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Computer Networks (CS F303)

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innovate achieve lead

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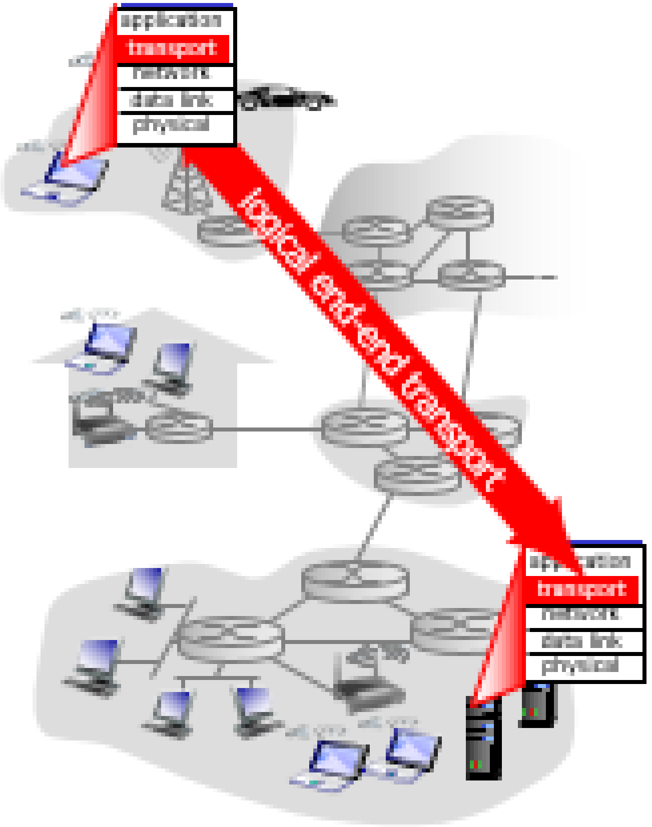
Second Semester 2020-2021
Module-3 <Transport layer>
Lecture: 10-13

Topics

- Transport Layer
  - Transport Layer Services
    - Multiplexing/Demultiplexing
      - Connectionless and Connection Oriented
        - » TCP and UDP
    - Reliable data transfer (Protocol design)
    - Flow control
    - Congestion control

Transport Layer Services and Protocols

- Provides logical communication between app processes
  - Apps processes sends msgs to each other using the logical communication
- Extend **host-to-host** delivery to **process-to-process** delivery



TP Layer vs. Network Layer

- Network layer: logical communication between hosts
- TP Layer: logical communication between processes
- TP layer services are constrained by the service model of underlying network-layer protocol
- But certain services can be offered by the TP layer even when the network layer doesn't offer
  - e.g., Reliable data transfer

Transport Layer Services

- Reliable in-order delivery (TCP)
  - Congestion control
  - Flow control
  - Connection setup
- Unreliable, unordered delivery (UDP)
  - Extension of “best-effort” IP

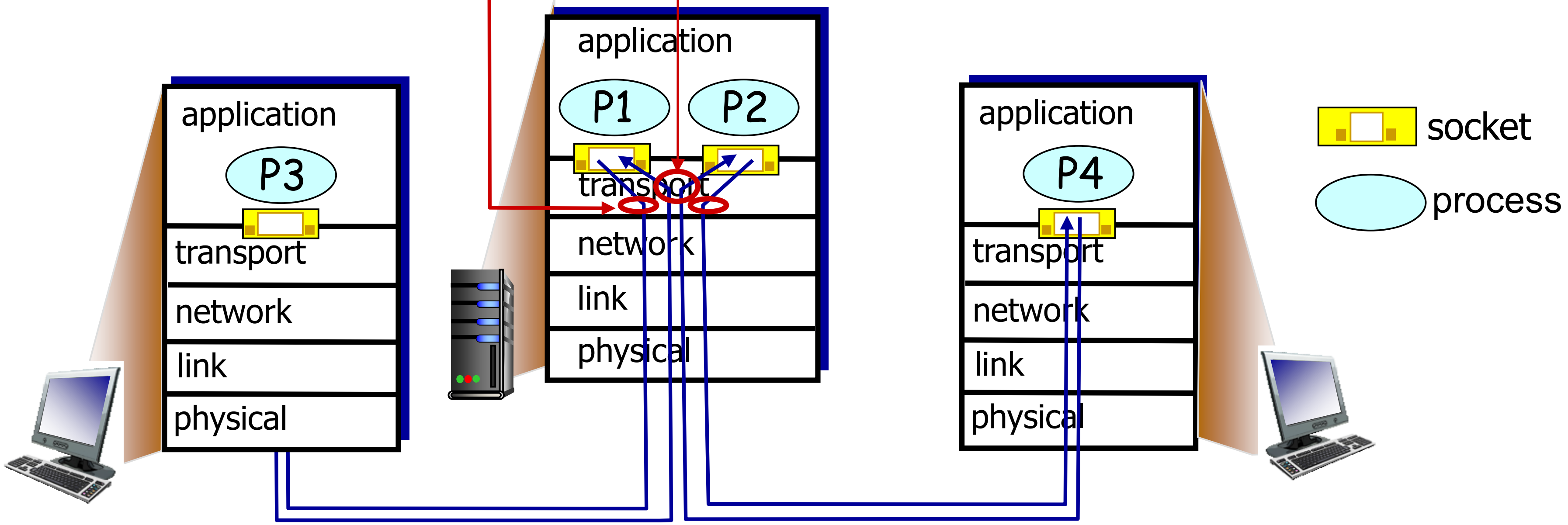
Process-to-Process Delivery Service

Multiplexing at sendening time:

handle data from multiple sockets, add transport **header**

Demux at receiving time:

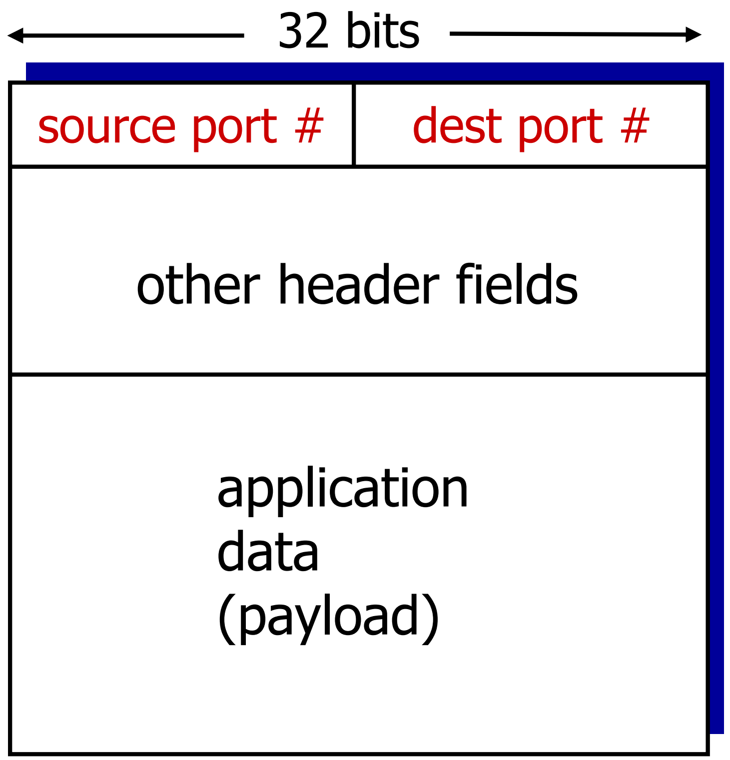
use **header** info to deliver received segments to correct socket



Demultiplexing at Receiver

- 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

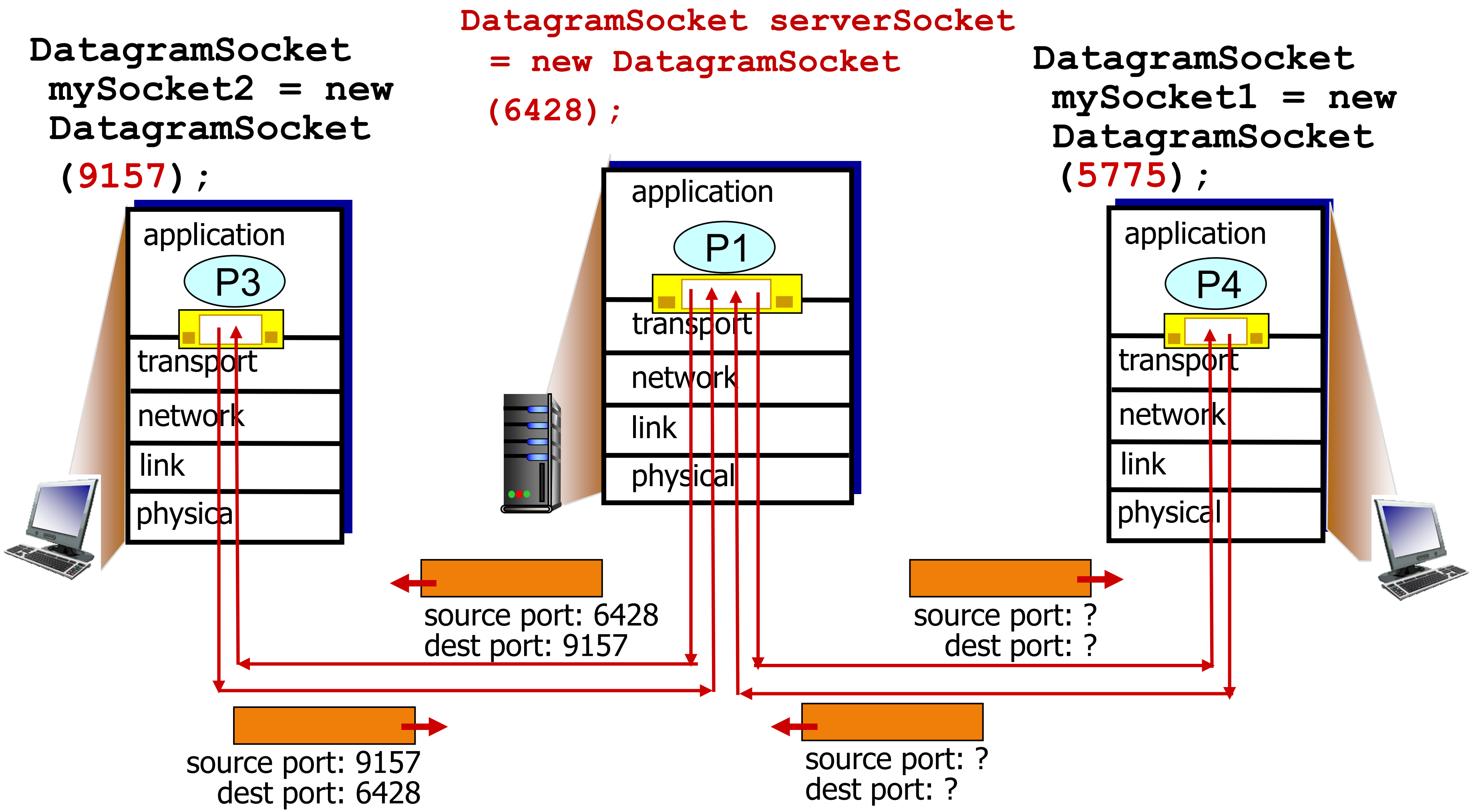
Connectionless (UDP) Demultiplexing

- When host receives UDP segment:
  - Checks destination **port #** in segment and directs segment to socket with **port #**

- *Recall:* when creating datagram to send into UDP socket, must specify
  - Destination IP address
  - Destination port #

- Important to note that
  - IP datagrams with *same destination* ***port #,*** but different source IP addresses and/or source port numbers will be directed to *same socket* at destination

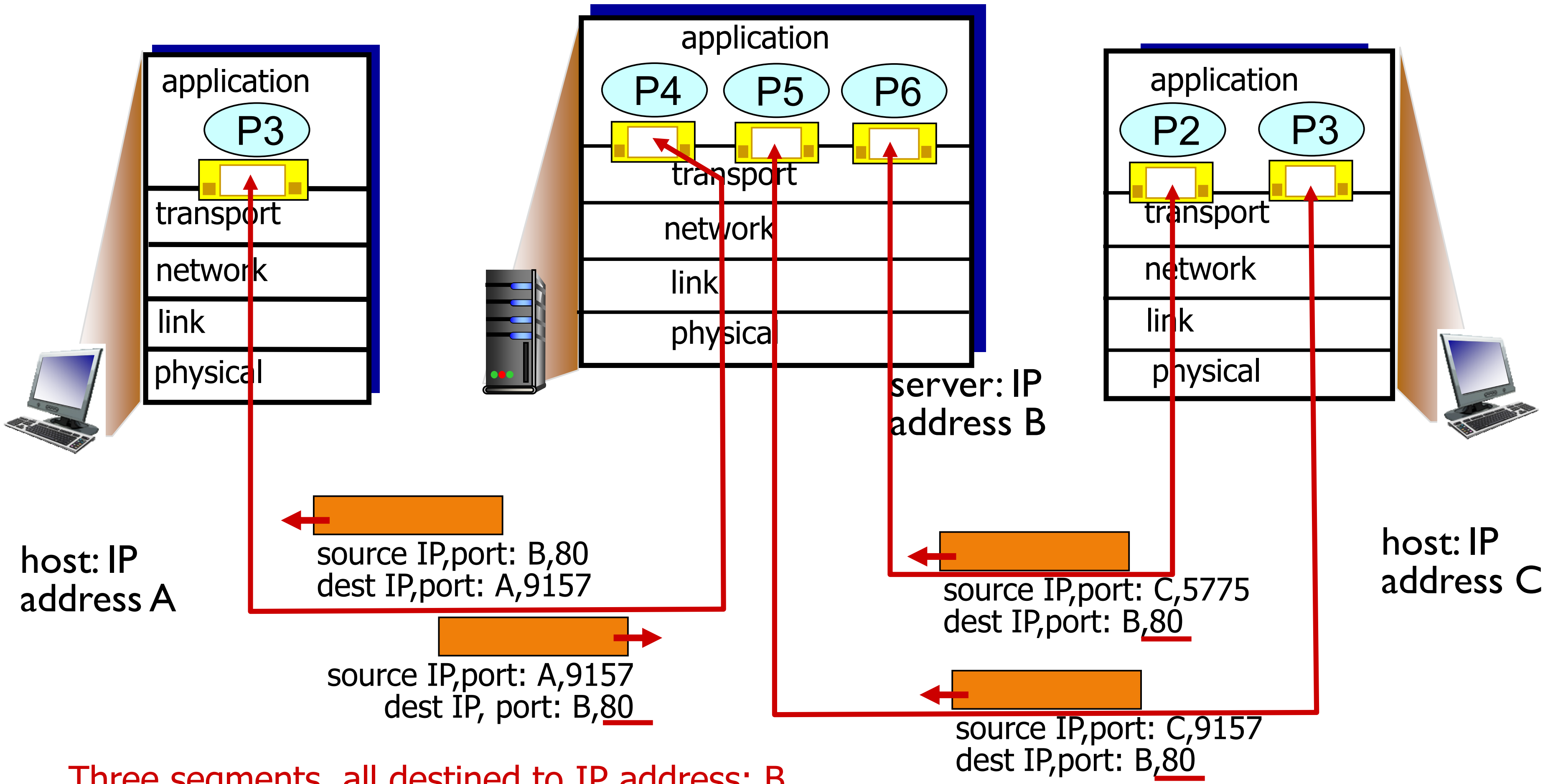
Example: Connectionless Demultiplexing



Connection Oriented Demultiplexing

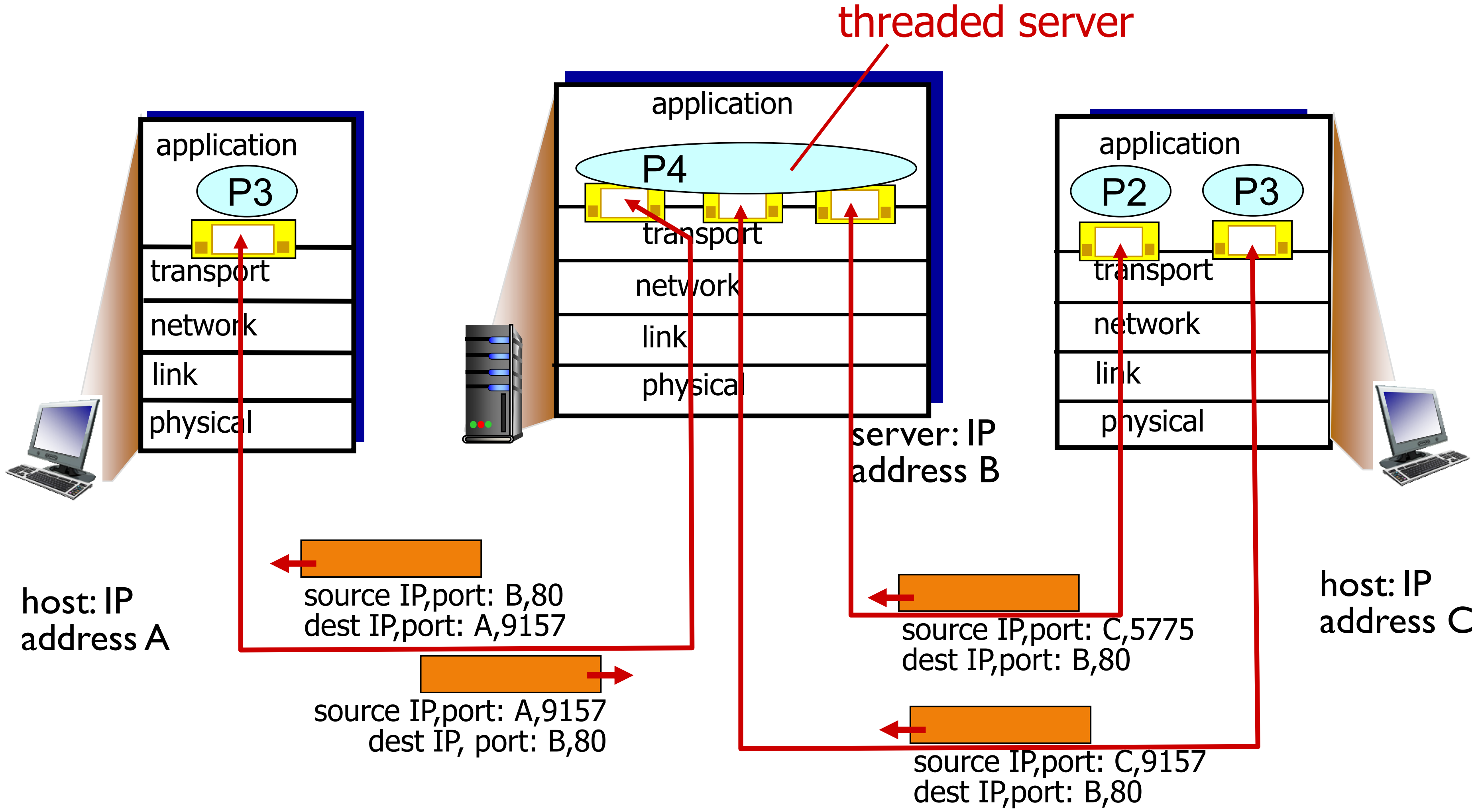
- TCP socket identified by 4-tuple:
  - Source IP address, source port #, dest IP address, dest port #
  - Demux: receiver uses all four values to direct segment to appropriate socket
- Server host may support many simultaneous TCP sockets:
  - Each socket identified by its own 4-tuple
- Web servers have different sockets for each connecting client
  - e.g., non-persistent HTTP will have different socket for each request

Example: Connection Oriented Demux



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

Example

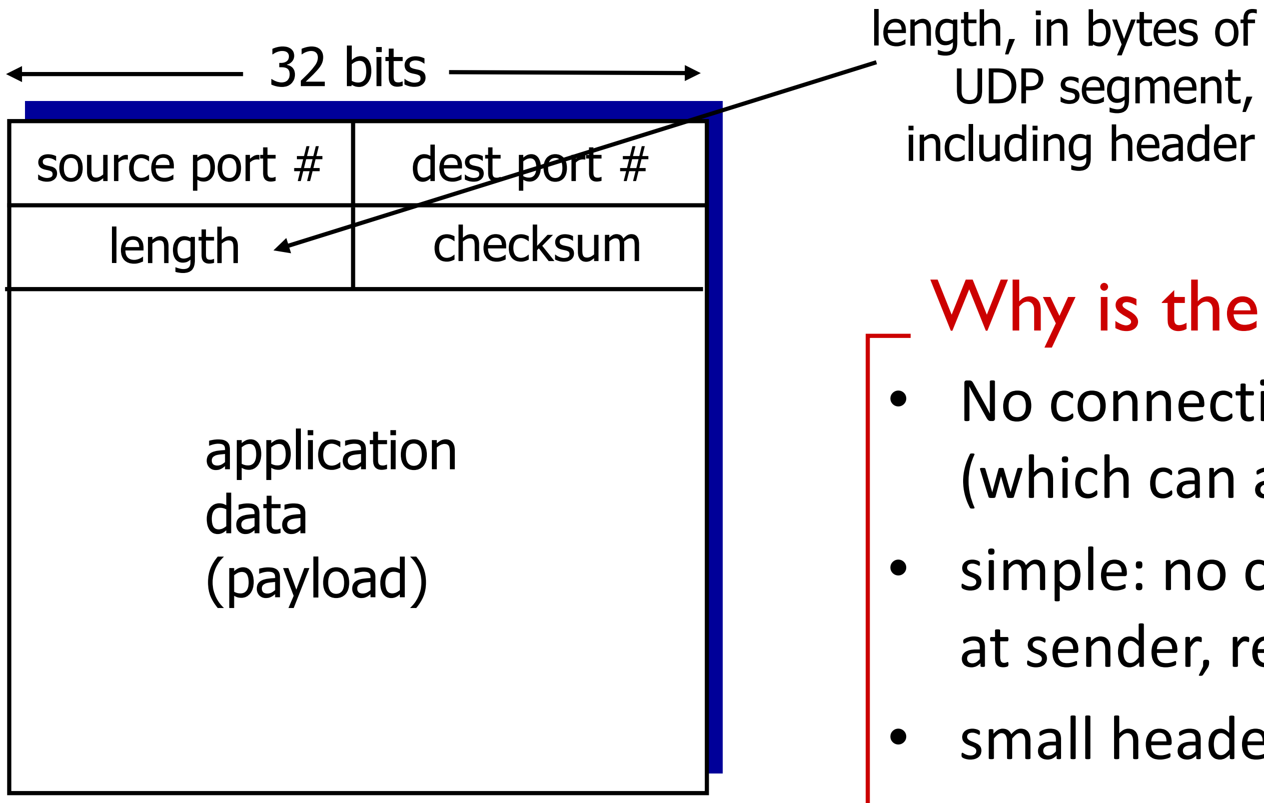

threaded server
application
P3
transport
network
link
physical
application
P4
transport
network
link
physical
server: IP
address B
application
P2
P3
transport
network
link
physical
host: IP
address A
source IP,port: B,80
dest IP,port: A,9157
source IP,port: A,9157
dest IP, port: B,80
source IP,port: C,5775
dest IP,port: B,80
source IP,port: C,9157
dest IP,port: B,80
host: IP
address C

User Datagram Protocol [RFC 768]

- Best effort service
  - UDP segment may lost, delivered out of order to app
- Connectionless
  - No handshaking between sender and receiver

- Each UDP segment handled independently of others

UDP Segment Header



UDP segment format

Why is there a UDP?

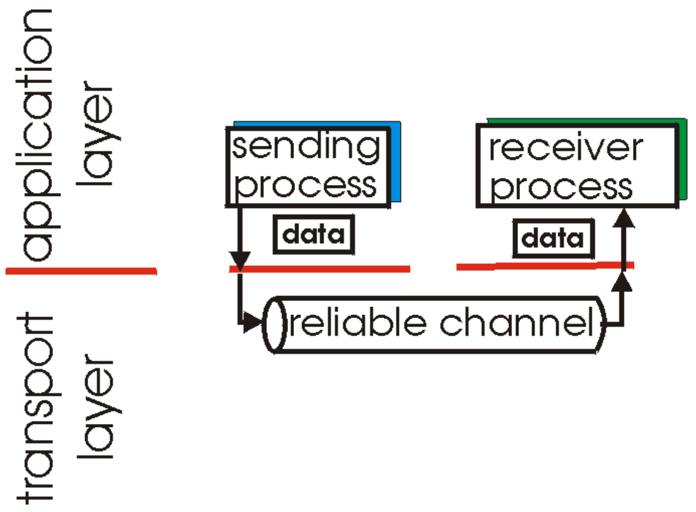
- No connection establishment (which can add delay)
- simple: no connection state at sender, receiver
- small header size
- no congestion control: UDP can blast away as fast as desired

UDP Checksum

- Treat segment contents (with header fields) as a sequence of 16-bit integers at sender
  - Sum all such 16-bit words in the segment
  - One’s complement of the sum is put in checksum field
- At the receiver, all 16-bit words are added (including checksum) to detect error in segment

Principles of Reliable Data Transfer

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



(a) provided service

Principles of Reliable Data Transfer

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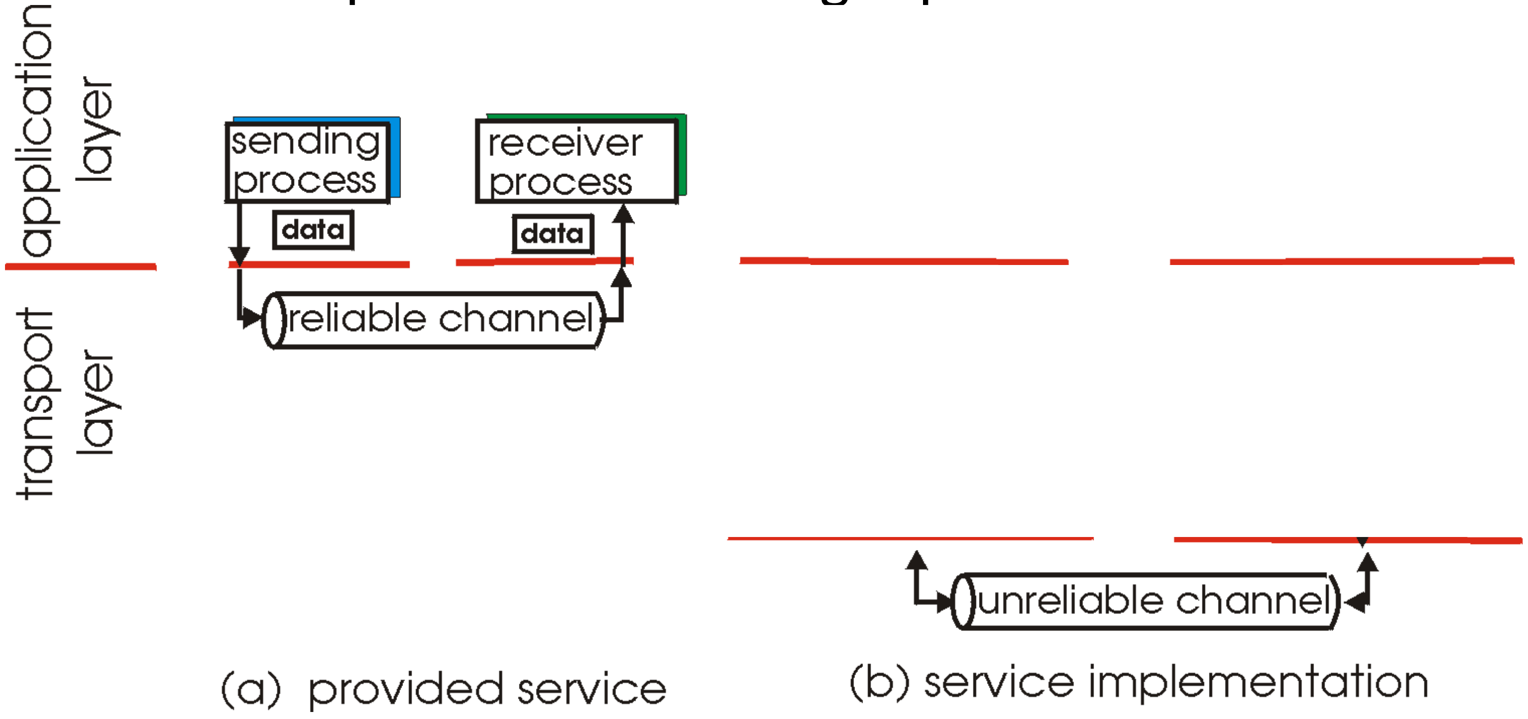
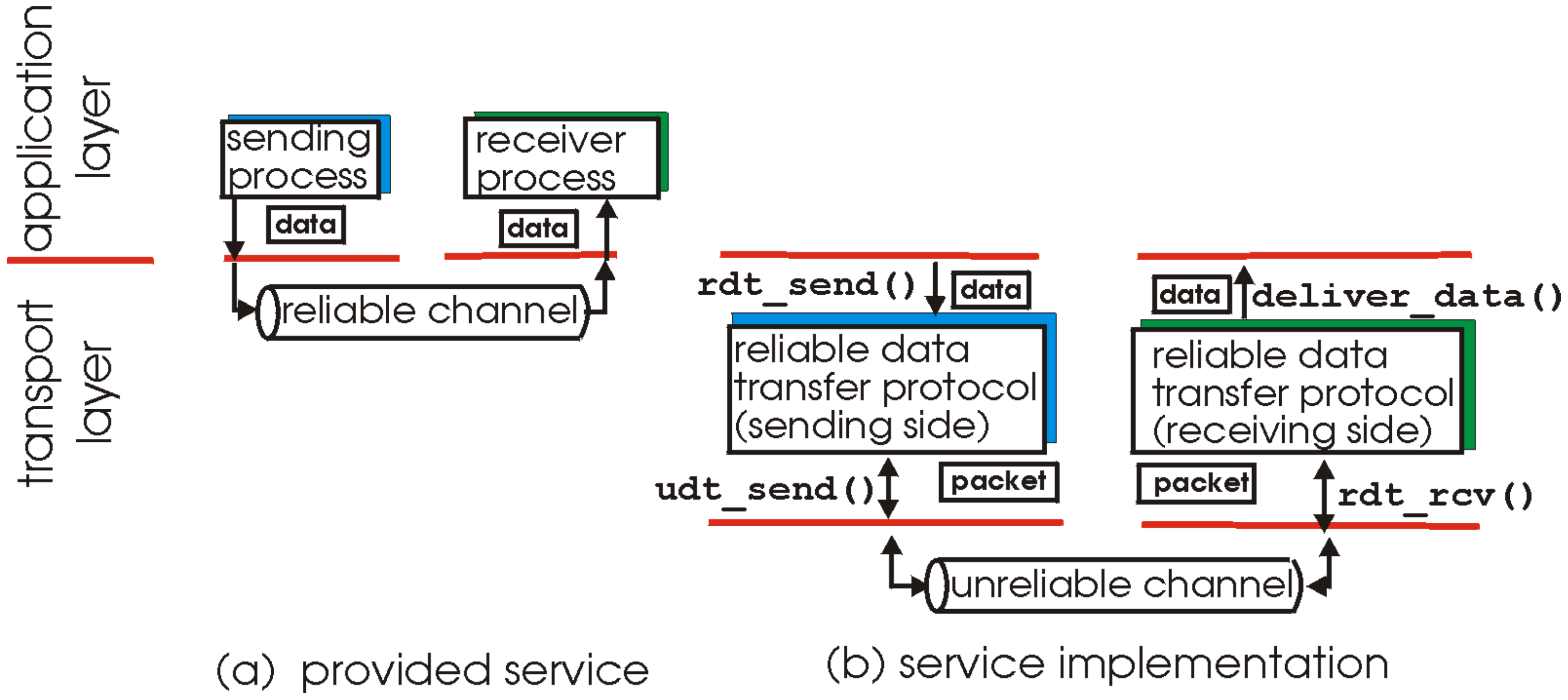


Principles of Reliable Data Transfer

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

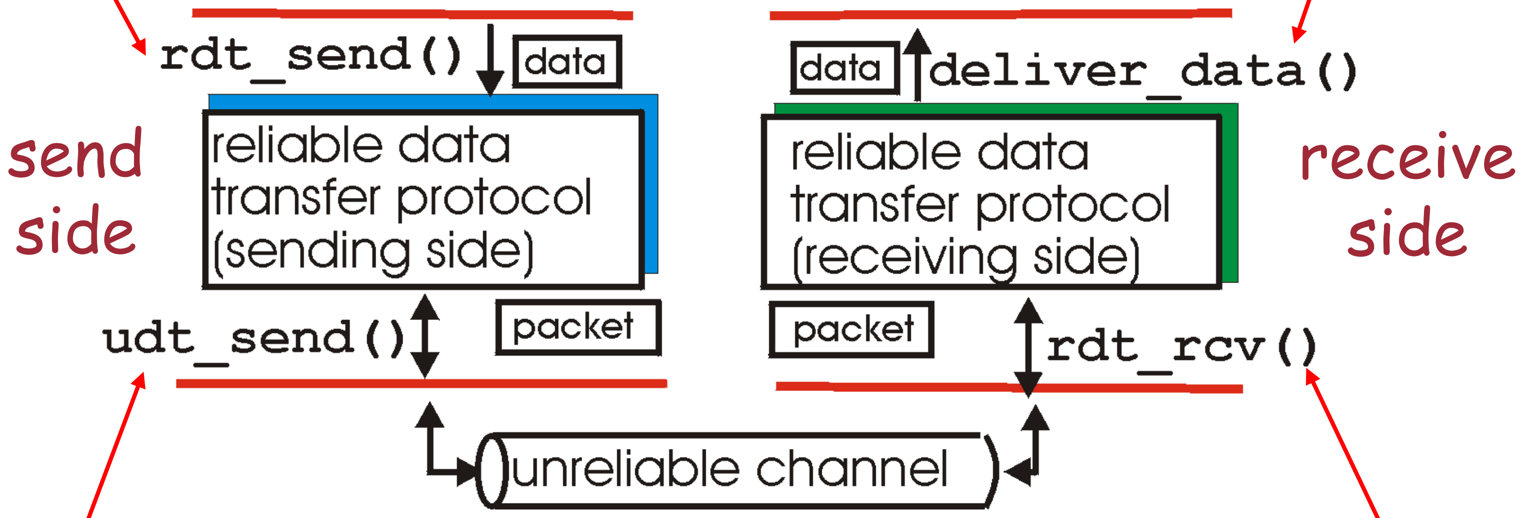


- Characteristics of unreliable channel will determine complexity of reliable data transfer protocol (rdt)

Reliable Data Transfer: getting started

rdt_send() : called from above, (e.g., by app.). Passed data to be delivered 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

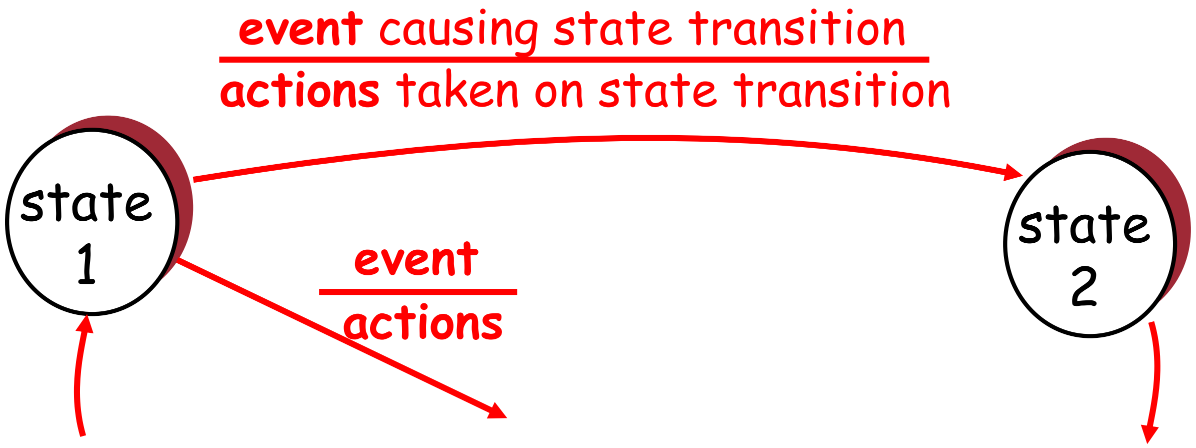
rdt_rcv() : called when packet arrives on rcv-side of channel

Reliable Data Transfer: getting started

We will:

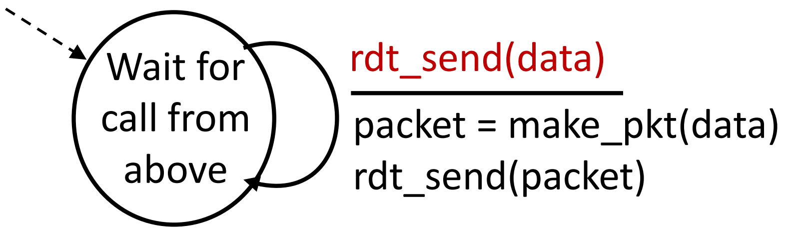
- Incrementally develop sender, receiver sides of reliable data transfer protocol (rdt)
- Consider only unidirectional data transfer
  - But control info will flow on both directions!
- Use finite state machines (FSM) to specify sender, receiver

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

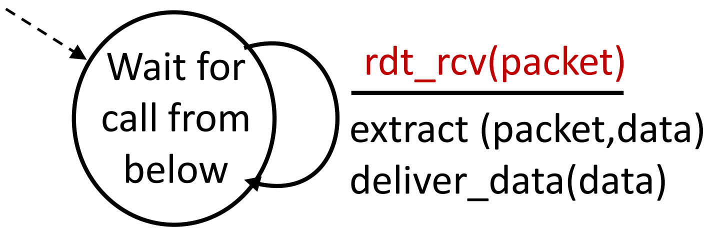


rdt1.0: reliable transfer over a reliable channel

- Underlying channel perfectly reliable
  - No bit errors, No loss of packets
- Separate FSMs for sender, receiver:
  - Sender sends data into underlying channel
  - Receiver read data from underlying channel



sender



receiver

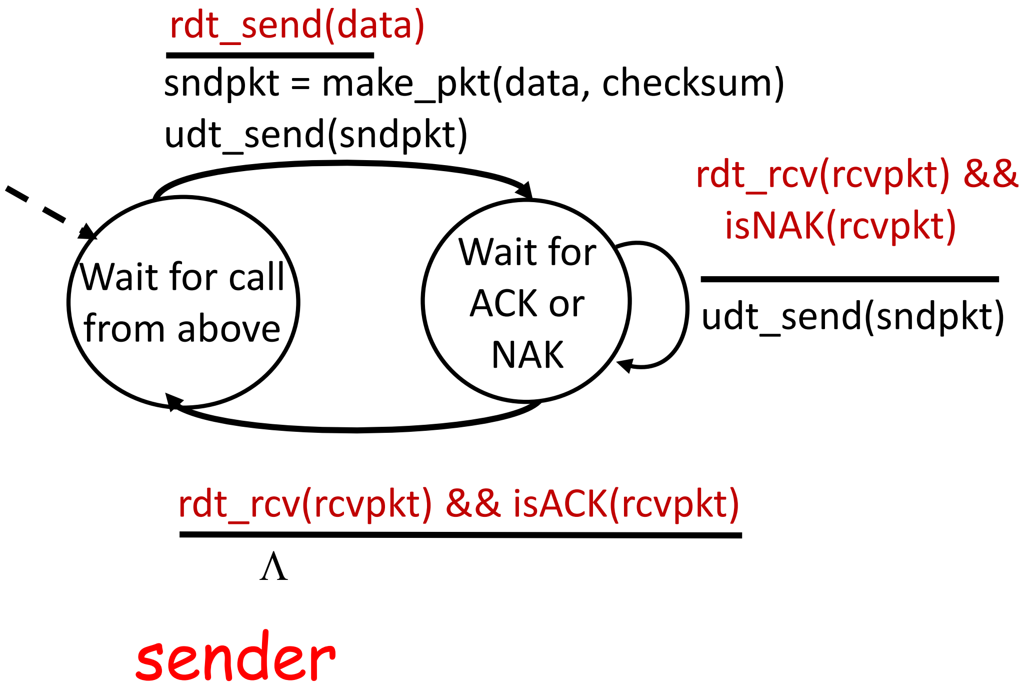
rdt2.0: channel with bit errors

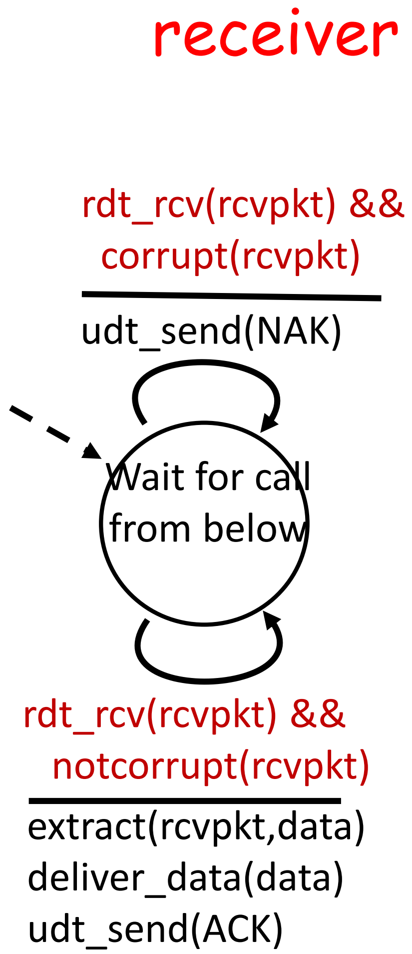
- Underlying channel may flip bits in packet
  - **Don't worry... Checksum is there to detect bit errors**

- **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

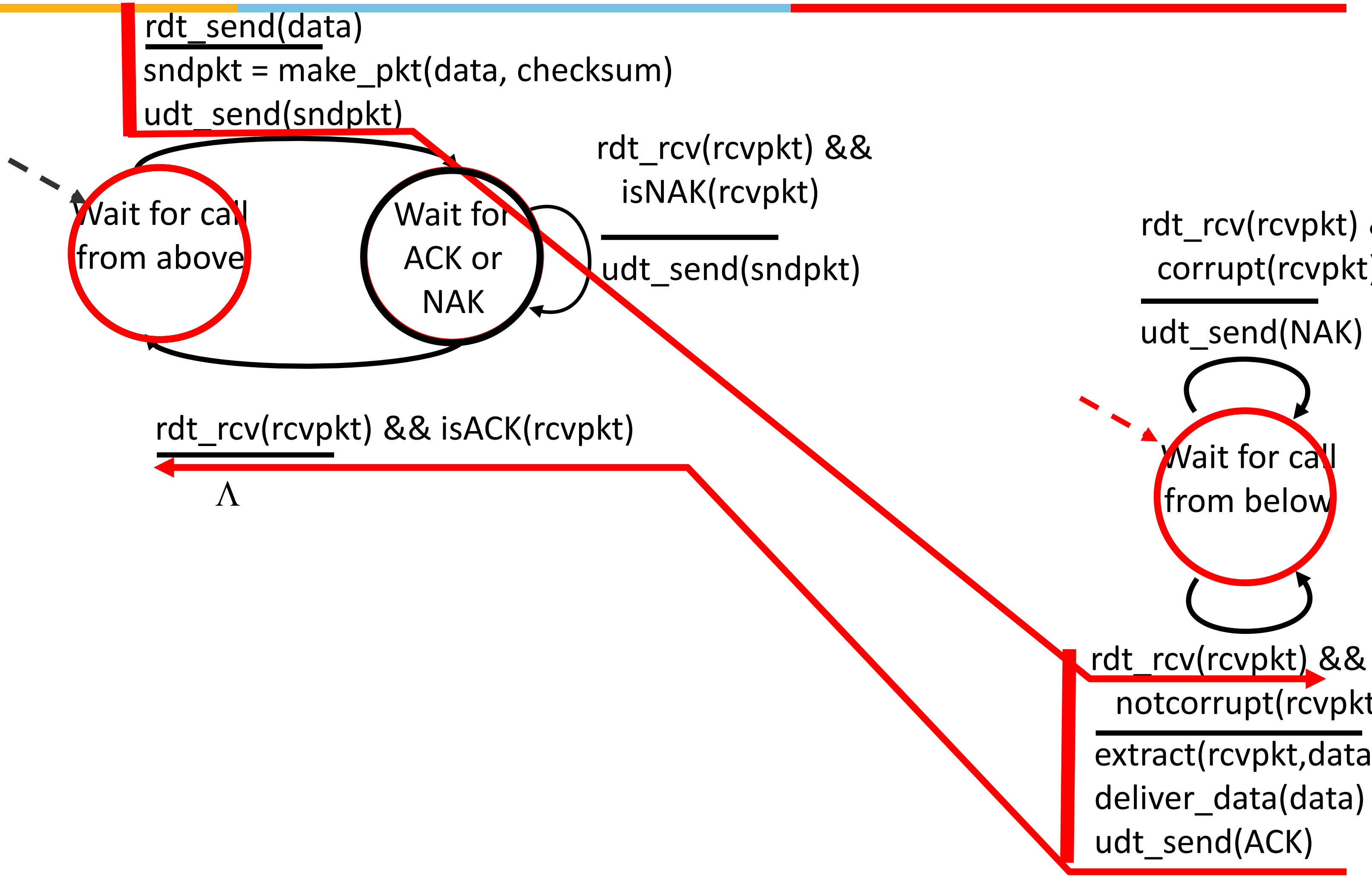
- New mechanisms in **`rdt2.0`** (beyond **`rdt1.0`**):
  - Error detection
  - Receiver feedback: control msgs (ACK,NAK) rcvr->sender

rdt2.0: FSM Specification

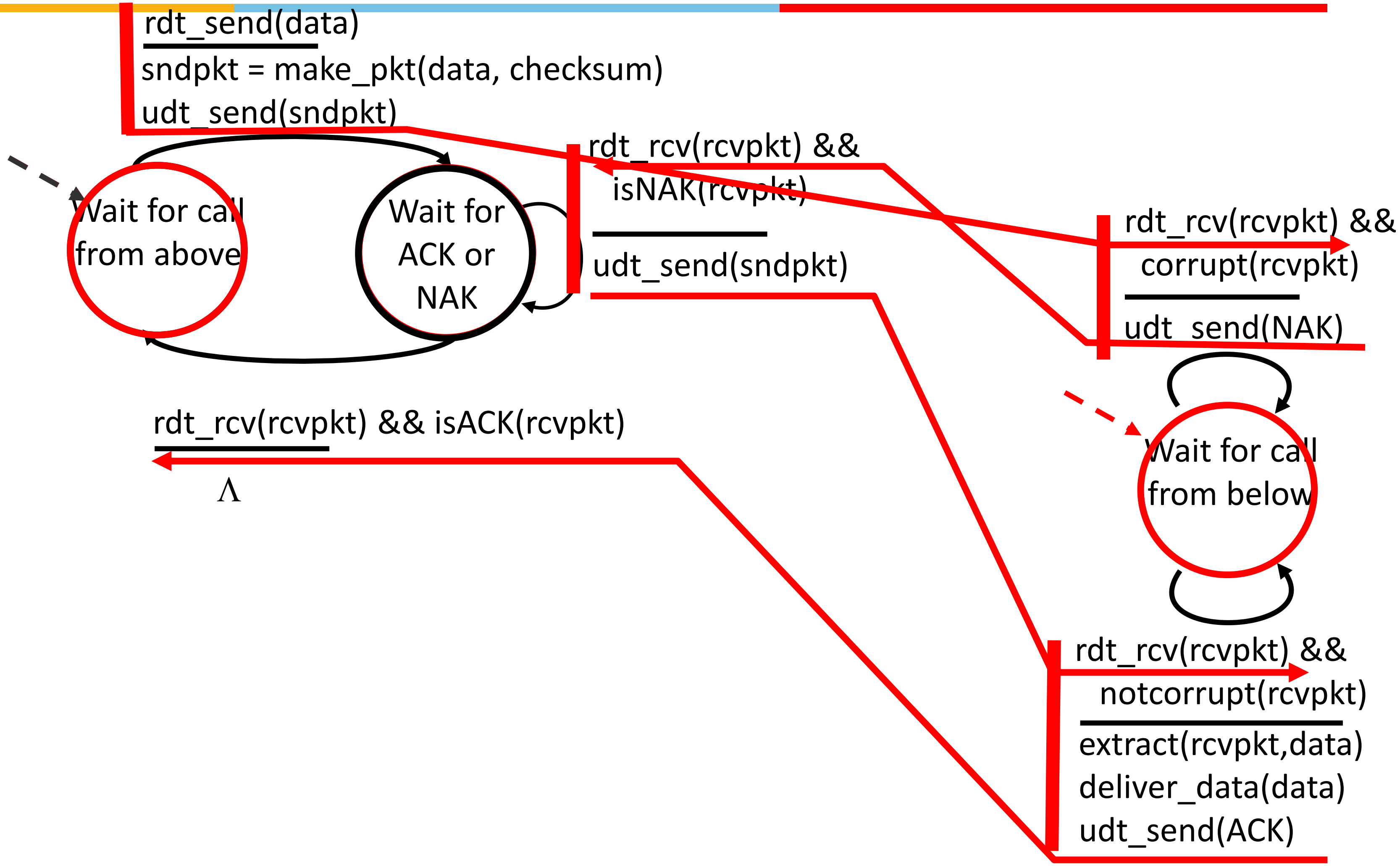




rdt2.0: Operation with no Errors



rdt2.0: Error Scenario

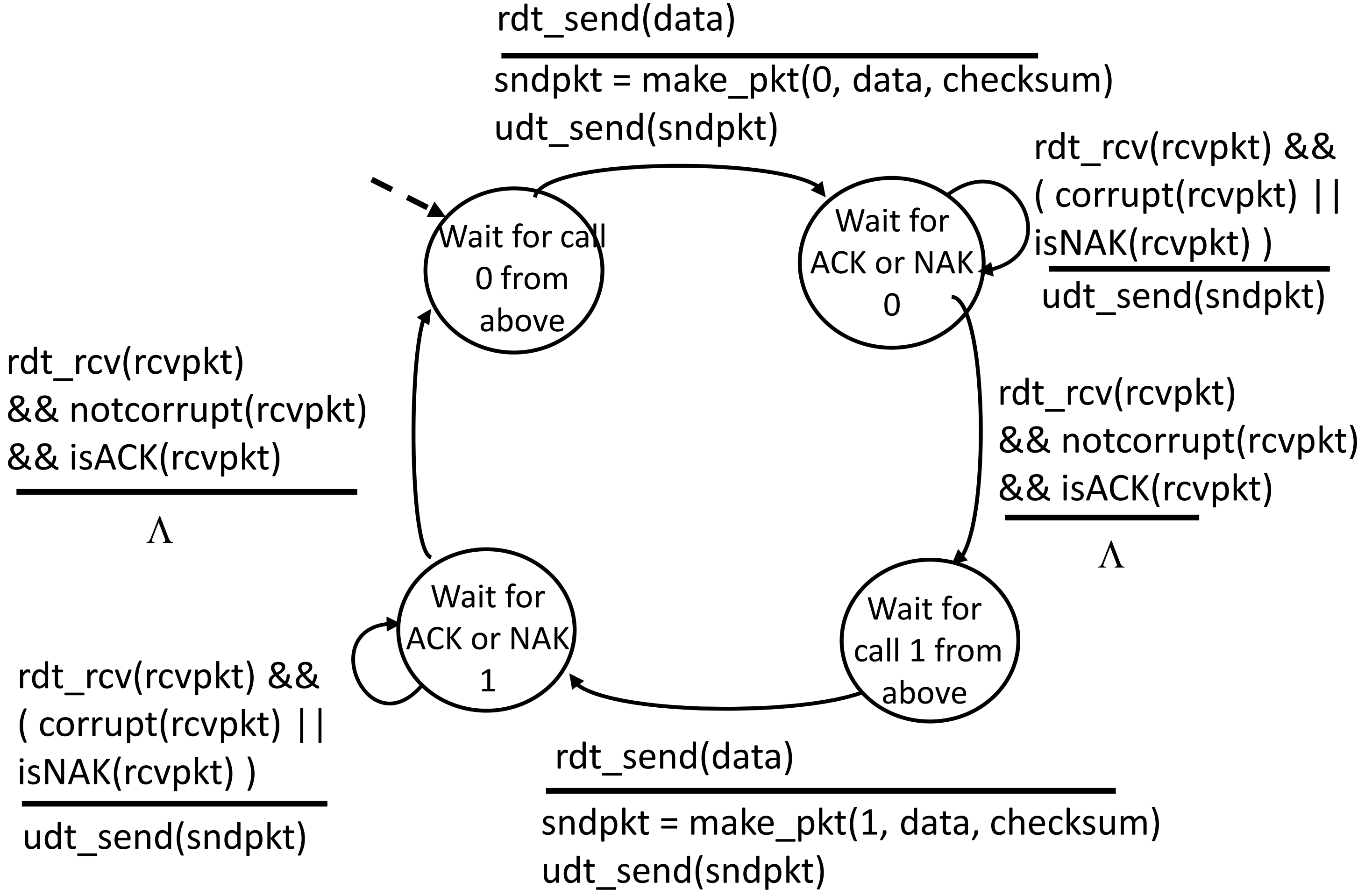


rdt2.0 Has a fatal flaw!

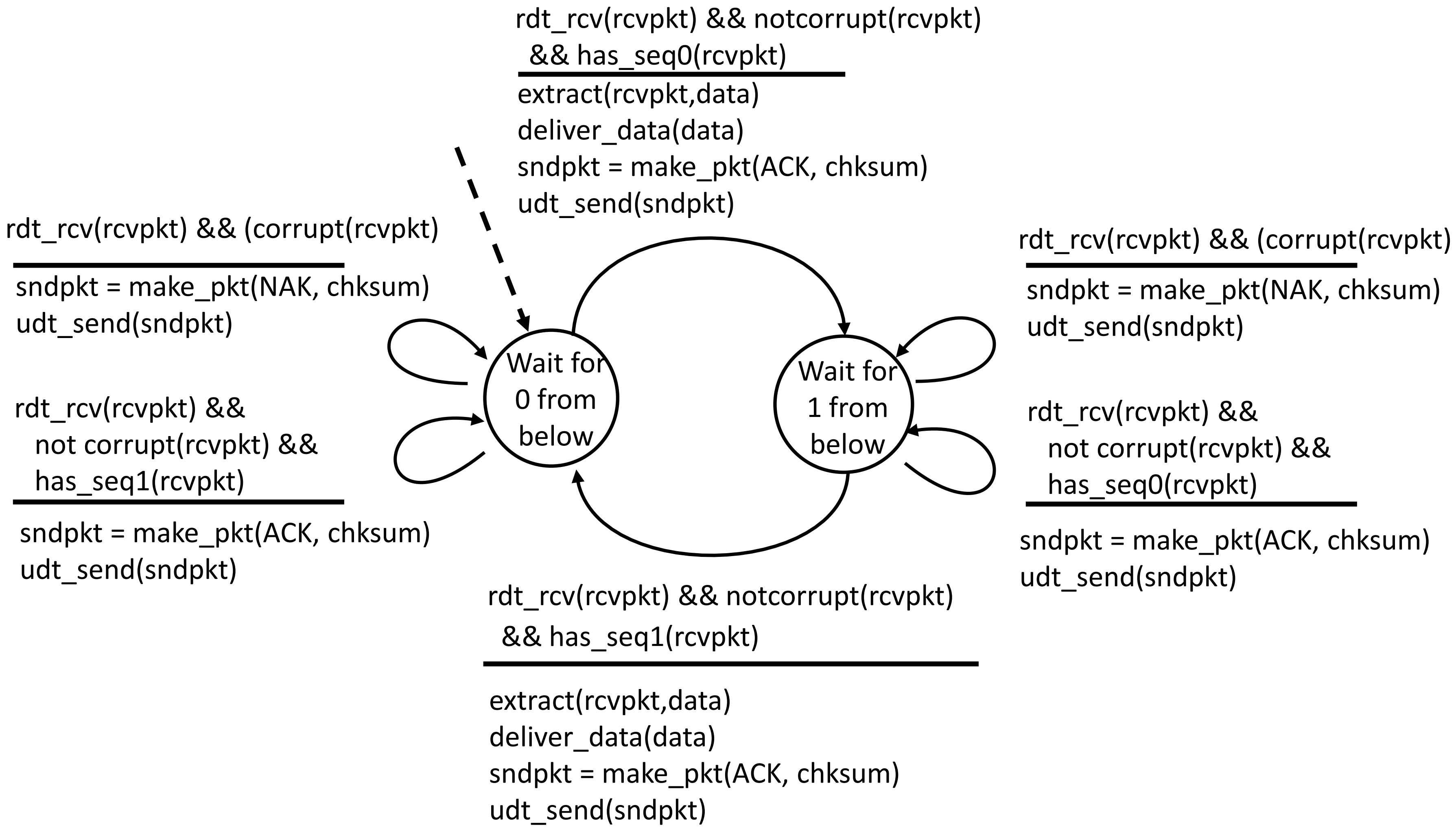
- What happens if ACK/NAK corrupted?
  - Sender doesn't know what happened at receiver!
  - Simple, just retransmit.

- How to handle duplicates?
  - Sender adds *sequence number* to each pkt
  - Receiver discards (doesn't deliver up) duplicate pkt

rdt2.1: Sender, handles garbled ACK/NAKs



rdt2.1: Receiver, handles garbled ACK/NAKs



rdt2.1: Discussion

Sender:

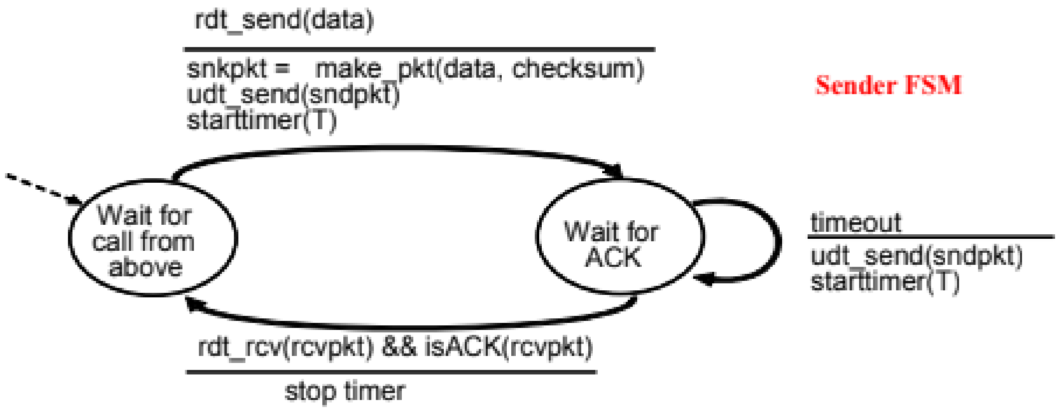
- Seq # added to pkt
- Two seq. #'s (0,1) will suffice. Why?
- Must check if received ACK/NAK corrupted
- Twice as many states
  - State must "remember" whether "current" pkt has 0 or 1 seq. #

Receiver:

- Must check if received packet is duplicate
  - State indicates whether 0 or 1 is expected pkt seq #
  - For an out of order received packet, it sends ACK for it

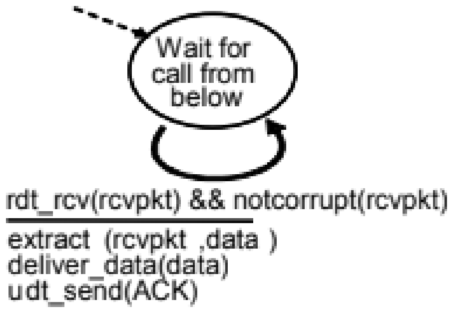
- Note: Receiver can *not* know if its last ACK/NAK received OK at sender

Sol:



Sender FSM

<5 marks for sender FSM>



<2 marks for receiver FSM>

Receiver FSM

Explanation:

Because the sender-to-receiver channel can corrupt packets, the data-sent on the sender-to-receiver channel will need a **checksum** to detect bit errors.
Because the sender-to-receiver channel can lose packets, we will need to have a **timer** to timeout and retransmit packets that have not been received by the receiver.
The receiver will need to indicate which packets it has received by using an **ACK message**; if a packet is not received or is received corrupted, no ACK is sent.
There is **no need for sequence numbers**, since there will be no unneeded (and unexpected at the receiver) retransmissions.
There is no need of **NAKs,** no response required by receiver for the arrival of corrupted packets. Such packets will be retransmitted by the sender when timer expires.
Checksum is not required at receiver side because receiver to sender channel is reliable.