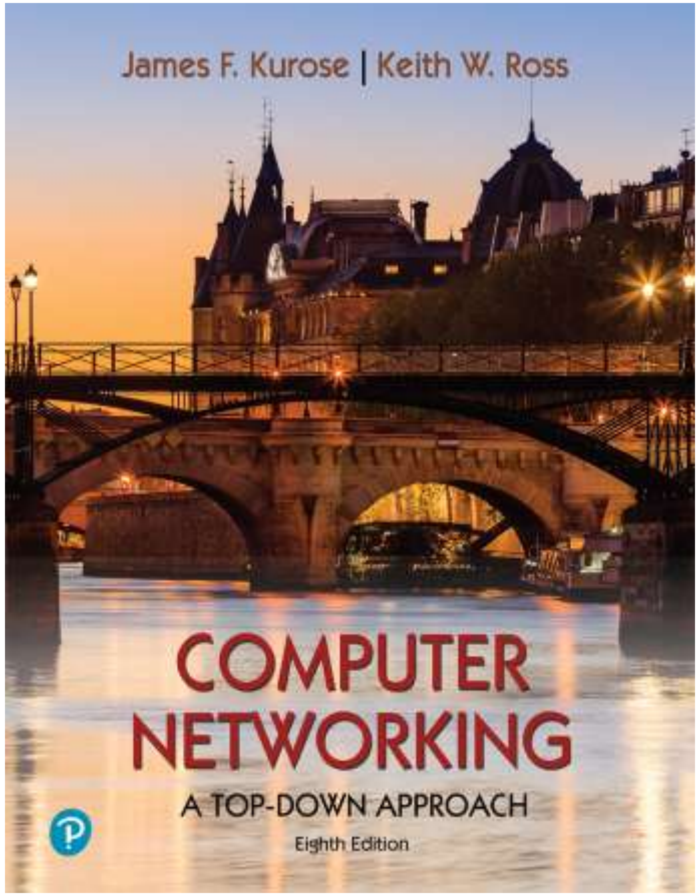
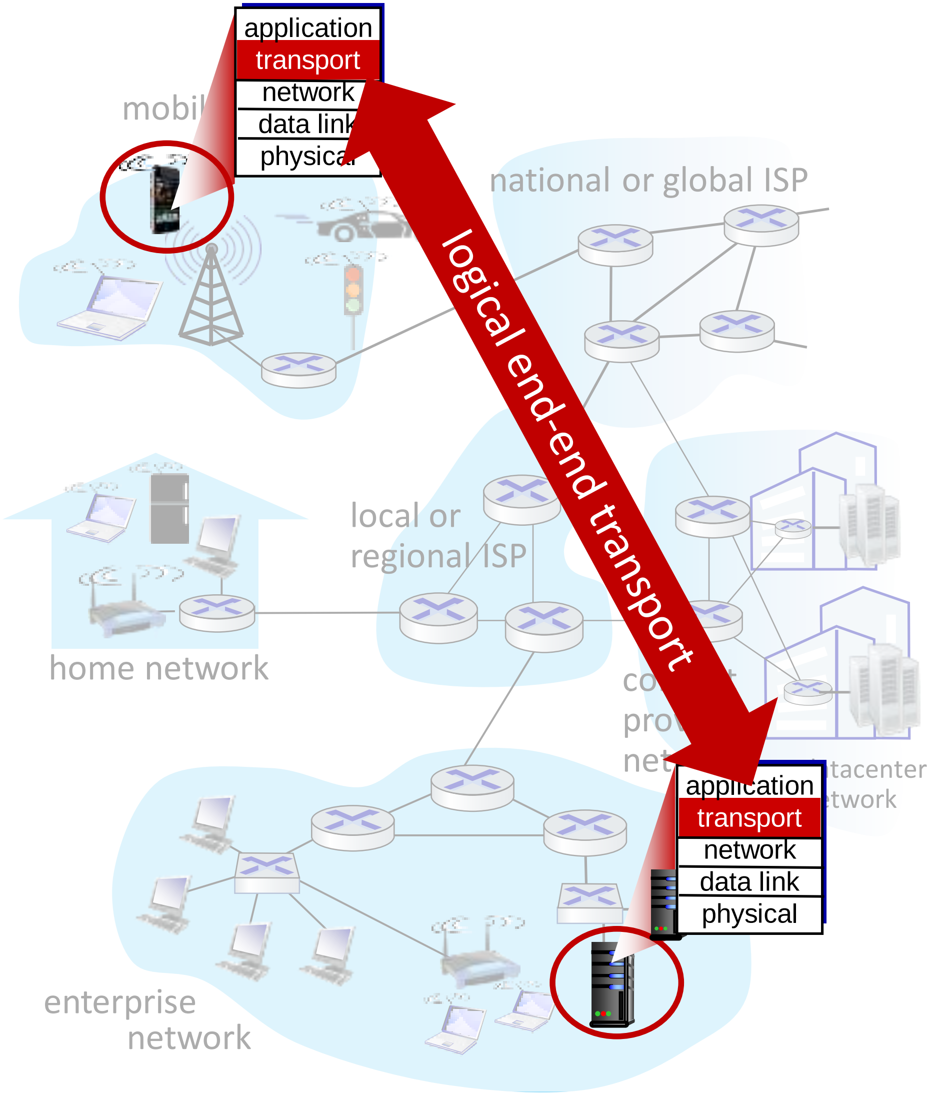
TCP
UDP
Congestion Control
Flow Control
TCP Tahoe
TCP Reno
TCP New Reno
TCP CUBIC



Computer Networking: A Top-Down Approach
8^{th} edition
Jim Kurose, Keith Ross
Pearson, 2020

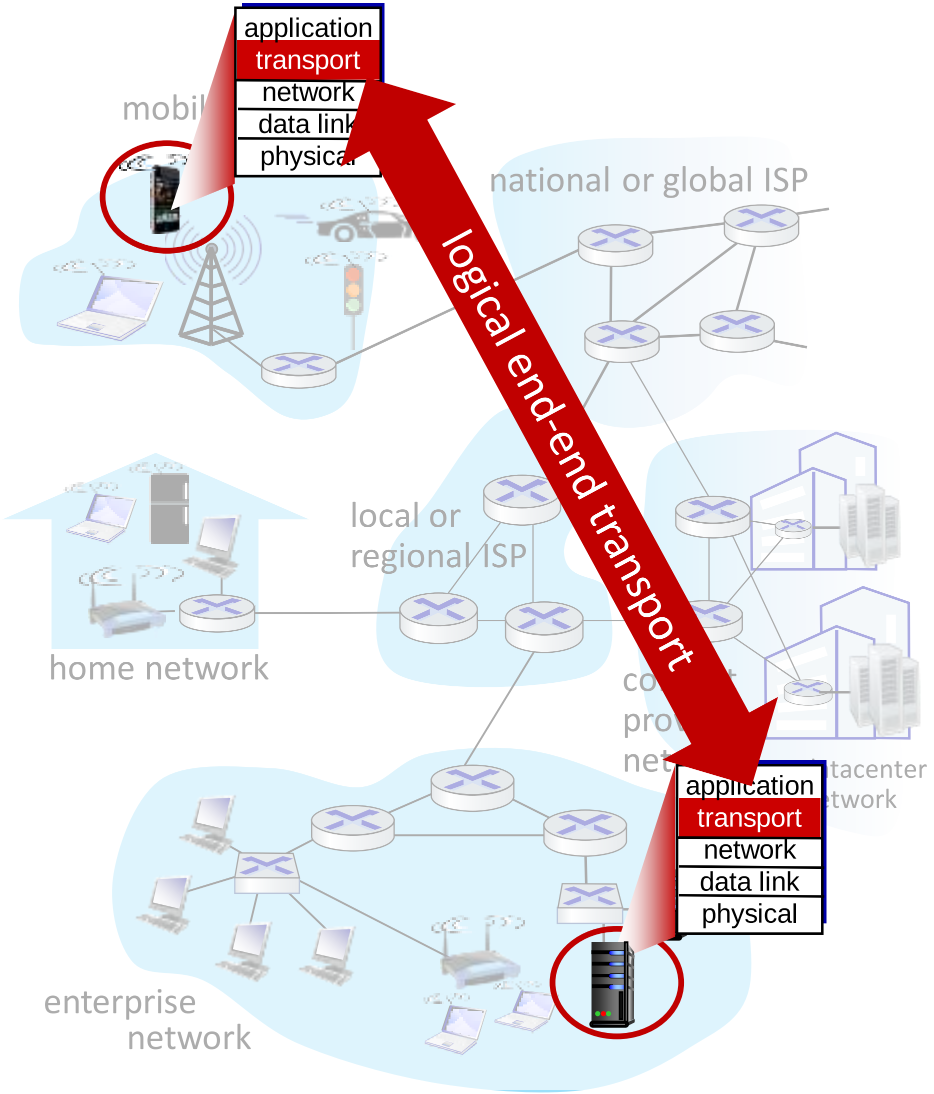
Transport services and protocols

- 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



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



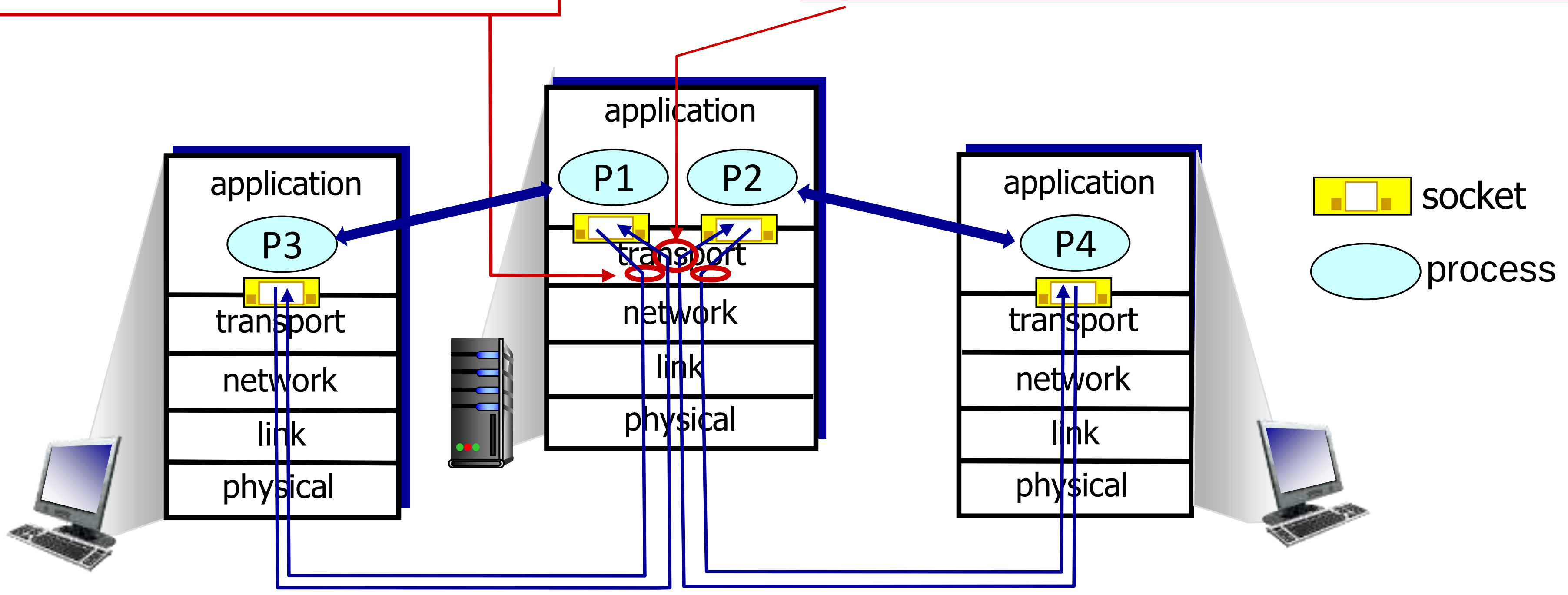
Multiplexing/demultiplexing

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

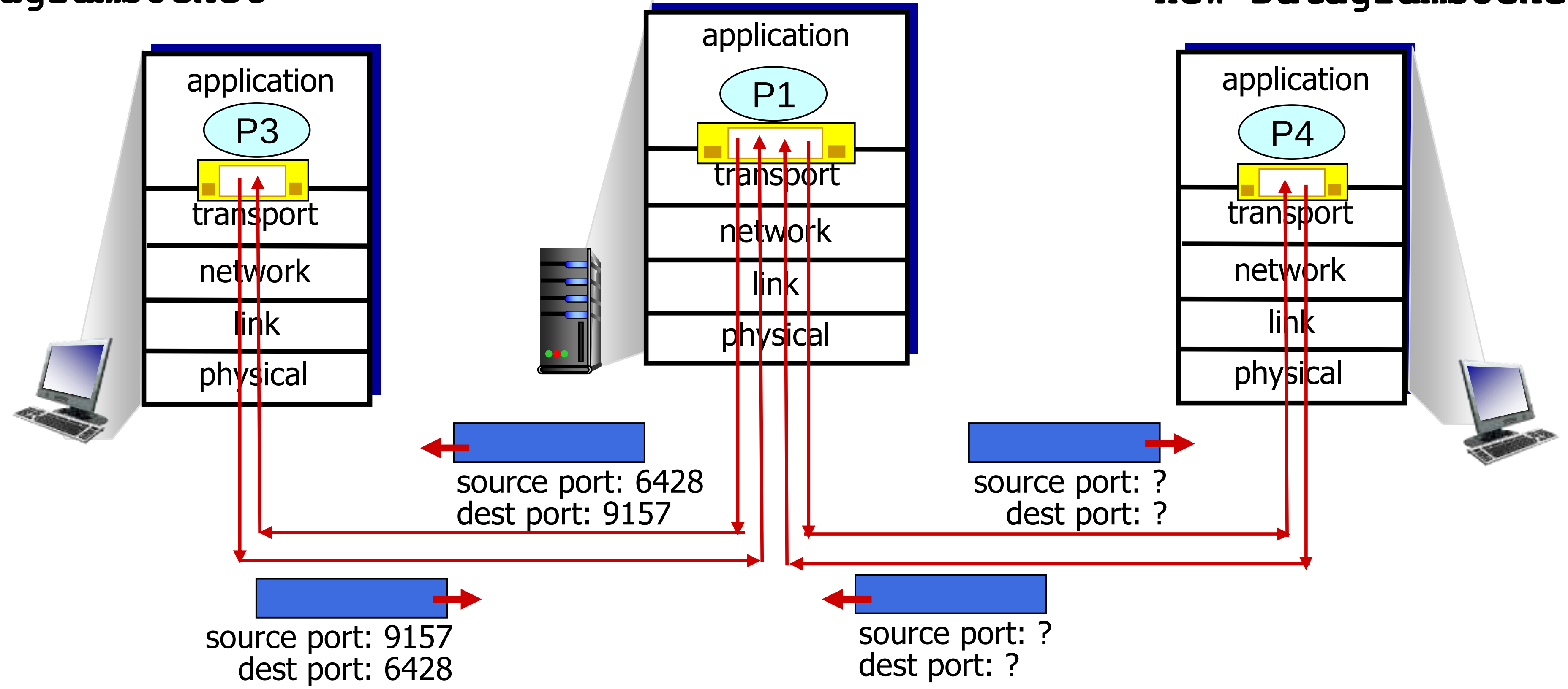


Connectionless demultiplexing: an example

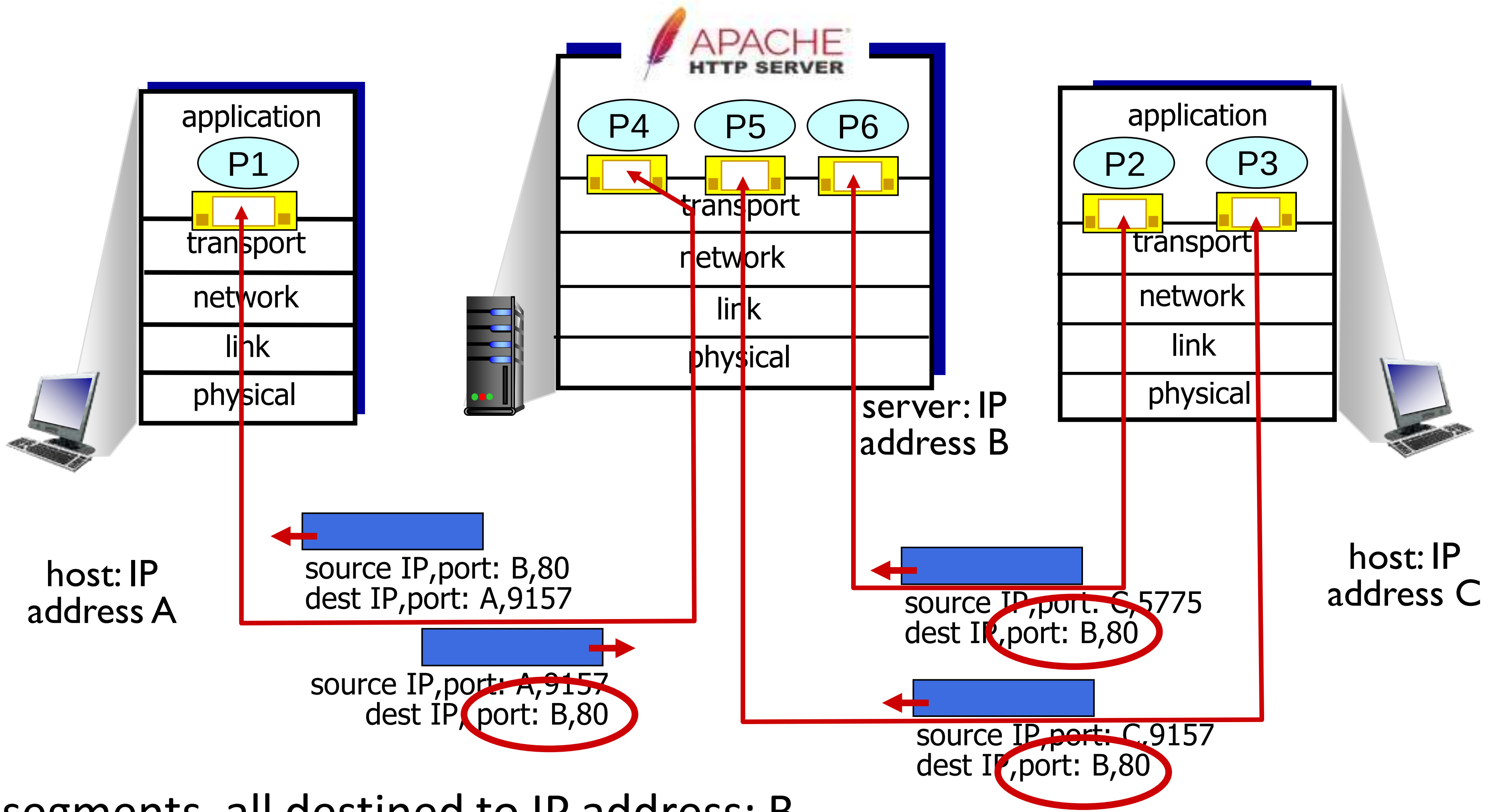
```
DatagramSocket mySocket2 =
 new DatagramSocket
 (9157);
```

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

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



Connection-oriented demultiplexing: example



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

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 [Do not handle Buffer size, congestion control parameters, Seq and Ack numbers]
- 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

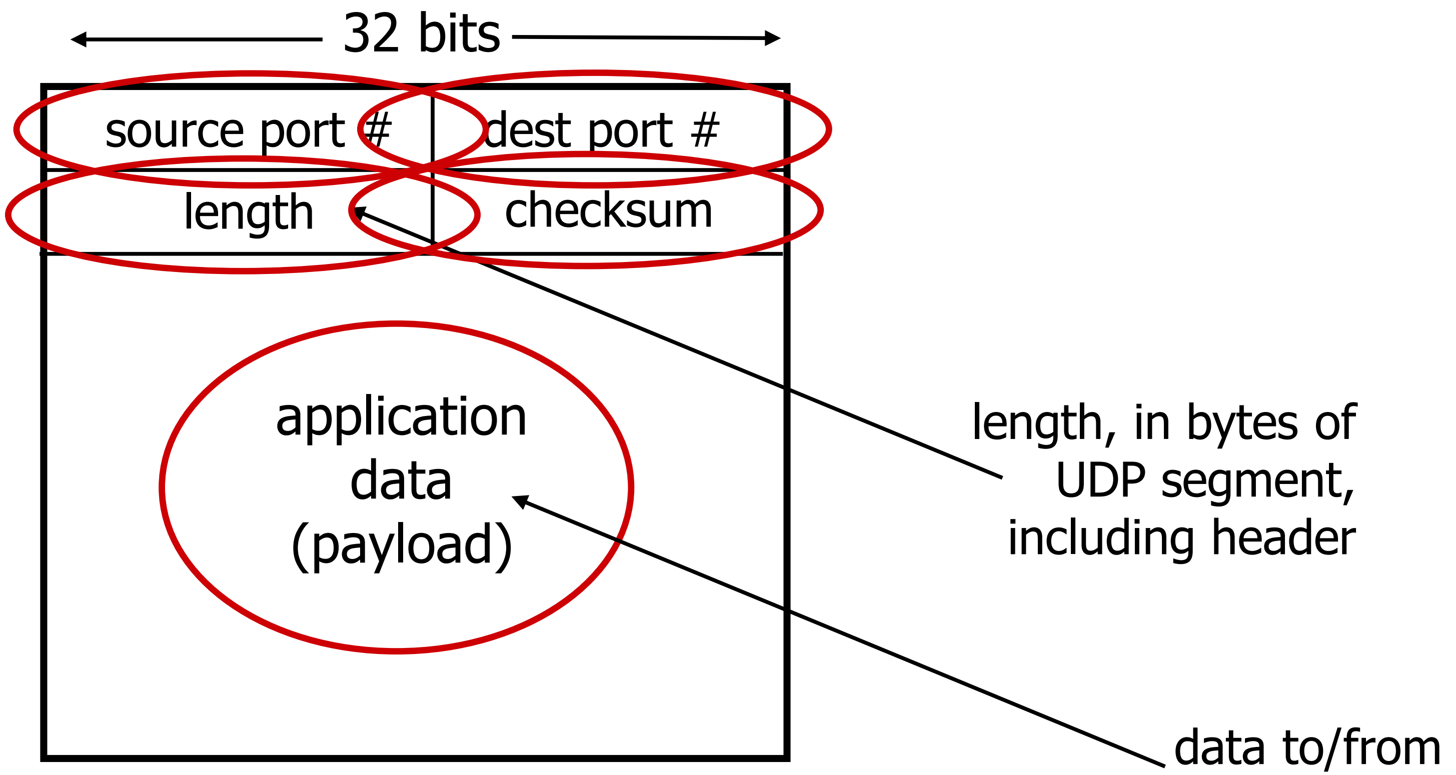
This User Datagram Protocol (UDP) is defined to make available a datagram mode of packet-switched computer communication in the environment of an interconnected set of computer networks. This protocol assumes that the Internet Protocol (IP) [1] is used as the underlying protocol.

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      |   Destination   |
|      Port       |      Port       |
+--------+--------+--------+--------+
|                 |                 |
|     Length      |    Checksum     |
+--------+--------+--------+--------+
|
|          data octets ...
+---------------- ...
```

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 $\neq$ 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 (one's complement sum) of segment content
- checksum value put into UDP checksum field

receiver:

- compute checksum of received segment
- check the checksum value calculated by receiver:
  - **checksum of receiver → all zero bits - no error**
  - **checksum of receiver → any bit non-zero – error present**

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?

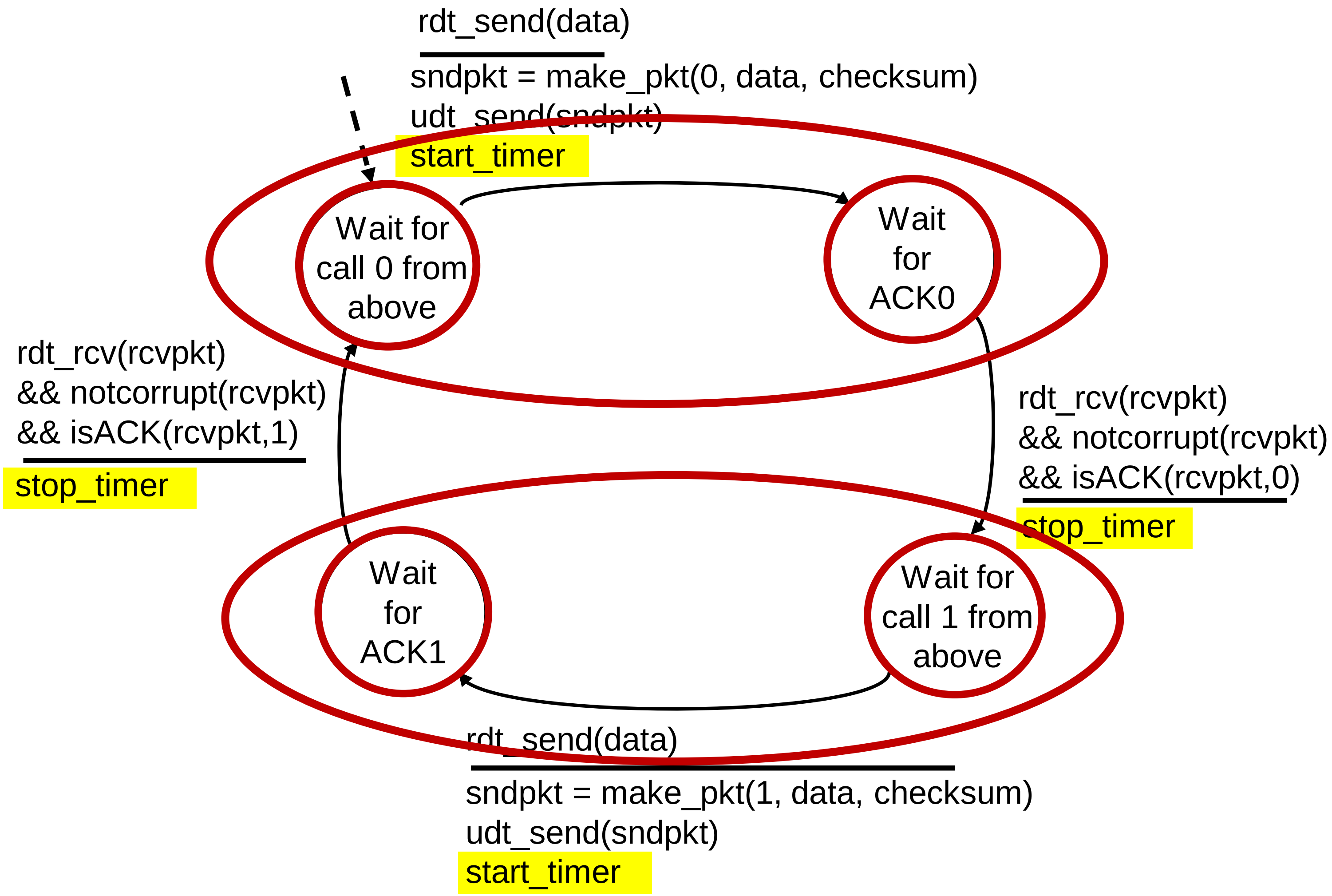
rdt3.0: channels with errors *and* loss

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

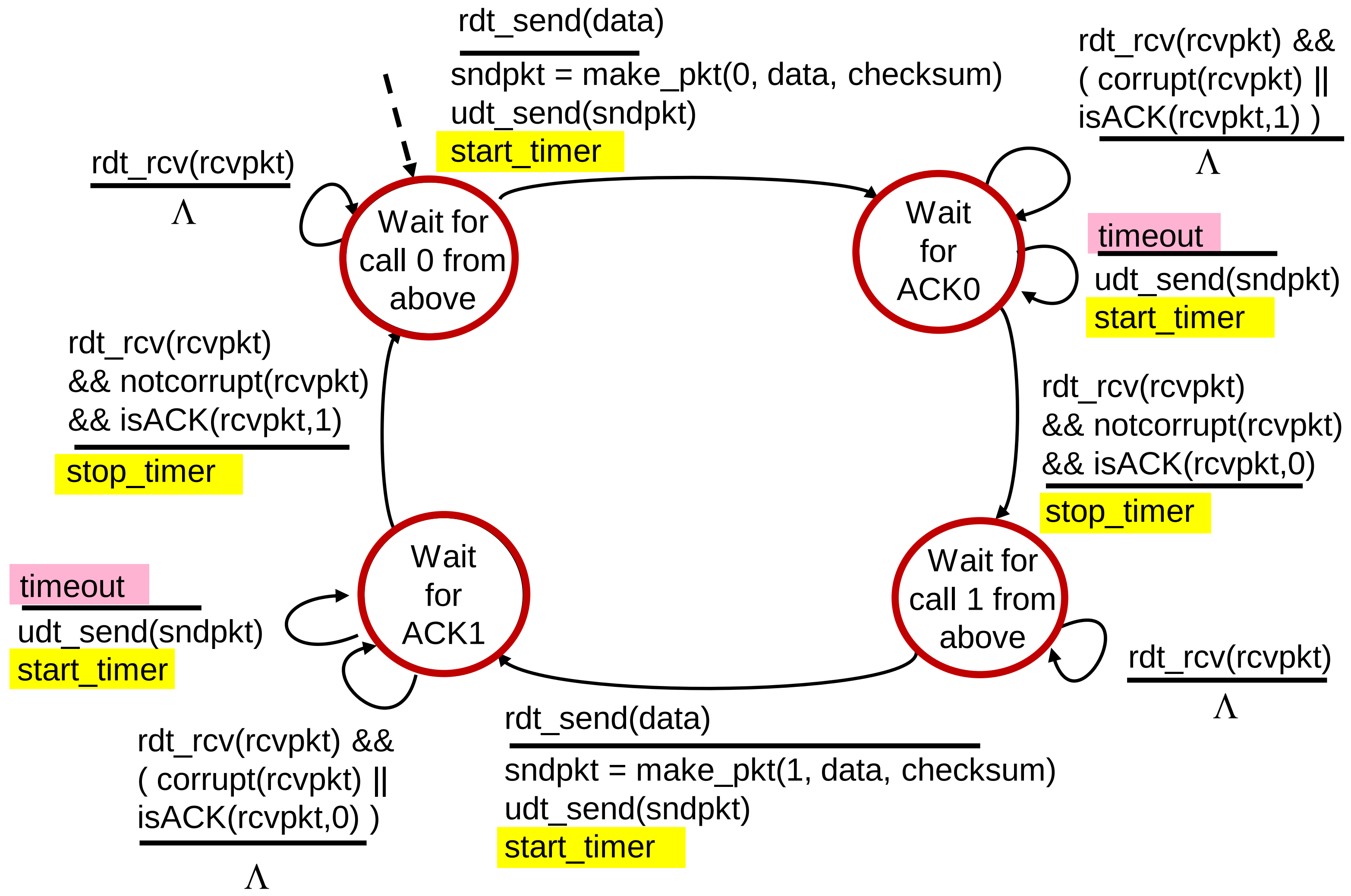
- retransmits if no ACK received in this time
- if pkt (or ACK) just delayed (not lost):
  - retransmission will be duplicate, but seq #s already handles this!
  - receiver must specify seq # of packet being ACKed
- use countdown timer to interrupt after "reasonable" amount of time

timeout

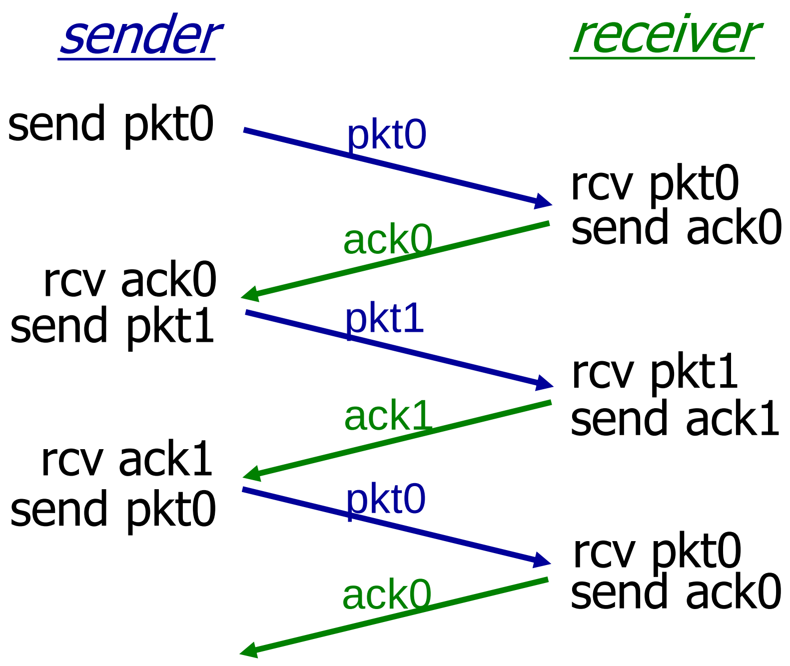
rdt3.0 sender



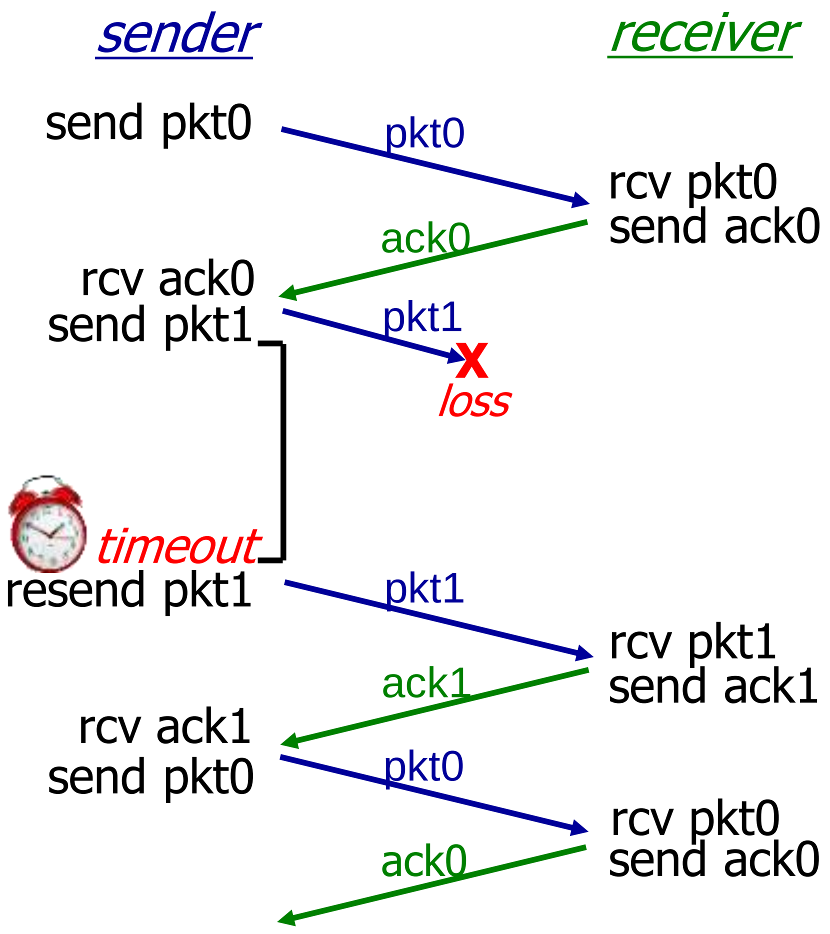
rdt3.0 sender



rdt3.0 in action

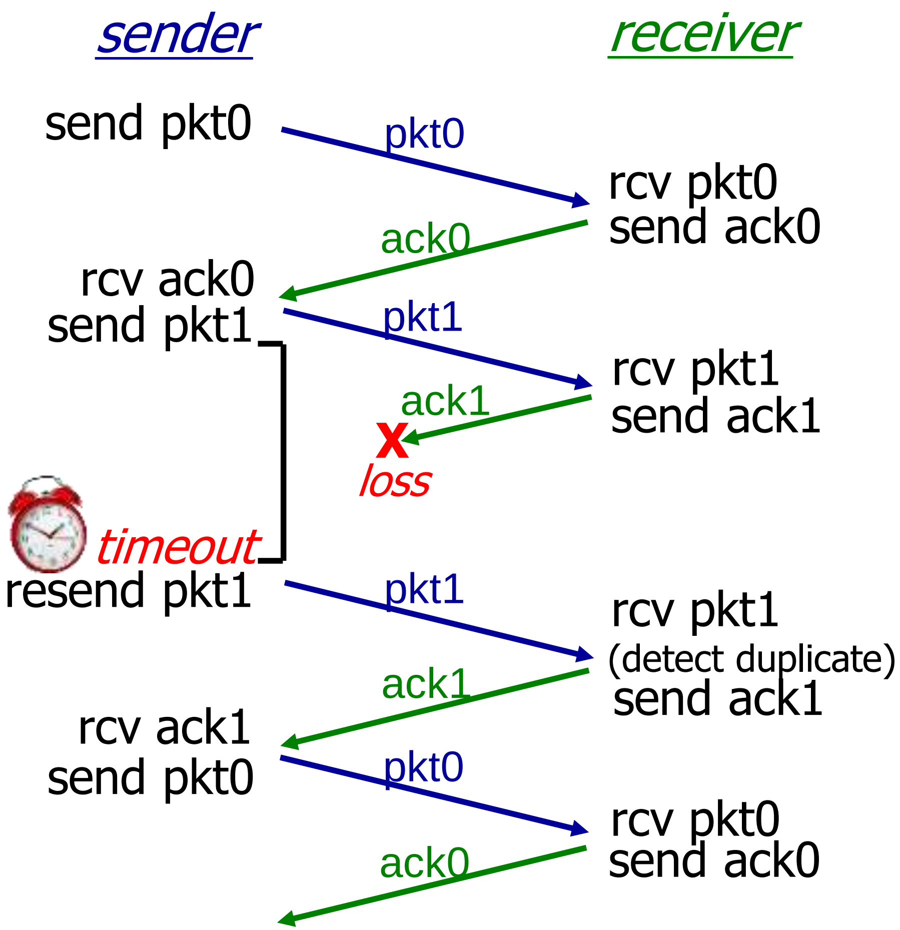


(a) no loss

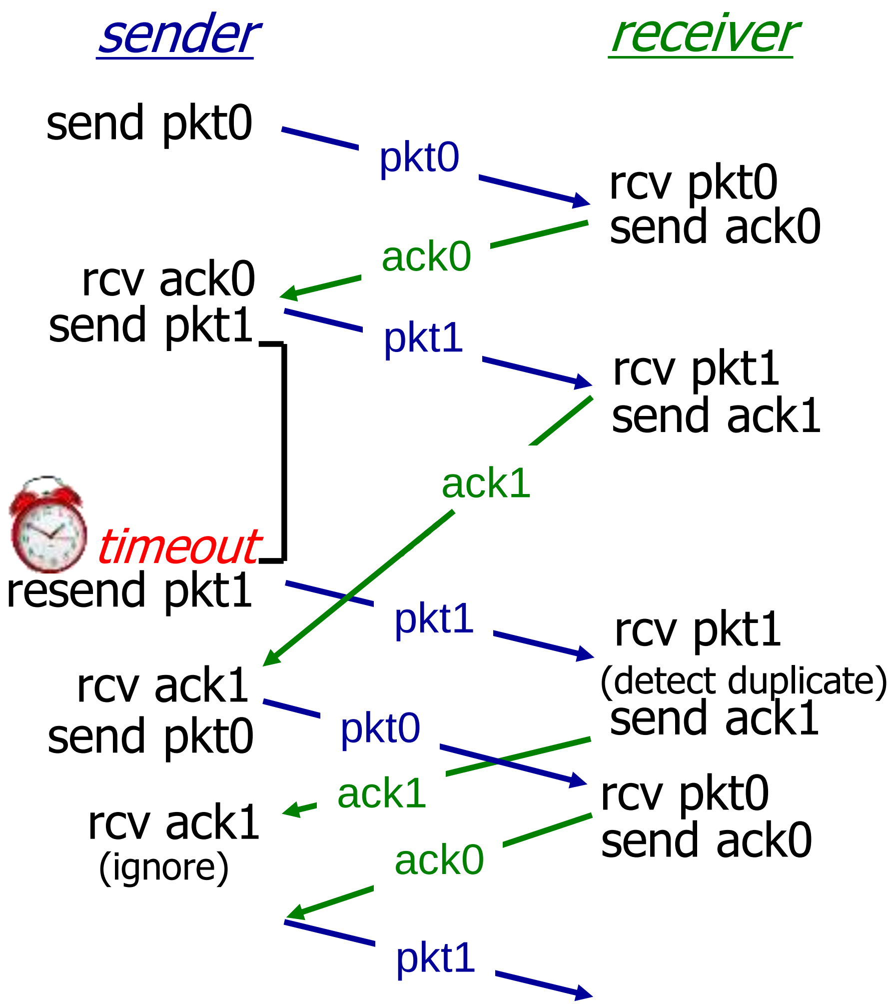


(b) packet loss

rdt3.0 in action



(c) ACK loss



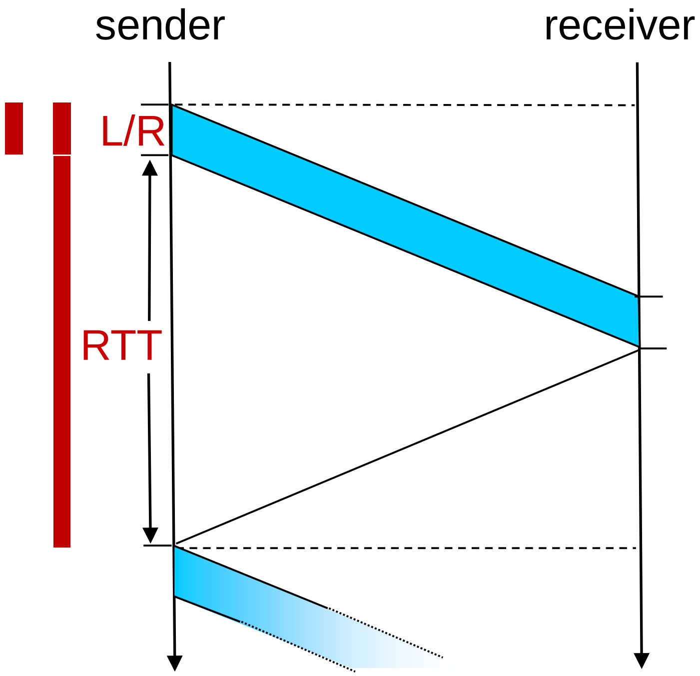
(d) premature timeout/ delayed ACK

rdt3.0: stop-and-wait operation

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

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

$$= 0.00027$$

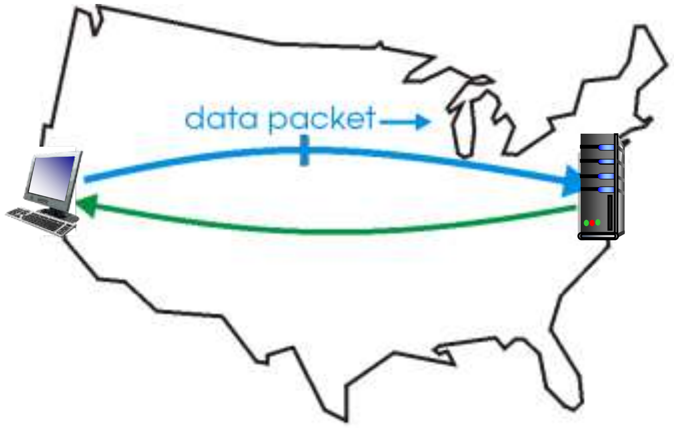


- rdt 3.0 protocol performance stinks!
- Protocol limits performance of underlying infrastructure (channel)

rdt3.0: pipelined protocols operation

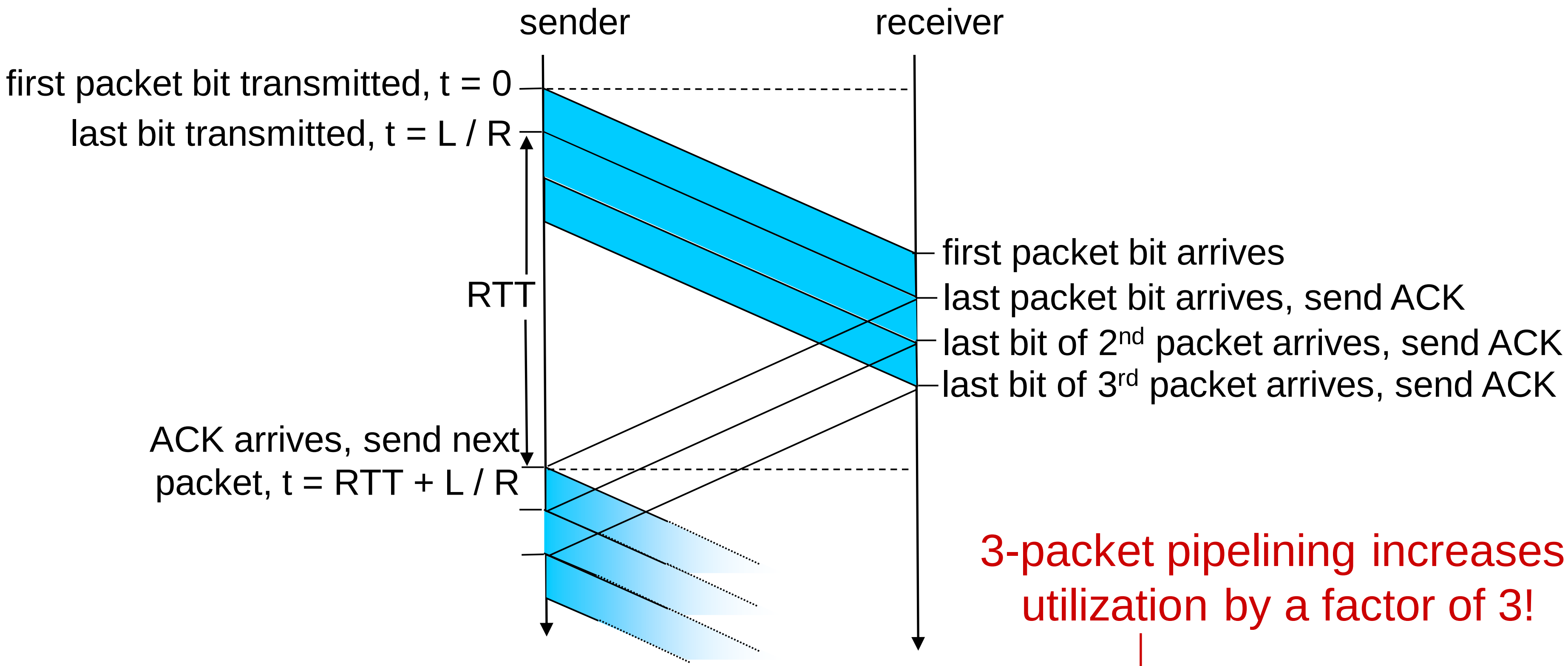
pipelining: sender allows multiple, “in-flight”, yet-to-be-acknowledged packets

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



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

Pipelining: increased utilization



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

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

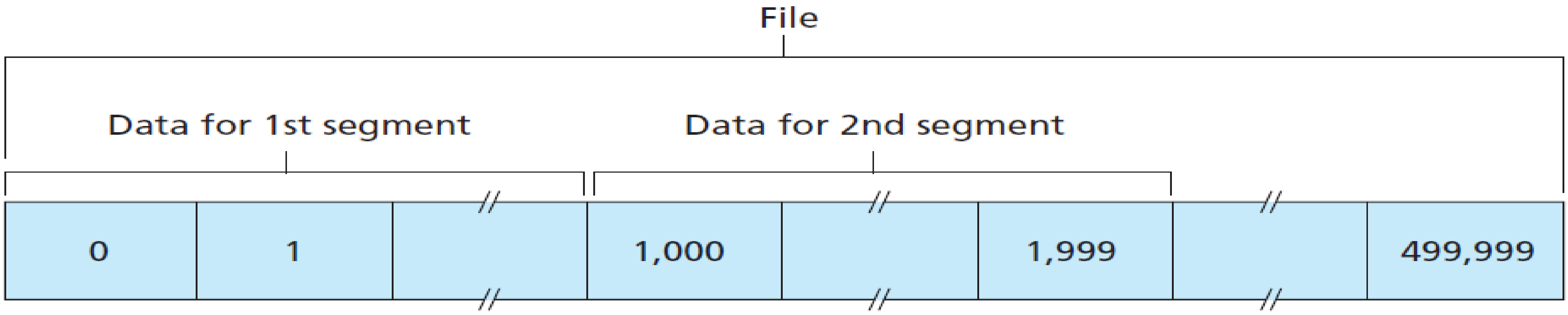
TCP: overview RFCs: 793,1122, 2018, 5681, 7323

- 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

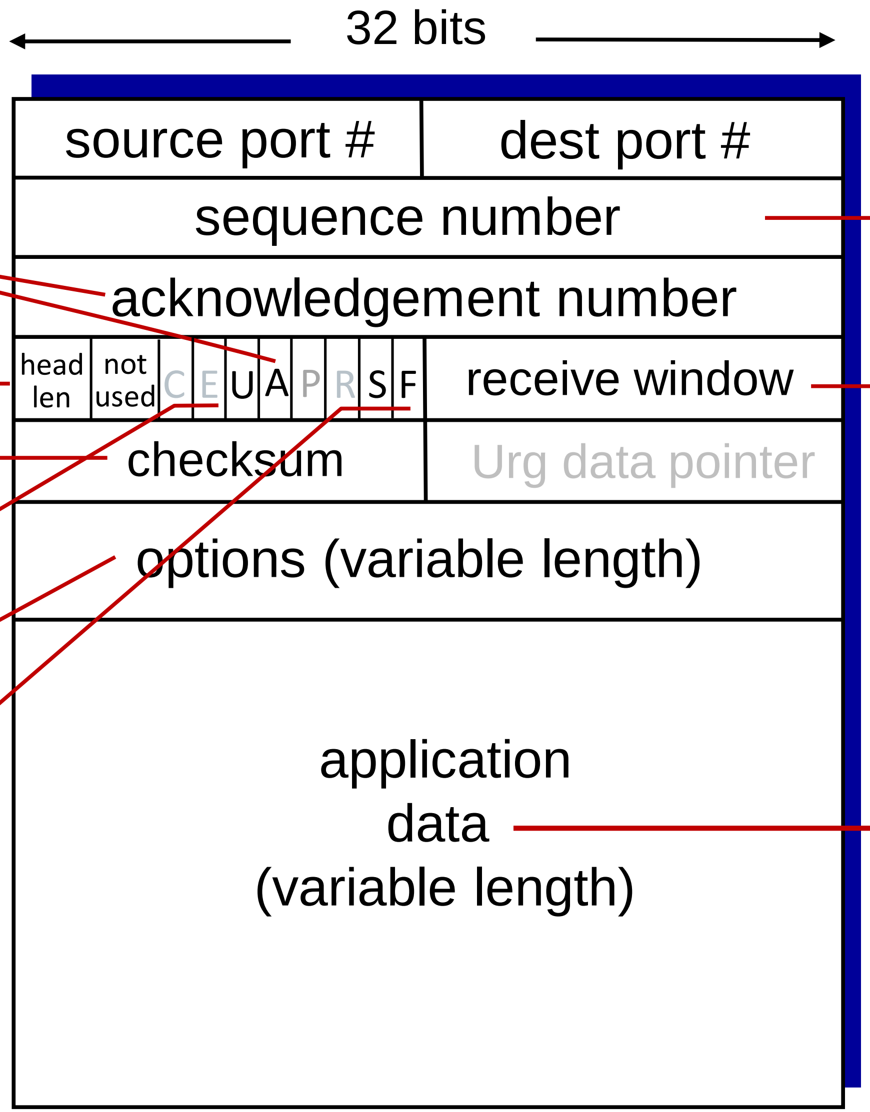
  MSS: maximum amount of application layer data in the segment.
- cumulative ACKs
- pipelining:
  - TCP congestion and flow control set window size
- connection-oriented:
  - handshaking (exchange of control messages) initializes sender, receiver state before data exchange
- flow controlled:
  - sender will not overwhelm receiver

TCP Seq numbers and Acks

- Suppose Host A wants to send a stream of data to a process in Host B over a TCP connection. Assume that, the data stream consists of a file consisting of 500,000 bytes, and that the MSS is 1,000 bytes. Then the segment looks like this:



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:

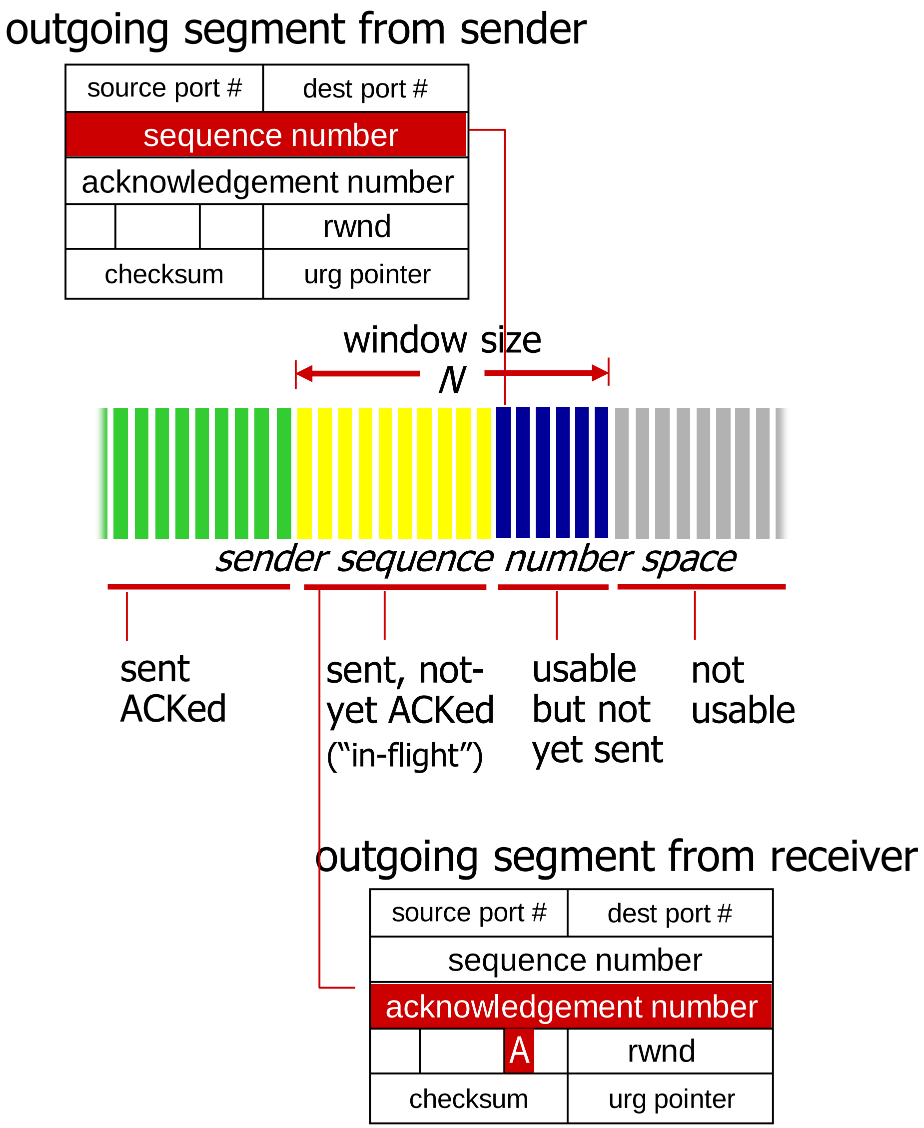
- byte stream "number" of first byte in segment's data

Acknowledgements:

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

Q: how the receiver handles out-of-order segments

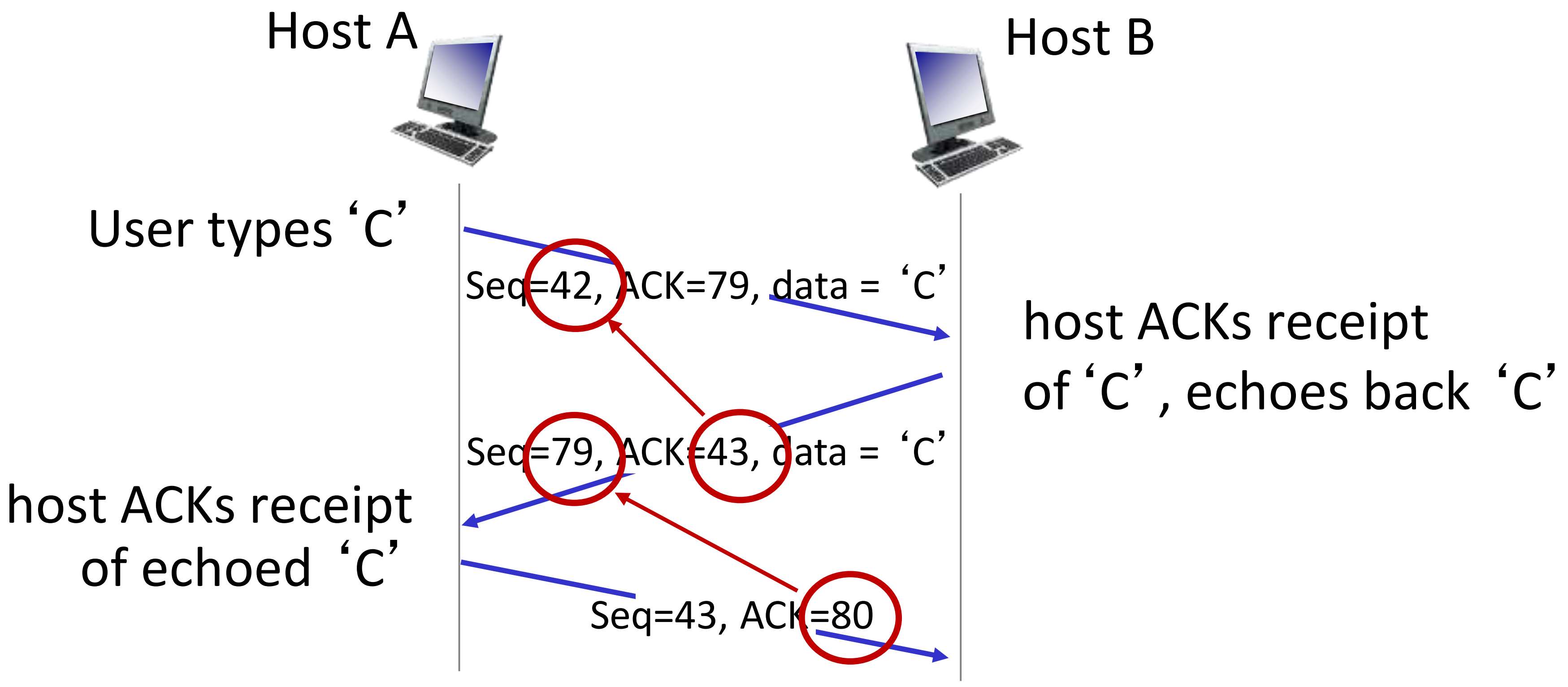
- *A:* Discards out-of-order segments
- *B:* Keeps the out-of-order bytes in the buffer
- *C:* TCP spec doesn't say, - up to the implementor



TCP sequence numbers, ACKs (Some Scenarios)

- Host A has received all data from 0 to 535 and Host A is expecting data 536 and all subsequent byte streams from B.
- Host A received one segment from 0 through 535 and another segment from 900 to 1000.
  - It has not received any segment from 536 to 899. Therefore, there has a gap
  - Cumulative acknowledgments.
- Host A received the segment from 900 to 1000 before receiving bytes 536 to 899. Therefore, out of order.

TCP sequence numbers, ACKs



simple telnet scenario

TCP round trip time, timeout

Q: how to set TCP timeout value?

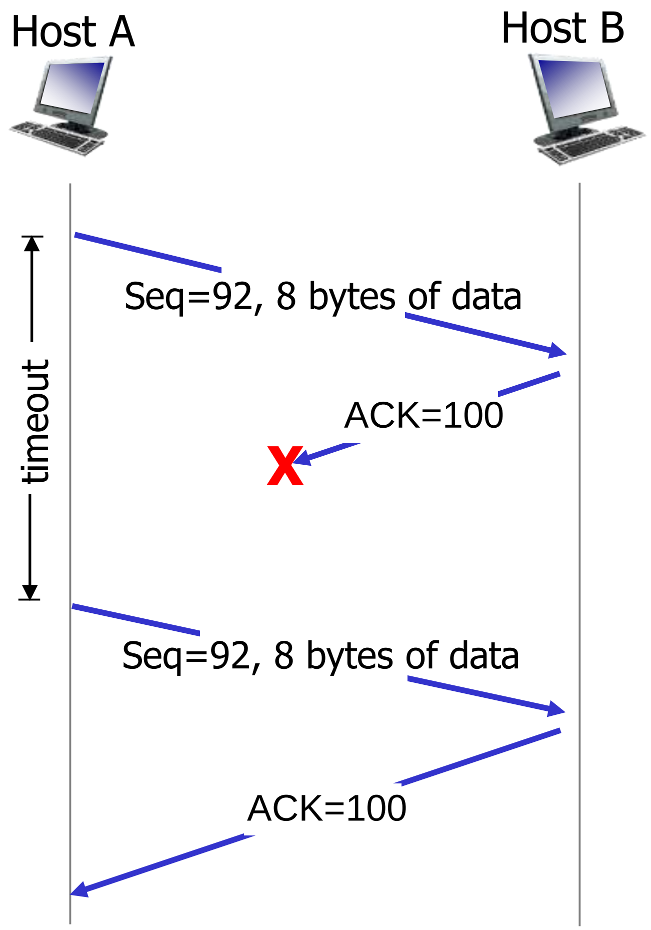
- longer than RTT, but RTT varies!
- *too short:* premature timeout, unnecessary retransmissions
- *too long:* slow reaction to segment loss

Q: how to estimate RTT?

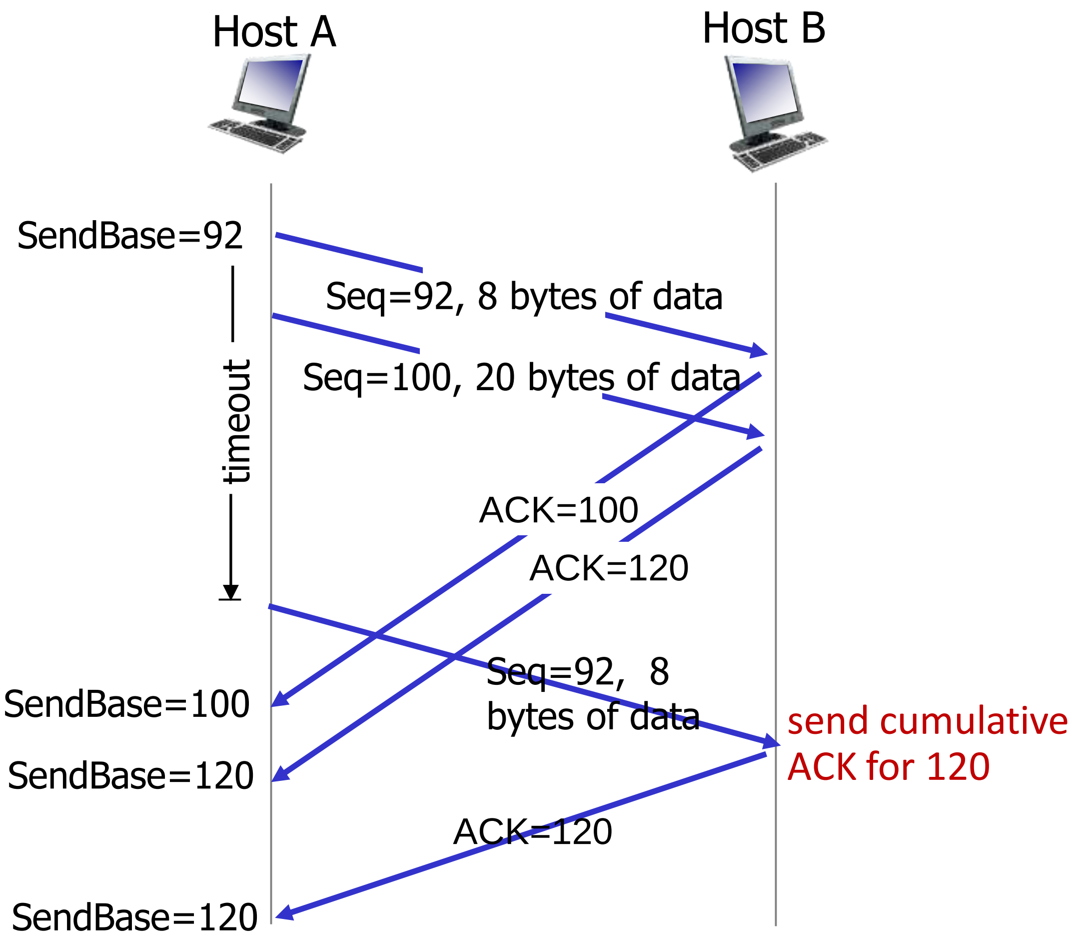
- `SampleRTT`: measured time from segment transmission until ACK receipt
  - ignore retransmissions
- `SampleRTT` will vary, want estimated RTT "smoother"
  - average several *recent* measurements, not just current `SampleRTT`

SampleRTT values will fluctuate from segment to segment due to congestion and load on the end systems.

TCP: retransmission scenarios

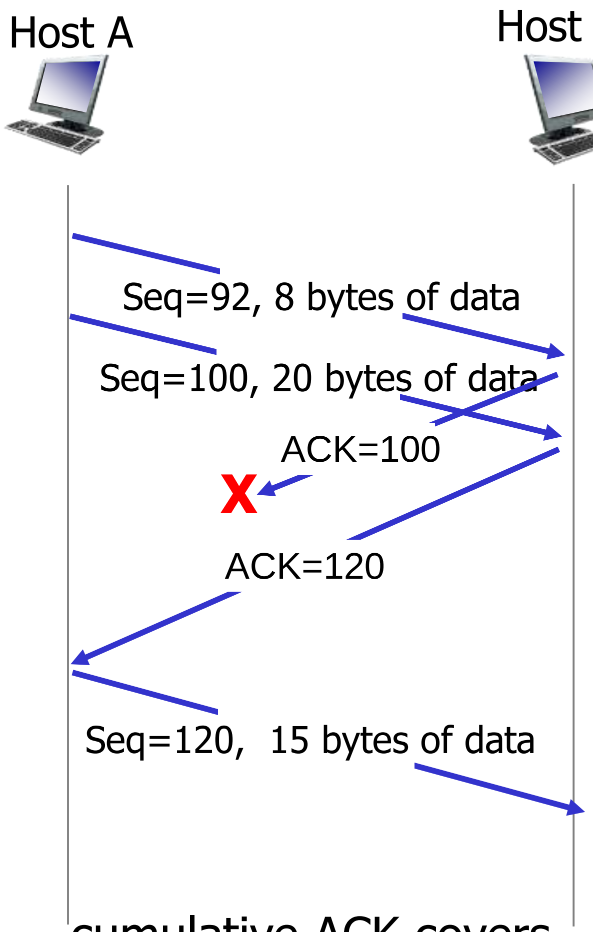


lost ACK scenario



premature timeout

TCP: retransmission scenarios



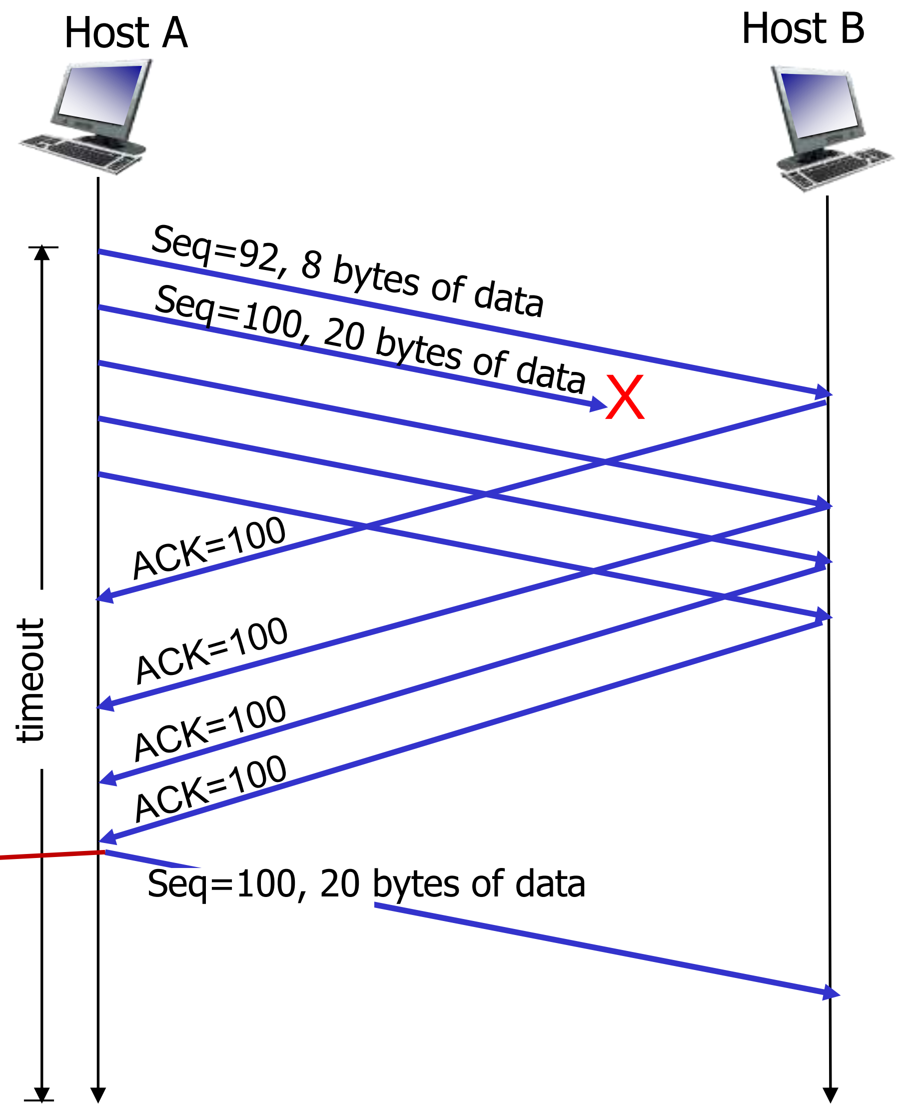
cumulative ACK covers
for earlier lost ACK

TCP fast retransmit

TCP fast retransmit

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

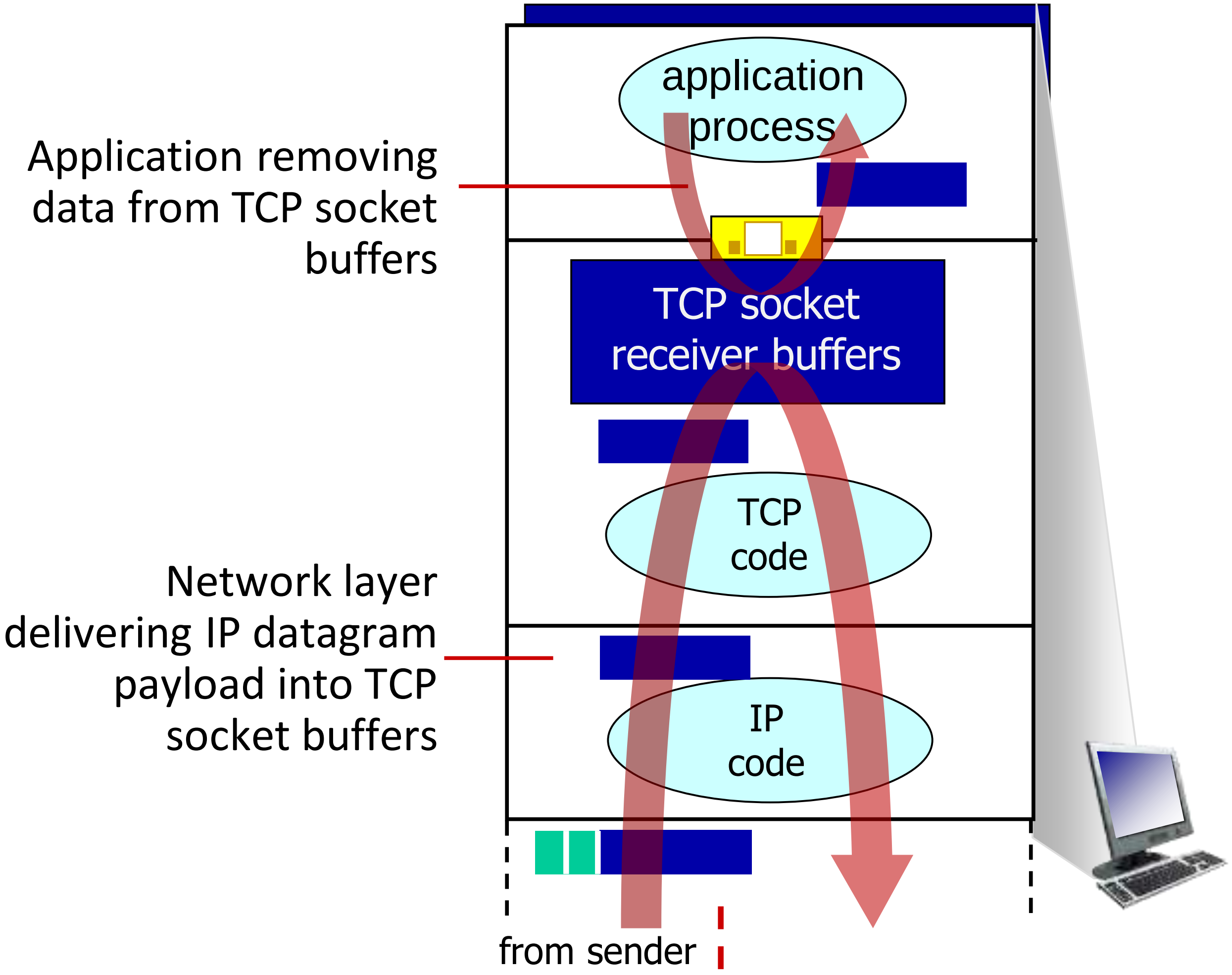
- likely that unACKed segment lost, so don’t wait for timeout



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

TCP flow control

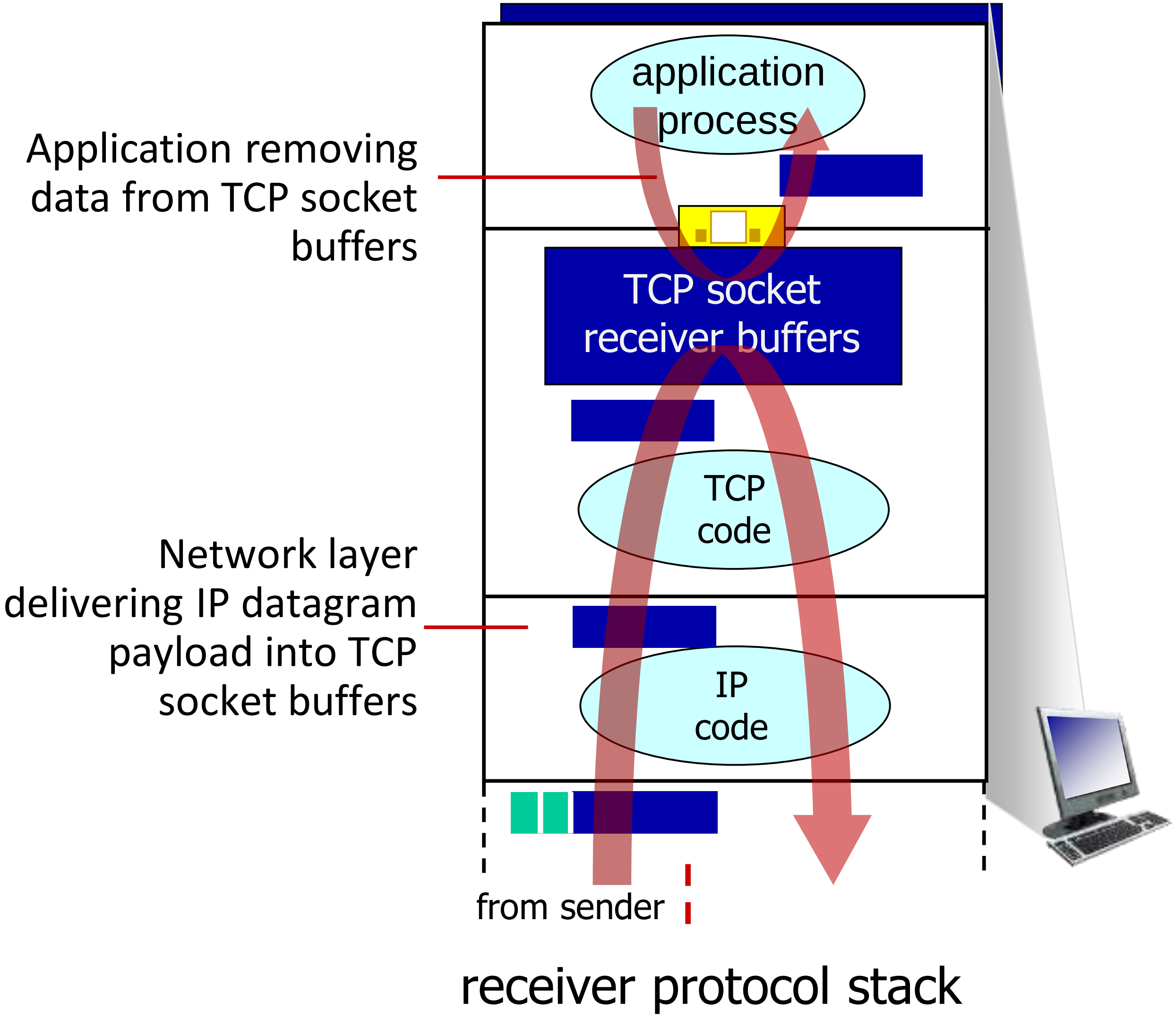
Q: What happens if network layer delivers data faster than application layer removes data from socket buffers?



receiver protocol stack

TCP flow control

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



TCP flow control

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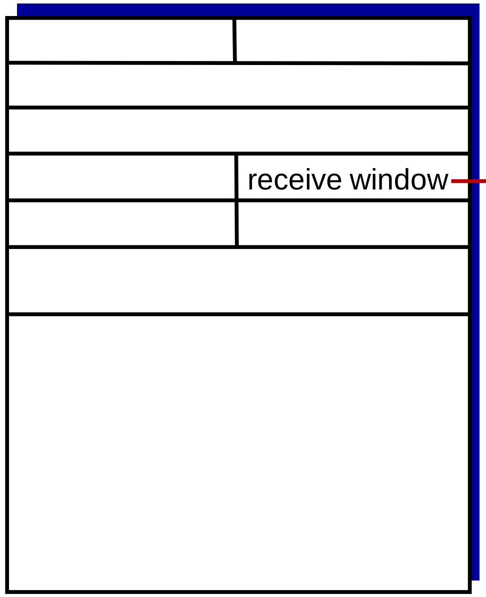
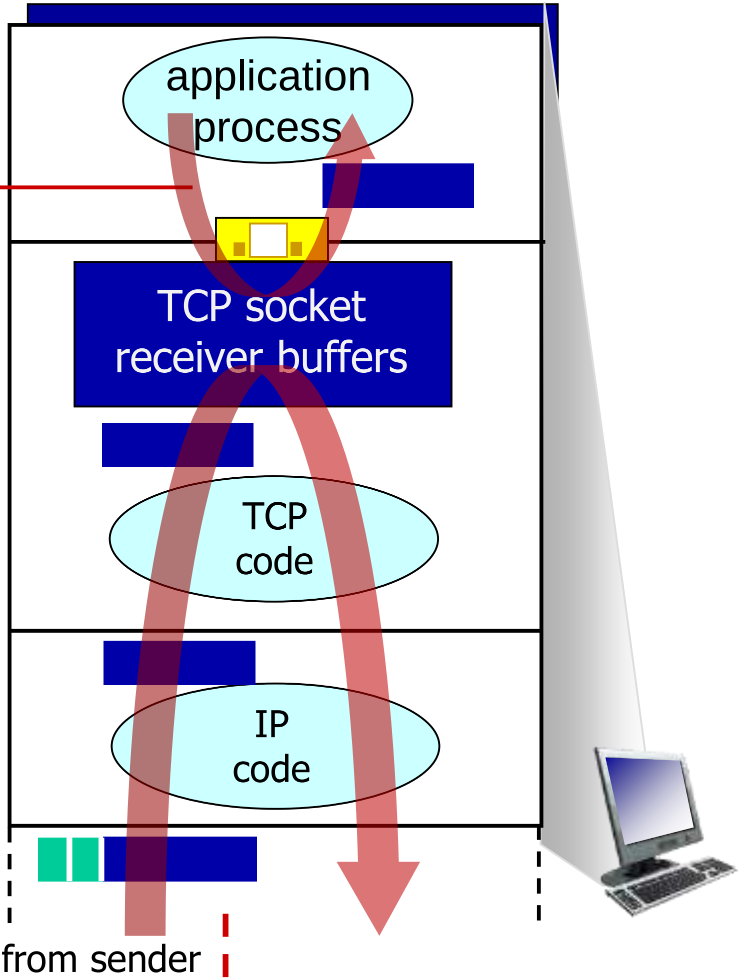


flow control: # bytes receiver willing to accept

Application removing data from TCP socket buffers



receiver protocol stack

TCP flow control

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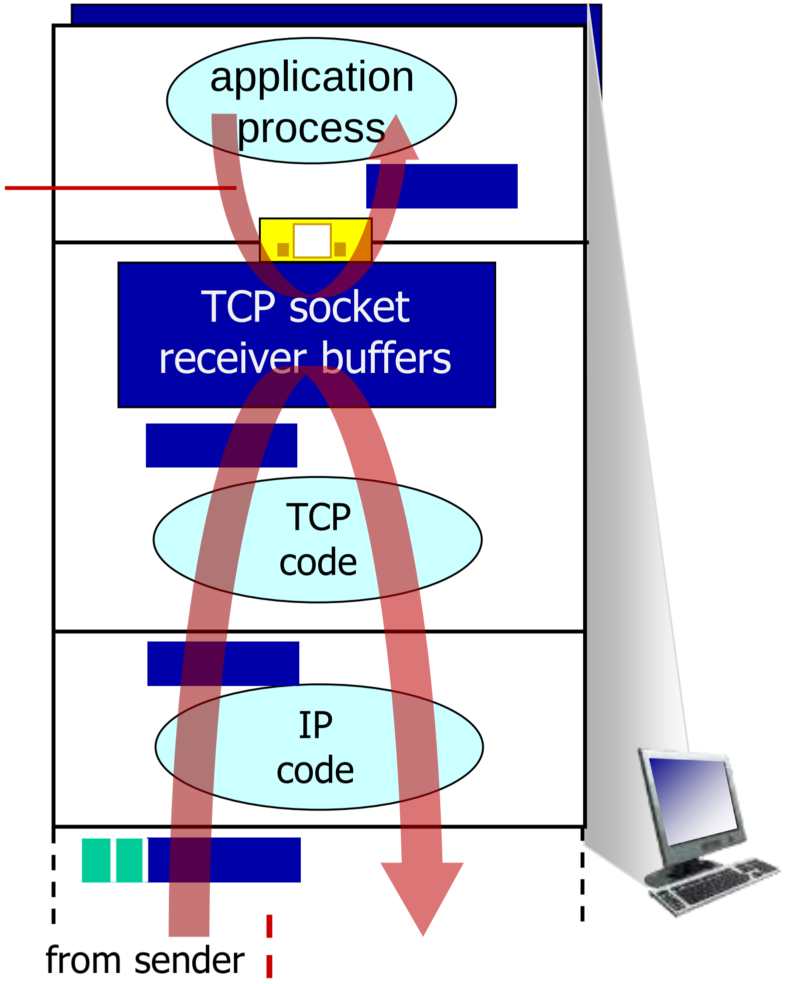
Application removing data from TCP socket buffers



receiver protocol stack

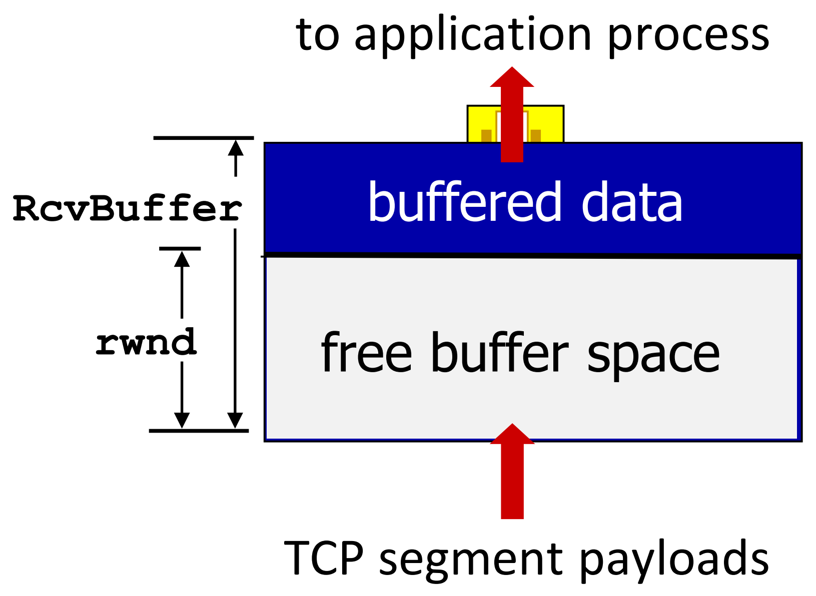
flow control

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

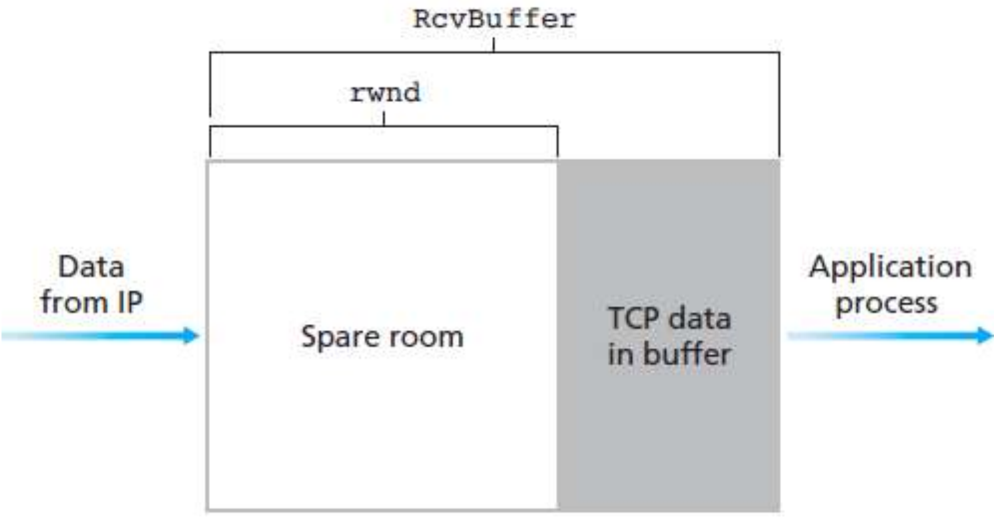
```
LastByteSent – LastByteAcked <= rwnd
```



TCP receiver-side buffering

TCP flow control

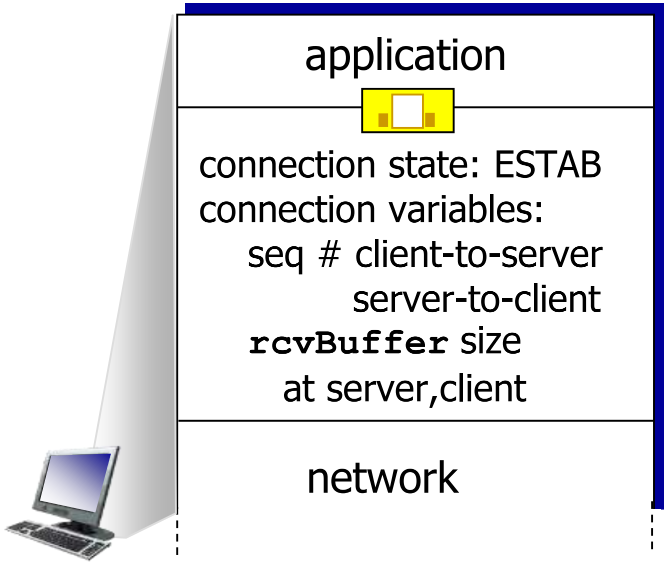
- LastByteRead: data stream read from the buffer
- LastByteRcvd: the data stream that has arrived from the network
- LastByteRcvd – LastByteRead <= RcvBuffer
- rwnd = RcvBuffer – [LastByteRcvd – LastByteRead]
- Initially rwnd = RcvBuffer
- LastByteSent – LastByteAcked ≤ rwnd
- Host A to continue to send segments with one data byte when B's receive window is zero. These segments will be acknowledged by the receiver



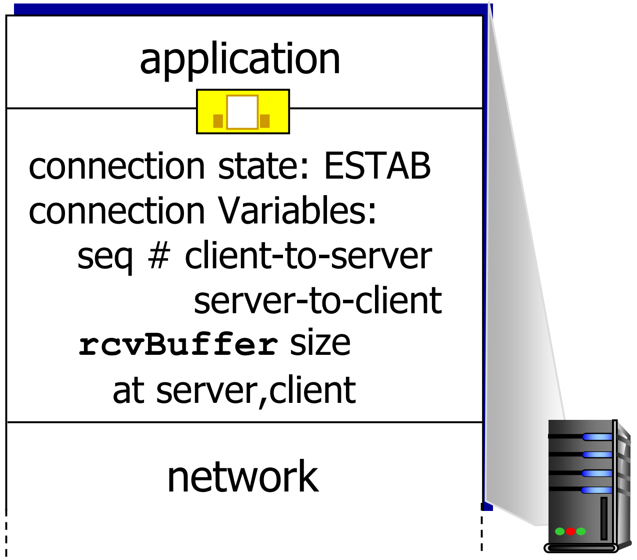
TCP connection management

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();
```

TCP 3-way handshake

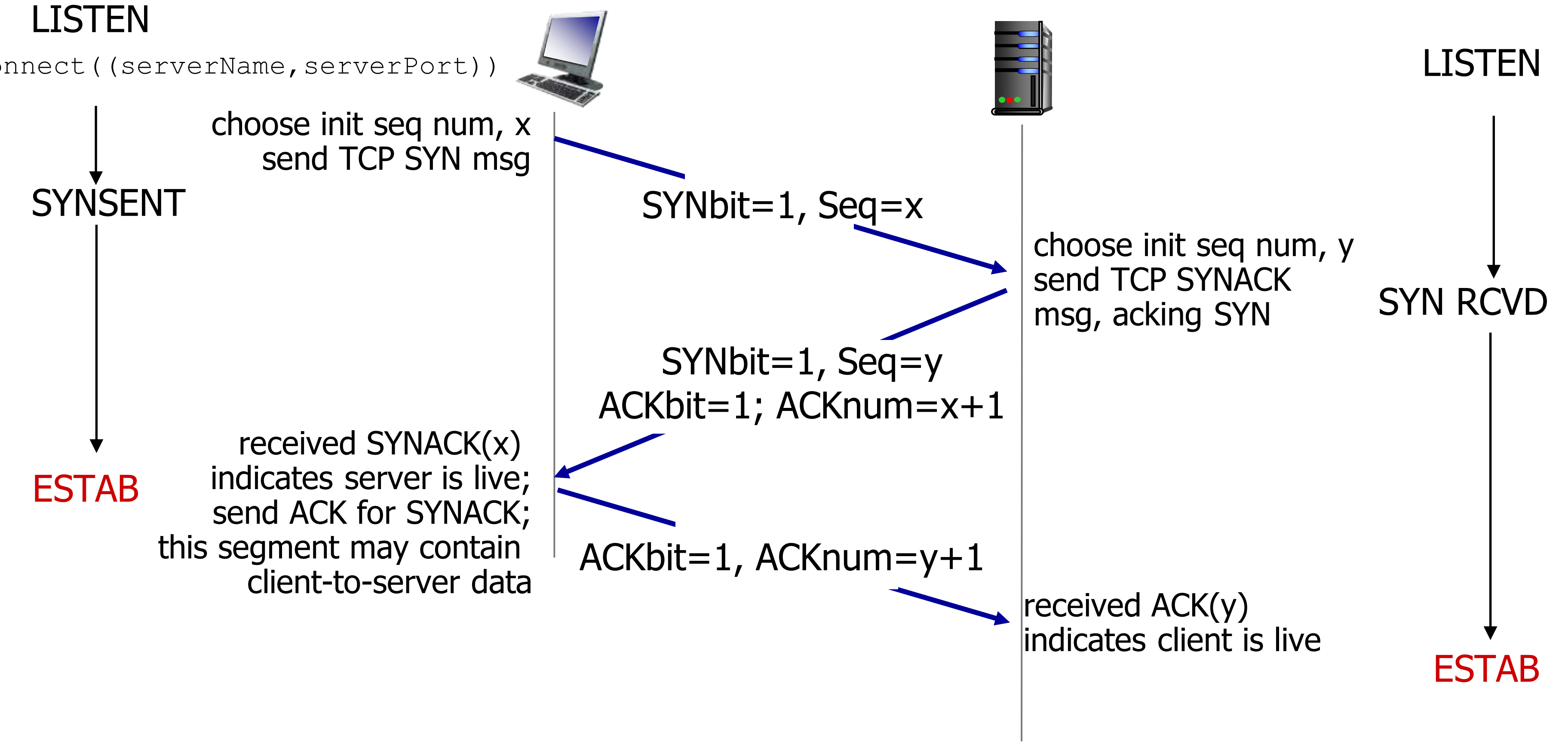
Client state

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

Server state

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

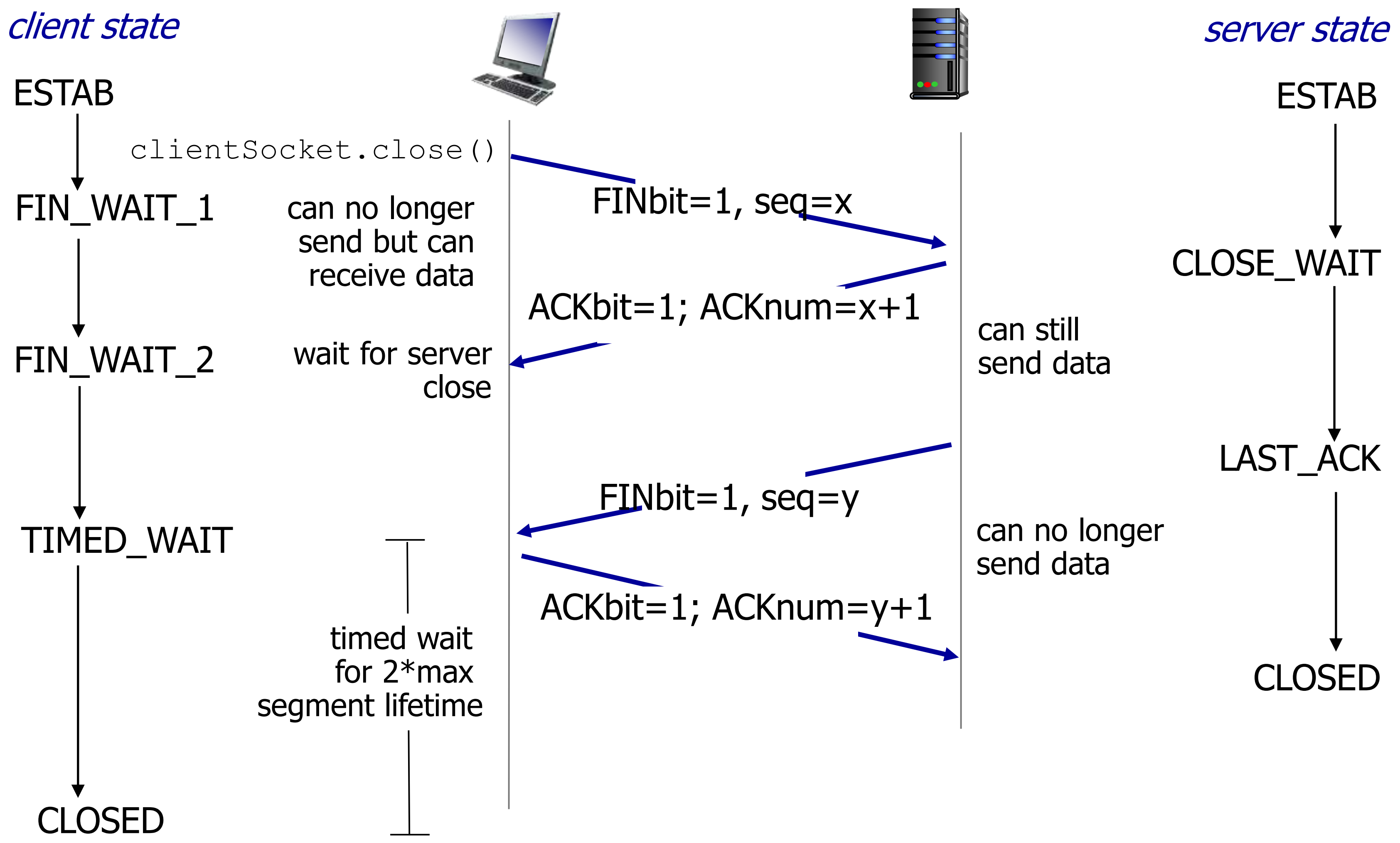
```
clientSocket.connect((serverName,serverPort))
```



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

TCP: closing a connection



TCP: closing a connection

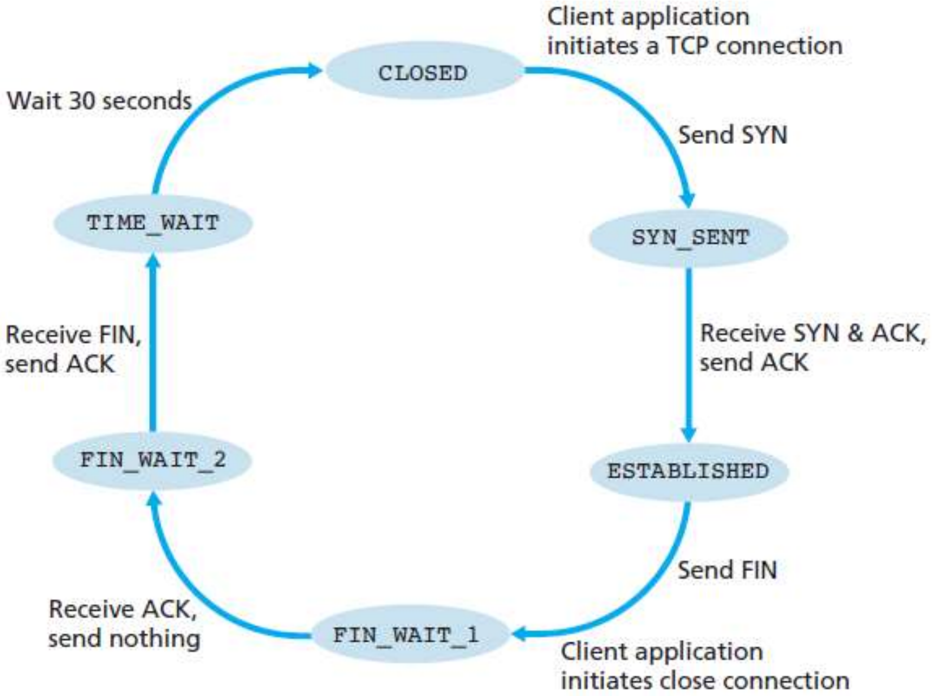


Fig: A typical sequence of TCP states visited by a client TCP

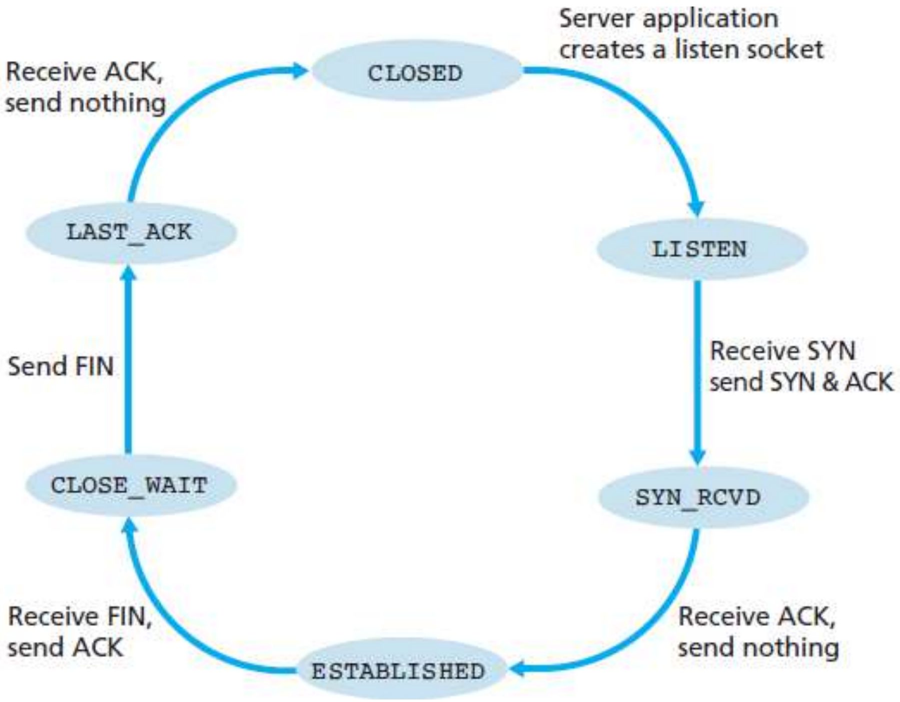


Fig: A typical sequence of TCP states visited by a server TCP

TCP congestion control: AIMD

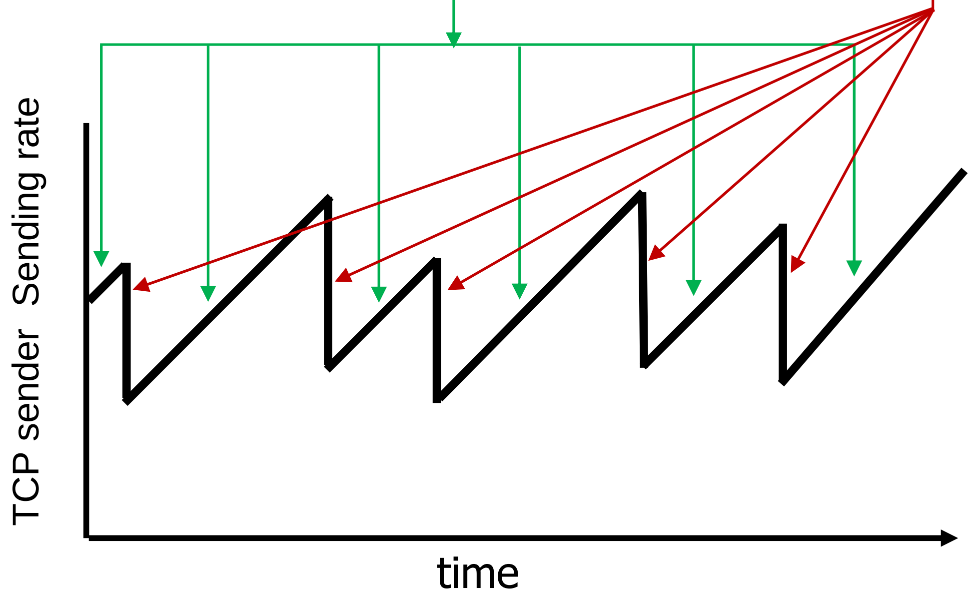
- *approach:* senders can increase sending rate until packet loss (congestion) occurs, then decrease sending rate on loss event

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

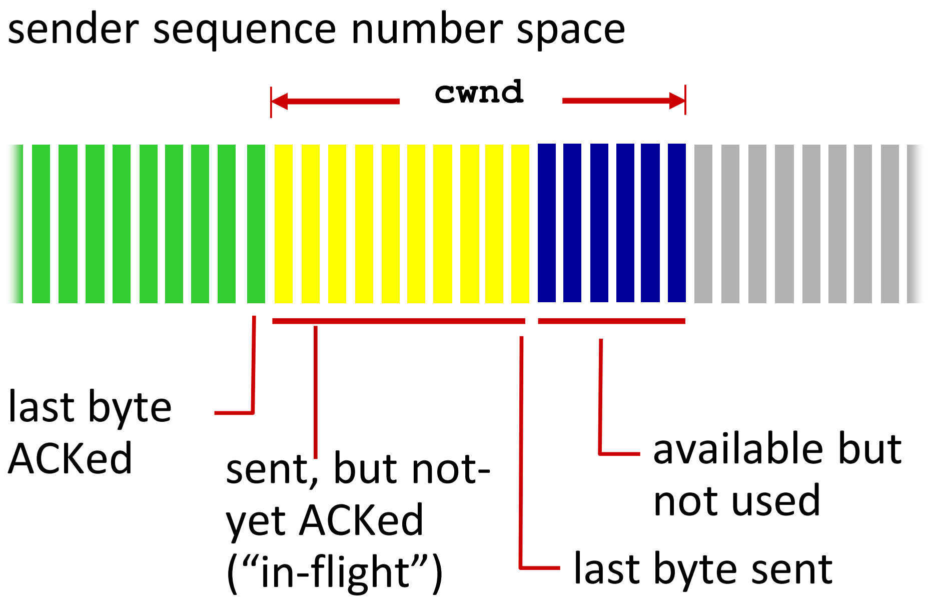
Multiplicative decrease detail: sending rate is

- 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:

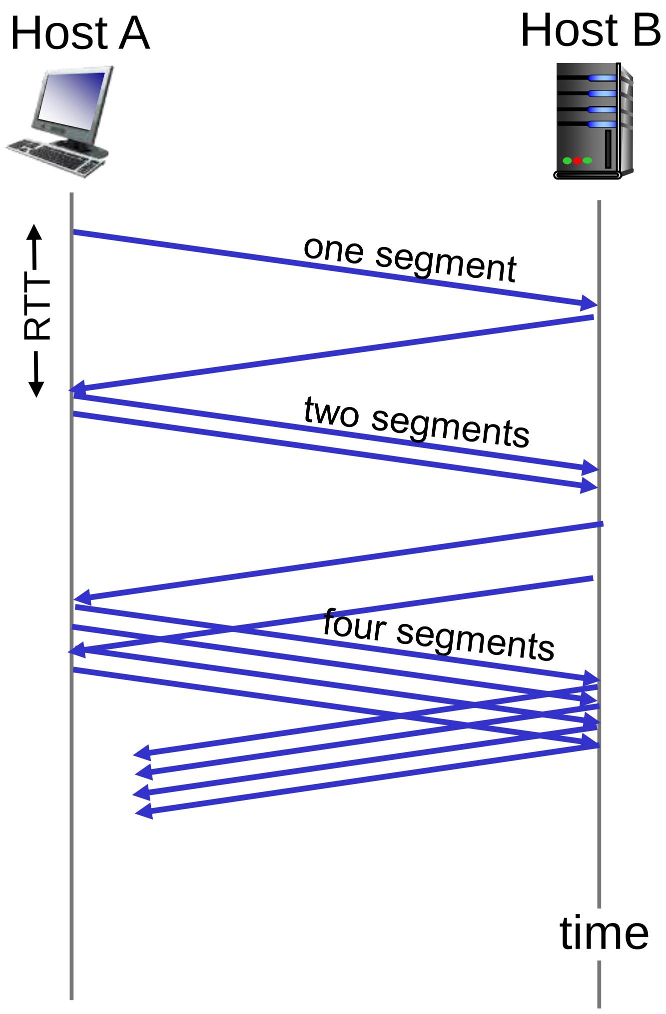
- *roughly:* send `cwnd` bytes, wait RTT for ACKS, then send more bytes

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

- 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 Slow Start

- loss indicated by timeout:
  - **`cwnd`** set to 1 MSS;
  - window then grows exponentially (as in slow start) to threshold, then grows linearly
  - Set the threshold value ssthresh is equal to cwnd/2
- When the value of cwnd >= ssthresh, Slow Start ends and Congestion Avoidance (CA) starts.
- loss indicated by 3 duplicate ACKs: TCP enters in the fast recovery mode.

TCP: Congestion Avoidance (CA)

- Rather than doubling the cwnd value, cwnd is increased by just a single MSS every RTT.
- TCP sender increase cwnd by MSS bytes (MSS/cwnd)
- **When the congestion avoidance ends?**
  - Depends on the timeout events and triple duplicates
  - dup ACKs indicate network capable of delivering some segments
- Fast Recovery: 3 dup ACKs
  - TCP Tahoe always sets **`cwnd`** to 1 then grows exponentially (timeout or 3 duplicate acks) [Earlier Style]
  - **`TCP Reno cut the cwnd`** in half window then grows linearly [New Version]

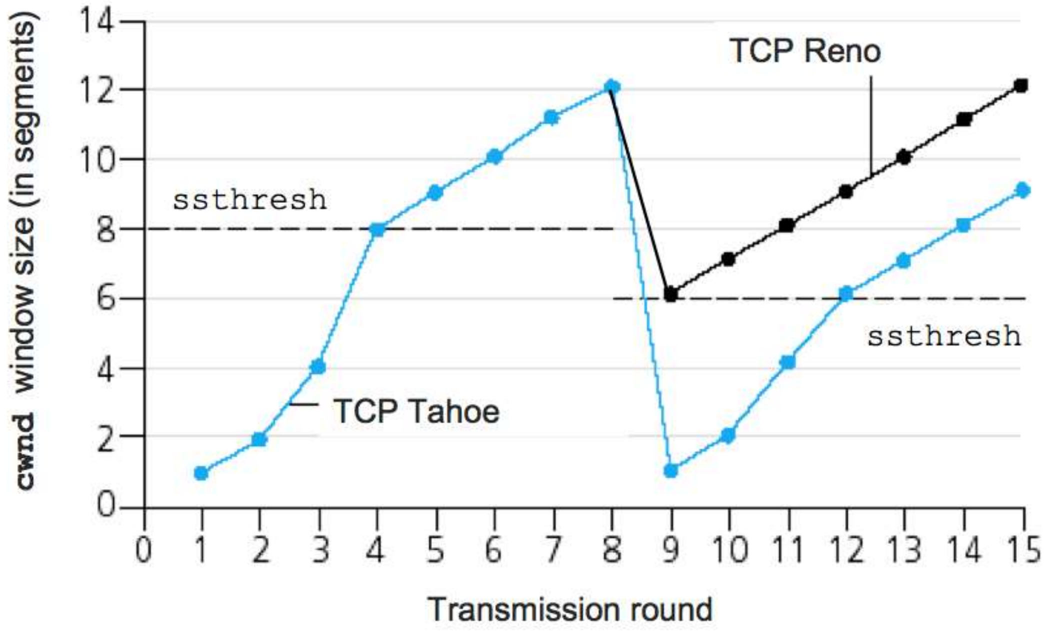
TCP: from slow start to congestion avoidance

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



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

Summary: TCP Congestion Control

- When CongWin is below Threshold, sender in slow start phase, window grows exponentially.
- When CongWin is above Threshold, sender is in congestion avoidance phase, window grows linearly.
- When a triple duplicate ACK occurs, Threshold set to CongWin/2 and CongWin set to Threshold + 3.
- When timeout occurs, Threshold set to CongWin/2 and CongWin is set to 1 MSS.

Summary: TCP Congestion Control

TCP sender congestion control

State	Event	TCP Sender Action	Commentary
Slow Start (SS)	ACK receipt for previously unacked data	CongWin = CongWin + MSS, If (CongWin > Threshold) set state to "Congestion Avoidance"	Resulting in a doubling of CongWin every RTT
Congestion Avoidance (CA)	ACK receipt for previously unacked data	CongWin = CongWin+MSS * (MSS/CongWin)	Additive increase, resulting in increase of CongWin by 1 MSS every RTT
SS or CA	Loss event detected by triple duplicate ACK	Threshold = CongWin/2, CongWin = Threshold, Set state to "Congestion Avoidance"	Fast recovery, implementing multiplicative decrease. CongWin will not drop below 1 MSS.
SS or CA	Timeout	Threshold = CongWin/2, CongWin = 1 MSS, Set state to "Slow Start"	Enter slow start
SS or CA	Duplicate ACK	Increment duplicate ACK count for segment being acked	CongWin and Threshold not changed