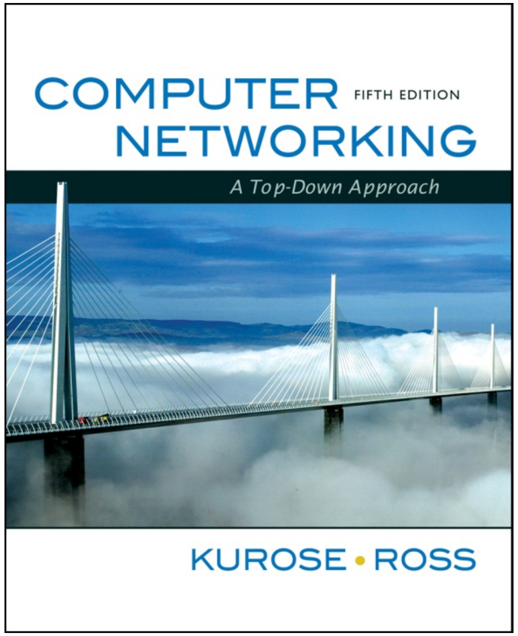
# Chapter 3
# Transport Layer

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*Computer Networking: A Top Down Approach ,*
6th edition.
Jim Kurose, Keith Ross
Addison-Wesley, March 2012.

# 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
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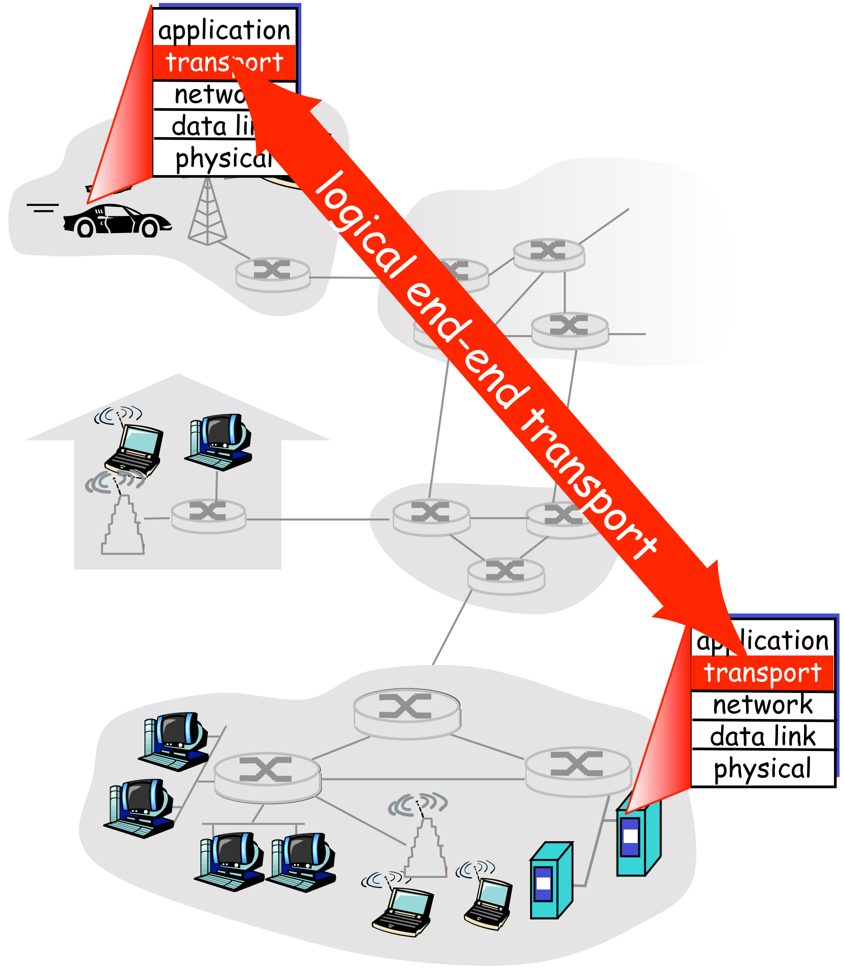
# Transport services and protocols

- 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 computers
- *transport layer:* logical communication between processes
  - relies on, enhances, network layer services

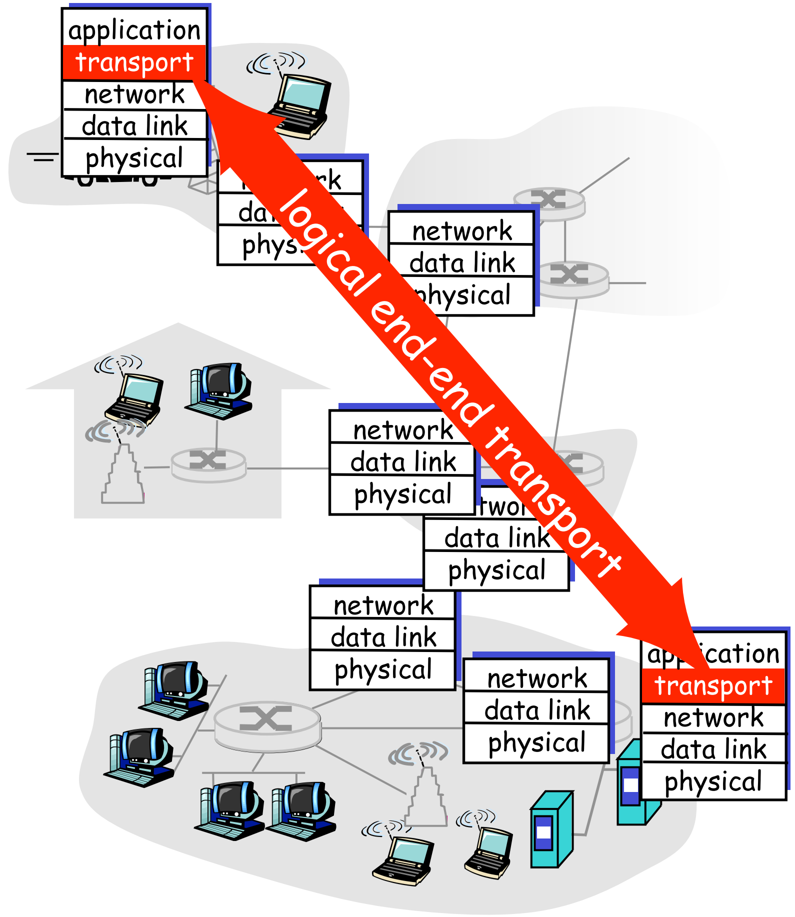
## Household analogy:

*12 kids sending letters to 12 kids*

- processes = kids
- app messages = letters in envelopes
- hosts = houses
- transport protocol = Ann and Bill (parents)
- 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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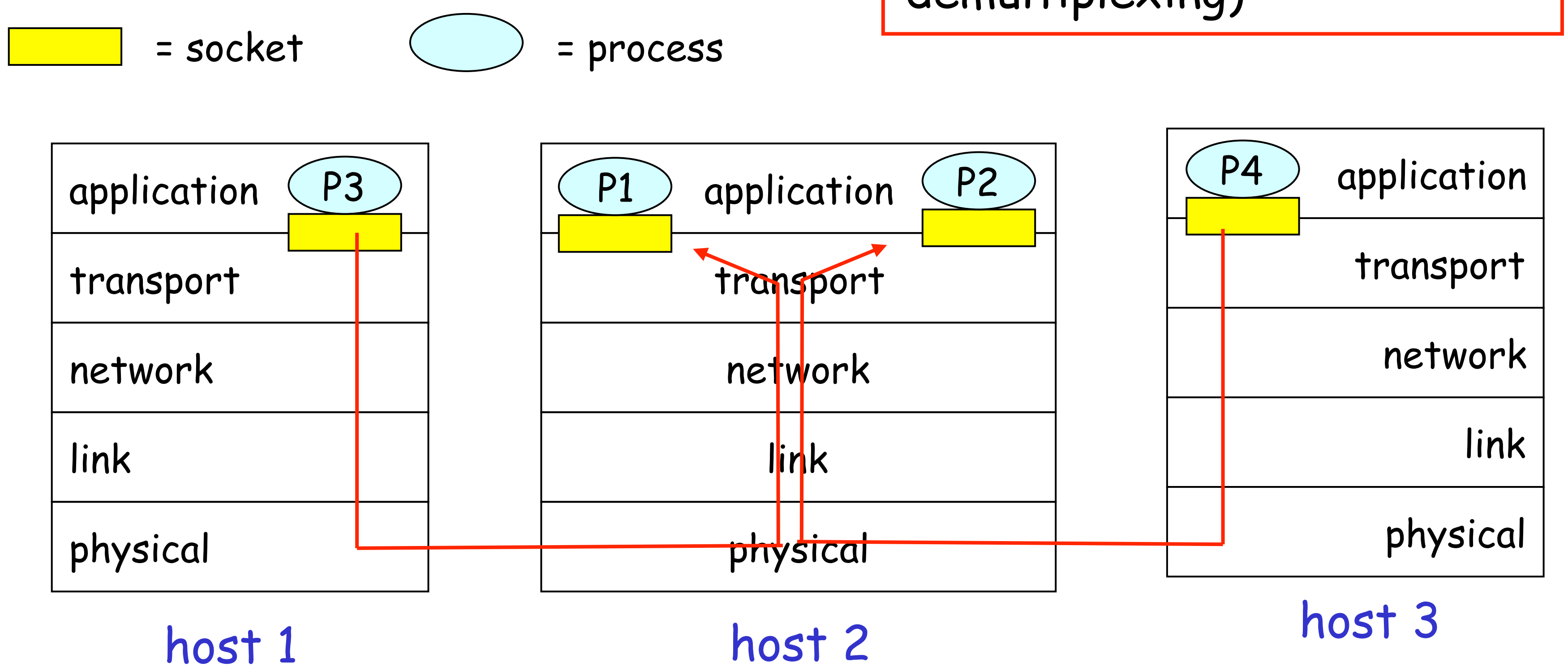
# Multiplexing/demultiplexing

**Demultiplexing at rcv host:**

delivering received segments to correct socket

**Multiplexing at send host:**

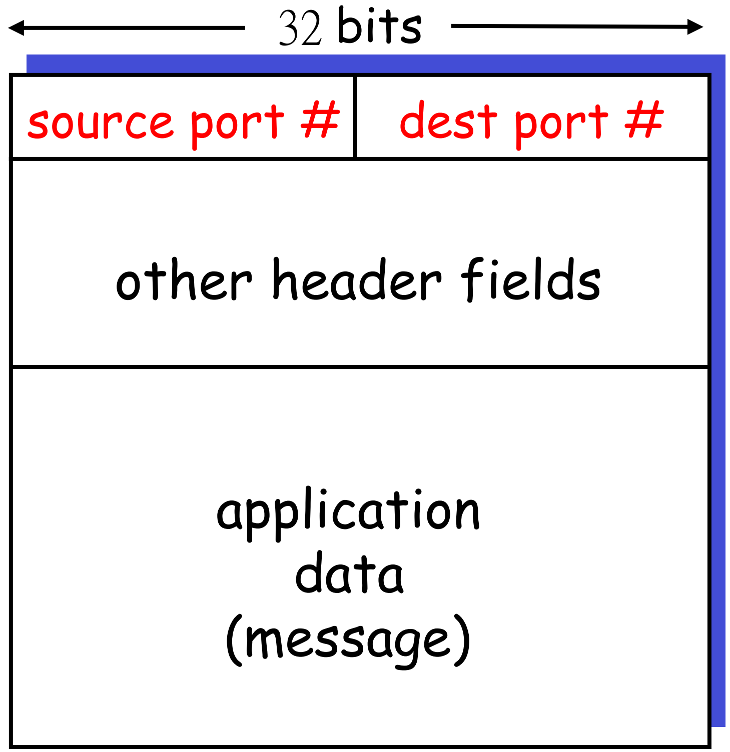
gathering data from multiple sockets, enveloping data with header (later used for demultiplexing)



# Quiz Time!

# 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 (recall: well-known port numbers for specific applications)
- host uses IP addresses & port numbers to direct segment to appropriate socket




TCP/UDP segment format

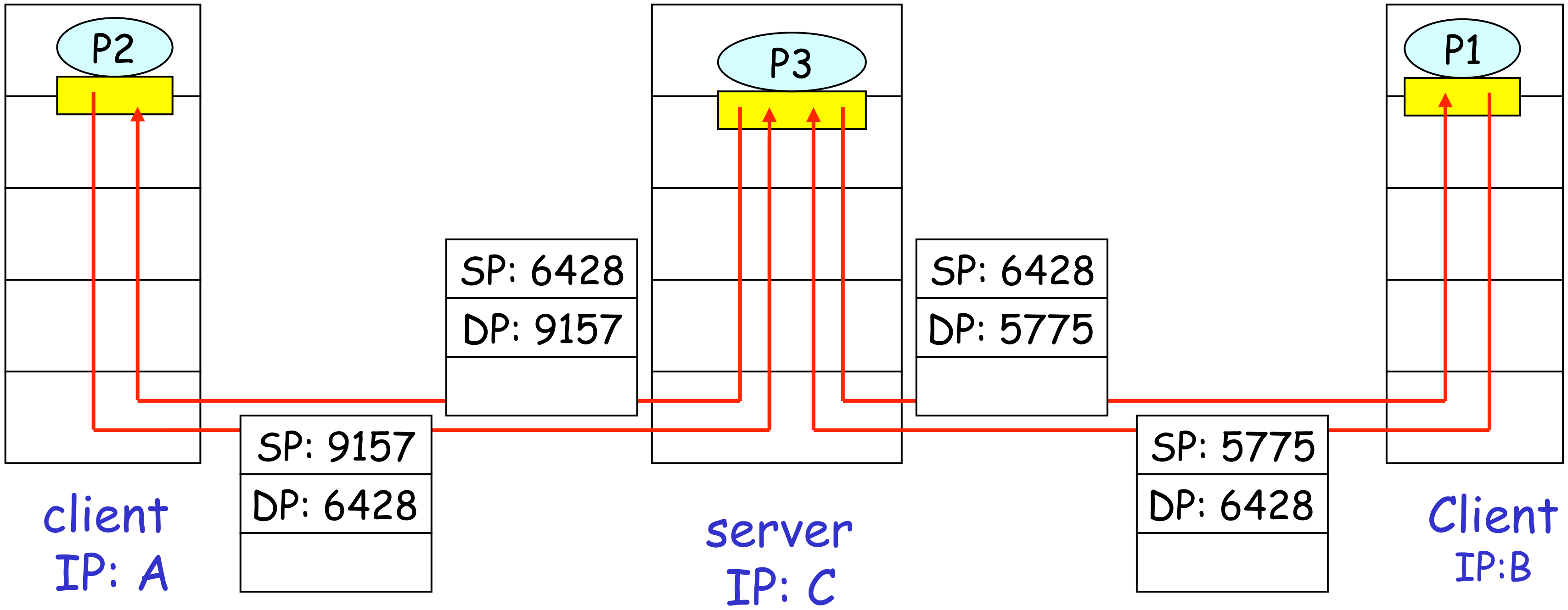
# Connectionless demultiplexing

- 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)



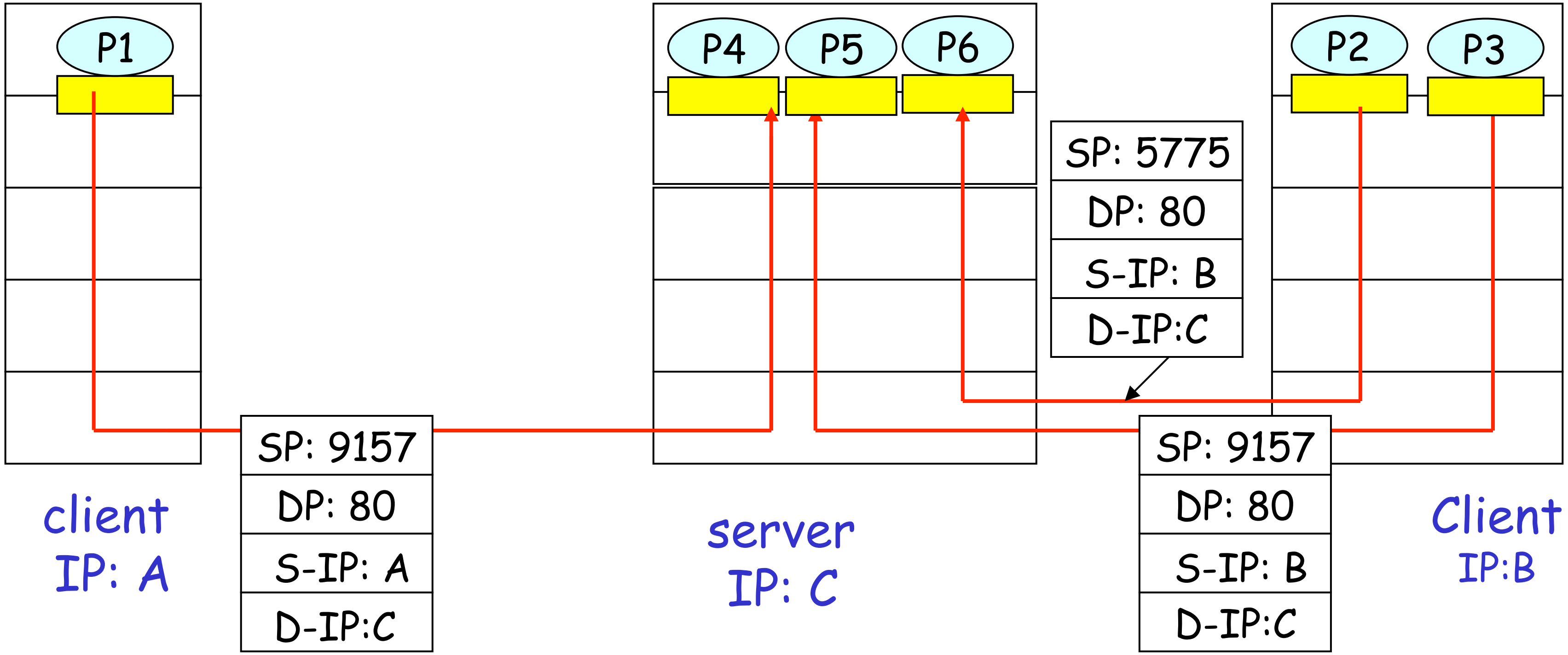

SP provides "return address"

# Connection-oriented demux

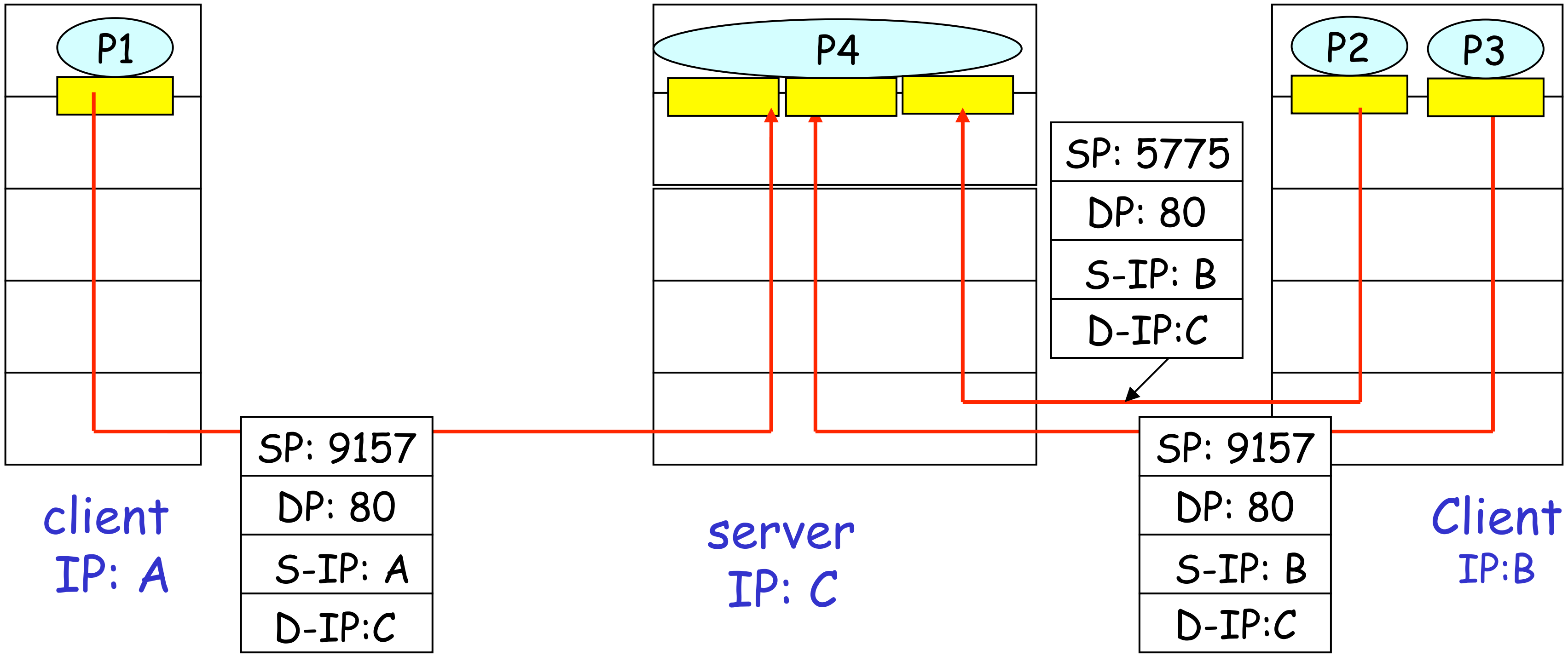
- TCP socket identified by 4-tuple:
  - source IP address
  - source port number
  - dest IP address
  - dest port number
- recv host uses all four values to direct segment to appropriate socket

- 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?**

# UDP: User Datagram Protocol [RFC 768]

- "no frills," "bare bones" Internet transport protocol
- "best effort" service, UDP segments may be:
  - lost
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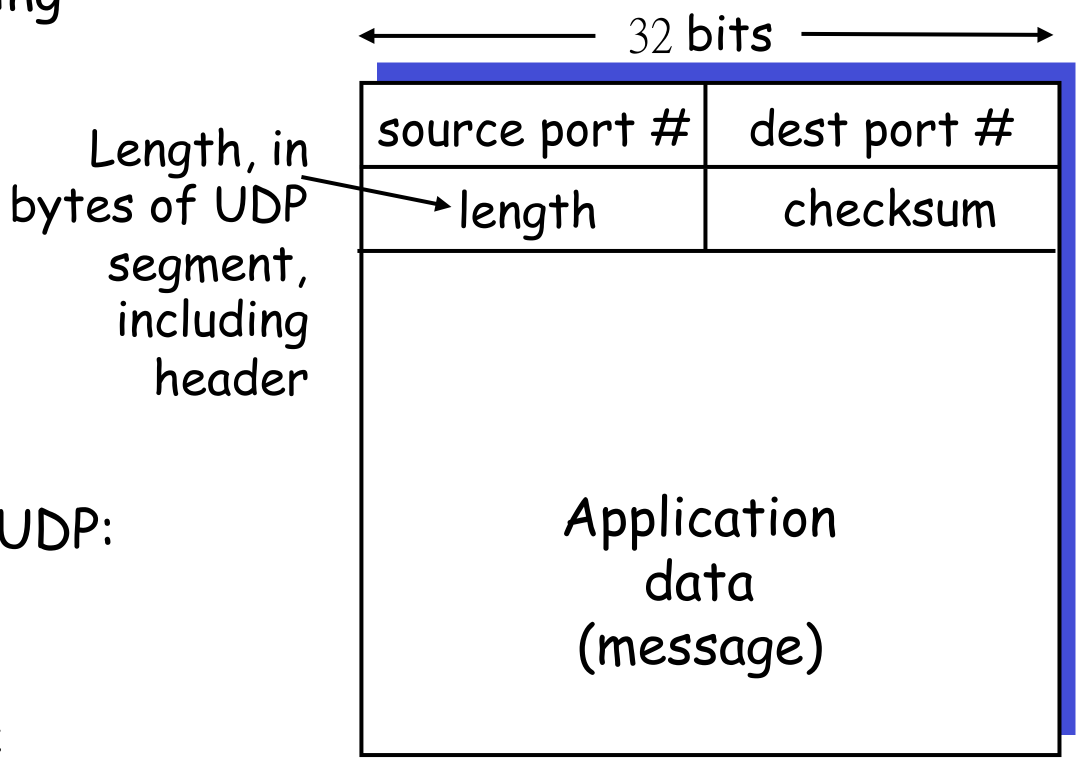
**Why is there a UDP?**

- 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 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

| | | |
|---|---|---|
| | | 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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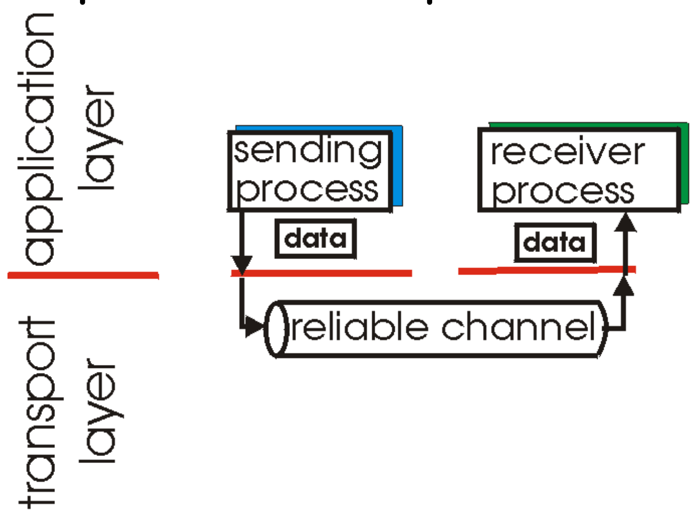
# Principles of Reliable data transfer

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




(a) provided service

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

# Principles of Reliable data transfer

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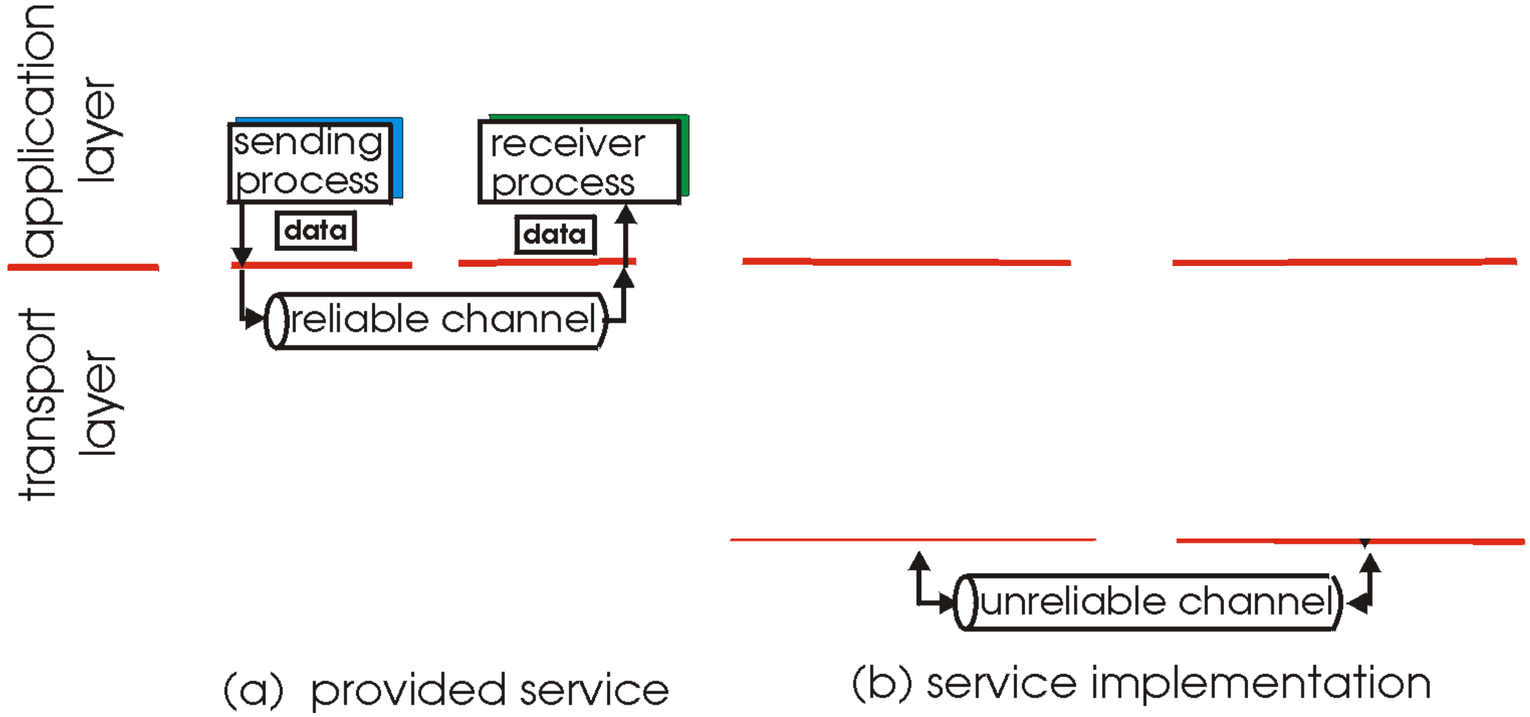


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# Principles of Reliable data transfer

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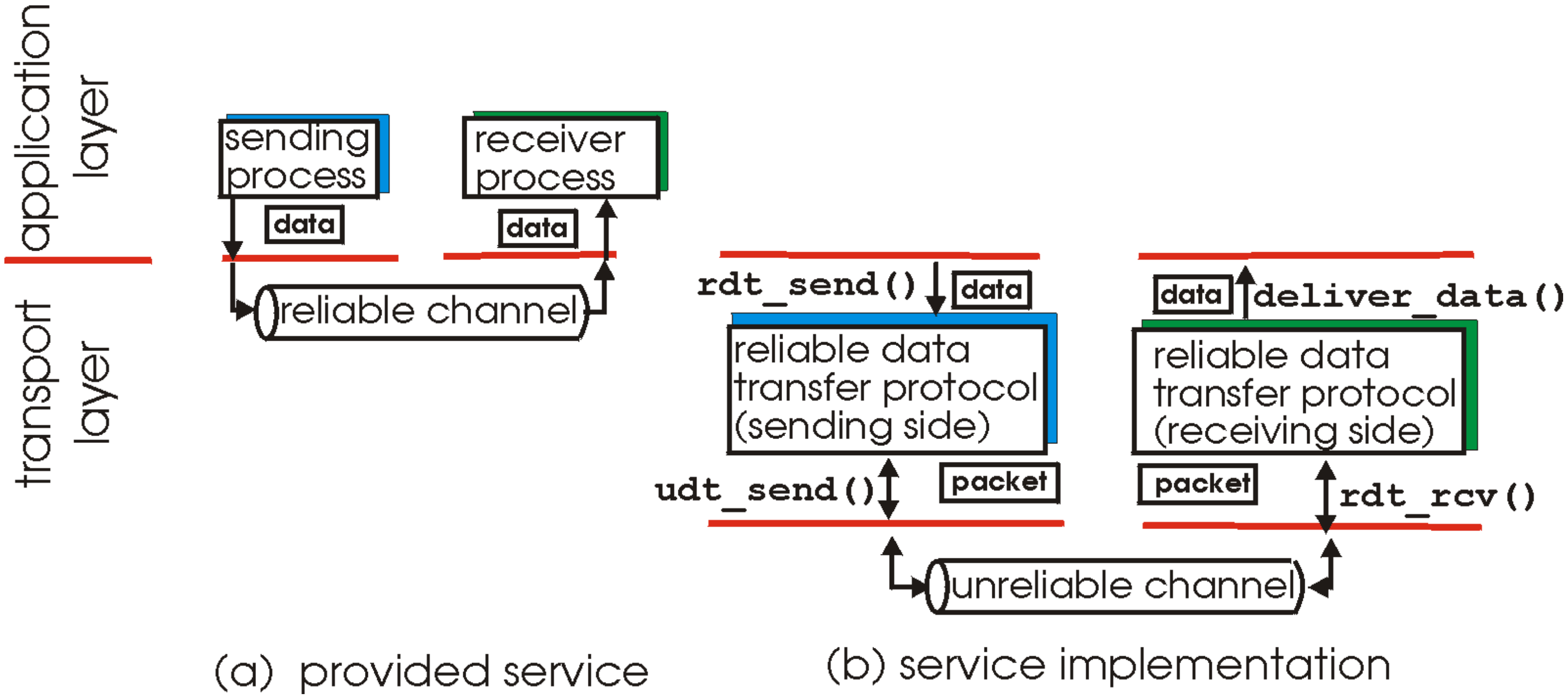
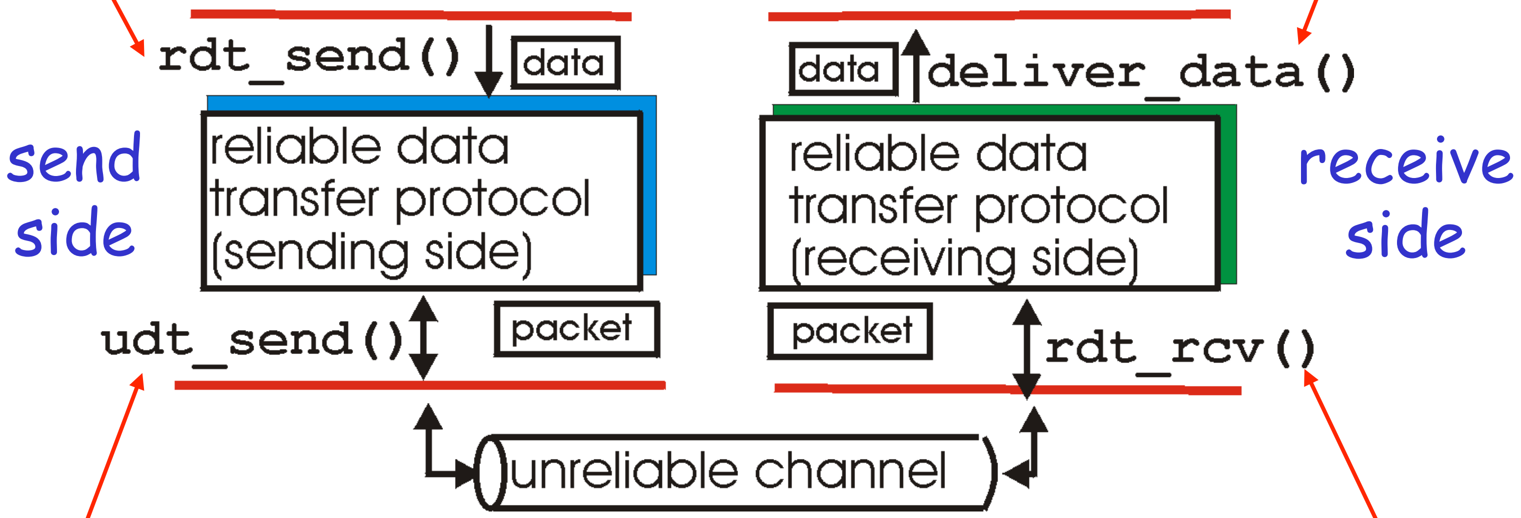



- characteristics of unreliable channel will determine complexity of reliable data transfer protocol (rdt)

# 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 layer



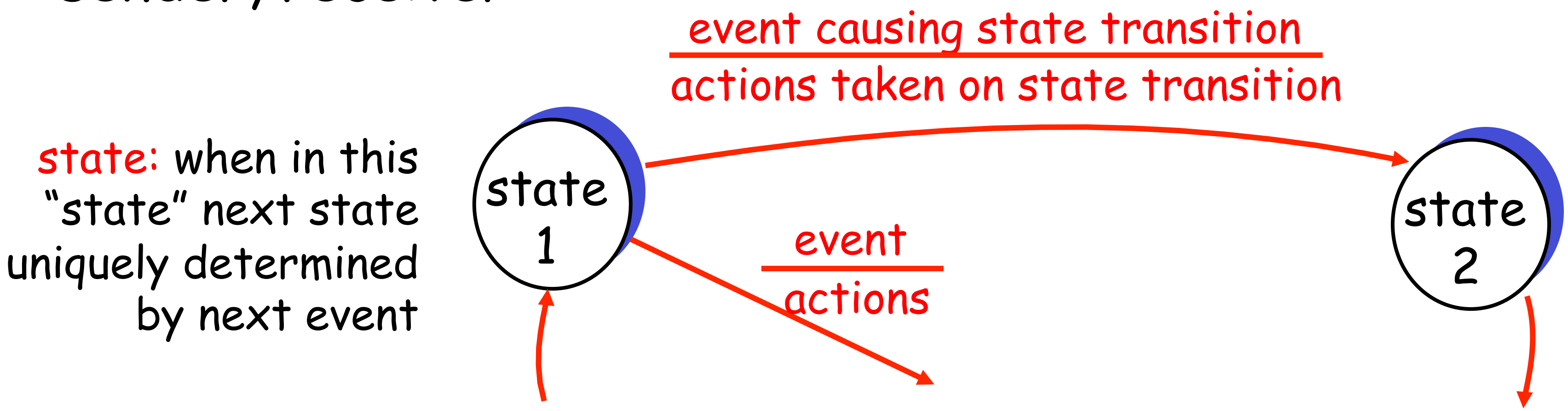

**udt_send():** called by rdt, to transfer packet over unreliable channel to receiver

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

# Reliable data transfer: getting started

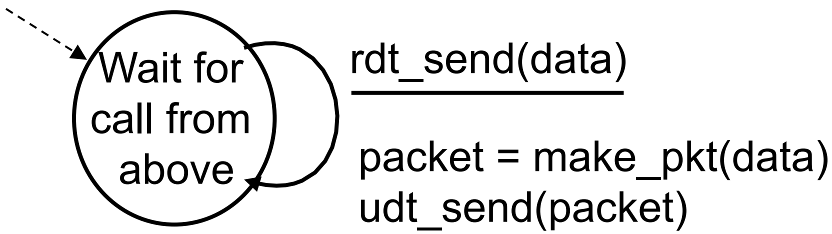
We'll:

- 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

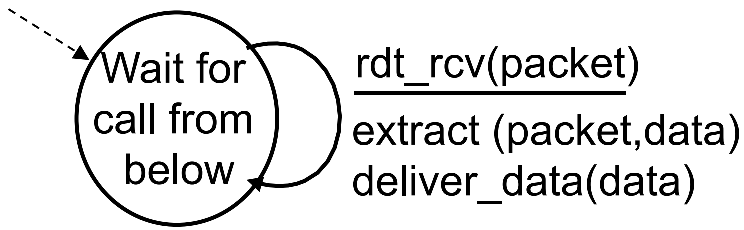


# Rdt1.0: reliable transfer over a reliable channel

- 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




sender

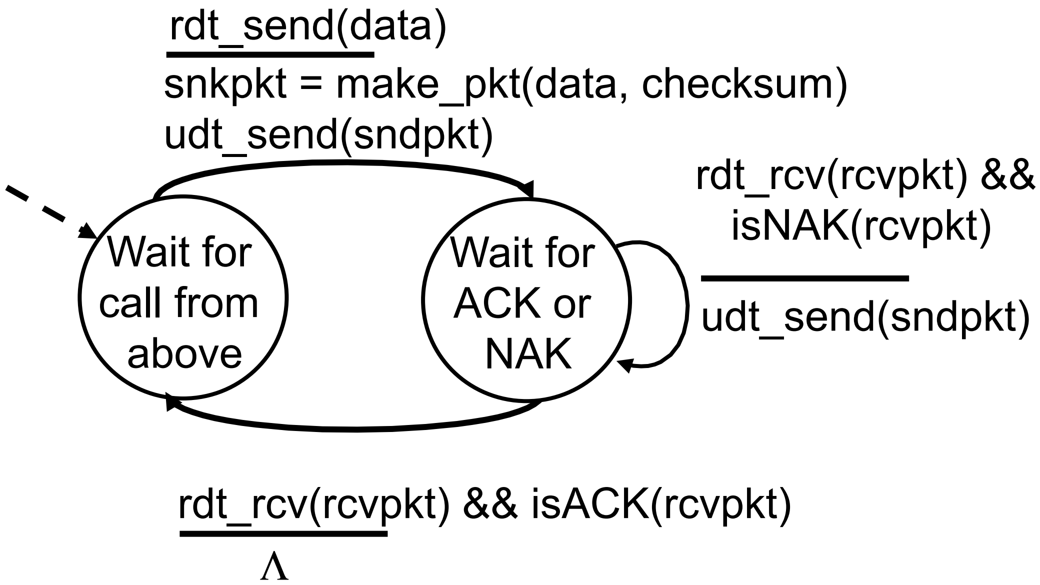



receiver

# Rdt2.0: channel with bit errors

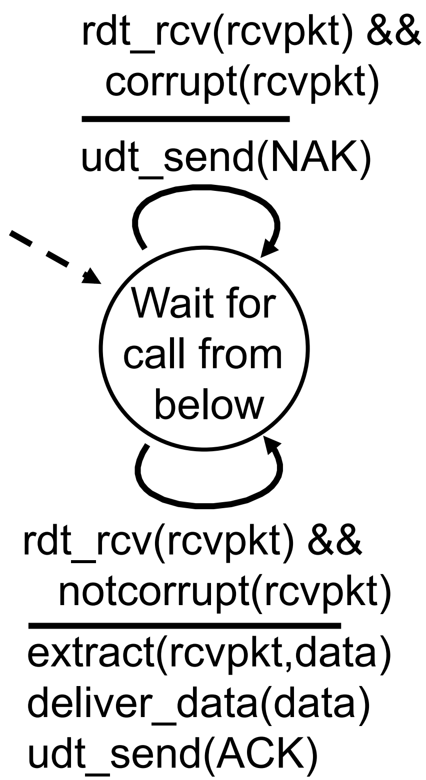
- underlying channel may flip bits in packet
  - Q: how to detect bit errors?
  - recall: UDP checksum to detect bit errors
- *the* question: how to recover from errors:
  - *acknowledgements (ACKs):* receiver explicitly tells sender that packet received OK
  - *negative acknowledgements (NAKs):* receiver explicitly tells sender that packet 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 messages (ACK,NAK) receiver->sender

# rdt2.0: FSM specification

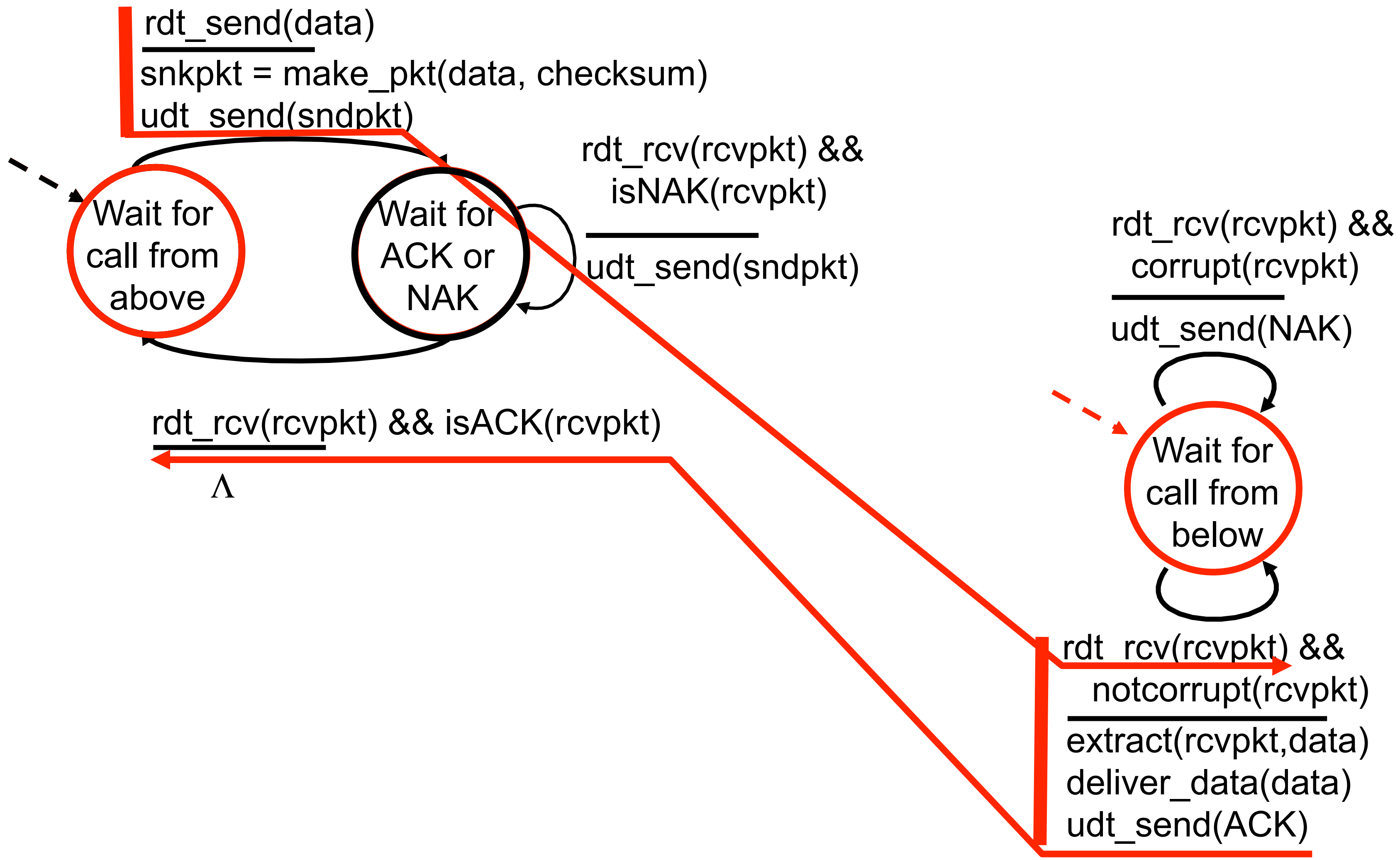



sender

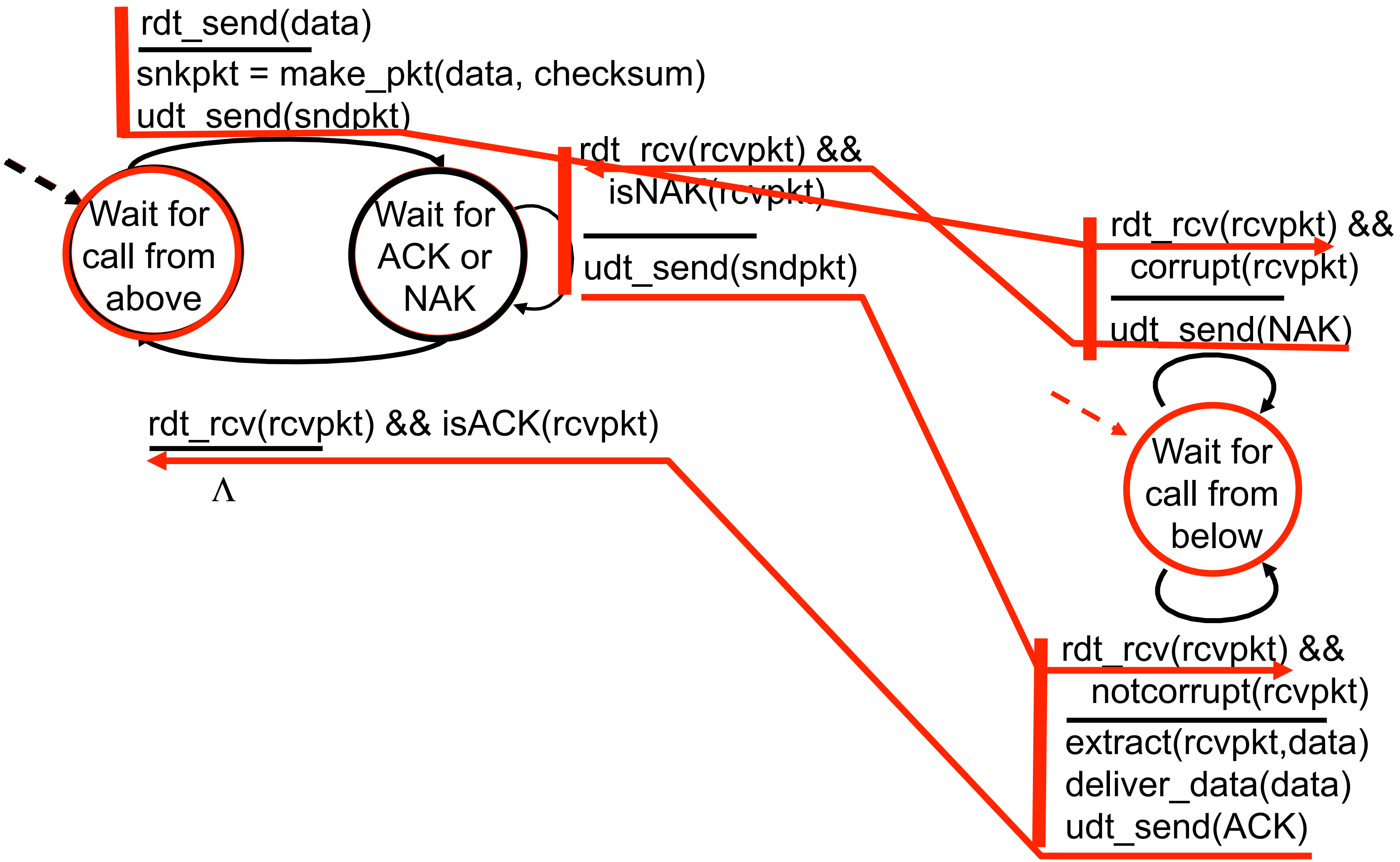
receiver



# rdt2.0: operation with no errors



# rdt2.0: error scenario



# rdt2.0 has a fatal flaw!

What happens if ACK/NAK corrupted?

- sender doesn't know what happened at receiver!

What to do?

- Q?

# rdt2.0 has a fatal flaw!

What happens if ACK/NAK corrupted?

- sender doesn't know what happened at receiver!

What to do?

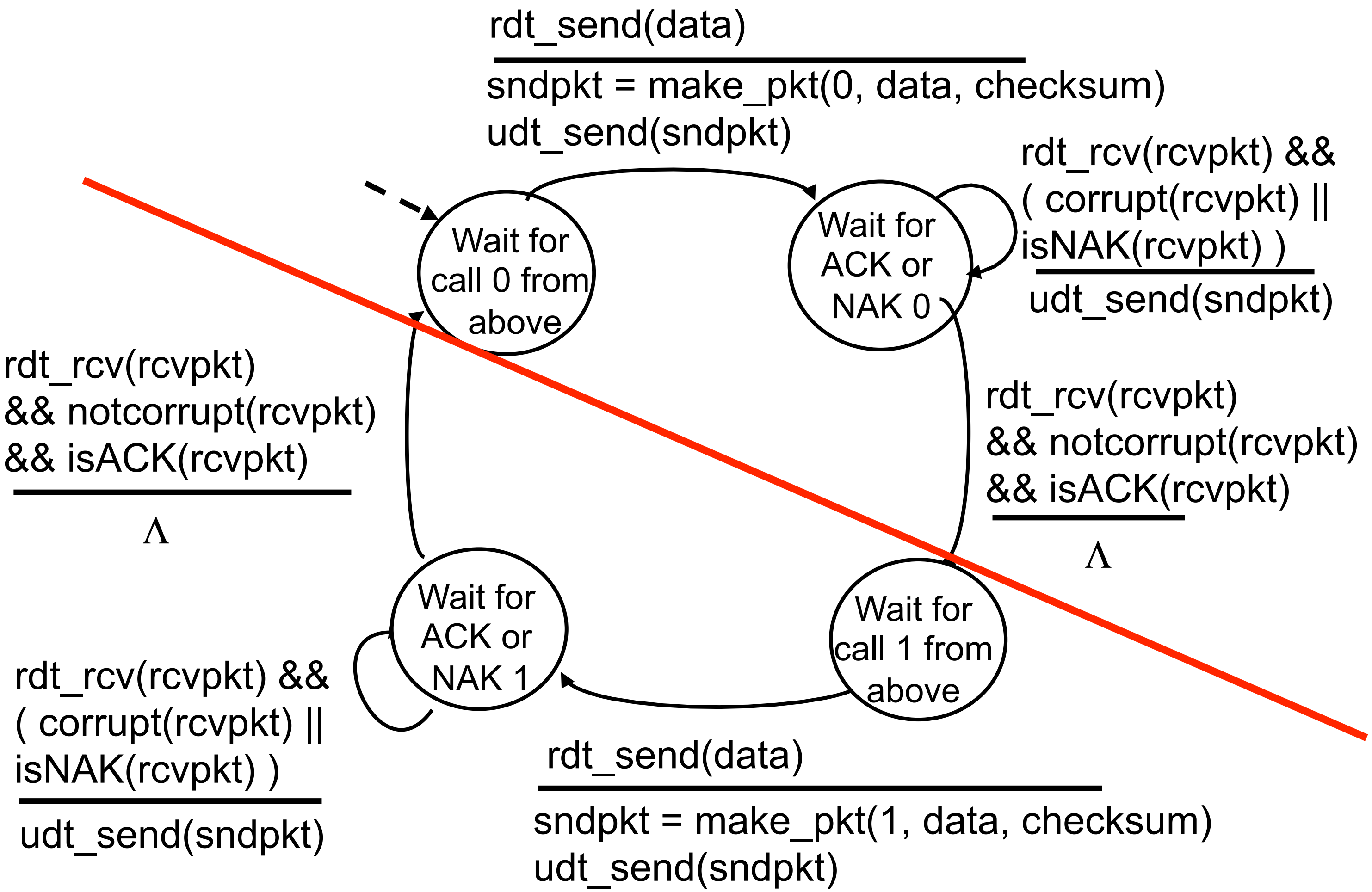
- sender ACKs/NAKs receiver's ACK/NAK? What if this sender ACK/NAK corrupted?... (see an end?)
- retransmit anyway, but this might cause retransmission of correctly received packet!

Handling duplicates:

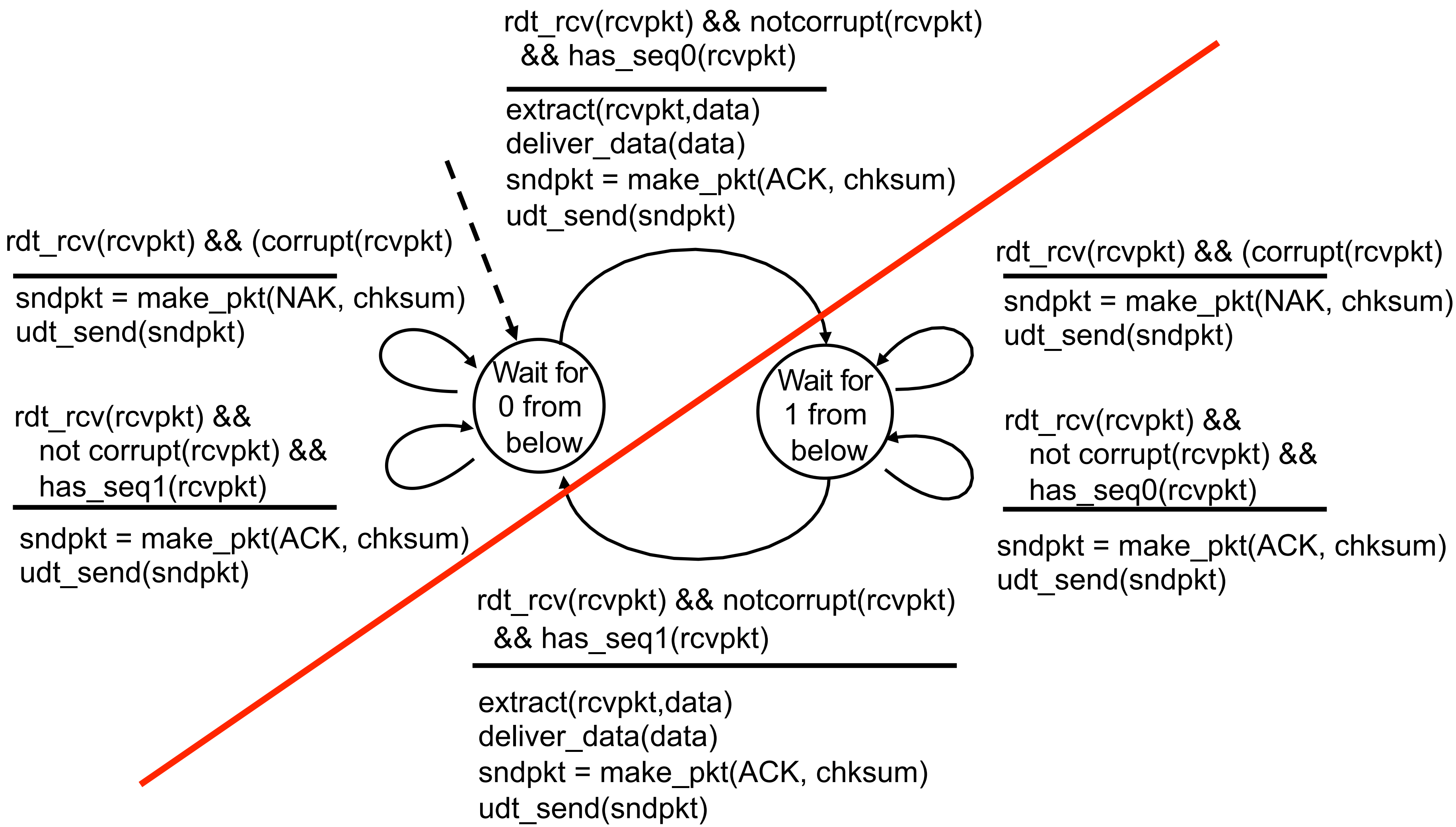
- sender adds *sequence number* to each packet
- sender retransmits current packet if ACK/NAK garbled
- receiver discards (doesn't deliver up) duplicate packet

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

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



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



# Quiz Time!

# 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 0 or 1 sequence #

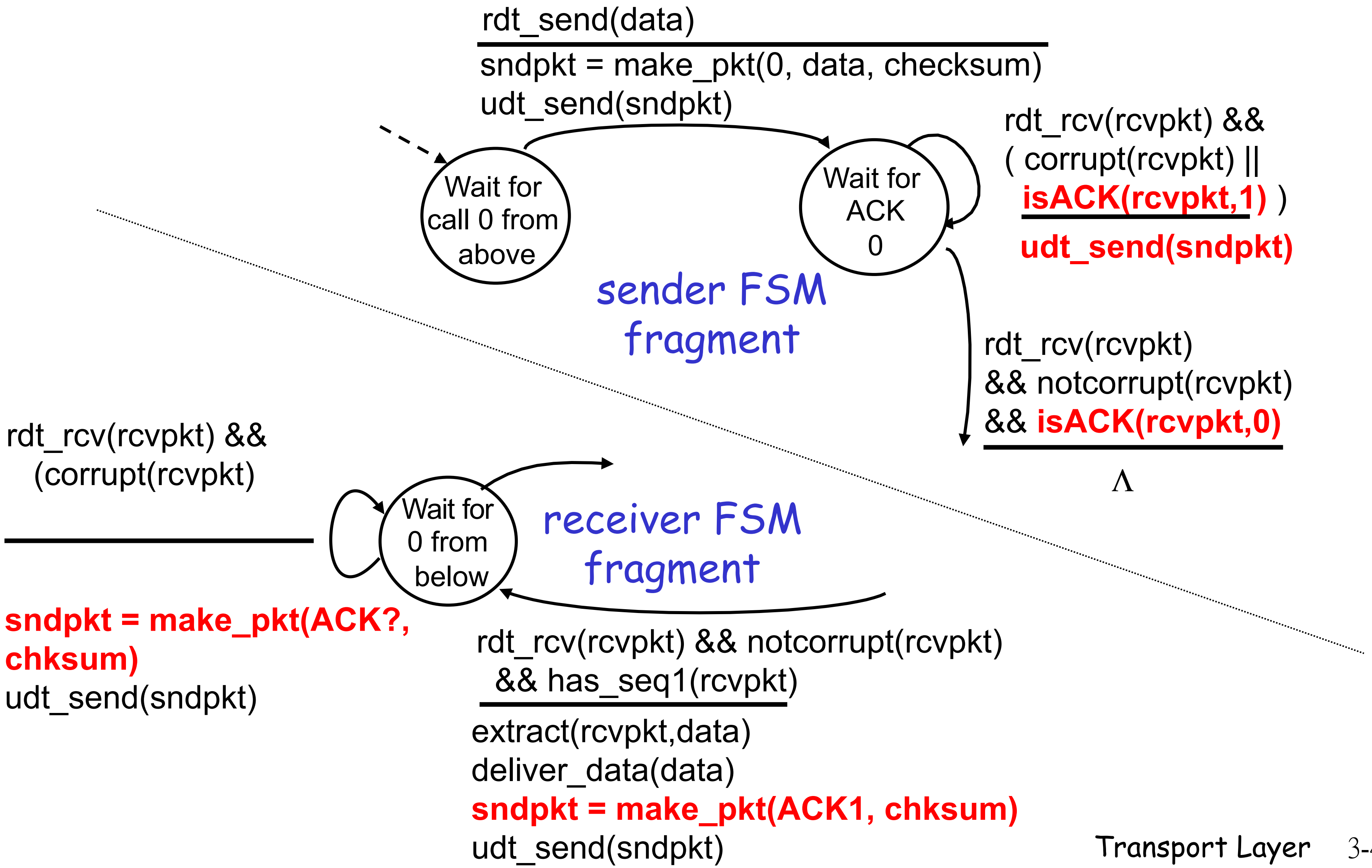
**Receiver:**

- must check if received packet is duplicate
  - state indicates whether 0 or 1 is expected packet sequence #
- note: receiver can *not* know if its last ACK/NAK received OK at sender

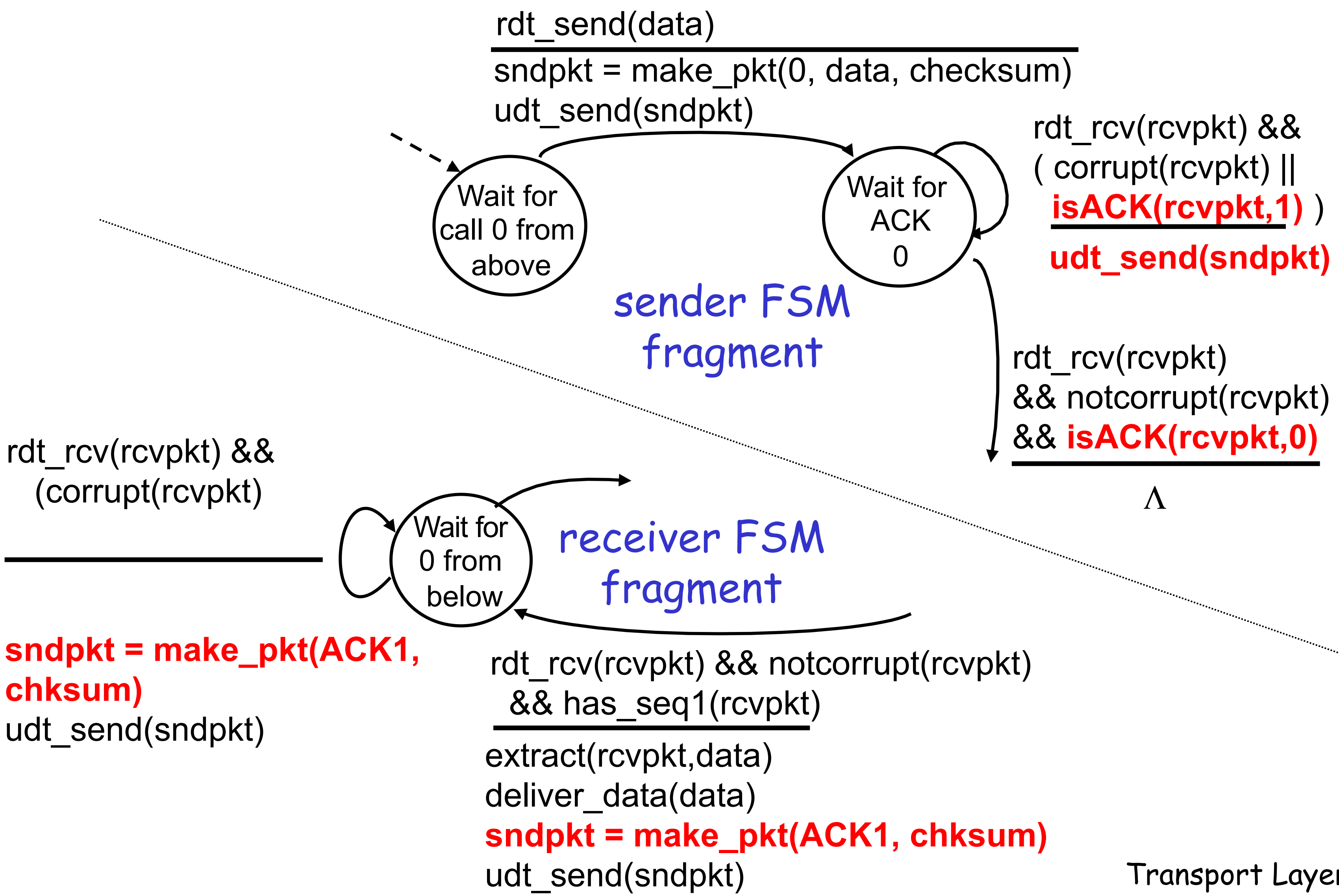
# rdt2.2: a NAK-free protocol

- same functionality as rdt2.1, using ACKs only
- instead of NAK, receiver sends ACK for last packet received OK
  - receiver must *explicitly* include sequence # of packet being ACKed
- duplicate ACK at sender results in same action as NAK: *retransmit current packet*

# rdt2.2: sender, receiver fragments



# rdt2.2: sender, receiver fragments



# Quiz Time!

# 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

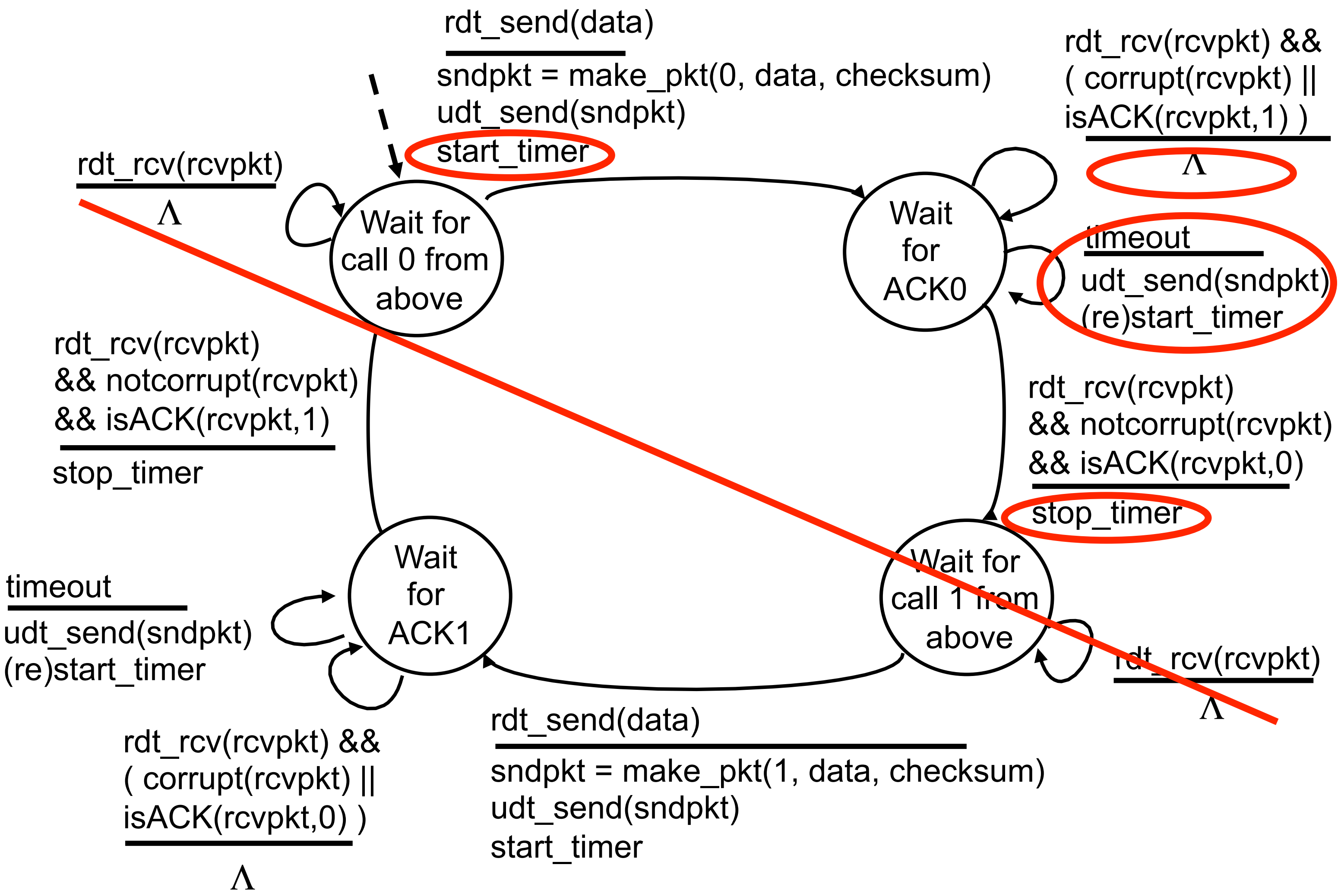
**Q:** how to deal with loss?

- sender waits until certain data or ACK lost, then retransmits
- yuck: drawbacks?

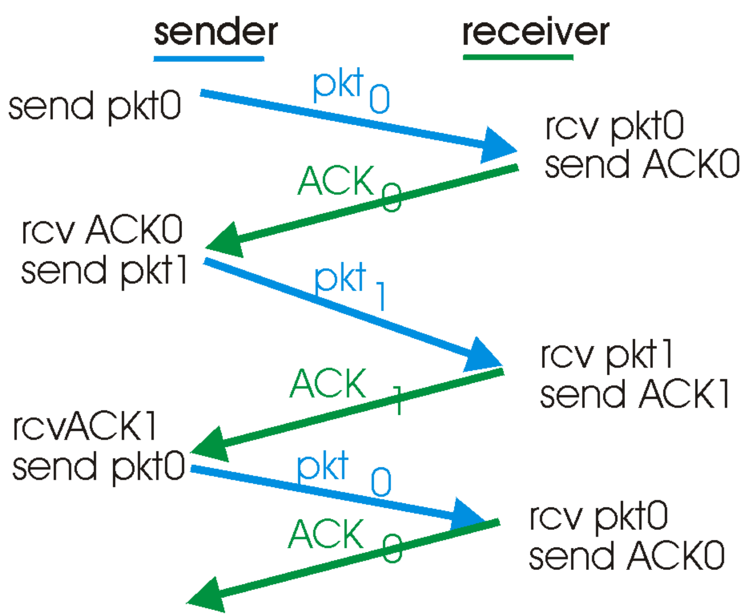
**Approach:** sender waits "reasonable" amount of time for ACK

- 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

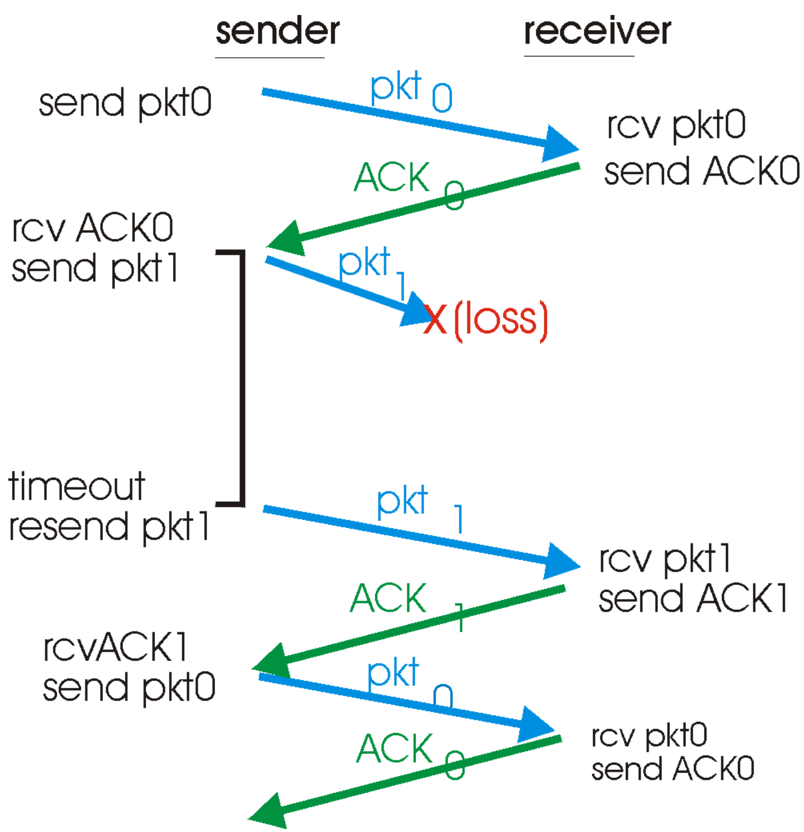
# rdt3.0 sender



# rdt3.0 in action

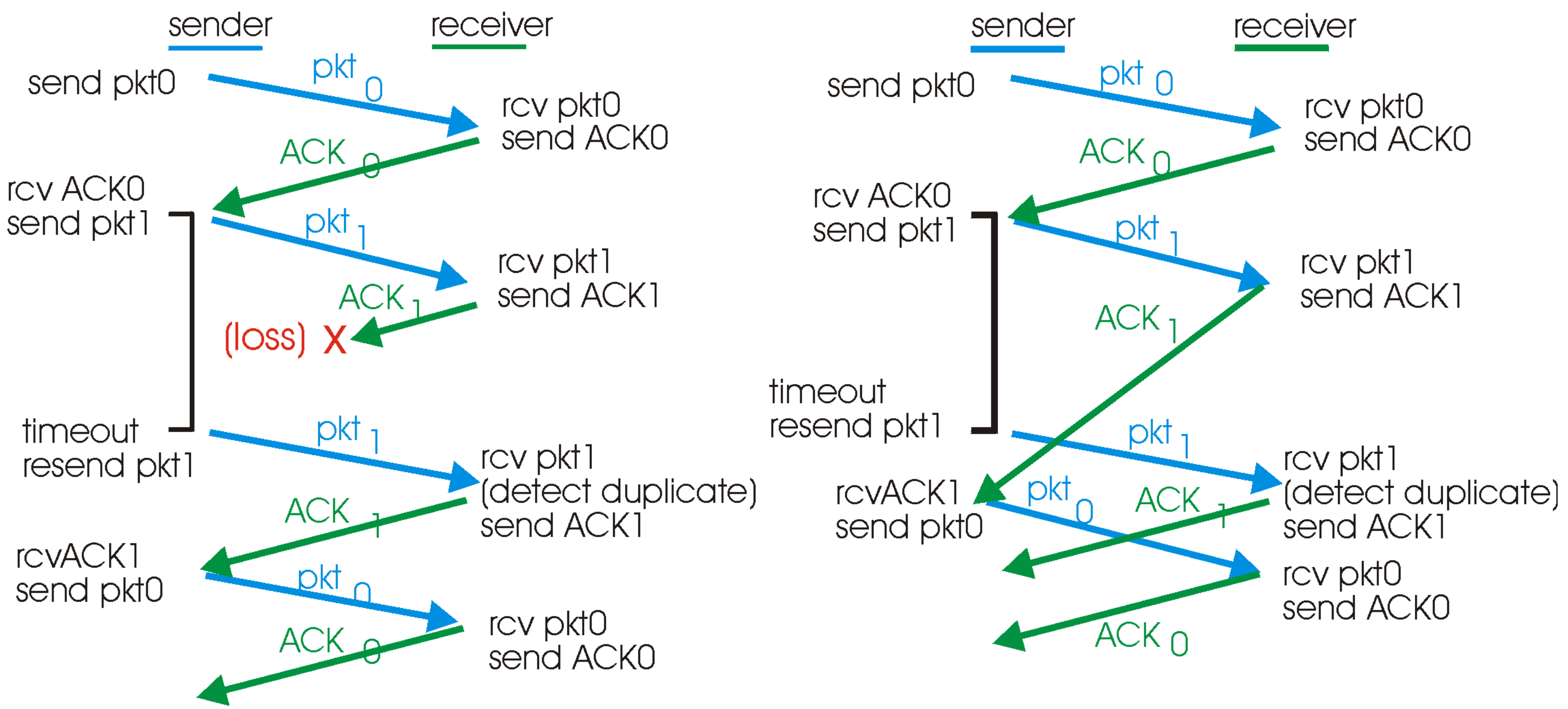



(a) operation with no loss




(b) lost packet

# rdt3.0 in action




(c) lost ACK

(d) premature timeout

# Performance of rdt3.0

- rdt3.0 works, but performance stinks
- example: 1 Gbps link, 15 ms e-e prop. delay, 1KB packet:

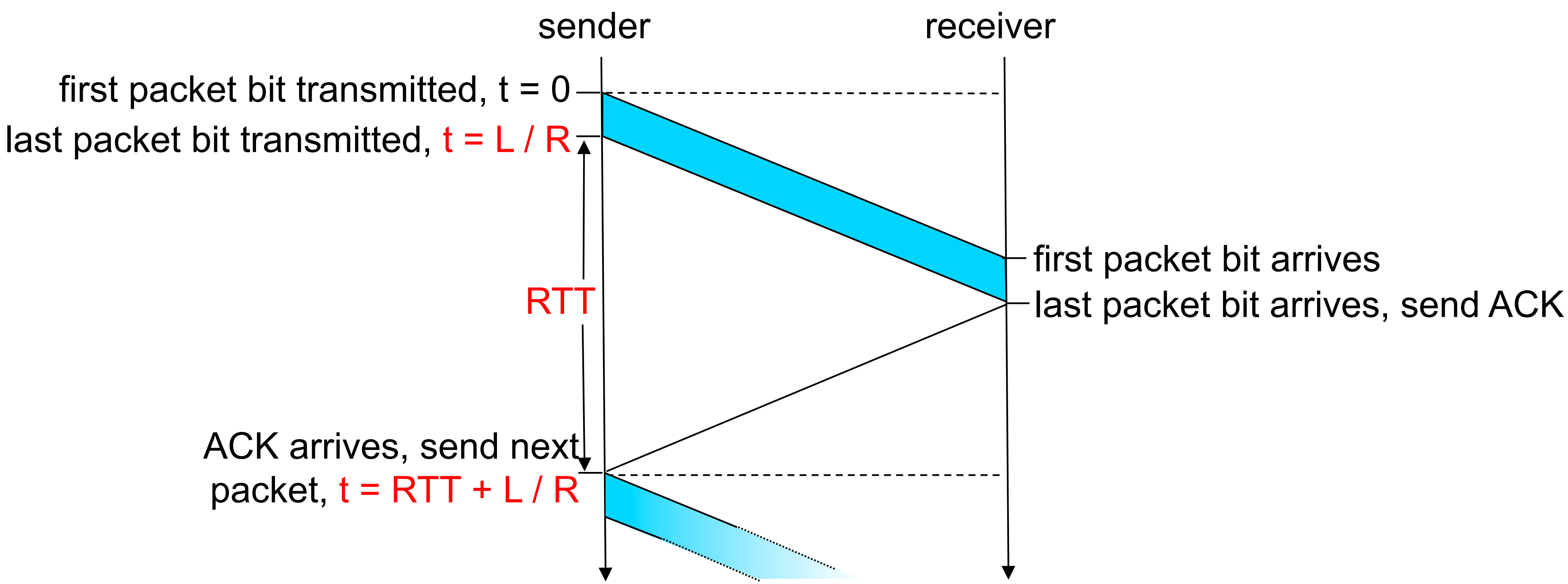
$$T_{transmit} = \frac{L \text{ (packet length in bits)}}{R \text{ (transmission rate, bps)}} = \frac{8\text{kb/pkt}}{10^{**}9 \text{ b/sec}} = 8 \text{ microsec}$$

  - $U_{sender}$: 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

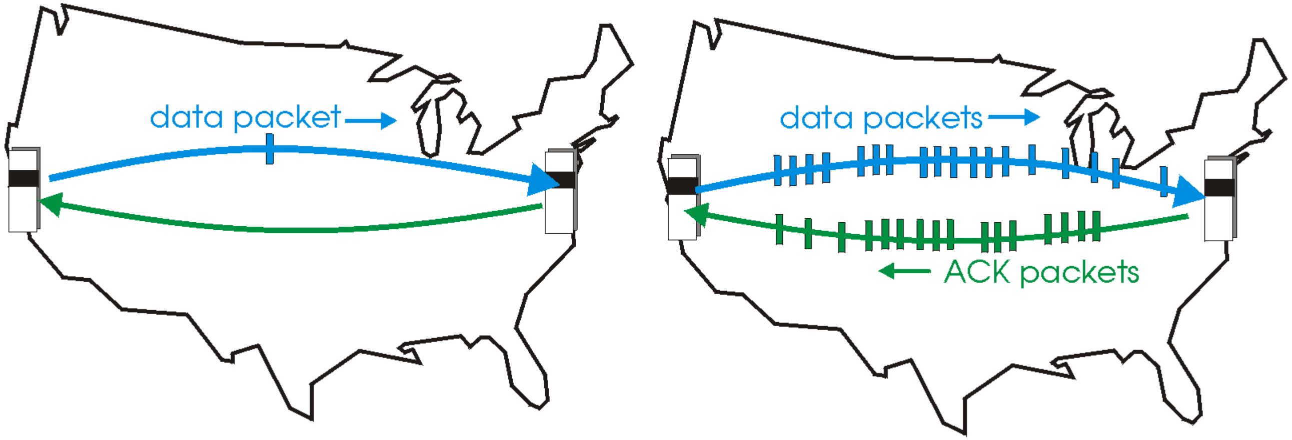



$$U_{sender} = \frac{L / R}{RTT + L / R} = \frac{.008}{30.008} = 0.00027$$

# Pipelined protocols

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

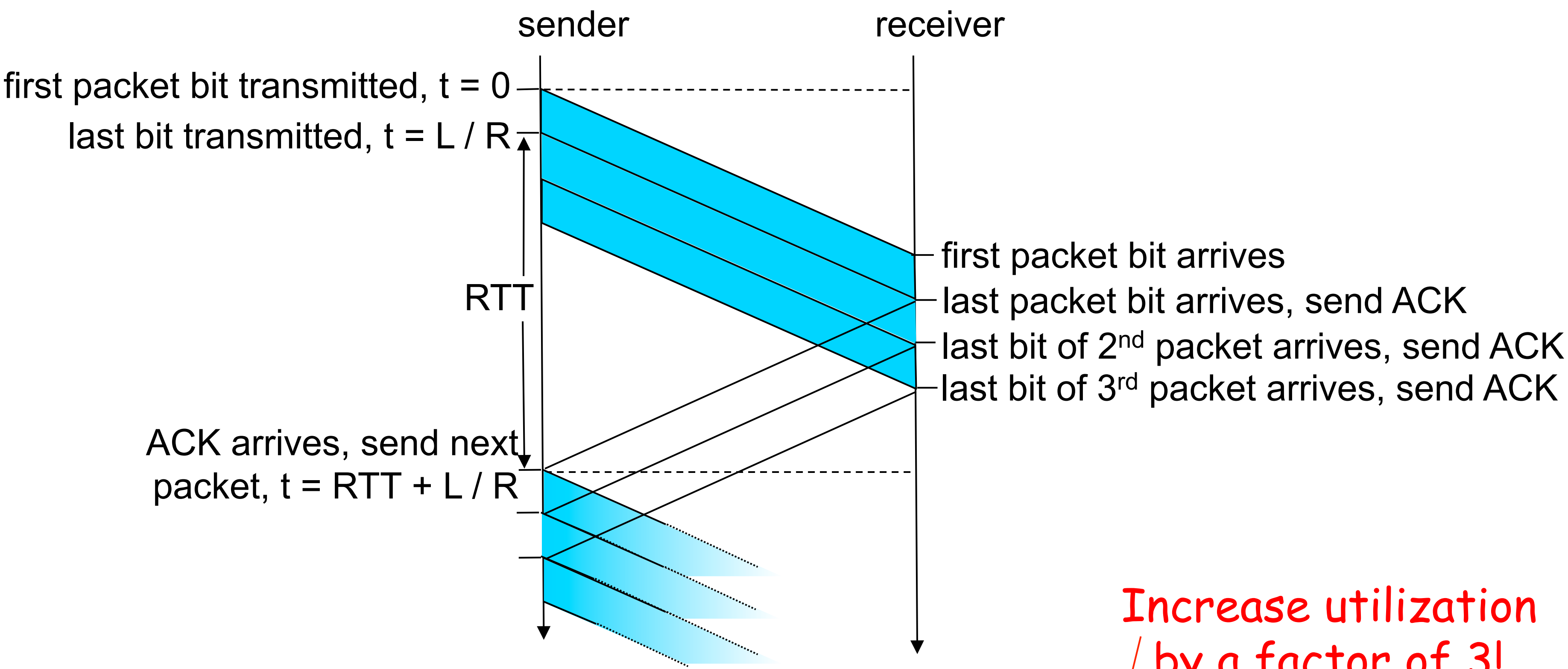
- range of sequence numbers must be increased
- buffering at sender and/or receiver




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

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

# Pipelining: increased utilization



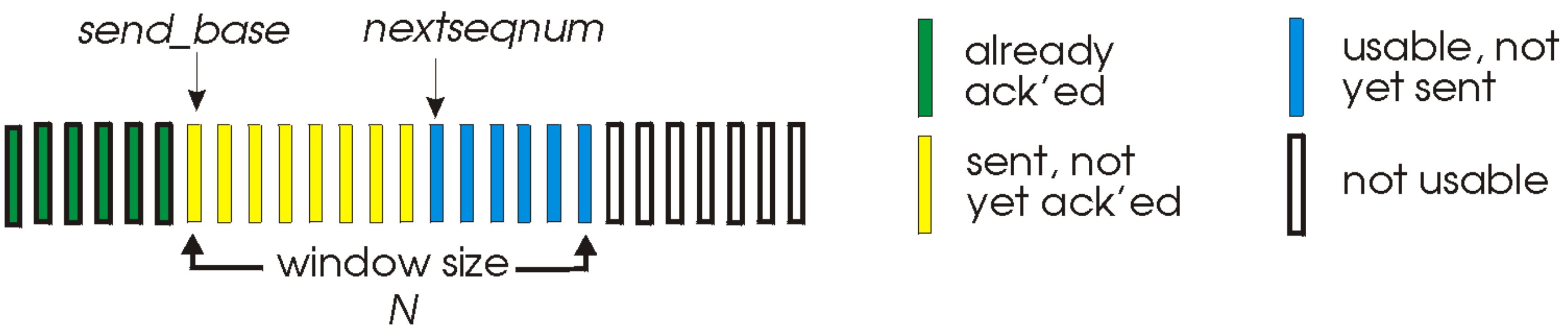

Increase utilization by a factor of 3!

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

# Go-Back-N

## Sender:

- Sequence # in packet header, k-bit
- "window" of up to N, consecutive unack'ed packets allowed




- ACK(n): ACKs all packets up to, including sequence # n
  - Cumulative ACK
- Timer for each in-flight packet batch (per send_base)
- *timeout(n):* retransmit packet n and all higher sequence # packets in window (send_base to nextseqnum)

# GBN: sender extended FSM

rdt_send(data)

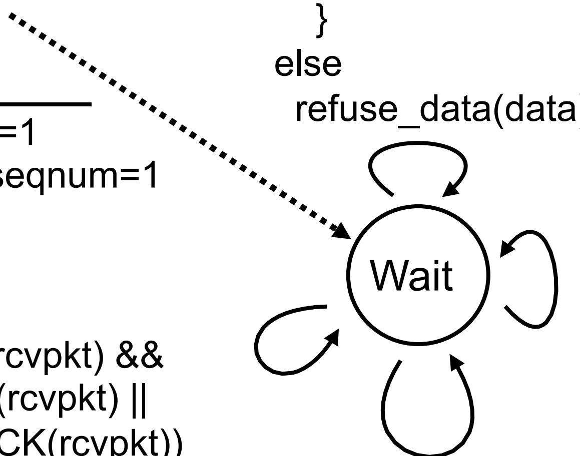
---

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)

$\Lambda$

---

base=1
nextseqnum=1




timeout

---

(re)start_timer
udt_send(sndpkt[base])
udt_send(sndpkt[base+1])
…
udt_send(sndpkt[nextseqnum-1])

rdt_rcv(rcvpkt) &&
(corrupt(rcvpkt) ||
isOLDACK(rcvpkt))

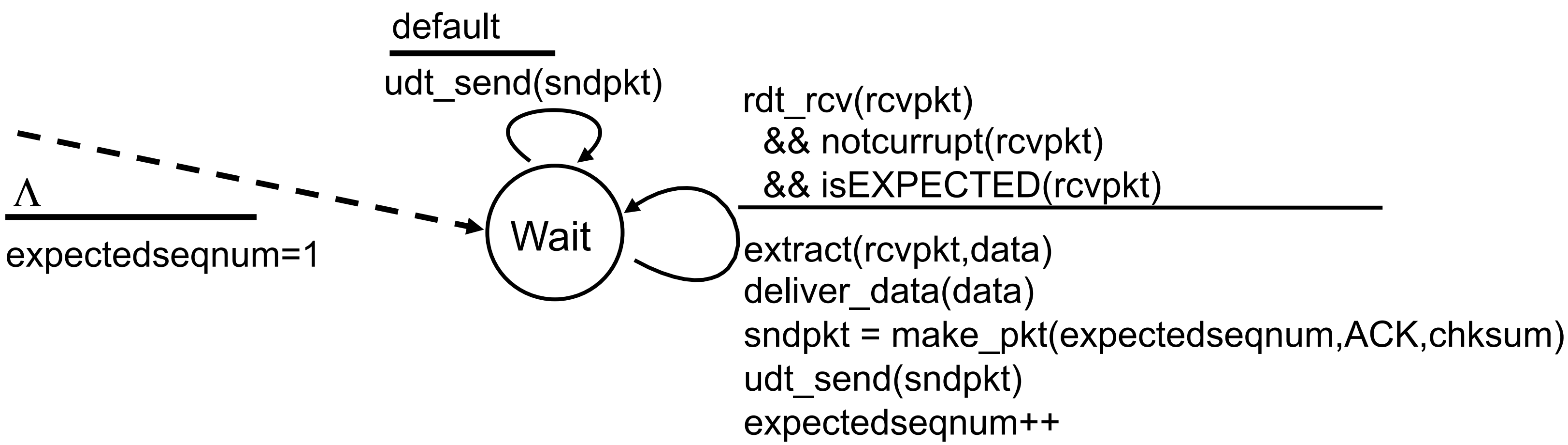
---

rdt_rcv(rcvpkt) &&
  notcorrupt(rcvpkt)  && isNEWACK (rcvpkt)

---

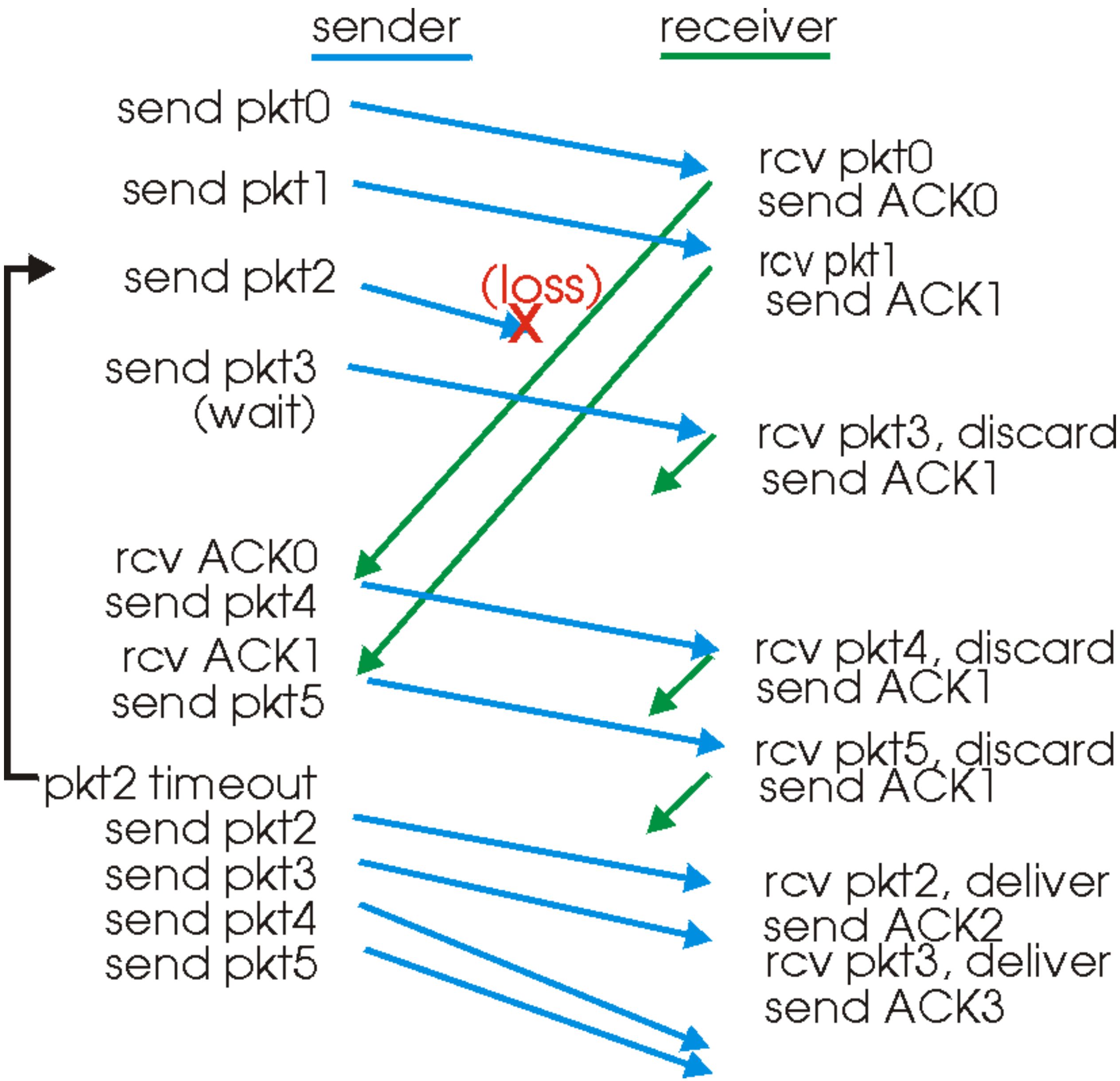
base = getacknum(rcvpkt)+1
If (base == nextseqnum)
    stop_timer **(no pkt outstanding)**
else {
    stop_timer **(for the old base)**
    start_timer **(for the new base)**
}

# GBN: receiver extended FSM




- Principle:
  - if it's the expected data packet, send ACK
  - Else, send NAK
- Making it ACK-only:
  - Send ACK for correctly-received packet with highest *in-order* sequence #
    - Need to remember **expectedseqnum**
  - For corrupted or out-of-order packet:
    - discard (don't buffer) -> no receiver buffering!
    - ACK packet with highest in-order sequence #

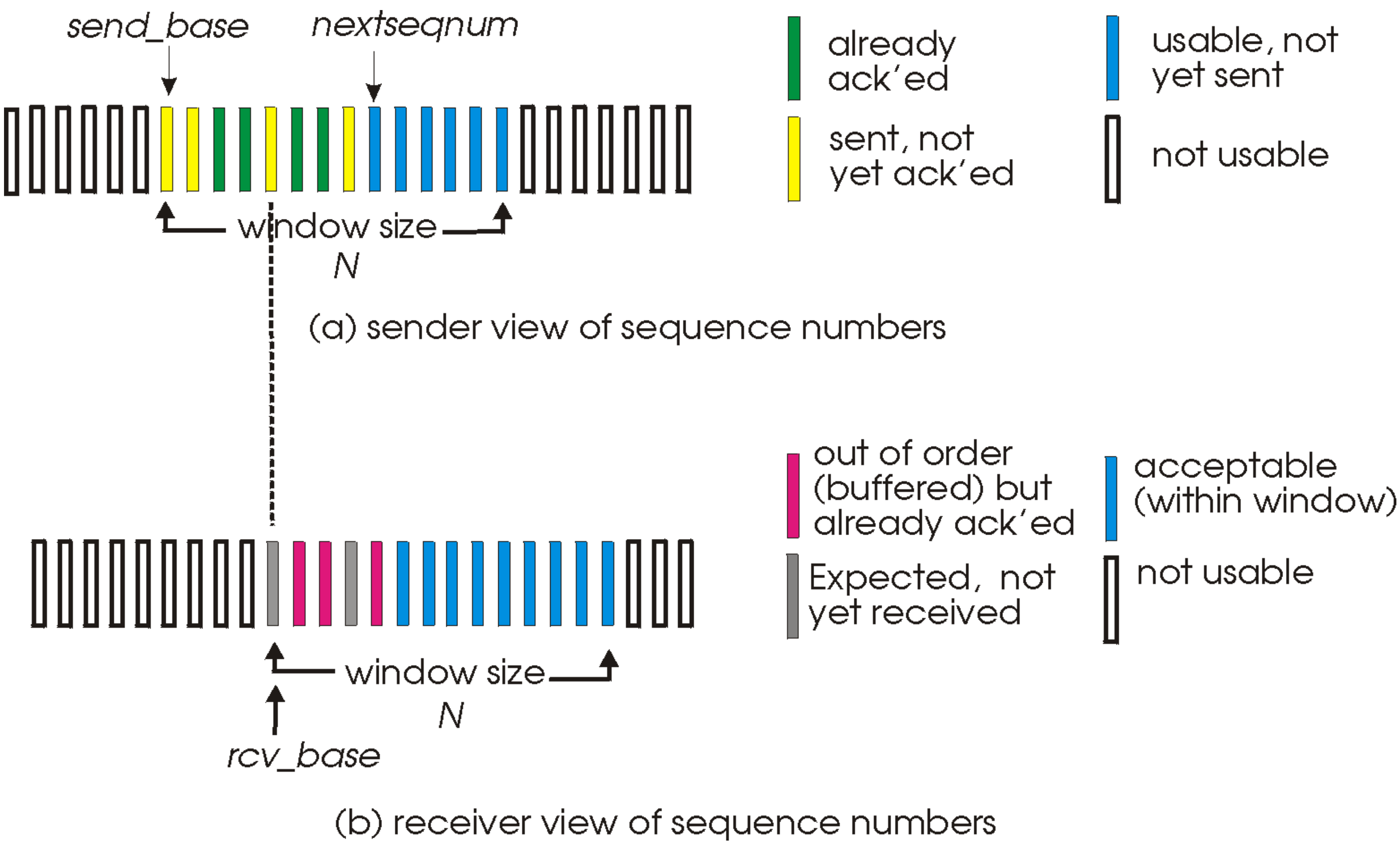
# GBN in action



# Selective Repeat

- Actually easier to understand
- Receiver *individually* acknowledges all correctly received packets
  - buffers packets, as needed, for eventual in-order delivery to upper layer
- Sender only re-sends packets for which ACK not received
  - sender timer for each unACKed packet
- Sender window
  - N consecutive sequence #'s
  - again limits sequence #s of sent, unACKed packets

# Selective repeat: sender, receiver windows



# Selective repeat

## sender

**Data from above :**

- If next available sequence # in window, send packet

**ACK(n)** in [sendbase,sendbase+N-1]:

- Mark packet n as received
- If n smallest unACKed packet, advance window base to next unACKed sequence #

**timeout(n):**

- Resend packet n, restart timer

## receiver

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

- Send ACK(n)
- Out-of-order: buffer
- In-order: deliver (also deliver buffered, in-order packets), advance window to next not-yet-received packet

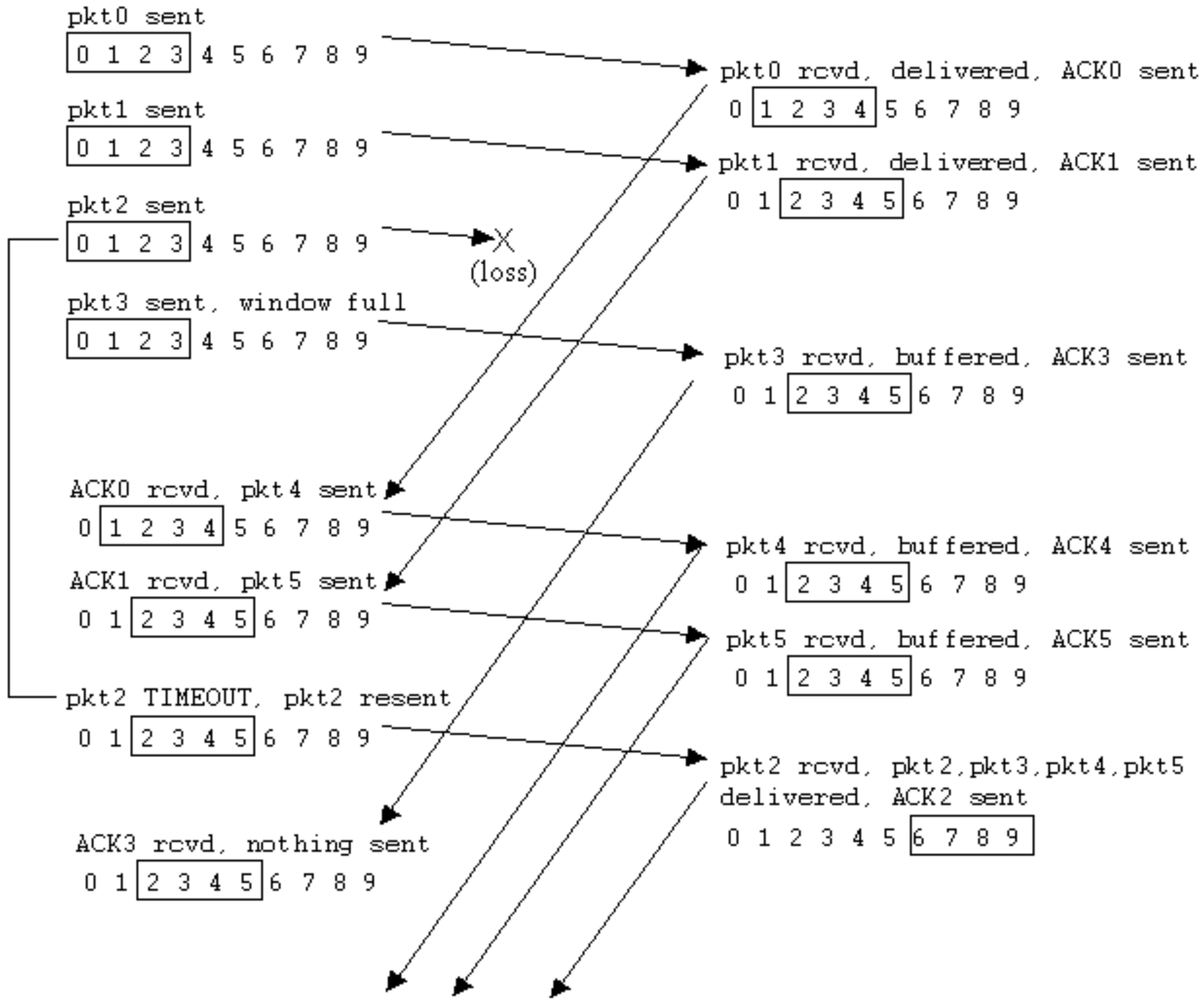
**pkt n in** [rcvbase-N, rcvbase-1]

- Send ACK(n)

**otherwise:**

- Ignore

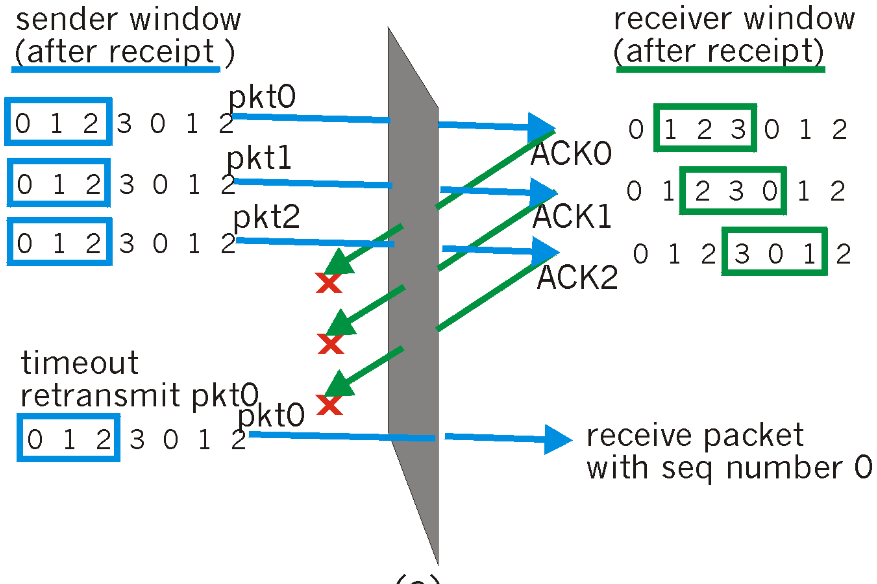
# Selective repeat in action



# Selective repeat: dilemma

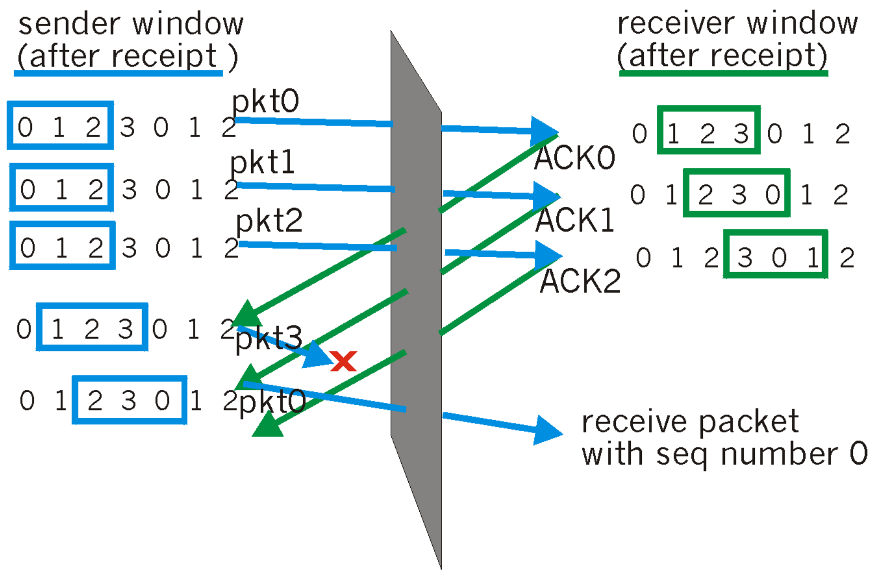
Example:

- sequence #'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 sequence # size and window size?

# Quiz Time!

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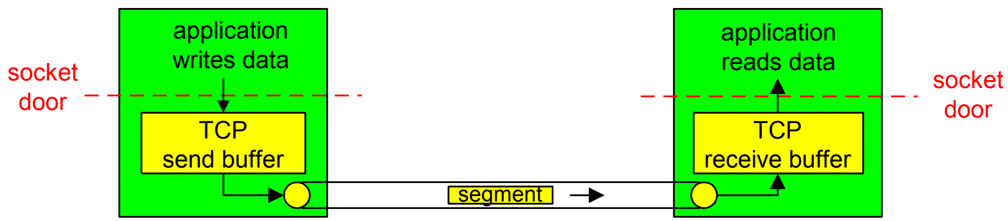
# TCP: Overview

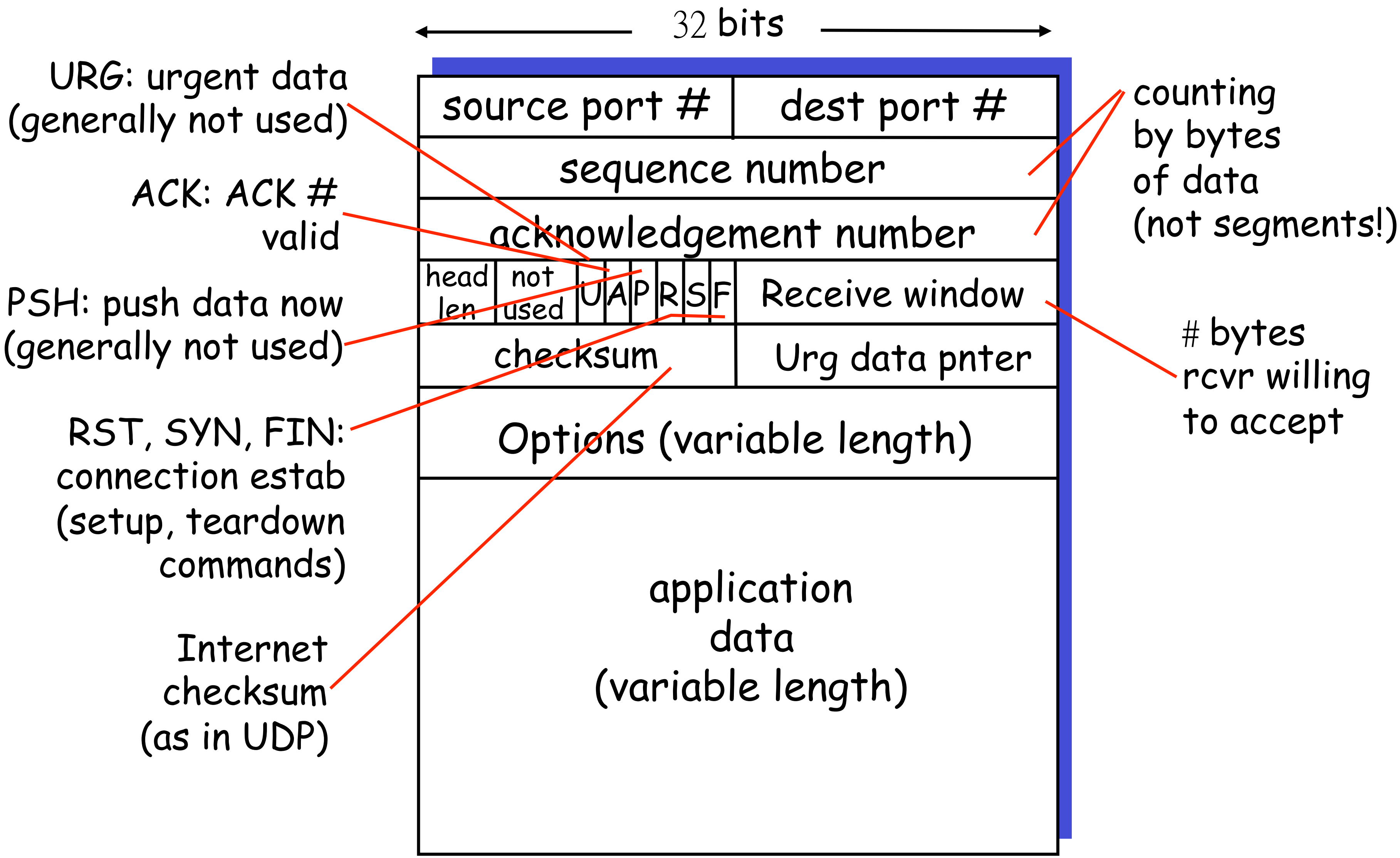
RFCs: 793, 1122, 1323, 2018, 2581

- point-to-point:
  - one sender, one receiver
- reliable, in-order *byte stream:*
  - no "message boundaries"
- pipelined:
  - TCP congestion and flow control set window size
- *send & receive buffers*




- full duplex data:
  - bi-directional data flow in same connection
  - MSS: maximum segment size
- connection-oriented:
  - handshaking (exchange of control messages) init's sender, receiver state before data exchange
- flow controlled:
  - sender will not overwhelm receiver

# TCP segment structure



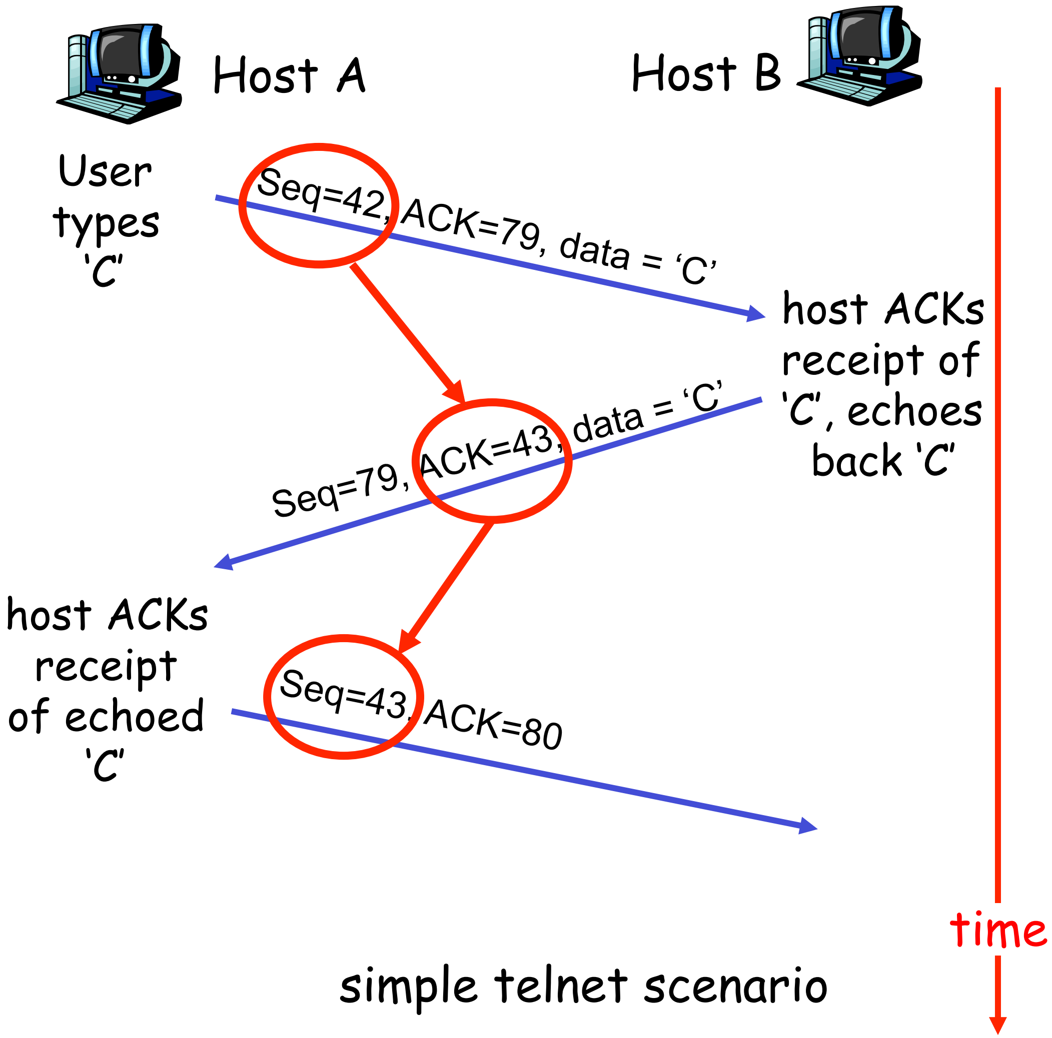
# TCP seq. #'s and ACKs

**Seq. #'s:**

- byte stream "number" of first byte in segment's data

**ACKs:**

- seq # of next byte expected from other side
- cumulative ACK




simple telnet scenario

# TCP Round Trip Time and Timeout

Q: how to set TCP timeout value?

- 1 sec? 1 min? Or else?
- too short?
- too long?

# 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?

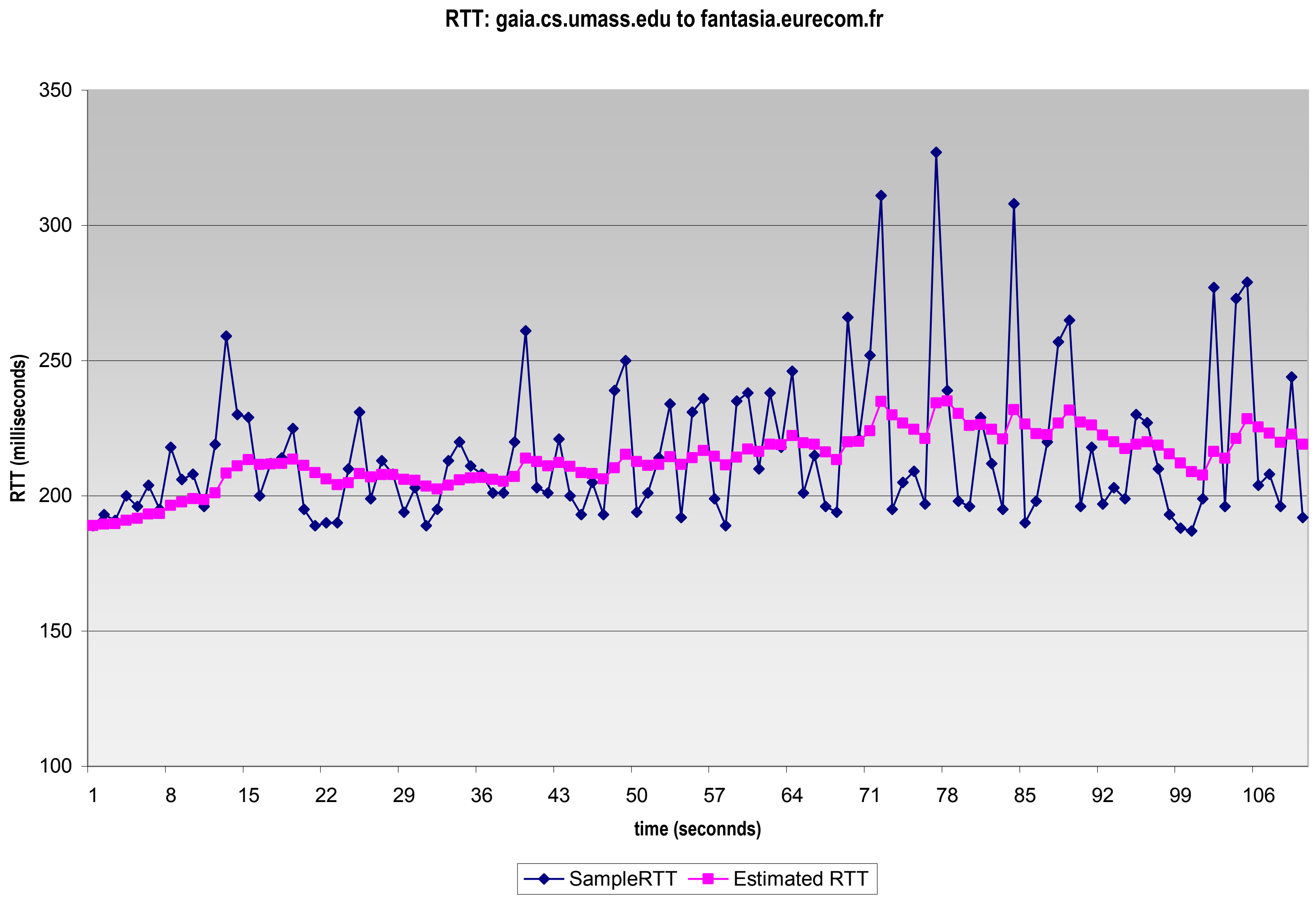
- **`SampleRTT`**: measured time from segment transmission until ACK receipt
  - ignore retransmissions
  - Why?
- **`SampleRTT`** will vary, want estimated RTT "smoother"
  - How?
  - average several recent measurements, not just current **`SampleRTT`**

# TCP Round Trip Time and Timeout

```
EstimatedRTT = (1- α)*EstimatedRTT + α*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

- `EstimtedRTT` plus "safety margin"
  - large variation in `EstimatedRTT` -> larger safety margin
- first estimate of how much SampleRTT deviates from EstimatedRTT:

```
DevRTT = (1-β)*DevRTT +
               β*|SampleRTT-EstimatedRTT|

(typically, β = 0.25)
```

Then set timeout interval:

```
TimeoutInterval = EstimatedRTT + 4*DevRTT
```

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# TCP reliable data transfer

- TCP creates rdt service on top of IP's unreliable service
- Pipelined segments
- Cumulative acks
- TCP uses single retransmission timer
- Retransmissions are triggered by:
  - timeout events
  - duplicate acks
- Initially consider simplified TCP sender:
  - ignore duplicate acks
  - ignore flow control, congestion control

# TCP sender events:

**data rcvd from app:**

- 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

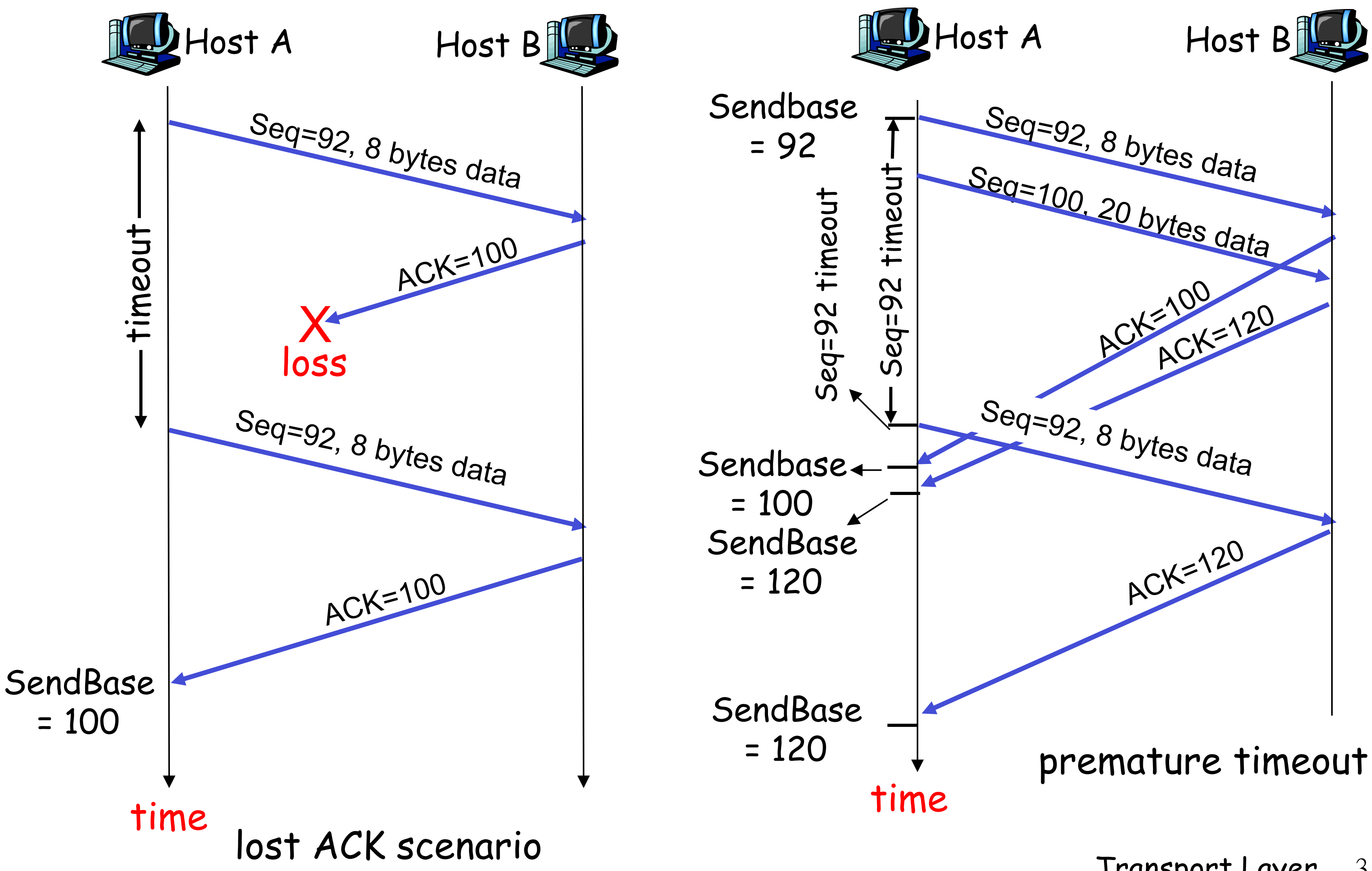
**Ack rcvd:**

- If acknowledges previously unacked segments
  - update what is known to be acked
  - start timer if there are outstanding segments

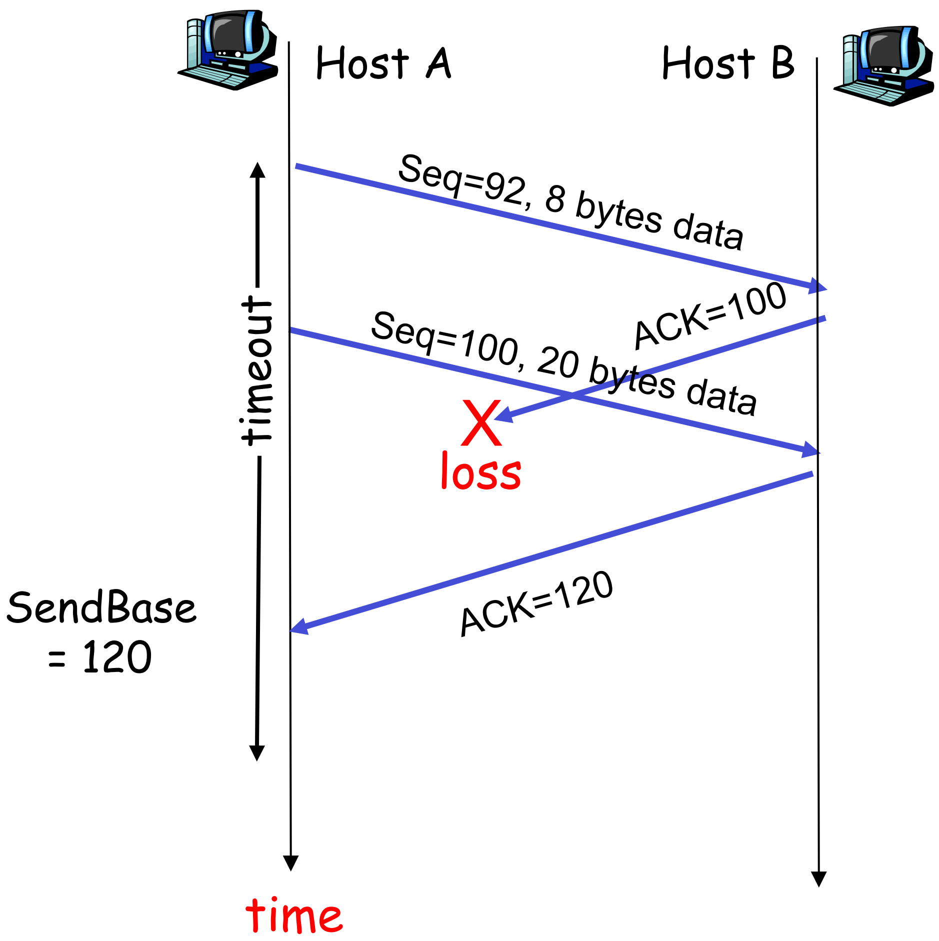
# TCP sender (simplified)

```
NextSeqNum = InitialSeqNum
SendBase = InitialSeqNum

loop (forever) {
  switch(event)

  event: data received from application above
      create TCP segment with sequence number NextSeqNum
      if (timer currently not running)
          start timer
      pass segment to IP
      NextSeqNum = NextSeqNum + length(data)

  event: timer timeout
      retransmit not-yet-acknowledged segment from
          smallest sequence number
      start timer

  event: ACK received, with ACK field value of y
      if (y > SendBase) {
          SendBase = y
          if (there are currently not-yet-acknowledged segments)
              (re)start timer
          }

} /* end of loop forever */
```

# TCP: retransmission scenarios



# TCP retransmission scenarios (more)




Cumulative ACK scenario

# TCP ACK generation [RFC 1122, RFC 2581]

| Event at Receiver | TCP Receiver action |
|---|---|
| Arrival of in-order segment with expected seq #. All data up to expected seq # already ACKed | Delayed ACK. Wait up to 500ms for next segment. If no next segment, send ACK |
| Arrival of in-order segment with expected seq #. One other segment has ACK pending | Immediately send single cumulative ACK, ACKing both in-order segments |
| Arrival of out-of-order segment higher-than-expect seq. # . Gap detected | Immediately send duplicate ACK, indicating seq. # of next expected byte |
| Arrival of segment that partially or completely fills gap | Immediate send ACK, provided that segment starts at lower end of gap |

# 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.
- If sender receives 3 dup ACKs for the same data, it supposes that segment after ACKed data was lost:
  - fast retransmit: resend segment before timer expires

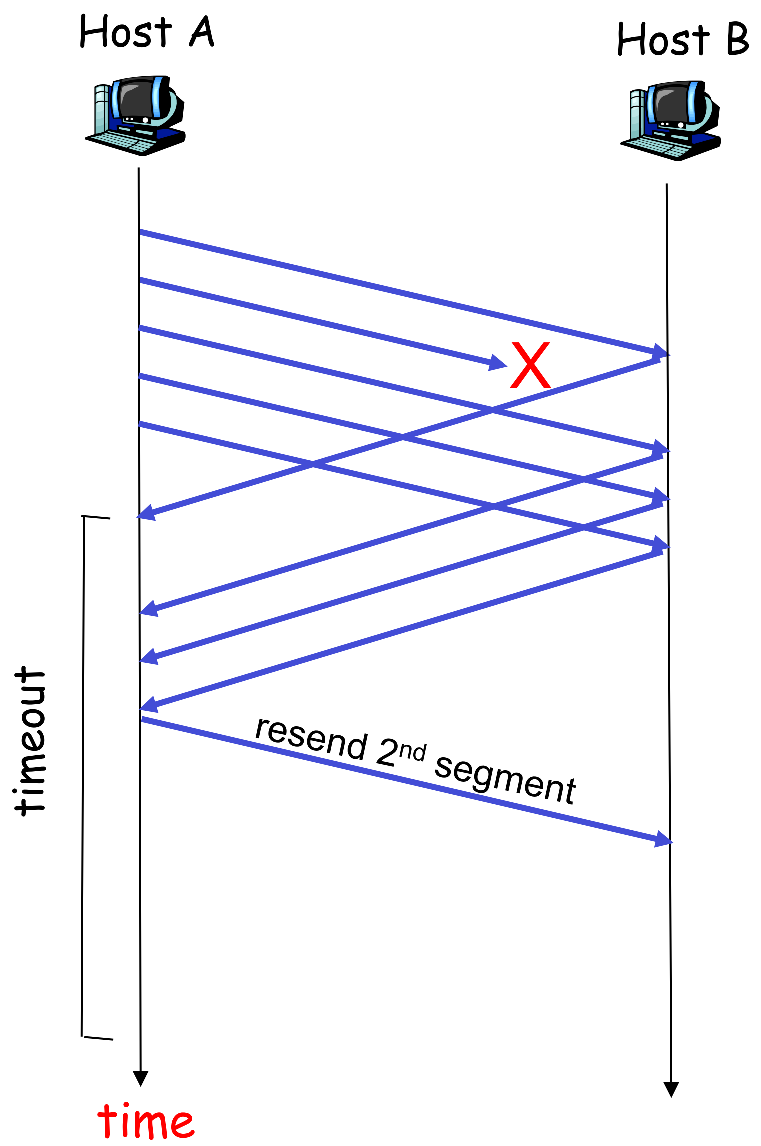



Figure 3.37 Resending a segment after triple duplicate ACK

# Fast retransmit algorithm:

```
event: ACK received, with ACK field value of y
           if (y > SendBase) {
               SendBase = y
               if (there are currently not-yet-acknowledged segments)
                    start timer
             }
           else {
                increment count of dup ACKs received for y
                if (count of dup ACKs received for y = 3) {
                     resend segment with sequence number y
                 }
```

a duplicate ACK for already ACKed segment

fast retransmit

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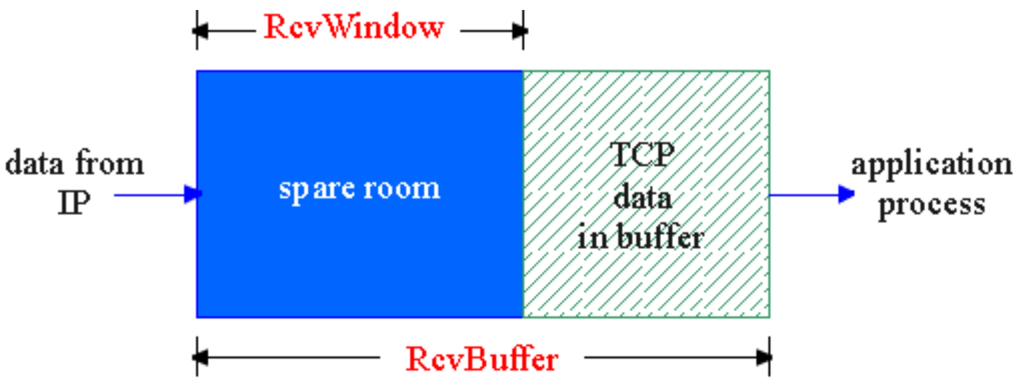
# TCP Flow Control

- receiver side of TCP connection has a receive buffer:



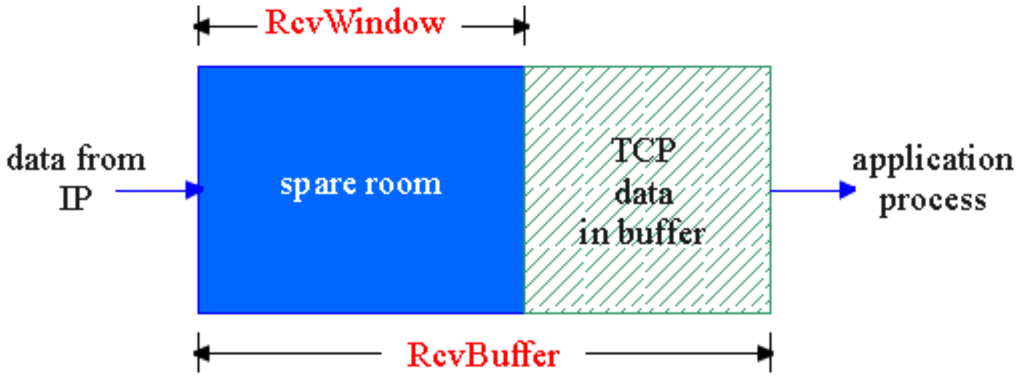

- app process may be slow at reading from buffer

**flow control**

sender won't overflow receiver's buffer by transmitting too much, too fast

- speed-matching service: matching the send rate to the receiving app's drain rate

# TCP Flow control: how it works




(Suppose TCP receiver discards out-of-order segments)

- spare room in buffer

= **`RcvWindow`**

= **`RcvBuffer-[LastByteRcvd - LastByteRead]`**

- Rcvr advertises spare room by including value of **`RcvWindow`** in segments
- Sender limits unACKed data to **`RcvWindow`**
  - guarantees receive buffer doesn't overflow

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# TCP Connection Management

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

- initialize TCP variables:
  - seq. #s
  - buffers, flow control info (e.g. `RcvWindow`)
- *client:* connection initiator
  `connect();`
- *server:* contacted by client
  `listen();`

## Three way handshake:

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

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

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

# TCP Connection Management (cont.)

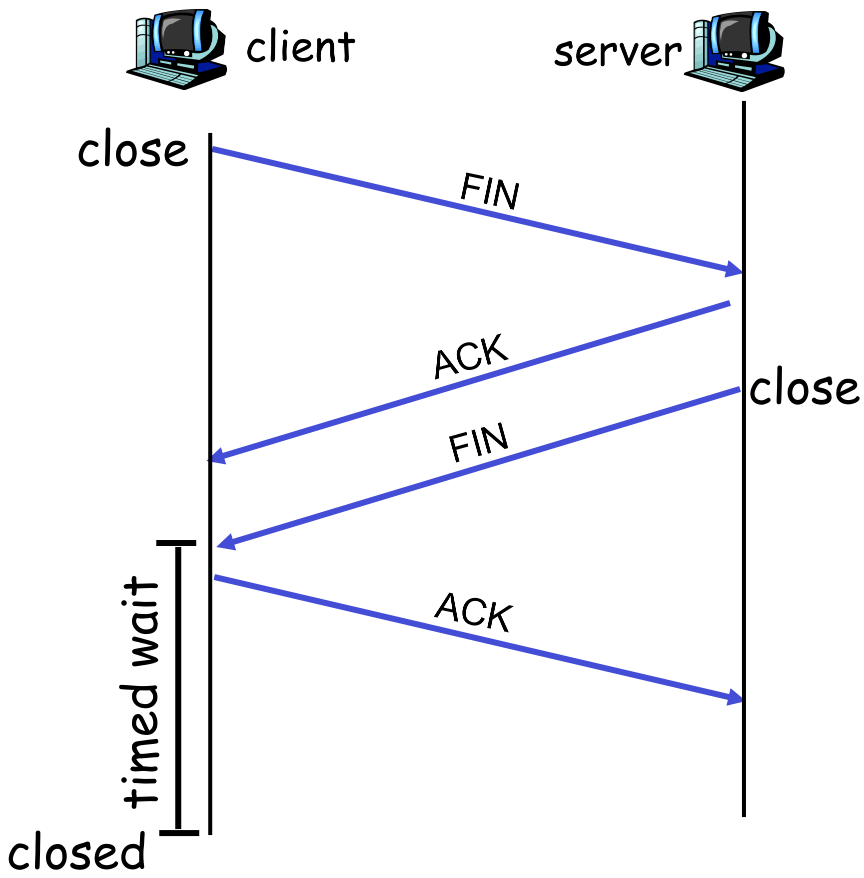
## Closing a connection:

client closes socket:

`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.



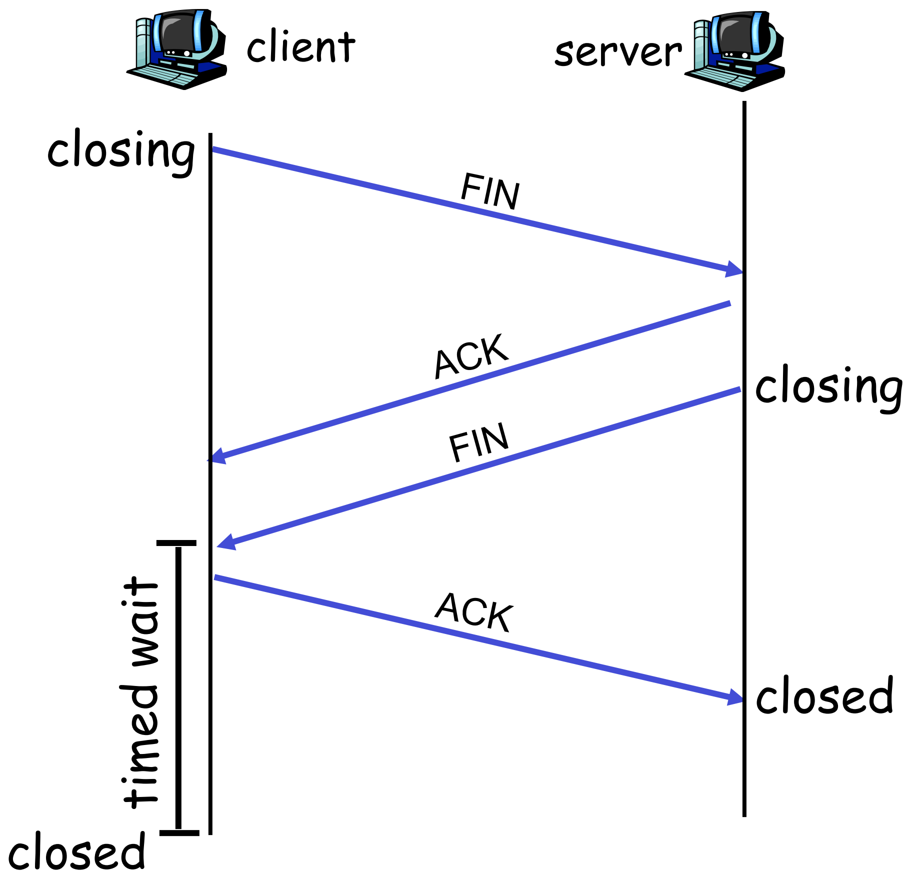
# TCP Connection Management (cont.)

**Step 3:** client receives FIN, replies with ACK.

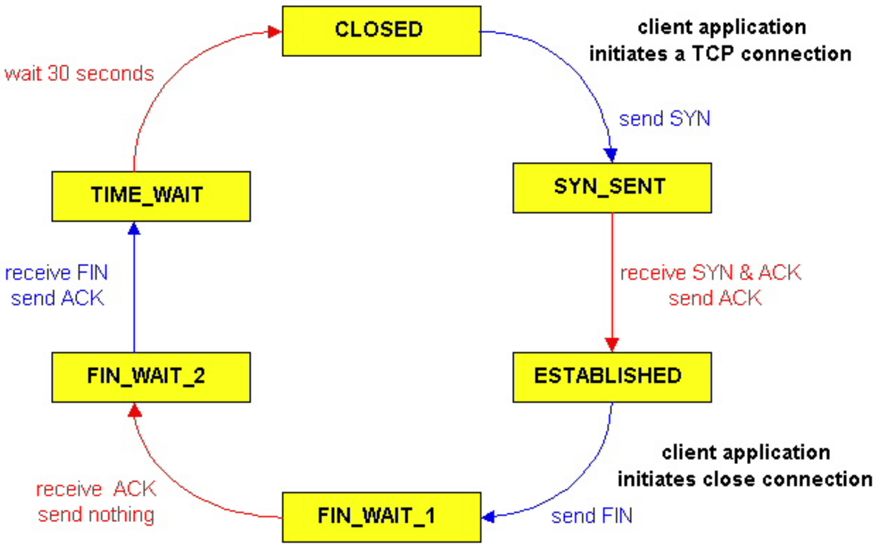
- Enters "timed wait" - will respond with ACK to received FINs

**Step 4:** server, receives ACK. Connection closed.

**Note:** with small modification, can handle simultaneous FINs.

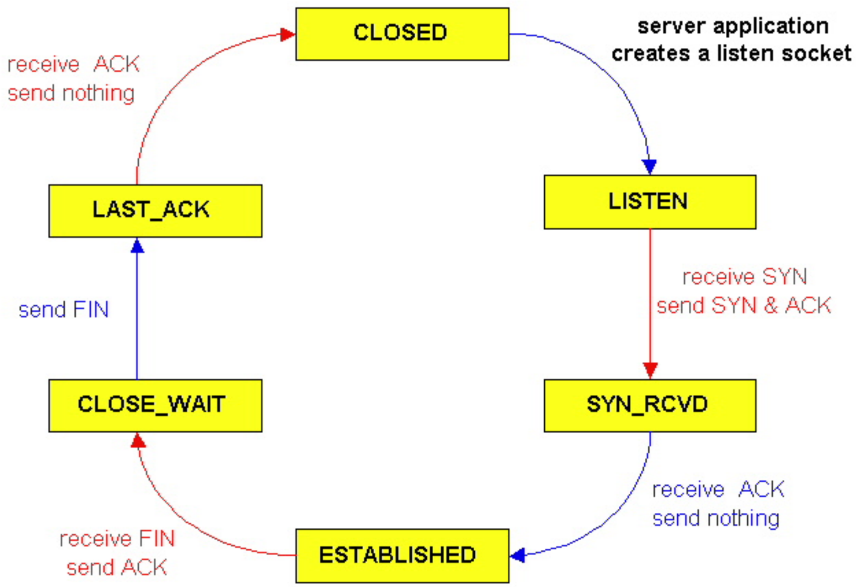


# TCP Connection Management (cont)




TCP client lifecycle

TCP server lifecycle



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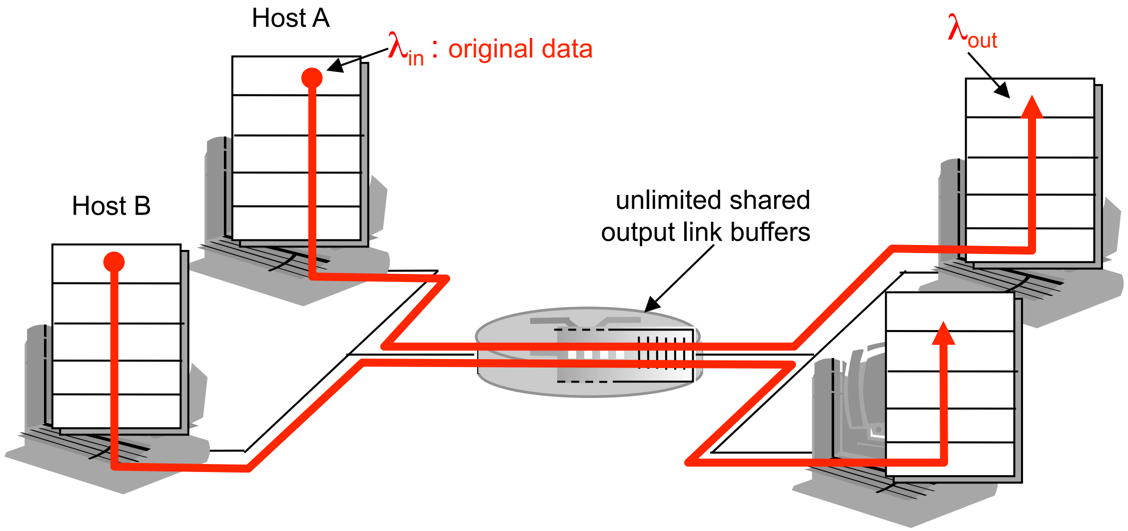
# Principles of Congestion Control

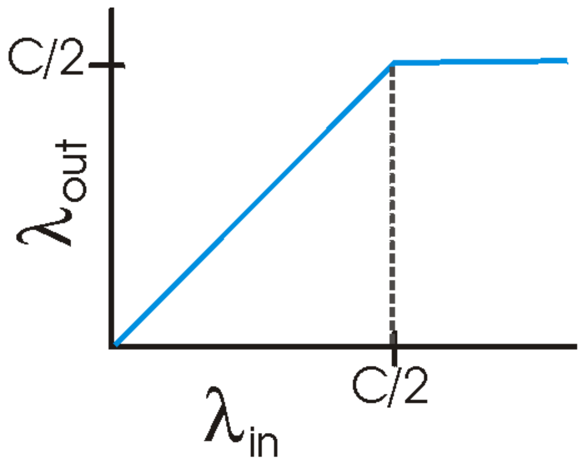
Congestion:

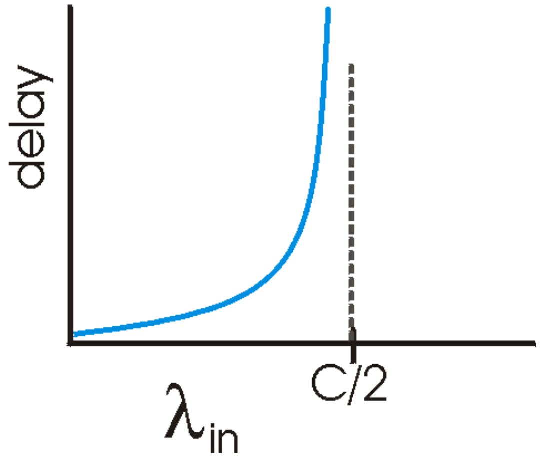
- informally: "too many sources sending too much data too fast for *network* to handle"
- different from flow control!
- manifestations:
  - lost packets (buffer overflow at routers)
  - long delays (queuing in router buffers)
- a top-10 problem!

# Causes/costs of congestion: scenario 1

- two senders, two receivers
- one router, infinite buffers
- no retransmission



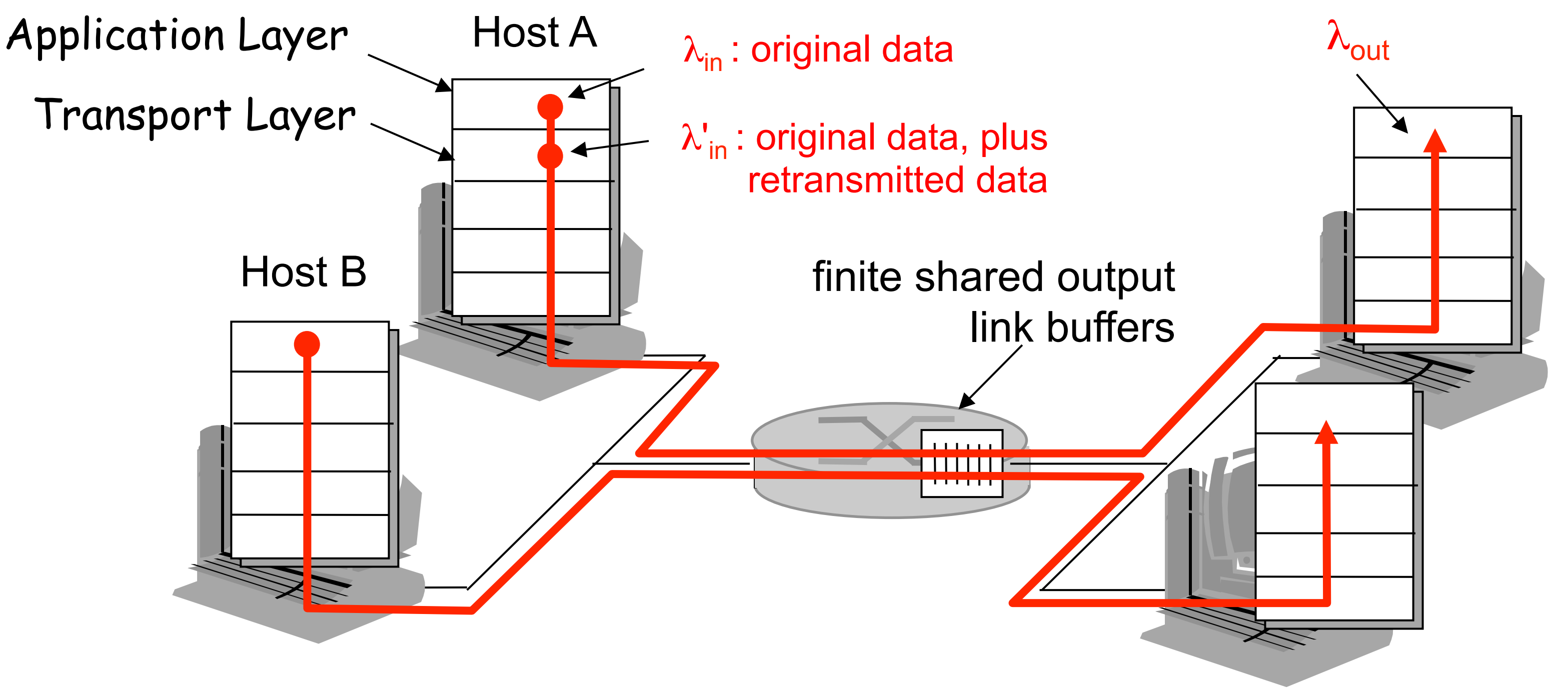




- maximum achievable throughput
- large delays when congested

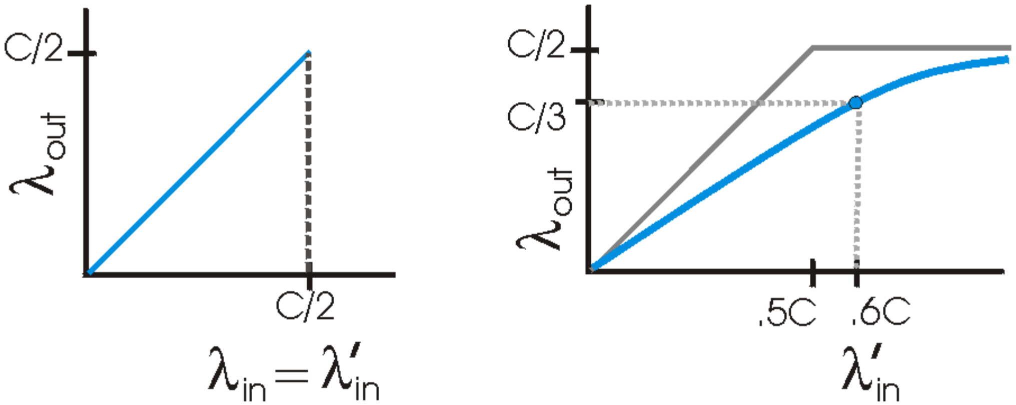
# Causes/costs of congestion: scenario 2

- one router, *finite* buffers
- sender retransmission of lost packet



# Causes/costs of congestion: scenario 2

- "perfect" case, always: $\lambda_{in} = \lambda_{out,}$ (goodput)
- retransmission only when loss: $\lambda'_{in} > \lambda_{out}$
- retransmission of lost packet makes $\lambda'_{in}$ larger (than perfect case) for same $\lambda_{out}$




(a) (b)

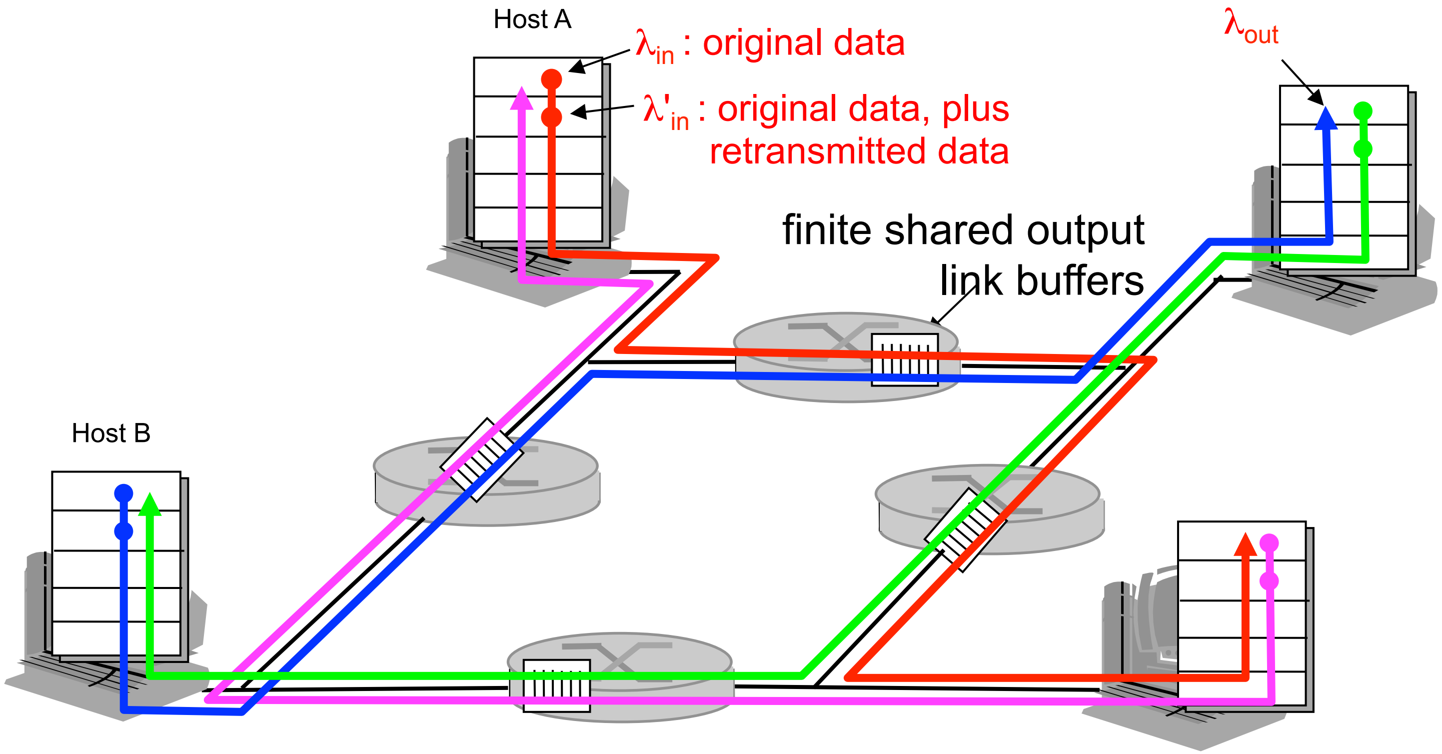
"costs" of congestion:

- more work (retransmission) for given "goodput"
- unneeded retransmissions: link carries multiple copies of pkt

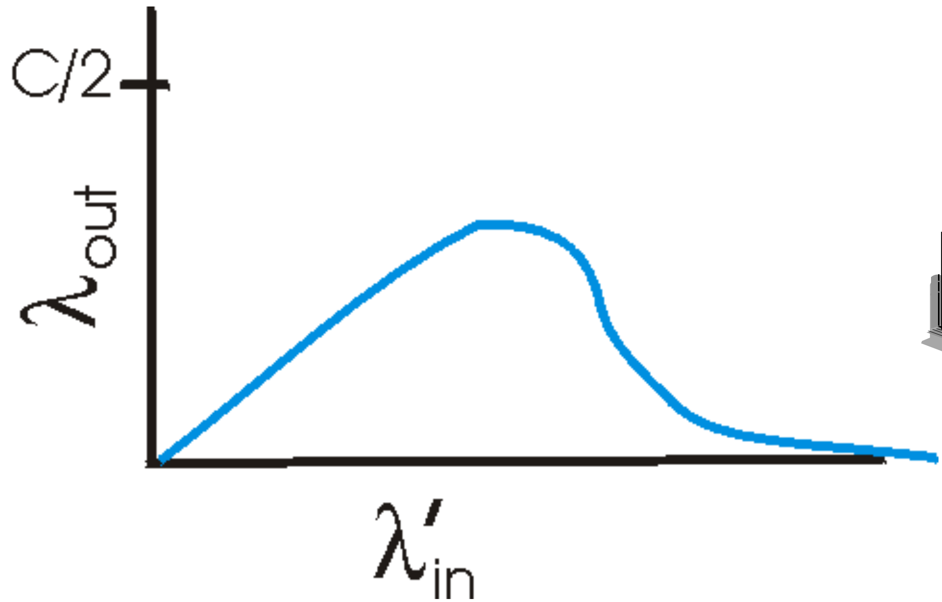
# Causes/costs of congestion: scenario 3

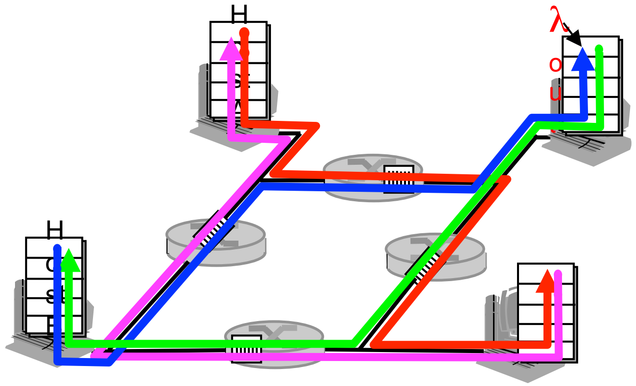
- Four senders
- Multi-hop paths
- Timeout/retransmit

Q: what happens as $\lambda_{in}$ and $\lambda'_{in}$ increase ?



# Causes/costs of congestion: scenario 3



Another "cost" of congestion:

- when packet gets dropped, any "upstream" transmission capacity used for that packet was wasted!

# Message: Congestion is bad

But what can we do about it?

# Quiz Time!

# Approaches towards congestion control

Two broad approaches towards congestion control:

**End-end congestion control:**

- no explicit feedback from network
- congestion inferred from end-system observed loss, delay
- approach taken by TCP

**Network-assisted congestion control:**

- routers provide feedback to end systems
  - single bit indicating congestion (SNA, DECbit, TCP/IP ECN, ATM)
  - tells explicit rate that sender should send at

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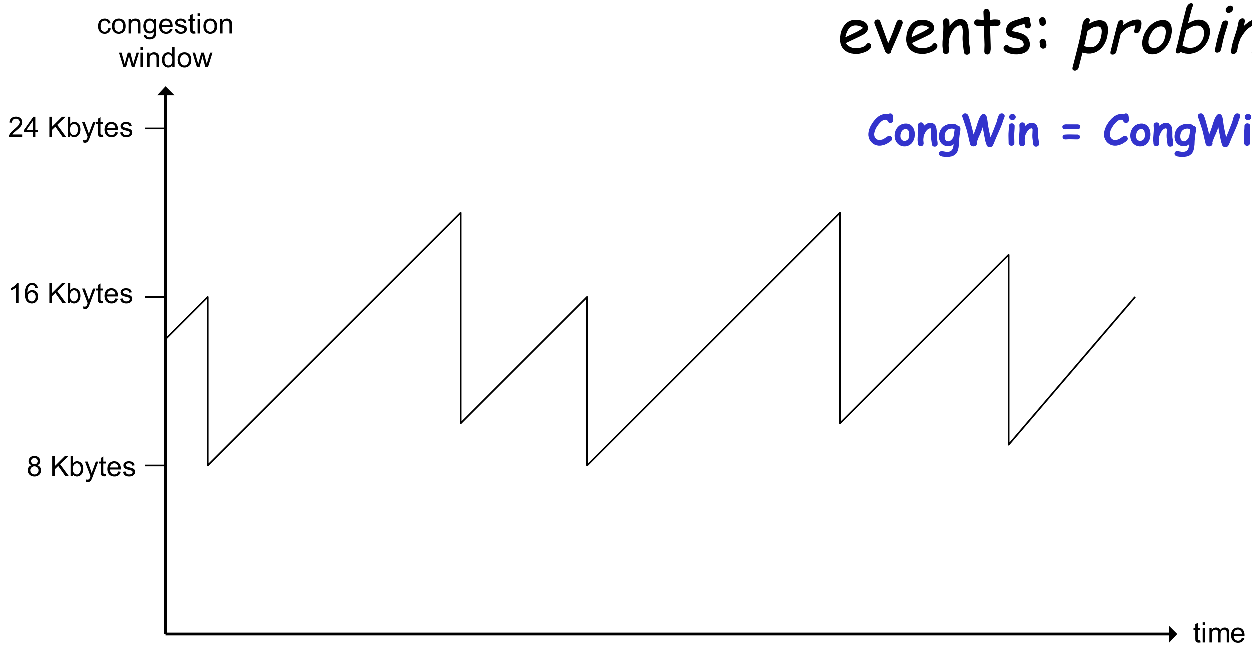
# TCP AIMD

multiplicative decrease: cut **CongWin** in half after loss event

CongWin = CongWin * 0.5

additive increase: increase **CongWin** by 1 MSS every RTT in the absence of loss events: *probing*

CongWin = CongWin + 1




Long-lived TCP connection

# TCP Congestion Control

- end-end control (no network assistance)
- sender limits transmission:

  **LastByteSent-LastByteAcked ≤ CongWin**

- Roughly,

$$\text{rate} = \frac{\text{CongWin}}{\text{RTT}} \text{ Bytes/sec}$$

- **CongWin** is dynamic, a function of perceived network congestion

How does sender perceive congestion?

- loss event
- How to tell whether there's a loss event?
- TCP sender reduces rate (**CongWin**) after loss event

three mechanisms:

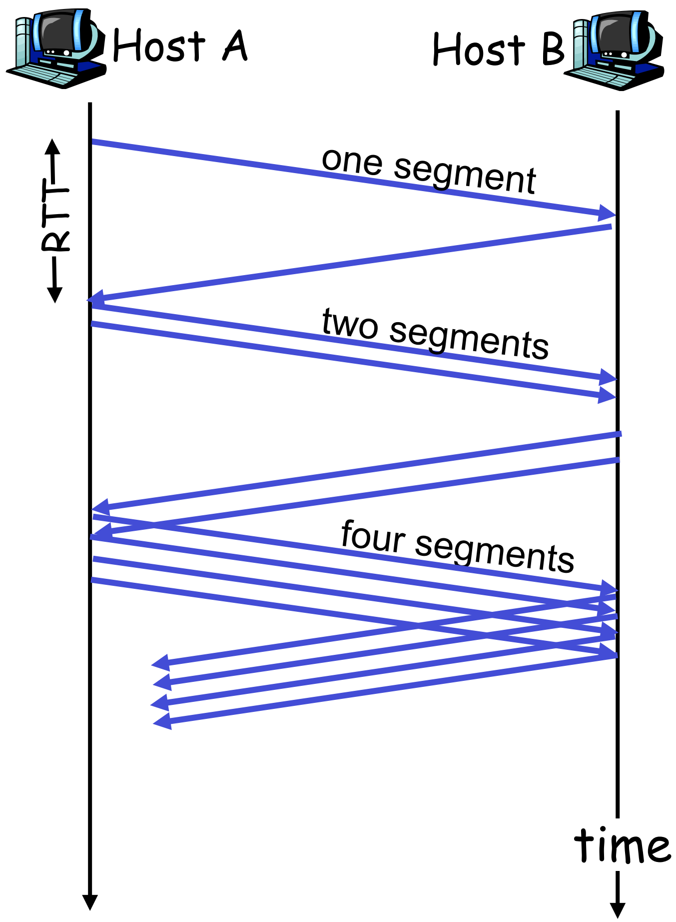
- AIMD
- slow start
- conservative after timeout events

# TCP Slow Start

- When connection begins, `CongWin` = 1 MSS
  - Example:
  - MSS = 500 bytes
  - RTT = 200 msec
  - initial rate = 20 kbps
- available bandwidth may be >> MSS/RTT
  - desirable to quickly ramp up to respectable rate
- When connection begins, increase rate exponentially fast until first loss event

# TCP Slow Start (more)

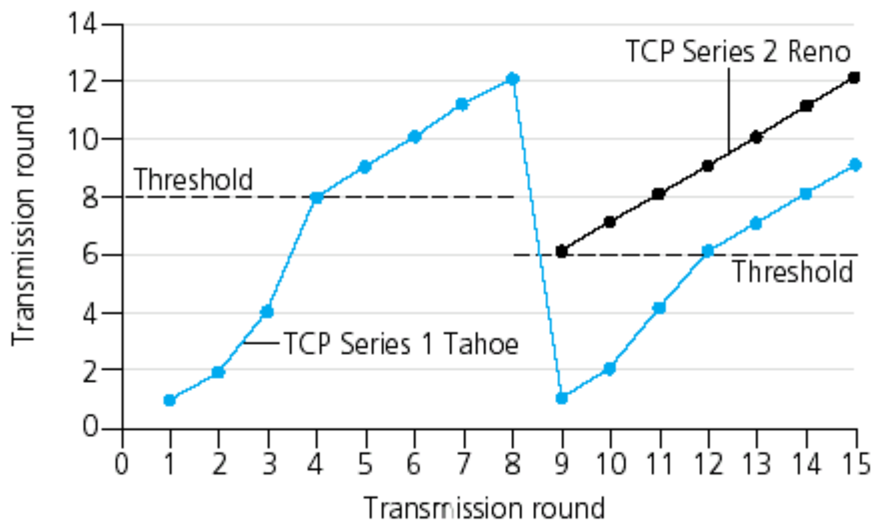
- When connection begins, increase rate exponentially:
  - double `CongWin` every RTT
  - done by incrementing `CongWin` for every ACK received
- Summary: initial rate is slow but ramps up exponentially fast



# Refinement

Q: When should the exponential increase switch to linear?

A: When **CongWin** gets to 1/2 of its value before timeout.




## Implementation:

- Variable Threshold
- At loss event, Threshold is set to 1/2 of CongWin just before loss event

# Refinement

- After 3 dup ACKs:
  - `CongWin` is cut in half
  - window then grows linearly
- <u>But</u> after timeout event:
  - `CongWin` instead set to 1 MSS;
  - window then grows exponentially
  - to a threshold, then grows linearly

**Philosophy:**

Why Half the CongWin vs. 1?

# Refinement

- After 3 dup ACKs:
  - `CongWin` is cut in half
  - window then grows linearly
- But after timeout event:
  - `CongWin` instead set to 1 MSS;
  - window then grows exponentially
  - to a threshold, then grows linearly

**Philosophy:**

- 3 dup ACKs indicates network capable of delivering some segments
- timeout before 3 dup ACKs is "more alarming"

## Quiz Time!

# Summary: TCP Congestion Control

- When **CongWin** is below **Threshold**, sender in slow-start phase, window grows exponentially.
- When **CongWin** is above **Threshold**, sender is in congestion-avoidance phase, window grows linearly.
- When a triple duplicate ACK occurs, **Threshold** set to **CongWin/2** and **CongWin** set to **Threshold**.
- When timeout occurs, **Threshold** set to **CongWin/2** and **CongWin** is set to 1 MSS.

# TCP sender congestion control

| State | Event | TCP Sender Action | Commentary |
|---|---|---|---|
| Slow Start (SS) | ACK receipt for previously unacked data | CongWin = CongWin + MSS, If (CongWin > Threshold) set state to “Congestion Avoidance” | Resulting in a doubling of CongWin every RTT |
| Congestion Avoidance (CA) | ACK receipt for previously unacked data | CongWin = CongWin+ MSS * (MSS/CongWin) | Additive increase, resulting in increase of CongWin by 1 MSS every RTT |
| SS or CA | Loss event detected by triple duplicate ACK | Threshold = CongWin/2, CongWin = Threshold, Set state to “Congestion Avoidance” | Fast recovery, implementing multiplicative decrease. CongWin will not drop below 1 MSS. |
| SS or CA | Timeout | Threshold = CongWin/2, CongWin = 1 MSS, Set state to “Slow Start” | Enter slow start |
| SS or CA | Duplicate ACK | Increment duplicate ACK count for segment being acked | CongWin and Threshold not changed |

# TCP throughput

- What's the average throughout of TCP as a function of window size and RTT?
  - Ignore slow start
- Let W be the window size when loss occurs.
- When window is W, throughput is W/RTT
- Just after loss, window drops to W/2, throughput to W/2RTT.
- Average throughout: .75 W/RTT

# TCP Futures: TCP over "long, fat pipes"

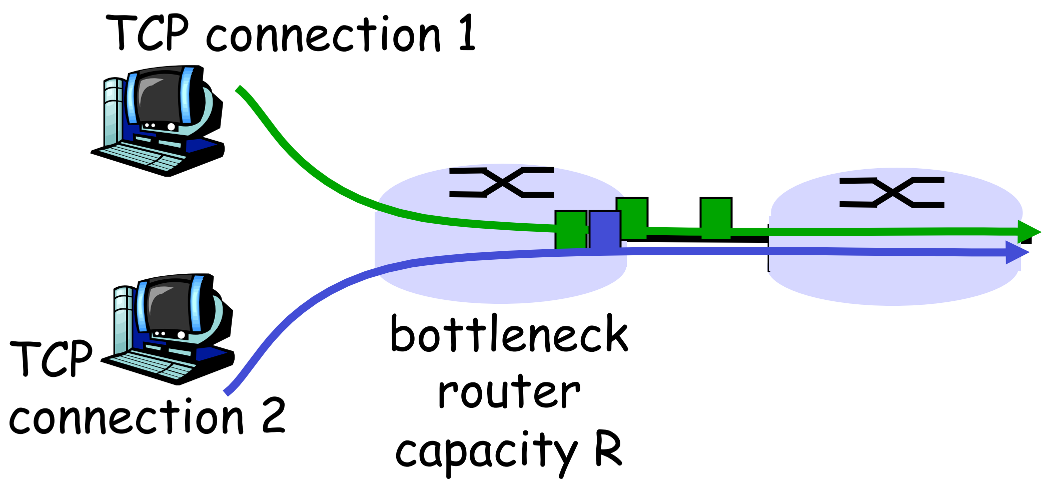
- Example: 1500 byte segments, 100ms RTT, want 10 Gbps throughput
- Requires window size W = 83,333 in-flight segments
- Throughput in terms of loss rate:

$$\frac{1.22 \cdot MSS}{RTT\sqrt{L}}$$

- ➜ $L = 2 \cdot 10^{-10}$ *Wow*
- New versions of TCP for high-speed

# TCP Fairness

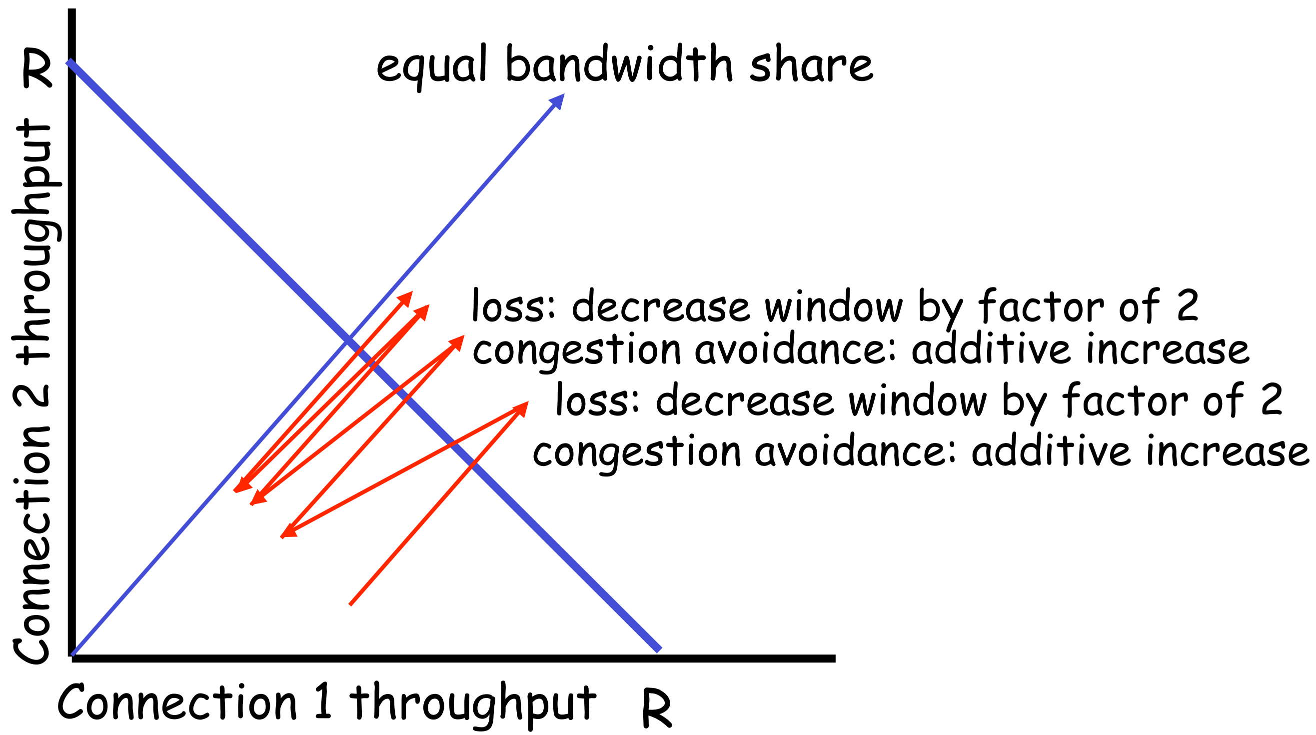
Fairness goal: if K TCP sessions share same bottleneck link of bandwidth R, each should have average rate of R/K



# Why is TCP fair?

Two competing sessions:

- Additive increase gives slope of 1, as throughout increases
- multiplicative decrease decreases throughput proportionally



# Fairness (more)

## Fairness and UDP

- Multimedia apps often do not use TCP
  - do not want rate throttled by congestion control
- Instead use UDP:
  - pump audio/video at constant rate, tolerate packet loss
- Research area: TCP friendly

## Fairness and parallel TCP connections

- nothing prevents app from opening parallel connections between 2 hosts.
- Web browsers do this
- Example: 9 users, link of rate R supporting 9 TCP connections;
  - new app/user asks for 1 TCP, gets rate R/10
  - new app/user asks for 9 TCPs, gets R/2 !

# Chapter 3: Summary

- principles behind transport layer services:
  - multiplexing, demultiplexing
  - reliable data transfer
  - flow control
  - congestion control
- instantiation and implementation in the Internet
  - UDP
  - TCP

## Next:

- leaving the network "edge" (application, transport layers)
- into the network "core"