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

Transport layer: Outline

- Transport-layer services
- Multiplexing and demultiplexing
- Connectionless transport: UDP
- Principles of reliable data transfer
- Connection-oriented transport: TCP
  - Segment structure
  - Reliable data transfer
  - Flow control
  - Connection management
- Principles of congestion control
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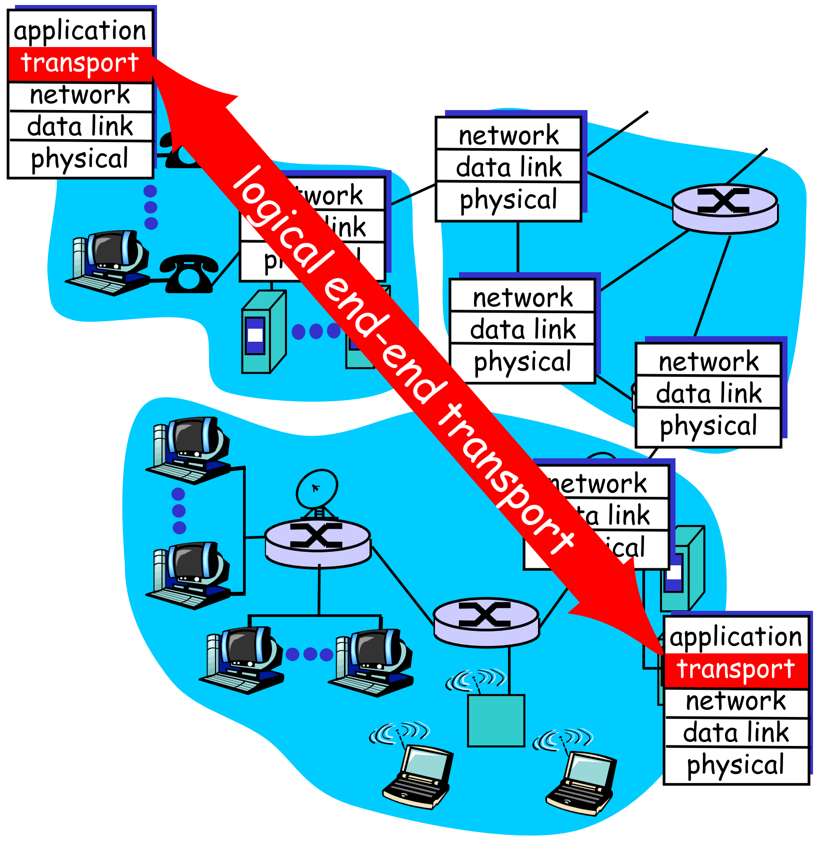
Transport services and protocols

- 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

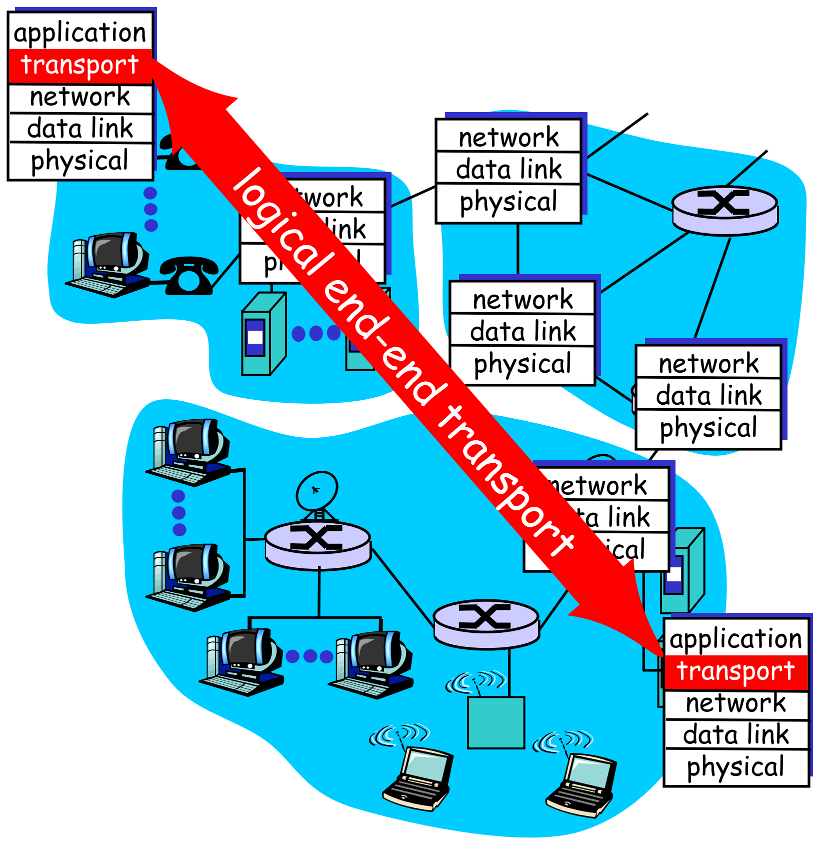
Household analogy:

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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Multiplexing/demultiplexing

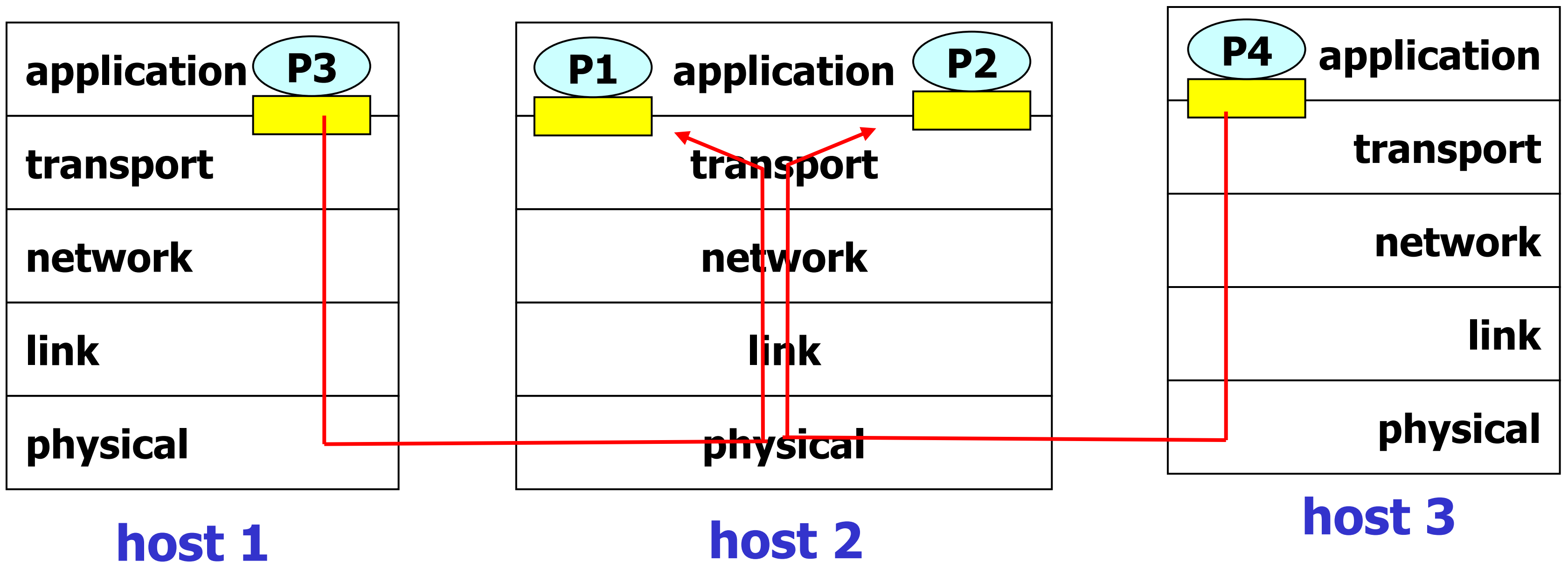
Demultiplexing at rcv host:

delivering received segments to correct socket

Multiplexing at send host:

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

🟨 = socket ⬭ = process



Connectionless demultiplexing

- Create sockets with port numbers:

```
DatagramSocket mySocket1 = new
   DatagramSocket(9111);
DatagramSocket mySocket2 = new
   DatagramSocket(9222);
```

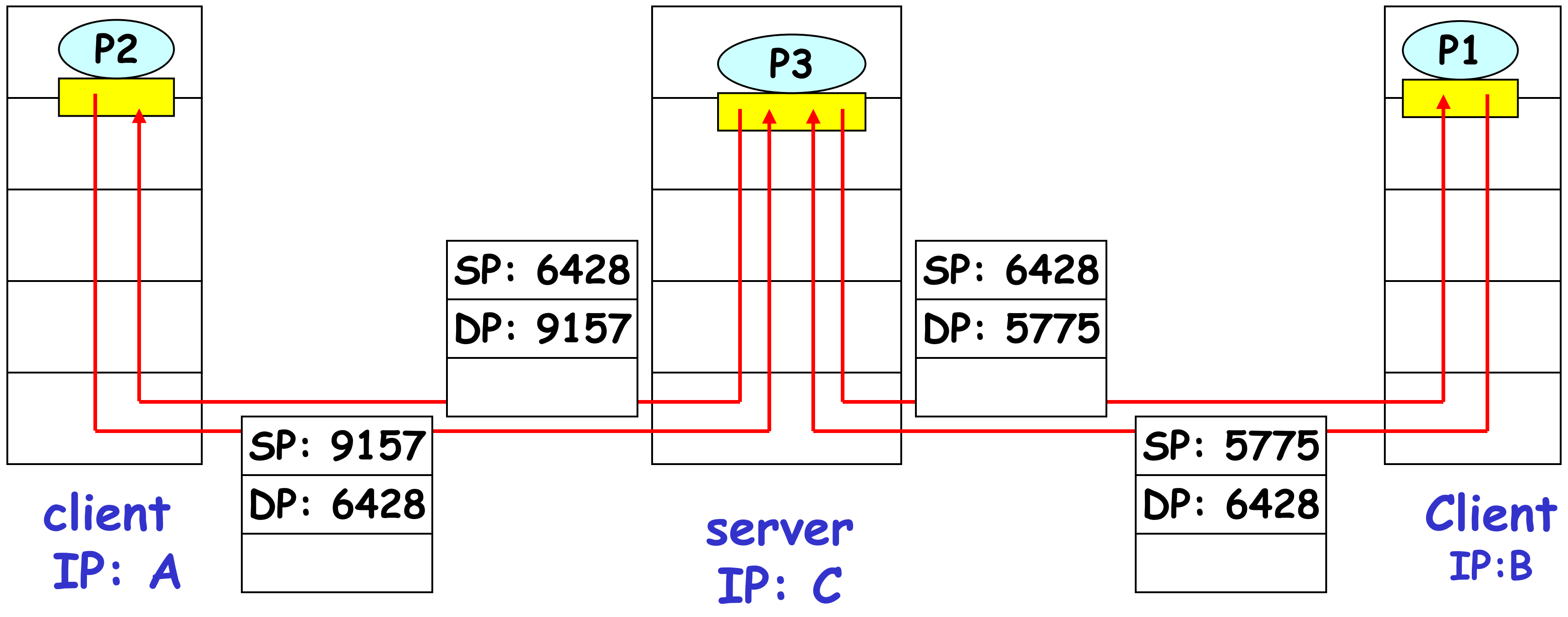
- UDP socket identified by two-tuple:

(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);
```

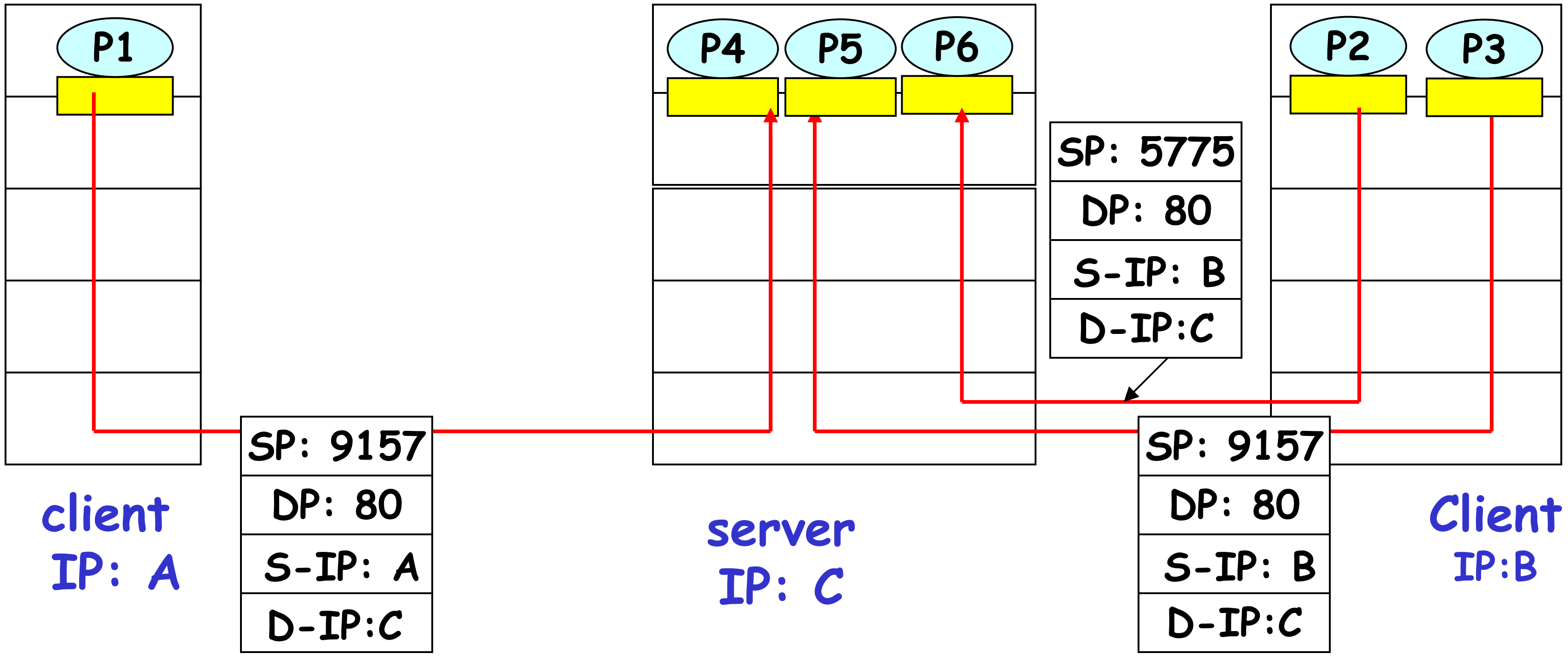


SP provides "return address"

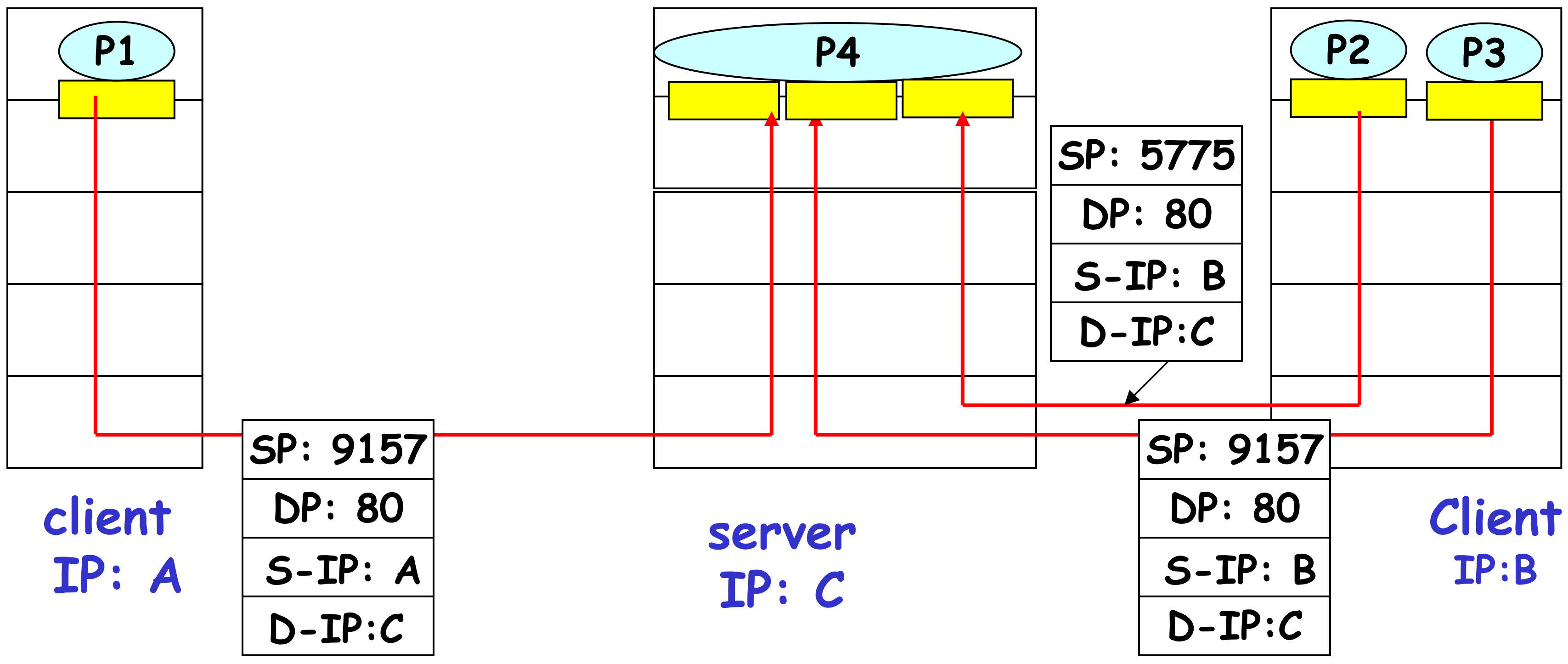
Connection-oriented demux

- 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]

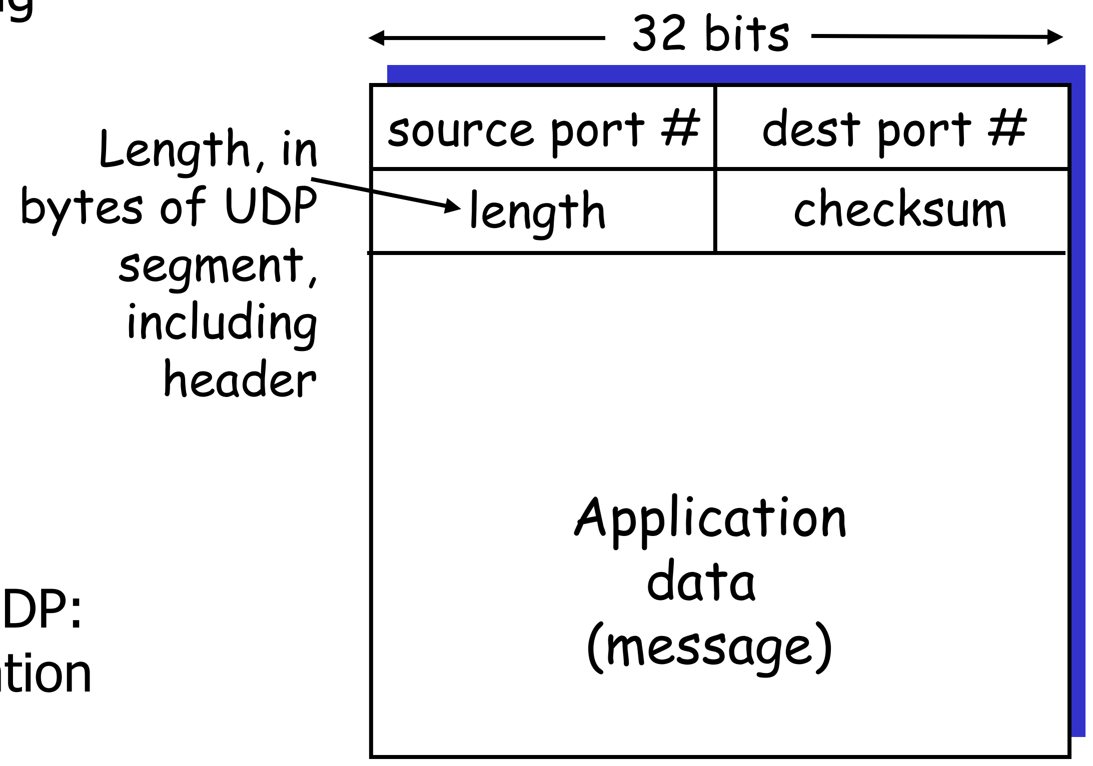
- "No frills," "bare bones" Internet transport protocol
- "Best effort" service, UDP segments may be:
  - Lost
  - Delivered out of order to app
- *Connectionless:*
  - No handshaking between UDP sender, receiver
  - Each UDP segment handled independently of others

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 (why?):
  - DNS
  - SNMP
- Reliable transfer over UDP: add reliability at application layer
  - Application-specific error recover!


32 bits
Length, in bytes of UDP segment, including header
source port #
dest port #
length
checksum
Application data (message)

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 nonethleess?* 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

		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	$\textcircled{1}$	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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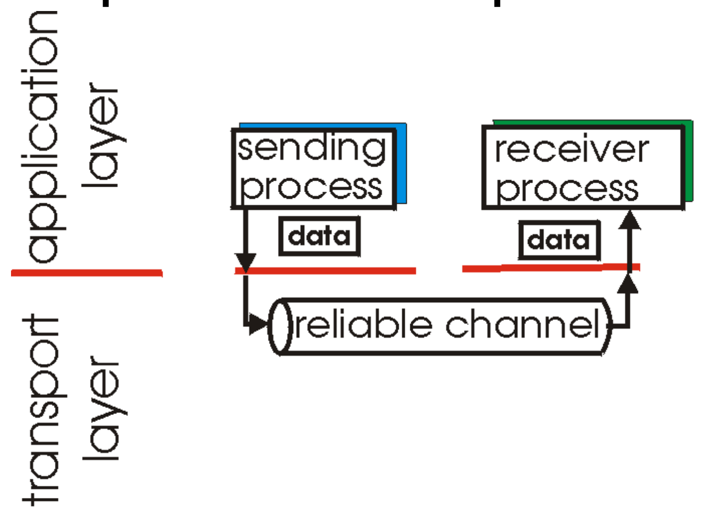
Principles of reliable data transfer

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

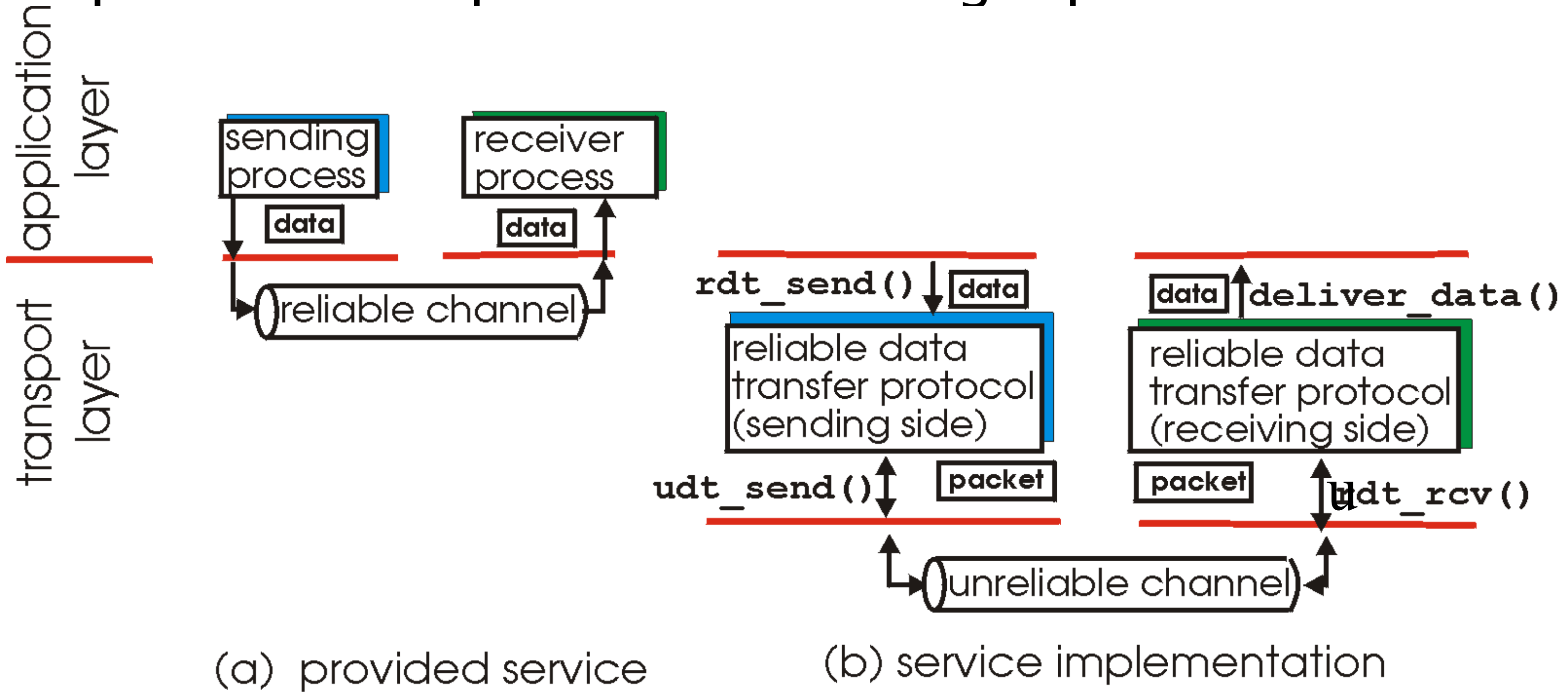


(a) provided service

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

Principles of reliable data transfer

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

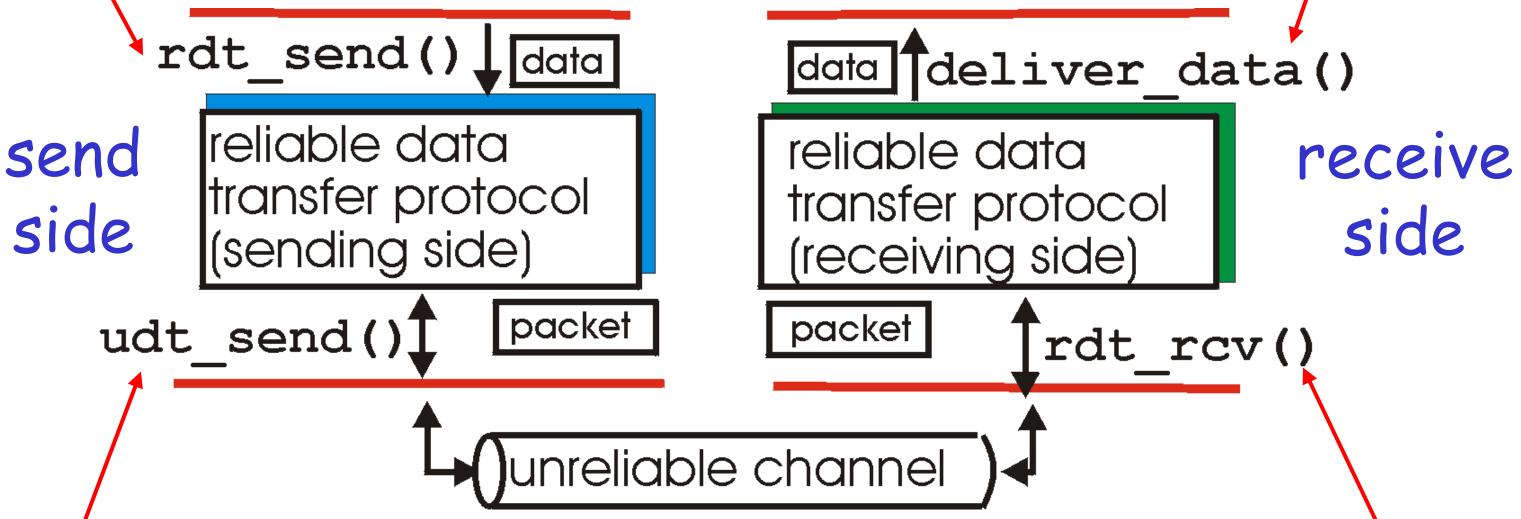


- Characteristics of unreliable channel will determine complexity of reliable data transfer protocol (rdt)

Reliable data transfer: Getting started

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



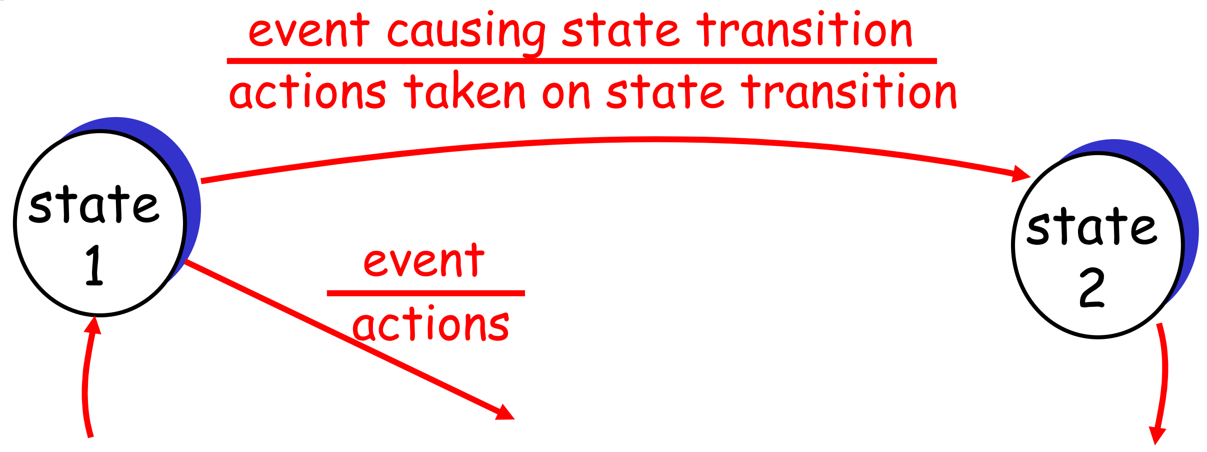
udt_send(): called by rdt, to transfer packet over unreliable channel to receiver

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

Reliable data transfer: Getting started

We'll:

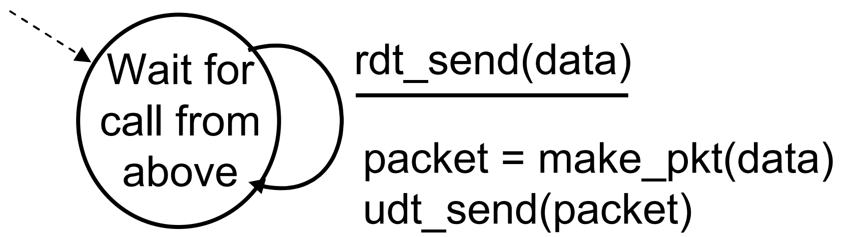
- Incrementally develop sender, receiver sides of reliable data transfer protocol (rdt)
- Consider only unidirectional data transfer
  - But control info will flow on both directions!
- Use finite state machines (FSM) to specify sender, receiver


event causing state transition
actions taken on state transition
state 1
state 2
event
actions

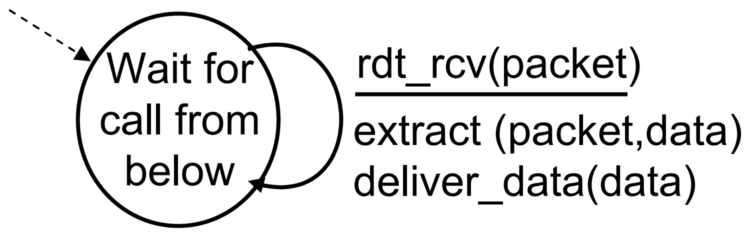
state: when in this "state" next state uniquely determined by next event

Rdt1.0: Reliable transfer over a reliable channel

- Underlying channel perfectly reliable
  - No bit erros
  - No loss of packets
- Separate FSMs for sender, receiver:
  - Sender sends data into underlying channel
  - Receiver read data from underlying channel



sender



receiver

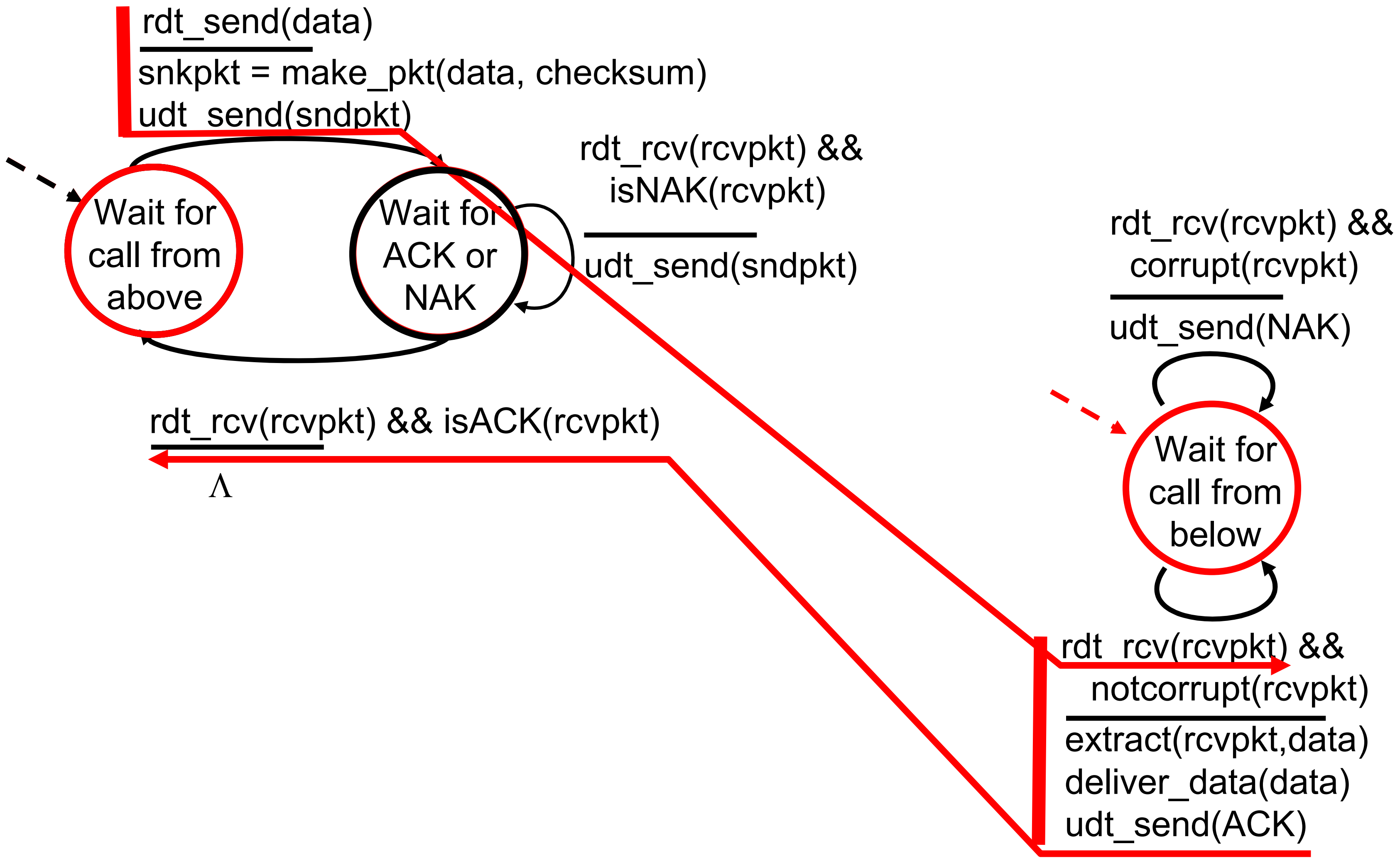
Rdt2.0: Channel with bit errors

- Underlying channel may flip bits in packet
  - Recall: UDP checksum to detect bit errors
- *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?

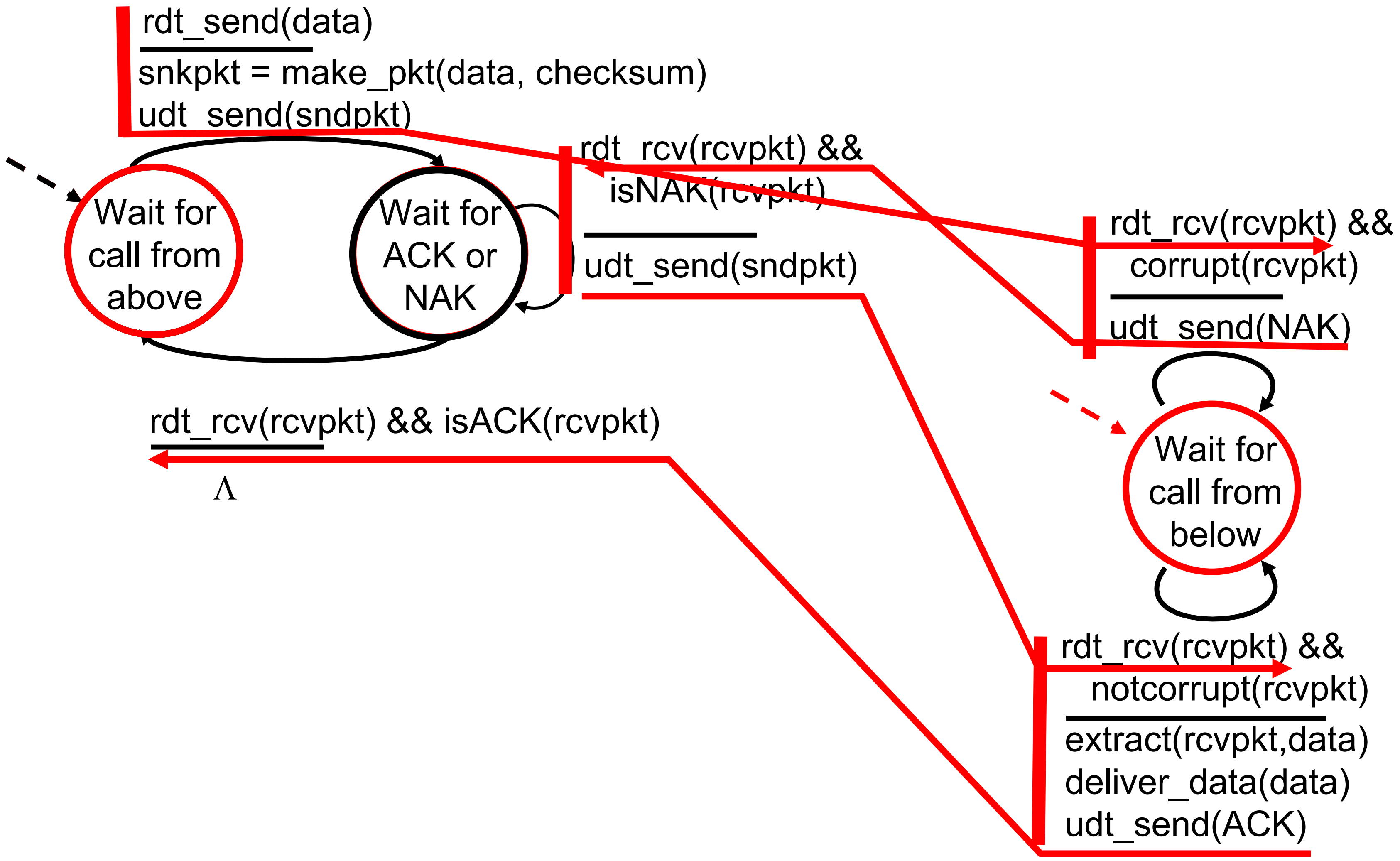
New mechanisms in **`rdt2.0`** (beyond **`rdt1.0`**):

- Error detection
- Receiver feedback: control msgs (ACK,NAK) rcvr->sender

Rdt2.0: 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?

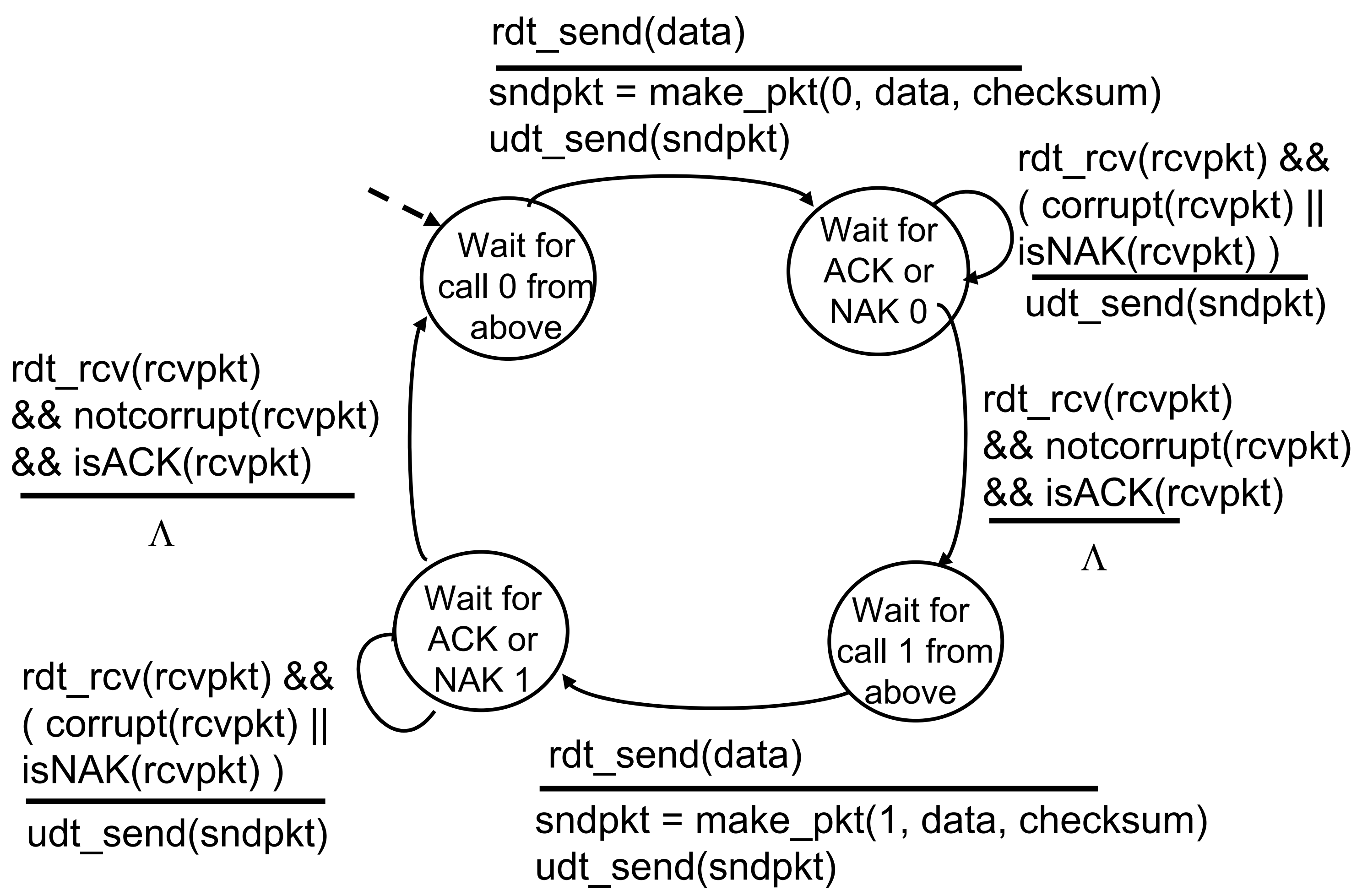
- Sender ACKs/NAKs receiver's ACK/NAK? What if sender ACK/NAK lost?
- Retransmit, but this might cause retransmission of correctly received pkt!

Handling duplicates:

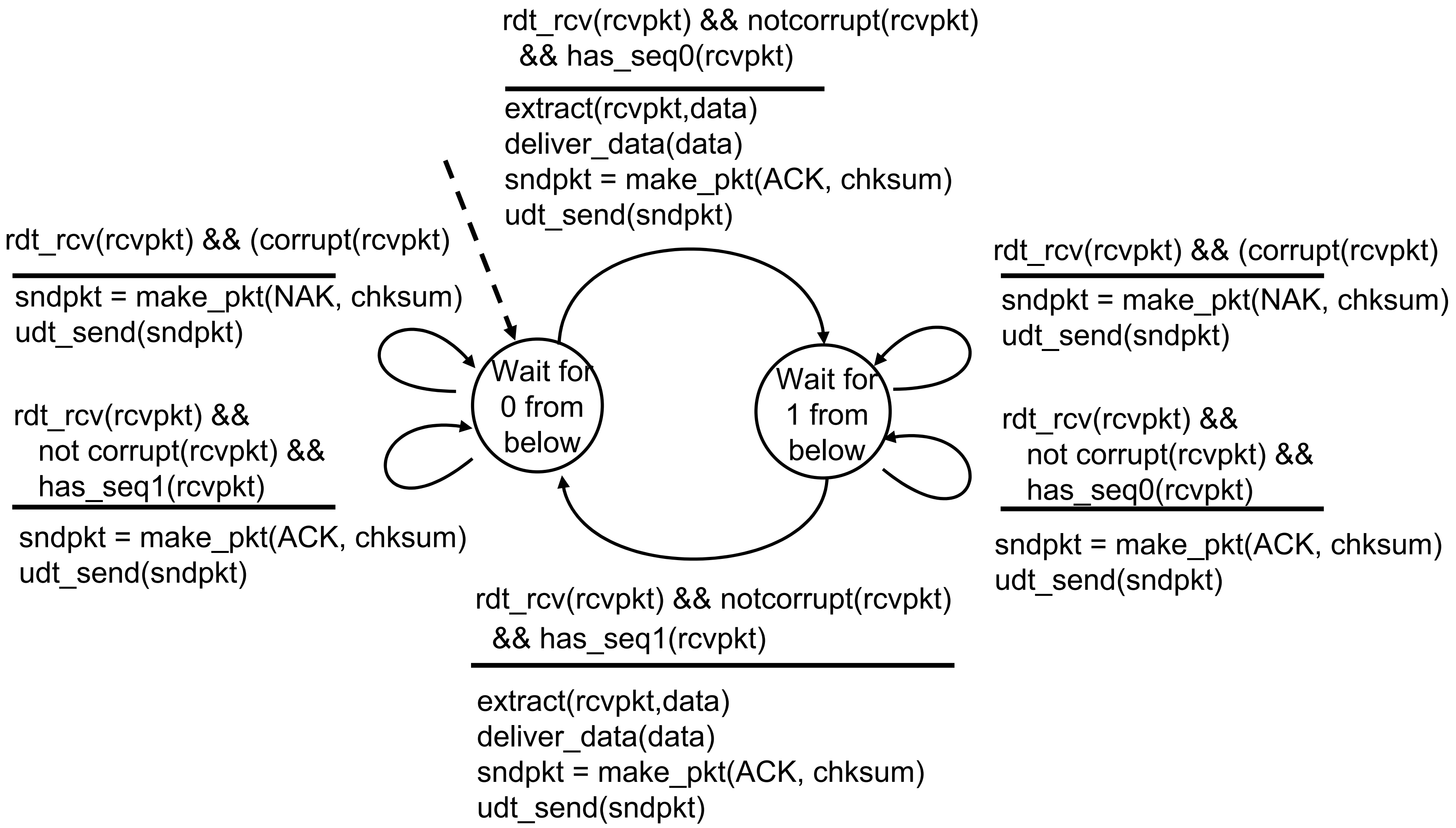
- Sender retransmits current pkt if ACK/NAK garbled
- Sender adds *sequence number* to each pkt
- Receiver discards (doesn't deliver up) duplicate pkt

stop and wait
Sender sends one packet, then waits for receiver response

Rdt2.1: Sender with garbled ACK/NAKs


rdt_send(data)
sndpkt = make_pkt(0, data, checksum)
udt_send(sndpkt)
Wait for call 0 from above
Wait for ACK or NAK 0
rdt_rcv(rcvpkt) && (corrupt(rcvpkt) || isNAK(rcvpkt))
udt_send(sndpkt)
rdt_rcv(rcvpkt) && notcorrupt(rcvpkt) && isACK(rcvpkt)
Λ
rdt_rcv(rcvpkt) && notcorrupt(rcvpkt) && isACK(rcvpkt)
Λ
Wait for ACK or NAK 1
Wait for call 1 from above
rdt_rcv(rcvpkt) && (corrupt(rcvpkt) || isNAK(rcvpkt))
udt_send(sndpkt)
rdt_send(data)
sndpkt = make_pkt(1, data, checksum)
udt_send(sndpkt)

Rdt2.1: Receiver with garbled ACK/NAKs



```
rdt_rcv(rcvpkt) && notcorrupt(rcvpkt)
  && has_seq0(rcvpkt)
_______________________________
extract(rcvpkt,data)
deliver_data(data)
sndpkt = make_pkt(ACK, chksum)
udt_send(sndpkt)
```

```
rdt_rcv(rcvpkt) && (corrupt(rcvpkt)
_______________________________
sndpkt = make_pkt(NAK, chksum)
udt_send(sndpkt)
```

```
rdt_rcv(rcvpkt) &&
  not corrupt(rcvpkt) &&
  has_seq1(rcvpkt)
_______________________________
sndpkt = make_pkt(ACK, chksum)
udt_send(sndpkt)
```

```
rdt_rcv(rcvpkt) && (corrupt(rcvpkt)
_______________________________
sndpkt = make_pkt(NAK, chksum)
udt_send(sndpkt)
```

```
rdt_rcv(rcvpkt) &&
  not corrupt(rcvpkt) &&
  has_seq0(rcvpkt)
_______________________________
sndpkt = make_pkt(ACK, chksum)
udt_send(sndpkt)
```

```
rdt_rcv(rcvpkt) && notcorrupt(rcvpkt)
  && has_seq1(rcvpkt)
_______________________________
extract(rcvpkt,data)
deliver_data(data)
sndpkt = make_pkt(ACK, chksum)
udt_send(sndpkt)
```

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. #

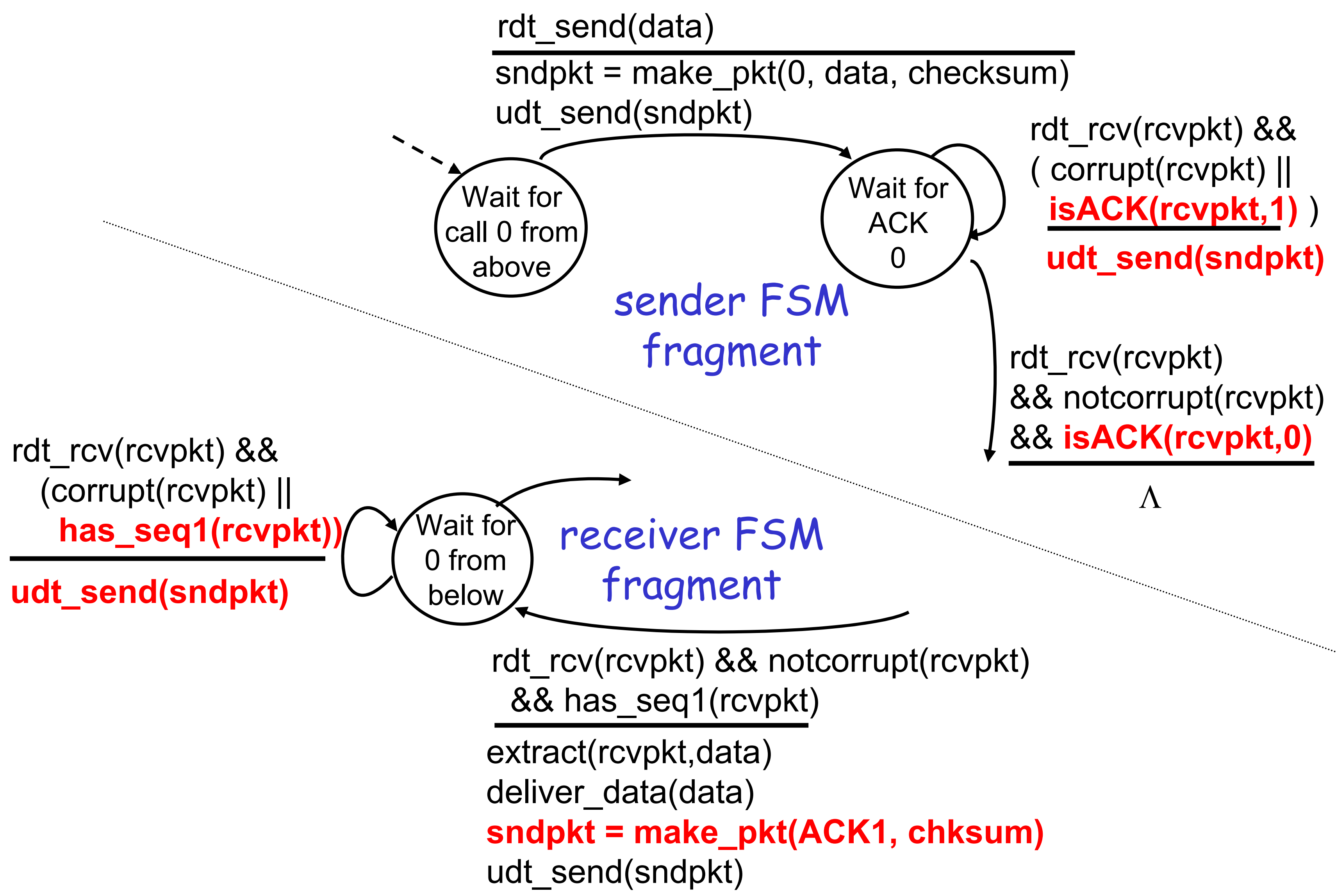
Receiver:

- 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 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



rdt_send(data)
sndpkt = make_pkt(0, data, checksum)
udt_send(sndpkt)
Wait for call 0 from above
Wait for ACK 0
rdt_rcv(rcvpkt) && (corrupt(rcvpkt) \|\| isACK(rcvpkt,1))
udt_send(sndpkt)
sender FSM fragment
rdt_rcv(rcvpkt) && notcorrupt(rcvpkt) && isACK(rcvpkt,0)
Λ
rdt_rcv(rcvpkt) && (corrupt(rcvpkt) \|\| has_seq1(rcvpkt))
udt_send(sndpkt)
Wait for 0 from below
receiver FSM fragment
rdt_rcv(rcvpkt) && notcorrupt(rcvpkt) && has_seq1(rcvpkt)
extract(rcvpkt,data)
deliver_data(data)
sndpkt = make_pkt(ACK1, chksum)
udt_send(sndpkt)

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

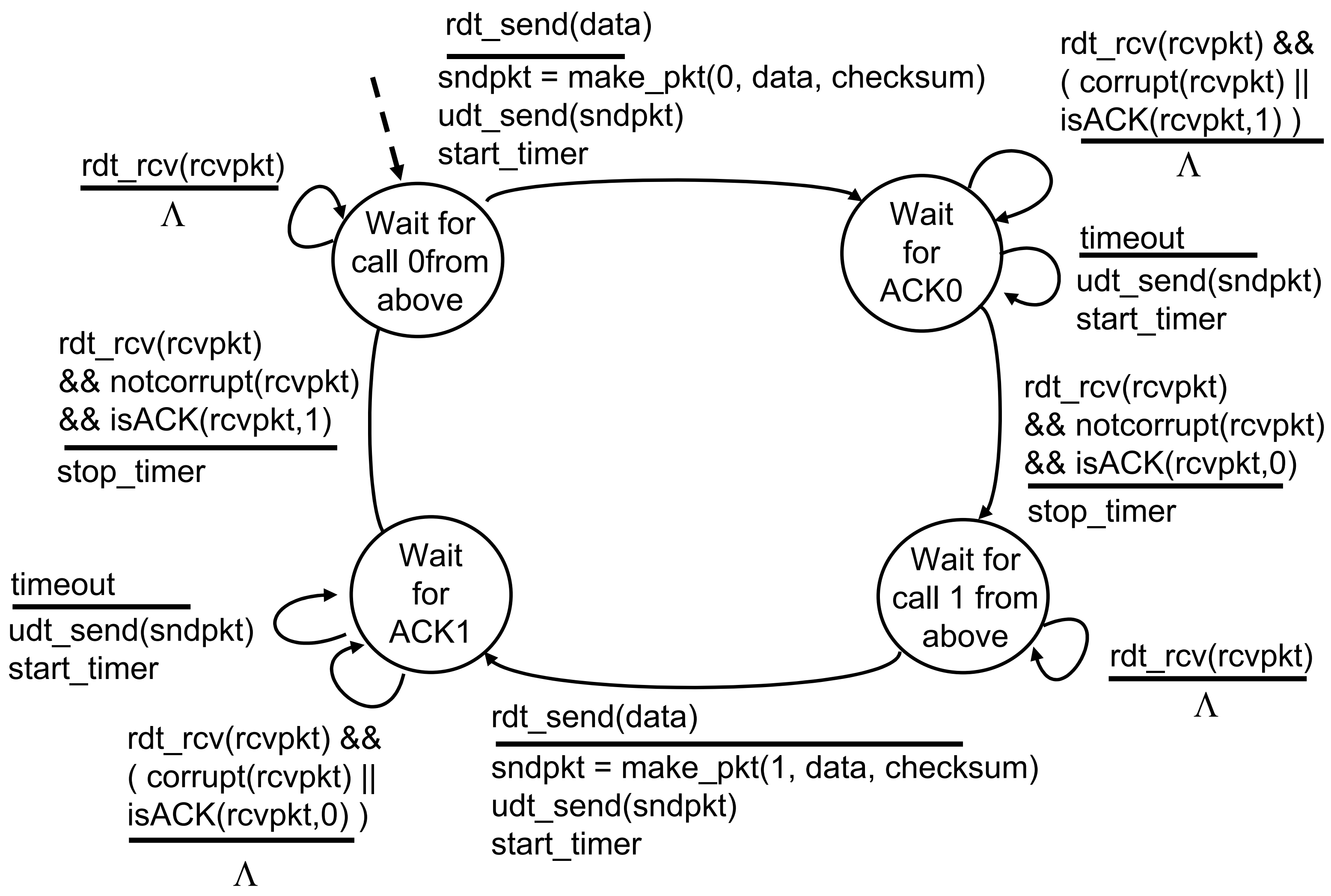
Q: How to deal with loss?

- Sender waits until certain data or ACK lost, then retransmits

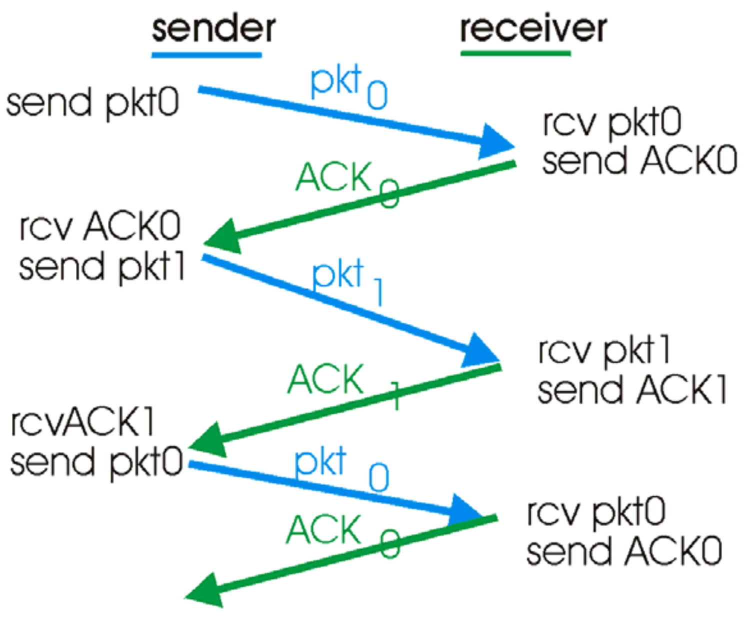
Approach: sender waits "reasonable" amount of time for ACK

- 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

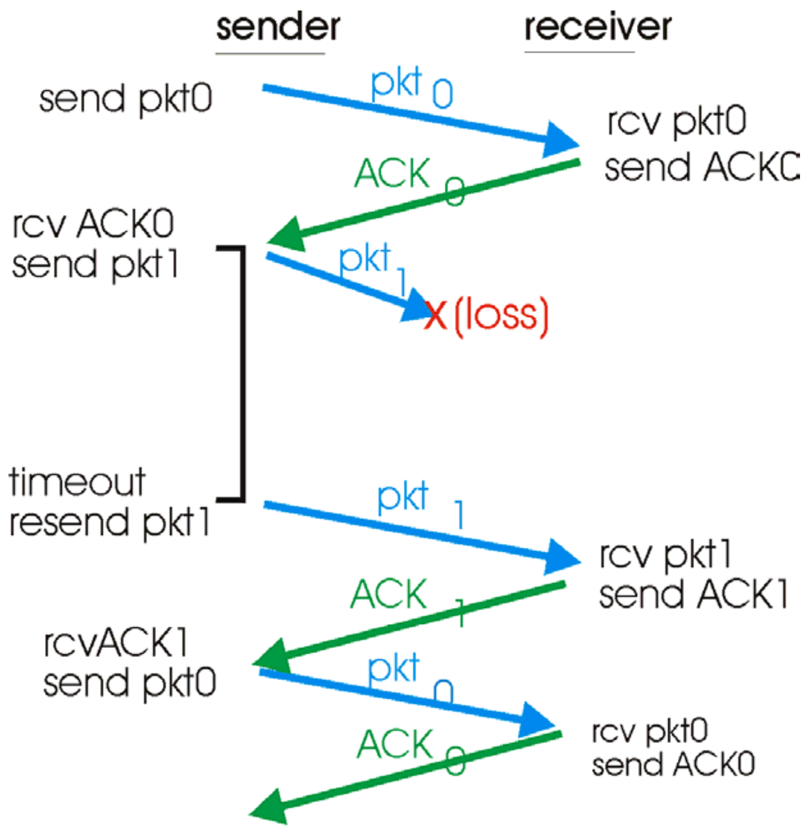
Rdt3.0 sender


rdt_send(data)
sndpkt = make_pkt(0, data, checksum)
udt_send(sndpkt)
start_timer
rdt_rcv(rcvpkt)
Λ
Wait for call 0from above
Wait for ACK0
rdt_rcv(rcvpkt) && (corrupt(rcvpkt) || isACK(rcvpkt,1))
Λ
timeout
udt_send(sndpkt)
start_timer
rdt_rcv(rcvpkt) && notcorrupt(rcvpkt) && isACK(rcvpkt,1)
stop_timer
rdt_rcv(rcvpkt) && notcorrupt(rcvpkt) && isACK(rcvpkt,0)
stop_timer
timeout
udt_send(sndpkt)
start_timer
Wait for ACK1
Wait for call 1 from above
rdt_rcv(rcvpkt)
Λ
rdt_rcv(rcvpkt) && (corrupt(rcvpkt) || isACK(rcvpkt,0))
Λ
rdt_send(data)
sndpkt = make_pkt(1, data, checksum)
udt_send(sndpkt)
start_timer

Rdt3.0 in action

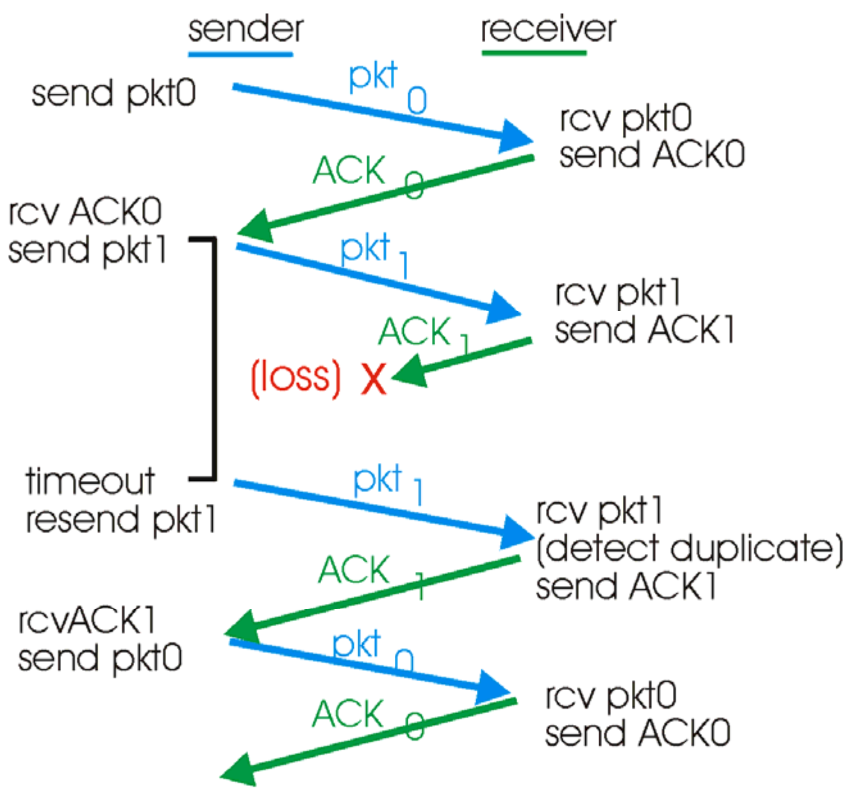


(a) operation with no loss

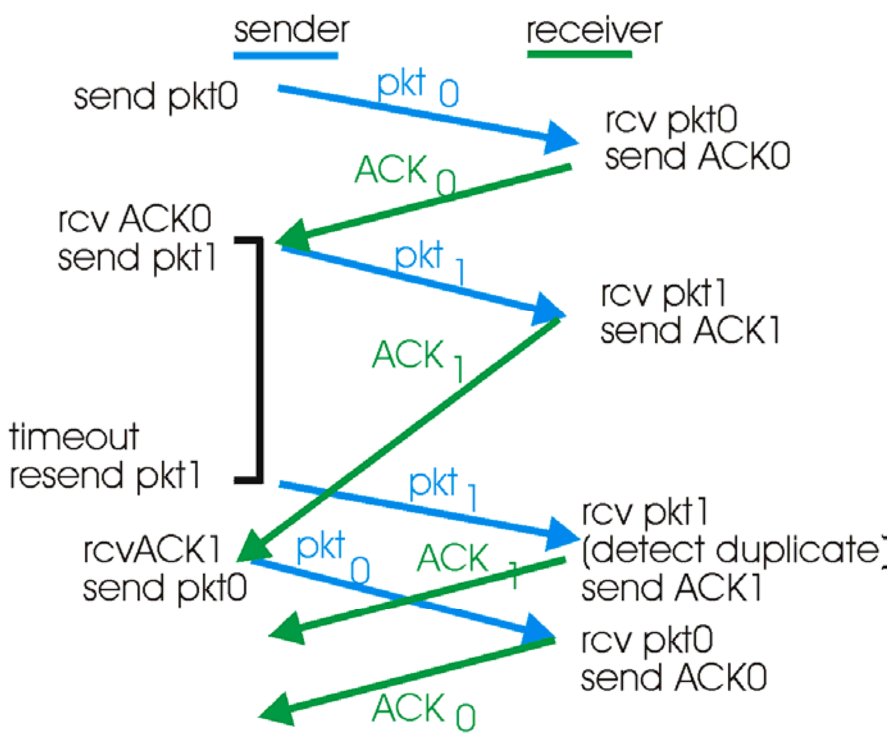


(b) lost packet

Rdt3.0 in action (cont.)



(c) lost ACK



(d) premature timeout

Performance of rdt3.0

- Rdt3.0 works, but performance stinks
- Example: 1 Gbps link, 15 ms e-e prop. delay, 1KB packet:

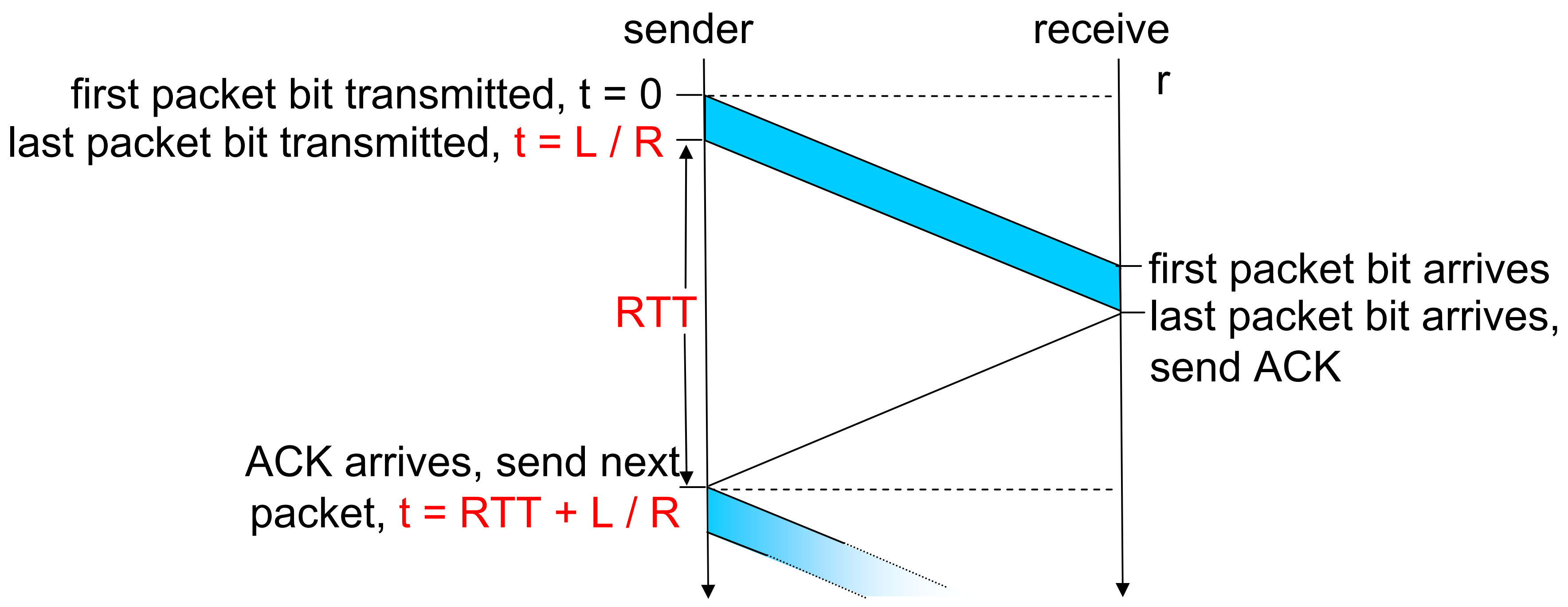
$$T_{transmit} = \frac{L \text{ (packet length in bits)}}{R \text{ (transmission rate, bps)}} = \frac{8\text{kb/pkt}}{10{*}{*}9 \text{ b/sec}} = 8 \text{ microsec}$$

  - 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

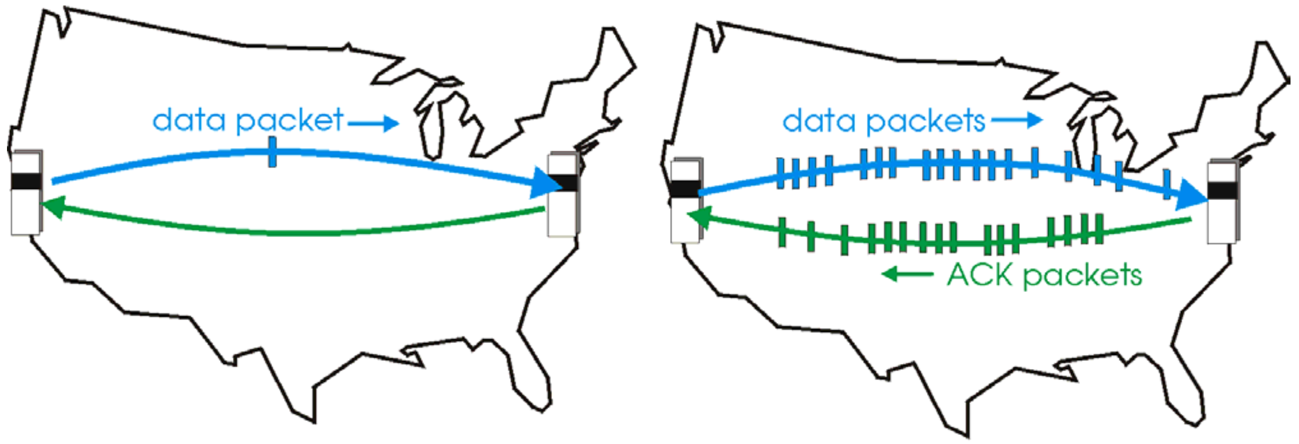

sender
receiver
first packet bit transmitted, t = 0
last packet bit transmitted, t = L / R
RTT
first packet bit arrives
last packet bit arrives, send ACK
ACK arrives, send next packet, t = RTT + L / R

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

Pipelined protocols

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

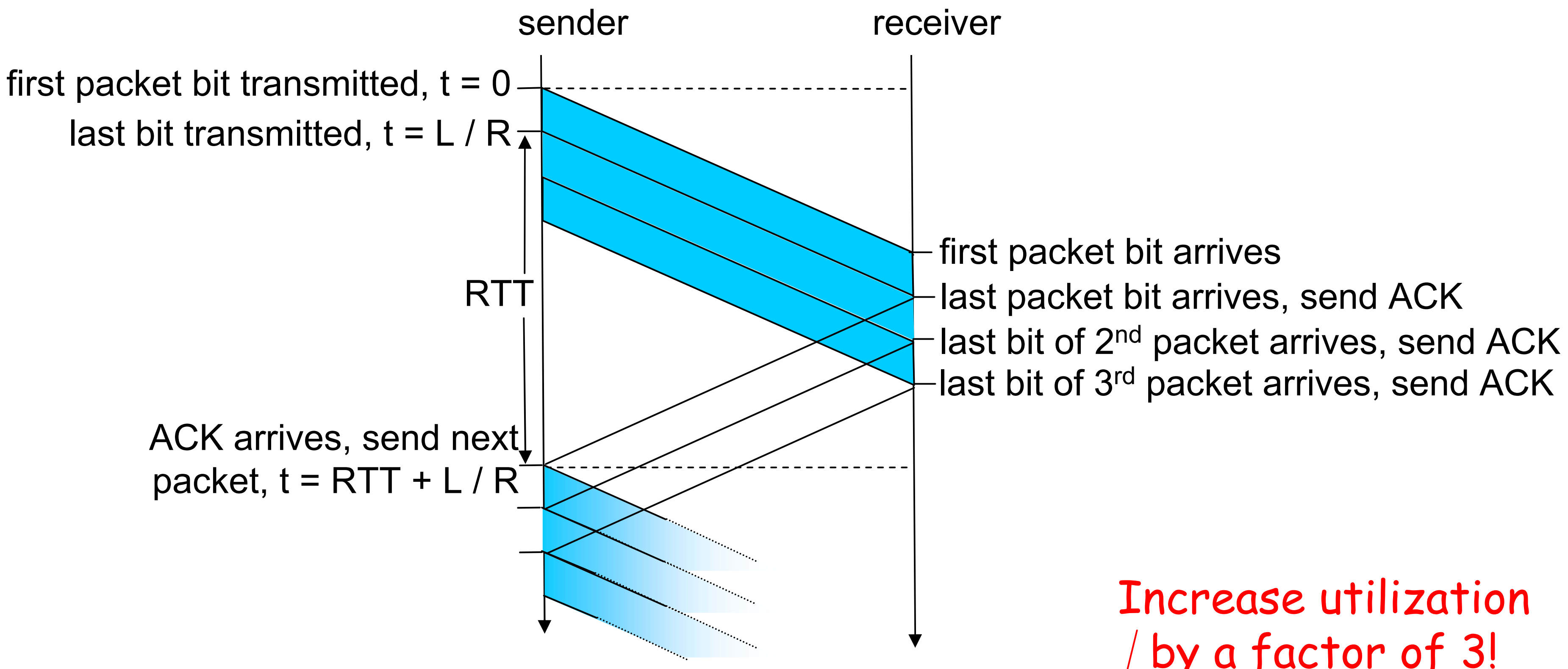
- Range of sequence numbers must be increased
- Buffering at sender and/or receiver



(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



sender
receiver
first packet bit transmitted, t = 0
last bit transmitted, t = L / R
RTT
ACK arrives, send next packet, t = RTT + L / R
first packet bit arrives
last packet bit arrives, send ACK
last bit of 2nd packet arrives, send ACK
last bit of 3rd packet arrives, send ACK

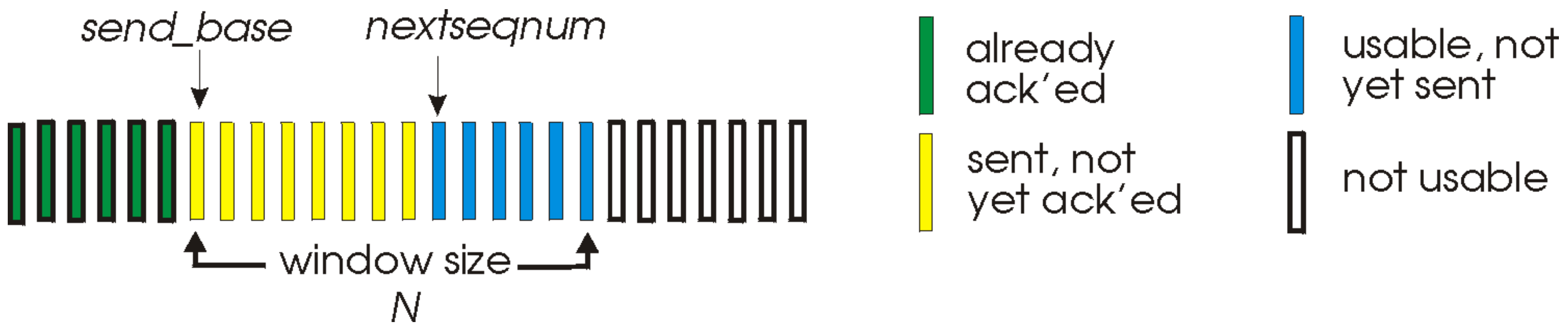
Increase utilization by a factor of 3!

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

Go-Back-N

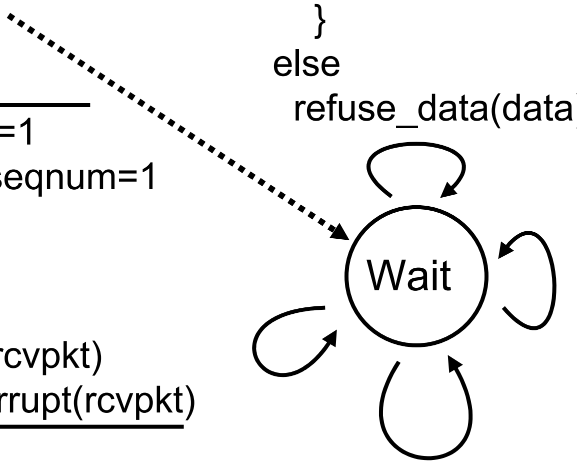
Sender:

- K-bit seq # in pkt header
- "Window" of up to N, consecutive unack'ed pkts allowed



- ACK(n): ACKs all pkts up to, including seq # n - "cumulative ACK"
  - May deceive duplicate ACKs (see receiver)
- Timer for each in-flight pkt
- *Timeout(n):* Retransmit pkt n and all higher seq # pkts in window

GBN: Sender extended FSM



```
rdt_send(data)
______________________________
if (nextseqnum < base+N) {
   sndpkt[nextseqnum] = make_pkt(nextseqnum,data,chksum)
   udt_send(sndpkt[nextseqnum])
   if (base == nextseqnum)
      start_timer
   nextseqnum++
   }
else
  refuse_data(data)
```

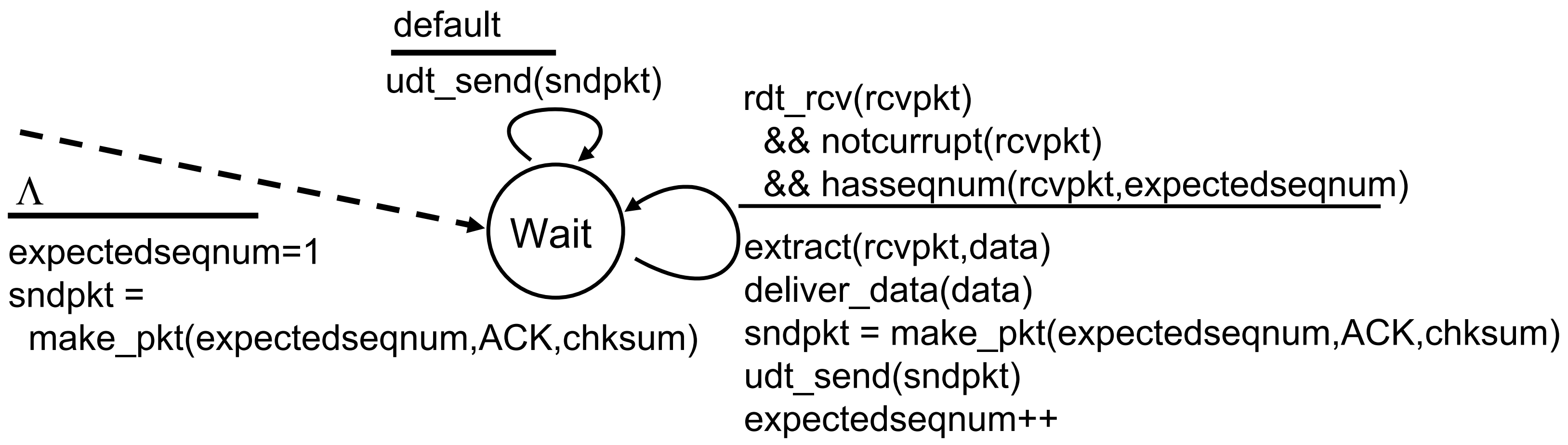
```
Λ
____________
base=1
nextseqnum=1
```

```
timeout
______________________________
start_timer
udt_send(sndpkt[base])
udt_send(sndpkt[base+1])
…
udt_send(sndpkt[nextseqnum-1])
```

```
rdt_rcv(rcvpkt)
  && corrupt(rcvpkt)
______________________
```

```
rdt_rcv(rcvpkt) &&
  notcorrupt(rcvpkt)
______________________
base = getacknum(rcvpkt)+1
If (base == nextseqnum)
   stop_timer
  else
   start_timer
```

GBN: Receiver extended FSM

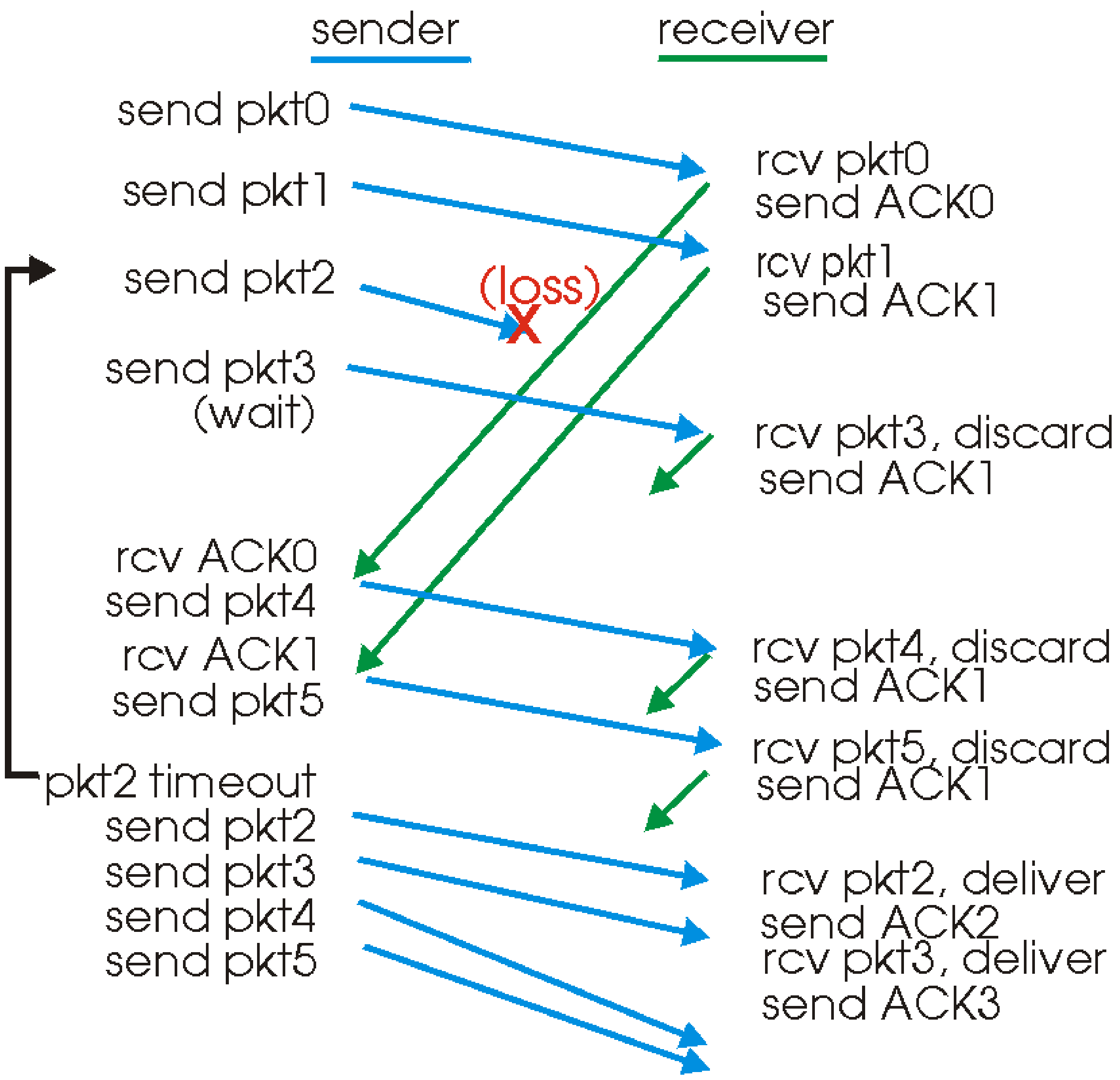


ACK-only: always send ACK for correctly-received pkt with highest *in-order* seq #

- 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 #

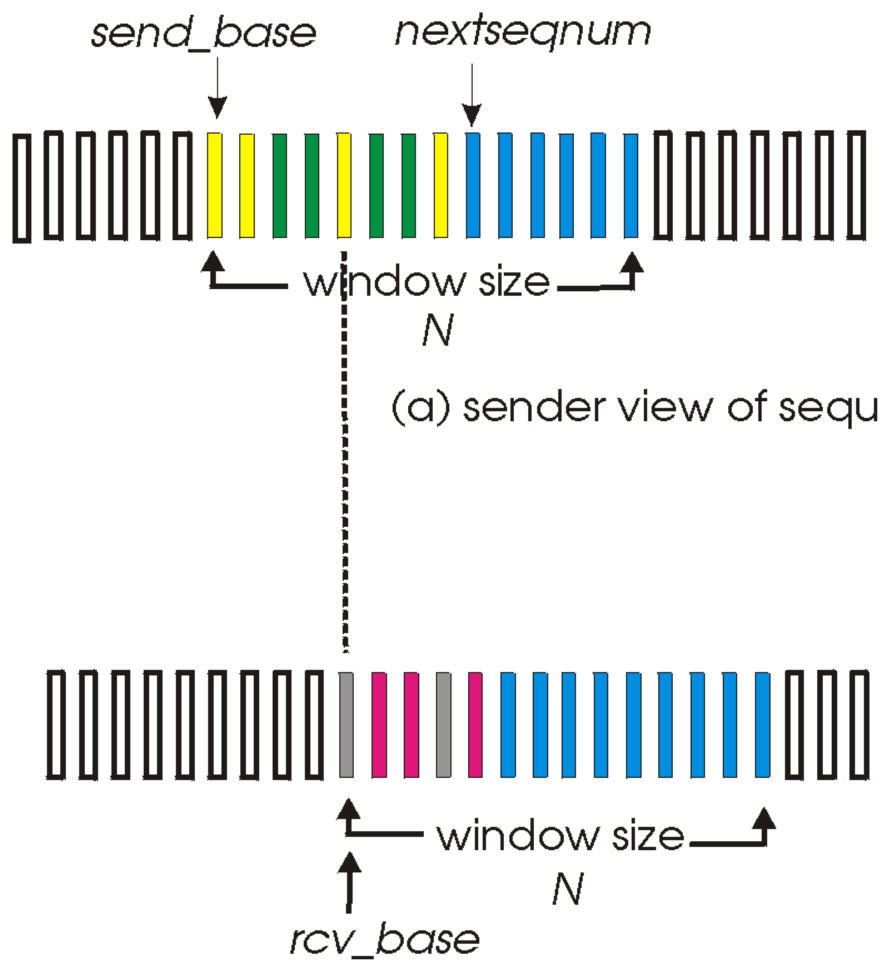
GBN in action



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

🟩	already ack'ed	🟦	usable, not yet sent
🟨	sent, not yet ack'ed	▯	not usable

🟥	out of order (buffered) but already ack'ed	🟦	acceptable (within window)
▨	Expected, not yet received	▯	not usable

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

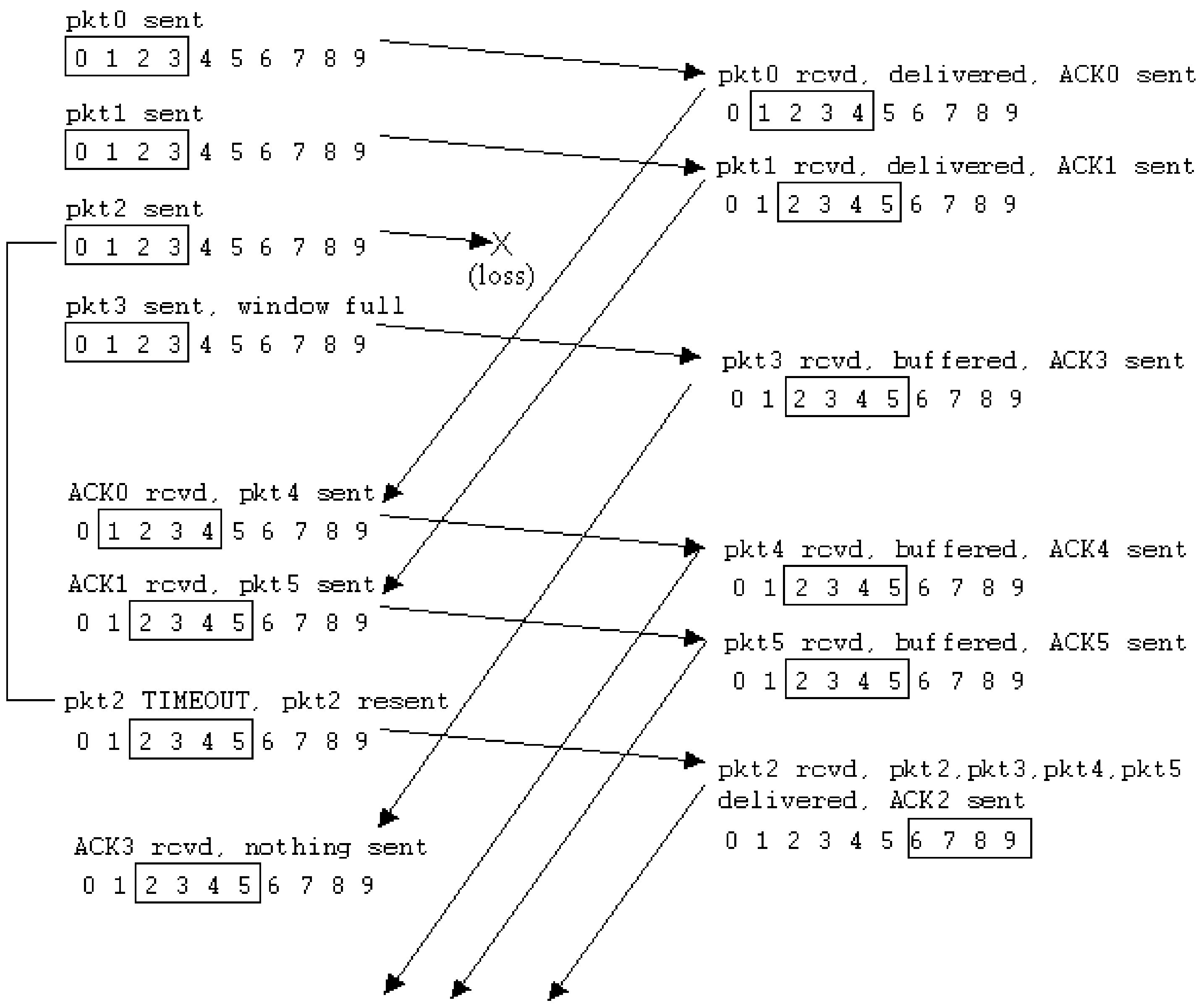
Pkt n in [rcvbase-N,rcvbase-1]

- ACK(n)

Otherwise:

- Ignore

Selective repeat in action



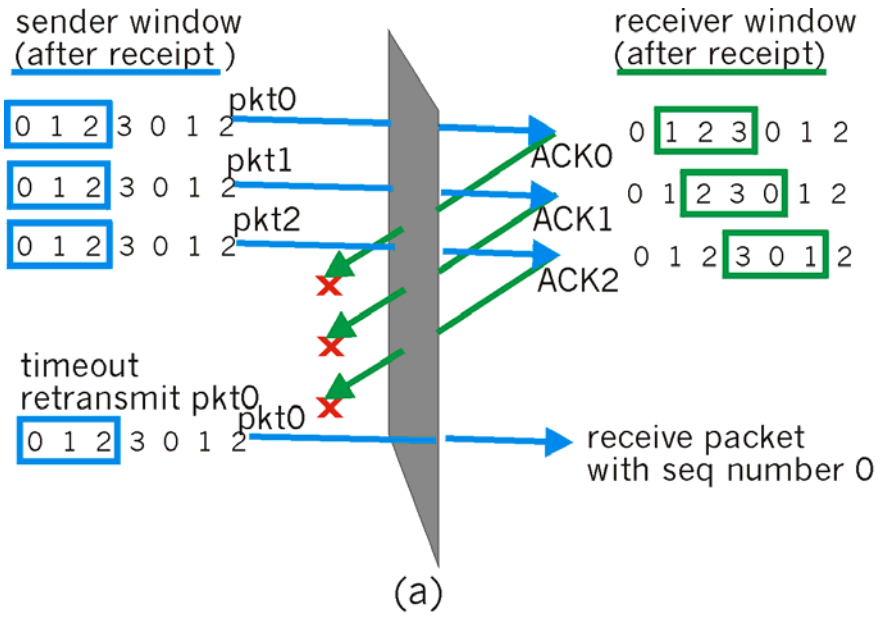
Selective repeat: dilemma

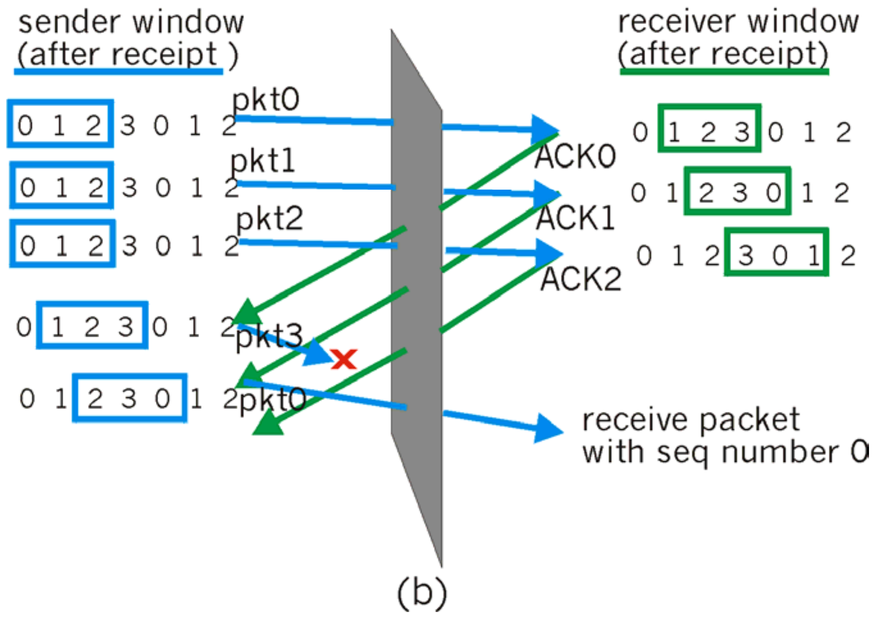
Example:

- 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?


sender window (after receipt)
receiver window (after receipt)
pkt0
pkt1
pkt2
ACK0
ACK1
ACK2
timeout retransmit pkt0
pkt0
receive packet with seq number 0
(a)


sender window (after receipt)
receiver window (after receipt)
pkt0
pkt1
pkt2
ACK0
ACK1
ACK2
pkt3
pkt0
receive packet with seq number 0
(b)