CS 305: Computer Networks

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Lecture 6：Transport Layer

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Chapter 2: outline

2.1 principles of network applications

2.2 Web and HTTP

2.3 electronic mail

- SMTP, POP3, IMAP

2.4 DNS

2.5 P2P applications

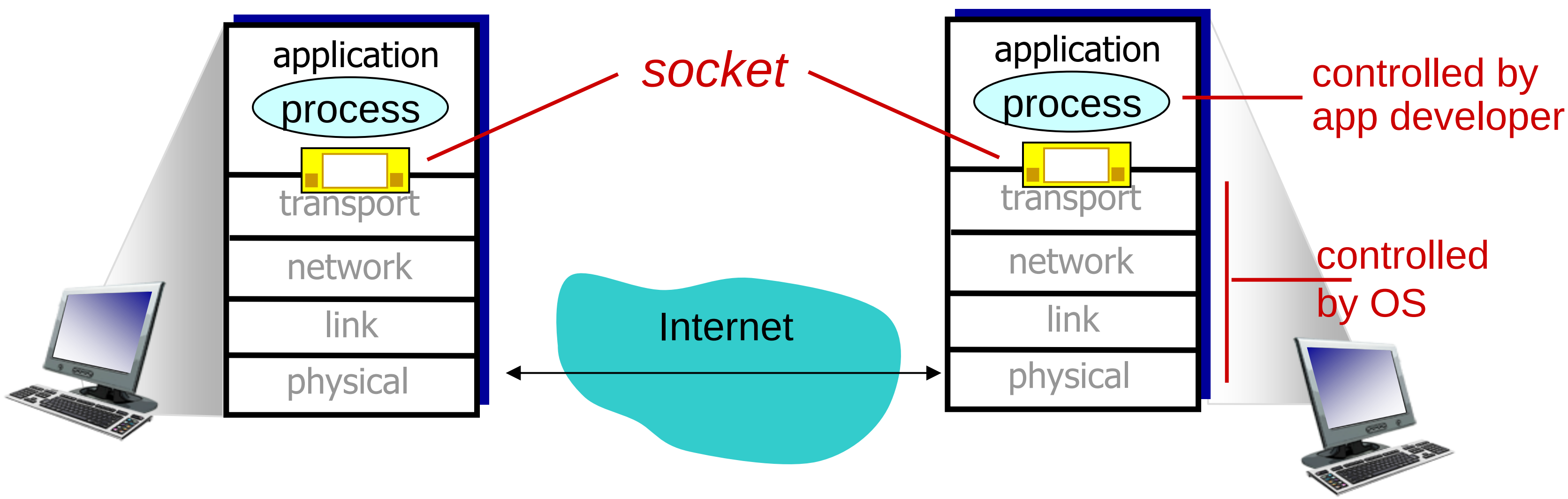
2.6 video streaming and content distribution networks

2.7 socket programming with UDP and TCP

Socket programming

Goal: learn how to build client/server applications that communicate using sockets

Socket: door between application process and end-end-transport protocol



Socket programming

Socket programming: how we can use socket API for creating communication between client and server processes.

Two socket types for two transport services:

- UDP: unreliable datagram
- TCP: reliable, connection-oriented

Application Example:

1. client reads a line of characters (data) from its keyboard and sends data to server
2. server receives the data and converts characters to uppercase
3. server sends modified data to client
4. client receives modified data and displays line on its screen

Socket programming with UDP

UDP: no "connection" between client & server

- no handshaking before sending data
- sender explicitly attaches destination IP address and port number to each packet
- receiver extracts sender IP address and port number from received packet

UDP: transmitted data may be lost or received out-of-order

Application viewpoint:

- UDP provides *unreliable* transfer of groups of bytes ("datagrams") between client and server

Client/server socket interaction: UDP

server (running on serverIP)

create socket, port= x:
serverSocket =
socket(AF_INET,SOCK_DGRAM)

read UPD datagram from
serverSocket

write reply to serverSocket
specifying client address,
port number

client

create socket:
clientSocket =
socket(AF_INET,SOCK_DGRAM)

Create datagram with server IP and
port=x; send datagram via
clientSocket

read datagram from
clientSocket

close
clientSocket

Segment: the transport-layer packet for TCP
Datagram: the packet for UDP

Example app: UDP server

Python UDPServer

include Python's socket library

create UDP socket

bind socket to local port number 12000

loop forever

Read from UDP socket into message, getting client's address (client IP and port)

send upper case string back to this client

IPv4

UDP socket

UDP socket is identified by destination IP address and port number

```
from socket import *
serverPort = 12000
serverSocket = socket(AF_INET, SOCK_DGRAM)
serverSocket.bind(('', serverPort))
print ("The server is ready to receive")
while True:
    message, clientAddress = serverSocket.recvfrom(2048)
    modifiedMessage = message.decode().upper()
    serverSocket.sendto(modifiedMessage.encode(),
                        clientAddress)
```

Example app: UDP client

Python UDPClient

We did not specify the client port number

either the IP address (e.g., "128.138.32.126") or the hostname (e.g., "cis.poly.edu")

```
from socket import *
serverName = 'hostname'
serverPort = 12000
clientSocket = socket(AF_INET,
                      SOCK_DGRAM)
message = raw_input('Input lowercase sentence:')
clientSocket.sendto(message.encode(),
                    (serverName, serverPort))
modifiedMessage, serverAddress =
                    clientSocket.recvfrom(2048)
print modifiedMessage.decode()
clientSocket.close()
```

Create the client's socket → clientSocket = socket(AF_INET, SOCK_DGRAM)

get user keyboard input → message = raw_input('Input lowercase sentence:')

Attach server name, port to message; send into socket → clientSocket.sendto(...)

IP + portnumber → serverAddress

read reply characters from socket into string → modifiedMessage, serverAddress = clientSocket.recvfrom(2048)

print out received string and close socket → print modifiedMessage.decode(); clientSocket.close()

Socket programming with TCP

Client must contact server

- server process must first be running
- server must have created socket (door) that welcomes client's contact

Client contacts server by:

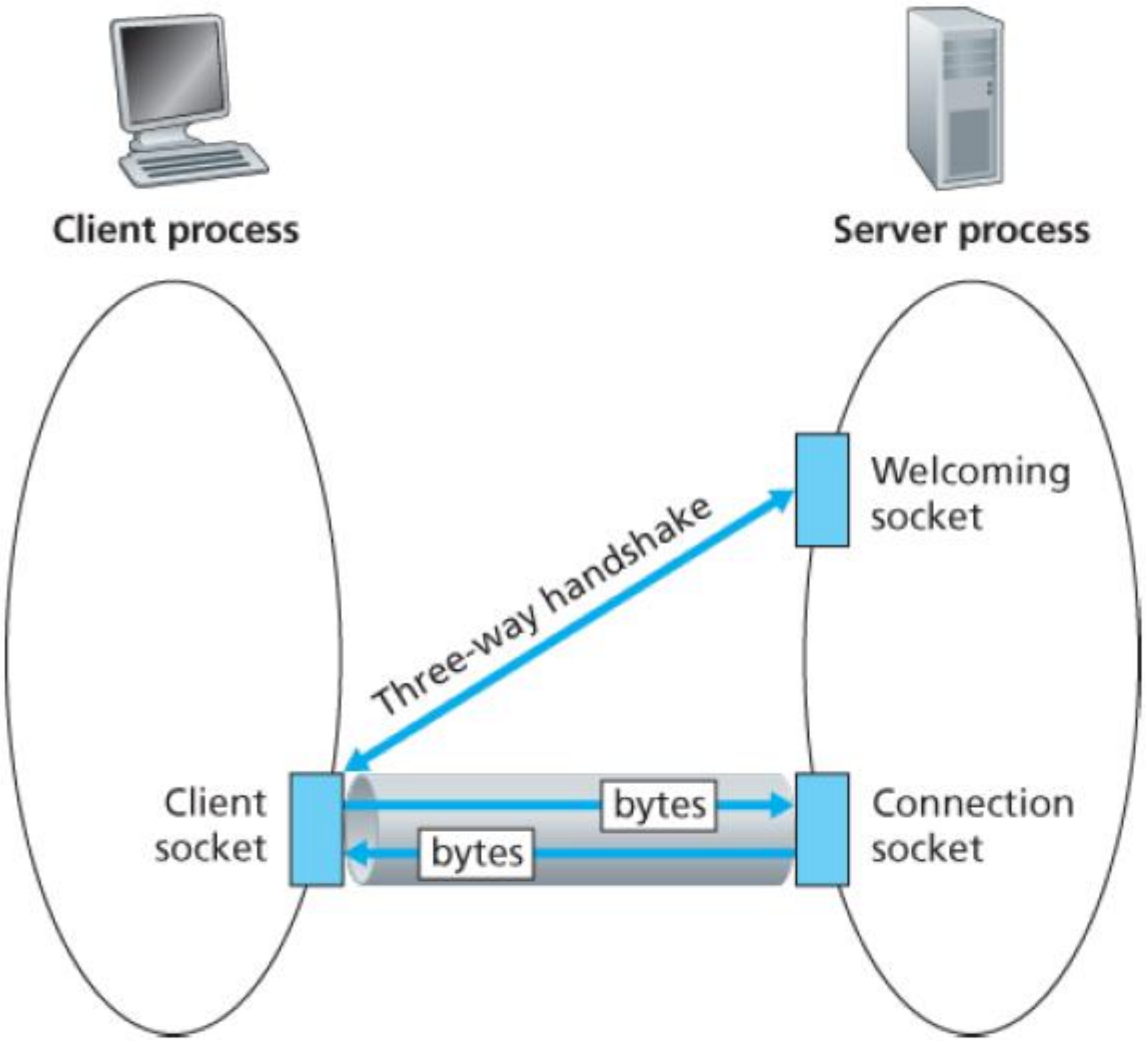
- Creating TCP socket, specifying IP address, port number of server process
- Client TCP establishes connection to server TCP

- when contacted by client, *server TCP creates new socket* for server process to communicate with that particular client
  - allows server to talk with multiple clients
  - source port numbers used to distinguish clients (more in Chap 3)

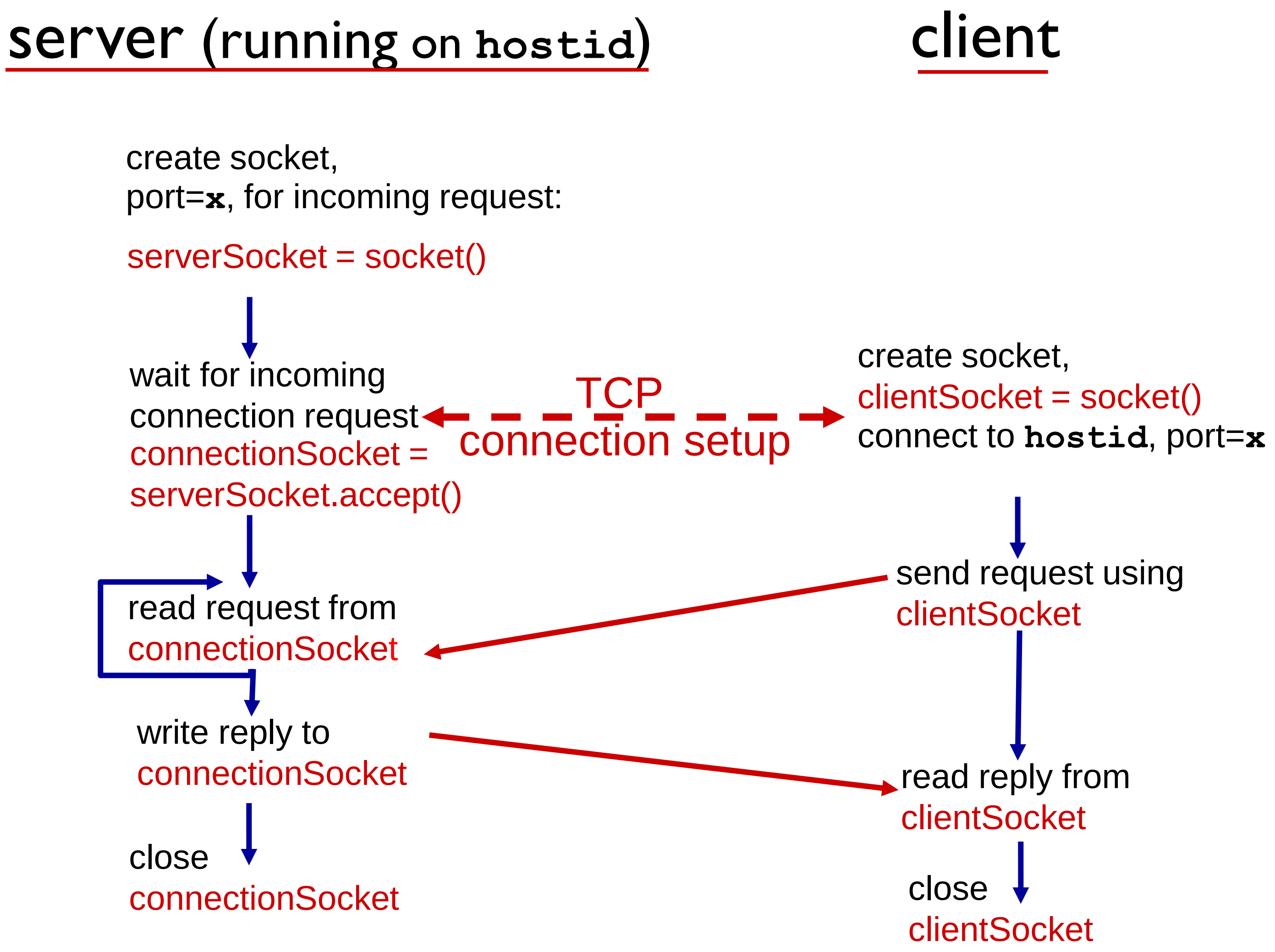
TCP socket is identified by (destination IP address, destination port number, source IP address, source port number)

Application viewpoint:
TCP provides reliable, in-order byte-stream transfer ("pipe") between client and server

Socket programming with TCP



Client/server socket interaction: TCP



Example app: TCP server

Python TCPServer

connectionSocket is identified by
- destination IP address and port number
- source IP address and port number

TCP socket

create TCP welcoming socket

server begins listening for incoming TCP requests

loop forever

server waits on accept() for incoming requests, new socket created on return

read bytes from socket (but not address as in UDP)

close connection to this client (but *not* welcoming socket)

```
from socket import *
serverPort = 12000
serverSocket = socket(AF_INET,SOCK_STREAM)
serverSocket.bind((‘’,serverPort))
serverSocket.listen(1)
print ‘The server is ready to receive’
while True:
    connectionSocket, addr = serverSocket.accept()

    sentence = connectionSocket.recv(1024).decode()
    capitalizedSentence = sentence.upper()
    connectionSocket.send(capitalizedSentence.
                                            encode())
    connectionSocket.close()
```

Example app: TCP client

Python TCPClient

create TCP socket for server

No need to attach server name, port

```
from socket import *
serverName = ’servername’
serverPort = 12000
clientSocket = socket(AF_INET, SOCK_STREAM)
clientSocket.connect((serverName,serverPort))
sentence = raw_input(‘Input lowercase sentence:’)
clientSocket.send(sentence.encode())
modifiedSentence = clientSocket.recv(1024)
print (‘From Server:’, modifiedSentence.decode())
clientSocket.close()
```

Chapter 2: summary

our study of network apps now complete!

- application architectures
  - client-server
  - P2P
- application service requirements:
  - reliability, bandwidth, delay
- Internet transport service model
  - connection-oriented, reliable: TCP
  - unreliable, datagrams: UDP
- specific protocols:
  - HTTP
  - SMTP, POP, IMAP
  - DNS
  - P2P: BitTorrent
- video streaming, CDNs
- socket programming:

  TCP, UDP sockets

Chapter 2: summary

most importantly: learned about protocols!

- typical request/reply message exchange:
  - client requests info or service
  - server responds with data, status code
- message formats:
  - *headers*: fields giving info about data
  - *data:* info being communicated

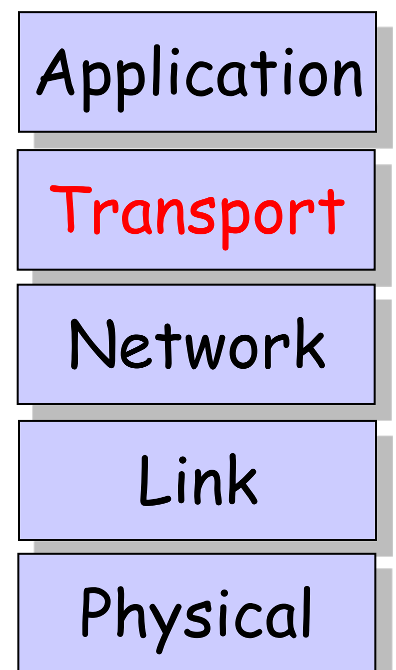
important themes:

- centralized vs. distributed
- stateless vs. stateful
- reliable vs. unreliable message transfer

Chapter 3: Transport Layer

Our goals:

- 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



Chapter 3 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

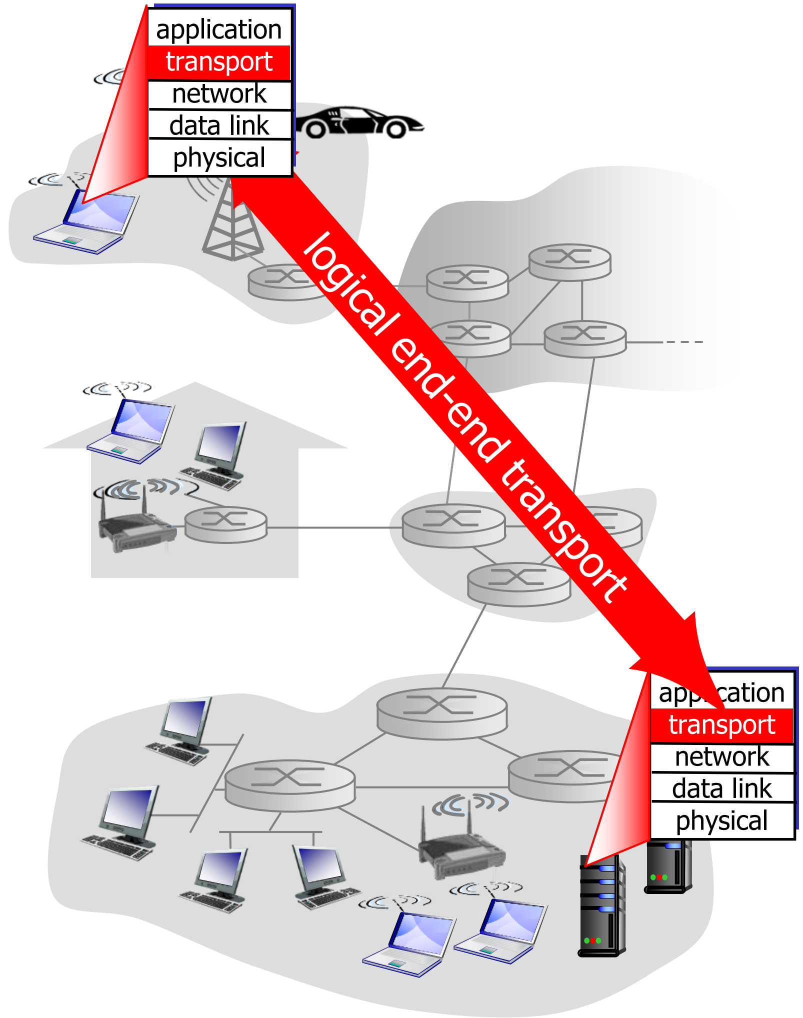
- segment structure
- reliable data transfer
- flow control
- connection management

3.6 principles of congestion control

3.7 TCP congestion control

Transport services and protocols

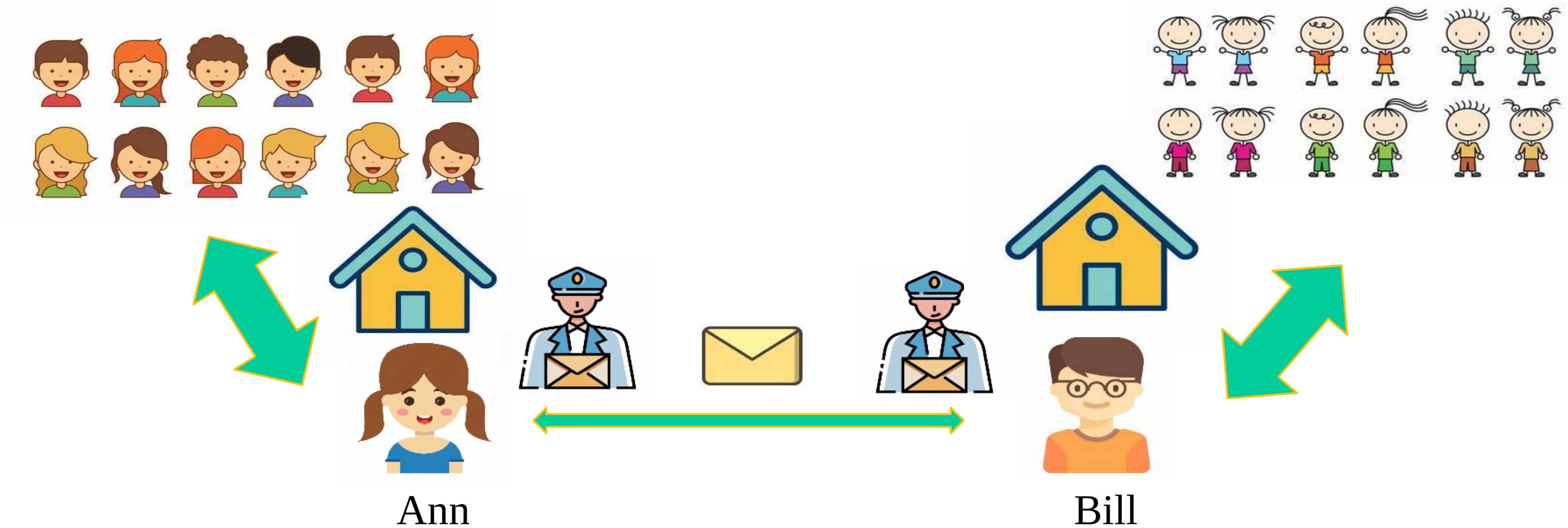
- Provide *logical communication* between app processes running on different hosts
- Transport protocols run in end systems
  - send side: breaks app messages into *segments*, passes to network layer
  - rcv side: reassembles segments into messages, passes to app layer
- more than one transport protocol 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

Transport vs. network layer



household analogy:

- hosts = houses
- processes = kids
- app messages = letters in envelopes
- transport protocol = Ann and Bill
- network-layer protocol = postal service

Susan and Harvey substitute for them and provide different delivery services?

Transport vs. network layer

- The services that a transport protocol can provide are often constrained by the service model of the underlying network-layer protocol.
  - delay or bandwidth guarantees

- Certain services can be offered by a transport protocol even when the underlying network protocol doesn't offer the corresponding service at the network layer.
  - Reliable data transfer; security

Internet transport-layer protocols

Network layer : Internet protocol (IP) is a best effort delivery service, unreliable

UDP: unreliable, unordered delivery:

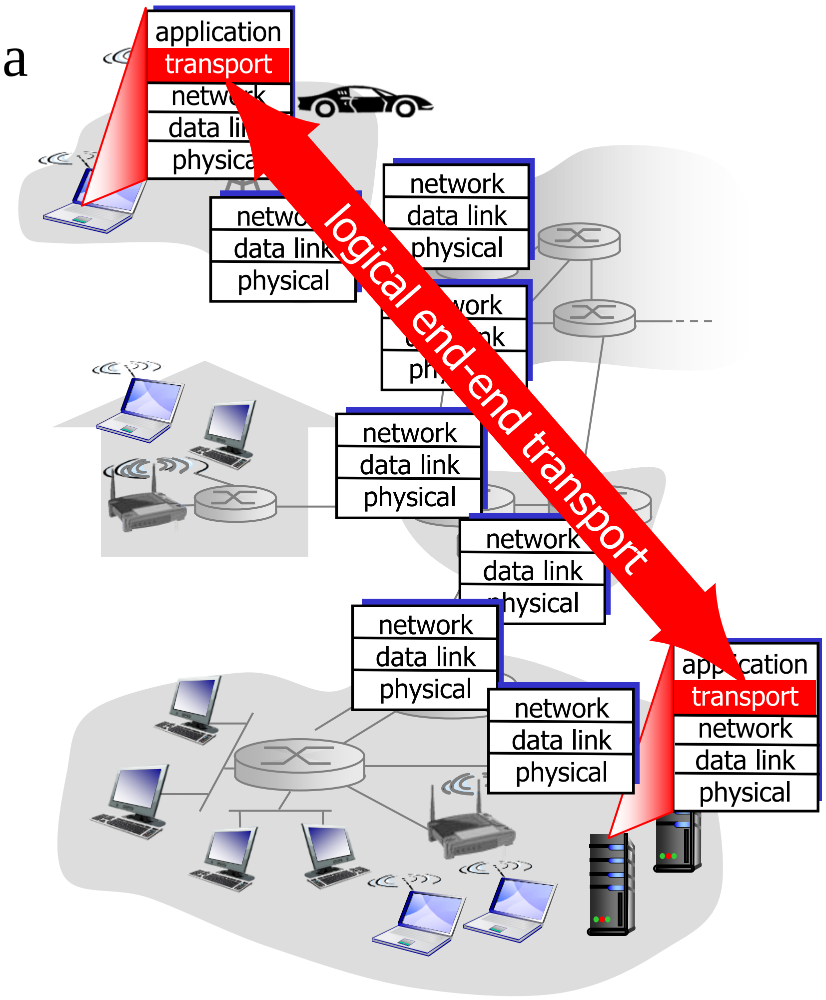
- no-frills extension
- process-to-process data delivery and error checking

TCP : reliable, in-order delivery

- congestion control
- flow control
- connection setup

Services not available:

- delay guarantees
- bandwidth guarantees



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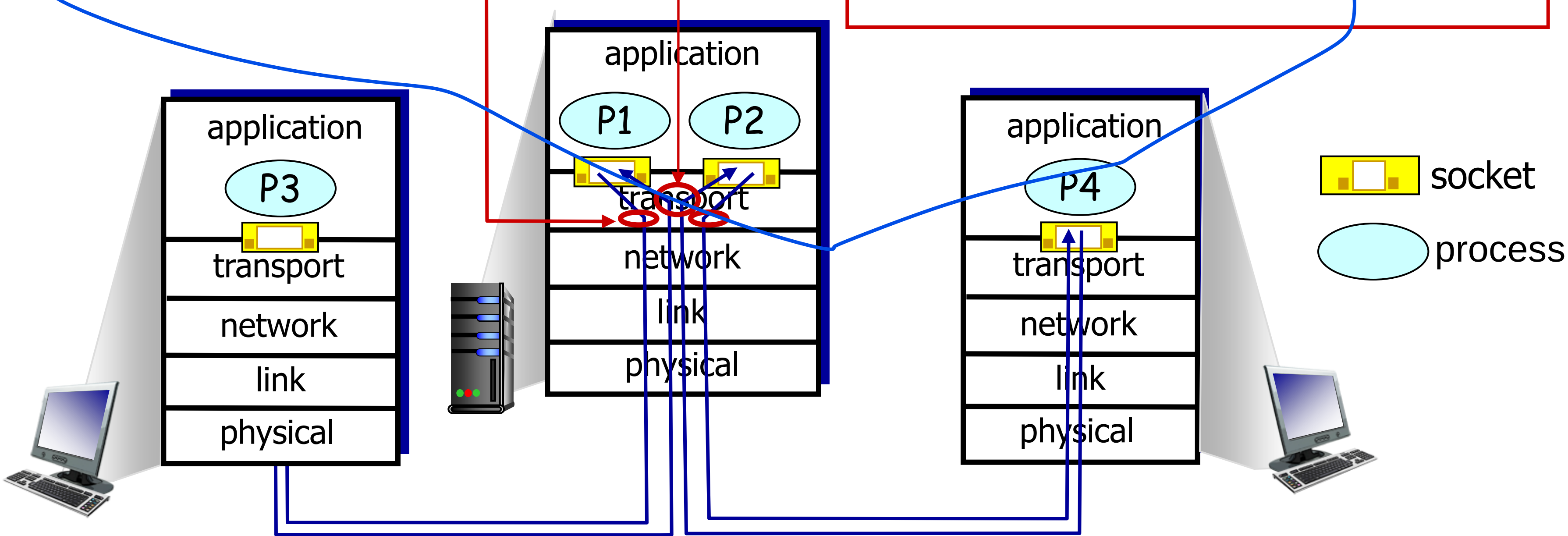
Multiplexing/demultiplexing

Multiplexing and demultiplexing: extending the host-to-host delivery service to a process-to-process delivery service for applications running on the hosts.

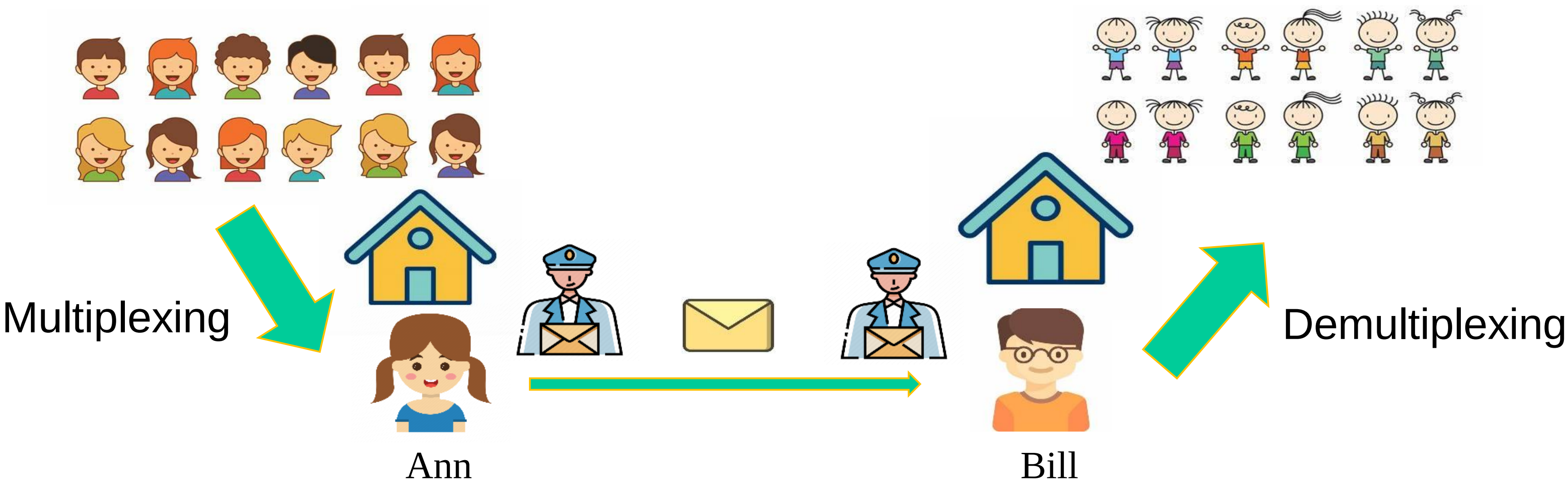
Multiplexing at sender:
handle data from multiple sockets, add transport header (later used for demultiplexing)

Demultiplexing at receiver:
use header info to deliver received segments to correct socket



Ann and Bill example?

Multiplexing/demultiplexing



How demultiplexing works？
- Each child must have an identifier（e.g.，name, ID）

How demultiplexing works

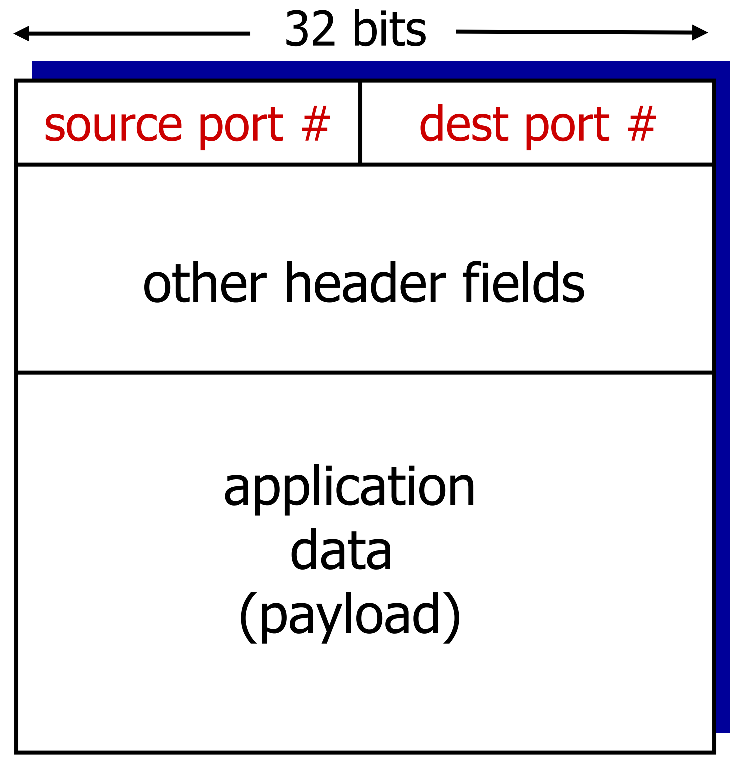
Multiplexing/demultiplexing:

- sockets have unique identifiers
- each segment have special fields that indicate the socket to which the segment is to be delivered.

Sending host: Host uses *IP addresses & port numbers* to direct segment to appropriate socket

Receiving host: Host receives IP datagrams from network layer

- each datagram has source IP address, destination IP address
- each datagram carries one transport-layer segment



TCP/UDP segment format

Port number: a 16-bit number, ranging from 0 to 65535:

- well-known port numbers: 0-1023

Overview

- UDP: Connectionless demux
- TCP: Connection-oriented demux

UDP: Connectionless demux

- *recall:* created socket has local port #:

  clientSocket = socket(AF_INET, SOCK_DGRAM)

  Automatically assigns a port number

  clientSocket.bind((‘’,19157))

  Optional; at the server side, usually assign port number

- *recall:* when creating datagram to send into UDP socket, must specify destination IP address and destination port #

  clientSocket.sendto(message.encode(), (serverName, serverPort))

UDP socket is fully identified by **a two-tuple** consisting of a destination IP address and a destination port number.

when host receives UDP segment:

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

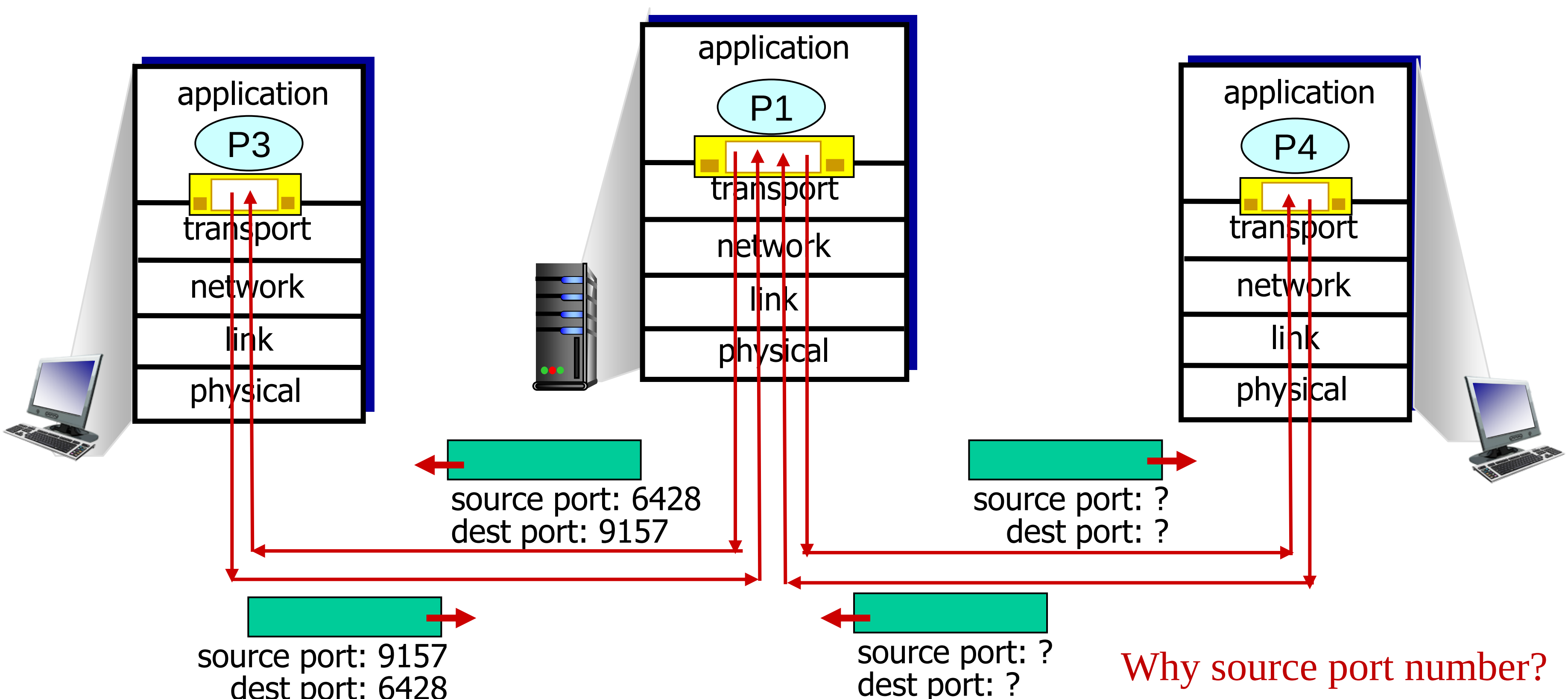
IP datagrams with *same dest. port #,* but different source IP addresses and/or source port numbers will be directed to *same socket* at destination

Connectionless demux: example

```
mysocket2 =
   socket(AF_INET,
   SOCK_DGRAM)
mysocket2.bind
   ((‘’,9157))
```

```
serversocket =
   socket(AF_INET,
   SOCK_DGRAM)
serversocket.bind
   ((‘’,6428))
```

```
mysocket1 =
   socket(AF_INET,
   SOCK_DGRAM)
mysocket1.bind
   ((‘’,5775))
```



Why source port number?

- As the “return address”

Overview

- UDP: Connectionless demux
- TCP: Connection-oriented demux

TCP: Connection-oriented demux

- Server creates a welcome socket with port no.12000

  serversocket = socket(AF_INET, SOCK_STREAM)

  serversocket.bind((‘’,12000))

- Client connects to the server, the request is a TCP segment with a flag bit = 1

  clientsocket = socket(AF_INET, SOCK_STREAM)

  clientsocket.connect((ServerName,12000))

- Server creates a new socket to accept the connection

  connectionsocket, addr = serversocket.accept()

The server may maintain TCP connections with multiple clients, each has a different **connectionsocket**. How to demux?

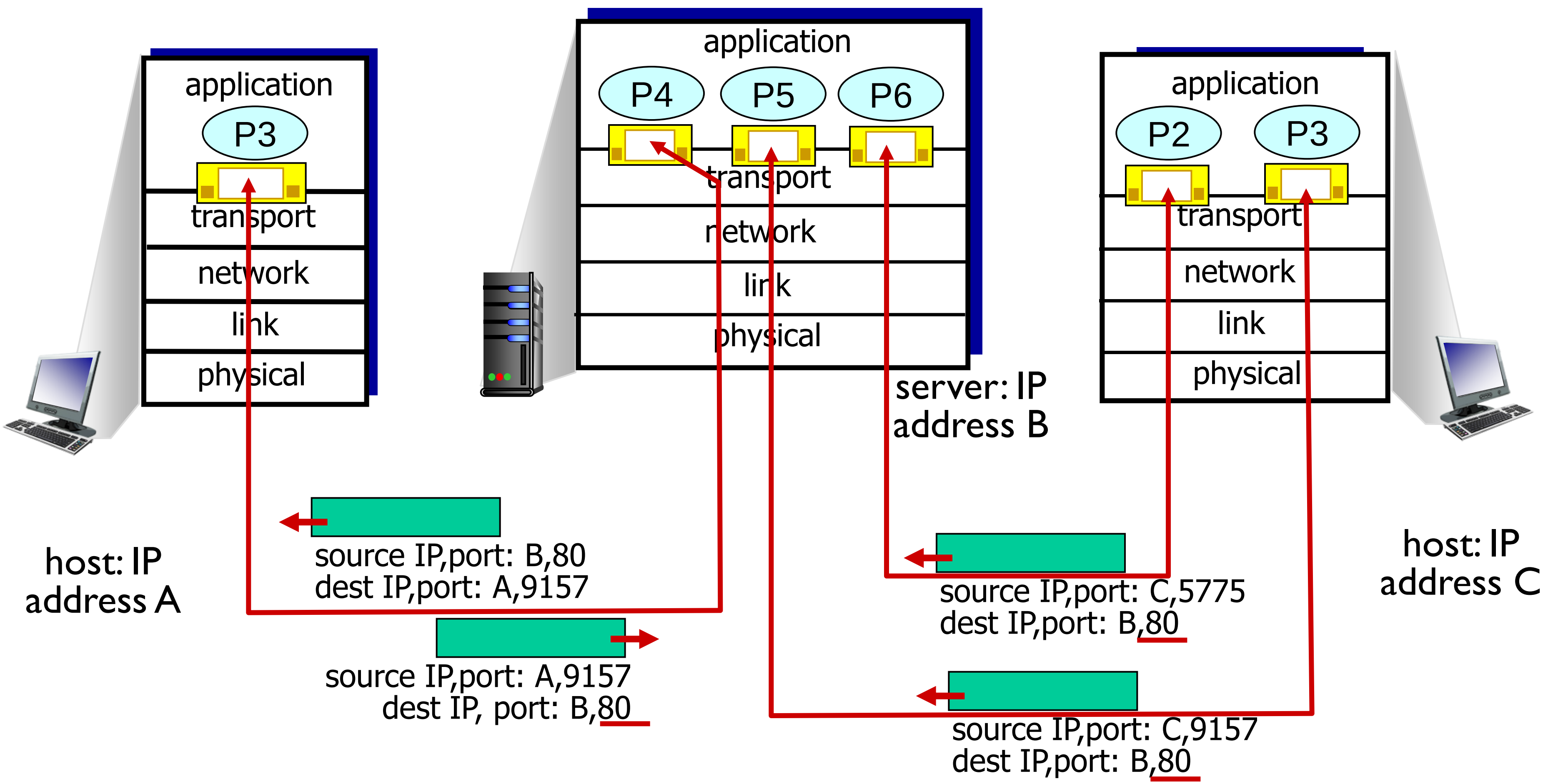
Connection-oriented demux

- 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

- Both the initial connection-establishment segments and the segments carrying HTTP requests will have destination port 80.
- non-persistent HTTP will have different socket for each request

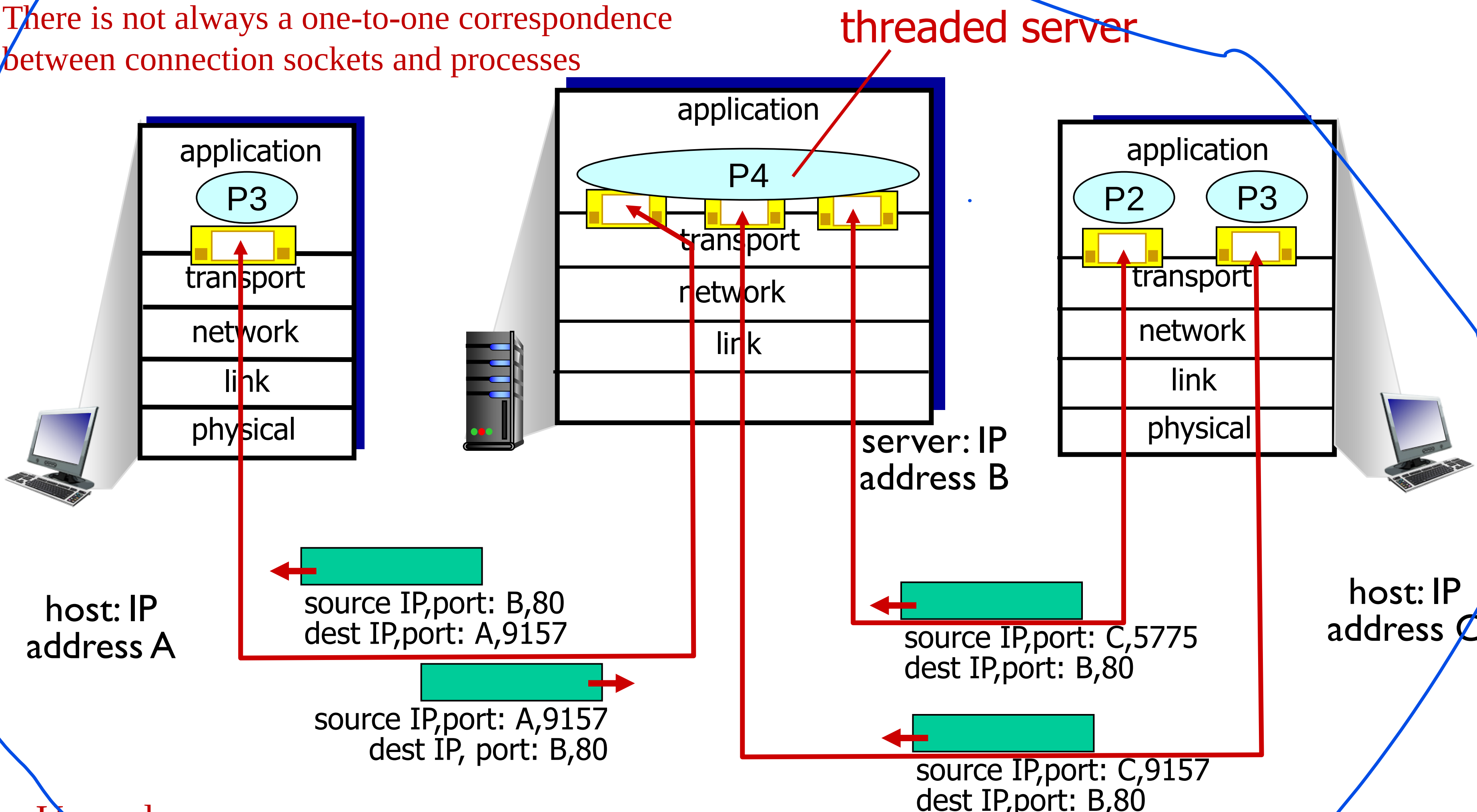
Connection-oriented demux: example



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

Connection-oriented demux: example

There is not always a one-to-one correspondence between connection sockets and processes



Use only one process

- Create a new thread with a new connection socket for each new client connection.
- A thread can be viewed as a lightweight subprocess.

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UDP: User Datagram Protocol

- "No frills," "bare bones" Internet transport protocol
  - Multiplexing/demultiplexing; light error checking
- "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
  - No congestion control

Advantage?

- No congestion control: Immediately pass the segment to network layer
- No connection-establish delay
- No connection state: server can support more clients
- Smaller packet overhead

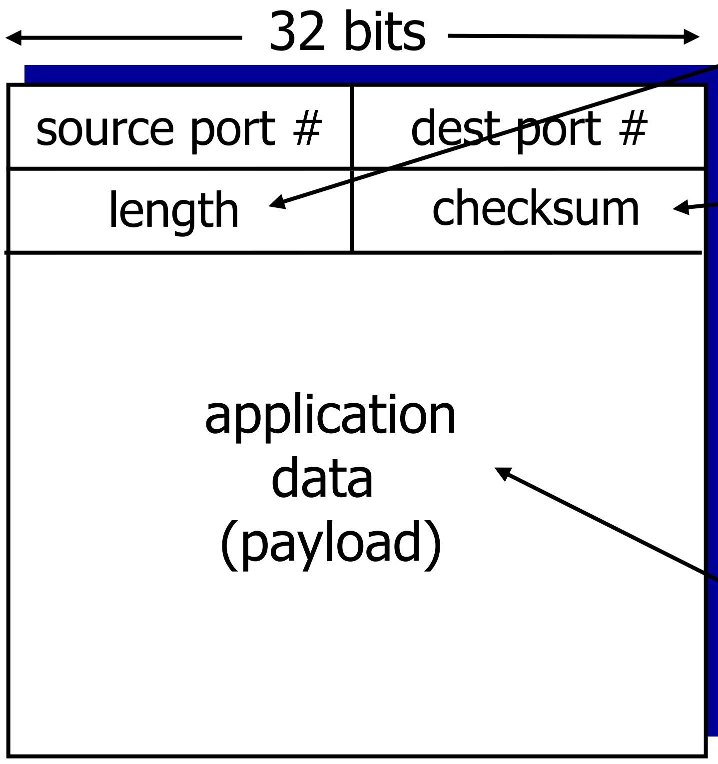
Disadvantage?

- No congestion control: congestion, overflow, fairness
- Not reliable

UDP: User Datagram Protocol

- UDP is used in:
  - streaming multimedia apps (loss tolerant, rate sensitive)
  - DNS
- reliable transfer over UDP:
  - add reliability at application layer
  - application-specific error recovery!

UDP: segment header



UDP segment format

length, in bytes of UDP segment (data + header)

Errors in segment (data + header)?

E.g., in DNS, query or response; in HTTP, objects

UDP checksum

Goal: detect "errors" (e.g., flipped bits) in transmitted segment (from source to destination)

Sender:

- treat segment contents, including header fields, as sequence of 16-bit integers
- checksum: 1s complement of the sum of segment contents
- sender puts checksum value into UDP checksum field

Receiver:

- check the sum of the segment
  - All bits are equal to 1 - no error detected. *But maybe errors nonetheless?* More later ….
  - Otherwise: error detected

Internet checksum: example

At the sender side, determine the check sum of the following three 16-bit words

0110011001100000
0101010101010101
1000111100001100

The sum of first two of these 16-bit words is

0110011001100000
0101010101010101
1011101110110101

Adding the third word to the above sum gives

1011101110110101
1000111100001100
10100101011000001

Wrap around

0100101011000010

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

The 1s complement is obtained by converting all the 0s to 1s and converting all the 1s to 0s.

Checksum：1011010100111101

Receiver Side？

Internet checksum: example

At the receiver side, all four 16-bit words are added, including the checksum:

- If no errors are introduced into the packet, then clearly the sum at the receiver will be 1111111111111111.
- If one of the bits is a 0, then we know that errors have been introduced into the packet.

Check the sum of the segment

- All bits are equal to 1 - **no error detected**. *But maybe errors nonetheless?* More later ….
- Otherwise: error detected

Internet checksum: example

Why UDP provides a checksum?

- no guarantee that all the links provide error checking
- bit errors could be introduced when segments are in memory
- “functions placed at the lower levels may be redundant or of little value when compared to the cost of providing them at the higher level.”

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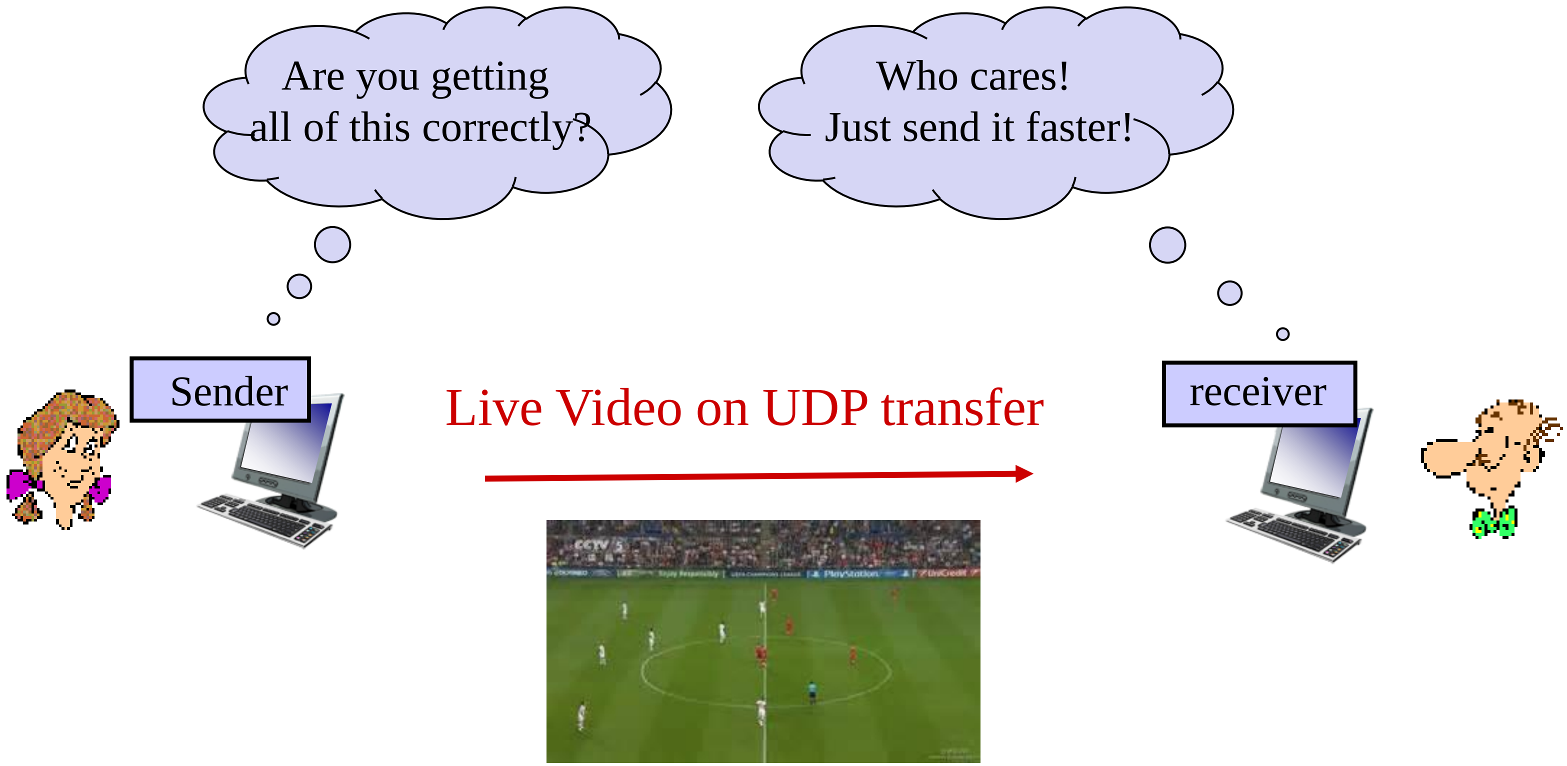
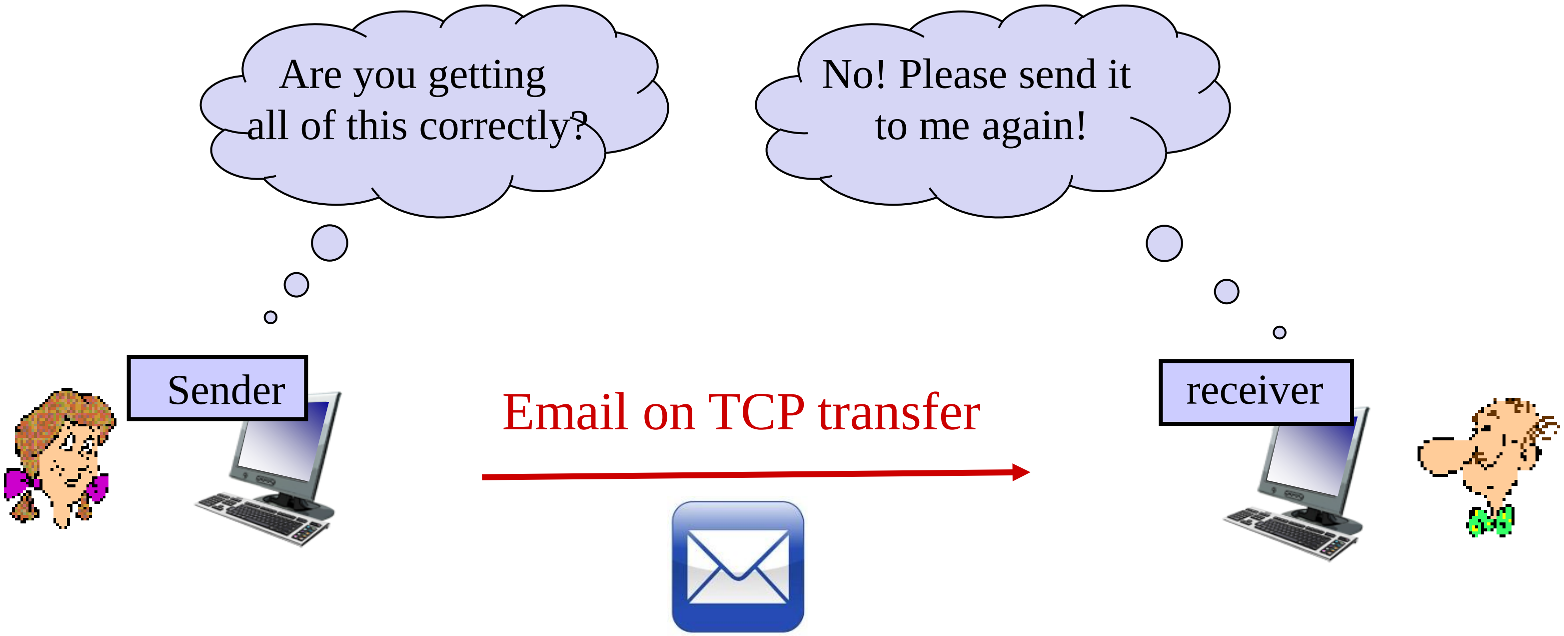
UDP Transfer

- UDP cannot guarantee reliable data transfer
- But, it's faster!



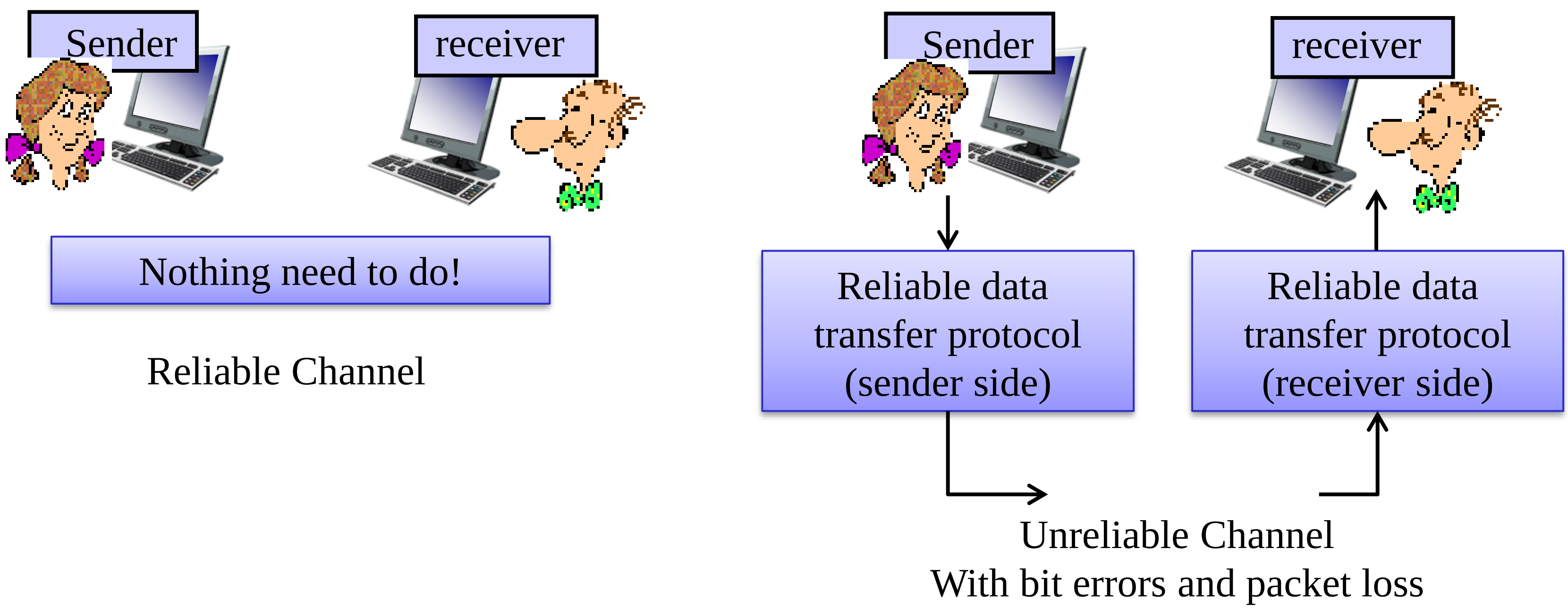
TCP Transfer

- TCP can guarantee reliable data transfer
- But, it's slower and more complex!

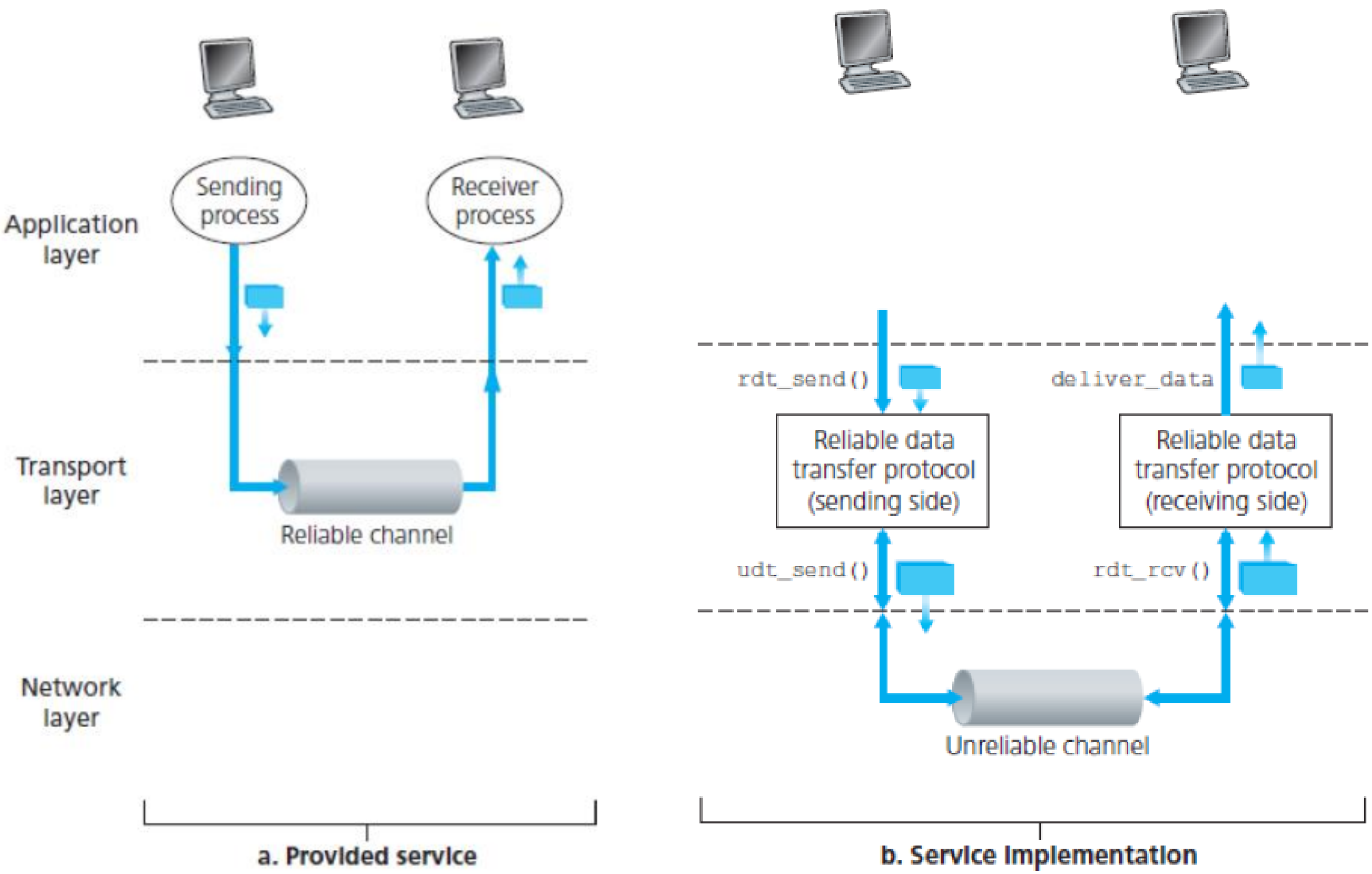


Reliable Data Transfer (rdt)

- In top-10 list of important networking topics!
- Reliable data transfer over unreliable channel :
  - Bit flip, lost, out-of-order
  - In this section, assume unreliable channel not reorder packets



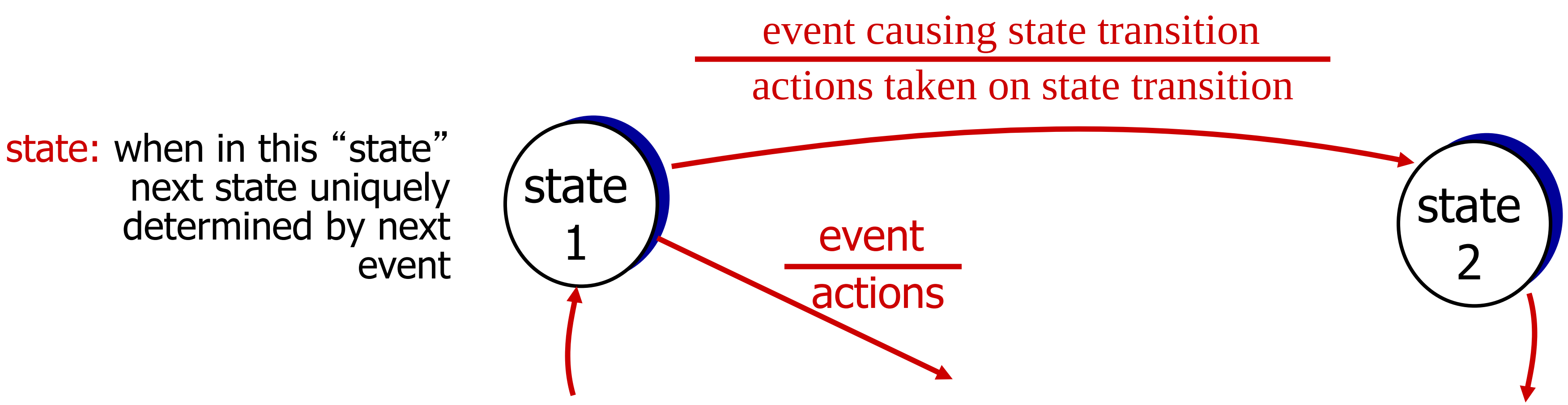
Reliable Data Transfer (rdt)



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



Overview

Roadmap:

- Perfectly reliable channel： rdt1.0
- Channel with bit error:
  - bit error in packet: rdt 2.0
  - bit error in ACK: 2.1
  - NAK-free: 2.2
- Lossy channel: rdt 3.0

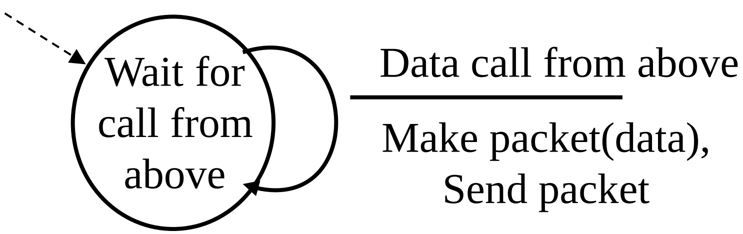
Summary of Techniques

- Checksum
- Sequence number
- ACK packets
- Retransmission
- Timeout

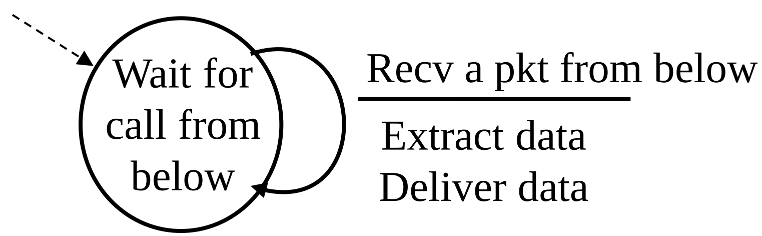
rdt1.0: reliable transfer over a reliable channel

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

- **Rdt 1.0:**
  - sender sends data into underlying channel
  - receiver reads data from underlying channel
  - Reliable channel, no need for feedback（no control message）

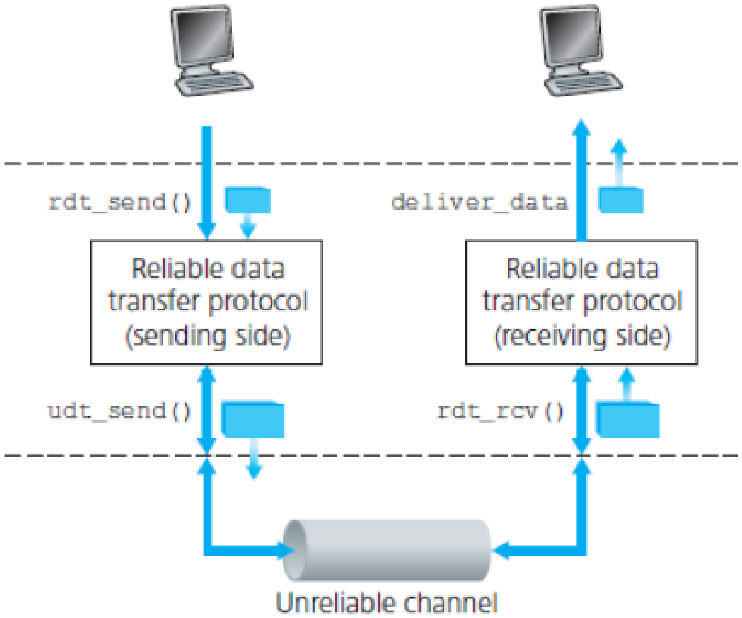


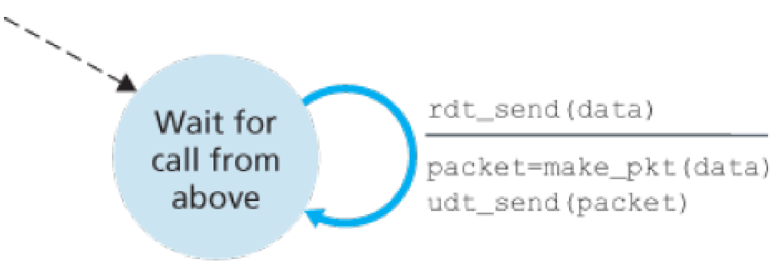
sender



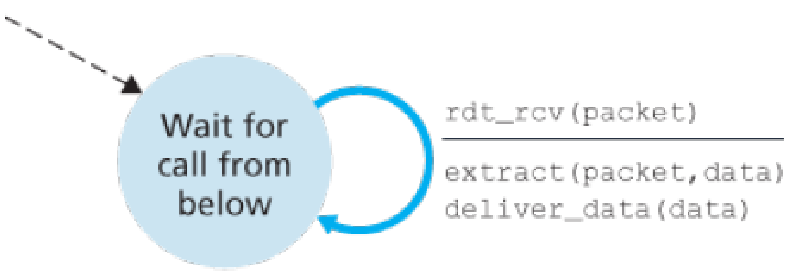
receiver

rdt1.0: reliable transfer over a reliable channel





sender

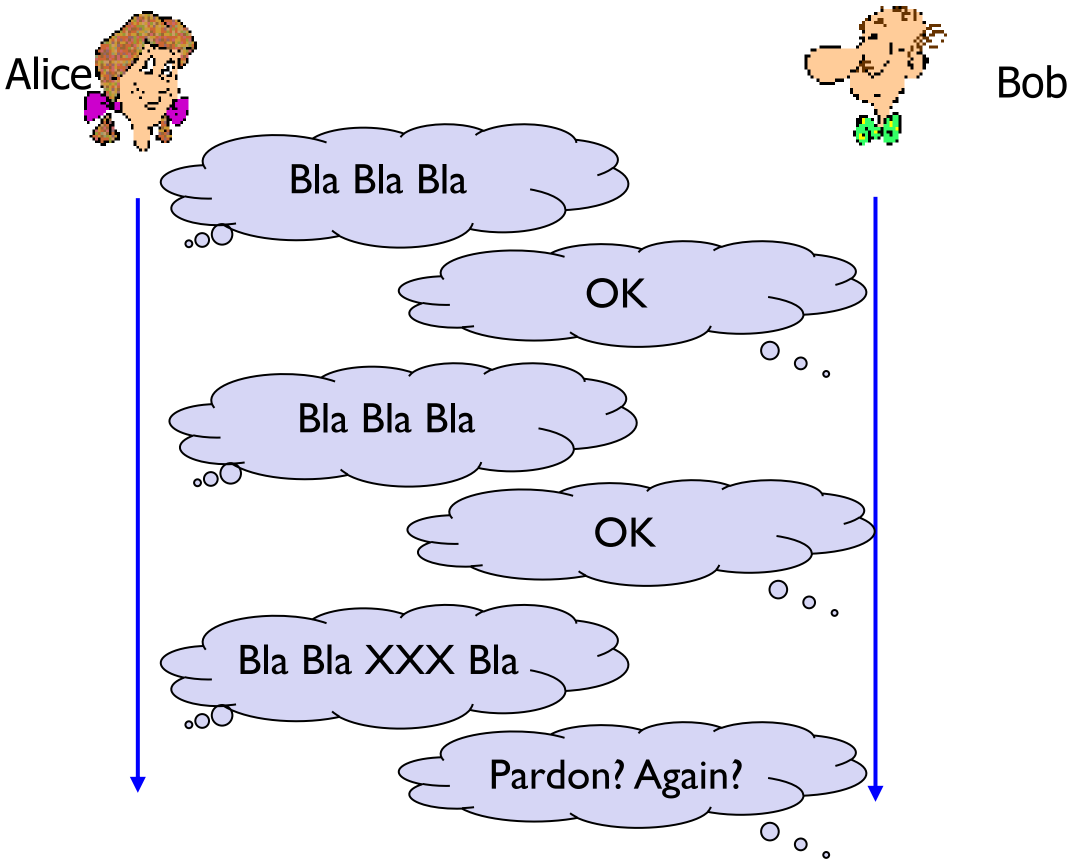


receiver

rdt2.0: channel with bit errors

- Underlying channel may flip bits (0 → 1) in packet

How do humans recover from "errors" during conversation?



rdt2.0: channel with bit errors

- Underlying channel may flip bits (0 → 1) in packet

How do humans recover from "errors" during conversation?

- 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

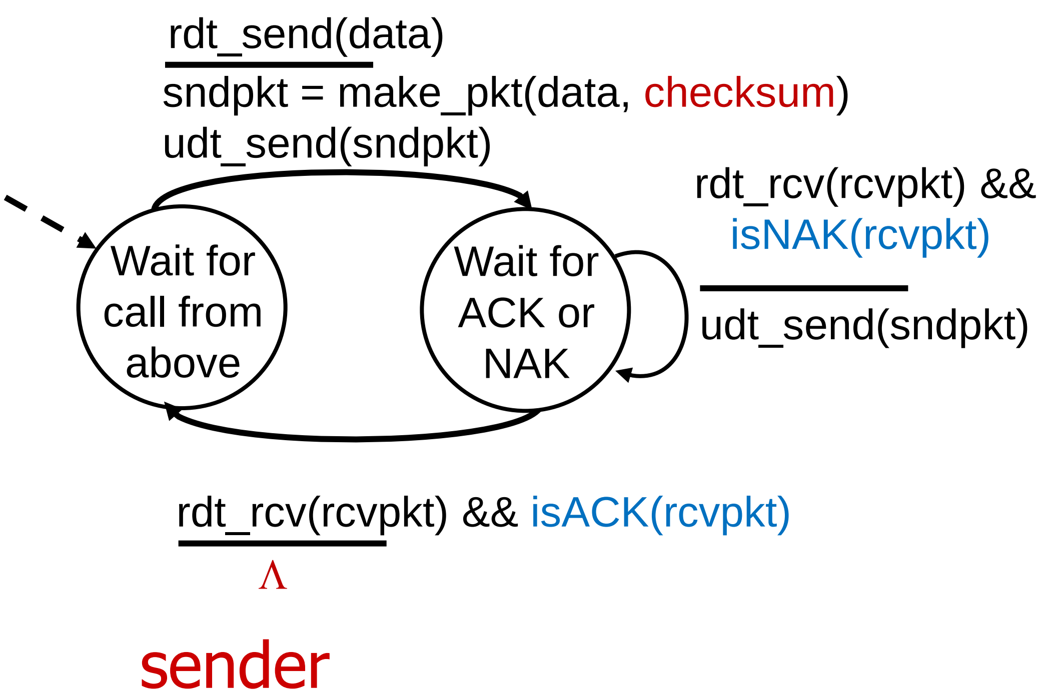
- new mechanisms in **rdt2.0** (beyond **rdt1.0**):
  - error detection
  - receiver feedback: control msgs (ACK,NAK) rcvr->sender
  - retransmission

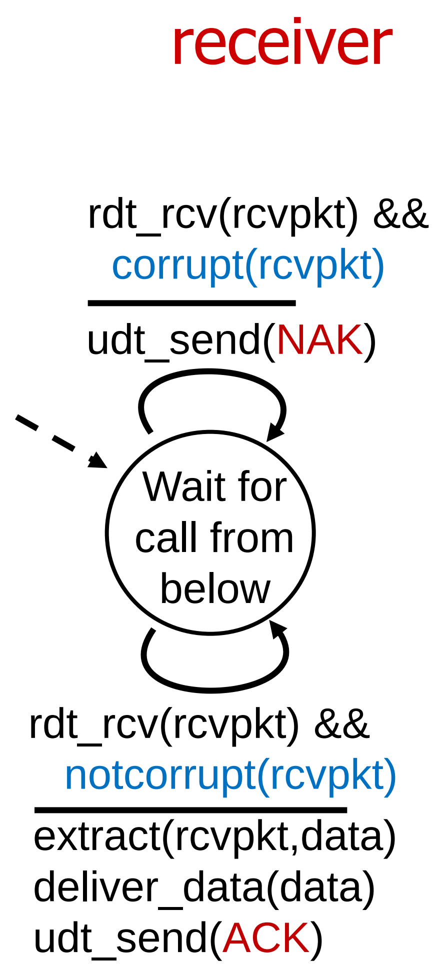
rdt2.0: channel with bit errors

- Key mechanisms:
  - error detection
  - feedback: control msgs (ACK, NAK) from receiver to sender
  - retransmission

- Error detection: checksum
- Feedback messages:
  - *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

rdt2.0: FSM specification

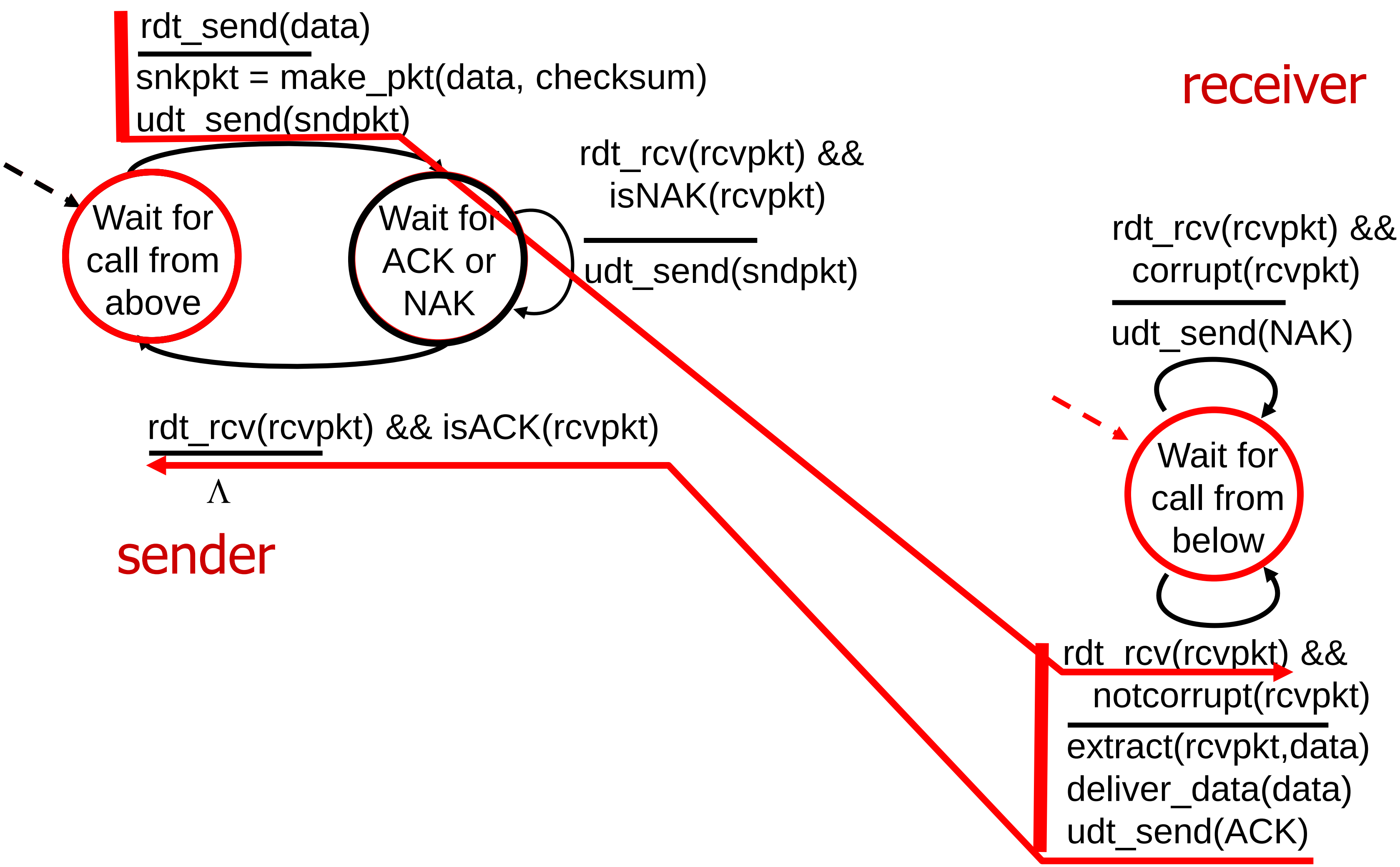




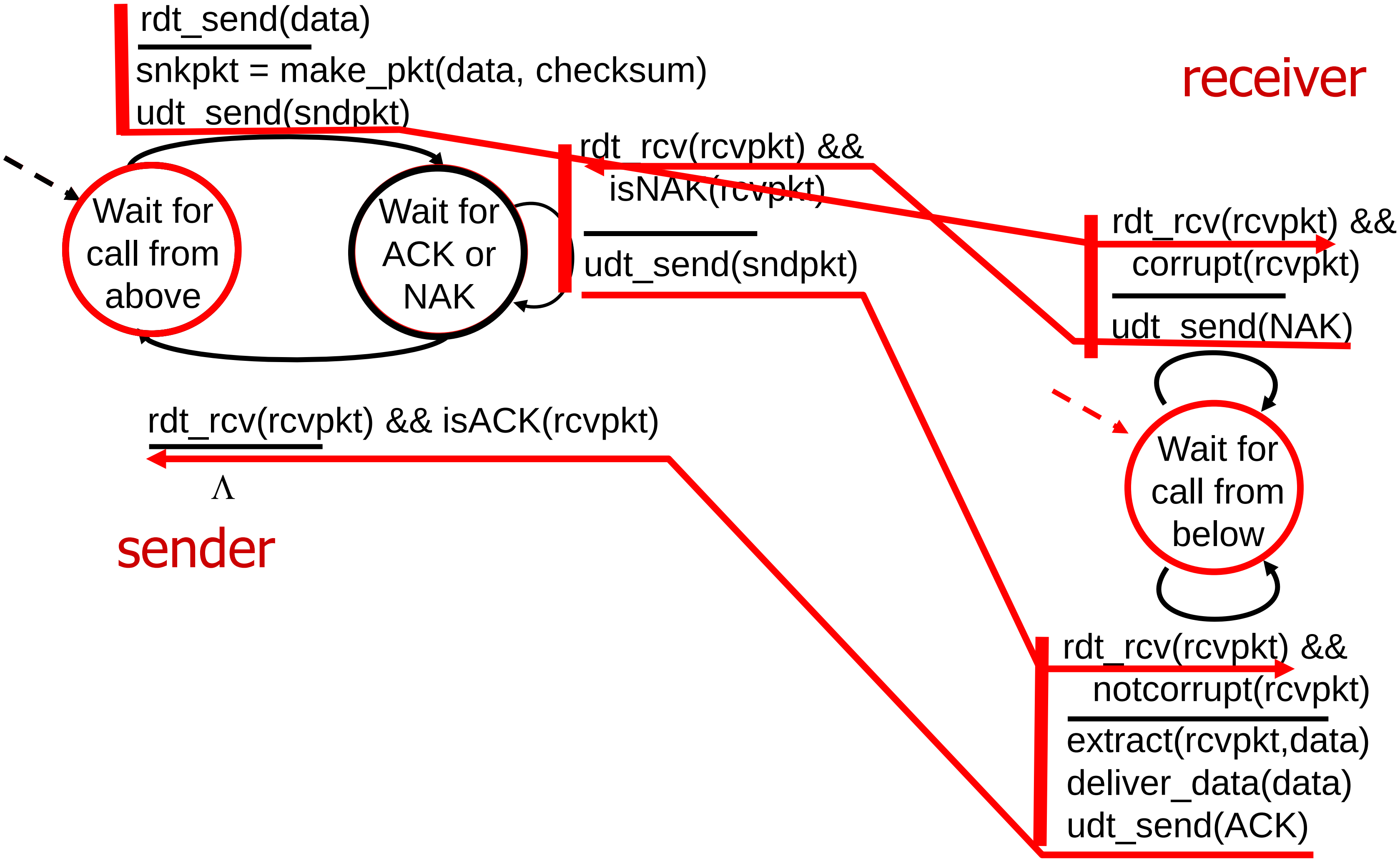
Stop and wait

Sender sends one packet, then waits for receiver response

rdt2.0: operation with no errors



rdt2.0: error scenario



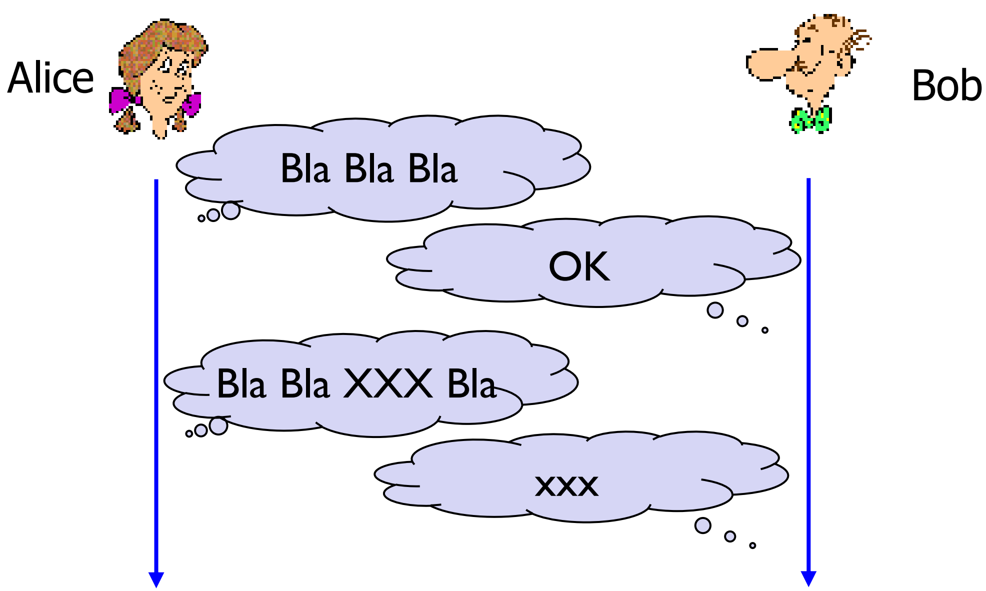
rdt2.0 has a fatal flaw!

The possibility that ACK or NAK packet could be corrupted:

- Checksum bits

Handling corrupted ACKs or NAKs:

- Option 1: “blabla…”, “OK”, “What did you say?”, “OK”
  - “What did you say?”, “What did you say?”, …
- Option 2: add enough checksum to recover
- Option 3: when garbled ACK or NAK, retransmit



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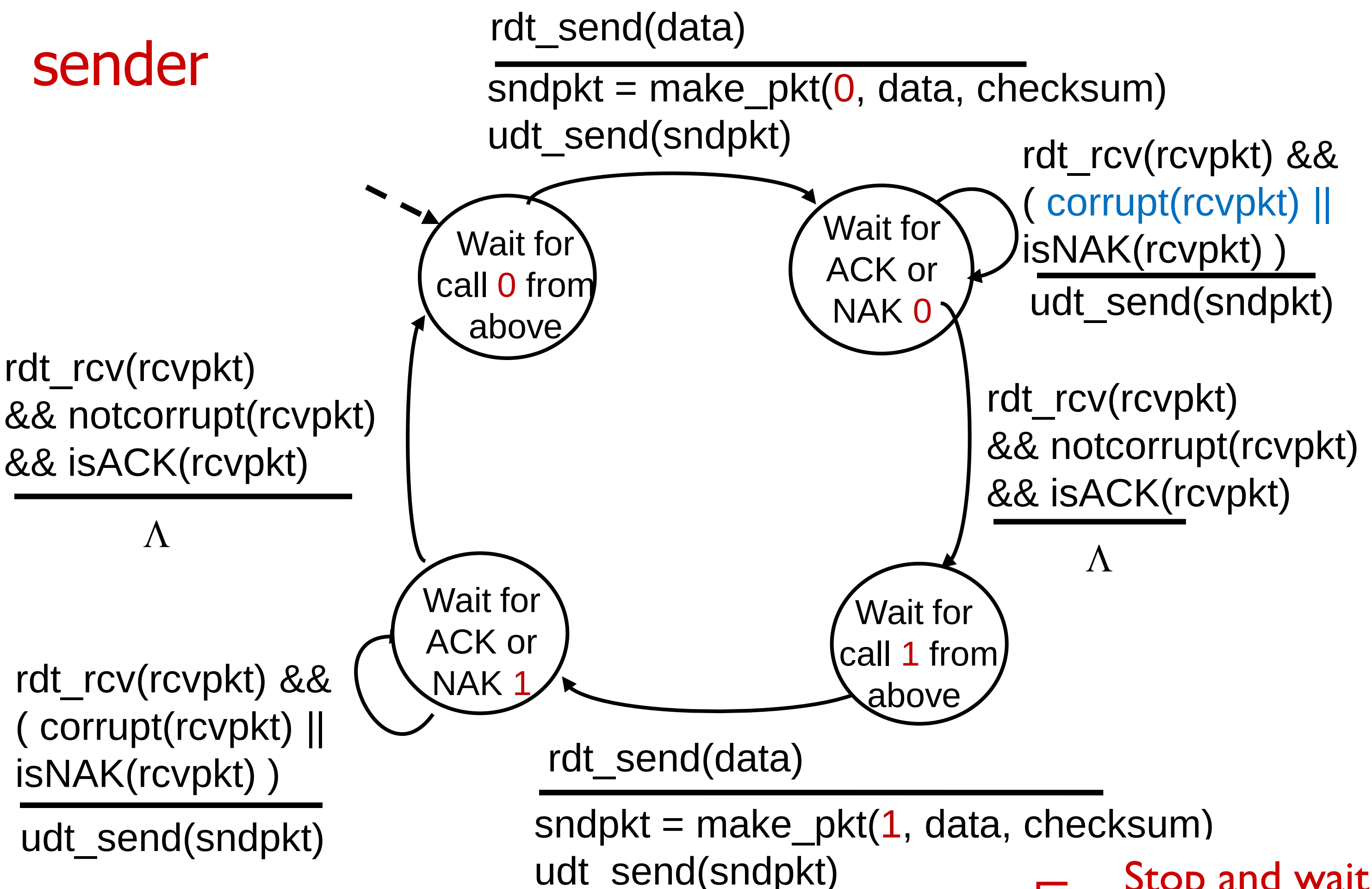
  - "What did you say?", "What did you say?", …
- Option 2: add enough checksum to recover
- **Option 3: when garbled ACK or NAK, retransmit**

Problem: can't just retransmit: new data or retransmission? 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

rdt2.1: sender, handles garbled ACK/NAKs



Stop and wait

Sender sends one packet, then waits for receiver response

Two sequence number would be sufficient!

rdt2.1: receiver, handles garbled ACK/NAKs

receiver

rdt_rcv(rcvpkt) && notcorrupt(rcvpkt)
&& has_seq0(rcvpkt)

---

extract(rcvpkt,data)
deliver_data(data)
sndpkt = make_pkt(ACK, chksum)
udt_send(sndpkt)

No sequence number

rdt_rcv(rcvpkt) && (corrupt(rcvpkt)

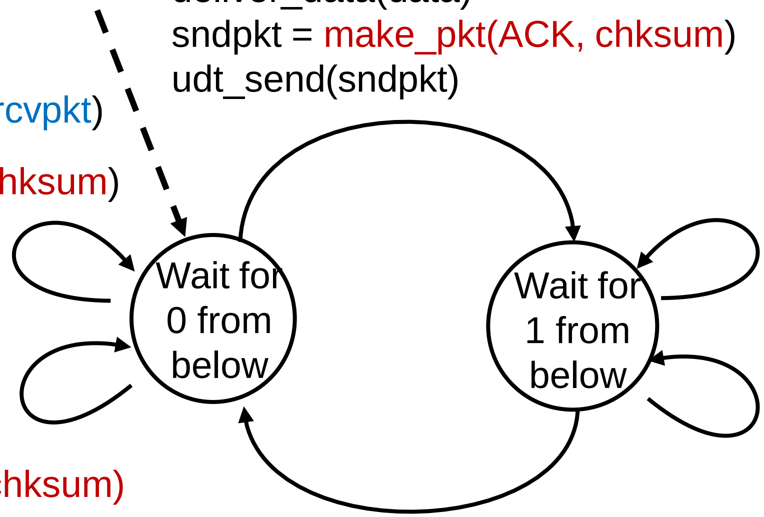
---

sndpkt = make_pkt(NAK, chksum)
udt_send(sndpkt)

rdt_rcv(rcvpkt) &&
not corrupt(rcvpkt) &&
has_seq1(rcvpkt)

---

sndpkt = make_pkt(ACK, chksum)
udt_send(sndpkt)



rdt_rcv(rcvpkt) && (corrupt(rcvpkt)

---

sndpkt = make_pkt(NAK, chksum)
udt_send(sndpkt)

rdt_rcv(rcvpkt) &&
not corrupt(rcvpkt) &&
has_seq0(rcvpkt)

---

sndpkt = make_pkt(ACK, chksum)
udt_send(sndpkt)

rdt_rcv(rcvpkt) && notcorrupt(rcvpkt)
&& has_seq1(rcvpkt)

---

extract(rcvpkt,data)
deliver_data(data)
sndpkt = make_pkt(ACK, chksum)
udt_send(sndpkt)

Saying seq1 has been received, please send the next one

rdt2.1: discussion

sender:

- seq # added to pkt
- 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:

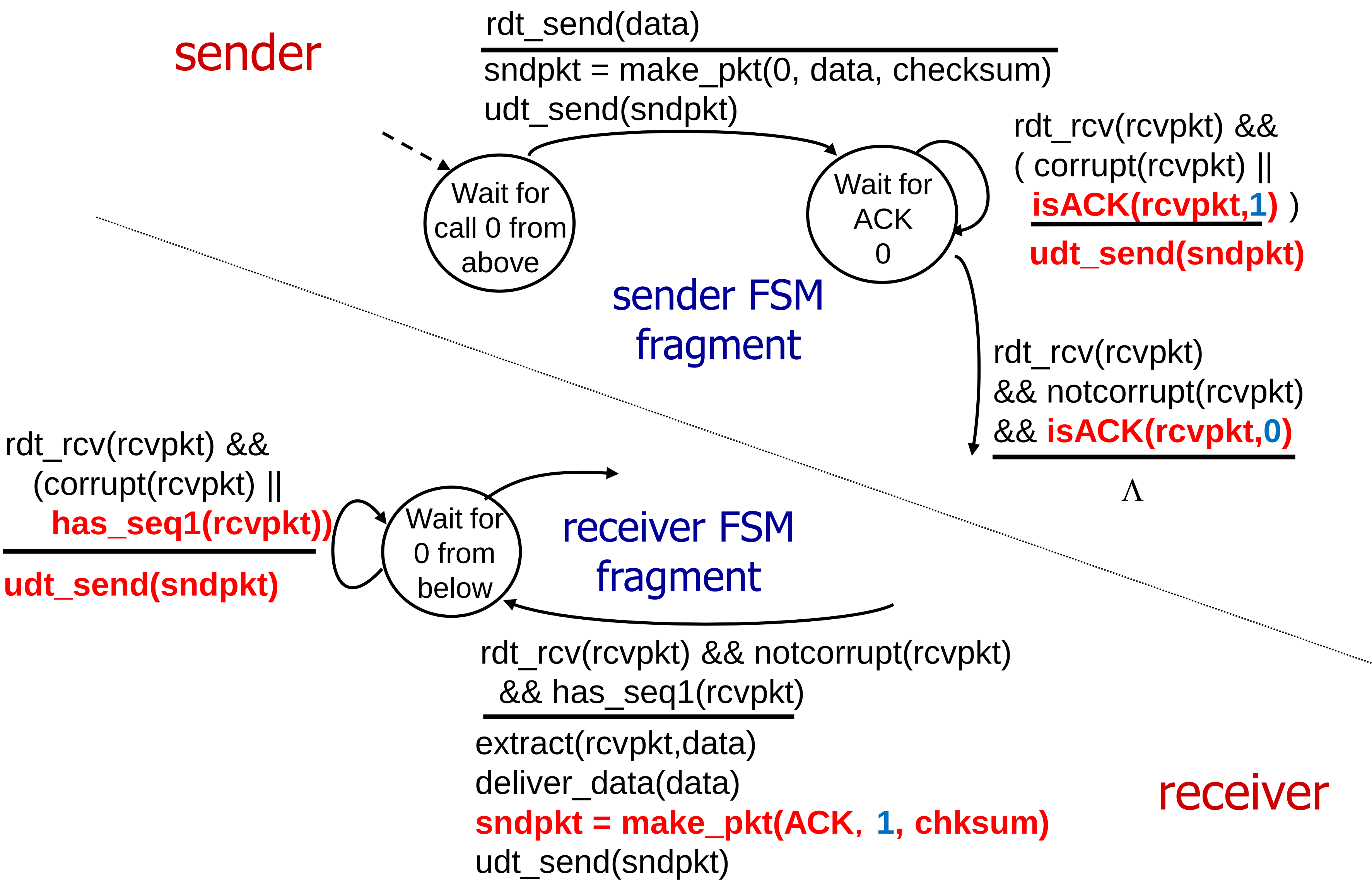
- must check if received packet is duplicate
  - state indicates whether 0 or 1 is expected pkt seq #
- note: receiver can *not* know if its last ACK/NAK received OK at sender

Two seq. #'s (0,1) will suffice. Why?

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)

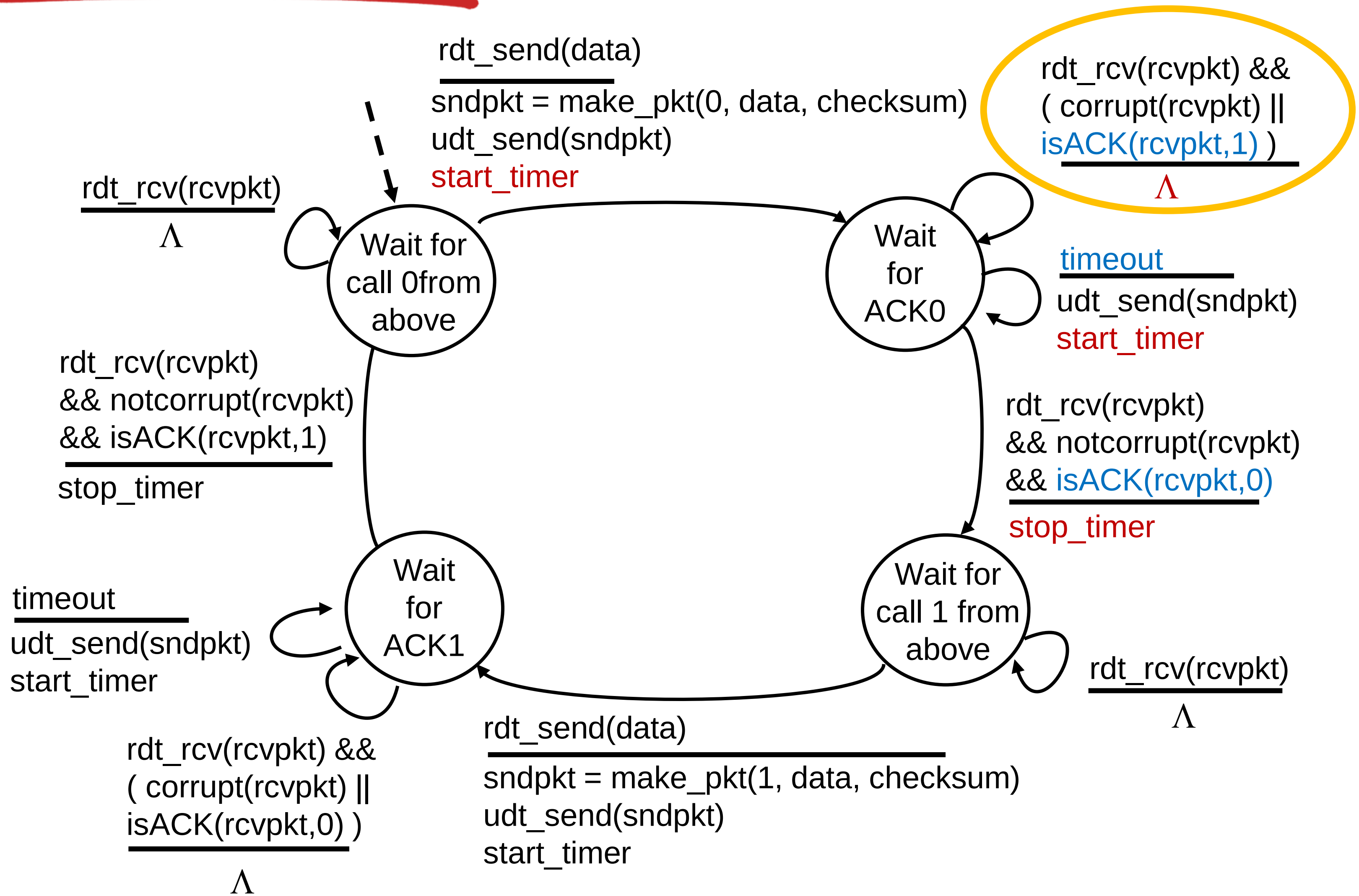
- checksum, seq. #, ACKs, retransmissions will be of help … but not enough

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

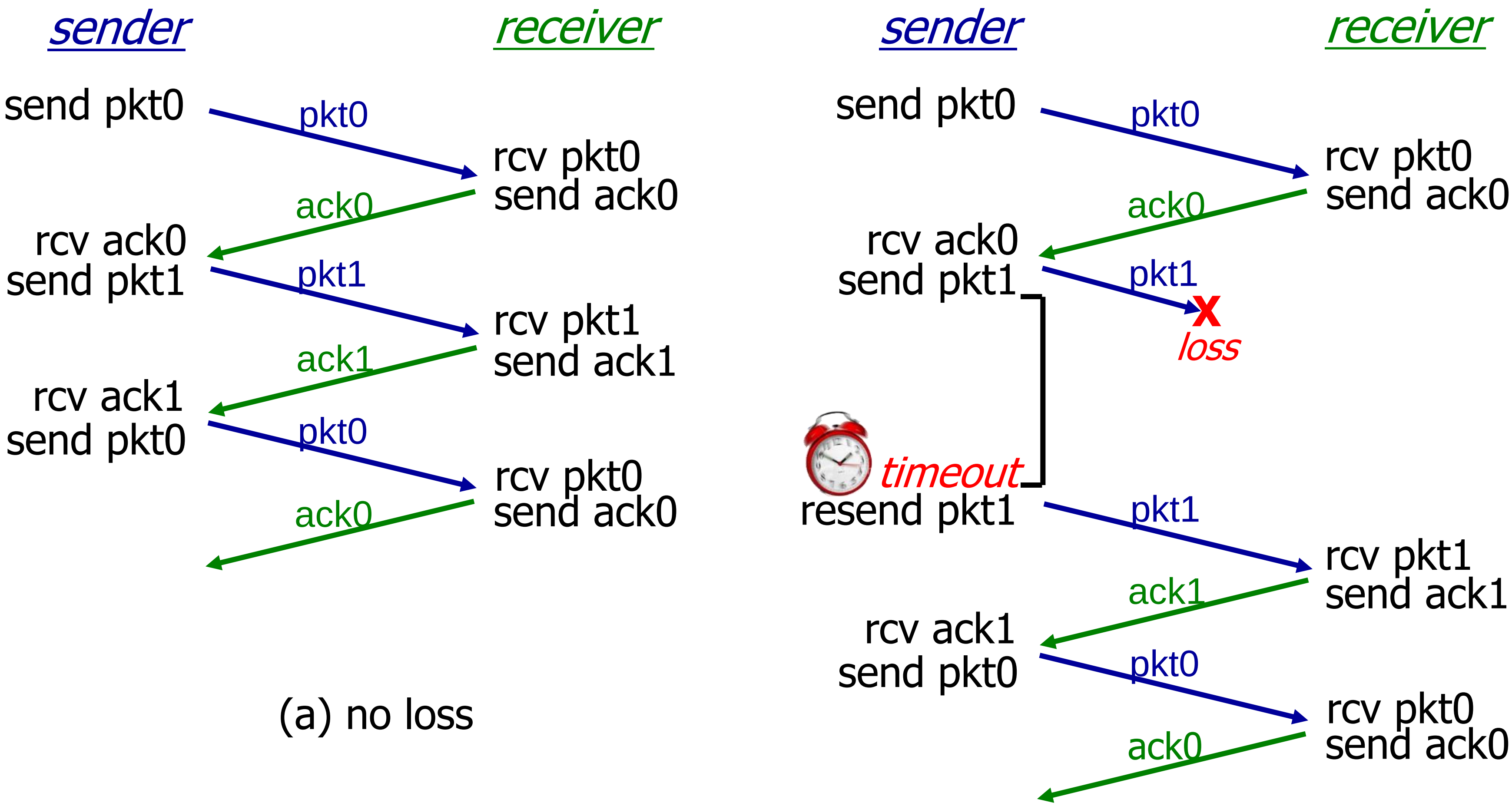
- retransmits if no ACK received in this time
- if pkt (or ACK) just delayed (not lost):
  - retransmission will be duplicate, but seq. #'s already handles this
  - receiver must specify seq # of pkt being ACKed
- requires countdown timer
  - start timer, timer interrupt, stop timer

How long should the sender wait?

rdt3.0 sender



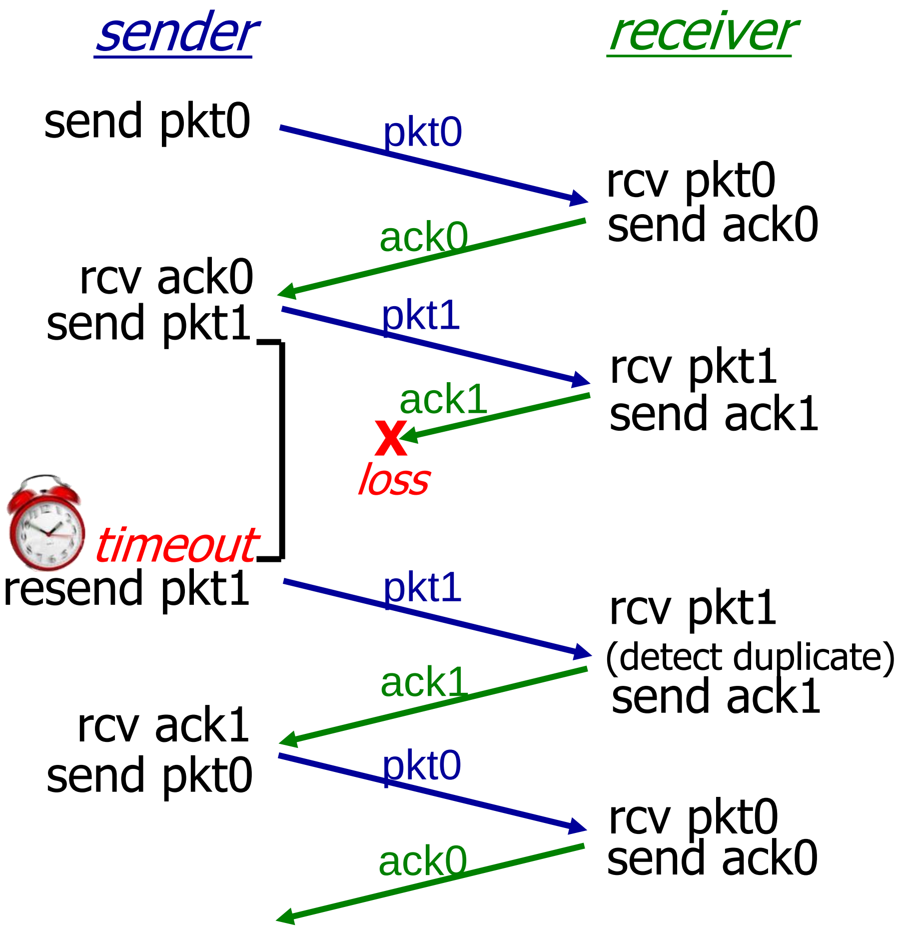
rdt3.0 in action



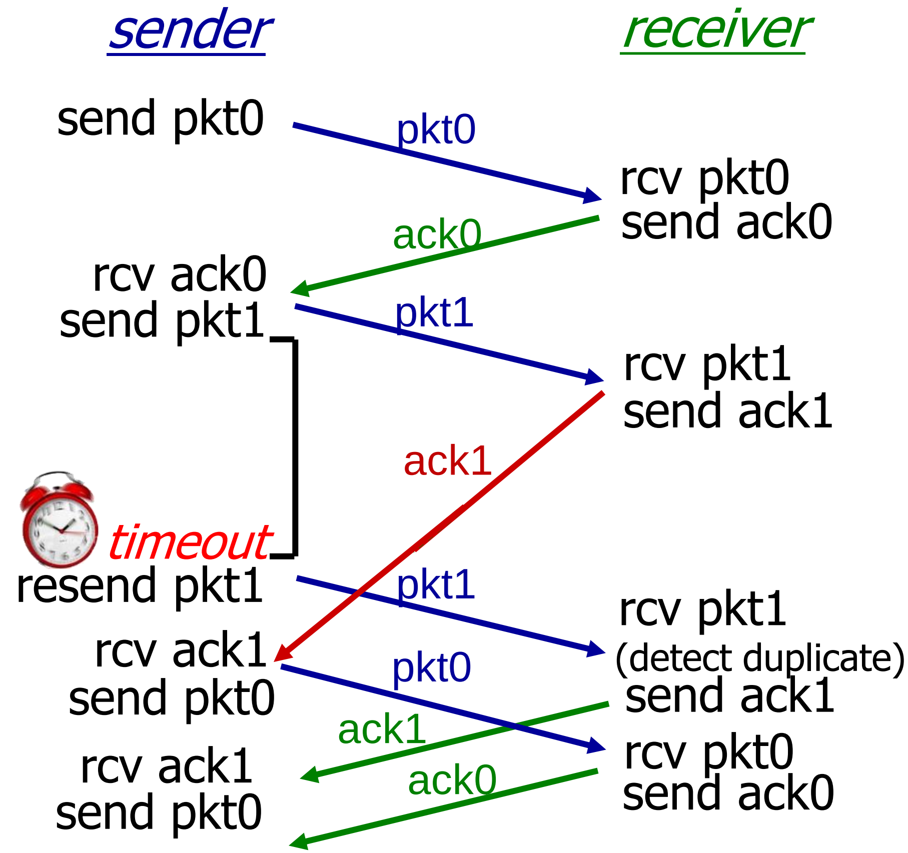
(a) no loss

(b) packet loss

rdt3.0 in action



(c) ACK loss



(d) premature timeout/ delayed ACK

Summary

Roadmap:

- Perfectly reliable channel: rdt1.0
- Channel with bit error:
  - bit error in packet: rdt 2.0
  - bit error in ACK: 2.1
  - NAK-free: 2.2
- Lossy channel: rdt 3.0

Summary of Techniques

- Checksum
- Sequence number
- ACK packets
- Retransmission
- Timeout

Next Lecture

3.1 transport-layer services
3.2 multiplexing and demultiplexing
3.3 connectionless transport: UDP
3.4 principles of reliable data transfer (continue)
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