



College of Information Studies

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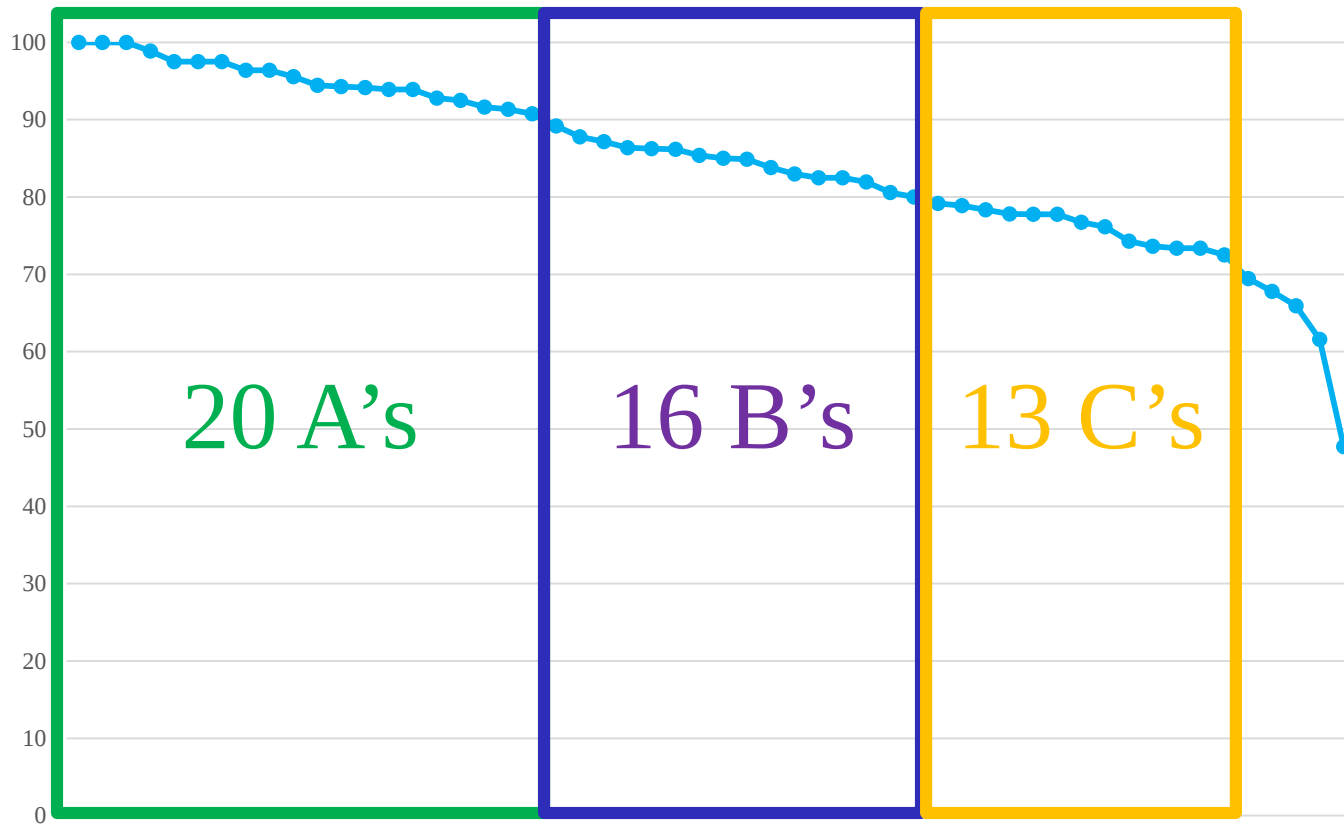
UDP

Session 21

INST 346

Technologies, Infrastructure and Architecture

Midterm Grades

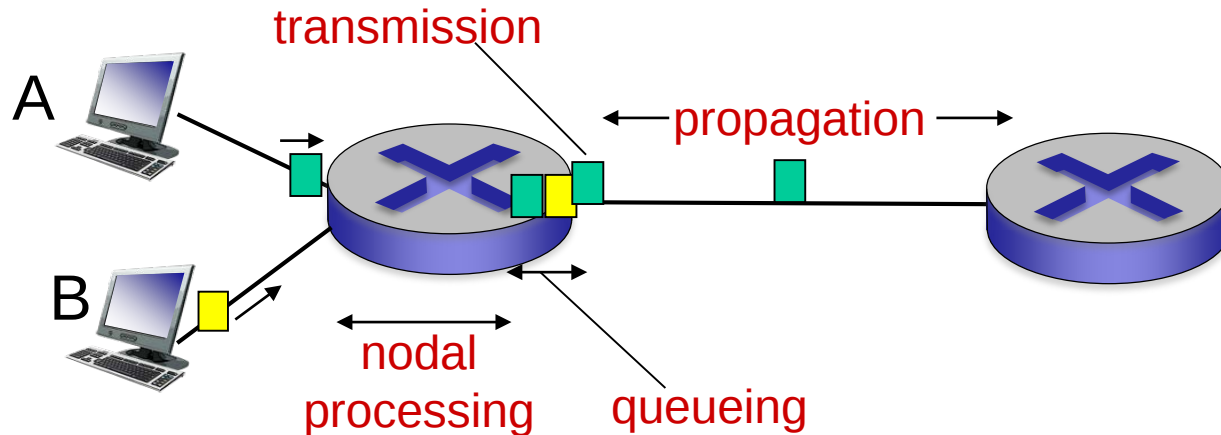


Based on best 4 of 5 homeworks, best 4.5 of 5 quizzes, best 1 of 1 exams
Assumes full remaining analysis team presentation and comment credit

Goals for Today

- Delays (finishing up the Network Layer)
- Transport layer
- UDP
- Getahead: TCP

Four sources of packet delay



$$d_{\text{nodal}} = d_{\text{proc}} + d_{\text{queue}} + d_{\text{trans}} + d_{\text{prop}}$$

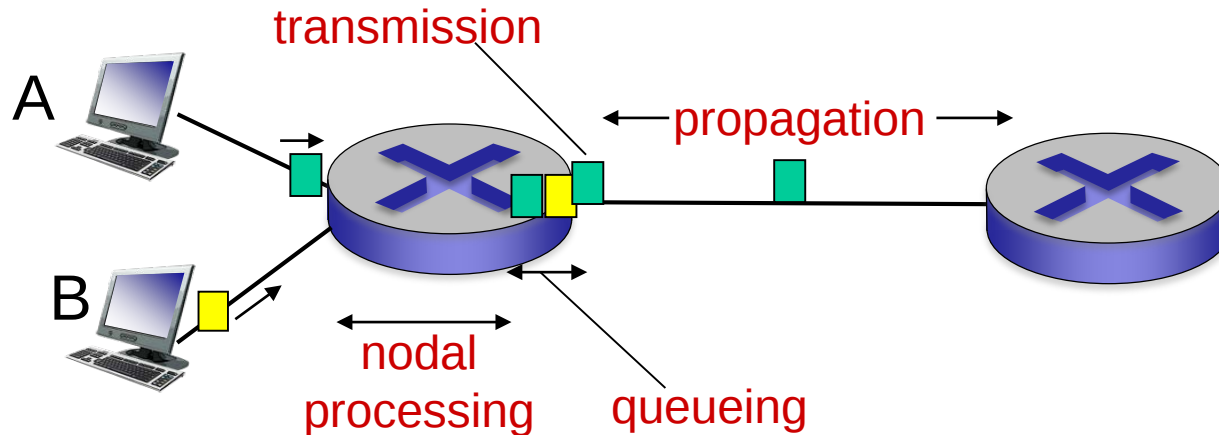
d_{proc} : nodal processing

- check bit errors
- determine output link
- typically < msec

d_{queue} : queueing delay

- time waiting at output link for transmission
- depends on congestion level of router

Four sources of packet delay



$$d_{\text{nodal}} = d_{\text{proc}} + d_{\text{queue}} + d_{\text{trans}} + d_{\text{prop}}$$

d_{trans} : transmission delay:

- L : packet length (bits)
- R : link bandwidth (bps)

▪ $d_{\text{trans}} = L/R$ ← d_{trans} and d_{prop} →
very different

d_{prop} : propagation delay:

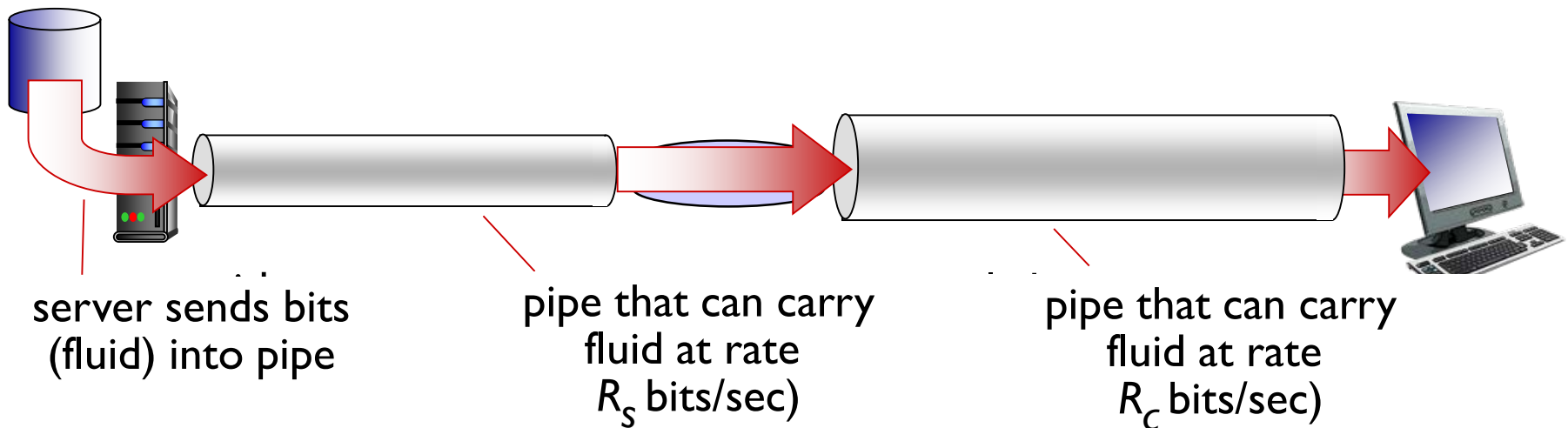
- d : length of physical link
 - s : propagation speed ($\sim 2 \times 10^8$ m/sec)
- $d_{\text{prop}} = d/s$

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

* Check out the Java applet for an interactive animation on trans vs. prop delay

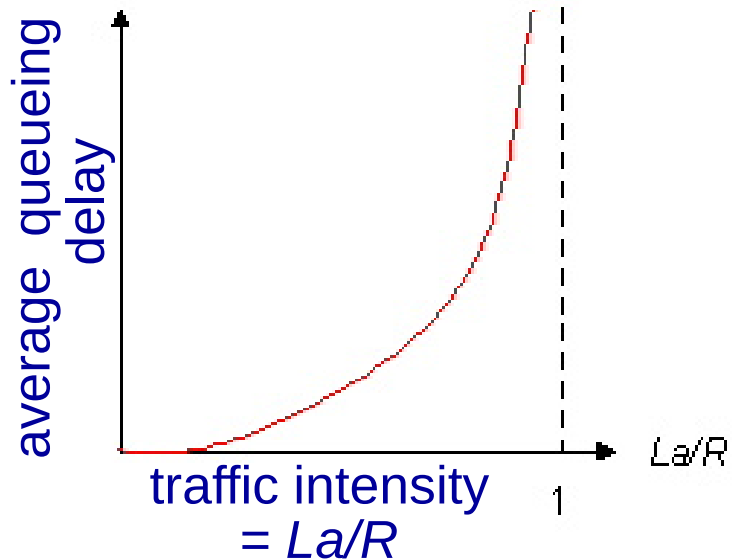
Throughput

- *throughput*: rate (bits/time unit) at which bits transferred between sender/receiver
 - *instantaneous*: rate at given point in time
 - *average*: rate over longer period of time



Queueing delay

- R : link bandwidth (bps)
- L : packet length (bits)
- a : average packet arrival rate



- $La/R \sim 0$: avg. queueing delay small
- $La/R \rightarrow 1$: avg. queueing delay large
- $La/R > 1$: more “work” arriving than can be serviced, average delay infinite!



$La/R \sim 0$

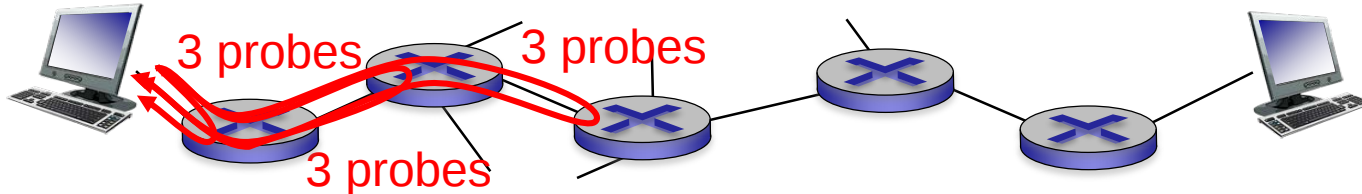


$La/R \rightarrow 1$

* Check online interactive animation on queueing and loss

“Real” Internet delays and routes

- what do “real” Internet delay & loss look like?
- **traceroute** program: provides delay measurement from source to router along end-end Internet path towards destination. For all i :
 - sends three packets that will reach router i on path towards destination
 - router i will return packets to sender
 - sender times interval between transmission and reply.



“Real” Internet delays, routes

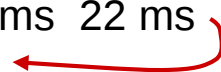
traceroute: gaia.cs.umass.edu to www.eurecom.fr

3 delay measurements from
gaia.cs.umass.edu to cs-gw.cs.umass.edu



1 cs-gw (128.119.240.254) 1 ms 1 ms 2 ms
2 border1-rt-fa5-1-0.gw.umass.edu (128.119.3.145) 1 ms 1 ms 2 ms
3 cht-vbns.gw.umass.edu (128.119.3.130) 6 ms 5 ms 5 ms
4 jn1-at1-0-0-19.wor.vbns.net (204.147.132.129) 16 ms 11 ms 13 ms
5 jn1-so7-0-0-0.wae.vbns.net (204.147.136.136) 21 ms 18 ms 18 ms
6 abilene-vbns.abilene.ucaid.edu (198.32.11.9) 22 ms 18 ms 22 ms
7 nycm-wash.abilene.ucaid.edu (198.32.8.46) 22 ms 22 ms 22 ms
8 62.40.103.253 (62.40.103.253) 104 ms 109 ms 106 ms
9 de2-1.de1.de.geant.net (62.40.96.129) 109 ms 102 ms 104 ms
10 de.fr1.fr.geant.net (62.40.96.50) 113 ms 121 ms 114 ms
11 renater-gw.fr1.fr.geant.net (62.40.103.54) 112 ms 114 ms 112 ms
12 nio-n2.cssi.renater.fr (193.51.206.13) 111 ms 114 ms 116 ms
13 nice.cssi.renater.fr (195.220.98.102) 123 ms 125 ms 124 ms
14 r3t2-nice.cssi.renater.fr (195.220.98.110) 126 ms 126 ms 124 ms
15 eurecom-valbonne.r3t2.ft.net (193.48.50.54) 135 ms 128 ms 133 ms
16 194.214.211.25 (194.214.211.25) 126 ms 128 ms 126 ms
17 * * *
18 * * *
19 fantasia.eurecom.fr (193.55.113.142) 132 ms 128 ms 136 ms

trans-oceanic link



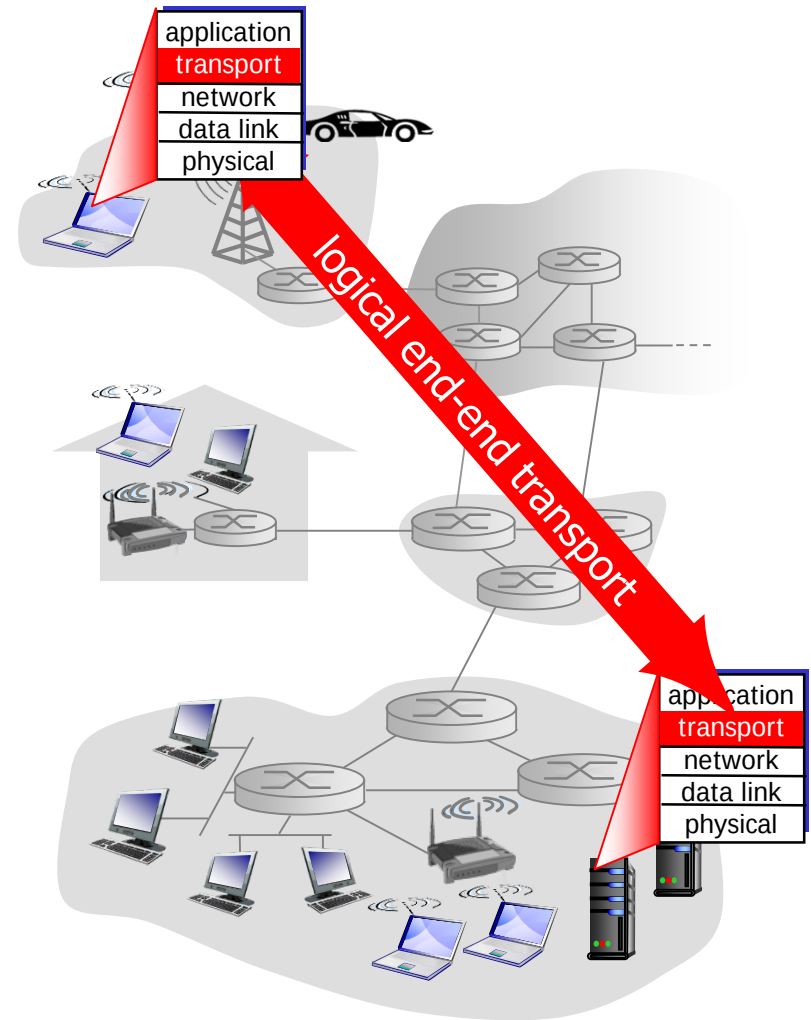
* means no response (probe lost, router not replying)



* Do some traceroutes from exotic countries at www.traceroute.org

Transport services and protocols

- provide *logical communication* between app processes running on different hosts
- transport protocols run in end systems
 - send side: breaks app messages into *segments*, passes to network layer
 - rcv side: reassembles segments into messages, passes to app layer
- more than one transport protocol available to apps
 - Internet: TCP and UDP



Transport vs. network layer

- *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 protocol = Ann and Bill who demux to in-house siblings
- network-layer protocol = postal service

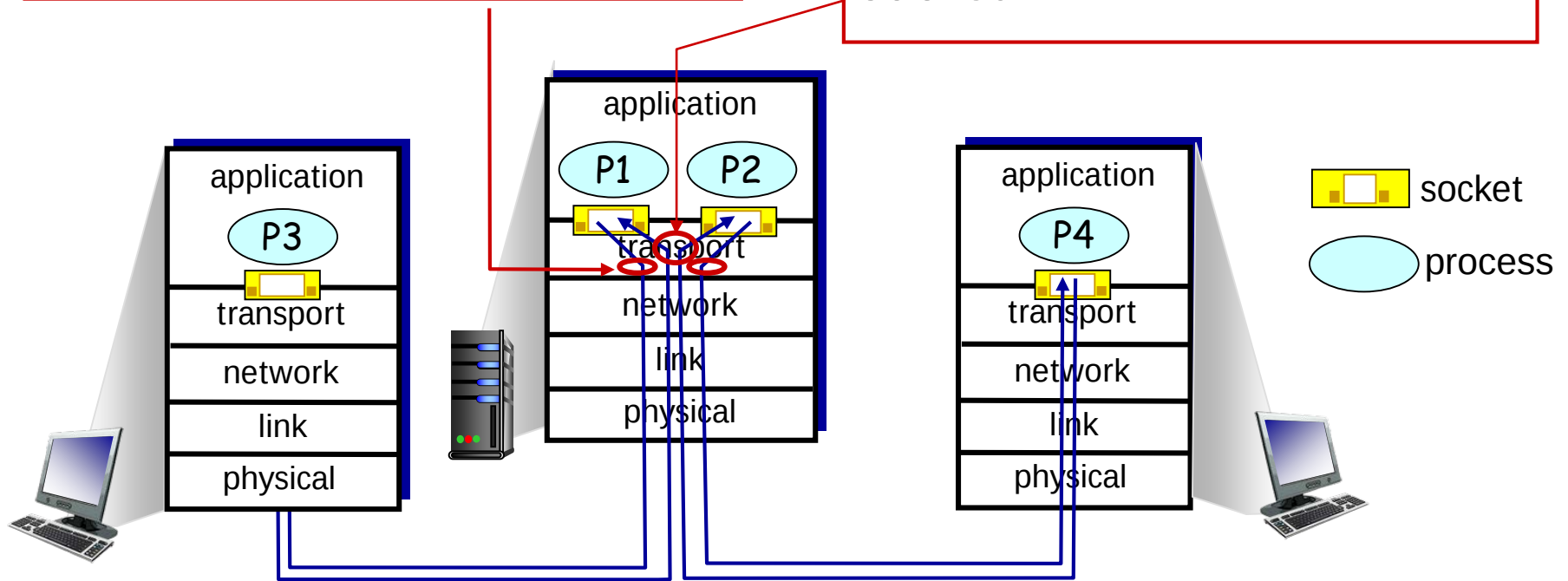
Multiplexing/demultiplexing

multiplexing at sender:

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

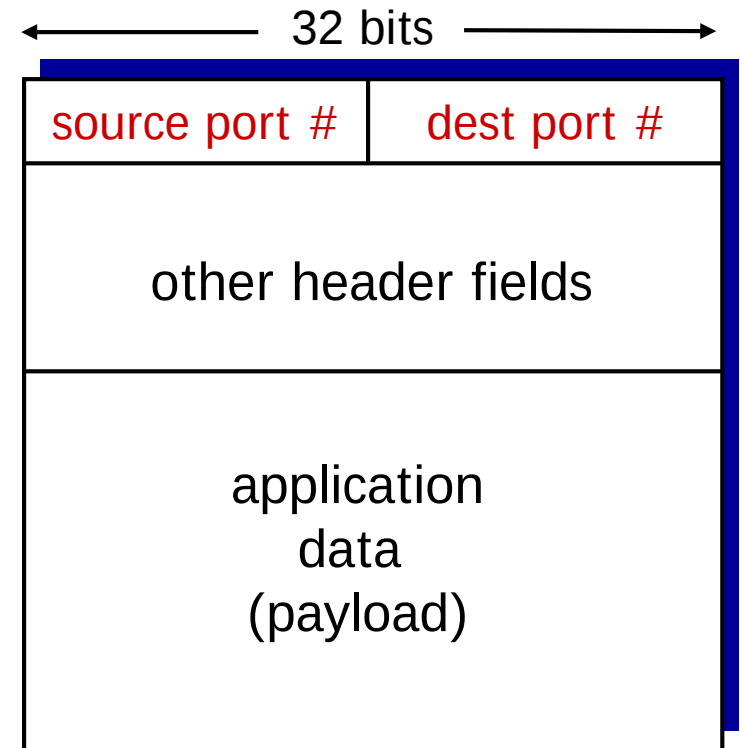
demultiplexing at receiver:

use header info to deliver received segments to correct socket



How demultiplexing works

- host receives IP datagrams
 - each datagram has source IP address, destination IP address
 - each datagram carries one transport-layer segment
 - each segment has source, destination port number
- host uses *IP addresses & port numbers* to direct segment to appropriate socket



TCP/UDP segment format

Connectionless demultiplexing

- created socket has host-local port #:

```
DatagramSocket mySocket1  
= new DatagramSocket(12534);
```

when creating datagram to send into UDP socket, must specify

- destination IP address
- destination port #

- when host receives UDP segment:

- checks destination port # in segment
- directs UDP segment to socket with that port #



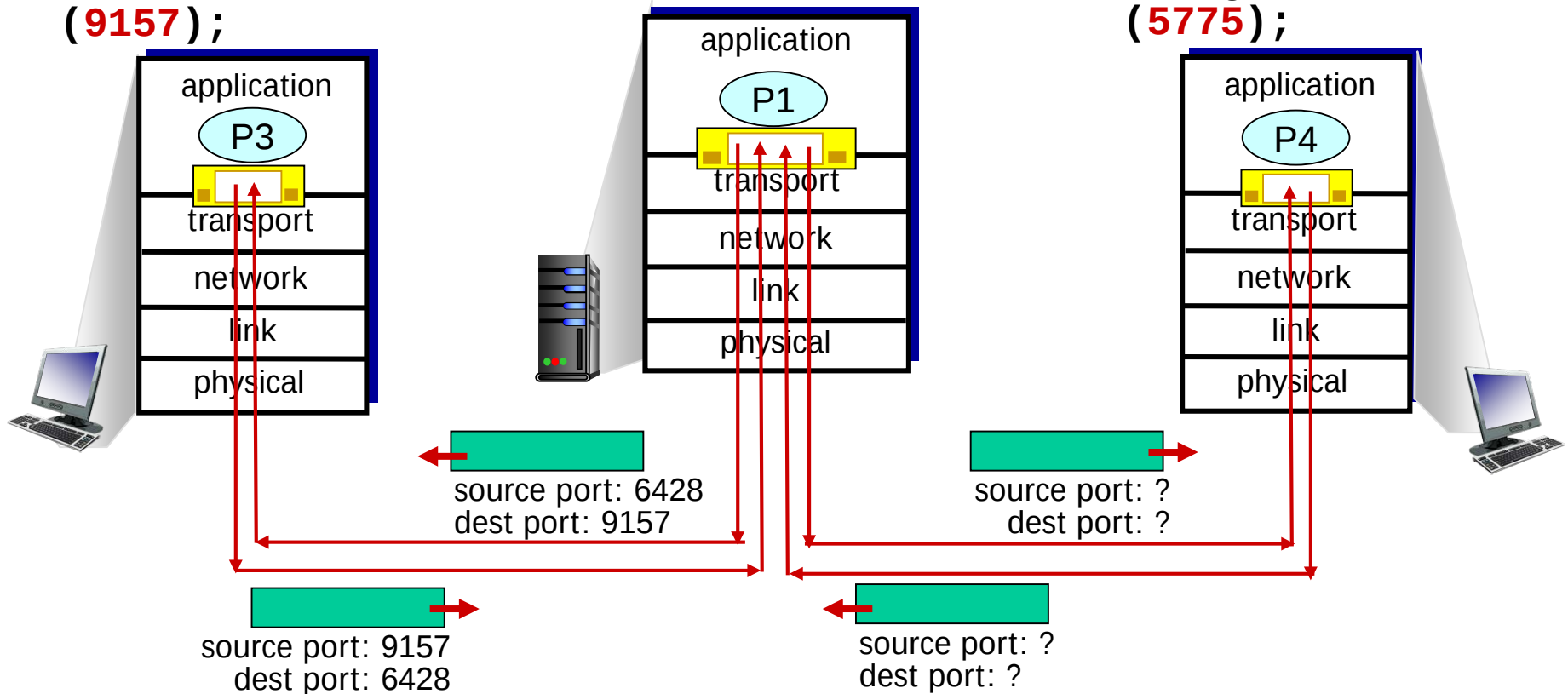
IP datagrams with *same destination port #*, but different source IP addresses and/or source port numbers will be directed to *same socket* at destination

Connectionless demux: example

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

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

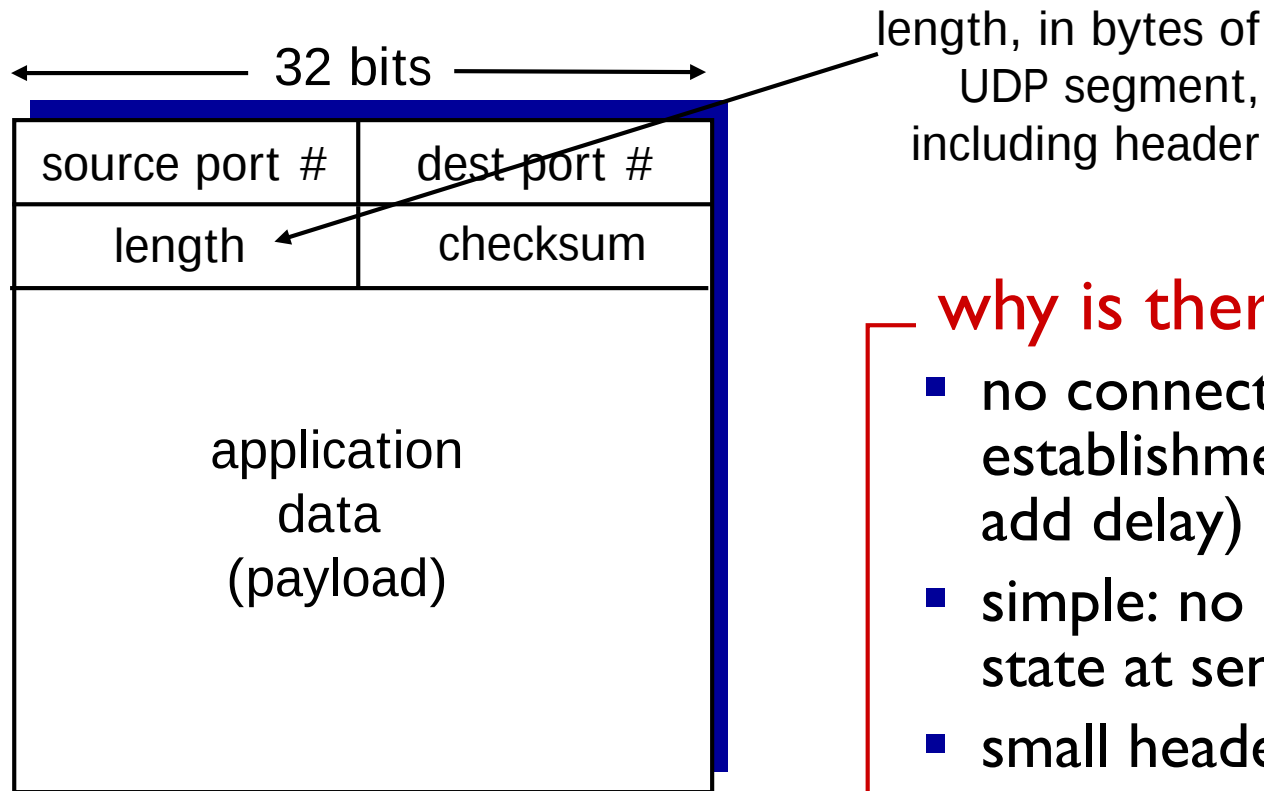
```
DatagramSocket  
mySocket1 = new  
DatagramSocket  
(5775);
```



UDP: User Datagram Protocol [RFC 768]

- “no frills” 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
- UDP use:
 - streaming multimedia apps (loss tolerant, rate sensitive)
 - DNS
 - VoIP
- reliable transfer over UDP:
 - add reliability at application layer
 - application-specific error recovery

UDP: segment header



UDP segment format

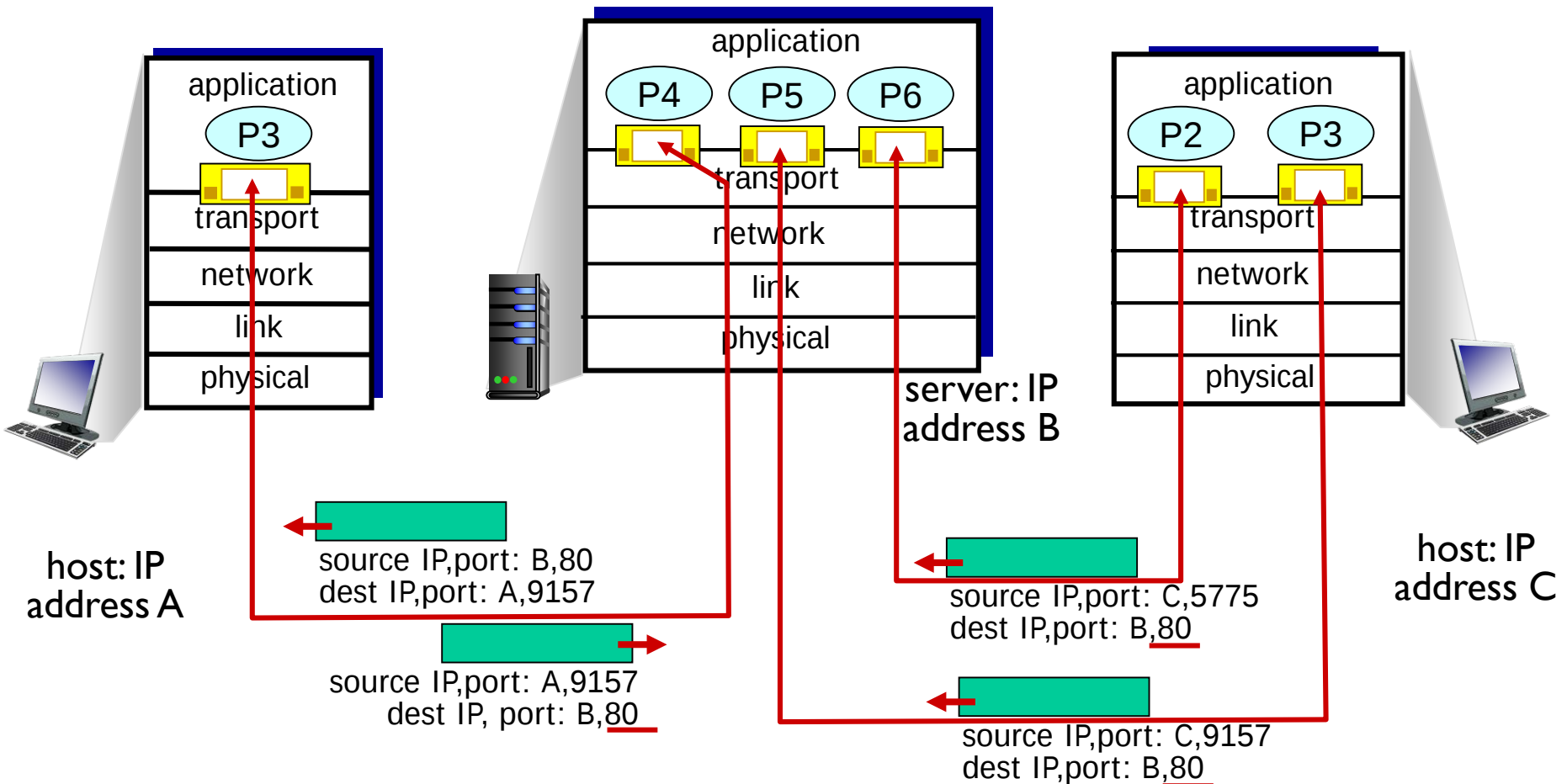
why is there a UDP?

- no connection establishment (which can add delay)
- simple: no connection state at sender, receiver
- small header size
- no congestion control: UDP can blast away as fast as desired

Connection-oriented demux

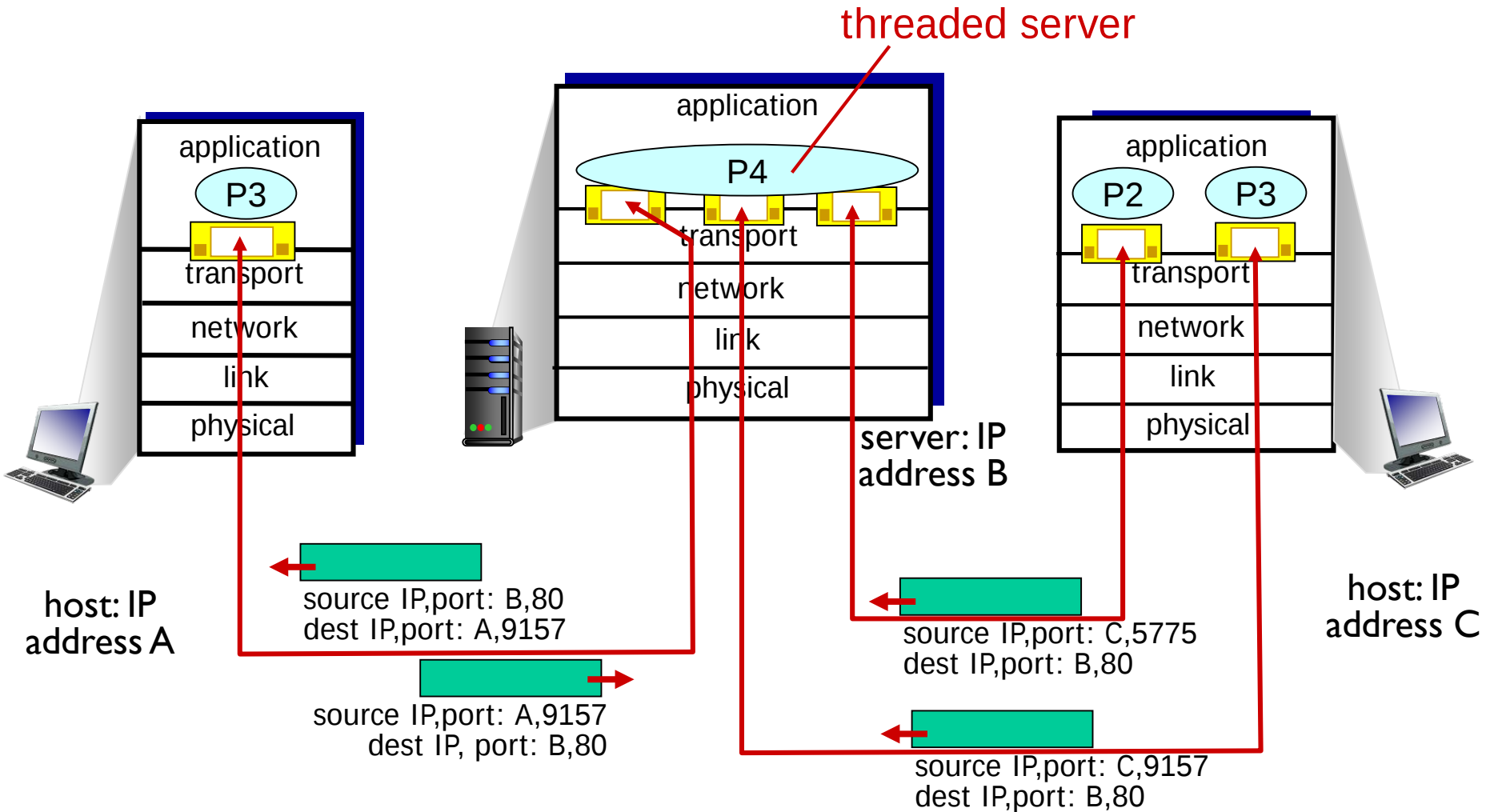
- TCP socket identified by 4-tuple:
 - source IP address
 - source port number
 - dest IP address
 - dest port number
- demux: receiver uses all four values to direct segment to appropriate socket
- server host may support many simultaneous TCP sockets:
 - each socket identified by its own 4-tuple
- web servers have different sockets for each connecting client
 - non-persistent HTTP will have different socket for each request

Connection-oriented demux: example



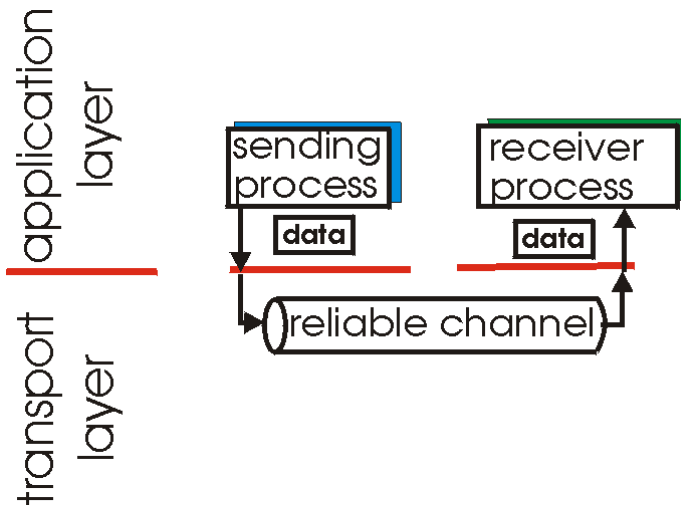
three segments, all destined to IP address: B,
dest port: 80 are demultiplexed to *different* sockets

Connection-oriented demux: example



Principles of reliable data transfer

- important in application, transport, link layers

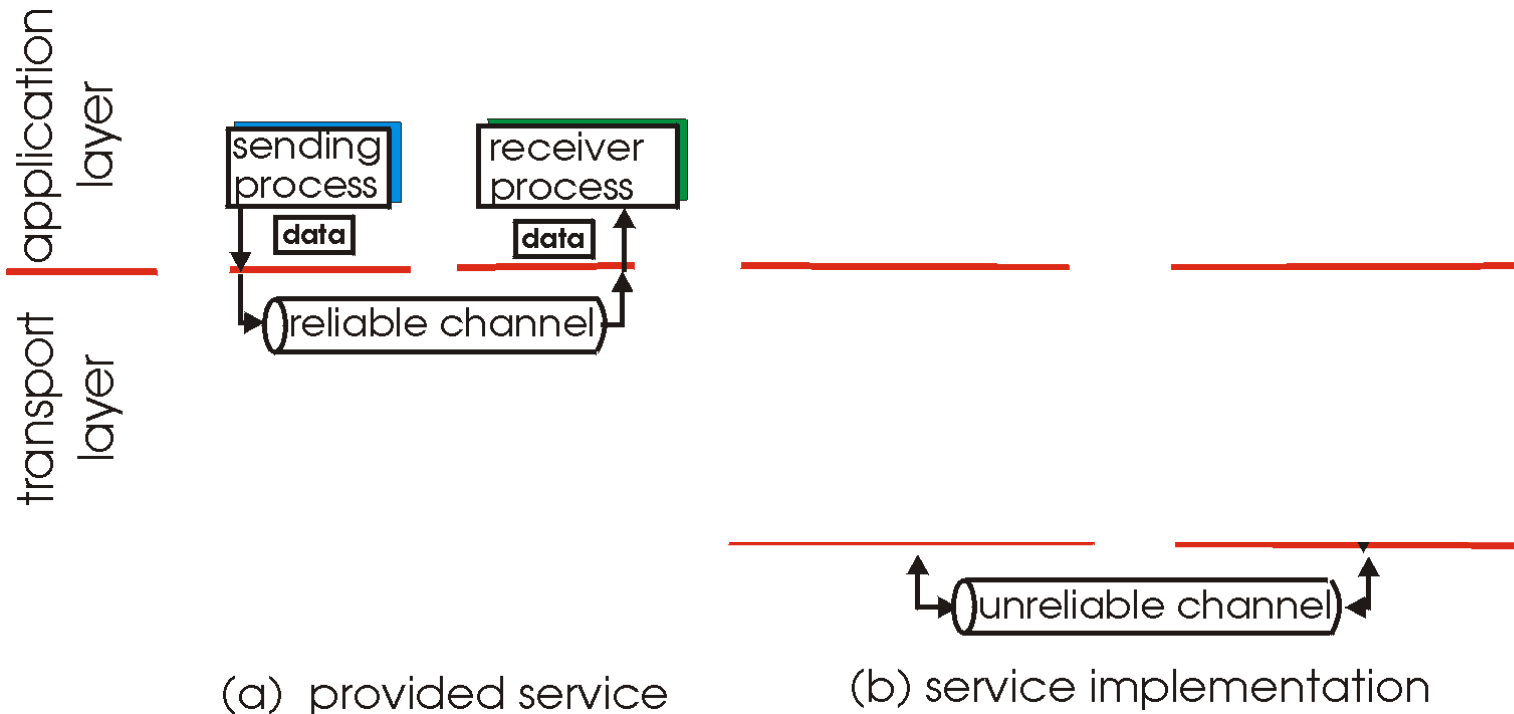


(a) provided service

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

Principles of reliable data transfer

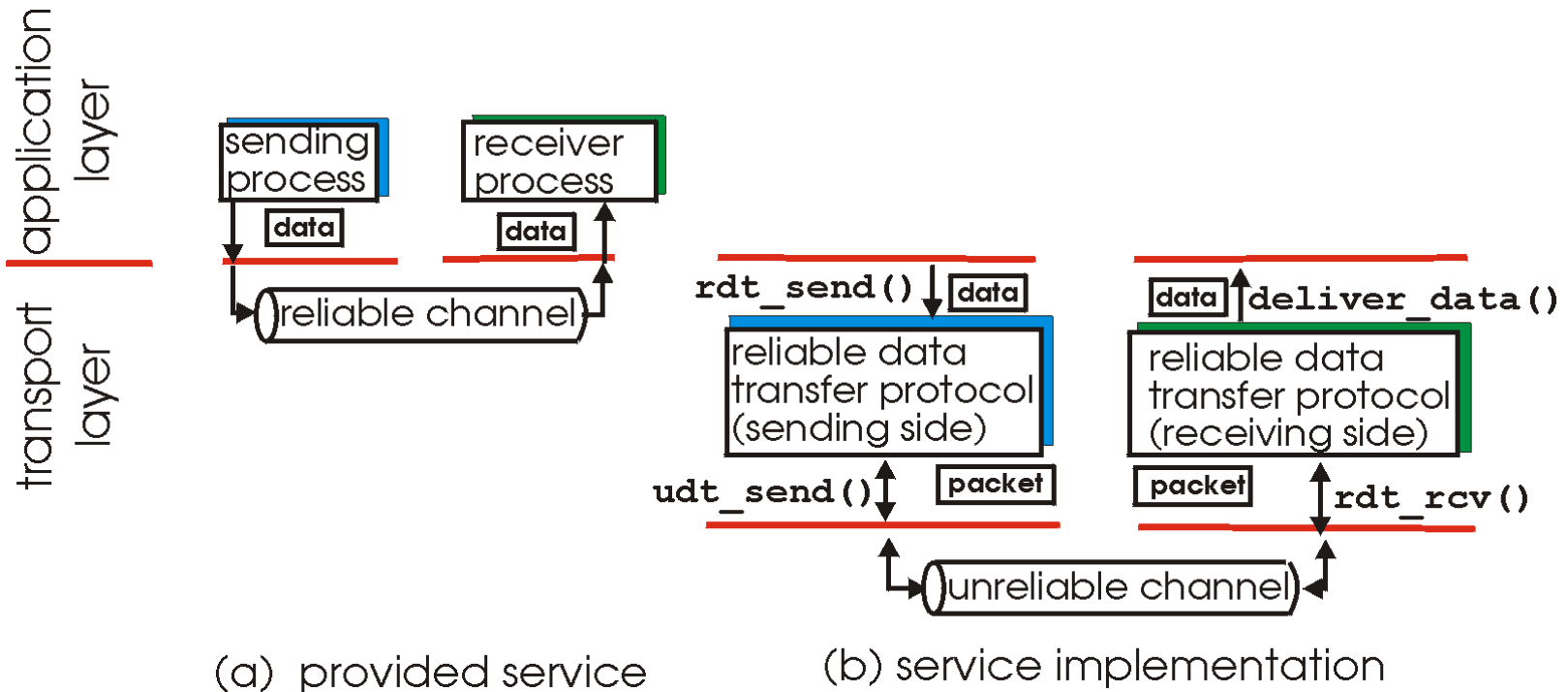
- important in application, transport, link layers
 - top-10 list of important networking topics!



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

Principles of reliable data transfer

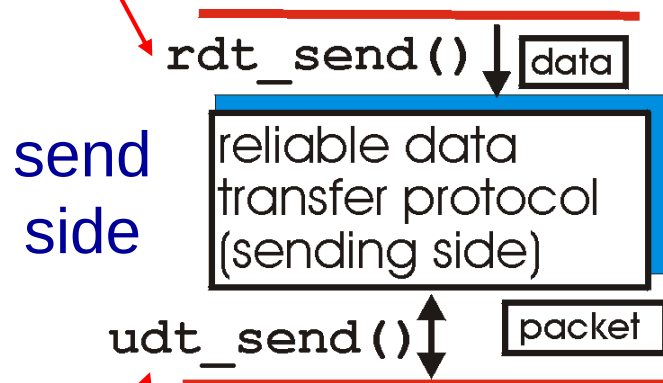
- important in application, transport, link layers
 - top-10 list of important networking topics!



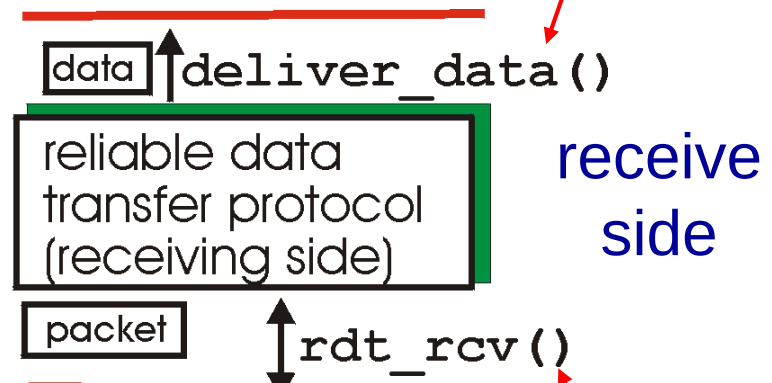
- characteristics of unreliable channel will determine complexity of reliable data transfer protocol (rdt)

Reliable data transfer: getting started

rdt_send(): called from above, (e.g., by app.). Passed data to deliver to receiver upper layer



deliver_data(): called by rdt to deliver data to upper



udt_send(): called by rdt, to transfer packet over unreliable channel to receiver

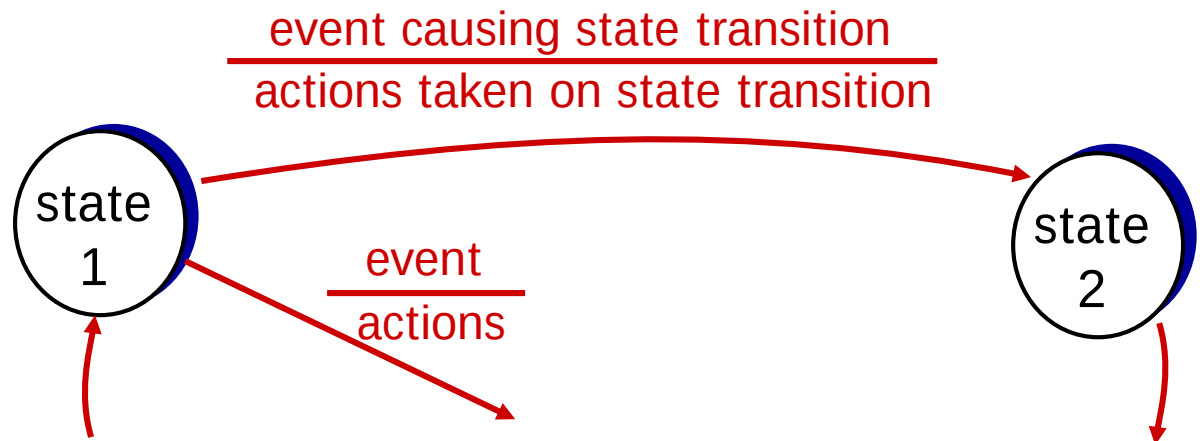
rdt_rcv(): called when packet arrives on rcv-side of channel

Reliable data transfer: getting started

we' ll:

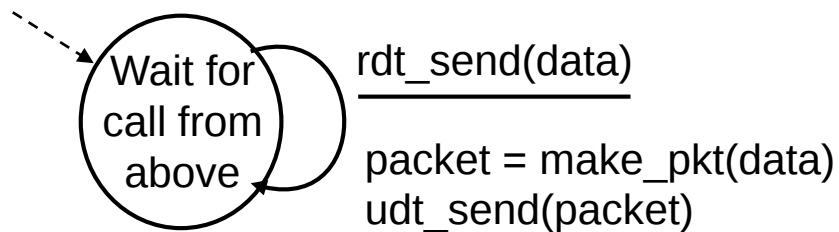
- incrementally develop sender, receiver sides of reliable data transfer protocol (rdt)
- consider only unidirectional data transfer
 - but control info will flow on both directions!
- use finite state machines (FSM) to specify sender, receiver

state: when in this “state” next state uniquely determined by next event

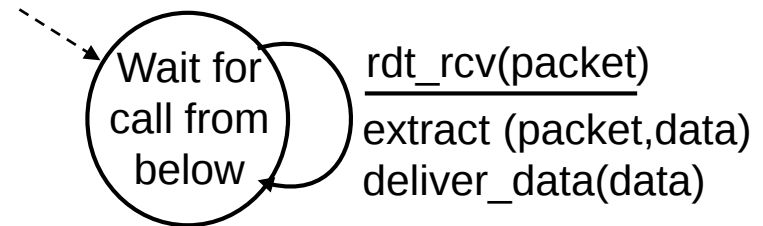


rdt1.0: reliable transfer over a reliable channel

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



sender



receiver

rdt2.0: channel with bit errors

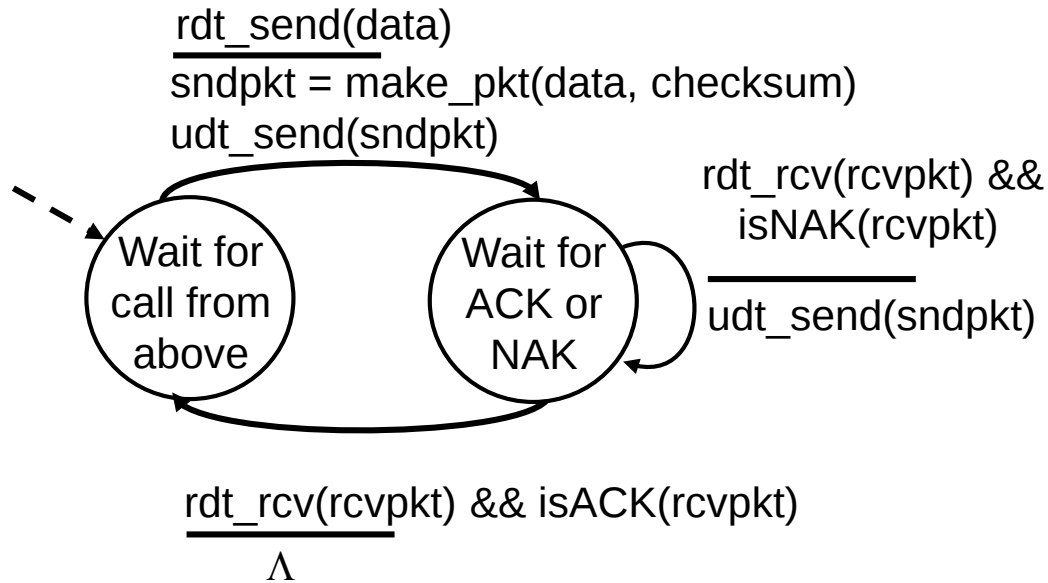
- underlying channel may flip bits in packet
 - checksum to detect bit errors
- *the question*: how to recover from errors:

*How do humans recover from “errors”
during conversation?*

rdt2.0: channel with bit errors

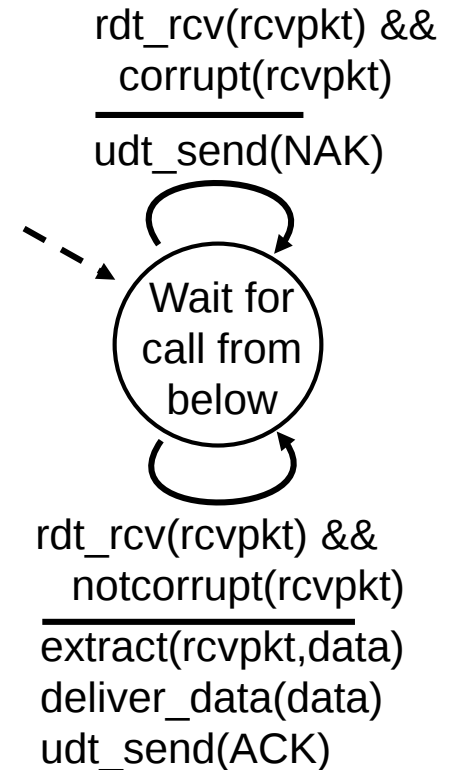
- underlying channel may flip bits in packet
 - checksum to detect bit errors
- *the question*: how to recover from errors:
 - *acknowledgements (ACKs)*: receiver explicitly tells sender that pkt received OK
 - *negative acknowledgements (NAKs)*: receiver explicitly tells sender that pkt had errors
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

rdt2.0: FSM specification

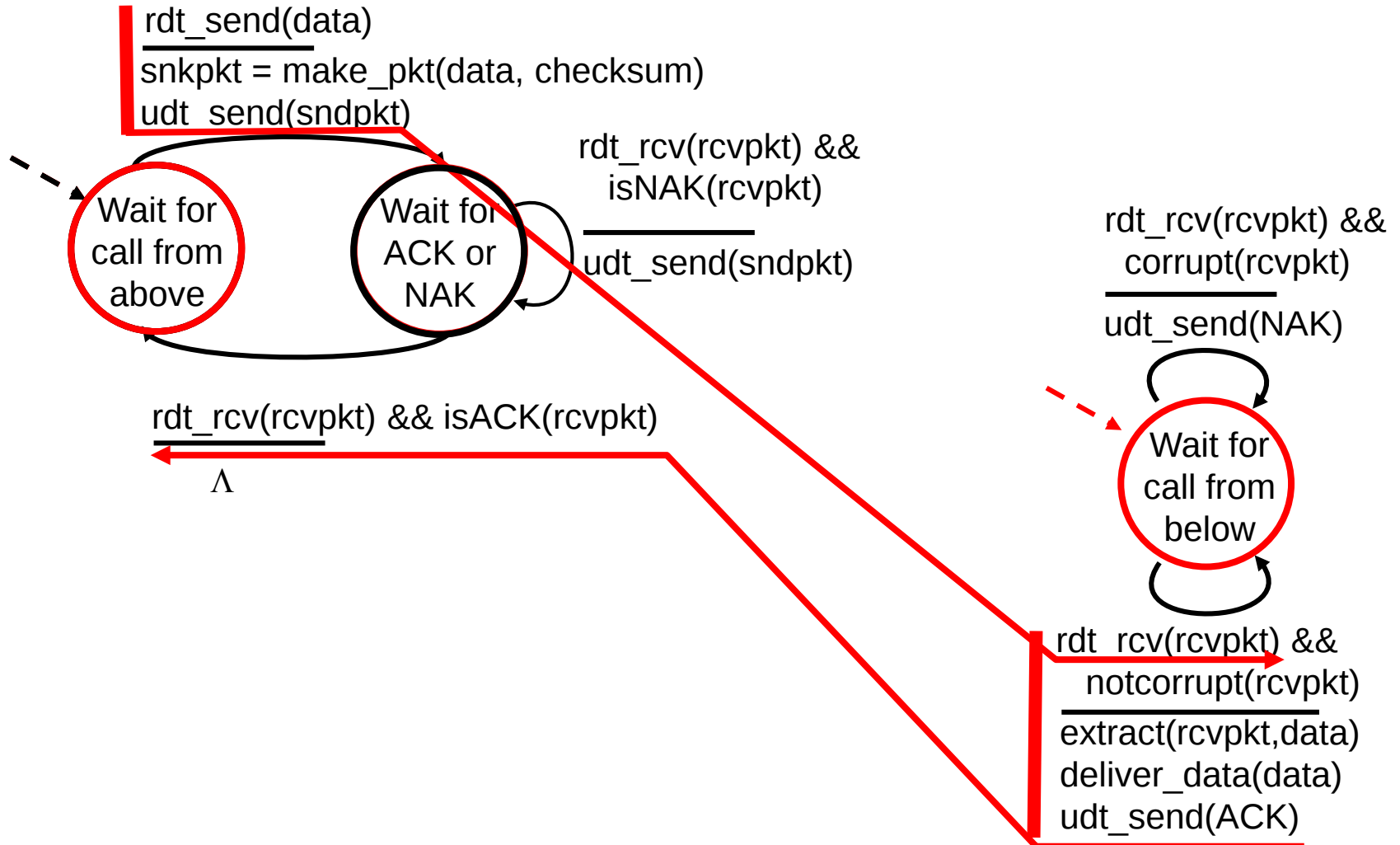


sender

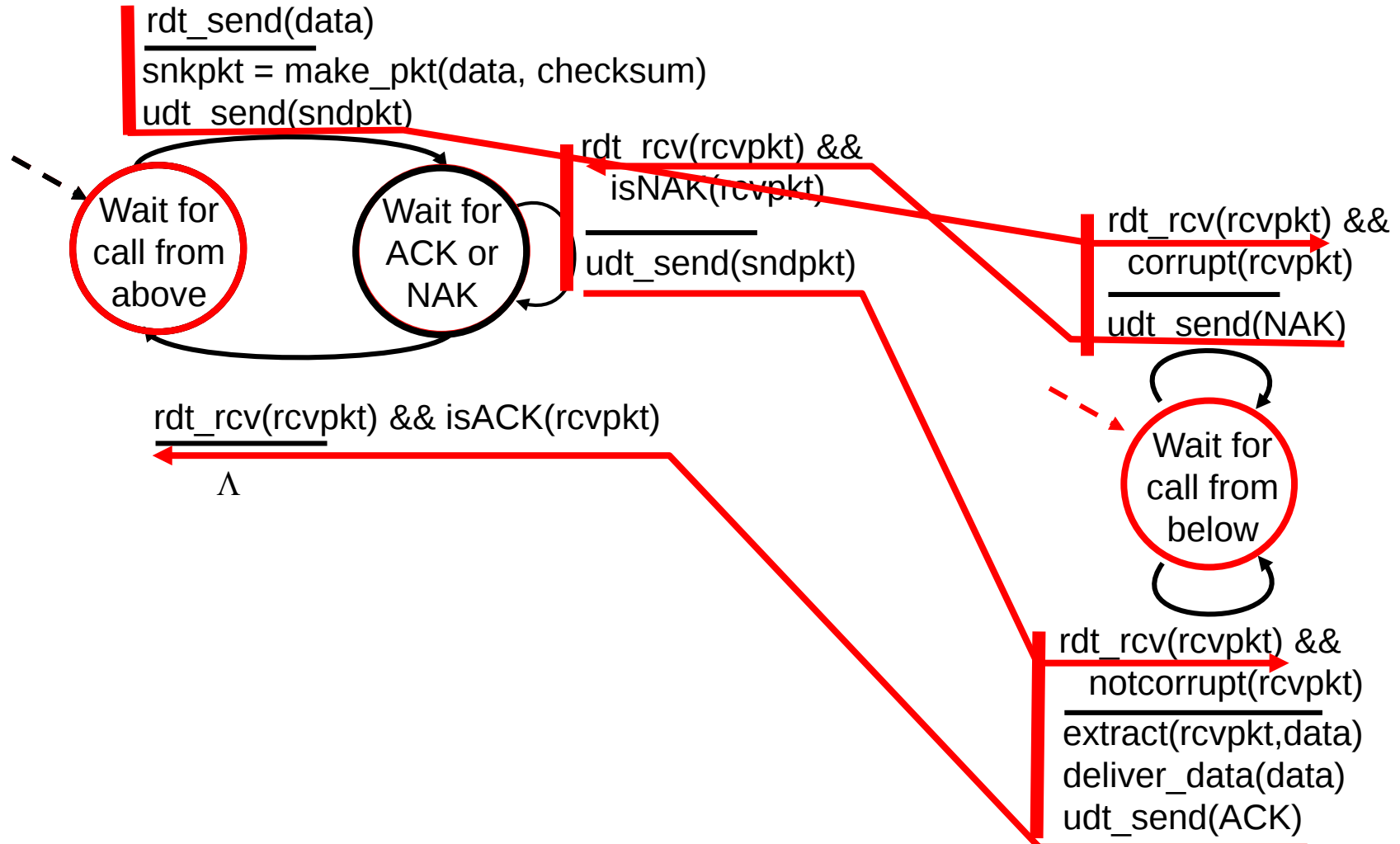
receiver



rdt2.0: operation with no errors



rdt2.0: error scenario



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

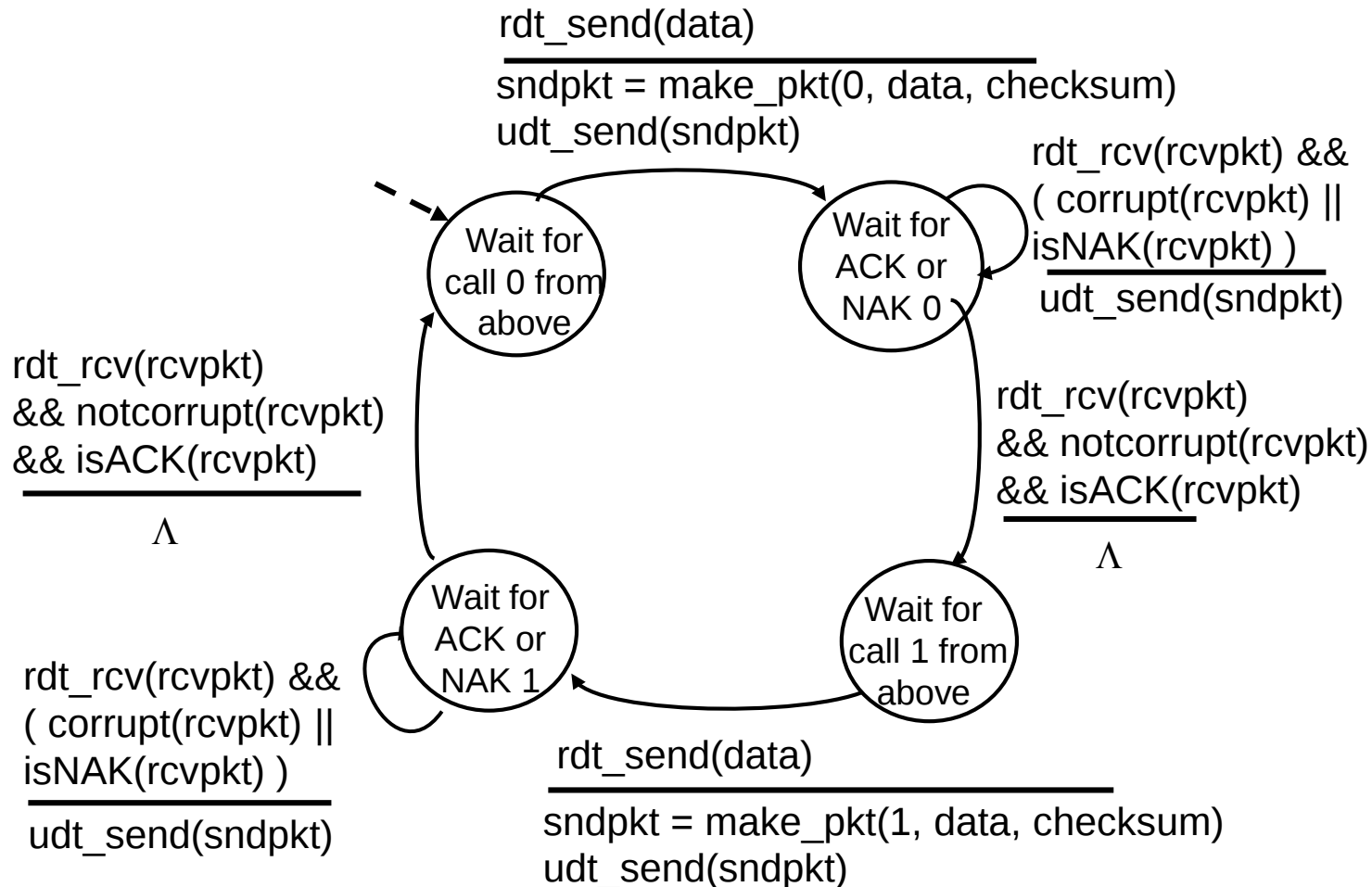
handling duplicates:

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

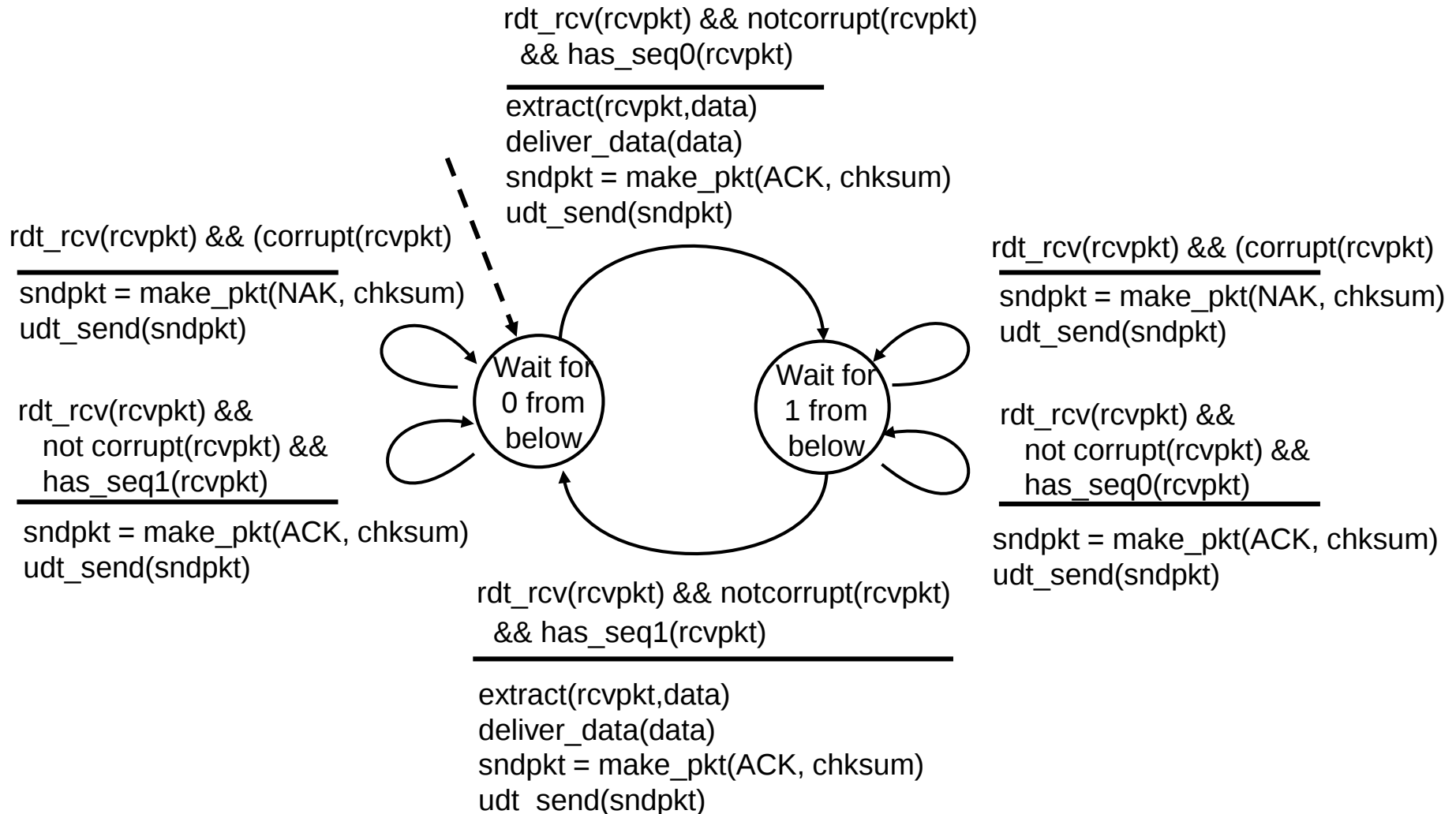
stop and wait

sender sends one packet,
then waits for receiver
response

rdt2.1: sender, handles garbled ACK/NAKs



rdt2.1: receiver, handles garbled ACK/NAKs



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

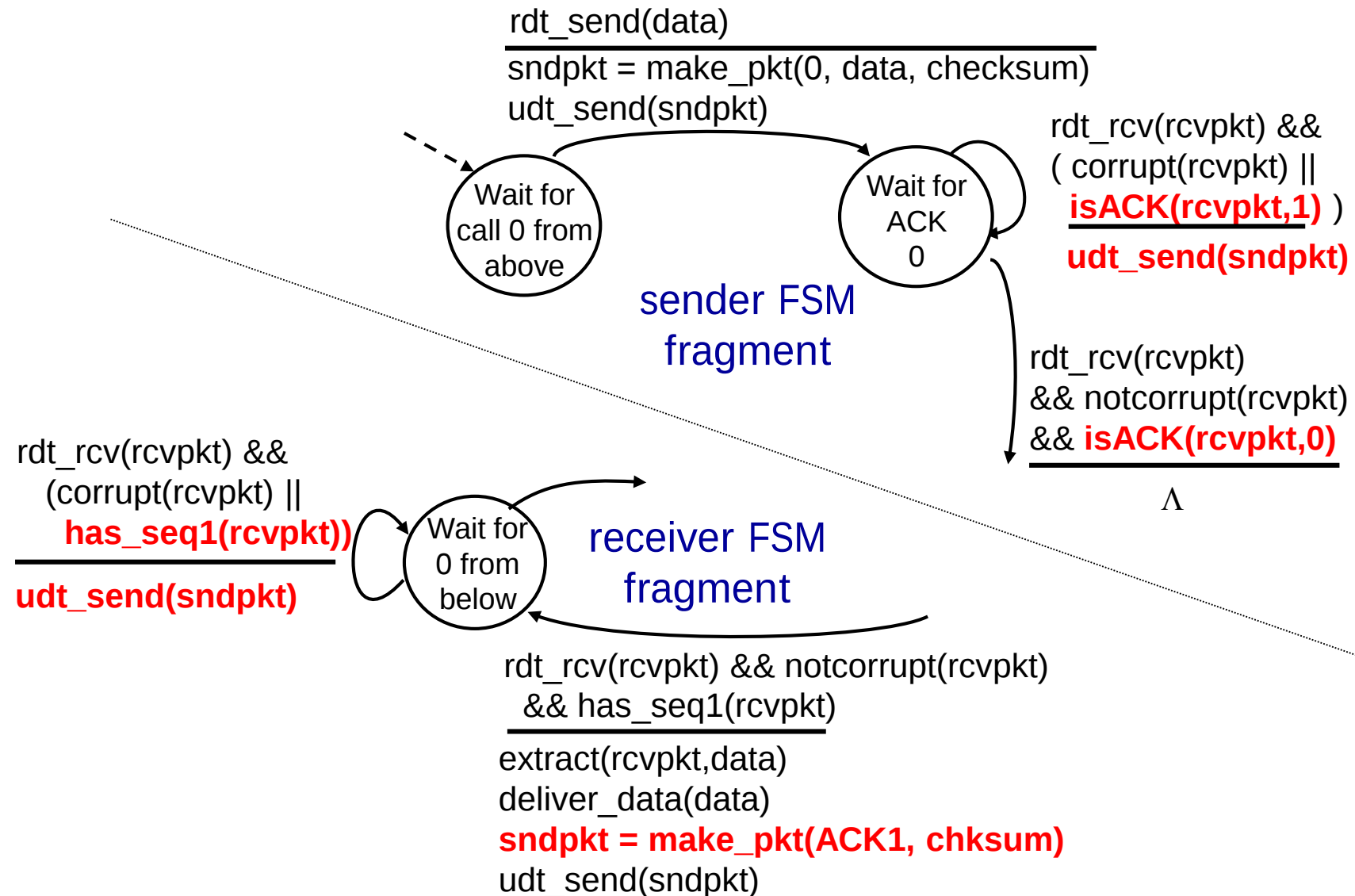
receiver:

- 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

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*

rdt2.2: sender, receiver fragments



rdt3.0: channels with errors *and* loss

new assumption:

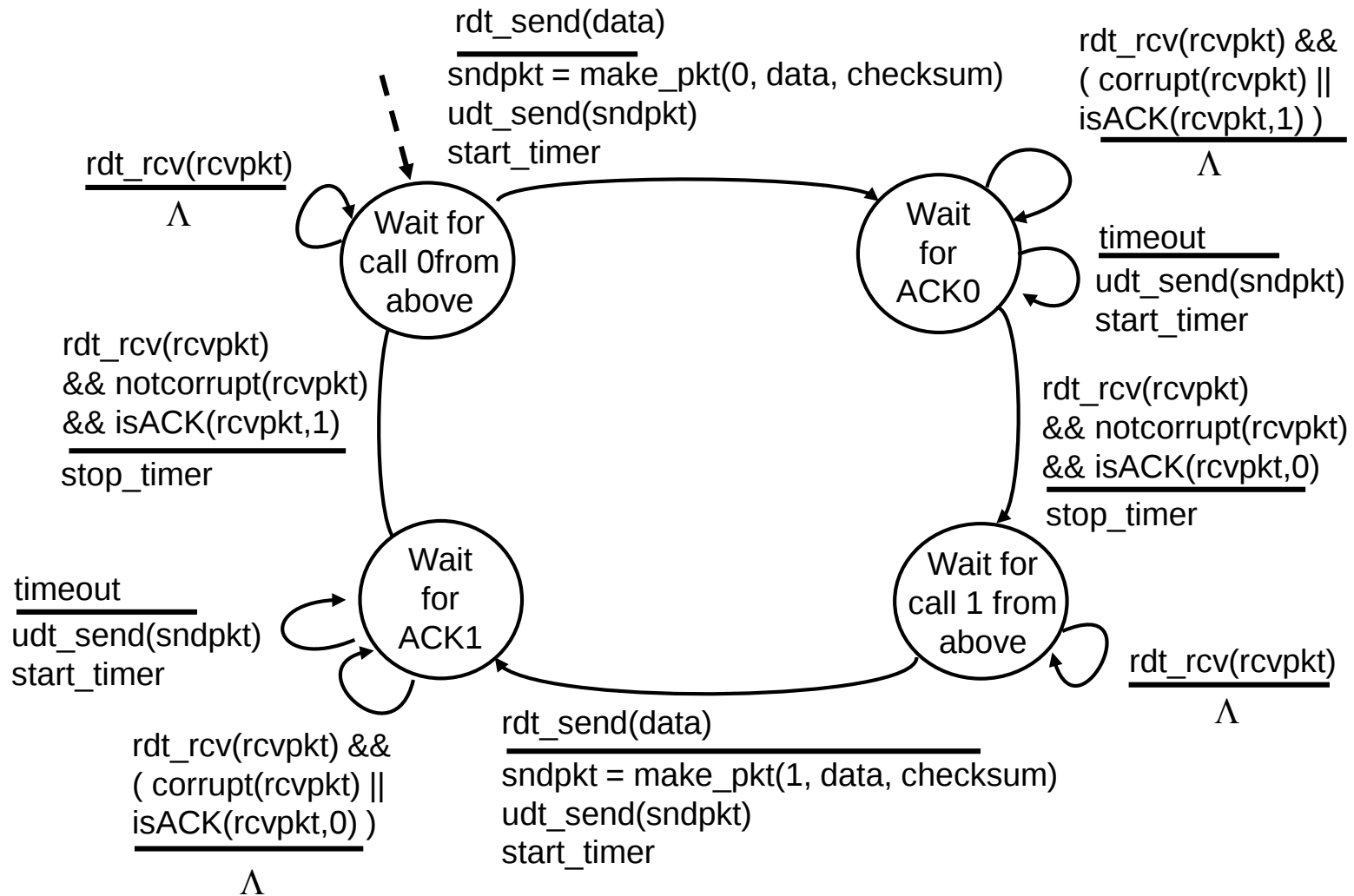
underlying channel can also lose packets (data, ACKs)

- checksum, seq. #, ACKs, retransmissions will be of help ... but not enough

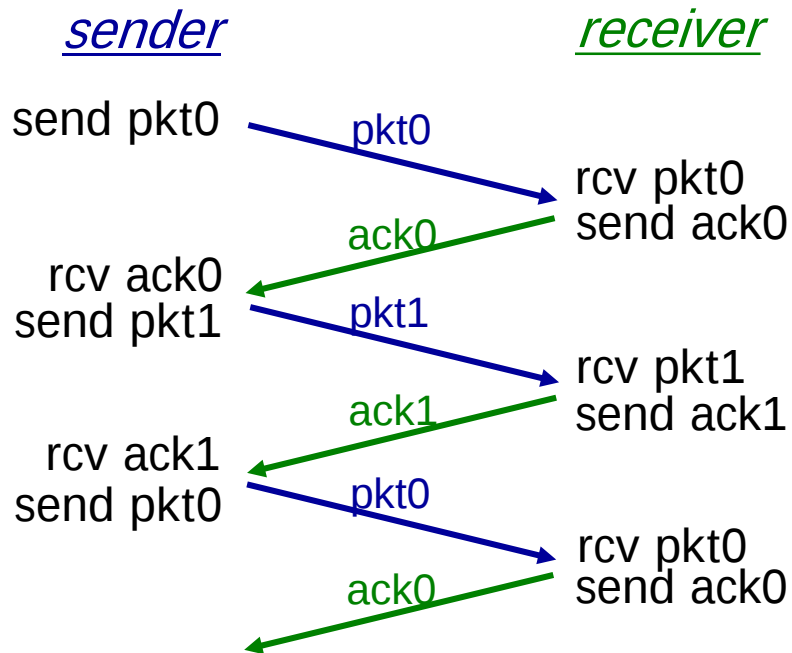
approach: sender waits “reasonable” amount of time for ACK

- 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 pkt being ACKed
- requires countdown timer

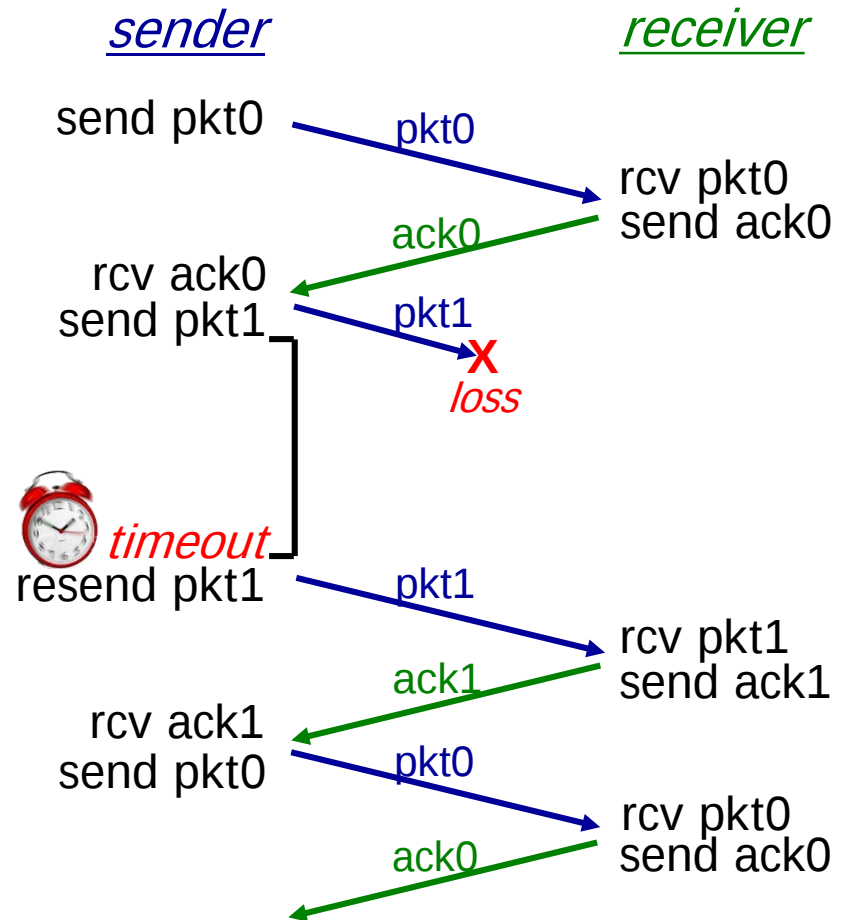
rdt3.0 sender



rdt3.0 in action

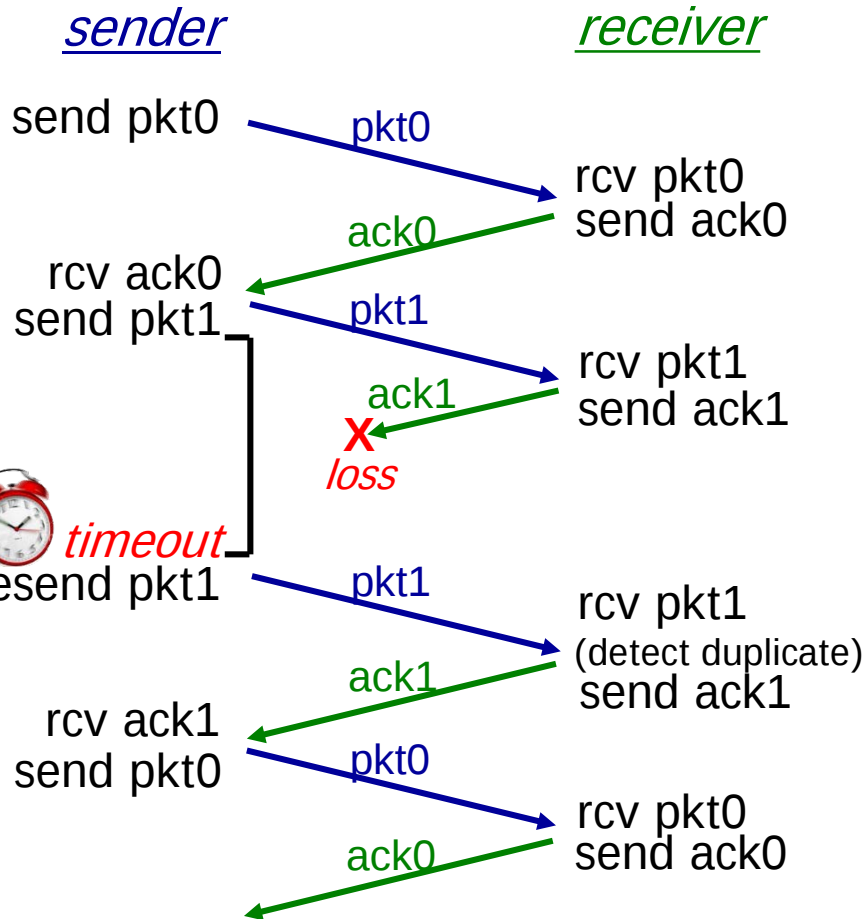


(a) no loss

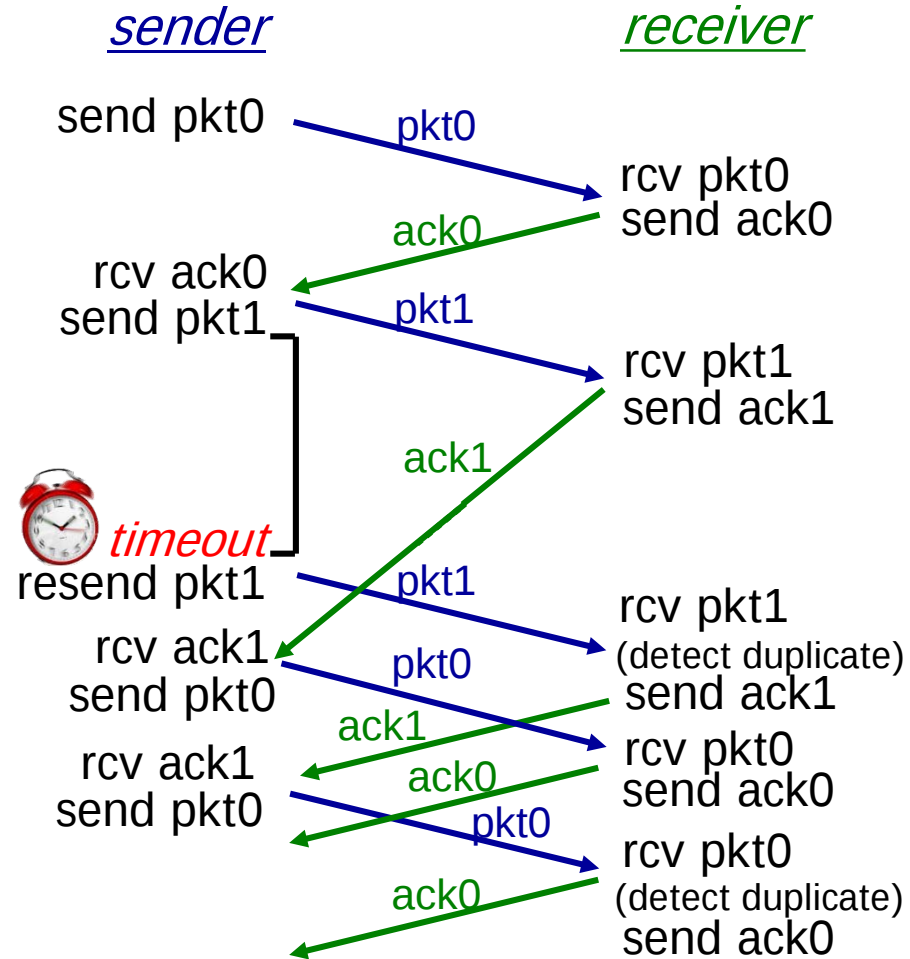


(b) packet loss

rdt3.0 in action



(c) ACK loss



(d) premature timeout/ delayed ACK

Performance of rdt3.0

- rdt3.0 is correct, but way too slow to use in practice
- e.g.: 1 Gbps link, 15 ms prop. delay, 8000 bit packet:

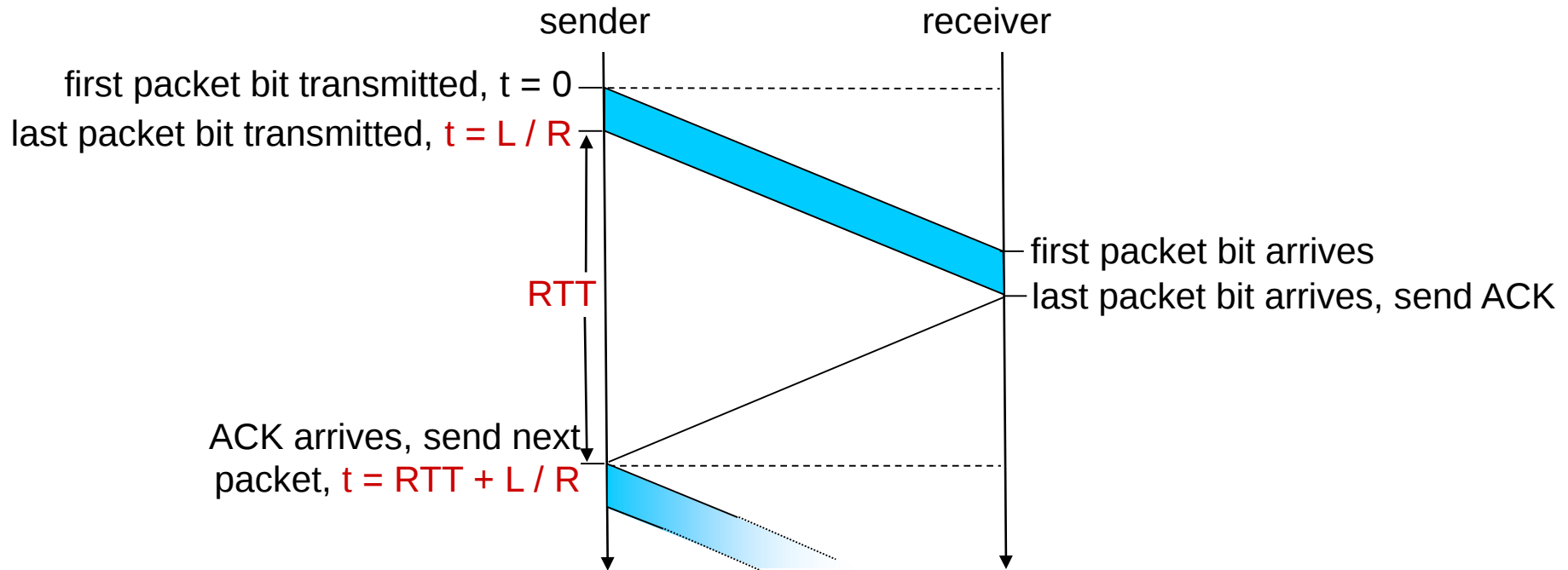
$$D_{trans} = \frac{L}{R} = \frac{8000 \text{ bits}}{10^9 \text{ bits/sec}} = 8 \text{ microseconds}$$

- U_{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, 1KB pkt every 30 msec: 33kB/sec thruput over 1 Gbps link
- network protocol limits use of physical resources!

rdt3.0: stop-and-wait operation



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

Before You Go

On a sheet of paper, answer the following (ungraded) question (no names, please):

What one thing could the instructor change to improve your learning?