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SAPIENZA
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Chapter 3
Transport Layer

Reti di Elaboratori
Corso di Laurea in Informatica
Università degli Studi di Roma "La Sapienza"
Canale A-L
Prof.ssa Chiara Petrioli

Parte di queste slide sono state prese dal materiale associato al libro *Computer Networking: A Top Down Approach* , 5th edition.

Thanks also to Antonio Capone, Politecnico di Milano, Giuseppe Bianchi and Francesco LoPresti, Un. di Roma Tor Vergata

Chapter 3: Transport Layer

Our goals:

- understand principles behind transport layer services:
  - multiplexing/demultiplexing
  - reliable data transfer
  - flow control
  - congestion control
- learn about transport layer protocols in the Internet:
  - UDP: connectionless transport
  - TCP: connection-oriented transport
  - TCP congestion control

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
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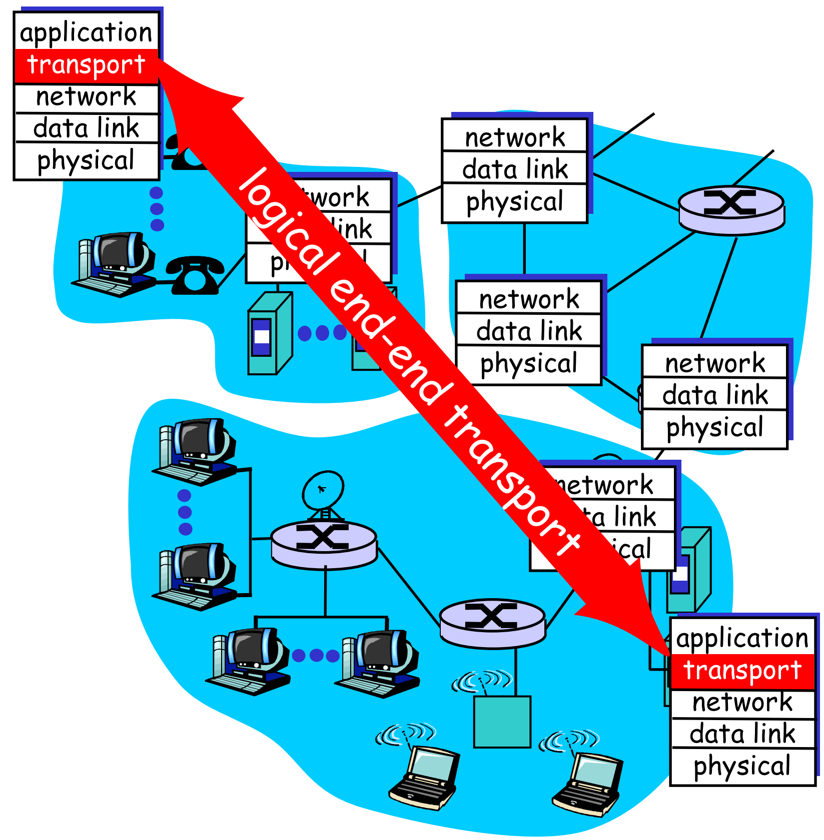
- 3.6 Principles of congestion control
- 3.7 TCP congestion control

Transport services and protocols

- provide *logical communication* between app processes running on different hosts
- transport protocols run in end systems
  - send side: breaks app messages into segments, passes to network layer
  - rcv side: reassembles segments into messages, passes to app layer
- more than one transport protocol available to apps
  - Internet: TCP and UDP



Transport vs. network layer

- *network layer:* logical communication between hosts
- *transport layer:* logical communication between processes
  - relies on, enhances, network layer services

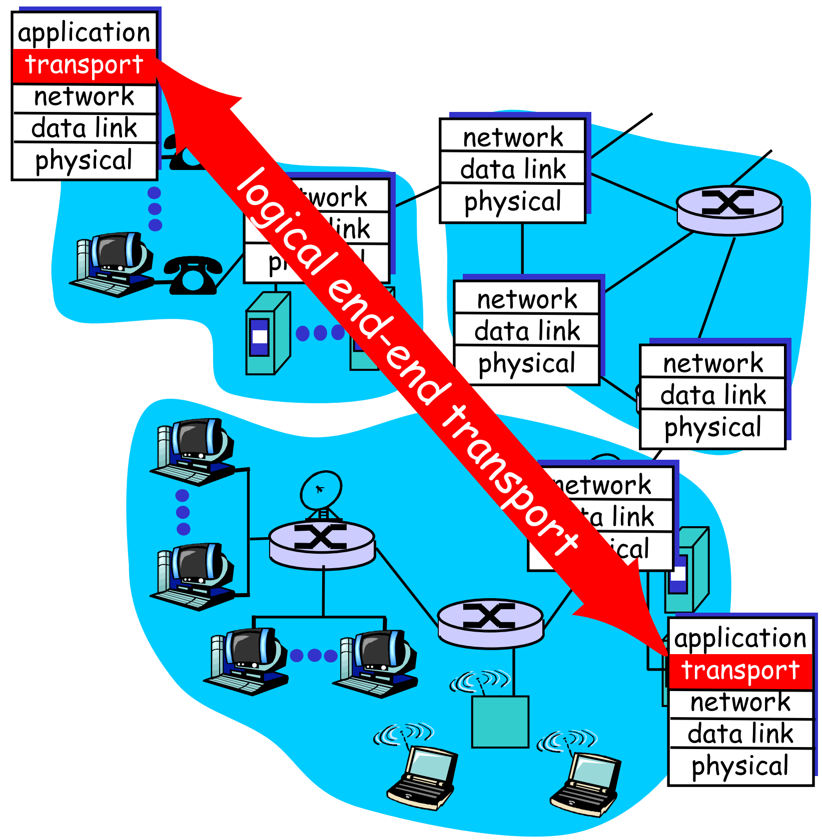
Household analogy:

12 kids sending letters to 12 kids

- processes = kids
- app messages = letters in envelopes
- hosts = houses
- transport protocol = Ann and Bill
- network-layer protocol = postal service

Internet transport-layer protocols

- reliable, in-order delivery (TCP)
  - congestion control
  - flow control
  - connection setup
- unreliable, unordered delivery: UDP
  - no-frills extension of "best-effort" IP
- services not available:
  - delay guarantees
  - bandwidth guarantees



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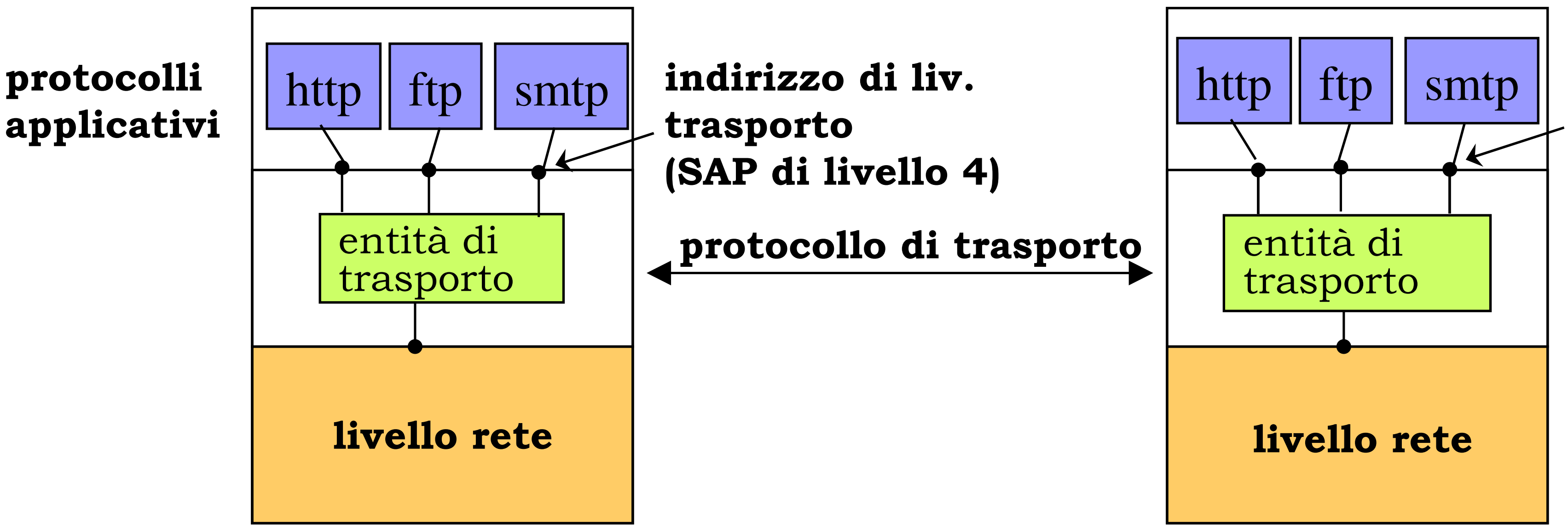
Servizio di trasporto

- Più applicazioni possono essere attive su un end system
  - il livello di trasporto svolge funzioni di multiplexing/demultiplexing
  - ciascun collegamento logico tra applicazioni è indirizzato dal livello di trasporto



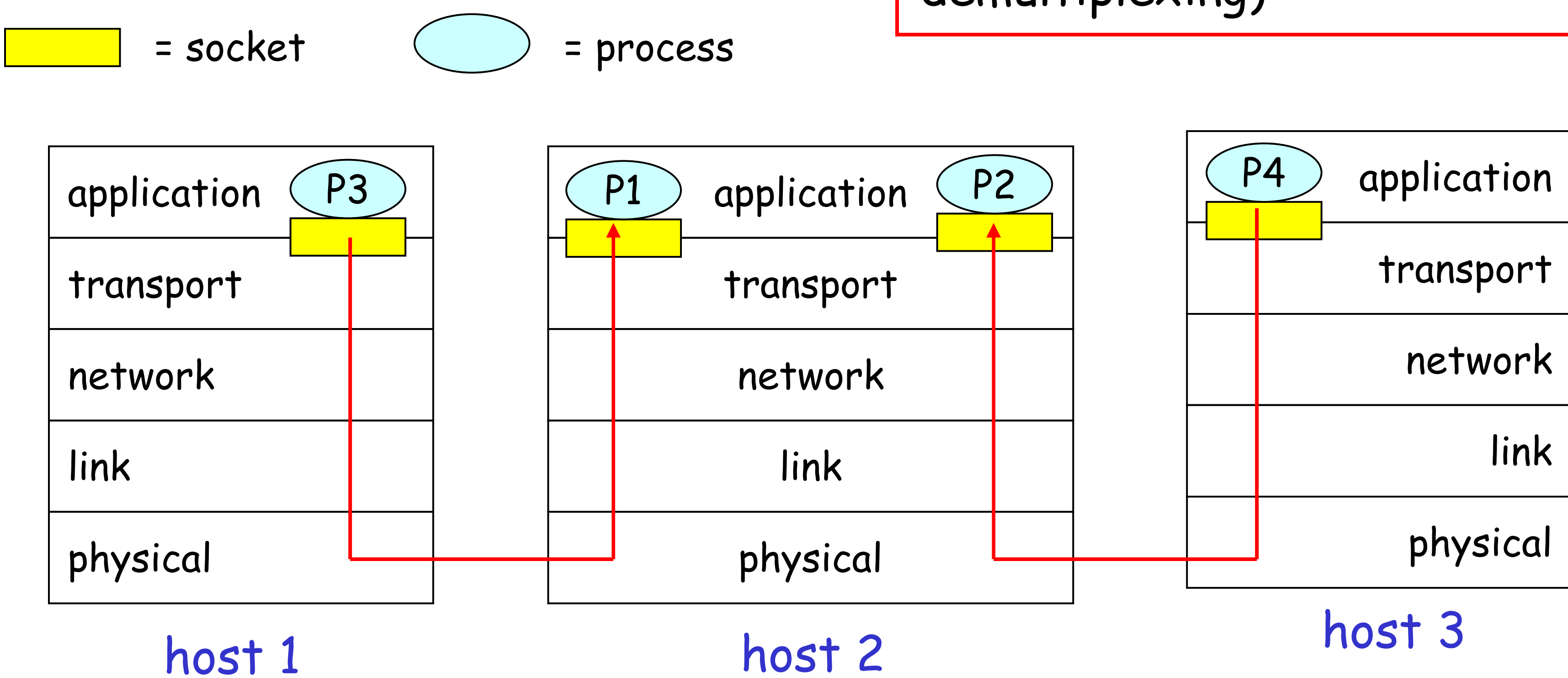
Multiplexing/demultiplexing

Demultiplexing at rcv host:

delivering received segments to correct socket

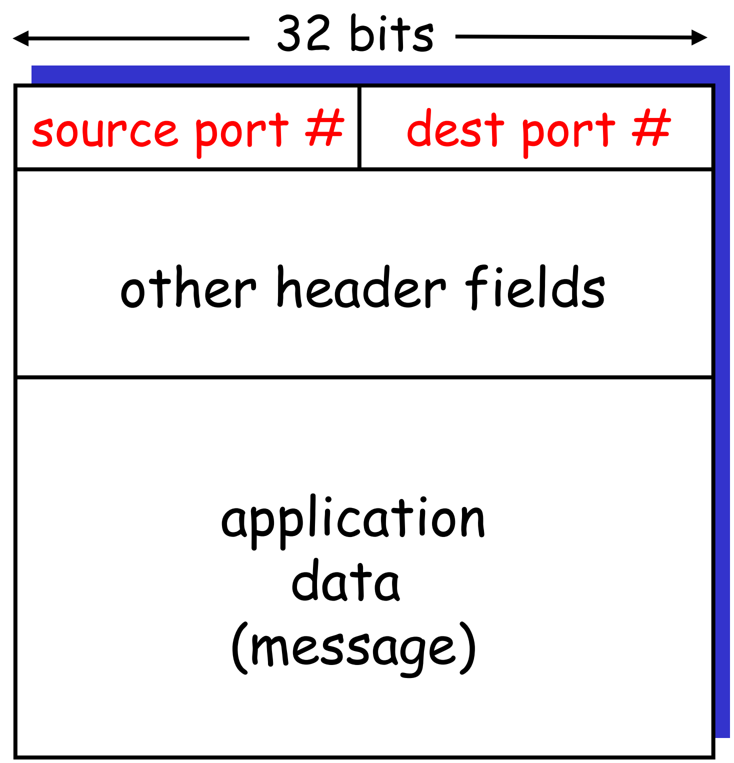
Multiplexing at send host:

gathering data from multiple sockets, enveloping data with header (later used for demultiplexing)



How demultiplexing works

- host receives IP datagrams
  - each datagram has source IP address, destination IP address
  - each datagram carries 1 transport-layer segment
  - each segment has source, destination port number (recall: well-known port numbers for specific applications)
- host uses IP addresses & port numbers to direct segment to appropriate socket



TCP/UDP segment format

Connectionless demultiplexing

- Create sockets with port numbers:

```
DatagramSocket mySocket1 = new
   DatagramSocket(99111);
DatagramSocket mySocket2 = new
   DatagramSocket(99222);
```

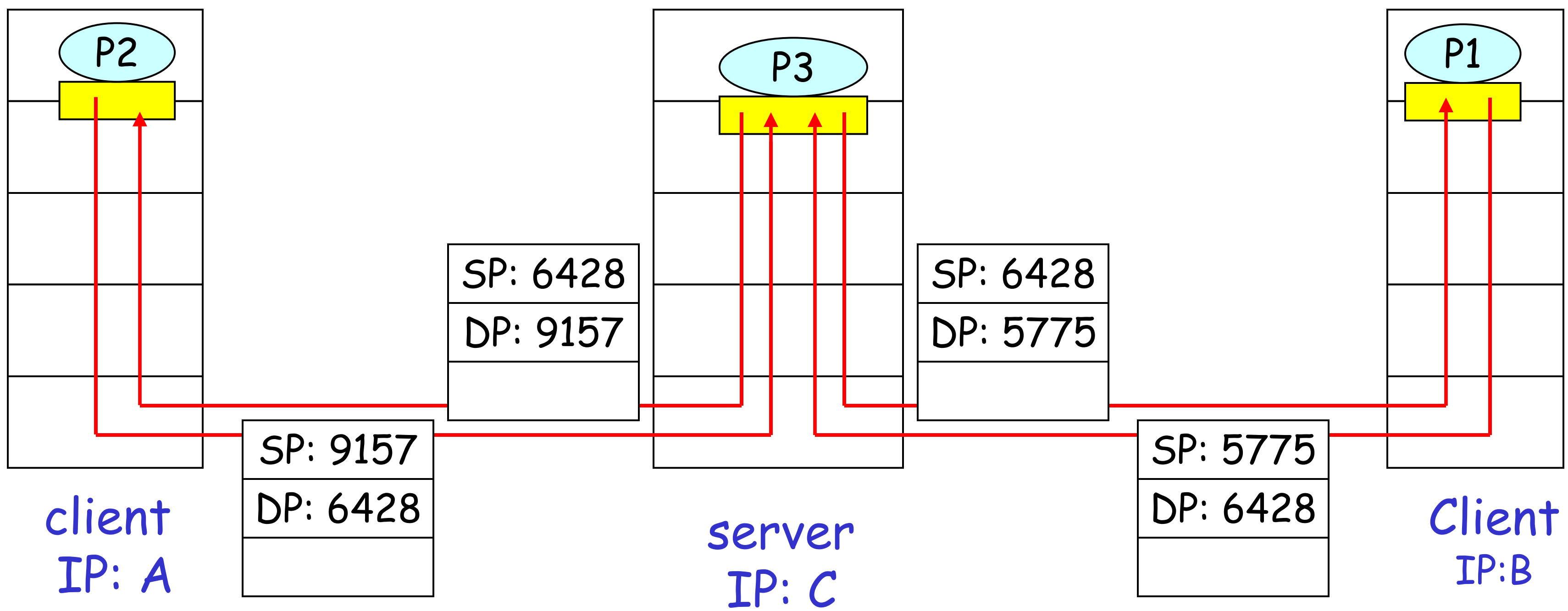
- UDP socket identified by two-tuple:

(dest IP address, dest port number)

- When host receives UDP segment:
  - checks destination port number in segment
  - directs UDP segment to socket with that port number
- IP datagrams with different source IP addresses and/or source port numbers directed to same socket

Connectionless demux (cont)

```
DatagramSocket serverSocket = new DatagramSocket(6428);
```

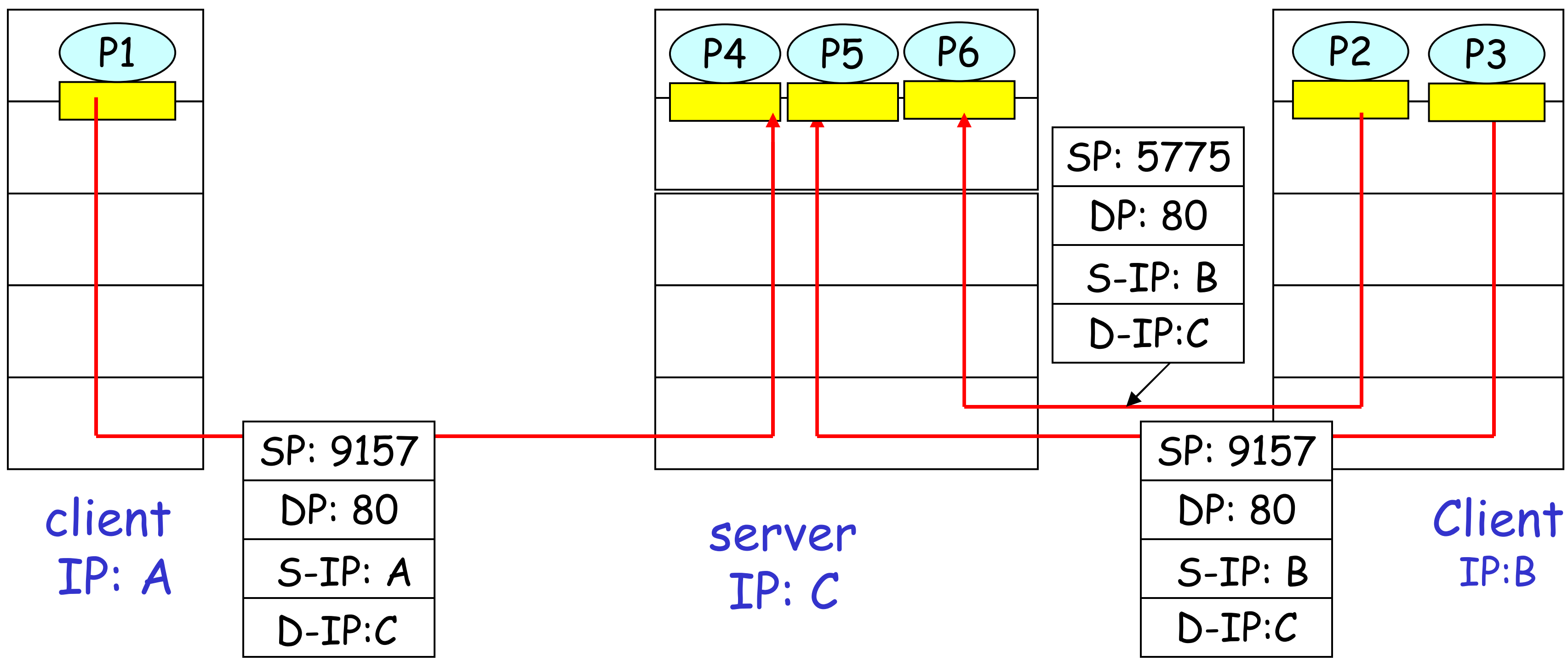


SP provides "return address"

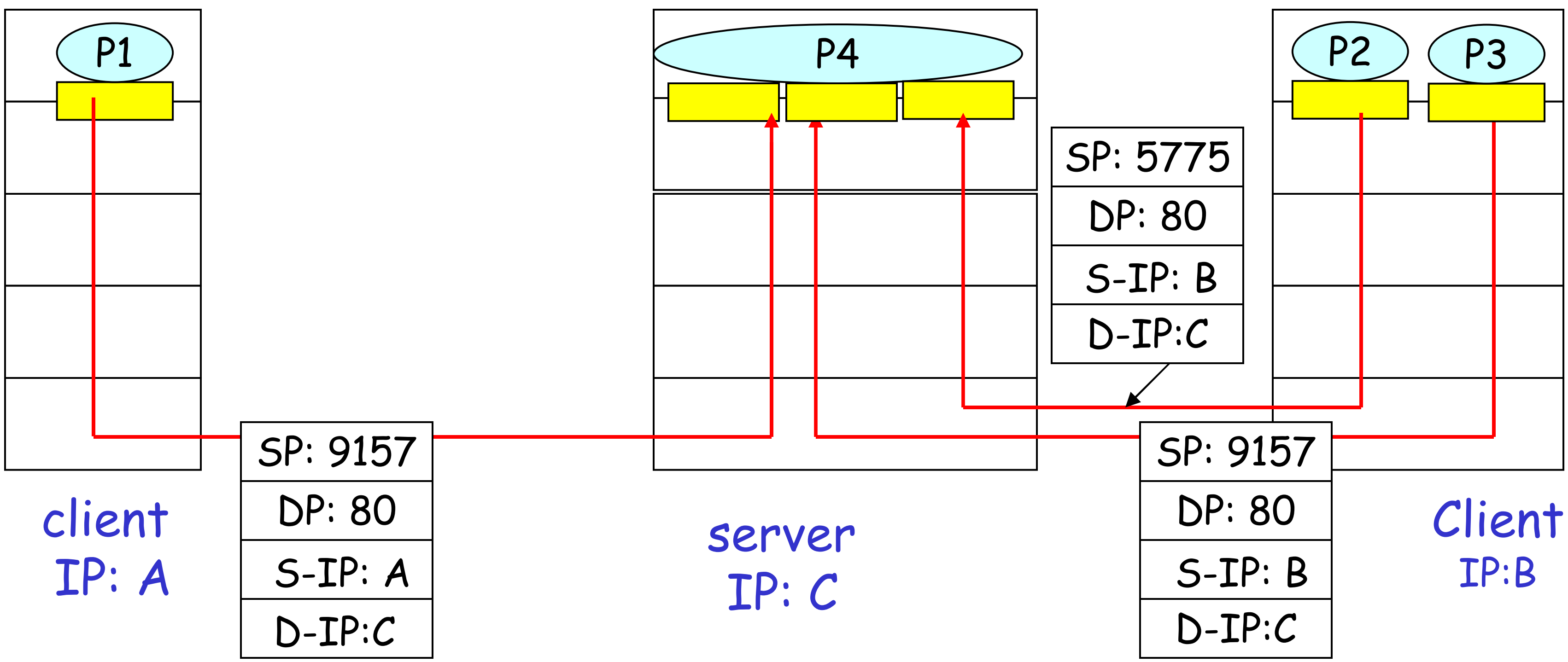
Connection-oriented demux

- TCP socket identified by 4-tuple:
  - source IP address
  - source port number
  - dest IP address
  - dest port number
- recv host 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
  - non-persistent HTTP will have different socket for each request

Connection-oriented demux (cont)



Connection-oriented demux: Threaded Web Server



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UDP: User Datagram Protocol [RFC 768]

- "no frills," "bare bones" Internet transport protocol
- "best effort" service, UDP segments may be:
  - lost
  - delivered out of order to app

reliable transfer over UDP: add reliability at application layer
  - application-specific error recovery!

- *connectionless:*
  - no handshaking between UDP sender, receiver
  - each UDP segment handled independently of others

Why is there a UDP?

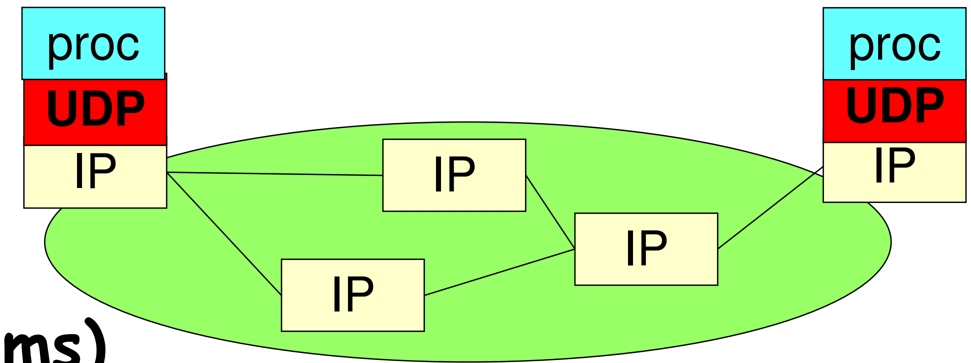
- no connection establishment (which can add delay)
- **simple: no connection state at sender, receiver**
- **small segment header (8 byte)**
- no congestion control: UDP can blast away as fast as desired

- often used for streaming multimedia apps
  - loss tolerant
  - rate sensitive

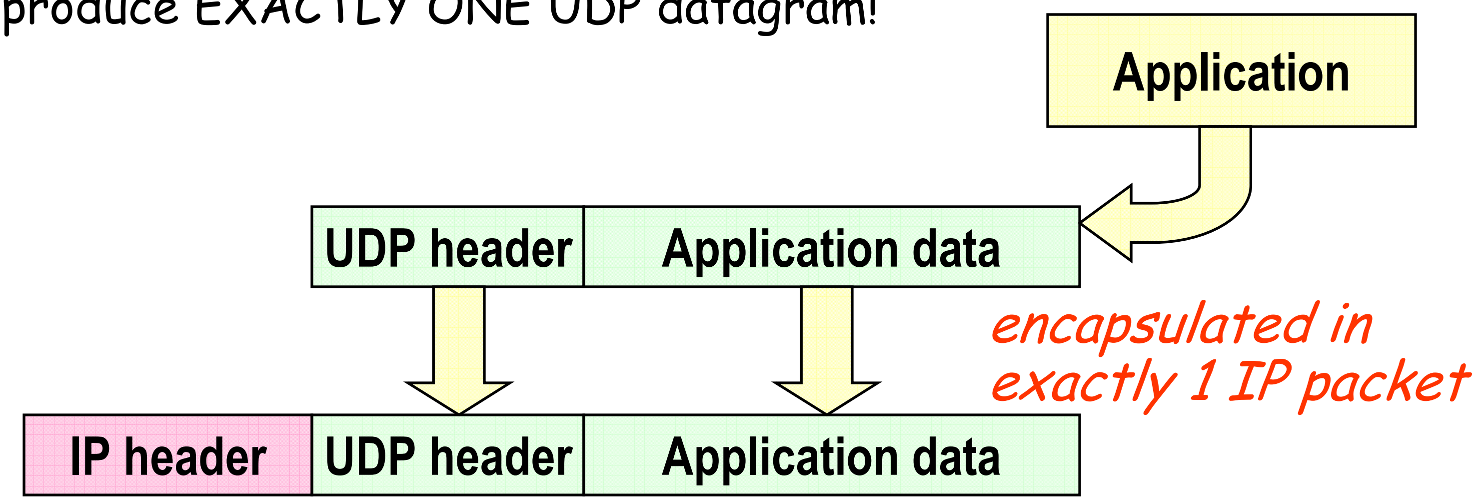
other UDP uses: DNS, SNMP..

UDP Packets

- **Connection-Less**
  - (no handshaking)



- **UDP packets (Datagrams)**
  - Each application interacts with UDP transport sw to produce EXACTLY ONE UDP datagram!



This is why, improperly, we use the term UDP packets

UDP datagram format

8 bytes header + variable payload

0	7	15	23	31
source port		destination port		
length (bytes)		Checksum		
Data				

- **UDP length field**
  - all UDP datagram
  - (header + payload)
- **payload sizes allowed:**
  - Empty
  - even size (bytes)

➔ **UDP functions limited to:**

⇨ **addressing**

→which is the only strictly necessary role of a transport protocol

⇨ **Error checking**

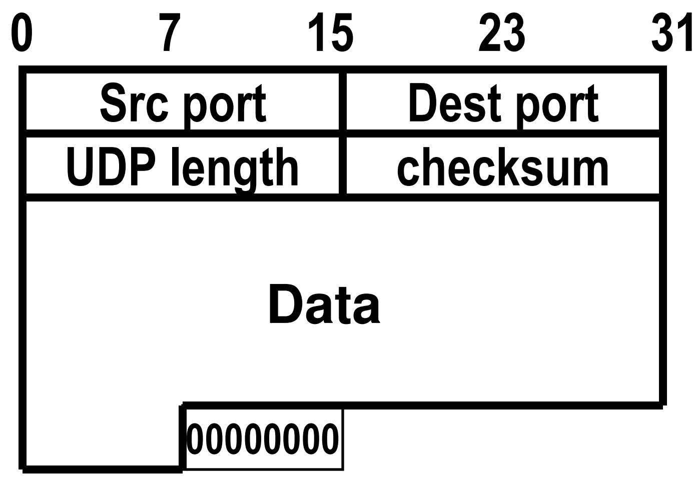
→which may even be **disabled** for performance

Maximum UDP datagram size

- 16 bit UDP length field:
  - Maximum up to 2^{16-1} = 65535 bytes
  - Includes 8 bytes UDP header (max data = 65527)
- But max IP packet size is also 65535
  - Minus 20 bytes IP header, minus 8 bytes UDP header
  - Max UDP_data = **<u>65507</u>** bytes!
- Moreover, most OS impose further limitations!
  - most systems provide 8192 bytes maximum (max size in NFS)
  - some OS had (still have?) internal implementation features (bugs?) that limit IP packet size
    - SunOS 4.1.3 had 32767 for max tolerable IP packet transmittable (but 32786 in reception…) – bug fixed only in Solaris 2.2
- Finally, subnet Maximum Transfer Unit (MTU) limits may fragment datagram – annoying for reliability!
  - E.g. ethernet = 1500 bytes; PPP on your modem = 576

Error checksum

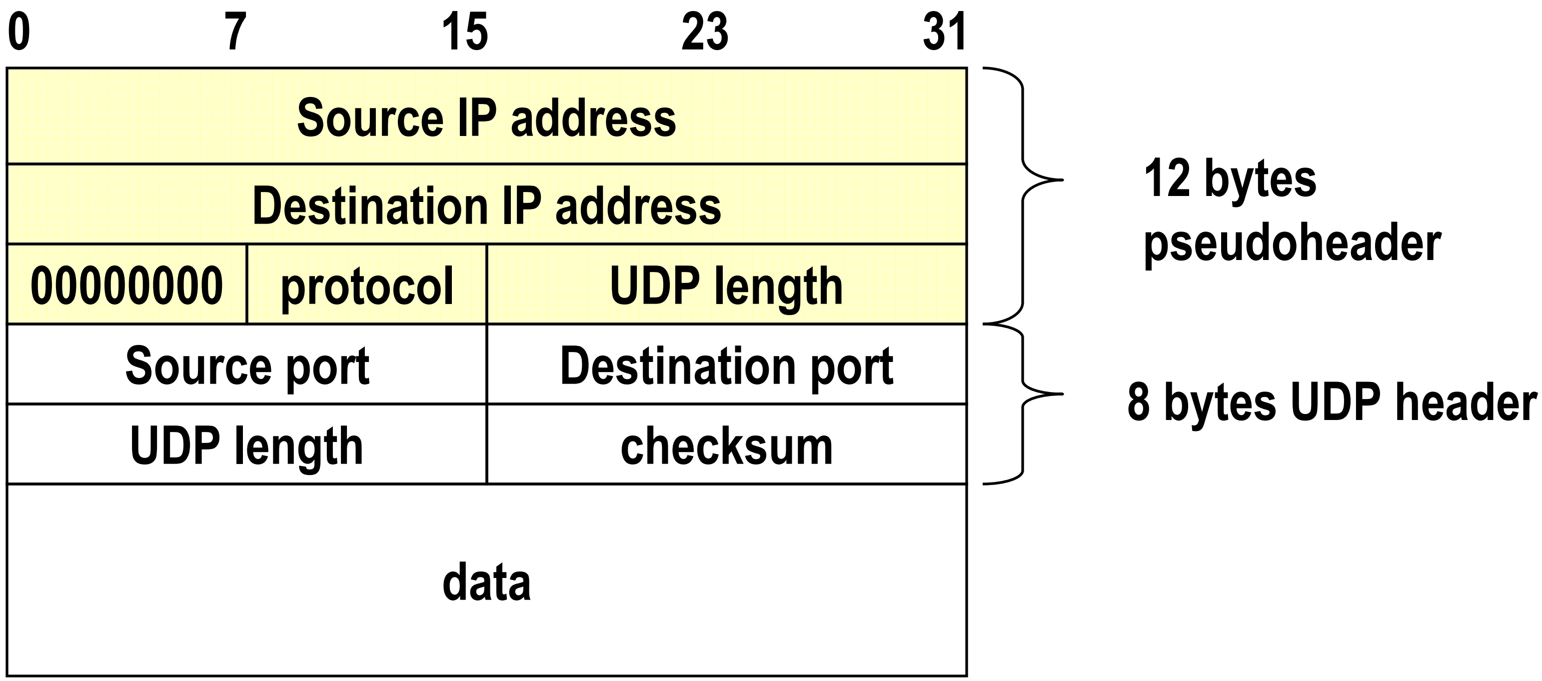
- 16 bit checksum field, obtained by:
  - summing up all 16 bit words in header data and **pseudoheader**, in 1's complement (checksum fields filled with 0s initially)
  - take 1's complement of result
  - if result is 0, set it to 111111…11 (65535==0 in 1's complement)
  - Sender puts checksum value into UDP checksum field
- at destination:
  - 1's complement sum should return 0, otherwise error detected
  - upon error, no action (just packet discard)
- efficient implementation RFC 1071



- Zero padding
  - To multiple of 16 bits
- checksum disabled
  - by source, by setting 0 in the checksum field

Pseudo header

- Is not transmitted!
  - But it is information available at transmitter and at receiver
  - intention: double check that packet has arrived at correct destination



Protocol field (TCP=6,UDP=17) necessary, as same checksum calculation used in TCP. UDP length duplicated.

disabling checksum

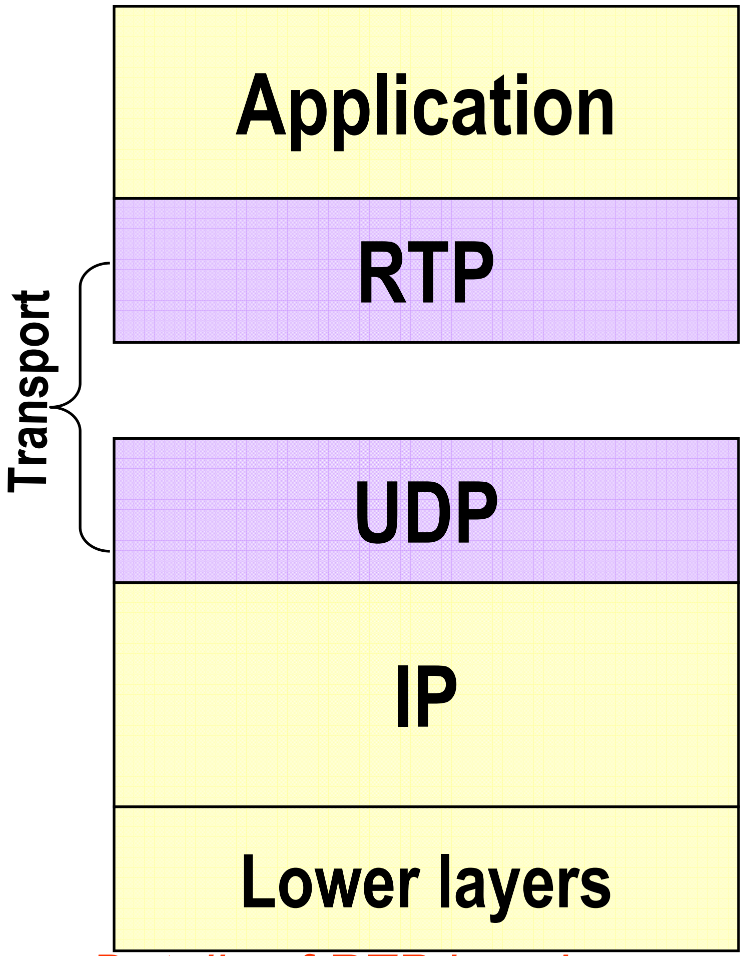
- In principle never!
  - Remember that IP packet checksum DOES NOT include packet payload.
- In practice, often done in NFS
  - sun was the first, to speed up implementation
- may be tolerable in LANs under one's control.
- Definitely dangerous in the wide internet
  - Exist layer 2 protocols without error checking

UDP: a lightweight protocol

- No connection establishment
  - no initial overhead due to handshaking
- No connection state
  - greater number of supported connections by a server!
- Small packet header overhead
  - 8 bytes only vs 20 in TCP
- originally intended for simple applications, oriented to short information exchange
  - DNS
  - management (e.g. SNMP)
  - etc
- No rate limitations
  - No throttling due to congestion & flow control mechanisms
  - No retransmission (for certain application loss tolerable)
- extremely important features for today multimedia applications! Expecially for real time applications which can tolerate some packet loss but require a minimum send rate.

RTP as seen from Application

Be careful: UDP ok for multimedia because it does not provide anything at all (no features = no limits!). Application developers have to provide supplementary transport capabilities at the application layer!



Solution for audio/video:
Real Time Protocol
(RTP, RFC 1889)

SOCKET INTERFACE

Application developer integrates RTP into the application by:
- writing code which creates the RTP encapsulating packets;
- sends the RTP packets into a UDP socket interface.

Details of RTP in subsequent courses – unless we are ahead of schedule

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A MUCH more complex transport
for three main reasons

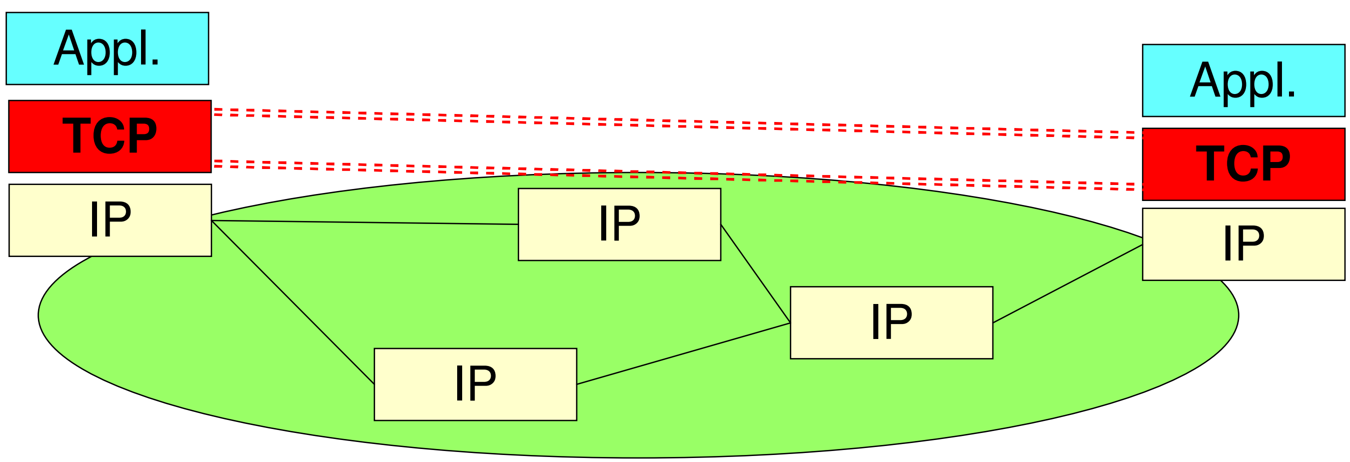
- Connection oriented
  - implements mechanisms to setup and tear down a full duplex connection between end points
- Reliable
  - implements mechanisms to guarantee error free and ordered delivery of information
- Flow & Congestion controlled
  - implements mechanisms to control traffic

TCP services

- connection oriented
  - TCP connections
- *reliable* transfer service
  - all bytes sent are received

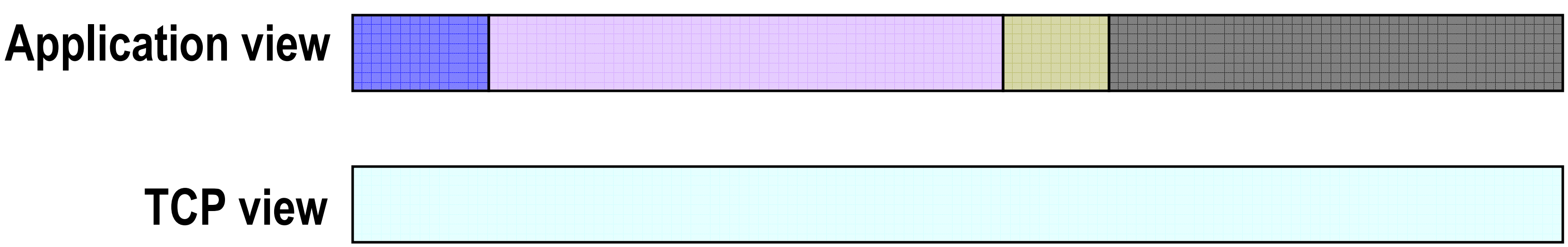
→ TCP functions

- → application addressing (ports)
- → error recovery (acks and retransmission)
- → reordering (sequence numbers)
- → flow control
- → congestion control



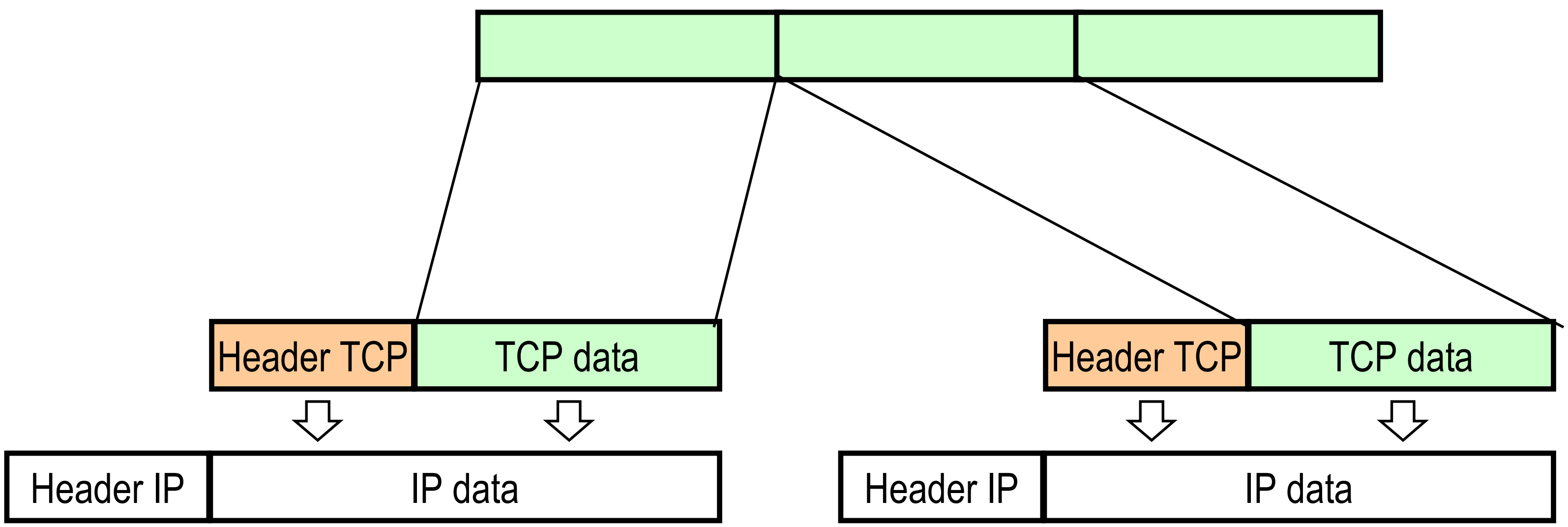
Byte stream service

- TCP exchange data between applications as a stream of bytes.
- It does not introduce any data delimiter (an application duty)
  - source application may enter 10 bytes followed by 1 and 40 (grouped with some semantics)
  - data is buffered at source, and transmitted
  - at receiver, may be read in the sequence 25 bytes, 22 bytes and 4 bytes...



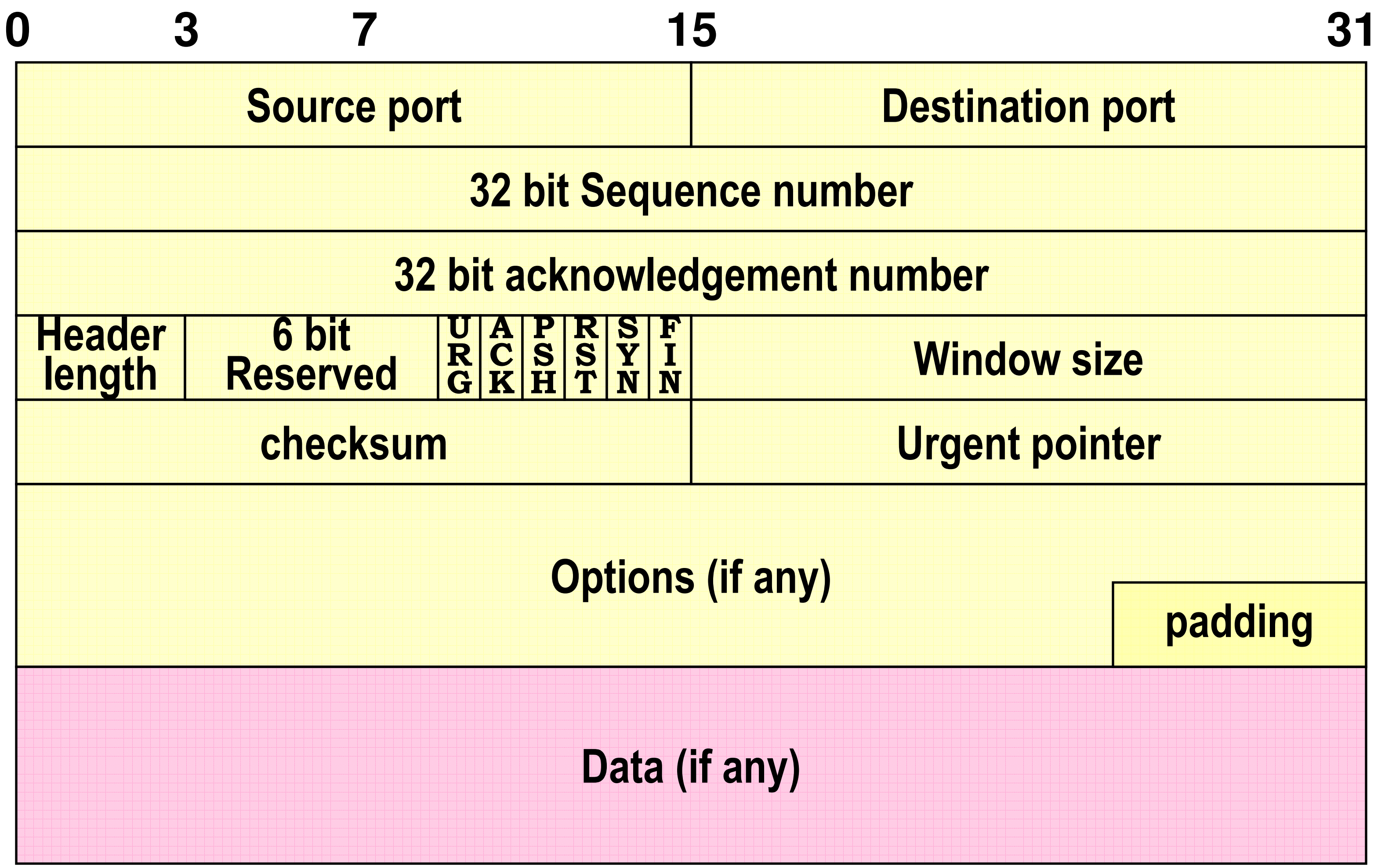
TCP segments

- Application data broken into segments for transmission
- segmentation totally up to TCP, according to what TCP considers being the best strategy
- each segment placed into an IP packet
- very different from UDP!!



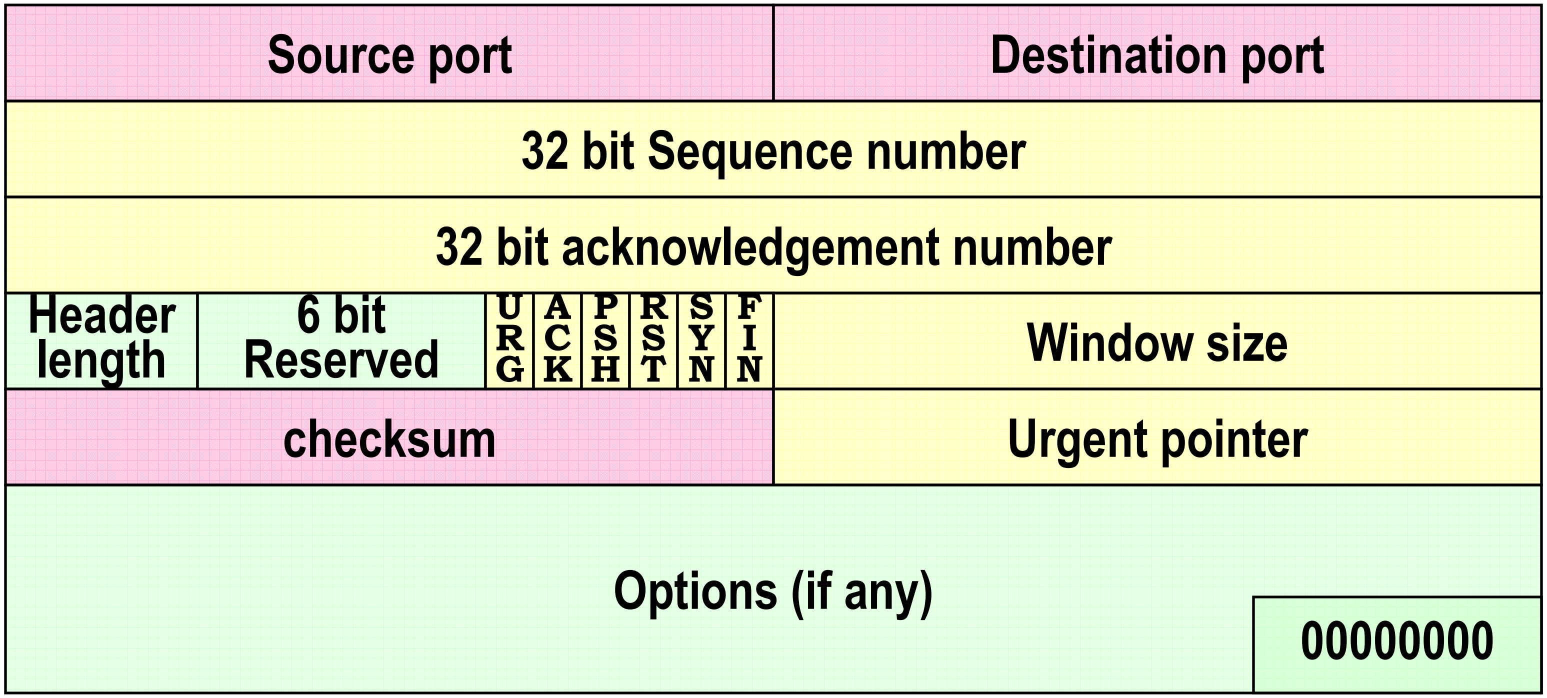
TCP segment format

20 bytes header (minimum)



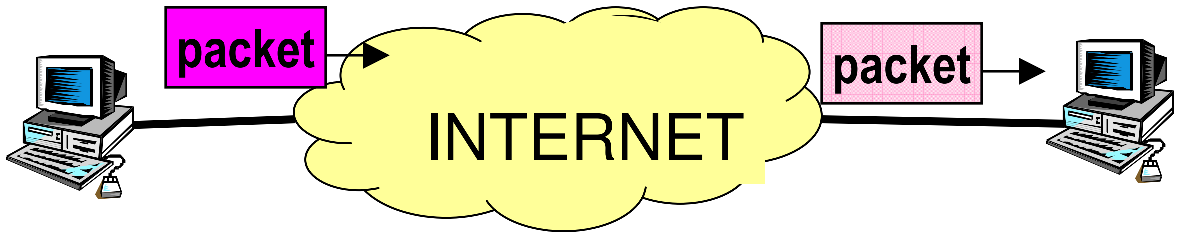
Source port								Destination port
32 bit Sequence number								
32 bit acknowledgement number								
Header length	6 bit Reserved	URG	ACK	PSH	RST	SYN	FIN	Window size
checksum								Urgent pointer

- Source & destination port + source and destination IP addresses
  - univocally determine TCP connection
- checksum as in UDP
  - same calculation including same pseudoheader
- no explicit segment length specification



- Header length: 4 bits
  - specifies the header size (n*4byte words) for options
  - maximum header size: 60 (15*4)
  - option field size must be multiple of 32bits: zero padding when not.
- Reserved: 000000 (still today!)

Reliable data transfer: issues



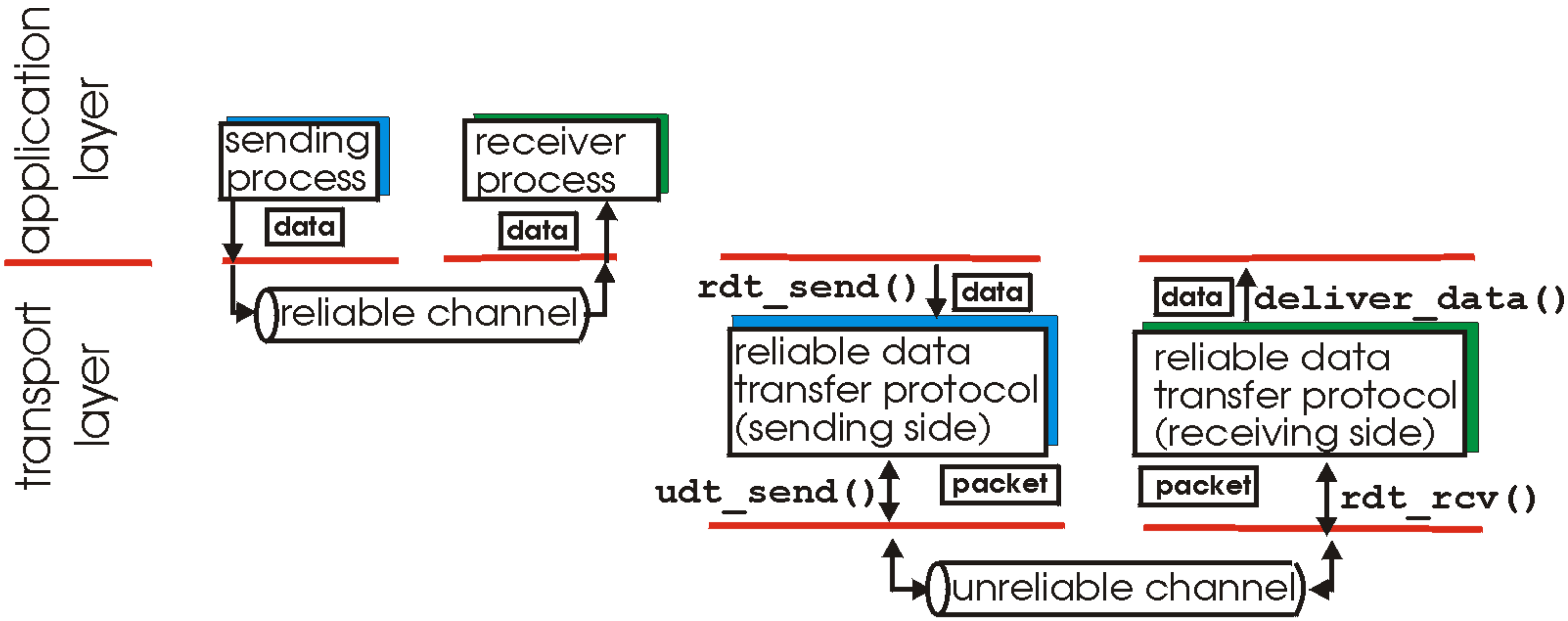
PROBLEMS:
1) Packet received with errors
2) Packet not received at all

Same problem considered at DATA LINK LAYER
(although it is less likely that a whole packet is lost at data link)

- mechanisms to guarantee correct reception:
  - Forward Error Correction (FEC) coding schemes
    - Powerful to correct bits affected by error, not effective in case of packet loss
    - Mostly used at link layer
  - Error detection (e.g. checksum used in UDP)
  - Retransmission – issues:
    - ACK
    - NACK
    - TIMEOUT

Principles of Reliable data transfer

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



(a) provided service

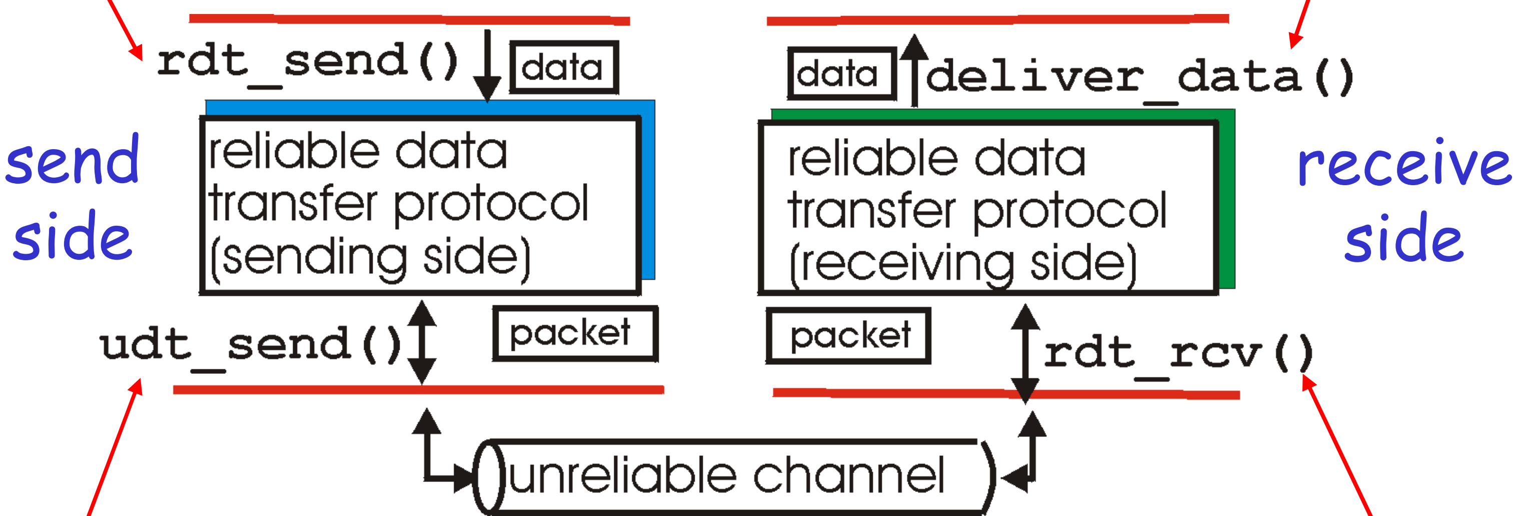
(b) service implementation

- 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

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



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

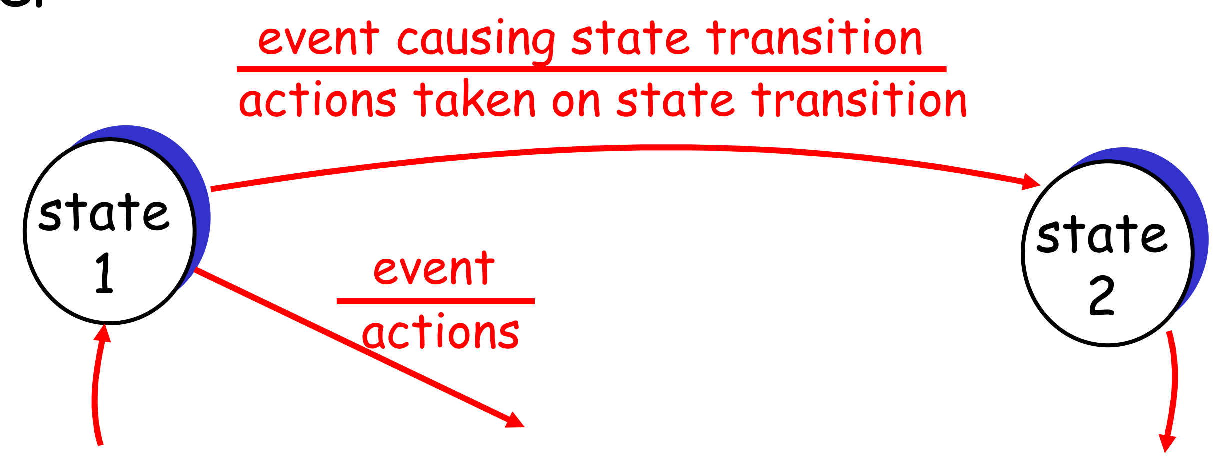
rdt_rcv(): called when packet arrives on rcv-side of channel

Reliable data transfer: getting started

We'll:

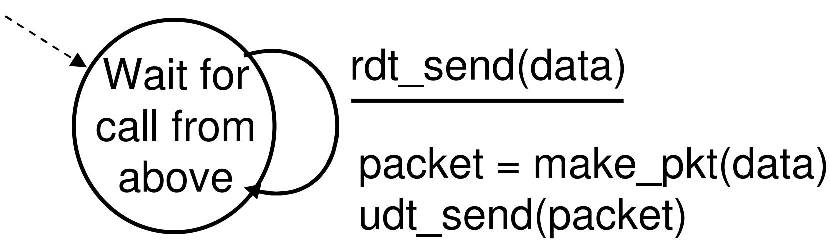
- incrementally develop sender, receiver sides of reliable data transfer protocol (rdt)
- consider only unidirectional data transfer
  - but control info will flow on both directions!
- use finite state machines (FSM) to specify sender, receiver

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

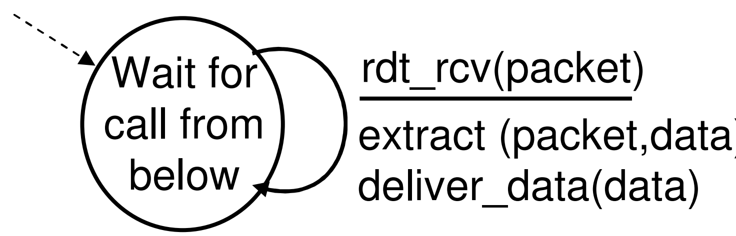


Rdt1.0: reliable transfer over a reliable channel

- underlying channel perfectly reliable
  - no bit errors
  - no loss of packets (→no congestion, no buffer overflows)
- separate FSMs for sender, receiver:
  - sender sends data into underlying channel
  - receiver read data from underlying channel



sender

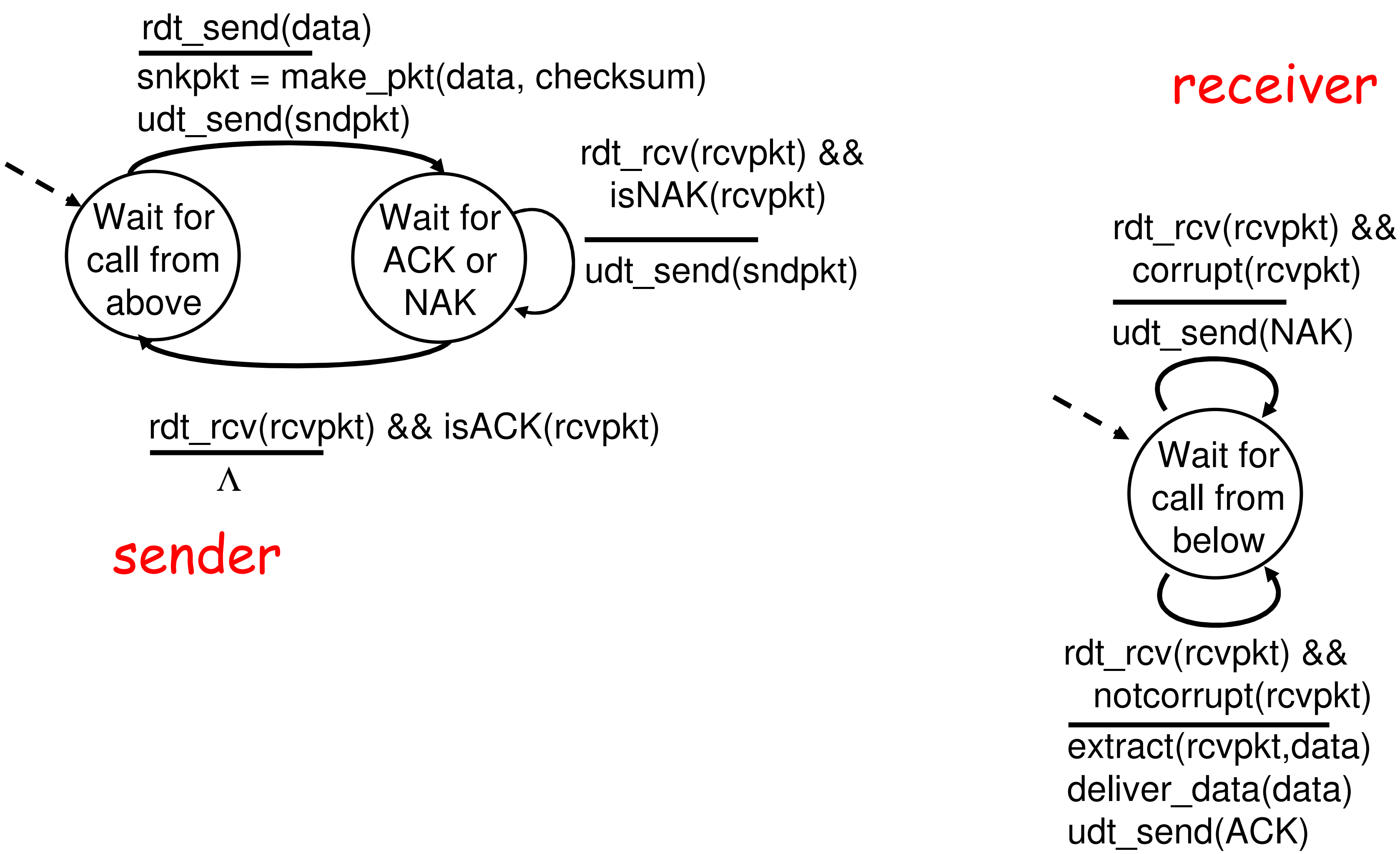


receiver

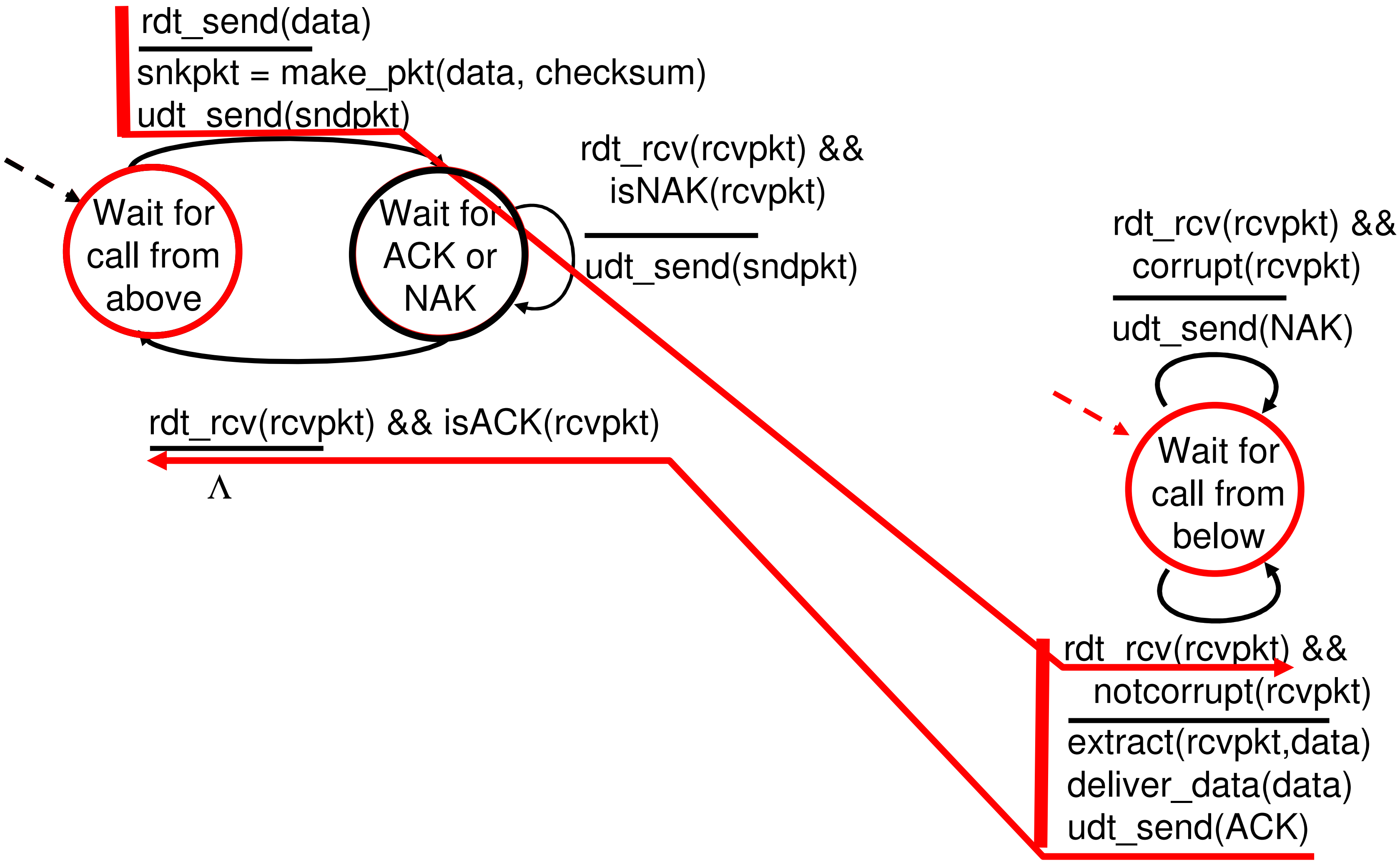
Rdt2.0: channel with bit errors

- underlying channel may flip bits in packet
  - recall: UDP checksum to detect bit errors
- **Still no loss!!**
- *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
  - human scenarios using ACKs, NAKs?
- new mechanisms in **`rdt2.0`** (beyond **`rdt1.0`**):
  - error detection
  - receiver feedback: control msgs (ACK,NAK) rcvr->sender

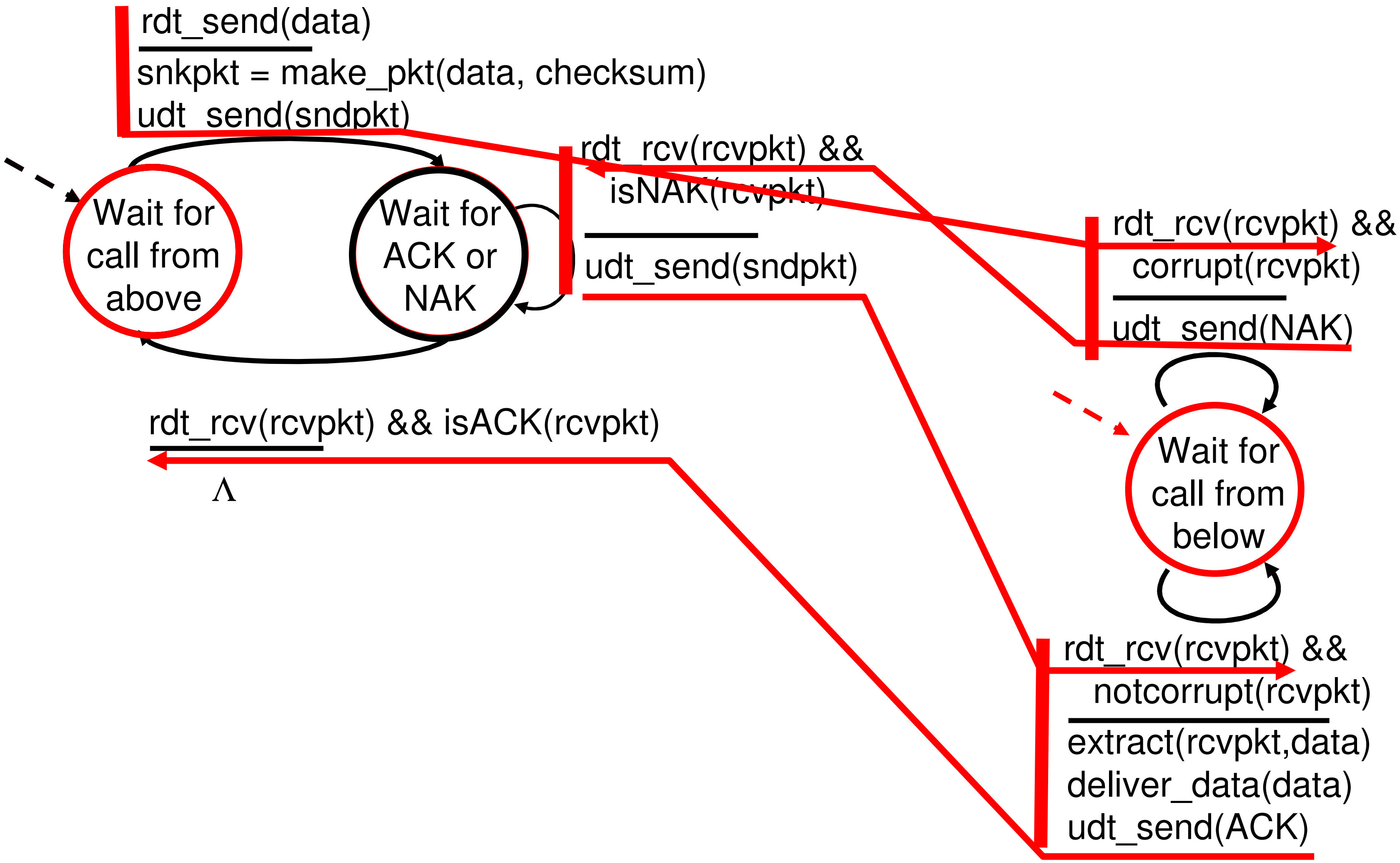
rdt2.0: FSM specification



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!
- can't just retransmit: possible duplicate

What to do?

- sender ACKs/NAKs receiver's ACK/NAK? What if sender ACK/NAK lost?
- retransmit, but this might cause retransmission of correctly received pkt!

Handling duplicates:

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

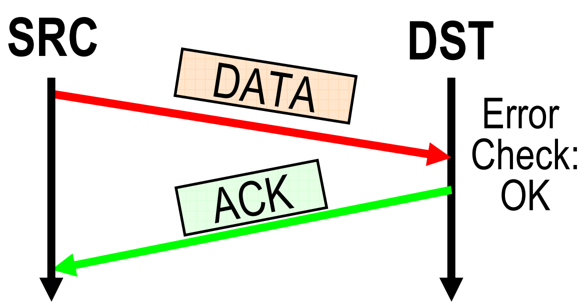
stop and wait

Sender sends one packet, then waits for receiver response

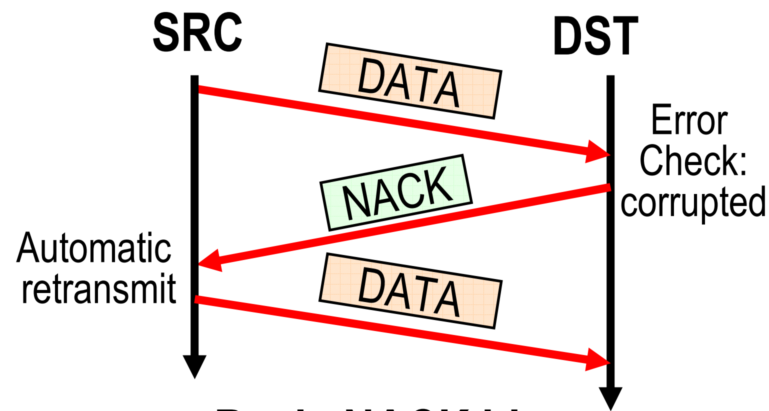
Retransmission scenarios

referred to as ARQ schemes (Automatic Retransmission reQuest)

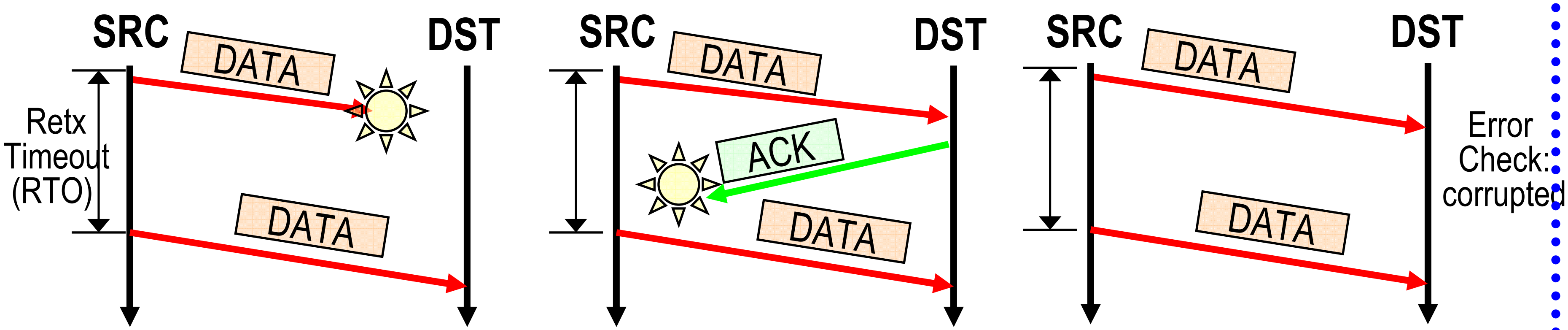
COMPONENTS: a) error checking at receiver; b) feedback to sender; c) retx



Basic ACK idea

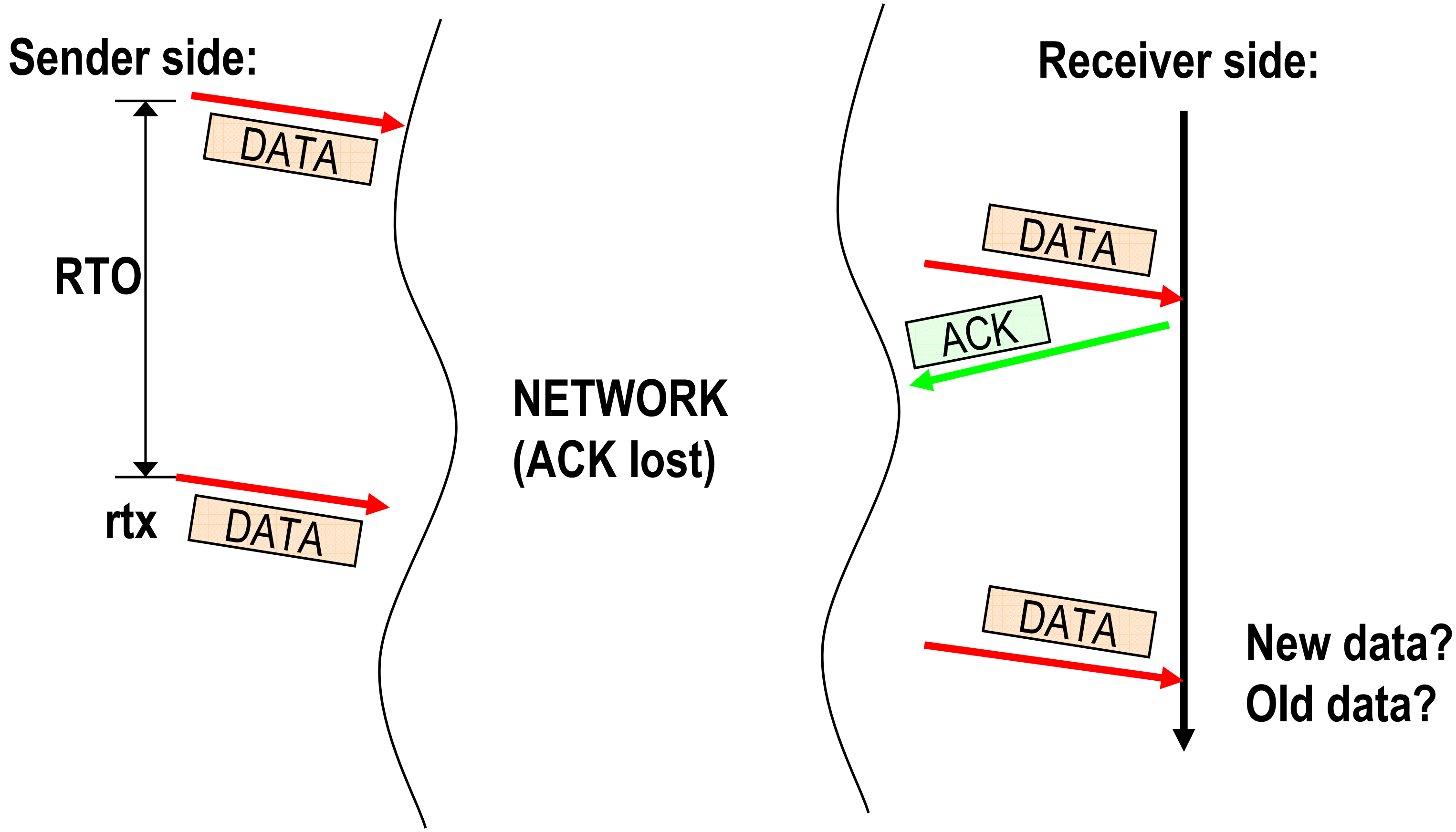


Basic NACK idea



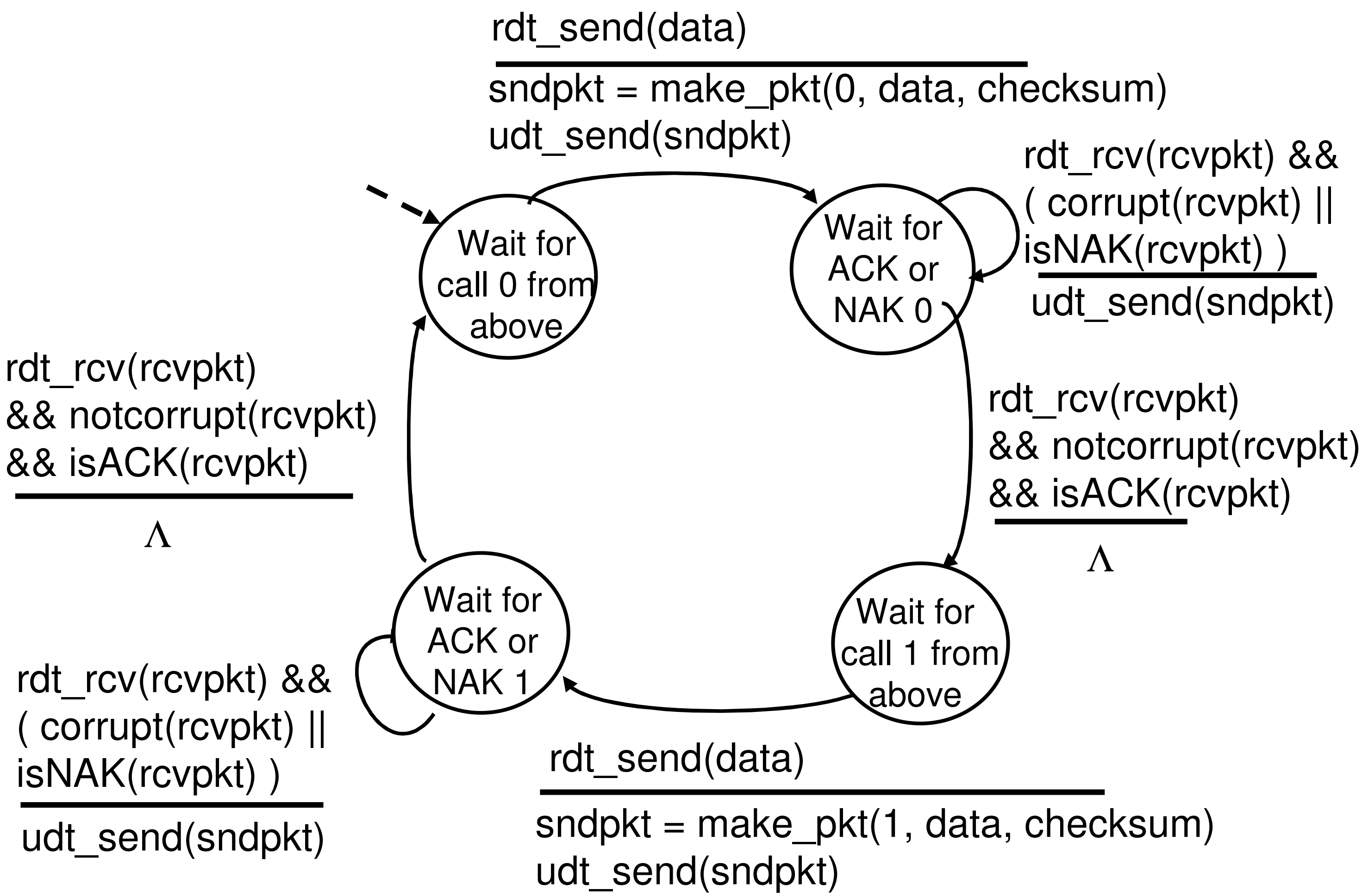
Basic ACK/Timeout idea

Why sequence numbers?
(on data)

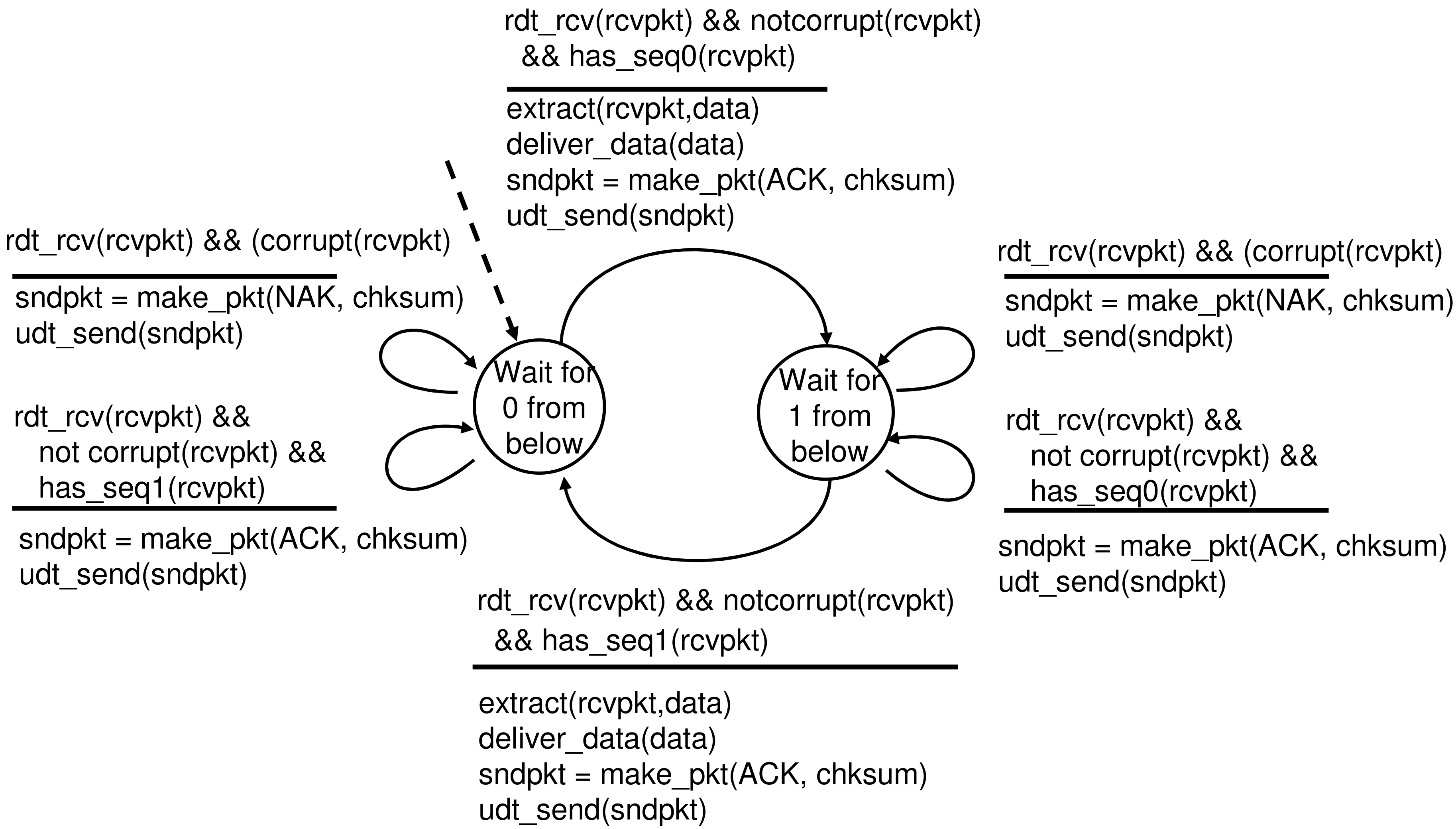


Need to univocally "label" all packets circulating in the network between two end points.
1 bit (0-1) enough for Stop-and-wait

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. #

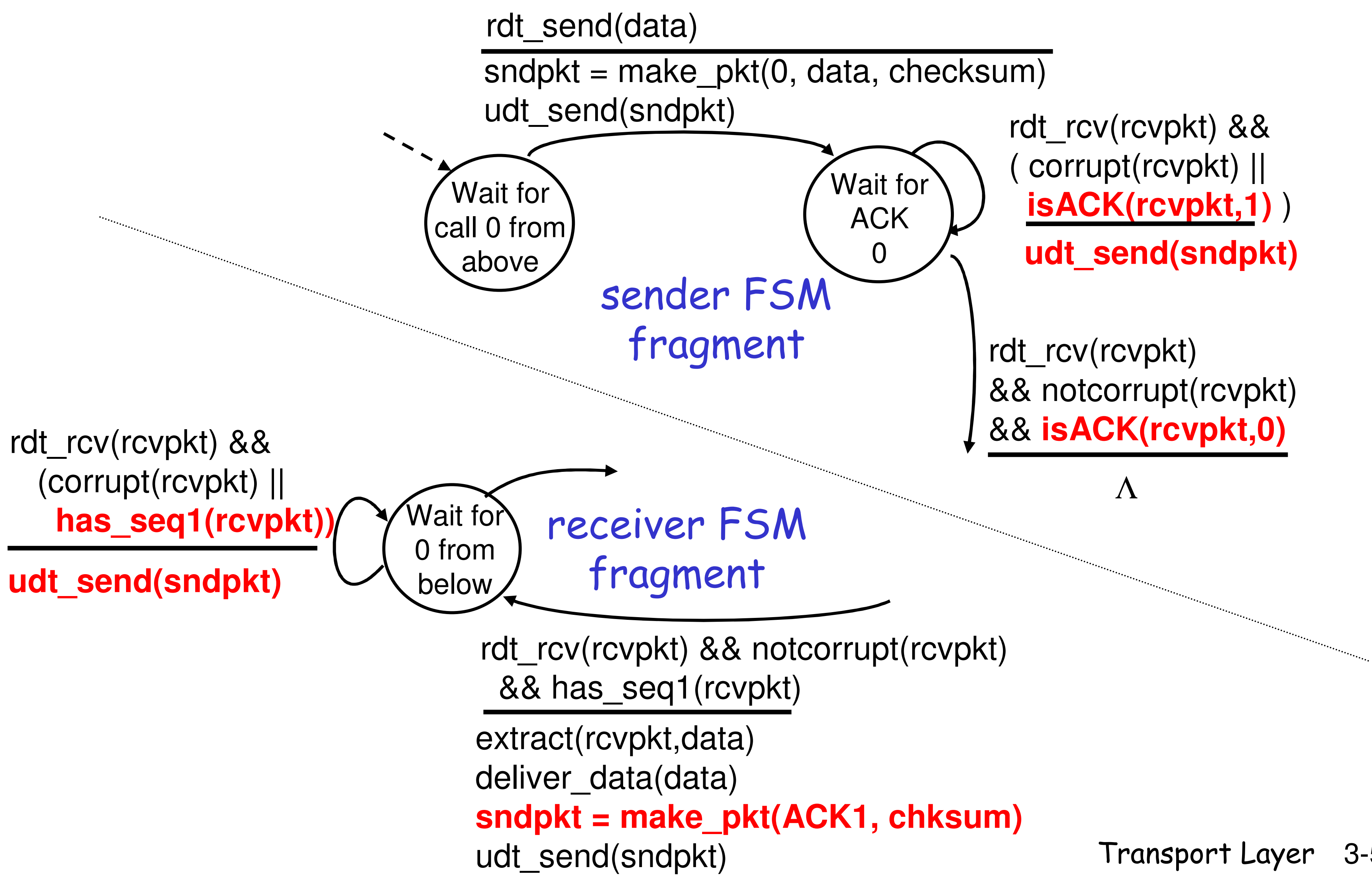
Receiver:

- must check if received packet is duplicate
  - state indicates whether 0 or 1 is expected pkt seq #
- 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 NAKs 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*

rdt2.2: sender, receiver fragments



rdt3.0: channels with errors *and* loss

New assumption: underlying channel can also loose packets (data or ACKs)

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

Q: how to deal with loss?

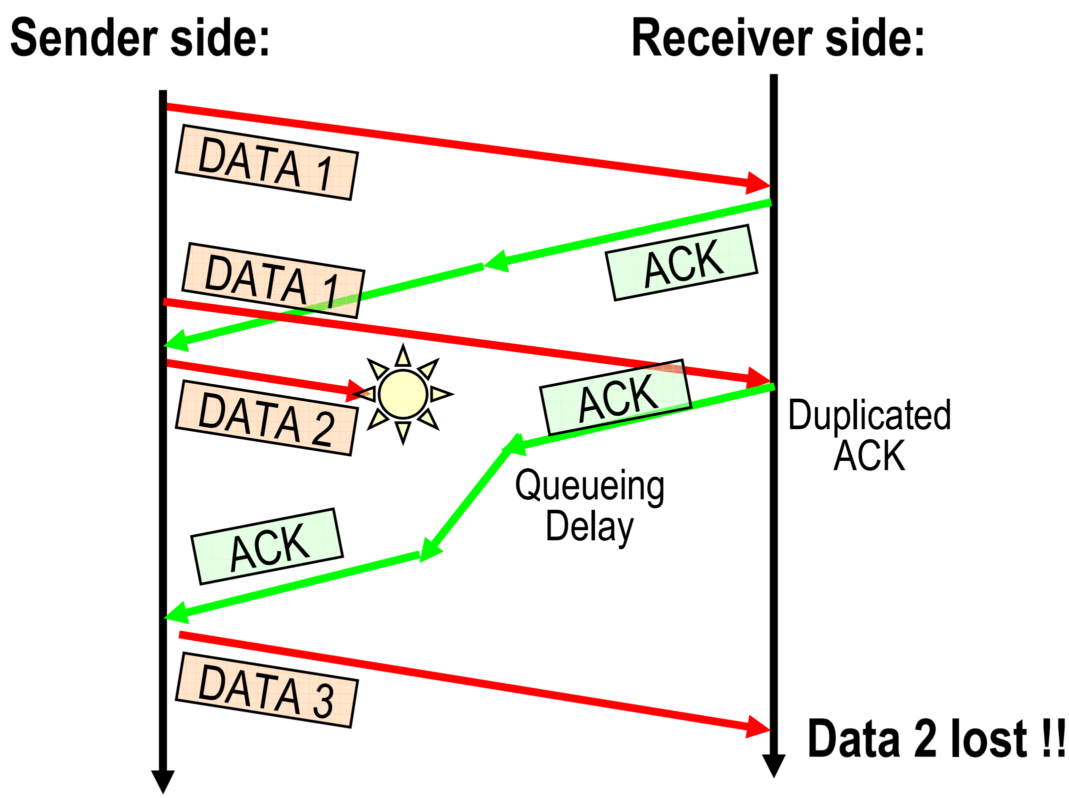
- sender waits until certain data or ACK lost, then retransmits
- yuck: drawbacks?

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 use of seq. #'s already handles this
  - receiver must specify seq # of pkt being ACKed
- requires countdown timer

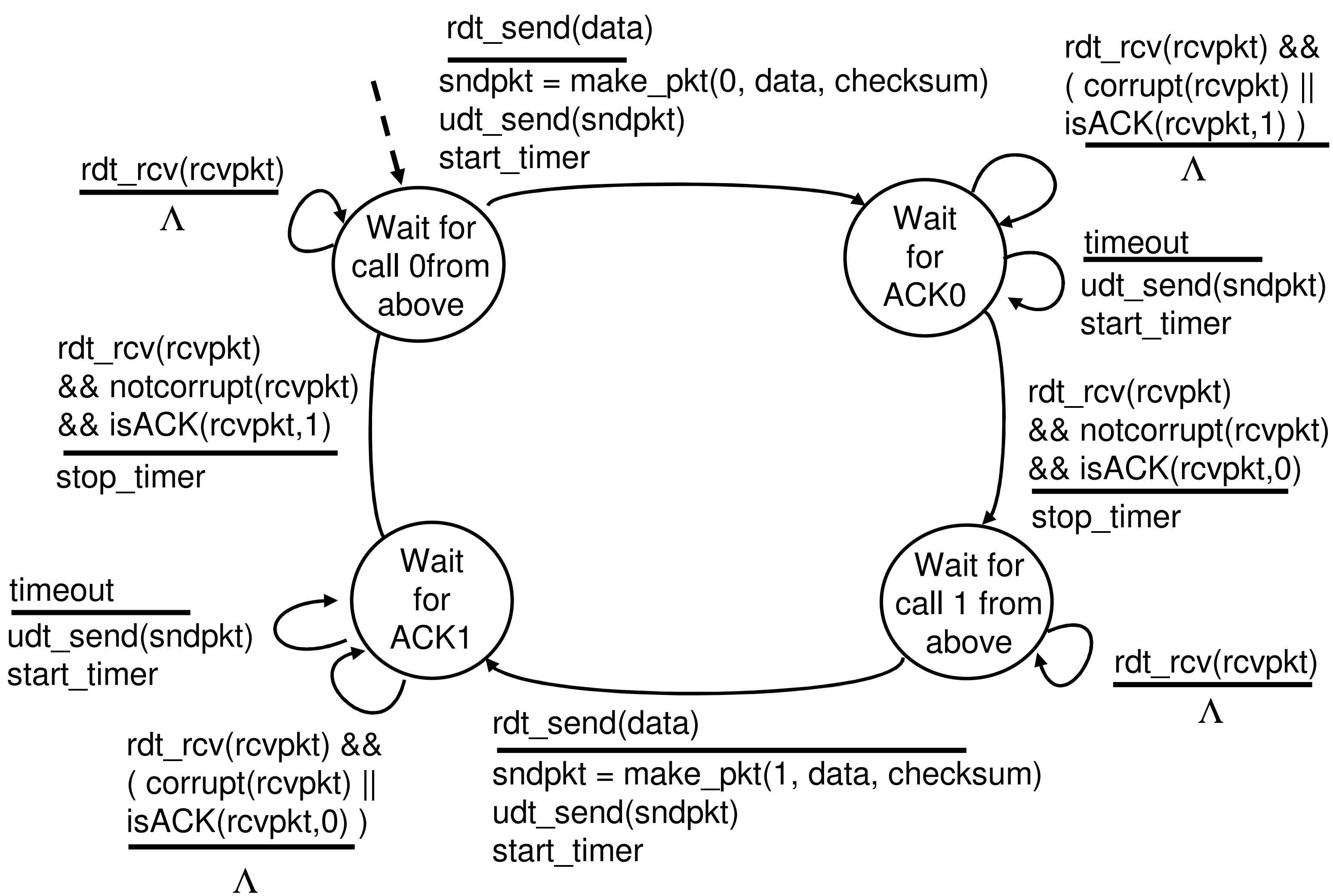
Why sequence numbers?

(on ack)

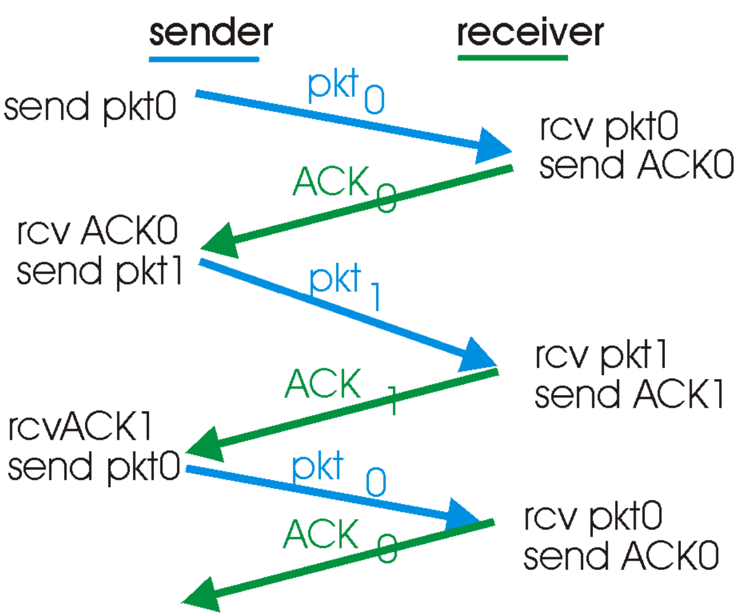


With pathologically critical network (as the Internet!)
also need to univocally "label" all acks circulating
in the network between two end points.
1 bit (0-1) enough for Stop-and-wait ?

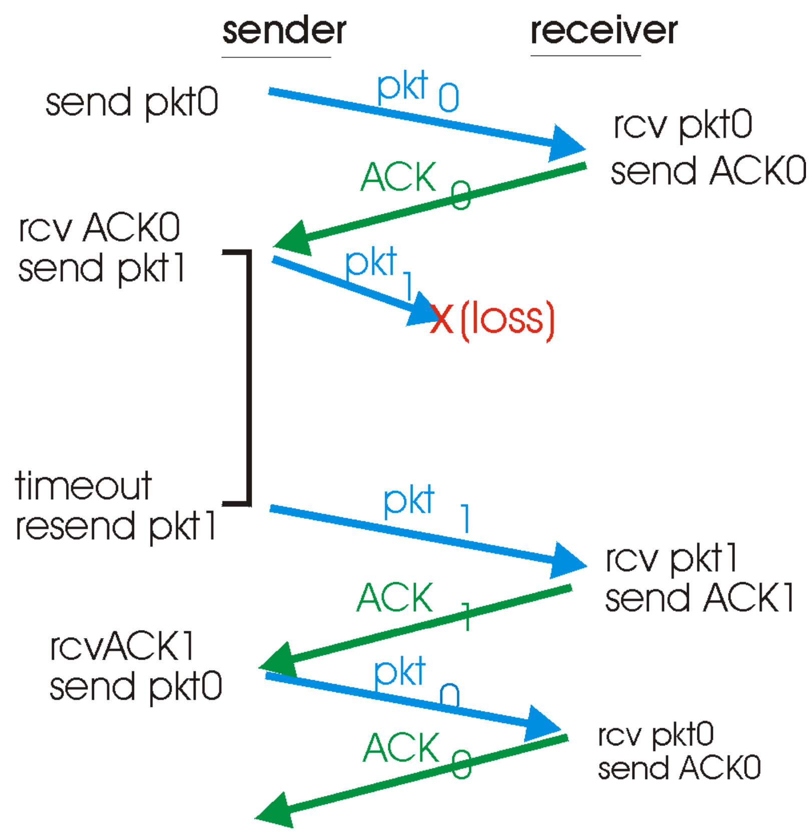
rdt3.0 sender



rdt3.0 in action

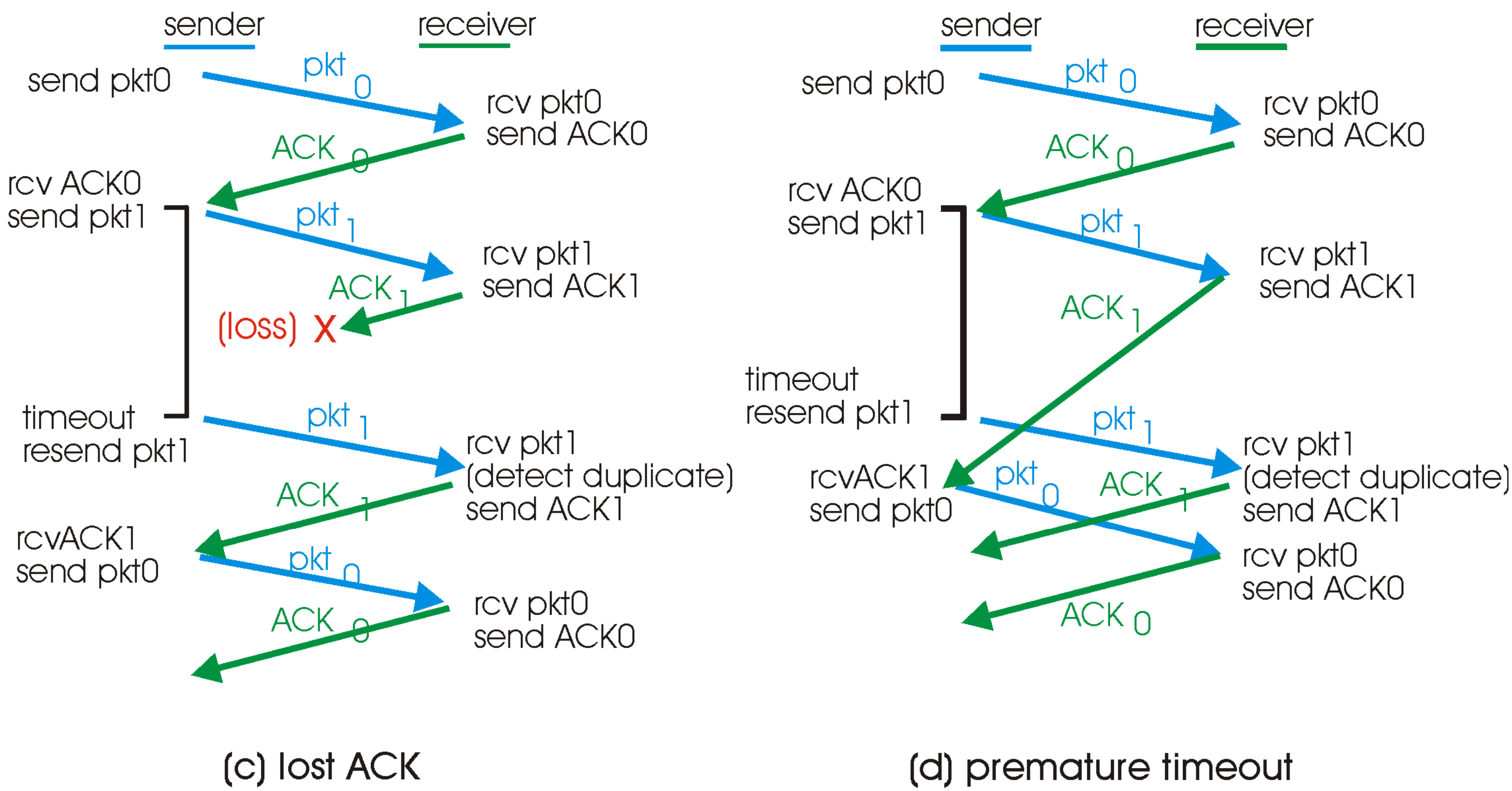


(a) operation with no loss



(b) lost packet

rdt3.0 in action



Performance of rdt3.0

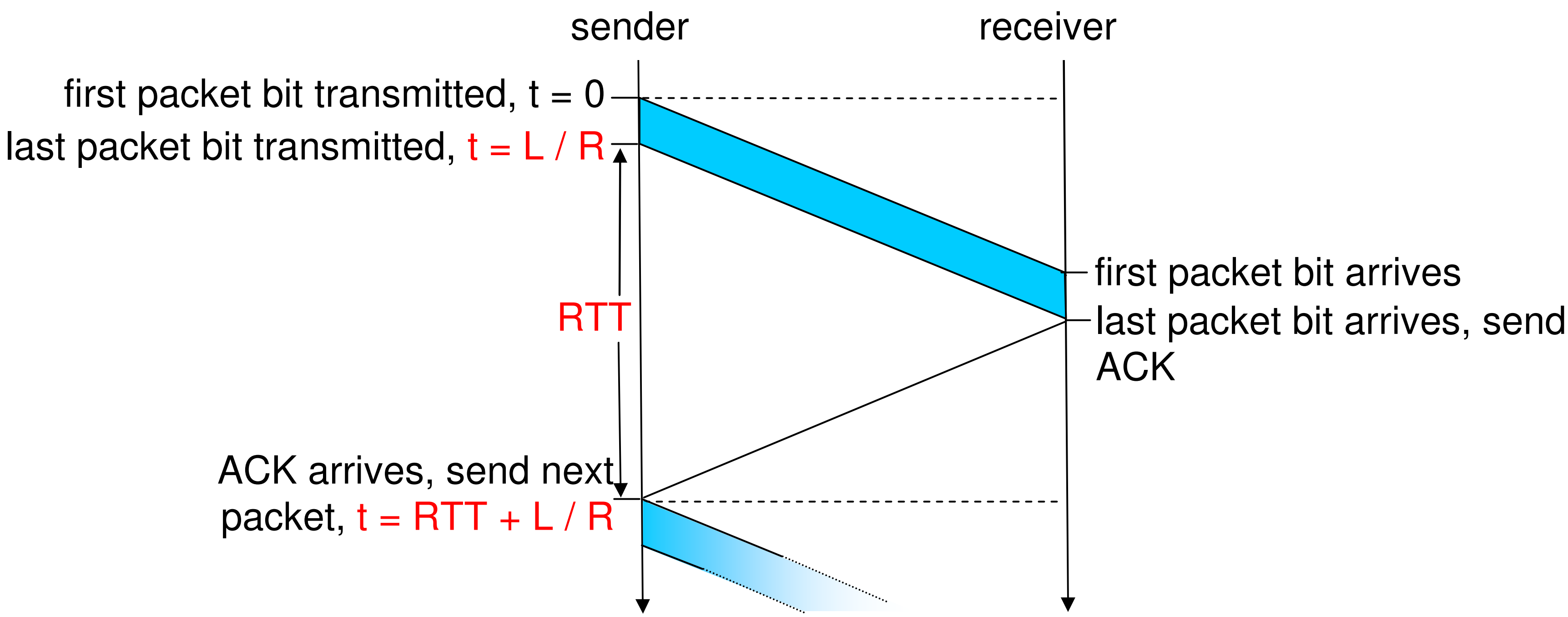
- rdt3.0 works, but performance stinks
- example: 1 Gbps link, 15 ms e-e prop. delay, 1KB packet:

$$T_{transmit} = \frac{L \text{ (packet length in bits)}}{R \text{ (transmission rate, bps)}} = \frac{8\text{kb/pkt}}{10^{**}9 \text{ b/sec}} = 8 \text{ microsec}$$

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

  - U_{sender}: utilization – fraction of time sender busy sending
  - 1KB pkt every 30 msec -> 33kB/sec throuput over 1 Gbps link
  - network protocol limits use of physical resources!

rdt3.0: stop-and-wait operation

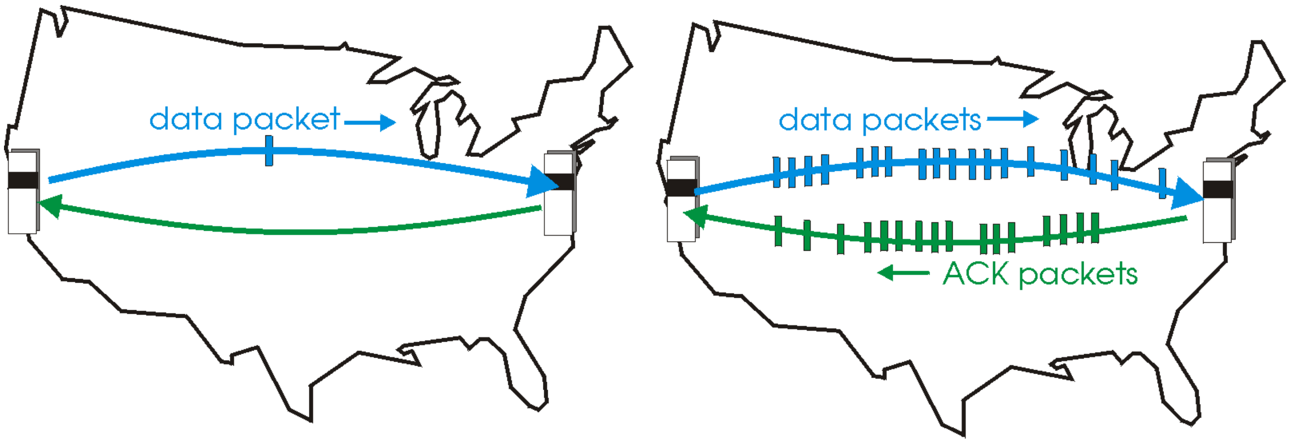


$$U_{sender} = \frac{L / R}{RTT + L / R} = \frac{.008}{30.008} = 0.00027$$

Pipelined protocols

Pipelining: sender allows multiple, “in-flight”, yet-to-be-acknowledged pkts

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



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

(b) a pipelined protocol in operation

- Two generic forms of pipelined protocols: *go-Back-N, selective repeat*