Chapter 3 Transport Layer

Layering in Internet protocol stack

Applications

… built on …

Reliable (or unreliable) transport

… built on …

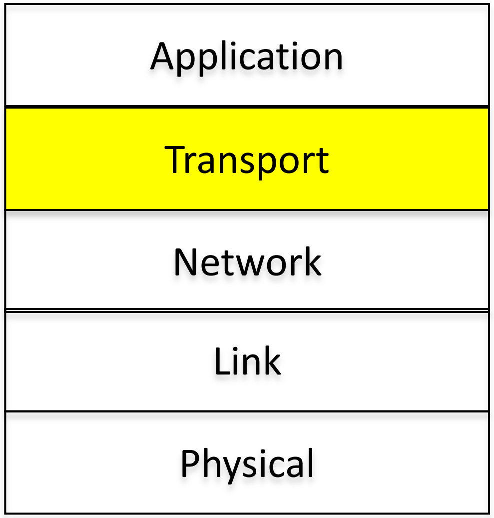
Best-effort global packet delivery

… built on …

Best-effort local packet delivery

… built on …

Physical transfer of bits



Modified from Scott Shenker (UC Berkeley): The Future of Networking, and the Past of Protocols

Chapter 3: Our Goals

- understand principles behind transport layer services:
  - multiplexing, de-multiplexing
  - reliable data transfer
  - flow control
  - congestion control
- Learn about transport layer protocols:
  - UDP: connectionless transport
  - TCP: connection-oriented reliable 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

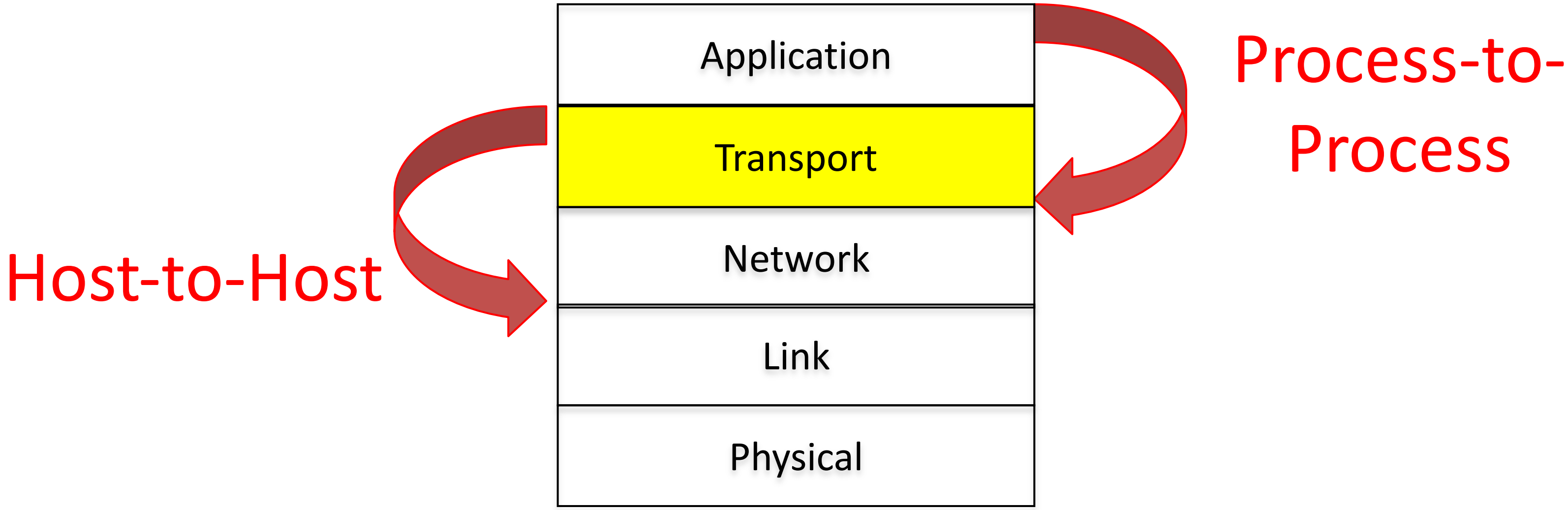
3.5 connection-oriented transport: TCP

- segment structure
- reliable data transfer
- flow control
- connection management

3.6 Principles of congestion control

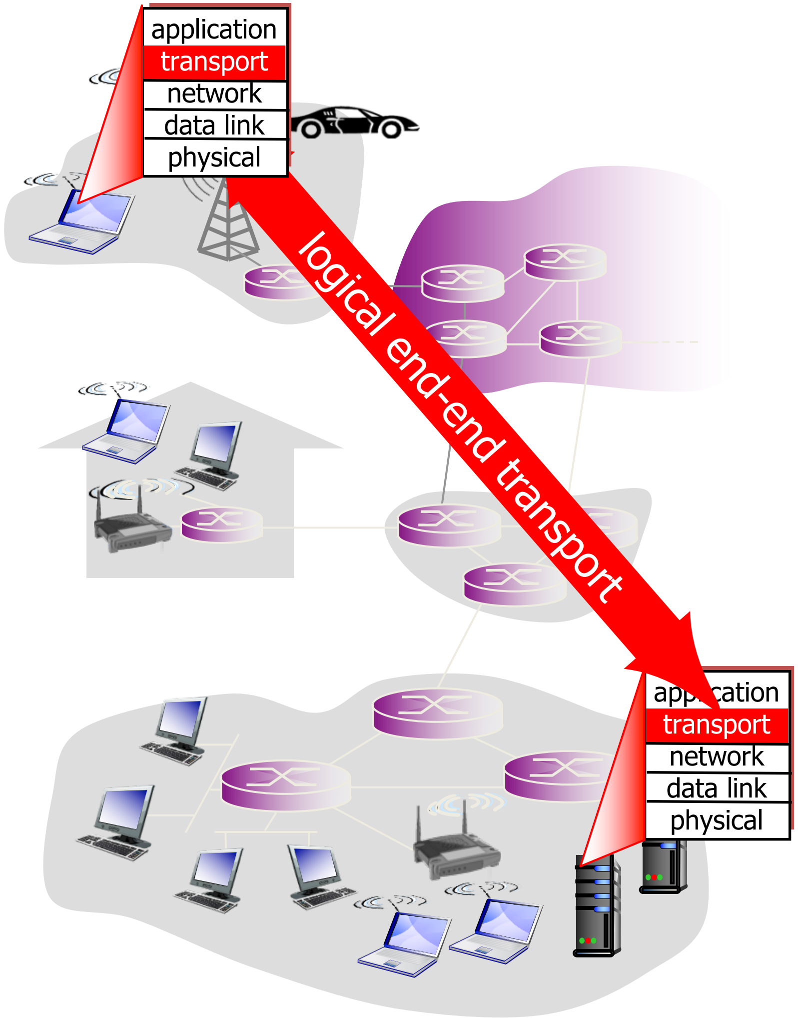
3.7 TCP congestion control

Upper and Lower Layers



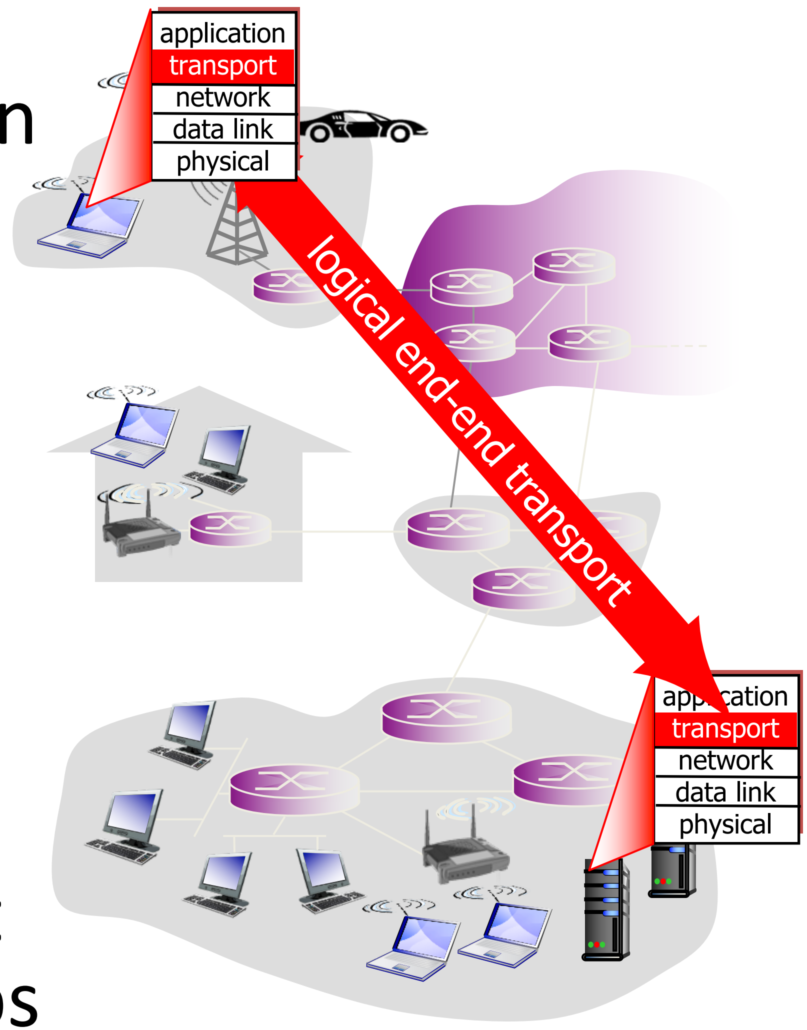
Transport services and protocols

- provide *logical communication* between app processes running on different hosts



Transport services and protocols

- 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



Transport vs. network layer

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

- *network layer:* logical communication between hosts

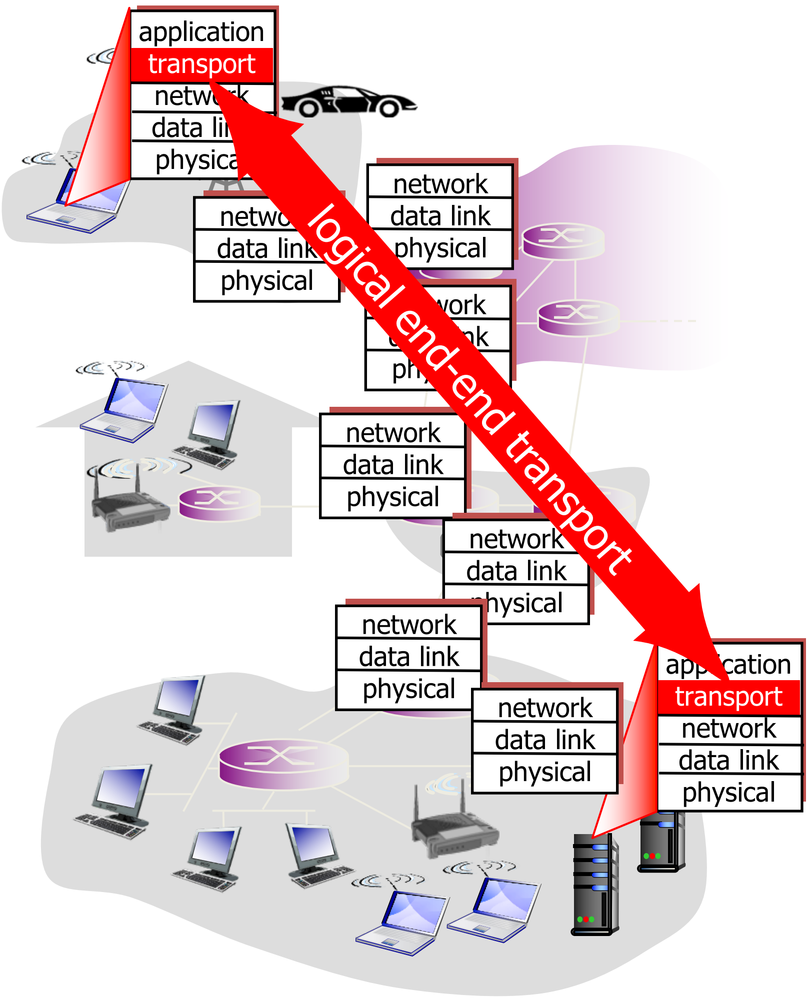
household analogy:

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

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

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



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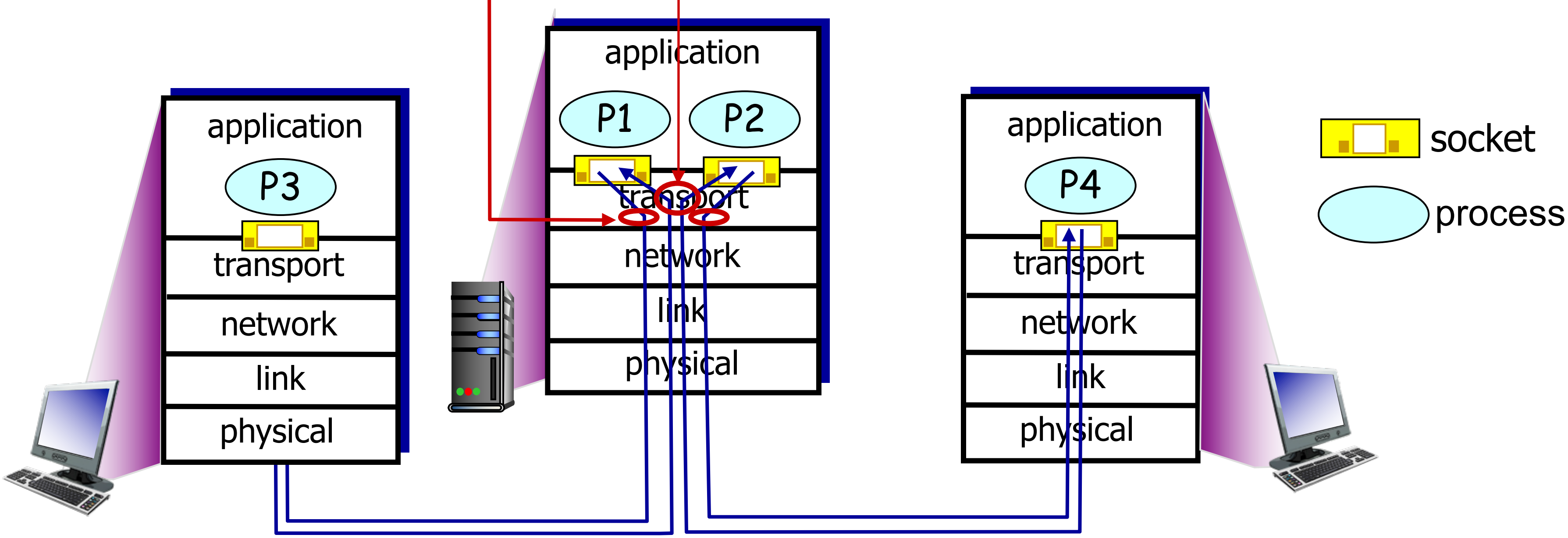
Multiplexing/demultiplexing

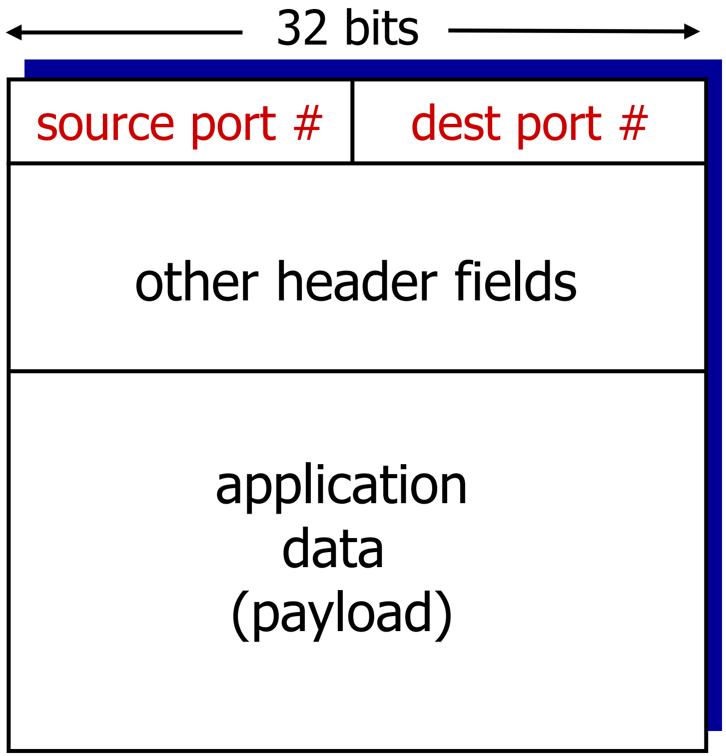
multiplexing at sender:
handle data from multiple sockets, add transport header (later used for demultiplexing)

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:* create and bind a socket:

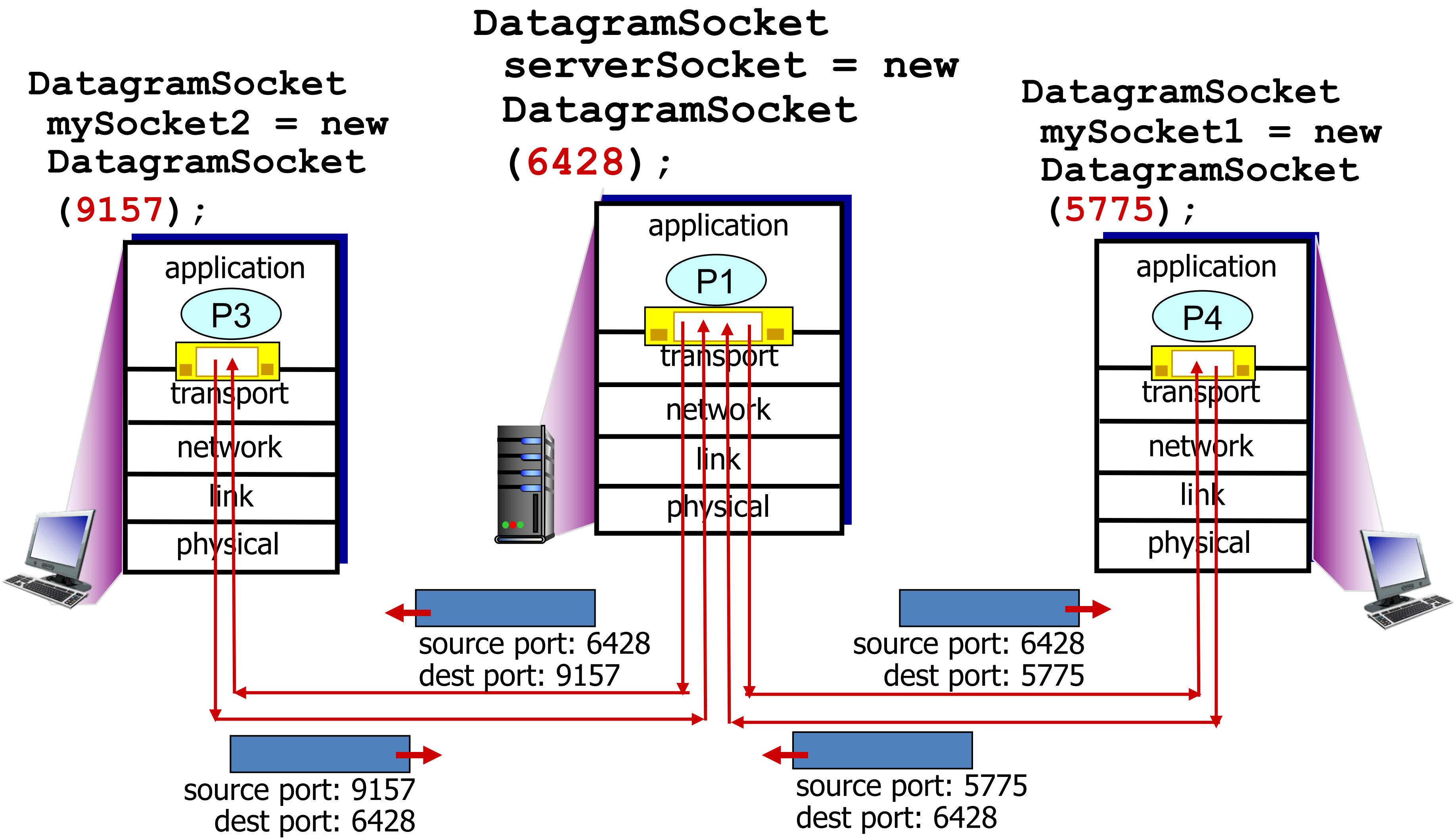
```
sockfd = socket(AF_INET , SOCK_DGRAM,1)  //
serv_addr.sin_port = htons(portno);  // local port # is specified
bind(sockfd, (struct sockaddr *) &serv_addr, sizeof(serv_addr));
```

Note: both destination IP address and destination port # are used

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

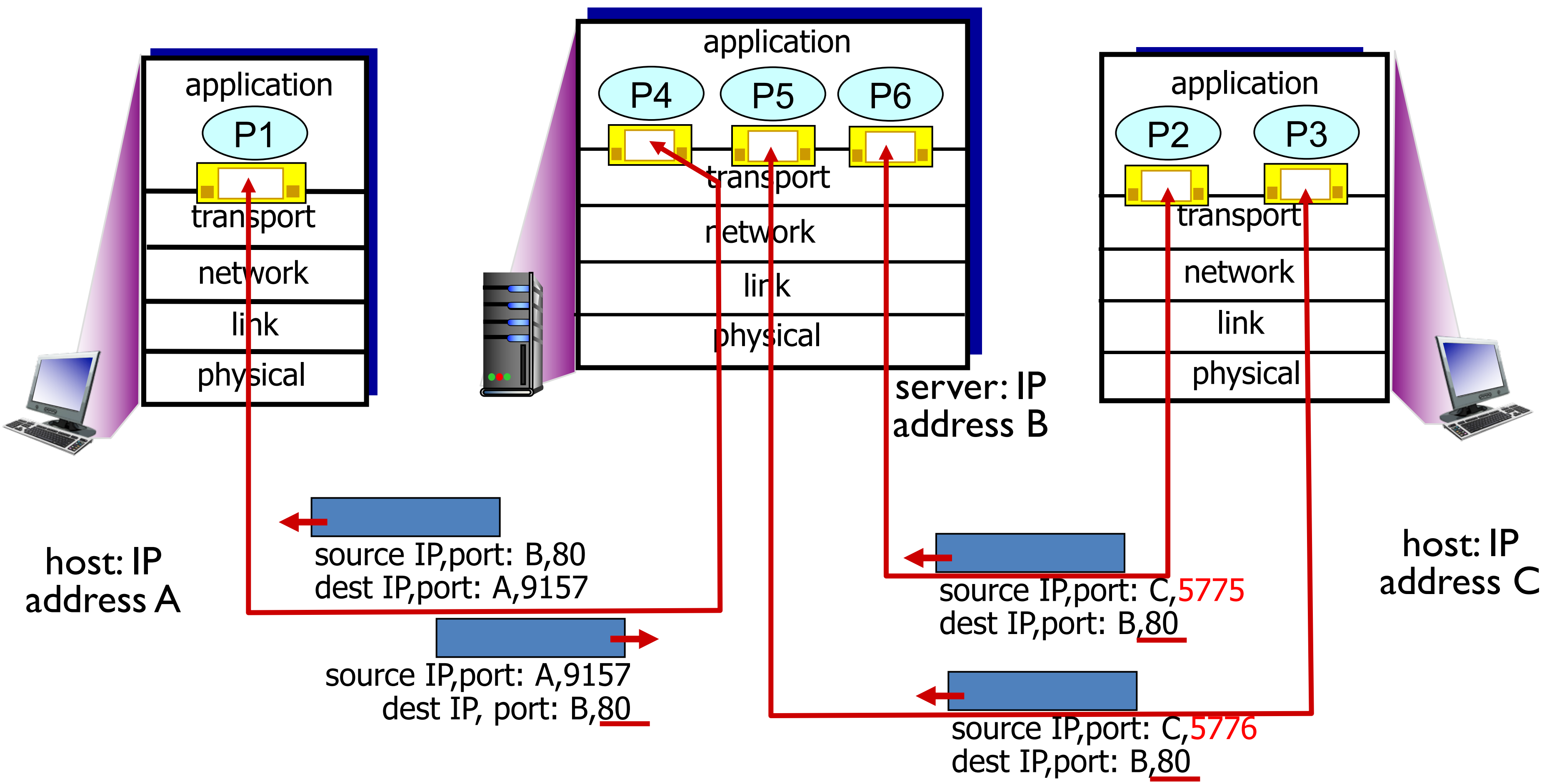
Connectionless demux: example



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

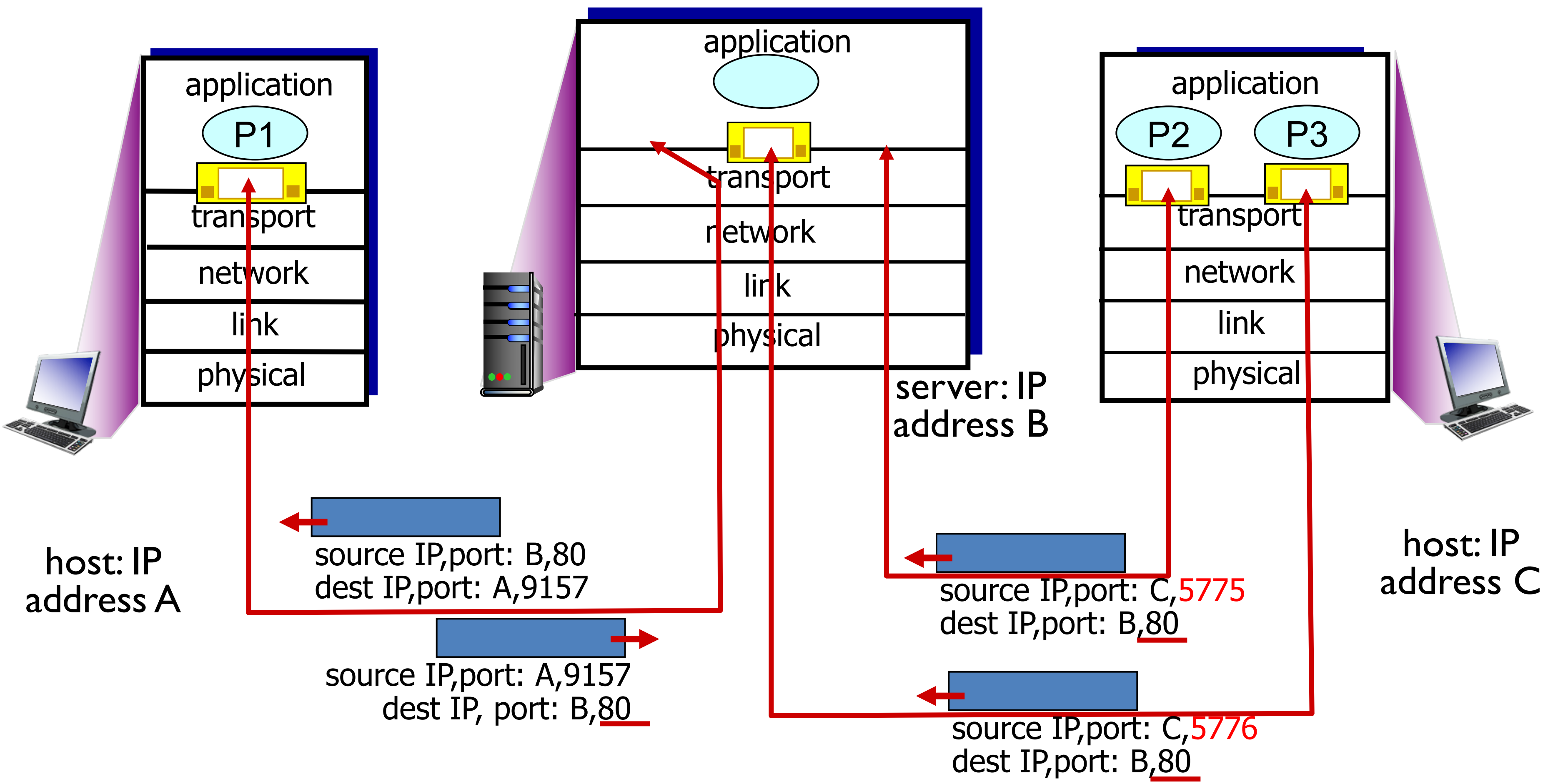
Connection-oriented demux: example



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

Question: What if UDP

How many sockets on the server side?



Question: why no checking of dest IP?

- check port# only in the UDP example
- Check port# + source IP in the TCP example
  - Correctness of IP address is ensured on the Networking layer
  - Destination IP: (not delivered to the node)

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UDP: User Datagram Protocol [RFC 768]

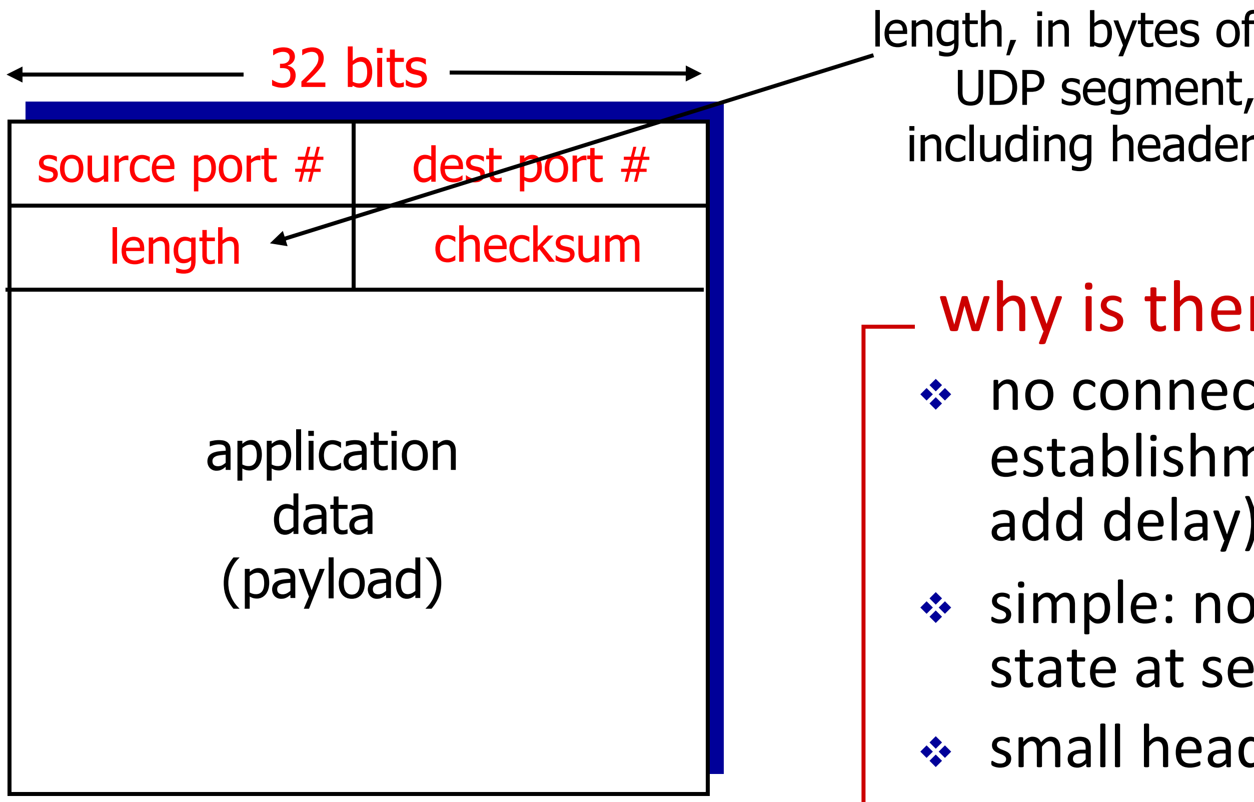
- "no frills," "bare bones" Internet transport protocol
- "best effort" service, UDP segments may be:
  - lost
  - delivered out-of-order to app
- *connectionless:*
  - no handshaking between UDP sender, receiver
  - each UDP segment handled independently of others

UDP: User Datagram Protocol

- UDP usage:
  - 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

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?* More later ….

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

After-class practice: UDP checksum

- 1st: 0110
- 2nd: 0101
- 3rd: 1000
- Calculate UDP checksum of 1st + 2nd + 3rd
- sum = 10011, -> 0011 + 1 (carryout) = 0100
- checksum = 1s complement = 1011
- Check: receiving 1011?
- Check: receiving 1001?
- Errors if receiving 1011??
  - See the notes for this slide for answers

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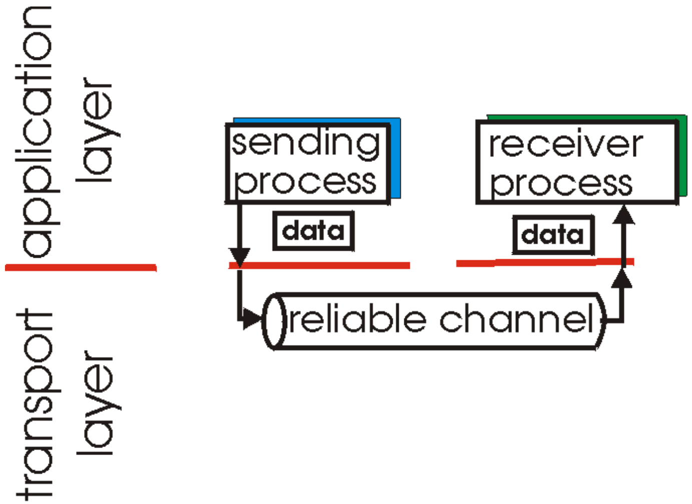
Principles of reliable data transfer

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

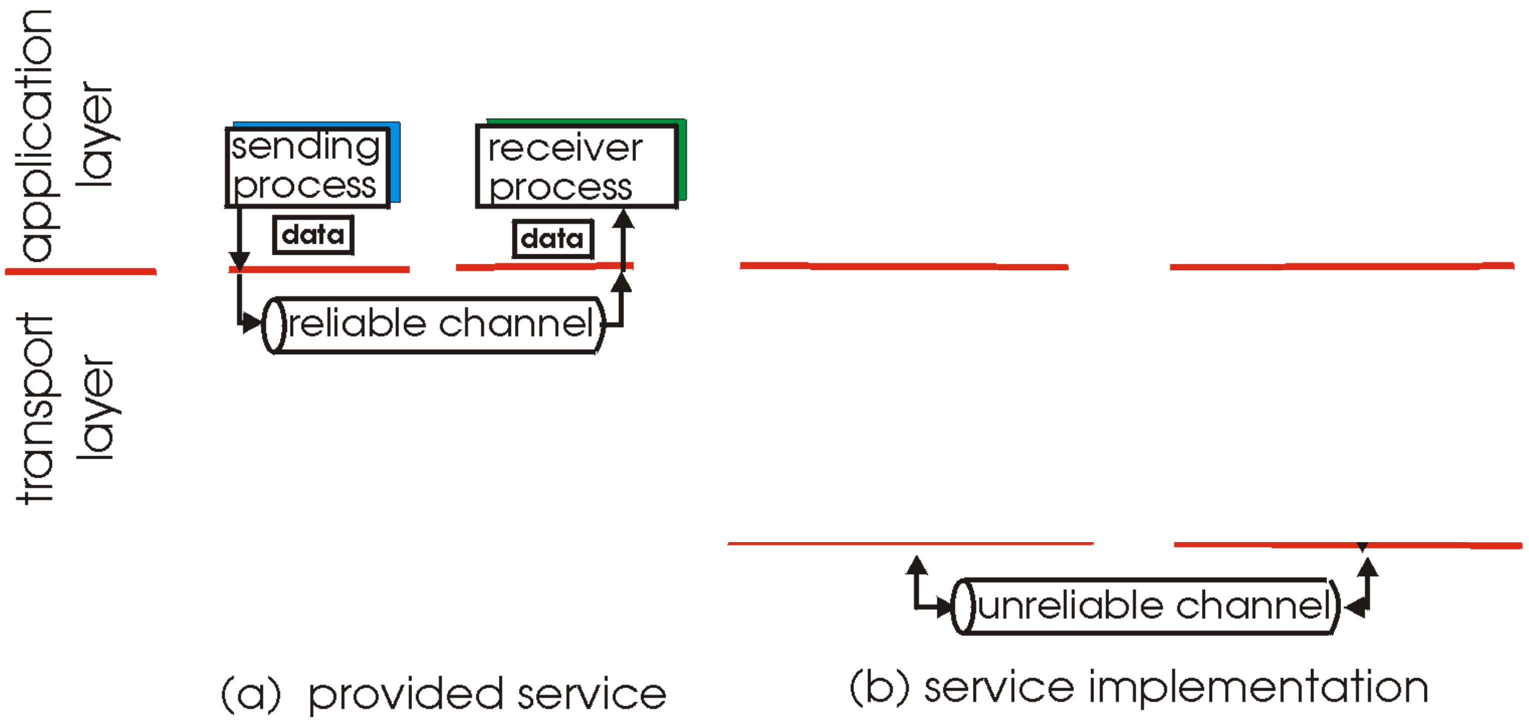


(a) provided service

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

Principles of reliable data transfer

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



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Principles of reliable data transfer

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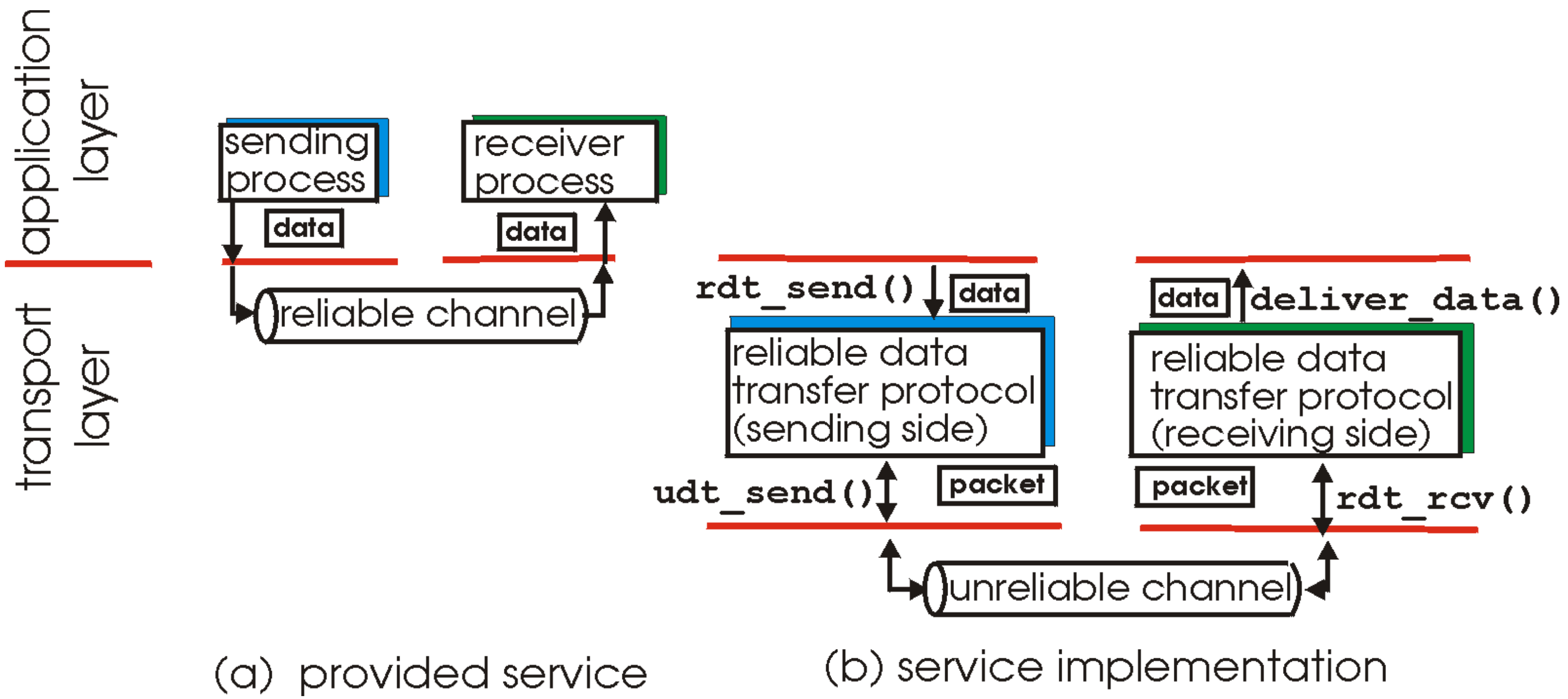
  - top-10 list of important networking topics!

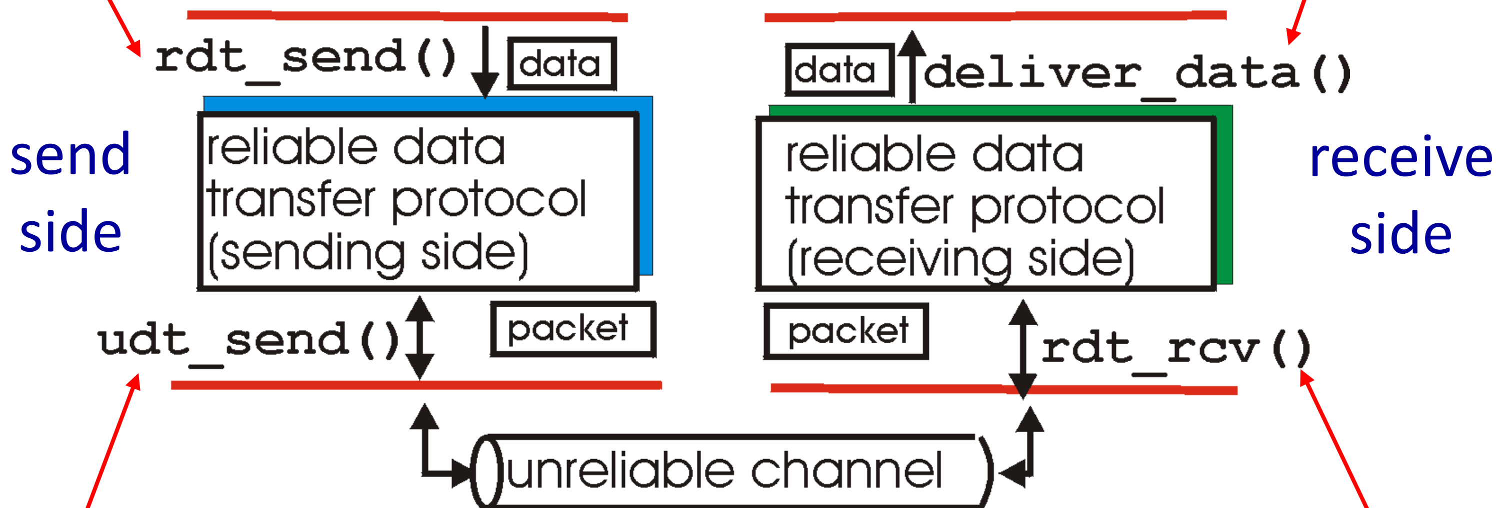


- characteristics of unreliable channel will determine complexity of reliable data transfer protocol (rdt)

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

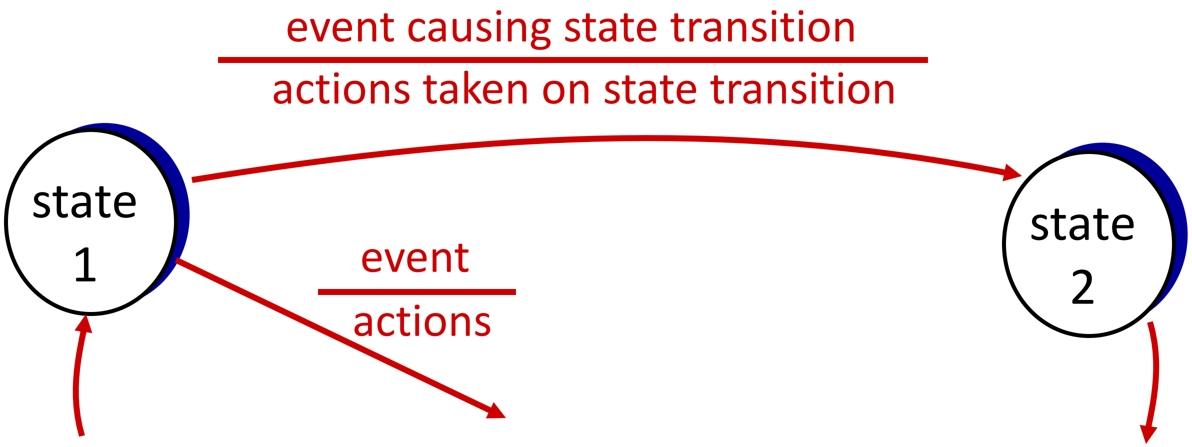
rdt_rcv(): called when packet arrives on rcv-side of channel

Reliable data transfer: getting started

we' ll:

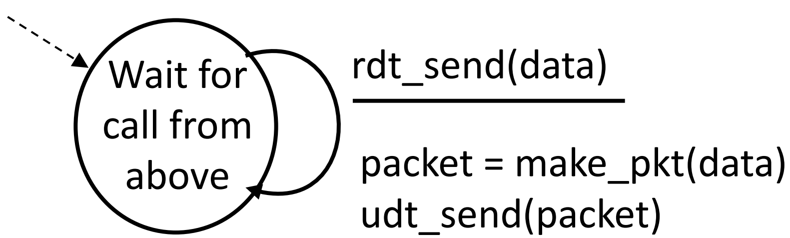
- incrementally develop sender, receiver sides of <u>r</u>eliable <u>d</u>ata <u>t</u>ransfer protocol (rdt)
- consider only unidirectional data transfer
  - but control info will flow on both directions!
- use finite state machines (FSM) to specify sender, receiver

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

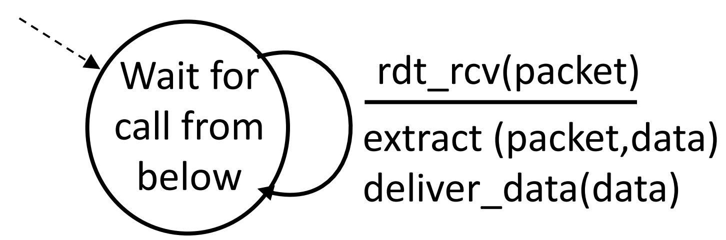


rdt1.0: reliable transfer over a reliable channel

- 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



sender



receiver

rdt2.0: channel with bit errors

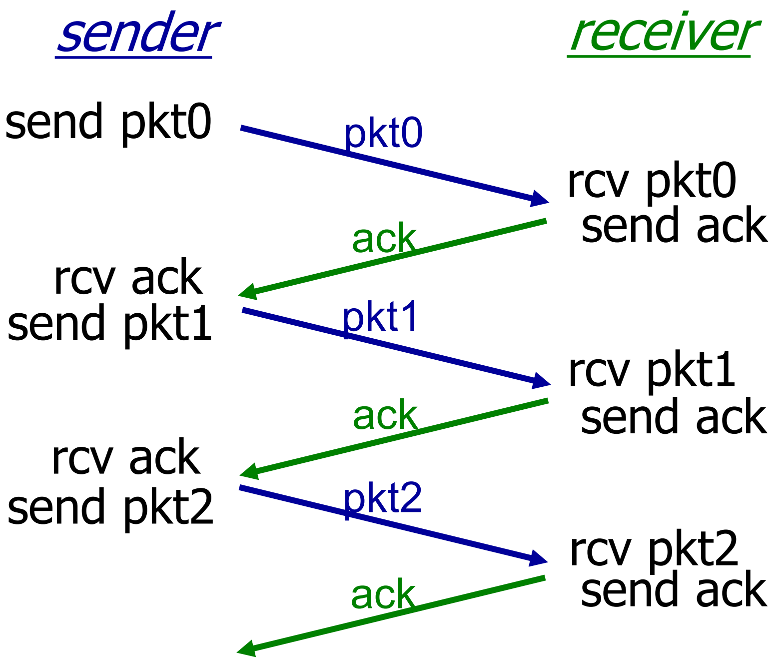
- underlying channel may flip bits in packet
  - checksum to detect bit errors
- question: how to recover from errors?

How do humans recover from "errors" during conversation?

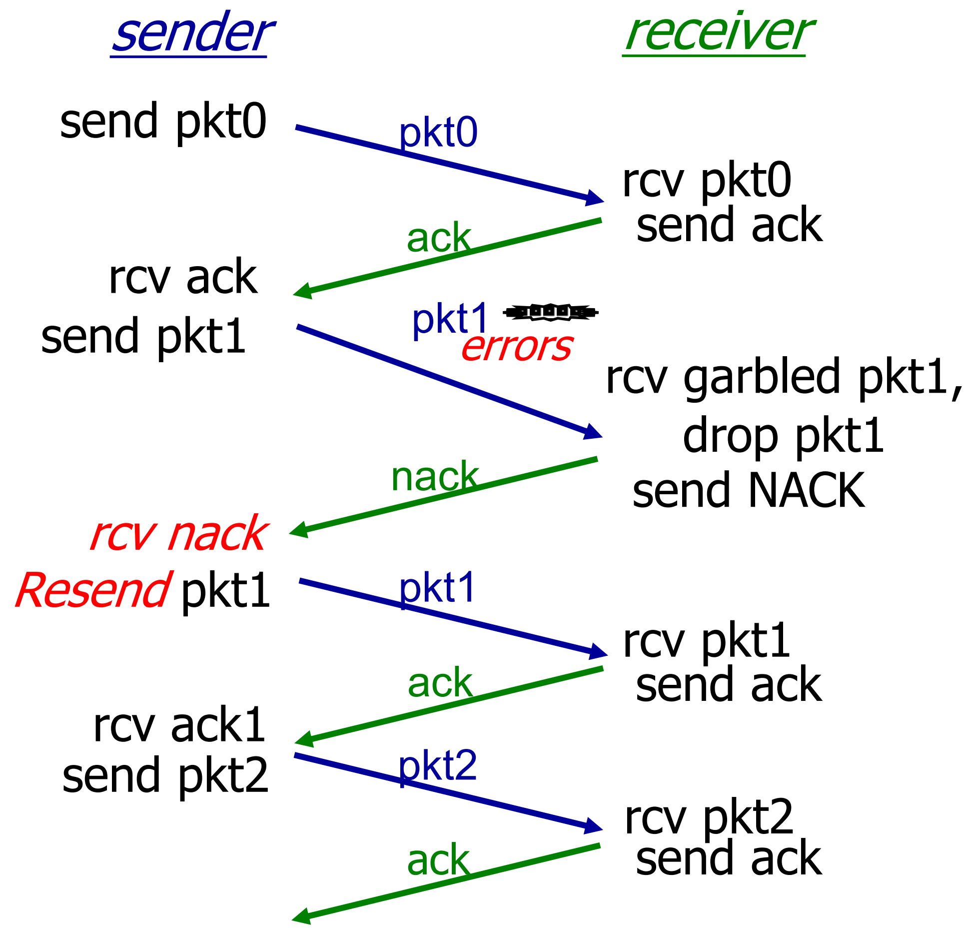
rdt2.0: channel with bit errors

- underlying channel may flip bits in packet
  - checksum to detect bit errors
- 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 at receiver
  - Feedback from receiver: control msgs (ACK,NAK) from receiver to sender
  - Retransmission at the sender

rdt2.0 in action

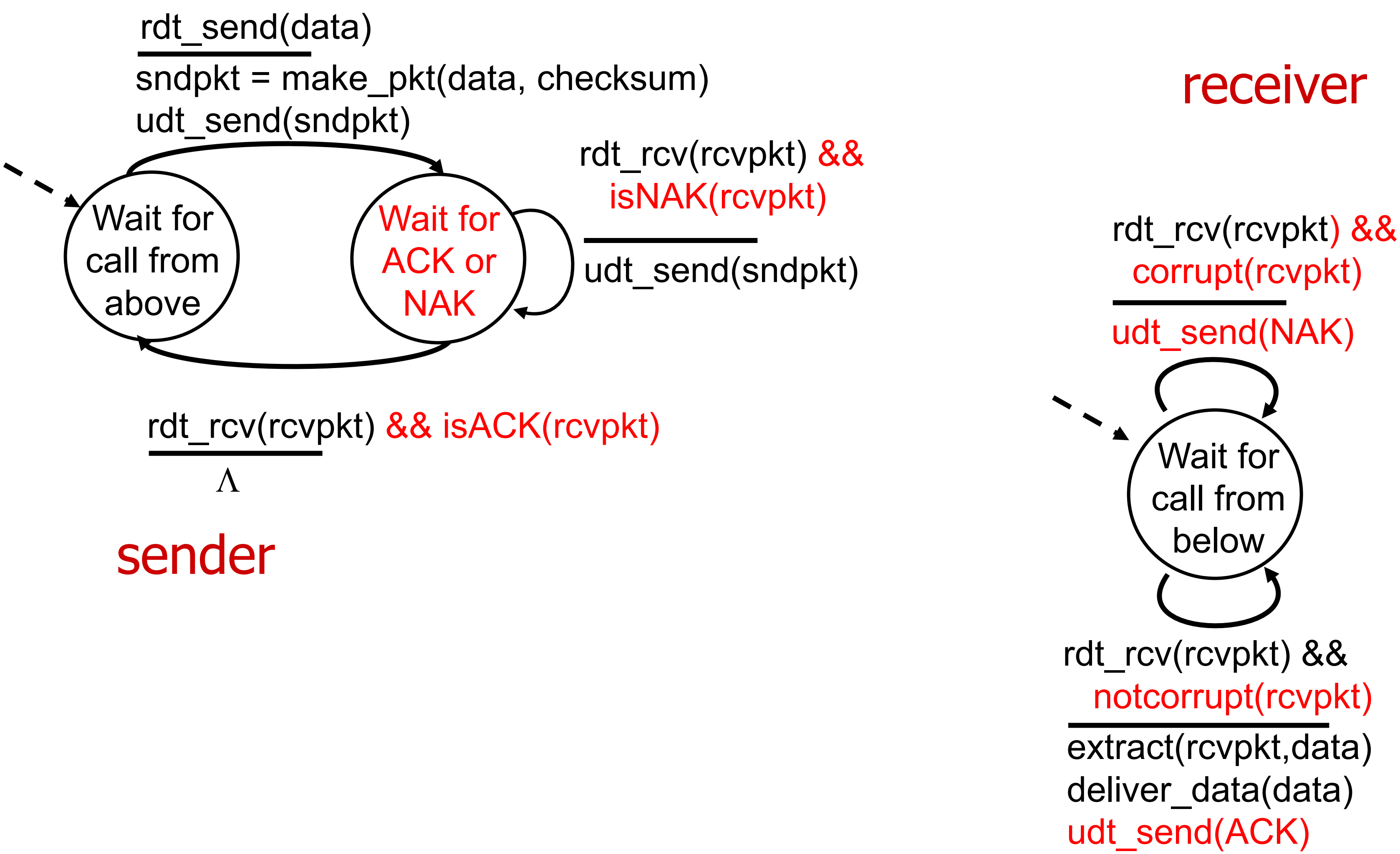


(a) no error

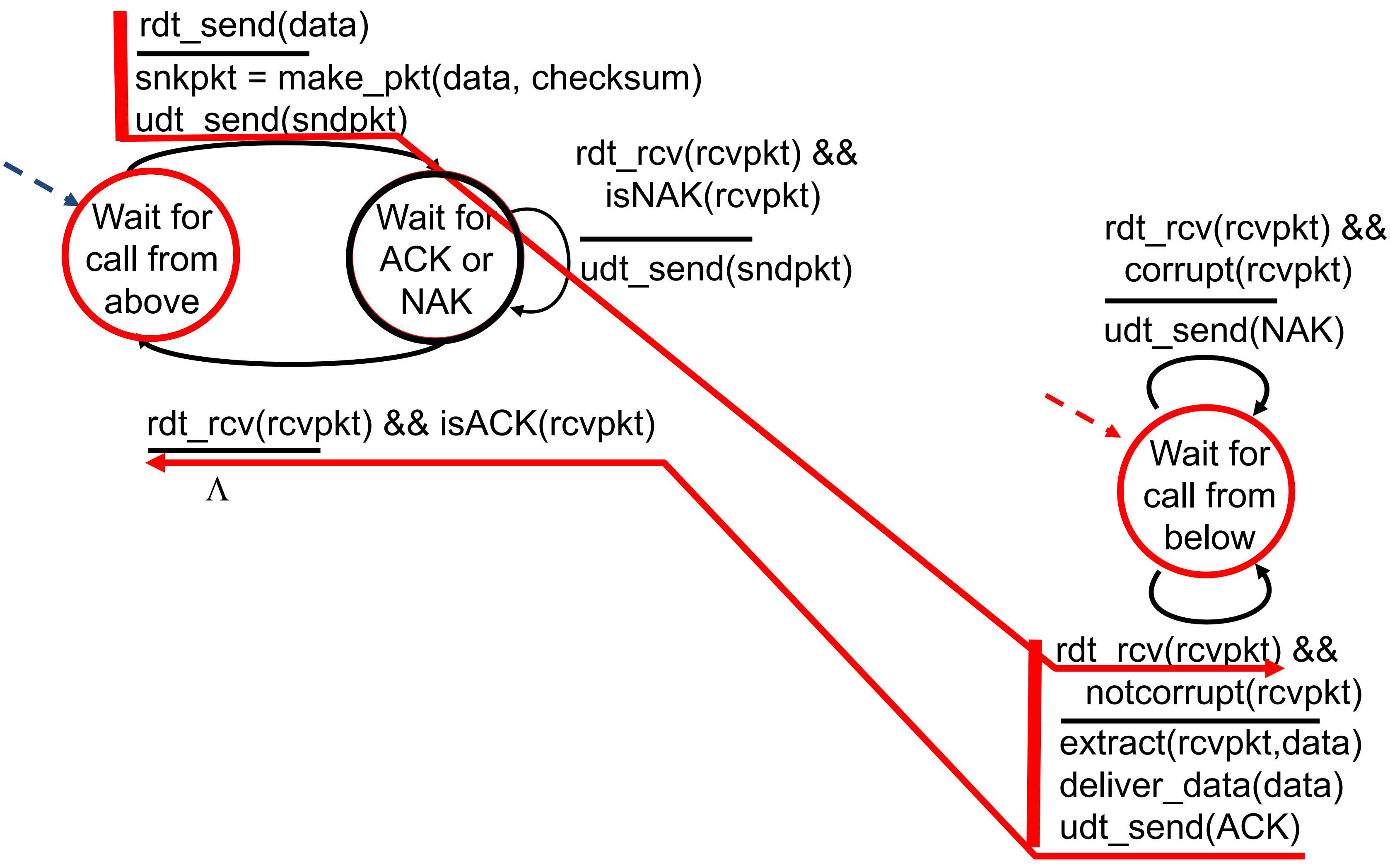


(b) packet with bit errors

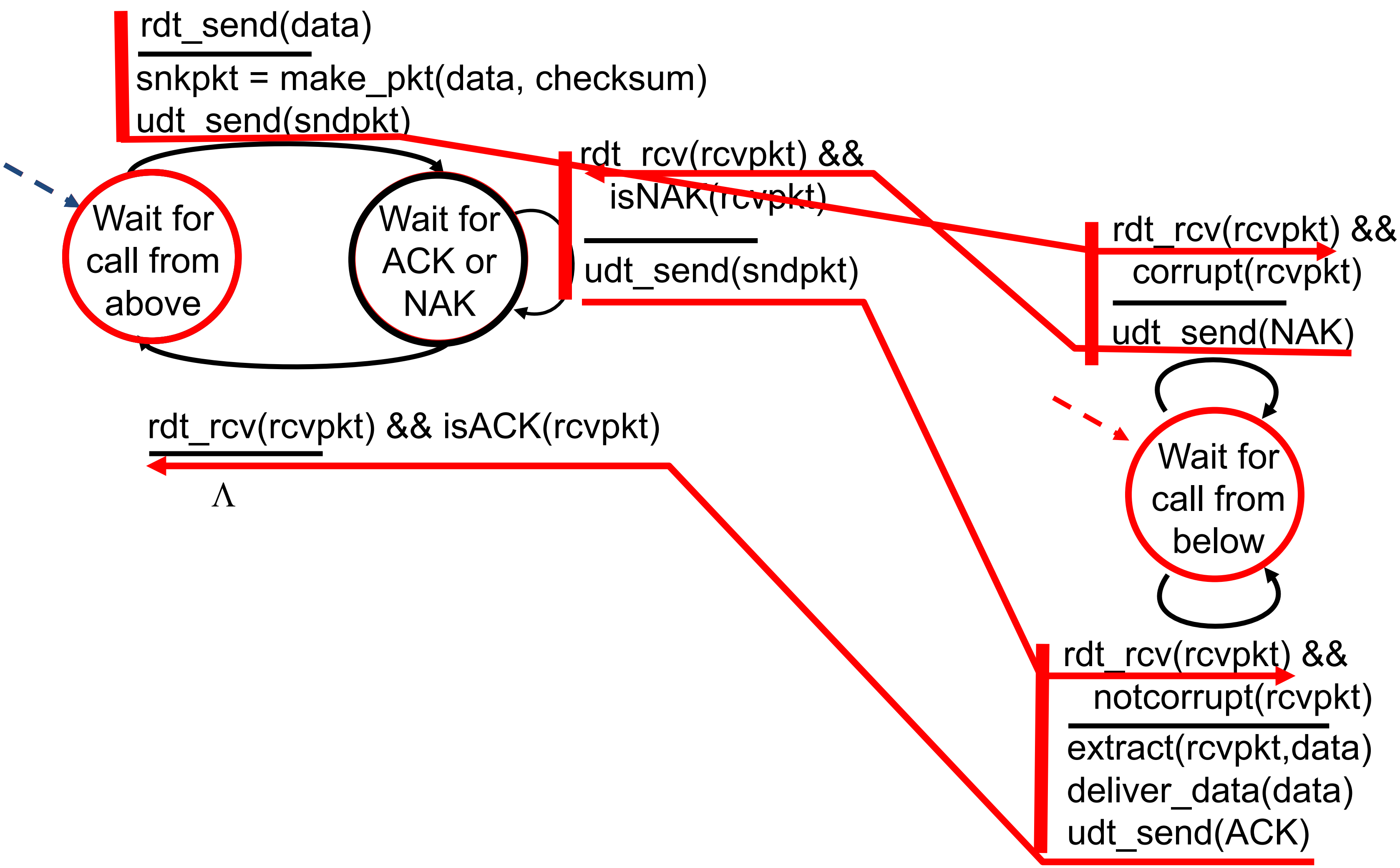
rdt2.0: FSM specification



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

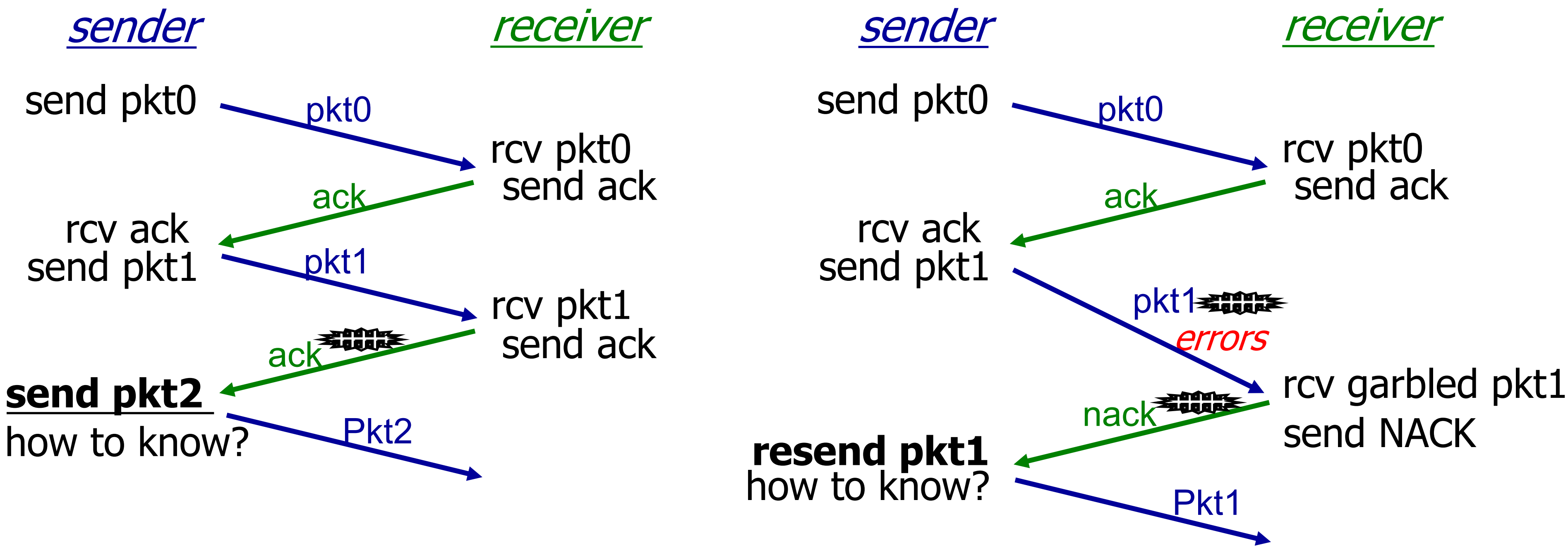
handling duplicates:

- 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

rdt2.0’s flaw: garbled ACK/NACK

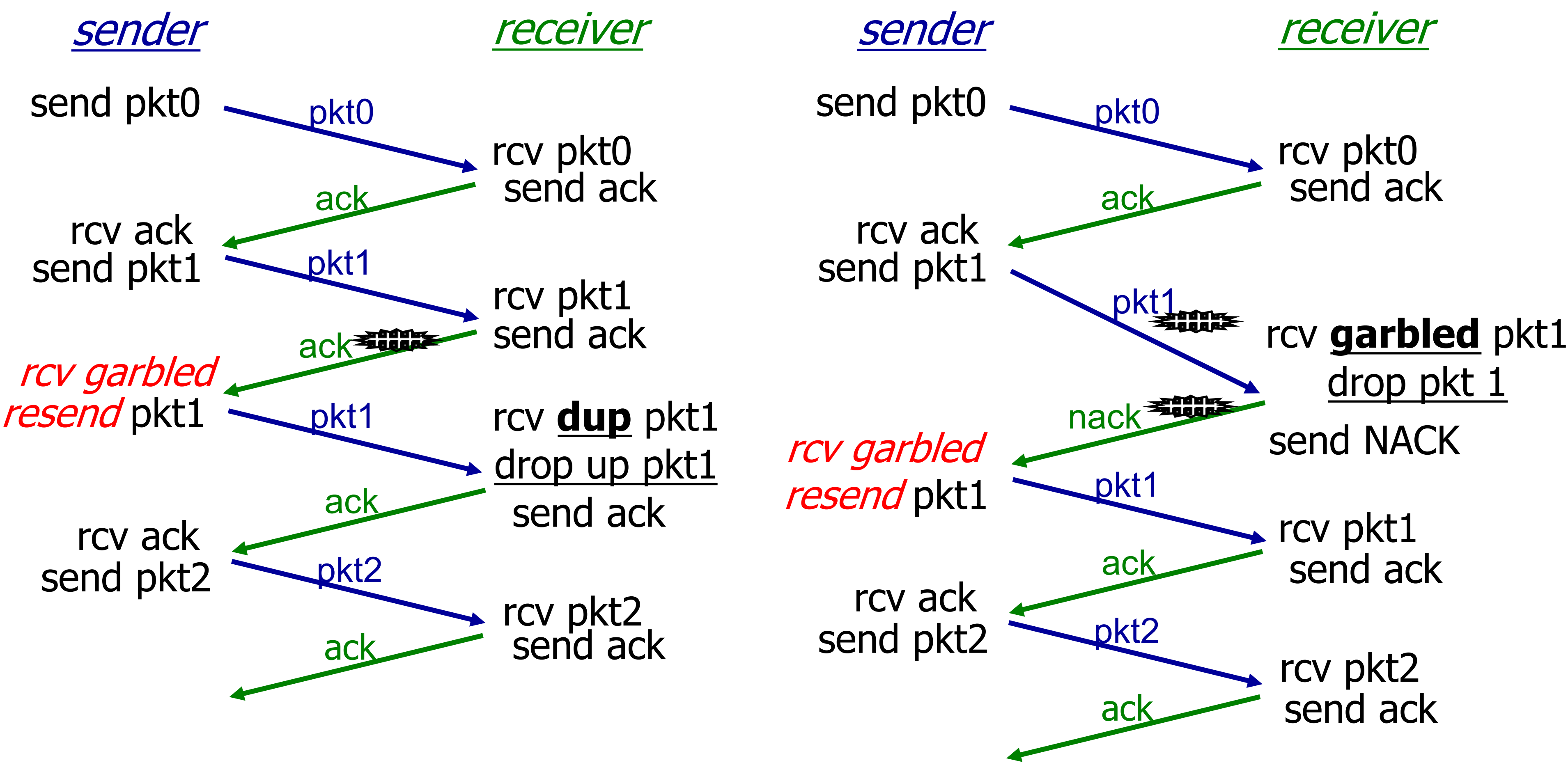


(a) Corrupted ack

(b) Corrupted NACK

Sender cannot tell whether it is corrupted ACK or NACK!

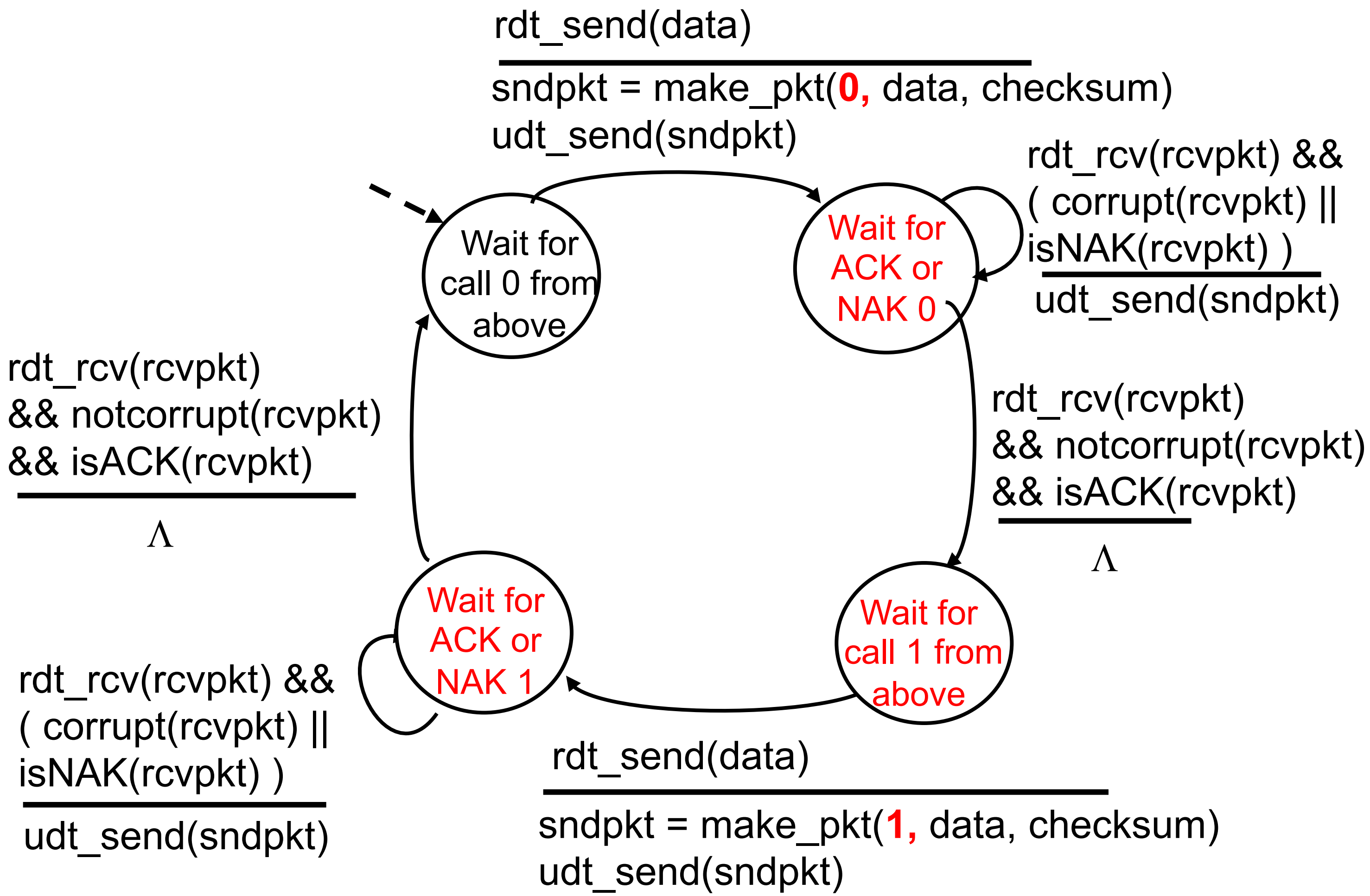
rdt2.1: need seq #!



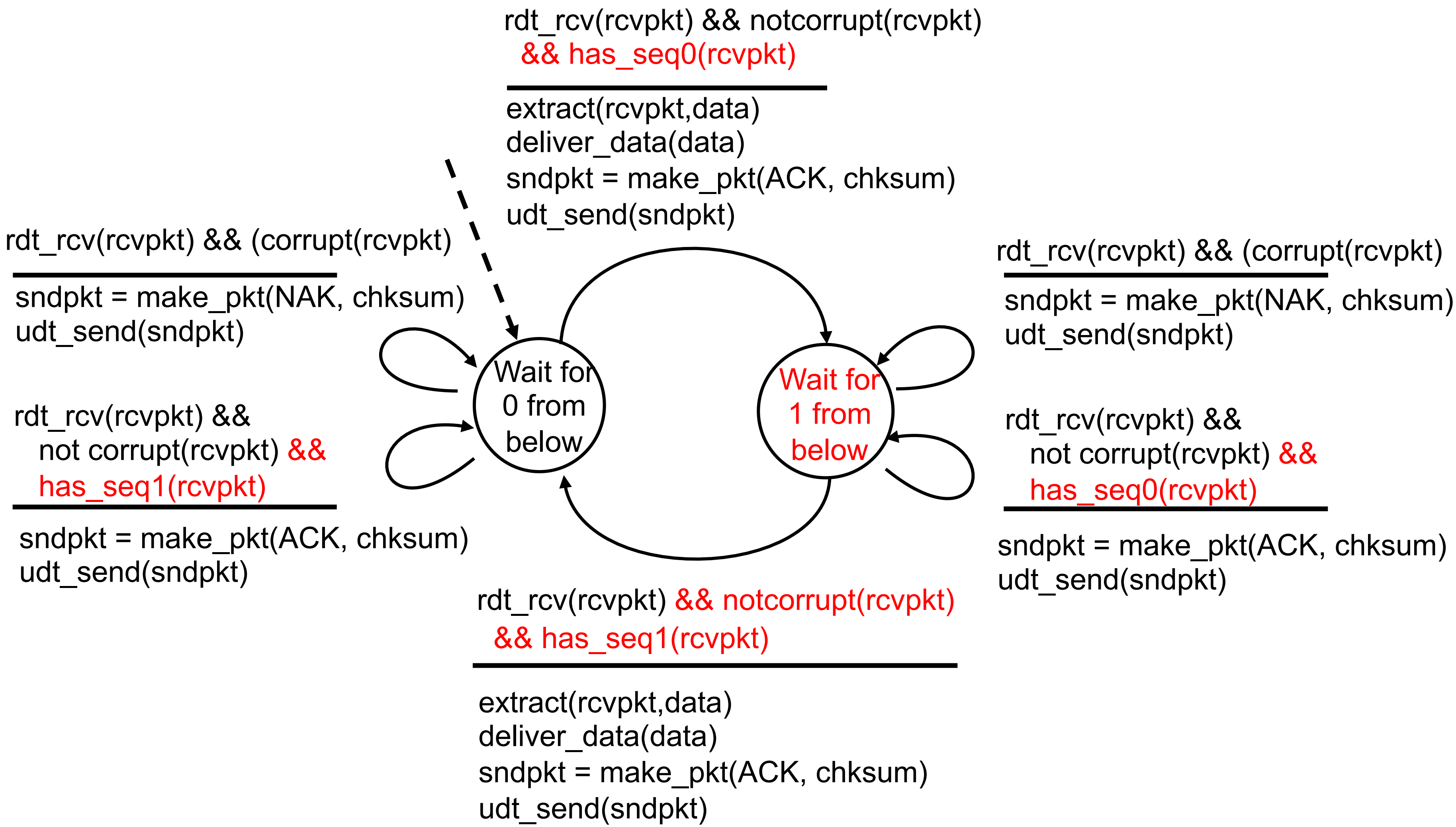
(a) Corrupted ack

(b) Corrupted NACK

rdt2.1: sender, handles garbled ACK/NAKs



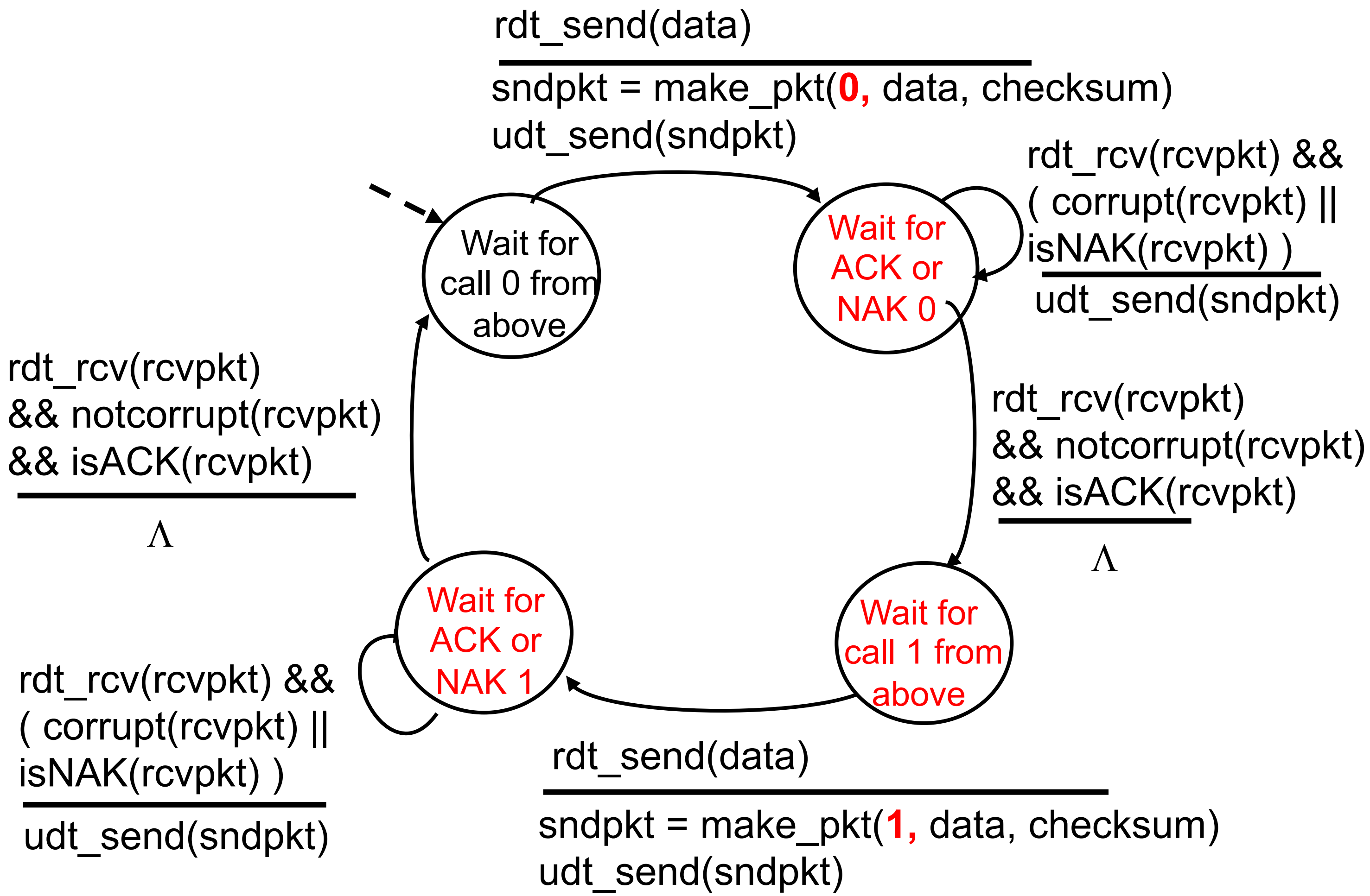
rdt2.1: receiver, handles garbled ACK/NAKs



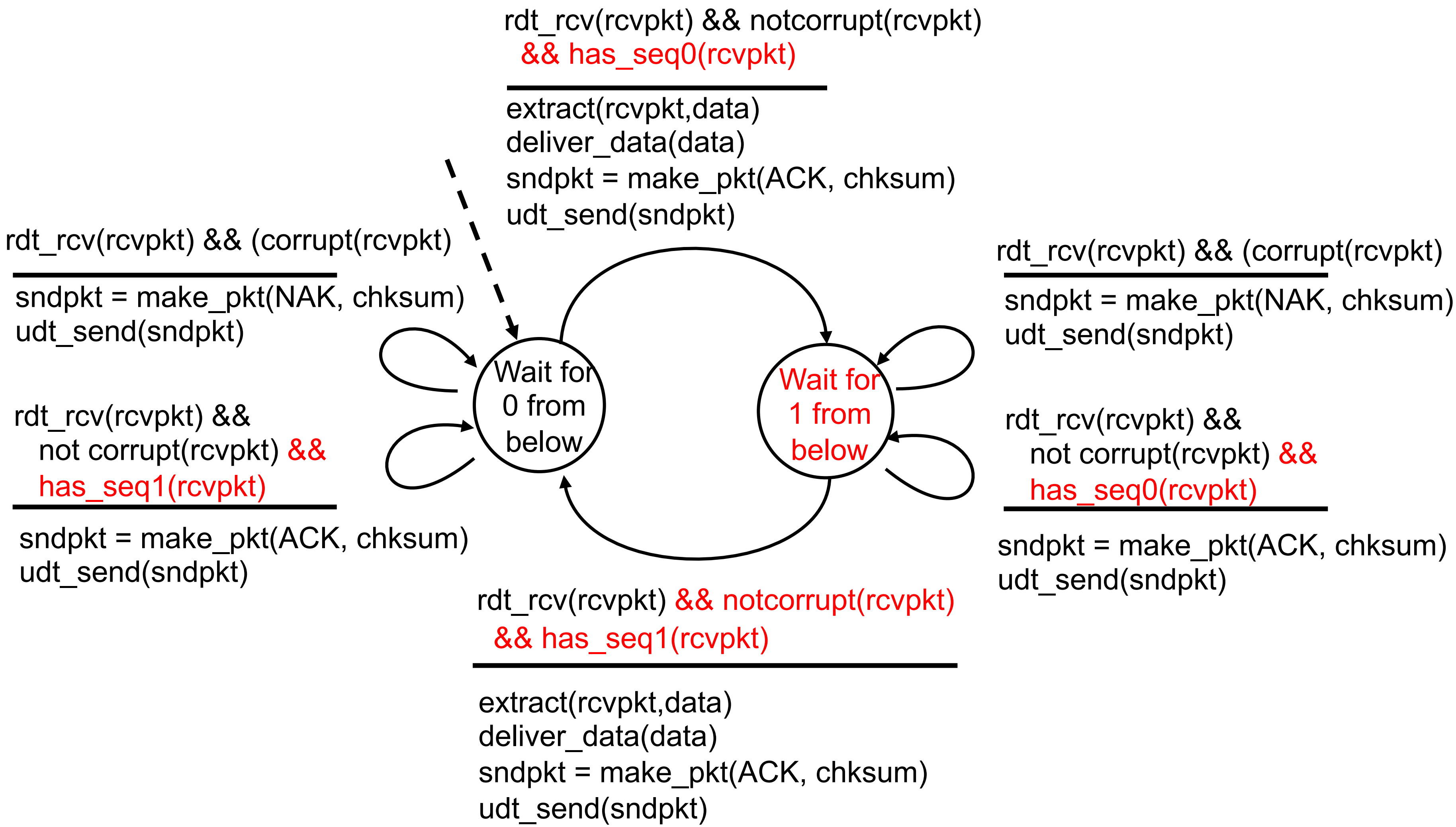
Summary: reliable data transfer

Version	Channel	Mechanism
rdt1.0	Reliable channel	nothing
rdt2.0	bit errors (no loss)	(1)error detection via checksum (2)receiver feedback (ACK/NAK) (3)retransmission
rdt2.1	Same as 2.0	handling fatal flaw with rdt 2.0: (4) need seq #.

rdt2.1: sender, handles garbled ACK/NAKs



rdt2.1: receiver, handles garbled ACK/NAKs



Rdt2.1 discussion

- Rdt2.1 mechanisms
  - Error detection (checksum)
  - Feedback (ACK and NAK)
  - Retransmission
  - Seq number (fresh or duplicate packets)
- Q1: How many bits are needed for seq#?
  - two seq. #'s (0,1) will suffice. Why?
  - Under various scenarios to send 3 packets: (1) all ACK, no error, (2) ACK 0 (1st time) corrupted, (3) NAK 0 (1st time) corrupted, (4) ACK 0 and ACK 1, both corrupted for the first time
- Q2: Do we still need NAK? If not, how?

Rdt2.1 discussion: how many bits for a seq number?

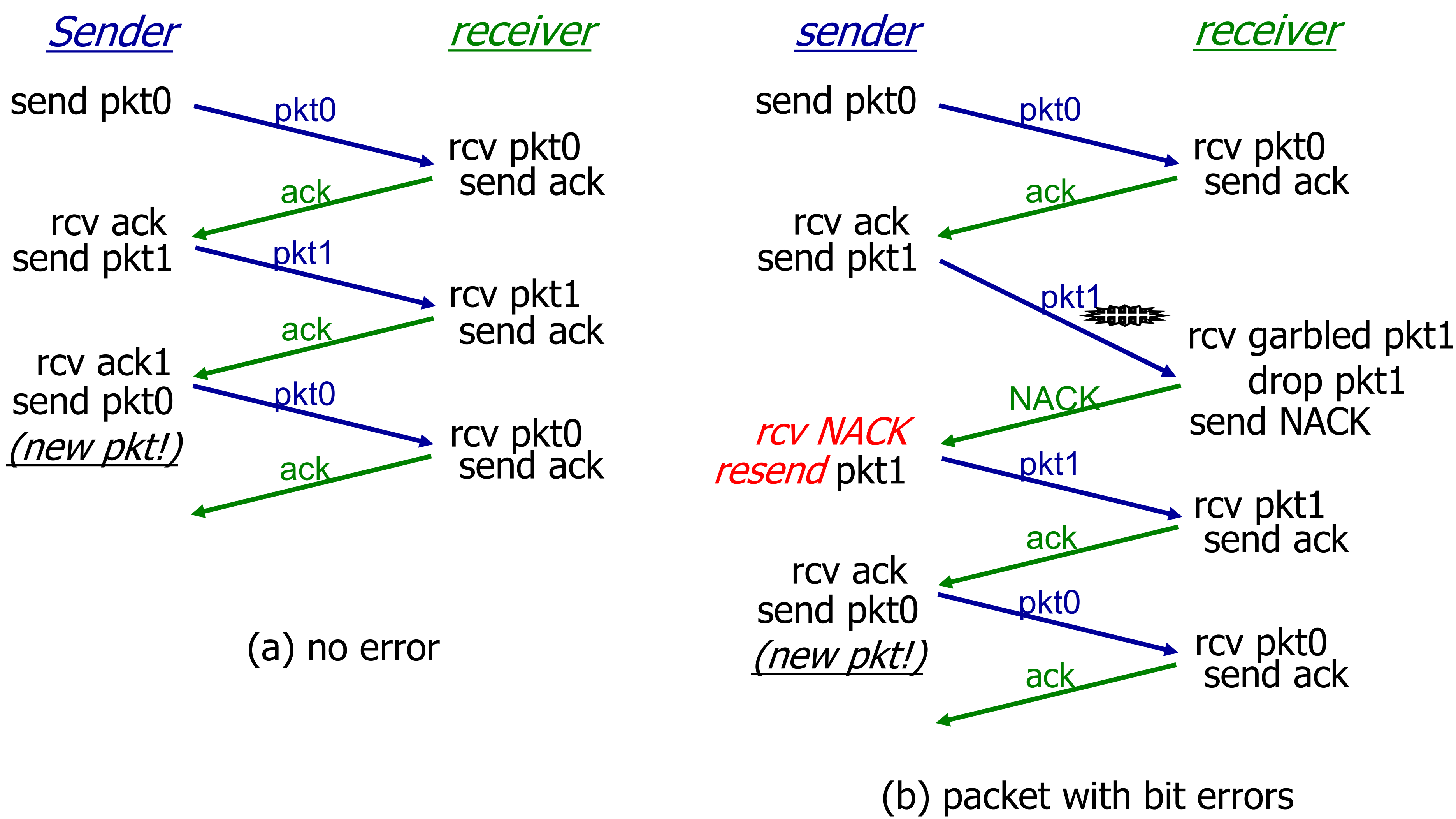
sender:

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

receiver:

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

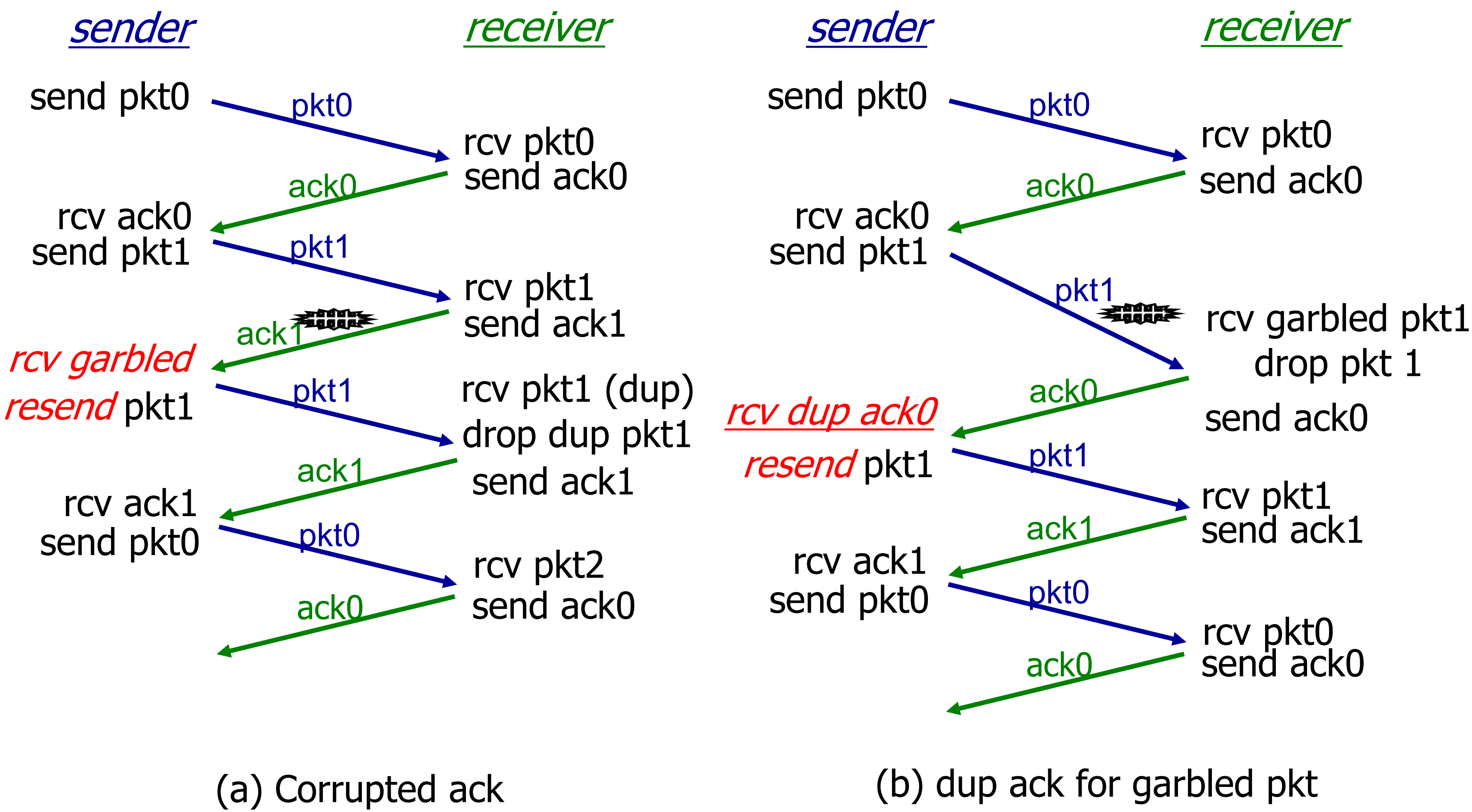
rdt2.1: 1-bit seq # is enough!



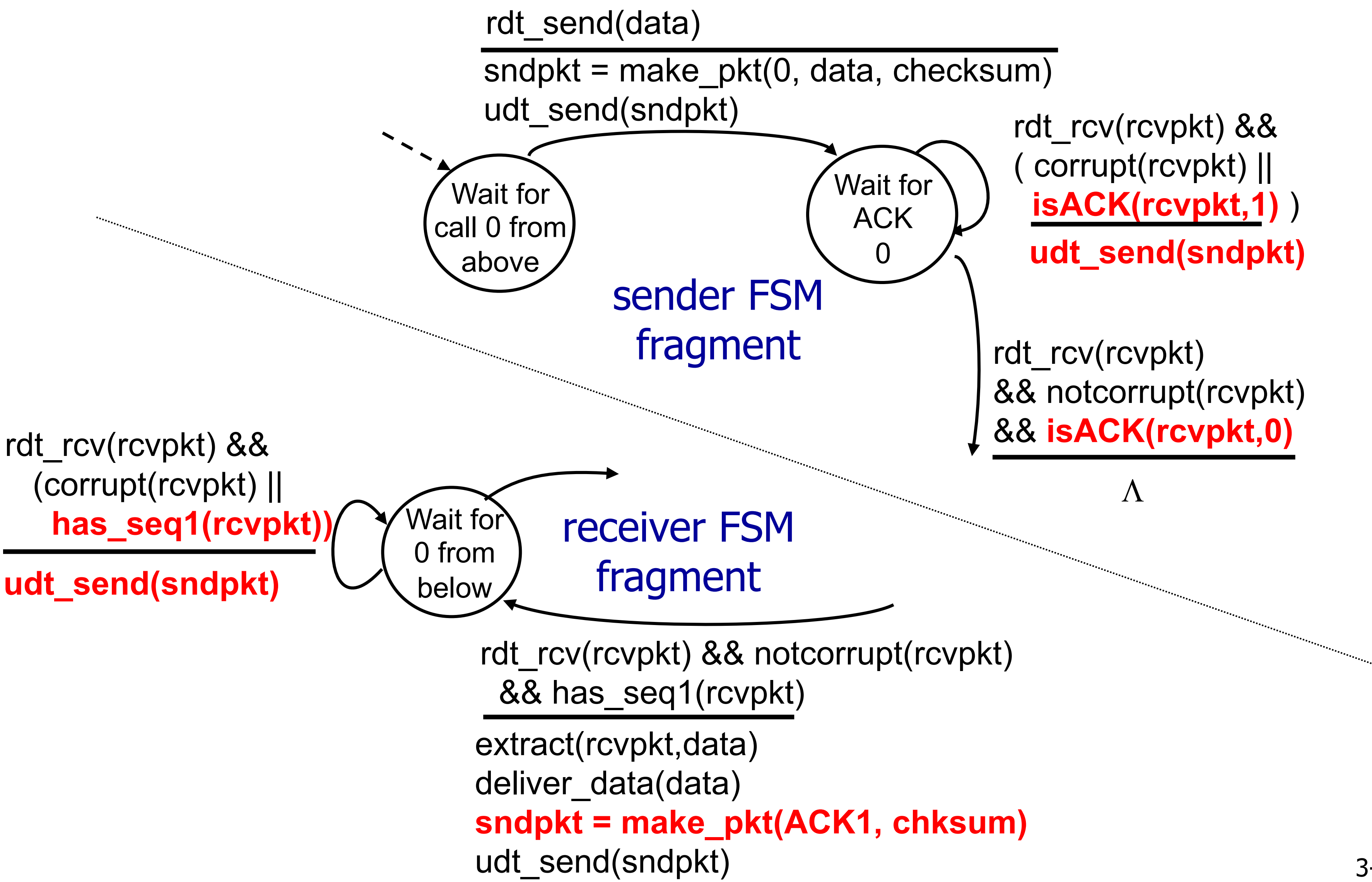
rdt2.2: a NAK-free protocol

- 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: NAK-free



rdt2.2: sender, receiver fragments



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rdt2.1	Same as 2.0 (fatal flaw)	(4)Seq# (1 bit, 0/1)
rdt2.2	Same as 2.0	A variant to rdt2.1 (no NAK) Duplicate ACK = NAK

rdt3.0: channels with errors *and* loss

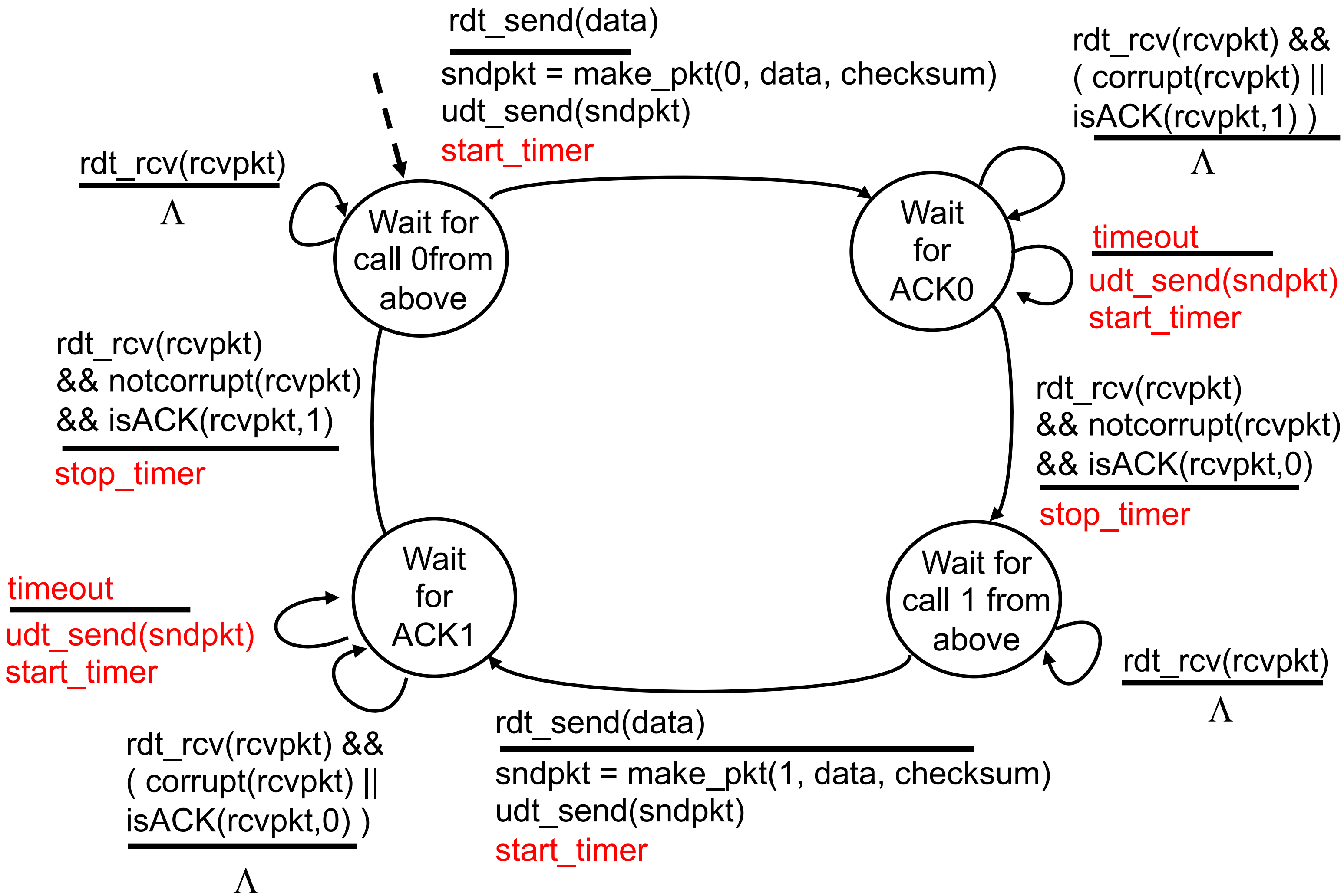
new assumption: underlying channel can also lose packets (data, ACKs)

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

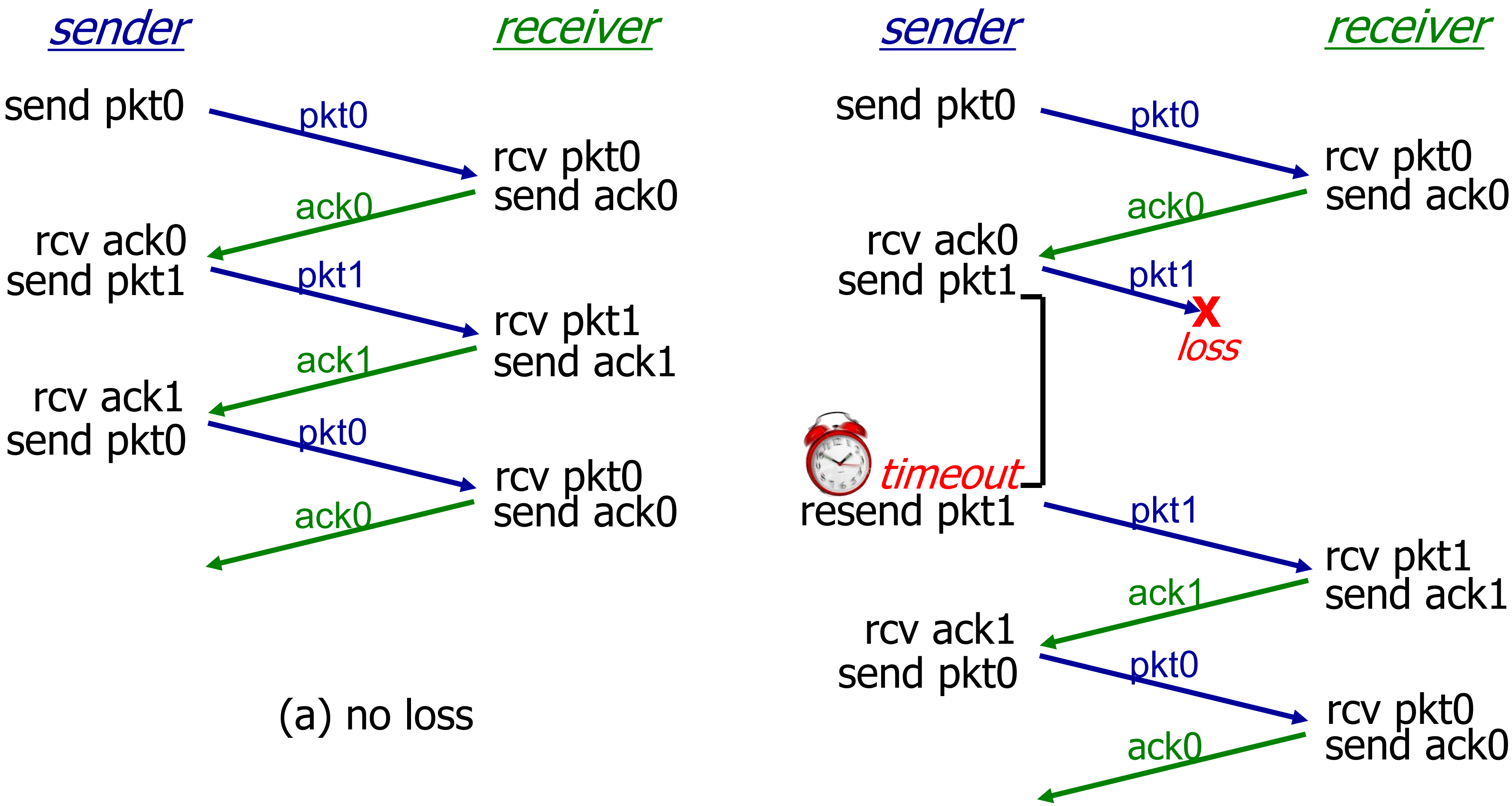
approach: sender waits "reasonable" amount of time for ACK (timer)

- 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



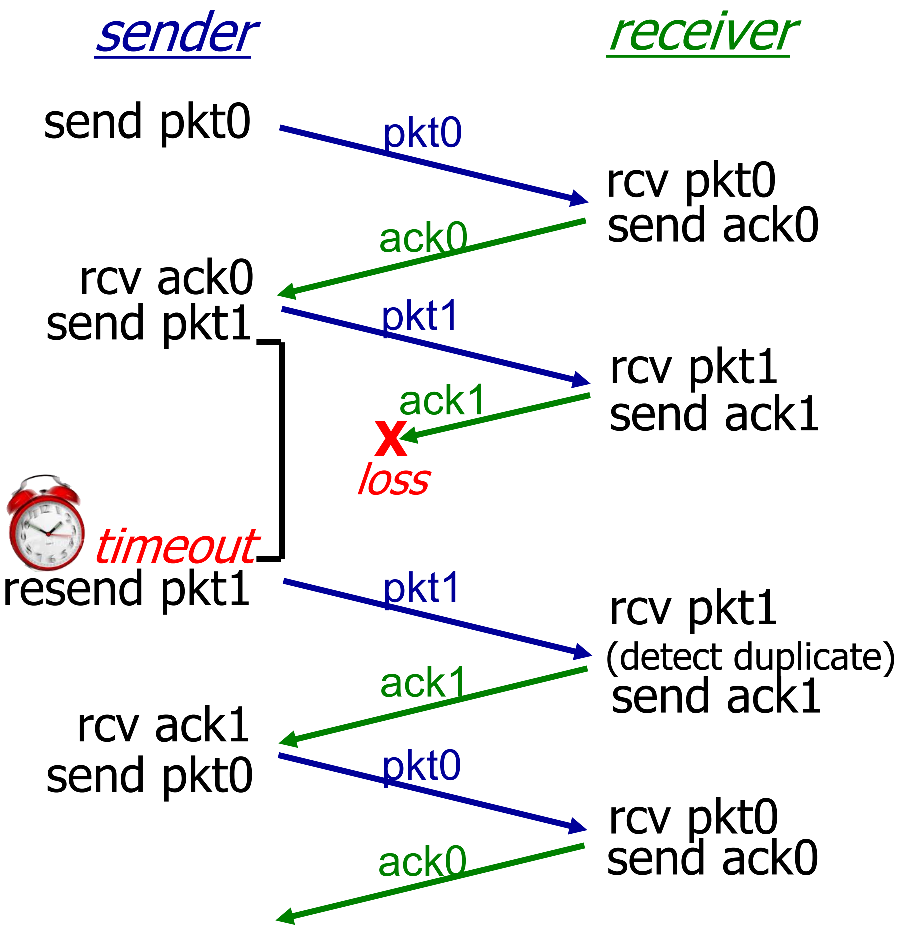
rdt3.0 in action



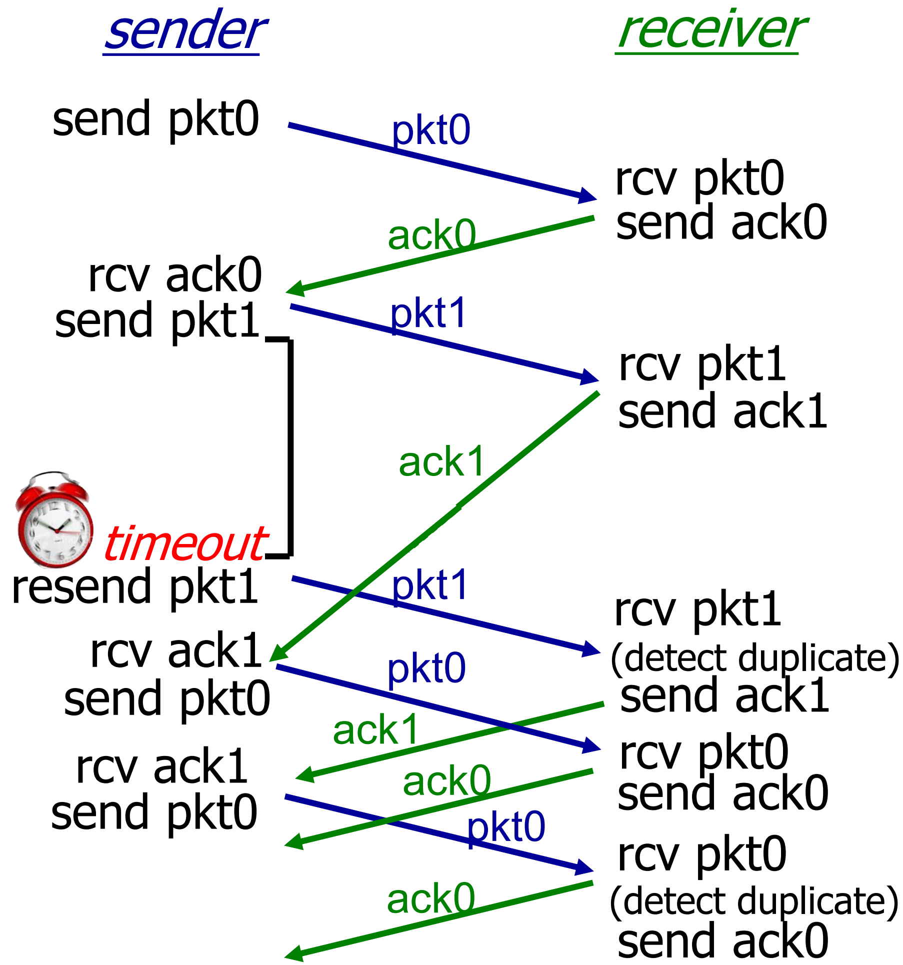
(a) no loss

(b) packet loss

rdt3.0 in action



(c) ACK loss



(d) premature timeout/ delayed ACK

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rdt2.1	Same as 2.0	(4)Seq# (1 bit)
rdt2.2	Same as 2.0	A variant to rdt2.1 (no NAK) Unexpected ACK = NAK ACK0 = ACK for pkt0, NAK for pkt1
Rdt3.0	Bit errors + loss	(5) Retransmission timer No NAK, only ACK

Performance of rdt3.0

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

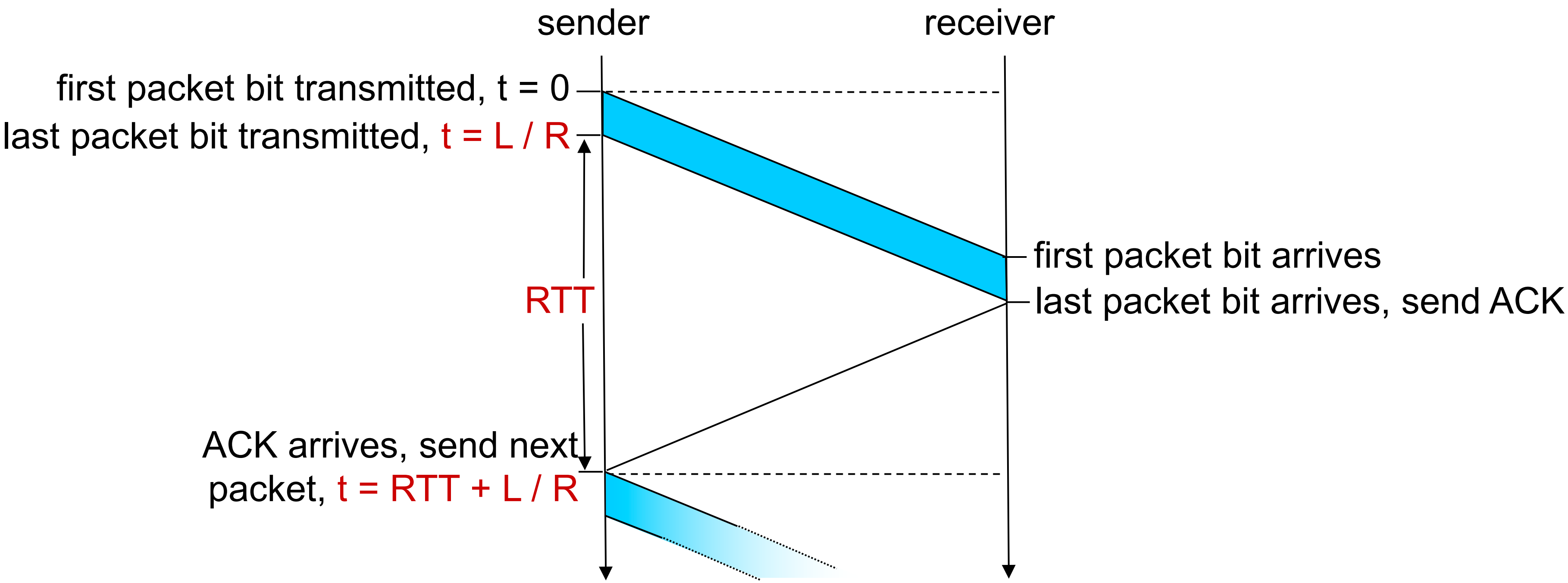
$$D_{trans} = \frac{L}{R} = \frac{8000 \text{ bits}}{10^9 \text{ bits/sec}} = 8 \text{ microsecs}$$

  - 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

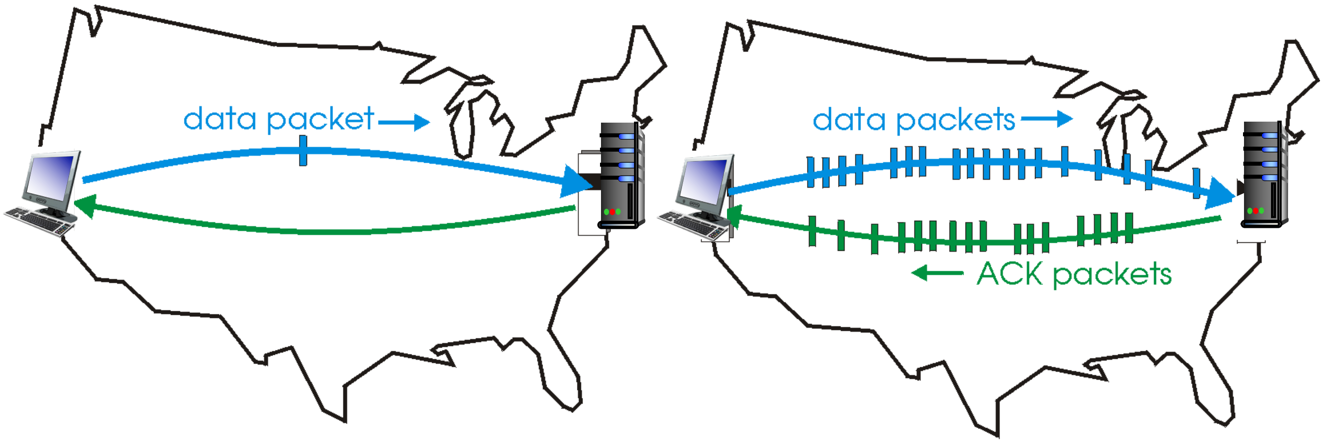


$$U_{sender} = \frac{L / R}{RTT + L / R} = \frac{.008}{30.008} = 0.00027$$

Pipelined protocols

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

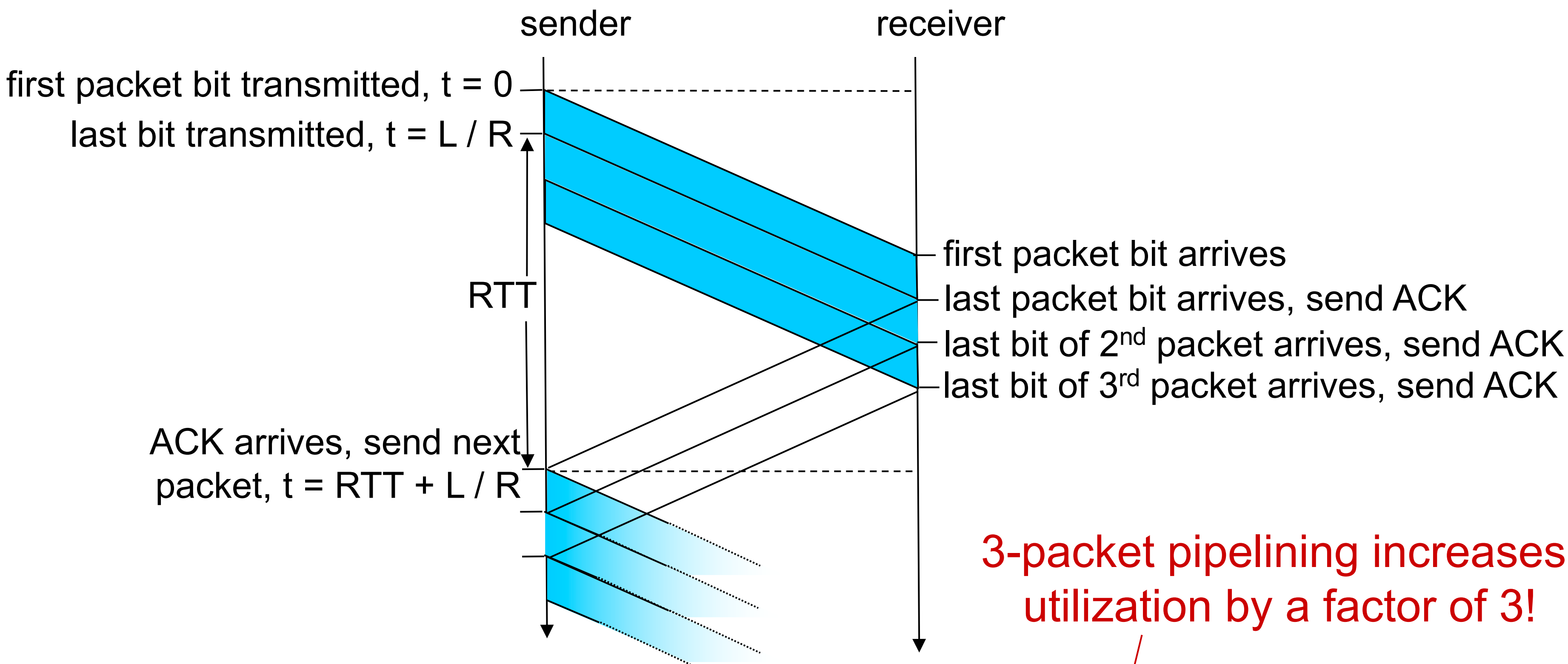
- range of sequence numbers must be increased
- buffering at sender and/or receiver



(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

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

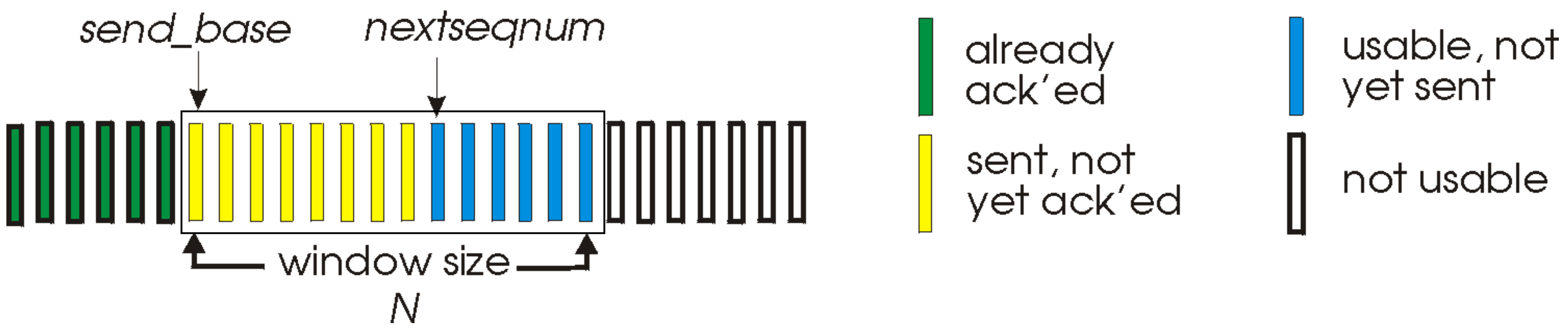
- sender can have up to N unacked packets in pipeline
- receiver only sends *cumulative ack*
  - doesn't ack packet if there's a gap
- sender has timer for oldest unacked packet
  - when timer expires, retransmit *all* unacked packets

<u>Selective Repeat:</u>

- 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

- sender can have up to N unacked packets in pipeline
- if data from the above AND nextseqnum < base+N,
  - Send(packet)
  - Nextseqnum++
  - Start timer (for one oldest unacked packet)
- If Timeout, retransmit *all* unacked packets
  - [base, nextseqnum-1]
- If receiving ACK, update base
  - base = getacknum(rcvpkt)+1
  - If base==nextseqnum, stop timer
- If receiving corrupted ACK, do nothing

GBN: sender extended FSM

rdt_send(data)

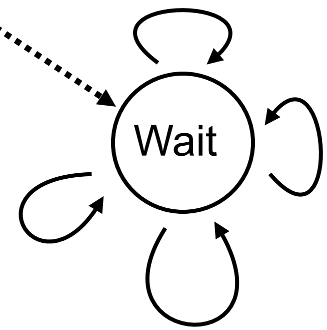
---

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)

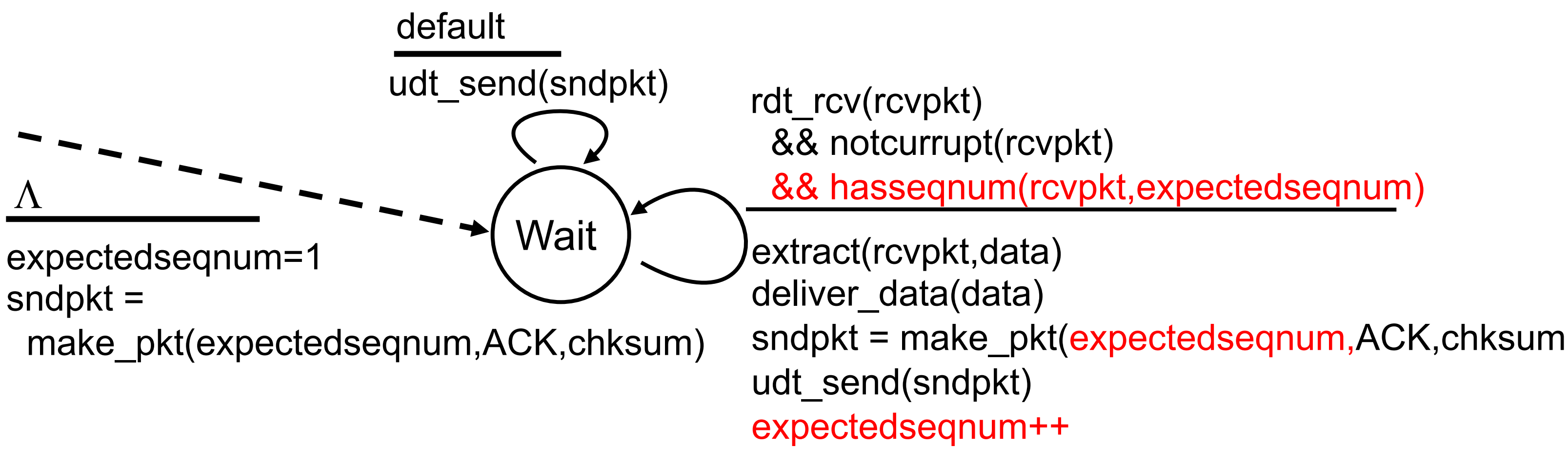
---

rdt_rcv(rcvpkt) &&
  notcorrupt(rcvpkt)

---

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

GBN: receiver extended FSM



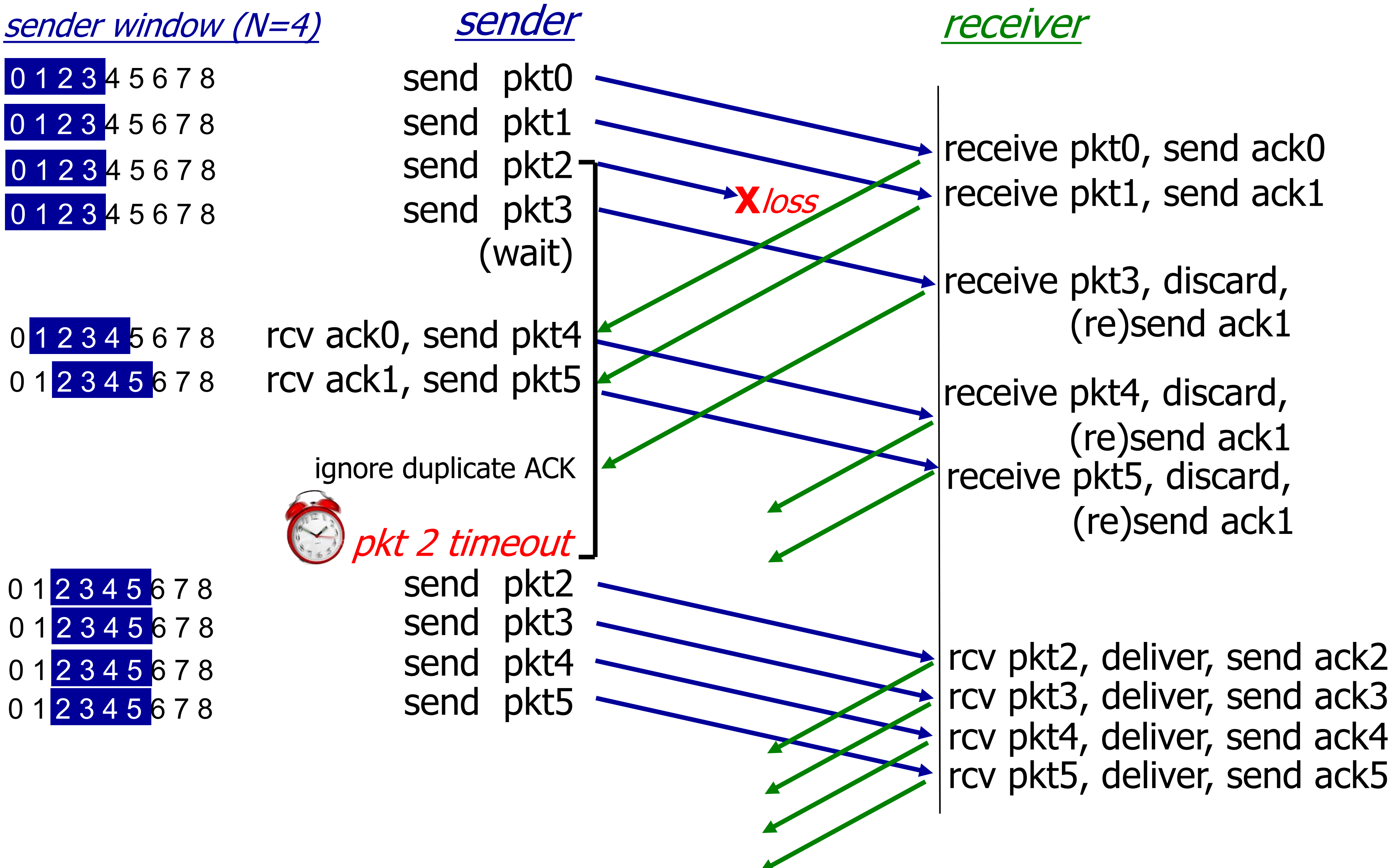
ACK-only: always send ACK for correctly-received pkt with highest *in-order* seq #

- 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 #

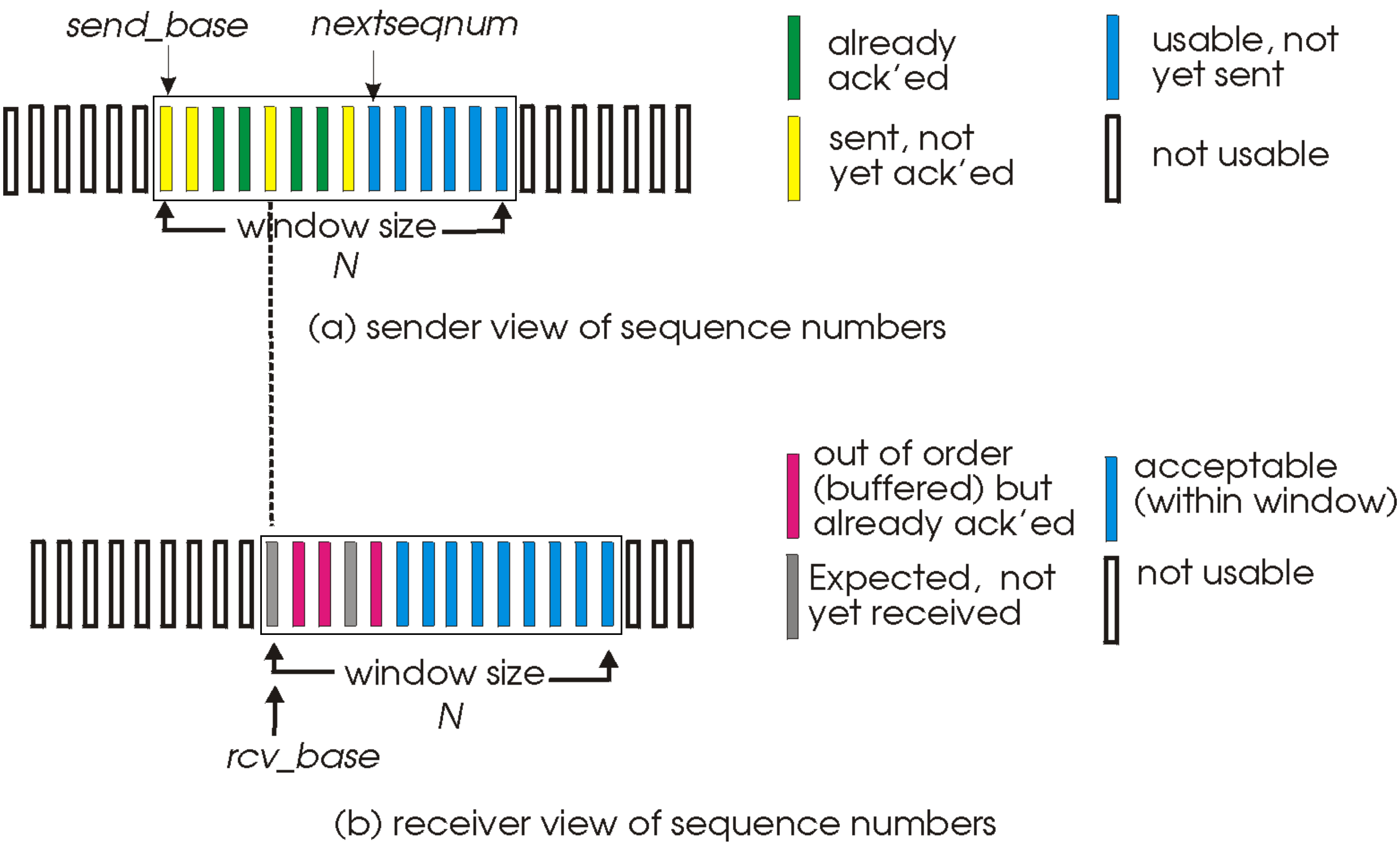
GBN in action



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
  - 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

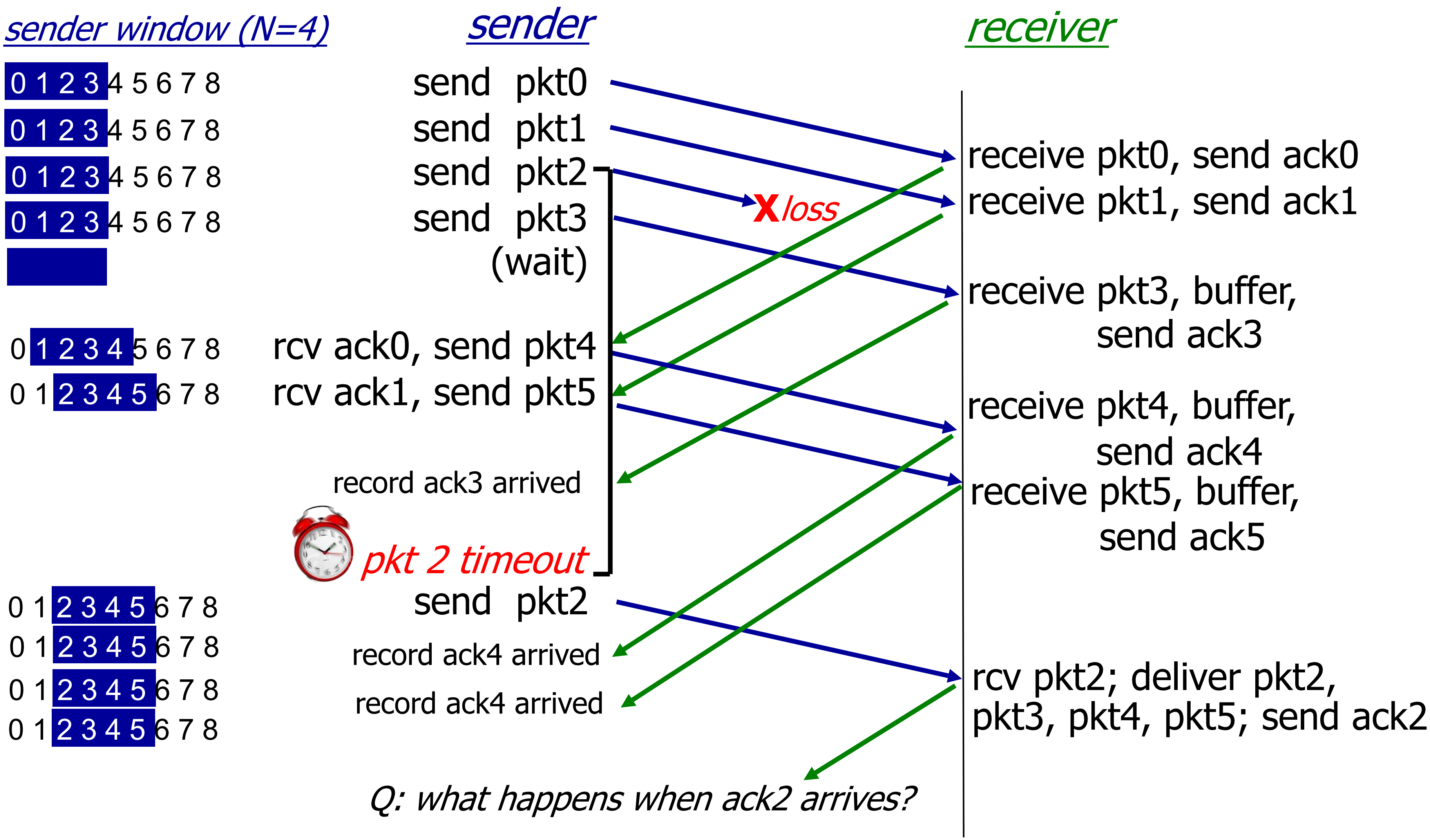
pkt n in [rcvbase-N,rcvbase-1]

- ACK(n)

otherwise:

- ignore

Selective repeat in action



After-class Practice: GBN vs SR

- How many unique seq# may appear in GBN and SR, respectively?
  - N = 2
  - GBN: sender [4,5], what is the expected number at the receiver? **[4, 5, 6]**
    - No error
    - ACK 4 is lost
    - ACK 4 and ACK 5 are lost

GBN: give the expected number x, the sender window will be [x-2, x-1], [x-1, x], [x, x+1]

  - Given the expected number 6, how to infer the sender window?
- How about SR (expected window)? **[4,5], [5,6], [6,7]**
- What if we have N+1 sequence numbers for SR?

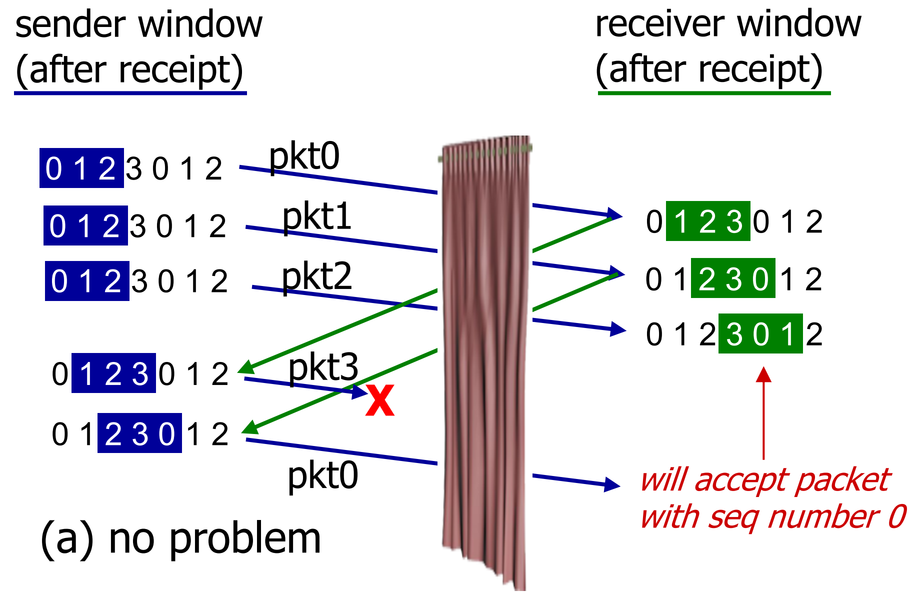
Selective repeat: dilemma (N+1)

example:

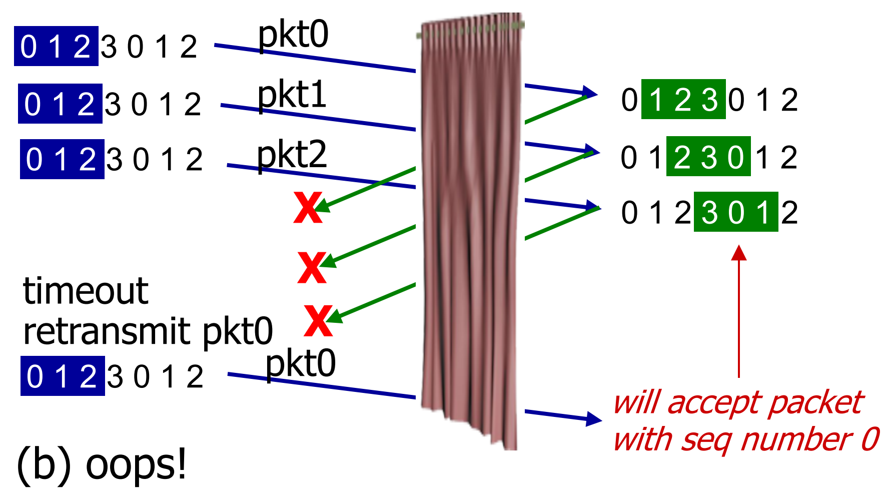
- window size=3
- seq #'s: 0, 1, 2, 3
- receiver sees no difference in two scenarios!
- duplicate data accepted as new in (b)

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

2N



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



Summary: reliable data transfer

Version	Channel	Mechanism
rdt1.0	No error/loss	nothing
rdt2.0	bit errors (no loss)	(1)error detection via checksum (2)receiver feedback (ACK/NAK) (3)retransmission
rdt2.1	Same as 2.0	(4)Seq# (1 bit)
rdt2.2	Same as 2.0	(no NAK): Unexpected ACK = NAK
Rdt3.0	errors + loss	(5) Timer; ACK-only
Performance issue: low utilization		
Go-back-N	Same as 3.0	N sliding window (pipeline) Discard out-of-order pkts (recovery)
Selective Repeat	Same as 3.0	N sliding window, selective recovery

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3.3 connectionless transport: UDP

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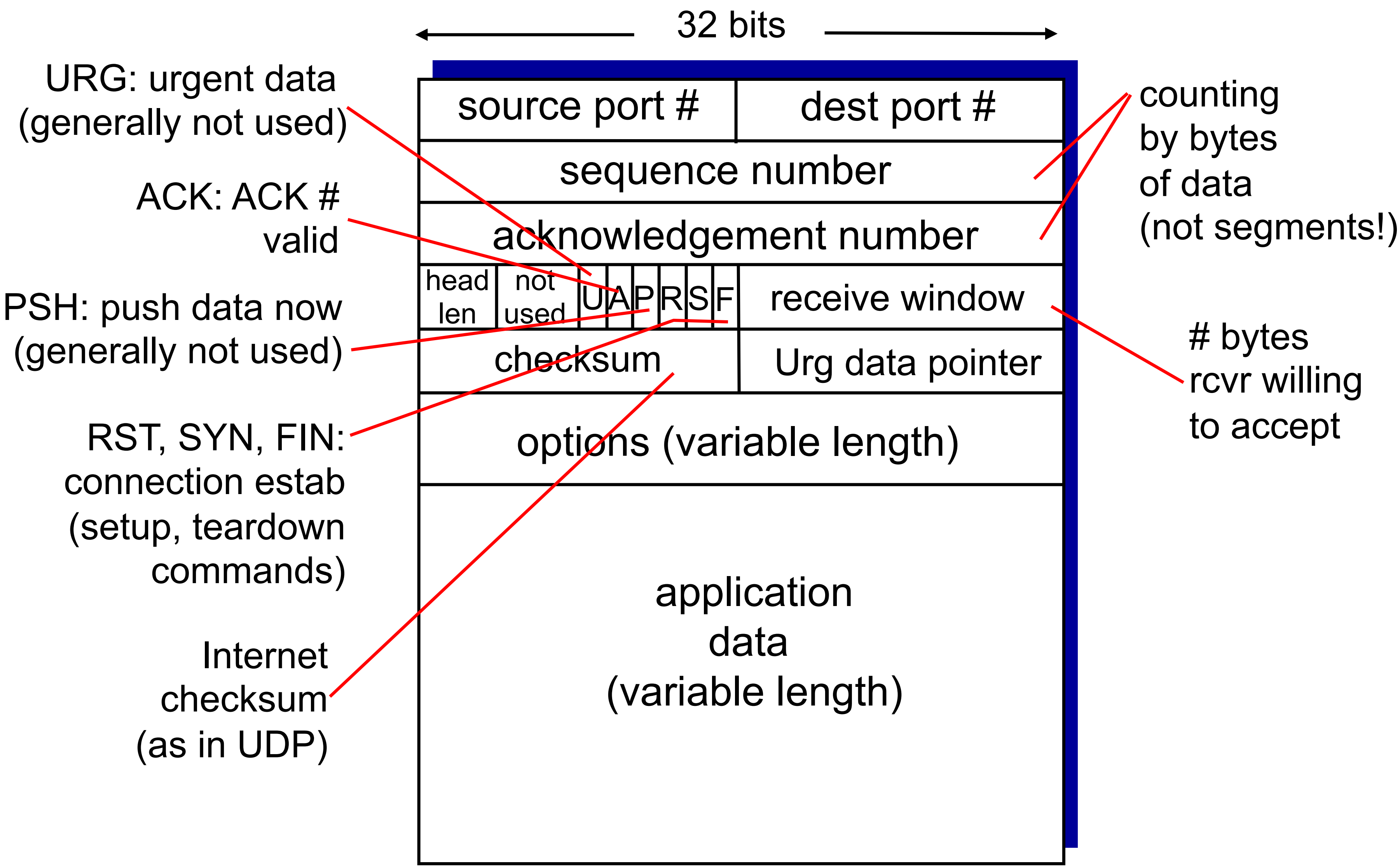
TCP: Overview RFCs: 793,1122,1323, 2018, 2581

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

TCP segment structure



TCP seq. numbers, ACKs

sequence number:

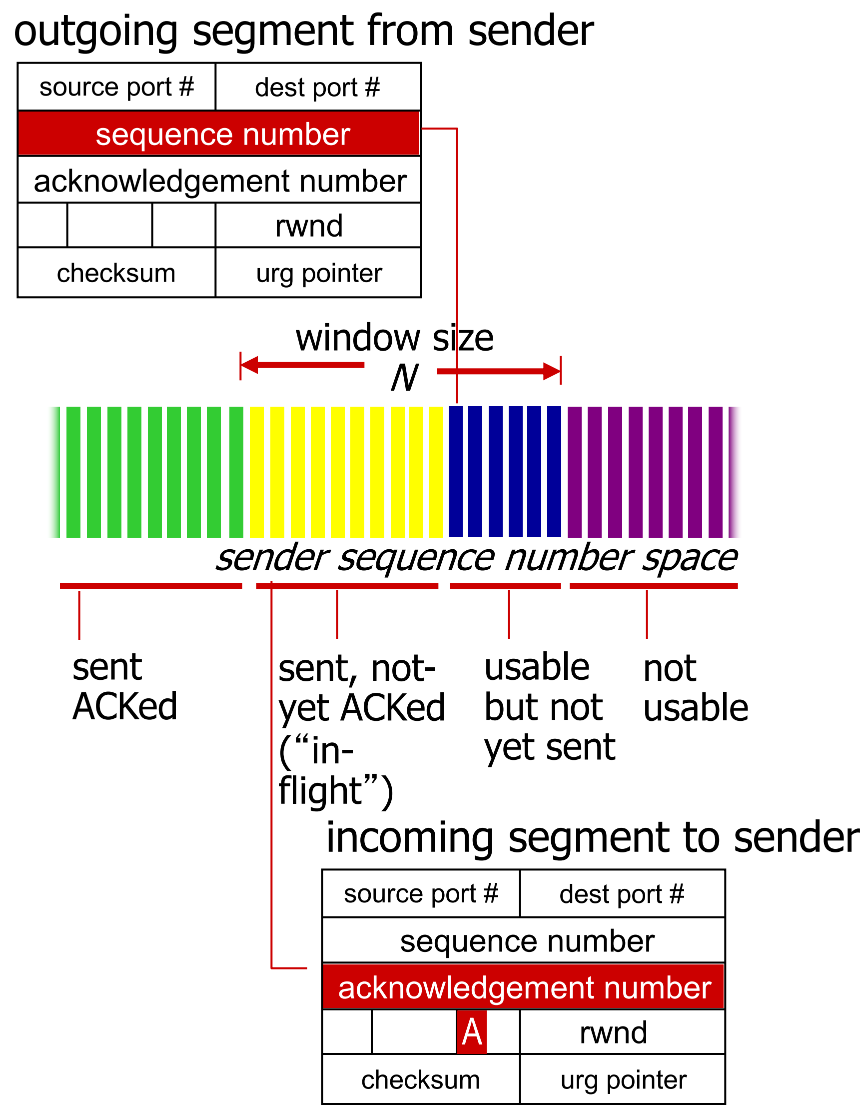
- byte stream "number" of first byte in segment's data

Acknowledgement #.:

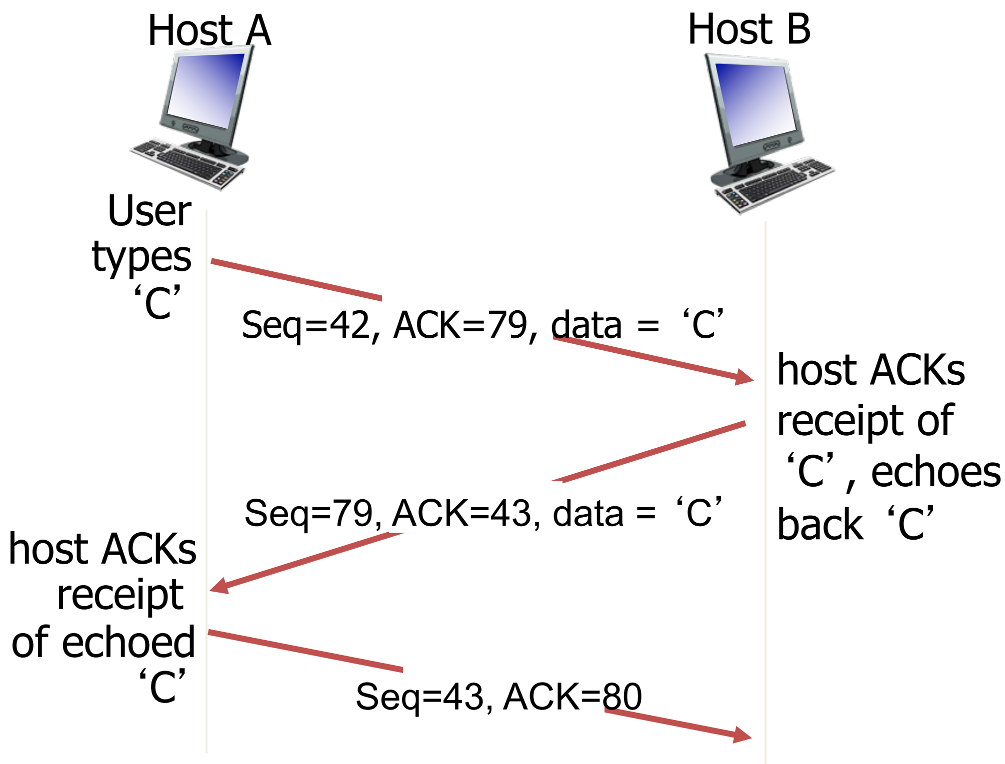
- seq # of next byte expected from other side
- cumulative ACK

Q: how receiver handles out-of-order segments

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



TCP seq. numbers, ACKs



simple telnet scenario

What if sending "ABC"?

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

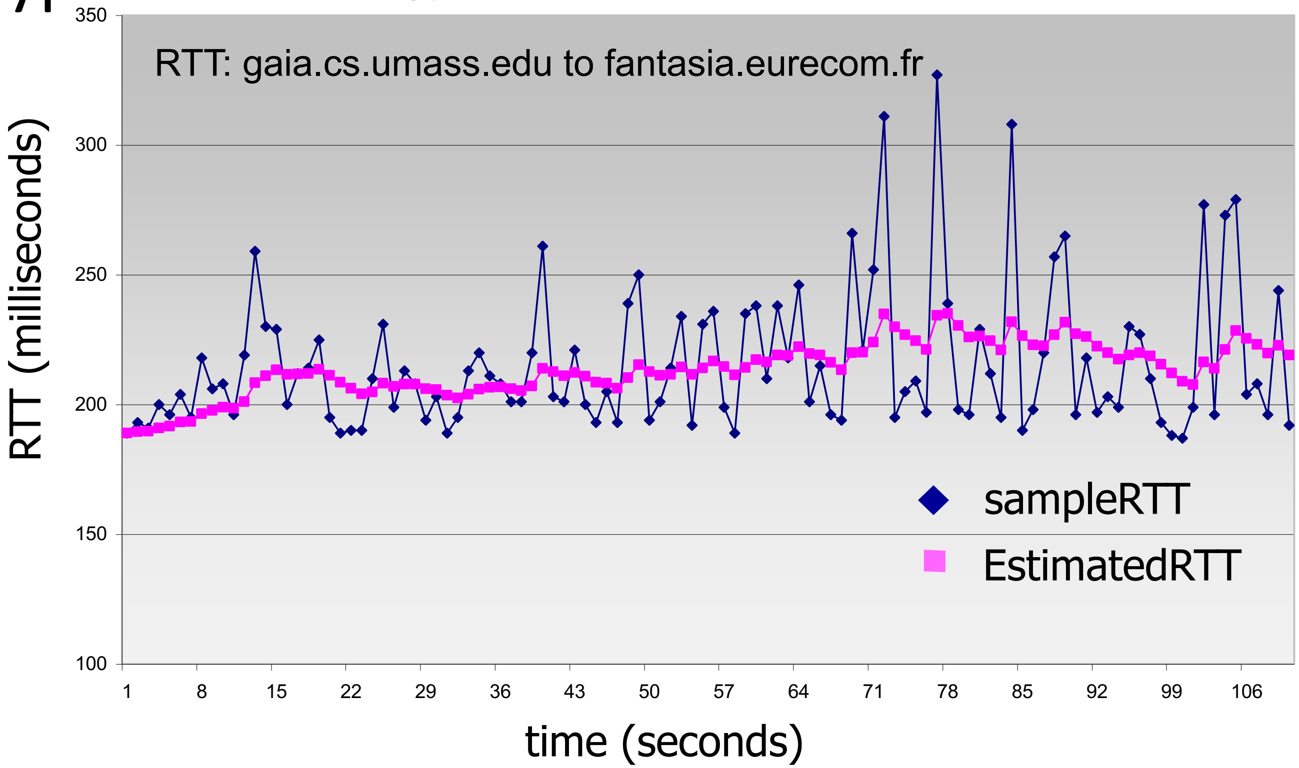
Q: how to estimate RTT?

- **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



TCP round trip time, timeout

- timeout interval: **`EstimatedRTT`** plus "safety margin"
  - large variation in **`EstimatedRTT`** -> larger safety margin
- estimate SampleRTT deviation from EstimatedRTT:

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

(typically, β = 0.25)

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

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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**

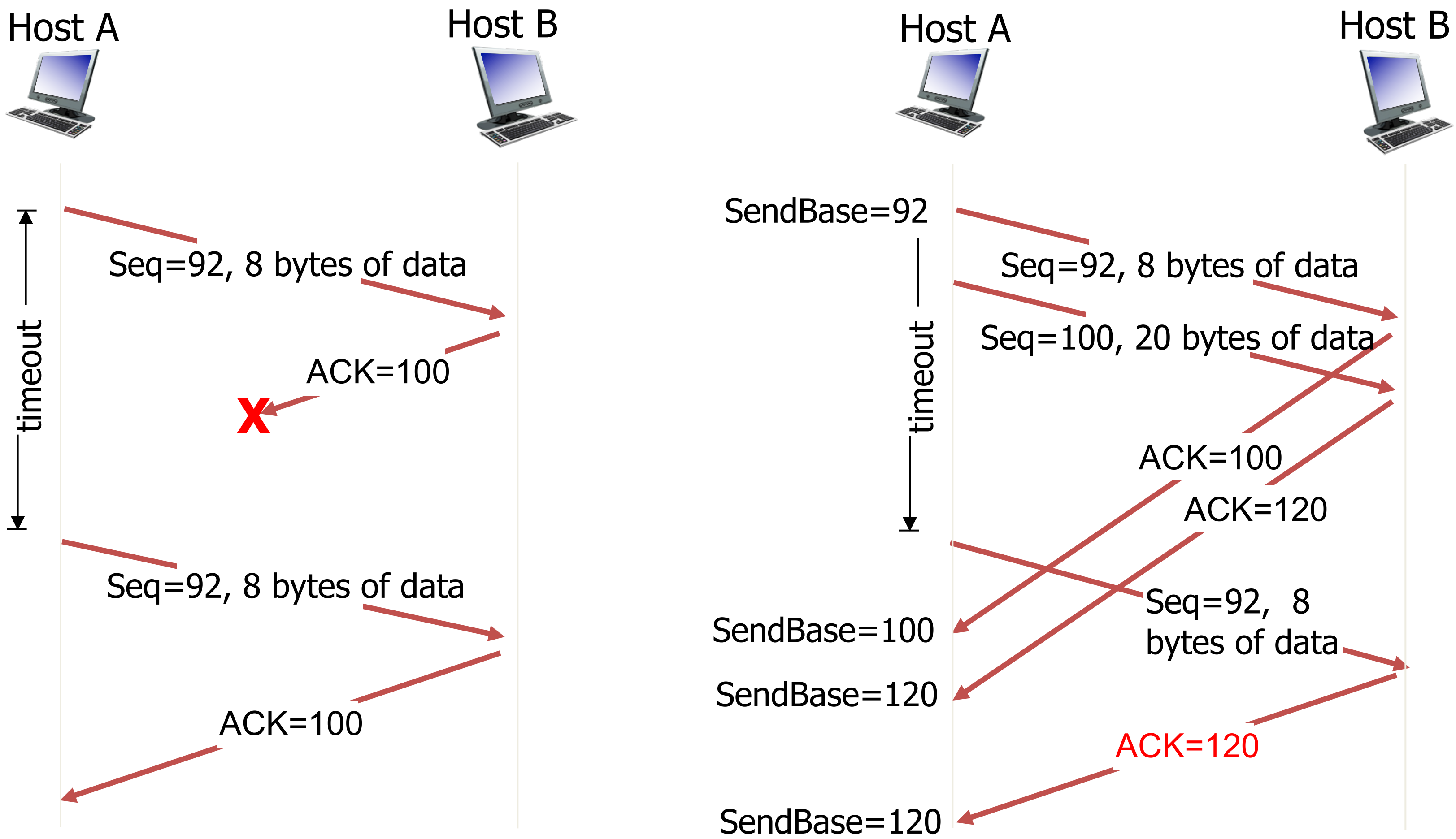
timeout:

- 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

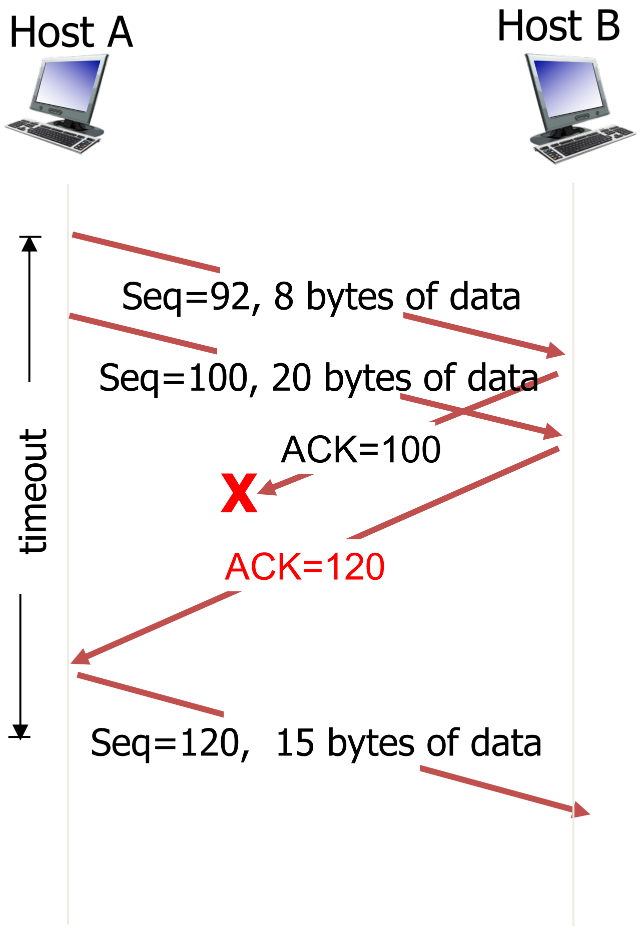
TCP: retransmission scenarios



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.

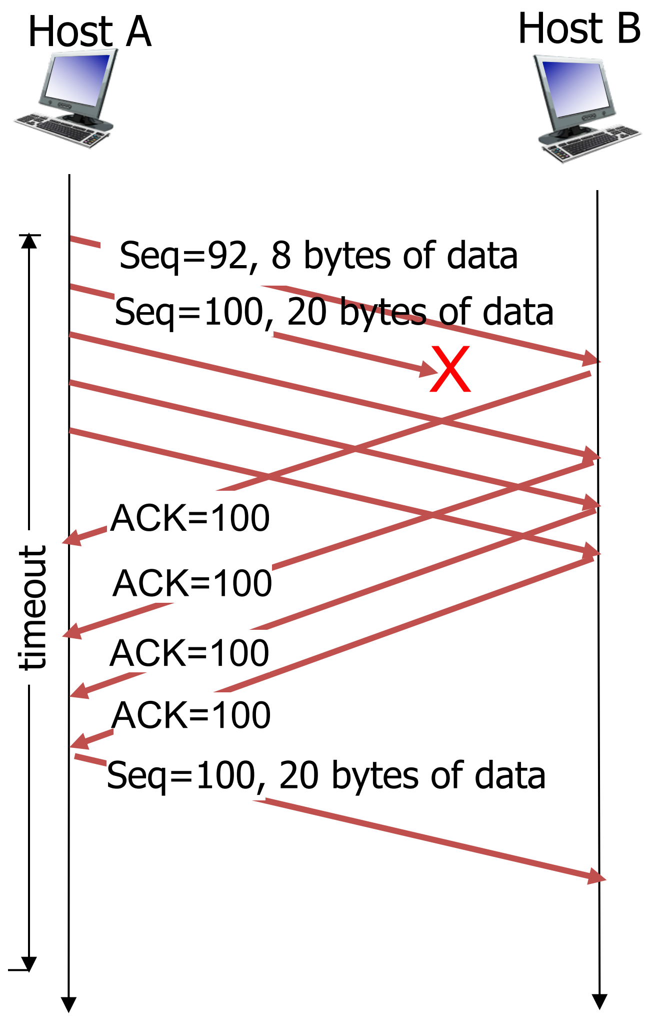
TCP fast retransmit

if sender receives 3 dup 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

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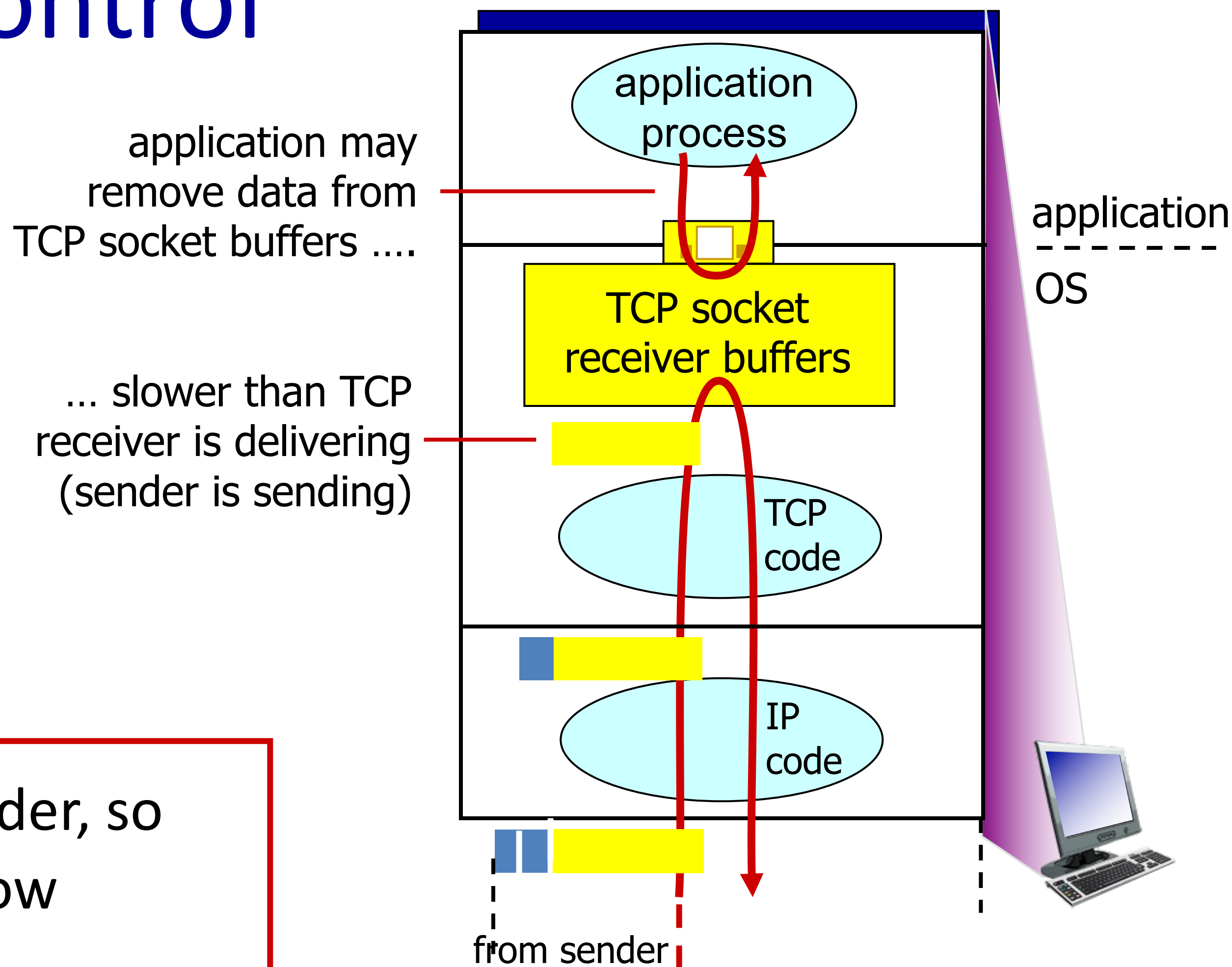
lectures (not textbook)

TCP flow control



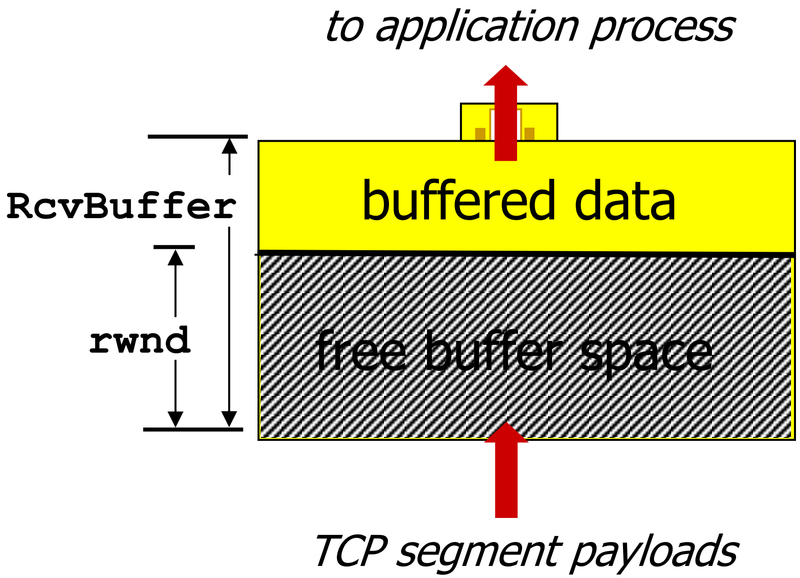
receiver protocol stack

flow control

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

TCP flow control

- receiver "advertises" free buffer space by including **rwnd** 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

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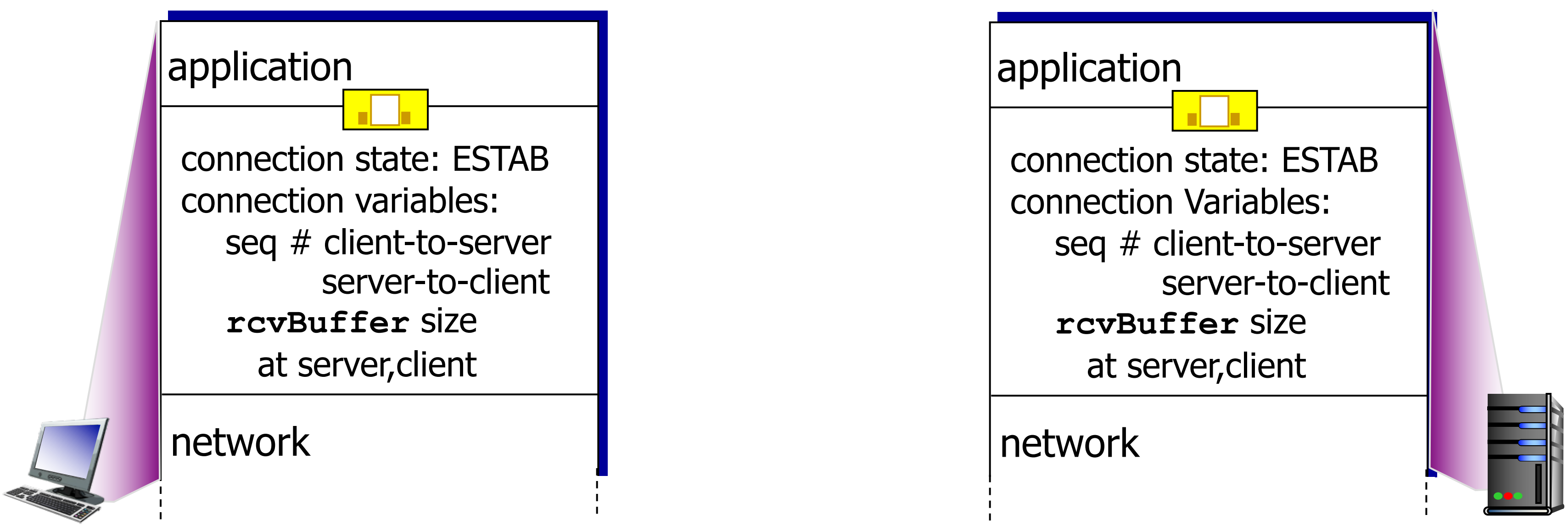
lectures (not textbook)

Connection Management

before exchanging data, sender/receiver "handshake":

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

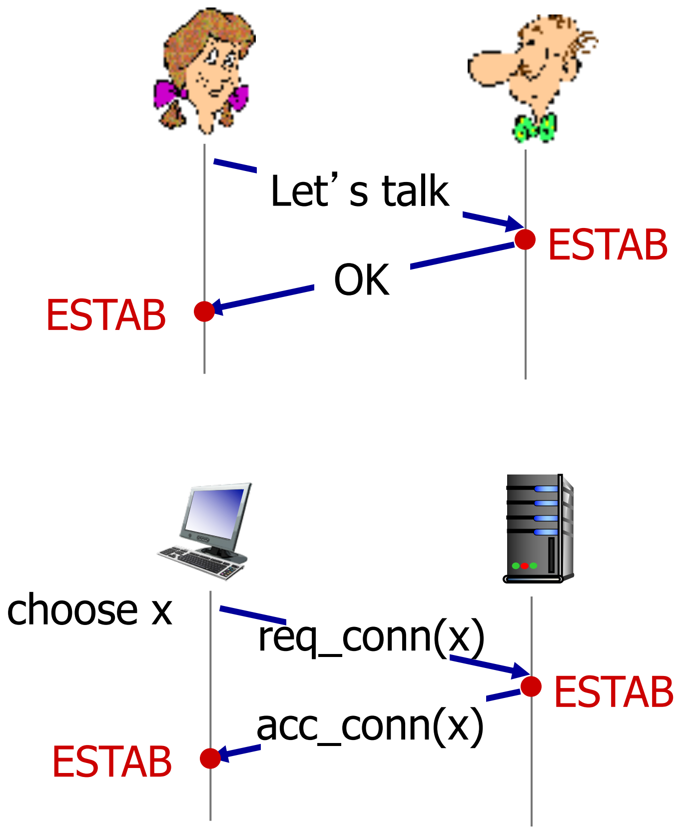



```
Socket clientSocket =
  newSocket("hostname","port
   number");
```

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

Agreeing to establish a connection

2-way handshake:

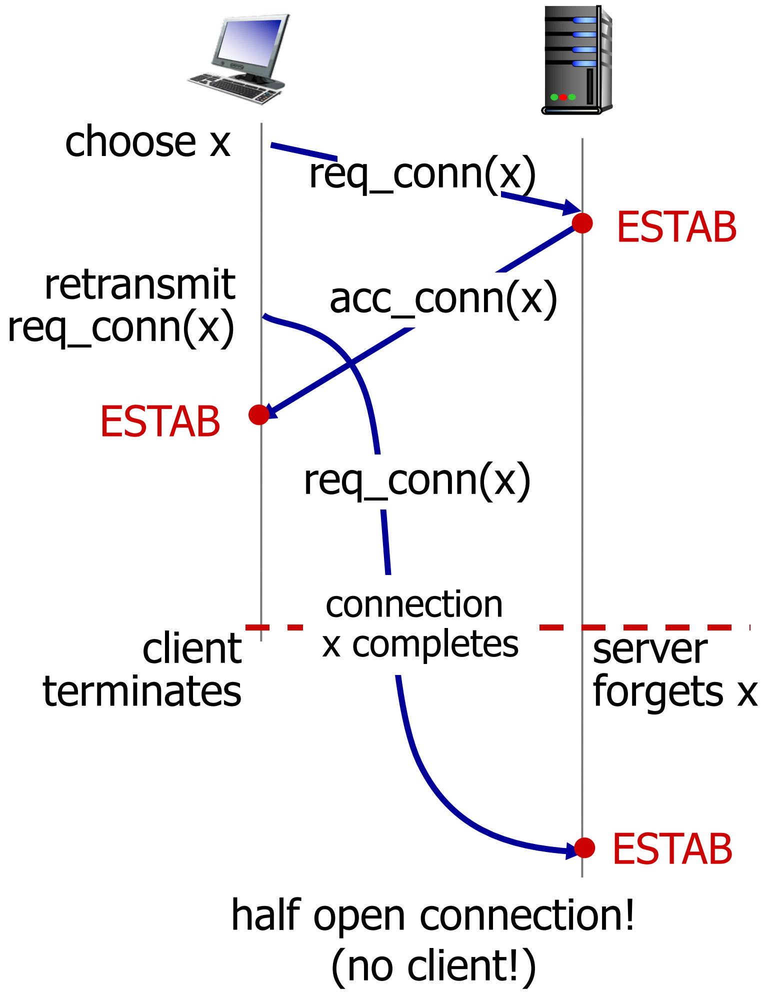


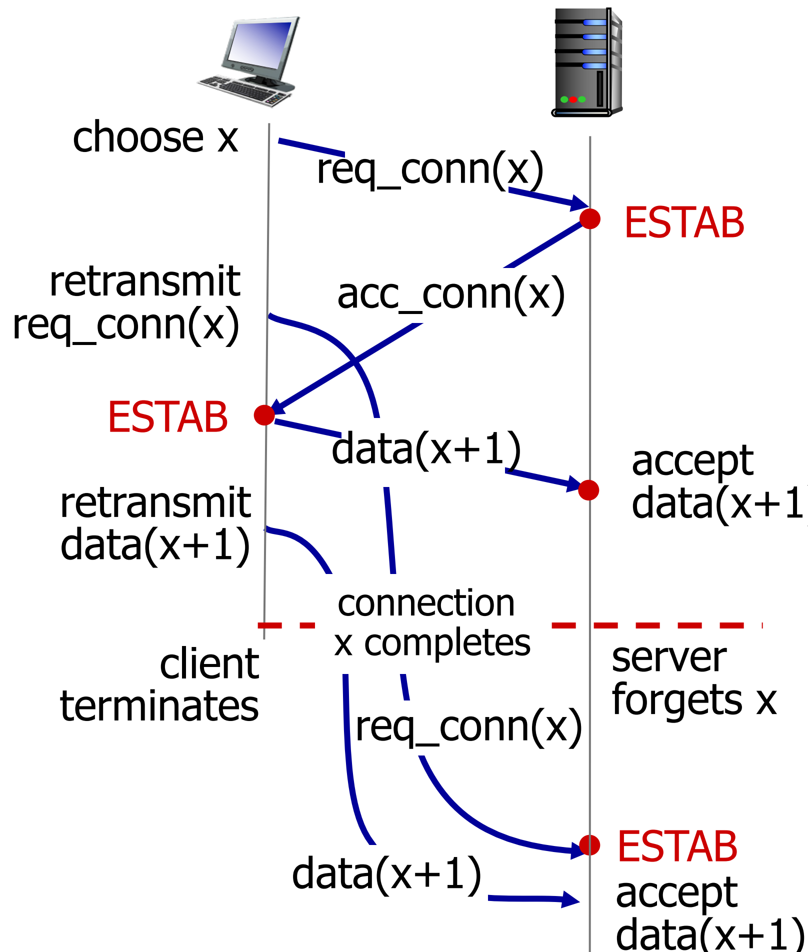
Q: will 2-way handshake always work in network?

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

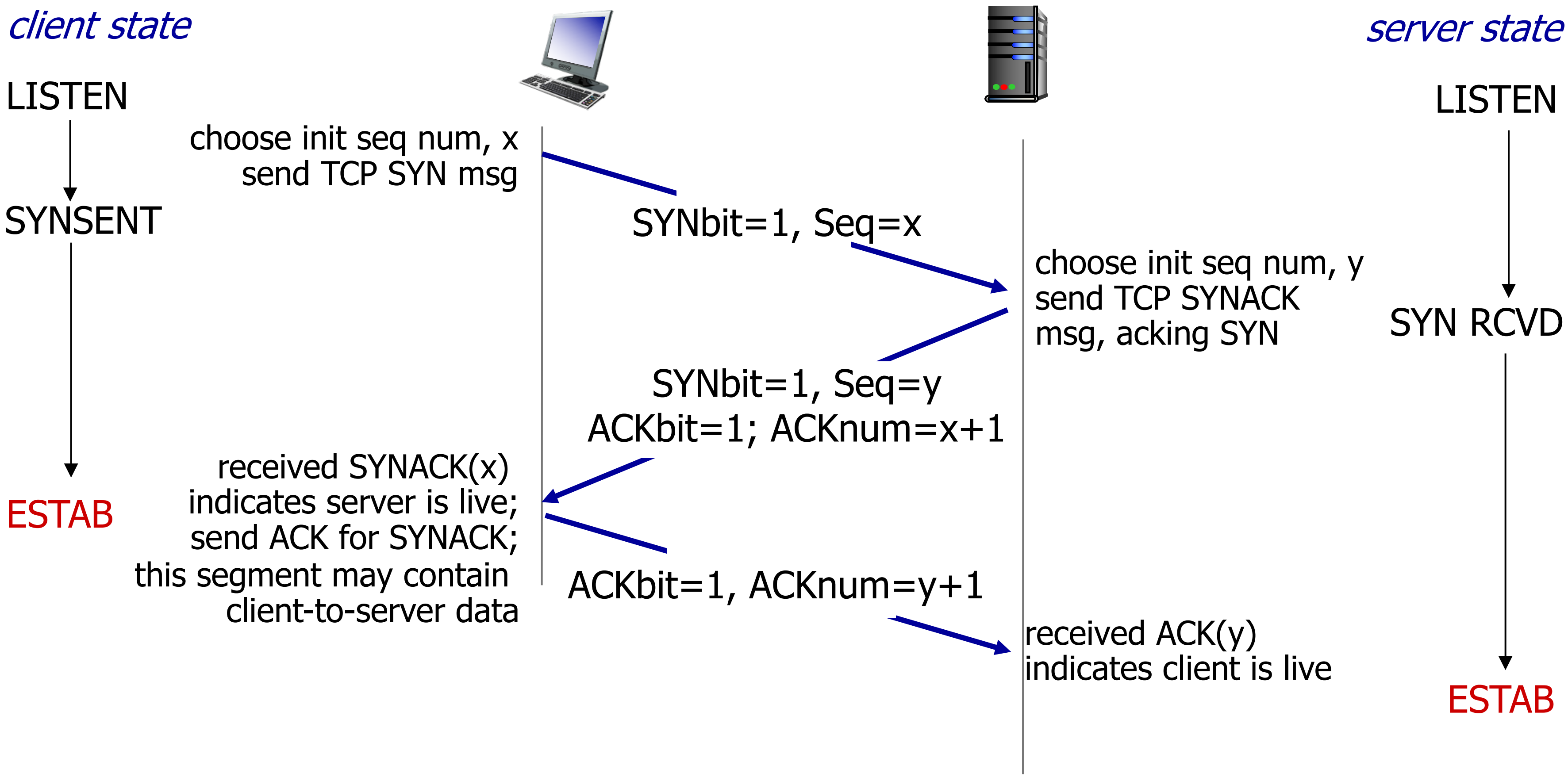
Agreeing to establish a connection

2-way handshake failure scenarios:

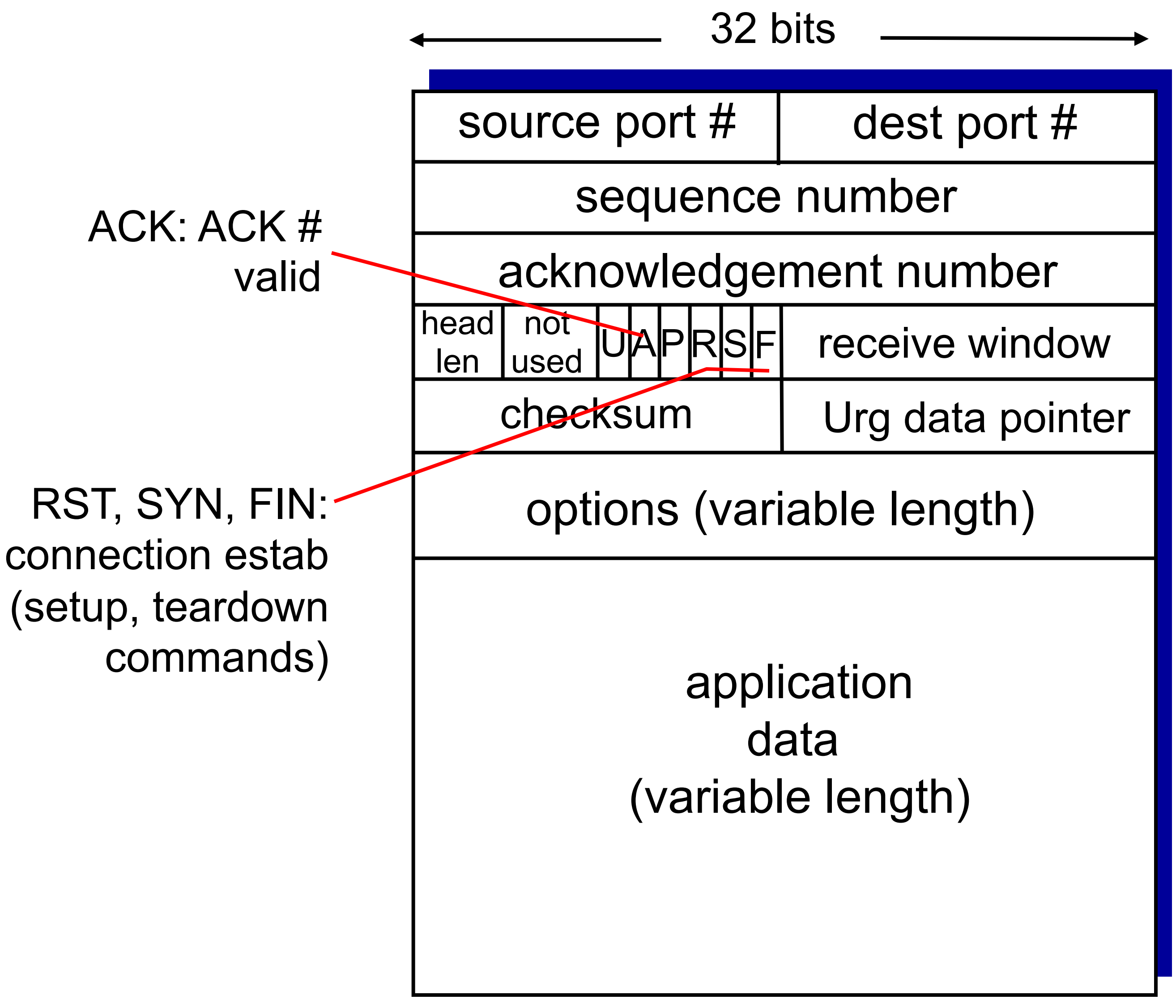




TCP 3-way handshake



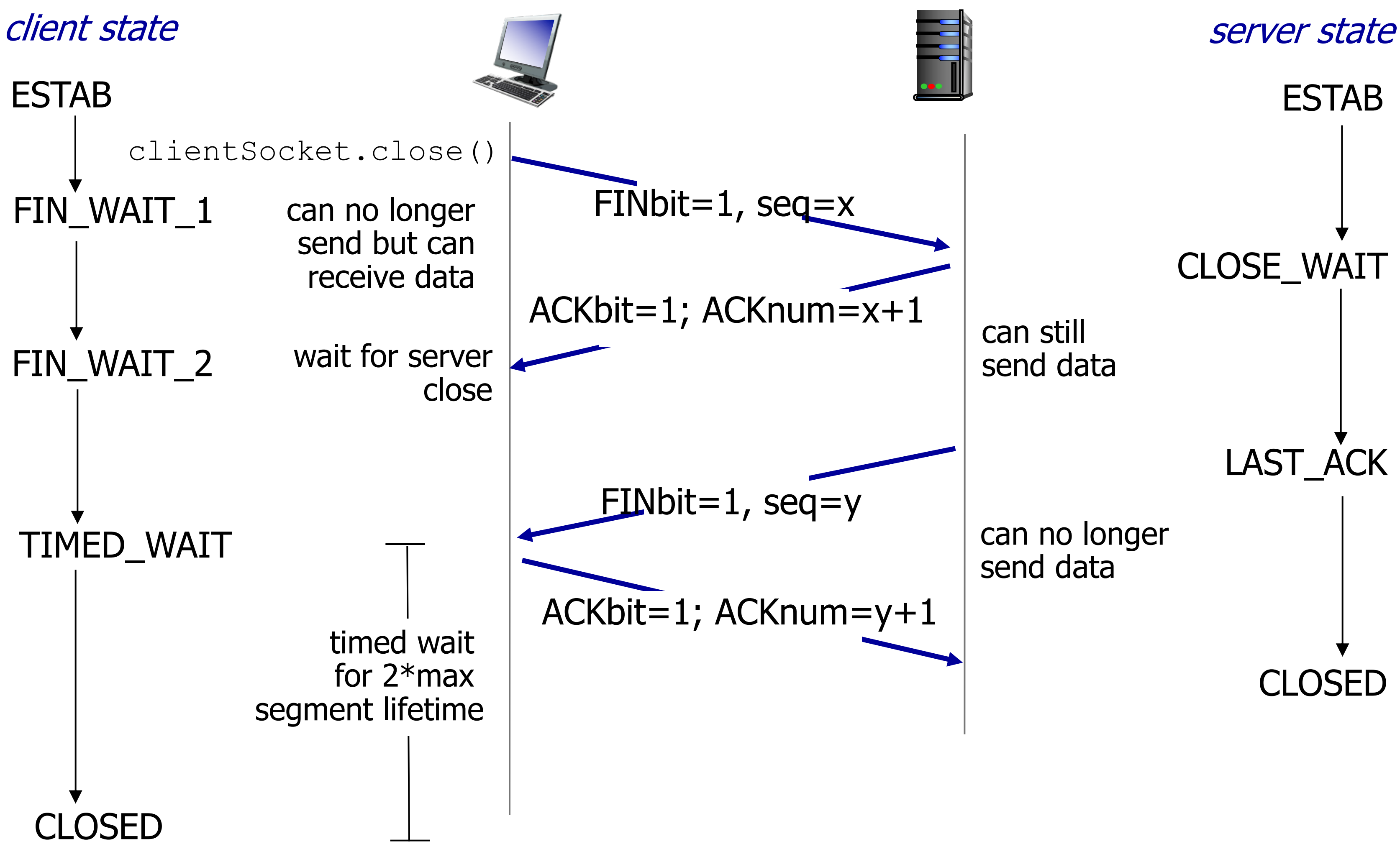
How to set SYNC, ACK bit?



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



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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 (queuing 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

TCP Congestion Control

- Idea
  - Assumes best-effort network
  - Each source determines network capacity for itself
  - Implicit feedback via ACKs or timeout events
  - ACKs pace transmission (self-clocking)

- Challenge
  - Determining initial available capacity
  - Adjusting to changes in capacity in a timely manner

TCP Congestion Control

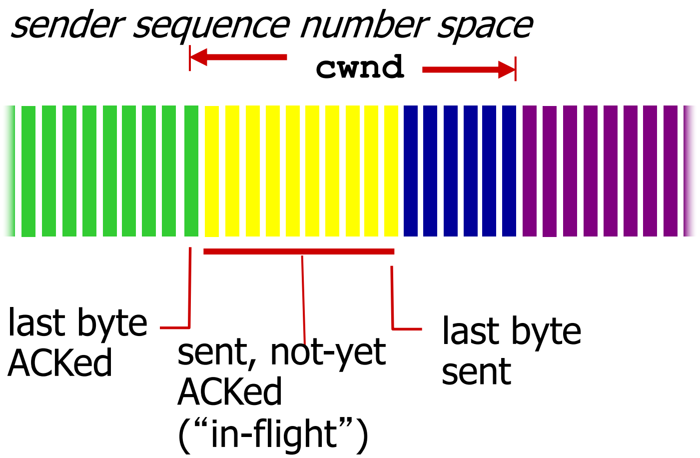
- Basic idea
  - Add notion of congestion window
    - Effective window is the smaller of
    - Advertised window (flow control) rwnd
    - Congestion window (congestion control) cwnd
  - Changes in congestion window size
    - Slow increases to absorb new bandwidth
    - Quick decreases to eliminate congestion

TCP Congestion Control

- sender limits transmission:

LastByteSent-LastByteAcked

≤ cwnd



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

How does sender perceive congestion?

- loss event = timeout *or* 3 duplicate acks
- TCP sender reduces rate (**cwnd**) after loss event

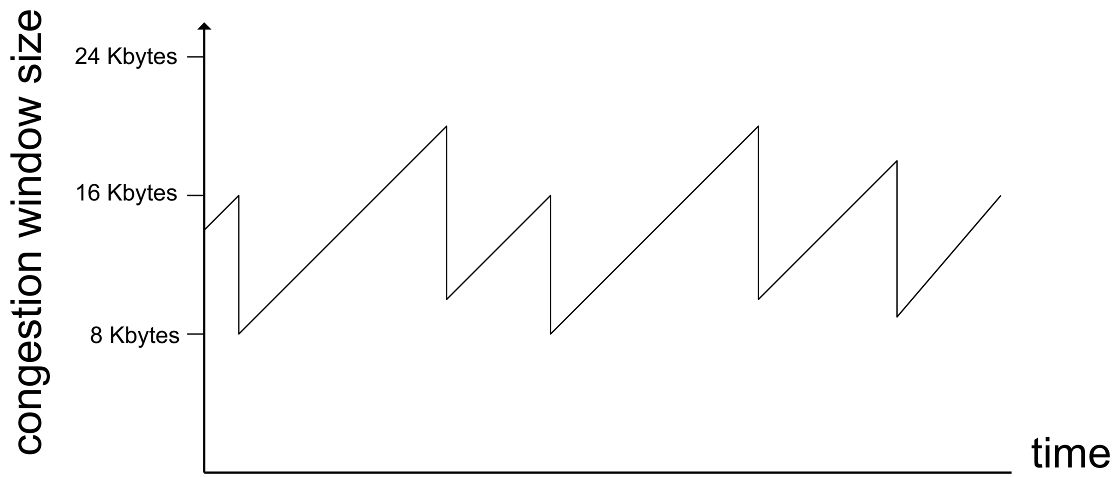
three mechanisms:

- AIMD: how to grow cwnd
- slow start: startup
- conservative after loss (timeout, duplicate ACKs) events

AIMD Rule: additive increase, multiplicative decrease

- *Approach:* increase 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** by 50% after loss

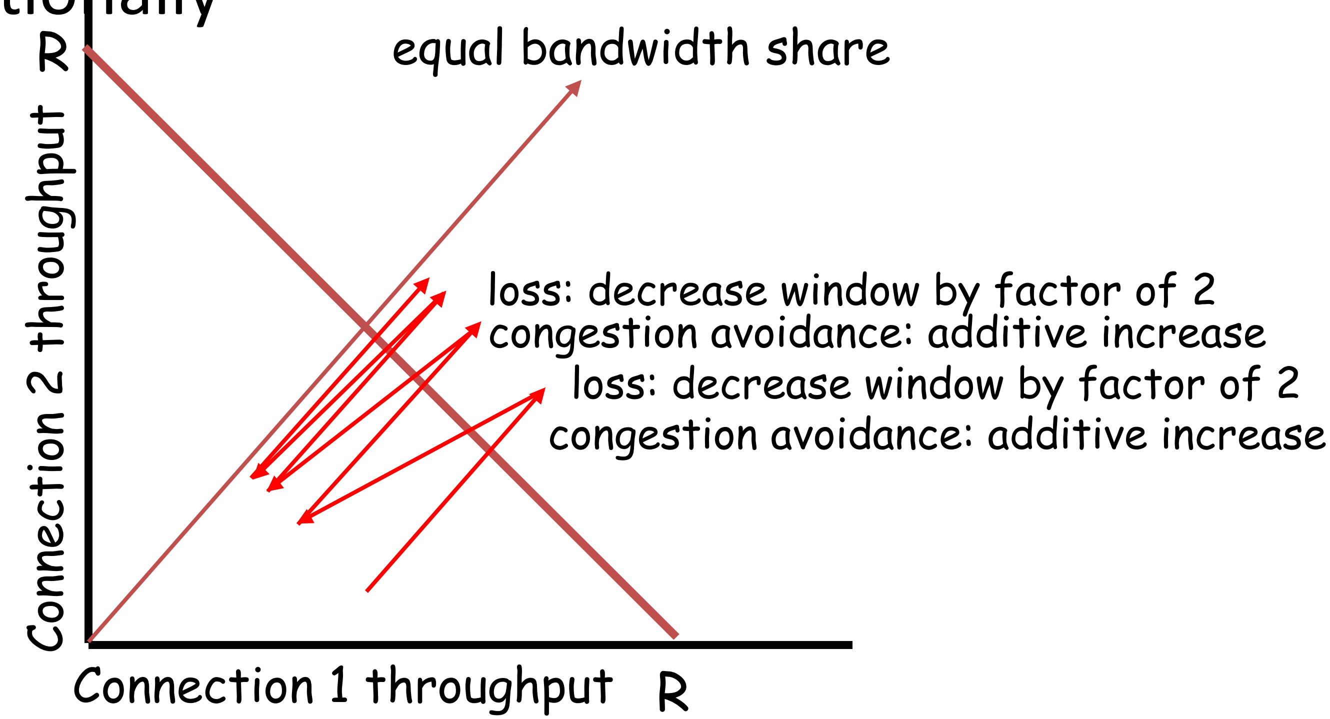
Saw tooth behavior: probing for bandwidth



What AIMD? TCP Fairness

Two competing sessions:

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



TCP Congestion Control (RFC 5681)

How to implement TCP Congestion Control?

Multiple algorithms work together:

- slow start: **how to jump-start**
- congestion avoidance: **additive increase**
- fast retransmit/fast recovery: recover from single packet loss: **multiplicative decrease**
- retransmission upon timeout: **conservative loss/failure handling**

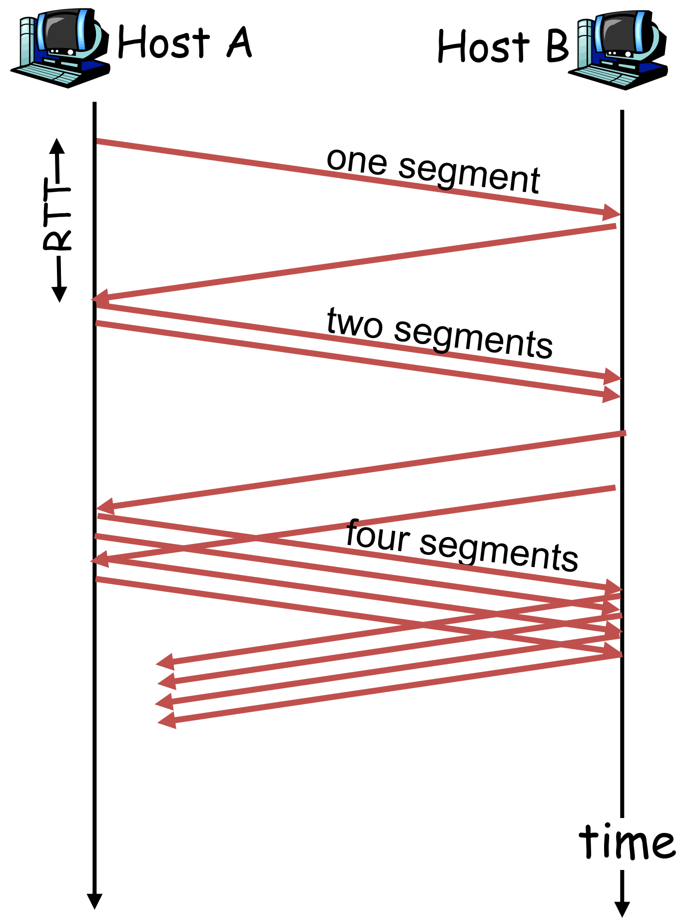
TCP Slow Start

- When connection begins, **cwnd** ≤ **2 MSS**, typically, set cwnd = 1MSS
  - 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

- When connection begins, increase rate exponentially fast until cwnd reaches a threshold value: slow-start-threshold *ssthresh*
  - cwnd < ssthresh

TCP Slow Start (more)

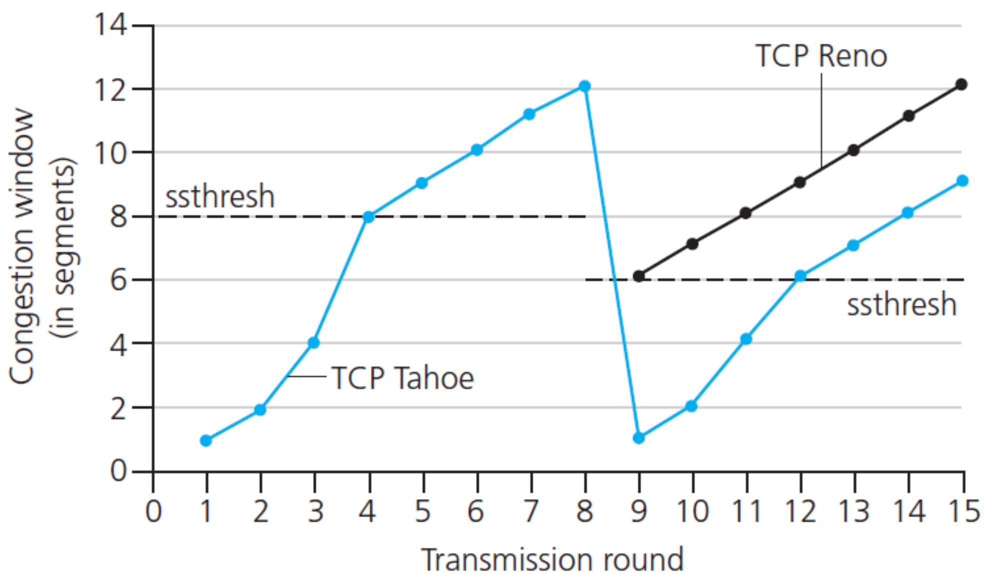
- When connection begins, increase rate exponentially when <u>cwnd<ssthresh</u>
  - Goal: double **cwnd** every RTT by setting
  - <u>**Action: cwnd** += 1 MSS</u> for every ACK received
- <u>Summary:</u> initial rate is slow but ramps up exponentially fast



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

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

Congestion Avoidance

- Goal: increase cwnd by 1 MSS per RTT until congestion (loss) is detected

  - Conditions: when cwnd > ssthresh and no loss occurs

  - Actions: cwnd += (MSS/cwnd)*MSS (bytes) upon every incoming non-duplicate ACK

TCP Congestion Control

Algoritms	condition	Design	action
Slow Start	cwnd <= ssthres;	cwnd doubles per RTT	cwnd+=1MSS per ACK
Congestion Avoidance	cwnd > ssthres	cwnd++ per RTT (additive increase)	cwnd+=1/cwnd * MSS per ACK

When loss occurs

- Detecting losses and reacting to them:
  - through duplicate ACKs
    - fast retransmit / fast recovery
      - Goal: multiplicative decrease cwnd upon loss
  - through retransmission timeout
      - Goal: reset everything

Fast Retransmit/Fast Recovery

- fast retransmit: to detect and repair loss, based on incoming duplicate ACKs
  - **use** 3 duplicate ACKs to infer packet loss
  - set ssthresh = max(cwnd/2, 2MSS)
  - cwnd = ssthresh + 3MSS
  - retransmit the lost packet
- fast recovery: governs the transmission of new data until a non-duplicate ACK arrives
  - **increase** cwnd by 1 MSS upon every duplicate ACK

Philosophy:

- 3 dup ACKs to infer losses and differentiate from transient out-of-order delivery
- receiving each duplicate ACK indicates one more packet left the network and arrived at the receiver

Putting them together

- Initially, fastretx = false;
- If upon 3rd duplicate ACK
  - ssthresh = max (cwnd/2, 2*MSS)
  - cwnd = ssthresh + 3*MSS
    - why add 3 packets here?
  - retransmit the lost TCP packet
  - Set fastretx = true;
- If fastretx == true; upon each *additional* duplicate ACK
  - cwnd += 1 MSS
  - transmit a new packet if allowed
    - by the updated cwnd and rwnd
- If fastretx == true; upon a new (i.e., non-duplicate) ACK
  - cwnd = ssthresh
  - Fastretx = false;   // After fast retx/fast recovery, cwnd decreases by half

Retransmission Timeout

when retransmission timer expires

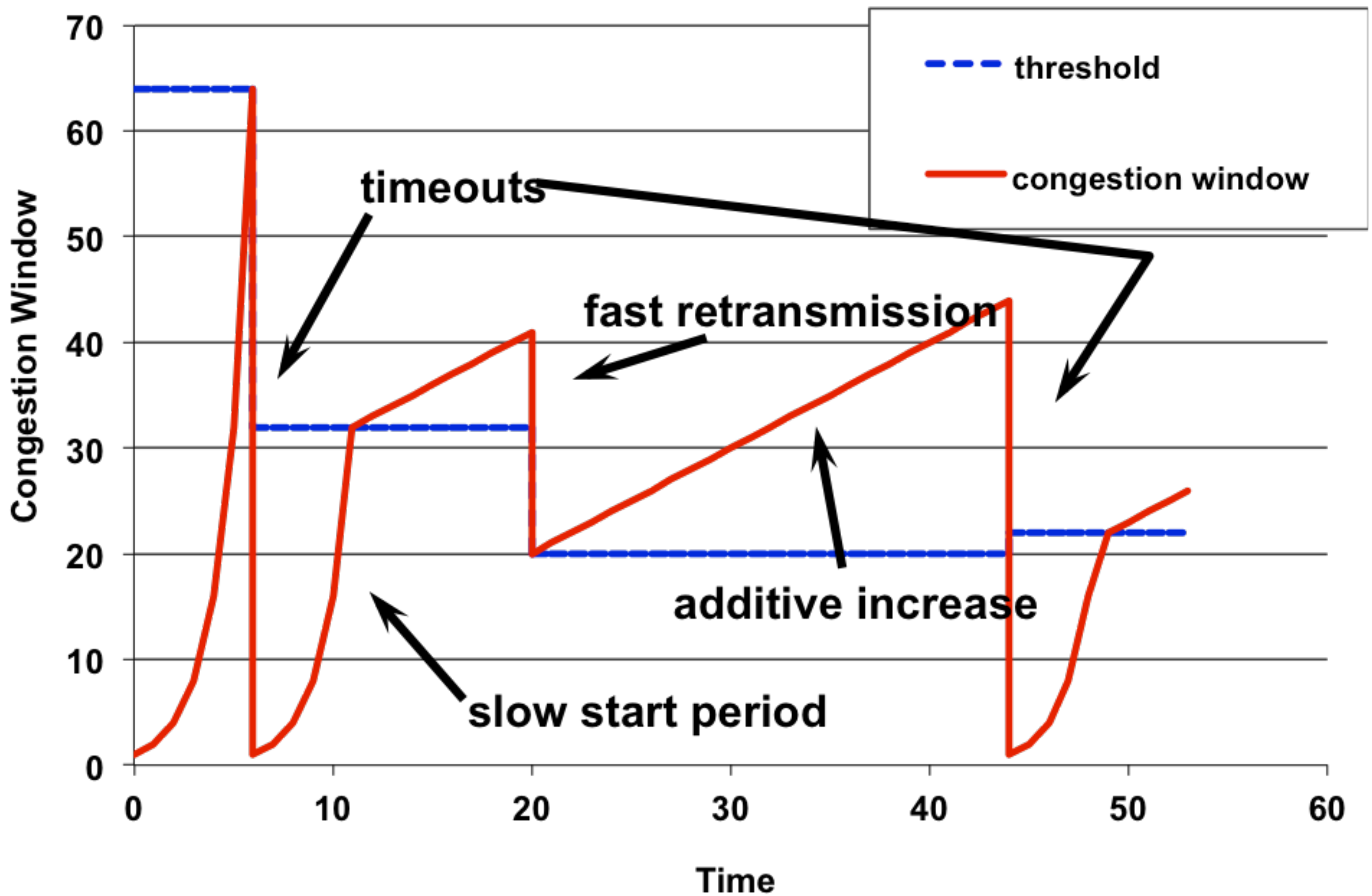
- ssthresh = max (cwnd/2, 2*MSS)
  - cwnd should be flight size to be more accurate
  - see RFC 2581

- cwnd = 1 MSS

- retransmit the lost TCP packet

- why resetting?
  - heavy loss detected

TCP Congestion Window Trace



TCP Congestion Control

Algoritms	condition	Design	action
Slow Start	cwnd <= ssthres;	cwnd doubles per RTT	cwnd+=1MSS per ACK
Congestion Avoidance	cwnd > ssthres	cwnd++ per RTT (additive increase)	cwnd+=1/cwnd * MSS per ACK
fast retx	3 duplicate ACK	reduce the cwnd by half (multicative decreasing)	ssthres = max(cwnd/2,2) cwnd = ssthres + 3 MSS; retx the lost packet
fast recovery	receiving a new ACK after fast retx	finish the 1/2 reduction of cwnd in fast retx/fast recovery	cwnd = ssthres; tx if allowed by cwnd
	upon a dup ACK after fast retx before fast recovery	("transition phrase)	cwnd +=1MSS; Note: it is different from slow start.
RTO timeout	time out	reset	ssthres = max(cwnd/2,2) cwnd = 1; retx the lost packet

Practice

- The receiver acknowledges every segment, and the sender always has data to transmit.
- Initially ***ssthresh*** at the sender is set to 4. Assume ***cwnd*** and ***ssthresh*** are measured in segments.

- Assumptions for simplification
  - When ssthresh ==cwnd, use slow start algorithm
  - In congestion avoidance, let us set cwnd = cwnd + 1/[cwnd]
  - All data delivery is done in segments, so we can send the interger number of segments (for example, cwnd = 2.5MSS, we can send 2 segments)
  - All out-of-order segments will be buffered at the receiver side

Practice

- Retransmission timeout (RTO) is initially set to 150ms at the sender and is unchanged during the connection lifetime.
- Latency in sending a segment: 30ms; latency in sending an ACK; 20ms; Ignore processing delays at sender & receiver.
- Data transmission starts at time t = 0, and the initial sequence# starts from 1. Segment with sequence#=5 is lost once. No other segments are lost.
- How long does it take, in milliseconds, for the sender to receive the *ack* for the segment with the sequence number 14? show your diagram.

Solution (Diagram in next slide)

Algorithm | **After algo runs**

ssh =4 cwnd =1

1					

slow start (upon ack 1)

ssh =4 cwnd =2

2	3				

slow start (upon ack 2, ack 3)

ssh =4 cwnd =4

4	5	6	7		

slow start (upon ack4) ssh =4 cwnd =5

5	6	7	8	9	

Do nothing upon ack4 (1st dup)
Do nothing upon ack4 (2nd dup)

Fast retx (upon 3 dup ack4)

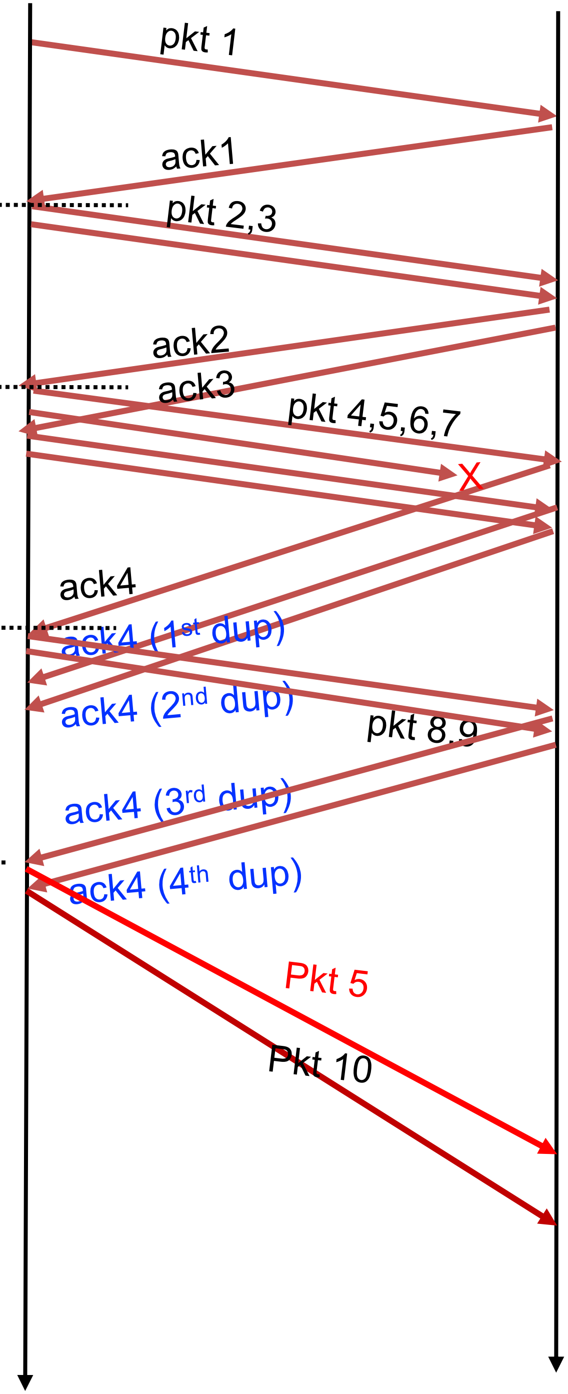
ssh = max(2, 5/2) = 2.5
cwnd = ssh + 3 = 5.5
Retx pkt 5

5	6	7	8	9	

Fast recovery w/ additional dup ACK (upon 4th dup)

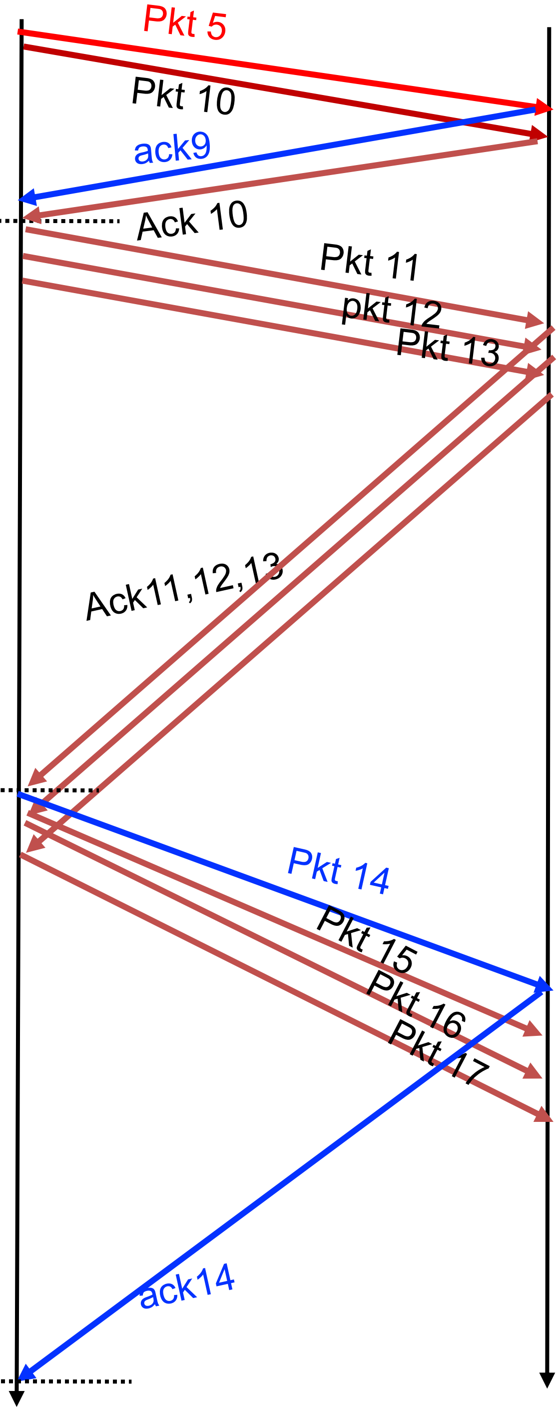
ssh = 2.5, cwnd = 5.5 +1 =6.5
send pkt 10

5	6	7	8	9	10



Algorithm

After algo runs



Fast recovery w/ a new ACK (upon ack 9)

ssh = 2.5

cwnd = ssh = 2.5

Fast retx/fast recovery is over

10	11				

Slow start upon ack 10

ssh = 2.5

cwnd =2.5 + 1 = 3.5

Send new packets 12,13

11	12	13			

Congestion avoidance upon ack 11

ssh = 2.5

cwnd = 3.5 + 1/3 < 4

Send new packets 14

12	13	14			

Congestion avoidance upon ack 12, 13

ssh = 2.5

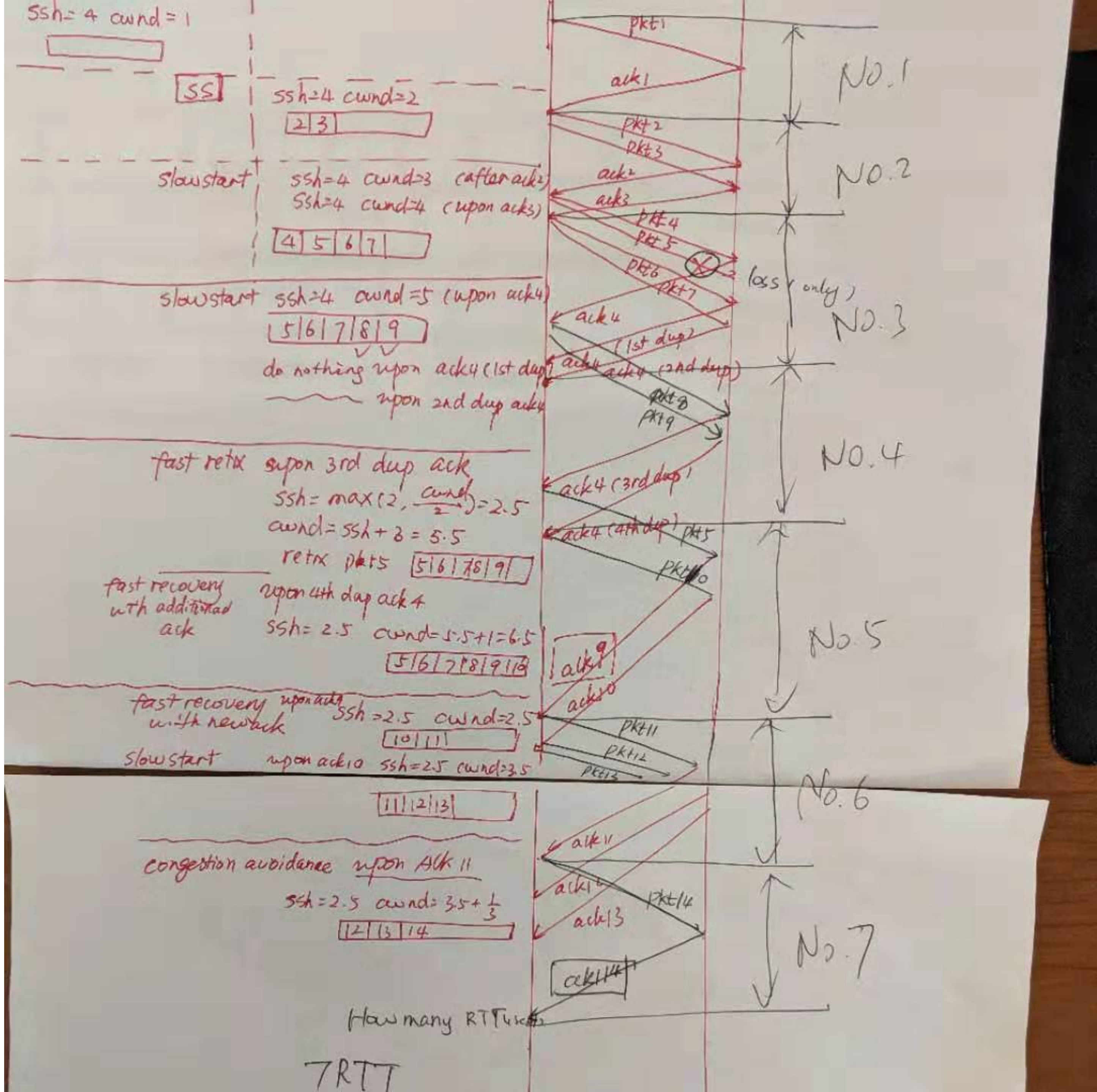
cwnd = 3.5 + 1/3 + 1/3 + 1/4

Send new packets 14

14	15	16	17		

Solution

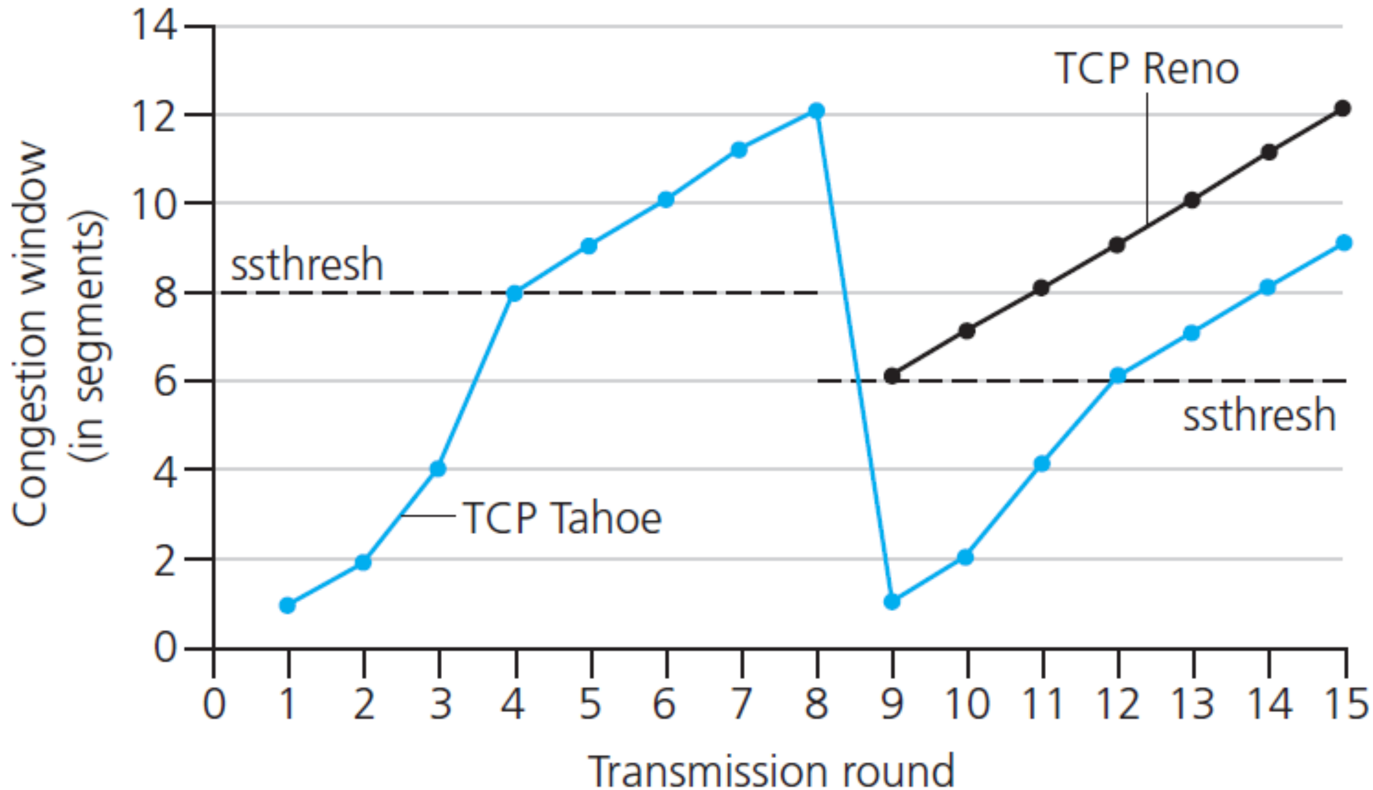
7 RTTs needed: 7 * (30+20) = 350ms



More Practice

- What if the retransmitted segment 5 gets loss again?
- Name other scenarios you want to play with

Practice:
TCP Congestion Window Trace



Putting Things Together in TCP

- use selective repeat to do reliable data transfer for a window of packets *win* at any time

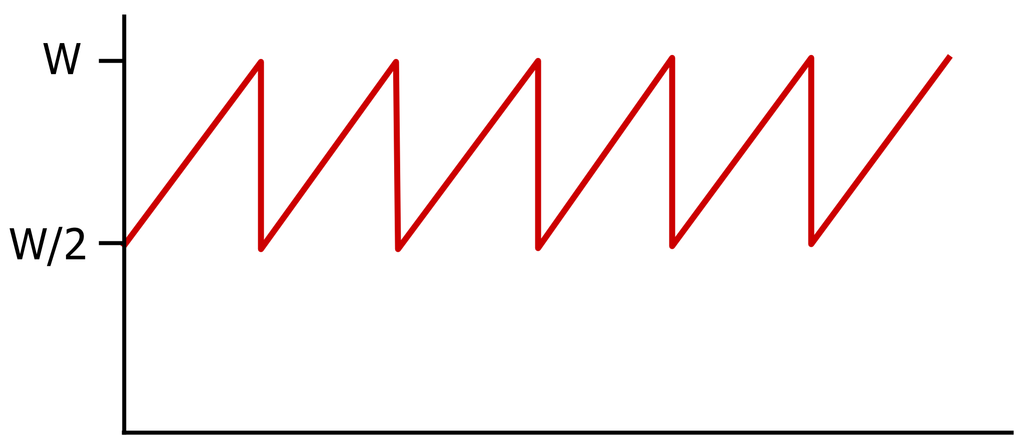
- update win = min (cwnd, rwnd)
  - cwnd is updated by TCP congestion control
  - rwnd is updated by TCP flow control

- Example: cwnd = 20; rwnd = 10

TCP throughput

- avg. TCP thruput as function of window size, RTT?
  - ignore slow start, assume always data to send
- W: window size (measured in bytes) where loss occurs
  - avg. window size (# in-flight bytes) is ¾ W
  - avg. thruput is 3/4W per RTT

$$\text{avg TCP thruput} = \frac{3}{4}\frac{W}{RTT} \text{ bytes/sec}$$



TCP Futures: TCP over "long, fat pipes"

- example: 1500 byte segments, 100ms RTT, want 10 Gbps throughput
- requires W = 83,333 in-flight segments
- throughput in terms of segment loss probability, L [Mathis 1997]:

$$\text{TCP throughput} = \frac{1.22 \cdot \text{MSS}}{\text{RTT}\sqrt{L}}$$

➜ to achieve 10 Gbps throughput, need a loss rate of L = $2 \cdot 10^{-10}$ *– a very small loss rate!*

- new versions of TCP for high-speed