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CSCI 4760 - Computer Networks
Fall 2016

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These slides are adapted from the textbook slides by J.F. Kurose and K.W. Ross

Chapter 3: Transport Layer

Our goals:

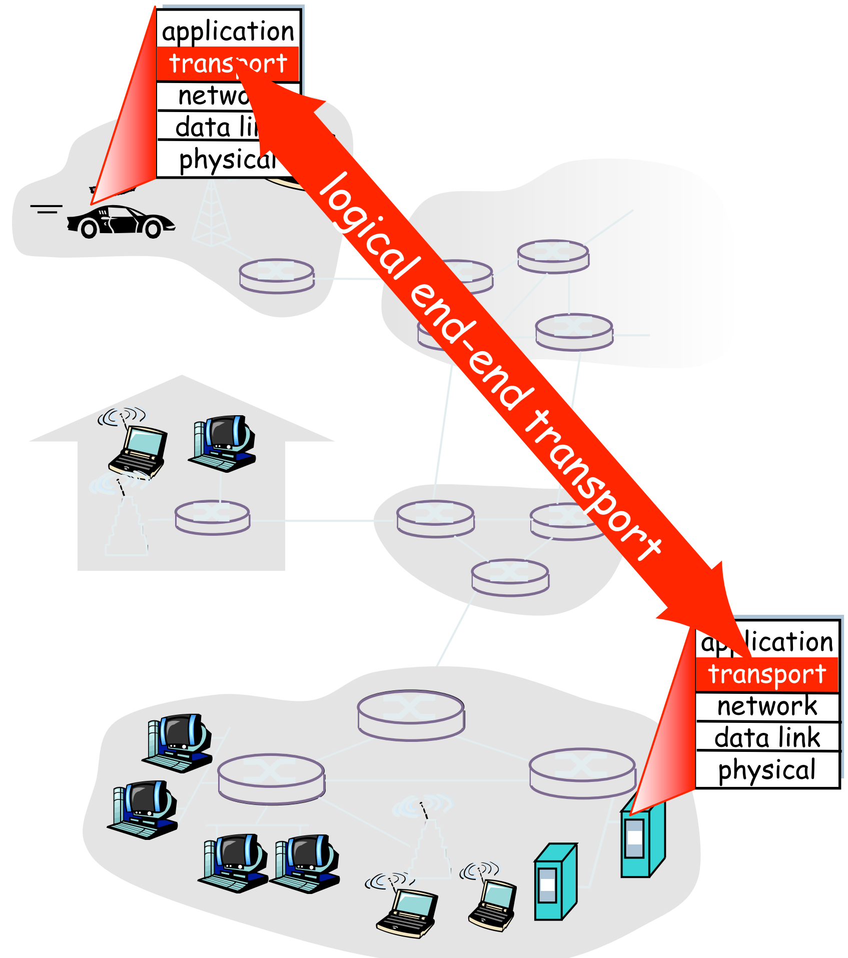
- understand principles behind transport layer services:
  - multiplexing/demultiplexing
  - reliable data transfer
  - flow control
  - congestion control
- learn about transport layer protocols in the Internet:
  - UDP: connectionless transport
  - TCP: connection-oriented transport
  - TCP congestion control

Chapter 3 outline

- 3.1 Transport-layer services
- 3.2 Multiplexing and demultiplexing
- 3.3 Connectionless transport: UDP
- 3.4 Principles of reliable data transfer
- 3.5 Connection-oriented transport: TCP
  - segment structure
  - reliable data transfer
  - flow control
  - connection management
- 3.6 Principles of congestion control
- 3.7 TCP congestion control

Transport services and protocols

- provide *logical communication* between app processes running on different hosts
- transport protocols run in end systems
  - send side: breaks long 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



Transport vs. network layer

- *network layer:* logical communication between hosts
- *transport layer:* logical communication between processes
  - relies on and enhances network layer services

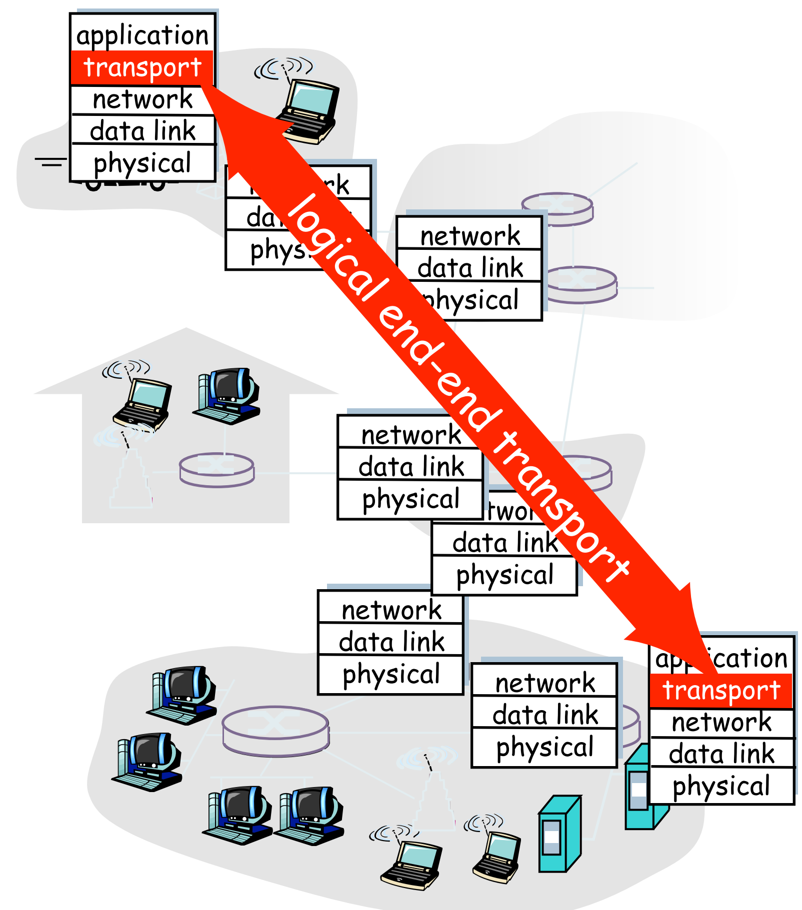
Household analogy:

12 kids sending letters to 12 kids

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

Internet transport-layer protocols

- reliable, in-order delivery (TCP)
  - connection setup
  - flow control
  - congestion control
- unreliable, unordered delivery: UDP
  - extension of "best-effort" IP
- Do ***not*** provide:
  - delay guarantees
  - bandwidth guarantees



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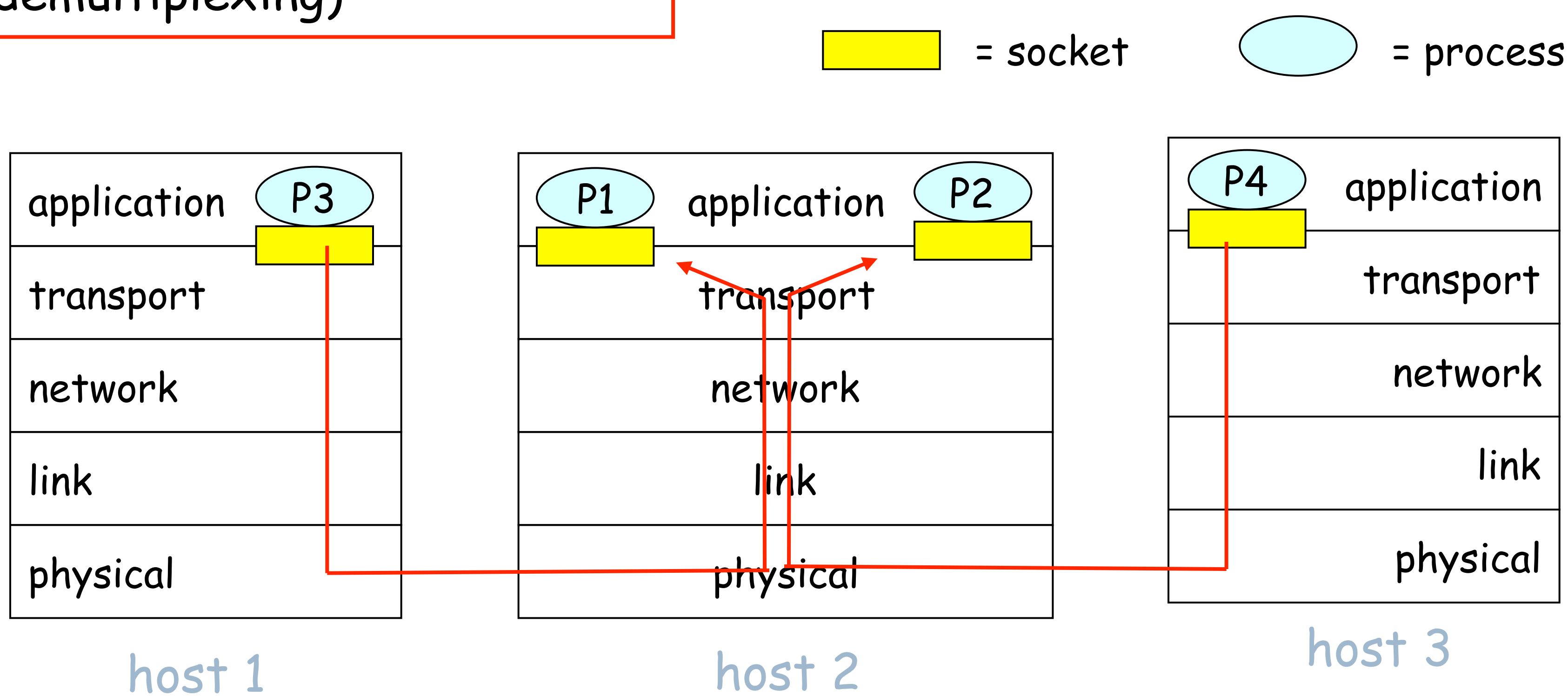
Multiplexing/demultiplexing

Multiplexing at send host:

gathering data from multiple sockets, enveloping data with header (later used for demultiplexing)

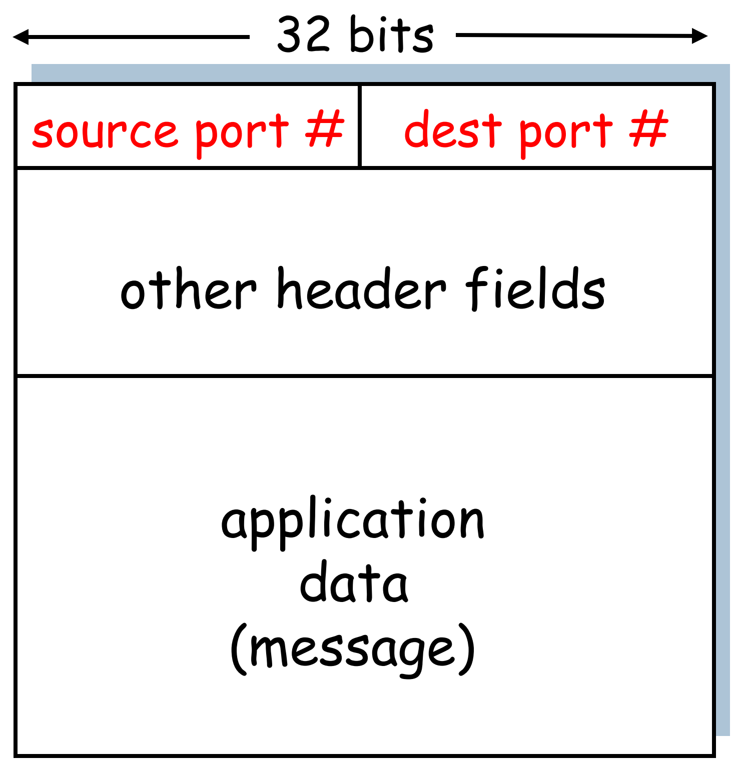
Demultiplexing at rcv host:

delivering received segments to correct socket



How demultiplexing works

- host receives IP datagrams
  - each datagram has source IP address, destination IP address
  - each datagram carries 1 transport-layer segment
  - each segment has source and destination port number
- host uses IP addresses & port numbers to direct segment to appropriate socket



TCP/UDP segment format

Connectionless demultiplexing

- Create sockets with port numbers:

```
DatagramSocket mySocket1 = new
  DatagramSocket(12534);
DatagramSocket mySocket2 = new
  DatagramSocket(12535);
```

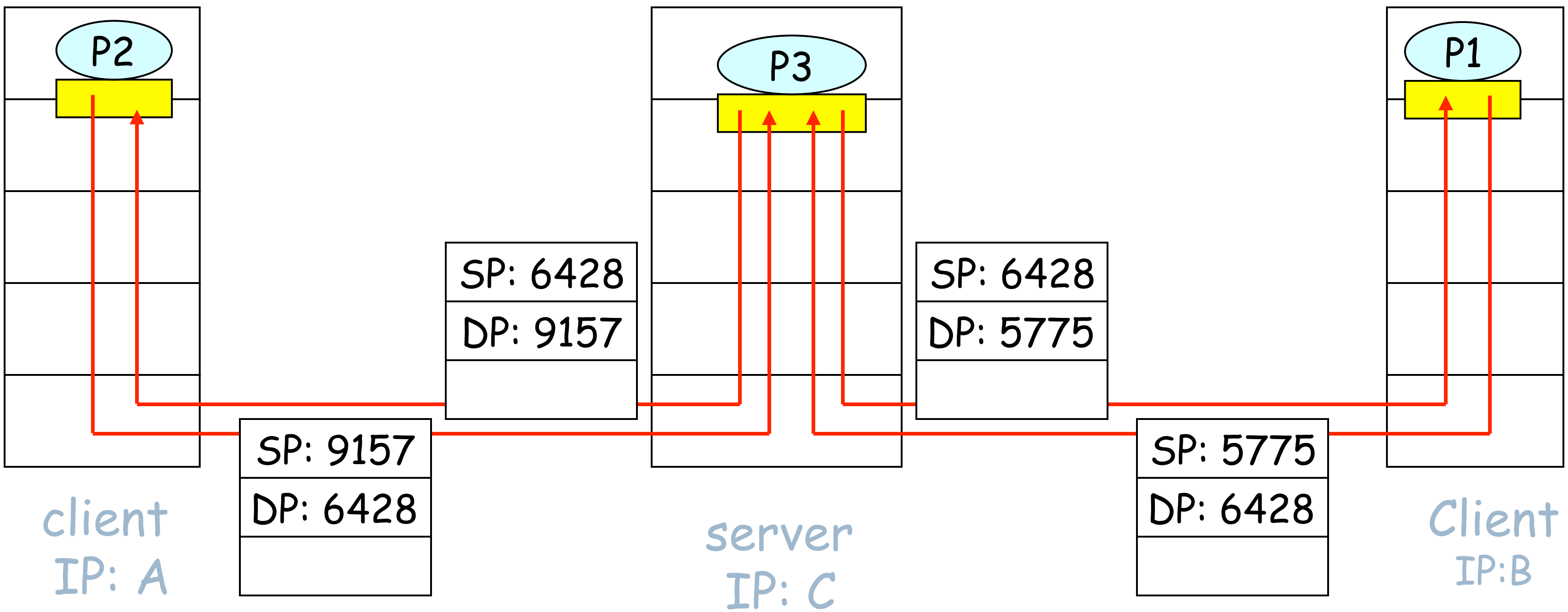
- UDP socket identified by two-tuple:

(dest IP address, dest port number)

- When host receives UDP segment:
  - checks destination port number in segment
  - directs UDP segment to socket with that port number
- IP datagrams with different source IP addresses and/or source port numbers directed to same socket

Connectionless demux (cont)

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

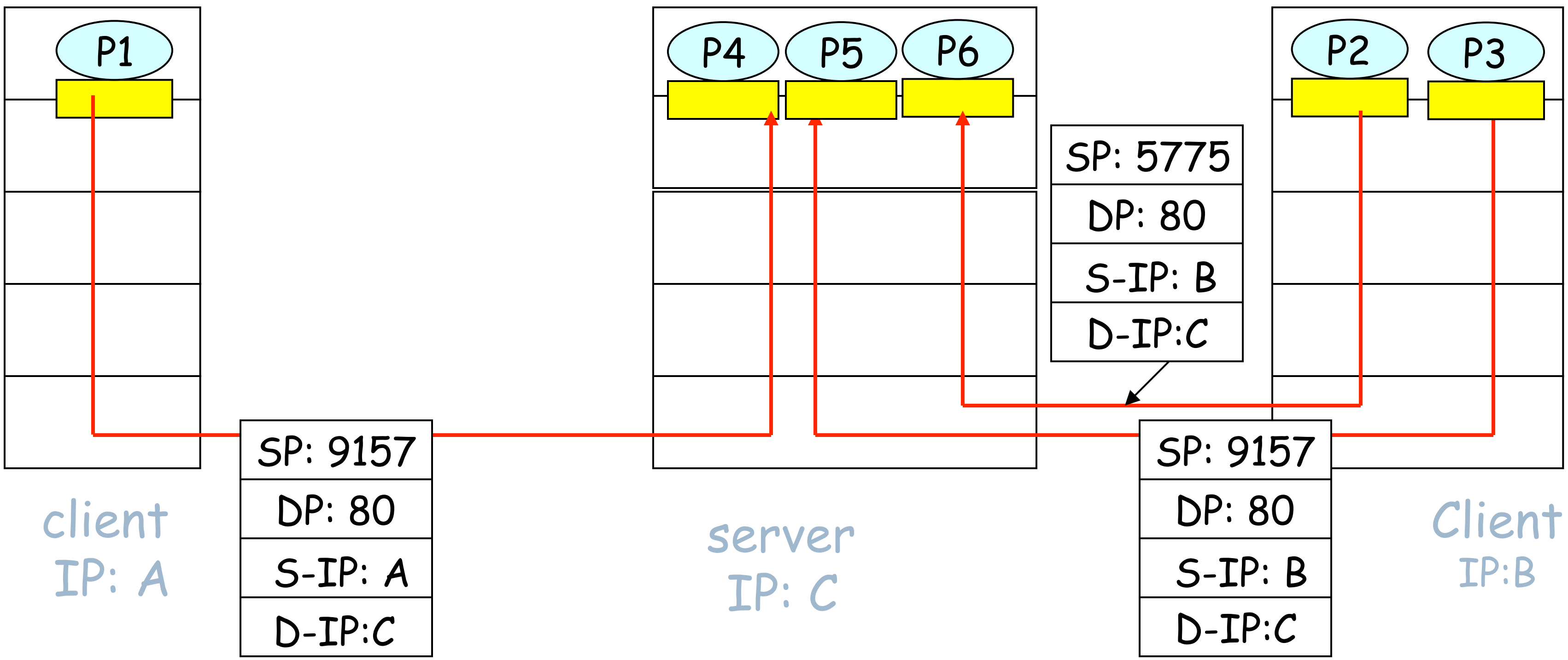


SP provides "return address"

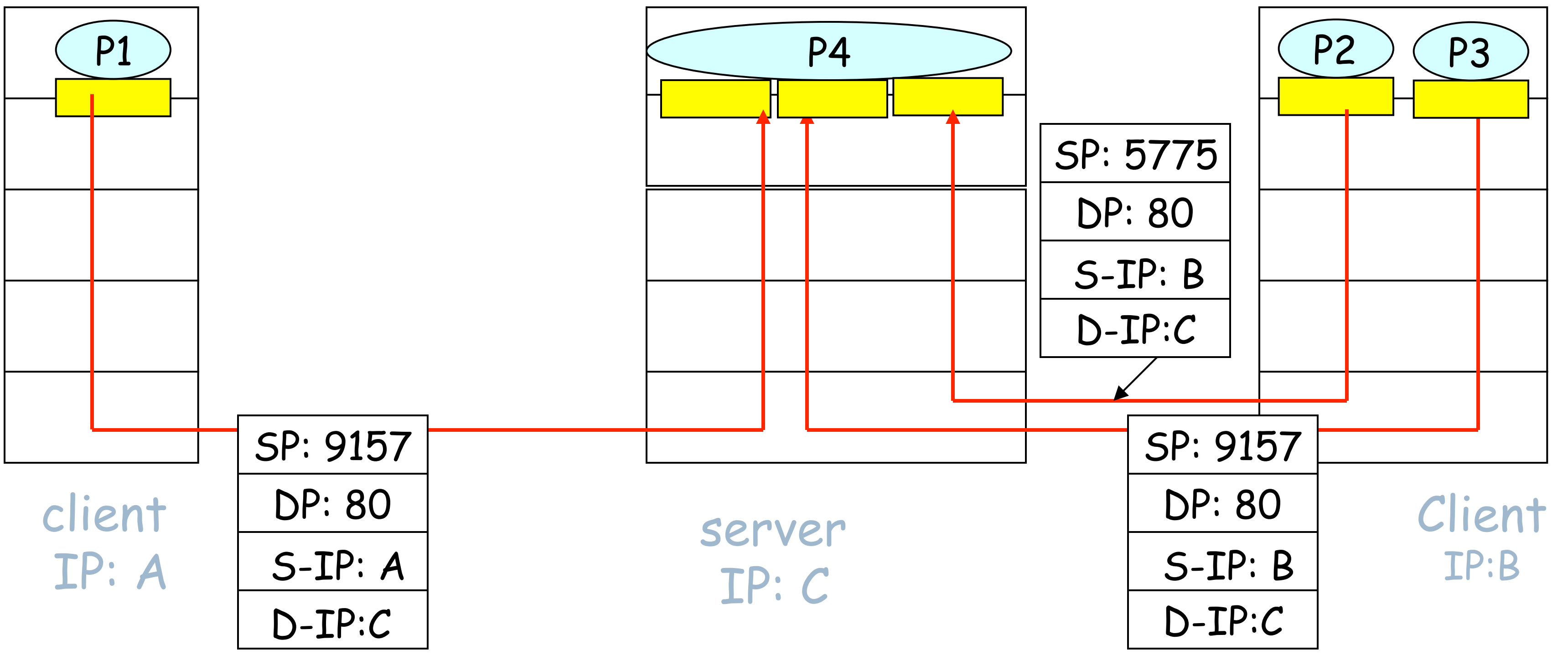
Connection-oriented demux

- TCP socket identified by 4-tuple:
  - source IP address
  - source port number
  - dest IP address
  - dest port number
- receiving host 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 (cont)



Connection-oriented demux: Threaded Web Server



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UDP: User Datagram Protocol [RFC 768]

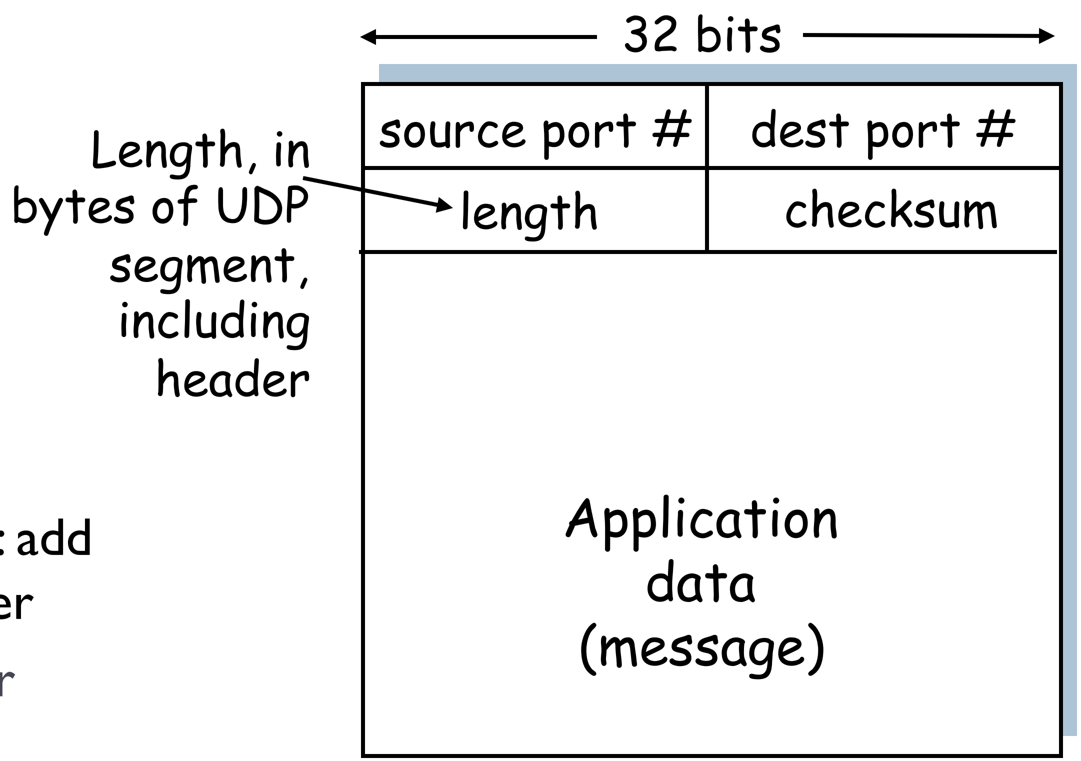
- "no frills," "bare bones" Internet transport protocol
- "best effort" service, UDP segments may be:
  - lost
  - delivered out of order to app
- *connectionless:*
  - no handshaking between UDP sender, receiver
  - each UDP segment handled independently of others

Why is there a UDP?

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

UDP: more

- often used for streaming multimedia apps
  - loss tolerant
  - rate sensitive
- other UDP uses
  - DNS
  - SNMP
- reliable transfer over UDP: add reliability at application layer
  - application-specific error recovery!



UDP segment format

UDP checksum

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

Sender:

- treat segment contents as sequence of 16-bit integers
- checksum: addition (1'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?* More later ….

Internet Checksum Example

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

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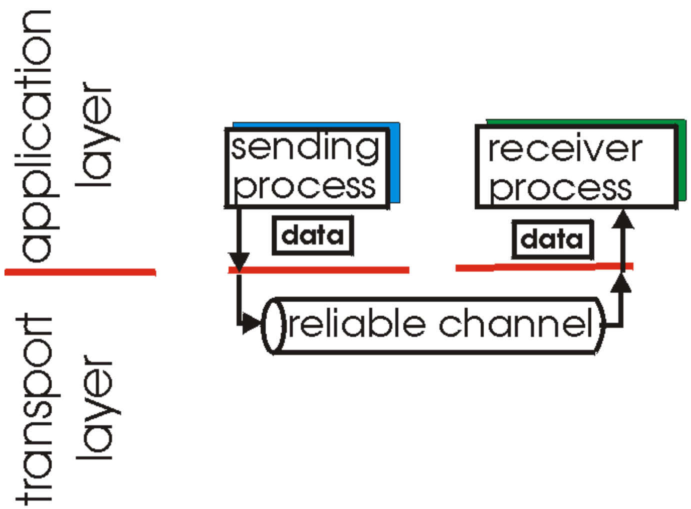
Principles of Reliable data transfer

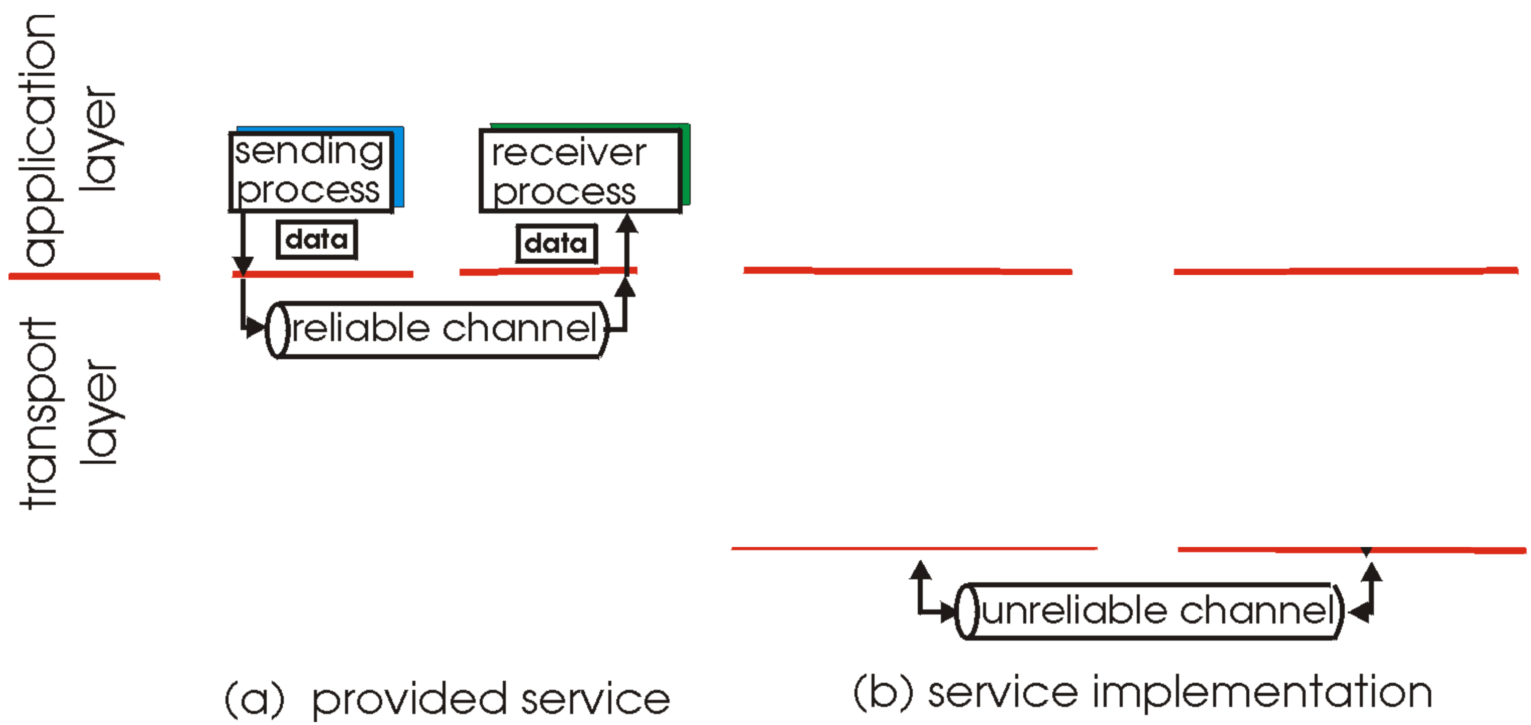
- important in app., transport, link layers
- top-10 list of important networking topics!



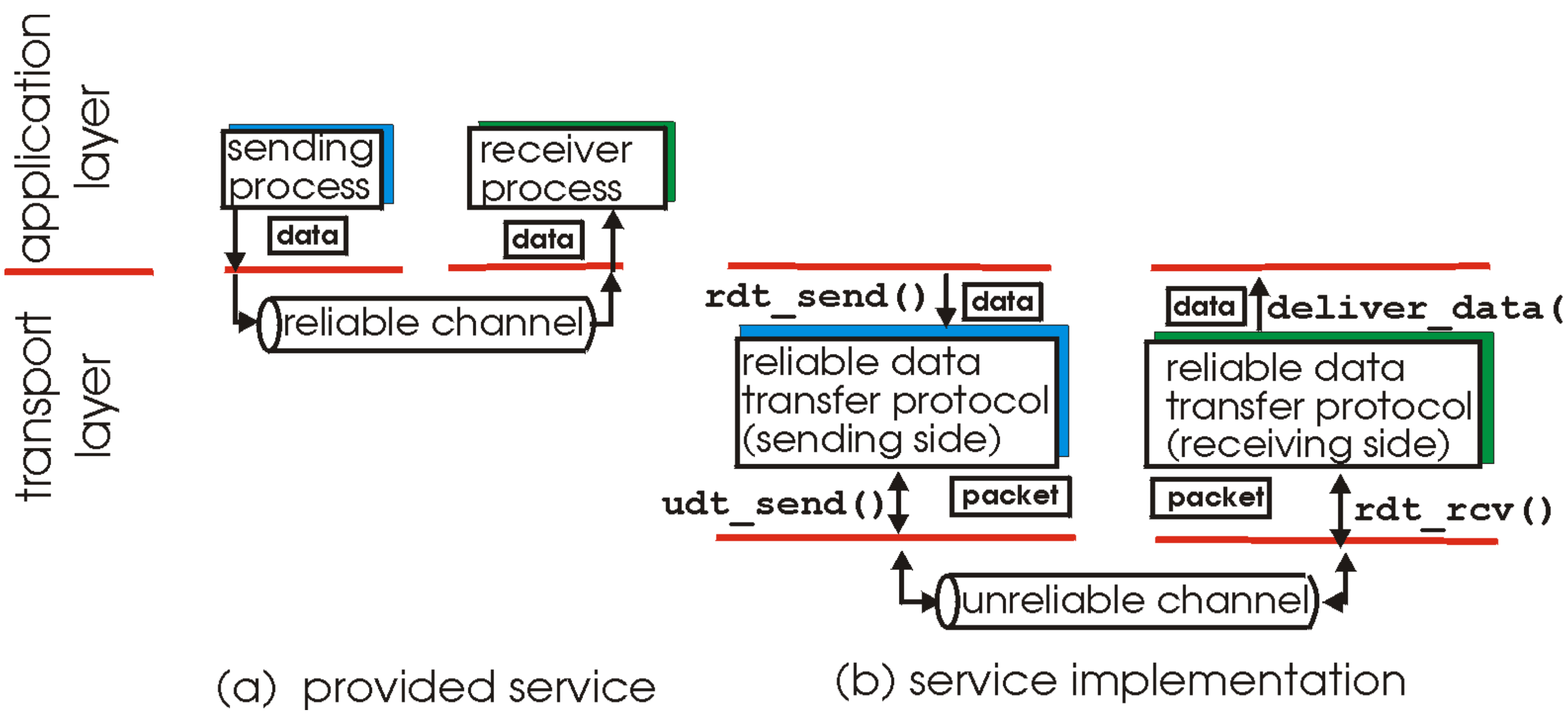
(a) provided service

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

Principles of Reliable data transfer



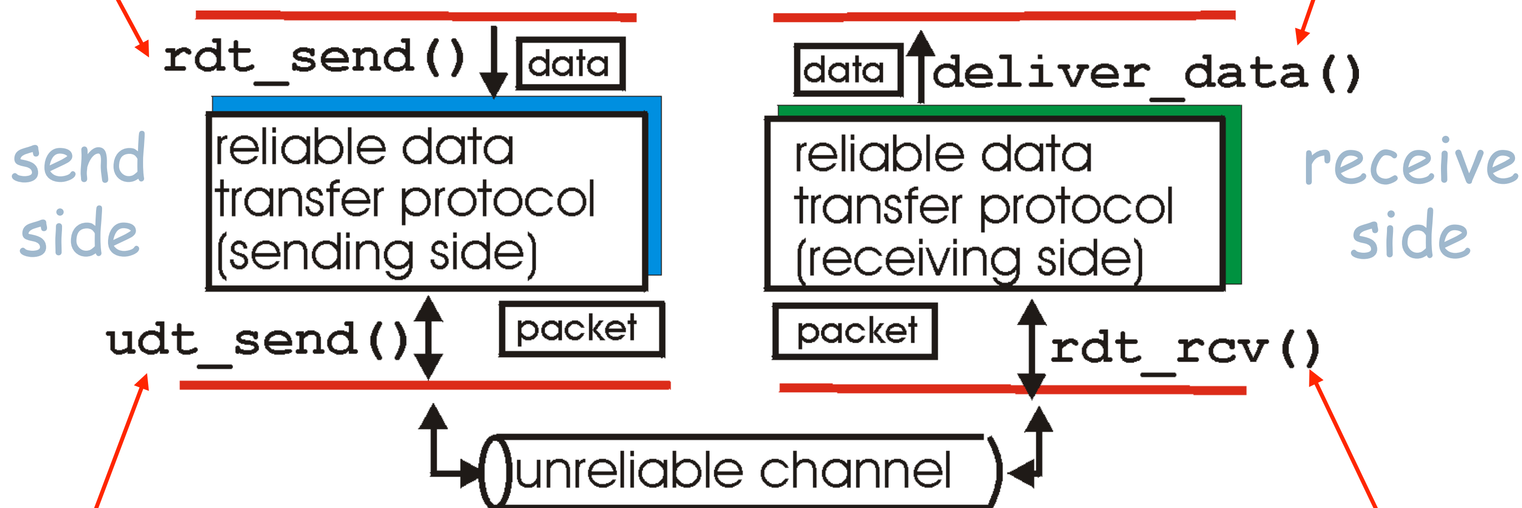
Principles of Reliable data transfer



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



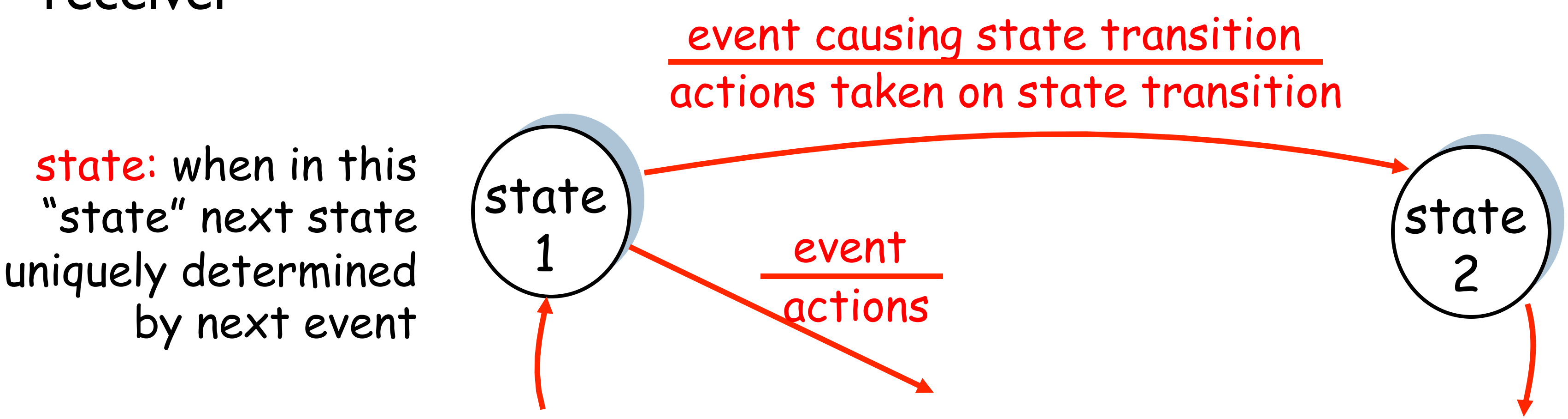
udt_send(): called by rdt, to transfer packet over unreliable channel to receiver

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

Reliable data transfer: getting started

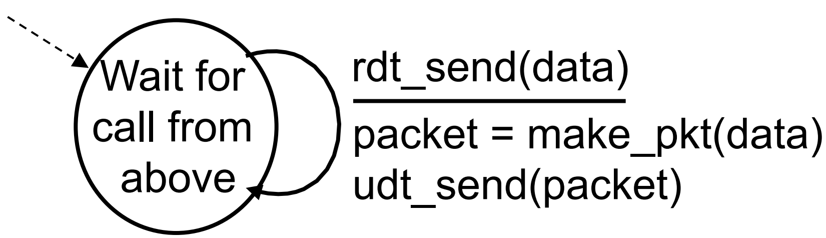
We'll:

- 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

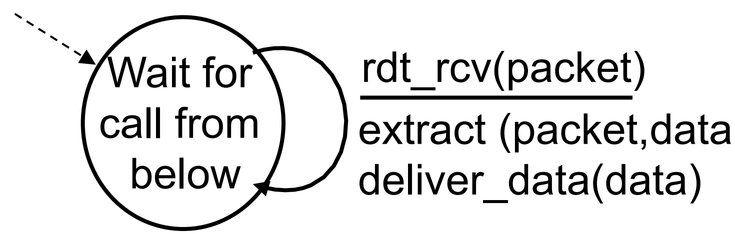


Rdt1.0: reliable transfer over a reliable channel

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



sender

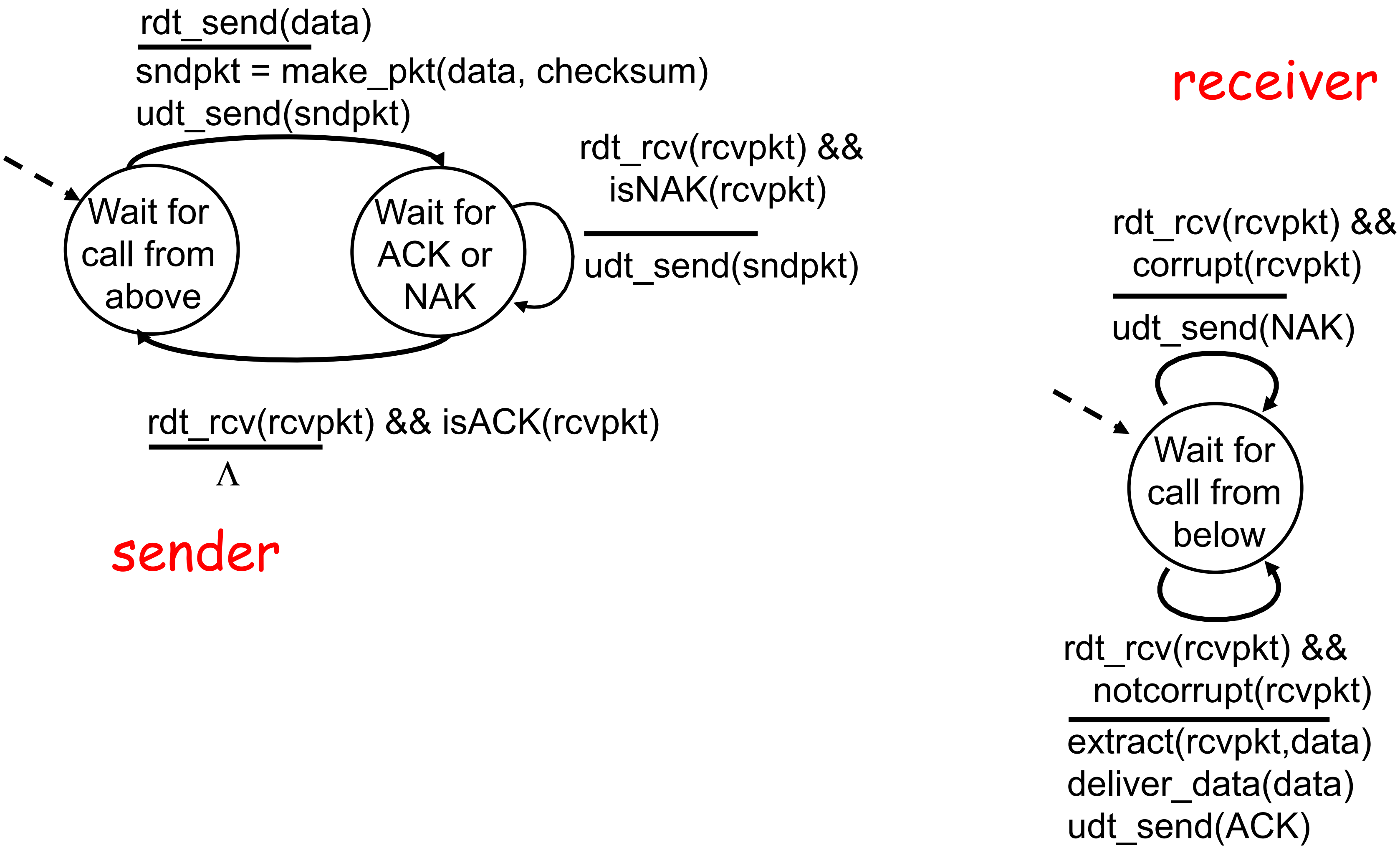


receiver

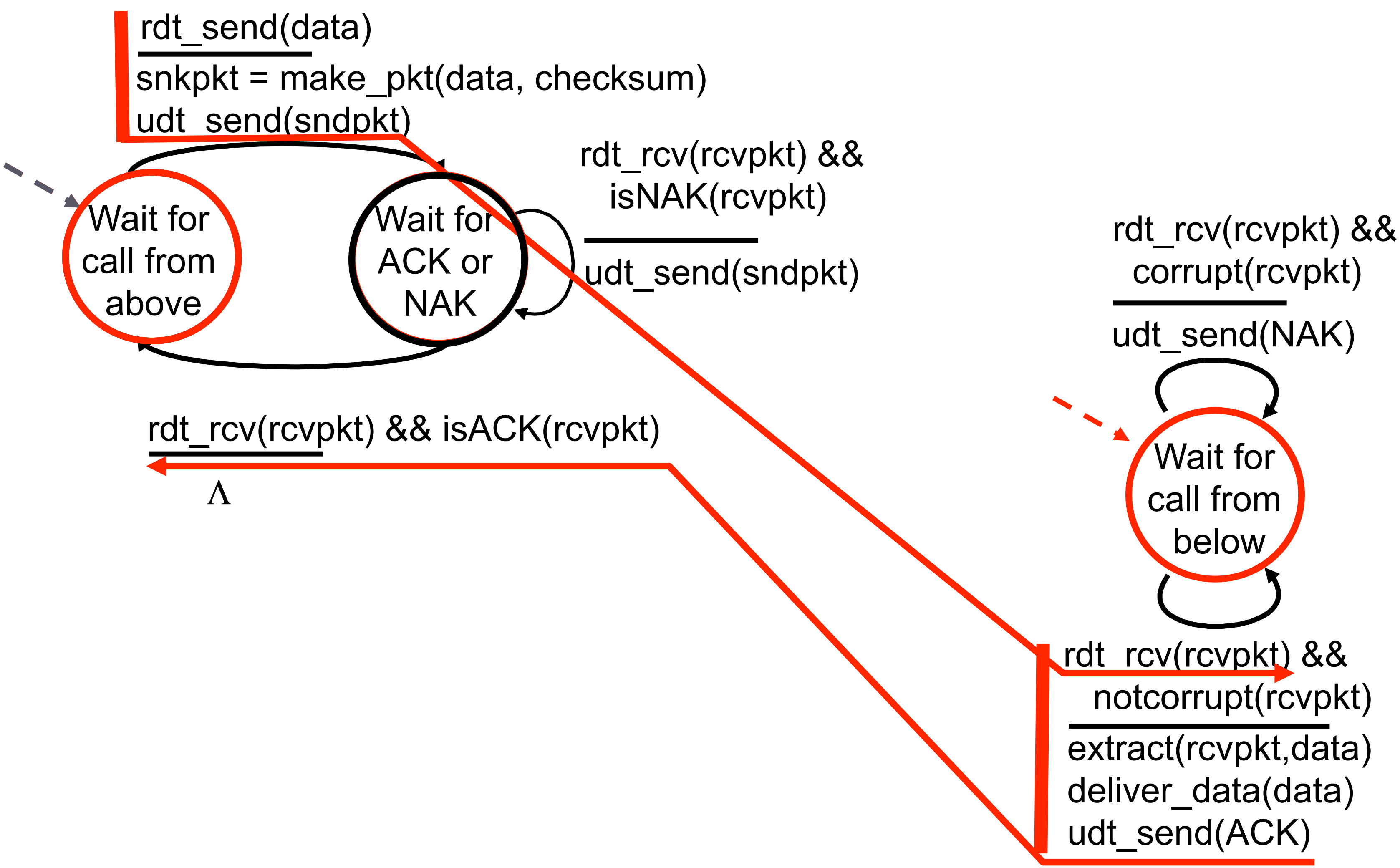
Rdt2.0: channel with bit errors

- 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
  - receiver feedback: control msgs (ACK,NAK) rcvr->sender

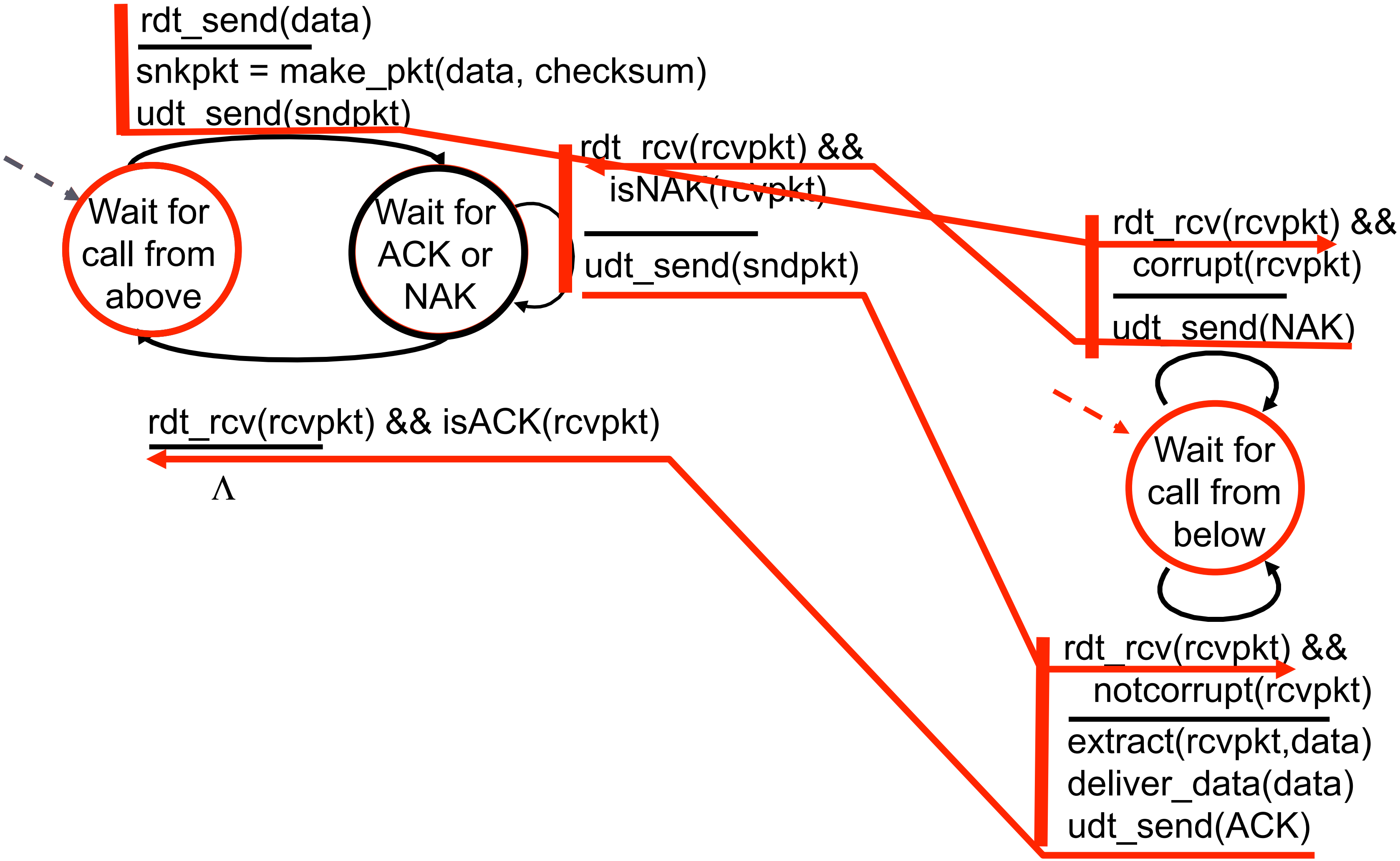
rdt2.0: FSM specification



rdt2.0: operation with no errors



rdt2.0: error scenario



rdt2.0 has a fatal flaw!

What happens if ACK/NAK corrupted?

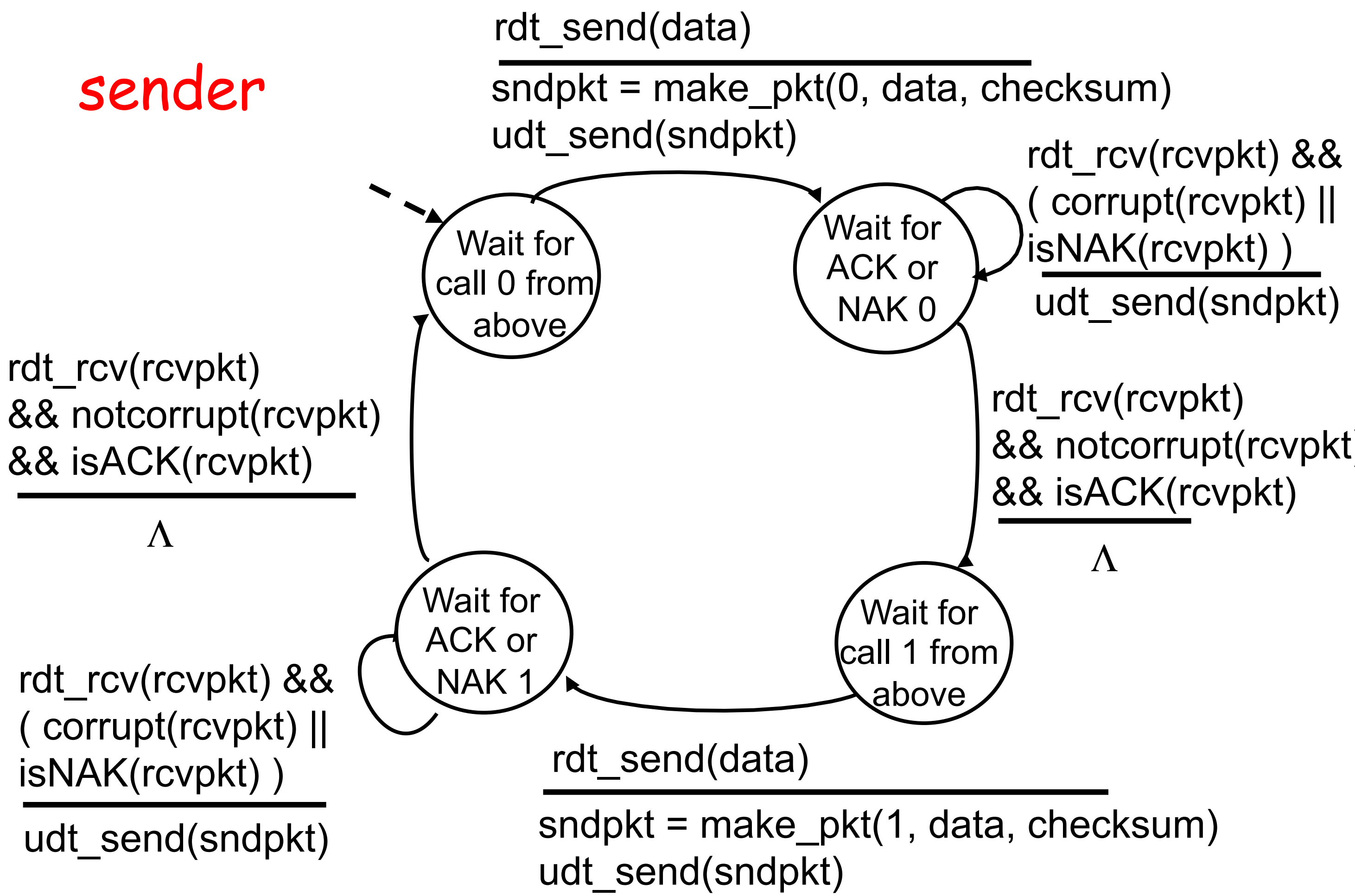
- sender doesn't know what happened at receiver!
- can't just retransmit: possible duplicate

Handling duplicates:

- sender retransmits current pkt if ACK/NAK garbled
- 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: handles garbled ACK/NAKs



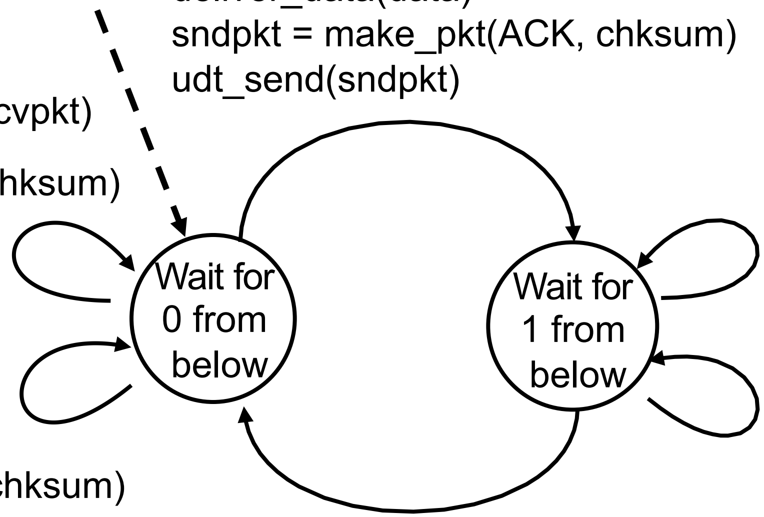
rdt2.1: handles garbled ACK/NAKs

receiver

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

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
- two seq. #'s (0,1) will suffice. Why?
- must check if received ACK/NAK corrupted
- twice as many states
  - state must "remember" whether "current" pkt has 0 or 1 seq. #

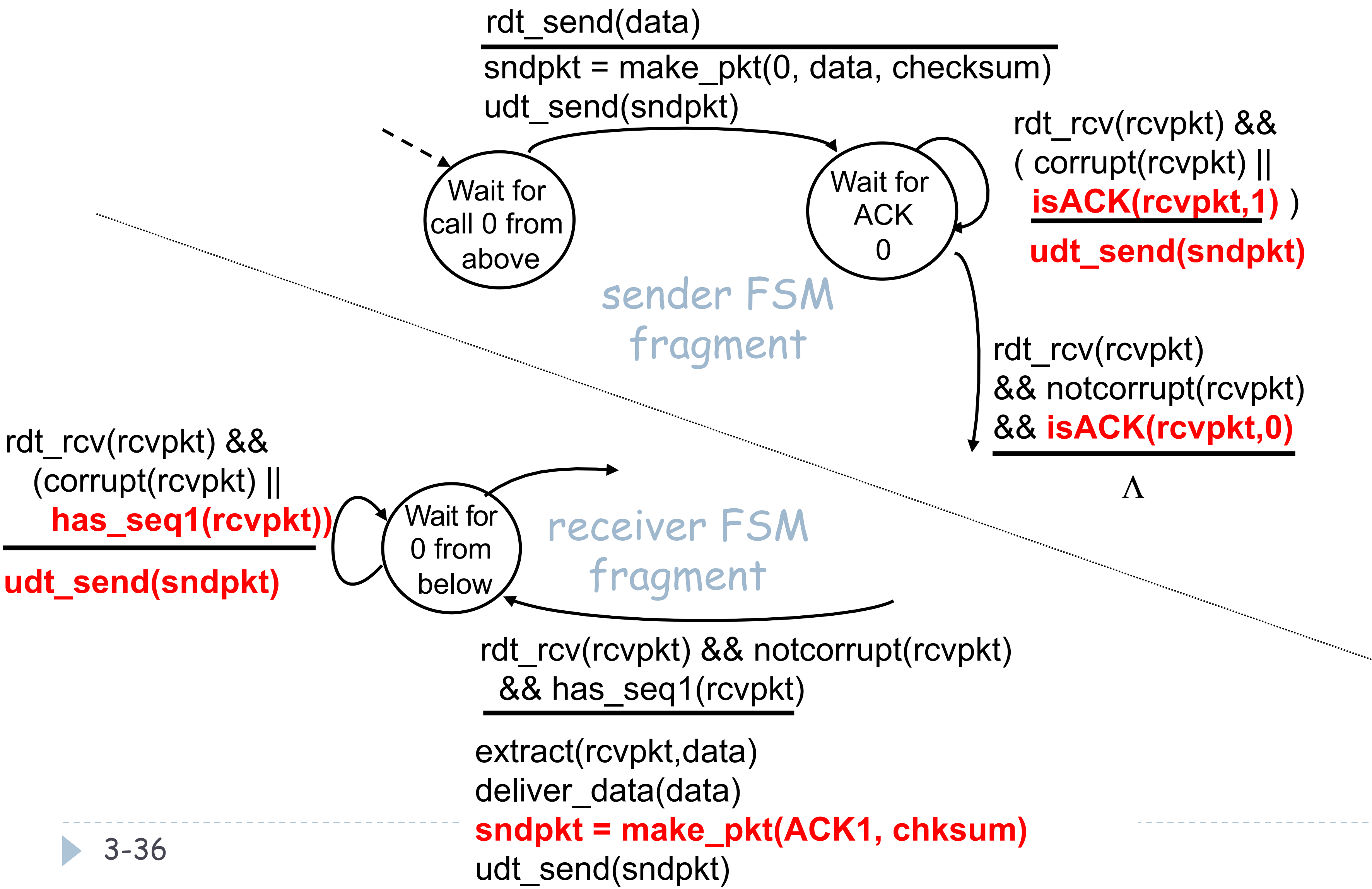
Receiver:

- 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

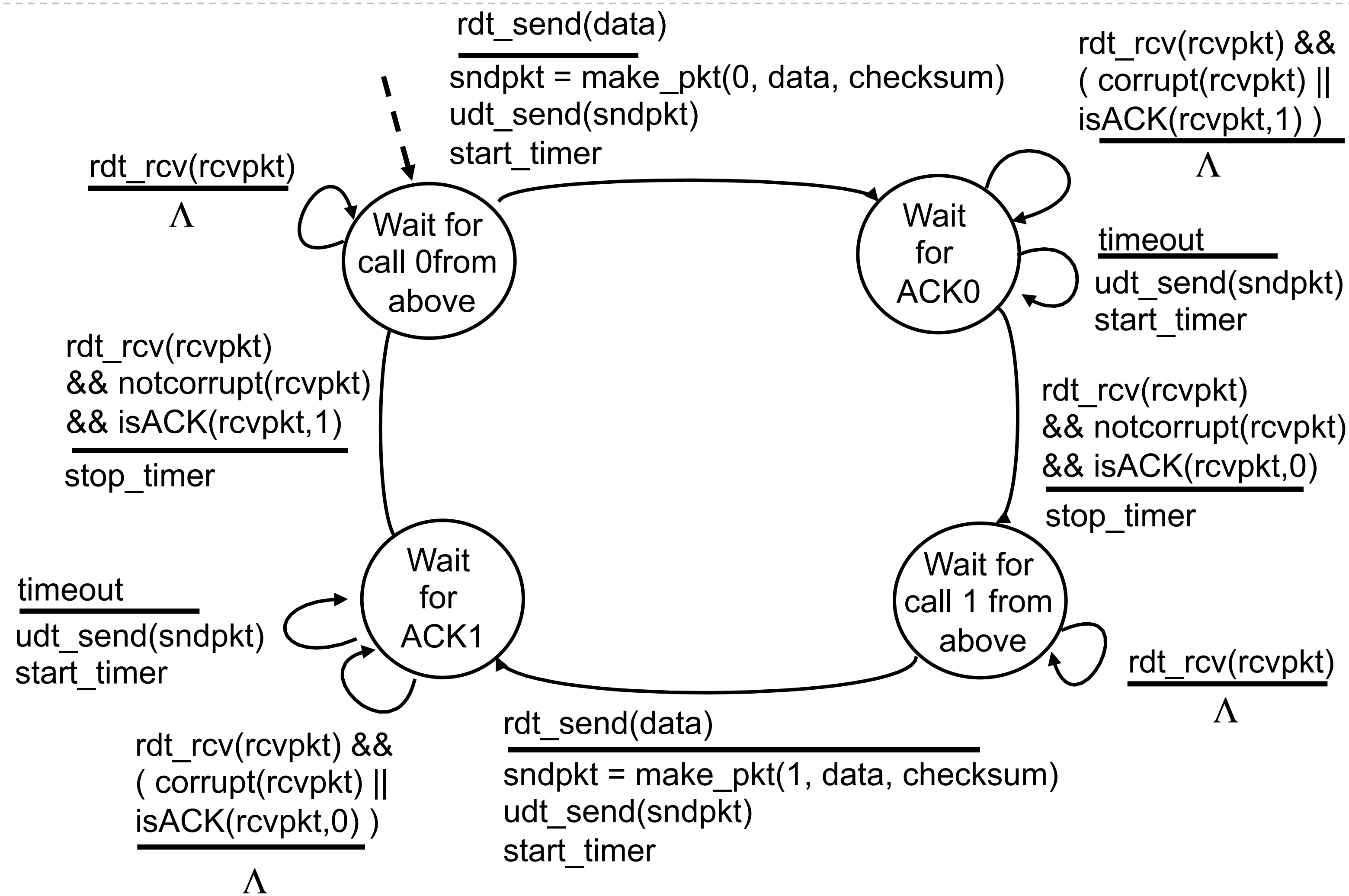
New assumption: underlying channel can also lose packets (data or ACKs)

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

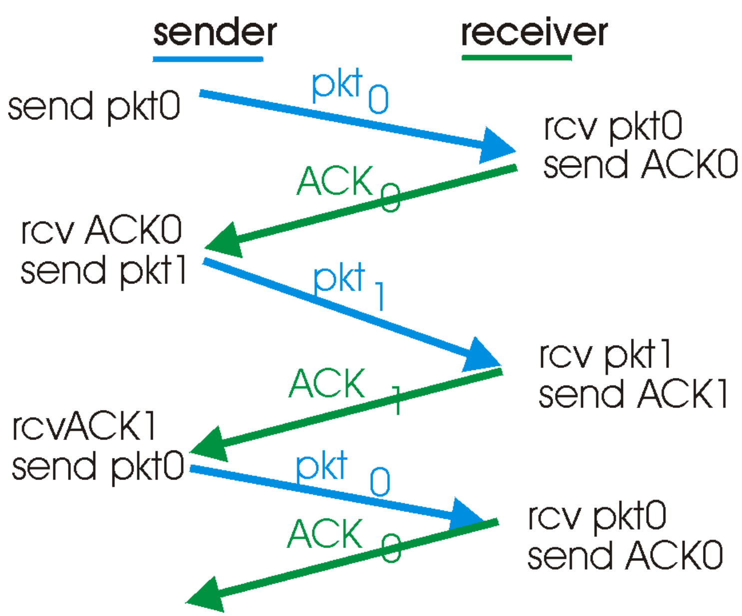
Approach: sender waits "reasonable" amount of time for ACK

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

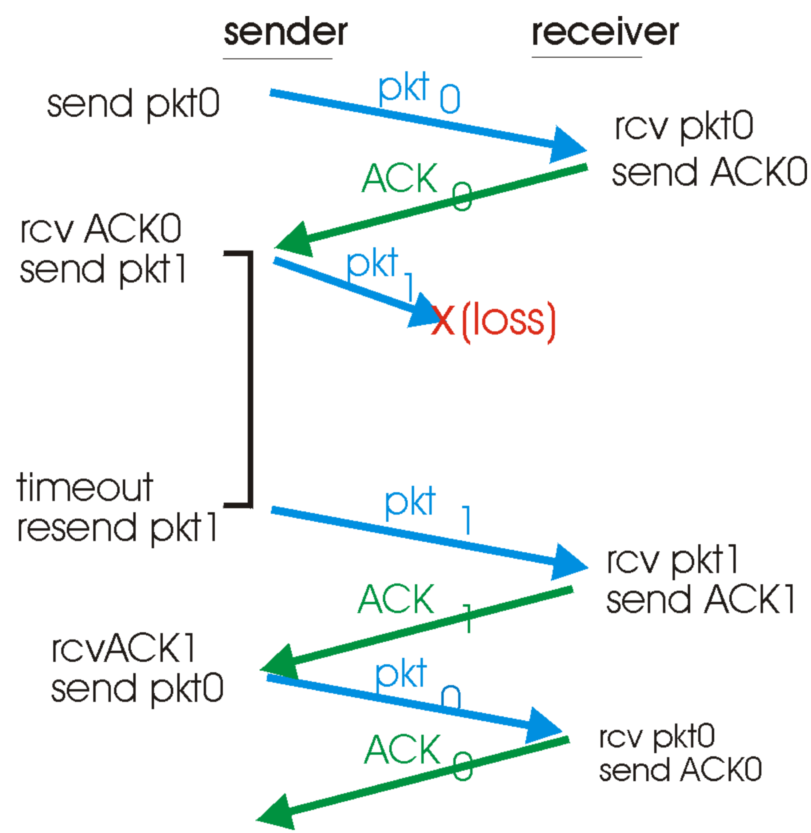
rdt3.0 sender



rdt3.0 in action

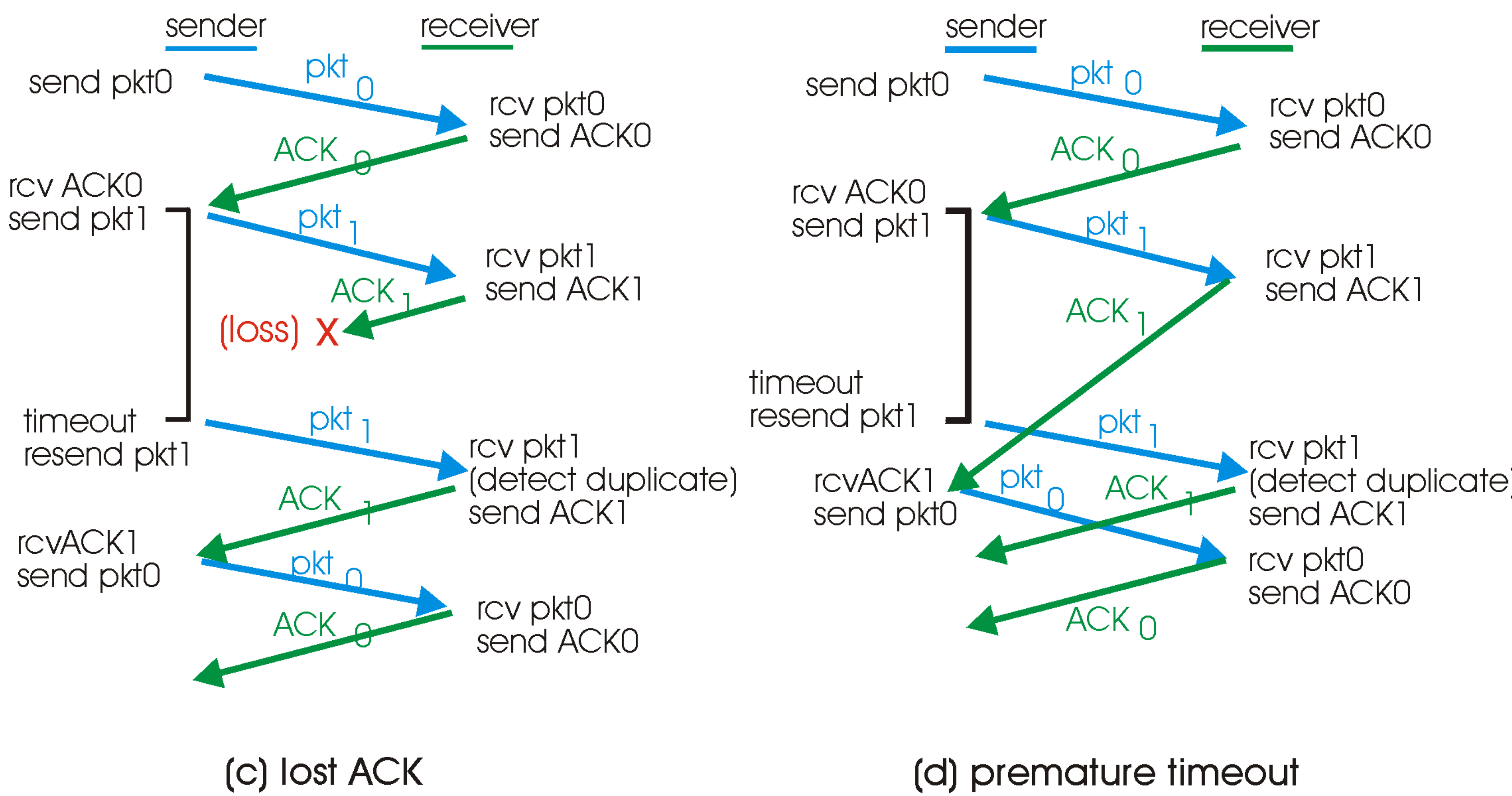


(a) operation with no loss

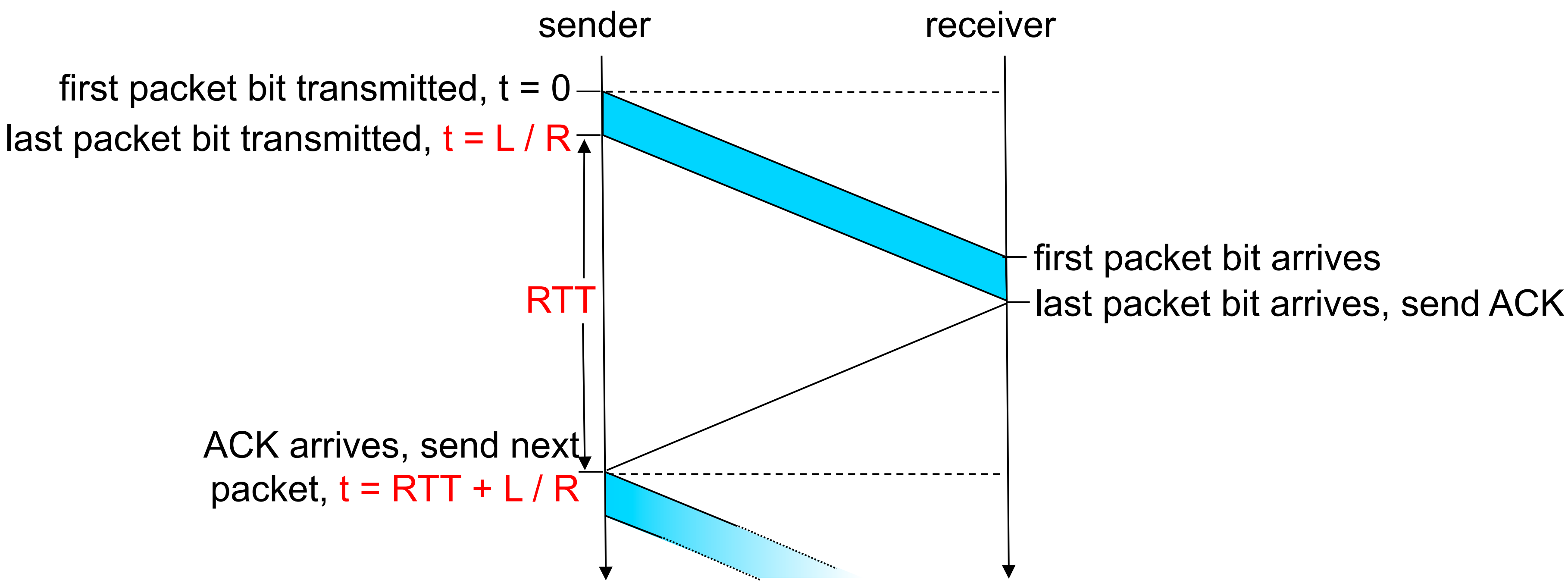


(b) lost packet

rdt3.0 in action



rdt3.0: stop-and-wait operation



Link Utilization →

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

Fraction of time the sender is busy sending traffic

Performance of rdt3.0

- rdt3.0 works, but performance is very bad
- ex: 1 Gbps link, 15 ms prop. delay, 8000 bit packet:

$$d_{trans} = \frac{L}{R} = \frac{8000\text{bits}}{10^9\text{bps}} = 8\,\text{microseconds}$$

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

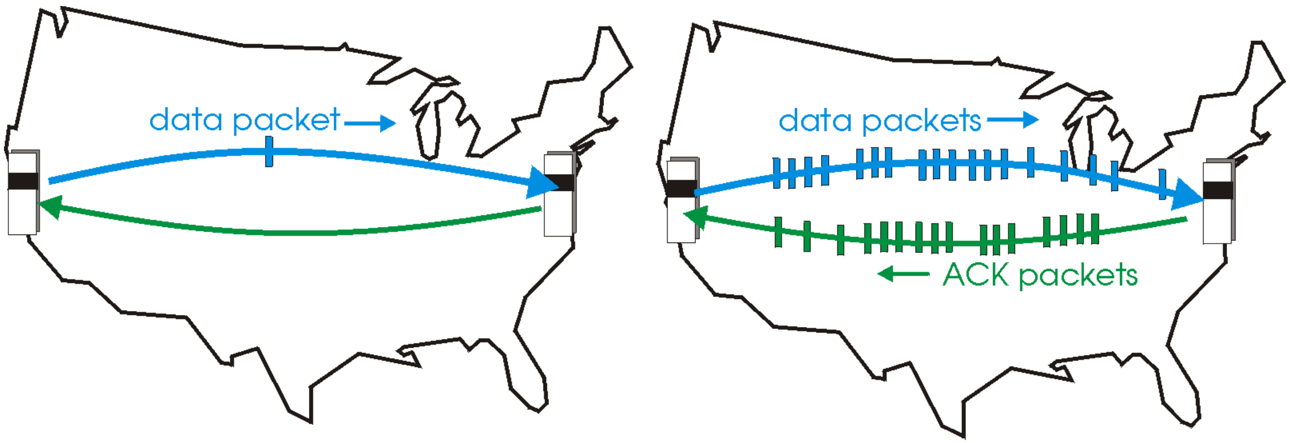
$$U_{sender} = \frac{L/R}{RTT + L/R} = \frac{.008}{30.008} = 0.00027$$

  - ~1kB pkt/30 msec -> 33kB/sec throughput over 1 Gbps link
  - network protocol limits use of physical resources!

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

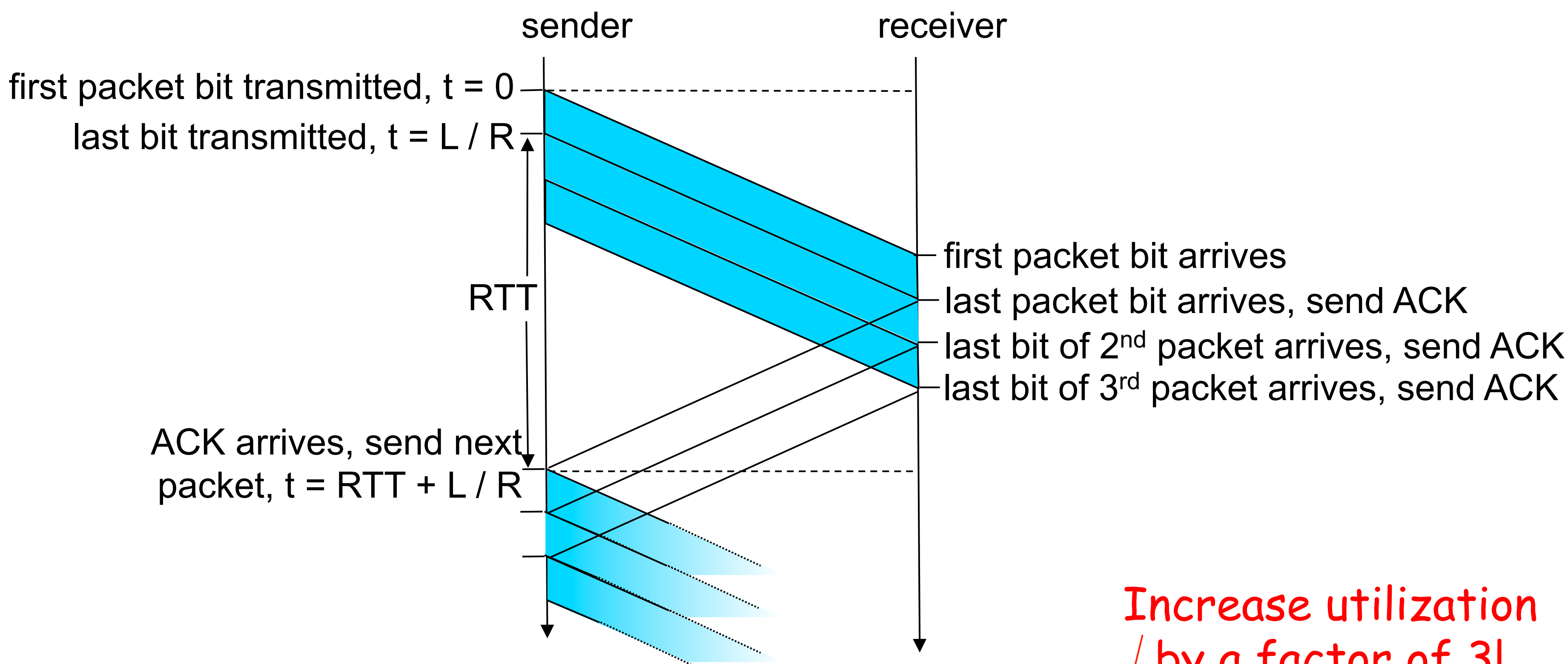


(a) a stop-and-wait protocol in operation

(b) a pipelined protocol in operation

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

Pipelining: increased utilization



Increase utilization by a factor of 3!

$$U_{sender} = \frac{3 * L / R}{RTT + L / R} = \frac{.024}{30.008} = 0.0008$$

Pipelining Protocols

<u>Go-back-N: overview</u>

- *sender:* up to N unACKed pkts in pipeline
- *receiver:* only sends cumulative ACKs
  - doesn't ACK pkt if there's a gap
- *sender:* has timer for oldest unACKed pkt
  - if timer expires: retransmit ***all*** unACKed packets

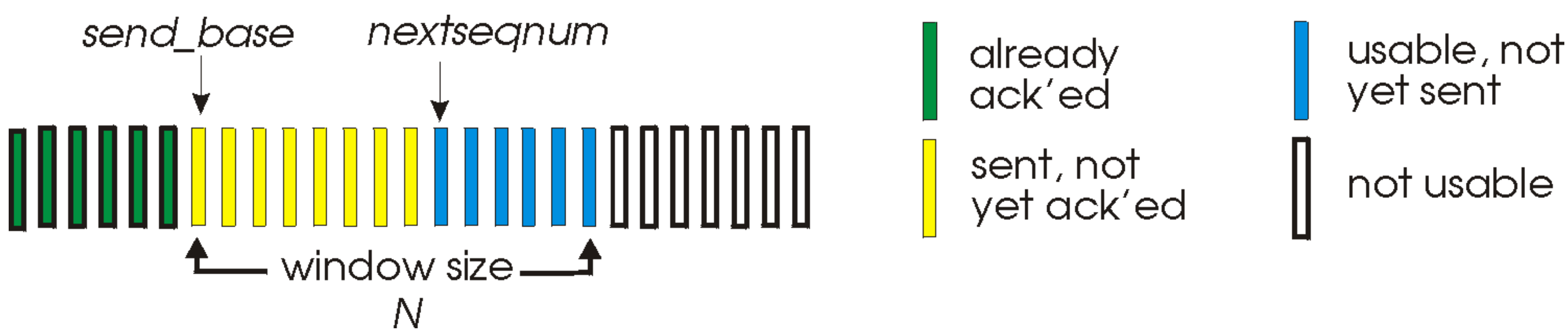
<u>Selective Repeat: overview</u>

- *sender:* up to N unACKed packets in pipeline
- *receiver:* ACKs individual pkts
- *sender:* maintains timer for each unACKed pkt
  - if timer expires: retransmit ***only*** unACKed packet

Go-Back-N

Sender:

- k-bit seq # in pkt header (not limited to 0/1)
- "**sliding window**" of up to N, consecutive unACKed pkts allowed



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

GBN: sender extended FSM

rdt_send(data)

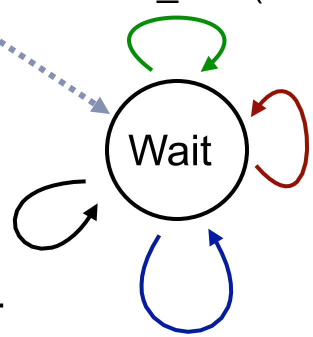
---

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)

Λ

---

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)

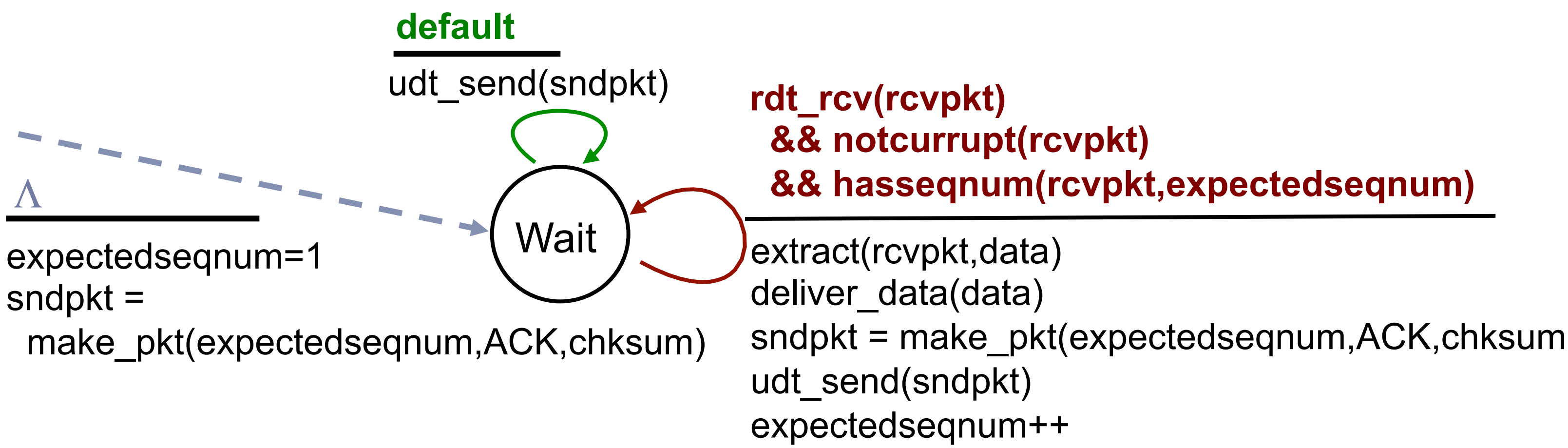
---

rdt_rcv(rcvpkt) &&
  notcorrupt(rcvpkt)

---

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

GBN: receiver extended FSM

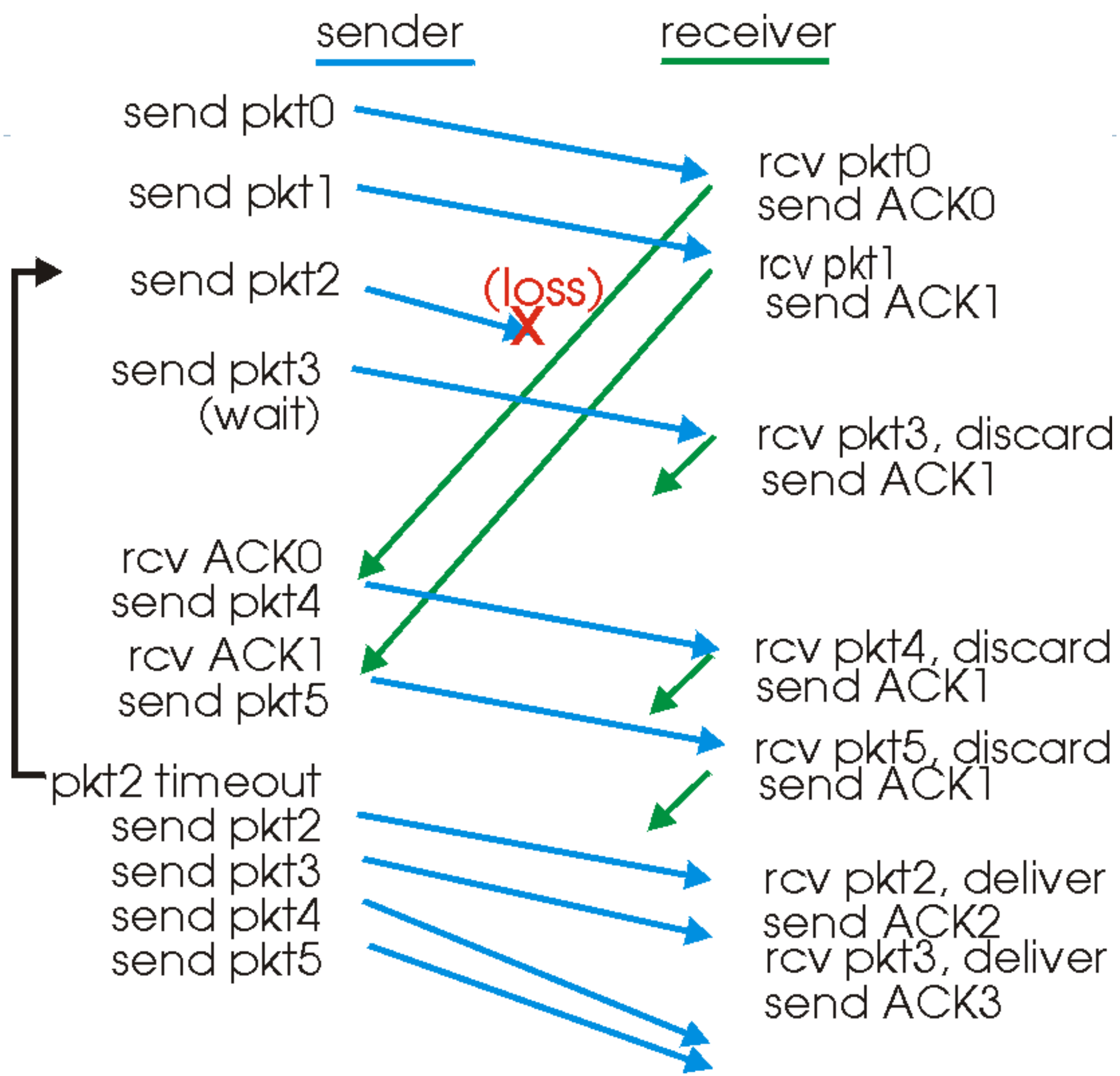


ACK-only: always send ACK for correctly-received pkt with highest *in-order* seq #

- 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 #

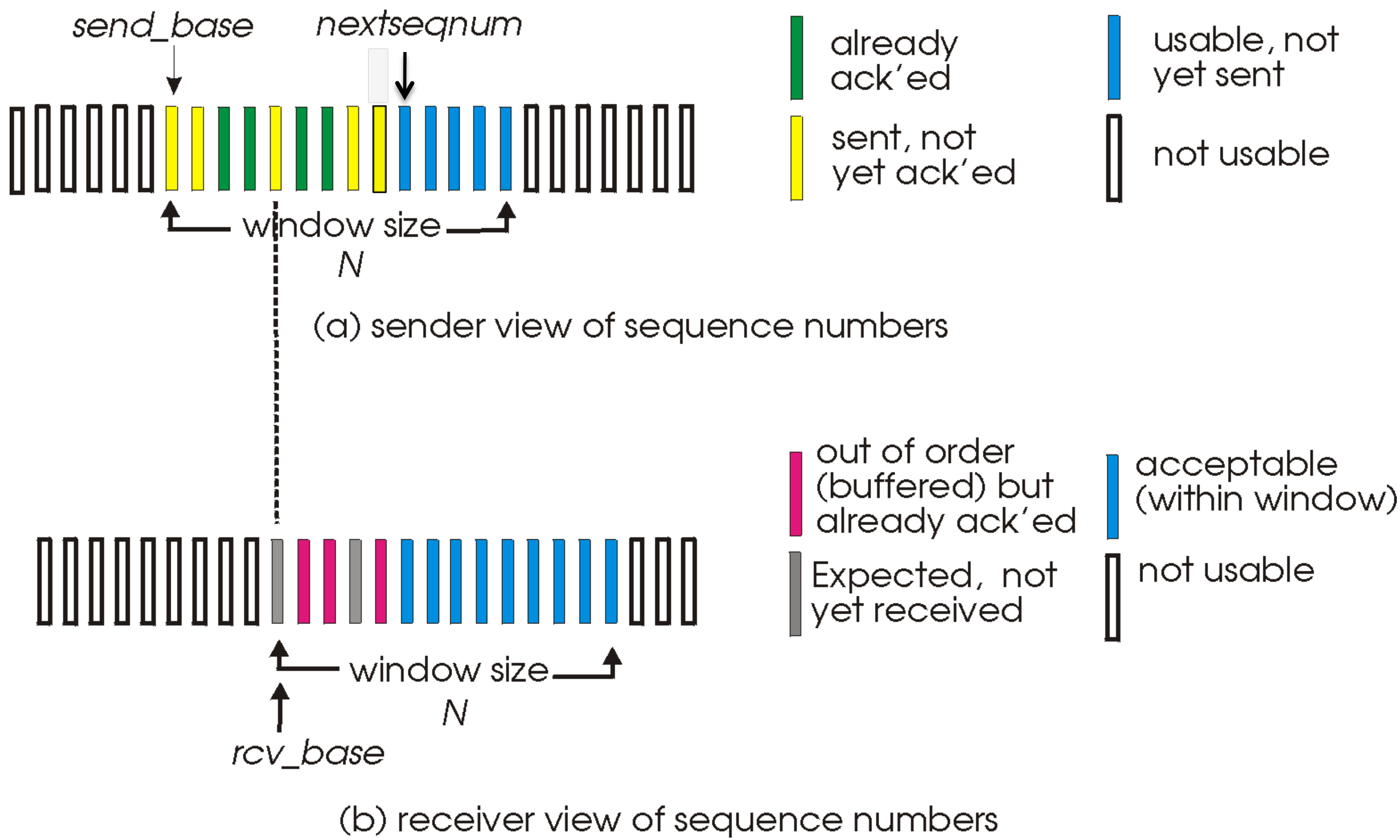
GBN in action



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
  - again 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

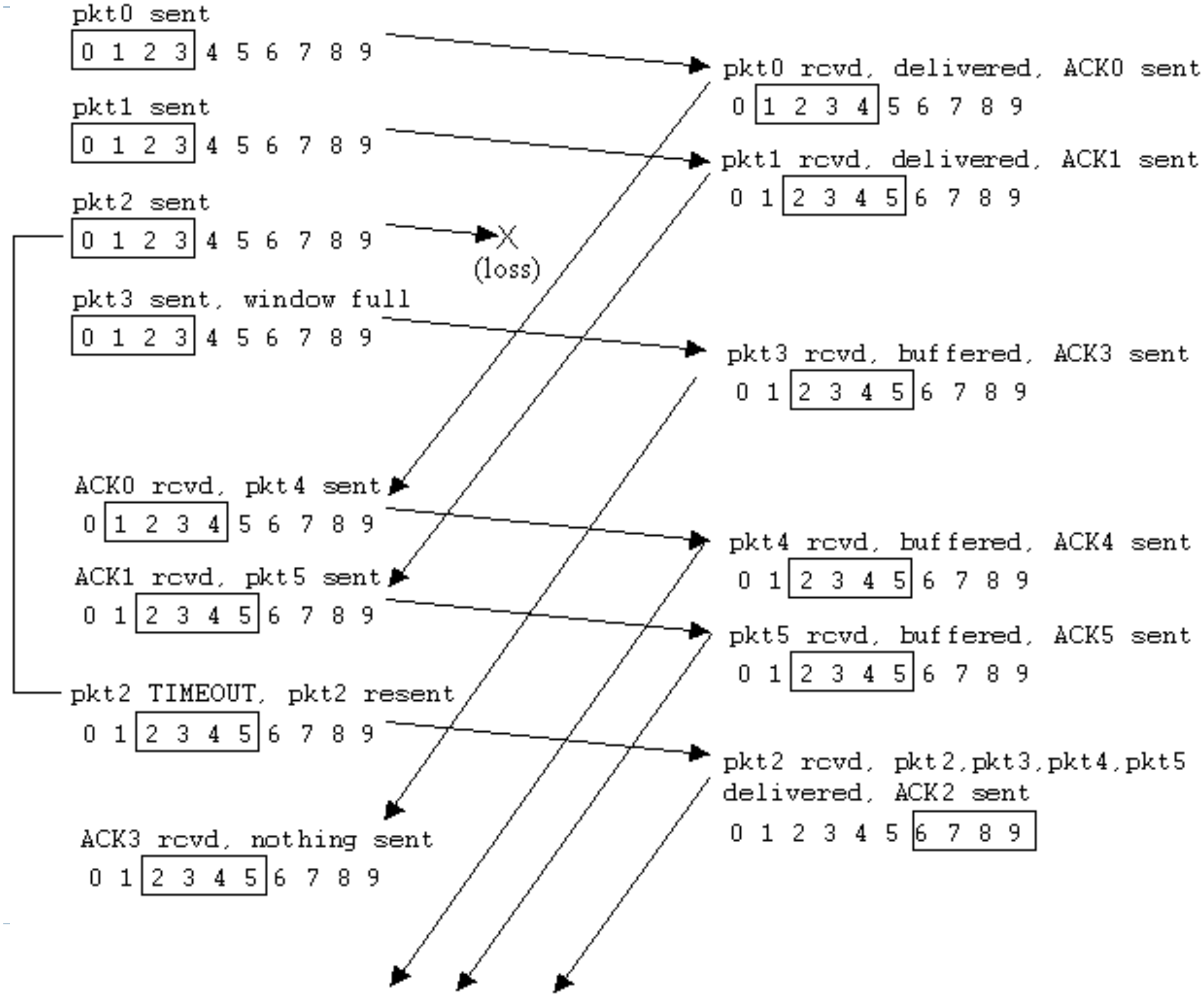
pkt n in [rcvbase-N,rcvbase-1]

- ACK(n)

otherwise:

- ignore

Selective repeat in action



Selective repeat: dilemma

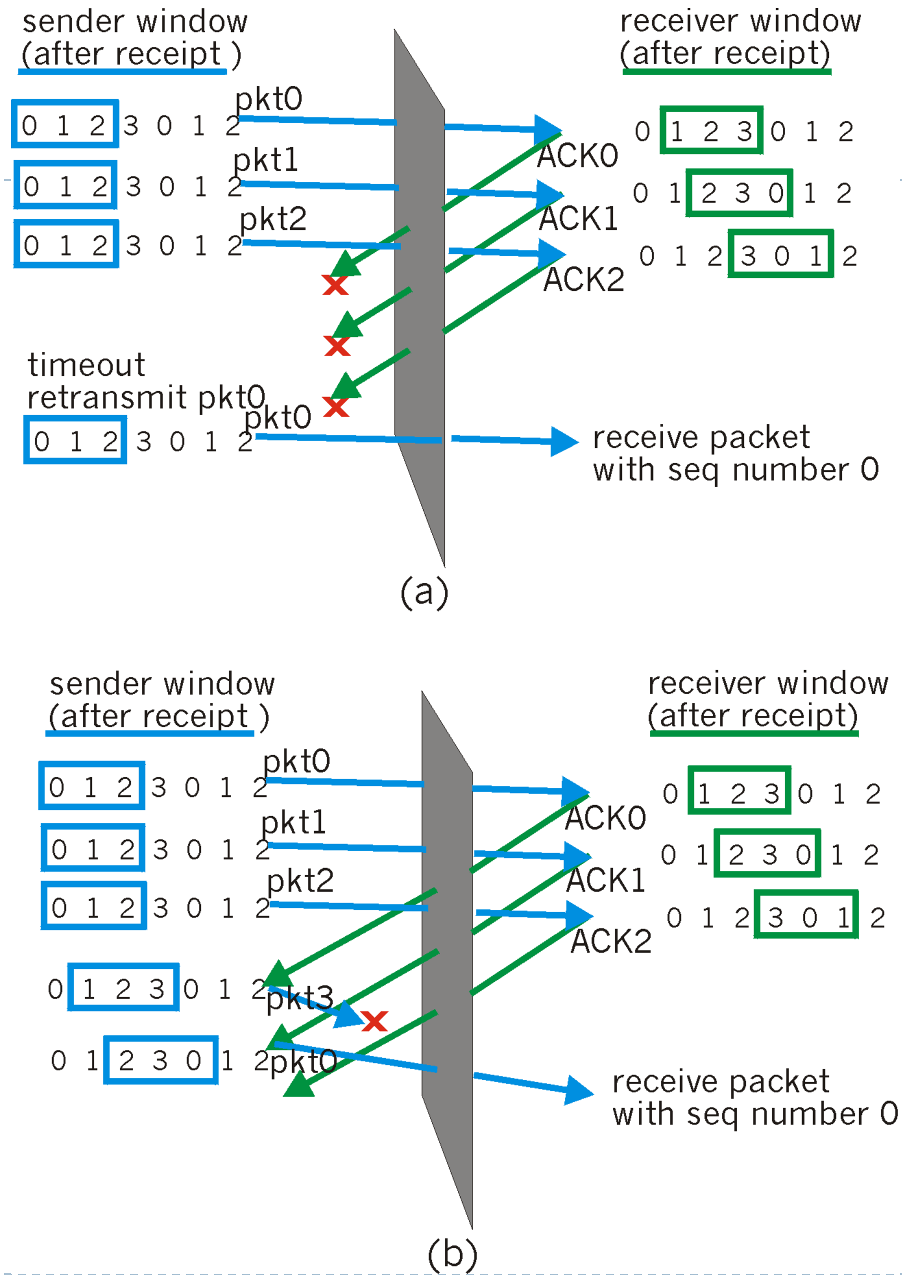
Example:

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

- receiver sees no difference in two scenarios!
- incorrectly passes duplicate data as new in (a)

Q: what relationship between seq # size and window size?

window size <= $\frac{1}{2}$Seq# size



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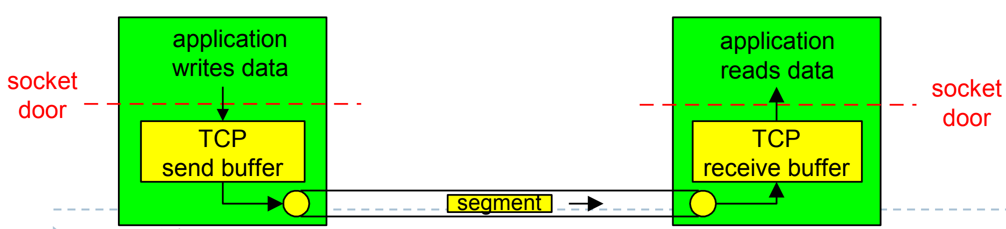
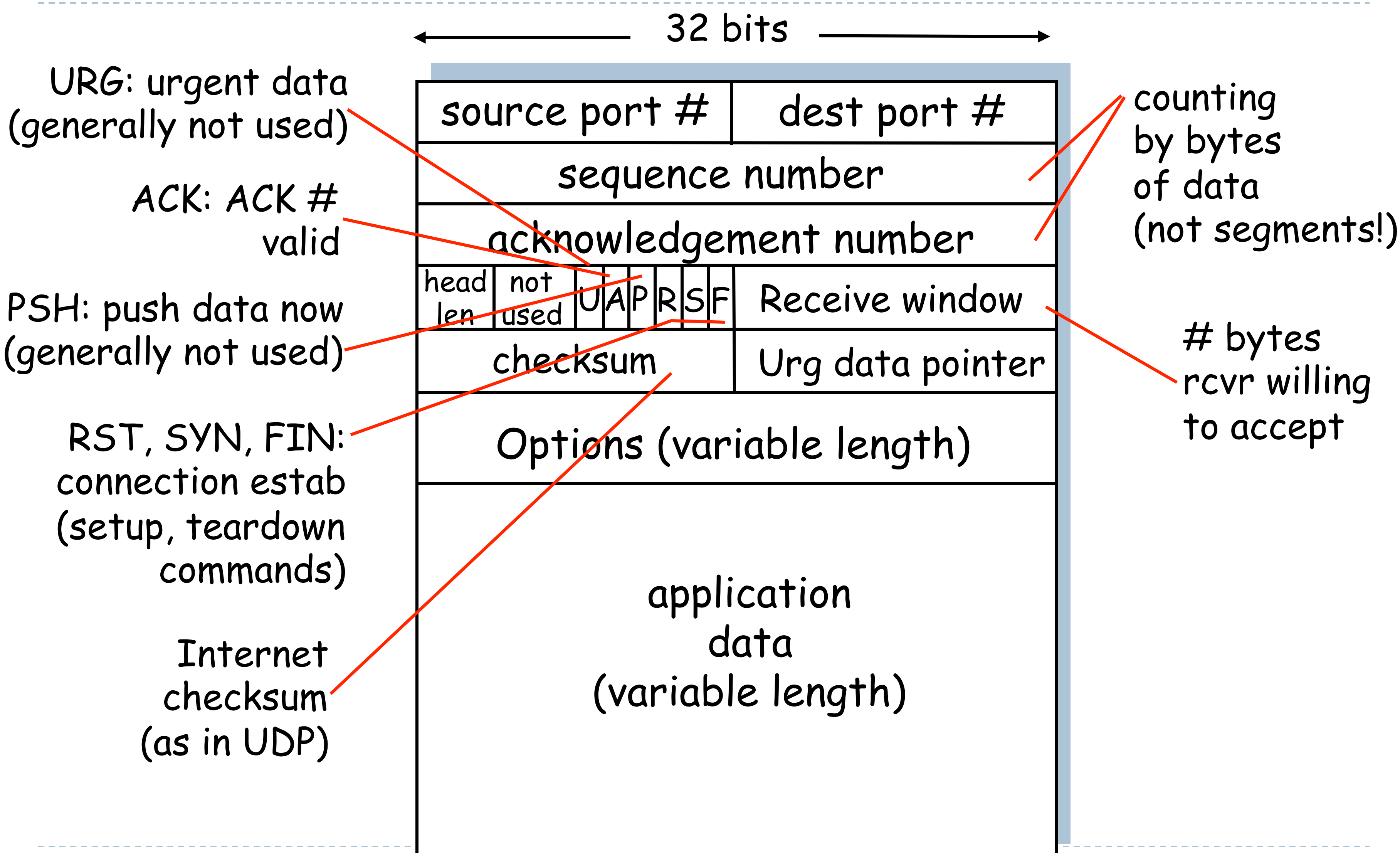
TCP: Overview RFCs: 793, 1122, 1323, 2018, 2581

- point-to-point:
  - one sender, one receiver
- reliable, in-order *byte stream:*
  - no "message boundaries"
- pipelined:
  - TCP congestion and flow control set window size
- *send & receive buffers*



- full duplex data:
  - bi-directional data flow in same connection
  - MSS: maximum segment size
- connection-oriented:
  - handshaking (exchange of control msgs) init's sender, receiver state before data exchange
- flow controlled:
  - sender will not overwhelm receiver

TCP segment structure



TCP seq. #'s and ACKs

Seq. #'s:

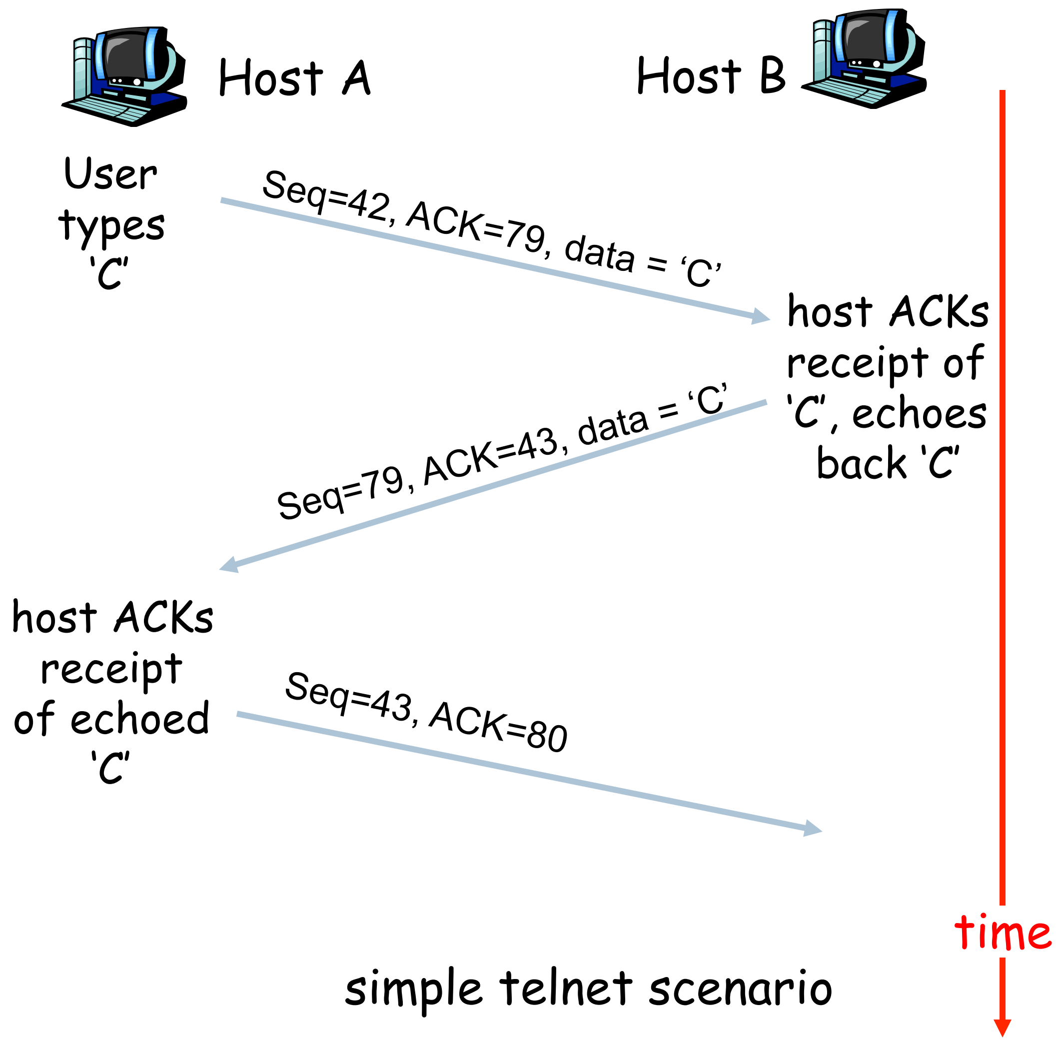
- byte stream "number" of first byte in segment's data

ACKs:

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

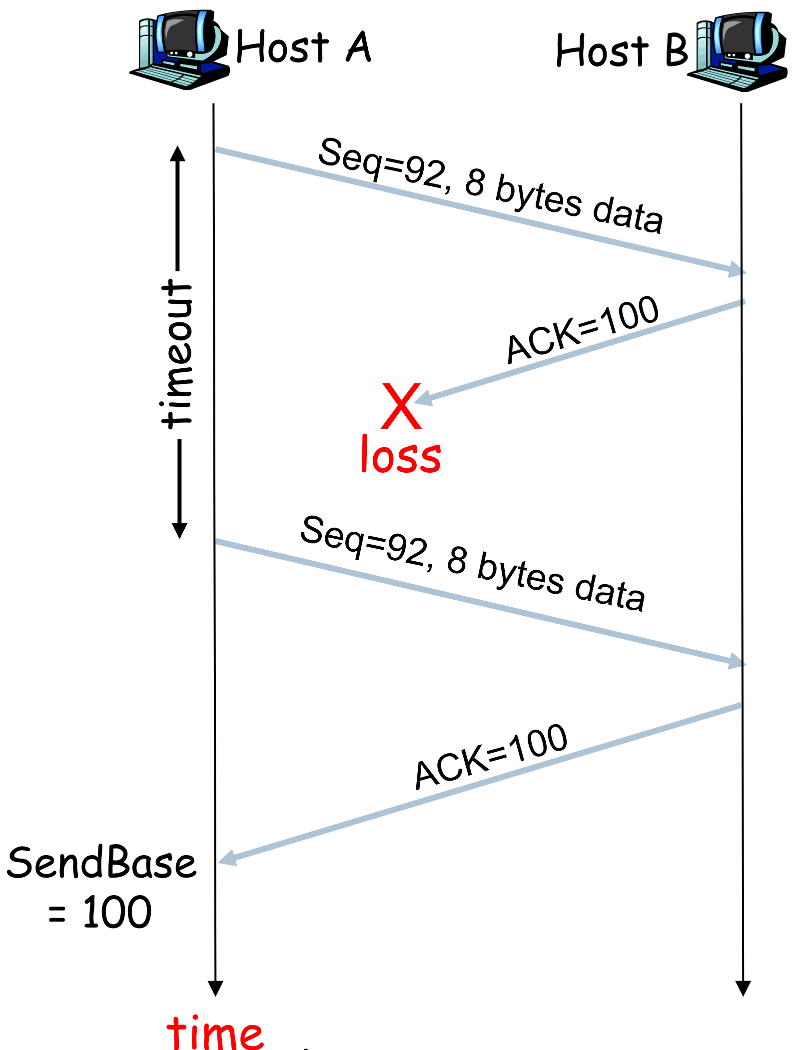
Q: how receiver handles out-of-order segments

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

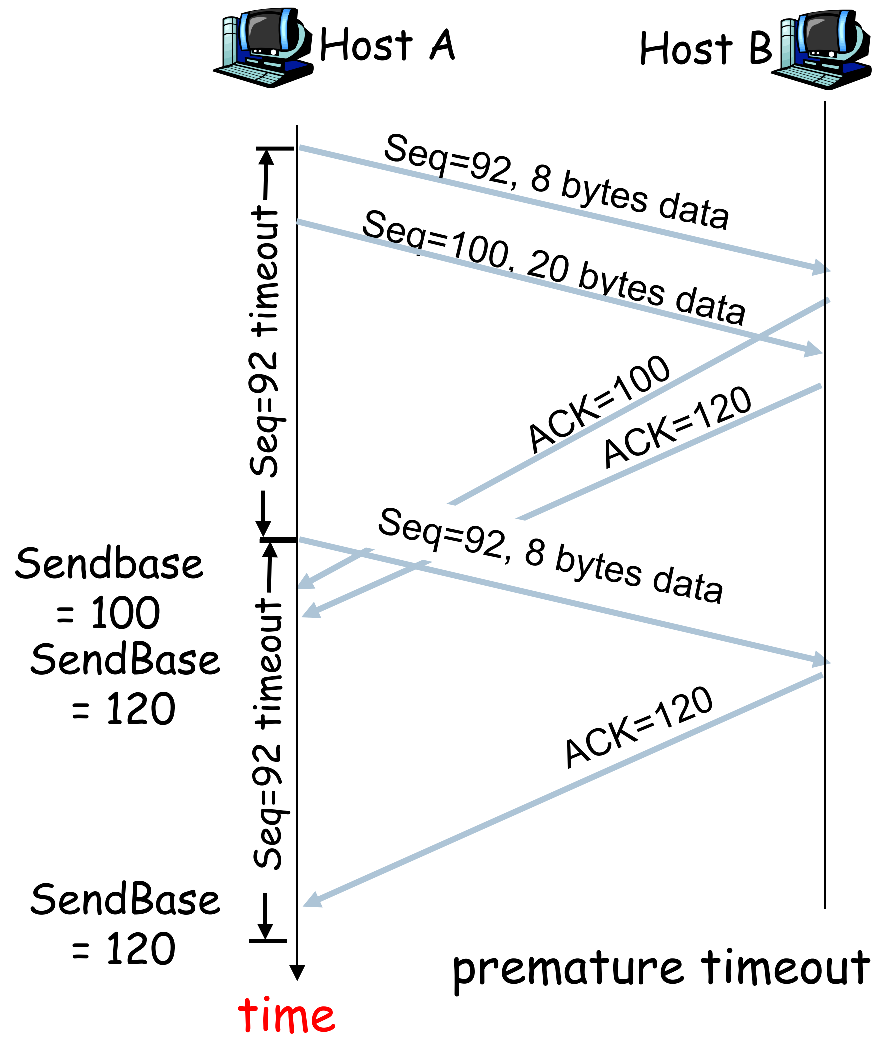


simple telnet scenario

TCP: retransmission scenarios



lost ACK scenario



premature timeout

TCP Round Trip Time and 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

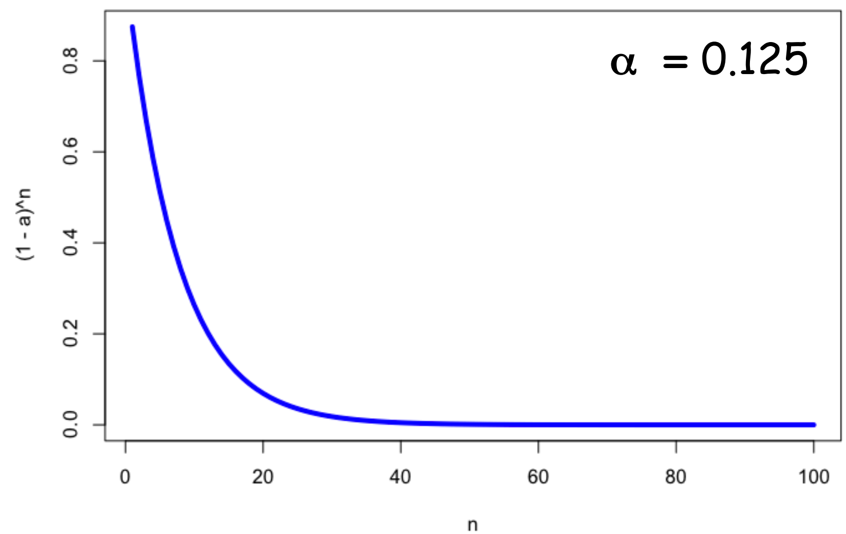
Q: how to estimate RTT?

- **`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`**

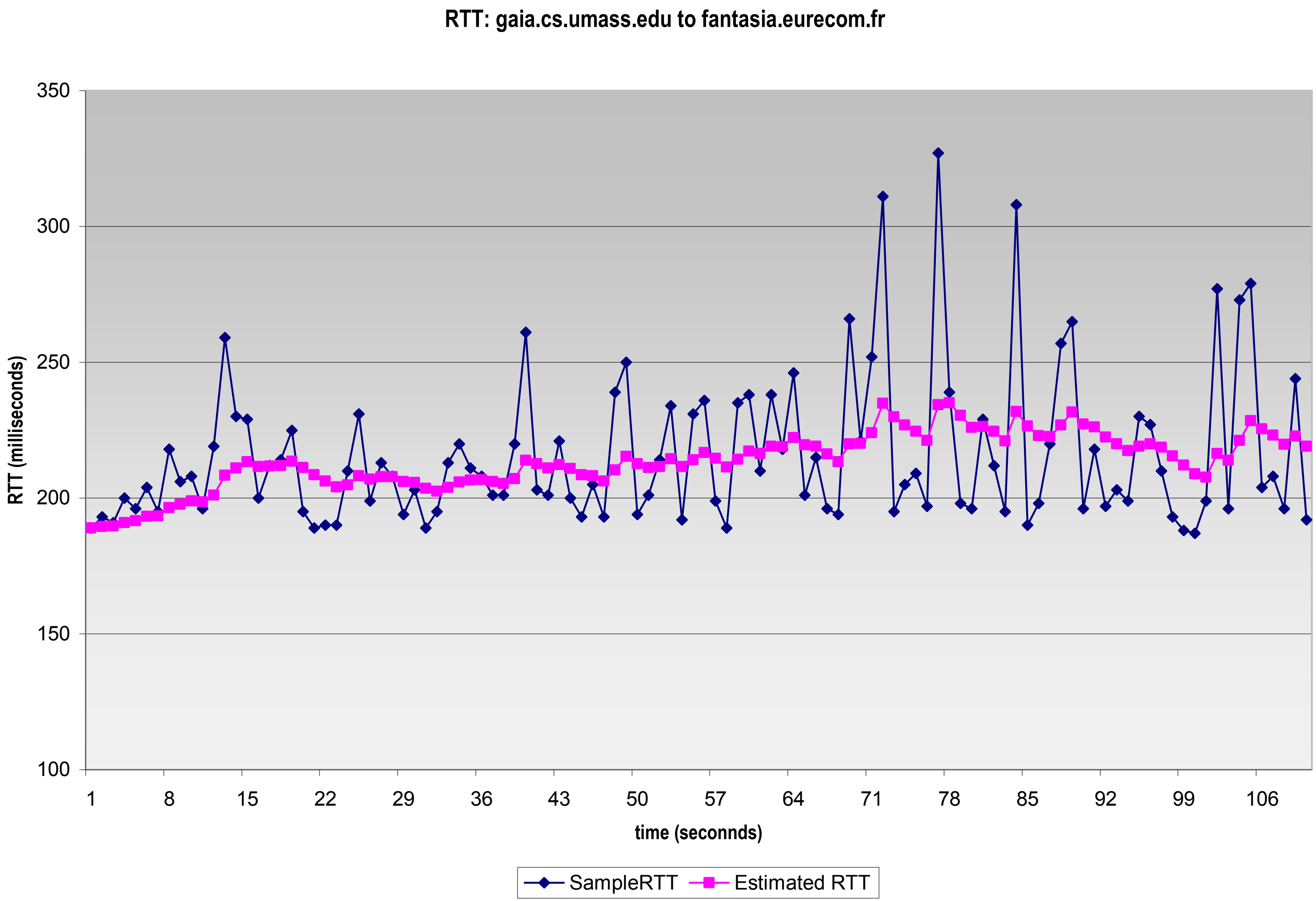
TCP Round Trip Time and Timeout

$$\texttt{EstimatedRTT}_{(i+1)} = (1-\alpha)*\texttt{EstimatedRTT}_{(i)} + \alpha*\texttt{SampleRTT}_{(i+1)}$$

- Exponential weighted moving average
- typical value: $\alpha = 0.125$
- influence of past sample decreases exponentially fast



Example RTT estimation:



TCP Round Trip Time and Timeout

Setting the timeout

- **`EstimtedRTT`** plus "safety margin"
  - large variation in **`EstimatedRTT`** -> larger safety margin
- first estimate of how much SampleRTT deviates from EstimatedRTT:

EWMA of |SampleRTT - Estimated RTT|

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

(typically, β = 0.25)
```

Then set timeout interval:

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

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TCP reliable data transfer

- TCP creates rdt service on top of IP's unreliable service
- pipelined segments
- cumulative ACKs
- TCP uses single retransmission timer
- retransmissions are triggered by:
  - timeout events
  - duplicate ACKs
- 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 acknowledges previously unACKed segments
  - update what is known to be ACKed
  - start timer if there are outstanding segments

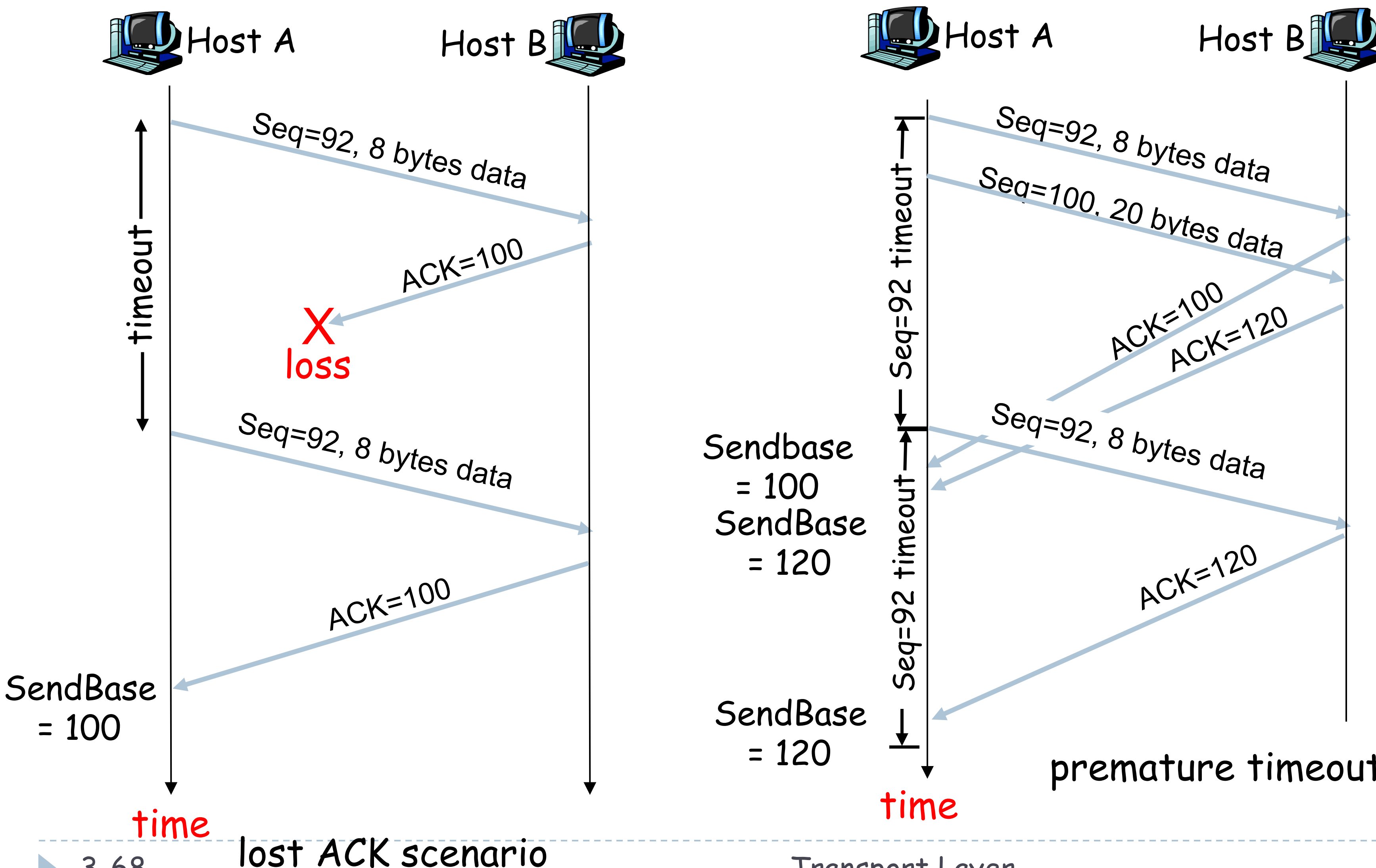
TCP sender (simplified)

```
NextSeqNum = InitialSeqNum
SendBase = InitialSeqNum

loop (forever) {
  switch(event)

  event: data received from application above
       create TCP segment with sequence number NextSeqNum
       if (timer currently not running)
            start timer
       pass segment to IP
       NextSeqNum = NextSeqNum + length(data)

  event: timer timeout
       retransmit not-yet-acknowledged segment with
            smallest sequence number
       start timer

  event: ACK received, with ACK field value of y
       if (y > SendBase) {
            SendBase = y
            if (there are currently not-yet-acknowledged segments)
                 start timer
            }

} /* end of loop forever */
```

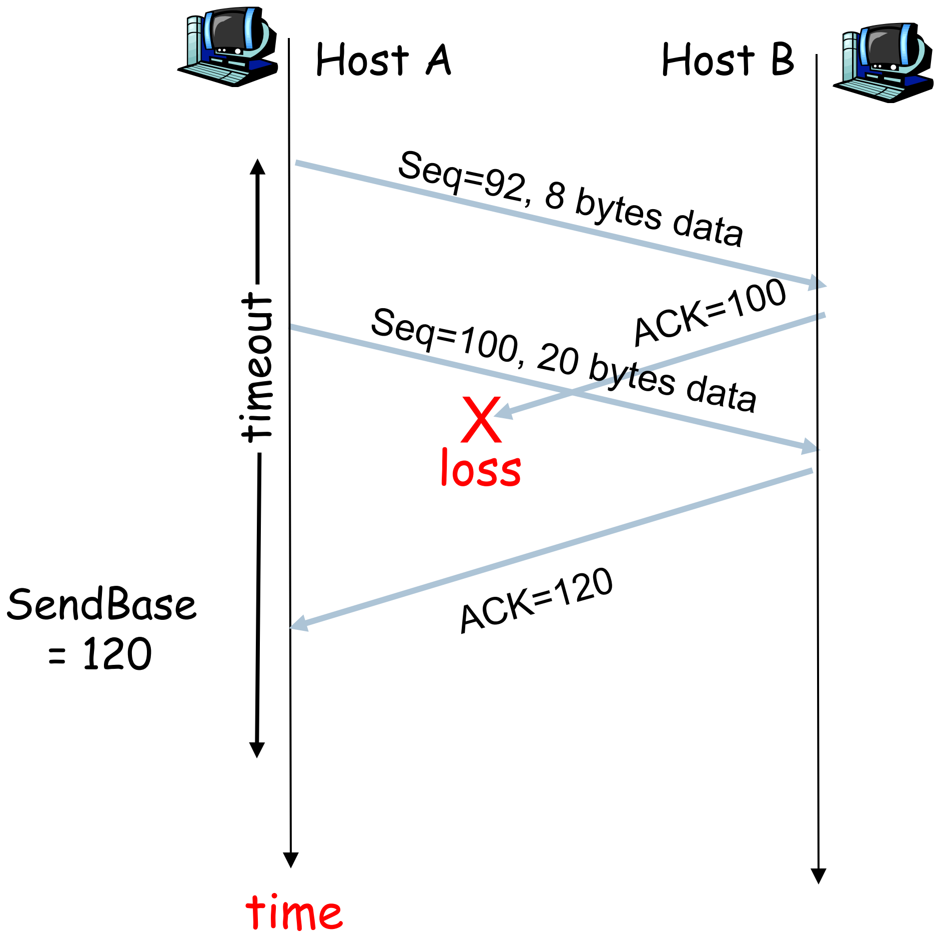
Comment:
- SendBase-1: last cumulatively ACKed byte

Example:
- SendBase-1 = 71; y= 73, so the rcvr wants 73+ ; y > SendBase, so that new data is ACKed

TCP: retransmission scenarios



TCP retransmission scenarios (more)



Cumulative ACK scenario

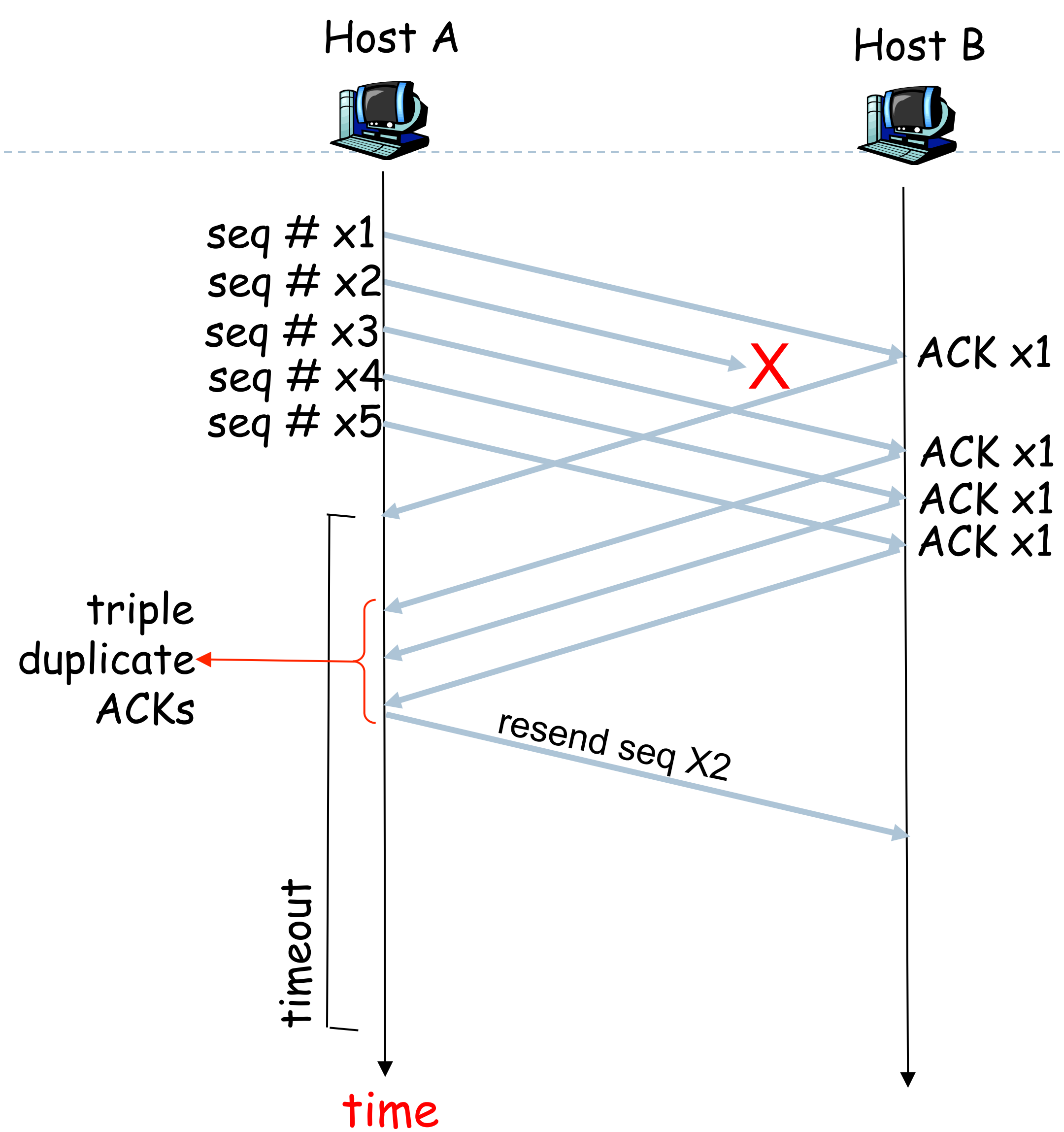
TCP ACK generation [RFC 1122, RFC 2581]

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

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 for that segment

- If sender receives 3 ACKs for same data, it assumes that segment after ACKed data was lost:
  - fast retransmit: resend segment before timer expires


Host A
Host B
seq # x1
seq # x2
seq # x3
seq # x4
seq # x5
X
ACK x1
ACK x1
ACK x1
ACK x1
triple
duplicate
ACKs
resend seq X2
timeout
time

Fast retransmit algorithm:

```
event: ACK received, with ACK field value of y
          if (y > SendBase) {
              SendBase = y
              if (there are currently not-yet-acknowledged segments)
                  start timer
             }
          else {
                increment count of dup ACKs received for y
                if (count of dup ACKs received for y = 3) {
                    resend segment with sequence number y
                 }
```

a duplicate ACK for already ACKed segment

fast retransmit

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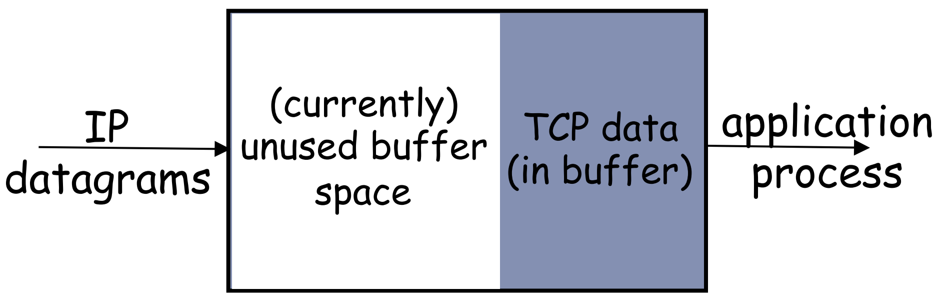
TCP Flow Control

- receive side of TCP connection has a receive buffer:



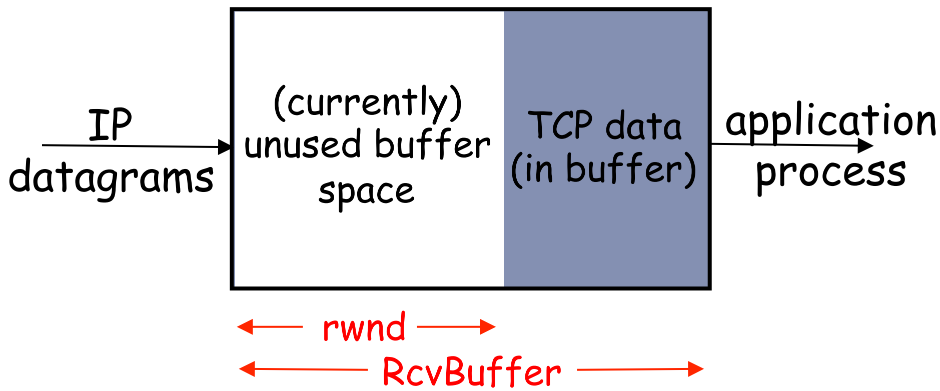
- app process may be slow at reading from buffer

flow control

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

- *speed-matching service:* matching send rate to receiving application's drain rate

TCP Flow control: how it works



(suppose TCP receiver discards out-of-order segments)

- unused buffer space:

= rwnd

= RcvBuffer-[LastByteRcvd - LastByteRead]

- receiver: advertises unused buffer space by including `rwnd` value in segment header
- sender: limits # of unACKed bytes to `rwnd`
  - guarantees receiver's buffer doesn't overflow

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TCP Connection Management

Recall: TCP sender, receiver establish "connection" before exchanging data segments

- initialize TCP variables:
  - seq. #s
  - buffers, flow control info (e.g. `RcvWindow`)
- *client:* connection initiator

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

- *server:* contacted by client

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

Three way handshake:

Step 1: client host sends TCP SYN segment to server

- specifies initial seq #
- no data

Step 2: server host receives SYN, replies with SYNACK segment

- server allocates buffers
- specifies server initial seq. #

Step 3: client receives SYNACK, replies with ACK segment, which may contain data

TCP Connection Management (cont.)

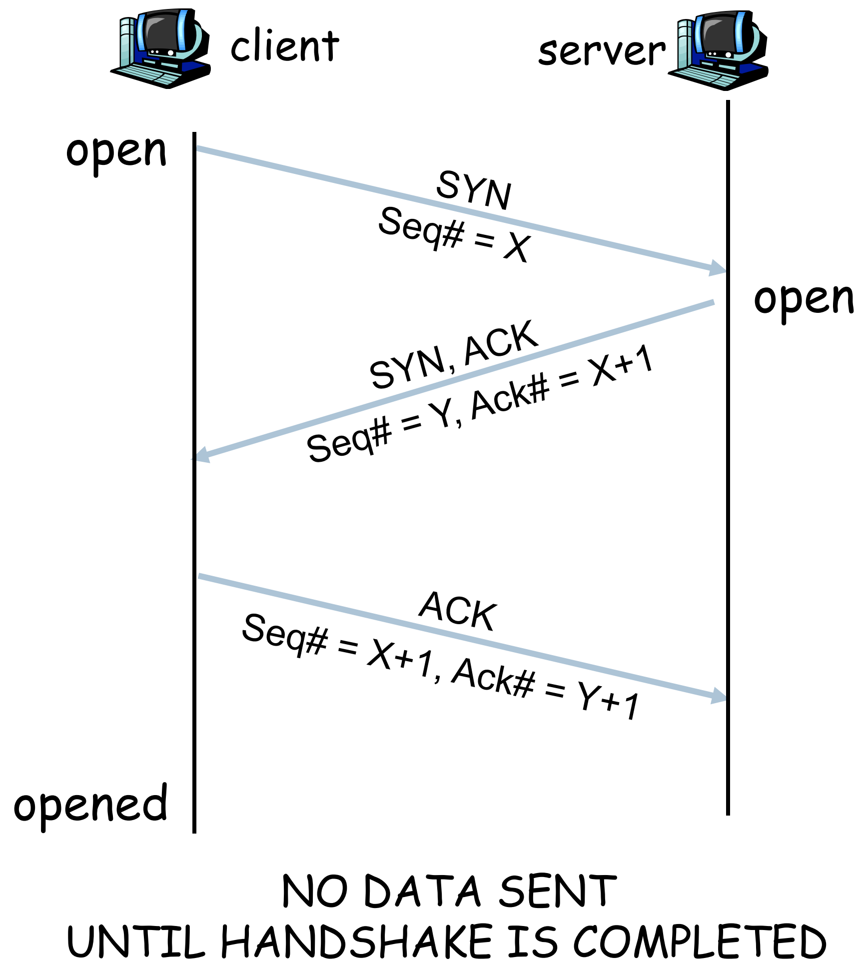
Starting a connection:

client opens a Socket:
    three-way handshake

Step 1: client end system sends TCP SYN control segment to server

Step 2: server receives SYN, replies with SYN, ACK

Step 3: client replies with ACK



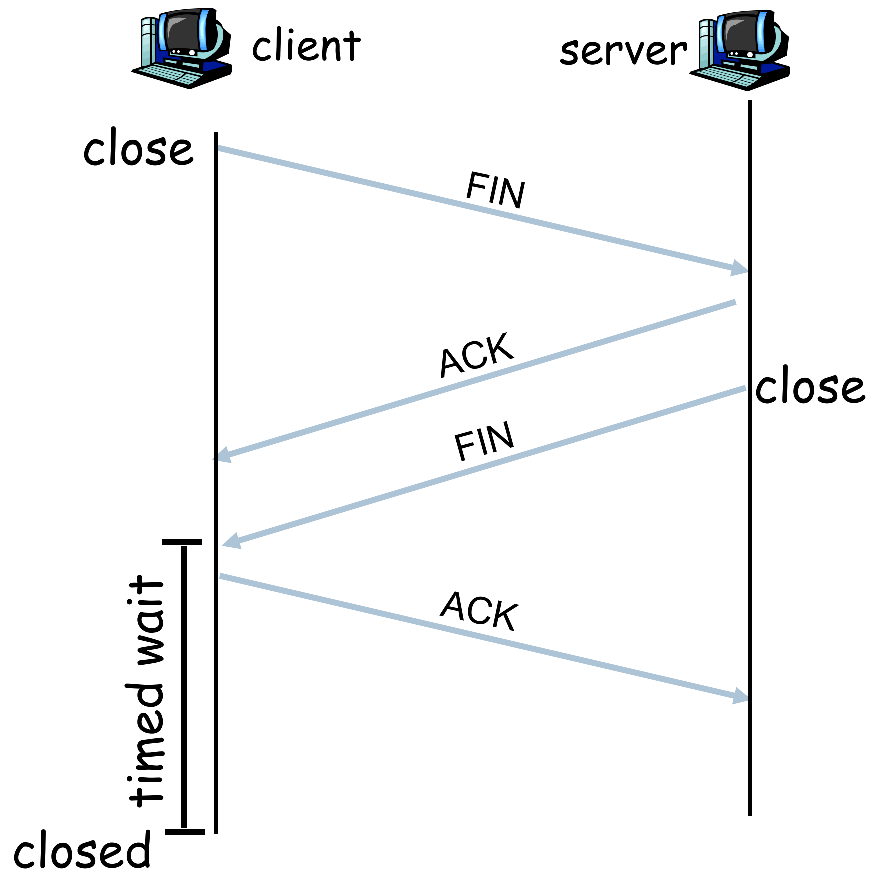
TCP Connection Management (cont.)

<u>Closing a connection:</u>

client closes socket:
`clientSocket.close();`

<u>Step 1:</u> client end system sends TCP FIN control segment to server

<u>Step 2:</u> server receives FIN, replies with ACK. Closes connection, sends FIN.



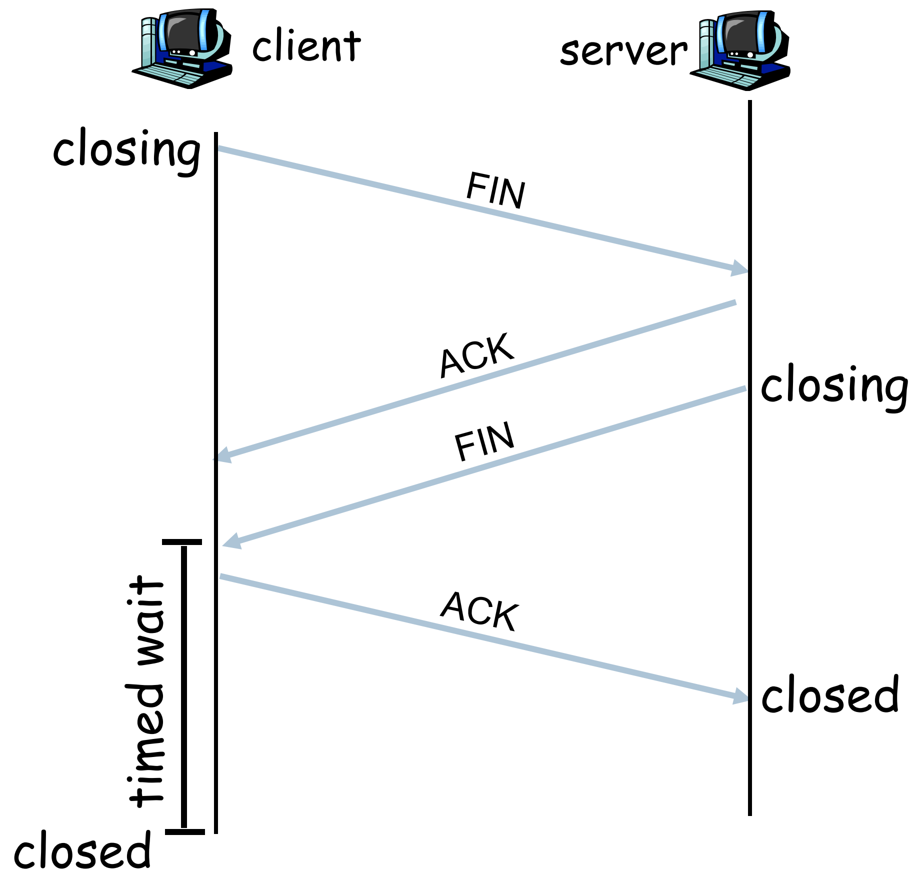
TCP Connection Management (cont.)

Step 3: client receives FIN, replies with ACK.

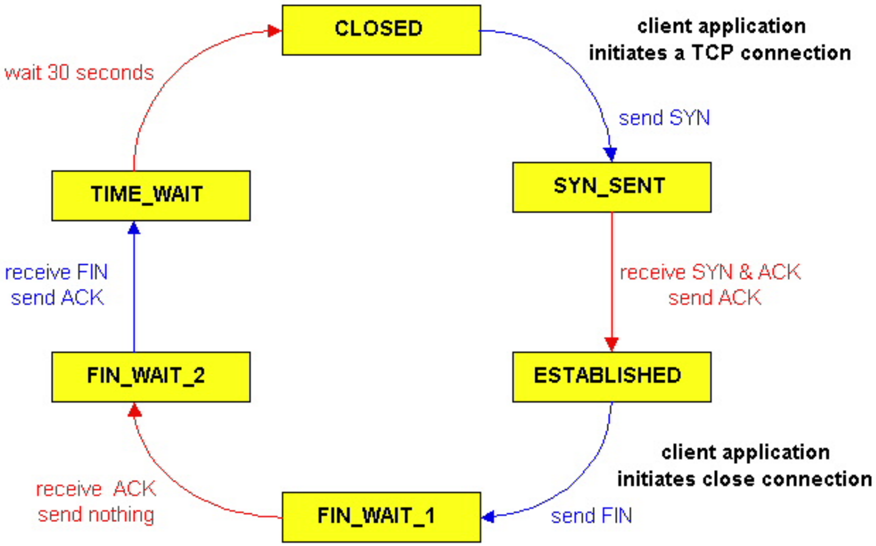
- Enters "timed wait" - will respond with ACK to received FINs

Step 4: server, receives ACK. Connection closed.

Note: with small modification, can handle simultaneous FINs.

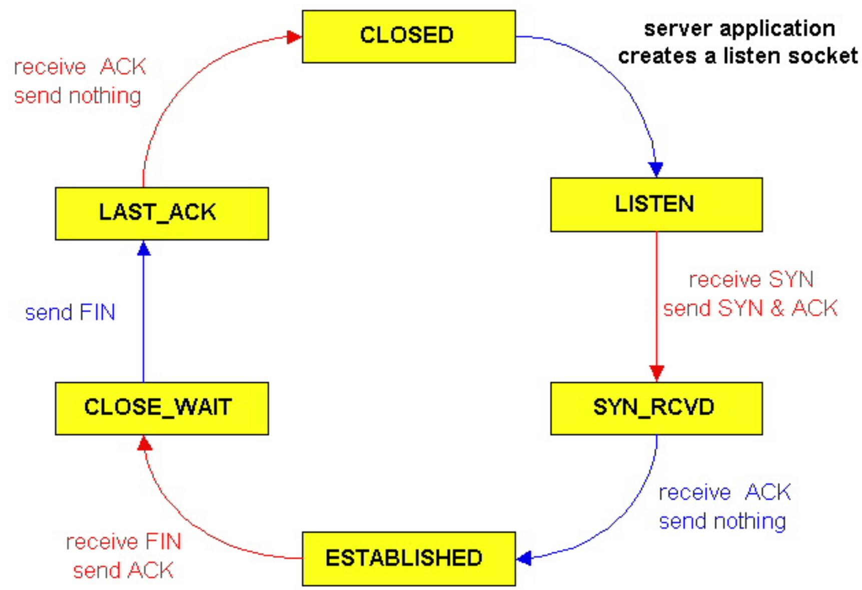


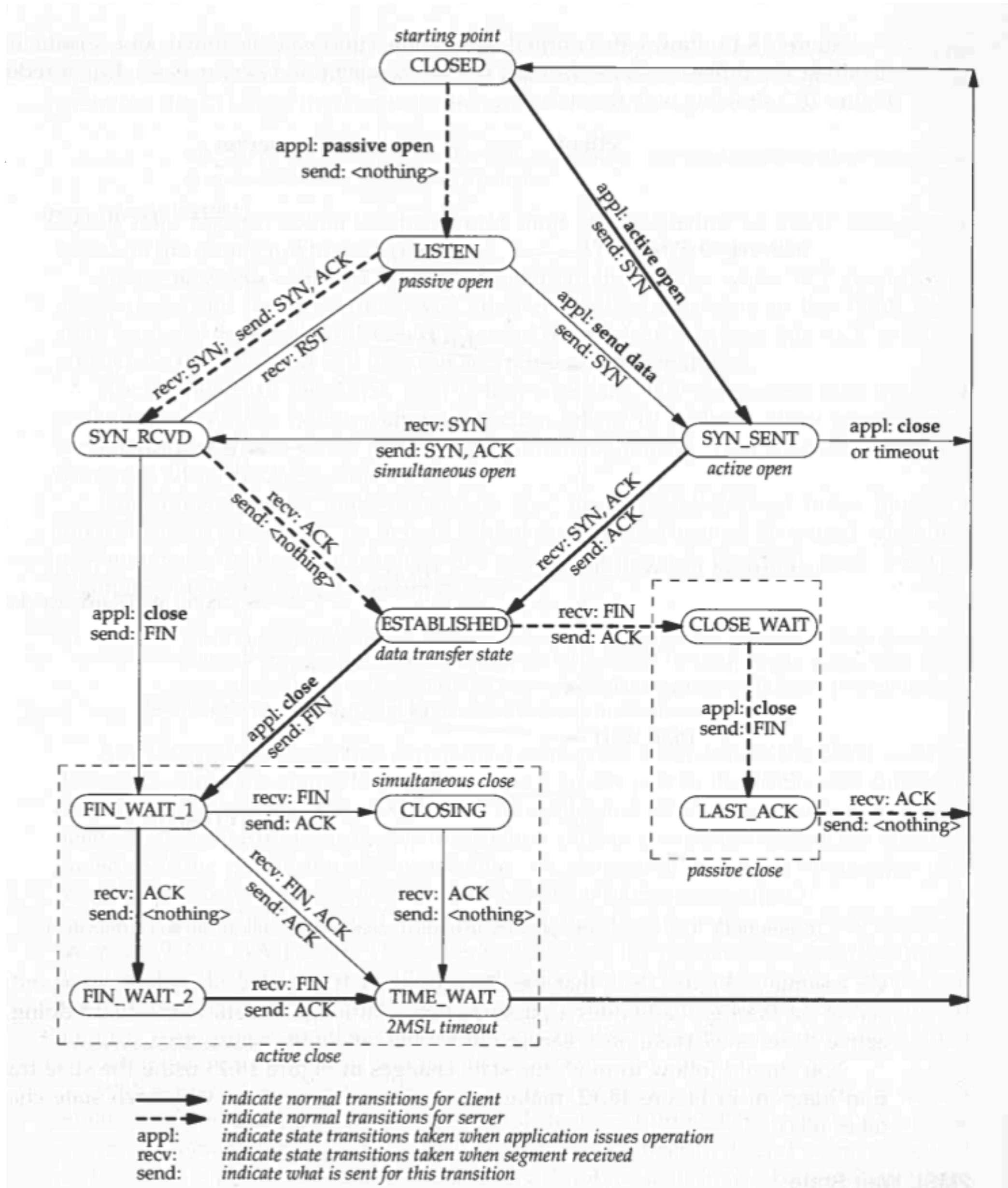
TCP Connection Management (cont)



TCP client lifecycle

TCP server lifecycle





Source: TCP/IP Illustrated, Volume1

TCP Syn and Security

- Port Scanning
  - Send Syn packets to many ports
  - Enumerate open ports
  - "Fingerprint" listening applications to find vulnerabilities

- Syn flood
  - DDoS attack (attack on Availability)
  - Exhausts resources
  - Can be prevented using "Syn cookies"
    - Serv_Sin_Seq# = H(C_Seq, SrcPort, SrcIP, DstPort, DstIP, Salt)

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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!

Approaches towards congestion control

two broad approaches towards congestion control:

end-end congestion control:

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

network-assisted congestion control:

- routers provide feedback to end systems
  - single bit indicating congestion (SNA, DECbit, TCP/IP ECN, ATM)
  - explicit rate sender should send at

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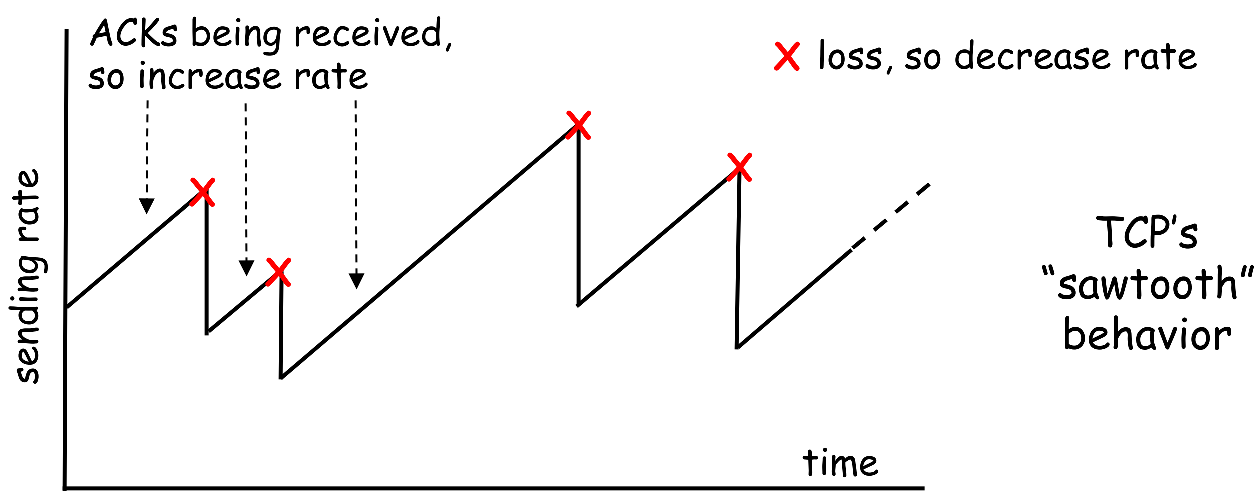
TCP congestion control:

- *goal:* TCP sender should transmit as fast as possible, but without congesting network
  - Q: how to find rate *just* below congestion level
- decentralized: each TCP sender sets its own rate, based on *implicit* feedback:
  - *ACK:* segment received (a good thing!), network not congested, so increase sending rate
  - *lost segment:* assume loss due to congested network, so decrease sending rate

TCP congestion control: bandwidth probing

- "probing for bandwidth": increase transmission rate on receipt of ACK, until eventually loss occurs, then decrease transmission rate
  - continue to increase on ACK, decrease on loss (since available bandwidth is changing, depending on other connections in network)



- Q: how fast to increase/decrease?
  - details to follow

TCP Congestion Control: details

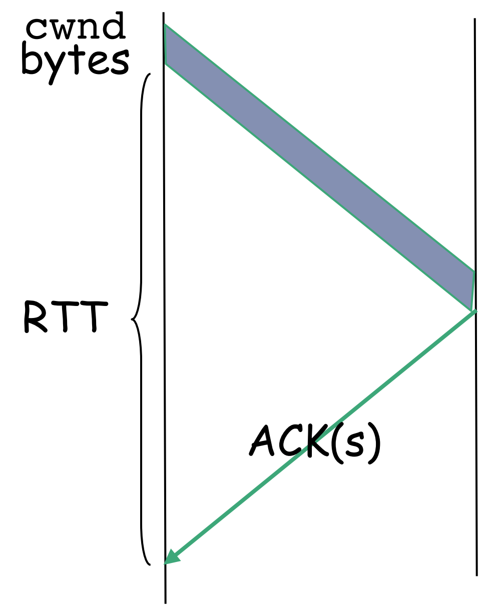
- sender limits rate by limiting number of unACKed bytes "in pipeline":

  `LastByteSent-LastByteAcked ≤ cwnd`

  - **`cwnd`**: differs from **`rwnd`** (how, why?)
  - sender limited by **`min(cwnd,rwnd)`**
  - For simplicity we assume cwnd << rwnd
- roughly,

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

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



TCP Congestion Control: more details

segment loss event: reducing `cwnd`

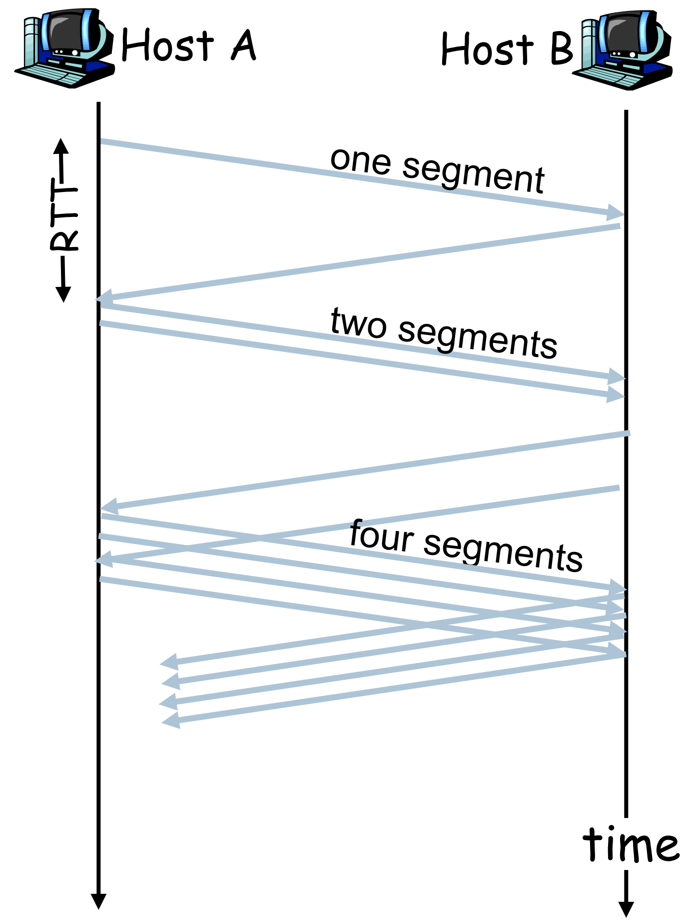
- timeout: no response from receiver
  - cut **`cwnd`** to 1 MSS
- 3 duplicate ACKs: at least some segments getting through (recall fast retransmit)
  - cut **`cwnd`** in half, less aggressively than on timeout

ACK received: increase `cwnd`

- slowstart phase:
  - increase exponentially fast (despite name) at connection start, or following timeout
- congestion avoidance:
  - increase linearly

TCP Slow Start

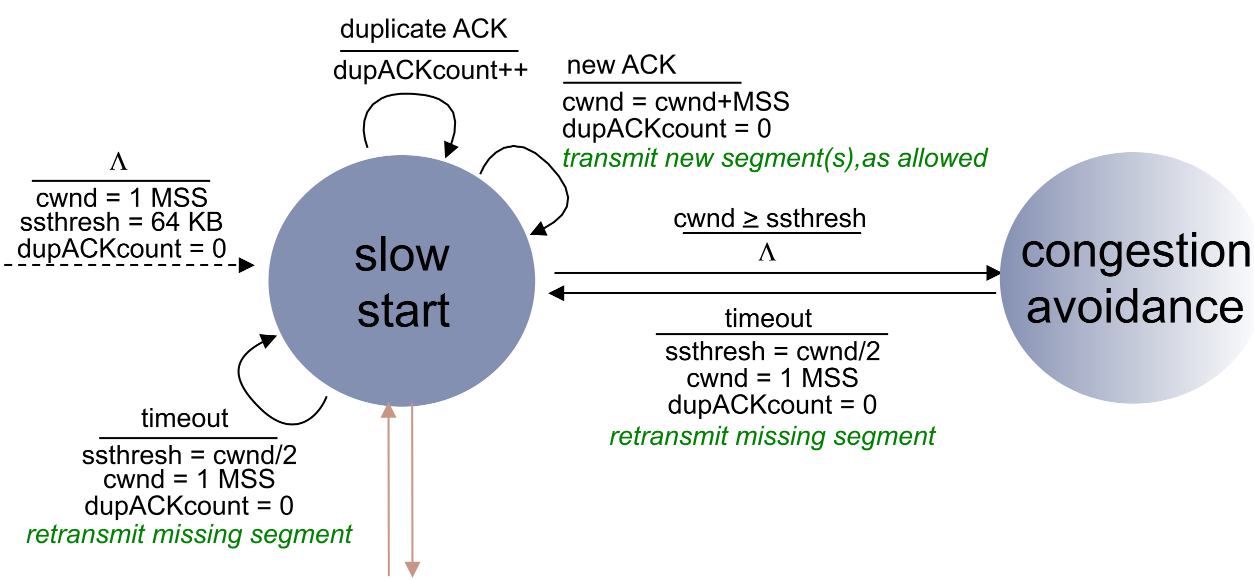
- when connection begins, **cwnd** = 1 MSS
  - example: MSS = 500 bytes & RTT = 200 msec
  - initial rate = 20 kbps
- available bandwidth may be >> MSS/RTT
  - desirable to quickly ramp up to respectable rate
- increase rate exponentially until first loss event or when threshold reached
  - double **cwnd** every RTT
  - done by incrementing **cwnd** by 1 for every ACK received



Transitioning into/out of slowstart

`ssthresh:` **`cwnd`** threshold maintained by TCP

- on loss event: set **`ssthresh`** to **`cwnd/2`**
  - remember (half of) TCP rate when congestion last occurred
- when **`cwnd >= ssthresh`**: transition from slowstart to congestion avoidance phase



Increase rate more cautiously

TCP: congestion avoidance

- when **cwnd > ssthresh** grow **cwnd** linearly
  - increase **cwnd** by 1 MSS per RTT
  - approach possible congestion slower than in slowstart
  - implementation: **cwnd = cwnd + MSS/cwnd** for each ACK received

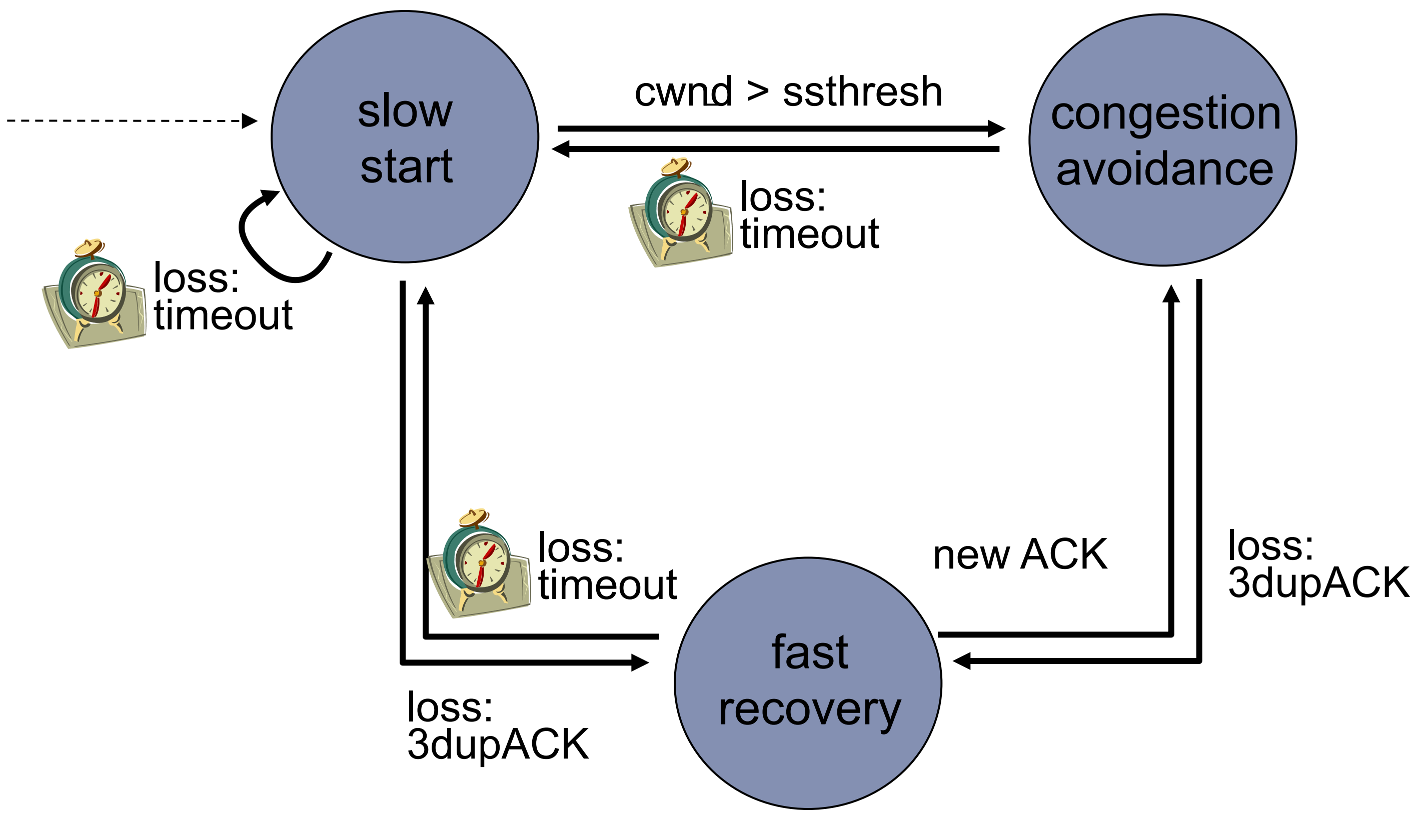
AIMD

- ACKs: increase **cwnd** by 1 MSS per RTT: additive increase
- loss: cut **cwnd** in half (non-timeout-detected loss): multiplicative decrease

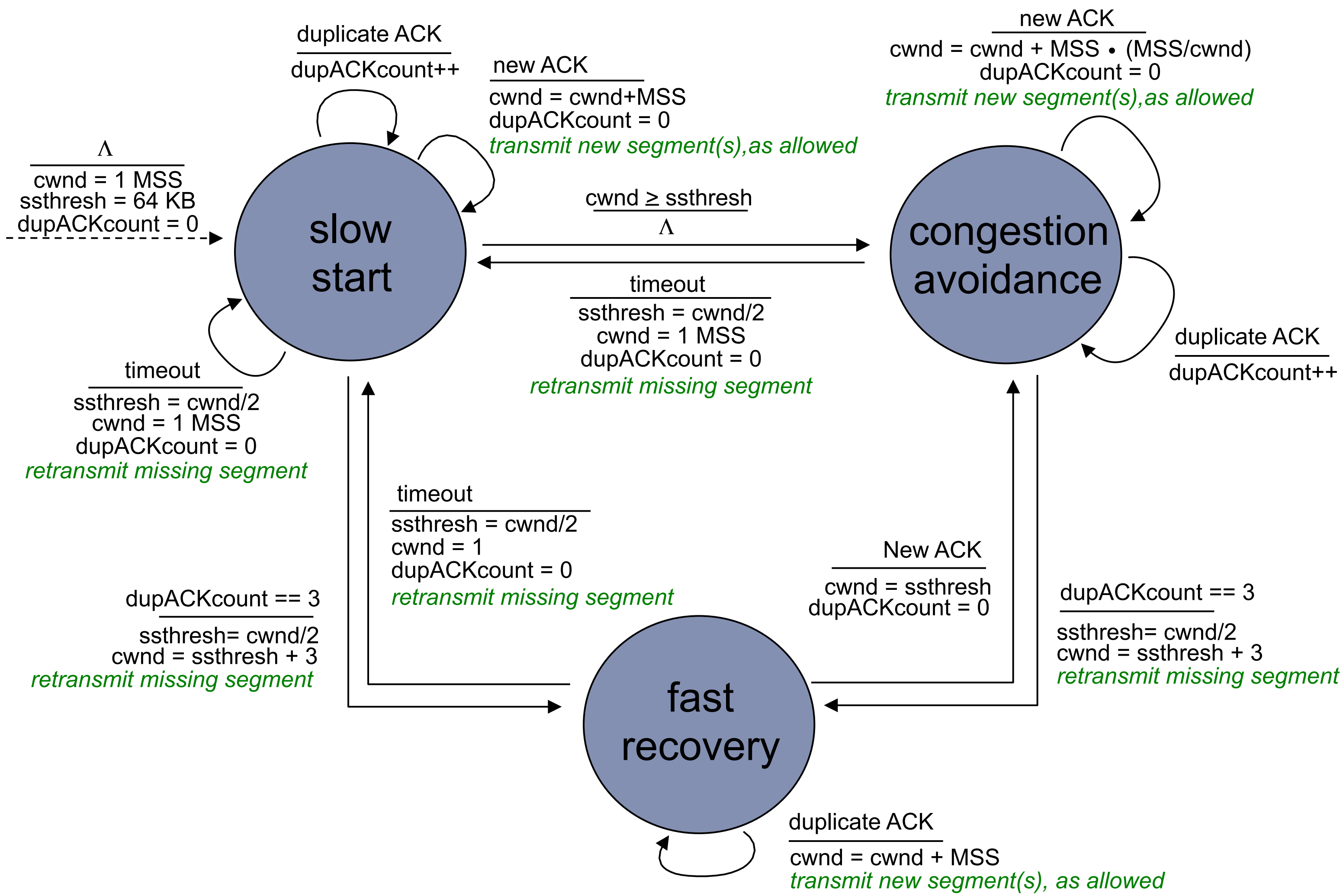
AIMD: Additive Increase Multiplicative Decrease

TCP congestion control FSM: overview

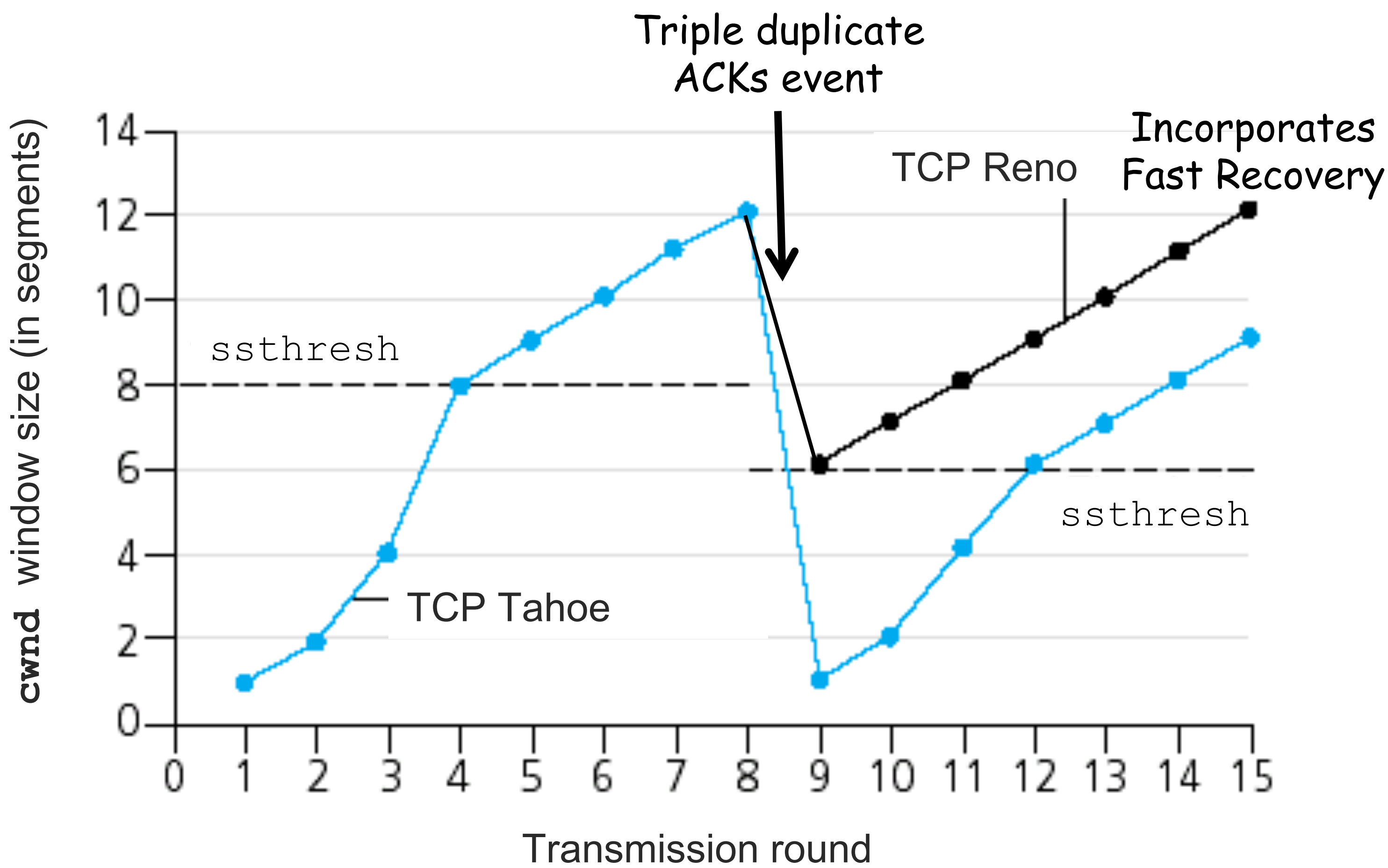
Fast Recovery introduced in 1997 (RFC2001)



TCP congestion control FSM: details



Popular "flavors" of TCP



Some more details on the slow start phase

- Things have changed since 1989 or 1997
  - 1989: RFC1122 – "Requirements for Internet Hosts -- Communication Layers"
  - 1997: RFC2001 – "TCP Slow Start, Congestion Avoidance, Fast Retransmit, and Fast Recovery Algorithms"

RFC 3390 Increasing TCP's Initial Window October 2002

The upper bound for the initial window is given more precisely in (1):

min (4*MSS, max (2*MSS, 4380 bytes)) (1)

Note: Sending a 1500 byte packet indicates a maximum segment size (MSS) of 1460 bytes (assuming no IP or TCP options). Therefore, limiting the initial window's MSS to 4380 bytes allows the sender to transmit three segments initially in the common case when using 1500 byte packets.

Note: some applications ***cheat*** on the slow start!
http://blog.benstrong.com/2010/11/google-and-microsoft-cheat-on-slow.html

Other cases of "cheating"

- Optimizations in Google Chrome:
  - Goal: make pages load as fast as possible
  - Reduce loading time to "instantaneous"
  - http://www.igvita.com/posa/high-performance-networking-in-google-chrome/

Summary: TCP Congestion Control

- when **`cwnd < ssthresh`**, sender in slow-start phase, window grows exponentially.
- when **`cwnd >= ssthresh`**, sender is in congestion-avoidance phase, window grows linearly.
- when triple duplicate ACK occurs, **`ssthresh`** set to **`cwnd/2`**, **`cwnd`** set to ~ **`ssthresh`**
- when timeout occurs, **`ssthresh`** set to **`cwnd/2`**, **`cwnd`** set to 1 MSS.

TCP throughput (highly simplified!)

- *Q:* what's average throughout of TCP as function of window size, RTT?
  - ignoring slow start
- let W be window size (in bytes) when loss occurs.
  - when window is W, throughput is W/RTT
  - just after loss, window drops to W/2, throughput to W/2RTT.
  - average throughout: .75 W/RTT
    - Assuming TCP is in a "steady state"
    - One packet drop when rate increases to W/2

TCP Futures: TCP over "long, fat pipes"

- example: 1500 byte segments, 100ms RTT, want 10 Gbps throughput
  - high bandwidth-delay product
- requires window size W = 83,333 in-flight segments
- Average throughput in terms of loss rate L:

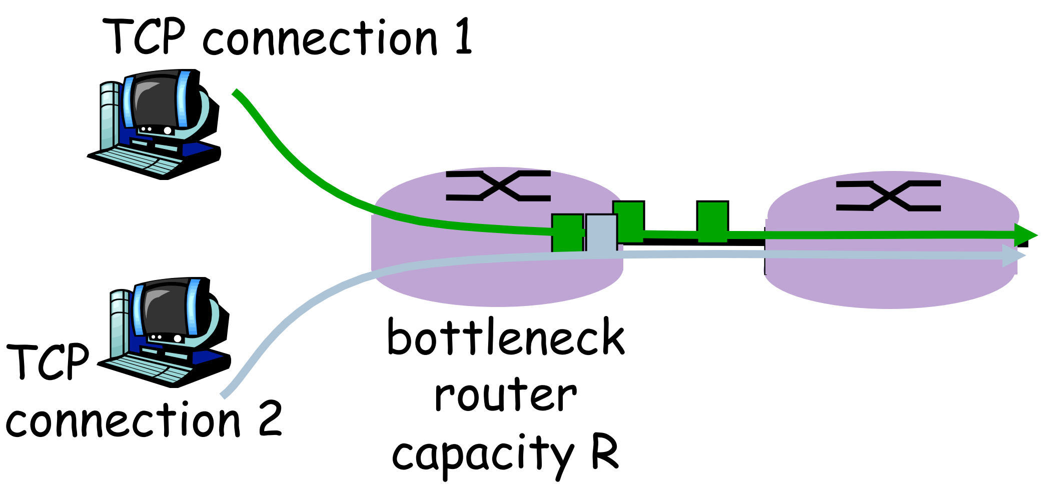
$$\frac{1.22 \cdot MSS}{RTT\sqrt{L}}$$

- ➜ $L = 2 \cdot 10^{-10}$ Wow!!! Probability of Loss needed to have high utilization

- new versions of TCP for high-speed
  - e.g., TCP CUBIC (default in Linux kernels > 2.6.19)
  - Modified congestion control algorithm

TCP Fairness

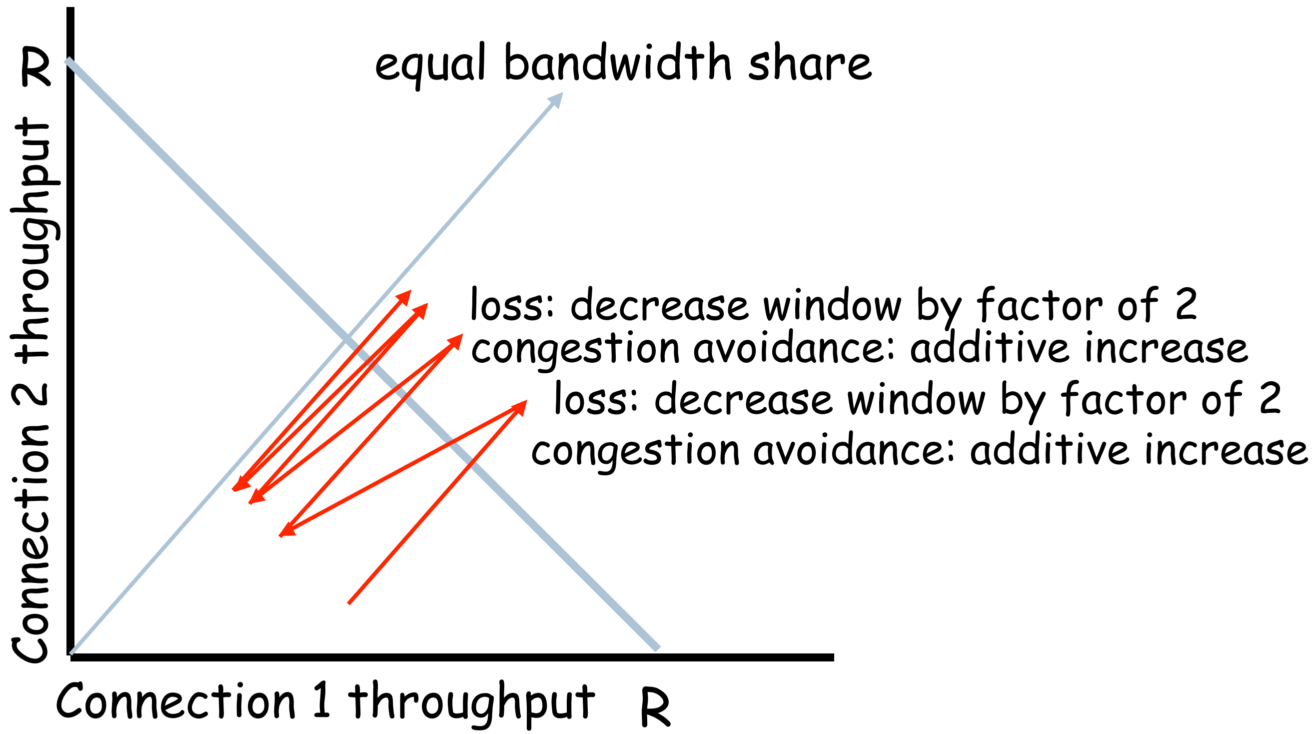
fairness goal: if K TCP sessions share same bottleneck link of bandwidth R, each should have average rate of R/K



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:
  - pump audio/video at constant rate, tolerate packet loss

Fairness and parallel TCP connections

- nothing prevents app from opening parallel connections between 2 hosts.
- web browsers do this
- example: link of rate R supporting 9 connections;
  - new app asks for 1 TCP, gets rate R/10
  - new app asks for 11 TCPs, gets ~R/2 !

Chapter 3: Summary

- principles behind transport layer services:
  - multiplexing, demultiplexing
  - reliable data transfer
  - flow control
  - congestion control
- instantiation and implementation in the Internet
  - UDP
  - TCP

Next:

- leaving the network "edge" (application, transport layers)
- into the network "core"