Computer Communication Networks

Transport

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Where to find in book?

- Materials covered in this section are in Chapter 5 and 6 of "Computer Networks: A Systems Approach", Larry Peterson and Bruce Davie, Elsevier

End-to-end Protocols

- Common properties that a *transport protocol* can be expected to *provide*
  - Guarantees message delivery
  - Delivers messages in the same order they were sent
  - Delivers at most one copy of each message
  - Supports arbitrarily large messages
  - Supports synchronization between the sender and the receiver
  - Allows the receiver to apply flow control to the sender
  - Supports multiple application processes on each host

End-to-end Protocols

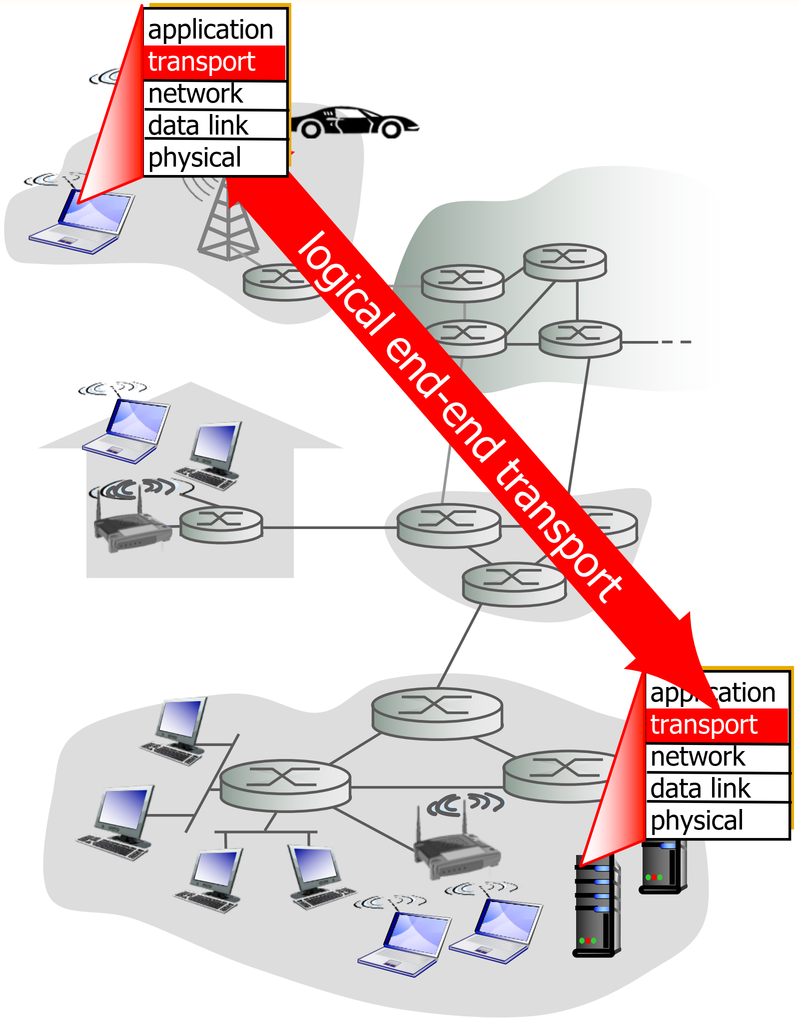
- Typical *limitations of the network* on which transport protocol will operate
  - Drop messages
  - Reorder messages
  - Deliver duplicate copies of a given message
  - Limit messages to some finite size
  - Deliver messages after an arbitrarily long delay

End-to-end Protocols

- *Challenge* for Transport Protocols
  - Develop algorithms that turn the less-than-desirable properties of the underlying network into the high level of service required by application programs

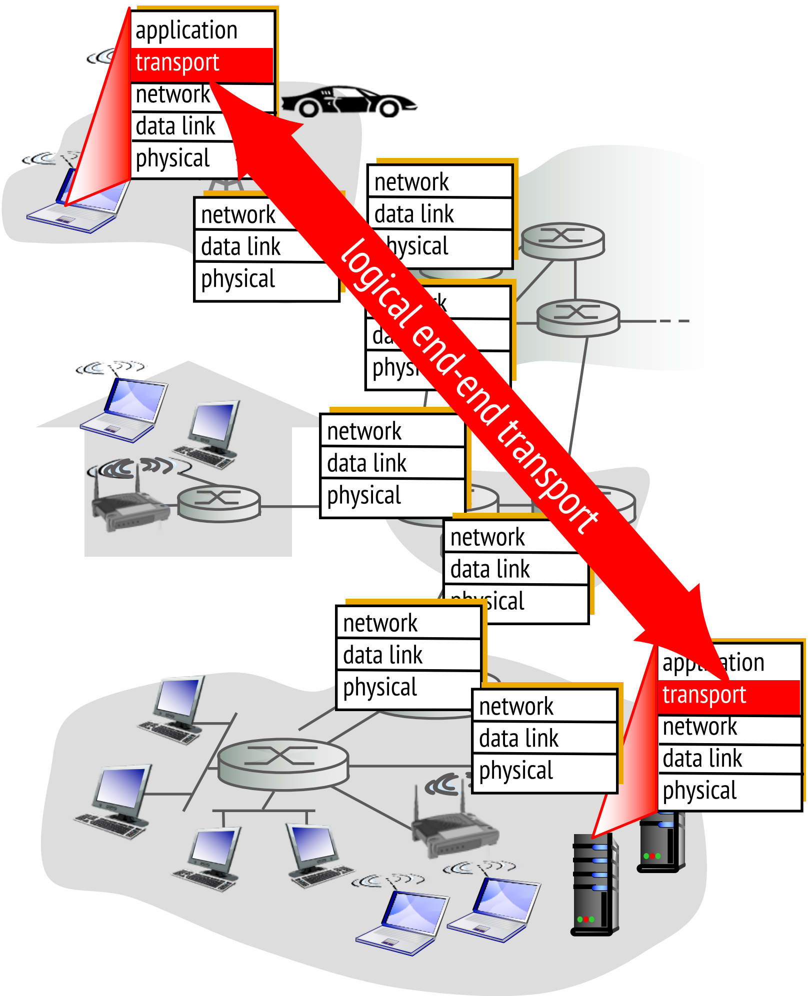
Transport services & protocols

- provide logical communication between app processes running on different hosts
- transport protocols run in end systems
  - send side: breaks app messages into segments, passes to network layer
  - rcv side: reassembles segments into messages, passes to app layer
- more than one transport protocol available to apps
  - Internet: TCP and UDP

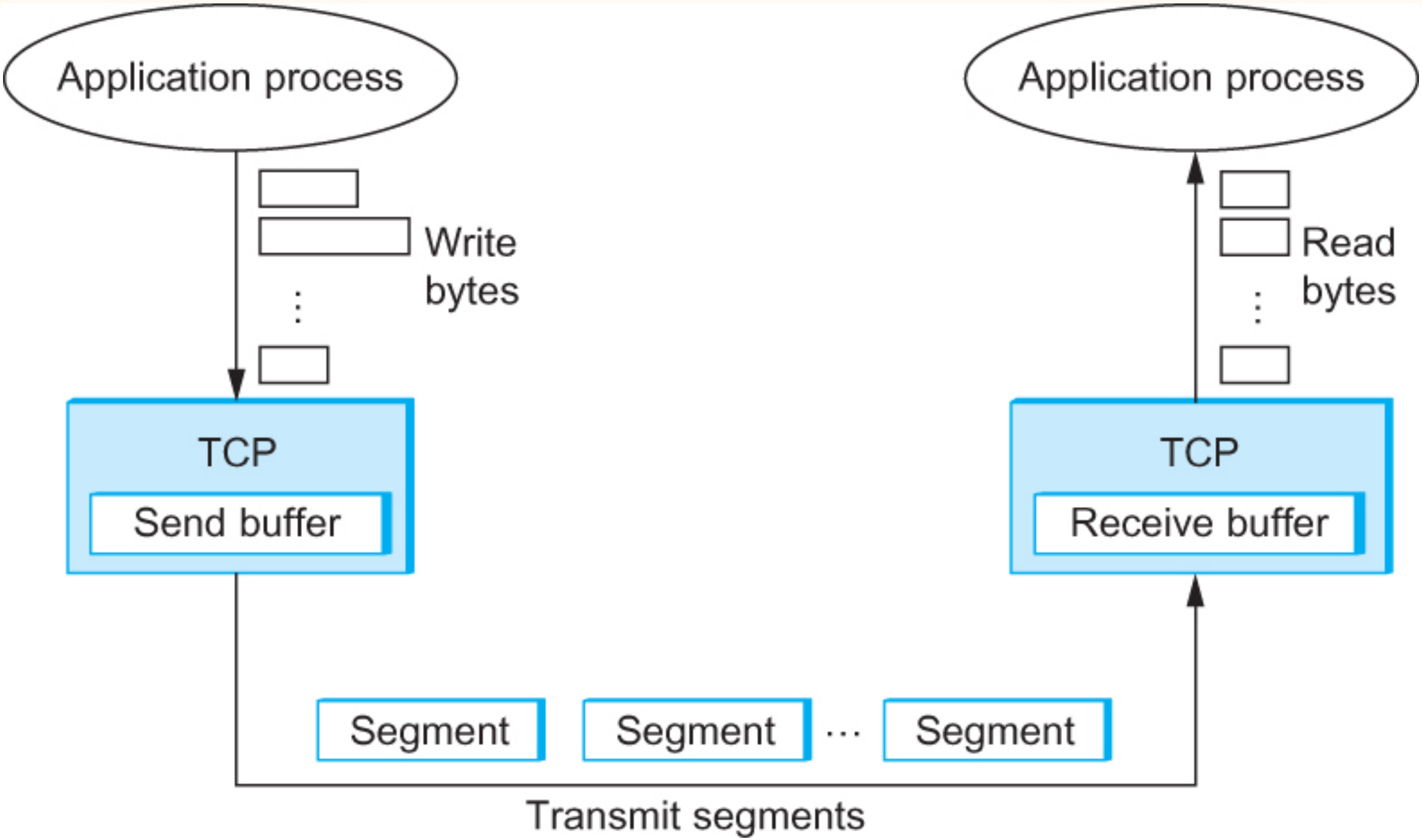


Internet transport-layer protocols

- reliable, in-order delivery (TCP)
  - congestion control
  - flow control
  - connection setup
- unreliable, unordered delivery: UDP
  - no-frills extension of "best-effort" IP
- services not available:
  - delay guarantees
  - bandwidth guarantees



Transport Layer Segmentation

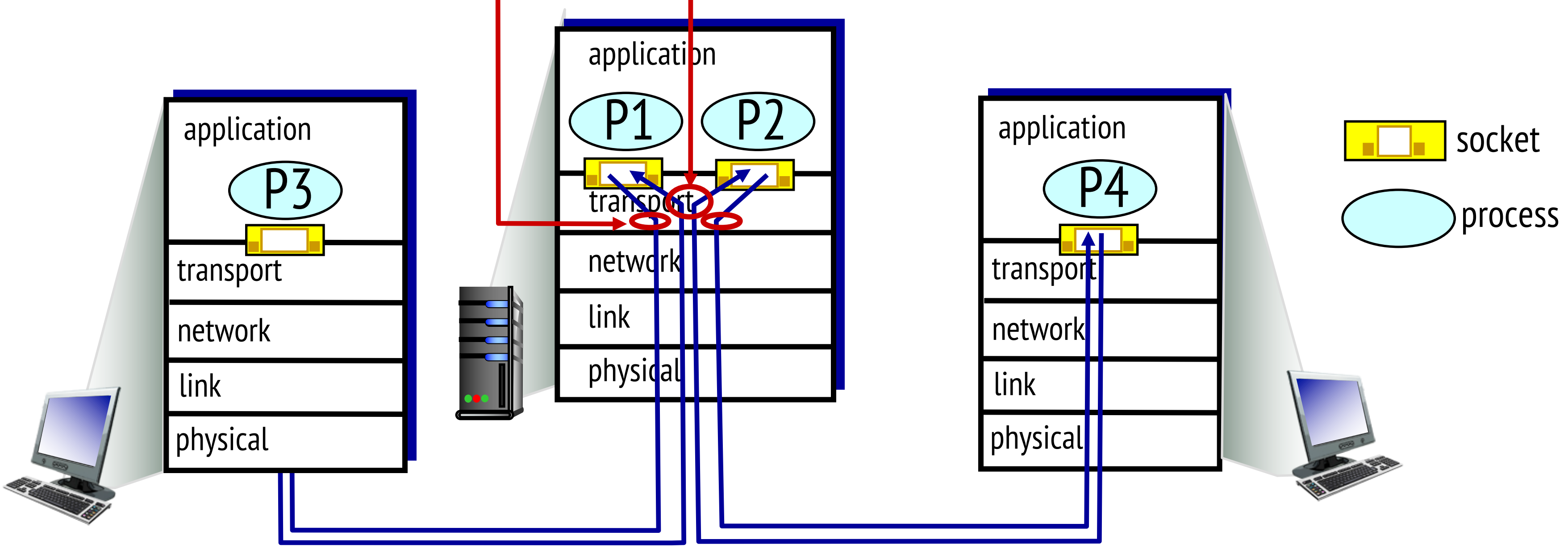


Multiplexing / Demultiplexing

Multiplexing/demultiplexing

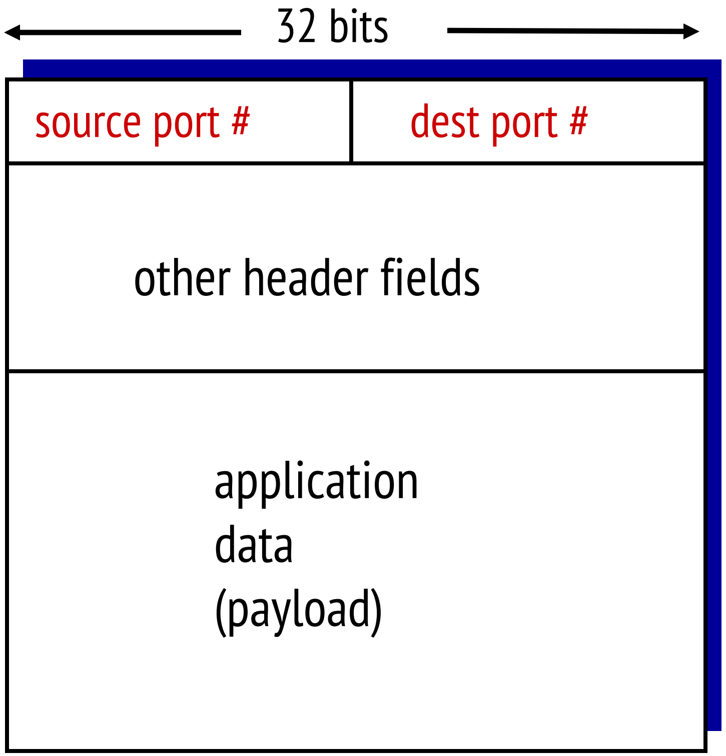
multiplexing at sender:
handle data from multiple sockets, add transport header (later used for demultiplexing)

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



How demultiplexing works

- host receives IP datagrams
  - each datagram has source IP address, destination IP address
  - each datagram carries one transport-layer segment
  - each segment has source, destination port number
- host uses *IP addresses & port numbers* to direct segment to appropriate socket



TCP/UDP segment format

Connectionless demultiplexing

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

serverSocket.bind(('', serverPort))

- when host receives UDP segment:
  - checks destination port # in segment
  - directs UDP segment to socket with that port #

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

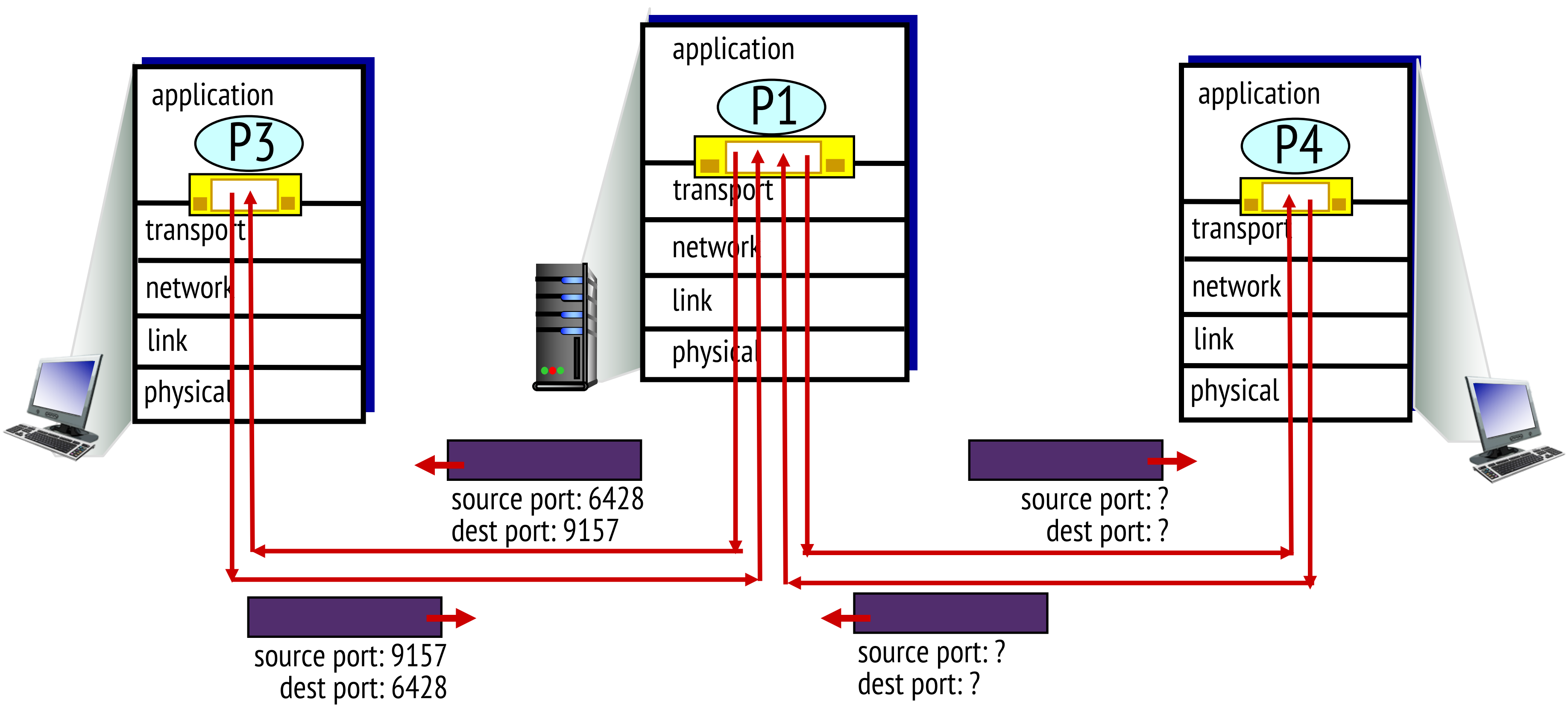
→ IP datagrams with *same dest. port #,* but different source IP addresses and/or source port numbers will be directed to *same socket* at dest

Connectionless demux: example

serverSocket.bind(('',
(6428))

clientSocket.bind(('', 9157))

clientSocket.bind(('', 5775))

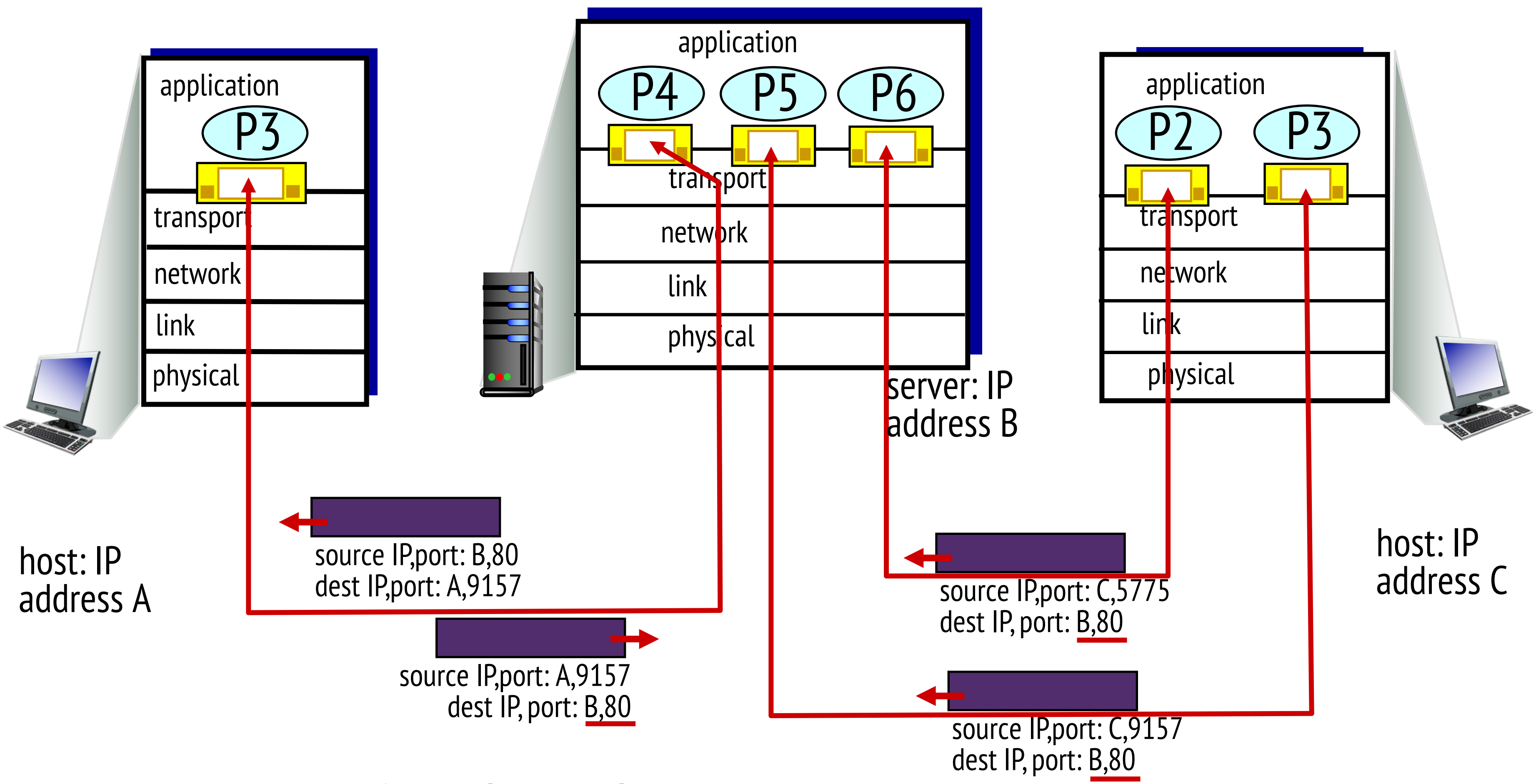


Connection-oriented demux

- TCP socket identified by 4-tuple:
  - source IP address
  - source port number
  - dest IP address
  - dest port number
- demux: receiver uses all four values to direct segment to appropriate socket

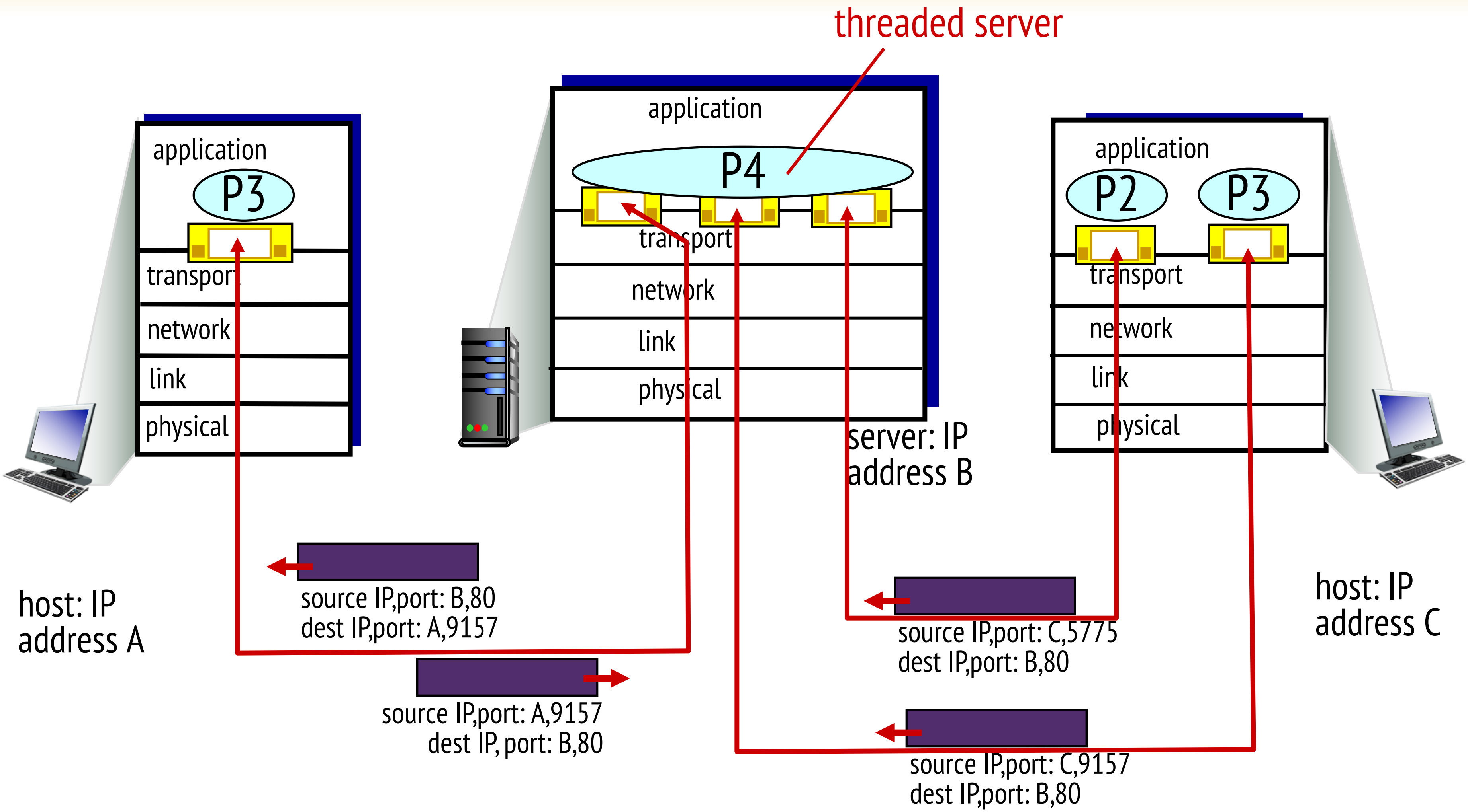
- server host may support many simultaneous TCP sockets:
  - each socket identified by its own 4-tuple
- web servers have different sockets for each connecting client
  - non-persistent HTTP will have different socket for each request

Connection-oriented demux: example



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

Connection-oriented demux: example



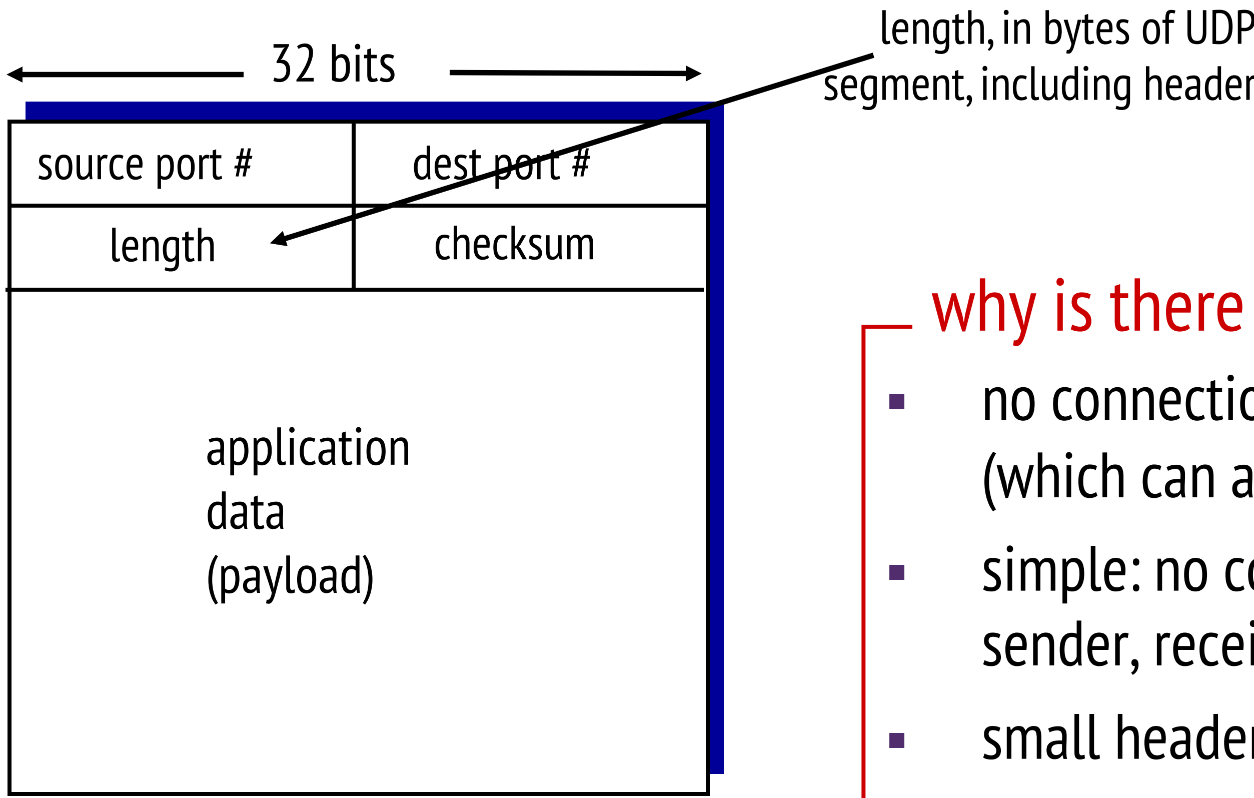
Connectionless Transport: UDP

UDP: User Datagram Protocol [RFC 768]

- "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

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

UDP: segment header



UDP segment format

why is there a UDP?

- no connection establishment (which can add delay)
- simple: no connection state at sender, receiver
- small header size
- no congestion control: UDP can blast away as fast as desired

UDP checksum [RFC 1071]

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

sender:

- treat segment contents, including header fields, as sequence of 16-bit integers
- checksum: addition (one's complement sum) of segment contents
- sender puts checksum value into UDP checksum field

receiver:

- compute checksum of received segment
- check if computed checksum equals checksum field value:
- NO - error detected
- YES - no error detected. But maybe errors nonetheless?

Internet checksum: example

example: add two 16-bit integers

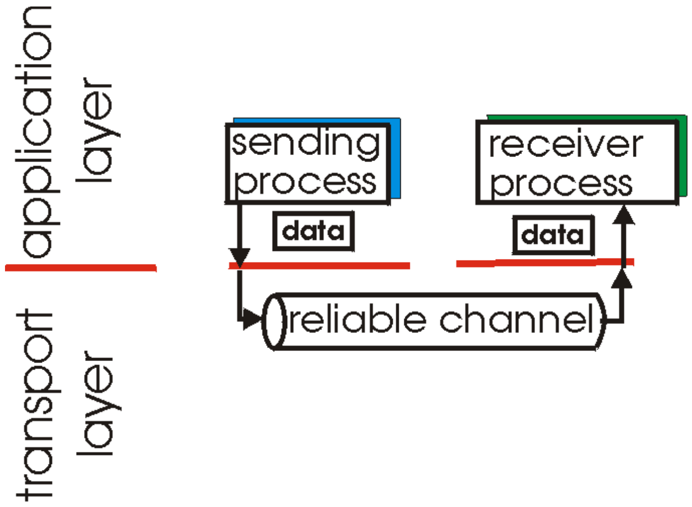
		1 1 1 0 0 1 1 0 0 1 1 0 0 1 1 0
		1 1 0 1 0 1 0 1 0 1 0 1 0 1 0 1
wraparound	1	1 0 1 1 1 0 1 1 1 0 1 1 1 0 1 1
sum		1 0 1 1 1 0 1 1 1 0 1 1 1 1 0 0
checksum		0 1 0 0 0 1 0 0 0 1 0 0 0 0 1 1

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

Principles of Reliable Data Transfer

Principles of reliable data transfer

- important in application, transport, link layers
  - top-10 list of important networking topics!



(a) provided service

- characteristics of unreliable channel will determine complexity of reliable data transfer protocol (rdt)

Principles of reliable data transfer

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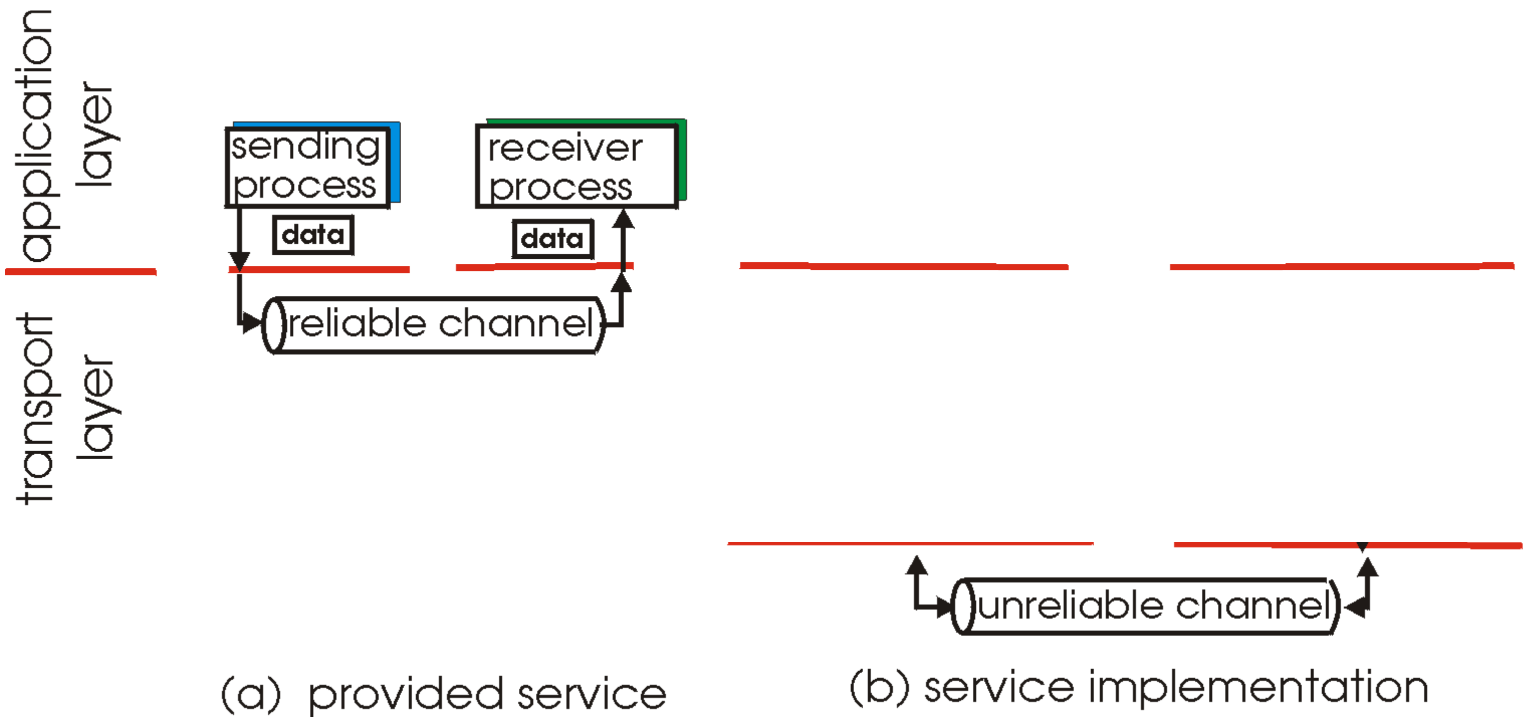

- characteristics of unreliable channel will determine complexity of reliable data transfer protocol (rdt)

Principles of reliable data transfer

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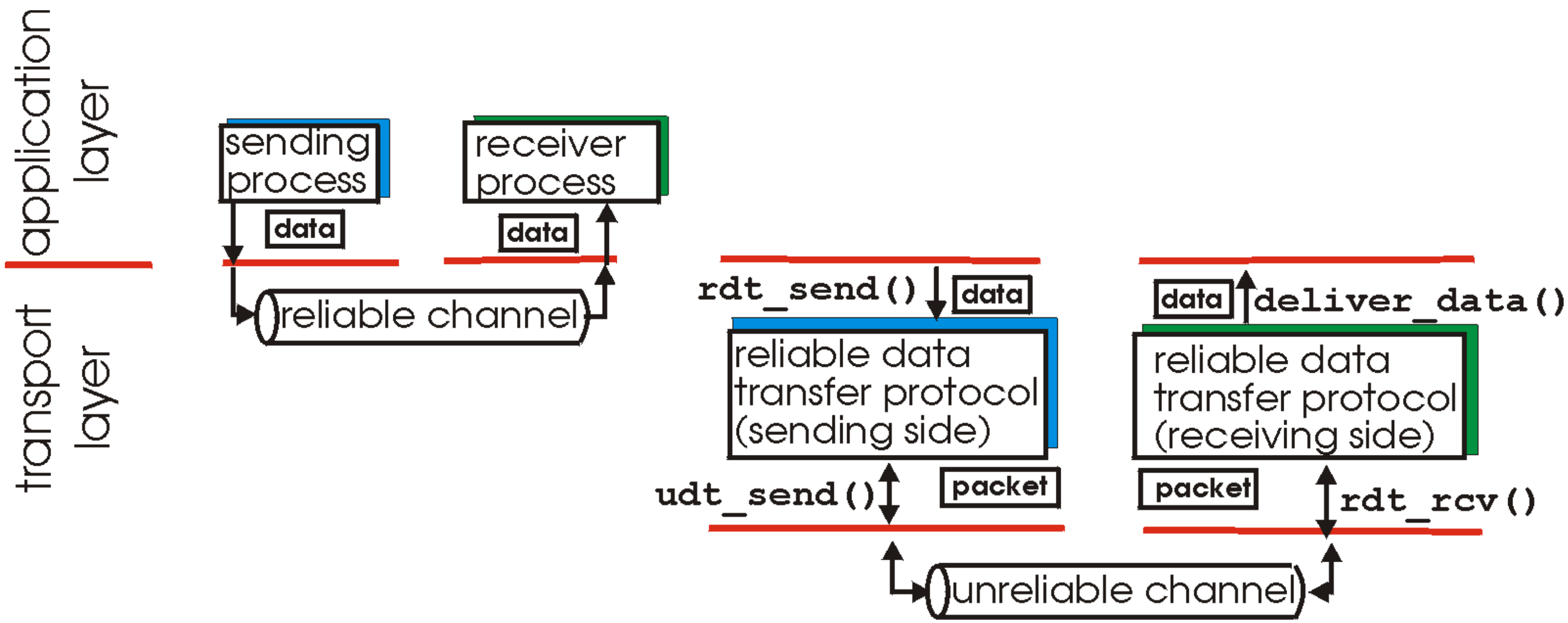


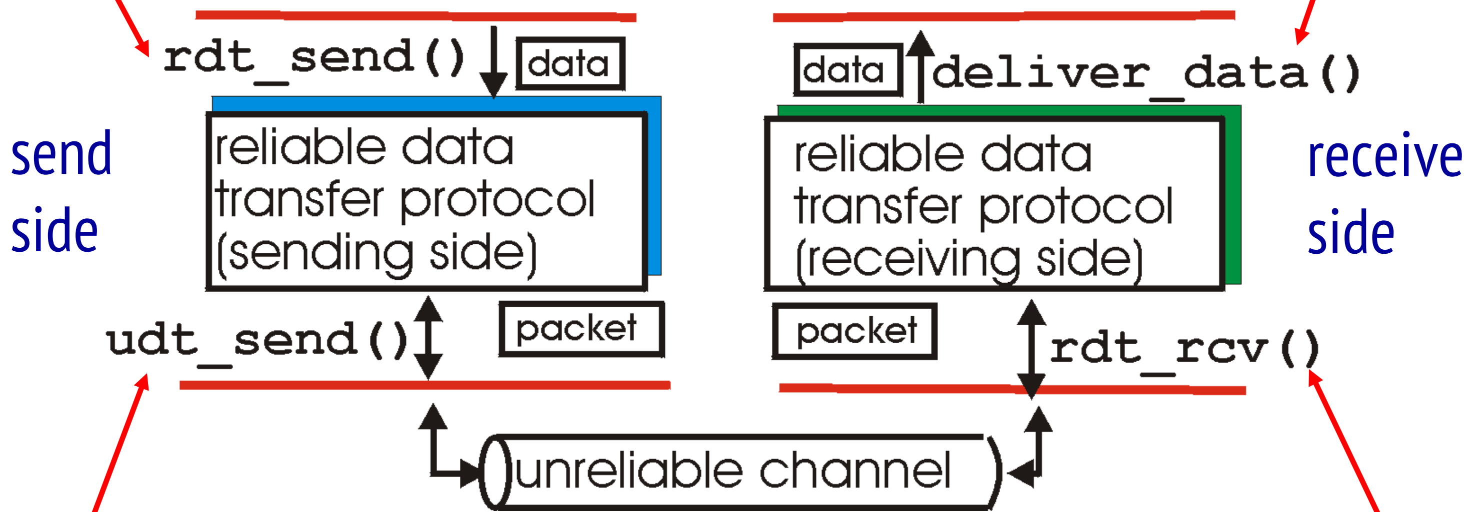
(a) provided service (b) service implementation

- characteristics of unreliable channel will determine complexity of reliable data transfer protocol (rdt)

Reliable data transfer: getting started

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

deliver_data(): called by **rdt** to deliver data to upper



udt_send(): called by rdt, to transfer packet over unreliable channel to receiver

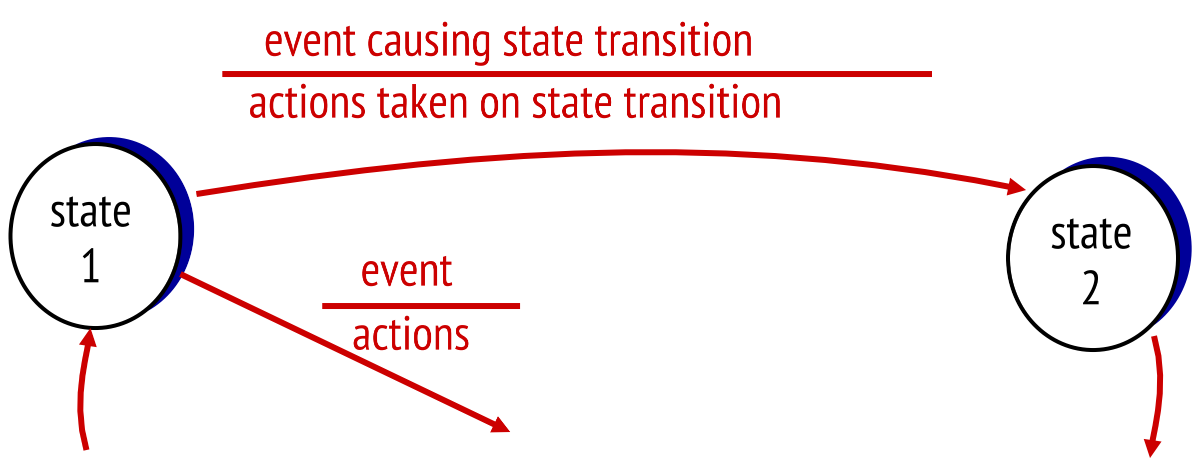
rdt_rcv(): called when packet arrives on rcv-side of channel

Reliable data transfer: getting started

we'll:

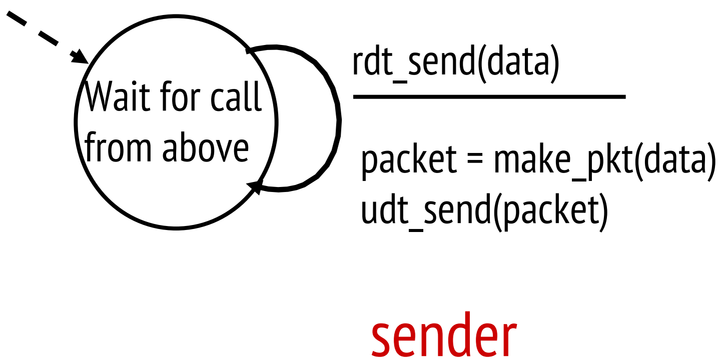
- incrementally develop sender, receiver sides of reliable data transfer protocol (rdt)
- consider only unidirectional data transfer
  - but control info will flow on both directions!
- use finite state machines (FSM) to specify sender, receiver

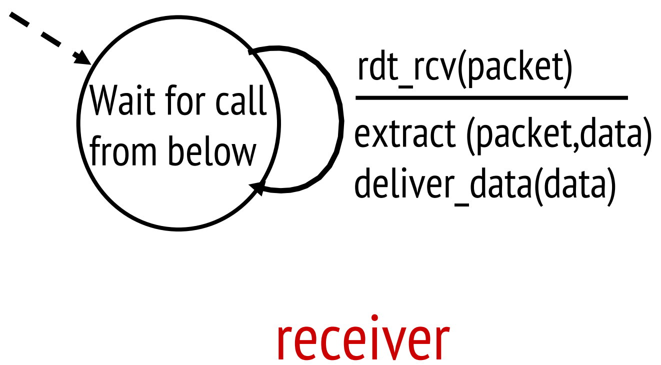
state: when in this "state" next state uniquely determined by next event



rdt1.0: reliable transfer over a reliable channel

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





rdt2.0: channel with bit errors

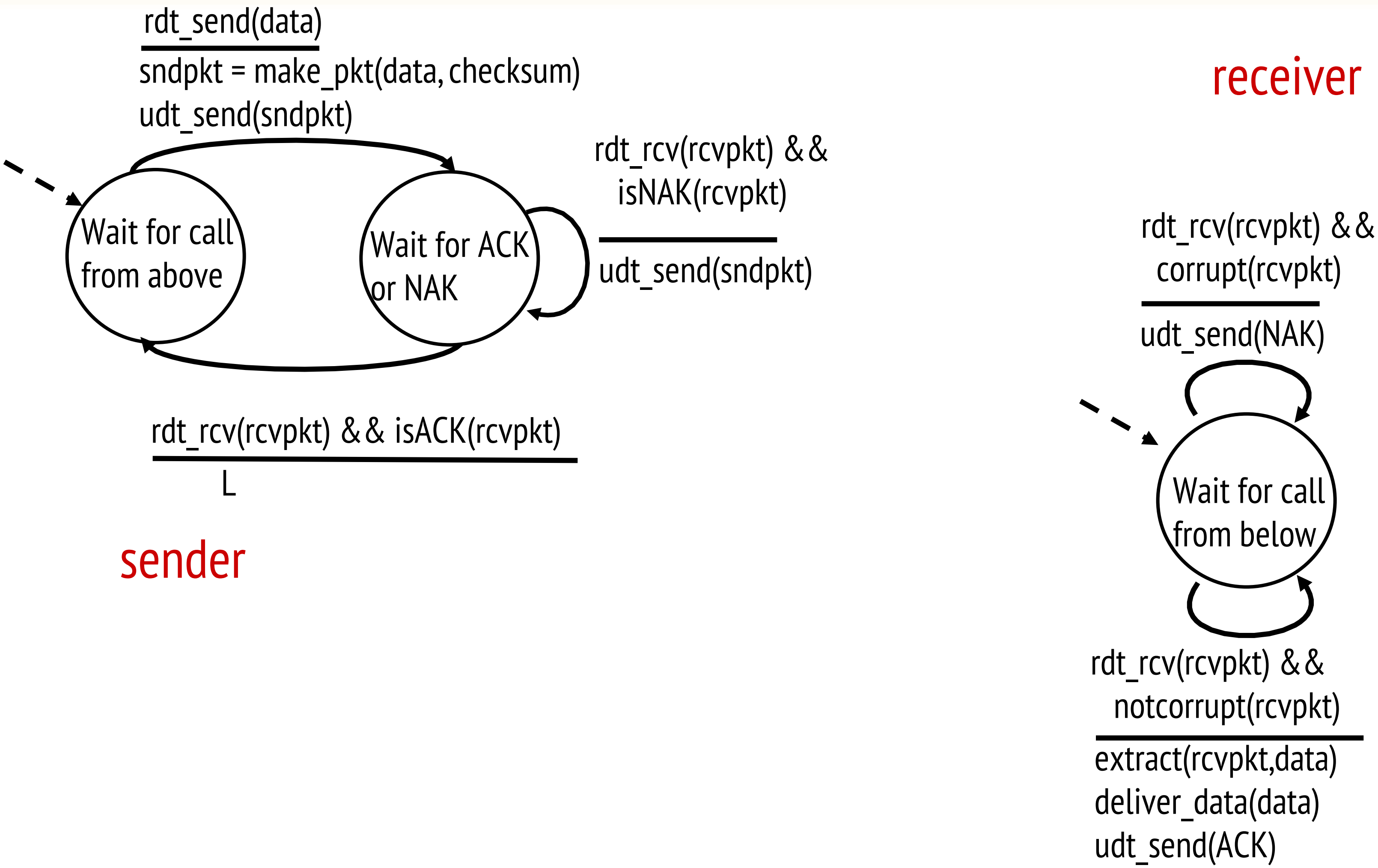
- underlying channel may flip bits in packet
  - checksum to detect bit errors
- *the* question: how to recover from errors:

How do humans recover from "errors" during conversation?

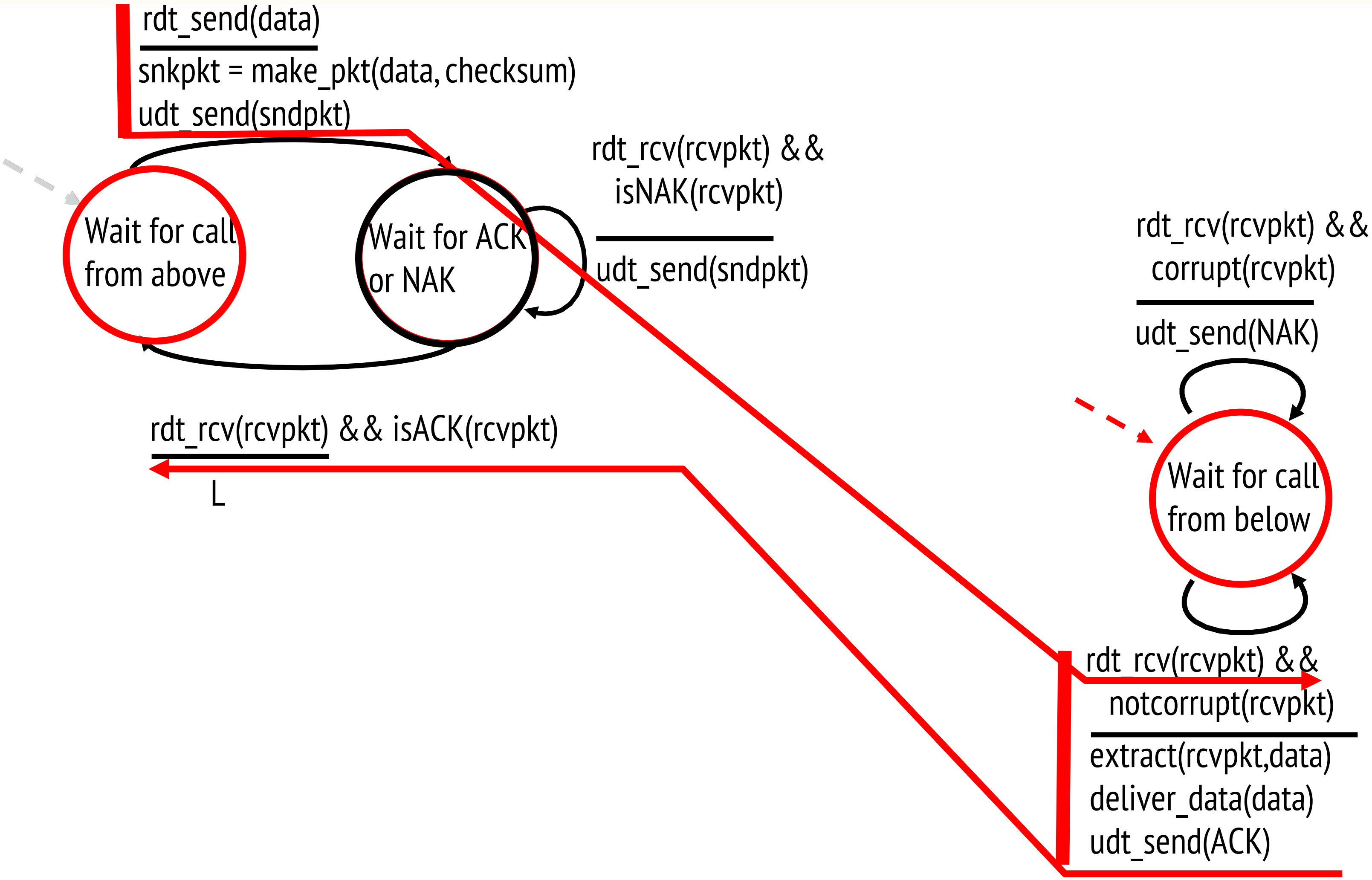
rdt2.0: channel with bit errors

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

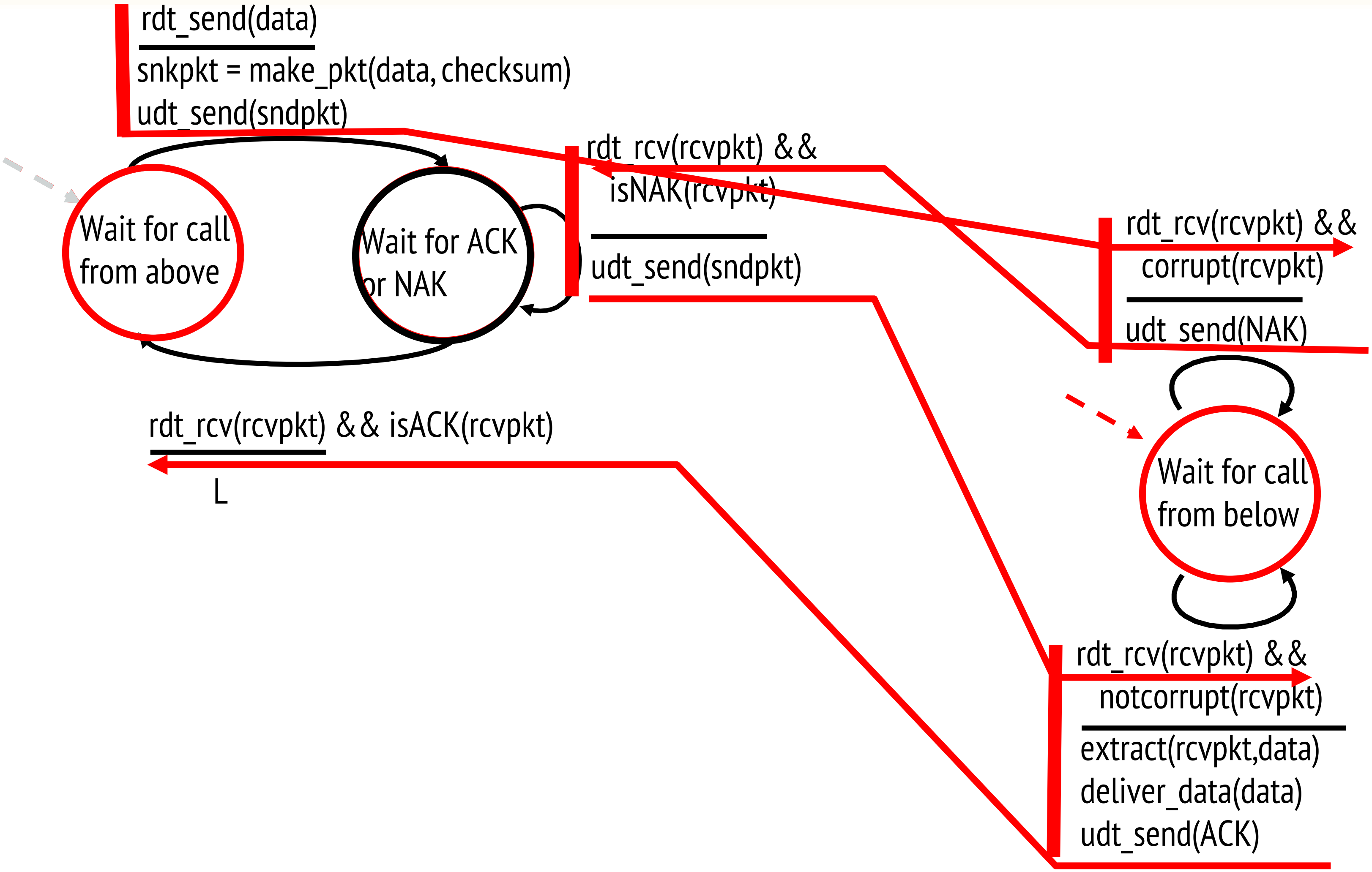
rdt2.0: FSM specification



rdt2.0: operation with no errors



rdt2.0: error scenario



rdt2.0 has a fatal flaw!

what happens if ACK/NAK corrupted?

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

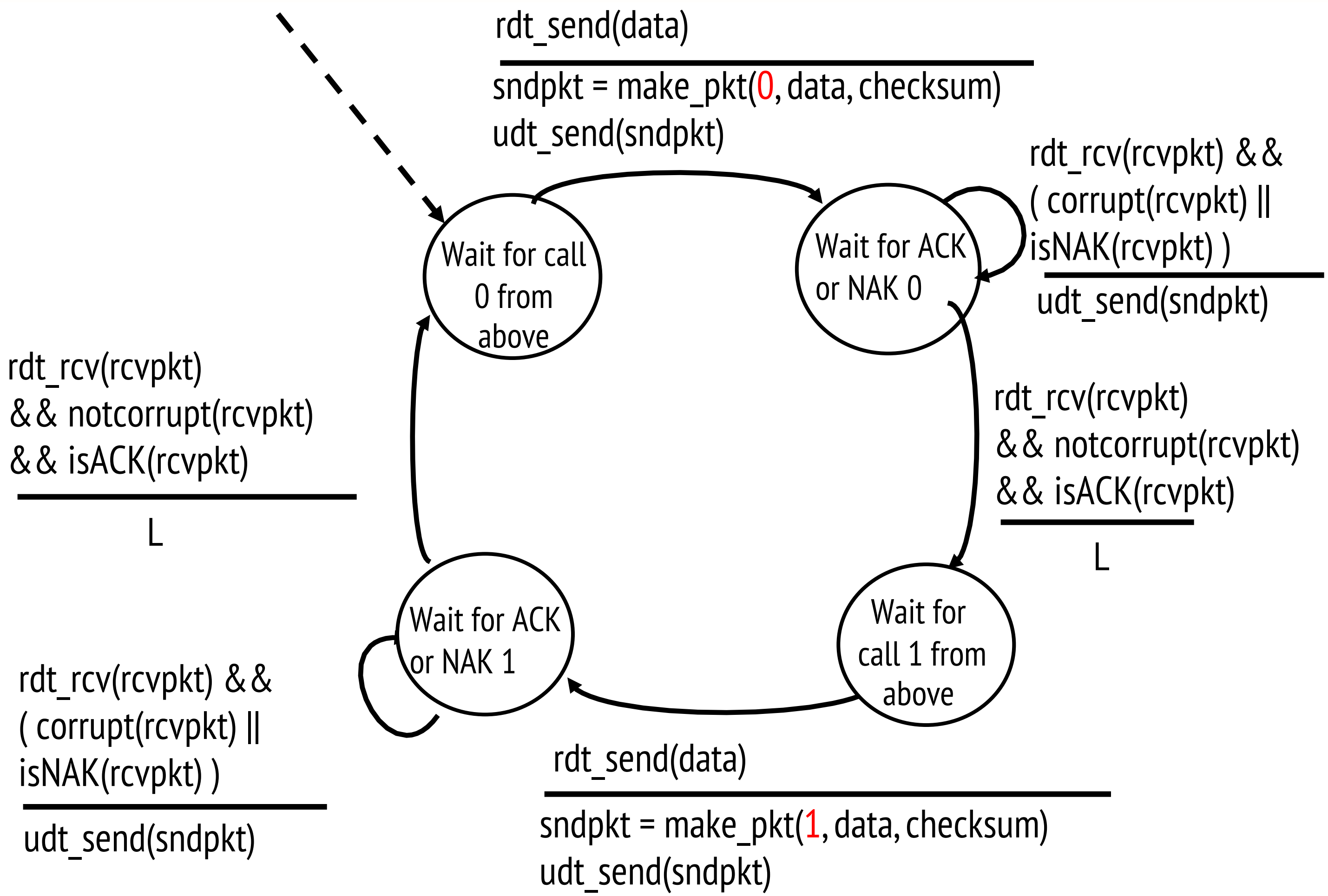
handling duplicates:

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

stop and wait

sender sends one packet, then waits for receiver response

rdt2.1: sender, handles garbled ACK/NAKs



rdt2.1: receiver, handles garbled ACK/NAKs

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

---

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

rdt_rcv(rcvpkt) && corrupt(rcvpkt)

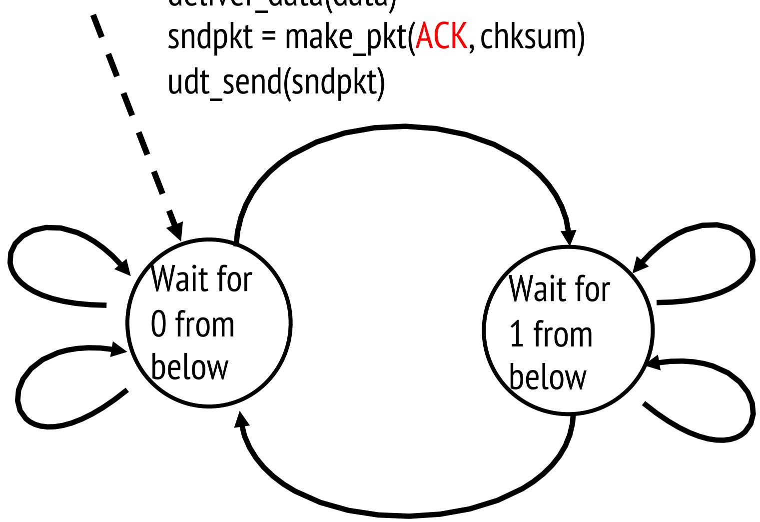
---

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

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

---

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



rdt_rcv(rcvpkt) && corrupt(rcvpkt)

---

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

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

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sndpkt = make_pkt(ACK, chksum)
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rdt_rcv(rcvpkt) && notcorrupt(rcvpkt)
&& **has_seq1(rcvpkt)**

---

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

rdt2.1: discussion

sender:

- seq # added to pkt
- must check if received ACK/NAK corrupted
- twice as many states
  - state must "remember" whether "expected" pkt should have seq # of 0 or 1

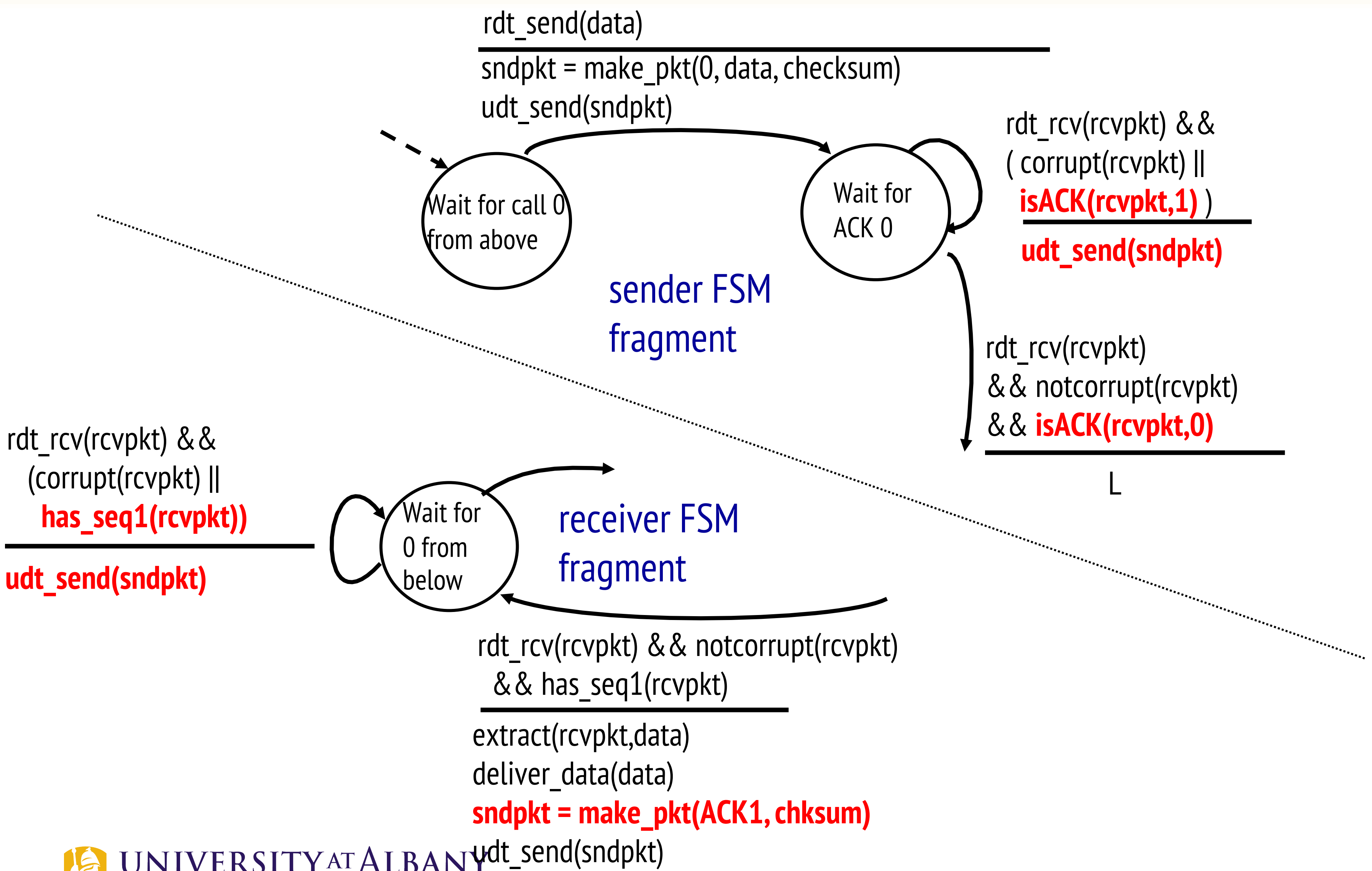
receiver:

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

rdt2.2: a NAK-free protocol

- same functionality as rdt2.1, using ACKs only
- instead of NAK, receiver sends ACK for last pkt received OK
  - receiver must *explicitly* include seq # of pkt being ACKed
- duplicate ACK at sender results in same action as NAK: *retransmit current pkt*

rdt2.2: sender, receiver fragments



rdt3.0: channels with errors *and* loss

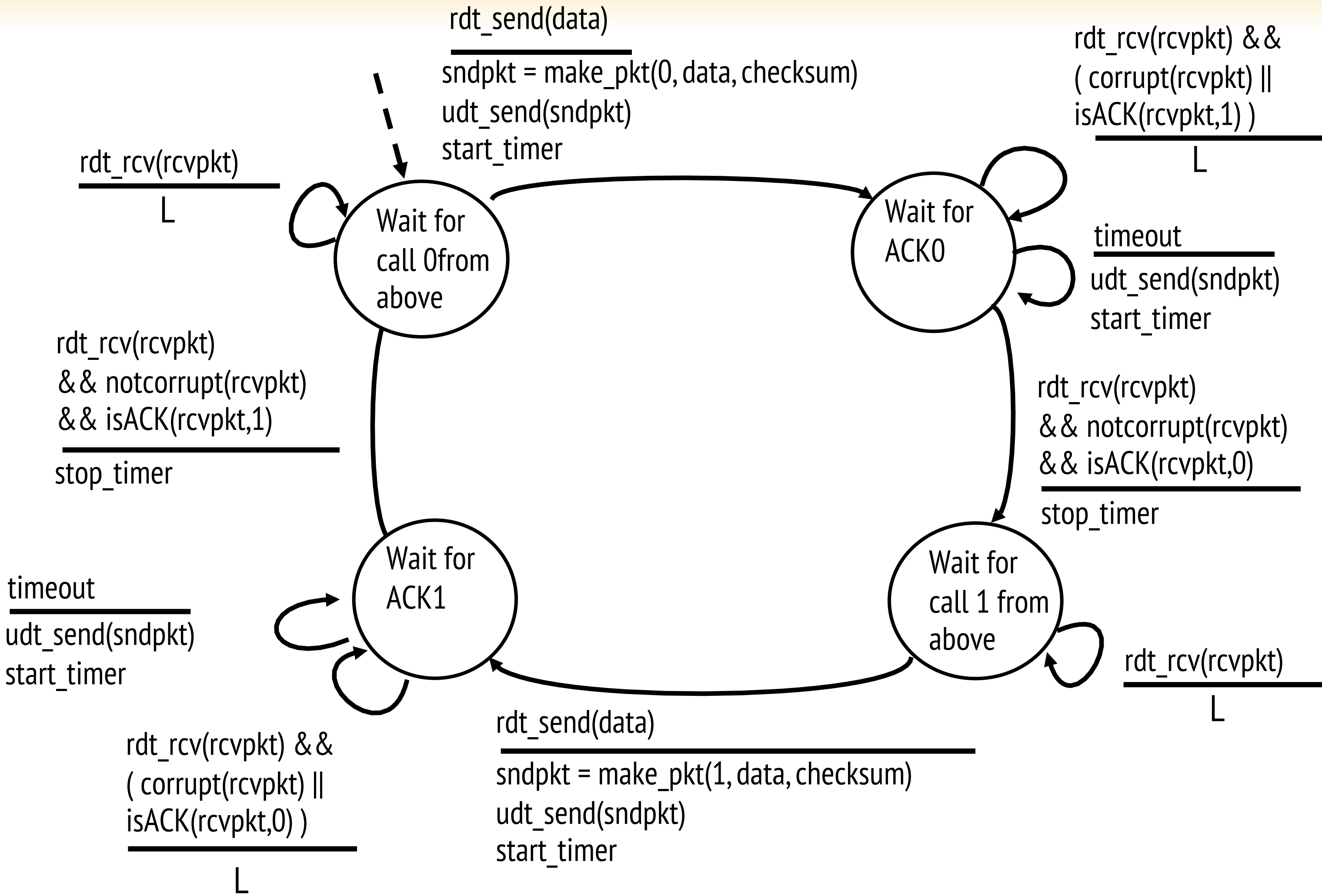
new assumption: underlying channel can also lose packets (data, ACKs)

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

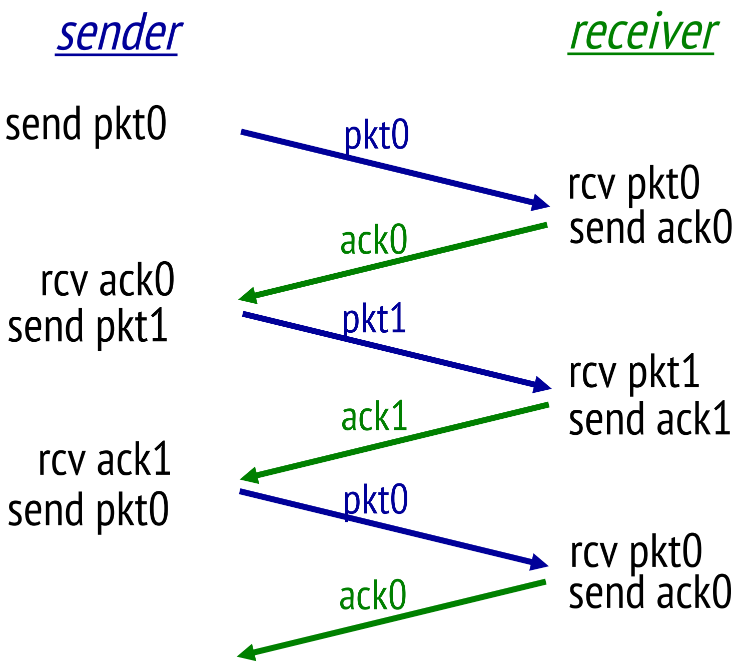
approach: sender waits "reasonable" amount of time for ACK

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

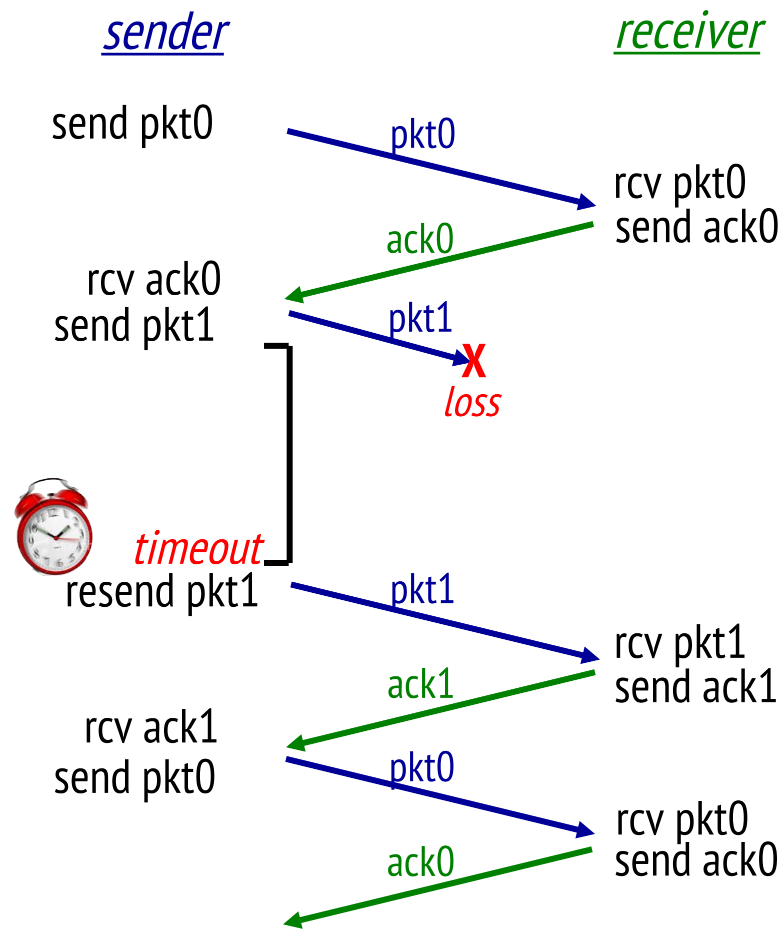
rdt3.0 sender



rdt3.0 in action

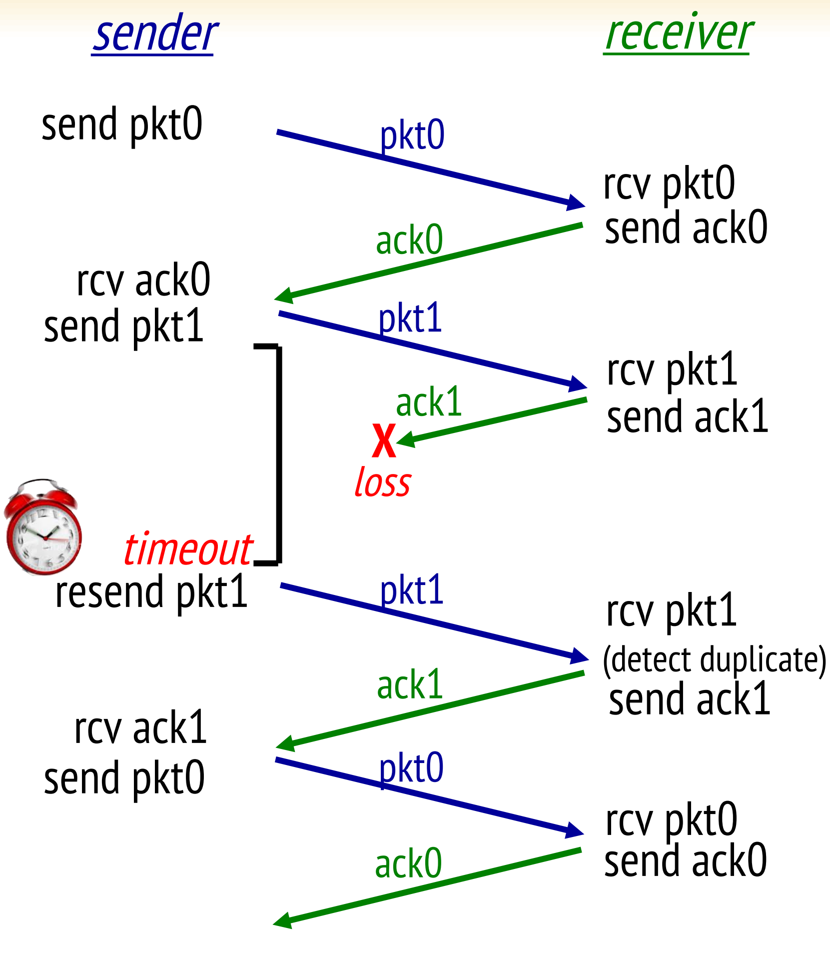


(a) no loss

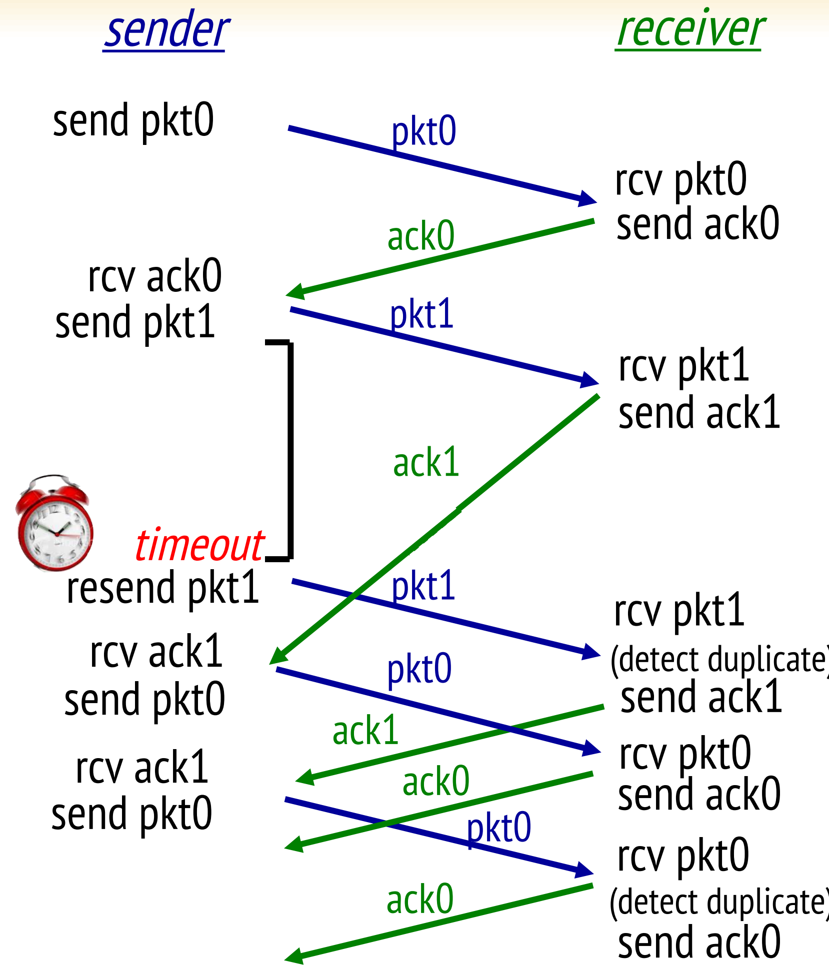


(b) packet loss

rdt3.0 in action



(c) ACK loss



(d) premature timeout/ delayed ACK

Performance of rdt3.0

- rdt3.0 is correct, but performance stinks
- e.g.: 1 Gbps link, 15 ms prop. delay, 8000 bit packet:

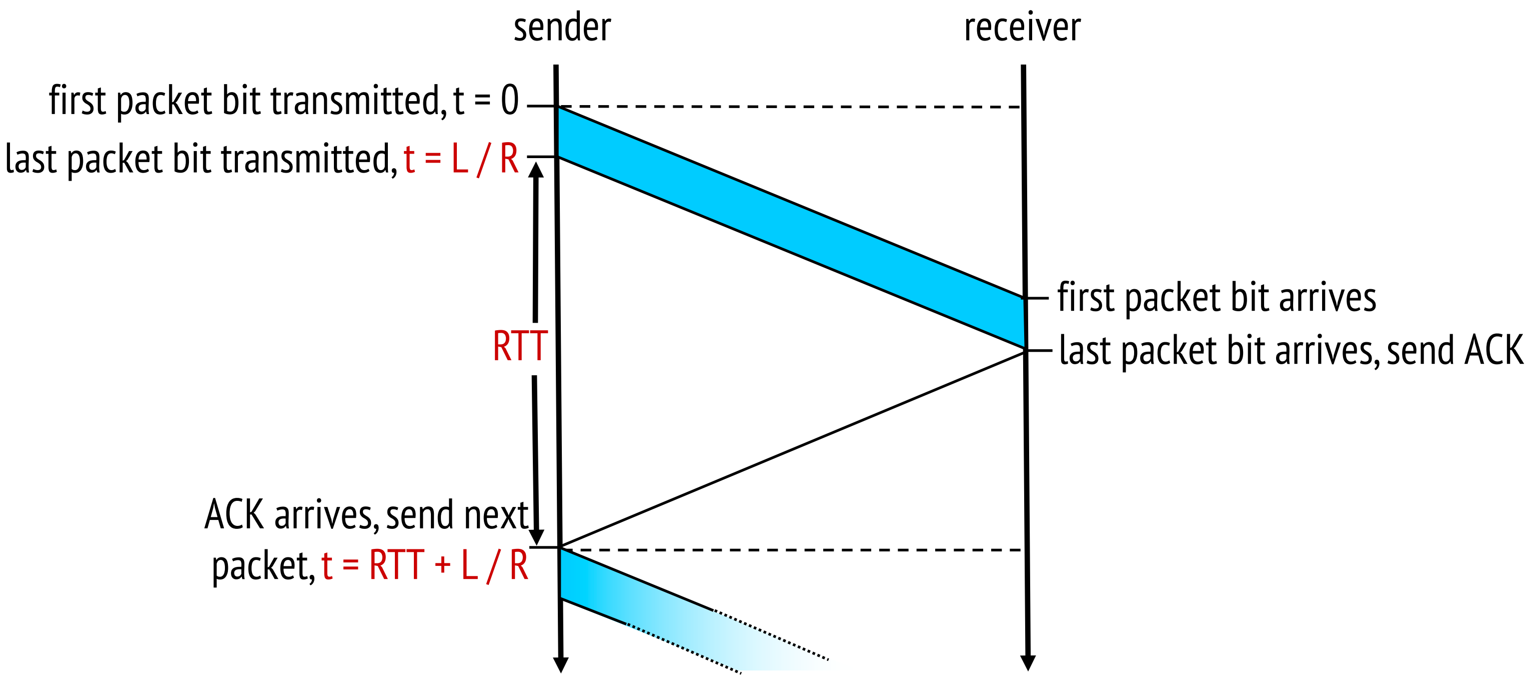
$$D_{trans} = \frac{L}{R} = \frac{8000 \text{ bits}}{10^9 \text{ bits/sec}} = 8 \text{ microsecs}$$

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

$$U_{sender} = \frac{L/R}{RTT + L/R} = \frac{.008}{30.008} = 0.00027$$

- if RTT=30 msec, 1KB pkt every 30 msec: 33kB/sec thruput over 1 Gbps link
- network protocol limits use of physical resources!

rdt3.0: stop-and-wait operation

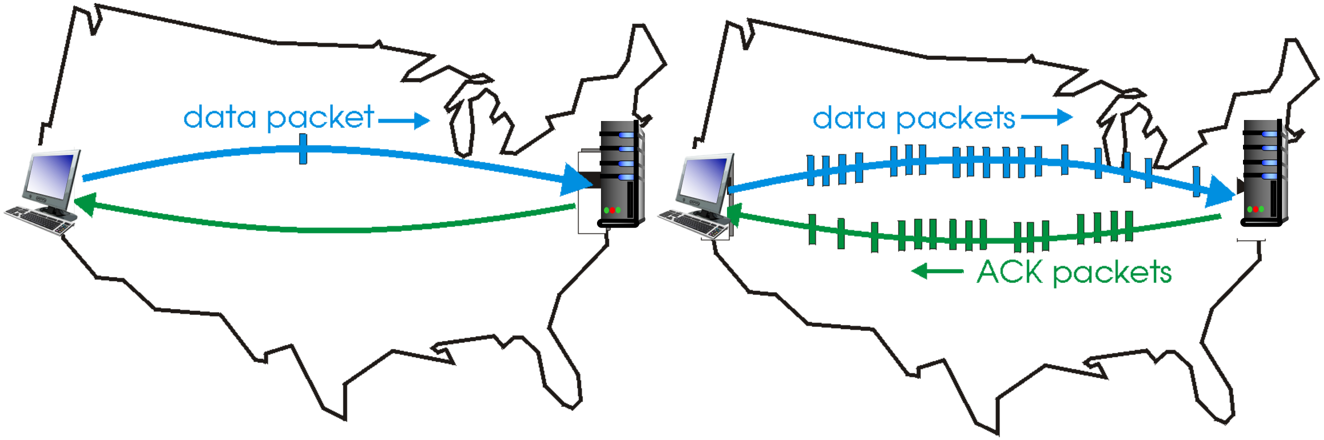


$$U_{sender} = \frac{L / R}{RTT + L / R} = \frac{.008}{30.008} = 0.00027$$

Pipelined protocols

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

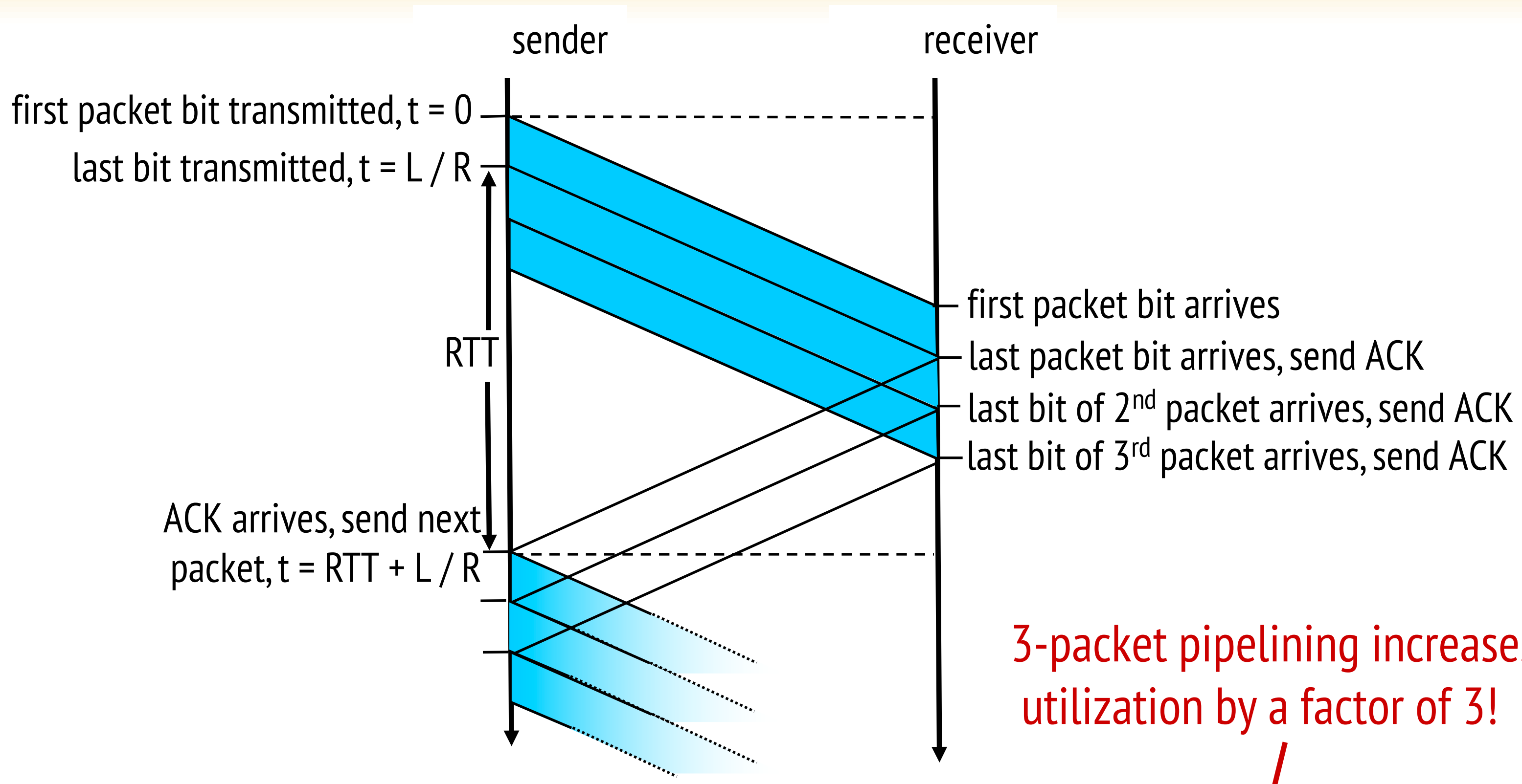
- range of sequence numbers must be increased
- buffering at sender and/or receiver



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

- two generic forms of pipelined protocols: *go-Back-N, selective repeat*

Pipelining: increased utilization



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

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

Pipelined protocols: overview

Go-back-N:

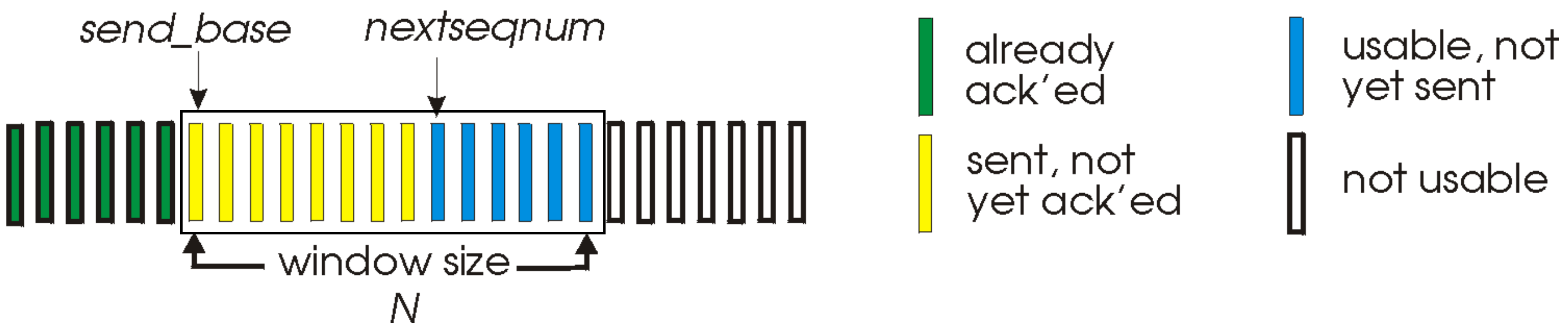
- sender can have up to N unacked packets in pipeline
- receiver only sends *cumulative ack*
  - doesn't ack packet if there's a gap
- sender has timer for oldest unacked packet
  - when timer expires, retransmit *all* unacked packets

Selective Repeat:

- sender can have up to N unack'ed packets in pipeline
- rcvr 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

- k-bit seq # in pkt header
- "window" of up to N, consecutive unack'ed pkts allowed



- ACK(n): ACKs all pkts up to, including seq # n - *"cumulative ACK"*
  - may receive duplicate ACKs (see receiver)
- timer for oldest in-flight pkt
- *timeout(n):* retransmit packet n and all higher seq # pkts in window

GBN: sender extended FSM

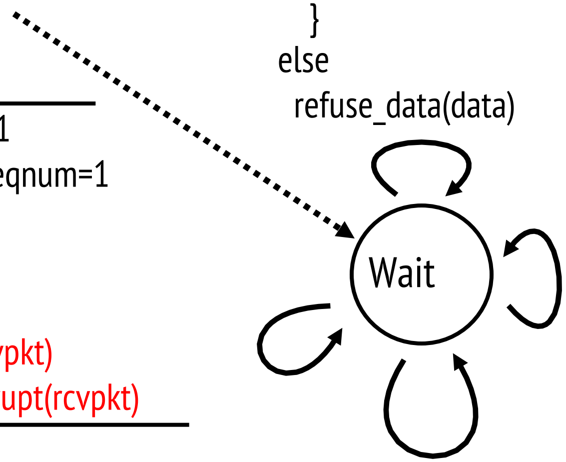
rdt_send(data)

```
if (nextseqnum < base+N) {
   sndpkt[nextseqnum] = make_pkt(nextseqnum,data,chksum)
   udt_send(sndpkt[nextseqnum])
   if (base == nextseqnum)
     start_timer
   nextseqnum++
   }
else
  refuse_data(data)
```

L

---

base=1
nextseqnum=1



timeout

```
start_timer
udt_send(sndpkt[base])
udt_send(sndpkt[base+1])
...
udt_send(sndpkt[nextseqnum-1])
```

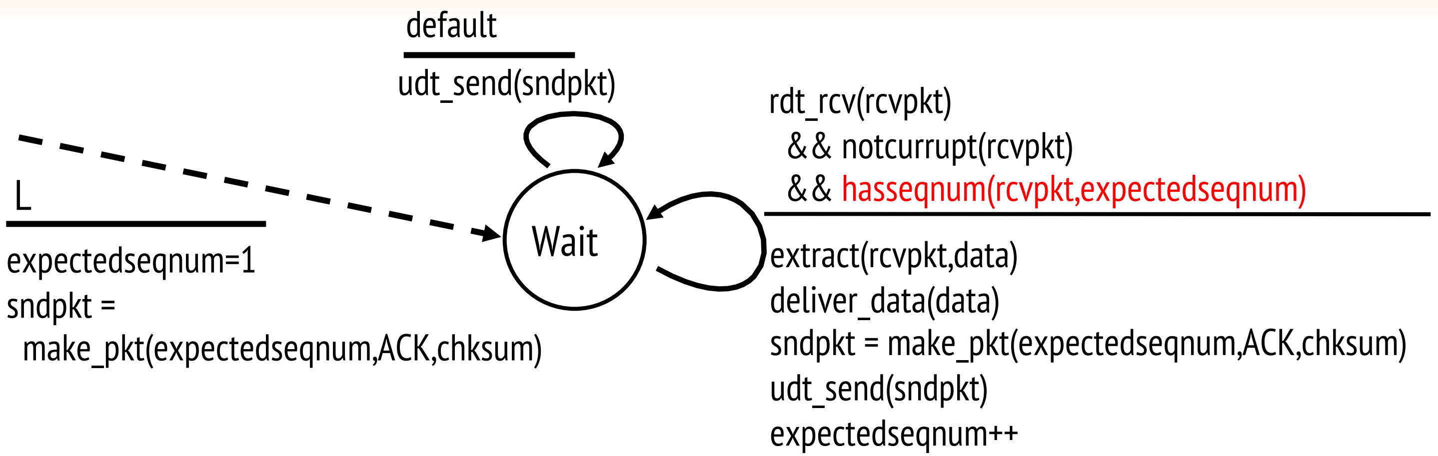
rdt_rcv(rcvpkt)
  && corrupt(rcvpkt)

---

rdt_rcv(rcvpkt) &&
  notcorrupt(rcvpkt)

```
base = getacknum(rcvpkt)+1
If (base == nextseqnum)
   stop_timer
  else
   start_timer
```

GBN: receiver extended FSM

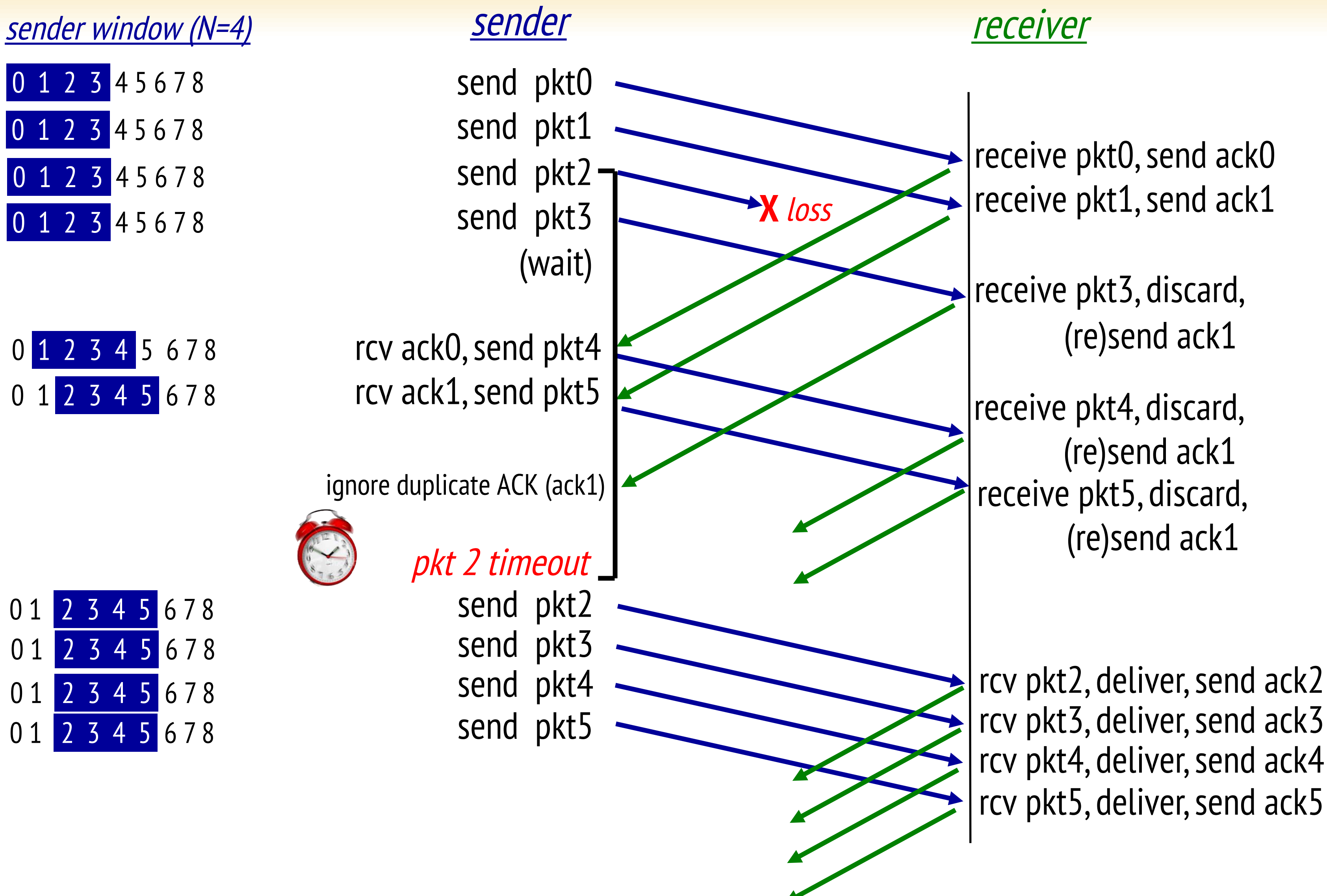


ACK-only: always send ACK for correctly-received pkt with highest *in-order* seq #

- may generate duplicate ACKs
- need only remember **expectedseqnum**

- out-of-order pkt:
  - discard (don't buffer): *no receiver buffering!*
  - re-ACK pkt with highest in-order seq #

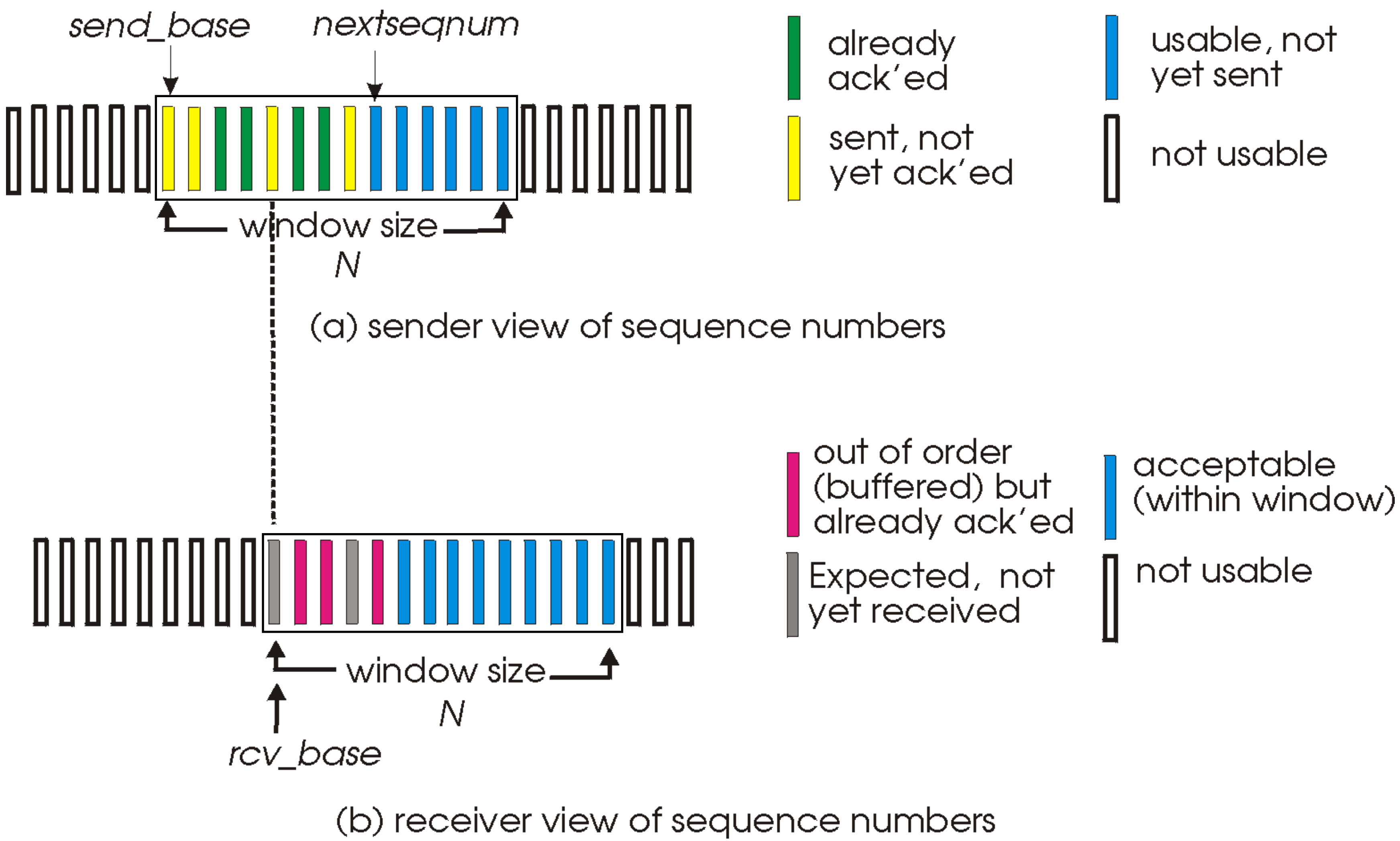
GBN in action



Selective repeat

- receiver *individually* acknowledges all correctly received pkts
  - buffers pkts, as needed, for eventual in-order delivery to upper layer
- sender only resends pkts for which ACK not received
  - sender timer for each unACKed pkt
- sender window
  - *N* consecutive seq #'s
  - limits seq #s of sent, unACKed pkts

Selective repeat: sender, receiver windows



Selective repeat

sender

data from above:

- if next available seq # in window, send pkt

timeout(n):

- resend pkt n, restart timer

ACK(n) in [sendbase,sendbase+N]:

- mark pkt n as received
- if n smallest unACKed pkt, advance window base to next unACKed seq #

receiver

pkt n in [rcvbase, rcvbase+N-1]

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

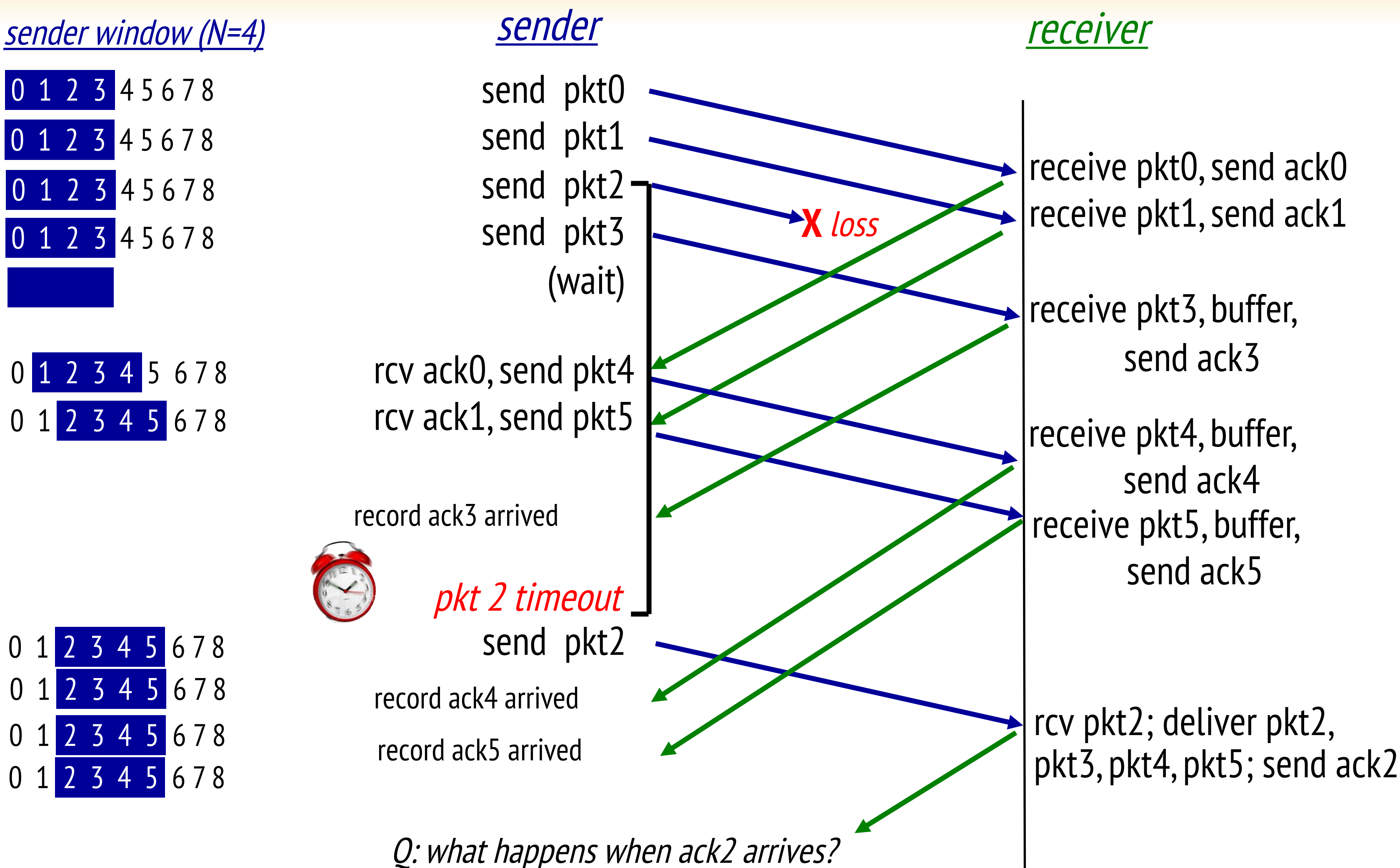
pkt n in [rcvbase-N,rcvbase-1]

- ACK(n)

otherwise:

- ignore

Selective repeat in action

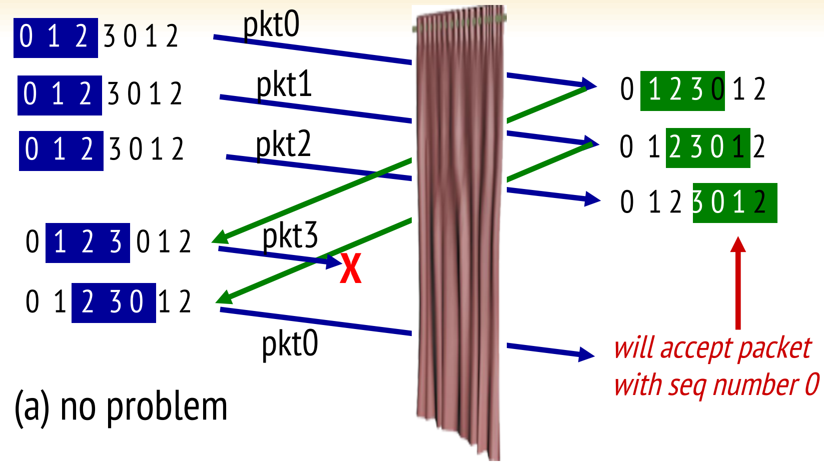


Selective repeat

sender window (after receipt)

receiver window (after receipt)

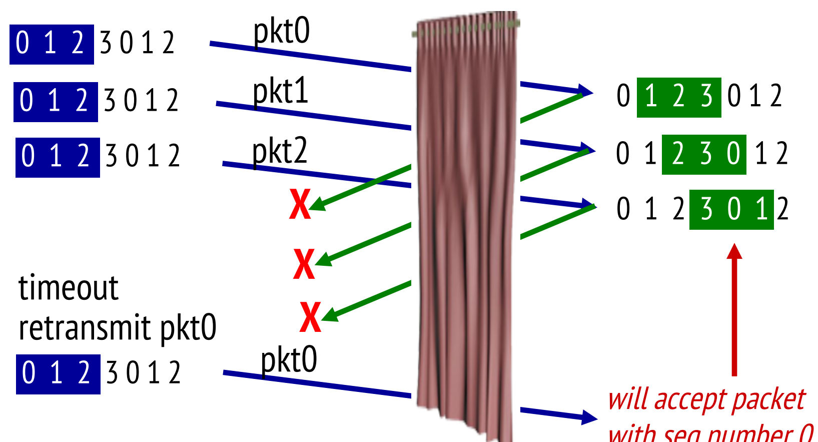
- Dilemma example
  - seq #'s: 0, 1, 2, 3
  - window size=3



(a) no problem

  - receiver sees no difference in two scenarios!
  - duplicate data accepted as new in (b)

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



(b) oops!

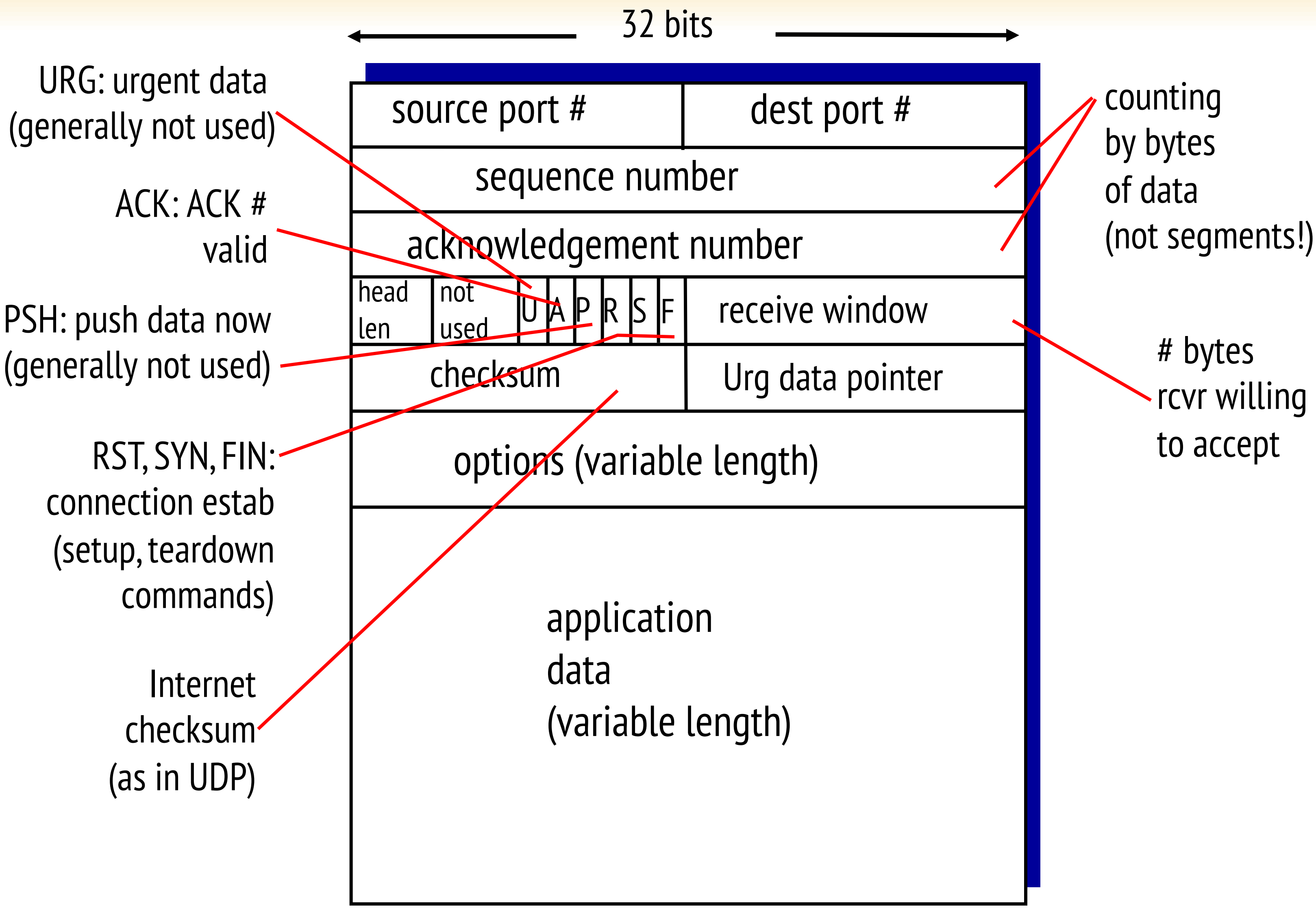
  - Q: what relationship between seq # size and window size to avoid problem in (b)?

Connection-oriented Transport: TCP

TCP: Overview RFCs: 793,1122,1323, 2018, 2581

- full duplex data:
  - bi-directional data flow in same connection
  - MSS: maximum segment size
- connection-oriented:
  - handshaking (exchange of control msgs) inits sender, receiver state before data exchange
- flow controlled:
  - sender will not overwhelm receiver
- point-to-point:
  - one sender, one receiver
- reliable, in-order *byte steam:*
  - no "message boundaries"
- pipelined:
  - TCP congestion and flow control set window size

TCP segment structure



TCP seq. numbers, ACKs

sequence numbers:

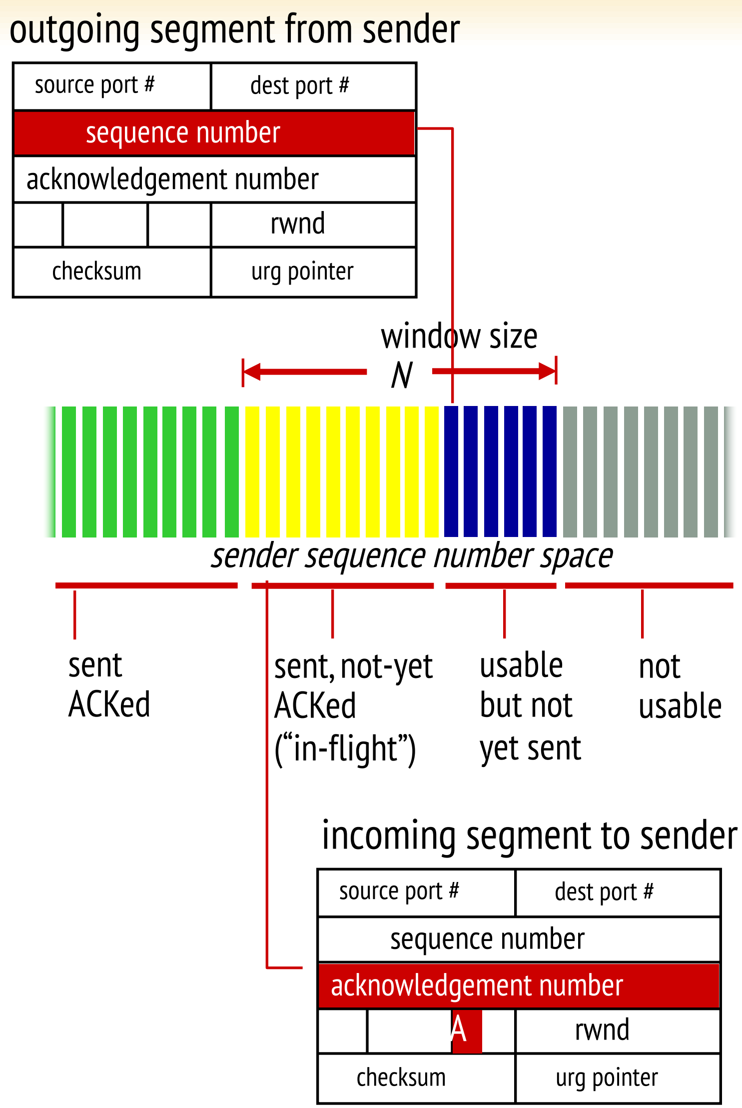
- byte stream "number" of first byte in segment's data

acknowledgements:

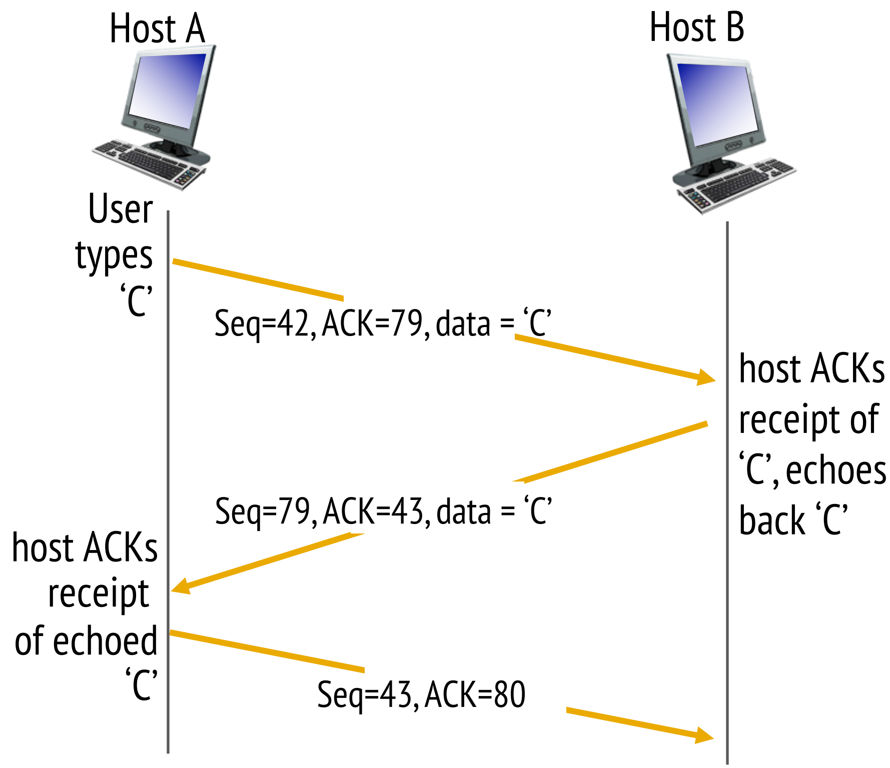
- seq # of next byte expected from other side
- cumulative ACK

Q: how receiver handles out-of-order segments

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



TCP seq. 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

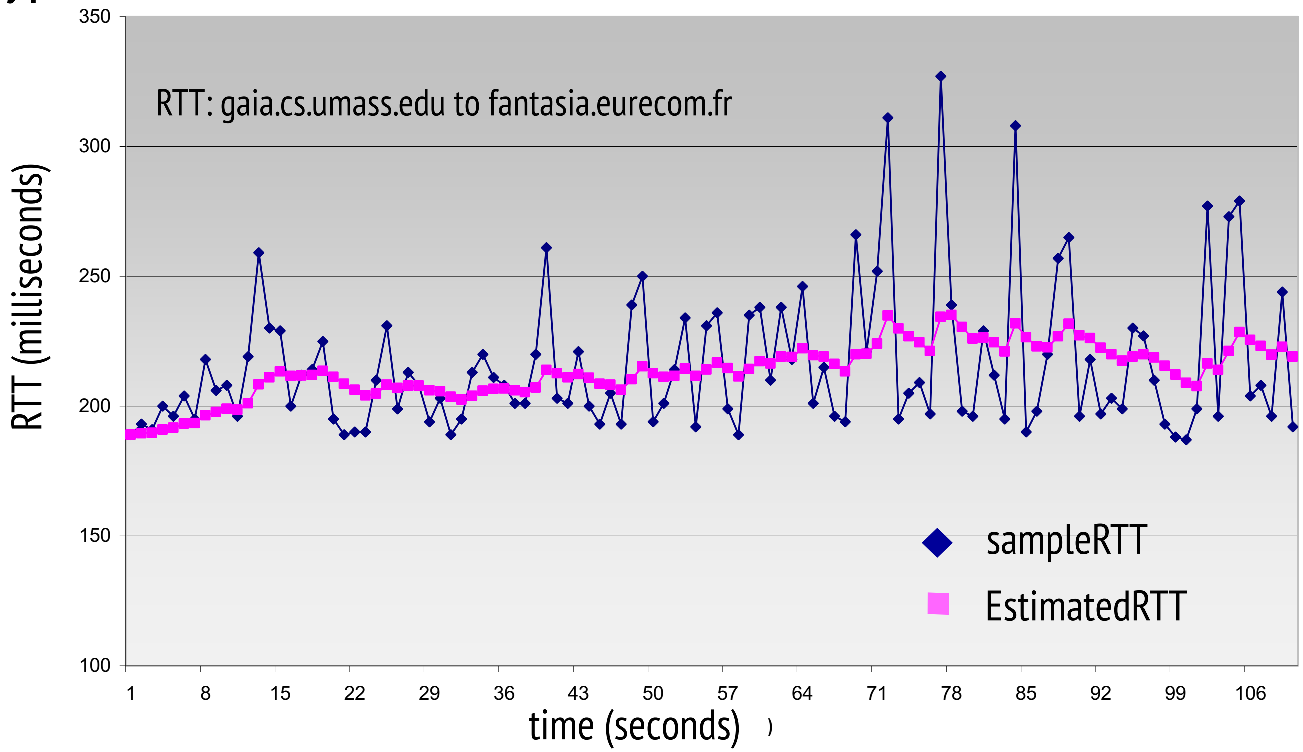
Q: how to estimate RTT?

- **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

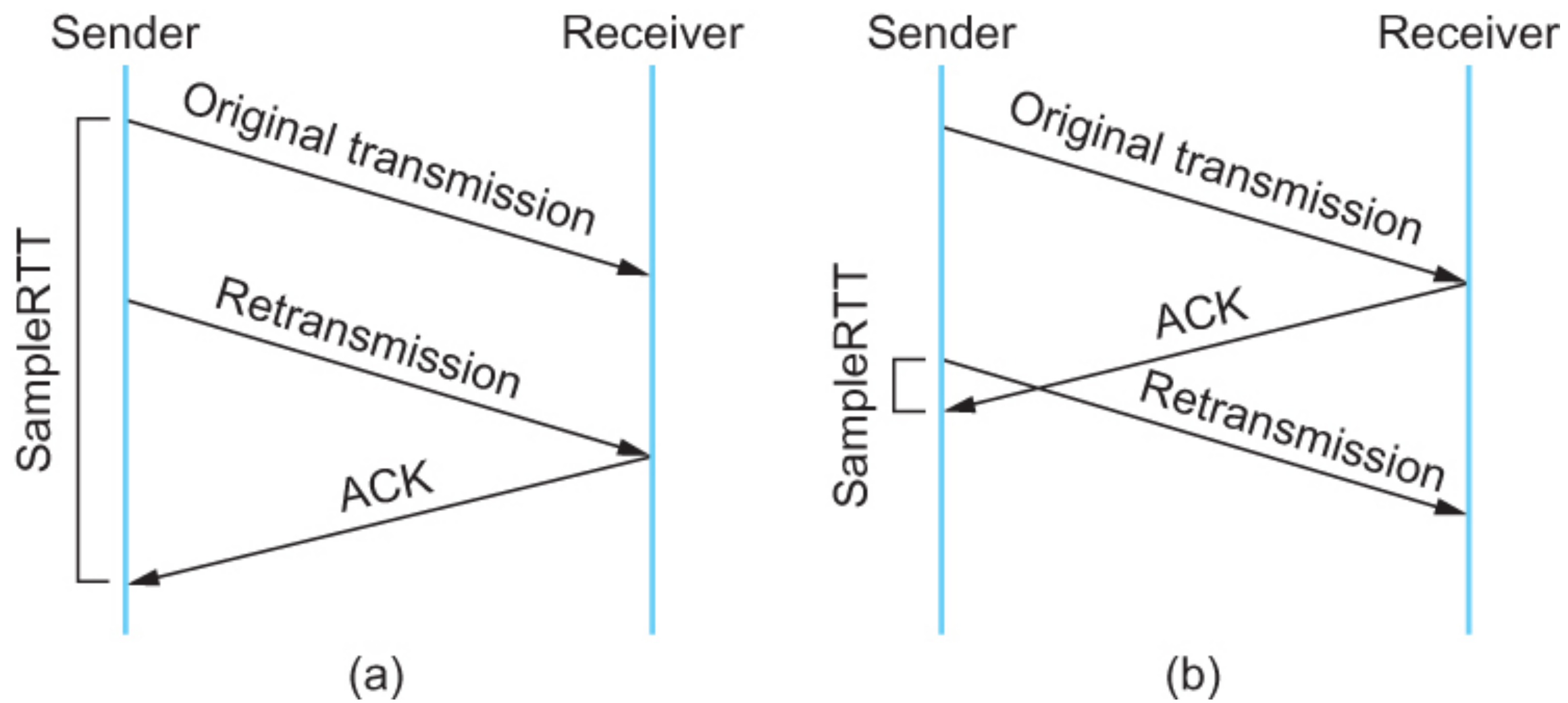
EstimatedRTT = (1- α)*EstimatedRTT + α*SampleRTT

- exponential weighted moving average
- influence of past sample decreases exponentially fast
- typical value: α = 0.125



Timeout = 2*EstimatedRTT

How to calculate SampleRTT?



Associating the ACK with (a) original transmission versus (b) retransmission

Karn/Partridge Algorithm

- Do not sample RTT when retransmitting

- Karn-Partridge algorithm was an improvement over the original approach, but it does not eliminate congestion

- We need to understand how timeout is related to congestion
  - If you timeout too soon, you may unnecessarily retransmit a segment which adds load to the network

Karn/Partridge Algorithm

- Main problem with the original computation is that it does not take variance of Sample RTTs into consideration.
- If the variance among Sample RTTs is small
  - Then the Estimated RTT can be better trusted
  - There is no need to multiply this by 2 to compute the timeout

Karn/Partridge Algorithm

- On the other hand, a large variance in the samples suggest that timeout value should not be tightly coupled to the Estimated RTT

- Jacobson/Karels proposed a new scheme for TCP retransmission

Jacobson/Karels Algorithm

- timeout interval: **EstimatedRTT** plus “safety margin”
  - large variation in **EstimatedRTT ->** larger safety margin
- estimate SampleRTT deviation from EstimatedRTT:
- RFC 6298

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

(typically, $\beta = 0.25$)

Measure of variability

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

estimated RTT

“safety margin”

TCP reliable data transfer

- TCP creates rdt service on top of IP's unreliable service
  - pipelined segments
  - cumulative acks
  - single retransmission timer
- retransmissions triggered by:
  - timeout events
  - duplicate acks

let's initially consider simplified TCP sender:

- ignore duplicate acks
- ignore flow control, congestion control

TCP sender events:

data rcvd from app:

- 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**

timeout:

- retransmit segment that caused timeout
- restart timer

ack rcvd:

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

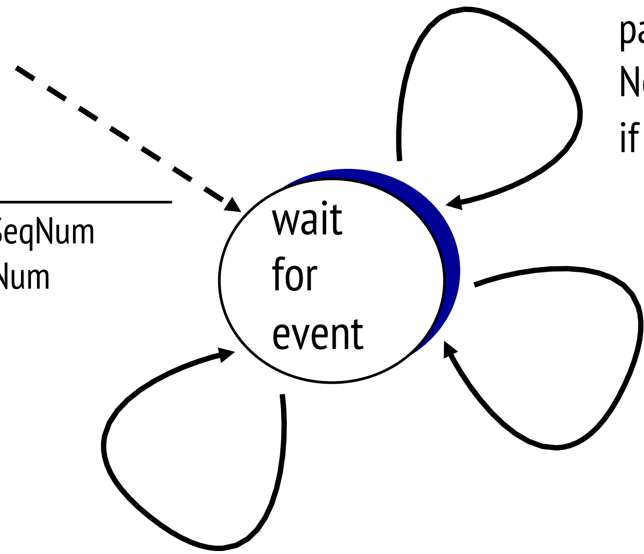
TCP sender (simplified)

data received from application above

create segment, seq. #: NextSeqNum
pass segment to IP (i.e., "send")
NextSeqNum = NextSeqNum + length(data)
if (timer currently not running)
  start timer

L

NextSeqNum = InitialSeqNum
SendBase = InitialSeqNum



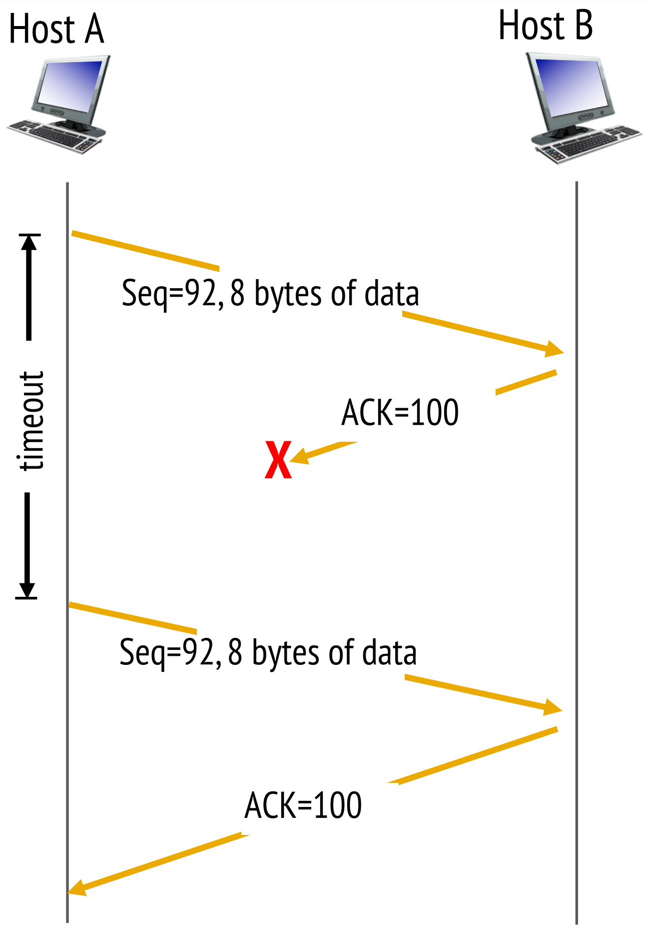
timeout

retransmit not-yet-acked segment
  with smallest seq. #
start timer

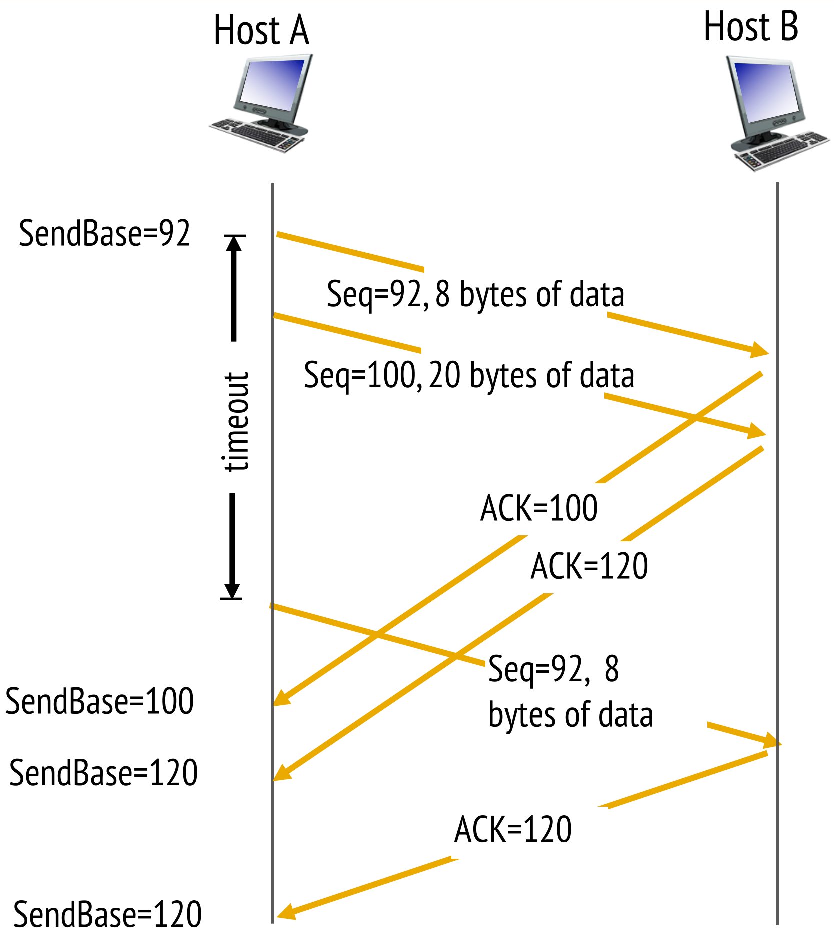
ACK received, with ACK field value y

```
if (y > SendBase) {
   SendBase = y
   /* SendBase–1: last cumulatively ACKed byte */
   if (there are currently not-yet-acked segments)
       start timer
     else stop timer
  }
```

TCP: retransmission scenarios

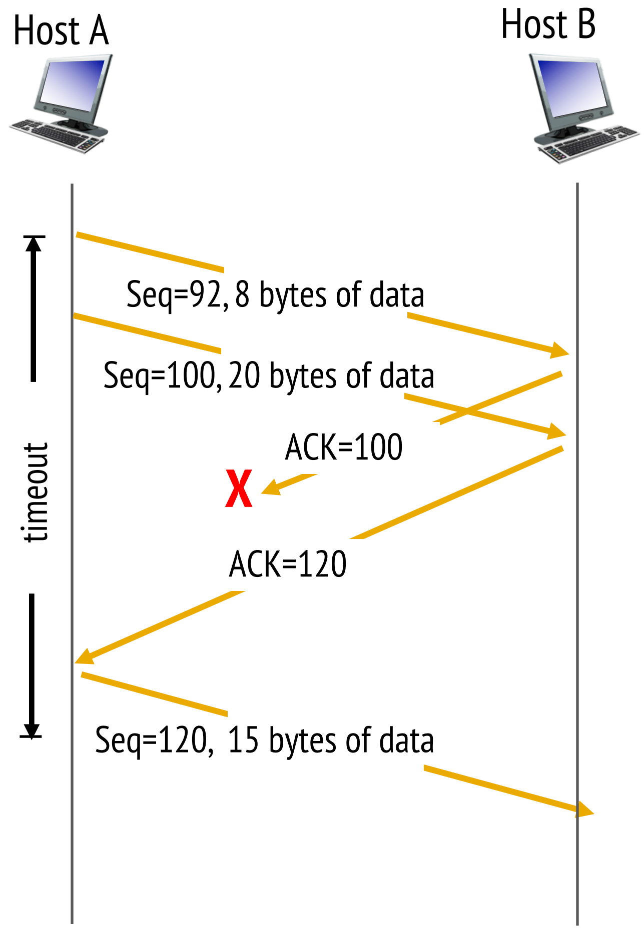


lost ACK scenario



premature timeout

TCP: retransmission scenarios



cumulative ACK

TCP ACK generation [RFC 1122, RFC 2581]

event at receiver	*TCP receiver action*
arrival of in-order segment with expected seq #. All data up to expected seq # already ACKed	delayed ACK. Wait up to 500ms for next segment. If no next segment, send ACK
arrival of in-order segment with expected seq #. One other segment has ACK pending	immediately send single cumulative ACK, ACKing both in-order segments
arrival of out-of-order segment higher-than-expect seq. # . Gap detected	immediately send *duplicate ACK,* indicating seq. # of next expected byte
arrival of segment that partially or completely fills gap	immediate send ACK, provided that segment starts at lower end of gap

TCP fast retransmit

- time-out period often relatively long:
  - long delay before resending lost packet
- detect lost segments via duplicate ACKs.
  - sender often sends many segments back-to-back
  - if segment is lost, there will likely be many duplicate ACKs.

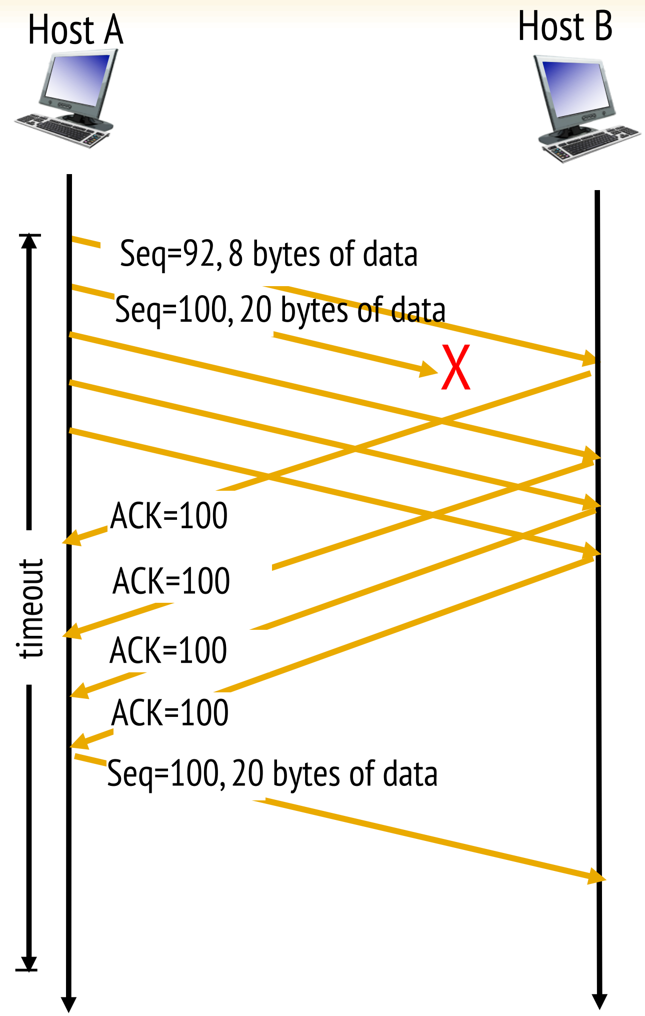
TCP fast retransmit

if sender receives 3 ACKs for same data

("triple duplicate ACKs"), resend unacked segment with smallest seq #

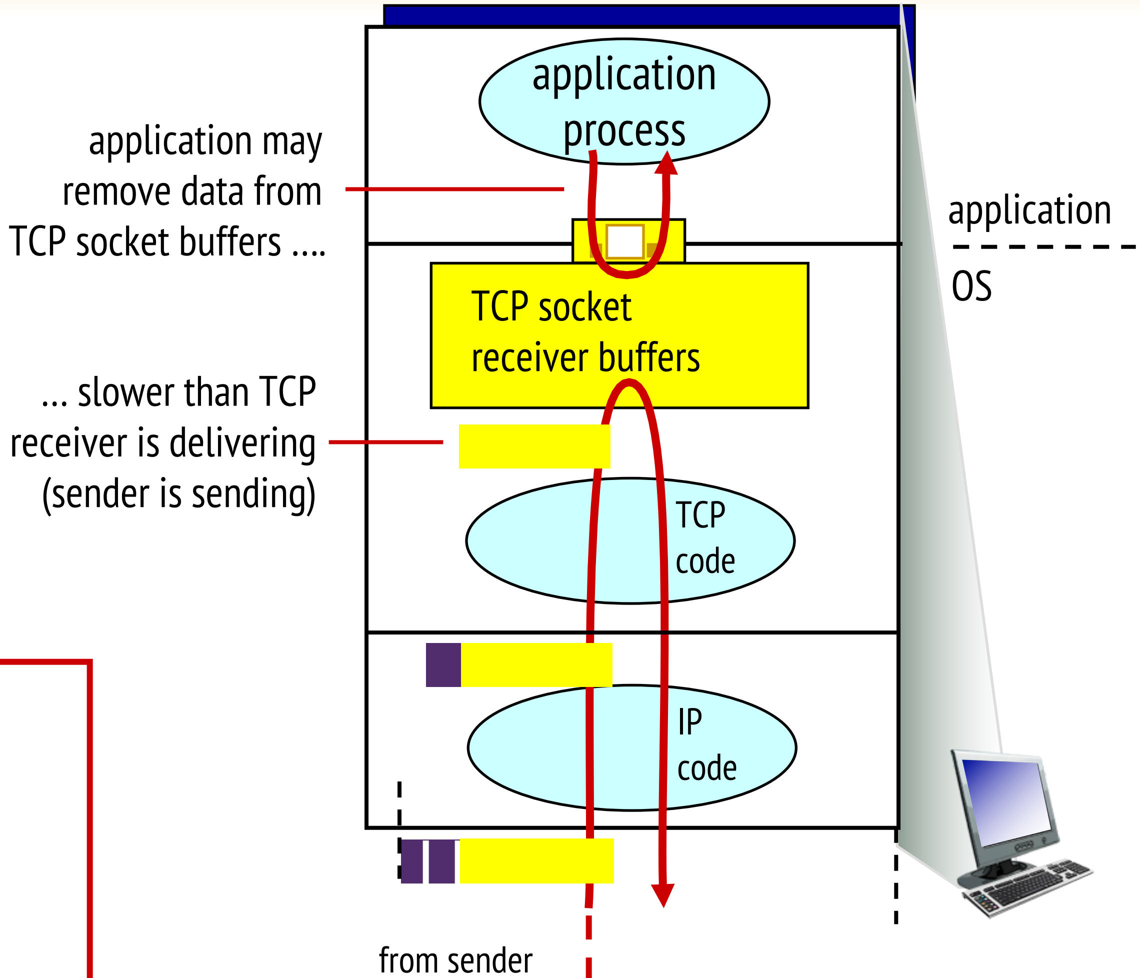
- likely that unacked segment lost, so don't wait for timeout

TCP fast retransmit



fast retransmit after sender receipt of triple duplicate ACK

TCP flow control



receiver protocol stack

flow control

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

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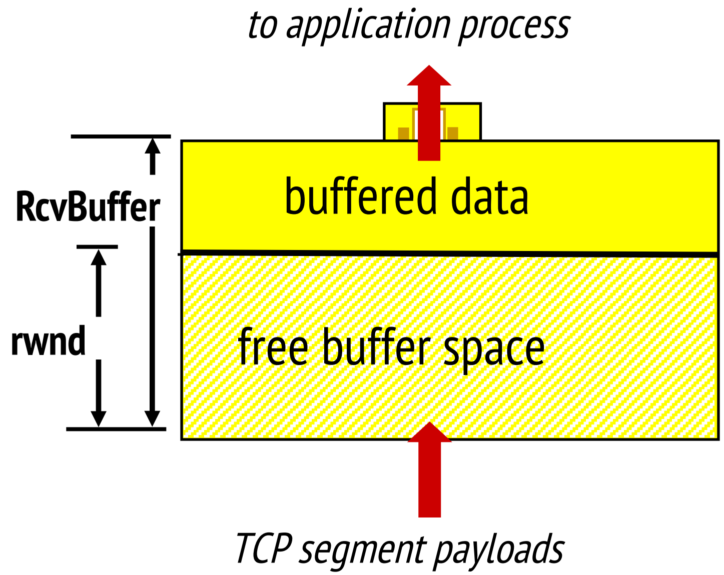
TCP flow control

- receiver "advertises" free buffer space by including **rwnd** (receiver window) value in TCP header of receiver-to-sender segments
  - **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 receiver's **rwnd** value
- guarantees receive buffer will not overflow

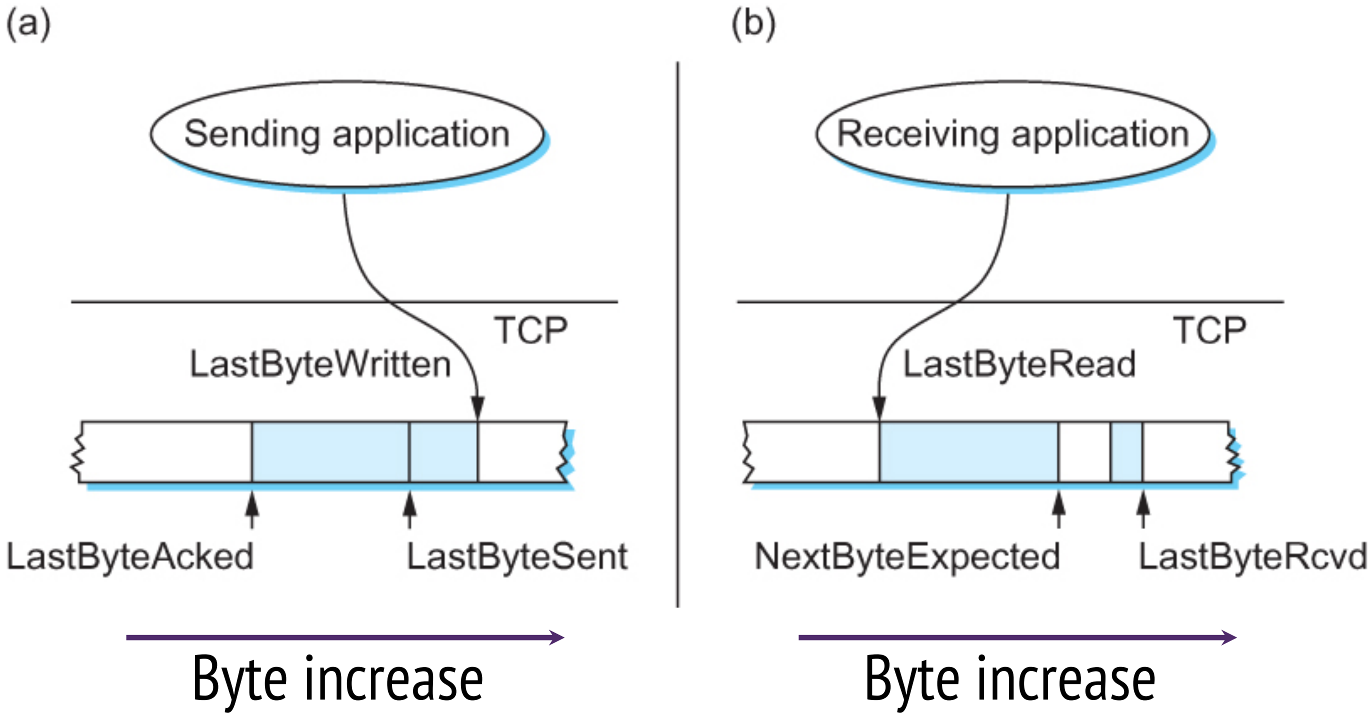


receiver-side buffering

Sliding Window Protocol

- TCP's variant of the sliding window algorithm, which serves several purposes:
  - it guarantees the reliable delivery of data,
  - it ensures that data is delivered in order, and
  - it enforces flow control between the sender and the receiver.

Sliding Window



Relationship between TCP send buffer (a) and receive buffer (b).

TCP Sliding Window

- Sending Side
  - LastByteAcked ≤ LastByteSent
  - LastByteSent ≤ LastByteWritten
- Receiving Side
  - LastByteRead < NextByteExpected
  - NextByteExpected ≤ LastByteRcvd + 1

TCP Flow Control

- LastByteRcvd − LastByteRead ≤ MaxRcvBuffer
- AdvertisedWindow = MaxRcvBuffer − ((NextByteExpected − 1) − LastByteRead)
- LastByteSent − LastByteAcked ≤ AdvertisedWindow
- EffectiveWindow = AdvertisedWindow − (LastByteSent − LastByteAcked)
- LastByteWritten − LastByteAcked ≤ MaxSendBuffer
- If the sending process tries to write y bytes to TCP, but

  (LastByteWritten − LastByteAcked) + y > MaxSendBuffer

  then TCP blocks the sending process and does not allow it to generate more data.

Protecting against Wraparound

- SequenceNum: 32 bits longs
- AdvertisedWindow: 16 bits long
  - TCP has satisfied the requirement of the sliding
  - window algorithm that is the sequence number
  - space be twice as big as the window size
  - $2^{32} >> 2 \times 2^{16}$

Protecting against Wraparound

- Relevance of the 32-bit sequence number space
  - The sequence number used on a given connection might wraparound
  - A byte with sequence number *x* could be sent at one time, and then at a later time a second byte with the same sequence number *x* could be sent
  - Packets cannot survive in the Internet for longer than the **MSL** (maximum segment lifetime)
  - **MSL** is set to 120 sec [recommended RFC 793]
  - Make sure that the sequence number does not wrap around within a 120-second period of time
  - Depends on how fast data can be transmitted over the Internet

Protecting against Wraparound

Bandwidth	Time until Wraparound
T1 (1.5 Mbps)	6.4 hours
Ethernet (10 Mbps)	57 minutes
T3 (45 Mbps)	13 minutes
Fast Ethernet (100 Mbps)	6 minutes
OC-3 (155 Mbps)	4 minutes
OC-12 (622 Mbps)	55 seconds
OC-48 (2.5 Gbps)	14 seconds

Time until 32-bit sequence number space wraps around.

Keeping the Pipe Full

- 16-bit AdvertisedWindow field must be big enough to allow the sender to keep the pipe full
- 16-bit field translates to max 64KB advertised window
- Clearly the receiver is free not to open the window as large as the AdvertisedWindow field allows
- If the receiver has enough buffer space
  - The window needs to be opened far enough to allow a full delay × bandwidth product's worth of data
  - Assuming an RTT of 100 ms

Keeping the Pipe Full

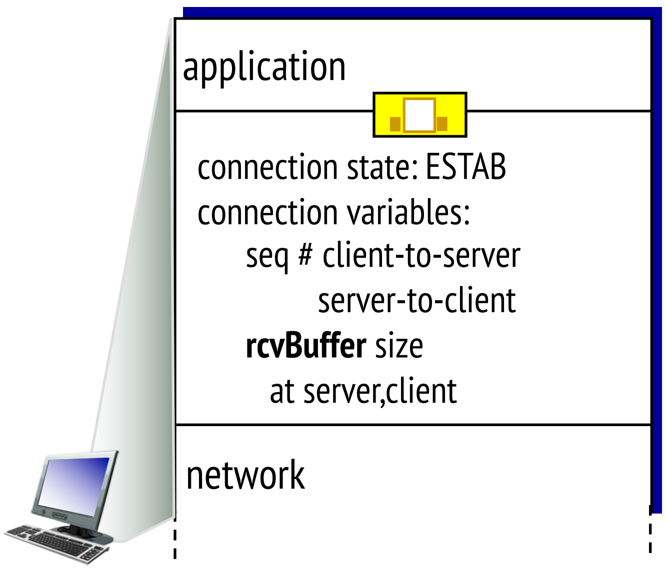
Bandwidth	Delay × Bandwidth Product
T1 (1.5 Mbps)	18 KB
Ethernet (10 Mbps)	122 KB
T3 (45 Mbps)	549 KB
Fast Ethernet (100 Mbps)	1.2 MB
OC-3 (155 Mbps)	1.8 MB
OC-12 (622 Mbps)	7.4 MB
OC-48 (2.5 Gbps)	29.6 MB

Required window size for 100-ms RTT.

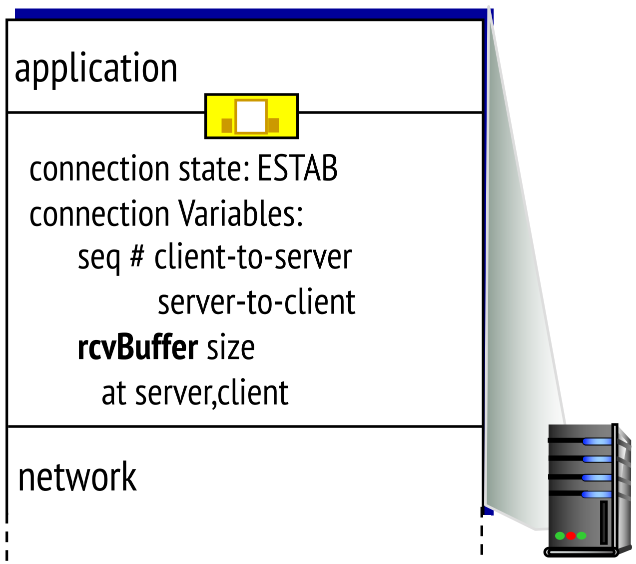
Connection Management

before exchanging data, sender/receiver “handshake”:

- agree to establish connection (each knowing the other willing to establish connection)
- agree on connection parameters

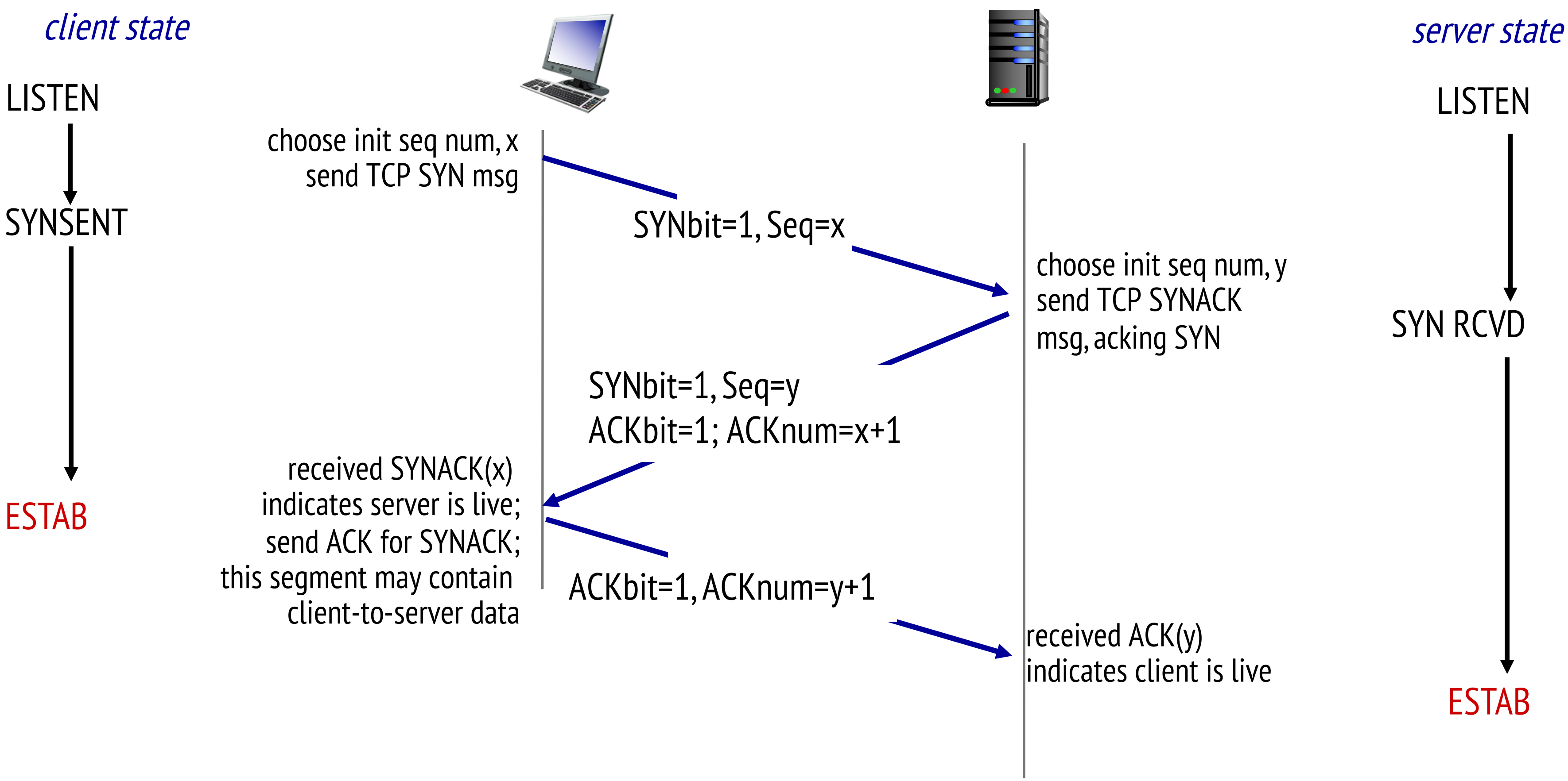


Socket clientSocket =
newSocket("hostname","port number");



Socket connectionSocket =
welcomeSocket.accept();

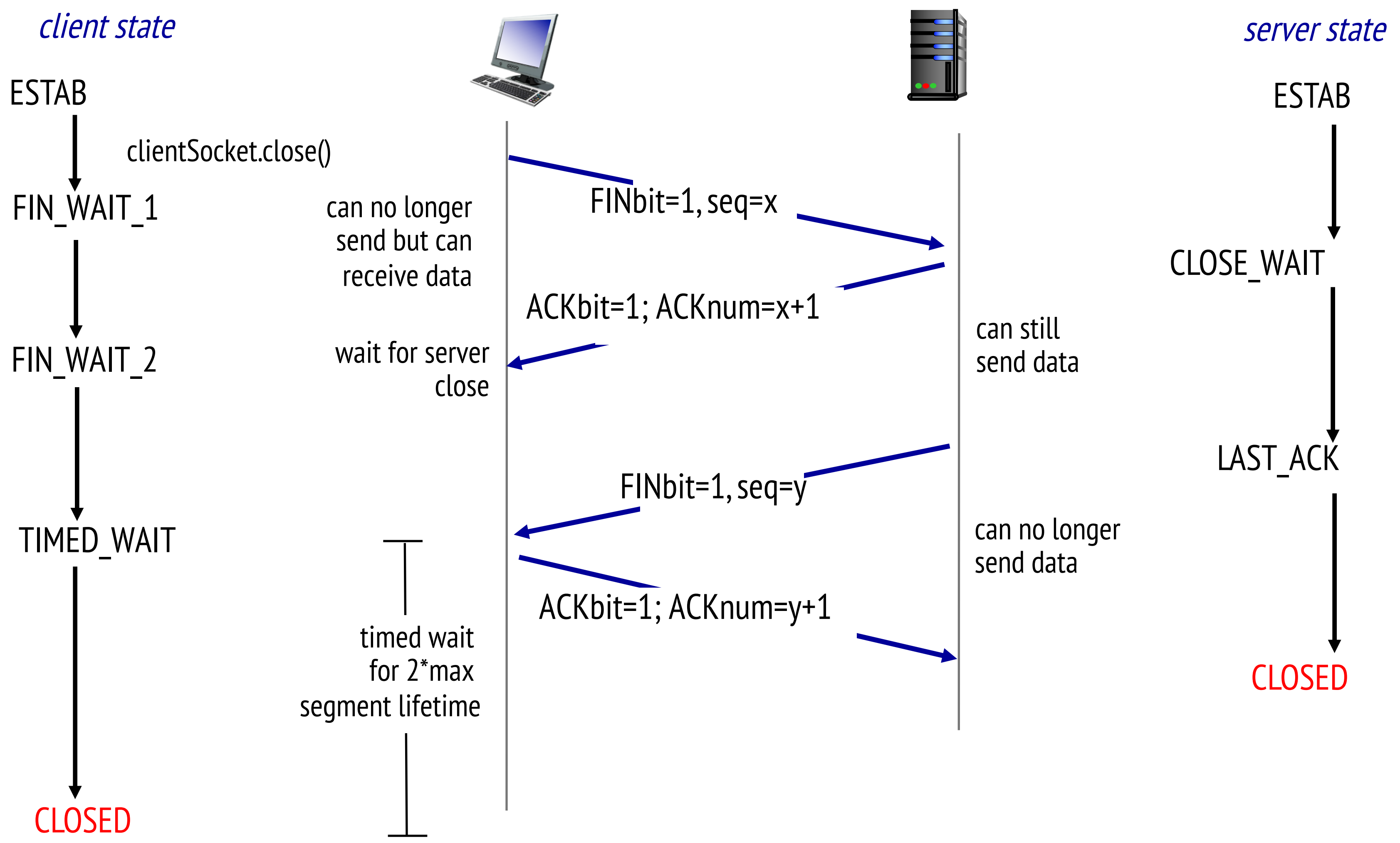
TCP 3-way handshake



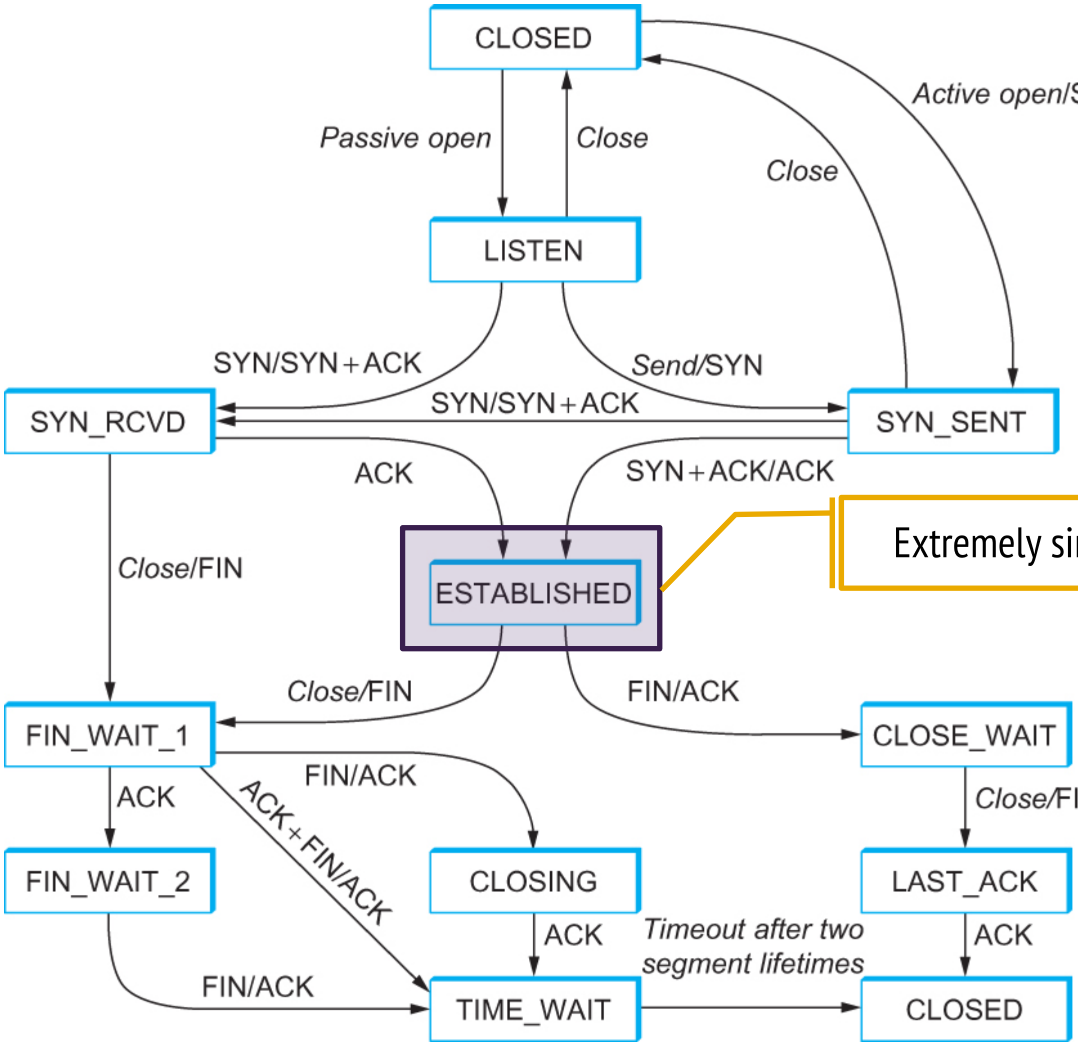
TCP: closing a connection

- 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 State Transition Diagram



Extremely simplified in this diagram

Principles of Congestion Control

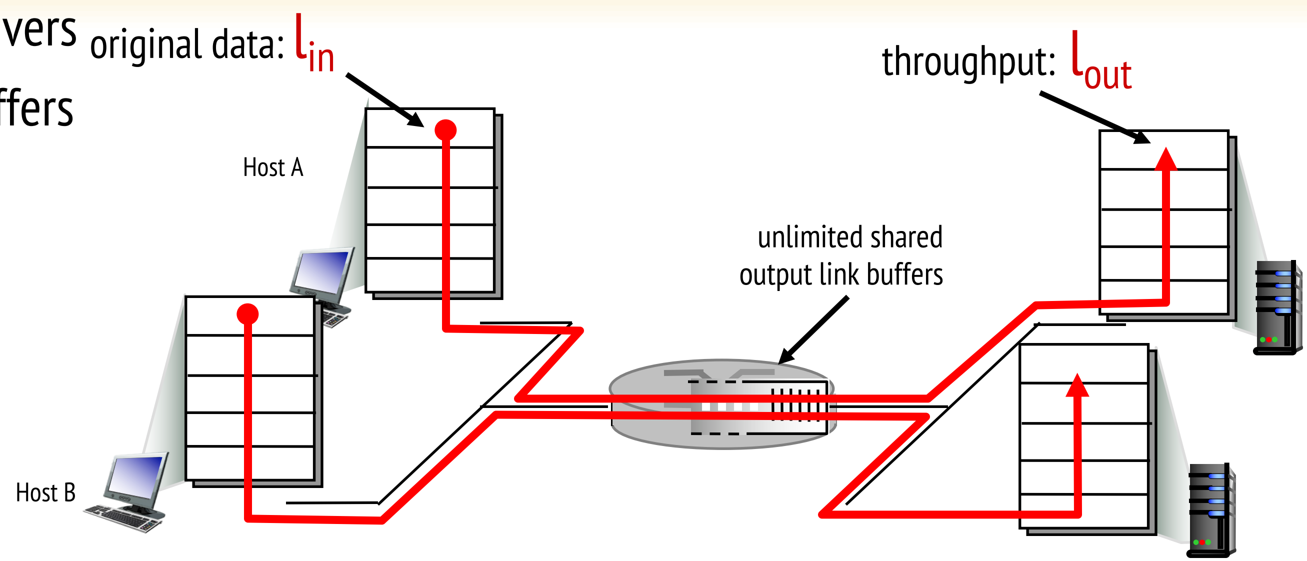
Principles of congestion control

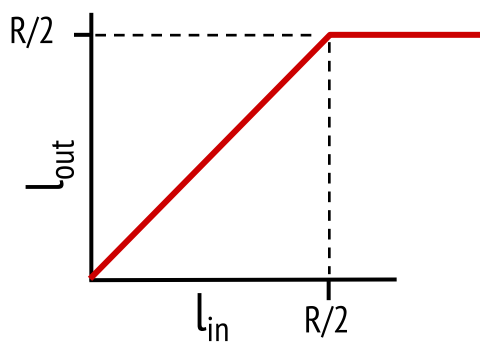
congestion:

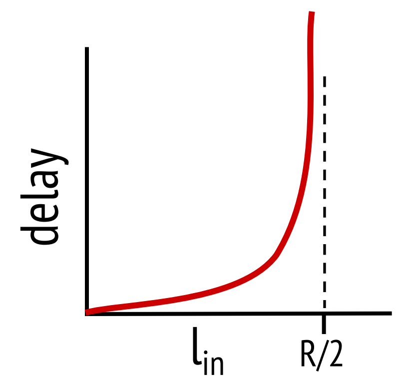
- Informally:
  - "too many sources sending too much data too fast for *network* to handle"
- Different from flow control!
- Manifestations:
  - lost packets (buffer overflow at routers)
  - long delays (queueing in router buffers)
- a top-10 problem!

Causes/costs of congestion: scenario 1

- two senders, two receivers
- one router, infinite buffers
- output link capacity: R
- no retransmission





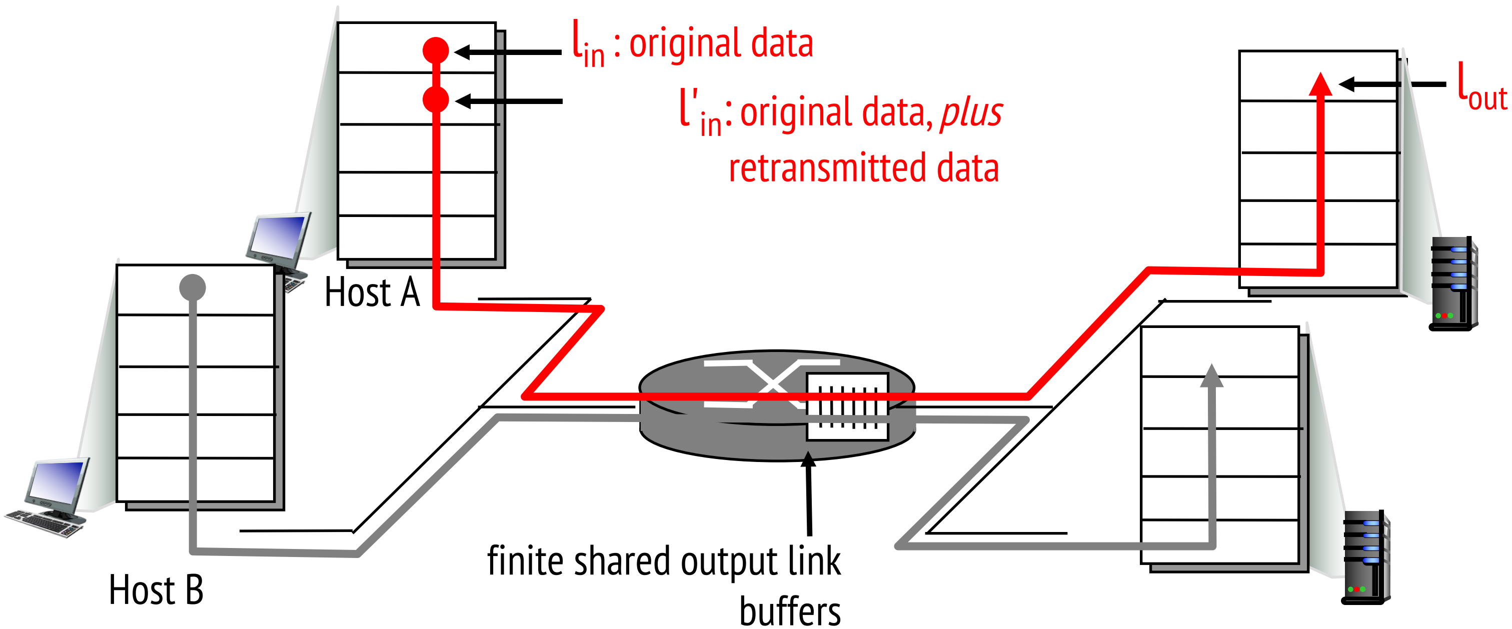


- maximum per-connection throughput: R/2

- large delays as arrival rate, l_{in}, approaches capacity

Causes/costs of congestion: scenario 2

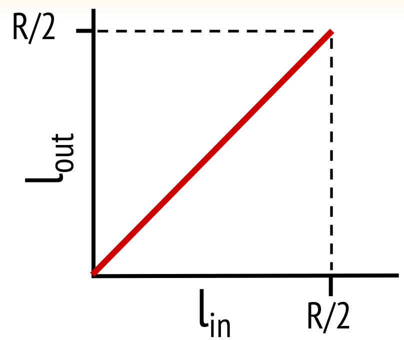
- one router, *finite* buffers
- sender retransmission of timed-out packet
  - application-layer input = application-layer output: $l_{in} = l_{out}$
  - transport-layer input includes *retransmissions* : $l'_{in} \geq l_{in}$

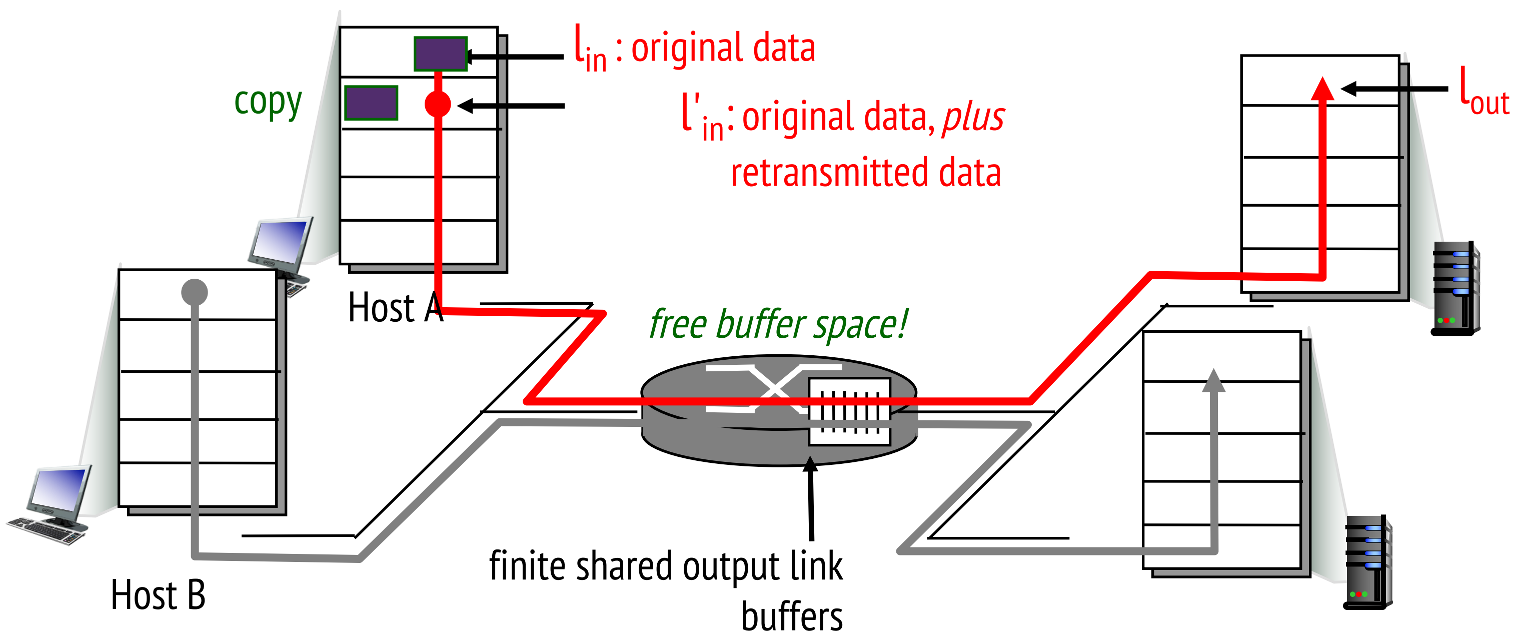


Causes/costs of congestion: scenario 2

idealization: perfect knowledge

- sender sends only when router buffers available

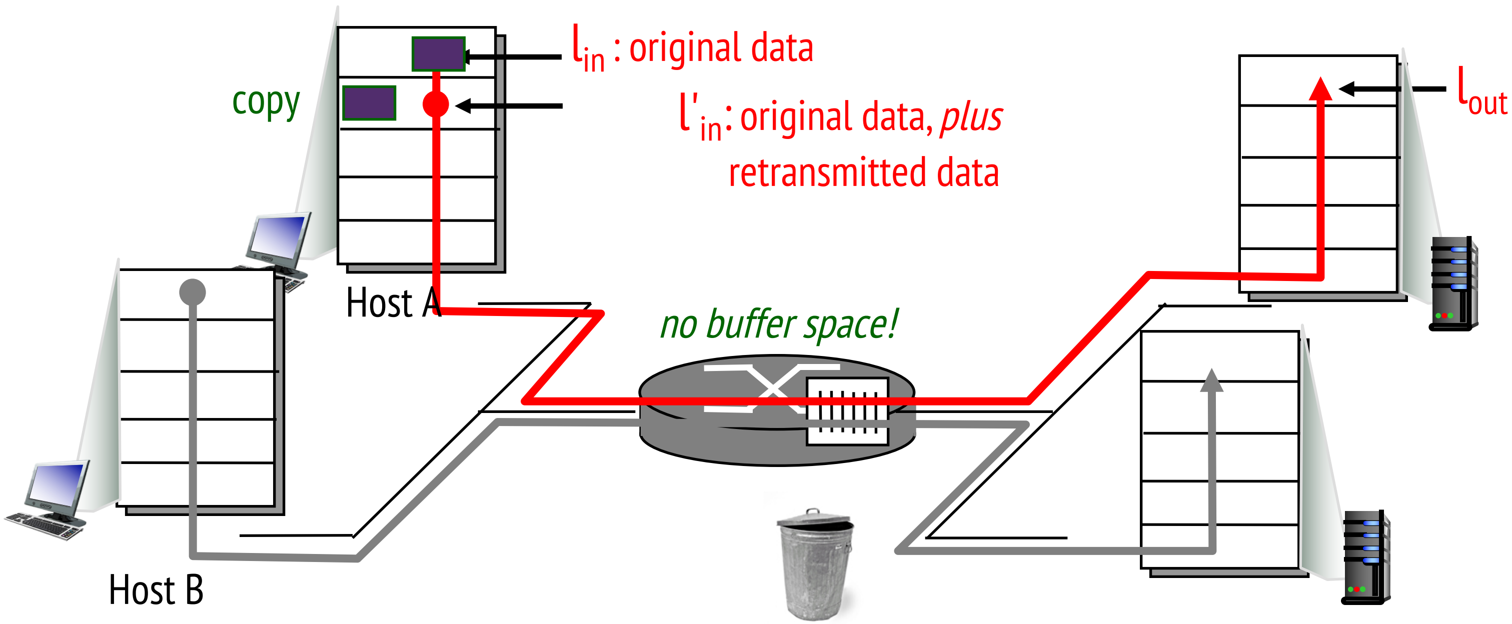




Causes/costs of congestion: scenario 2

Idealization: known loss packets can be lost, dropped at router due to full buffers

- sender only resends if packet *known* to be lost



Causes/costs of congestion: scenario 2

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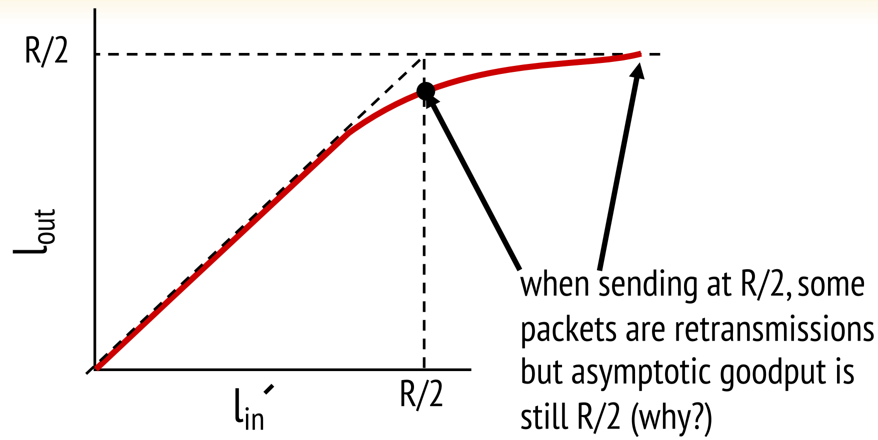
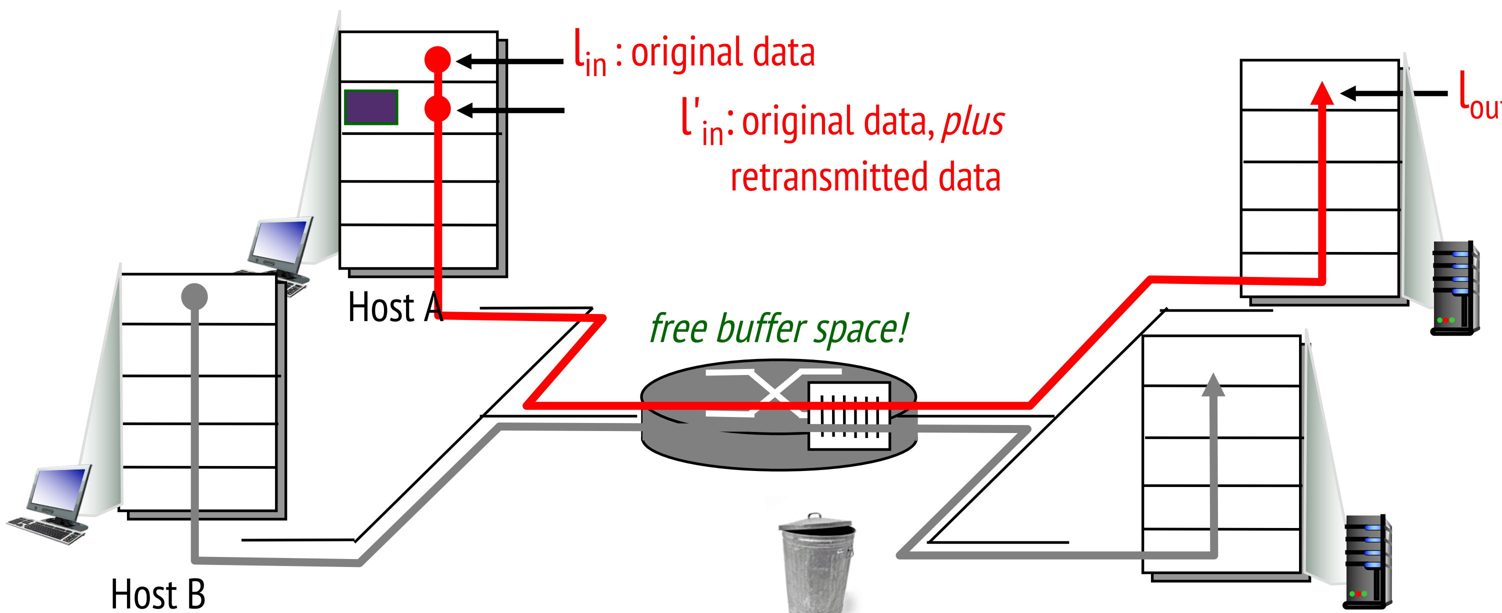
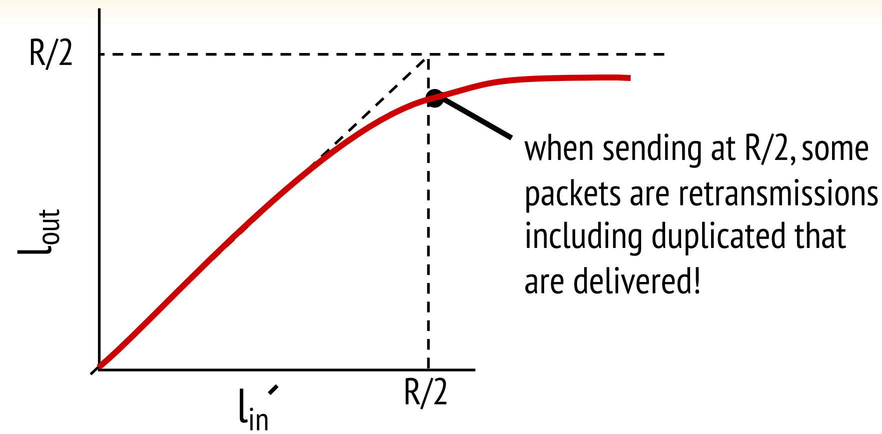


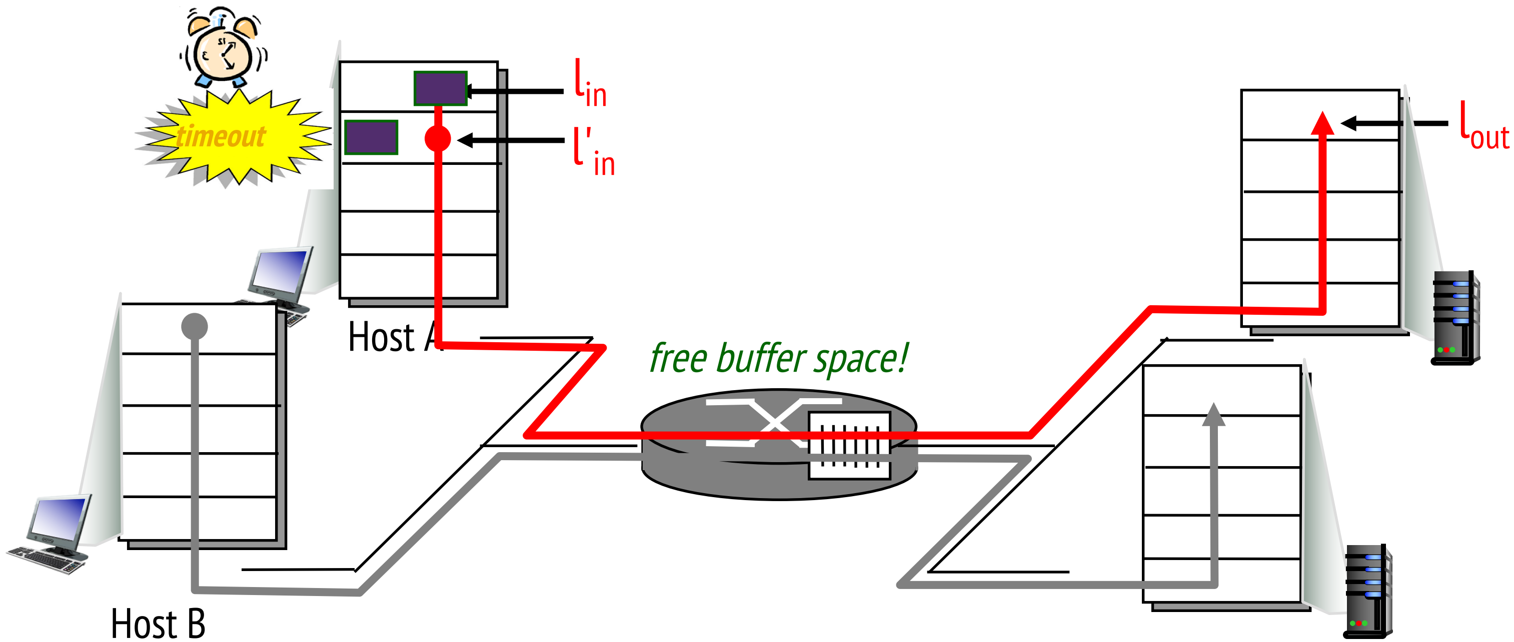


Causes/costs of congestion: scenario 2

Realistic: duplicates

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





Causes/costs of congestion: scenario 2

Realistic: duplicates

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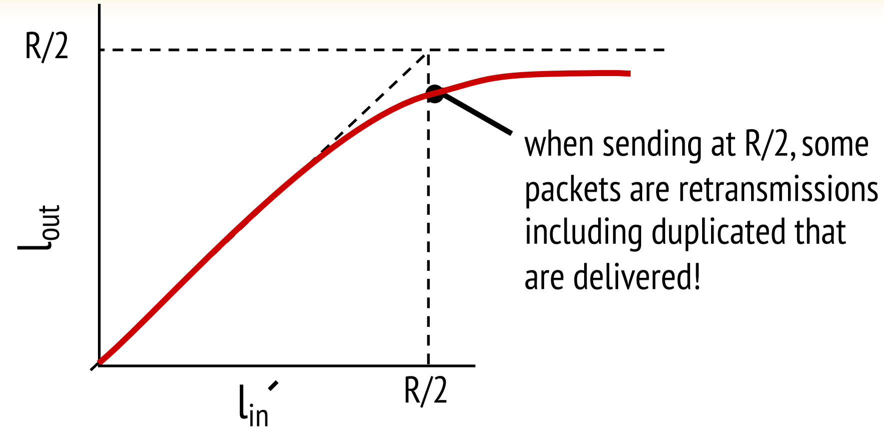


"costs" of congestion:

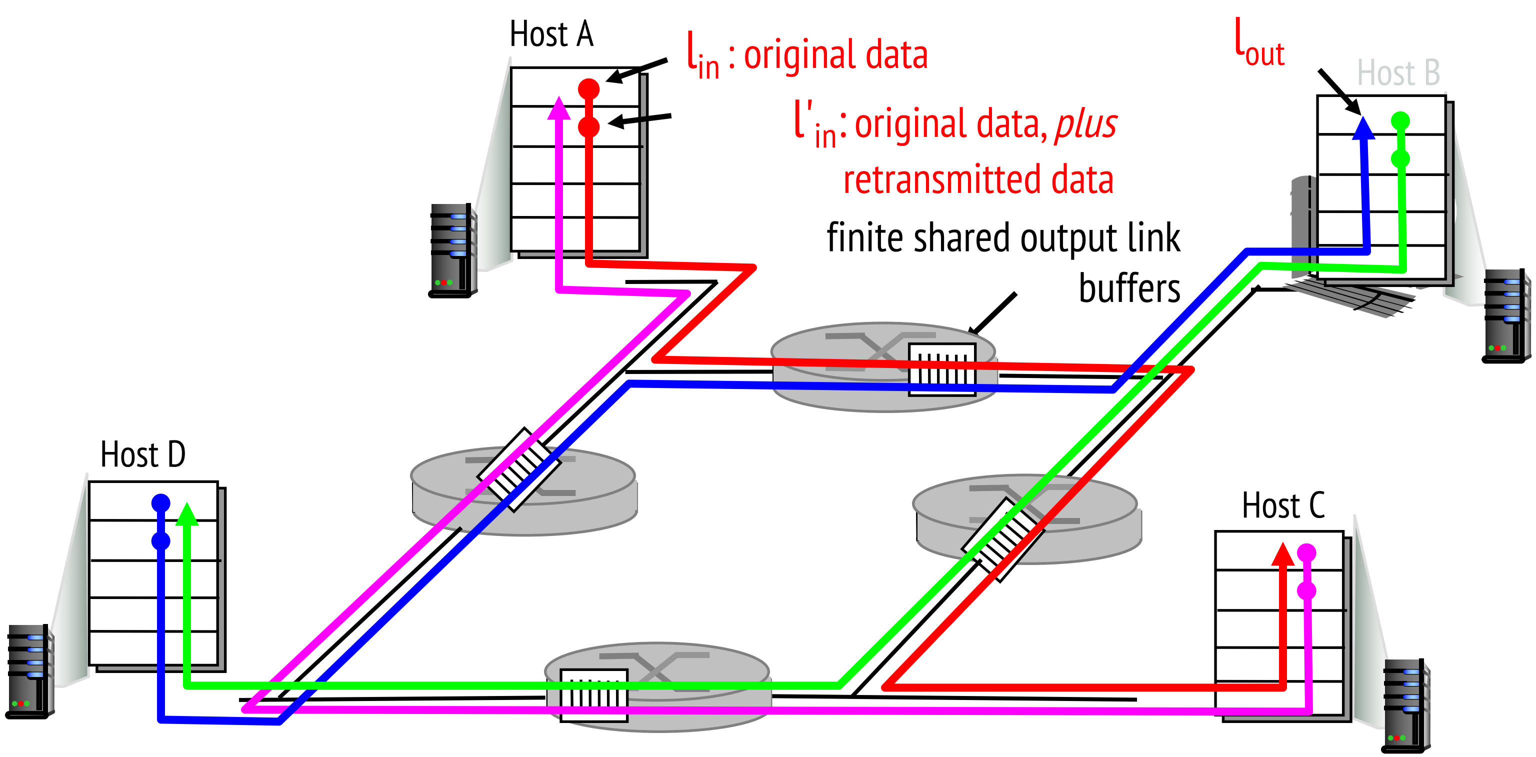
- more work (retransmission) for given "goodput"
- unneeded retransmissions: link carries multiple copies of pkt
  - decreasing goodput

Causes/costs of congestion: scenario 3

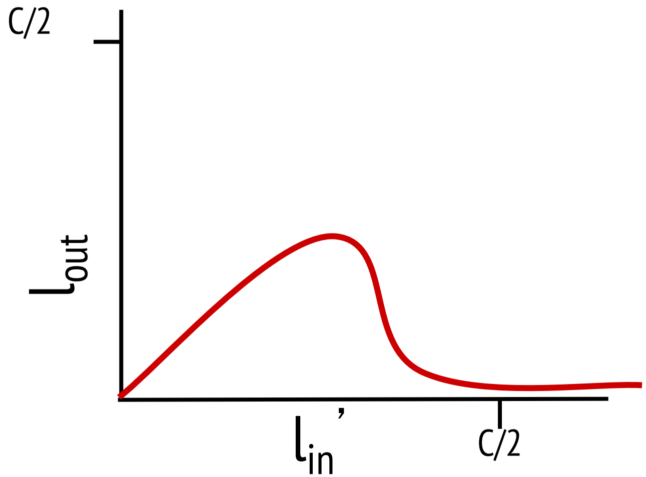
- four senders
- multihop paths
- timeout/retransmit

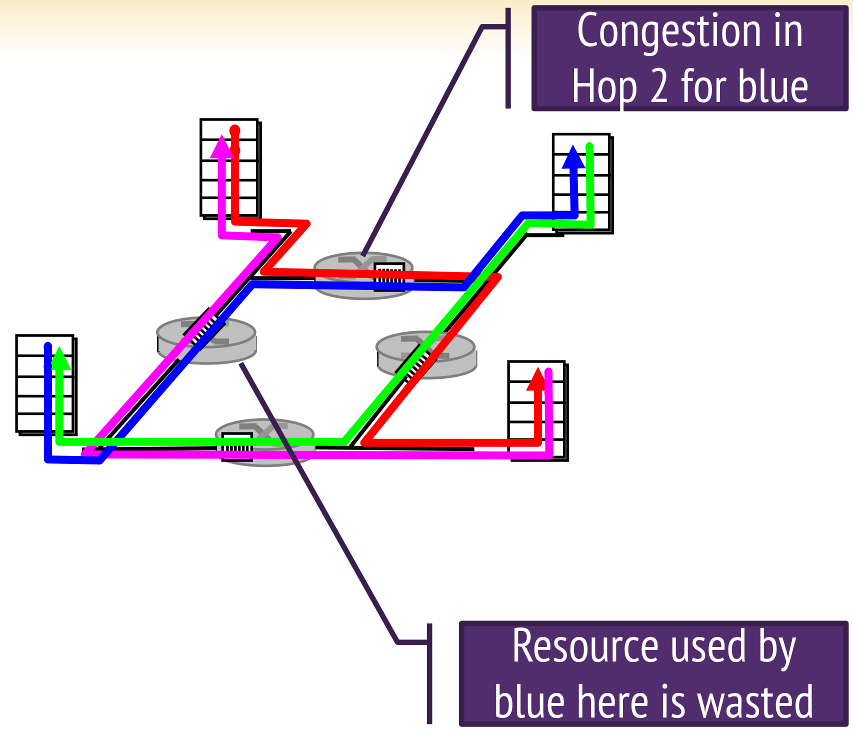
Q: what happens as l_{in} and l_{in}' increase ?

A: as red l_{in}' increases, all arriving blue pkts in queue are dropped, blue throughput goes down



Causes/costs of congestion: scenario 3





another "cost" of congestion:

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

TCP Congestion Control

TCP Congestion Control

- TCP congestion control was introduced into the Internet in the late 1980s by Van Jacobson, roughly eight years after the TCP/IP protocol stack had become operational.

- Immediately preceding this time, the Internet was suffering from congestion collapse–
  - hosts would send their packets into the Internet as fast as the advertised window would allow, congestion would occur at some router (causing packets to be dropped), and the hosts would time out and retransmit their packets, resulting in even more congestion

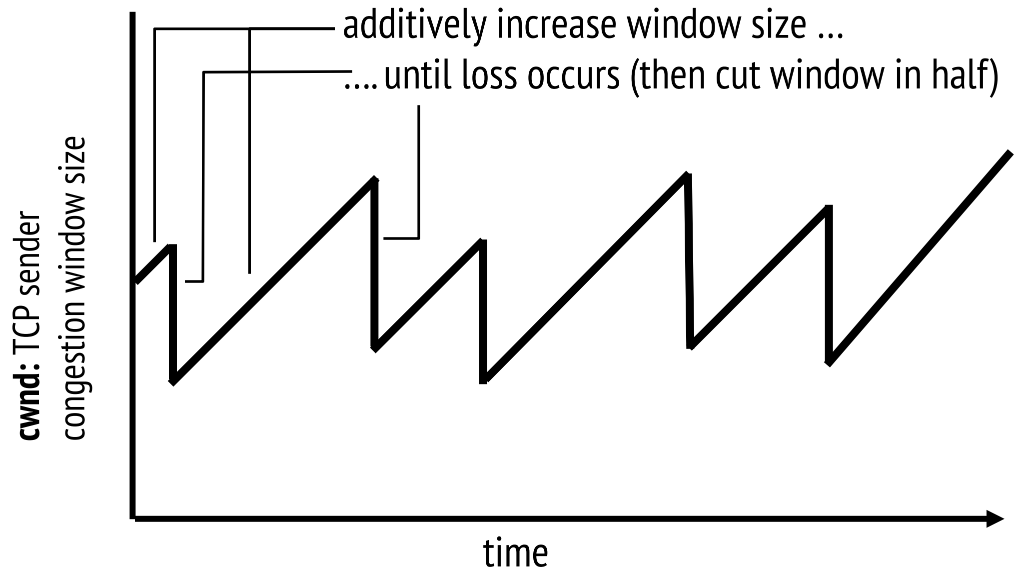
Congestion Window

- TCP maintains a new state variable for each connection, called *CongestionWindow*, which is used by the source to limit how much data it is allowed to have in transit at a given time.
- The congestion window is congestion control's counterpart to flow control's advertised window.
- TCP is modified such that the maximum number of bytes of unacknowledged data allowed is now the minimum of the congestion window and the advertised window

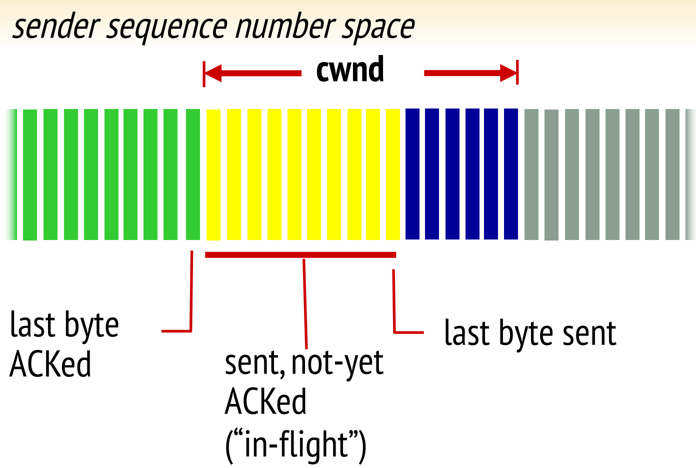
TCP congestion control

- Additive Increase Multiplicative Decrease
- *approach:* sender increases transmission rate (window size), probing for usable bandwidth, until loss occurs
  - *additive increase:* increase **cwnd** by 1 MSS every RTT until loss detected
  - *multiplicative decrease*: cut **cwnd** in half after loss

AIMD saw tooth behavior: probing for bandwidth



TCP Congestion Control: details



- sender limits transmission:

LastByteSent – LastByteAcked <= cwnd

- **cwnd** is dynamic, function of perceived network congestion

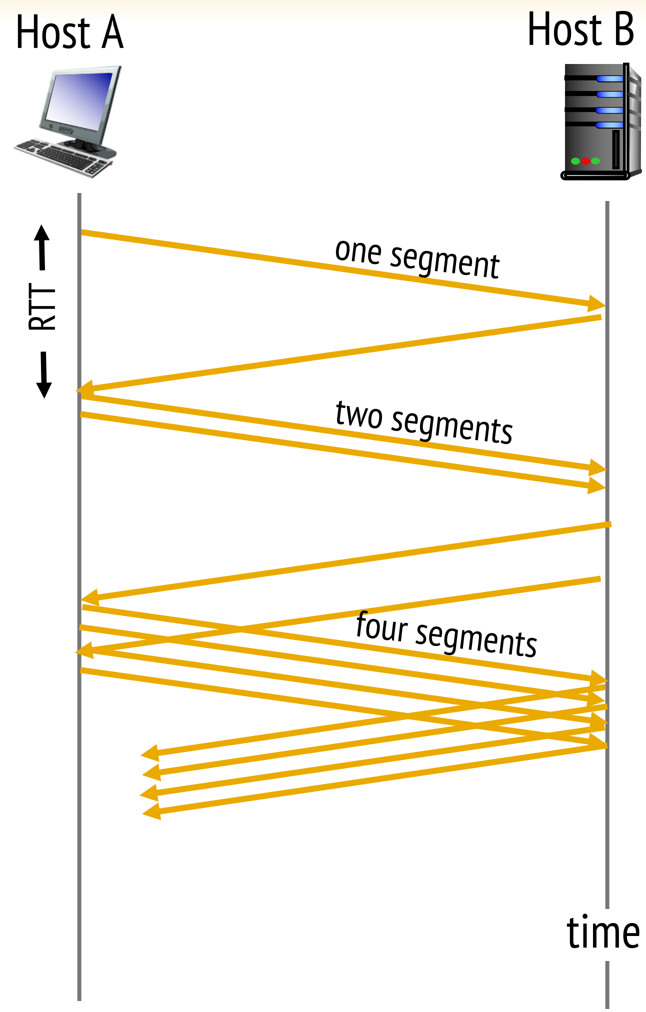
TCP sending rate:

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

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

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

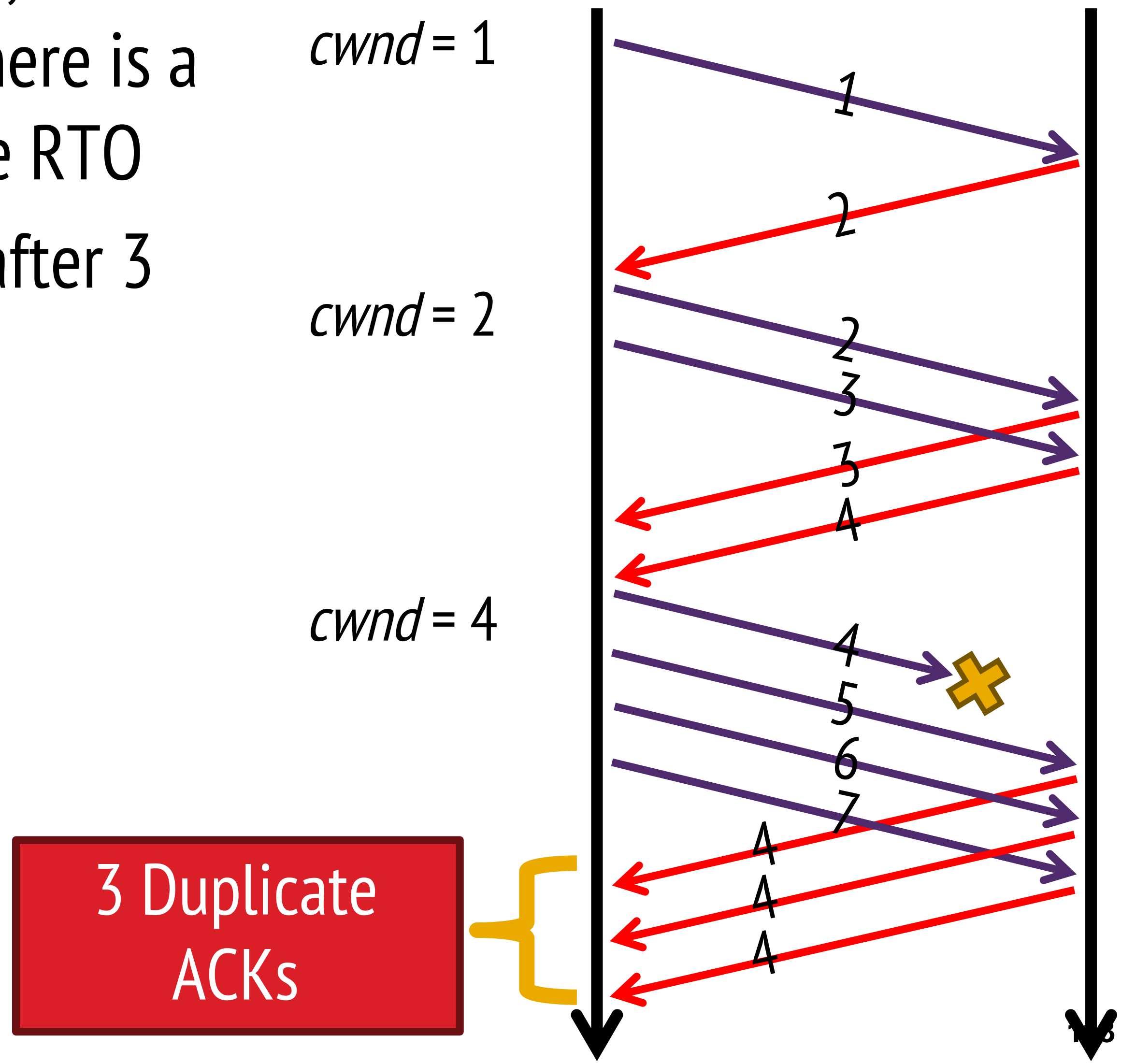
- loss indicated by timeout:
  - **cwnd** set to 1 MSS;
  - window then grows exponentially (as in slow start) to threshold, then grows linearly
- loss indicated by 3 duplicate ACKs: TCP RENO
  - dup ACKs indicate network capable of delivering some segments
  - **cwnd** is cut in half window then grows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

The Evolution of TCP

- Thus far, we have discussed TCP Tahoe
  - Original version of TCP
- However, TCP was invented in 1974!
  - Today, there are many variants of TCP
- Early, popular variant: TCP Reno (1990)
  - Tahoe features, plus…
  - Fast retransmit
    - 3 duplicate ACKs? -> retransmit (don't wait for RTO)
  - Fast recovery
    - On loss: set cwnd = cwnd/2 (ssthresh = new cwnd value)

TCP Reno: Fast Retransmit

- Problem: in Tahoe, if segment is lost, there is a long wait until the RTO
- Reno: retransmit after 3 duplicate ACKs



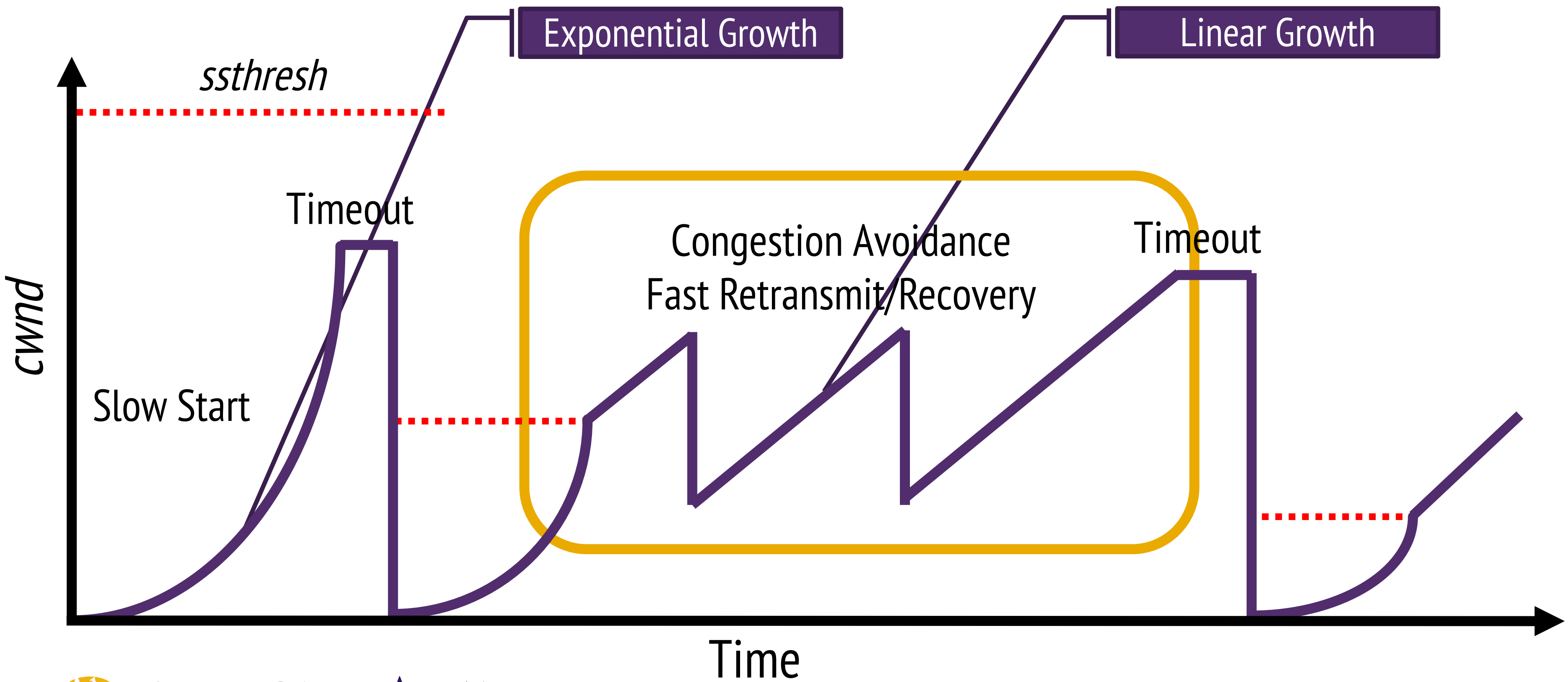
TCP Reno: Fast Recovery

- After a fast-retransmit set *cwnd* to *cwnd/2*
  - Also reset (slow start threshold) ssthresh to the new halved cwnd value
  - i.e. don't reset *cwnd* to 1
  - Avoid unnecessary return to slow start
  - Prevents expensive timeouts

- But when RTO expires still do *cwnd* = 1
  - Return to slow start, same as Tahoe
  - Indicates packets aren't being delivered at all
  - i.e. congestion must be really bad

Fast Retransmit and Fast Recovery

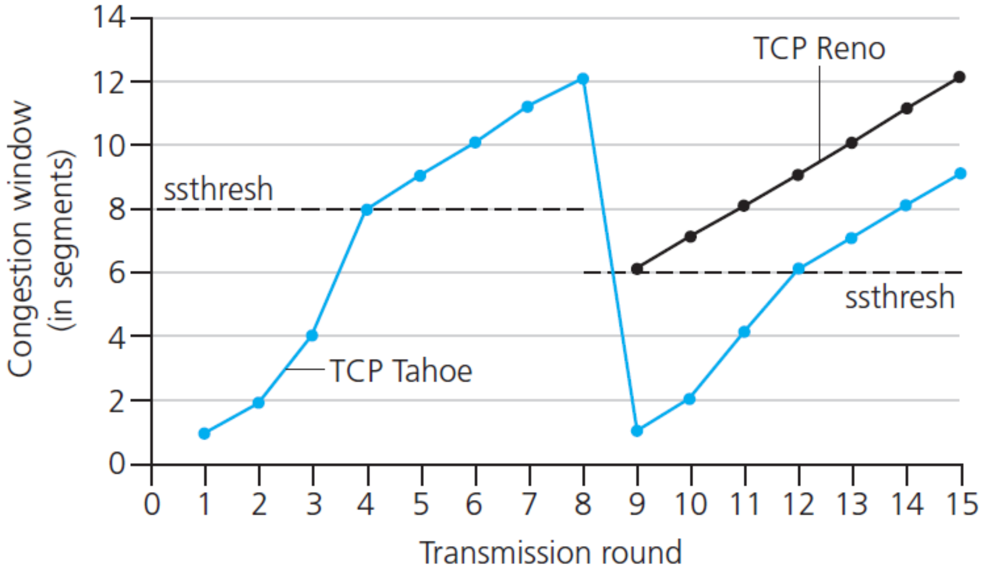
- At steady state, *cwnd* oscillates around the optimal window size
- TCP always forces packet drops



TCP: switching from slow start to CA

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

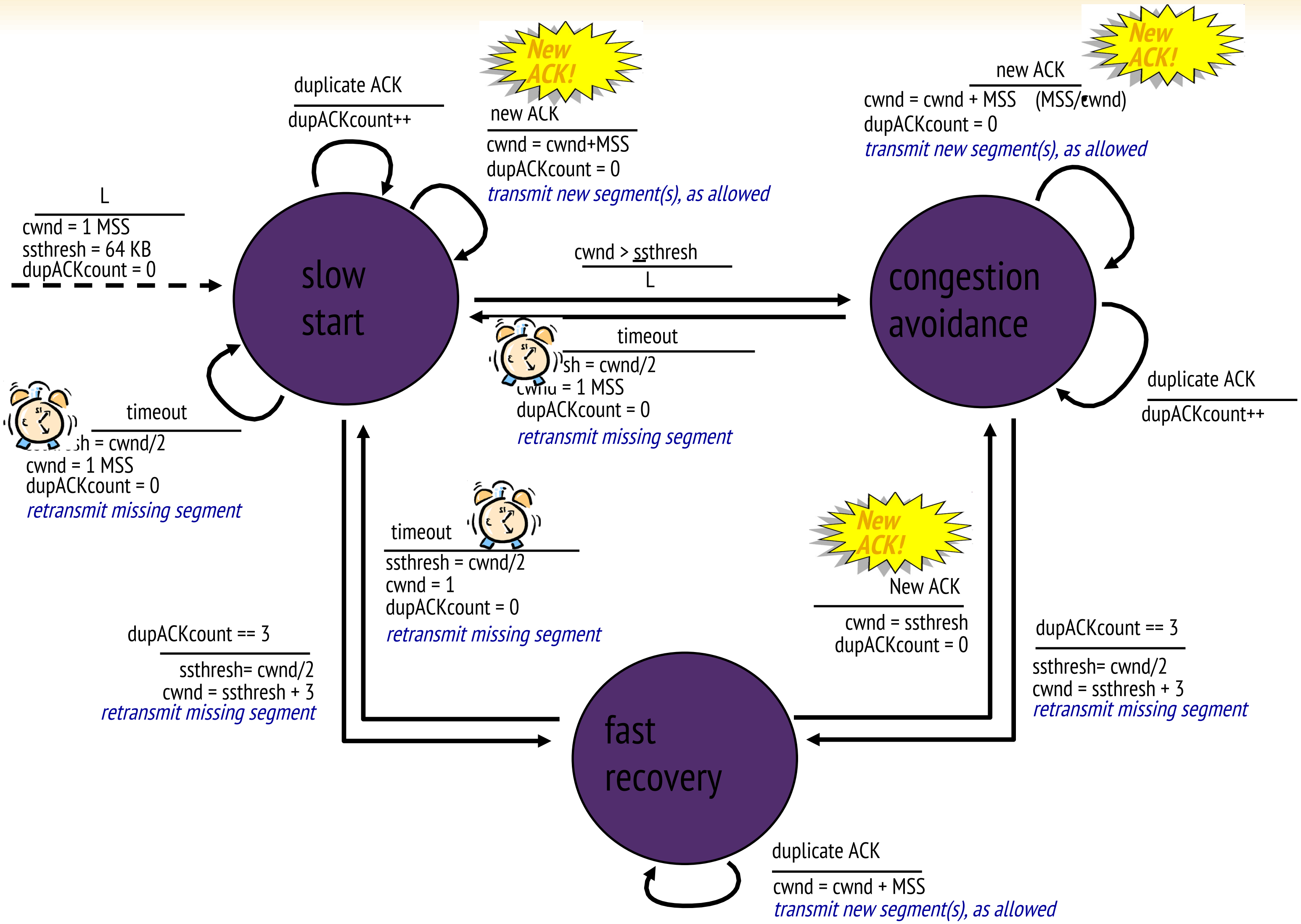
Many TCP Variants…

- Tahoe: the original
  - Slow start with (Additive Increase Multiplicative Decrease) AIMD
  - Dynamic RTO based on RTT estimate
- Reno:
  - fast retransmit (3 dupACKs)
  - fast recovery (cwnd = cwnd/2 on loss)
- NewReno: improved fast retransmit
  - Each duplicate ACK triggers a retransmission
  - Problem: >3 out-of-order packets causes pathological retransmissions
- Vegas: delay-based congestion avoidance
- And many, many, many more…

TCP in the Real World

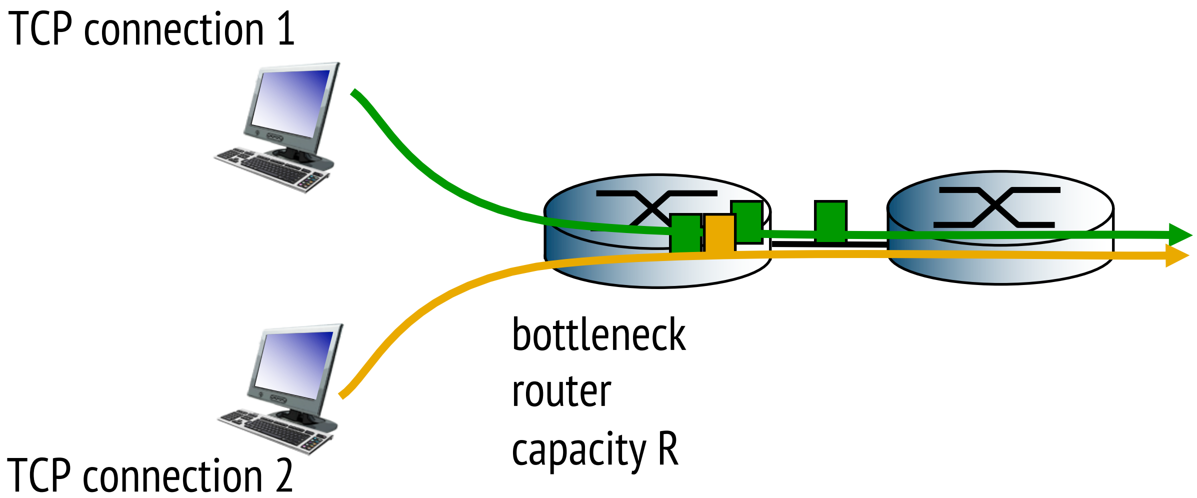
- What are the most popular variants today?
  - Key problem: TCP performs poorly on high bandwidth-delay product networks (like the modern Internet)
  - Compound TCP (Windows)
    - Based on Reno
    - Uses two congestion windows: delay based and loss based
    - Thus, it uses a *compound* congestion controller
  - TCP CUBIC (Linux)
    - Enhancement of BIC (Binary Increase Congestion Control)
    - Window size controlled by cubic function
    - Parameterized by the time *T* since the last dropped packet

Summary: TCP Congestion Control



TCP Fairness

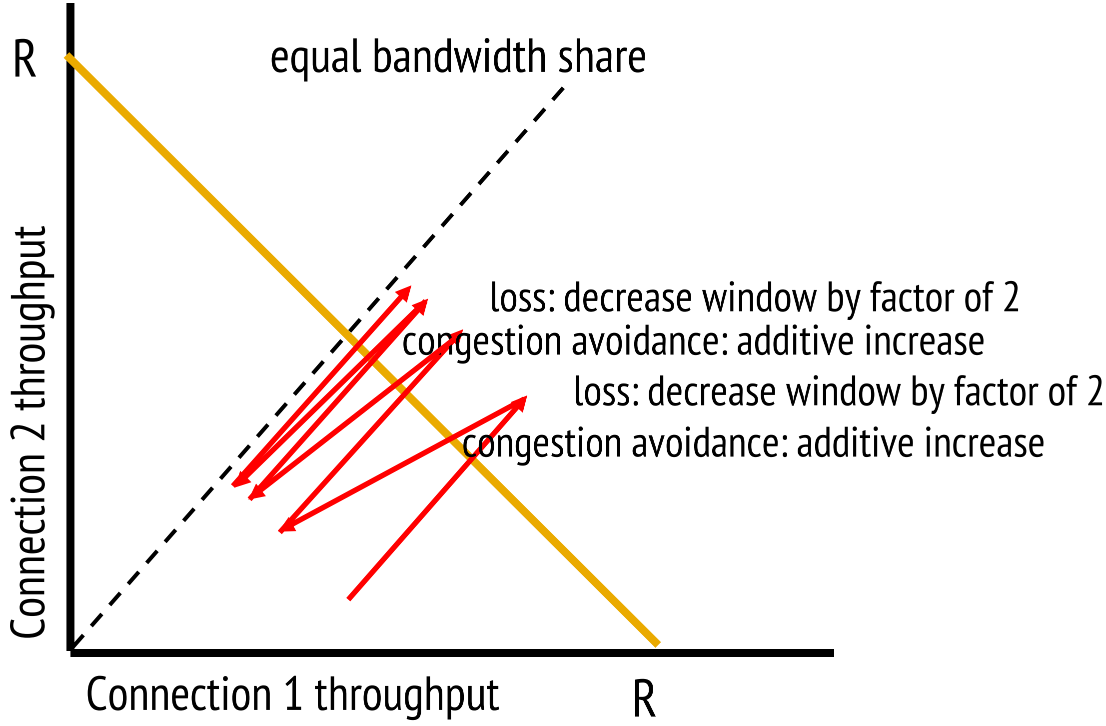
fairness goal: if K TCP sessions share same bottleneck link of bandwidth R, each should have average rate of R/K



Why is TCP fair?

two competing sessions:

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



Fairness (more)

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

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

Summary

- We have discussed
  - how to convert host-to-host packet delivery service to process-to-process communication channel.
  - UDP
  - TCP
  - Flow control
  - Congestion Control