

Computer Networks

CS3001

(Section BDS-7A)

Lecture 12

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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



UDP: User Datagram Protocol

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

Why is there a UDP?

- no connection establishment (which can add RTT delay)
- simple: no connection state at sender, receiver
- small header size
- no congestion control
 - UDP can blast away as fast as desired!
 - can function in the face of congestion

UDP: User Datagram Protocol

- UDP use:
 - streaming multimedia apps (loss tolerant, rate sensitive)
 - DNS
 - SNMP
 - HTTP/3
- if reliable transfer needed over UDP (e.g., HTTP/3):
 - add needed reliability at application layer
 - add congestion control at application layer

UDP: User Datagram Protocol [RFC 768]

INTERNET STANDARD

RFC 768

J. Postel

ISI

28 August 1980

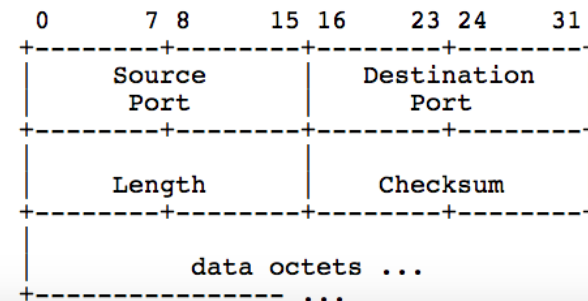
User Datagram Protocol

Introduction

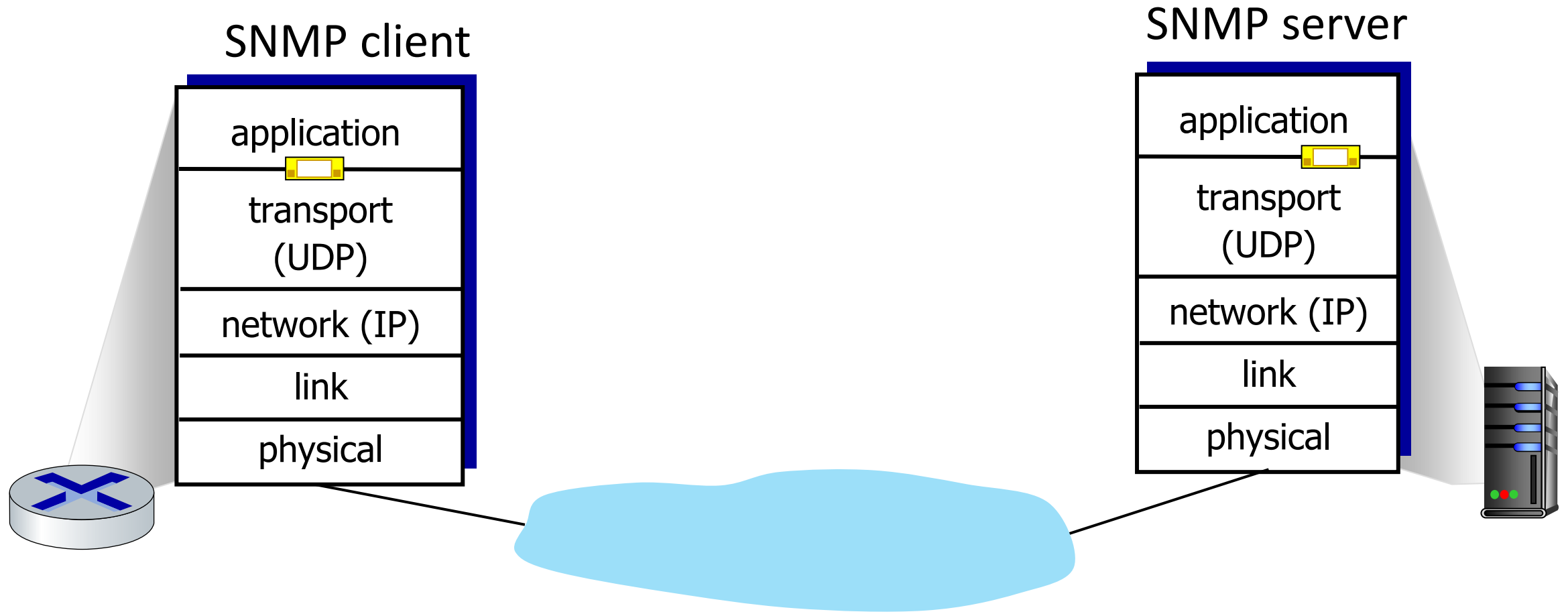
This User Datagram Protocol (UDP) is defined to make available a datagram mode of packet-switched computer communication in the environment of an interconnected set of computer networks. This protocol assumes that the Internet Protocol (IP) [1] is used as the underlying protocol.

This protocol provides a procedure for application programs to send messages to other programs with a minimum of protocol mechanism. The protocol is transaction oriented, and delivery and duplicate protection are not guaranteed. Applications requiring ordered reliable delivery of streams of data should use the Transmission Control Protocol (TCP) [2].

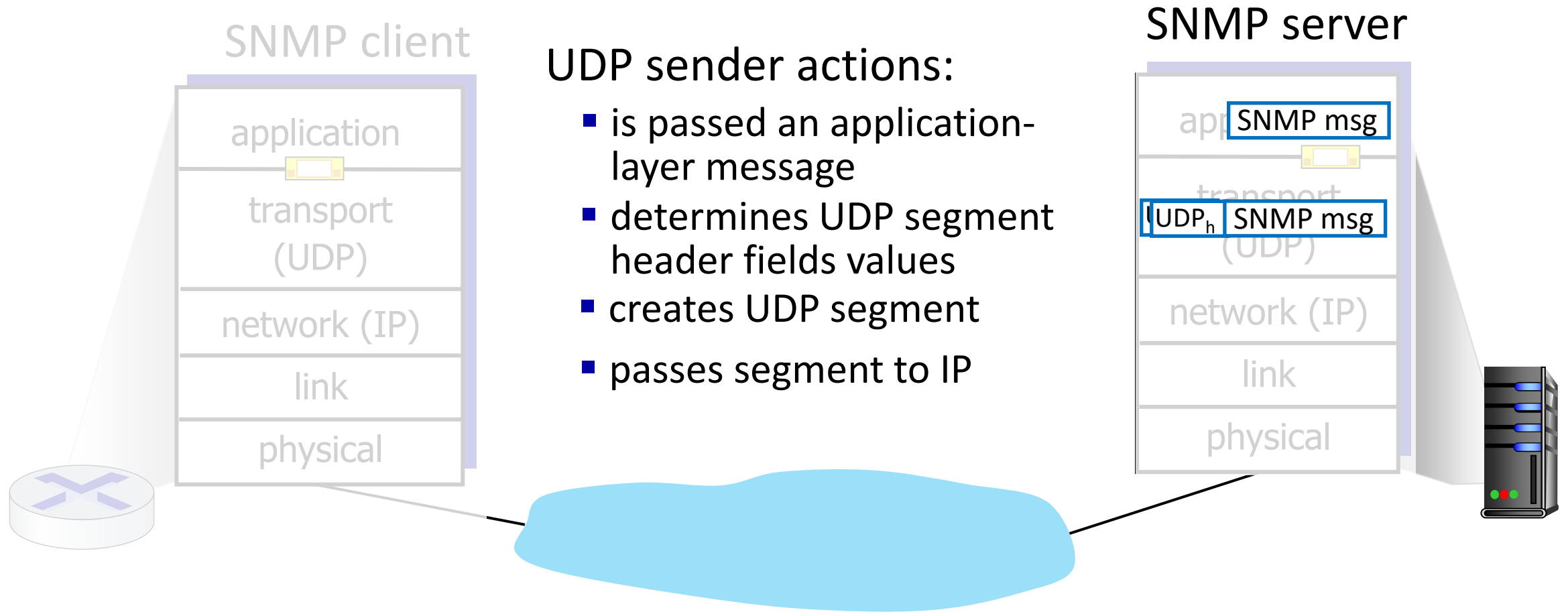
Format



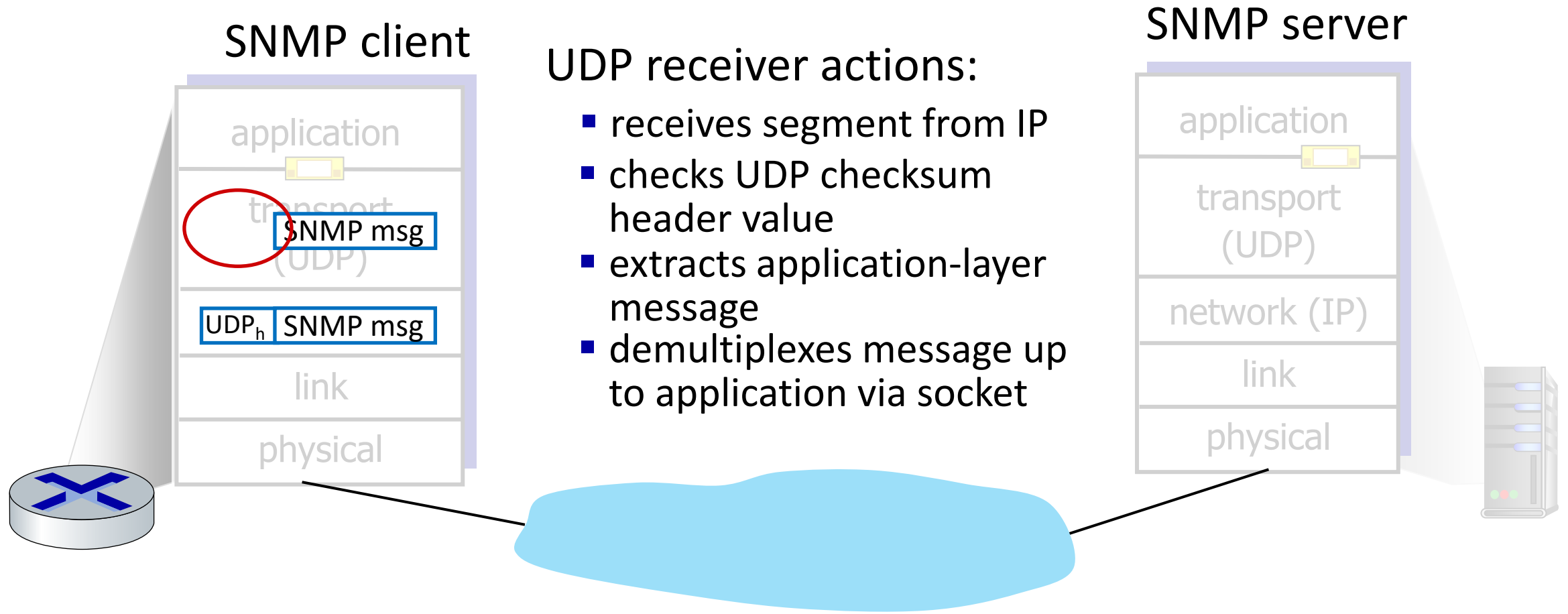
UDP: Transport Layer Actions



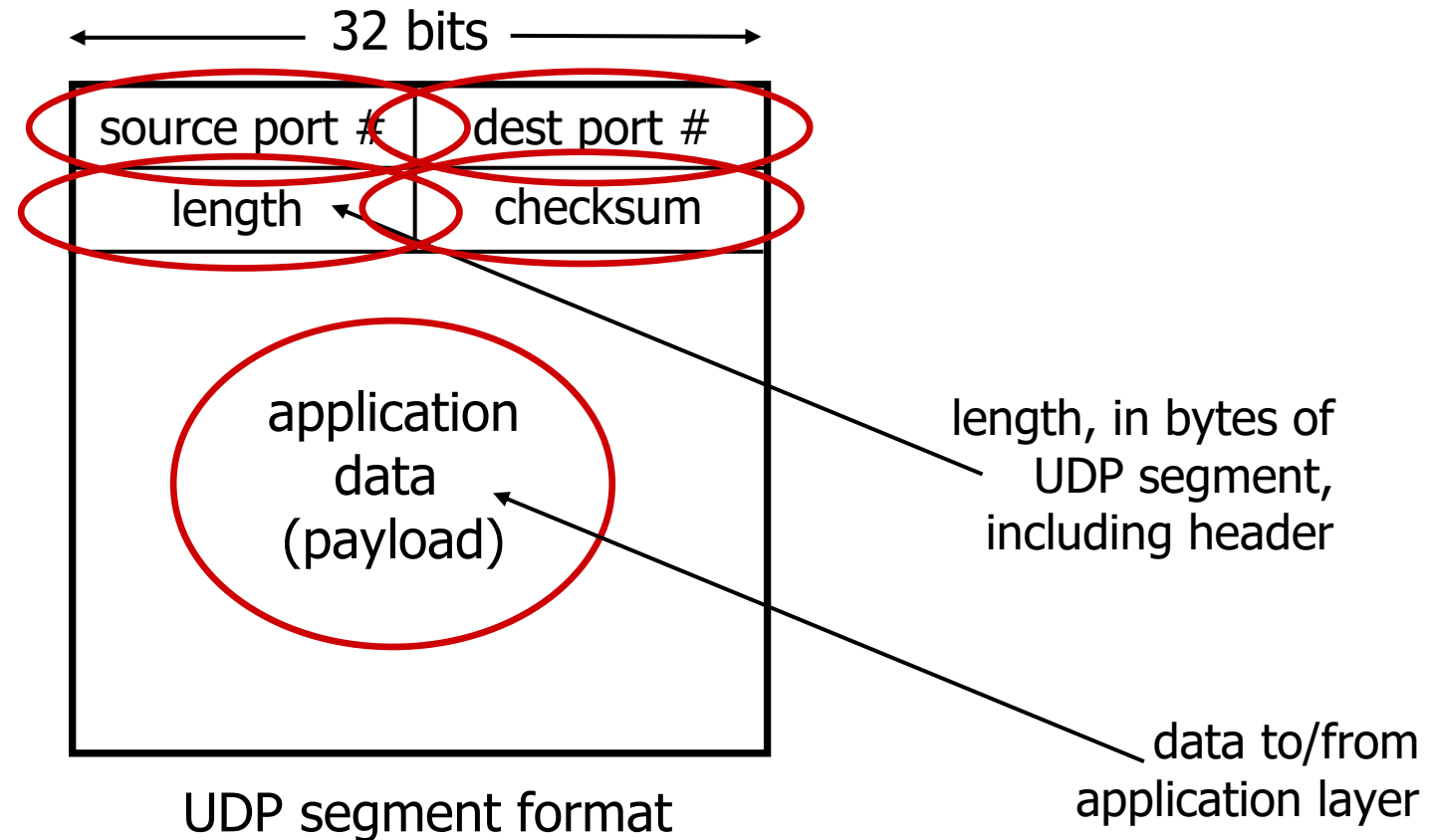
UDP: Transport Layer Actions



UDP: Transport Layer Actions

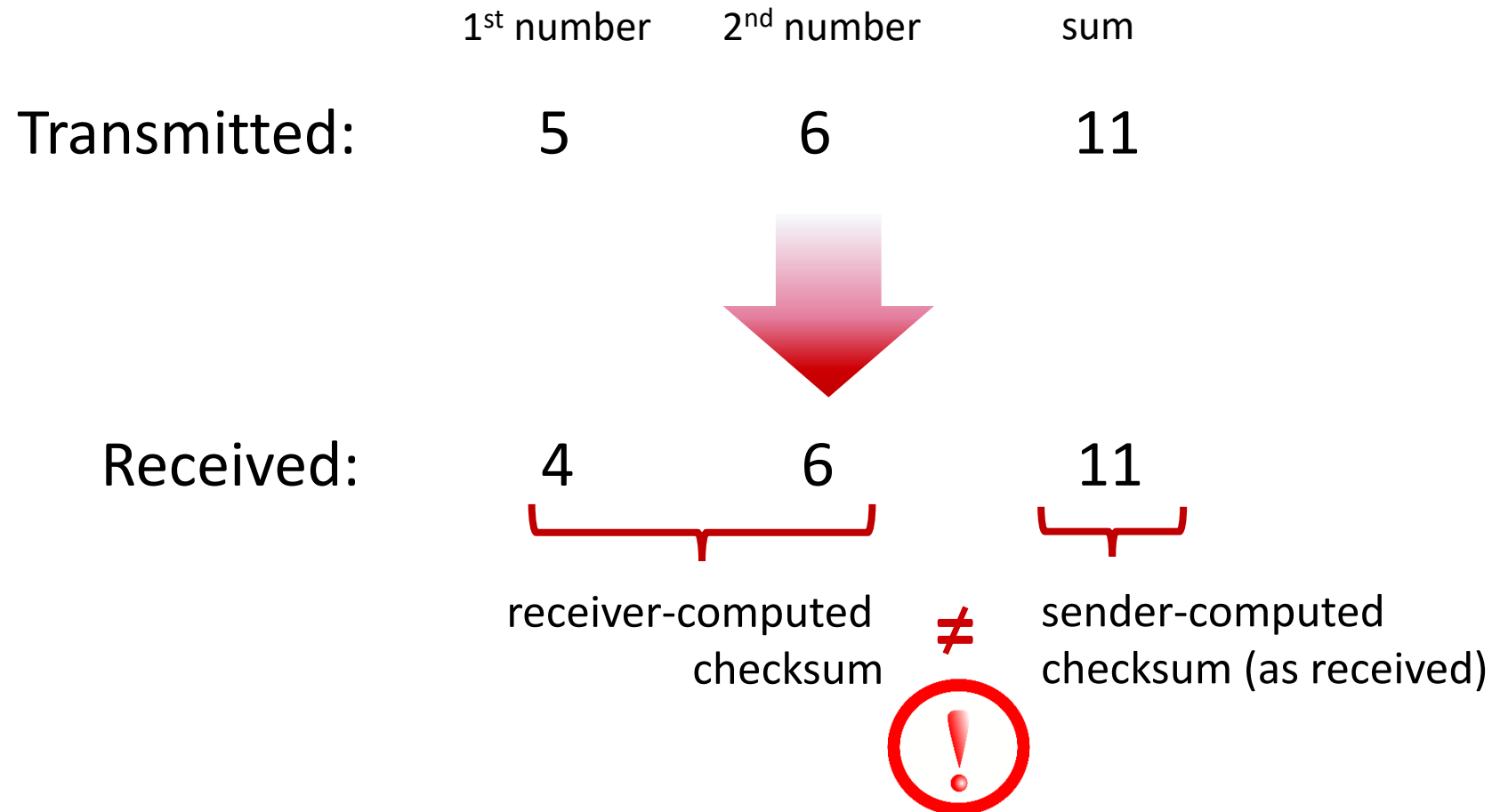


UDP segment header



UDP checksum

Goal: detect errors (*i.e.*, flipped bits) in transmitted segment



Internet checksum

Goal: detect errors (*i.e.*, flipped bits) in transmitted segment

sender:

- treat contents of UDP segment (including UDP header fields and IP addresses) as sequence of 16-bit integers
- **checksum:** addition (one's complement sum) of segment content
- checksum value put into UDP checksum field

receiver:

- compute checksum of received segment
- check if computed checksum equals checksum field value:
 - not equal - error detected
 - equal - no error detected. *But maybe errors nonetheless?* More later

Internet checksum: an 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	
<hr/>																	
wraparound	1	1	0	1	1	1	0	1	1	1	0	1	1	1	0	1	1
<hr/>																	
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	0	1	1

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

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

Internet checksum: weak protection!

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
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wraparound	1	1	0	1	1	1	0	1	1	1	0	1	1	1	0	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

Even though numbers have changed (bit flips), *no* change in checksum!

Summary: UDP

- “no frills” protocol:
 - segments may be lost, delivered out of order
 - best effort service: “send and hope for the best”
- UDP has its plusses:
 - no setup/handshaking needed (no RTT incurred)
 - can function when network service is compromised
 - helps with reliability (checksum)
- build additional functionality on top of UDP in application layer (e.g., HTTP/3)

Chapter 3: roadmap

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Principles of reliable data transfer

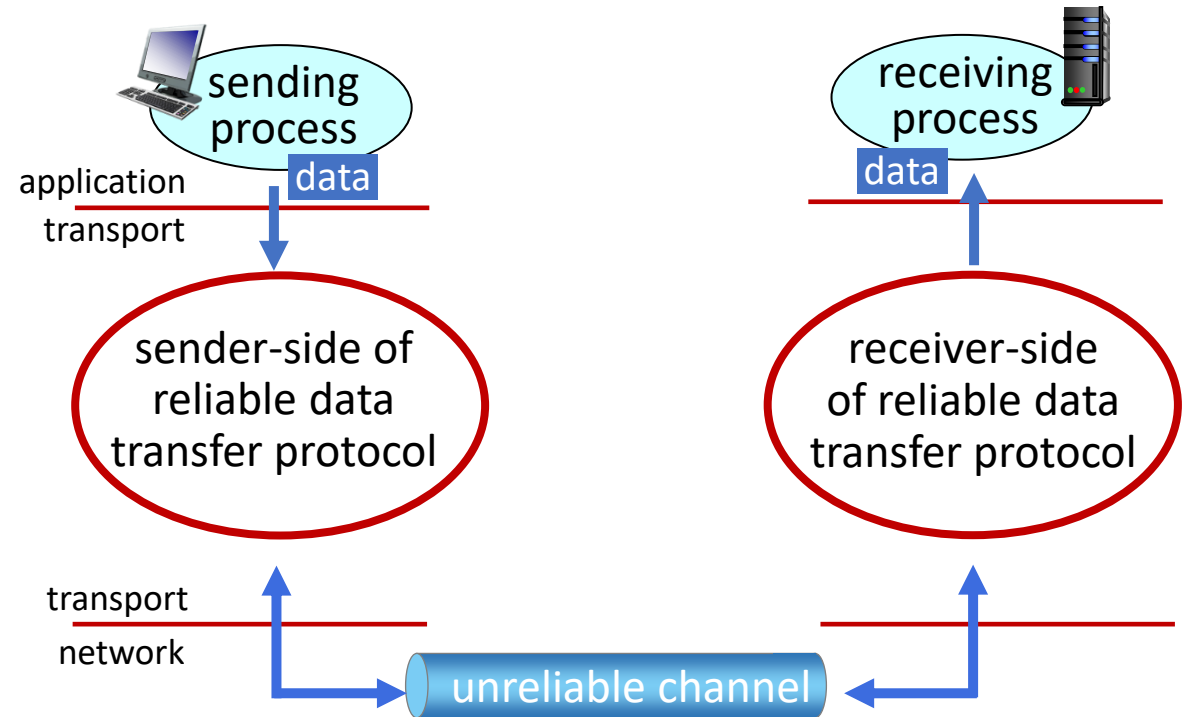
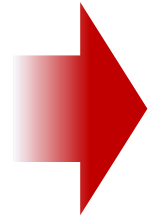


reliable service *abstraction*

Principles of reliable data transfer



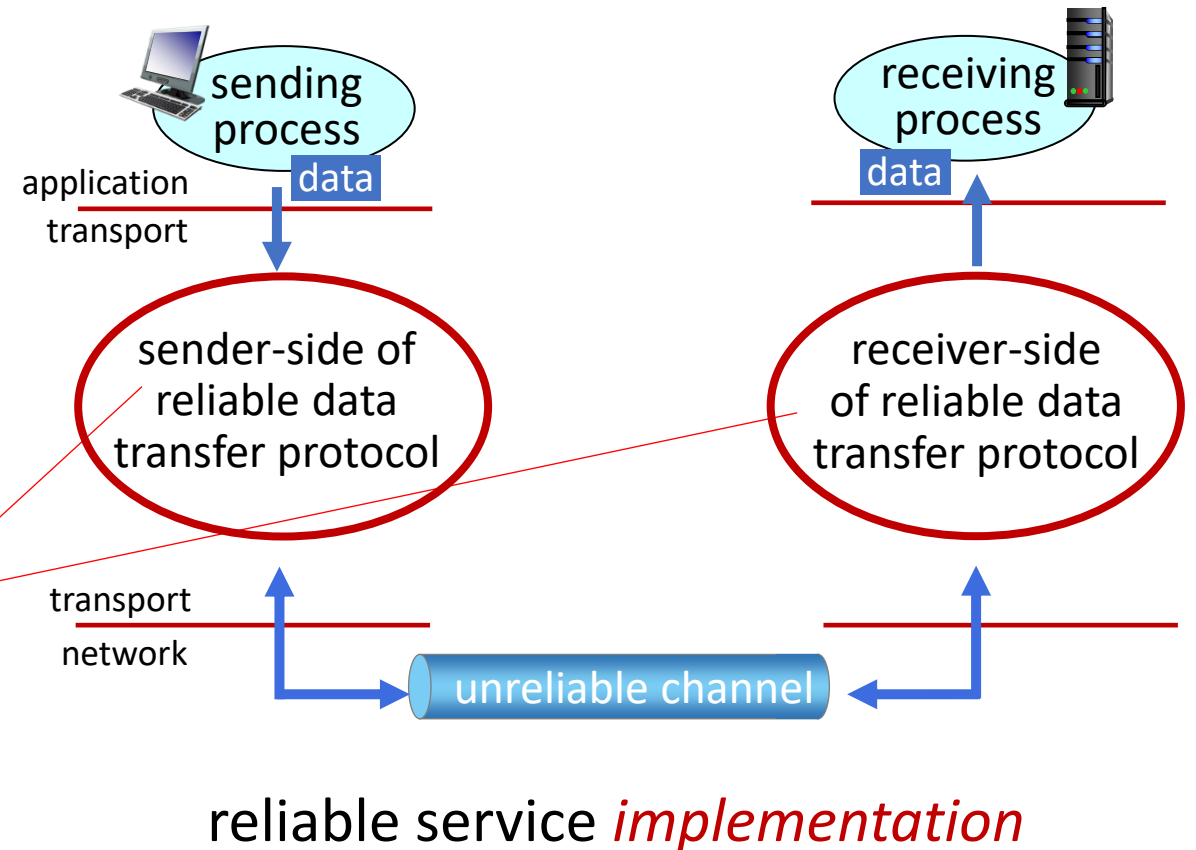
reliable service *abstraction*



reliable service *implementation*

Principles of reliable data transfer

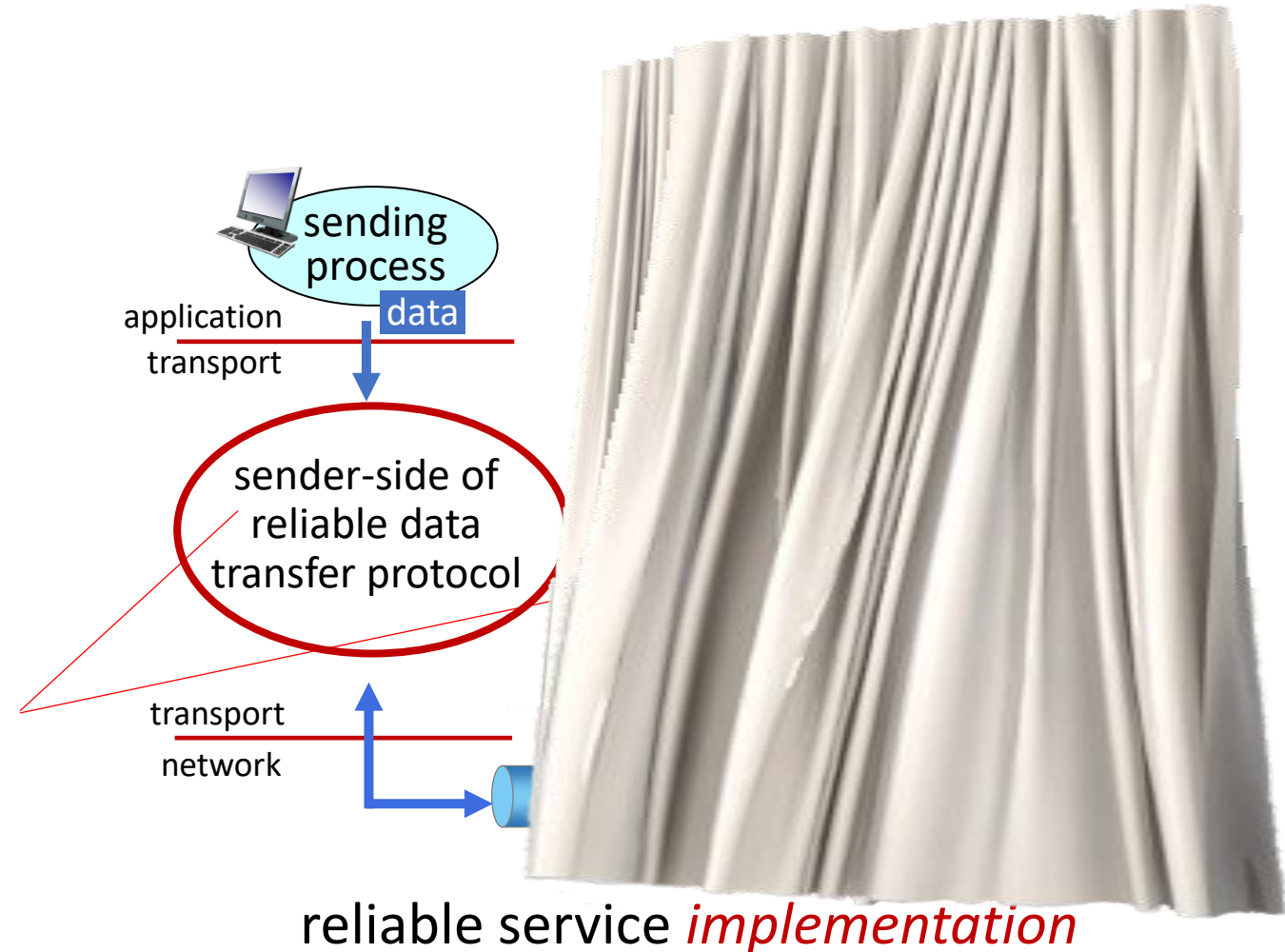
Complexity of reliable data transfer protocol will depend (strongly) on characteristics of unreliable channel (lose, corrupt, reorder data?)



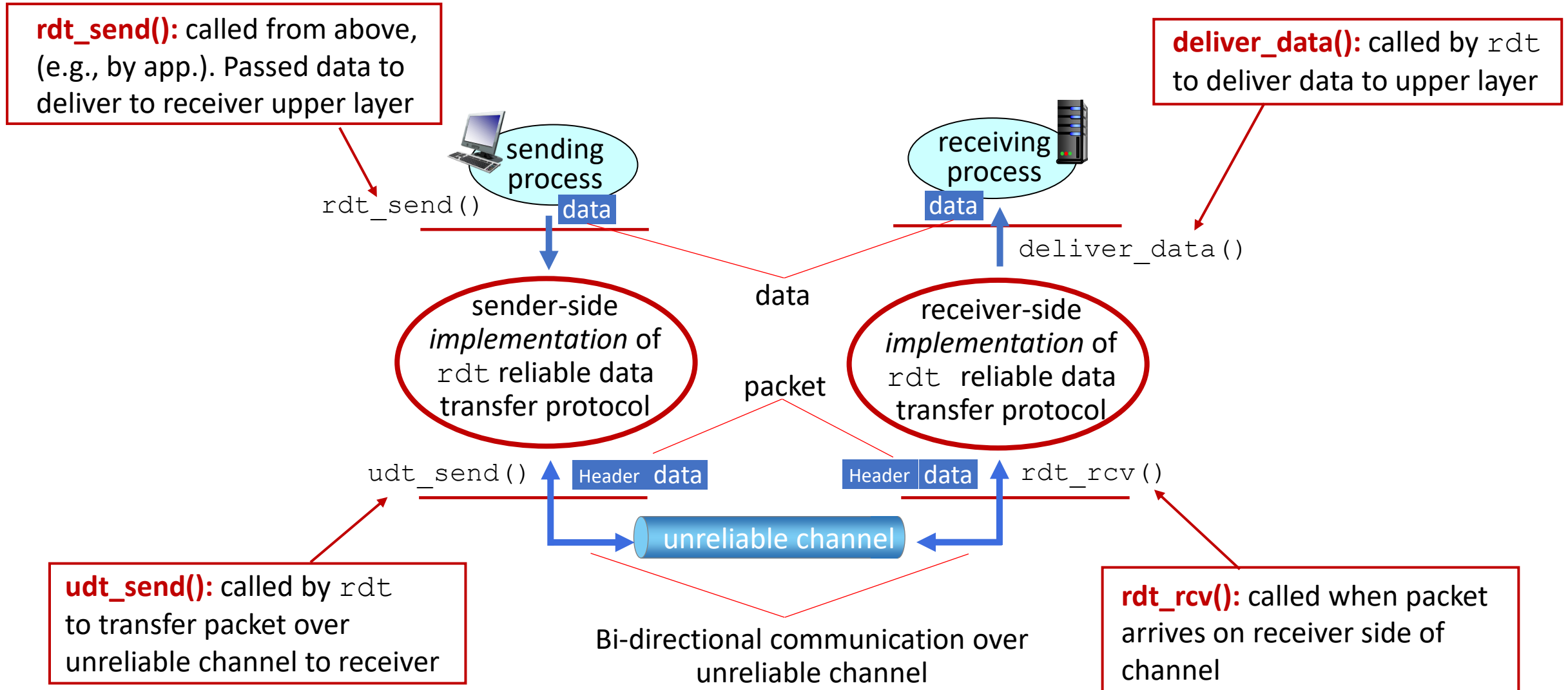
Principles of reliable data transfer

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

- unless communicated via a message



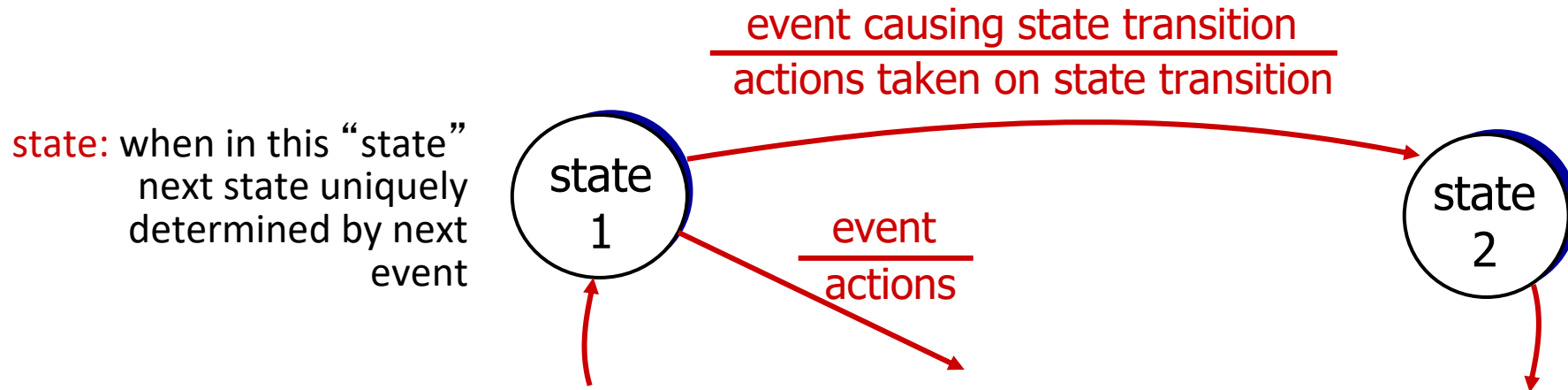
Reliable data transfer protocol (rdt): interfaces



Reliable data transfer: getting started

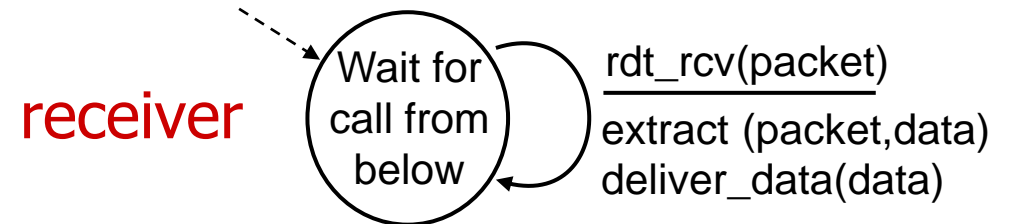
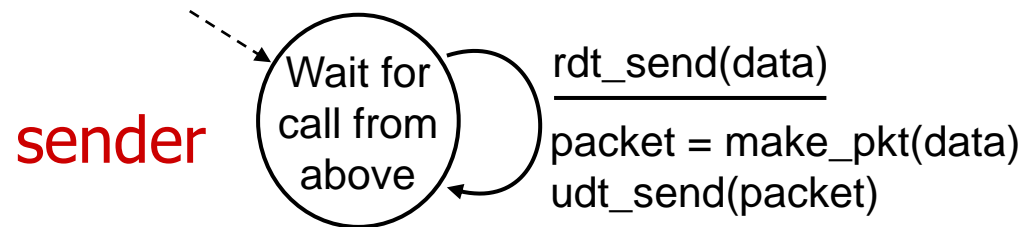
We will:

- 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?

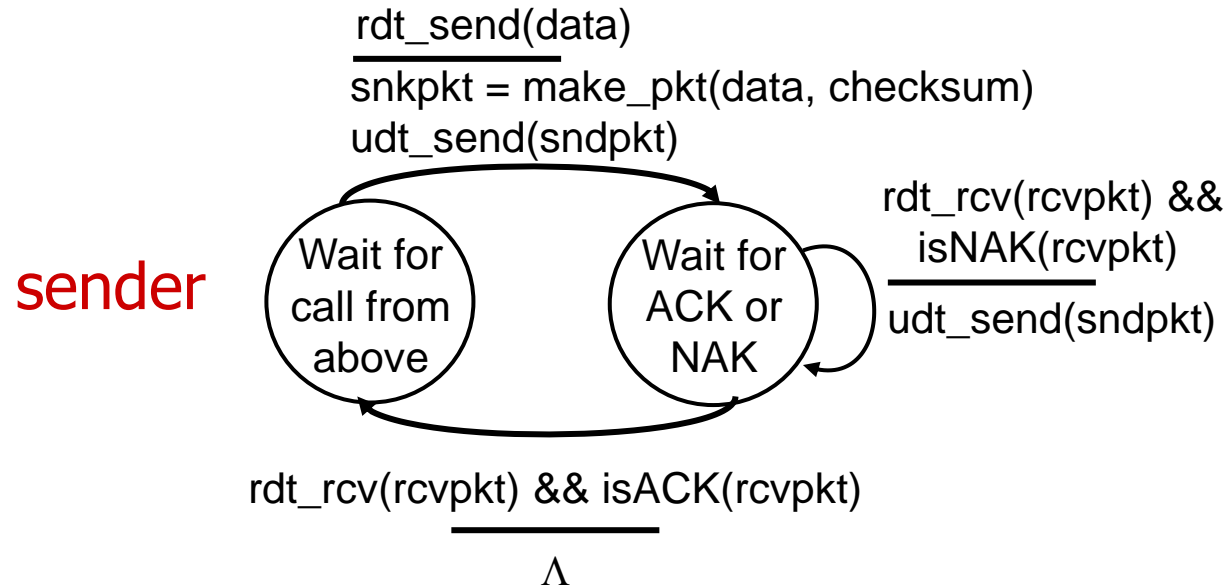
rdt2.0: channel with bit errors

- 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

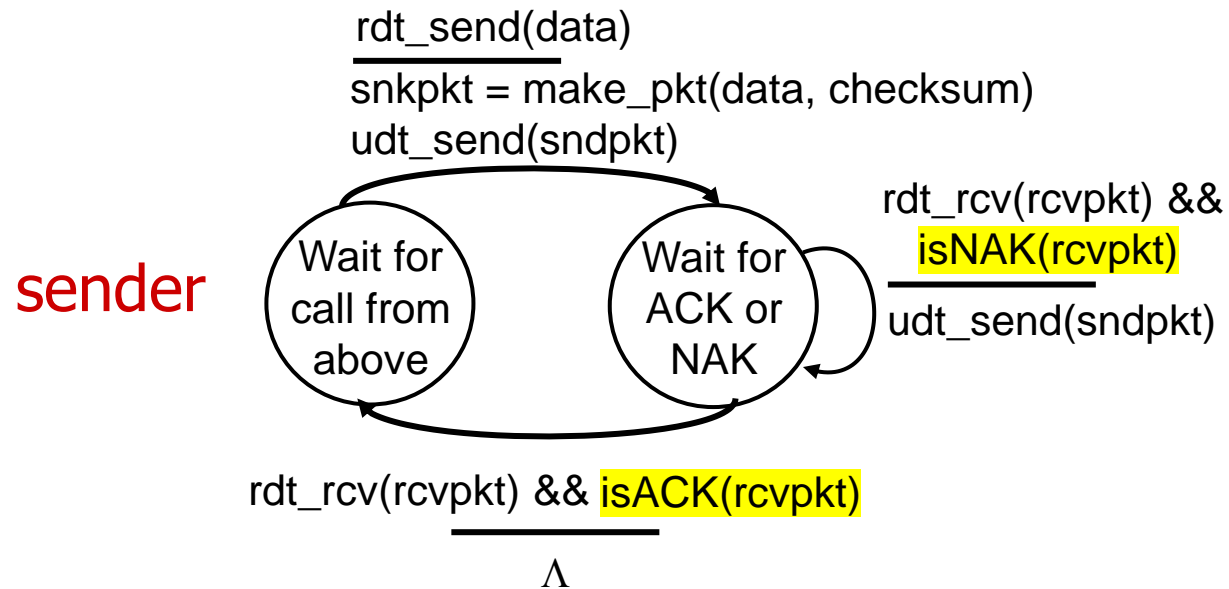
stop and wait

sender sends one packet, then waits for receiver response

rdt2.0: FSM specifications

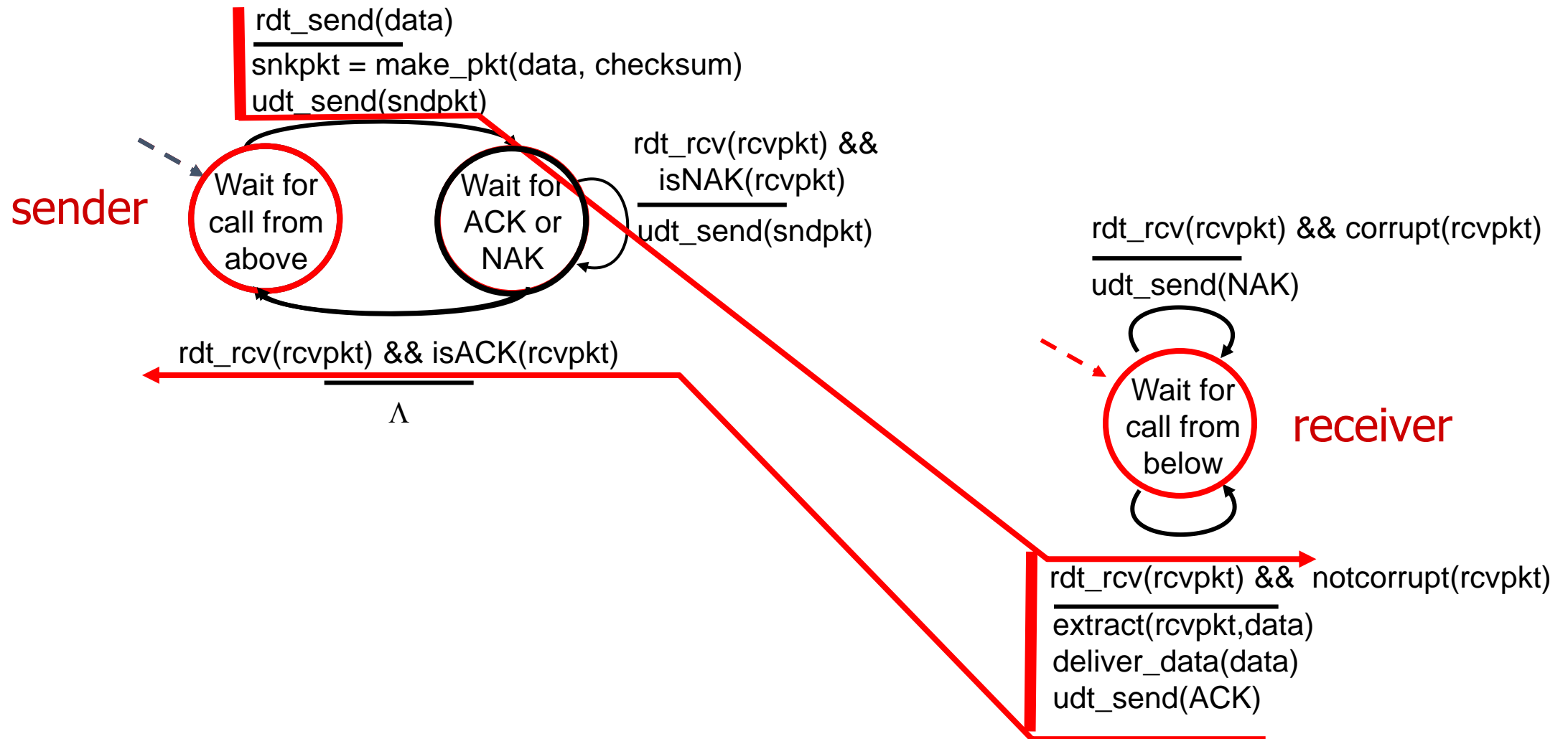


rdt2.0: FSM specification

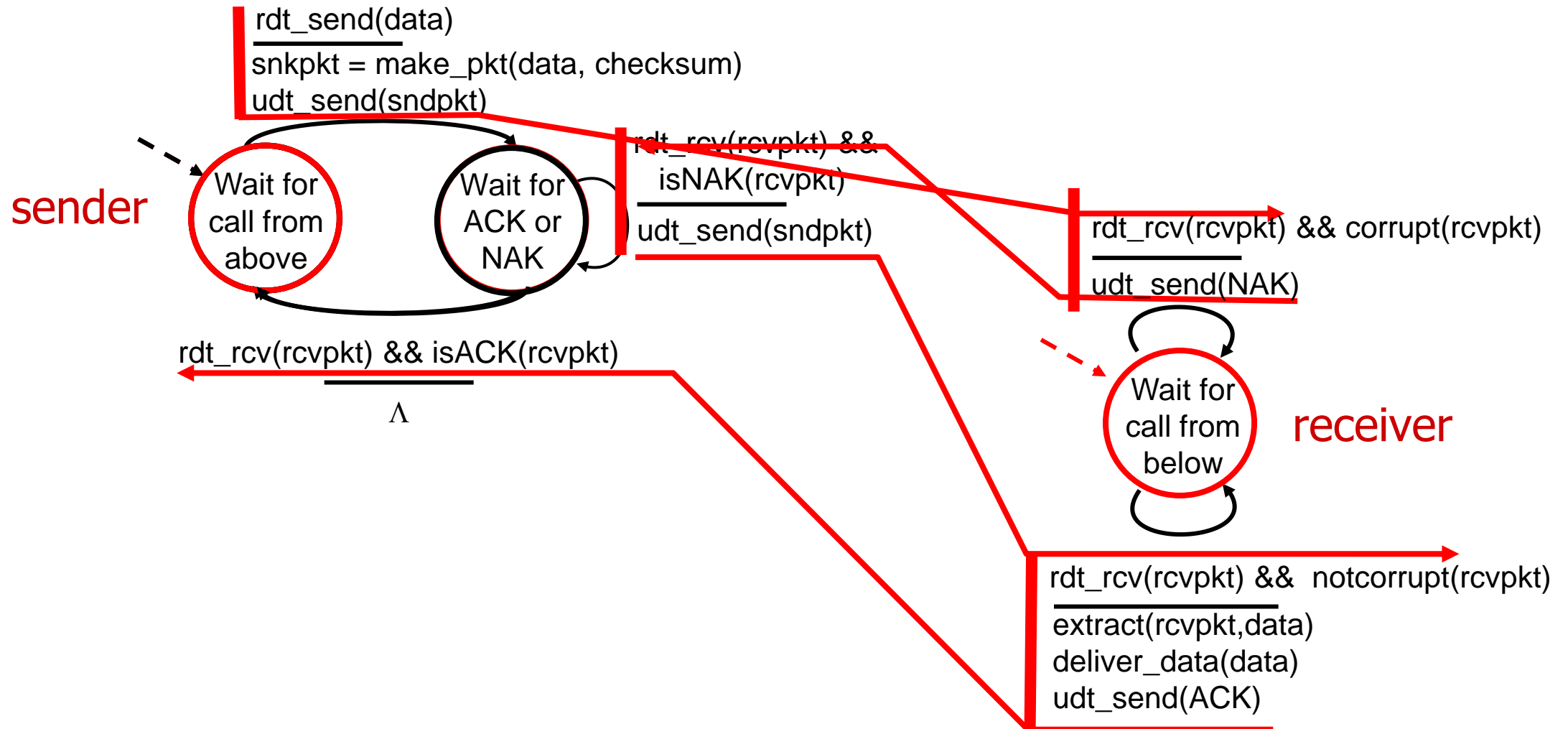


- Note:** “state” of receiver (did the receiver get my message correctly?) isn’t known to sender unless somehow communicated from receiver to sender
- that’s why we need a protocol!

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

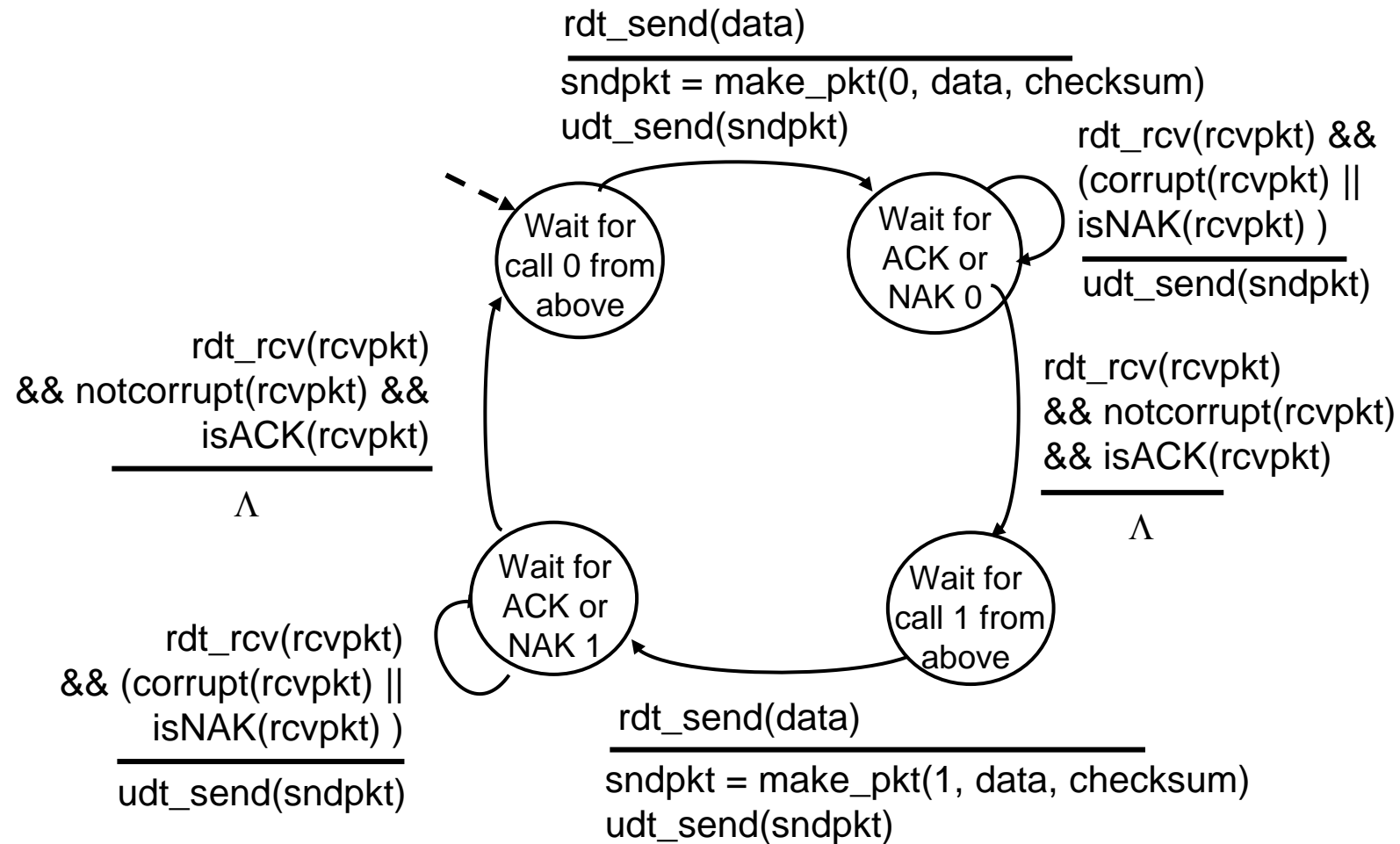
handling duplicates:

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

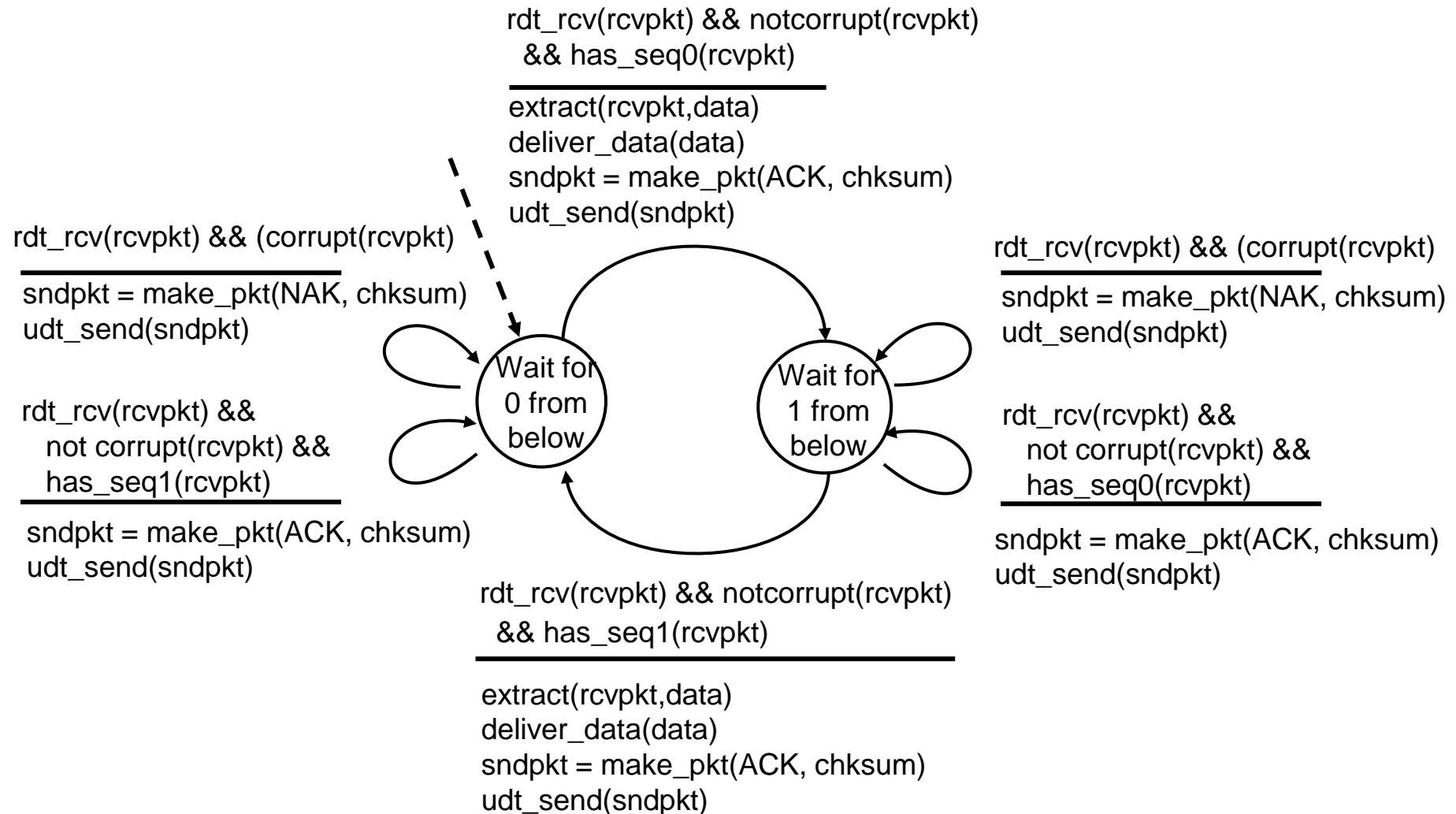
stop and wait

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