

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

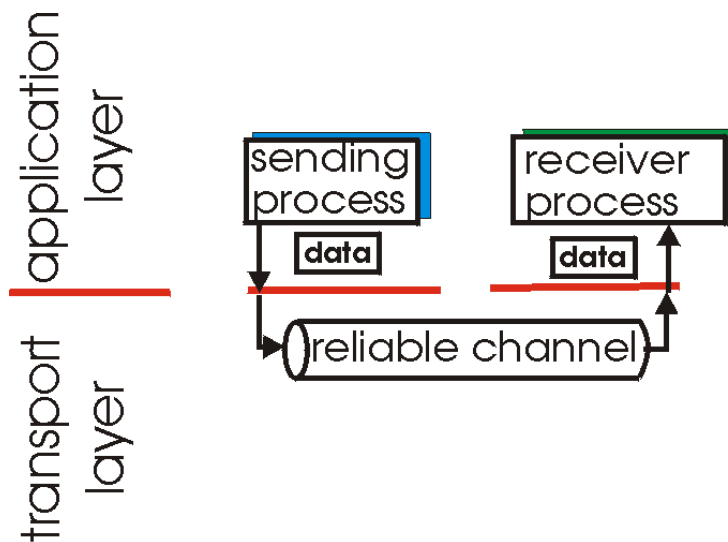
- segment structure
- reliable data transfer
- flow control
- connection management

3.6 principles of congestion control

3.7 TCP congestion control

Principles of reliable data transfer

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

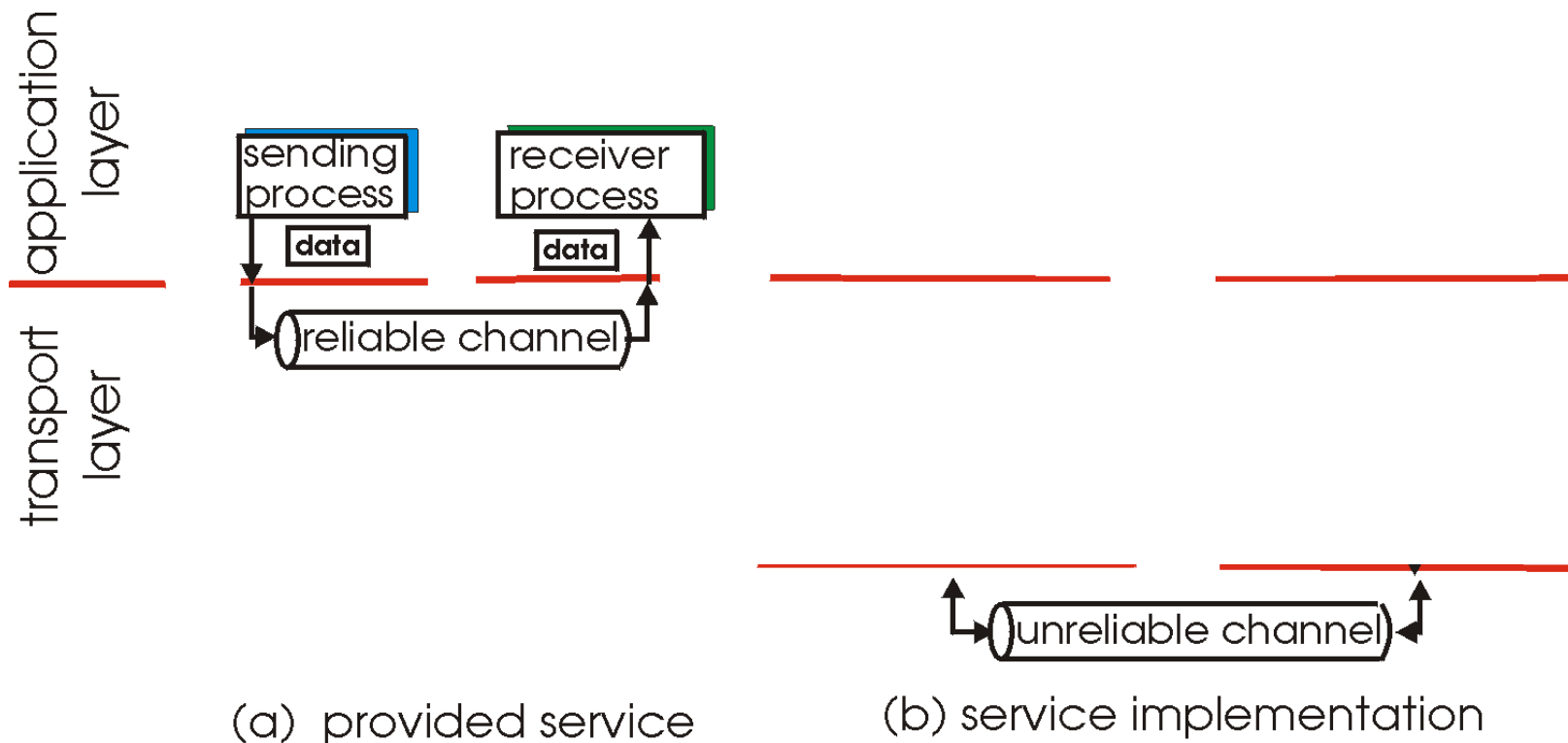


(a) provided service

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

Principles of reliable data transfer

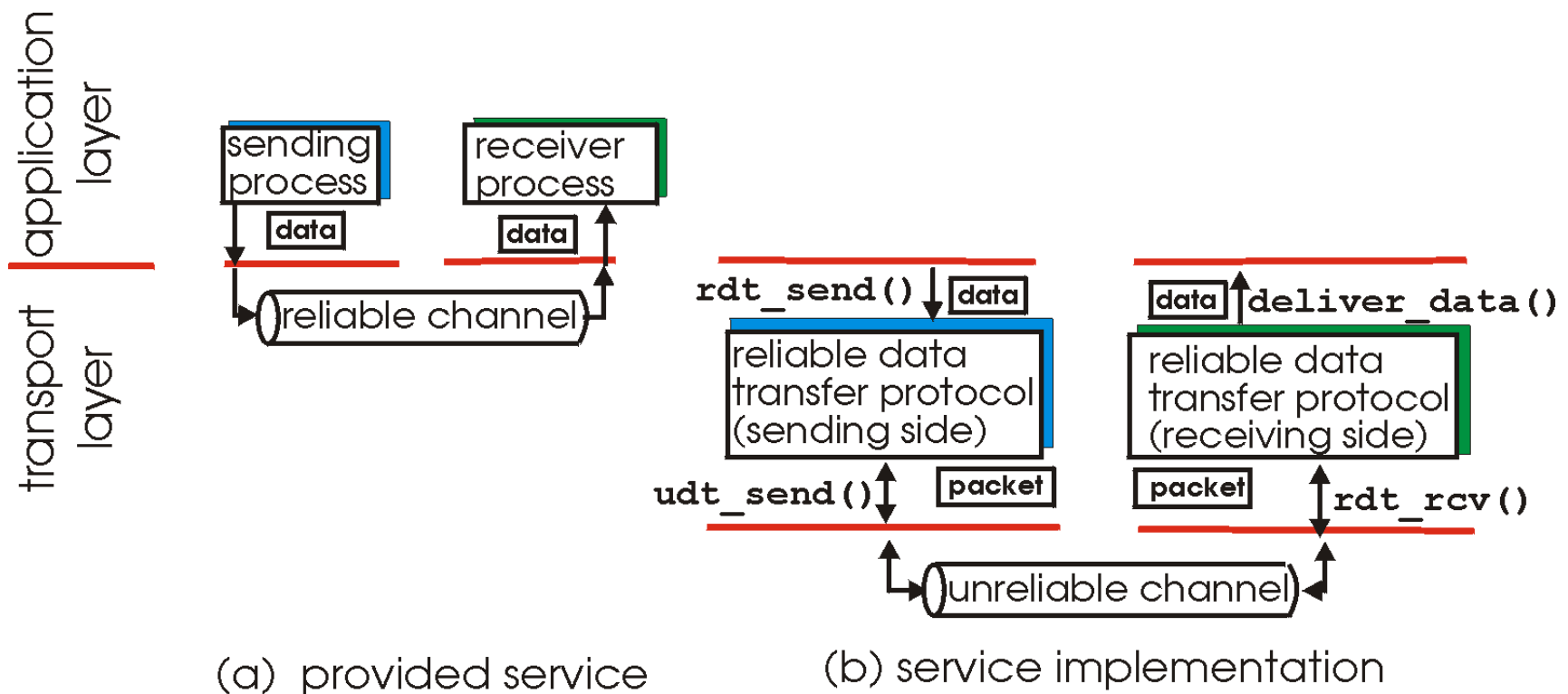
- important in application, transport, link layers
 - top-10 list of important networking topics!



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Principles of reliable data transfer

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

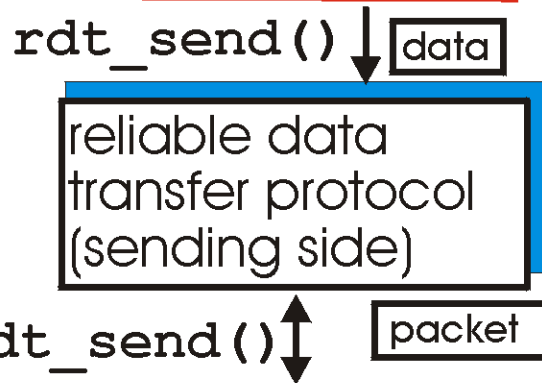


- characteristics of unreliable channel will determine complexity of reliable data transfer protocol (rdt)

Reliable data transfer: getting started

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

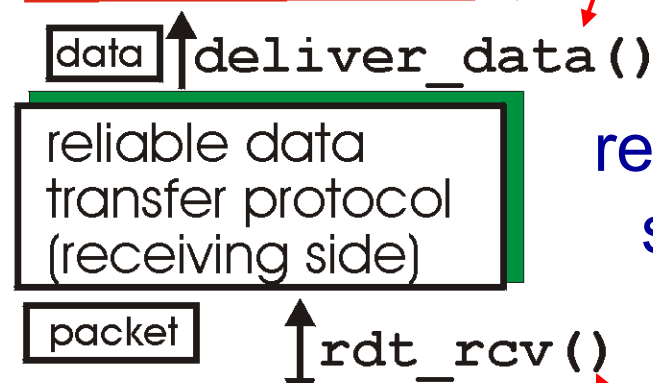
send
side



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

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

receive
side

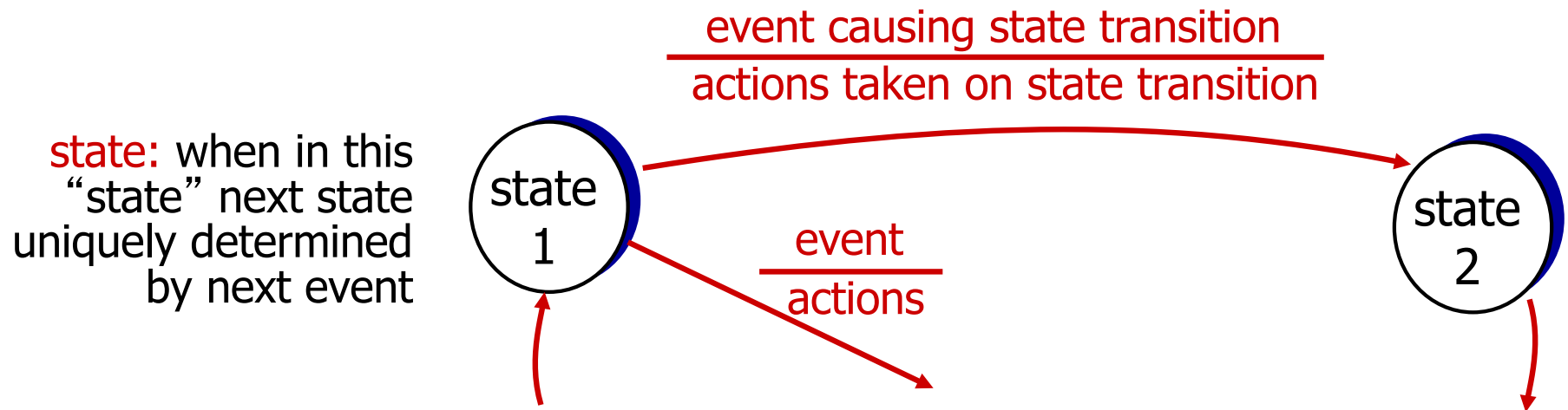


rdt_rcv() : called when packet
arrives on rcv-side of channel

Reliable data transfer: getting started

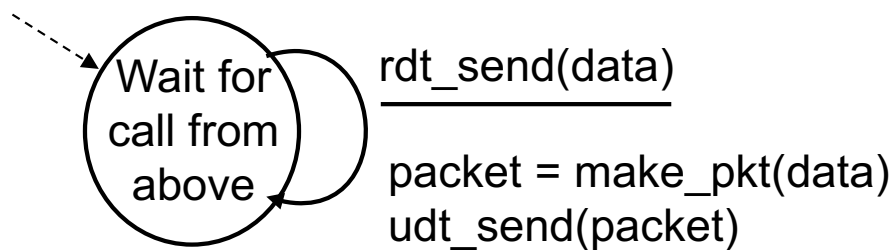
We will:

- 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

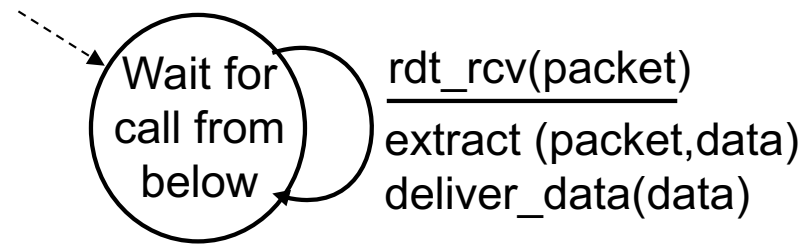


rdt1.0: reliable transfer over a reliable channel

- 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
- clearly RDT is useless here 😊



sender

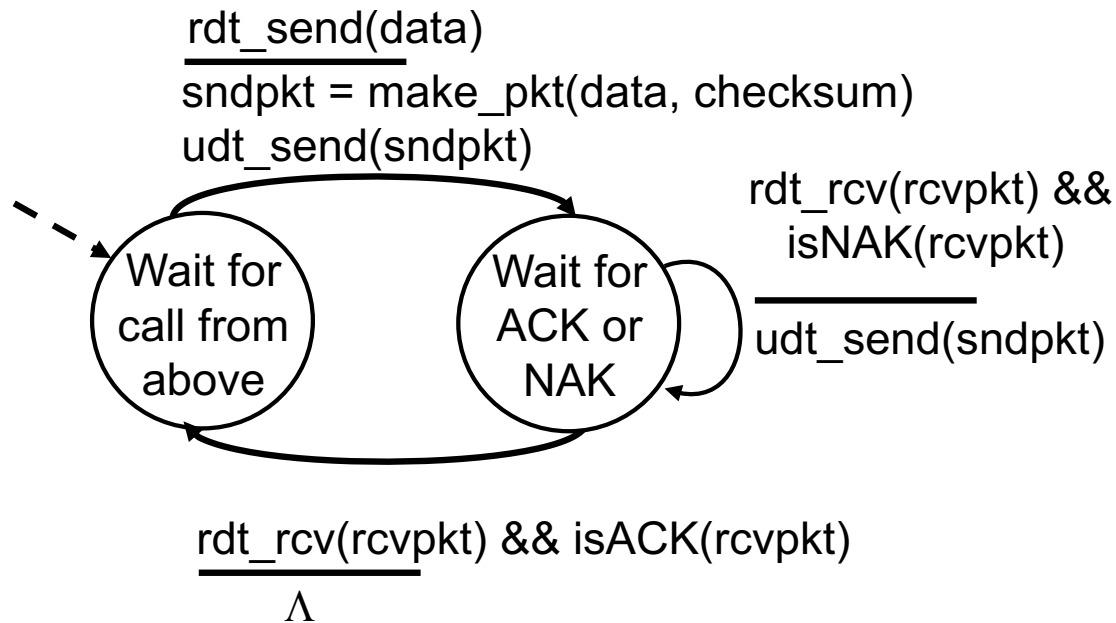


receiver

rdt2.0: channel with bit errors

- underlying channel may flip bits in packet
 - *error detection*: e.g., user checksum to detect bit errors
- Q: how to recover from errors?
- A: new mechanisms needed:
 - *feedback (control messages from receiver to sender)*
 - *acknowledgements (ACKs)*: receiver explicitly tells sender that pkt received OK
 - *negative acknowledgements (NAKs)*: receiver explicitly tells sender that pkt had errors
 - *retransmission*
 - sender retransmits pkt on receipt of NAK

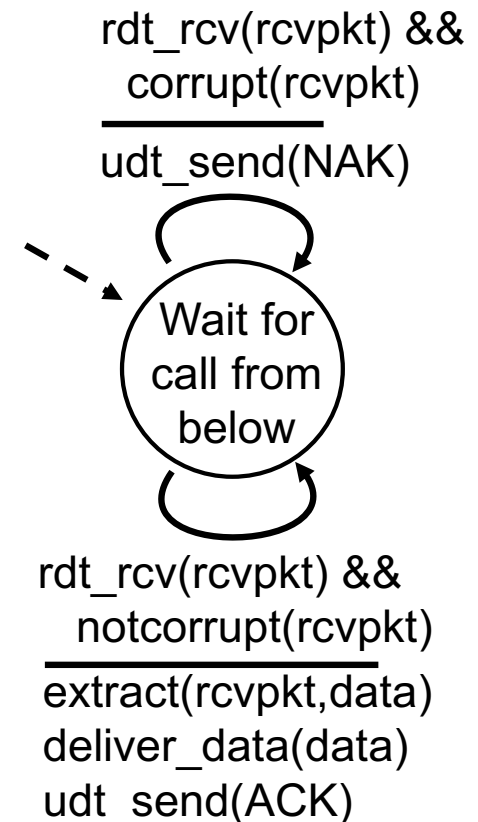
rdt2.0: FSM specification



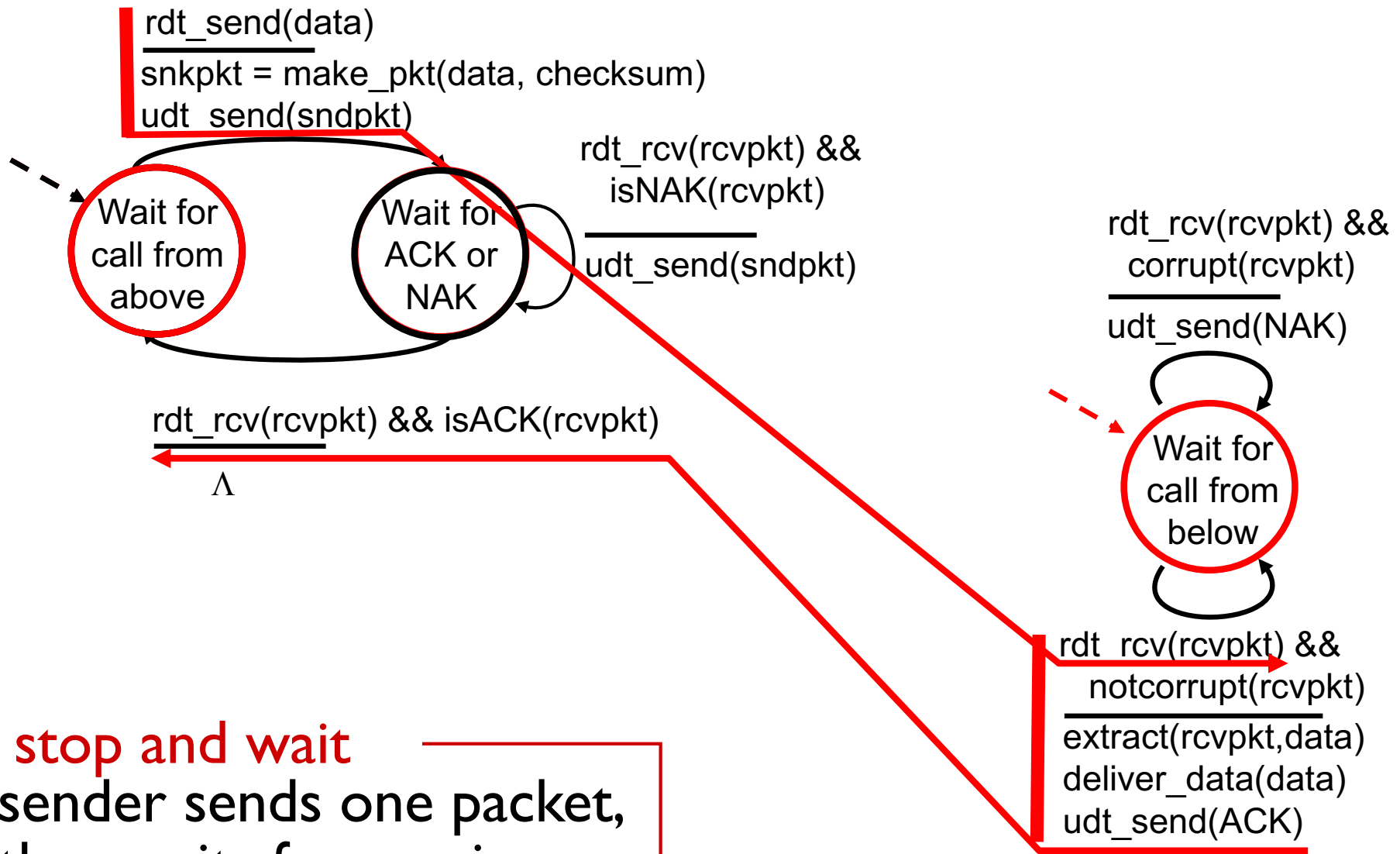
sender

stop and wait
sender sends one packet,
then waits for receiver
response

receiver

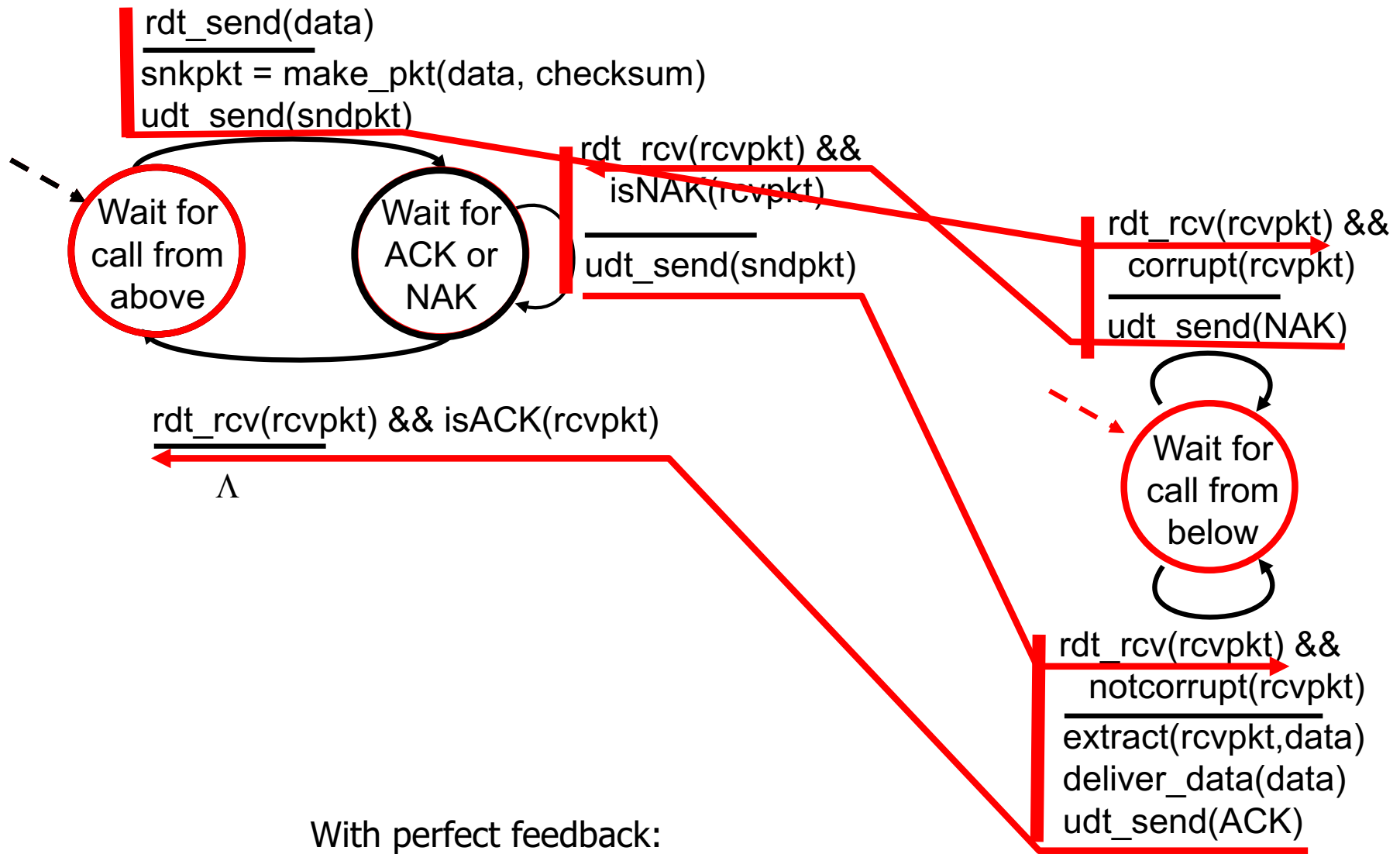


rdt2.0: operation with no errors



stop and wait
sender sends one packet,
then waits for receiver
response

rdt2.0: error scenario



With perfect feedback:
sender's state and receiver's state are in perfect sync

rdt2.0 has a flaw

what happens if ACK/NAK corrupted?

- sender doesn't know what happened at receiver!
 - Is packet received or corrupted?
 - should we retransmit the same (may cause duplicates) or send the next (missing one)
- solutions
 - Feedback on feedback...
 - may be corrupted but we can detect that (e.g. through checksum).
 - Error detection+correction
 - Cannot handle lost acks
 - Retransmissions
 - cant just retransmit: possible duplicates

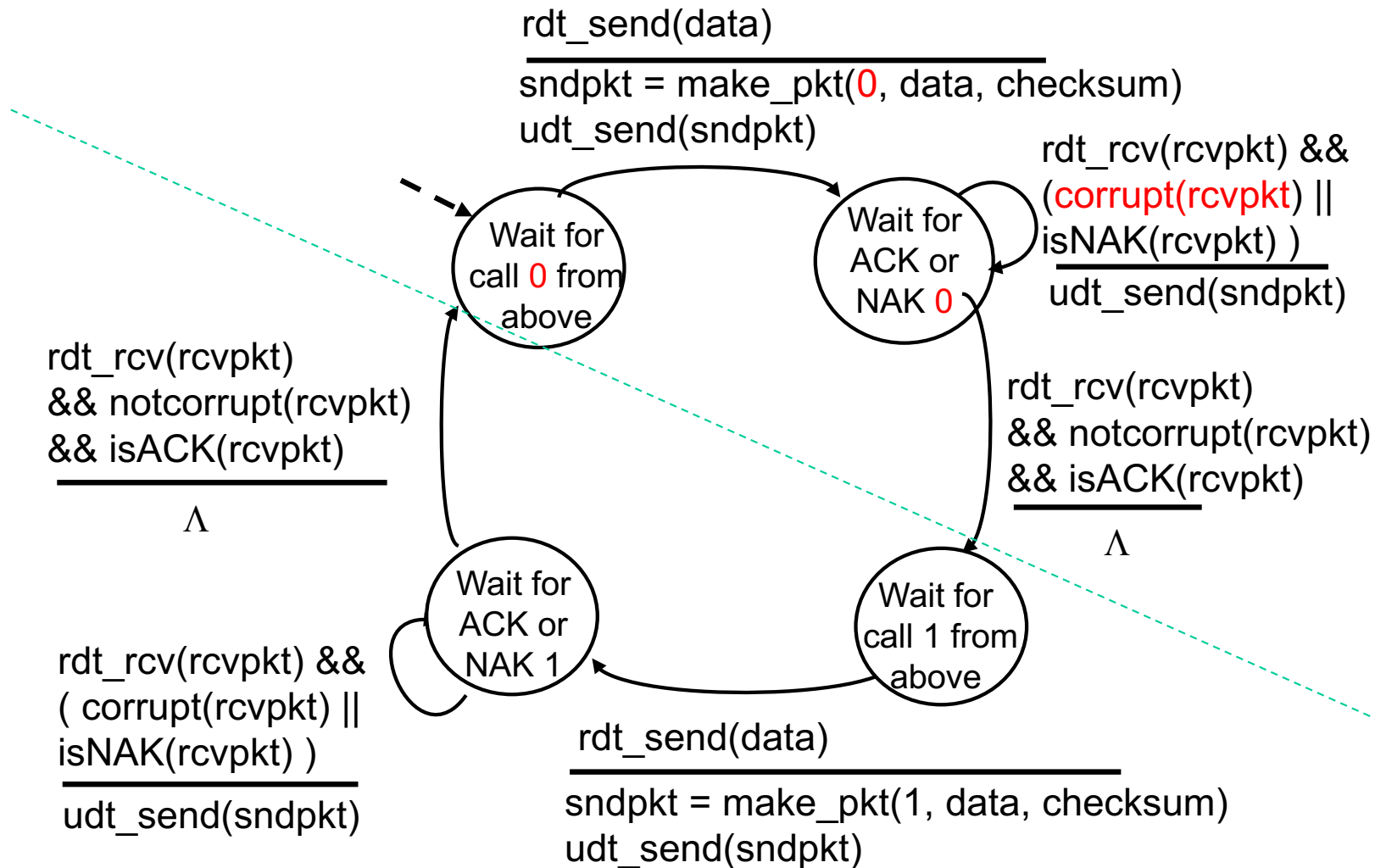
handling duplicates:

- sender **conservatively** retransmits current pkt if ACK/NAK corrupted
- sender adds **sequence number** to each pkt
- receiver discards (doesn't deliver up to app) duplicate pkt
- **but receiver always ACKs/NAKs received packet (to help the server move on to next seqno)**

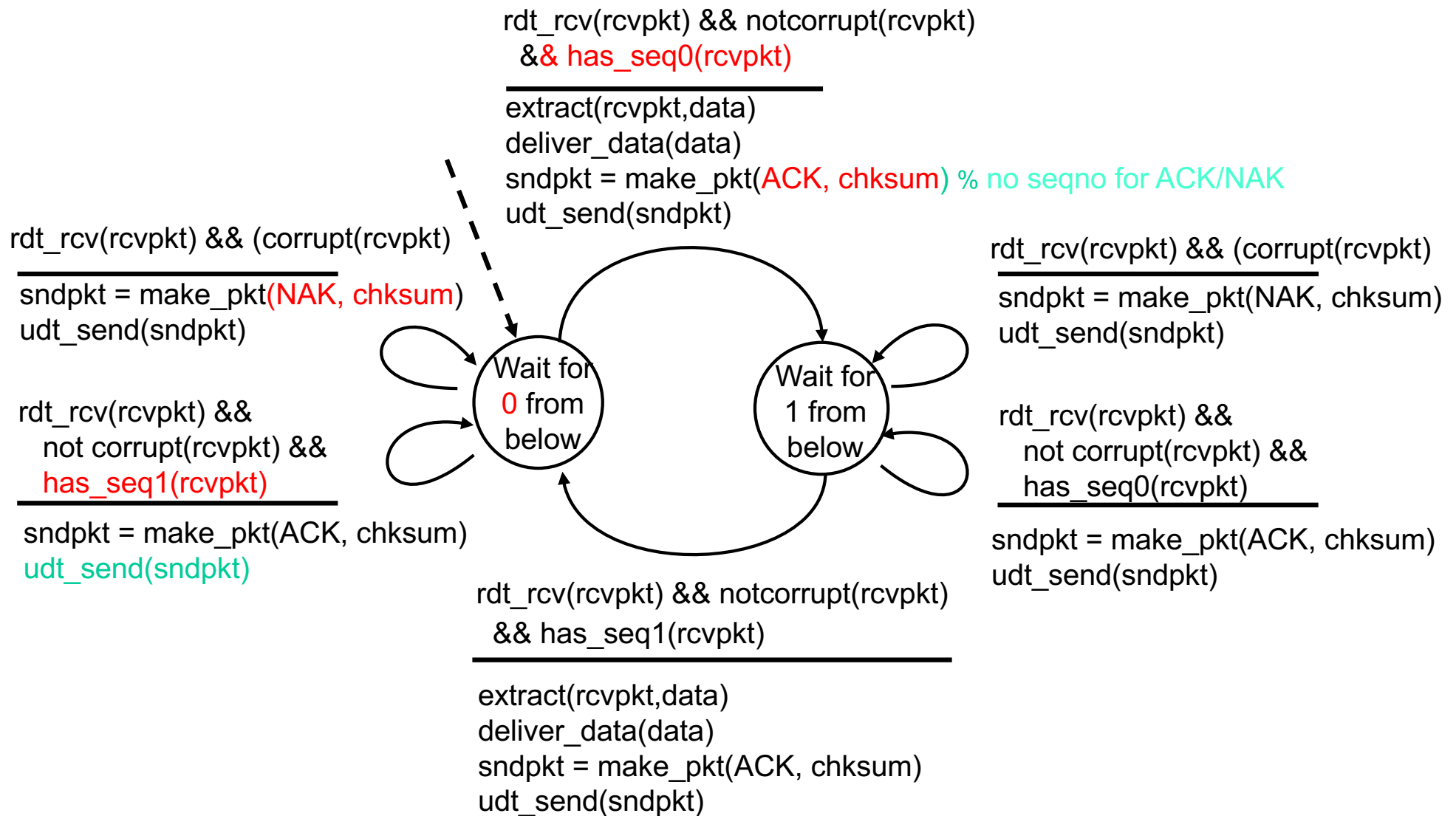
stop and wait

Sender **still** sends one packet, then waits for receiver response. **But** now both Sender and Receiver have double the number of states

rdt2.1: sender, handles garbled ACK/NAKs



rdt2.1: receiver, handles garbled ACK/NAKs



rdt2.1: discussion

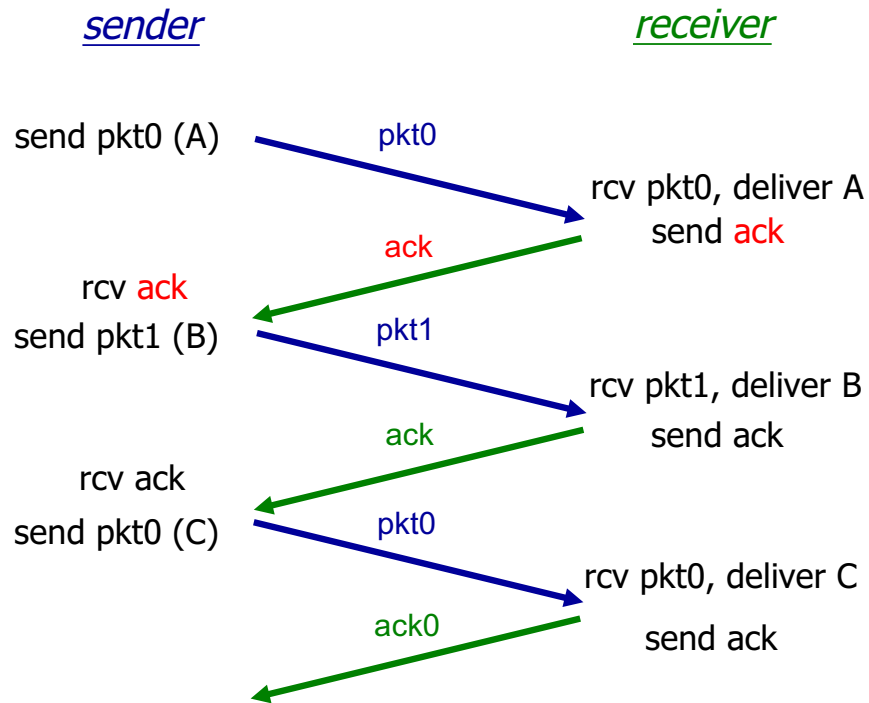
sender:

- must check if received ACK/NAK corrupted
- seq # added to pkt
 - two seq. #'s (0,1) will suffice for stop&wait
 - state indicates whether sender retransmits previous or transmits new pkt
 - twice as many states

receiver:

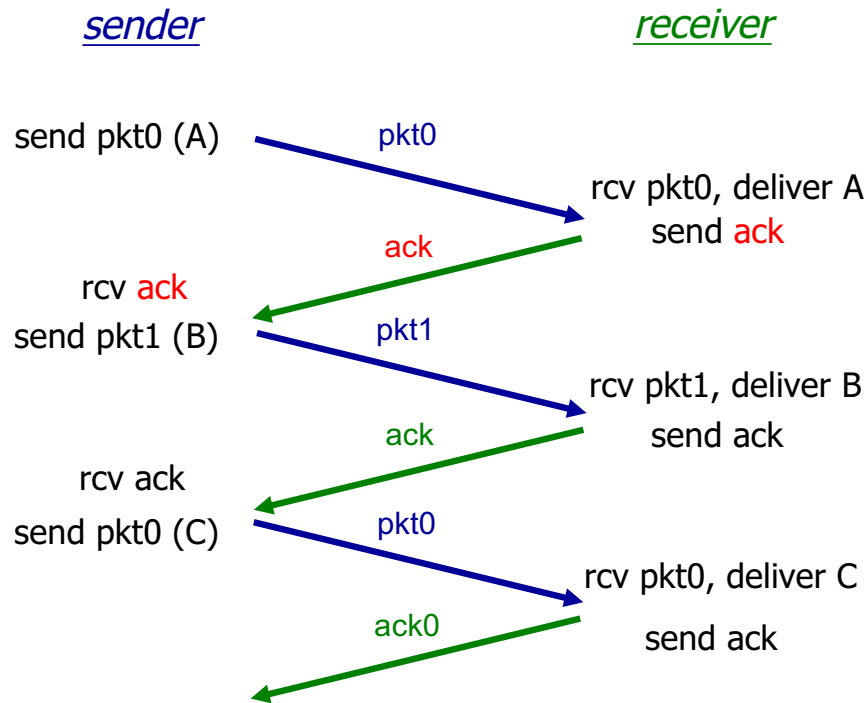
- must check if received packet is duplicate
 - state indicates whether 0 or 1 is expected pkt seq #
 - note: receiver can *not* know if its last ACK/NAK received OK at sender
 - If duplicate is received: do not pass up, but send ACK/NAK to help sender move on
- ACKs/NAKs don't need seq# for channel that doesn't drop packets
 - Always refer to most recently sent pkt
 - Seqnos necessary in ACK-only protocols and when channel drops packets

Rdt2.1 in action

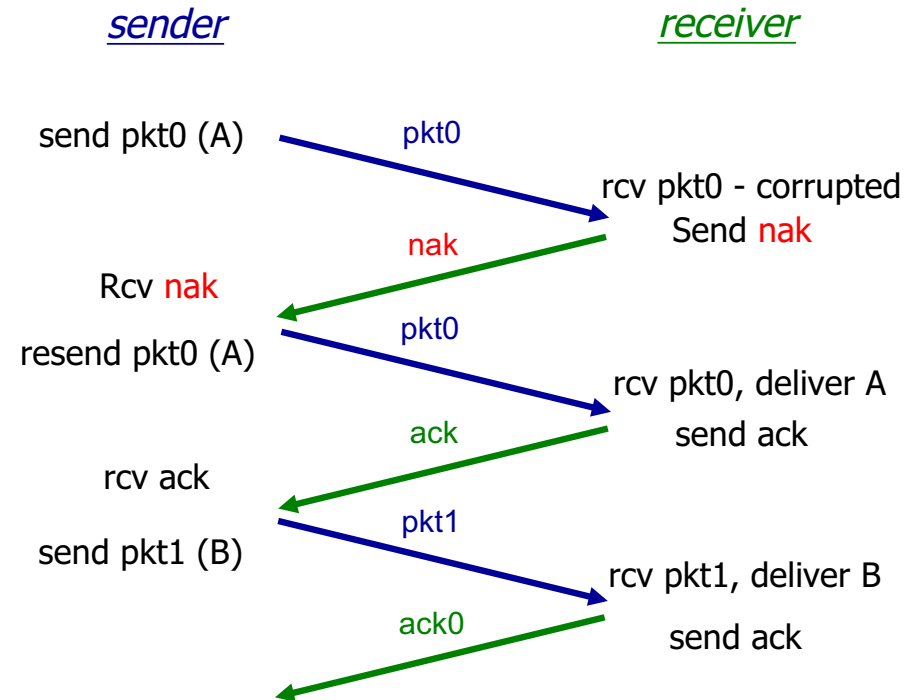


(a) No message or ACK corrupted

Rdt2.1 in action - continued

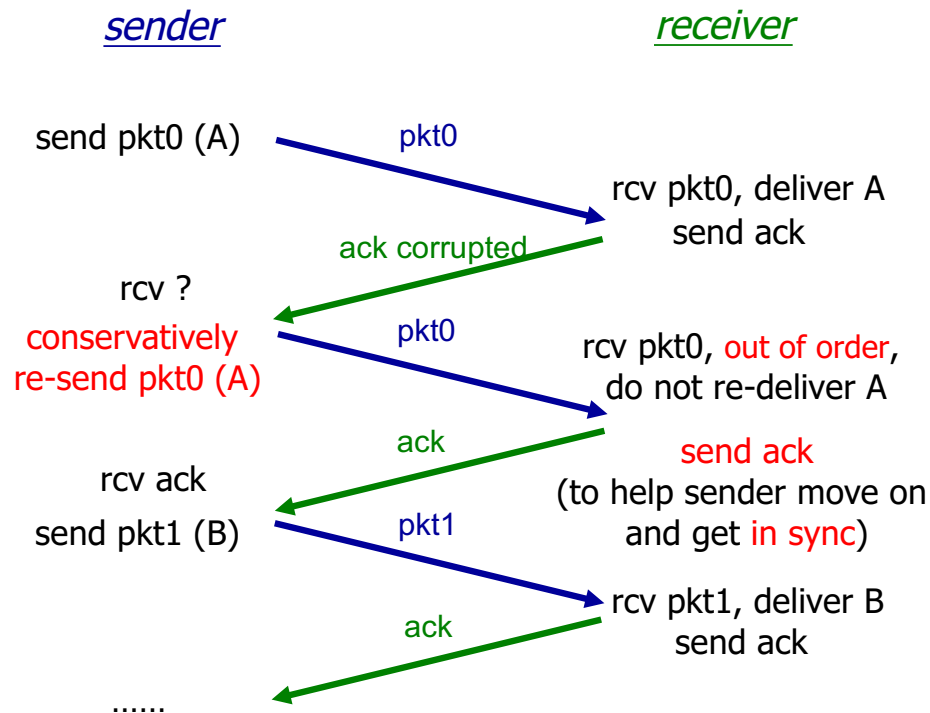


(a) No message or ACK corrupted

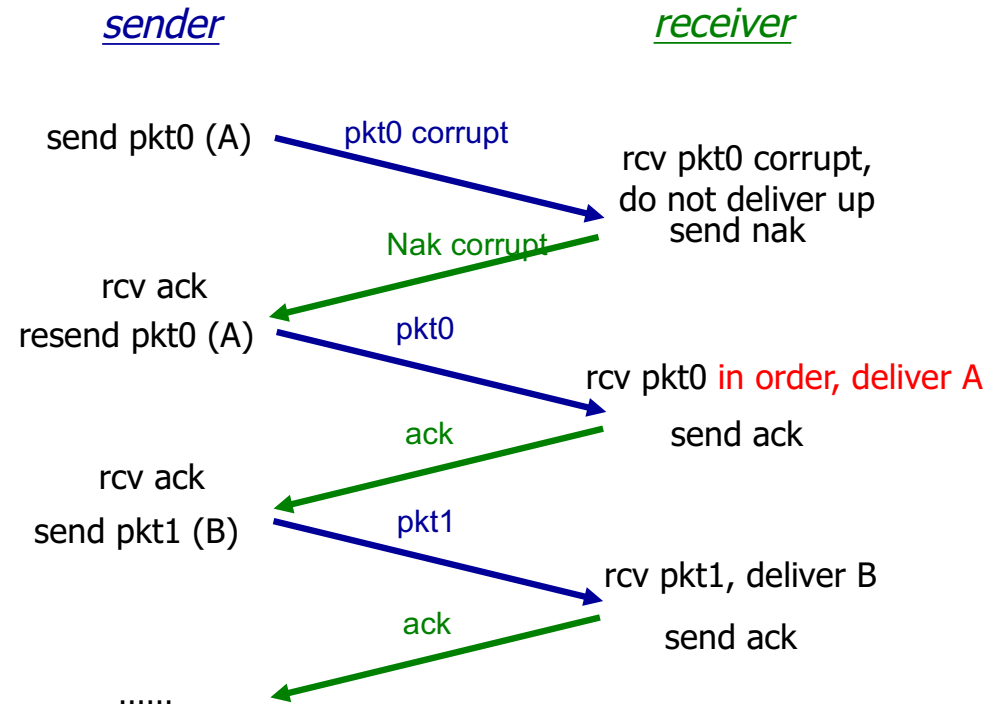


(b) Message corrupted, feedback ok

Rdt2.1 in action - continued



(c) Message ok, ACK corrupted

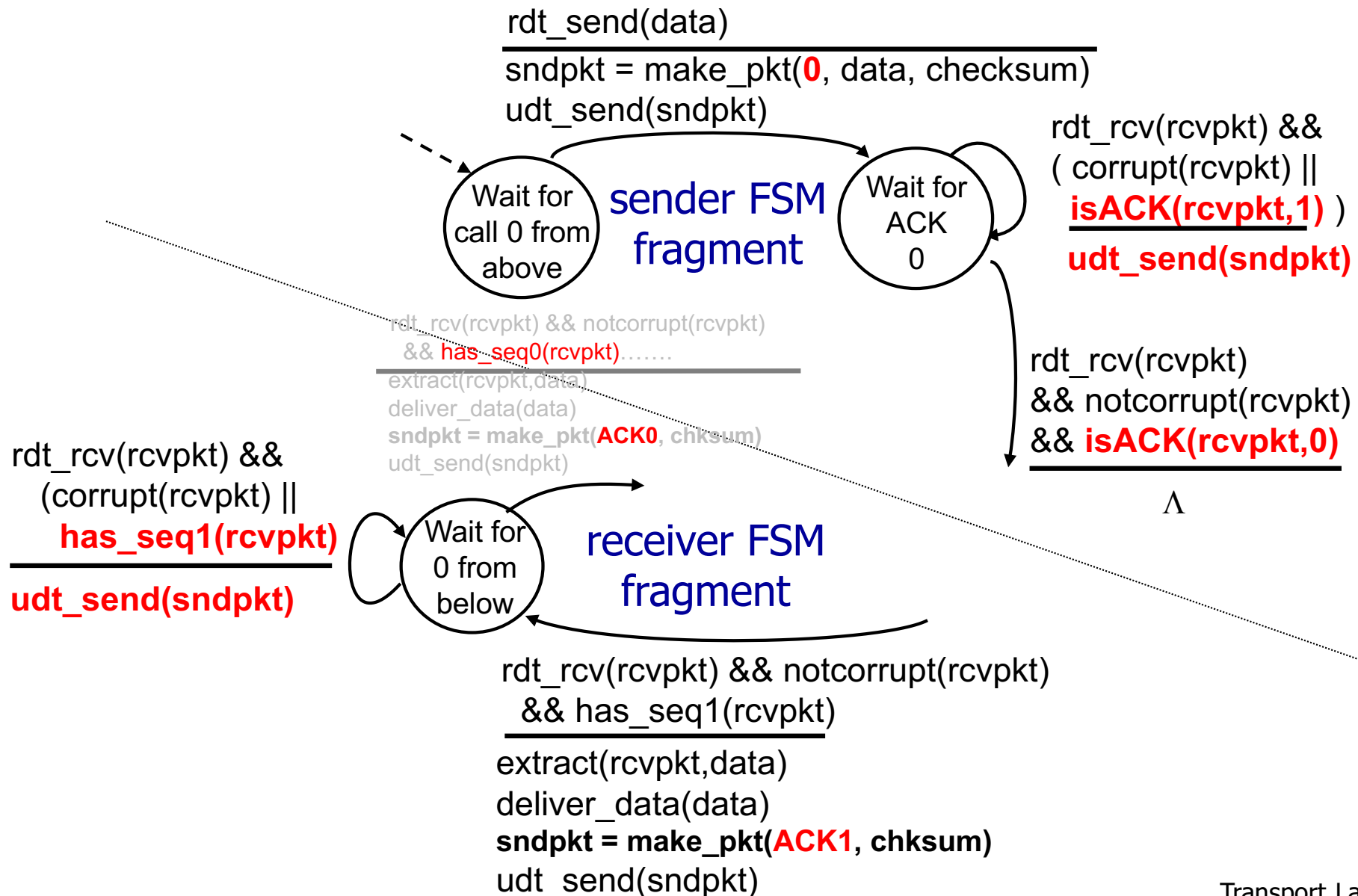


(d) Message corrupted, NAK corrupted

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** (=“duplicate ACK”)
 - receiver must *explicitly* include seq # of pkt being ACKed
- **duplicate ACK** at sender results in same action as NAK: *retransmit current pkt*

rdt2.2: sender, receiver fragments



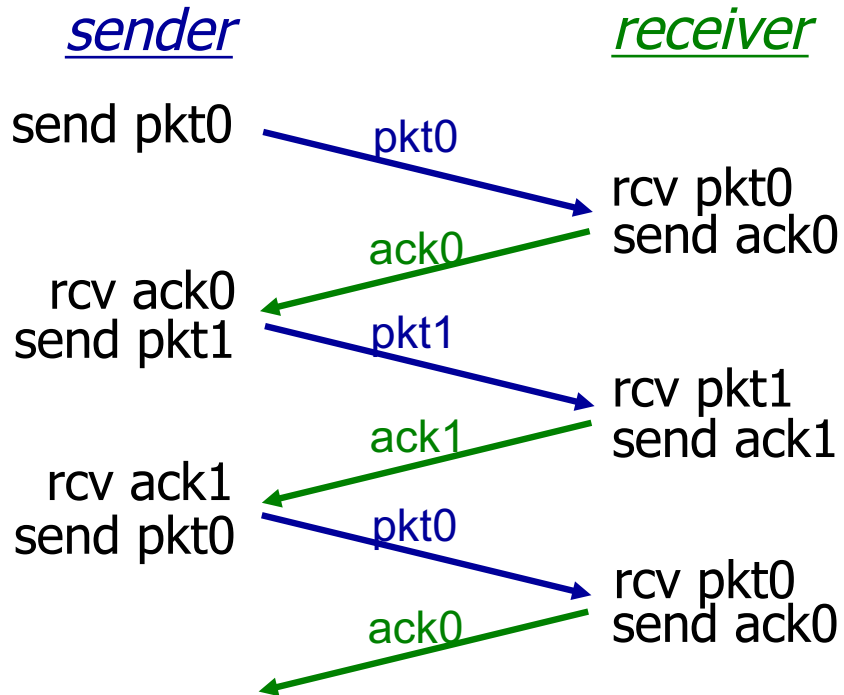
rdt3.0: channels with errors *and* loss

new assumption:

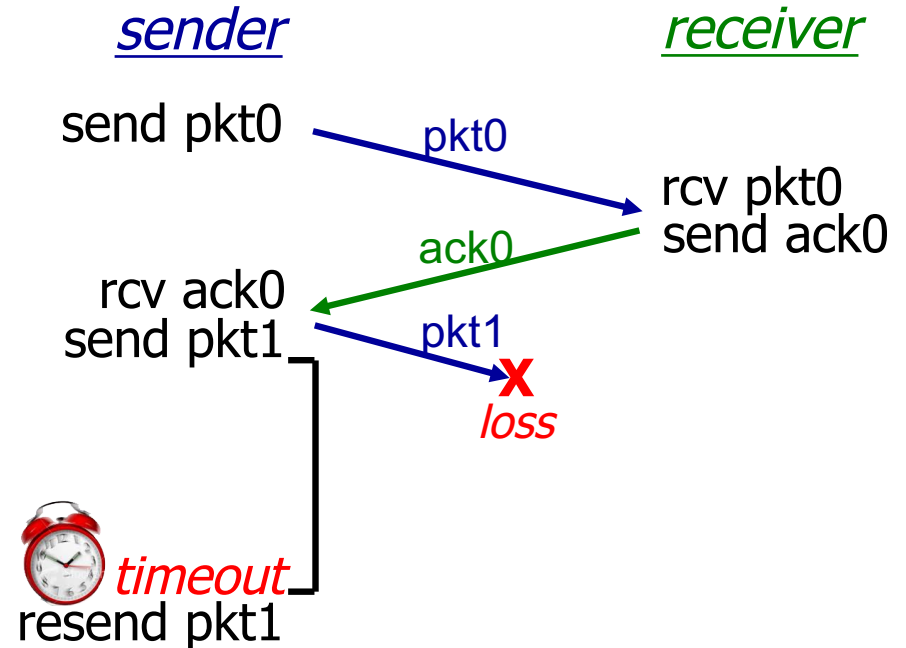
underlying channel can also lose packets (data, ACKs)

- How to detect loss?
- What to do in case of loss?
- checksum, seq. #, ACKs, retransmissions will be of help here, but not enough [why?]

rdt3.0 in action



(a) no loss



(b) packet loss

rdt3.0: channels with errors *and* loss

new assumption:

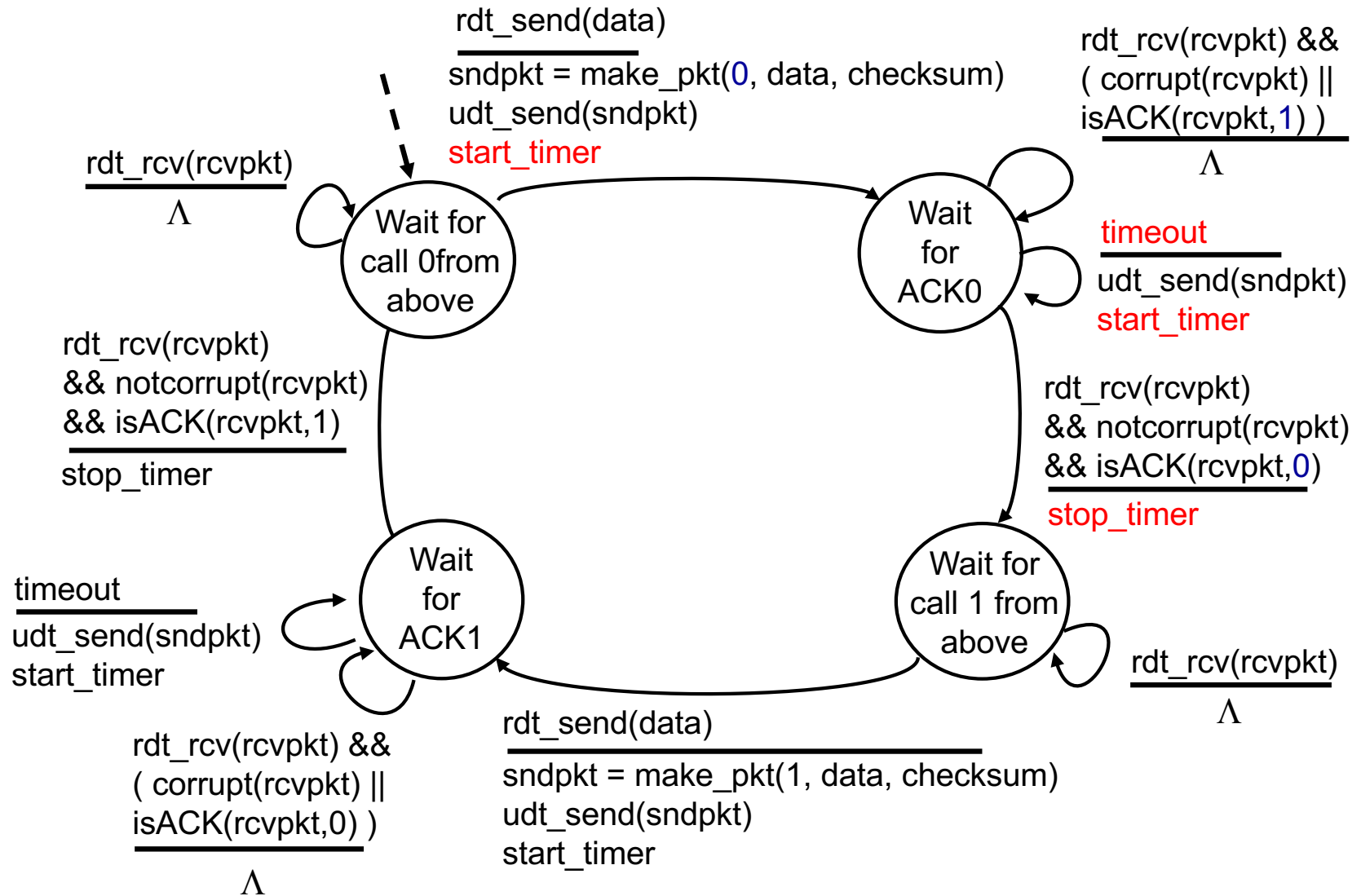
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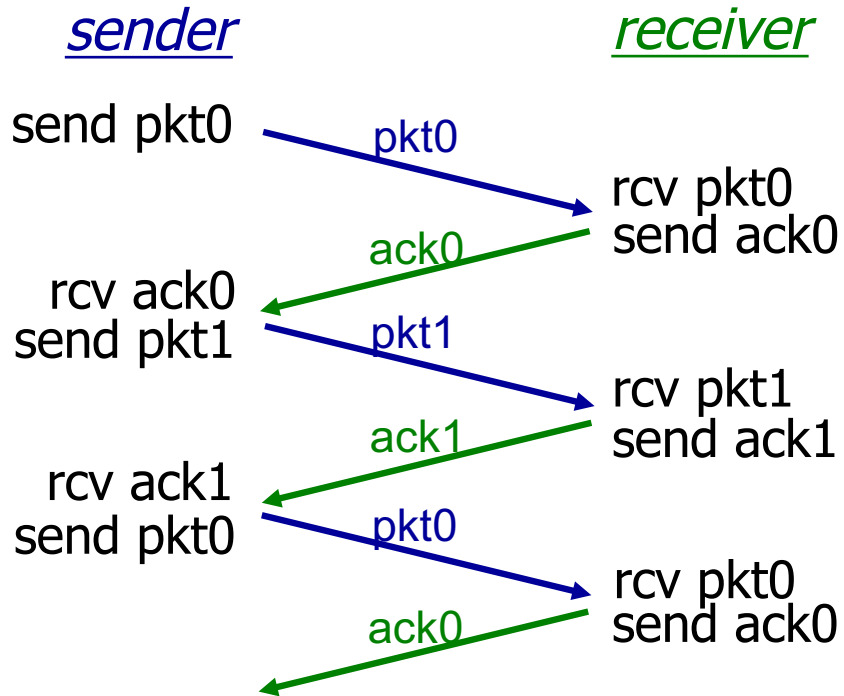
approach: detect loss at server with Timeout

- ❖ 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

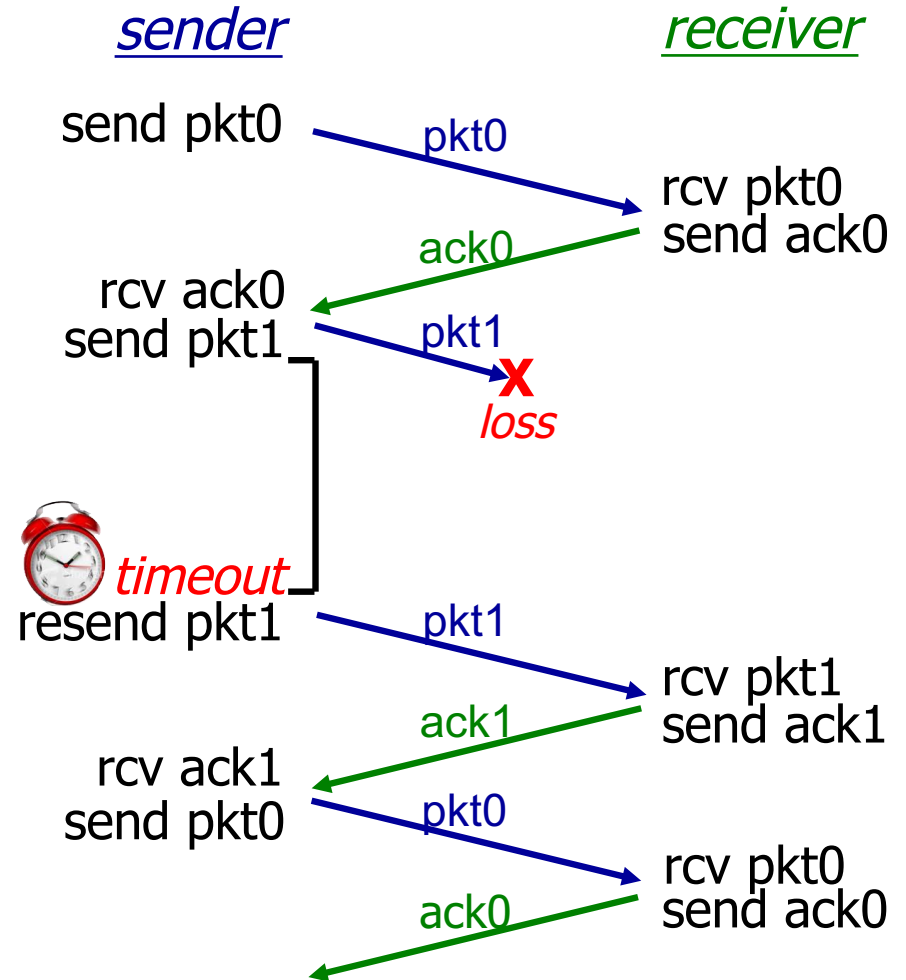
rdt3.0 sender (“alternating bit protocol”)



rdt3.0 in action

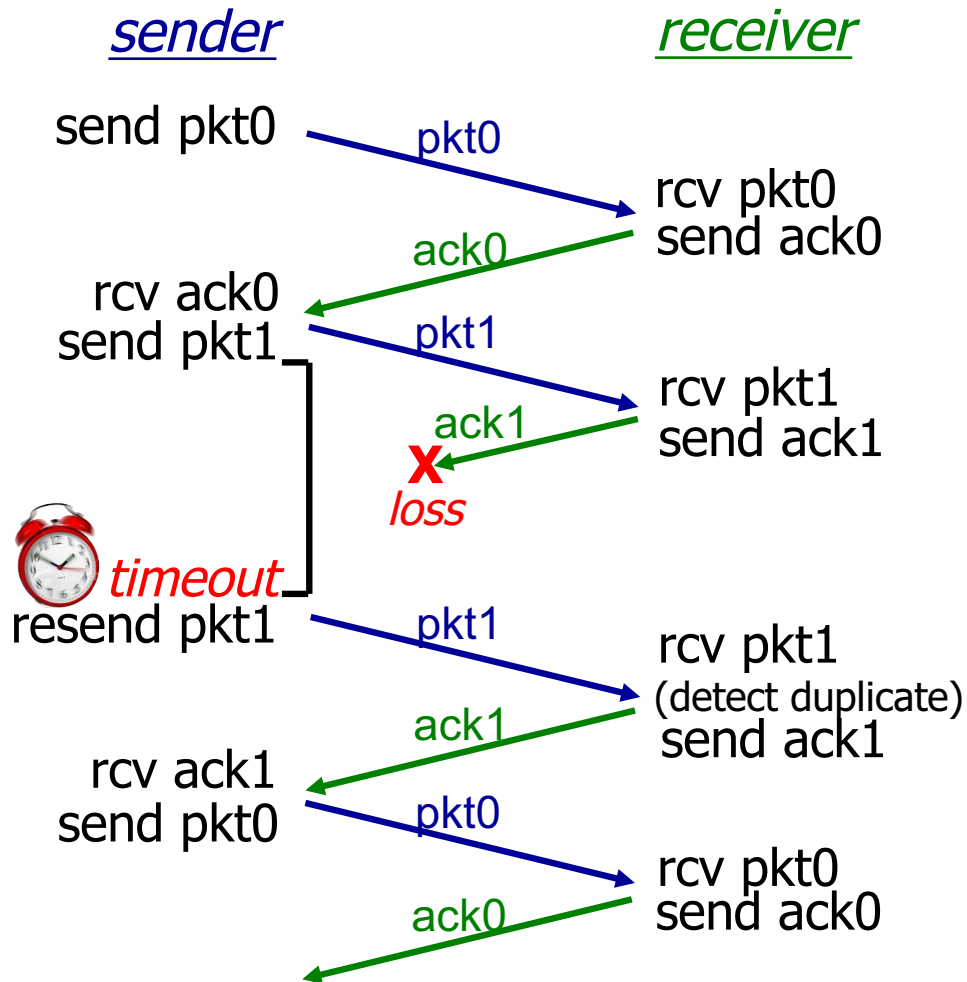


(a) no loss

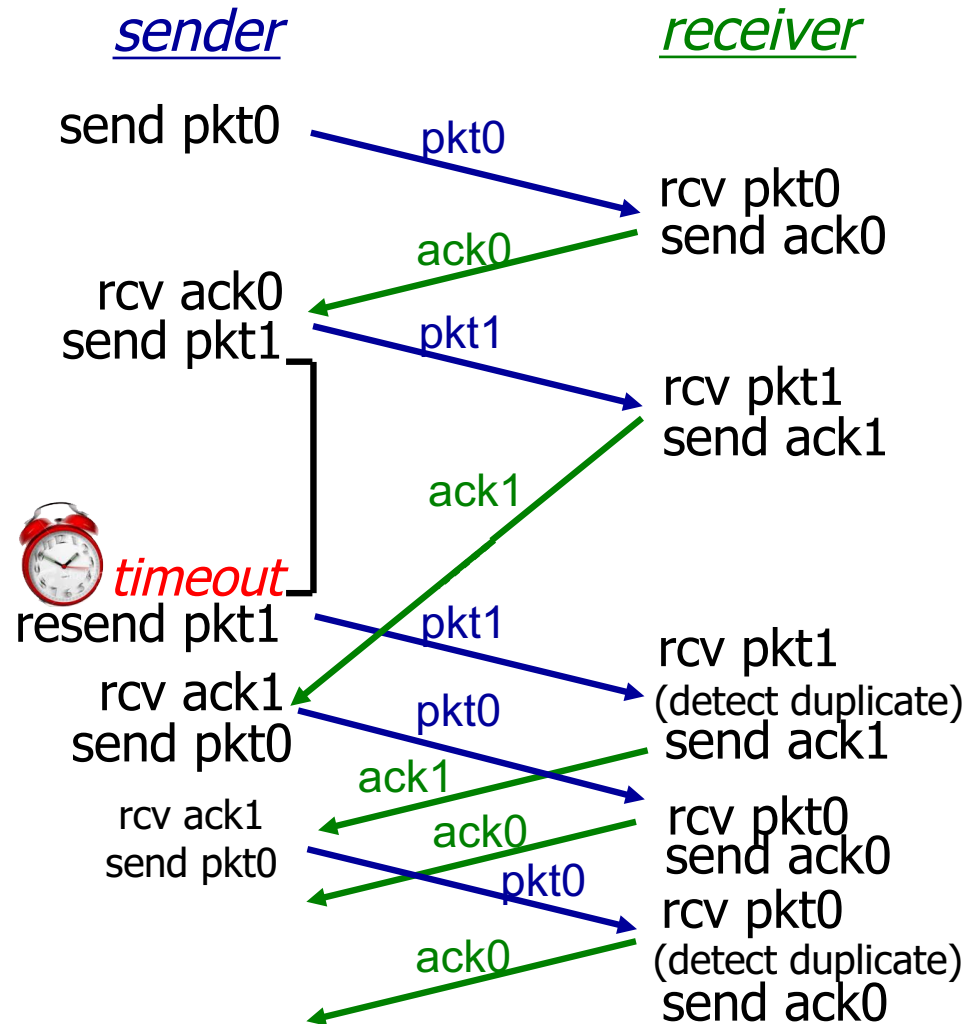


(b) packet loss

rdt3.0 in action

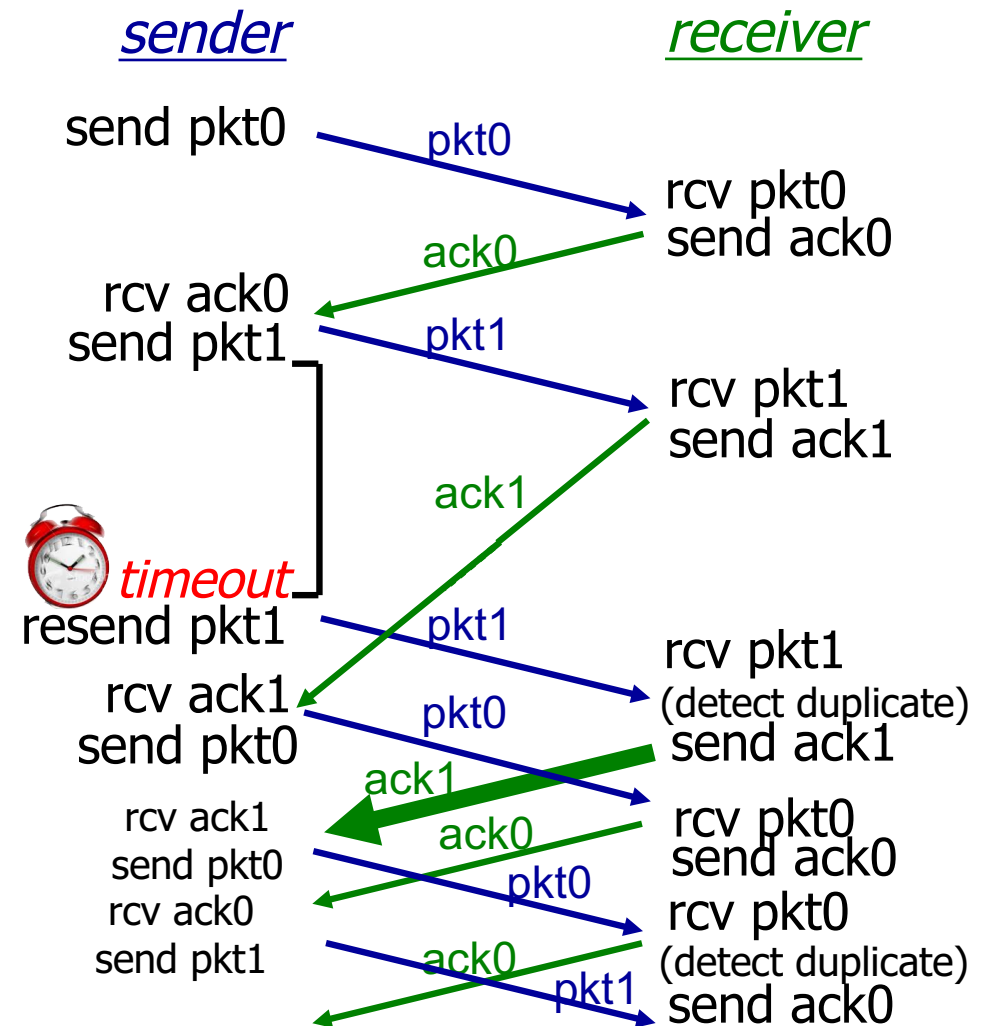


(c) ACK loss



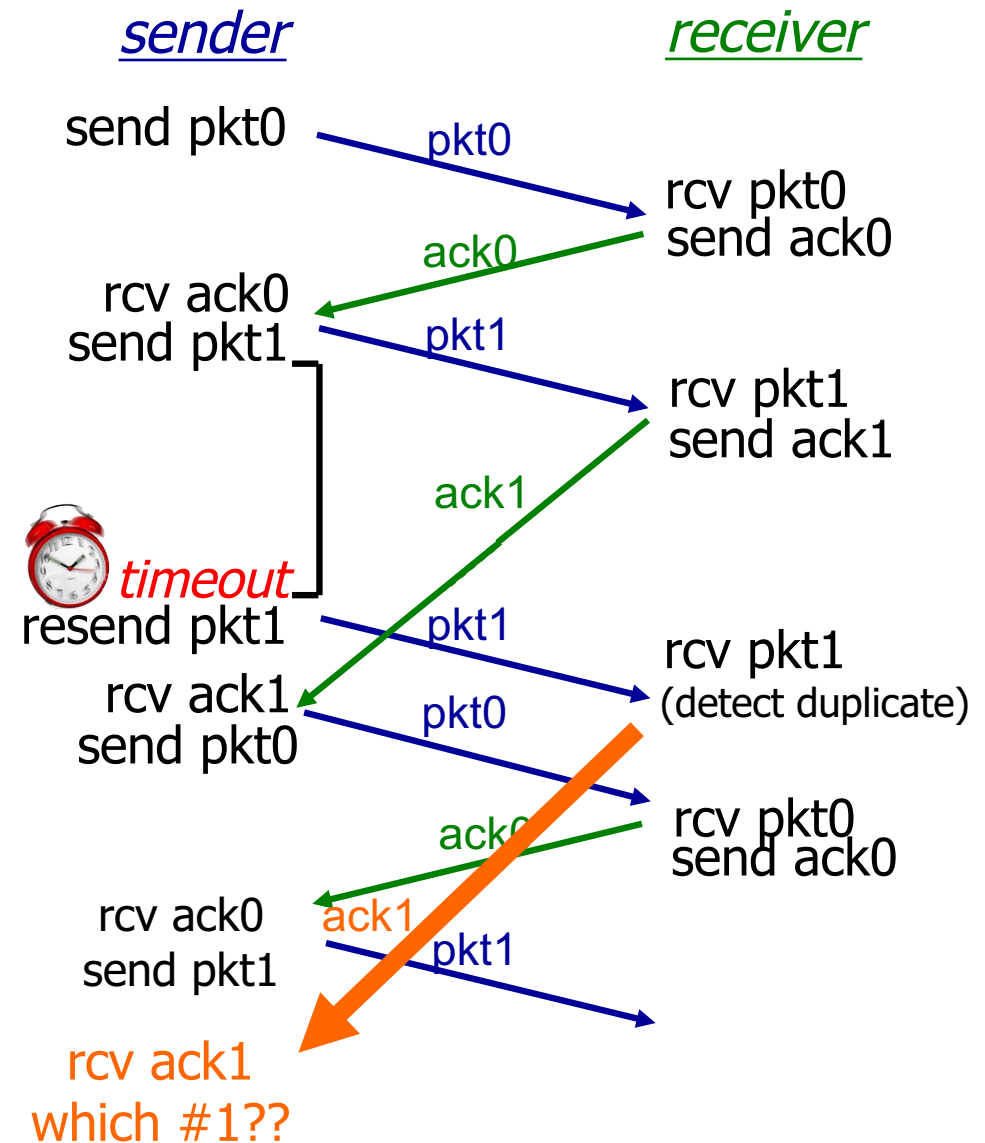
(d) premature timeout/ delayed ACK

rdt3.0 in action



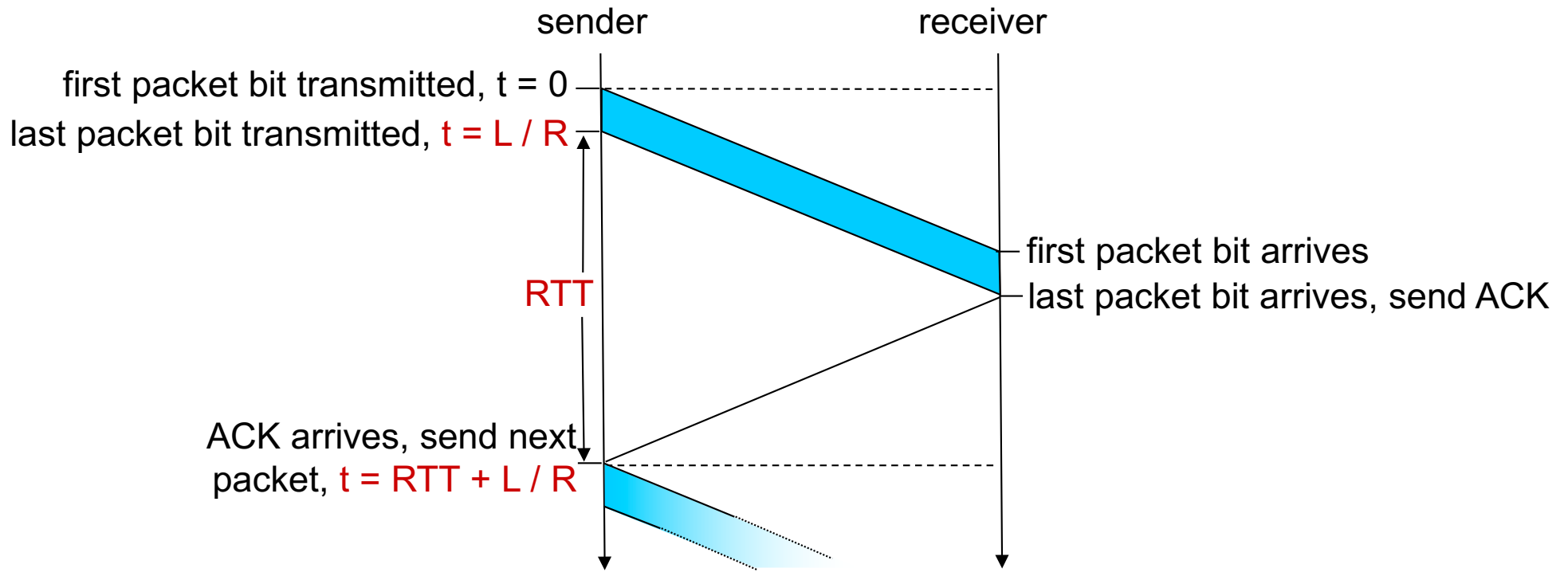
(d) premature timeout/ delayed ACK

rdt3.0: delayed ack- what if?



A: possible only if network reorders packets

rdt3.0: stop-and-wait operation



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