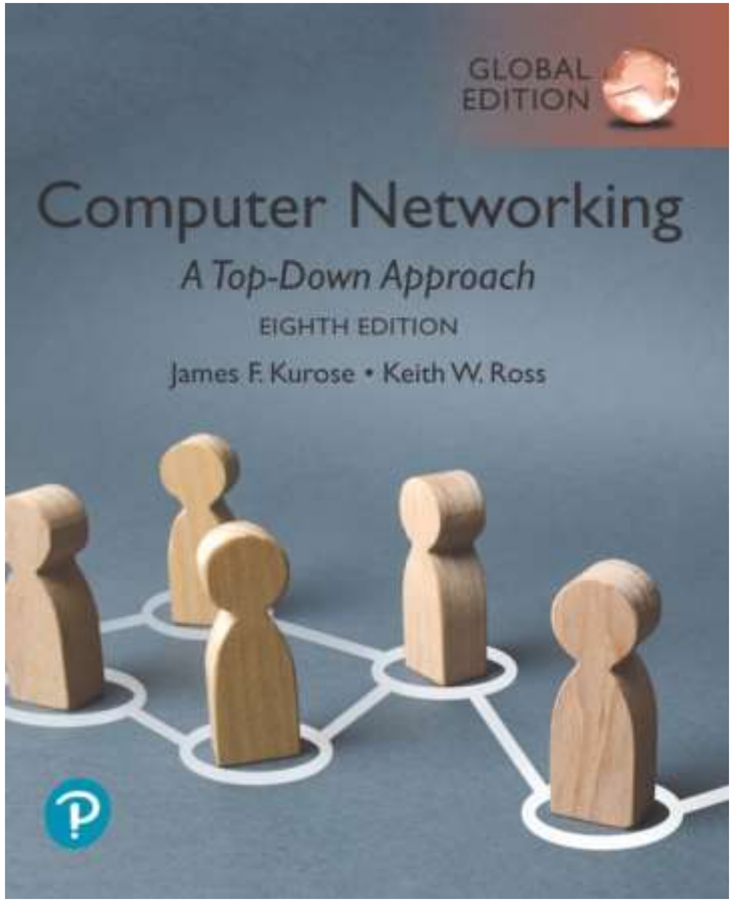
Chapter 3
Transport Layer

Ren Ping Liu
renping.liu@uts.edu.au

adapted from textbook slides by JFK/KWR

18 Mar 2024



Computer Networking: A Top-Down Approach
8th Edition, Global Edition
Jim Kurose, Keith Ross

Transport layer: roadmap

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

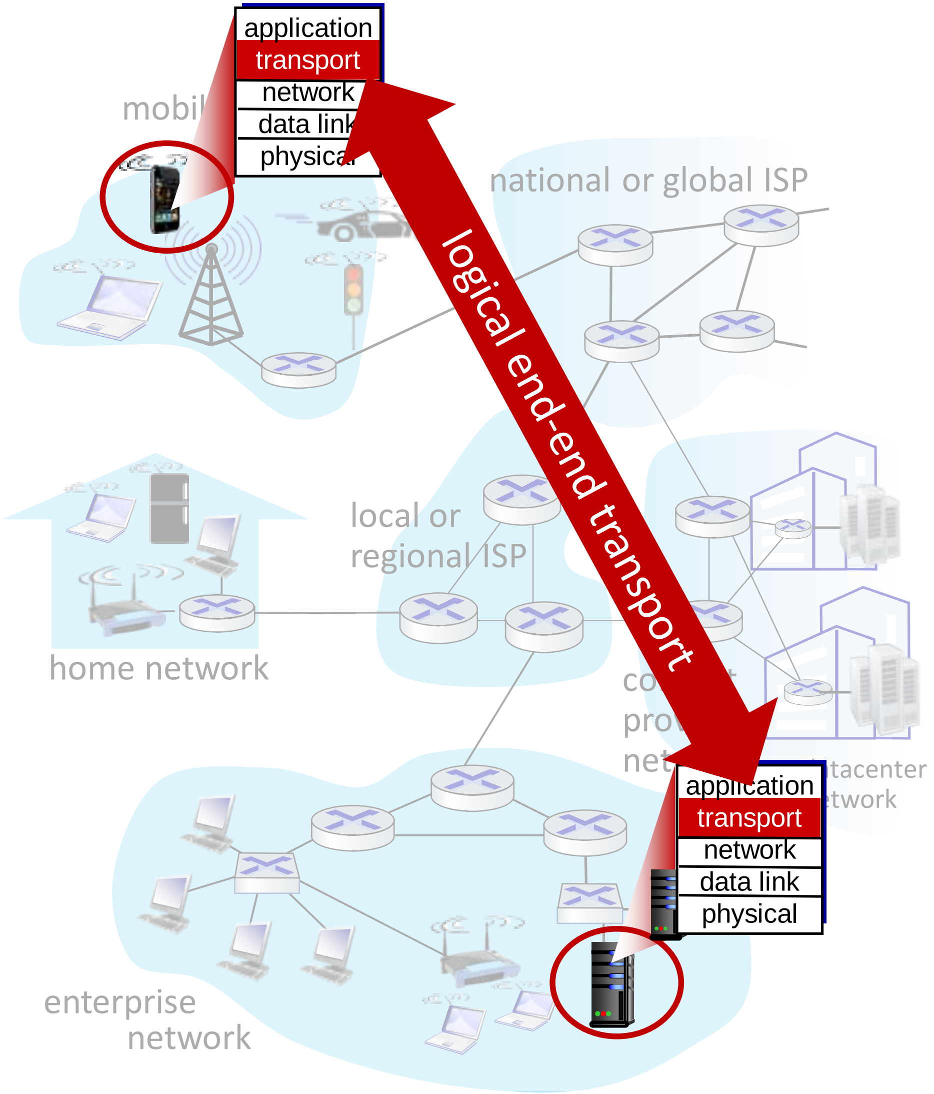
3.6 Principles of congestion control

3.7 TCP congestion control

~~3.8 Evolution of transport-layer functionality~~

Transport services and protocols

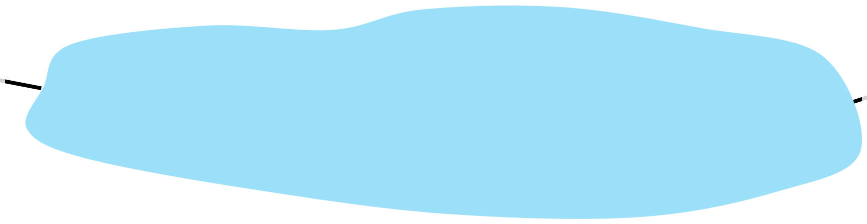
- provide *logical communication* between application processes running on different hosts
- transport protocols actions in end systems:
  - sender: breaks application messages into *segments*, passes to network layer
  - receiver: reassembles segments into messages, passes to application layer
- two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

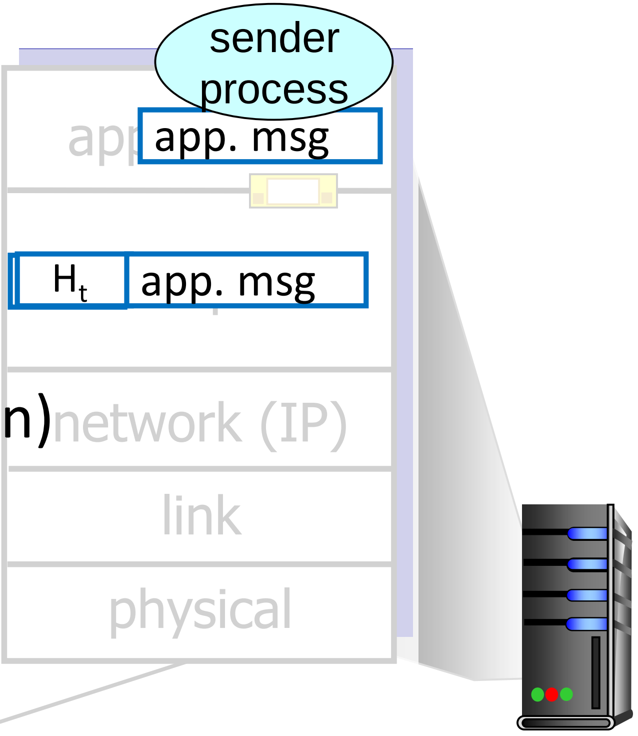


Transport Layer Actions

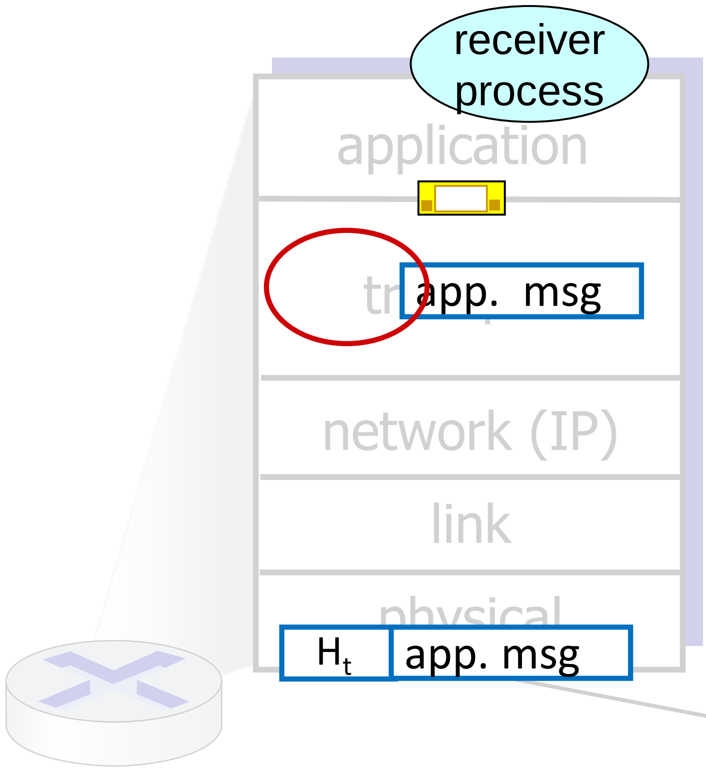
Sender:

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



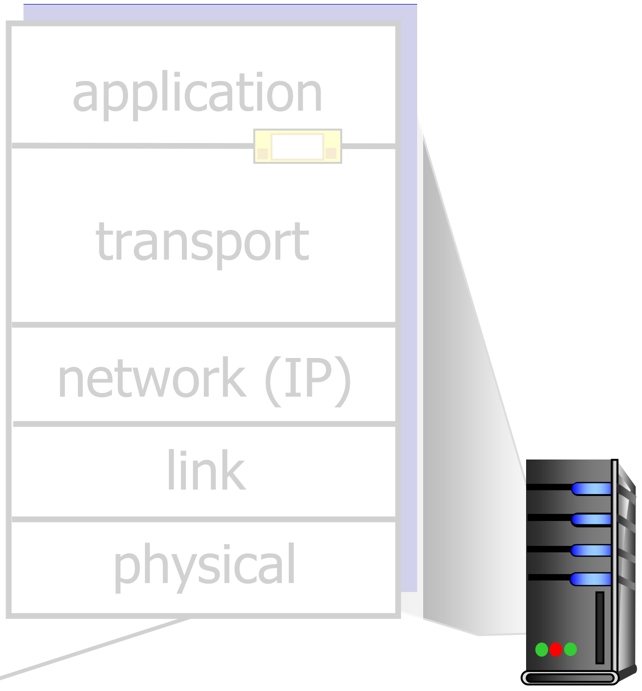


Transport Layer Actions



Receiver:

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



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 vs. network layer services and protocols

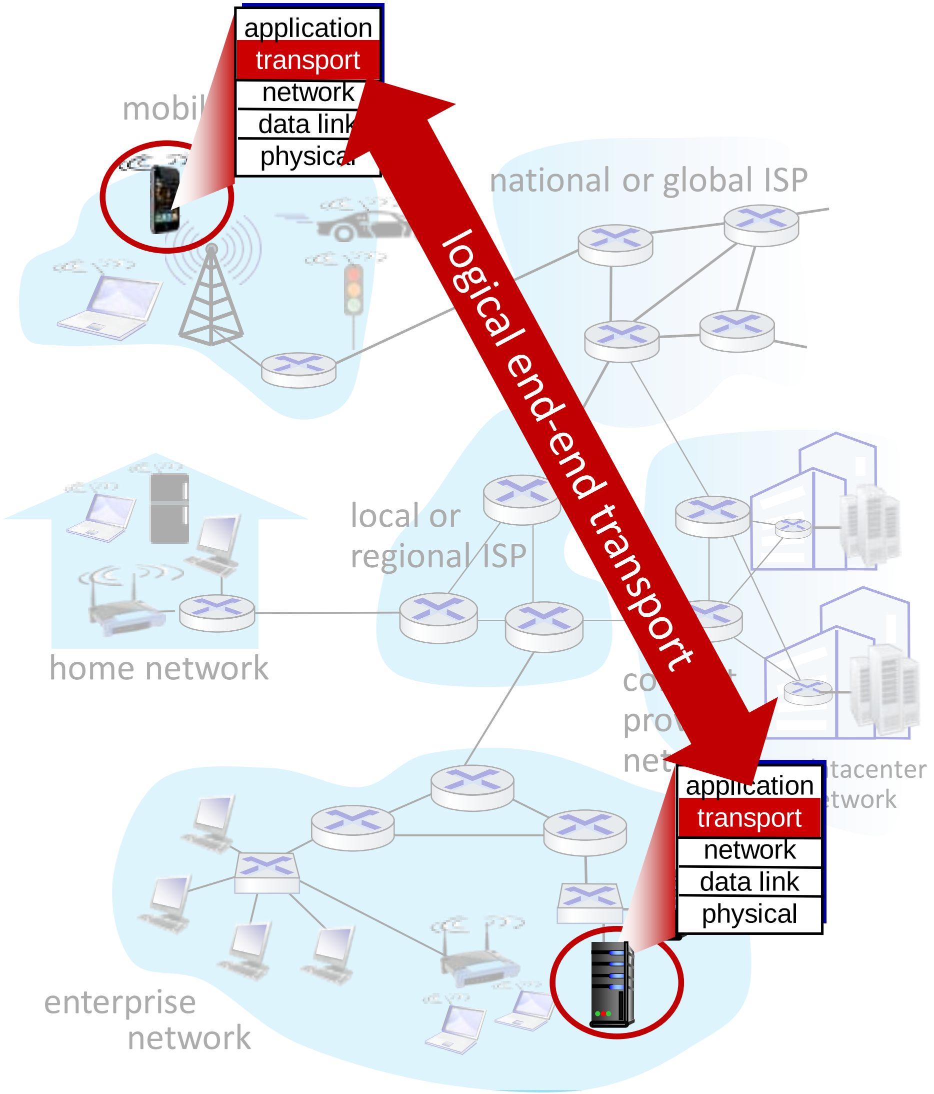
household analogy:

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

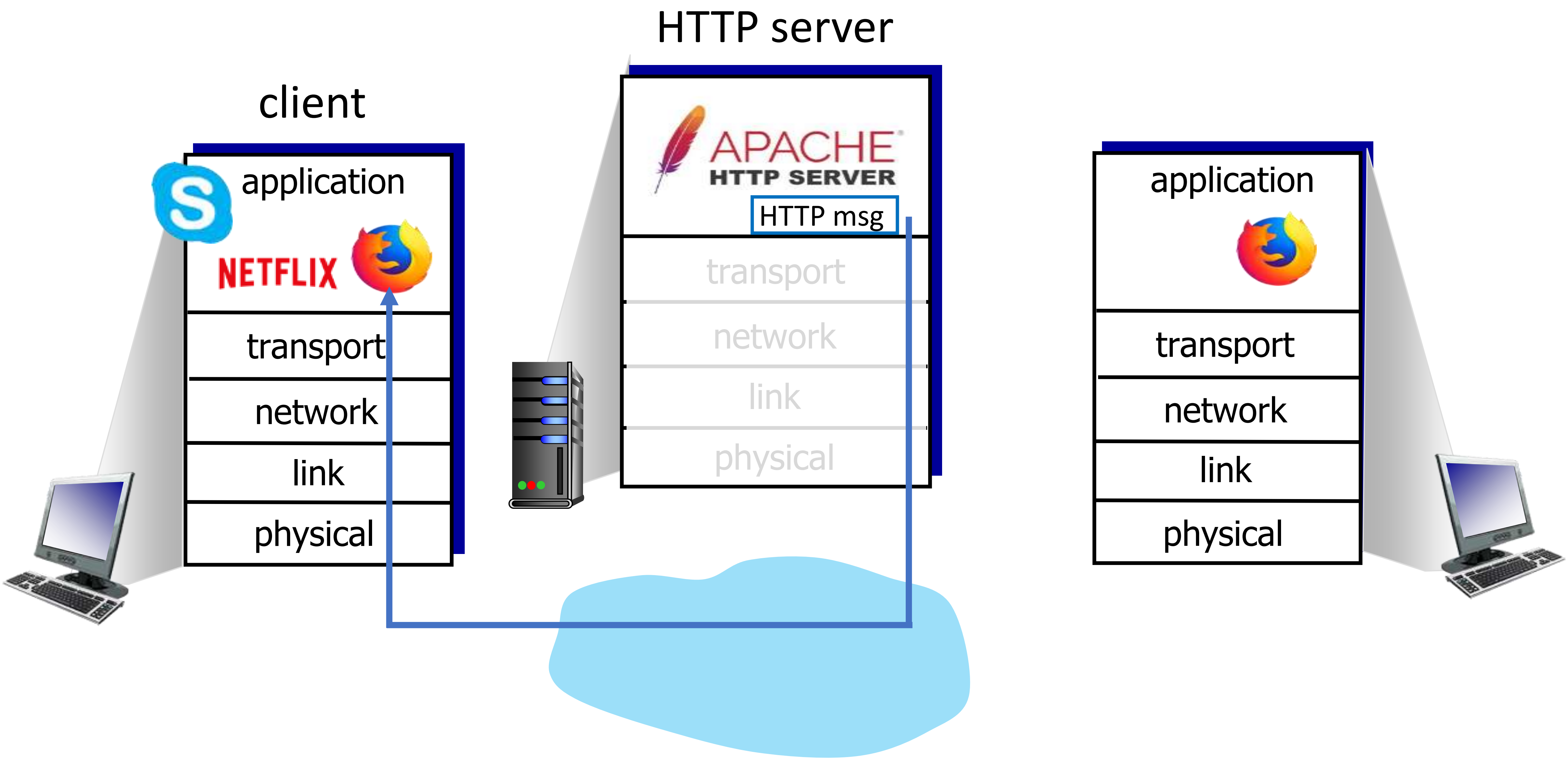
- hosts = houses
- processes = kids
- app messages = letters in envelopes

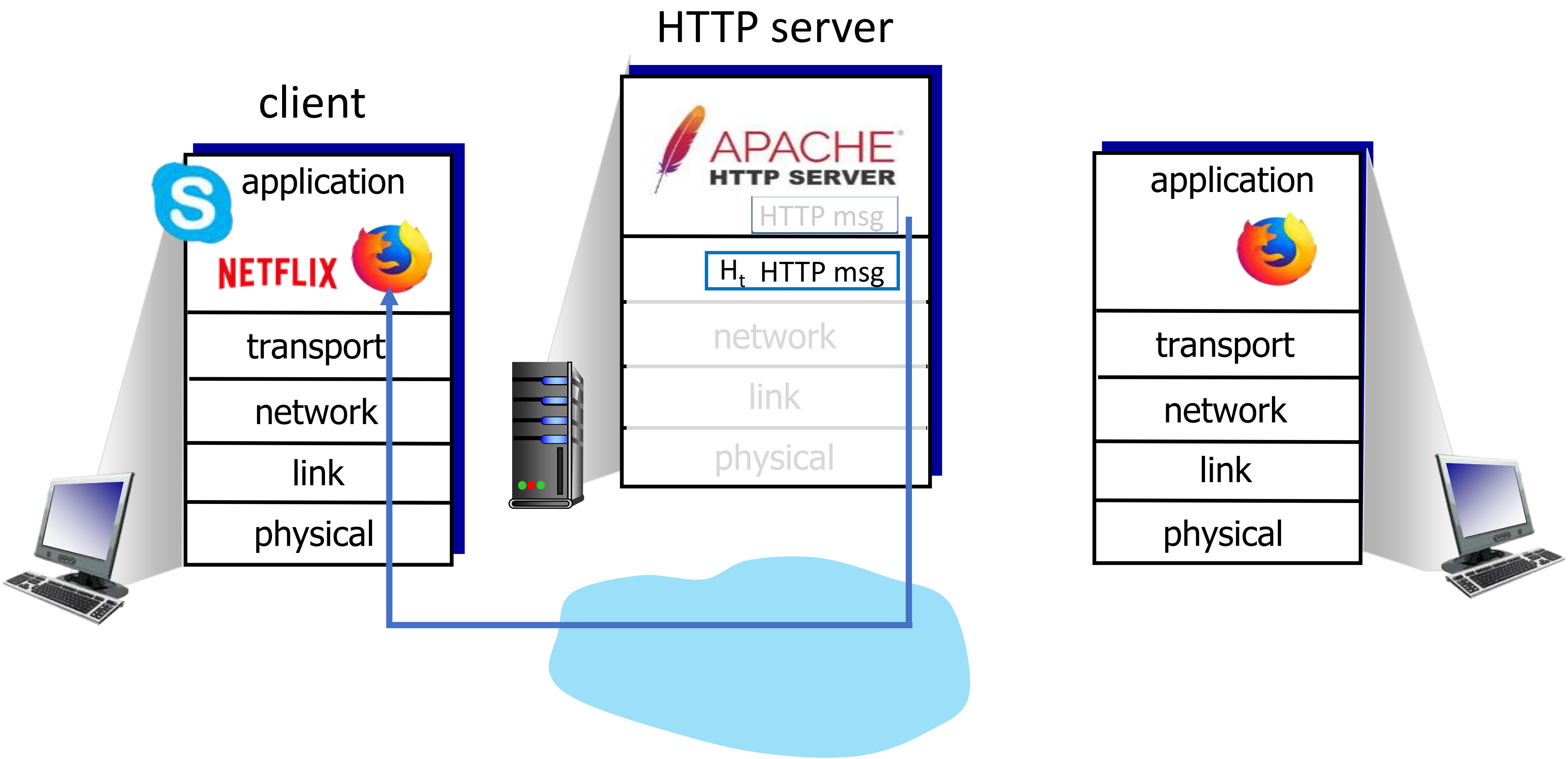
Two principal Internet transport protocols

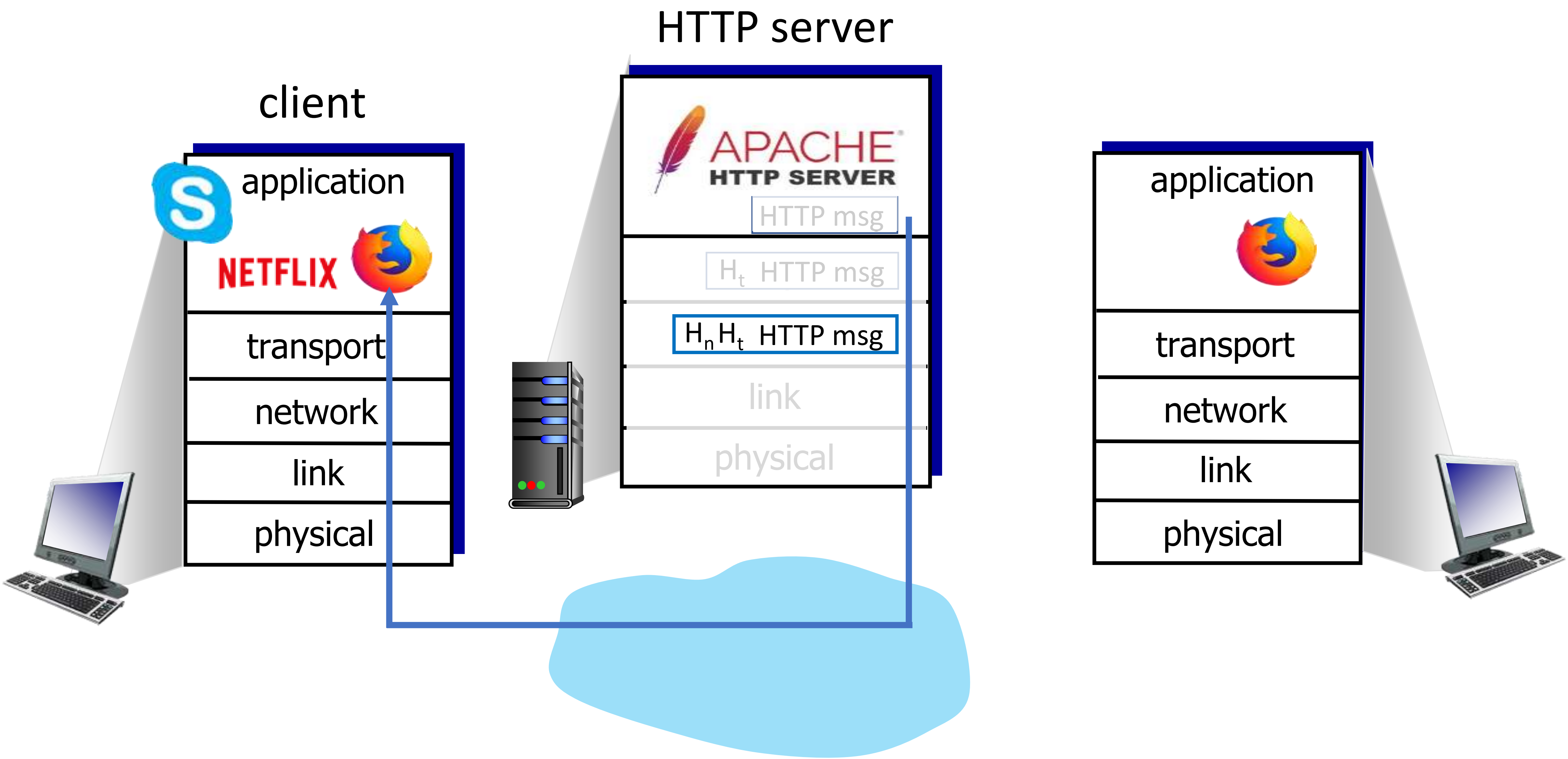
- **TCP:** Transmission Control Protocol
  - reliable, in-order delivery
  - congestion control
  - flow control
  - connection setup
- **UDP:** User Datagram Protocol
  - unreliable, unordered delivery
  - no-frills extension of “best-effort” IP
- services not available:
  - delay guarantees
  - bandwidth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

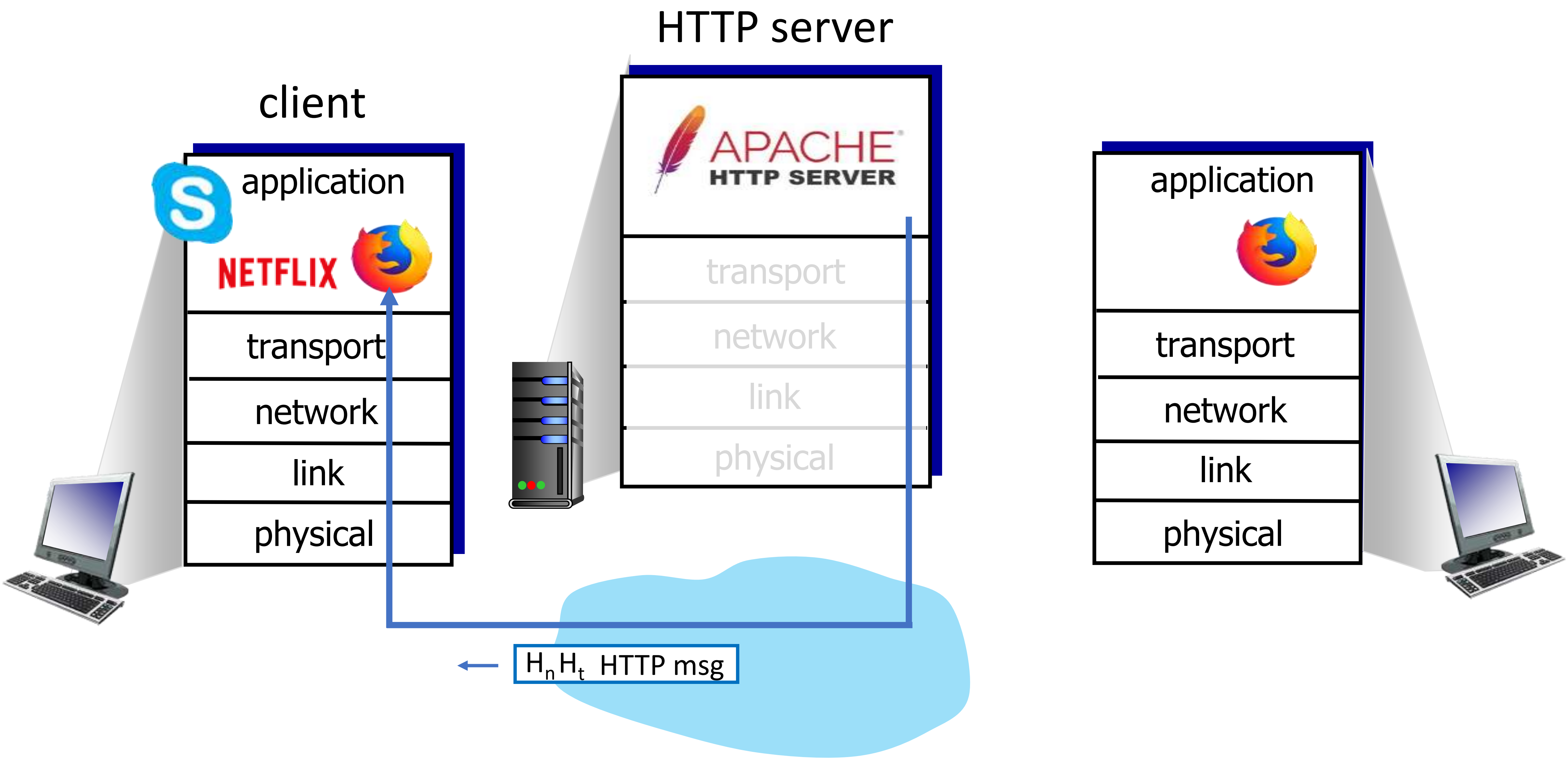


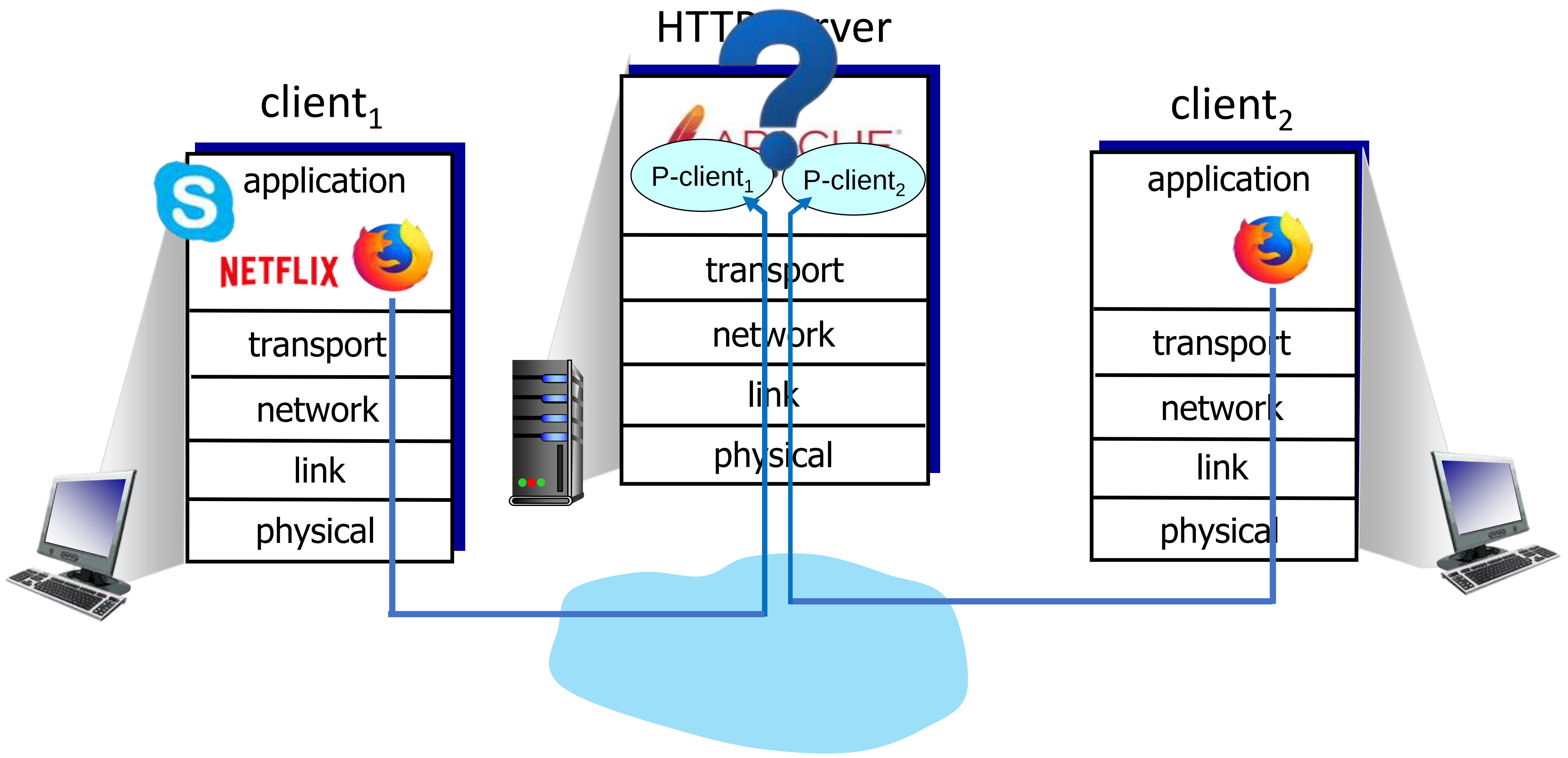
Transport layer: roadmap


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


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


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


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


HTTP server
client1
client2
application
NETFLIX
transport
network
link
physical
P-client1
P-client2
transport
network
link
physical
application
transport
network
link
physical

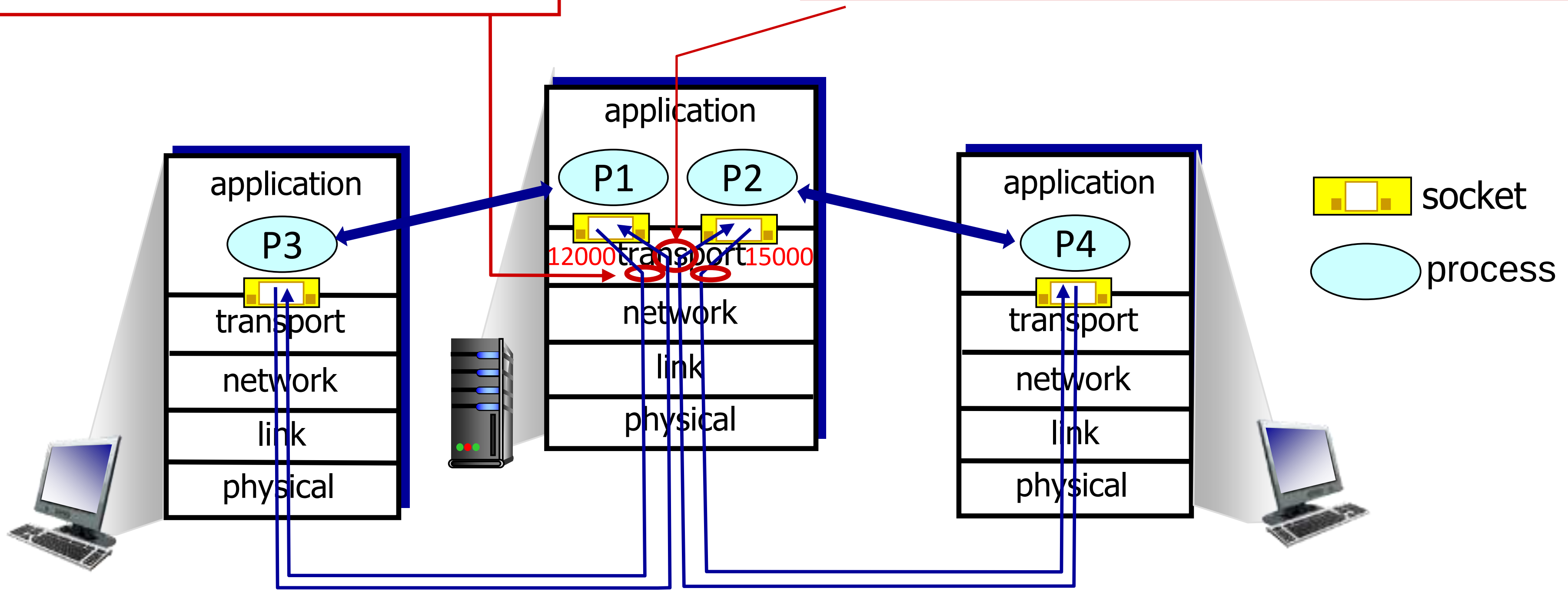
Multiplexing/demultiplexing

multiplexing at sender:

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

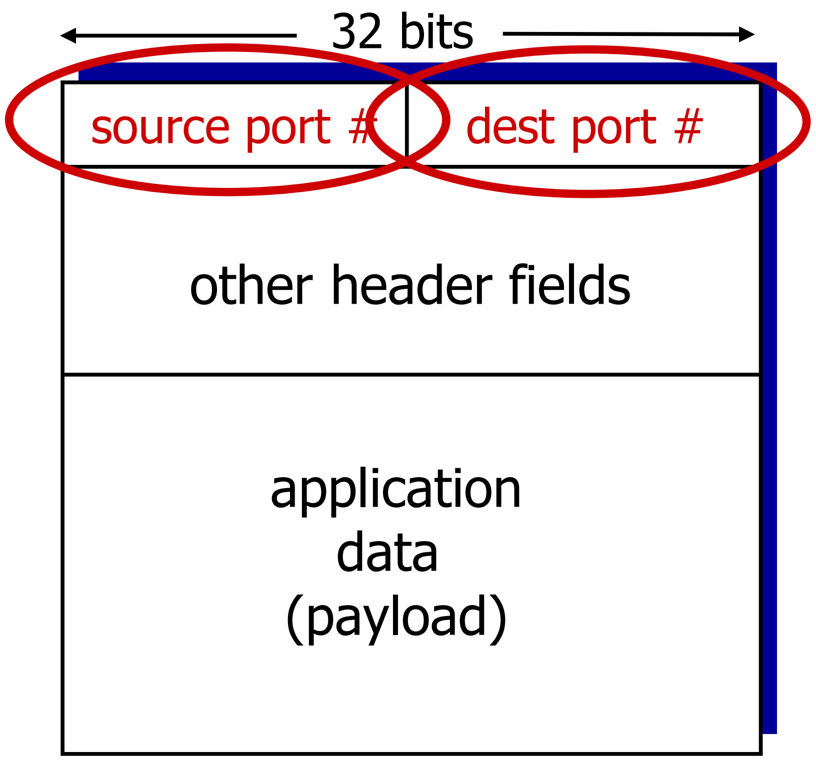
demultiplexing at receiver:

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 (UDP) demultiplexing

Recall:

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

```
serverSock.bind((’’, 12000));
```

- 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 #

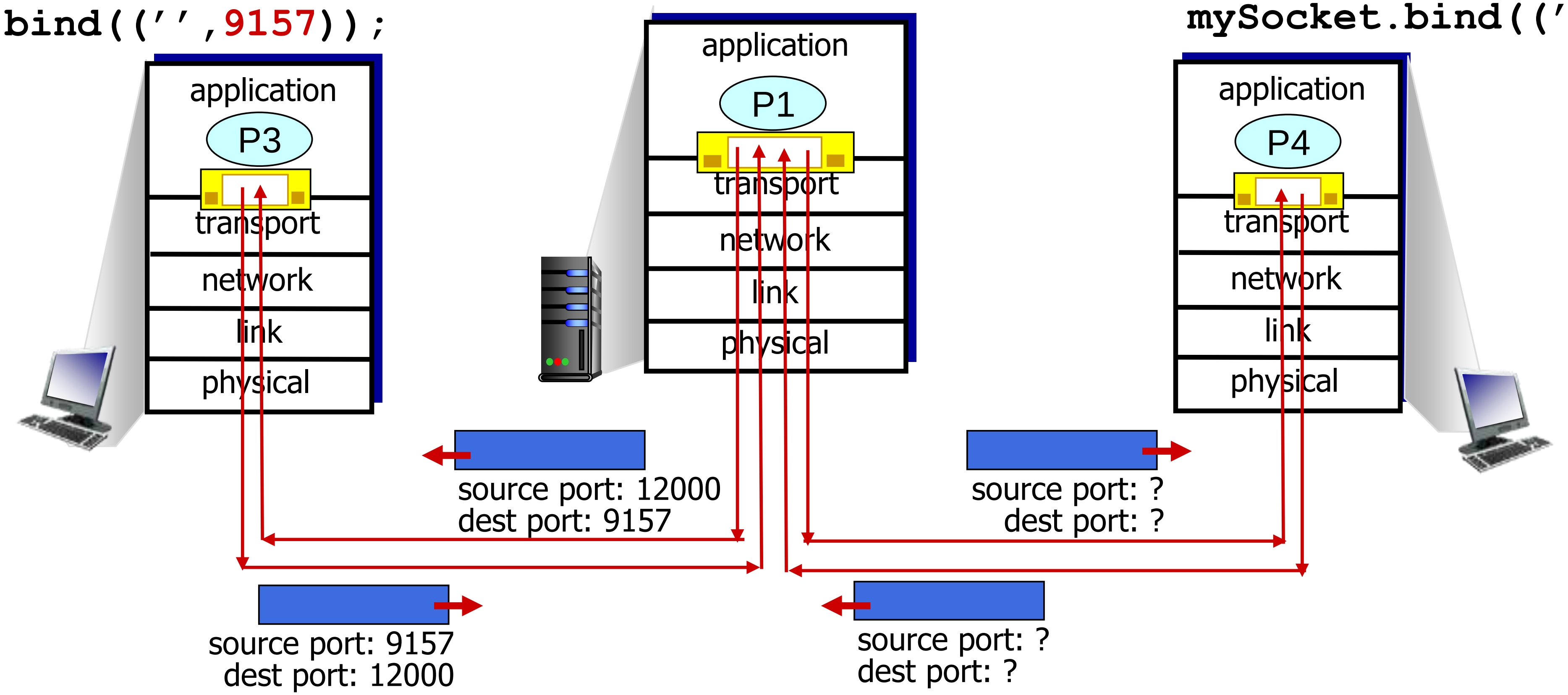
IP/UDP datagrams with *same dest. port #,* but different source IP addresses and/or source port numbers will be directed to *same socket* at receiving host

Connectionless (UDP) demultiplexing: an example

```
mySocket2=socket
 (AF_INET,SOCKET_DGRAM)
mySocket.bind(('',9157));
```

```
serverSocket=socket
 (AF_INET,SOCKET_DGRAM)
serverSocket.bind(('',12000));
```

```
mySocket1=socket
 (AF_INET,SOCKET_DGRAM
mySocket.bind(('',5775));
```

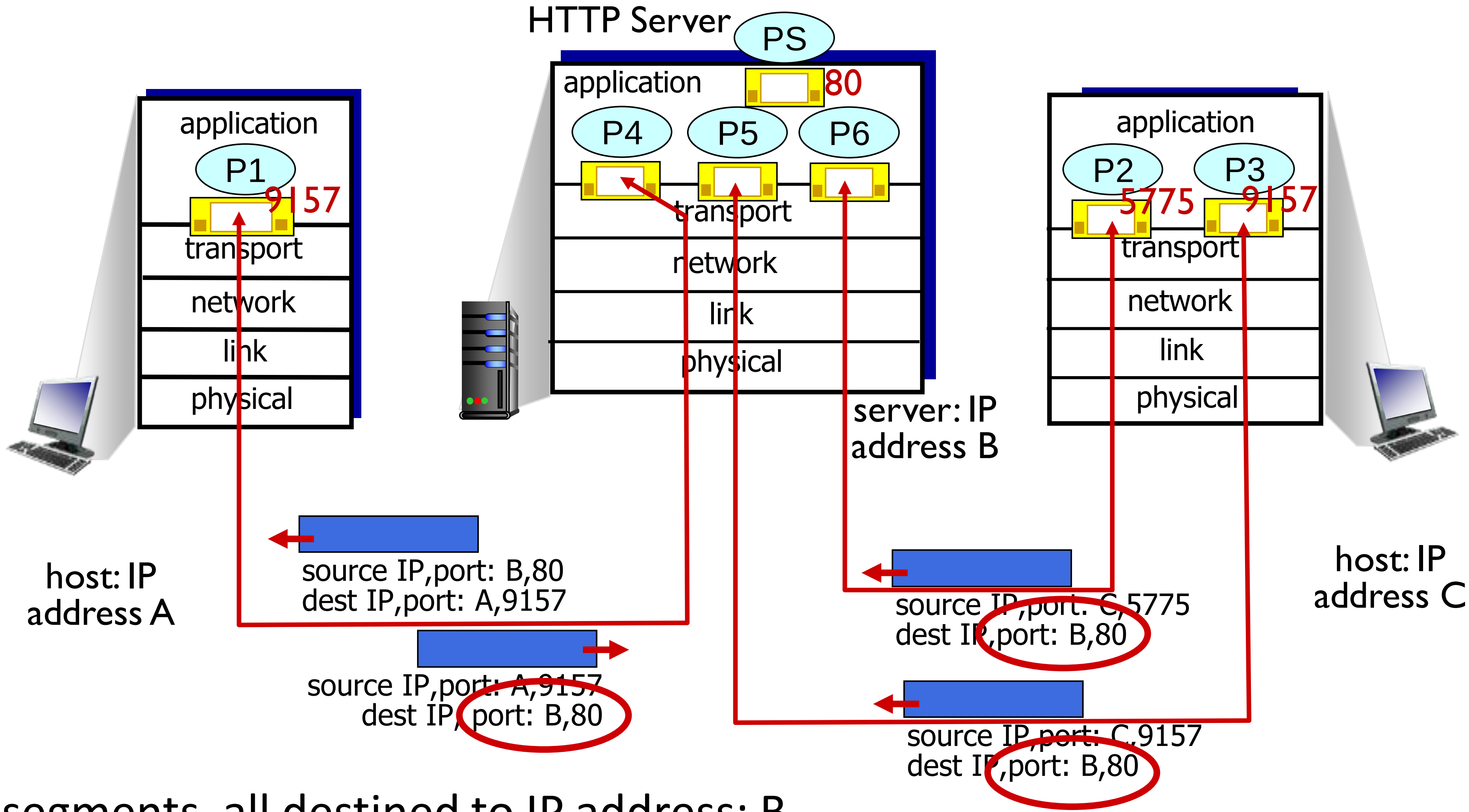


Connection-oriented (TCP) demultiplexing

- 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 (TCP) demultiplexing: example



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

Summary of MUX/DEMUX

- 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

Transport layer: roadmap

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

3.6 Principles of congestion control

3.7 TCP congestion control

3.8 Evolution of transport-layer functionality

UDP: User Datagram Protocol

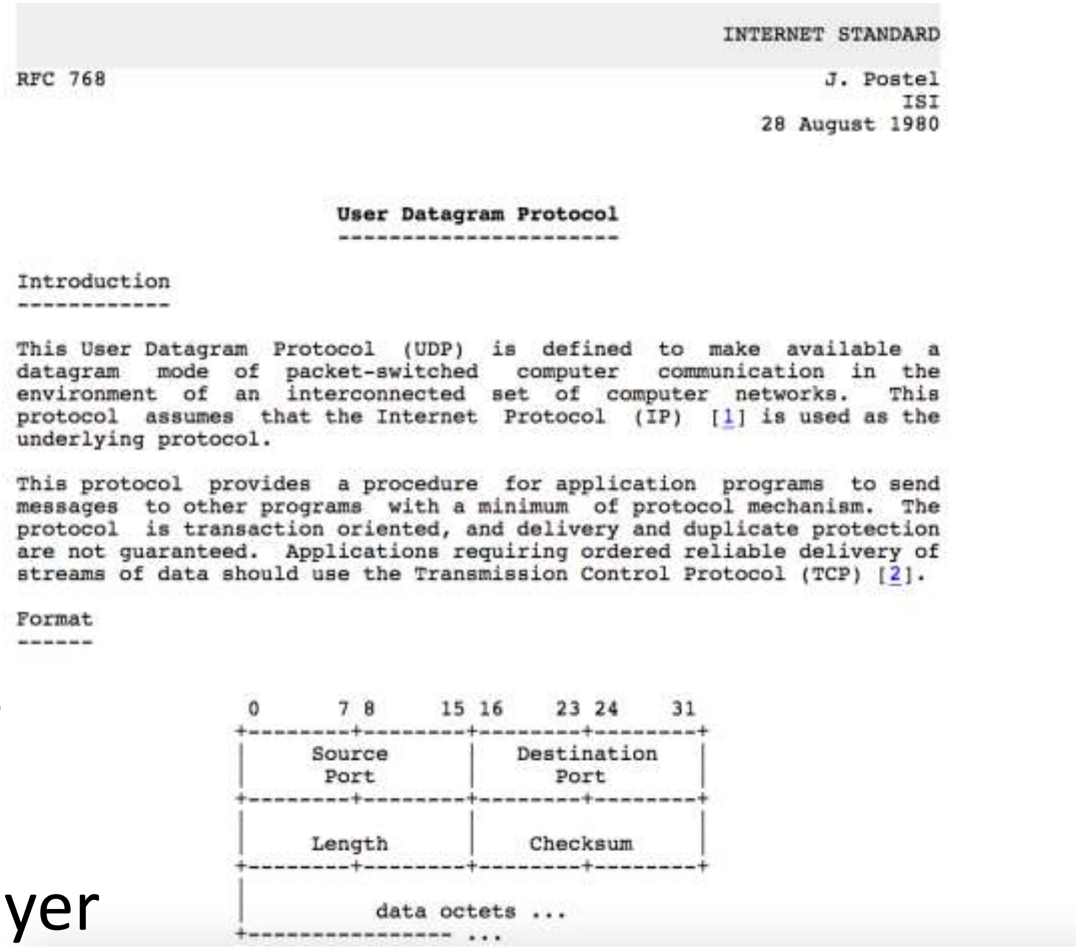
- "no frills," "bare bones" Internet transport protocol
- "best effort" service, UDP segments may be:
  - lost
  - delivered out-of-order to app
- *connectionless:*
  - no handshaking between UDP sender, receiver
  - each UDP segment handled independently of others

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 [RFC 768]

- 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



INTERNET STANDARD

RFC 768 J. Postel
ISI
28 August 1980

User Datagram Protocol

Introduction

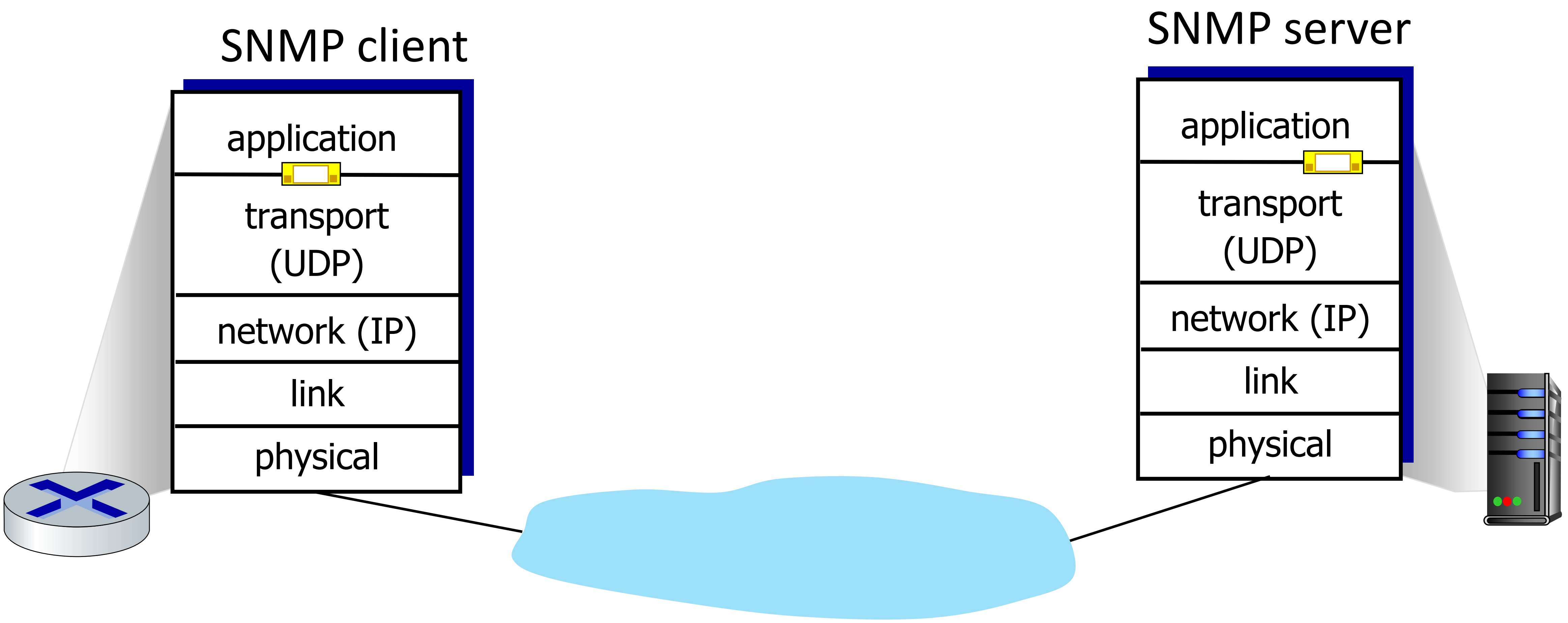
This User Datagram Protocol (UDP) is defined to make available a datagram mode of packet-switched computer communication in the environment of an interconnected set of computer networks. This protocol assumes that the Internet Protocol (IP) [1] is used as the underlying protocol.

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

```
 0      7 8     15 16    23 24    31
+--------+--------+--------+--------+
|     Source      |   Destination   |
|      Port       |      Port       |
+--------+--------+--------+--------+
|                 |                 |
|     Length      |    Checksum     |
+--------+--------+--------+--------+
|
|          data octets ...
+---------------- ...
```

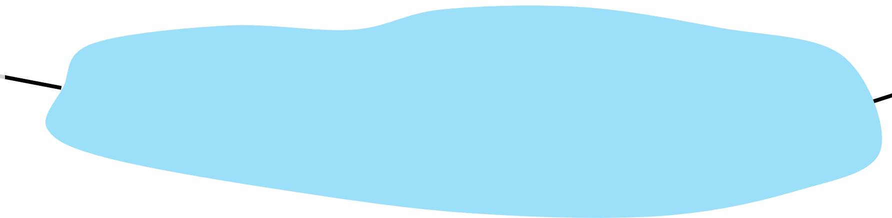
UDP: Transport Layer Actions

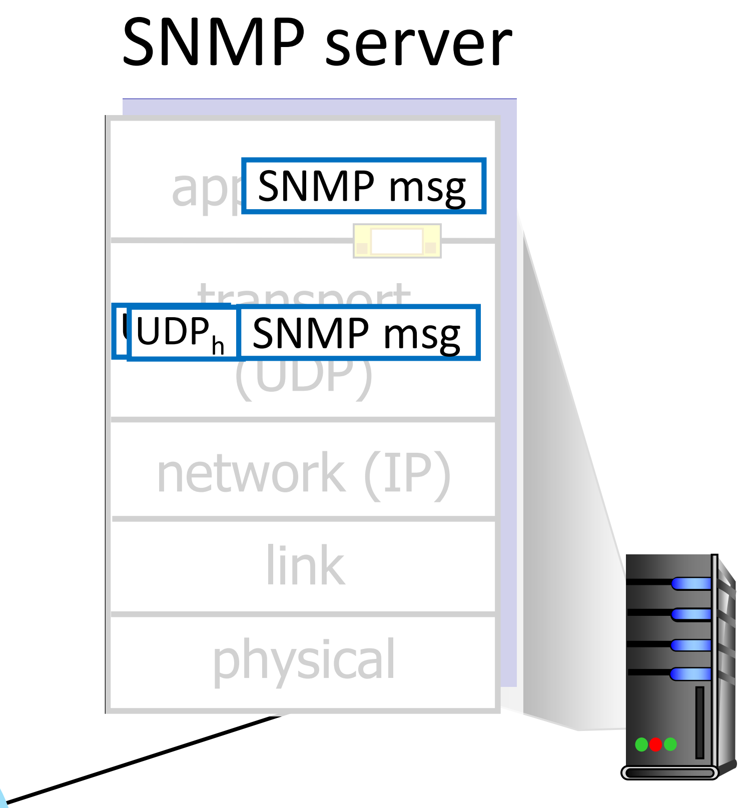


UDP: Transport Layer Actions

UDP sender actions:

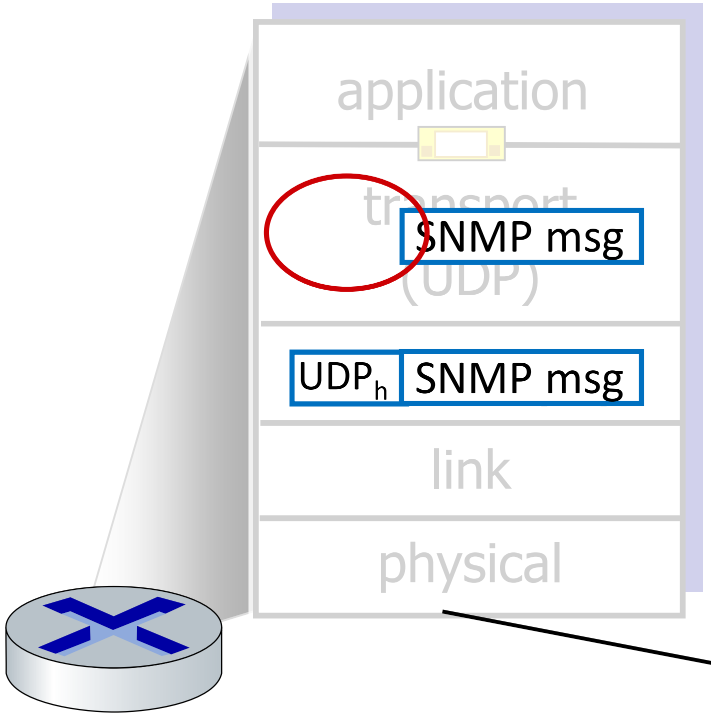
- is passed an application-layer message
- determines UDP segment header fields values
- creates UDP segment (segmentation
- passes segment to IP





UDP: Transport Layer Actions

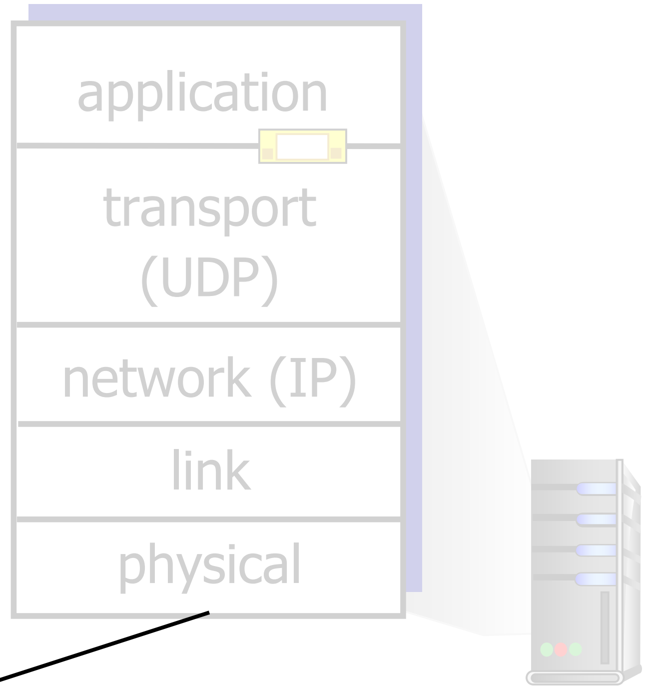
SNMP client



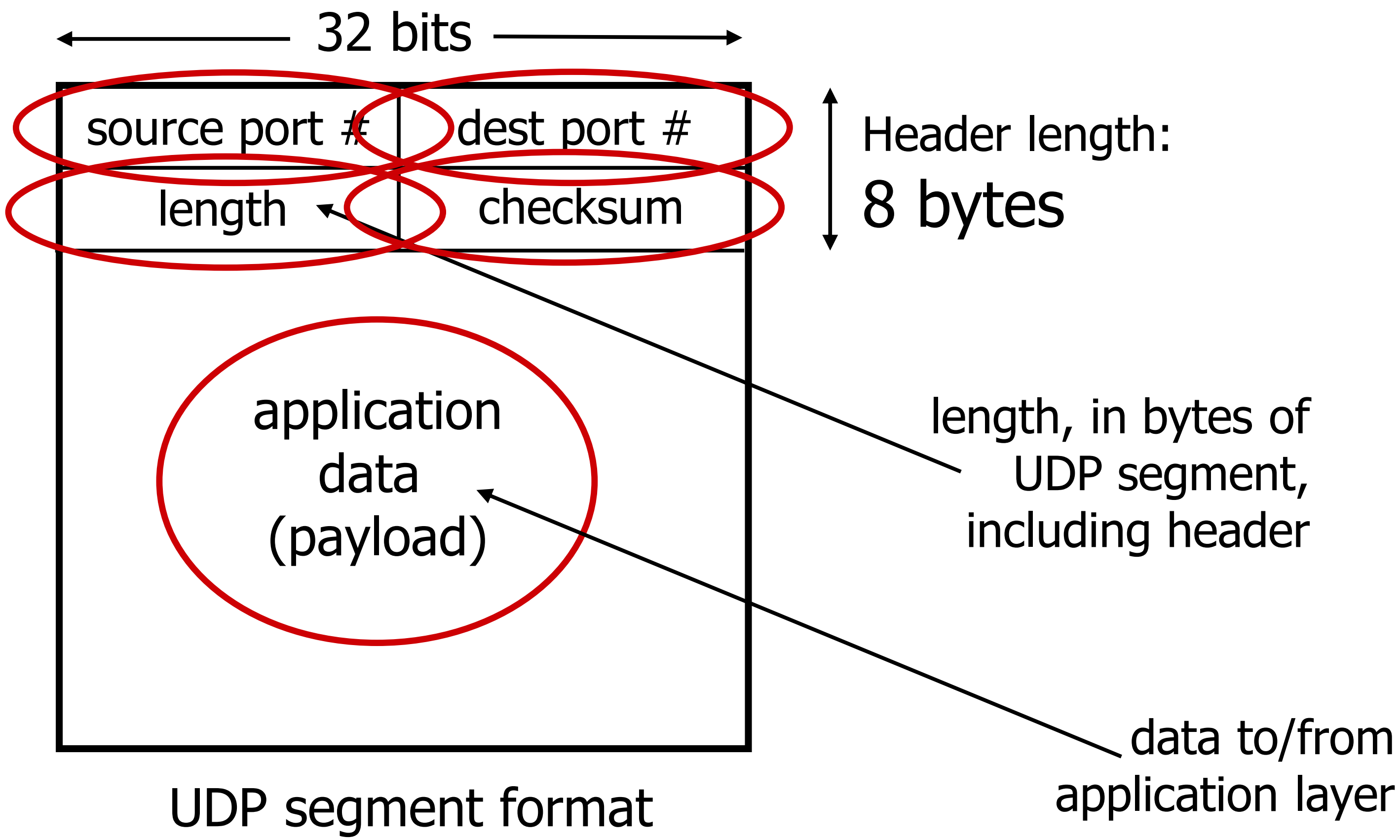
UDP receiver actions:

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

SNMP server



UDP segment header



UDP segment format

UDP checksum

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

	1st number	2nd number	sum
Transmitted:	3	6	9
Received:	2	6	9

receiver-computed checksum ≠ sender-computed checksum (as received)

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

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

At the Transmitter end:

1. compute the sum from data (above)
2. send data with checksum (complement of sum) to Receiver

Error Checking at the Receiver end:

1. compute the "sum" from the received data
2. Add "sum" with received "checksum" == all '1' ➔Correct

* 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

		0
	1 1 1 0 0 1 1 0 0 1 1 0 0 1 1 0	1
	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	

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

Summary of 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)

Mid-break

☕

- Q & A

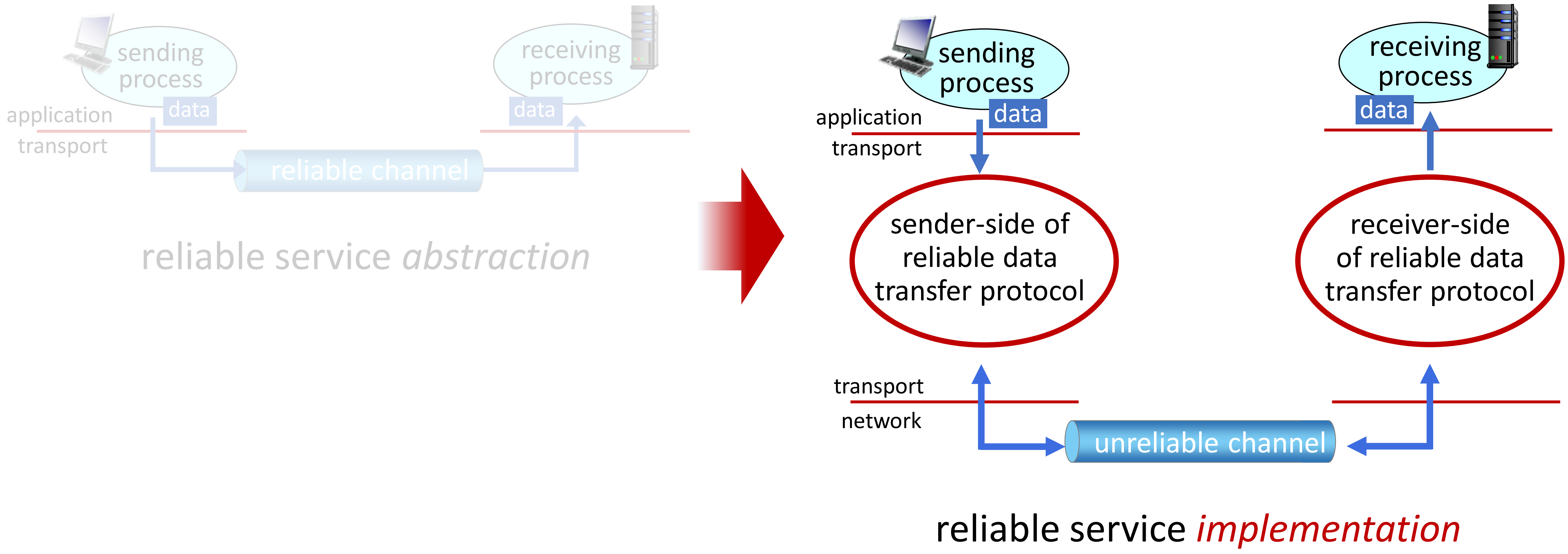
Transport layer: roadmap

- 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
- 3.6 Principles of congestion control
- 3.7 TCP congestion control
- 3.8 Evolution of transport-layer functionality

Principles of reliable data transfer

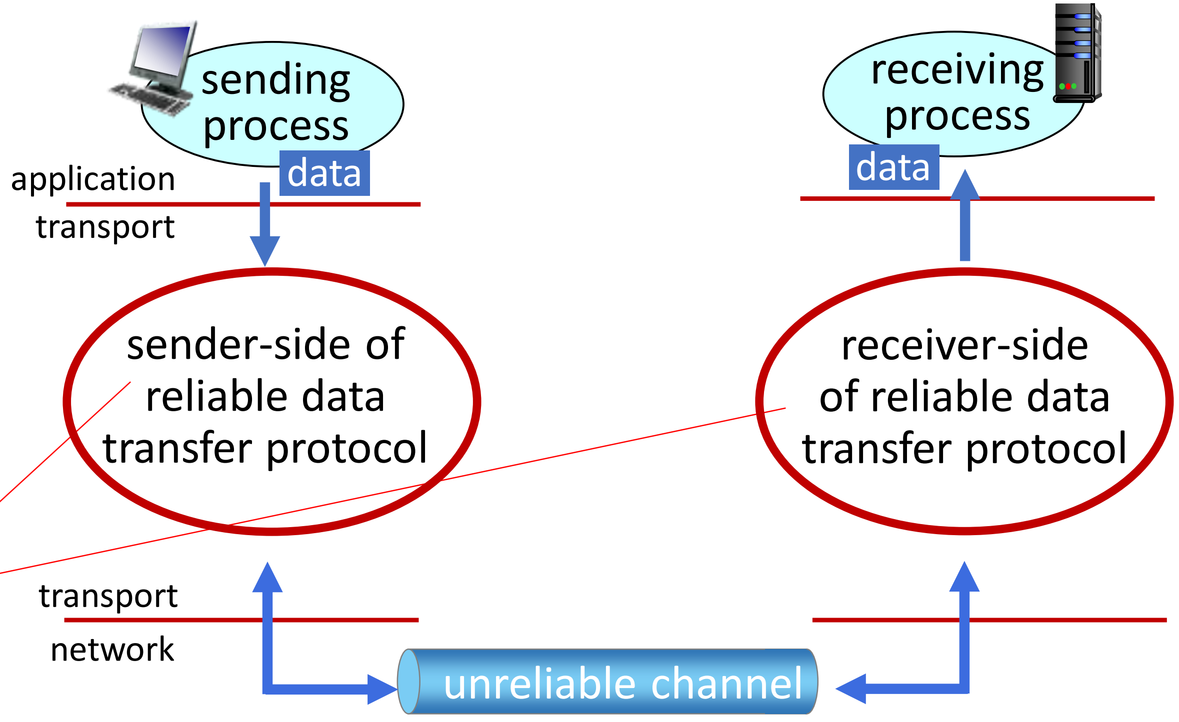
reliable service *abstraction*

Principles of reliable data transfer



Principles of reliable data transfer

Complexity of reliable data transfer protocol will depend (strongly) on characteristics of unreliable channel (lose, corrupt, reorder data?)

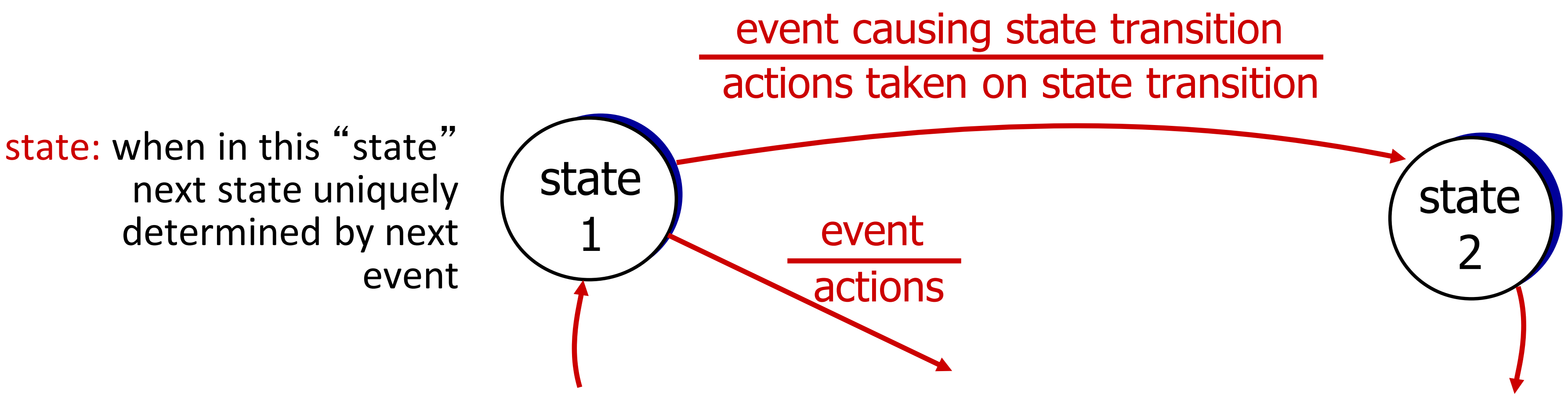


reliable service *implementation*

Reliable data transfer: getting started

We will:

- incrementally develop sender, receiver sides of reliable data transfer protocol (`rdt`)
- consider only unidirectional data transfer
  - but control info will flow in both directions!
- ~~use finite state machines (FSM) to specify sender, receiver~~



rdt1.0: reliable transfer over a reliable channel

- underlying channel perfectly reliable
  - no bit errors
  - no loss of packets

sender *receiver*

send pkt → pkt → rcv pkt

rdt2.0: channel with bit errors

- underlying channel may flip bits in packet
  - checksum (e.g., Internet checksum) to detect bit errors
- *the* question: how to recover from errors?

How do humans recover from "errors" during conversation?

rdt2.0: channel with bit errors

- underlying channel may flip bits in packet
  - checksum to detect bit errors
- *the* question: how to recover from errors?
  - *acknowledgements (ACKs):* receiver explicitly tells sender that pkt received OK
  - *negative acknowledgements (NAKs):* receiver explicitly tells sender that pkt had errors
  - sender *retransmits* pkt on receipt of NAK

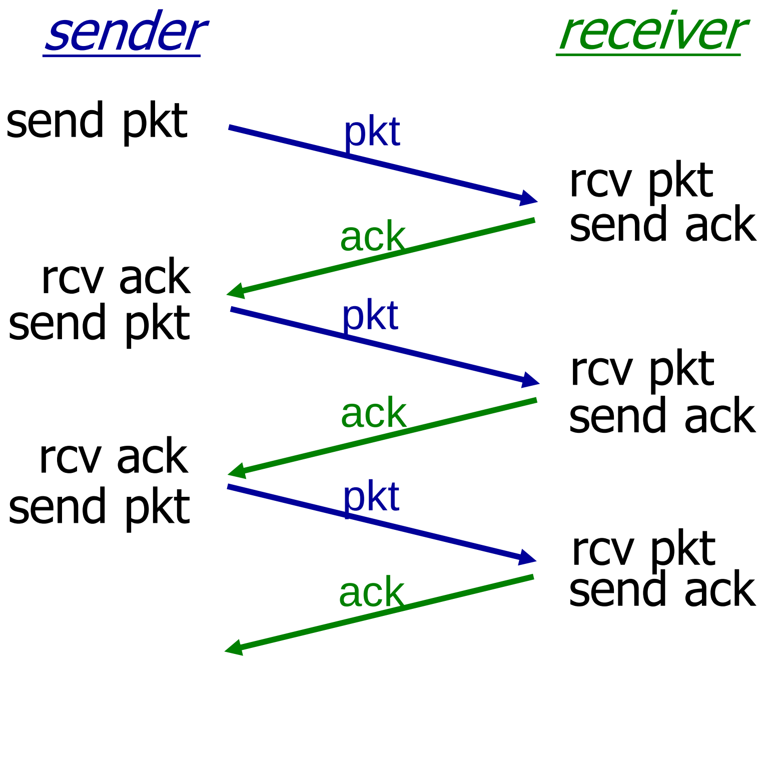
stop and wait

sender sends one packet, then waits for receiver response

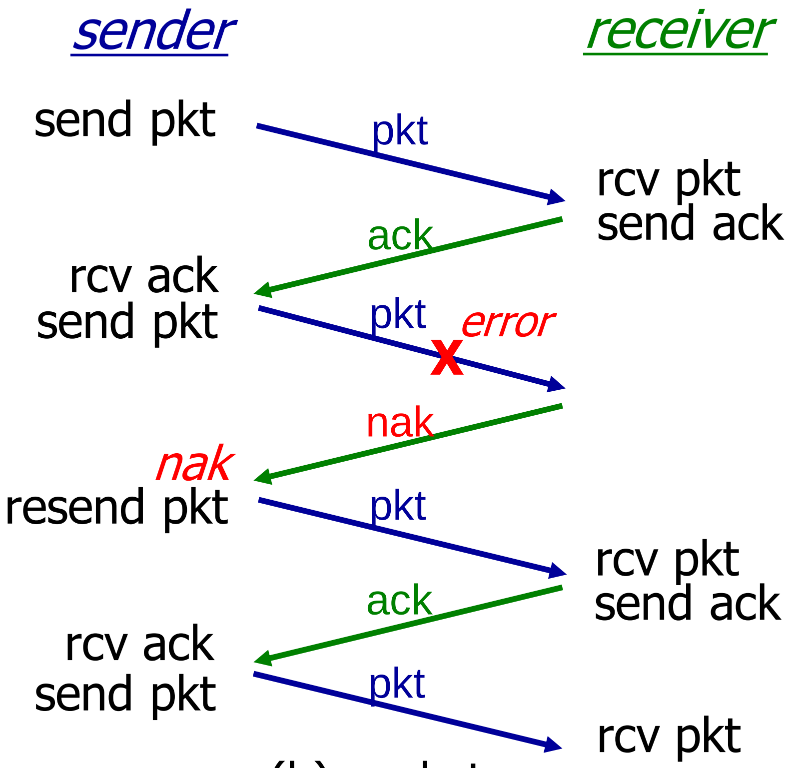
rdt2.0: in action

stop and wait

sender sends one packet, then waits for receiver response, then transmit the next packet



(a) no error

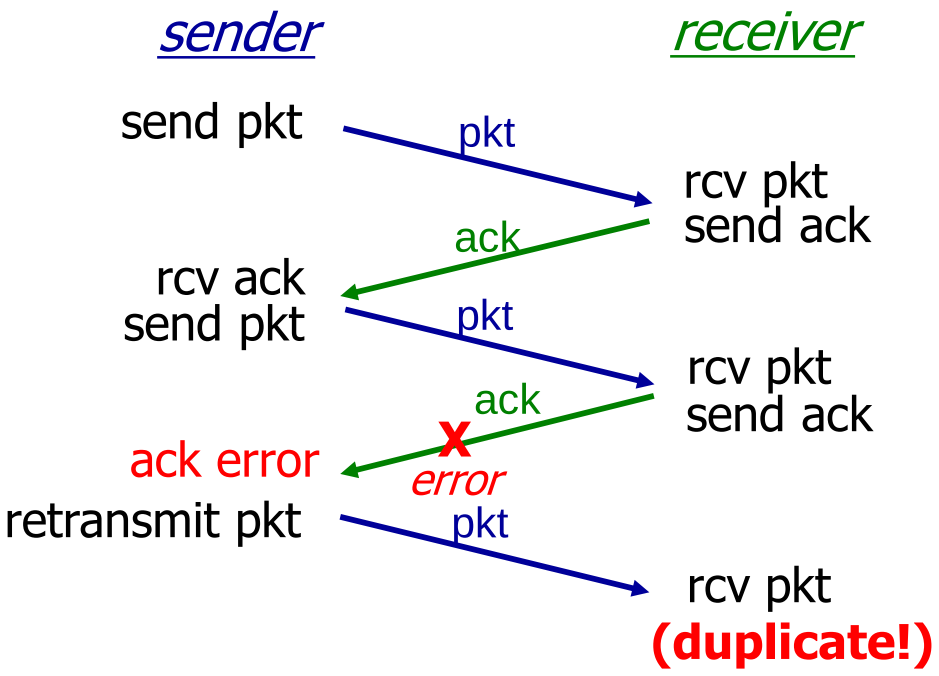


(b) packet error

rdt2.0 has a fatal flaw!

what happens if ACK/NAK corrupted?

- sender doesn't know what happened at receiver!
- can't just retransmit: possible duplicate!



rdt2.1: discussion

sender:

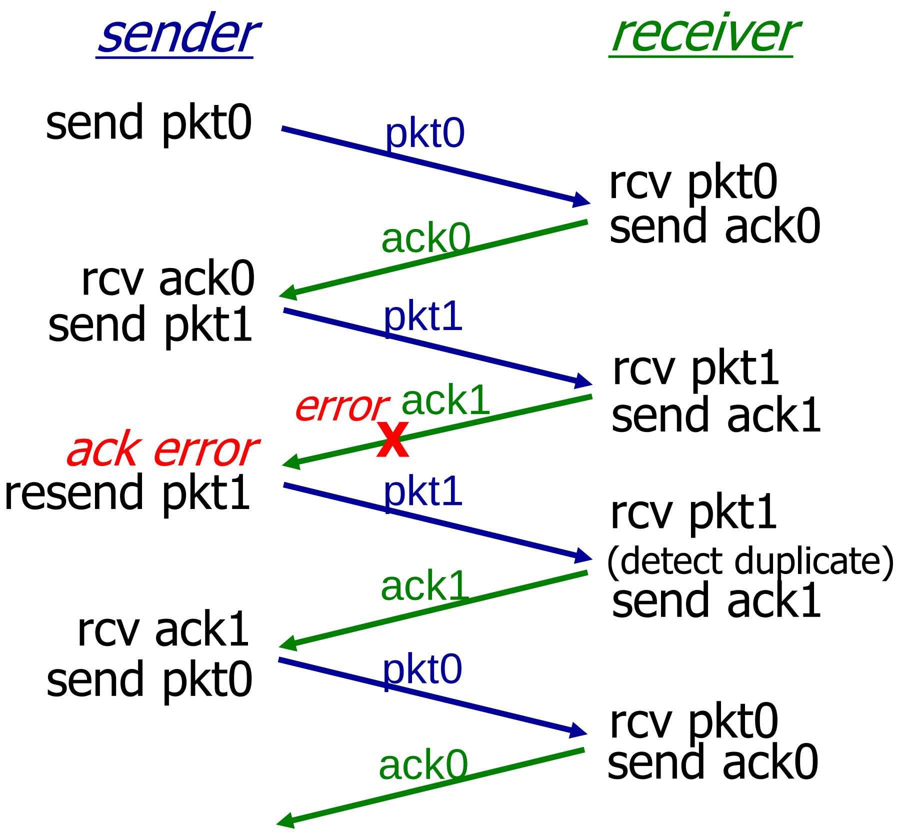
- seq # added to pkt
- two seq. #s (0,1) will suffice. Why?
- must check if received ACK/NAK corrupted
- twice as many states
  - state must "remember" whether "expected" pkt should have seq # of 0 or 1

receiver:

- must check if received packet is duplicate, Old or New
  - state indicates whether 0 or 1 is expected pkt seq #
- note: receiver can *not* know if its last ACK/NAK received OK at sender

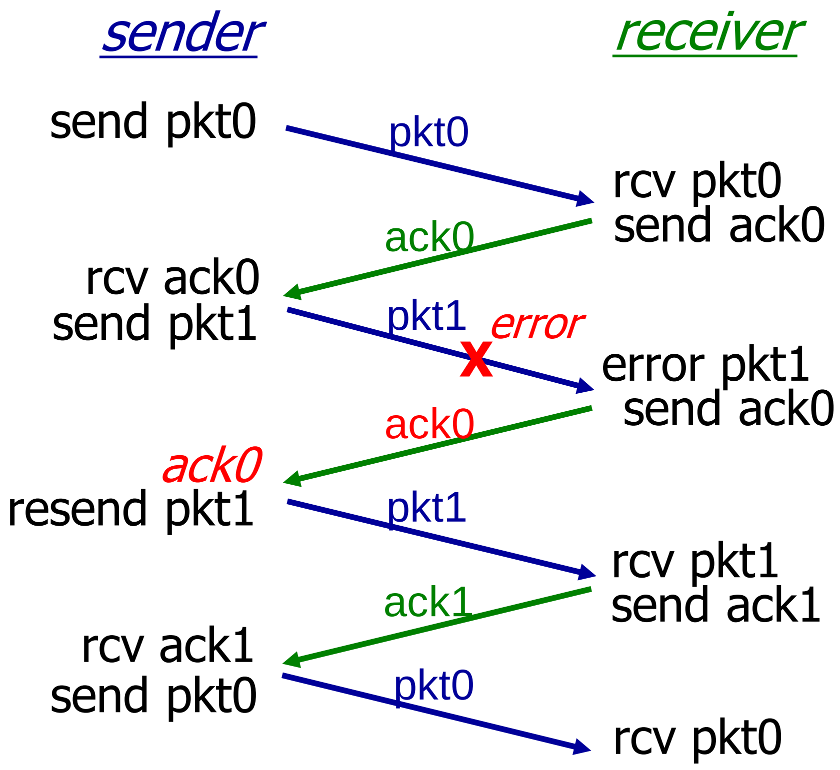
rdt2.1: Add sequence number

- sender adds *sequence number* to each pkt
- receiver discards (doesn't deliver up) duplicate pkt



rdt2.2: a NAK-free protocol

- same functionality as rdt2.1, using ACKs only
- instead of NAK, receiver sends ACK for last pkt received OK
  - receiver must *explicitly* include seq # of pkt being ACKed
- Duplicate ACK at sender results in same action as NAK: *retransmit current pkt*



Duplicate ACK

As we will see, TCP uses this approach to be NAK-free

rdt3.0: channels with errors *and* loss

New channel assumption: underlying channel can also *lose* packets (data, ACKs)

- checksum, sequence #s, ACKs, retransmissions will be of help … but not quite enough

Q: How do *humans* handle lost sender-to-receiver words in conversation?

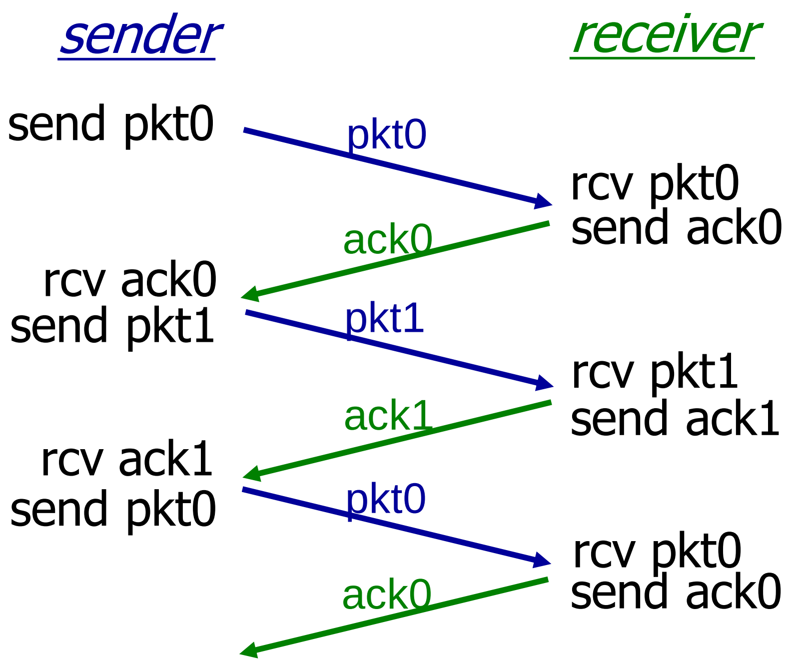
rdt3.0: channels with errors *and* loss

Approach: sender waits "reasonable" amount of time for ACK

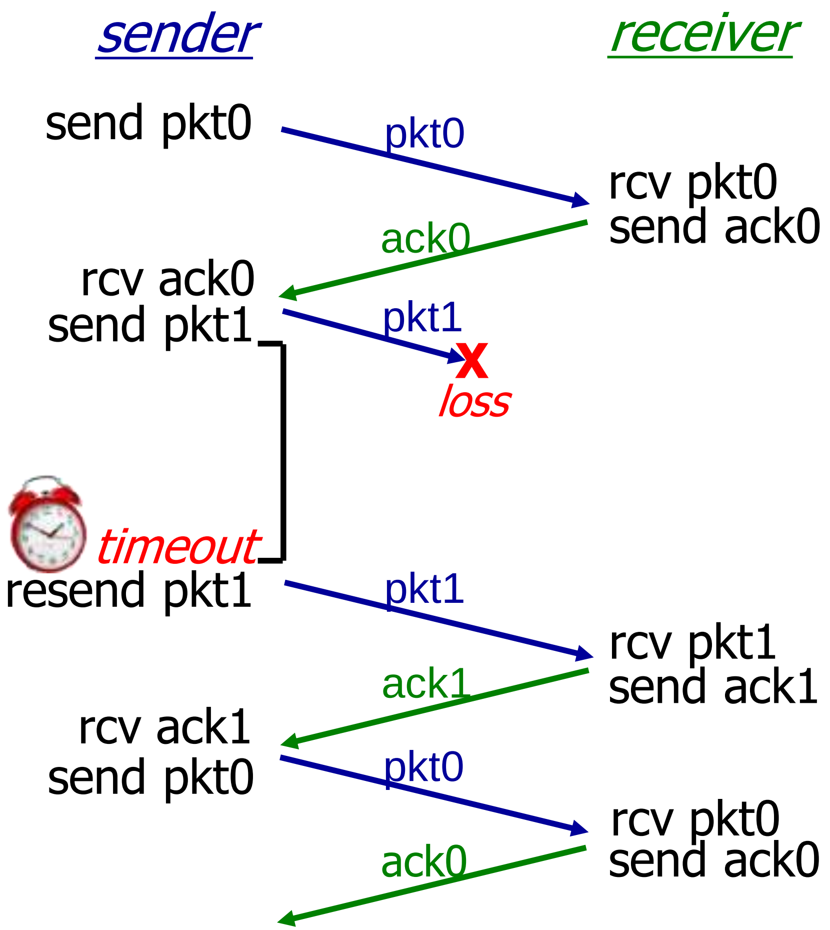
- retransmits if no ACK received in this time
- if pkt (or ACK) just delayed (not lost):
  - retransmission will be duplicate, but seq #s already handles this!
  - receiver must specify seq # of packet being ACKed
- use countdown timer to interrupt after "reasonable" amount of time

timeout

rdt3.0 in action

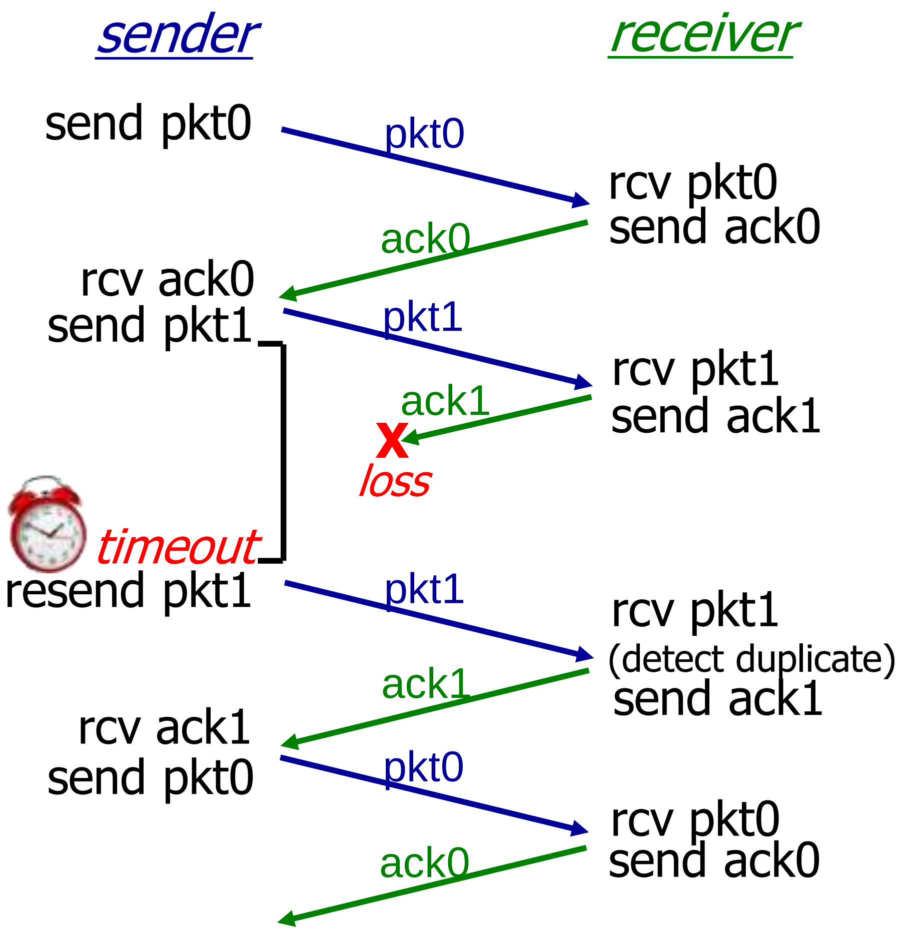


(a) no loss

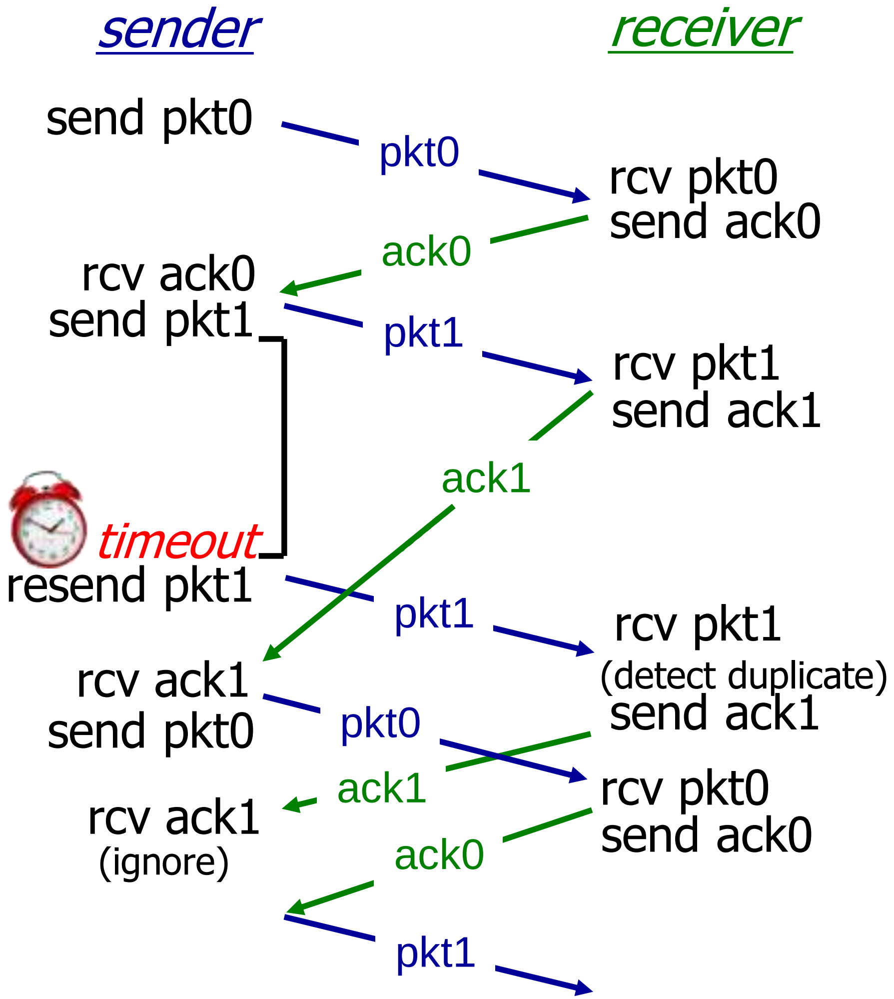


(b) packet loss

rdt3.0 in action



(c) ACK loss



(d) premature timeout/ delayed ACK

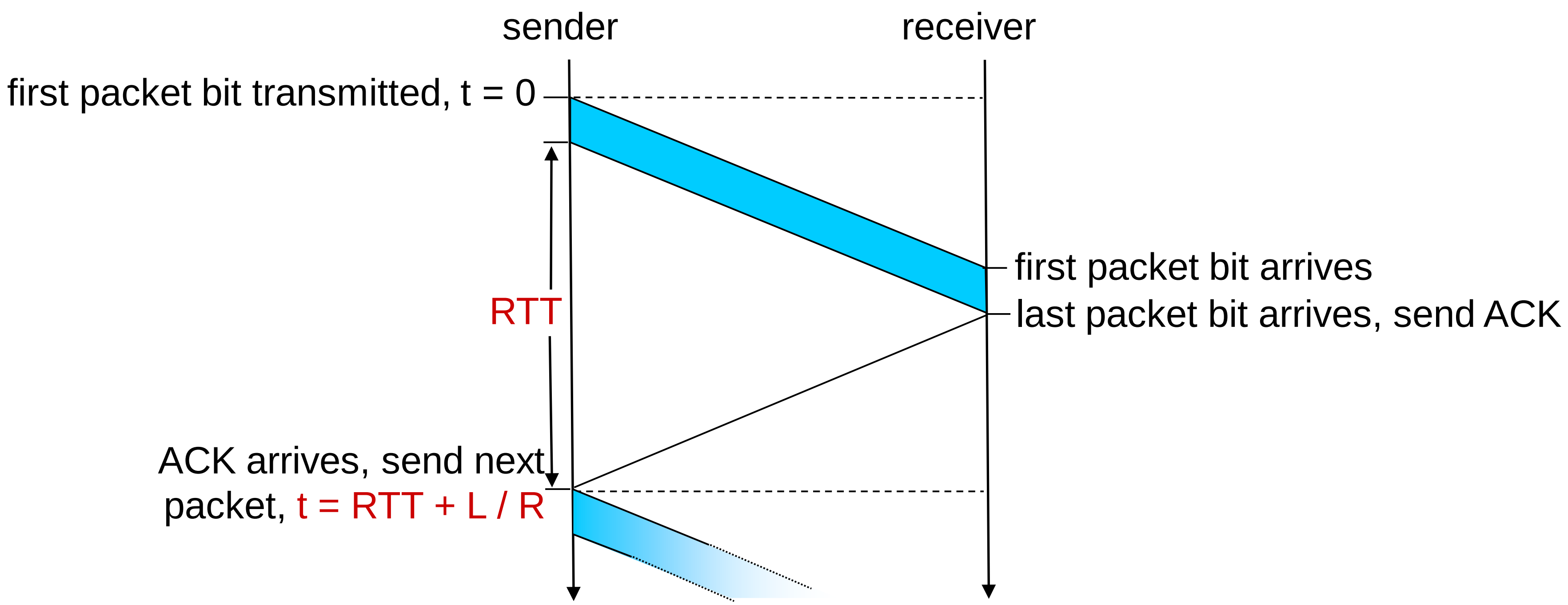
Performance of rdt3.0 (stop-and-wait)

- U_{sender}: *utilization* – fraction of time sender busy sending

- example: 1 Gbps link, 15 ms prop. delay, 8000 bit packet
  - time to transmit packet into channel:

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

rdt3.0: stop-and-wait operation

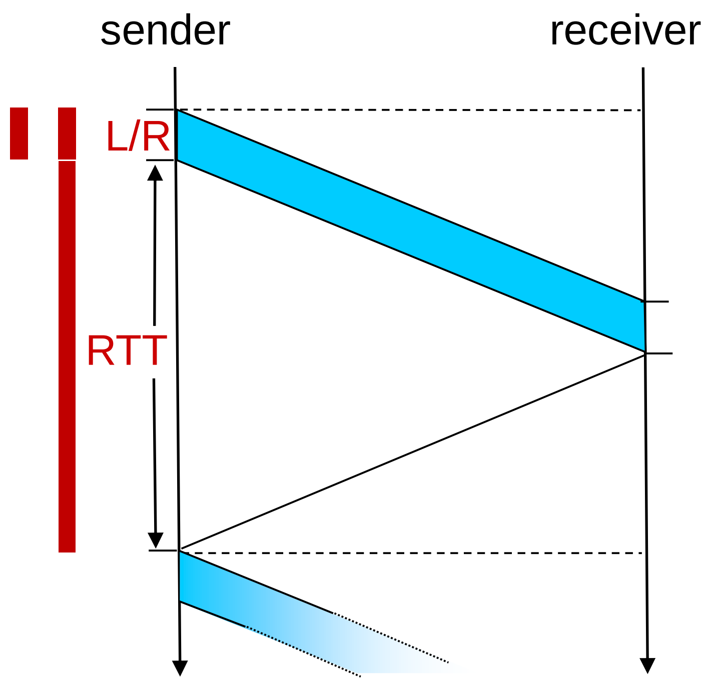


rdt3.0: stop-and-wait operation

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

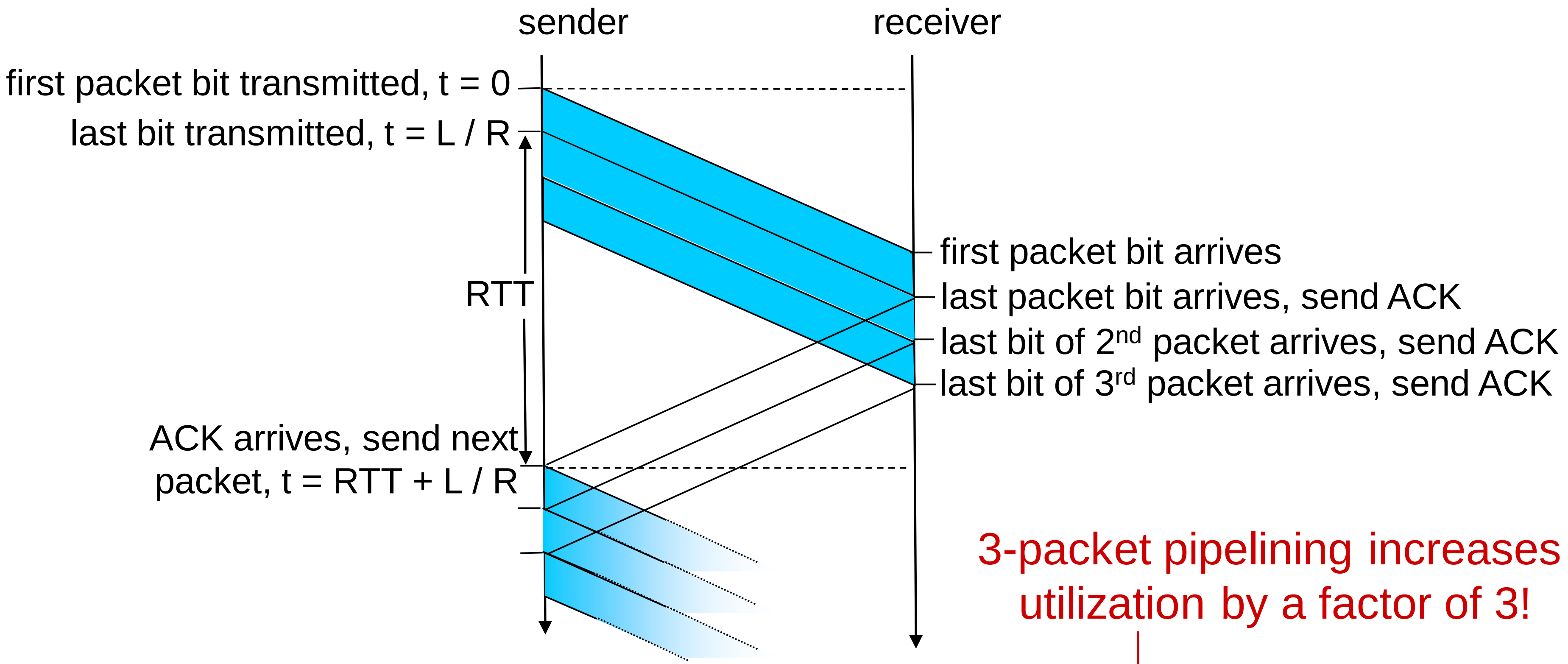
$$= \frac{.008}{30.008}$$

$$= 0.00027$$



- Achieved Throughput = 1Gbps * 0.00027 = 0.27Mbps!
- rdt 3.0 protocol performance stinks!
- Protocol limits performance of underlying infrastructure (channel)

Pipelining: increased utilization



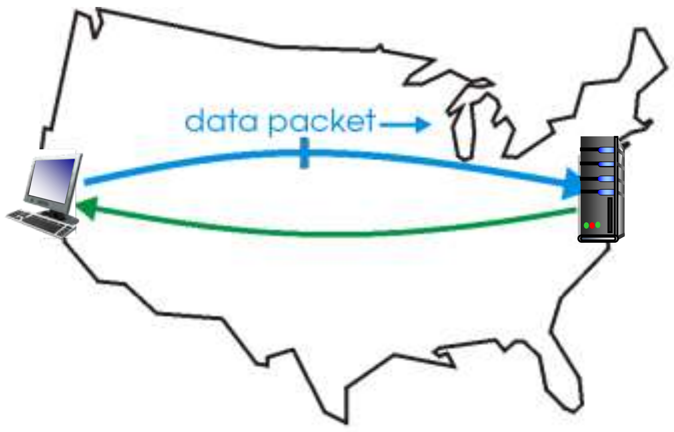
3-packet pipelining increases utilization by a factor of 3!

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

rdt3.0: pipelined protocols operation

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

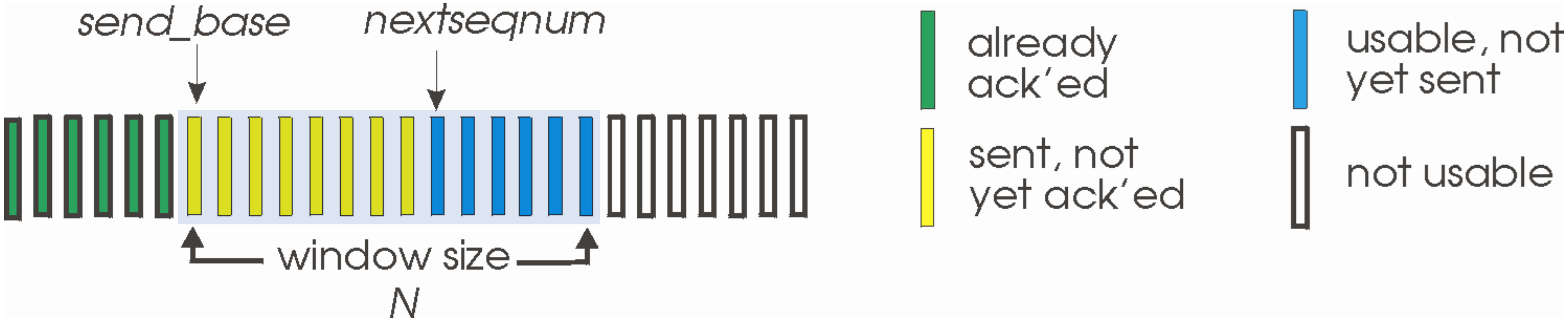
- range of sequence numbers must be increased
- buffering at sender and/or receiver



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

Go-Back-N: sender

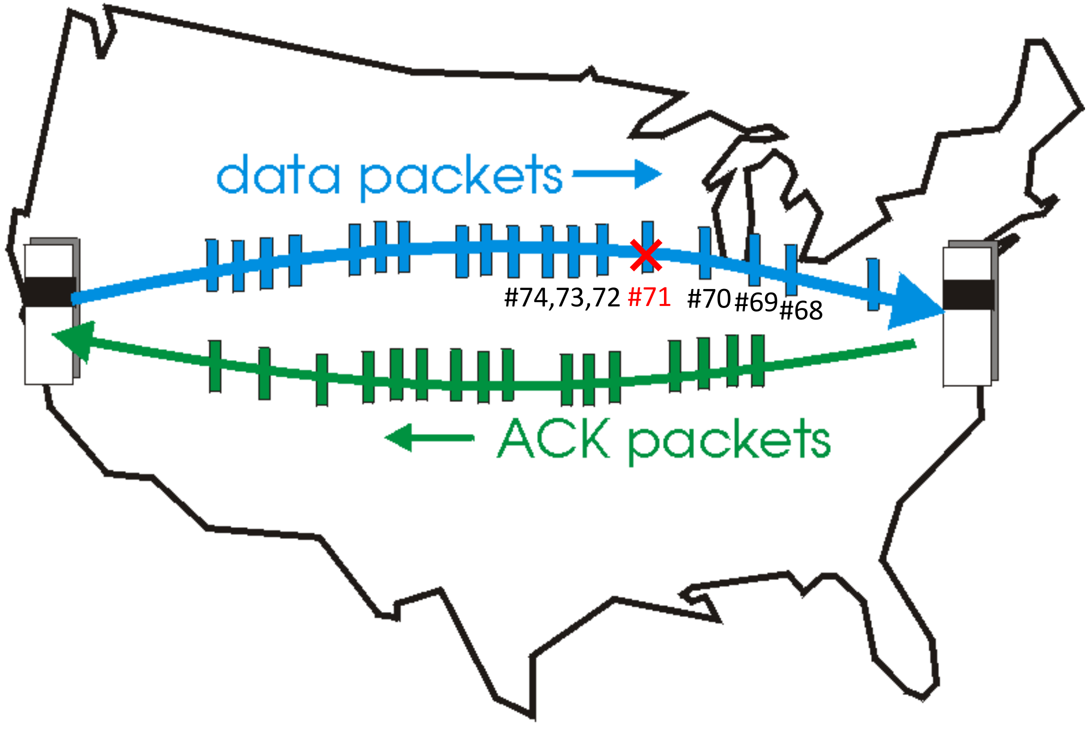
- sender: “window” of up to N, consecutive transmitted but unACKed pkts
  - k-bit seq # in pkt header



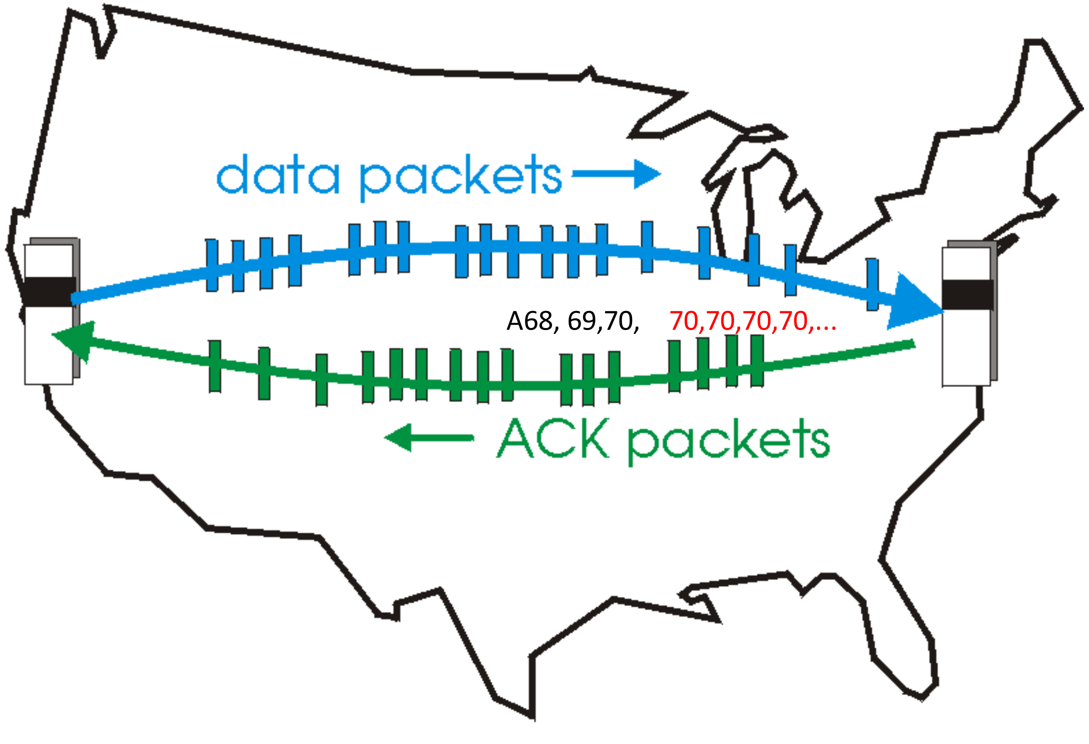
- *cumulative ACK:* ACK(*n*): ACKs all packets up to, including seq # *n*
  - on receiving ACK(*n*): move window forward to begin at *n+1*
- timer for oldest in-flight packet
- *timeout(n):* retransmit packet n and all higher seq # packets in window

Go-Back-N: receiver

- ACK-only: always send ACK for correctly-received packet so far, with highest *in-order* seq #
  - may generate duplicate ACKs

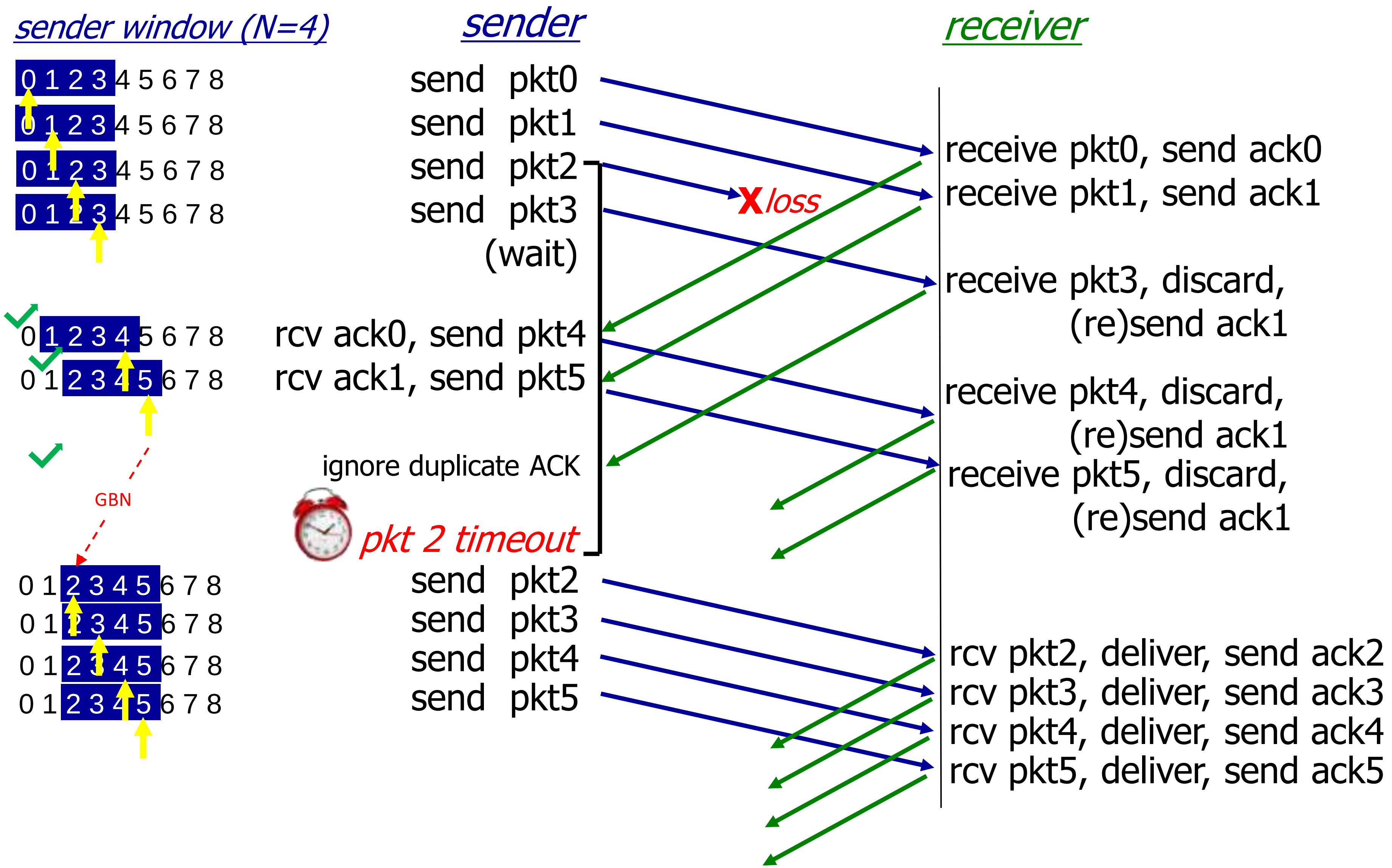


(b) a pipelined protocol in operation



(b) a pipelined protocol in operation

Go-Back-N in action



Selective repeat

- receiver *individually* acknowledges all correctly received packets
  - buffers packets, as needed, for eventual in-order delivery to upper layer
- sender times-out/retransmits individually for unACKed packets
  - sender maintains timer for each unACKed pkt
- sender window
  - *N* consecutive seq #s
  - limits seq #s of sent, unACKed packets

Optional – not tested

Week5 summary

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

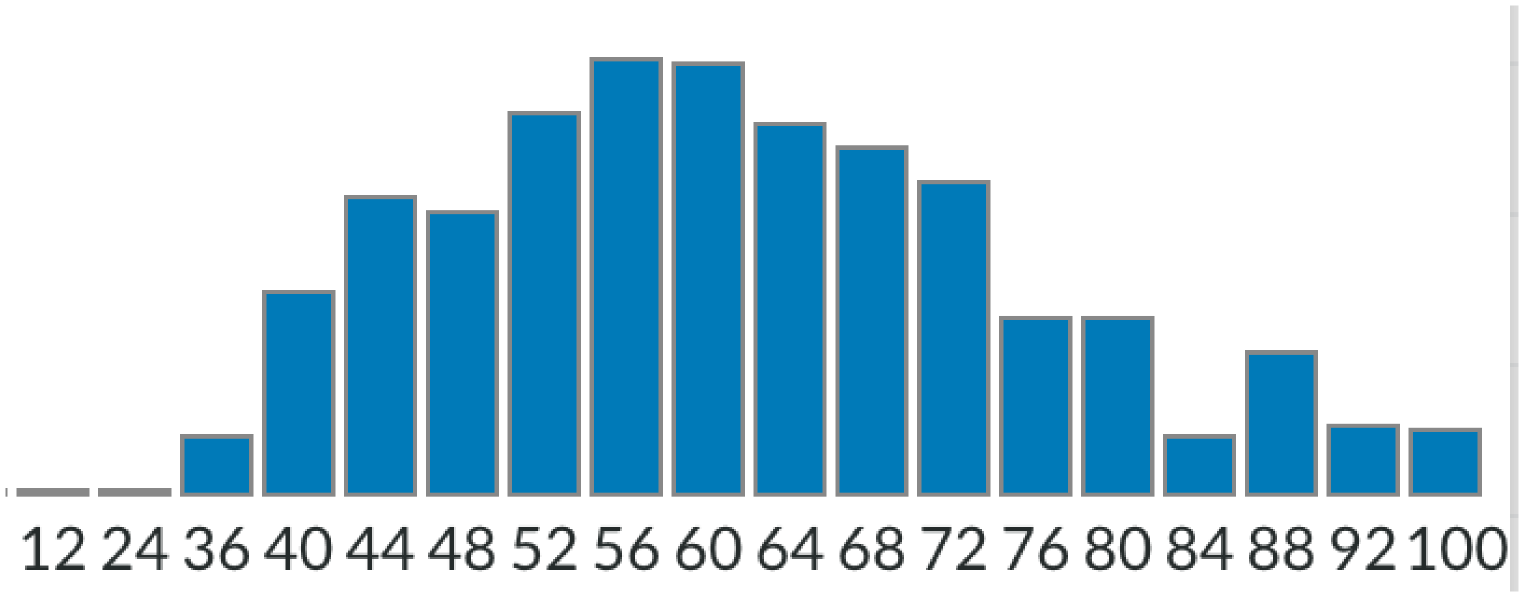
Up next:

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

Project 1

- Project1 – 5%
  - Task Sheet in Canvas Week5 module
  - example web page: simpleWeb.html
- Project1 due in Week5 in your lab class
  - Group work, individually assessed.
  - demonstrate running code as a group – group work
  - show your individual work – individual assessment
  - Please complete UDP lab while tutor is marking other groups

Quiz results



- Quiz Results
  - Average: 73
  - Pass: 718 (86%)
- Quiz answers will NOT be released
  - similar questions appeared in Assignment1 and Practice Quiz
- Weeks 1-4 contents will NOT be tested in final exam!

Lecture done ✓

- Q & A