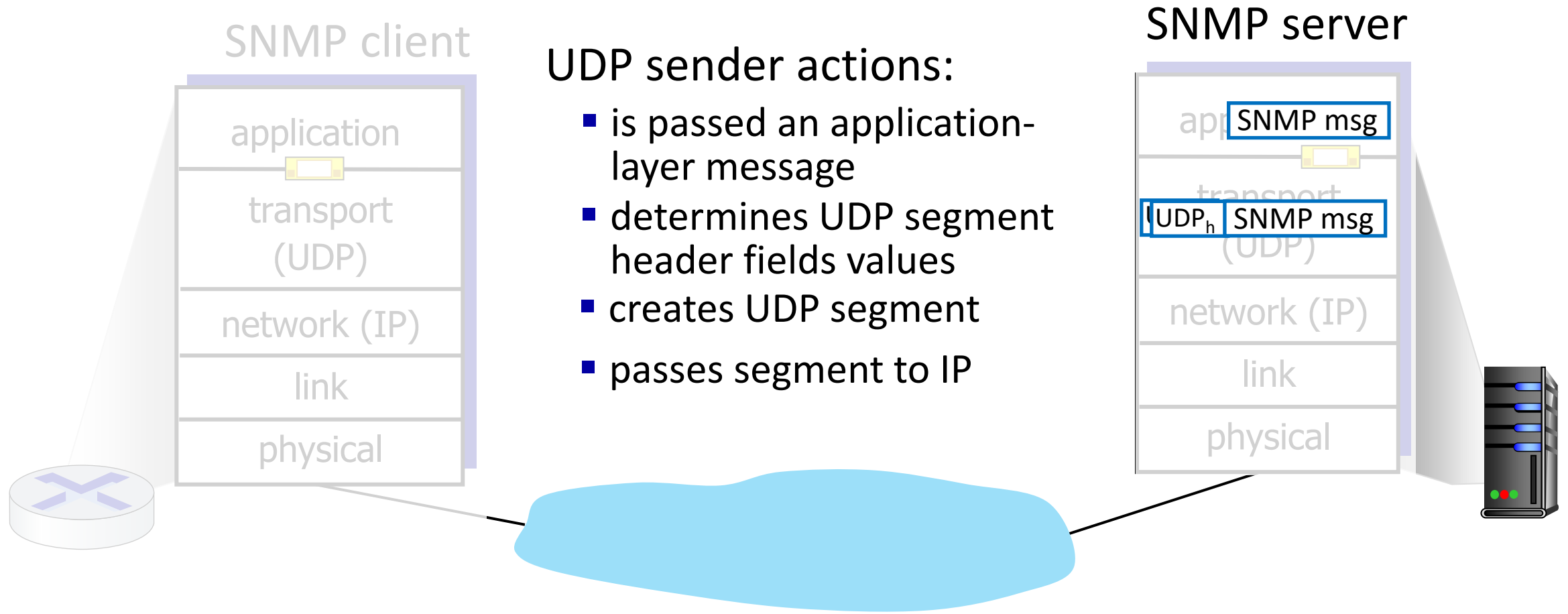
Format



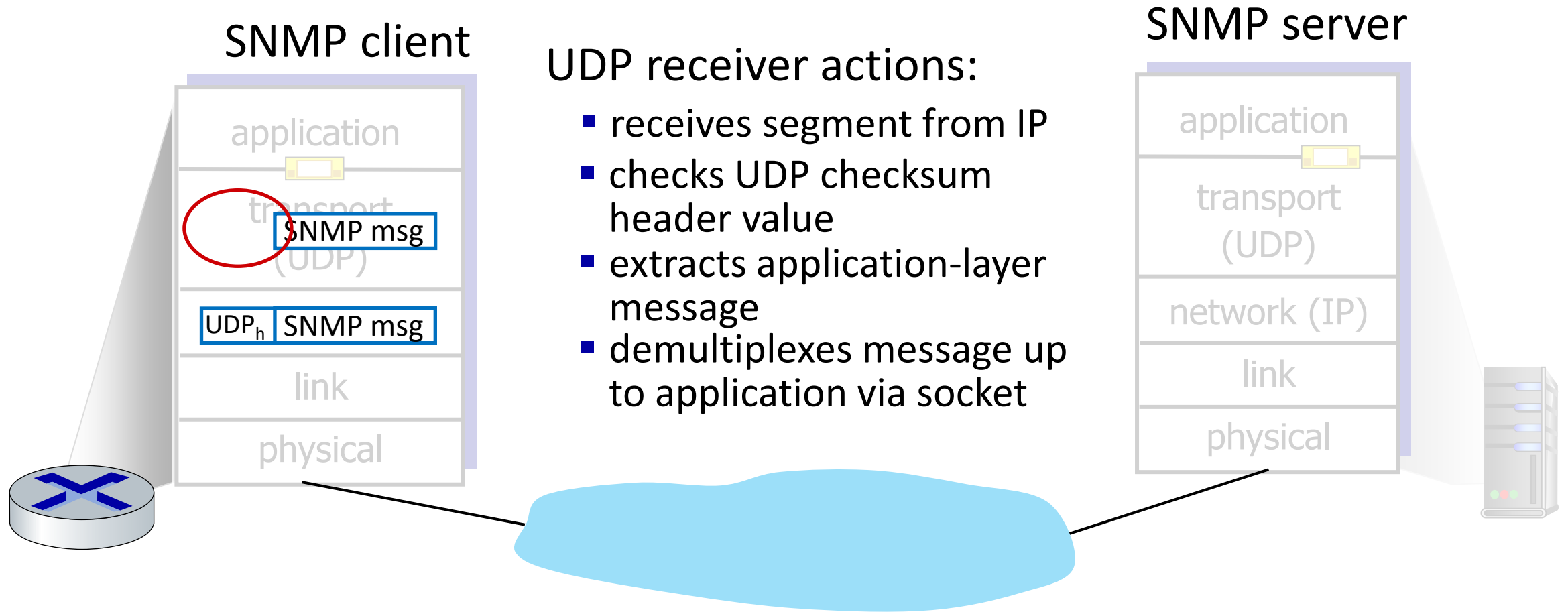
UDP: Transport Layer Actions



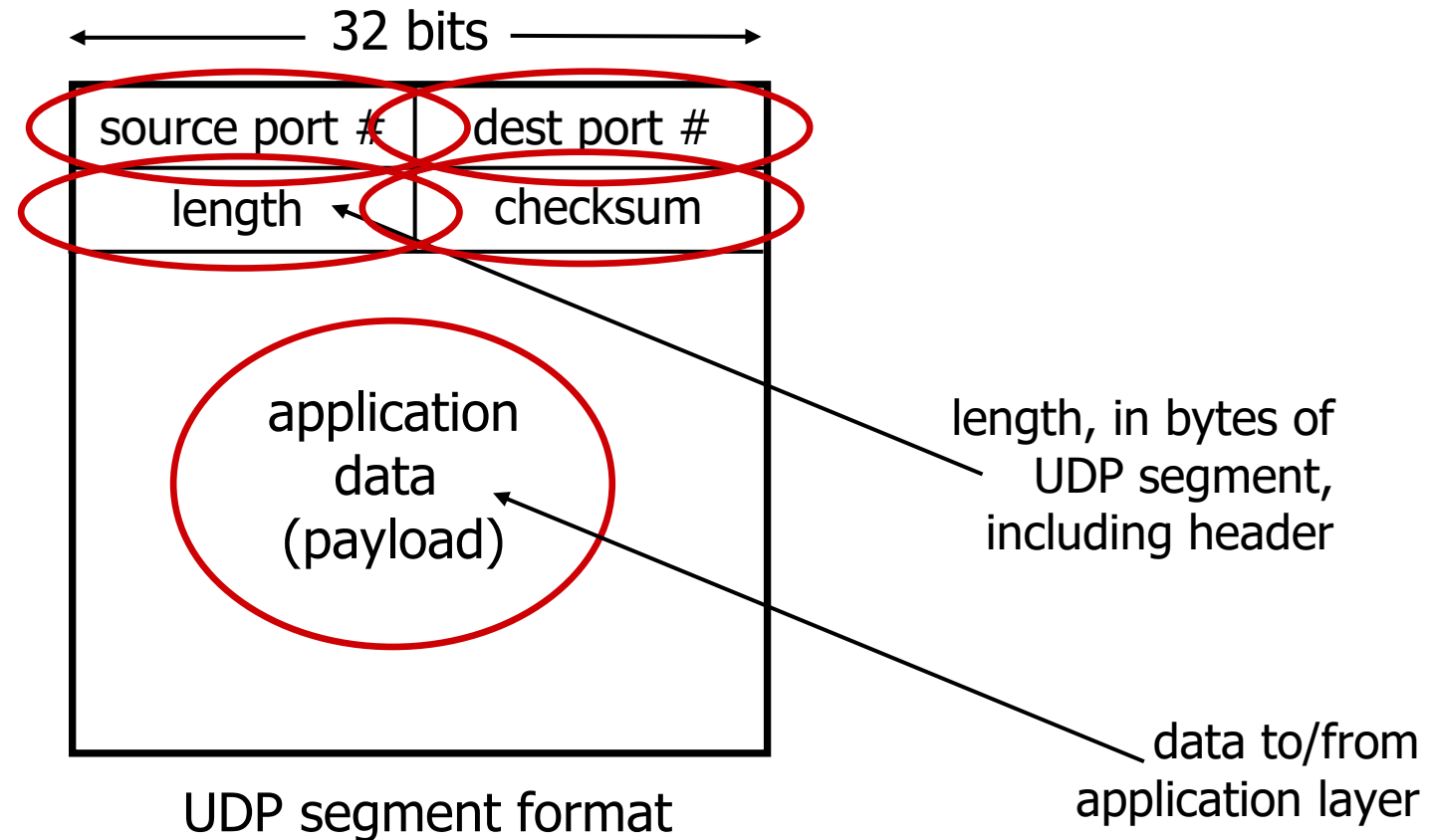
UDP: Transport Layer Actions



UDP: Transport Layer Actions

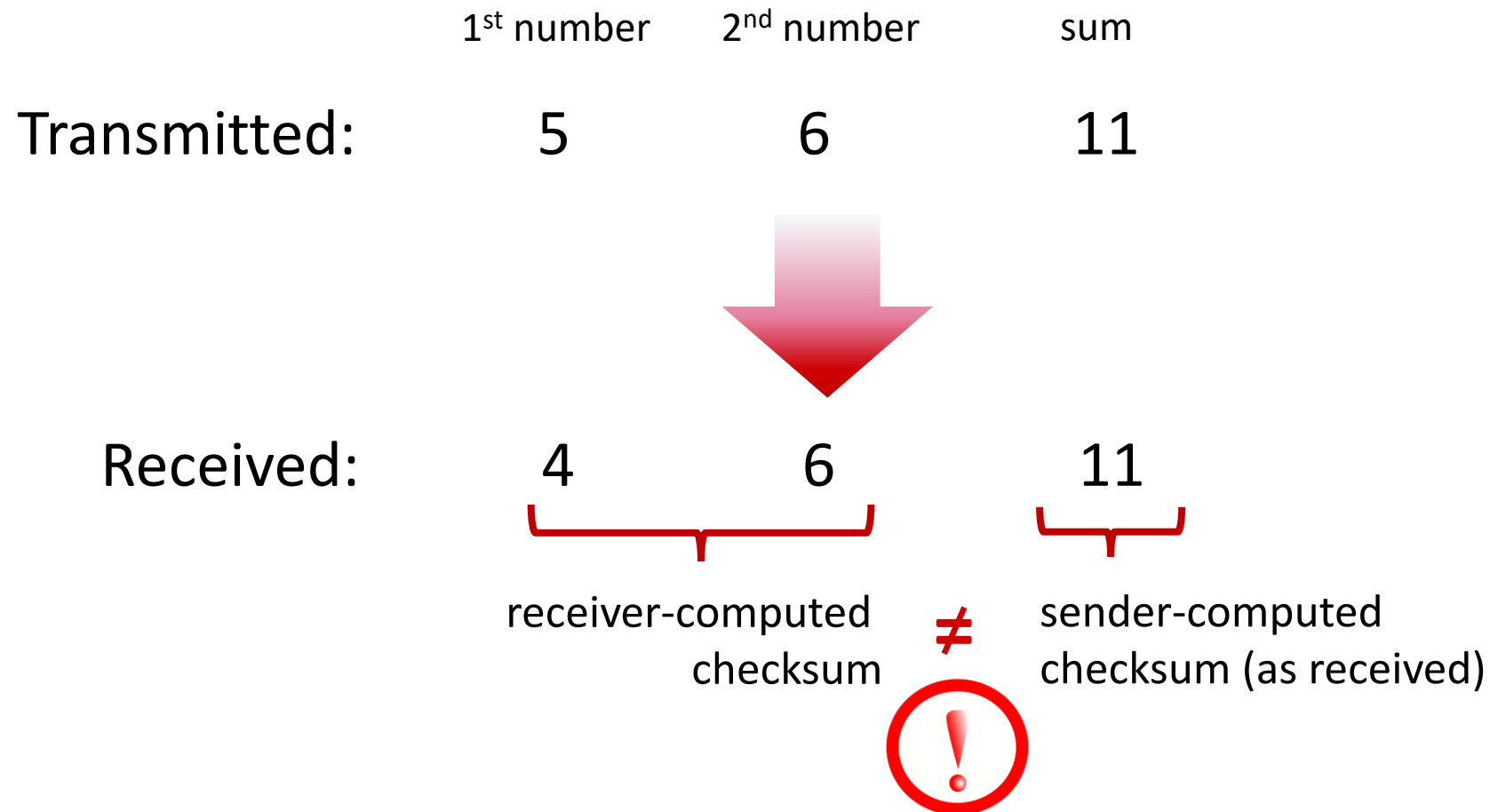


UDP segment header



UDP checksum

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



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	
<hr/>																	
wraparound	1	1	0	1	1	1	0	1	1	1	0	1	1	1	0	1	1
<hr/>																	
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	0	1	1

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

* Check out the online interactive exercises for more examples: http://gaia.cs.umass.edu/kurose_ross/interactive/

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
	1	1	0	1	0	1	0	1	0	1	0	1	0	1	0	1
	<hr/>															
wraparound	1	1	0	1	1	1	0	1	1	1	0	1	1	1	0	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

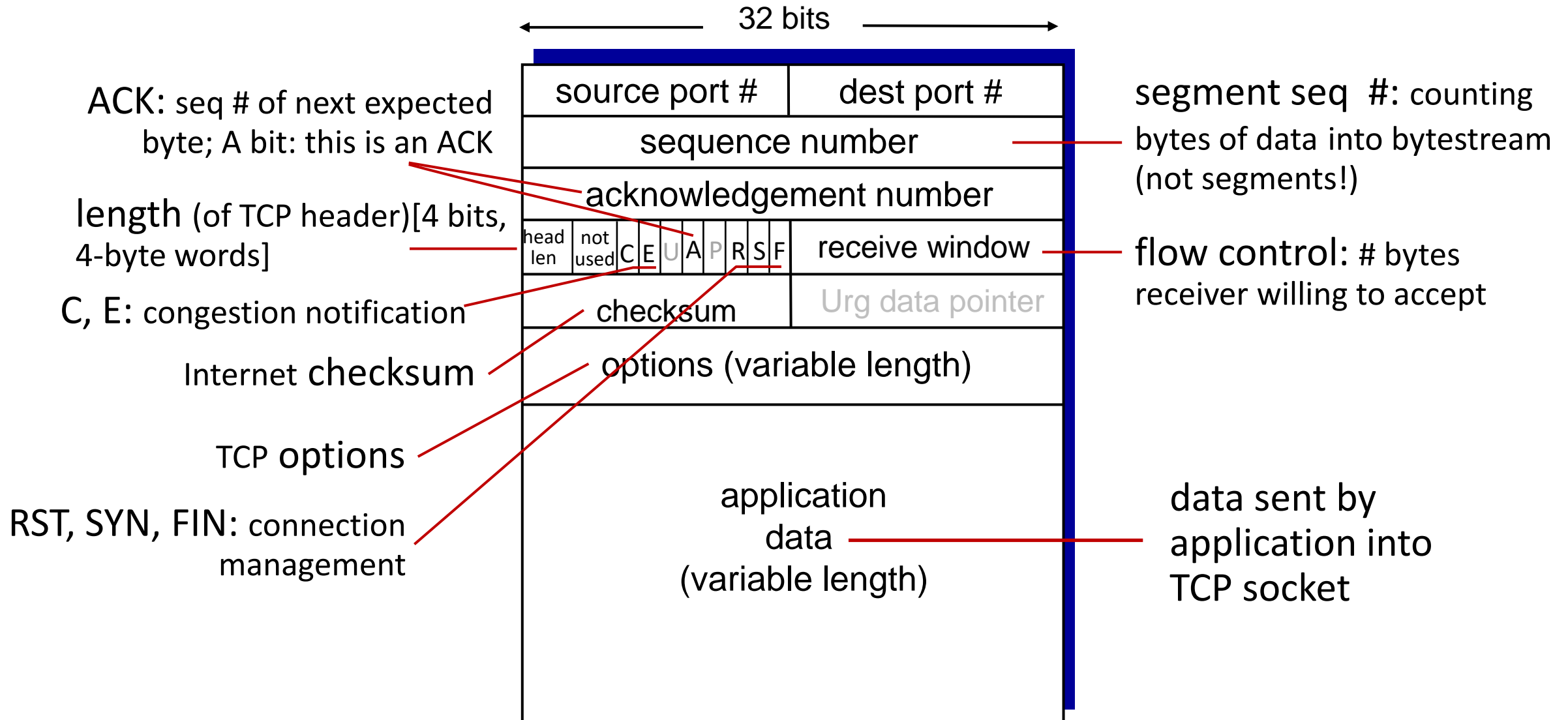
- “no frills” protocol:
 - segments may be lost, delivered out of order
 - best effort service: “send and hope for the best”
- UDP has its plusses:
 - no setup/handshaking needed (no RTT incurred)
 - can function when network service is compromised
 - helps with reliability (checksum)
- build additional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

TCP: overview

RFCs: 793, 1122, 2018, 5681, 7323

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

TCP segment structure



TCP sequence numbers, ACKs

Sequence numbers:

- byte stream “number” of first byte in segment’s data

Acknowledgements:

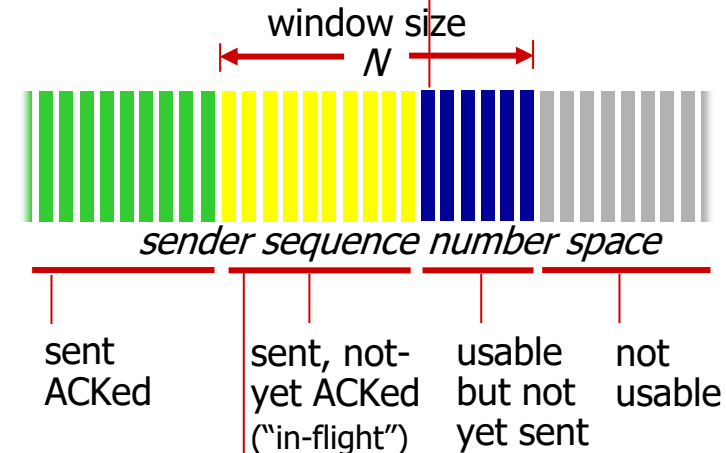
- seq # of next byte expected from other side
- cumulative ACK

Q: how receiver handles out-of-order segments

- A: TCP spec doesn’t say, - up to implementor

outgoing segment from sender

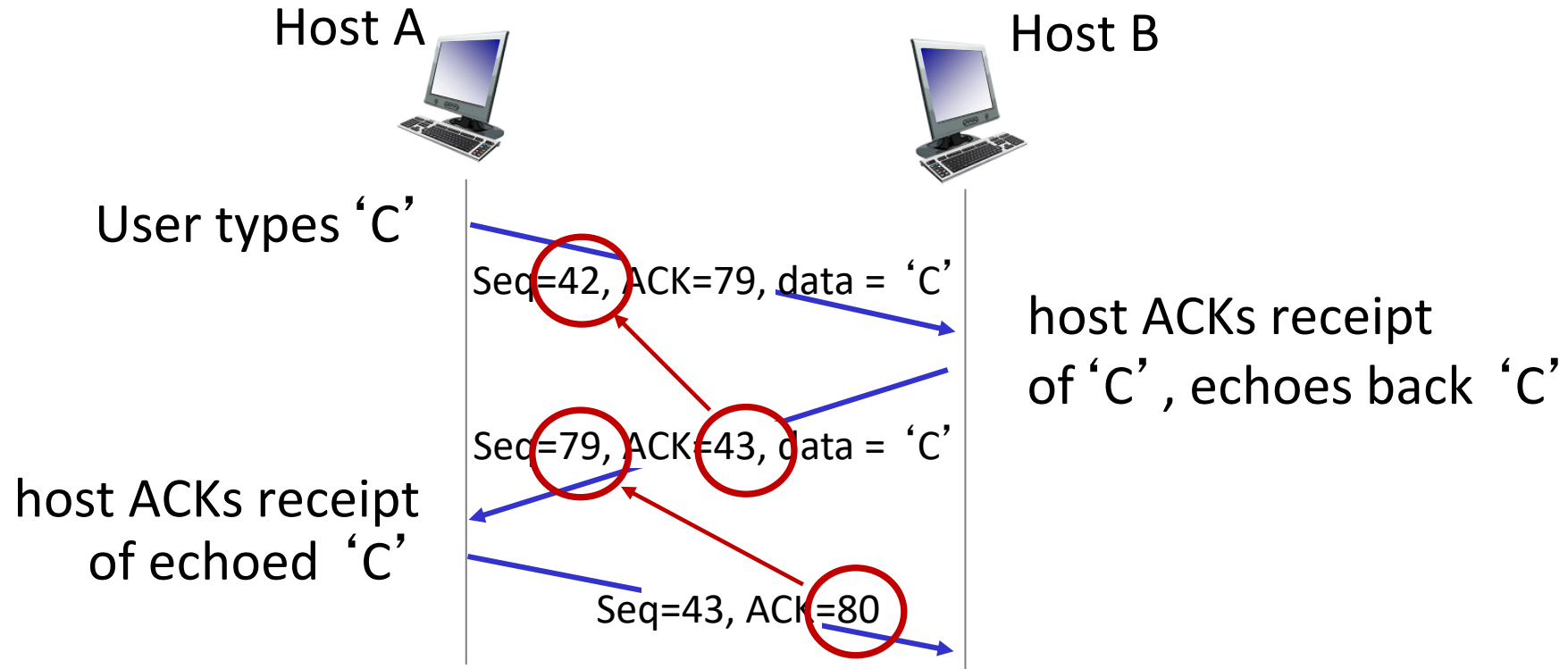
source port #	dest port #
sequence number	
acknowledgement number	
	rwnd
checksum	urg pointer



outgoing segment from receiver

source port #	dest port #
sequence number	
acknowledgement number	
	A
checksum	urg pointer

TCP sequence numbers, ACKs



simple telnet scenario

TCP round trip time, timeout

Q: how to set TCP timeout value?

- longer than RTT, but RTT varies!
- *too short*: premature timeout, unnecessary retransmissions
- *too long*: slow reaction to segment loss

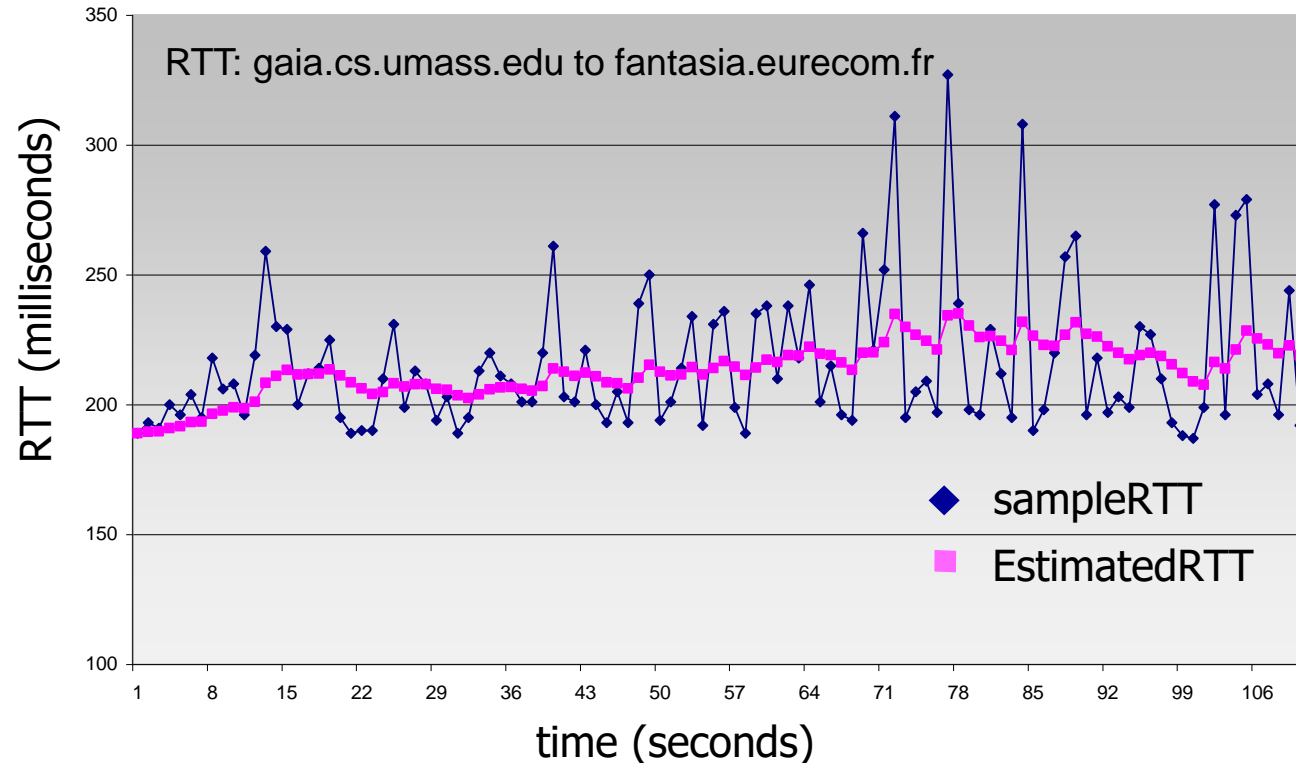
Q: how to estimate RTT?

- *SampleRTT*: measured time from segment transmission until ACK receipt
 - ignore retransmissions
- *SampleRTT* will vary, want estimated RTT “smoother”
 - average several *recent* measurements, not just current *SampleRTT*

TCP round trip time, timeout

$$\text{EstimatedRTT} = (1 - \alpha) * \text{EstimatedRTT} + \alpha * \text{SampleRTT}$$

- exponential weighted moving average (EWMA)
- influence of past sample decreases exponentially fast
- typical value: $\alpha = 0.125$



TCP round trip time, timeout

- timeout interval: **EstimatedRTT** plus “safety margin”
 - large variation in **EstimatedRTT**: want a larger safety margin

$$\text{TimeoutInterval} = \text{EstimatedRTT} + 4 * \text{DevRTT}$$



↑
estimated RTT

↑
“safety margin”

- **DevRTT**: EWMA of **SampleRTT** deviation from **EstimatedRTT**:

$$\text{DevRTT} = (1 - \beta) * \text{DevRTT} + \beta * |\text{SampleRTT} - \text{EstimatedRTT}|$$

(typically, $\beta = 0.25$)

TCP Sender (simplified)

event: data received from application

- create segment with seq #
- seq # 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

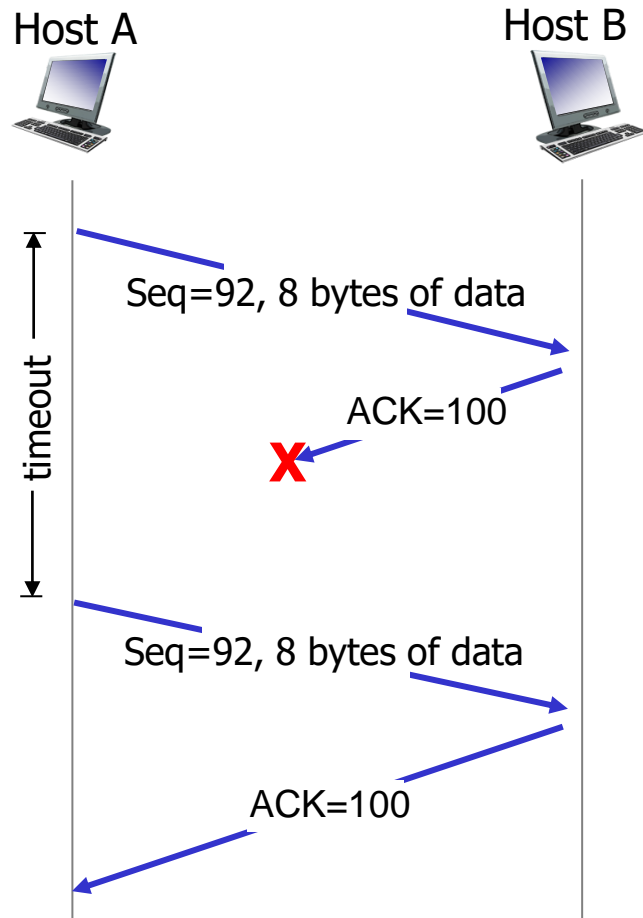
event: *ACK received*

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

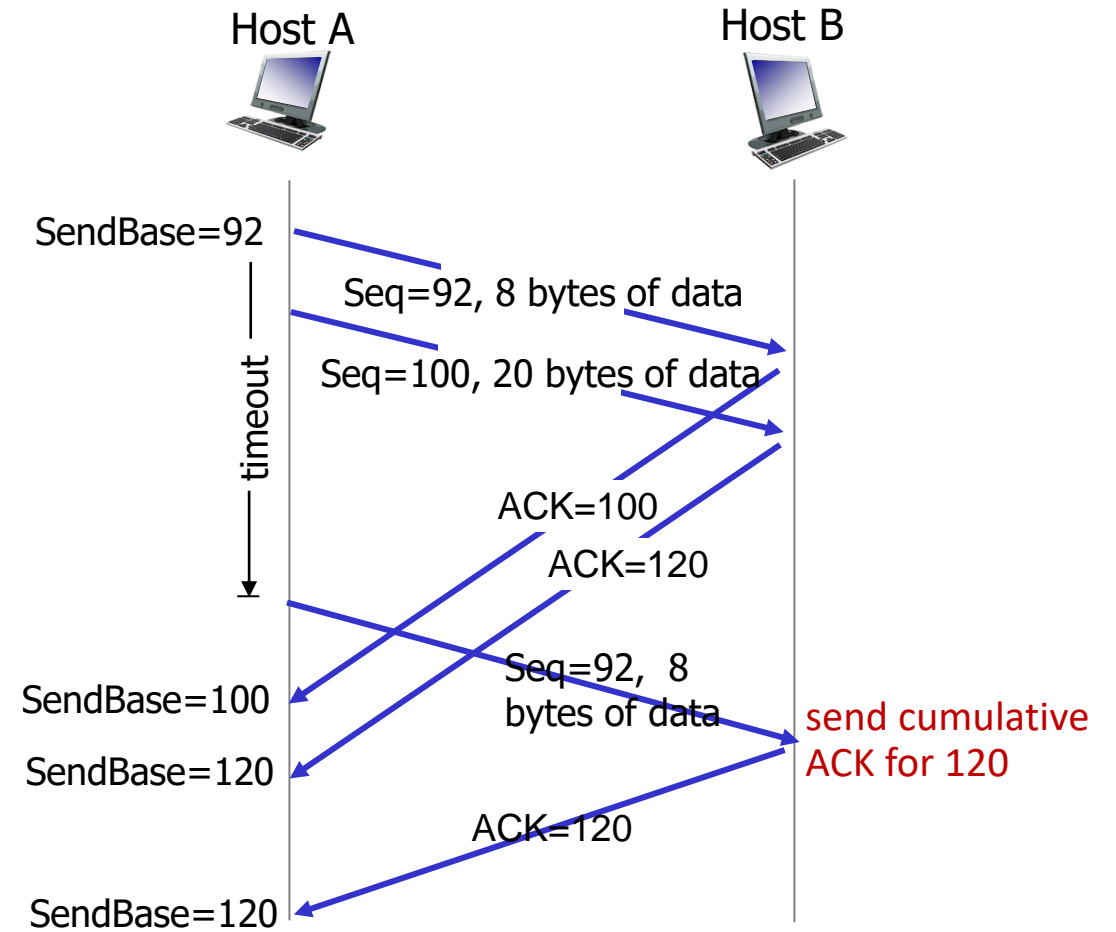
TCP Receiver: ACK generation [RFC 5681]

<i>Event at receiver</i>	<i>TCP receiver action</i>
arrival of in-order segment with expected seq #. All data up to expected seq # already ACKed	delayed ACK. Wait up to 500ms for next segment. If no next segment, send ACK
arrival of in-order segment with expected seq #. One other segment has ACK pending	immediately send single cumulative ACK, ACKing both in-order segments
arrival of out-of-order segment higher-than-expect seq. # . Gap detected	immediately send <i>duplicate ACK</i> , indicating seq. # of next expected byte

TCP: retransmission scenarios

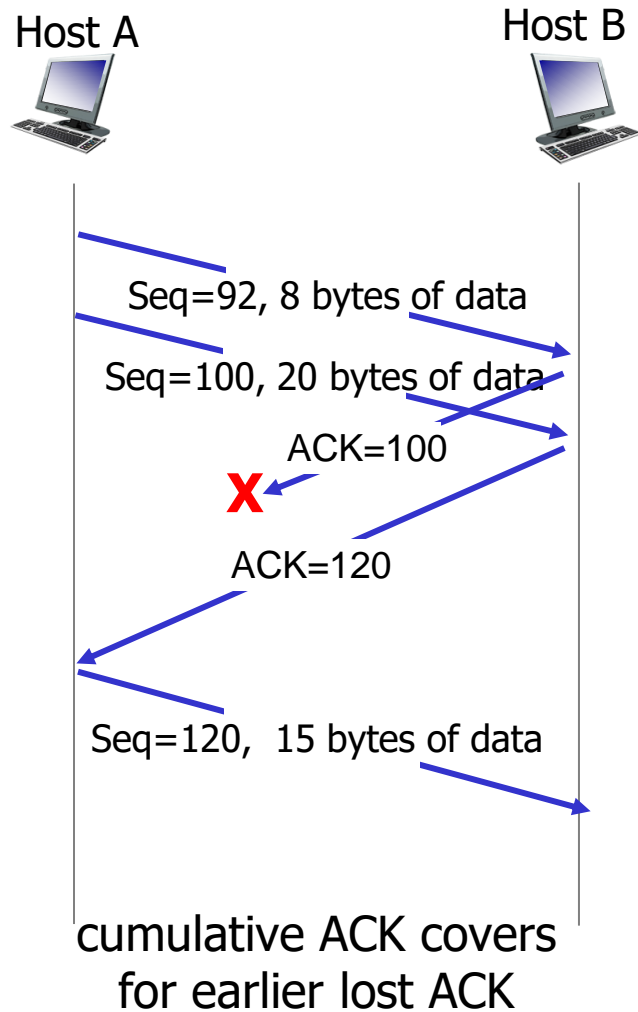


lost ACK scenario



premature timeout

TCP: retransmission scenarios



TCP fast retransmit

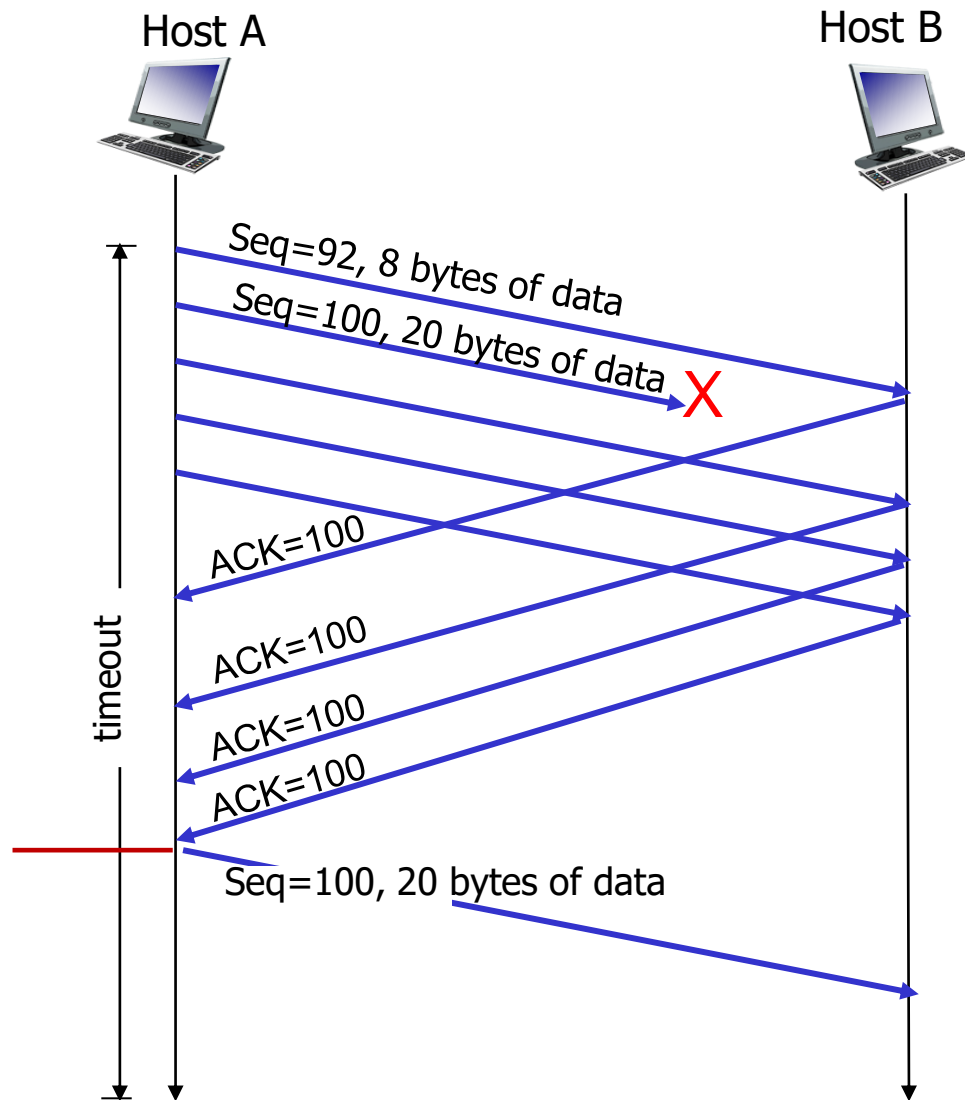
TCP fast retransmit

if sender receives 3 additional ACKs for same data (“triple duplicate ACKs”), resend unACKed segment with smallest seq #

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

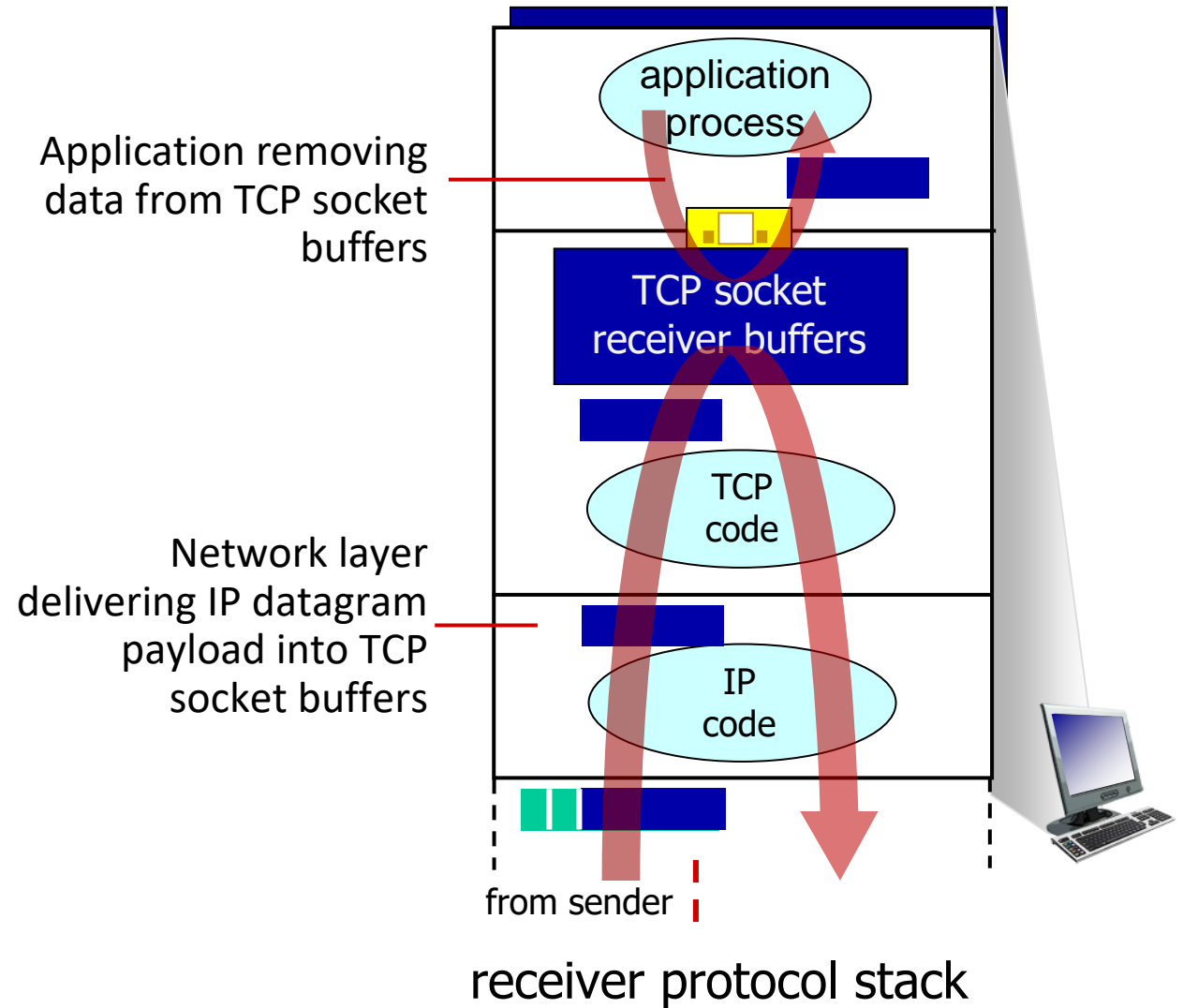


Receipt of three duplicate ACKs indicates 3 segments received after a missing segment – lost segment is likely. So retransmit!



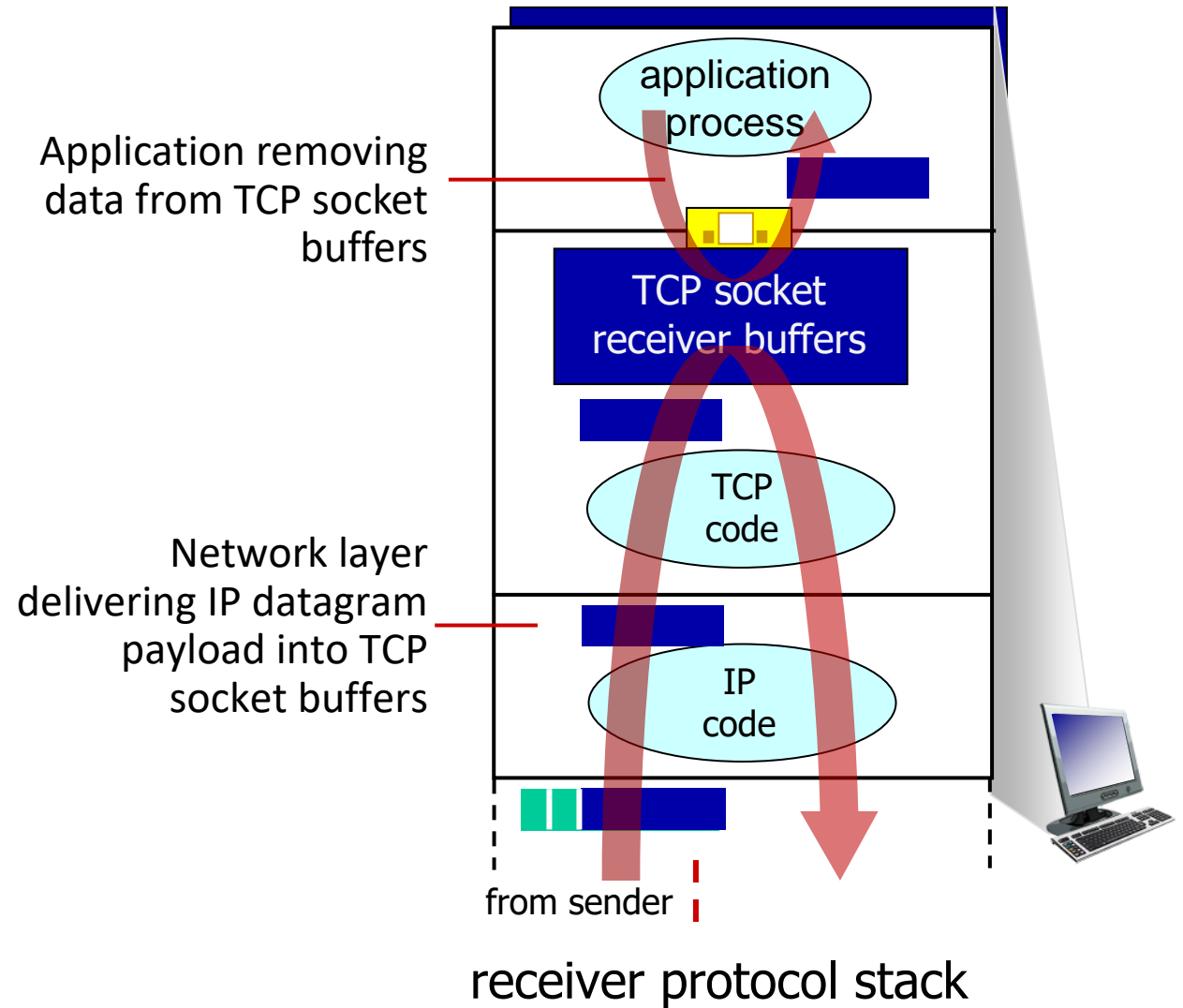
TCP flow control

Q: What happens if network layer delivers data faster than application layer removes data from socket buffers?



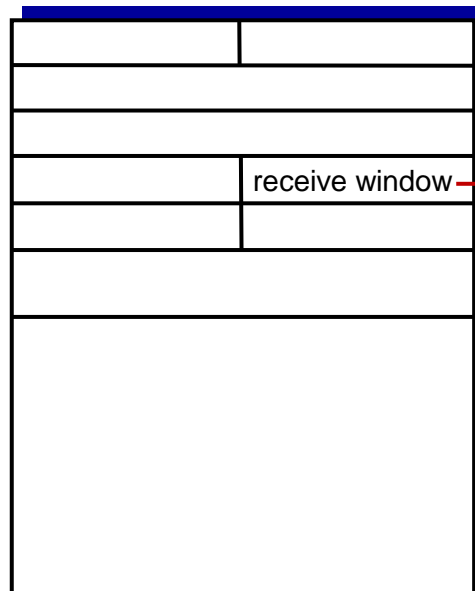
TCP flow control

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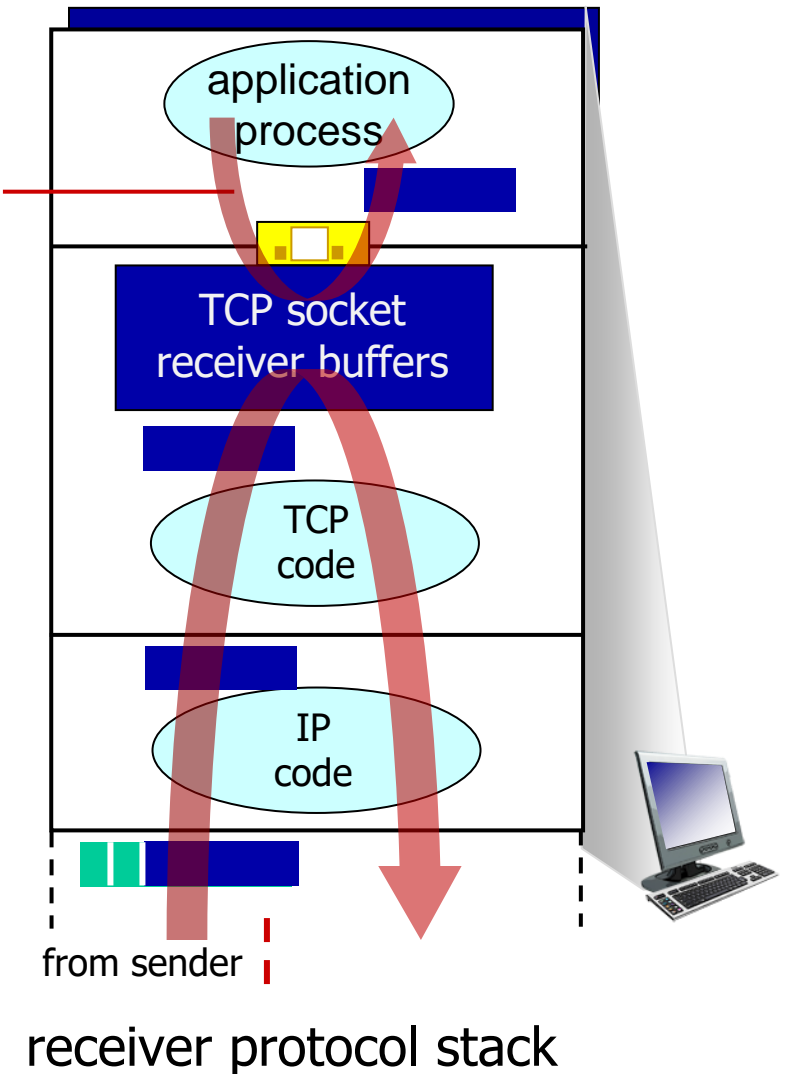
TCP flow control

Q: What happens if network layer delivers data faster than application layer removes data from socket buffers?



flow control: # bytes
receiver willing to accept

Application removing
data from TCP socket
buffers

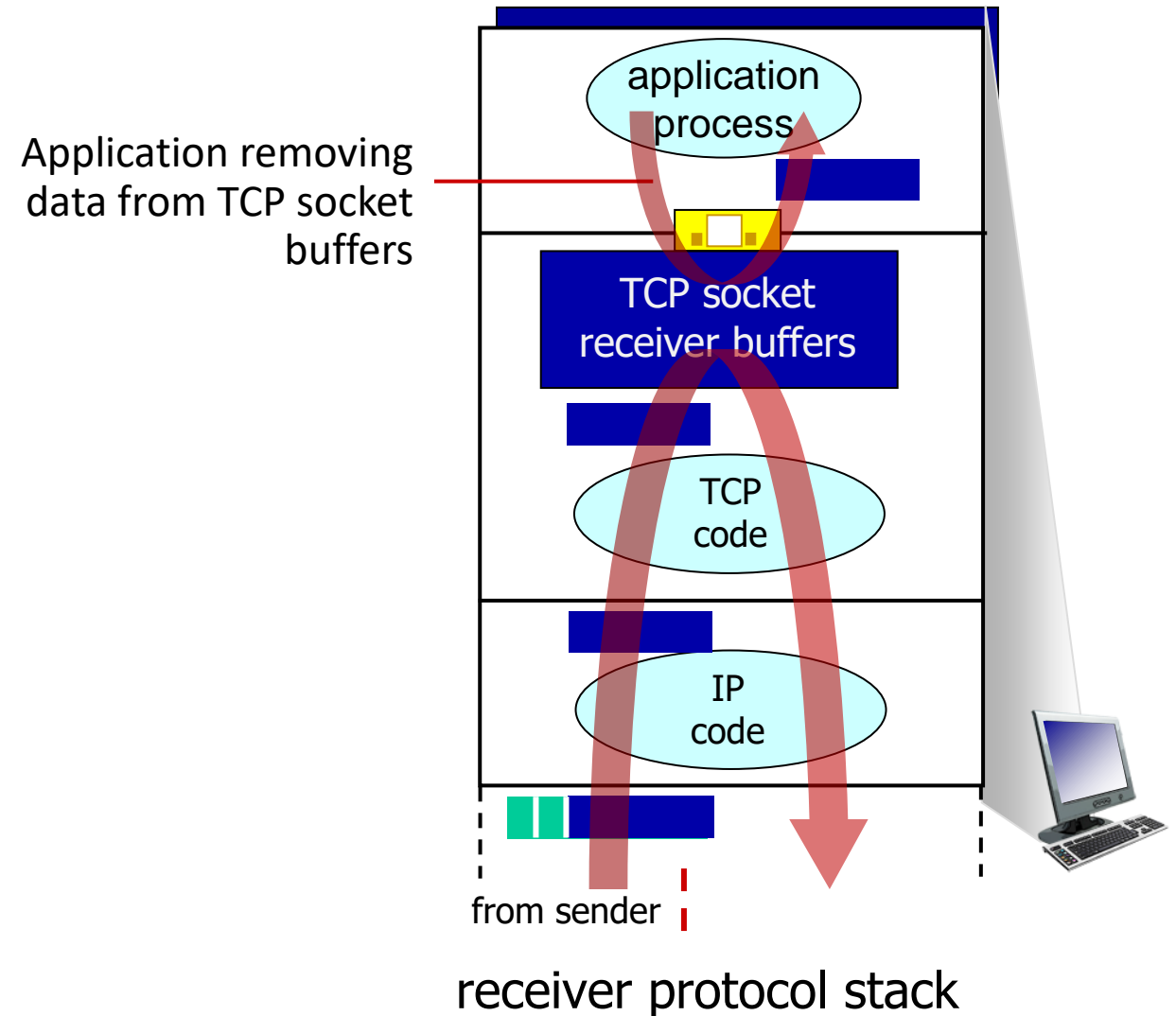


TCP flow control

Q: What happens if network layer delivers data faster than application layer removes data from socket buffers?

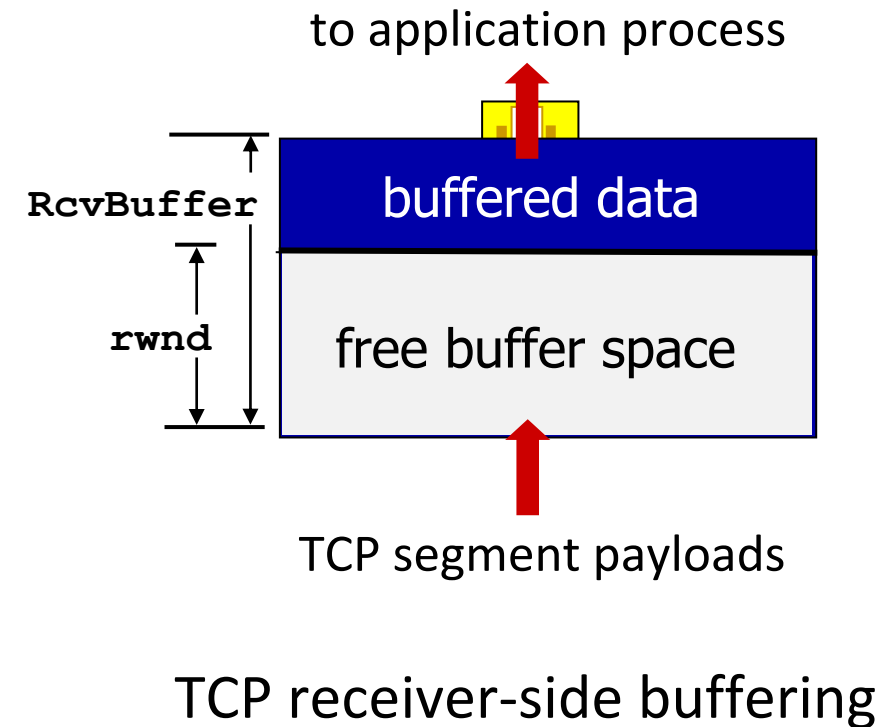
—flow control—

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



TCP flow control

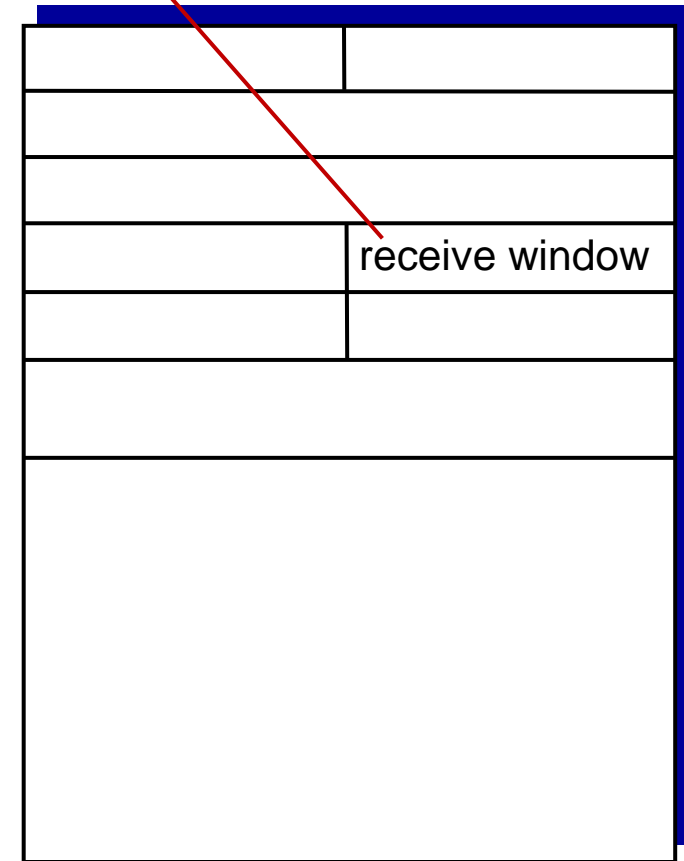
- TCP receiver “advertises” free buffer space in **rwnd** field in TCP header
 - **RcvBuffer** size set via socket options (typical default is 4096 bytes)
 - many operating systems autoadjust **RcvBuffer**
- sender limits amount of unACKed (“in-flight”) data to received **rwnd**
- guarantees receive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



TCP 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**
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flow control: # bytes receiver willing to accept

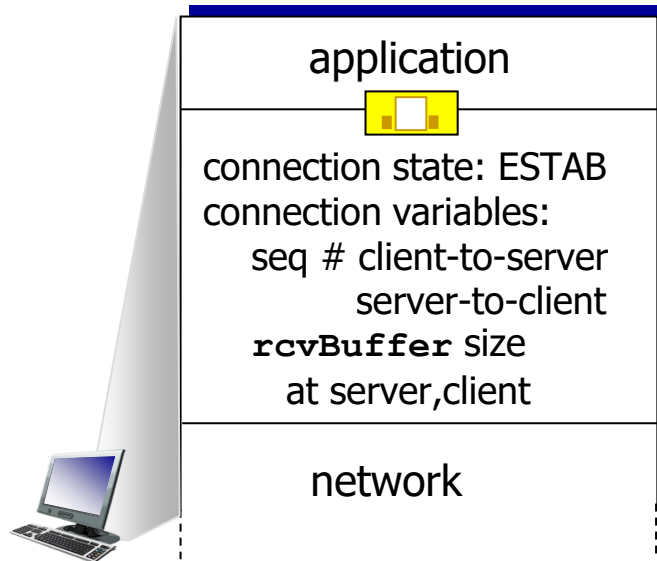


TCP segment format

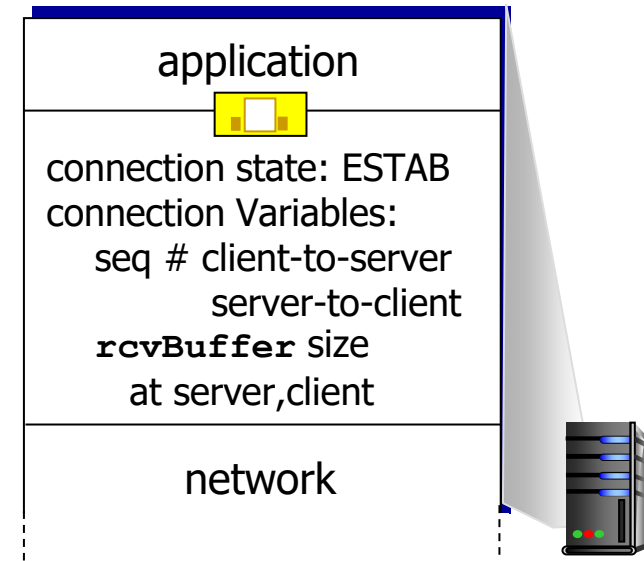
TCP connection management

before exchanging data, sender/receiver “handshake”:

- agree to establish connection (each knowing the other willing to establish connection)
- agree on connection parameters (e.g., starting seq #s)



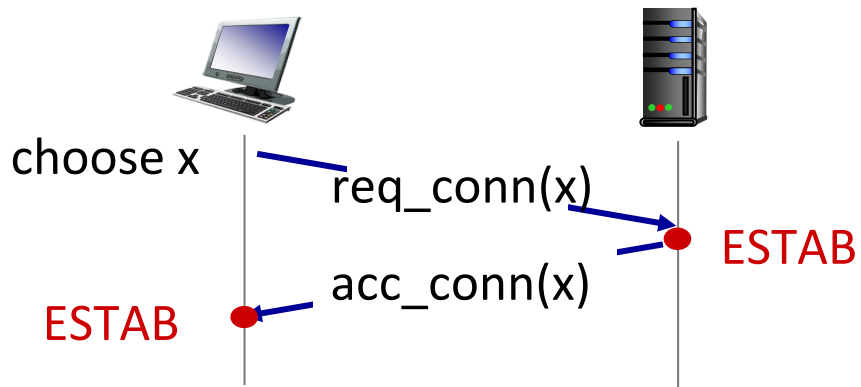
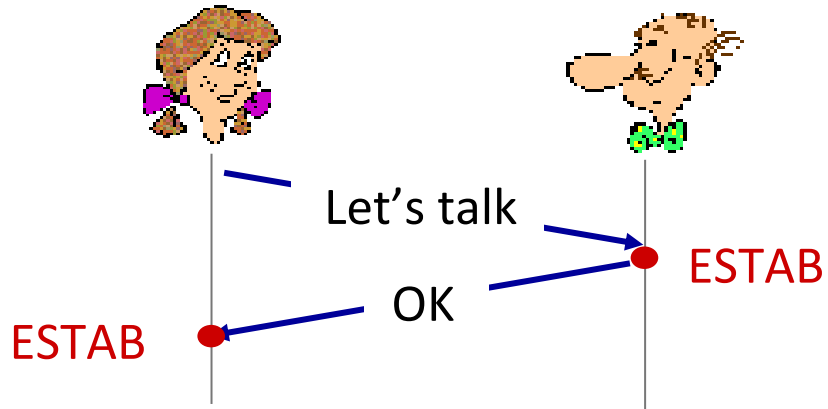
```
Socket clientSocket =  
    newSocket("hostname", "port number");
```



```
Socket connectionSocket =  
    welcomeSocket.accept();
```

Agreeing to establish a connection

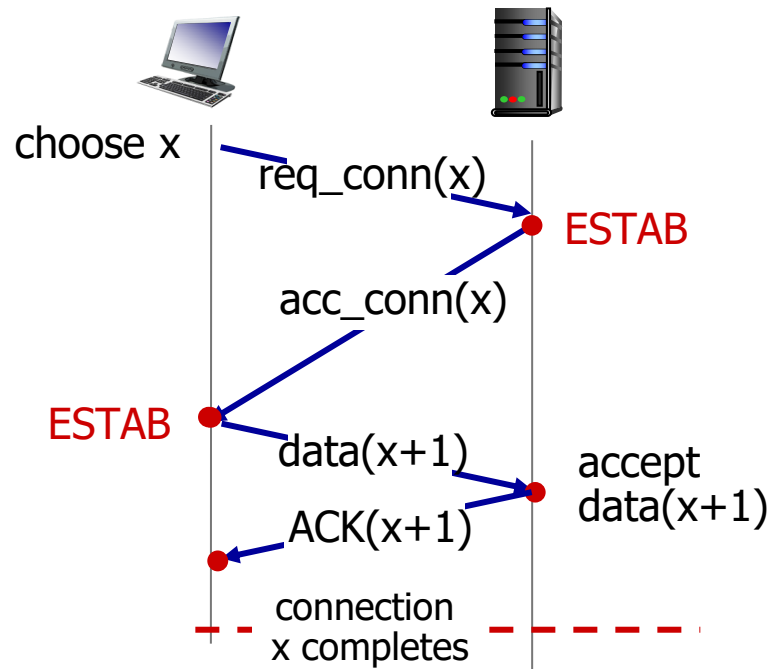
2-way handshake:



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

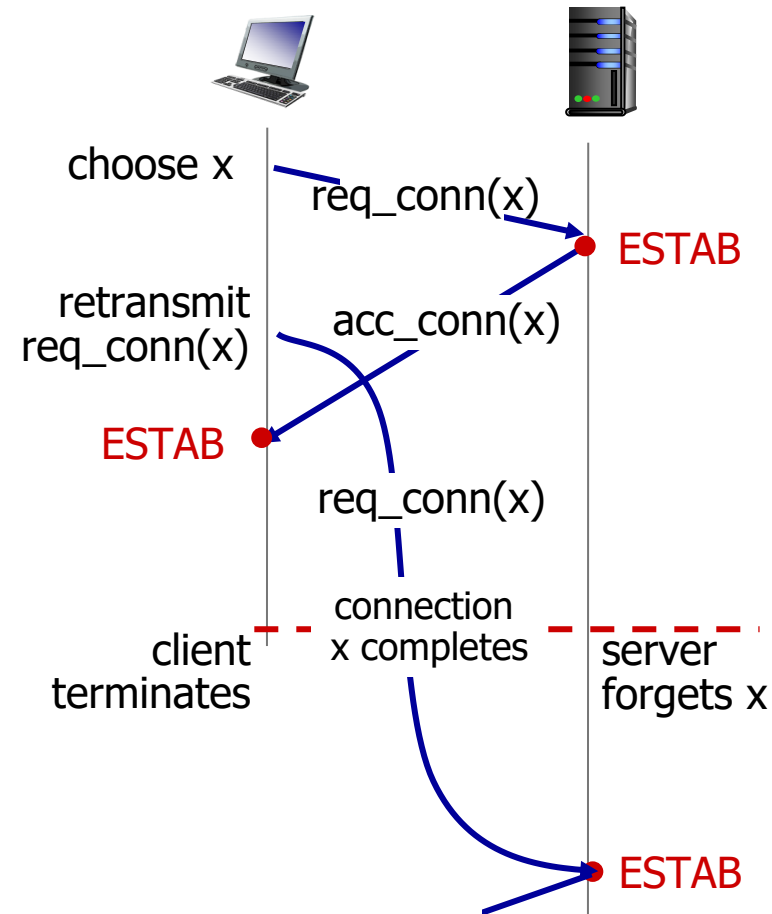
2-way handshake scenarios




No problem!

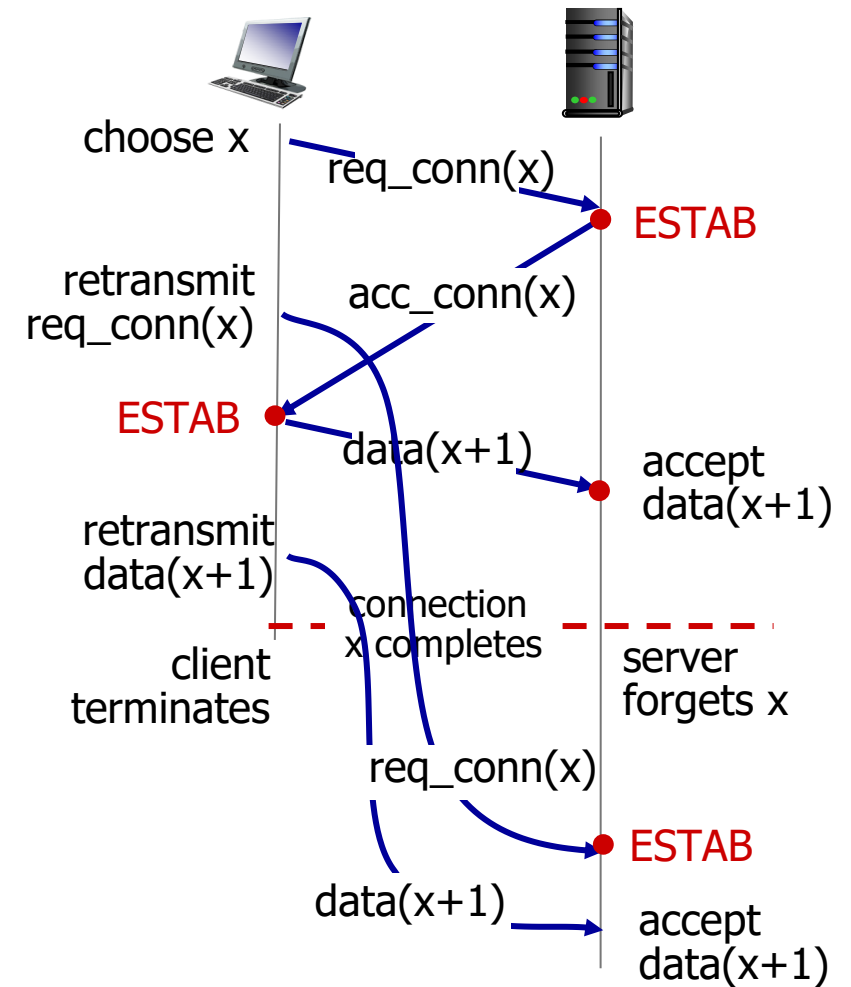


2-way handshake scenarios



 Problem: half open connection! (no client)

2-way handshake scenarios



 Problem: dup data accepted!

TCP 3-way handshake

Client state

```
clientSocket = socket(AF_INET, SOCK_STREAM)
```

LISTEN

```
clientSocket.connect((serverName, serverPort))
```

SYNSENT

ESTAB

choose init seq num, x
send TCP SYN msg

received SYNACK(x)
indicates server is live;
send ACK for SYNACK;
this segment may contain
client-to-server data



SYNbit=1, Seq=x

SYNbit=1, Seq=y
ACKbit=1; ACKnum=x+1

ACKbit=1, ACKnum=y+1



choose init seq num, y
send TCP SYNACK
msg, acking SYN

received ACK(y)
indicates client is live

Server state

```
serverSocket = socket(AF_INET, SOCK_STREAM)  
serverSocket.bind(('', serverPort))  
serverSocket.listen(1)  
connectionSocket, addr = serverSocket.accept()
```

LISTEN

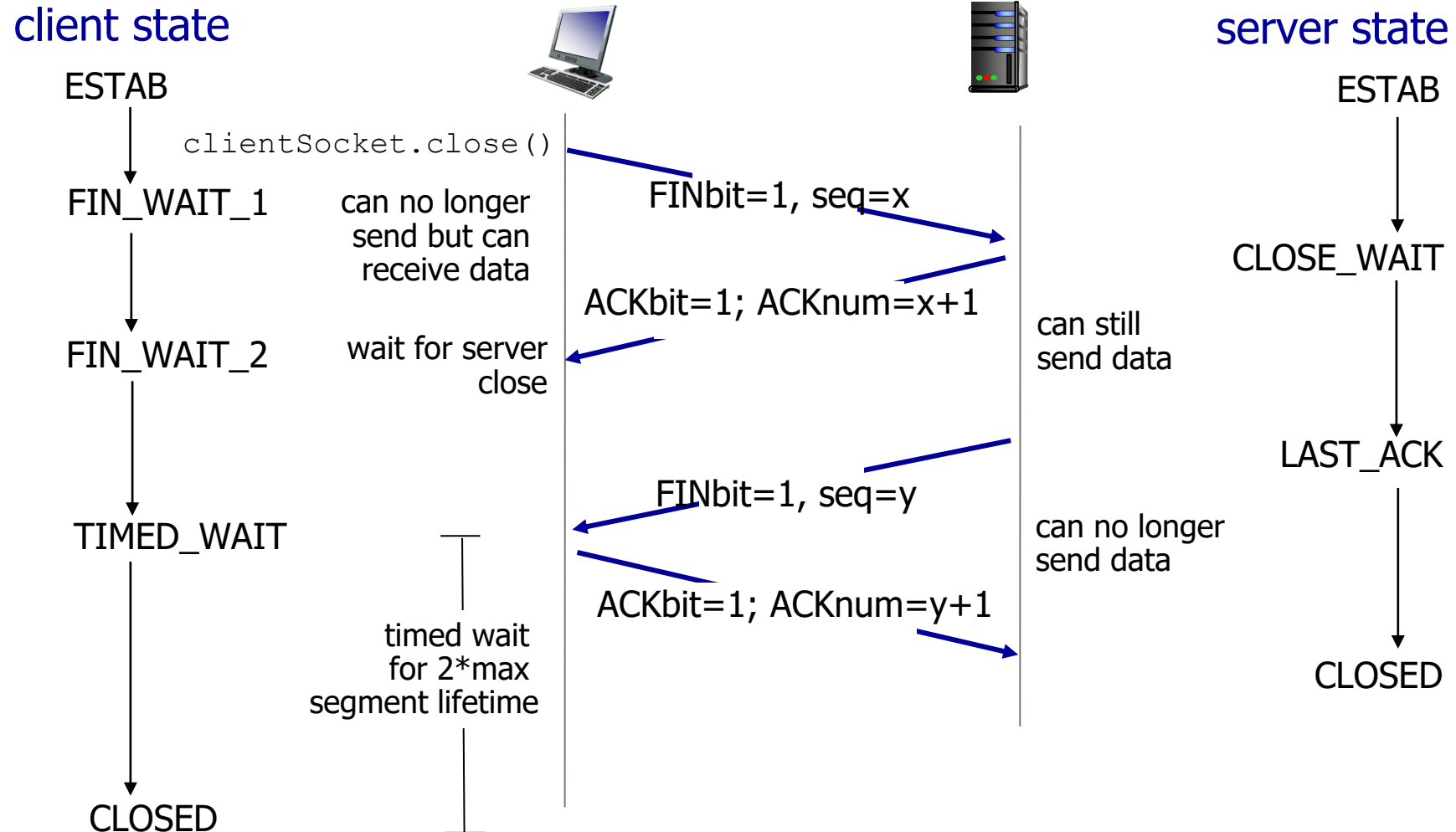
SYN RCVD

ESTAB

Closing a TCP connection

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

Closing a TCP connection



Principles of congestion control

Congestion:

- informally: “too many sources sending too much data too fast for *network* to handle”
- manifestations:
 - long delays (queueing in router buffers)
 - packet loss (buffer overflow at routers)
- different from flow control!
- a top-10 problem!



congestion control:

too many senders,
sending too fast

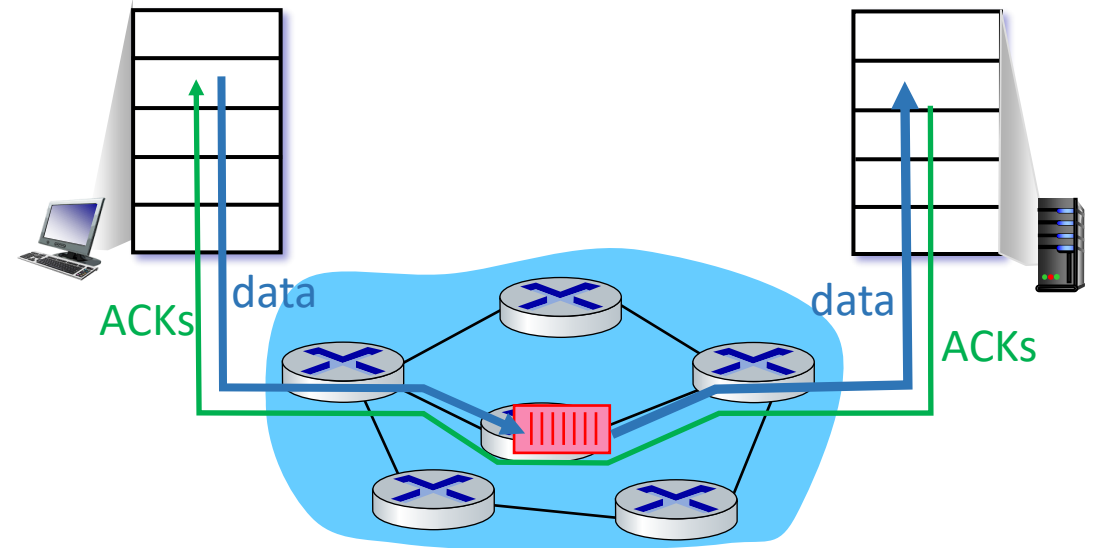


flow control: one sender
too fast for one receiver

Approaches towards congestion control

End-end congestion control:

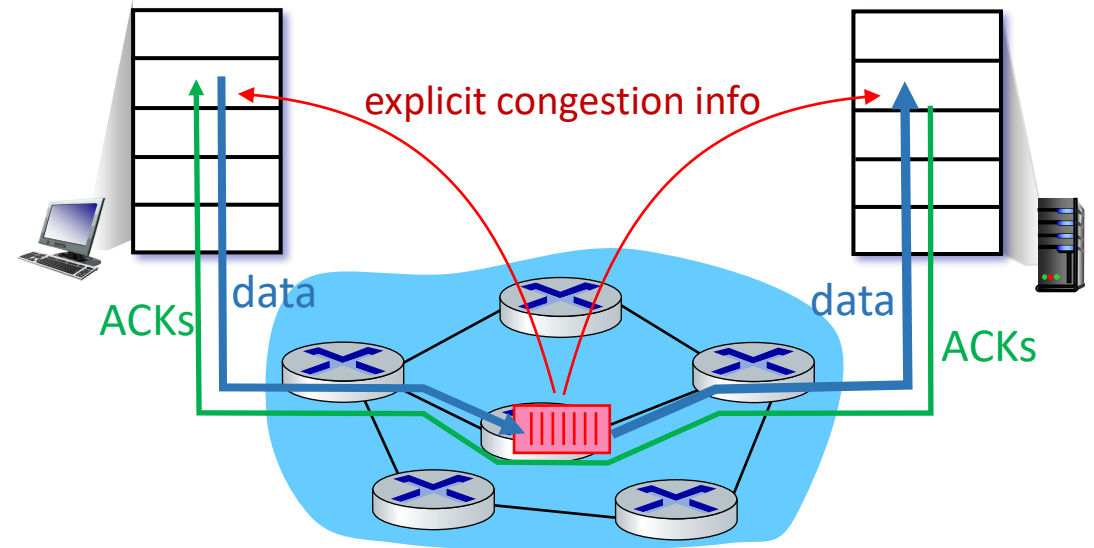
- no explicit feedback from network
- congestion *inferred* from observed loss, delay
- approach taken by TCP



Approaches towards congestion control

Network-assisted congestion control:

- routers provide *direct* feedback to sending/receiving hosts with flows passing through congested router
- may indicate congestion level or explicitly set sending rate
- TCP ECN, ATM, DECbit protocols



TCP congestion control: AIMD

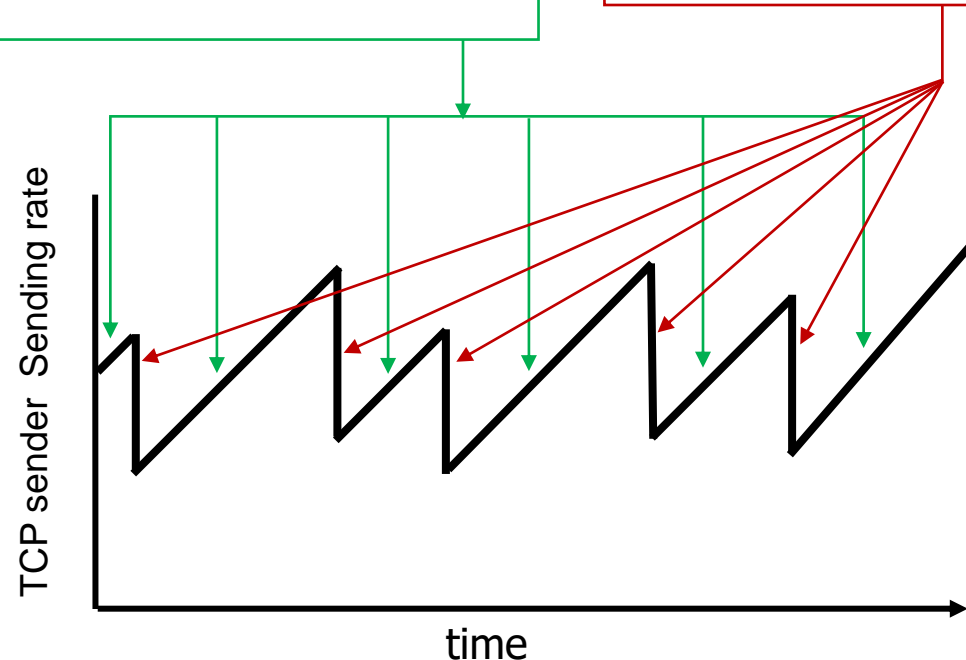
- *approach*: senders can increase sending rate until packet loss (congestion) occurs, then decrease sending rate on loss event

Additive Increase

increase sending rate by 1 maximum segment size every RTT until loss detected

Multiplicative Decrease

cut sending rate in half at each loss event



AIMD sawtooth behavior: *probing* for bandwidth

TCP AIMD: more

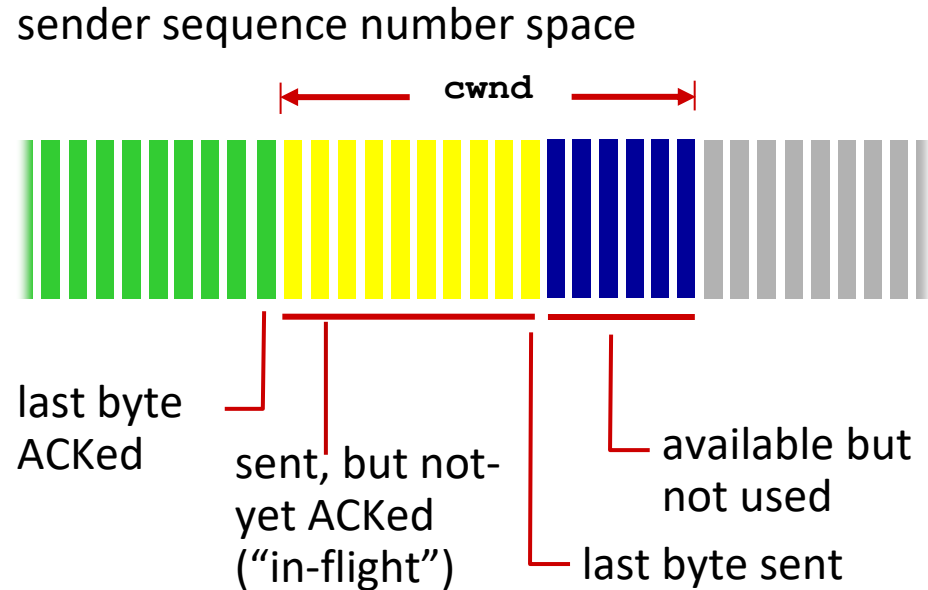
Multiplicative decrease detail: sending rate is

- Cut in half on loss detected by triple duplicate ACK (TCP Reno)
- Cut to 1 MSS (maximum segment size) when loss detected by timeout (TCP Tahoe)

Why AIMD?

- AIMD – a distributed, asynchronous algorithm – has been shown to:
 - optimize congested flow rates network wide!
 - have desirable stability properties

TCP congestion control: details



TCP sending behavior:

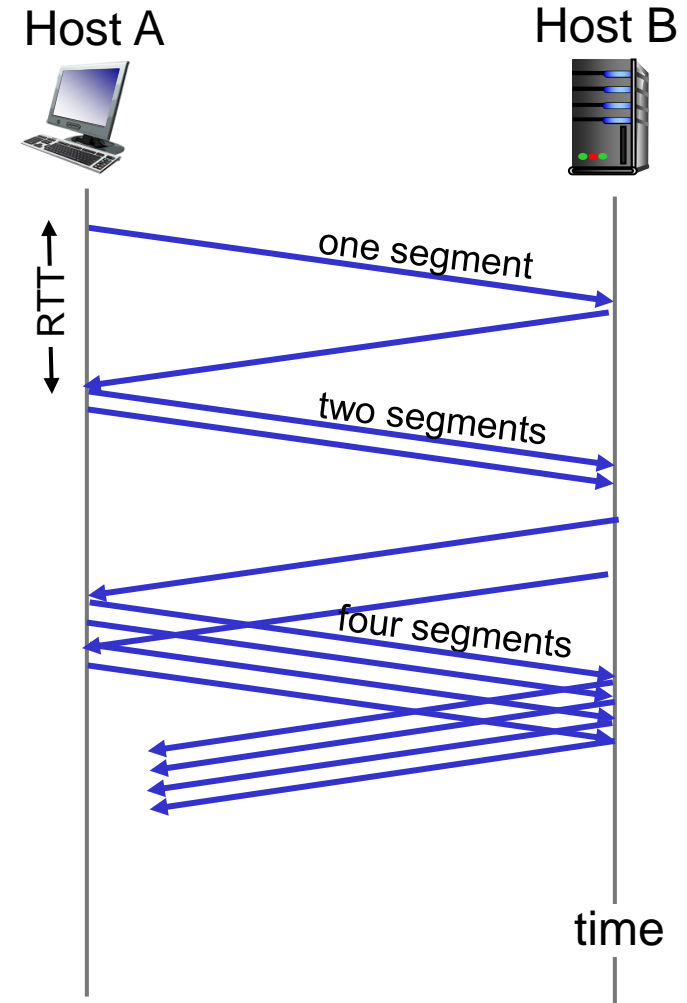
- *roughly*: send `cwnd` bytes, wait RTT for ACKS, then send more bytes

$$\text{TCP rate} \approx \frac{\text{cwnd}}{\text{RTT}} \text{ bytes/sec}$$

- TCP sender limits transmission: $\text{LastByteSent} - \text{LastByteAcked} \leq \min(\text{cwnd}, \text{rwnd})$
- `cwnd` is dynamically adjusted in response to observed network congestion (implementing TCP congestion control)

TCP slow start

- when connection begins, increase rate exponentially until first loss event:
 - initially **cwnd** = 1 MSS
 - double **cwnd** every RTT
 - done by incrementing **cwnd** for every ACK received
- *summary*: initial rate is slow, but ramps up exponentially fast



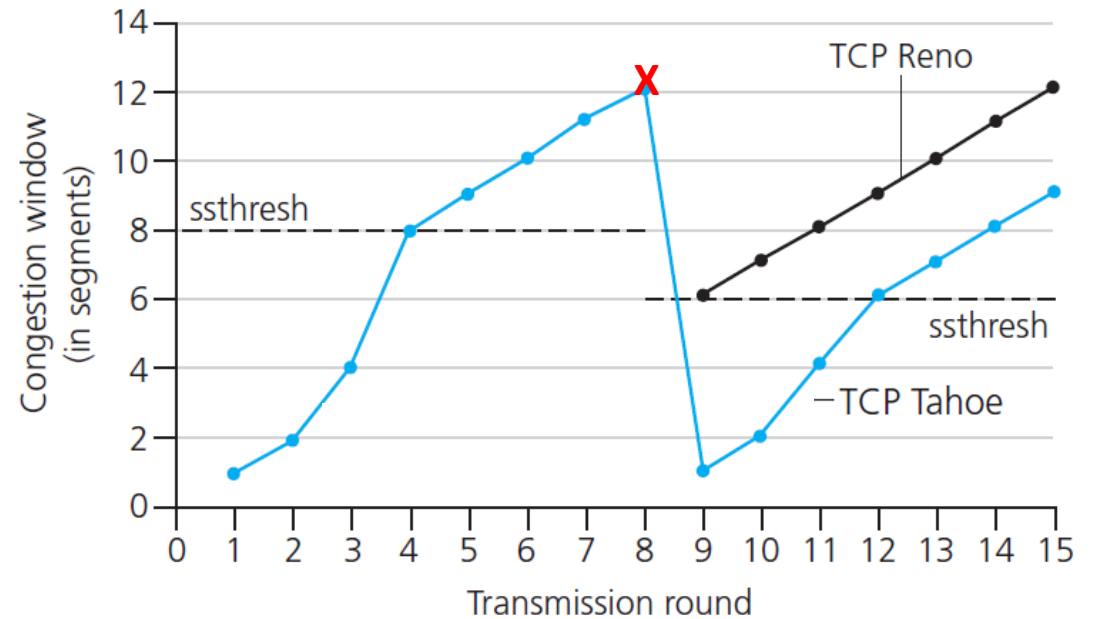
TCP: from slow start to congestion avoidance

Q: when should the exponential increase switch to linear?

A: when **cwnd** gets to 1/2 of its value before timeout.

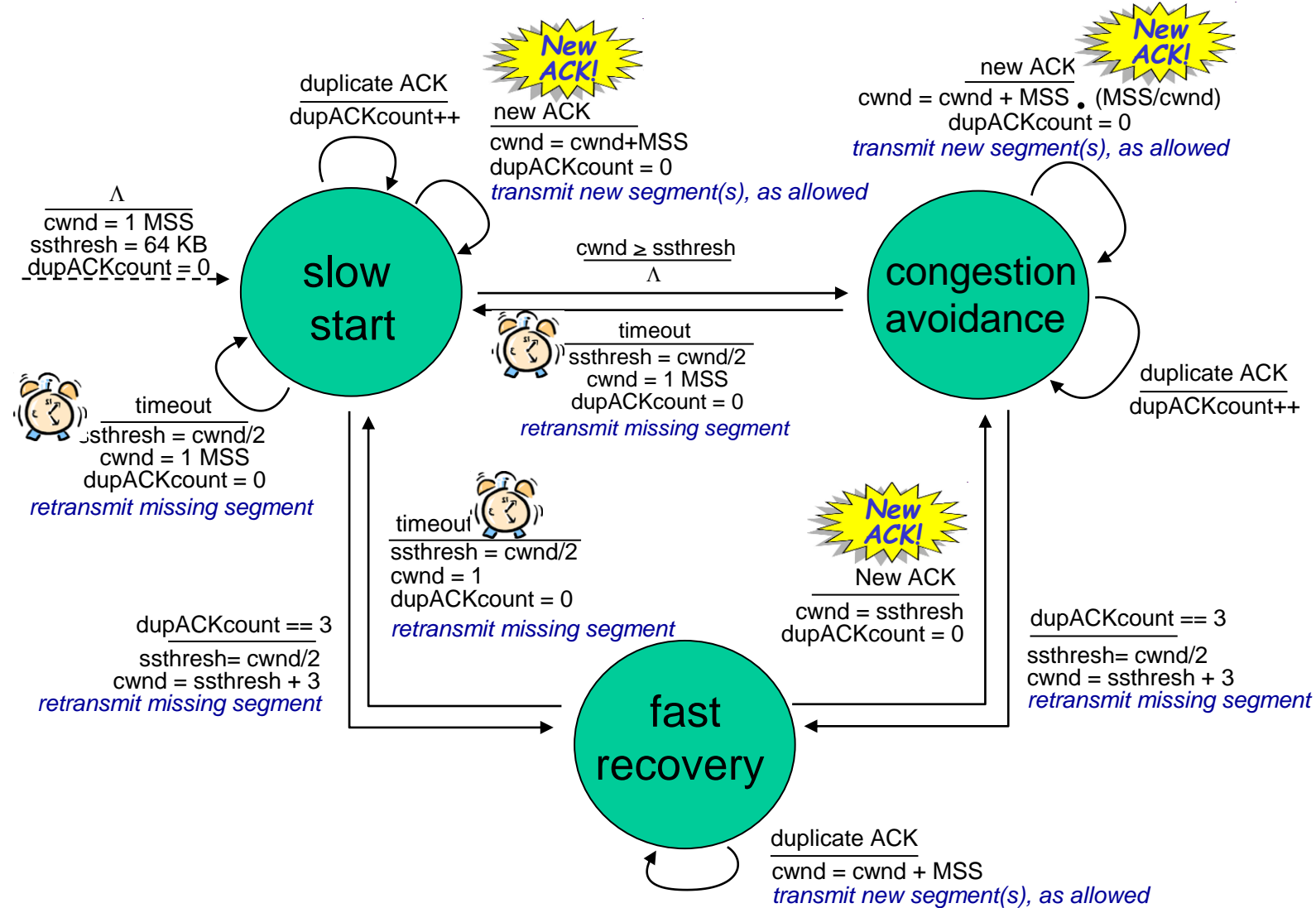
Implementation:

- variable **ssthresh**
- on loss event, **ssthresh** is set to 1/2 of **cwnd** just before loss event



* Check out the online interactive exercises for more examples: http://gaia.cs.umass.edu/kurose_ross/interactive/

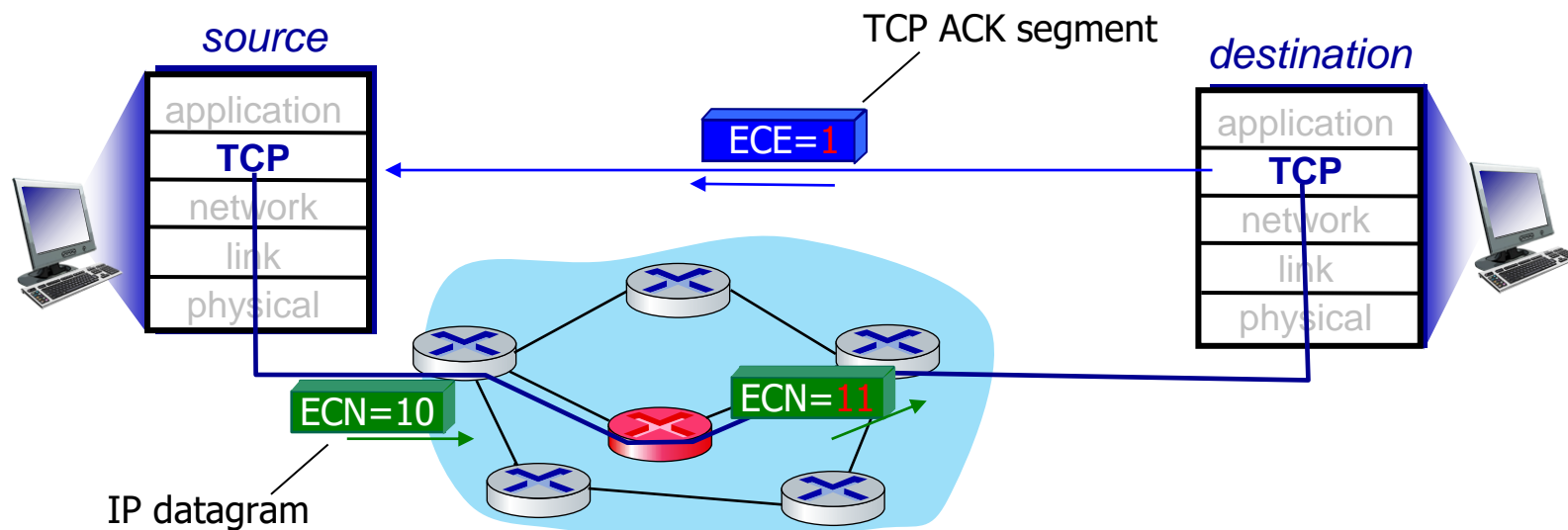
Summary: TCP congestion control



Explicit congestion notification (ECN)

TCP deployments often implement *network-assisted* congestion control:

- two bits in IP header (ToS field) marked *by network router* to indicate congestion
 - *policy* to determine marking chosen by network operator
- congestion indication carried to destination
- destination sets ECE bit on ACK segment to notify sender of congestion
- involves both IP (IP header ECN bit marking) and TCP (TCP header C,E bit marking)



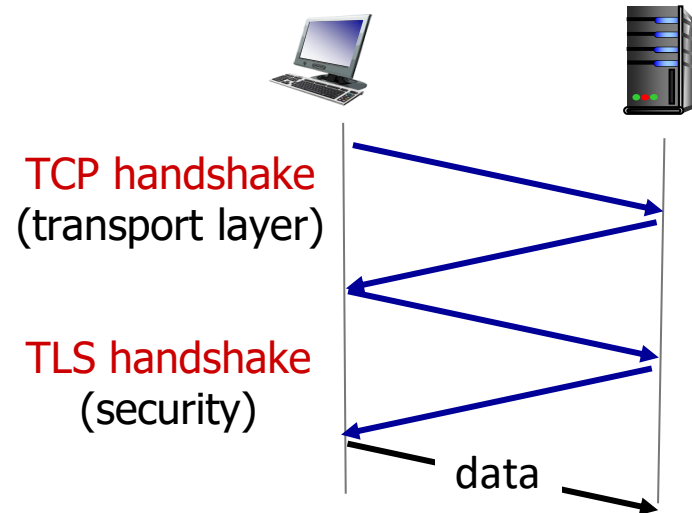
Evolving transport-layer functionality

- TCP, UDP: principal transport protocols for 40 years
- different “flavors” of TCP developed, for specific scenarios:

Scenario	Challenges
Long, fat pipes (large data transfers)	Many packets “in flight”; loss shuts down pipeline
Wireless networks	Loss due to noisy wireless links, mobility; TCP treat this as congestion loss
Long-delay links	Extremely long RTTs
Data center networks	Latency sensitive
Background traffic flows	Low priority, “background” TCP flows

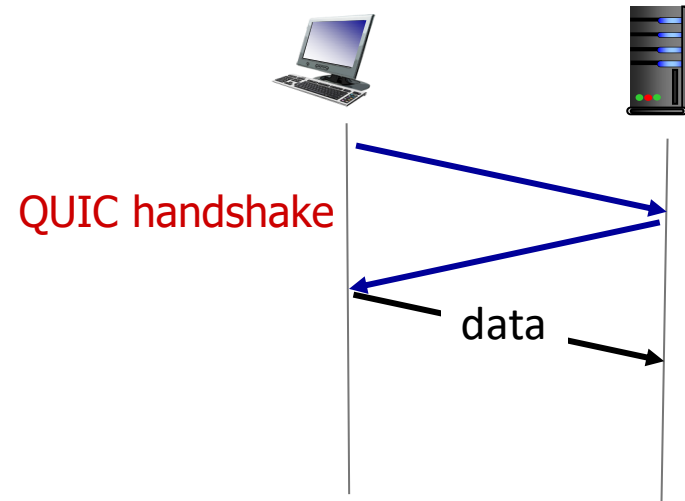
- moving transport–layer functions to application layer, on top of UDP
 - HTTP/3: QUIC

QUIC: Connection establishment



TCP (reliability, congestion control state) + TLS (authentication, crypto state)

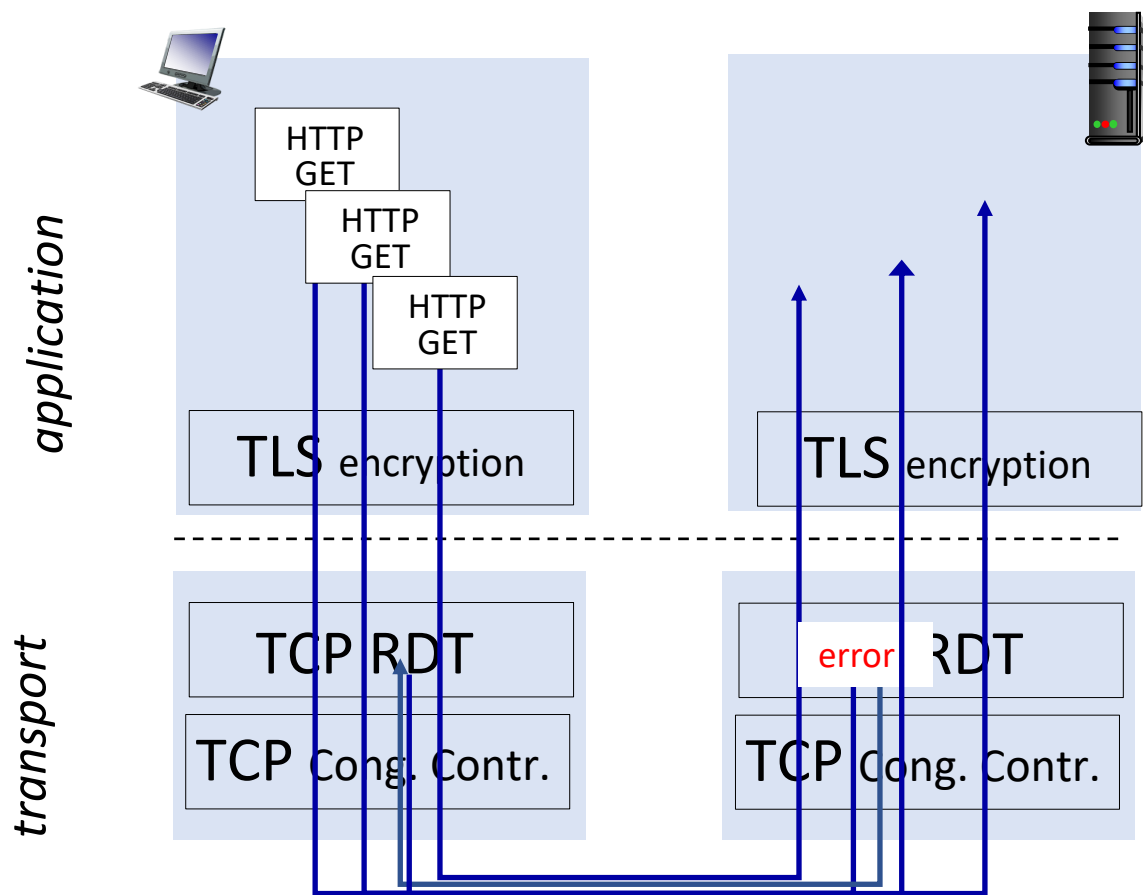
- 2 serial handshakes



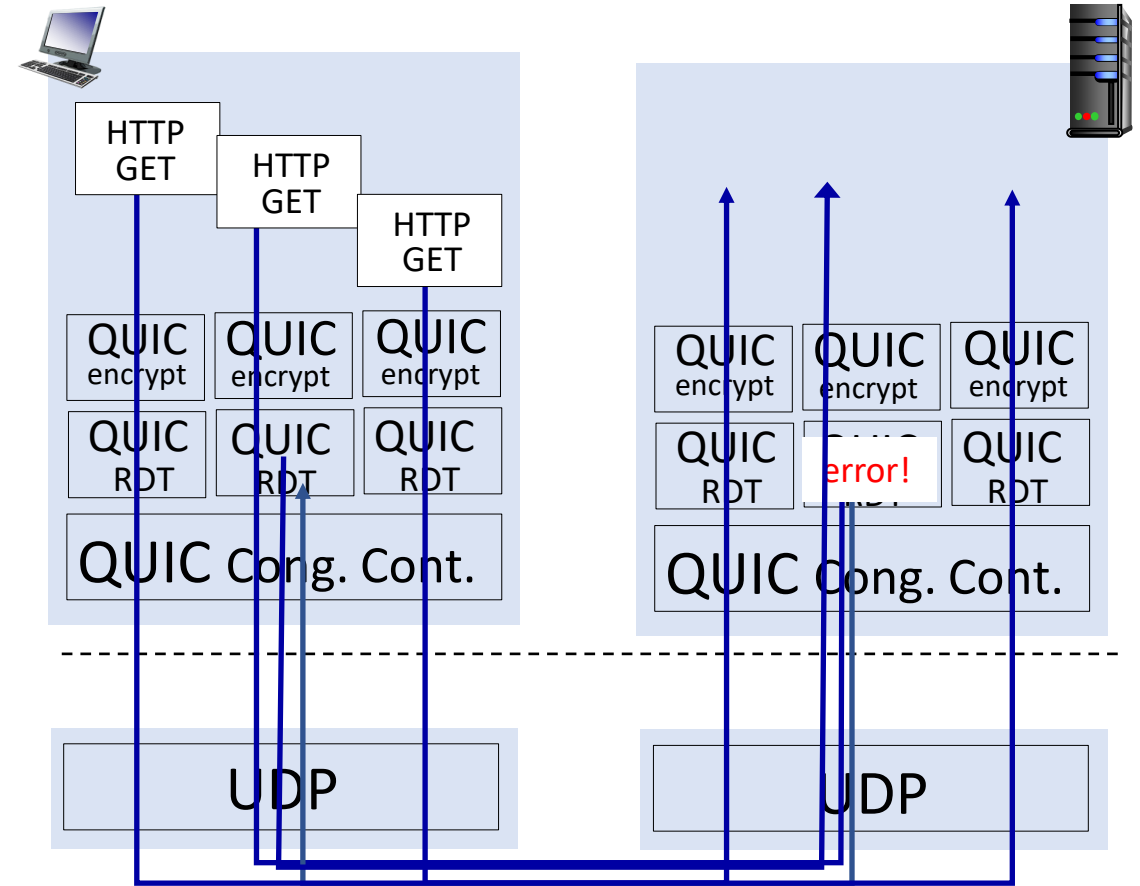
QUIC: reliability, congestion control, authentication, crypto state

- 1 handshake

QUIC: streams: parallelism, no HOL blocking



(a) HTTP 1.1



(b) HTTP/2 with QUIC: no HOL blocking