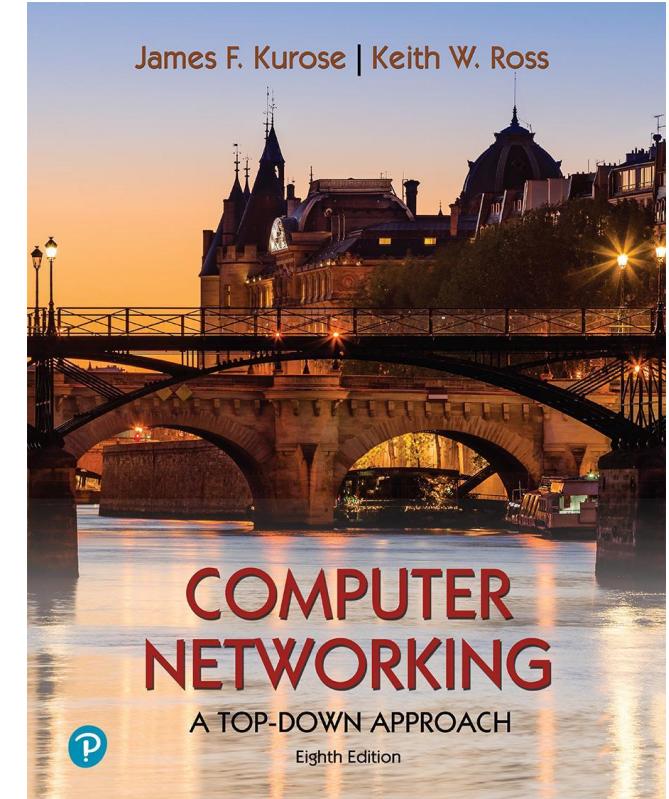


Chapter 3

Transport Layer



*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

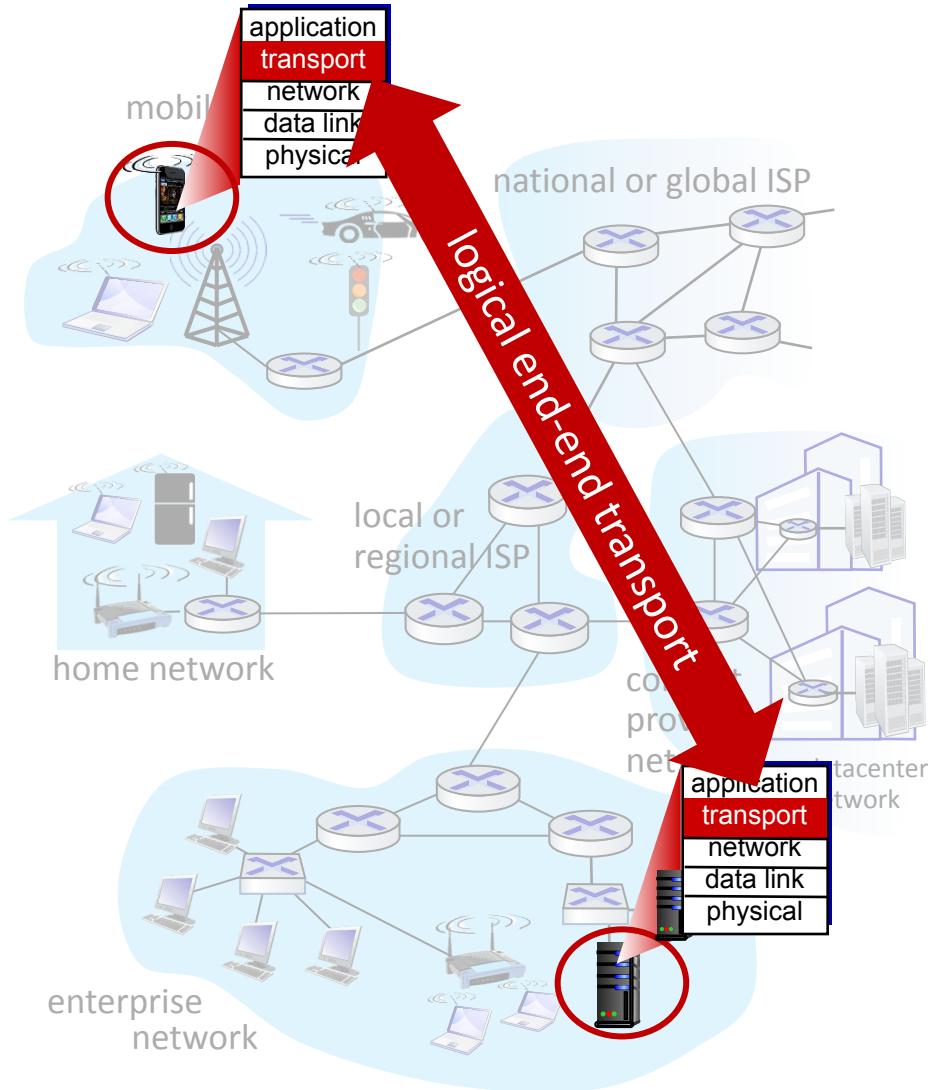
Transport layer: roadmap

- 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



household analogy:

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

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

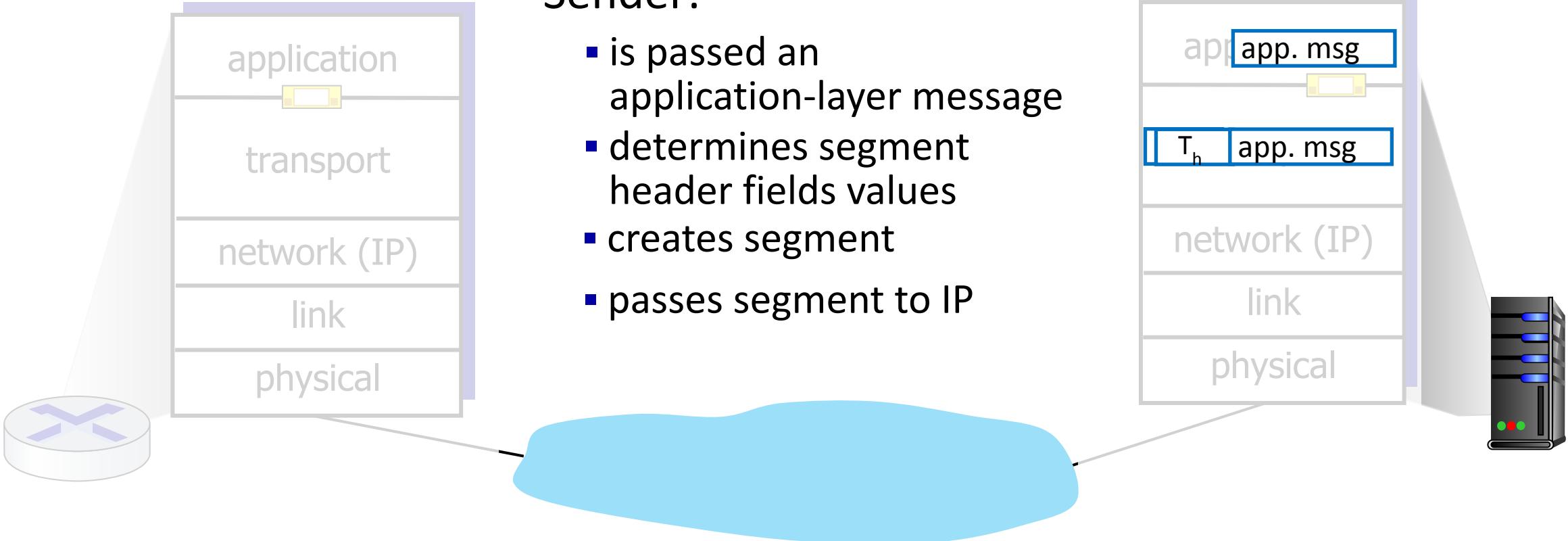
Transport vs. network layer services and protocols

- **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

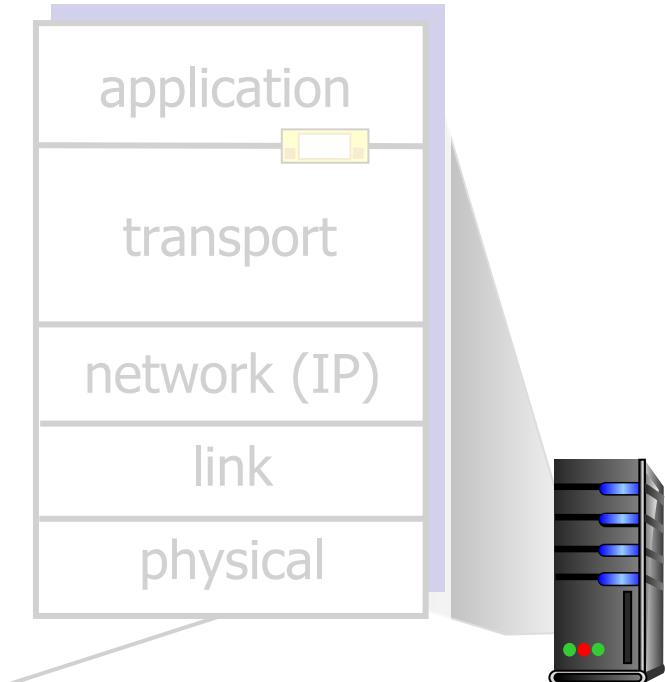
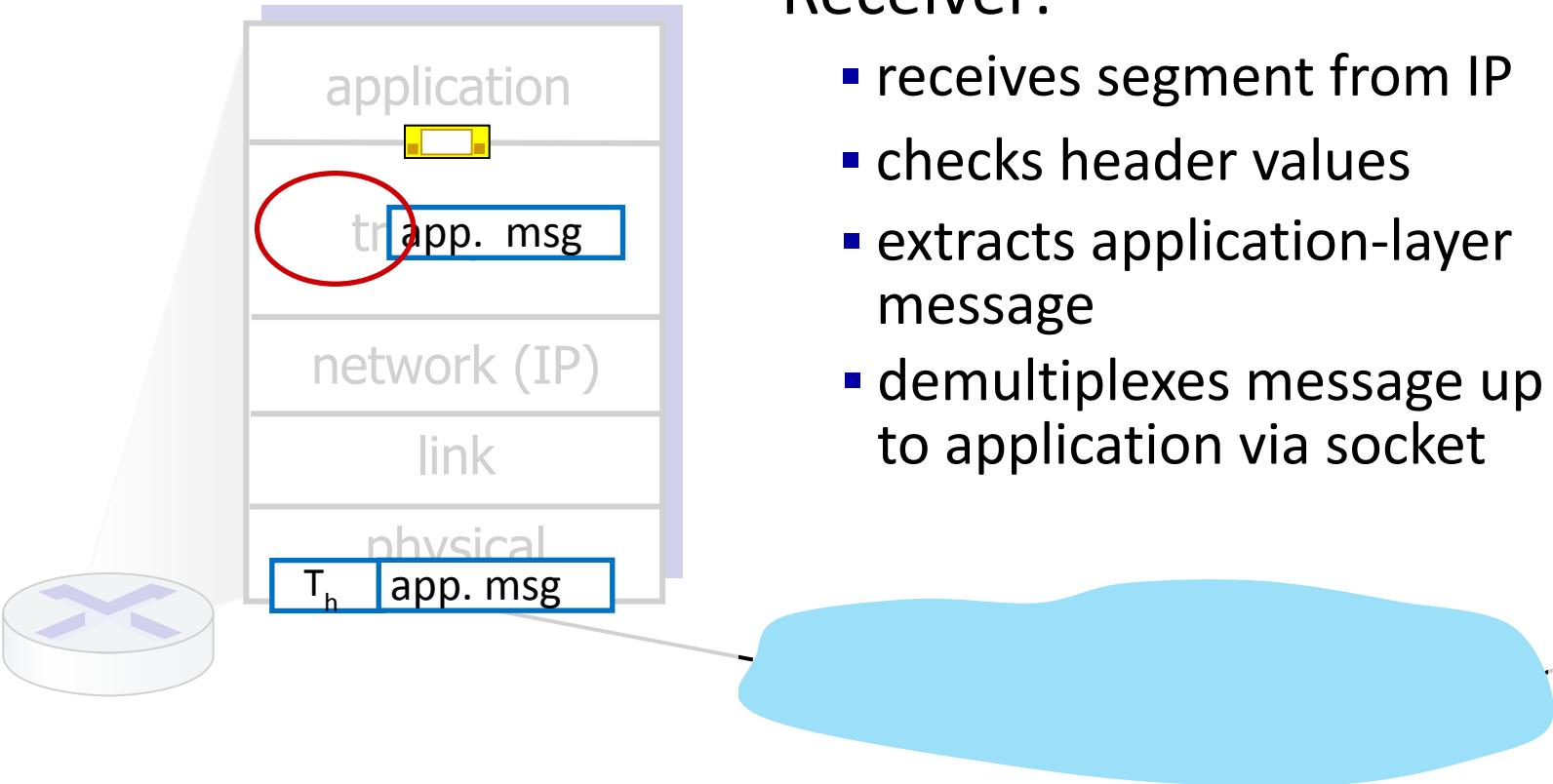
Transport Layer Actions



Transport Layer Actions

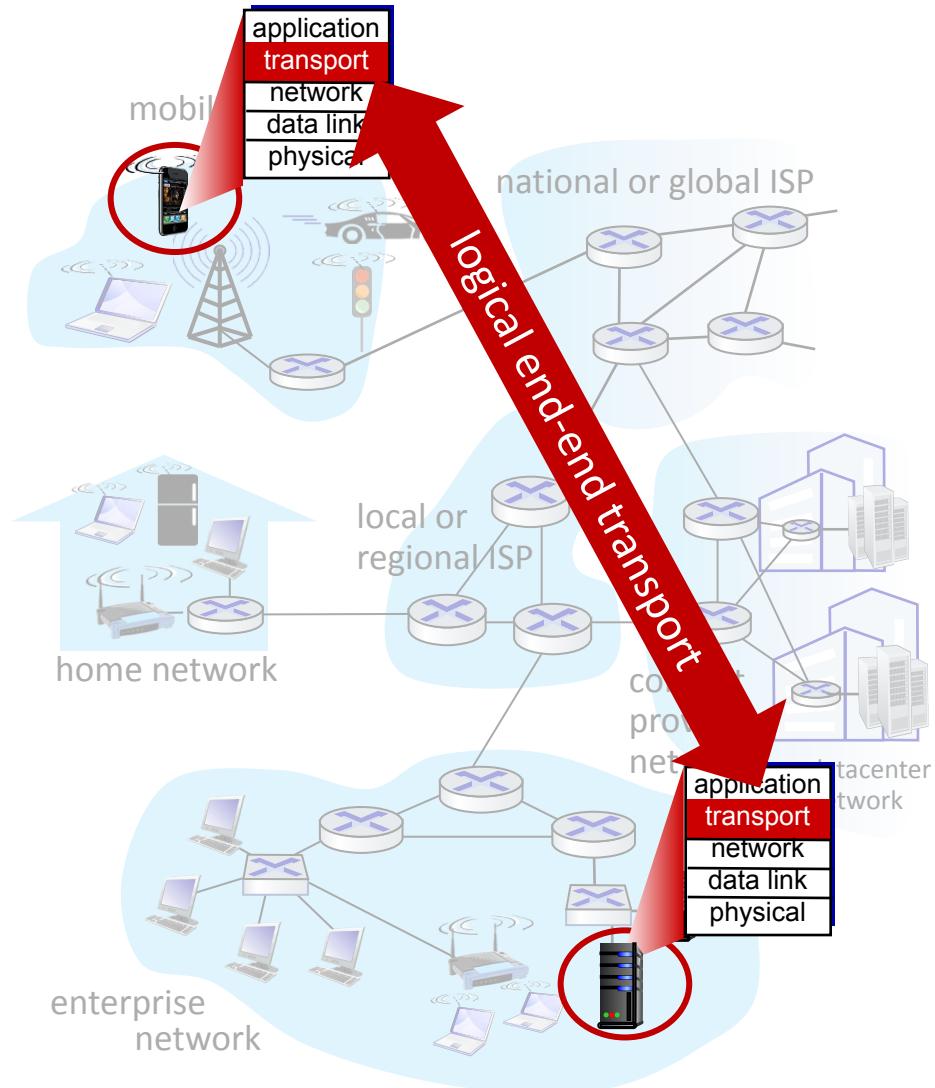
Receiver:

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



Two principal Internet transport protocols

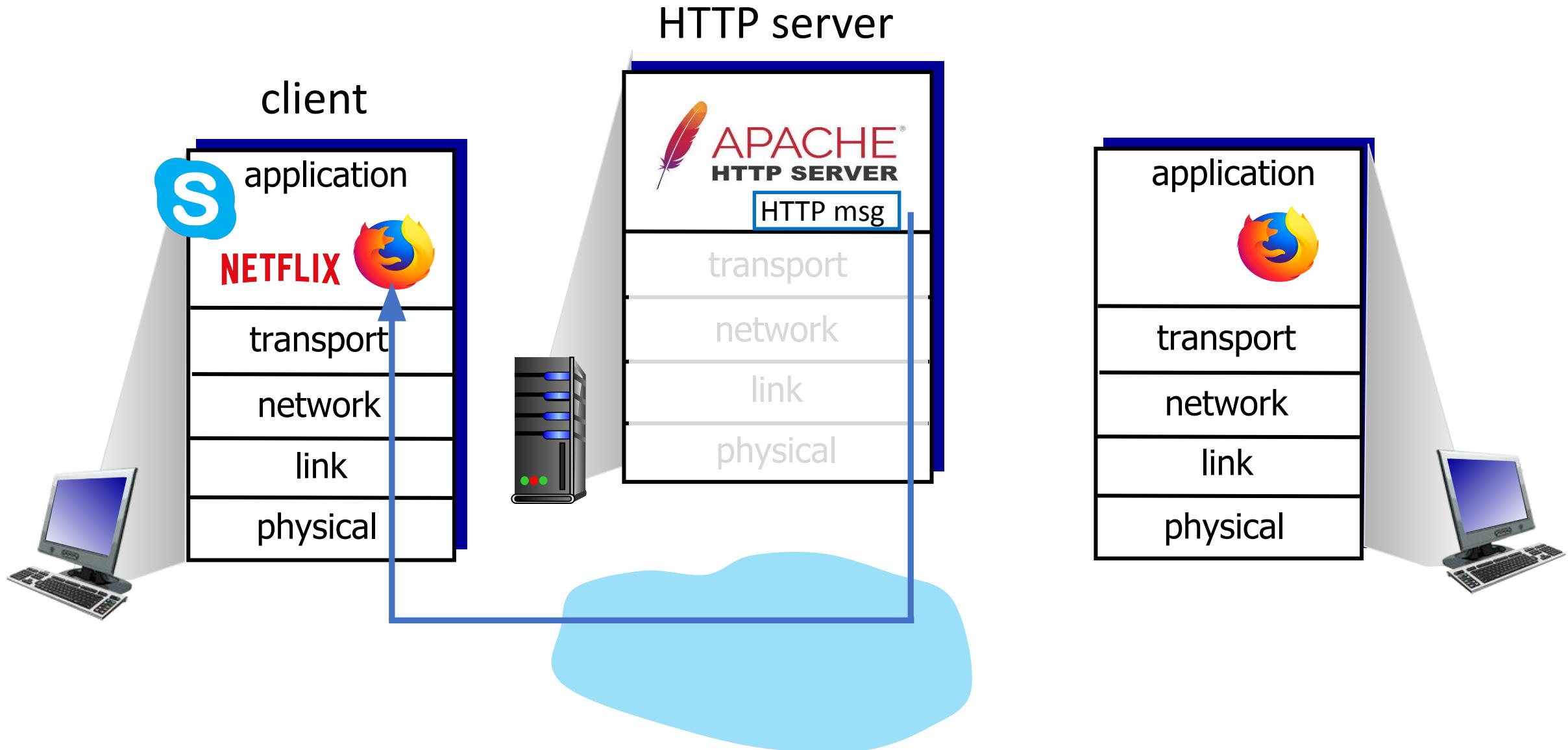
- **TCP:** Transmission Control Protocol
 - reliable, in-order delivery
 - congestion control
 - flow control
 - connection setup
- **UDP:** User Datagram Protocol
 - unreliable, unordered delivery
 - no-frills extension of “best-effort” IP
- services not available:
 - delay guarantees
 - bandwidth guarantees

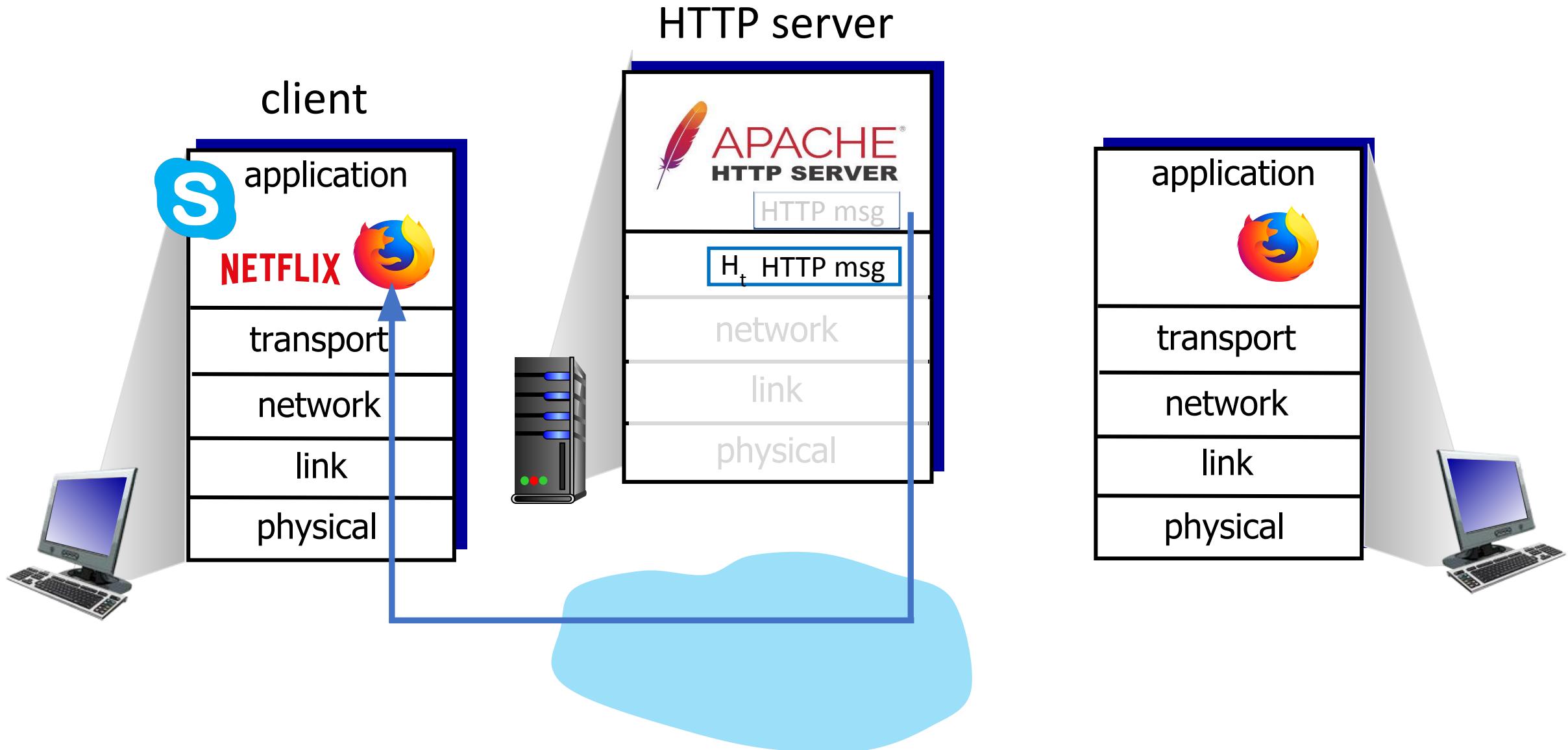


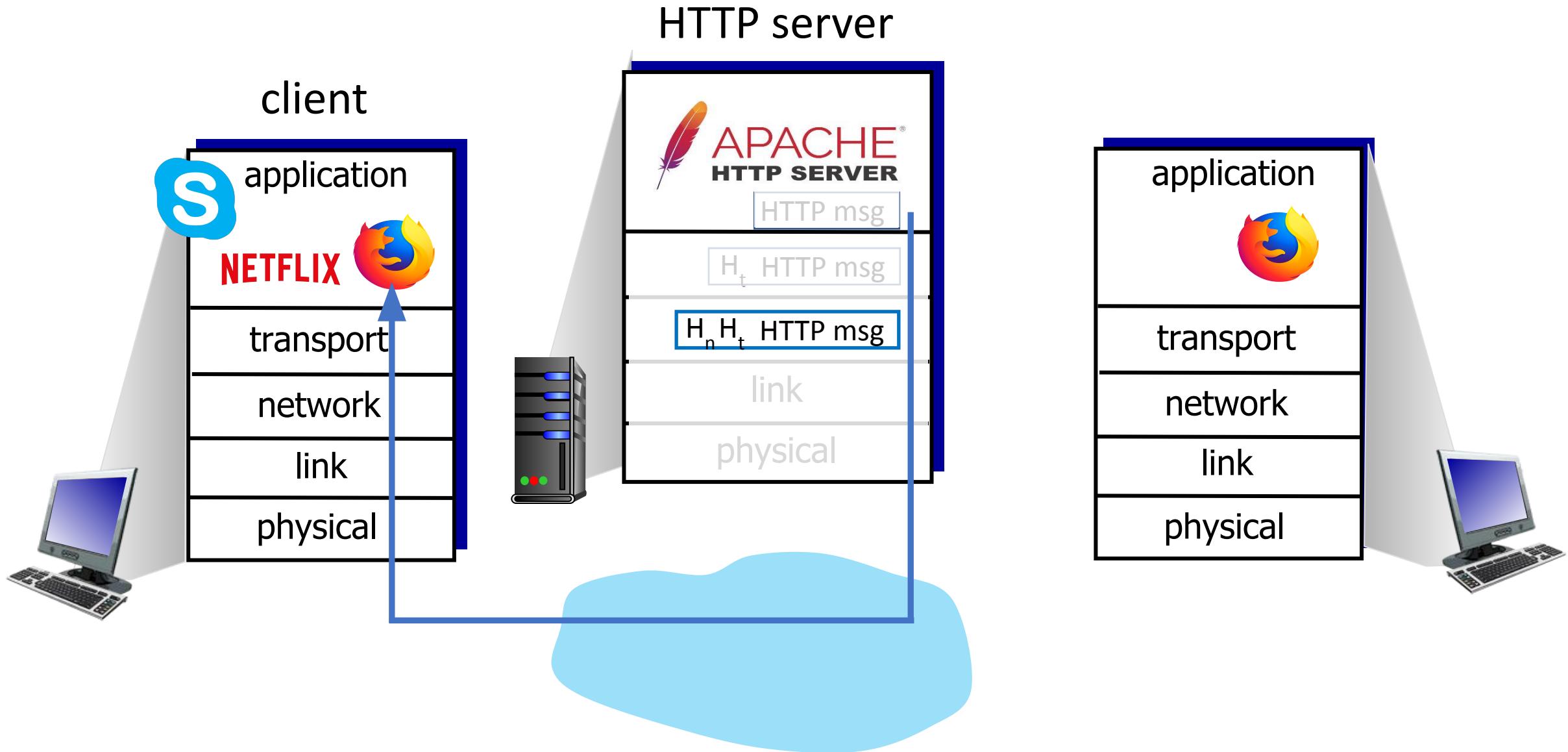
Chapter 3: roadmap

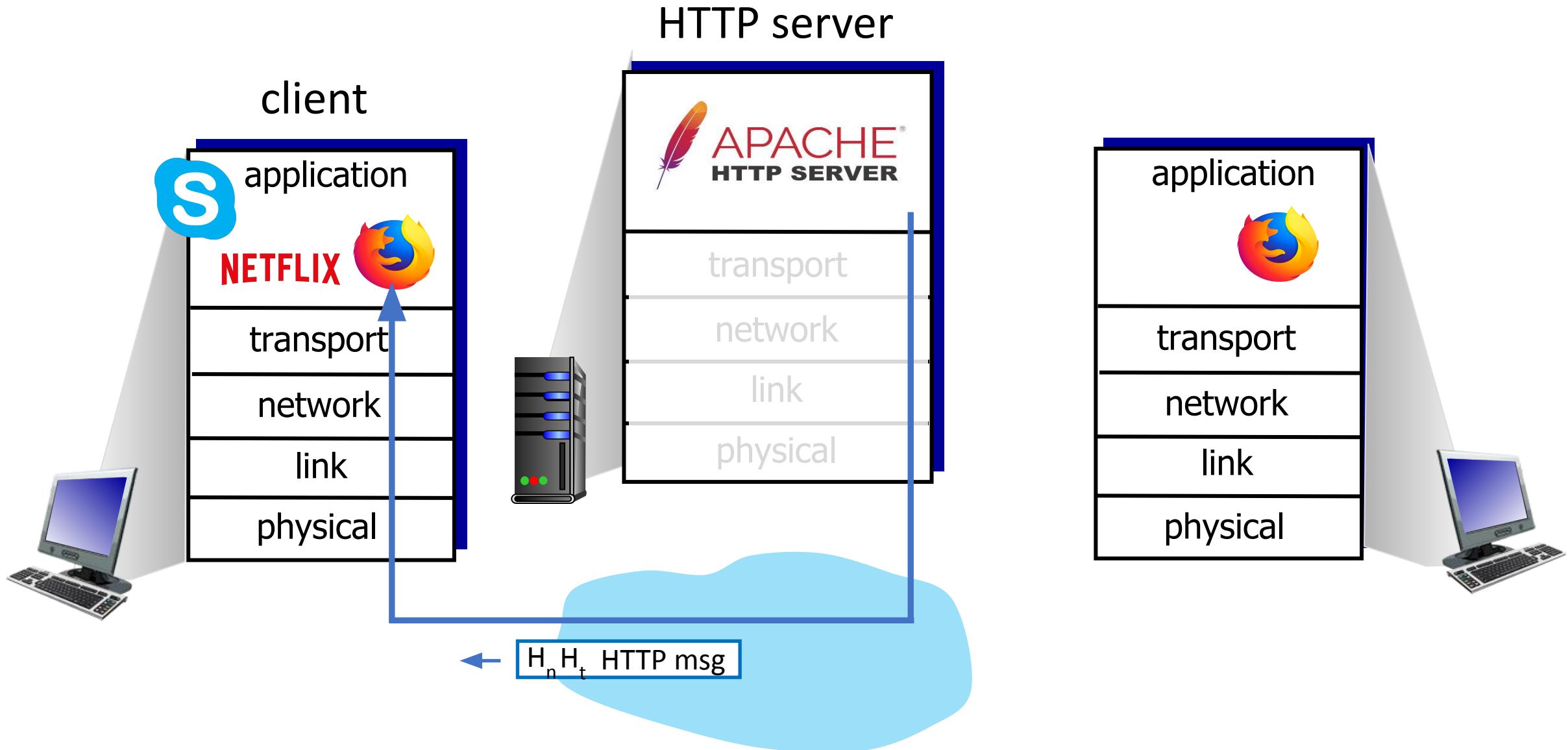
- Transport-layer services
- **Multiplexing and demultiplexing**
- Connectionless transport: UDP
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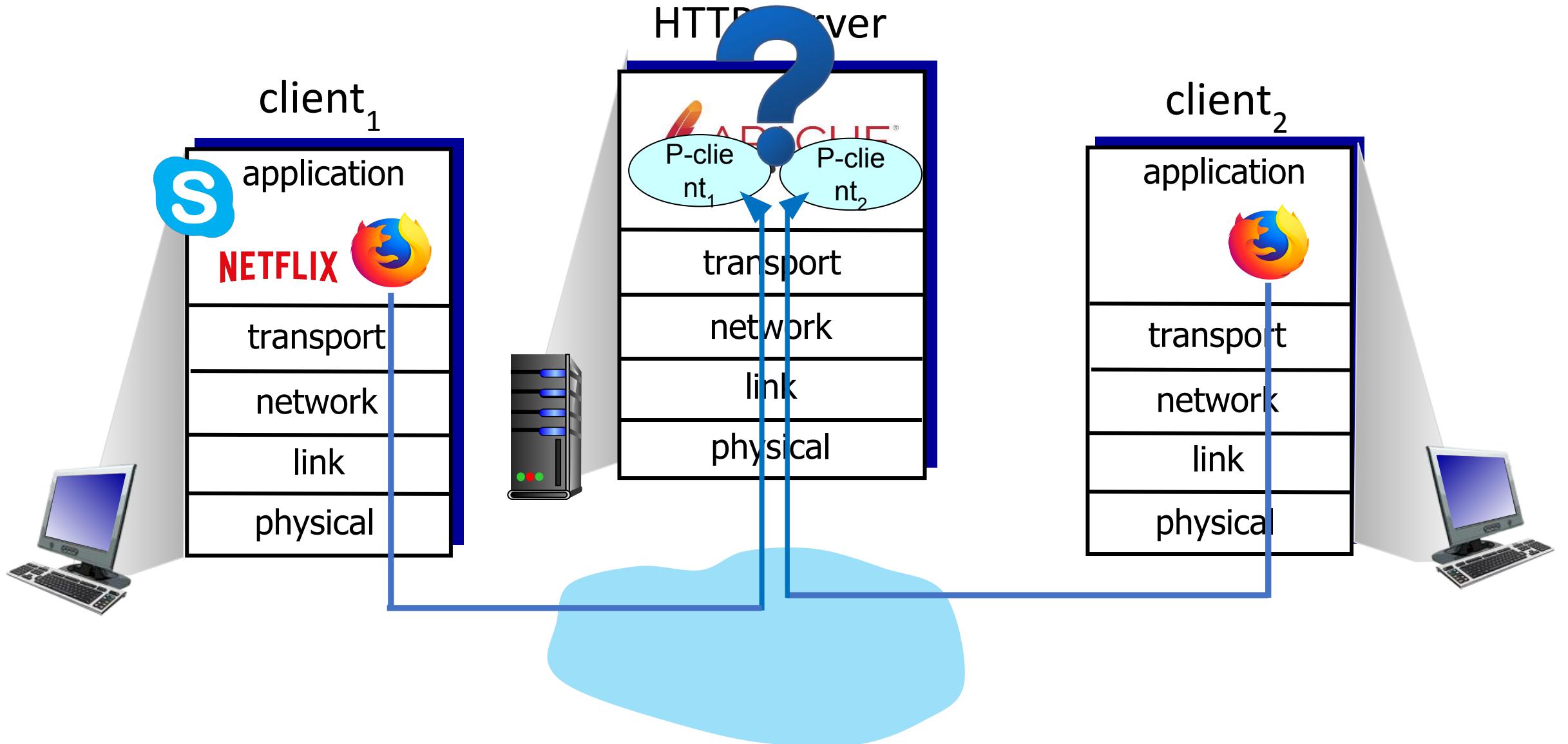












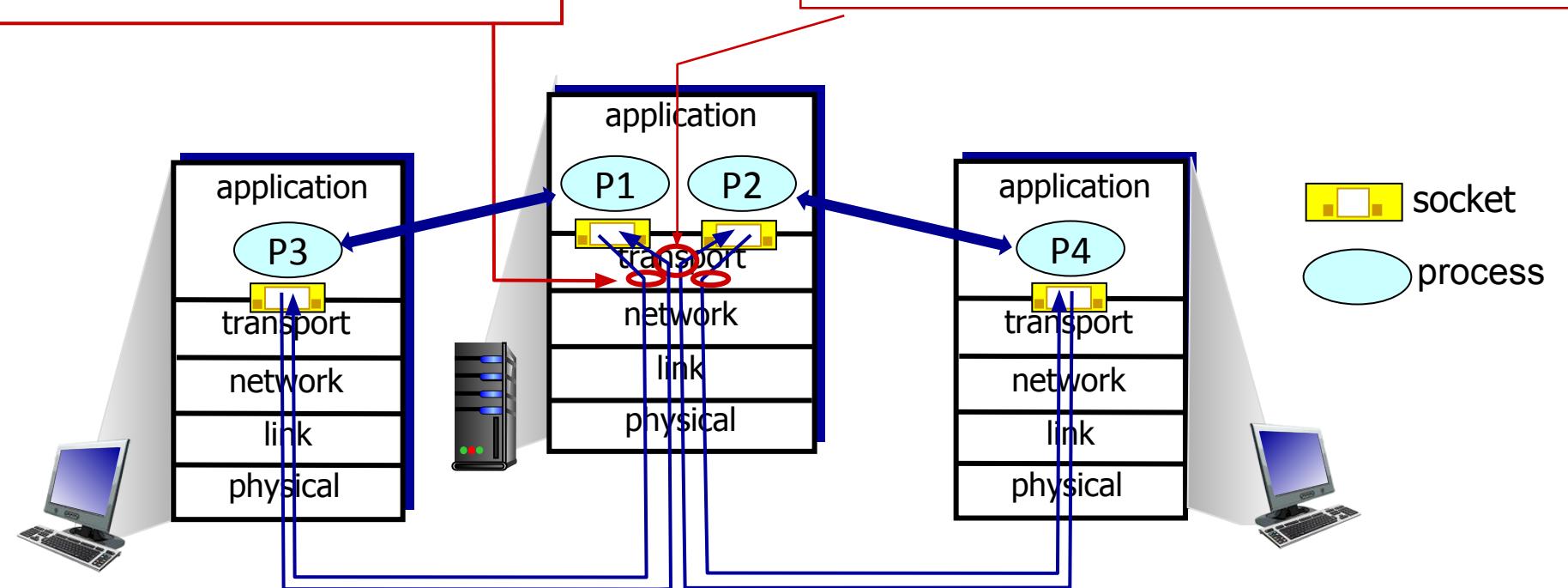
Multiplexing/demultiplexing

multiplexing at sender:

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

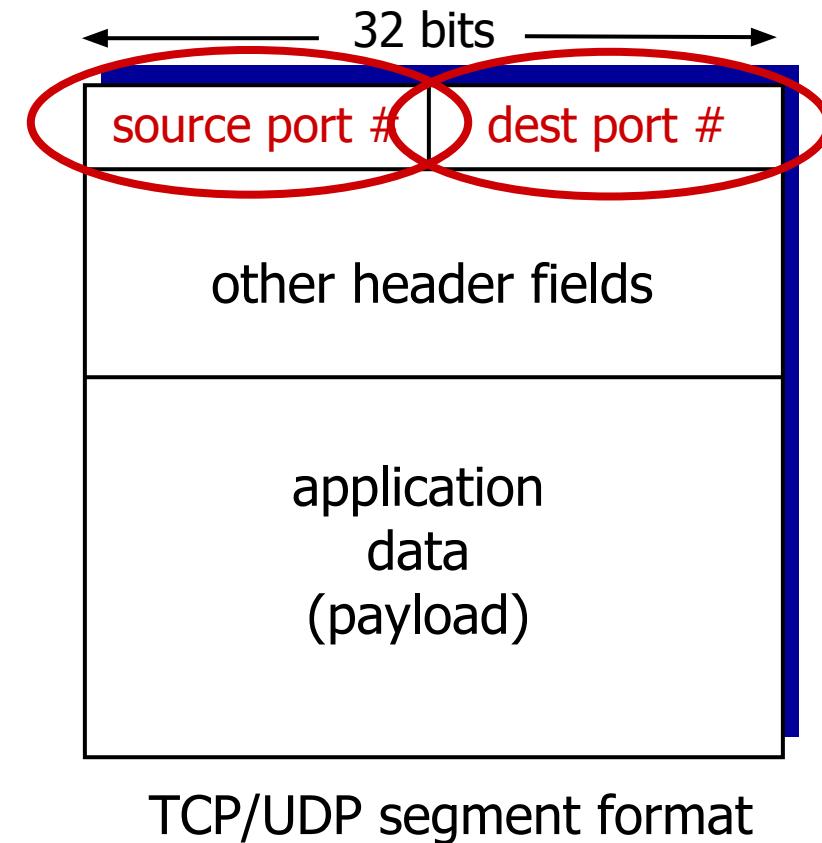
demultiplexing at receiver:

use header info to deliver received segments to correct socket



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 #:

```
DatagramSocket mySocket1  
= new  
DatagramSocket (1234);
```

- 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 #



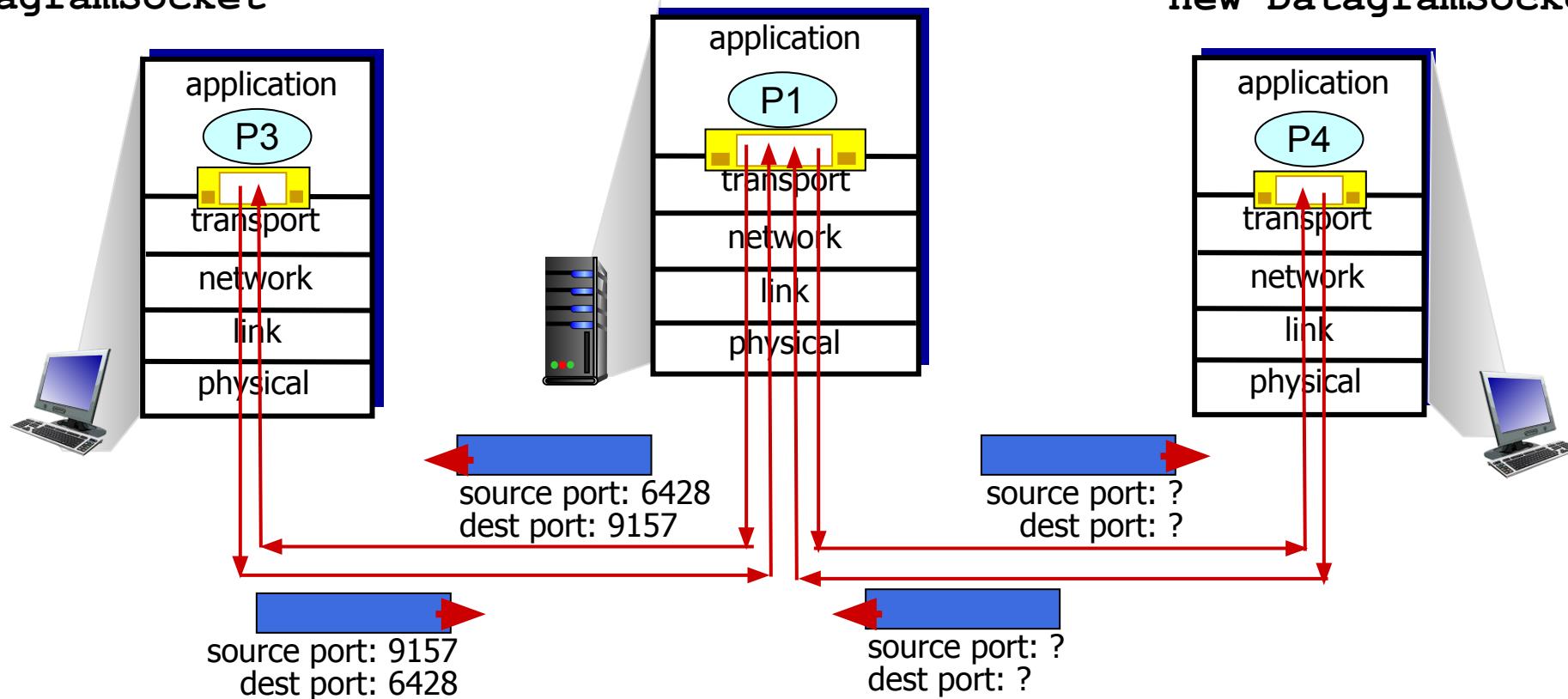
IP/UDP datagrams with *same dest. port #*, but different source IP addresses and/or source port numbers will be directed to *same socket* at receiving host

Connectionless demultiplexing: an example

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

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

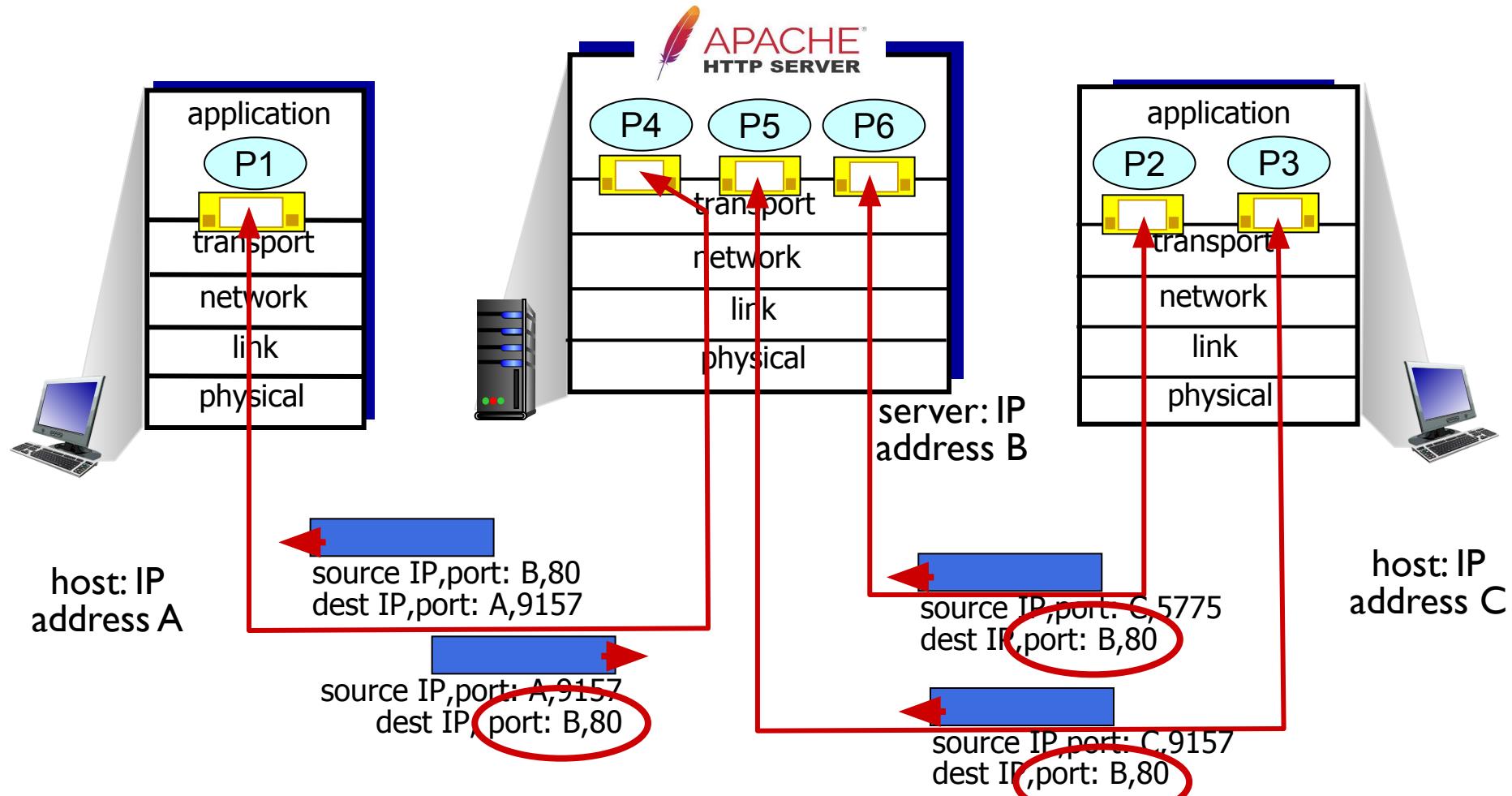
```
DatagramSocket mySocket1 =  
new DatagramSocket (5775);
```



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

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

Chapter 3: 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



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

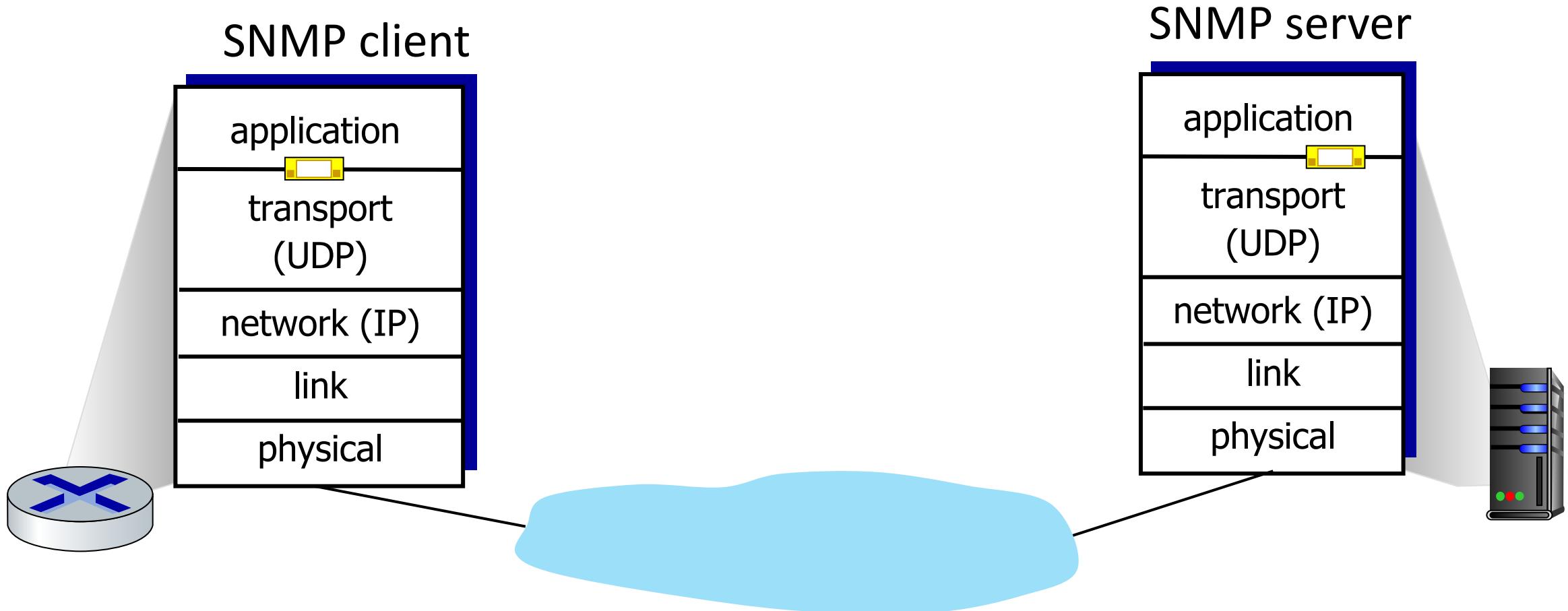
This User Datagram Protocol (UDP) is defined to make available a datagram mode of packet-switched computer communication in the environment of an interconnected set of computer networks. This protocol assumes that the Internet Protocol (IP) [1] is used as the underlying protocol.

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].

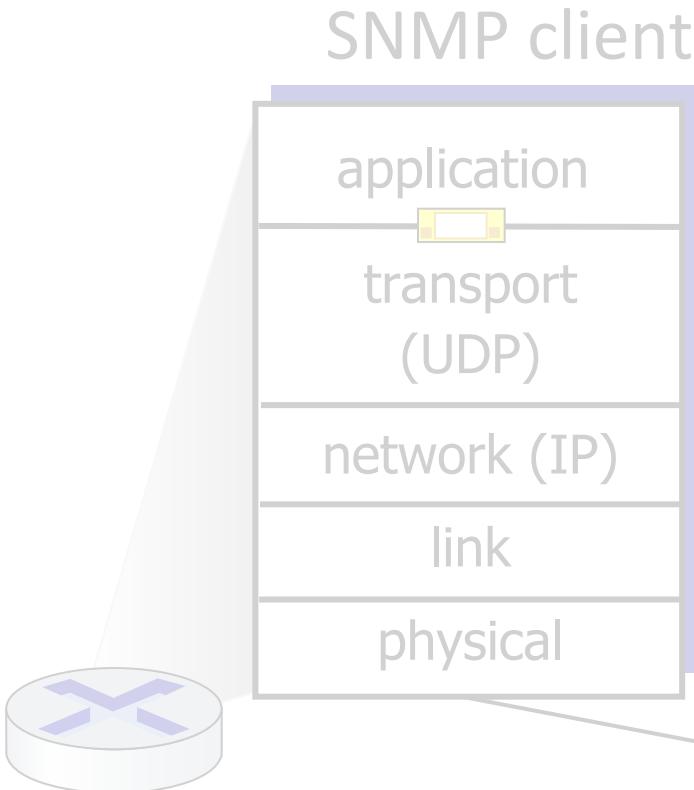
Format

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

UDP: Transport Layer Actions



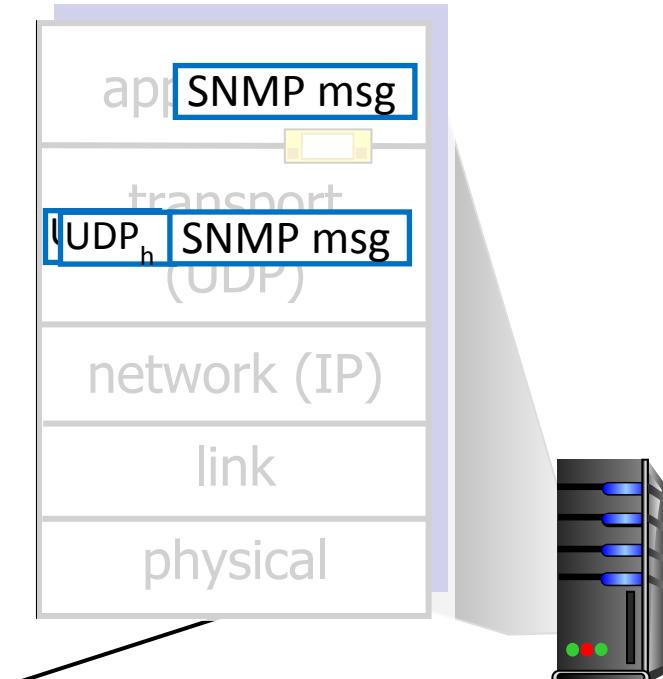
UDP: Transport Layer Actions



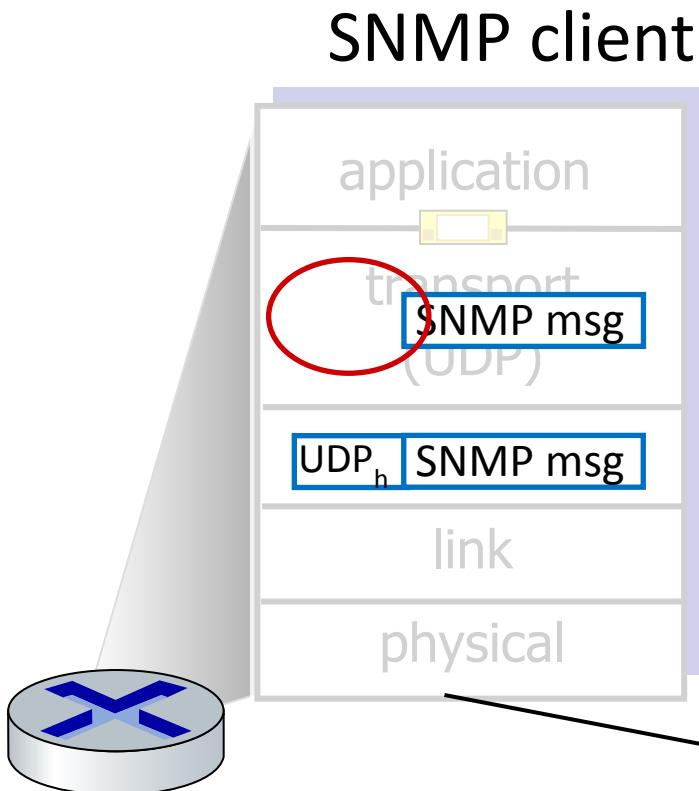
UDP sender actions:

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

SNMP server



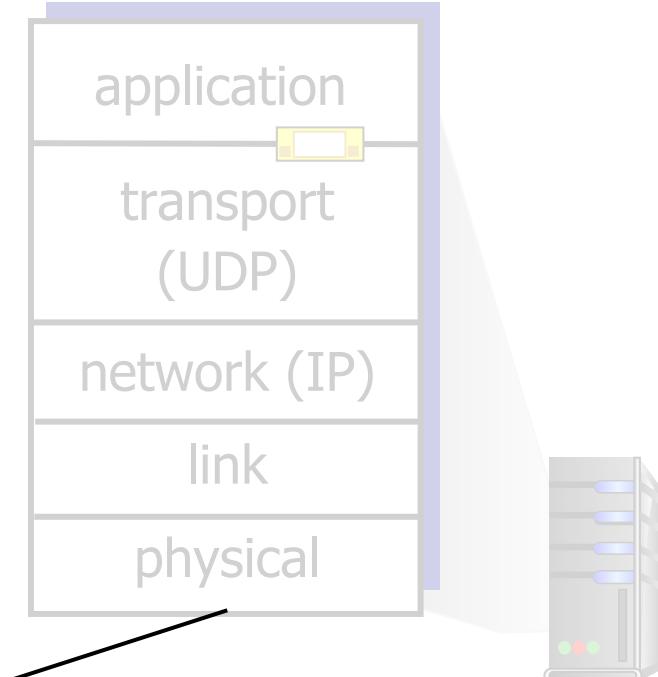
UDP: Transport Layer Actions



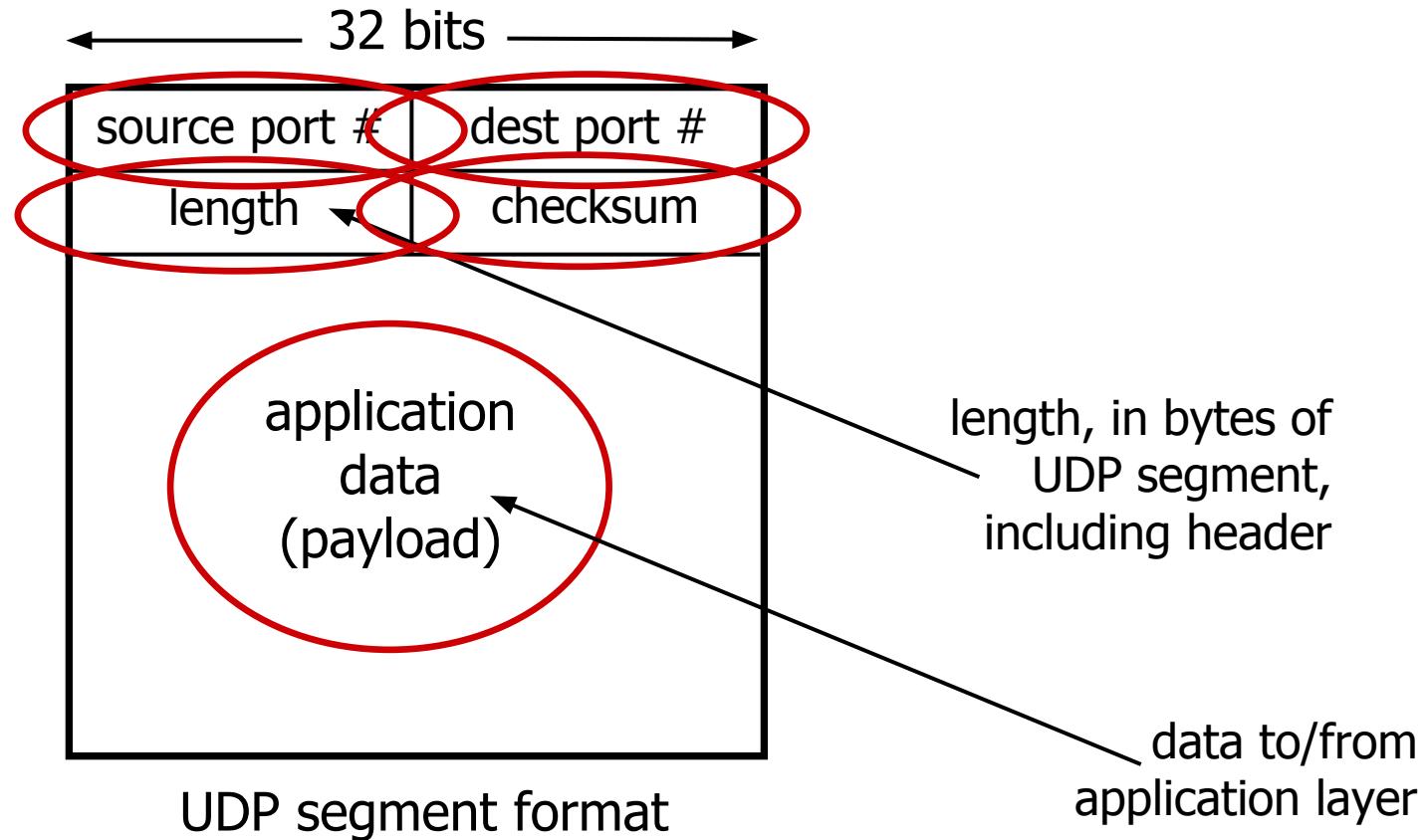
UDP receiver actions:

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

SNMP server

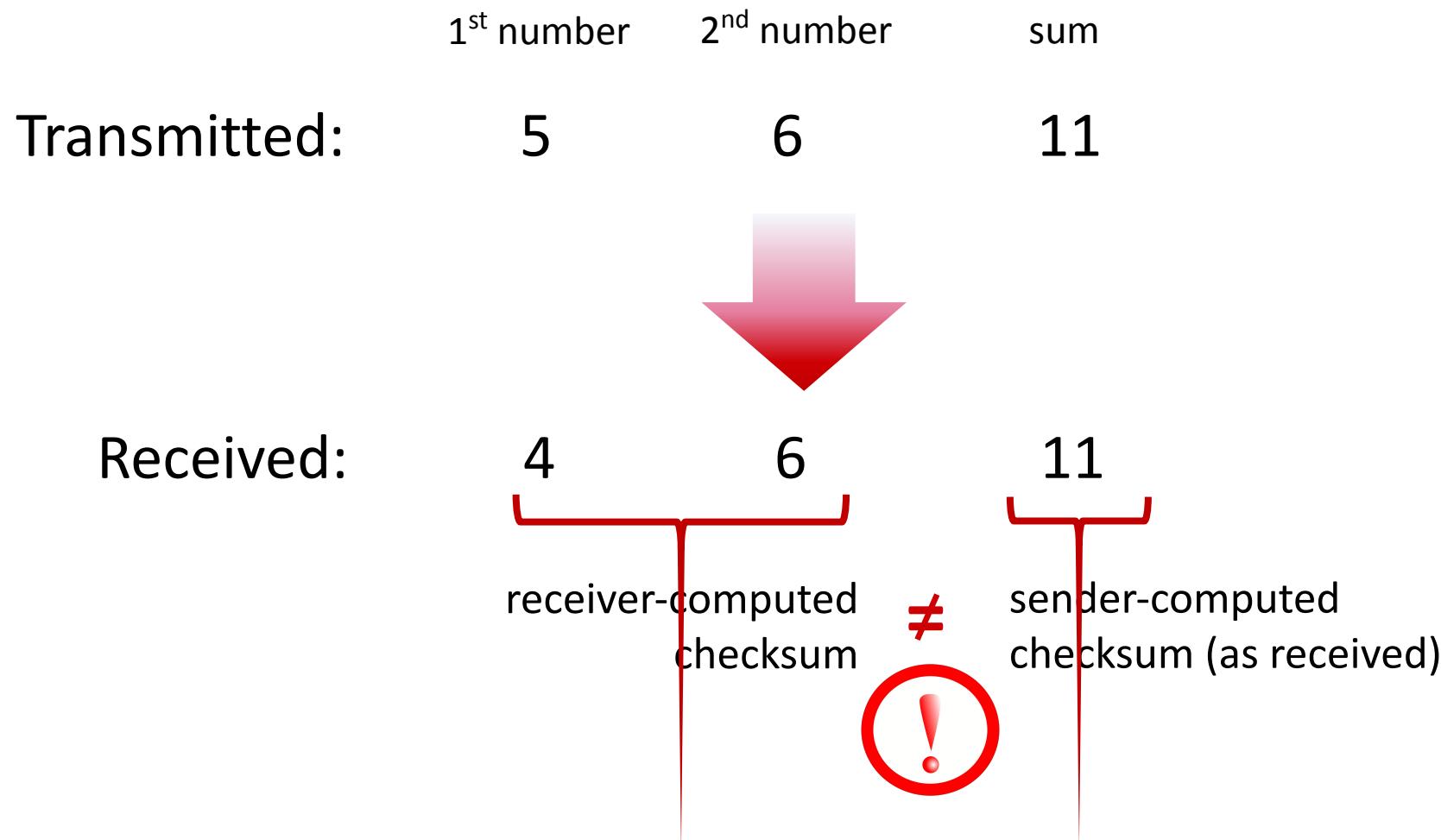


UDP segment header



UDP checksum

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



UDP 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
<hr/>																
wraparound	1	1	0	1	1	1	0	1	1	1	0	1	1	1	0	1
<hr/>																
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	0
	1	1	0	1	0	1	0	1	0	1	0	1	0	1	0	1	0
<hr/>																	
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	0
checksum	0	1	0	0	0	1	0	0	0	1	0	0	0	0	1	1	0

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

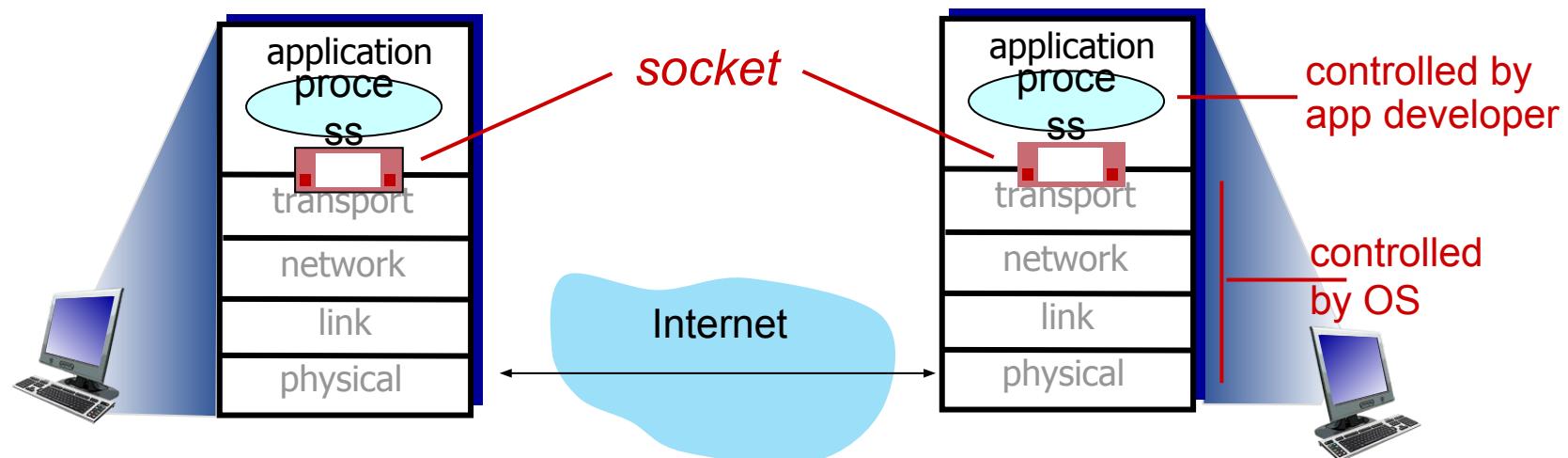
Summary: UDP

- “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)

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

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 and server:

- no handshaking before sending data
- sender explicitly attaches IP destination address and port # to each packet
- receiver extracts sender IP address and port# 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 processes

Client/server socket interaction: UDP



server (running on serverIP)

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

read datagram from
serverSocket

write reply to
serverSocket
specifying
client address,
port number



client

```
create socket:  
clientSocket =  
socket(AF_INET,SOCK_DGRAM)
```

Create datagram with serverIP address
And port=x; send datagram via
clientSocket

read datagram from
clientSocket
close
clientSocket

Example app: UDP client

Python UDPCClient

```
include Python's socket library → 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 UDP socket for server →

get user keyboard input →

attach server name, port to message; send into socket →

read reply characters from socket into string →

print out received string and close socket →

Example app: UDP server

Python UDPServer

```
from socket import *
serverPort = 12000
create UDP socket → serverSocket = socket(AF_INET, SOCK_DGRAM)
bind socket to local port number 12000 → serverSocket.bind(("", serverPort))
loop forever → print ("The server is ready to receive")
Read from UDP socket into message, getting → while True:
client's address (client IP and port) →     message, clientAddress = serverSocket.recvfrom(2048)
send upper case string back to this client →     modifiedMessage = message.decode().upper()
                                                serverSocket.sendto(modifiedMessage.encode(),
                                                               clientAddress)
```

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:

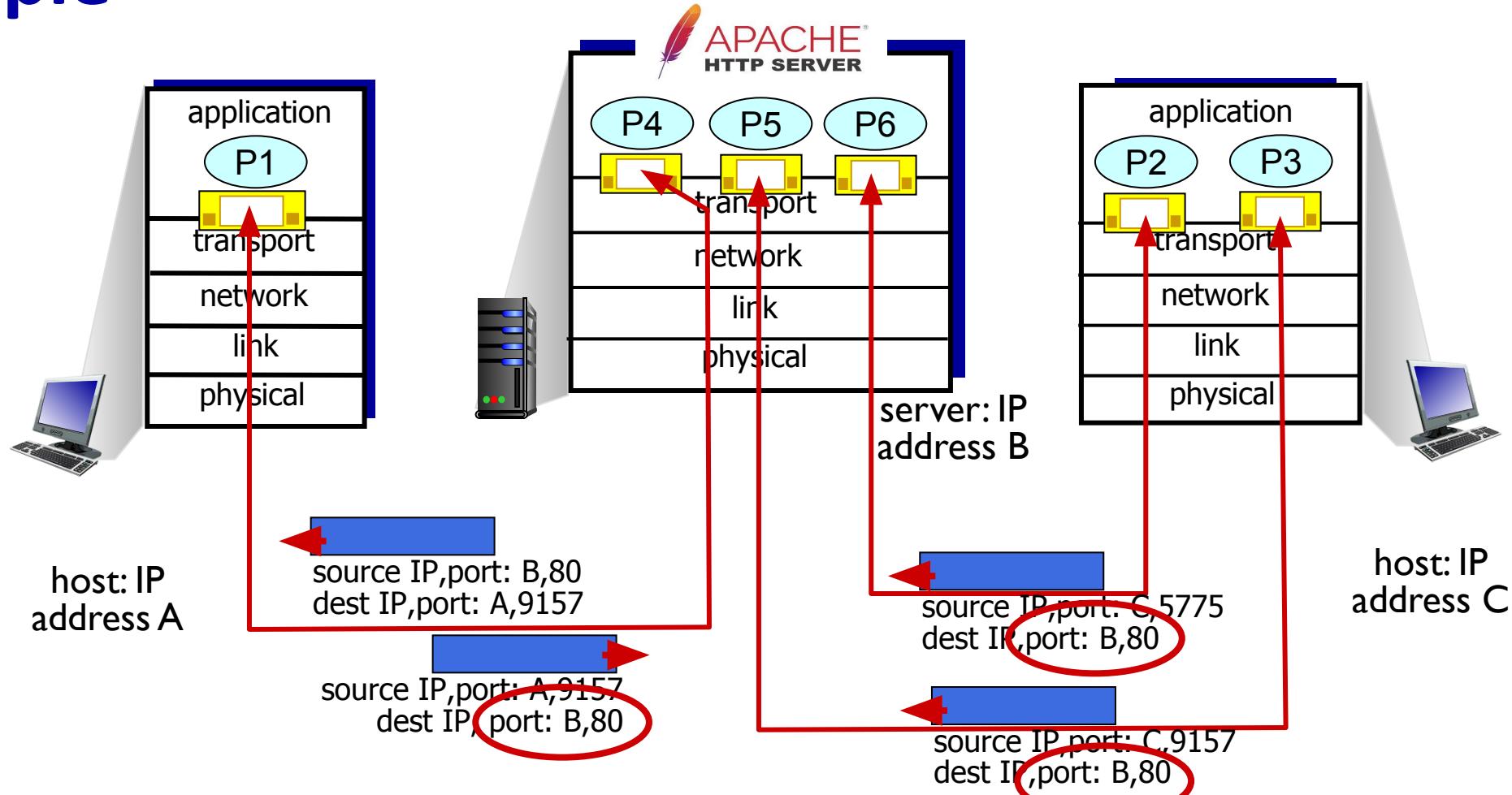
- Creating TCP socket, specifying IP address, port number of server process
- *when client creates socket*: client TCP establishes connection to server TCP

- 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)

Application viewpoint

TCP provides reliable, in-order byte-stream transfer (“pipe”) between client and server processes

(Recap) Connection-oriented demultiplexing: example



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

Client/server socket interaction: TCP



server (running on hostid)



client

create socket,
port=**x**, for incoming
request:
serverSocket = socket()

wait for incoming
connection request
connectionSocket = serverSocket.accept()

read request from
connectionSocket

write reply to
connectionSocket

close
connectionSocket

create socket,
connect to **hostid**, port=**x**
clientSocket = socket()

send request using
clientSocket

read reply from
clientSocket

close
clientSocket

TCP
connection setup

Example app: TCP client

Python TCPClient

create TCP socket for server,
remote port 12000

```
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()
```

No need to attach server name, port

→ clientSocket.send(sentence.encode())

Example app: TCP server

Python TCP Server

```
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()
```

create TCP welcoming socket → from socket import *

server begins listening for incoming TCP requests → serverPort = 12000
→ serverSocket = socket(AF_INET,SOCK_STREAM)
→ serverSocket.bind(("","serverPort"))
→ serverSocket.listen(1)

loop forever → print 'The server is ready to receive'
→ while True:

server waits on accept() for incoming requests, new socket created on return → connectionSocket, addr = serverSocket.accept()

read bytes from socket (but not address as in UDP) → sentence = connectionSocket.recv(1024).decode()
→ capitalizedSentence = sentence.upper()
→ connectionSocket.send(capitalizedSentence.
 encode())

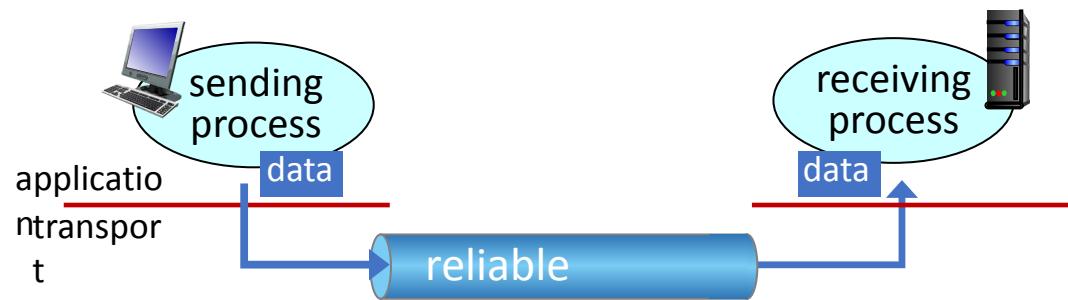
close connection to this client (but *not* welcoming socket) → connectionSocket.close()

Chapter 3: 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

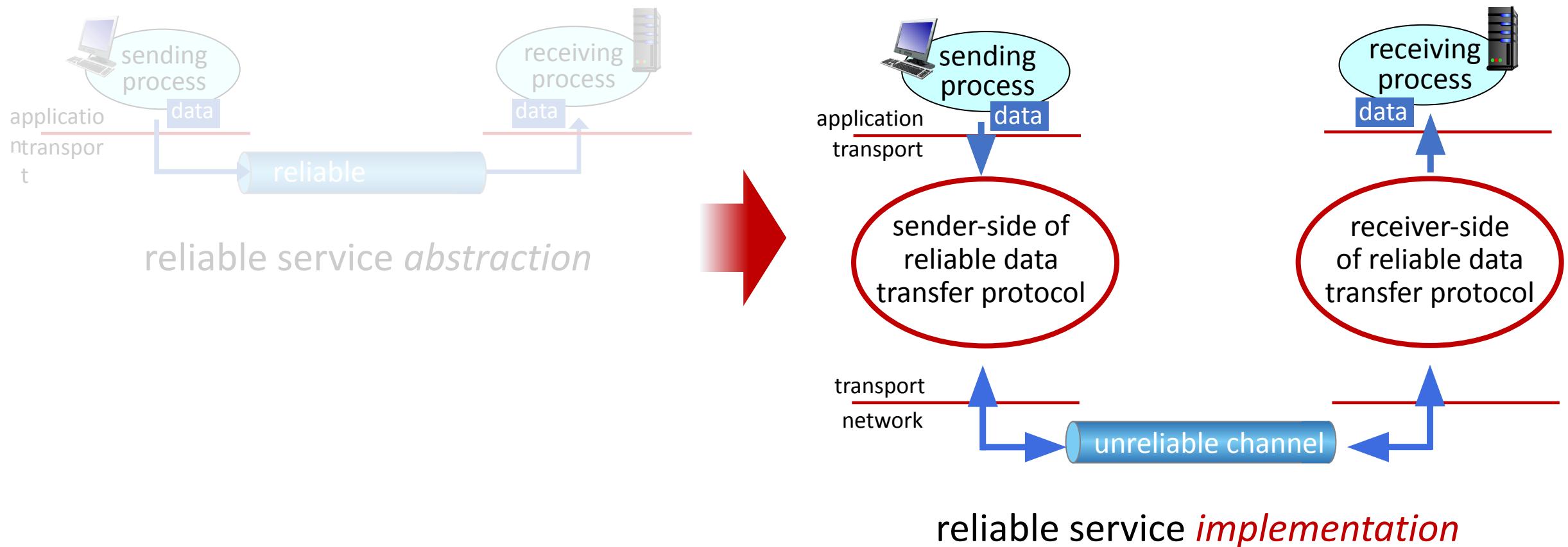


Principles of reliable data transfer



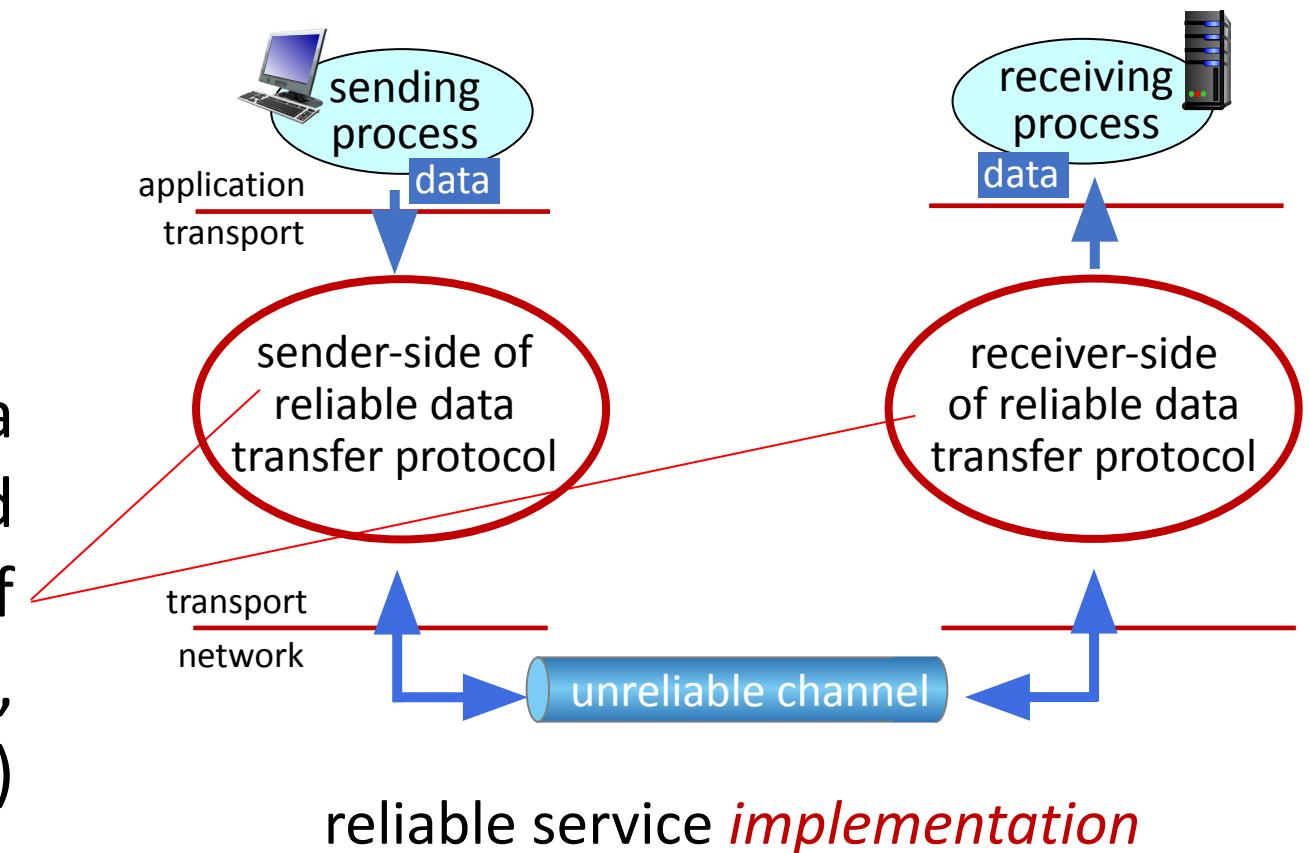
reliable service *abstraction*

Principles of reliable data transfer



Principles of reliable data transfer

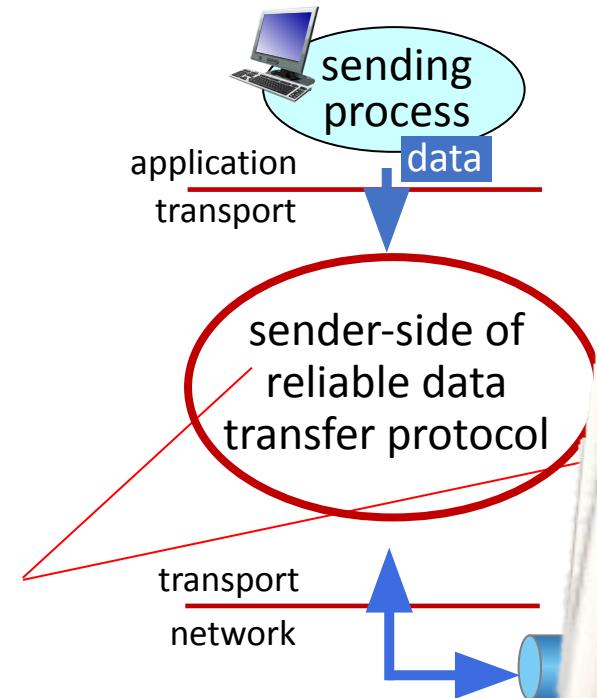
Complexity of reliable data transfer protocol will depend (strongly) on characteristics of unreliable channel (lose, corrupt, reorder data?)



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

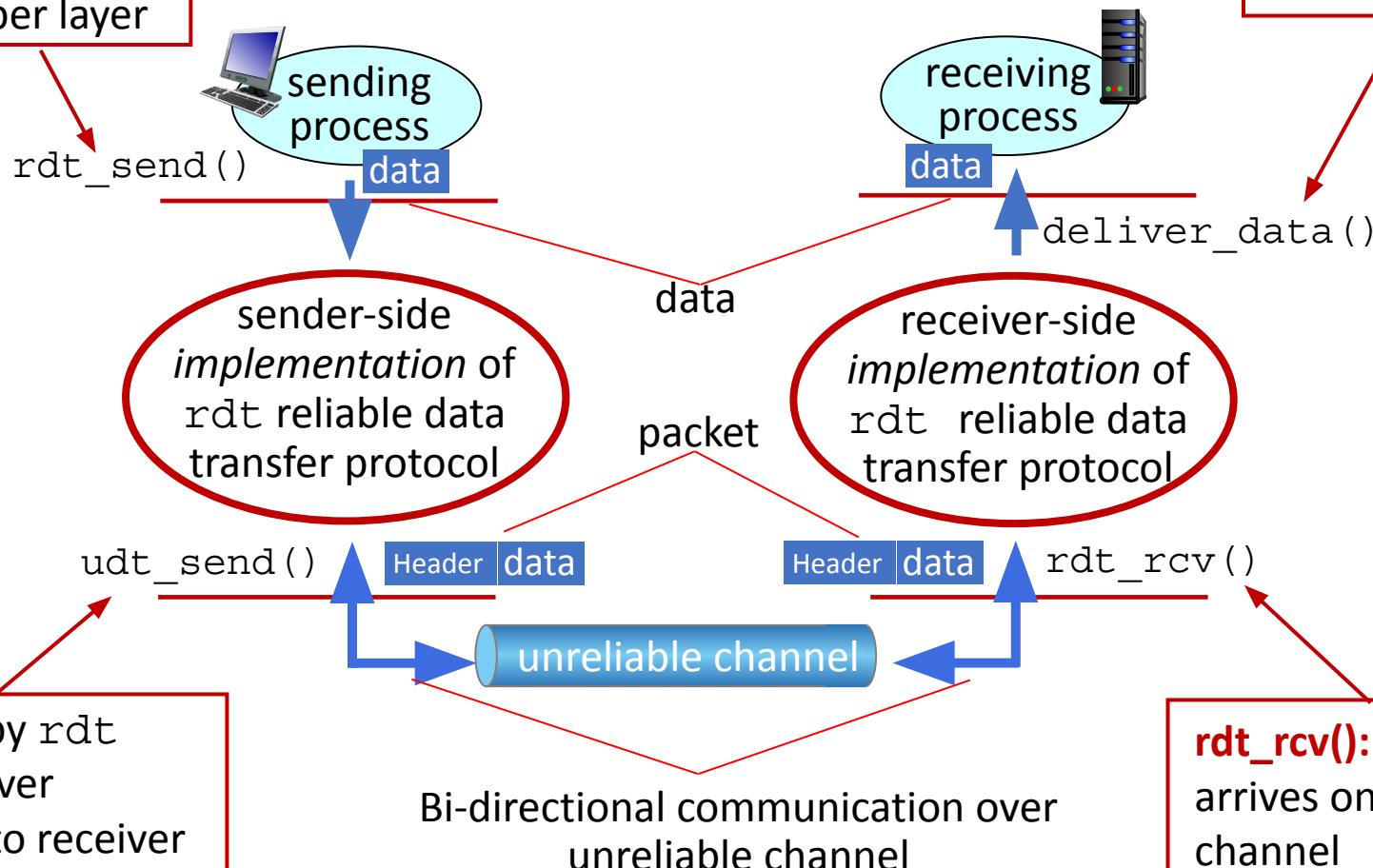


reliable service *implementation*



Reliable data transfer protocol (rdt): interfaces

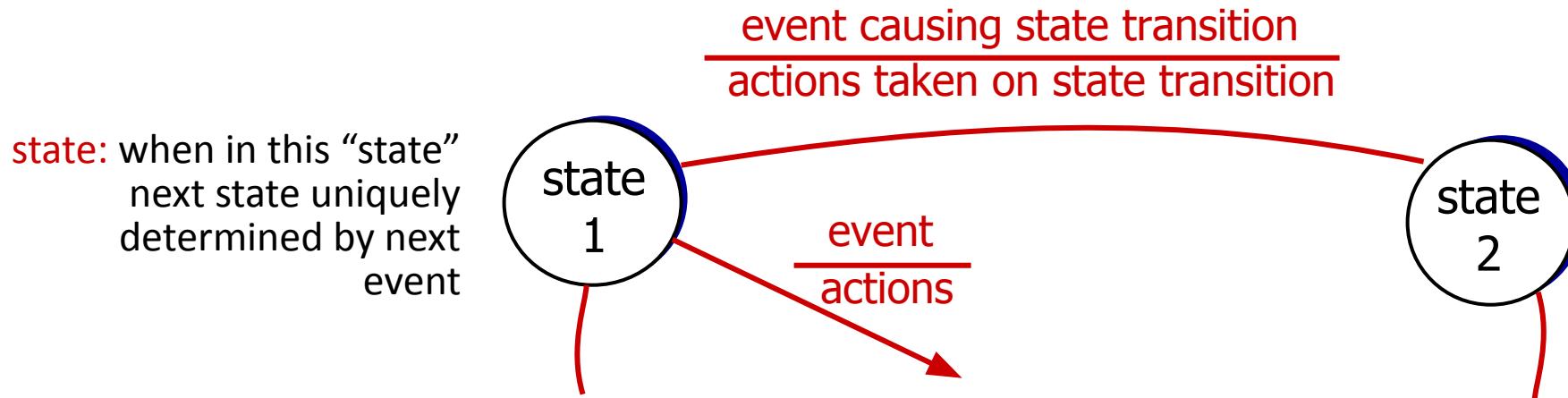
rdt_send(): called from above, (e.g., by app.). Passed data to deliver to receiver upper layer



Reliable data transfer: getting started

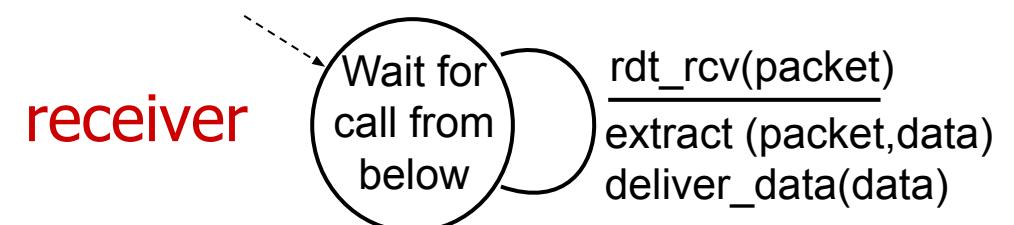
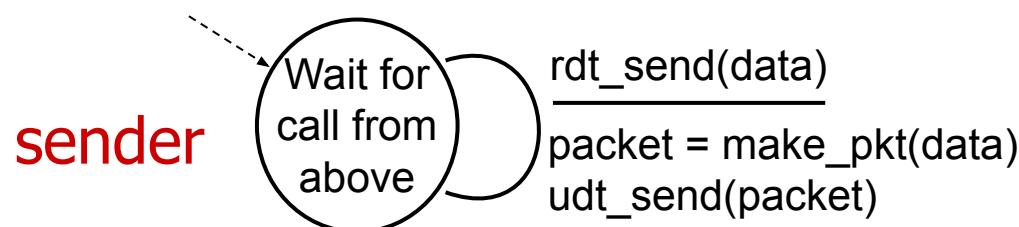
We will:

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



rdt1.0: reliable transfer over a reliable channel

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



rdt2.0: channel with bit errors

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

How do humans recover from “errors” during conversation?

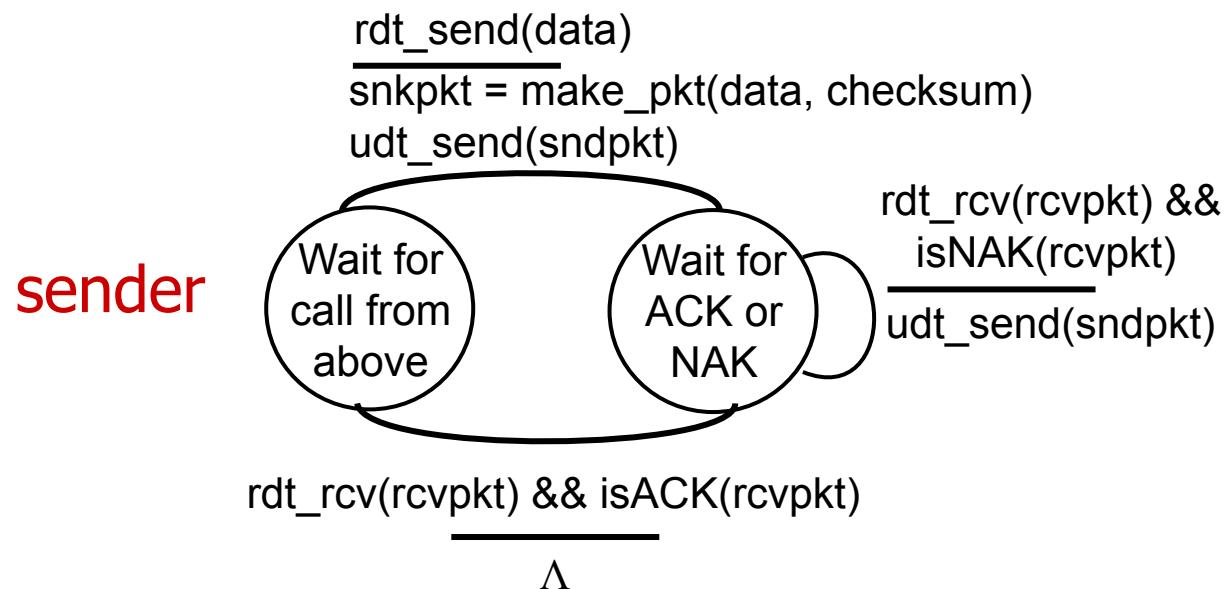
rdt2.0: channel with bit errors

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

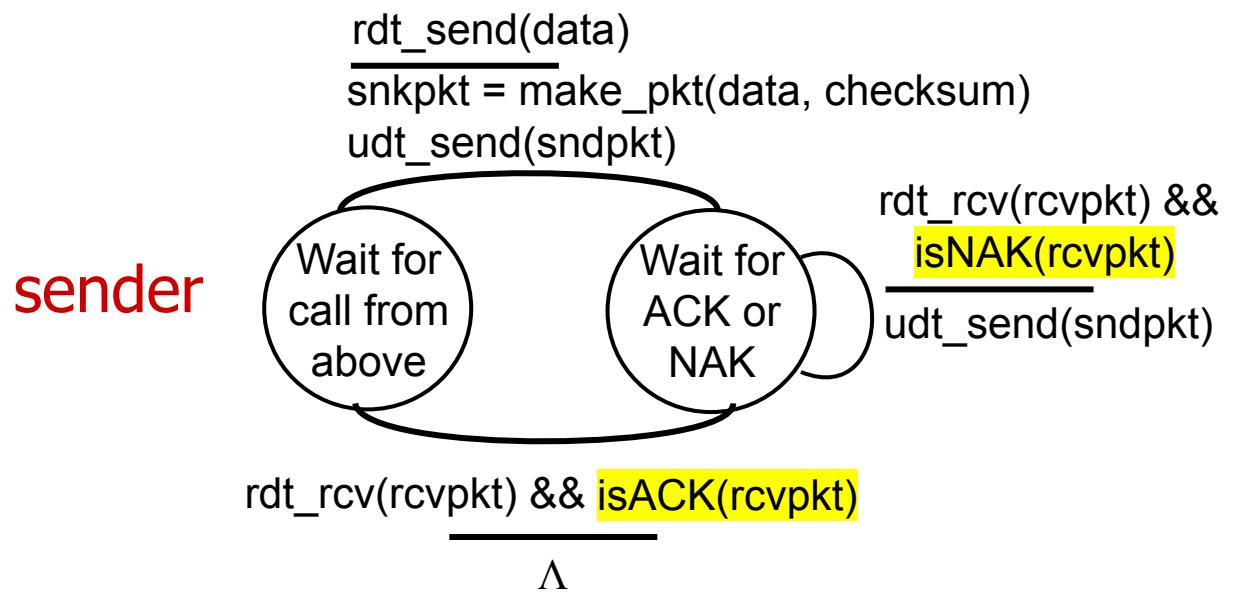
stop and wait

sender sends one packet, then waits for receiver response

rdt2.0: FSM specifications



rdt2.0: FSM specification

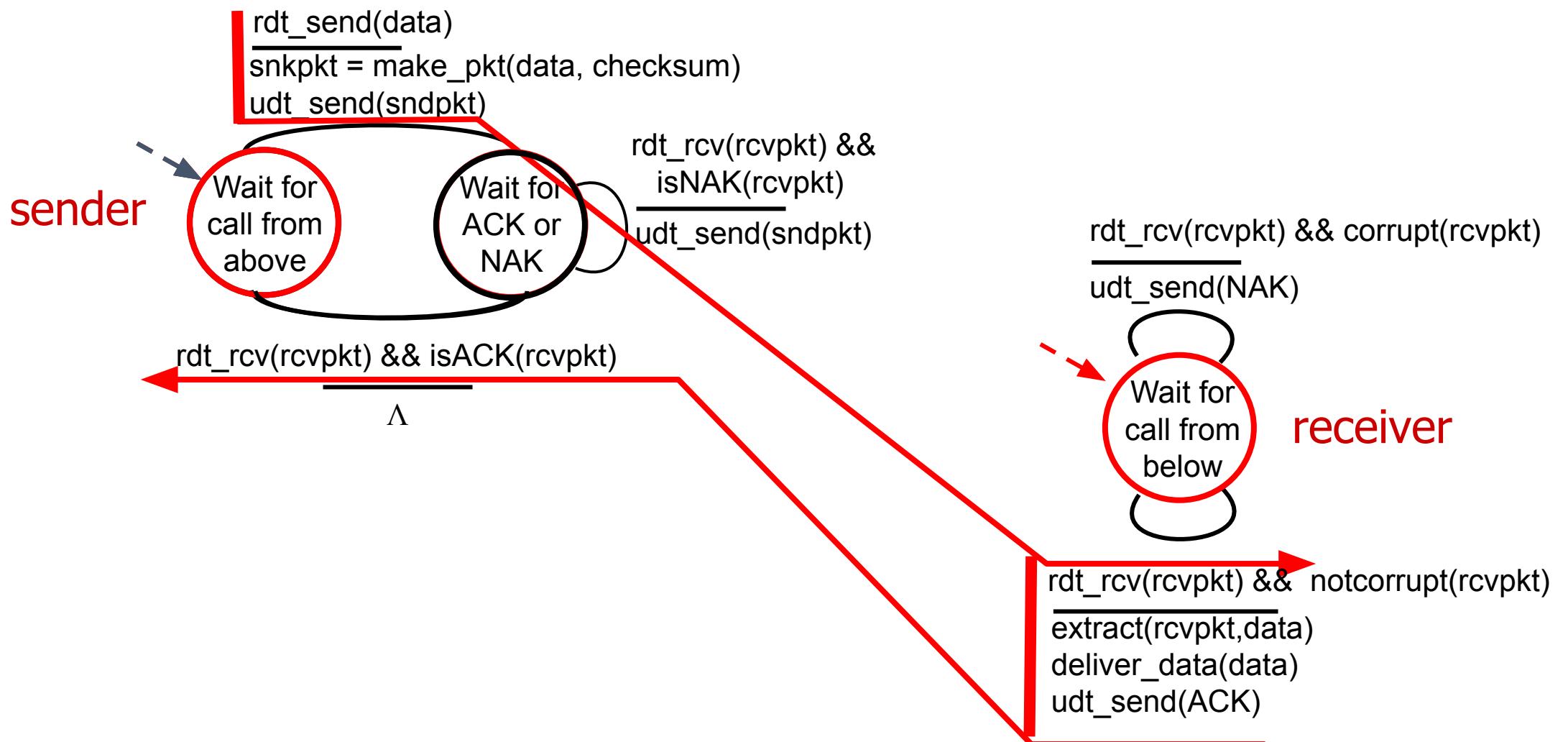


Note: “state” of receiver (did the receiver get my message correctly?) isn’t known to sender unless somehow communicated from receiver to sender

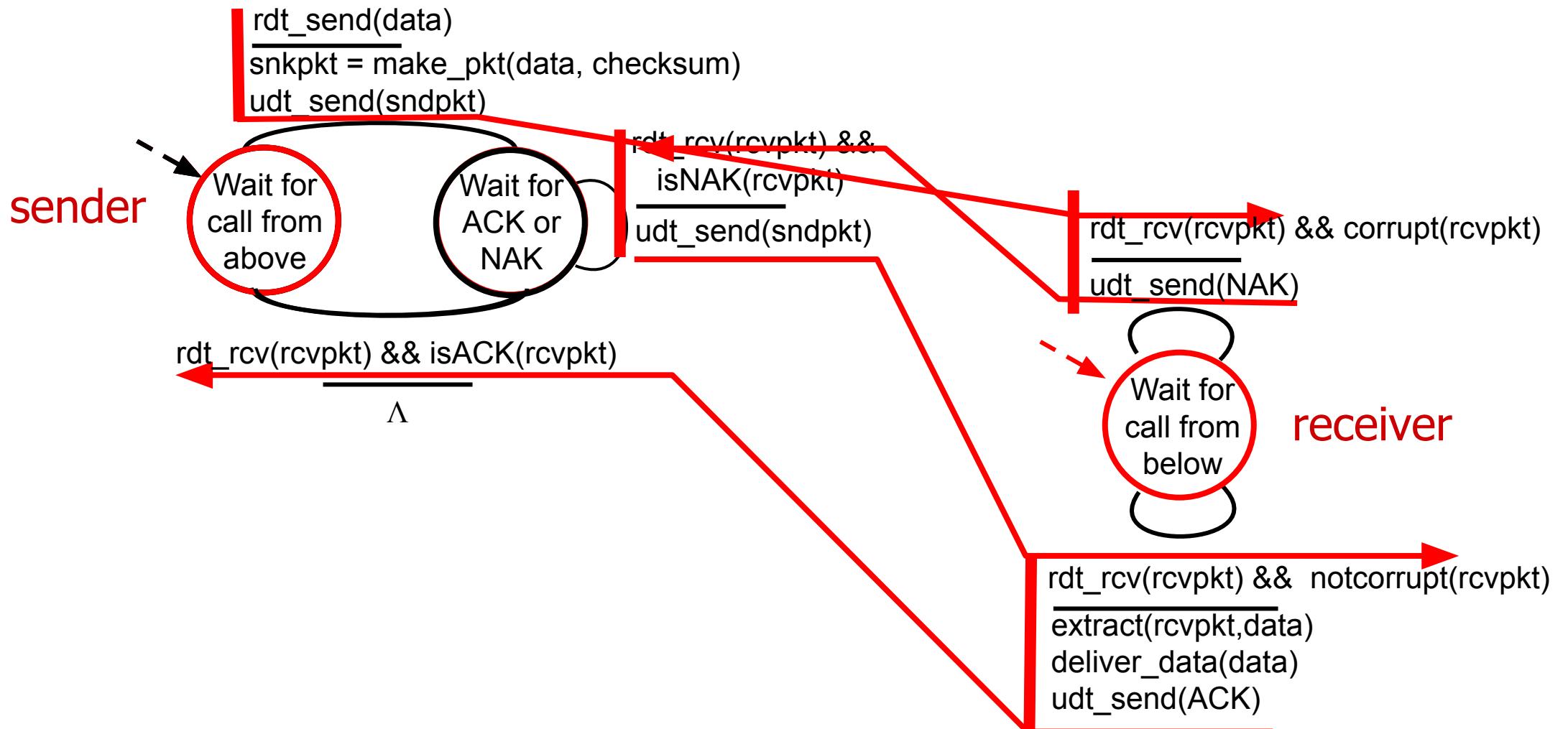
- that’s why we need a protocol!



rdt2.0: operation with no errors



rdt2.0: corrupted packet 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

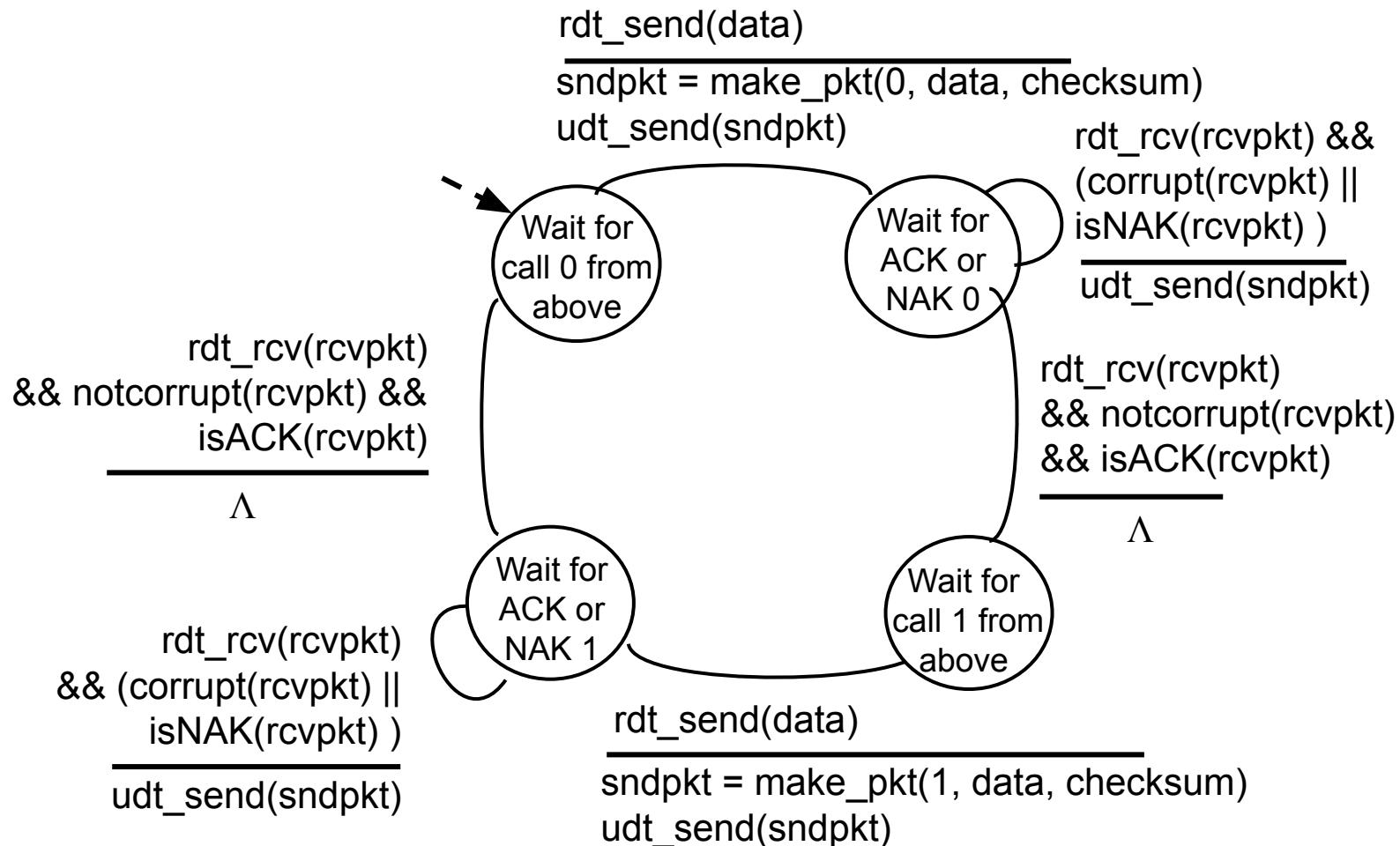
handling duplicates:

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

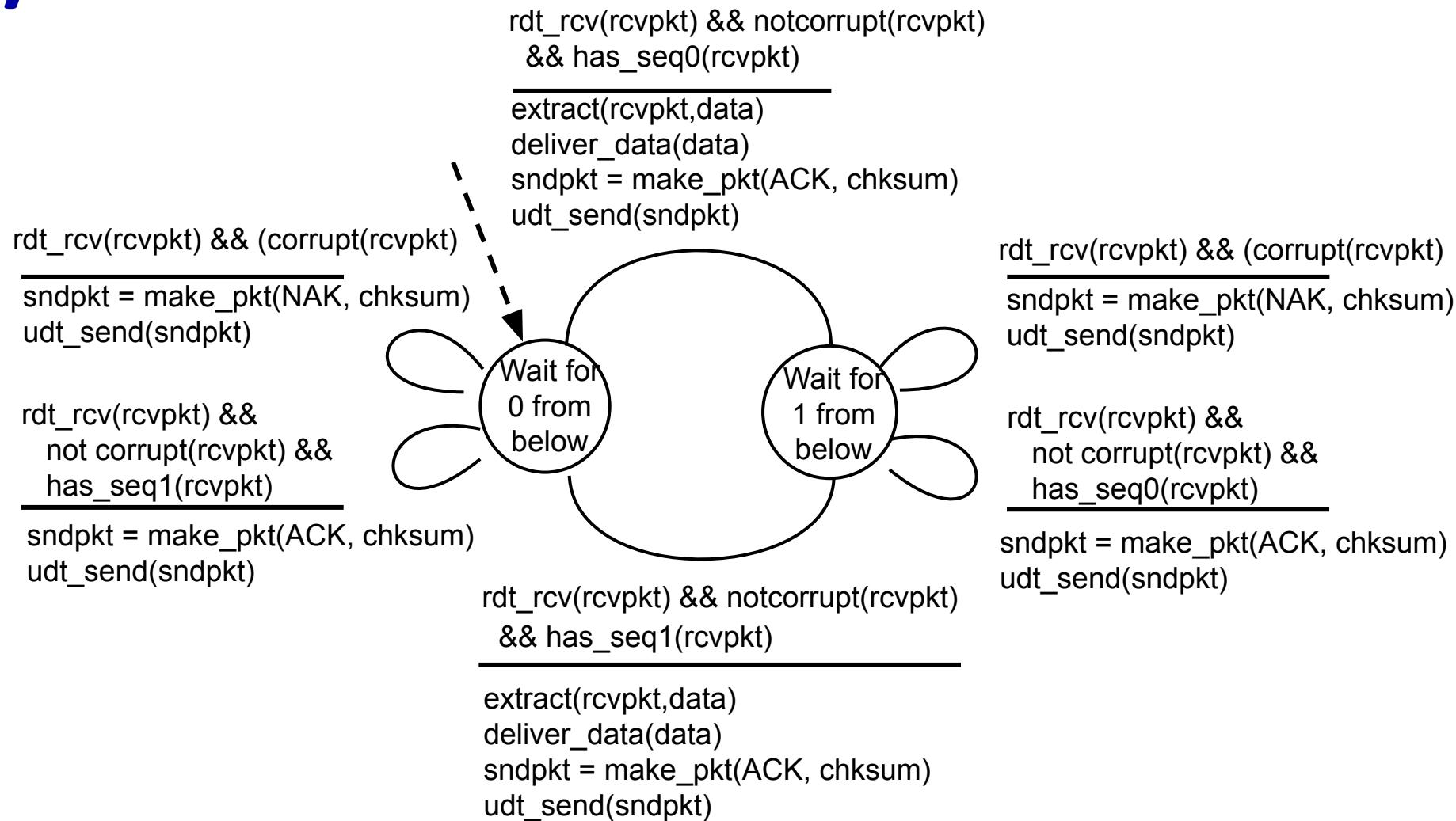
stop and wait

sender sends one packet, then waits for receiver response

rdt2.1: sender, handling garbled ACK/NAKs



rdt2.1: receiver, handling 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:

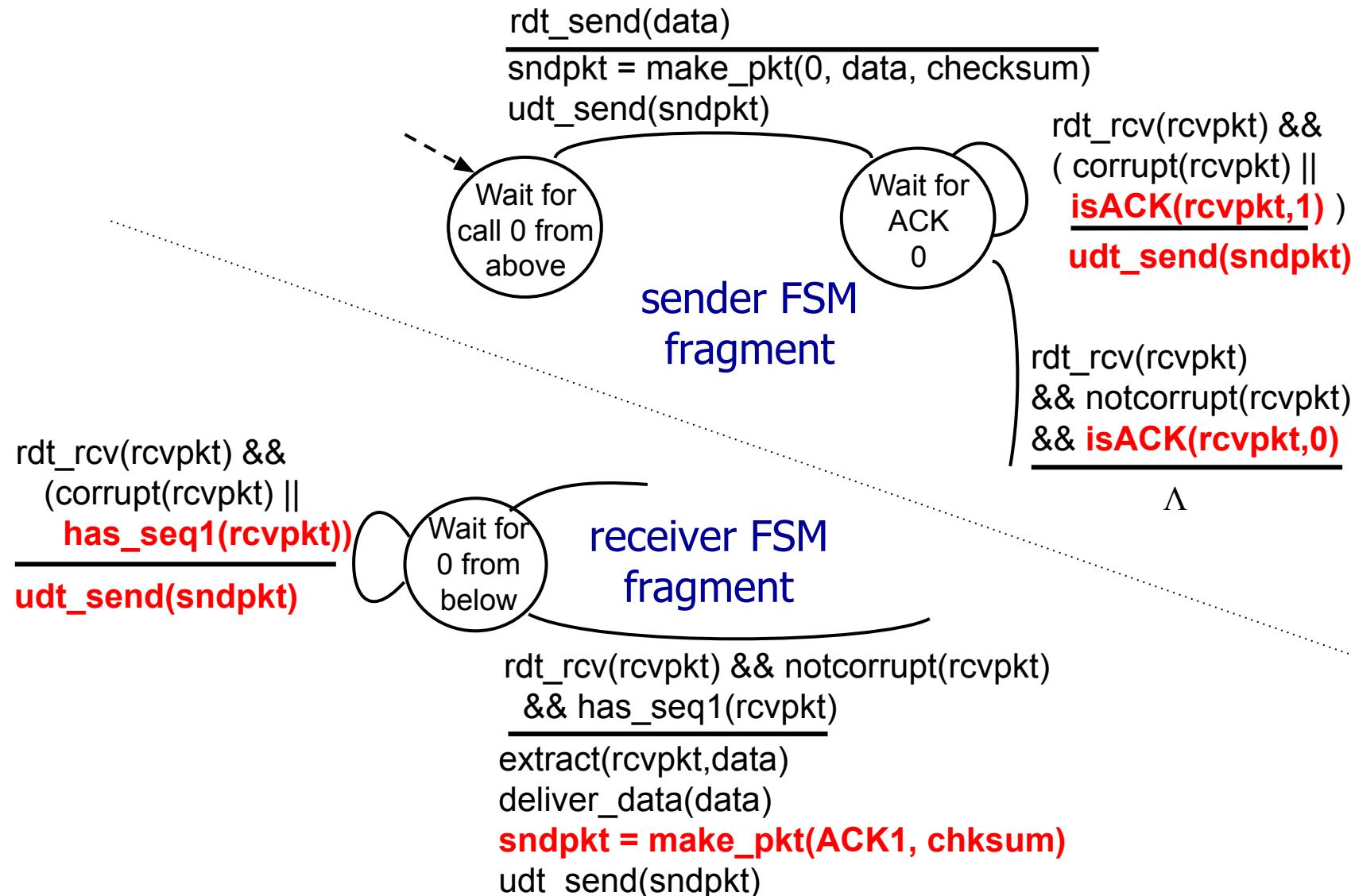
- must check if received packet is duplicate
 - state indicates whether 0 or 1 is expected pkt seq #
- note: receiver can *not* know if its last ACK/NAK received OK at sender

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

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

rdt2.2: sender, receiver fragments



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?

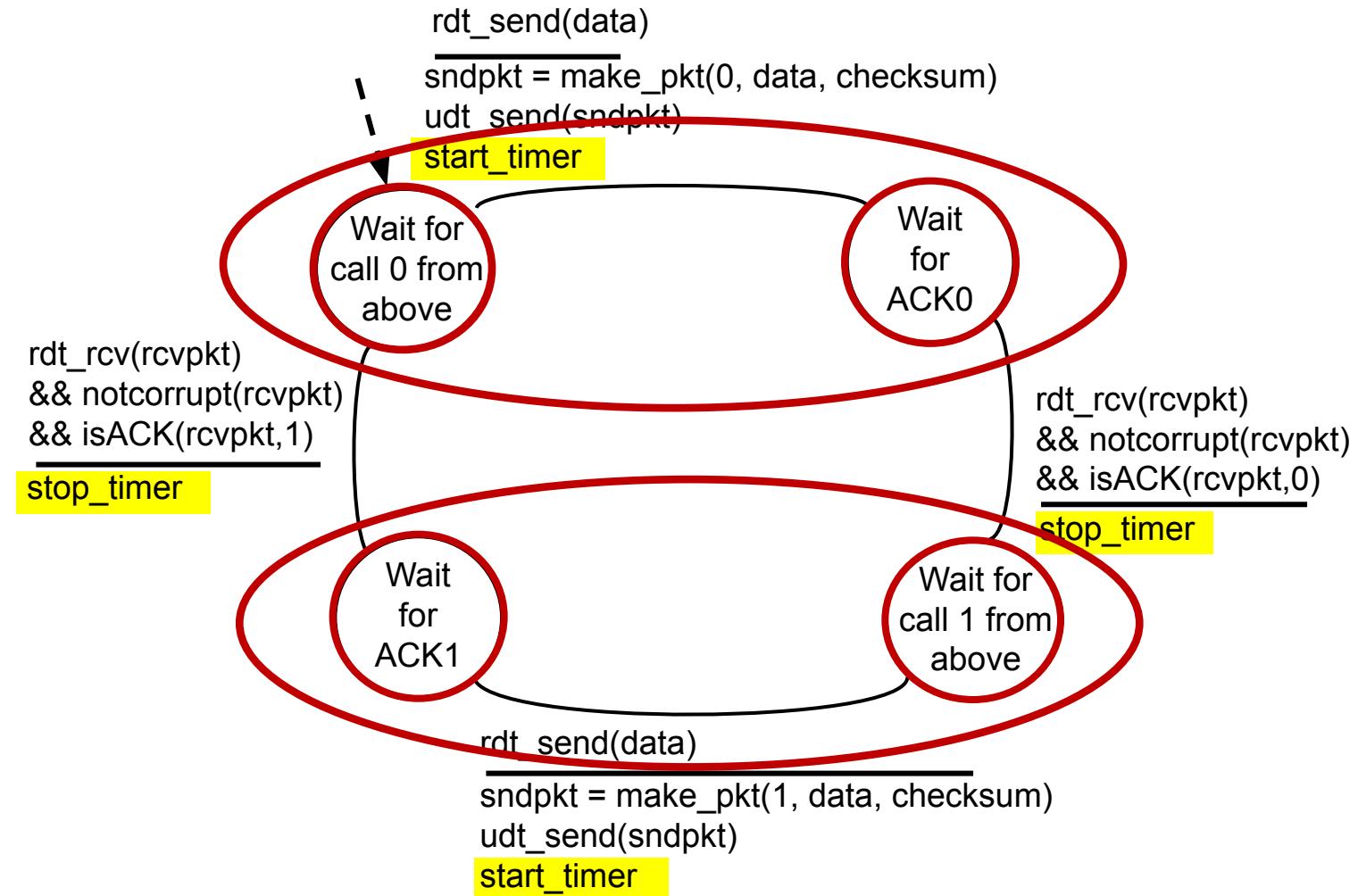
rdt3.0: channels with errors *and* loss

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

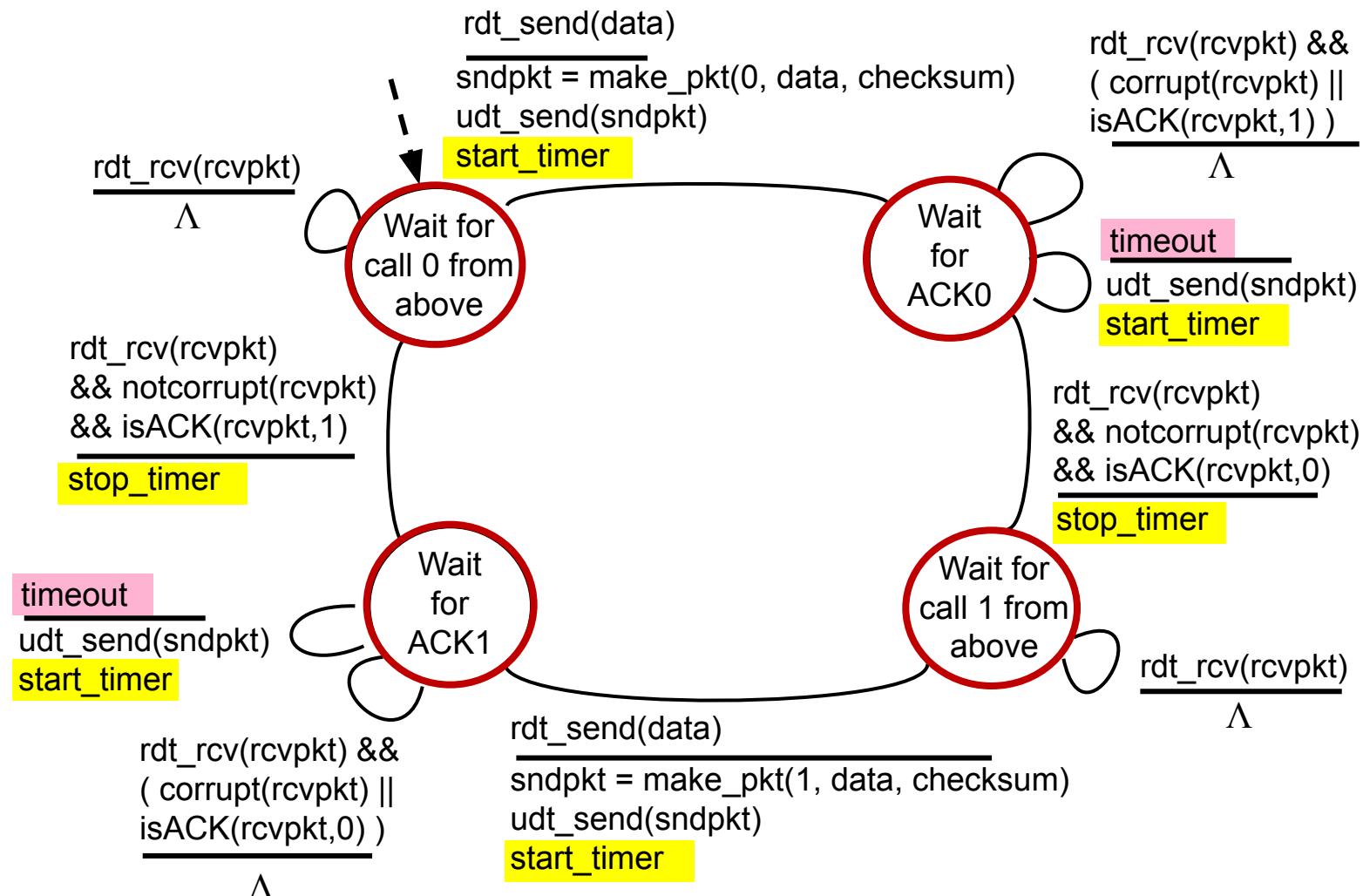
- retransmits if no ACK received in this time
- if pkt (or ACK) just delayed (not lost):
 - retransmission will be duplicate, but seq #s already handles this!
 - receiver must specify seq # of packet being ACKed
- use countdown timer to interrupt after “reasonable” amount of time



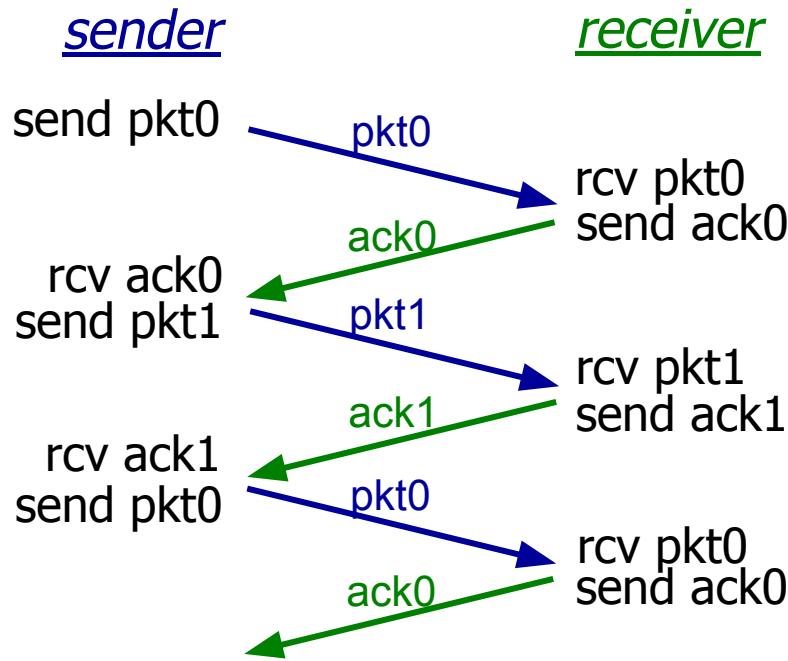
rdt3.0 sender



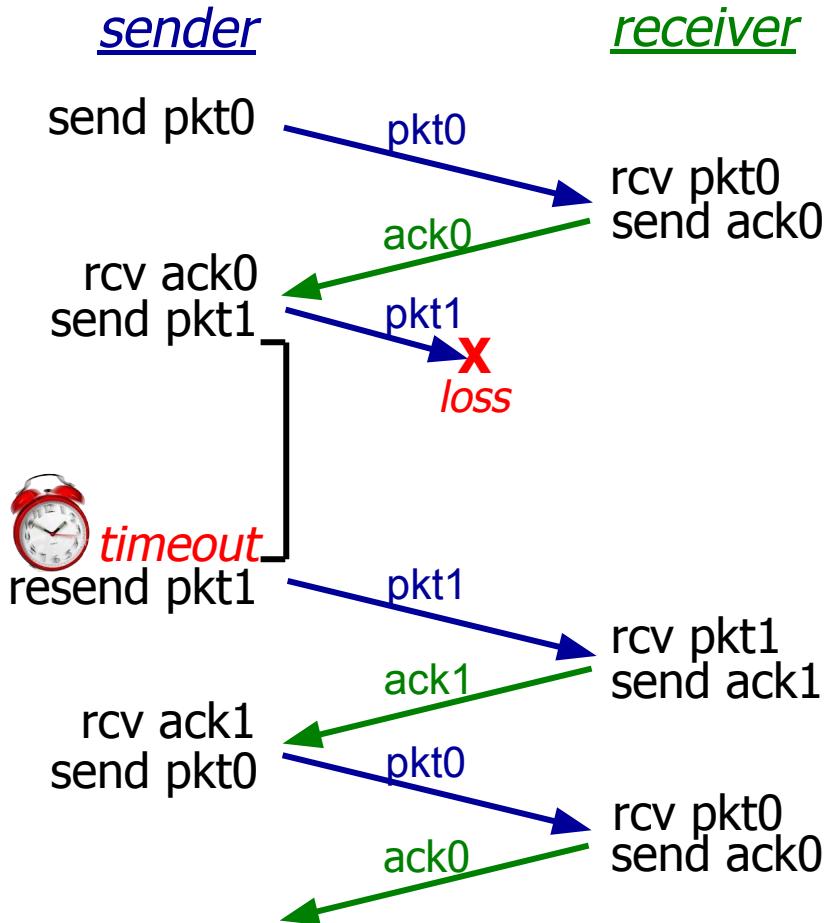
rdt3.0 sender



rdt3.0 in action

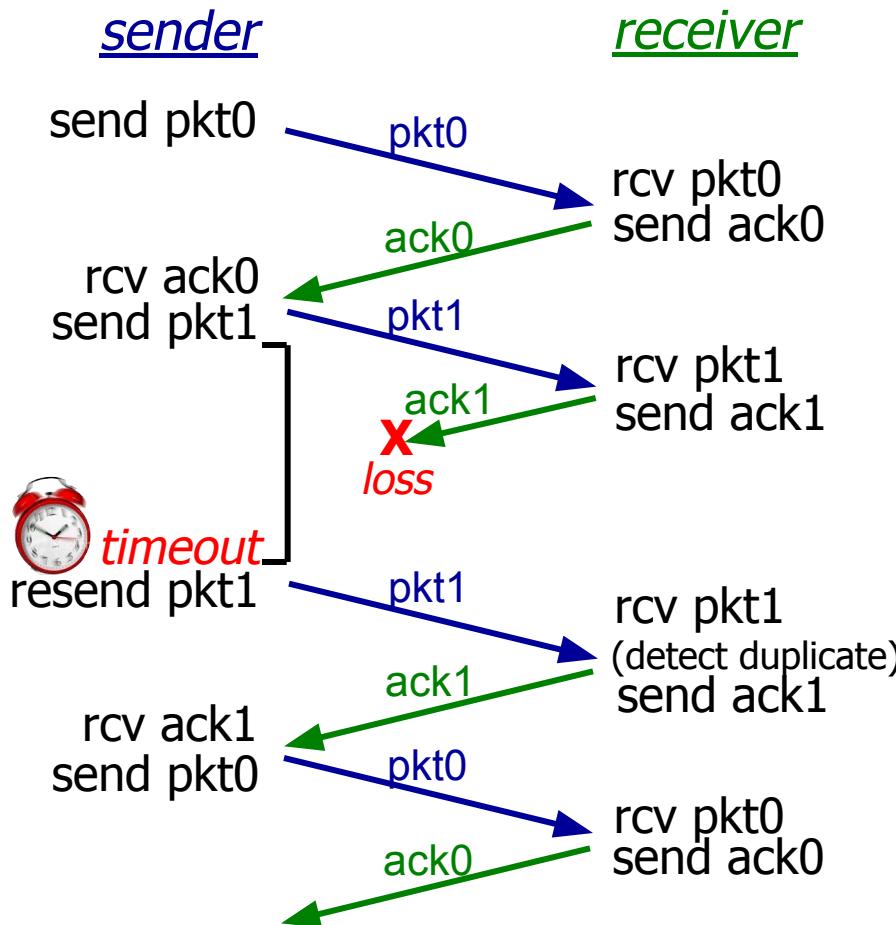


(a) no loss

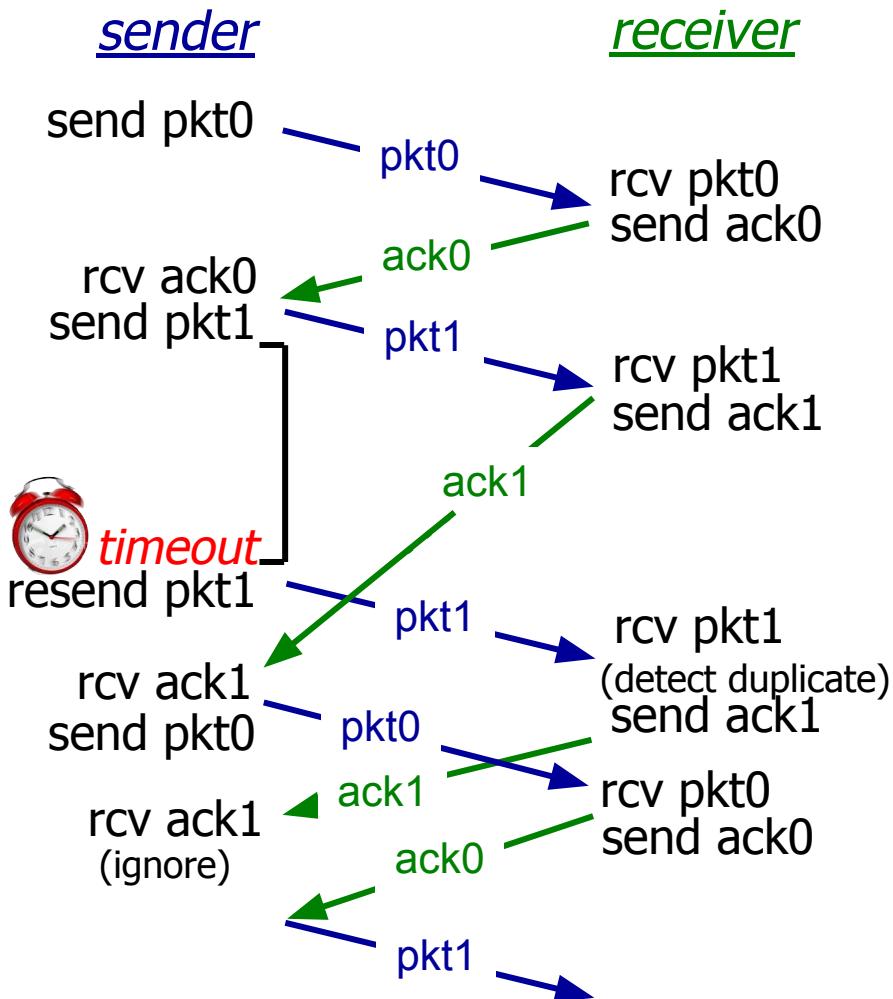


(b) packet loss

rdt3.0 in action



(c) ACK loss



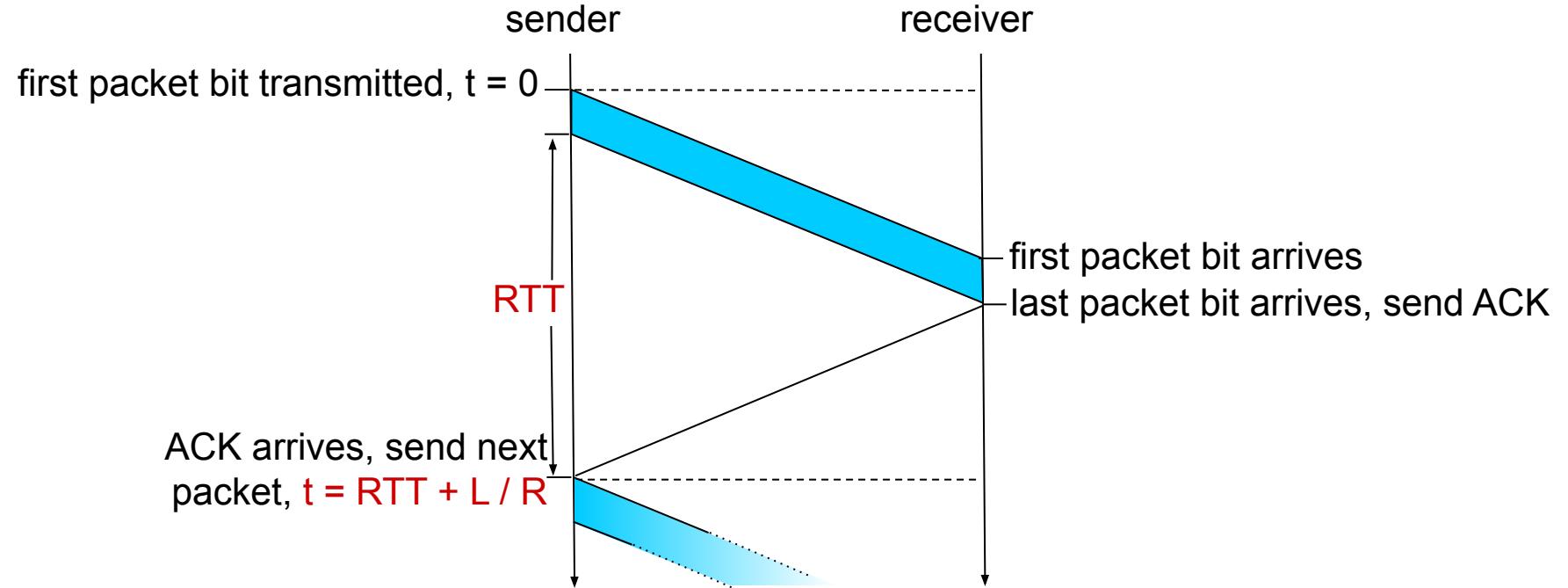
(d) premature timeout/ delayed ACK

Performance of rdt3.0 (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:

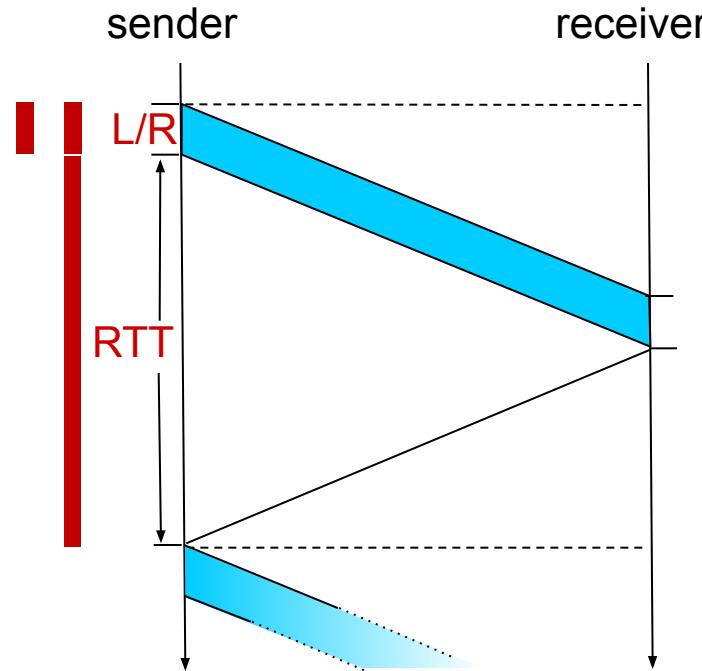
$$D_{\text{trans}} = \frac{L}{R} = \frac{8000 \text{ bits}}{10^9 \text{ bits/sec}} = 8 \text{ microsecs}$$

rdt3.0: stop-and-wait operation



rdt3.0: stop-and-wait operation

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

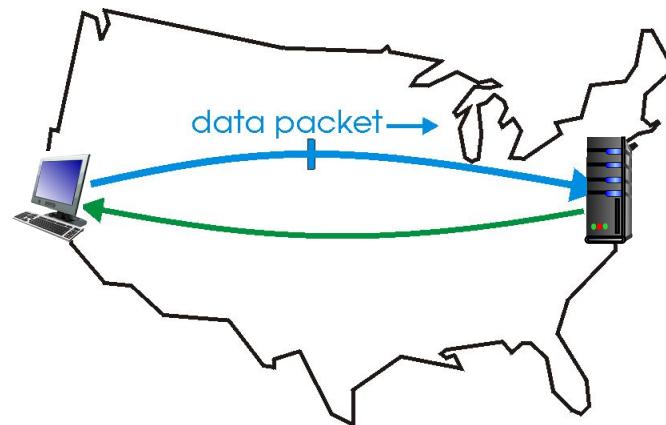


- rdt 3.0 protocol performance stinks!
- Protocol limits performance of underlying infrastructure (channel)

rdt3.0: pipelined protocols operation

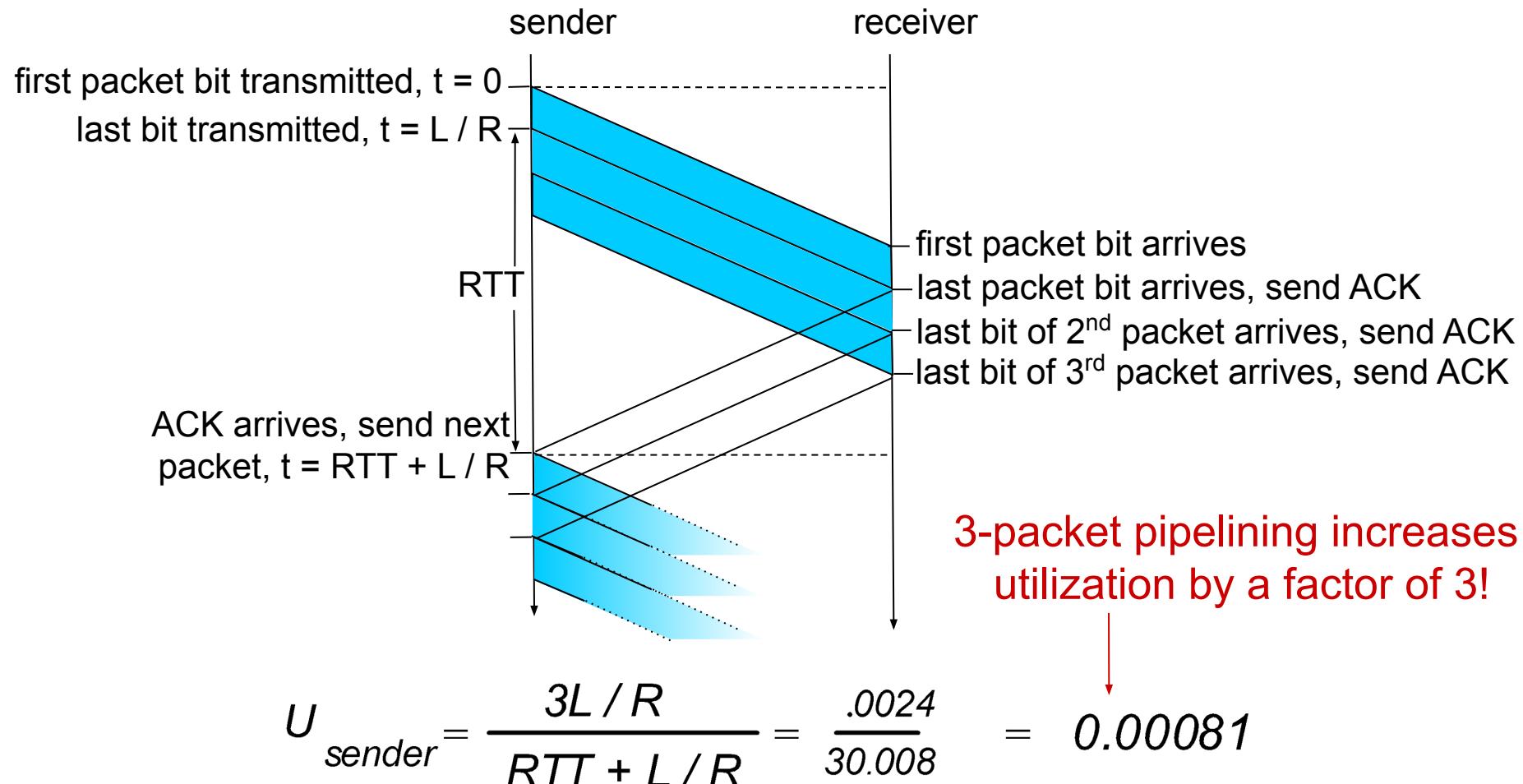
pipelining: sender allows multiple, “in-flight”, yet-to-be-acknowledged packets

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



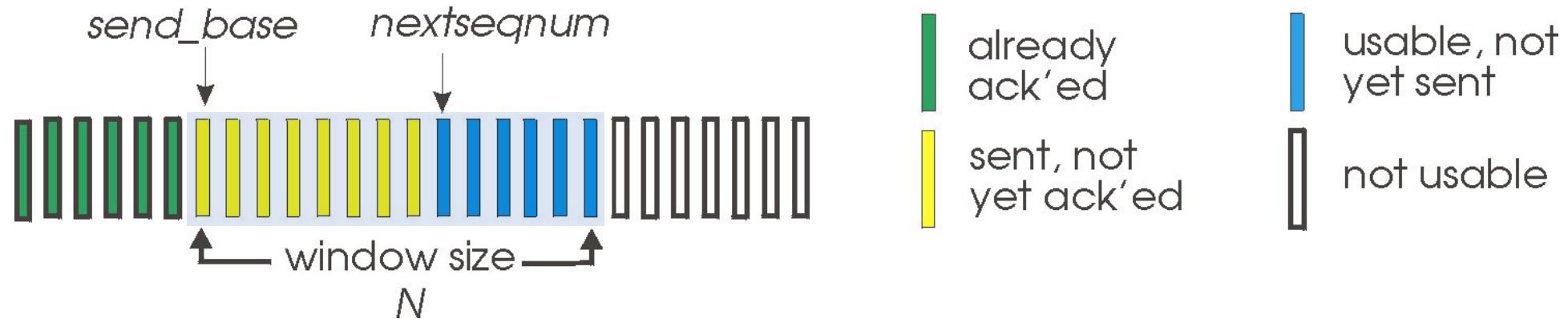
(a) a stop-and-wait protocol in operation

Pipelining: increased utilization



Go-Back-N: sender

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

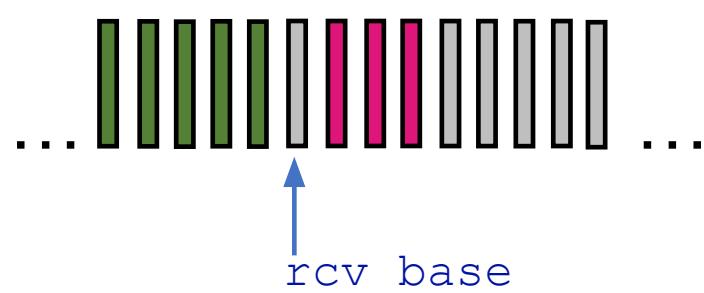


- *cumulative ACK*: $\text{ACK}(n)$: ACKs all packets up to, including seq # n
 - on receiving $\text{ACK}(n)$: move window forward to begin at $n+1$
- timer for oldest in-flight packet
- $\text{timeout}(n)$: retransmit packet n and all higher seq # packets in window

Go-Back-N: receiver

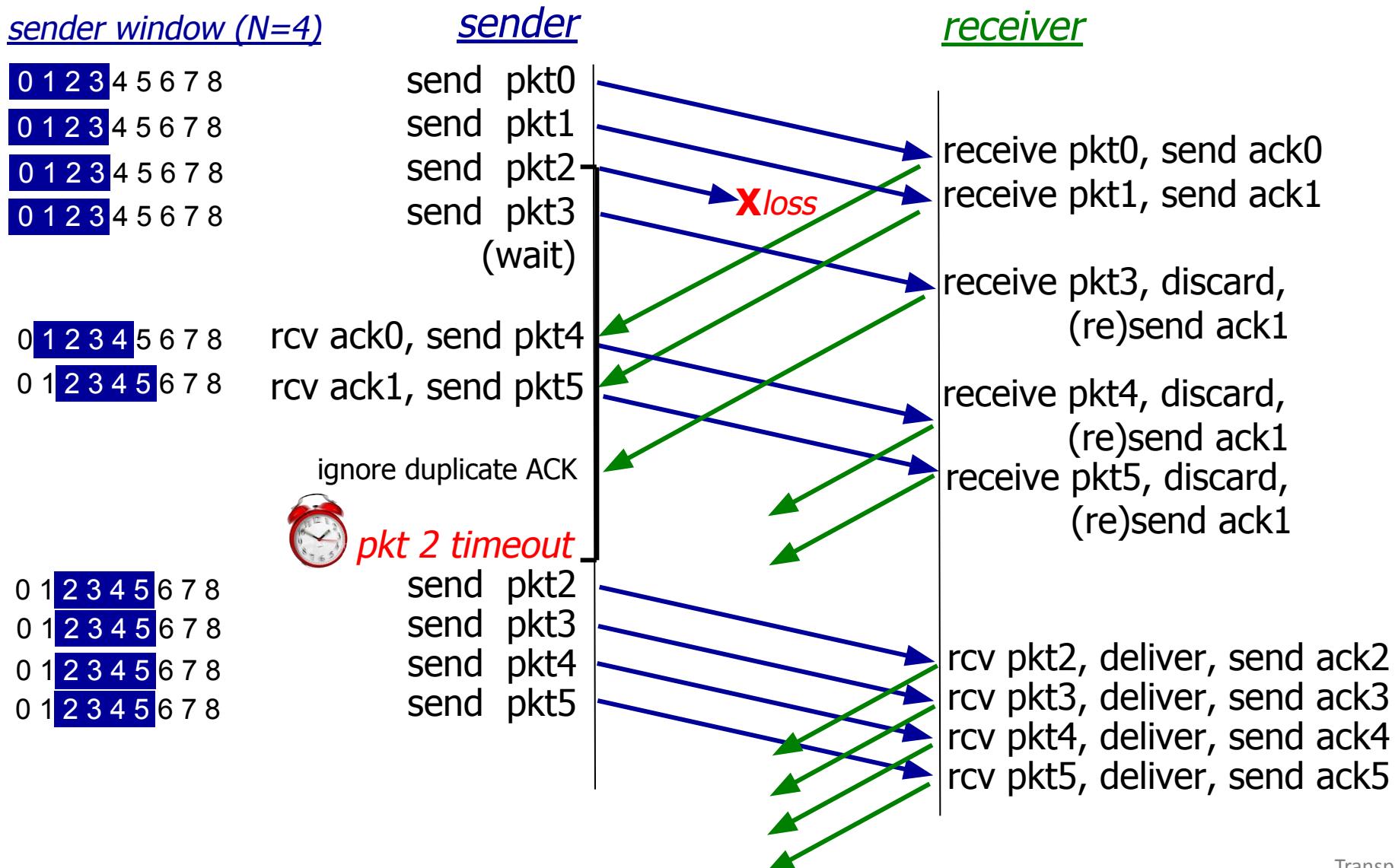
- ACK-only: always send ACK for correctly-received packet so far, with highest *in-order* seq #
 - may generate duplicate ACKs
 - need only remember `rcv_base`
- on receipt of out-of-order packet:
 - can discard (don't buffer) or buffer: an implementation decision
 - re-ACK pkt with highest in-order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

Receiver view of sequence number space:



	received and ACKed
	Out-of-order: received but not ACKed
	Not received

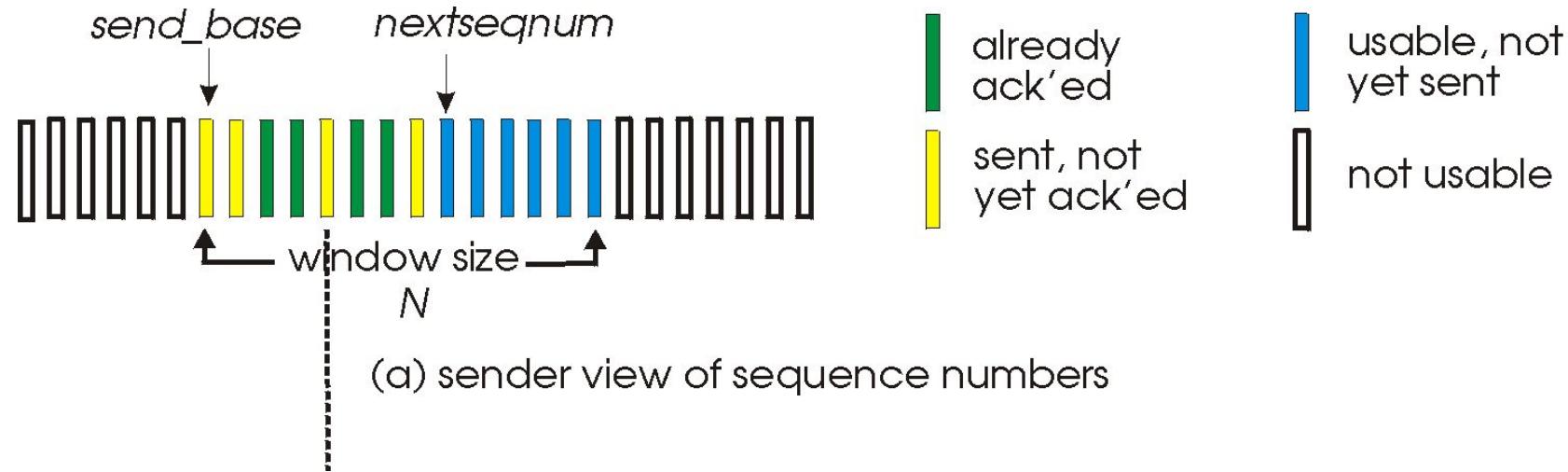
Go-Back-N in action



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

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

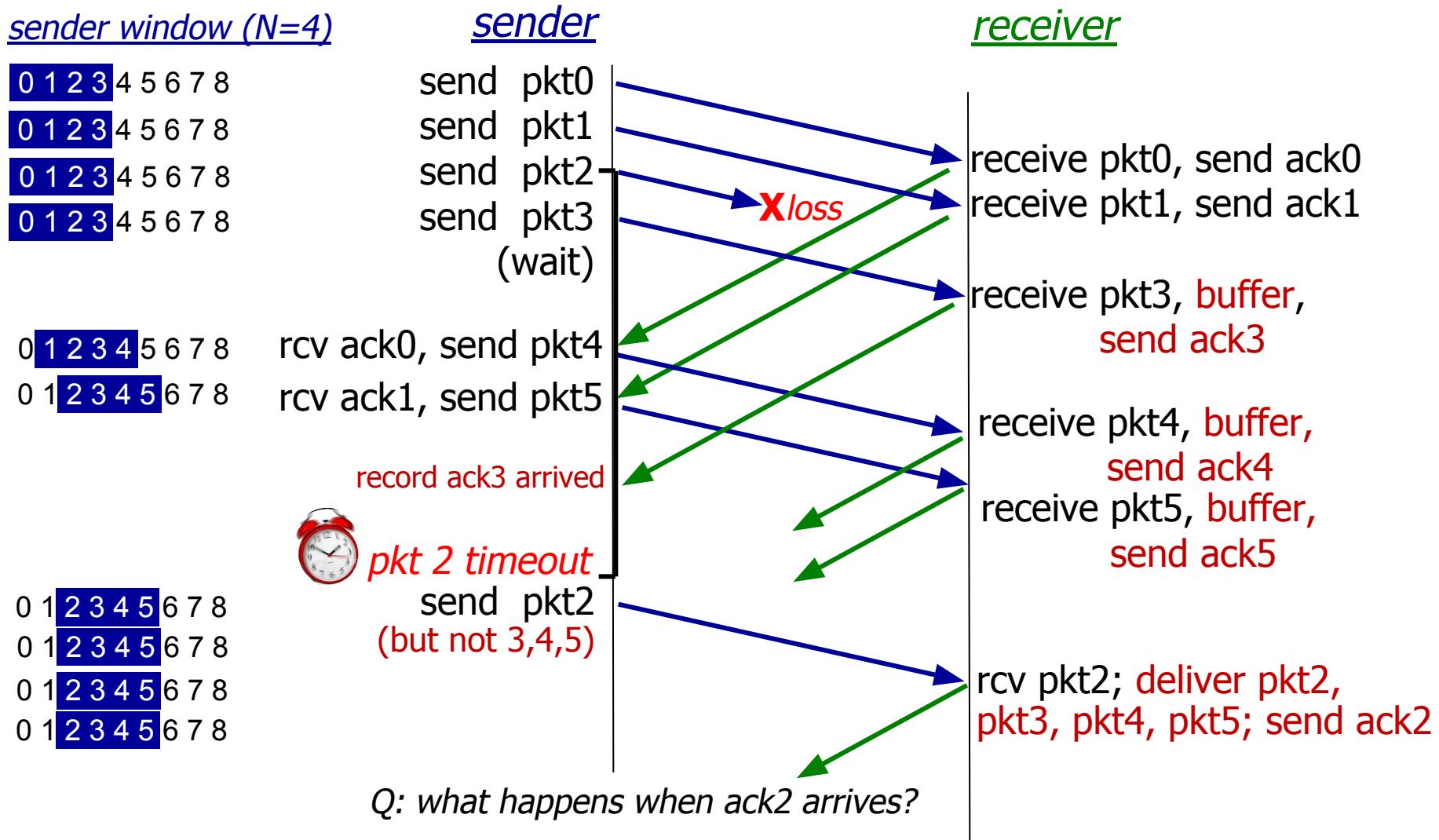
packet n in [rcvbase-N,rcvbase-1]

- ACK(n)

otherwise:

- ignore

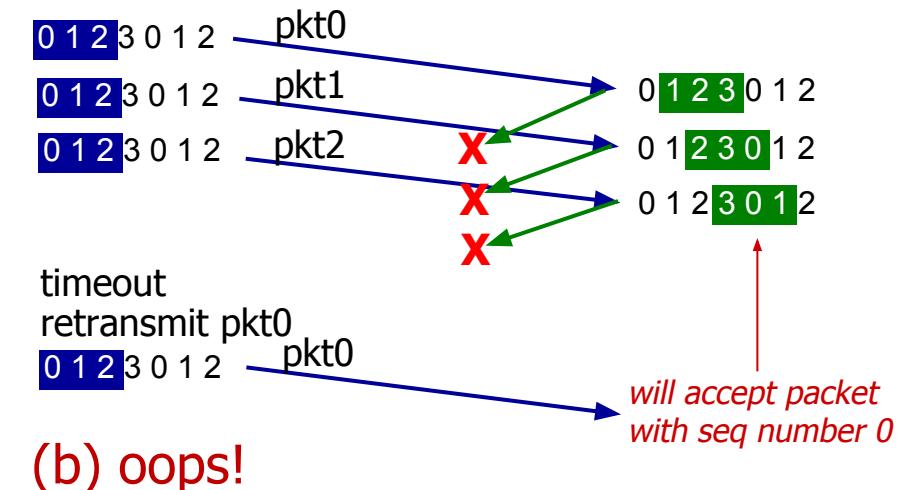
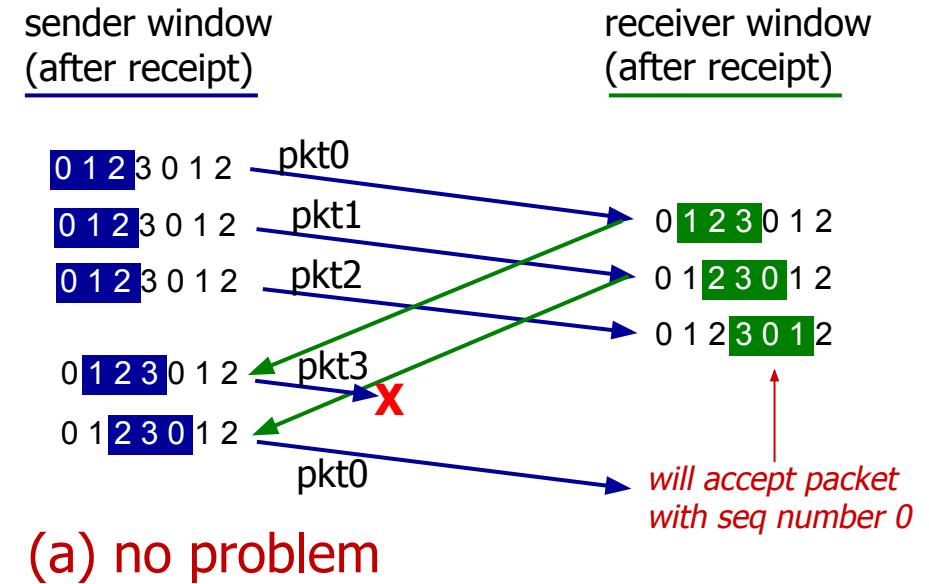
Selective Repeat in action



Selective repeat: a dilemma!

example:

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



Selective repeat: a dilemma!

example:

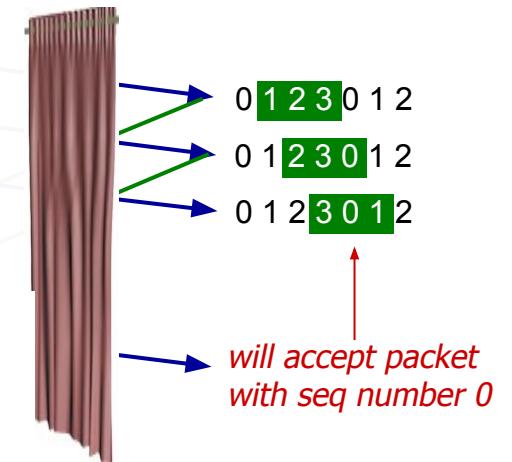
- seq #s: 0, 1, 2, 3 (base 4 counting)
- window size=3

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

sender window
(after receipt)



receiver window
(after receipt)



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



(b) oops!

Chapter 3: roadmap

- Transport-layer services
- Multiplexing and demultiplexing
- Connectionless transport: UDP
- Principles of reliable data transfer
- **Connection-oriented transport: TCP**
 - segment structure
 - reliable data transfer
 - flow control
 - connection management
- Principles of congestion control
- TCP congestion control



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

ACK: seq # of next expected byte; A bit: this is an ACK

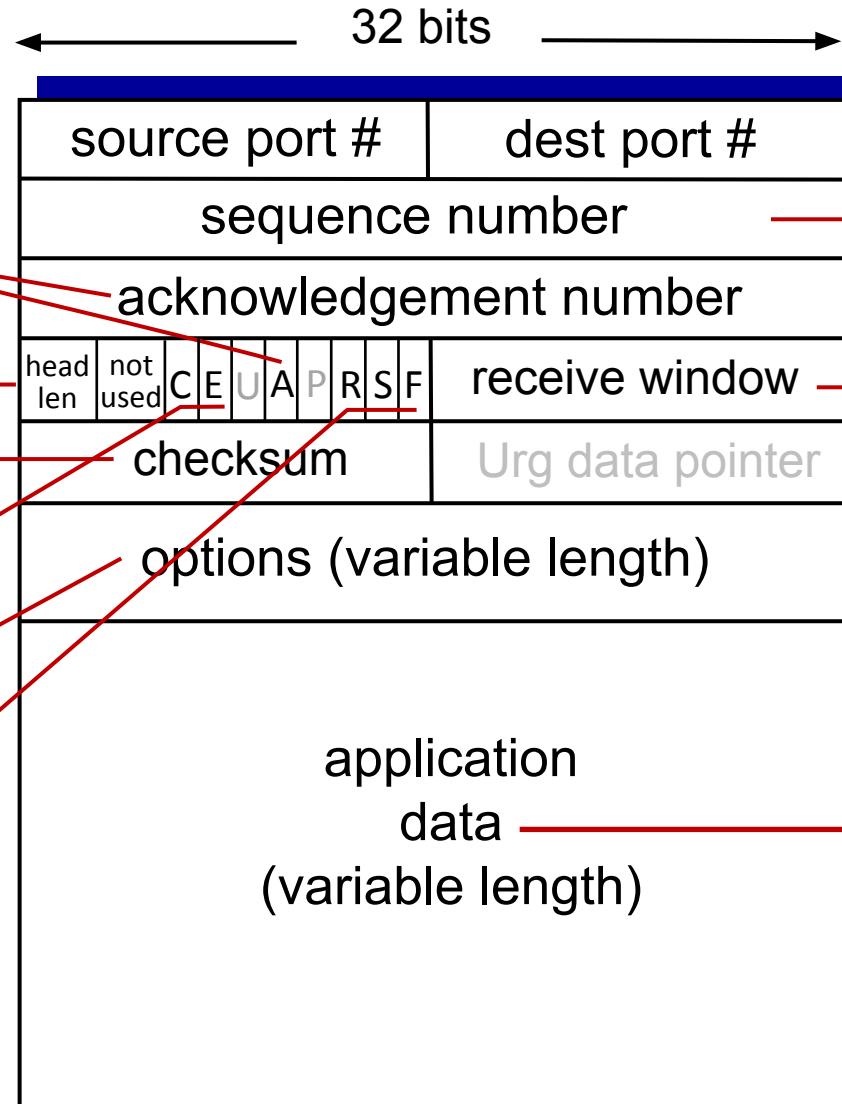
length (of TCP header)

Internet checksum

C, E: congestion notification

TCP options

RST, SYN, FIN: connection management



segment seq #: counting bytes of data into bytestream (not segments!)

flow control: # bytes receiver willing to accept

data sent by application into TCP socket

TCP sequence numbers, ACKs

Sequence numbers:

- byte stream “number” of first byte in segment’s data

Acknowledgements:

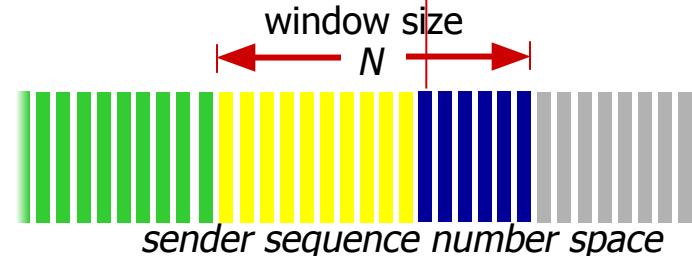
- seq # of next byte expected from other side
- cumulative ACK

Q: how receiver handles out-of-order segments

- A: TCP spec doesn’t say, - up to implementor

outgoing segment from sender

source port #	dest port #
sequence number	
acknowledgement number	
	rwnd
checksum	urg pointer



sender sequence number space

sent
ACKed

sent,
not-yet
ACKed
("in-flight")

usable
but not
yet sent

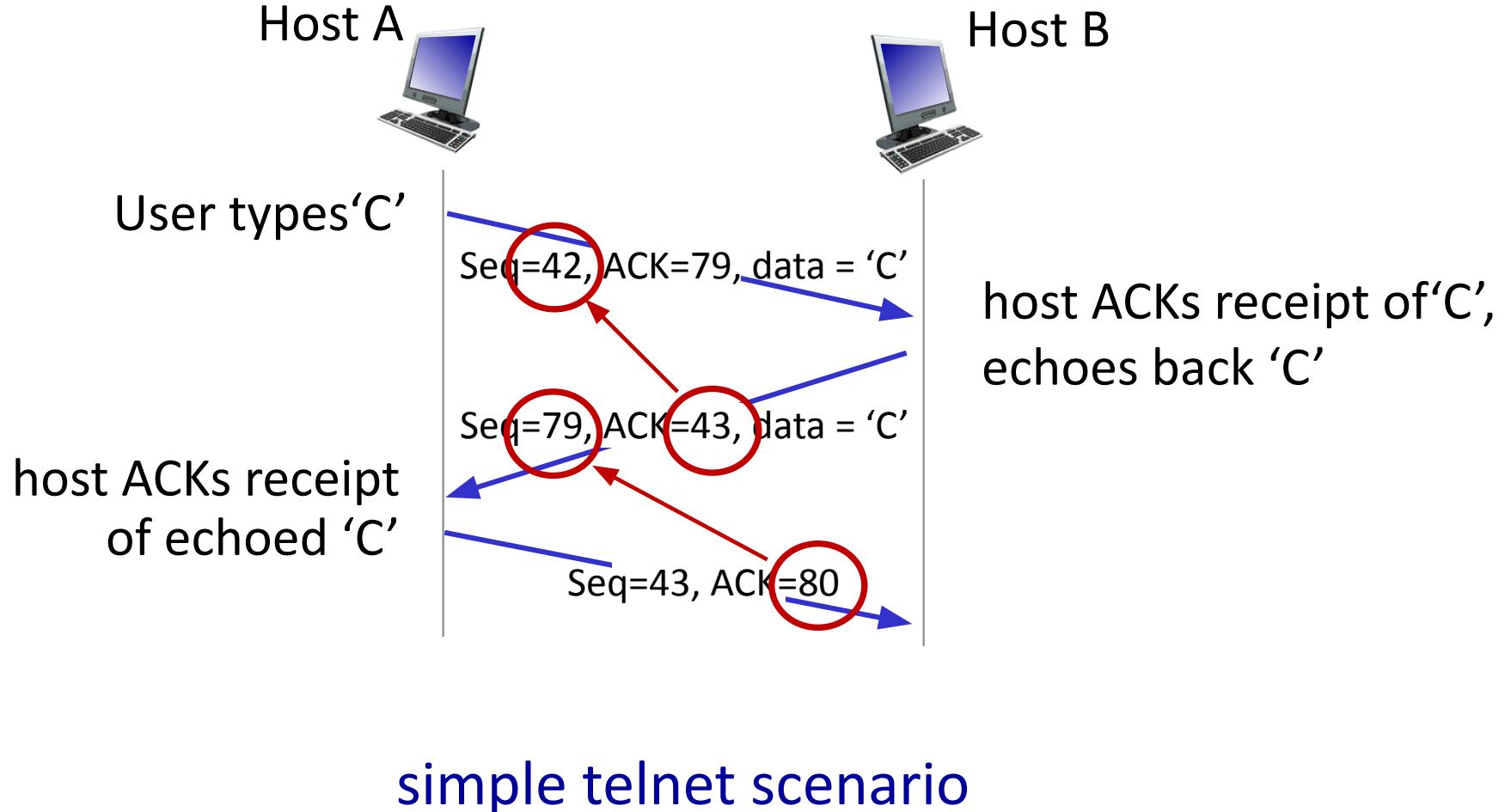
not
usable

outgoing segment from receiver

source port #	dest port #
sequence number	
acknowledgement number	
	rwnd
checksum	urg pointer

A

TCP sequence numbers, ACKs



TCP round trip time, timeout

Q: how to set TCP timeout value?

- longer than RTT, but RTT varies!
- ***too short:*** premature timeout, unnecessary retransmissions
- ***too long:*** slow reaction to segment loss

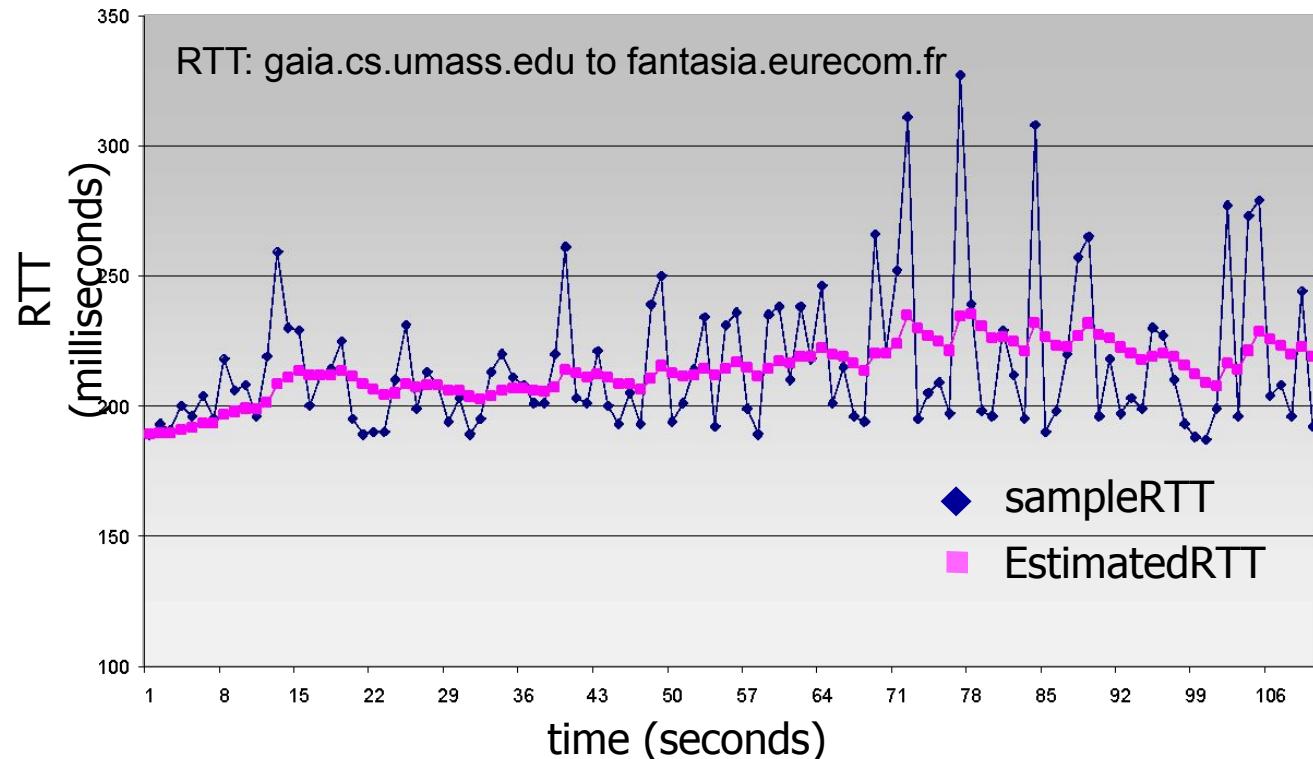
Q: how to estimate RTT?

- **SampleRTT:** measured time from segment transmission until ACK receipt
 - ignore retransmissions
- **SampleRTT will vary,** want estimated RTT “smoother”
 - average several *recent* measurements, not just current SampleRTT

TCP round trip time, timeout

$$\text{EstimatedRTT} = (1 - \alpha) * \text{EstimatedRTT} + \alpha * \text{SampleRTT}$$

- exponential weighted moving average (EWMA)
- influence of past sample decreases exponentially fast
- typical value: $\alpha = 0.125$



TCP round trip time, timeout

- timeout interval: **EstimatedRTT** plus “safety margin”
 - large variation in **EstimatedRTT**: want a larger safety margin

$$\text{TimeoutInterval} = \text{EstimatedRTT} + 4 * \text{DevRTT}$$



estimated RTT

“safety margin”

- DevRTT**: EWMA of **SampleRTT** deviation from **EstimatedRTT**:

$$\text{DevRTT} = (1 - \beta) * \text{DevRTT} + \beta * |\text{SampleRTT} - \text{EstimatedRTT}|$$

(typically, $\beta = 0.25$)

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

TCP Sender (simplified)

event: data received from application

- create segment with seq #
- seq # is byte-stream number of first data byte in segment
- start timer if not already running
 - think of timer as for oldest unACKed segment
 - expiration interval:
TimeOutInterval

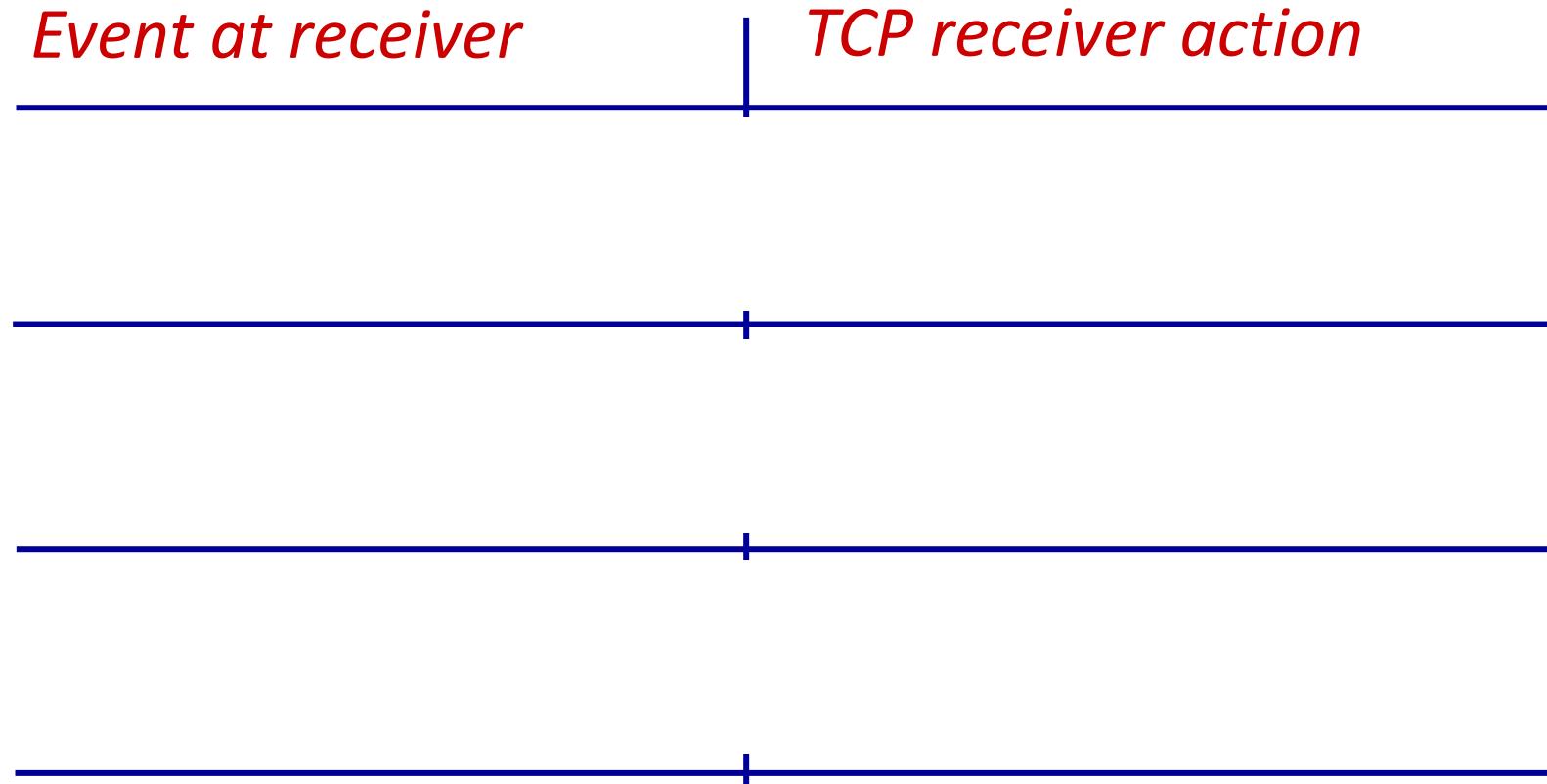
event: timeout

- retransmit segment that caused timeout
- restart timer

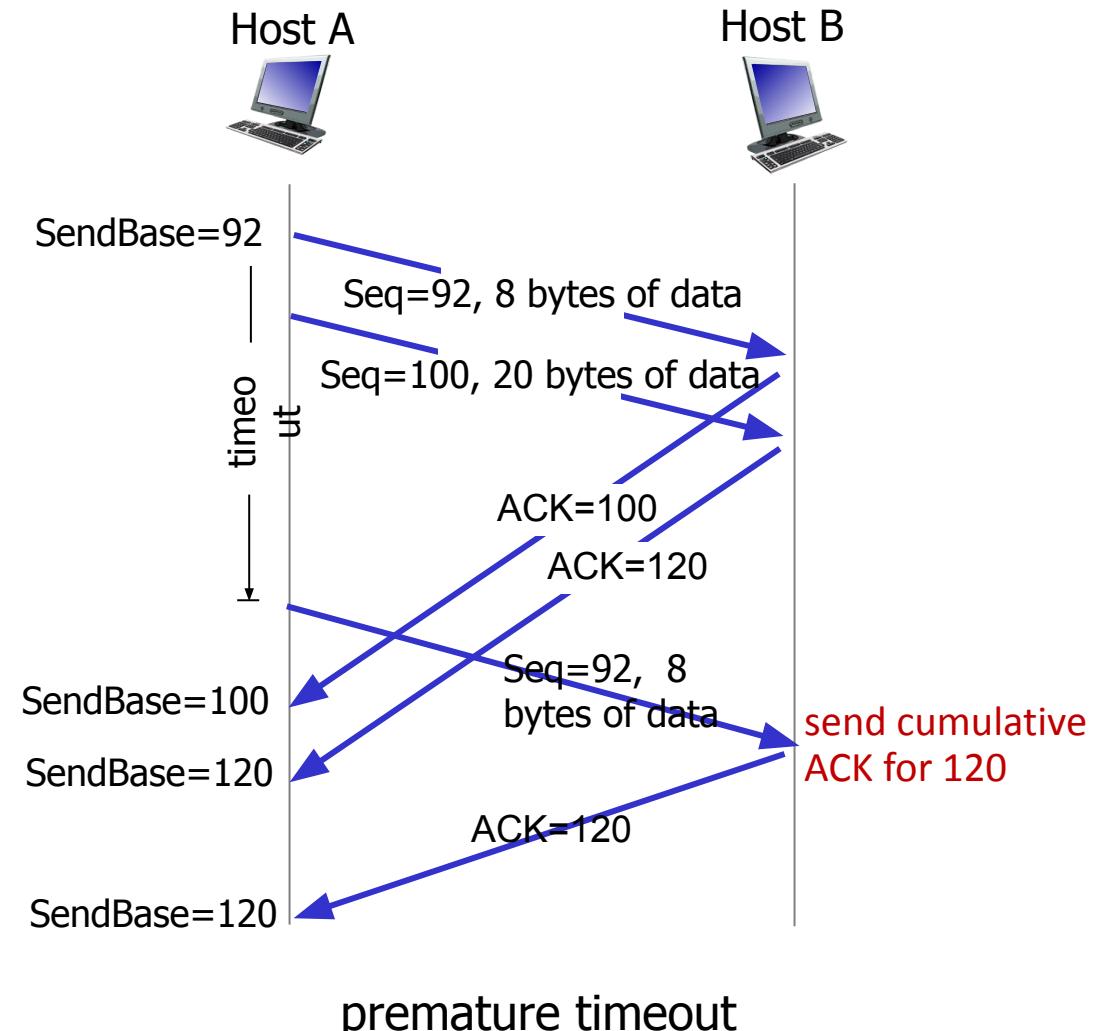
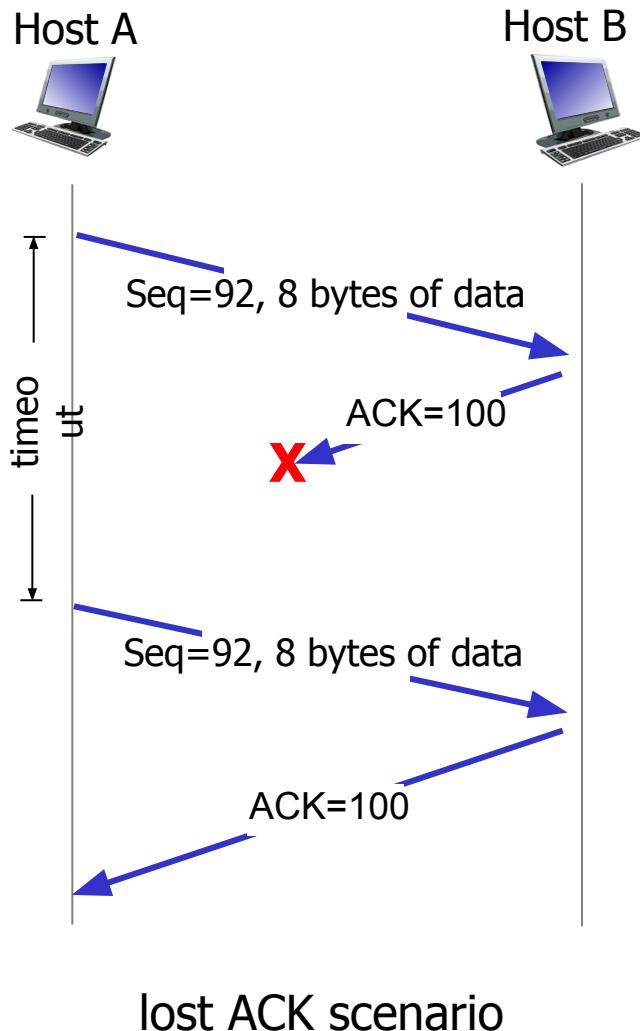
event: ACK received

- if ACK acknowledges previously unACKed segments
 - update what is known to be ACKed
 - start timer if there are still unACKed segments

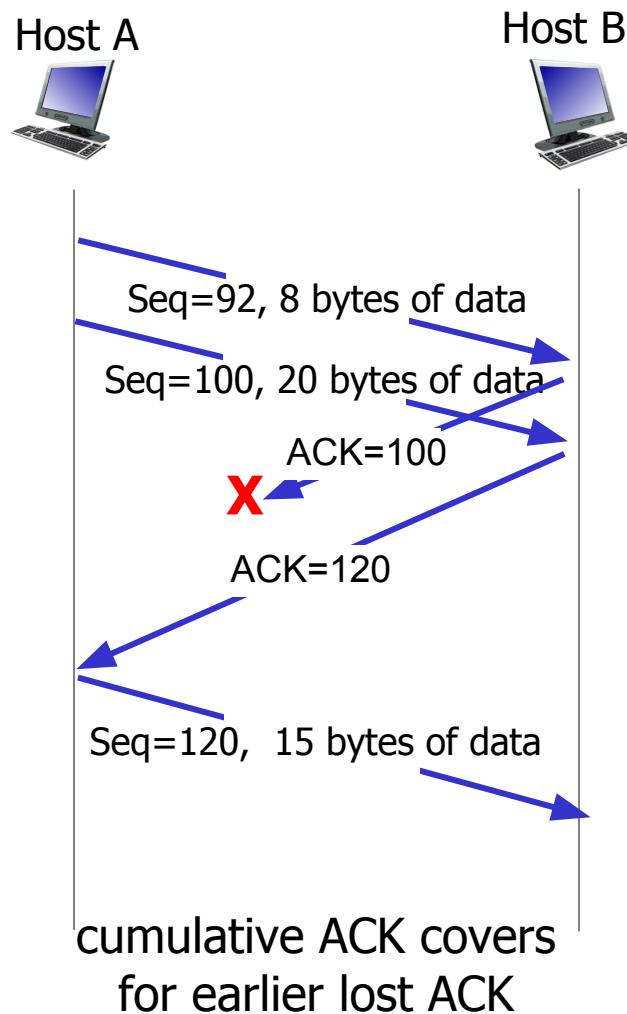
TCP Receiver: ACK generation [RFC 5681]



TCP: retransmission scenarios



TCP: retransmission scenarios



TCP fast retransmit

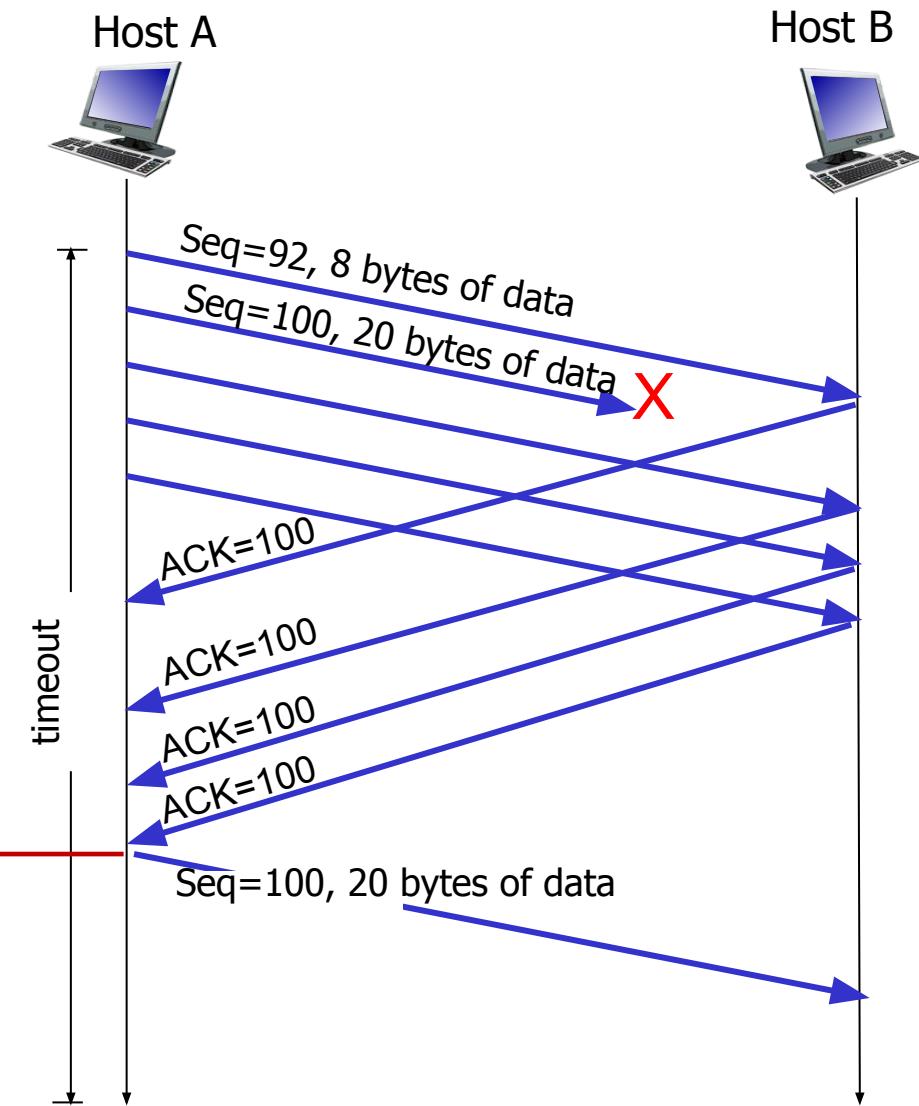
TCP fast retransmit

if sender receives 3 additional ACKs for same data (“triple duplicate ACKs”), resend unACKed segment with smallest seq #

- likely that unACKed segment lost, so don’t wait for timeout



Receipt of three duplicate ACKs indicates 3 segments received after a missing segment – lost segment is likely. So retransmit!



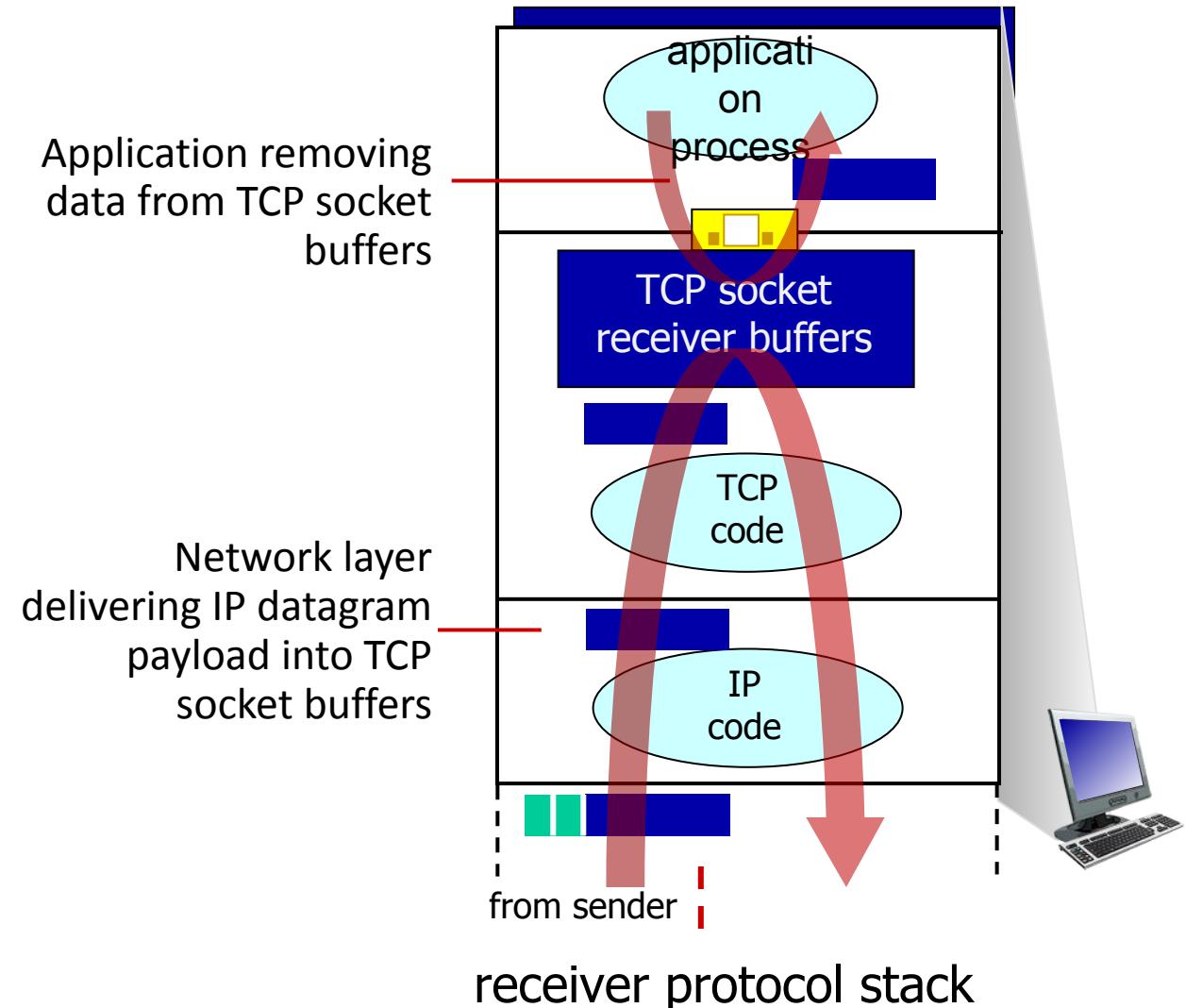
Chapter 3: roadmap

- Transport-layer services
- Multiplexing and demultiplexing
- Connectionless transport: UDP
- Principles of reliable data transfer
- **Connection-oriented transport: TCP**
 - segment structure
 - reliable data transfer
 - **flow control**
 - connection management
- Principles of congestion control
- TCP congestion control



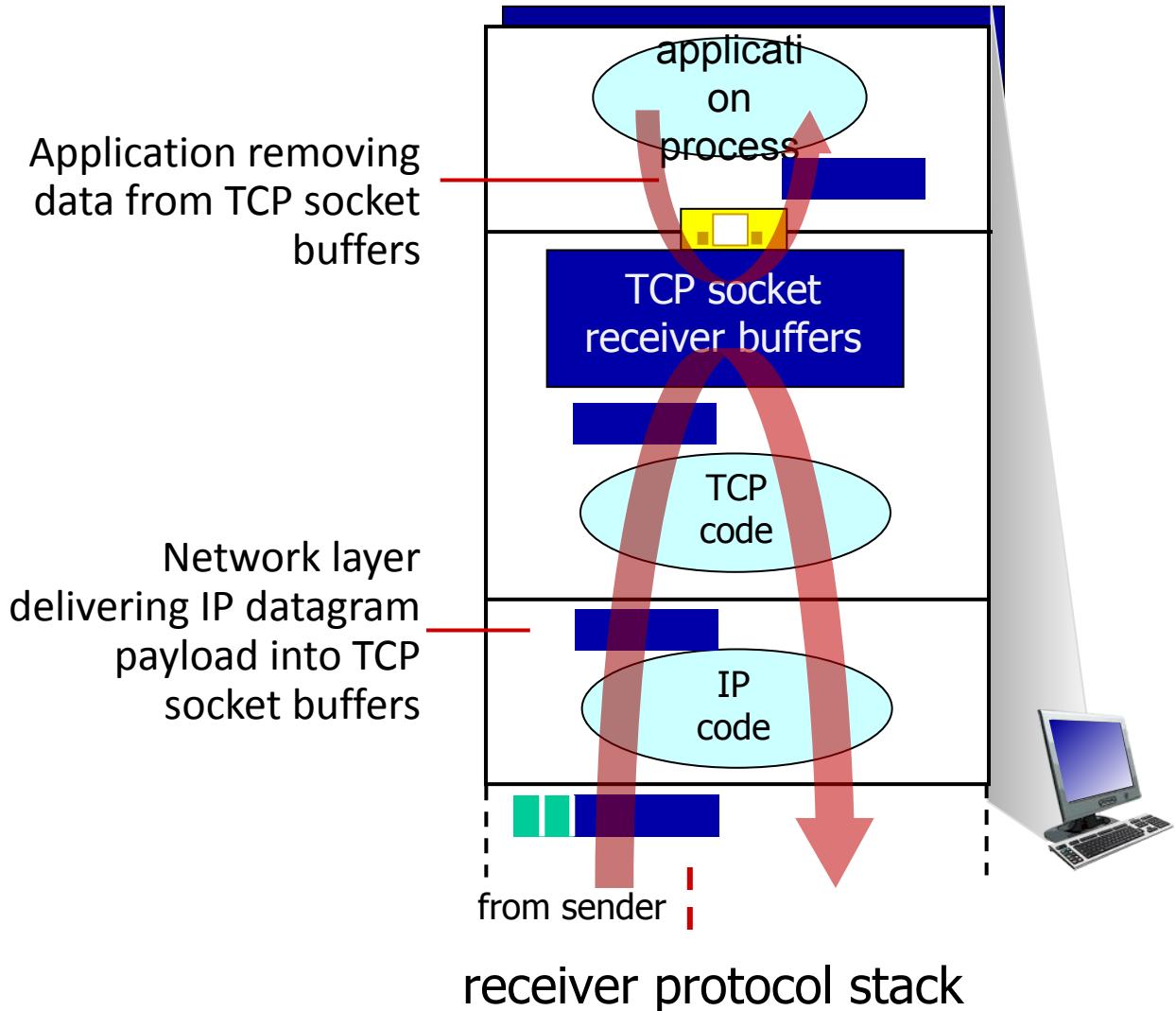
TCP flow control

Q: What happens if network layer delivers data faster than application layer removes data from socket buffers?



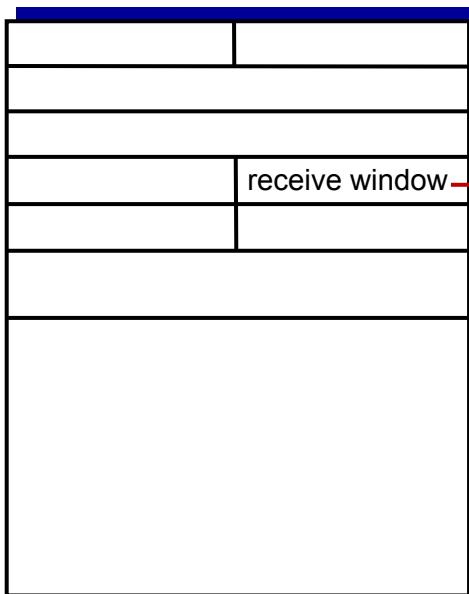
TCP flow control

Q: What happens if network layer delivers data faster than application layer removes data from socket buffers?



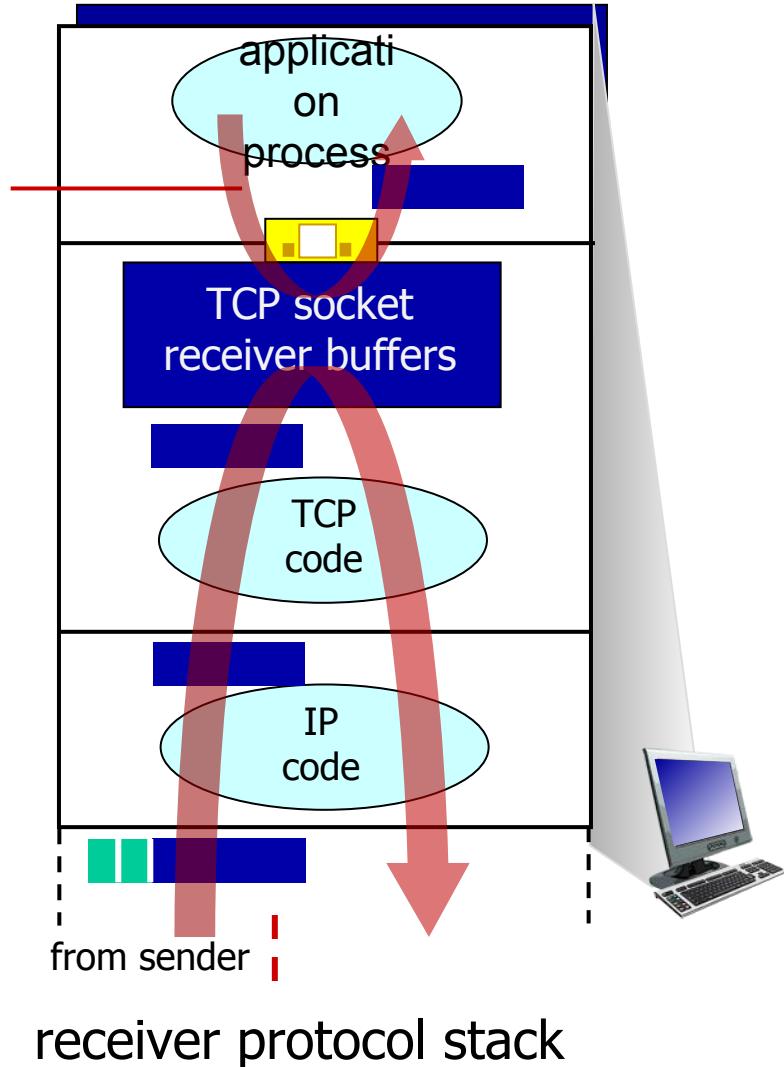
TCP flow control

Q: What happens if network layer delivers data faster than application layer removes data from socket buffers?



flow control: # bytes receiver willing to accept

Application removing data from TCP socket buffers



receiver protocol stack

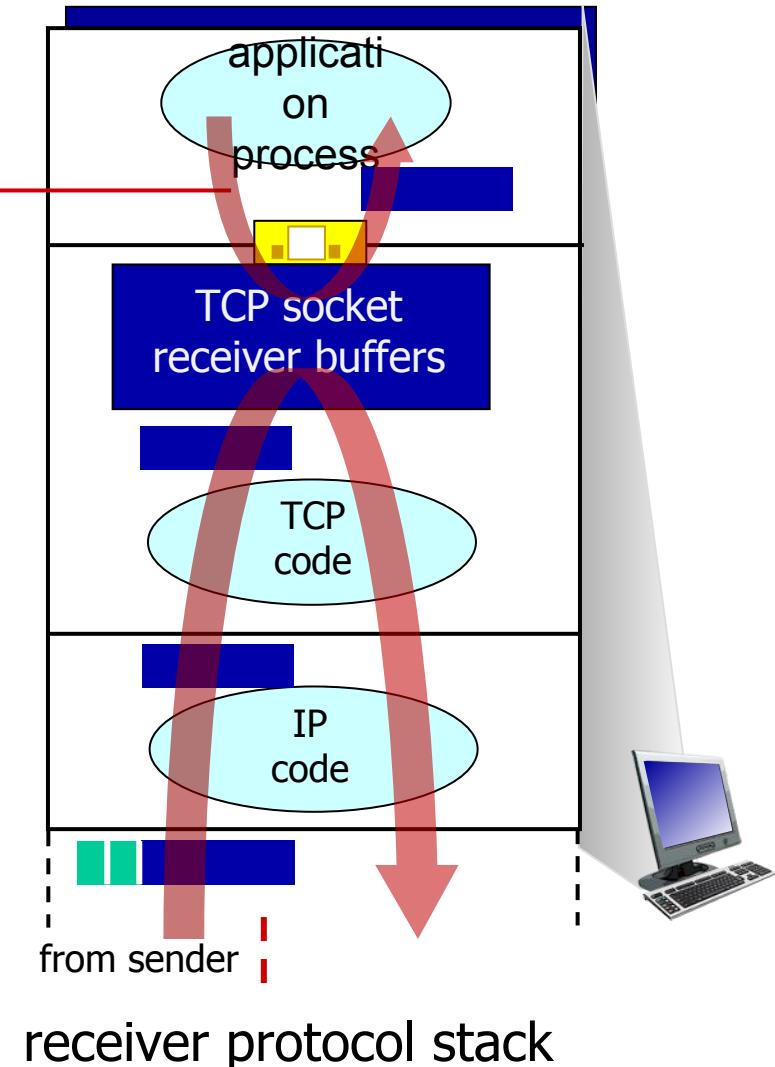
TCP flow control

Q: What happens if network layer delivers data faster than application layer removes data from socket buffers?

flow control

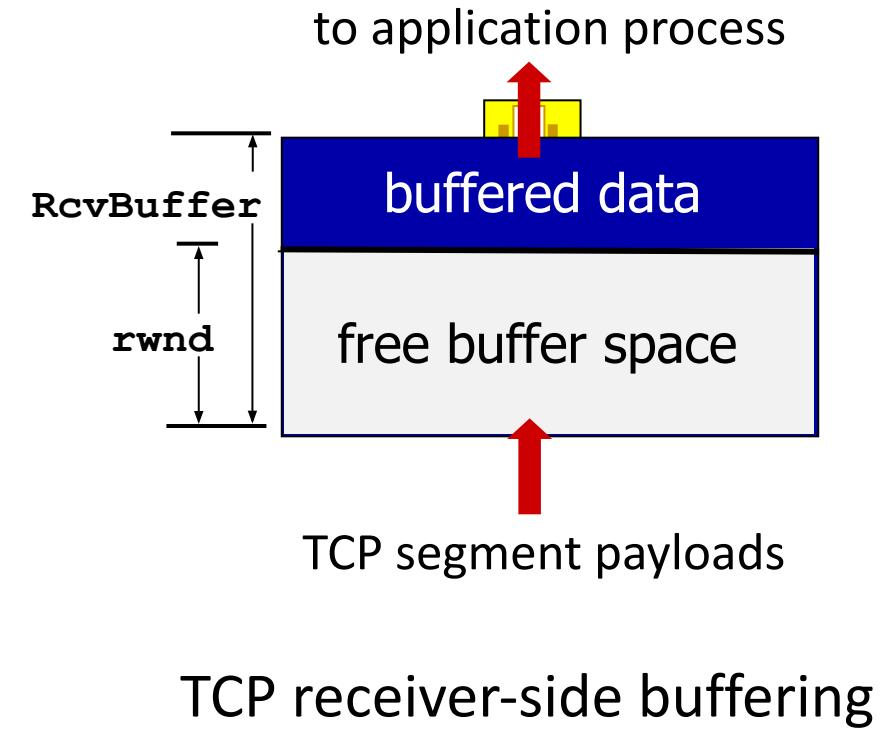
receiver controls sender, so sender won't overflow receiver's buffer by transmitting too much, too fast

Application removing data from TCP socket buffers



TCP flow control

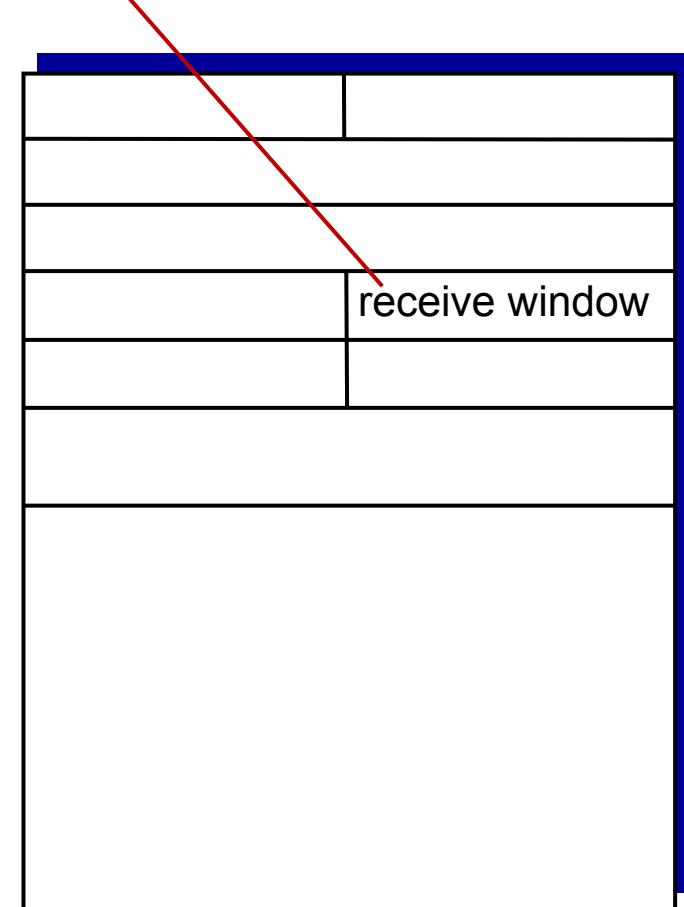
- TCP receiver “advertises” free buffer space in **rwnd** field in TCP header
 - **RcvBuffer** size set via socket options (typical default is 4096 bytes)
 - many operating systems autoadjust **RcvBuffer**
- sender limits amount of unACKed (“in-flight”) data to received **rwnd**
- guarantees receive buffer will not overflow



TCP flow control

- TCP receiver “advertises” free buffer space in **rwnd** field in TCP header
 - **RcvBuffer** size set via socket options (typical default is 4096 bytes)
 - many operating systems autoadjust **RcvBuffer**
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flow control: # bytes receiver willing to accept

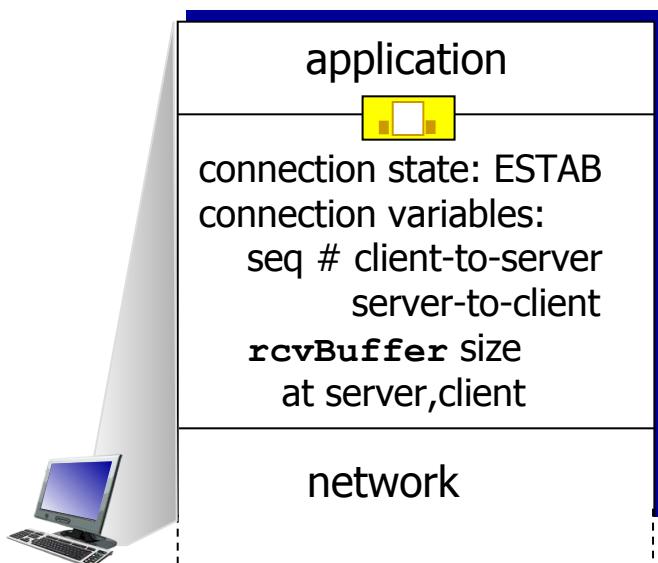


TCP segment format

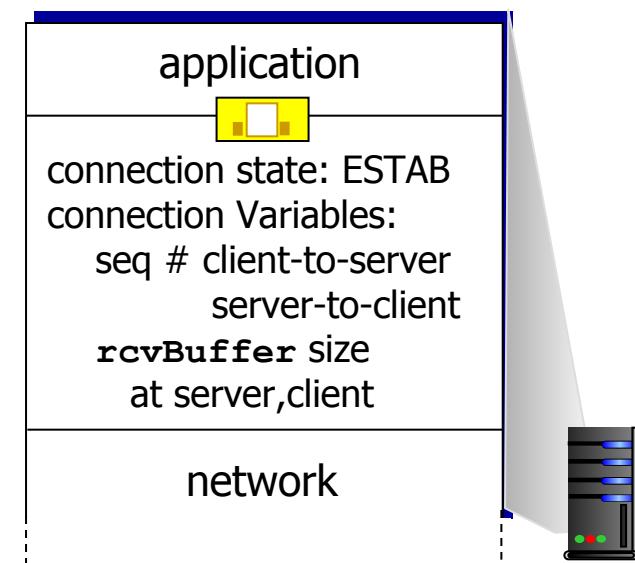
TCP connection management

before exchanging data, sender/receiver “handshake”:

- agree to establish connection (each knowing the other willing to establish connection)
- agree on connection parameters (e.g., starting seq #s)



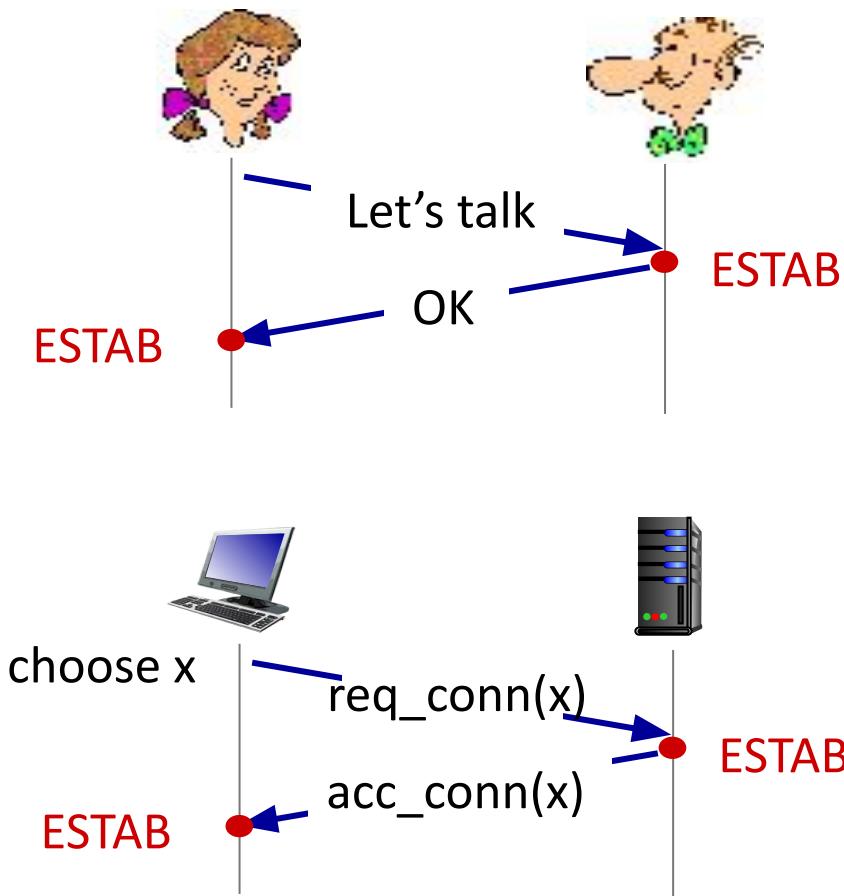
```
Socket clientSocket =  
    newSocket("hostname", "port number");
```



```
Socket connectionSocket =  
    welcomeSocket.accept();
```

Agreeing to establish a connection

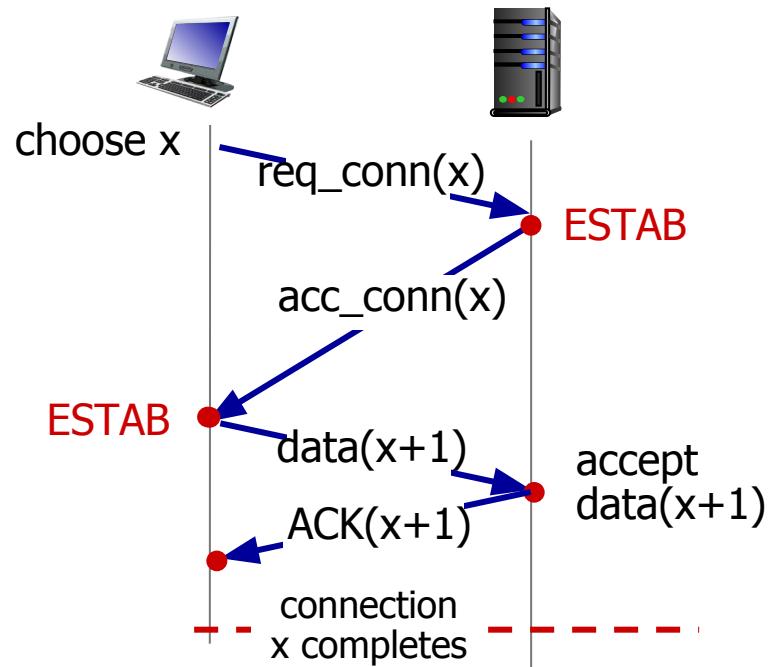
2-way handshake:



Q: will 2-way handshake always work in network?

- variable delays
- retransmitted messages (e.g. $\text{req_conn}(x)$) due to message loss
- message reordering
- can't “see” other side

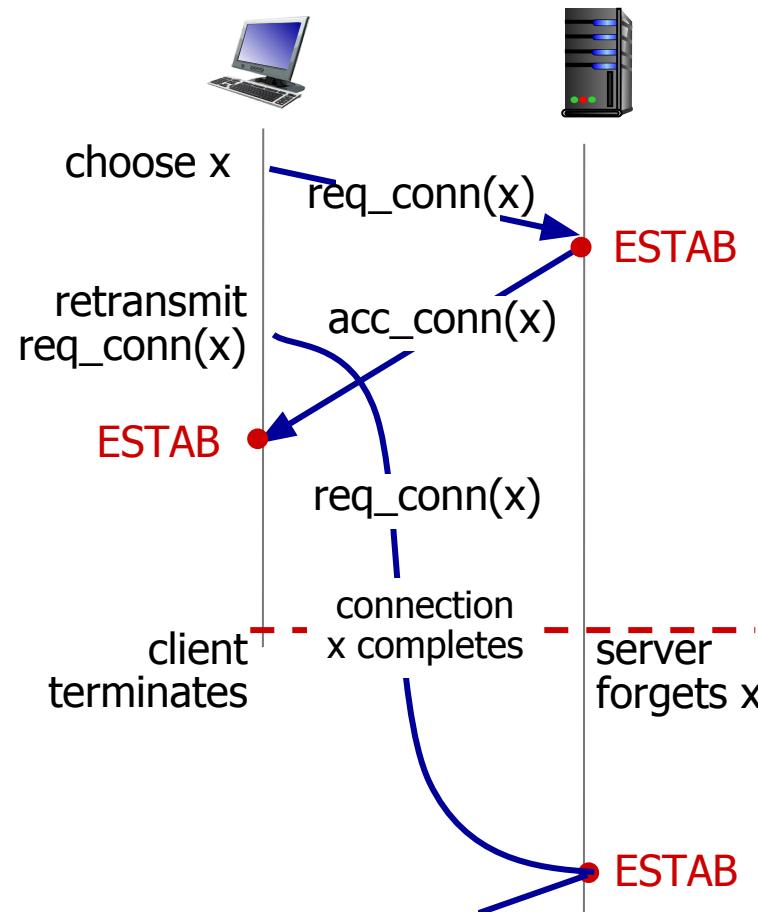
2-way handshake scenarios



No problem!

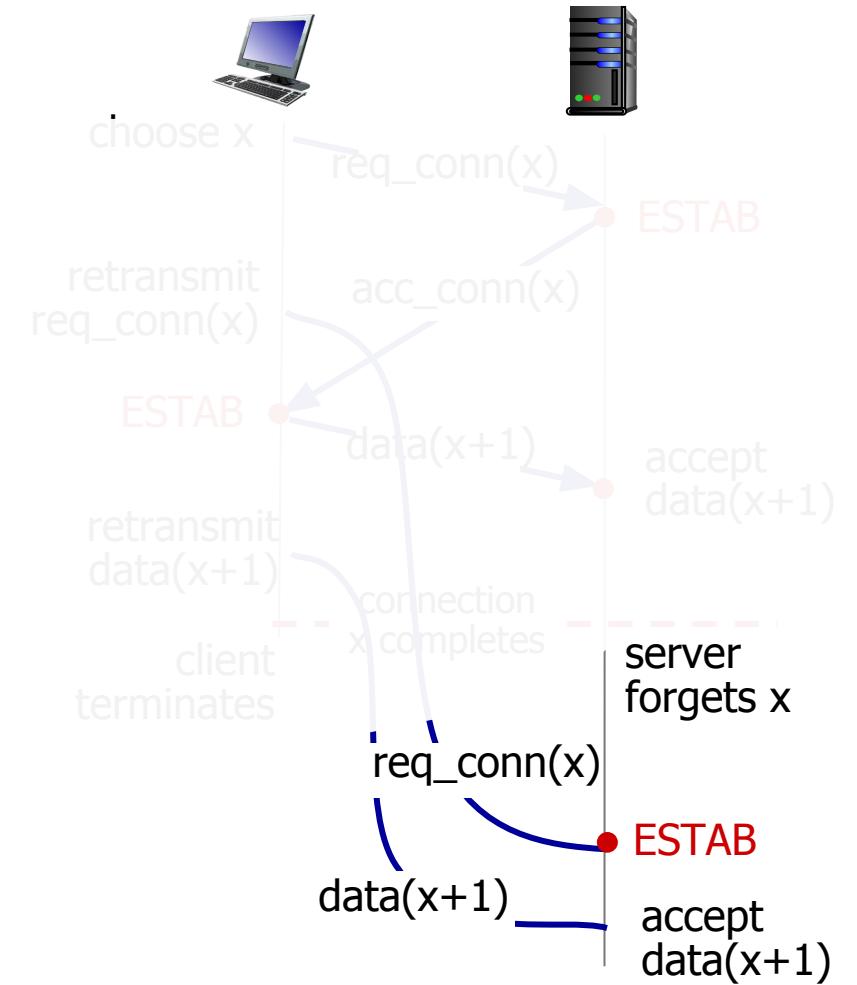


2-way handshake scenarios



Problem: half open connection! (no client)

2-way handshake scenarios



Problem: dup data accepted!

TCP 3-way handshake

Client state

```
clientSocket = socket(AF_INET, SOCK_STREAM)
```

LISTEN

```
clientSocket.connect((serverName, serverPort))
```

SYNSENT

choose init seq num, x
send TCP SYN msg



SYNbit=1, Seq=x

ESTAB

received SYNACK(x)
indicates server is live;
send ACK for SYNACK;
this segment may contain
client-to-server data

SYNbit=1, Seq=y
ACKbit=1; ACKnum=x+1

ACKbit=1, ACKnum=y+1

Server state

```
serverSocket = socket(AF_INET, SOCK_STREAM)  
serverSocket.bind(('', serverPort))  
serverSocket.listen(1)  
connectionSocket, addr = serverSocket.accept()
```

LISTEN

SYN RCVD

choose init seq num, y
send TCP SYNACK
msg, acking SYN

received ACK(y)
indicates client is live

ESTAB

A human 3-way handshake protocol



Closing a TCP connection

- client, server each close their side of connection
 - send TCP segment with FIN bit = 1
- respond to received FIN with ACK
 - on receiving FIN, ACK can be combined with own FIN
- simultaneous FIN exchanges can be handled

Chapter 3: 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



Principles of congestion control

Congestion:

- informally: “too many sources sending too much data too fast for *network* to handle”
- manifestations:
 - long delays (queueing in router buffers)
 - packet loss (buffer overflow at routers)
- different from flow control!
- a top-10 problem!



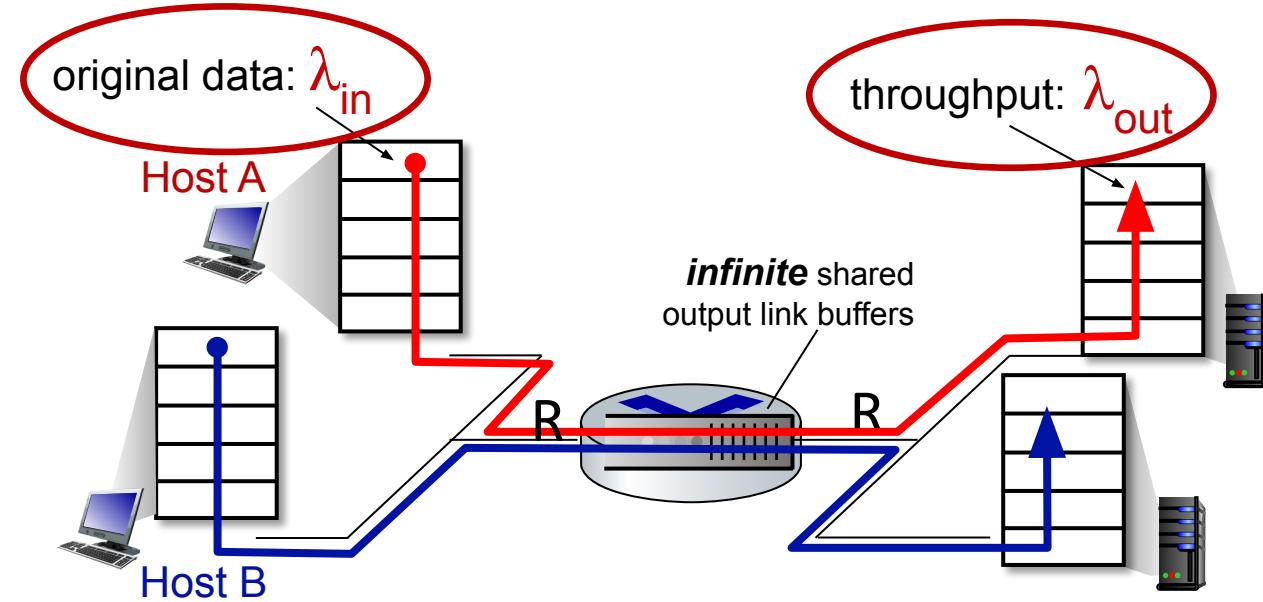
congestion control:
too many senders,
sending too fast

flow control: one sender
too fast for one receiver

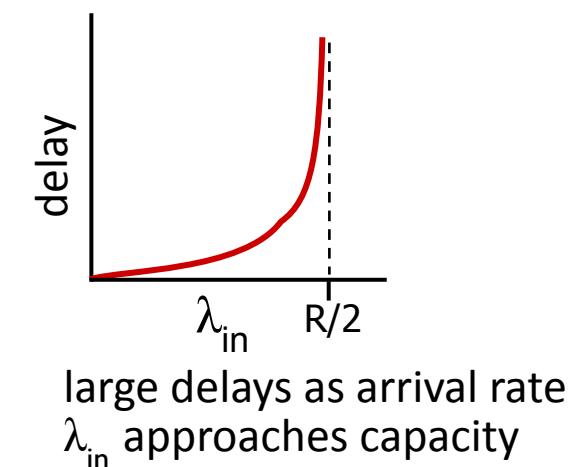
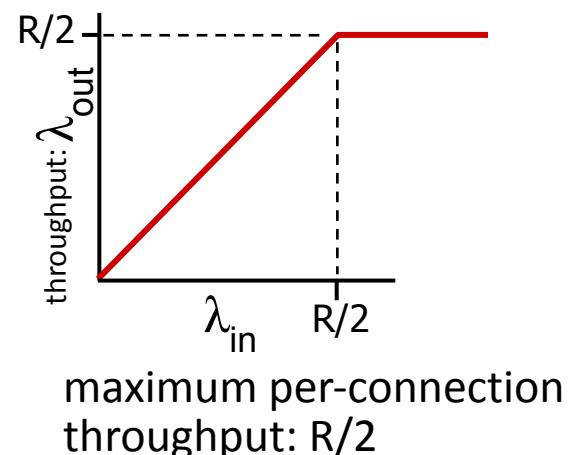
Causes/costs of congestion: scenario 1

Simplest scenario:

- one router, infinite buffers
- input, output link capacity: R
- two flows
- no retransmissions needed

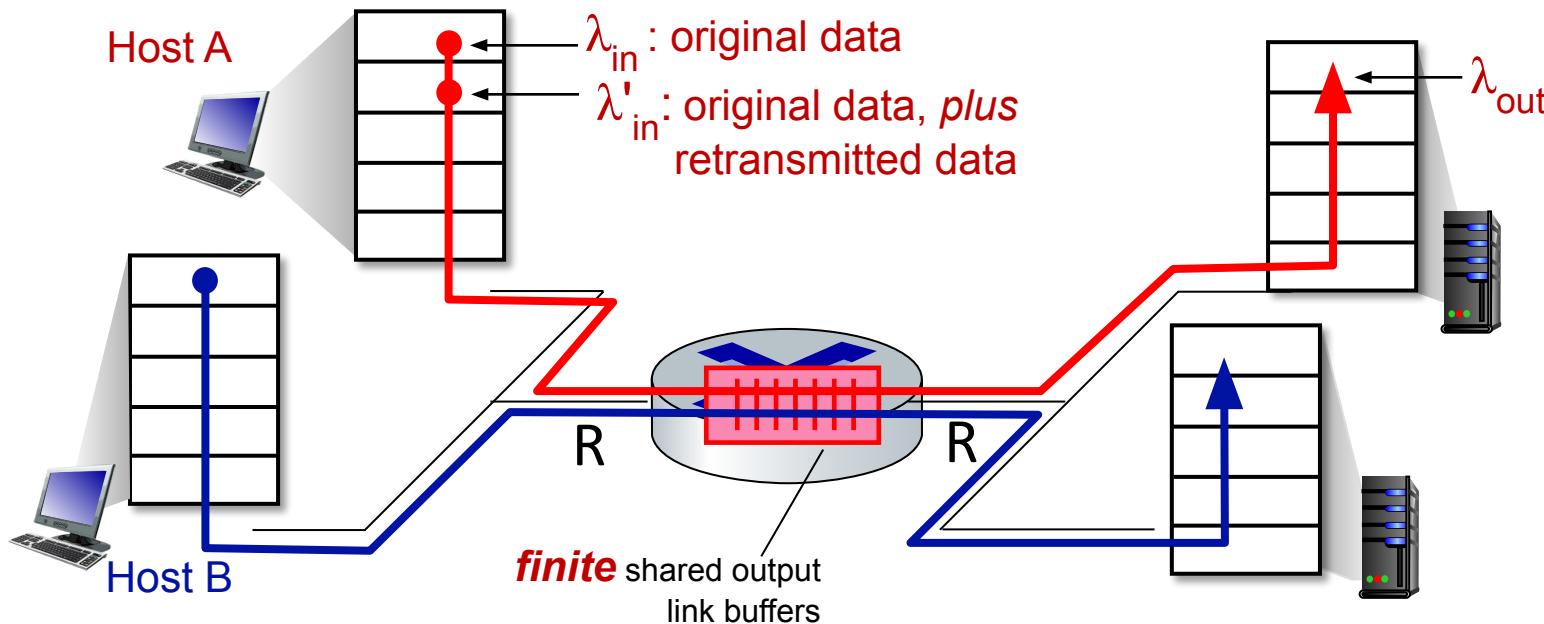


Q: What happens as arrival rate λ_{in} approaches $R/2$?



Causes/costs of congestion: scenario 2

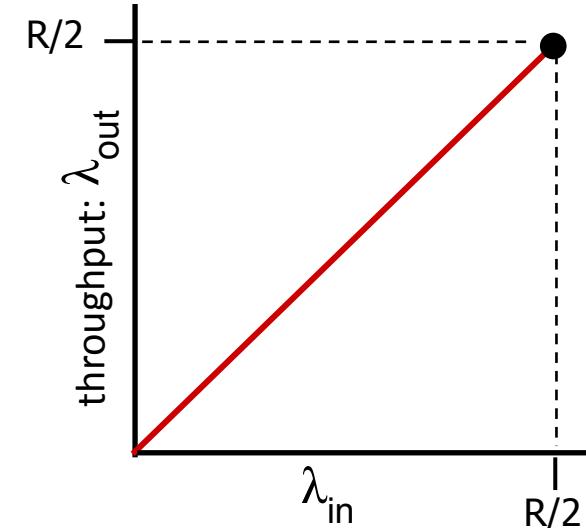
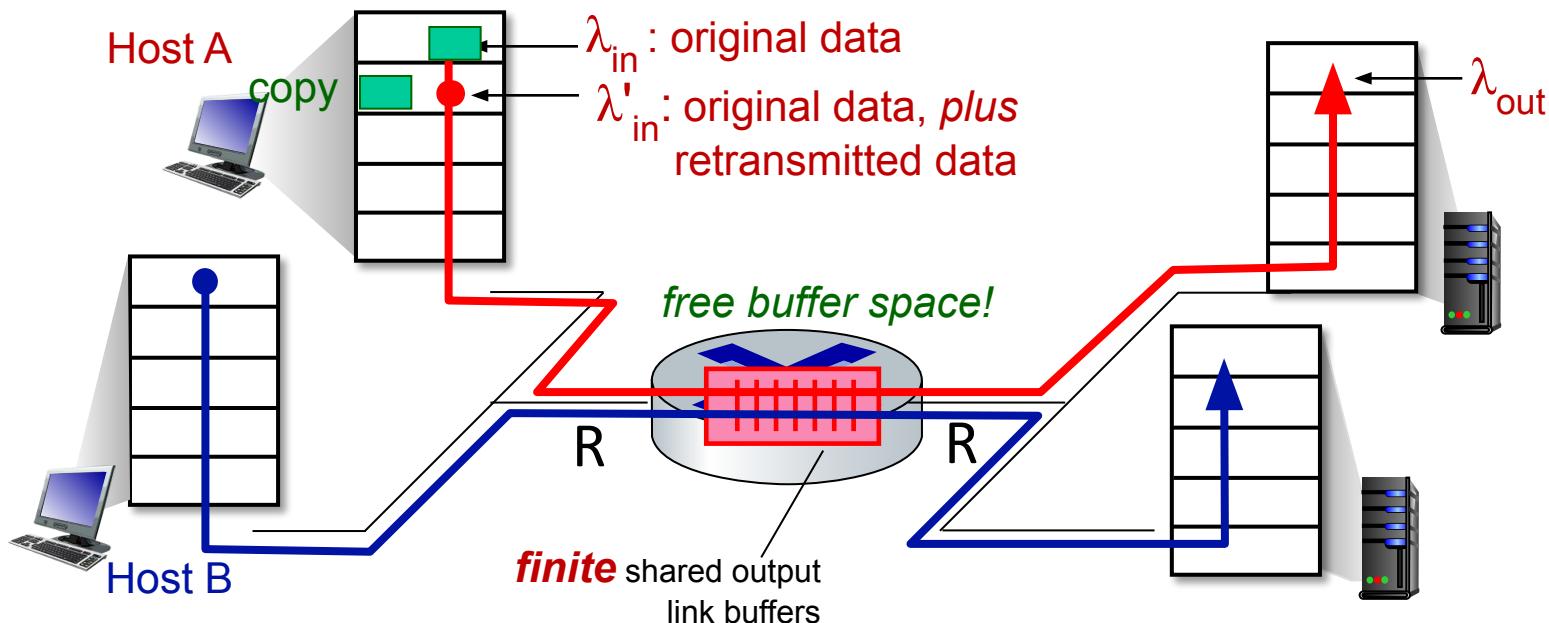
- one router, *finite* buffers
- sender retransmits lost, timed-out packet
 - application-layer input = application-layer output: $\lambda_{in} = \lambda_{out}$
 - transport-layer input includes *retransmissions* : $\lambda'_{in} \geq \lambda_{in}$



Causes/costs of congestion: scenario 2

Idealization: perfect knowledge

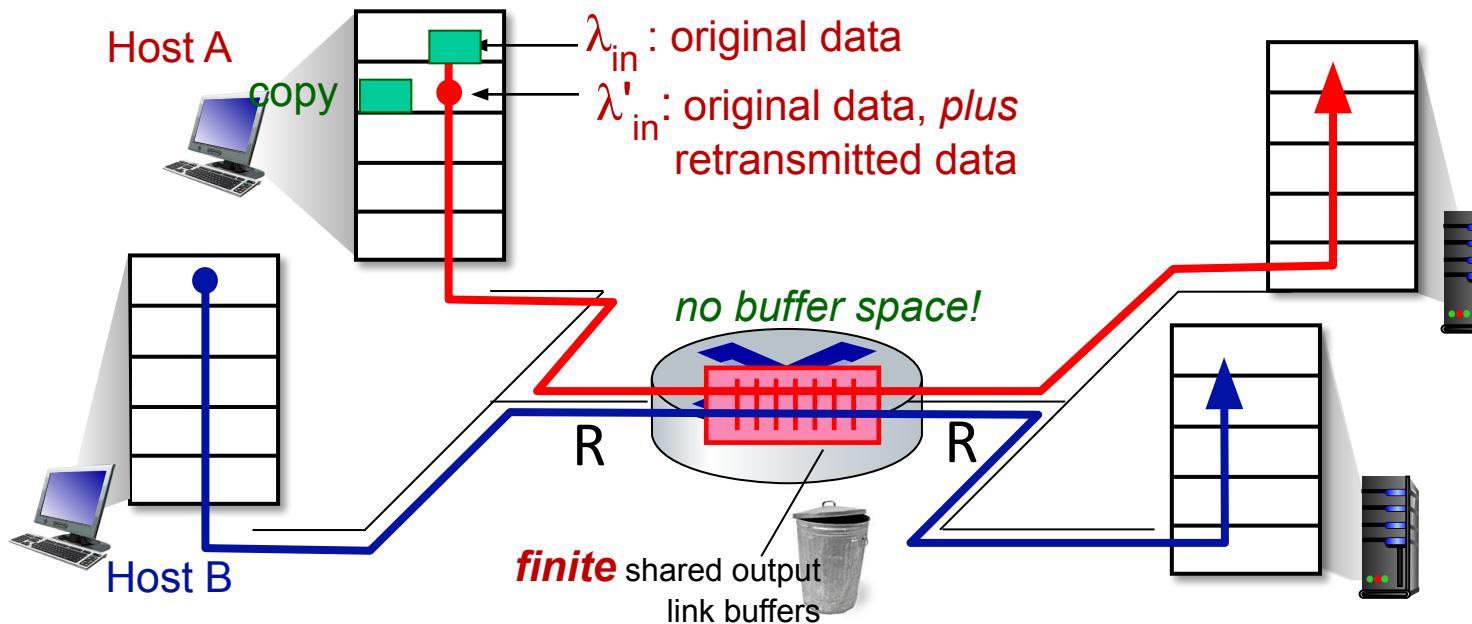
- sender sends only when router buffers available



Causes/costs of congestion: scenario 2

Idealization: *some* perfect knowledge

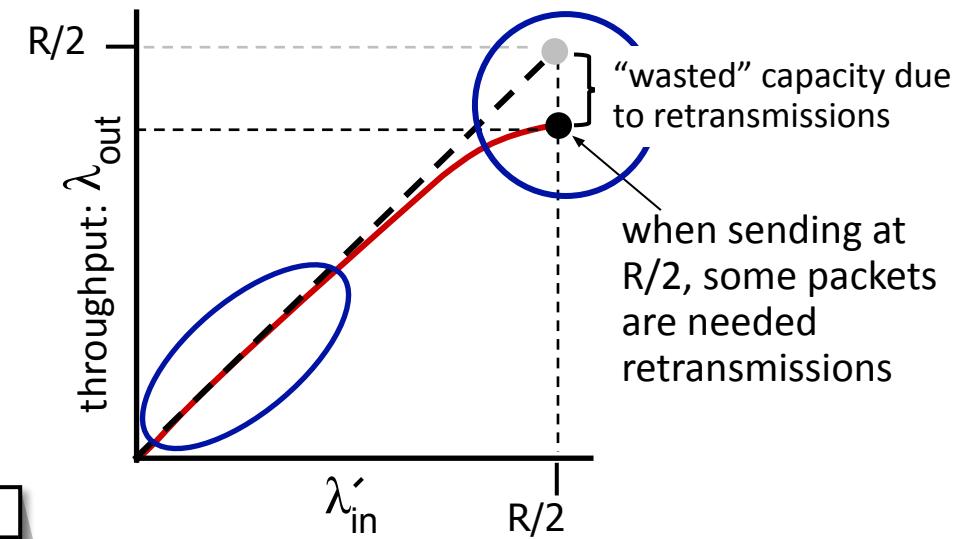
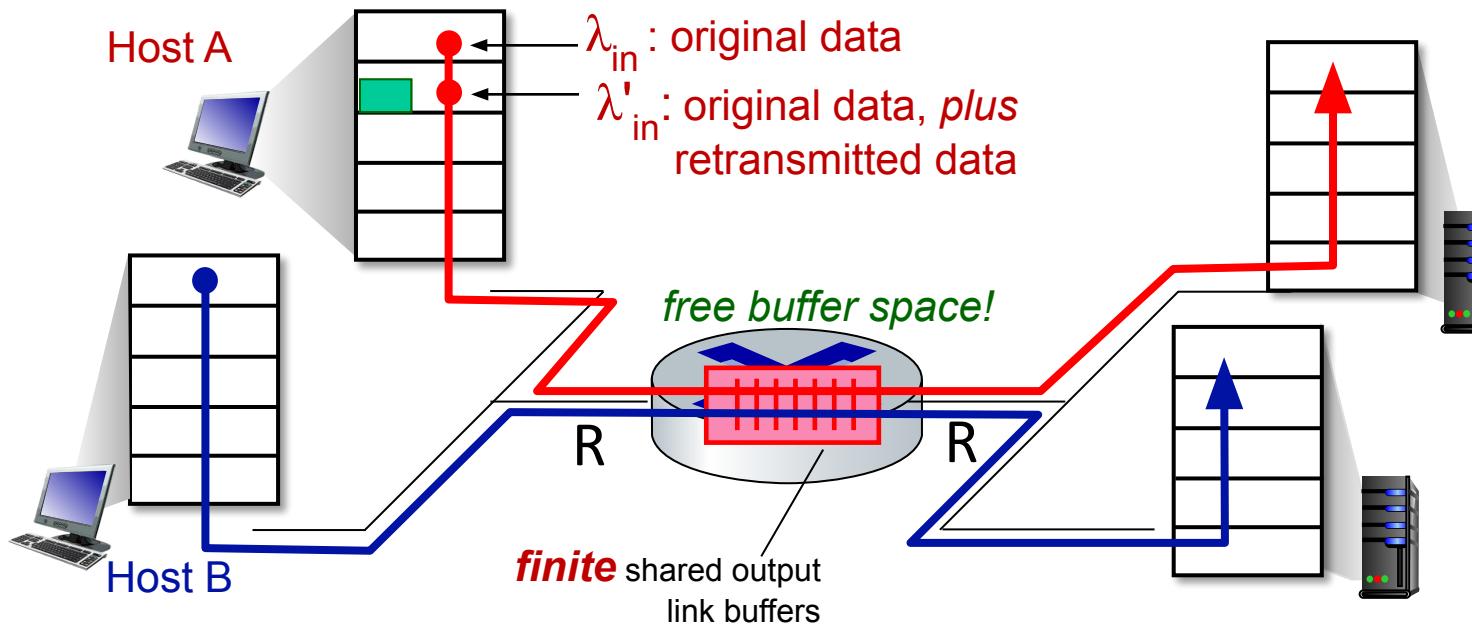
- packets can be lost (dropped at router) due to full buffers
- sender knows when packet has been dropped: only resends if packet *known* to be lost



Causes/costs of congestion: scenario 2

Idealization: *some* perfect knowledge

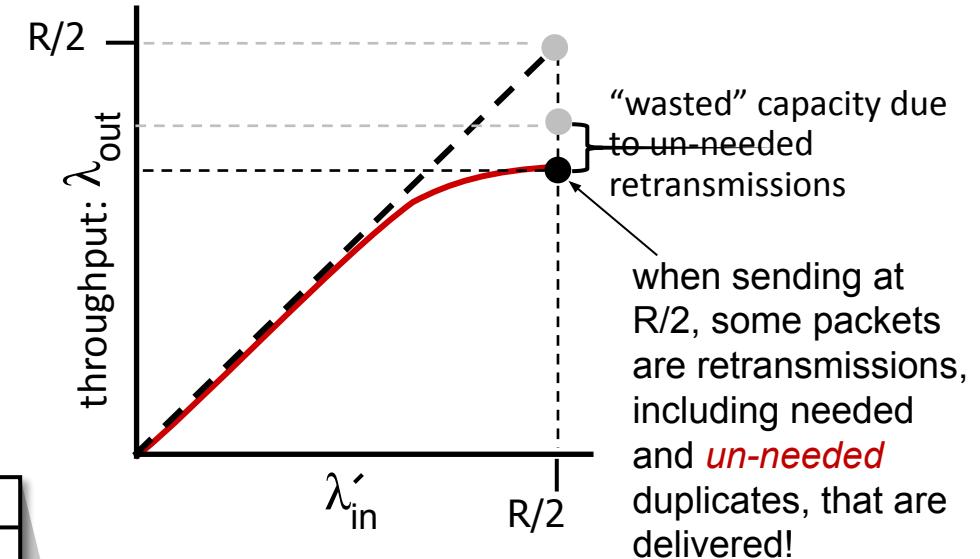
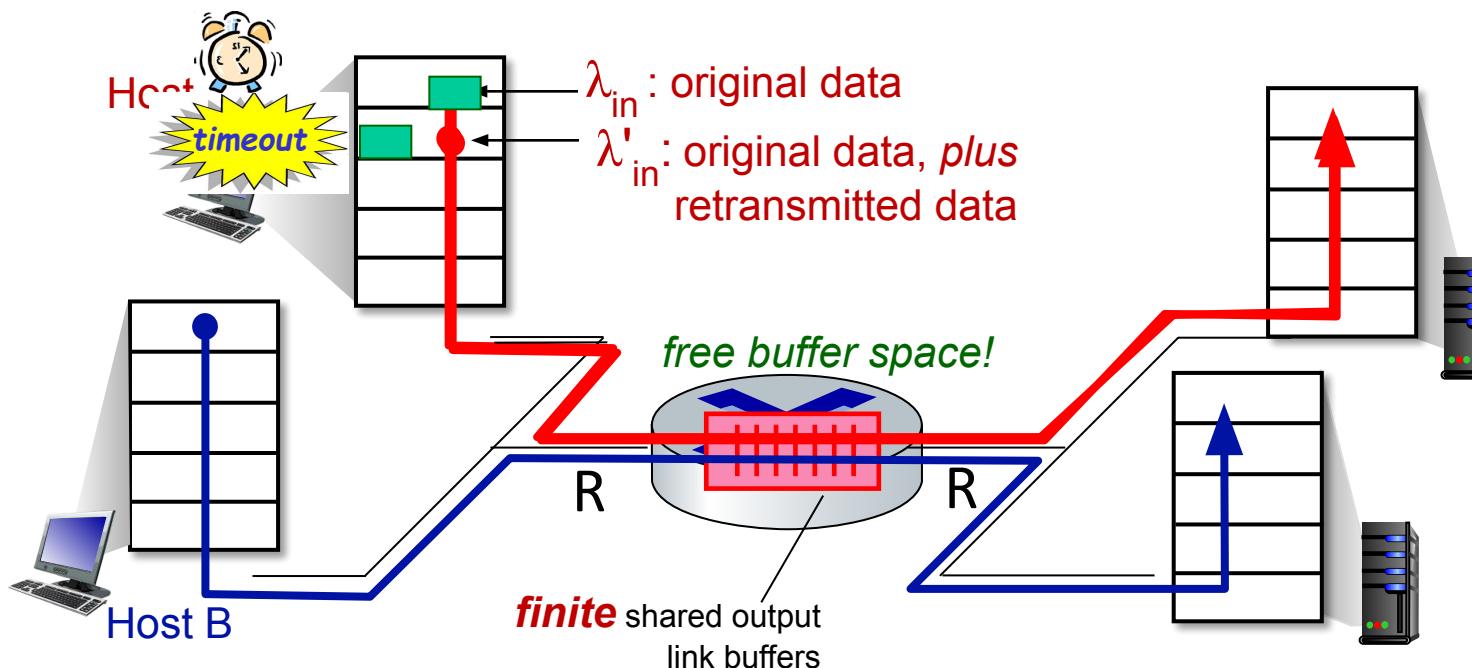
- packets can be lost (dropped at router) due to full buffers
- sender knows when packet has been dropped: only resends if packet *known* to be lost



Causes/costs of congestion: scenario 2

Realistic scenario: *un-needed duplicates*

- packets can be lost, dropped at router due to full buffers – requiring retransmissions
- but sender times can time out prematurely, sending *two* copies, *both* of which are delivered



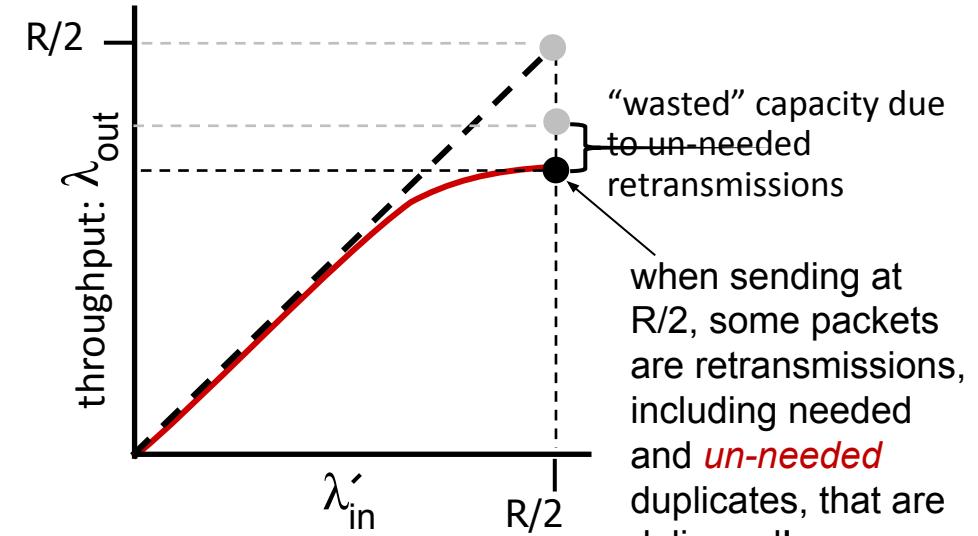
Causes/costs of congestion: scenario 2

Realistic scenario: *un-needed duplicates*

- packets can be lost, dropped at router due to full buffers – requiring retransmissions
- but sender times can time out prematurely, sending *two* copies, *both* of which are delivered

“costs” of congestion:

- more work (retransmission) for given receiver throughput
- unneeded retransmissions: link carries multiple copies of a packet
 - decreasing maximum achievable throughput

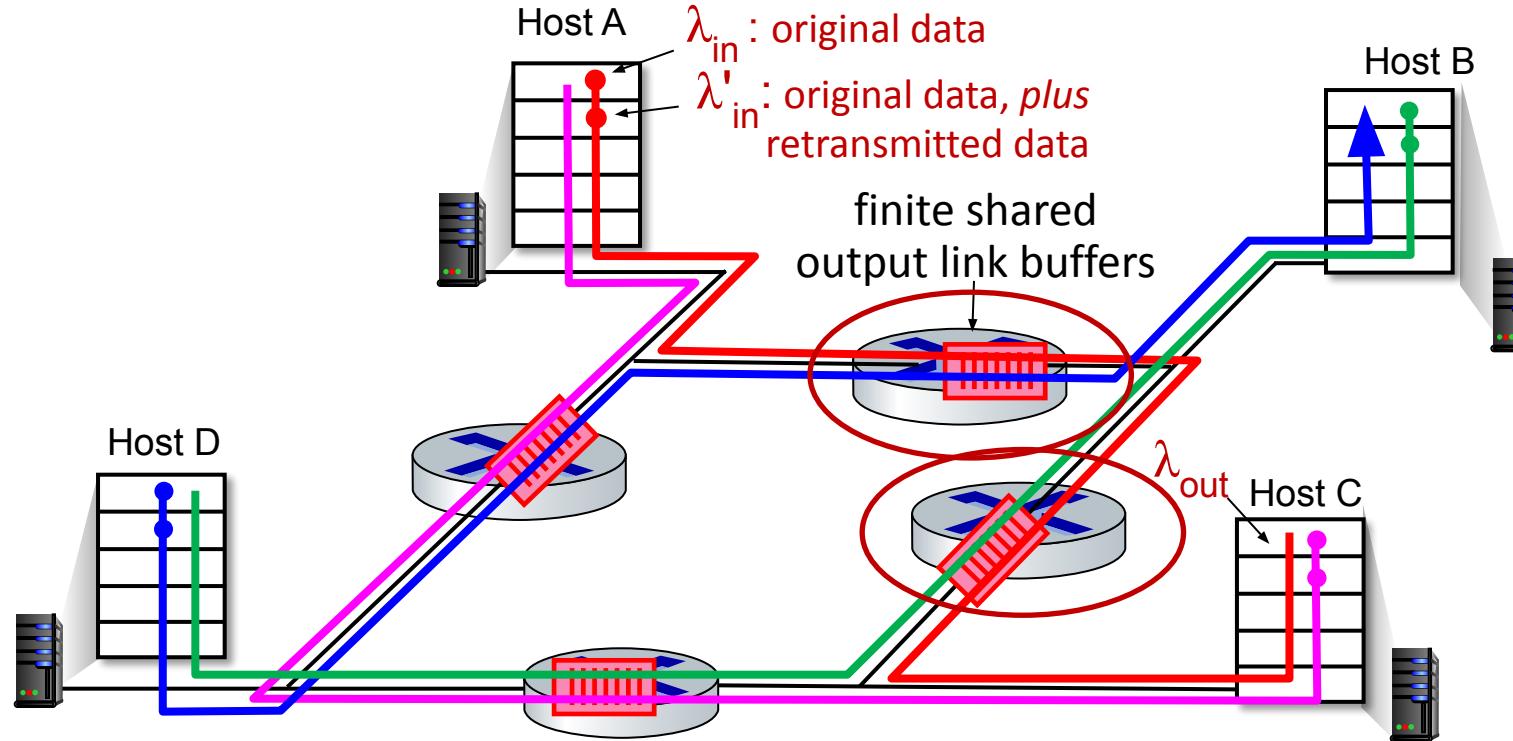


Causes/costs of congestion: scenario 3

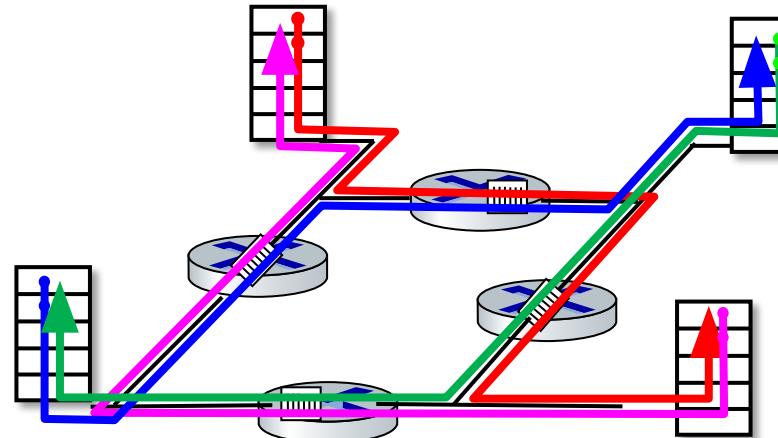
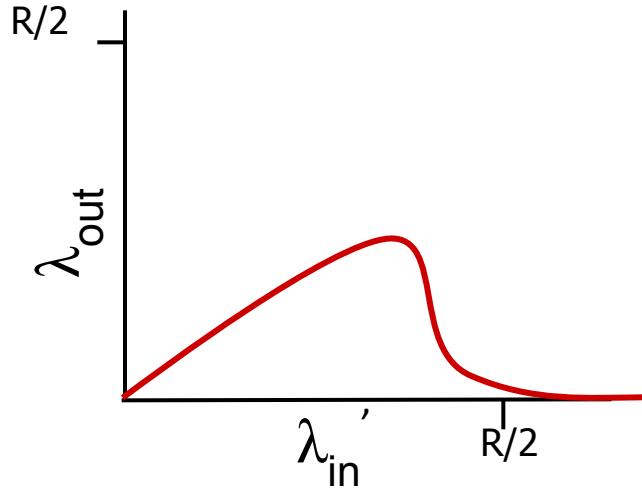
- four senders
- multi-hop paths
- timeout/retransmit

Q: what happens as λ_{in} and λ'_{in} increase ?

A: as red λ_{in} increases, all arriving blue pkts at upper queue are dropped, blue throughput $\rightarrow 0$



Causes/costs of congestion: scenario 3

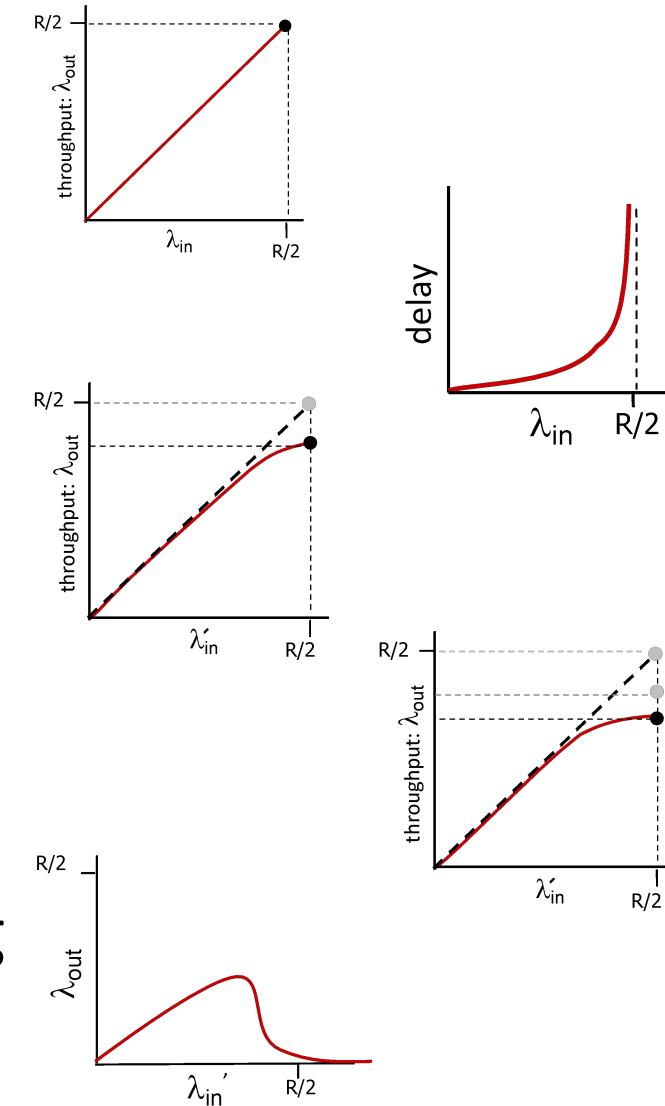


another “cost” of congestion:

- when packet dropped, any upstream transmission capacity and buffering used for that packet was wasted!

Causes/costs of congestion: insights

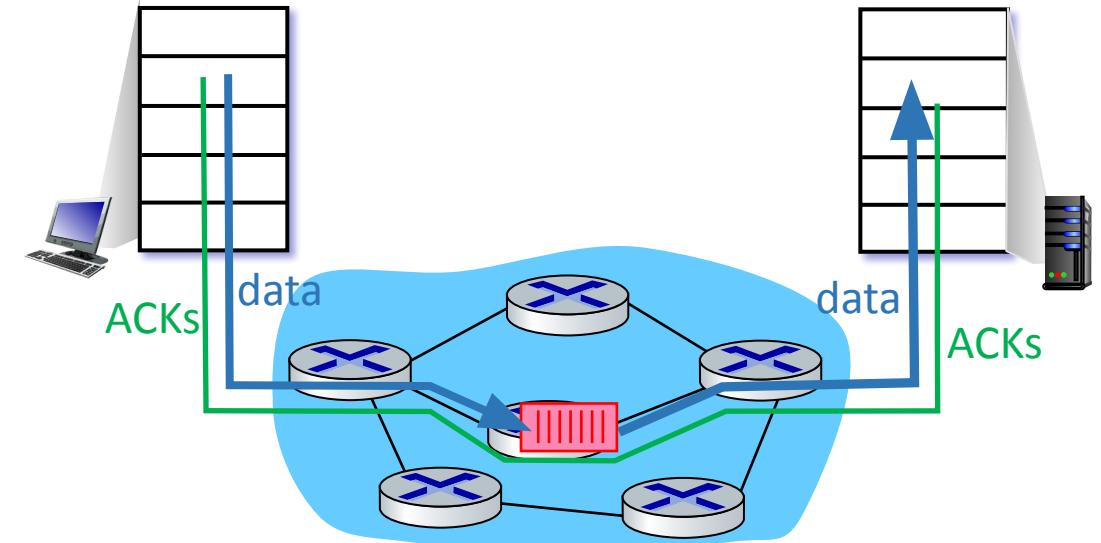
- throughput can never exceed capacity
- delay increases as capacity approached
- loss/retransmission decreases effective throughput
- un-needed duplicates further decreases effective throughput
- upstream transmission capacity / buffering wasted for packets lost downstream



Approaches towards congestion control

End-end congestion control:

- no explicit feedback from network
- congestion *inferred* from observed loss, delay
- approach taken by TCP



Approaches towards congestion control

Network-assisted congestion control:

- routers provide *direct* feedback to sending/receiving hosts with flows passing through congested router
- may indicate congestion level or explicitly set sending rate
- TCP ECN, ATM, DECbit protocols

