# Lecture 4

Transport Layer, Part I

# Homework #1

Due: 4/15 before Noon

Upload the report file at KLAS

1. Problem solving
   1) From Chapter 1: R11, P6, P11, P13, P25, P31
   2) From Chapter 2: R19, P8, P9
2. Wireshark Labs
   1) Introduction
   2) HTTP
   3) DNS

# Chapter 3: Transport Layer

Our goals:

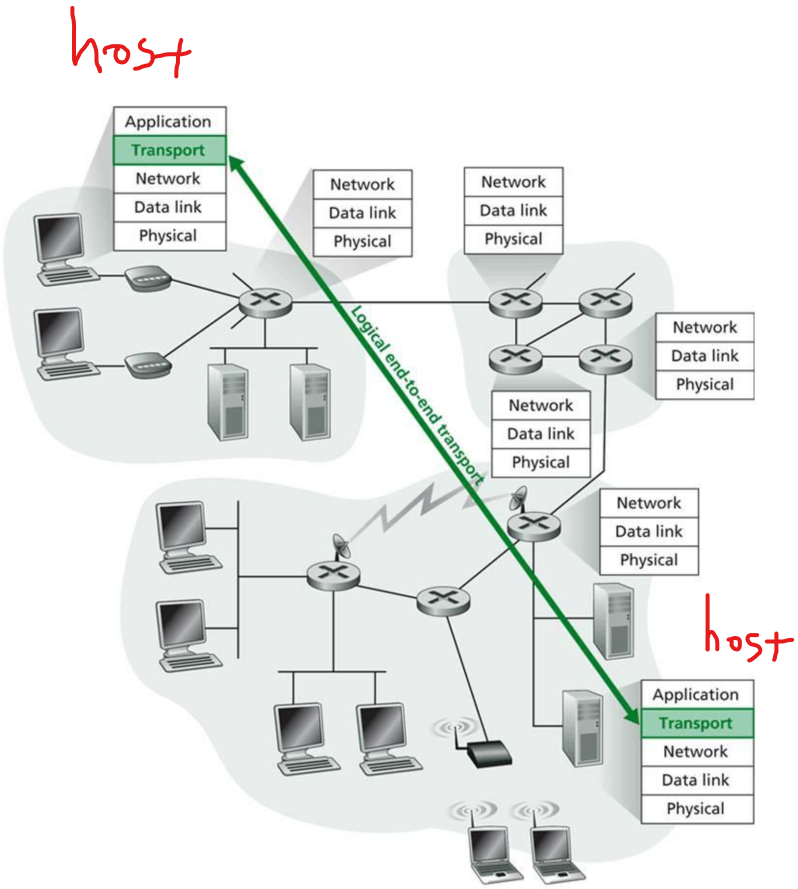
- understand principles behind transport layer services:
  - multiplexing/demultiplexing
  - reliable data transfer
  - flow control
  - congestion control
- learn about transport layer protocols in the Internet:
  - UDP: connectionless transport
  - TCP: connection-oriented transport
  - TCP congestion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

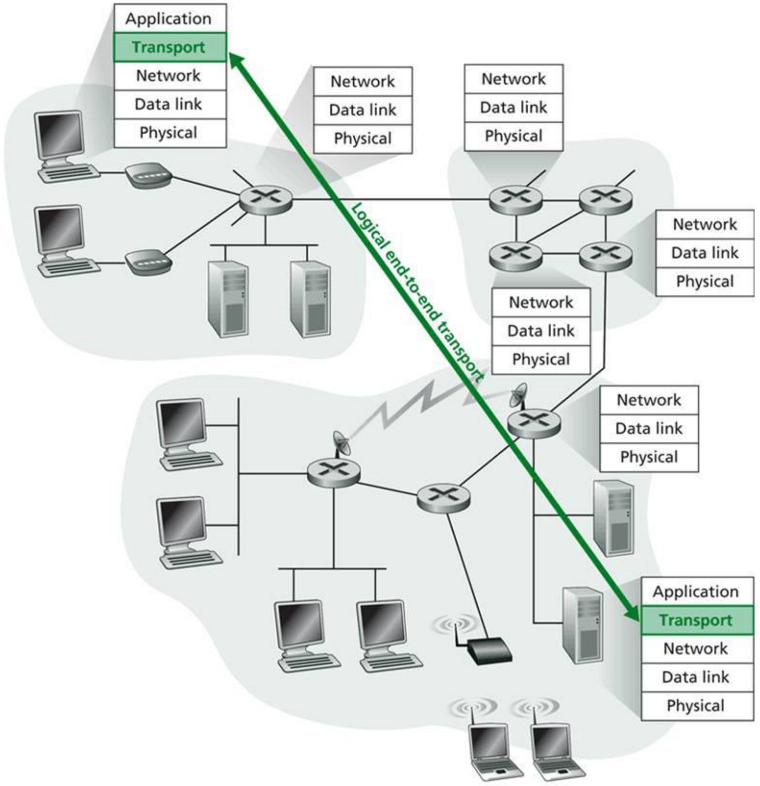
<u>Household analogy:</u>

*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

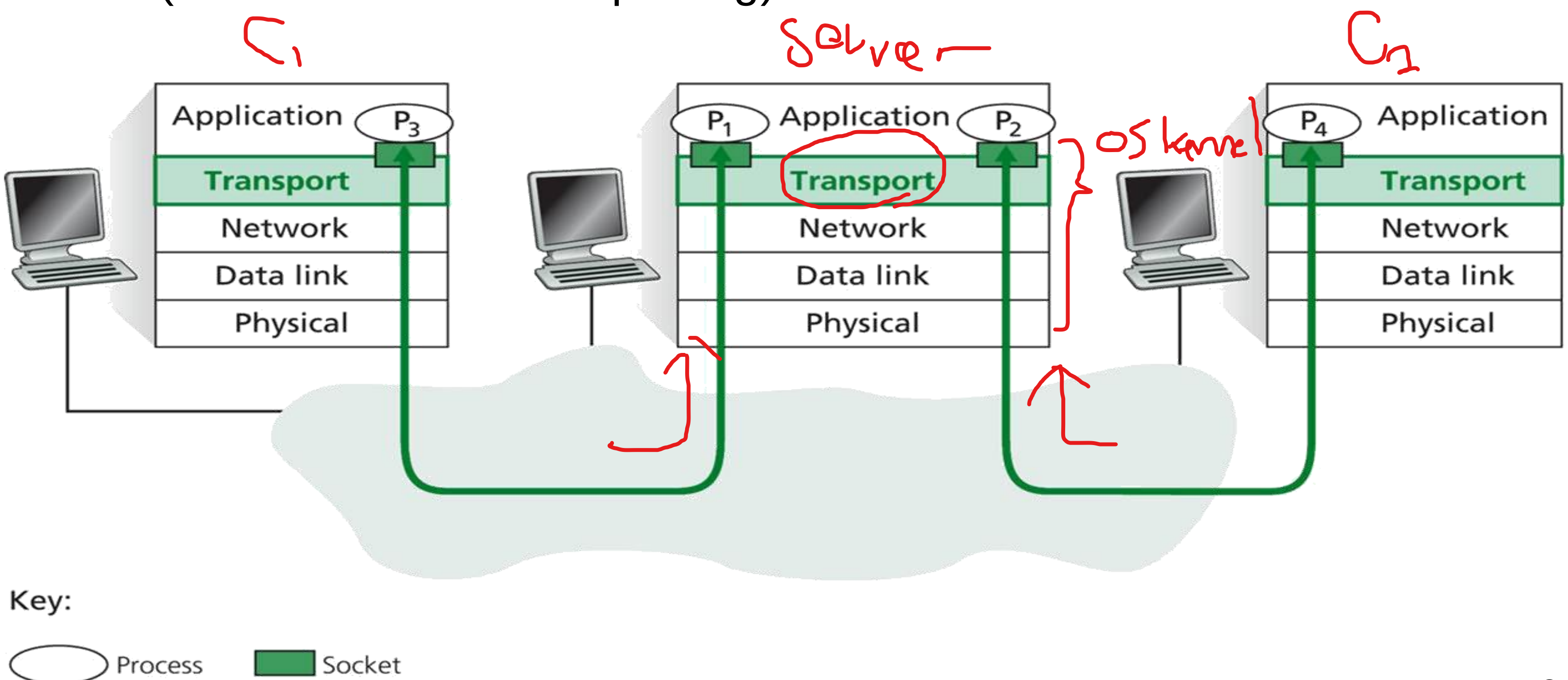
# Internet transport-layer protocols

- 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



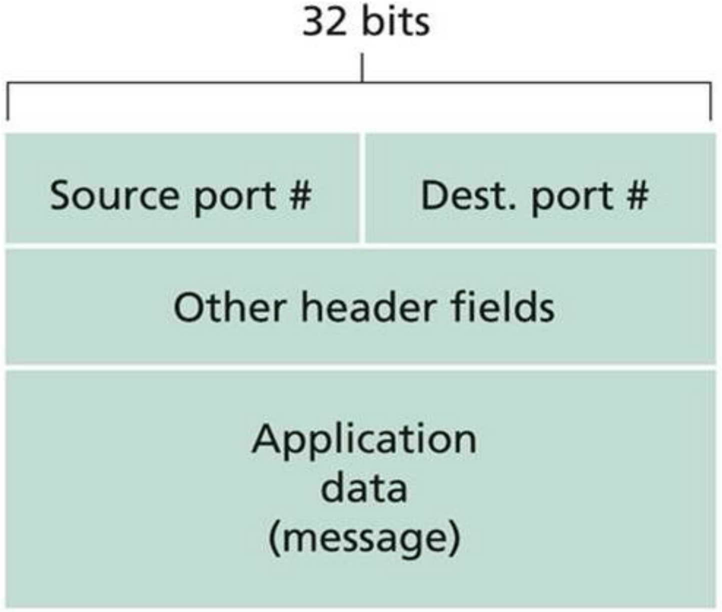
# Multiplexing/demultiplexing

- 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
- host uses IP addresses & port numbers to direct segment to appropriate socket




TCP/UDP segment format

# Connectionless demultiplexing

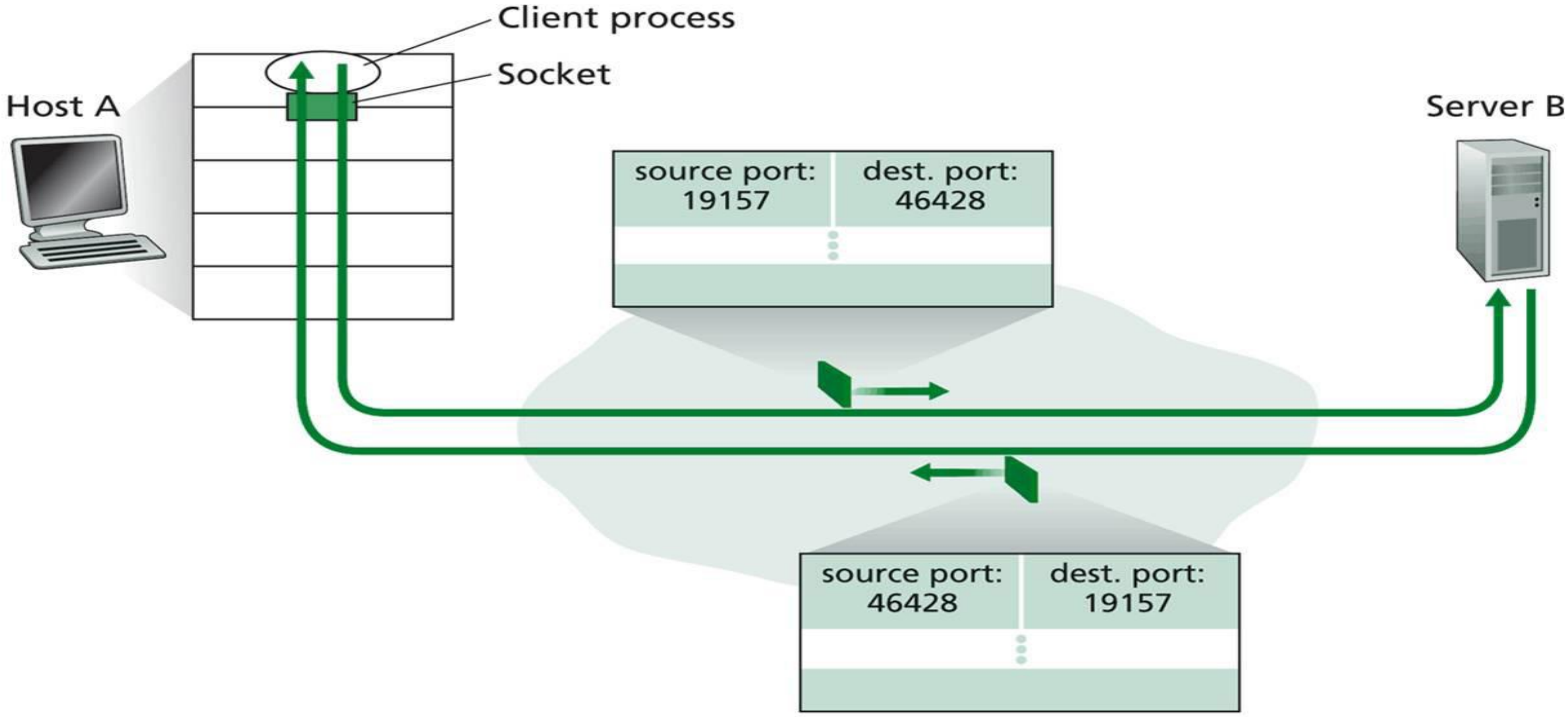
- Create sockets with port numbers:

```
DatagramSocket mySocket1 = new DatagramSocket(19157);
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);
```

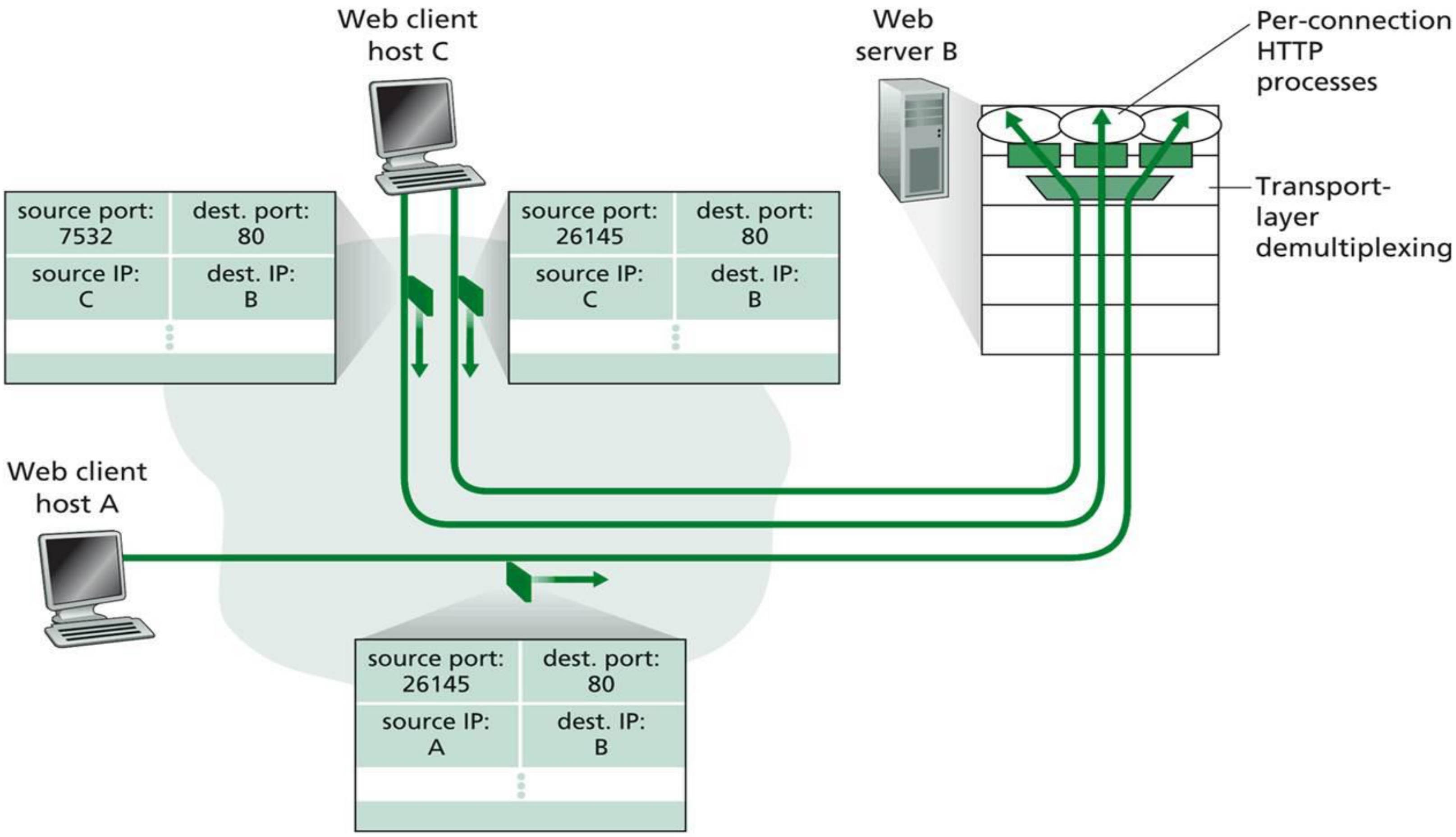



SP provides "return address"

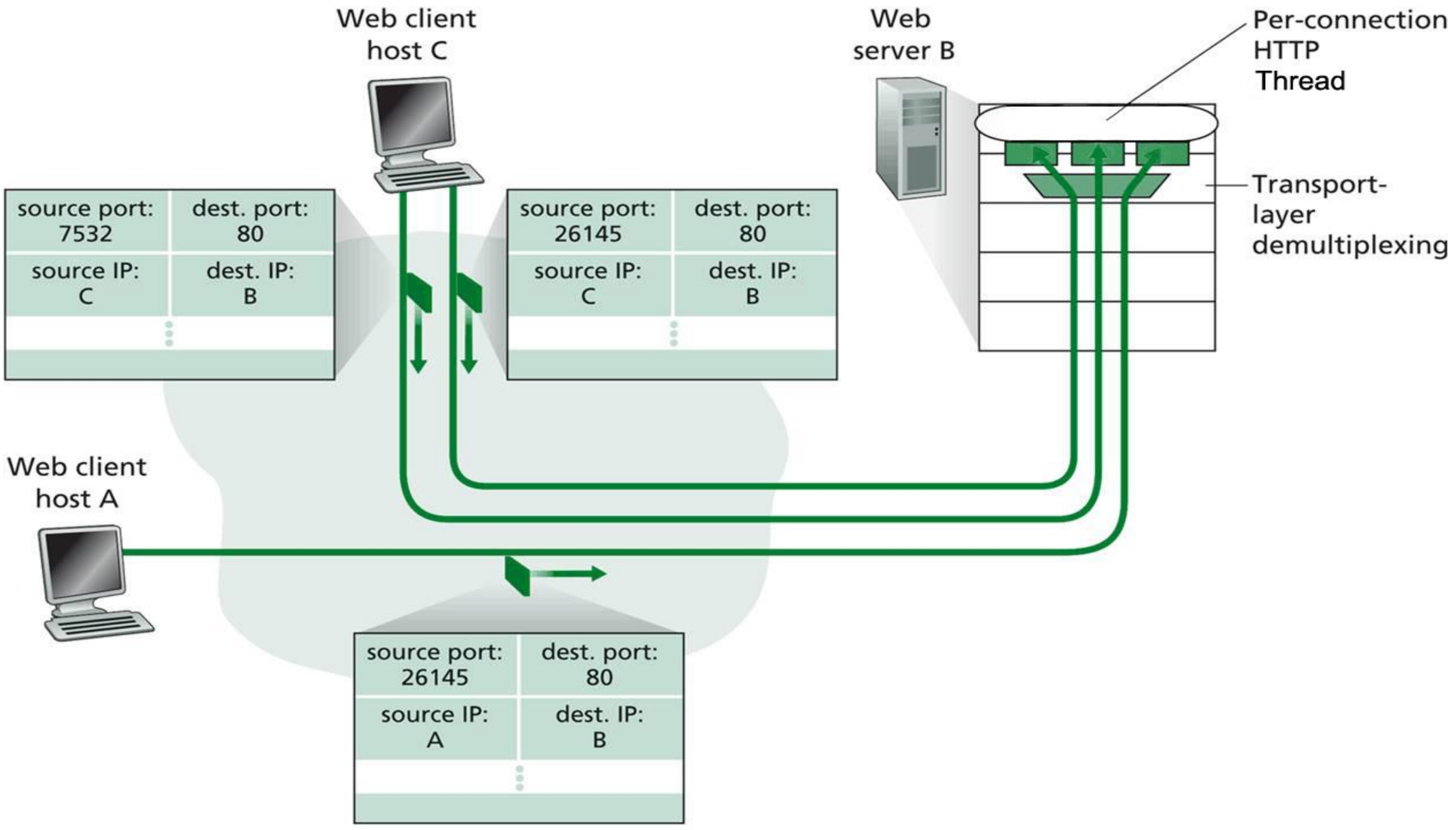
# Connection-oriented demux

- 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



# UDP: User Datagram Protocol [RFC 768]

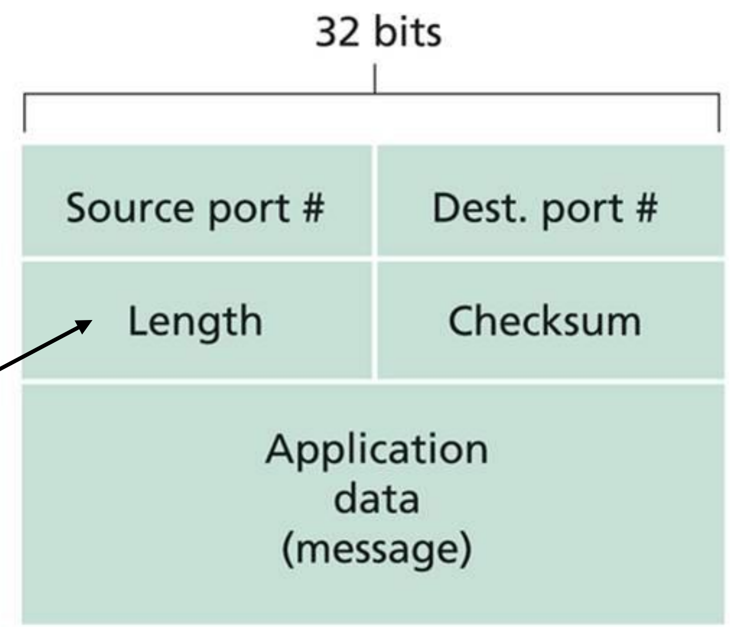
- "no frills," "bare bones" Internet transport protocol
- "best effort" service, UDP segments may be:
  - lost
  - delivered out of order to app
- *connectionless:*
  - no handshaking between UDP sender, receiver
  - each UDP segment handled independently of others

*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: more

- often used for streaming multimedia apps
  - loss tolerant
  - rate sensitive
- other UDP uses
  - DNS
  - SNMP
- reliable transfer over UDP: add reliability a
  - application-specific error recovery!

Length, in bytes of UDP segment,
including header




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?* More later ....

# Internet checksum: 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 |
| 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 |

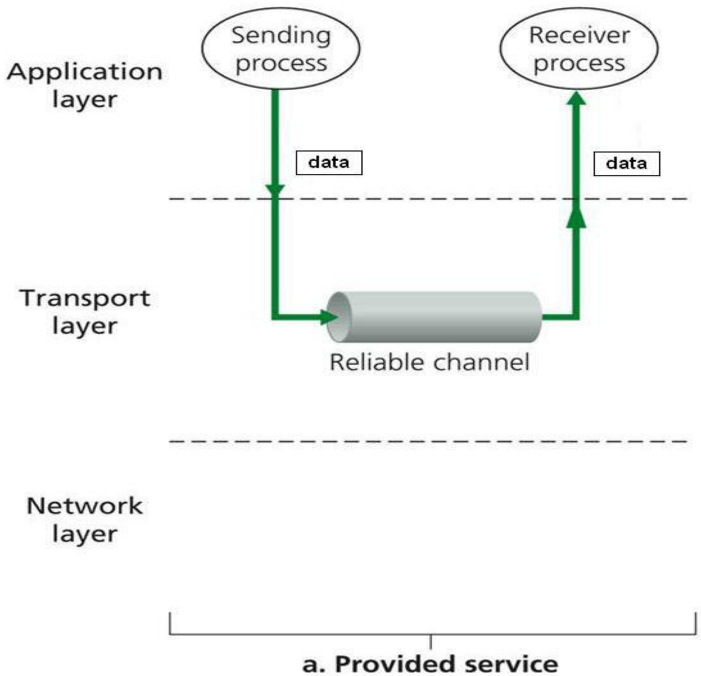
*Note:* when adding numbers, a carryout from the most significant bit needs to be added to the result

*Sender transmits both of 16-bit numbers and checksum!*

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

# Principles of Reliable data transfer

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




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

# Principles of Reliable data transfer

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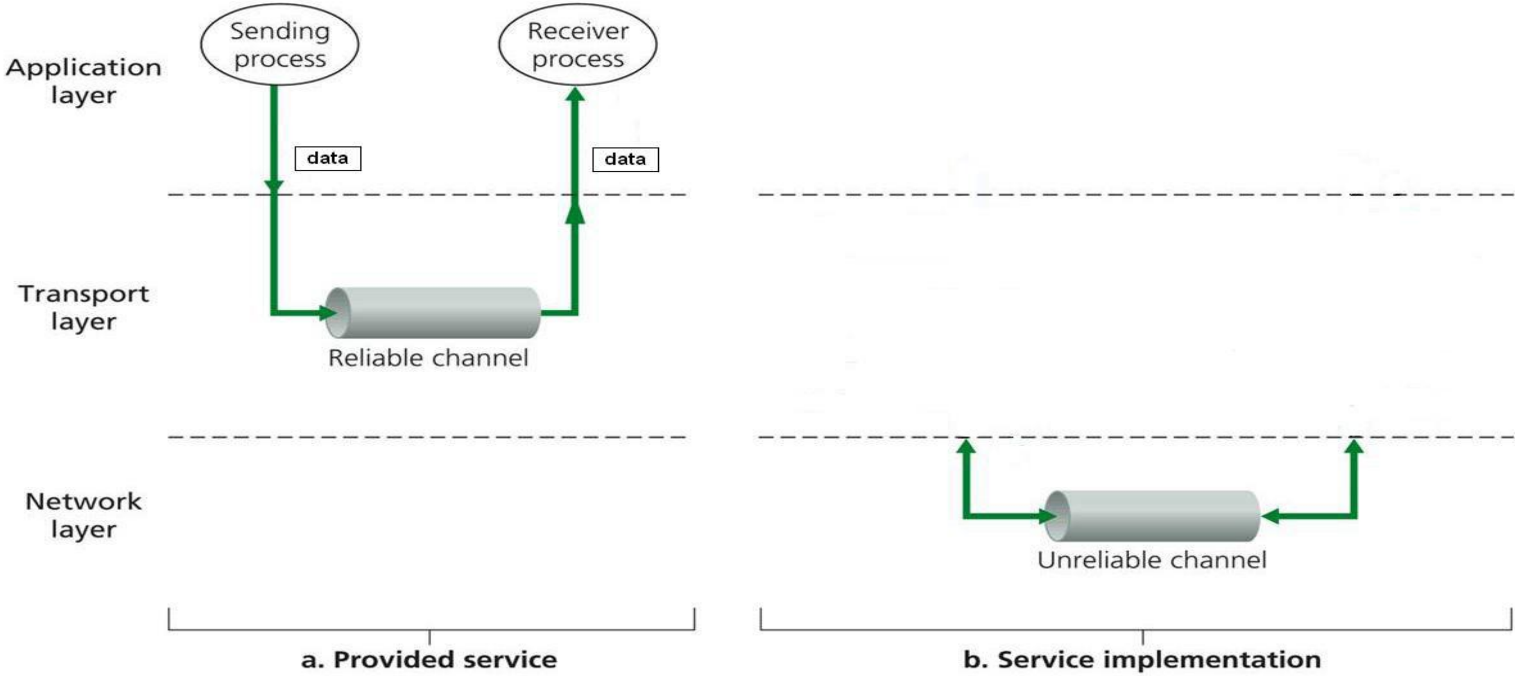


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# Principles of Reliable data transfer

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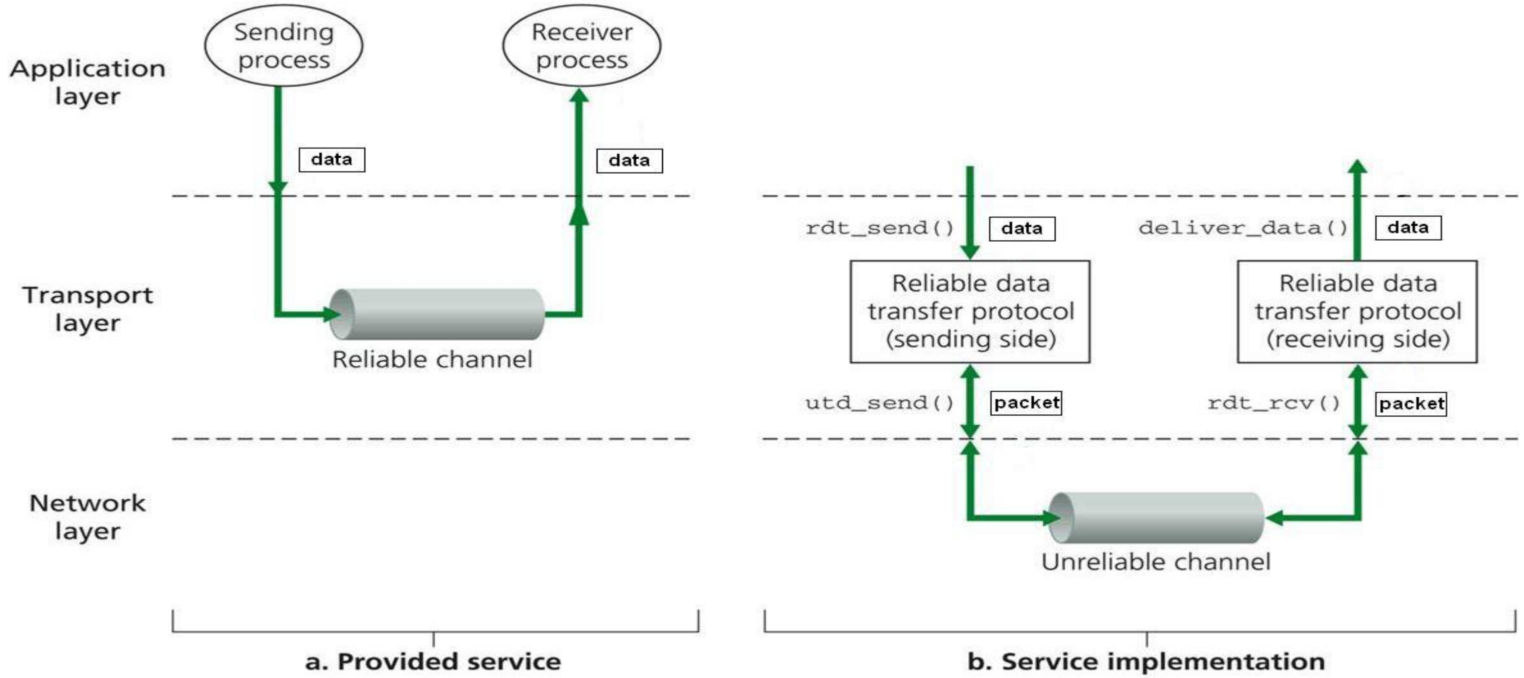


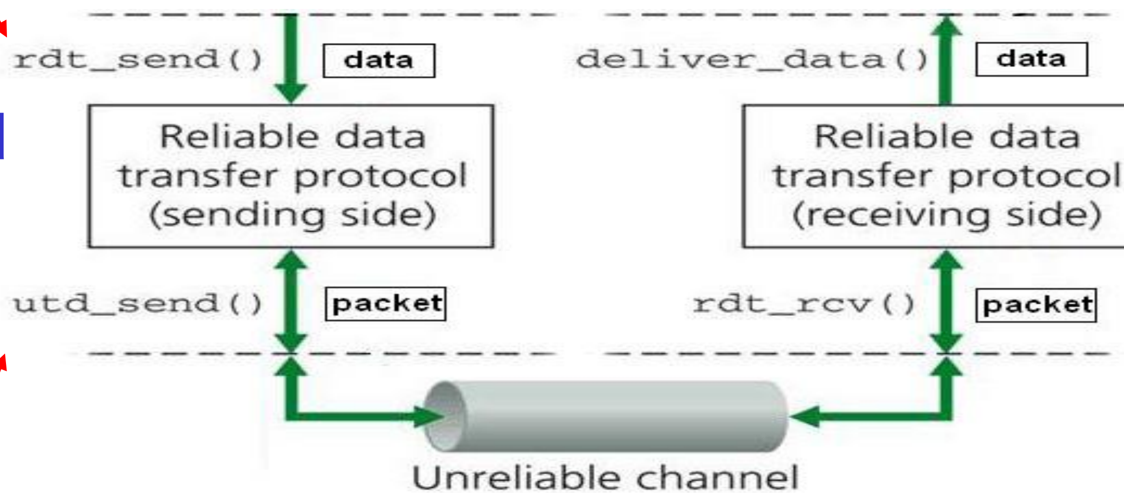

- characteristics of unreliable channel will determine complexity of reliable data transfer protocol (rdt)

# 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

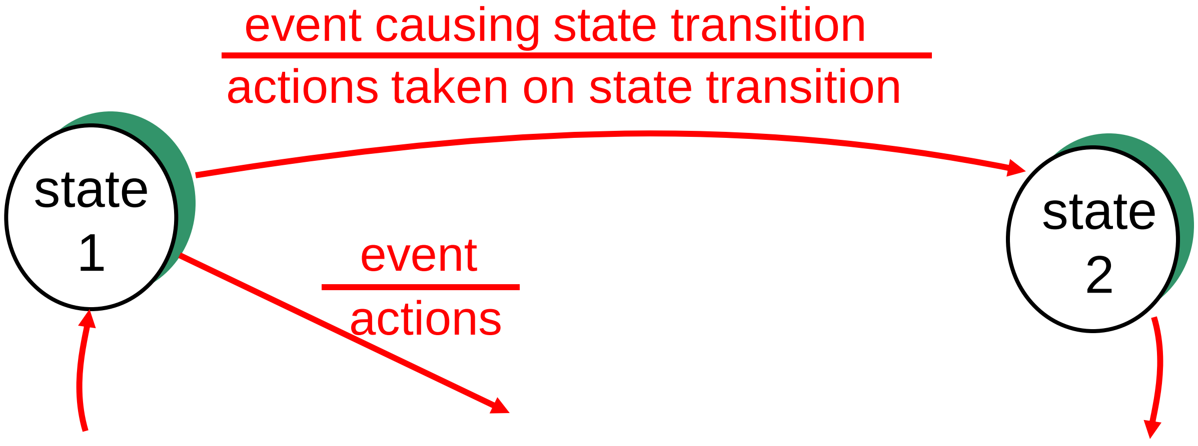
**rdt_rcv():** called when packet arrives on rcv-side of channel

# Reliable data transfer: getting started

We'll:

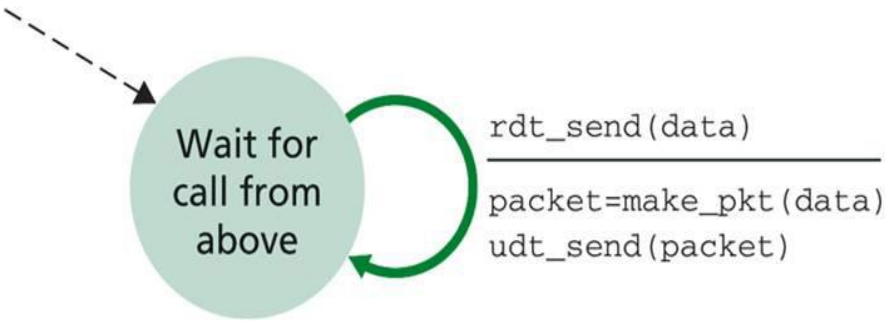
- incrementally develop sender, receiver sides of reliable data transfer protocol (rdt)
- consider only unidirectional data transfer
  - but control info will flow on both directions!
- use finite state machines (FSM)  to specify sender, receiver

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

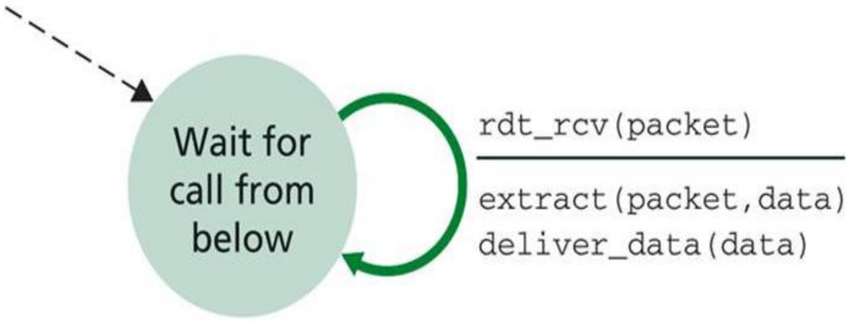


# rdt1.0: reliable transfer over a reliable channel

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




a. rdt1.0: sending side

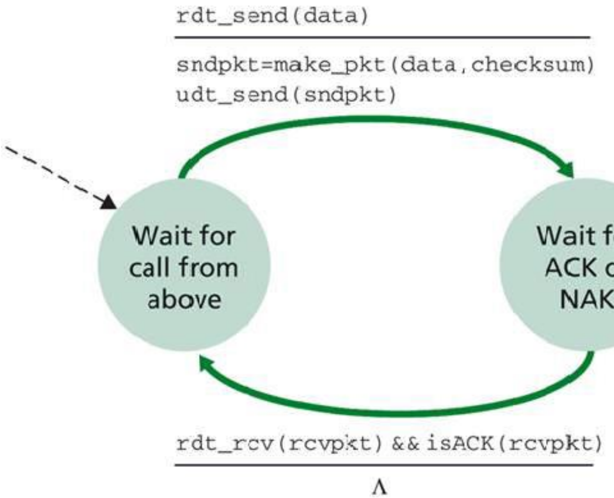



b. rdt1.0: receiving side

# rdt2.0: channel with bit errors

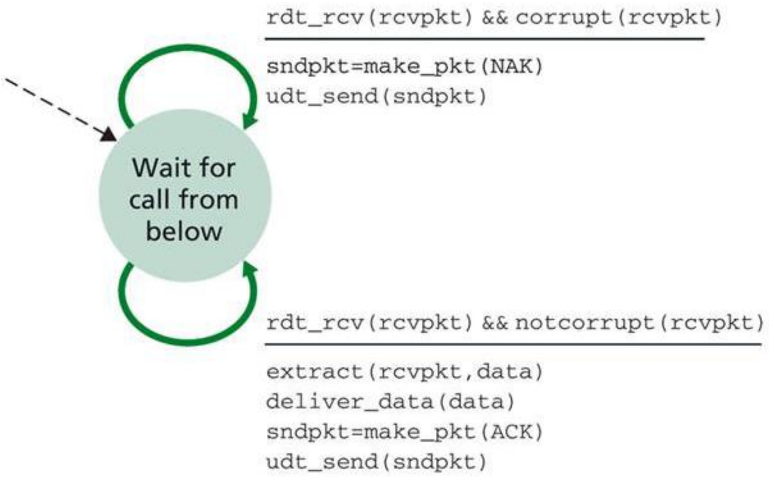
- underlying channel may flip bits in packet
  –
- *the* question: how to recover from errors:
  –
  –
  –
- new mechanisms in **`rdt2.0`** (beyond **`rdt1.0`**):
  - error detection
  - receiver feedback: control msgs (ACK,NAK) rcvr->sender

# rdt2.0: FSM specification

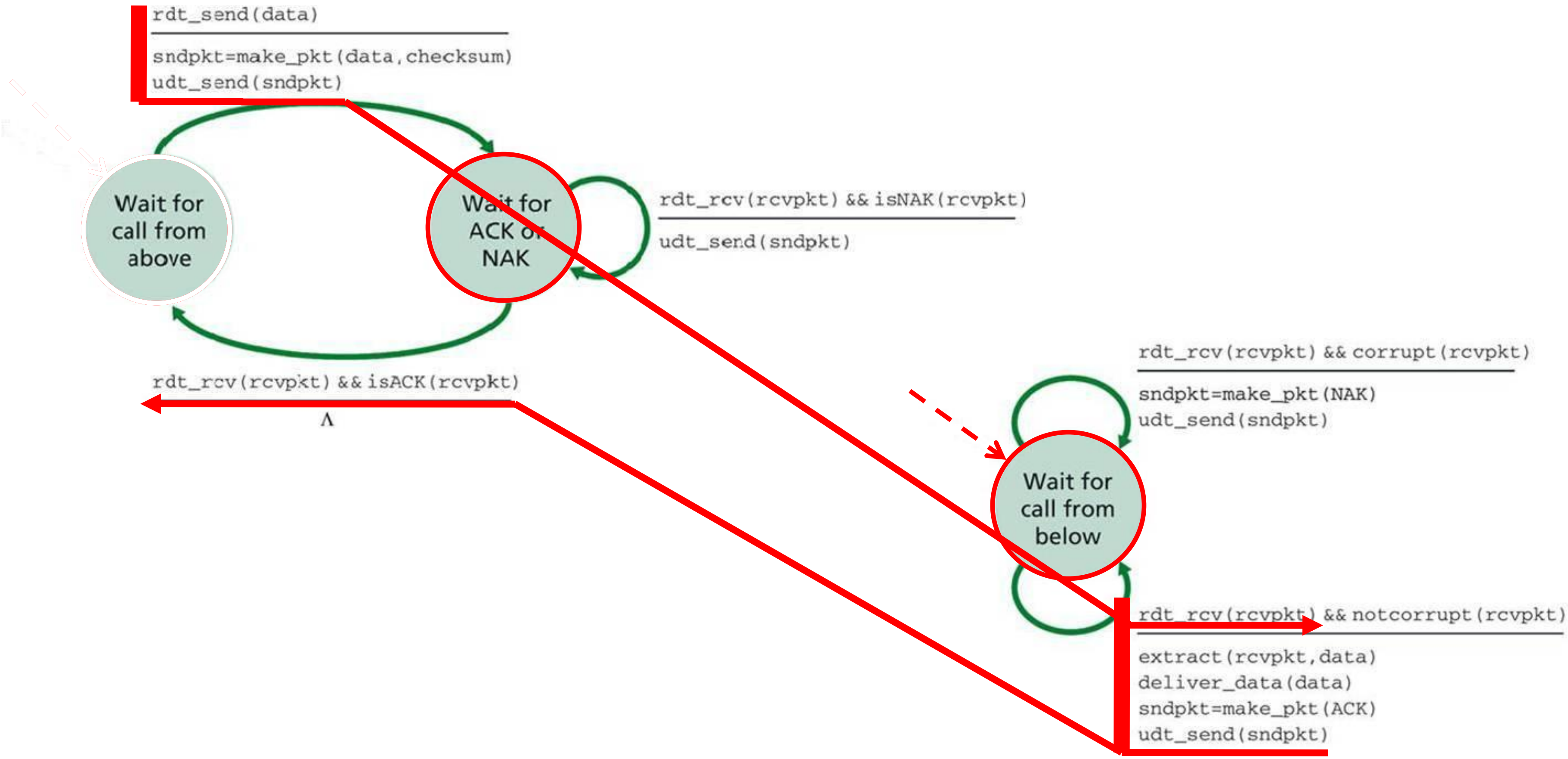



sender

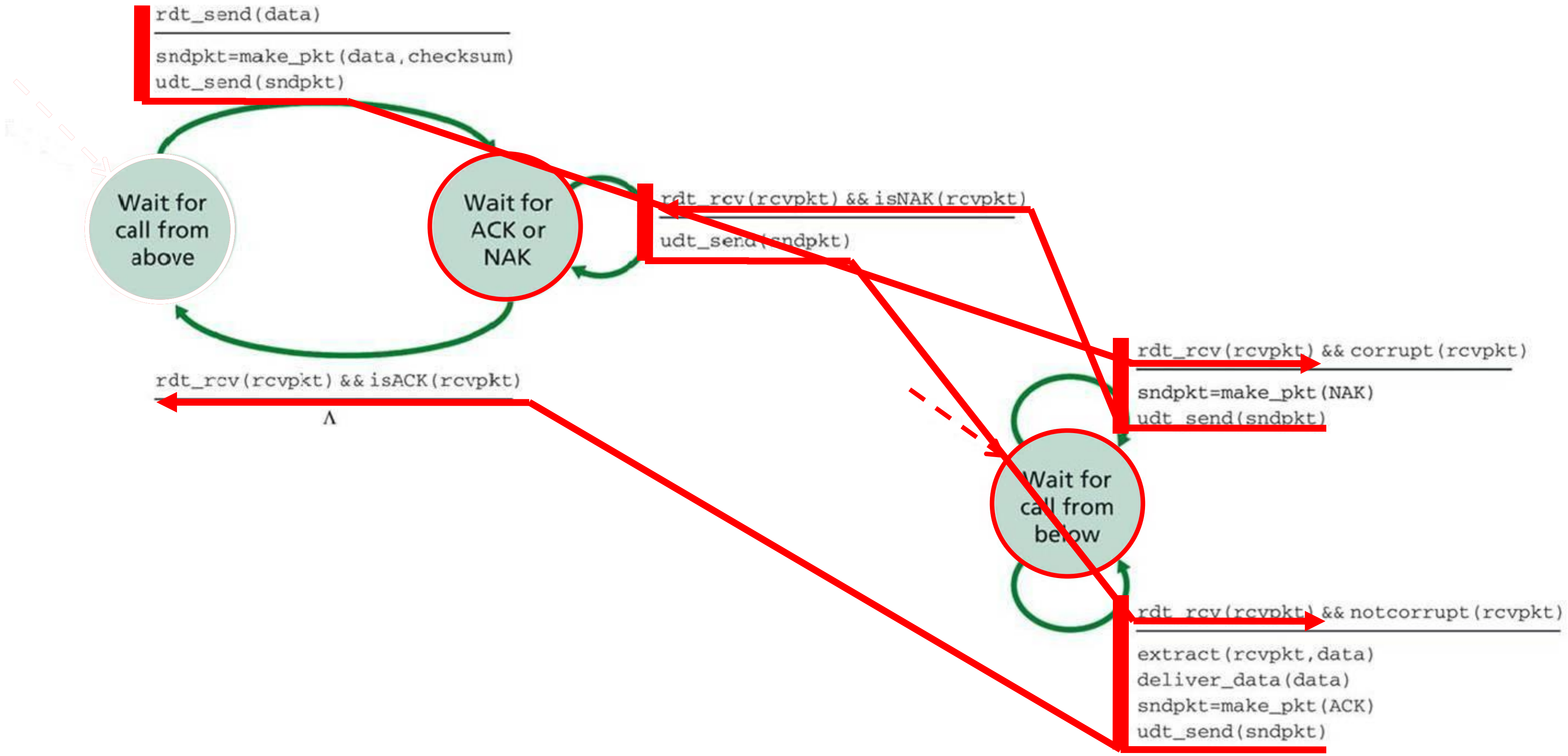
receiver



# rdt2.0: operation with no errors



# rdt2.0: error scenario



# rdt2.0 has a fatal flaw!

What happens if ACK/NAK corrupted?

- 
- 

Handling duplicates:

- 
- 
- 

Stop and wait

- Sender sends one packet, then waits for receiver response

# rdt2.1: discussion

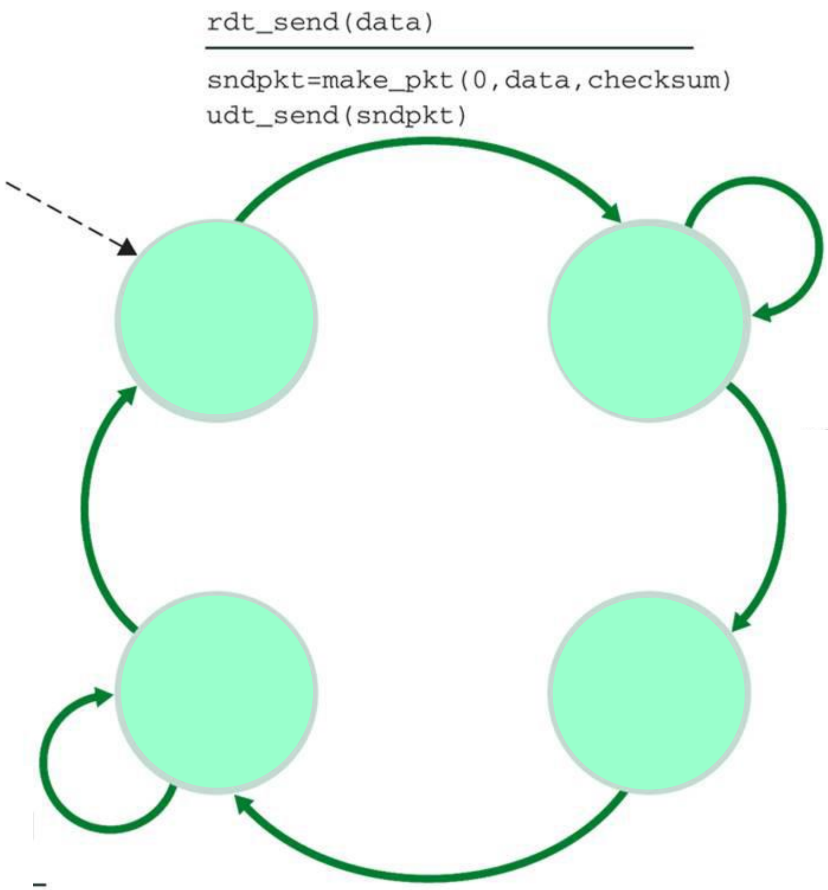
Sender:

- 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. #

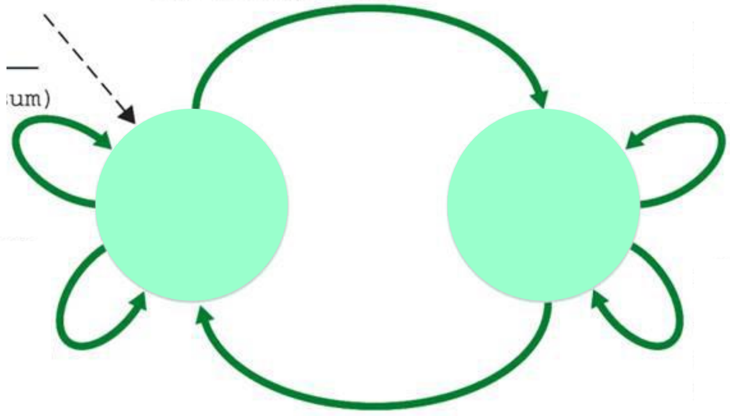
Receiver:

- 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

# rdt2.1: sender, handles garbled ACK/NAKs



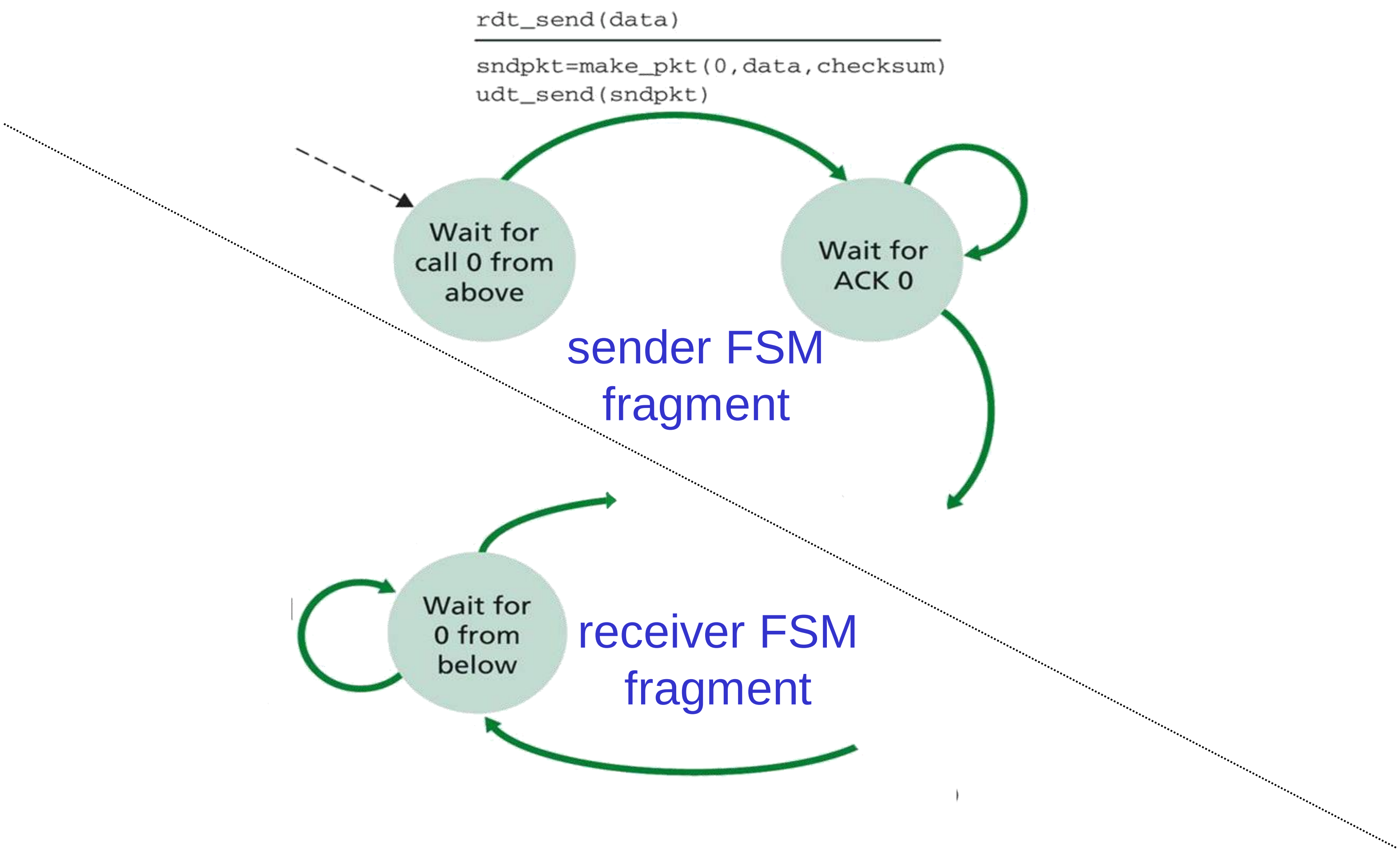
# rdt2.1: receiver, handles garbled ACK/NAKs



# 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
  - 
- 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 lose packets (data or ACKs)

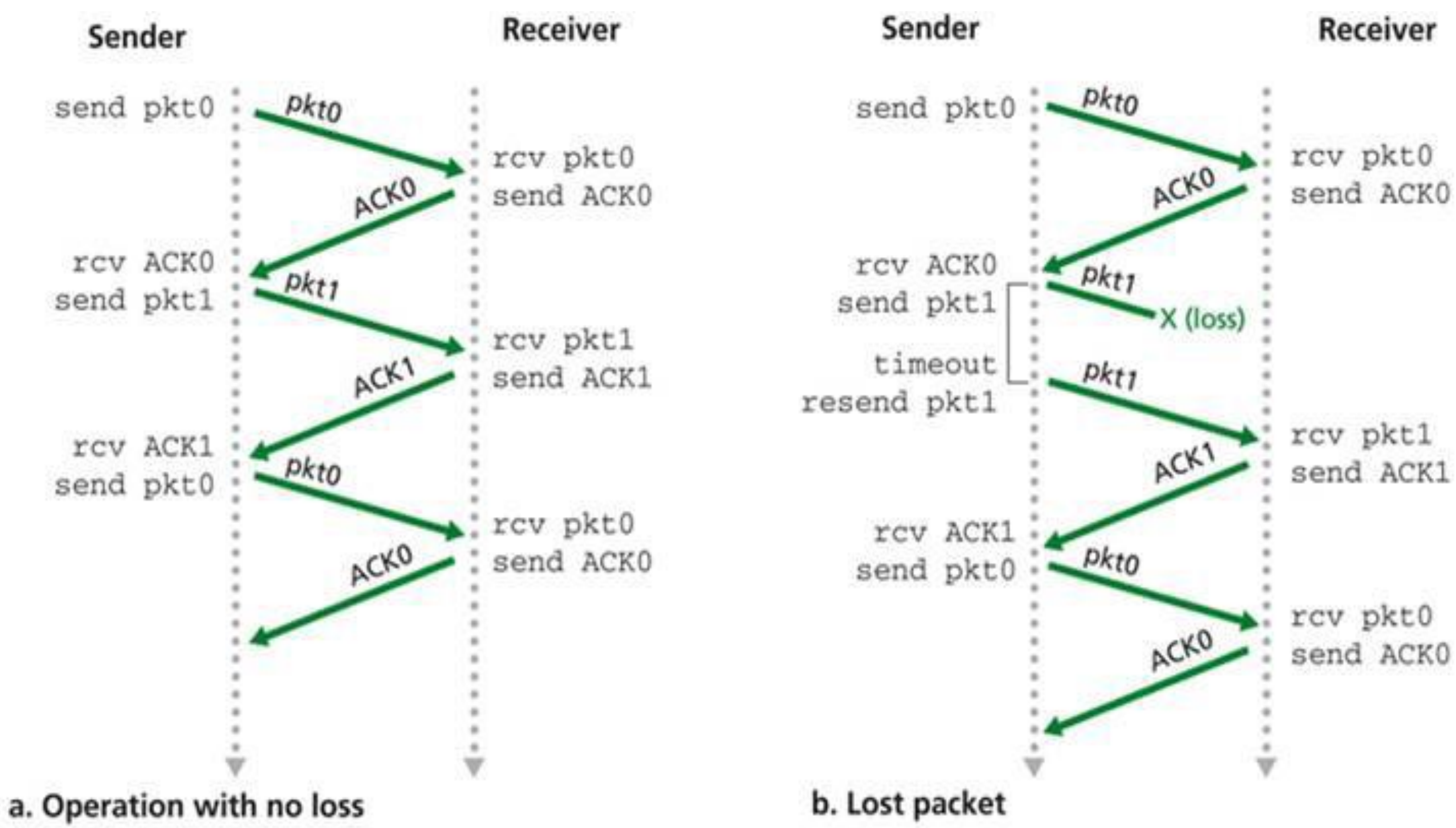
- checksum, seq. #, ACKs, retransmissions will be of help, but not enough

Approach:

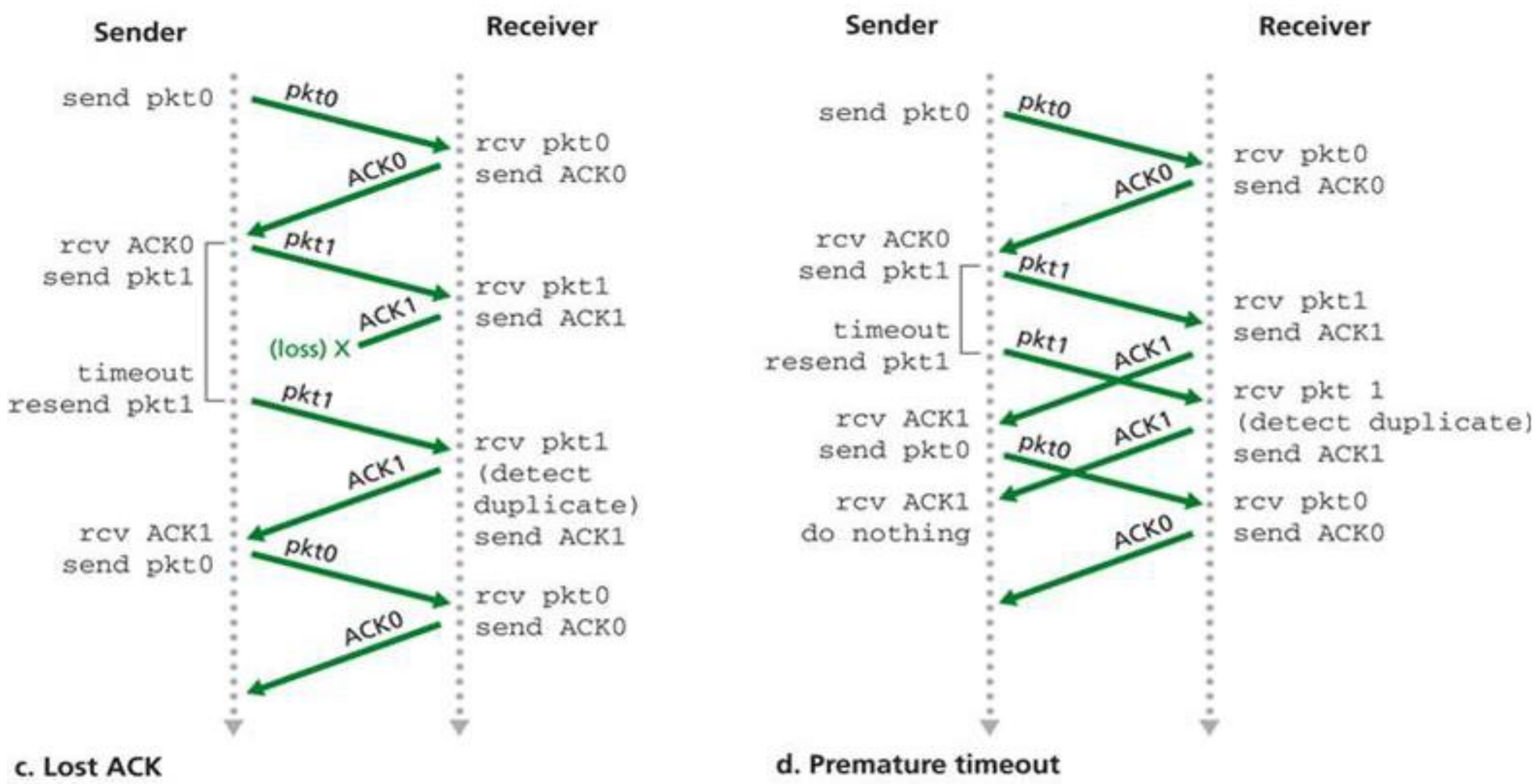
- if pkt (or ACK) just delayed (not lost):
  - 
  - 
- requires

# rdt3.0 sender

# rdt3.0 in action



# 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:

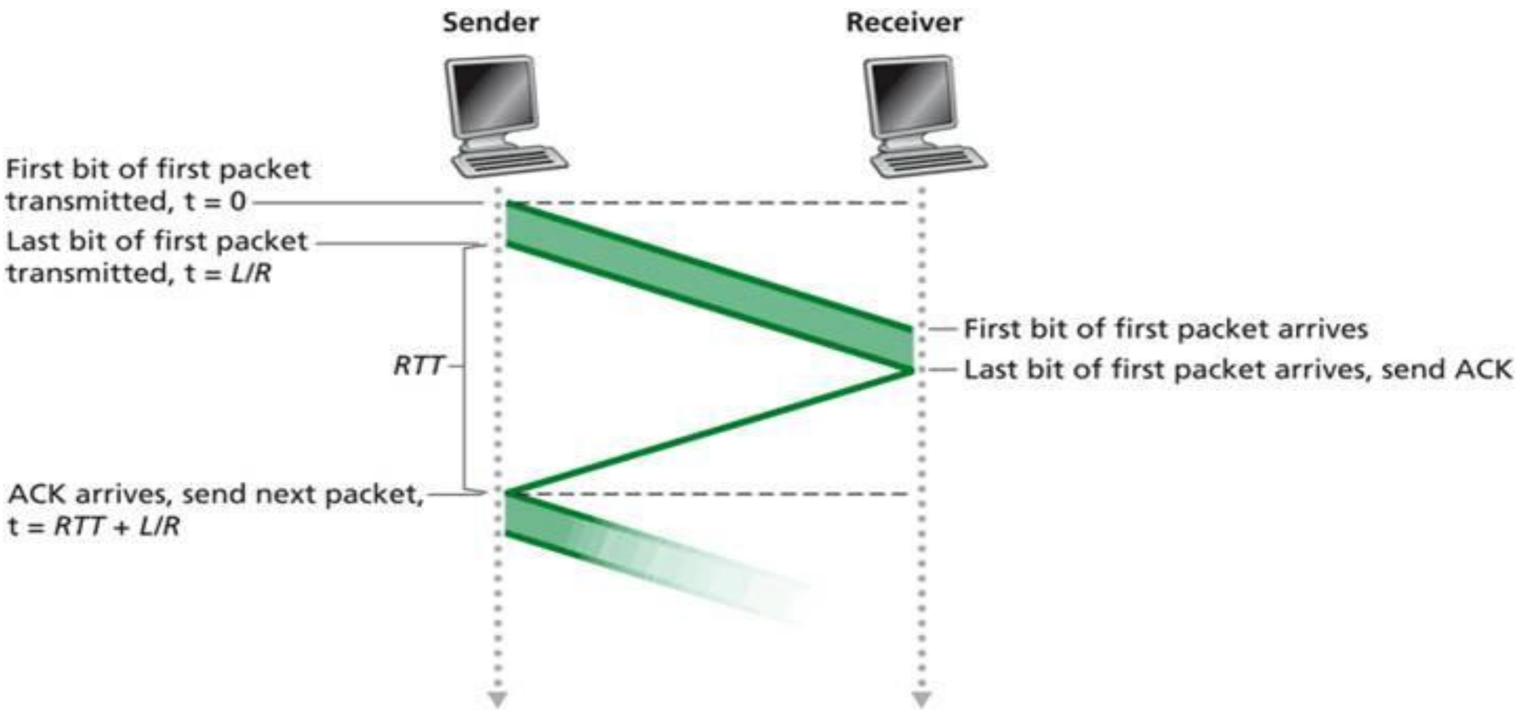
$$T_{transmit} = \frac{L \text{ (packet length in bits)}}{R \text{ (transmission rate, bps)}} = \frac{8\text{kb/pkt}}{10^{**}9 \text{ b/sec}} = 8 \text{ microsec}$$

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

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

- 1KB pkt every 30 msec -> 33kB/sec thruput over 1 Gbps link
- network protocol limits use of physical resources!

# rdt3.0: stop-and-wait operation



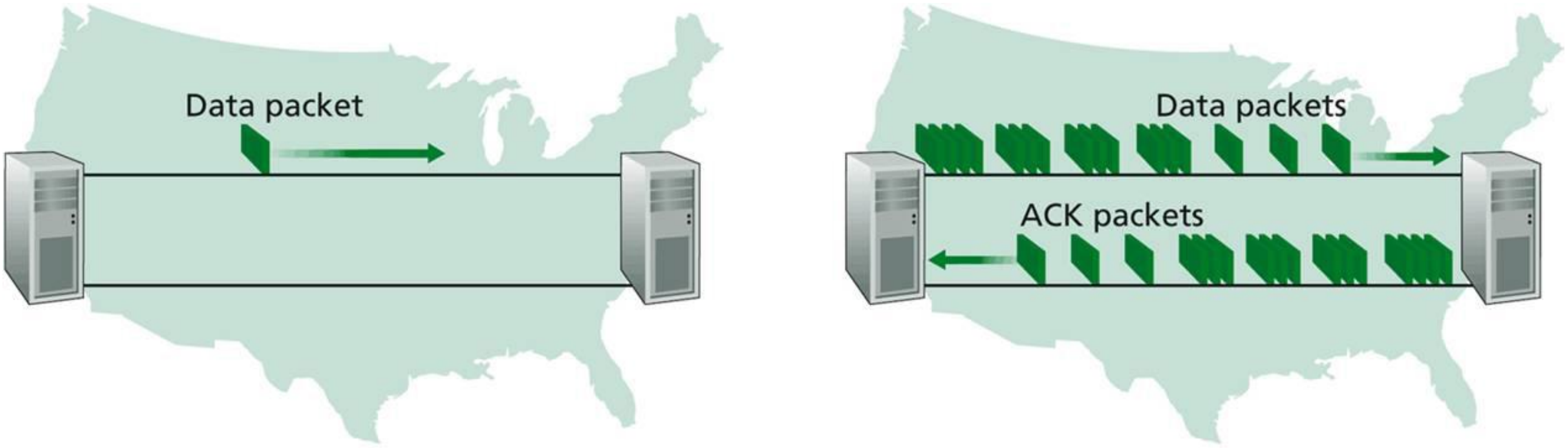

a. Stop-and-wait operation

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

# 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

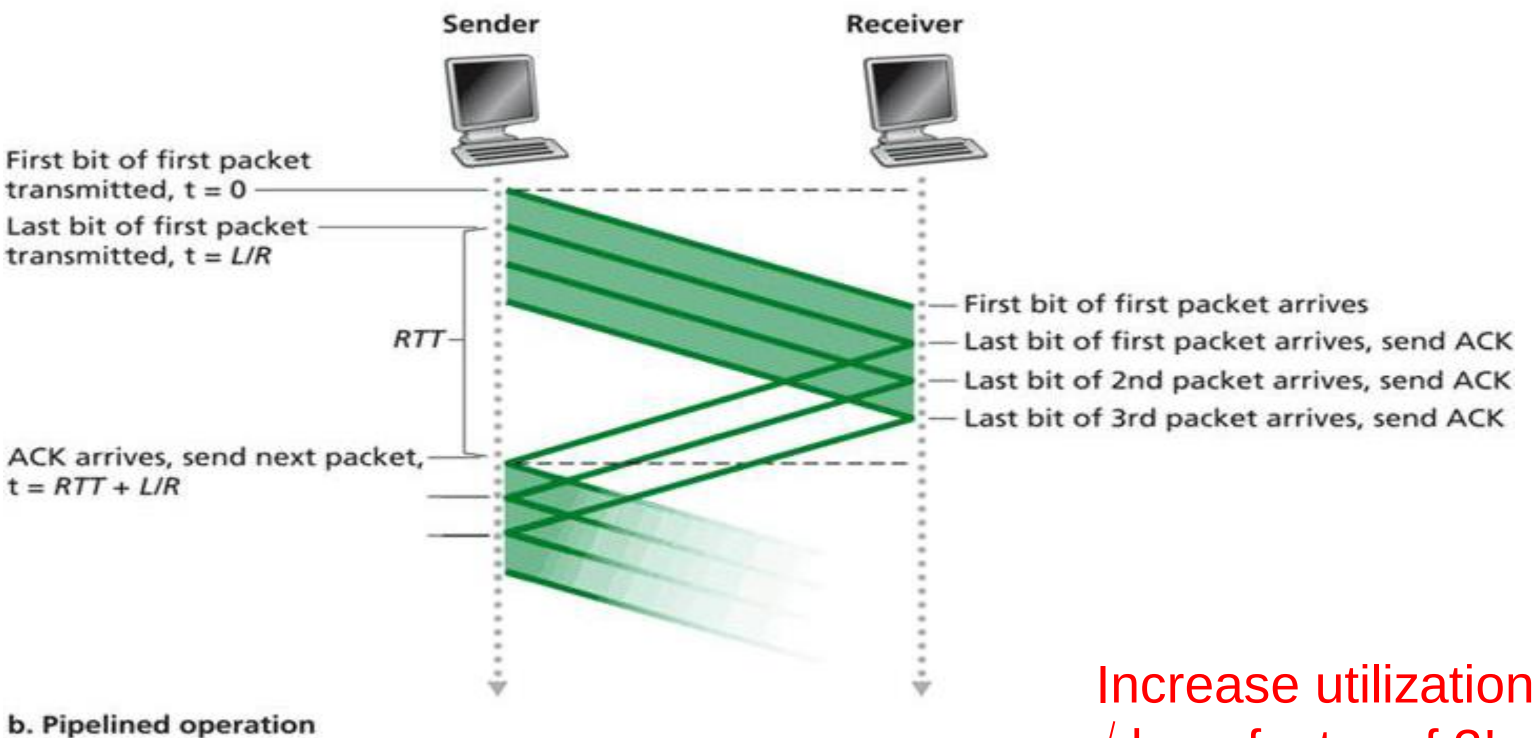



a. A stop-and-wait protocol in operation

b. A pipelined protocol in operation

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

# Pipelining: increased utilization

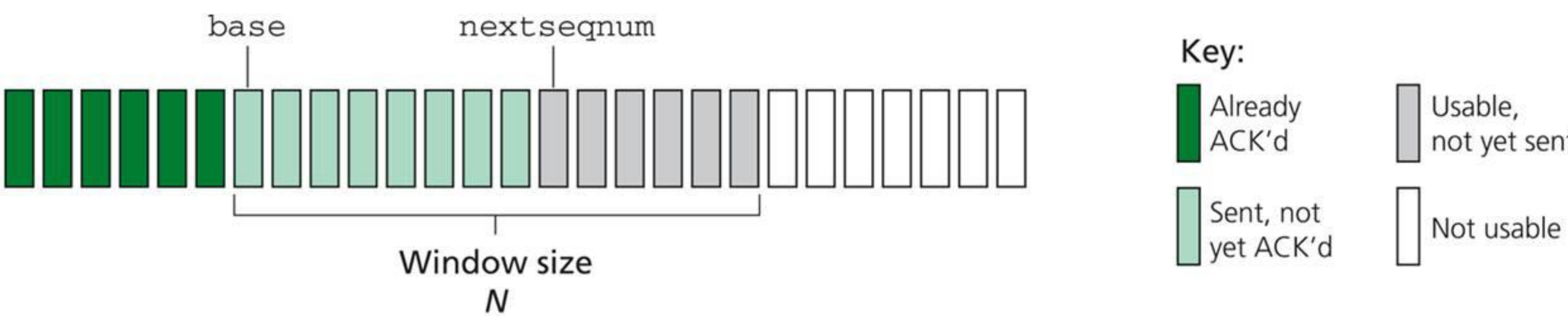



Increase utilization by a factor of 3!

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

# Go-Back-N

Sender:




- ACK :
  - –
- timer
- *timeout* :

# GBN: sender extended FSM

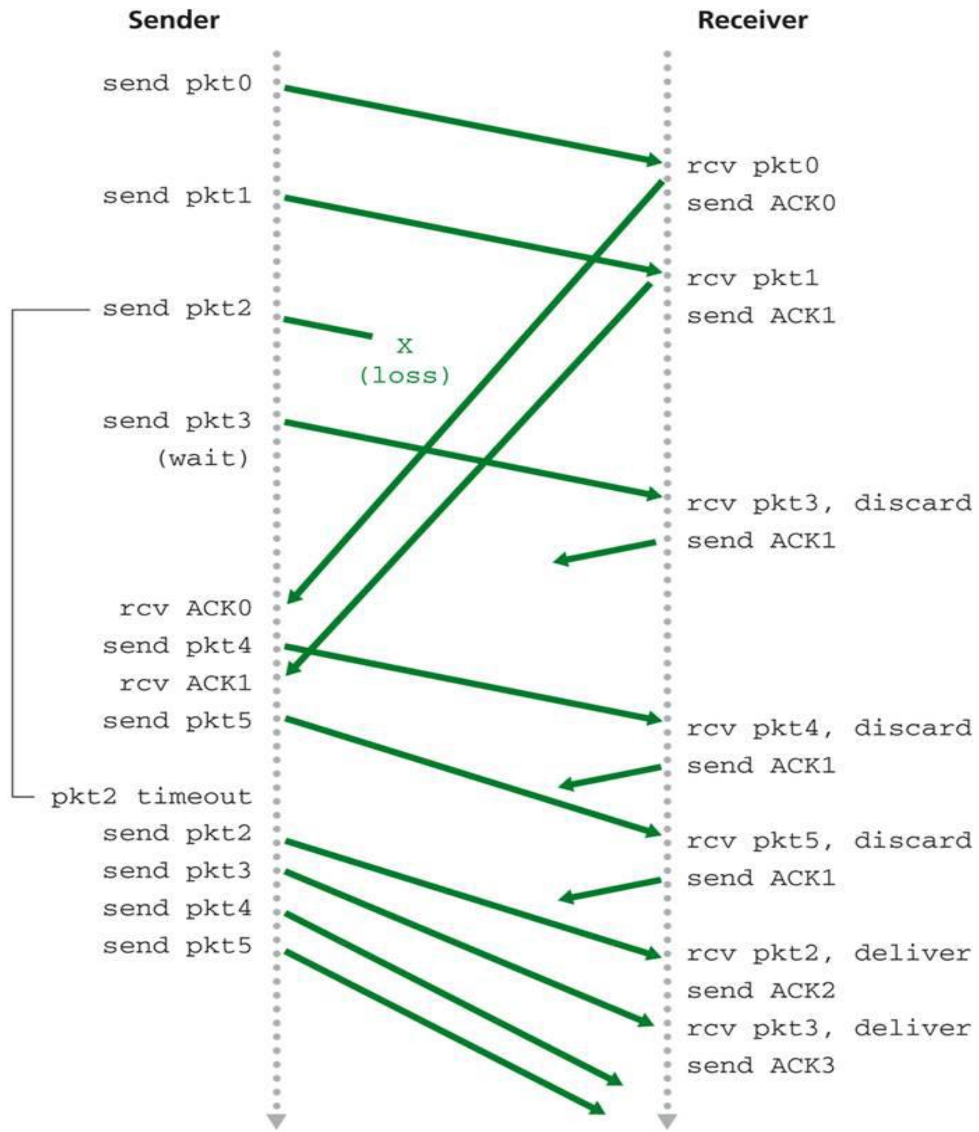
# GBN: receiver extended FSM

ACK-only:

- duplicate ACKs?
- 

- out-of-order pkt:
  - 
  - Ack?

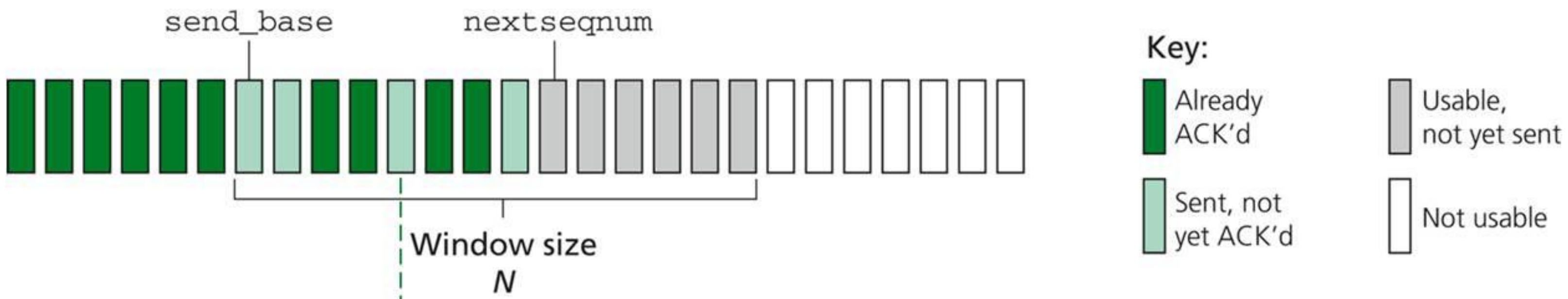
# GBN in action



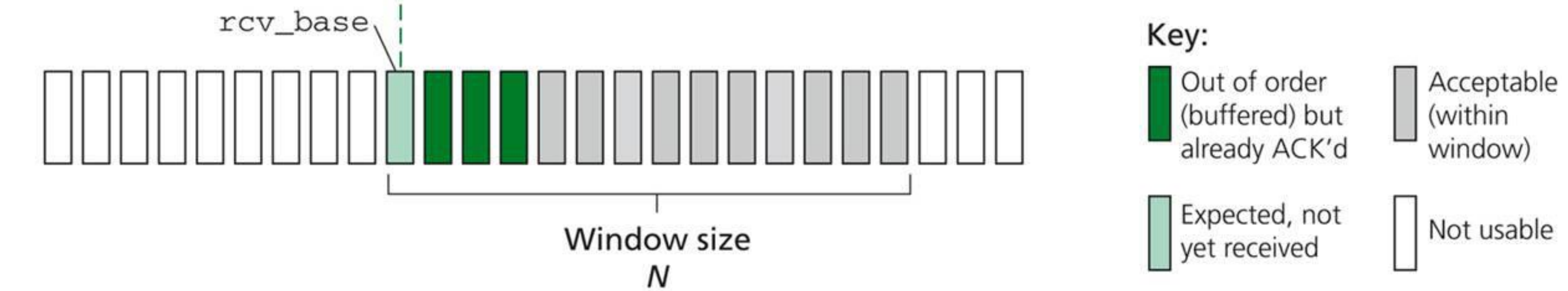
# Selective Repeat

- 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 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 #

# Selective repeat

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

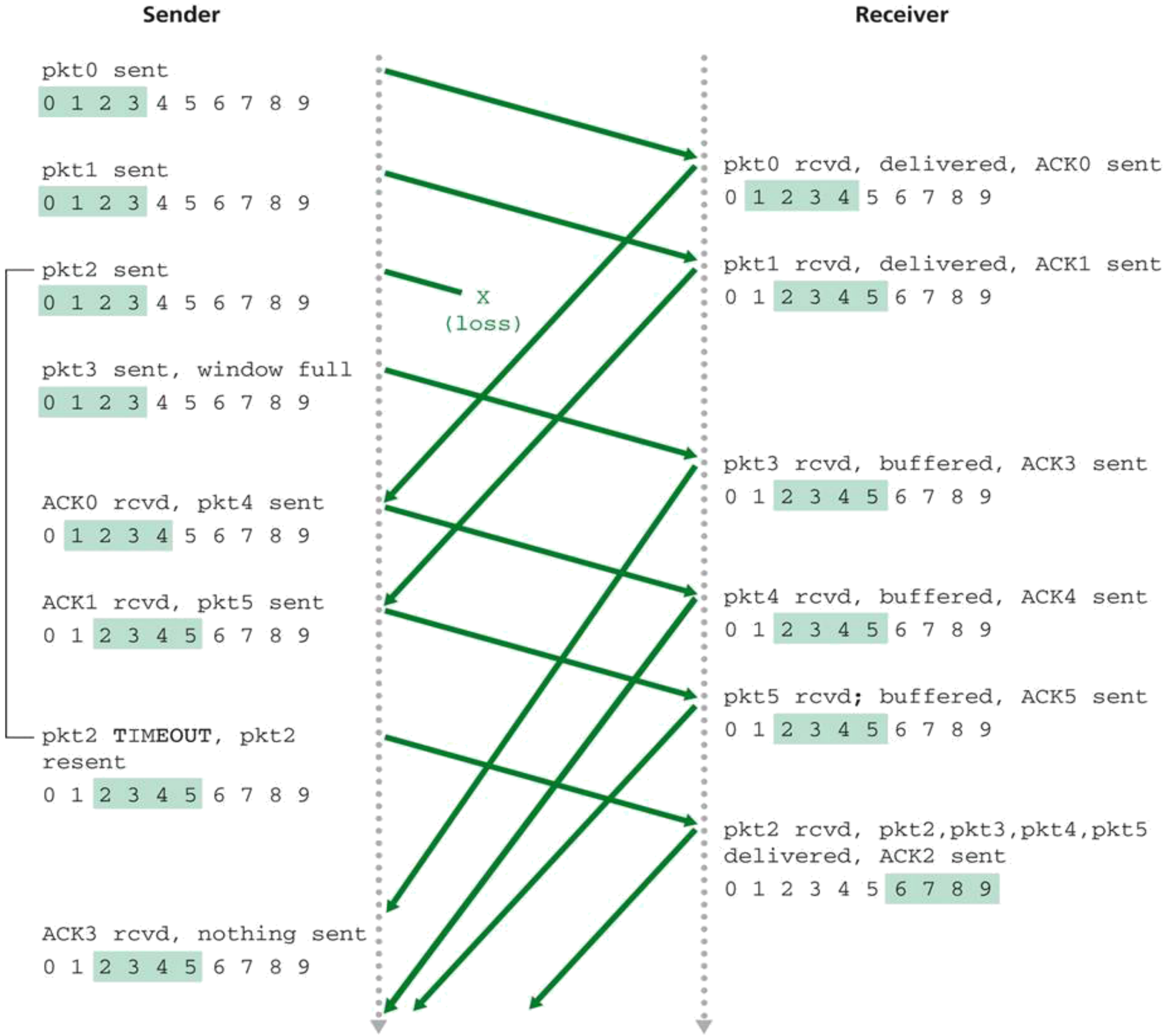
pkt n in [rcvbase-N,rcvbase-1]

- 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 between seq # size and window size?