Distributed Systems

5. Transport Protocols

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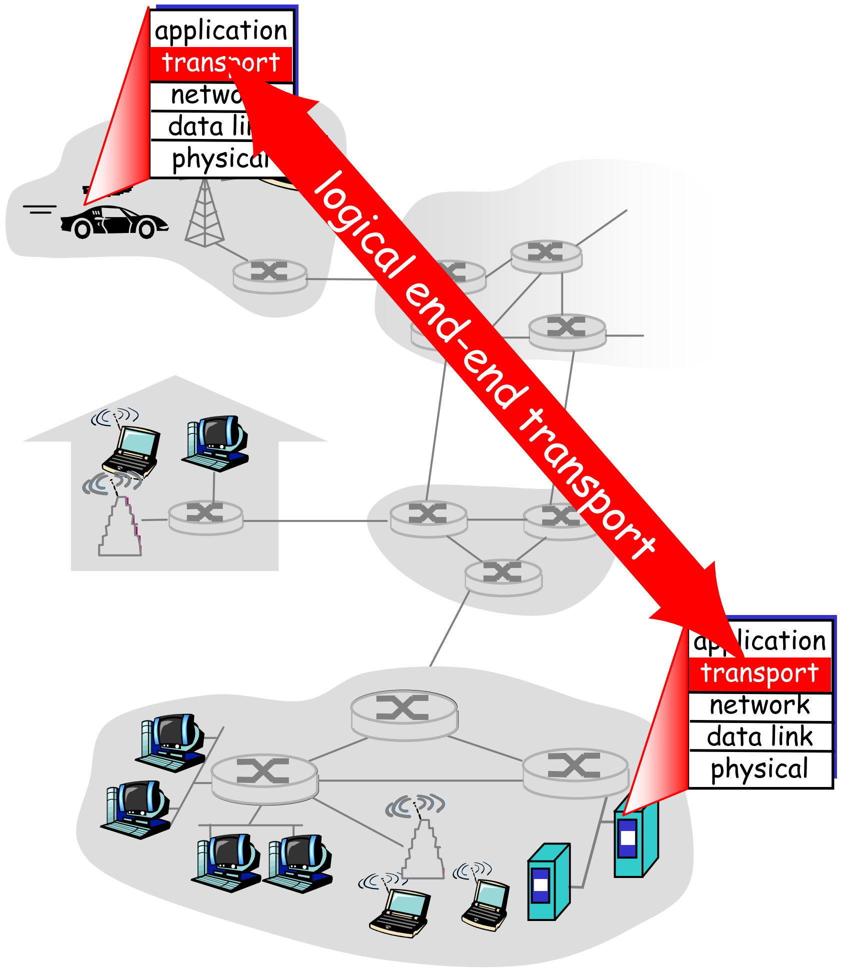
5. Transport Protocols

5.1 Transport-layer Services

Transport Services and Protocols

- Provide *communication* between application processes running on different hosts
- Transport protocols run in end systems
  - send side: breaks application messages into segments, passes to network layer
  - receive side: reassembles segments into messages, passes to application layer
- Two transport protocol available to Internet applications
  - TCP and UDP



Transport vs. Network Layer

- *Network layer:* communication between hosts
- *Transport layer:* communication between processes
  - relies on, enhances, network layer services

5. Transport Protocols

5.2 Multiplexing and Demultiplexing

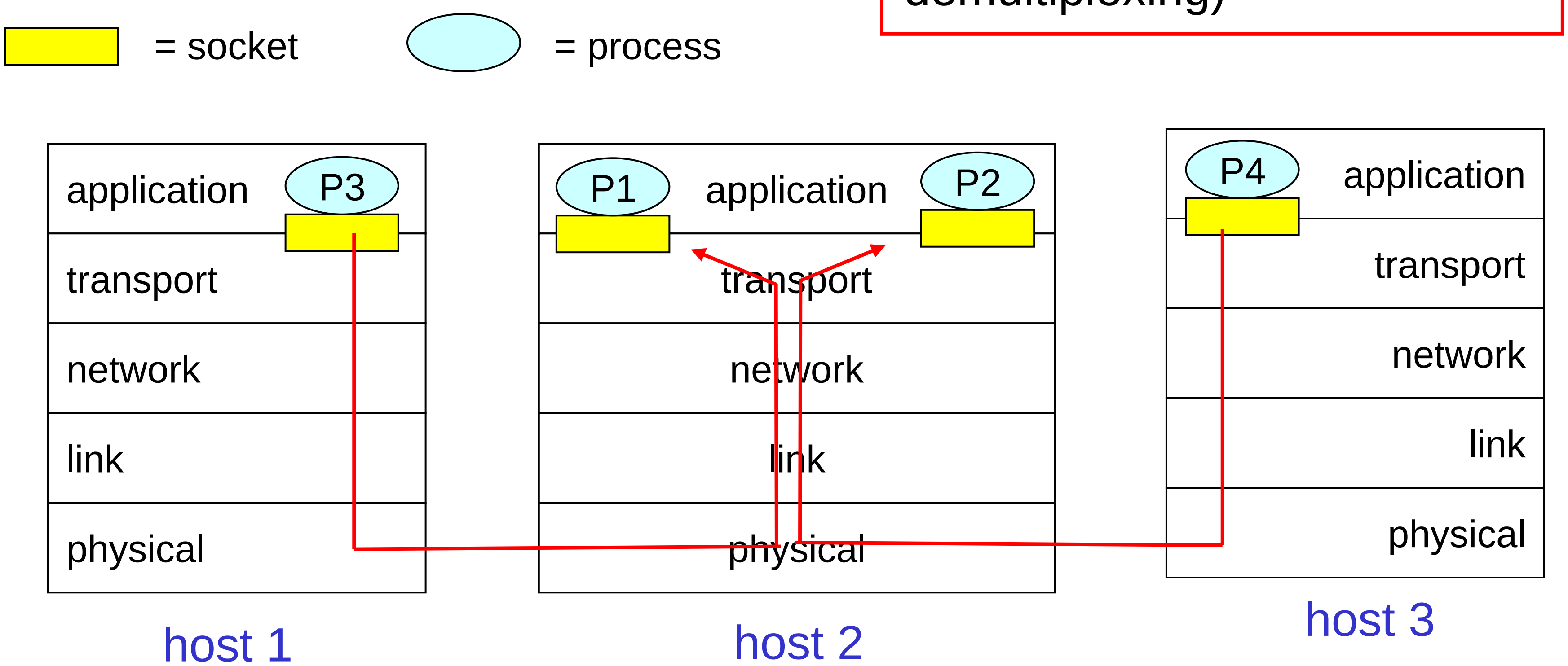
Multiplexing/Demultiplexing

Demultiplexing at receive host:

delivering received segments to correct socket

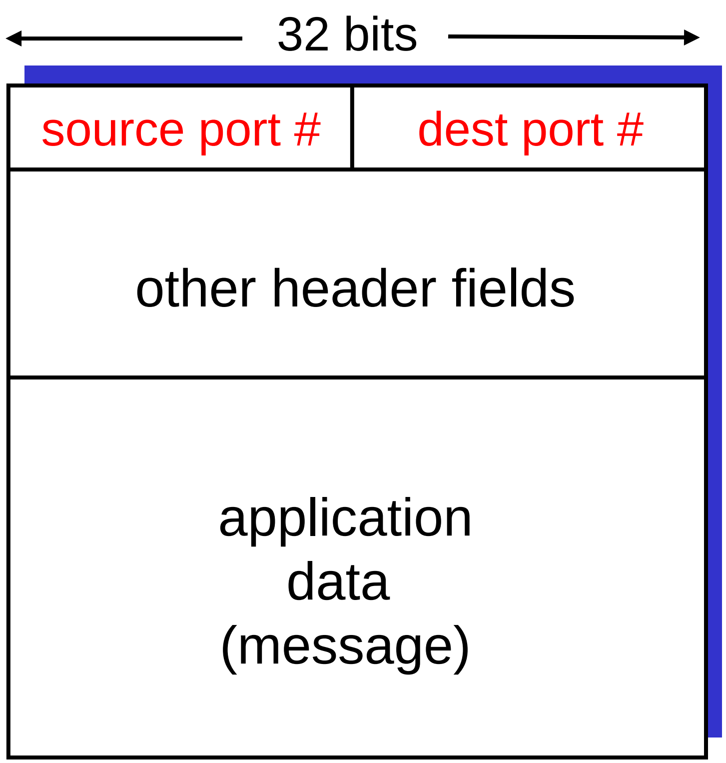
Multiplexing at send host:

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
- Host uses IP addresses and port numbers to direct segment to appropriate socket



TCP/UDP segment format

Connectionless Demultiplexing

- Create sockets with port numbers:

```
DatagramSocket mySocket1 =
  new DatagramSocket(12534);
DatagramSocket mySocket2 =
  new DatagramSocket(12535);
```

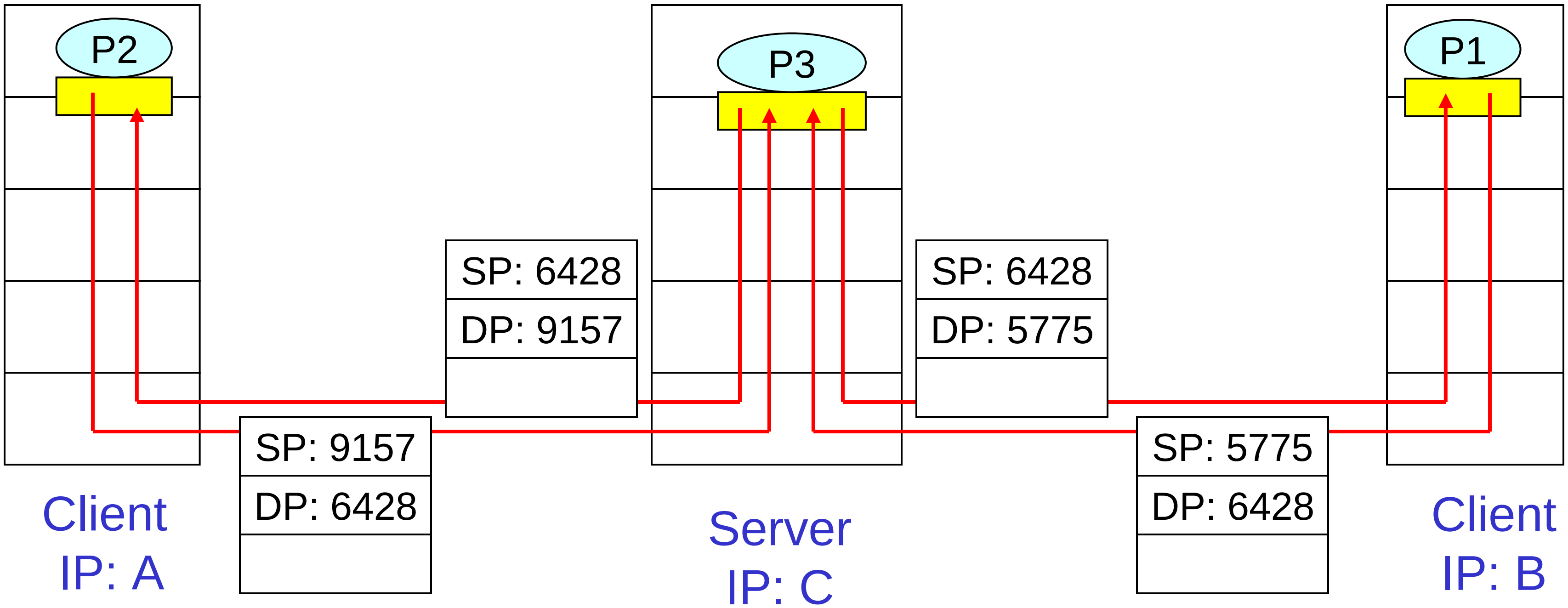
- UDP socket identified by 2-tuple:

  (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 Demultiplexing (cntd)

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



SP provides “return address”

Connection-oriented Demultiplexing

- A TCP socket is identified by a 4-tuple:
  - source IP address
  - source port number
  - dest IP address
  - dest port number
- Receiving 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 a different socket for each request

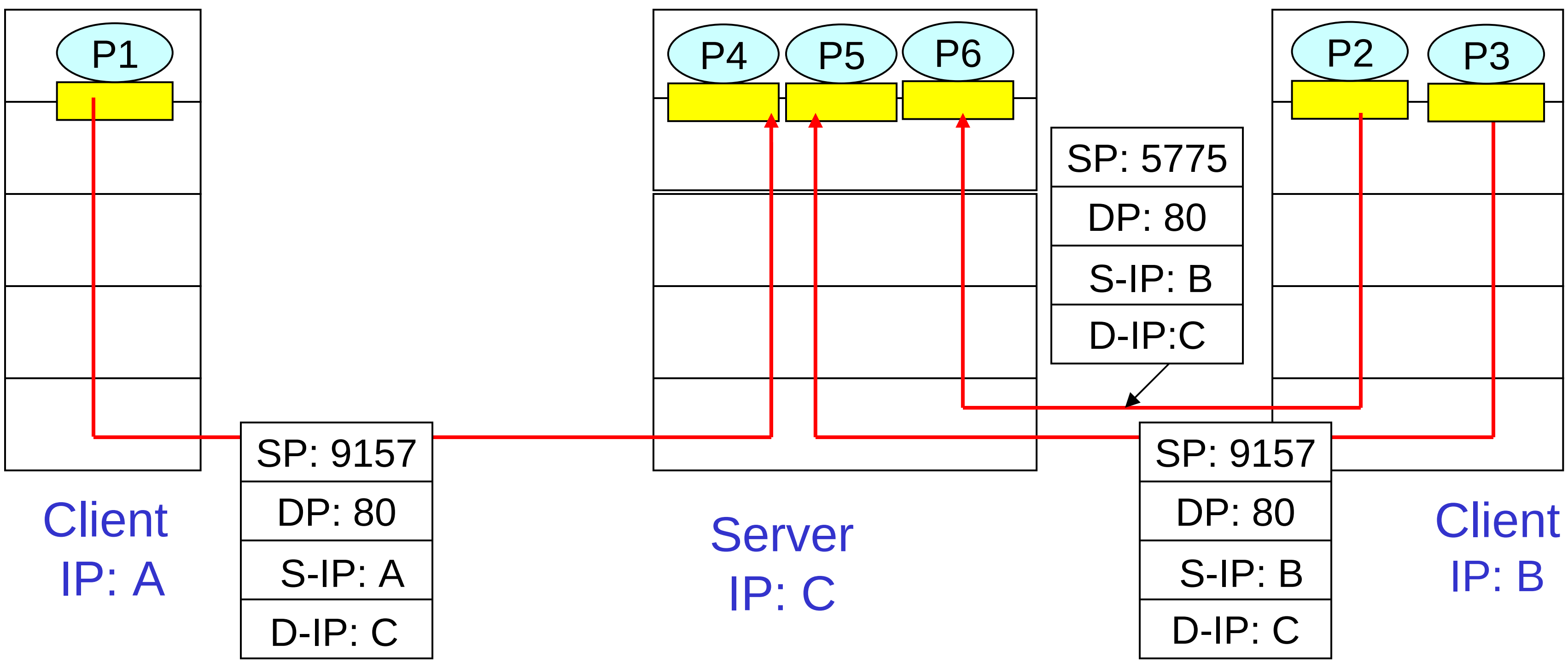
Exercise: In Firefox, type

  `about:config`

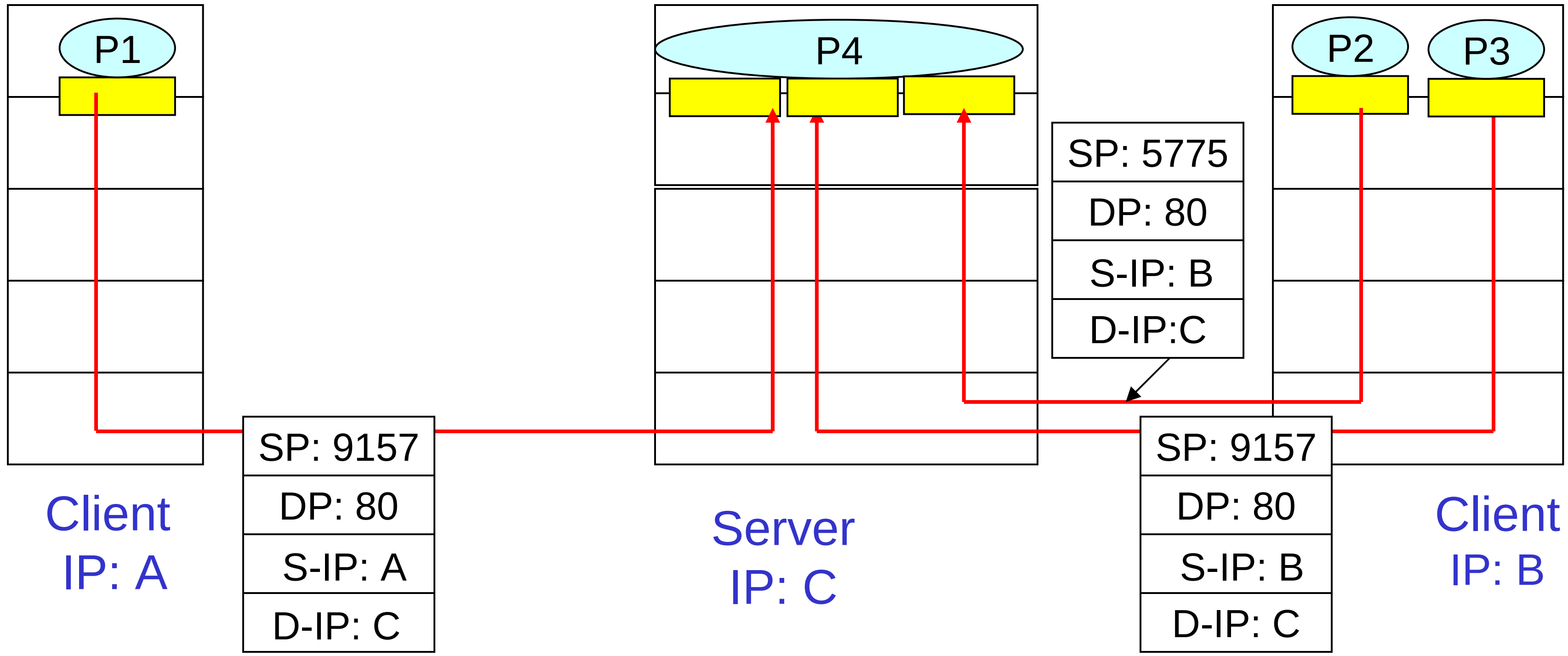
and check out

  `network.http.max-connections-per-server`

Connection-oriented Demultiplexing (cntd)



Connection-oriented Demultiplexing: Threaded Server



5. Transport Protocols

5.3 Connectionless Transport: UDP

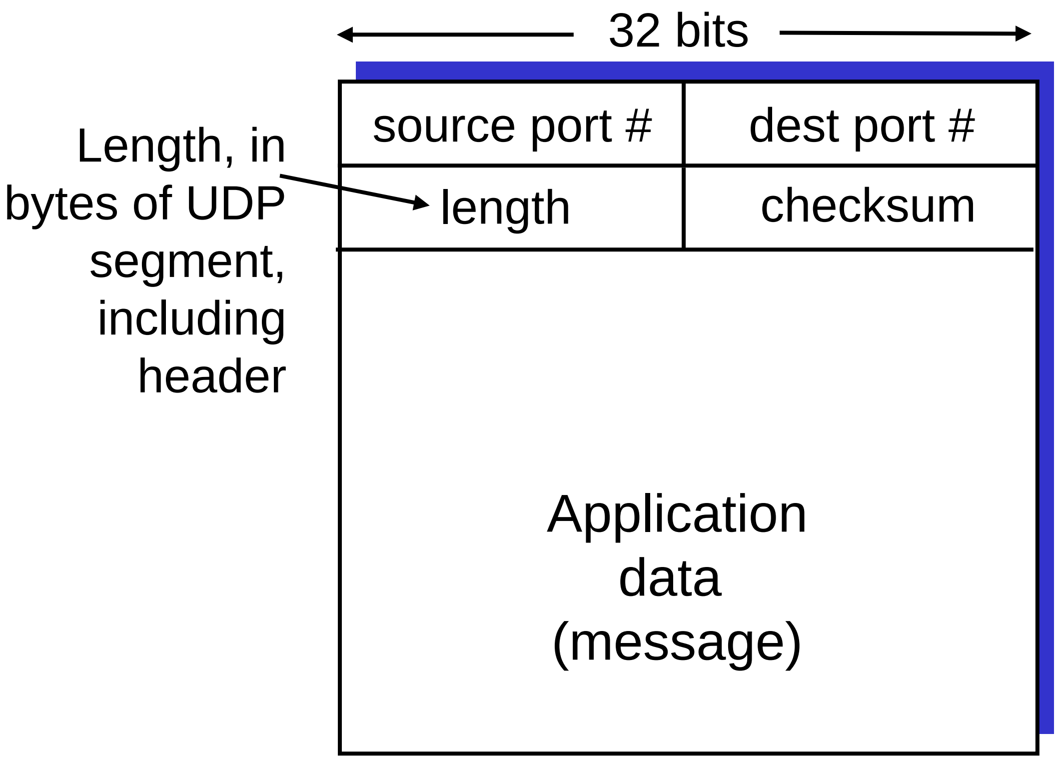
UDP: User Datagram Protocol [RFC 768]

- "No frills" Internet transport protocol
- "Best effort" service, UDP segments may be:
  - lost
  - delivered out of order to application
- *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
- no congestion control: UDP can blast away as fast as desired

UDP Segment Format



UDP Checksum

Goal: Detect "errors" (e.g., flipped bits) in transmitted segment

Sender:

- Treat segment contents as sequence of 16-bit integers
- checksum: addition (1's complement sum) of segment contents
- sender puts checksum value into UDP checksum field

Receiver:

- Compute checksum of received segment
- Check if computed checksum equals checksum field value:
  - NO - error detected
  - YES - no error detected. *But maybe errors nonetheless?*

Internet Checksum Example

- Note

  when adding numbers, a carry from the most significant bit needs to be added to the result
- 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
wraparound	1	1	0	1	1	1	0	1	1	1	0	1	1	1	0	1	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

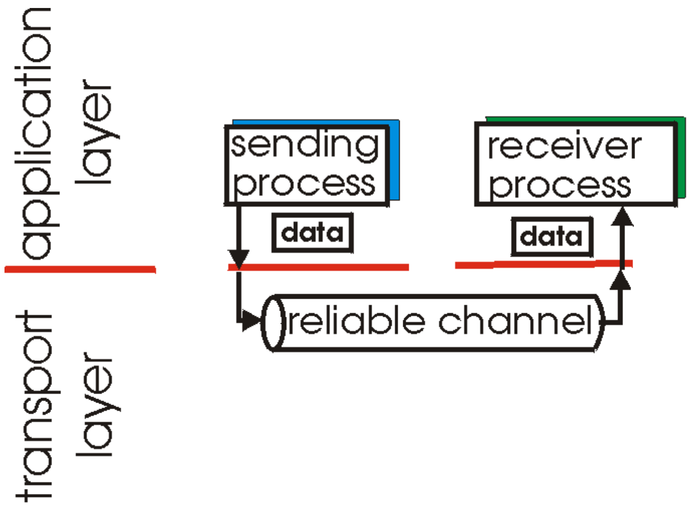
5. Transport Protocols

5.4 Principles of Reliable Data Transfer

5.1 Transport-layer Services
5.2 Multiplexing and Demultiplexing
5.3 Connectionless Transport: UDP
5.4 Principles of Reliable Data Transfer
5.5 Connection-oriented Transport: TCP

Principles of Reliable Data Transfer

- Important in application, transport, data link layers

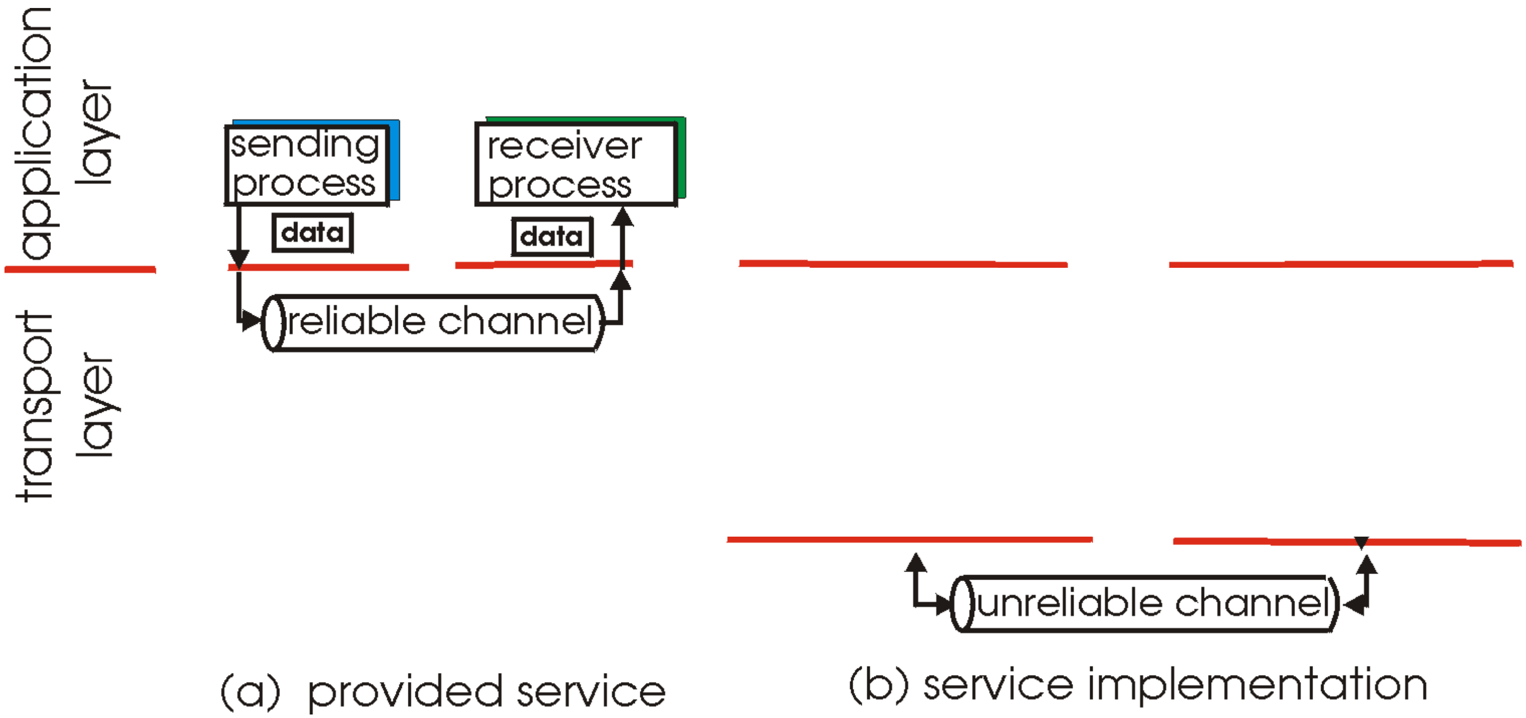


(a) provided service

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

Principles of Reliable Data Transfer

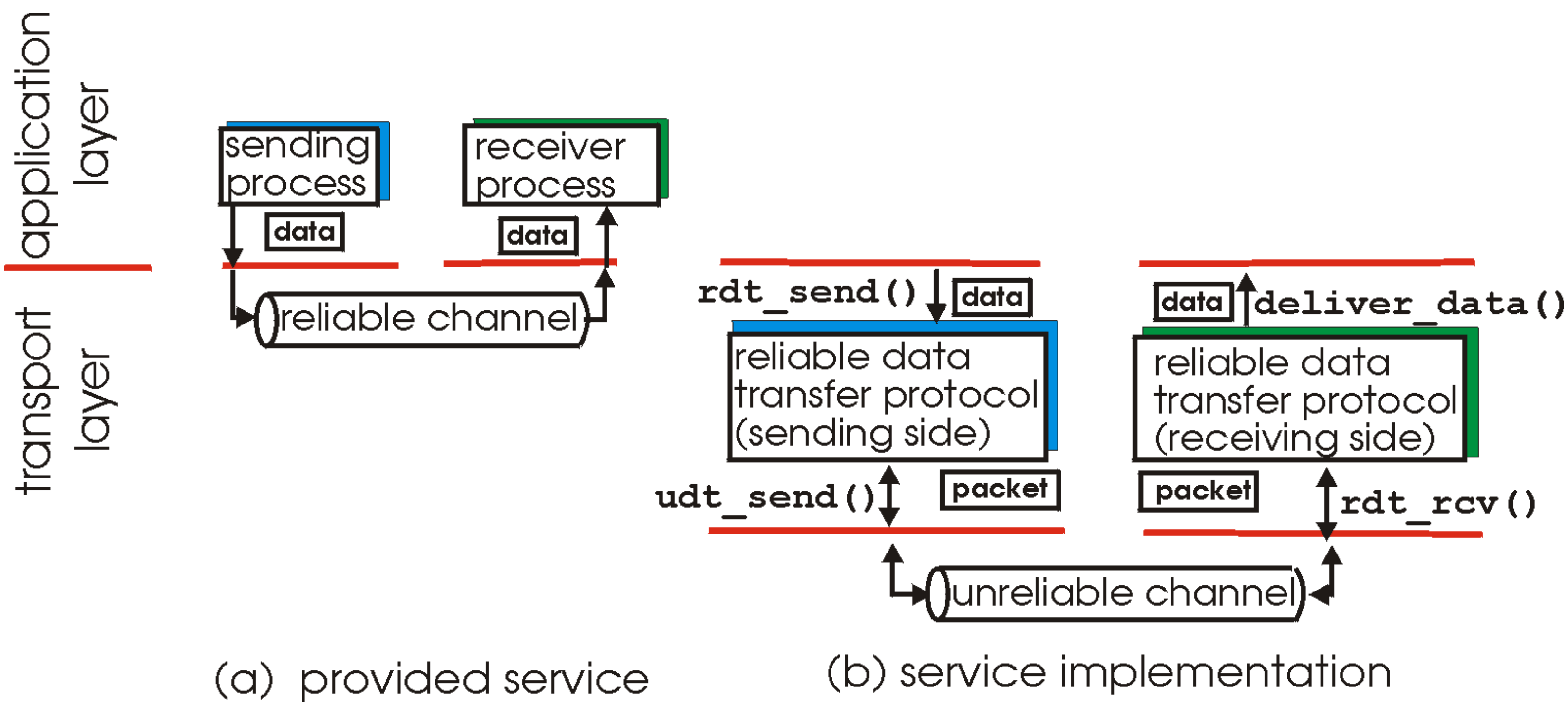
- Important in application, transport, data link layers



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

Principles of Reliable Data Transfer

- Important in application, transport, data link layers



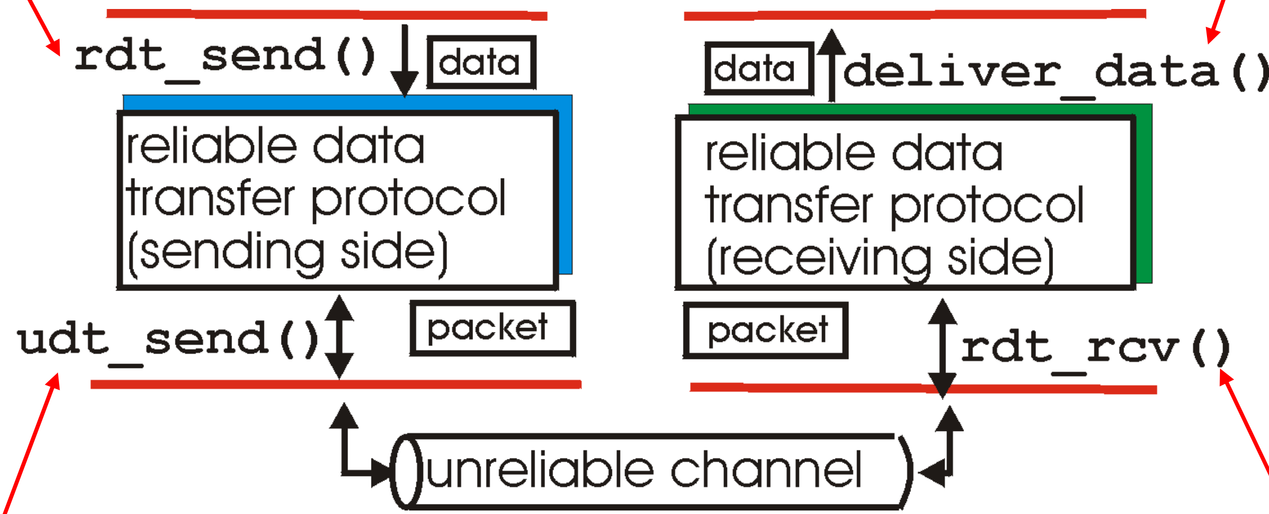
- Characteristics of unreliable channel will determine complexity of reliable data transfer protocol (RDT)

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

send side



receive side

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

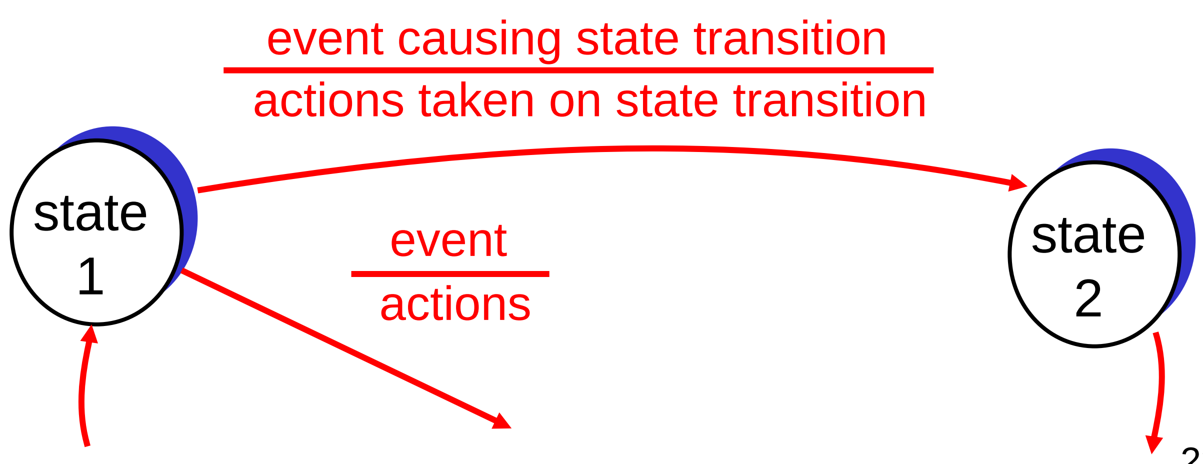
rdt_rcv(): called when packet arrives on rcv-side of channel

Reliable Data Transfer: Getting Started

We will:

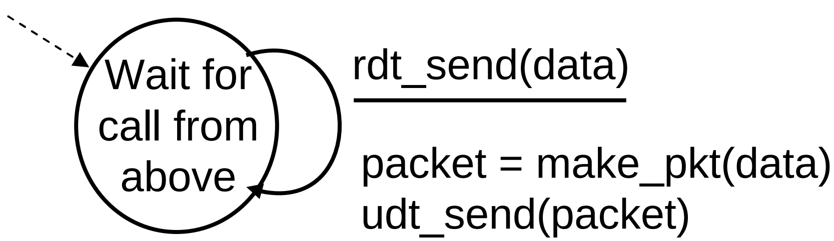
- incrementally develop the sender, receiver sides of a reliable data transfer protocol (RDT)
- consider only unidirectional data transfer
  - but control info will flow on both directions!
- use finite state machines (FSM) to specify sender, receiver

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

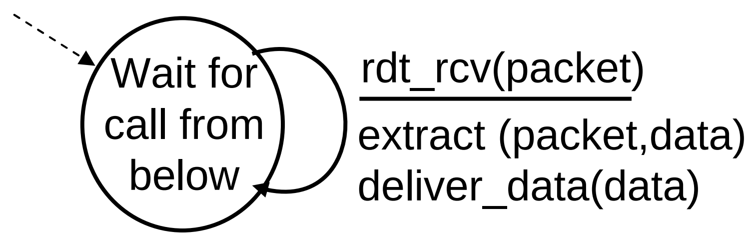


RDT1.0: Reliable Transfer over a Reliable Channel

- Underlying channel perfectly reliable
  - no bit errors
  - no loss of packets
- Separate FSMs for sender, receiver:
  - sender sends data into underlying channel
  - receiver reads data from underlying channel



Sender

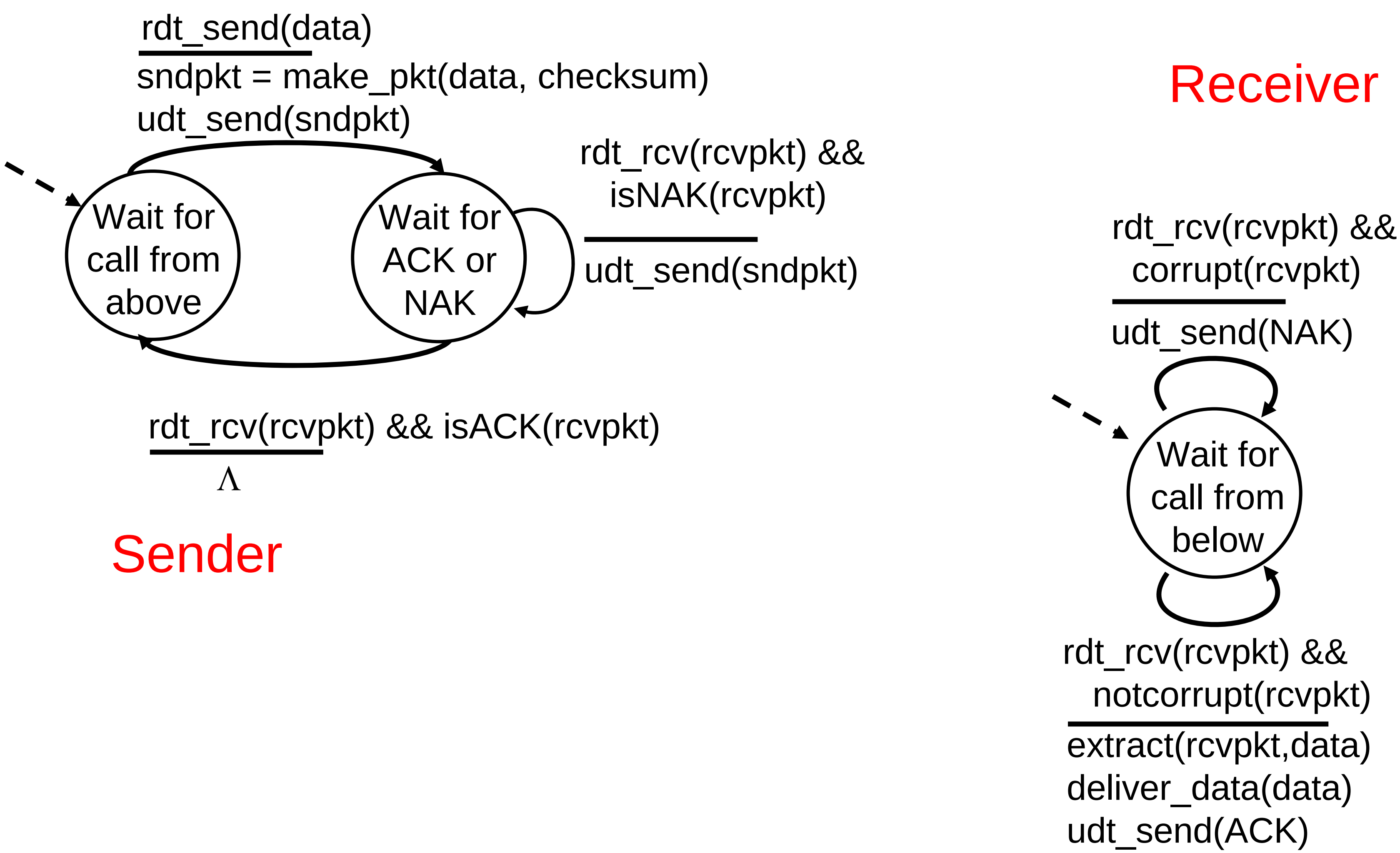


Receiver

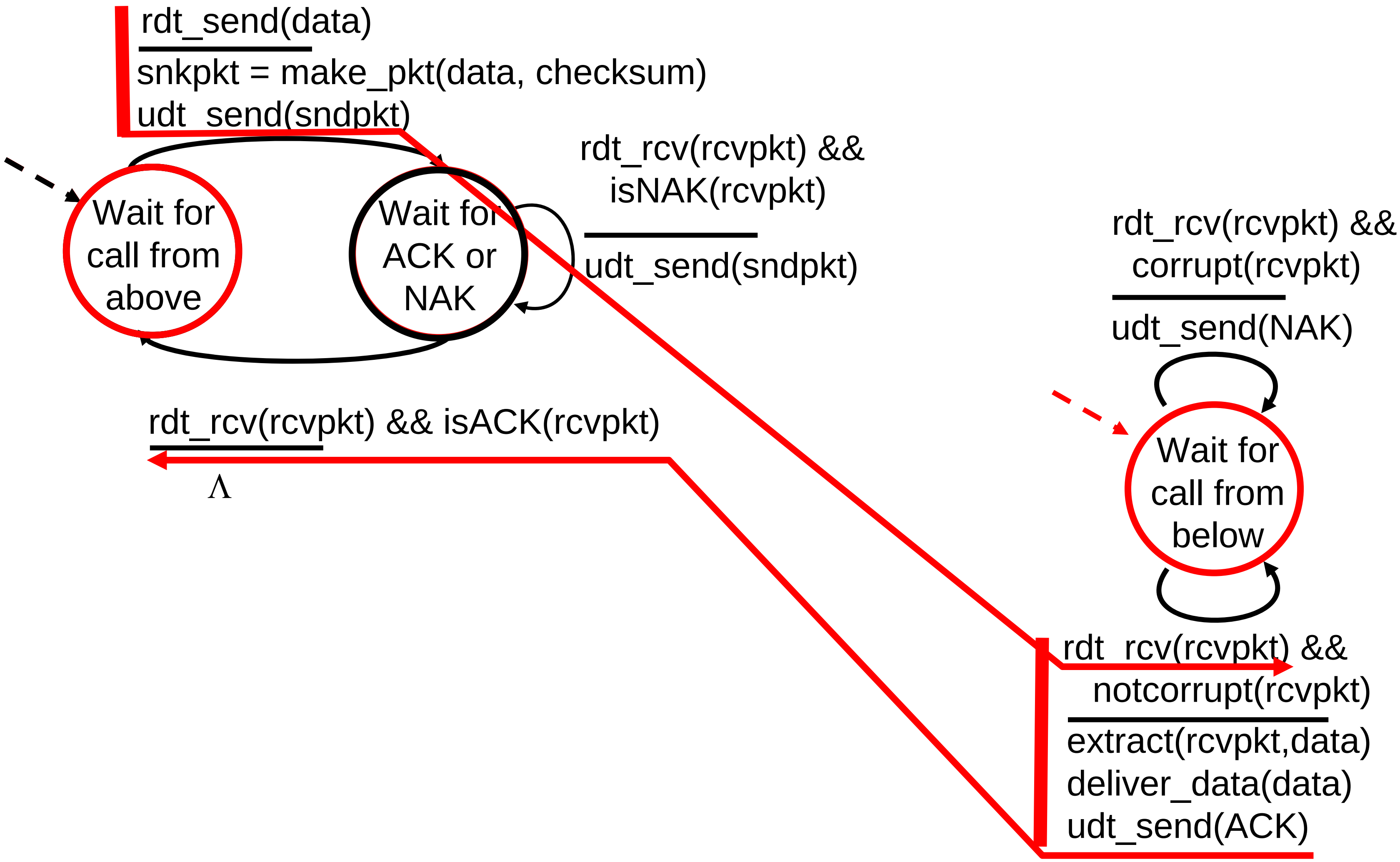
RDT2.0: Channel with Bit Errors

- 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 packet was received OK
  - *negative acknowledgements (NAKs):* receiver explicitly tells sender that packet had errors
  - sender retransmits packet on receipt of NAK
- New mechanisms in **RDT2.0** (beyond **RDT1.0**):
  - error detection
  - receiver feedback: control messages (ACK,NAK) receiver $\rightarrow$ sender

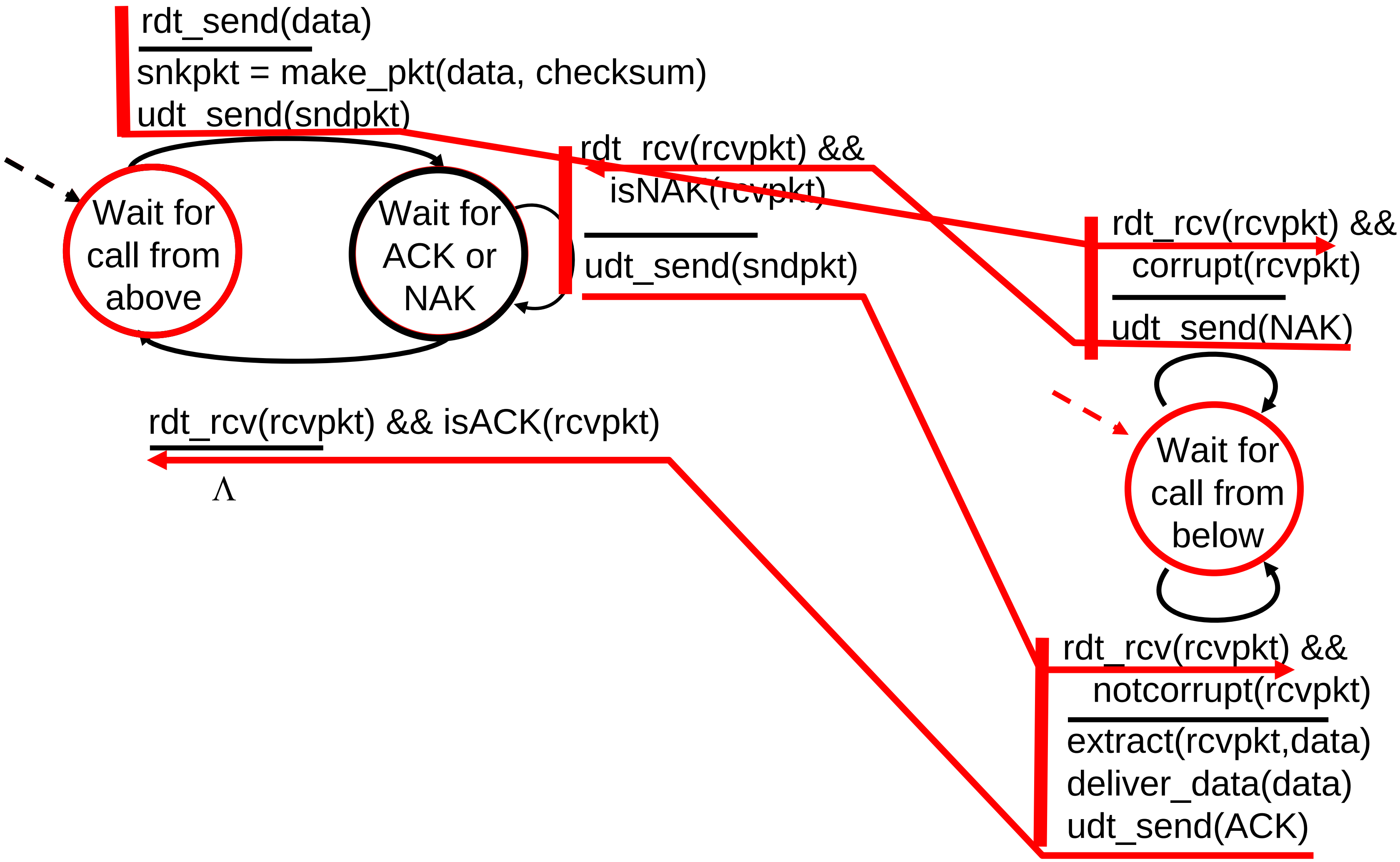
RDT2.0: FSM Specification



RDT2.0: Operation without Errors



RDT2.0: Error Scenario



RDT2.0 Has a Fatal Flaw!

- What happens if ACK/NAK is corrupted?
- Sender doesn't know what happened at the receiver!
- It can't just retransmit: possible duplicate

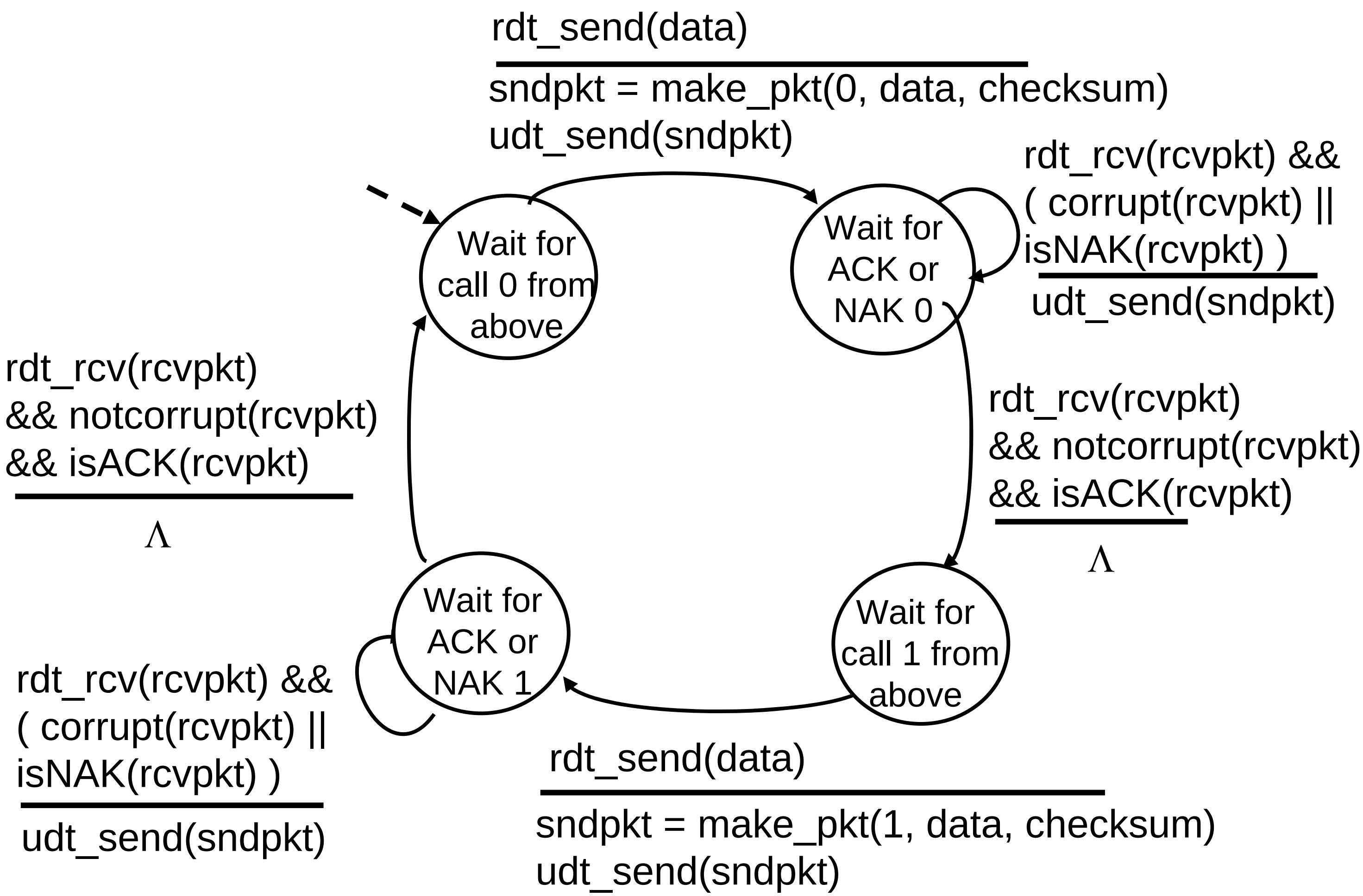
Handling duplicates:

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

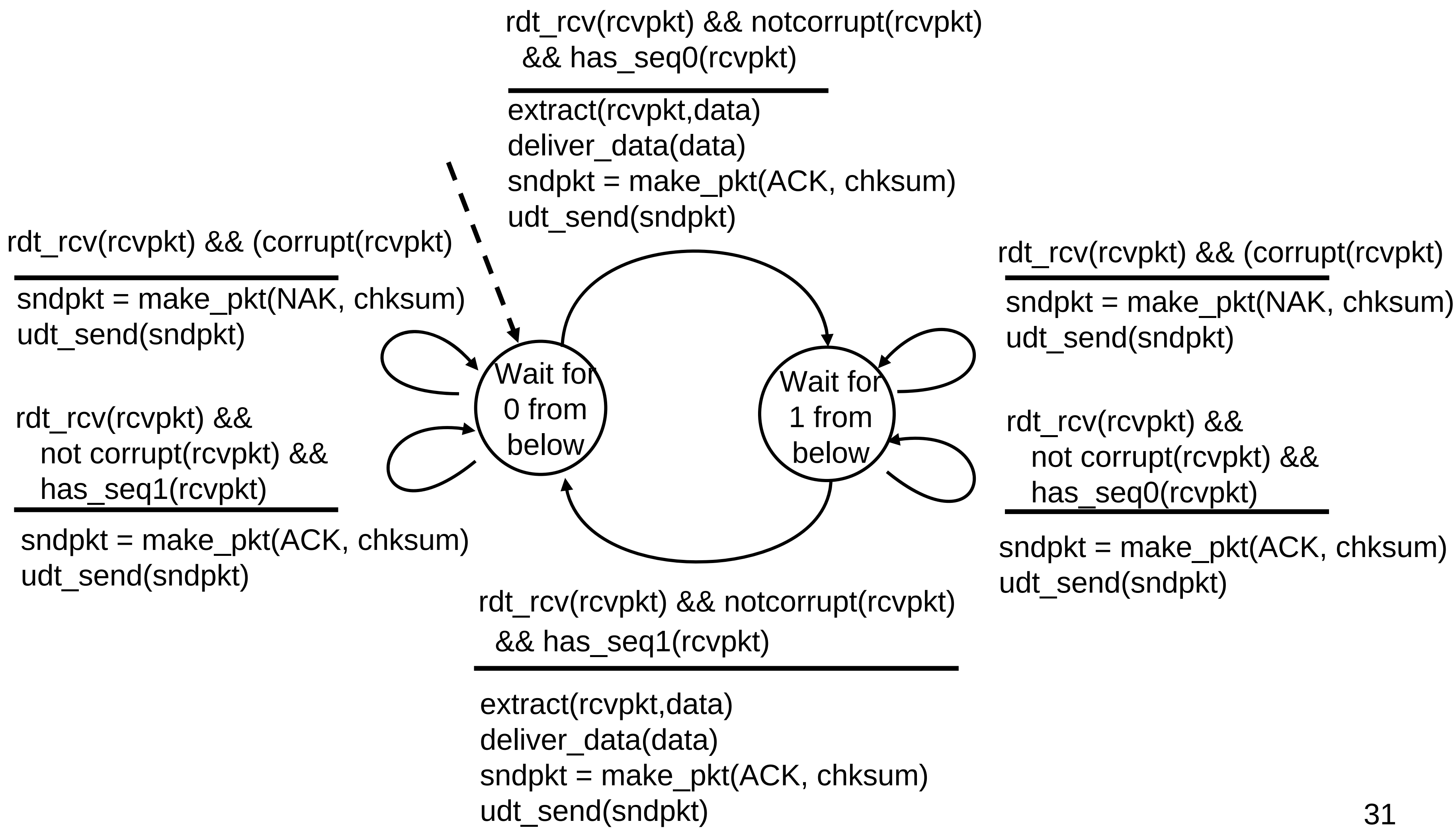
"Stop and Wait" Protocol

Sender sends one packet, then waits for receiver response

RDT2.1: Sender, Handles Corrupted ACK/NAKs



RDT2.1: Receiver, Handles Corrupted ACK/NAKs



RDT2.1: Discussion

Sender:

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

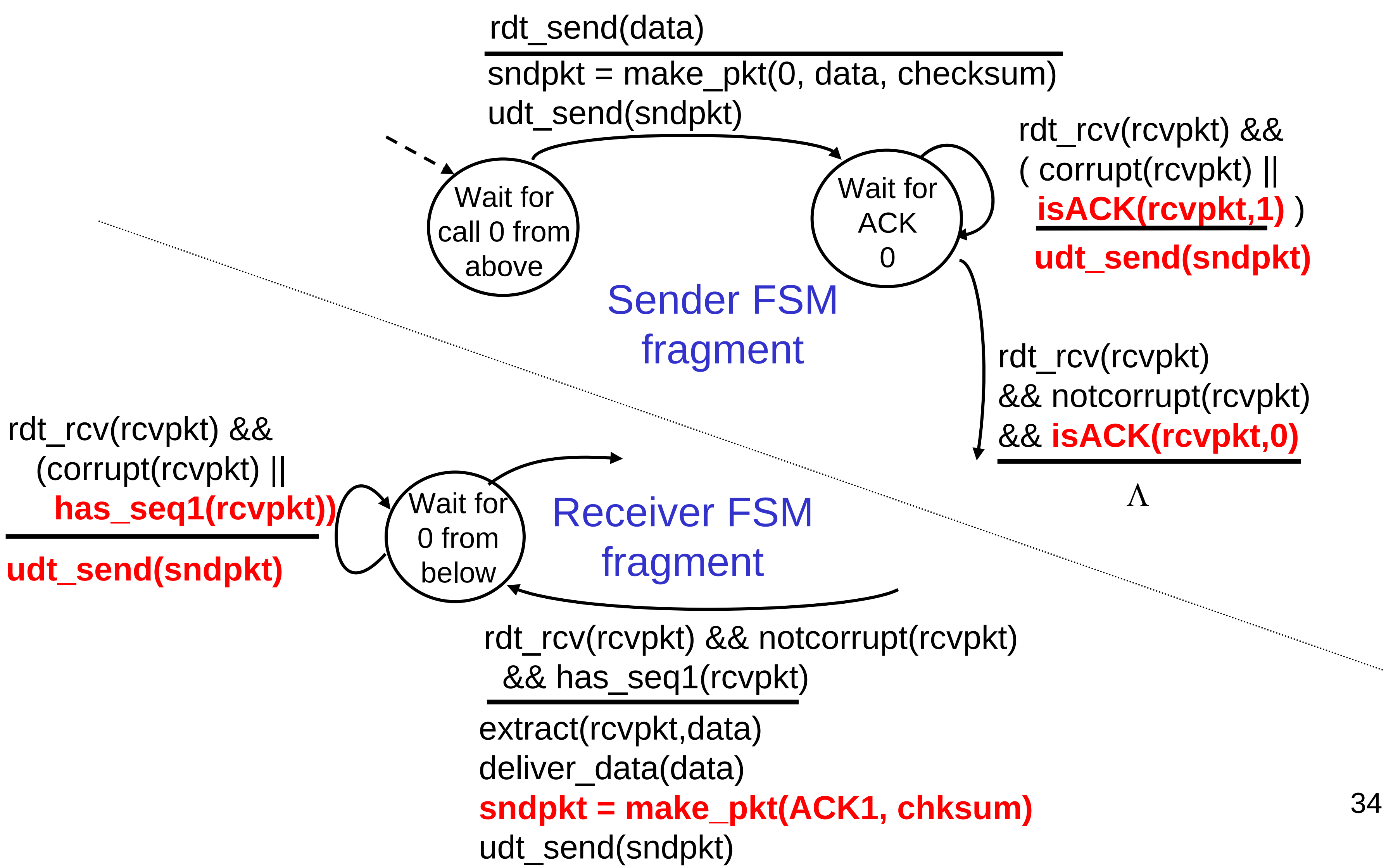
Receiver:

- Must check if received packet is duplicate
  - state indicates whether 0 or 1 is expected packet sequence #
- Note: receiver *cannot* know if sender received its last ACK/NAK OK

RDT2.2: A Protocol w/o NAK

- Same functionality as RDT2.1, using ACKs only
- Instead of NAK, receiver sends ACK for last packet received OK
  - receiver must *explicitly* include seq# of packet being ACKed
- duplicate ACK at sender results in same action as NAK: *retransmit current packet*

RDT2.2: Sender, Receiver (Fragments)



RDT3.0: Channels with Errors *and* Loss

New assumption:

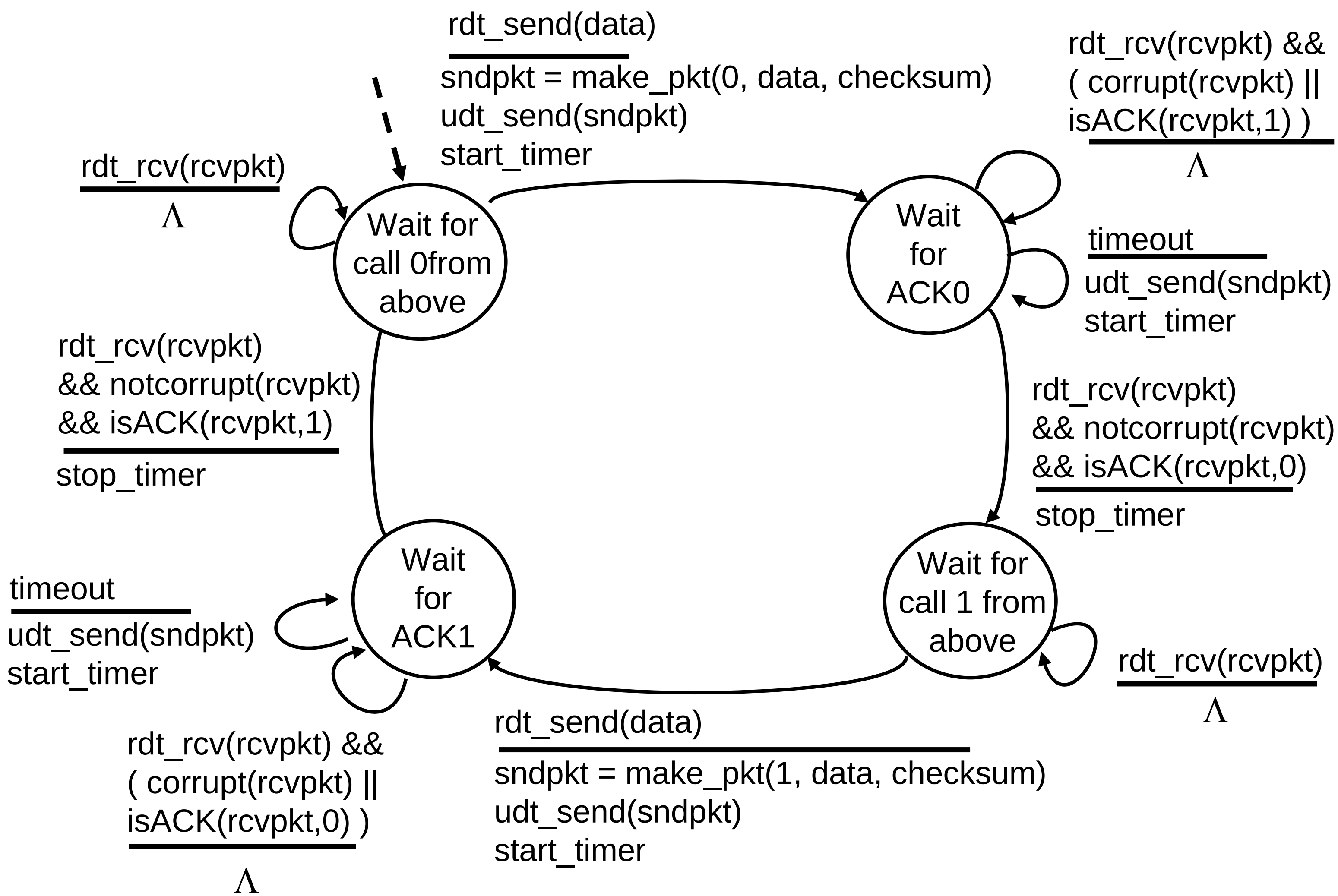
- Underlying channel can also lose packets (data or ACKs):
  - checksum
  - sequence #s
  - ACKs
  - retransmissions

  will be of help, but not enough

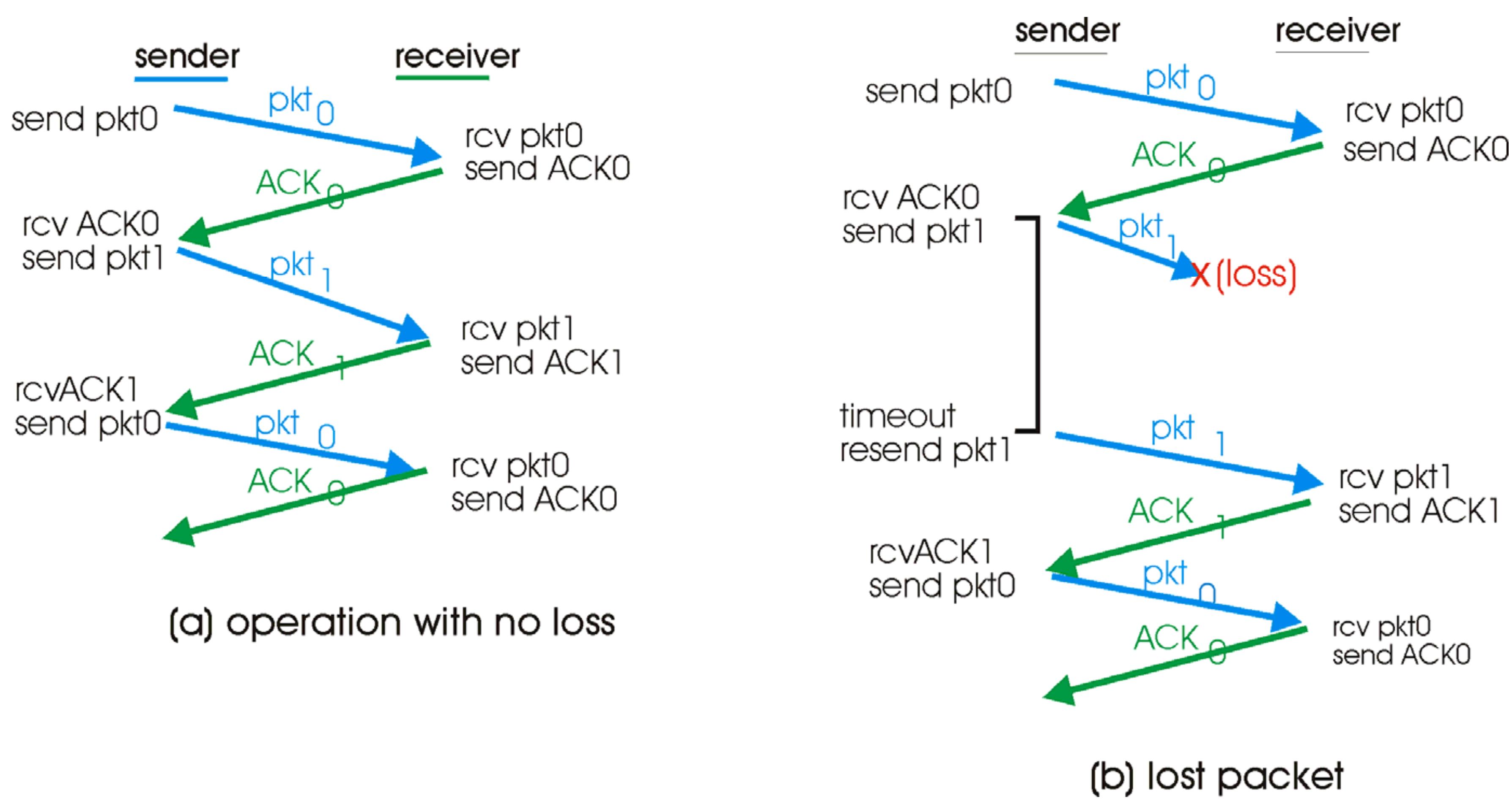
Approach:

- Sender waits “reasonable” amount of time for ACK
- Retransmits if no ACK received in this time
- If packet (or ACK) is just delayed (not lost):
  - retransmission will be duplicate, but use of seq #’s already handles this
  - receiver must specify seq # of packet being ACKed
- Requires countdown timer

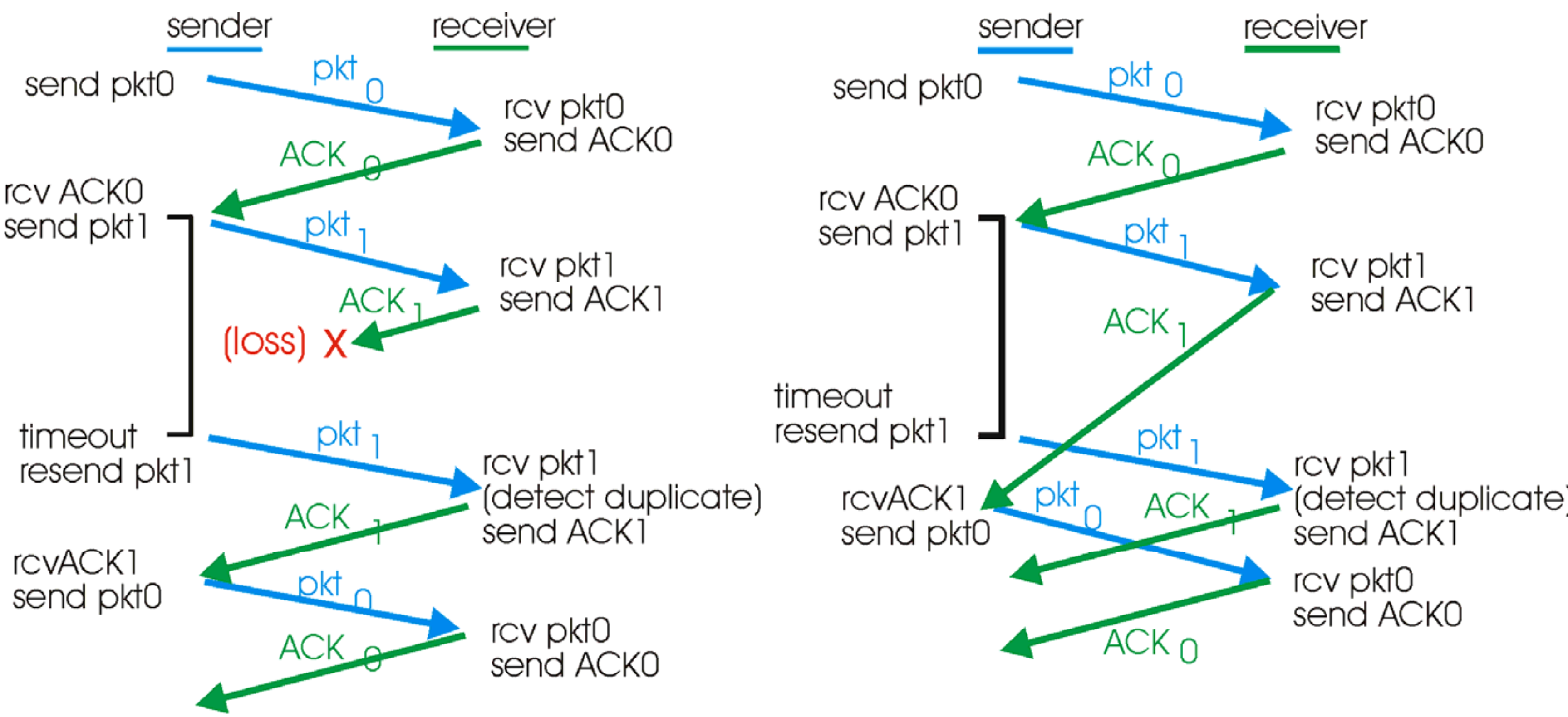
RDT3.0 Sender



RDT3.0 in Action



RDT3.0 in Action



(c) lost ACK

(d) premature timeout

Performance of RDT3.0

- RDT3.0 works, but performance is poor
- Example: 1 Gbps link, 15 ms propagation delay, 8000 bit packet:

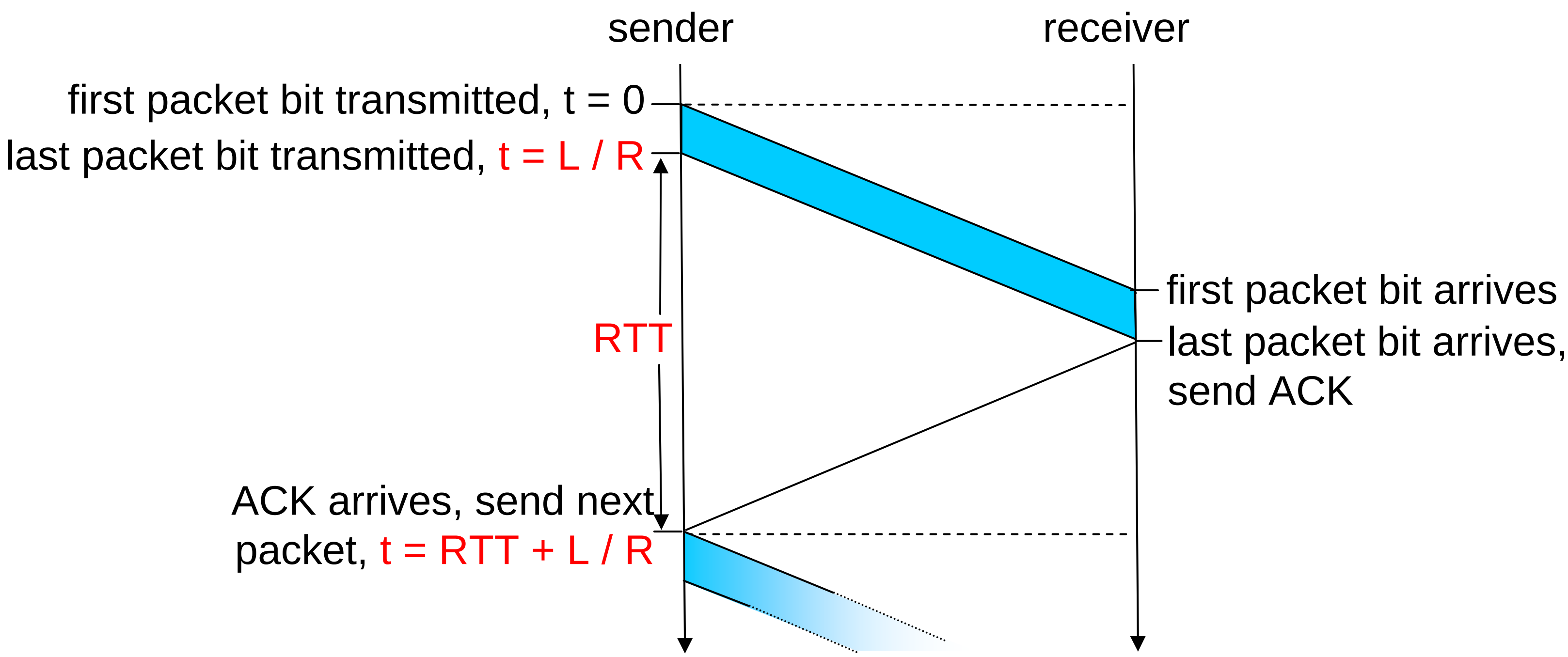
$$d_{trans} = \frac{L}{R} = \frac{8000\text{bits}}{10^9\text{bps}} = 8\,\text{microseconds}$$

U_{sender}: utilization – fraction of time sender is busy sending

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

- 1KB packet every 30 msec
  $\rightarrow$ 33KB/sec throughput over 1 Gbps link
- Network protocol limits use of physical resources!

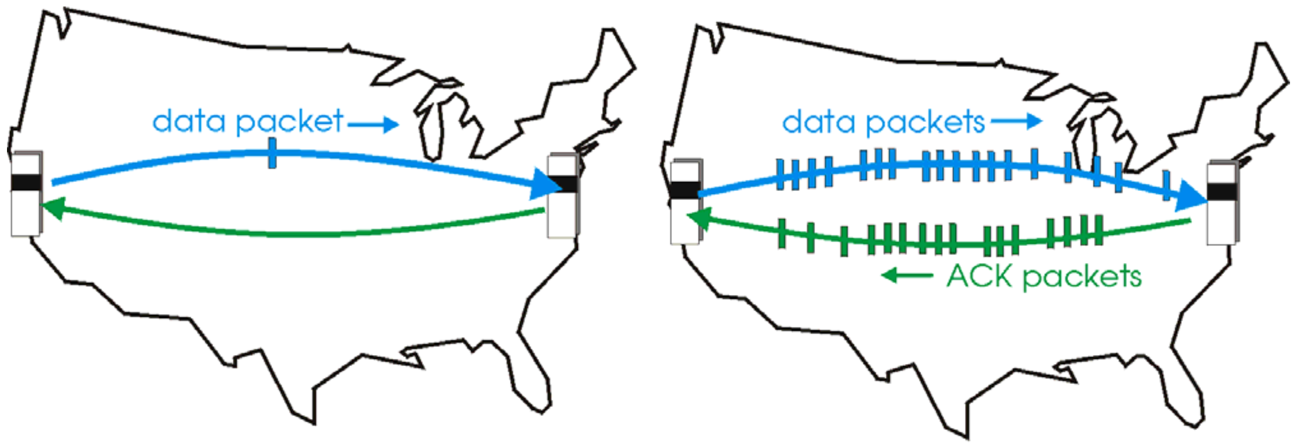
RDT3.0: Stop-and-wait Operation



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

Pipelined Protocols

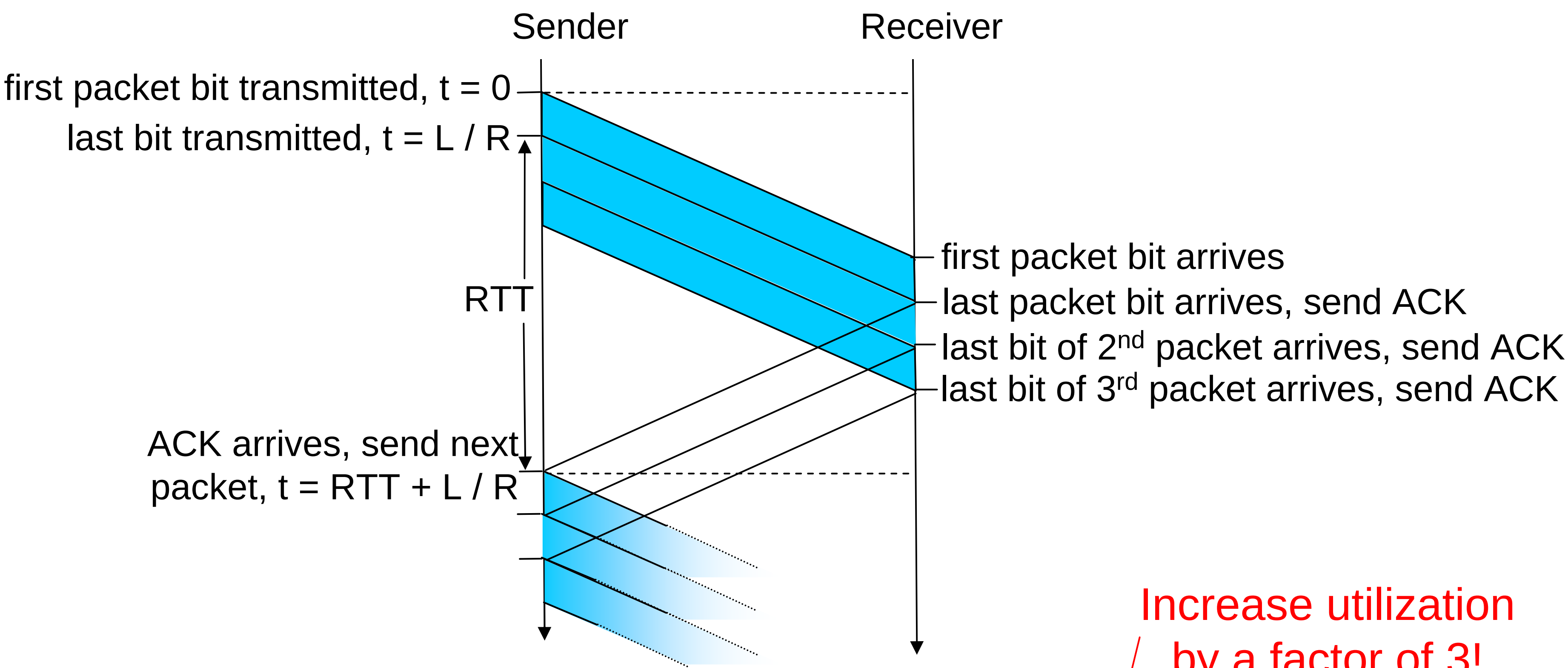
- Pipelining: sender allows multiple, “in-flight”, yet-to-be-acknowledged packets
  - range of sequence numbers must be increased
  - buffering at sender and/or receiver



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

- Two generic forms of pipelined protocols: *Go-Back-N* and *Selective Repeat*

Pipelining: Increased Utilization



Increase utilization by a factor of 3!

$$U_{sender} = \frac{3 * L / R}{RTT + L / R} = \frac{.024}{30.008} = 0.0008$$

Pipelining Protocols

Go-back-N: Overview

- *Sender:* up to N unACKed packets in pipeline
- *Receiver:* only sends cumulative ACKs
  - does not ACK packet if there is a gap
- *Sender:* has timer for oldest unACKed packet
  - if timer expires: retransmit all unACKed packets

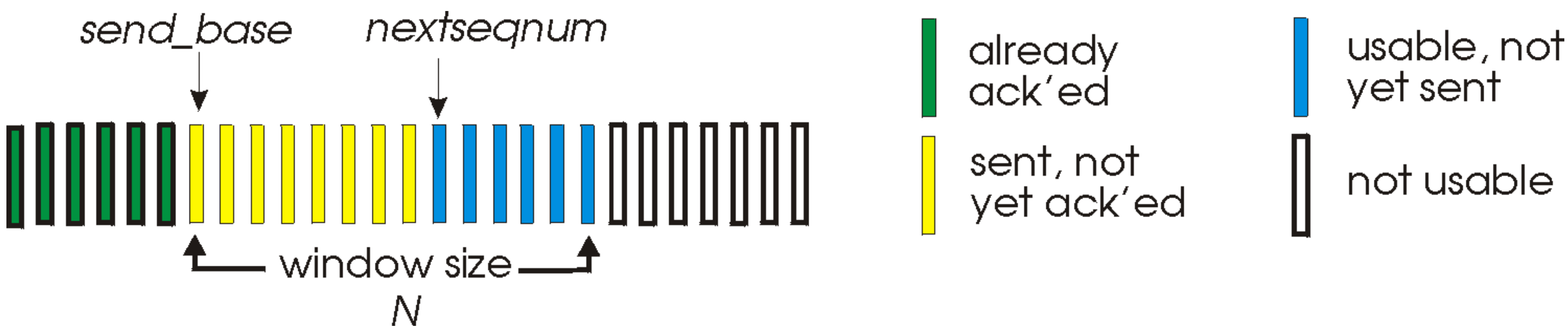
Selective Repeat: Overview

- *Sender:* up to N unACKed packets in pipeline
- *Receiver:* ACKs individual pkts
- *Sender:* maintains timer for each unACKed packet
  - if timer expires: retransmit only unACKed packet

Go-Back-N

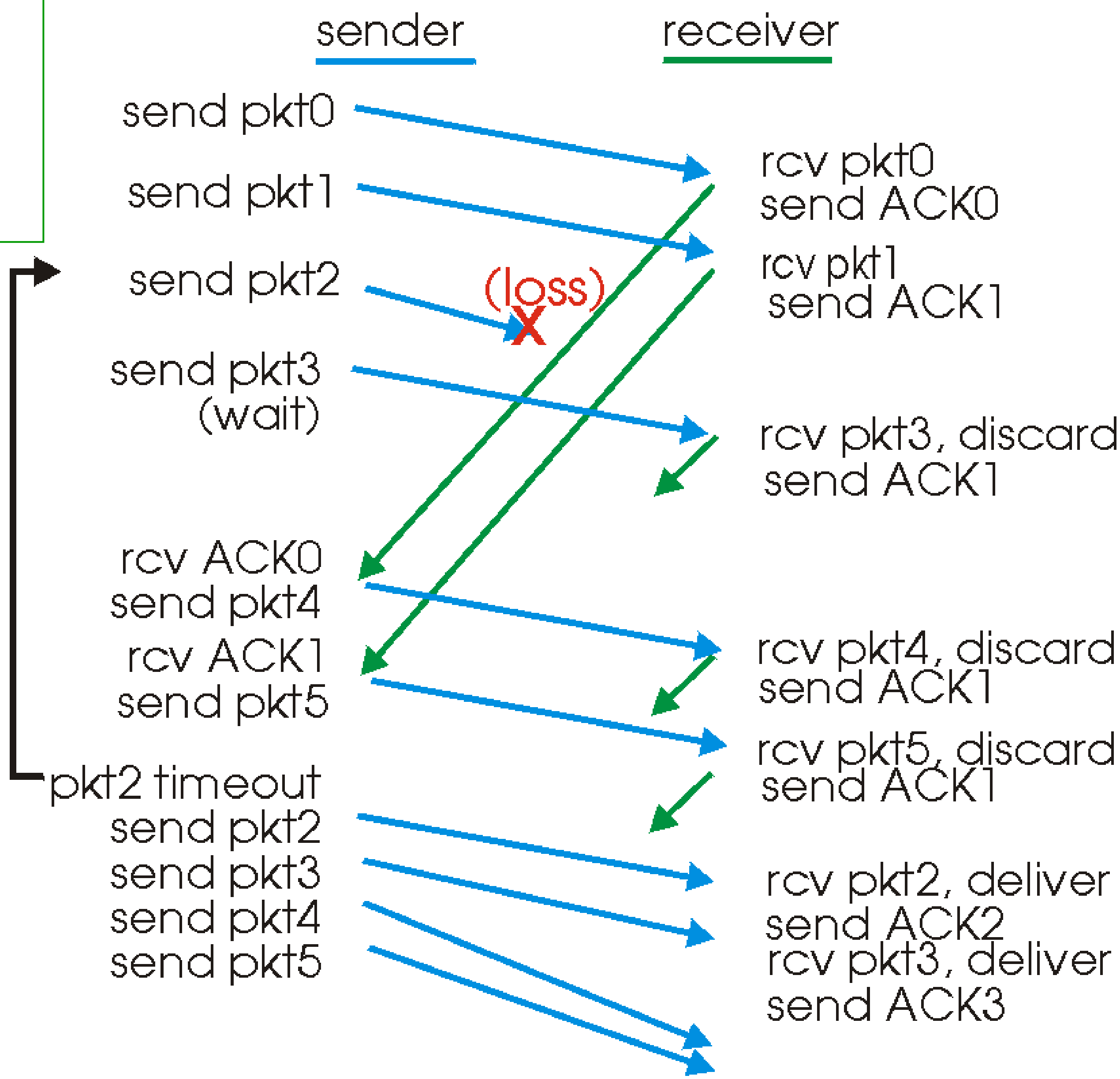
Sender:

- k-bit sequence # in packet header
- “window” of up to N, consecutive unACKed packets allowed



- ACK(n): ACKs all packets up to, including seq # n — “cumulative ACK”
  – may receive duplicate ACKs (see receiver)
- Timer for each in-flight pkt
- *Timeout(n):* retransmit packet n and all higher seq # packets in window

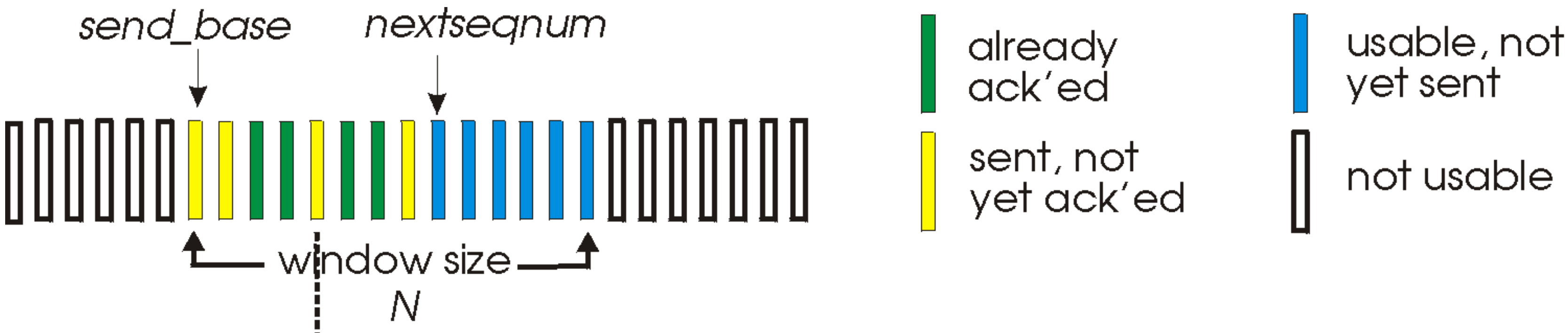
Go-Back-N in Action



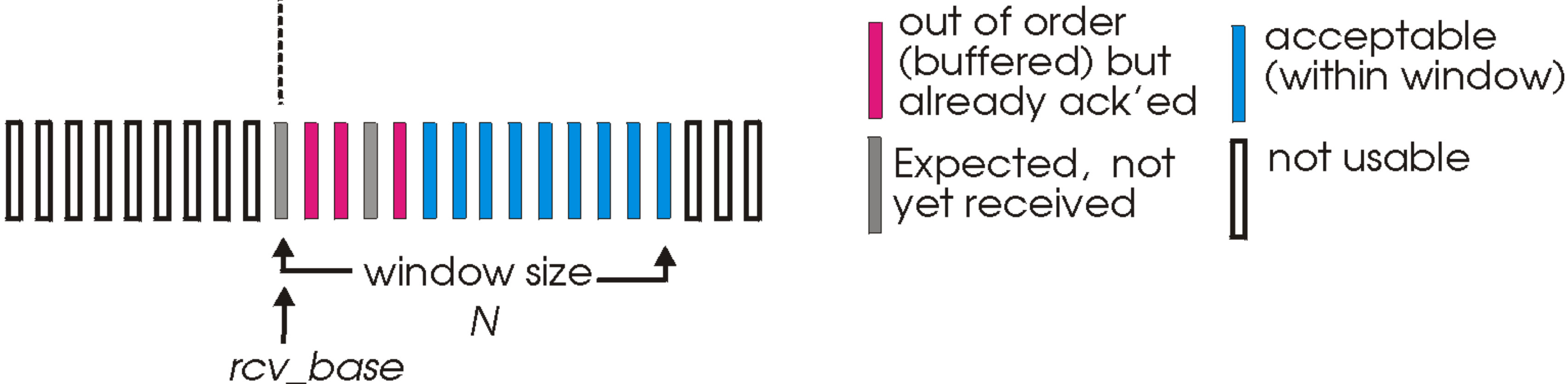
Selective Repeat

- Receiver *individually* acknowledges all correctly received pkts
  - buffers pkts, as needed, for eventual in-order delivery to upper layer
- Sender only resends pkts for which ACK not received
  - sender timer for each unACKed pkt
- Sender window
  - N consecutive seq #’s
  - again limits seq #s of sent, unACKed pkts

Selective Repeat: Sender, Receiver Windows



(a) sender view of sequence numbers



(b) receiver view of sequence numbers

Selective Repeat

Sender

Data from application (above):

- if next available seq # in window, send pkt

Timeout(n):

- resend pkt n, restart timer

ACK(n) in [sendbase,sendbase+N]:

- mark pkt n as received
- if n smallest unACKed pkt, advance window base to next unACKed seq #

Receiver

Pkt n in [rcvbase, rcvbase+N-1]

- send ACK(n)
- out-of-order: buffer
- in-order: deliver (also deliver buffered, in-order pkts), advance window to next not-yet-received pkt

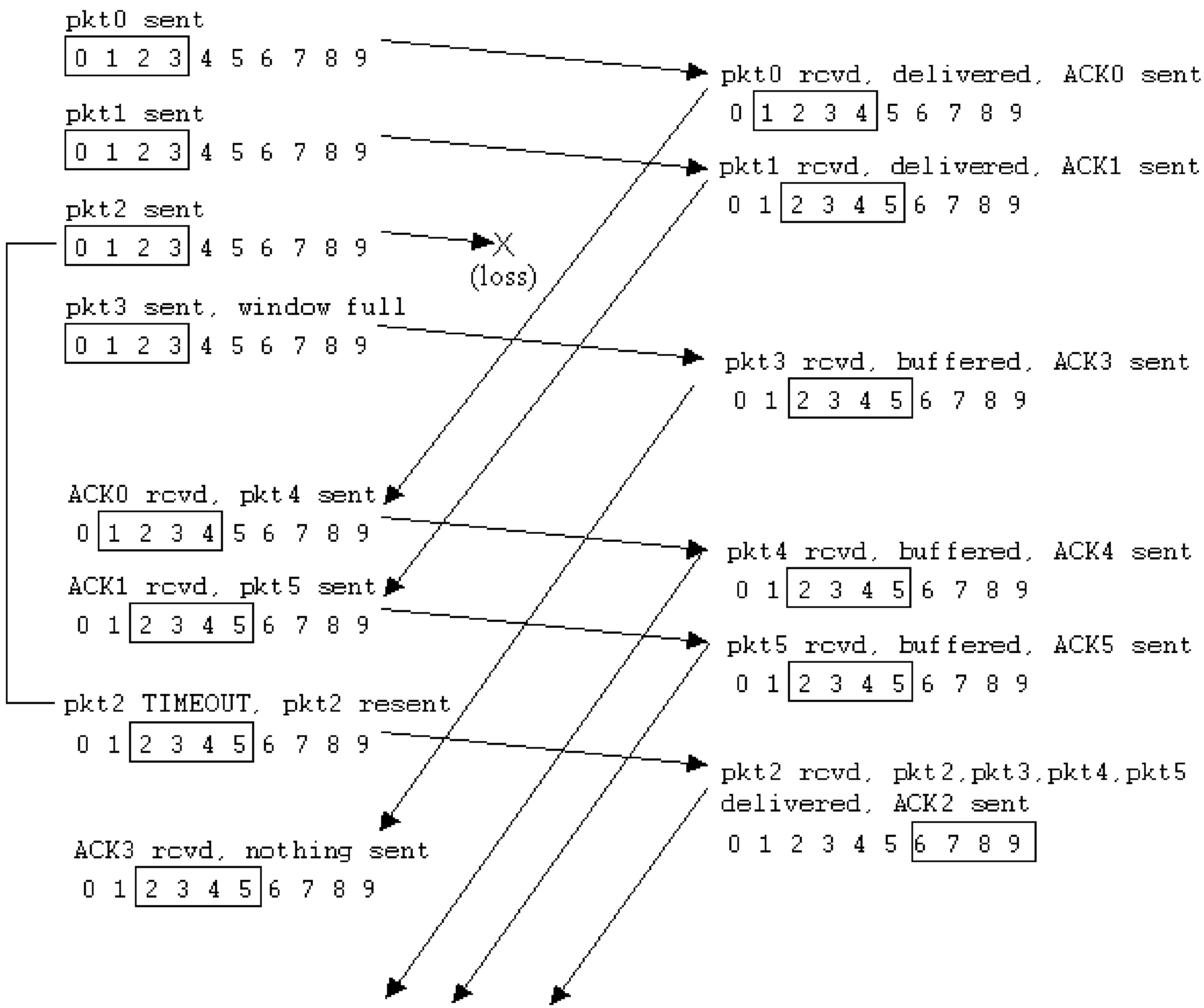
Pkt n in [rcvbase-N,rcvbase-1]

- ACK(n)

Otherwise:

- ignore

Selective Repeat in Action



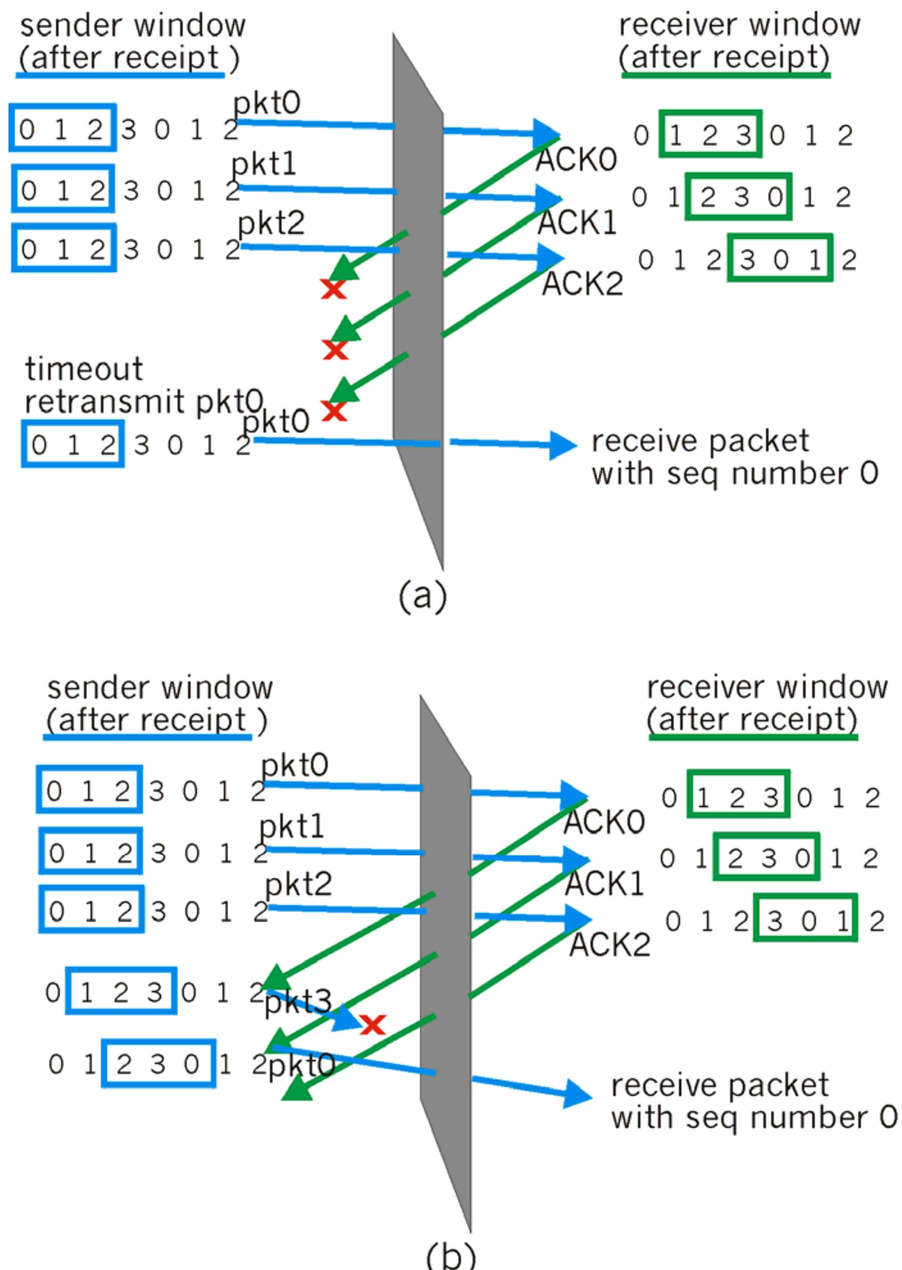
Selective Repeat: Dilemma

Example:

- seq #'s: 0, 1, 2, 3
- window size=3

- Receiver sees no difference in two scenarios!
- Incorrectly passes duplicate data as new in (a)

Q: what relationship should hold between seq # size and window size?



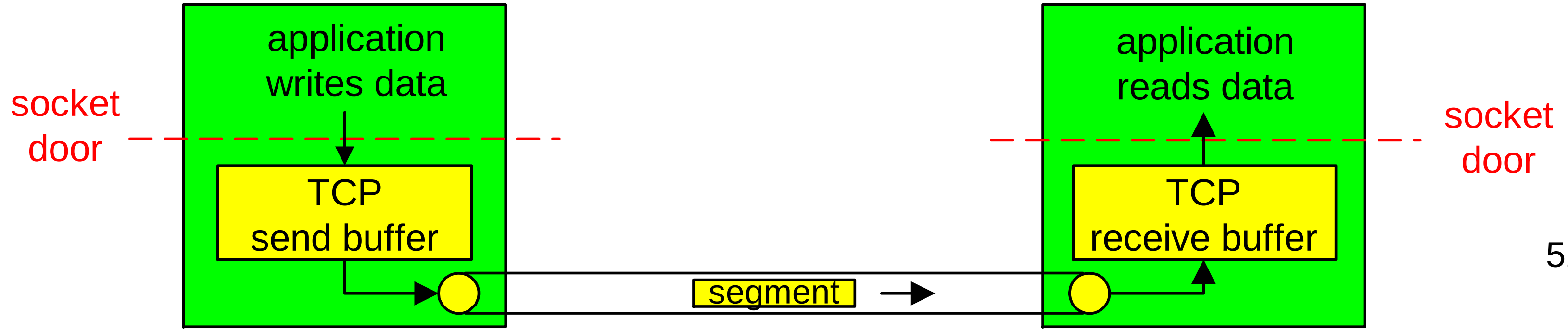
5. Transport Protocols

5.5 Connection-oriented Transport: TCP

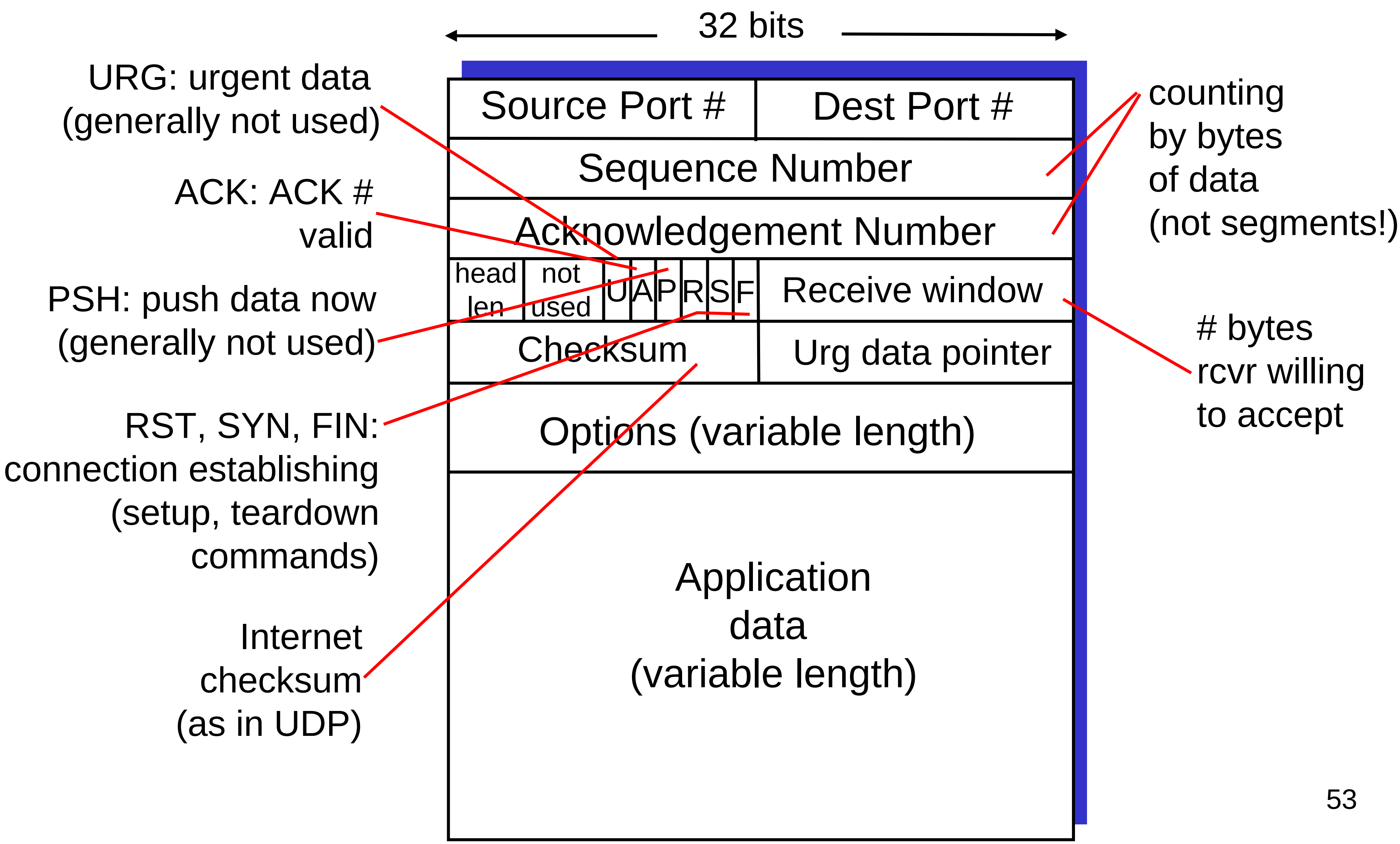
TCP: Overview

RFCs: 793, 1122, 1323, 2018, 2581

- Point-to-point:
  - one sender, one receiver
- Reliable, in-order *byte steam:*
  - no “message boundaries”
- Pipelined:
  - TCP congestion and flow control set window size
- *Send & receive buffers*
- Flow controlled:
  - sender will not overwhelm receiver
- Full duplex data:
  - bi-directional data flow in same connection
  - MSS: maximum segment size
- Connection-oriented:
  - handshaking (exchange of control msgs) initialises sender, receiver state before data exchange

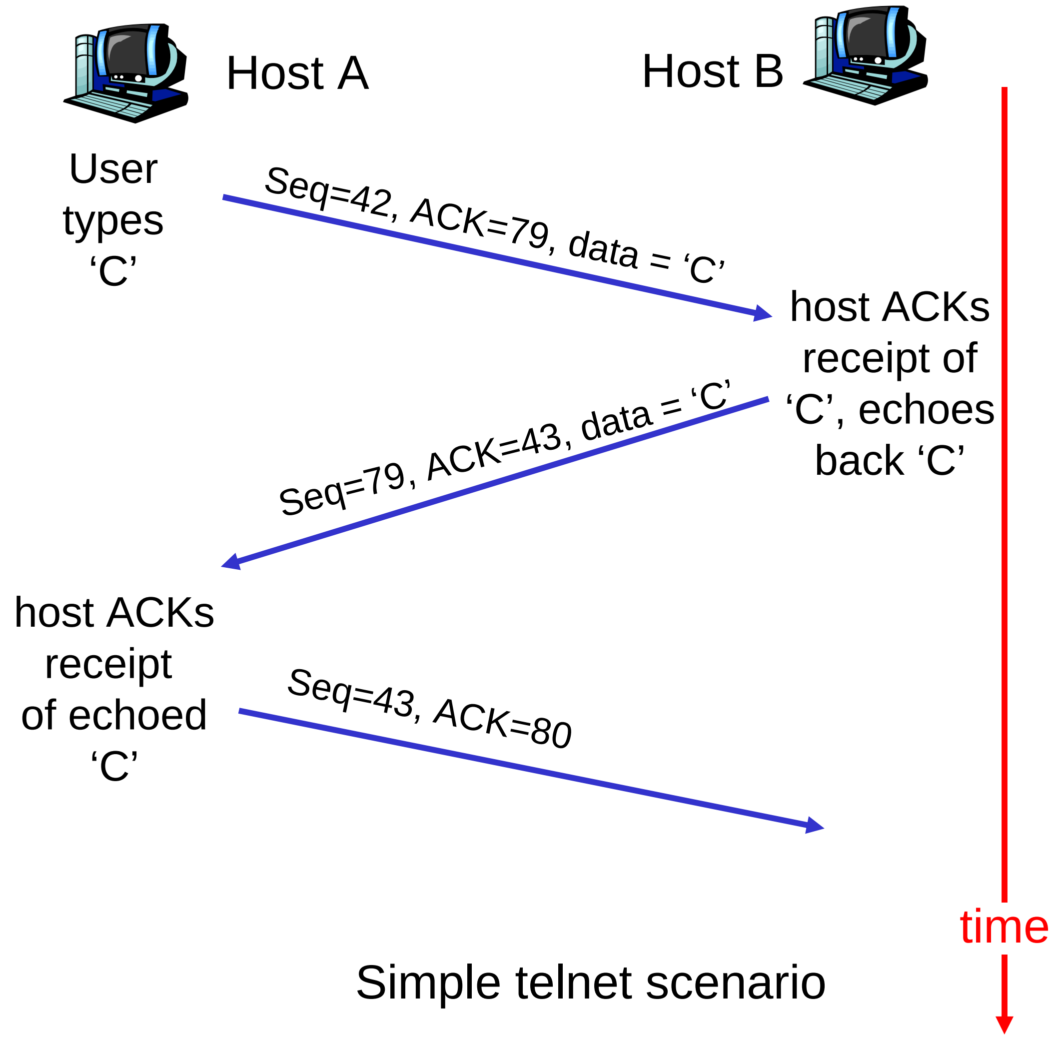


TCP Segment Structure



TCP Sequence #'s and ACKs

- Seq. #'s:
  - byte stream "number" of first byte in segment's data
- ACKs:
  - seq # of next byte expected from other side
  - cumulative ACK
- Q: how receiver handles out-of-order segments
  - A: TCP spec doesn't say, - up to implementer



Simple telnet scenario

TCP Round Trip Time and Timeout

Q: How to set TCP timeout value?

- longer than RTT
  - but RTT varies
- too short: premature timeout
  - unnecessary retransmissions
- too long: slow reaction to segment loss

Q: How to estimate RTT?

- **`SampleRTT`**: measured time from segment transmission until ACK receipt
  - ignore retransmissions
- **`SampleRTT`** will vary, want estimated RTT "smoother"
  - average several recent measurements, not just current **`SampleRTT`**

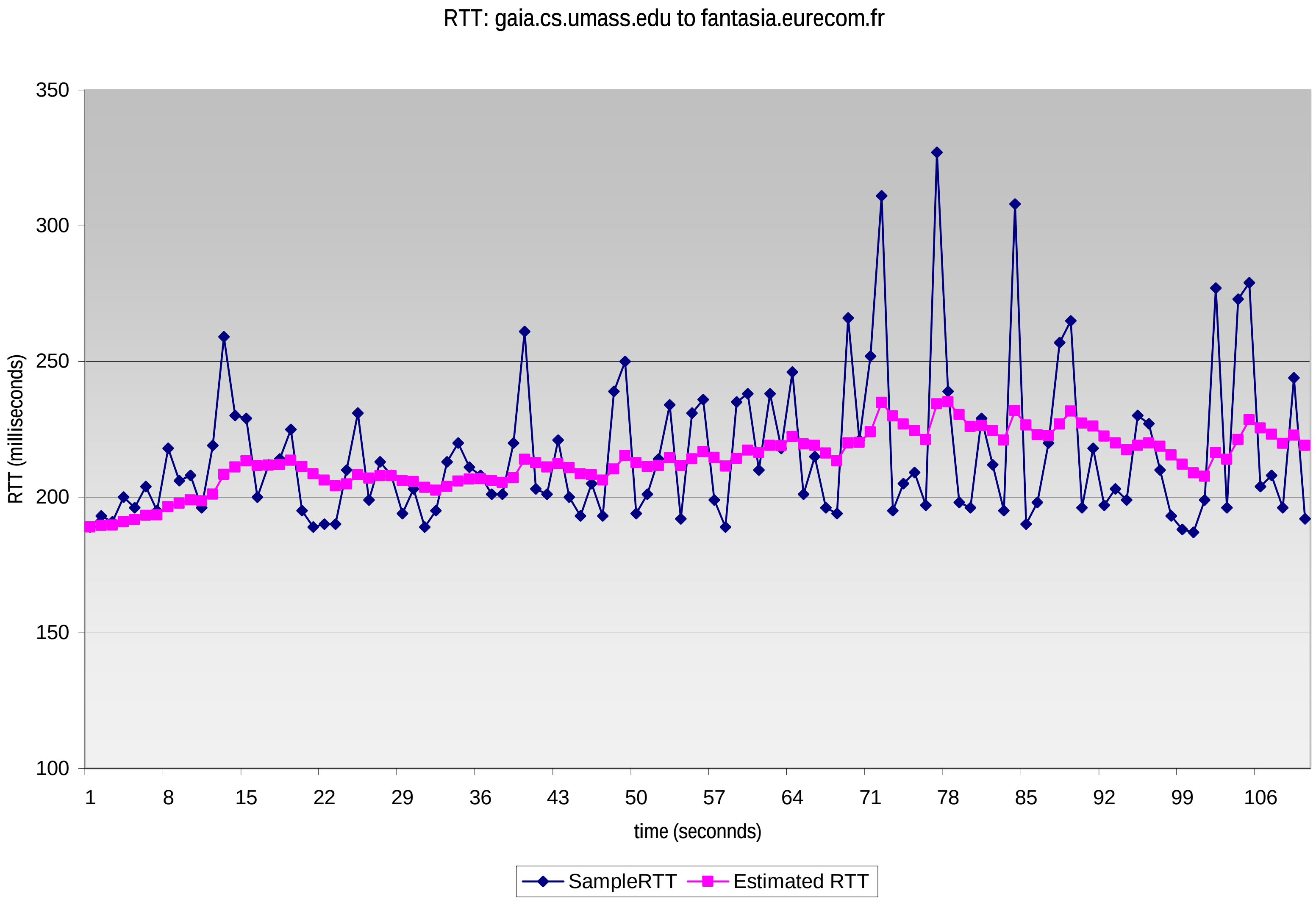
TCP Round Trip Time and Timeout

$$\texttt{EstimatedRTT} = (1-\alpha) * \texttt{EstimatedRTT} + \alpha * \texttt{SampleRTT}$$

Exponential weighted moving average

- influence of past sample decreases exponentially fast
- typical value: $\alpha = 0.125$

Example RTT Estimation



TCP Round Trip Time and Timeout

Setting the timeout

- **`EstimatedRTT`** plus "safety margin"
  - large variation in **`EstimatedRTT`** $\rightarrow$ larger safety margin
- First, estimate of how much SampleRTT deviates from **`EstimatedRTT`**:

$$\texttt{DevRTT} = (1-\beta)*\ \texttt{DevRTT} + \beta * |\texttt{SampleRTT - EstimatedRTT}|$$

(typically, $\beta = 0.25$)

Then set timeout interval:

$$\texttt{TimeoutInterval} = \texttt{EstimatedRTT} + 4*\texttt{DevRTT}$$

TCP Reliable Data Transfer

- TCP creates RDT service on top of IP's unreliable service
- TCP features
  - pipelined segments
  - cumulative ACKs
- TCP uses *by default* single retransmission timer
- Retransmissions are triggered by:
  - timeout events
  - duplicate ACKs

- Consider simplified TCP sender:
  - ignore congestion control

TCP Sequence Numbers

Sequence number of a segment:

Byte stream number of first byte in segment

Example: A sends to B over TCP

- 500k image with MSS = 1k,
  initial sequence number = 0

➔ 500 segments,
with sequence numbers 0, 1024, 2048, ...

TCP Acknowledgement Numbers

Acknowledgement number in segment sent from B to A:

Sequence number of next byte B is expecting from A

Example:

- B has received segments 1, 2, and 4, but not 3.
- Acknowledgement number is 2048
  (= 1st byte of segment 3)

Example shows:

- Acknowledgement is cumulative
  (acknowledges all bytes up to Ack - 1)
- No mention of out-of-order segments

TCP Sender Events:

Data received from application:

- create segment with seq #
- seq # is byte-stream number of first data byte in segment
- start timer if not already running (think of timer as for oldest unACKed segment)
- expiration interval: `TimeOutInterval`

Timeout:

- retransmit segment that caused timeout
- restart timer

ACK received:

- if acknowledges previously unACKed segments
  - update what is known to be ACKed
  - start timer if there are outstanding segments

TCP Sender Actions

Client variables

```
ackSNo  = initialSequenceNumber  // ack'ed sequence #
nextSNo = initialSequenceNumber // next sequence #
```

Loop through the following cases:

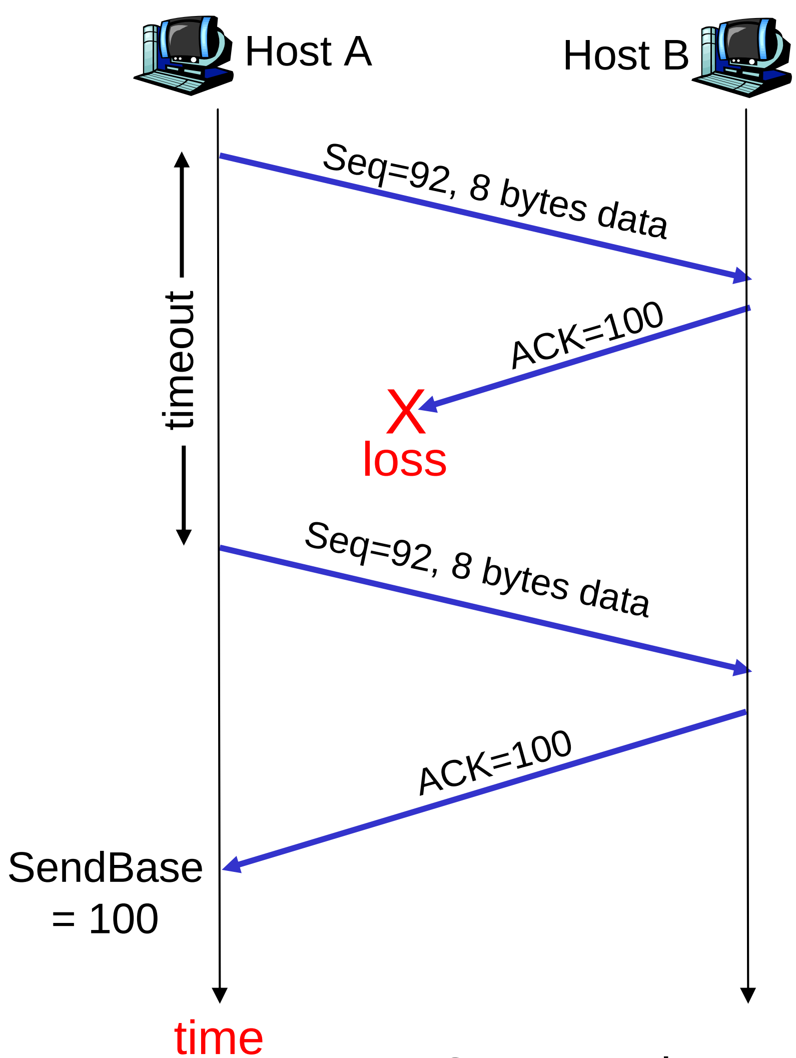
```
if (data received from application){
   create segment with sequence number nextSNo;
   start timer for segment nextSNo;
   pass segment to IP;
   nextSNo = nextSNo + data.length}

if (timeout for segment with sNo y){
   retransmit segment y;
   restart timer for segment y}
```

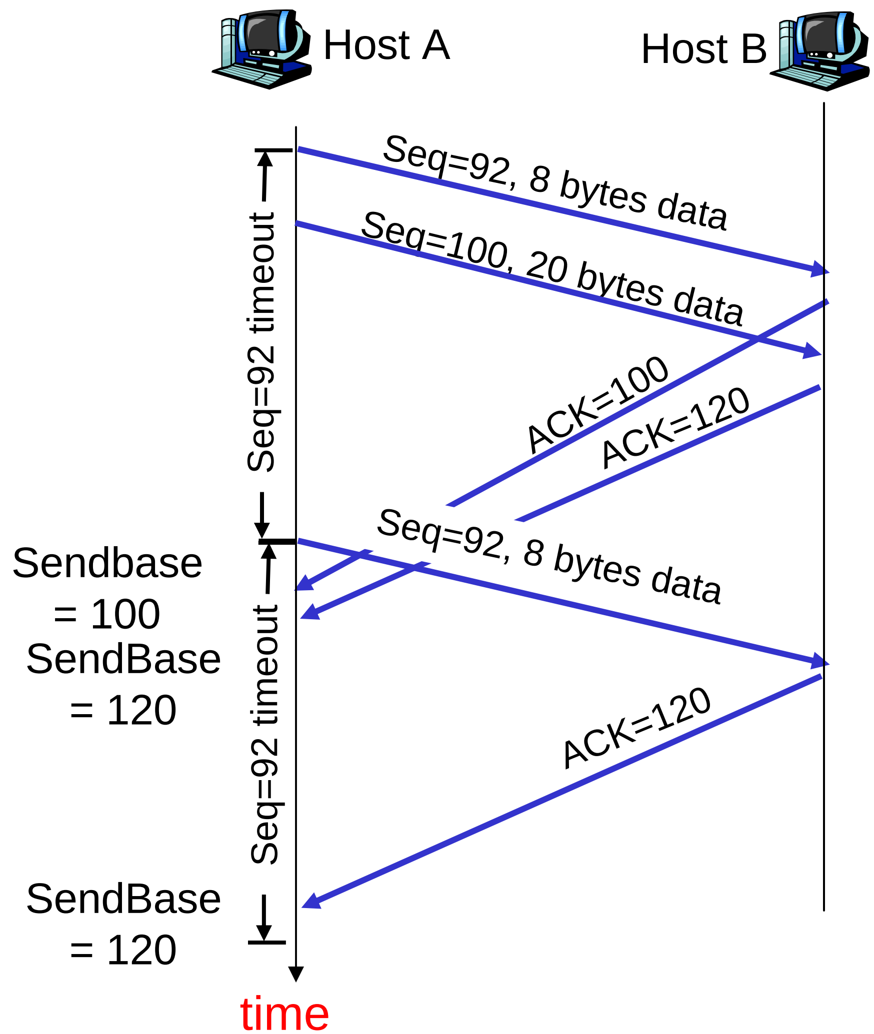
TCP Sender Actions (cntd)

```
if (ACK received with AckNo = y)
    if (y > ackSNo){ // cumulative ack
        cancel timers for segments with lower SNos;
        ackSNo = y}
    else { // duplicate ack
        increment counter for duplicate acks for y;
        if (number of duplicate acks for y == 3) {
            retransmit segment y;
            restart timer for segment y
            }
          }
```

TCP: Retransmission Scenarios

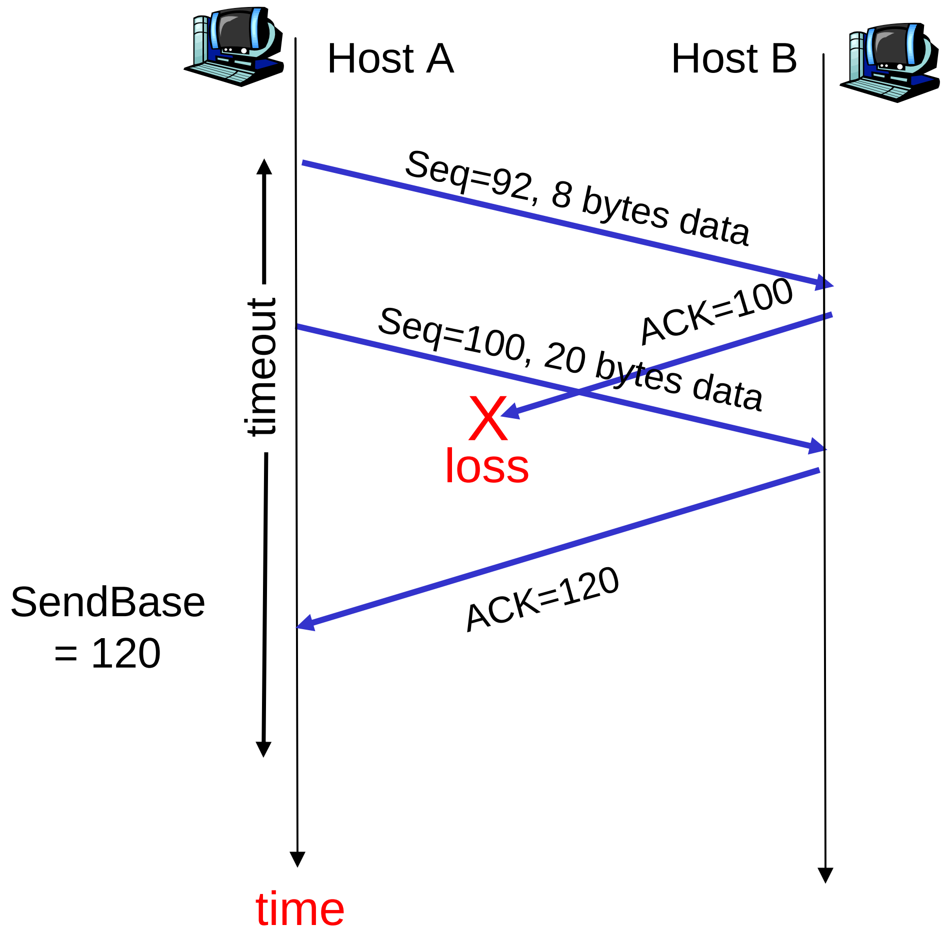


Lost ACK scenario



Premature timeout

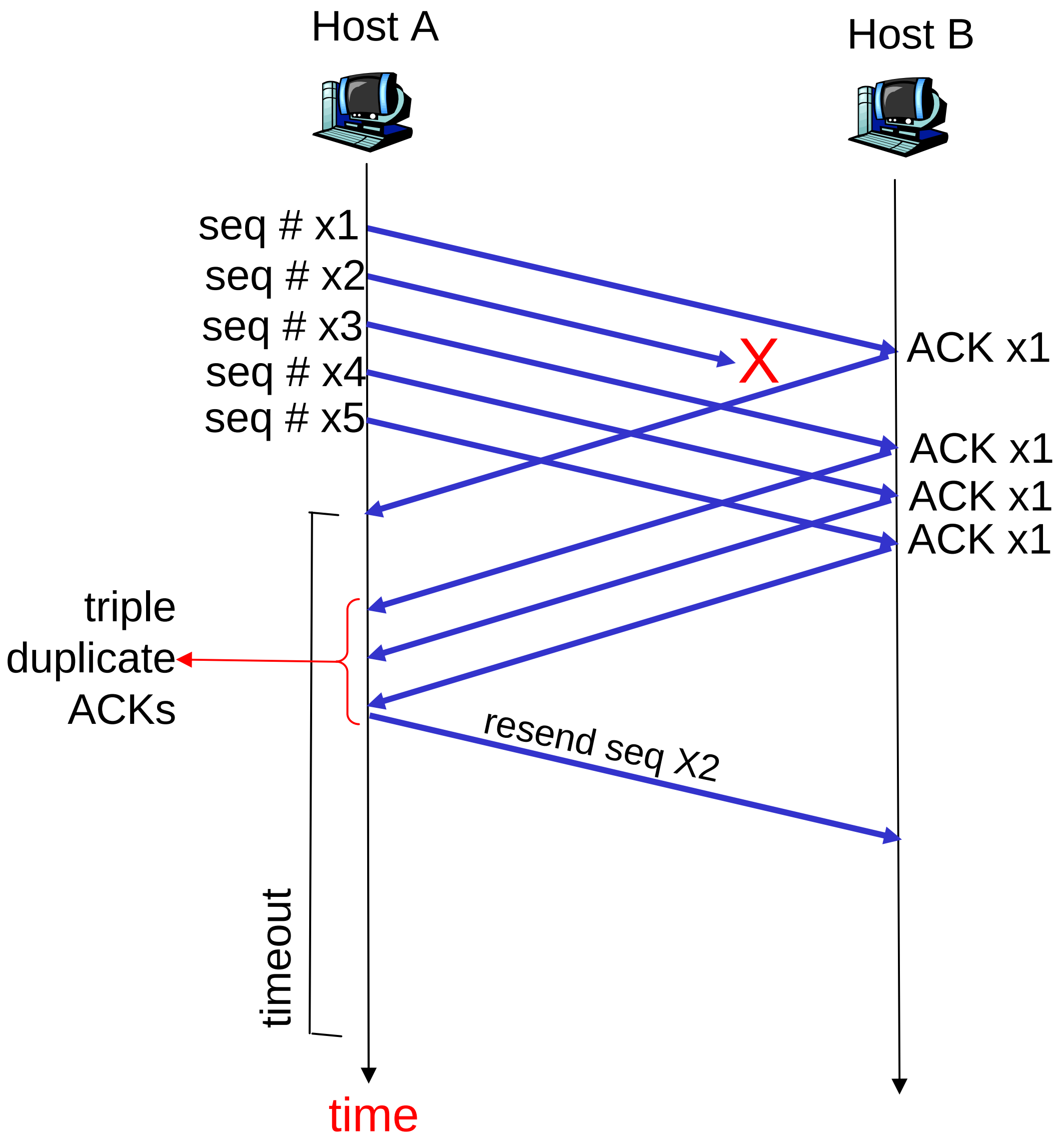
TCP retransmission scenarios (cntd)



Cumulative ACK scenario

Fast Retransmit

- Time-out period often relatively long:
  - long delay before resending lost packet
- Detect lost segments via duplicate ACKs
  - sender often sends many segments back-to-back
  - if segment is lost, there will likely be many duplicate ACKs for that segment
- If sender receives 3 ACKs for same data, it assumes that the segment after ACKed data was lost:
  - fast retransmit: resend segment before timer expires



Host A
Host B
seq # x1
seq # x2
seq # x3
seq # x4
seq # x5
X
ACK x1
ACK x1
ACK x1
ACK x1
triple
duplicate
ACKs
resend seq X2
timeout
time

TCP Receiver Actions

Event	Action
Segment arrives with expected SNo, all previous data already ack'ed	Wait up to 500 ms for arrival of another segment. Then send ack
Segment arrives with expected SNo, preceding segment received, but not ack'ed	Send cumulative ack
Out-of-order segment arrives with higher SNo than expected	Send duplicate ack, indicating SNo of next expected byte
Out-of-order segment arrives with lower SNo than expected	Send duplicate ack, indicating SNo of next expected byte

Flow Control

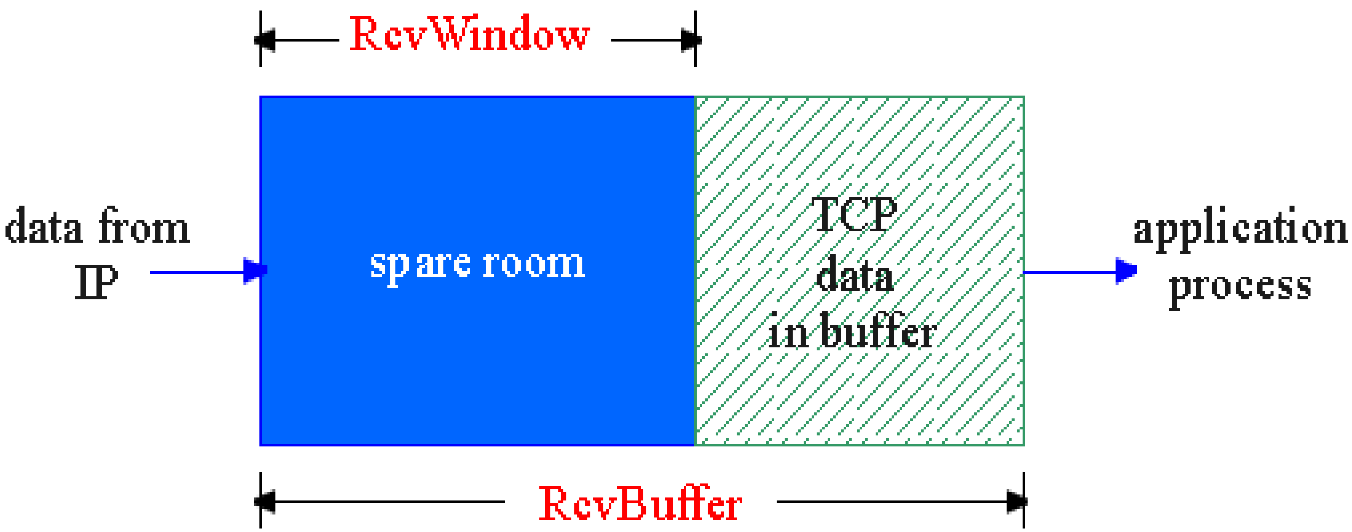
Receiver's buffer has size **`RcvBuffer`**

Receiver maintains variables

- **`LastByteRead`**
- **`LastByteReceived`**

Constraint:

`LastByteReceived - LastByteRead <= RcvBuffer`



Flow Control (cntd)

Receiver communicates to sender

```
RcvWindow =
        RcvBuffer - (LastByteReceived - LastByteRead)
```

Sender maintains variables

```
LastByteSent
LastByteAcked
```

Sender makes sure

```
LastByteSent - LastByteAcked <= RcvWindow
```

TCP Connection Management

Recall: TCP sender, receiver establish "connection" before exchanging data segments

- initialize TCP variables:
  - sequence #s
  - buffers, flow control info (e.g. `RcvWindow`)
- *Client:* connection initiator

```
Socket clientSocket =
new Socket("hostname",
       "port number");
```

- *Server:* contacted by client

```
Socket connectionSocket
= serverSocket.accept();
```

Three Way Handshake

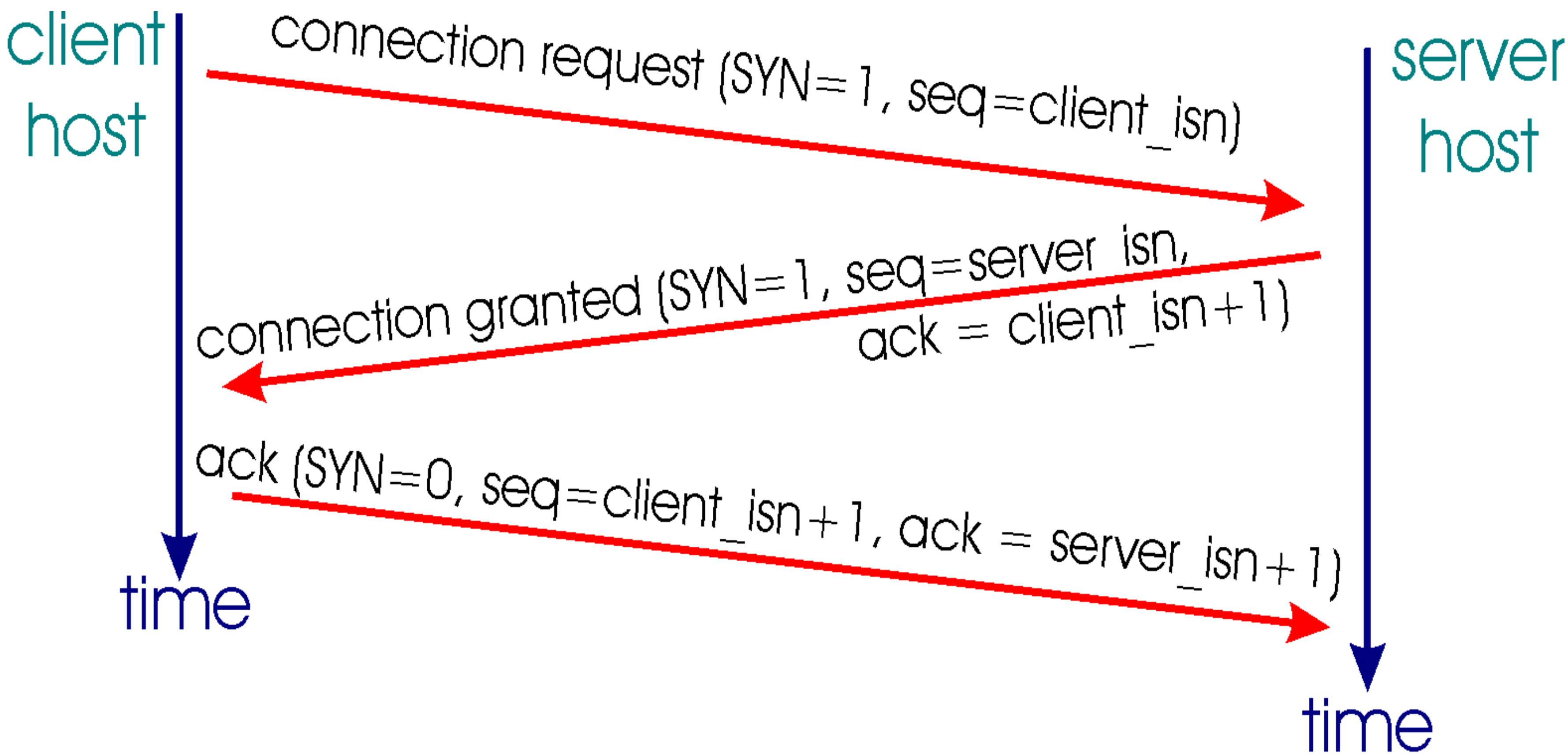
Step 1: client host sends TCP SYN segment to server
- specifies initial sequence #
- no data

Step 2: server host receives SYN, replies with SYNACK segment
- server allocates buffers
- specifies server initial sequence #

Step 3: client receives SYNACK, replies with ACK segment, which may contain data

Establishing a TCP Connection

"Three way handshake"



Why are sequence numbers exchanged?

Why does the sender acknowledge?

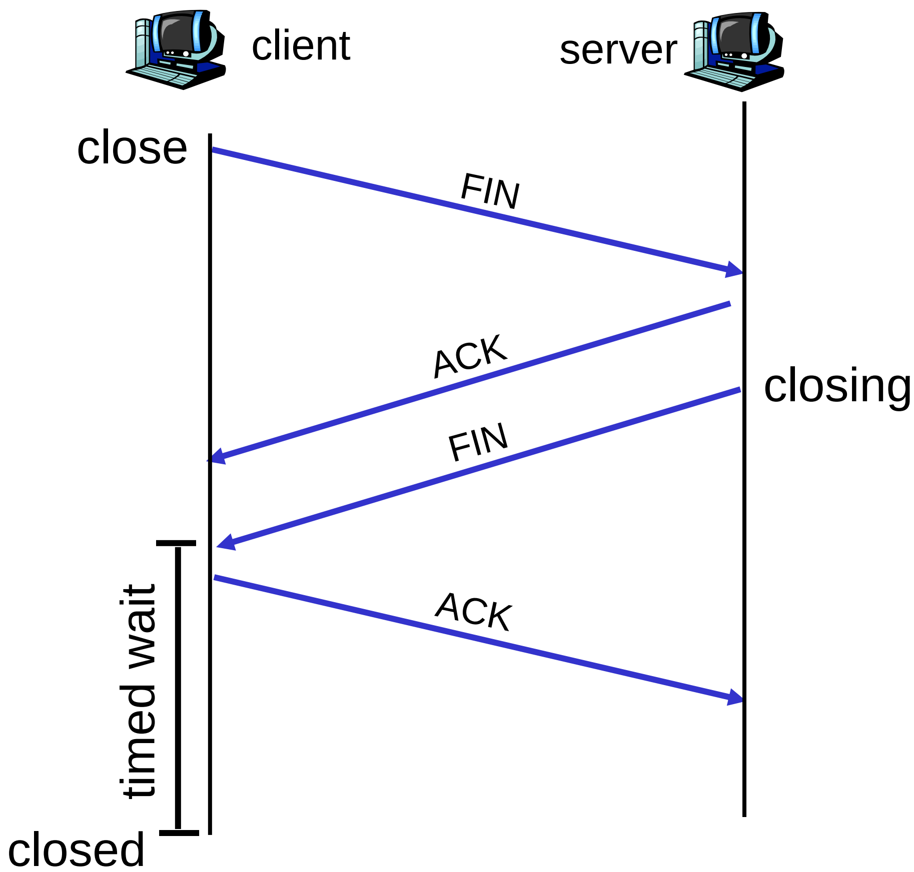
TCP Connection Management (cntd)

Closing a connection:

client closes socket:
`clientSocket.close();`

Step 1: client end system sends TCP FIN control segment to server

Step 2: server receives FIN, replies with ACK. Closes connection, sends FIN.

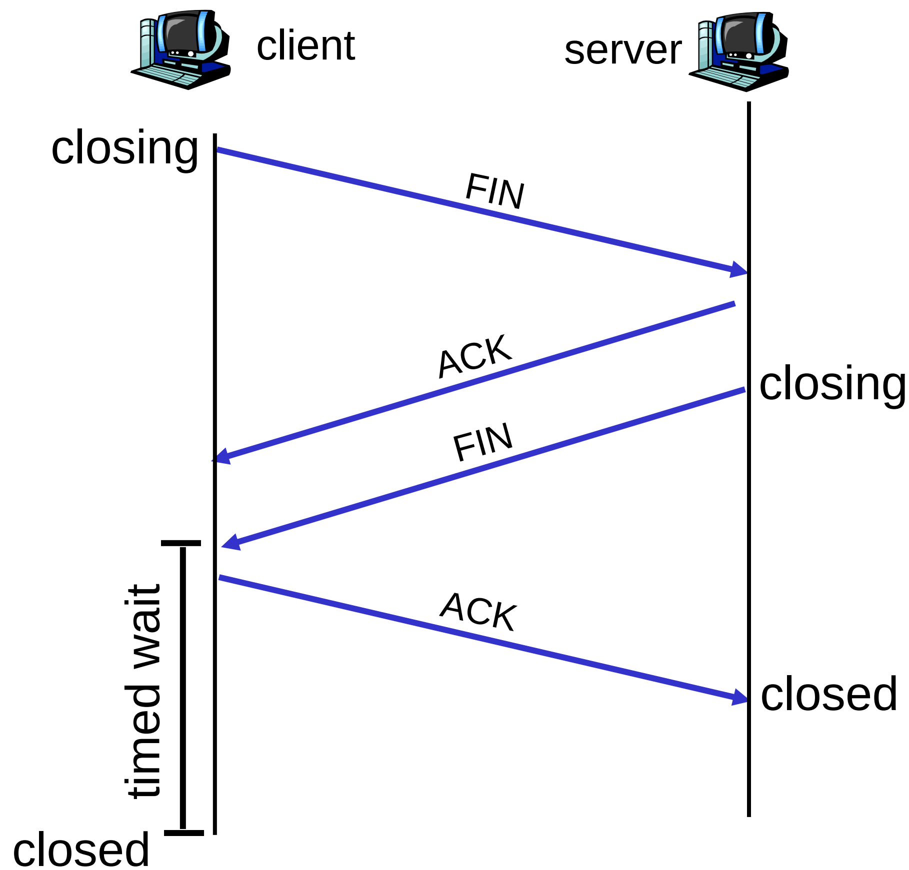


TCP Connection Management (cntd)

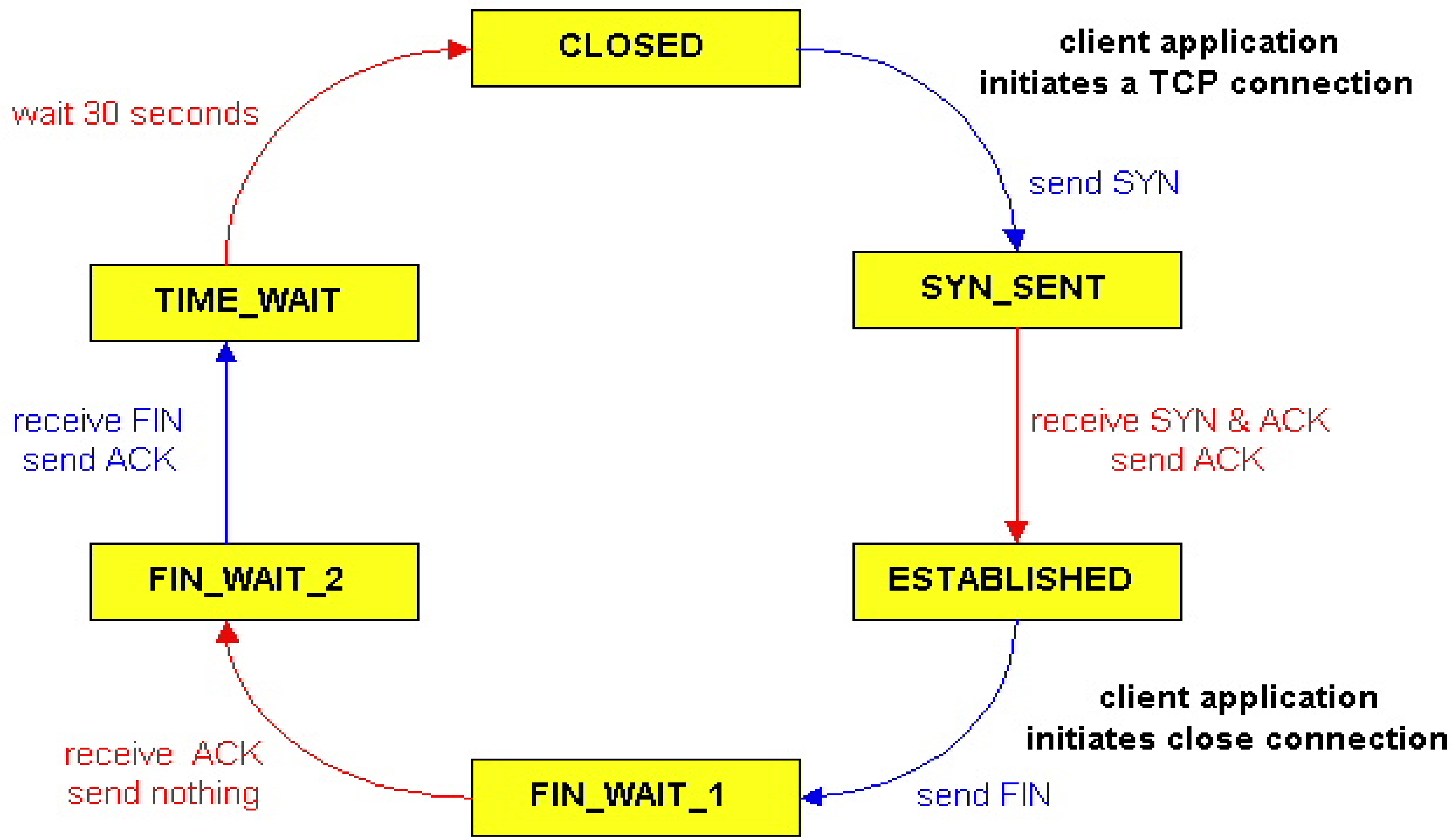
Step 3: client receives FIN, replies with ACK

- Enters "timed wait" - will respond with ACK to received FINs

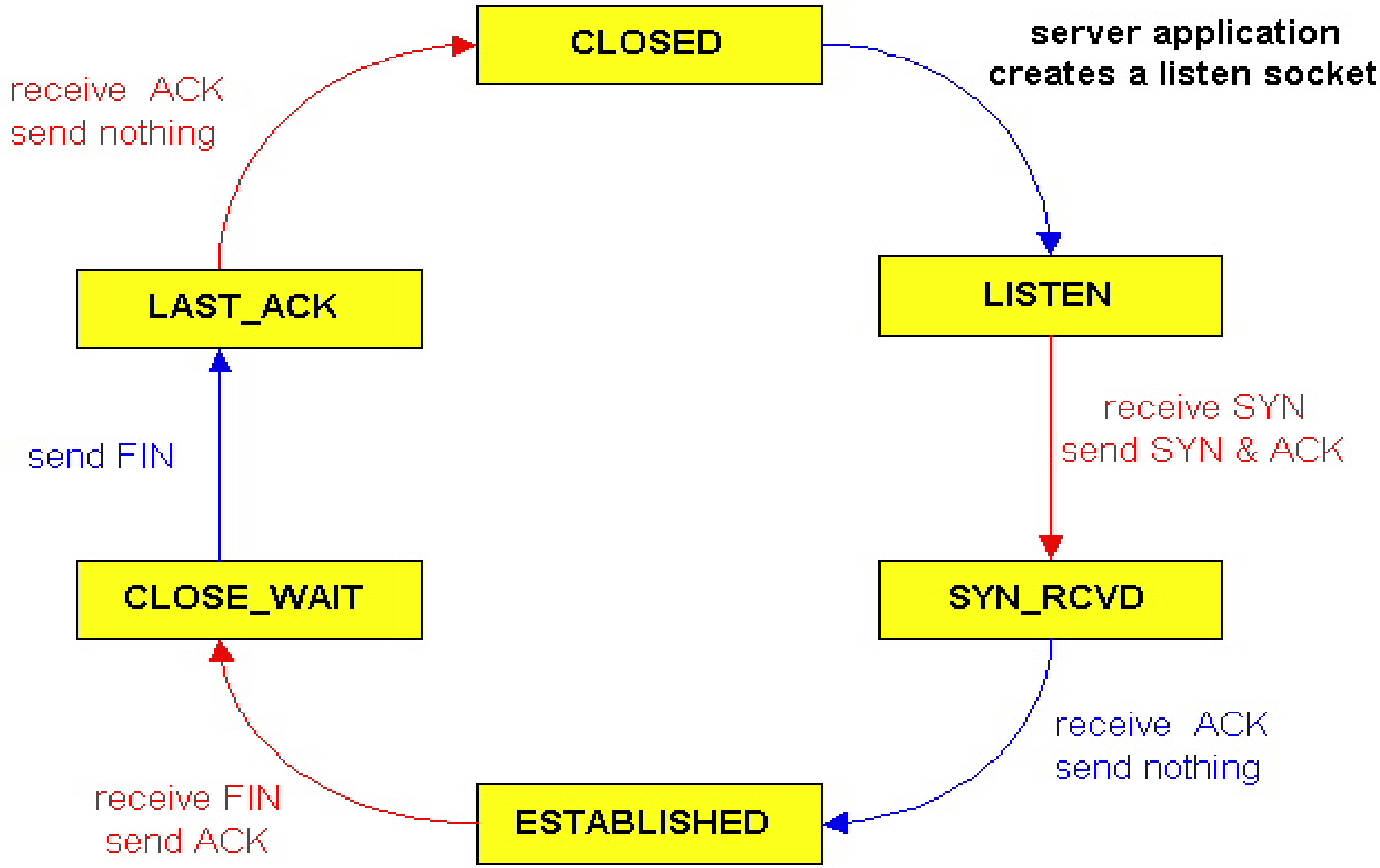
Step 4: server, receives ACK. Connection closed



TCP Life Cycle of a Client



TCP Life Cycle of a Server



References

The slides of this lecture are almost exclusively based on

Books:

- Kurose/Ross. Computer Networking: A Top-Down Approach

Slides:

- Kurose/Ross, Material for lecturers