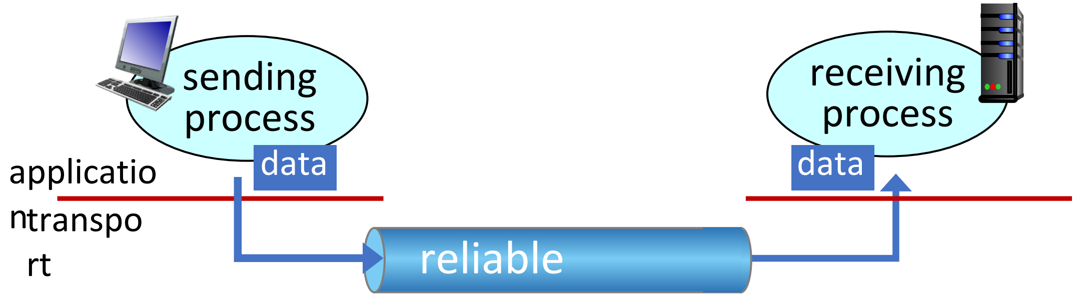
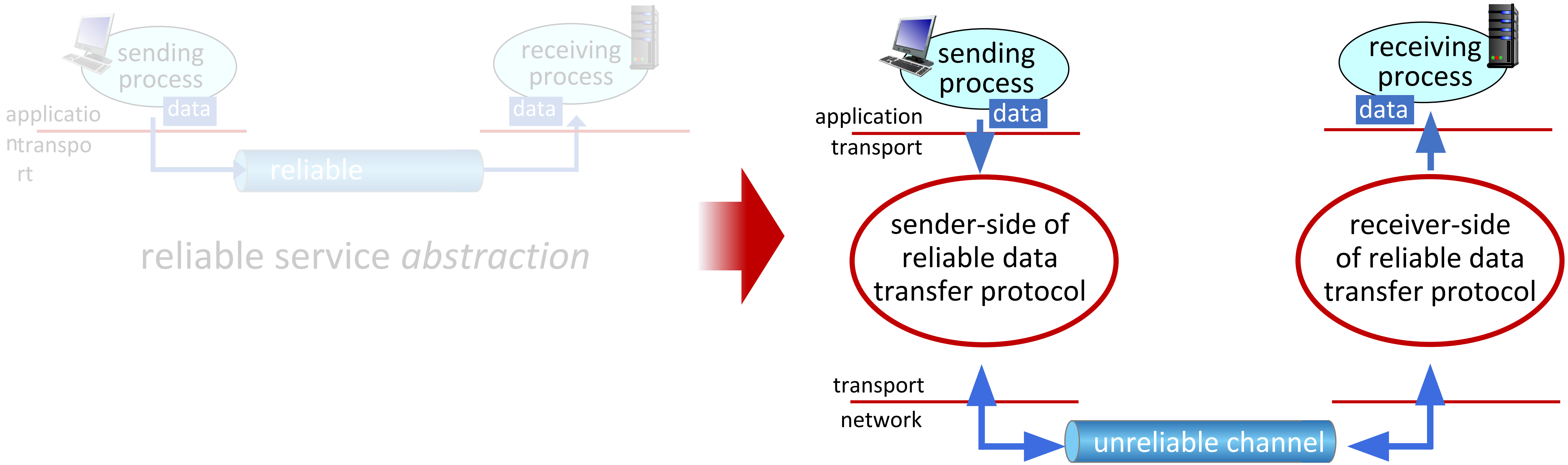
Principles of reliable data transfer



reliable service *abstraction*

Principles of reliable data transfer

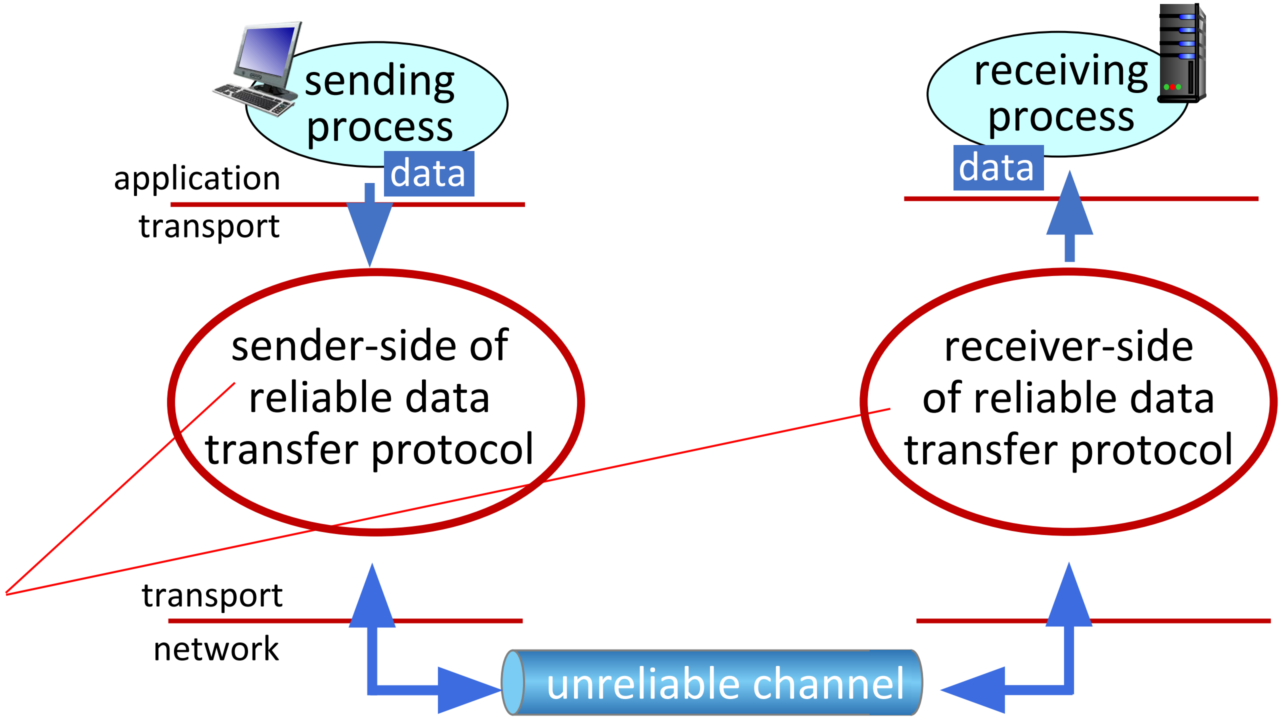


reliable service *abstraction*

reliable service *implementation*

Principles of reliable data transfer

Complexity of reliable data transfer protocol will depend (strongly) on characteristics of unreliable channel (lose, corrupt, reorder data?)

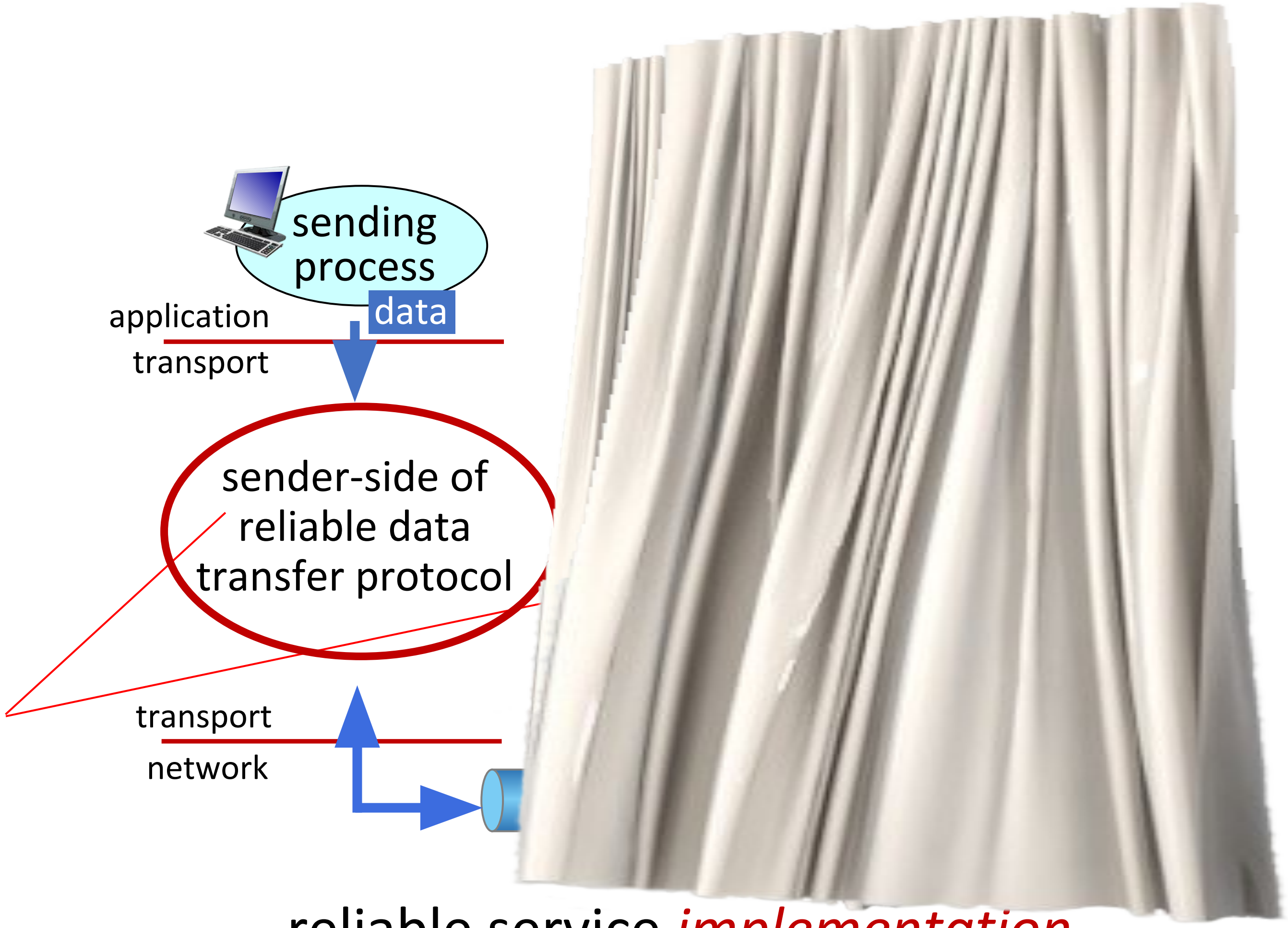

sending process
data
application
transport
sender-side of reliable data transfer protocol
transport
network
unreliable channel
receiving process
data
receiver-side of reliable data transfer protocol

reliable service *implementation*

Principles of reliable data transfer

Sender, receiver do *not* know the “state” of each other, e.g., was a message received?

- unless communicated via a message

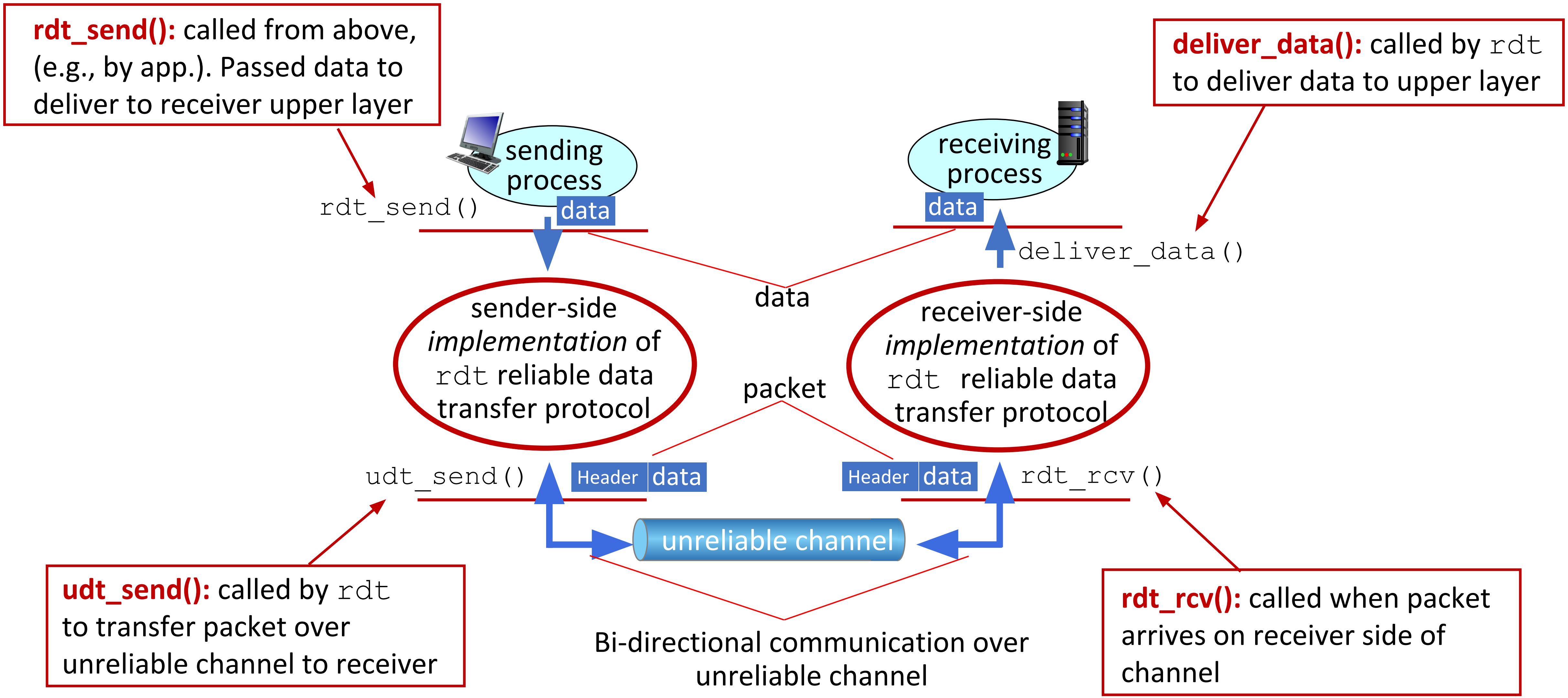


reliable service *implementation*

rdt protocol mechanisms:

- error detection (e.g., checksum)
- ACKs, NAKs
- retransmission
- sequence numbers (duplicate detection)

Reliable data transfer protocol (rdt): interfaces



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 layer

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

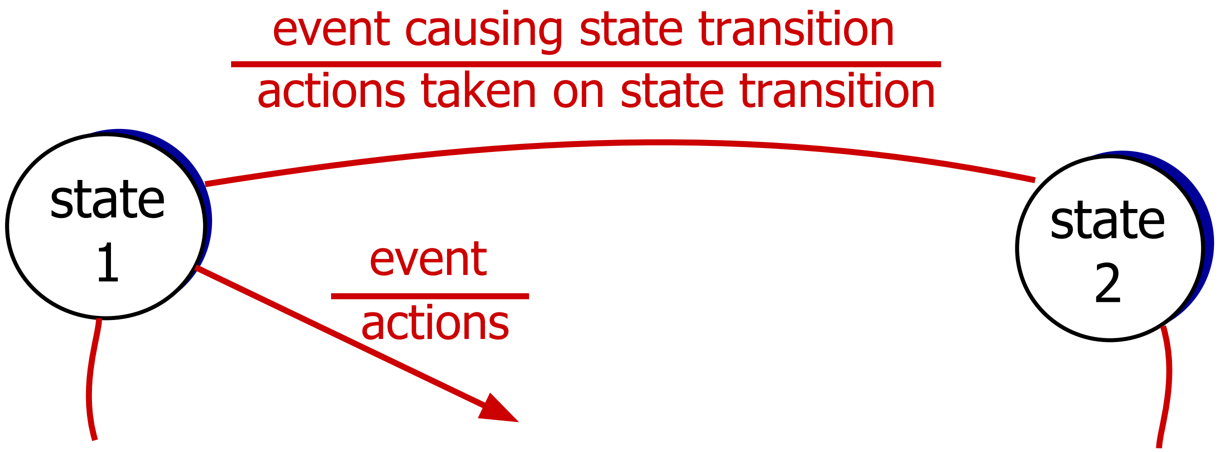
rdt_rcv(): called when packet arrives on receiver side of channel

sending process
data
rdt_send()
receiving process
data
deliver_data()
data
sender-side *implementation* of rdt reliable data transfer protocol
receiver-side *implementation* of rdt reliable data transfer protocol
packet
udt_send()
Header data
Header data
rdt_rcv()
unreliable channel
Bi-directional communication over unreliable channel

Reliable data transfer: getting started

We will:

- 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

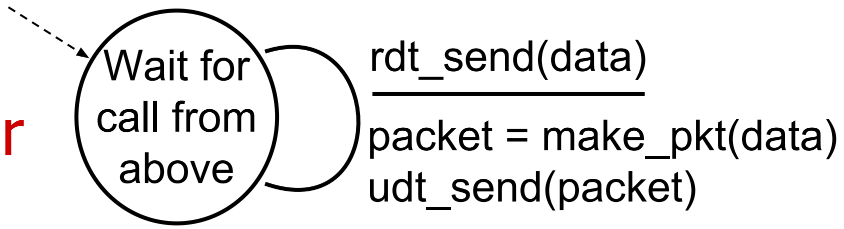

event causing state transition
actions taken on state transition
state 1
event
actions
state 2

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

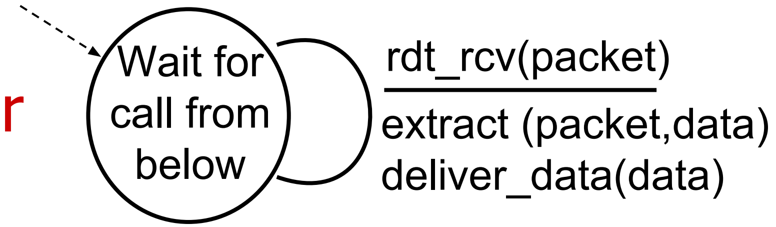
rdt1.0: reliable transfer over a reliable channel

- 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 (e.g., Internet 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

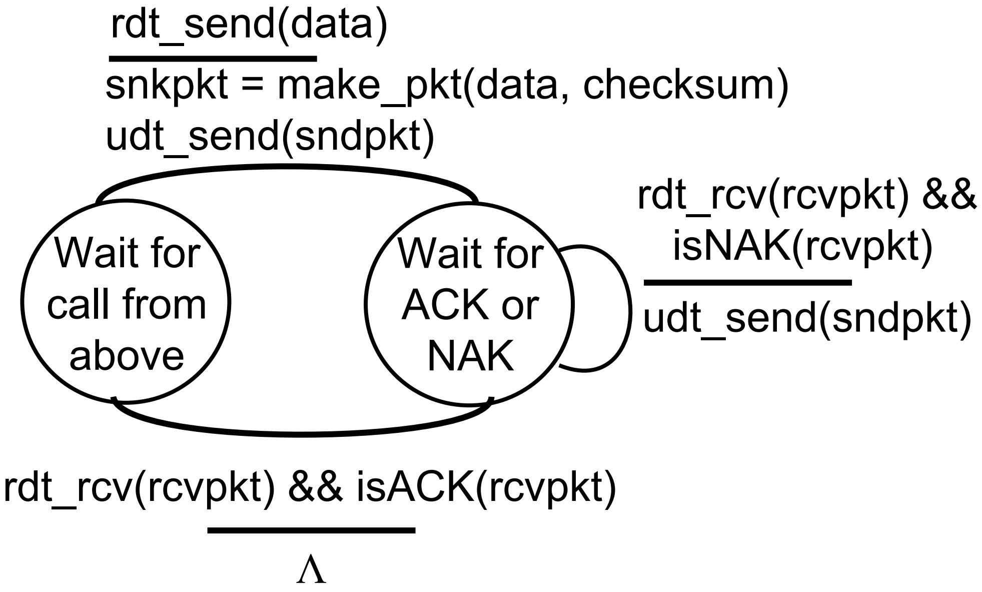
- underlying channel may flip bits in packet
  - checksum to detect bit errors
- *the* question: how to recover from errors?
  - *acknowledgements (ACKs):* receiver explicitly tells sender that pkt received OK
  - *negative acknowledgements (NAKs):* receiver explicitly tells sender that pkt had errors
  - sender *retransmits* pkt on receipt of NAK

stop and wait

sender sends one packet, then waits for receiver response

rdt2.0: FSM specifications

sender



rdt_send(data)

snkpkt = make_pkt(data, checksum)
udt_send(sndpkt)

Wait for call from above

Wait for ACK or NAK

rdt_rcv(rcvpkt) &&
isNAK(rcvpkt)

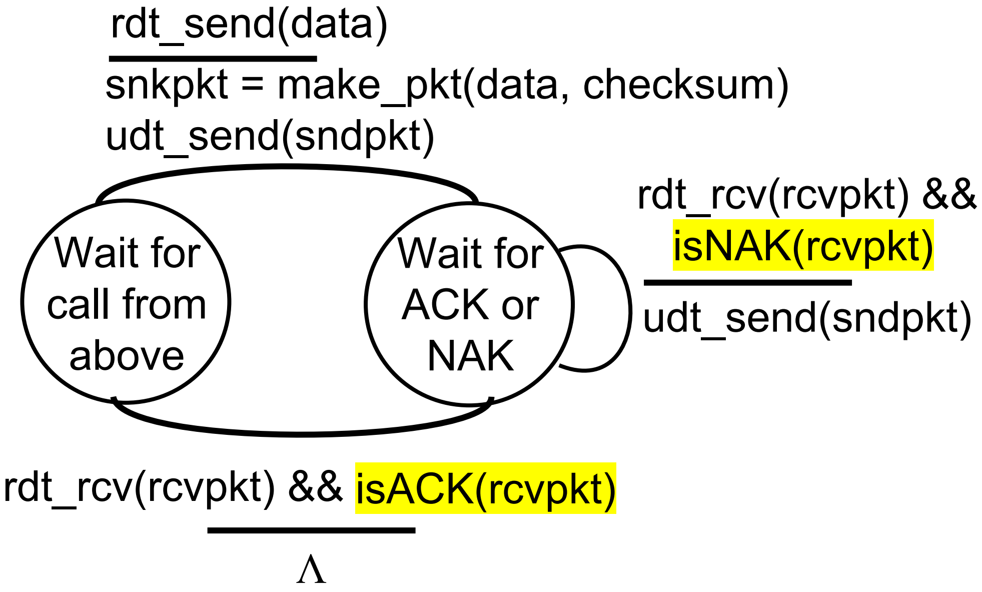
udt_send(sndpkt)

rdt_rcv(rcvpkt) && isACK(rcvpkt)

Λ

rdt2.0: FSM specification

sender



rdt_send(data)

snkpkt = make_pkt(data, checksum)
udt_send(sndpkt)

Wait for call from above

Wait for ACK or NAK

rdt_rcv(rcvpkt) && isNAK(rcvpkt)

udt_send(sndpkt)

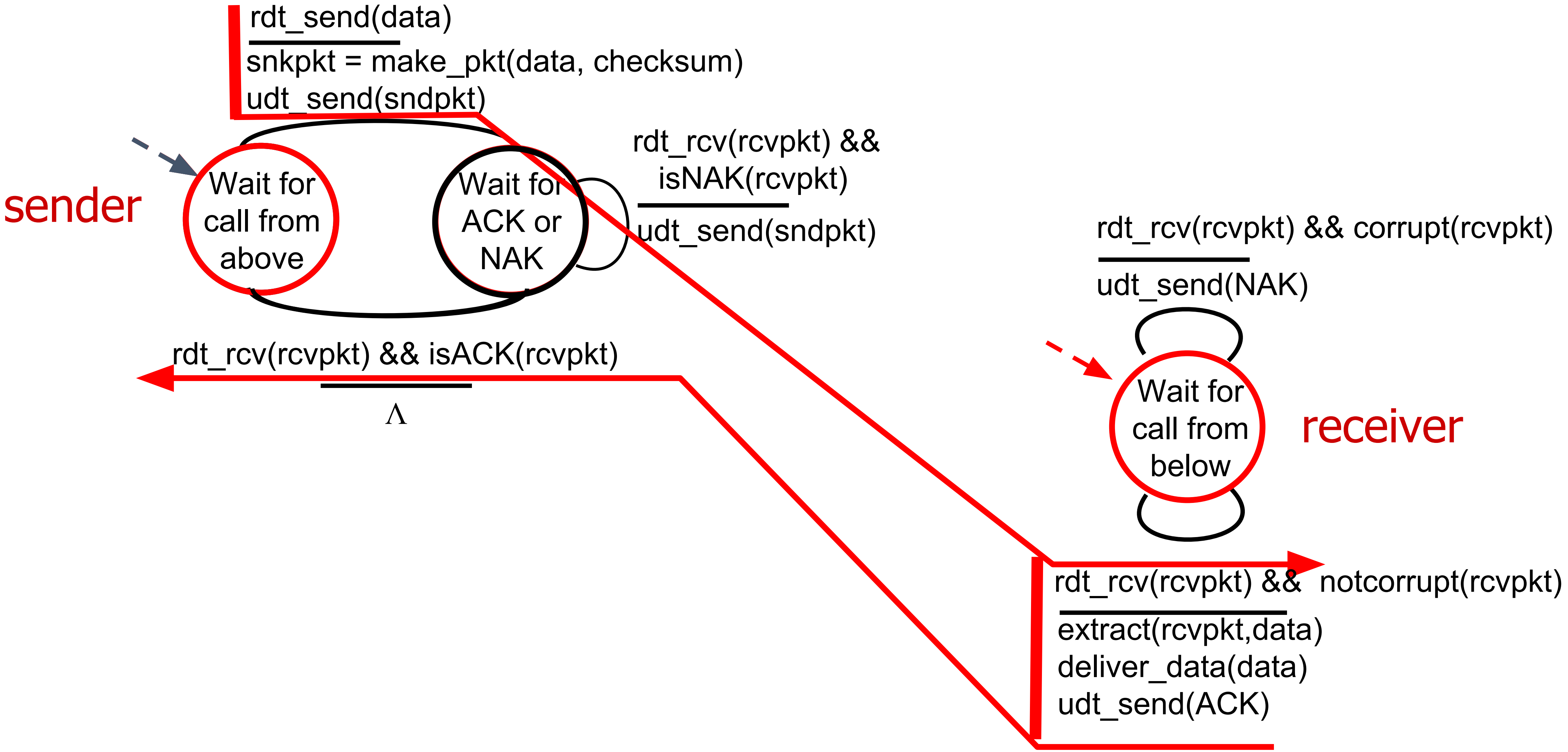
rdt_rcv(rcvpkt) && isACK(rcvpkt)

Λ

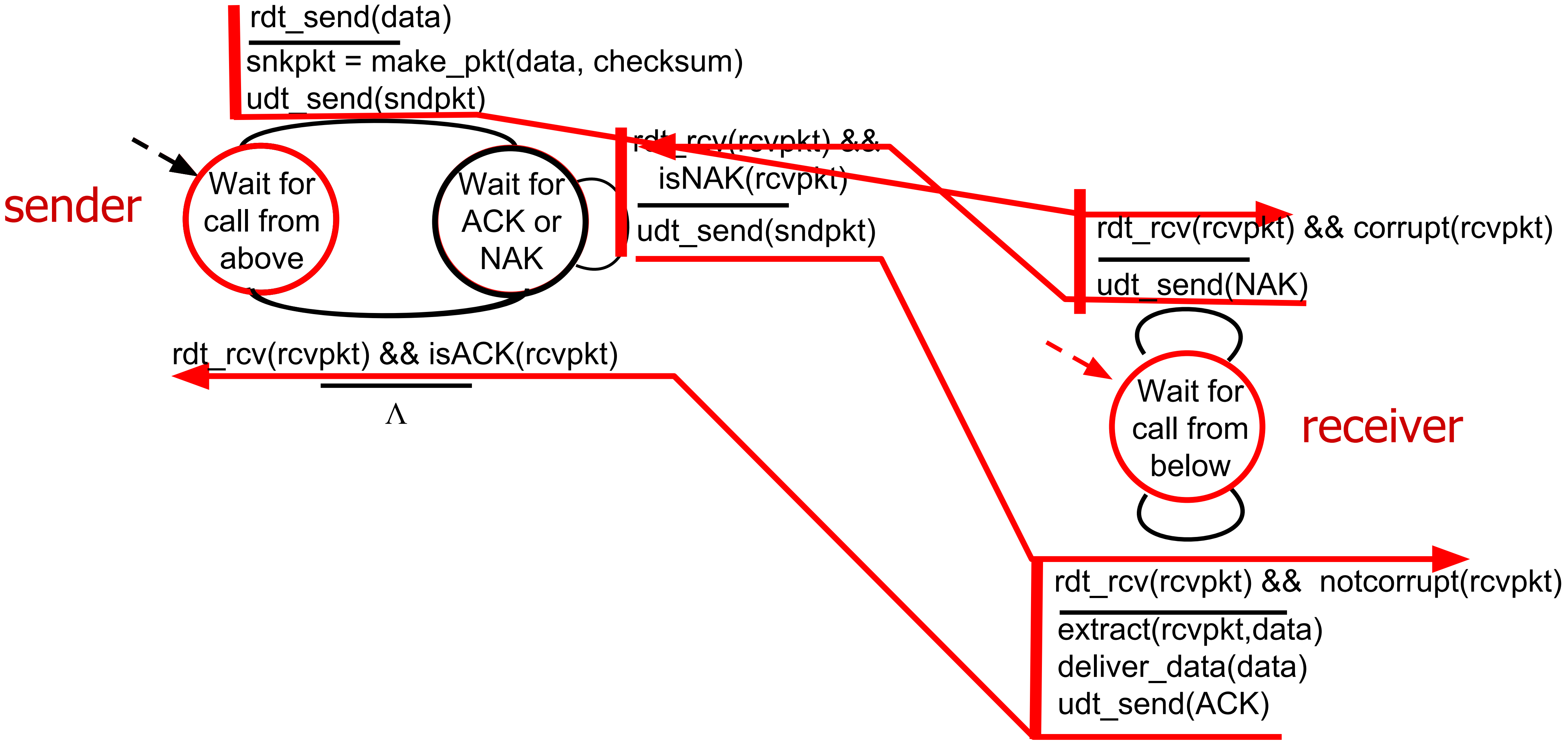
Note: “state” of receiver (did the receiver get my message correctly?) isn’t known to sender unless somehow communicated from receiver to sender

- that’s why we need a protocol!

rdt2.0: operation with no errors



rdt2.0: corrupted packet 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

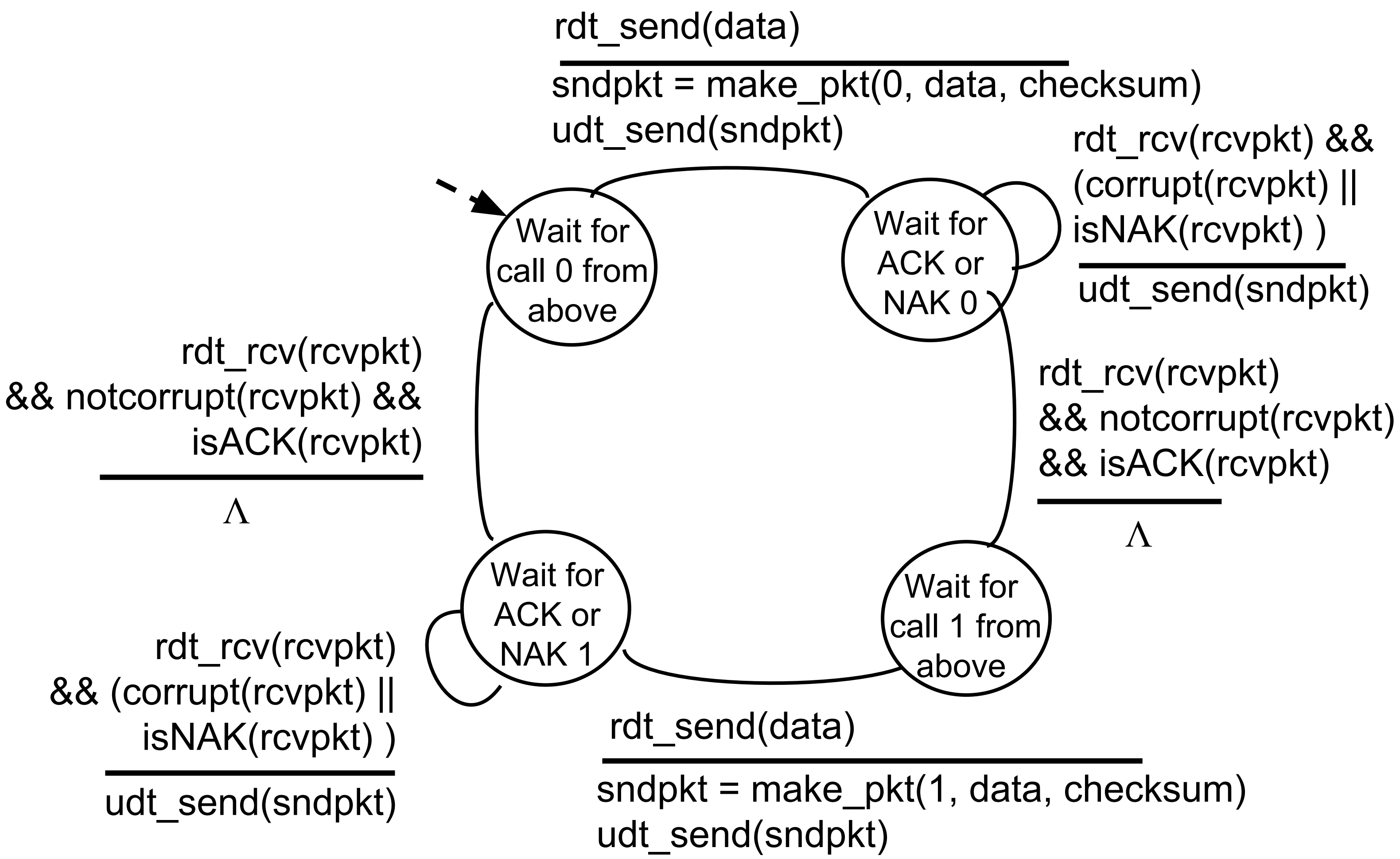
handling duplicates:

- 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, handling garbled ACK/NAKs



Wait for call 0 from above

Wait for ACK or NAK 0

Wait for call 1 from above

Wait for ACK or NAK 1

rdt_send(data)
sndpkt = make_pkt(0, data, checksum)
udt_send(sndpkt)

rdt_rcv(rcvpkt) &&
(corrupt(rcvpkt) ||
isNAK(rcvpkt))
udt_send(sndpkt)

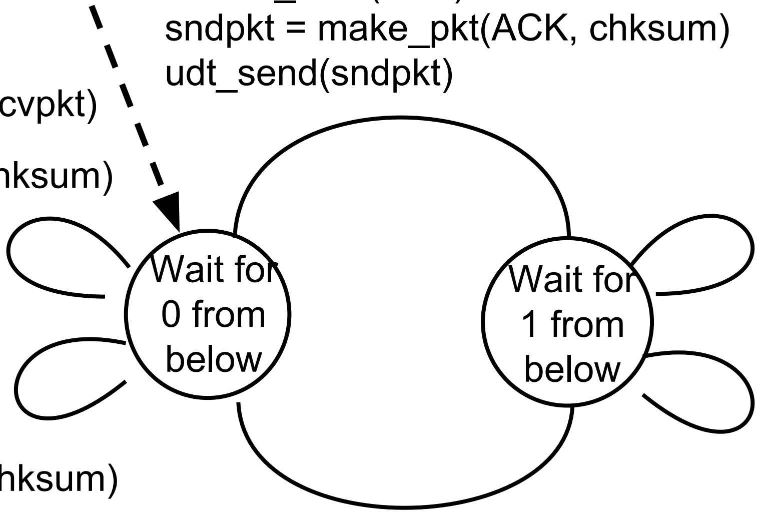
rdt_rcv(rcvpkt)
&& notcorrupt(rcvpkt)
&& isACK(rcvpkt)
Λ

rdt_send(data)
sndpkt = make_pkt(1, data, checksum)
udt_send(sndpkt)

rdt_rcv(rcvpkt)
&& (corrupt(rcvpkt) ||
isNAK(rcvpkt))
udt_send(sndpkt)

rdt_rcv(rcvpkt)
&& notcorrupt(rcvpkt) &&
isACK(rcvpkt)
Λ

rdt2.1: receiver, handling 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)

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 "expected" pkt should have seq # of 0 or 1

receiver:

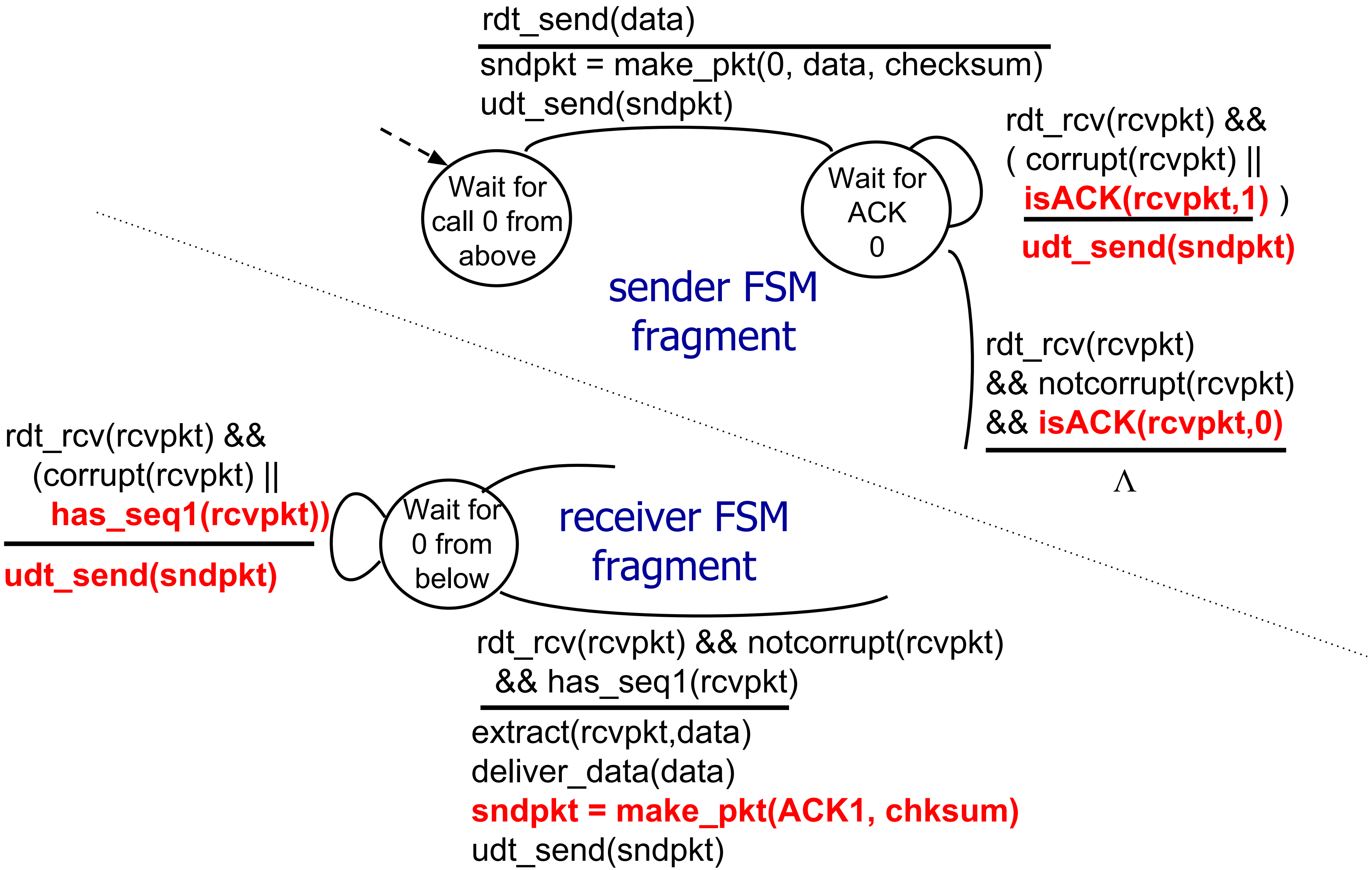
- must check if received packet is duplicate
  - state indicates whether 0 or 1 is expected pkt seq #
- note: receiver can *not* know if its last ACK/NAK received OK at sender

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*

As we will see, TCP uses this approach to be NAK-free

rdt2.2: sender, receiver fragments



rdt_send(data)

sndpkt = make_pkt(0, data, checksum)
udt_send(sndpkt)

Wait for call 0 from above

Wait for ACK 0

rdt_rcv(rcvpkt) &&
(corrupt(rcvpkt) ||
isACK(rcvpkt,1))

udt_send(sndpkt)

sender FSM fragment

rdt_rcv(rcvpkt)
&& notcorrupt(rcvpkt)
&& **isACK(rcvpkt,0)**

Λ

rdt_rcv(rcvpkt) &&
(corrupt(rcvpkt) ||
has_seq1(rcvpkt))

udt_send(sndpkt)

Wait for 0 from below

receiver FSM fragment

rdt_rcv(rcvpkt) && notcorrupt(rcvpkt)
&& has_seq1(rcvpkt)

extract(rcvpkt,data)
deliver_data(data)
sndpkt = make_pkt(ACK1, chksum)
udt_send(sndpkt)

rdt3.0: channels with errors *and* loss

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

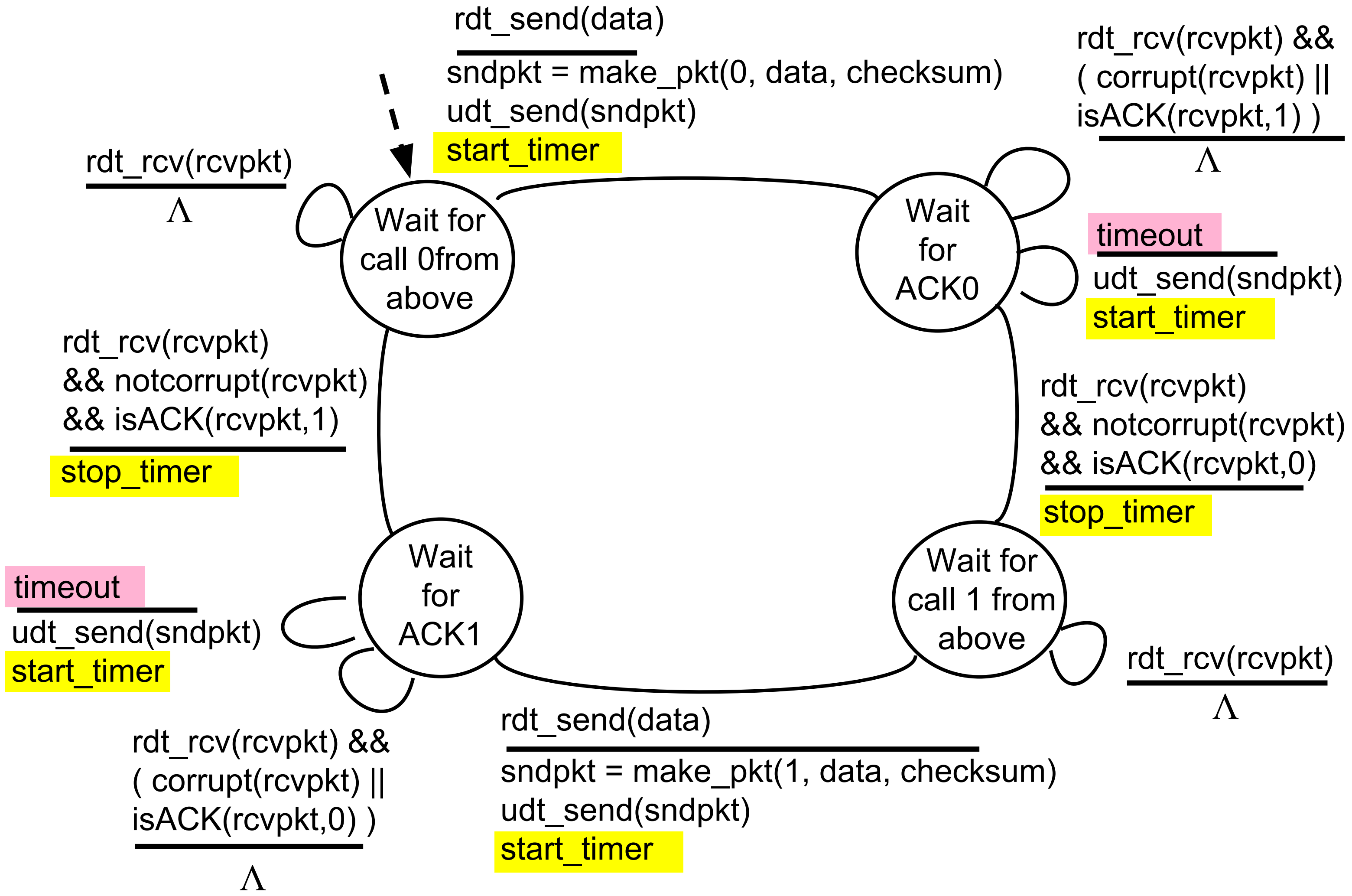
- checksum, seq. #, ACKs, retransmissions will be of help … but not enough

How do humans handle lost sender-to-receiver words in conversation?

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



Wait for call 0from above

Wait for ACK0

Wait for call 1 from above

Wait for ACK1

rdt_send(data)
sndpkt = make_pkt(0, data, checksum)
udt_send(sndpkt)
start_timer

rdt_rcv(rcvpkt)
Λ

rdt_rcv(rcvpkt) &&
(corrupt(rcvpkt) ||
isACK(rcvpkt,1))
Λ

timeout
udt_send(sndpkt)
start_timer

rdt_rcv(rcvpkt)
&& notcorrupt(rcvpkt)
&& isACK(rcvpkt,0)
stop_timer

rdt_rcv(rcvpkt)
Λ

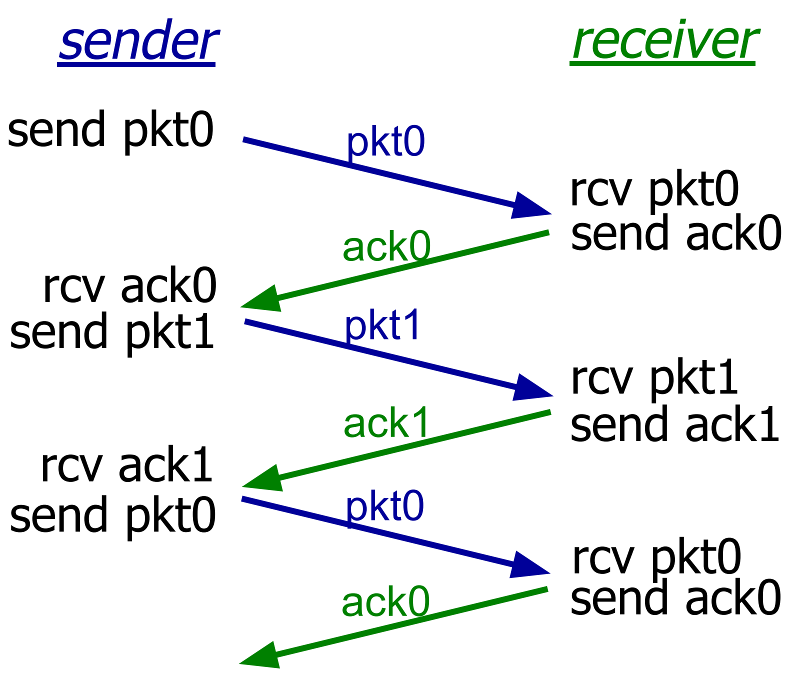
rdt_send(data)
sndpkt = make_pkt(1, data, checksum)
udt_send(sndpkt)
start_timer

rdt_rcv(rcvpkt) &&
(corrupt(rcvpkt) ||
isACK(rcvpkt,0))
Λ

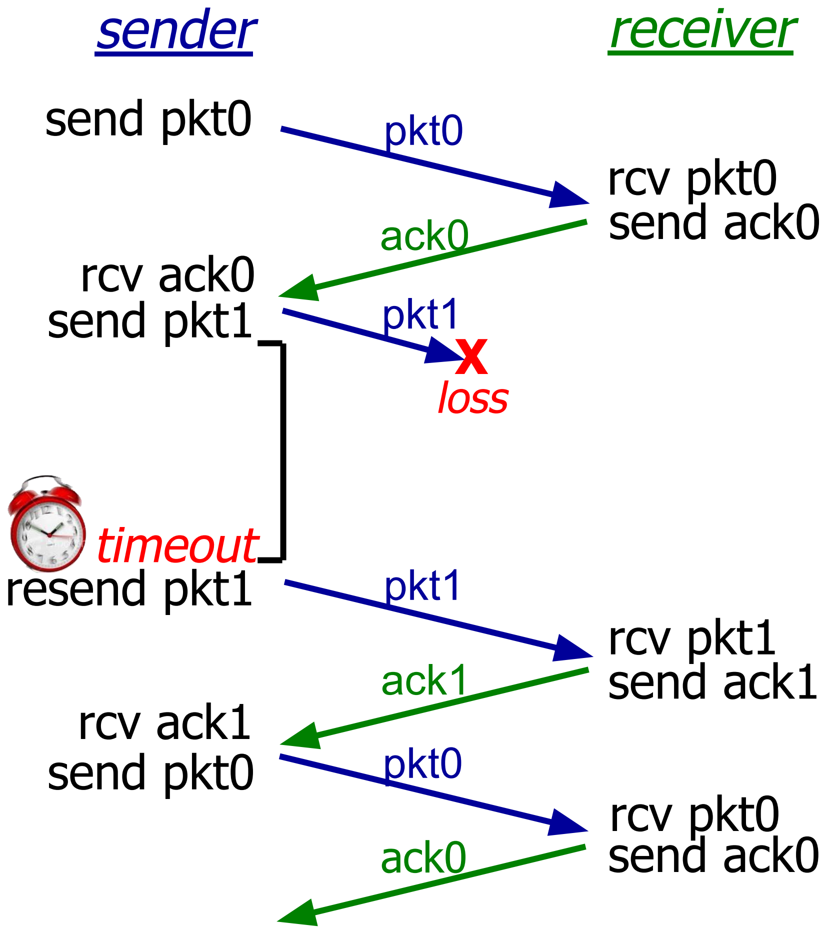
timeout
udt_send(sndpkt)
start_timer

rdt_rcv(rcvpkt)
&& notcorrupt(rcvpkt)
&& isACK(rcvpkt,1)
stop_timer

rdt3.0 in action

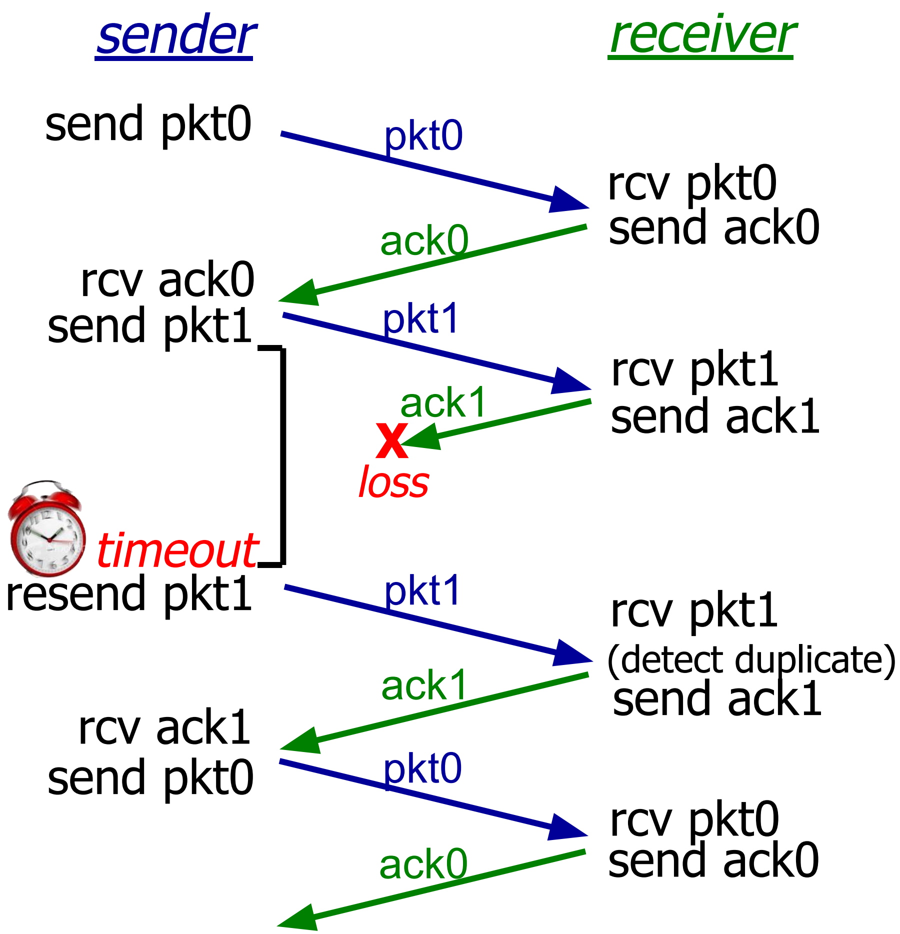


(a) no loss

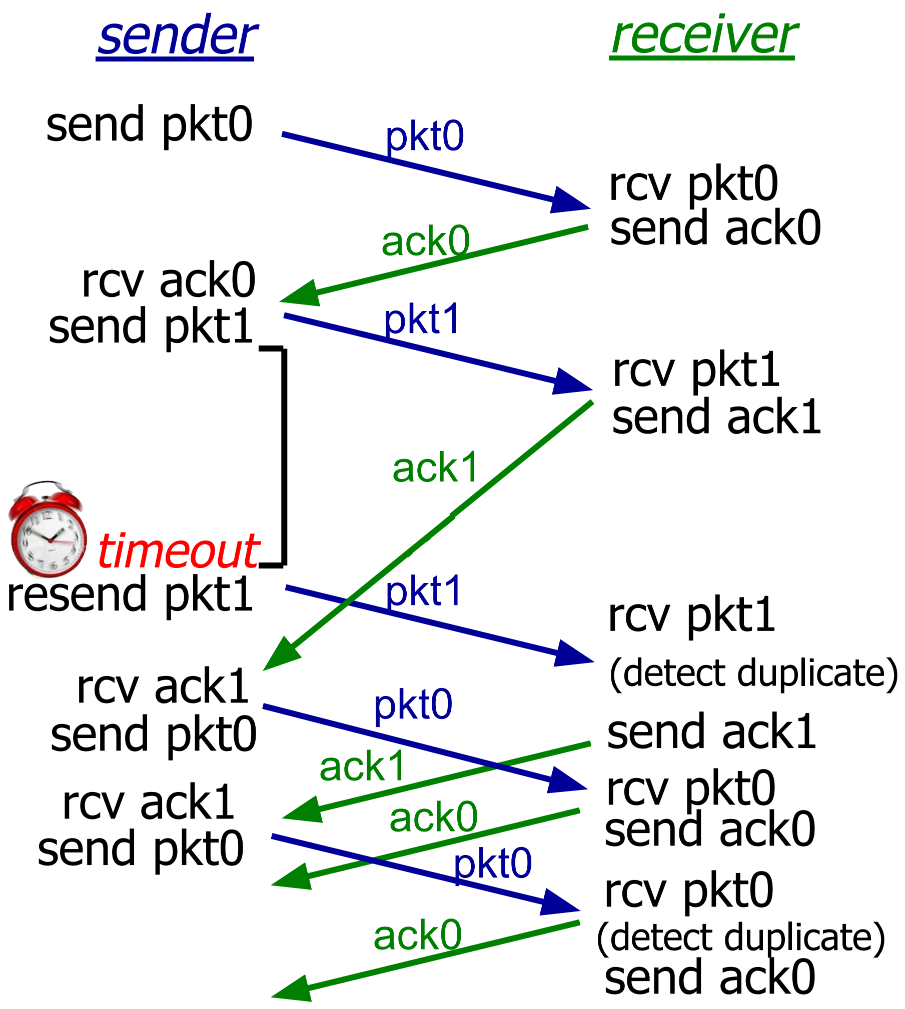


(b) packet loss

rdt3.0 in action



(c) ACK loss



(d) premature timeout/ delayed ACK

rdt3.0: channels with errors *and* loss

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

- checksum, sequence #s, ACKs, retransmissions will be of help … but not quite enough

Q: How do *humans* handle lost sender-to-receiver words in conversation?

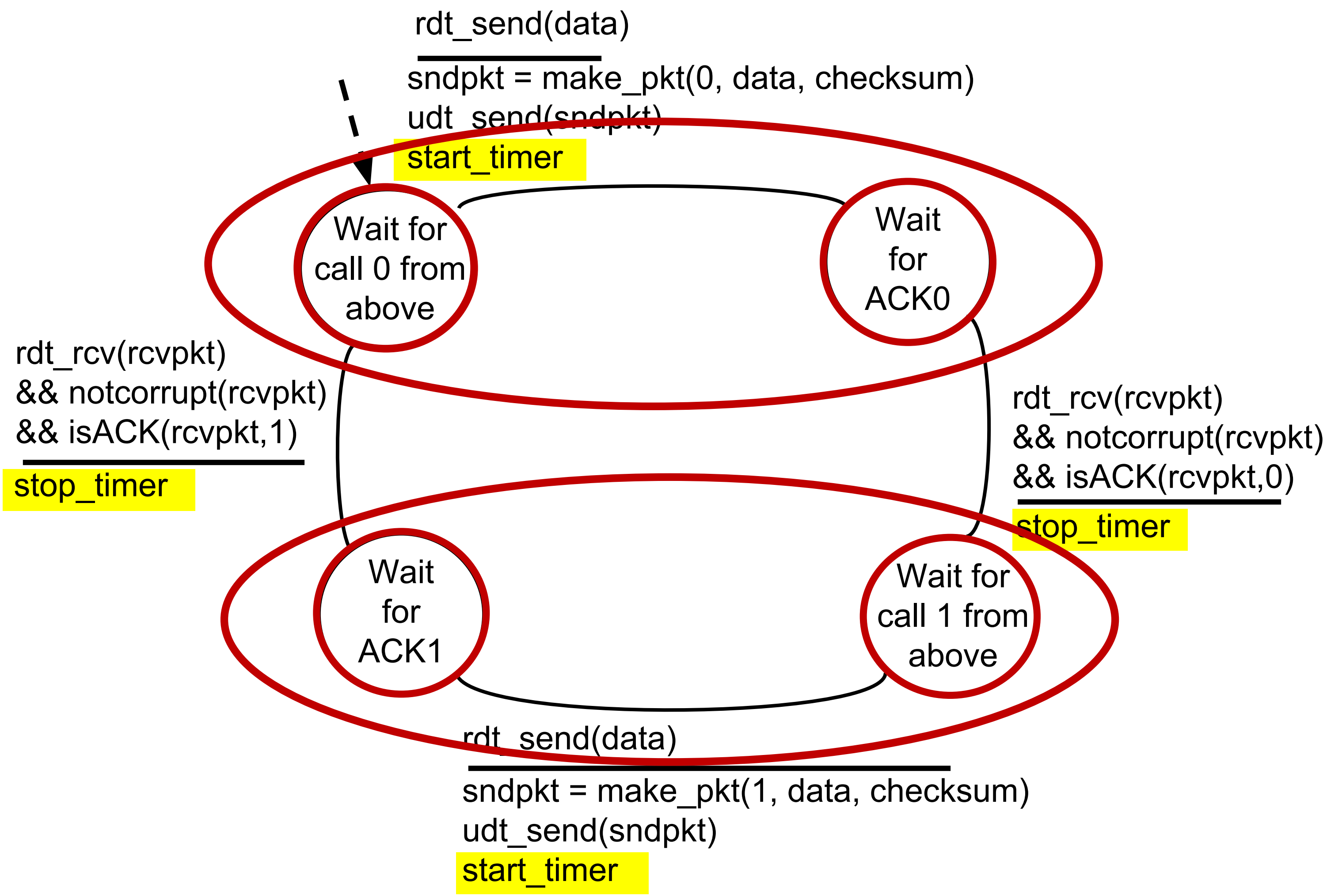
rdt3.0: channels with errors *and* loss

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

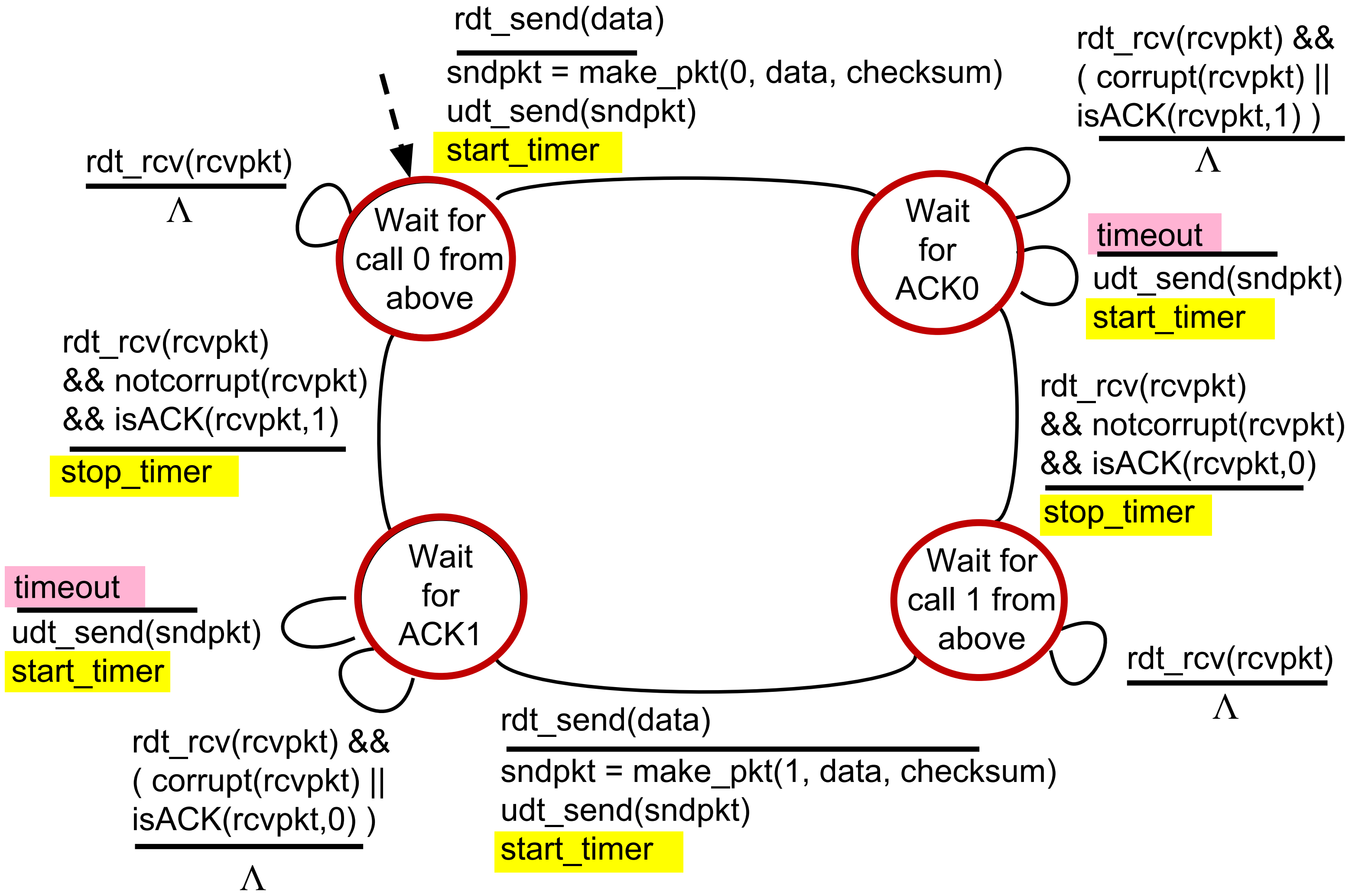
- retransmits if no ACK received in this time
- if pkt (or ACK) just delayed (not lost):
  - retransmission will be duplicate, but seq #s already handles this!
  - receiver must specify seq # of packet being ACKed
- use countdown timer to interrupt after "reasonable" amount of time

timeout

rdt3.0 sender



rdt3.0 sender



rdt_send(data)

sndpkt = make_pkt(0, data, checksum)
udt_send(sndpkt)
start_timer

rdt_rcv(rcvpkt) &&
(corrupt(rcvpkt) ||
isACK(rcvpkt,1))

Λ

rdt_rcv(rcvpkt)

Λ

timeout

udt_send(sndpkt)
start_timer

rdt_rcv(rcvpkt)
&& notcorrupt(rcvpkt)
&& isACK(rcvpkt,1)

stop_timer

rdt_rcv(rcvpkt)
&& notcorrupt(rcvpkt)
&& isACK(rcvpkt,0)

stop_timer

timeout

udt_send(sndpkt)
start_timer

rdt_rcv(rcvpkt)

Λ

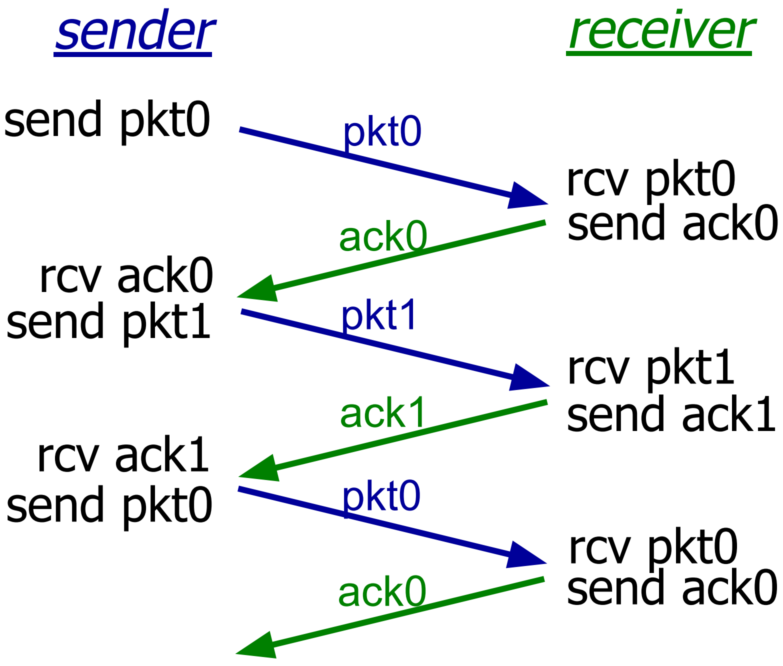
rdt_rcv(rcvpkt) &&
(corrupt(rcvpkt) ||
isACK(rcvpkt,0))

Λ

rdt_send(data)

sndpkt = make_pkt(1, data, checksum)
udt_send(sndpkt)
start_timer

rdt3.0 in action



sender

receiver

send pkt0

pkt0

rcv pkt0
send ack0

ack0

rcv ack0
send pkt1

pkt1

rcv pkt1
send ack1

ack1

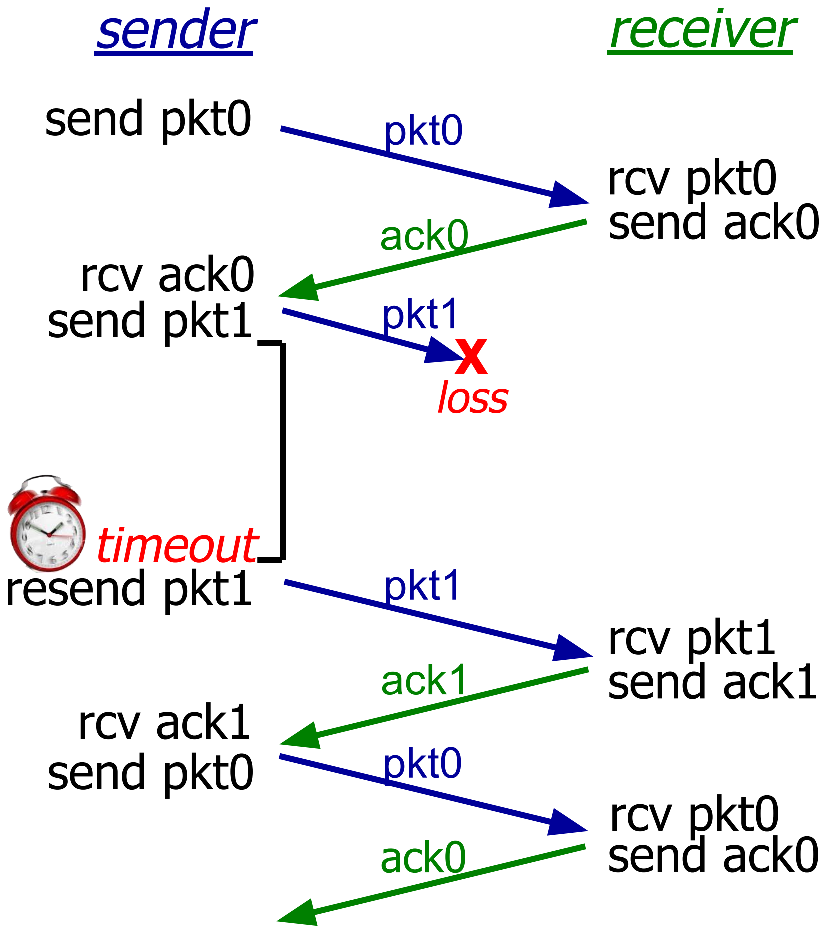
rcv ack1
send pkt0

pkt0

rcv pkt0
send ack0

ack0

(a) no loss



sender

receiver

send pkt0

pkt0

rcv pkt0
send ack0

ack0

rcv ack0
send pkt1

pkt1

X

loss

timeout

resend pkt1

pkt1

rcv pkt1
send ack1

ack1

rcv ack1
send pkt0

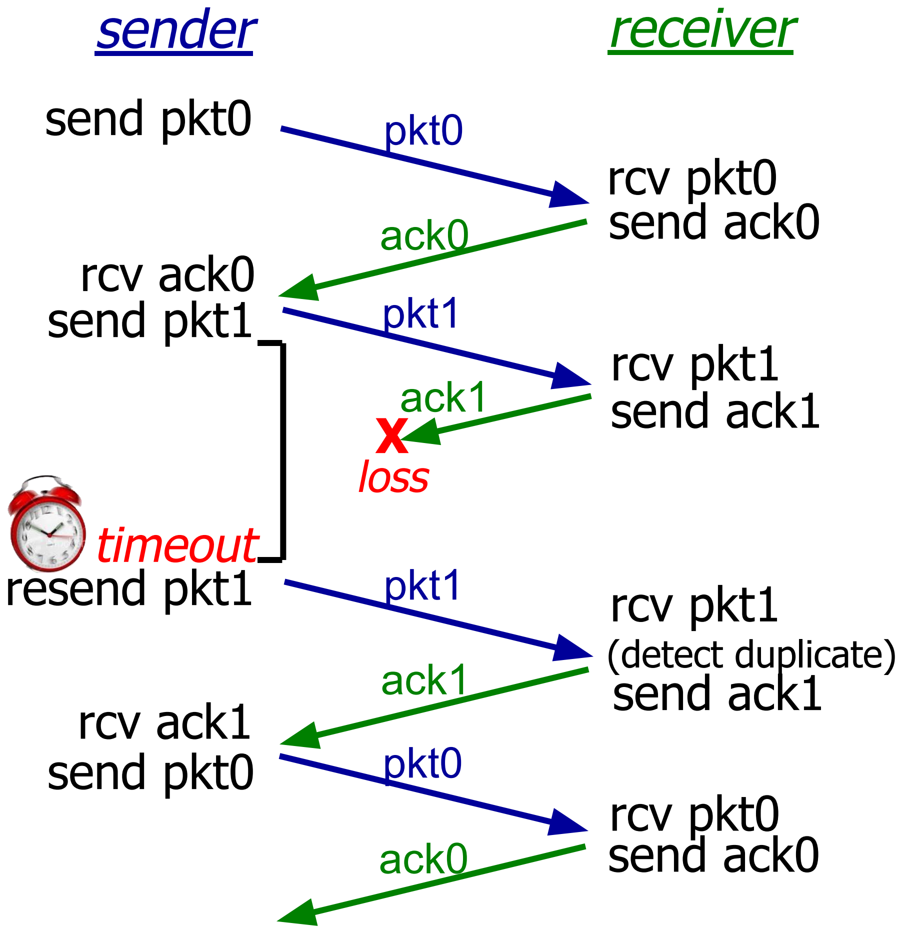
pkt0

rcv pkt0
send ack0

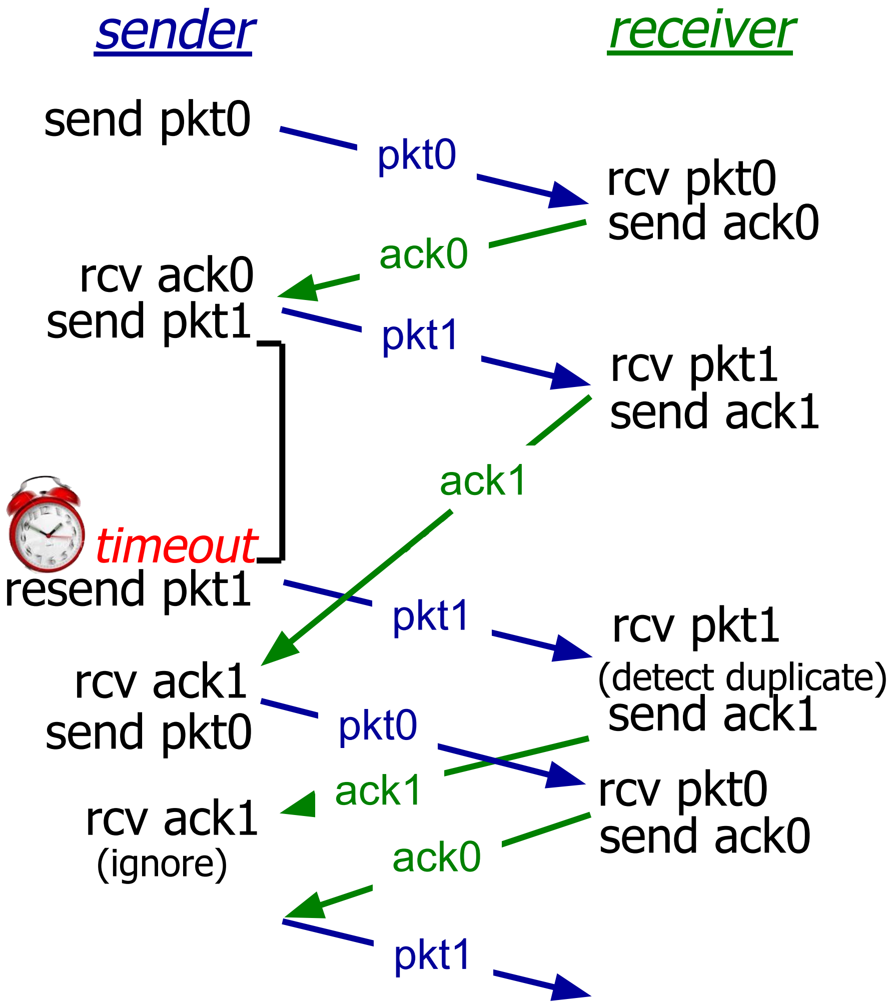
ack0

(b) packet loss

rdt3.0 in action



(c) ACK loss



(d) premature timeout/ delayed ACK

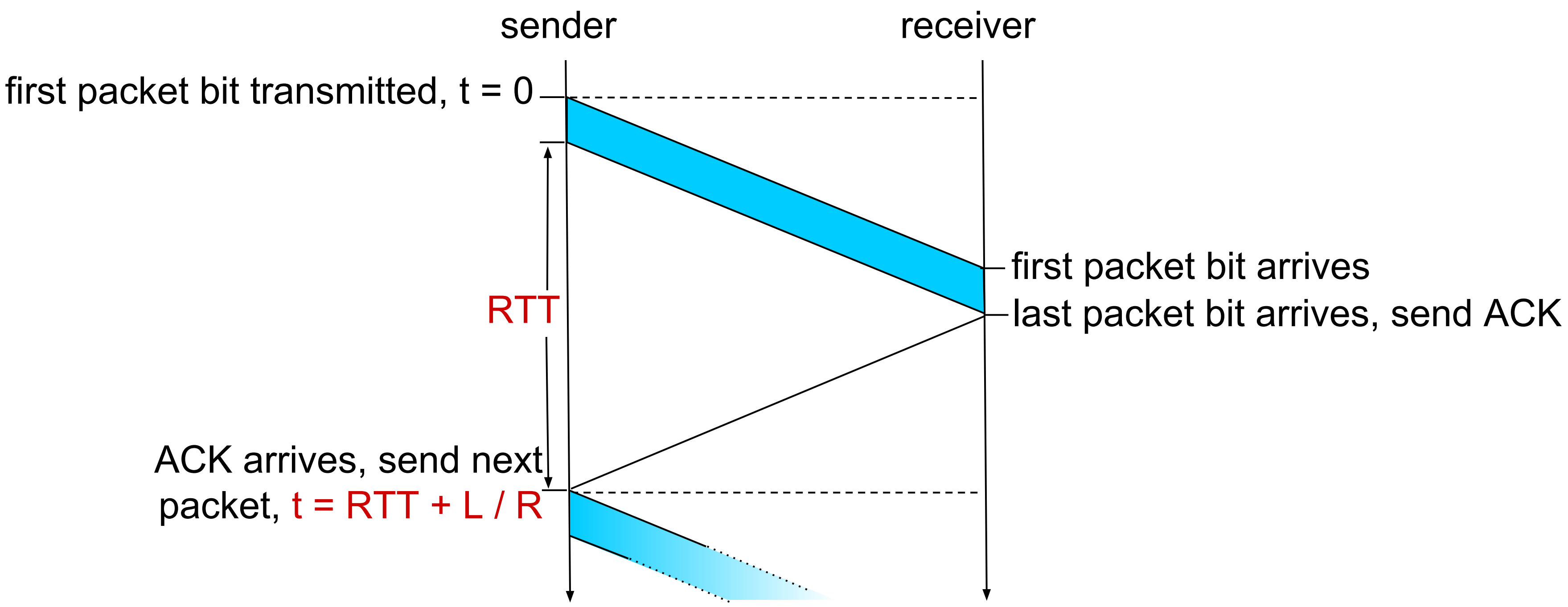
Performance of rdt3.0 (stop-and-wait)

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

- example: 1 Gbps link, 15 ms prop. delay, 8000 bit packet
  - time to transmit packet into channel:

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

rdt3.0: stop-and-wait operation

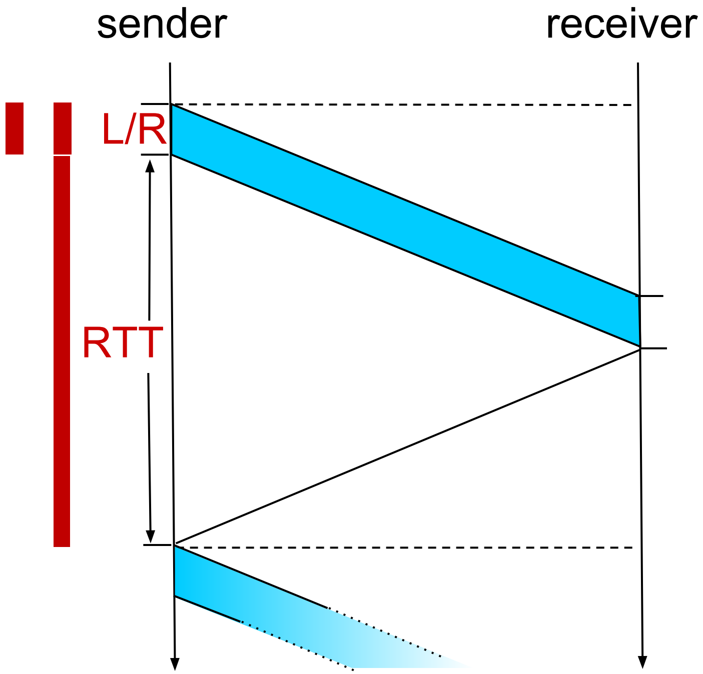

sender
receiver
first packet bit transmitted, t = 0
first packet bit arrives
last packet bit arrives, send ACK
RTT
ACK arrives, send next packet, t = RTT + L / R

rdt3.0: stop-and-wait operation

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

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

$$= 0.00027$$

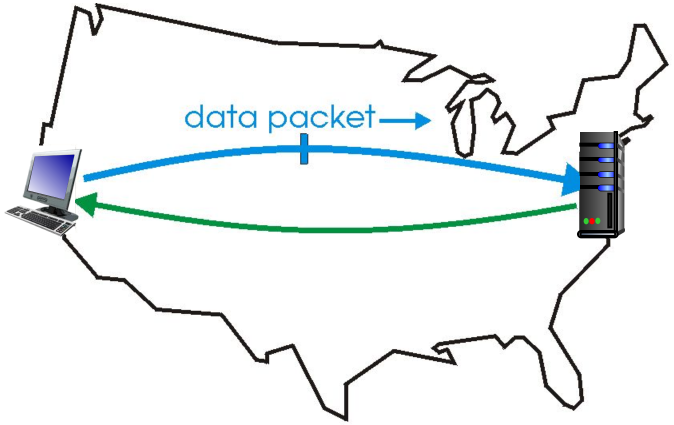


- rdt 3.0 protocol performance stinks!
- Protocol limits performance of underlying infrastructure (channel)

rdt3.0: pipelined protocols operation

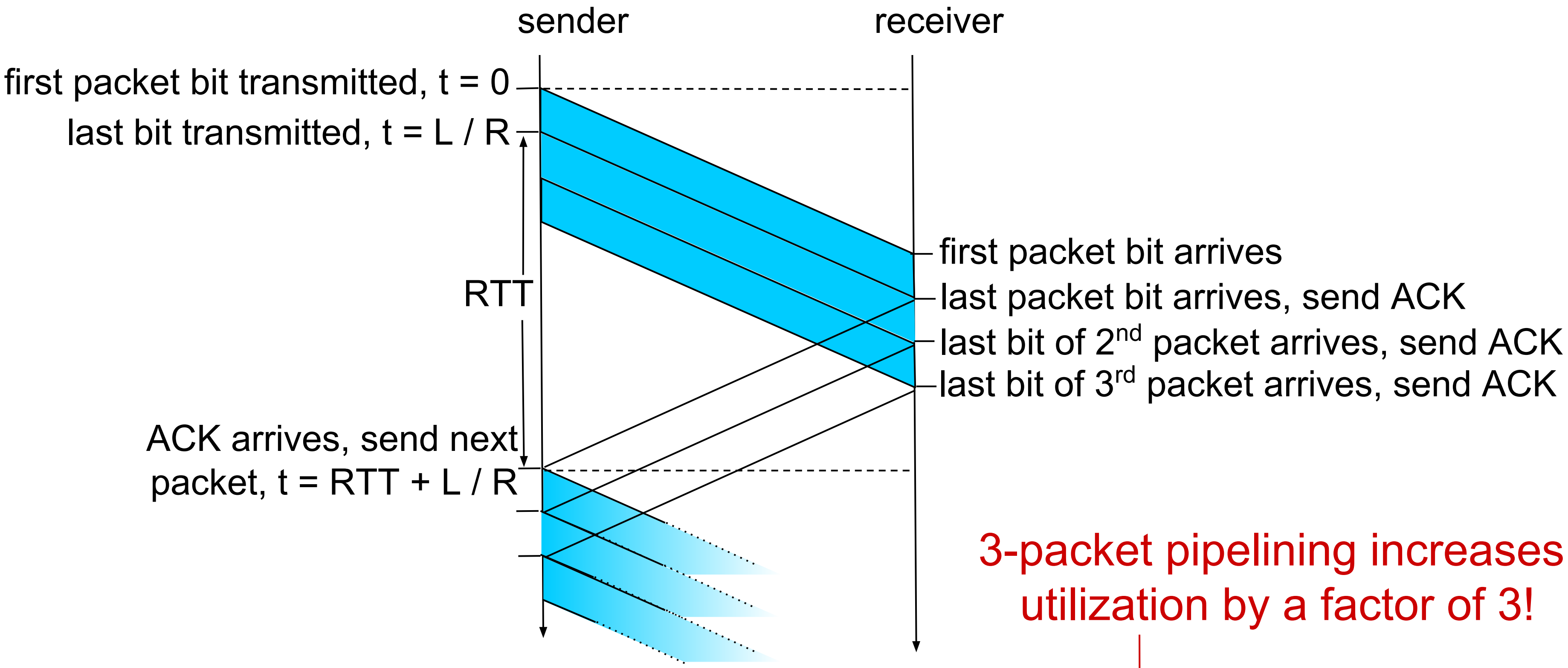
pipelining: sender allows multiple, "in-flight", yet-to-be-acknowledged packets

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



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

Pipelining: increased utilization

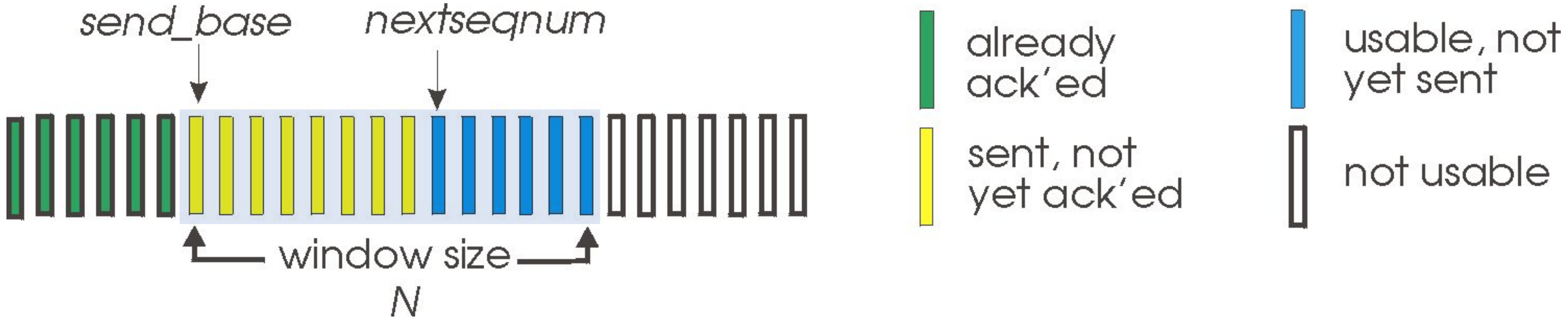

sender
receiver
first packet bit transmitted, t = 0
last bit transmitted, t = L / R
RTT
first packet bit arrives
last packet bit arrives, send ACK
last bit of 2nd packet arrives, send ACK
last bit of 3rd packet arrives, send ACK
ACK arrives, send next packet, t = RTT + L / R

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

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

Go-Back-N: sender

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

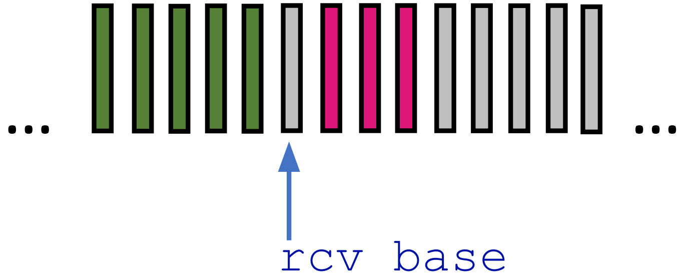


- *cumulative ACK:* ACK(*n*): ACKs all packets up to, including seq # *n*
  - on receiving ACK(*n*): move window forward to begin at *n+1*
- timer for oldest in-flight packet
- *timeout(n):* retransmit packet n and all higher seq # packets in window

Go-Back-N: receiver

- ACK-only: always send ACK for correctly-received packet so far, with highest *in-order* seq #
  - may generate duplicate ACKs
  - need only remember `rcv_base`
- on receipt of out-of-order packet:
  - can discard (don't buffer) or buffer: an implementation decision
  - re-ACK pkt with highest in-order seq #

Receiver view of sequence number space:

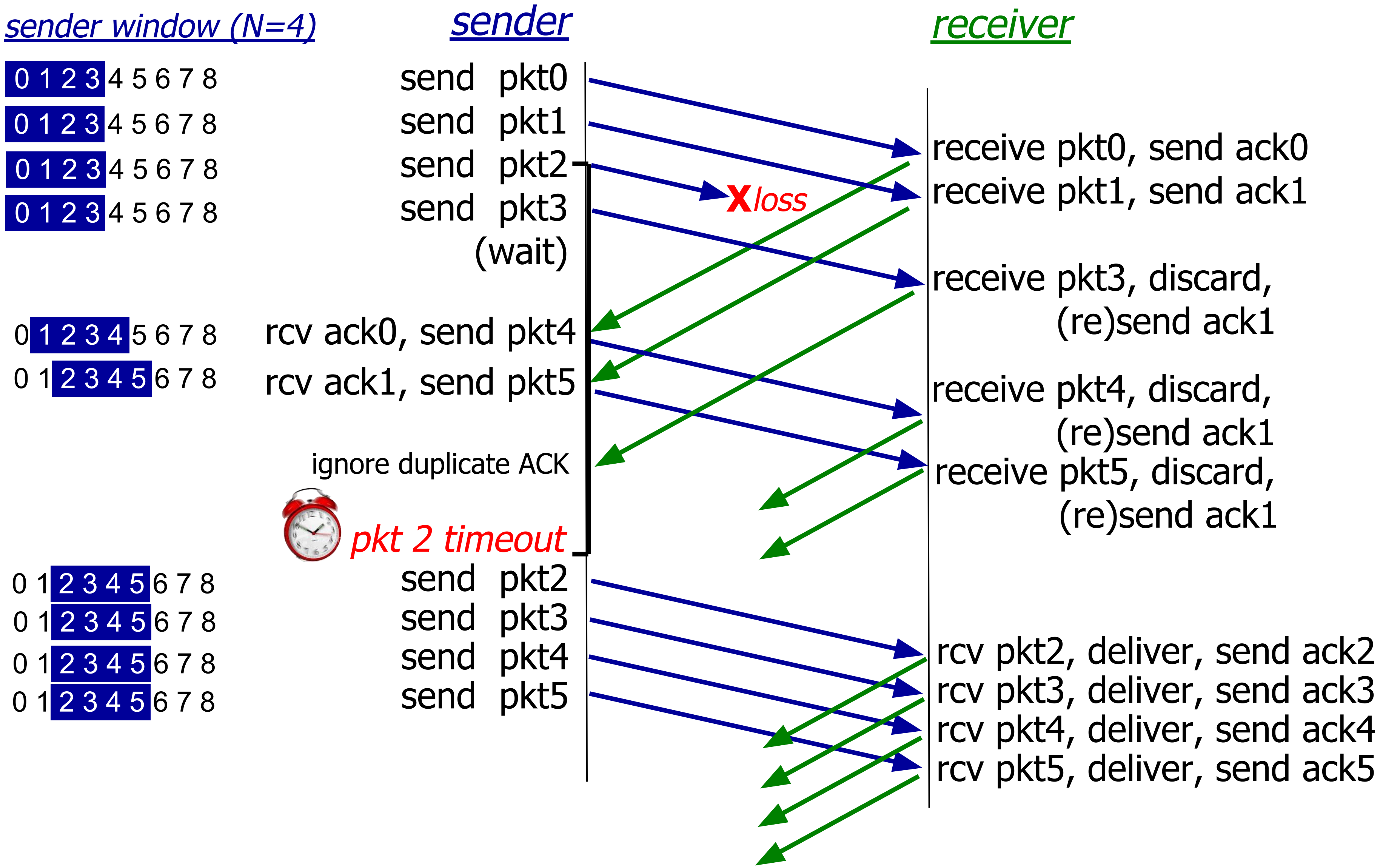


🟩 received and ACKed

🟥 Out-of-order: received but not ACKed

▨ Not received

Go-Back-N in action



sender window (N=4)

0 1 2 3 4 5 6 7 8

0 1 2 3 4 5 6 7 8

0 1 2 3 4 5 6 7 8

0 1 2 3 4 5 6 7 8

0 1 2 3 4 5 6 7 8

0 1 2 3 4 5 6 7 8

0 1 2 3 4 5 6 7 8

0 1 2 3 4 5 6 7 8

0 1 2 3 4 5 6 7 8

0 1 2 3 4 5 6 7 8

sender

send pkt0
send pkt1
send pkt2
send pkt3
(wait)

rcv ack0, send pkt4
rcv ack1, send pkt5

ignore duplicate ACK

pkt 2 timeout

send pkt2
send pkt3
send pkt4
send pkt5

receiver

receive pkt0, send ack0
receive pkt1, send ack1

X *loss*

receive pkt3, discard,
(re)send ack1

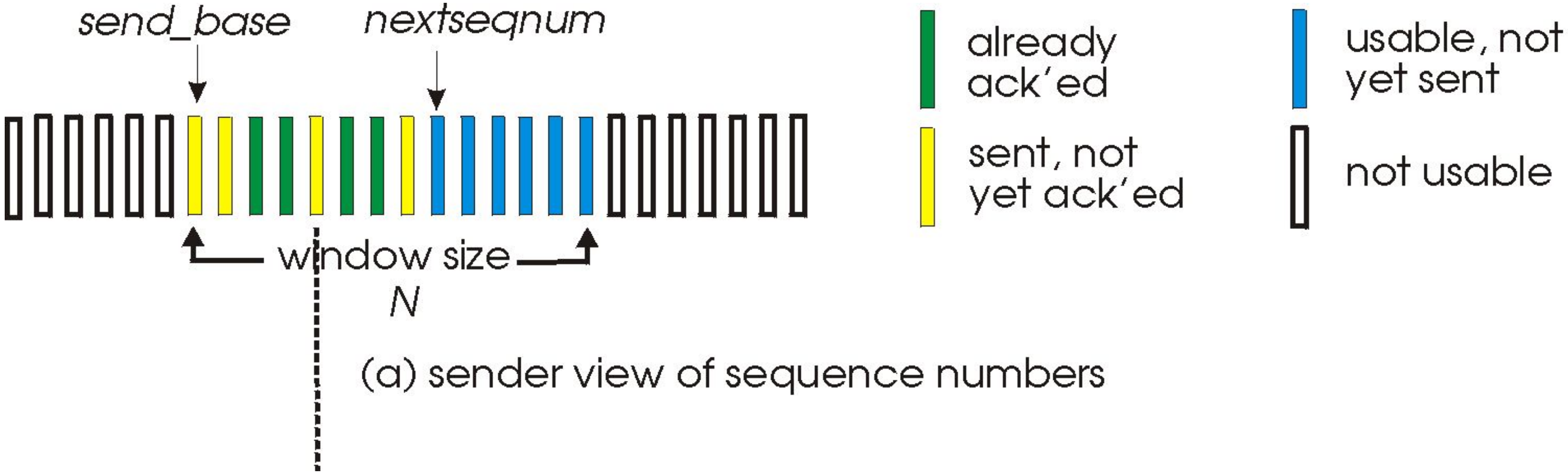
receive pkt4, discard,
(re)send ack1
receive pkt5, discard,
(re)send ack1

rcv pkt2, deliver, send ack2
rcv pkt3, deliver, send ack3
rcv pkt4, deliver, send ack4
rcv pkt5, deliver, send ack5

Selective repeat

- receiver *individually* acknowledges all correctly received packets
  - buffers packets, as needed, for eventual in-order delivery to upper layer
- sender times-out/retransmits individually for unACKed packets
  - sender maintains timer for each unACKed pkt
- sender window
  - *N* consecutive seq #s
  - limits seq #s of sent, unACKed packets

Selective repeat: sender, receiver windows



(a) sender view of sequence numbers

Selective repeat: sender and receiver

sender

data from above:

- if next available seq # in window, send packet

timeout(*n*):

- resend packet *n*, restart timer

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

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

receiver

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

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

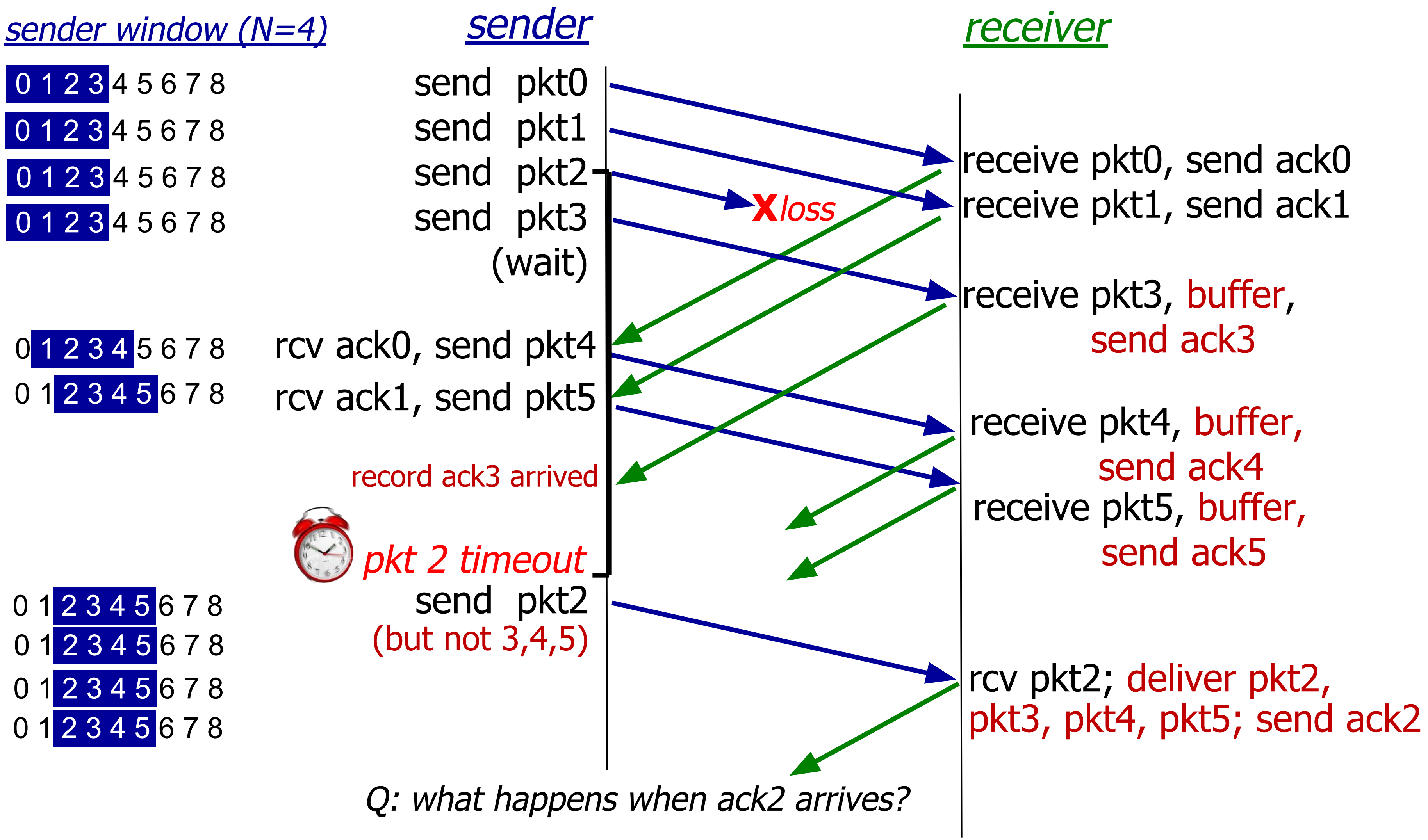
packet *n* in [rcvbase-N,rcvbase-1]

- ACK(*n*)

otherwise:

- ignore

Selective Repeat in action



sender window (N=4)

0 1 2 3 4 5 6 7 8
0 1 2 3 4 5 6 7 8
0 1 2 3 4 5 6 7 8
0 1 2 3 4 5 6 7 8

0 1 2 3 4 5 6 7 8
0 1 2 3 4 5 6 7 8

0 1 2 3 4 5 6 7 8
0 1 2 3 4 5 6 7 8
0 1 2 3 4 5 6 7 8
0 1 2 3 4 5 6 7 8

sender

send pkt0
send pkt1
send pkt2
send pkt3
(wait)

rcv ack0, send pkt4
rcv ack1, send pkt5

record ack3 arrived

pkt 2 timeout
send pkt2
(but not 3,4,5)

receiver

receive pkt0, send ack0
receive pkt1, send ack1

receive pkt3, buffer,
send ack3

receive pkt4, buffer,
send ack4
receive pkt5, buffer,
send ack5

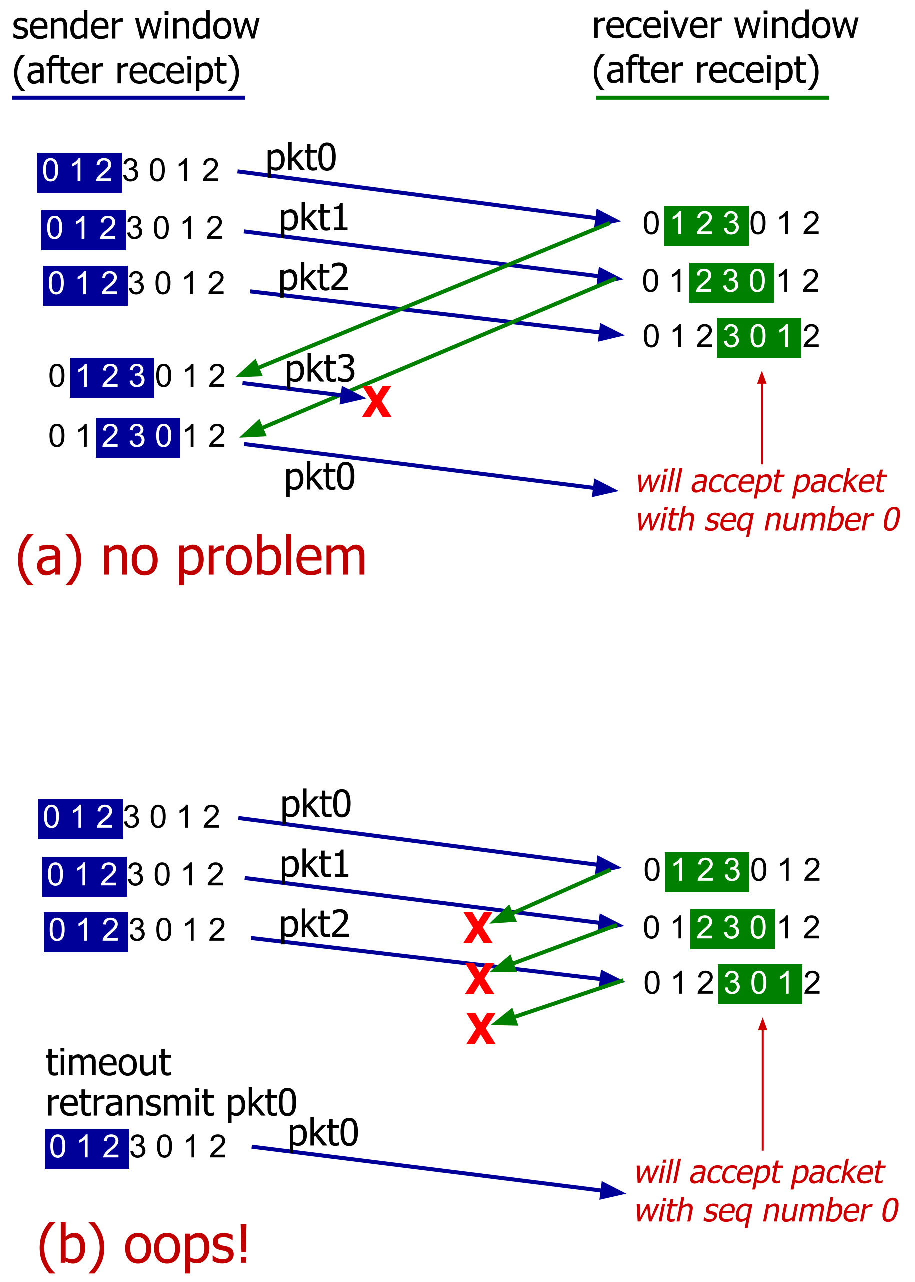
rcv pkt2; deliver pkt2,
pkt3, pkt4, pkt5; send ack2

Q: what happens when ack2 arrives?

Selective repeat: a dilemma!

example:

- seq #s: 0, 1, 2, 3 (base 4 counting)
- window size=3

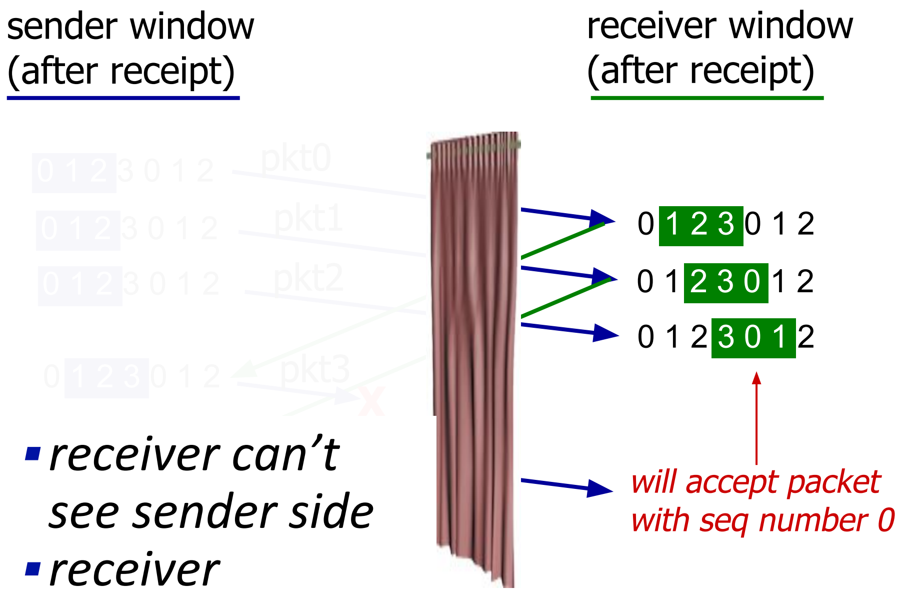


sender window (after receipt)
receiver window (after receipt)
0 1 2 3 0 1 2
pkt0
pkt1
pkt2
pkt3
X
pkt0
will accept packet with seq number 0
(a) no problem
timeout
retransmit pkt0
will accept packet with seq number 0
(b) oops!

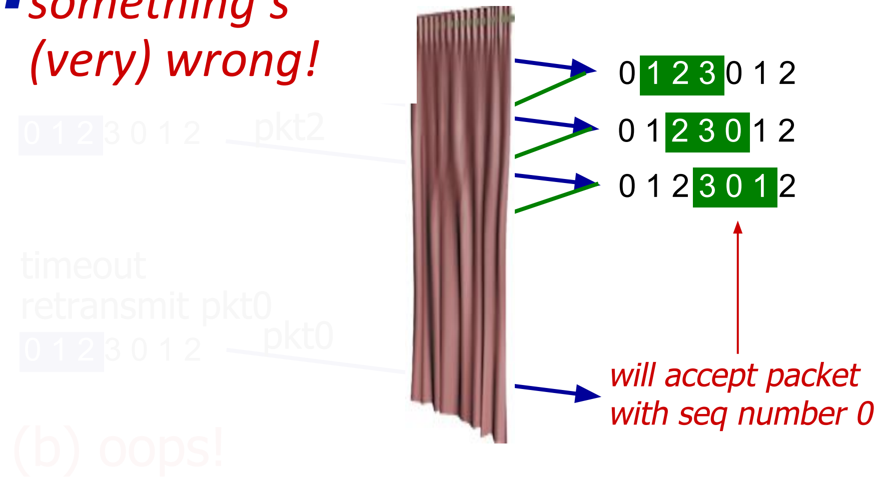
Selective repeat: a dilemma!

example:

- seq #s: 0, 1, 2, 3 (base 4 counting)
- window size=3



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



Q: what relationship is needed between sequence # size and window size to avoid problem in scenario (b)?