CS5222 Computer Networks and Internets

Transport Layer

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Slides based on book *Computer Networking: A Top-Down Approach.*

Transport layer: overview

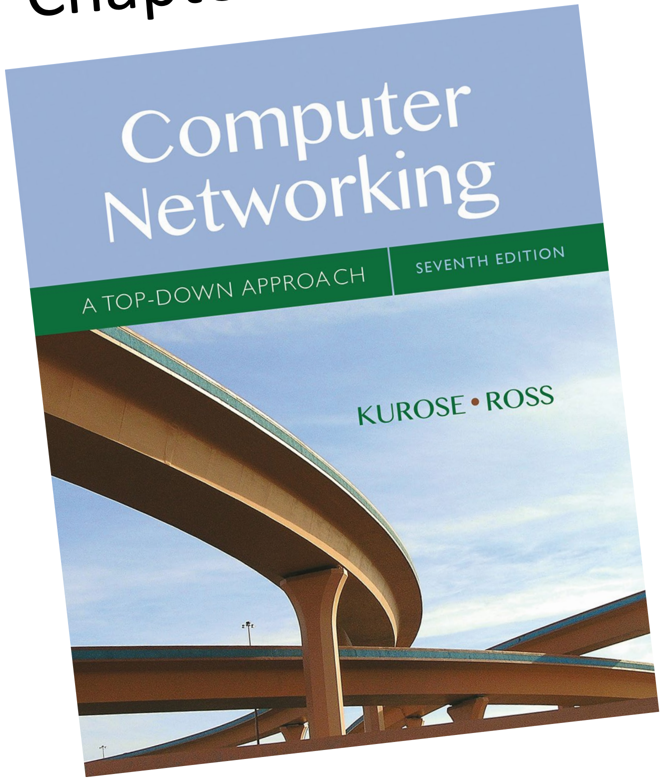
Our goal:

- 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

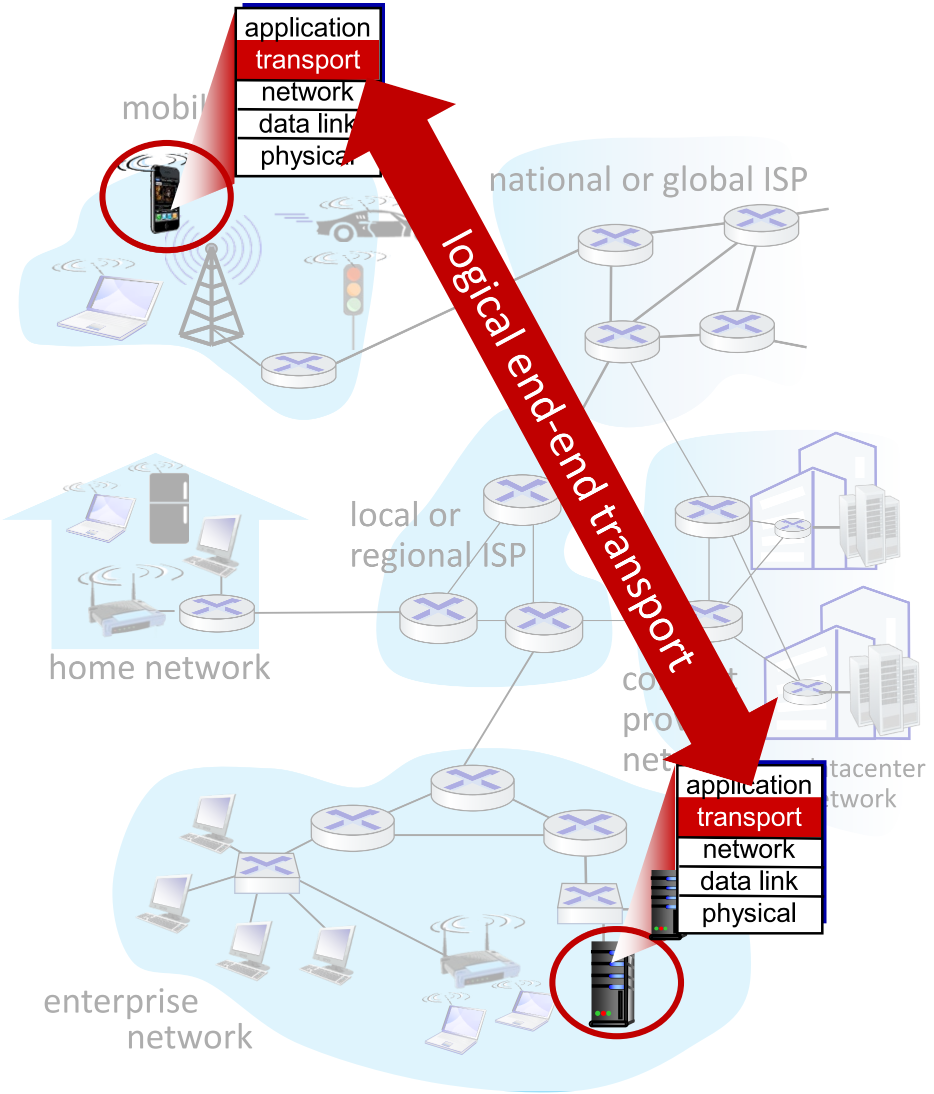
- Transport-layer services
- Multiplexing and demultiplexing
- Connectionless transport: UDP
- Principles of reliable data transfer
- Connection-oriented transport: TCP
- Principles of congestion control
- TCP congestion control

Chapter 3 in



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

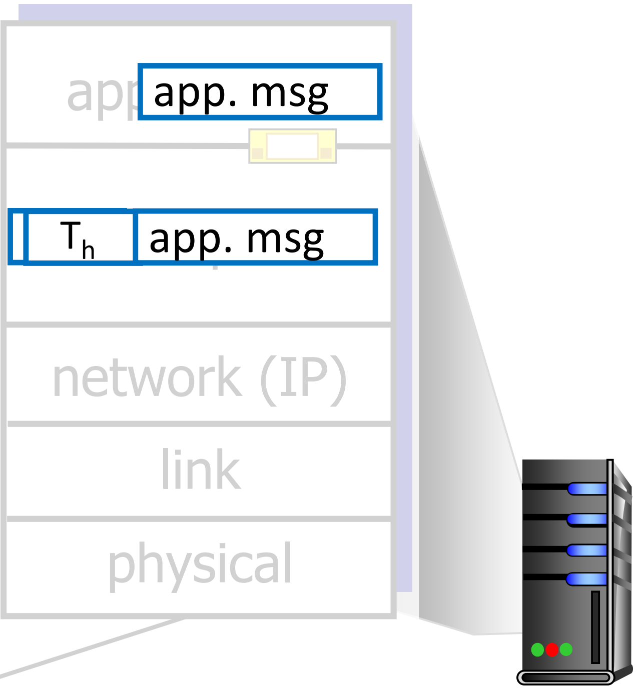
- network layer:
  logical communication between *hosts*

- transport layer*:*
  logical communication between *processes*

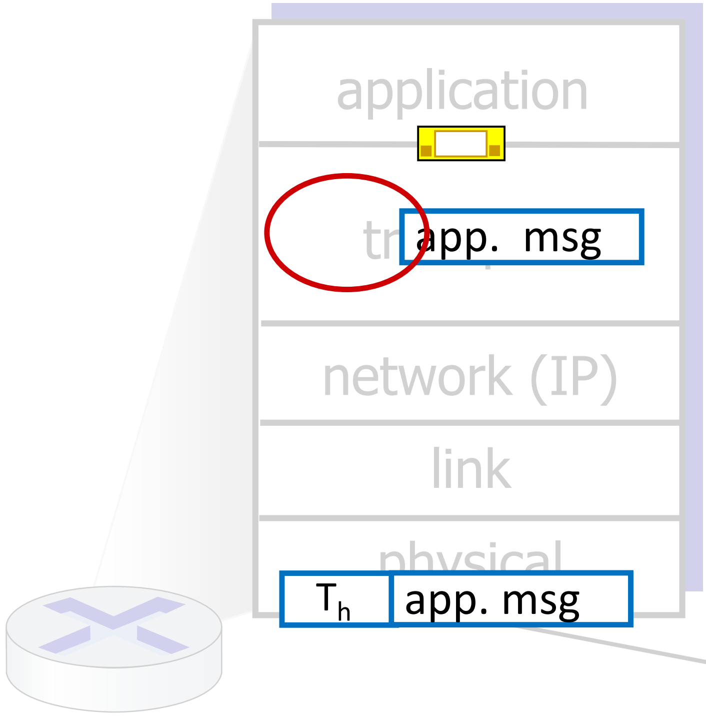
Transport Layer Actions

Sender:

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

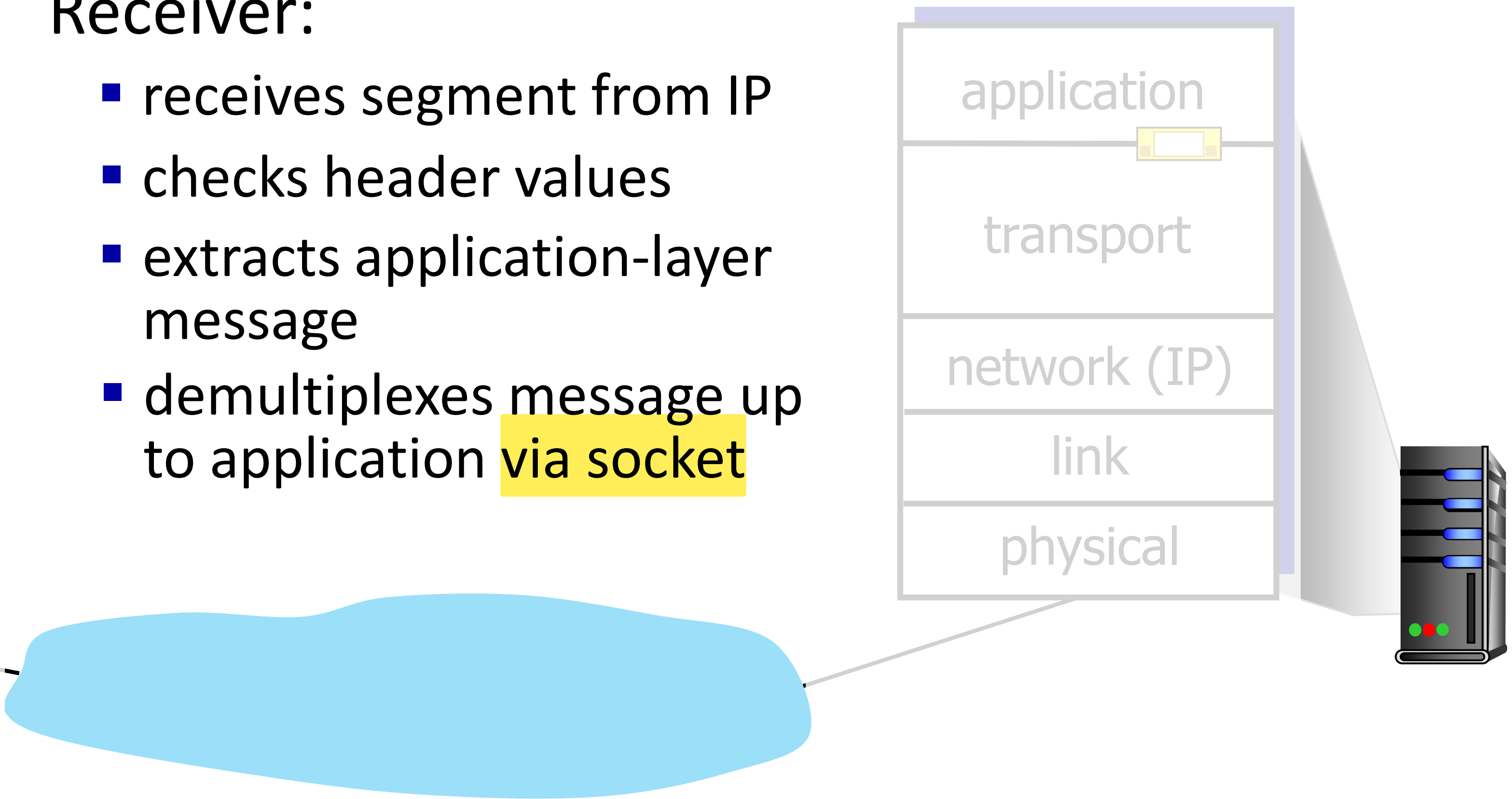


Transport Layer Actions



Receiver:

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



→ Canvas: Quiz (ungraded) - A Household Analogy

Quiz on Canvas: A Household Analogy

- Consider two houses, one on the East Coast and the other on the West Coast, with each house being home to a dozen kids. The kids in the East Coast household are cousins of the kids in the West Coast household.

- The kids in the two households love to write to each other—each kid writes each cousin every week, with each letter delivered by the traditional postal service.

- In each of the households there is one kid—**Ann** in the West Coast house and **Bill** in the East Coast house—responsible for mail collection and mail distribution.

- Each week Ann visits all her brothers and sisters, collects the mail, and gives the mail to a postal-service mail carrier, who makes daily visits to the house. When letters arrive at the West Coast house, Ann also has the job of distributing the mail to her brothers and sisters. Bill has a similar job on the East Coast.

Solution: Household Analogy

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

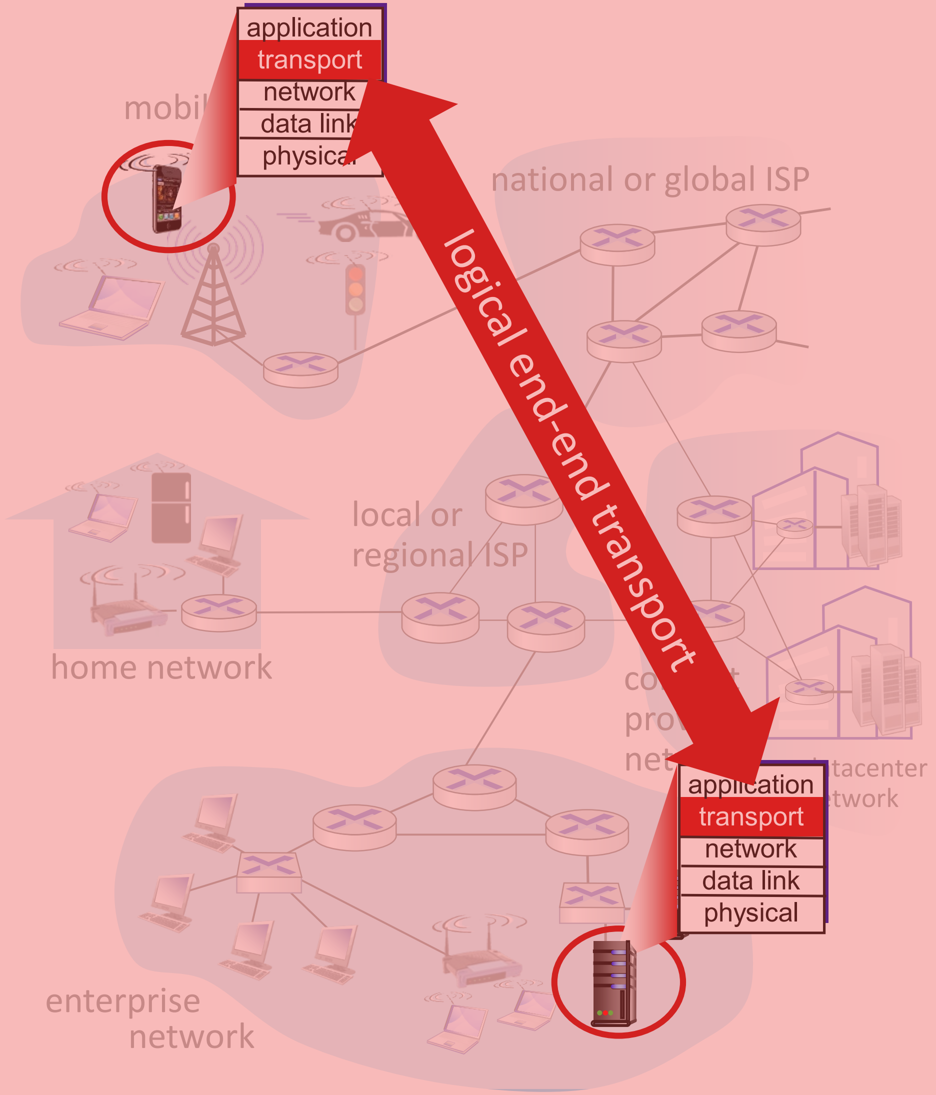
- hosts = houses
- processes = kids
- app messages = letters in envelopes
- transport protocol = Ann and Bill

这个很乐

- network-layer protocol = postal service

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



Transport Layer: Roadmap

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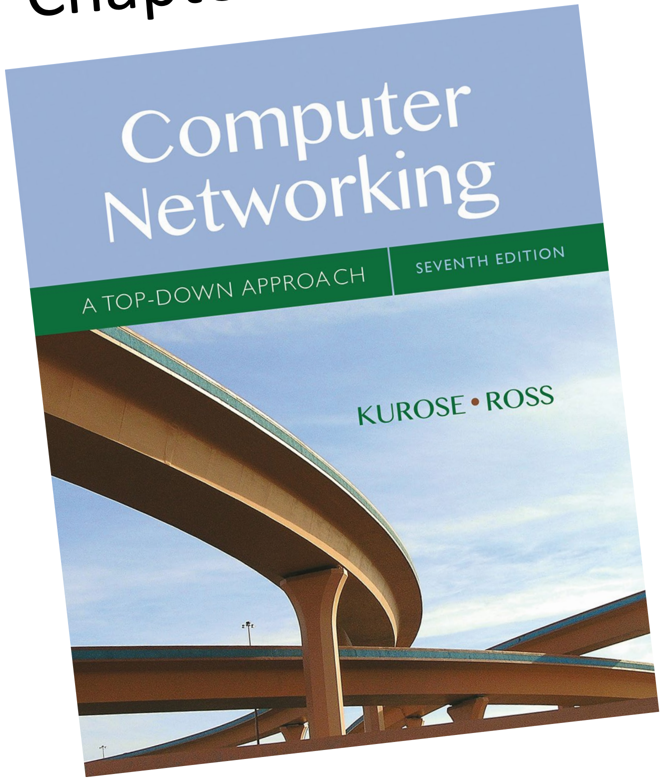

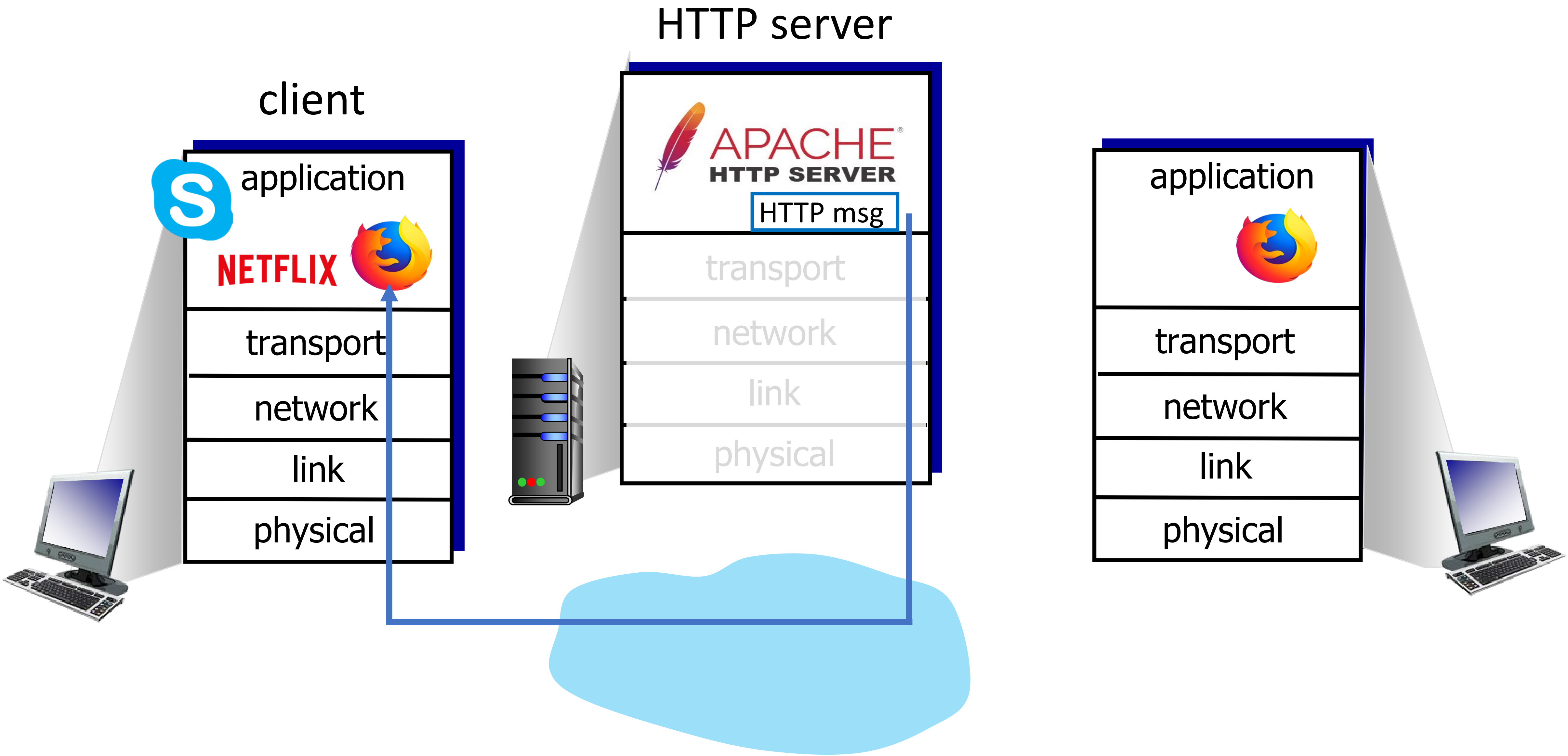

HTTP server
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transport
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APACHE
HTTP SERVER
HTTP msg
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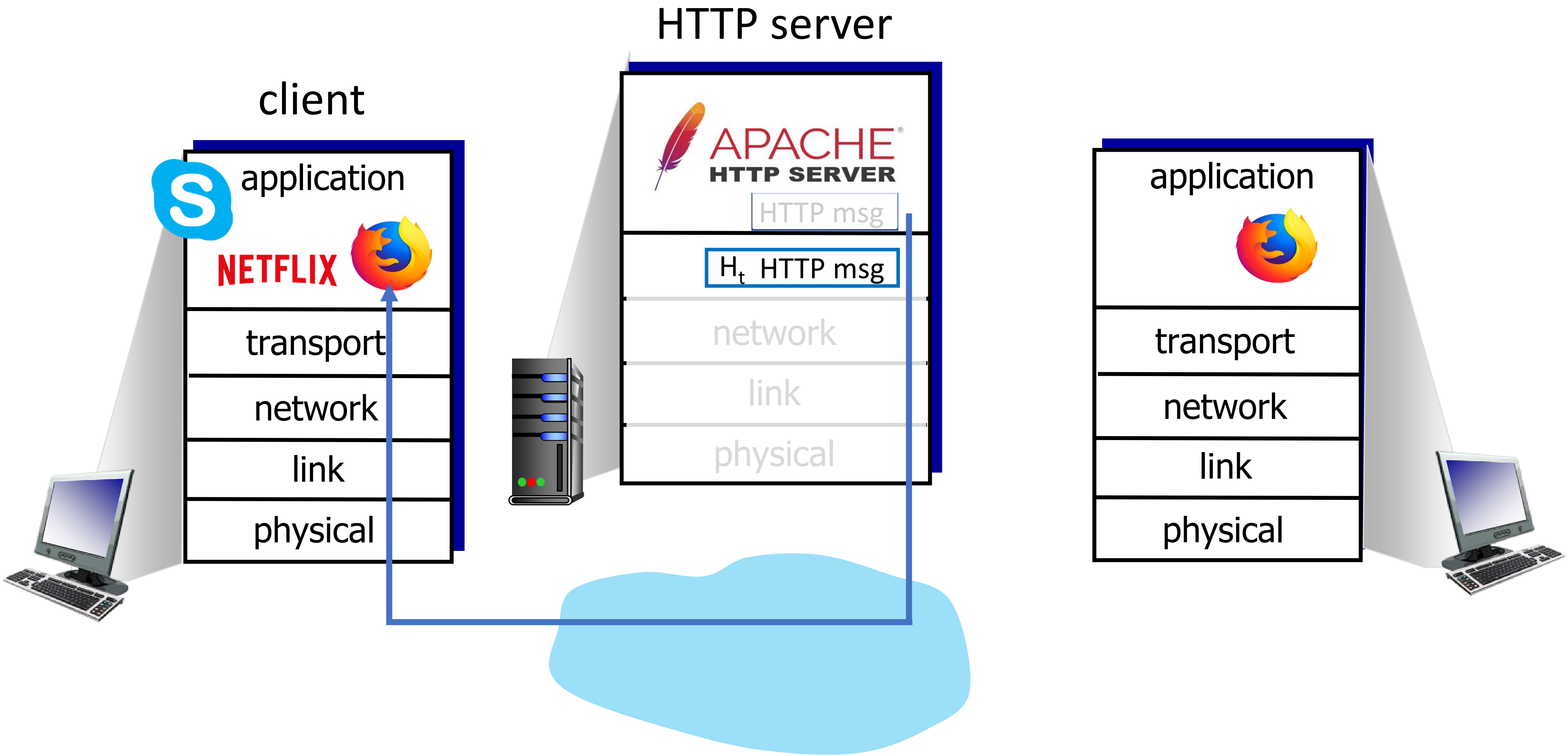
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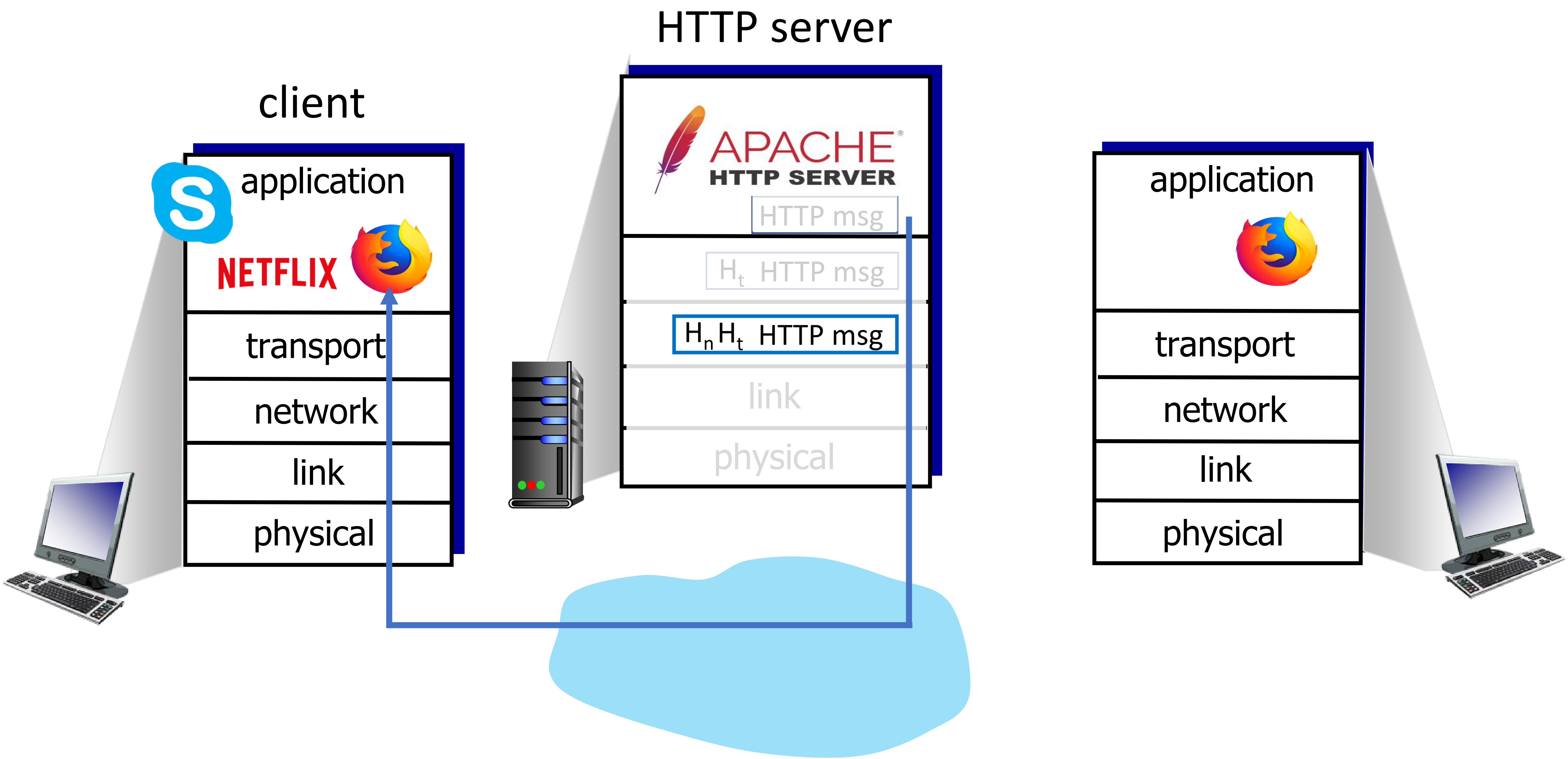
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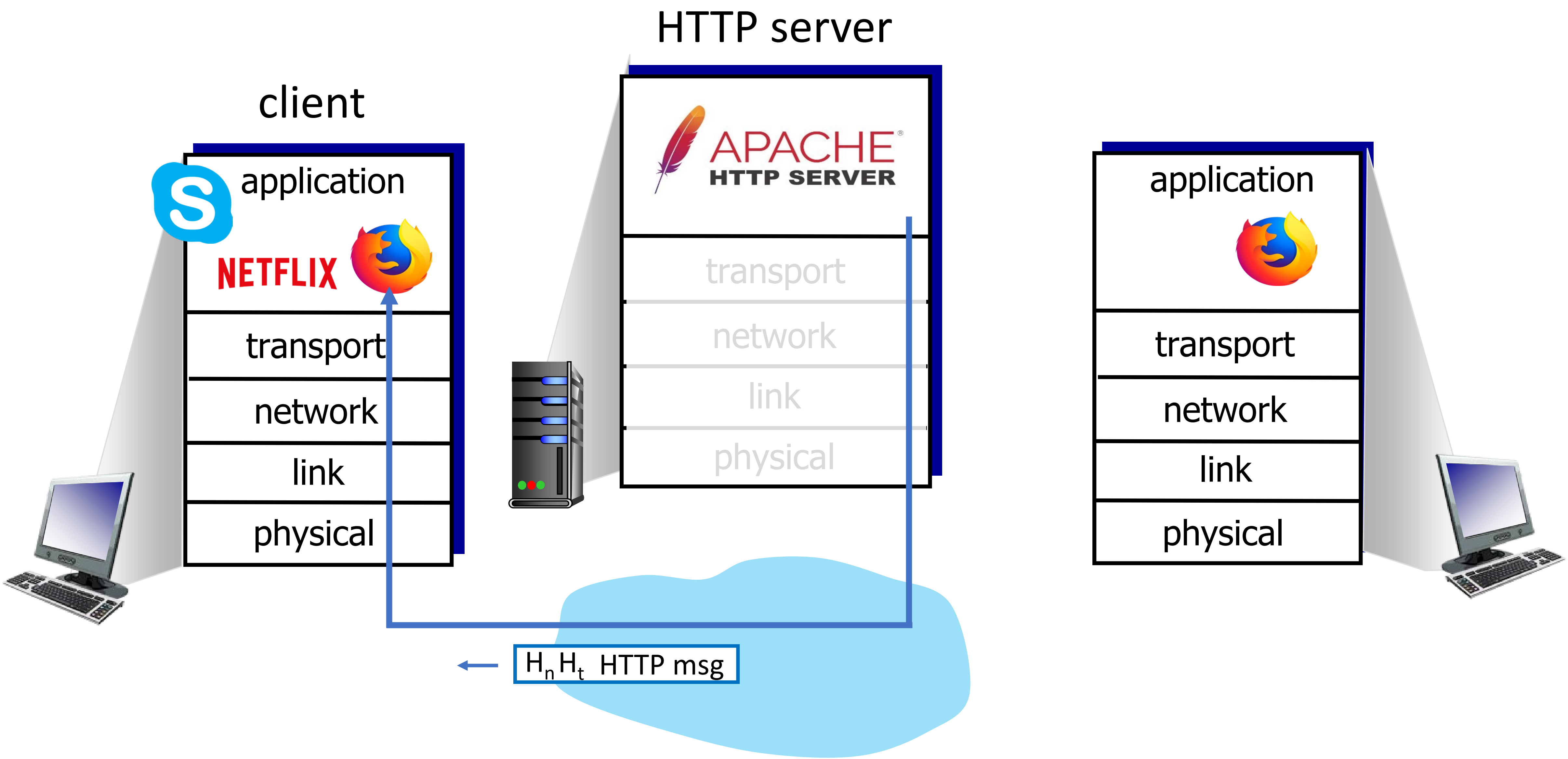
HTTP server
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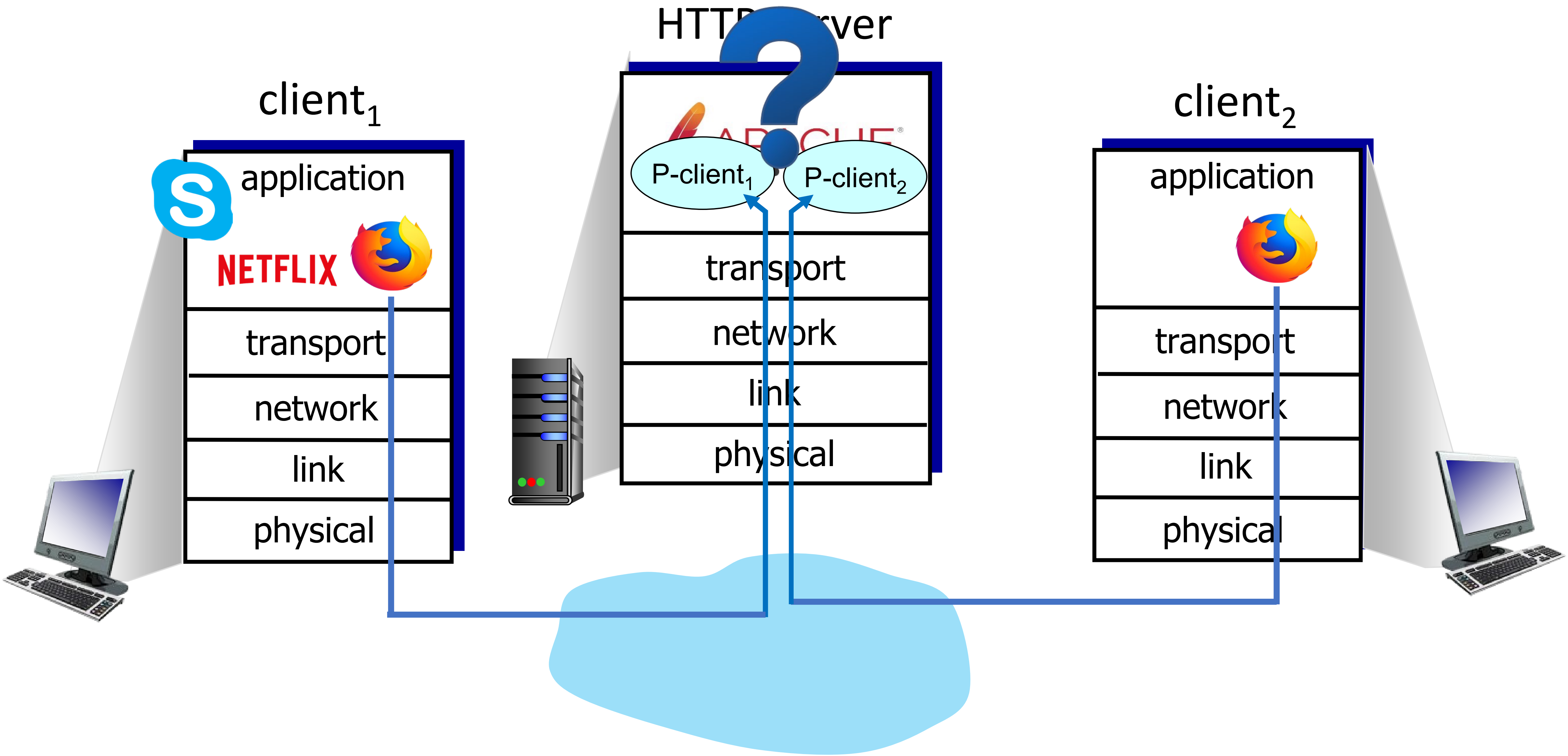
$H_n H_t$ HTTP msg


HTTP server
client1
client2
application
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network
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P-client1
P-client2
NETFLIX

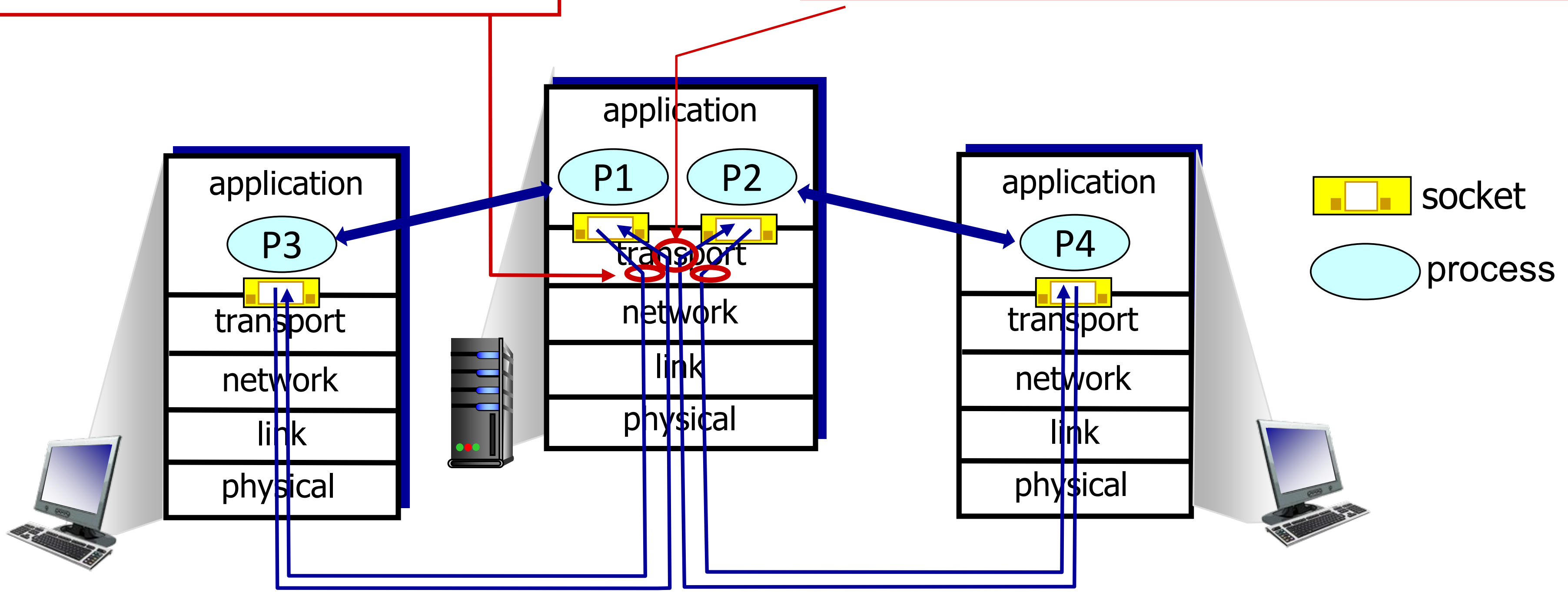
Multiplexing/demultiplexing

multiplexing at sender:

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

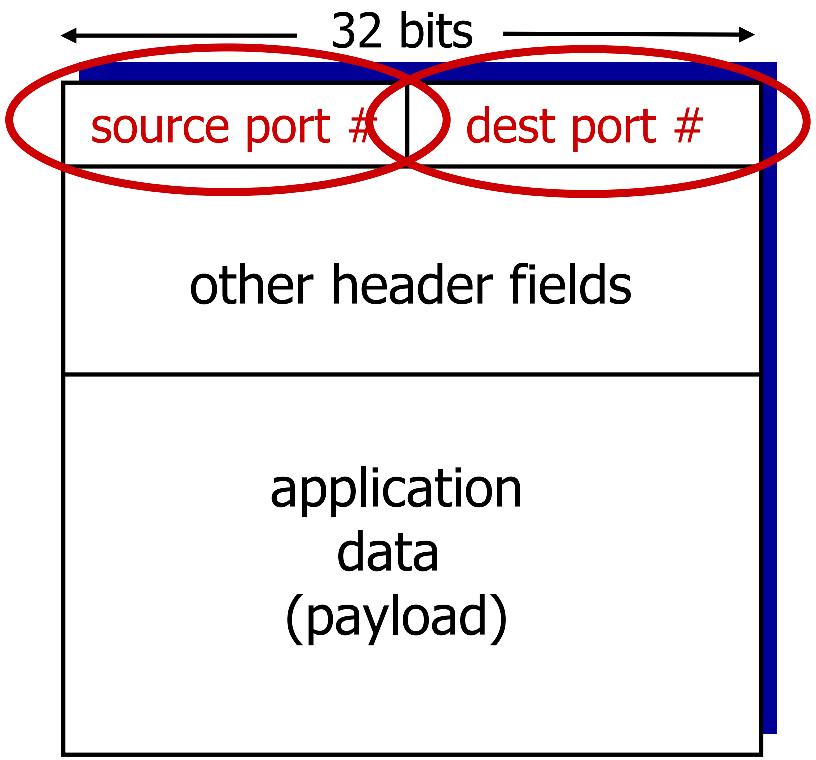
demultiplexing at receiver:

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

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

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

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

对，这里定义套接字的时候只
看了目标IP地址和端口，没有
审核来源的IP地址和端口

when receiving host receives *UDP* segment:

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

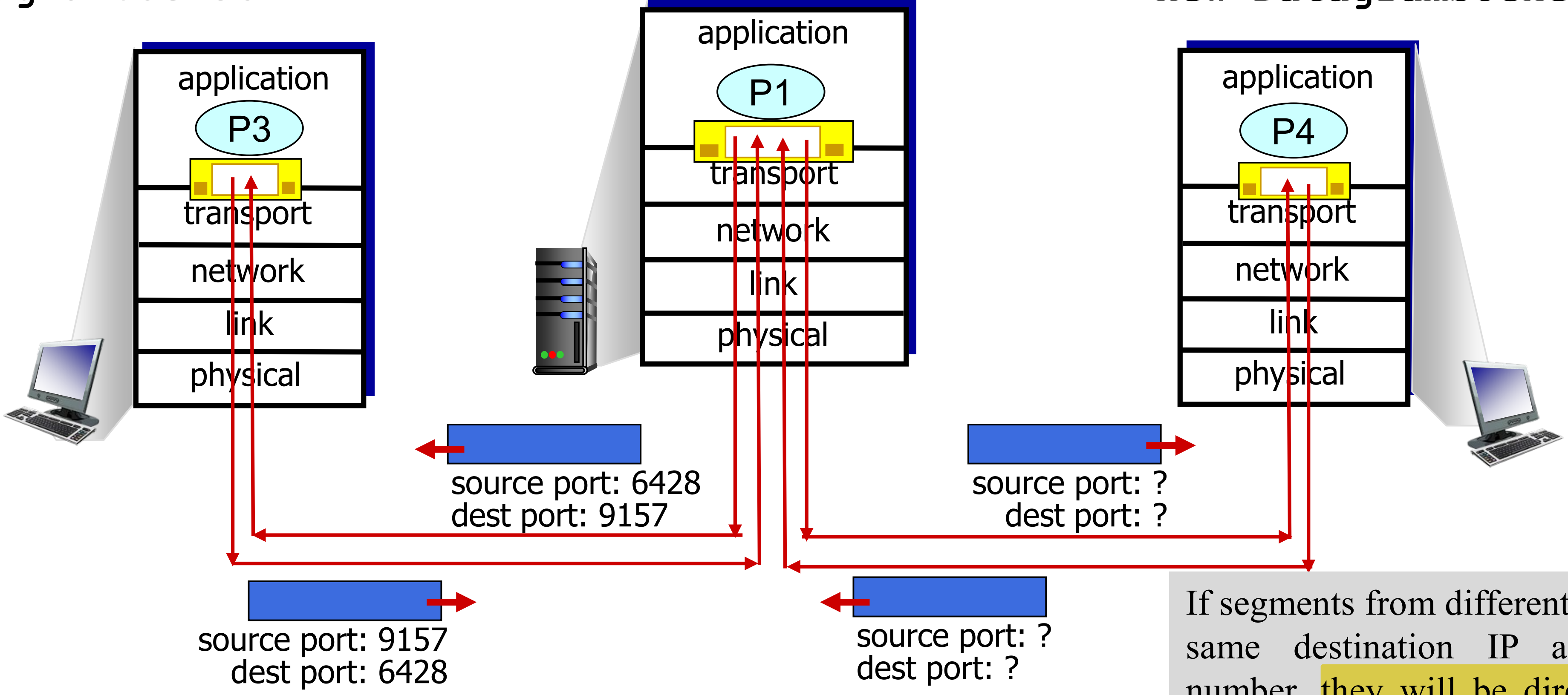
IP/UDP datagrams with *same dest. port #,* but different source IP addresses and/or source port numbers will be directed to *same socket* at receiving host

Connectionless demultiplexing: an example

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

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

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



If segments from different IP addresses have same destination IP address and port number, they will be directed to the same process

这不就出错了吗？所以要怎么解决呢

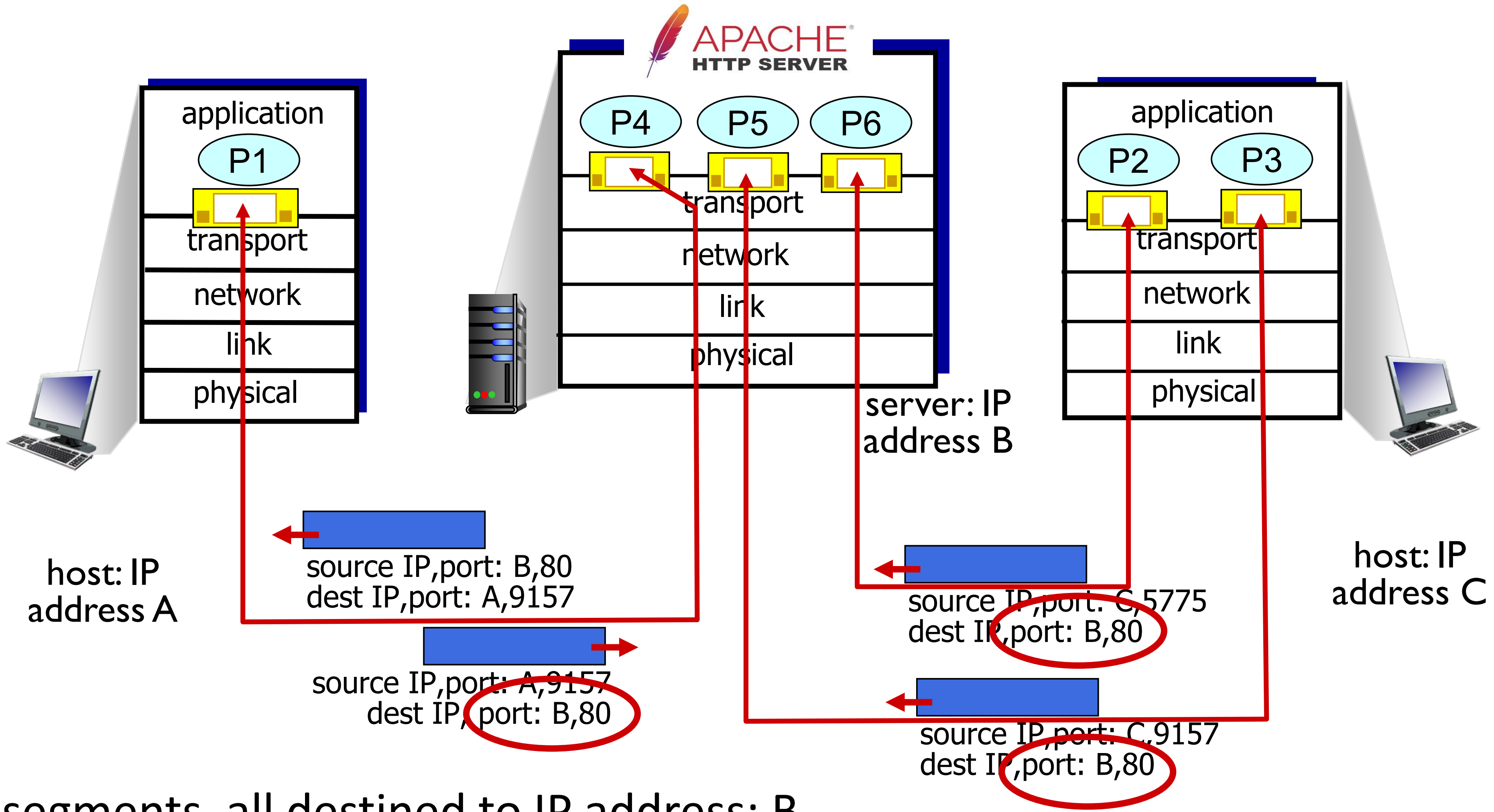
Connection-oriented demultiplexing

- 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

这就防止了上一页的出错可能性

- 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

还有IP address啦

- **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

Transport Layer: Roadmap

- Transport-layer services
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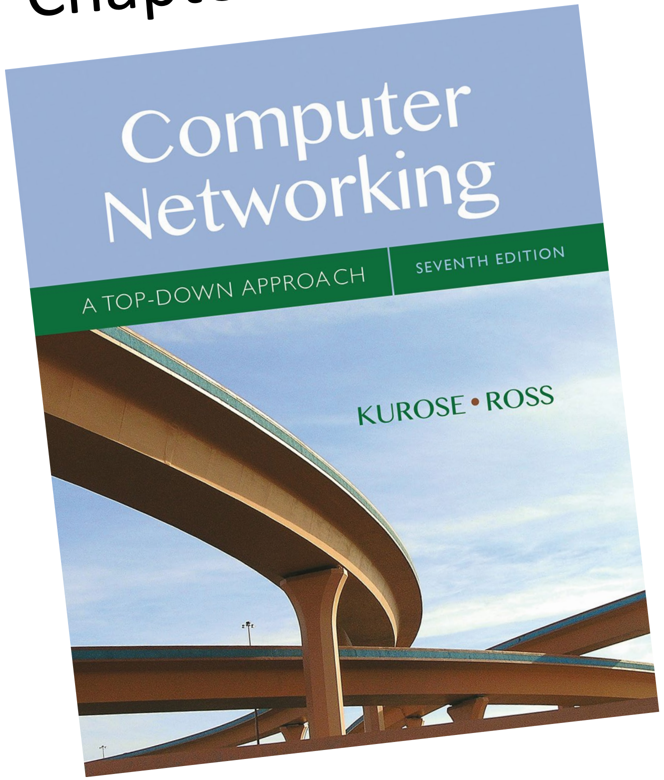


UDP: User Datagram Protocol

- “no frills,” “bare bones” Internet transport protocol
- “best effort” service, UDP segments may be:
  - lost
  - 不按顺序
    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 delay (no extra RTT)
- simple: no connection state at sender, receiver
- small header size
- no congestion control 这个有点意思，只要够快，容错率就高
  - UDP can blast away as fast as desired

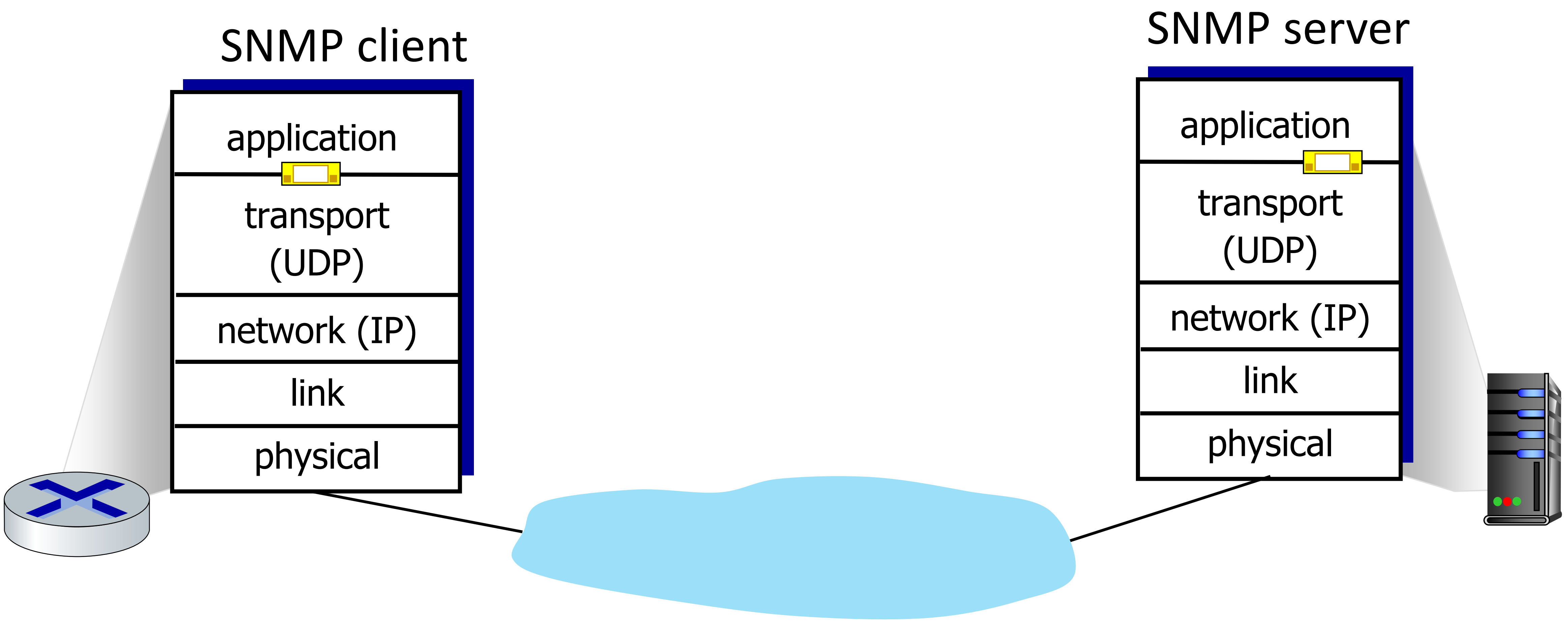
UDP: User Datagram Protocol

- UDP use:
  - streaming multimedia apps (loss tolerant, rate sensitive)
  - DNS
  - SNMP

Simple Network Management Protocol
应用层协议

- if reliable transfer needed over UDP:
  - add needed reliability at application layer
  - add congestion control at application layer

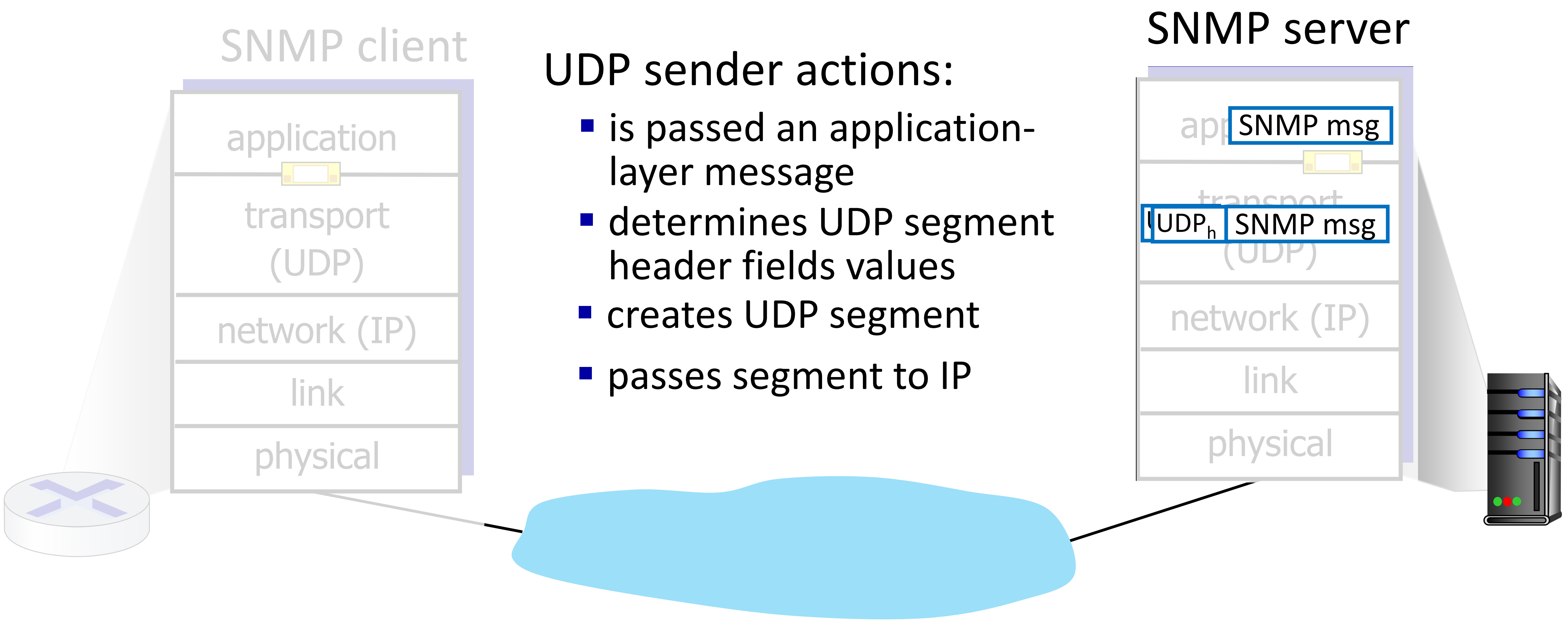
UDP: Transport Layer Actions



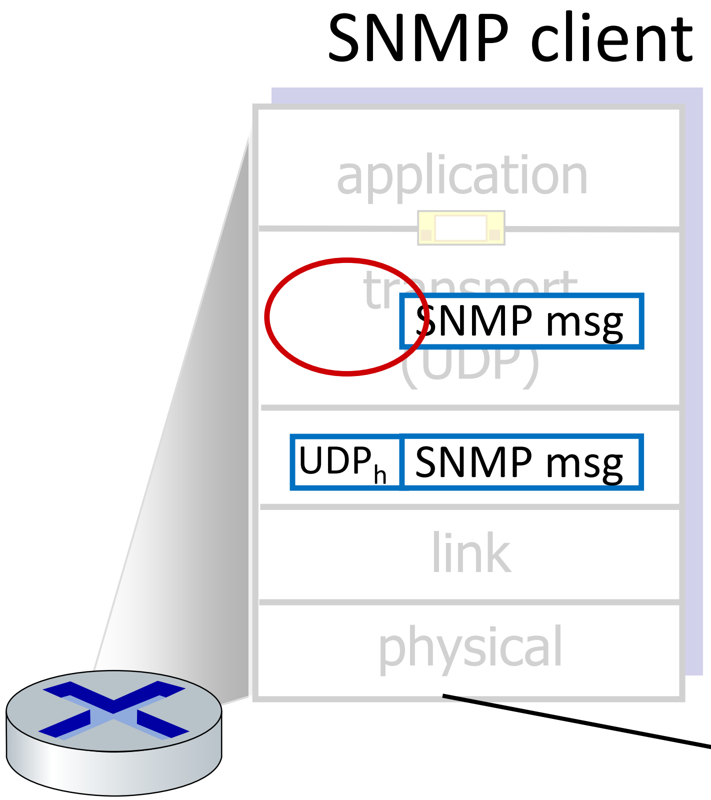
UDP: Transport Layer Actions

UDP sender actions:

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

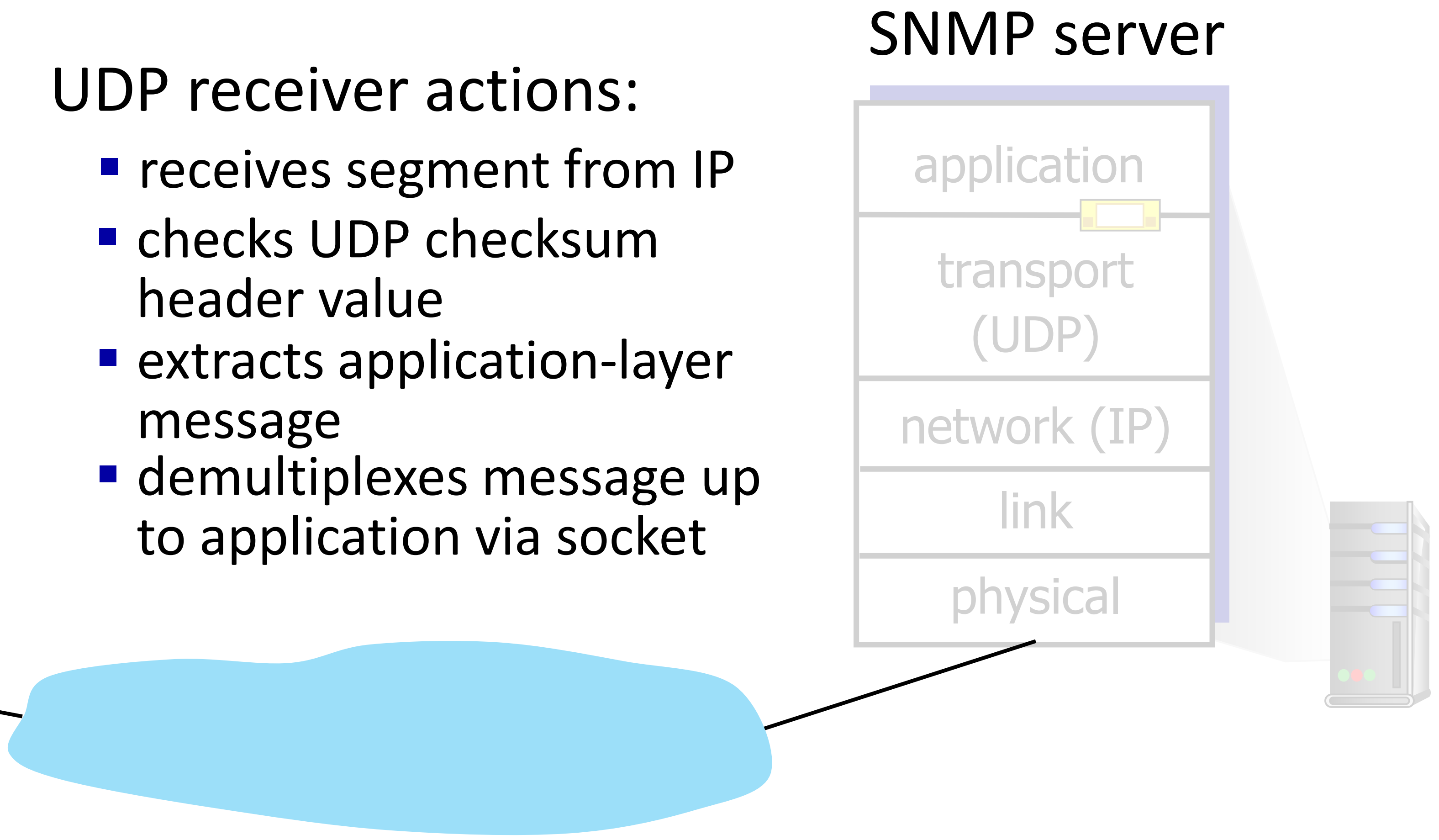


UDP: Transport Layer Actions

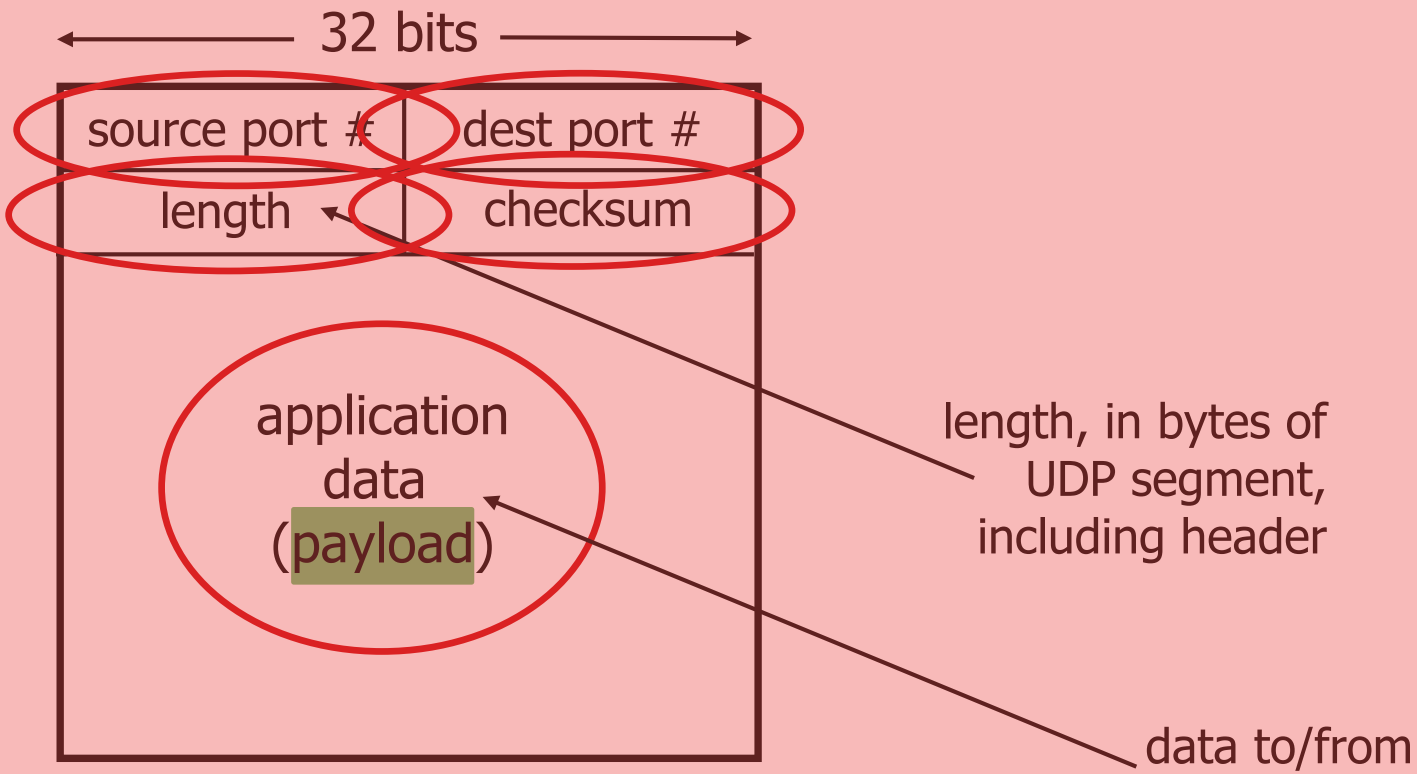


UDP receiver actions:

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



UDP segment header



UDP segment format

UDP checksum

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

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

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

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 (**1s 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?*

Internet checksum: an example

example: add two 16-bit integers

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

这个意思是说把进的一位加回到个位去从而获得sum

至于checksum的值就是sum取反

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

Transport Layer: Roadmap

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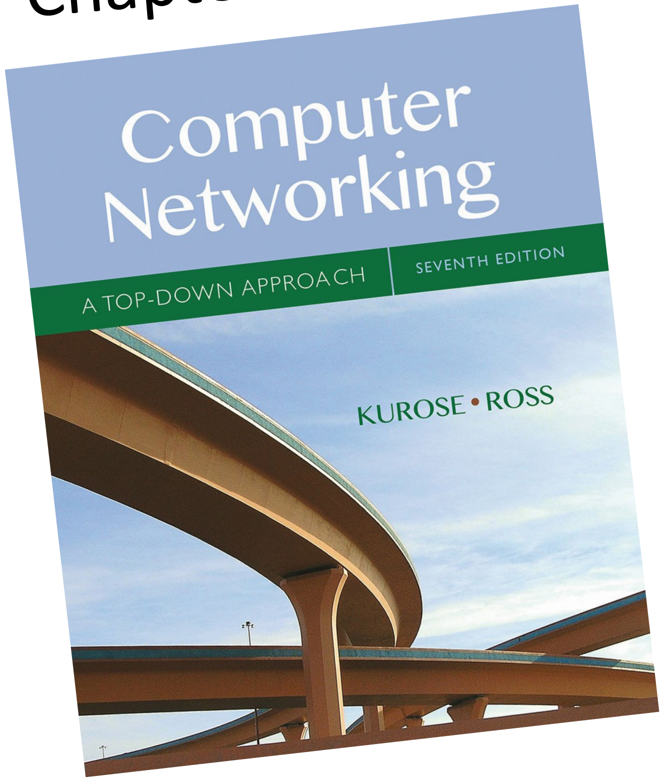


An Analogy: Talking on a Cell Phone

- Alice and Bob on their cell phones
  - Both Alice and Bob are talking
- What if Bob couldn't understand Alice?
  - Bob asks Alice to repeat what she said
- What if Bob hasn't heard Alice for a while?
  - Is Alice just being quiet?
  - Or, have Bob and Alice lost reception?
  - How long should Bob just keep on talking?
  - Maybe Alice should periodically say "I'm still here"
  - … or Bob should ask "Can you hear me now?"

这里的意思应该时"失去接受能力，也就是失去联通能力"

Some Take-Aways from the Example

- Acknowledgments from receiver
  - Positive: “okay” or “ACK”
  - Negative: “please repeat that” or “NACK”

- Timeout by the sender (“stop and wait”)
  - Don’t wait indefinitely without receiving some response

- Retransmission by the sender
  - After receiving a “NACK” from the receiver
  - After receiving no feedback from the receiver

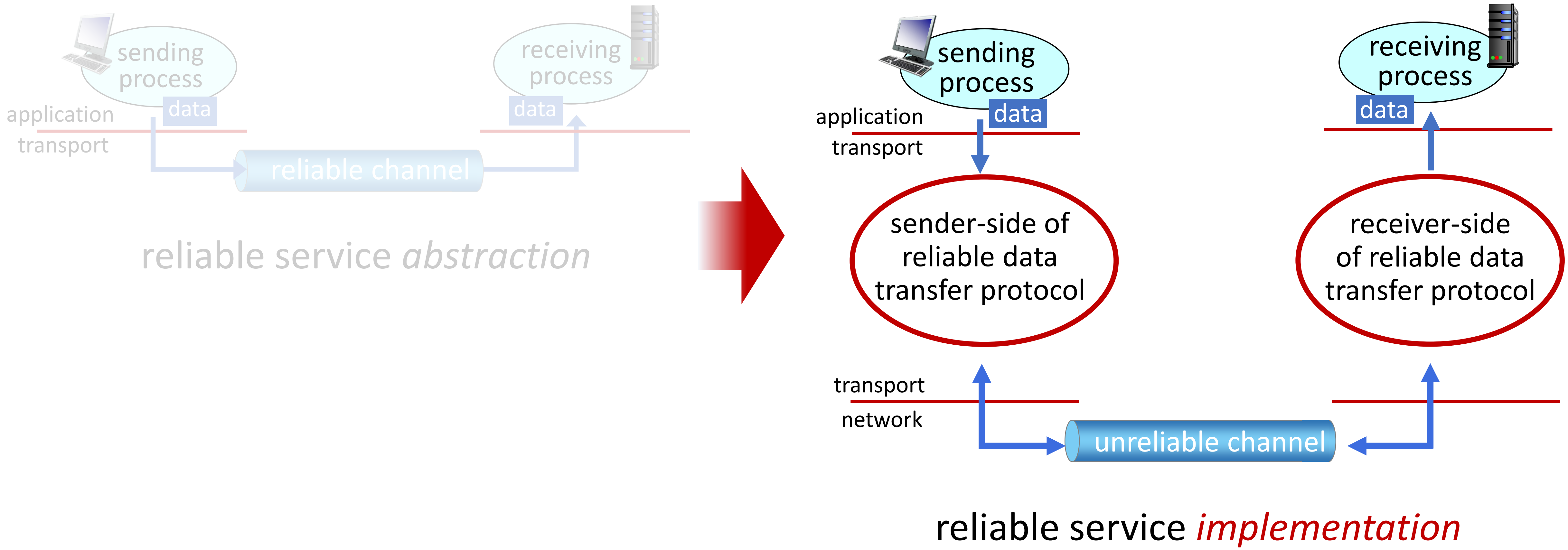
How do we design a protocol for reliable data transfer?

Principles of reliable data transfer

reliable service *abstraction*

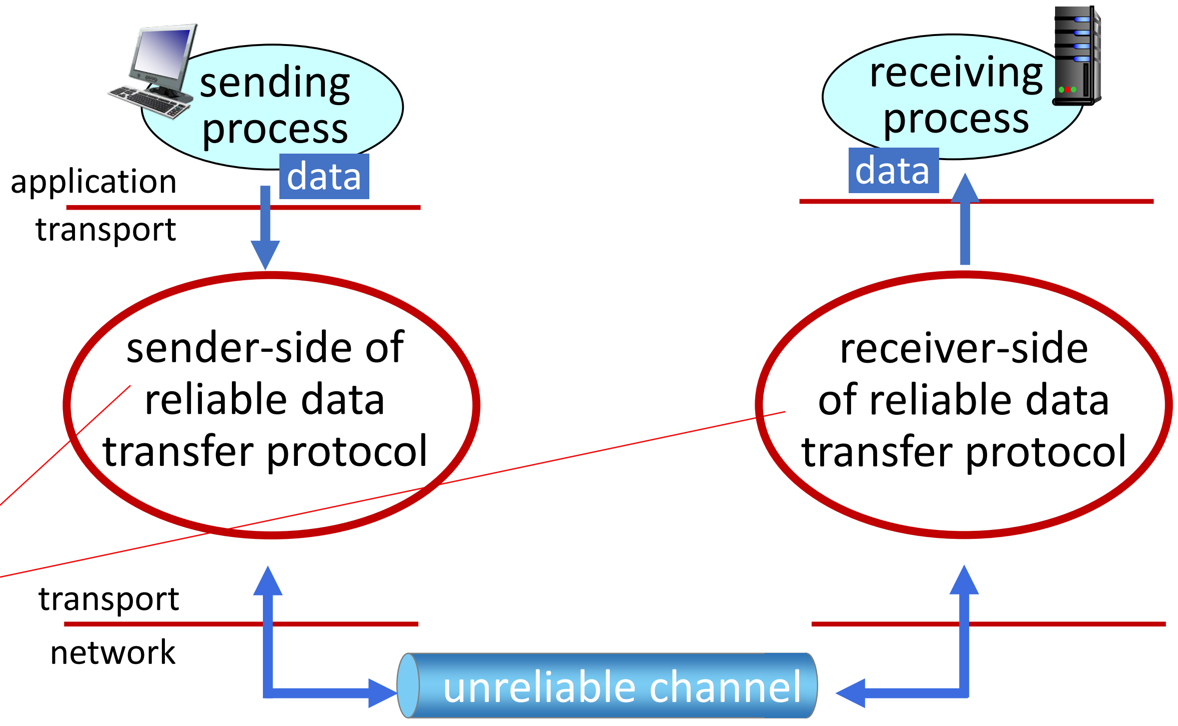
这里应该理解成“概念”，是相对于application的说法

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?)

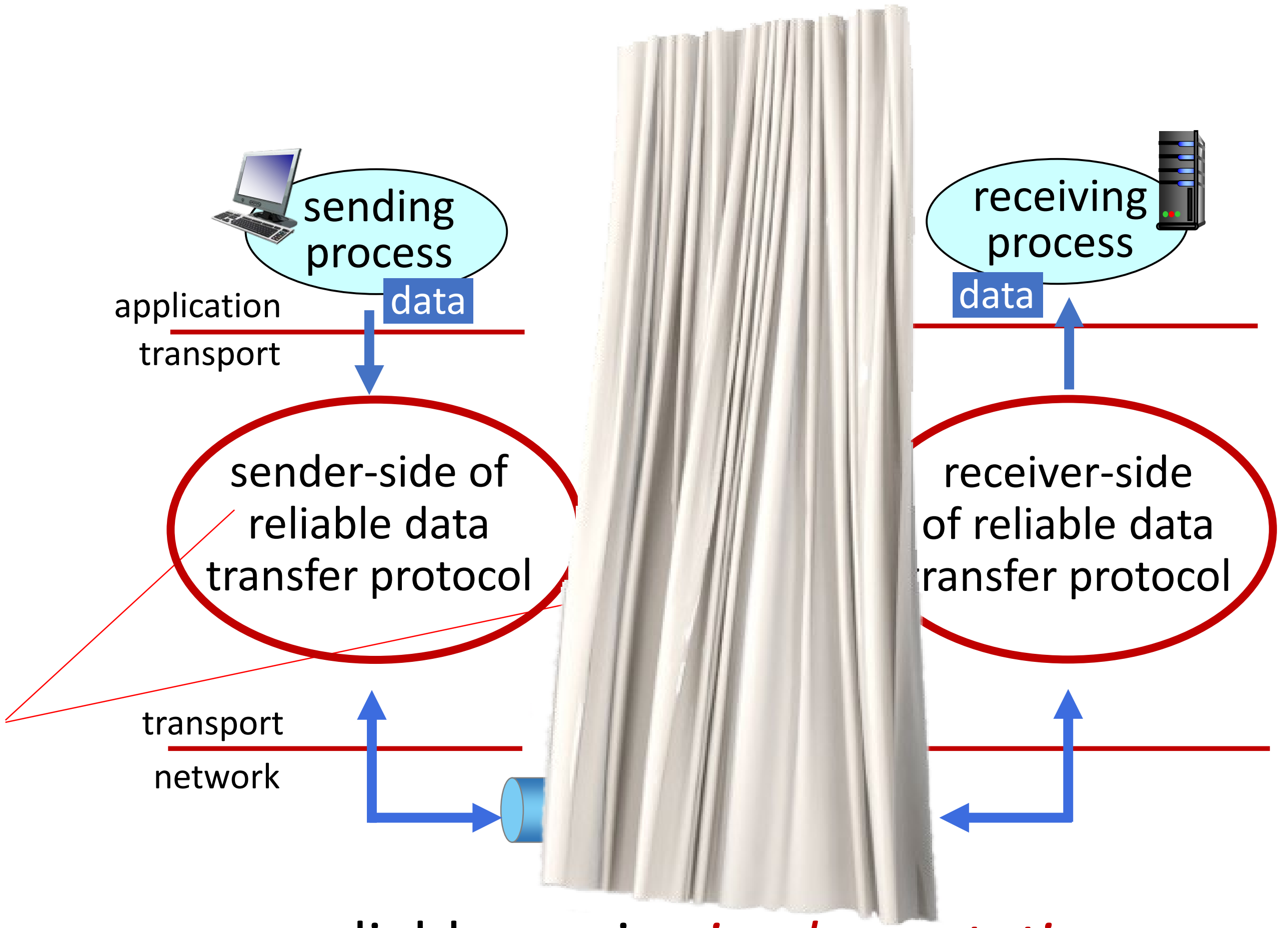


reliable service *implementation*

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



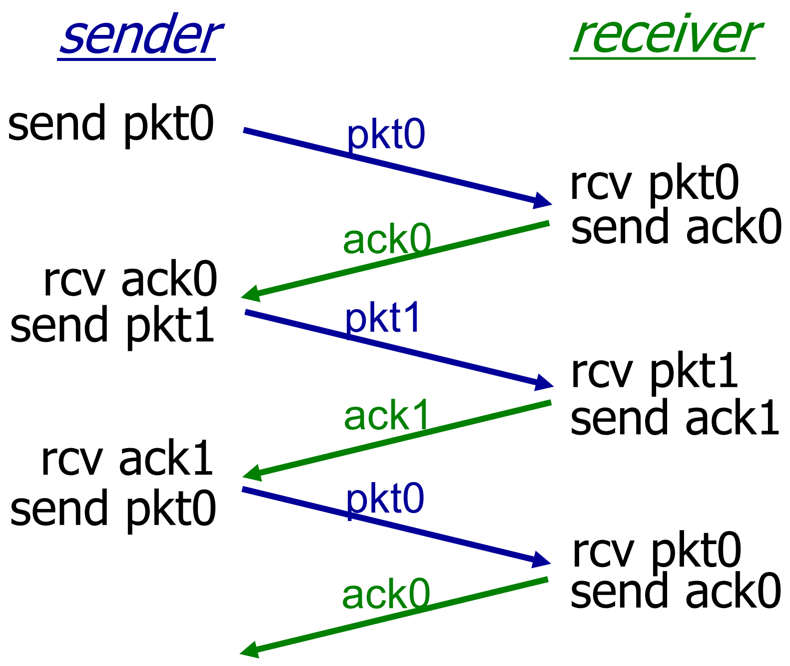
reliable service *implementation*

Challenges of Reliable Data Transfer

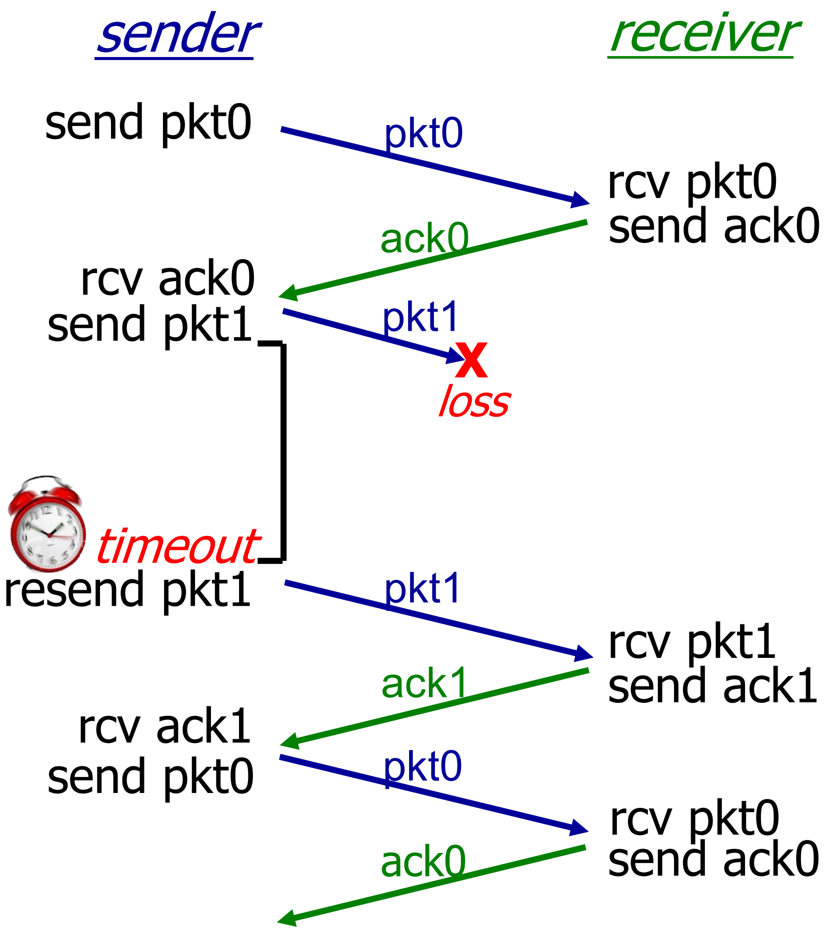
- Over a perfectly reliable channel
  - Simple: sender sends data, and receiver receives data
- Over a channel with bit errors
  - All of the data arrives in order, but some bits corrupted
  - Receiver detects errors and says “please repeat that”
  - Sender retransmits the data that were corrupted
- Over a lossy channel with bit errors
  - Some data are missing, and some bits are corrupted
  - Receiver detects errors but cannot always detect loss
  - Sender must wait for acknowledgment (“ACK” or “OK”)
  - … and retransmit data after some time if no ACK arrives

丢失和错误是两回事。可以检测错误，但不好检测丢失

The Stop-and-Wait Protocol

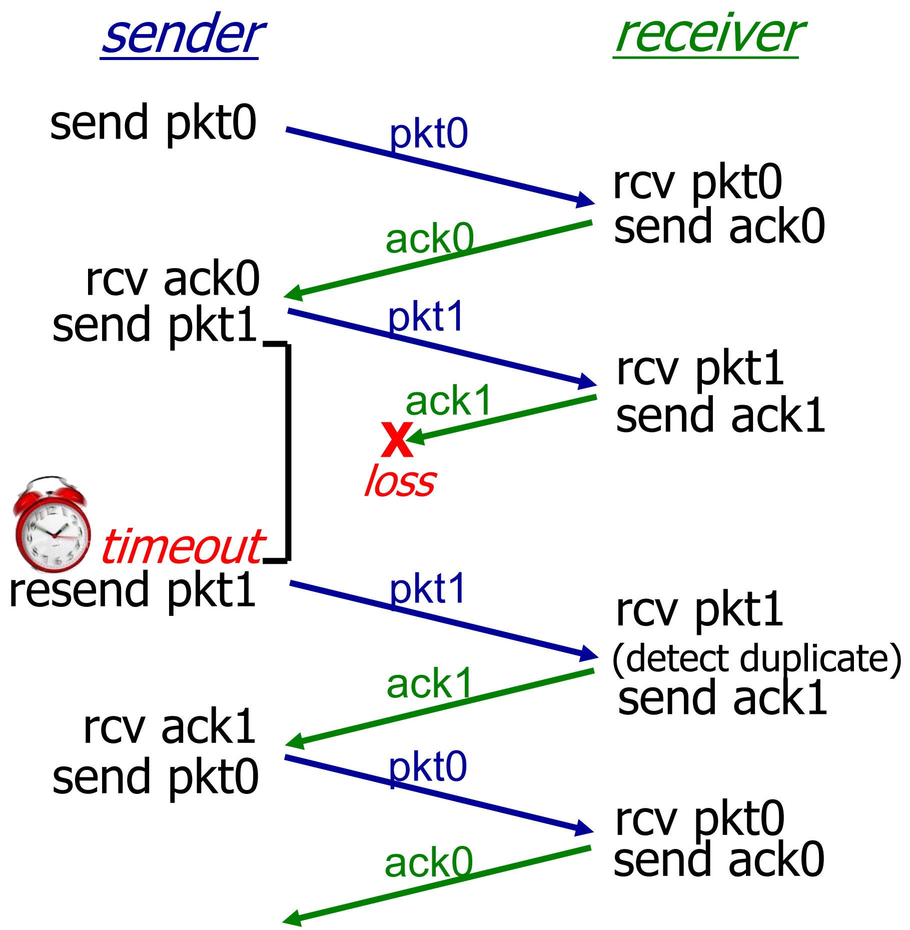


(a) no loss

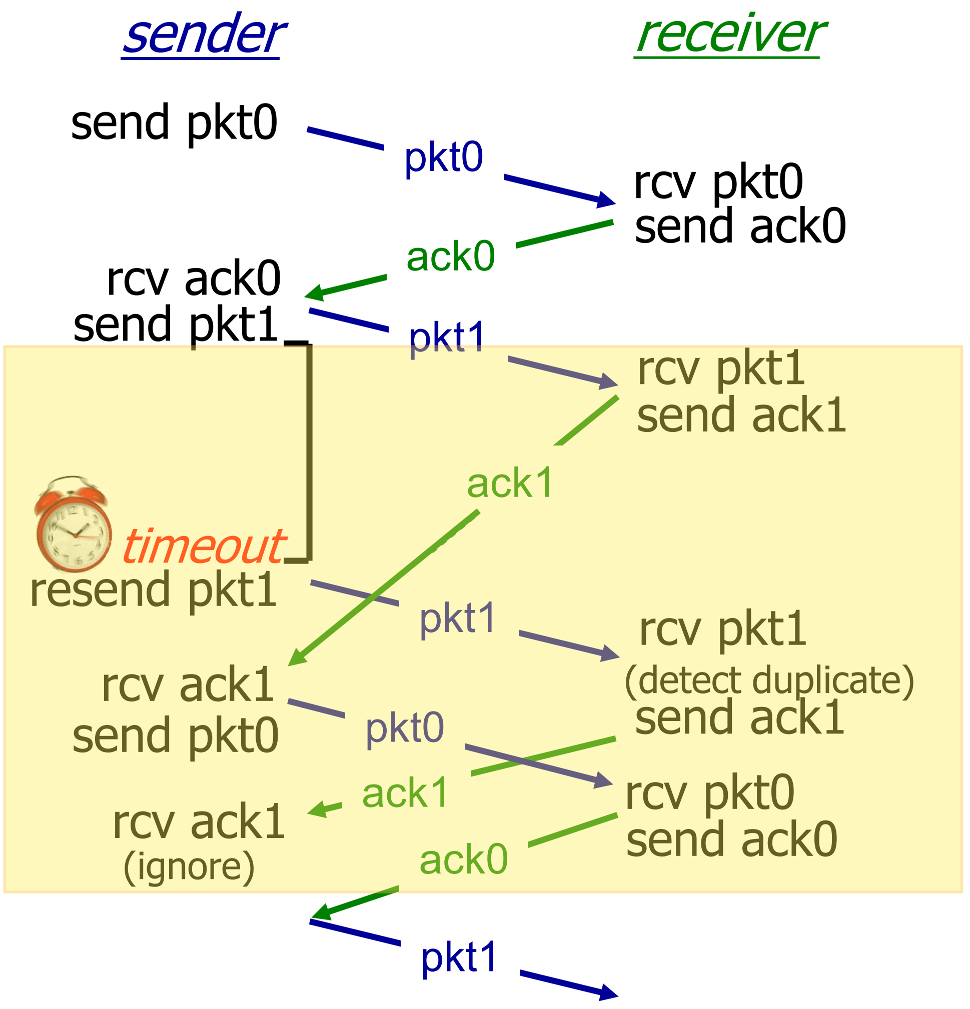


(b) packet loss

The Stop-and-Wait Protocol



(c) ACK loss



(d) premature timeout/ delayed ACK

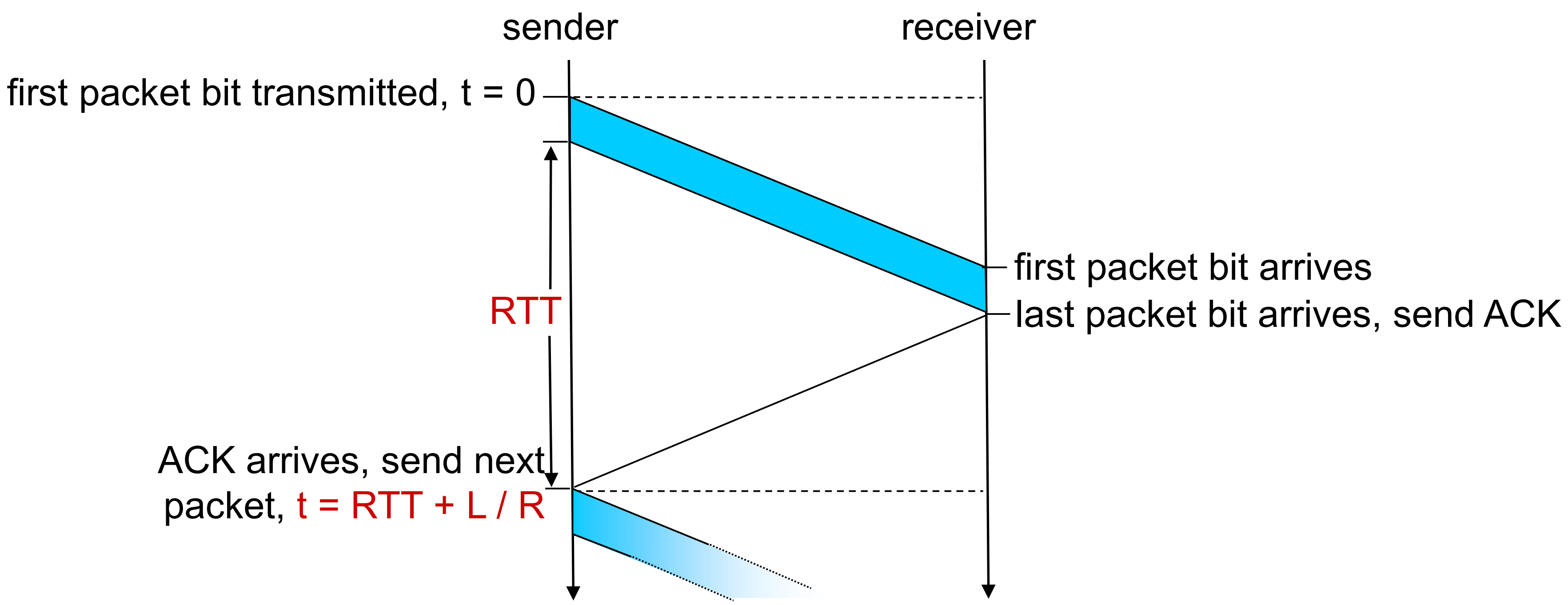
Performance of stop-and-wait

- 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:

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

stop-and-wait operation

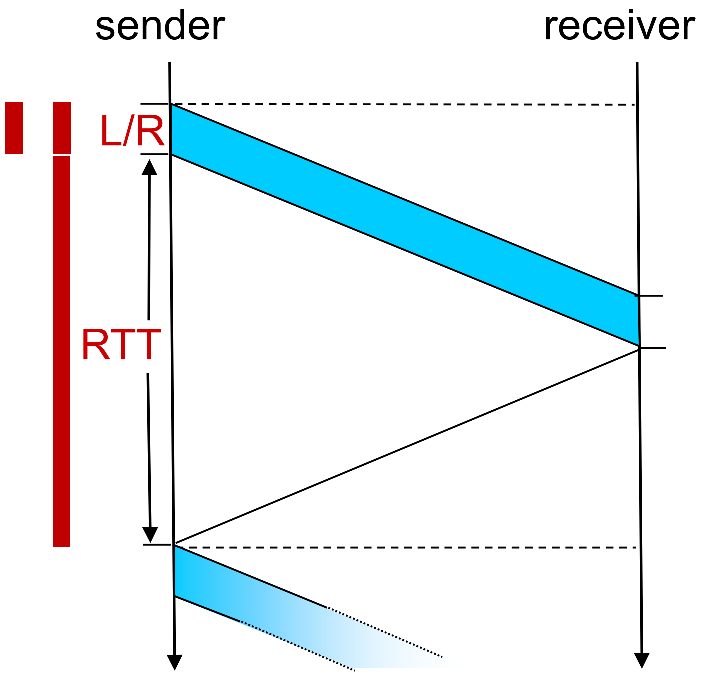


stop-and-wait operation

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

$$= \frac{.008}{30.008}$$

$$= 0.00027$$



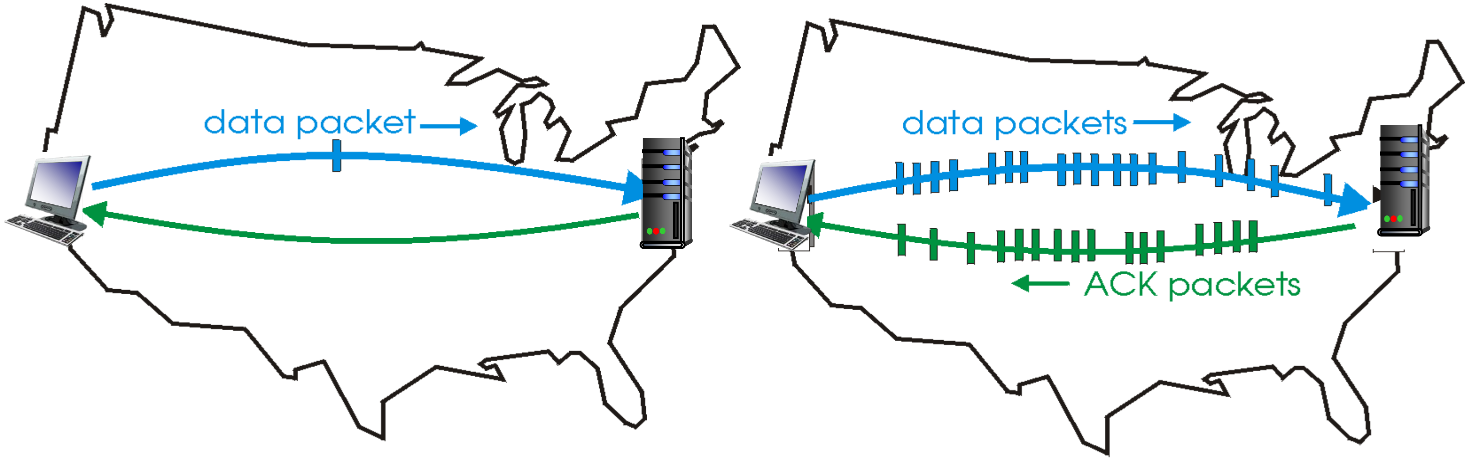
- Protocol doesn’t make good use of resources!
- Limits performance of underlying infrastructure (channel)

底层设施，就理解成基础设施就好了，但是不晓得为啥要用这么拗口的词汇

Pipelined protocols

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

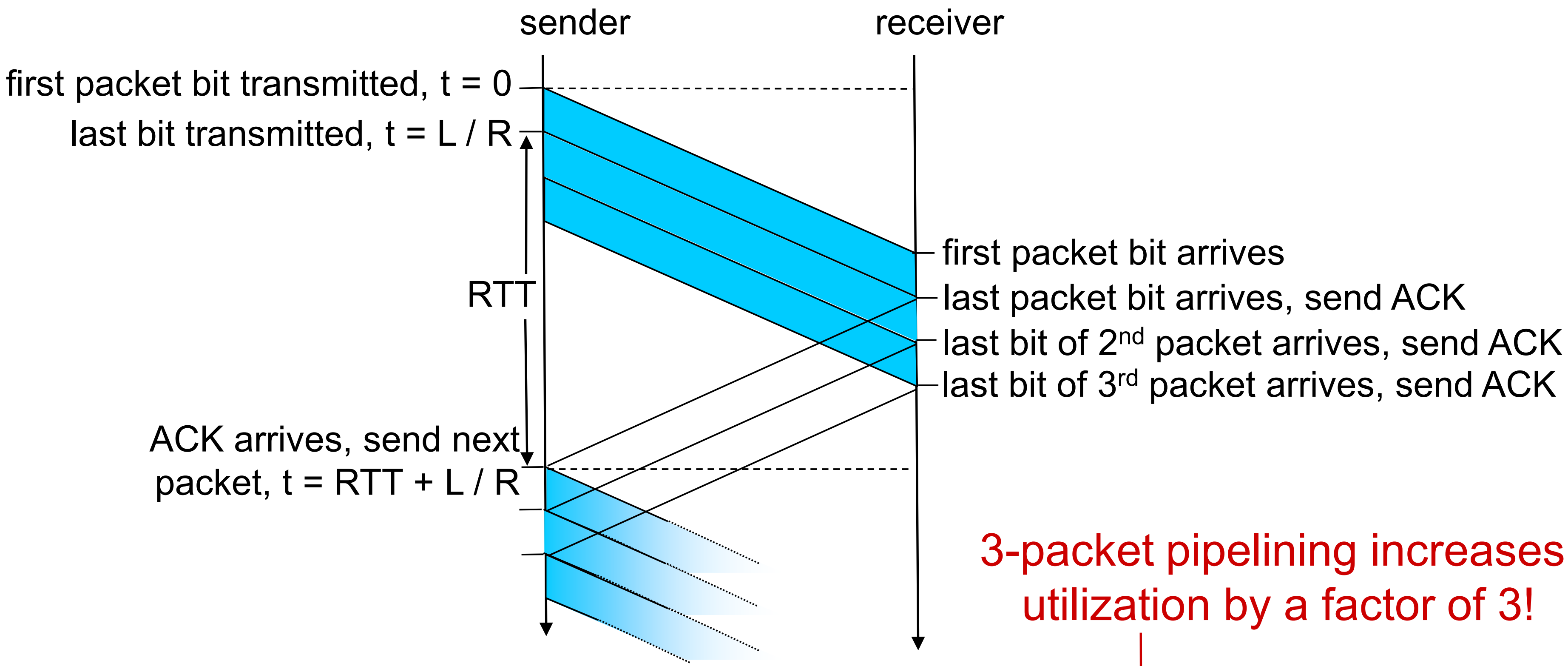
- range of sequence numbers must be increased
- buffering at sender and/or receiver
- two generic pipelined protocols:
  - *Go-Back-N*
  - *Selective Repeat*



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

(b) a pipelined protocol in operation

Pipelining: increased utilization



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

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

书本上也没有明说有这个公式，不过我理解是他一个计算周期就是RTT+L/R，其中发了3个RTT，你把这个图再往下画一个周期就明白了，第一个ack收到后就开始进入L/R以及RTT也就是开始进入下一个周期了

Pipelined protocols: overview

Go-back-N:

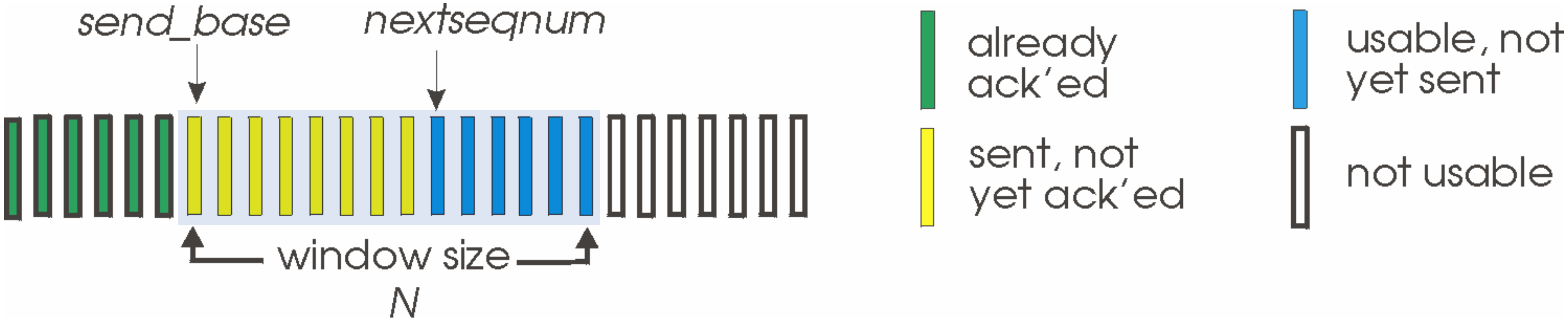
- sender can have up to N unacked packets in pipeline
- receiver only sends *cumulative ack*
  - doesn't ack packet if there's a gap
- sender has timer for oldest unacked packet
  - when timer expires, retransmit *all* unacked packets

Selective Repeat:

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

Go-Back-N: sender

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

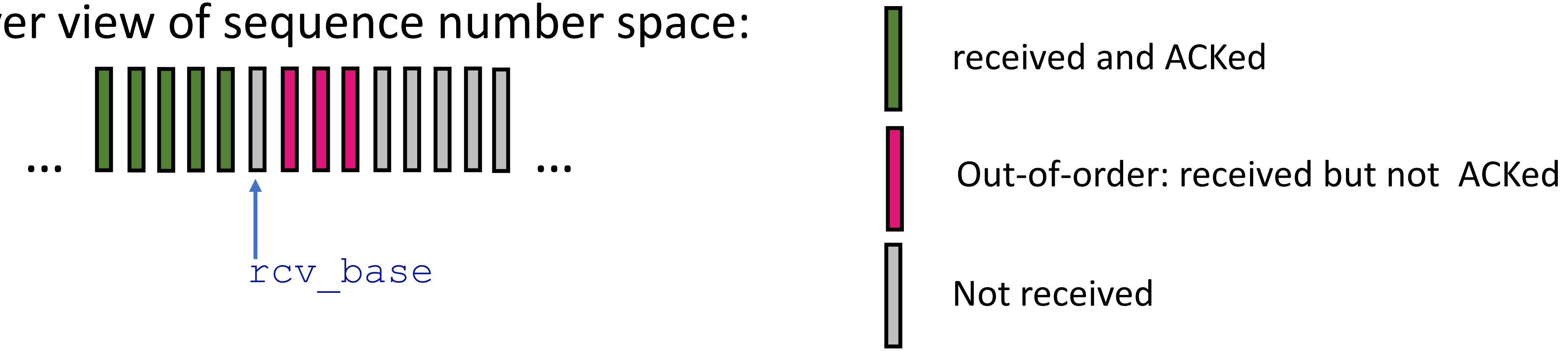


- *cumulative ACK:* ACK(*n*): ACKs all packets up to, including seq # *n*
  - on receiving ACK(*n*): move window forward to begin at *n+1*
- timer for oldest in-flight/unacked packet
- *timeout(n):* retransmit packet n and all higher seq # packets in window

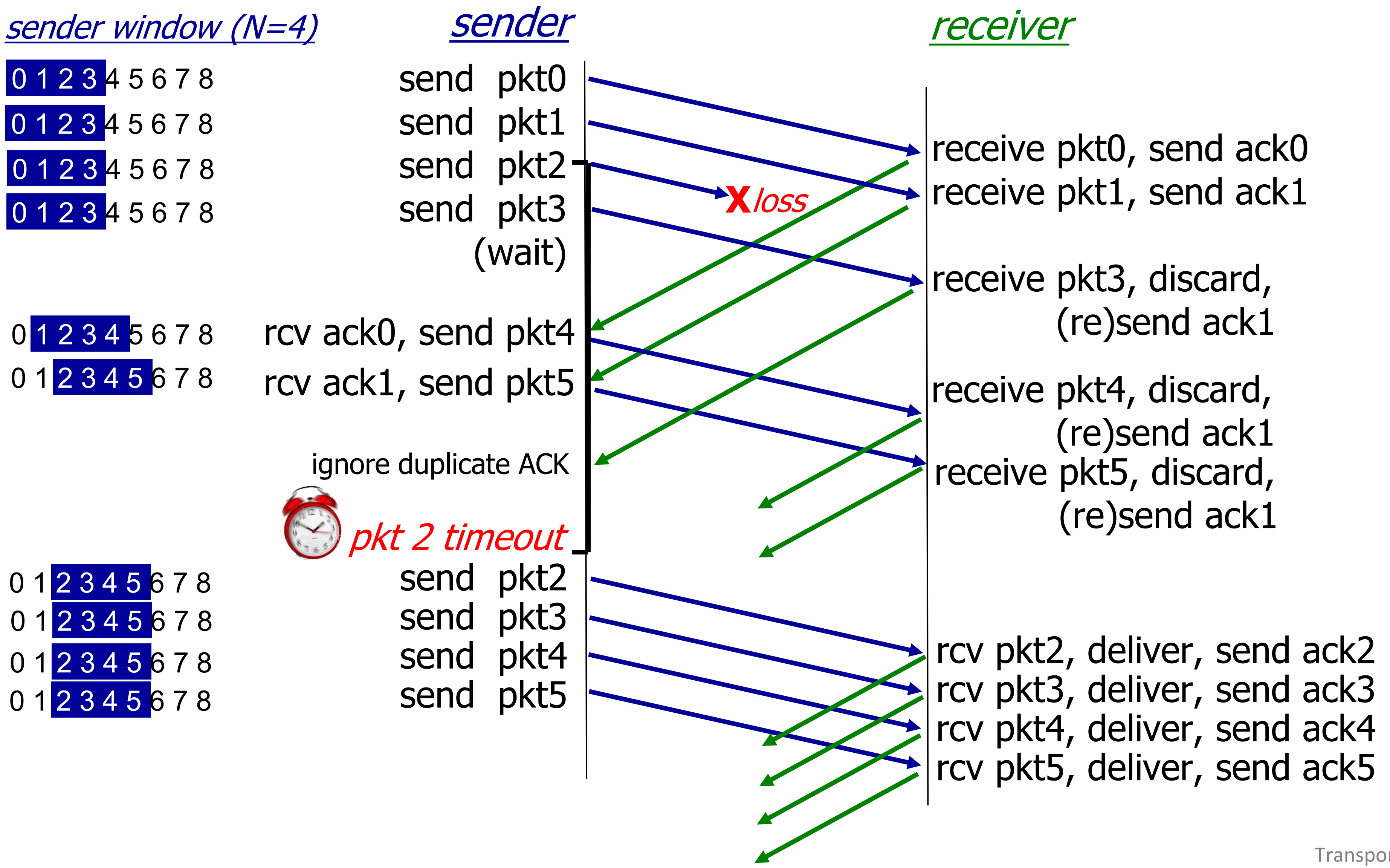
Go-Back-N: receiver side

- 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:
  - either discard (i.e. don't buffer) or buffer: depends on implementation!
  - re-ACK pkt with highest in-order seq #

Receiver view of sequence number space:



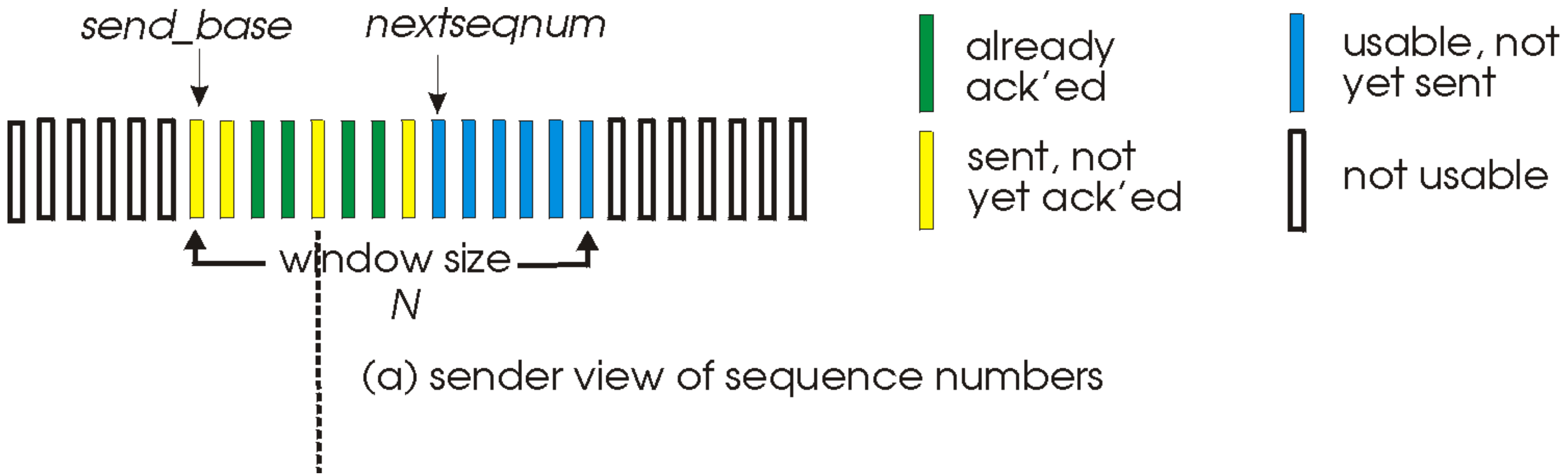
Go-Back-N in action



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 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]

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

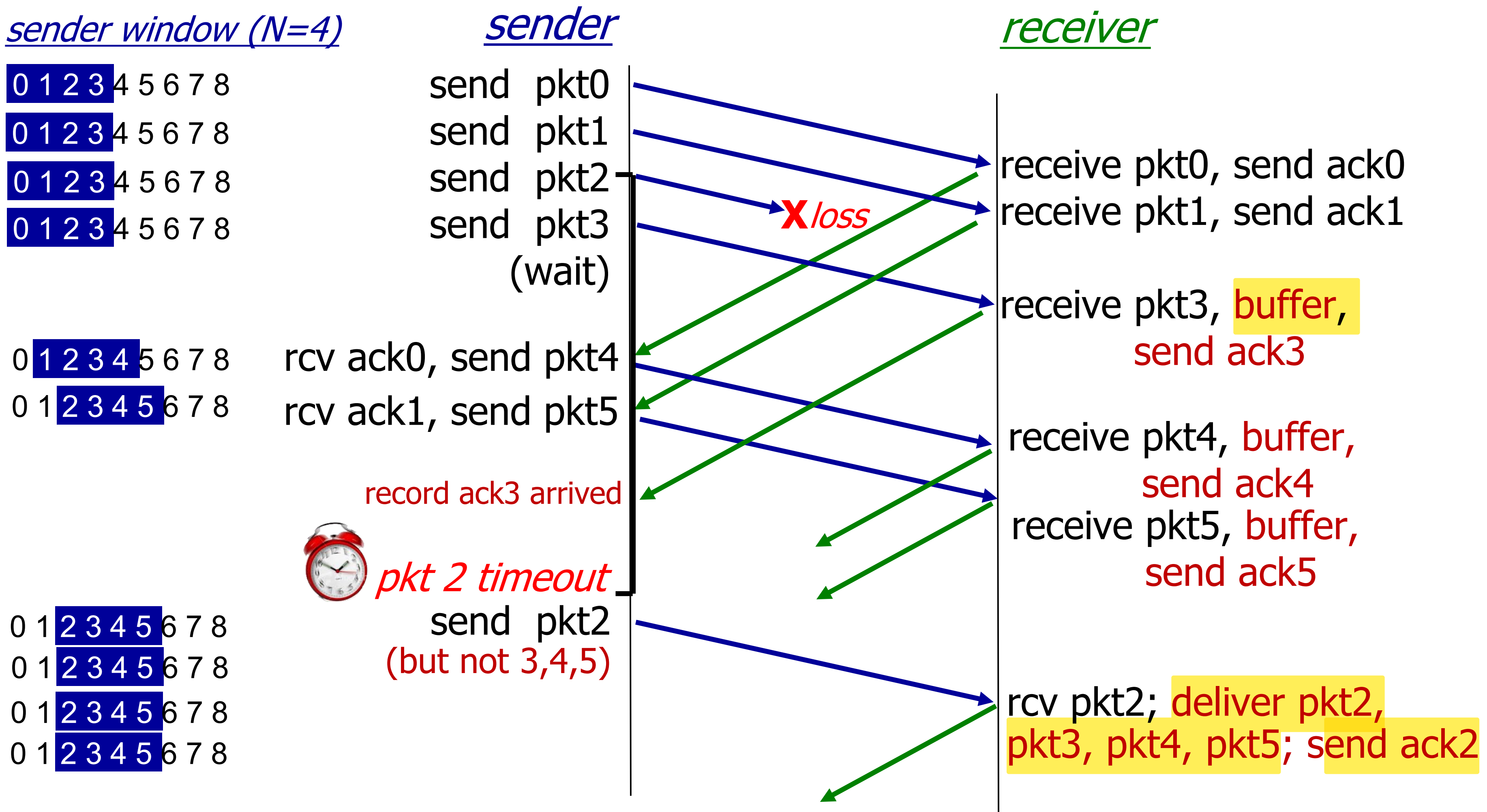
packet *n* in [rcvbase-N,rcvbase-1]

- ACK(*n*)

otherwise:

- ignore

Selective Repeat in action



Q: what happens when ack2 arrives?

Transport Layer: Roadmap

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  - reliable data transfer
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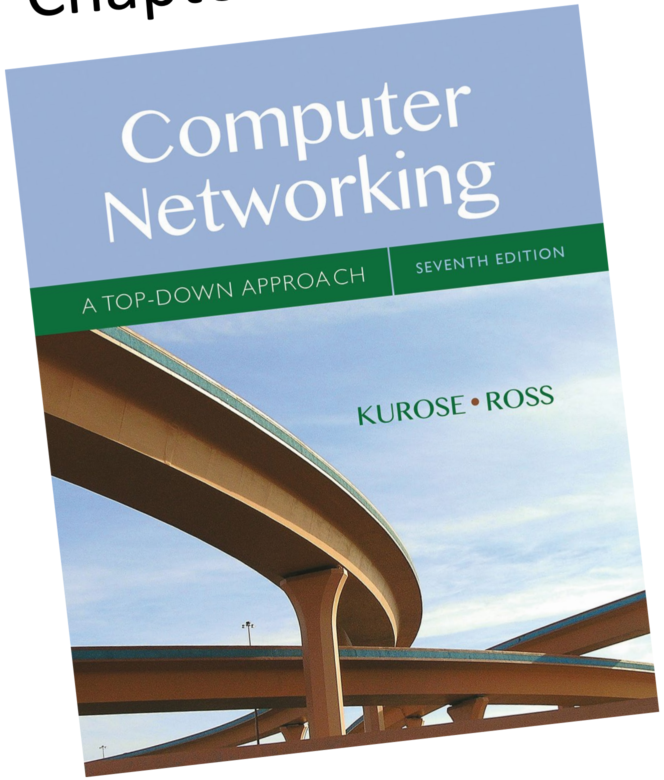


TCP: overview RFCs: 793,1122, 2018, 5681, 7323

别光罗列啊，有啥内容，这几个条例有啥区别啊？？

- 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

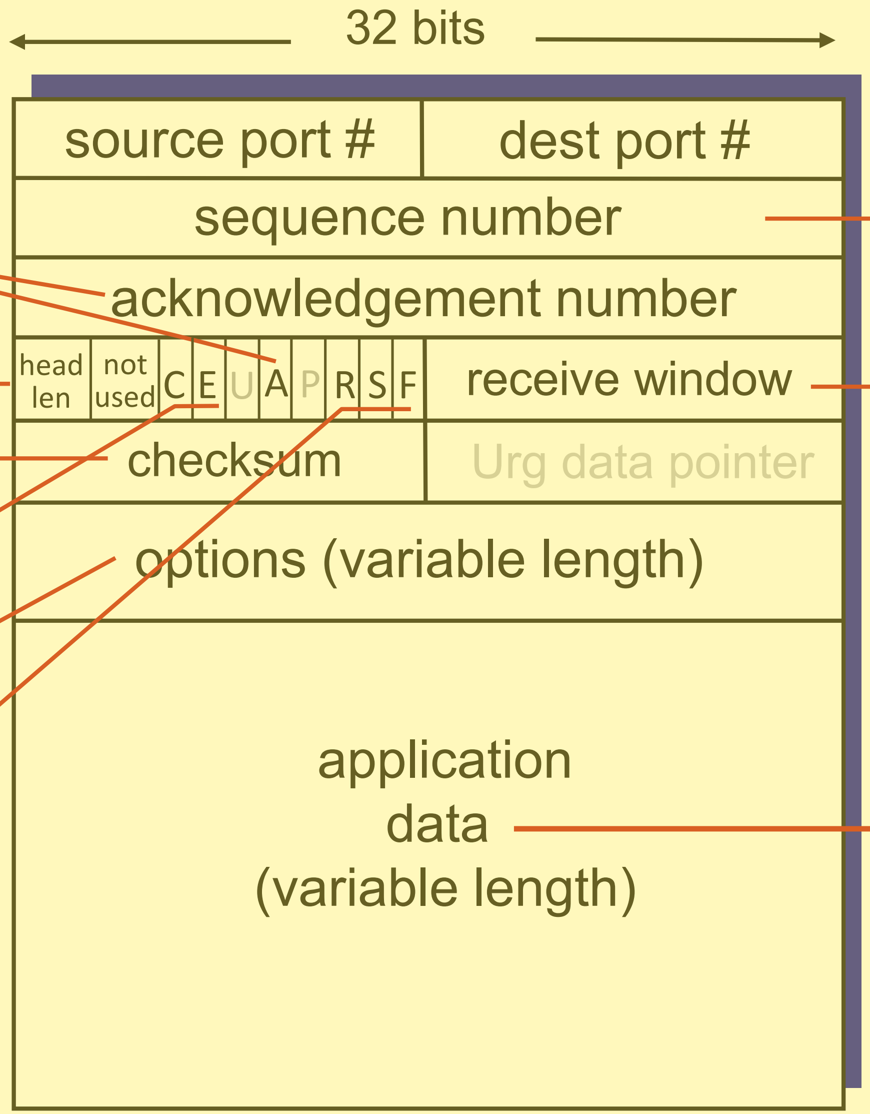
length (of TCP header)

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:

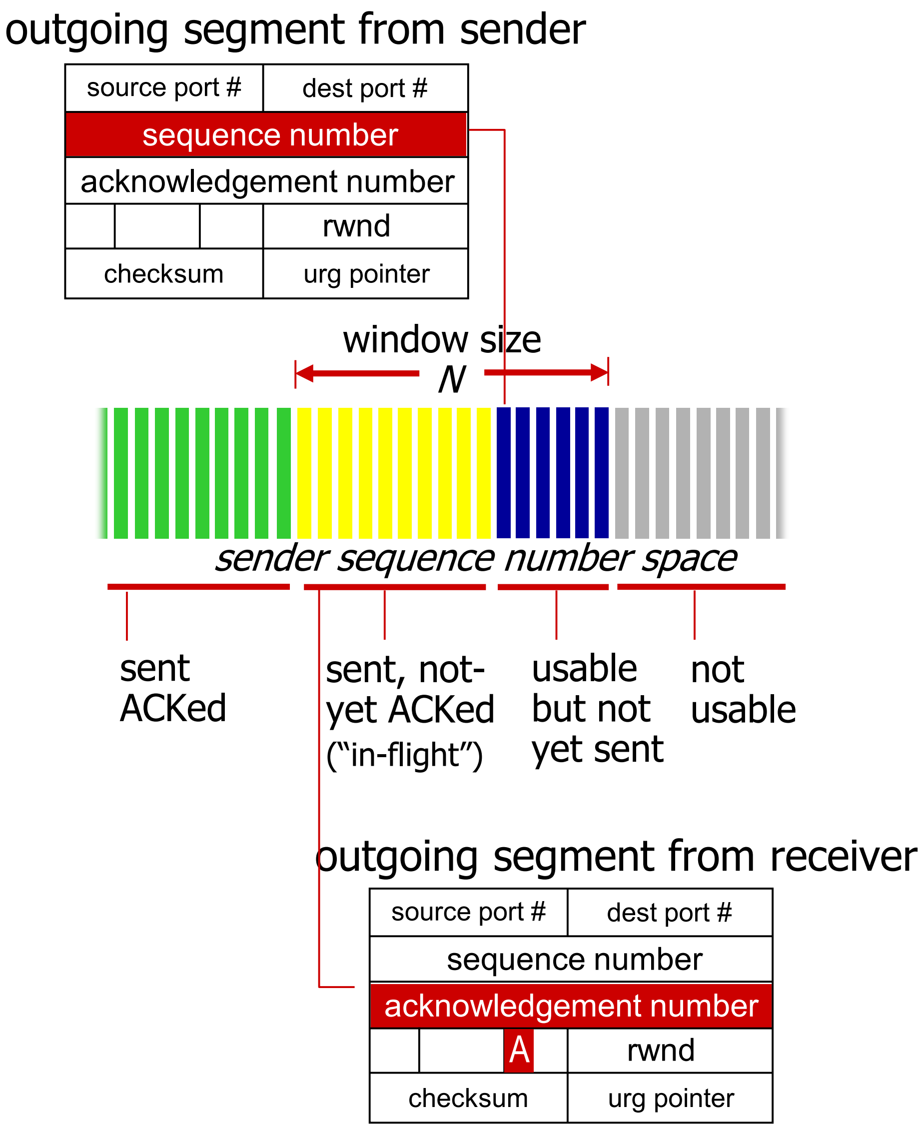
- byte stream "number" of first byte in segment's data

Acknowledgements:

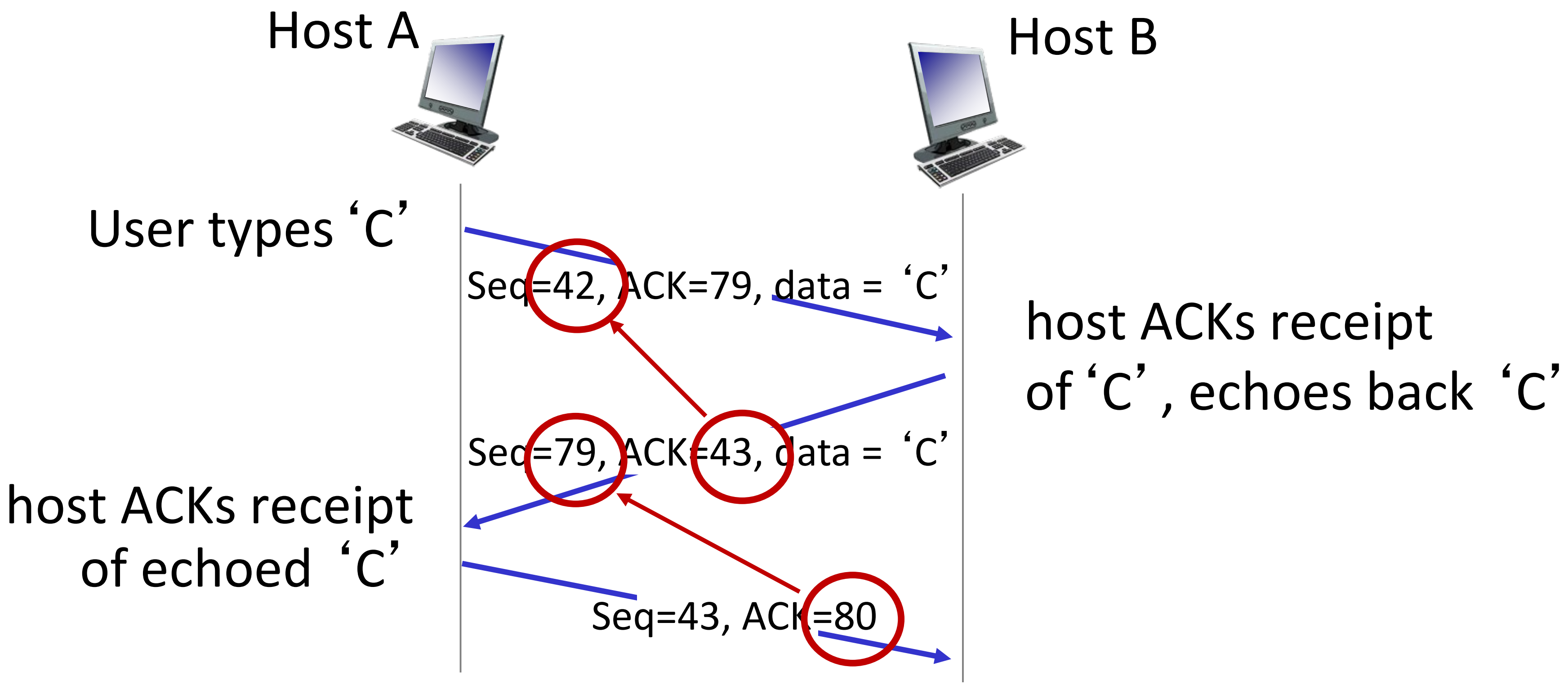
- seq # of next byte expected from other side
- cumulative ACK

Q: how receiver handles out-of-order segments

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



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

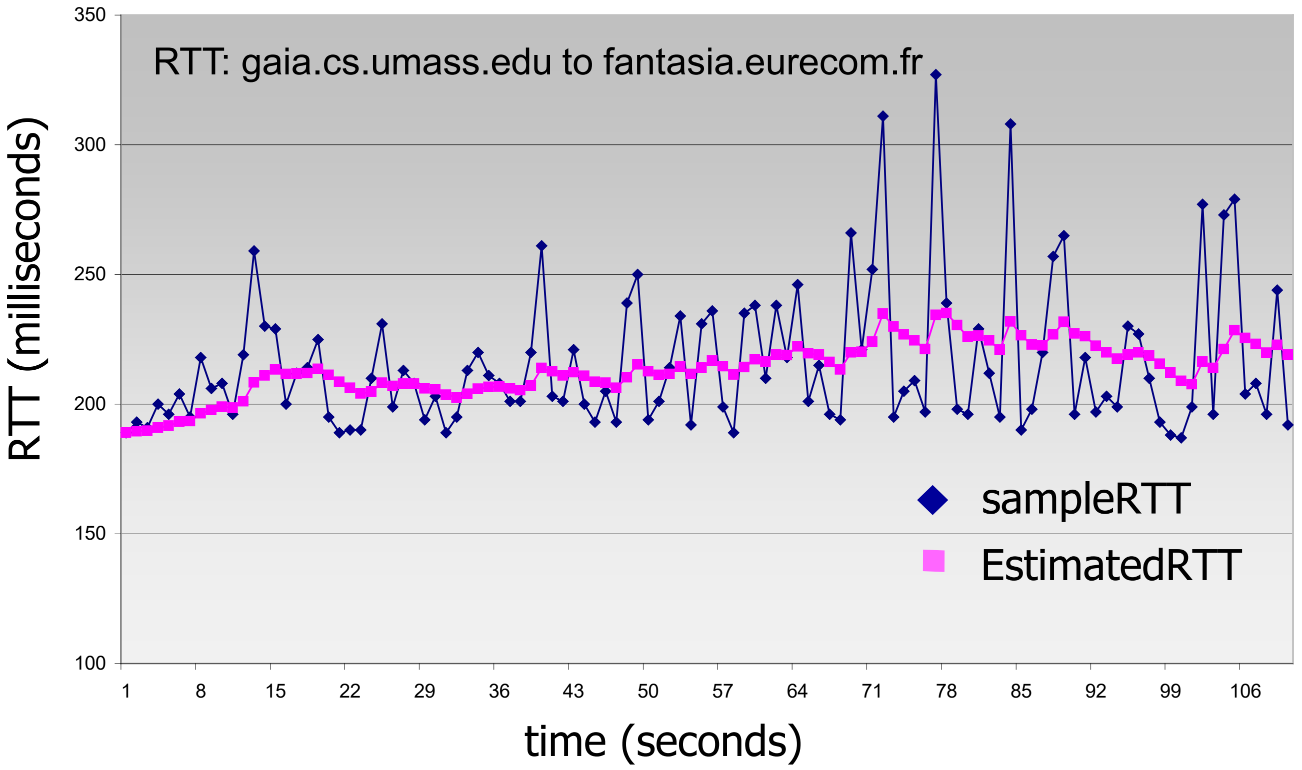
Q: how to estimate RTT?

- `SampleRTT`: measured time from segment transmission until ACK receipt
  - ignore retransmissions
- `SampleRTT` will vary, → we need "smooth" estimated RTT:
  - average several *recent* measurements, not just current `SampleRTT`

TCP round trip time, timeout

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

- exponential weighted moving average (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`** → large safety margin

```
TimeoutInterval = EstimatedRTT + 4*DevRTT
```

estimated RTT

"safety margin"

- **`DevRTT`**: EWMA of |**`SampleRTT - EstimatedRTT`**|:

```
DevRTT = (1-β)*DevRTT + β*|SampleRTT-EstimatedRTT|
```

(typically, β = 0.25)

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

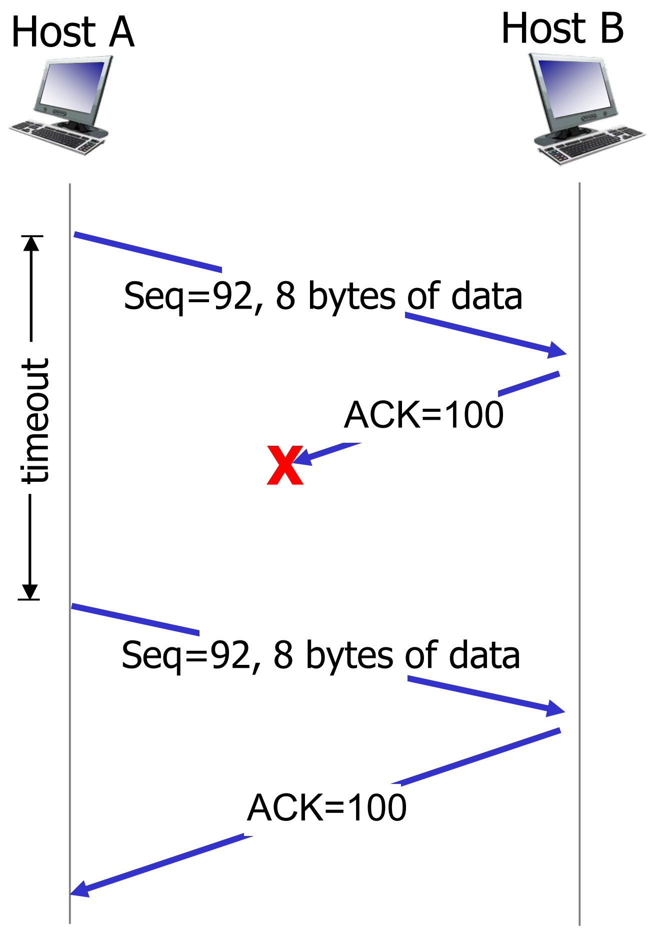
event: ACK received

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

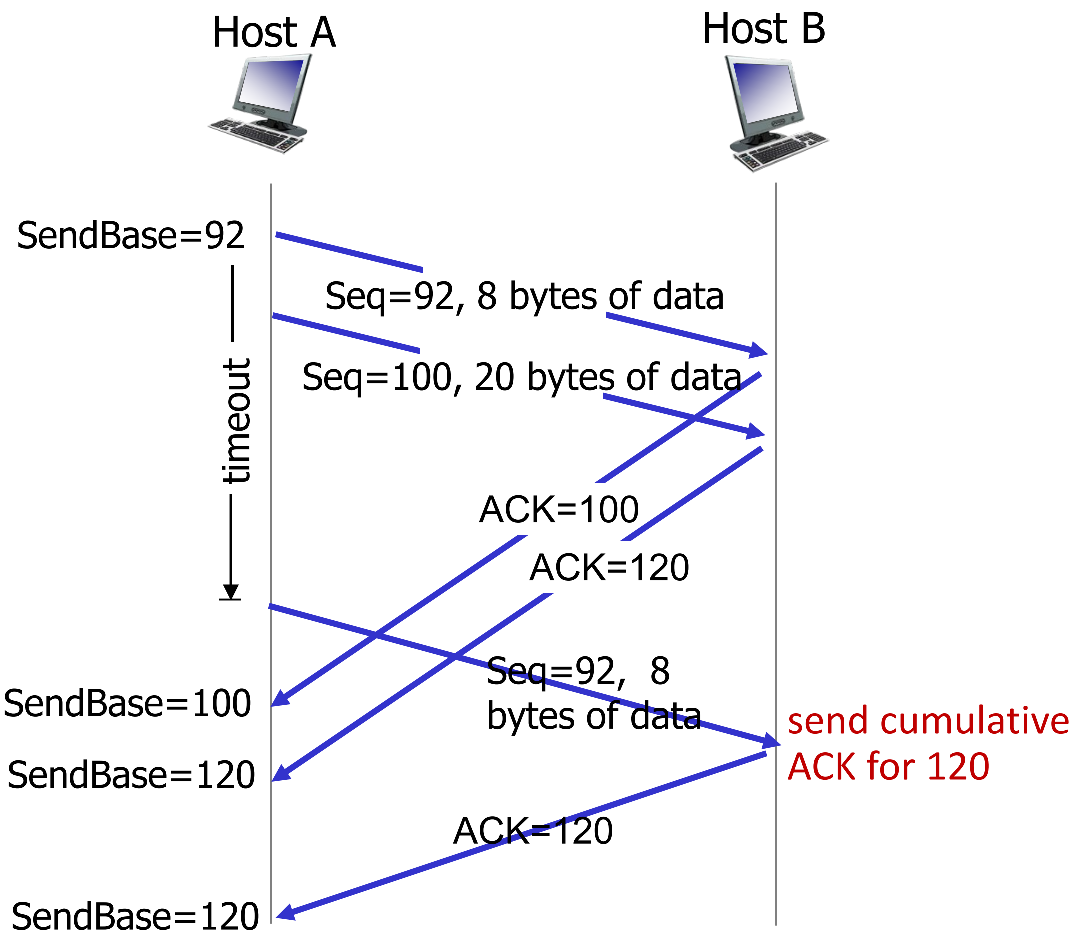
TCP Receiver: ACK generation [RFC 5681]

Event at receiver	*TCP receiver action*

TCP: retransmission scenarios

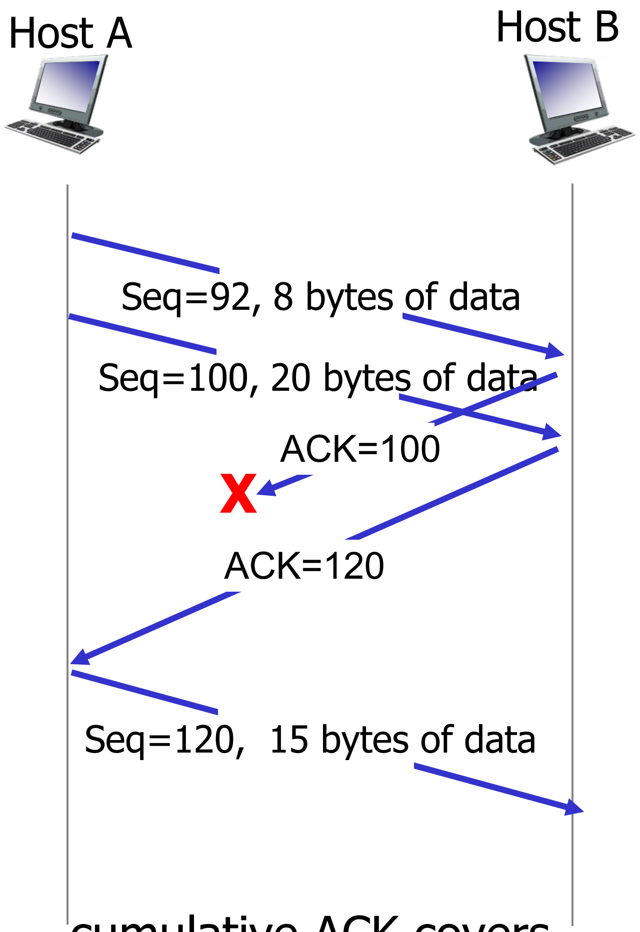


lost ACK scenario



premature timeout

TCP: retransmission scenarios



cumulative ACK covers
for earlier lost ACK

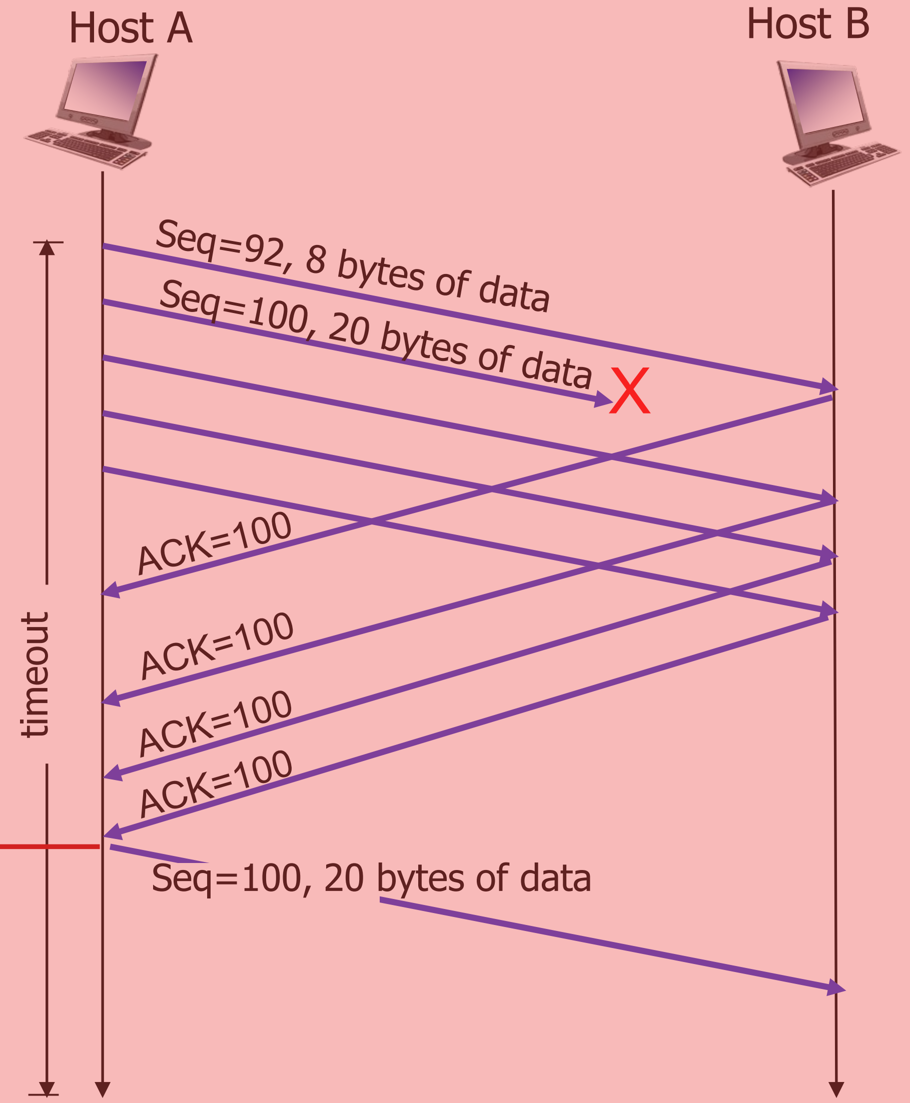
TCP fast retransmit

后面的seq是120，140，160这种吗？所以不是类似选择重传协议么

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!

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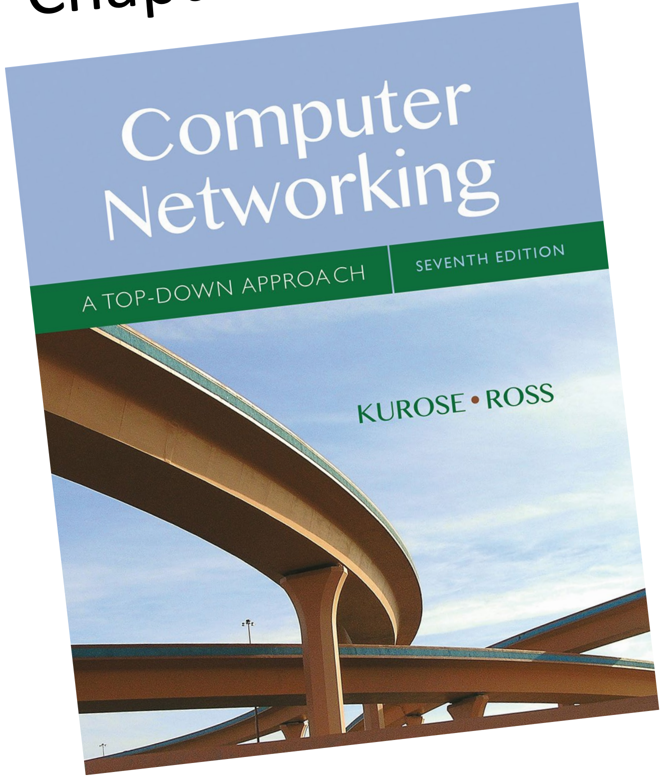
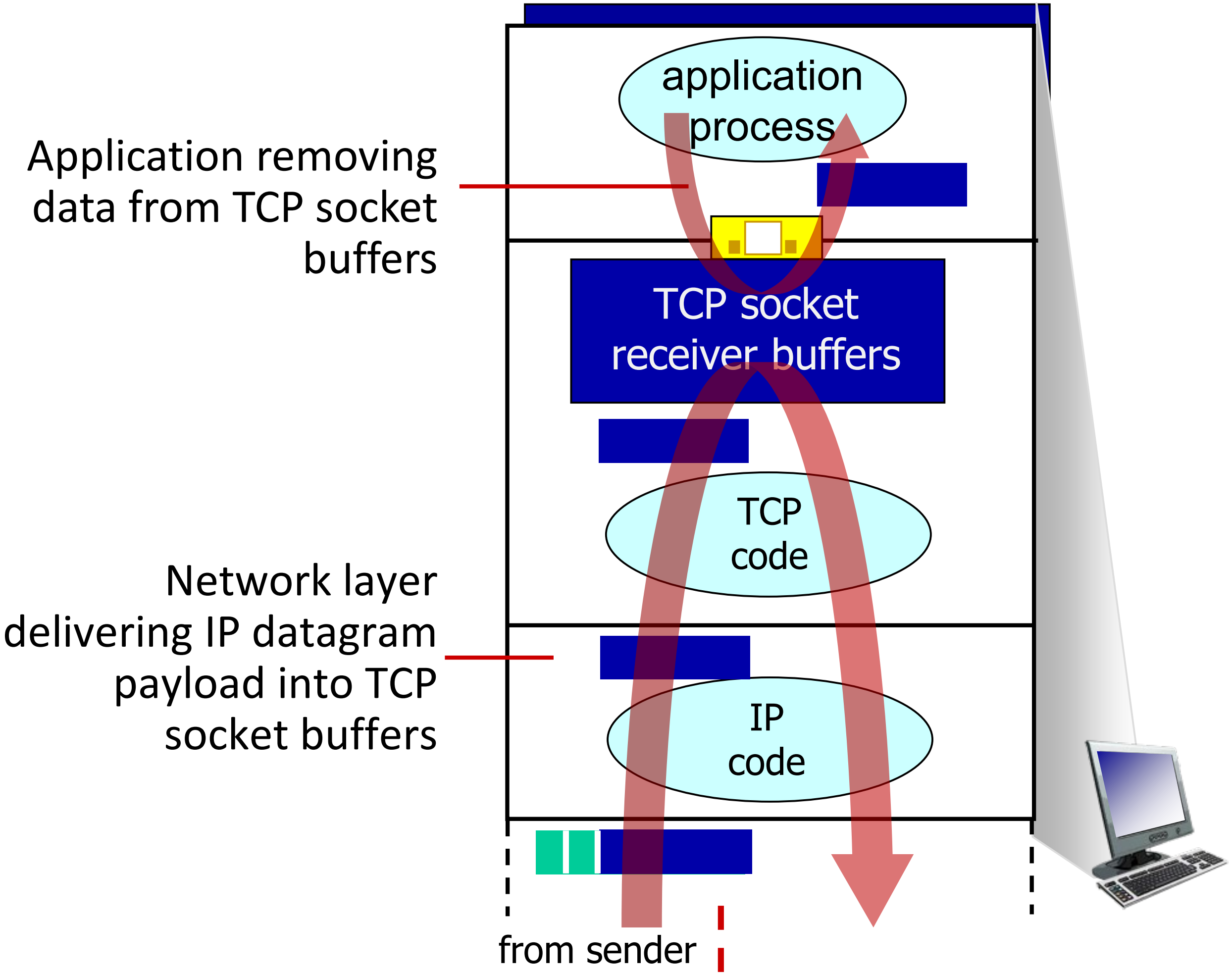


TCP flow control

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



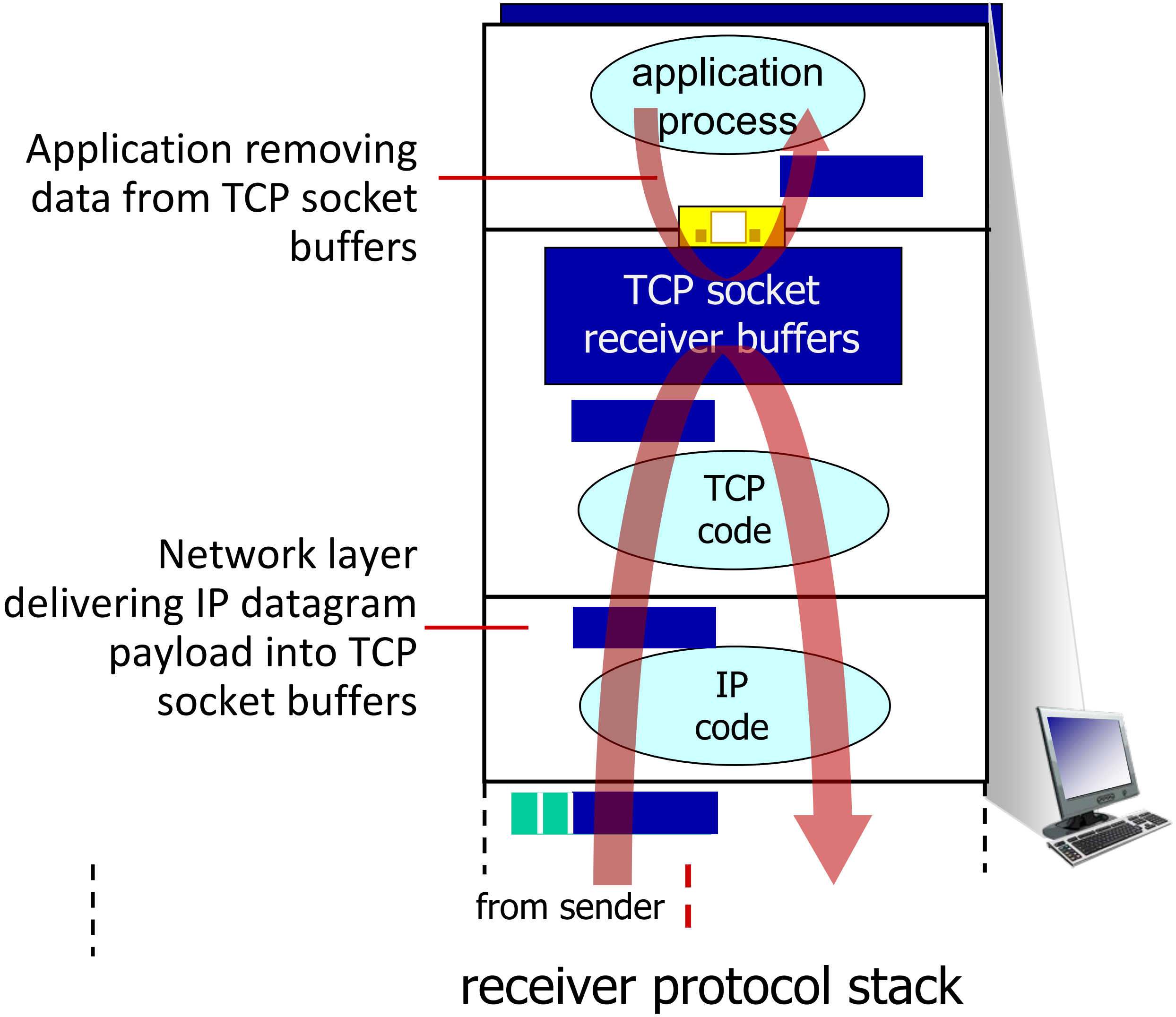
receiver protocol stack

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



TCP flow control

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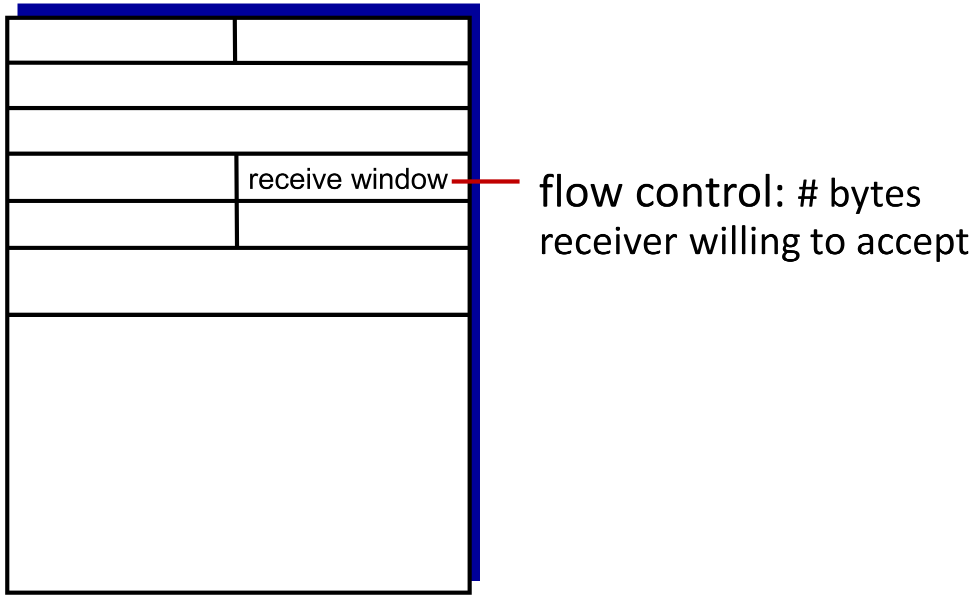

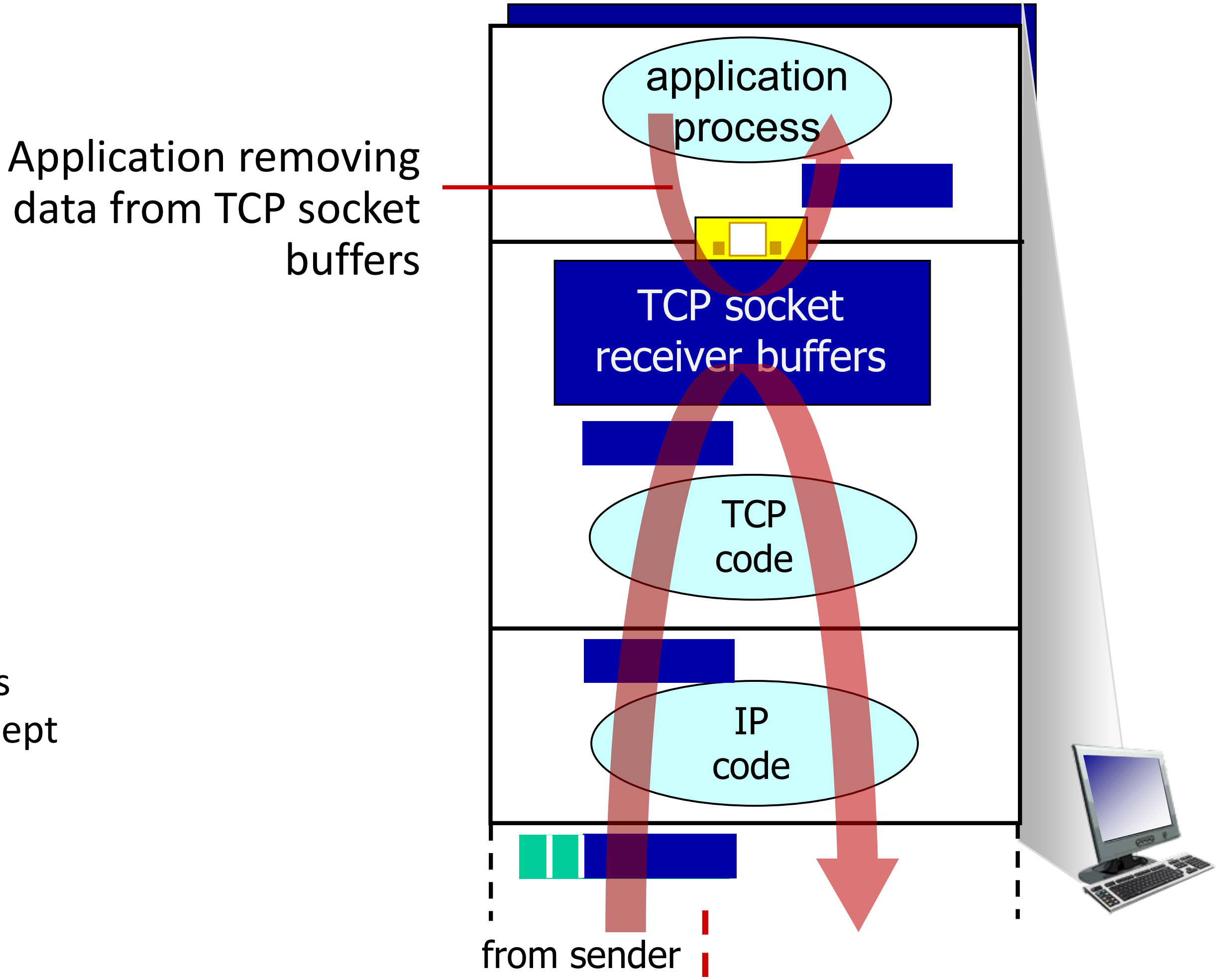

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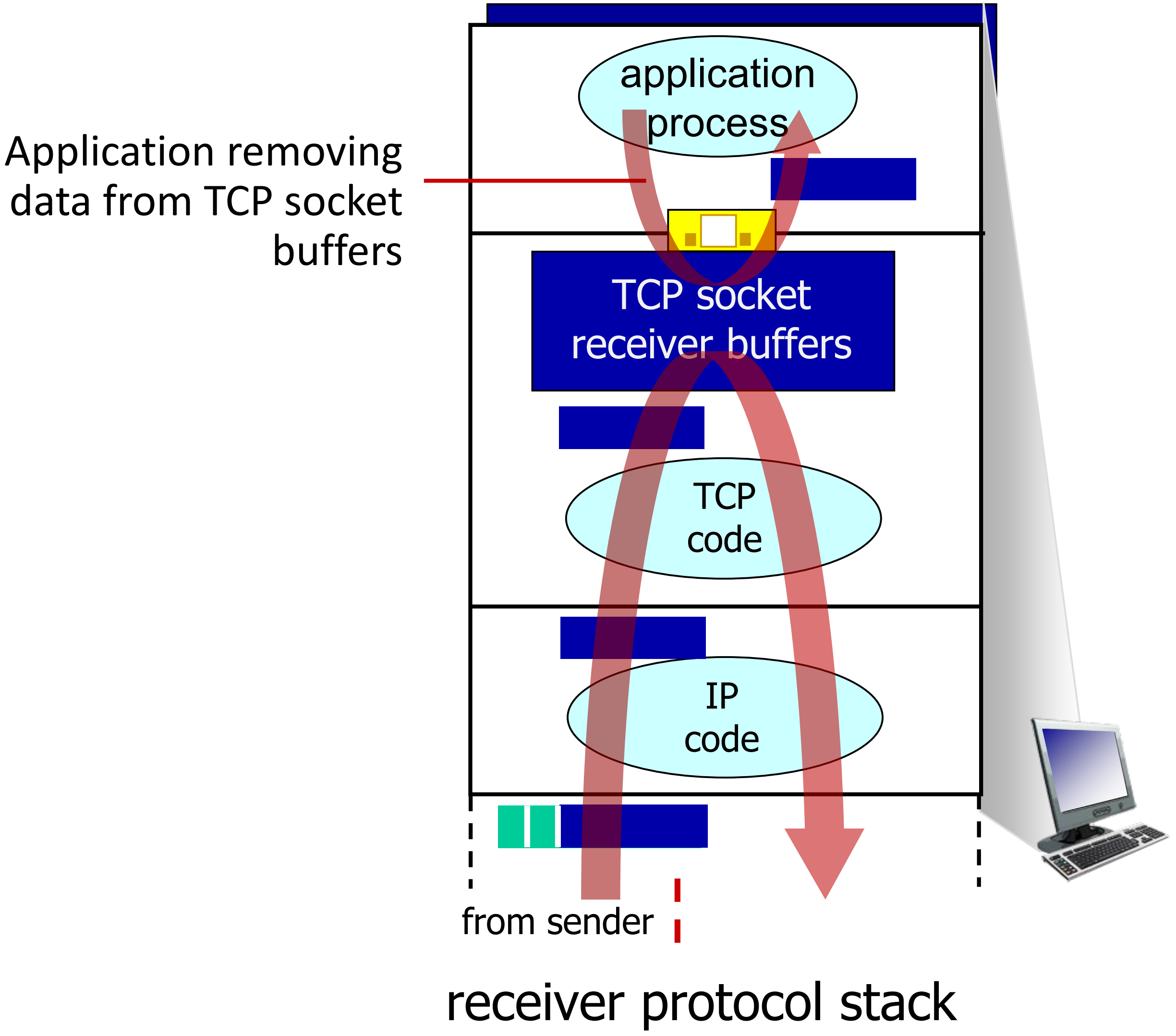
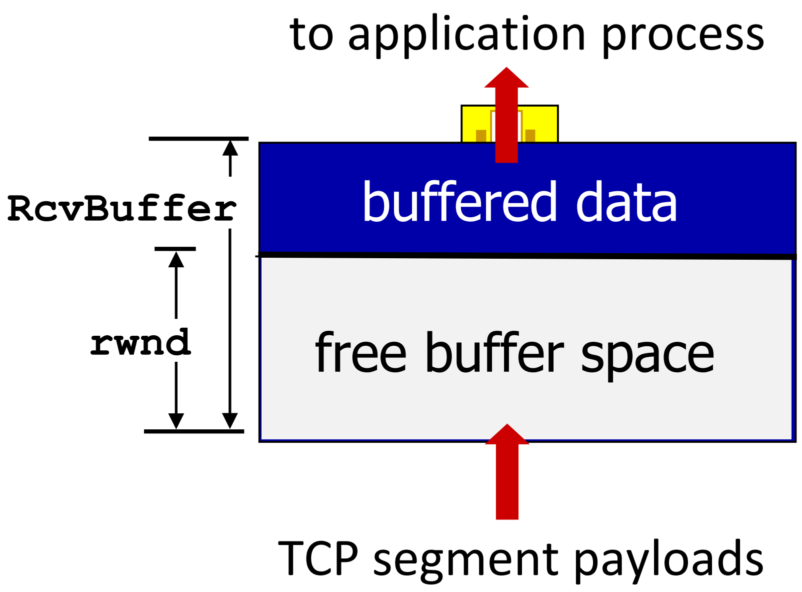


flow control

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

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

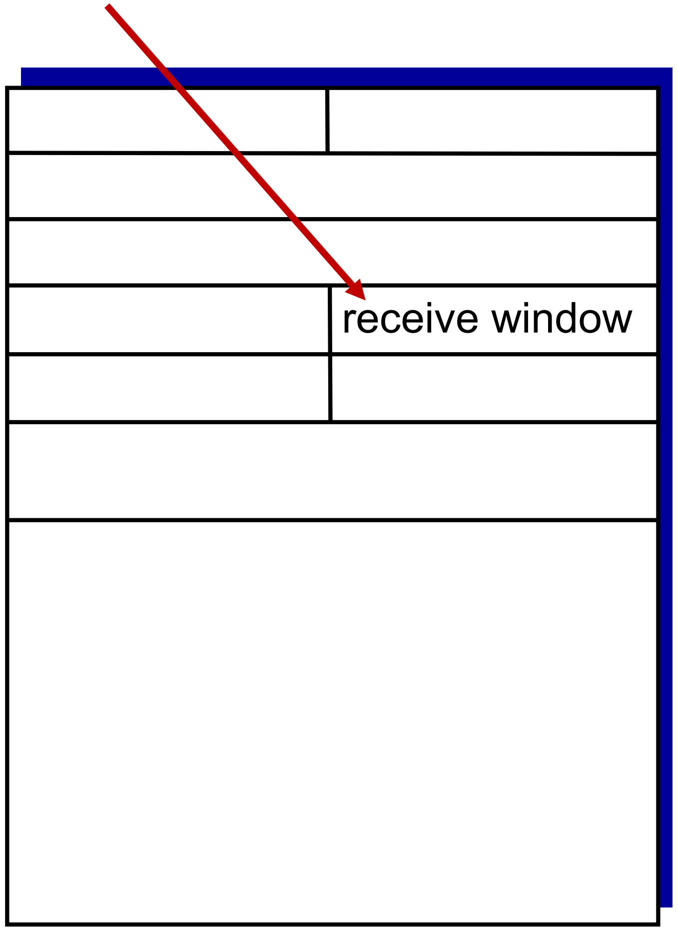


TCP receiver-side buffering

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

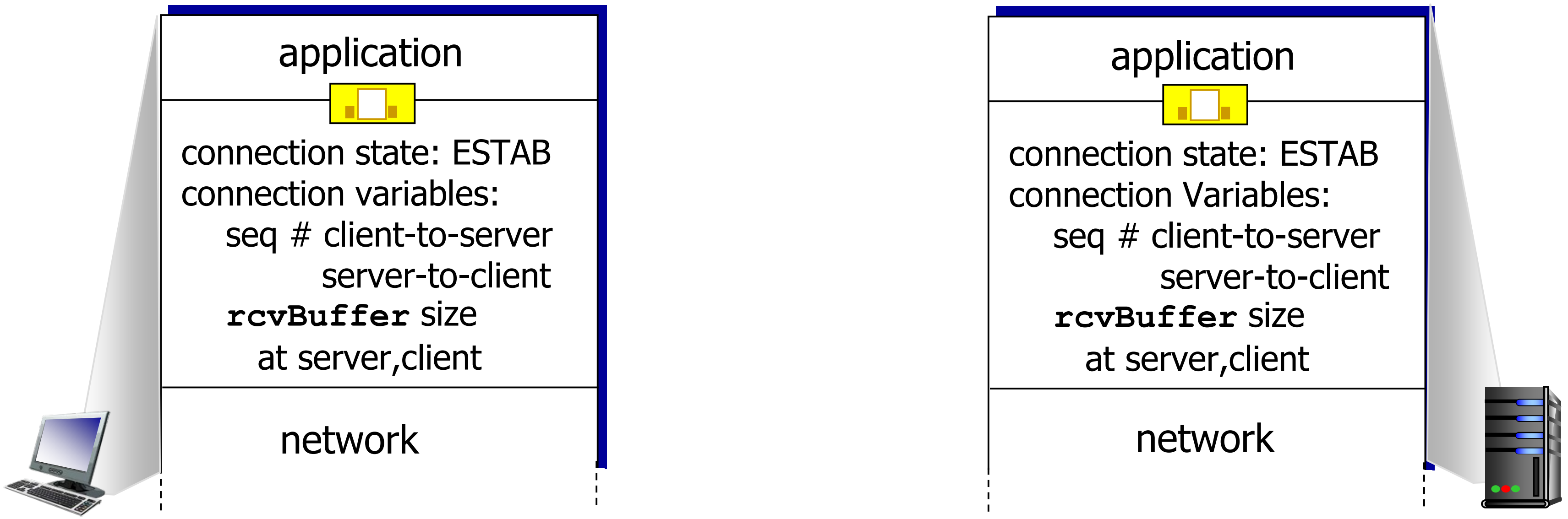


TCP segment format

TCP connection management

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)

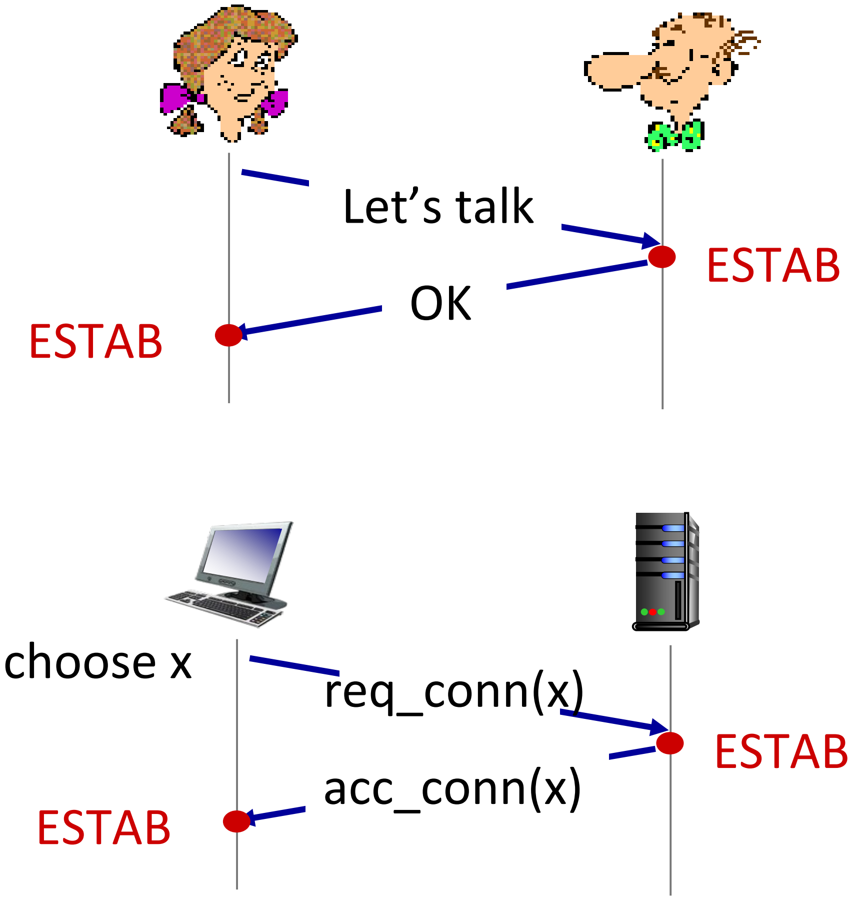



```
Socket clientSocket =
  newSocket("hostname","port number");
```

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

Agreeing to establish a connection

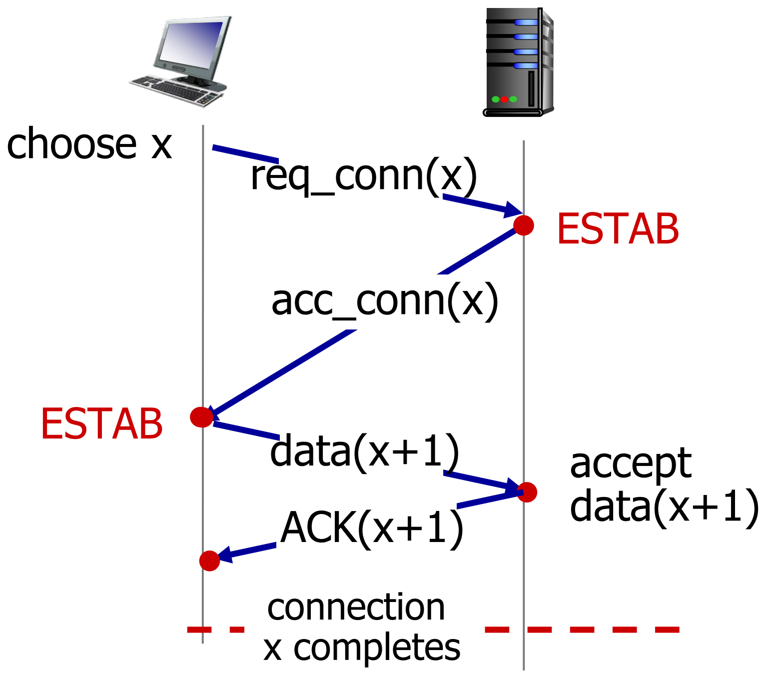
2-way handshake:



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

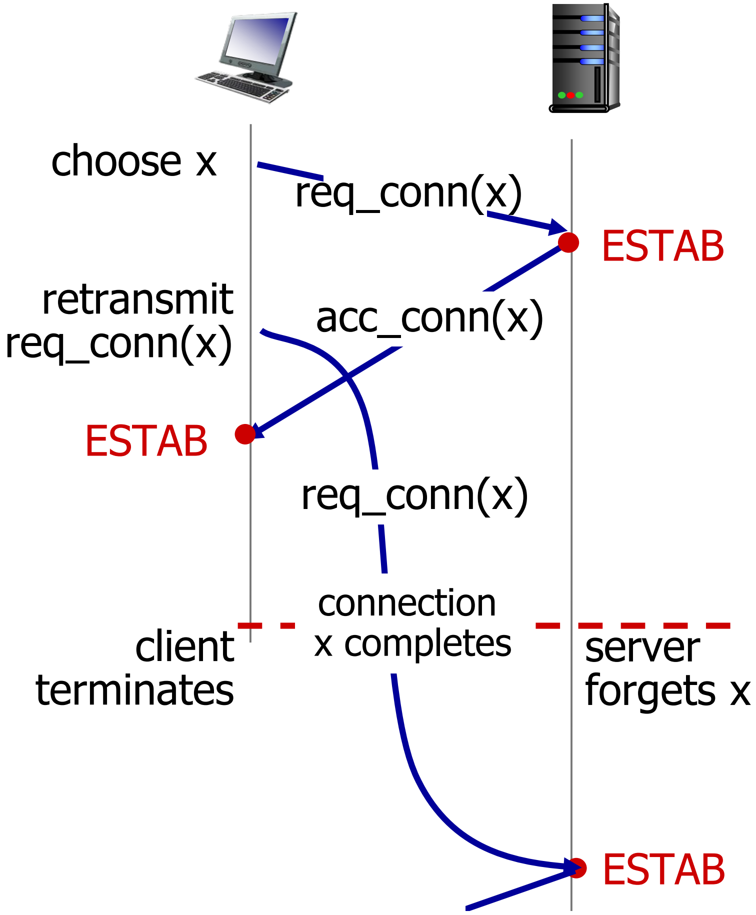
- variable delays
- retransmitted messages (e.g. req_conn(x)) due to message loss
- message reordering
- can't "see" other side

2-way handshake scenarios



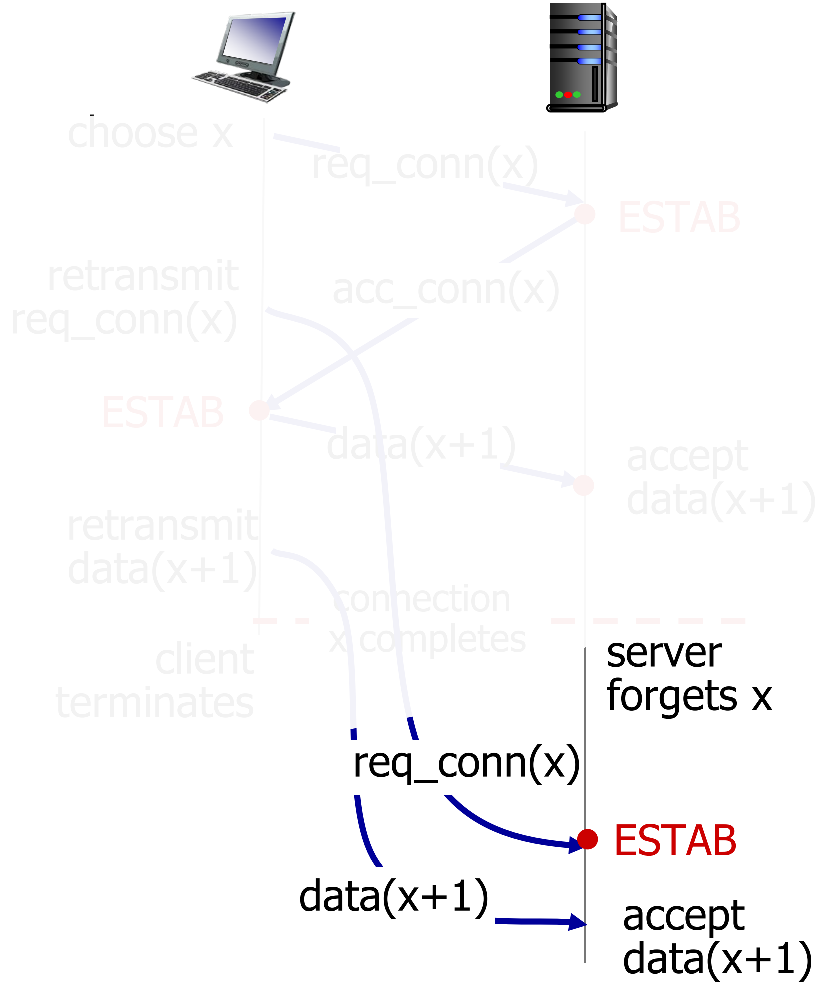
No problem!

2-way handshake scenarios



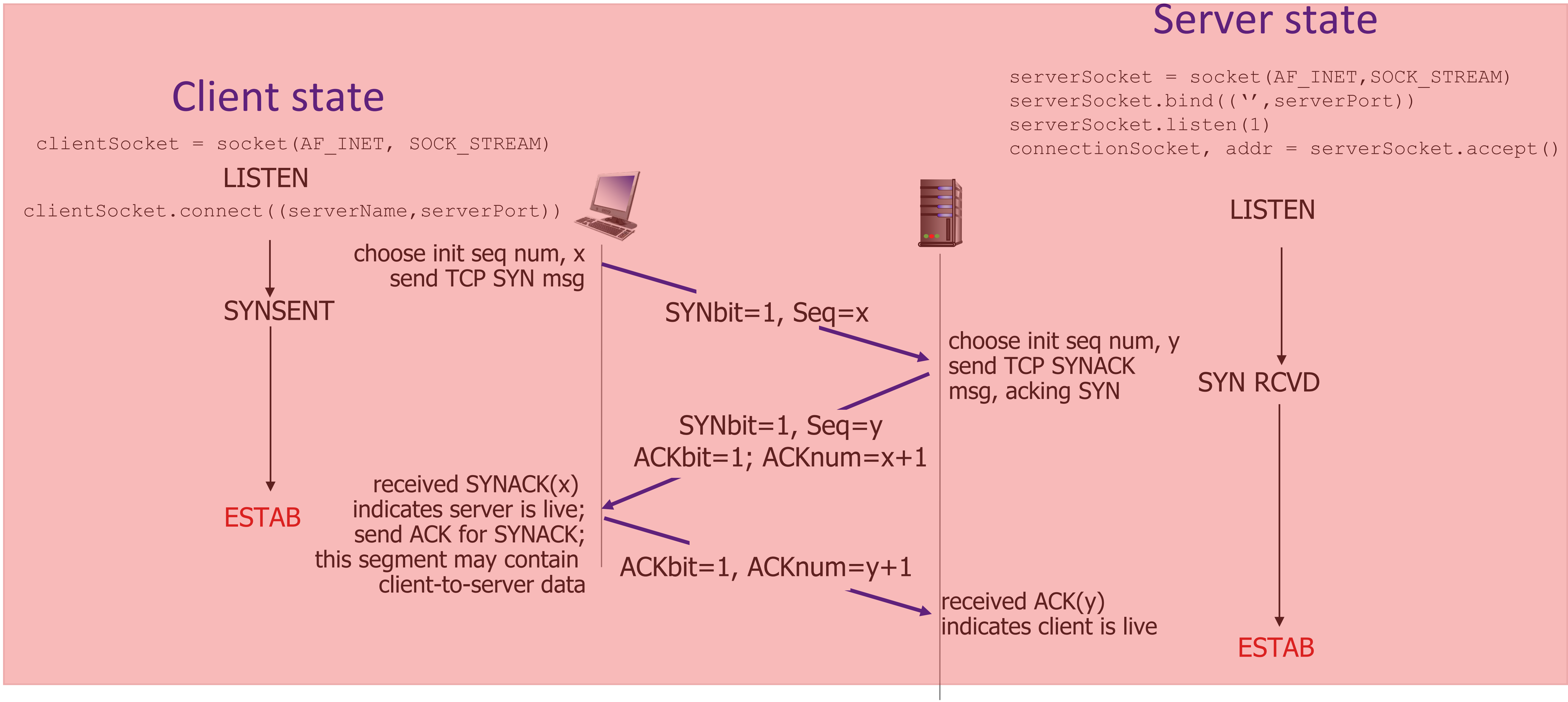
Problem: half open connection! (no client)

2-way handshake scenarios



Problem: dup data accepted!

TCP 3-way handshake



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

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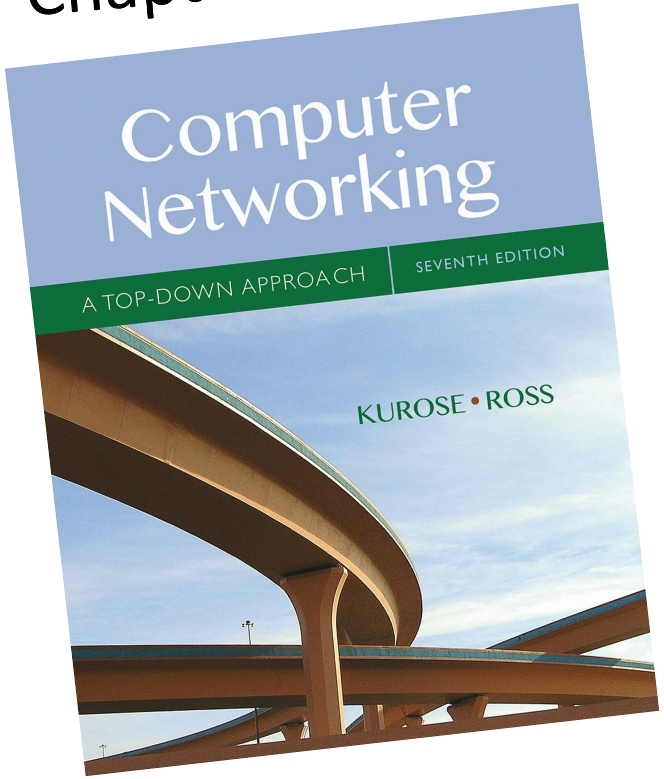


Principles of congestion control

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

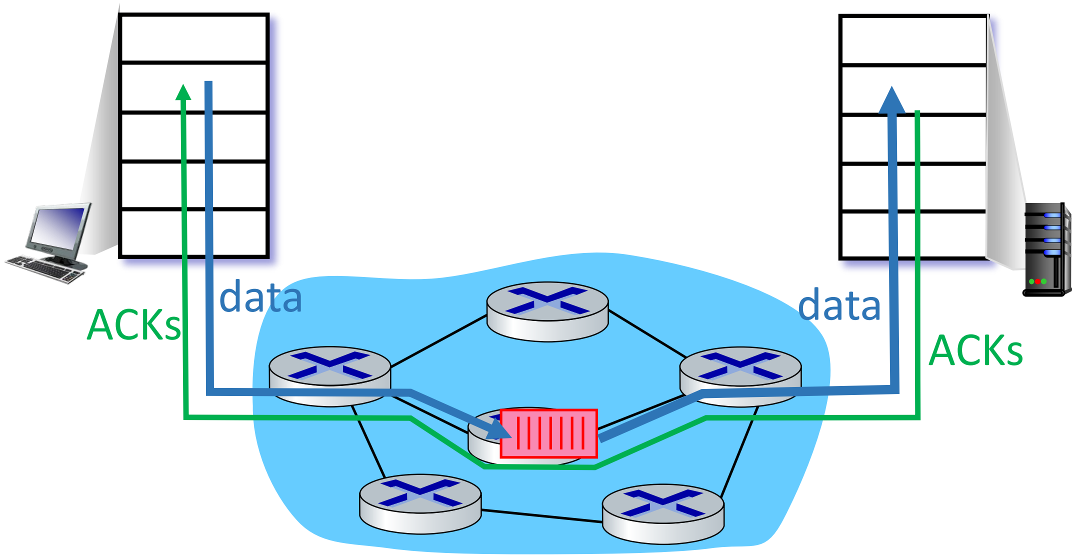
Flow Control vs. Congestion Control

- Flow control
  - Keeping *one fast sender* from overwhelming *a slow receiver*
- Congestion control
  - Keep a *set of senders* from overloading the *network*
    - E.g., persuade hosts to stop sending, or slow down
    - Typically has notions of fairness (i.e., sharing the pain)
- Different concepts, but similar mechanisms
  - TCP flow control: receiver window
  - TCP congestion control: congestion window
  - TCP window: *min{congestion window, receiver window}*

Approaches towards congestion control

End-end congestion control:

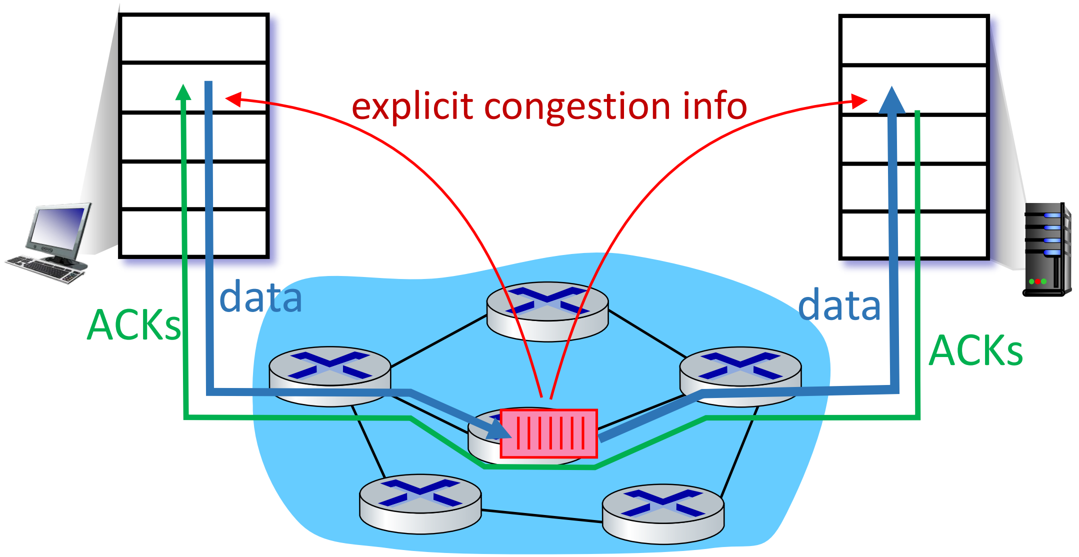
- no explicit feedback from network
- congestion *inferred* from observed loss, delay
- approach taken by TCP



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



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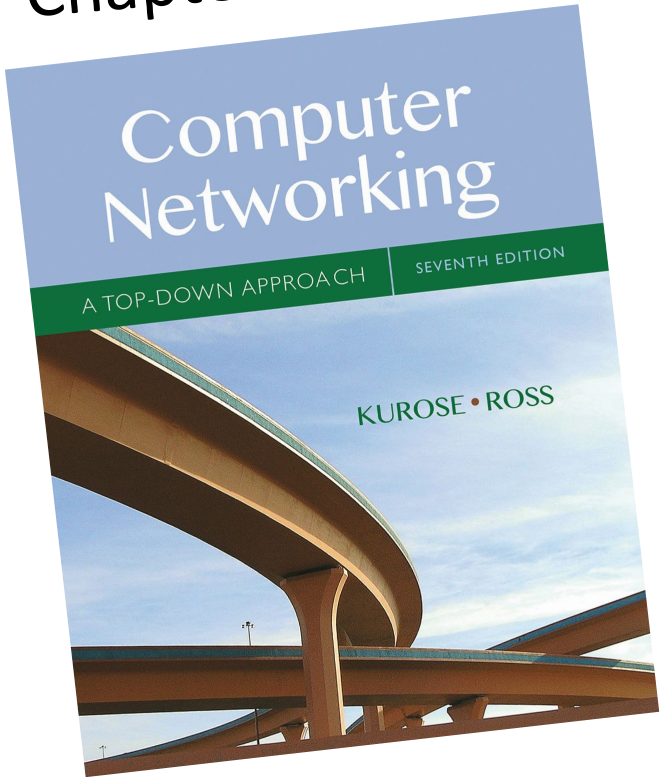


TCP congestion control: AIMD

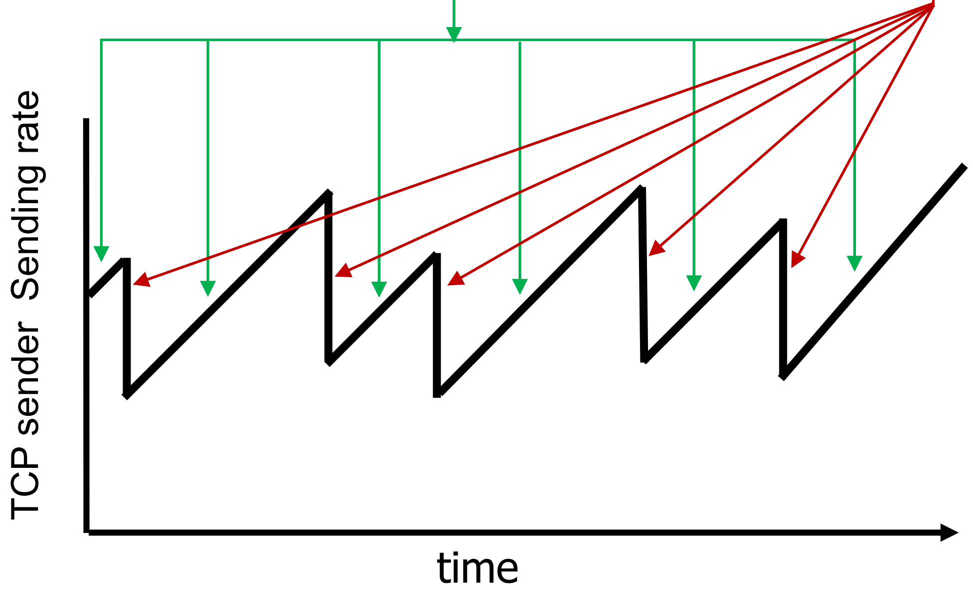
- *approach:* senders can increase sending rate until packet loss (congestion) occurs, then decrease sending rate on loss event

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



AIMD sawtooth behavior: *probing* for bandwidth

TCP AIMD: more

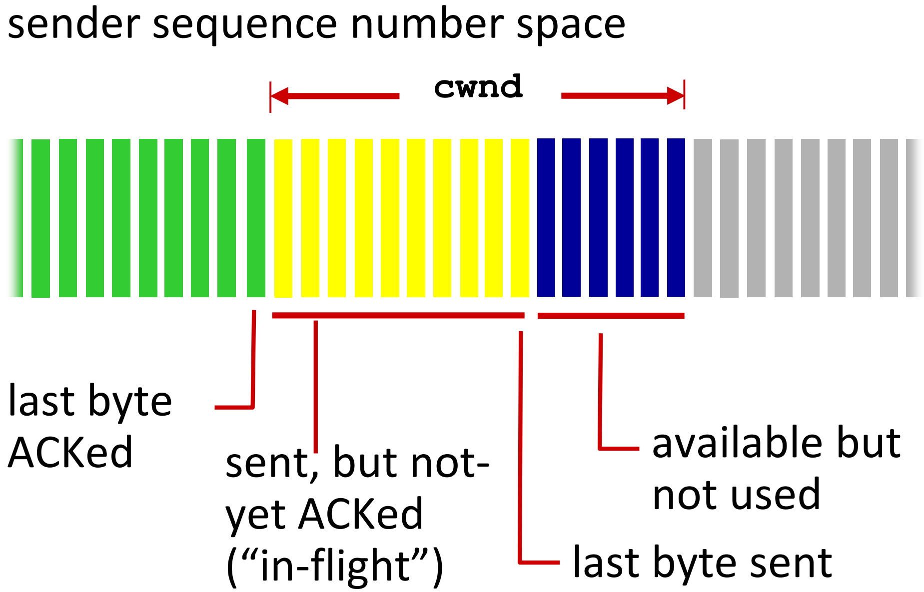
Multiplicative decrease detail: sending rate is

- 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:

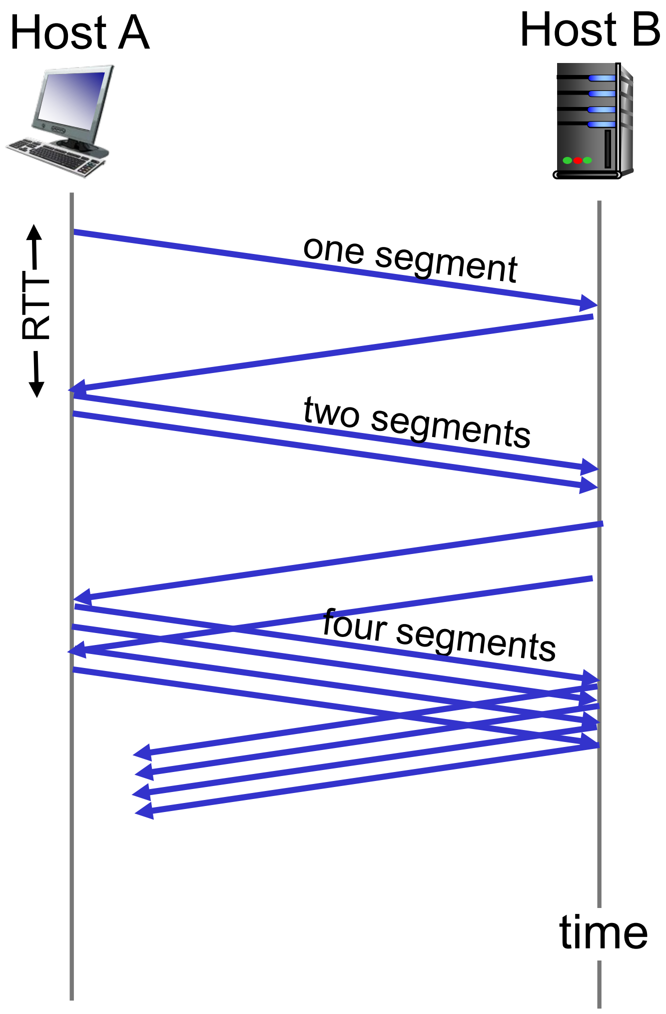
- *roughly:* send `cwnd` bytes, wait RTT for ACKS, then send more bytes

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

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

TCP slow start

- 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



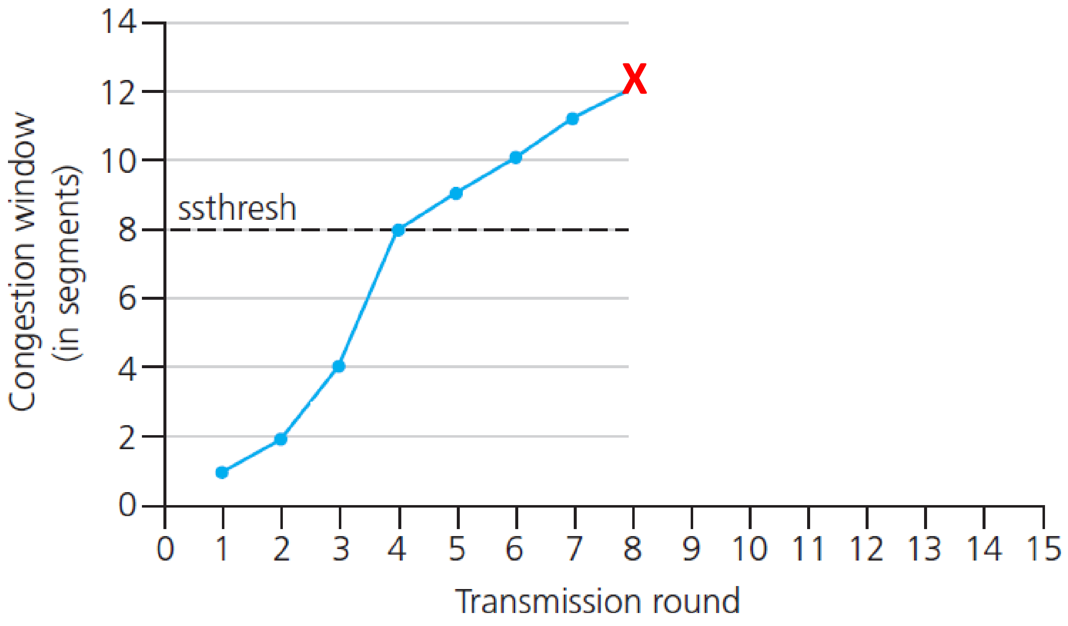
TCP: detecting, reacting to loss

- loss indicated by timeout:
  - **`cwnd`** set to 1 MSS;
  - window then grows exponentially (as in slow start) to threshold, then grows linearly
- loss indicated by 3 duplicate ACKs: **TCP RENO**
  - dup ACKs indicate network capable of delivering *some* segments
  - **`cwnd`** is cut in half window then grows linearly
- **TCP Tahoe** always sets **`cwnd`** to 1 (timeout or 3 duplicate acks)

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.

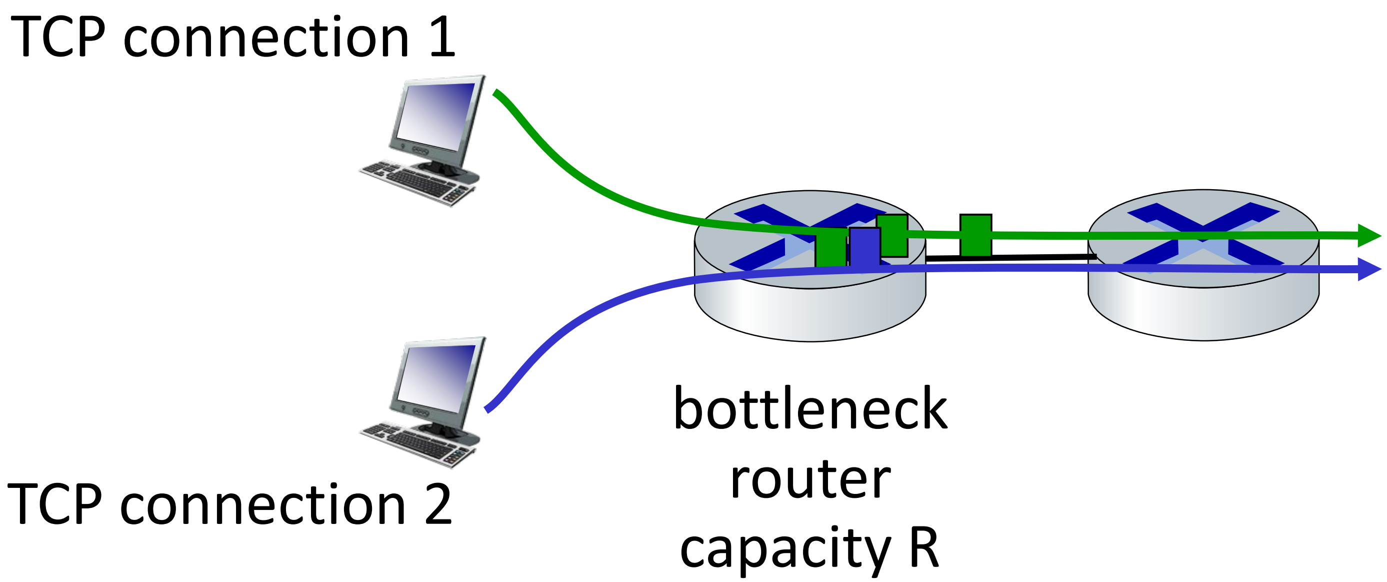


Implementation:

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

TCP fairness

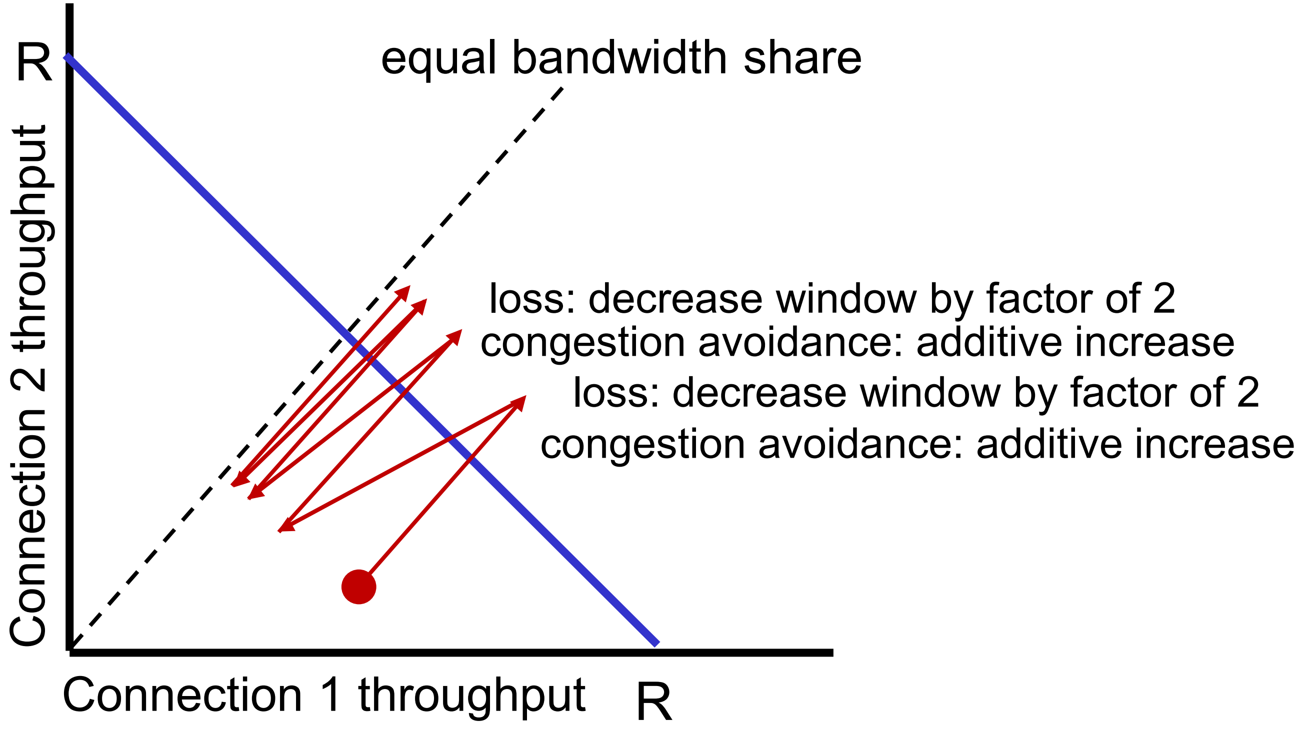
Fairness goal: if K TCP sessions share same bottleneck link of bandwidth R, each should have average rate of R/K



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

Fairness: must all network apps be "fair"?

Fairness and UDP

- multimedia apps often do not use TCP
  - do not want rate throttled by congestion control
- instead use UDP:
  - send audio/video at constant rate, tolerate packet loss
- there is no "Internet police" policing use of congestion control

Fairness, parallel TCP connections

- application can open *multiple* parallel connections between two hosts
- web browsers do this , e.g., link of rate R with 9 existing connections:
  - new app asks for 1 TCP, gets rate R/10
  - new app asks for 11 TCPs, gets >R/2