Computer Networks and Applications

COMP 3331/COMP 9331

Week 4

Transport Layer Part 1

Reading Guide:
Chapter 3, Sections 3.1 – 3.5

Transport layer: overview

Our goal:

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

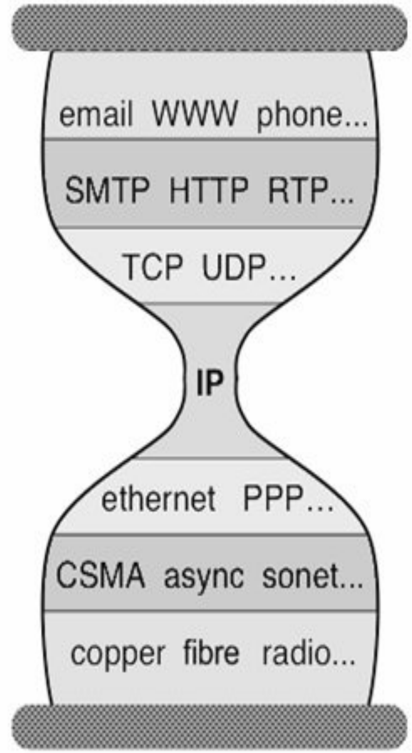
Transport layer: roadmap

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

Transport layer

- Moving "down" a layer

- Current perspective:
  - Application layer is the boss….
  - Transport layer usually executing within the OS Kernel
  - The network layer is ours to command !!

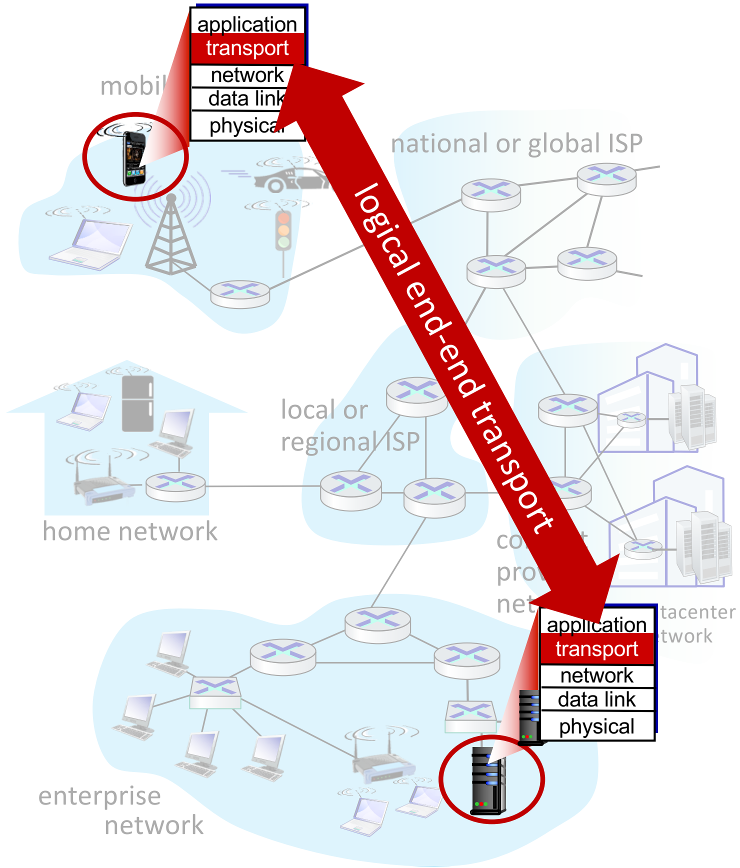


Network layer (some context)

- What it does: finds paths through network
  - Routing from one end host to another
- What it doesn't:
  - Reliable transfer: "best effort delivery"
  - Guarantee paths
  - Arbitrate transmission rates
- For now, think of the network layer as giving us an "API" with one function: *sendtohost(data, host)*
  - Promise: the data will go to that (usually!!)

Transport services and protocols

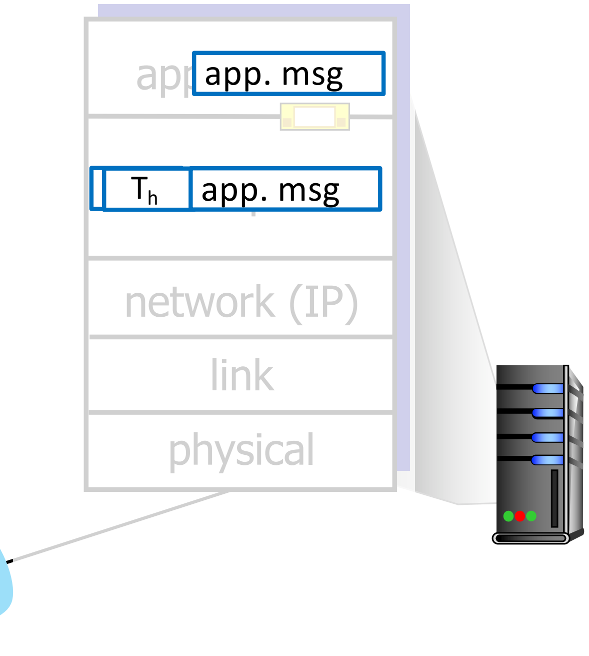
- provide *logical communication* between application processes running on different hosts
- transport protocols actions in end systems:
  - sender: breaks application messages into *segments*, passes to network layer
  - receiver: reassembles segments into messages, passes to application layer
- two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



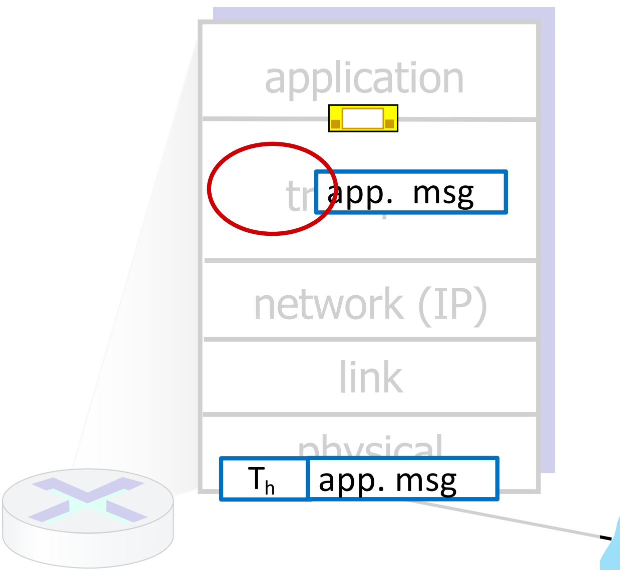
Transport Layer Actions

Sender:

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

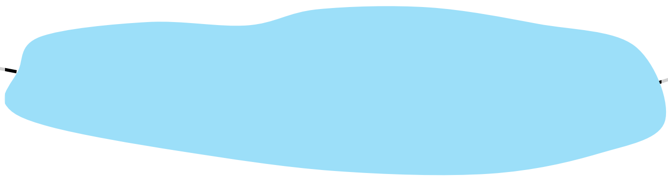


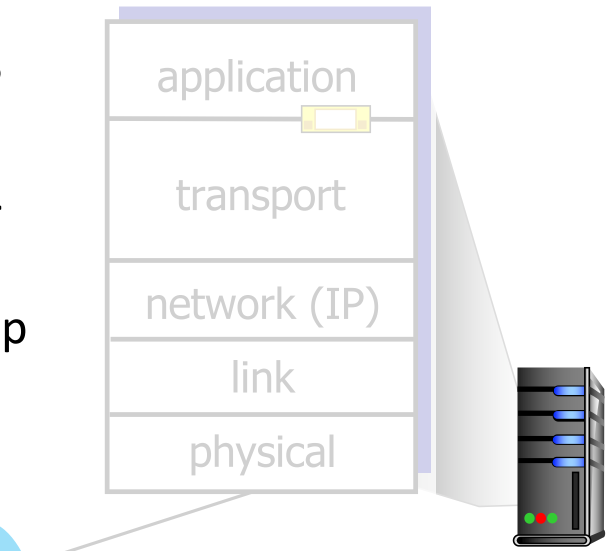
Transport Layer Actions



Receiver:

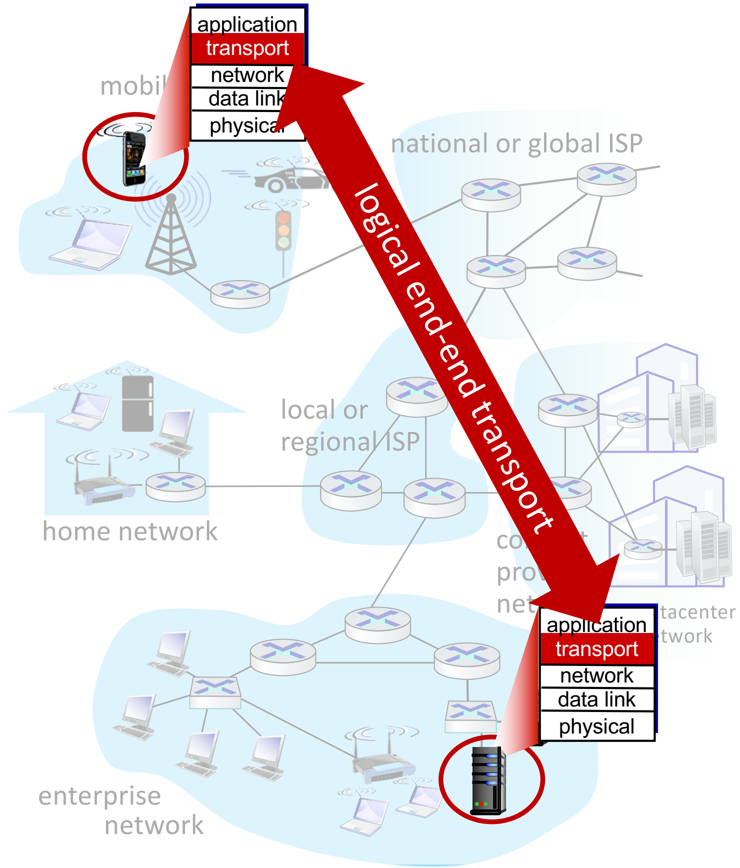
- receives segment from IP
- checks header values
- extracts application-layer message
- demultiplexes message up to application via socket





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



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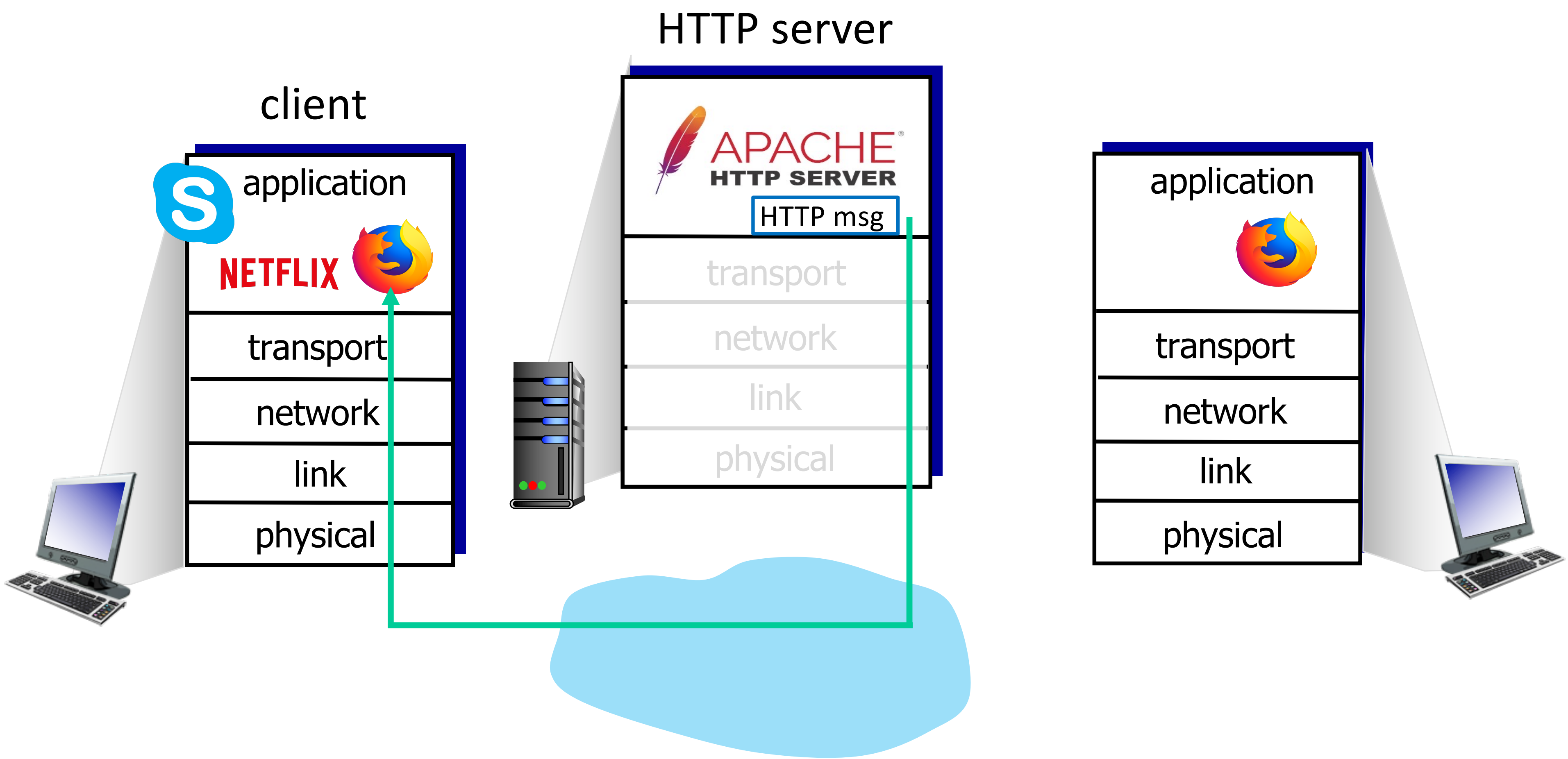
HTTP server
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NETFLIX
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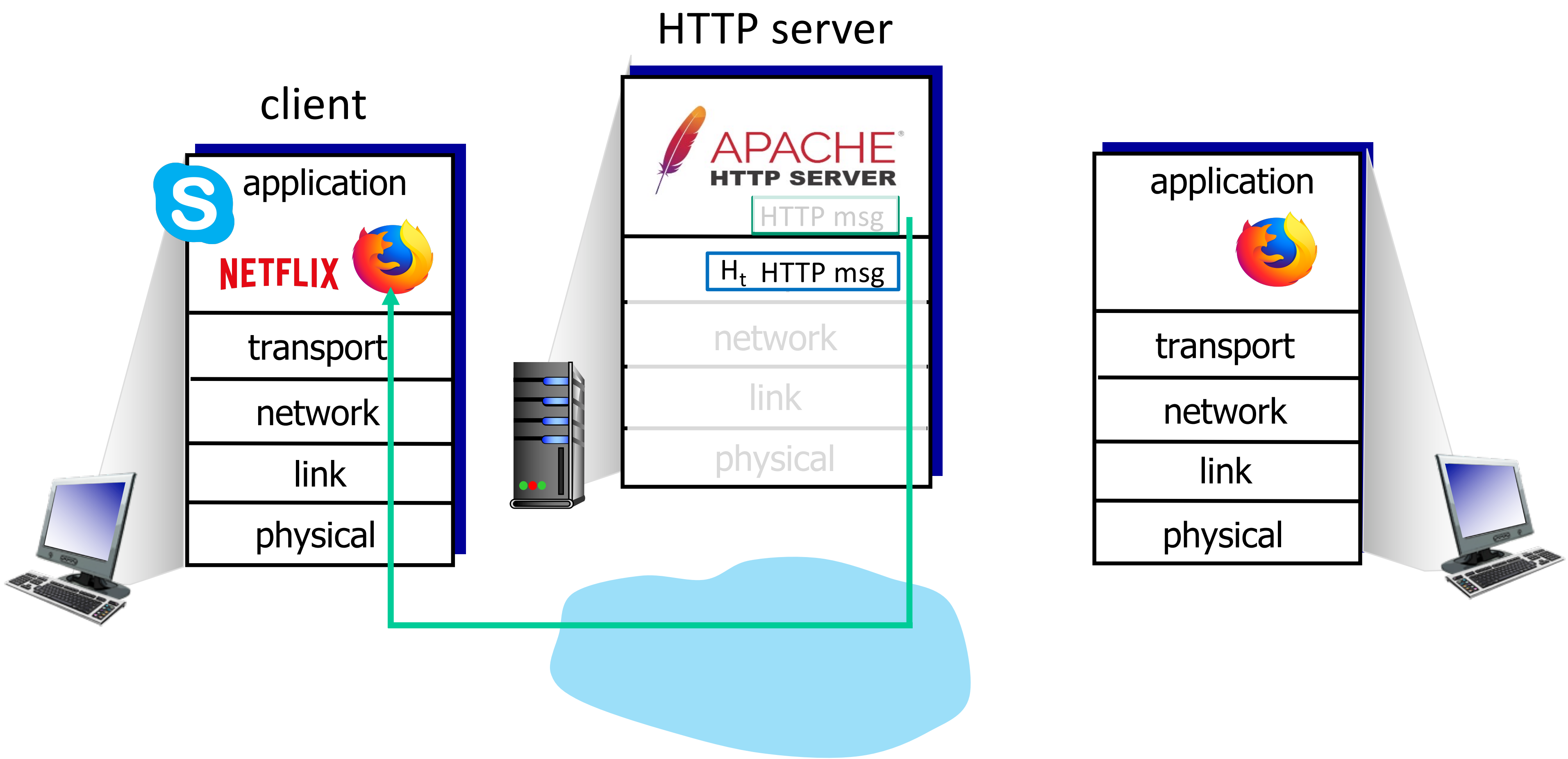
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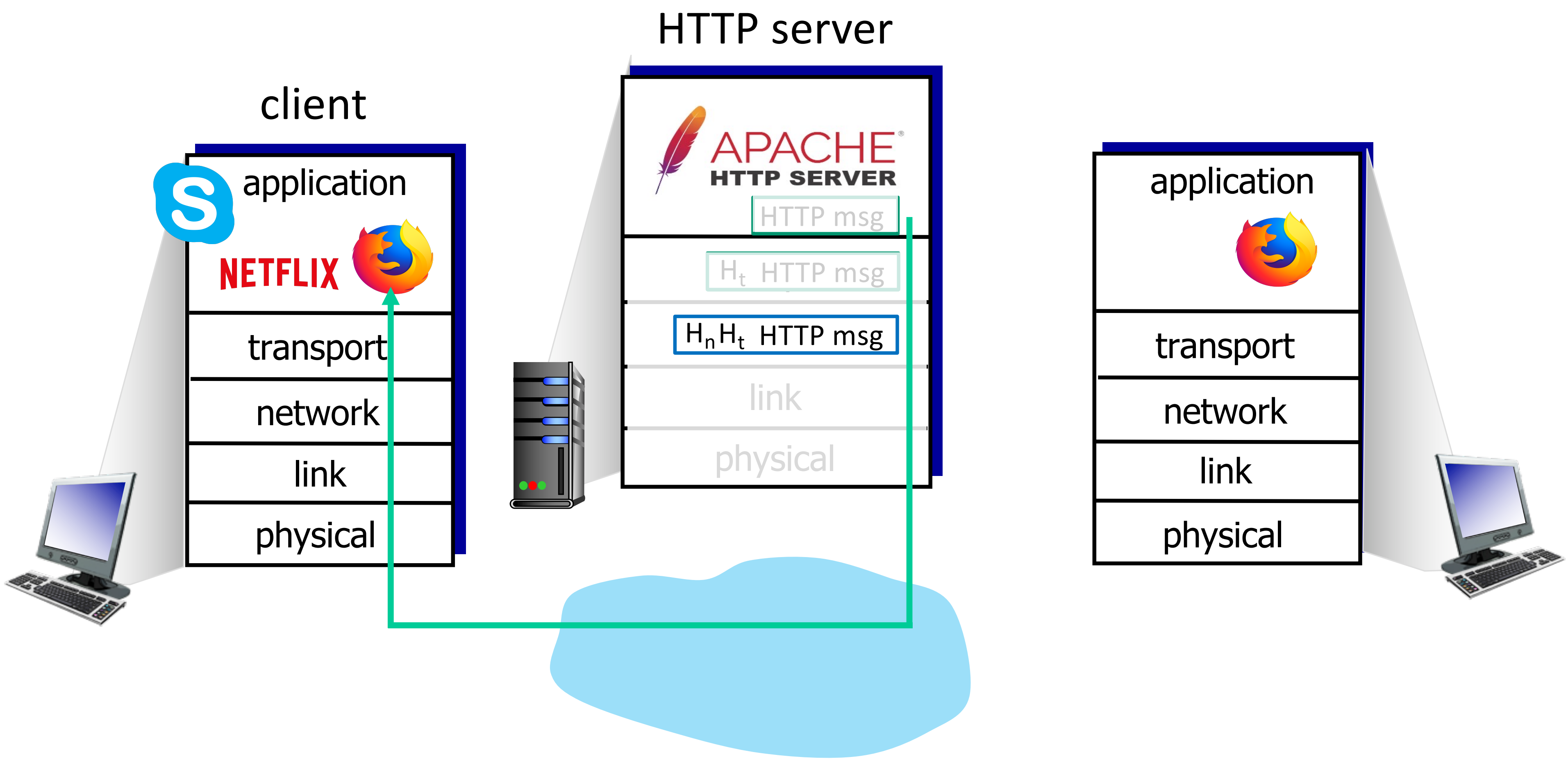

HTTP server
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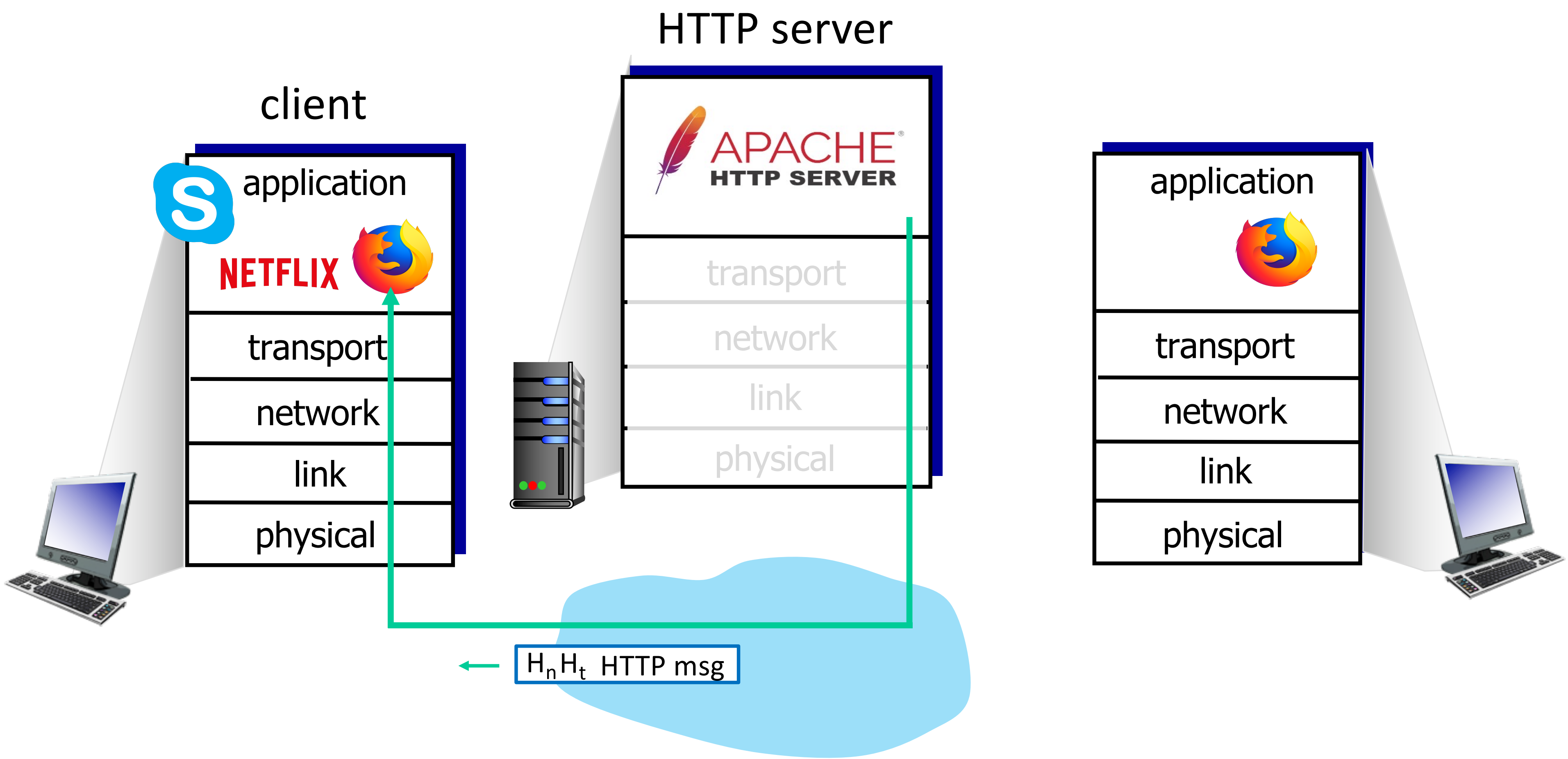

HTTP server
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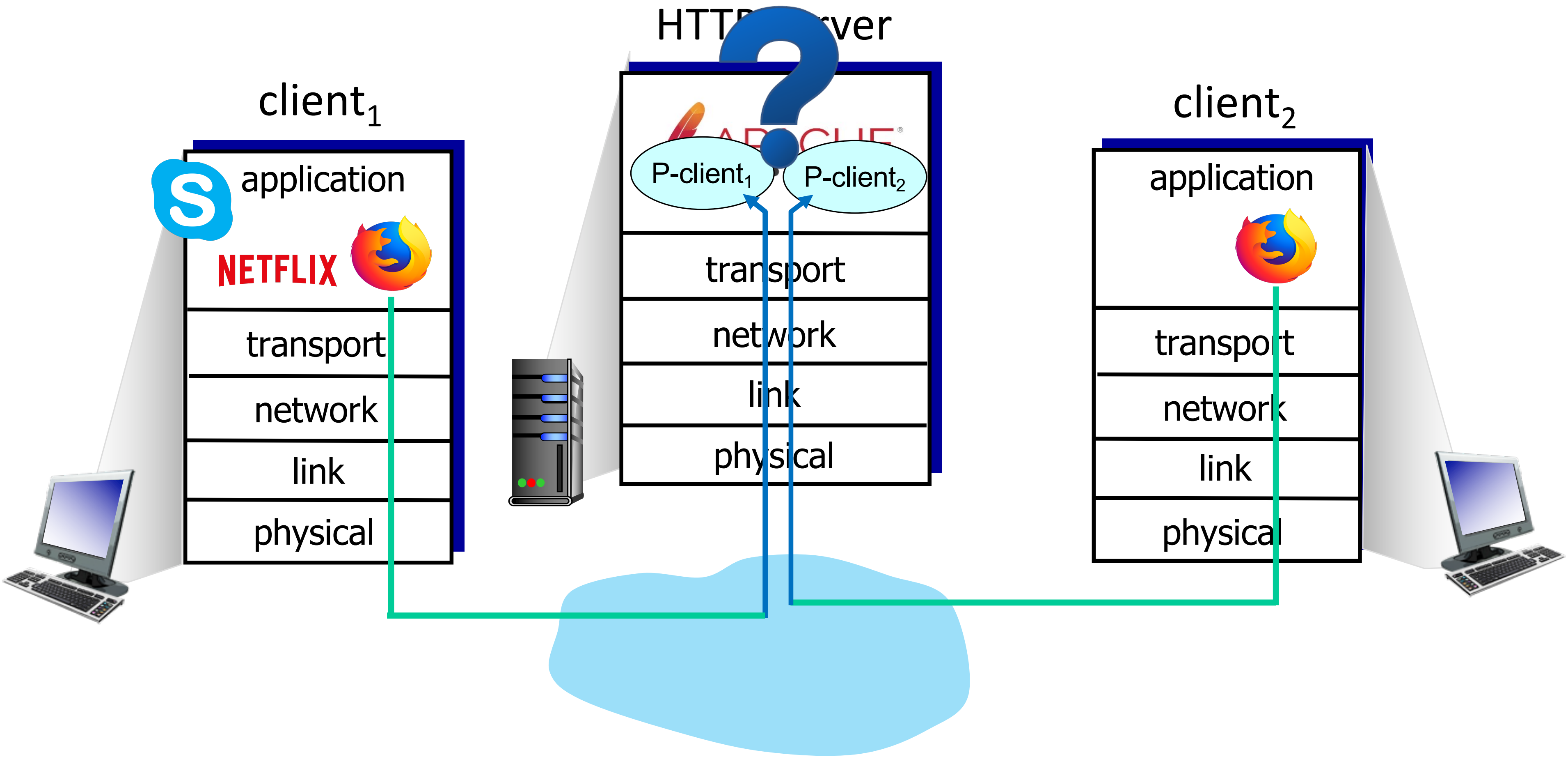
$H_n H_t$ HTTP msg


client1
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P-client1
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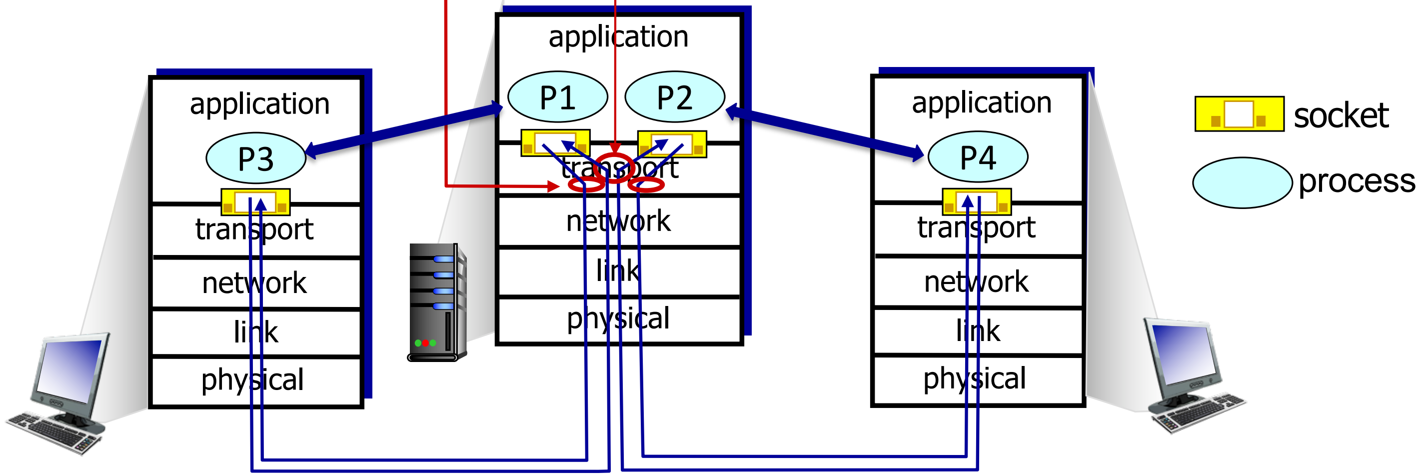
Multiplexing/demultiplexing

multiplexing at sender:

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

demultiplexing at receiver:

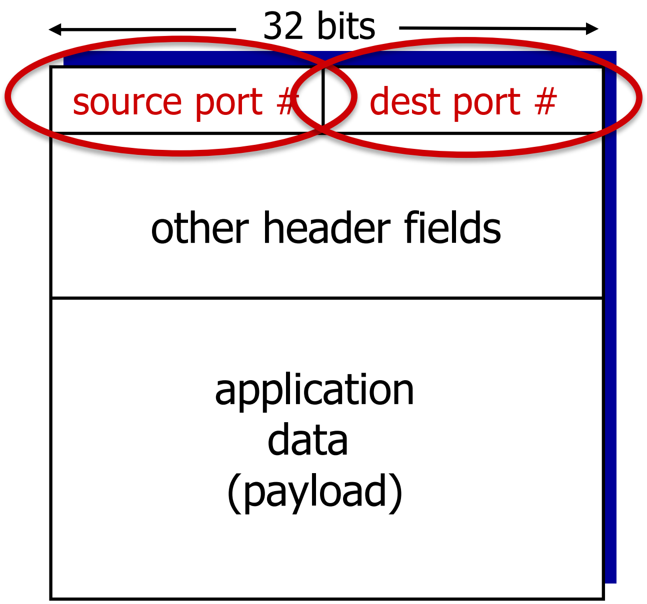
use header info to deliver received segments to correct socket



Note: The network is a shared resource. It does not care about your applications, sockets, etc.

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 # (or let OS pick random available 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 #

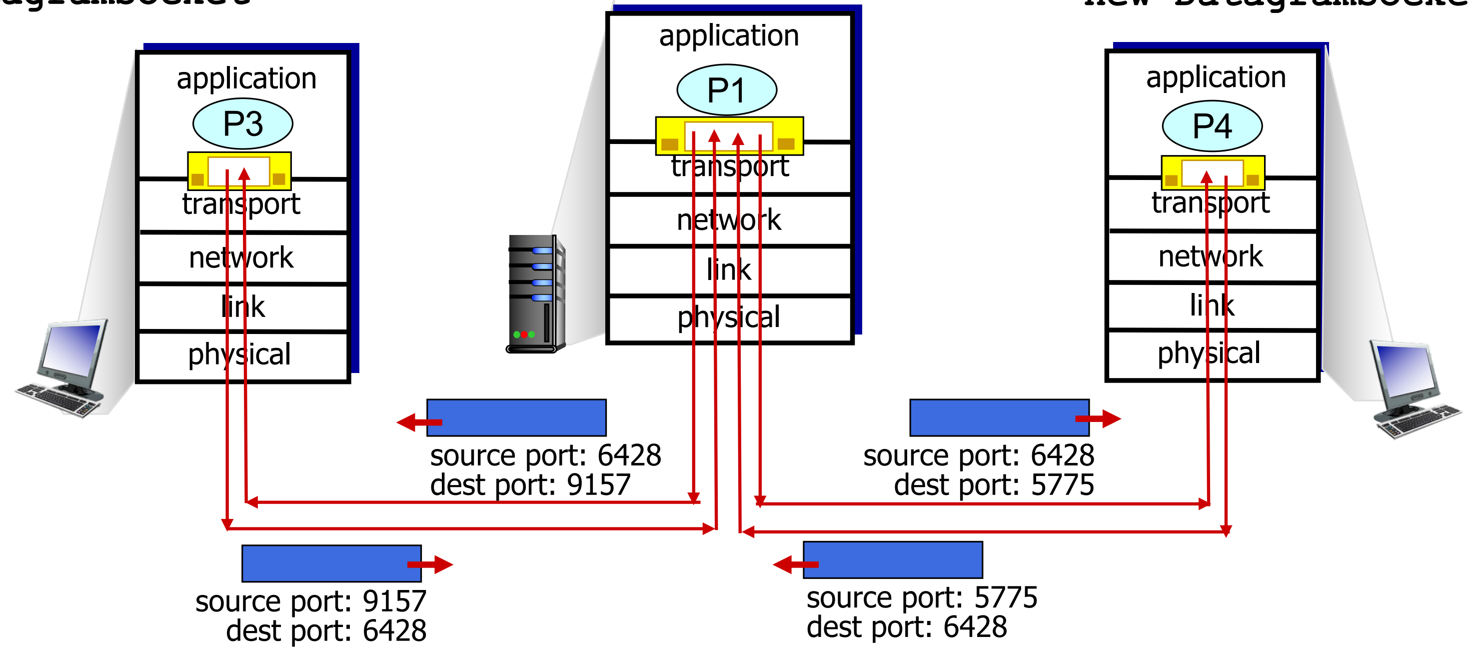
IP/UDP datagrams with *same dest. port #,* but different source IP addresses and/or source port numbers will be directed to *same socket* at receiving host

Connectionless demultiplexing: an example

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

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

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

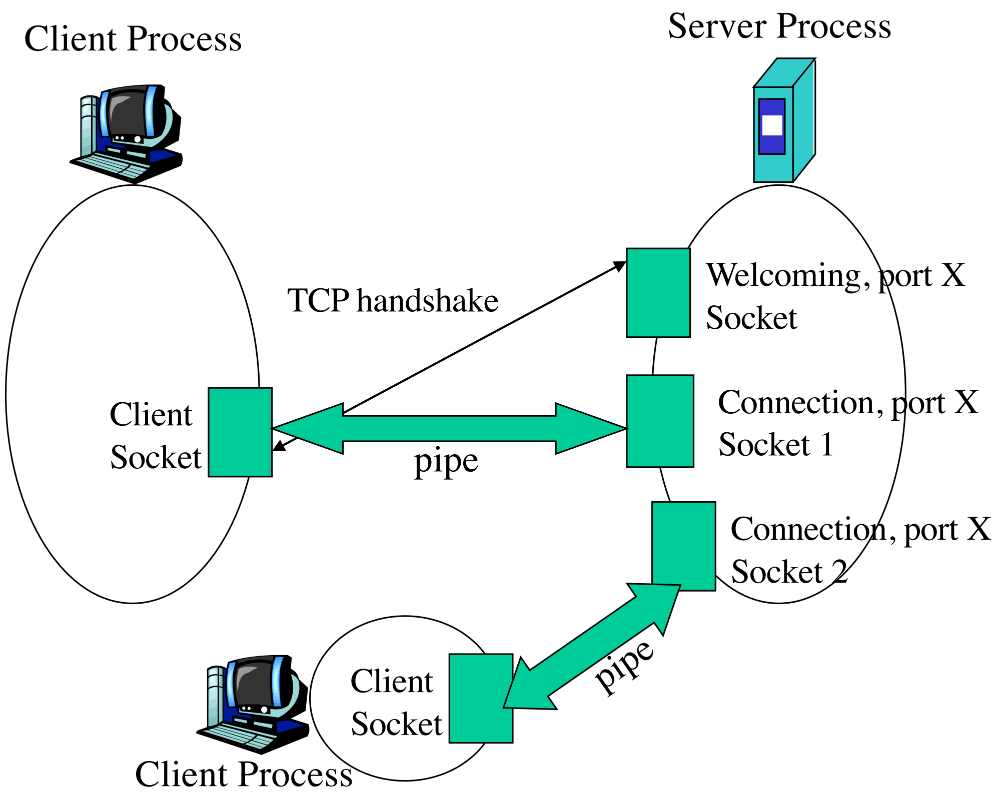


Connection-oriented demultiplexing

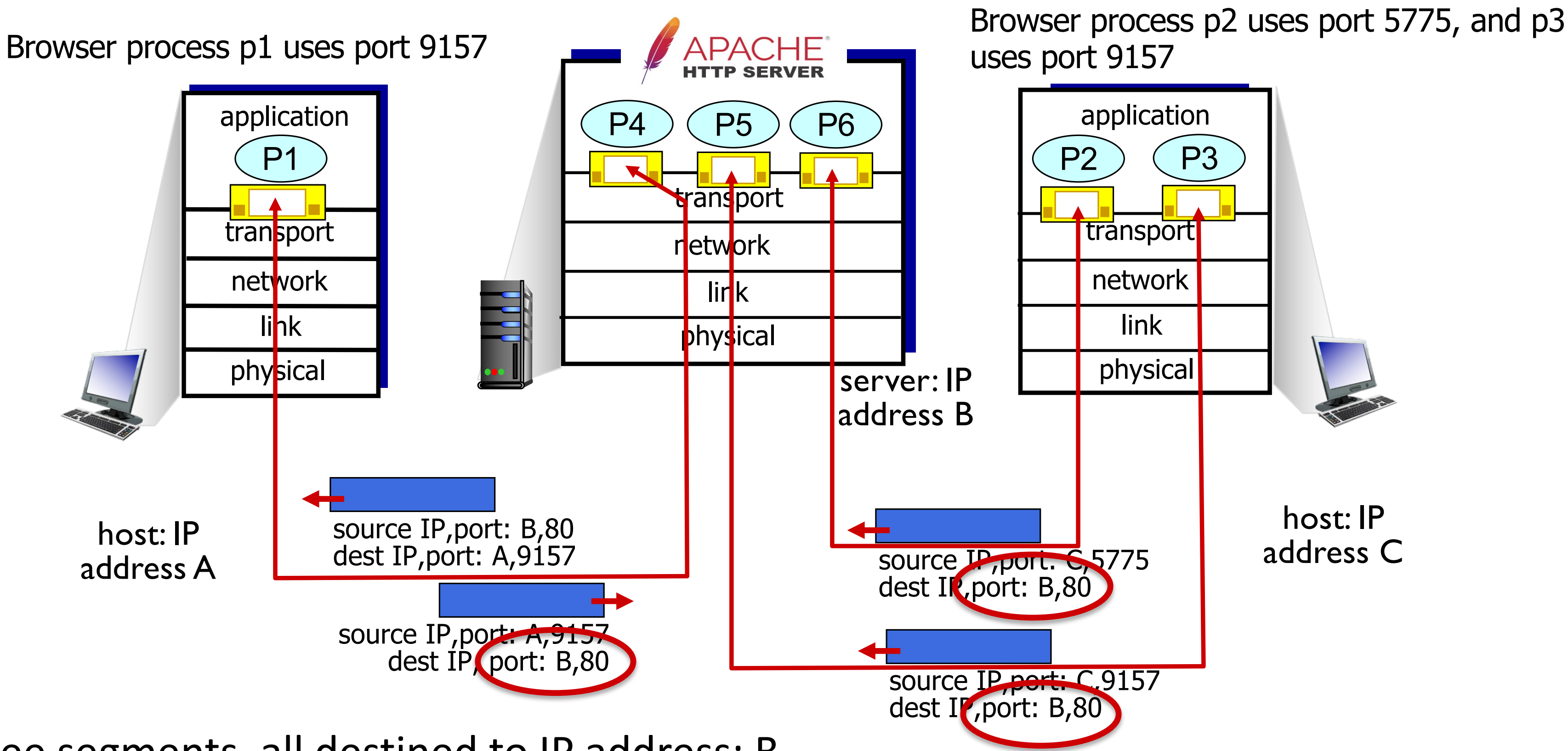
- TCP socket identified by 4-tuple:
  - source IP address
  - source port number
  - dest IP address
  - dest port number
- demux: receiver uses *all four values (4-tuple)* to direct segment to appropriate socket

- server may support many simultaneous TCP sockets:
  - each socket identified by its own 4-tuple
  - each socket associated with a different connecting client

Revisiting TCP Sockets



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 (more later in the course)

May I scan your ports?

http://netsecurity.about.com/cs/hackertools/a/aa121303.htm

- Servers wait at open ports for client requests
- Hackers often perform *port scans* to determine open, closed and unreachable ports on candidate victims
- Several ports are well-known
  - <1024 are reserved for well-known apps
  - Other apps also use known ports
    - MS SQL server uses port 1434 (udp)
    - Sun Network File System (NFS) 2049 (tcp/udp)
- Hackers can exploit known flaws with these known apps
  - Example: Slammer worm exploited buffer overflow flaw in the SQL server
- How do you scan ports?
  - Nmap, Superscan, etc

http://www.auditmypc.com/

https://www.grc.com/shieldsup

Quiz: UDP Sockets

Suppose we use UDP instead of TCP for communicating with a web server where all requests and responses fit in a single UDP segment. Suppose 100 clients are simultaneously communicating with this web server. How many sockets are respectively active at the server and each client?

a) 1, 1
b) 2, 1
c) 200, 2
d) 100, 1
e) 101, 1

ANSWER: a)
A UDP socket does not keep any information about the other end point, see slide 19

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Quiz: TCP Sockets

Suppose 100 clients are simultaneously communicating with a traditional HTTP/TCP web server. How many sockets are active respectively at the server and each client?

a) 1, 1
b) 2, 1
c) 200, 2
d) 100, 1
e) 101, 1

ANSWER: d) or e) depending on whether a welcoming socket is counted as a socket

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Quiz: TCP Sockets

Suppose 100 clients are simultaneously communicating with a traditional HTTP/TCP web server. Do all the TCP sockets at the server have the same server-side port number?

a) Yes
b) No

Answer: a), slide 22

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Transport layer: roadmap

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

- Applications that use UDP:
  - 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

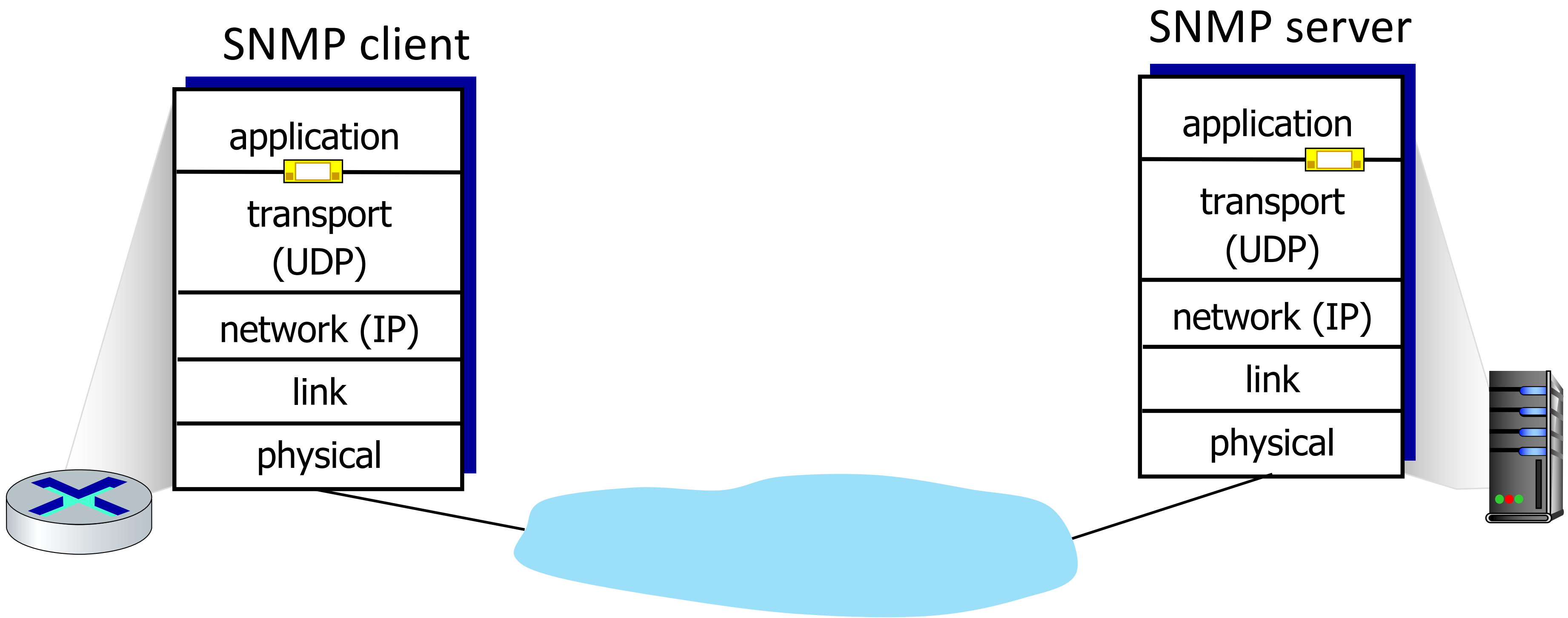
This User Datagram Protocol (UDP) is defined to make available a datagram mode of packet-switched computer communication in the environment of an interconnected set of computer networks. This protocol assumes that the Internet Protocol (IP) [1] is used as the underlying protocol.

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].

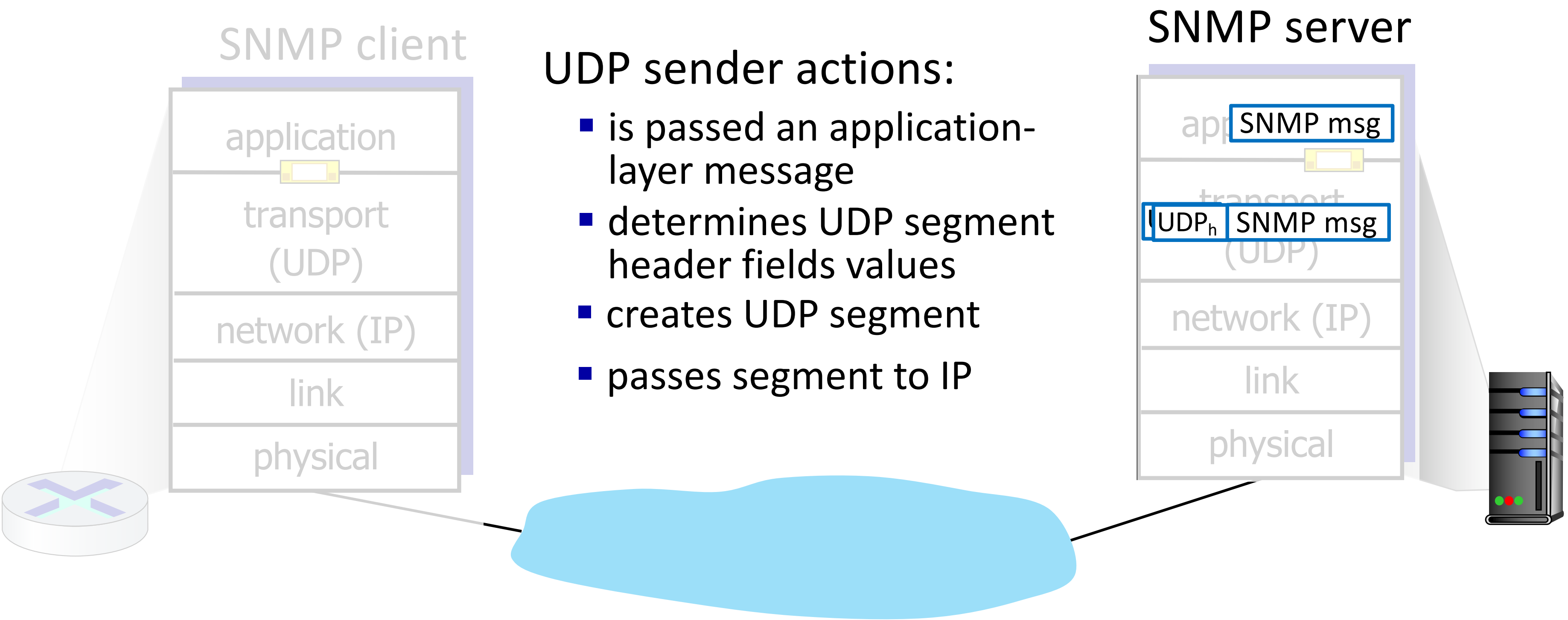
Format

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

UDP: Transport Layer Actions



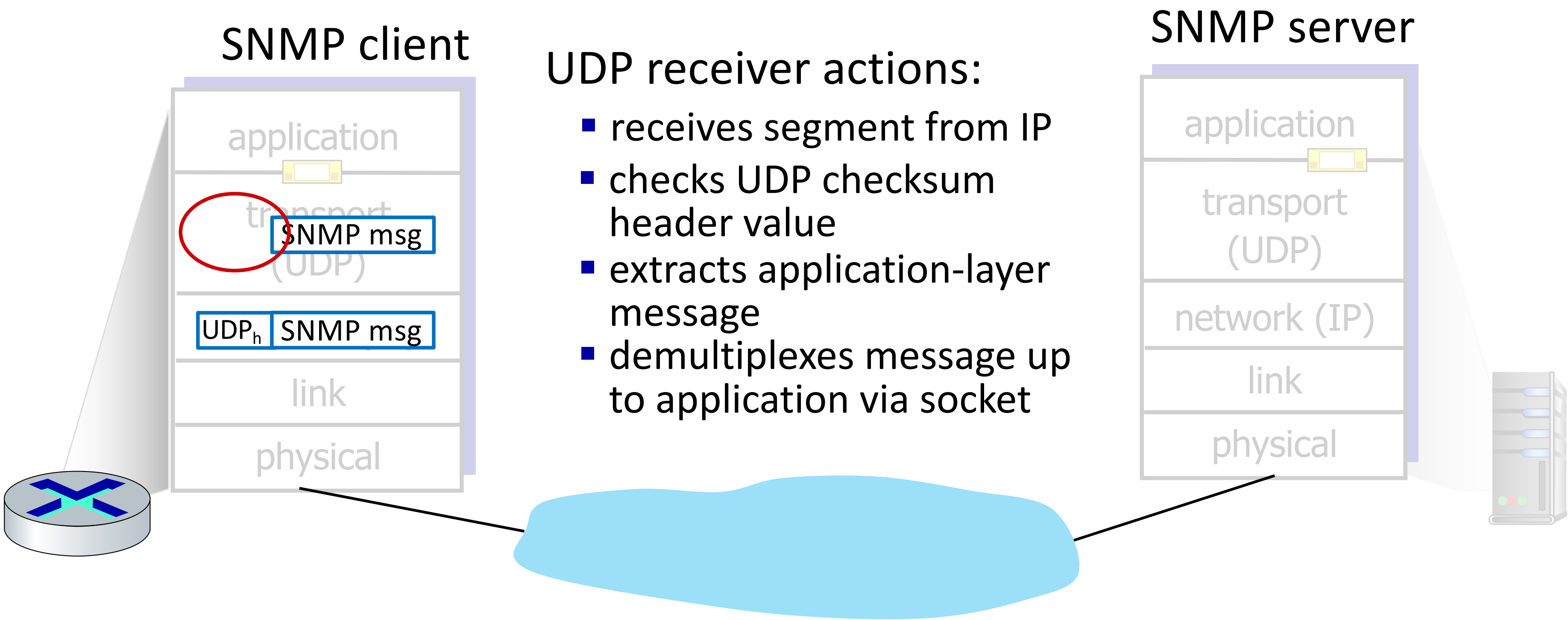
UDP: Transport Layer Actions



UDP sender actions:

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

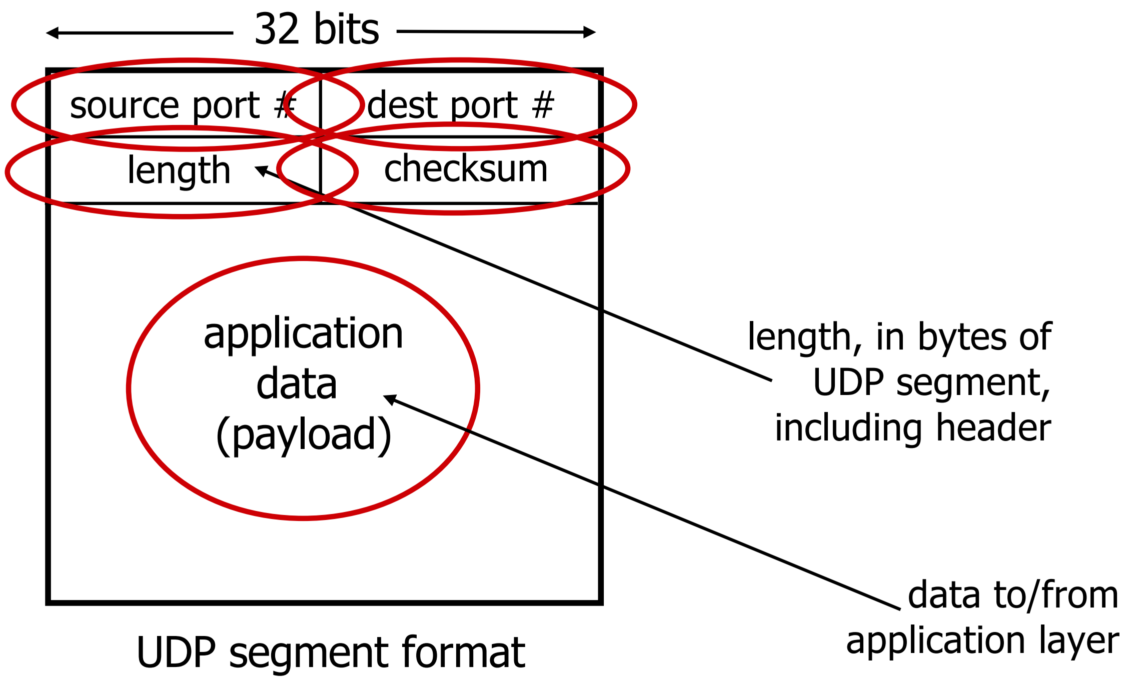
UDP: Transport Layer Actions



UDP receiver actions:

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

UDP segment header



UDP segment format

UDP checksum

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

	1st number	2nd number	sum
Transmitted:	5	6	11
Received:	4	6	11

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

Internet checksum: weak protection!

example: add two 16-bit integers

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

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

UDP Checksum in Practice

- Checksum is the 16-bit one's complement of the one's complement sum of a pseudo header of information from the IP header, the UDP header, and the data, padded with zero octets at the end (if necessary) to make a multiple of two octets.
- Checksum header, data and pre-pended IP pseudo-header (some fields from the IP header)
- But the header contains the checksum itself?

bits	0 – 7	8 – 15	16 – 23	24 – 31
0	Source address			
32	Destination address			
64	Zeros	Protocol	UDP length	
96	Source Port		Destination Port	
128	Length		Checksum	
160+	Data			

IP pseduo-header

UDP header

UDP payload

Checksum: example

10011001 00010010	→	153.18
00001000 01101001	→	8.105
10101011 00000010	→	171.2
00001110 00001010	→	14.10
00000000 00010001	→	0 and 17
00000000 00001111	→	15
00000100 00111111	→	1087
00000000 00001101	→	13
00000000 00001111	→	15
00000000 00000000	→	0 (checksum)
01010100 01000101	→	T and E
01010011 01010100	→	S and T
01001001 01001110	→	I and N
01000111 00000000	→	G and 0 (padding)
10010110 11101011	→	Sum
01101001 00010100	→	Checksum

Note: TCP Checksum computation is exactly similar

UDP Applications

- Latency sensitive/time critical
  - Quick request/response (DNS, DHCP)
  - Network management (SNMP)
  - Routing updates (RIP)
  - Voice/video chat
  - Gaming (especially FPS)
- Error correction managed by periodic messages

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)

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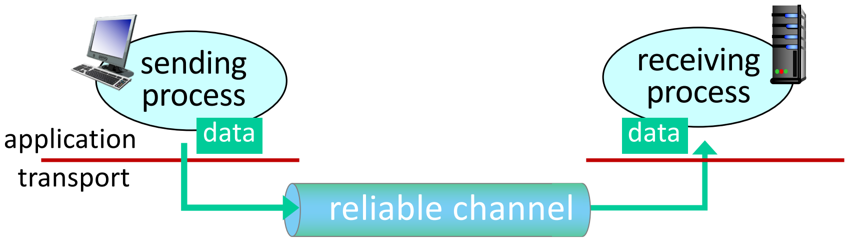
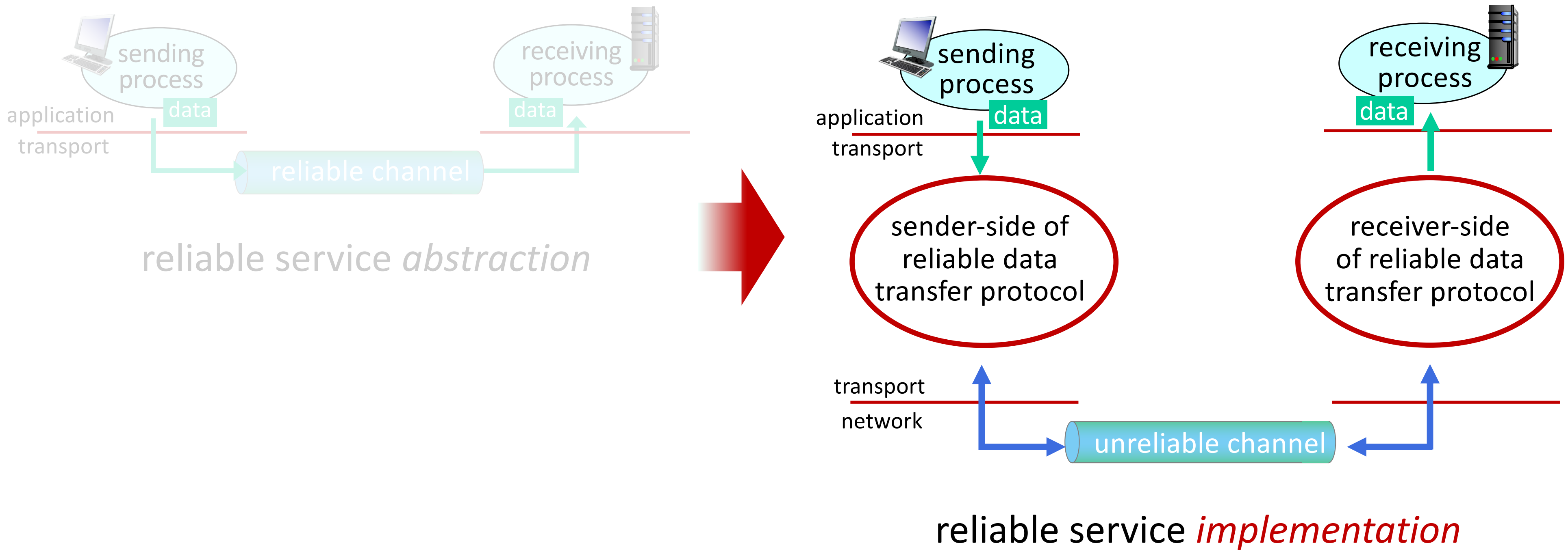
Principles of reliable data transfer



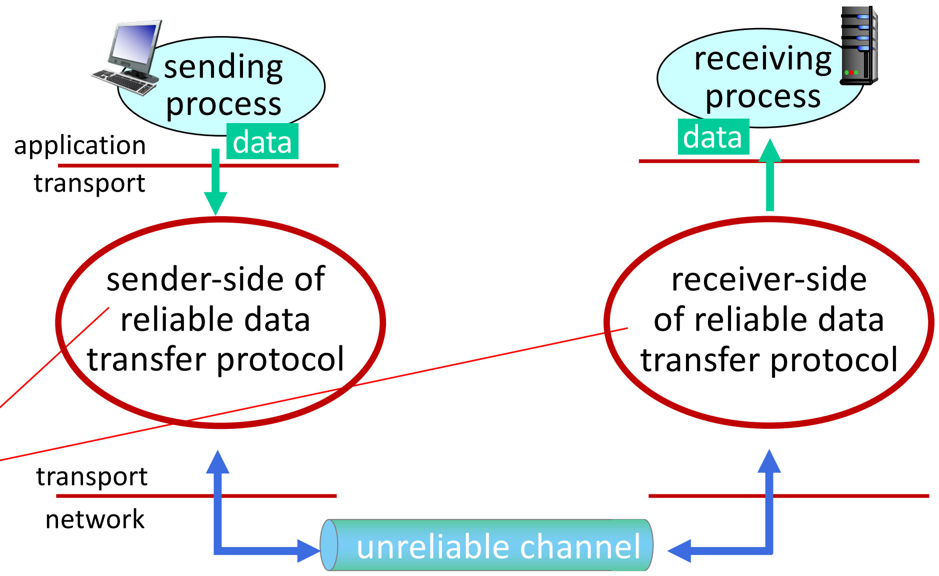
reliable service *abstraction*

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?)

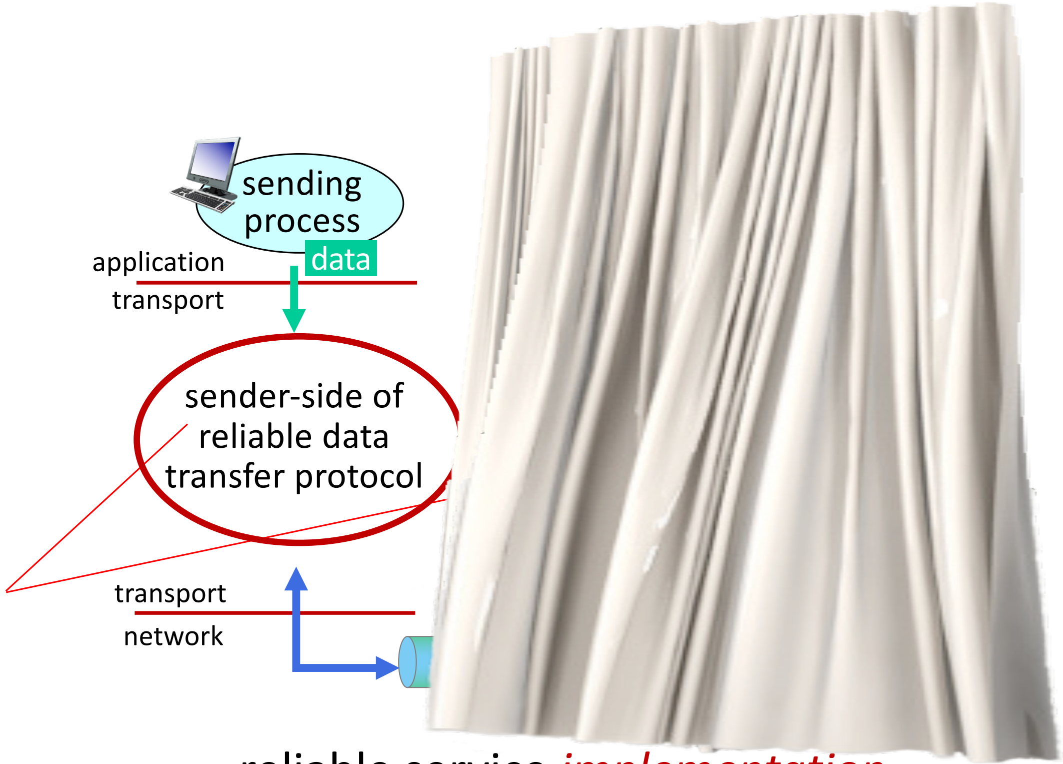


reliable service *implementation*

Principles of reliable data transfer

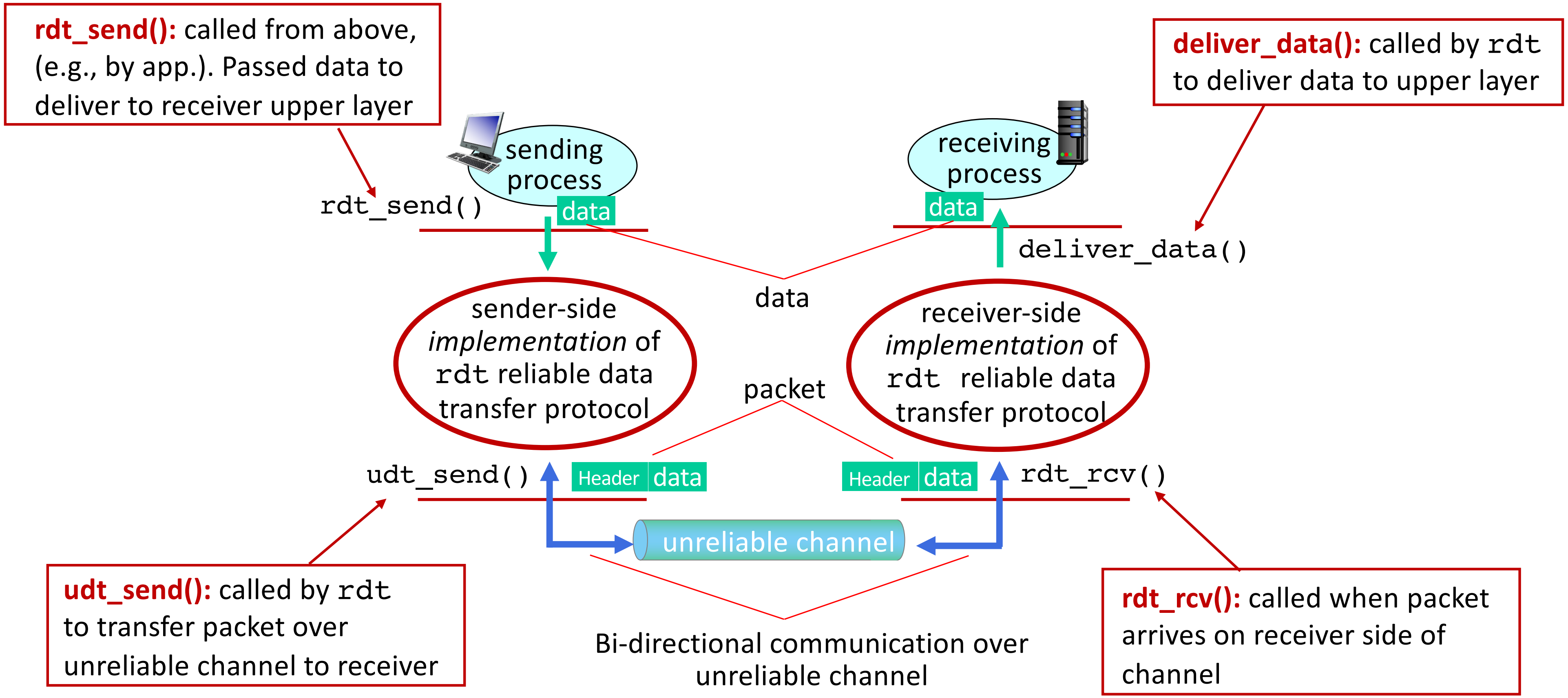
Sender, receiver do *not* know the “state” of each other, e.g., was a message received?

- unless communicated via a message



reliable service *implementation*

Reliable data transfer protocol (rdt): interfaces

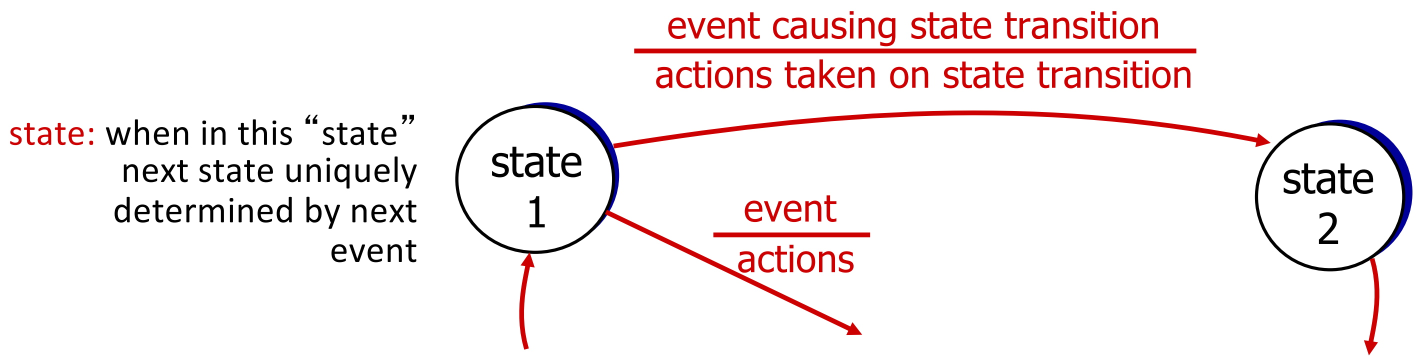


Reliable data transfer: getting started

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!

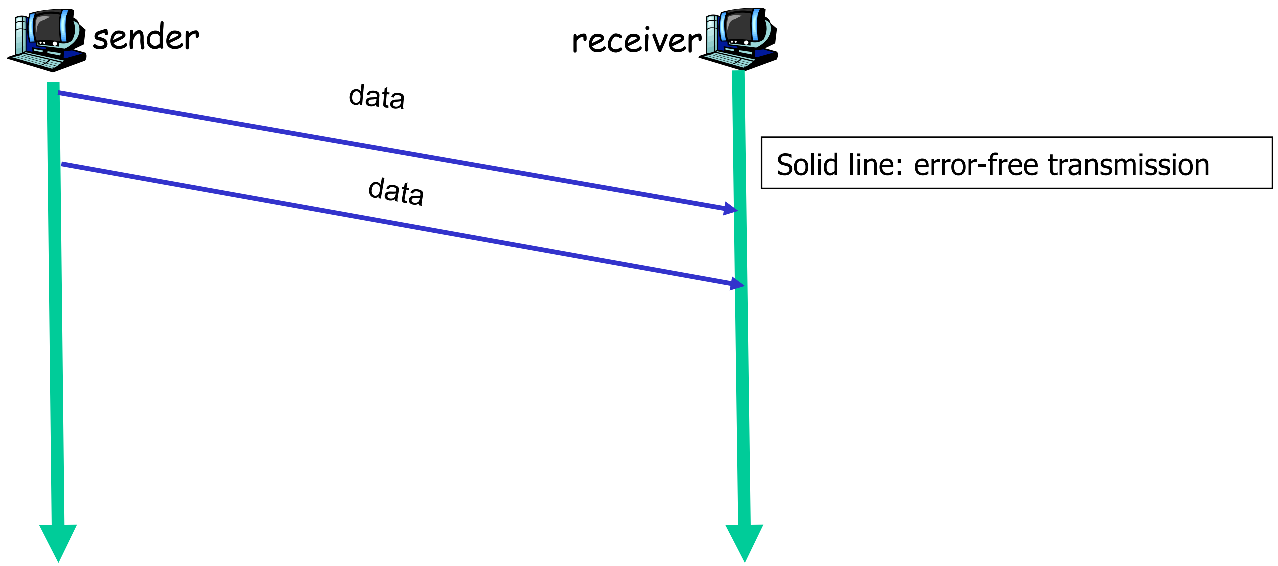
- Book uses finite state machines (FSM) to specify sender, receiver
  - We won't use them in the lecture, and you won't be asked exam questions on them



rdt1.0: reliable transfer over a reliable channel

- underlying channel perfectly reliable
  - no bit errors
  - no loss of packets
- Nothing to do

Global Picture of rdt1.0



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?

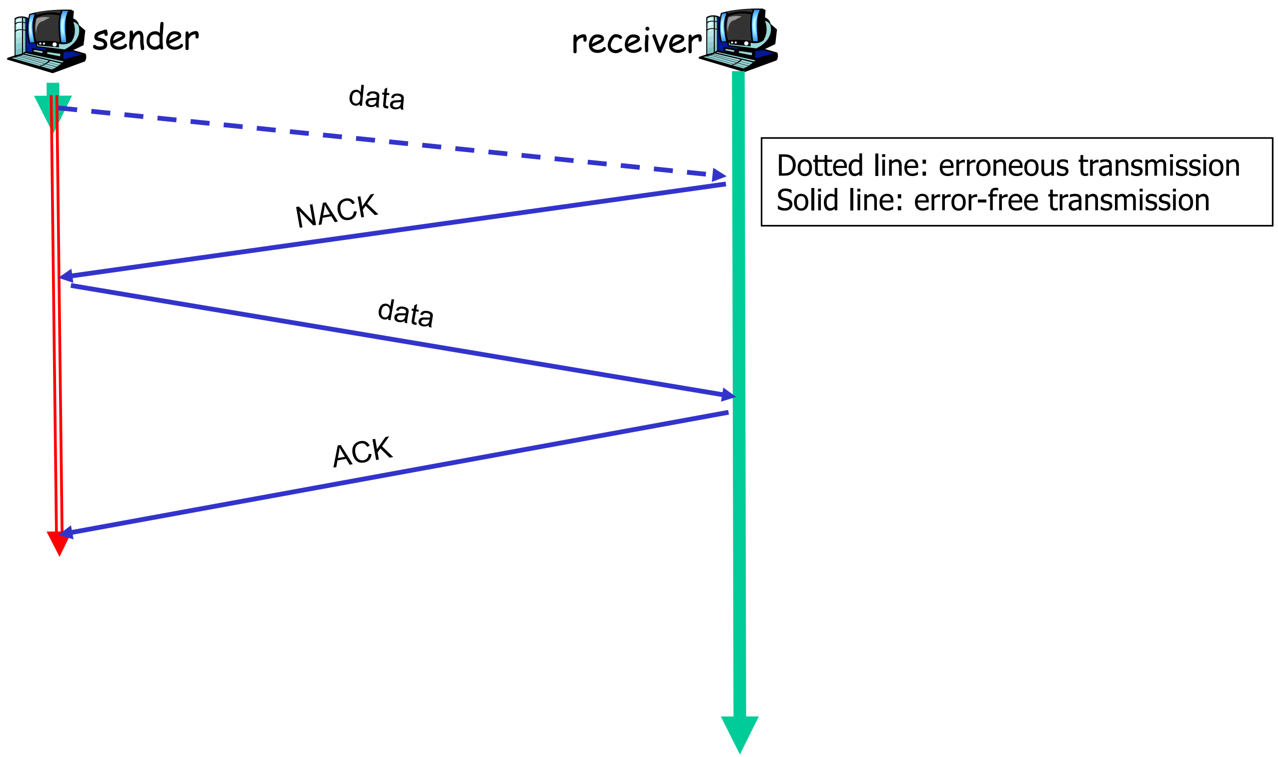
rdt2.0: channel with bit errors

- 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
- new mechanisms in **rdt2.0** (beyond **rdt1.0**):
  - error detection
  - feedback: control msgs (ACK,NAK) from receiver to sender
  - retransmission

stop and wait

sender sends one packet, then waits for receiver response

Global Picture of rdt2.0



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

handling duplicates:

- sender retransmits current pkt if ACK/NAK corrupted
- sender adds *sequence number* to each pkt
- receiver discards (doesn't deliver up) duplicate pkt

stop and wait

sender sends one packet, then waits for receiver response

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

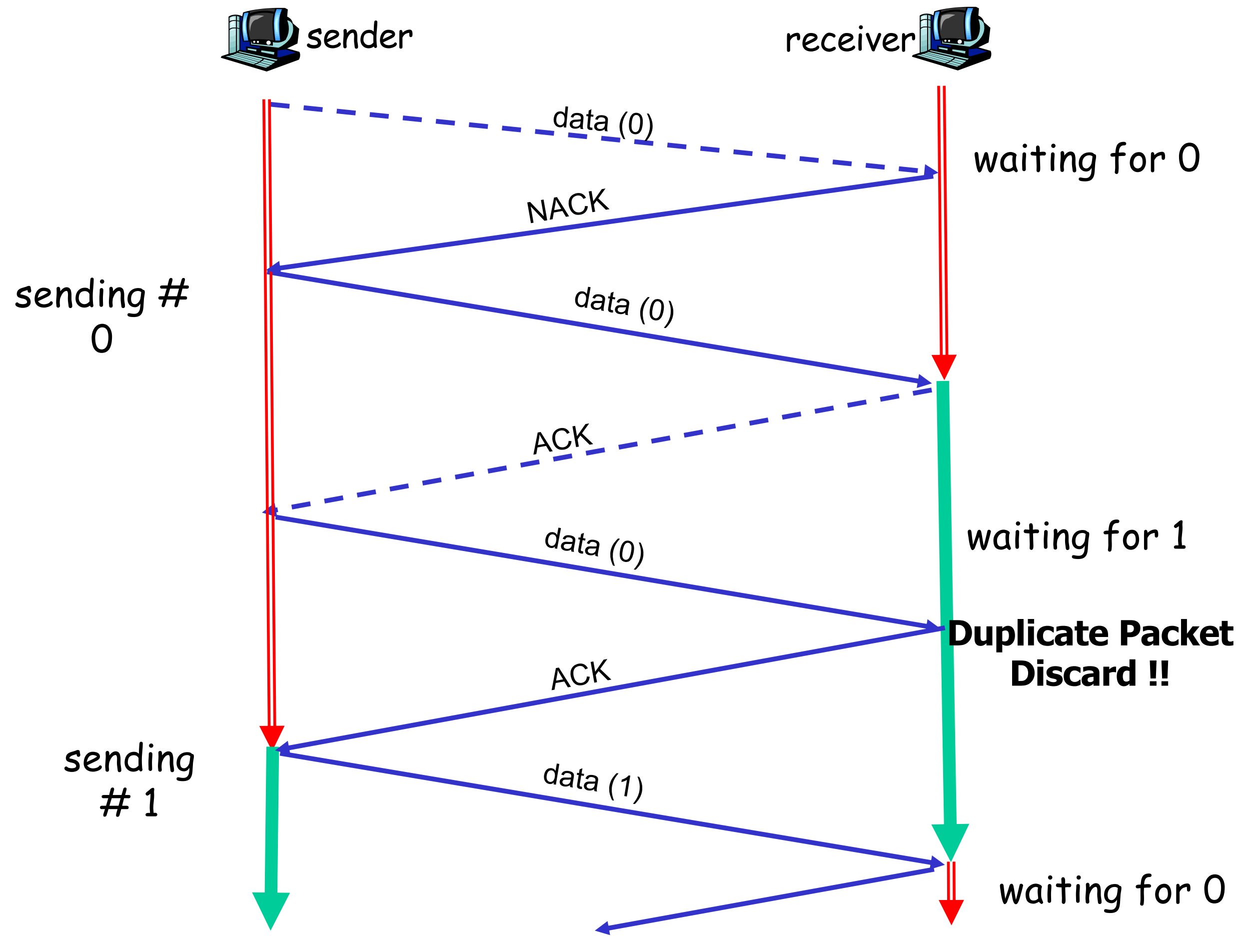
receiver:

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

New Measures: Sequence Numbers, Checksum for ACK/NACK, Duplicate detection

Another Look at rdt2.1

Dotted line: erroneous transmission
Solid line: error-free transmission



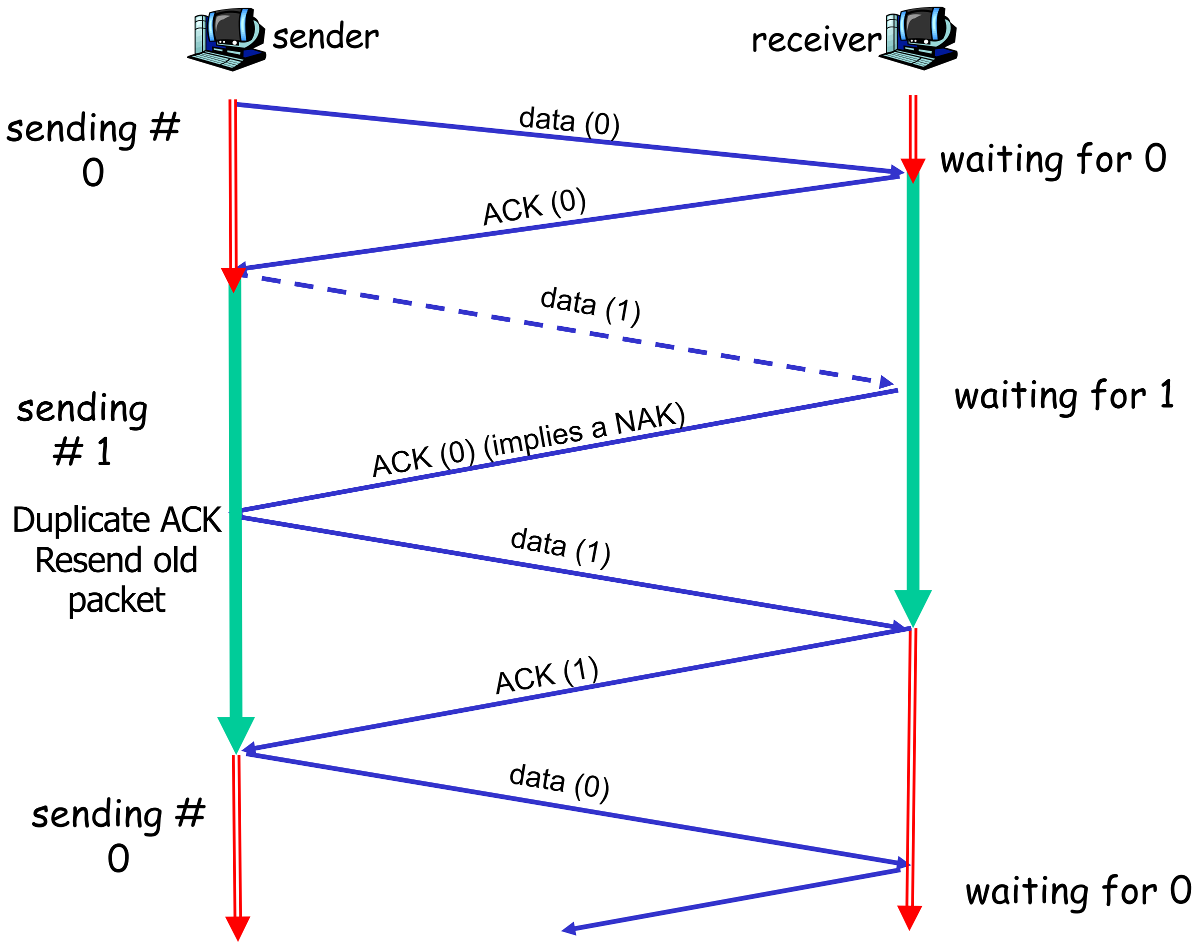
rdt2.2: a NAK-free protocol

- 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*

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

rdt2.2: Example

Dotted line: erroneous transmission
Solid line: error-free transmission



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

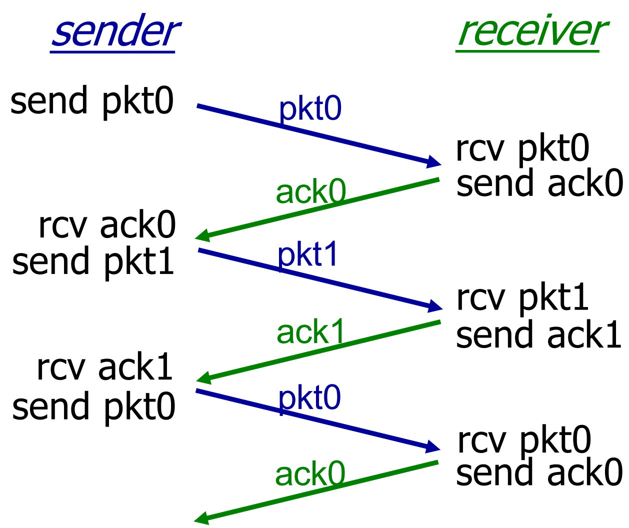
Q: How do *humans* handle lost sender-to-receiver words in conversation?

rdt3.0: channels with errors *and* loss

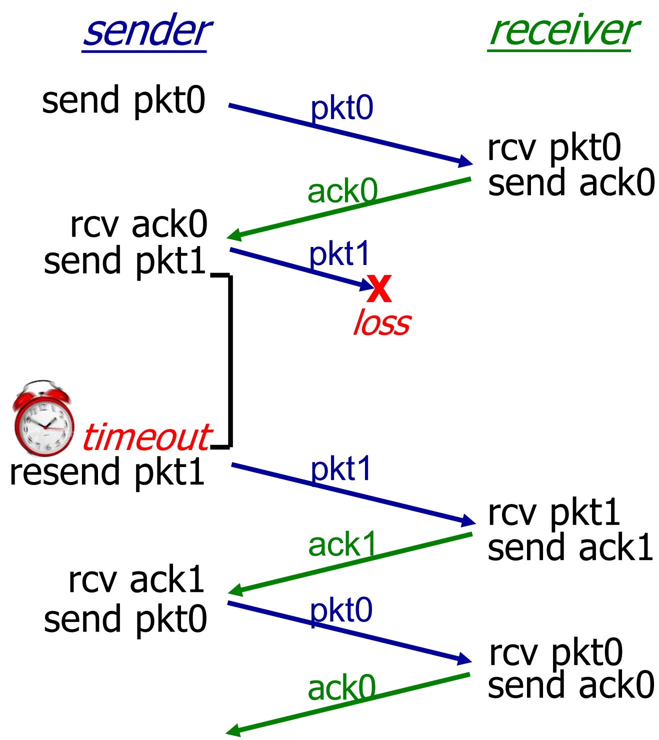
Approach: sender waits "reasonable" amount of time for ACK

- 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
- No retransmission on duplicate ACKs

rdt3.0 in action

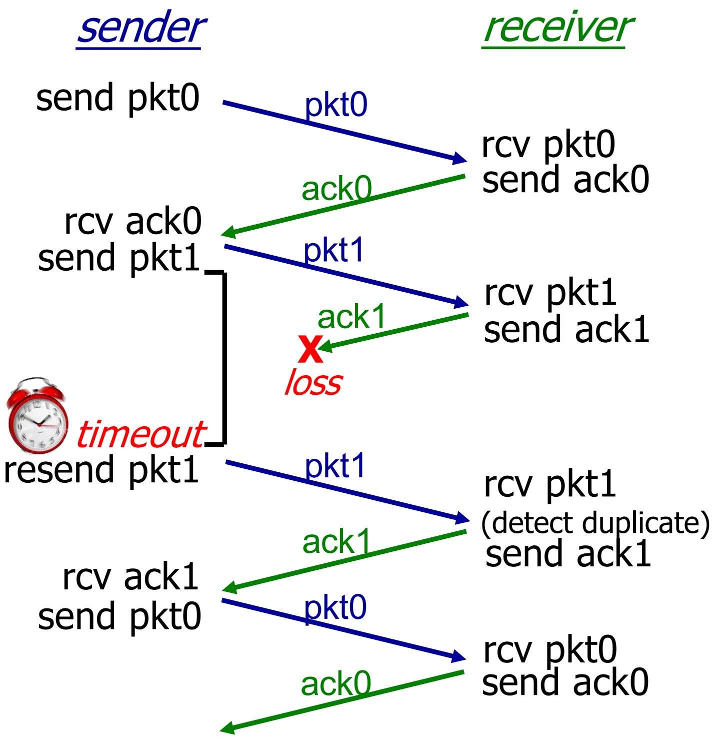


(a) no loss

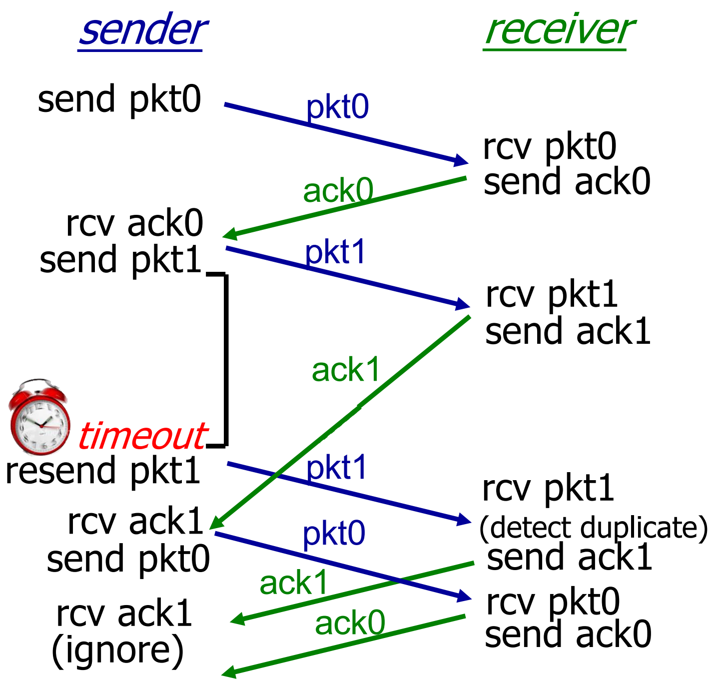


(b) packet loss

rdt3.0 in action



(c) ACK loss



(d) premature timeout/ delayed ACK

Quiz: Reliable Data Transfer

Which of the following are needed for reliable data transfer with only packet corruption (and no loss or reordering)? Use only as much as is strictly needed.

a) Checksums
b) Checksums, ACKs, NACKs
c) Checksums, ACKs
d) Checksums, ACKs, sequence numbers
e) Checksums, ACKs, NACKs, sequence numbers

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Answer: D (RDT 2.2)

Quiz: Reliable Data Transfer

If packets (and ACKs and NACKs) could be lost which of the following is true of RDT 2.1 (or 2.2)?

a) Reliable in-order delivery is still achieved
b) The protocol will get stuck
c) The protocol will continue making progress but may skip delivering some messages

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Answer: B

Quiz: Reliable Data Transfer

Which of the following are needed for reliable data transfer to handle packet corruption and loss? Use only as much as is strictly needed.

a) Checksums, timeouts
b) Checksums, ACKs, sequence numbers
c) Checksums, ACKs, timeouts
d) Checksums, ACKs, timeouts, sequence numbers
e) Checksums, ACKs, NACKs, timeouts, sequence numbers

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Answer: D (RDT 3.0)

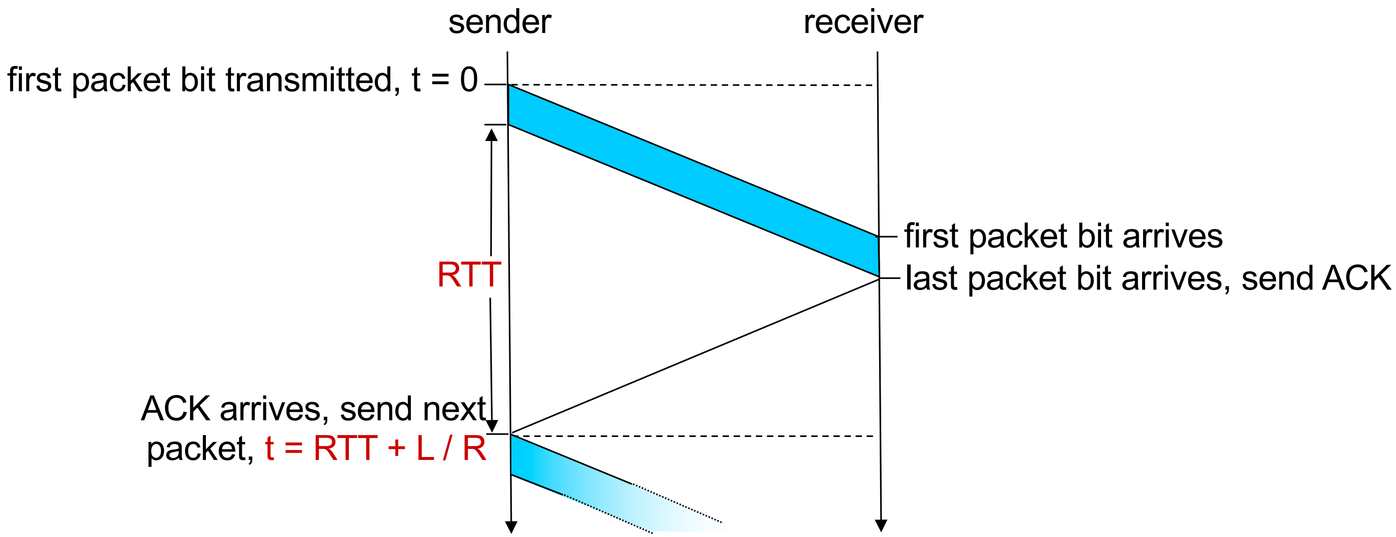
Performance of rdt3.0 (stop-and-wait)

- 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

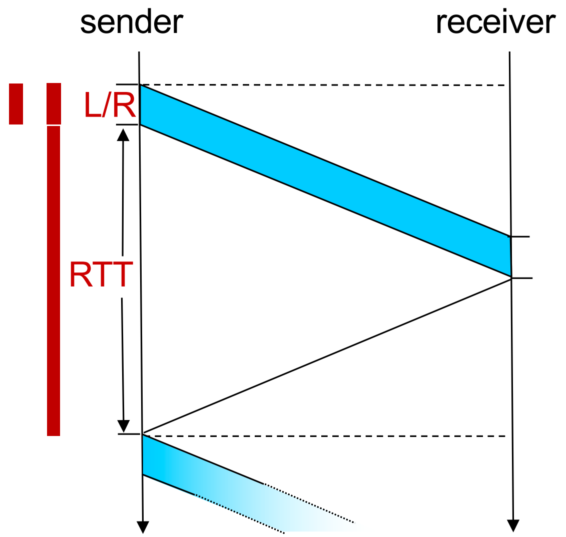


rdt3.0: stop-and-wait operation

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

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

$$= 0.00027$$

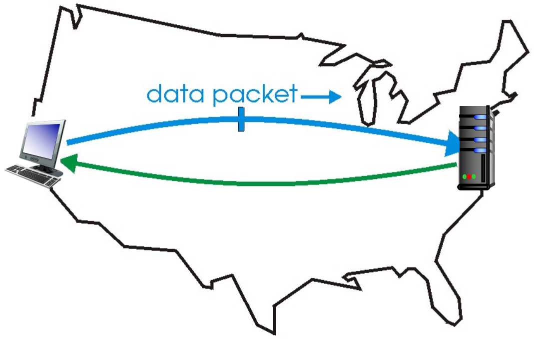


- rdt 3.0 protocol performance is very poor!
- Protocol limits performance of underlying infrastructure (channel)

rdt3.0: pipelined protocols operation

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

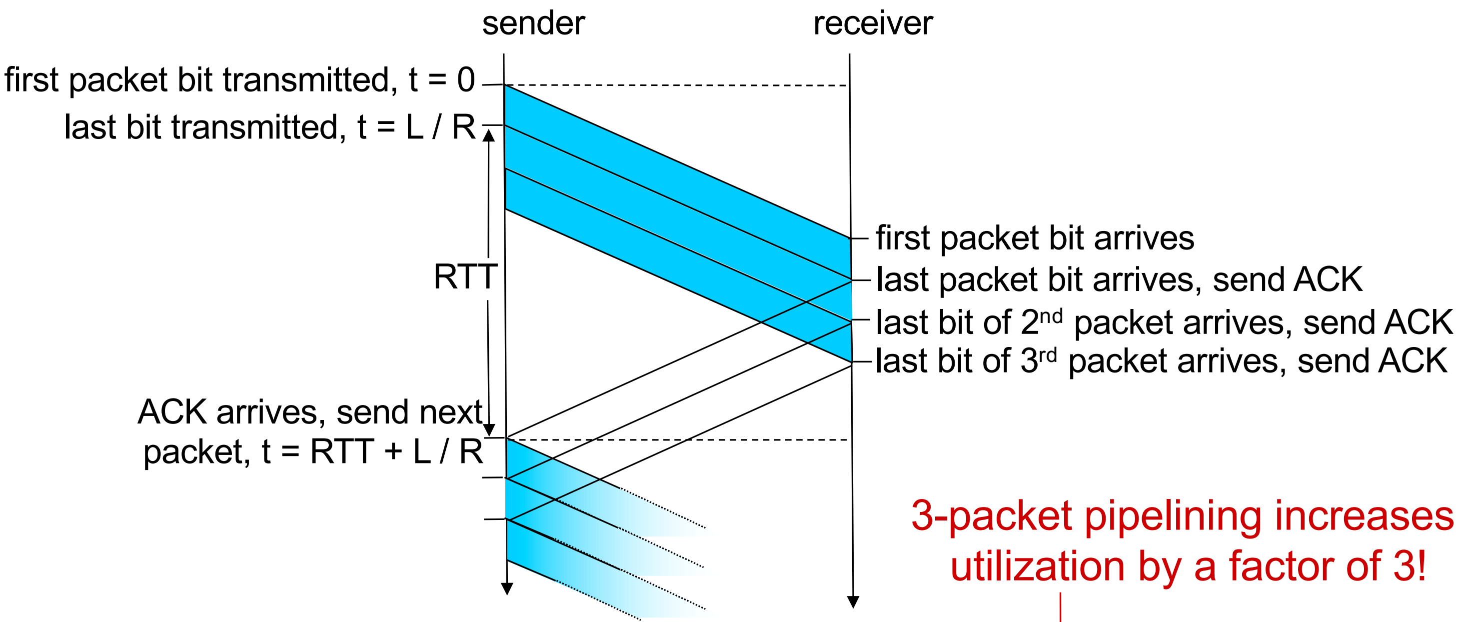
- range of sequence numbers must be increased
- buffering at sender and/or receiver



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

- Go Back N, Selective Repeat

Pipelining: increased utilization

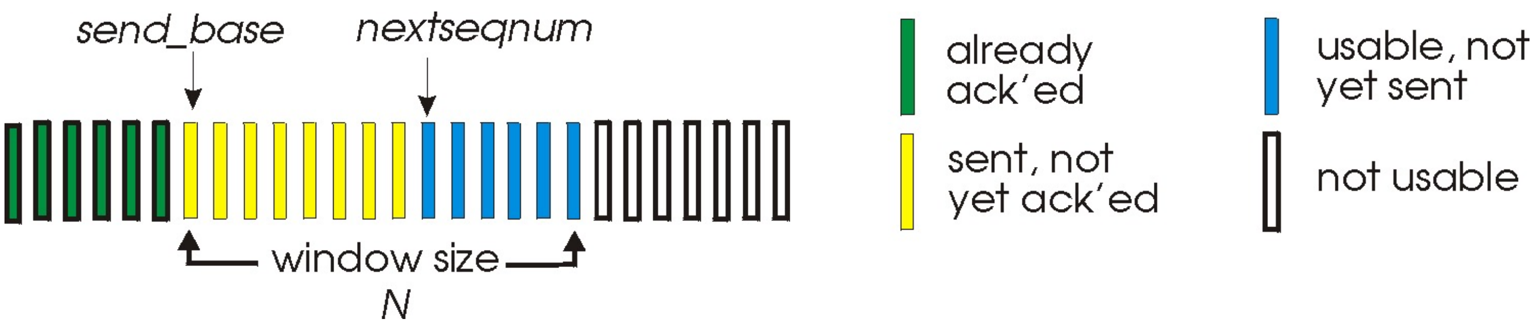


3-packet pipelining increases utilization by a factor of 3!

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

Go-Back-N: sender

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



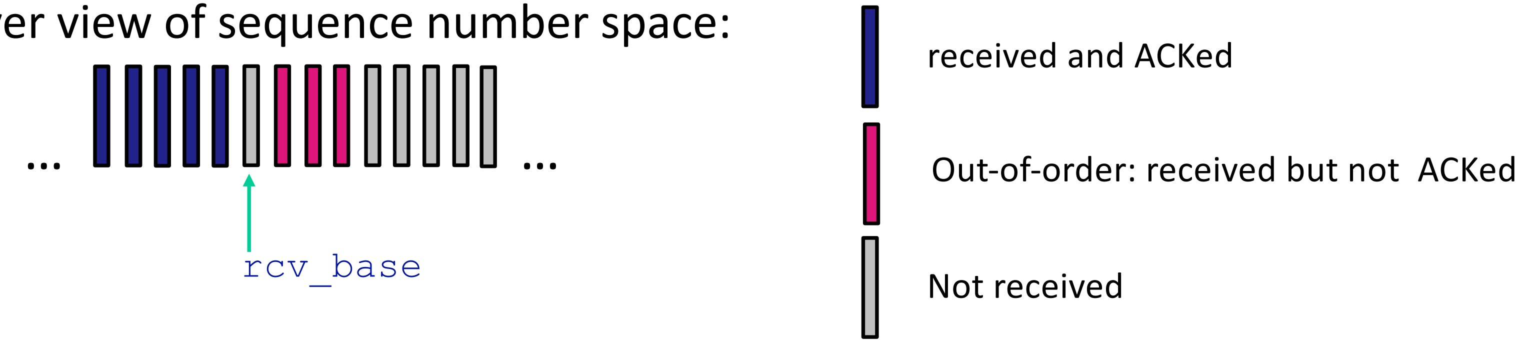
- *cumulative ACK:* ACK(*n*): ACKs all packets up to, including seq # *n*
  - on receiving ACK(*n*): move window forward to begin at *n+1*
- timer for oldest in-flight packet
- *timeout(n):* retransmit packet n and all higher seq # packets in window

Applets: http://media.pearsoncmg.com/aw/aw_kurose_network_2/applets/go-back-n/go-back-n.html
http://www.ccs-labs.org/teaching/rn/animations/gbn_sr/

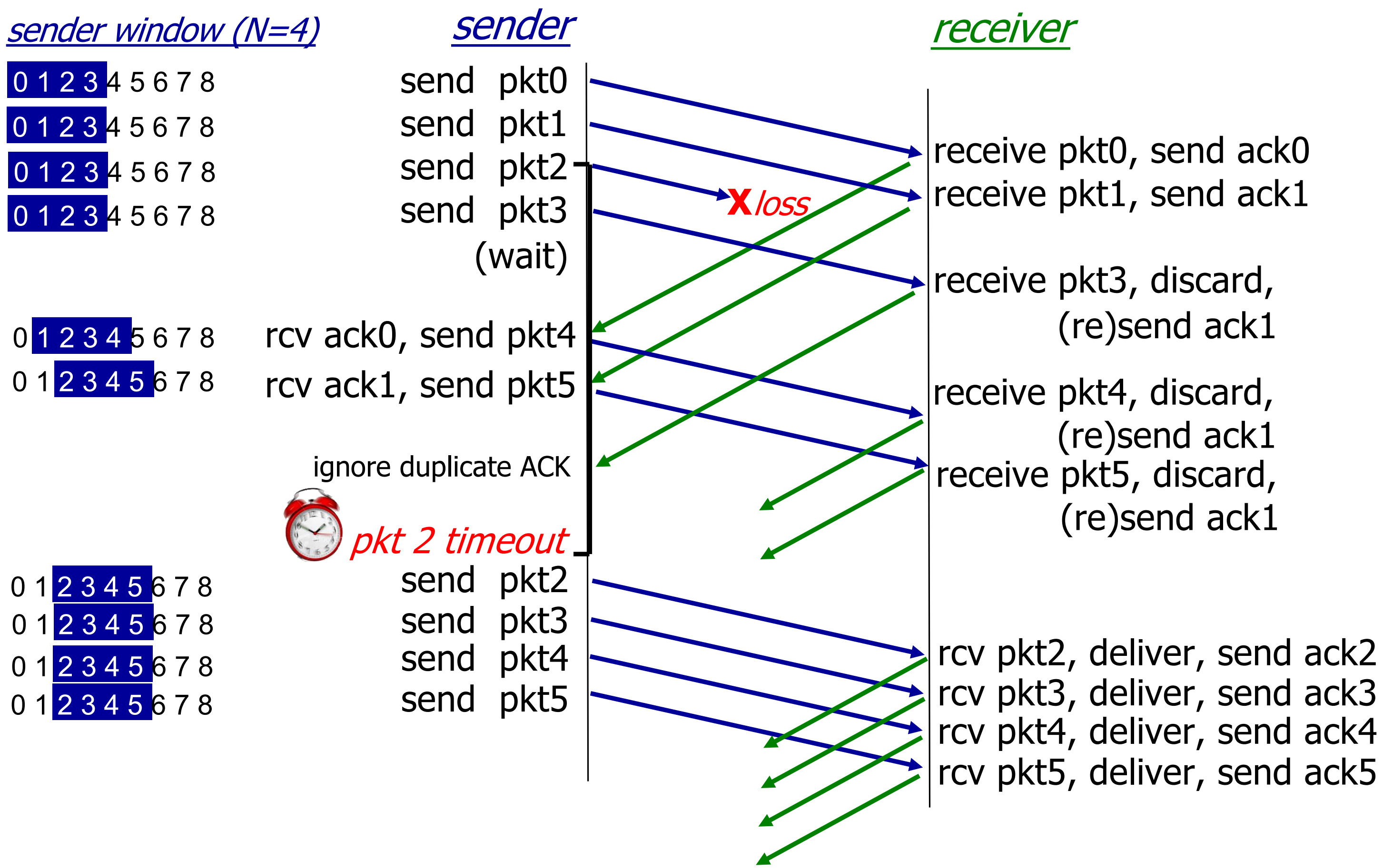
Go-Back-N: receiver

- ACK-only: always send ACK for correctly-received packet so far, with highest *in-order* seq #
  - may generate duplicate ACKs
  - need only remember `rcv_base`
- on receipt of out-of-order packet:
  - can discard (don't buffer) or buffer: an implementation decision
  - re-ACK pkt with highest in-order seq #

Receiver view of sequence number space:



Go-Back-N in action



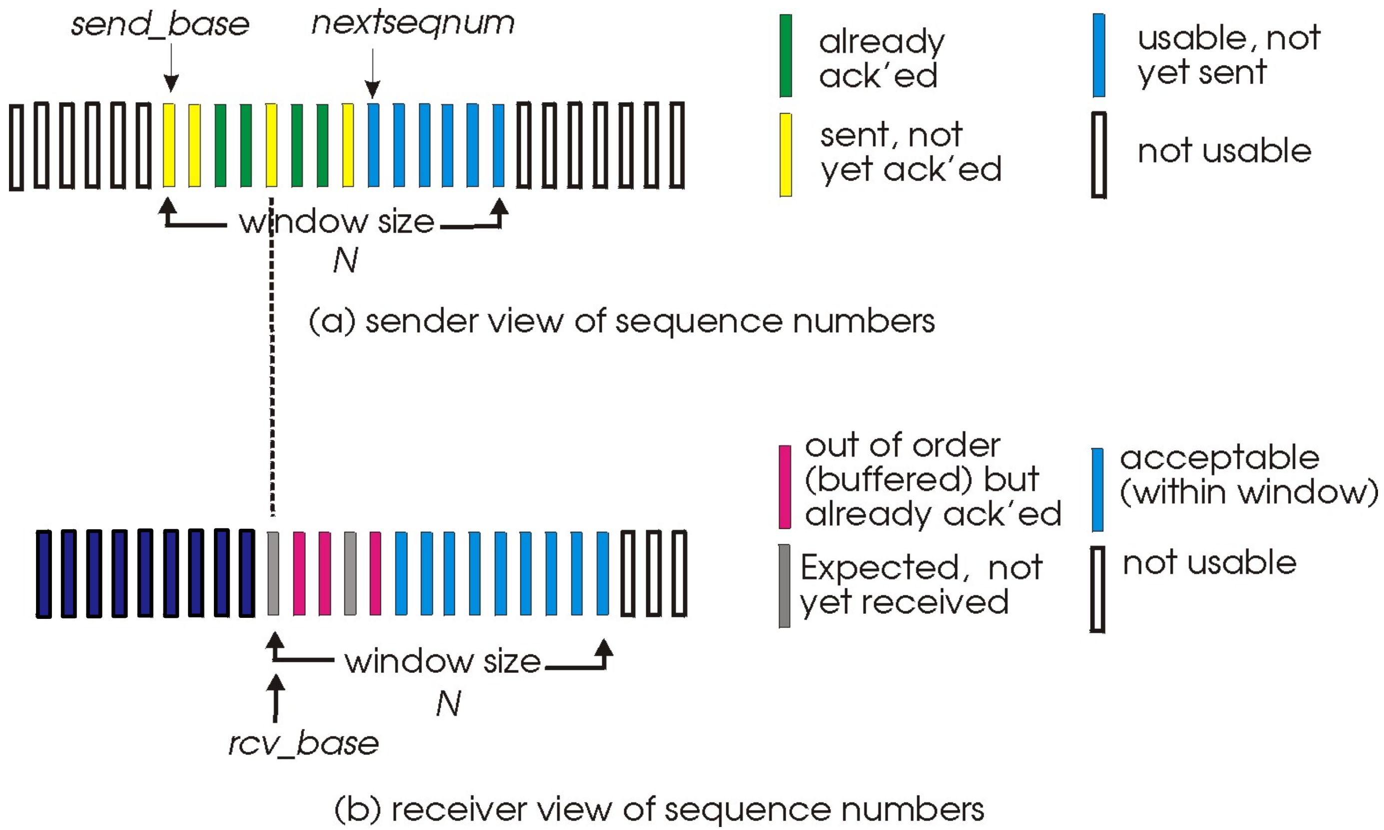
Selective repeat

- 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

Applet: http://media.pearsoncmg.com/aw/aw_kurose_network_3/applets/SelectRepeat/SR.html

http://www.ccs-labs.org/teaching/rn/animations/gbn_sr/

Selective repeat: sender, receiver windows



Selective repeat: sender and receiver

sender

data from above:

- if next available seq # in window, send packet

timeout(*n*):

- resend packet *n*, restart timer

ACK(*n*) in [sendbase,sendbase+N]:

- mark packet *n* as received
- if n smallest unACKed packet, advance window base to next unACKed seq #

receiver

packet *n* in [rcvbase, rcvbase+N-1]

- send ACK(*n*)
- out-of-order: buffer
- in-order: deliver (also deliver buffered, in-order packets), advance window to next not-yet-received packet

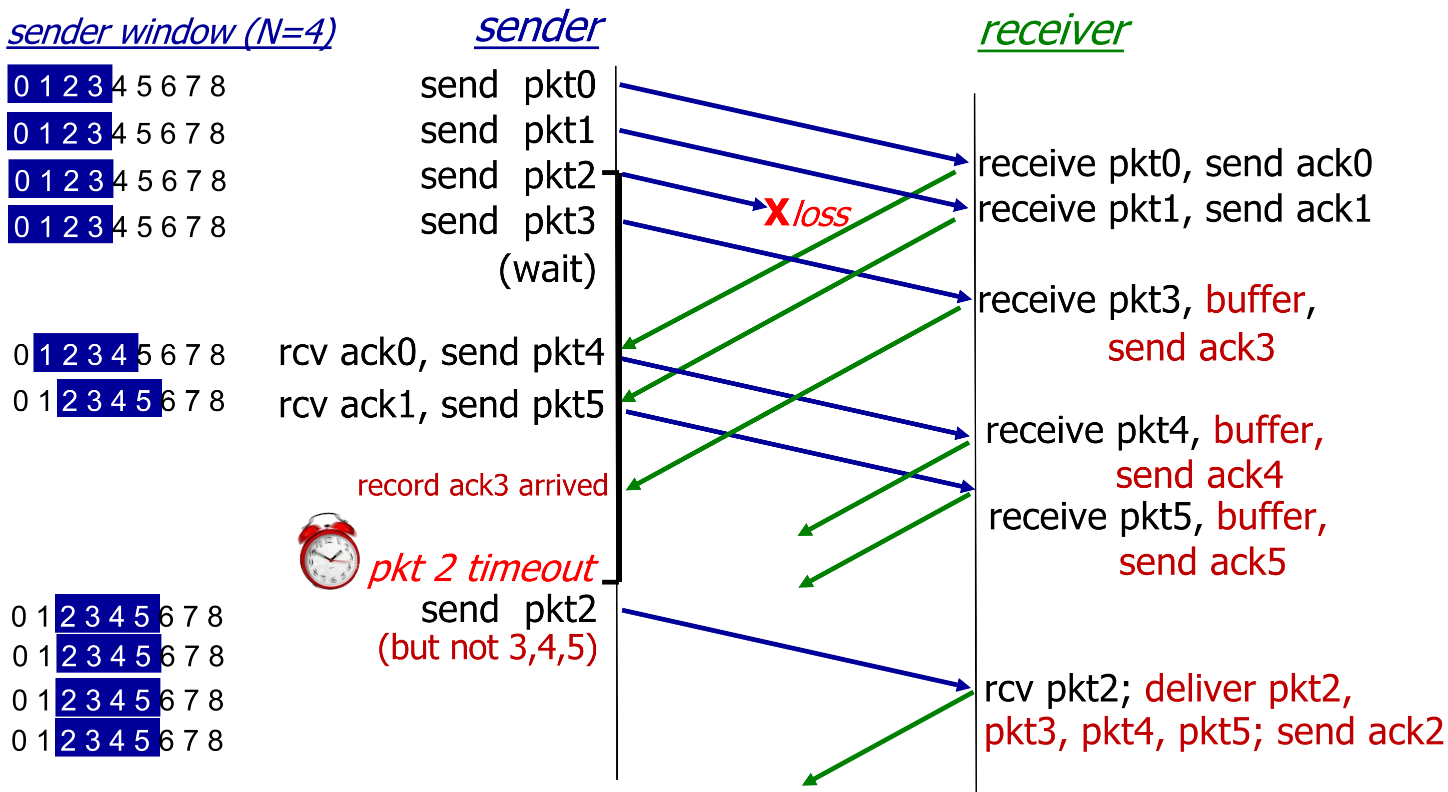
packet *n* in [rcvbase-N,rcvbase-1]

- ACK(*n*)

otherwise:

- ignore

Selective Repeat in action

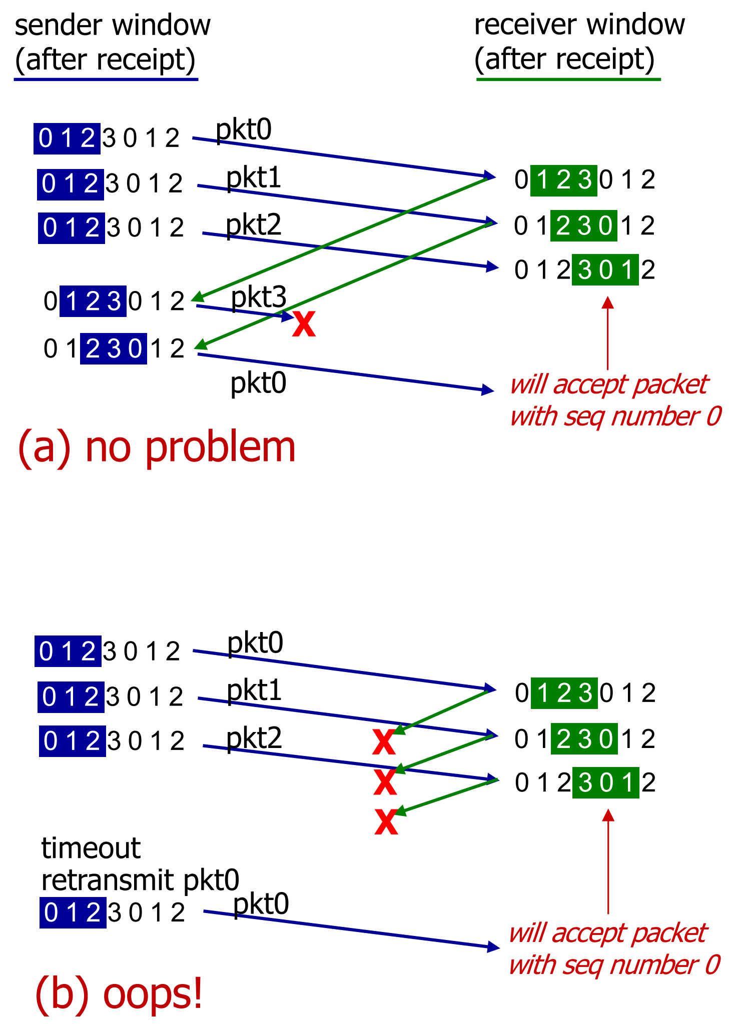


Q: what happens when ack2 arrives?

Selective repeat: a dilemma!

example:

- seq #s: 0, 1, 2, 3 (base 4 counting)
- window size=3



Selective repeat: a dilemma!

example:

- seq #s: 0, 1, 2, 3 (base 4 counting)
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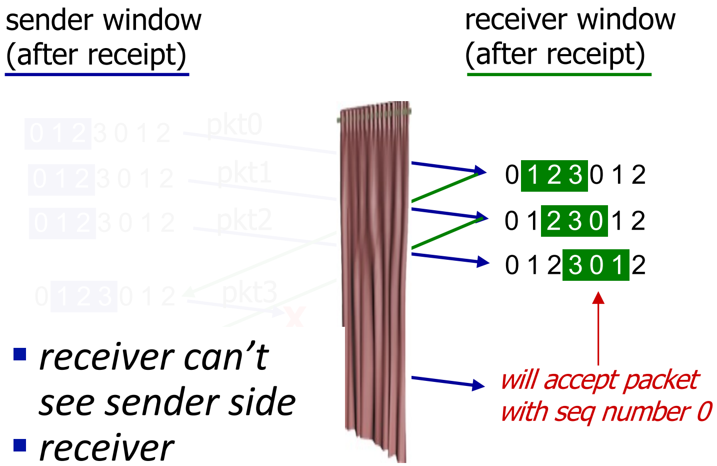
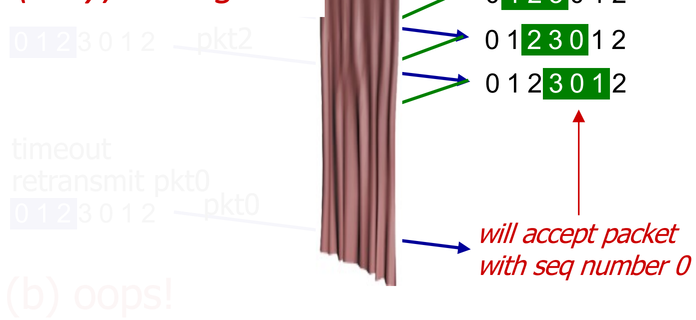


- *receiver can't see sender side*
- *receiver behavior identical in both cases!*
- *something's (very) wrong!*



Q: what relationship is needed between sequence # size and window size to avoid problem in scenario (b)?

A: Sender window size <= ½ of Sequence number space

Recap: components of a solution

- Checksums (for error detection)
- Timers (for loss detection)
- Acknowledgments
  - cumulative
  - selective
- Sequence numbers (duplicates, windows)
- Sliding Windows (for efficiency)

- Reliability protocols use the above to decide when and what to retransmit or acknowledge

Quiz: GBN, SR

Which of the following is not true?

a) GBN uses cumulative ACKs, SR uses individual ACKs
b) Both GBN and SR use timeouts to address packet loss
c) GBN maintains a separate timer for each outstanding packet
d) SR maintains a separate timer for each outstanding packet
e) Neither GBN nor SR use NACKs

Answer: C

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Quiz: GBN, SR

Suppose a receiver that has received all packets up to and including sequence number 24 and next receives packet 27 and 28. In response, what are the sequence numbers in the ACK(s) sent out by the GBN and SR receiver, respectively?

a) [27, 28], [28, 28]
b) [24, 24], [27, 28]
c) [27, 28], [27, 28]
d) [25, 25], [25, 25]
e) [nothing], [27, 28]

Answer: B

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Transport Layer 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

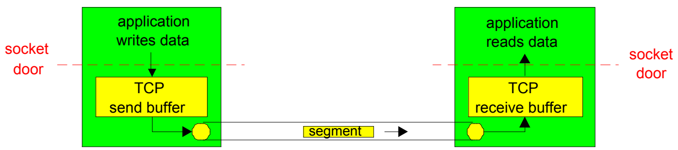
3.7 TCP congestion control

Practical Reliability Questions

- How do the sender and receiver keep track of outstanding pipelined segments?
- How many segments should be pipelined?
- How do we choose sequence numbers?
- What does connection establishment and teardown look like?
- How should we choose timeout values?

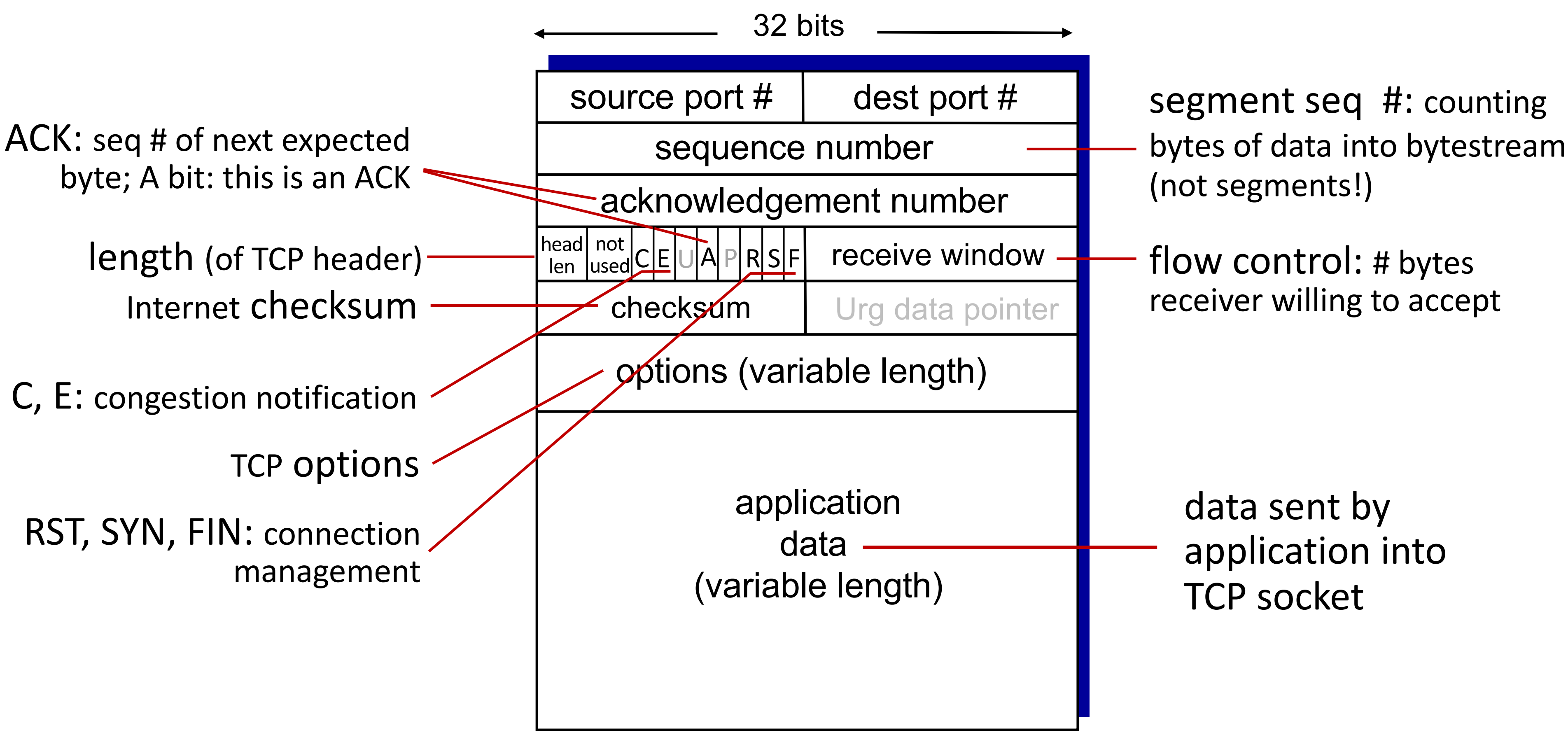
TCP: overview RFCs: 793,1122, 2018, 5681, 7323

- 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

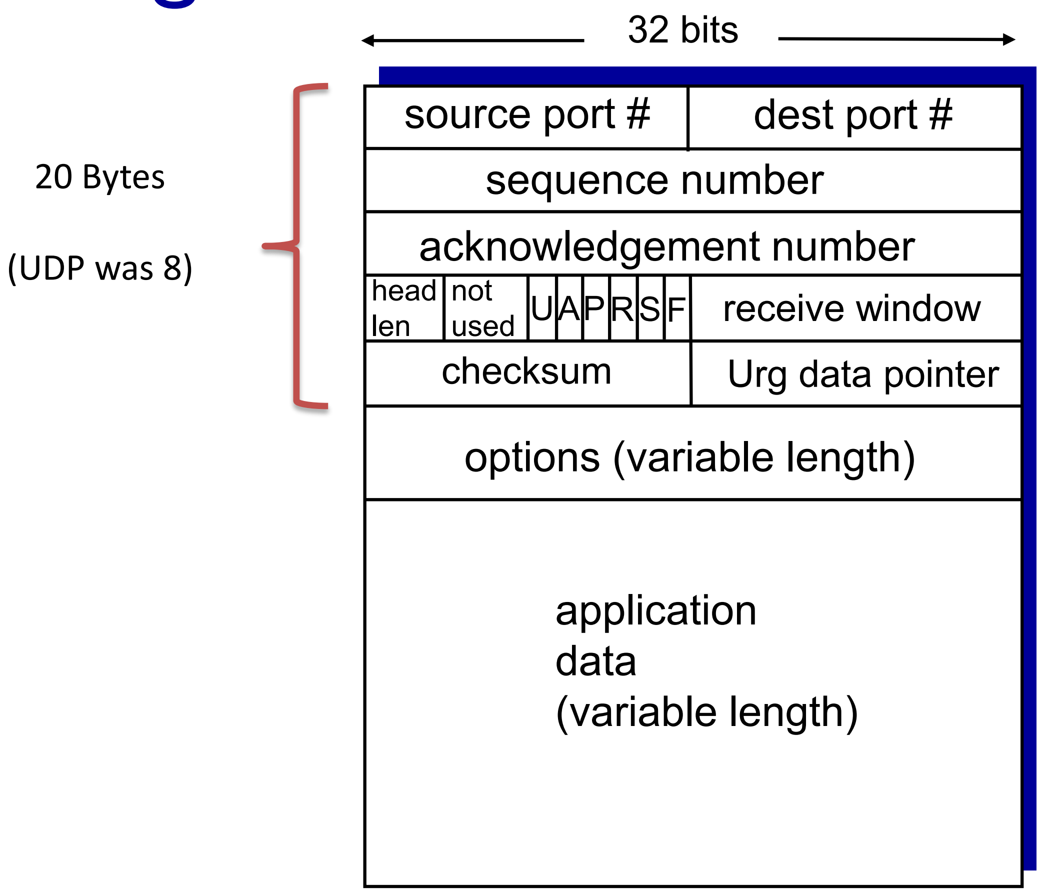


- cumulative ACKs
- pipelining:
  - TCP congestion and flow control set window size
- connection-oriented:
  - handshaking (exchange of control messages) initializes sender, receiver state before data exchange
- flow controlled:
  - sender will not overwhelm receiver

TCP segment structure



TCP segment structure



Summary

- Multiplexing/Demultimplexing
- UDP
- Reliable Data Transfer
  - Stop-and-wait protocols
  - Sliding winding protocols
- TCP - intro
- Up Next:
  - TCP – continued in more detail
  - Congestion Control