

Chapter 3

Transport Layer

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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

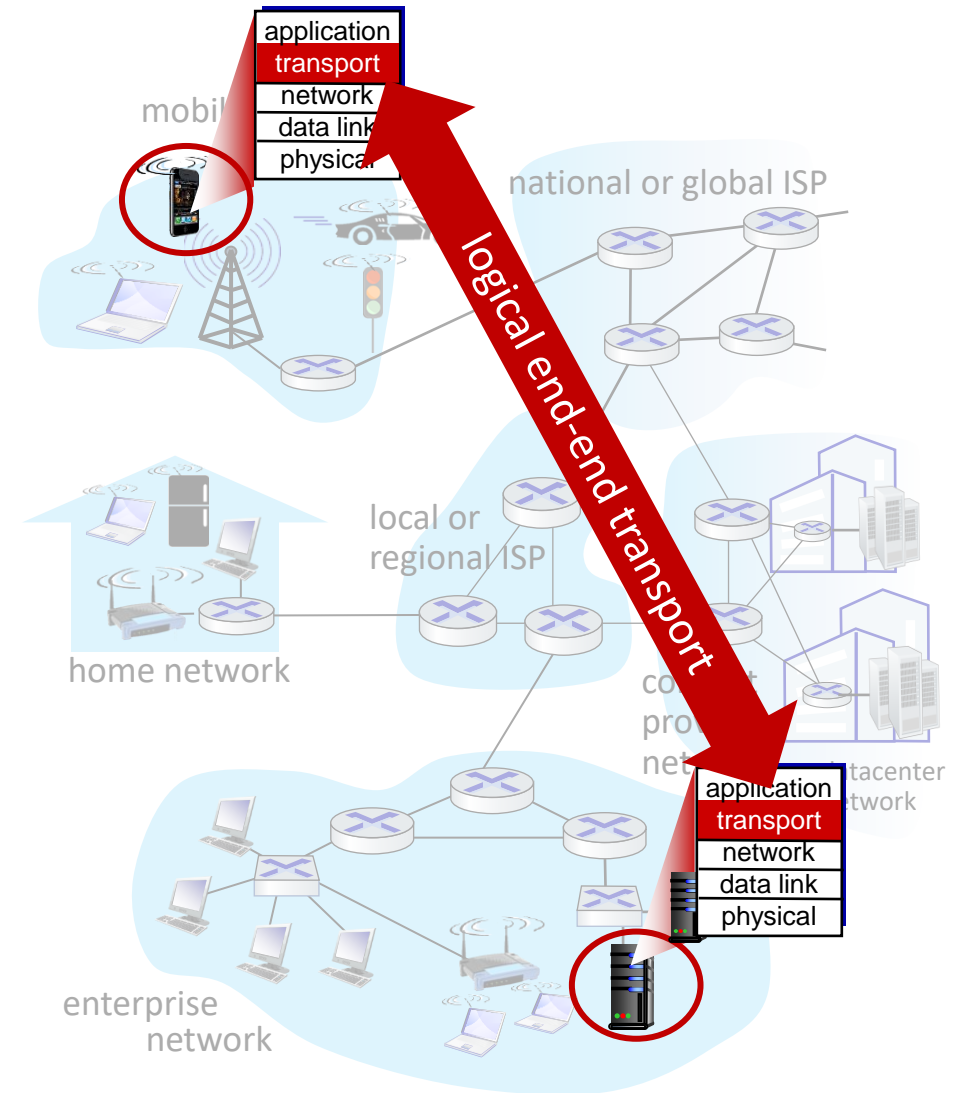
Transport layer: roadmap

- Transport-layer services
- Multiplexing and demultiplexing
- Connectionless transport: UDP
- Connection-oriented transport: TCP
- TCP congestion control



Transport services and protocols

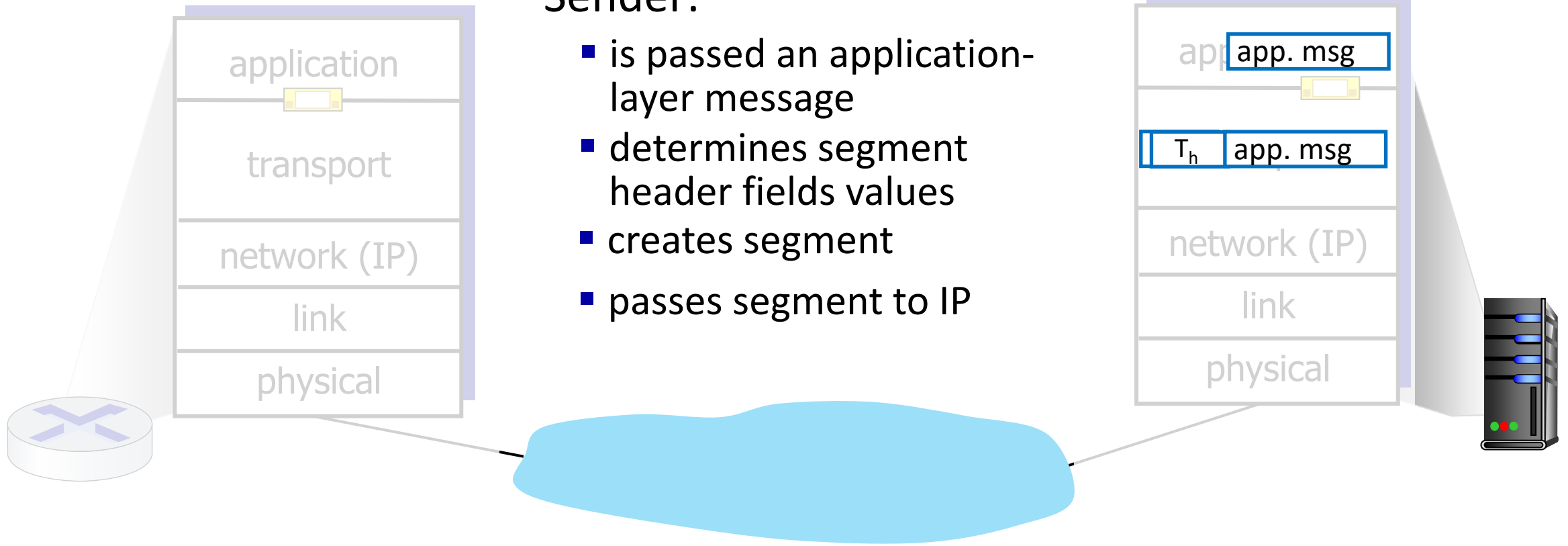
- provide *logical communication* between application processes running on different hosts
- transport protocols actions in end systems:
 - sender: breaks application messages into *segments*, passes to network layer
 - receiver: reassembles segments into messages, passes to application layer
- two transport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



Transport Layer Actions

Sender:

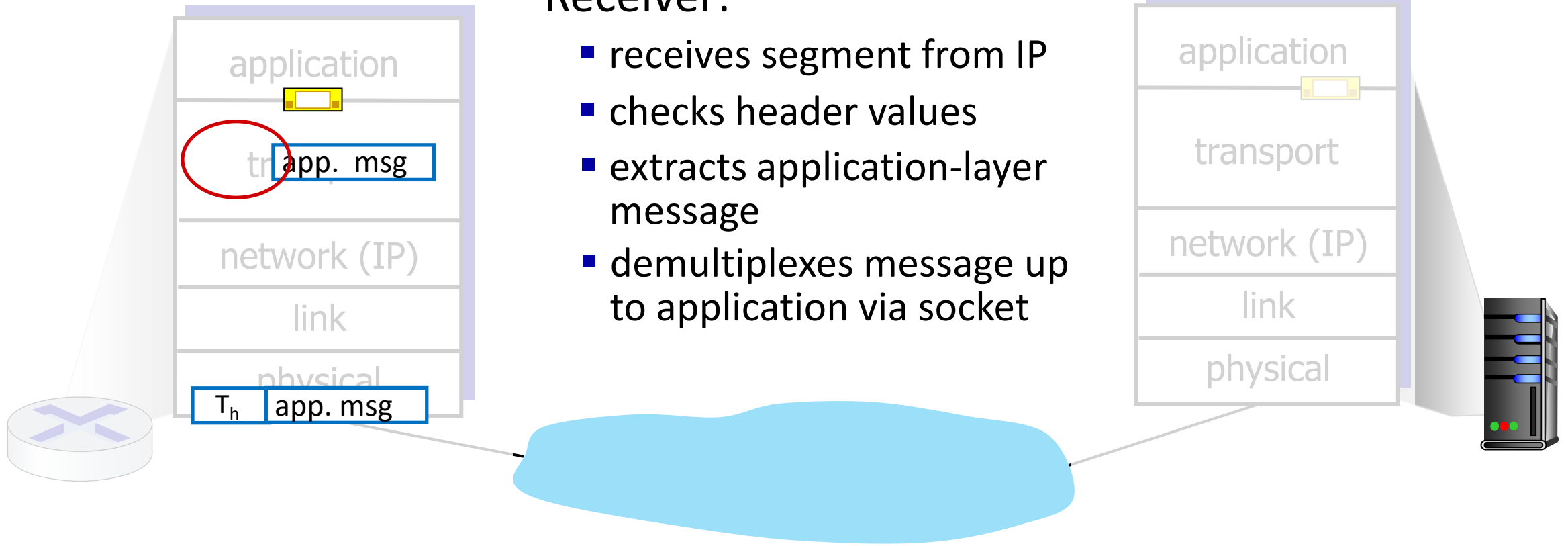
- is passed an application-layer message
- determines segment header fields values
- creates segment
- passes segment to IP



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

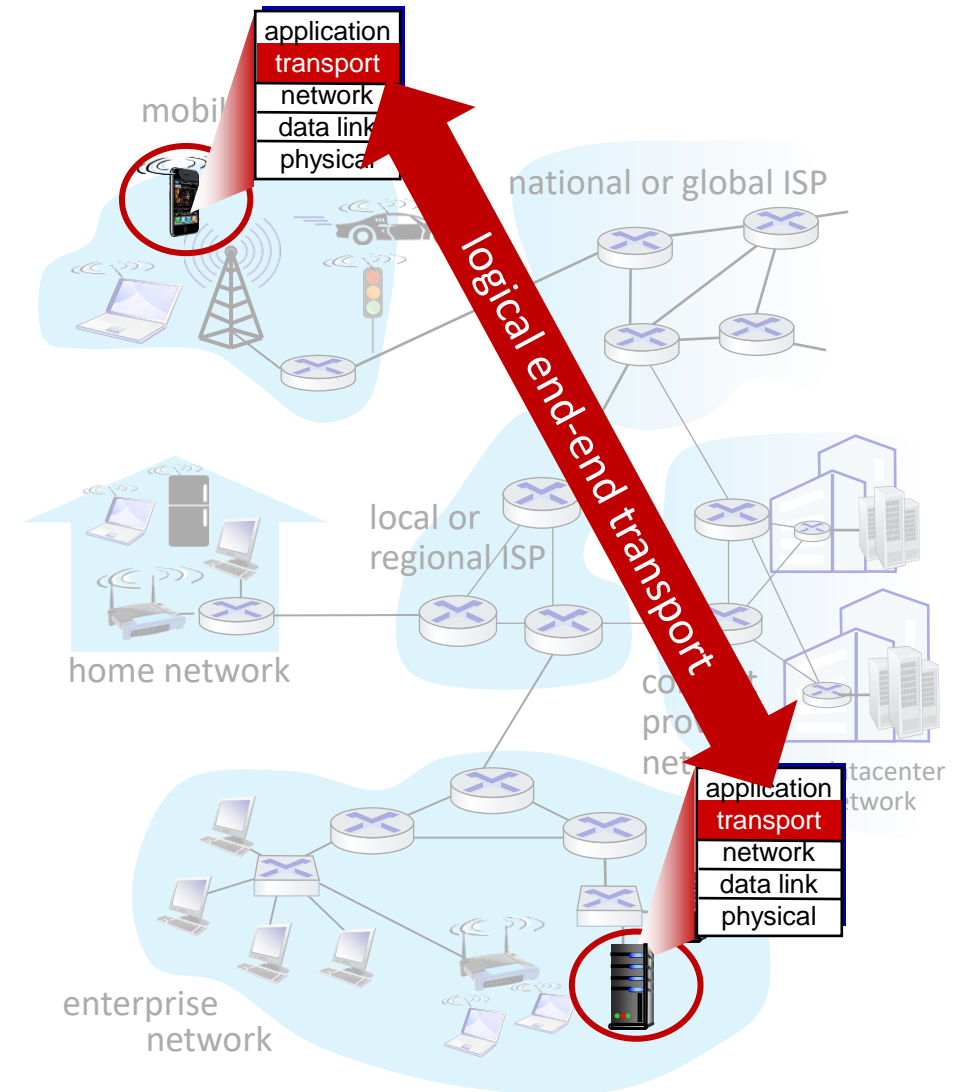
- reliable, in-order delivery
- congestion control
- flow control
- connection setup

- **UDP:** User Datagram Protocol

- unreliable, unordered delivery
- no-frills extension of “best-effort” IP

- services not available:

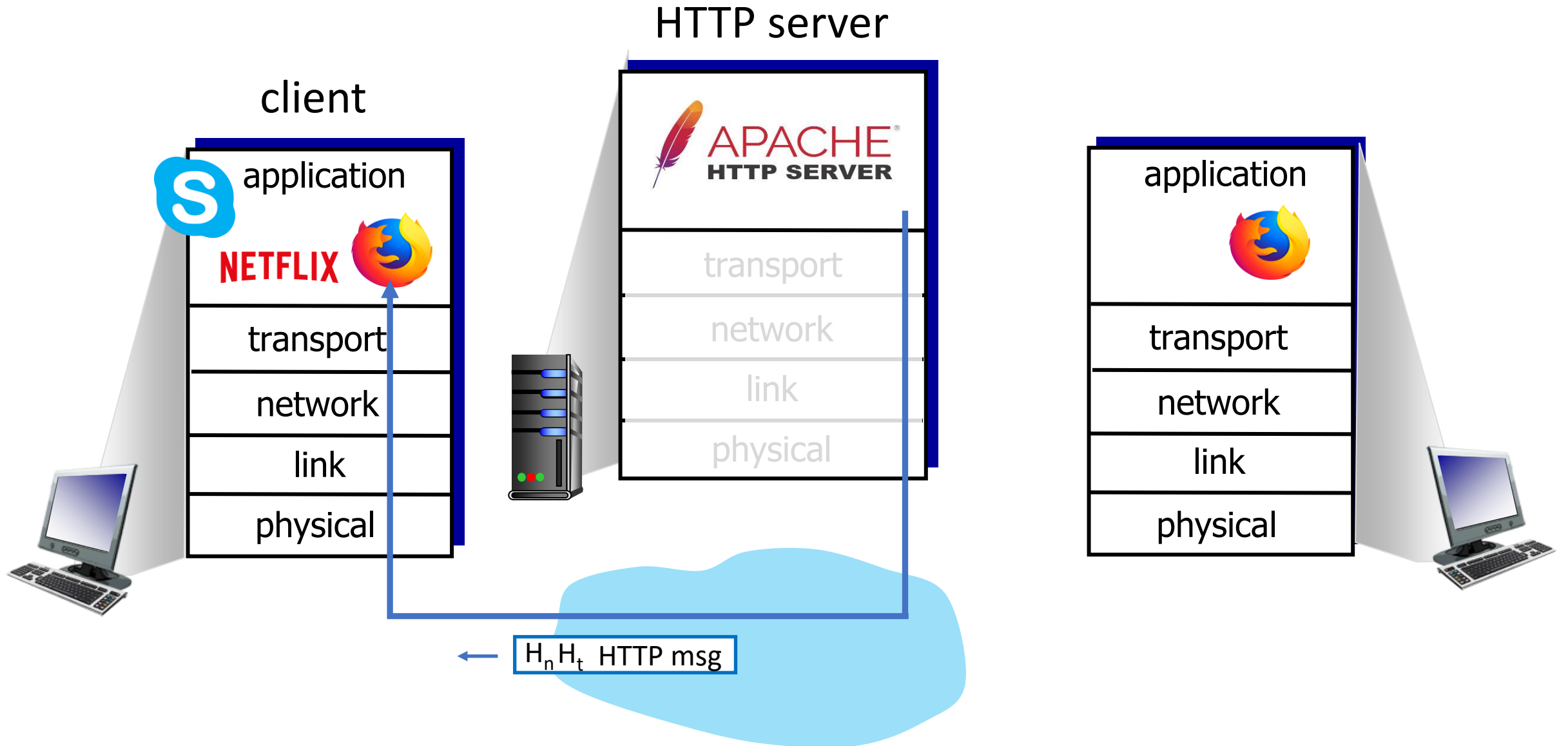
- delay guarantees
- bandwidth guarantees

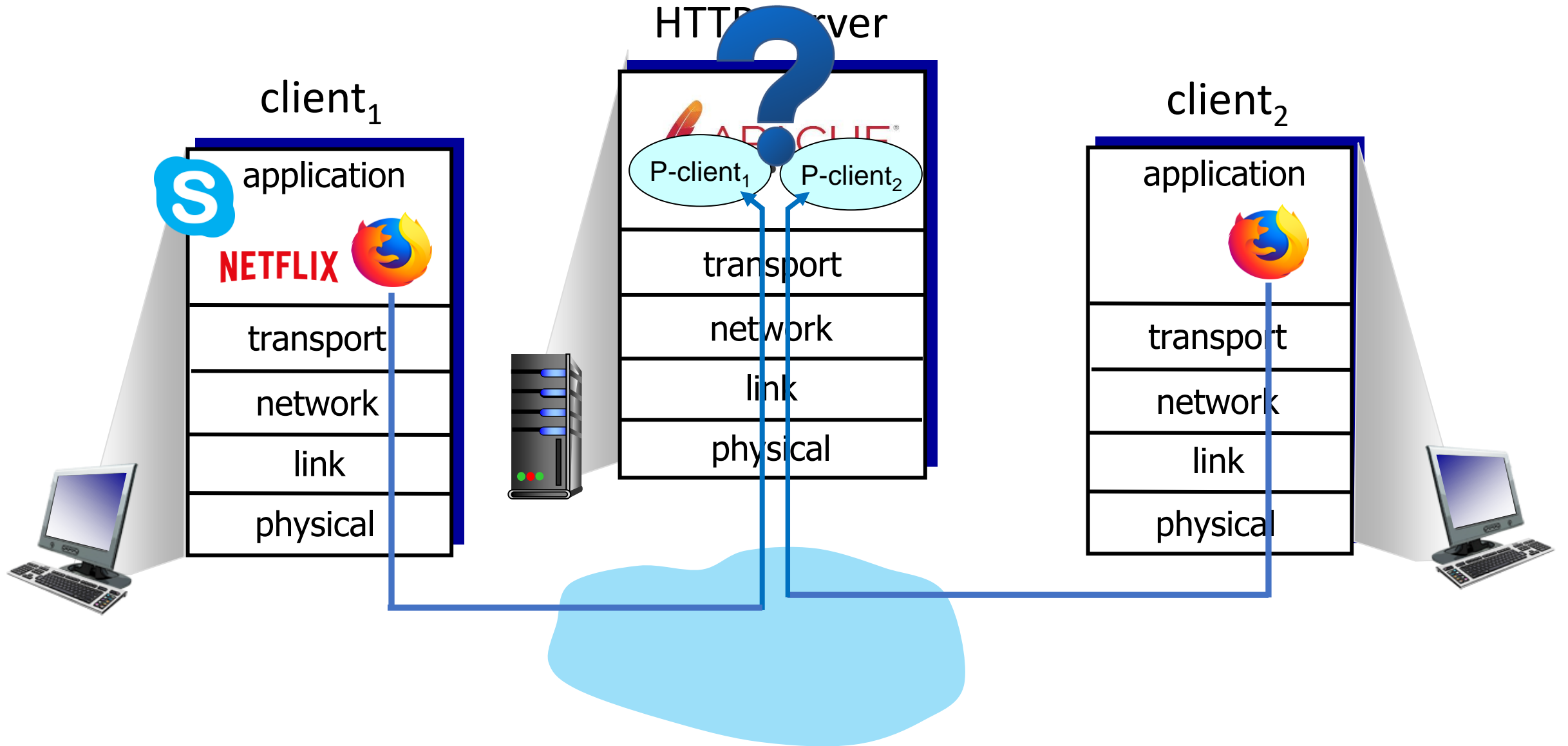


Chapter 3: roadmap

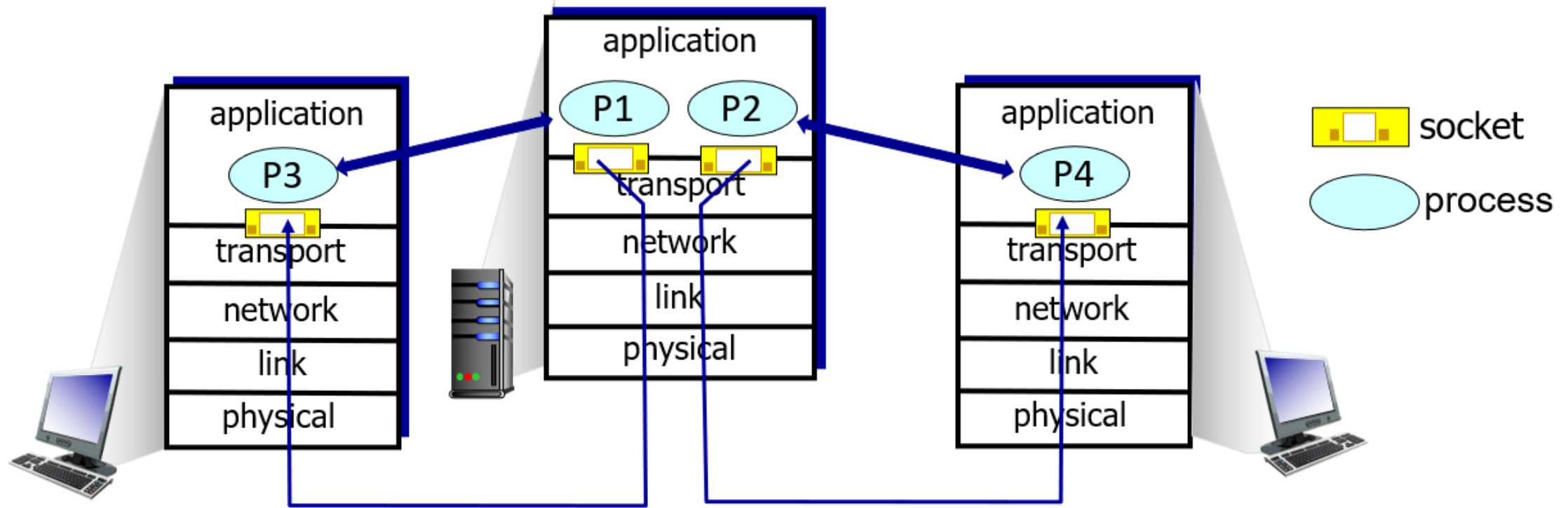
- Transport-layer services
- **Multiplexing and demultiplexing**
- Connectionless transport: UDP
- Connection-oriented transport: TCP
- TCP congestion control







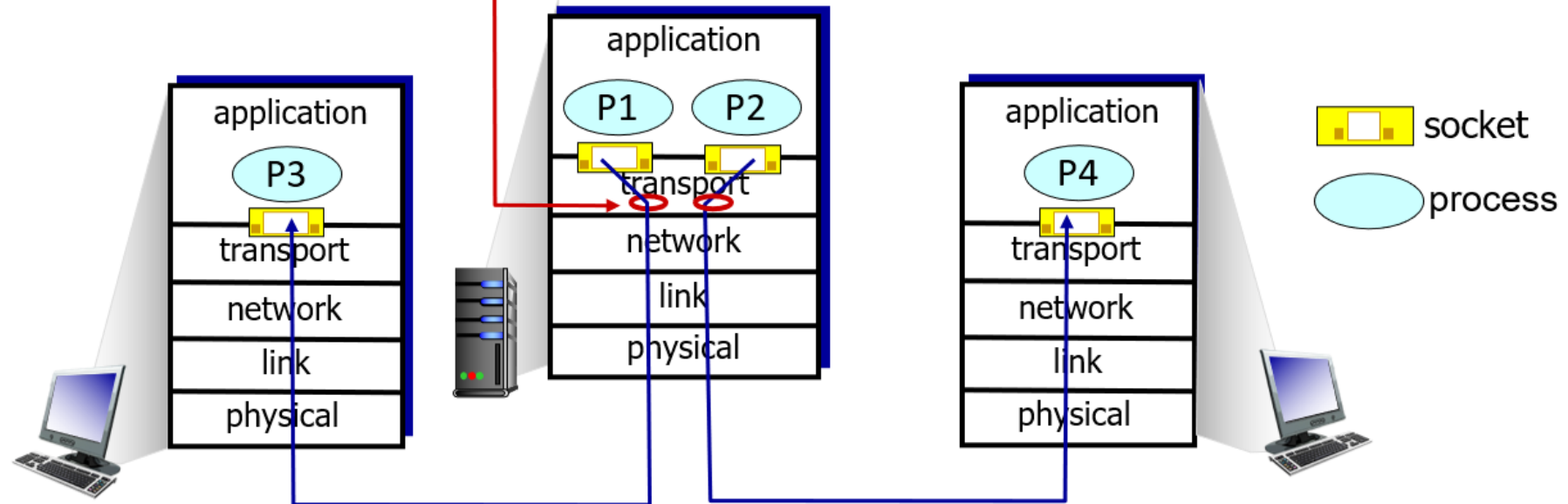
Multiplexing/demultiplexing



Multiplexing/demultiplexing

multiplexing at sender:

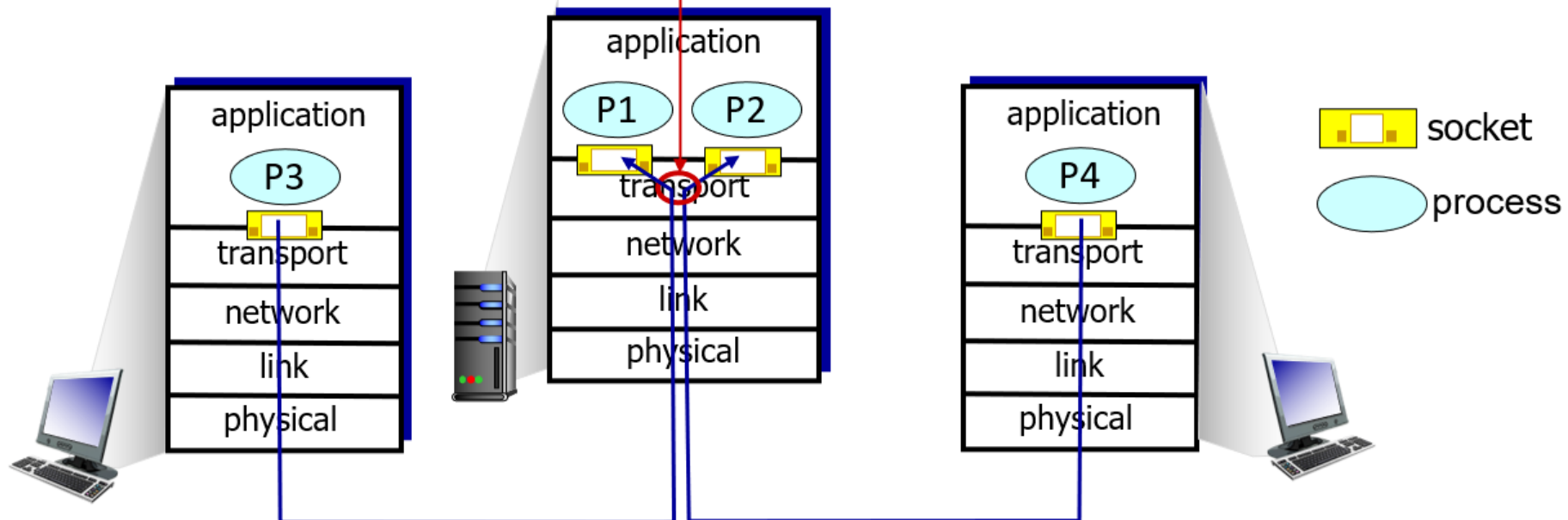
handle data from multiple sockets, add transport header (later used for demultiplexing)



Multiplexing/demultiplexing

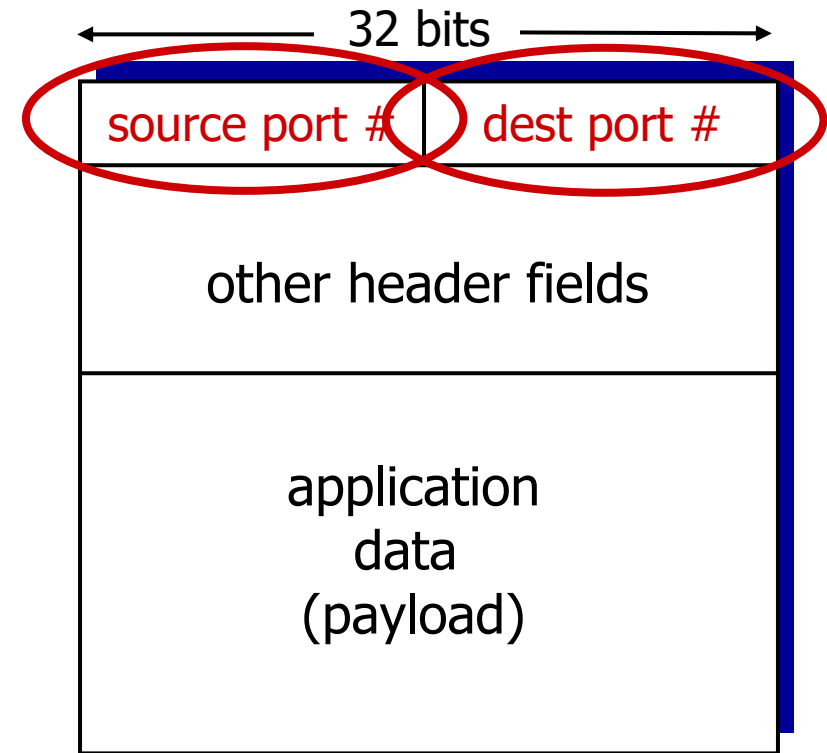
demultiplexing at receiver:

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 #



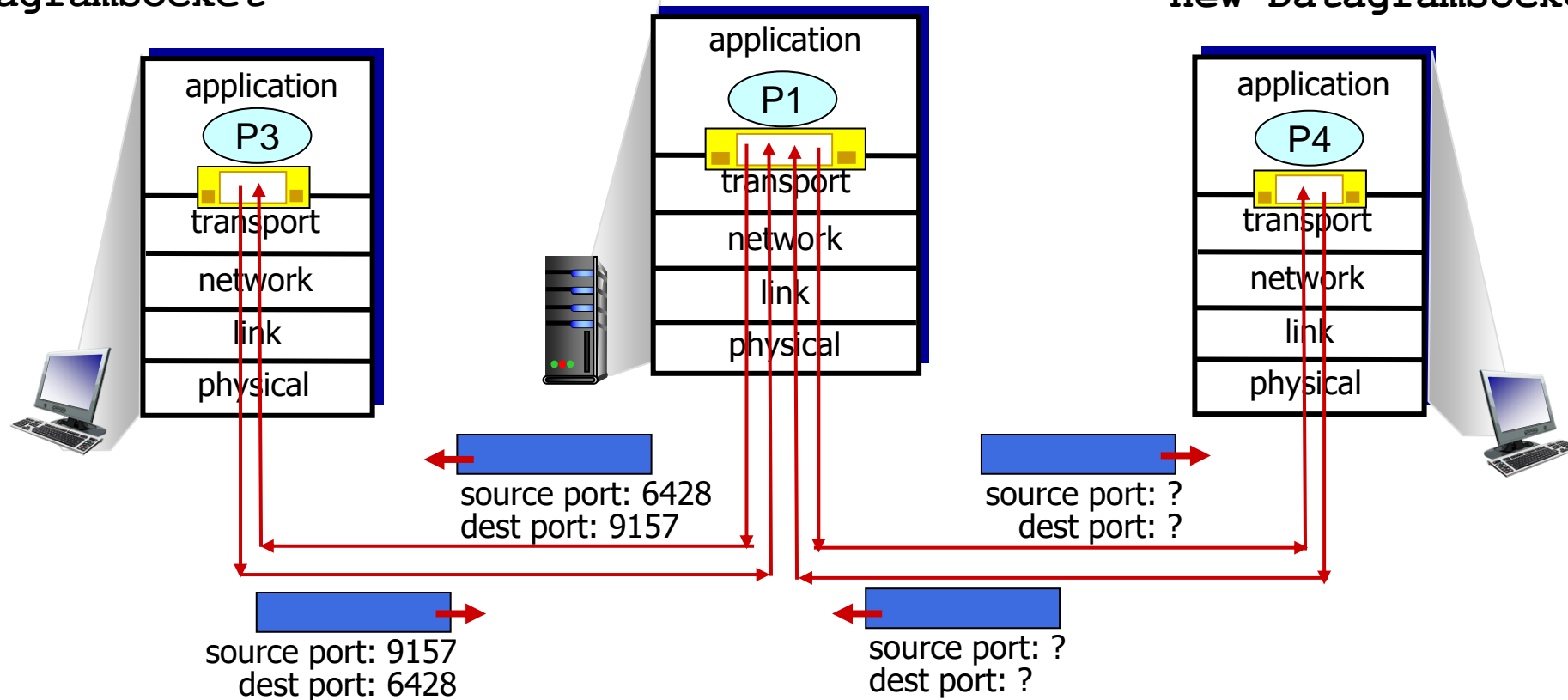
IP/UDP datagrams with *same dest. port #*, but different source IP addresses and/or source port numbers will be directed to *same socket* at receiving host

Connectionless demultiplexing: an example

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

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

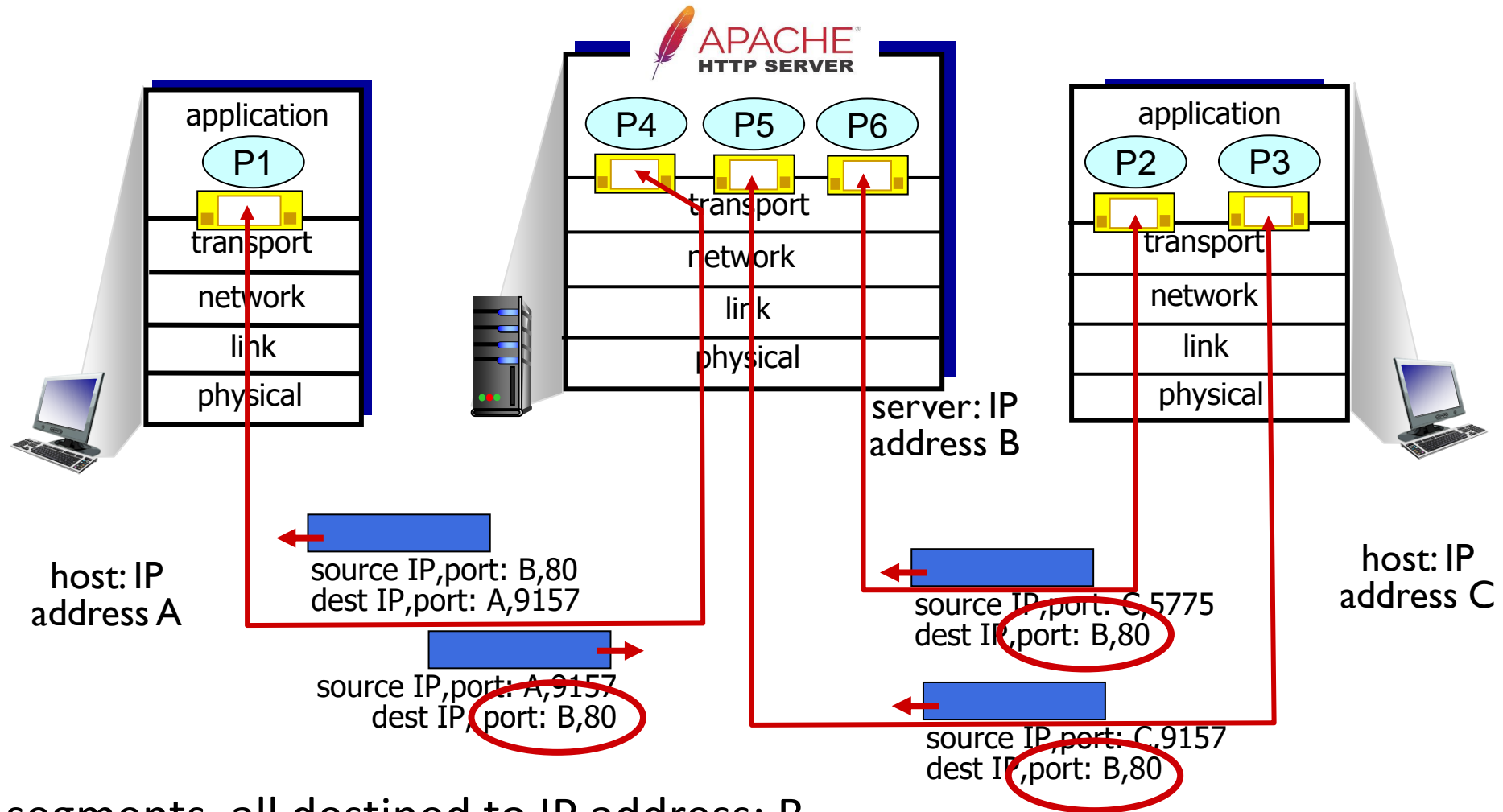
```
DatagramSocket mySocket1 =  
new DatagramSocket (5775) ;
```



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

Chapter 3: roadmap

- Transport-layer services
- Multiplexing and demultiplexing
- **Connectionless transport: UDP**
- Connection-oriented transport: TCP
- TCP congestion control



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

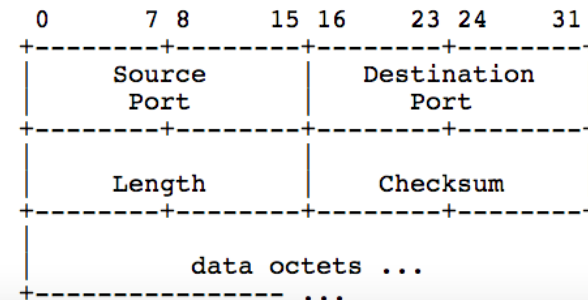
User Datagram Protocol

Introduction

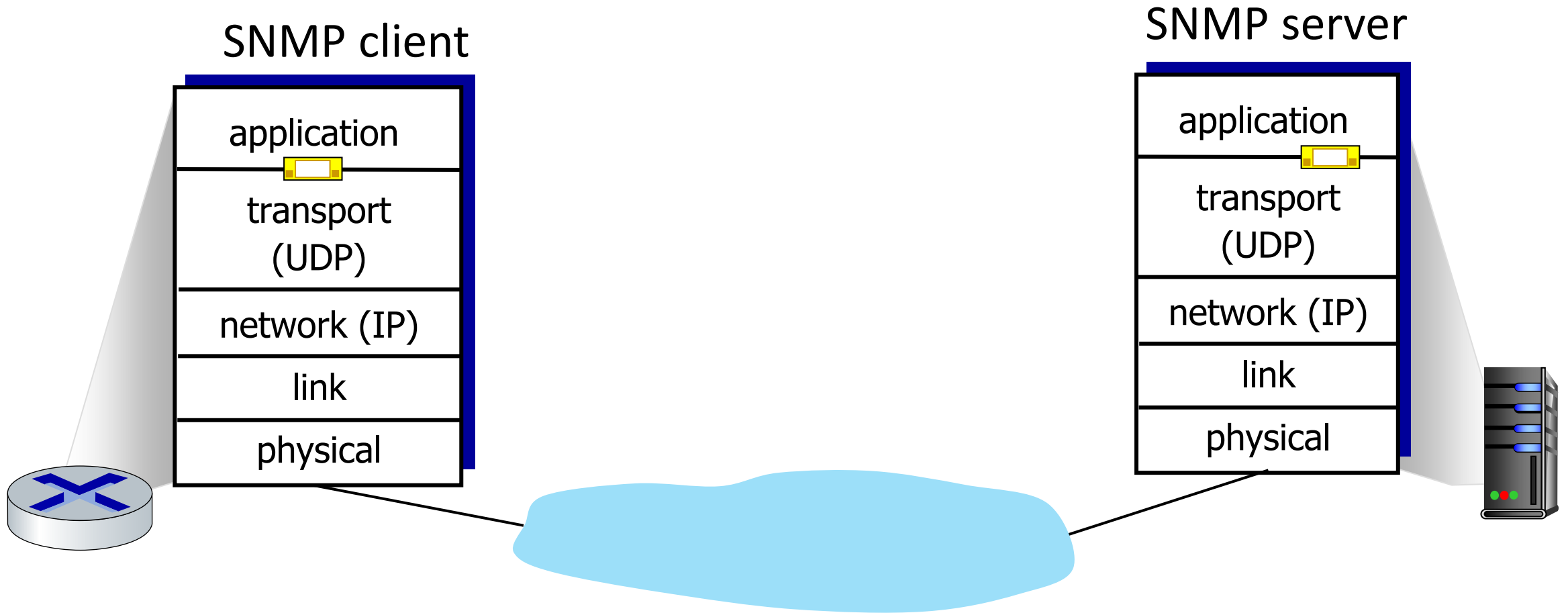
This User Datagram Protocol (UDP) is defined to make available a datagram mode of packet-switched computer communication in the environment of an interconnected set of computer networks. This protocol assumes that the Internet Protocol (IP) [1] is used as the underlying protocol.

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].

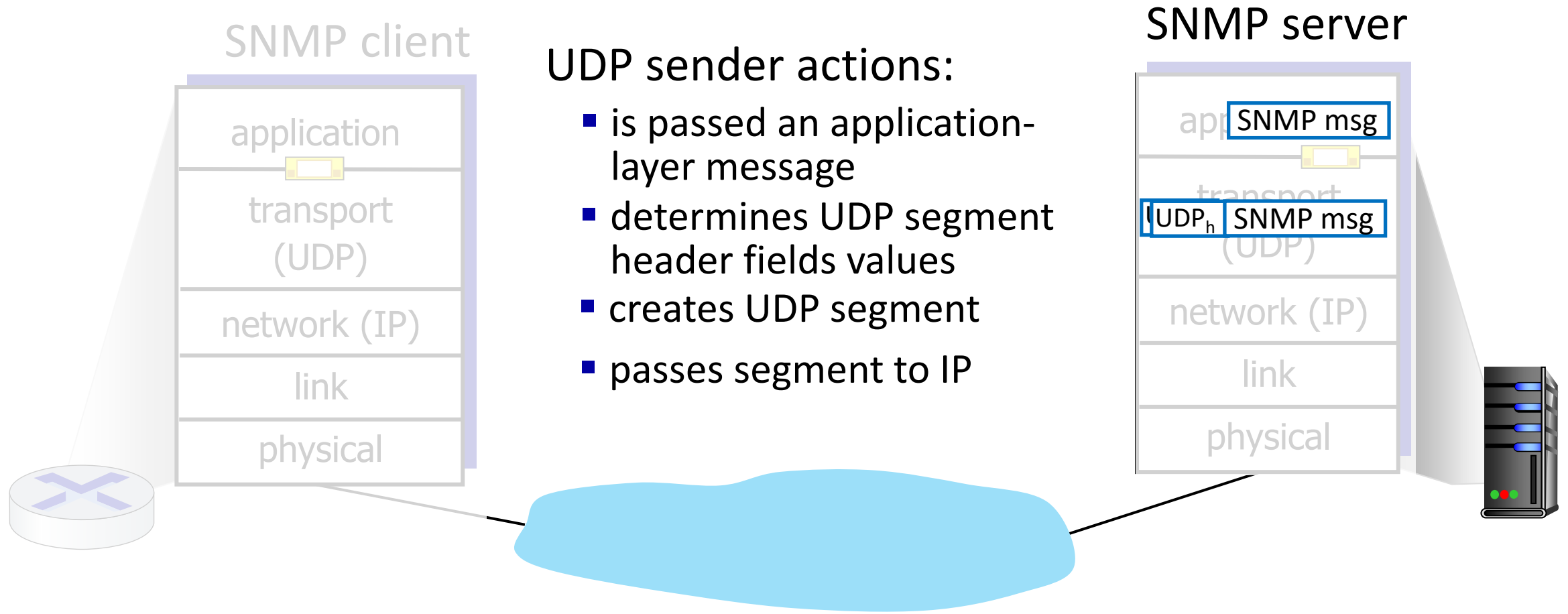
Format



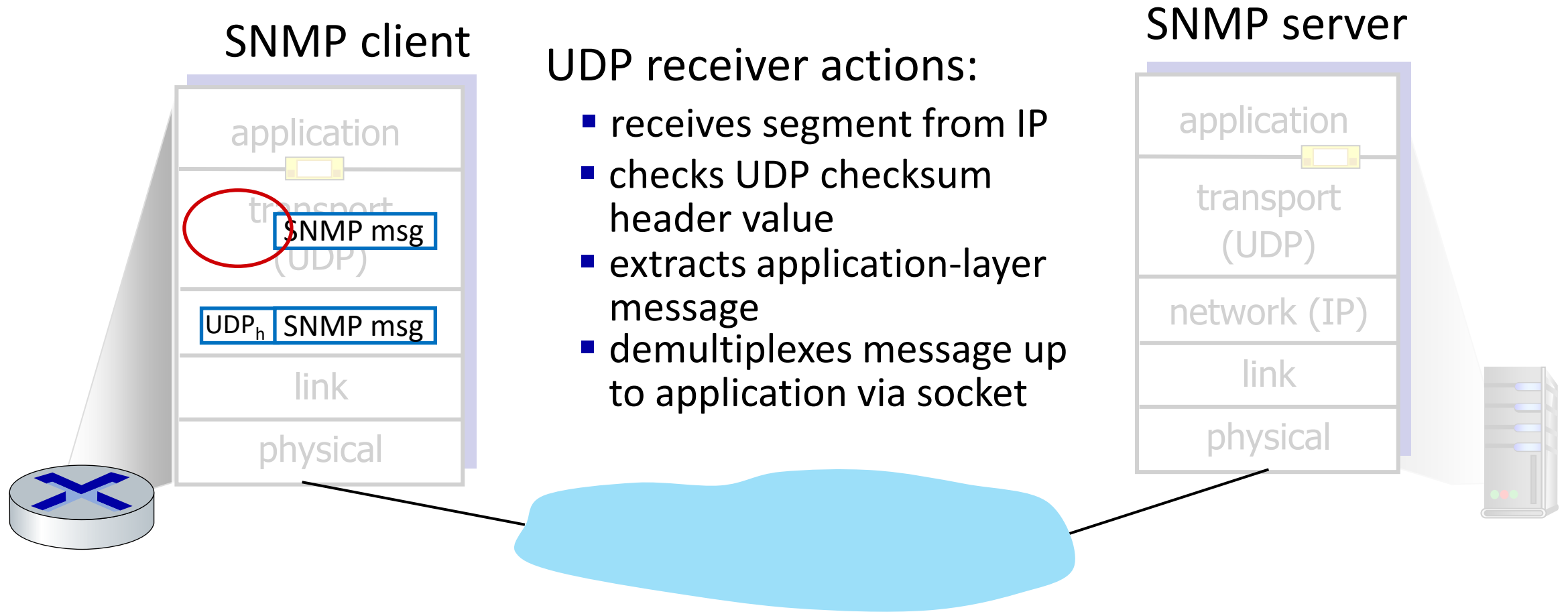
UDP: Transport Layer Actions



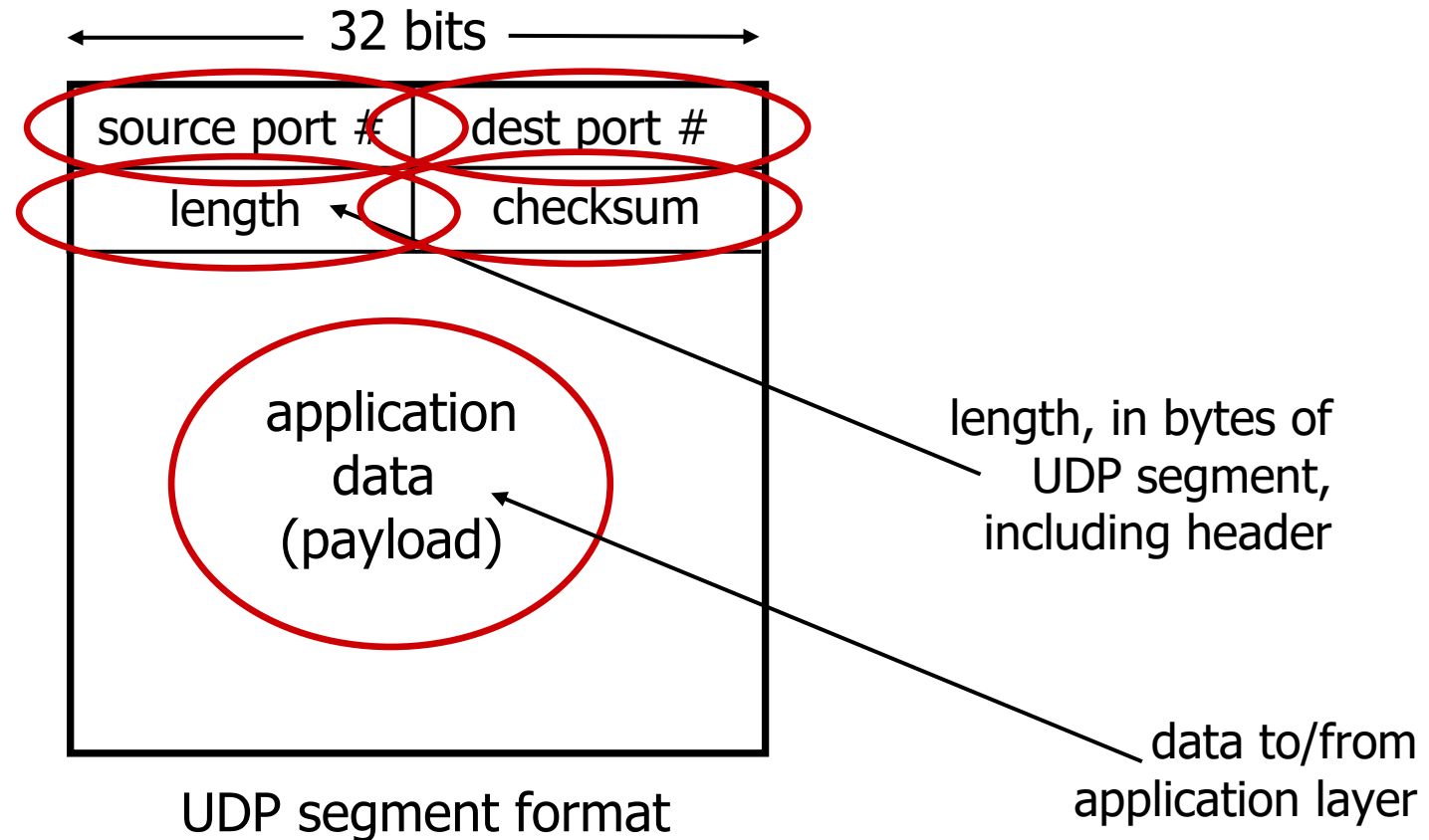
UDP: Transport Layer Actions



UDP: Transport Layer Actions



UDP segment header



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)

Chapter 3: roadmap

- Transport-layer services
- Multiplexing and demultiplexing
- Connectionless transport: UDP
- **Connection-oriented transport: TCP**
 - segment structure
 - reliable data transfer
 - flow control
 - connection management
- TCP congestion control

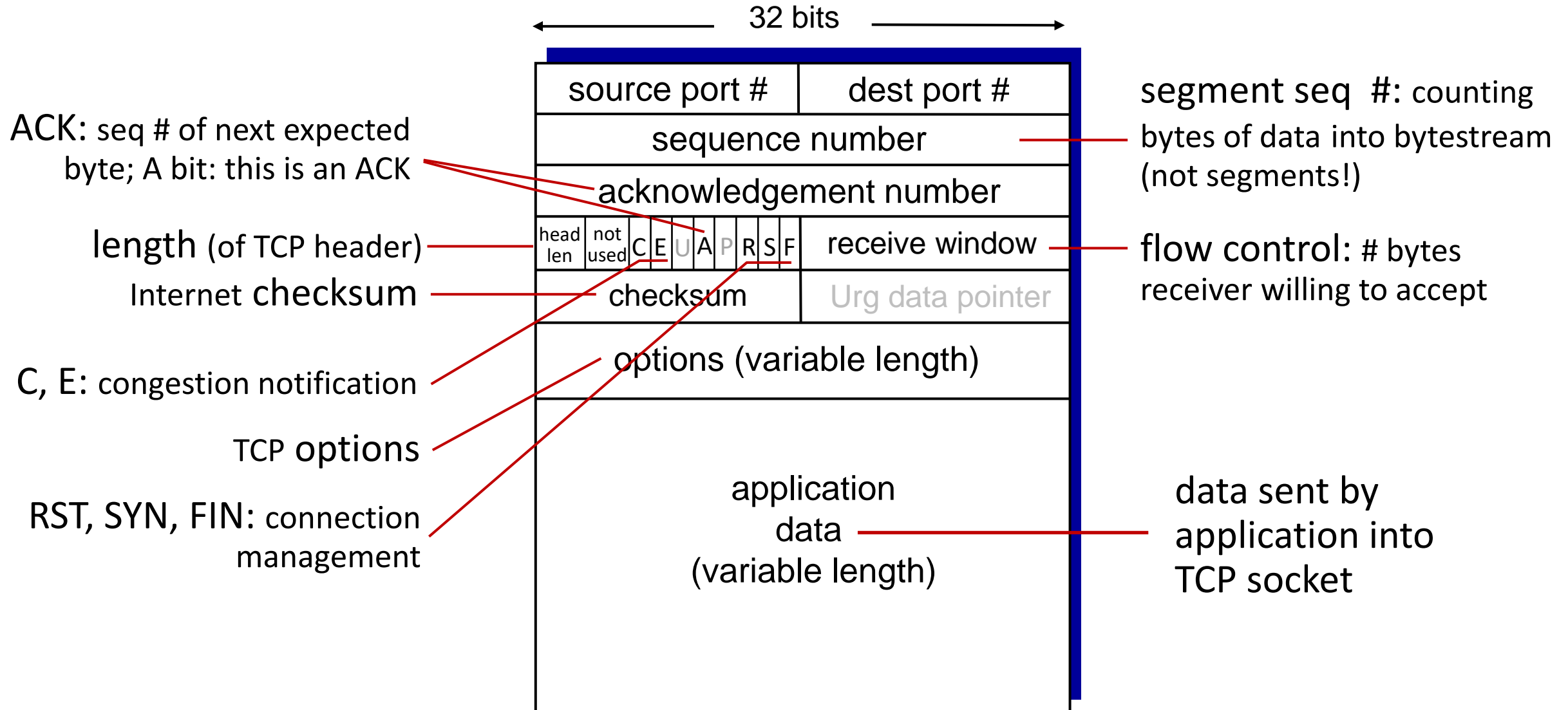


TCP: overview

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:

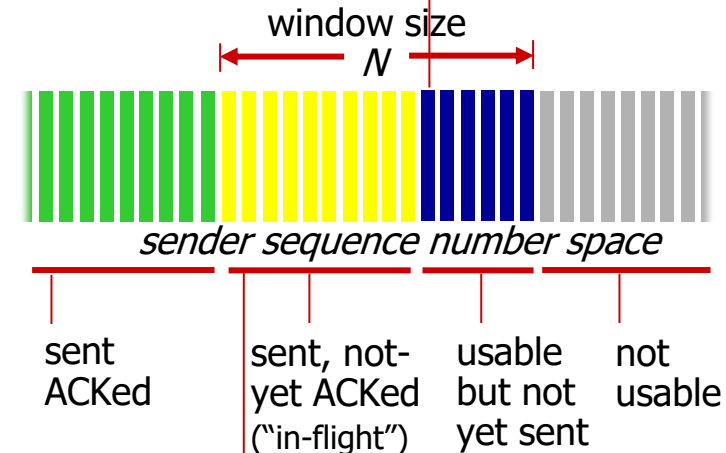
- seq # of next byte expected from other side
- cumulative ACK

Q: how receiver handles out-of-order segments

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

outgoing segment from sender

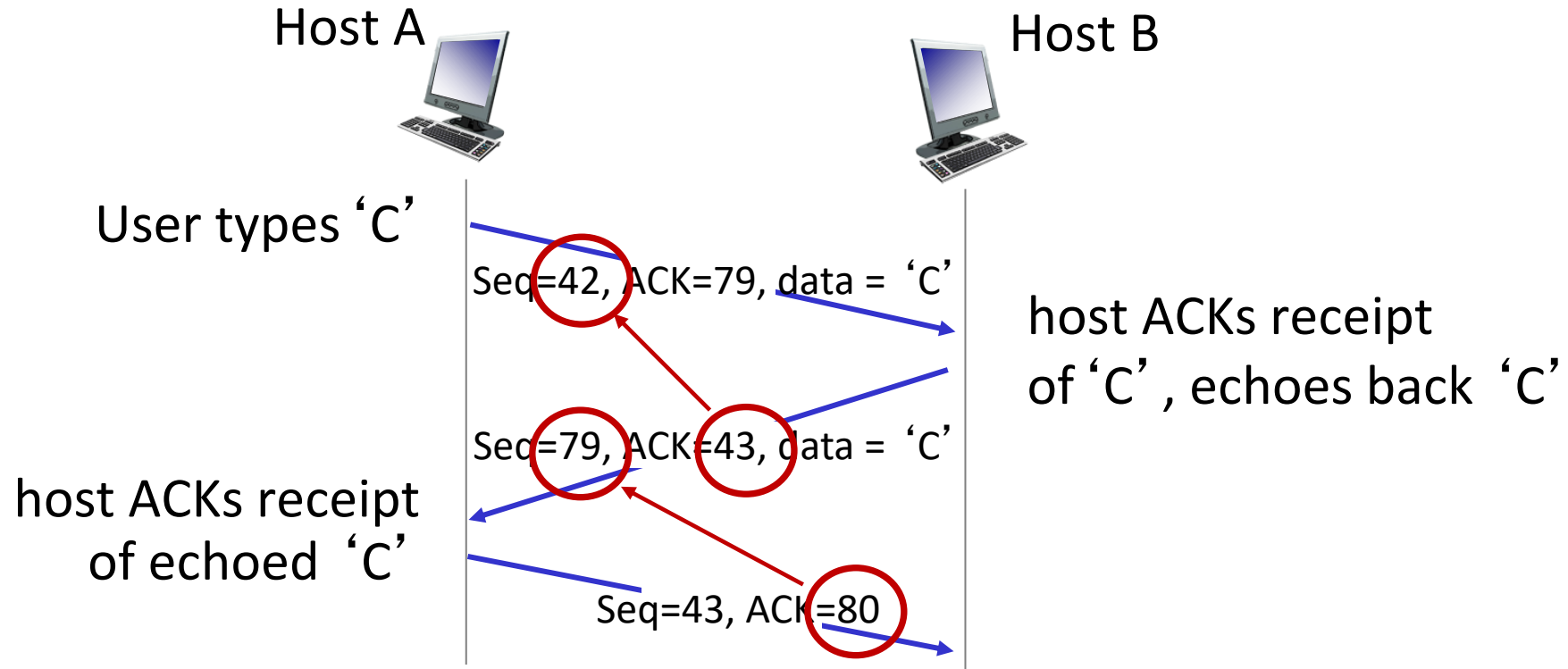
source port #	dest port #
sequence number	
acknowledgement number	
	rwnd
checksum	urg pointer



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 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**

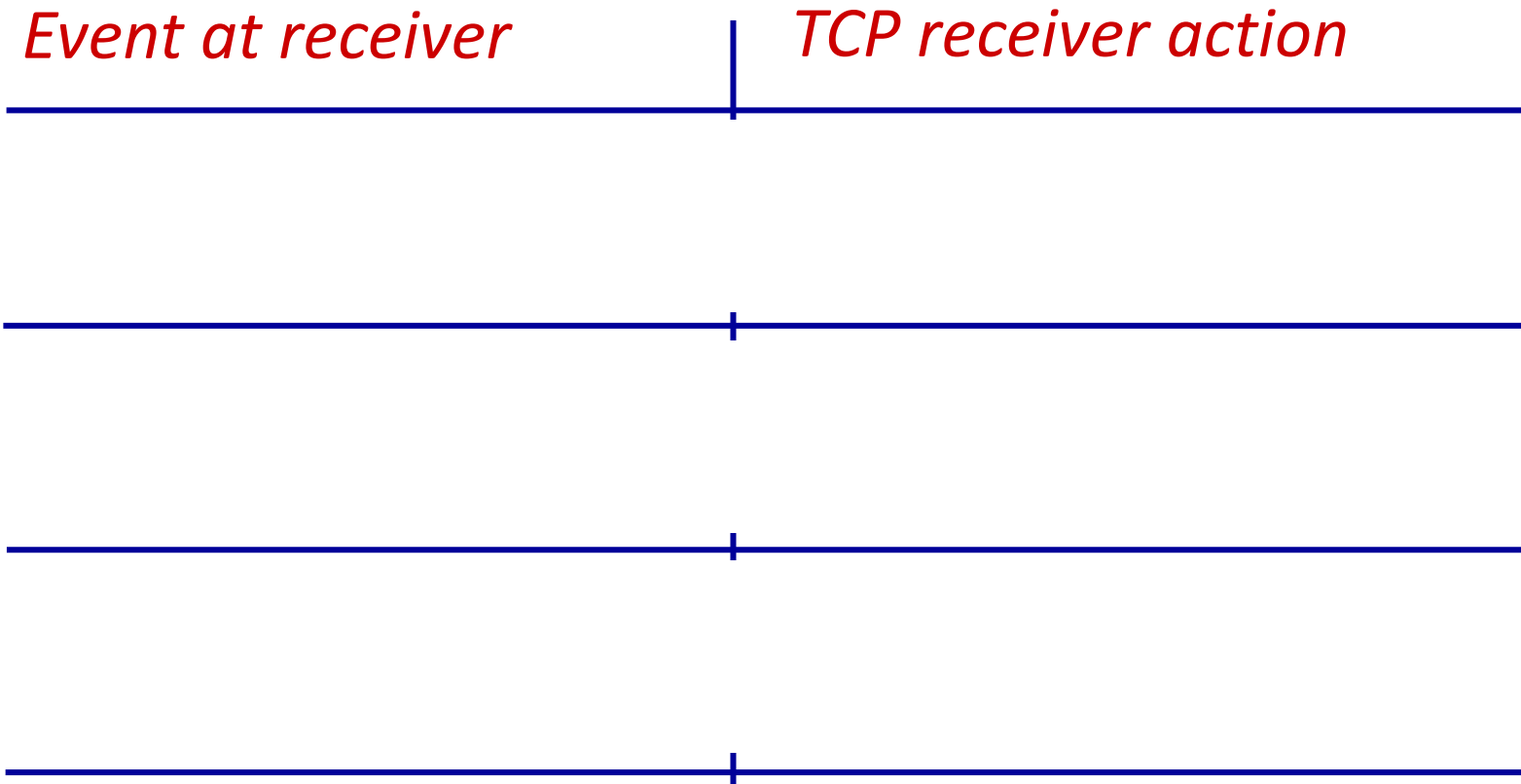
event: *timeout*

- retransmit segment that caused timeout
- restart timer

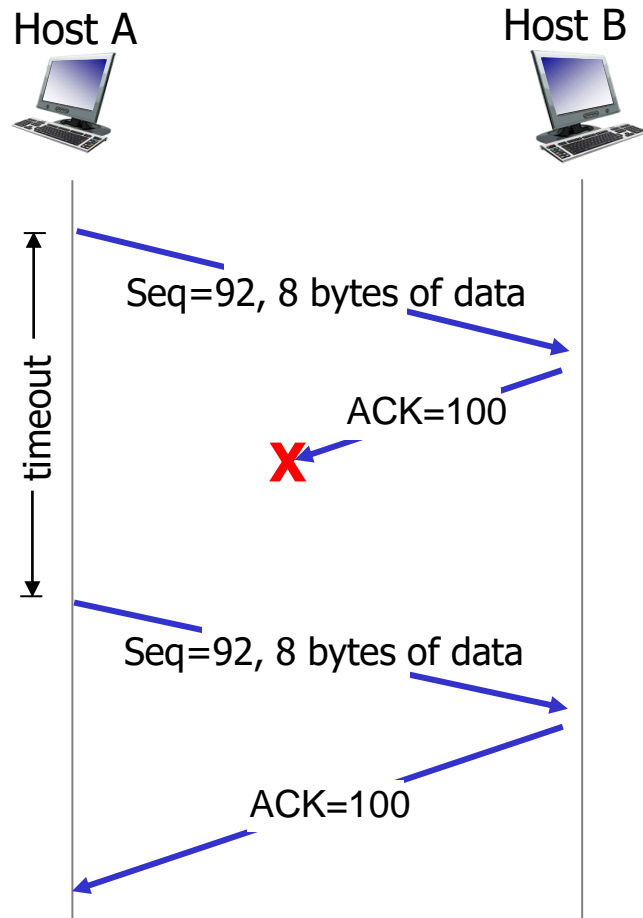
event: *ACK received*

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

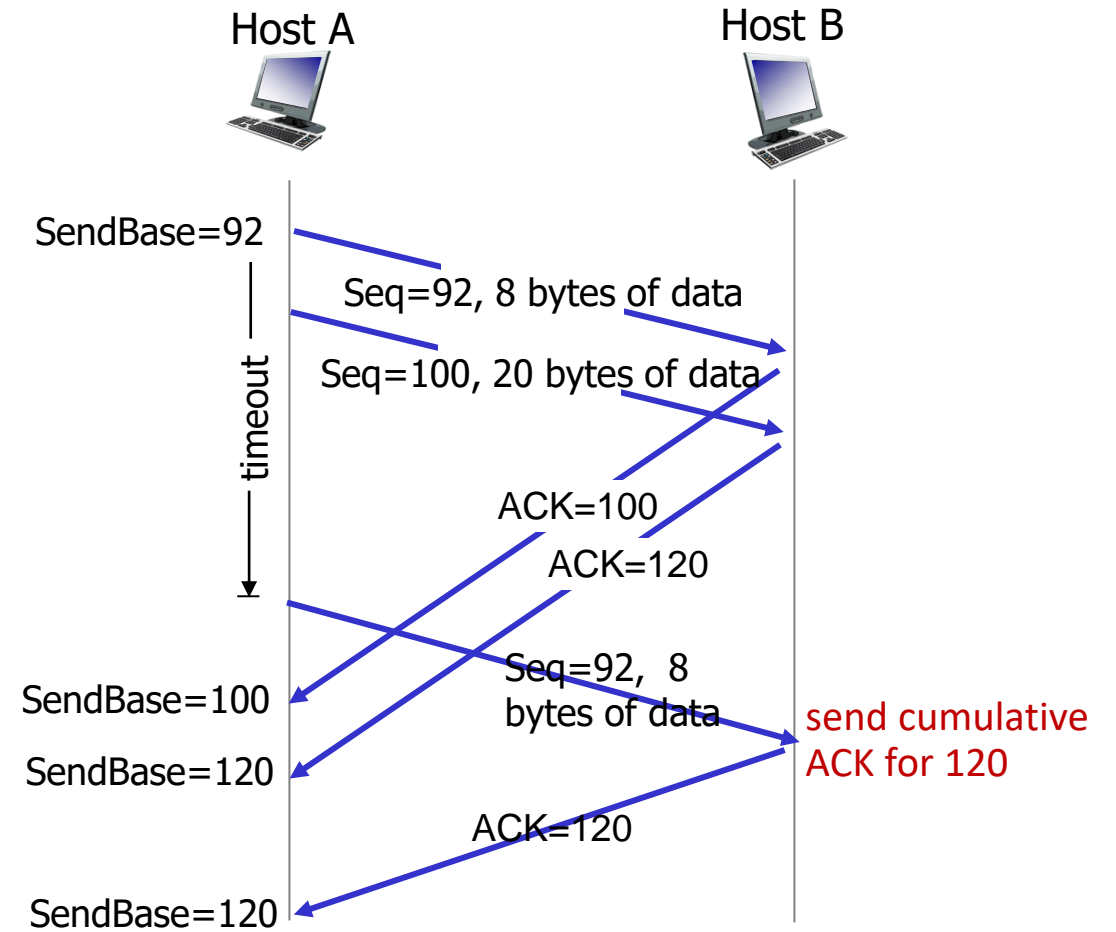
TCP Receiver: ACK generation [RFC 5681]



TCP: retransmission scenarios

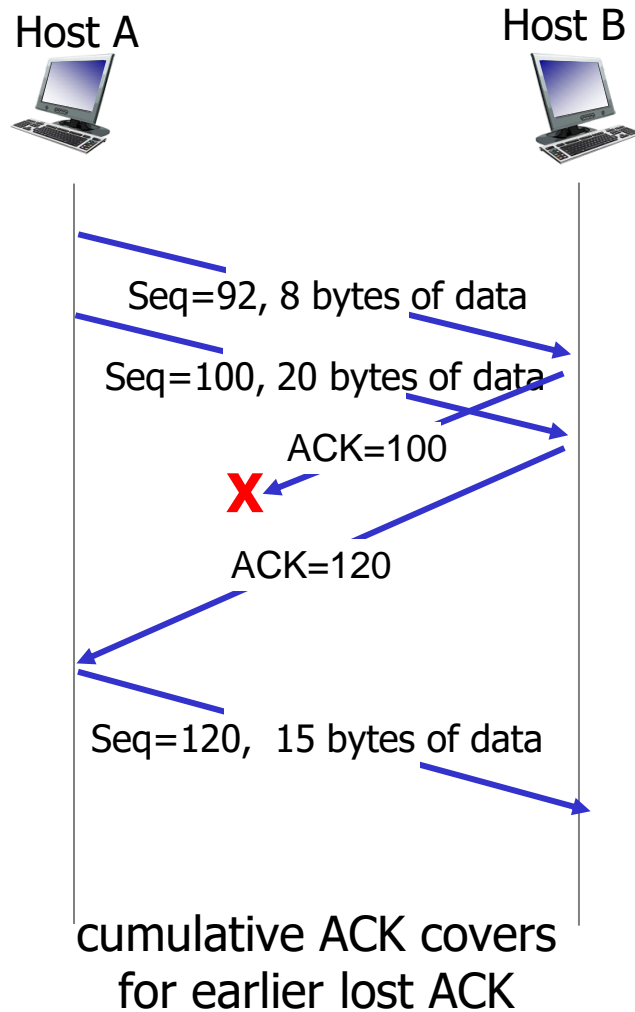


lost ACK scenario



premature timeout

TCP: retransmission scenarios



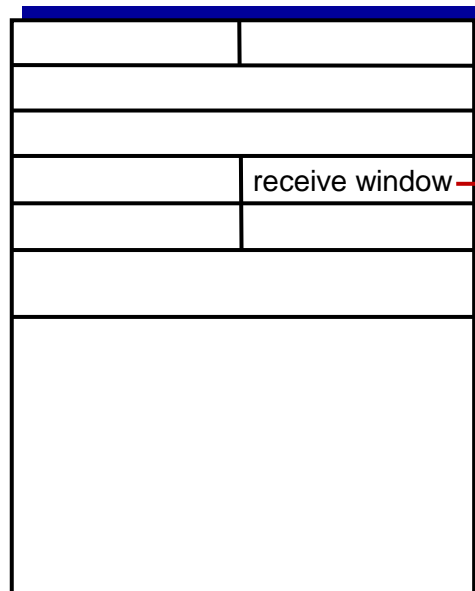
Chapter 3: roadmap

- Transport-layer services
- Multiplexing and demultiplexing
- Connectionless transport: UDP
- **Connection-oriented transport: TCP**
 - segment structure
 - reliable data transfer
 - flow control
 - connection management
- TCP congestion control



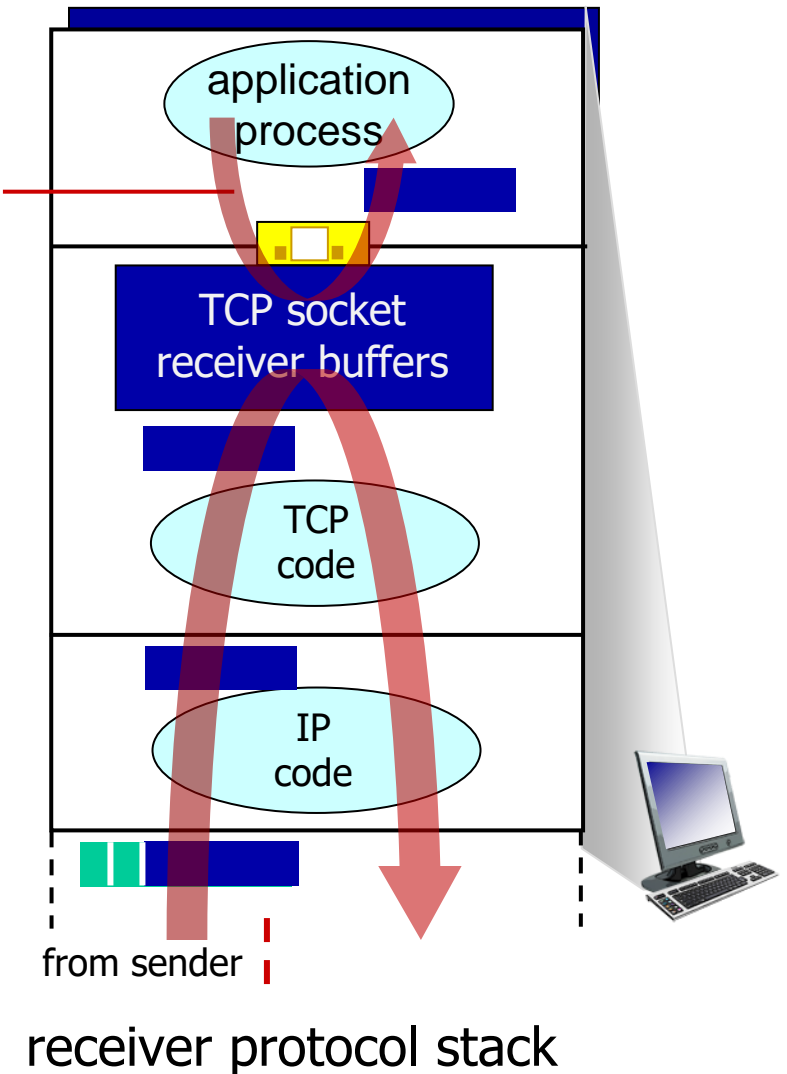
TCP flow control

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

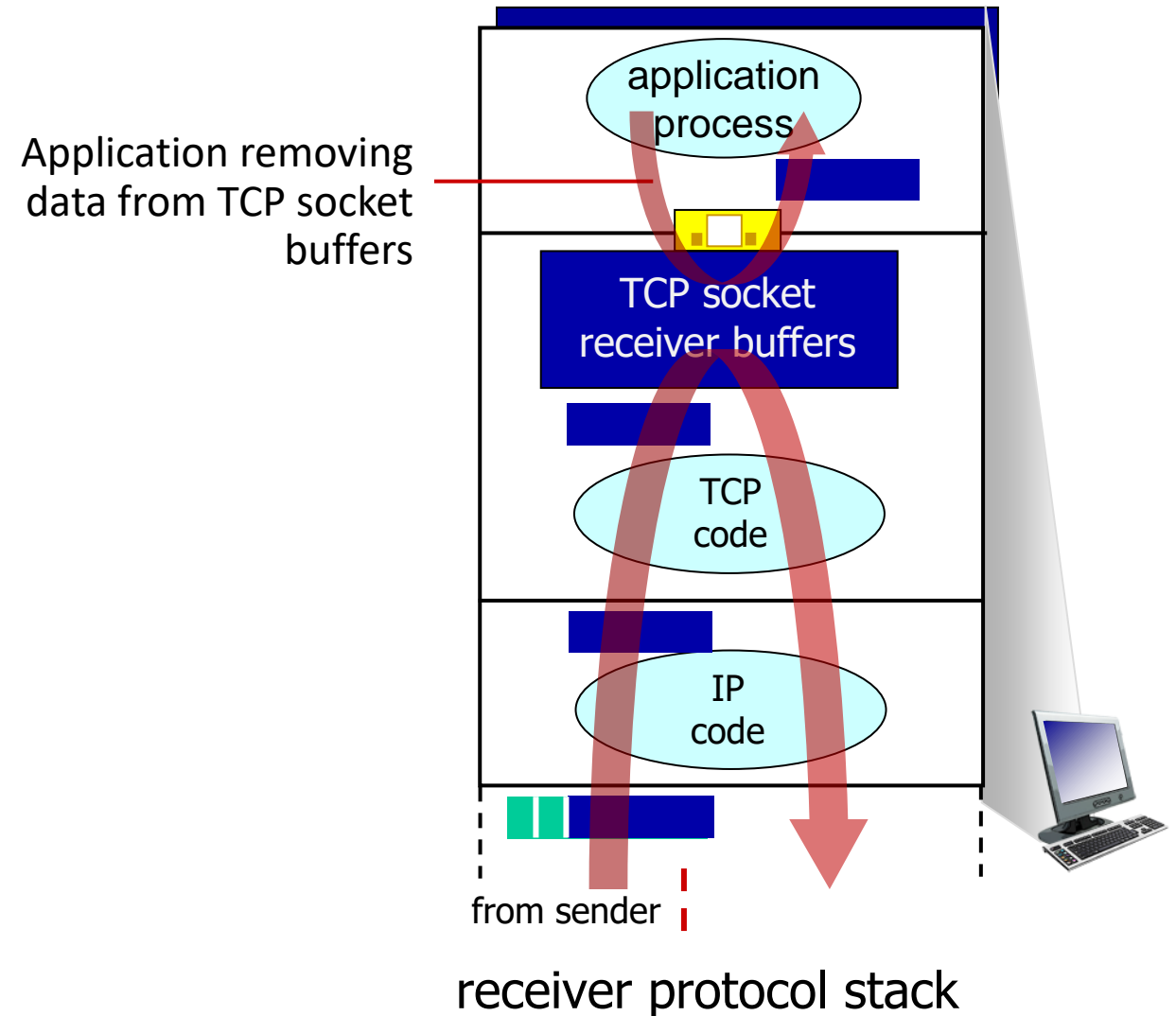


TCP flow control

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

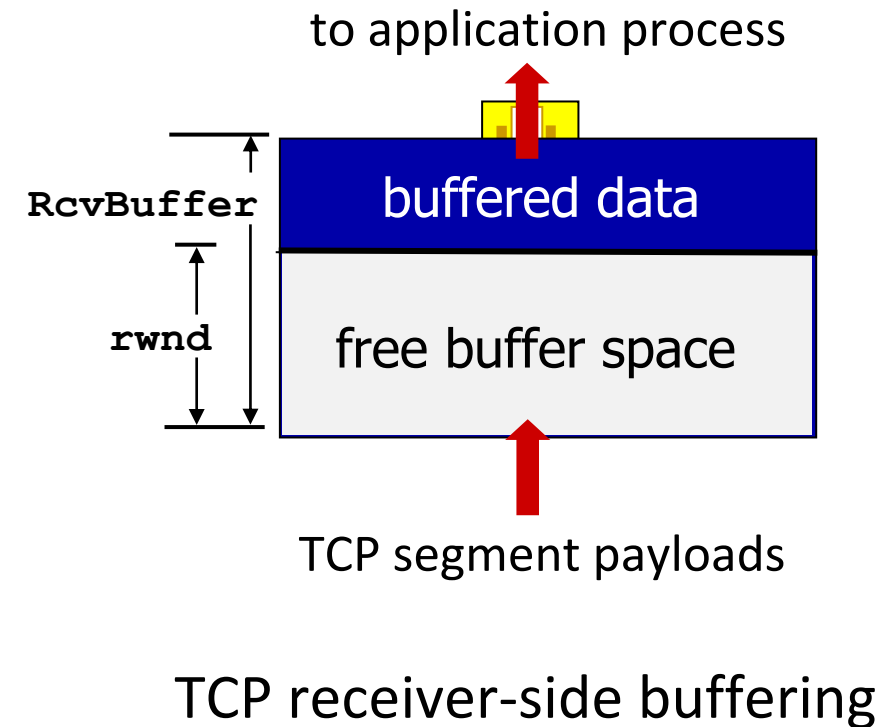
—flow control—

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



TCP flow control

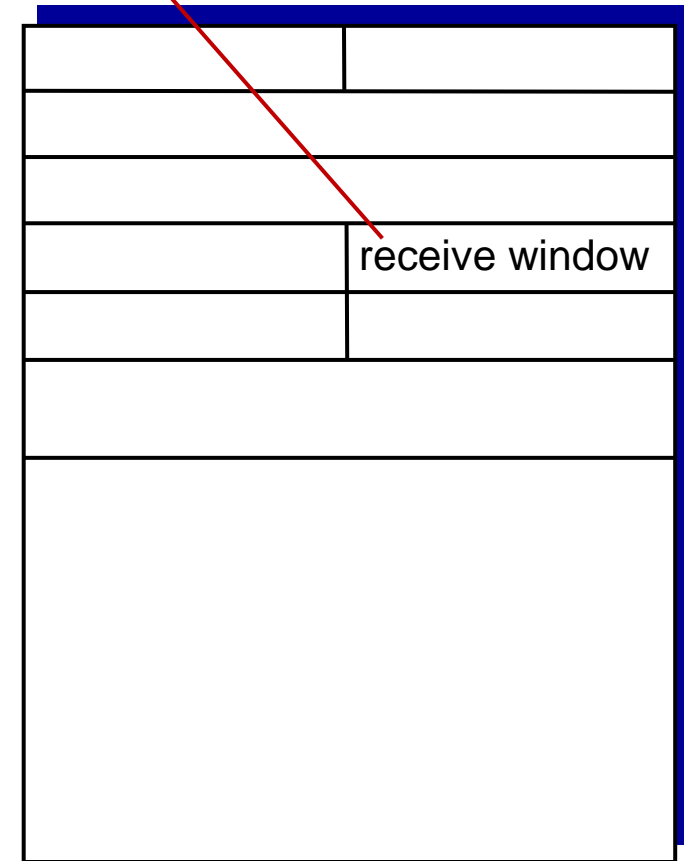
- TCP receiver “advertises” free buffer space in **rwnd** field in TCP header
 - **RcvBuffer** size set via socket options (typical default is 4096 bytes)
 - many operating systems autoadjust **RcvBuffer**
- sender limits amount of unACKed (“in-flight”) data to received **rwnd**
- guarantees receive 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**
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flow control: # bytes receiver willing to accept

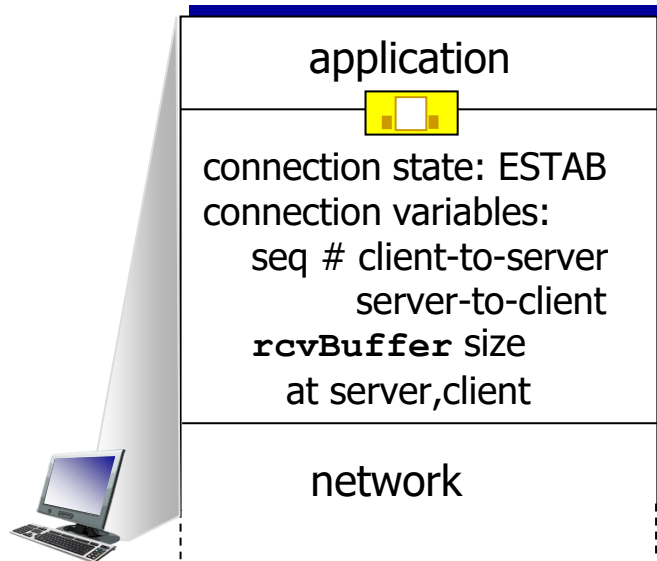


TCP segment format

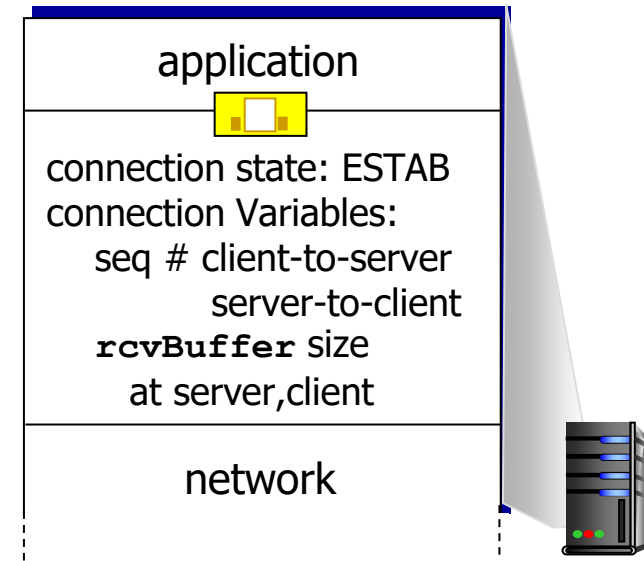
TCP connection management

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();
```


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

SYNbit=1, Seq=x

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

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

ACKbit=1, ACKnum=y+1

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

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

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

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TCP congestion control: AIMD

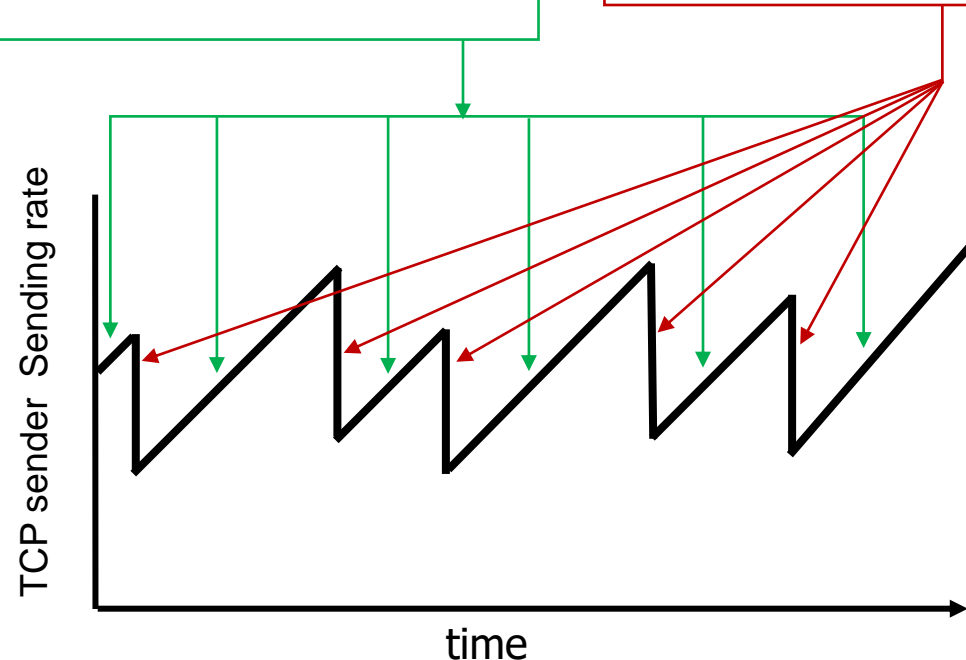
- *approach*: senders can increase sending rate until packet loss (congestion) occurs, then decrease sending rate on loss event

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

Chapter 3: summary

- principles behind transport layer services:
 - multiplexing, demultiplexing
 - reliable data transfer
 - flow control
 - congestion control
- instantiation, implementation in the Internet
 - UDP
 - TCP

Up next:

- leaving the network “edge” (application, transport layers)
- into the network “core”
- two network-layer chapters:
 - data plane
 - control plane