

Chapter 3

Transport Layer

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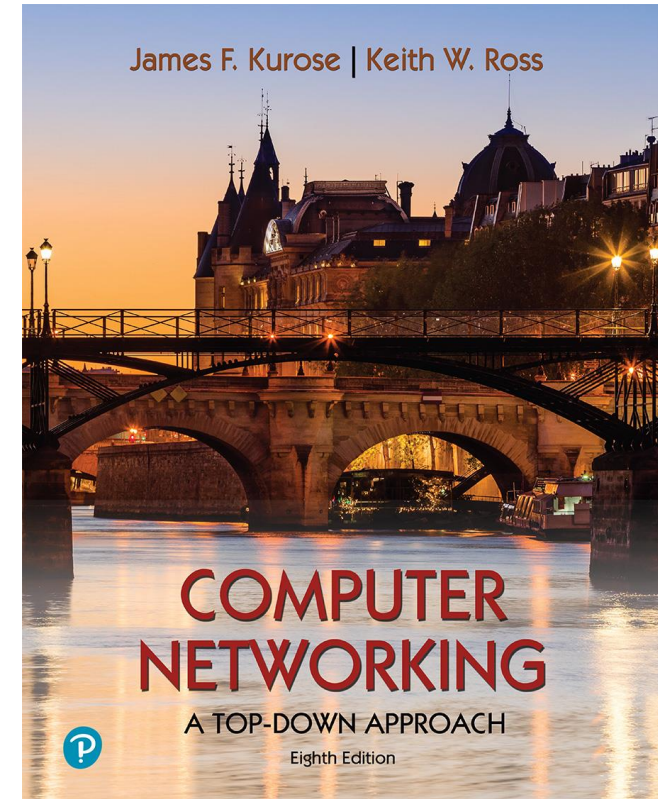
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Computer Networking: A Top-Down Approach

8th edition

Jim Kurose, Keith Ross
Pearson, 2020

Chapter 3: roadmap

- Transport-layer services
- Multiplexing and demultiplexing
- Connectionless transport: UDP
- **Principles of reliable data transfer**
- Connection-oriented transport: TCP
- Principles of congestion control
- TCP congestion control
- Evolution of transport-layer functionality

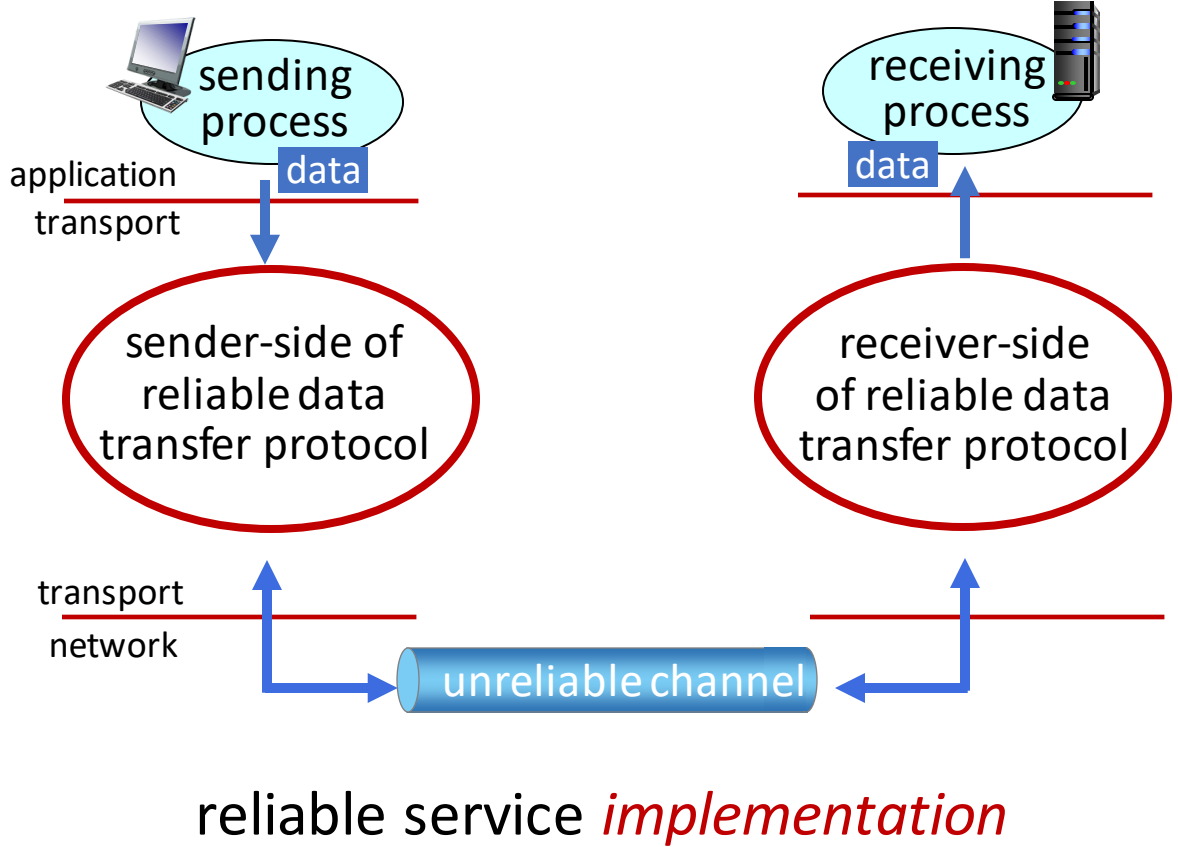
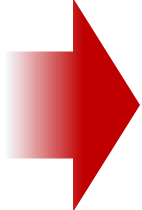
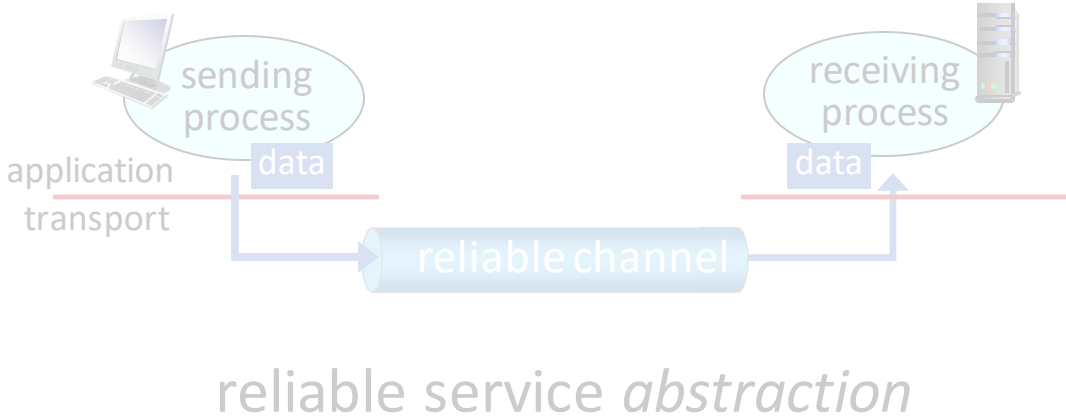


Principles of reliable data transfer



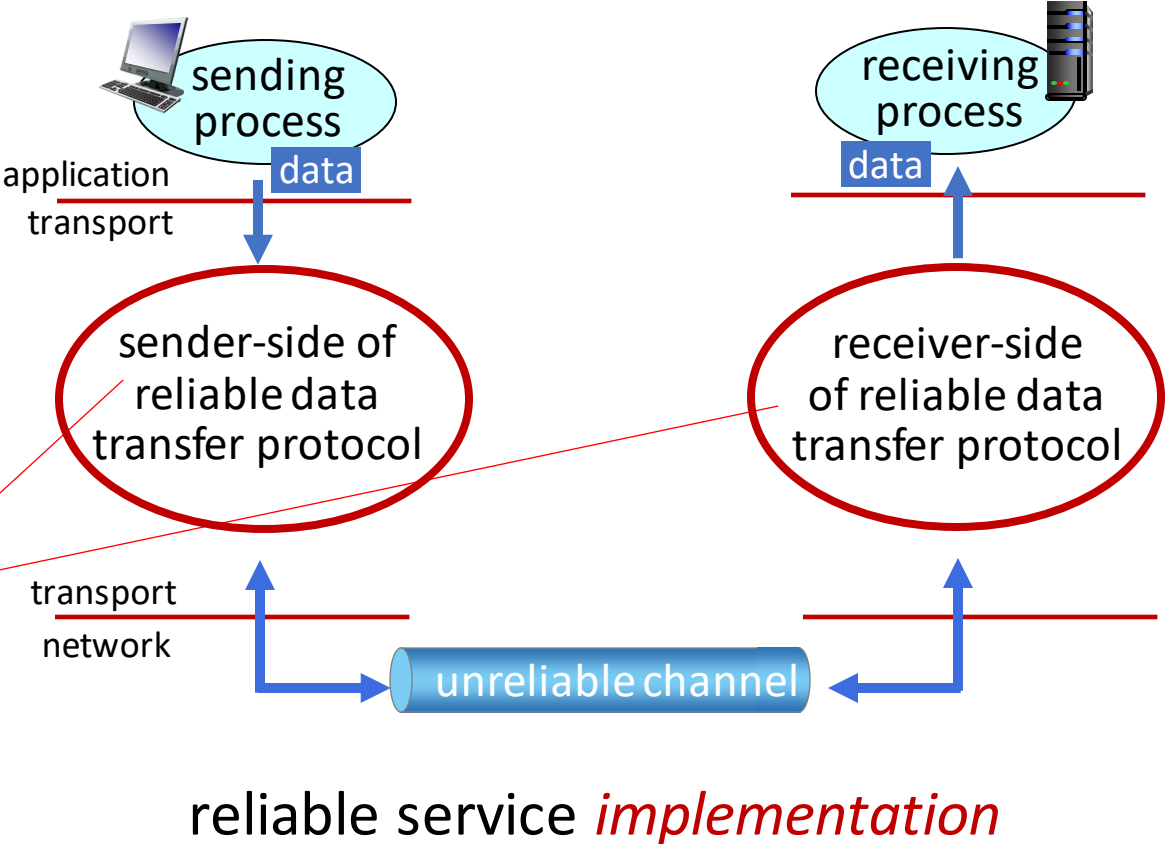
reliable service *abstraction*

Principles of reliable data transfer



Principles of reliable data transfer

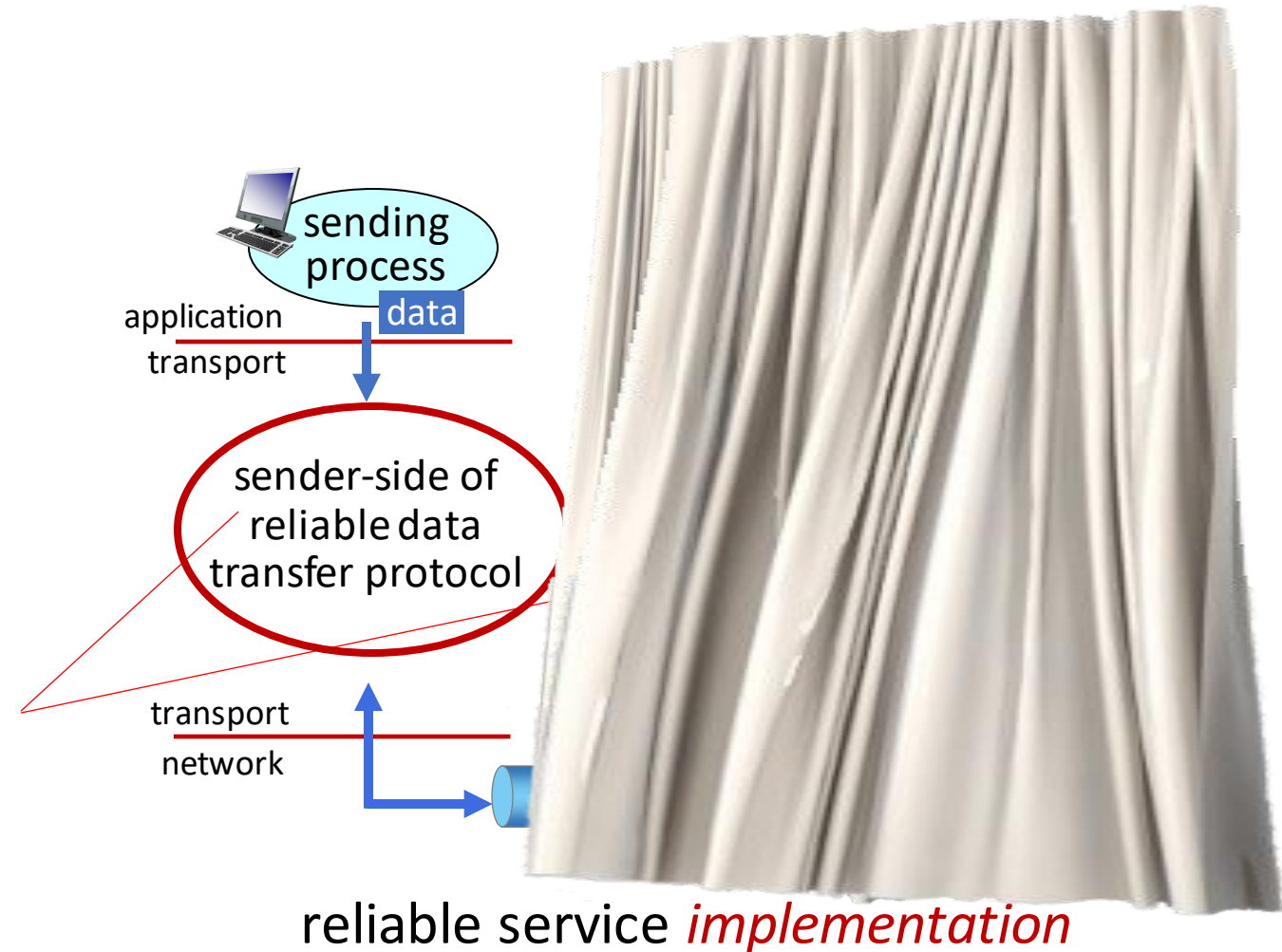
Complexity of reliable data transfer protocol will depend (strongly) on characteristics of unreliable channel (lose, corrupt, reorder data?)



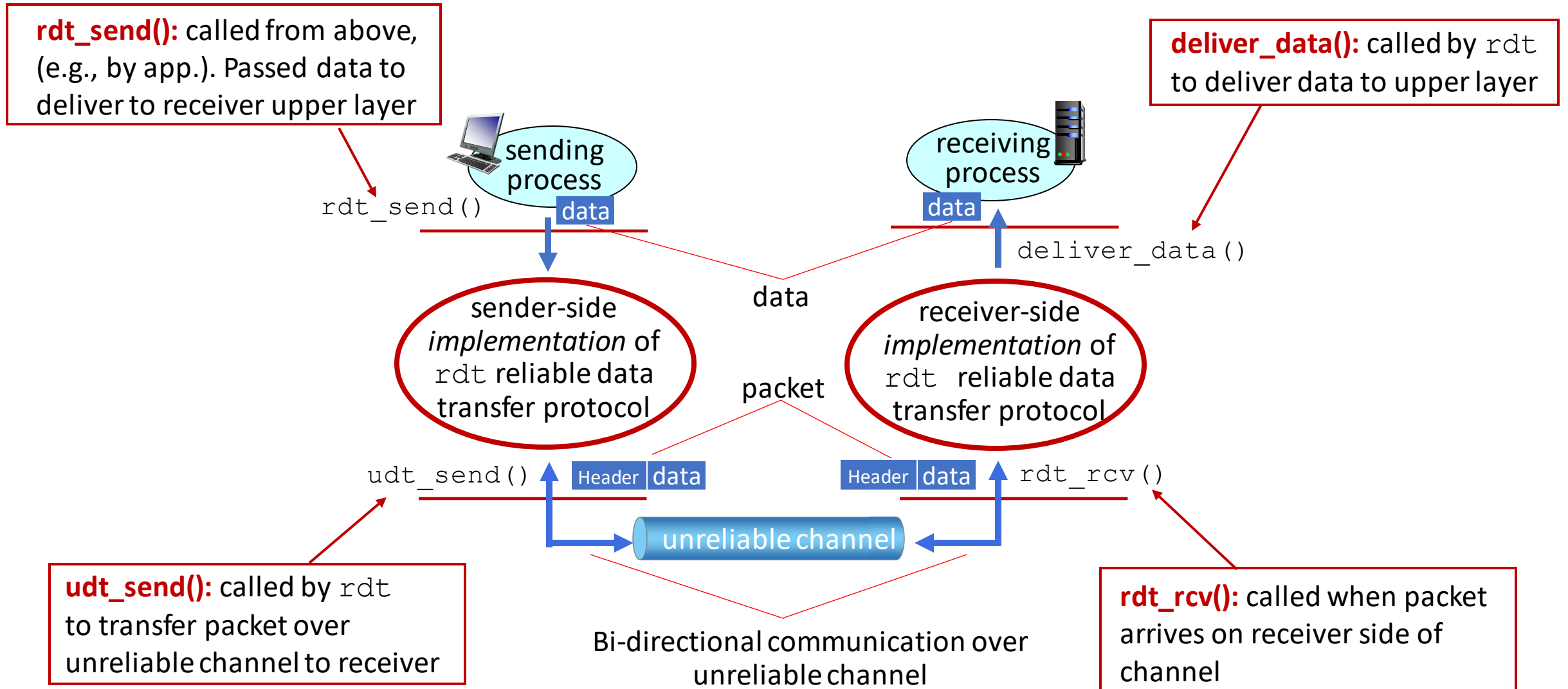
Principles of reliable data transfer

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

- unless communicated via a message



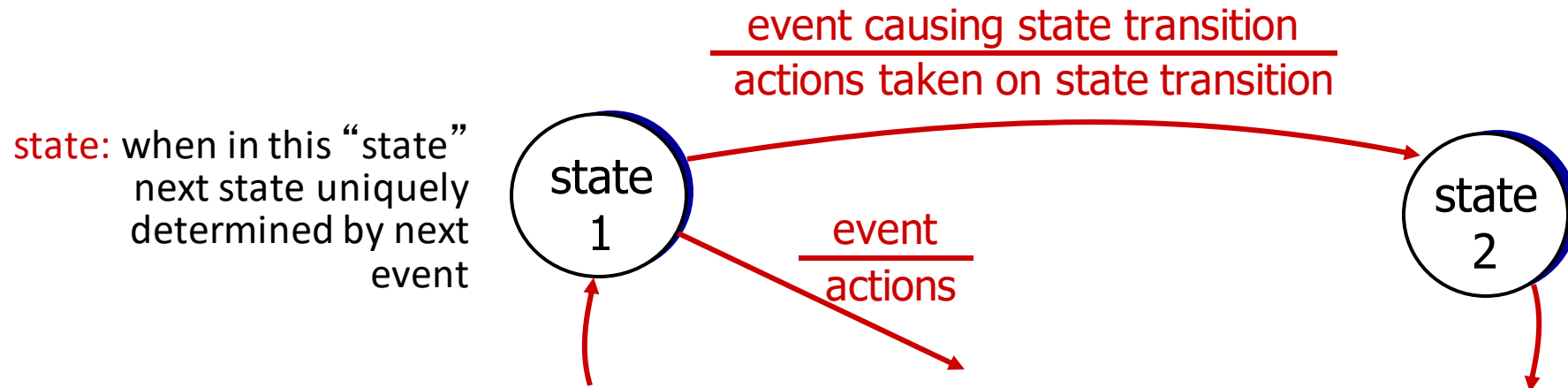
Reliable data transfer protocol (rdt): interfaces



Reliable data transfer: getting started

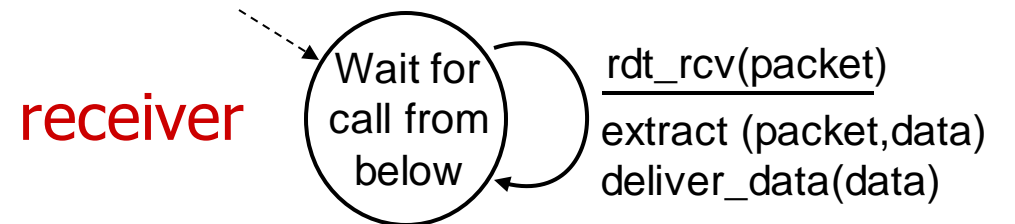
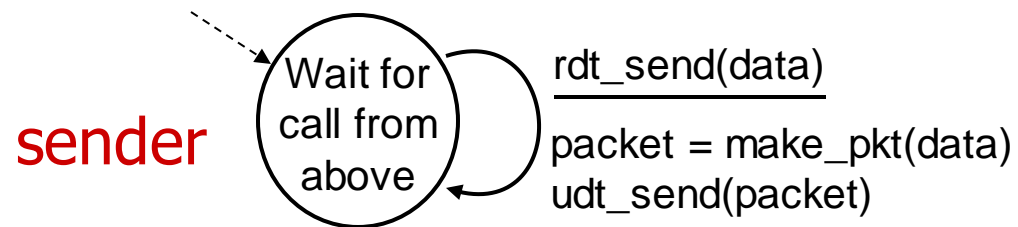
We will:

- incrementally develop sender, receiver sides of reliable data transfer protocol (rdt)
- consider only unidirectional data transfer
 - but control info will flow in 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



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?

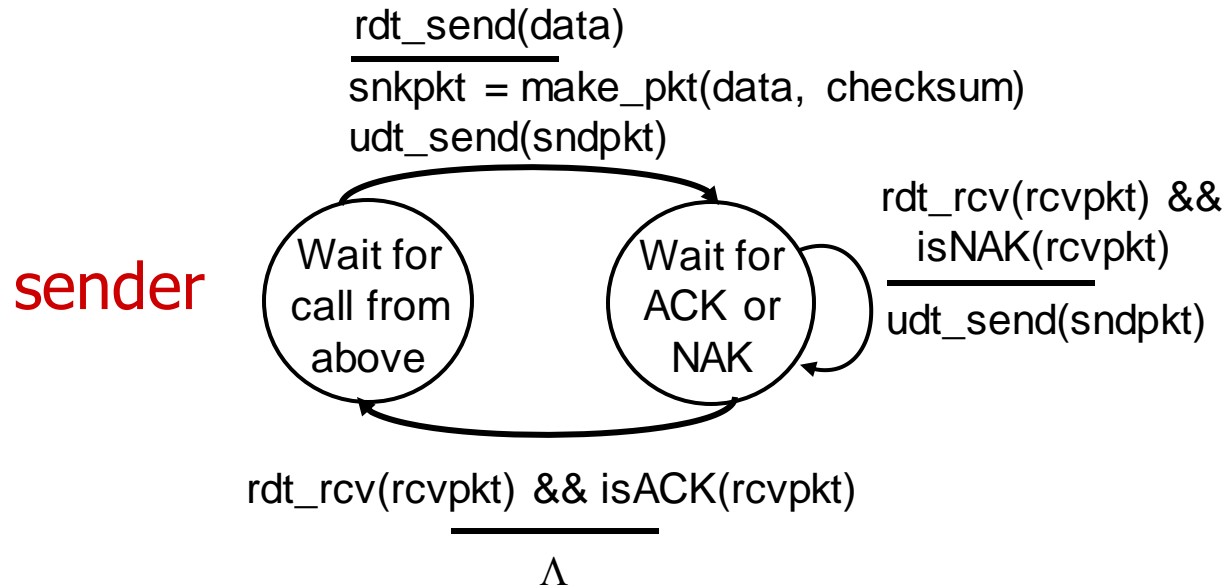
rdt2.0: channel with bit errors

- 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

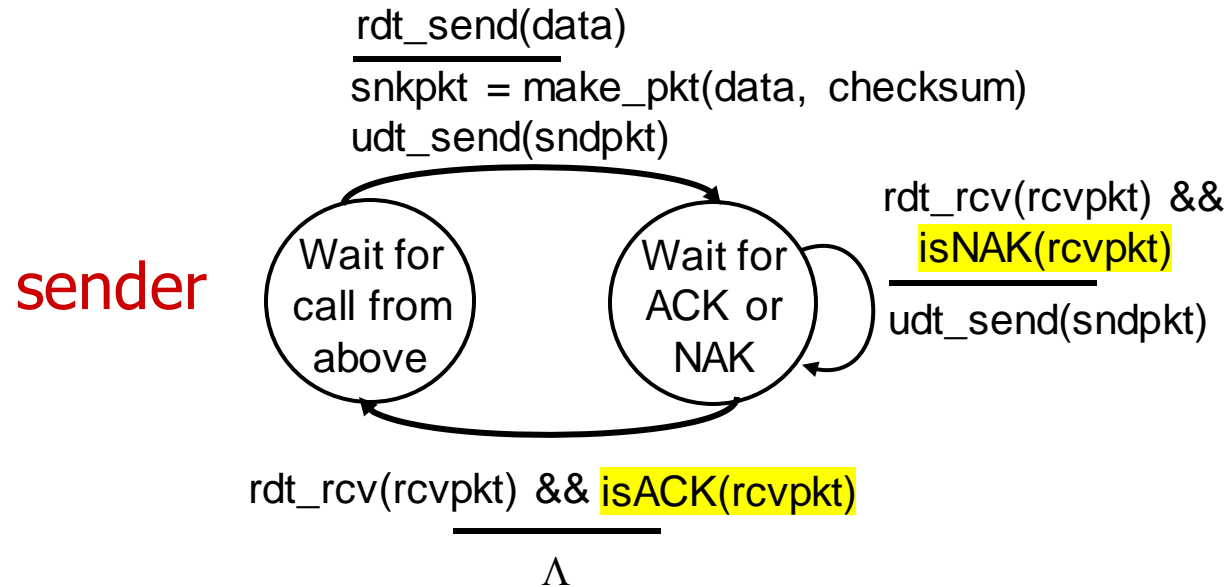
stop and wait

sender sends one packet, then waits for receiver response

rdt2.0: FSM specifications



rdt2.0: FSM specification

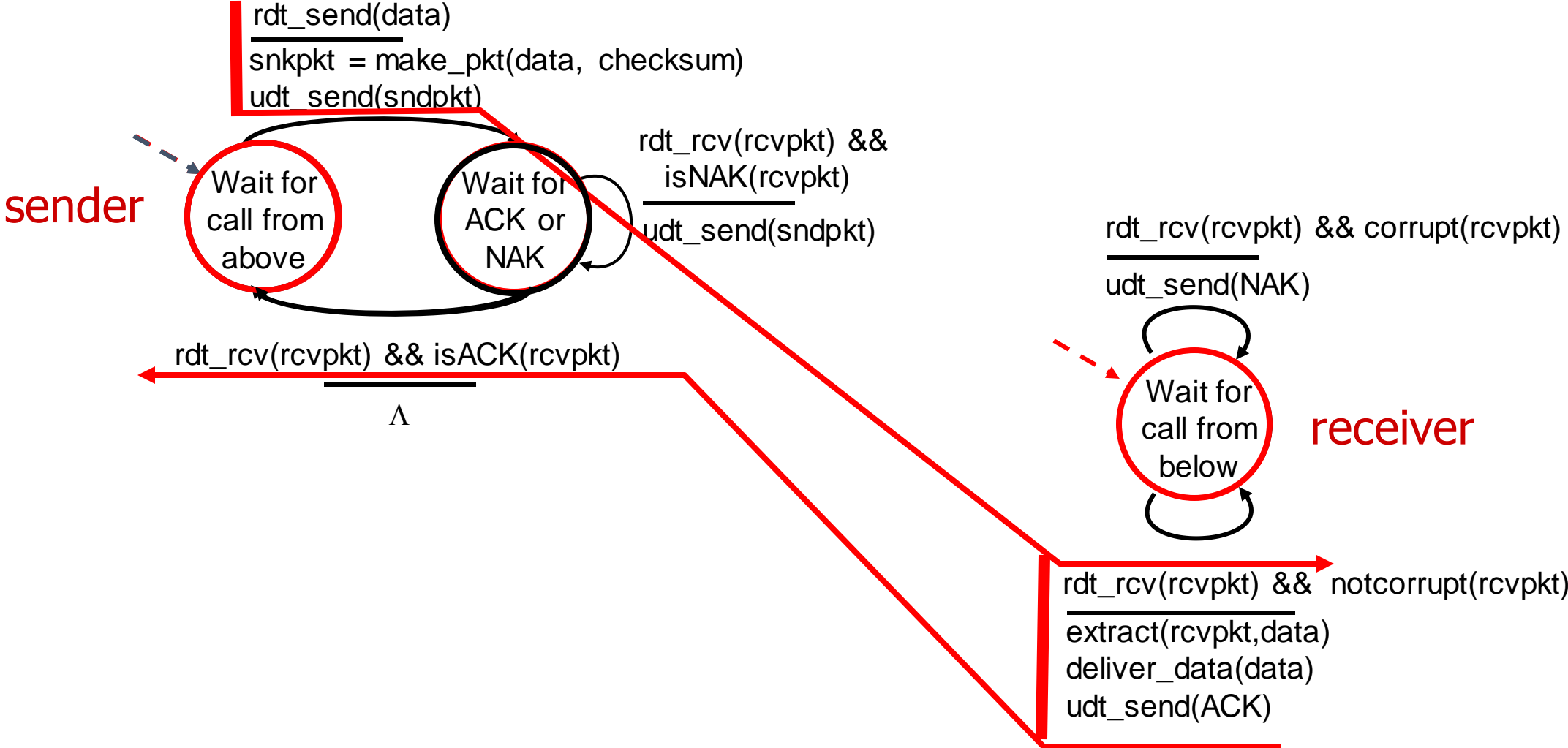


Note: “state” of receiver (did the receiver get my message correctly?) isn’t known to sender unless somehow communicated from receiver to sender

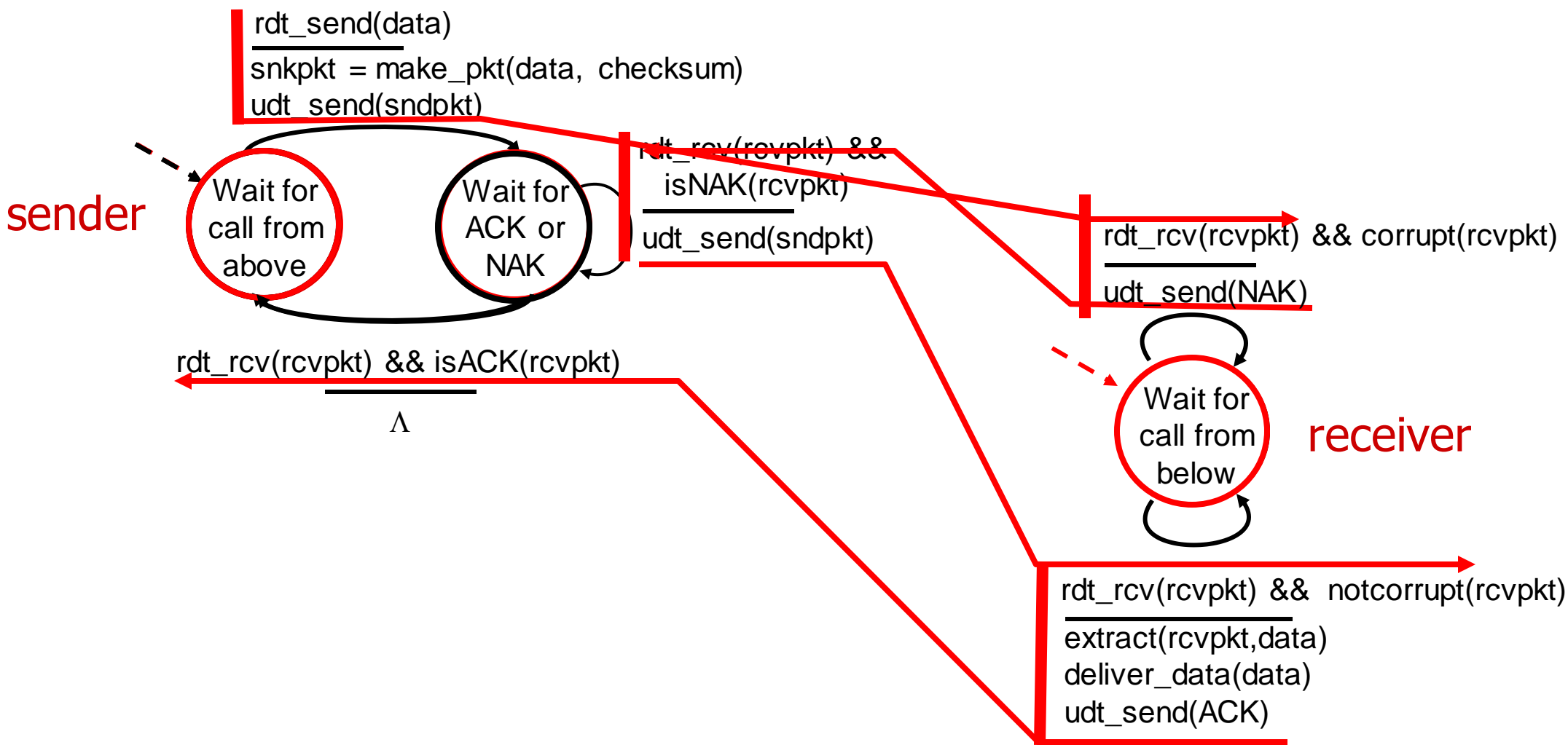
- that’s why we need a protocol!



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

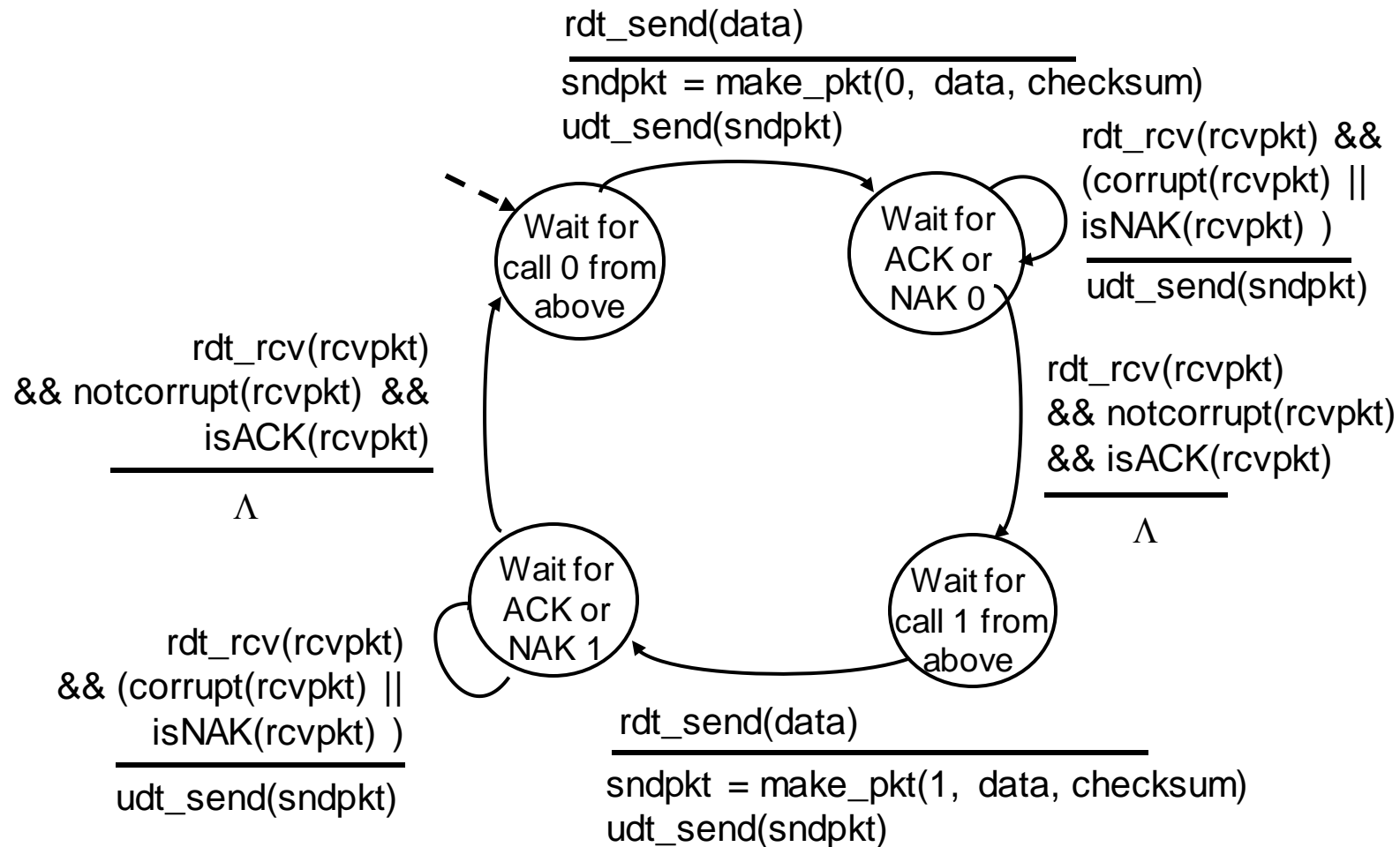
handling duplicates:

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

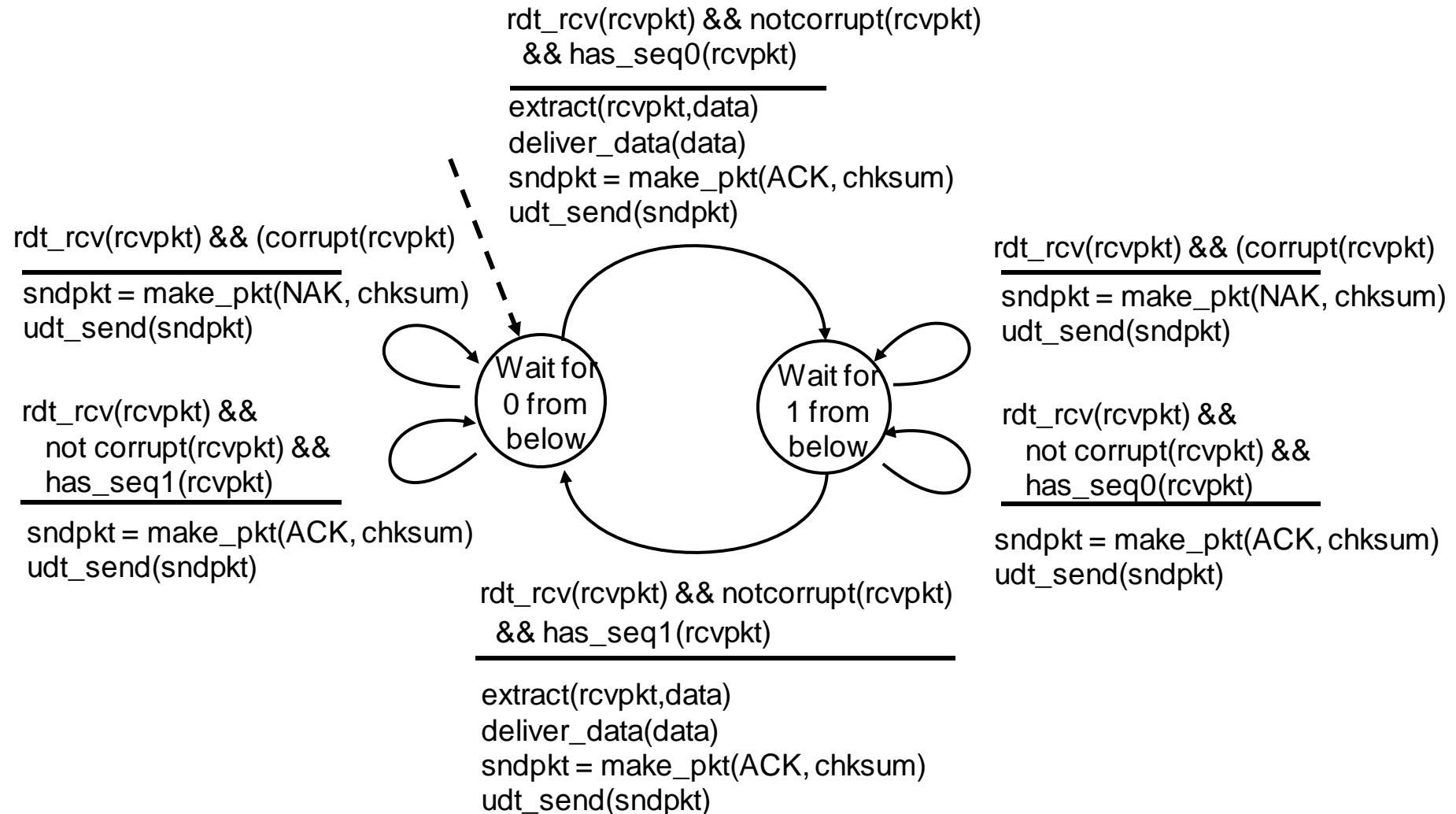
stop and wait

sender sends one packet, then waits for receiver response

rdt2.1: sender, handling garbled ACK/NAKs



rdt2.1: receiver, handling garbled ACK/NAKs



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:

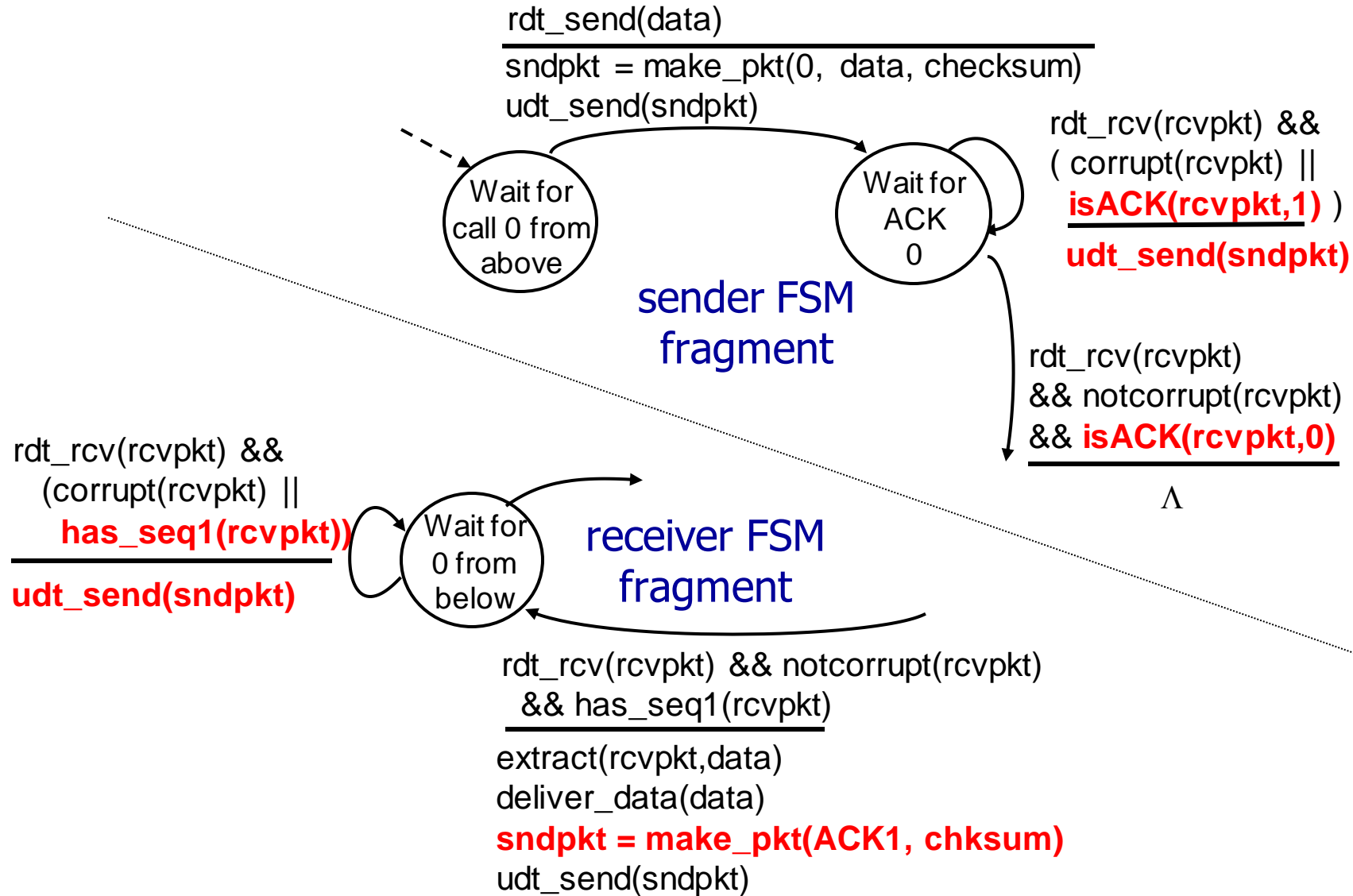
- must check if received packet is duplicate
 - state indicates whether 0 or 1 is expected pkt seq #
- note: receiver can *not* know if its last ACK/NAK received OK at sender

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



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

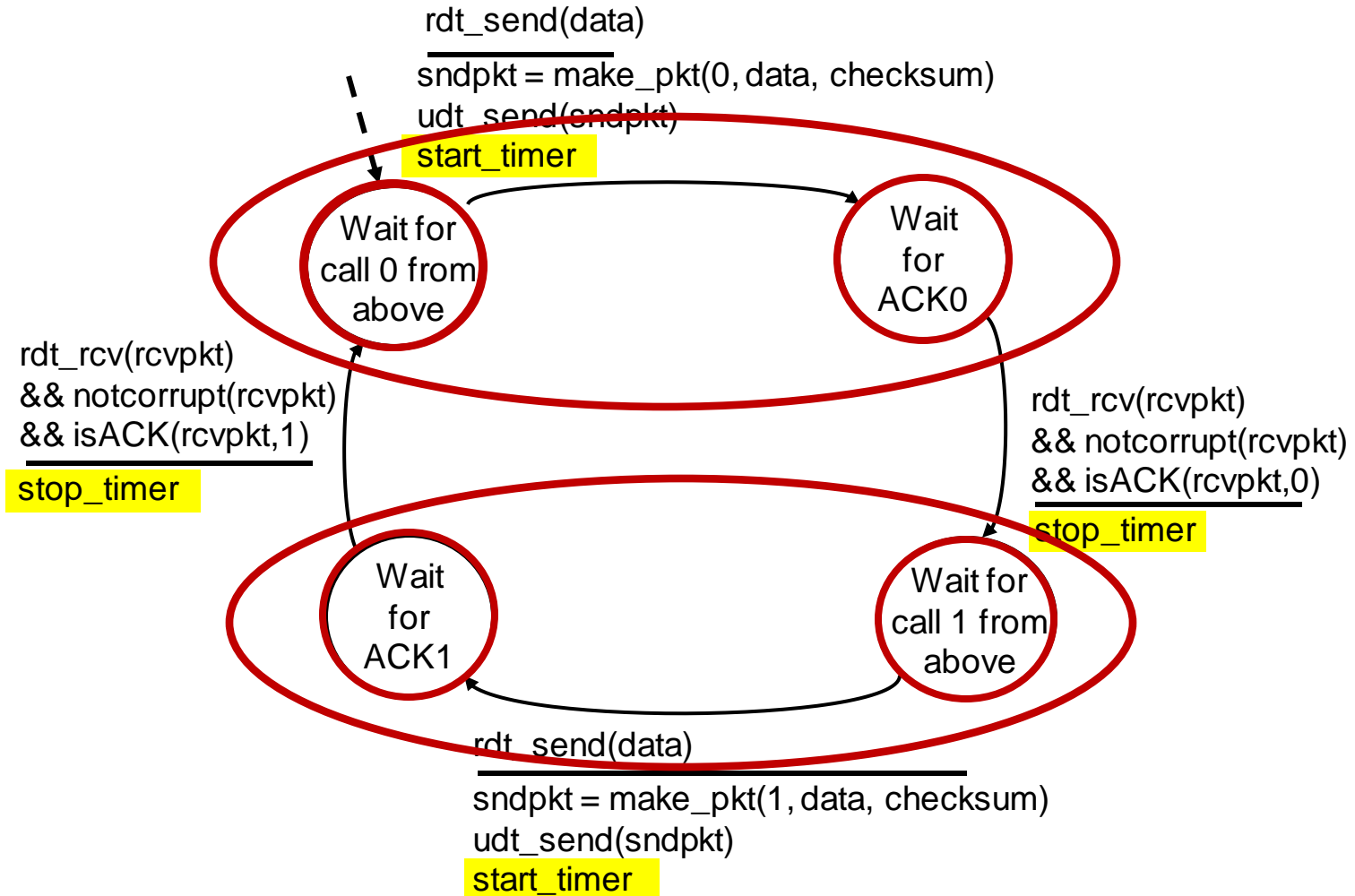
Approach: sender waits “reasonable” amount of time for ACK

- 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

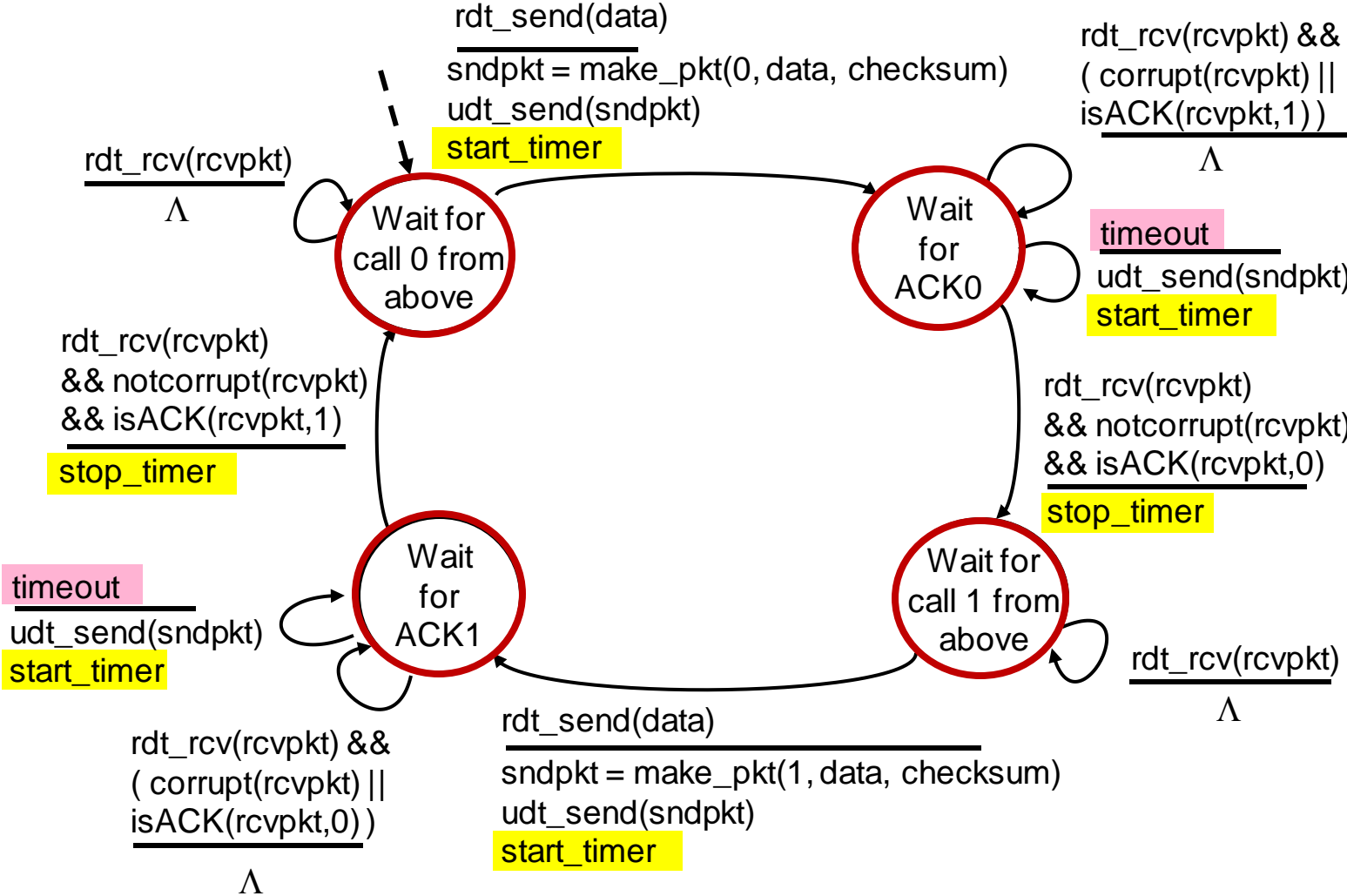


timeout

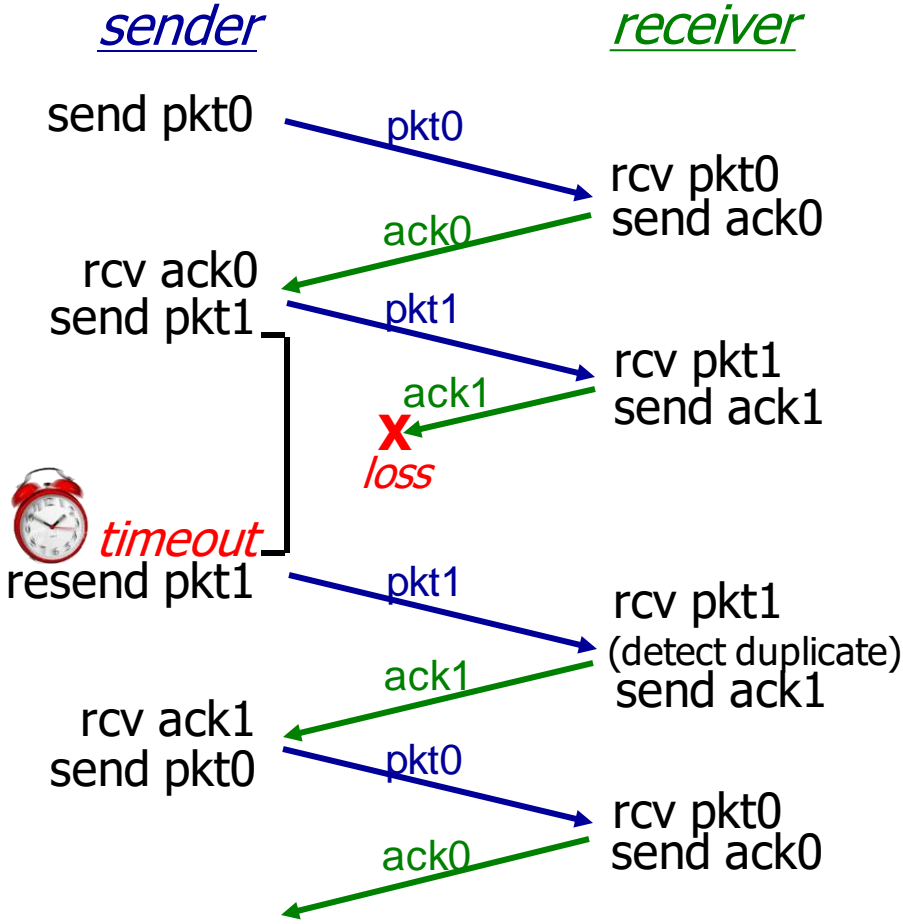
rdt3.0 sender



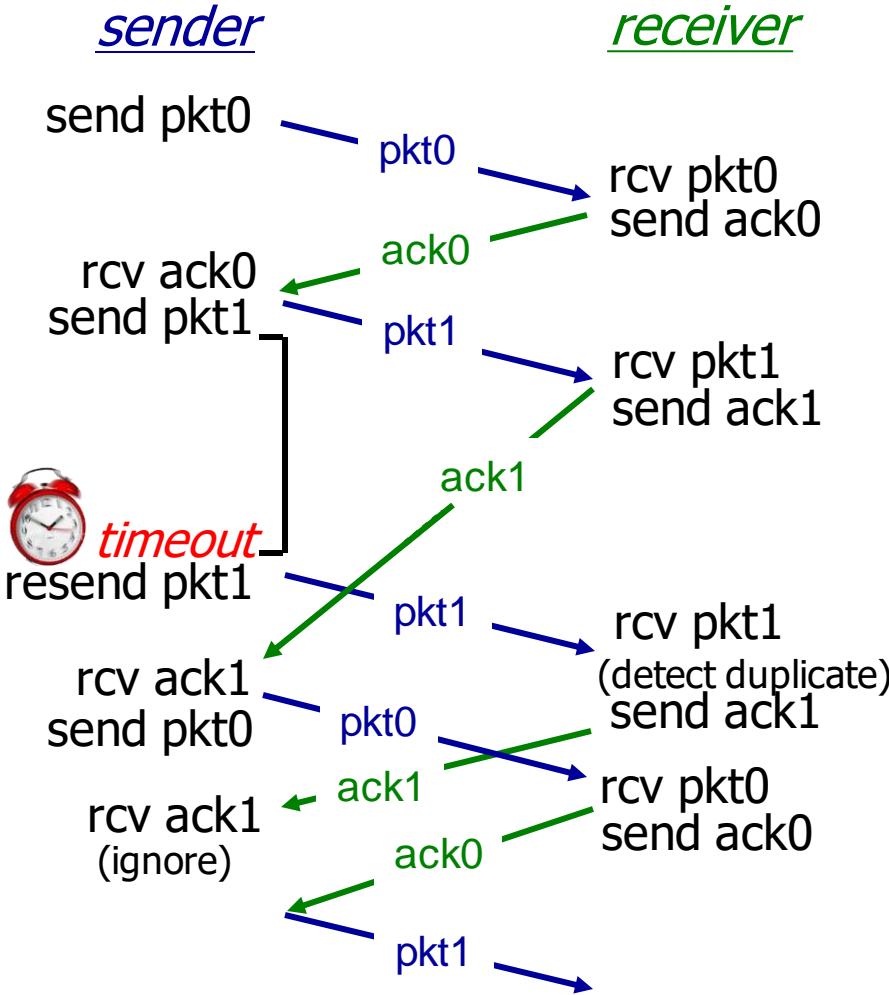
rdt3.0 sender



rdt3.0 in action



(c) ACK loss



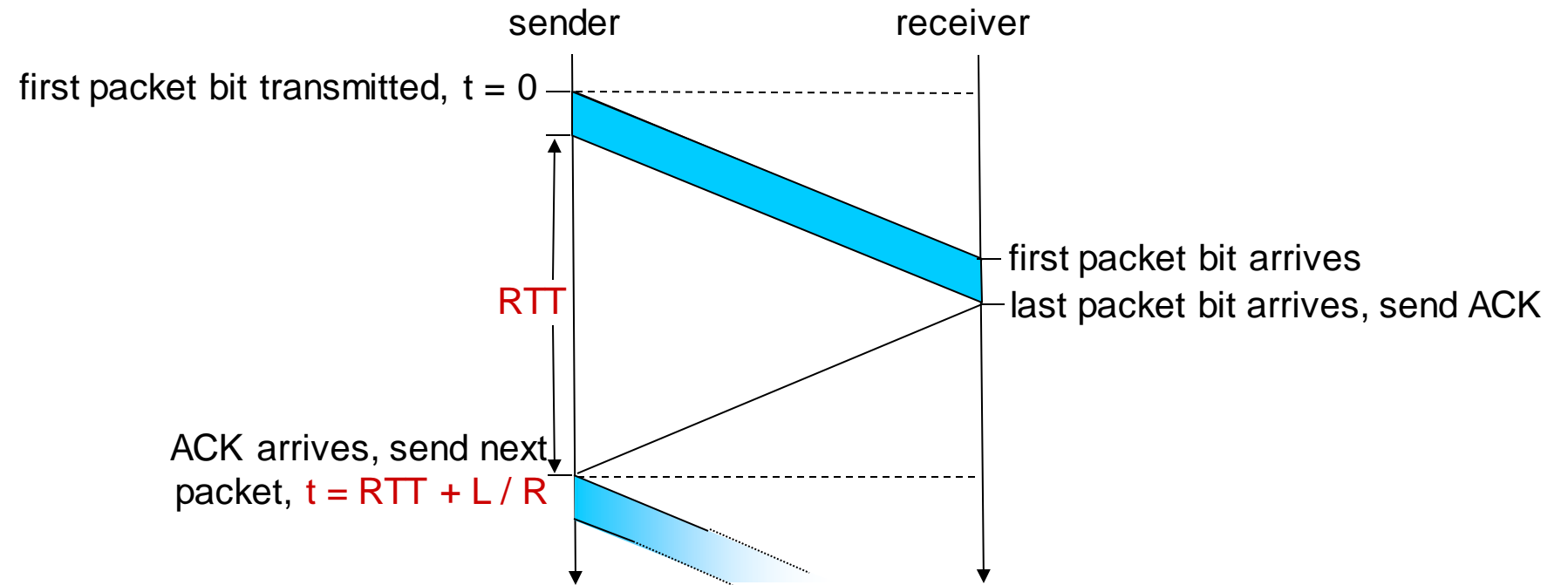
(d) premature timeout/ delayed ACK

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:

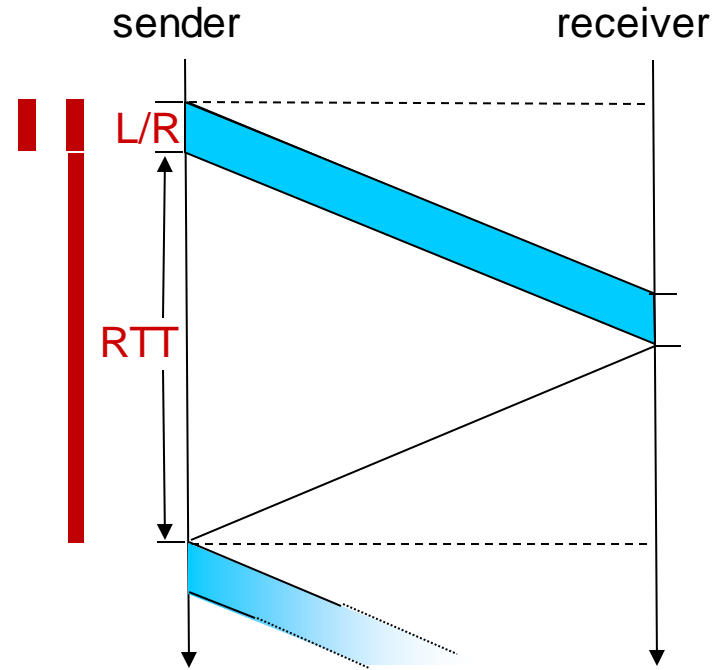
$$D_{trans} = \frac{L}{R} = \frac{8000 \text{ bits}}{10^9 \text{ bits/sec}} = 8 \text{ microsecs}$$

rdt3.0: stop-and-wait operation



rdt3.0: stop-and-wait operation

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

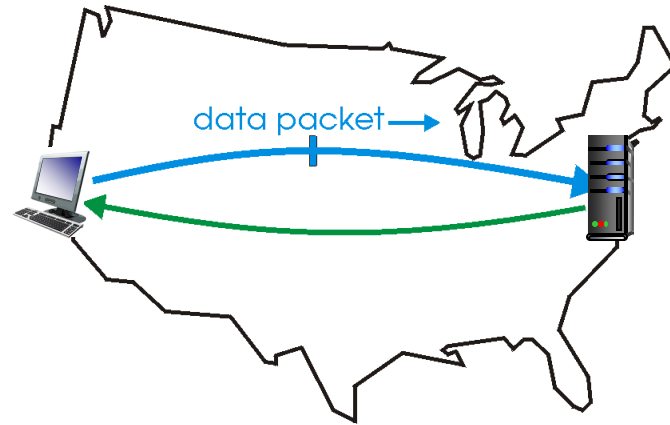


- rdt 3.0 protocol performance stinks!
- Protocol limits performance of underlying infrastructure (channel)

rdt3.0: pipelined protocols operation

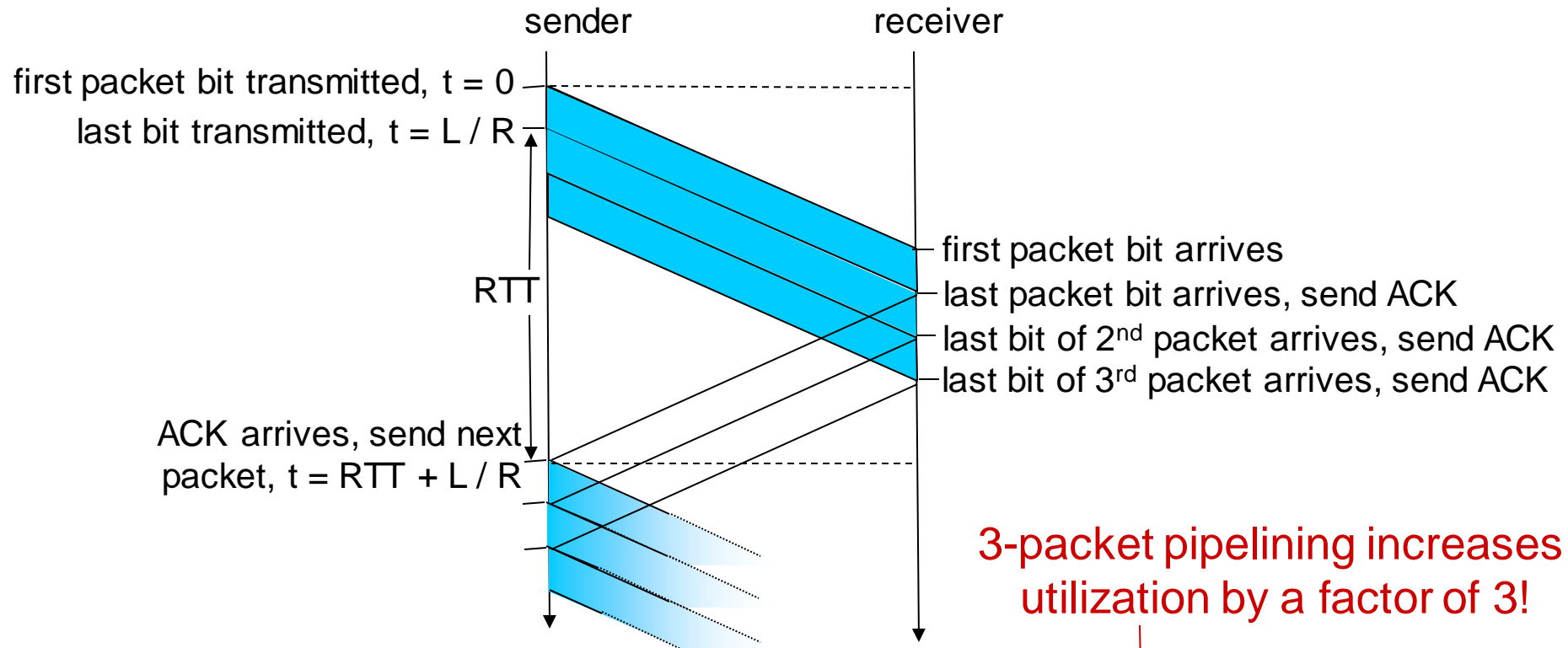
pipelining: sender allows multiple, “in-flight”, yet-to-be-acknowledged packets

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



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

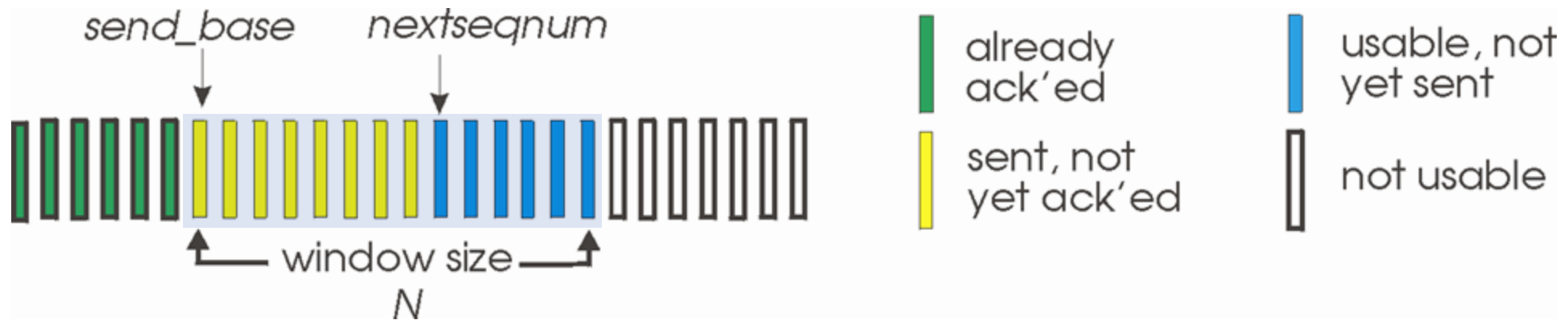
Pipelining: increased utilization



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

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:



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

receive pkt3, discard,
 (re)send ack1

receive pkt4, discard,
 (re)send ack1

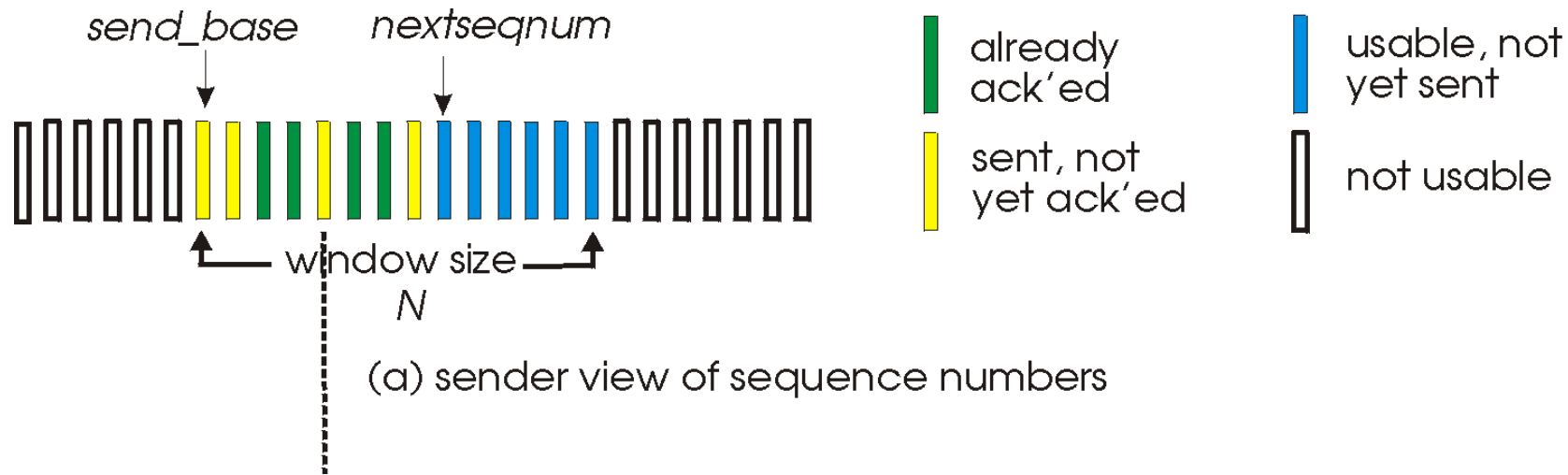
receive pkt5, discard,
 (re)send ack1

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



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

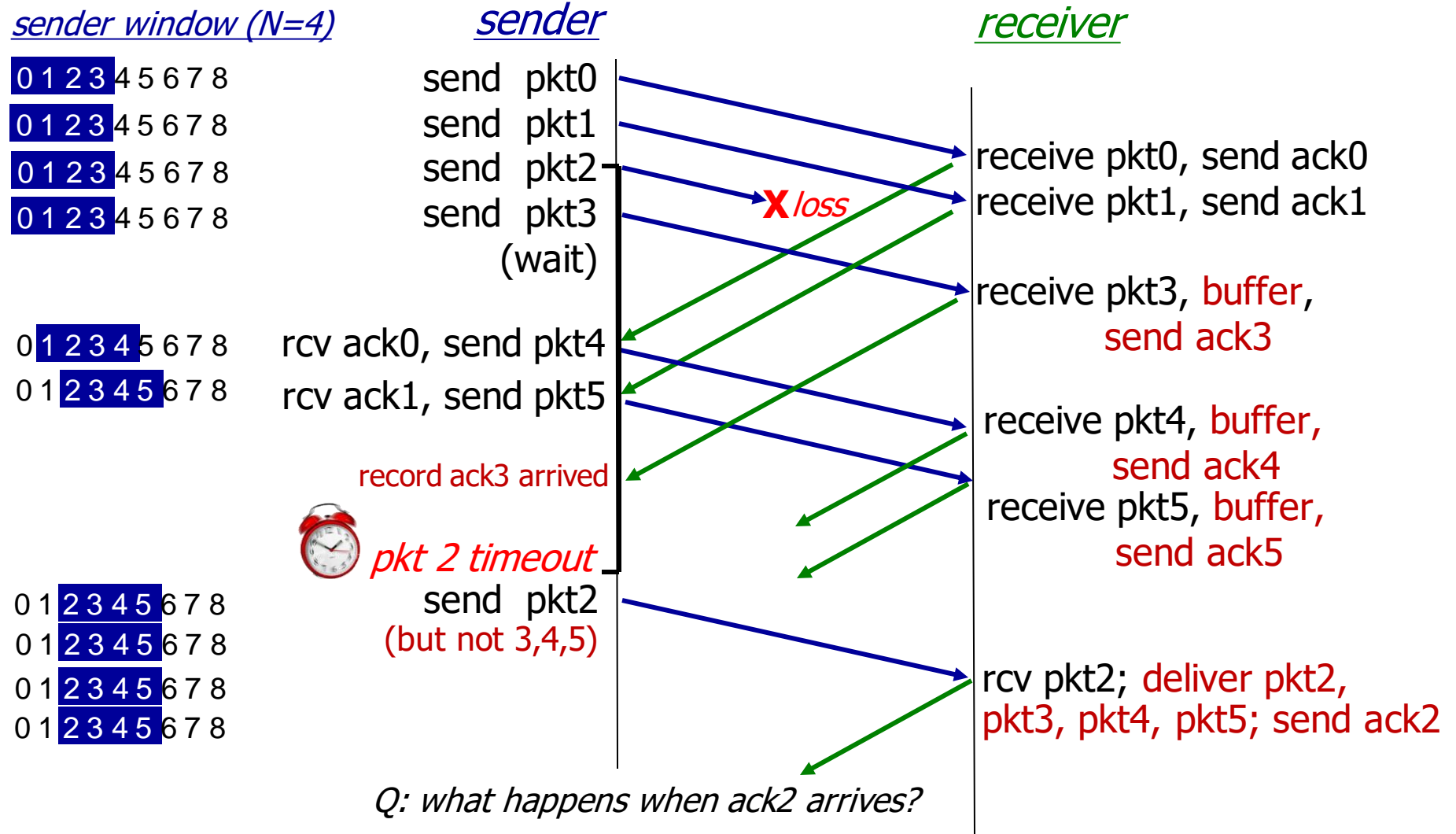
packet n in [rcvbase-N,rcvbase-1]

- ACK(n)

otherwise:

- ignore

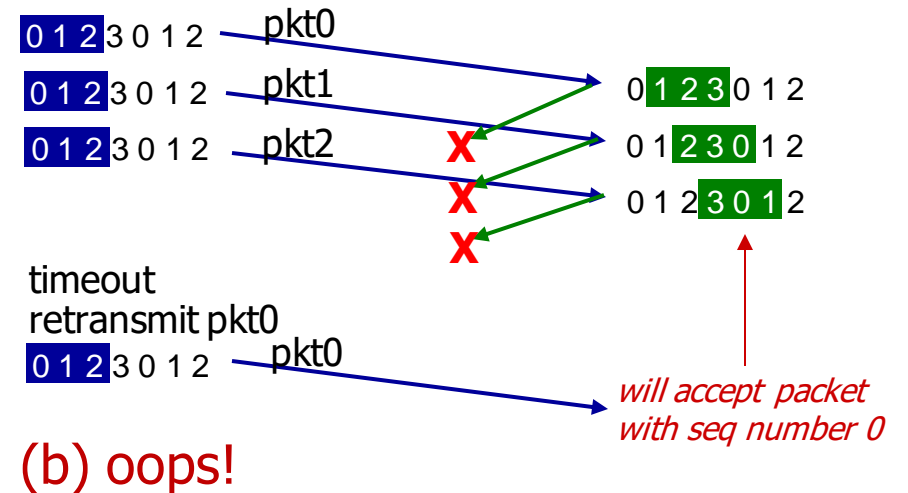
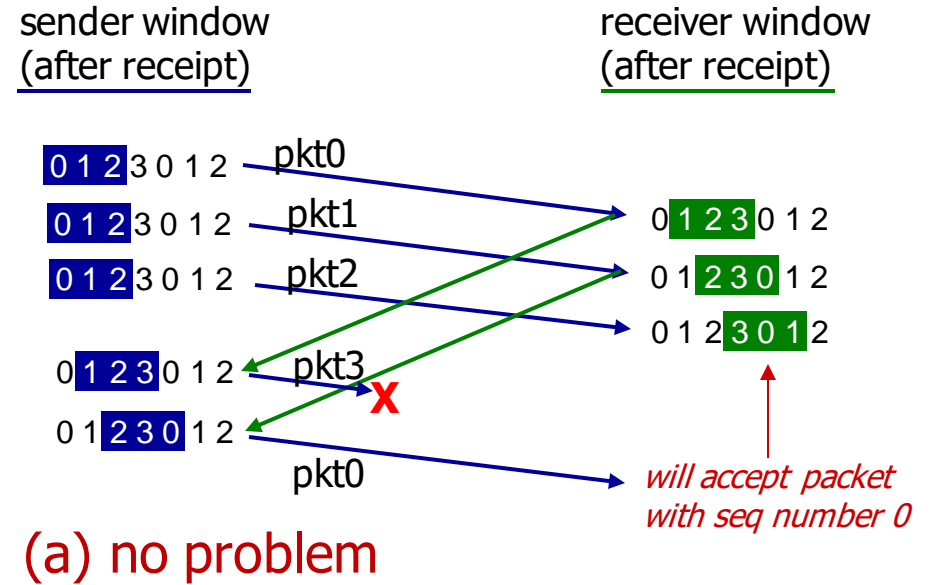
Selective Repeat in action



Selective repeat: a dilemma!

example:

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

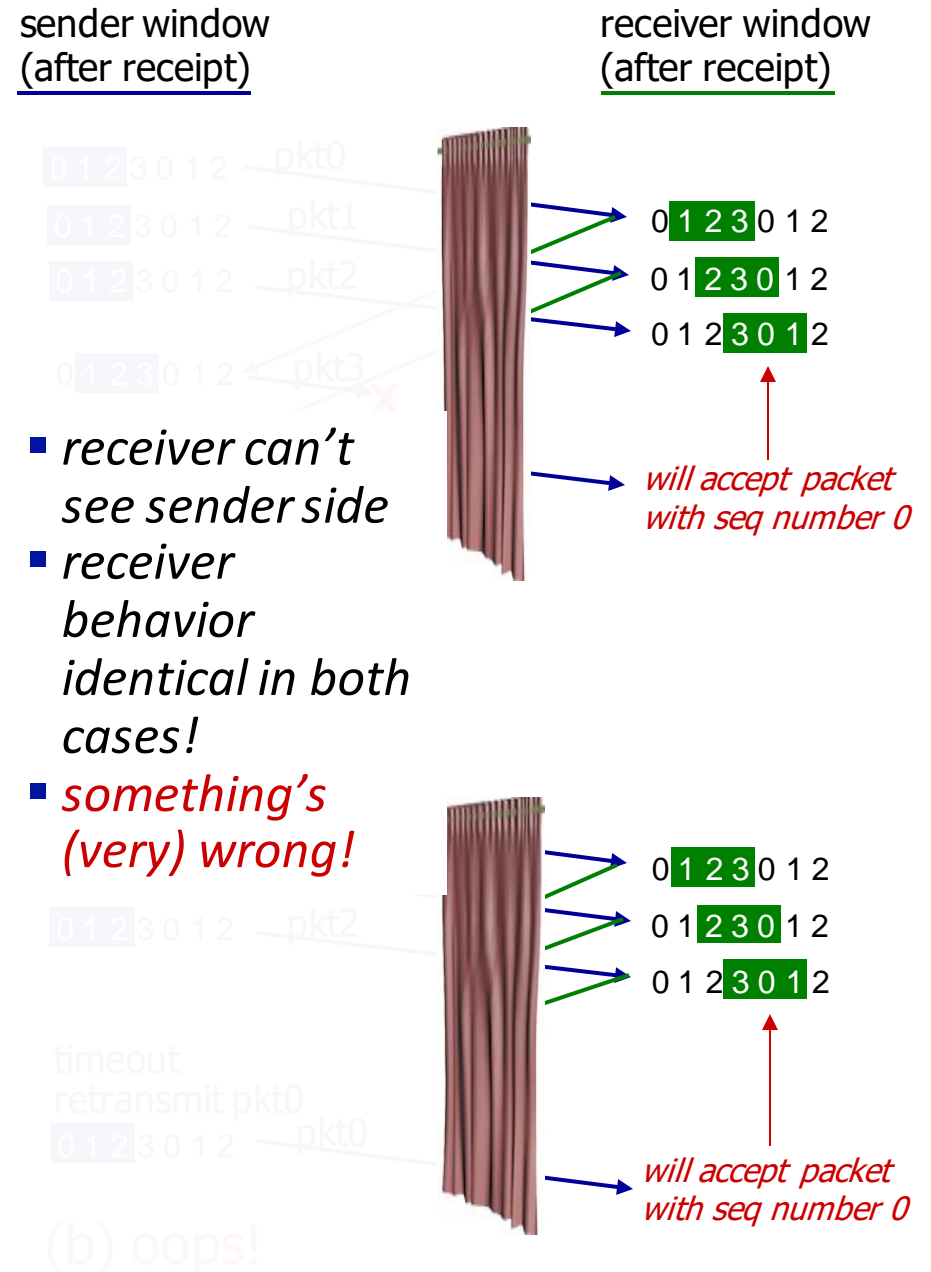


Selective repeat: a dilemma!

example:

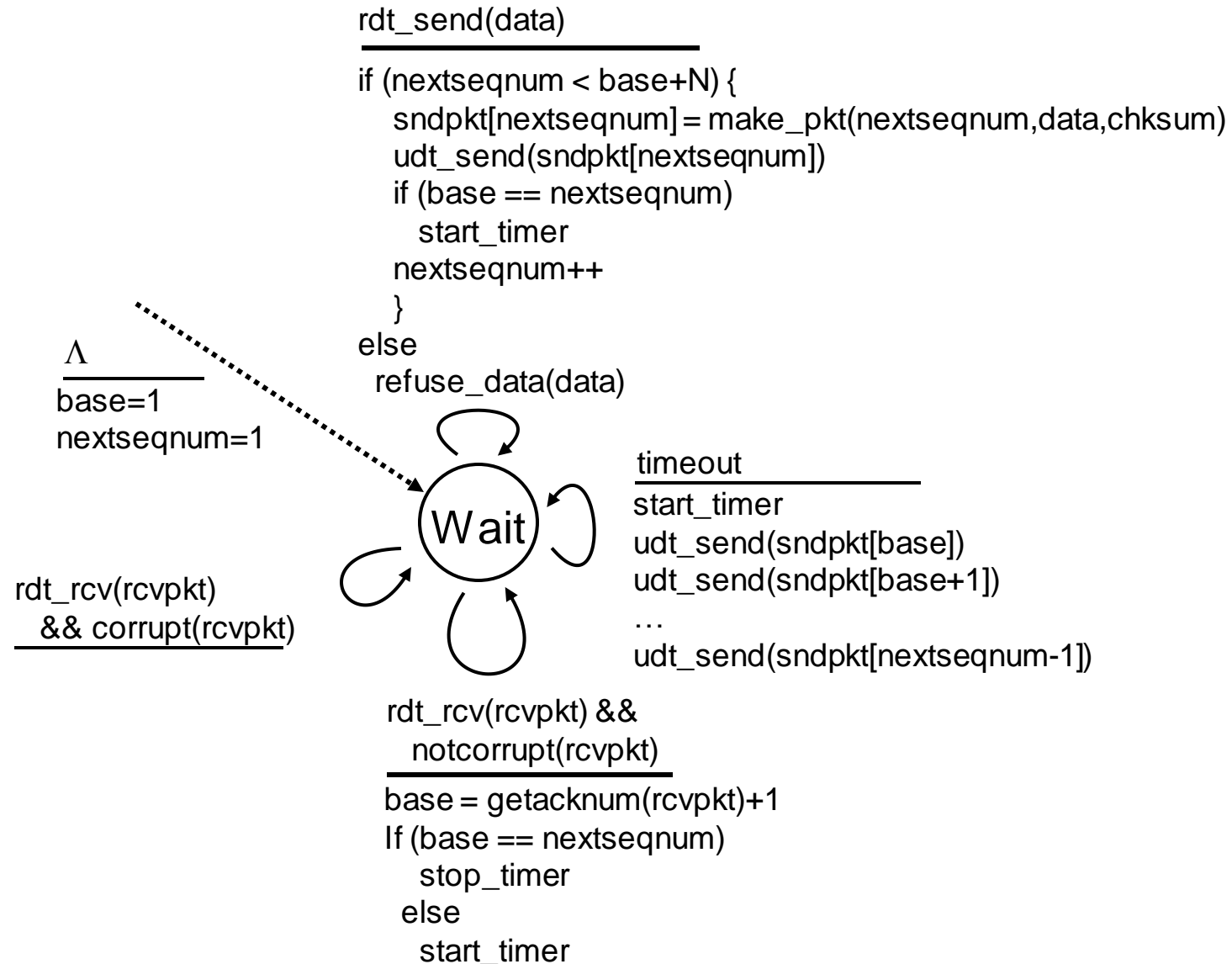
- seq #s: 0, 1, 2, 3 (base 4 counting)
- window size=3

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

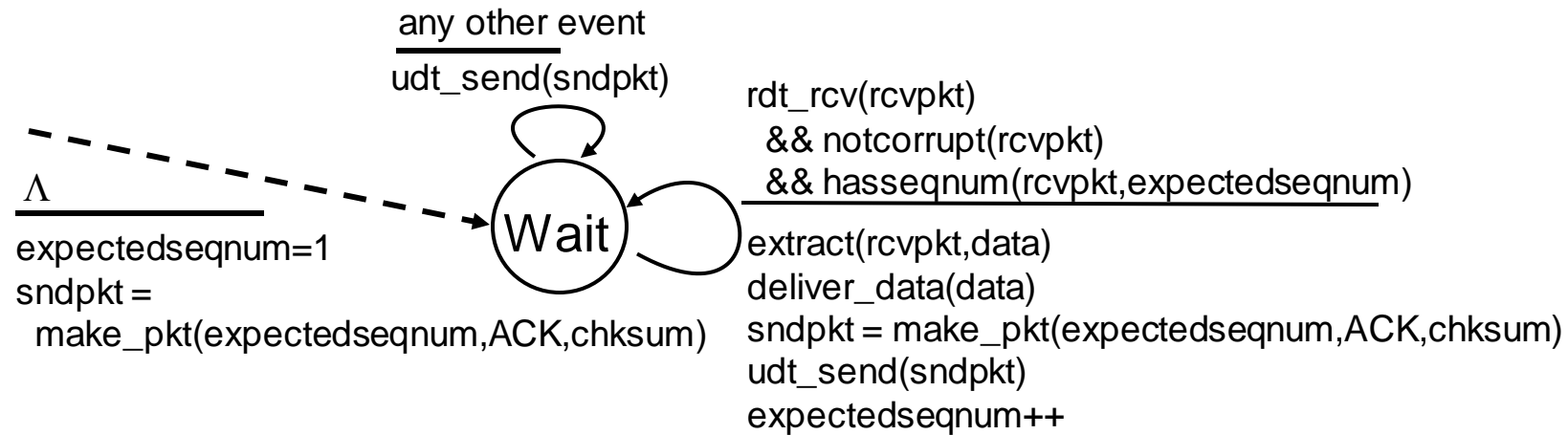


Additional Chapter 3 slides

Go-Back-N: sender extended FSM



Go-Back-N: receiver extended FSM



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

- may generate duplicate ACKs
- need only remember **expectedseqnum**
- out-of-order packet:
 - discard (don't buffer): *no receiver buffering!*
 - re-ACK pkt with highest in-order seq #