Chapter 3
Transport Layer

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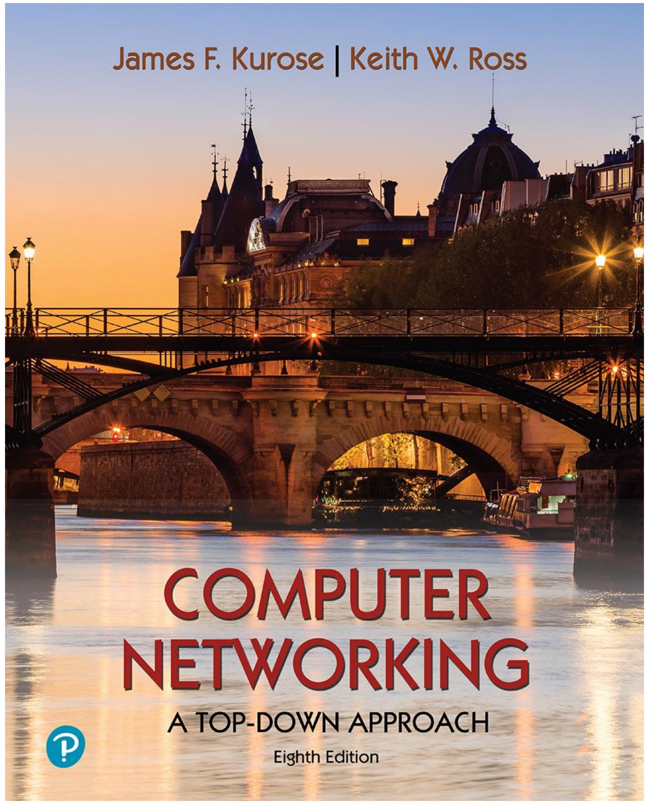


Computer Networking: A Top-Down Approach
8th edition
Jim Kurose, Keith Ross
Pearson, 2020

Transport layer: overview

Our goal:

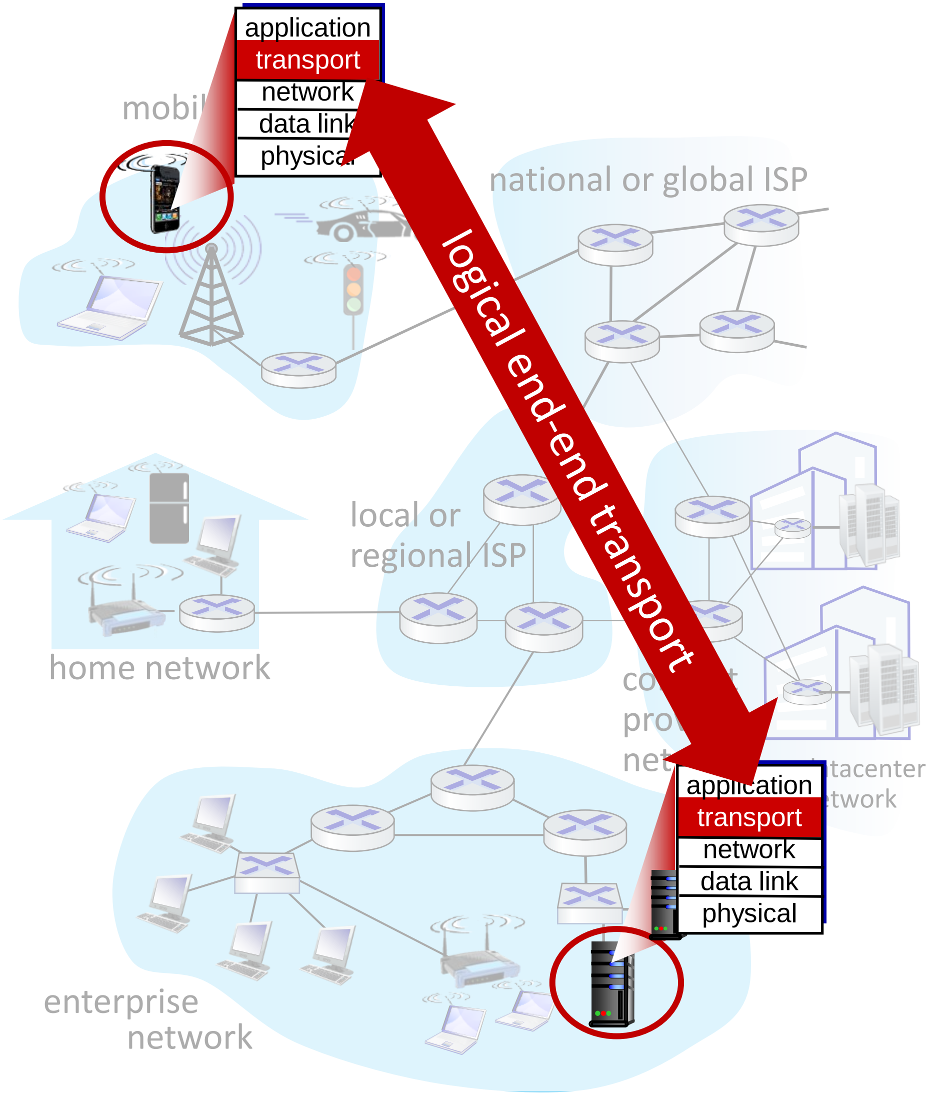
- understand principles behind transport layer services:
  - multiplexing, demultiplexing
  - reliable data transfer
  - flow control
  - congestion control
- learn about Internet transport layer protocols:
  - UDP: connectionless transport
  - TCP: connection-oriented reliable transport
  - TCP congestion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

Transport layer: roadmap

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

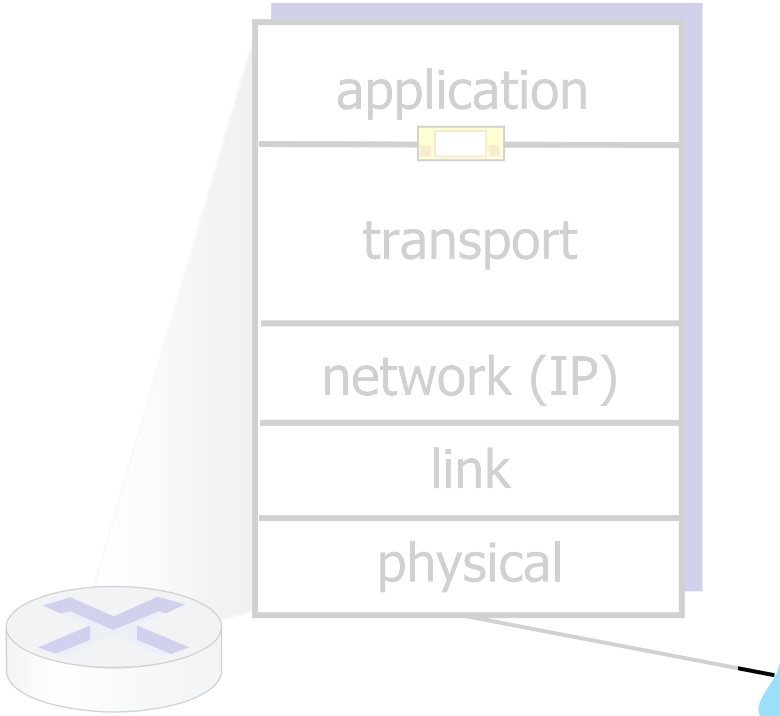


Transport vs. network layer services and protocols

- **transport layer:** logical communication between *processes*
  - relies on, enhances, network layer services
  - PDU: Segment
  - extends "host-to-host" communication to "process-to-process" communication

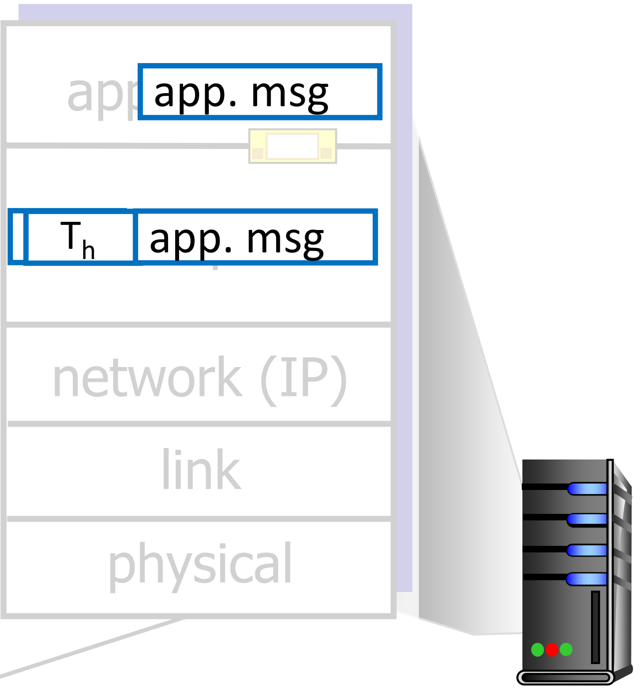
- **network layer:** logical communication between *hosts*
  - PDU: Datagram/Packet
  - Datagram's may be lost, duplicated, reordered in the Internet – "best effort" service

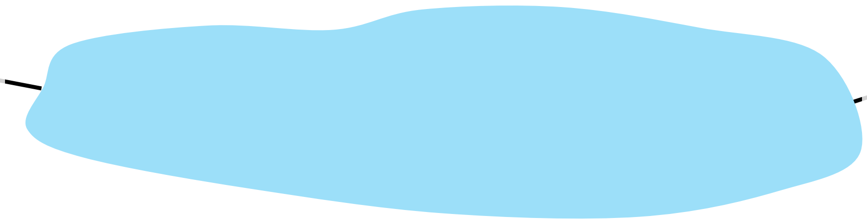
Transport Layer Actions



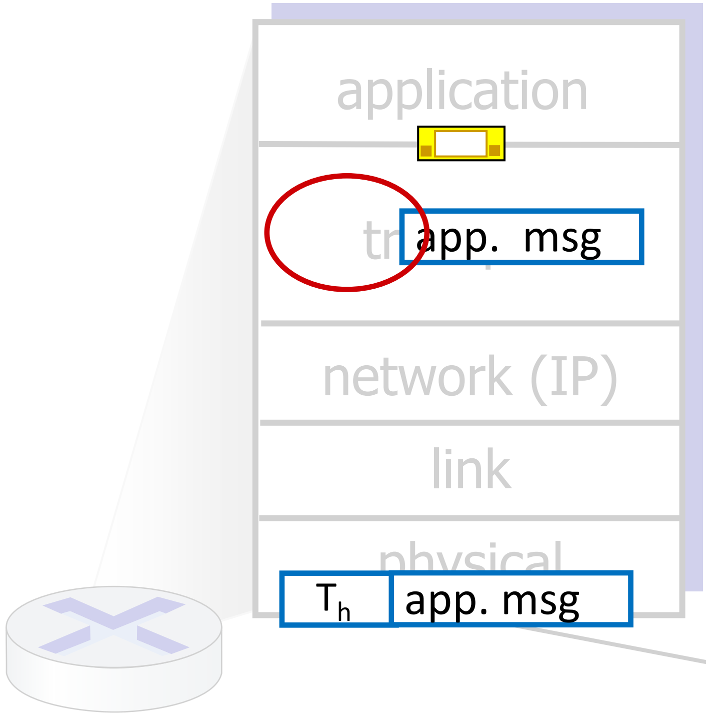
Sender:

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



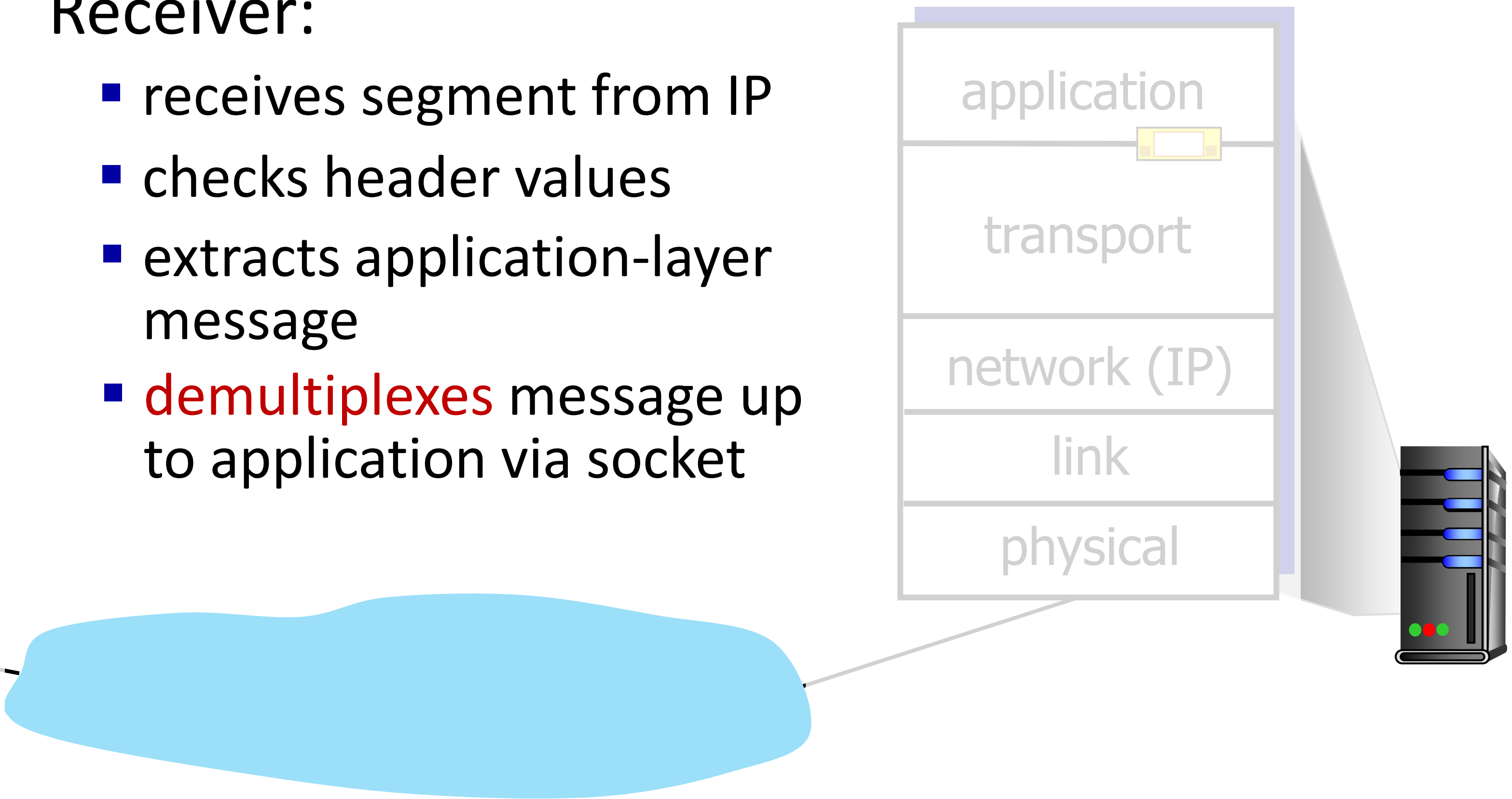


Transport Layer Actions



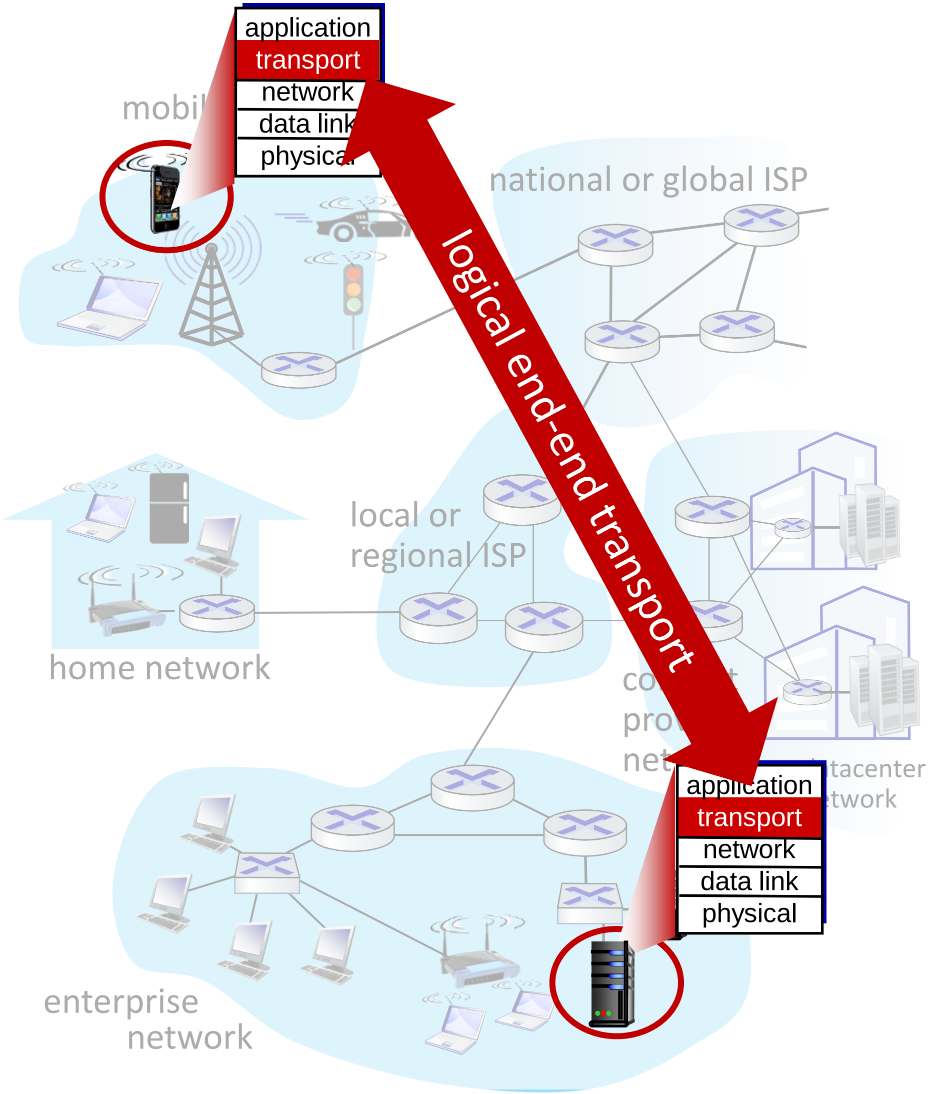
Receiver:

- 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

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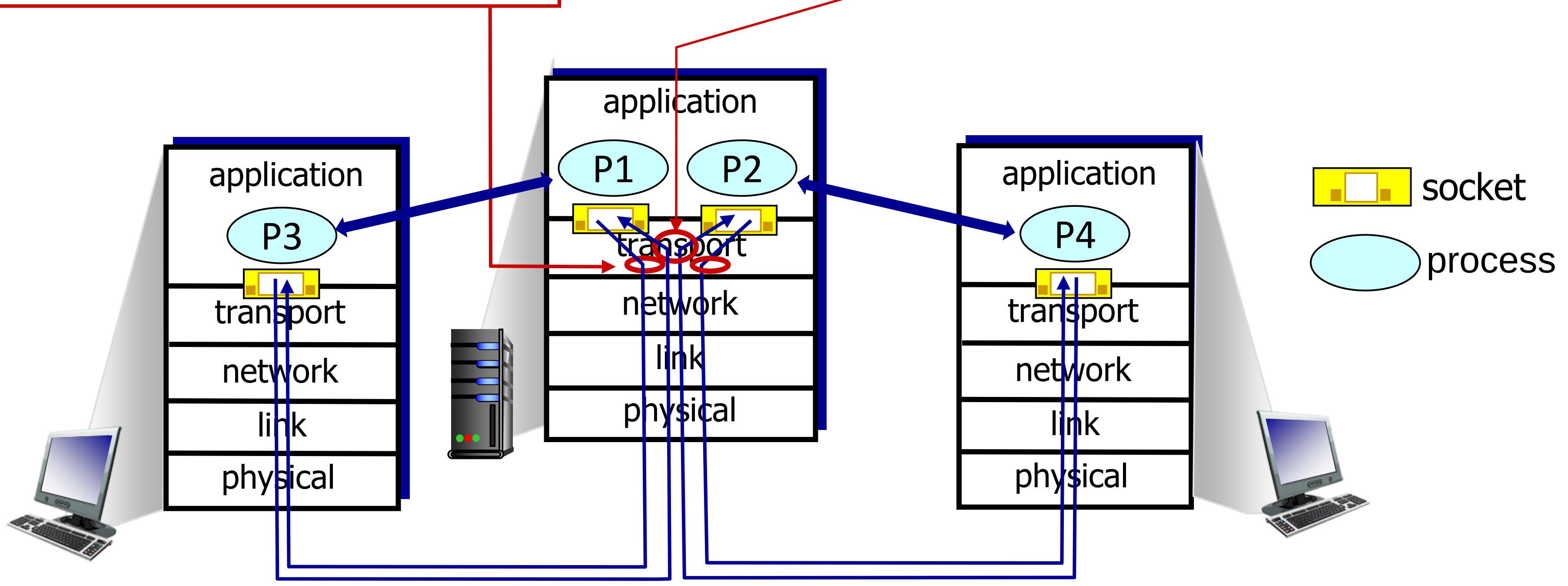
Multiplexing/demultiplexing

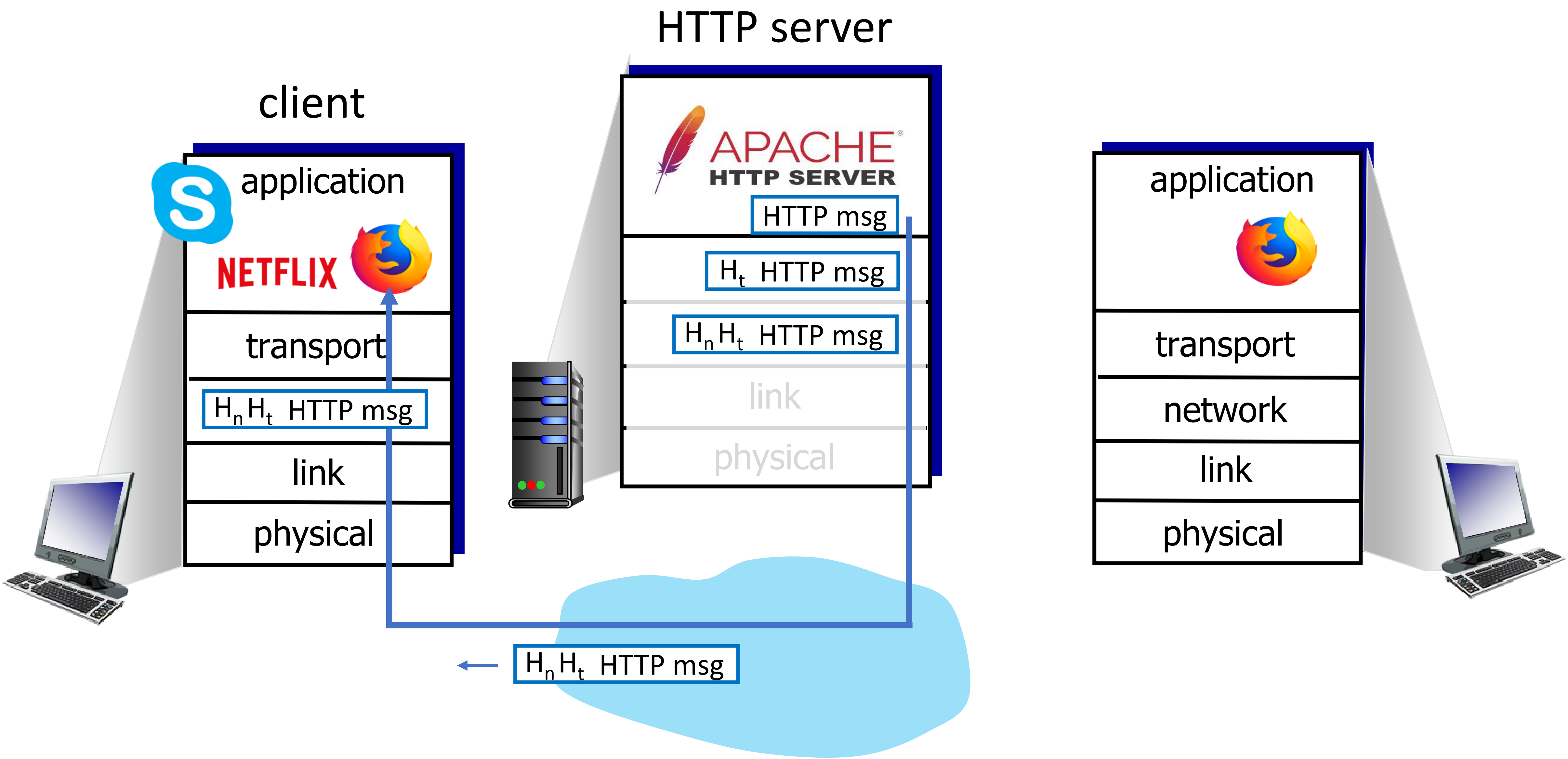
multiplexing as sender:

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

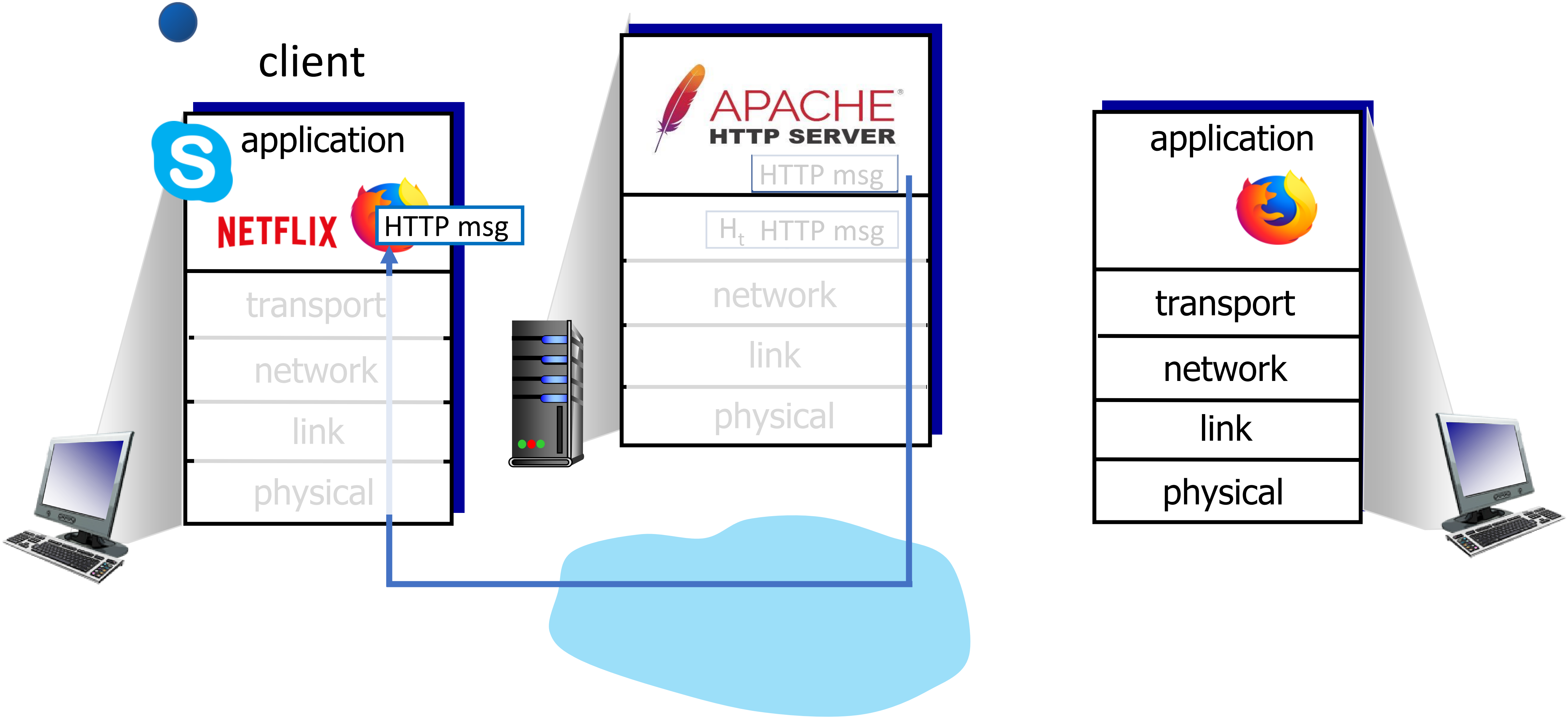
demultiplexing as receiver:

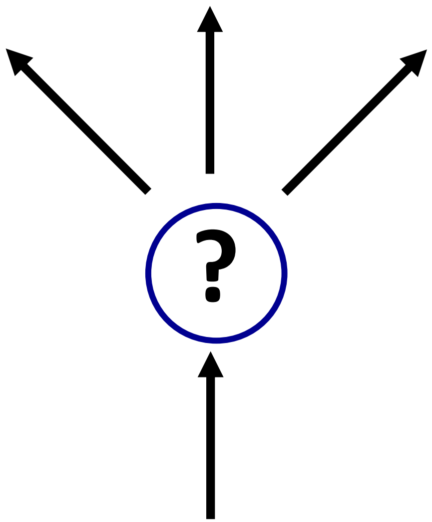
use header info to deliver received segments to correct socket



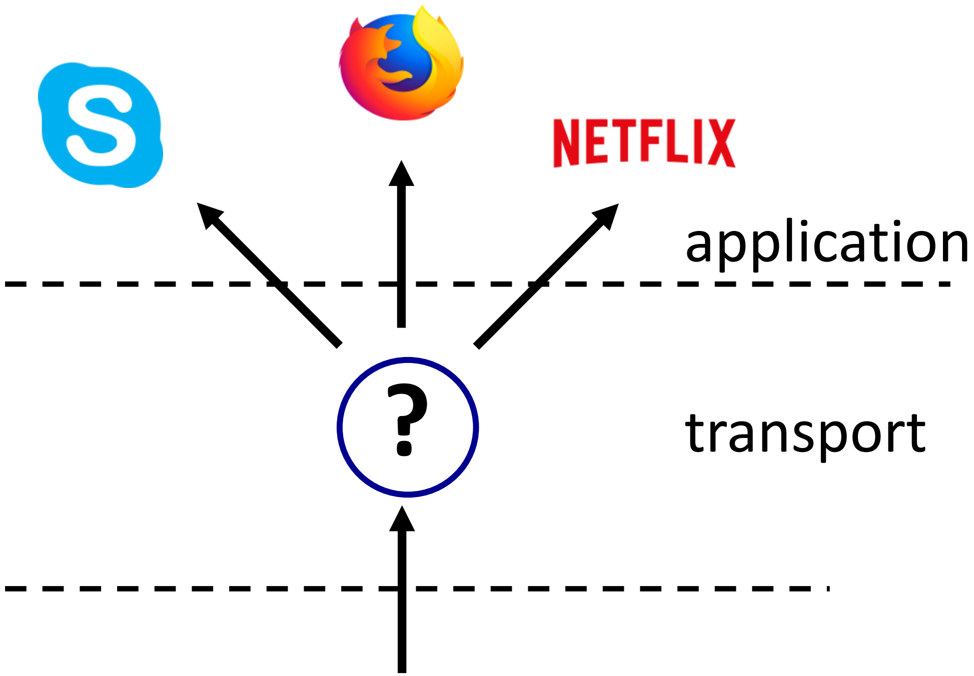

HTTP server
client
application
NETFLIX
transport
$H_n H_t$ HTTP msg
link
physical
APACHE
HTTP SERVER
HTTP msg
H_t HTTP msg
$H_n H_t$ HTTP msg
link
physical
application
transport
network
link
physical
$H_n H_t$ HTTP msg

Q: how did transport layer know to deliver message to Firefox browser process rather than Netflix process or Skype process?

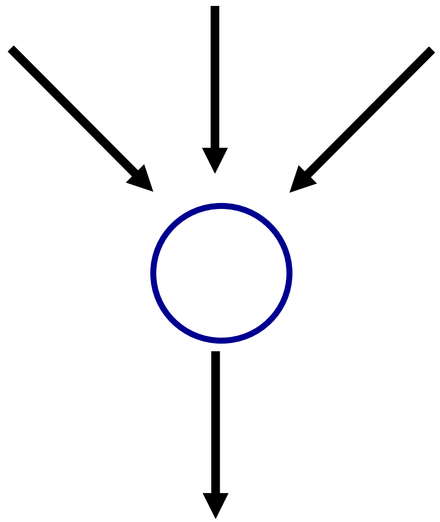




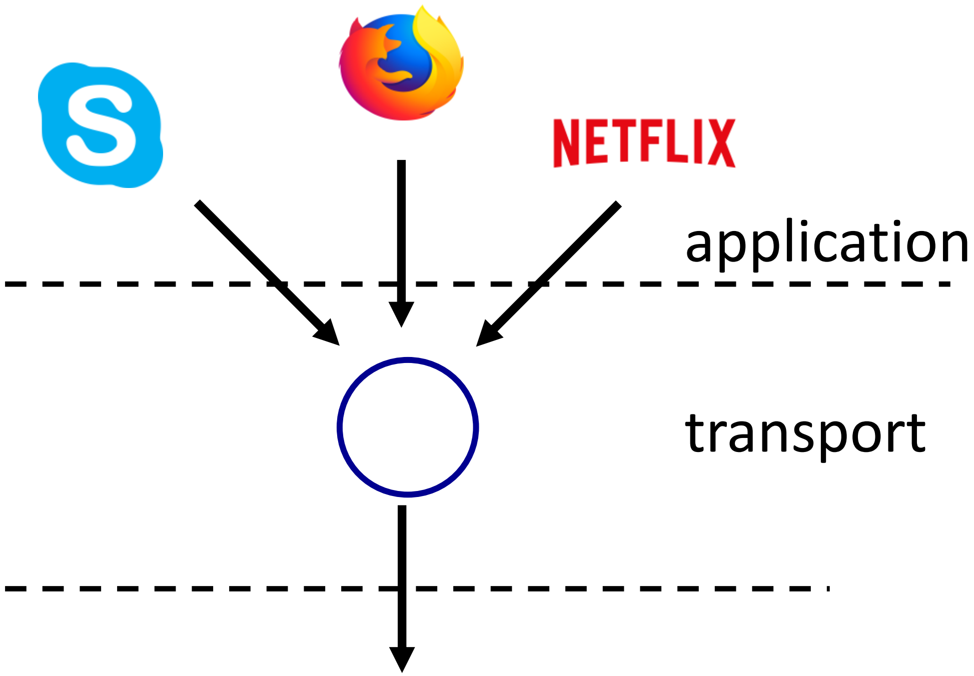
de-multiplexing



de-multiplexing



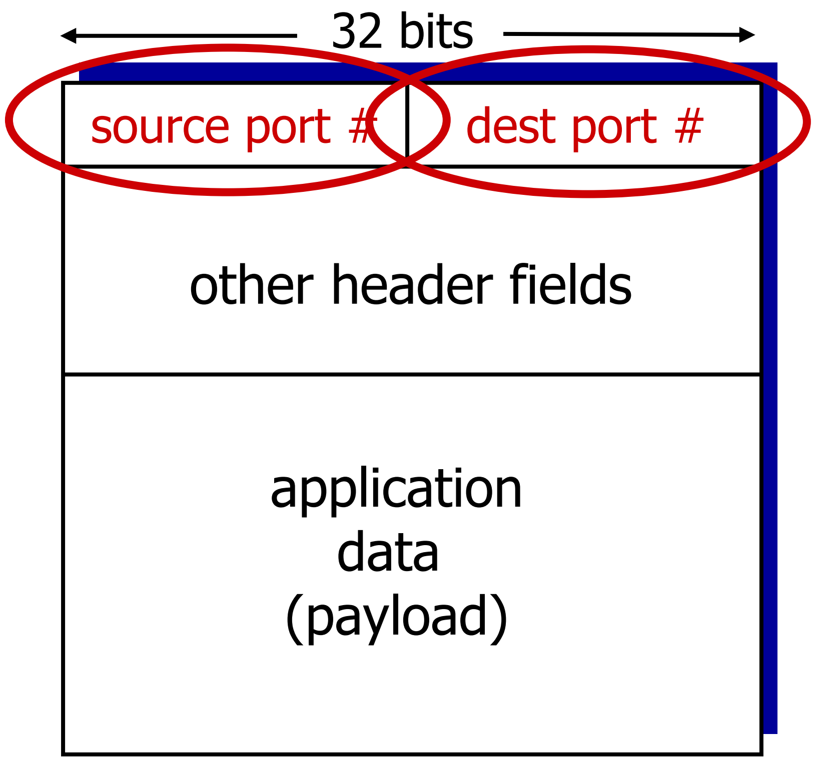
multiplexing



multiplexing

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

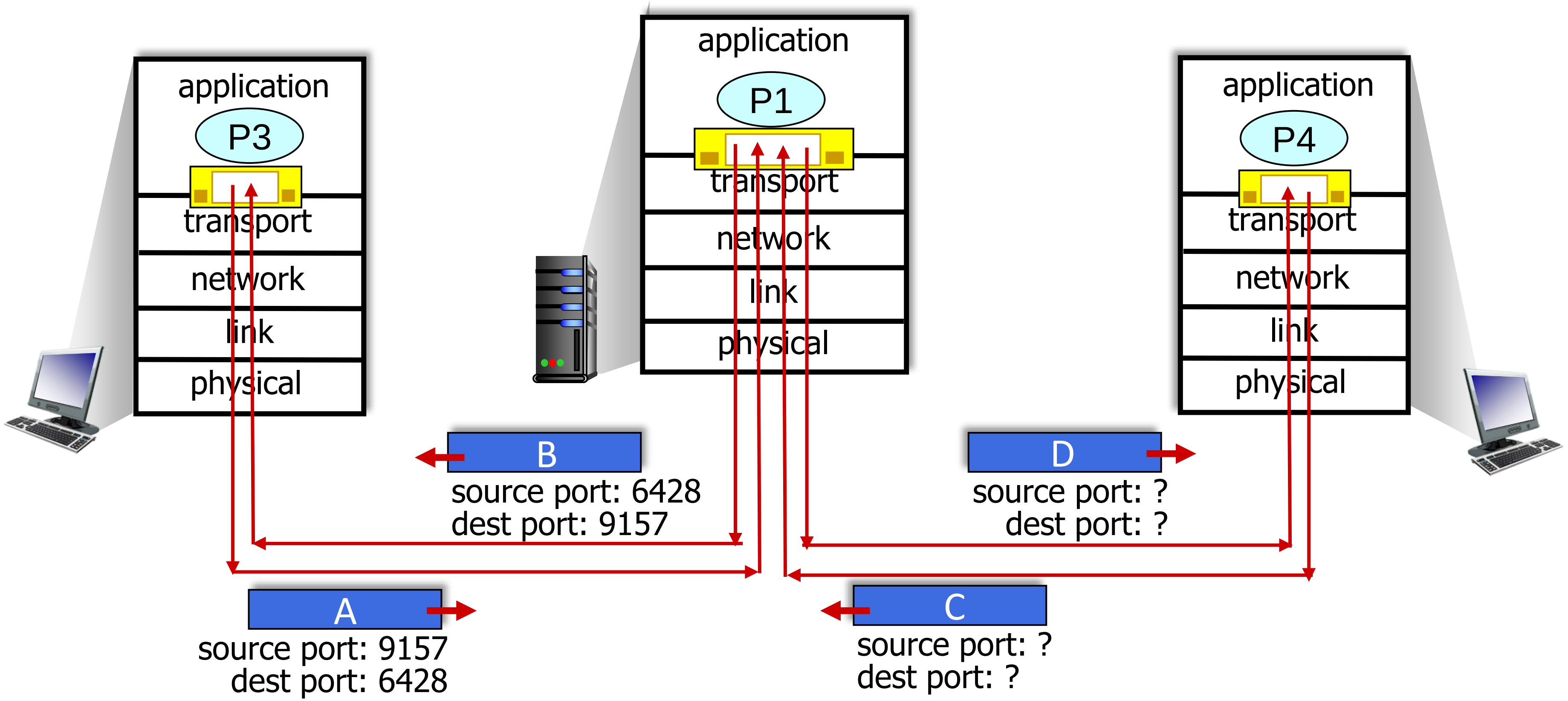
Connectionless demultiplexing

- UDP socket identified by two-tuple:
  - 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

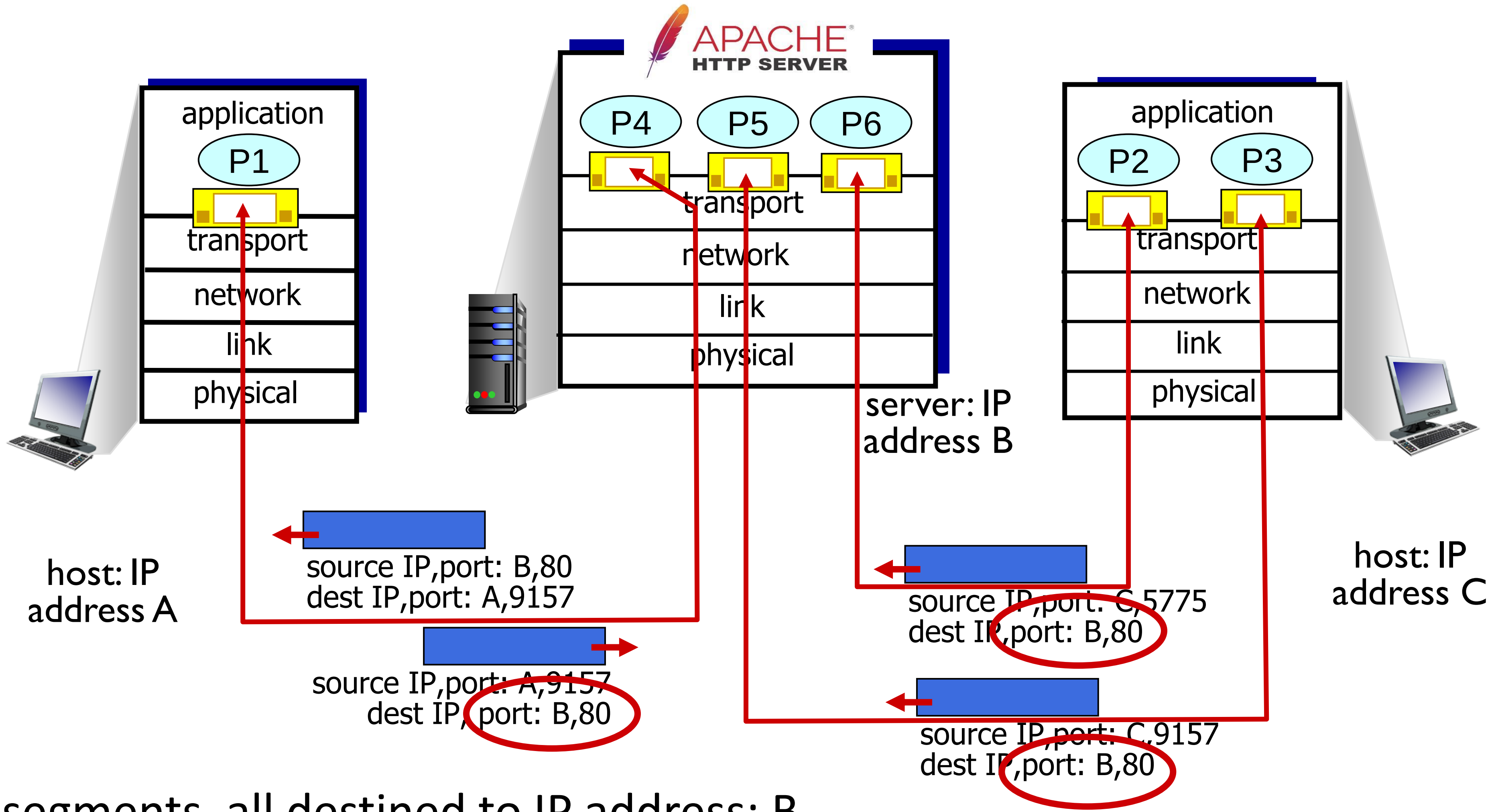
Connectionless demultiplexing: an example



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
- Web servers have different sockets for each connecting client.
  - non-persistent HTTP will have different socket for each request

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

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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

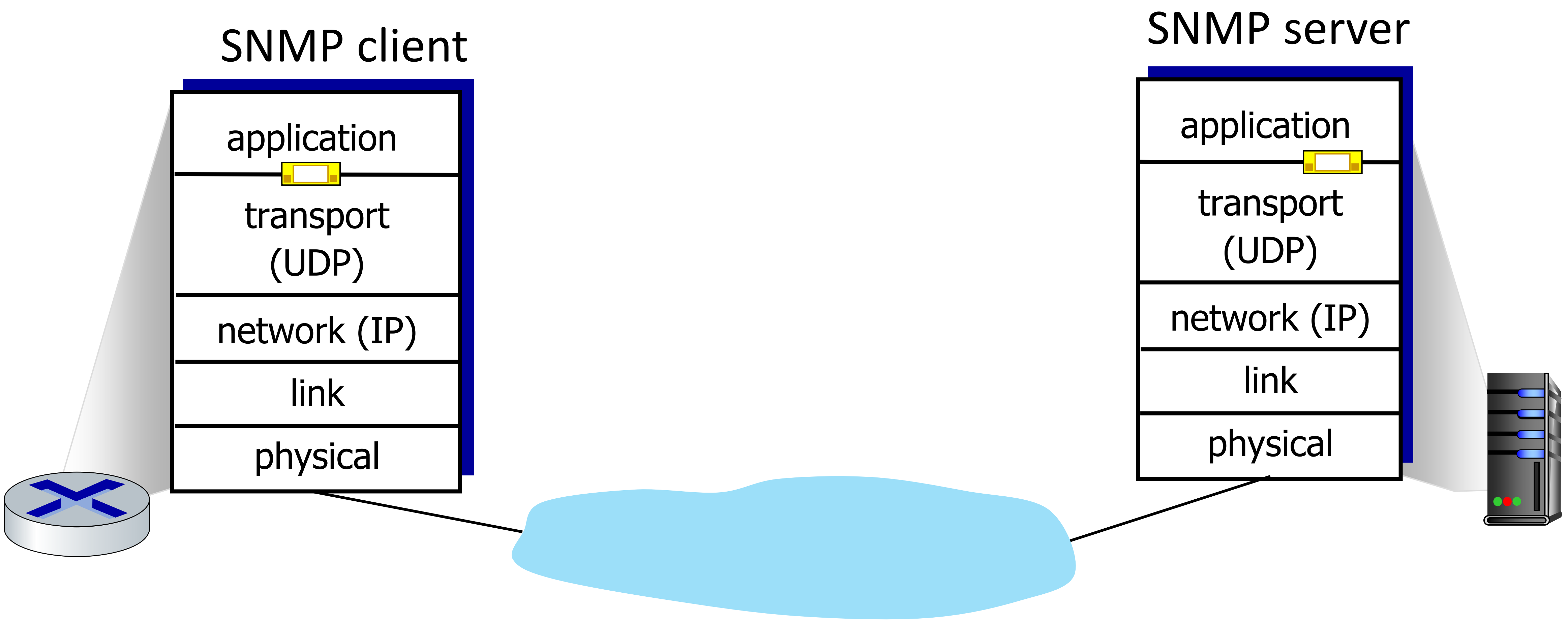
This User Datagram Protocol (UDP) is defined to make available a datagram mode of packet-switched computer communication in the environment of an interconnected set of computer networks. This protocol assumes that the Internet Protocol (IP) [1] is used as the underlying protocol.

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].

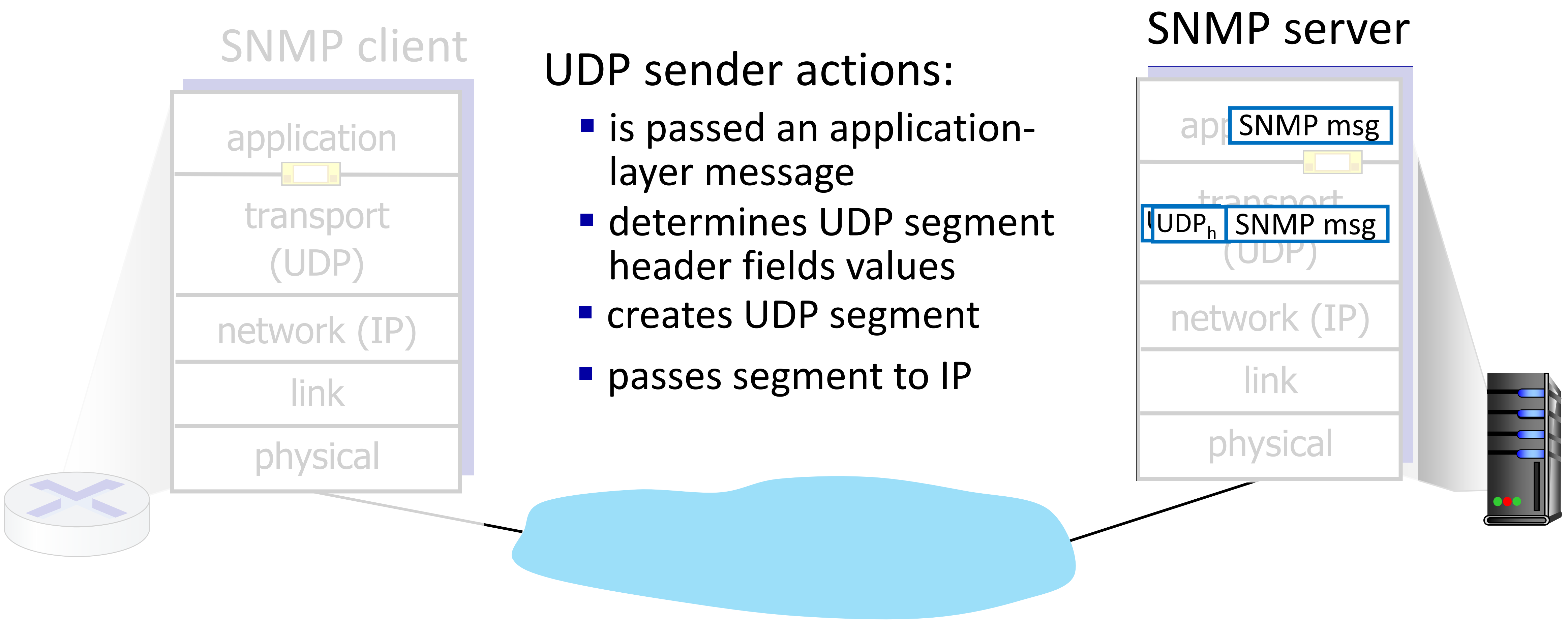
Format

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

UDP: Transport Layer Actions



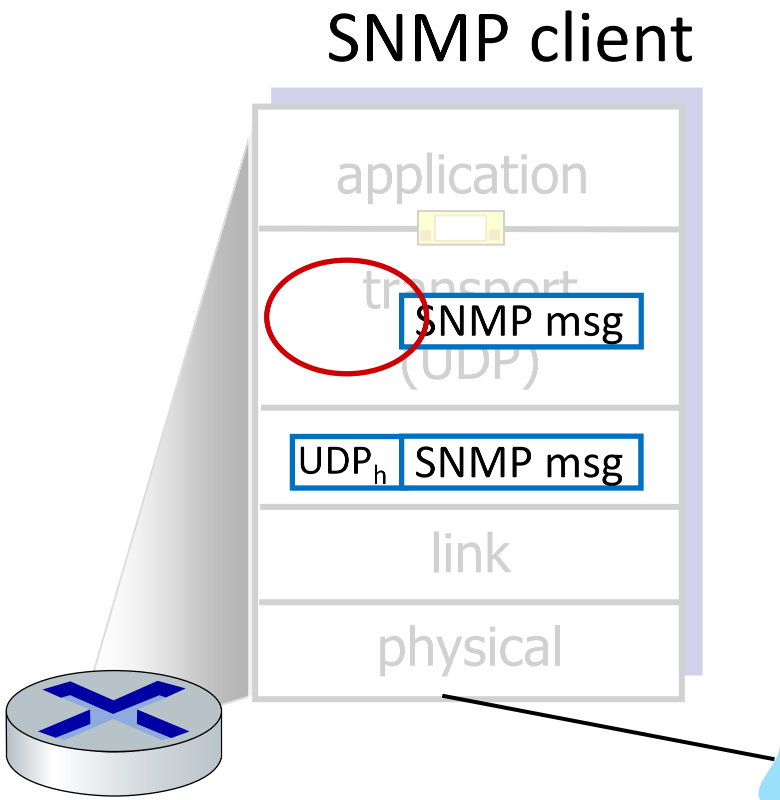
UDP: Transport Layer Actions



UDP sender actions:

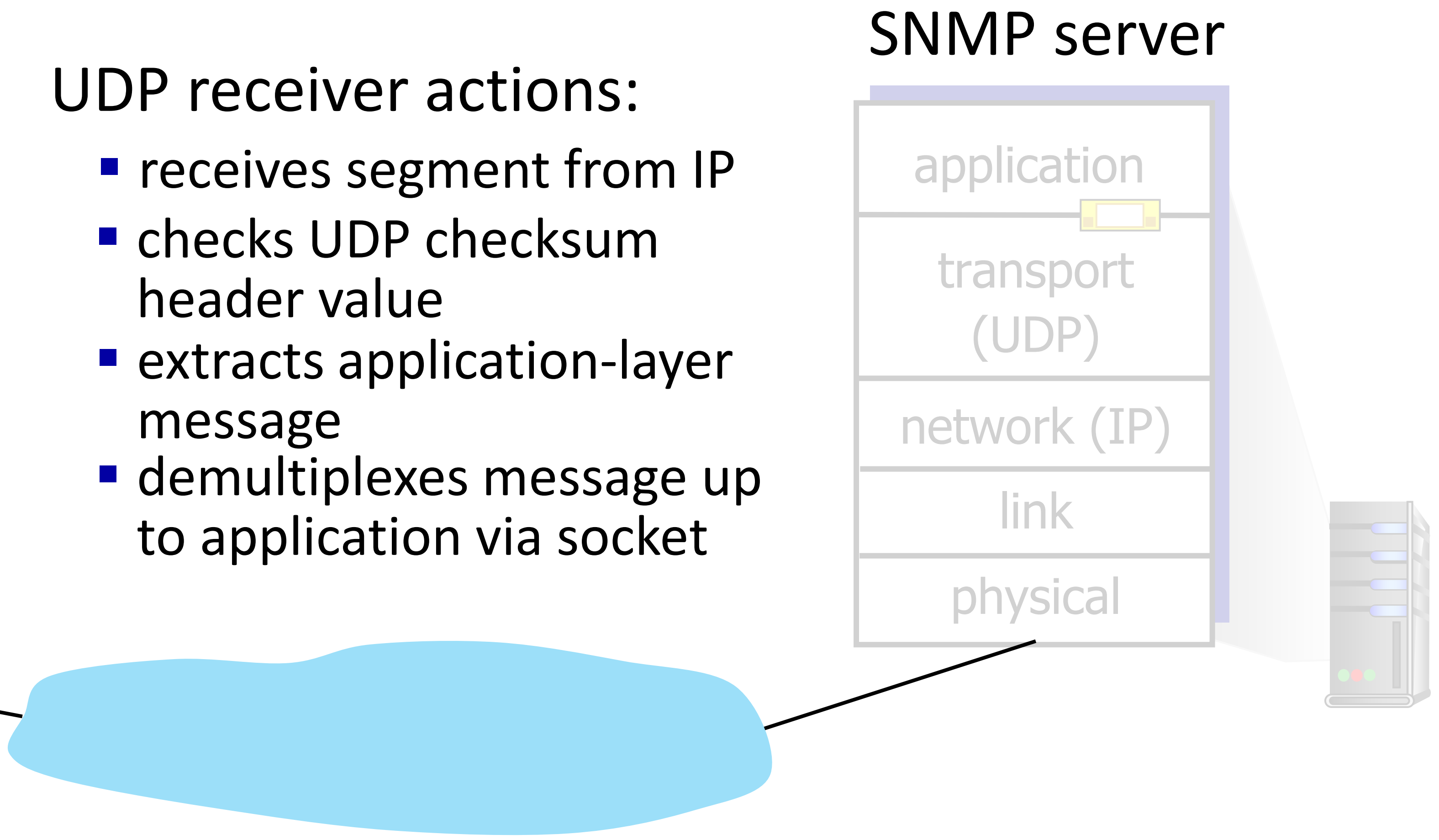
- is passed an application-layer message
- determines UDP segment header fields values
- creates UDP segment
- passes segment to IP

UDP: Transport Layer Actions

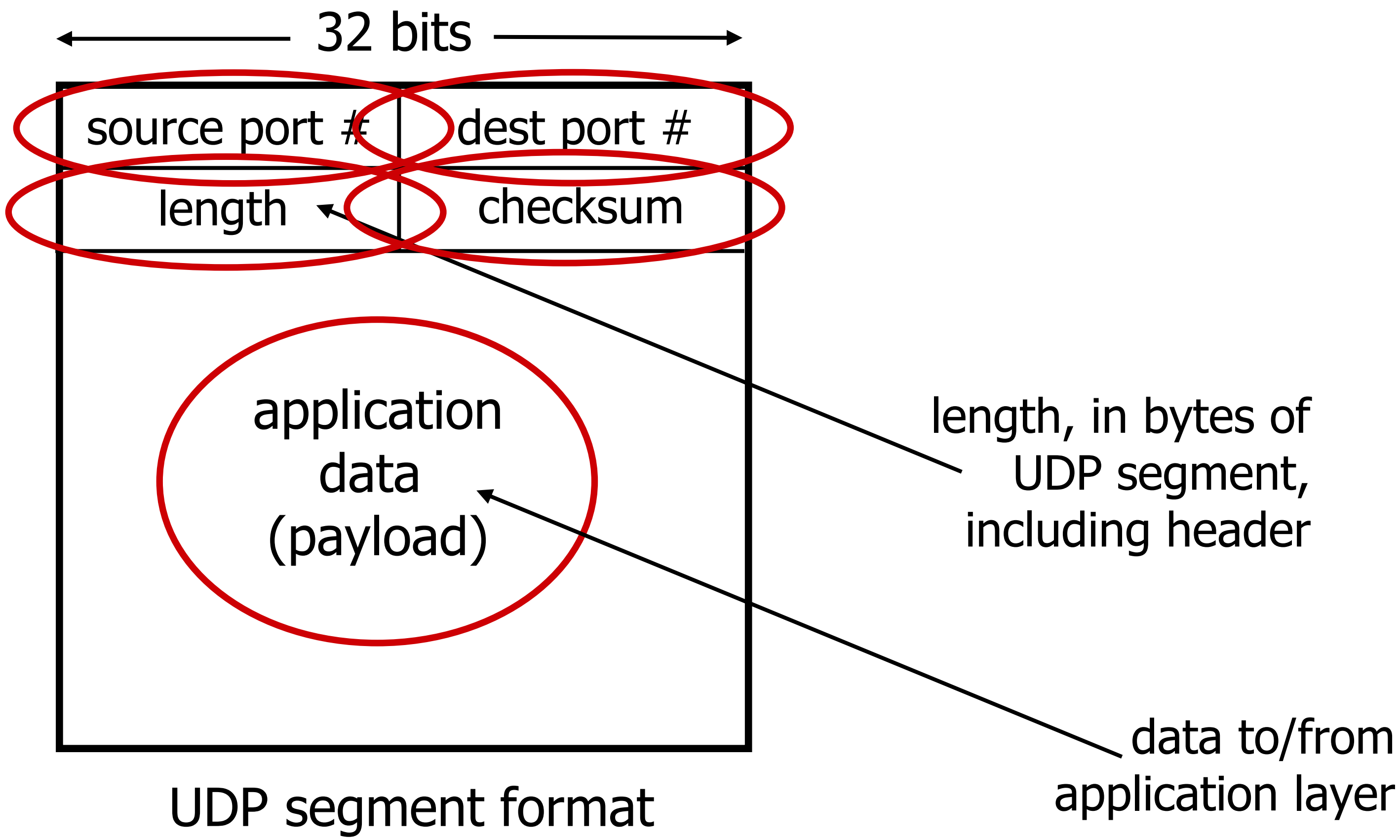


UDP receiver actions:

- 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

* 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
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

0 1
1 0

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)

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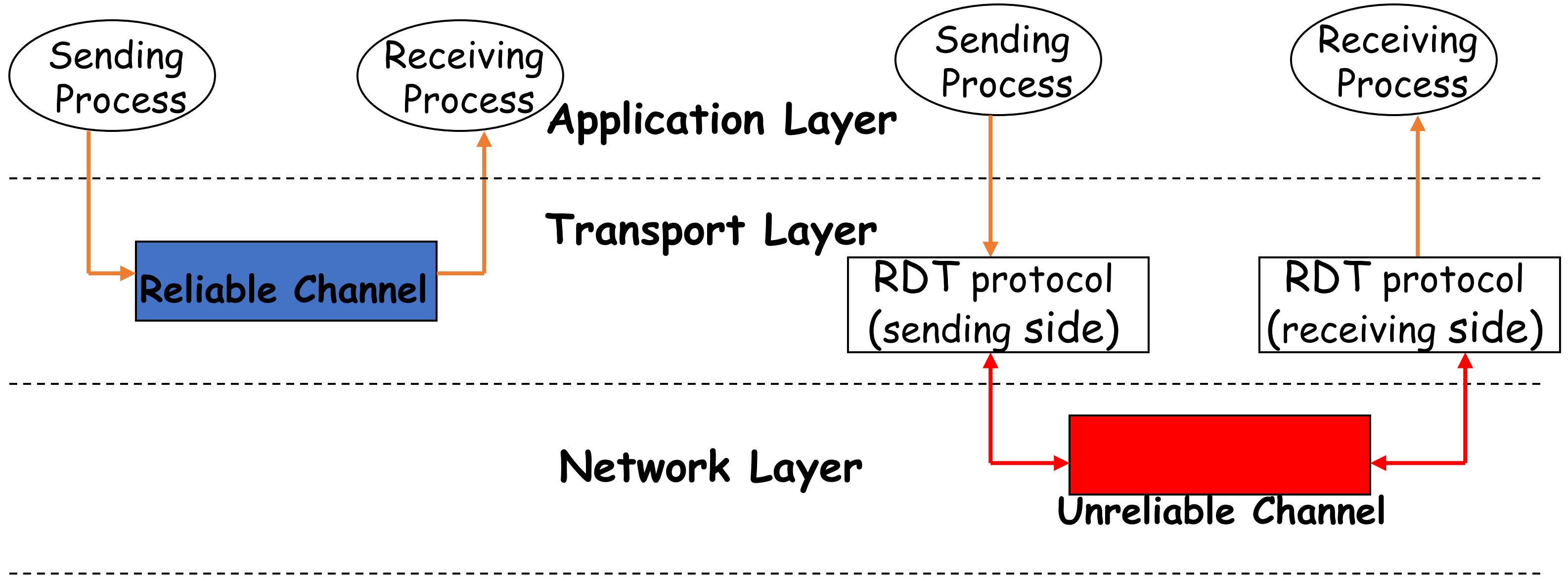
Principles of Reliable Data Transfer

- important in application, transport, and link layers



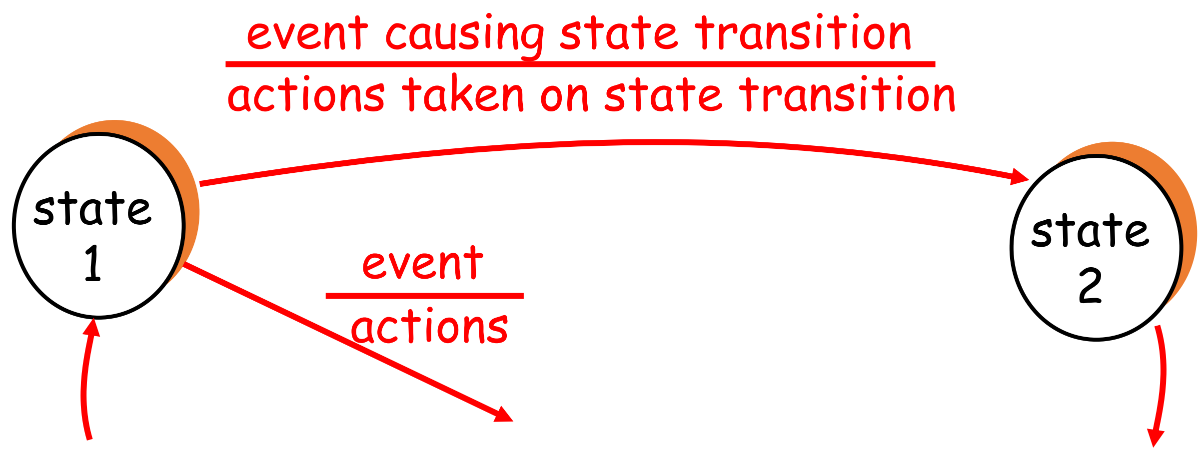
- characteristics of unreliable channel will determine complexity of reliable data transfer protocol (rdt)

Reliable Data Transfer with FSMs

We'll:

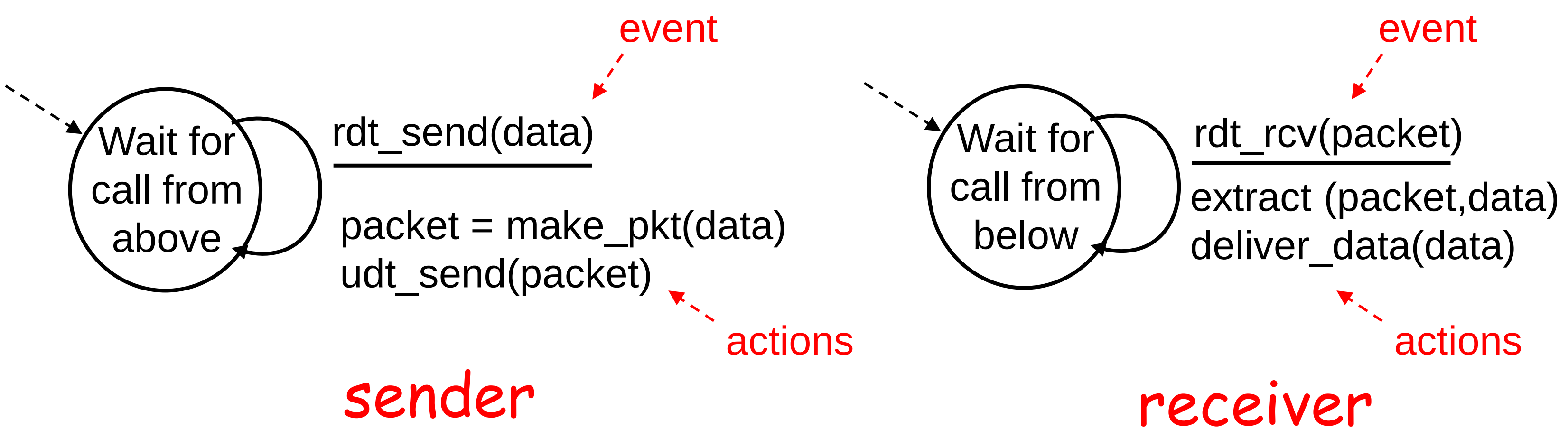
- incrementally develop sender, receiver sides of reliable data transfer protocol (rdt)
- consider only unidirectional data transfer
  - but control info will flow on both directions!
- use finite state machines (FSM) to specify sender, receiver

state: when in this "state" next state uniquely determined by next event



rdt1.0: Reliable Transfer over a Reliable (perfect) Channel

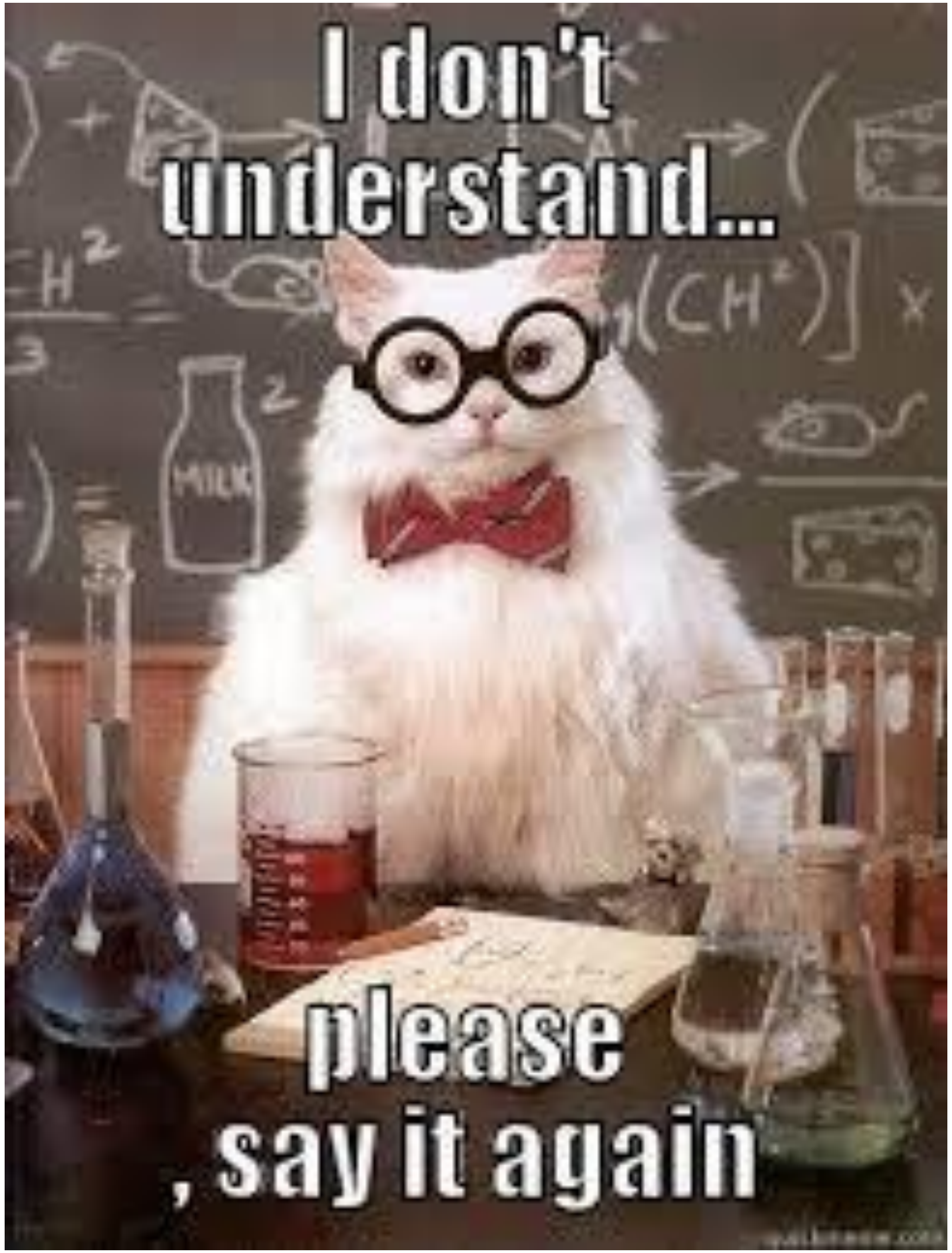
- underlying channel perfectly reliable
  - no bit errors
  - no loss of packets
- separate FSMs for sender, receiver:
  - sender sends data into underlying channel
  - receiver read data from underlying channel



rdt2.0: channel with bit errors
[stop & wait protocol]

- Assumptions
  - All packets are received
  - Packets may be corrupted (i.e., bits may be flipped)
  - Checksum to detect bit errors

How do humans recover from "errors" during conversation?


I don't understand...
please
, say it again

rdt2.0: channel with bit errors
[stop & wait protocol]

- Assumptions
  - All packets are received
  - Packets may be corrupted (i.e., bits may be flipped)
  - Checksum to detect bit errors

- How to recover from errors? ::: Use ARQ mechanism
  - *acknowledgements (ACKs):* receiver explicitly tells sender that packet received correctly
  - *negative acknowledgements (NAKs):* receiver explicitly tells sender that packet had errors
  - sender retransmits packet on receipt of NAK

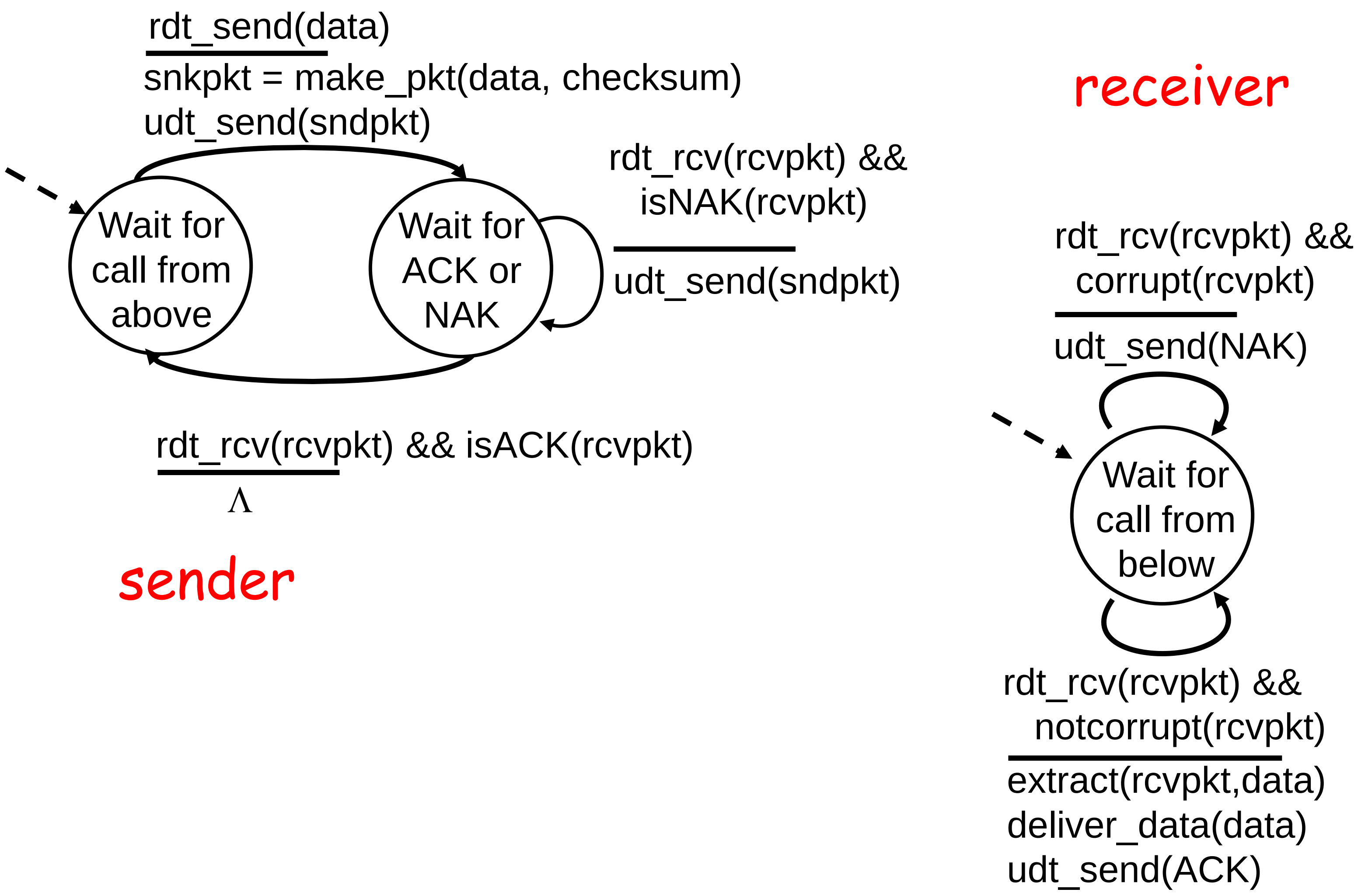
- in rdt2.0 (beyond rdt1.0):
  - error detection
  - feedback: control msgs (ACK,NAK) from receiver to sender

rdt2.0: channel with bit errors
[stop & wait protocol]

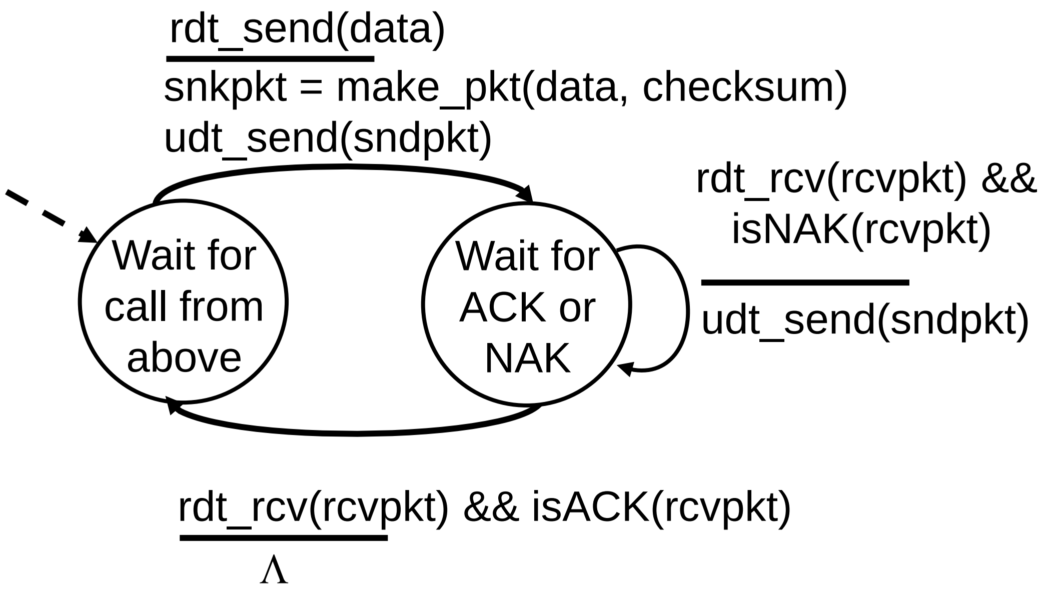
- Three additional protocol capabilities are required in ARQ protocols:
  (1) Error detection
    - Extra bits are required
  (2) Receiver feedback
    - Using ACK / NAK
  (3) Retransmission

rdt2.0: FSM specification

[stop & wait protocol]



rdt2.0: Observations



sender

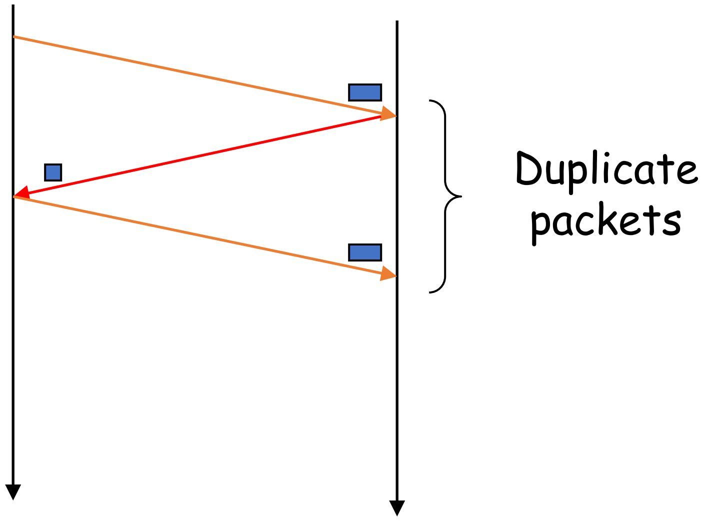
stop and wait

sender sends one packet,
then waits for receiver response

1. A stop-and-Wait protocol

2. What happens when ACK or NAK has bit errors?

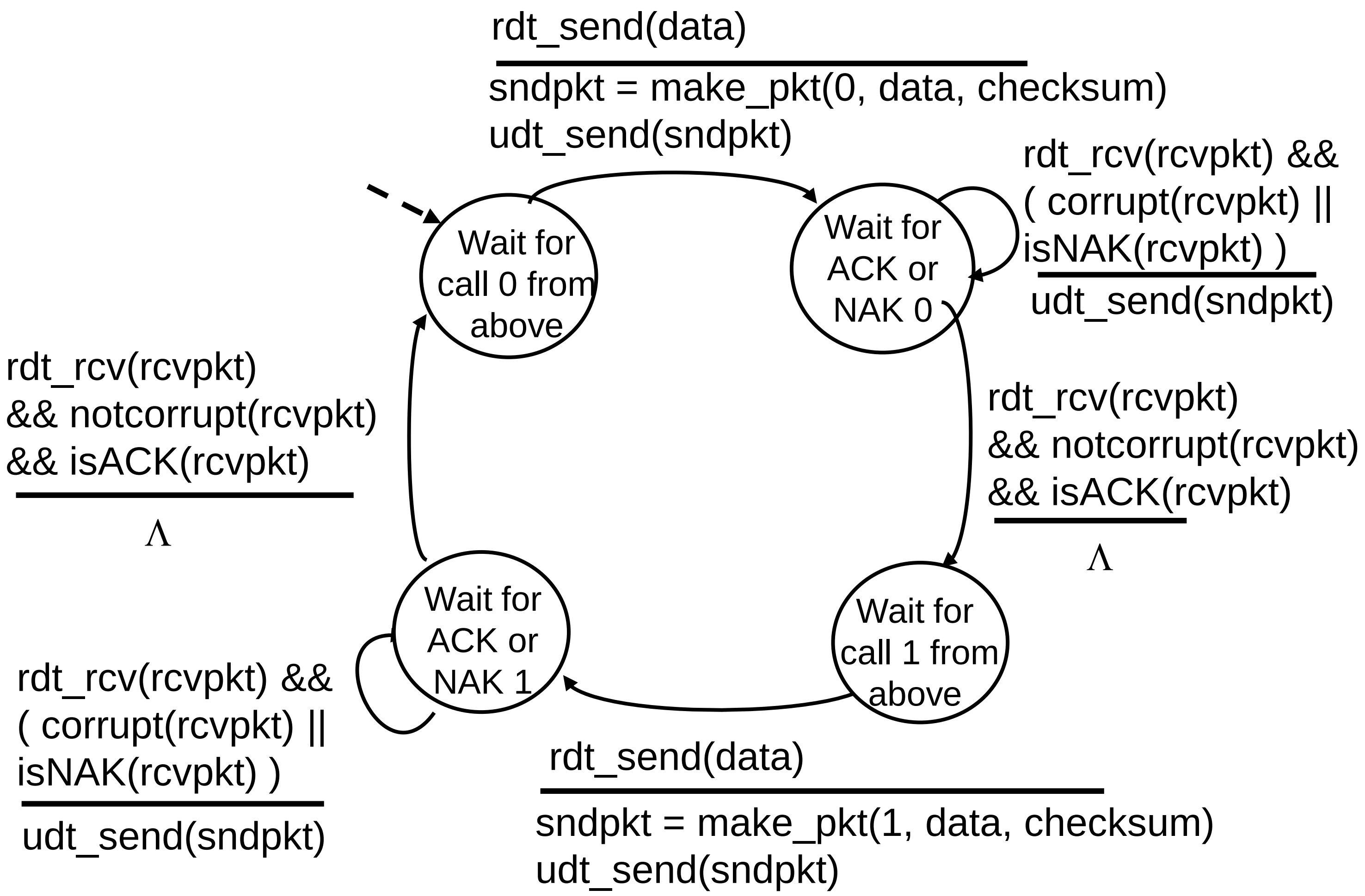
Approach 1: resend the current data packet?



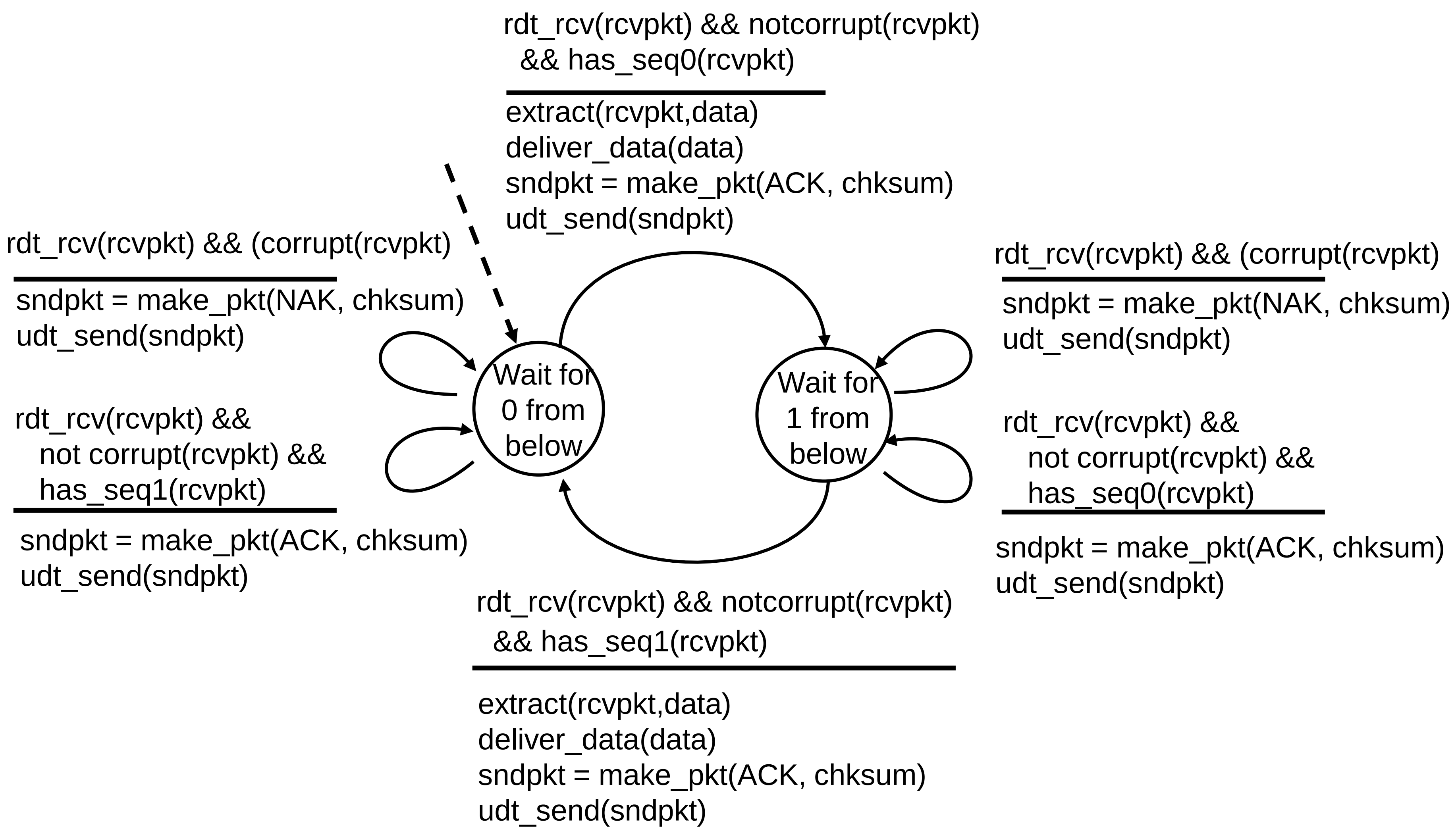
Handling Duplicate Packets

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

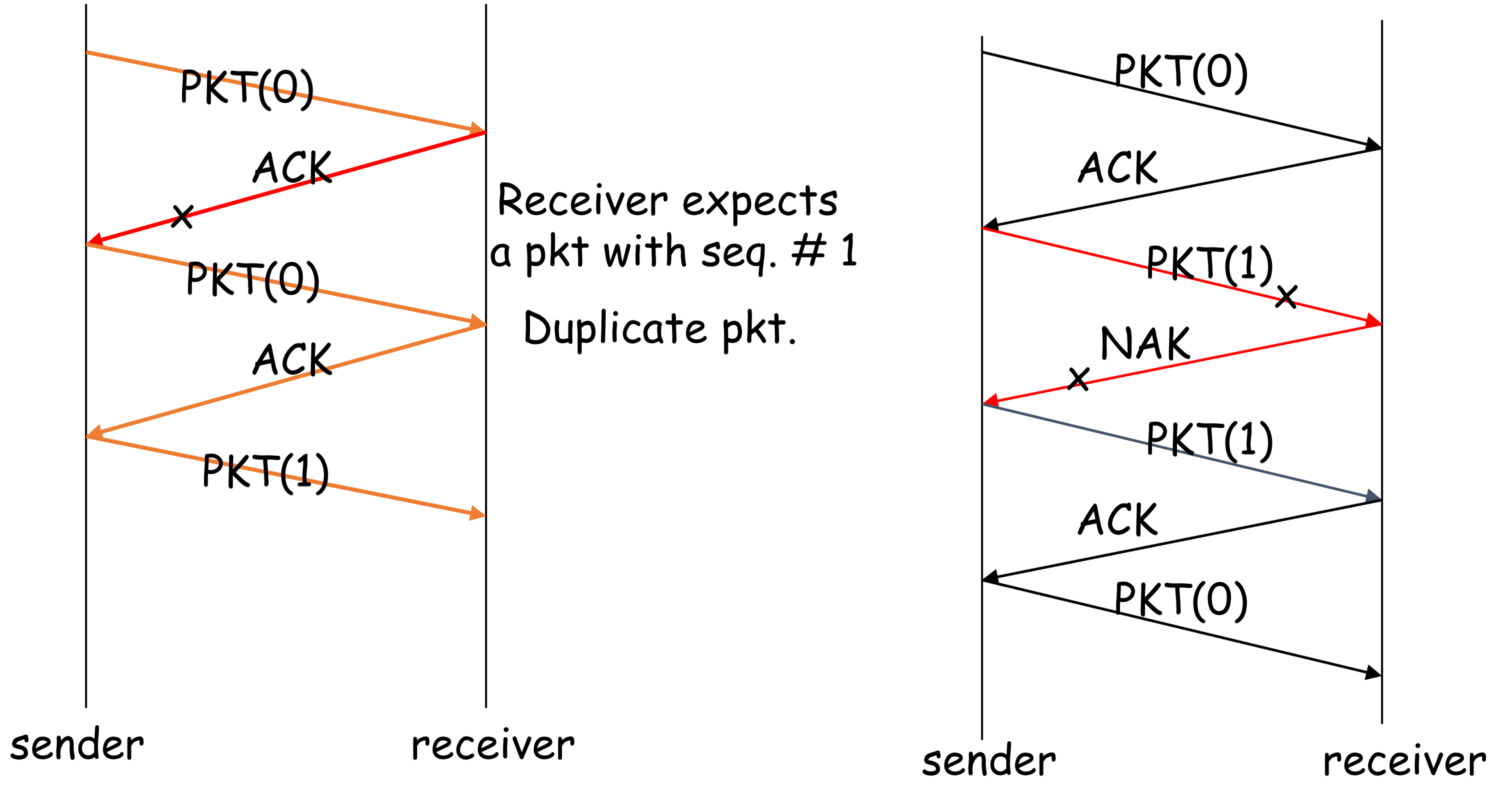
rdt2.1: sender, handles garbled ACK/NAKs



rdt2.1: receiver, handles garbled ACK/NAKs



rtd2.1: examples



rdt2.1: summary

Sender:

- seq # added to packet
- two seq. #'s (0,1) will suffice. Why?
- must check if received ACK/NAK corrupted
- twice as many states
  - state must "remember" whether "current" packet has 0 or 1 seq. #

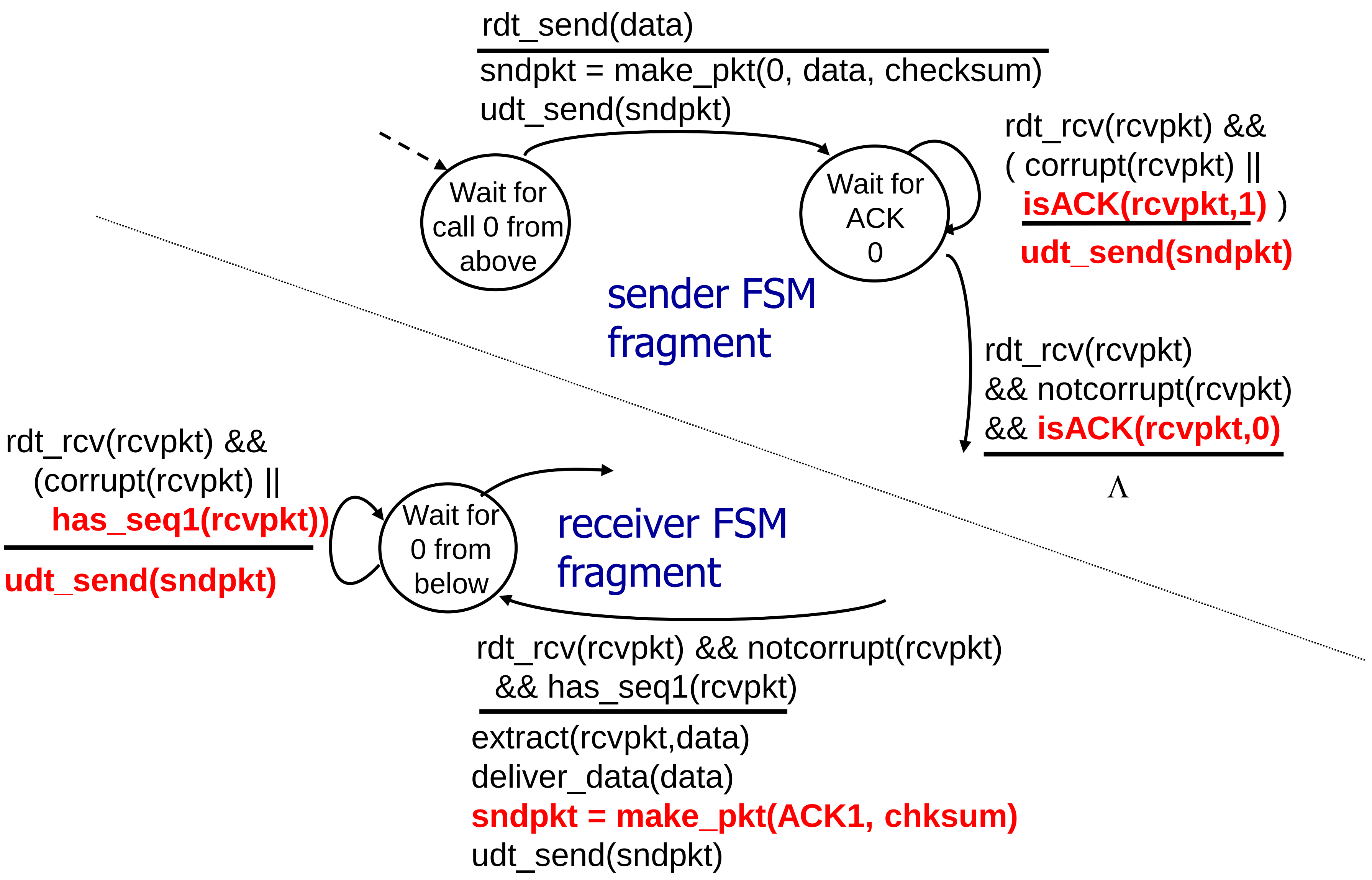
Receiver:

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

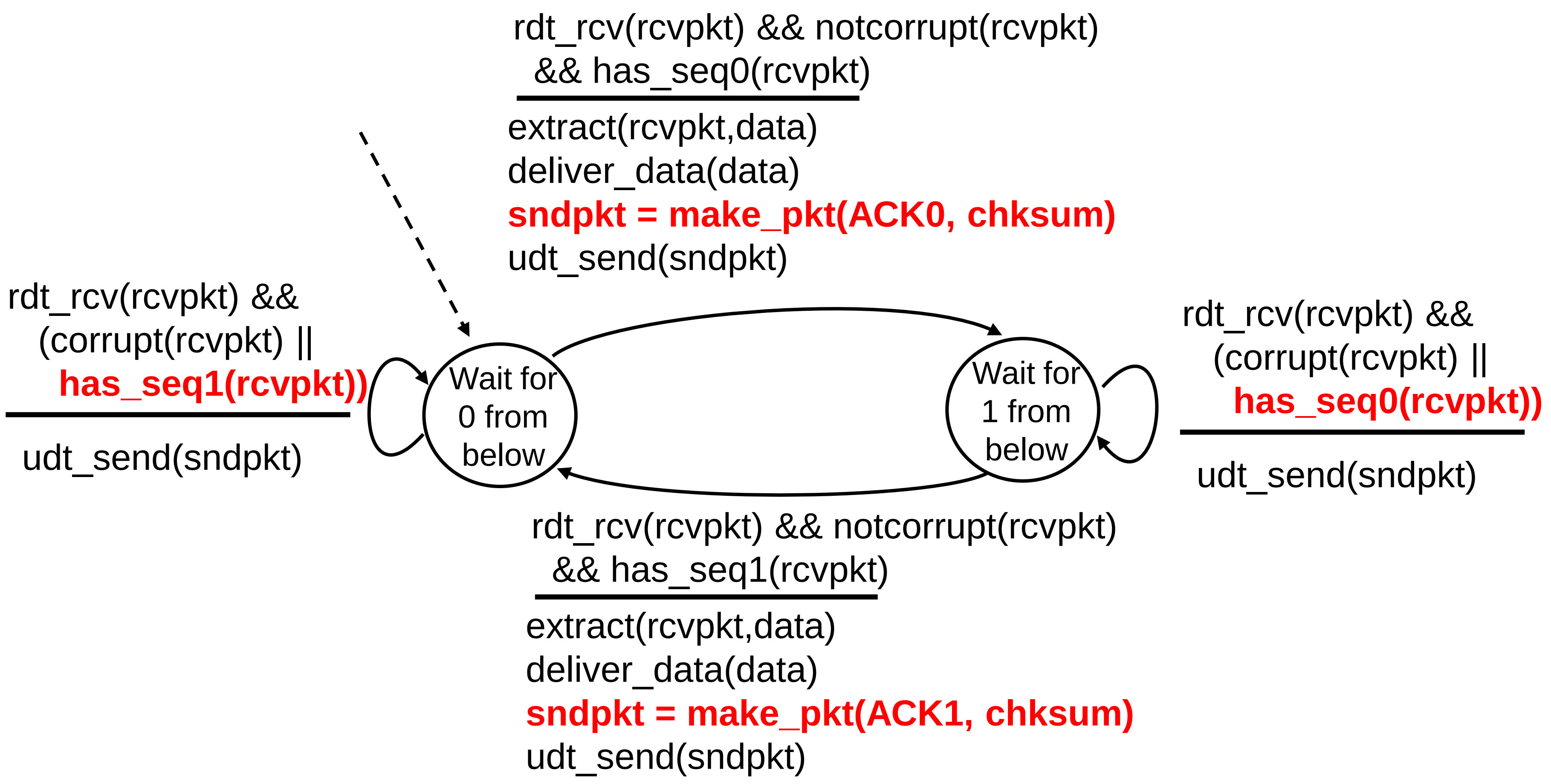
rdt2.2: a NAK-free protocol

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

rdt2.2: sender, receiver fragments



rdt2.2: Receiver



rdt3.0:The case of “Lossy” Channels

- Assumption: underlying channel can also lose packets (data or ACKs)
  - checksum, seq. #, ACKs, retransmissions will be of help … but not enough

- Approach: sender waits “reasonable” amount of time for ACK (a Time-Out)
  - Time-out value?
  - Possibility of duplicate packets/ACKs?

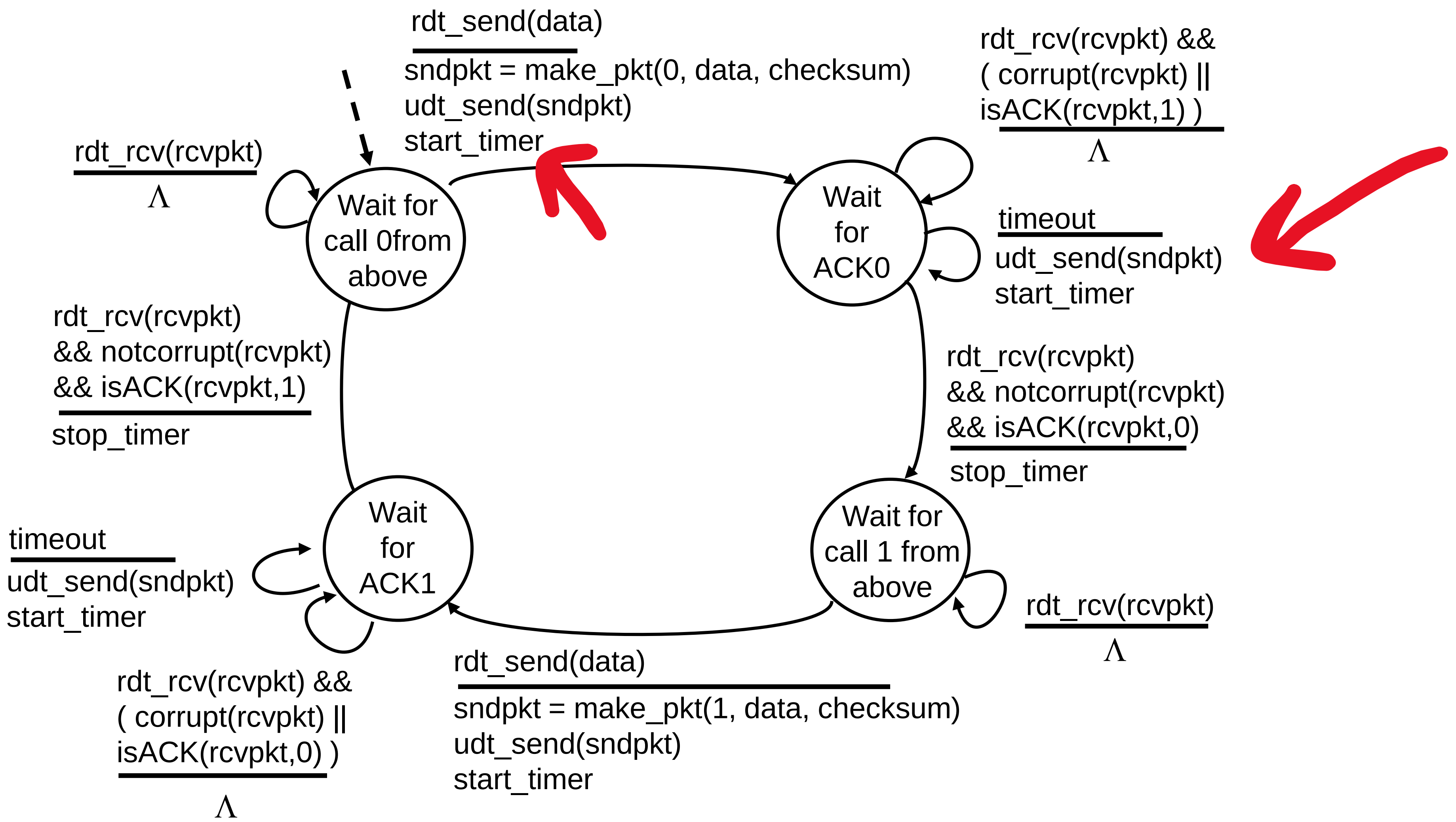
- if packet (or ACK) just delayed (not lost):
  - retransmission will be duplicate, but use of seq. #’s already handles this
  - receiver must specify seq # of packet being ACKed

- Countdown timer is required.

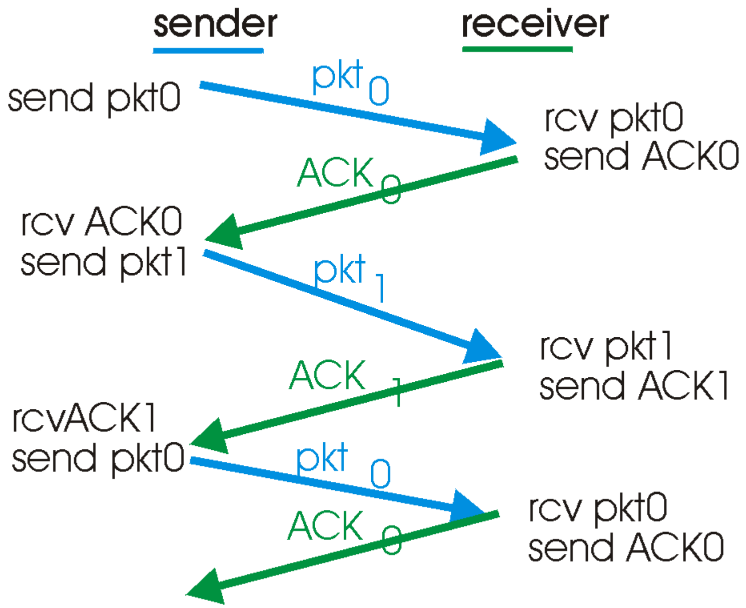
rdt3.0:The case of "Lossy" Channels

- The sender will need to be able
  1. (1) Start the timer each time a packet is sent
  2. (2) Respond to a timer interrupt
  3. (3) Stop the timer

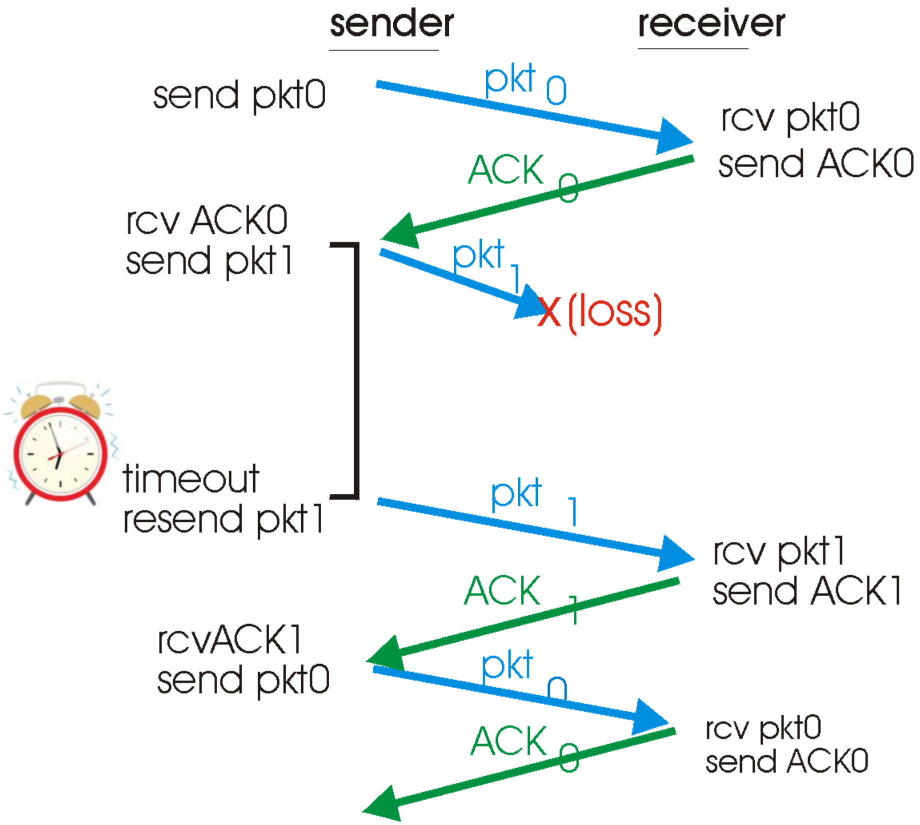
rdt3.0 sender



rdt3.0 in action

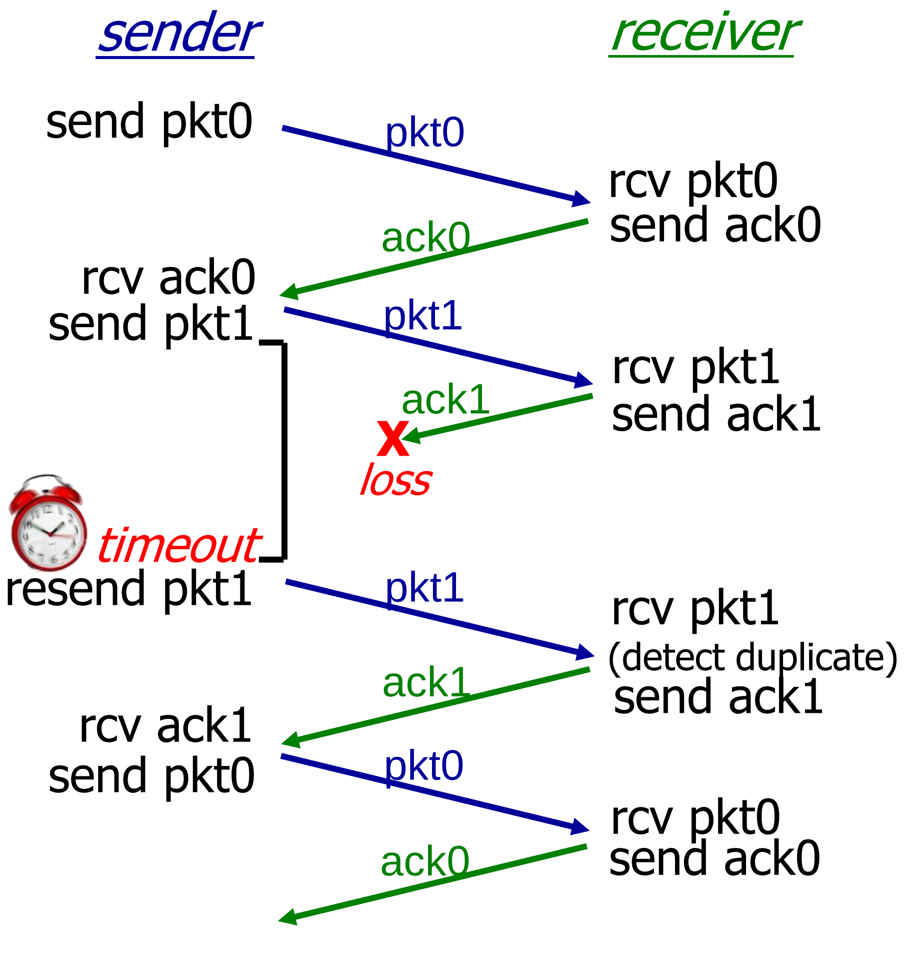


(a) operation with no loss

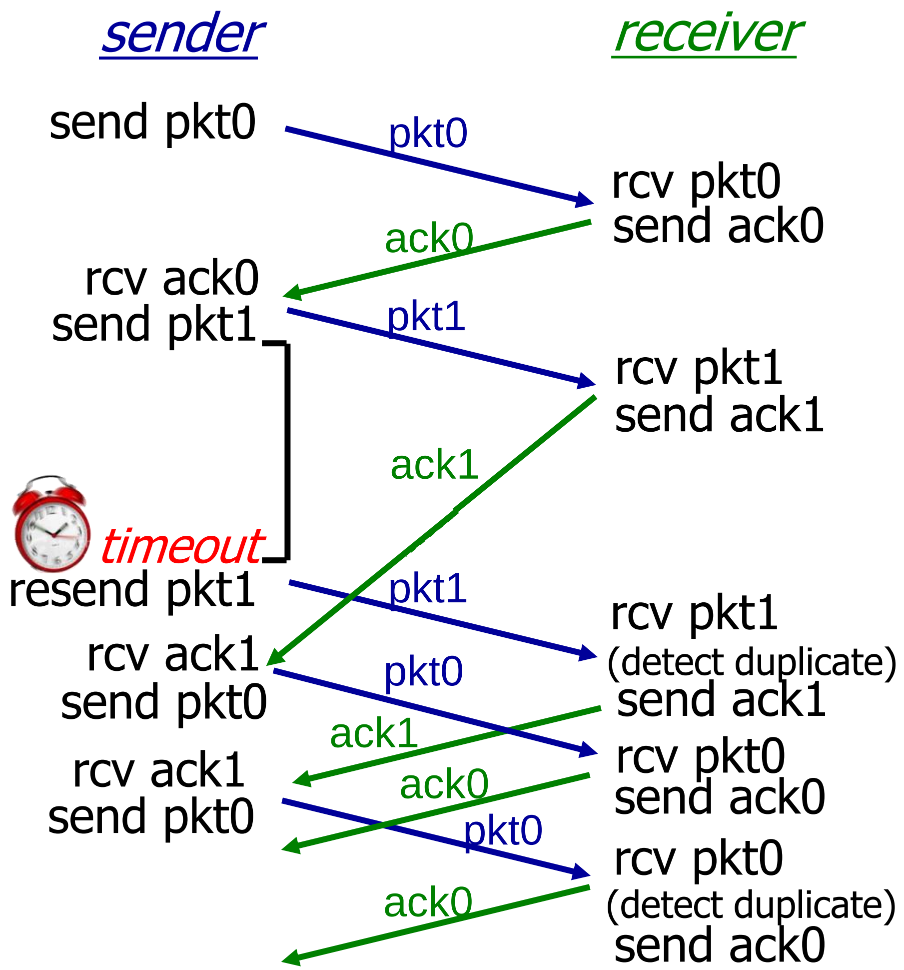


(b) lost packet

rdt3.0 in action

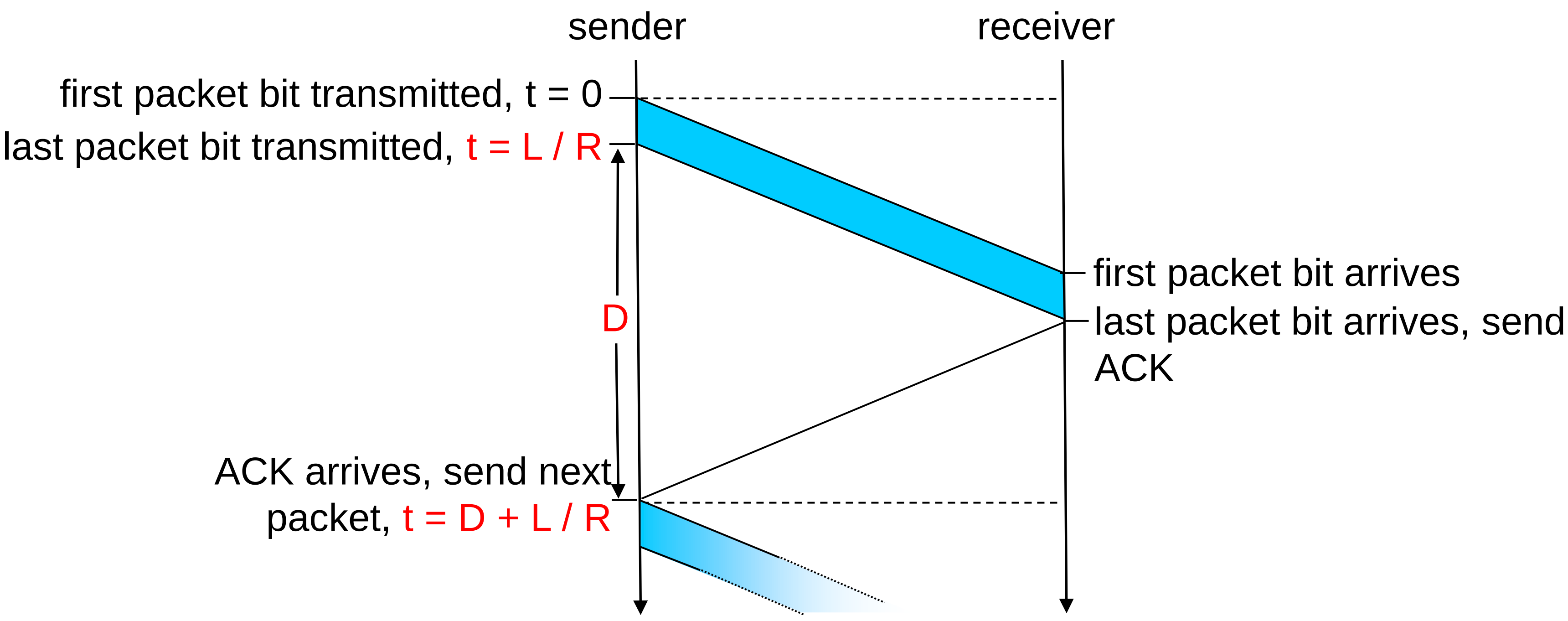


(c) ACK loss



(d) premature timeout/ delayed ACK

stop-and-wait operation



Protocol rdt3.0 is a stop-and-wait protocol.

Performance of rdt3.0

- rdt3.0 is correct, but performance stinks
- e.g.: 1 Gbps link, 15 ms prop. delay, 8000 bit packet:

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

  - U $_{\text{sender}}$: *utilization* – fraction of time sender busy sending

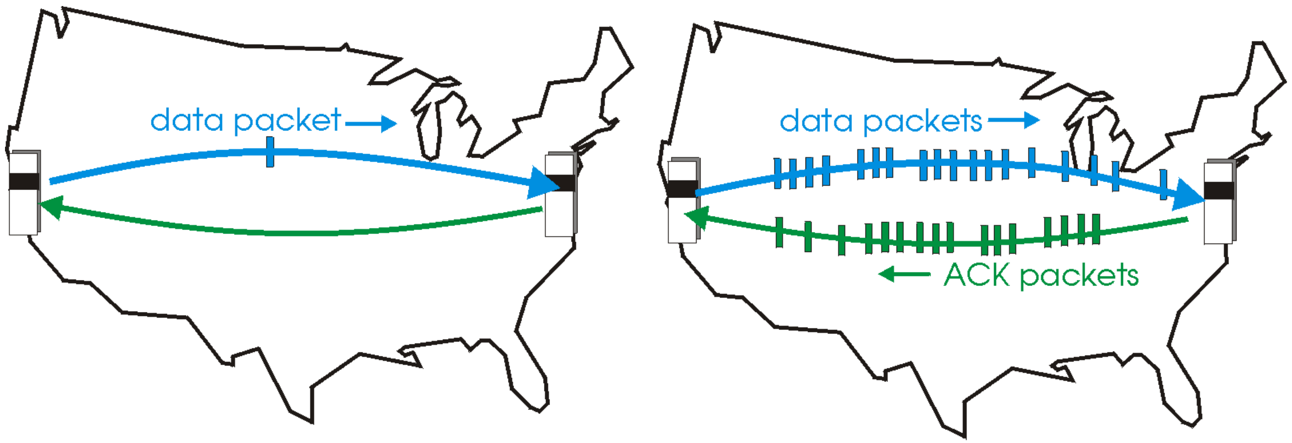
$$U_{sender} = \frac{L / R}{RTT + L / R} = \frac{.008}{30.008} = 0.00027$$

  - if RTT=30 msec, 1 KB pkt every 30 msec: 33 KB/sec throughput over 1 Gbps link
- network protocol limits use of physical resources!

*** U = Utilization ***

Pipelined Protocols

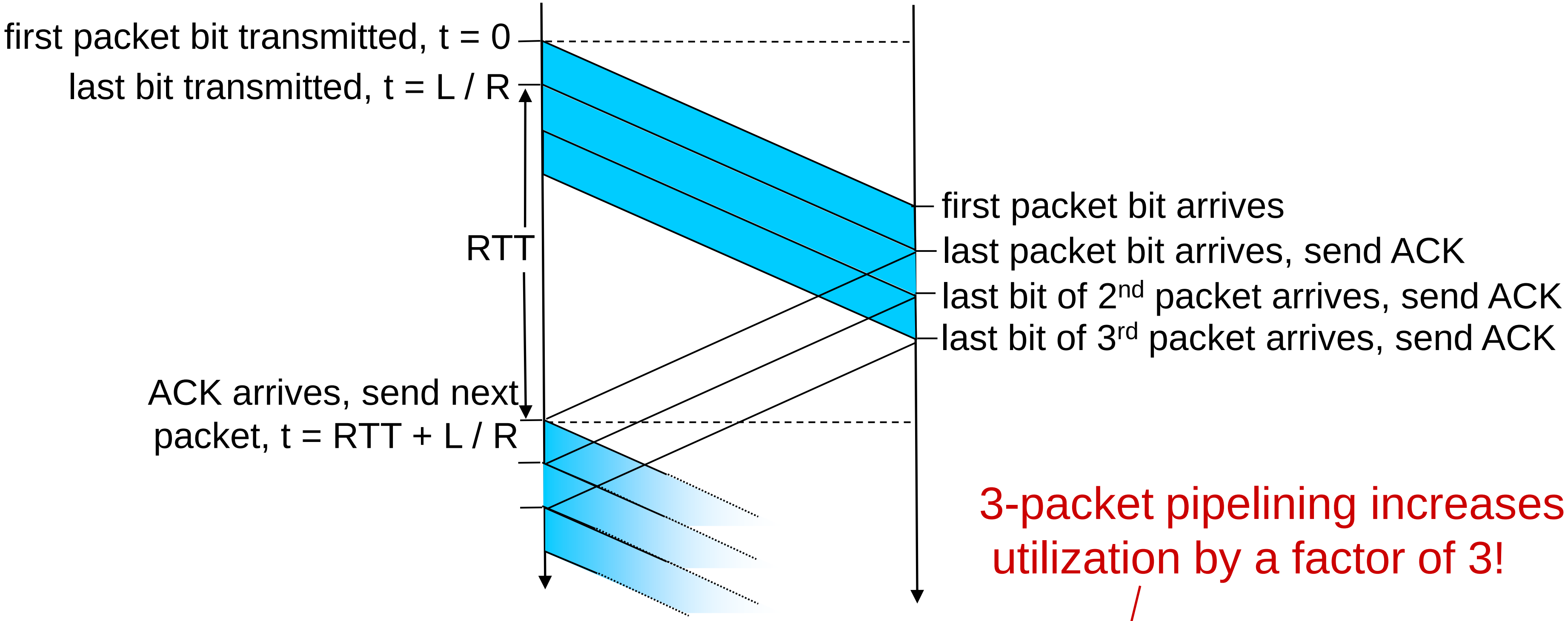
- Pipelining: sender allows multiple, "in-flight", yet-to-be-acknowledged pkts
  - range of sequence numbers must be increased
  - buffering at sender and/or receiver



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

- Two generic forms of pipelined protocols: *go-Back-N and selective repeat*

Pipelining: increased utilization



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

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

Pipelined protocols: overview

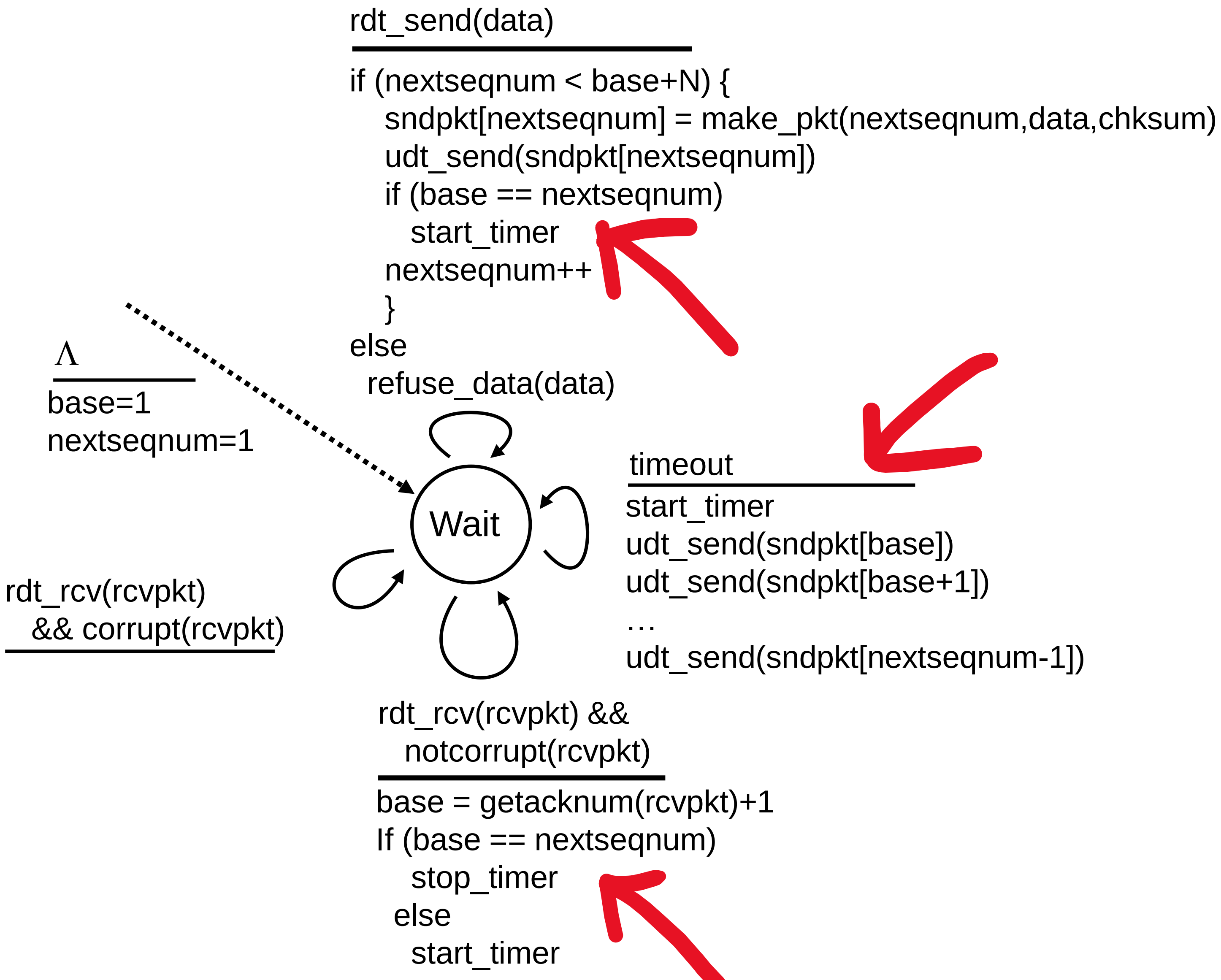
Go-back-N:

- sender can have up to N unacked packets in pipeline
- receiver only sends *cumulative ack*
  - doesn't ack packet if there's a gap
- sender uses a single timer for oldest unacked packet
  - when timer expires, retransmit *all* unacked packets

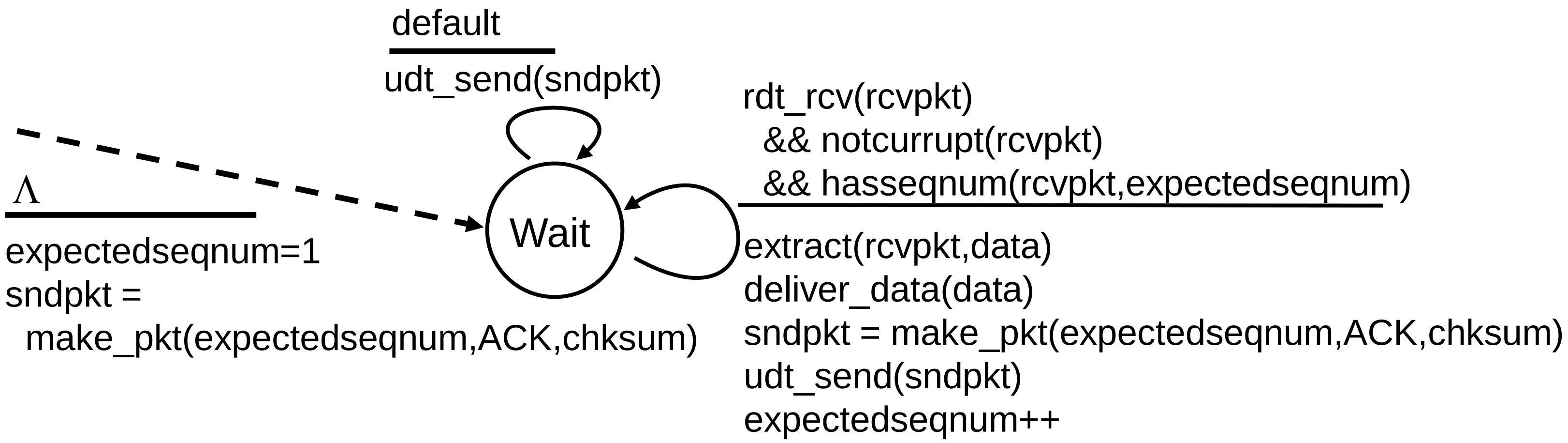
Selective Repeat:

- sender can have up to N unacked packets in pipeline
- receiver sends *individual ack* for each packet
- sender maintains timer for each unacked packet
  - when timer expires, retransmit only that unacked packet

GBN: sender extended FSM



GBN: receiver extended FSM

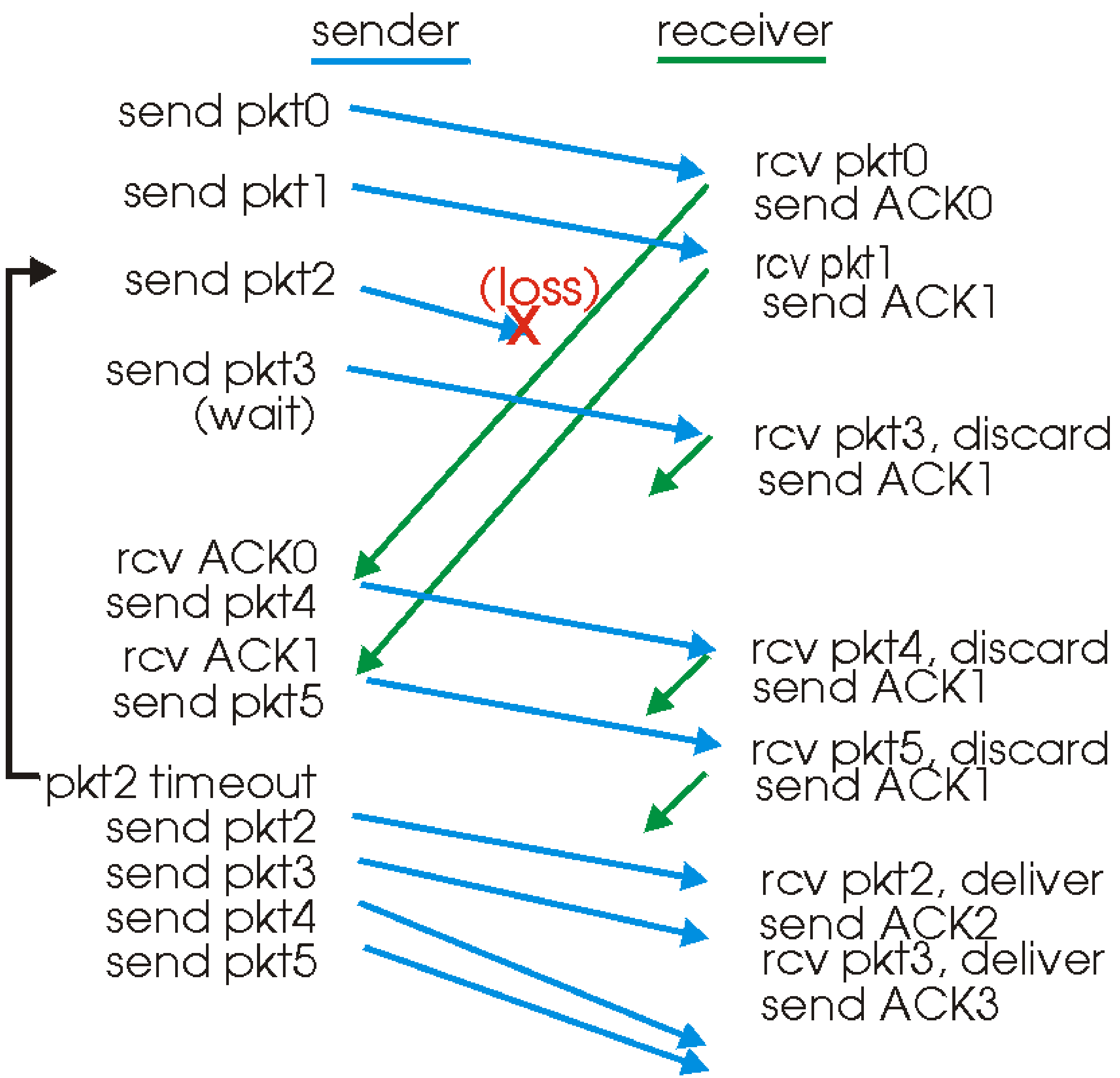


ACK-only: always send ACK for correctly-received pkt with highest *in-order* seq #

- may generate duplicate ACKs
- need only remember **`expectedseqnum`**

- out-of-order pkt:
  - discard (don't buffer) -> no receiver buffering!
  - Re-ACK pkt with highest in-order seq #

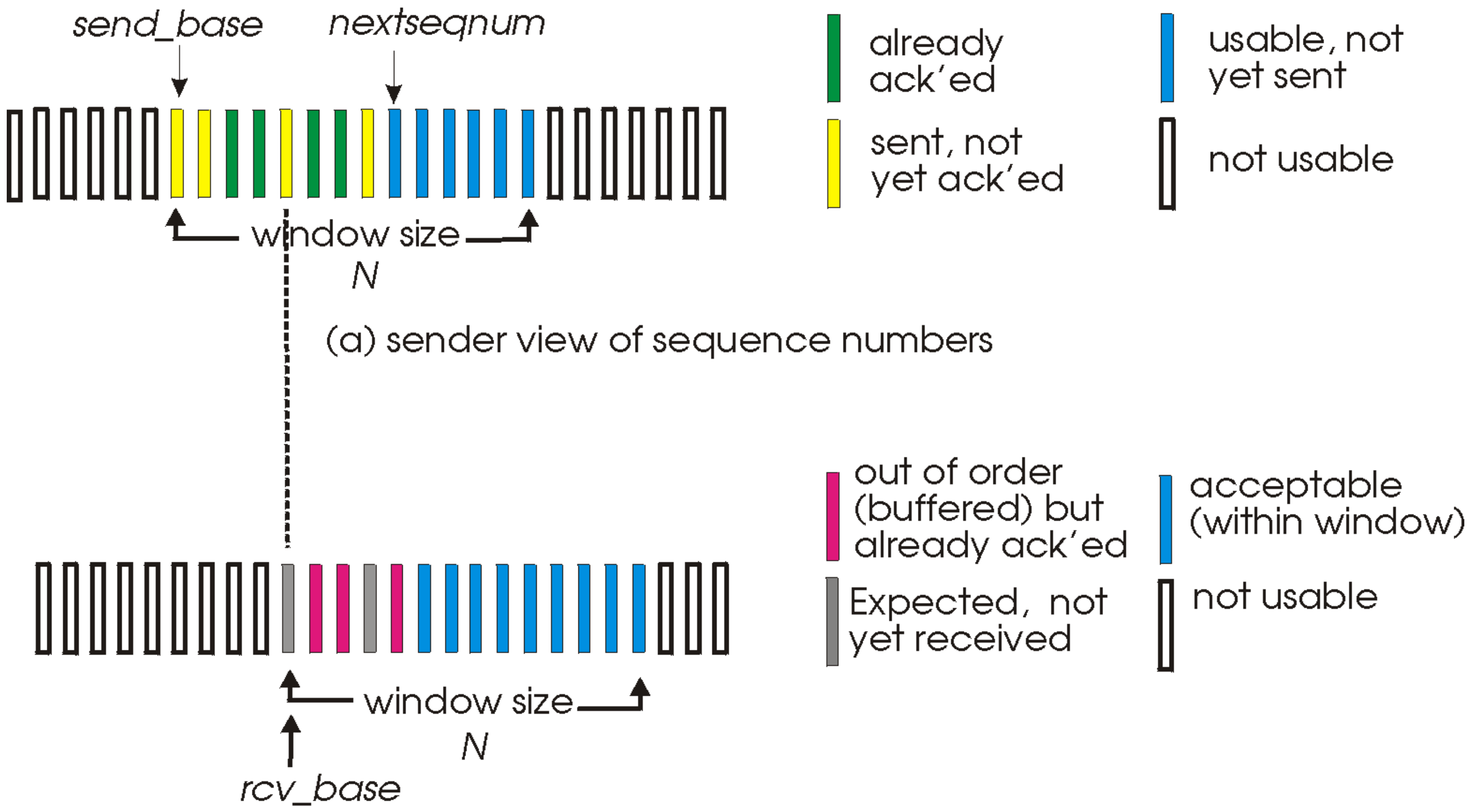
GBN in Action



Selective Repeat

- receiver *individually* acknowledges all correctly received pkts
  - buffers pkts, as needed, for eventual in-order delivery to upper layer
- sender only resends pkts for which ACK not received
  - sender timer for each unACKed pkt
- sender window
  - *N* consecutive seq #'s
  - again limits seq #'s of sent, unACKed pkts

Selective repeat: sender, receiver windows



(a) sender view of sequence numbers

(b) receiver view of sequence numbers

Selective Repeat

sender

data from above :

- if next available seq # in window, send pkt

timeout(n):

- resend pkt n, restart timer

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

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

receiver

pkt n in [rcvbase, rcvbase+N-1]

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

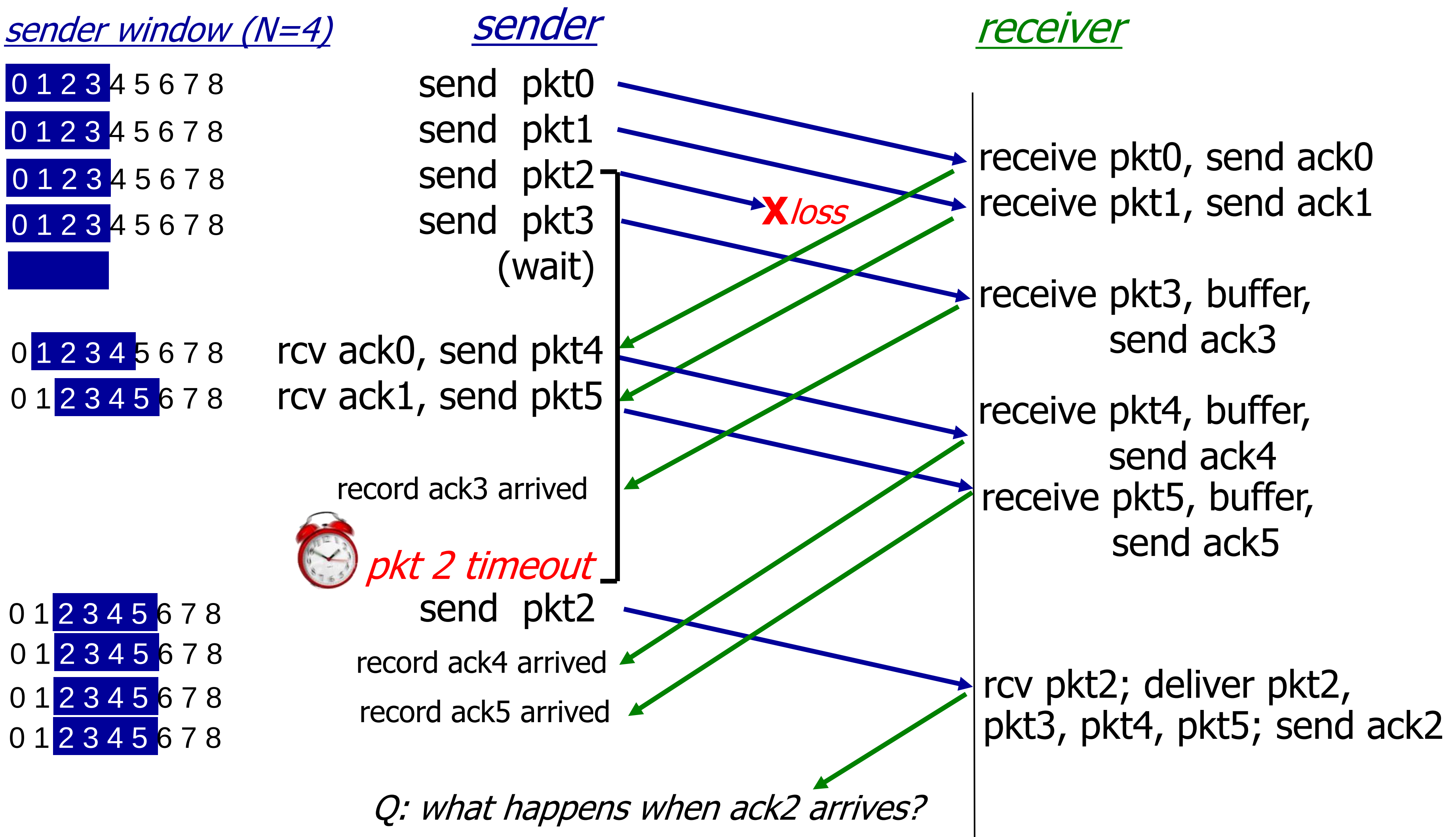
pkt n in [rcvbase-N,rcvbase-1]

- ACK(n)

otherwise:

- ignore

Selective Repeat in Action



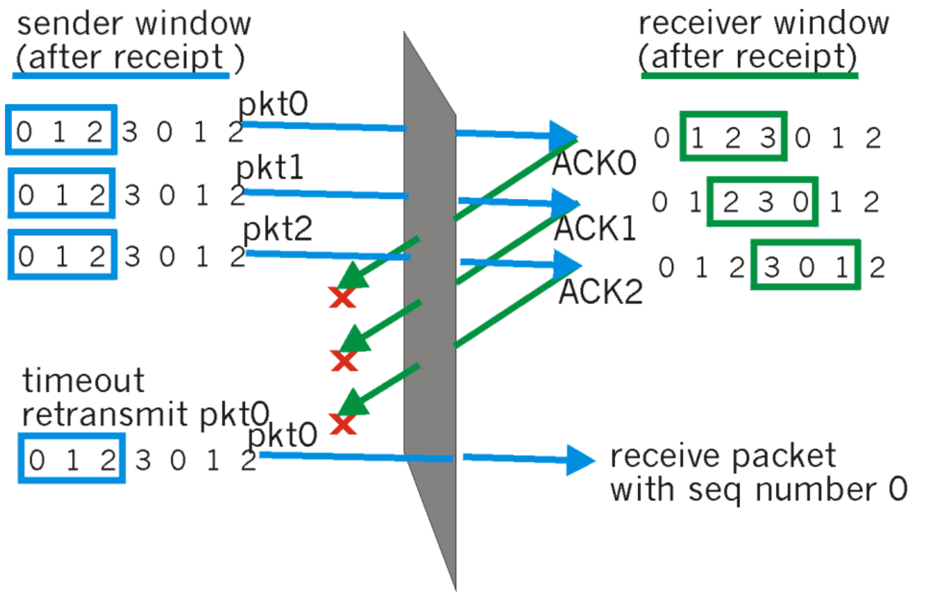
Selective Repeat: Dilemma

Example:

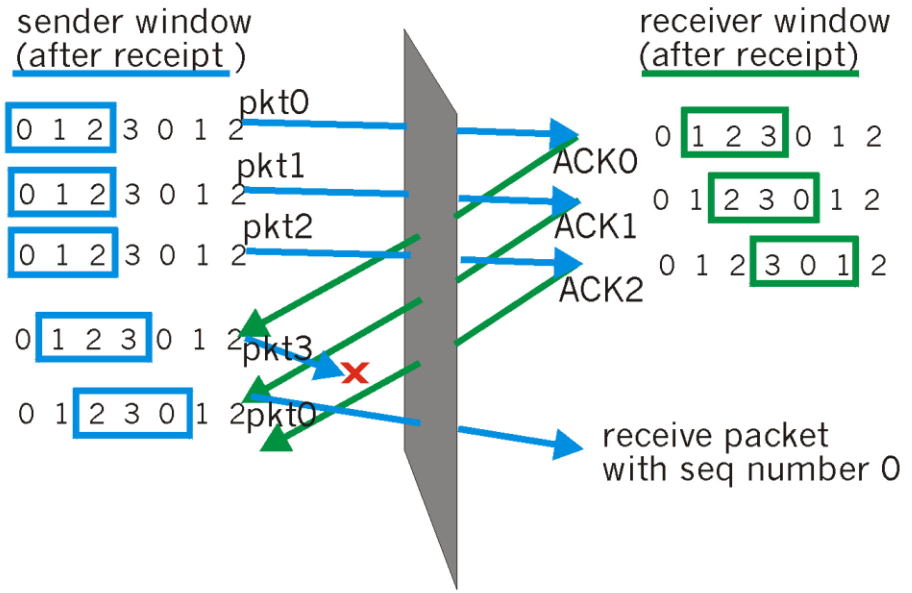
- seq #'s: 0, 1, 2, 3
- window size=3

- receiver sees no difference in two scenarios!
- incorrectly passes duplicate data as new in (a)

Q: what relationship between seq # size and window size?



(a)



(b)

To Be Continued

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Chapter 3: Roadmap

- Connection-oriented transport: TCP
- Principles of congestion control
- TCP congestion control
- Evolution of transport-layer functionality