CSCE 560
Introduction to Computer Networking

Dr. Barry Mullins
AFIT/ENG
Bldg 642, Room 209
255-3636 x7979

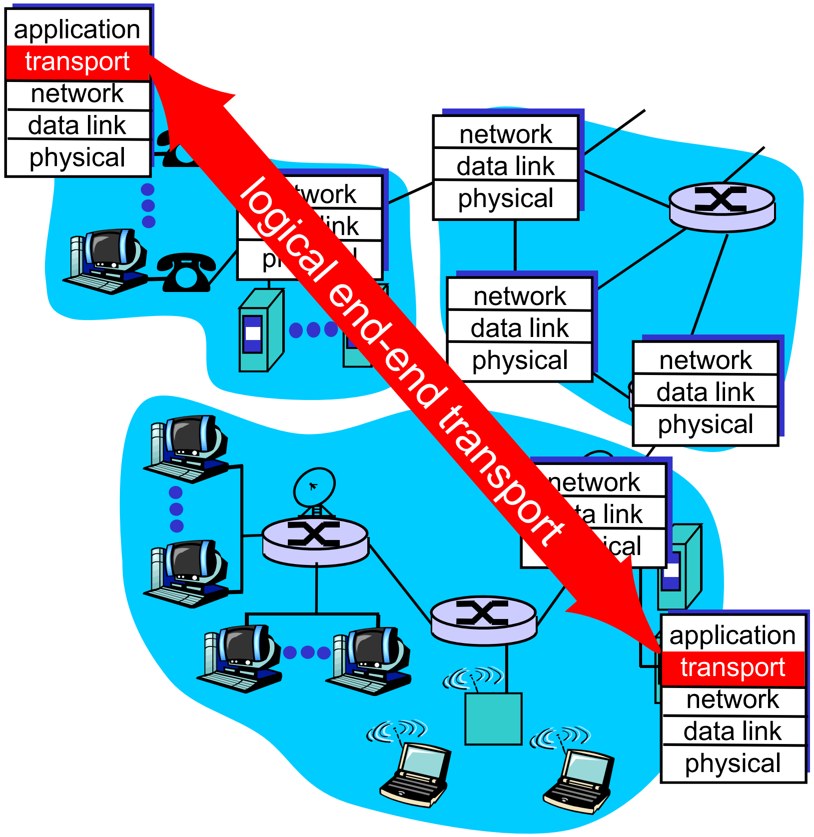
President George W. Bush stands with Presidential Medal of Freedom recipients, **Vinton G. Cerf** and **Robert E. Kahn**, Nov. 9, 2005, during ceremonies at the White House. Cerf and Kahn were co-designers of the TCP/IP Internet network protocol.

Chapter 3 Outline

- 3.1 Transport-layer services
- 3.2 Multiplexing and demultiplexing
- 3.3 Connectionless transport: UDP
- 3.4 Principles of reliable data transfer
- 3.5 Connection-oriented transport: TCP
  - Segment structure
  - Reliable data transfer
  - Flow control
  - Connection management
- 3.6 Principles of congestion control
- 3.7 TCP congestion control

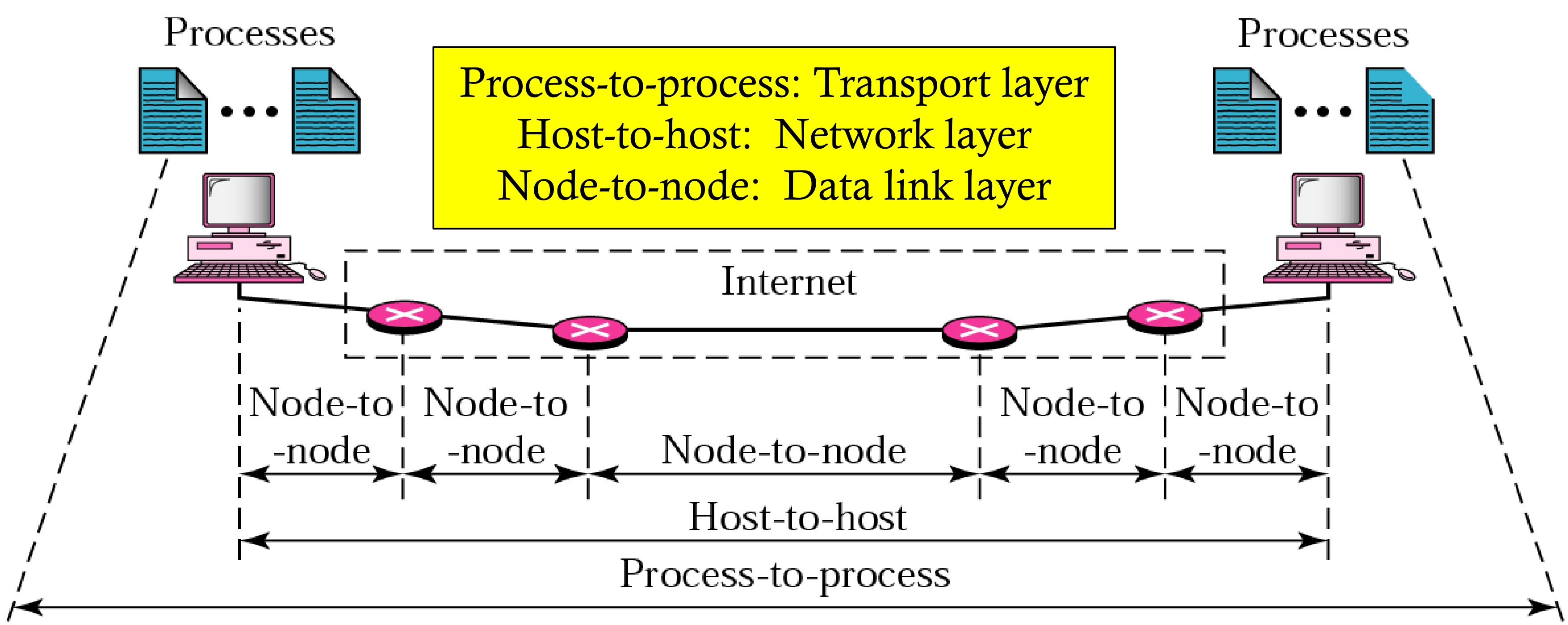
Transport Services and Protocols

- Provide logical communication between app processes running on different hosts
- Transport protocols run in end systems
  - Sender: breaks app messages into segments, passes to network layer
  - Rcvr: 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: logical end-to-end communication between processes
  - Relies on, enhances, network layer services
- Network: logical communication between hosts



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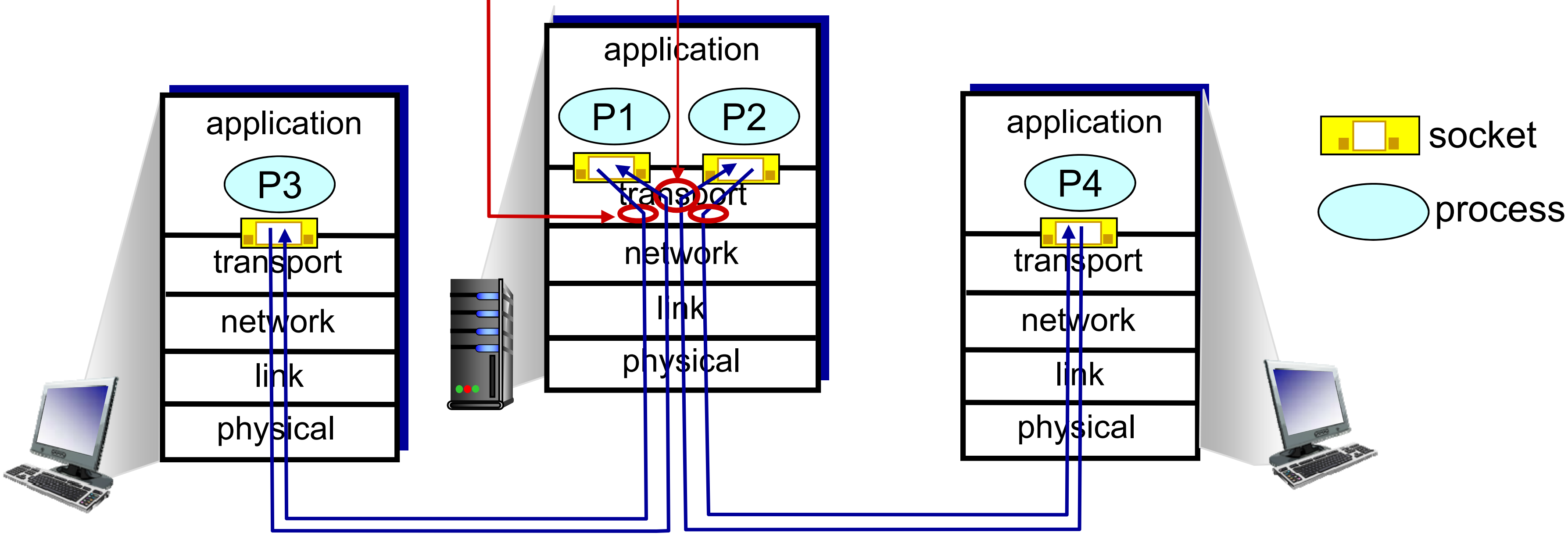
Multiplexing / Demultiplexing

Multiplexing at sender:

Handle data from multiple sockets, add transport header (later used for demultiplexing)

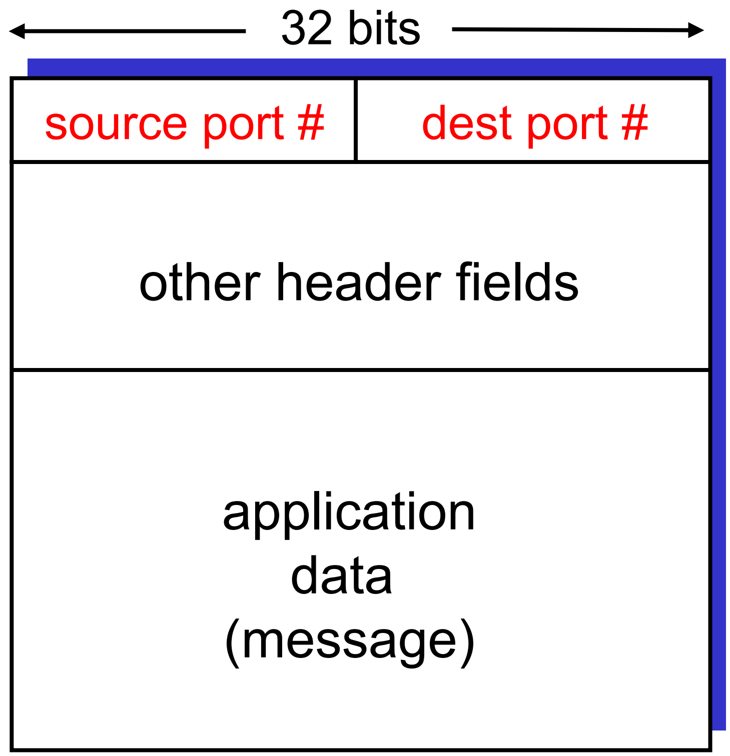
Demultiplexing at receiver:

Use header info to deliver received segments to correct socket



How Demultiplexing Works

- Host receives IP datagrams
  - Each datagram has
    - Source IP address
    - Destination IP address
  - Each datagram carries 1 transport-layer segment
- Each TCP/UDP segment has
  - Source port number
  - Destination port number
- Destination uses IP addresses & port numbers to direct segment to appropriate socket



TCP / UDP segment format

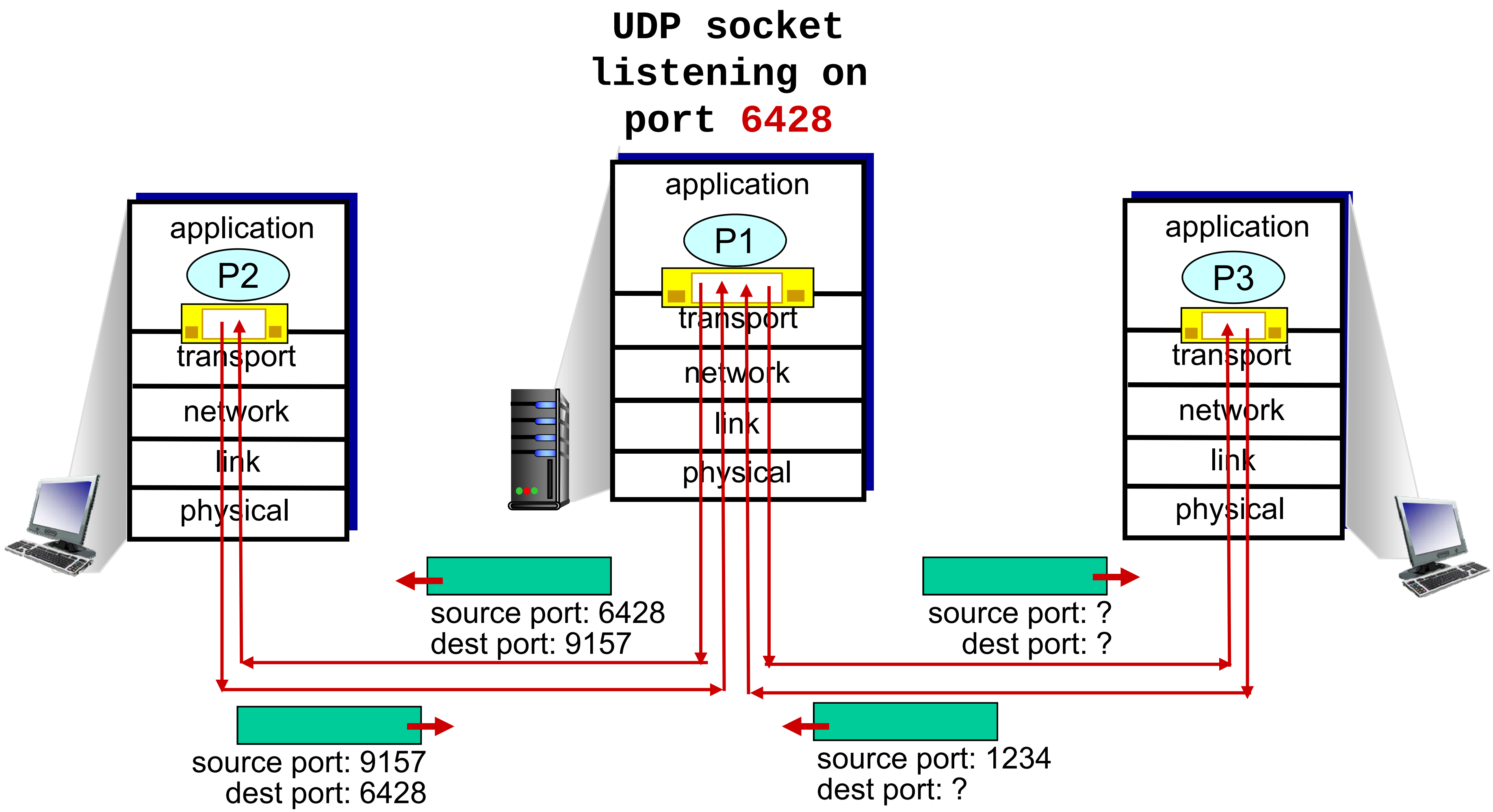
Connectionless (UDP) Demultiplexing

- Create sockets with port numbers:

```
serverSocket = socket(AF_INET, SOCK_DGRAM)
serverSocket.bind(('', 6428))
```

- 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 …
    - even if IP datagrams came from different sources
      - different source IP addresses and/or source port numbers

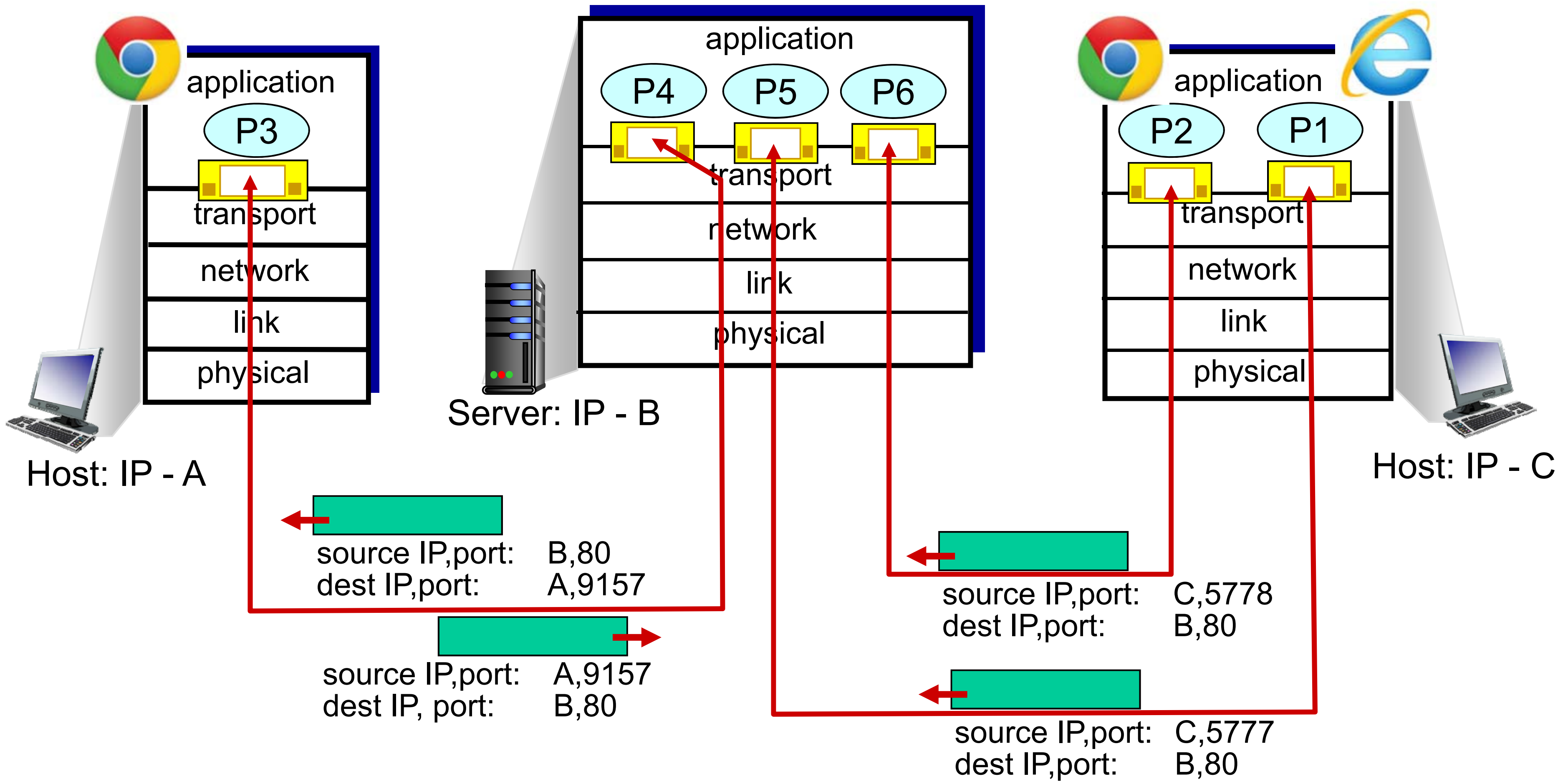
Connectionless Demux: Example



Connection-oriented (TCP) Demux

- TCP socket identified by 4-tuple:
  - Source IP address
  - Source port number
  - Dest IP address
  - Dest port number
- Receiver uses all four values to direct segment to appropriate socket
- Servers support many simultaneous TCP sockets:
  - Each socket identified by its own 4-tuple
- Web servers have different sockets for each TCP connection from a client
  - Non-persistent HTTP will have a different socket for each object request!

Connection-oriented Demux: Example



Three segments, all destined to IP address: B, dest port: 80
are demultiplexed to *different connection* sockets

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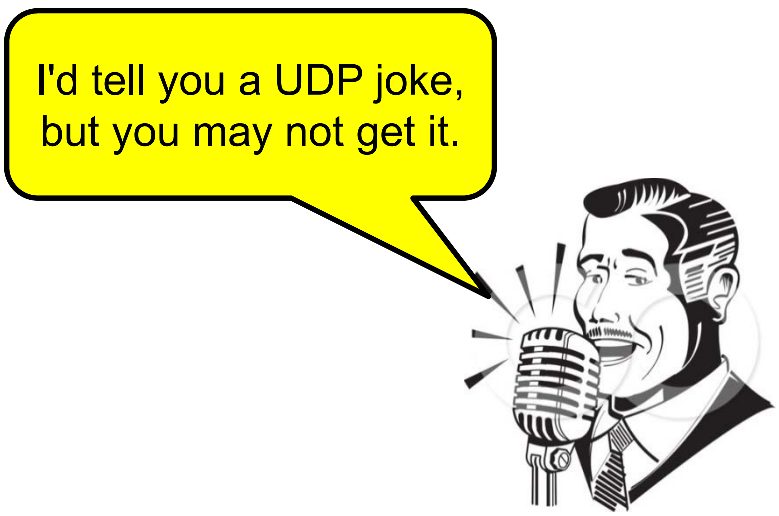
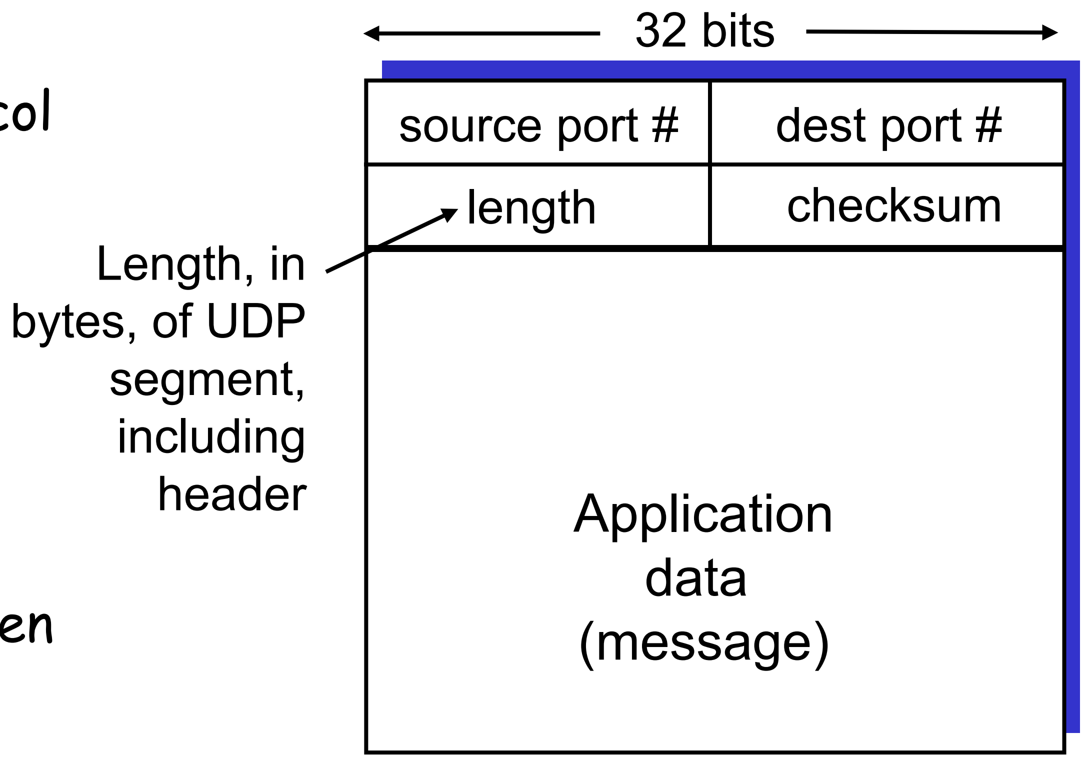


UDP: User Datagram Protocol [RFC 768]

- "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 independent of others



UDP segment format

So Why is There a UDP?

- Application can control exactly what data is sent and when
  - UDP passes data to network layer immediately
  - No congestion control: UDP can blast away as fast as desired
- No connection set up (which can add delay)
- Simple: no connection state at sender, receiver
  - No buffers and fewer variables to maintain
  - Server can support more UDP clients than TDP
- Small segment header
  - 8 bytes instead of 20 for TCP
- You want reliable transfer over UDP?
  - Add reliability at the application layer
  - Application-specific error recovery!

UDP Checksum

Goal: detect "errors" (e.g., flipped bits) in received segment

Sender:

- Treat entire segment contents as sequence of 16-bit integers
- Checksum = 1's complement of sum of segment contents
- Sender puts checksum value into UDP checksum field

Receiver:

- Add all 16-bit integers plus the checksum of received segment
- If result is FFFF:
  - No error detected
    - *But there may be errors nonetheless?* More later
- If result contains a 0: error!
  - Discard segment (UDP) or
  - Pass segment to app with warning
    - UDP Lite - RFC 3828

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
                                                1
                ----------------------------------
sum  -->          1 0 1 1 1 0 1 1 1 0 1 1 1 1 0 0

1's comp
of sum is  -->    0 1 0 0 0 1 0 0 0 1 0 0 0 0 1 1
checksum
```

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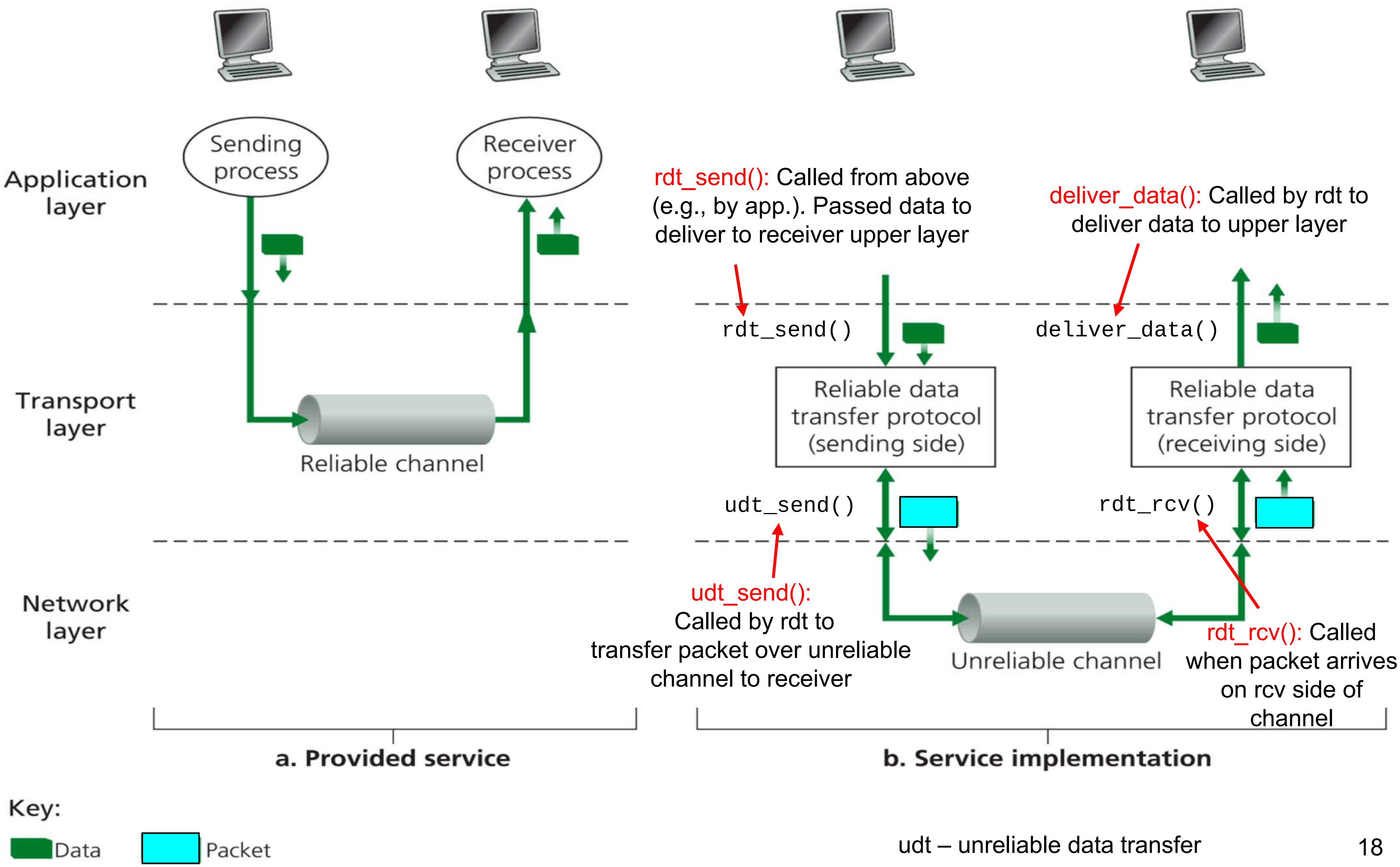
Principles of Reliable Data Transfer (rdt)

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

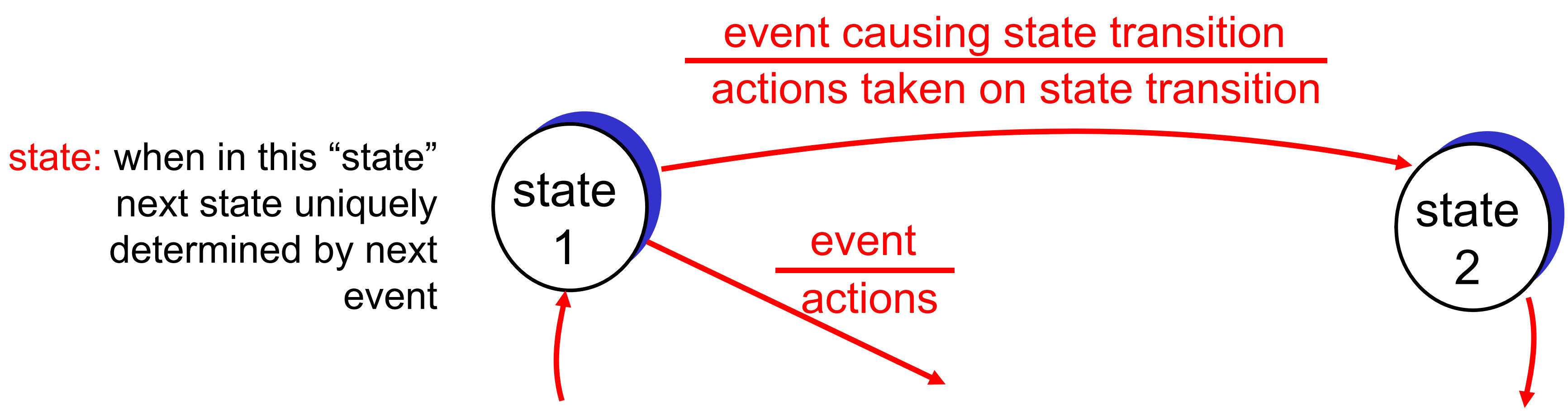


udt – unreliable data transfer

Reliable Data Transfer: Getting Started

We'll:

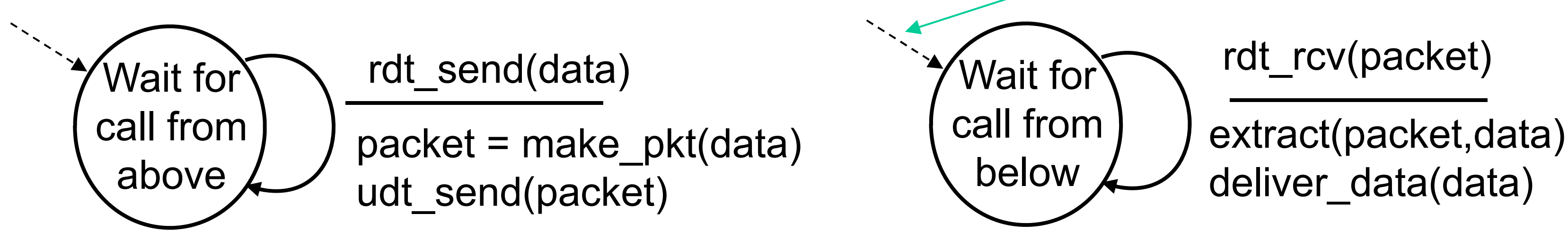
- Incrementally develop sender, receiver sides of reliable data transfer protocol (rdt)
- Consider only unidirectional data transfer
  - but control info will flow in both directions!
- Use finite state machines (FSM) to specify sender, receiver



rdt1.0: Reliable Transfer Over a Reliable Channel

- Underlying channel perfectly reliable
  - No bit errors
  - No loss of packets
  - No flow control required
- Separate FSMs for sender, receiver:
  - Sender sends data into underlying channel
  - Receiver reads data from underlying channel

What is the dotted line?



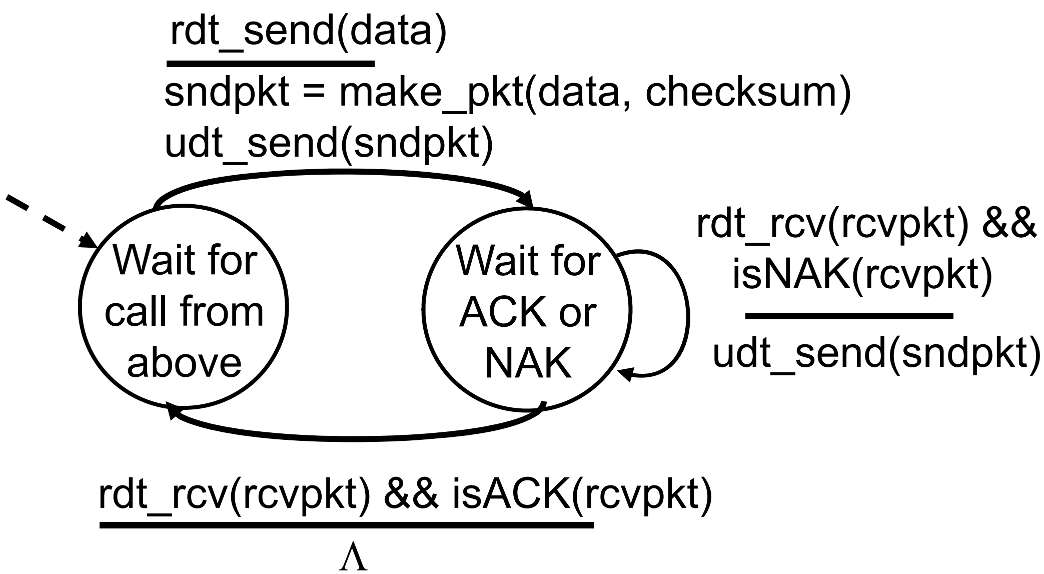
sender

receiver

rdt2.0: Channel With Bit Errors

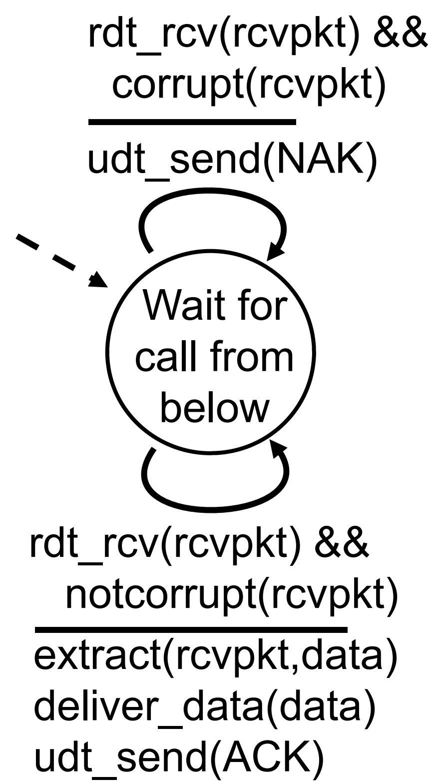
- Underlying channel may flip bits in packet
  - Use checksum to detect bit errors
- The question: how to recover from errors:
  - Acknowledgements (ACKs): receiver explicitly tells sender that pkt received OK
  - Negative acknowledgements (NAKs): receiver explicitly tells sender that pkt had errors
    - Sender retransmits pkt on receipt of NAK
- New mechanisms in **`rdt2.0`** (beyond **`rdt1.0`**):
  - Error detection
  - Receiver feedback: control msgs (ACK,NAK) rcvr to sender

rdt2.0: FSM Specification

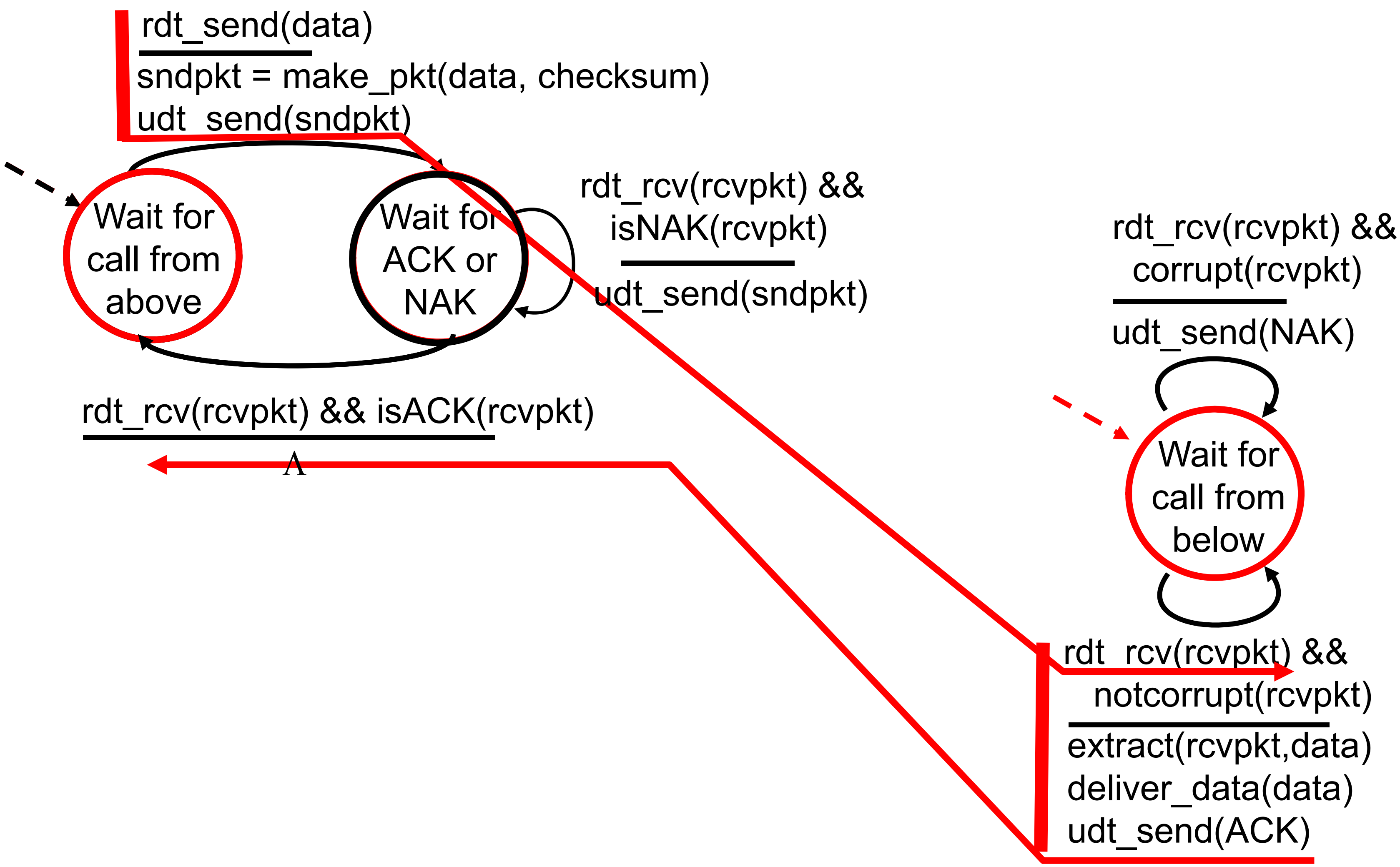


sender

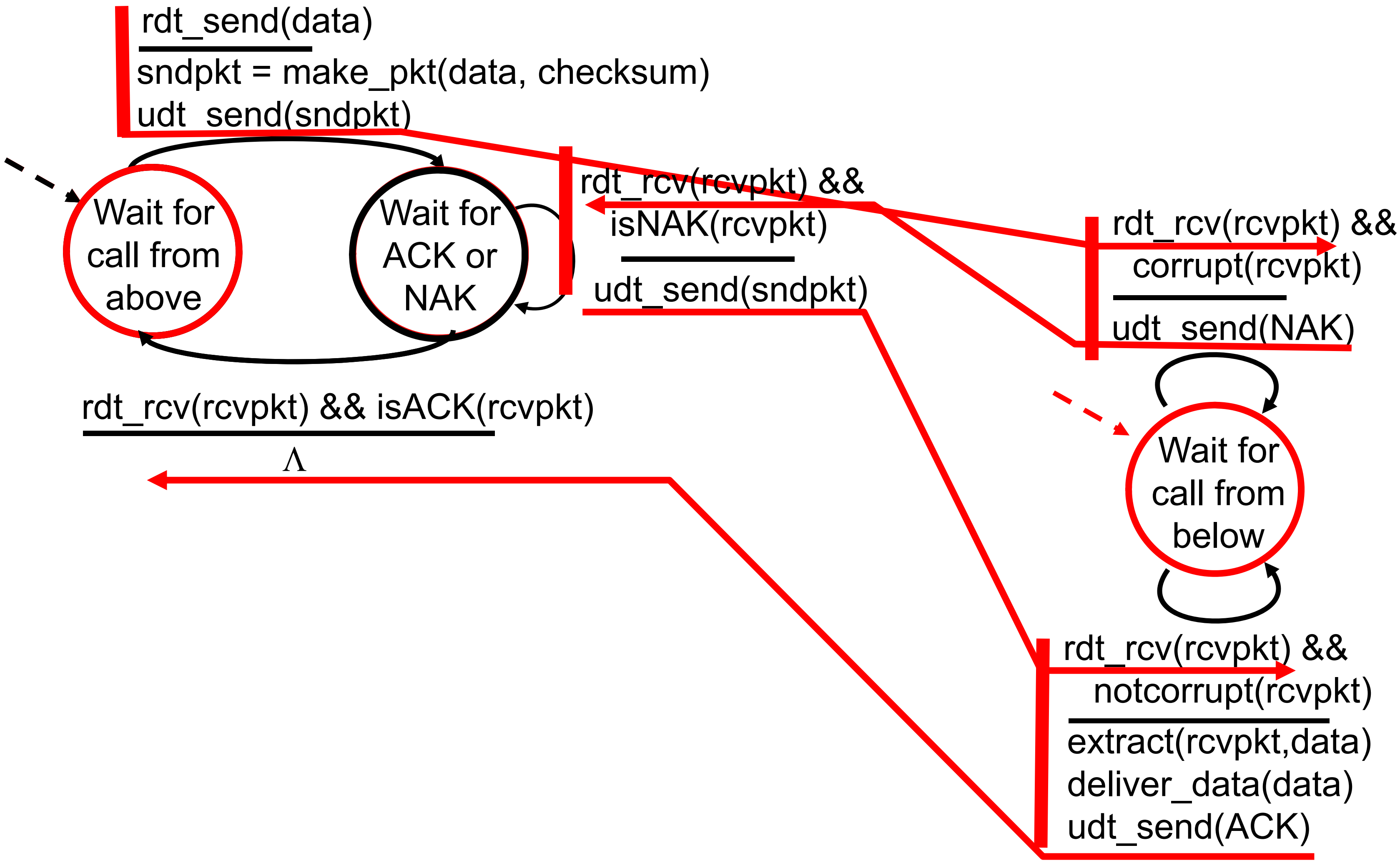
receiver



rdt2.0: Operation With No Errors



rdt2.0: Error Scenario



rdt2.0 Has a Fatal Flaw!

What if ACK/NAK corrupted?

- Sender doesn't know what happened at receiver!
- Can't just retransmit: possible duplicate

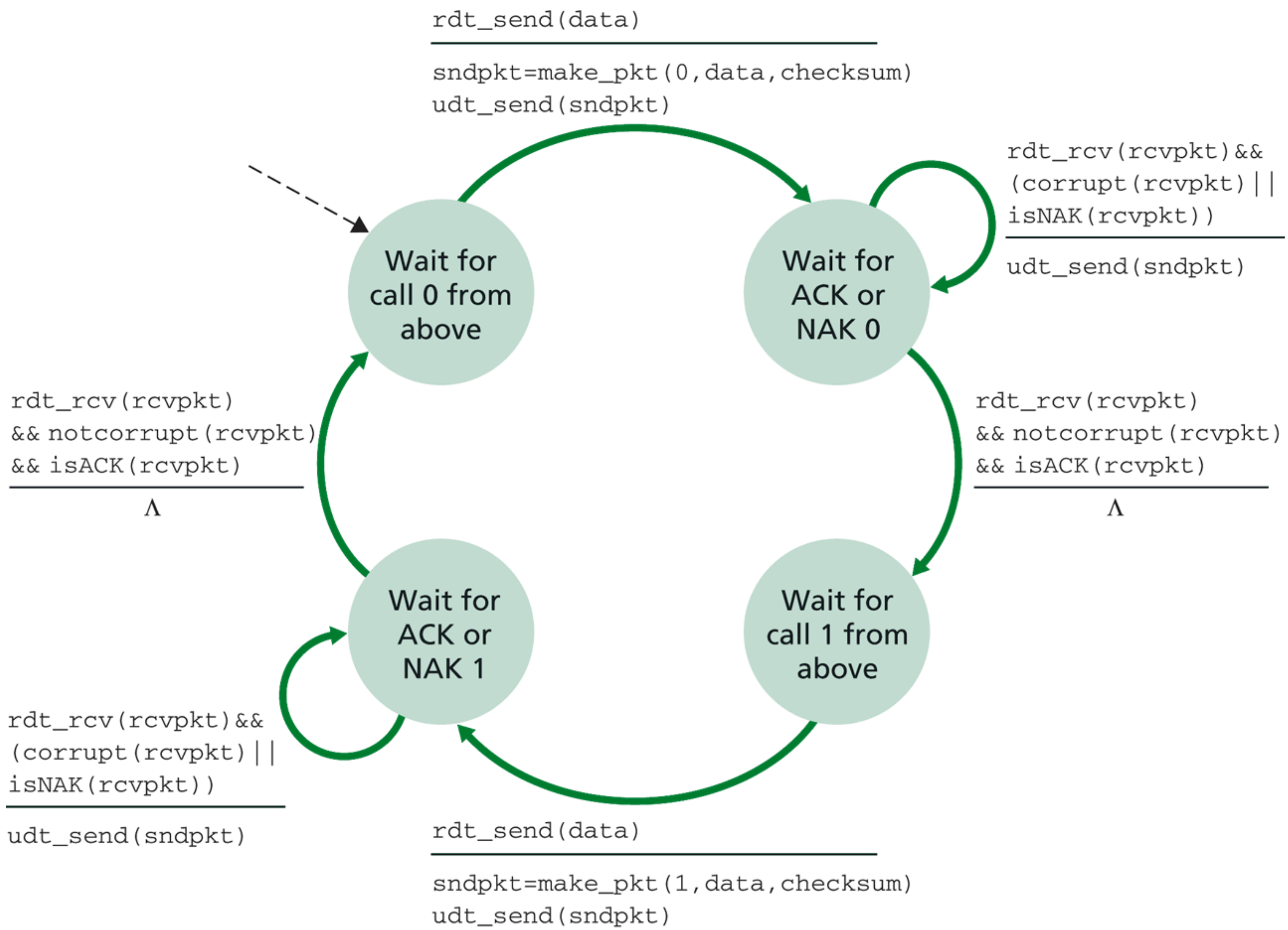
Handling duplicates:

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

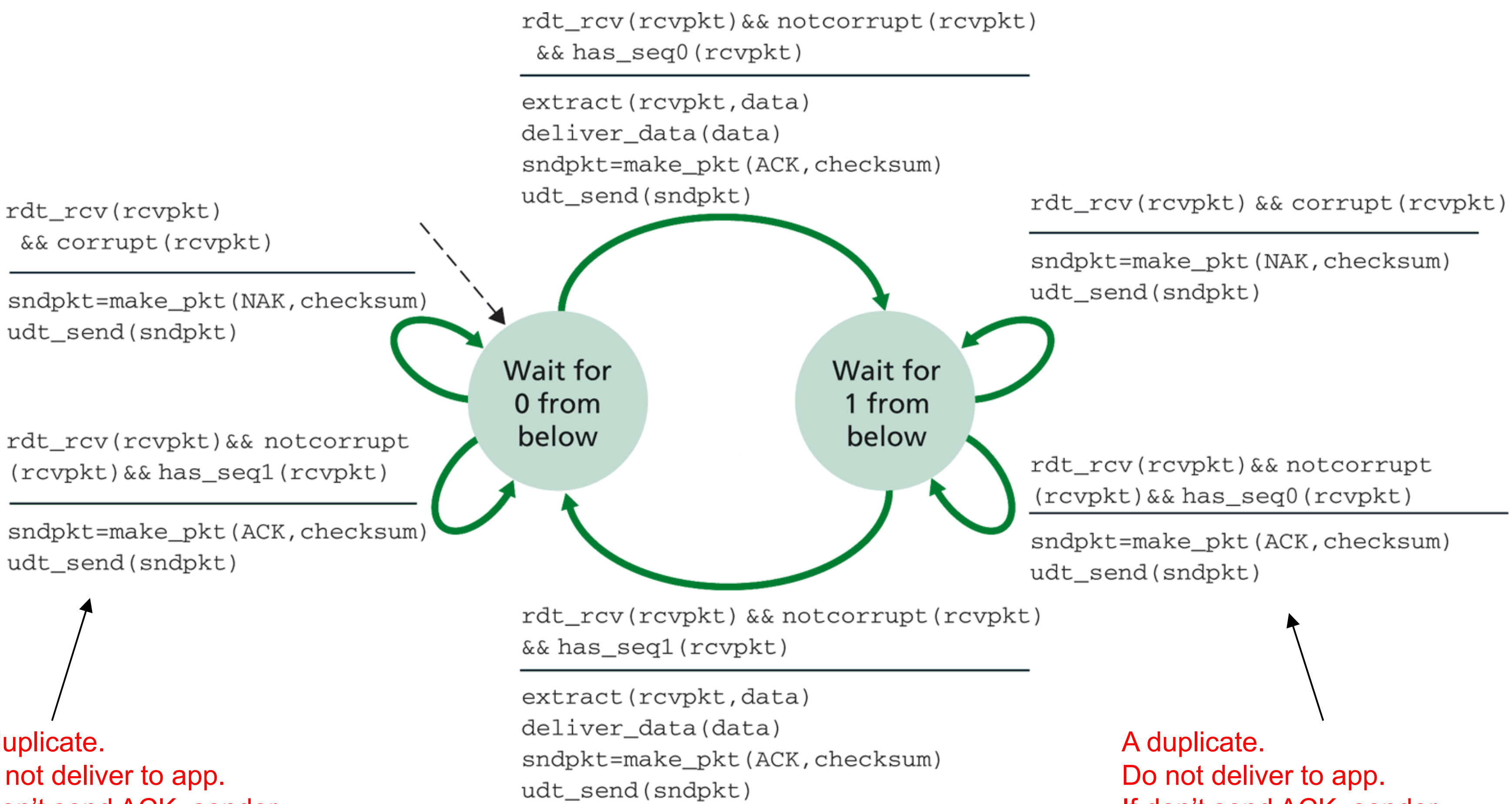
stop and wait

Sender sends one packet, then waits for receiver response before sending next packet

rdt2.1: Sender, Handles Garbled ACK/NAKs



rdt2.1: Receiver, Handles Garbled ACK/NAKs



A duplicate.
Do not deliver to app.
If don't send ACK, sender stuck in "Wait for ACK or NAK 1"

A duplicate.
Do not deliver to app.
If don't send ACK, sender stuck in "Wait for ACK or NAK 0"

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 cannot know if its last ACK/NAK received OK at sender

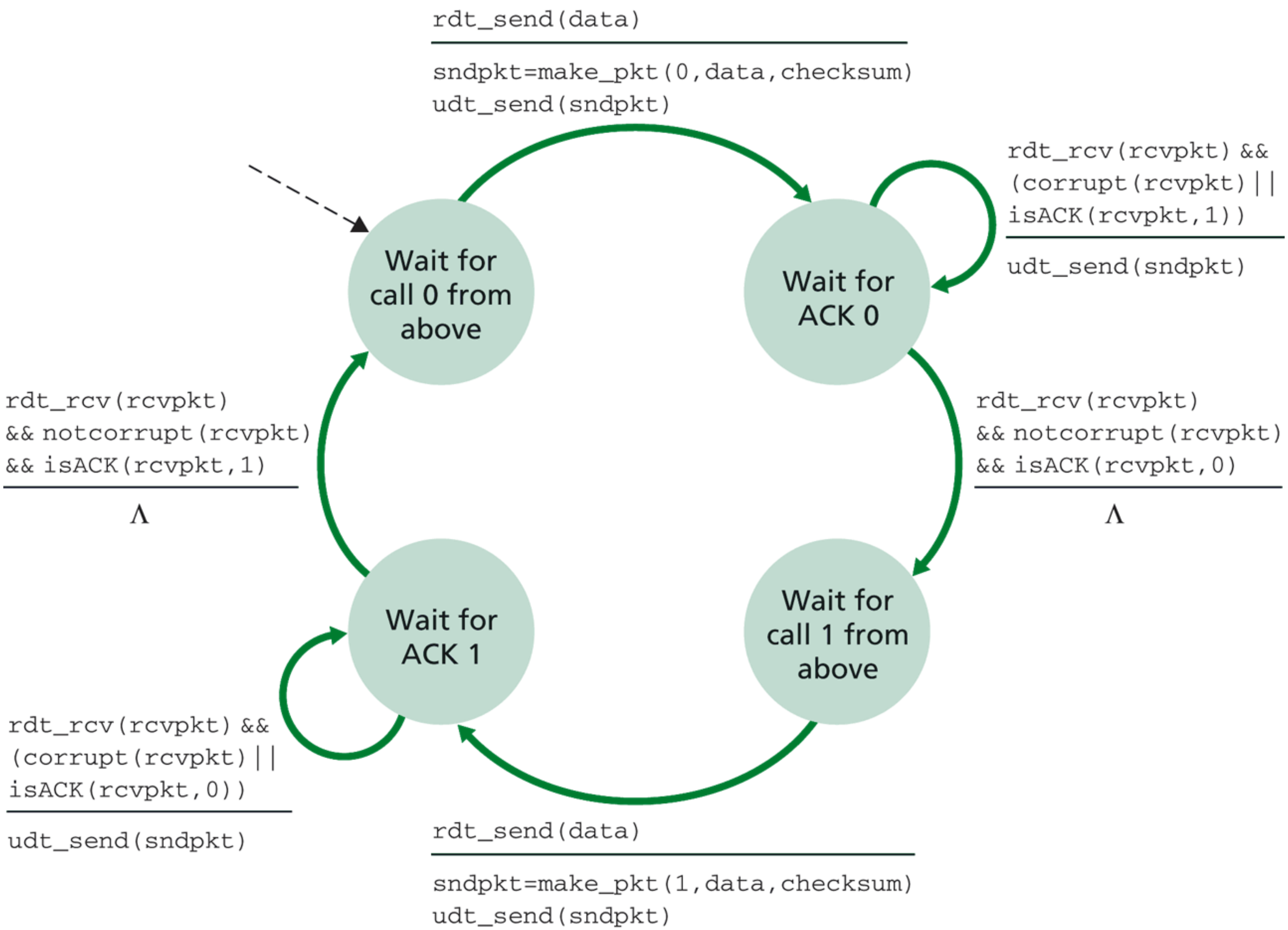
rdt2.2: A NAK-free Protocol

- Same functionality as rdt2.1, but using ACKs only

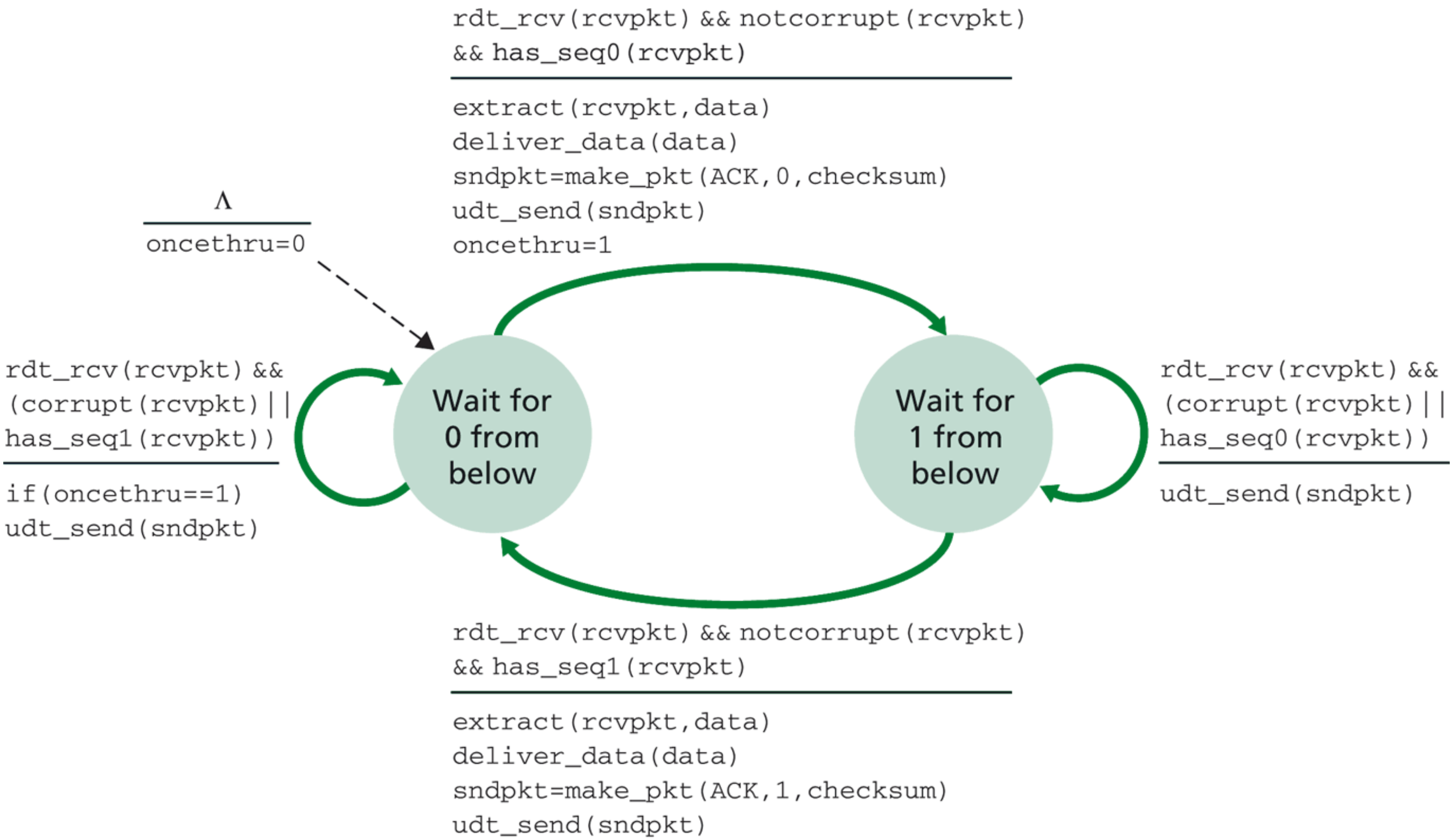
- Instead of NAK, receiver sends ACK for last pkt received OK
  - Receiver must *explicitly* include seq # of pkt being ACKed

- Duplicate ACK at sender results in same action as NAK: retransmit current pkt

rdt2.2: Sender



rdt2.2: Receiver



rdt3.0: Channels With Errors and Loss

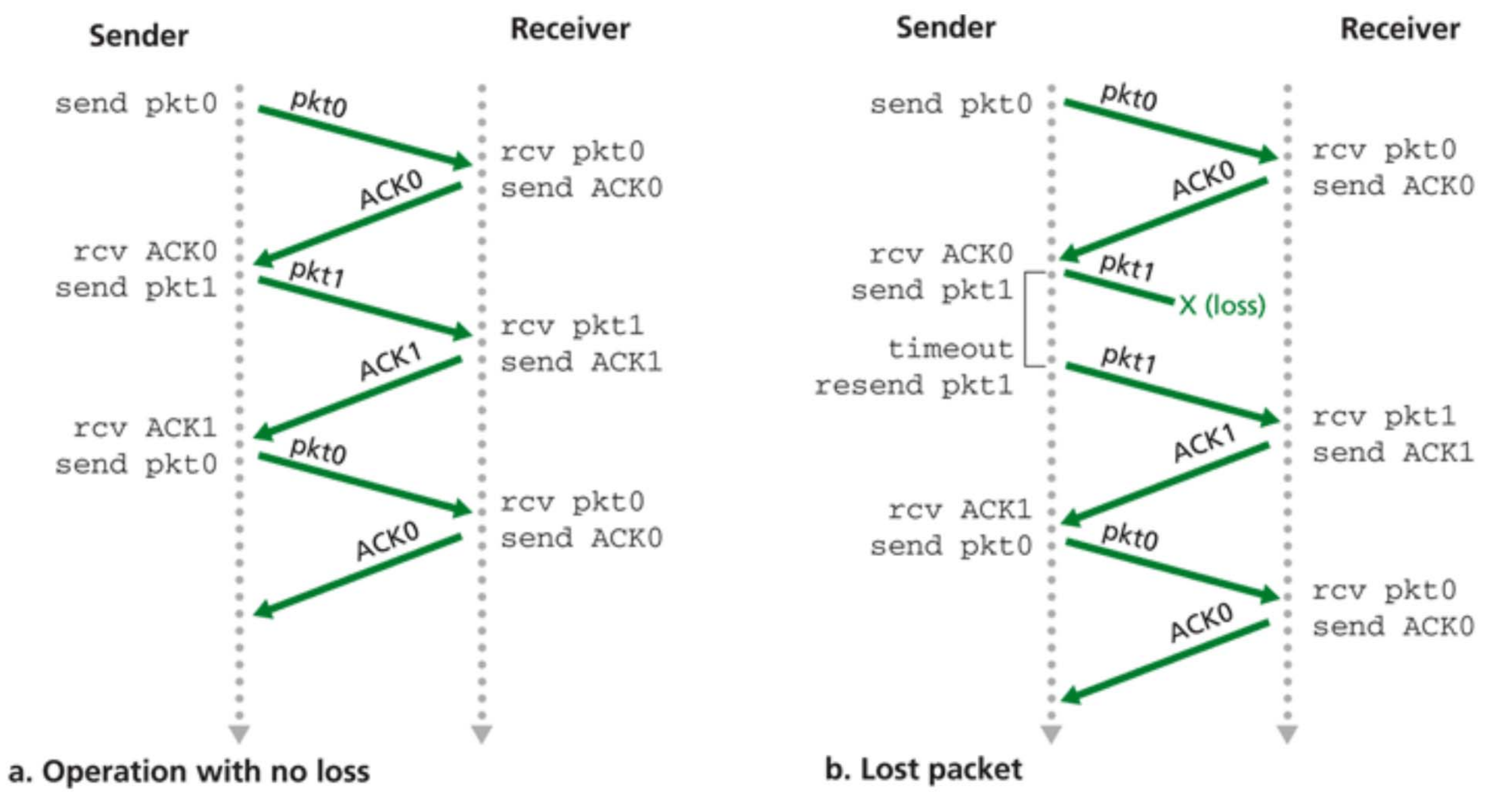
New assumption:

- Underlying channel can also lose packets (data or ACKs)
  - checksum, seq. #, ACKs, retransmissions will be of help, but not enough

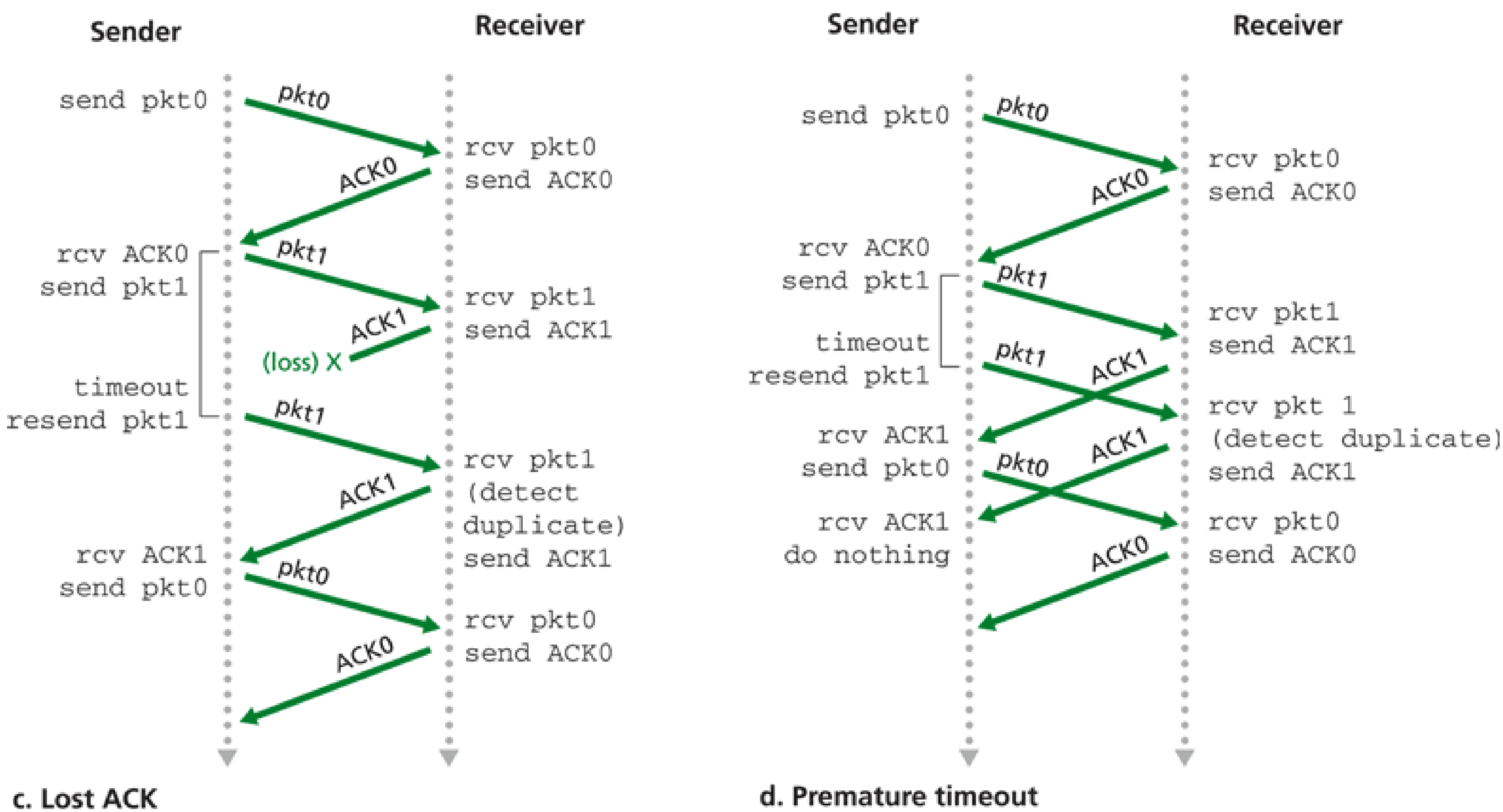
Approach:

- Sender waits "reasonable" amount of time for ACK
  - Requires countdown timer
- 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

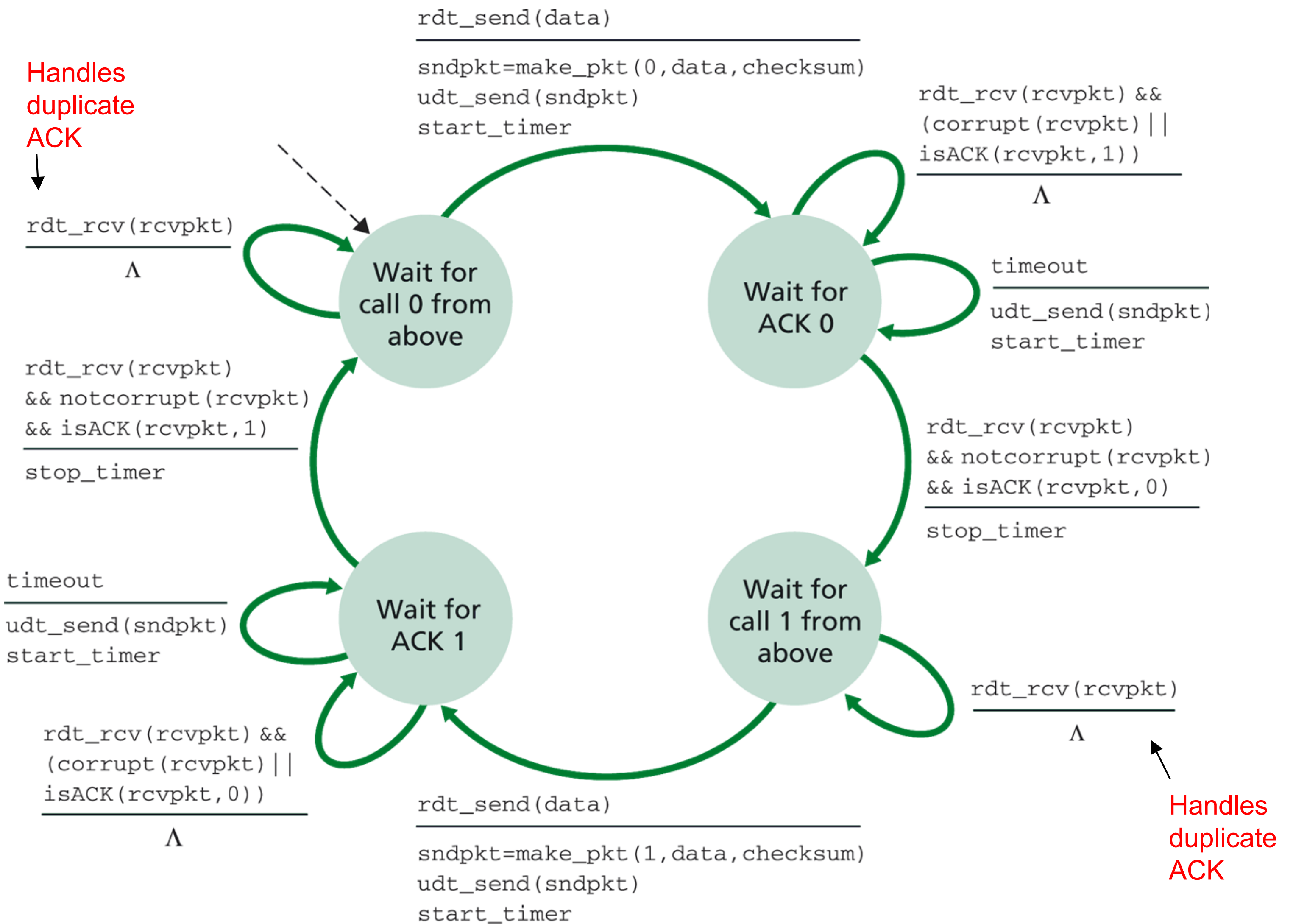
rdt3.0 In Action



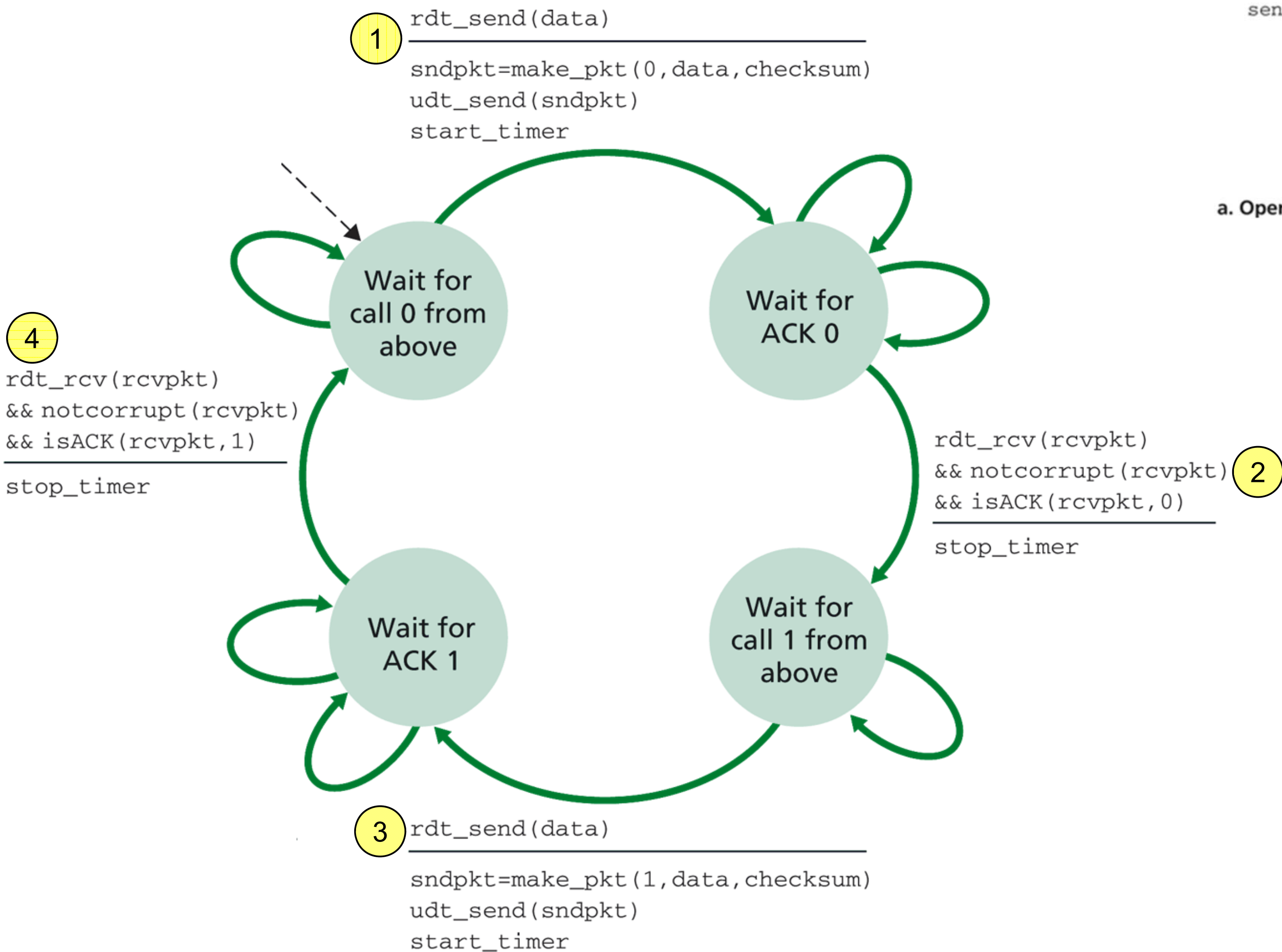
rdt3.0 In Action

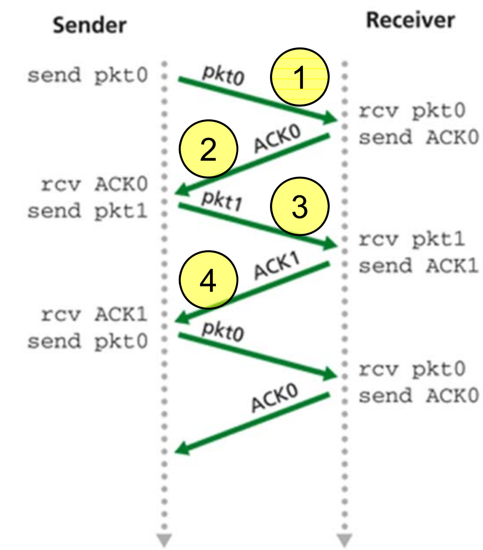


rdt3.0 Sender



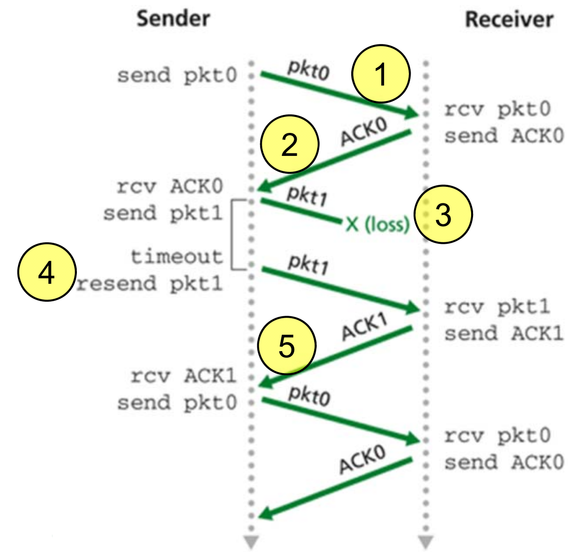
rdt3.0 Sender



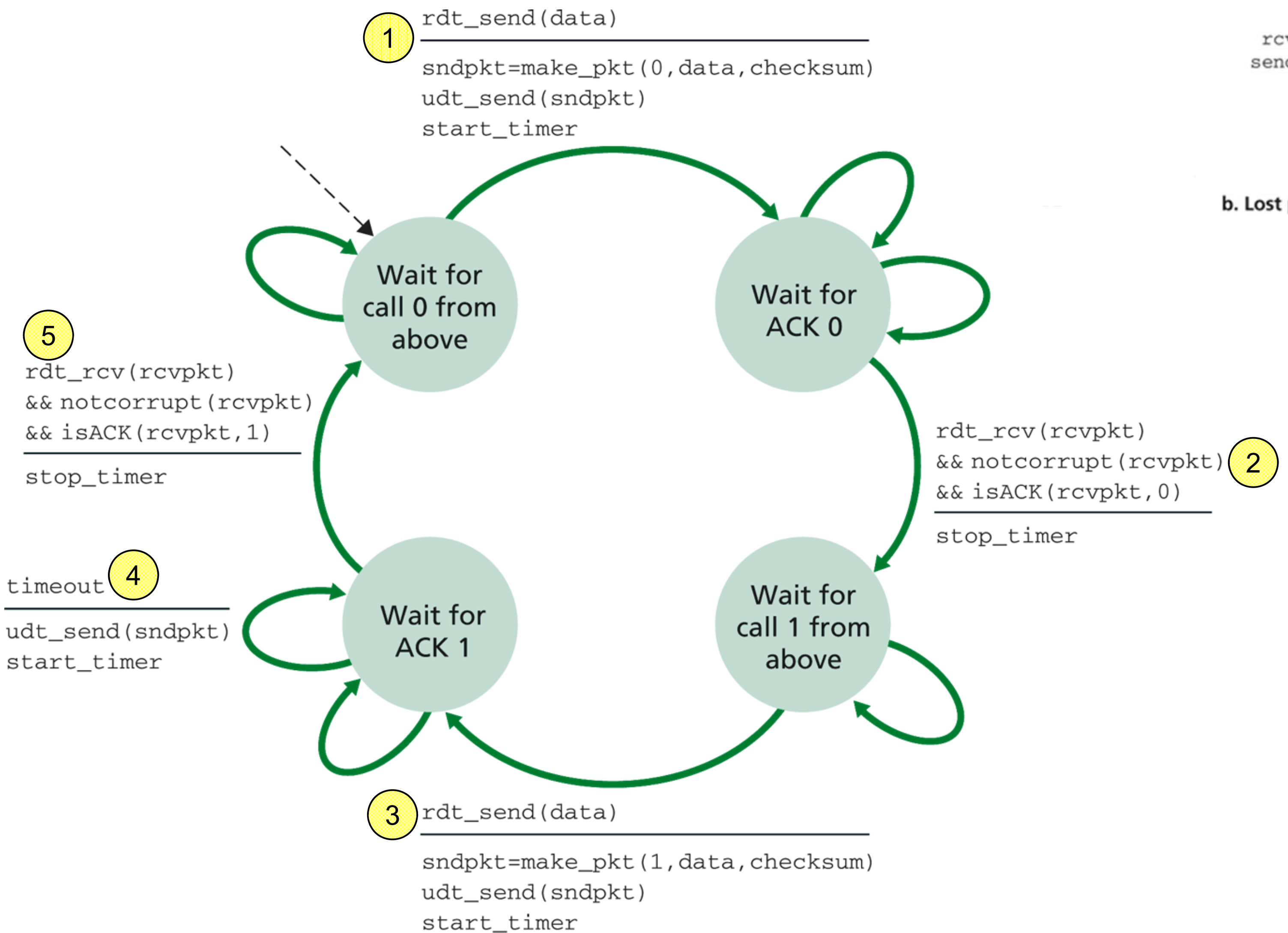


a. Operation with no loss

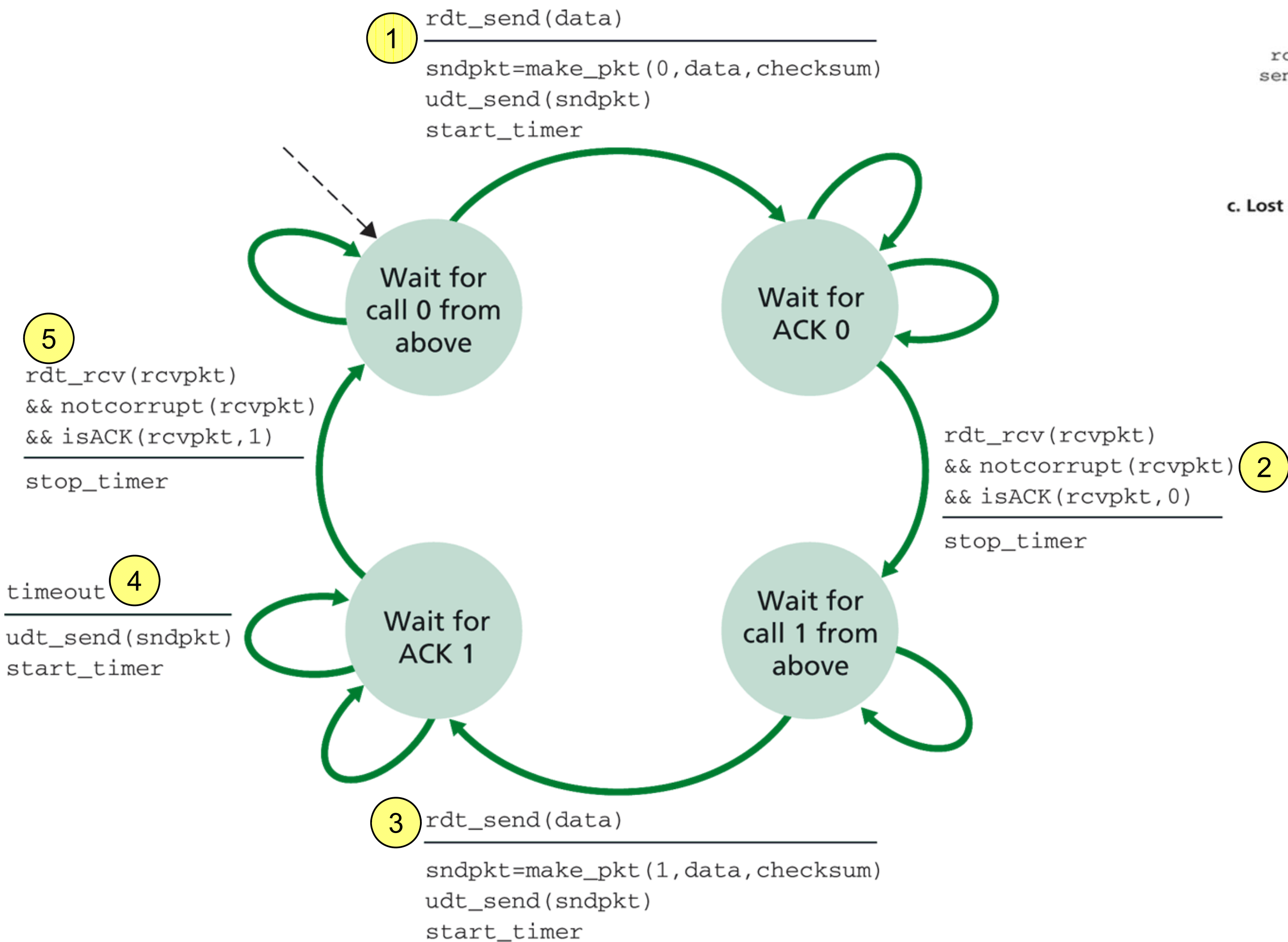
rdt3.0 Sender

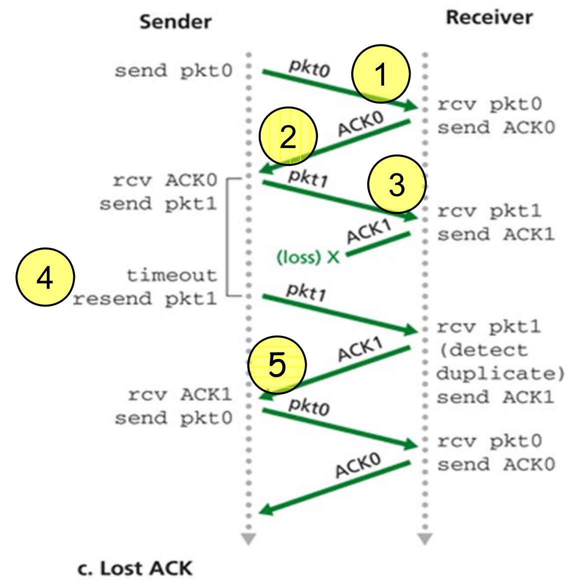


b. Lost packet

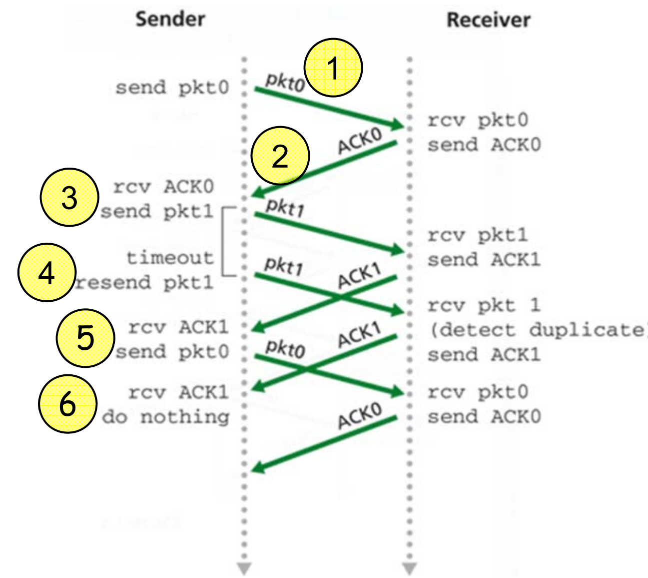


rdt3.0 Sender

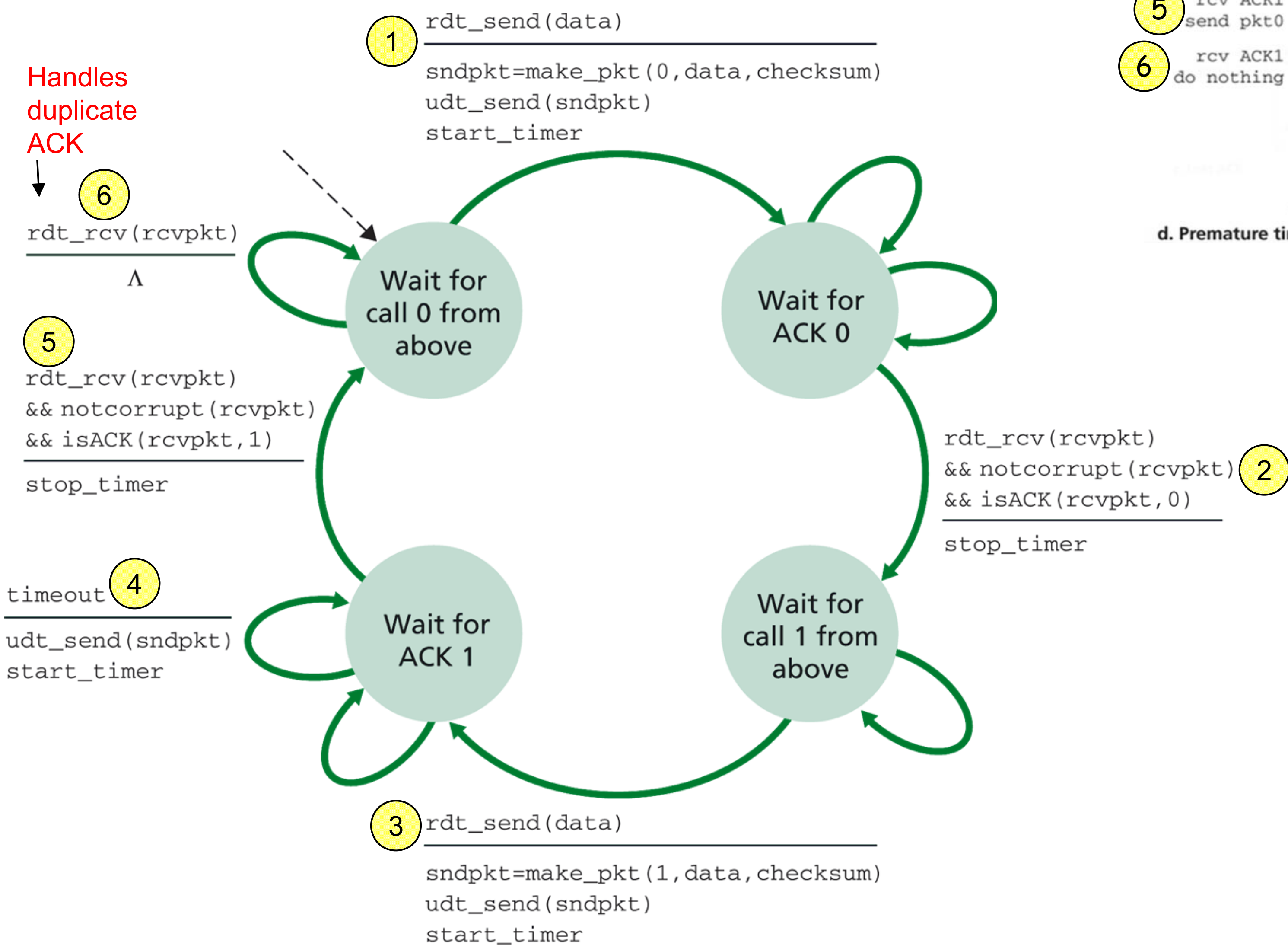




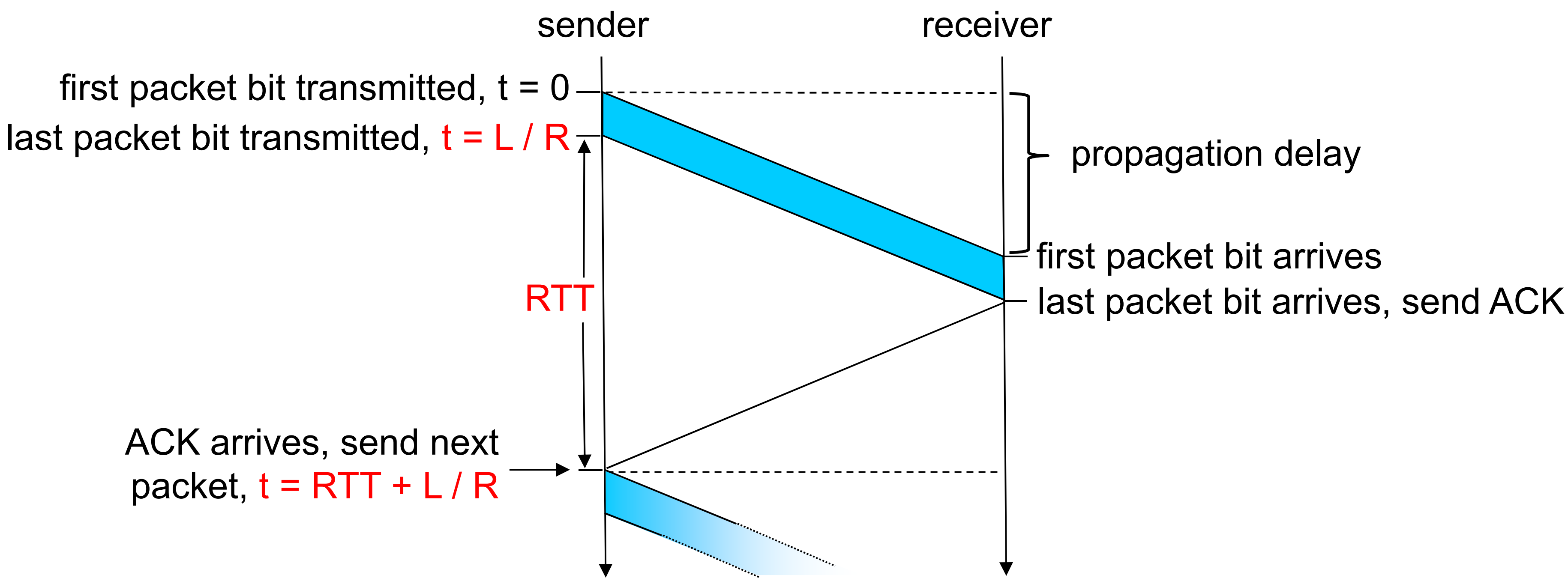
rdt3.0 Sender



d. Premature timeout



rdt3.0: Stop-and-Wait Operation / Performance



Performance of rdt3.0

- rdt3.0 works, but performance stinks
- Example: 1 Gbps link, 15 ms end-to-end prop. delay, 1000 byte packet:

$$T_{transmit} = \frac{L \text{ (packet length in bits)}}{R \text{ (transmission rate, bps)}} = \frac{8000\, b/pkt}{10^9\, b/\sec} = 8 \mu s/pkt$$

$$U = \frac{\frac{L}{R}}{\frac{L}{R} + RTT} = \frac{8 x 10^{-6}}{30.008 x 10^{-3}} = 0.00027$$

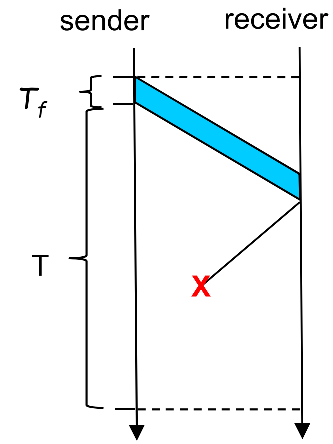
- U $_{sender}$: utilization → fraction of time sender busy sending
  - We want U $_{sender}$ to be closer to 1, not 0.00027!
- 1000 B pkt every 30 msec -> 267 kbps thruput over 1 Gbps link!
- Network protocol limits use of physical resources

Efficiency of Stop-and-Wait

- What length of time should the timeout (T) timer be set to?
  - Must compensate for:
    - Propagation time one way (τ)
    - Processing time at each station (T_p)
    - Time to transmit an acknowledgment (T_{ack})
  - Normally timeout (T) value is > $\underbrace{2\tau}_{RTT} + T_{ack} + 2T_p$
- Let's assume
  - processing time (T_p) is relatively negligible
  - acknowledgment frame time (T_{ack}) is small compared to data frame
  - Time to transmit one frame is T_f
- What is the efficiency of Stop-and-Wait in an errorless environment?

$$U = \frac{\frac{L}{R}}{\frac{L}{R} + RTT} = \frac{T_f}{T_f + 2\tau}$$

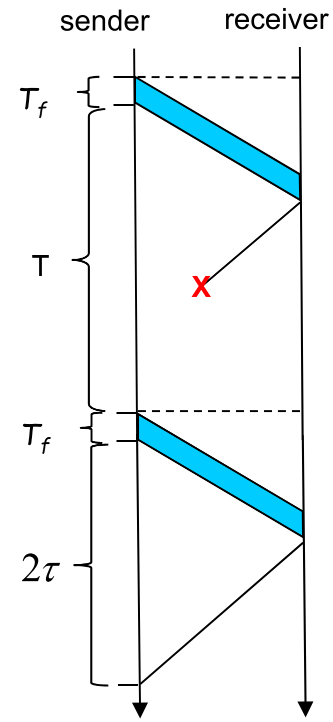
Problems with Stop-and-Wait (cont.)



- What happens in a noisy environment (i.e., errors)?
  - Let *P* be the probability of an unsuccessful transmission of a data or acknowledgment frame
  - Assume that errors on different frames are independent
  - Every unsuccessful transmission wastes $T_f + T$ seconds, where *T* is the timeout period

- Q: What is the probability of *i* transmissions per frame?
  - *A*: It is the probability of *i-1* unsuccessful transmissions followed by 1 successful transmission

Problems with Stop-and-Wait (cont.)



- We also need to know the average number of transmissions per frame, N_f

$$N_f = \sum_{i=1}^{\infty} i a_i = \sum_{i=1}^{\infty} i P^{i-1}(1-P) = \frac{1}{1-P}$$

  - first N_f - 1 transmissions waste $(T + T_f)(N_f - 1) = \dfrac{(T+T_f)P}{(1-P)}$ seconds
  - last (successful) transmission takes $T_f + 2\tau$
  - total time, T_t is the summation of the two times, so efficiency is

$$U = \frac{T_f}{\dfrac{(T+T_f)P}{(1-P)} + (T_f + 2\tau)}$$

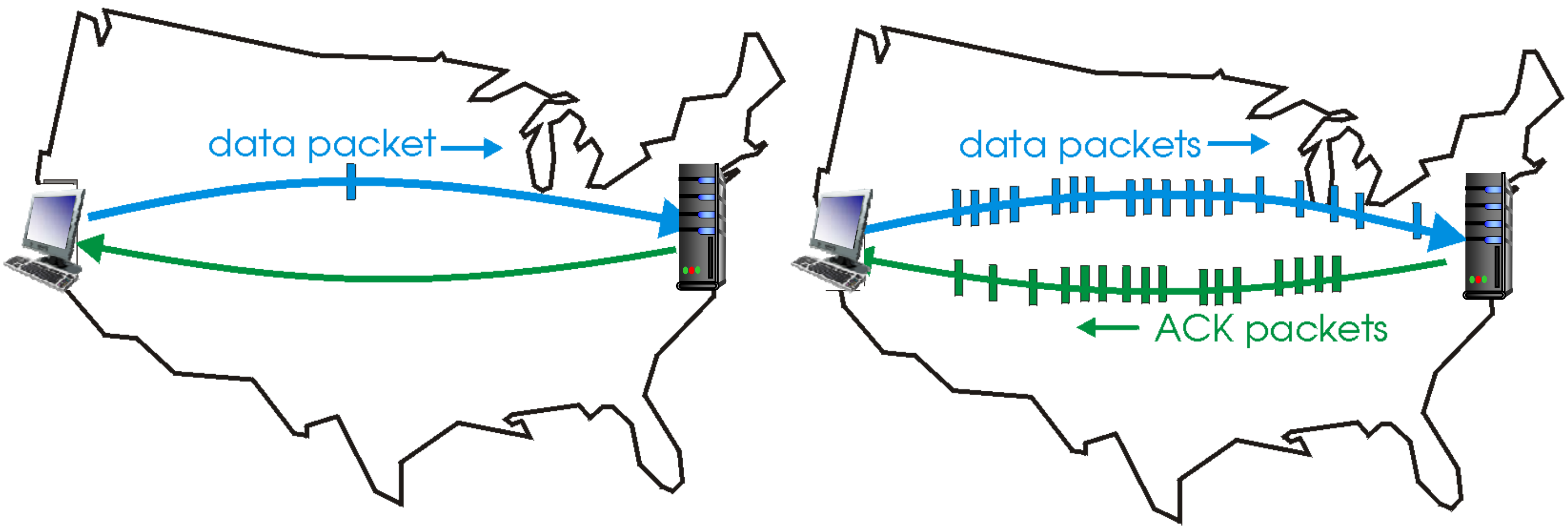
What if there are no errors (P=0)? $\longrightarrow$ $U = \dfrac{T_f}{T_f + 2\tau}$

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

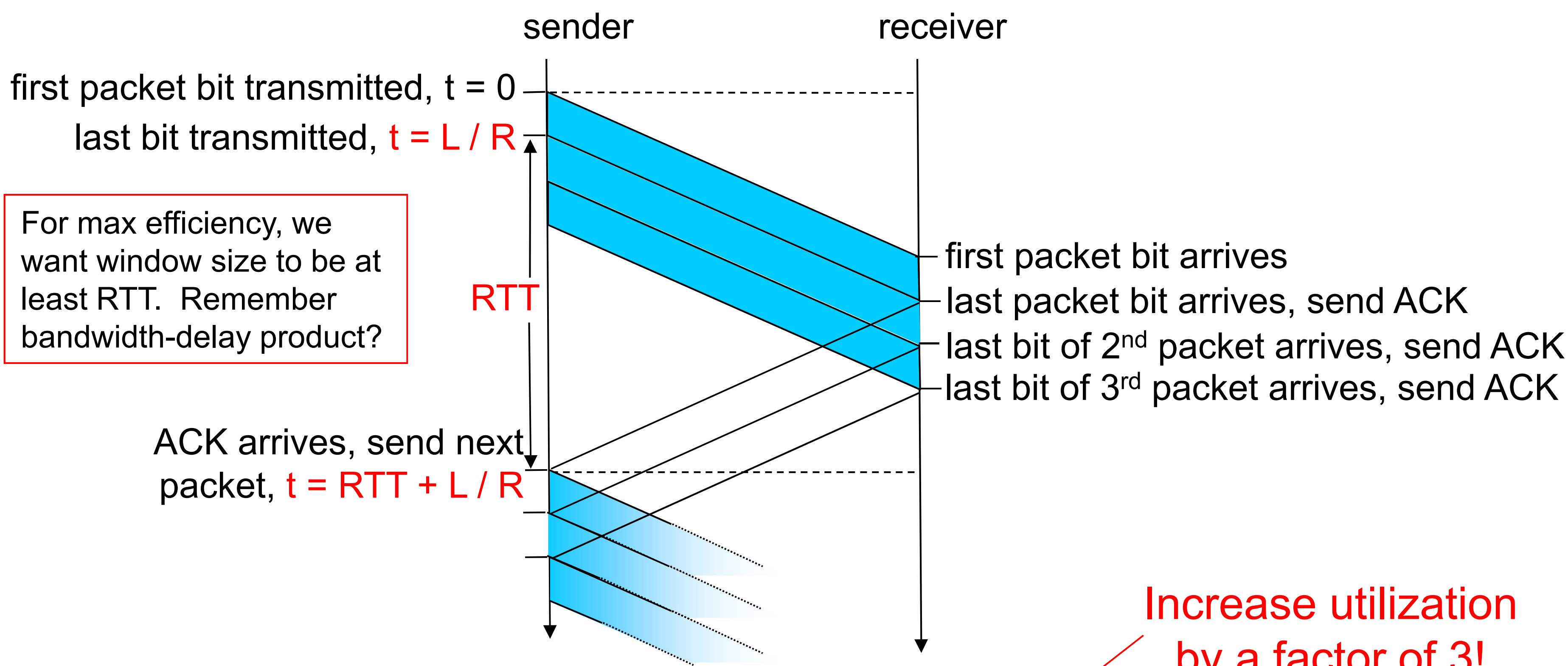
- Two generic forms of pipelined protocols: *go-Back-N, selective repeat*



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

(b) a pipelined protocol in operation

Pipelining: Increased Utilization



For max efficiency, we want window size to be at least RTT. Remember bandwidth-delay product?

$$U = \frac{3 * \frac{L}{R}}{\frac{L}{R} + RTT} = \frac{24x10^{-6}}{30.008x10^{-3}} = 0.0008$$

Increase utilization by a factor of 3!

$$U = \frac{3T_f}{T_f + 2\tau}$$

Pipelined Protocols: Overview

<u>Go-back-N:</u>

- Sender can have up to N unacked packets in pipeline

<u>Selective Repeat:</u>

- Sender can have up to N unacked packets in pipeline

Go-back-N:

- Receiver sends *cumulative ack*
  - Doesn't ack packet if there's a gap (missing packet)

Selective Repeat:

- Receiver sends *individual ack* for each packet

Go-back-N:

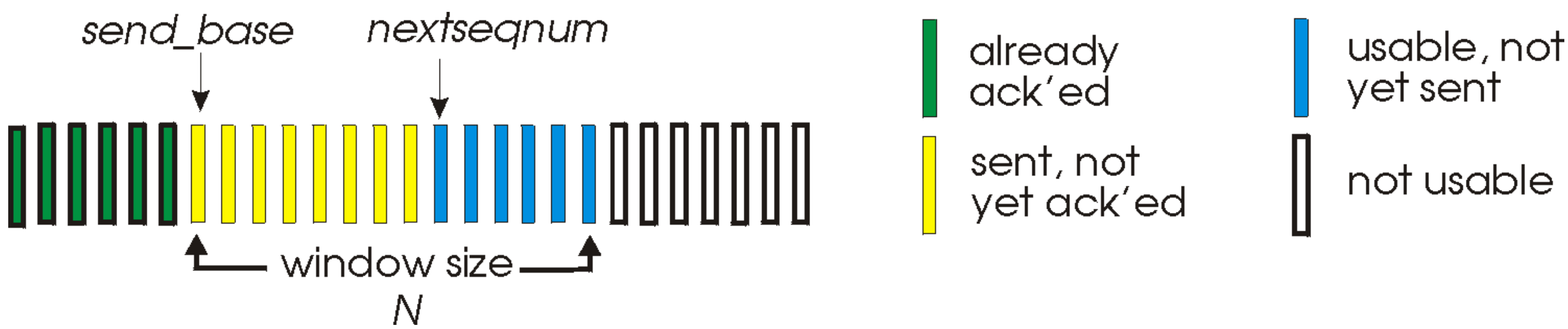
- Sender has timer for oldest unacked packet
  - When timer expires, retransmit *all* unacked packets

Selective Repeat:

- Sender has timer for each unacked packet
  - When timer expires, retransmit only that unacked packet

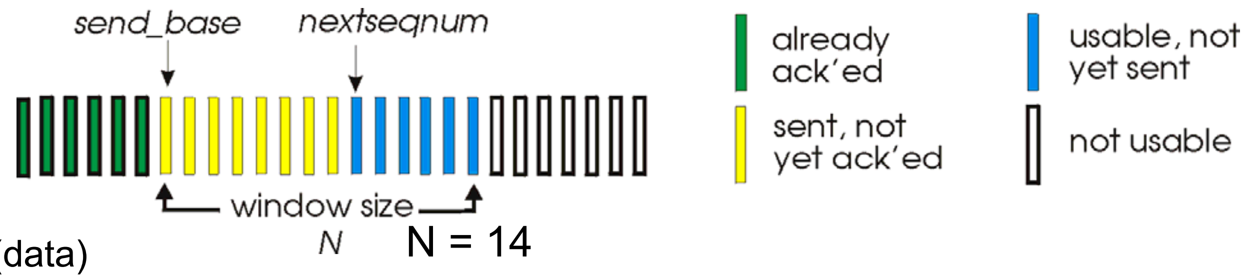
Go-Back-N (Sliding Window Protocol) Sender

- k-bit seq # in pkt header → 2^k packets
- "Window" of up to N, consecutive unack'ed pkts allowed
  - With N = 1, Go-Back-N becomes Stop-and-Wait ARQ



- ACK(n): ACKs all pkts up to, including seq # n - "cumulative ACK"
- Timer is used for in-flight packets
  - Really a timer for the oldest transmitted but not yet acked pkt
- Upon *timeout(n)*: retransmit pkt n and all higher seq # pkts in window

GBN: Sender Extended FSM



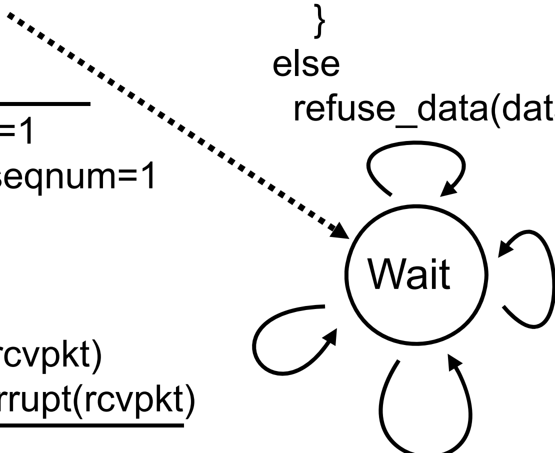
N = 14

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])
```

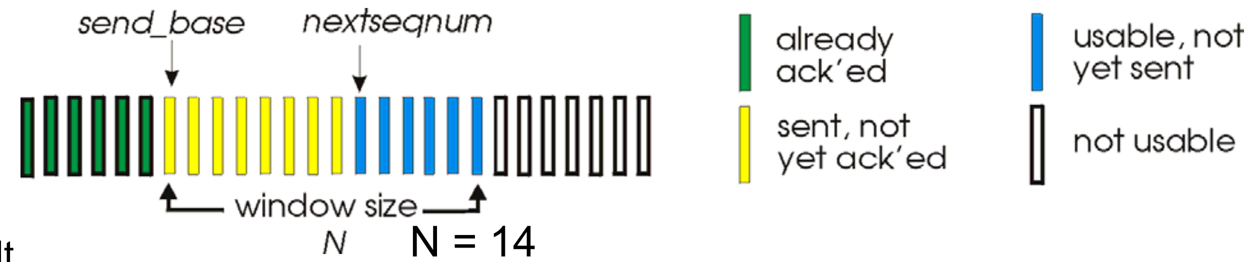
rdt_rcv(rcvpkt)
  && corrupt(rcvpkt)

Λ

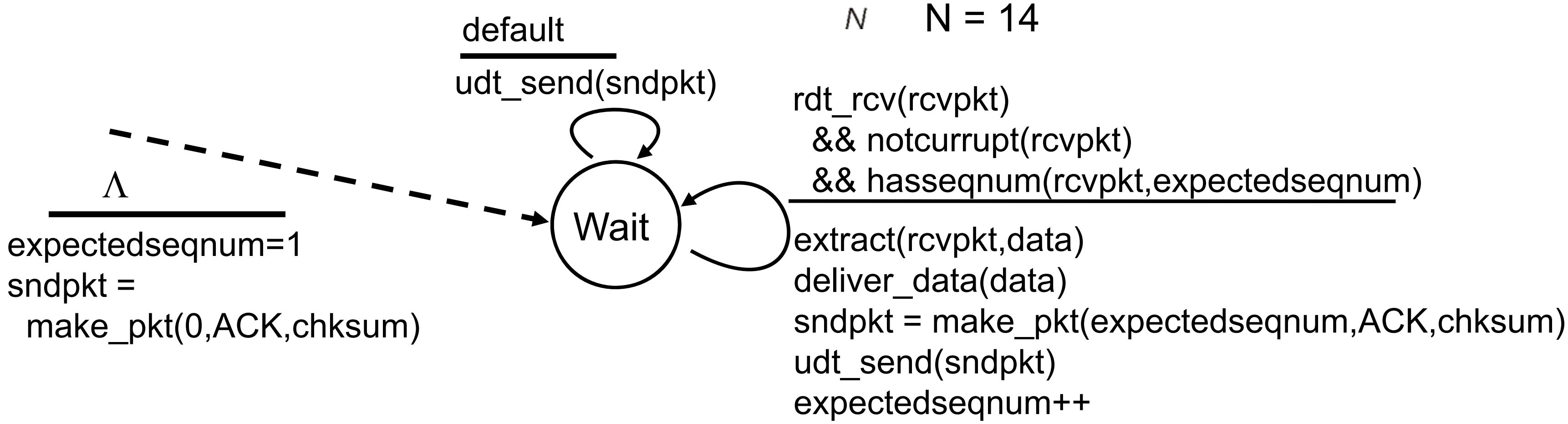
rdt_rcv(rcvpkt) &&
  notcorrupt(rcvpkt)

```
base = getacknum(rcvpkt)+1
If (base == nextseqnum)
    stop_timer
  else
    start_timer
```

GBN: Receiver Extended FSM



N = 14

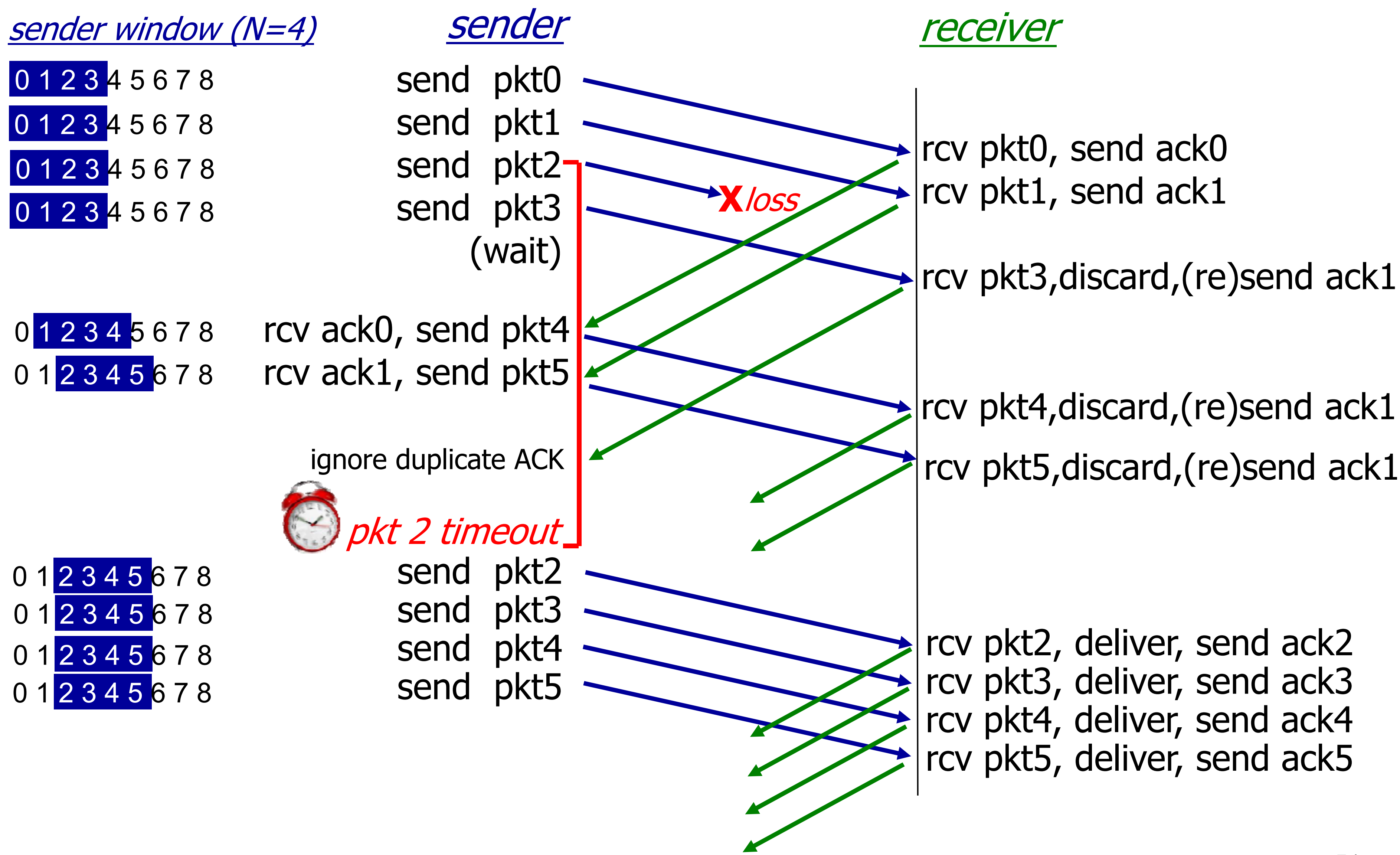


ACK-only: always send ACK for correctly-received pkt with highest *in-order* seq #

- 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

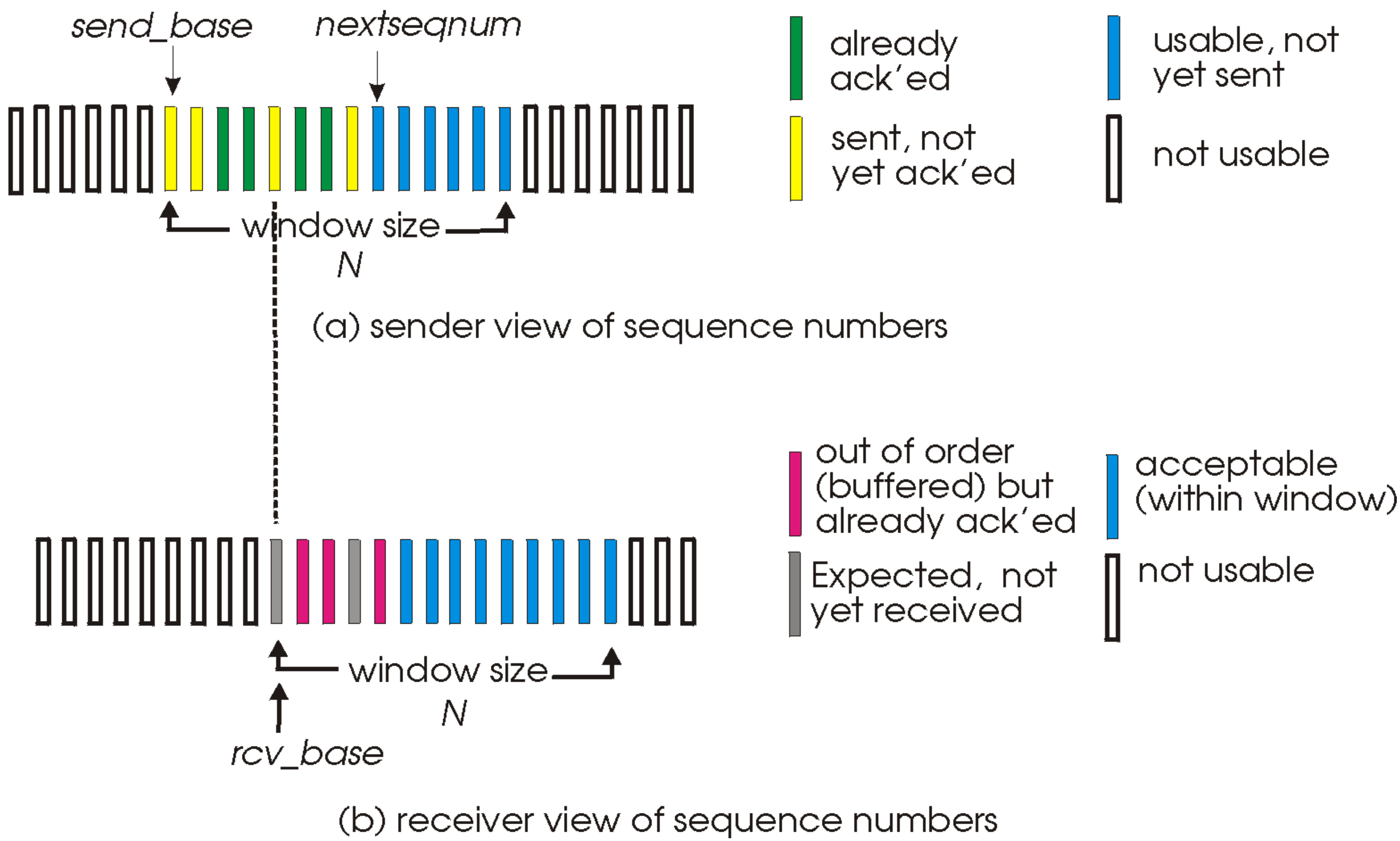


http://www.ccs-labs.org/teaching/rn/animations/gbn_sr/

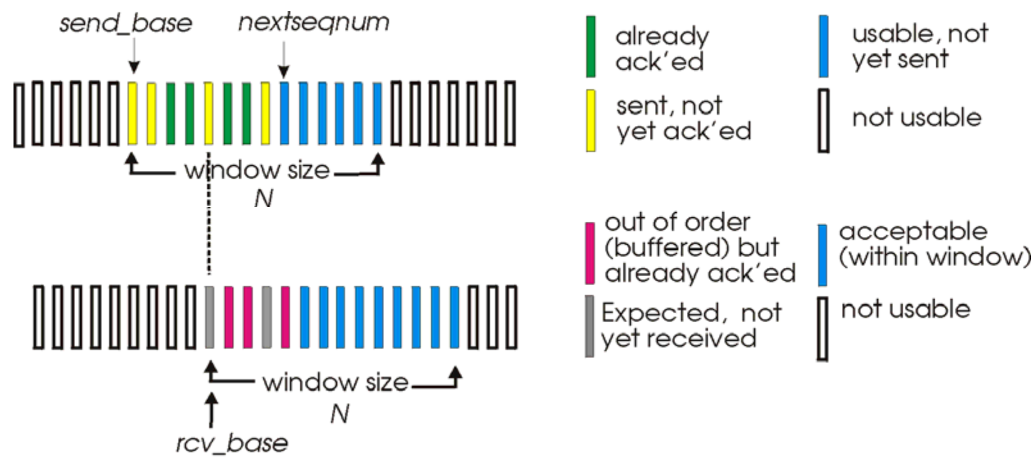
Selective Repeat

- Receiver individually acknowledges all correctly received pkts
  - Buffers pkts for eventual in-order delivery to upper layer
- Sender only resends pkts for which ACK not received
  - Sender has a timer for each unACKed pkt
- Sender window
  - N (window size) 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 # is in window, send pkt

ACK(n) in [sendbase,sendbase+N]:
- mark pkt n as received
- if n is the smallest unACKed pkt, advance window base to next unACKed seq #

timeout(n):
- resend only pkt n, restart timer

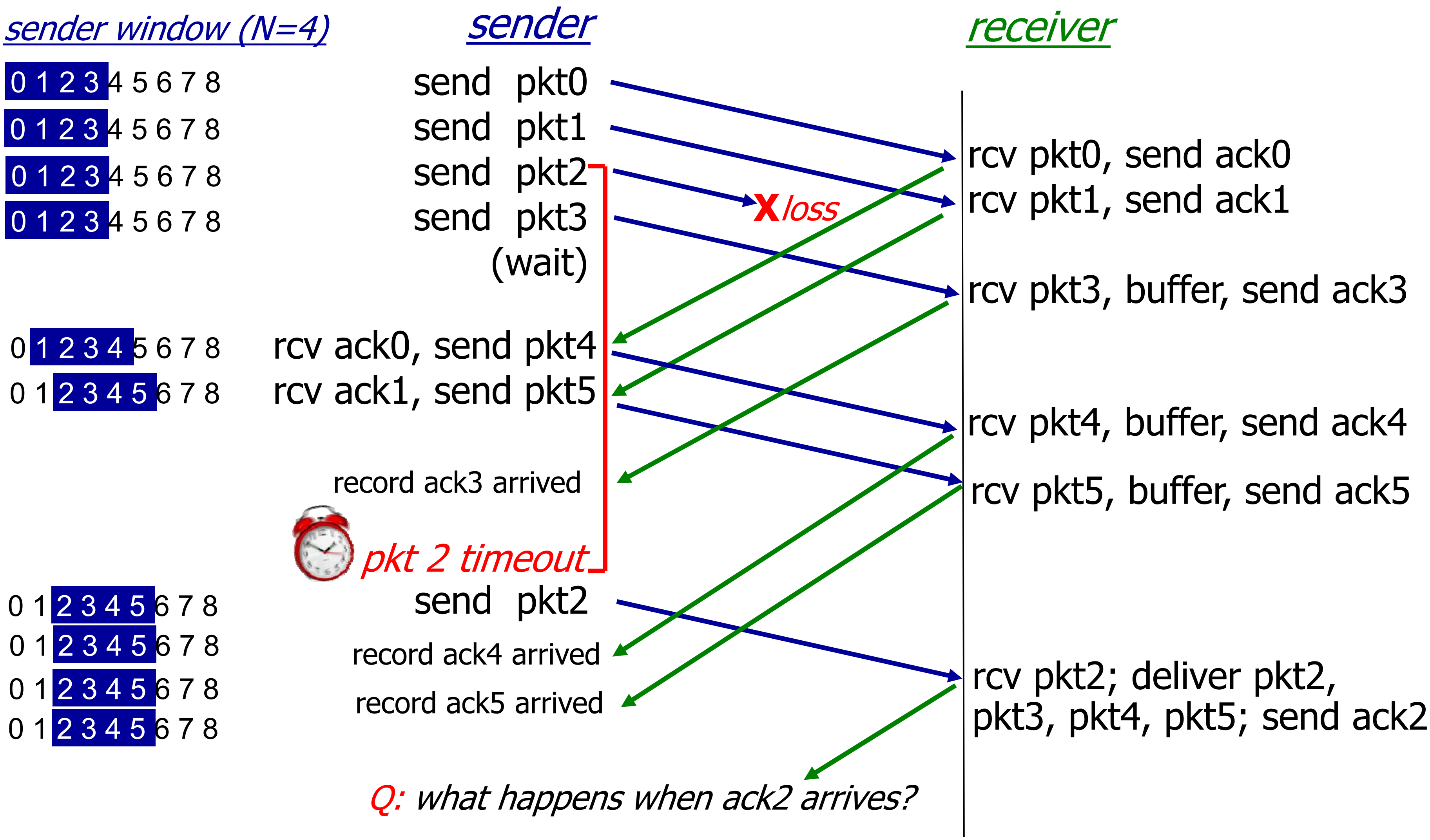
Receiver

pkt n in [rcvbase, rcvbase+N-1]
- send ACK(n)
- if out-of-order: buffer
- if in-order: deliver to app
  - also deliver buffered, in-order pkts
  - advance window to next not-yet-received pkt

pkt n in [rcvbase-N,rcvbase-1]
- ACK(n) again
- Covers the lost ACK case

otherwise:
- Ignore

Selective Repeat In Action



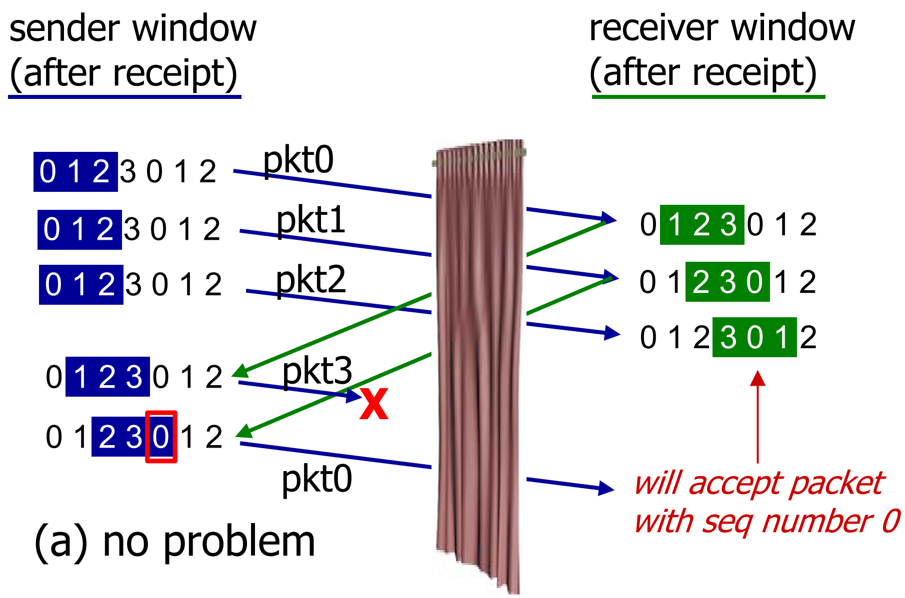
Selective Repeat: Dilemma

Example:

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

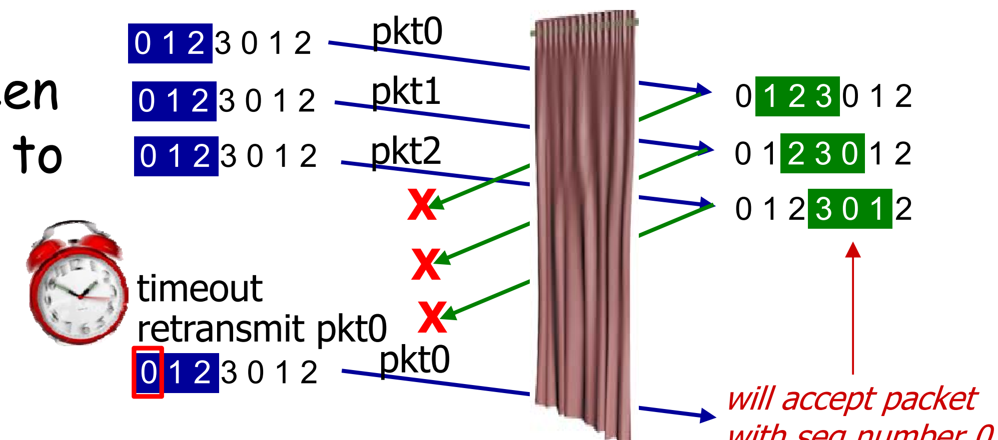
- Receiver sees no difference in two scenarios!
- Duplicate data accepted as new data in (b)

- Q: what relationship between seq # size and window size to avoid problem in (b)?
  - Seq # > 2*window



(a) no problem

receiver can't see sender side.
receiver behavior identical in both cases!
something's (very) wrong!



(b) oops!

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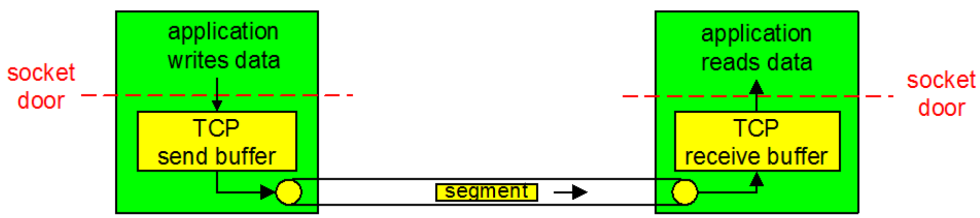
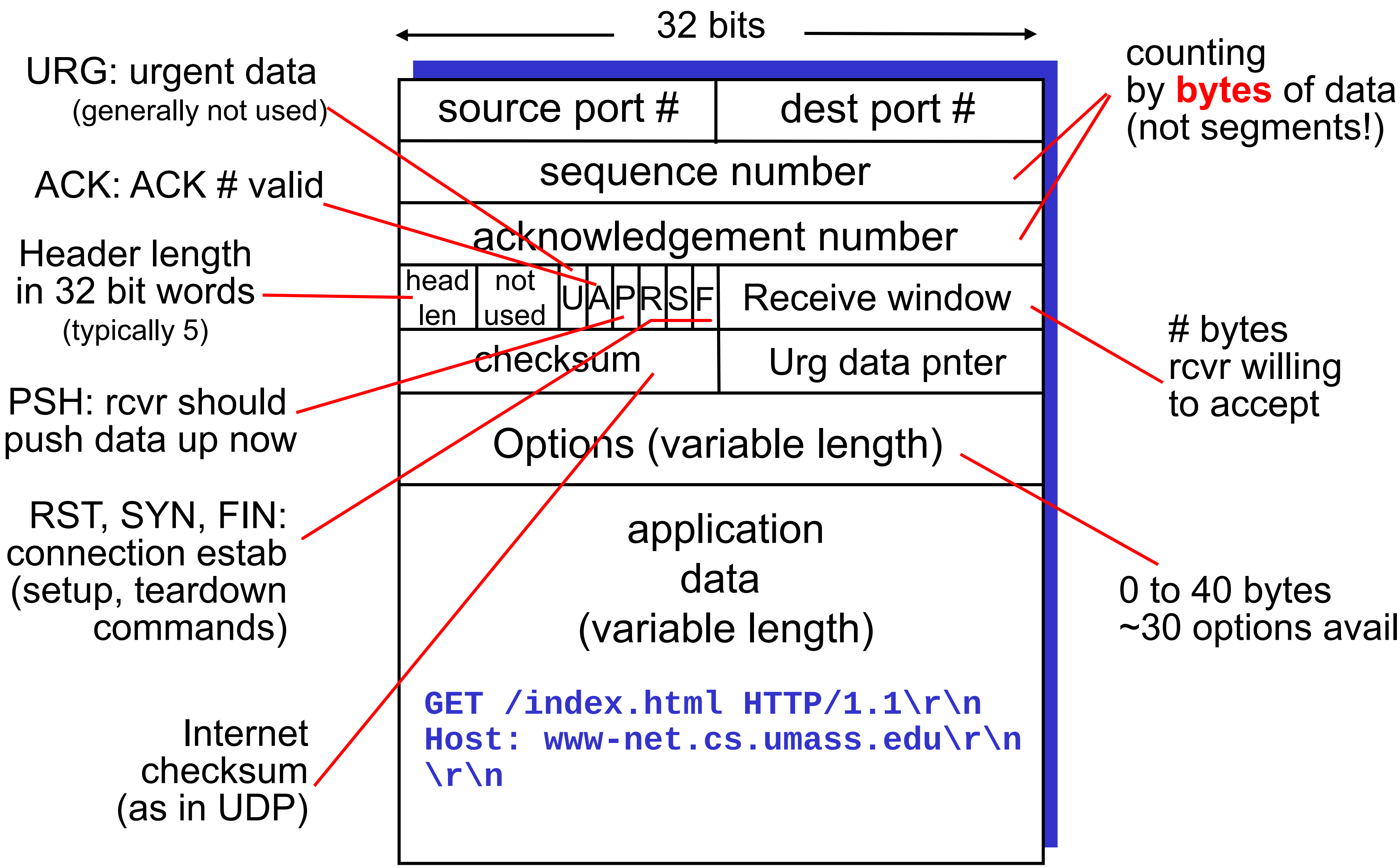
TCP: Overview RFCs: 793, 1122, 1323, 2018, 2581

- Point-to-point:
  - One sender, one receiver
- Reliable, in-order byte stream
- Pipelined (uses windowing):
  - 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 (data only)
- Connection-oriented:
  - Handshaking (exchange of control msgs) 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:

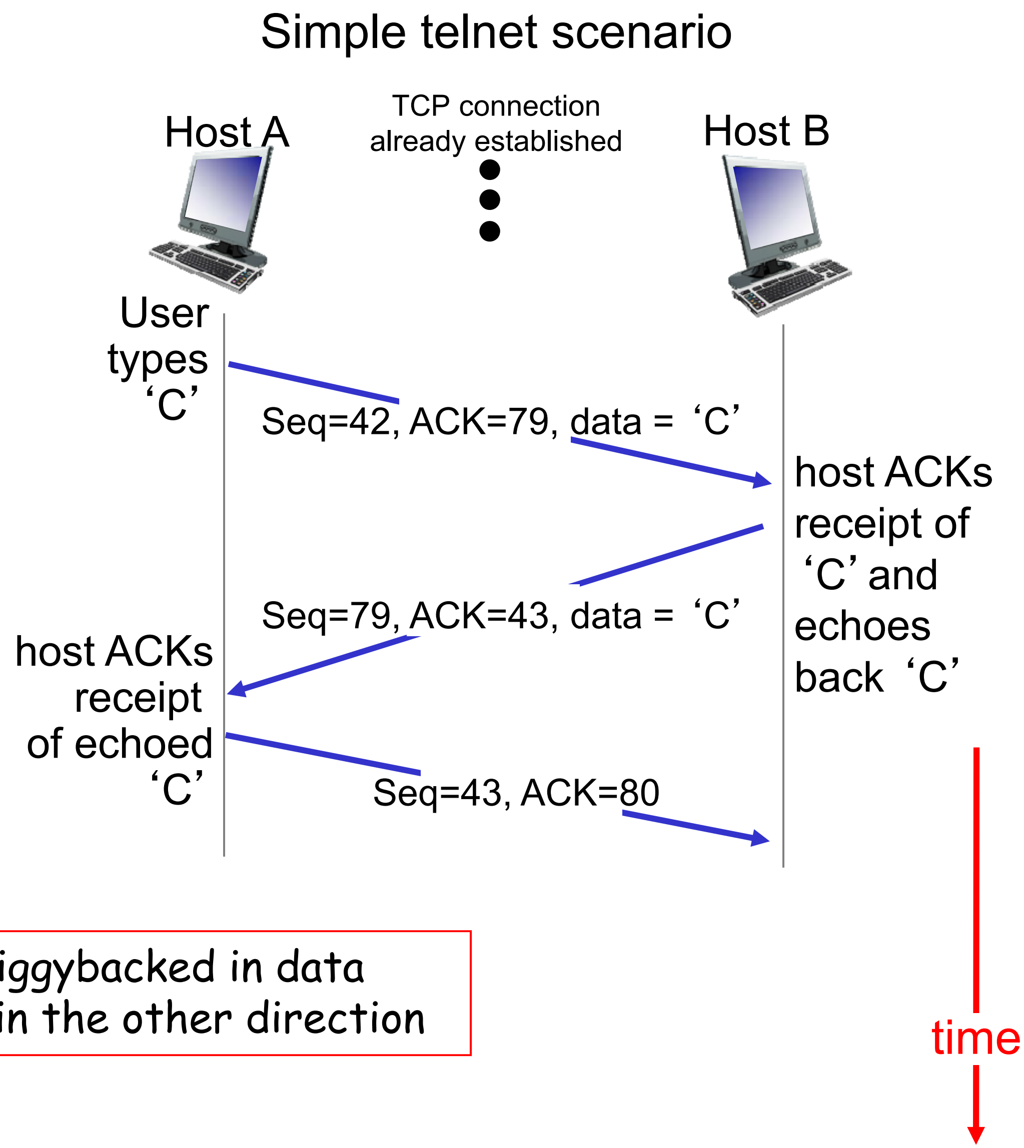
- Number of first byte in segment's data

ACKs:

- Seq # of next byte *expected* from other side
- Cumulative ACK

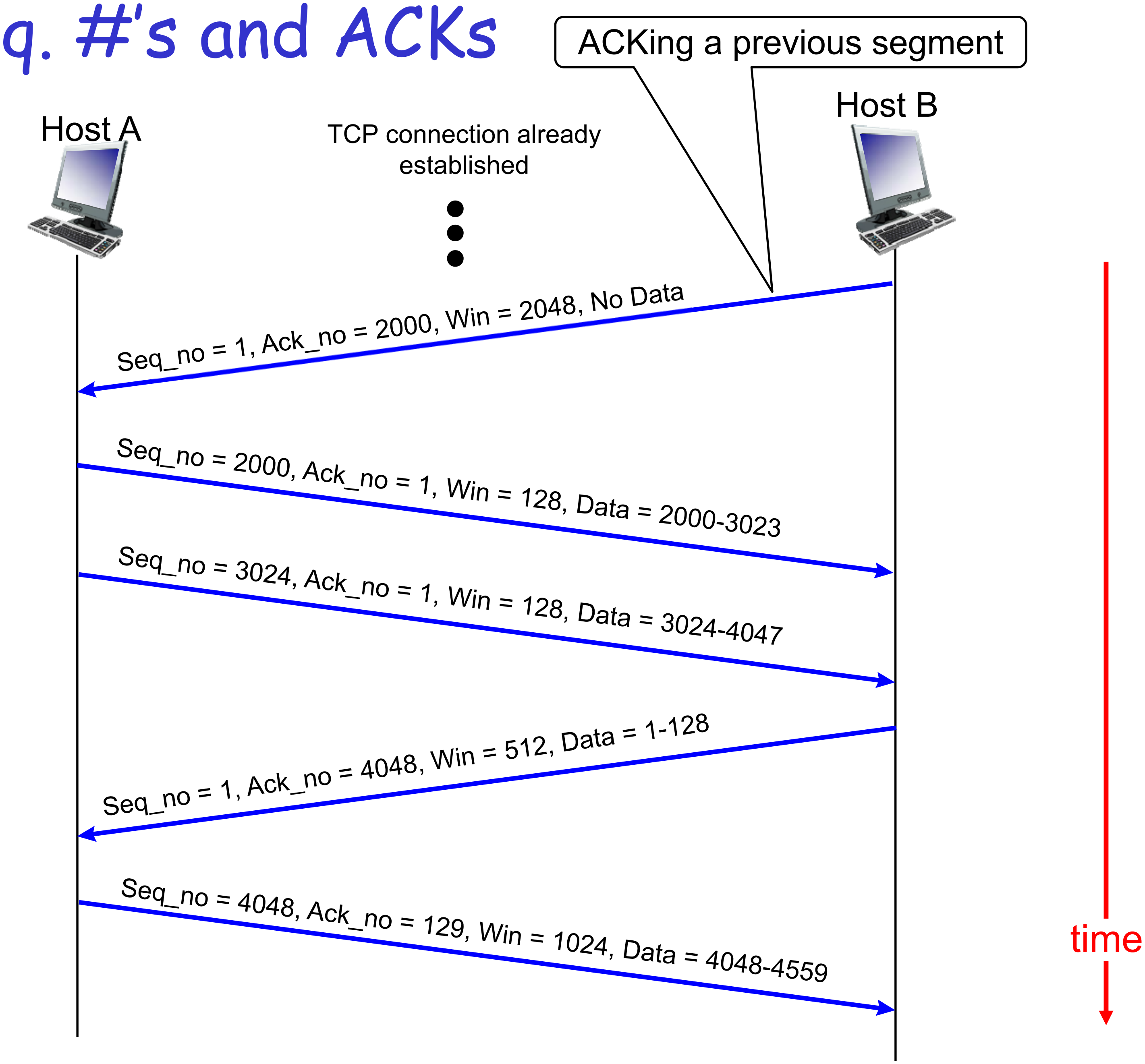
Q: How does receiver handle out-of-order segments?

- A: TCP spec doesn't say - up to implementer



ACKs piggybacked in data segments in the other direction

TCP Seq. #'s and ACKs



TCP Retransmission

- TCP calculates round trip time (RTT) for a segment and its ack
- From the RTT, TCP can guess how long it should wait before timing out on the next segment
- When a segment remains unacknowledged for a period of time, TCP assumes it is lost and retransmits it
  - Timer for retransmitted segments is doubled for each retransmission of that segment with a maximum timeout of 240 seconds
- What happens if retransmissions of a segment continue to fail?
  - **`TcpMaxDataRetransmissions`** registry value indicates the maximum number of retransmissions that can be sent on an existing connection
  - Default is 3

    `Registry location:`
    `HKEY_LOCAL_MACHINE\System\CurrentControlSet\Services\Tcpip\Parameters`

TCP RTT 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 end of segment transmission until ACK receipt
  - Ignore retransmissions
- SampleRTT will vary over time
  - We want an estimated RTT "smoother" to avoid short-term spikes
  - Average several recent measurements, not just current SampleRTT

TCP RTT and Timeout

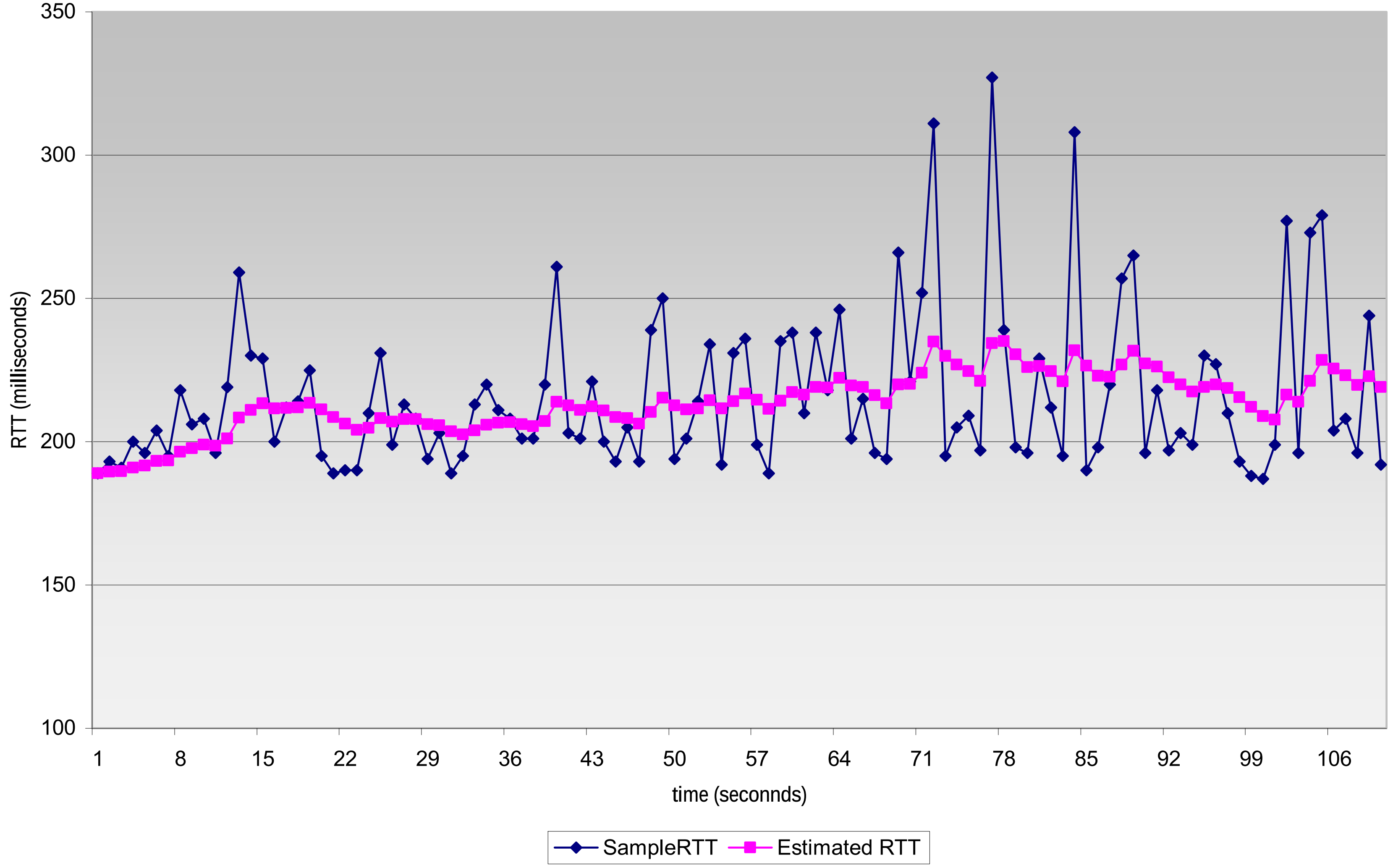
- Very first retransmission timeout value (1st segment)
  - **TimeoutInterval** set to 3 sec
  - Can be changed in Windows registry: **TcpInitialRtt**
- Subsequent timeout values are calculated based on sampled RTTs

```
EstimatedRTT = (1 - α)*EstimatedRTT + α*SampleRTT
EstimatedRTT = (0.875)*EstimatedRTT + 0.125*SampleRTT
```

- **EstimatedRTT = SampleRTT** for first calculation (2nd segment)
- "Exponential weighted moving average"
  - Influence of past sample decreases exponentially fast
- Typical value: α = 0.125

```
Registry location:
HKEY_LOCAL_MACHINE\System\CurrentControlSet\Services\Tcpip\Parameters
```

Example RTT Estimation



TCP RTT and Timeout

Setting the timeout

- `EstimatedRTT` plus "safety margin"
  - Large variation in `EstimatedRTT` → larger safety margin

- 1st estimate of how much `SampleRTT` deviates from `EstimatedRTT`:

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

typically, $\beta = 0.25$

- `DevRTT = [SampleRTT / 2]` for first calculation (2nd segment)

- Then set timeout interval:

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

Estimated RTT "Safety margin"

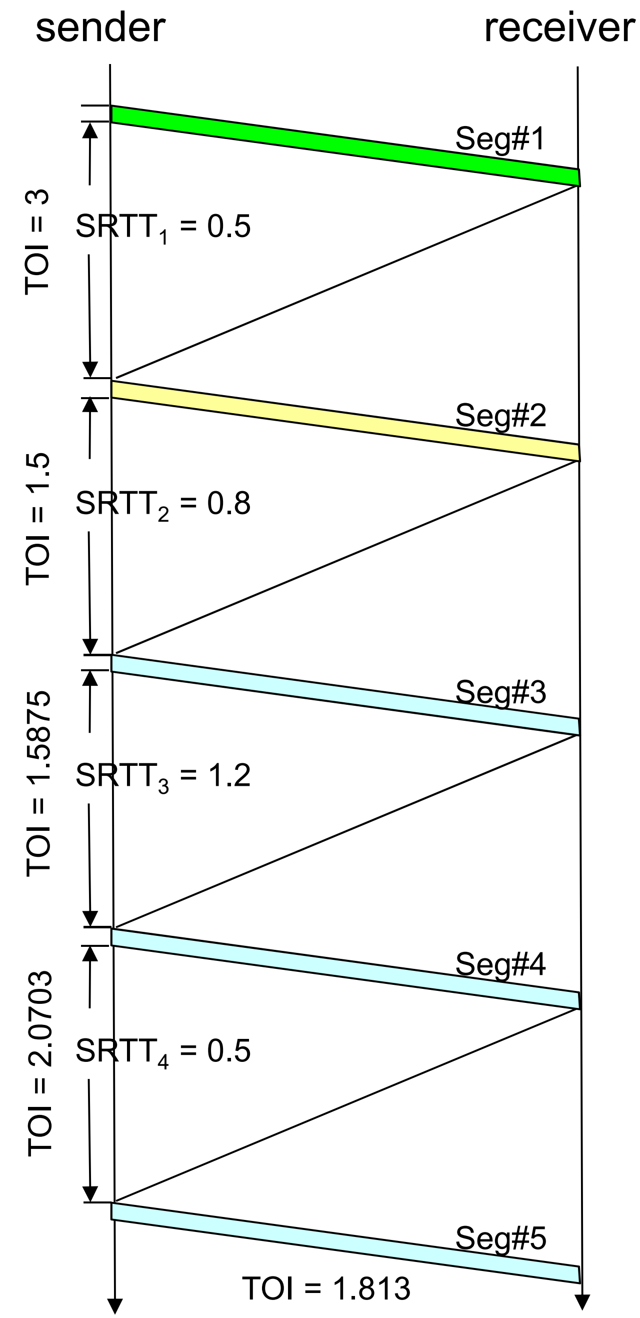
TCP RTT and TOI Example

First TimeoutInterval (TOI_1) = 3

For first calculation:
EstimateRTT$_2$ (ERTT$_2$) = SampleRTT$_1$ (SRTT$_1$)
DevRTT$_2$ = [SRTT$_1$ / 2]
TOI$_2$ = ERTT$_2$ + 4*DevRTT$_2$

For all subsequent calculations (compute in this order):
DevRTT$_n$ = 0.75*DevRTT$_{n-1}$ + 0.25*|SRTT$_{n-1}$-ERTT$_{n-1}$|
ERTT$_n$ = 0.875*ERTT$_{n-1}$ + 0.125*SRTT$_{n-1}$
TOI$_n$ = ERTT$_n$ + 4*DevRTT$_n$

Seg#	**DevRTT** Calculated	**ERTT** Calculated	**TOI** Calculated		**SRTT** Measured
1			3		0.5
2	0.25	0.5	1.5		0.8
3	0.2625	0.5375	1.5875		1.2
4	0.3625	0.6203	2.0703		0.5
5	0.30195	0.6053	1.813		



SRTT times are made up

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TCP Reliable Data Transfer

- TCP creates reliable data transfer service on top of IP's unreliable service
  - Pipelined segments
  - Cumulative acks
  - Single retransmission timer

- Retransmissions are triggered by:
  - Timeout events
  - Three duplicate acks

- Initially consider simplified TCP sender:
  - Ignore duplicate acks
  - Ignore flow control, congestion control

TCP Sender Events:

Data rcvd from app:

- Create TCP 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:

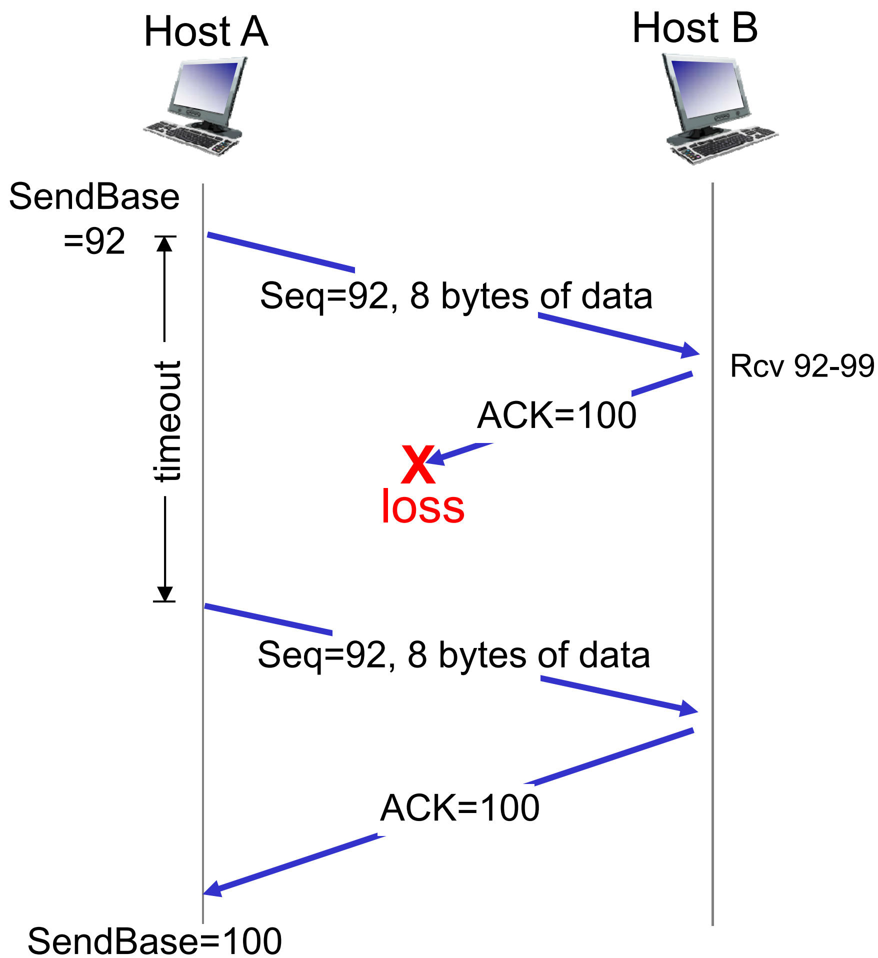
- If ack acknowledges previously unacked segments
  - Update what is known to be acked
  - Start timer if there are outstanding segments

```
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 with
             smallest sequence number
      start timer

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

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

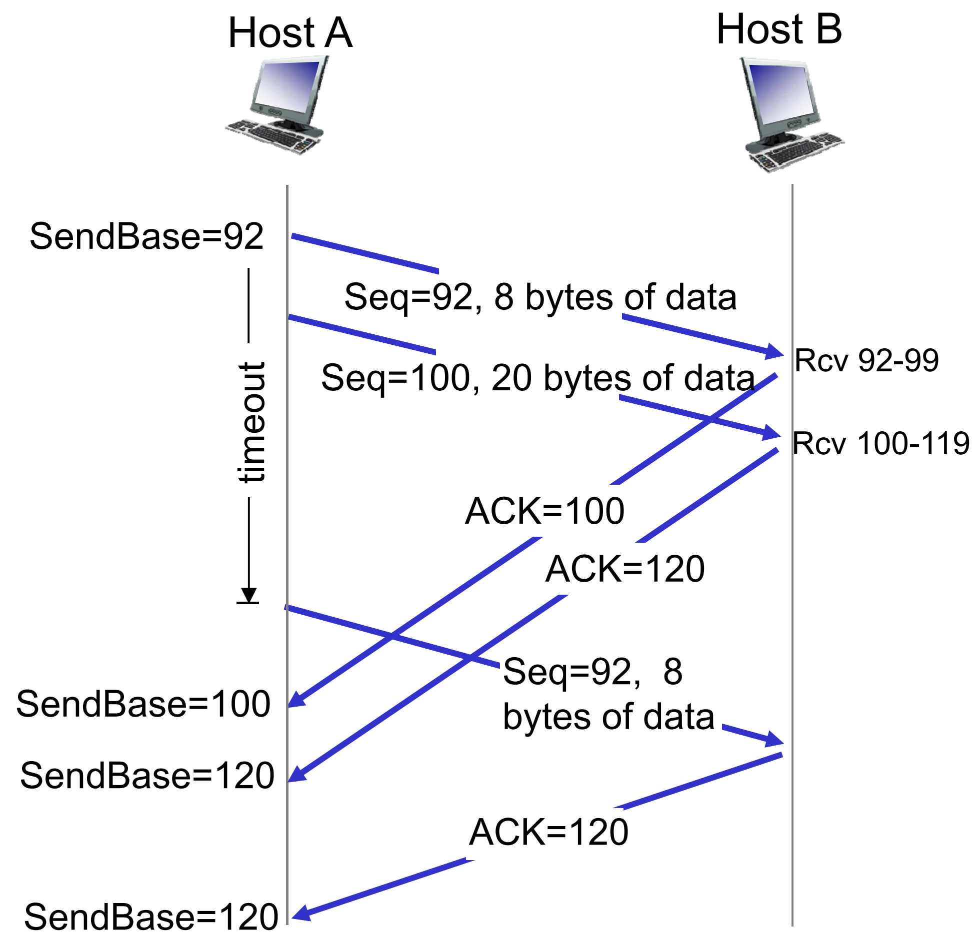
TCP Sender
(simplified)

<u>Comment:</u>
SendBase-1: last cumulatively ack'ed byte

TCP: Retransmission Scenarios

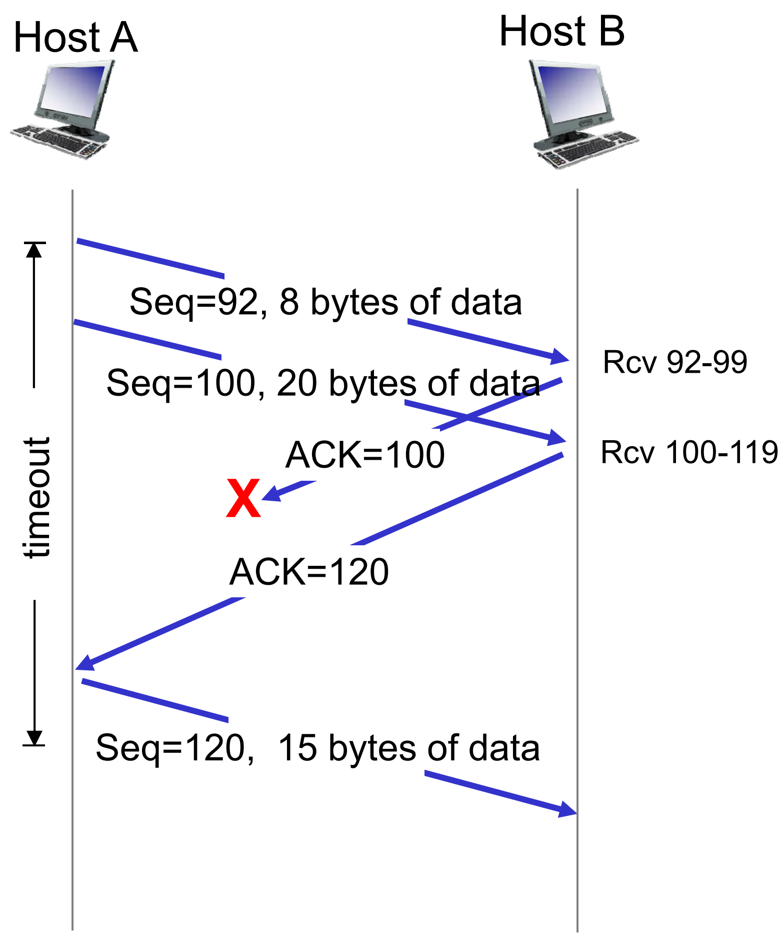


Lost ACK scenario



Premature timeout
segment 100 not retransmitted

TCP: Retransmission Scenarios



Cumulative ACK

TCP ACK Generation
[RFC 1122, RFC 2581]

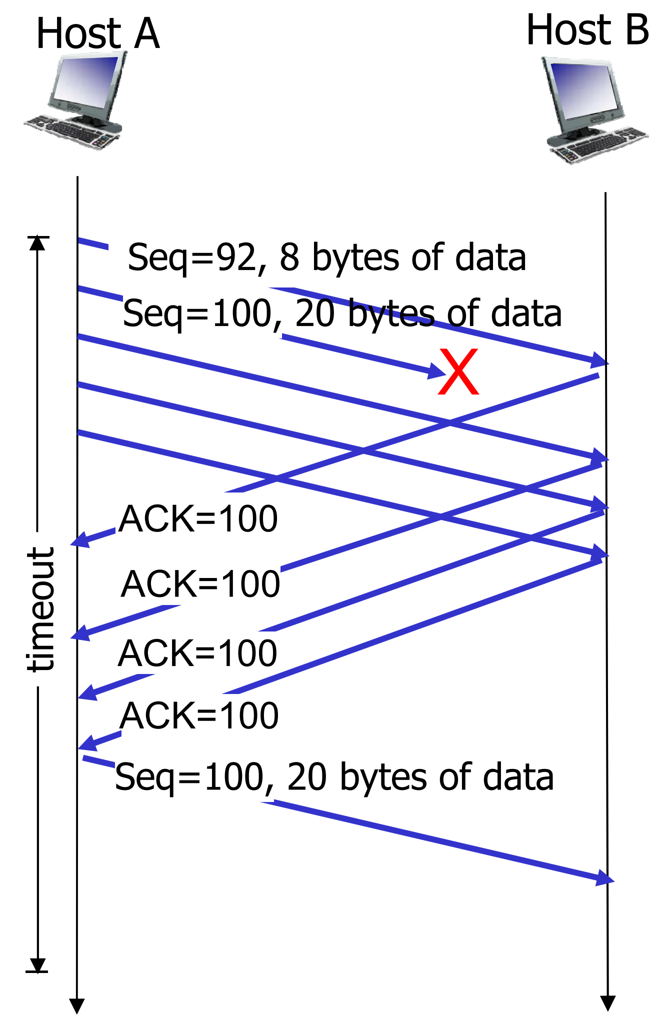
The hope is, within the delay, the receiver will have data ready to be sent to the receiver. Then, the ACK can be piggybacked with a data segment

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 200ms for next segment. If no next segment, send ACK. Registry: **`TcpAckFrequency`**
Arrival of in-order segment with expected seq #. One other segment has an 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

`Registry location:`
`HKEY_LOCAL_MACHINE\System\CurrentControlSet\Services\Tcpip\Parameters`

Fast Retransmit

- Timeout period can be 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 **duplicate** ACKs for the same data, it supposes that segment after ACKed data was lost:
  - <u>Fast retransmit:</u> resend segment before timer expires



Fast retransmit after receipt of 3rd duplicate ACK

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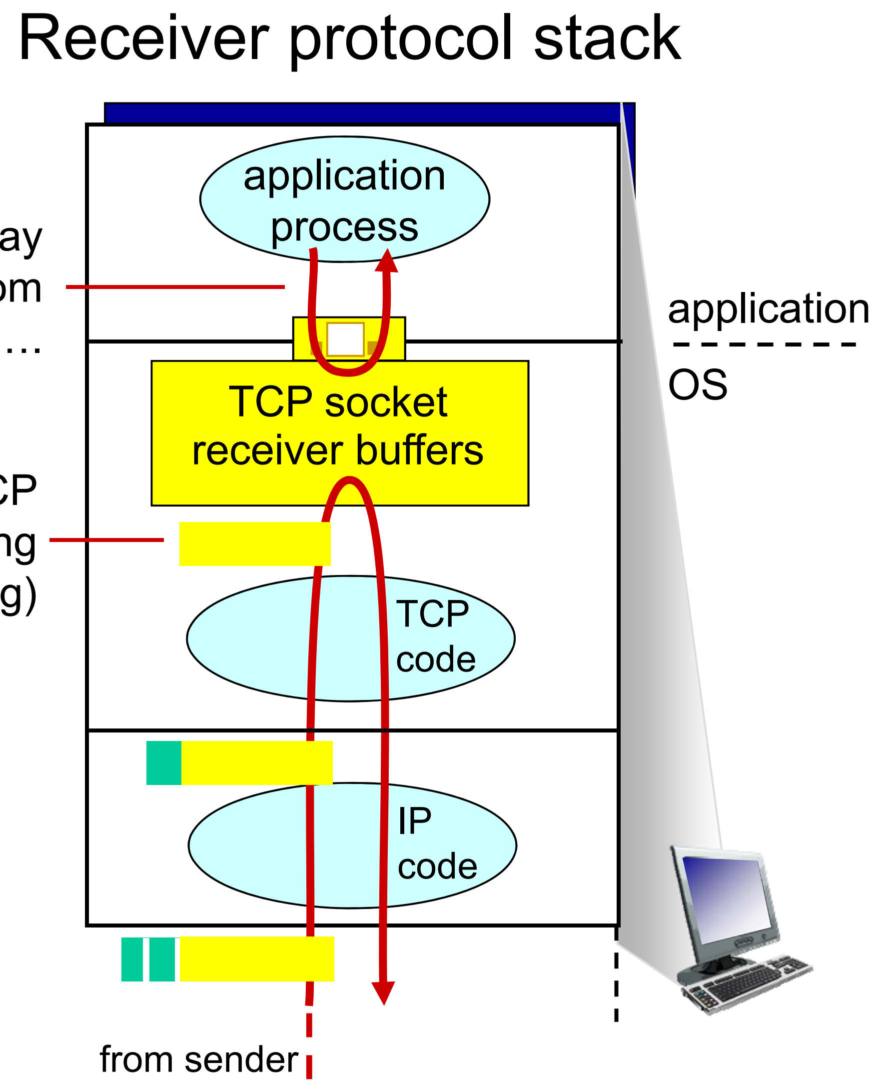
TCP Flow Control

Application may remove data from TCP socket buffers ….

… slower than TCP receiver is delivering (sender is sending)



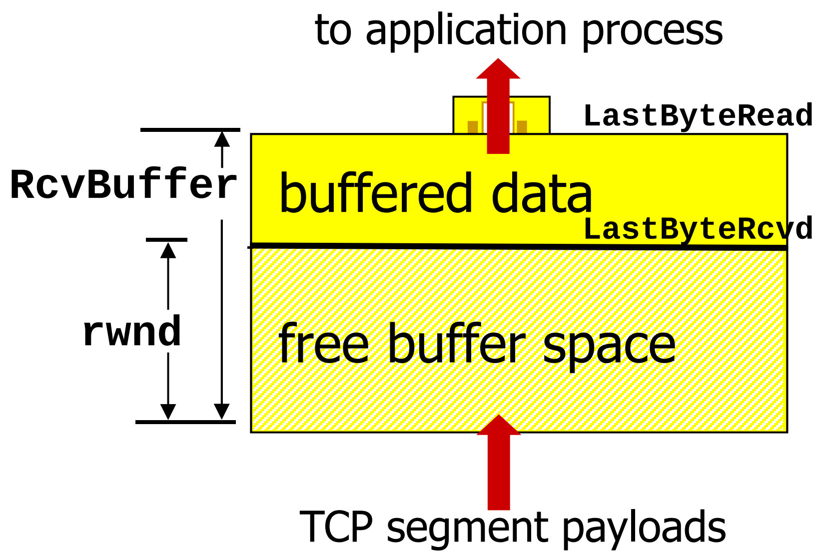
flow control

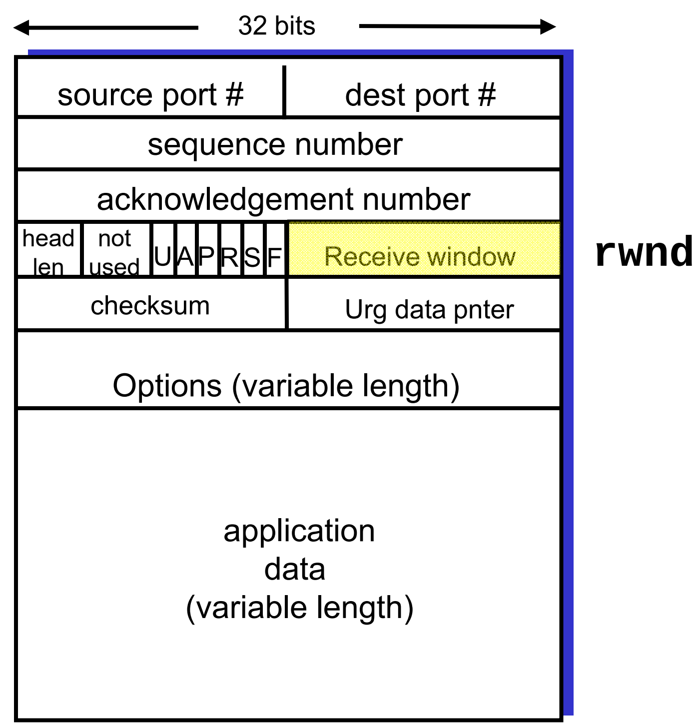
Receiver controls sender, so sender won't overflow receiver's buffer by transmitting too much, too fast

TCP Flow Control

- Receiver "advertises" free buffer space by including **rwnd** value in TCP header of receiver-to-sender segments
  - **RcvBuffer** size set via socket options (typical default is 4096 bytes)
- Sender limits amount of unacked ("in-flight") data to receiver's **rwnd** value
- Guarantees receive buffer will not overflow
- Spare room in buffer

rwnd = RcvBuffer- [LastByteRcvd – LastByteRead]

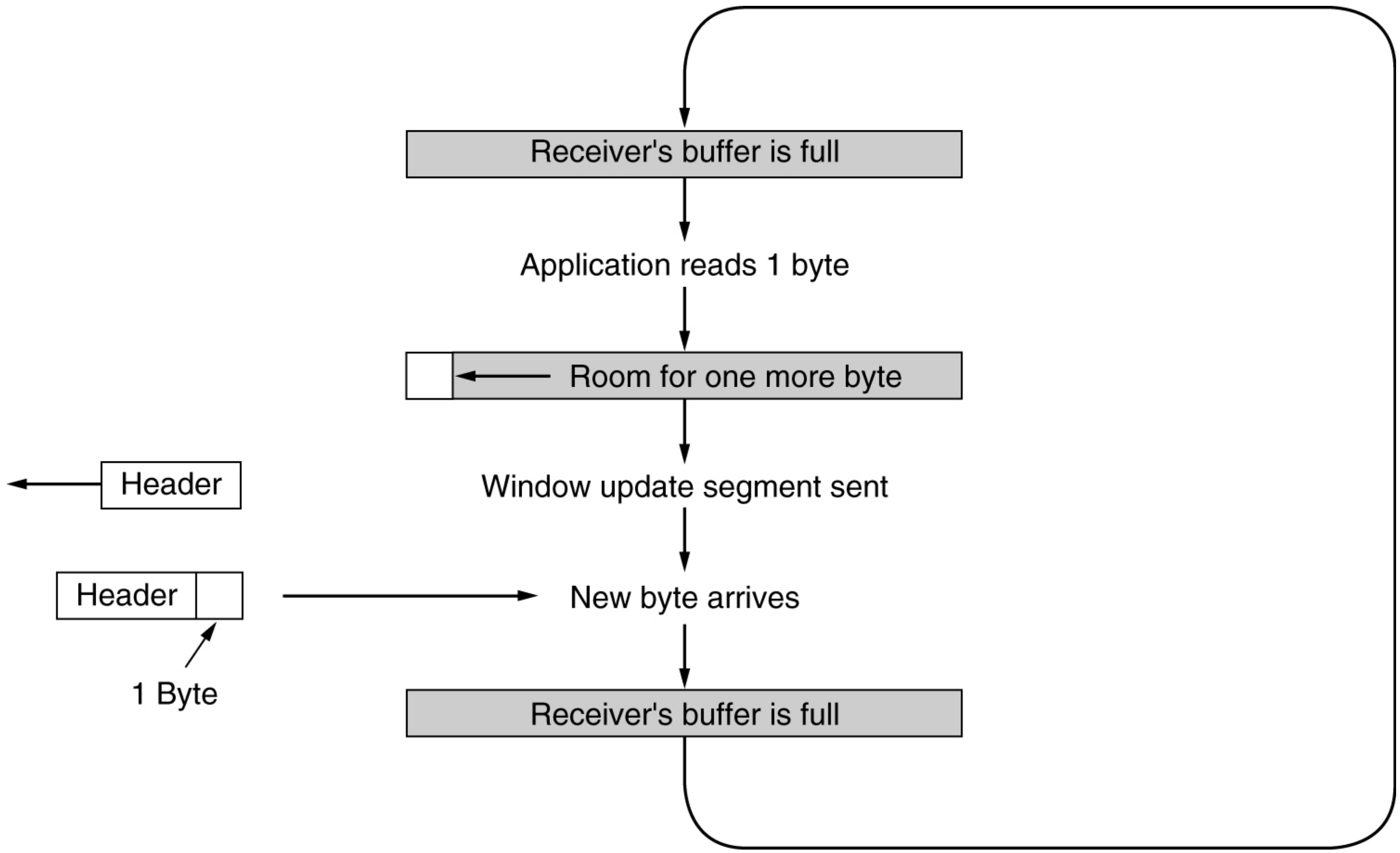




Persistence Timer

- Suppose sender is blocked because receiver hasn't read data from receiver buffer (**rwnd** = 0)
- Then receiver app reads n bytes
- New window is advertised with **rwnd** = n
- But this advertisement is lost
- Sender is stuck
  - Solution: persistence timer
  - Sender sends probe (the next data byte) while **rwnd** = 0
  - These segments are acked by receiver
  - After a while, receiver will advertise a non-zero window
  - **TCPPERSIST** values are 6, 12, 24, 48, 60 seconds
  - Persistent timer never stops until window update or connection terminated

Silly Window Syndrome



Clark's solution: don't advertise windows below a certain size

See this applet: http://media.pearsoncmg.com/aw/aw_kurose_network_2/applets/flow/flowcontrol.html

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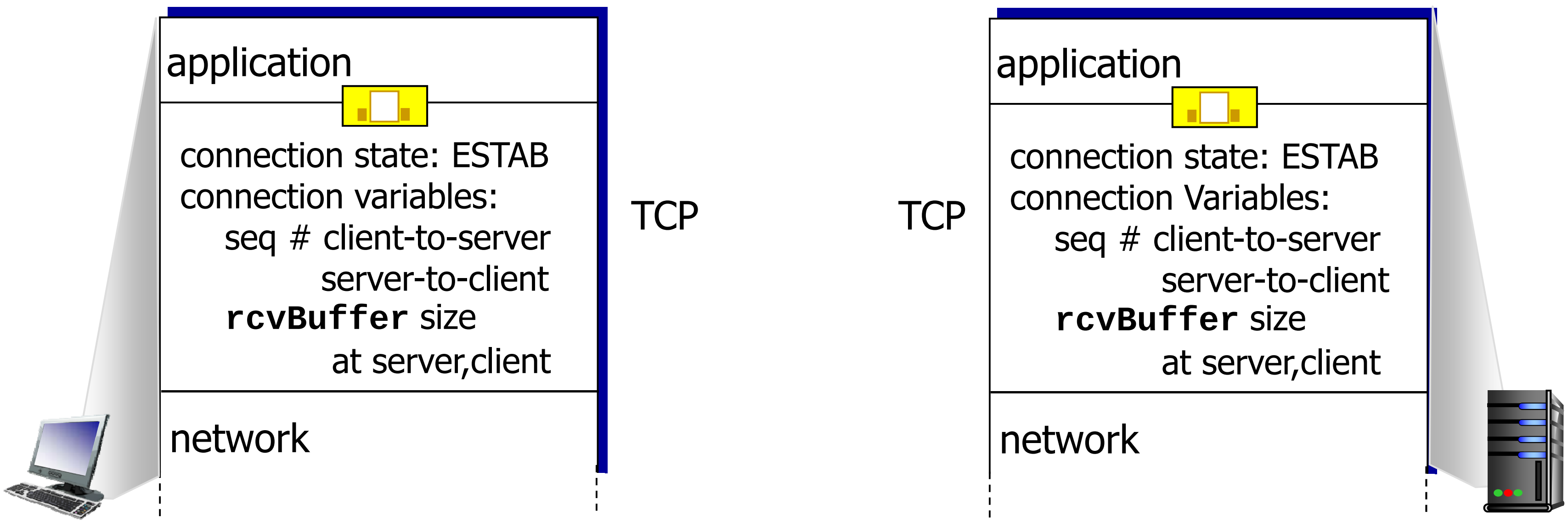
Connection Management

Before exchanging data, sender/receiver "handshake"

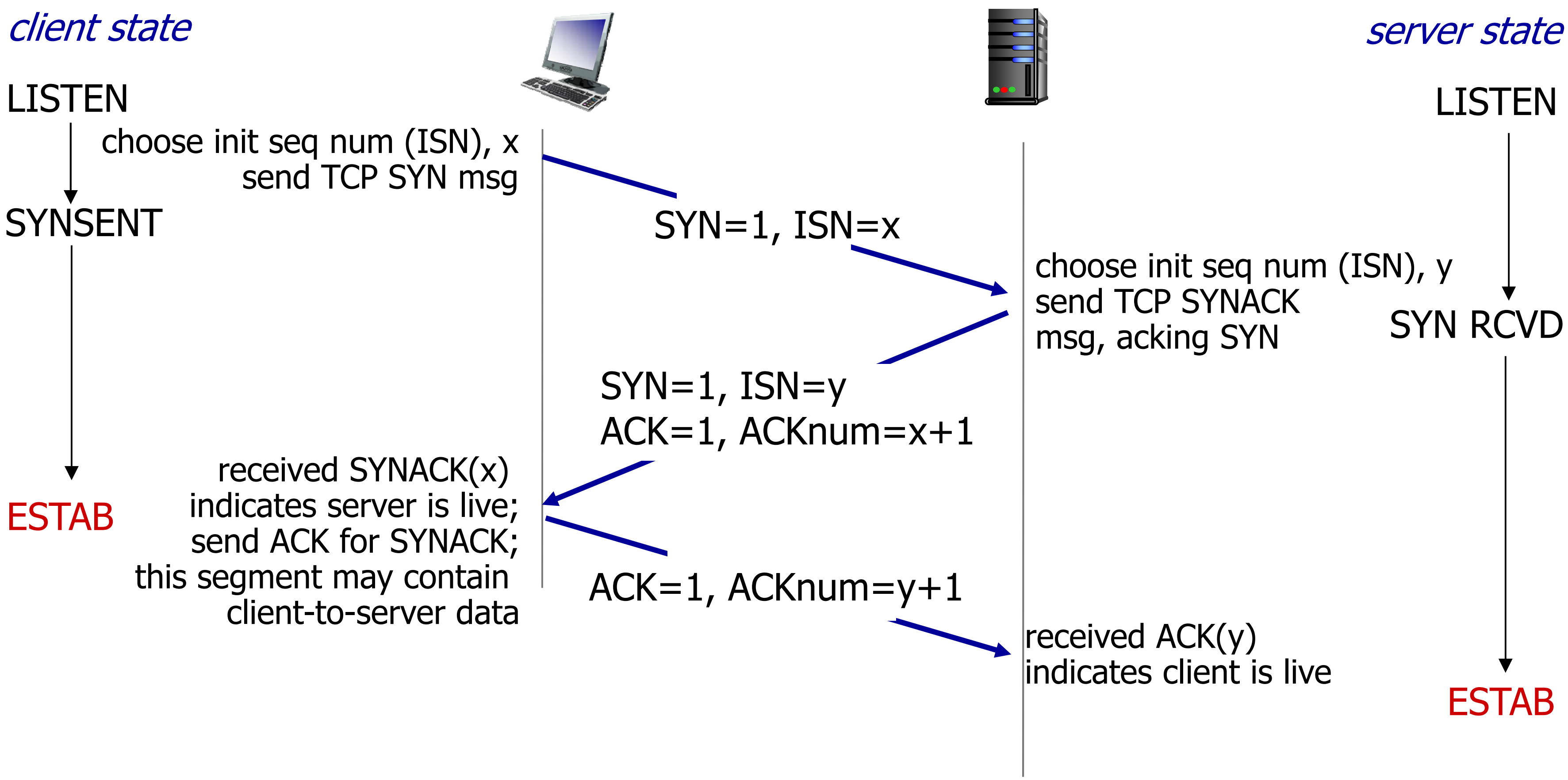
- Agree to establish connection
  - Each knowing the other willing to establish connection
- Agree on connection parameters



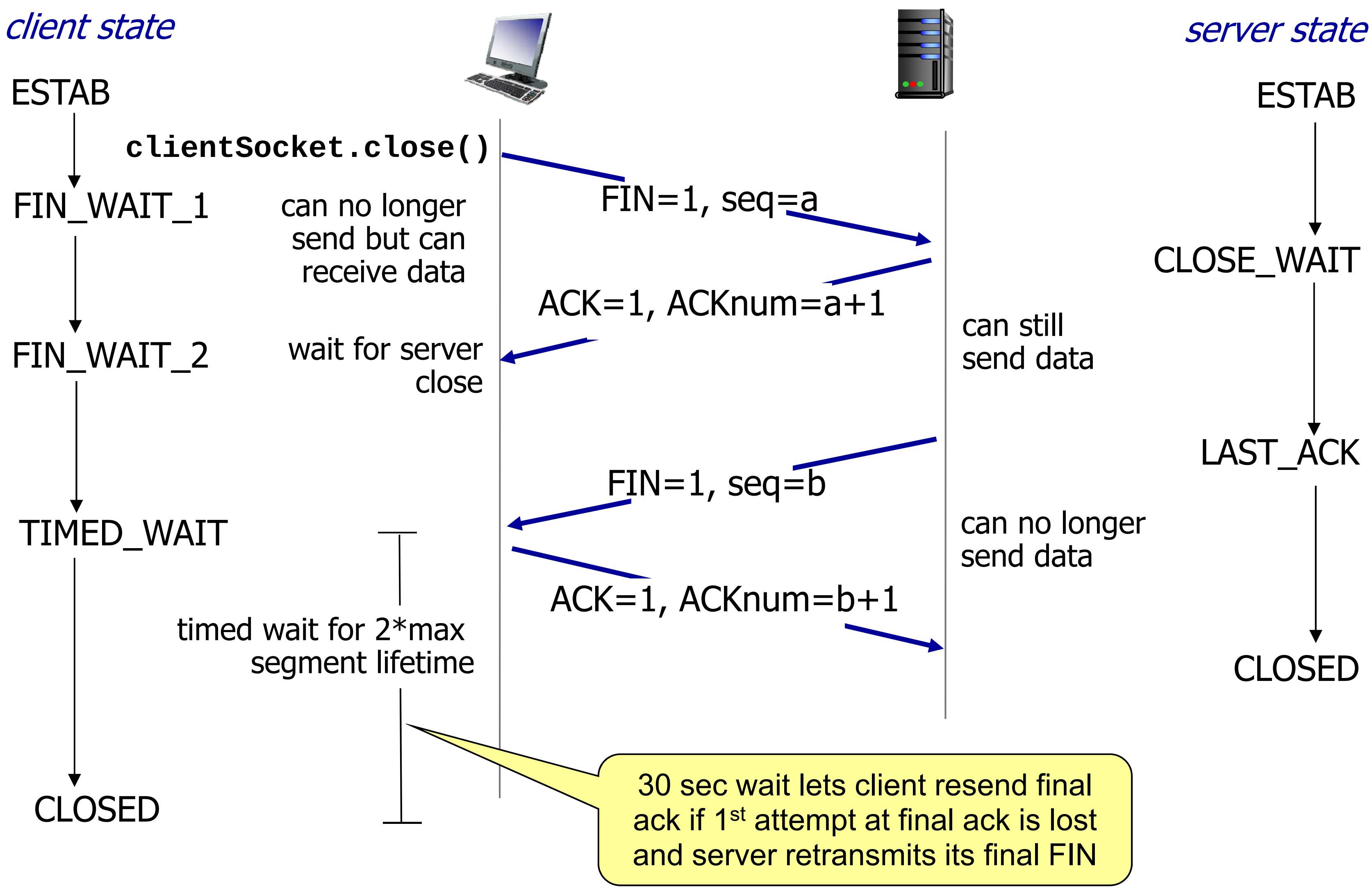

```
clientSocket = socket(AF_INET, SOCK_STREAM)
clientSocket.connect((serverName,serverPort))
```

```
serverSocket = socket(AF_INET,SOCK_STREAM)
serverSocket.bind(('',serverPort))
serverSocket.listen(1)
```

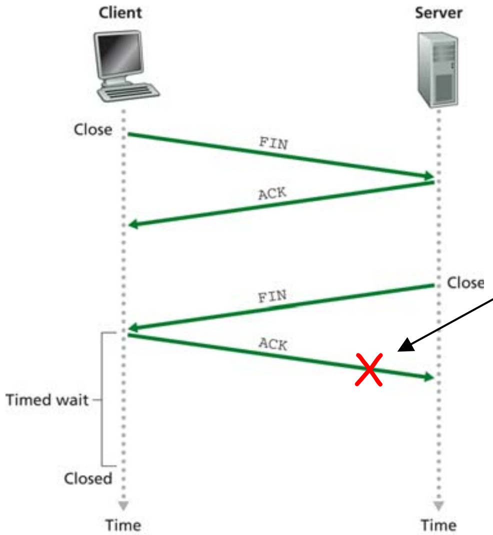
TCP 3-way Handshake



TCP: Closing A Connection

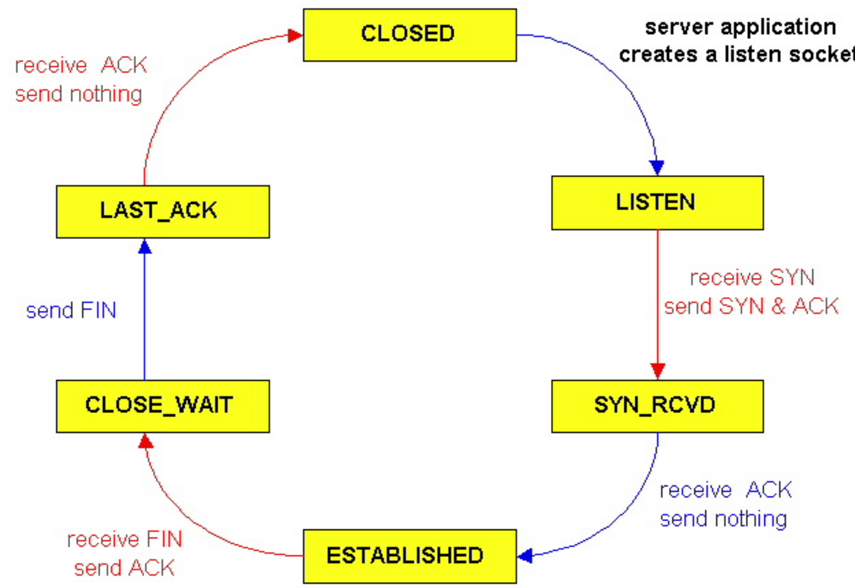


TCP Connection Lifecycles

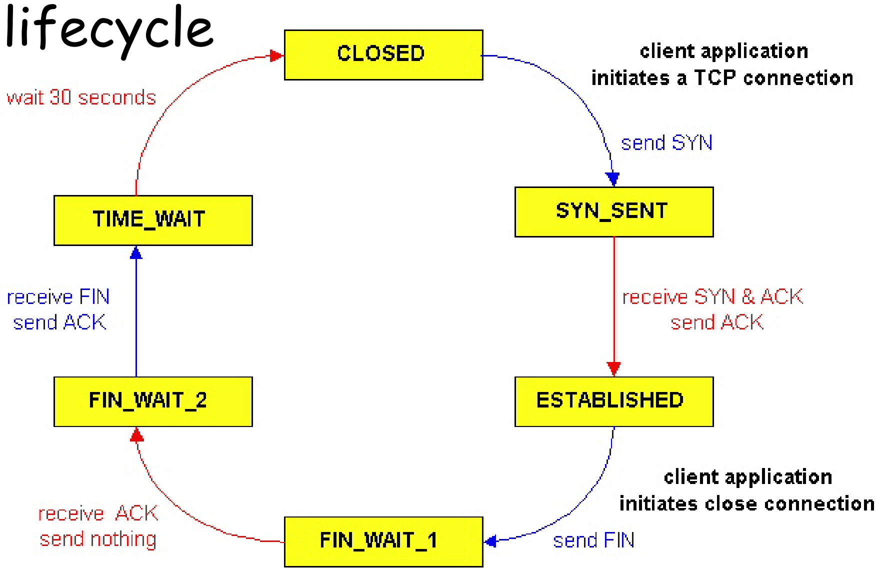


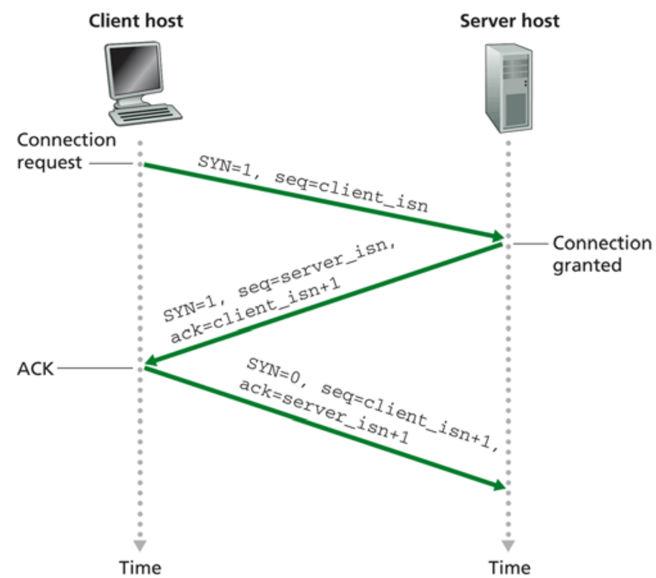
30 sec wait lets client resend final ack if 1st attempt at final ack is lost and server retransmits its final FIN

TCP server lifecycle



TCP client lifecycle





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Principles of Congestion Control

Congestion:

- Informally: "too many sources sending too much data too fast for the network to handle"
  - "Network" is not referring to the network layer but the network between sender and receiver

- Different from flow control!

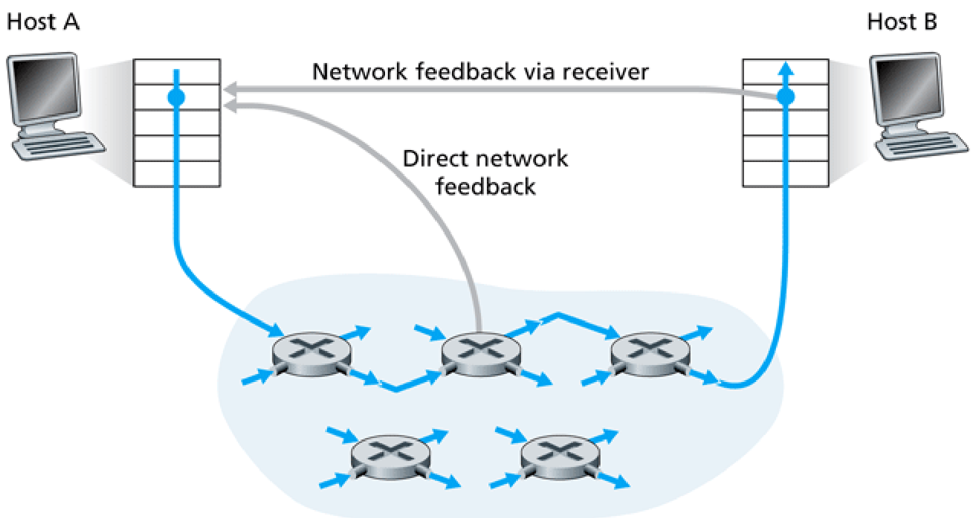
- Manifestations:
  - Lost packets (buffer overflow at routers)
  - Long delays (queuing in router buffers)

Approaches Towards Congestion Control

Two broad approaches towards congestion control:

End-end congestion control (via receiver):

- No explicit feedback from network layer
- Congestion inferred from end-system observed loss or delay
- Approach taken by TCP



Network-assisted congestion control (direct network feedback):

- Routers provide feedback to end systems
  - Single bit indicating congestion
    - IBM Systems Network Architecture (SNA)
    - DECnet
    - TCP/IP Explicit Congestion Notification (ECN)
    - ATM
  - Explicit rate sender should send at

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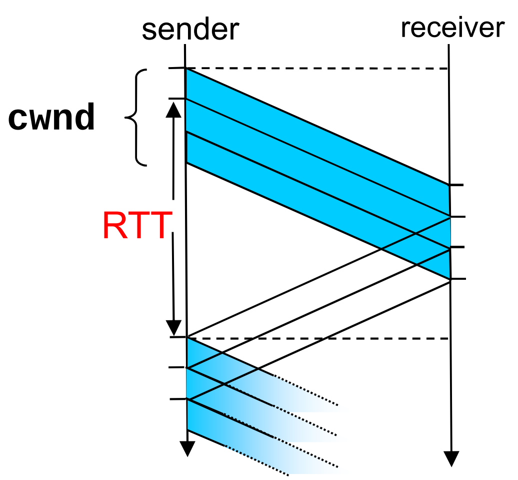
TCP Congestion Control

- End-end control (no network assist)
- Congestion window (**cwnd**) is dynamic function of perceived network congestion
  - Ideally make **cwnd** as large as possible
    - Transmit as fast as possible
- Sender limits transmission

  LastByteSent-LastByteAcked

  ≤ min{cwnd, rwnd}
- Roughly,

$$\text{rate} = \frac{\textbf{cwnd}}{\text{RTT}} = \frac{\text{W*MSS}}{\text{RTT}} \text{ Bytes/sec}$$



W = window size in segments
MSS = max segment size (data only)

How does sender perceive congestion?

- Loss event = timeout or 3 duplicate acks
- TCP sender reduces rate (**cwnd**) after loss event
- Acks regulate the send rate
  - Slower ack rate = slower increase in **cwnd**

TCP Congestion Control Algorithm has 2 phases:

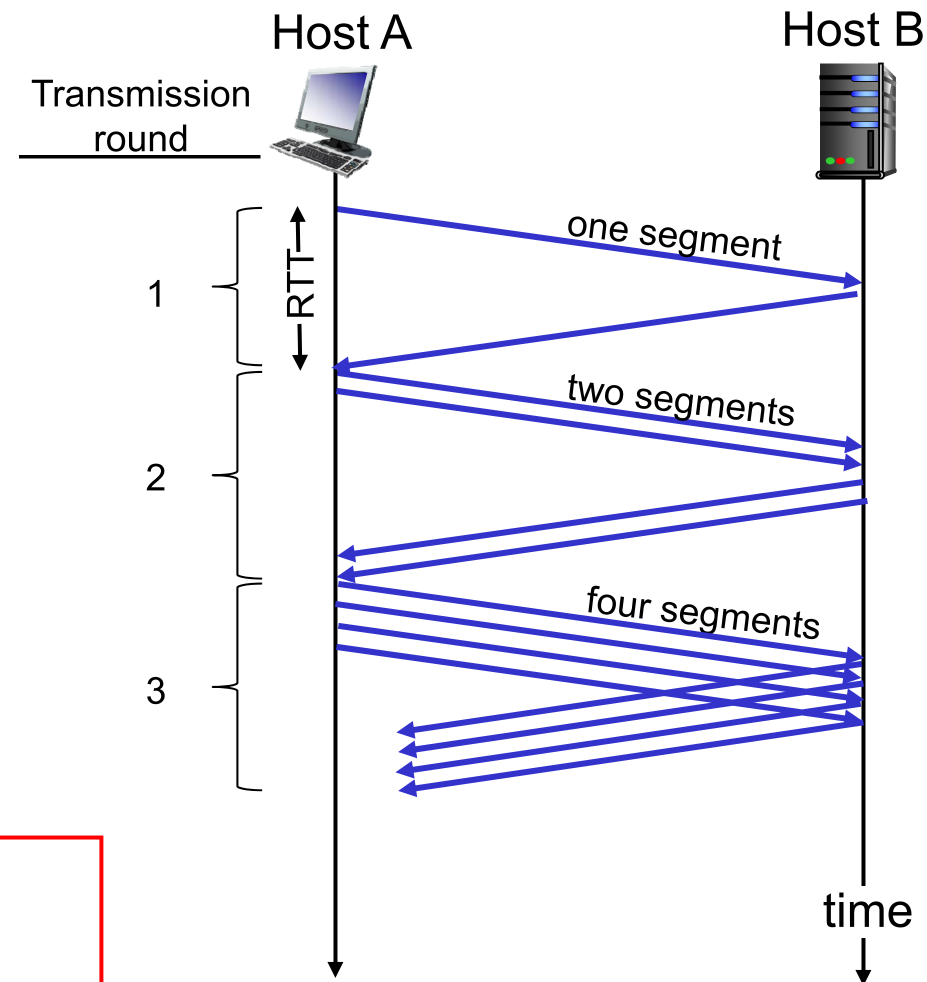
- Slow start
- Congestion Avoidance
  - AIMD (Additive Increase Multiplicative Decrease)

TCP Slow Start

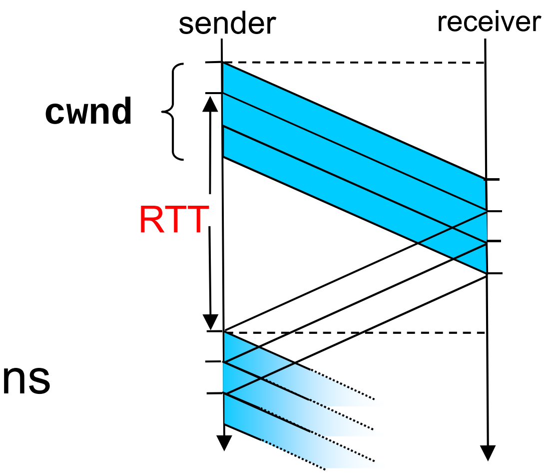
- When connection begins, increase rate exponentially until first loss event:
  - Initially **cwnd** = 1 MSS
  - Double **cwnd** every RTT
    - Done by incrementing **cwnd** for every ACK received
- Summary:
  - Initial rate is slow but ramps up exponentially fast

Slow Start algorithm

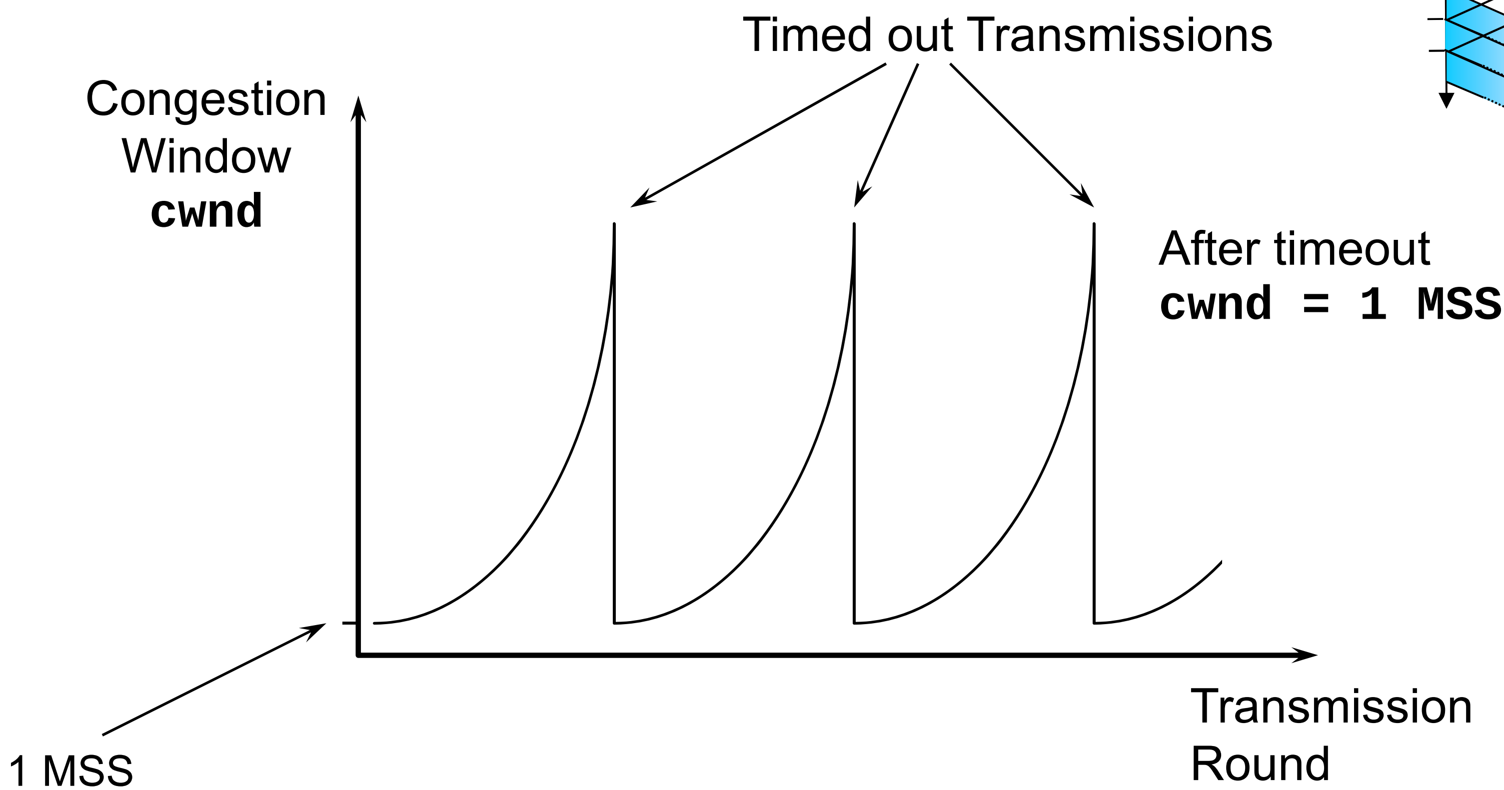
initialize: cwnd = 1 MSS
for (each segment ACKed)
    cwnd++
until (loss event OR cwnd > threshold)



TCP Slow Start



- It works but performance is not all that great



Refinement

- After timeout event:
  - **cwnd** set to 1 MSS
  - Window then grows exponentially (slow start)…
  - …to a threshold, then grows linearly (congestion avoidance)

Philosophy:

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

- After 3 dup ACKs: (TCP Reno)
  - **cwnd** is cut in half
  - Window then grows linearly

- Note: TCP Tahoe always sets **cwnd** to 1 (timeout or 3 duplicate acks)

TCP AIMD (Congestion Avoidance)

Additive increase

Increase **cwnd** a little each time an ACK is received with the goal of increasing **cwnd** by 1 MSS every RTT in the absence of loss events: *probing*

Multiplicative decrease

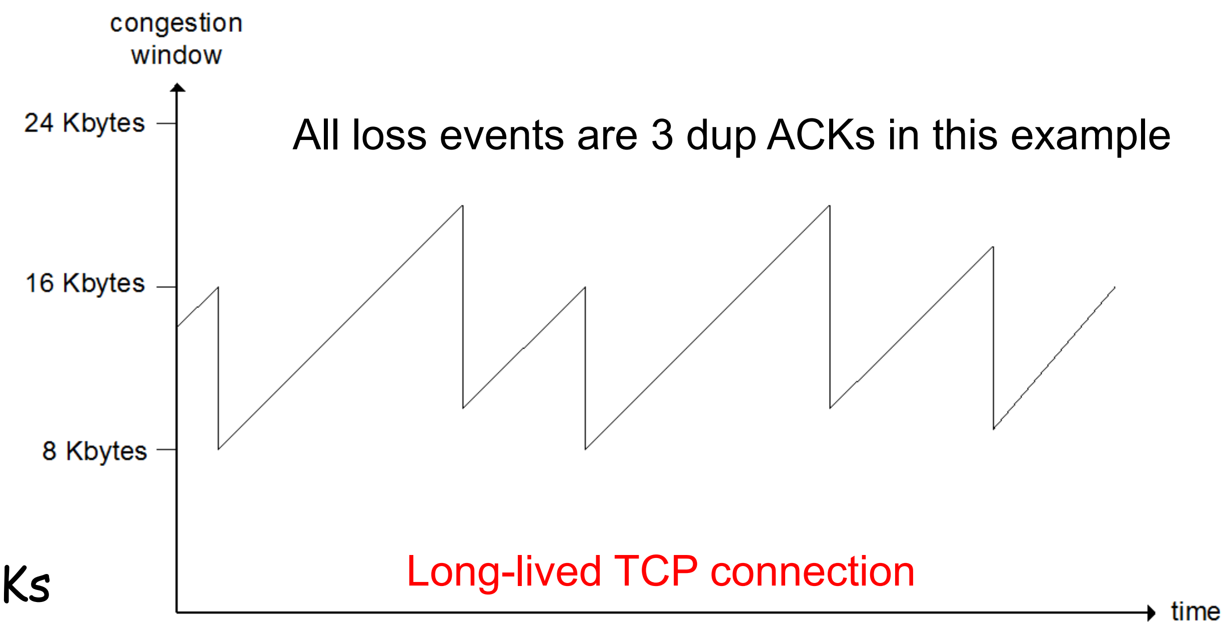
Cut **cwnd** in half after 3 dup acks

cwnd = **cwnd** + MSS * (MSS/**cwnd**)

Example:
MSS = 1,460 bytes

If window = 10,
  cwnd = 14,600 bytes

cwnd would increase
  by ~146 bytes per ACK
  or one MSS after 10 ACKs

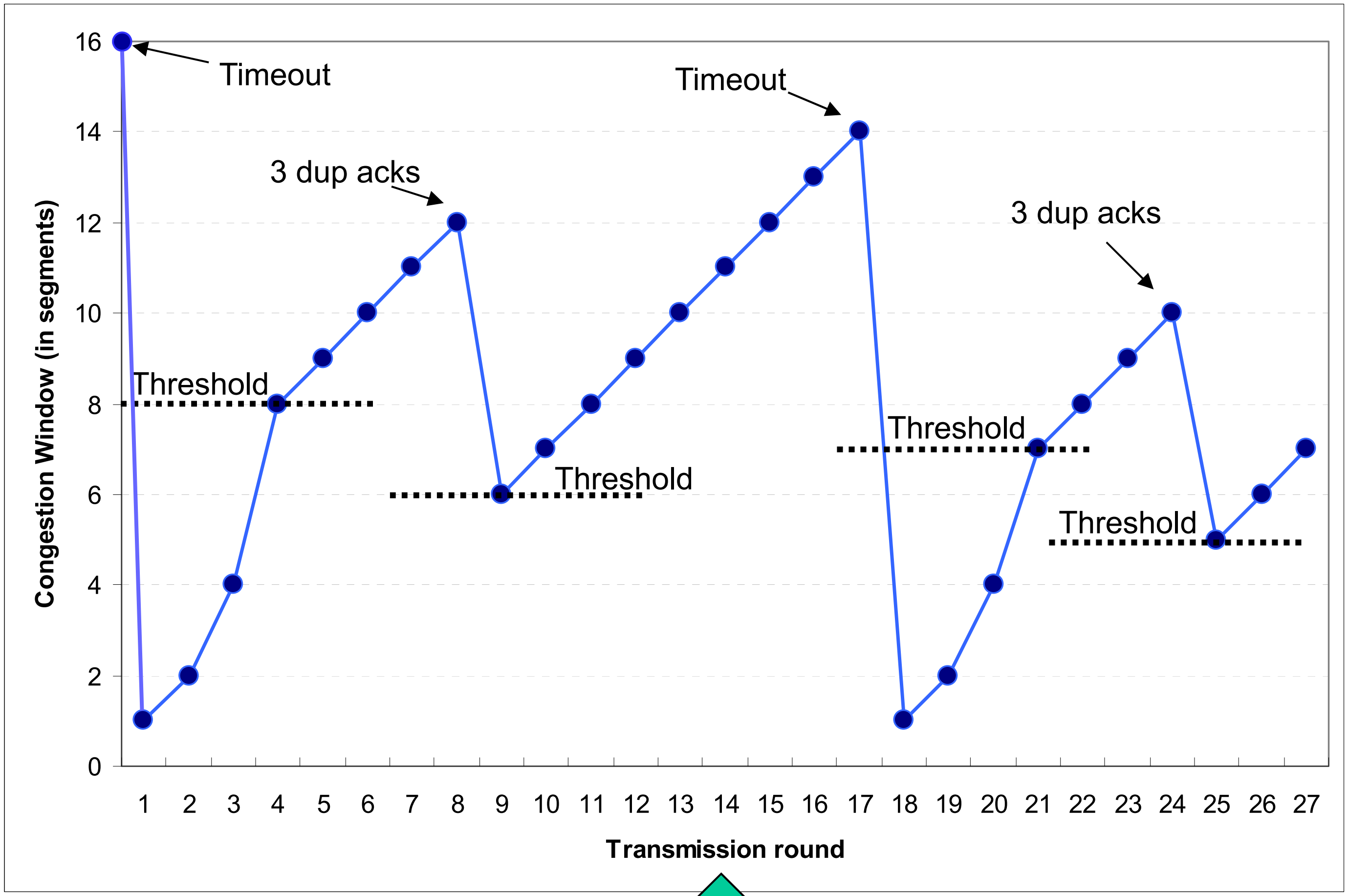


TCP Reno Performance

This trace is not the beginning of a connection but somewhat into it

Q: When should the exponential increase (SS) switch to linear (Additive increase)?

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

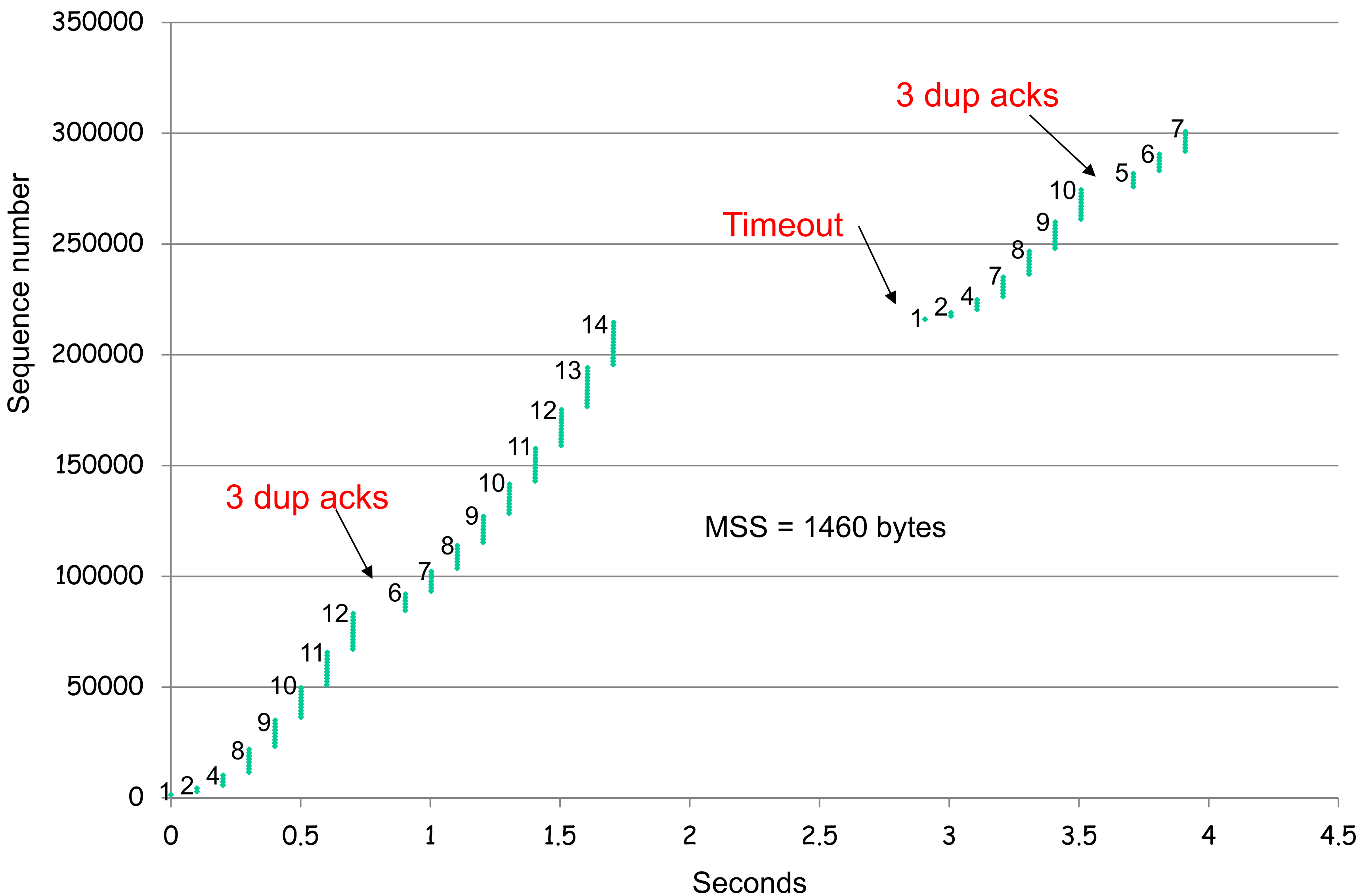


Note: This is NOT time!

Implementation:

- Variable Threshold
- Initial threshold set to 65 Kbytes
- At loss event, threshold is set to 1/2 of **cwnd** just before loss event

TCP Reno Performance - Stevens Graph



Summary: TCP Congestion Control

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

TCP Sender Congestion Control

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