Chapter 3: Transport Layer

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Chapter 3: roadmap

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- 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

Transport services and protocols

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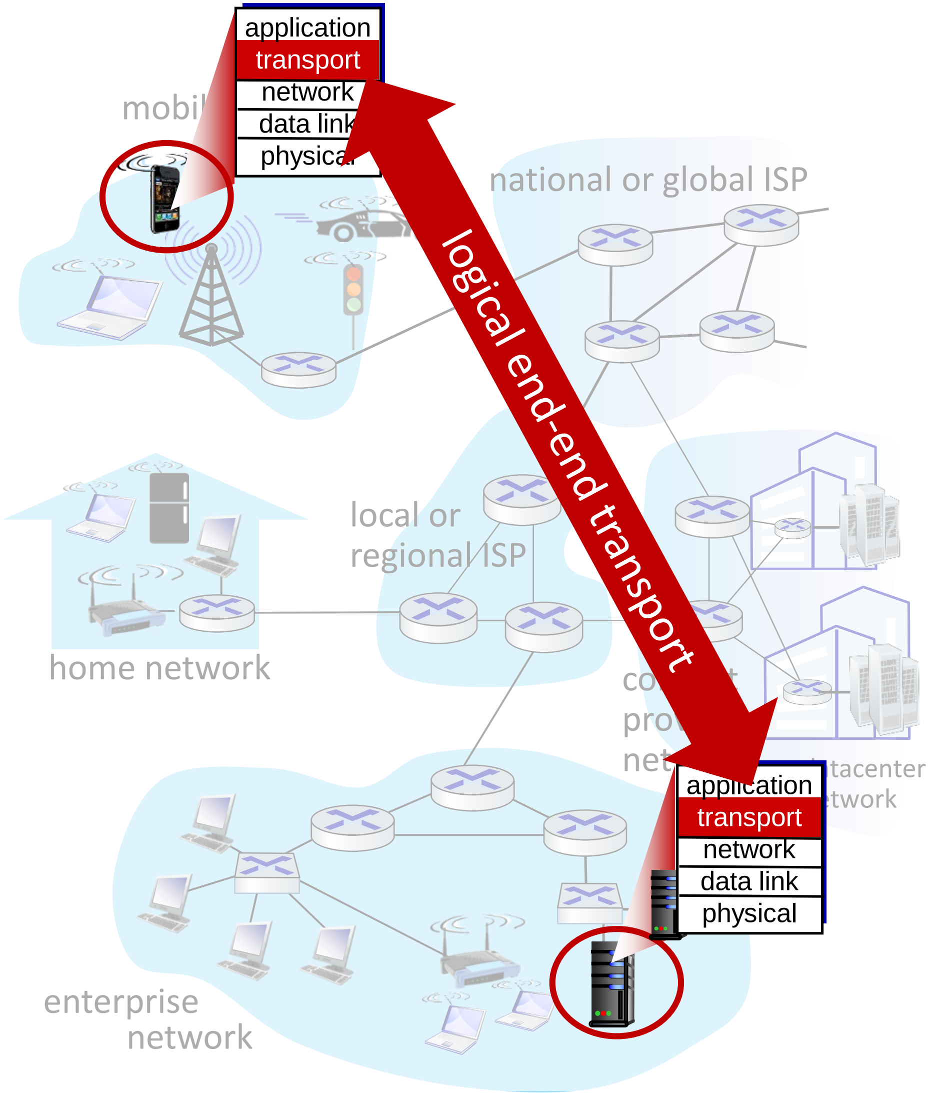
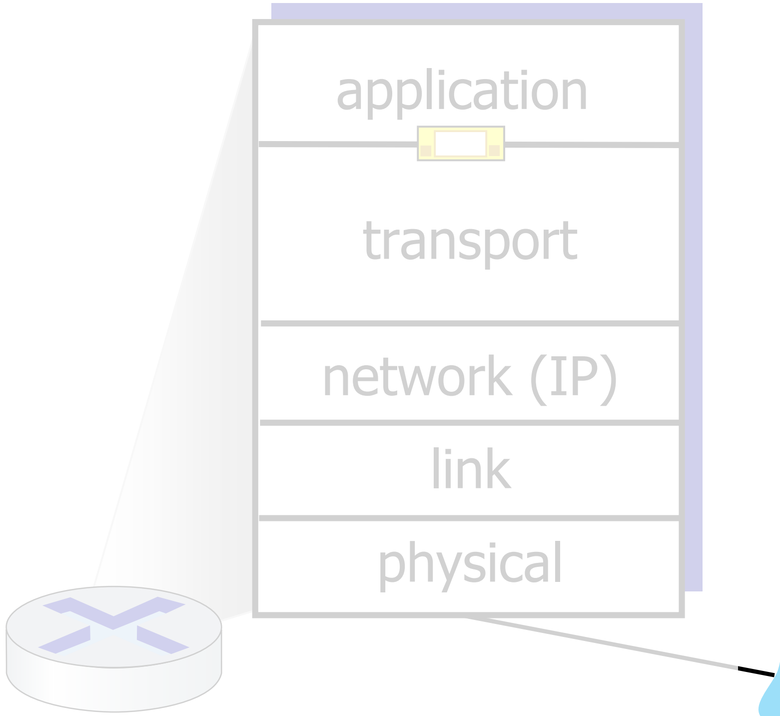
- provide *logical communication* between application processes running on different hosts
- transport protocols actions in end systems:
  - sender: breaks application messages into *segments*, passes to network layer
  - receiver: reassembles segments into messages, passes to application layer
- two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

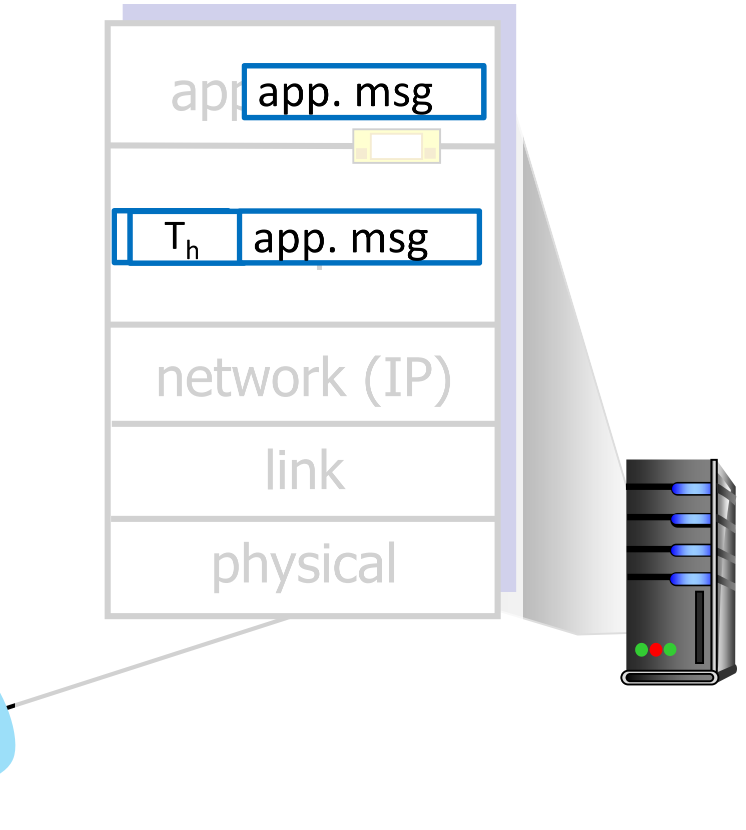


Transport Layer Actions

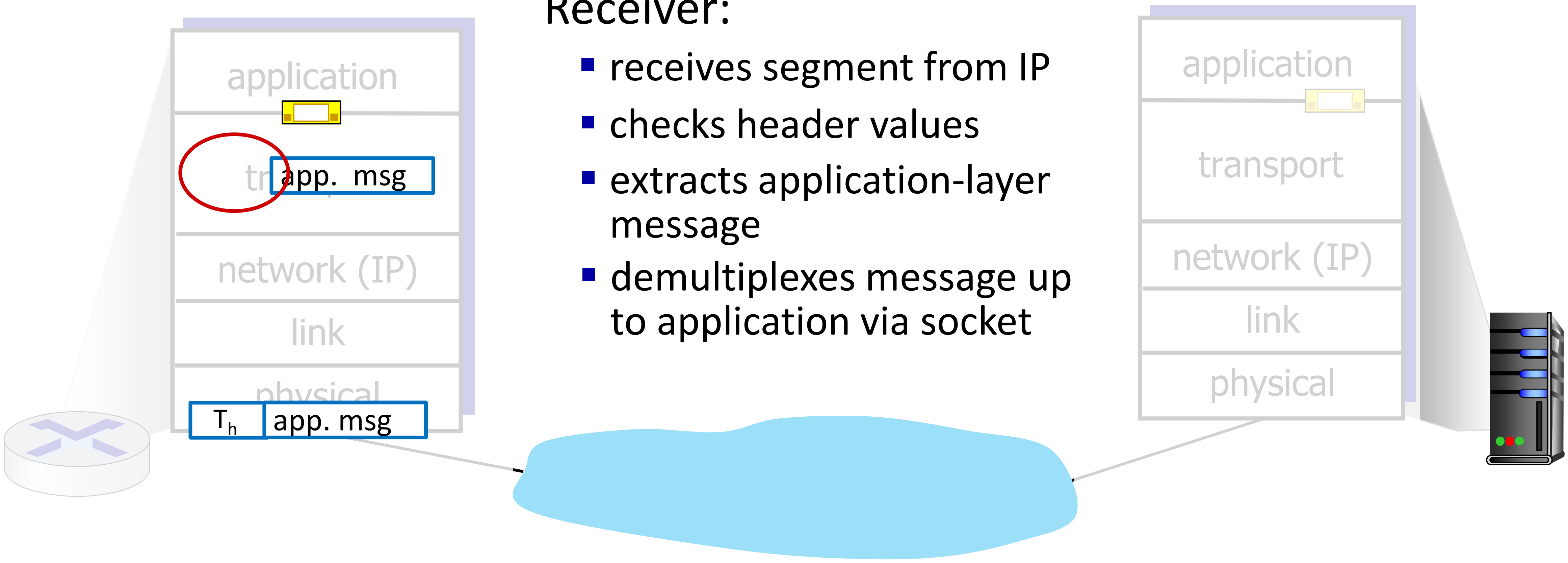


Sender:

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



Transport Layer Actions



Receiver:

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

Two principal Internet transport protocols

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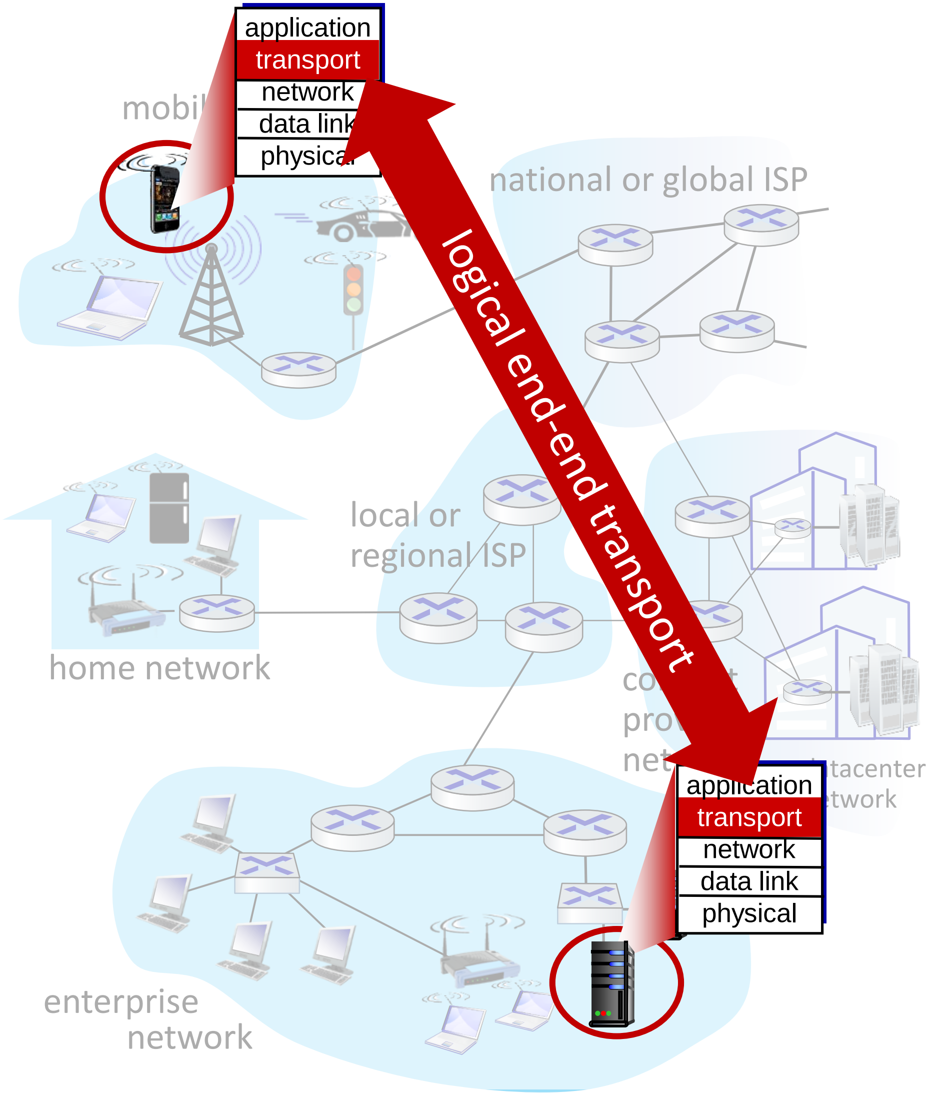
- **TCP:** Transmission Control Protocol
  - reliable, in-order delivery
  - congestion control
  - flow control
  - connection setup
- **UDP:** User Datagram Protocol
  - unreliable, unordered delivery
- services not available:
  - delay guarantees
  - bandwidth guarantees



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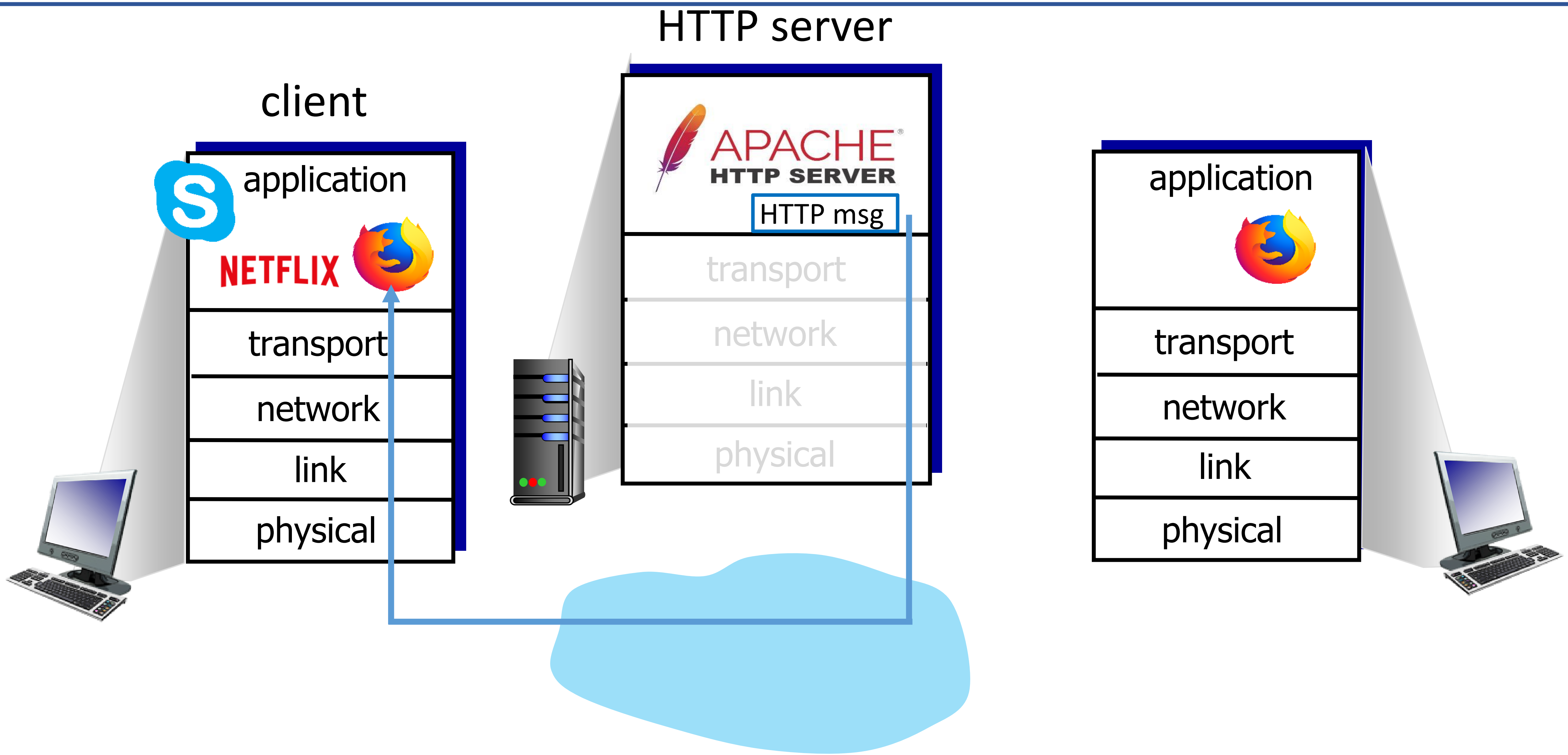
HTTP server
client
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NETFLIX
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APACHE
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HTTP msg
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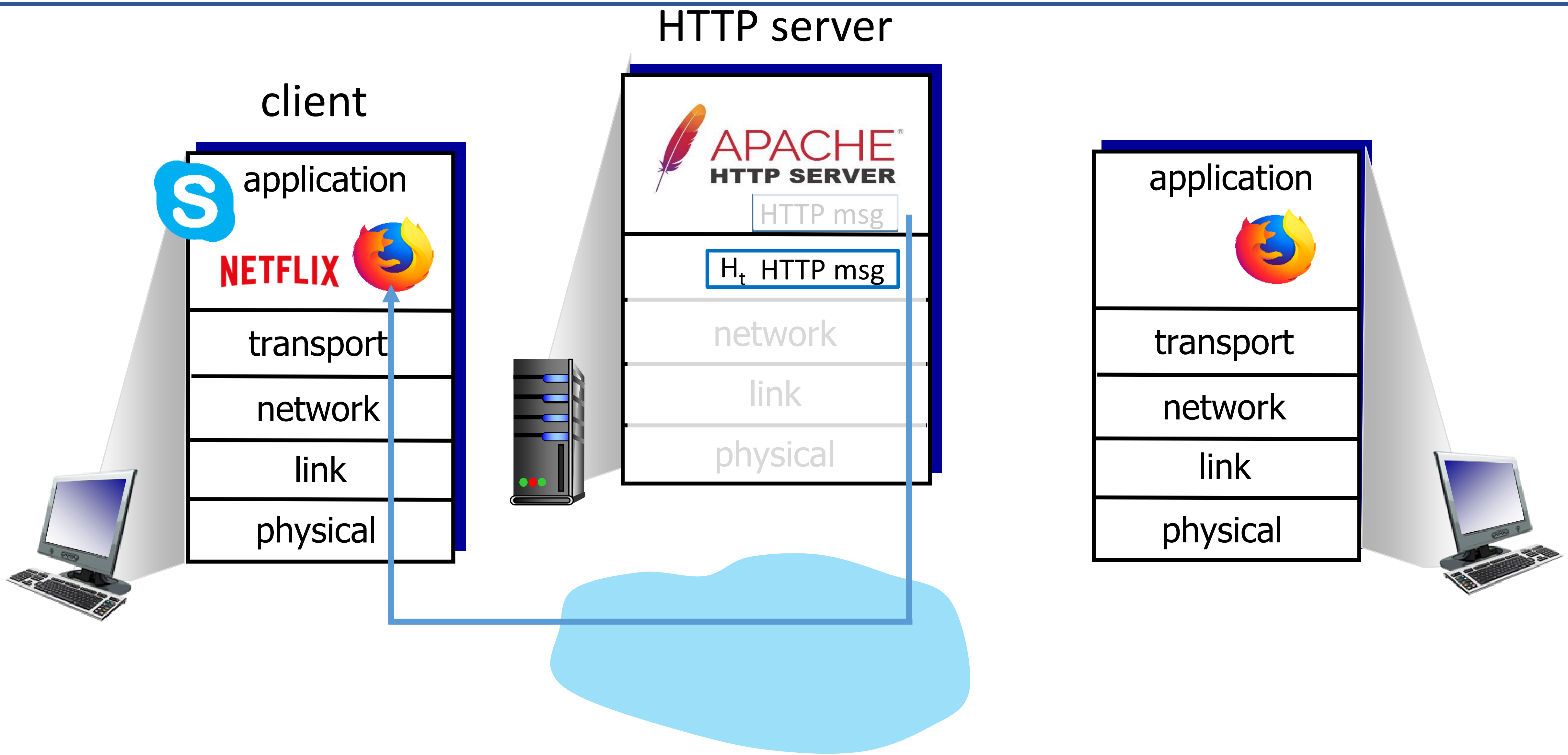
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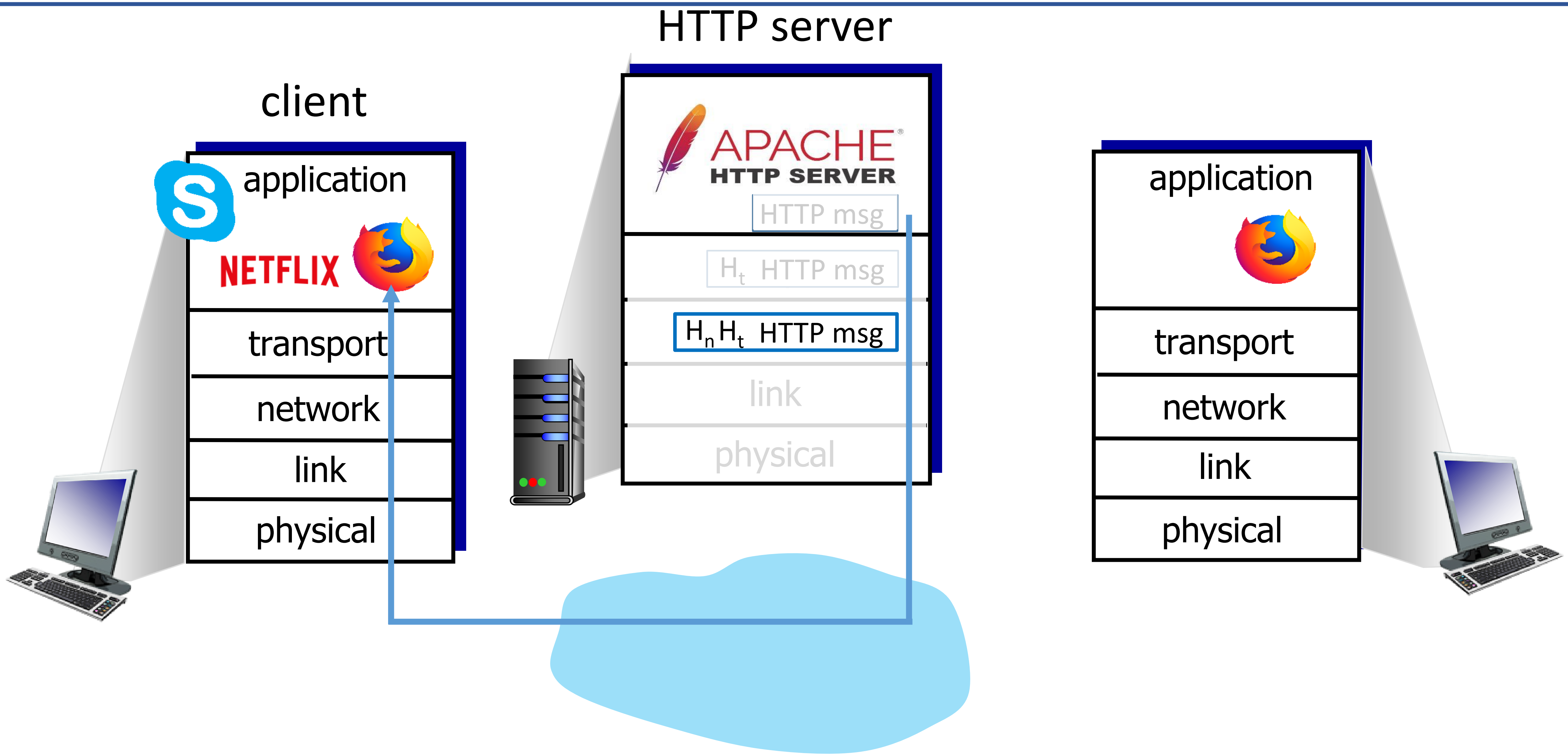
HTTP server
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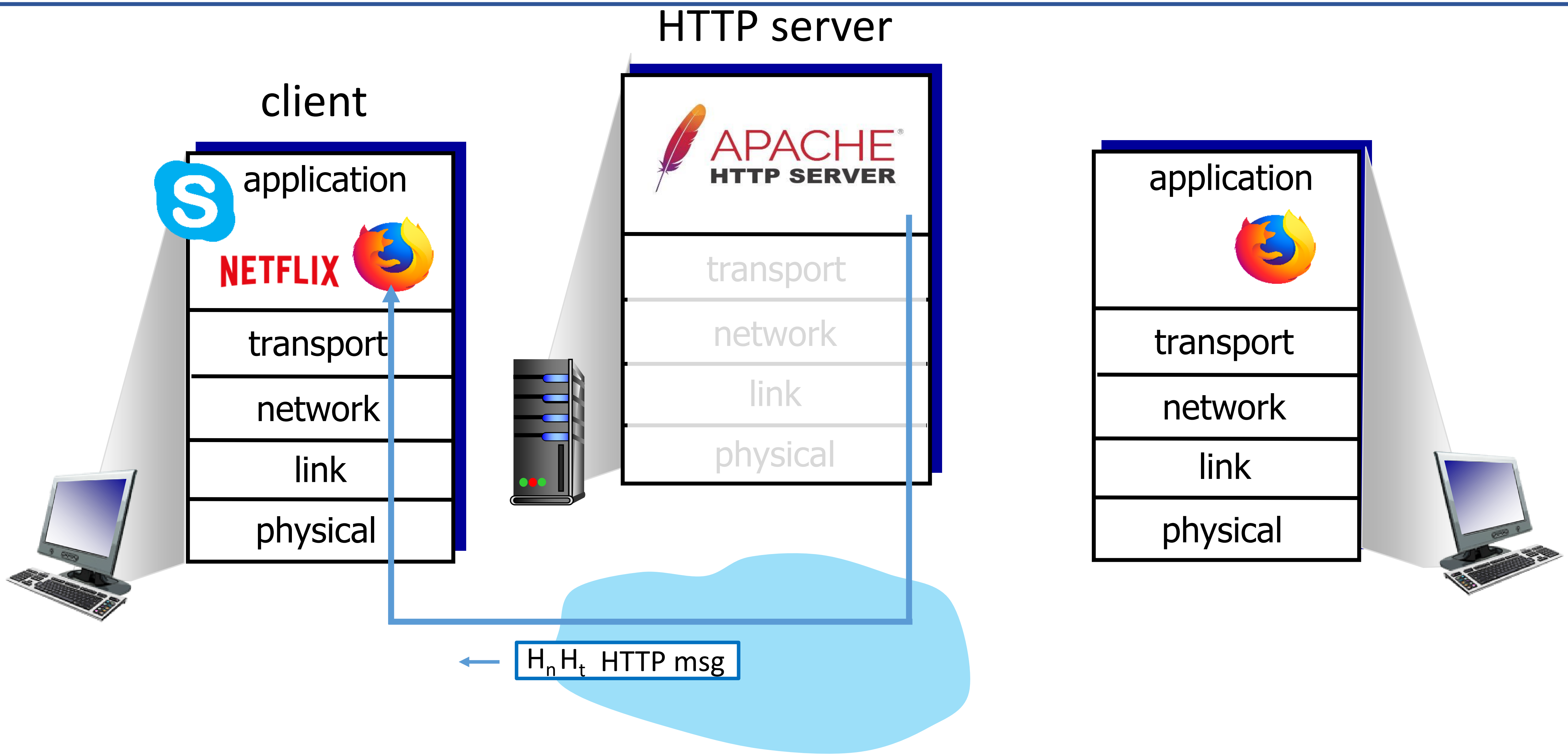
HTTP server
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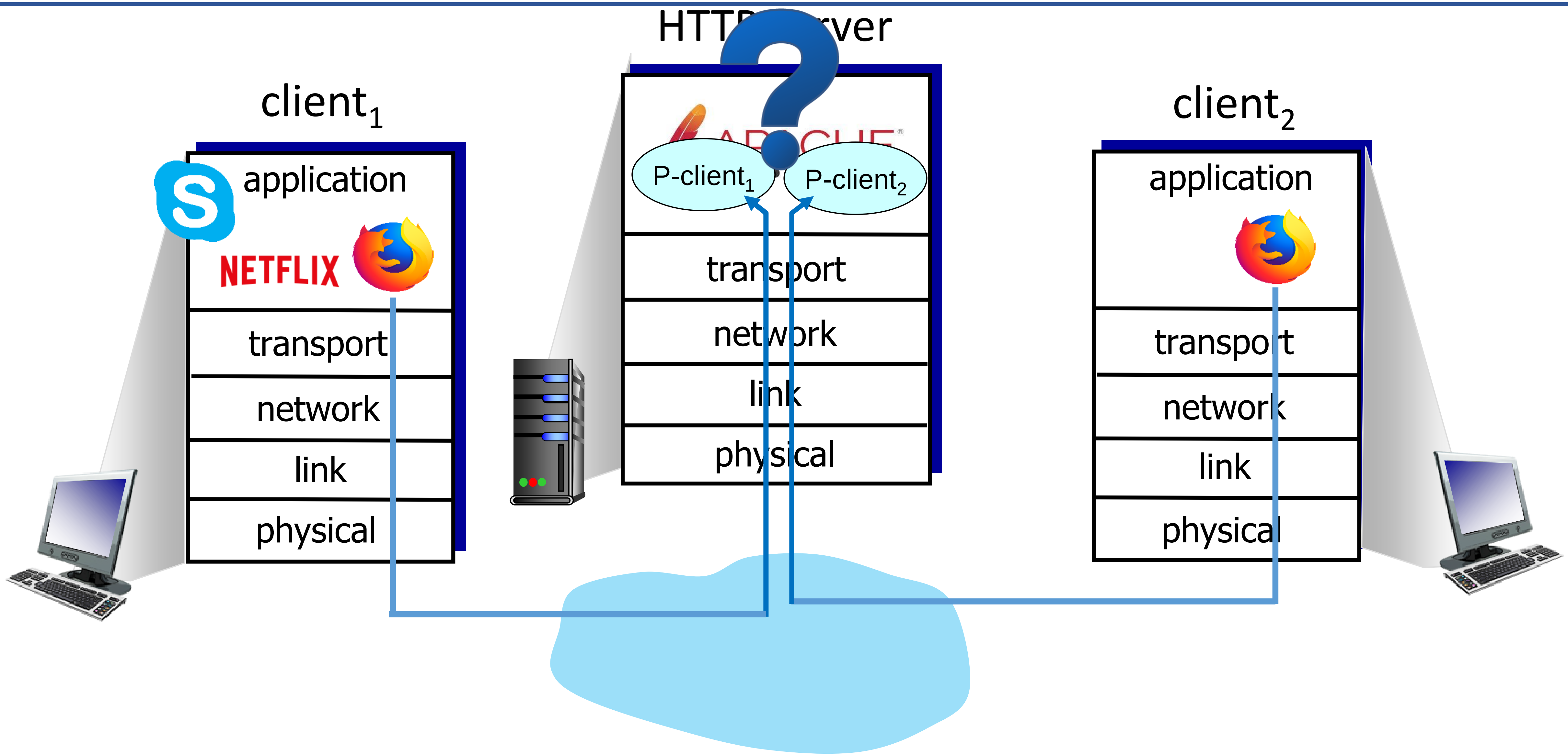
HTTP server
client$_1$
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P-client$_1$
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client$_2$
application
transport
network
link
physical

Multiplexing/demultiplexing

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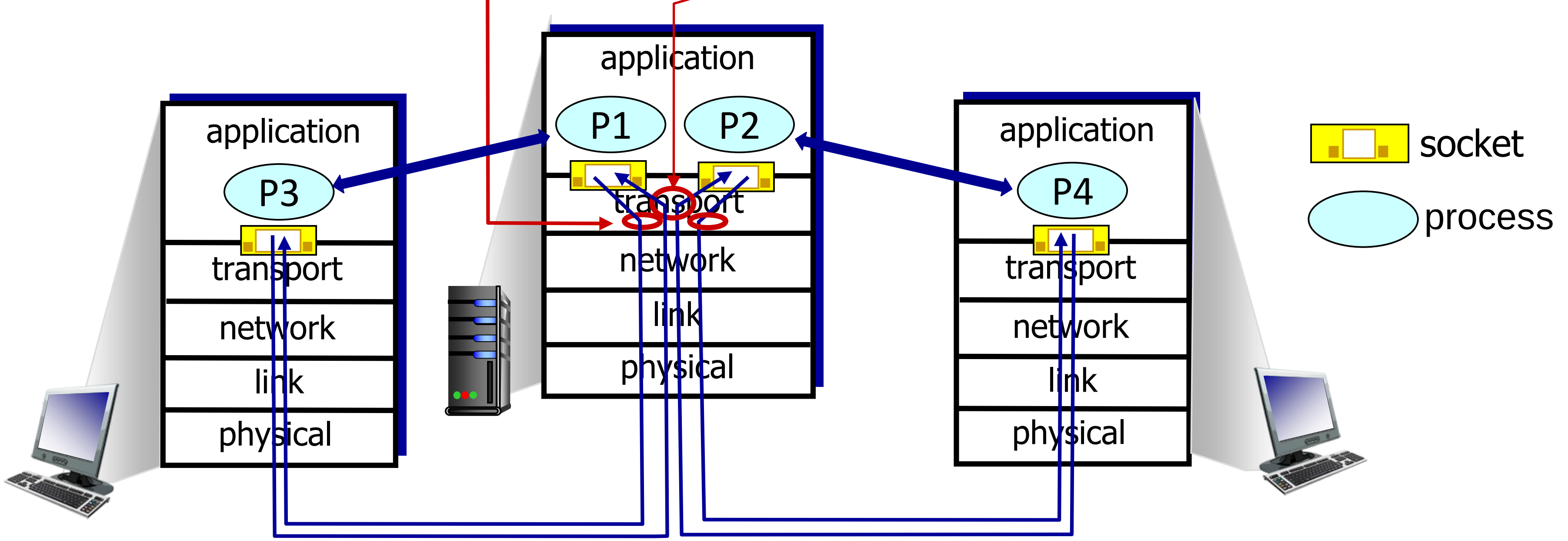
multiplexing at sender:

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

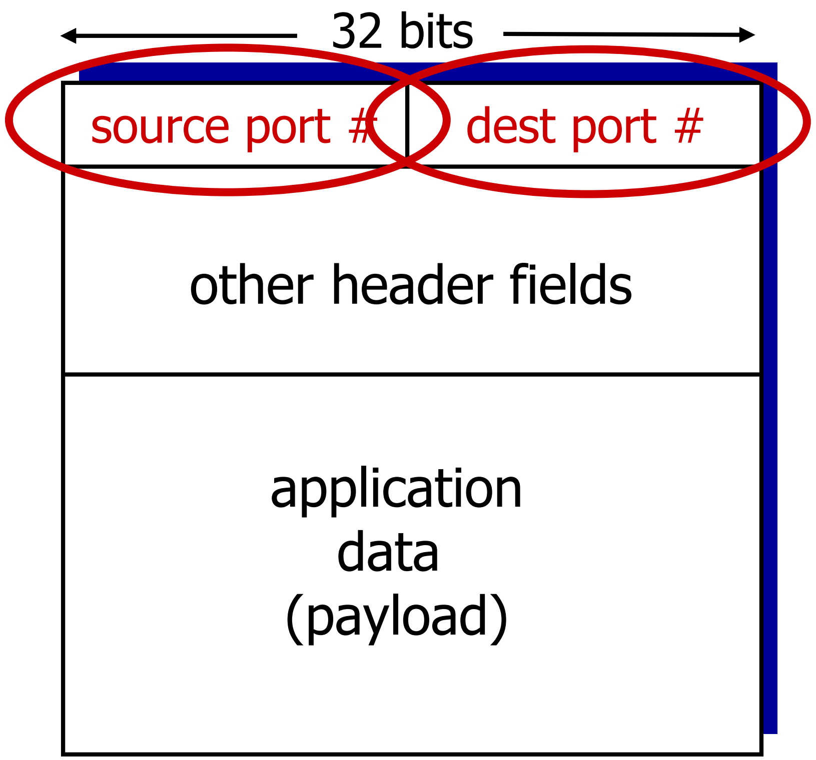
demultiplexing at receiver:

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

- when creating socket, must specify *host-local* port #:

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

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

when receiving host receives *UDP* segment:

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

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

Connectionless demultiplexing: an example

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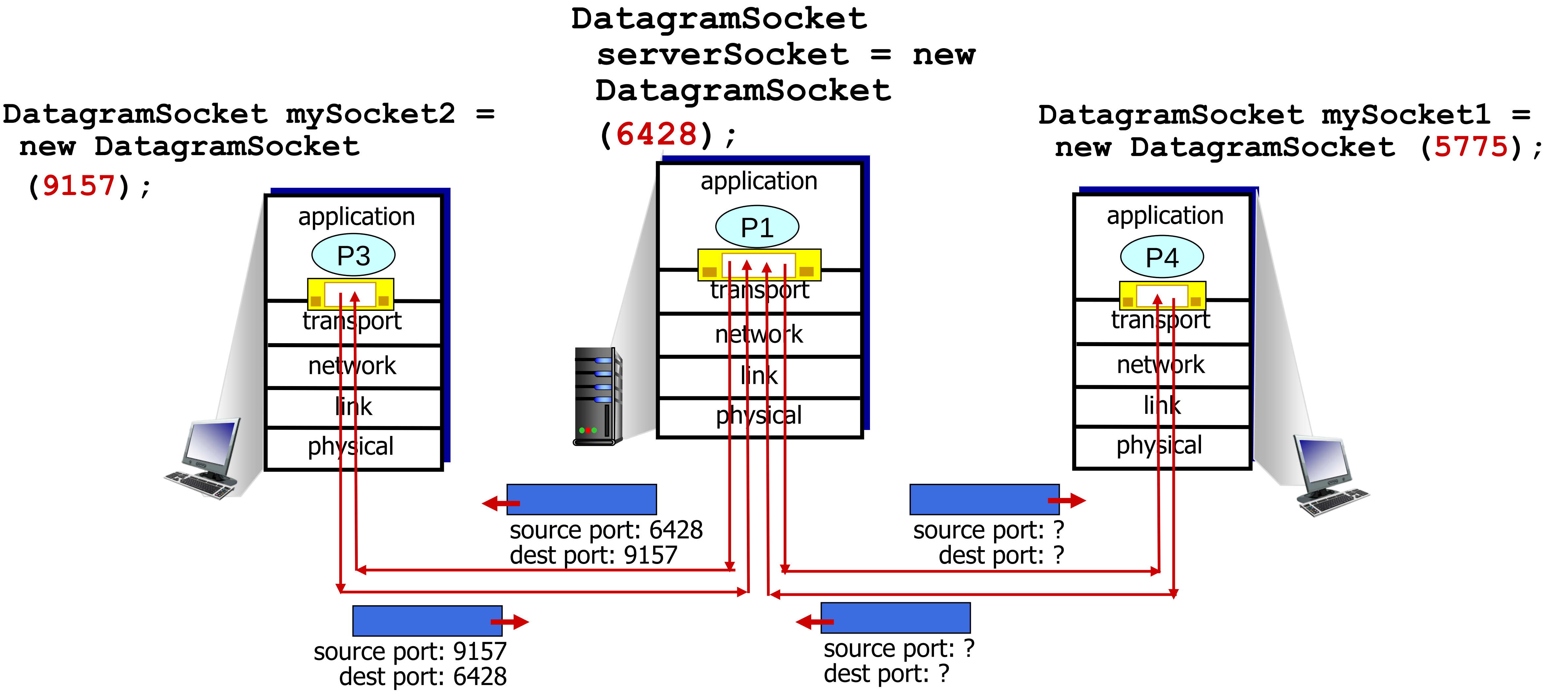
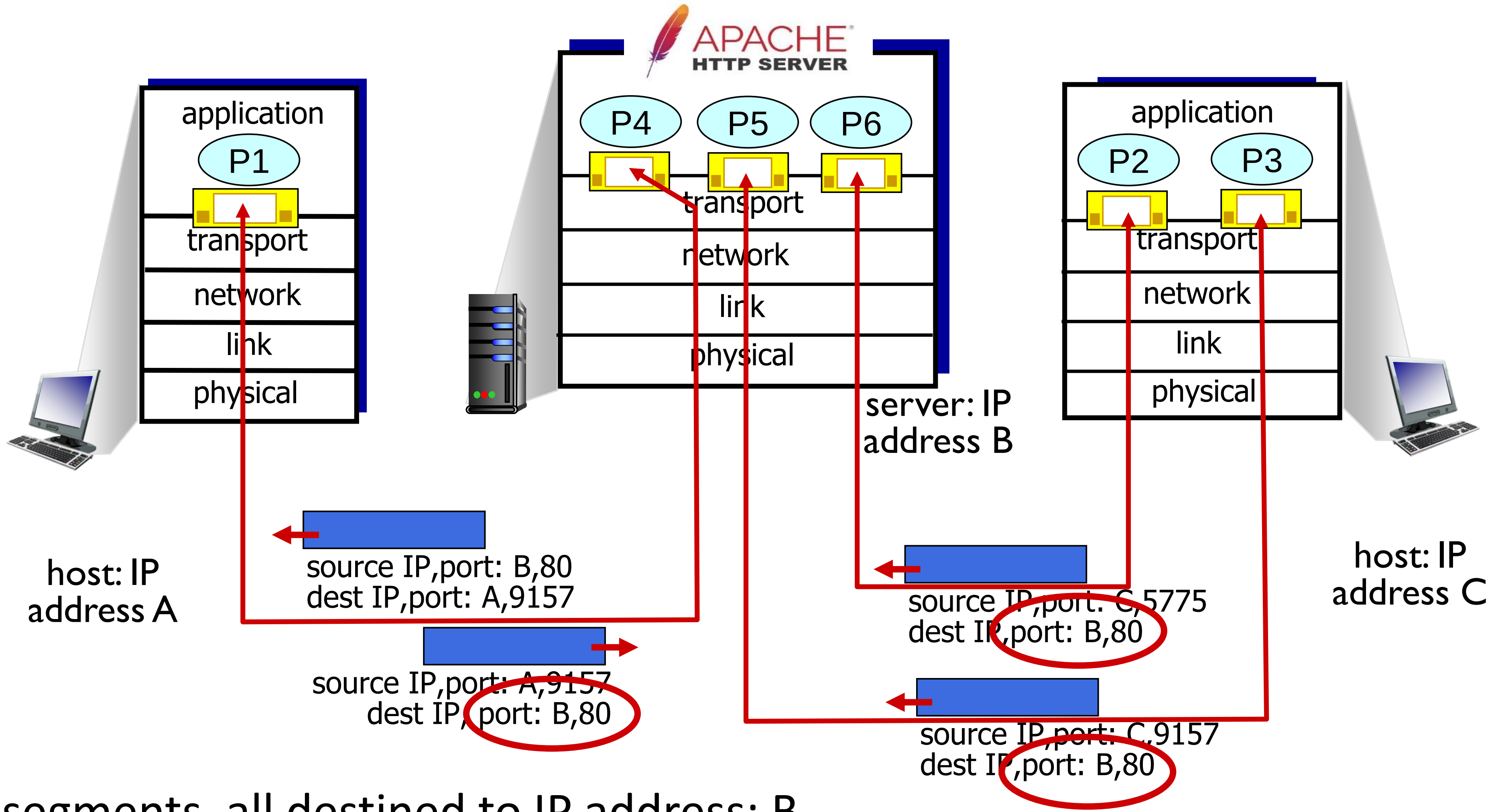


Connection-oriented demultiplexing

- 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

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UDP: User Datagram 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: Transport Layer Actions

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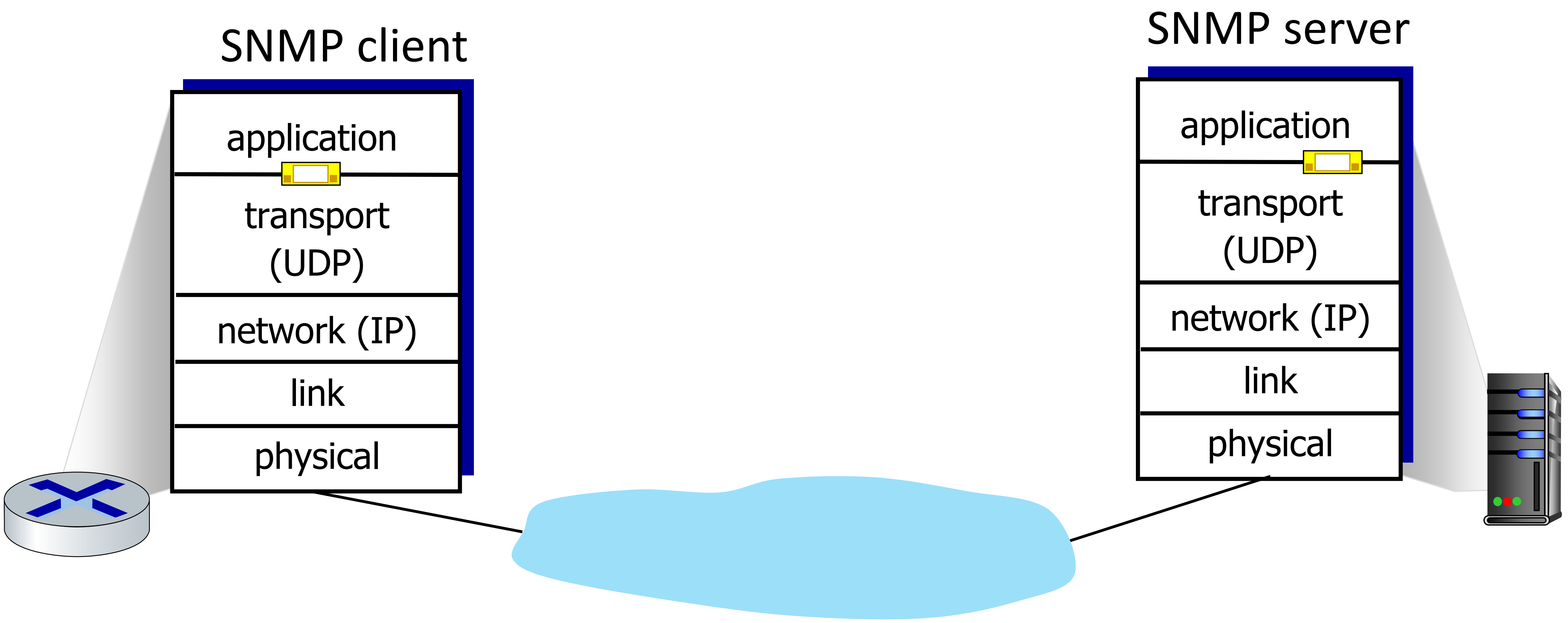
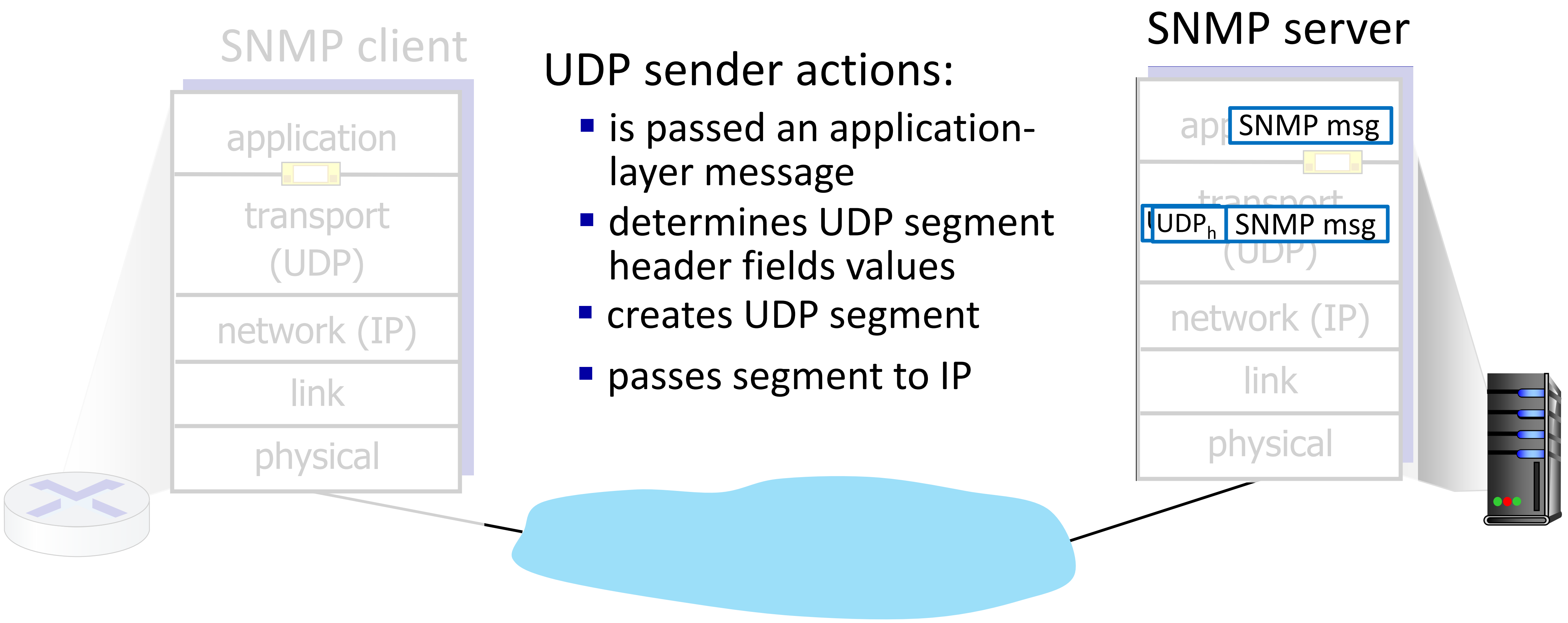


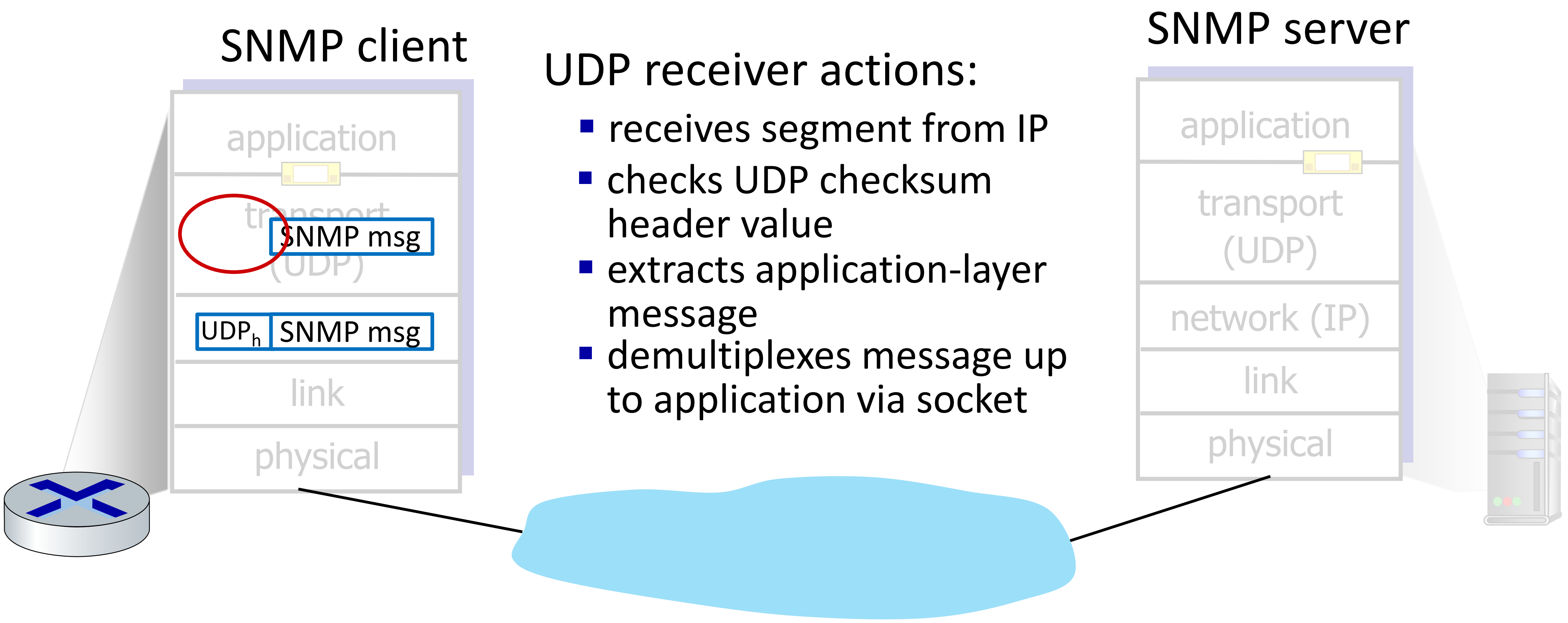
UDP: Transport Layer Actions



UDP sender actions:

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

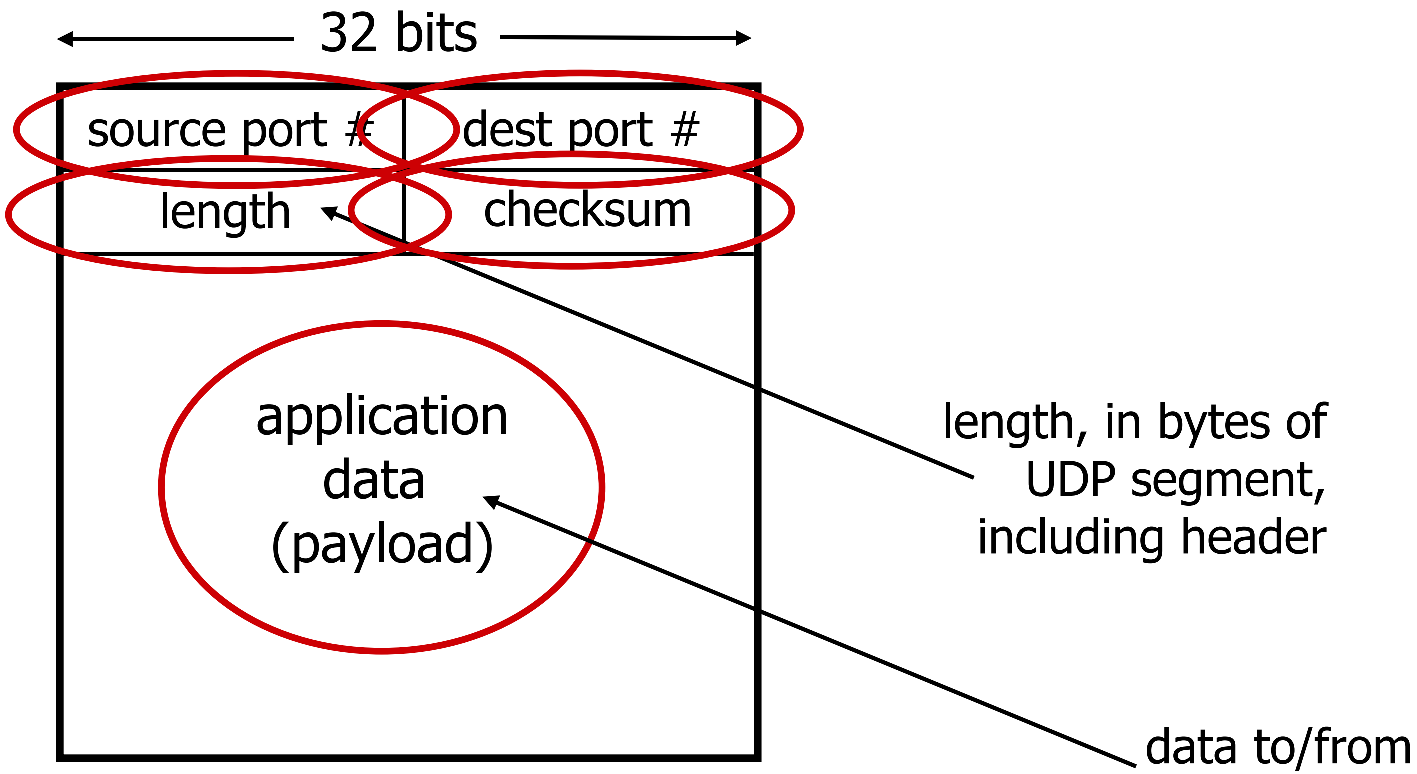
UDP: Transport Layer Actions



UDP receiver actions:

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

UDP Segment Structure



UDP segment format

UDP Checksum

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

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

receiver-computed checksum ≠ sender-computed checksum (as received)

!

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 of segment content (do one's complement)
- 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.

Internet Checksum: an example

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example: add two 16-bit integers

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

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

Internet Checksum: weak protection!

example: add two 16-bit integers

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

0 1
1 0

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

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- Transport-layer services
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- **Principles of reliable data transfer**
- Connection-oriented transport: TCP
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Principles of reliable data transfer

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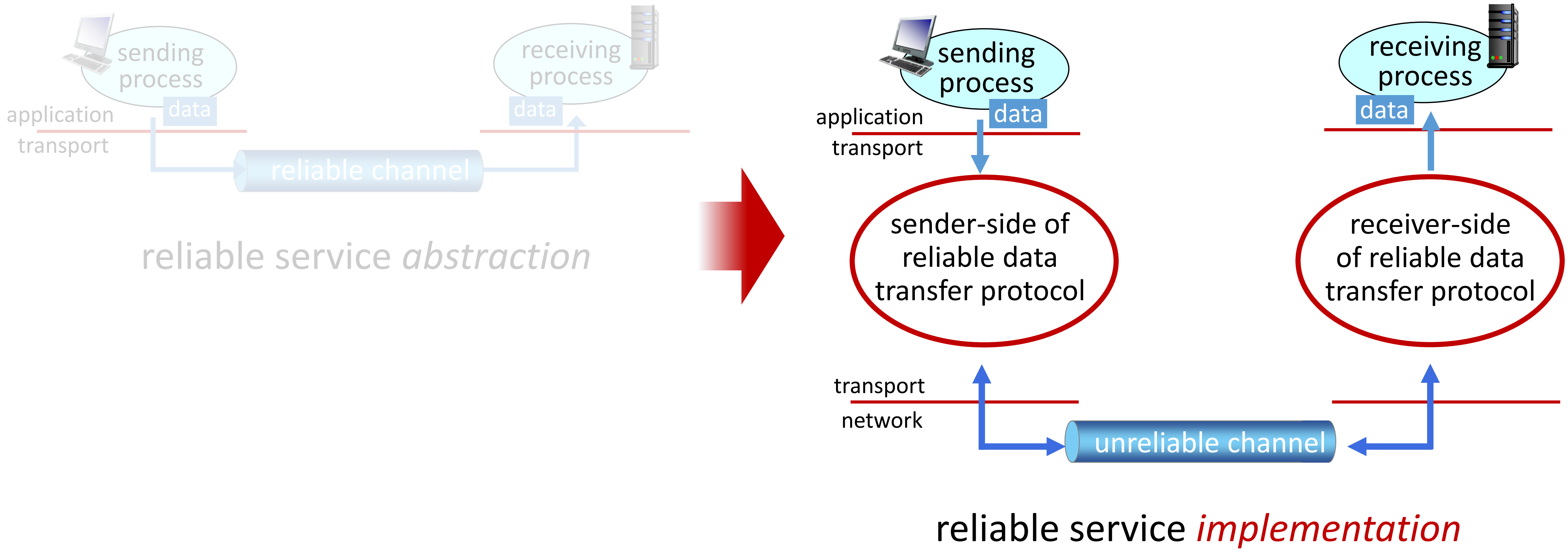
reliable service *abstraction*

Principles of reliable data transfer



Principles of reliable data transfer

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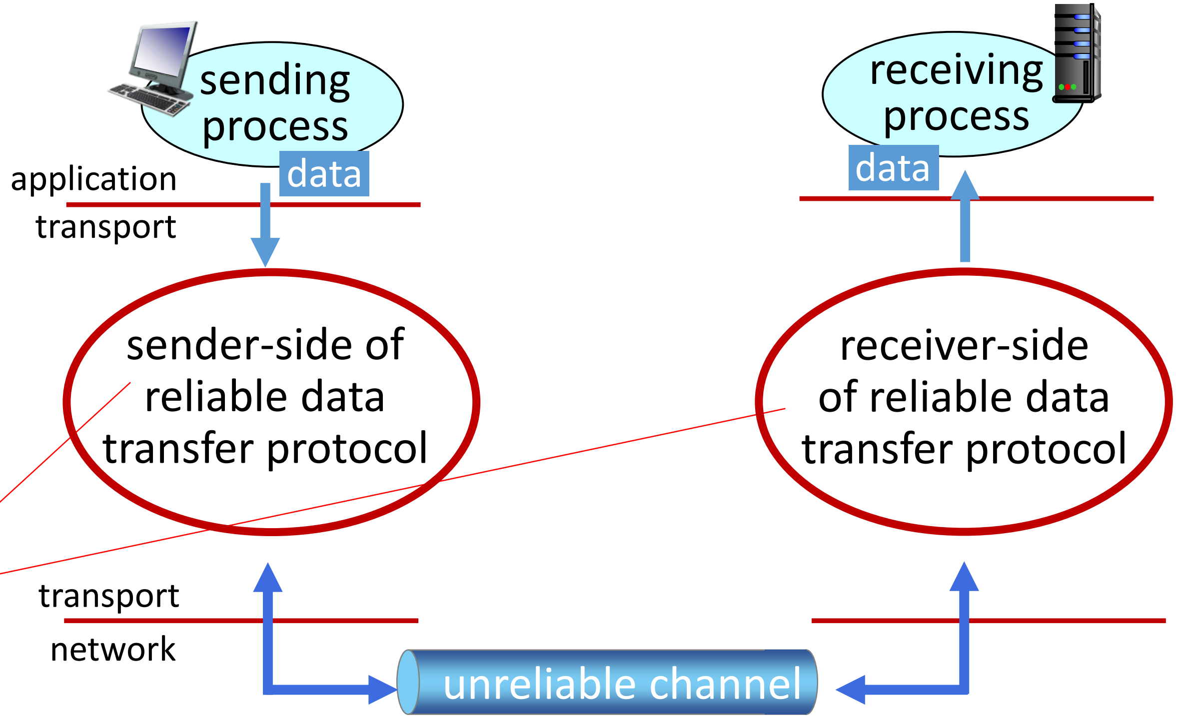
Complexity of reliable data transfer protocol will depend (strongly) on characteristics of unreliable channel (lose, corrupt, reorder data?)

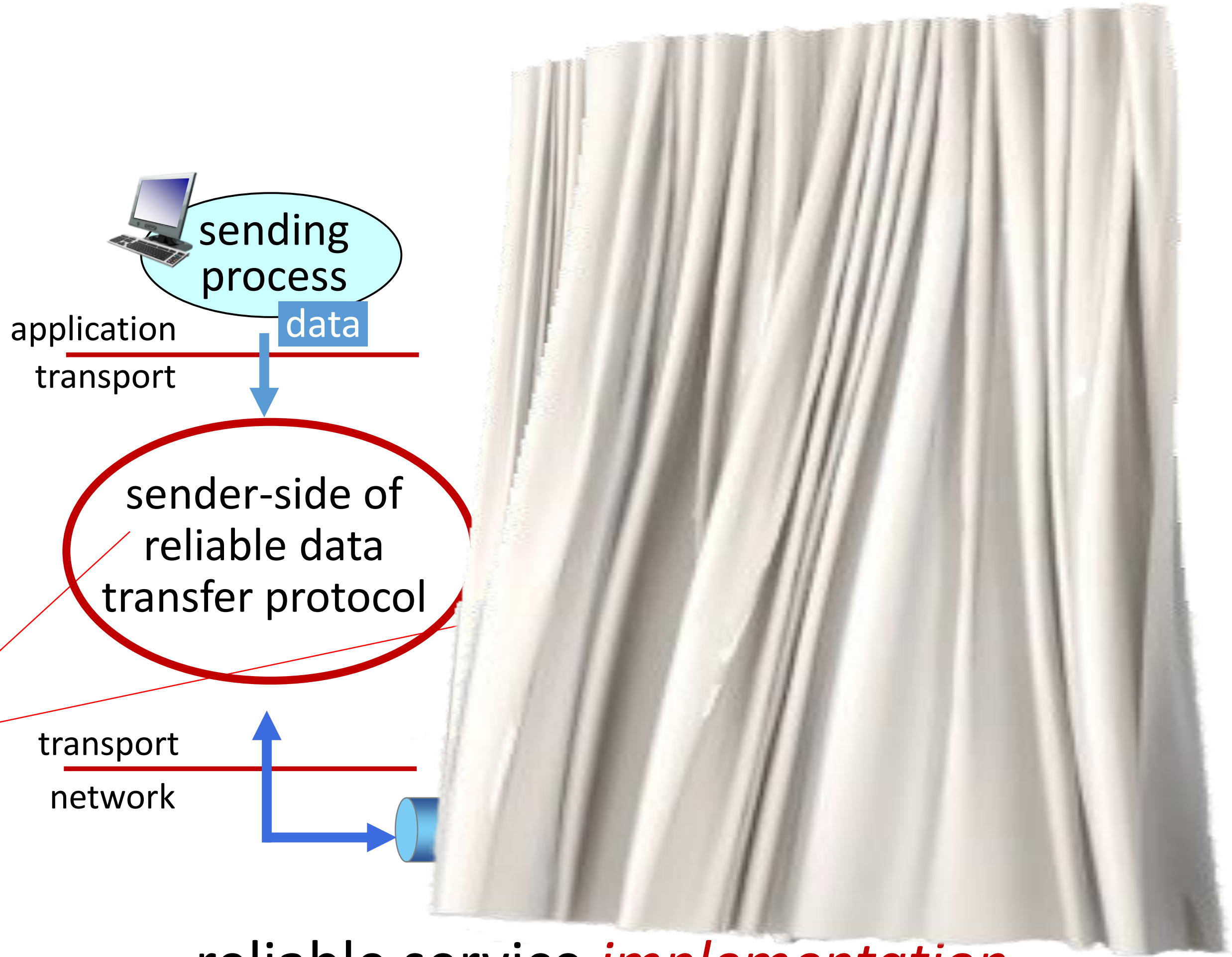


reliable service *implementation*

Principles of reliable data transfer

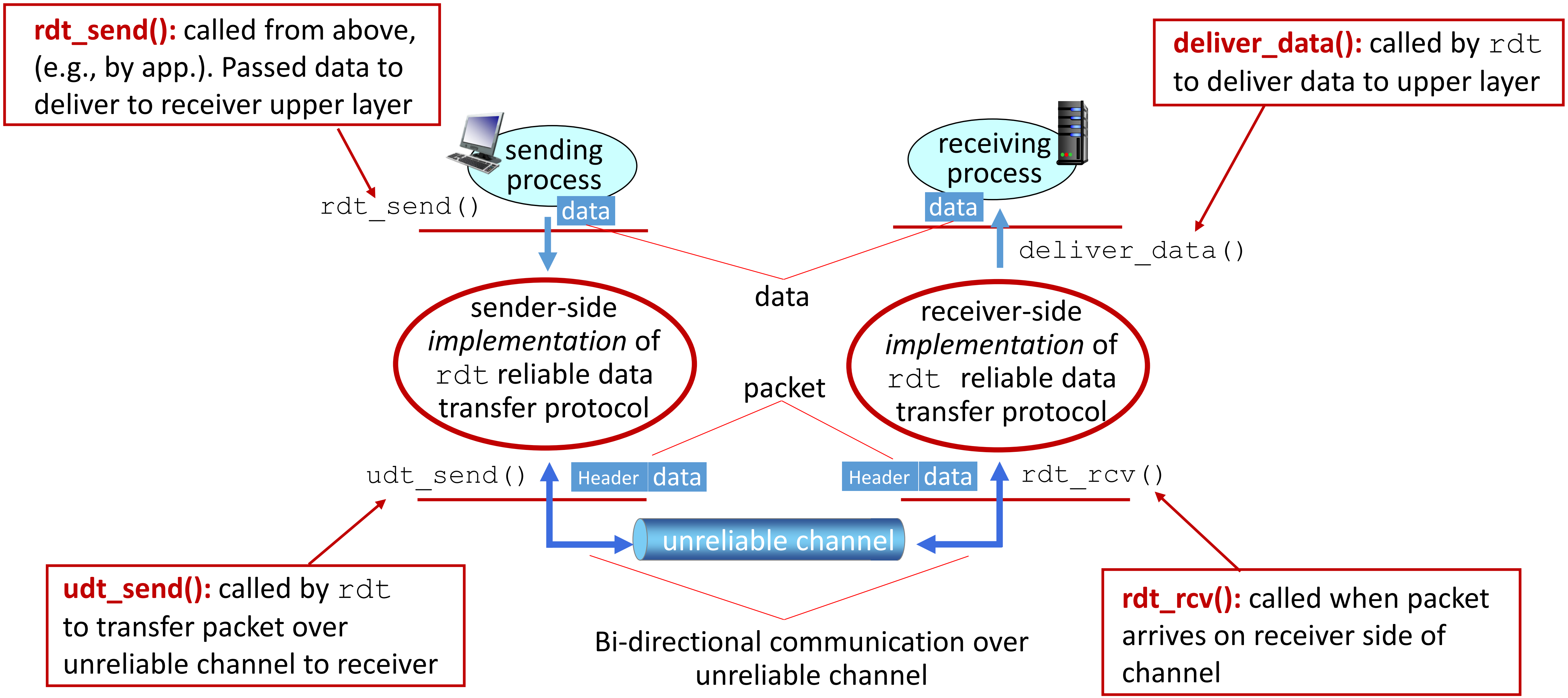
Sender, receiver do *not* know the “state” of each other, e.g., was a message received?

- unless communicated via a message



reliable service *implementation*

Reliable data transfer protocol (rdt): interfaces



Reliable data transfer: getting started

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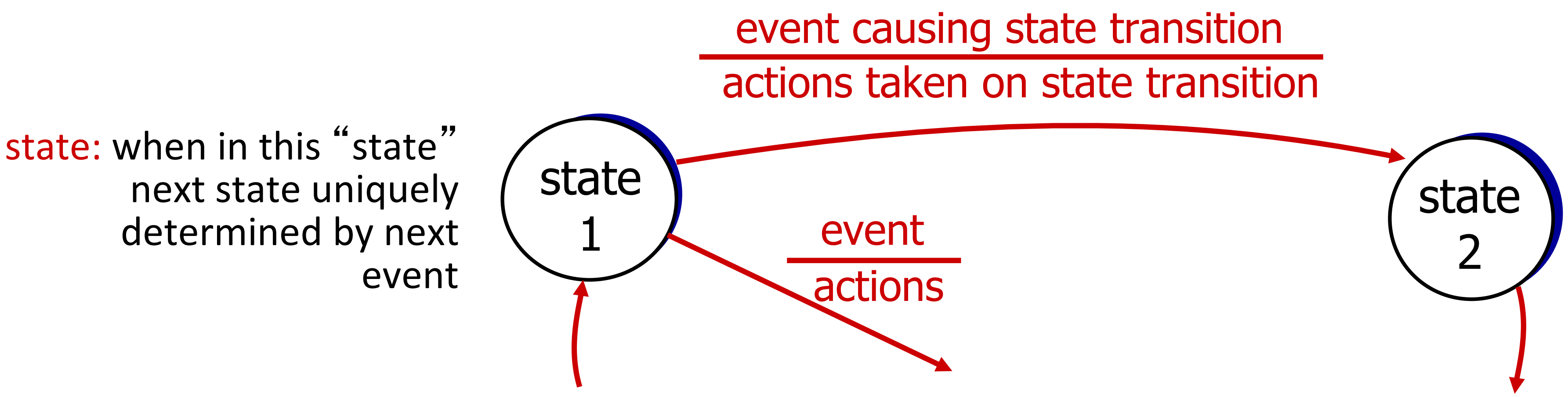
We will:

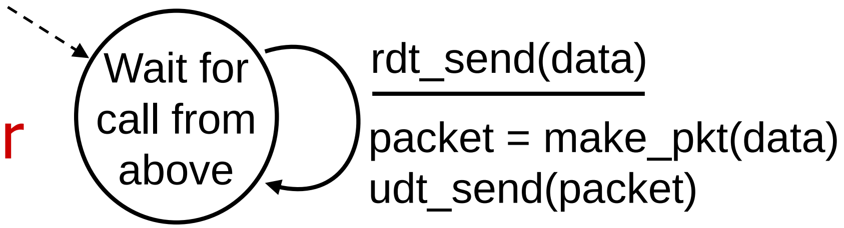
- incrementally develop sender, receiver sides of reliable data transfer protocol (`rdt`)
- consider only unidirectional data transfer
  - but control info will flow in both directions!
- use finite state machines (FSM) to specify sender, receiver



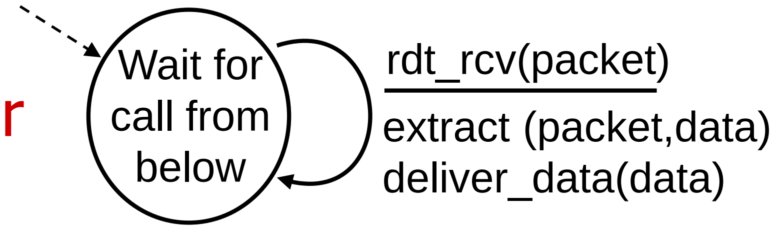
rdt1.0: reliable transfer over a reliable channel

- 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

sender



receiver



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

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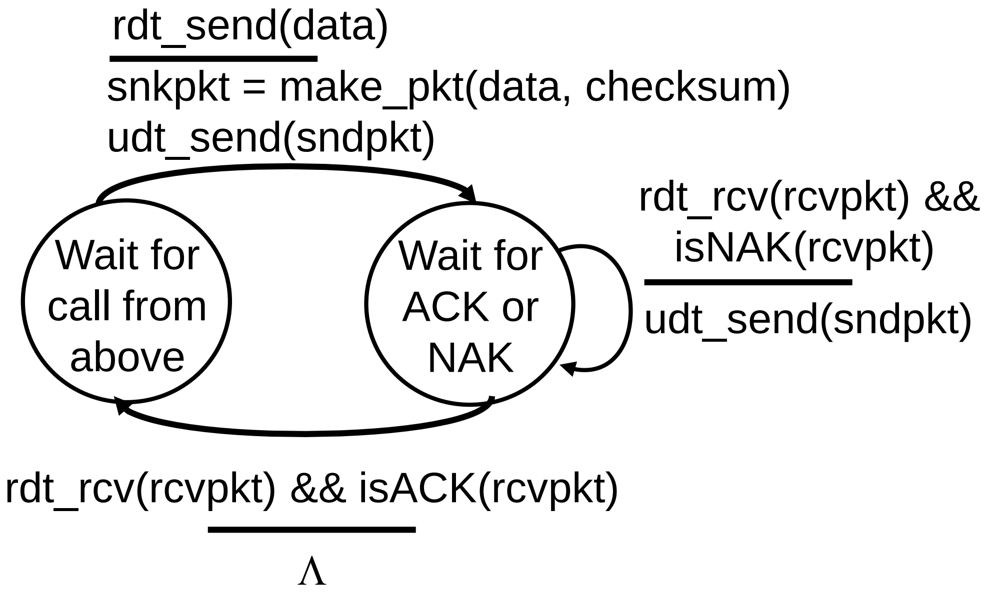
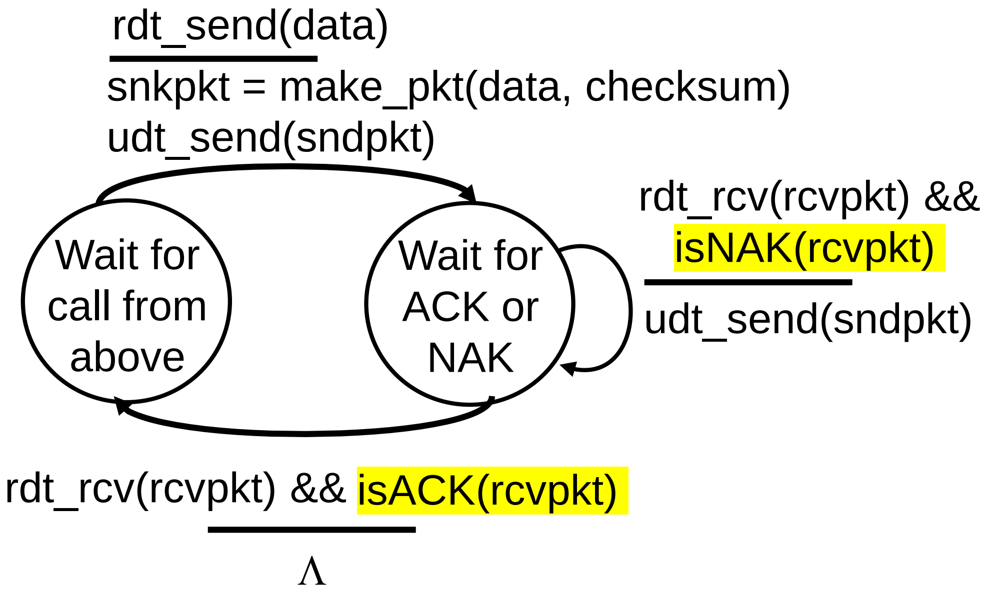
sender



rdt2.0: FSM specification

sender



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

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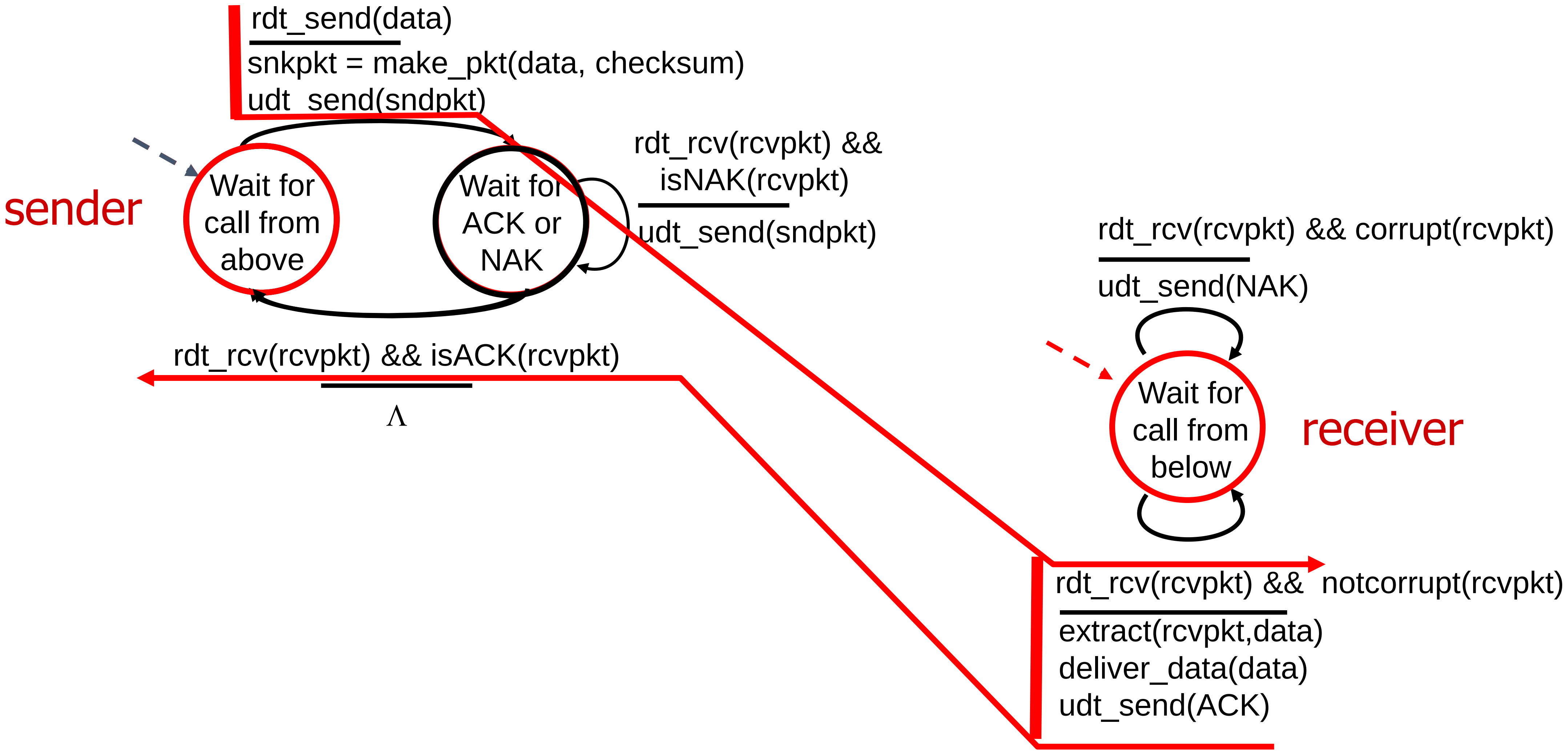

rdt2.0: corrupted packet scenario

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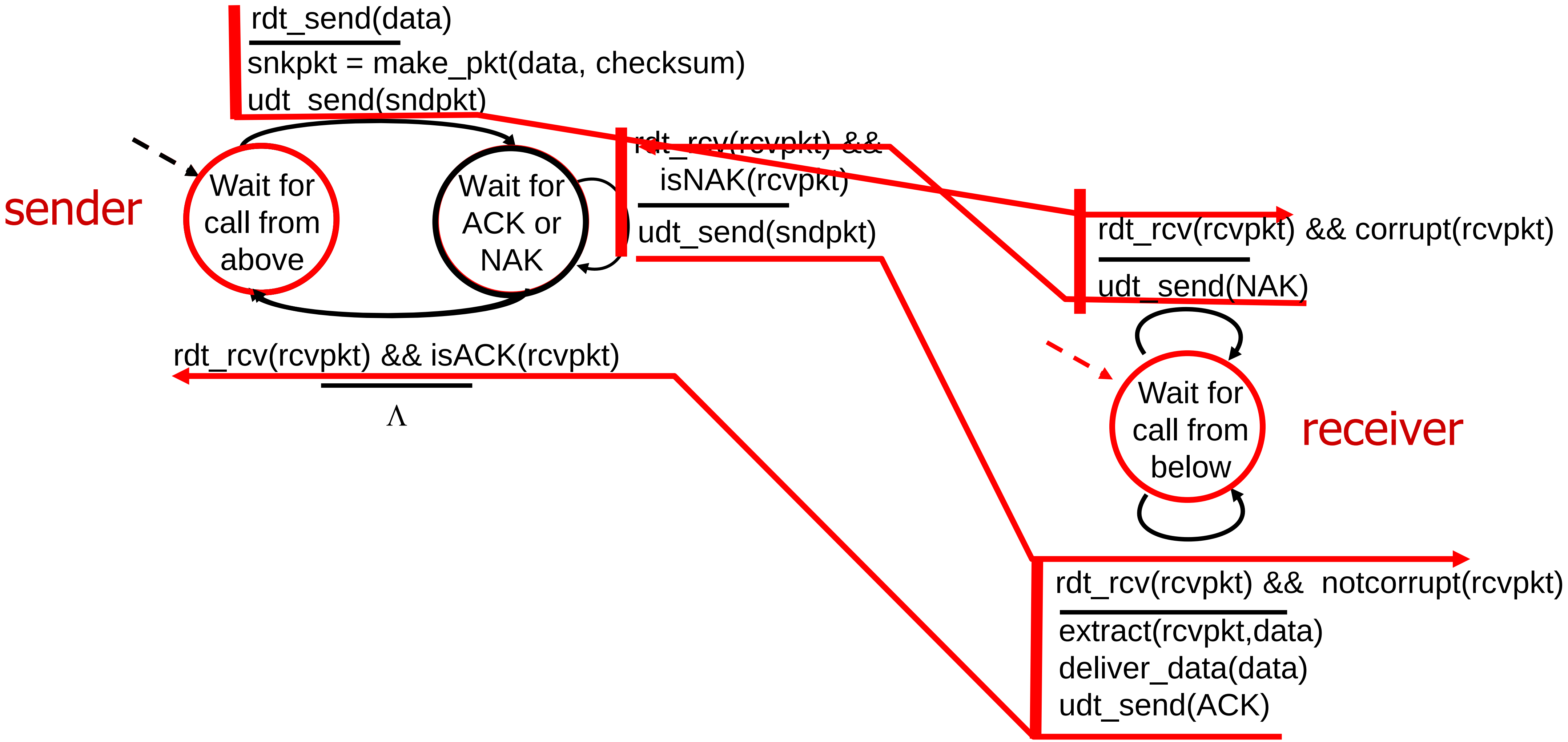


rdt2.0 has a fatal flaw!

what happens if ACK/NAK corrupted?

- sender doesn't know what happened at receiver!
- can't just retransmit: possible duplicate

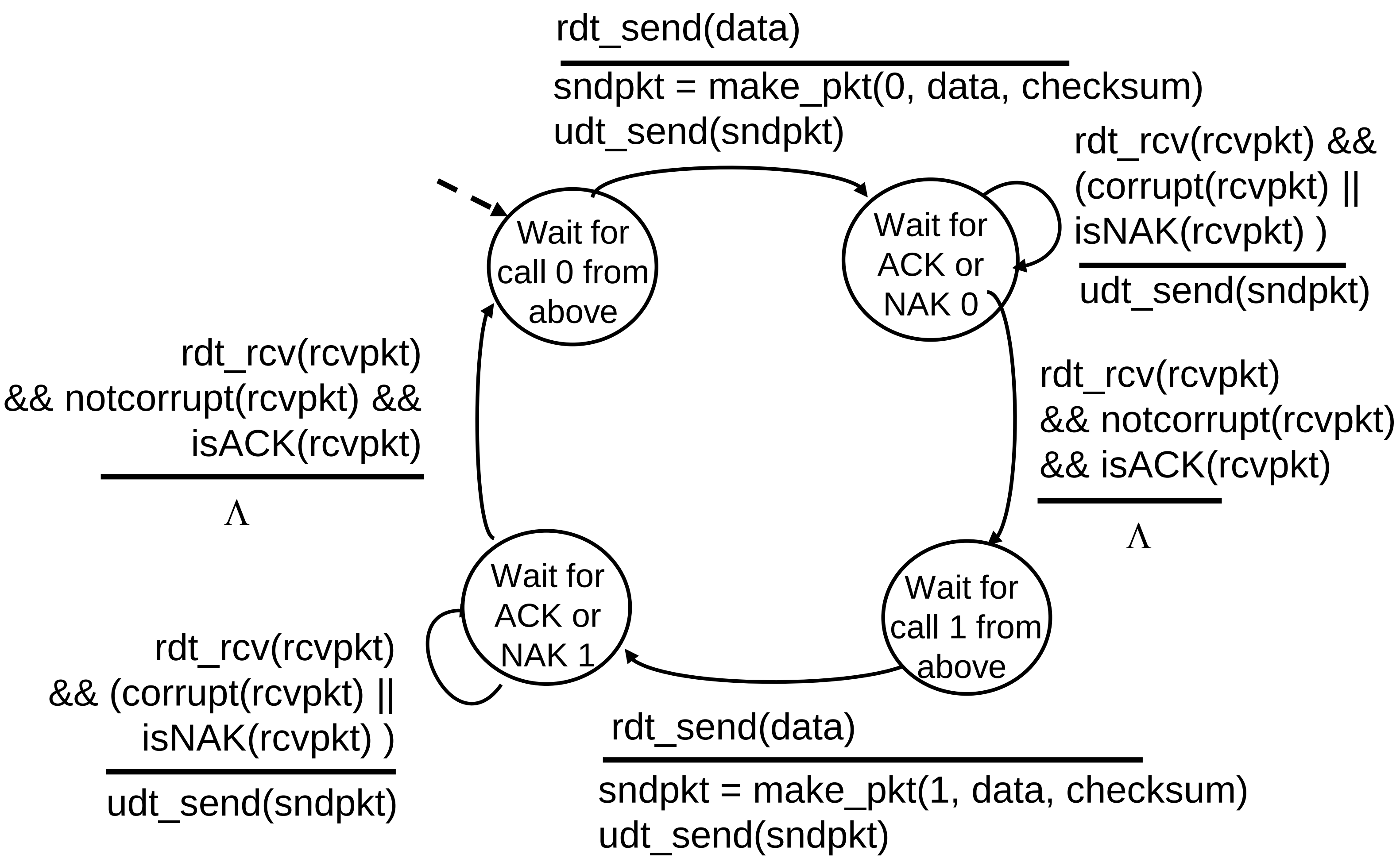
handling duplicates:

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

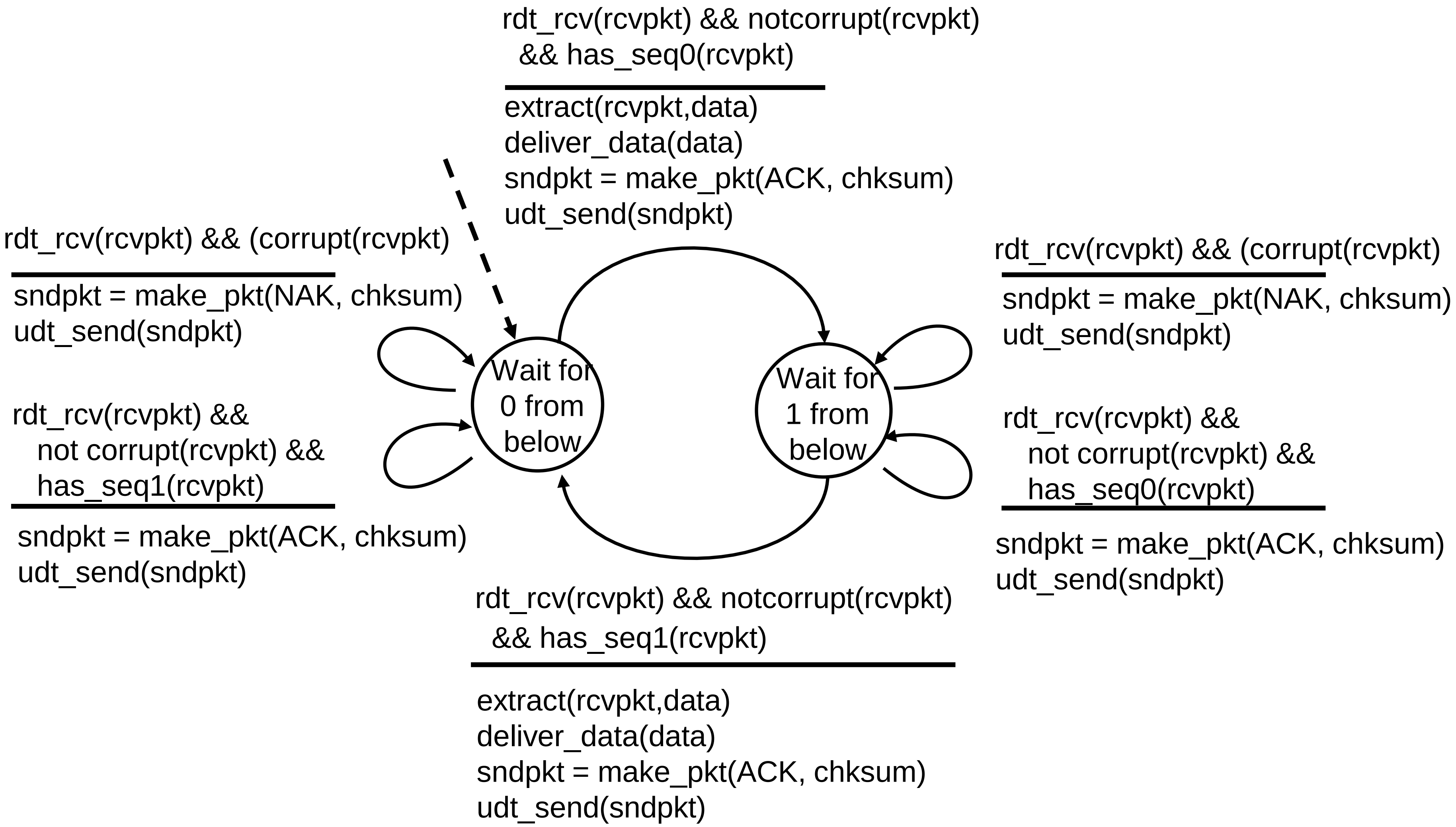
stop and wait

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:

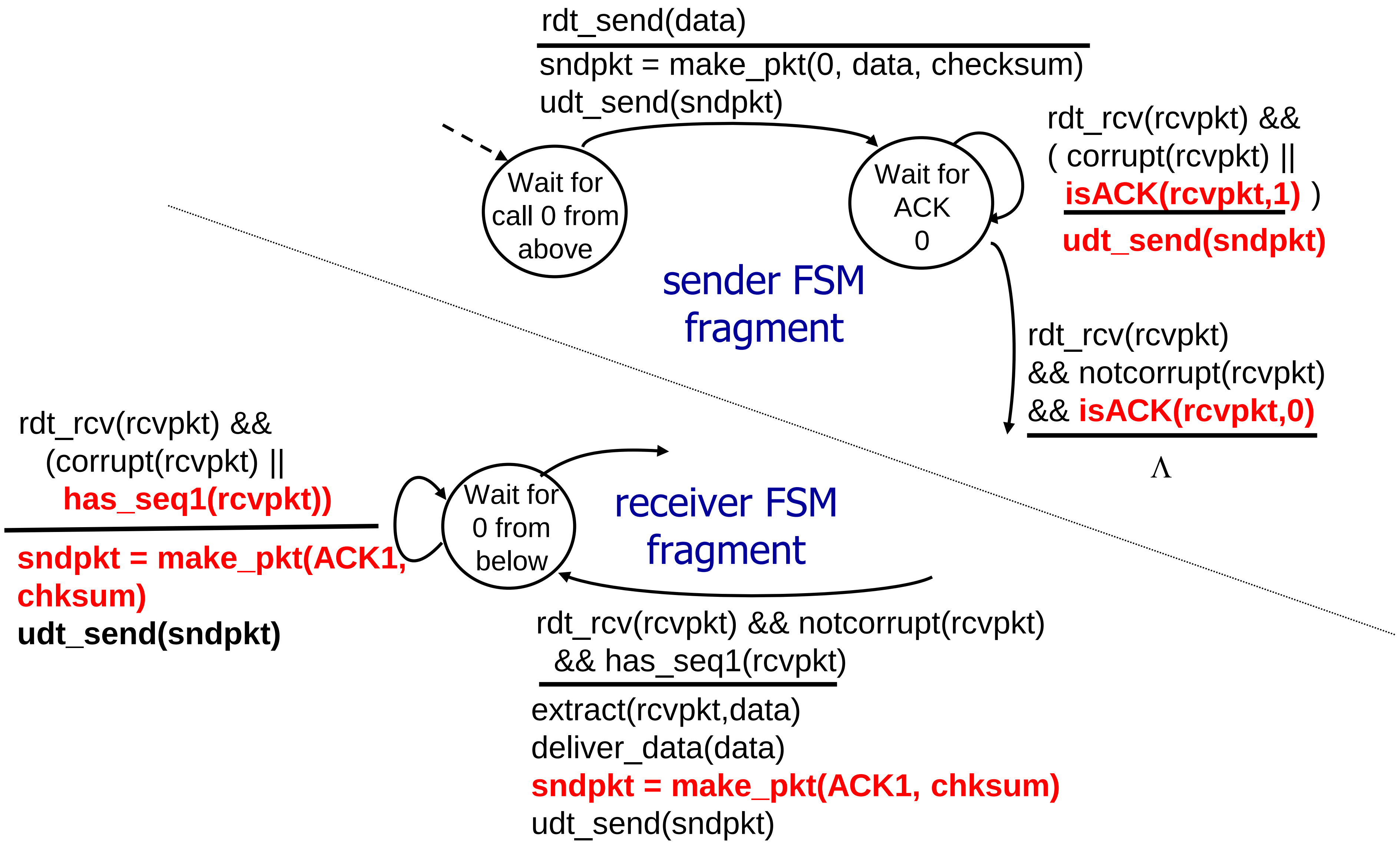
- must check if received packet is duplicate
  - state indicates whether 0 or 1 is expected pkt seq #
- note: receiver can *not* know if its last ACK/NAK received OK at sender

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

timeout

rdt3.0 sender

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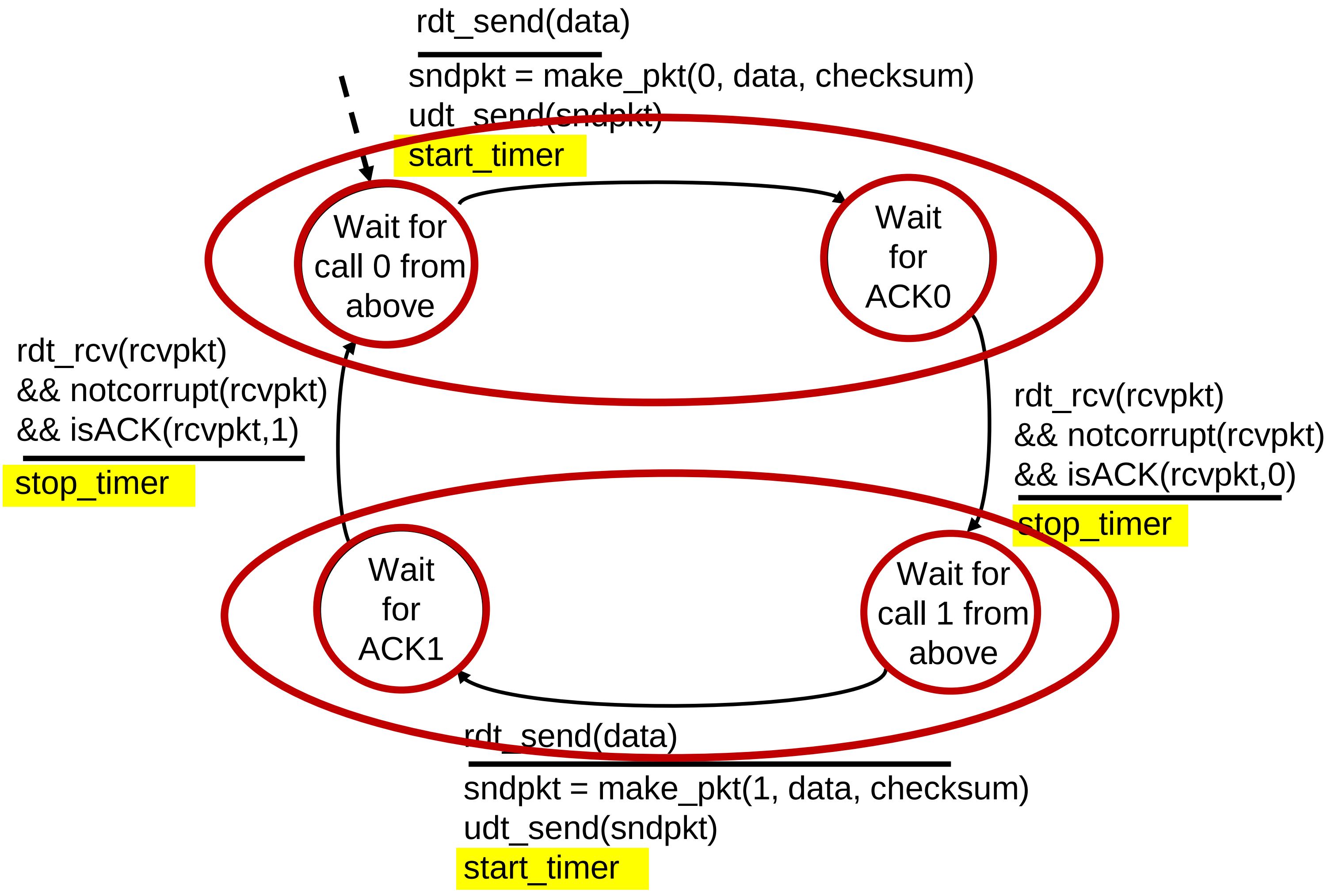
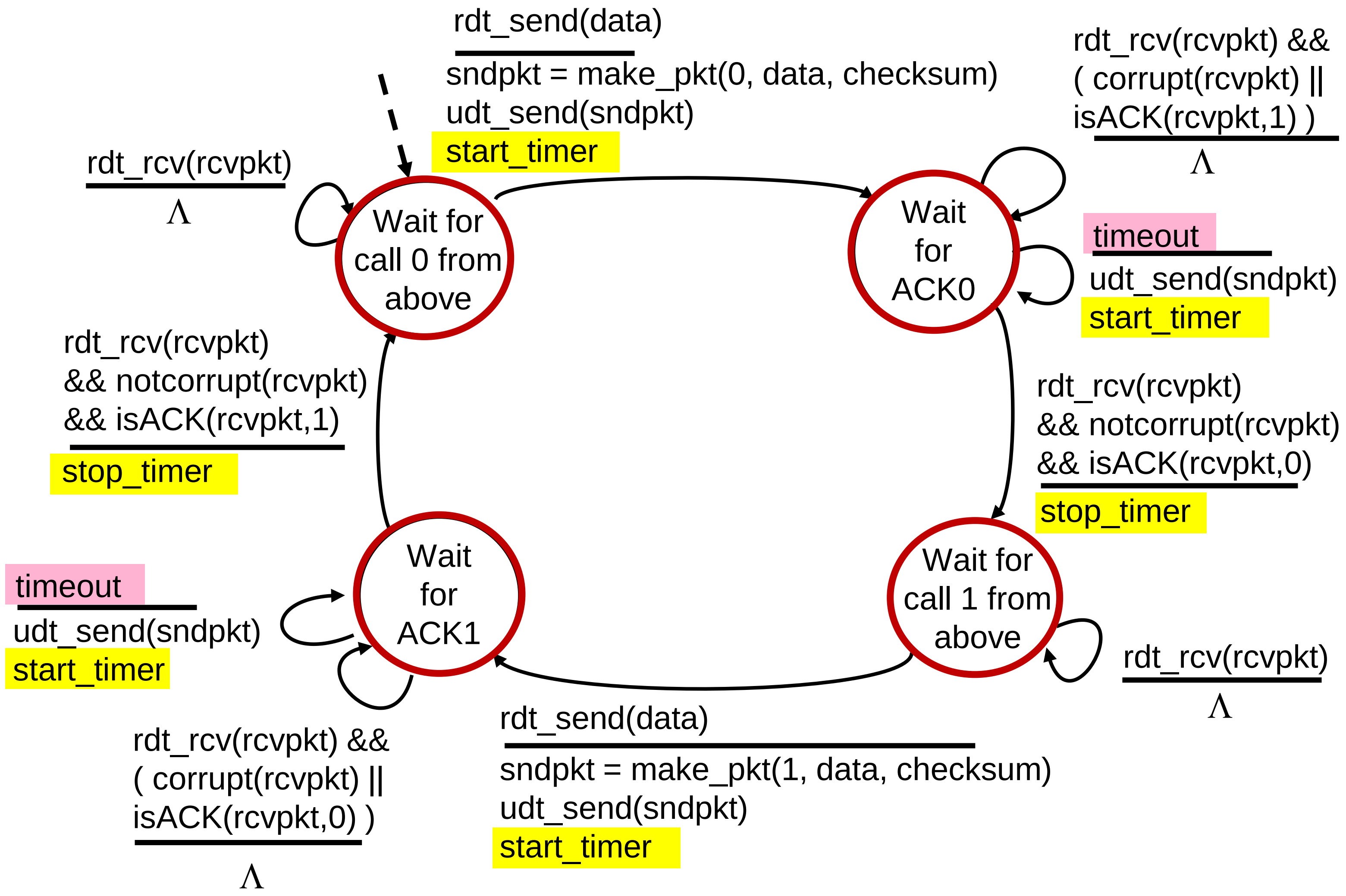


rdt3.0 sender



rdt3.0 in action

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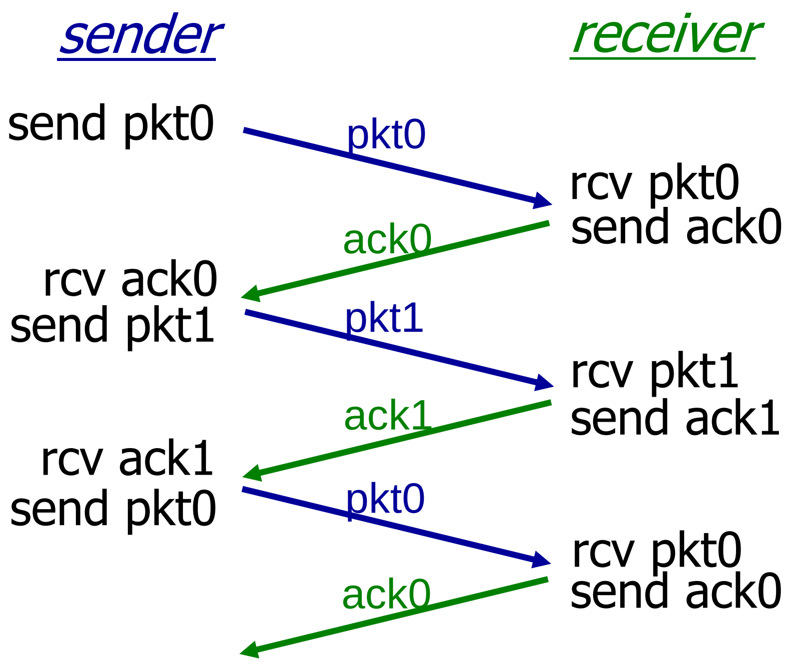
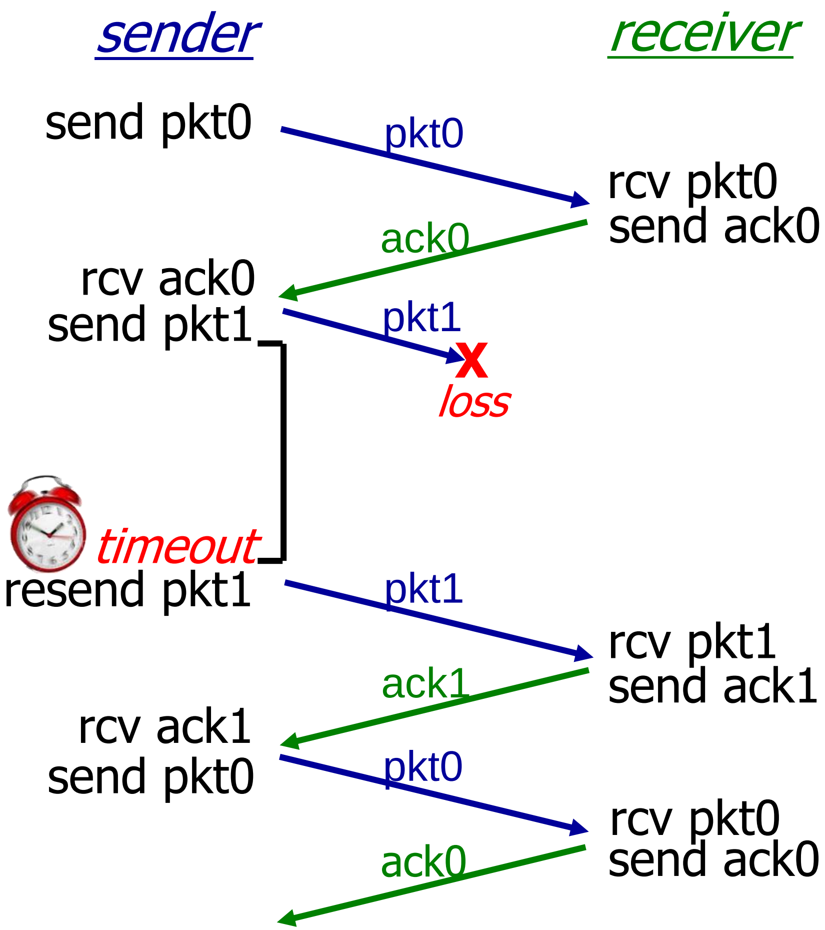
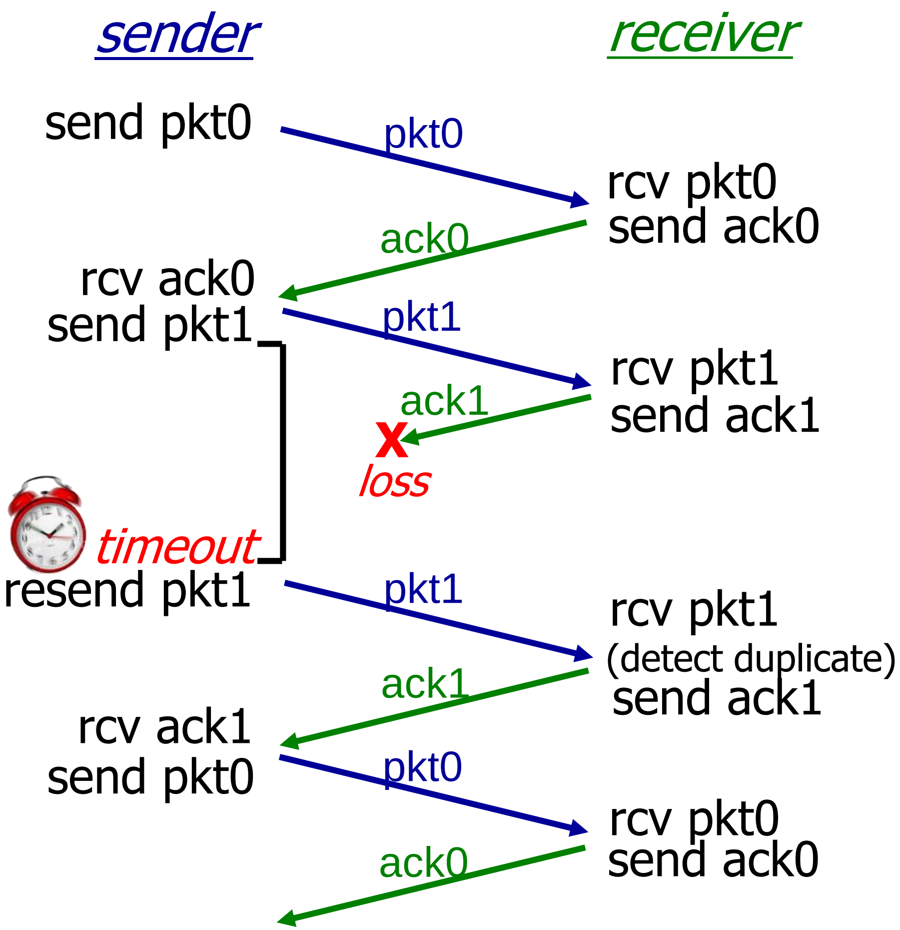


(a) no loss

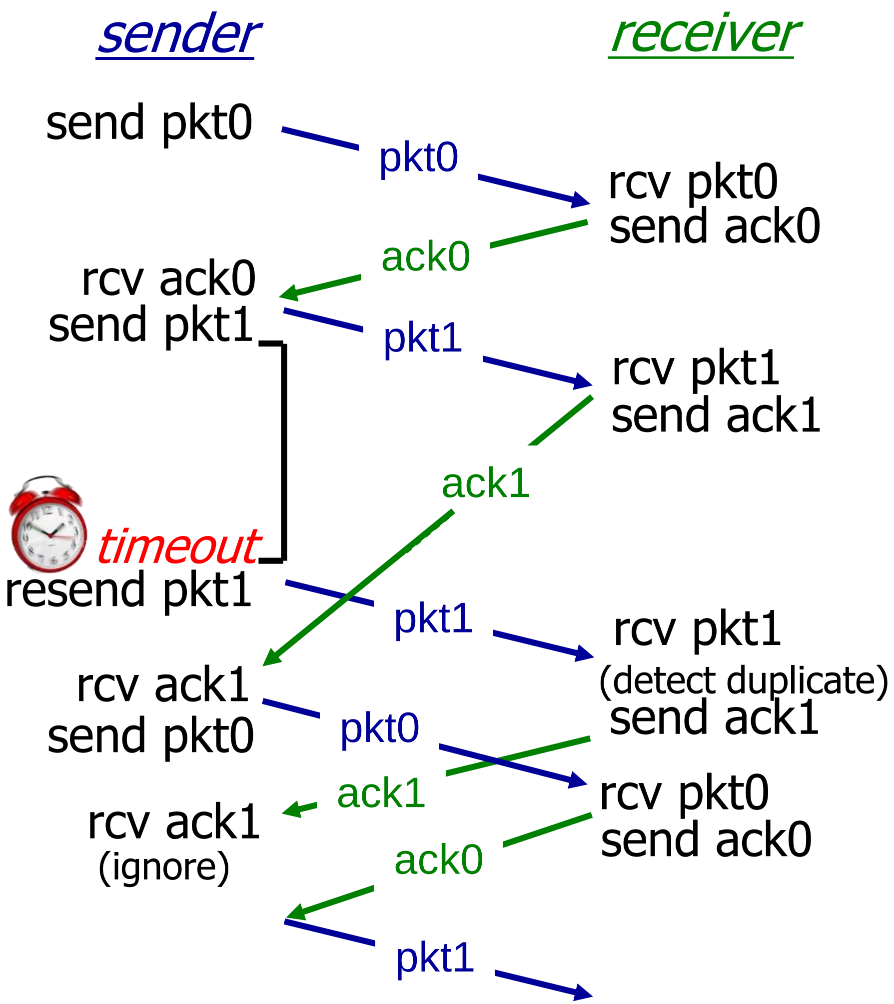


(b) packet loss

rdt3.0 in action



(c) ACK loss



(d) premature timeout/ delayed ACK

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

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- U_{sender}: *utilization* – fraction of time sender busy sending

- example: 1 Gbps link, 15 ms prop. delay, 8000 bit packet
  - time to transmit packet into channel:

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

rdt3.0: stop-and-wait operation

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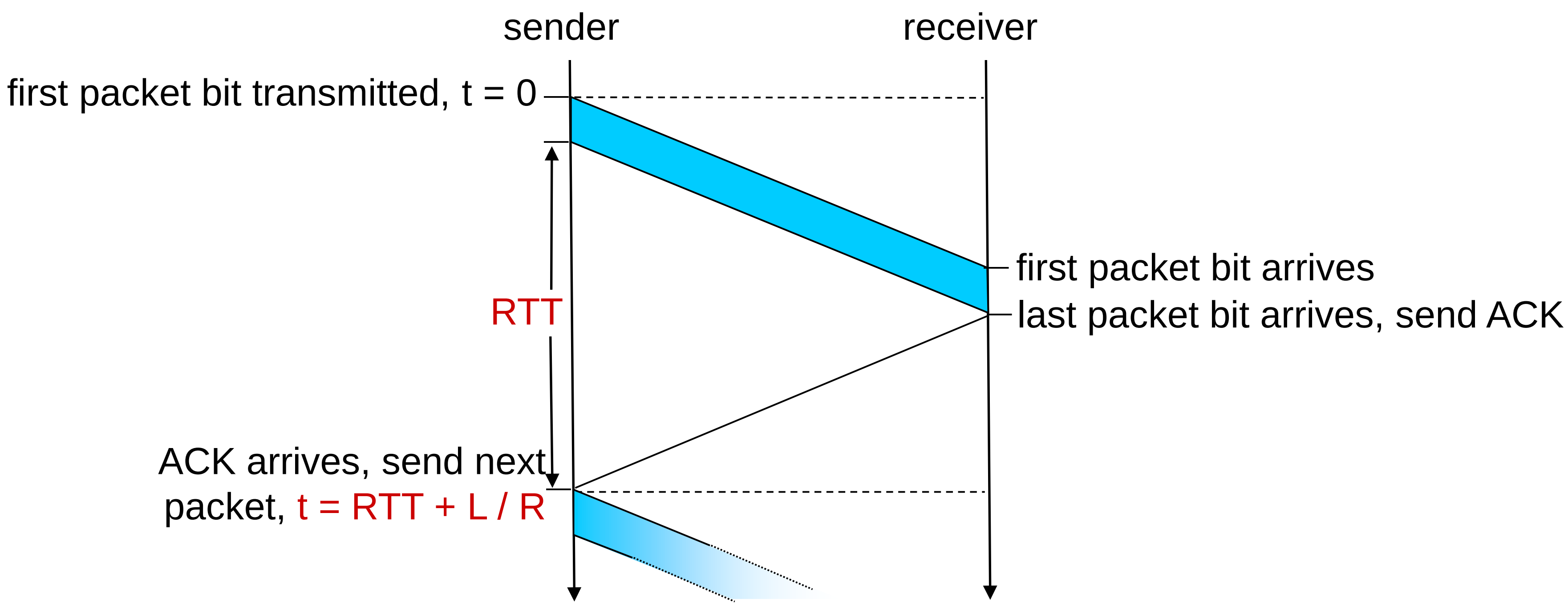
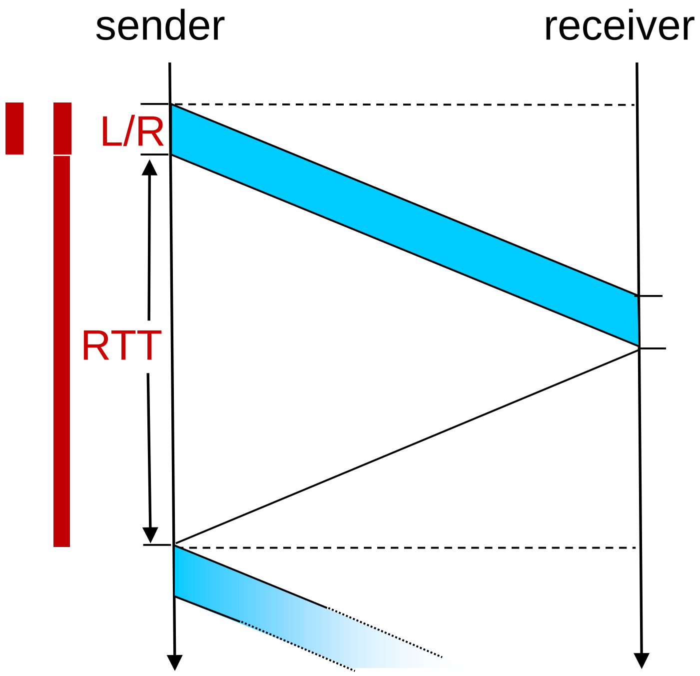


rdt3.0: stop-and-wait operation

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

$$= \frac{0.008}{30.008}$$

$$= 0.00027$$

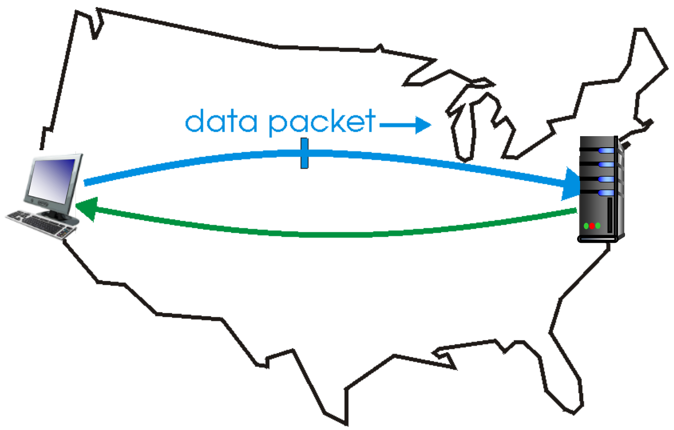


- rdt 3.0 protocol performance stinks!
- Protocol limits performance of underlying infrastructure (channel)

rdt3.0: pipelined protocols operation

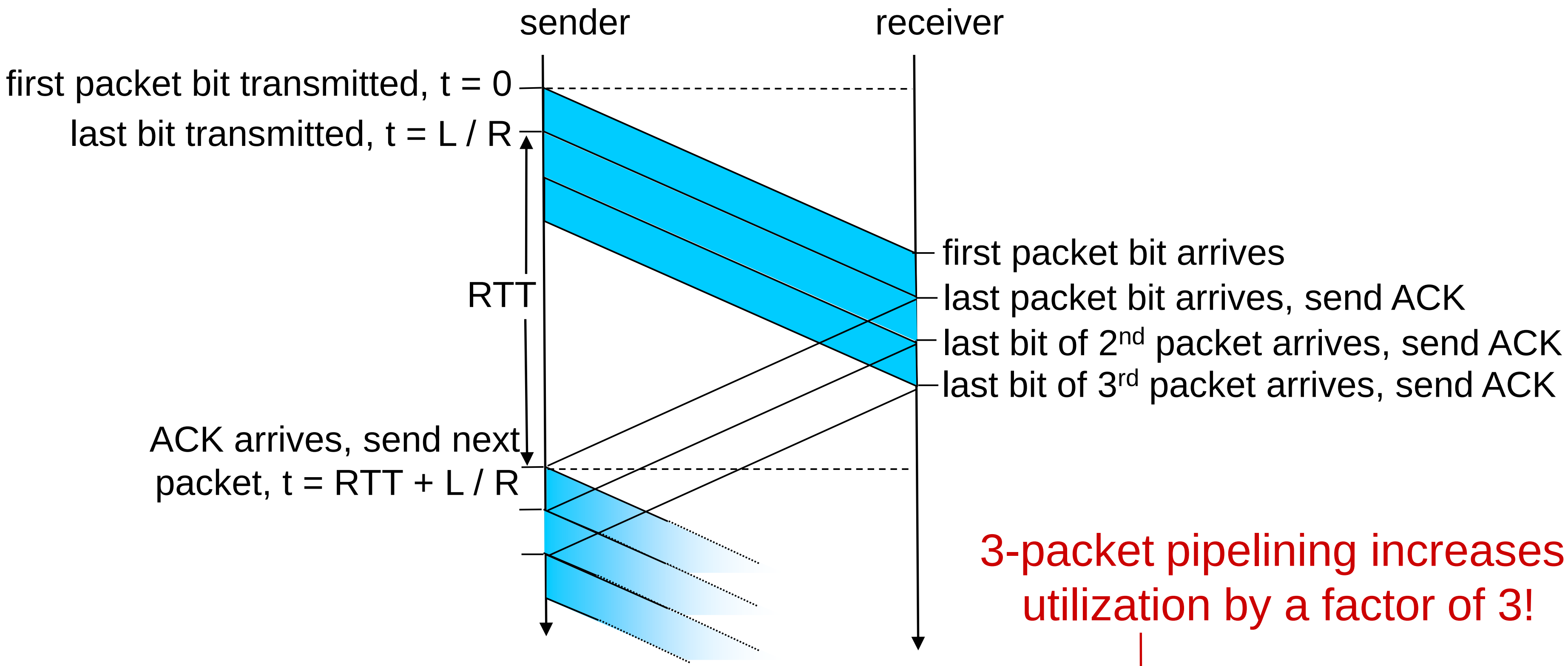
pipelining: sender allows multiple, yet-to-be-acknowledged packets

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



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

Pipelining: increased utilization

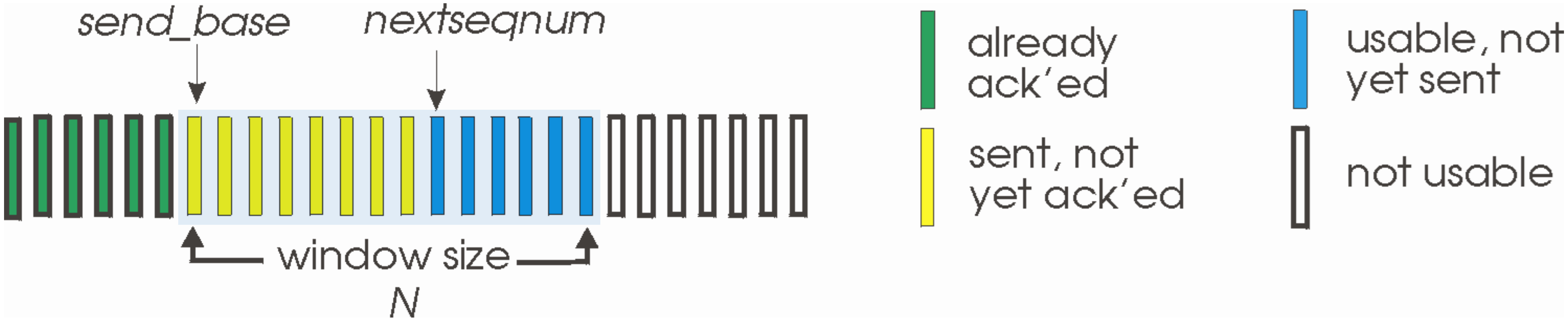


3-packet pipelining increases utilization by a factor of 3!

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

Go-Back-N: sender

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



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

Go-Back-N: receiver

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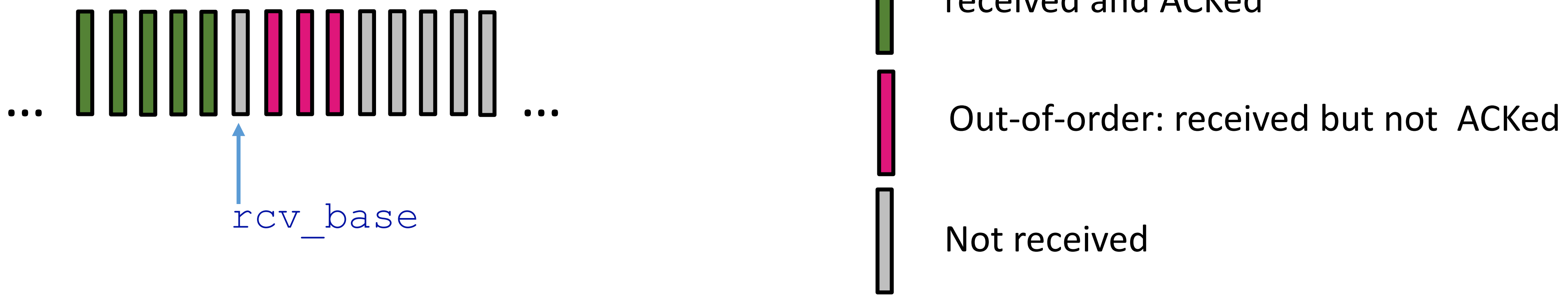
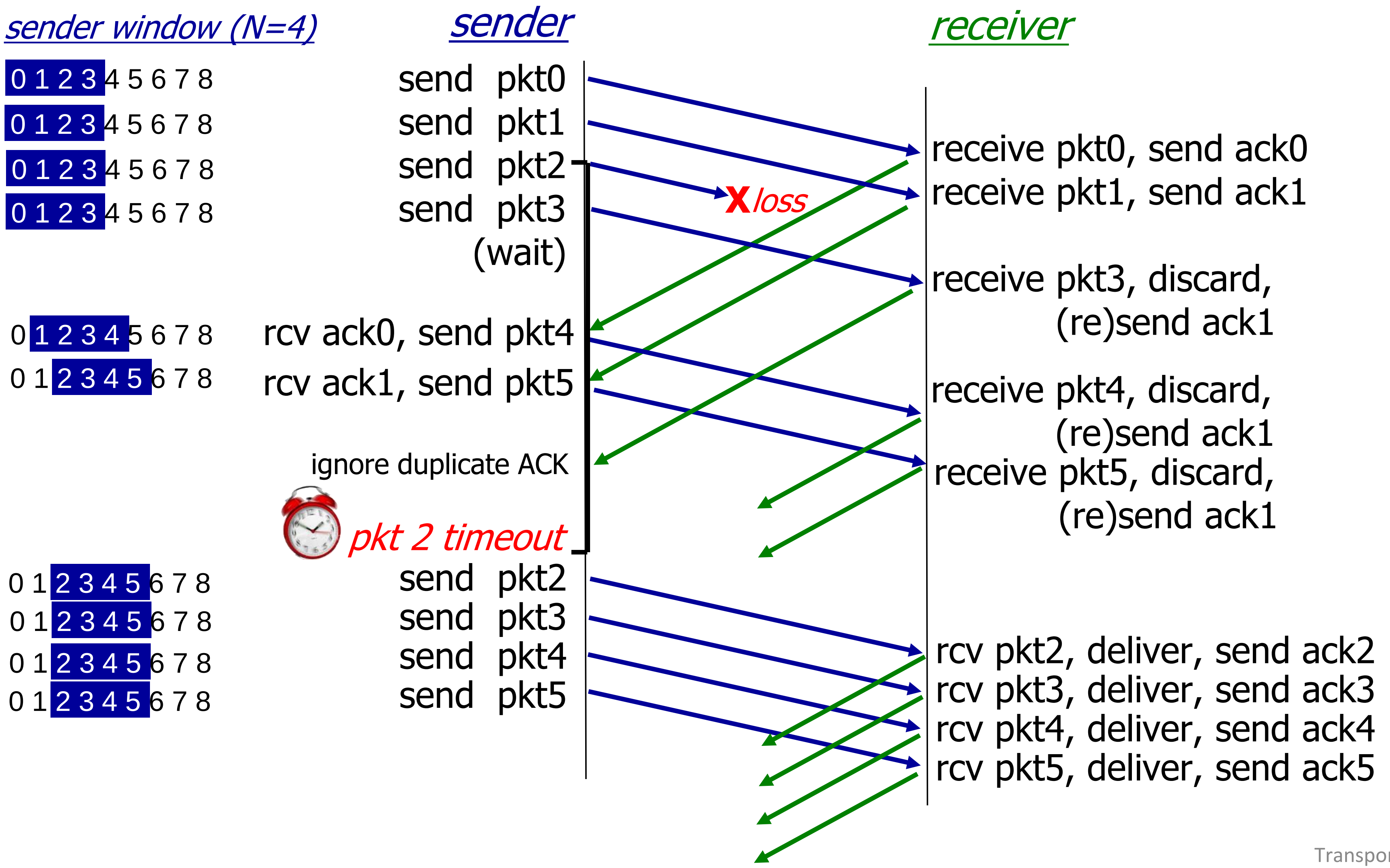
- ACK-only: always send ACK for correctly-received packet so far, with highest *in-order* seq #
  - may generate duplicate ACKs
- on receipt of out-of-order packet:
  - can discard (don't buffer) or buffer: an implementation decision

Receiver view of sequence number space:



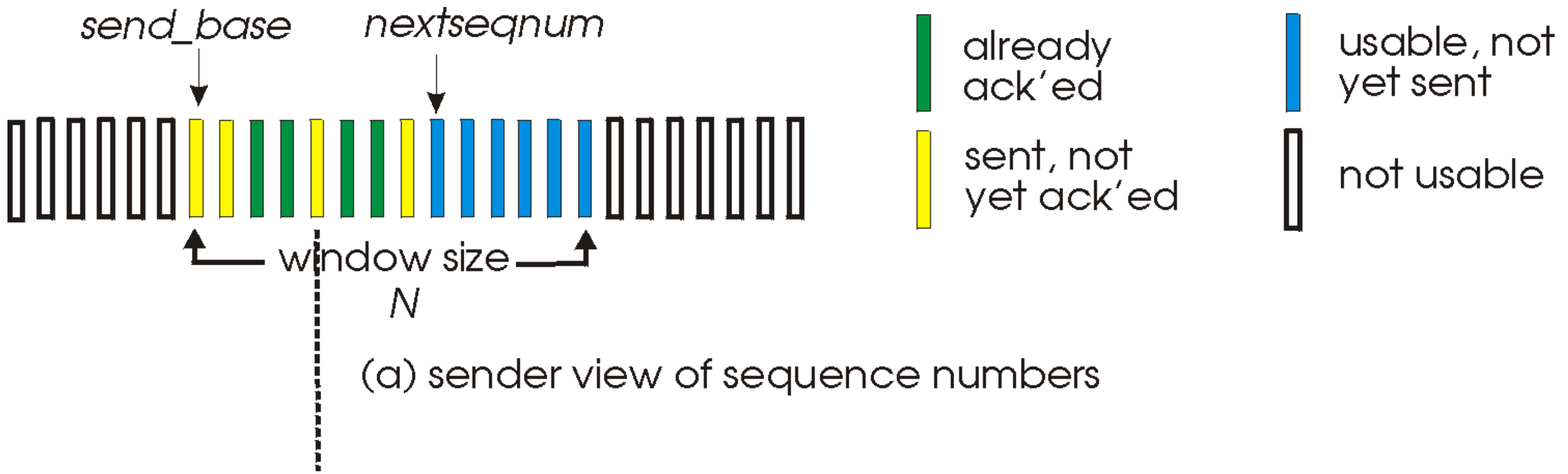
Go-Back-N in action



Selective repeat

- receiver *individually* acknowledges all correctly received packets
  - buffers packets, as needed, for eventual in-order delivery to upper layer
- sender times-out/retransmits individually for unACKed packets
  - sender maintains timer for each unACKed pkt
- sender window
  - *N* consecutive seq #s
  - limits seq #s of sent, unACKed packets

Selective repeat: sender, receiver windows



(a) sender view of sequence numbers

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

Chapter 3: roadmap

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- 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

TCP: overview

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- 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
- 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

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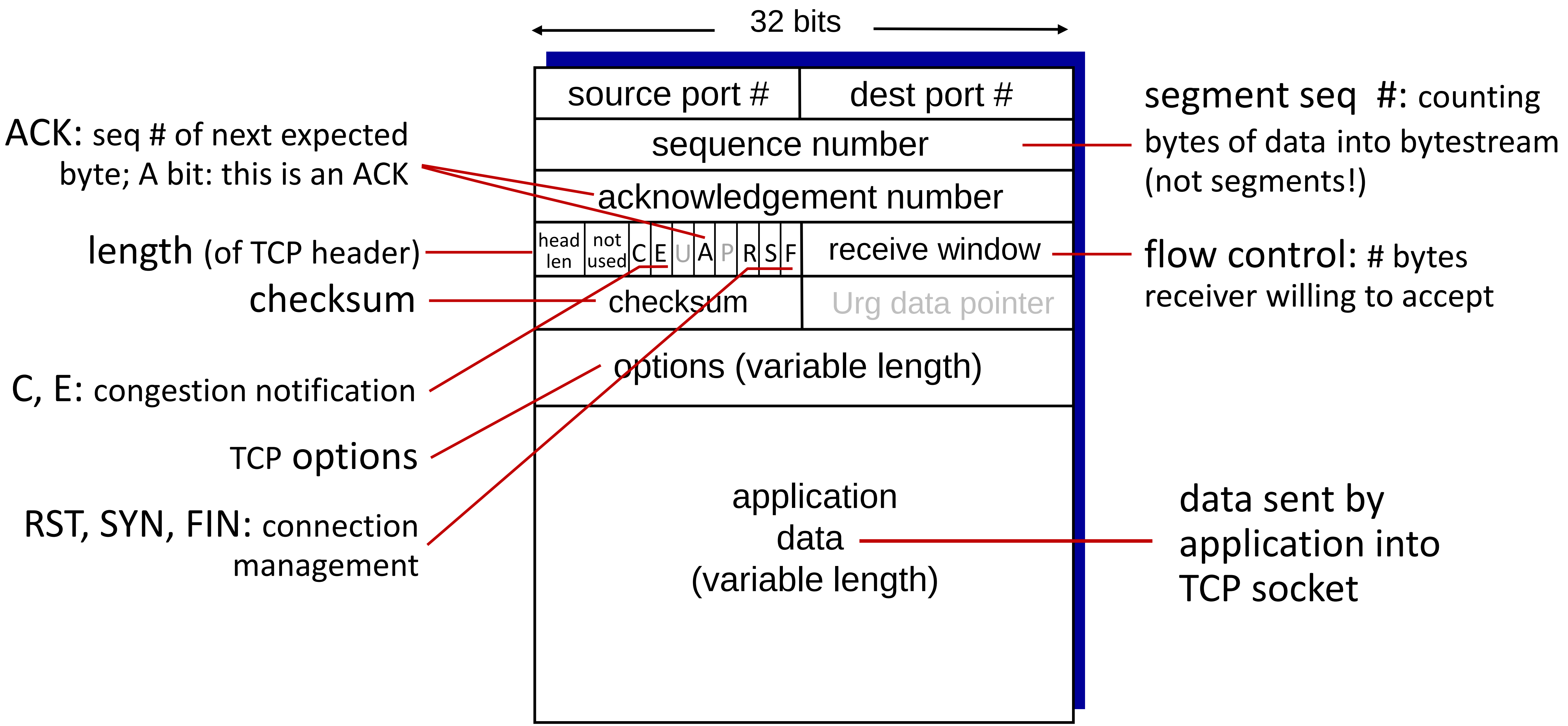


TCP sequence numbers, ACKs

Sequence numbers:

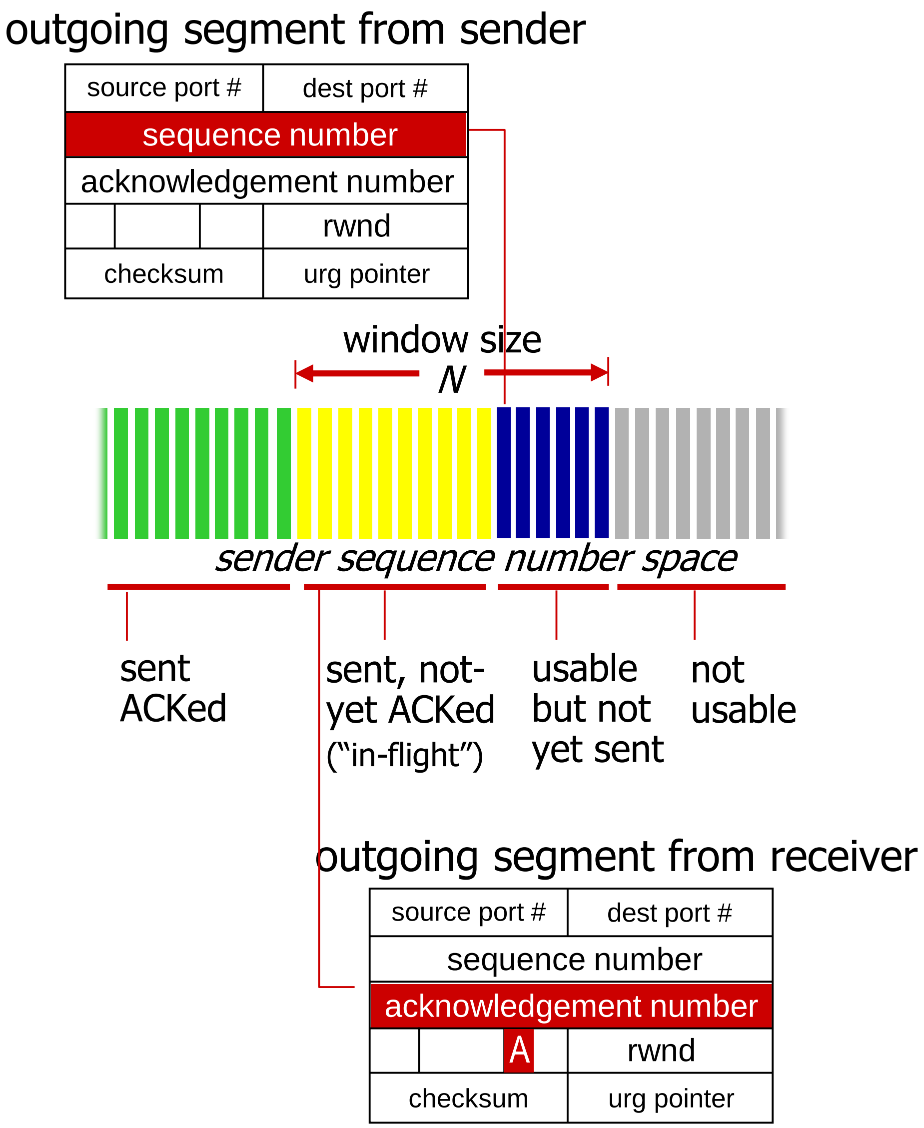
- byte stream "number" of first byte in segment's data

Acknowledgements:

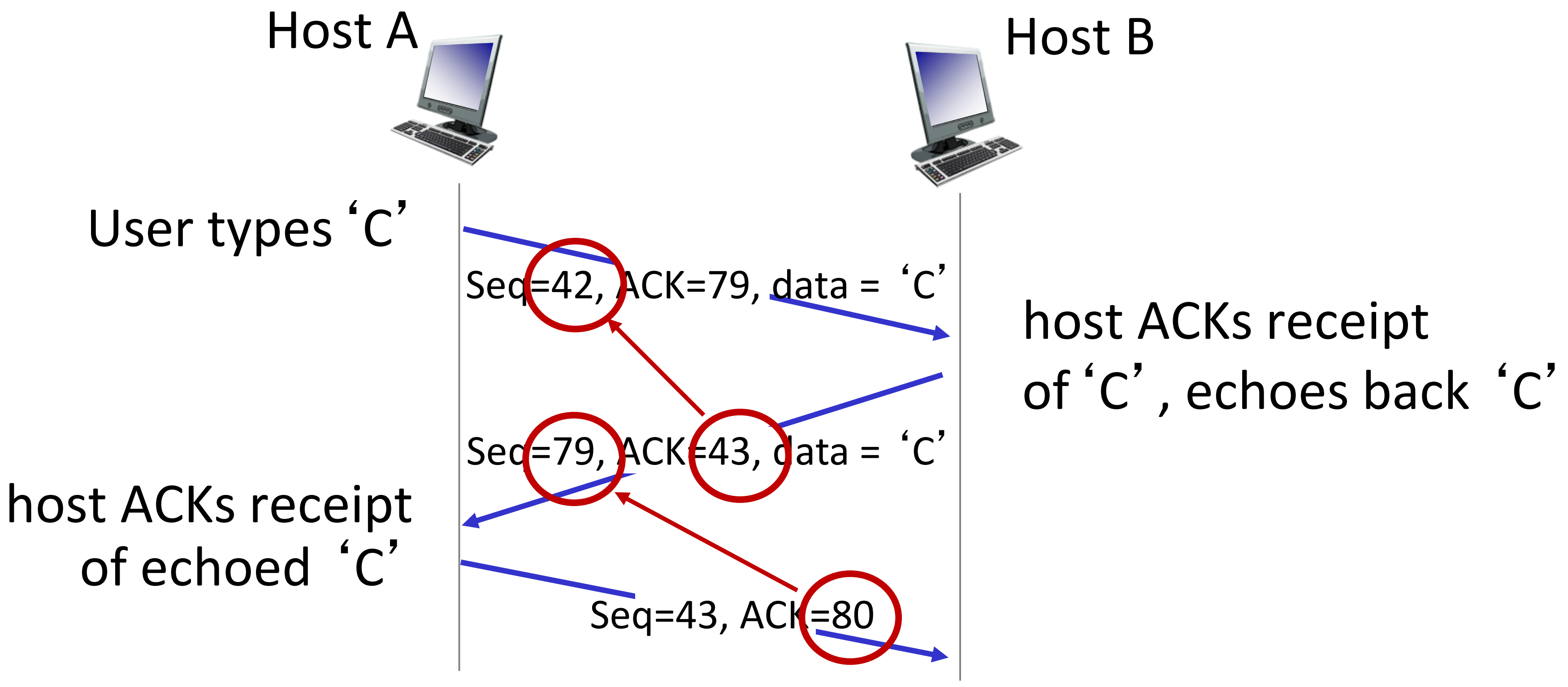
- seq # of next byte expected from other side
- cumulative ACK

Q: how receiver handles out-of-order segments

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



TCP sequence numbers, ACKs



simple telnet scenario

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

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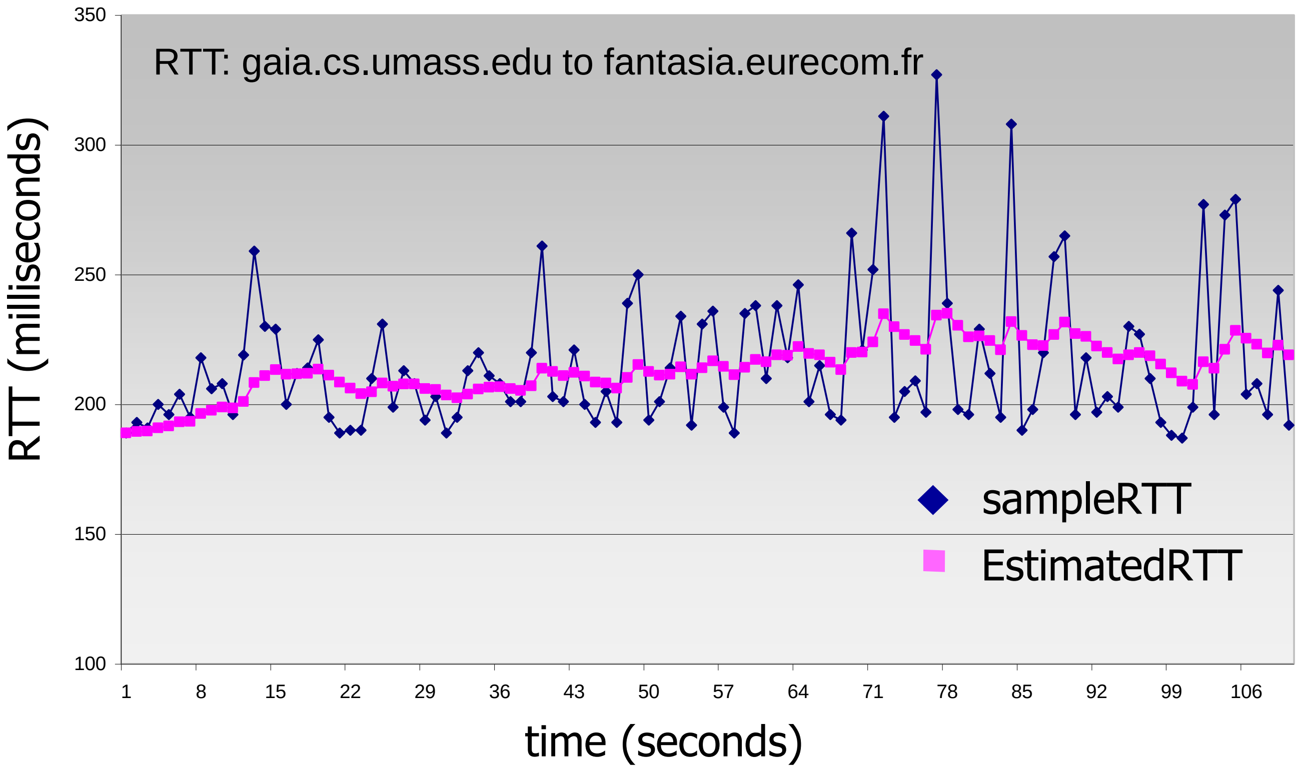
EstimatedRTT = (1- α)*EstimatedRTT + α*SampleRTT

- <u>e</u>xponential <u>w</u>eighted <u>m</u>oving <u>a</u>verage (EWMA)
- influence of past sample decreases exponentially fast
- typical value: α = 0.125



TCP round trip time, timeout

- timeout interval: **`EstimatedRTT`** plus "safety margin"
  - large variation in **`EstimatedRTT`**: want a larger safety margin

$$\texttt{TimeoutInterval} = \underbrace{\texttt{EstimatedRTT}}_{\text{estimated RTT}} + \underbrace{\texttt{4*DevRTT}}_{\text{"safety margin"}}$$

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

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

(typically, β = 0.25)

TCP Sender (simplified)

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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`**

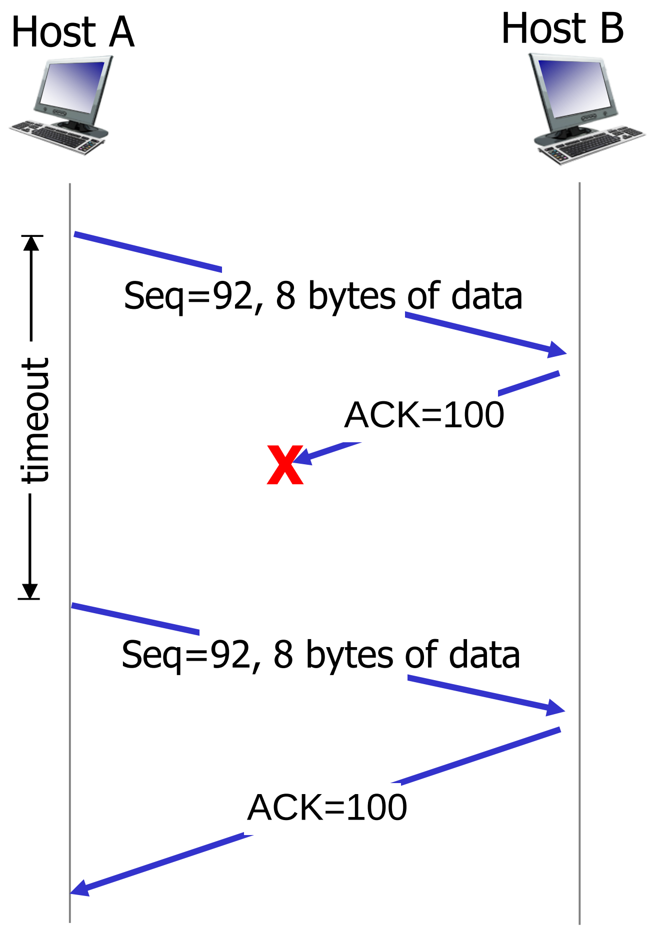
event: timeout

- retransmit segment that caused timeout
- restart timer

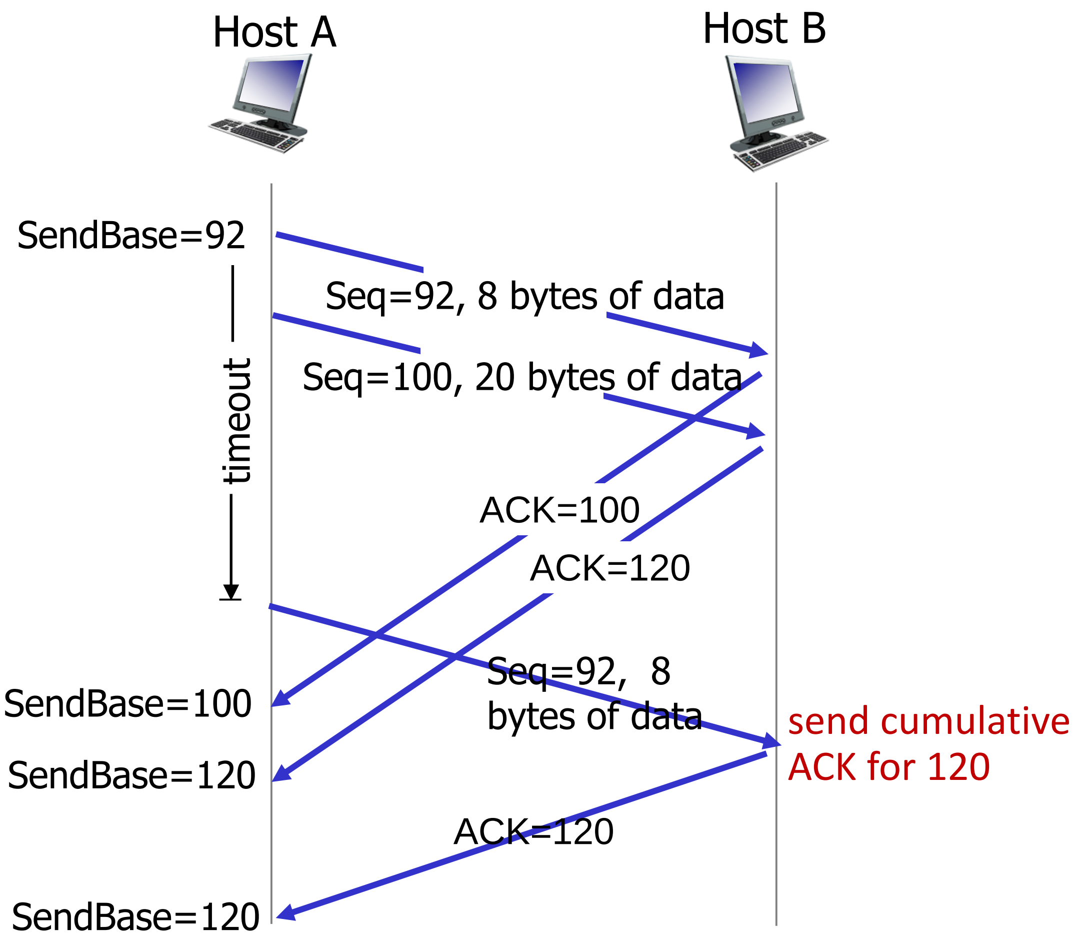
event: ACK received

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

TCP: retransmission scenarios

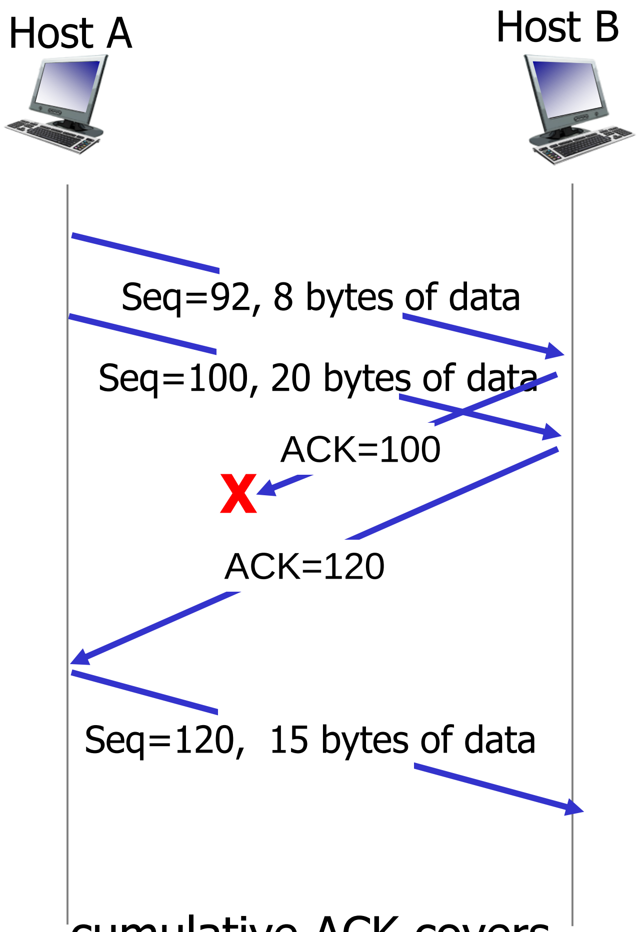


lost ACK scenario



premature timeout

TCP: retransmission scenarios



cumulative ACK covers for earlier lost ACK

TCP fast retransmit

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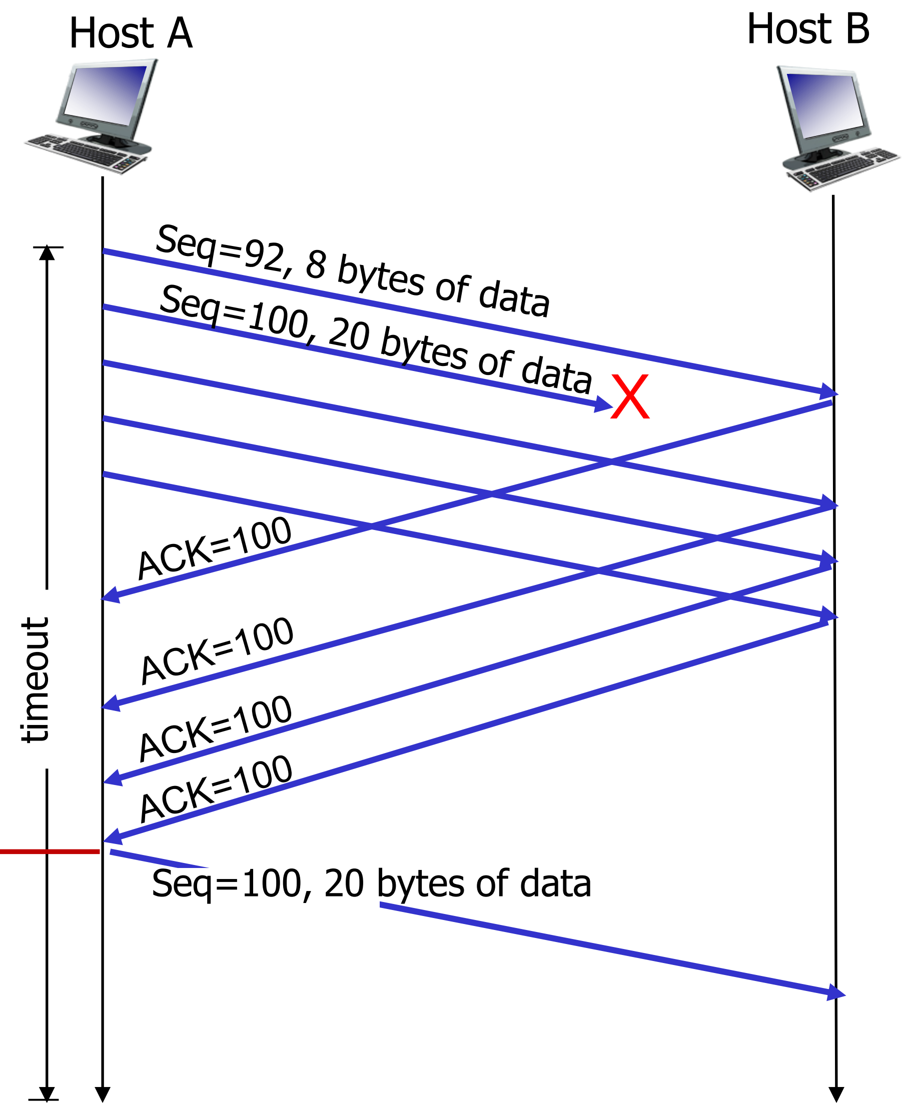
TCP fast retransmit

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

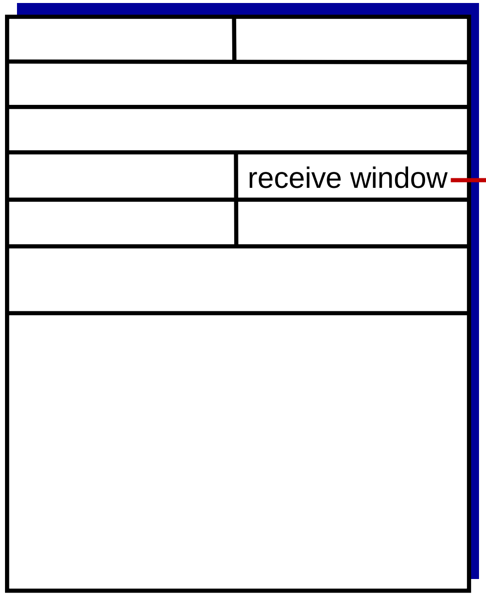
- likely that unACKed segment lost, so don't wait for timeout



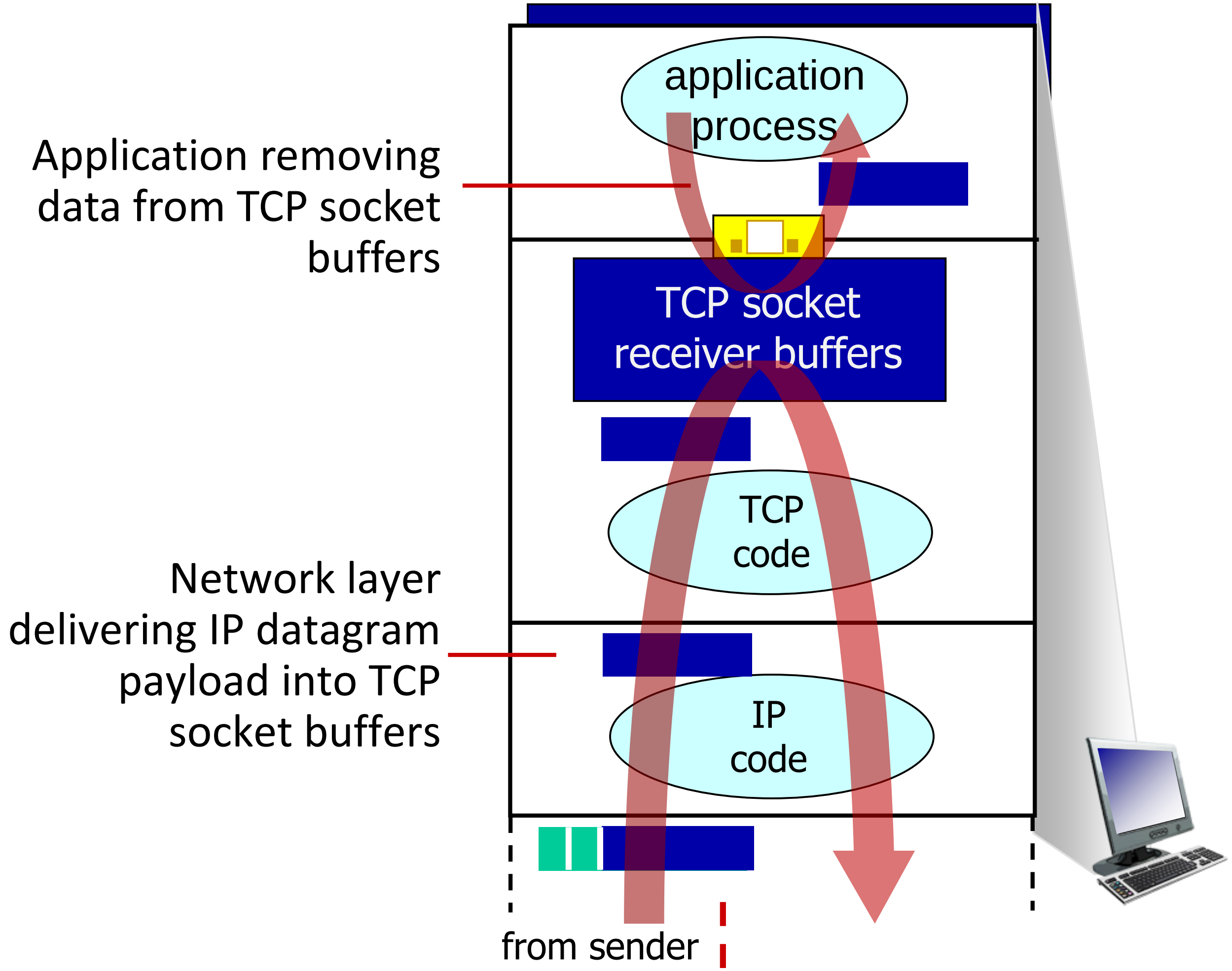
Receipt of three duplicate ACKs indicates 3 segments received after a missing segment – lost segment is likely. So retransmit!

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



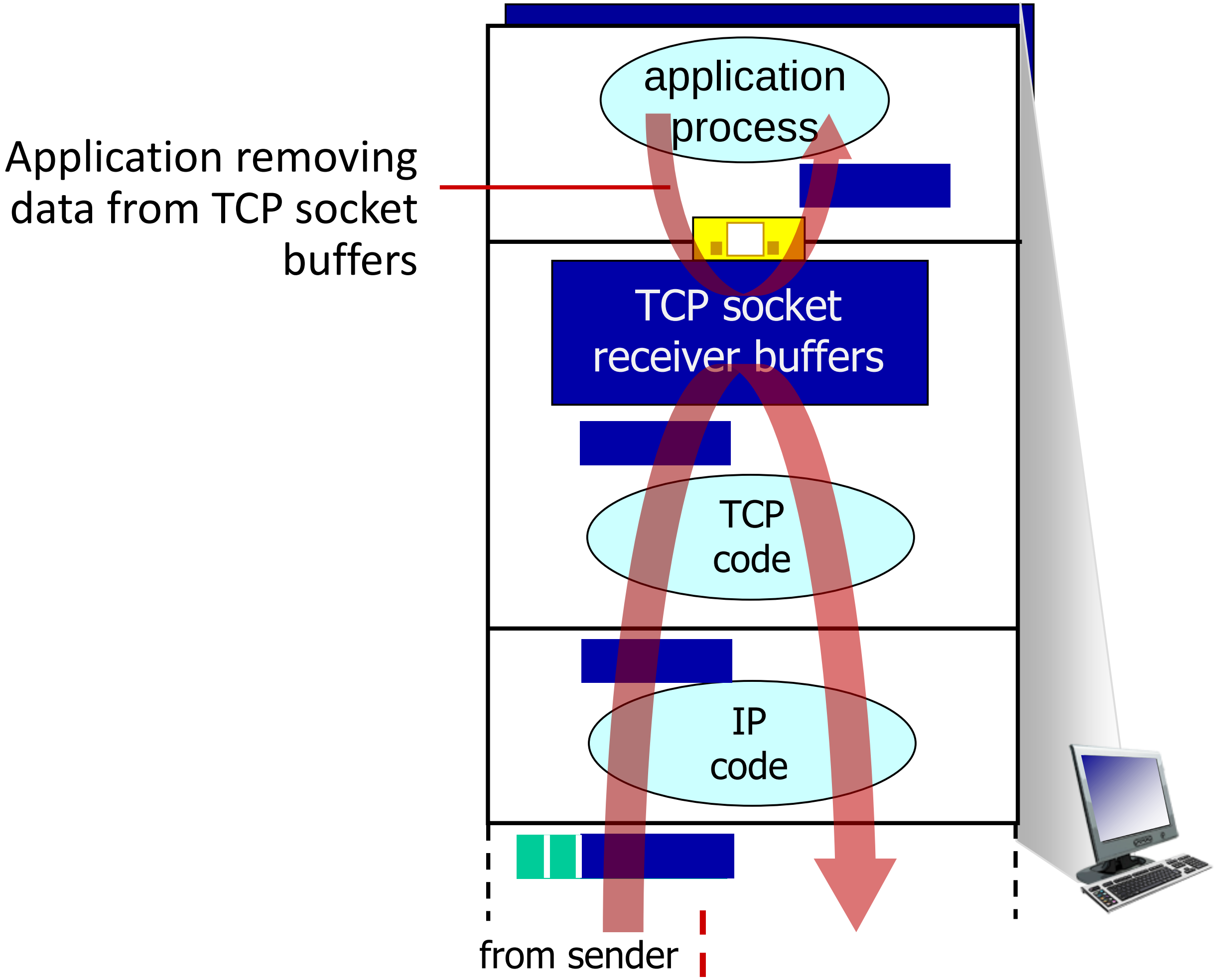
receiver protocol stack

TCP flow control

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

flow control

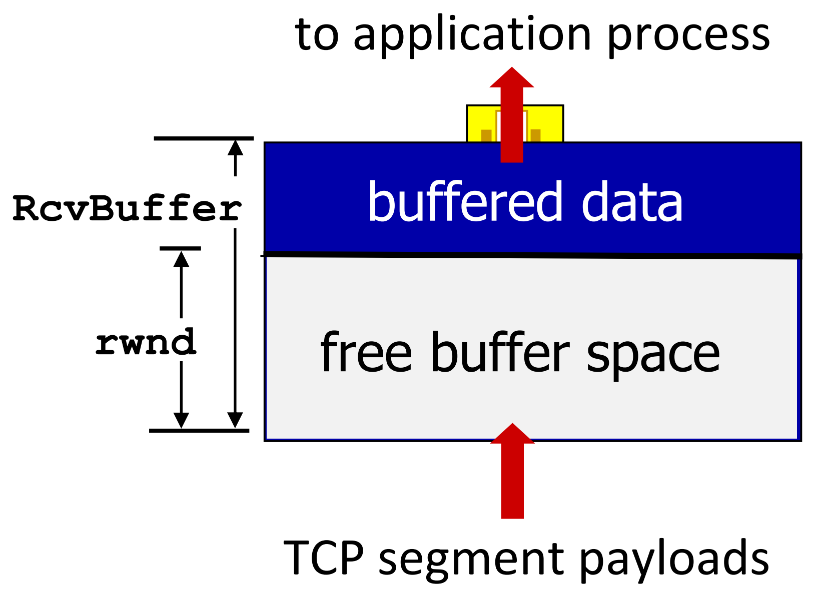
receiver controls sender, so sender won't overflow receiver's buffer by transmitting too much, too fast



receiver protocol stack

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

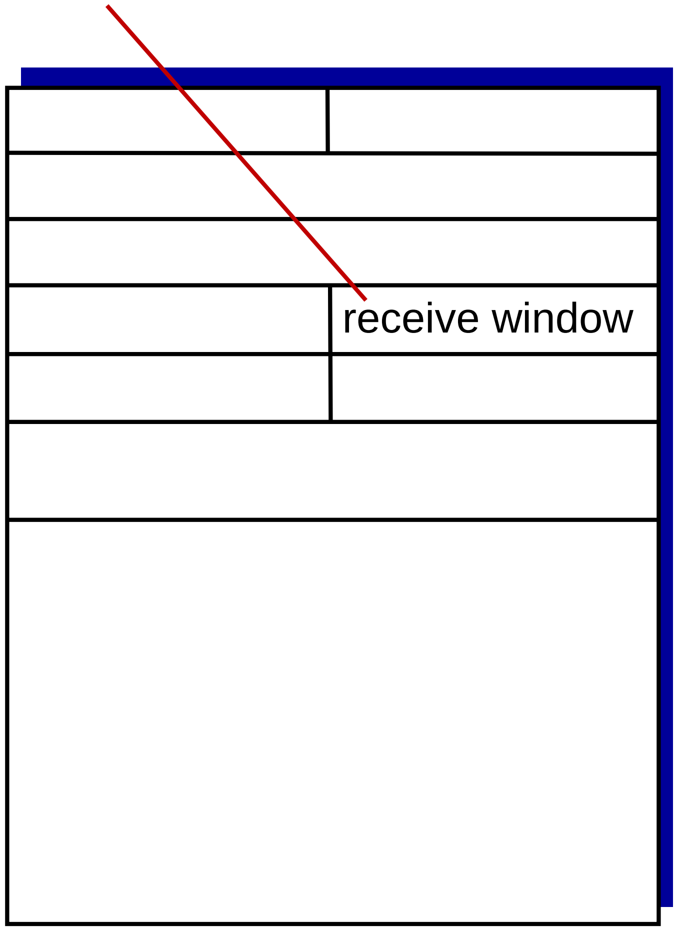


TCP receiver-side buffering

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

flow control: # bytes receiver willing to accept



TCP segment format

TCP connection management

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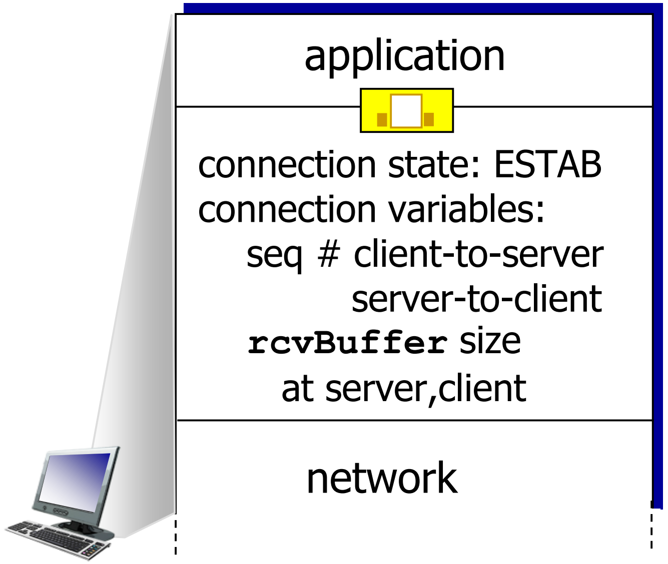
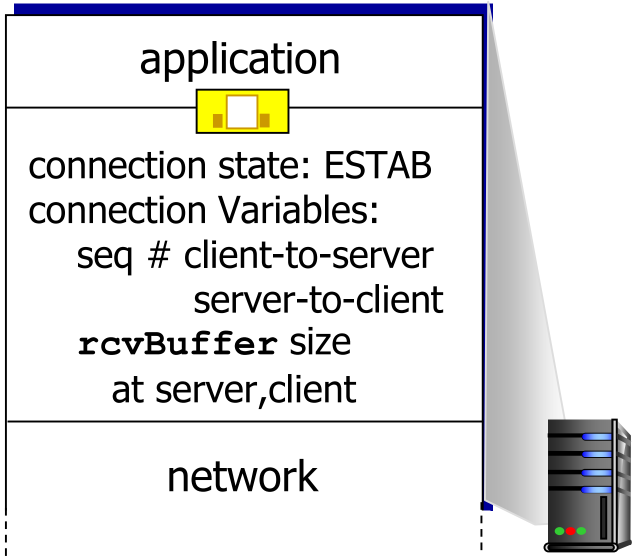
before exchanging data, sender/receiver "handshake":

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



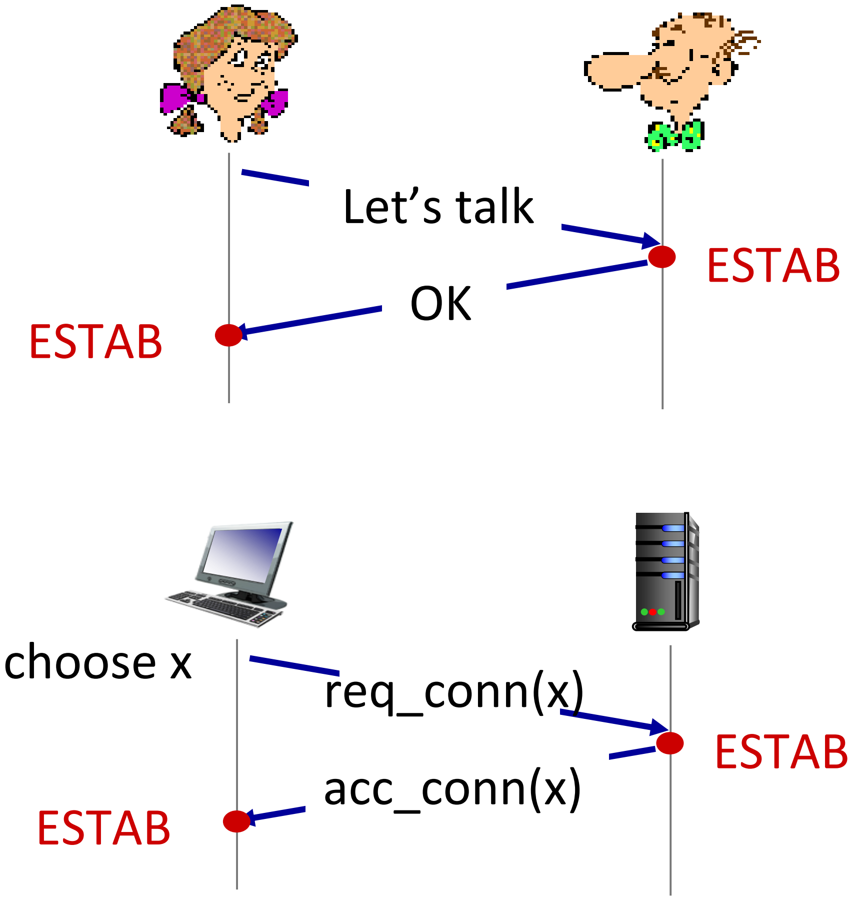



```
Socket clientSocket =
  newSocket("hostname","port number");
```

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

Agreeing to establish a connection

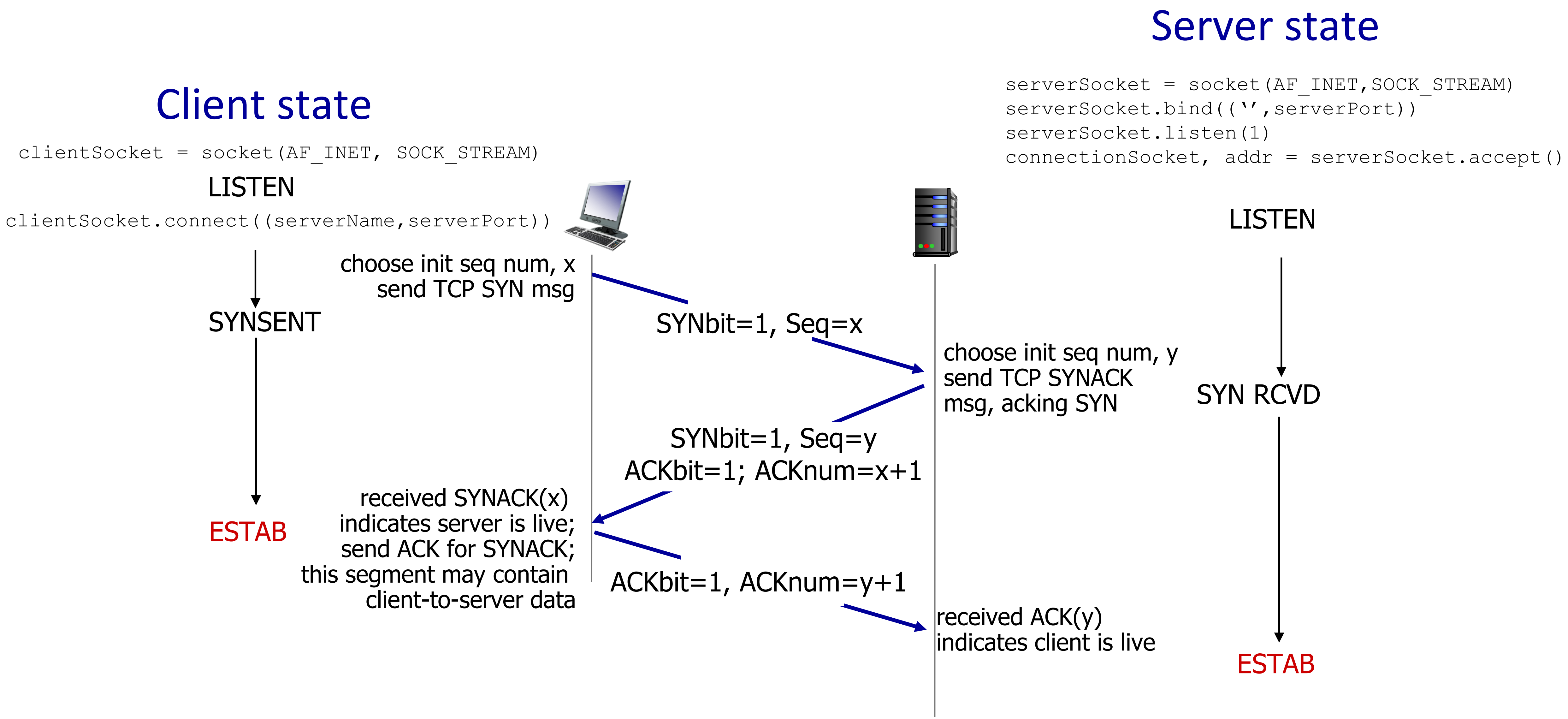
2-way handshake:



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

- variable delays
- retransmitted messages (e.g. req_conn(x)) due to message loss
- message reordering
- can't "see" other side

TCP 3-way handshake



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

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!

congestion control: too many senders, sending too fast

flow control: one sender too fast for one receiver

Approaches towards congestion control

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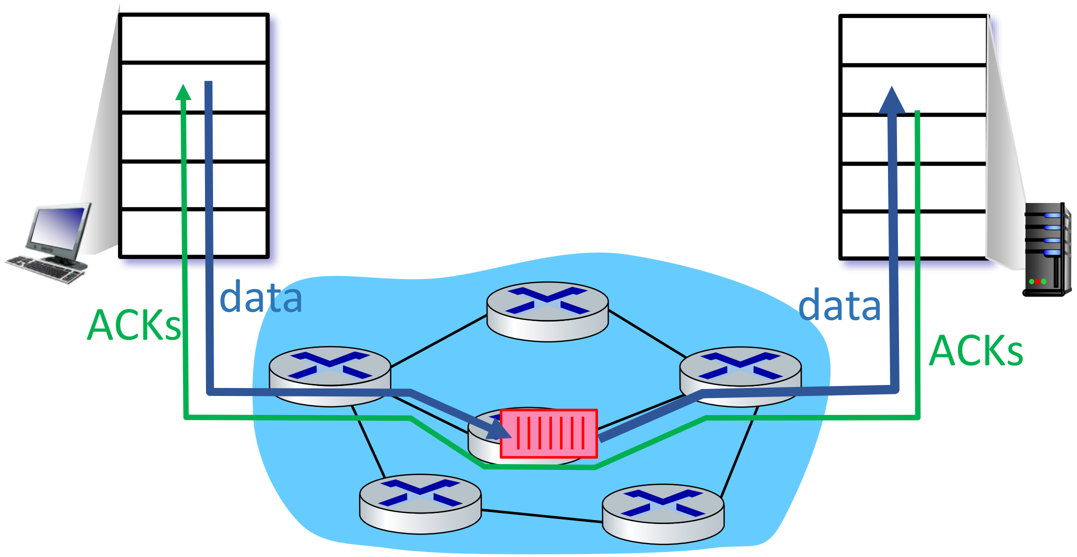
End-end congestion control:

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



Approaches towards congestion control

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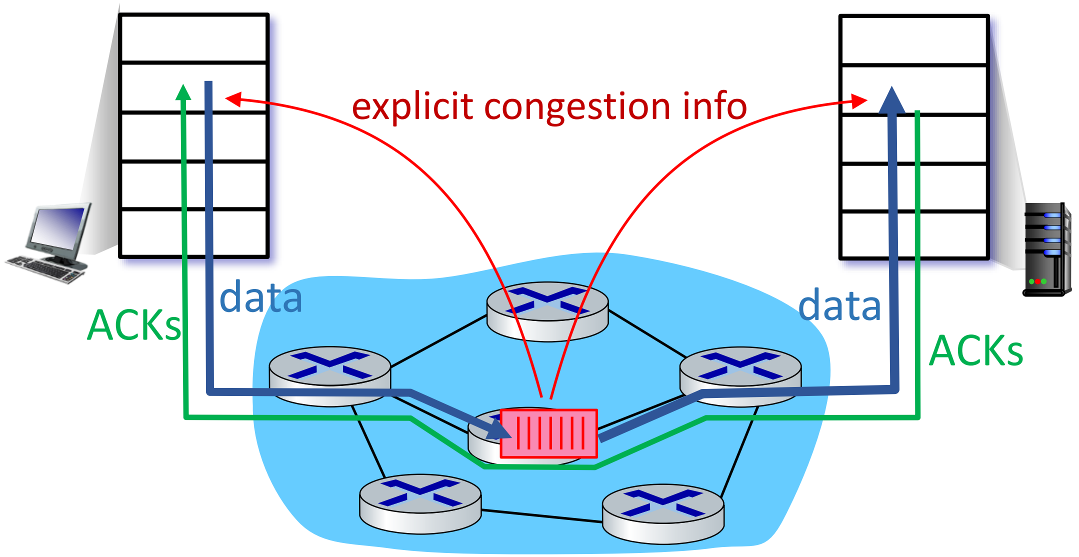
Network-assisted congestion control:

- 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



Chapter 3: roadmap

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- Transport-layer services
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- TCP congestion control

TCP congestion control: AIMD

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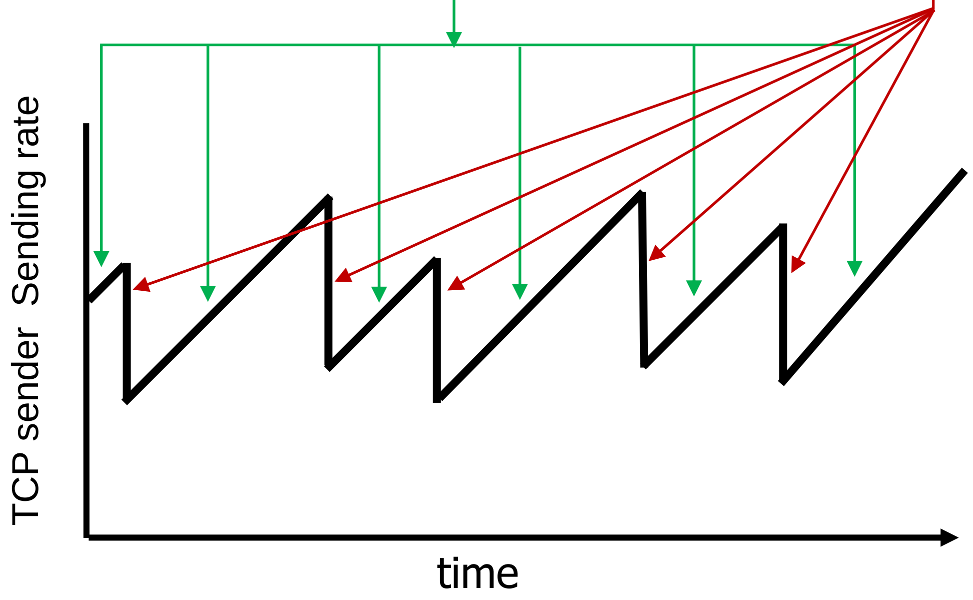
- *approach:* senders can increase sending rate until packet loss (congestion) occurs, then decrease sending rate on loss event

Additive Increase

increase sending rate by 1 maximum segment size every RTT until loss detected

Multiplicative Decrease

cut sending rate in half at each loss event (Reno)



TCP AIMD: more

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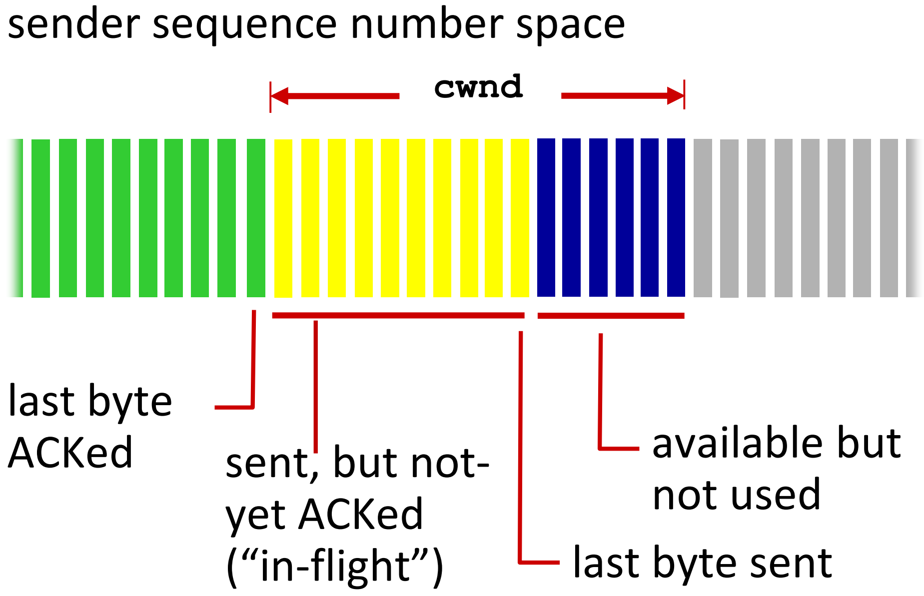
Multiplicative decrease detail: sending rate is

- Cut in half on loss detected by triple duplicate ACK (TCP Reno)
- Cut to 1 MSS (maximum segment size) when loss detected by timeout (TCP Tahoe)

Why AIMD?

- AIMD – a distributed, asynchronous algorithm – has been shown to:
  - optimize congested flow rates network wide!
  - have desirable stability properties

TCP congestion control: details



TCP sending behavior:

- *roughly:* send `cwnd` bytes, wait RTT for ACKS, then send more bytes

$$\text{TCP rate} \approx \frac{\texttt{cwnd}}{\text{RTT}} \text{ bytes/sec}$$

- TCP sender limits transmission: **`LastByteSent- LastByteAcked ≤ cwnd`**
- `cwnd` is dynamically adjusted in response to observed network congestion (implementing TCP congestion control)

TCP slow start

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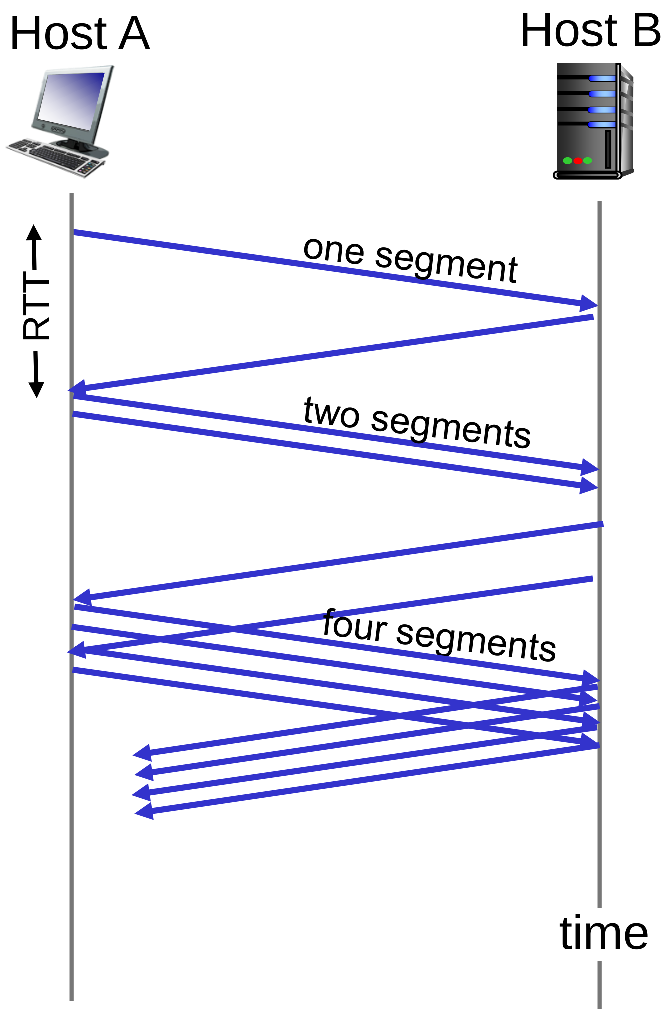
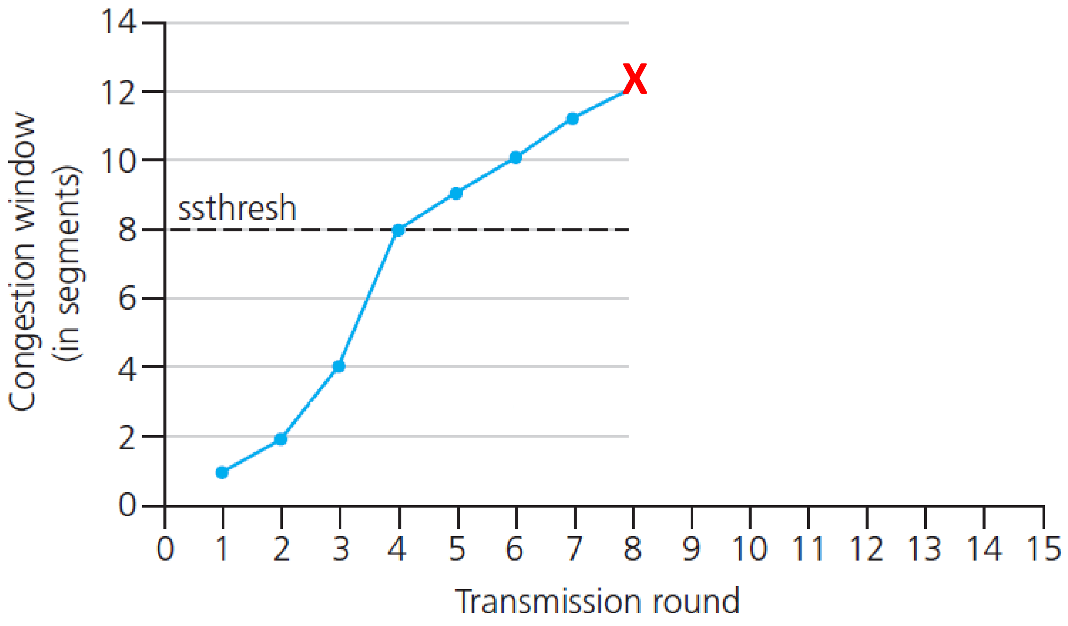
- when connection begins, increase rate exponentially until first loss event:
  - initially **`cwnd`** = 1 MSS
  - double **`cwnd`** every RTT
  - done by incrementing **`cwnd`** for every ACK received
- *summary:* initial rate is slow, but ramps up exponentially fast



TCP: from slow start to congestion avoidance

Q: when should the exponential increase switch to linear?

A: when **cwnd** gets to 1/2 of its value before timeout.



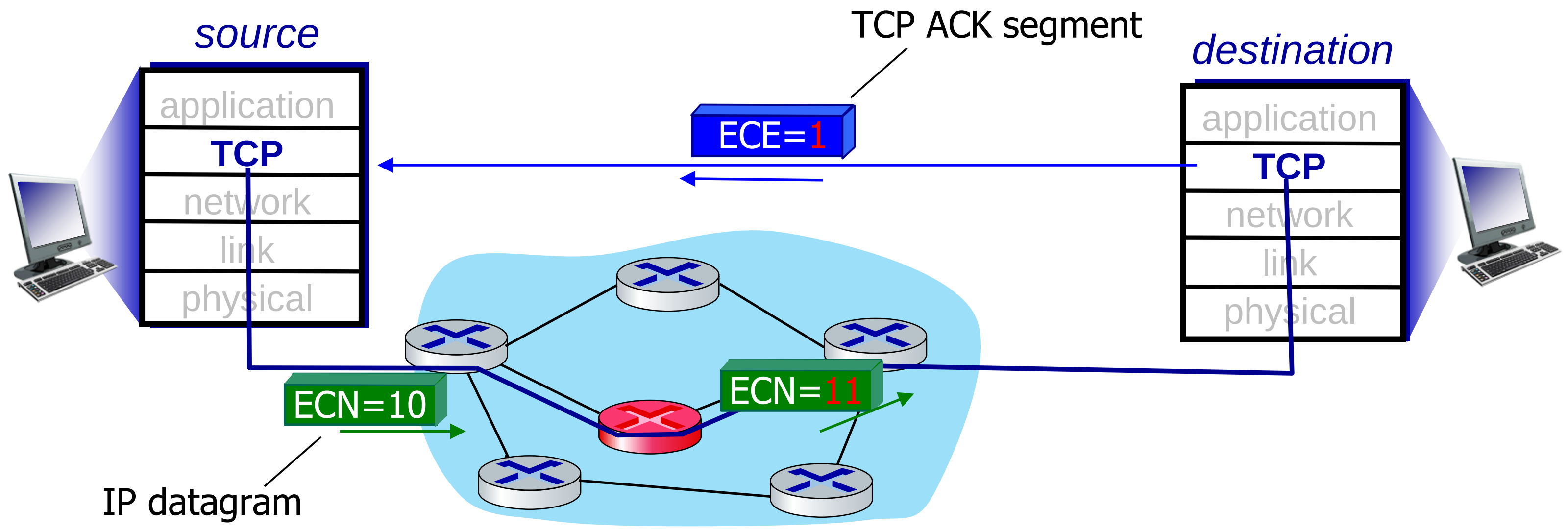
Implementation:

- variable **ssthresh**
- on loss event, **ssthresh** is set to 1/2 of **cwnd** just before loss event

Explicit congestion notification (ECN)

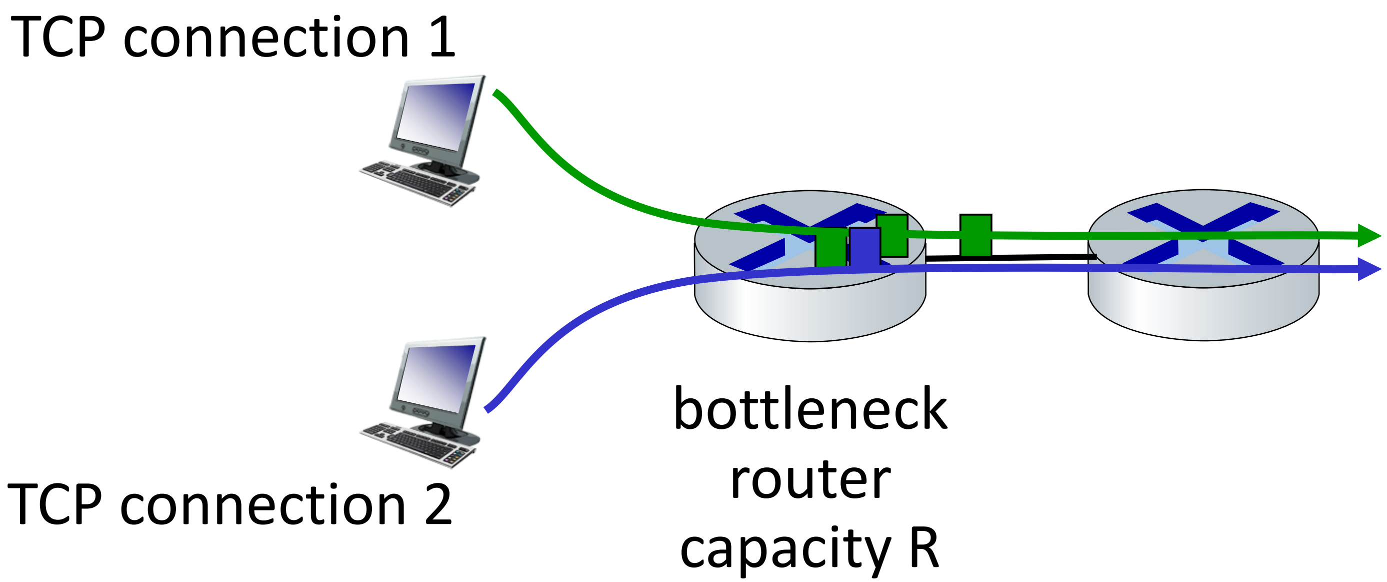
TCP deployments often implement *network-assisted* congestion control:

- two bits in IP header (ToS field) marked *by network router* to indicate congestion
  - *policy* to determine marking is chosen by network operator
- congestion indication carried to destination
- destination sets ECE bit on ACK segment to notify sender of congestion
- involves both IP (IP header ECN bit marking) and TCP (TCP header C,E bit marking)



TCP fairness

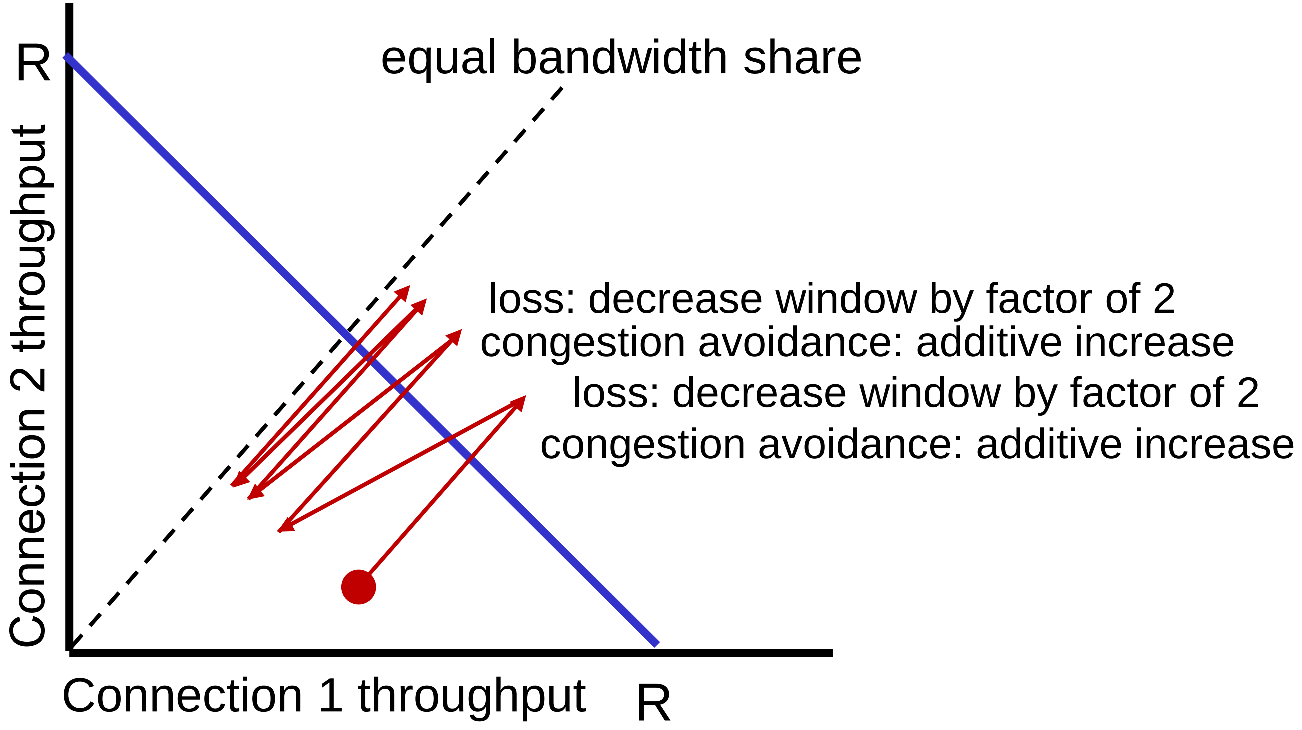
Fairness goal: if *K* TCP sessions share same bottleneck link of bandwidth *R*, each should have average rate of *R/K*



Q: is TCP Fair?

Example: two competing TCP sessions:

- additive increase gives slope of 1, as throughout increases
- multiplicative decrease decreases throughput proportionally



Is TCP fair?

A: Yes, under idealized assumptions:

- same RTT
- fixed number of sessions only in congestion avoidance