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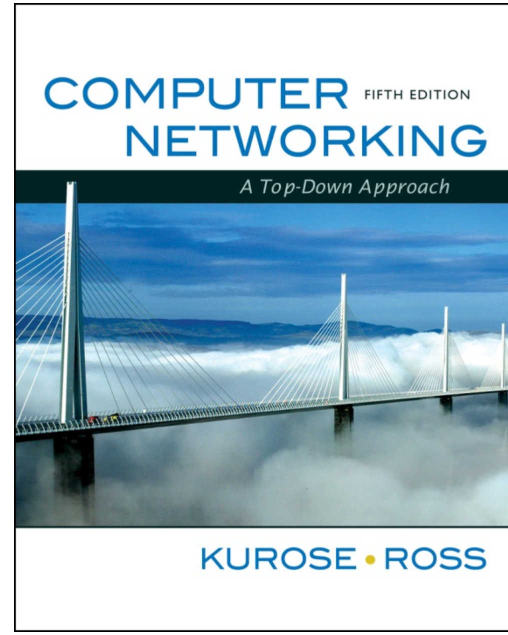
Transport Layer

TDTS04: Computer Networks & Distributed Systems
Instructor: Niklas Carlsson

Chapter 3
Transport Layer



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Computer Networking: A Top Down Approach
5th edition.
Jim Kurose, Keith Ross
Addison-Wesley, April 2009.

Transport Layer: Roadmap

3.1 Transport-layer services

3.2 Multiplexing and demultiplexing

3.3 Connectionless transport: UDP

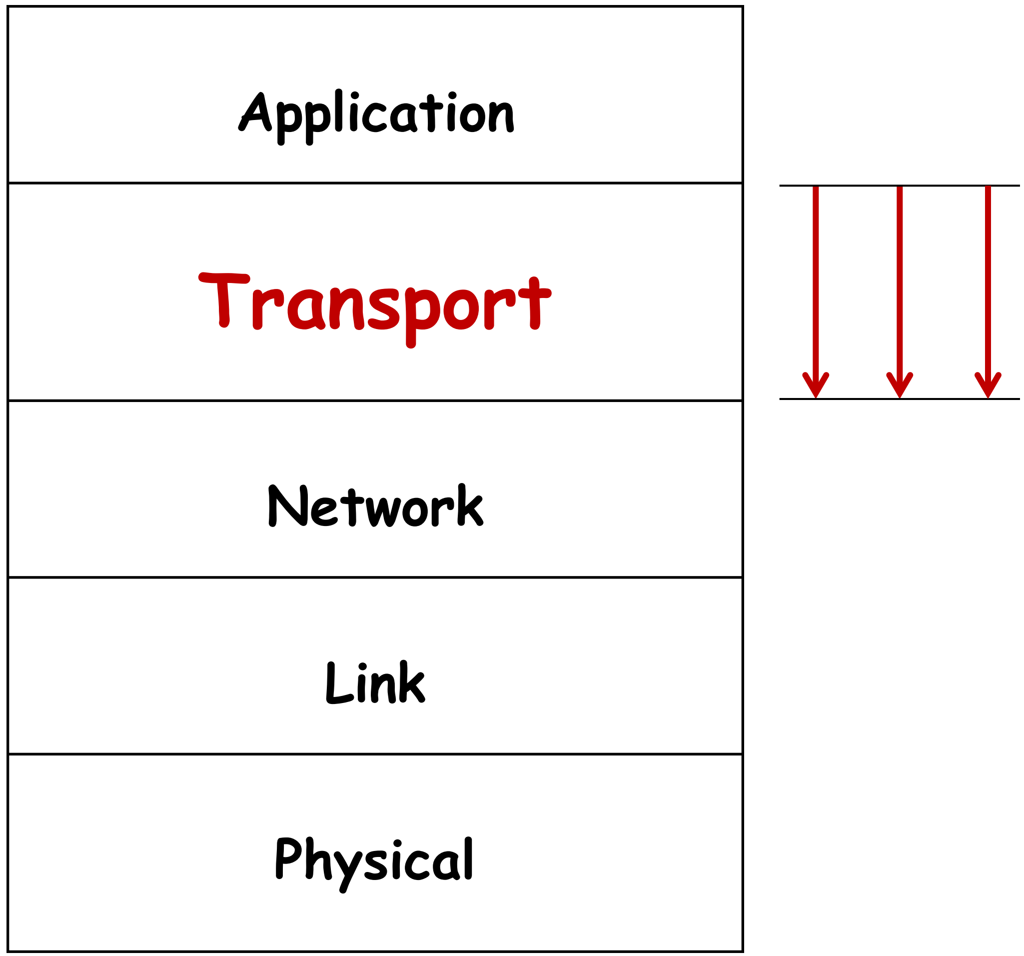
3.4 Principles of reliable data transfer

3.5 Connection-oriented transport: TCP
- segment structure
- reliable data transfer
- flow control
- connection management

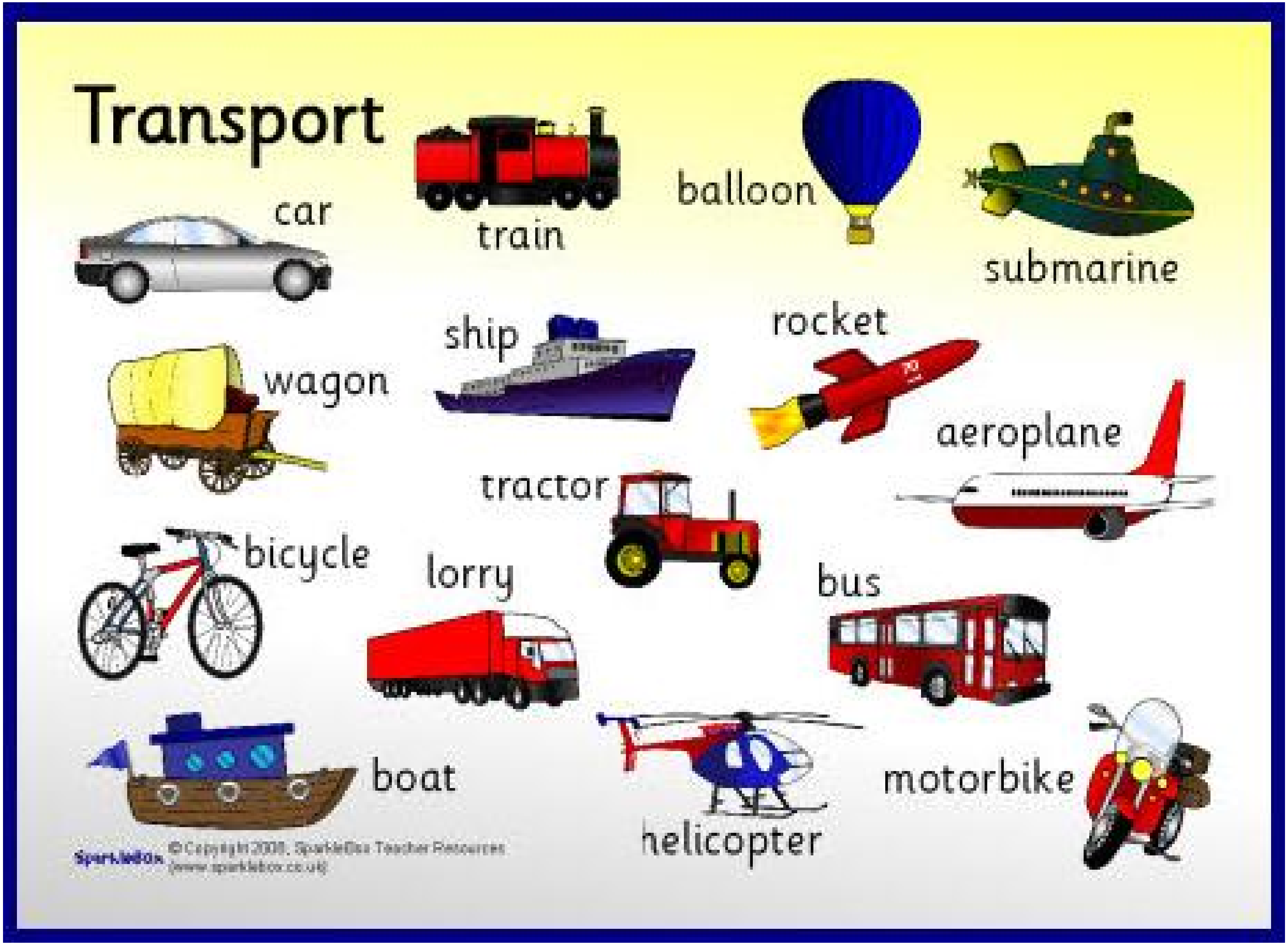
3.6 Principles of congestion control

3.7 TCP congestion control

TCP/IP protocol stack



Transport?


Transport
car
train
balloon
submarine
ship
rocket
wagon
aeroplane
tractor
bicycle
lorry
bus
boat
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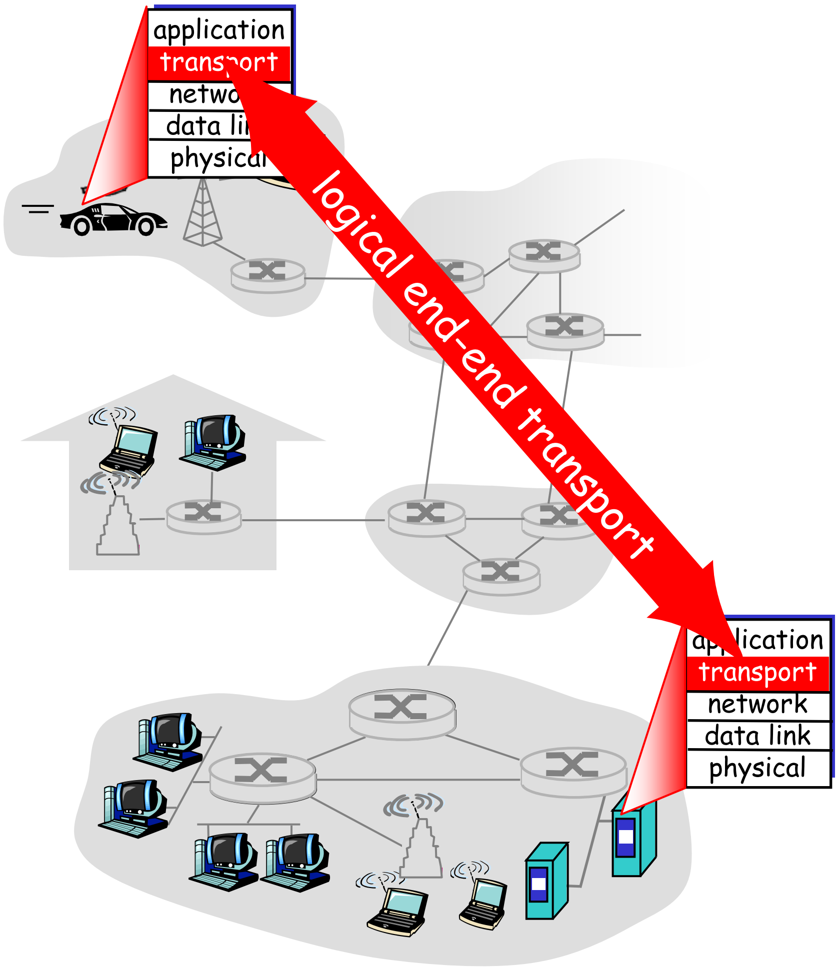
Transport services

- 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



Internet transport-layer protocols

- unreliable, unordered delivery: (UDP)
  - no-frills extension of "best-effort" IP

- reliable, in-order delivery (TCP)
  - congestion control
  - flow control
  - connection setup
  - …

Applications?

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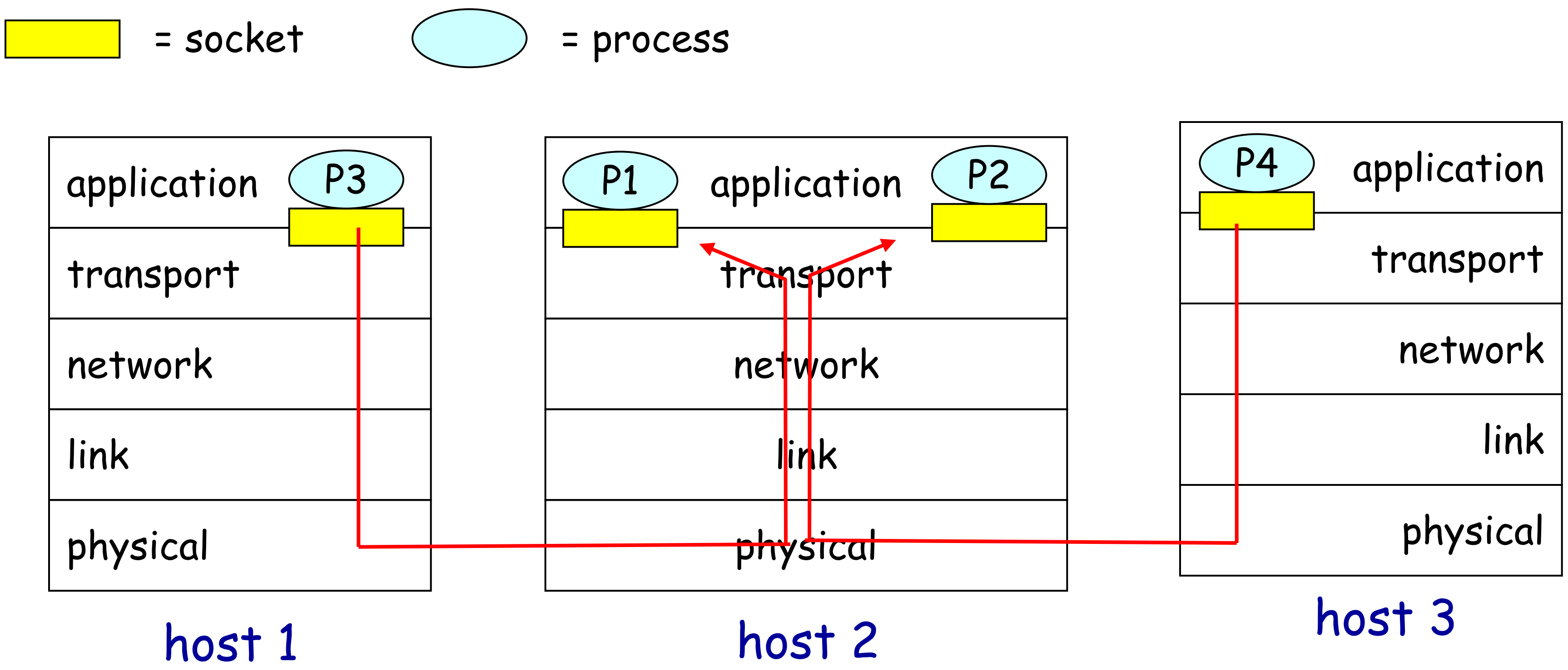
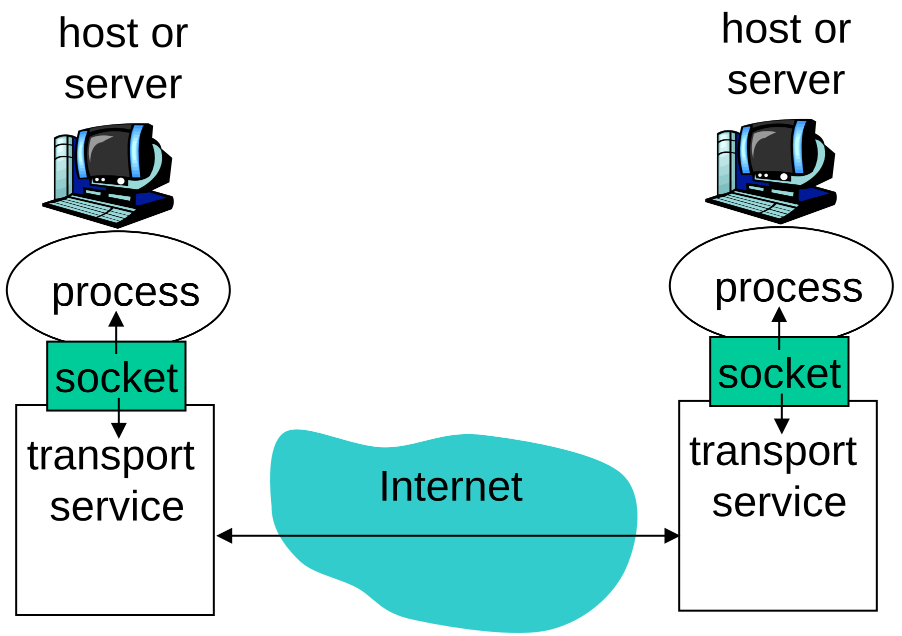
Multiplexing/demultiplexing



Recall: Sockets

- process sends/receives messages to/from its socket
- socket analogous to door
  - provides connection between the application and network



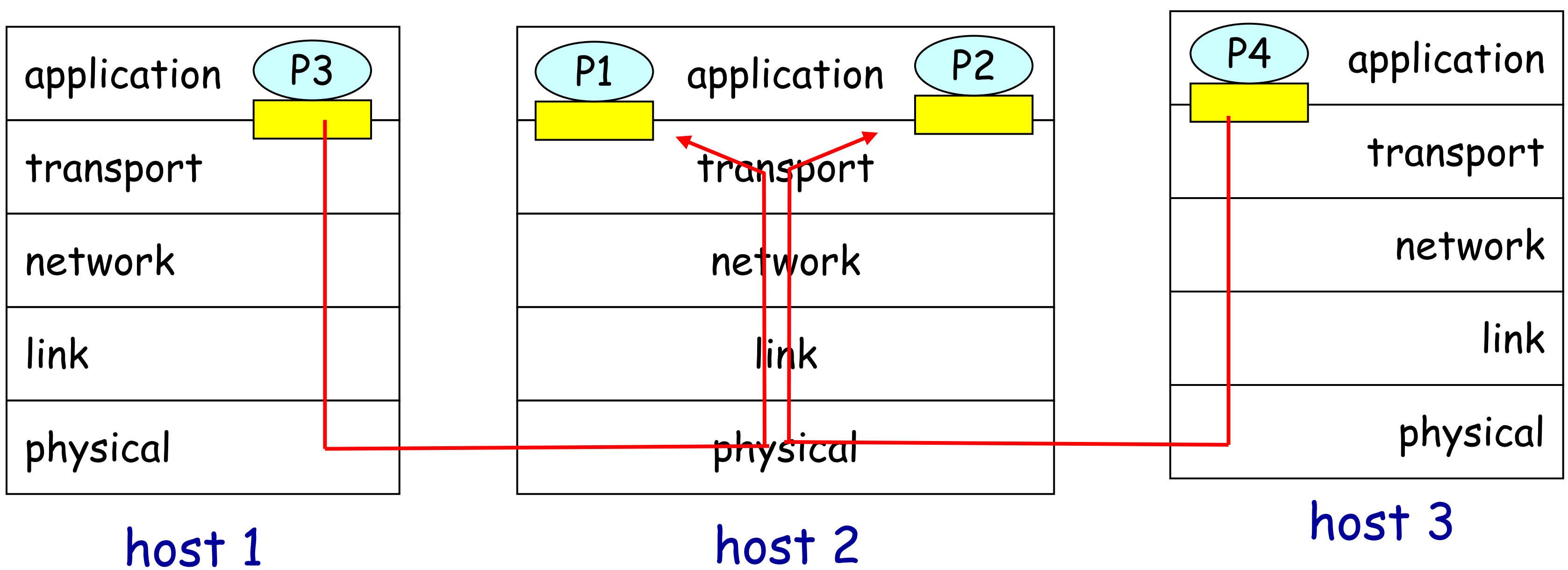
Multiplexing/demultiplexing

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

- each datagram has source IP address, destination IP address
- host uses IP addresses & port numbers to direct segment to appropriate socket

Connectionless demultiplexing

- *UDP* :

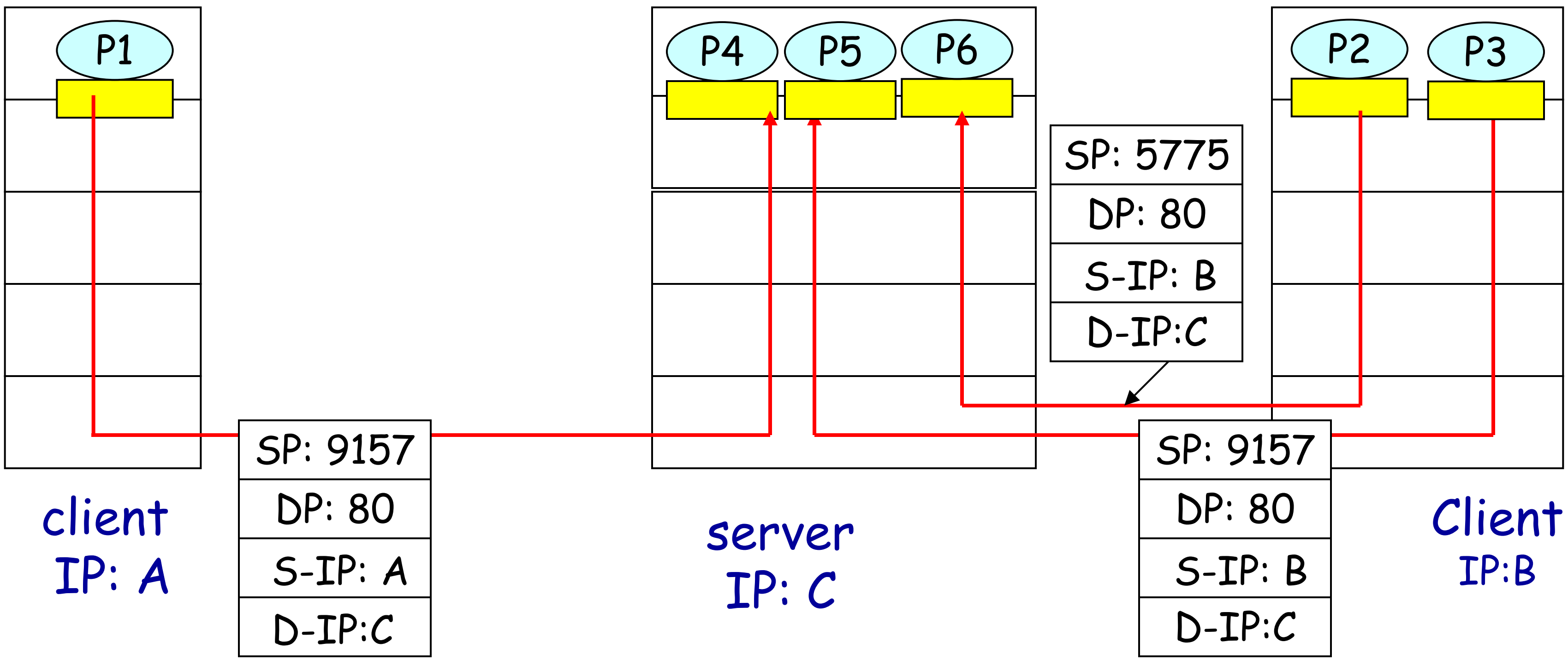
  when creating datagram to send into UDP socket, must specify the 2-tuple

  - dest IP address
  - dest port number

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

Connection-oriented demux (cont)



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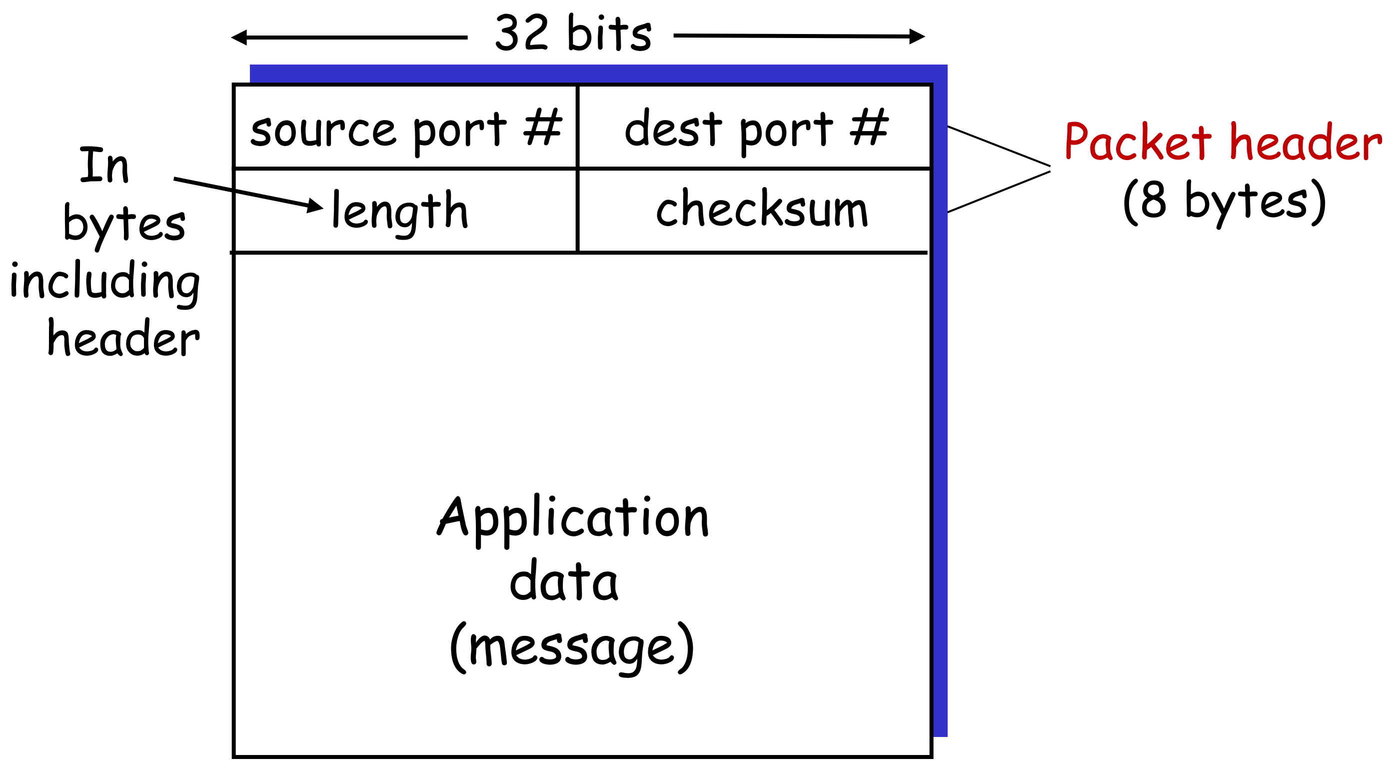
UDP: User Datagram Protocol [RFC 768]

- *connectionless:*
  - no handshaking between UDP sender and receiver

- *only 2 functions*
  - mult/demult
  - error detection (optional)

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.

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

		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

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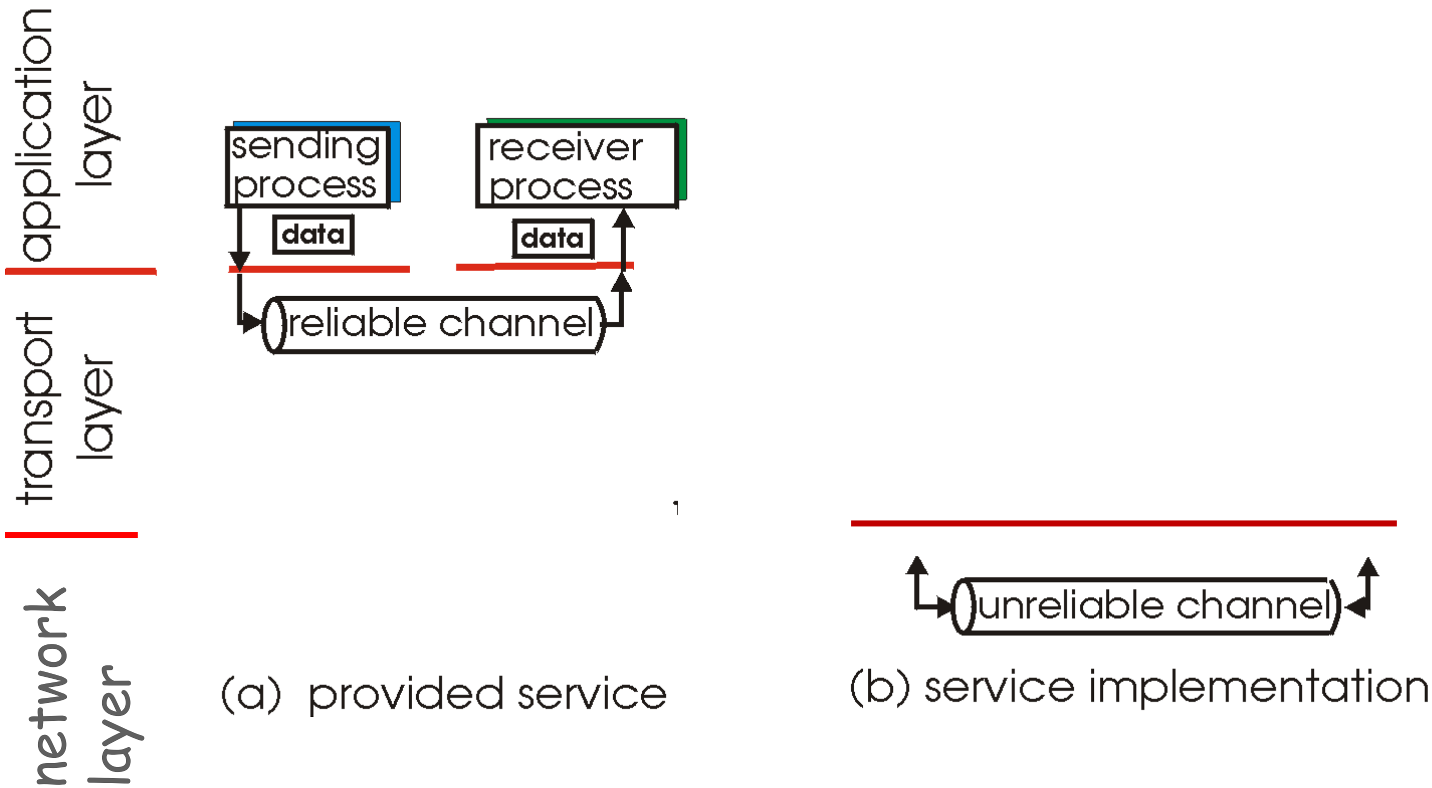
Principles of Reliable data transfer

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

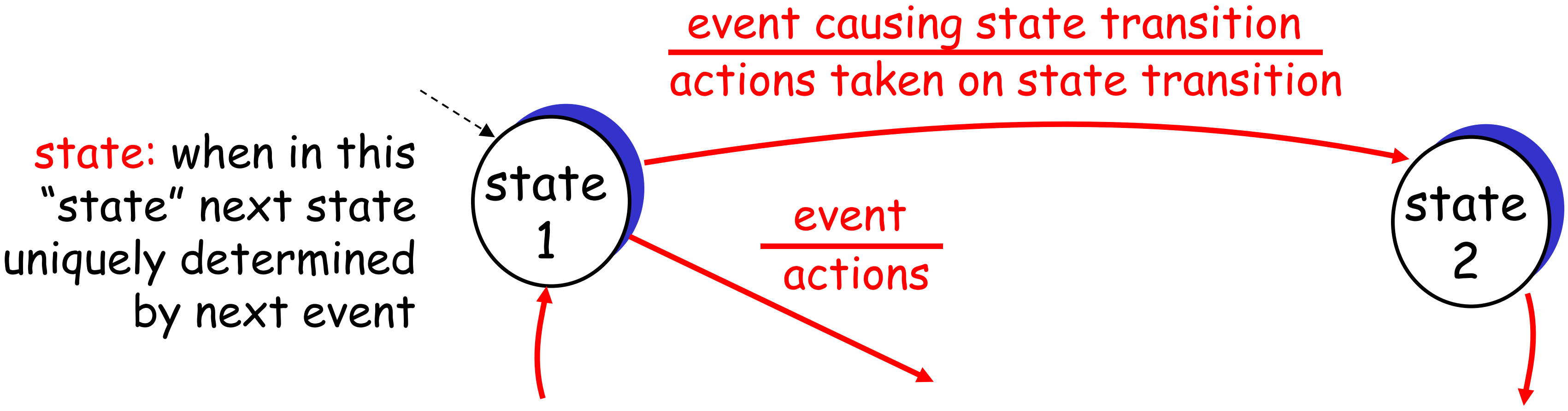


- characteristics of unreliable channel will determine complexity of reliable data transfer protocol (rdt)

Reliable data transfer: getting started

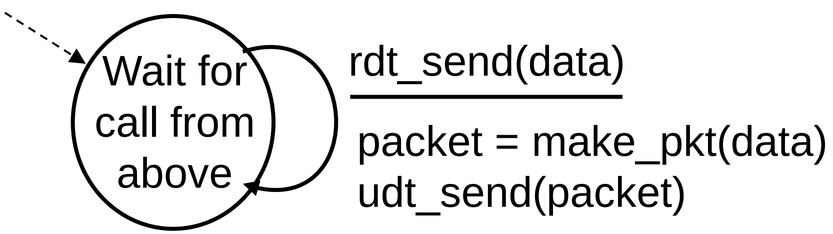
We'll:

- incrementally develop sender, receiver sides of reliable data transfer protocol (rdt)
- use finite state machines (FSM) to specify sender, receiver

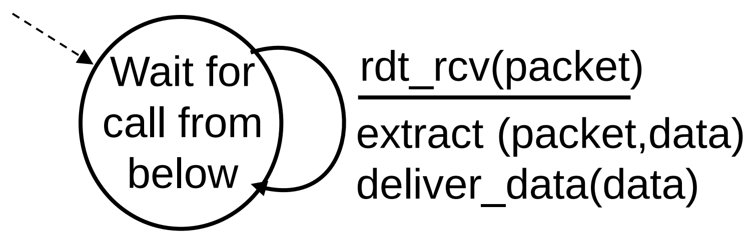


Rdt1.0: reliable transfer over a reliable channel

- underlying channel perfectly reliable
  - no bit errors
  - no loss of packets
- separate FSMs for sender, receiver:



sender



receiver

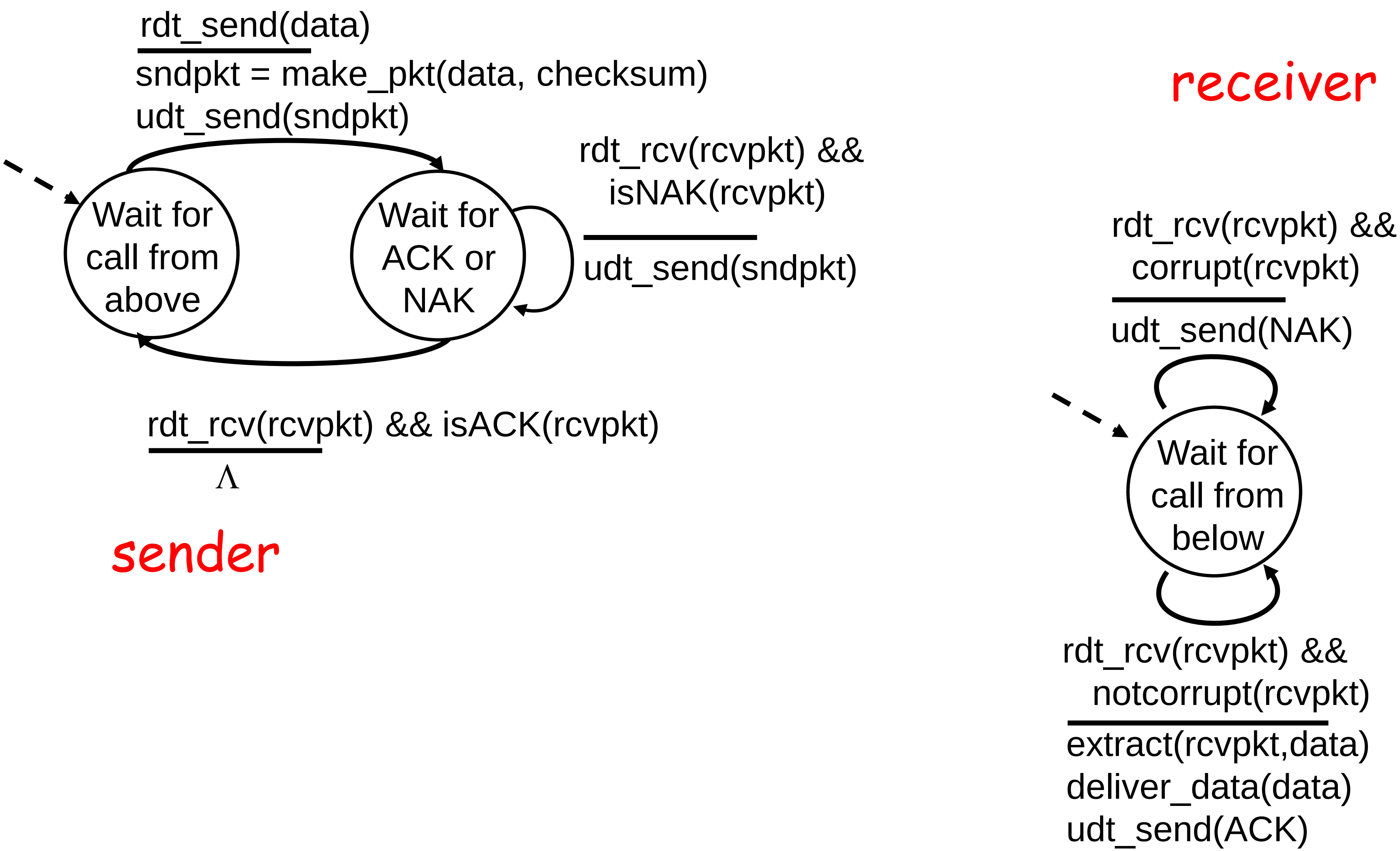
Rdt2.0: channel with bit errors

- underlying channel may flip bits in packet
- use checksum to detect bit errors
- how to recover from errors?

Rdt2.0: channel with bit errors

- Mechanisms for error recovery:
  - *acknowledgements (ACKs):* receiver explicitly tells sender that pkt received OK
  - *negative acknowledgements (NAKs):* receiver explicitly tells sender that pkt had errors
  - *retransmissions* - sender retransmits pkt on receipt of NAK

rdt2.0: FSM specification



rdt2.0 has a fatal flaw!

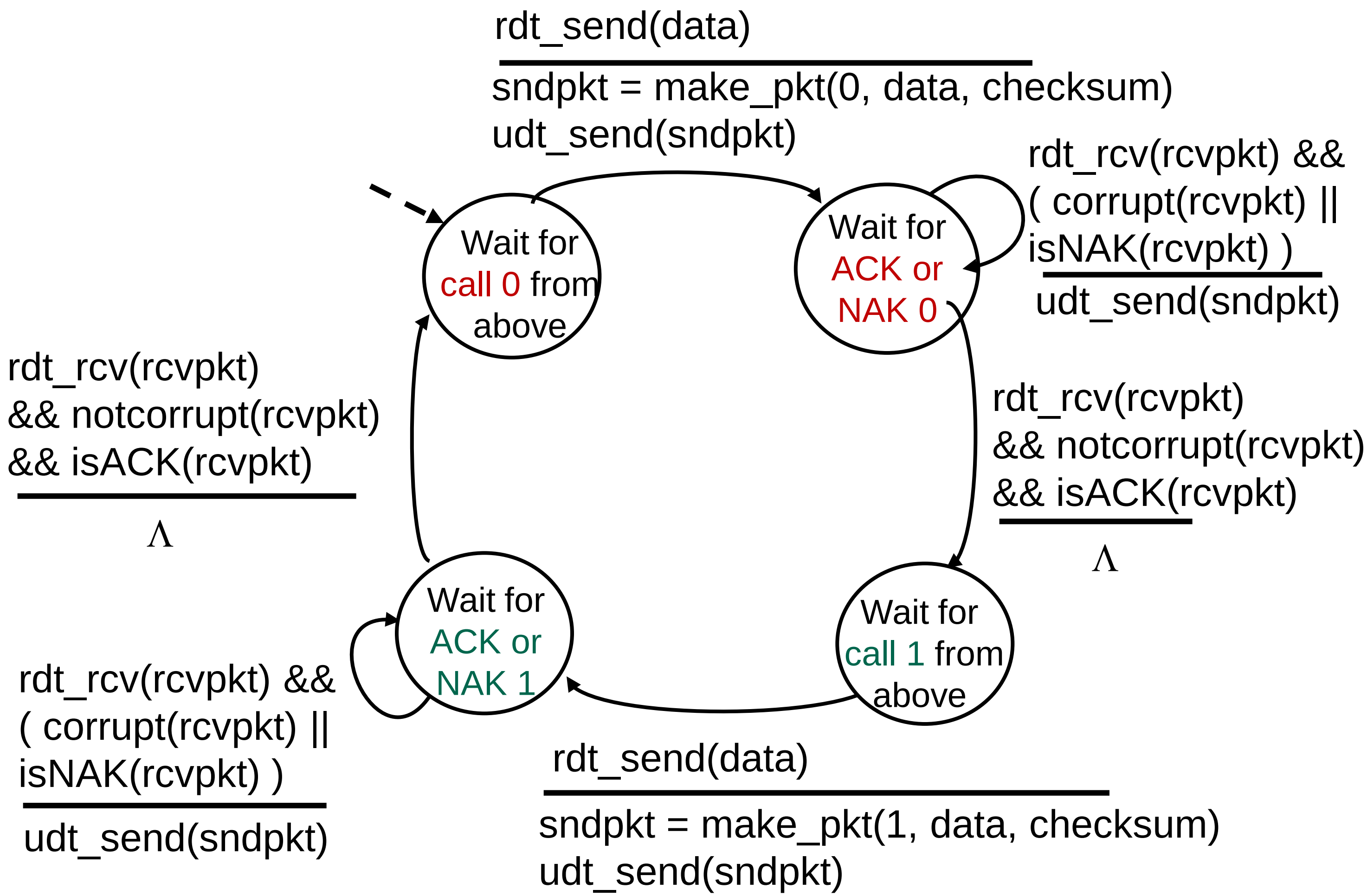
What happens if ACK/NAK corrupted?

- sender doesn't know what happened at receiver!
- can't just retransmit: possible duplicate

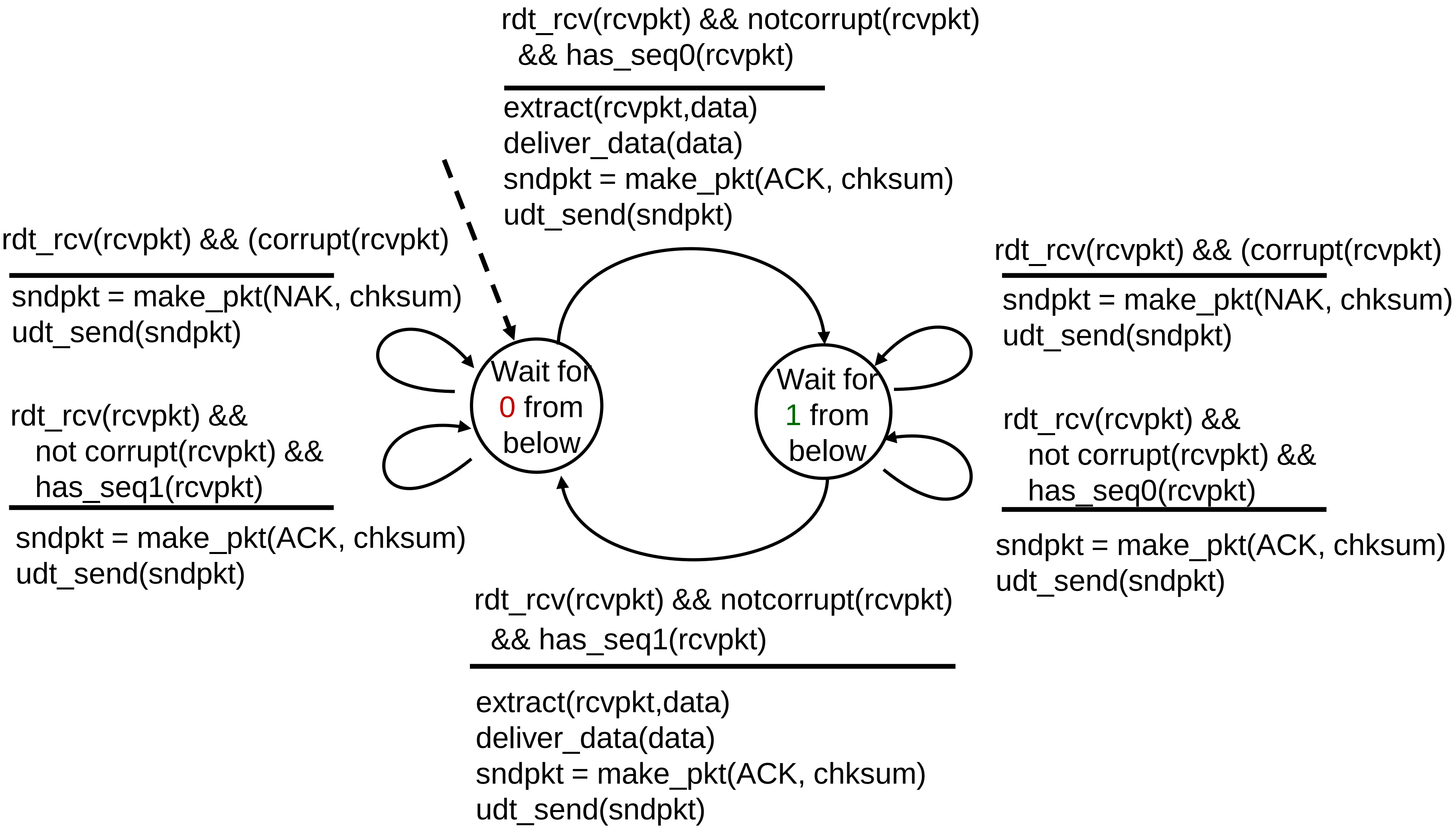
Handling duplicates:

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

rdt2.1: sender, handles garbled ACK/NAKs



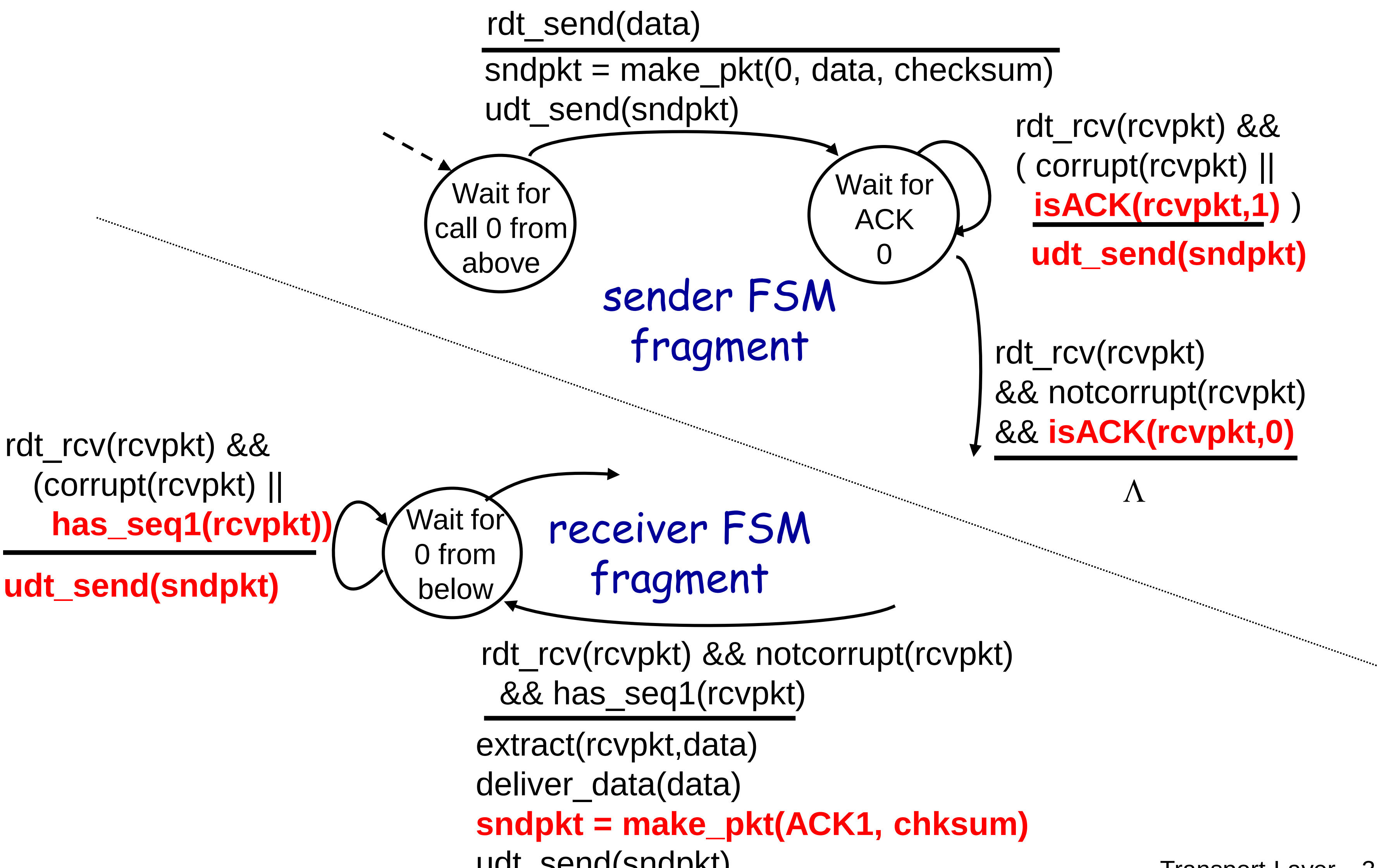
rdt2.1: receiver, handles garbled ACK/NAKs



rdt2.2: 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: channels with errors *and* loss

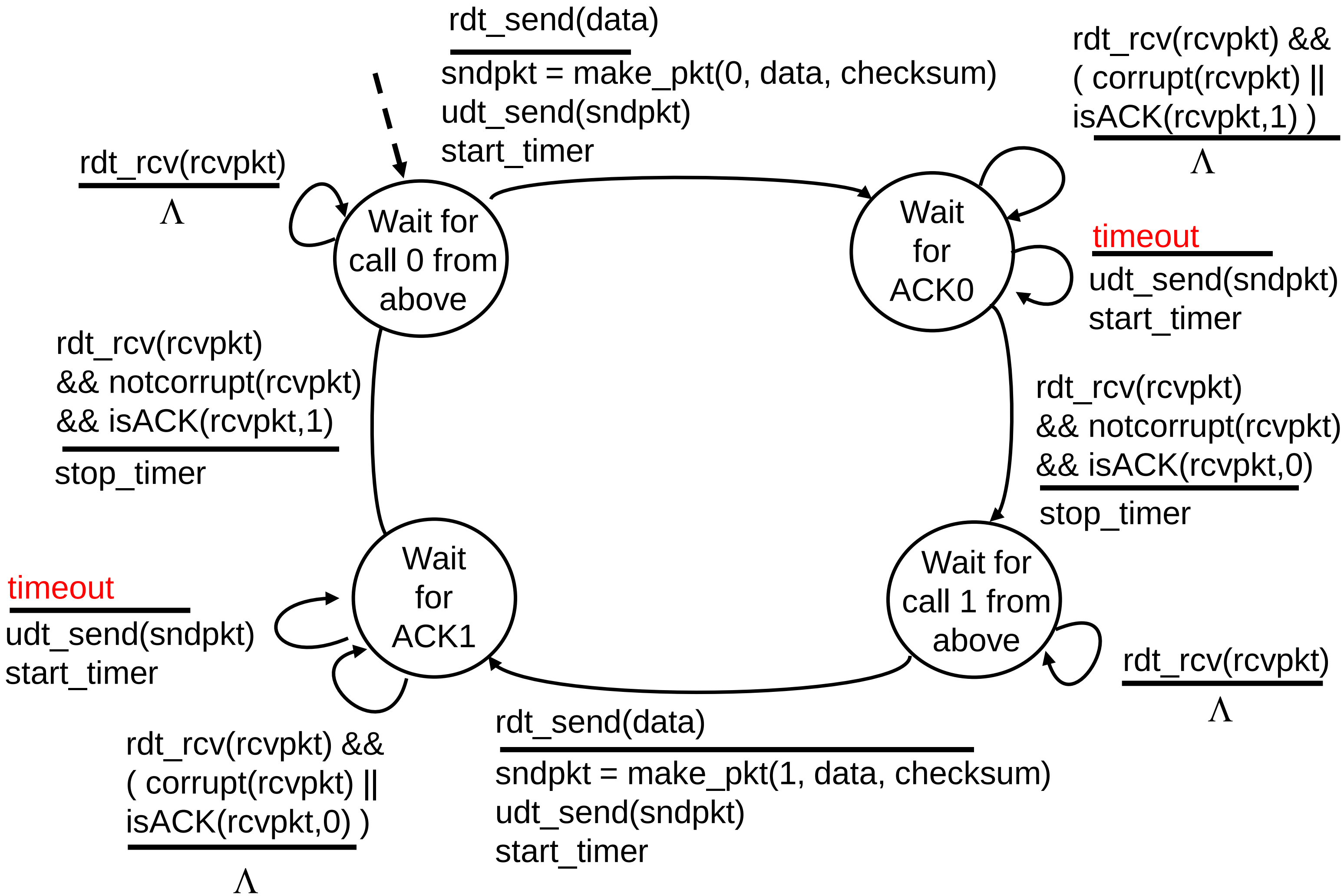
New assumption: underlying channel can lose packets (data or ACKs)

- checksum
- seq. #
- ACKs
- retransmissions
- ?

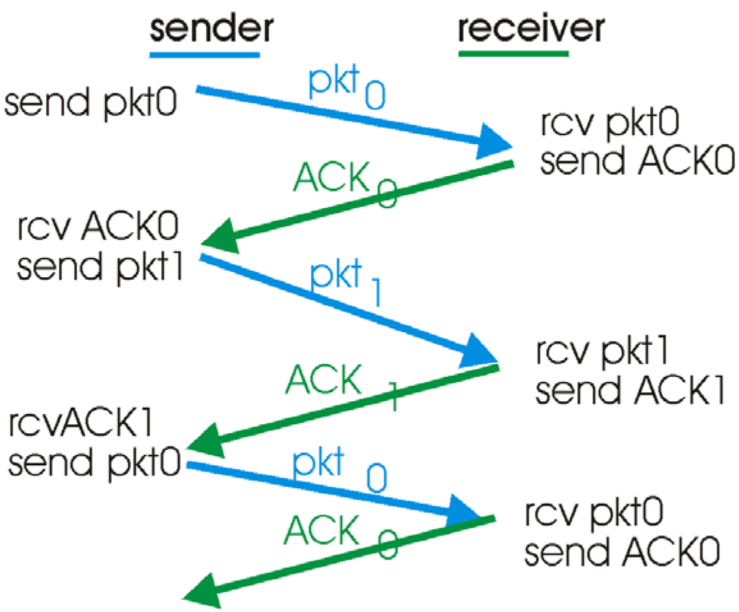
Approach: sender waits "reasonable" amount of time for ACK

- 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**

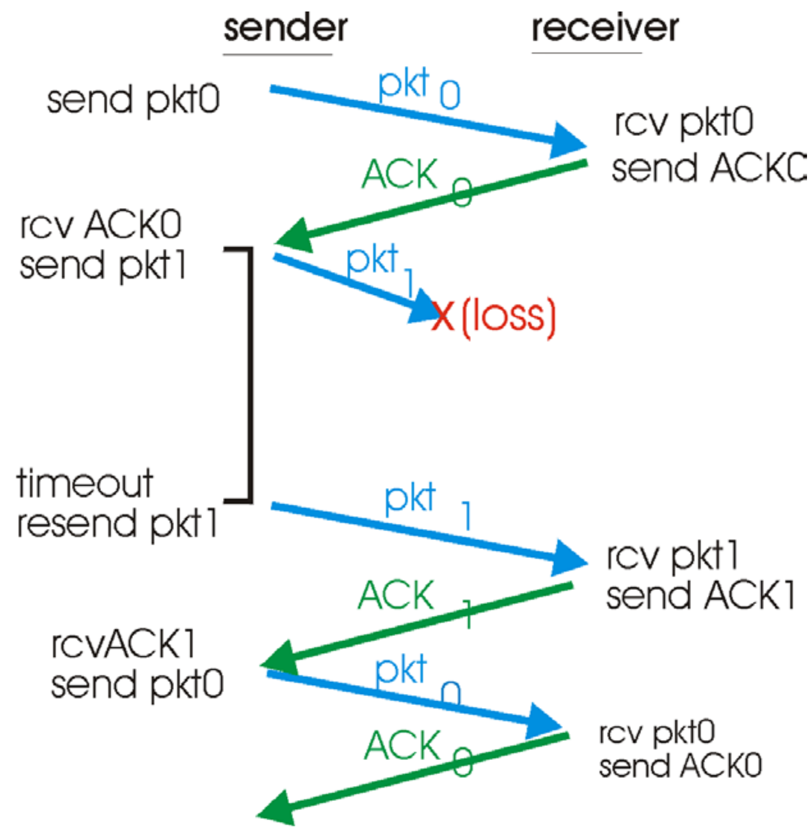
rdt3.0 sender



rdt3.0 in action

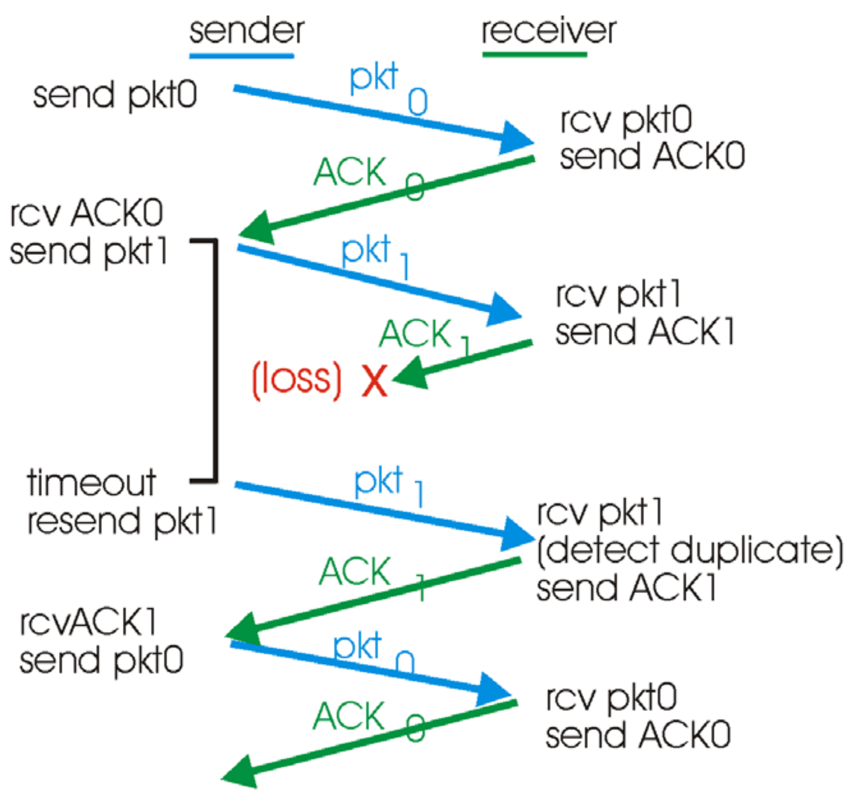


(a) operation with no loss

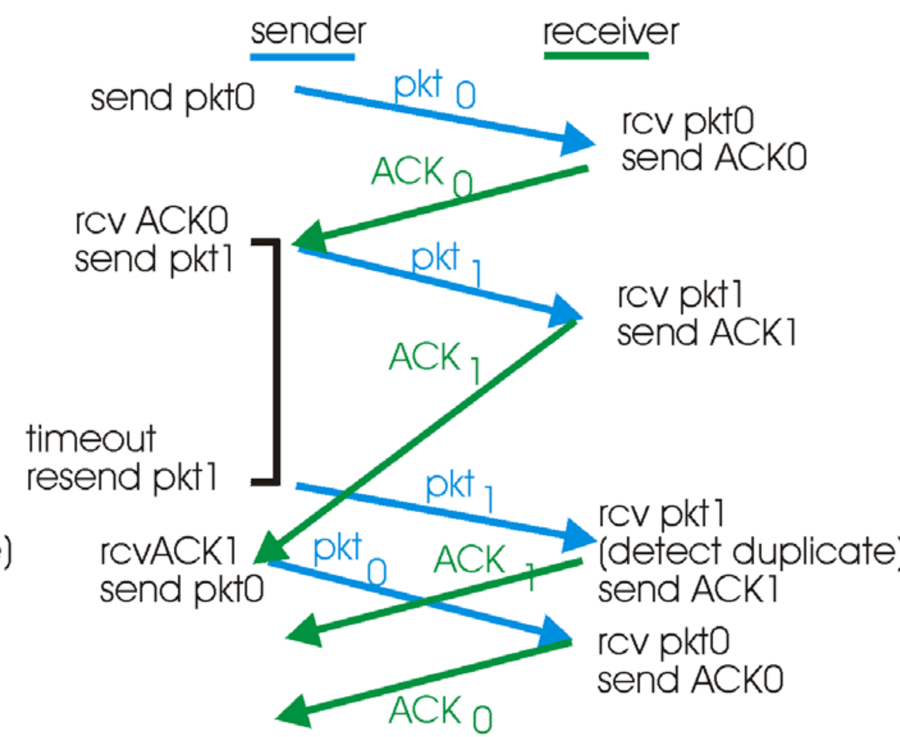


(b) lost packet

rdt3.0 in action

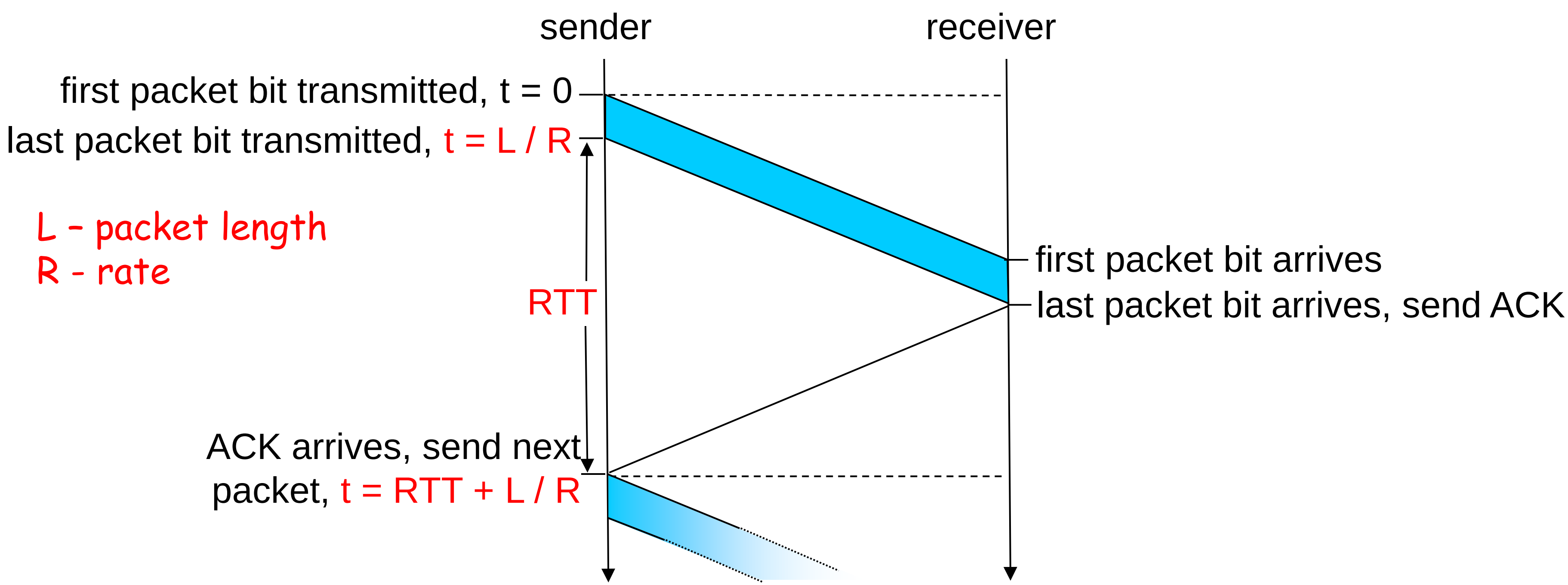


(c) lost ACK



(d) premature timeout

rdt3.0: stop-and-wait operation

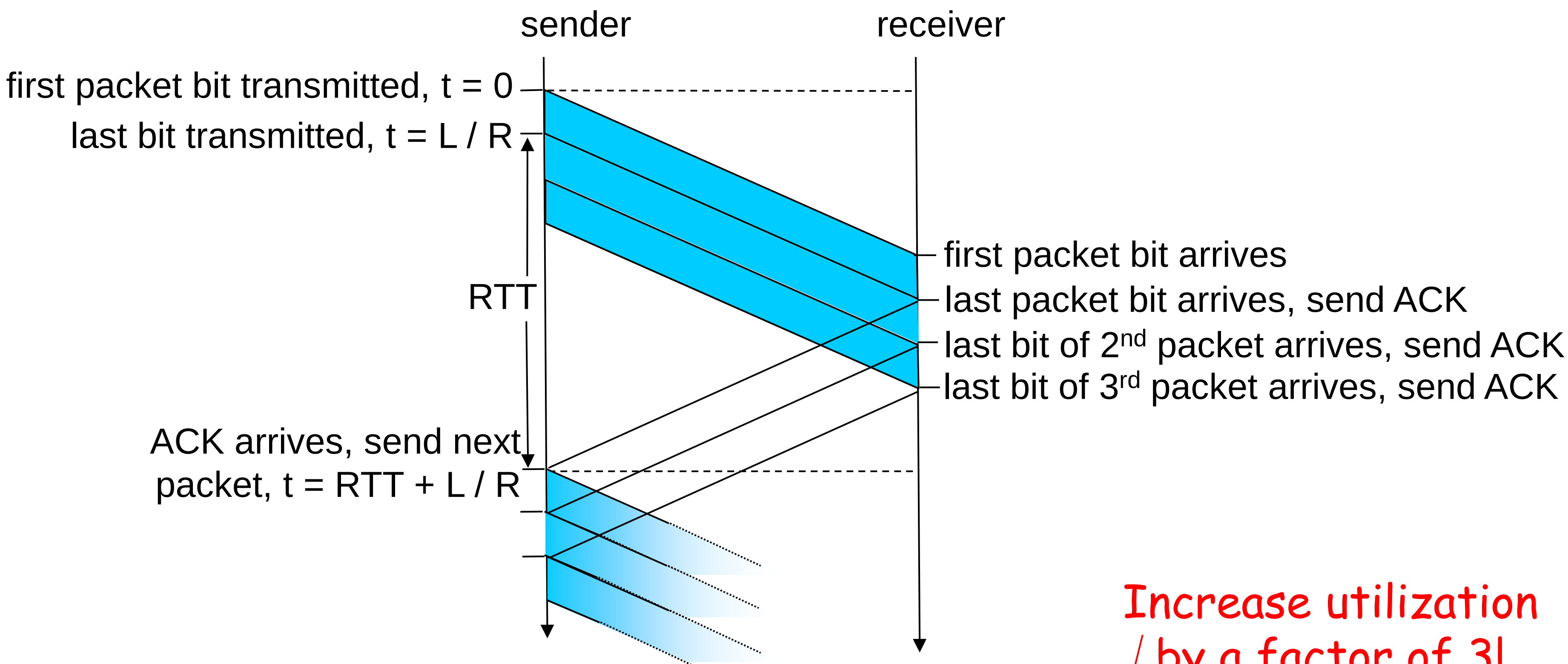


L - packet length
R - rate

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

Utilization

Pipelining: increased utilization



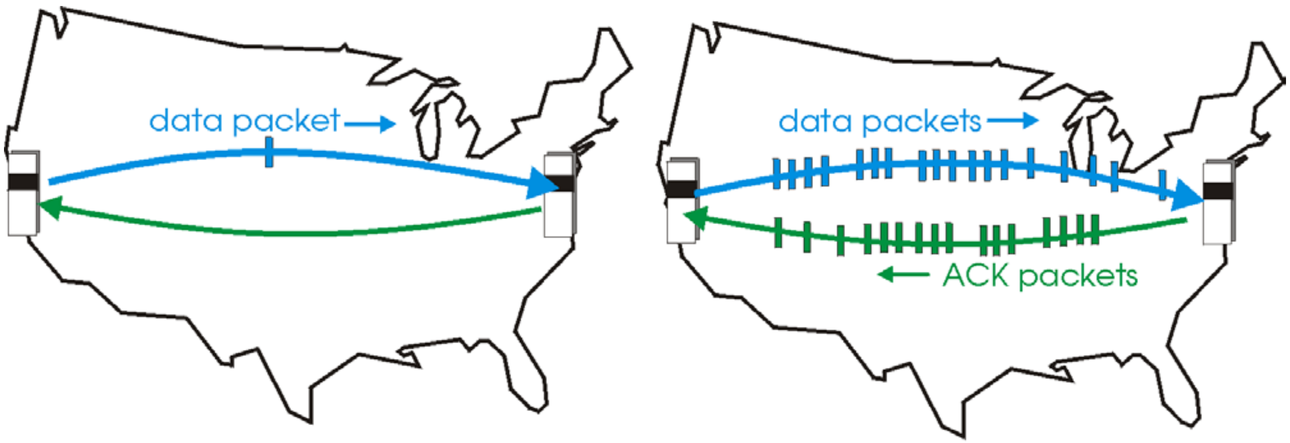
Increase utilization by a factor of 3!

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

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

Mechanisms for reliable data transfer

Stop-and-wait

- ✓ checksum
- ✓ seq. #
- ✓ ACKs
- ✓ retransmissions
- ✓ timeouts

Pipelined

- seq. # range increased
- windows
- Cumulative ACKs
- Send/recv buffers

Pipelined Protocols

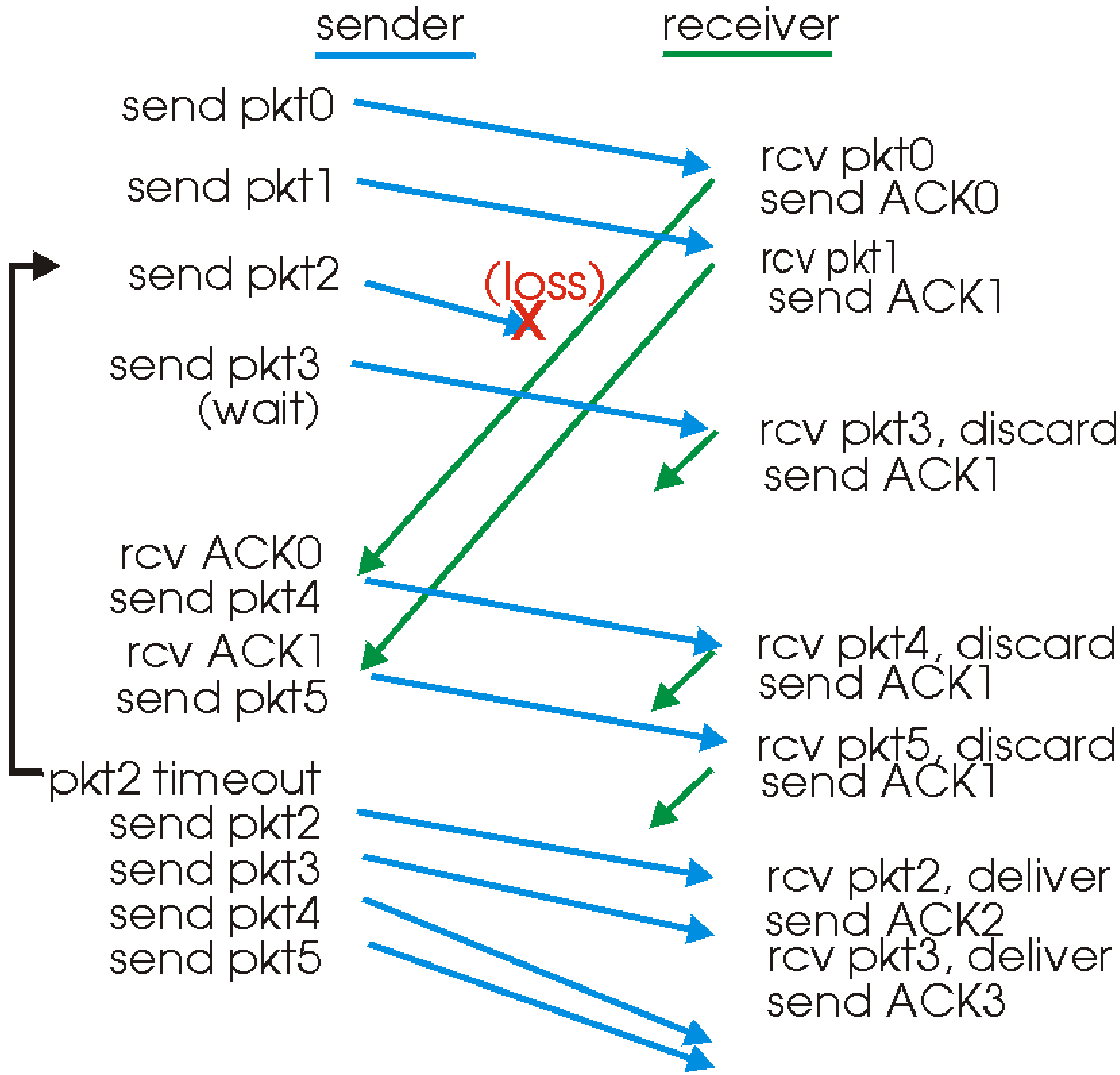
Go-back-N:

- N unacked packets in pipeline – Window
- *cumulative* acks
  - doesn't ack packet if there's a gap
- sender has timer for **oldest** unacked packet
  - if timer expires, retransmit **all** unack'ed packets

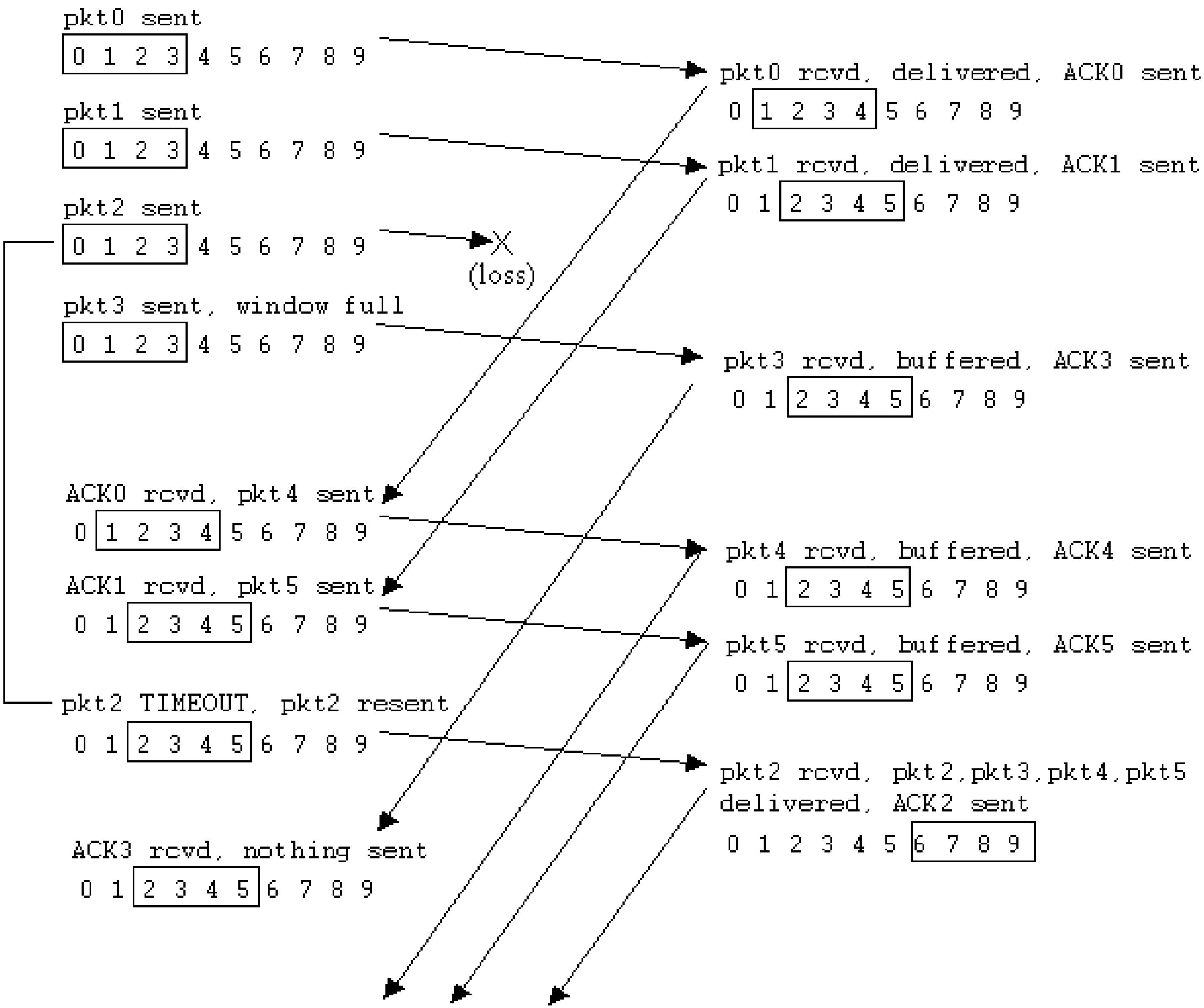
Selective Repeat:

- sender can have up to N unack'ed packets in pipeline
- rcvr sends *individual ack* for each packet
- sender maintains timer for each unacked packet
  - when timer expires, retransmit only unack'ed packet

GBN in action



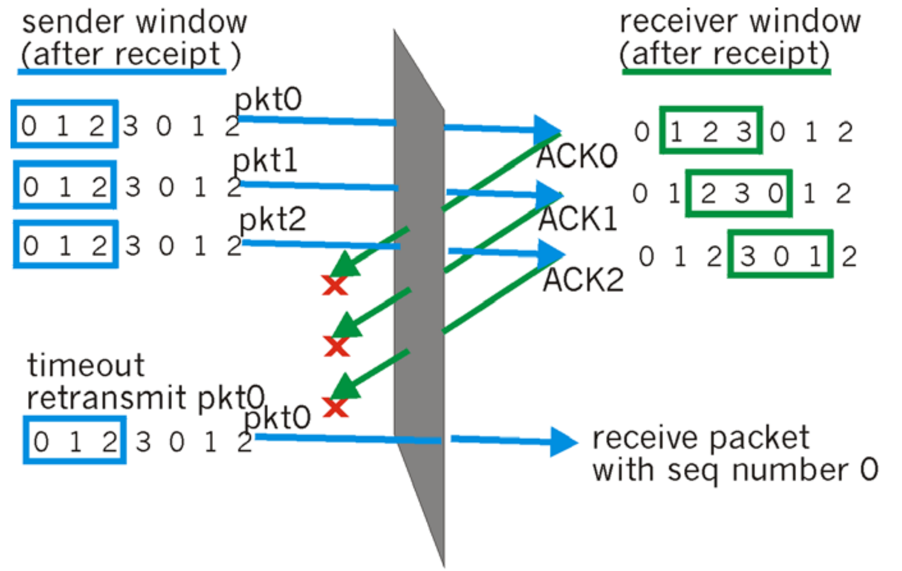
Selective repeat in action



Selective repeat: dilemma

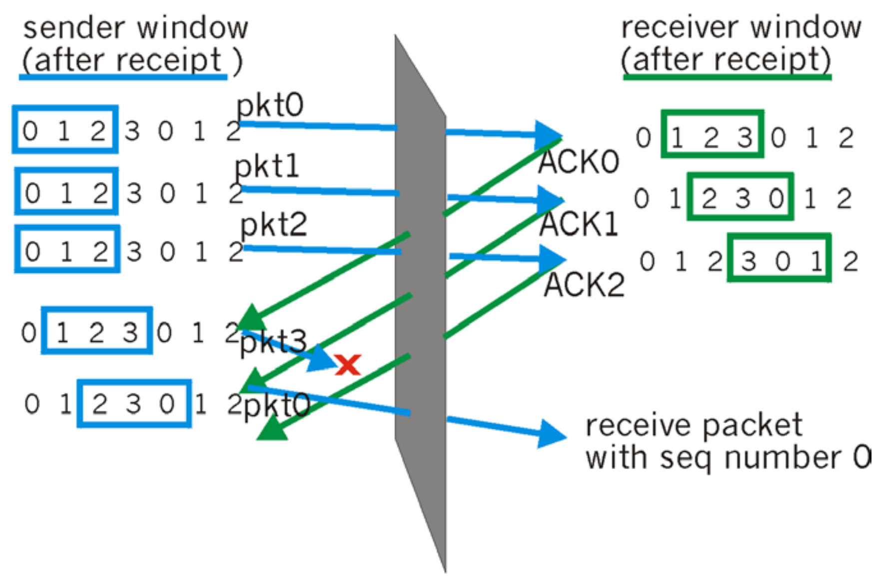
Example:

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



(a)

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



(b)

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

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