TDTS06: Computer Networks

Instructor: Niklas Carlsson

Email: niklas.carlsson@liu.se

Notes derived from "*Computer Networking: A Top Down Approach*", by Jim Kurose and Keith Ross, Addison-Wesley.

The slides are adapted and modified based on slides from the book's companion Web site, as well as modified slides by Anirban Mahanti and Carey Williamson.

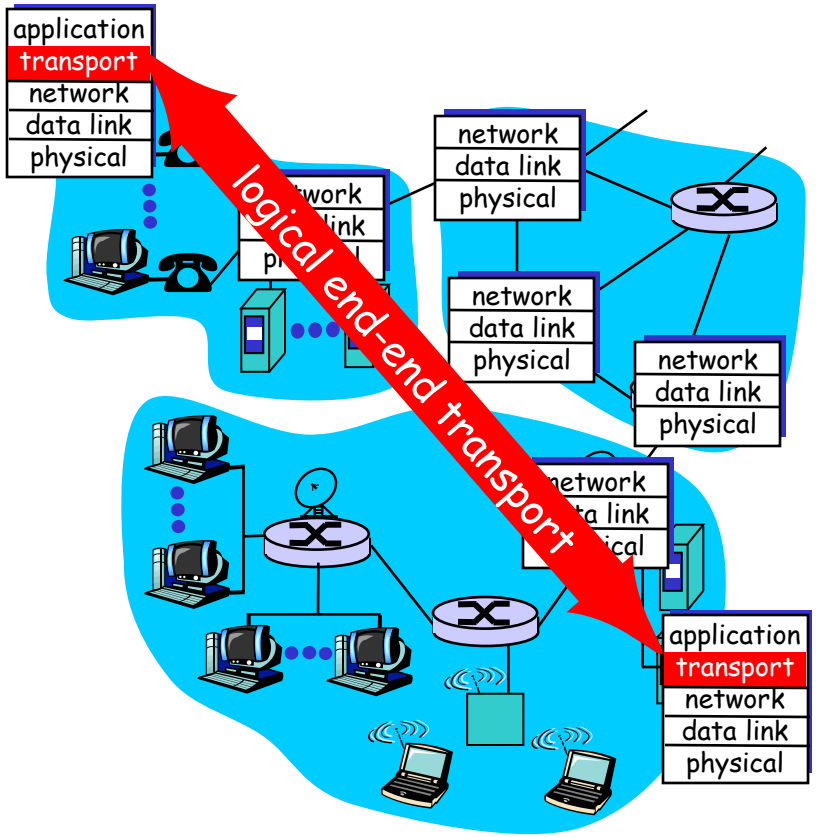
Chapter 3: Transport Layer

Our goals:

- understand principles behind transport layer services:
  - multiplexing and 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

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



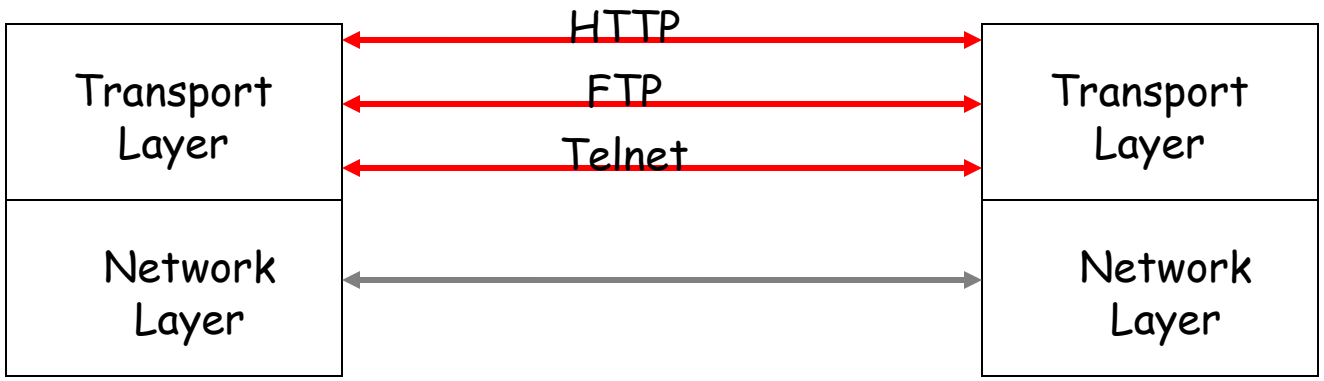
Transport vs. Network Layer

- *transport layer:* logical communication between processes
  - relies on, enhances, network layer services
  - PDU: Segment
  - extends "host-to-host" communication to "process-to-process" communication
- *network layer:* logical communication between hosts
  - PDU: Datagram
  - Datagrams may be lost, duplicated, reordered in the Internet – "best effort" service

TCP/IP Transport Layer Protocols

- reliable, in-order delivery (TCP)
  - connection setup
  - flow control
  - congestion control
- unreliable, unordered delivery: UDP
  - no-frills extension of "best-effort" IP
  - What does UDP provide in addition to IP?
- services not provided by IP (network layer):
  - delay guarantees
  - bandwidth guarantees

Multiplexing/Demultiplexing



- Use same communication channel between hosts for several logical communication processes
- How does Mux/DeMux work?
  - Sockets: doors between process & host
  - UDP socket: (dest. IP, dest. Port)
  - TCP socket: (src. IP, src. port, dest. IP, dest. Port)

Connectionless demux

- 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

Connection-oriented demux

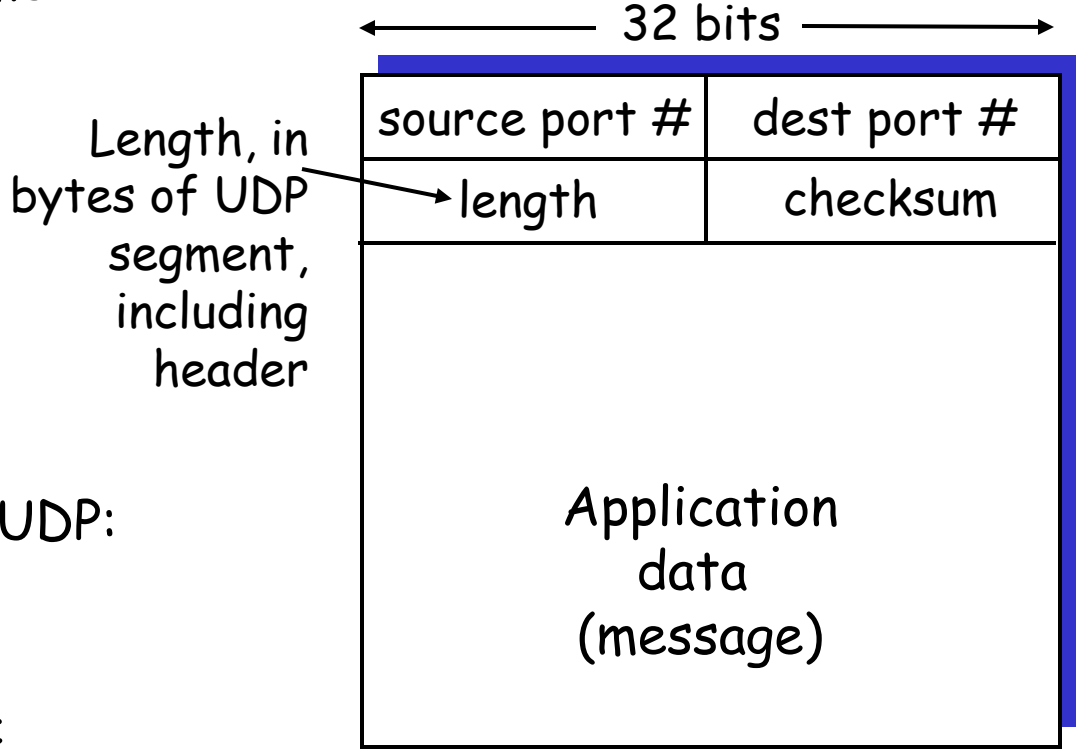
- TCP socket identified by 4-tuple:
  - source IP address
  - source port number
  - dest IP address
  - dest port number
- recv host uses all four values to direct segment to appropriate socket
- Server host may support many simultaneous TCP sockets:
  - each socket identified by its own 4-tuple
- Web servers have different sockets for each connecting client
  - non-persistent HTTP will have different socket for each request

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
- Why use UDP?
  - No connection establishment cost (critical for some applications, e.g., DNS)
  - No connection state
  - Small segment headers (only 8 bytes)
  - Finer application control over data transmission

UDP Segment Structure

- often used for real-time (streaming) apps
  - loss tolerant
  - rate sensitive
- other UDP uses
  - DNS
  - SNMP
- reliable transfer over UDP: add reliability at application layer
  - application-specific error recovery!



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

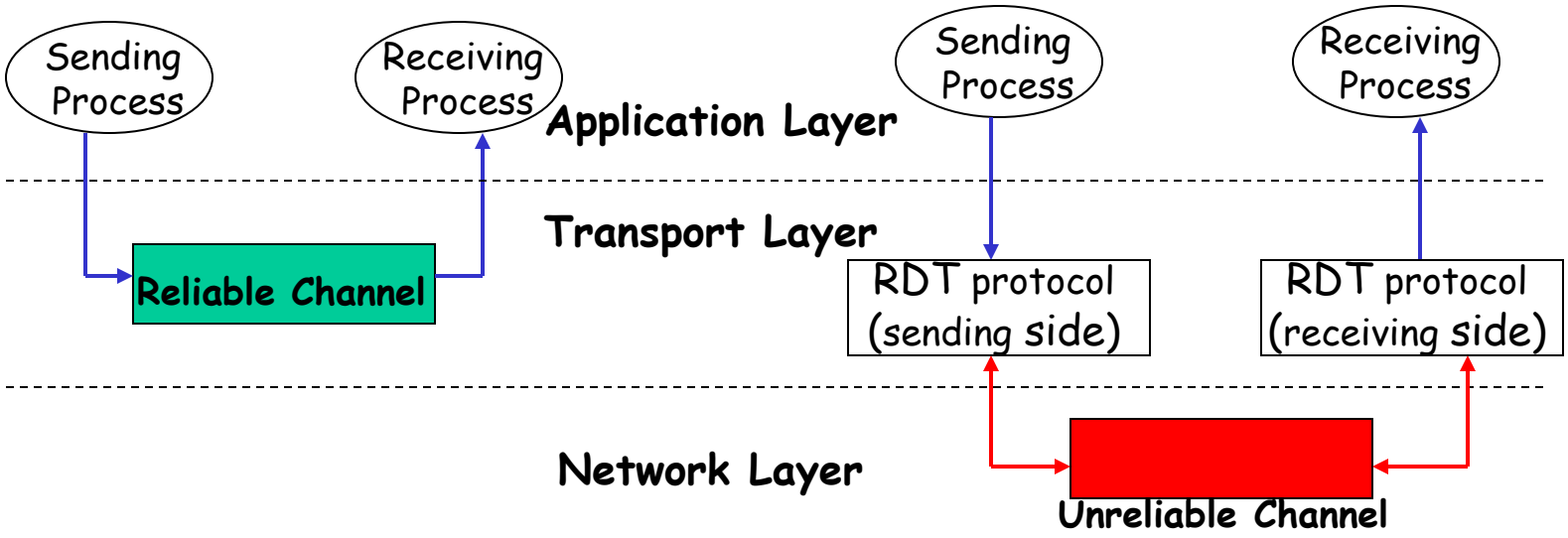
- Note: When adding numbers, a carryout from the most significant bit needs to be added to the result
- Example: add two 16-bit integers
- Weak error protection? Why is it useful?

	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

Reliable Data Transfer

Principles of Reliable Data Transfer

- important in application, transport, and link layers
- top-10 list of important networking topics!

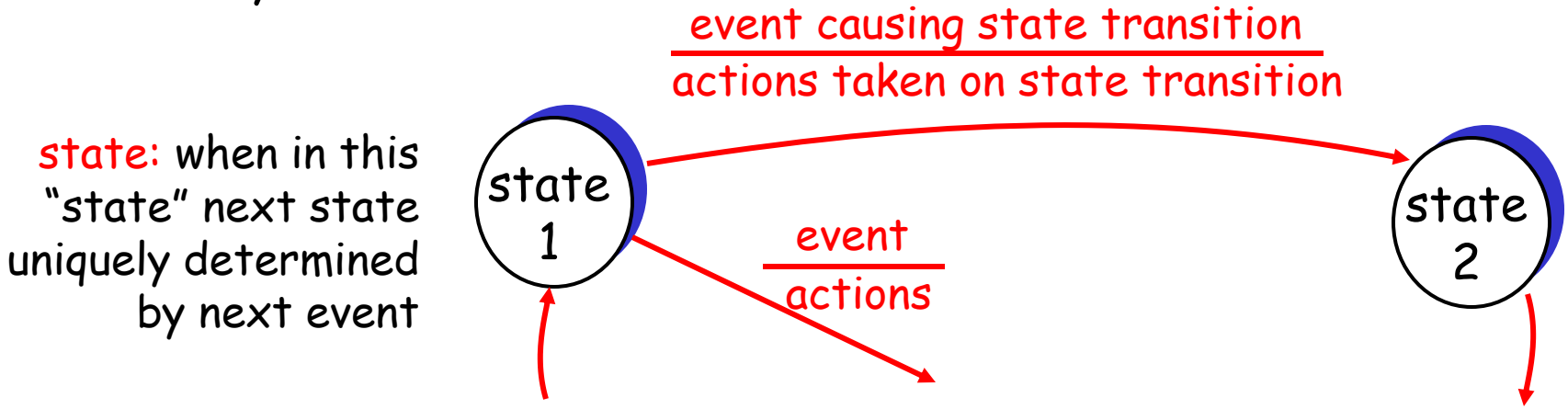


- characteristics of unreliable channel will determine complexity of reliable data transfer protocol (rdt)

Reliable Data Transfer: FSMs

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

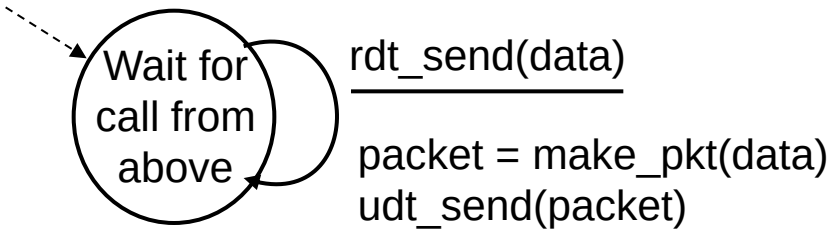


Rdt1.0: Data Transfer over a Perfect Channel

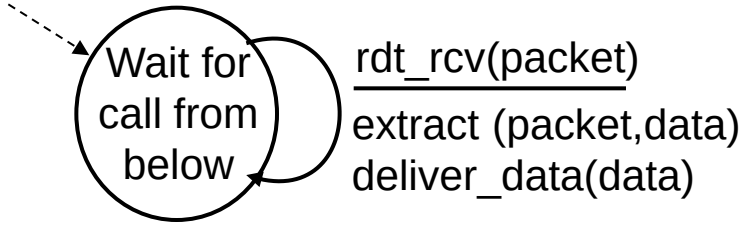
- underlying channel perfectly reliable
  - no bit errors
  - no loss of packets

Rdt1.0: Data Transfer over a Perfect 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

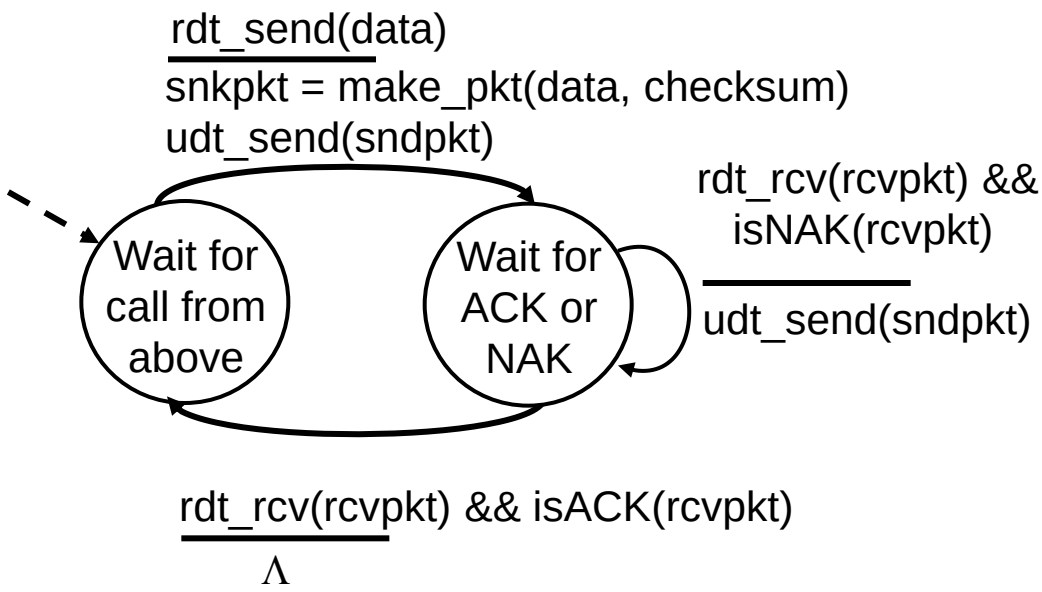
Rdt2.0: channel with bit errors [stop & wait protocol]

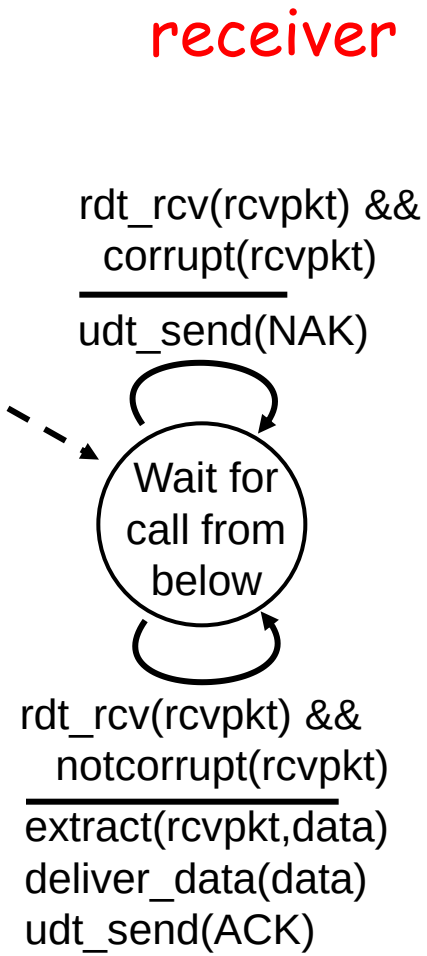
- Assumptions
  - All packets are received
  - Packets may be corrupted (i.e., bits may be flipped)
  - Checksum to detect bit errors

Rdt2.0: channel with bit errors [stop & wait protocol]

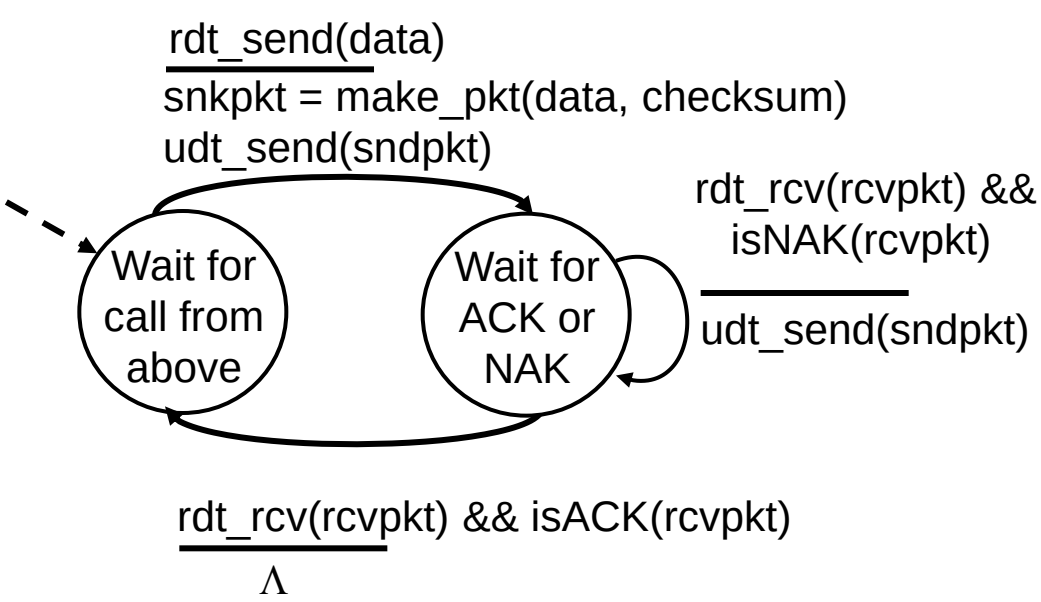
- Assumptions
  - All packets are received
  - Packets may be corrupted (i.e., bits may be flipped)
  - Checksum to detect bit errors
- How to recover from errors? Use ARQ mechanism
  - *acknowledgements (ACKs):* receiver explicitly tells sender that packet received correctly
  - *negative acknowledgements (NAKs):* receiver explicitly tells sender that packet had errors
  - sender retransmits pkt on receipt of NAK
- What about error correcting codes?

rdt2.0: FSM specification





rdt2.0: Observations

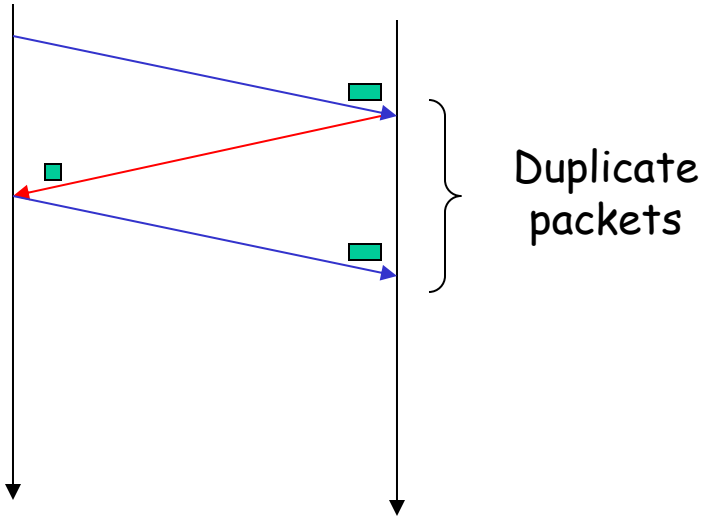


sender

1. A stop-and-Wait protocol

2. What happens when ACK or NAK has bit errors?

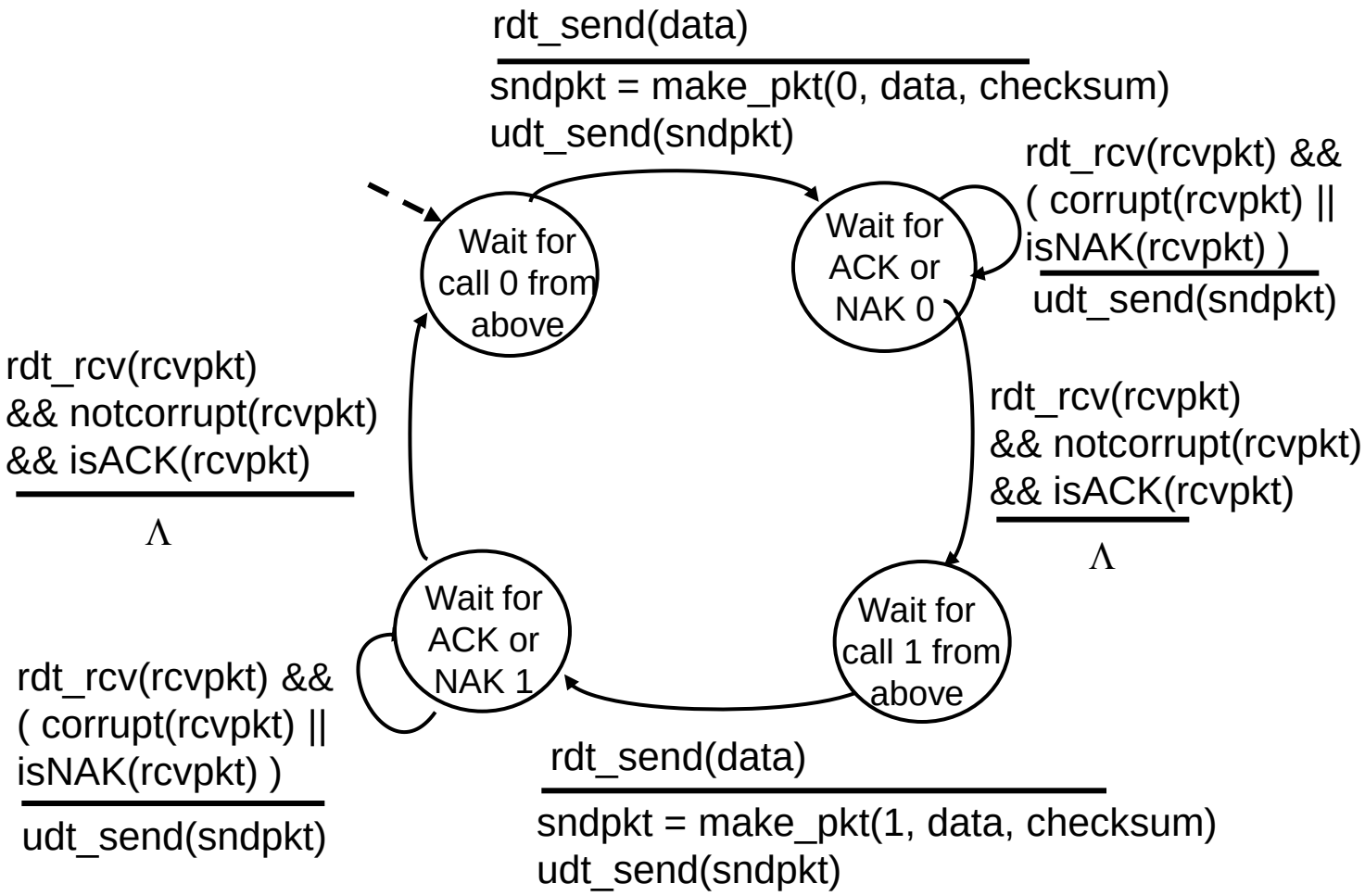
Approach 1: resend the current data packet?



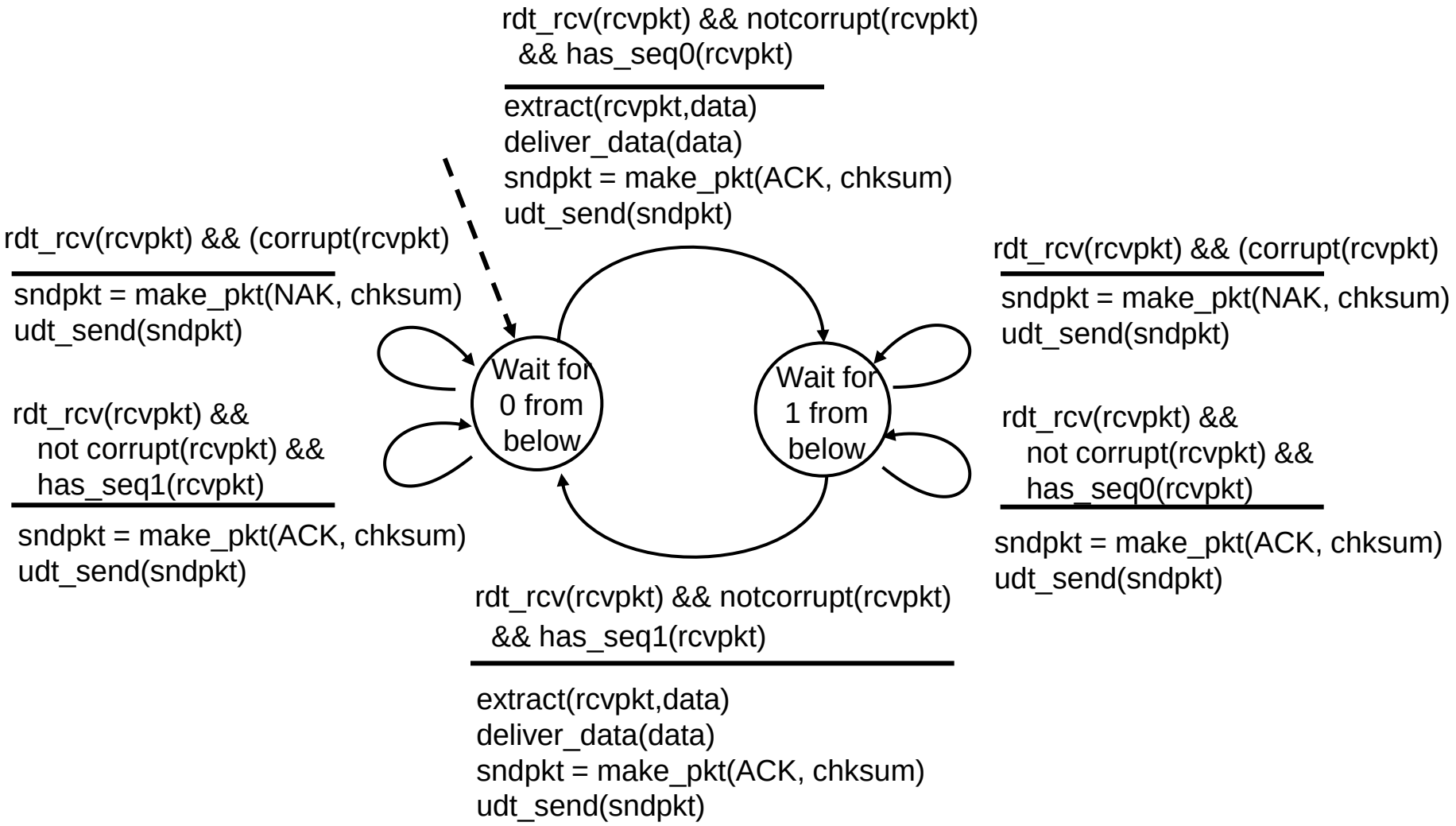
Handling Duplicate Packets

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

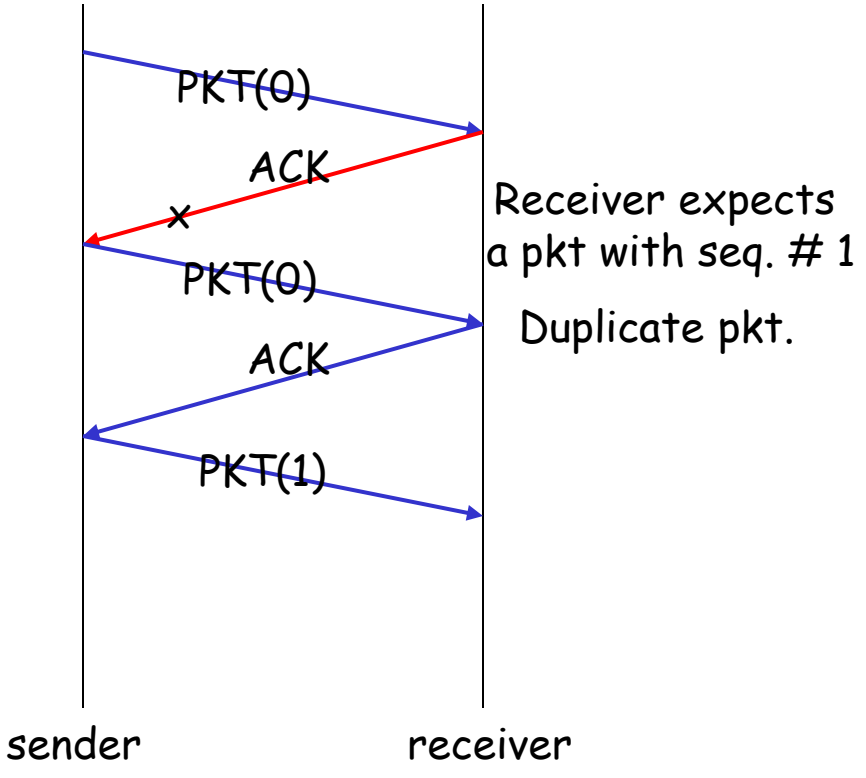
rdt2.1: sender, handles garbled ACK/NAKs

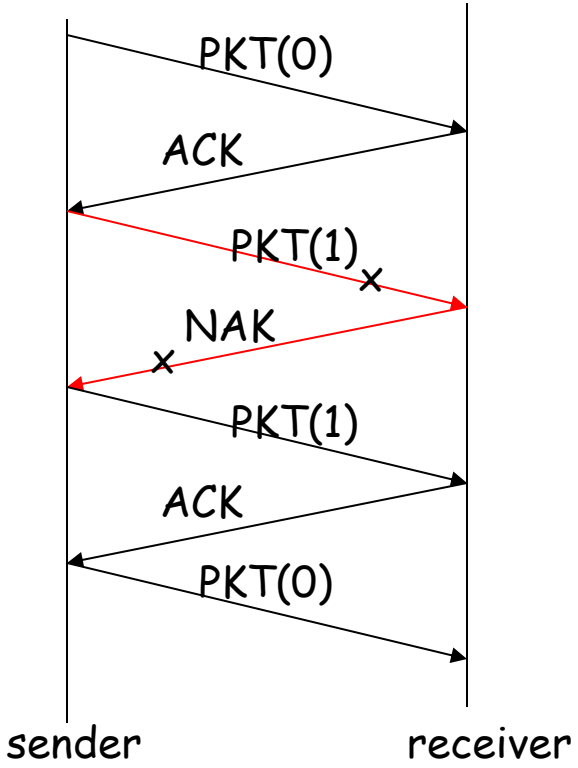


rdt2.1: receiver, handles garbled ACK/NAKs



rtd2.1: examples





rdt2.1: summary

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. #

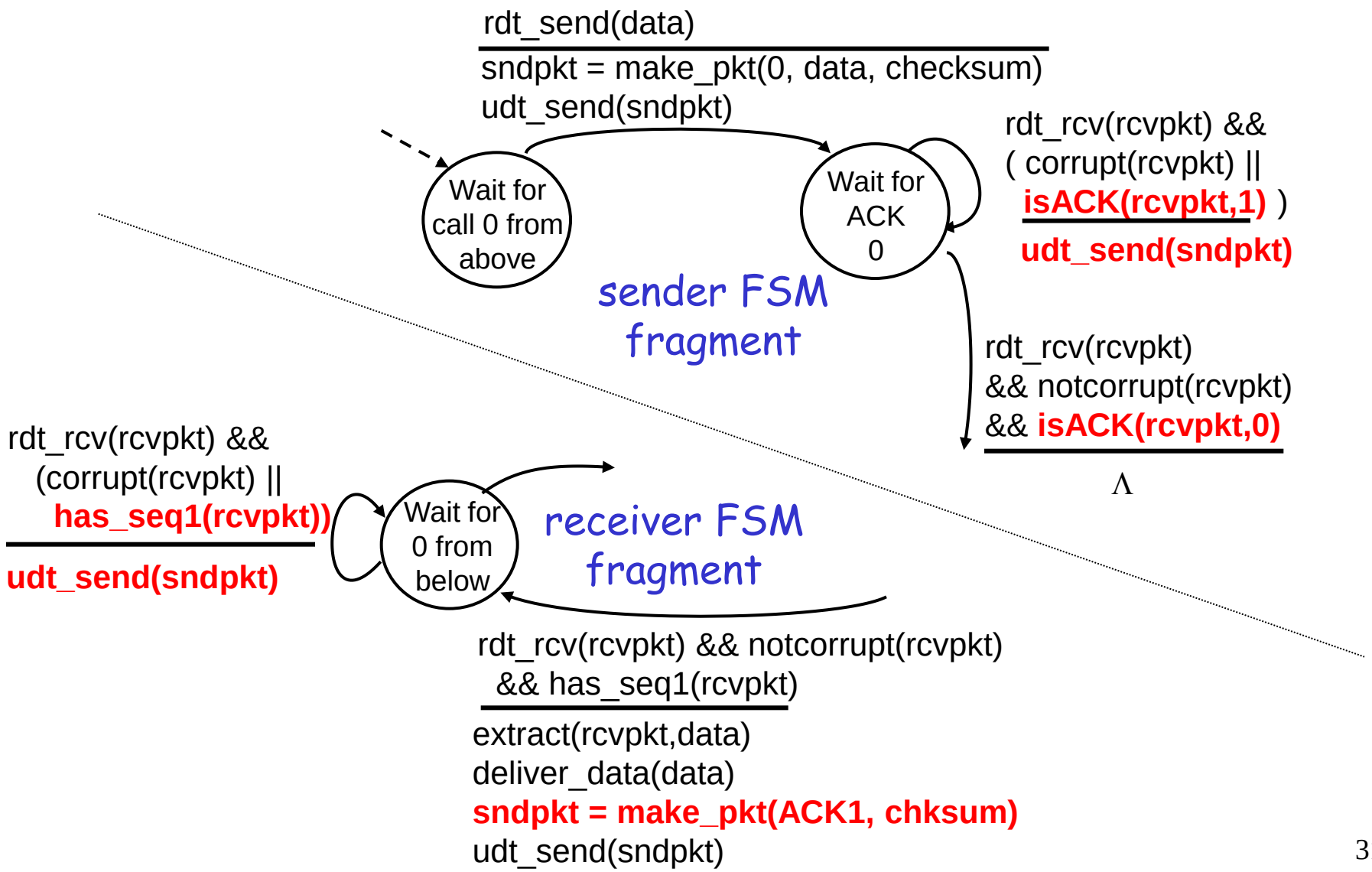
Receiver:

- 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 ACKs 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



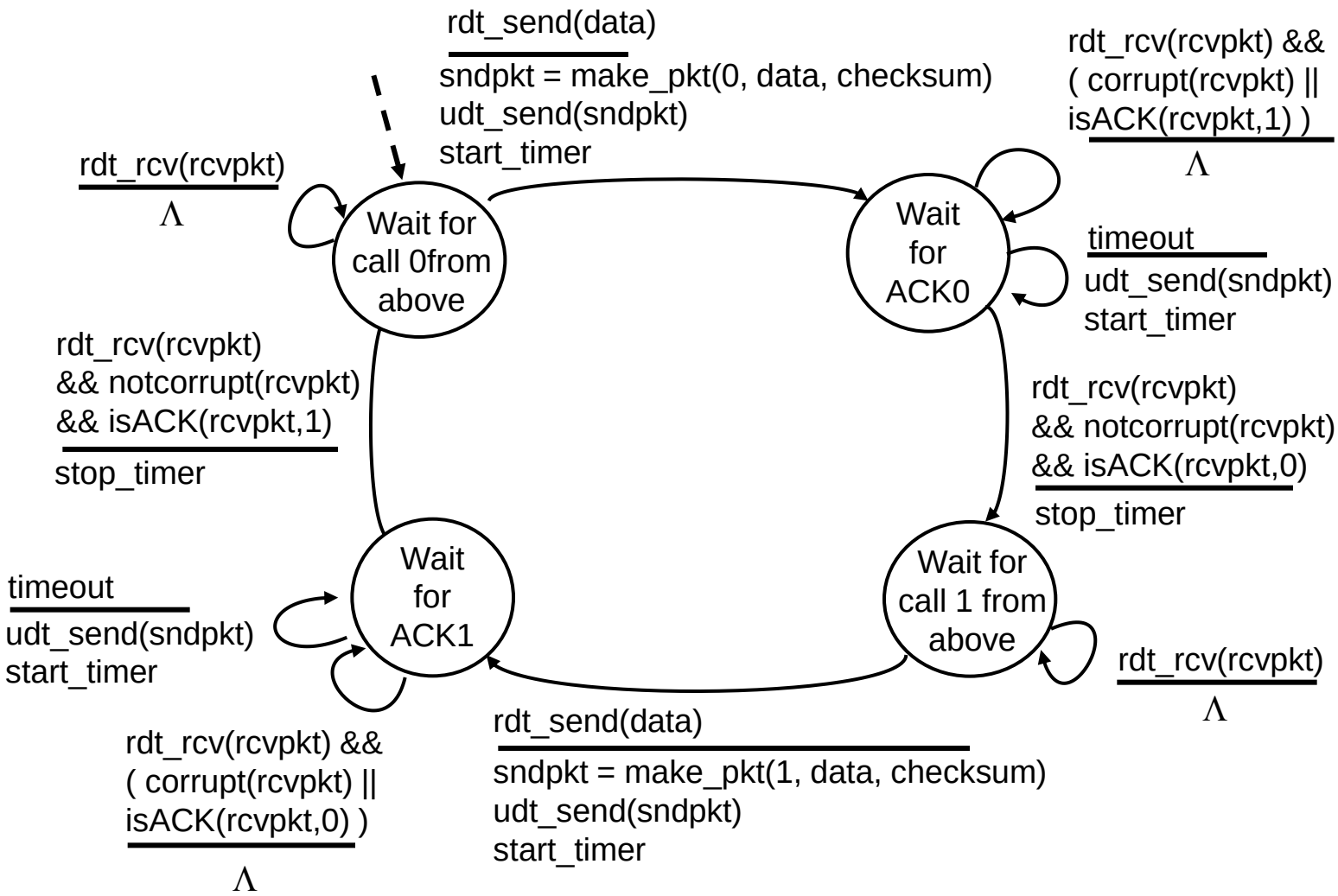
rdt3.0:The case of "Lossy" Channels

- Assumption: underlying channel can also lose packets (data or ACKs)

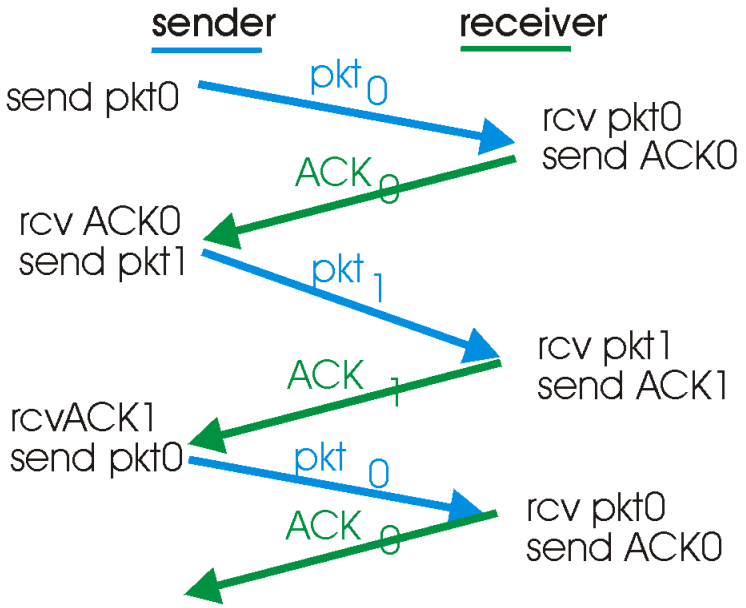
rdt3.0:The case of "Lossy" Channels

- Assumption: underlying channel can also lose packets (data or ACKs)
- Approach: sender waits "reasonable" amount of time for ACK (a Time-Out)
  - Time-out value?
  - Possibility of duplicate packets/ACKs?
- 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

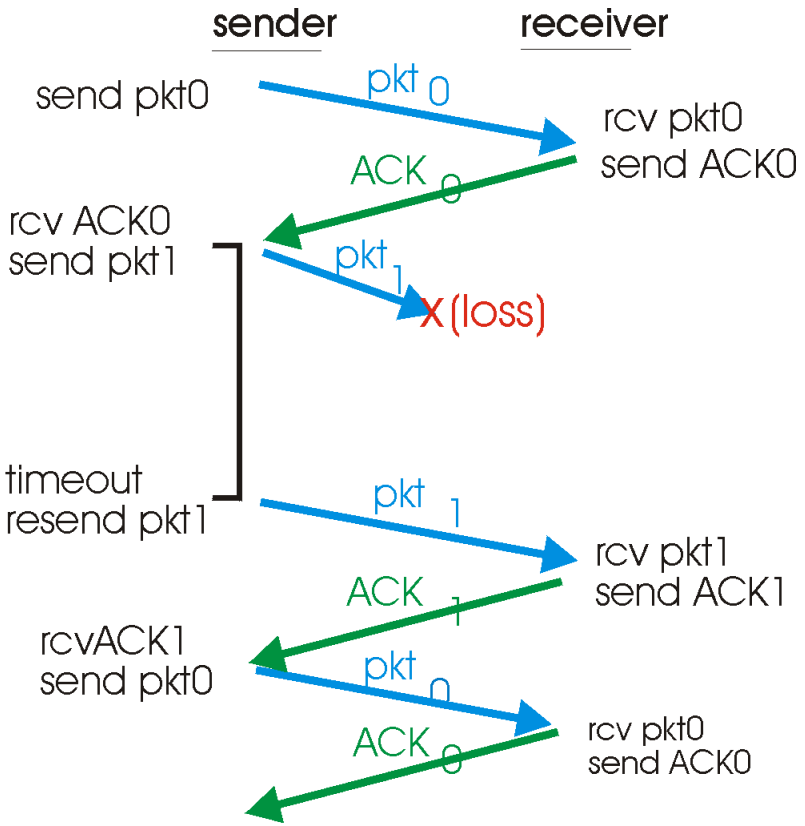
rdt3.0 sender



rdt3.0 in action

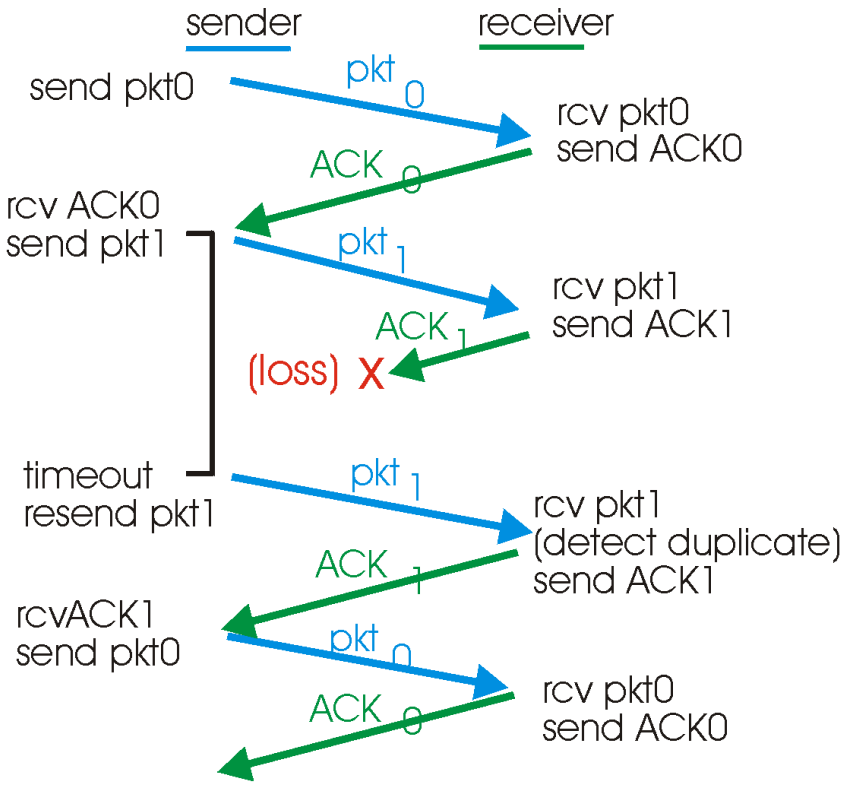


(a) operation with no loss

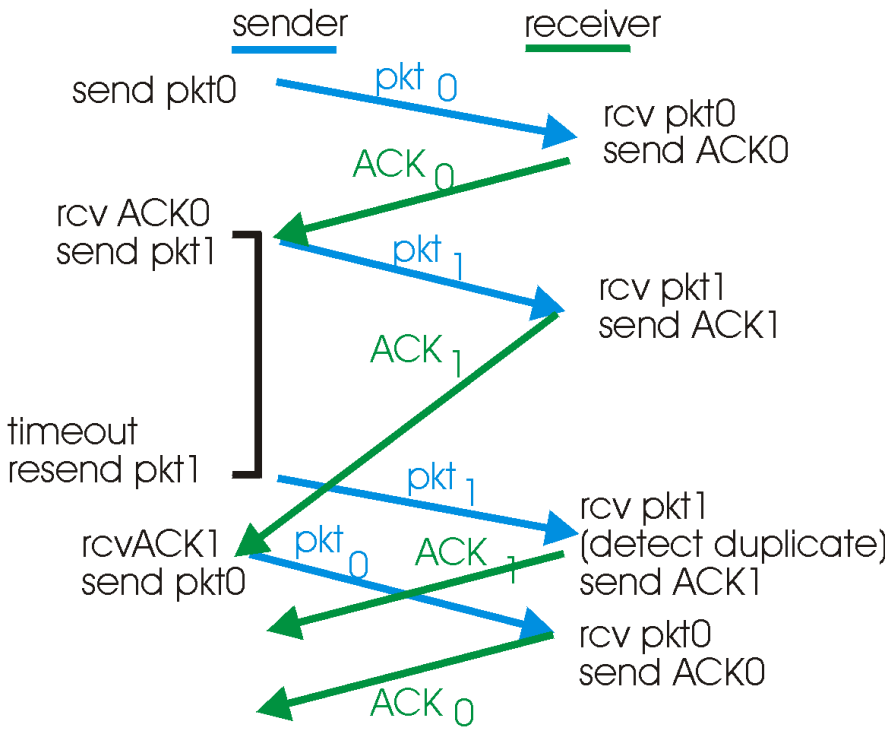


(b) lost packet

rdt3.0 in action

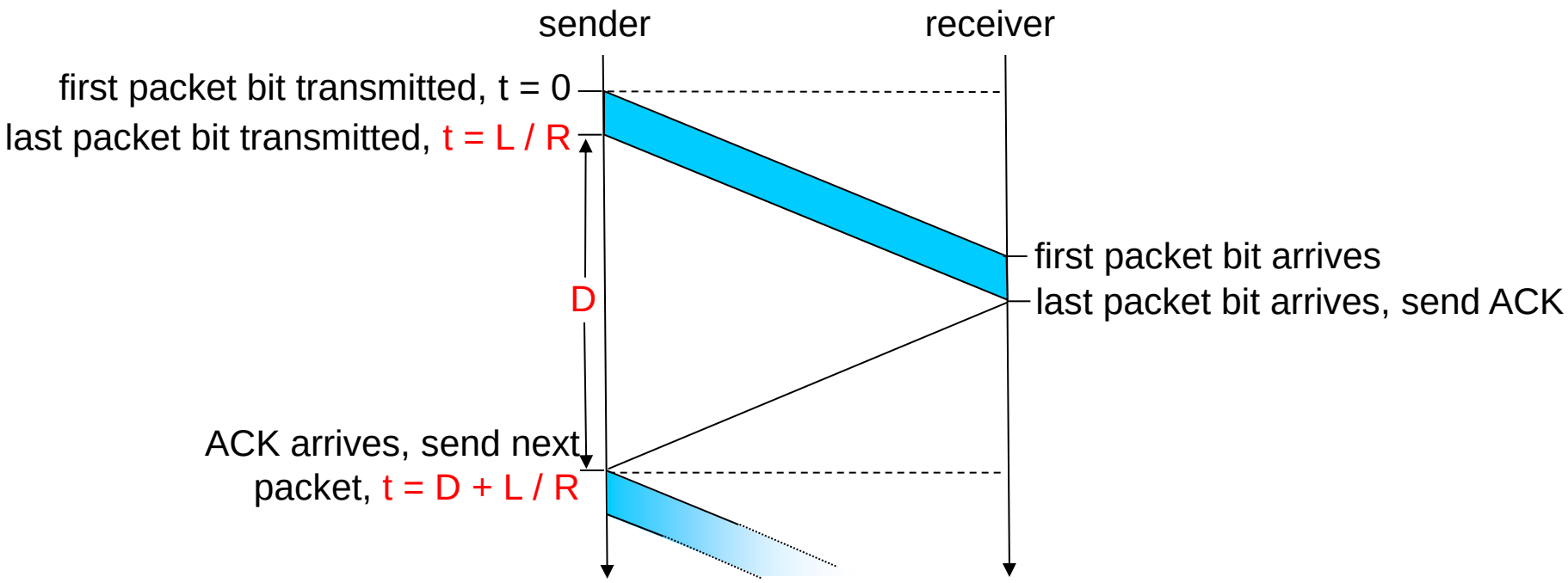


(c) lost ACK



(d) premature timeout

stop-and-wait operation



Pipelining: Motivation

- Stop-and-wait allows the sender to only have a single unACKed packet at any time
- example: 1 Mbps link (R), end-2-end round trip propagation delay (D) of 92ms, 1KB packet (L):

$$T_{transmit} = \frac{L \text{ (packet length in bits)}}{R \text{ (transmission rate, bps)}} = \frac{8kb}{10^3 \text{ kb/sec}} = 8 \text{ ms}$$

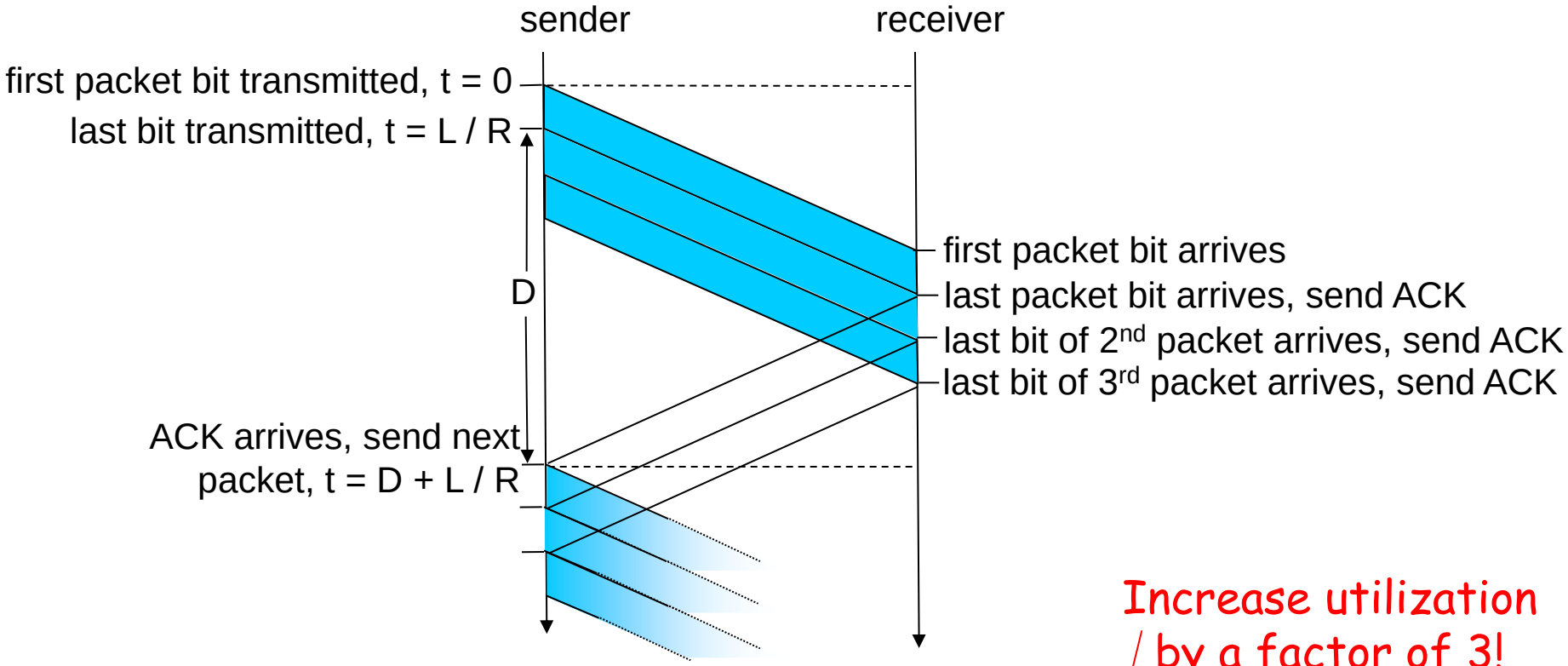
$$U_{sender} = \frac{L / R}{D + L / R} = \frac{8ms}{100ms} = 0.08$$

  - 1KB pkt every 100 ms -> 80Kbps throughput on a 1 Mbps link
  - What does bandwidth x delay product tell us?

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

Pipelining: increased utilization



Increase utilization by a factor of 3!

$$U_{sender} = \frac{3 * L / R}{D + L / R} = \frac{24ms}{100ms} = 0.24$$

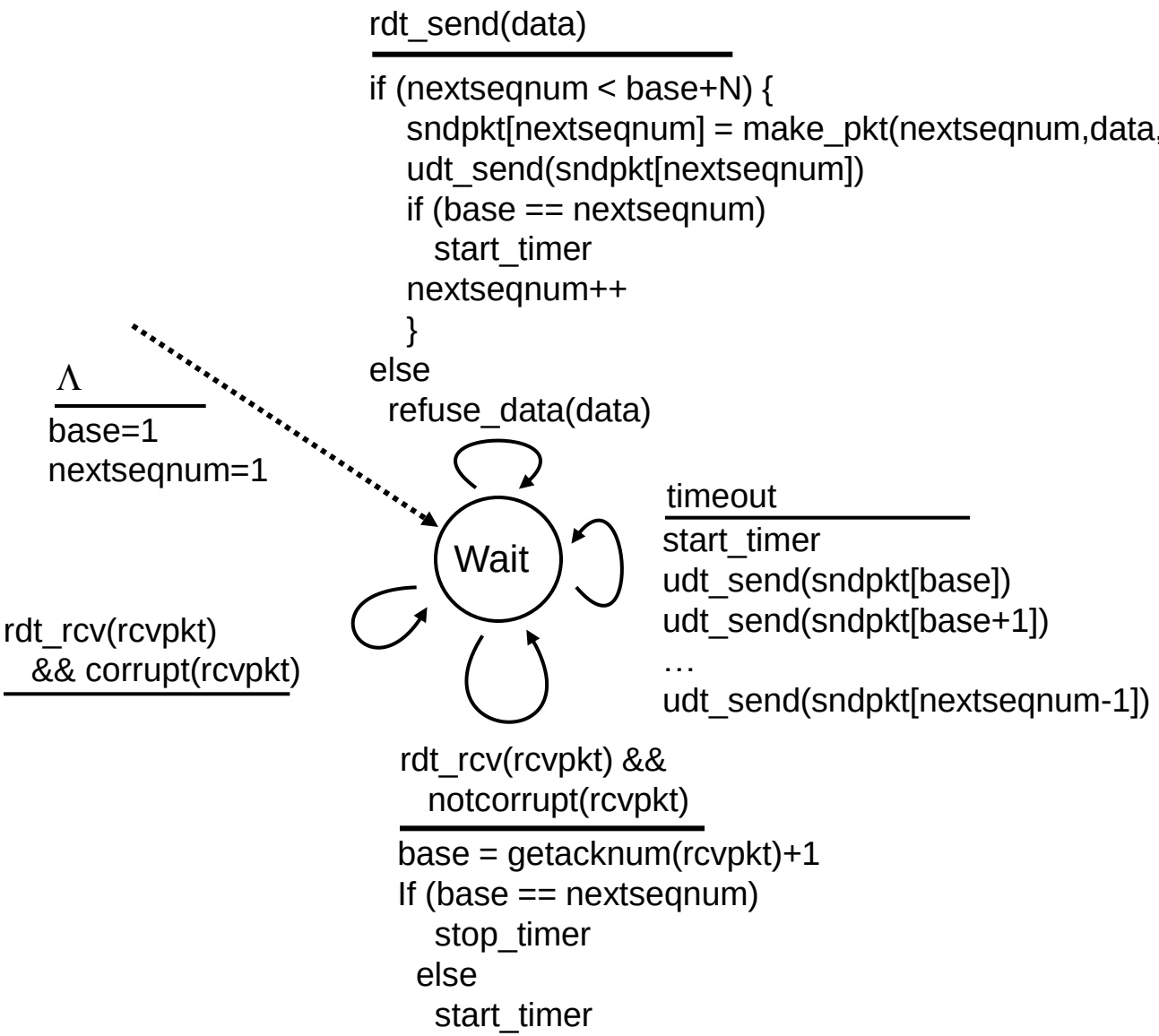
Pipelined protocols

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

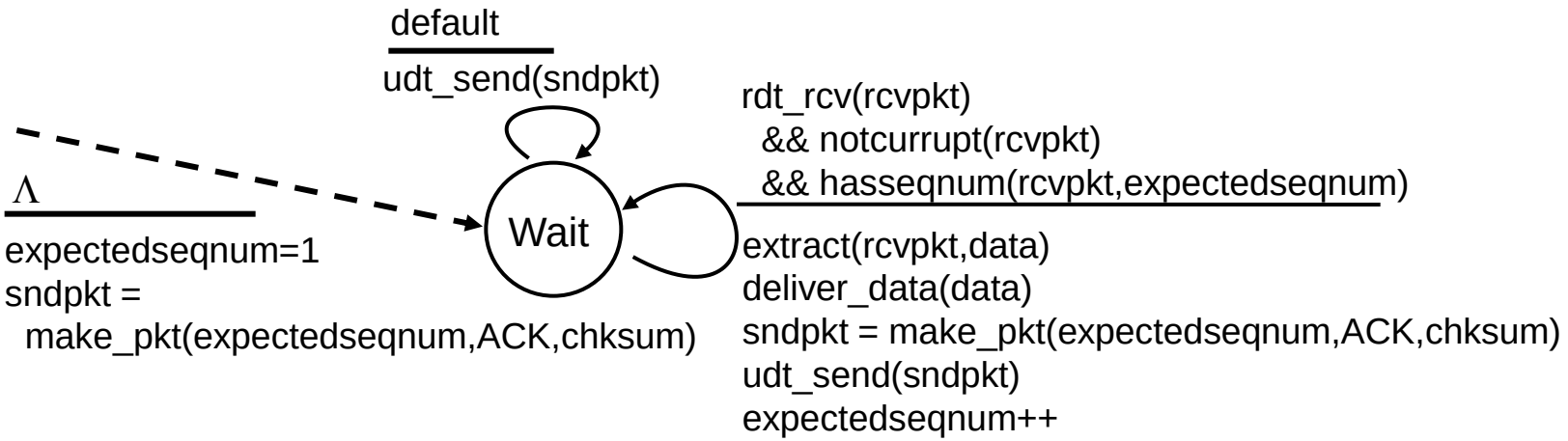
Go-Back-N

- Allow up to N unACKed pkts in the network
  - N is the Window size
- Sender Operation:
  - If window not full, transmit
  - ACKs are cumulative
  - On timeout, send all packets previously sent but not yet ACKed.
  - Uses a single timer – represents the oldest transmitted, but not yet ACKed pkt

GBN: sender extended FSM



GBN: receiver extended FSM



ACK-only: always send ACK for correctly-received pkt with highest *in-order* seq #

- may generate duplicate ACKs
- need only remember **`expectedseqnum`**

- out-of-order pkt:
  - discard (don't buffer) -> no receiver buffering!
  - Re-ACK pkt with highest in-order seq #

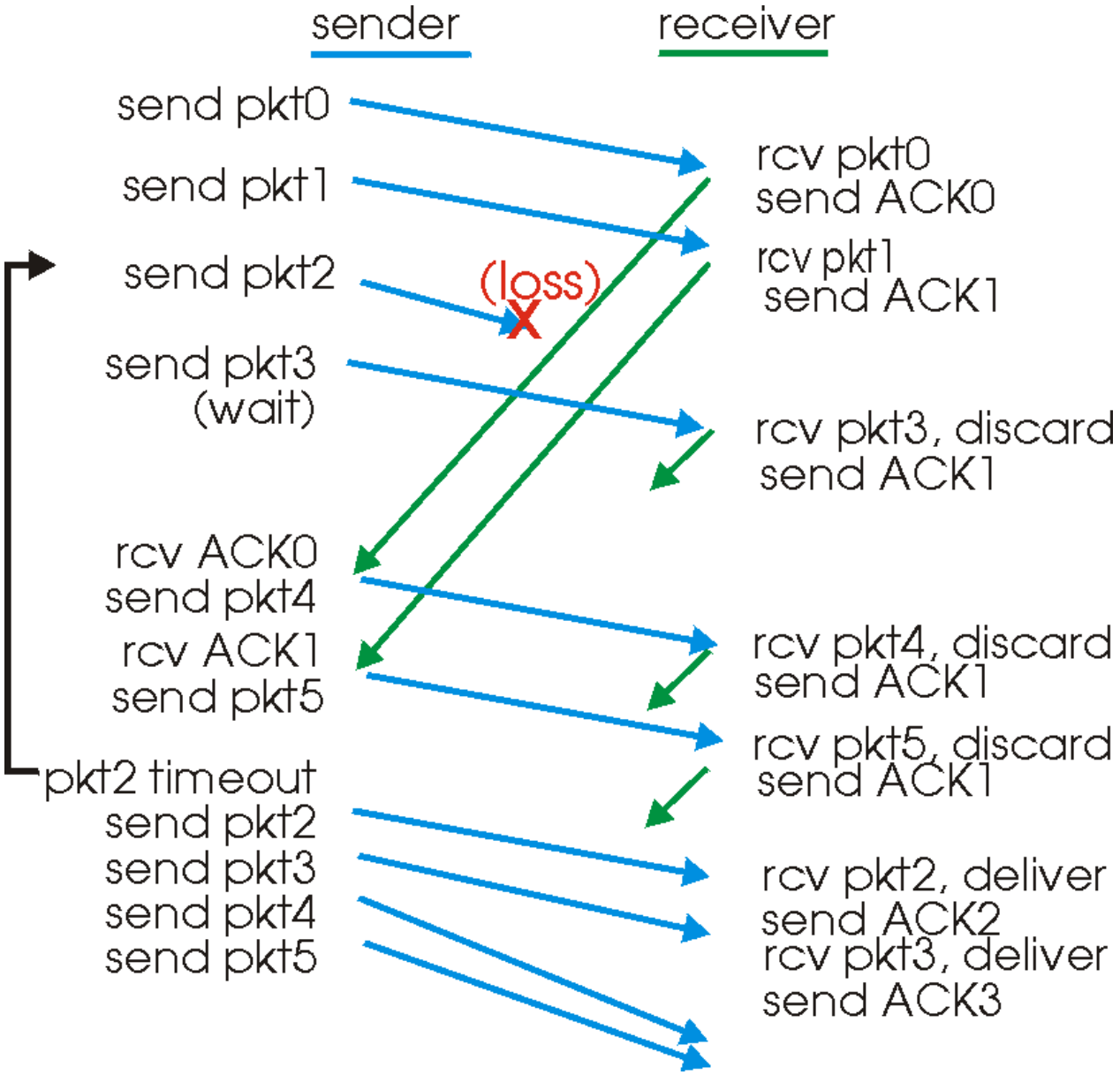
GBN: ...

ACK-only: always send ACK for correctly-received pkt with highest *in-order* seq #

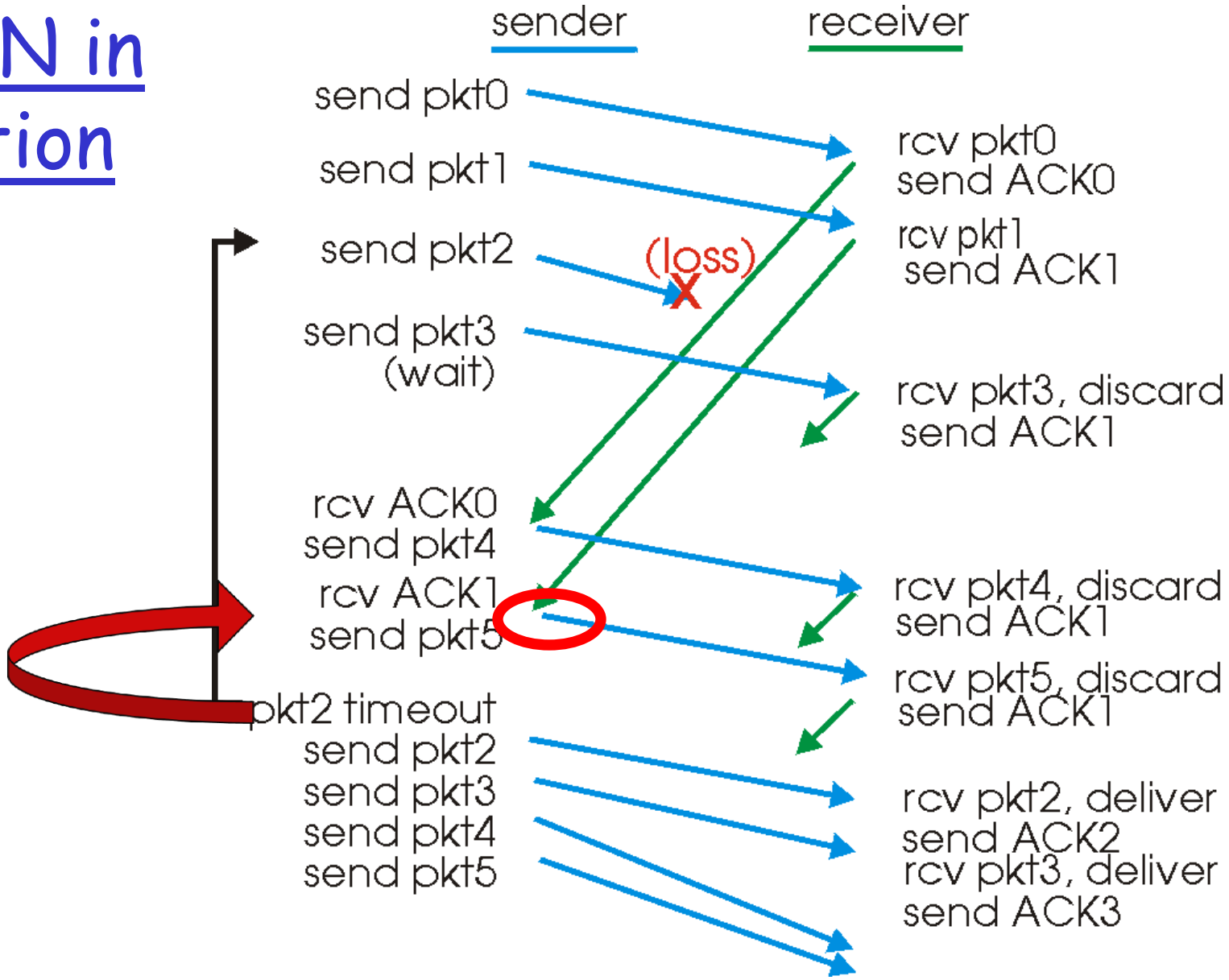
- may generate duplicate ACKs
- need only remember **`expectedseqnum`**

- out-of-order pkt:
  - discard (don't buffer) -> no receiver buffering!
  - Re-ACK pkt with highest in-order seq #

GBN in action



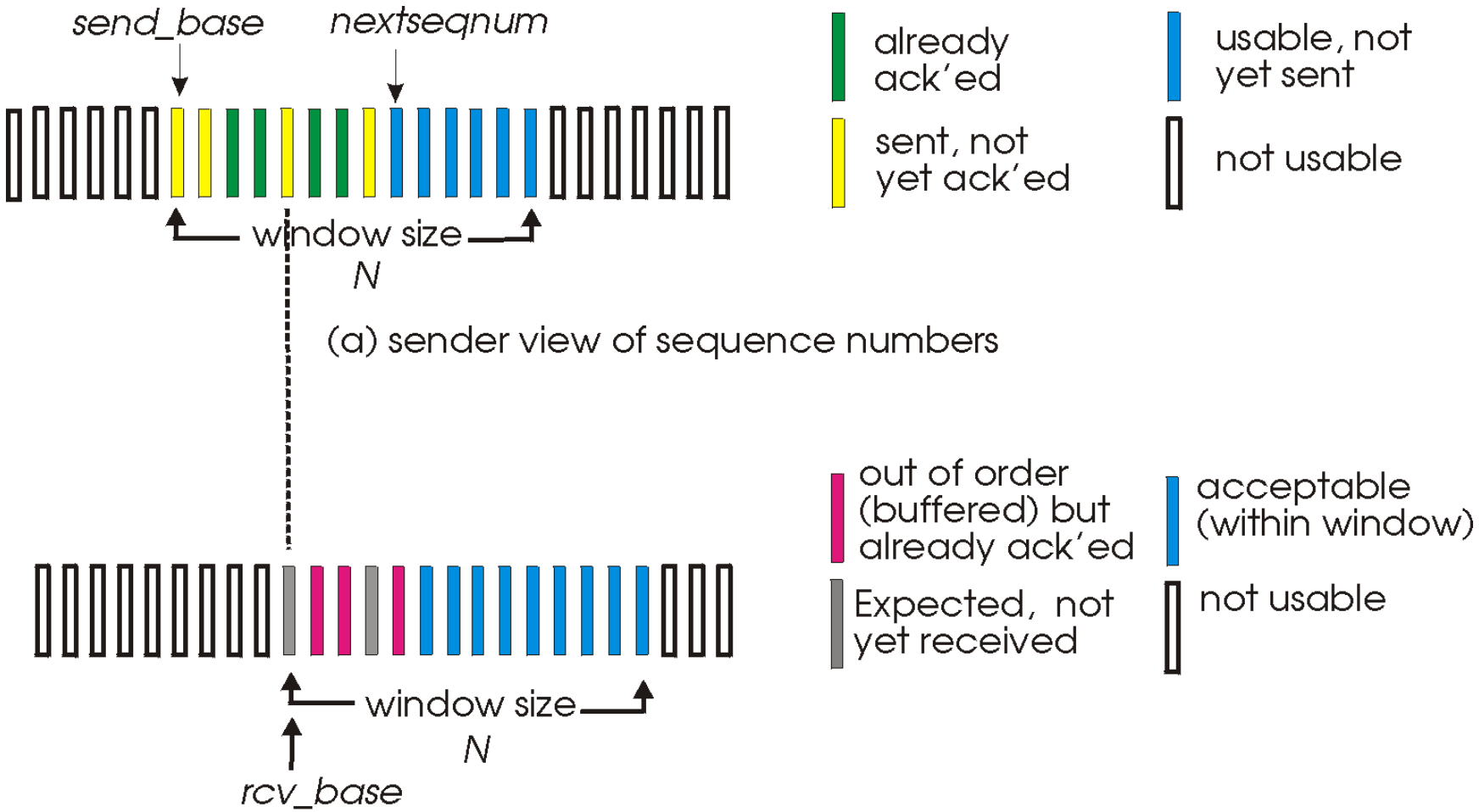
GBN in action



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



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 #

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

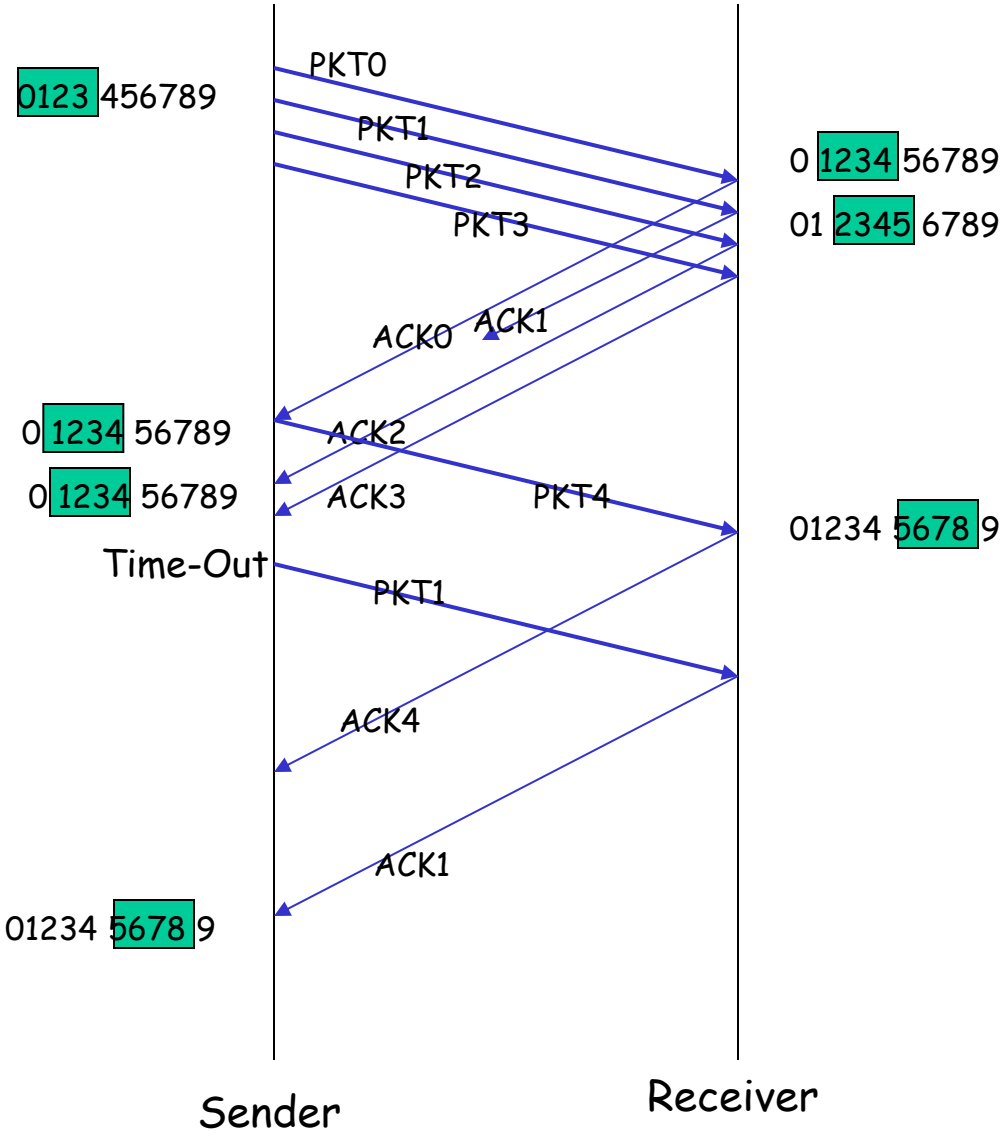
pkt n in [rcvbase-N,rcvbase-1]

- ACK(n)

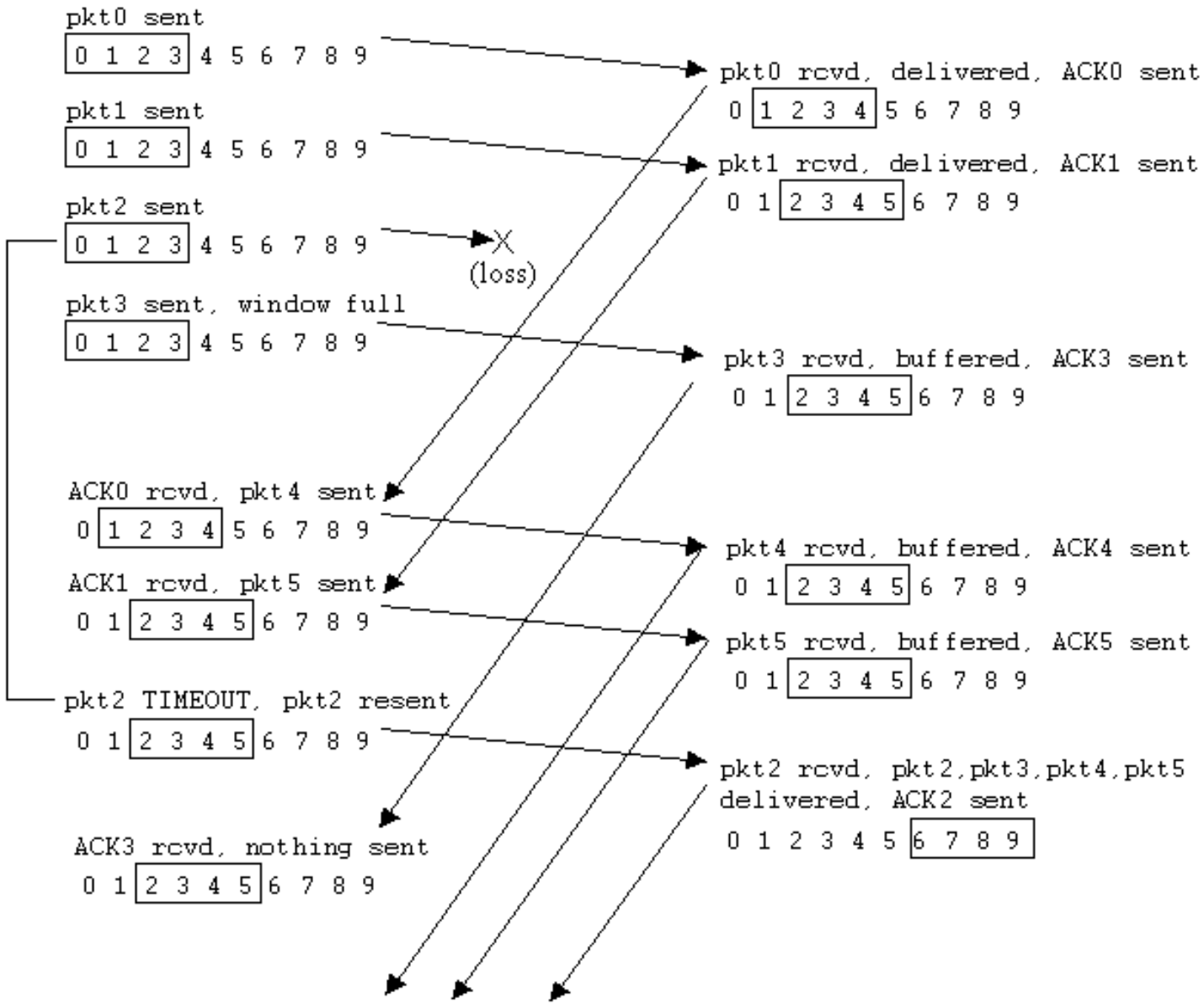
otherwise:

- ignore

Selective Repeat Example



Another Example



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?