Network Transport Layer: Sliding Window, TCP

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https://sngroup.org.cn/courses/cnns-xmuf25/index.shtml

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This deck of slides are heavily based on CPSC 433/533 at Yale University, by courtesy of Dr. Y. Richard Yang.

Outline

- Admin and recap
- Reliable data transfer,
  - stop-and-wait
  - sliding window
- TCP

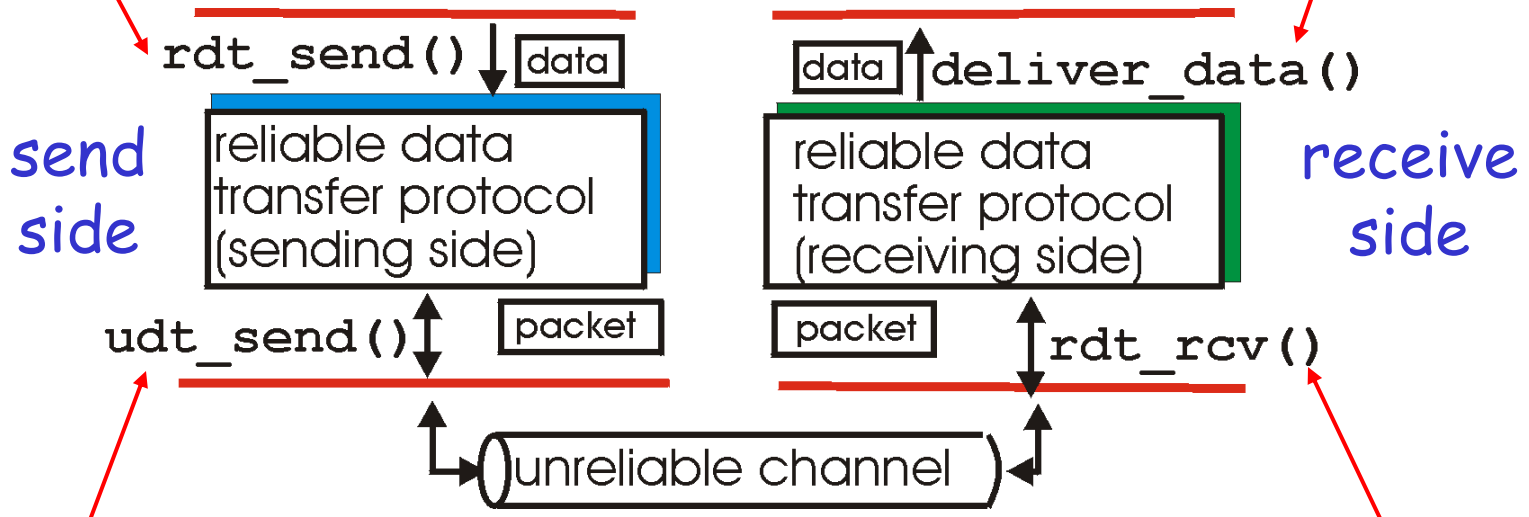
Admin

- Lab 4 to be posted this week

Recap: Reliable Data Transfer Context

rdt_send(): called from above, (e.g., by app.)

deliver_data(): called by rdt to deliver data to upper



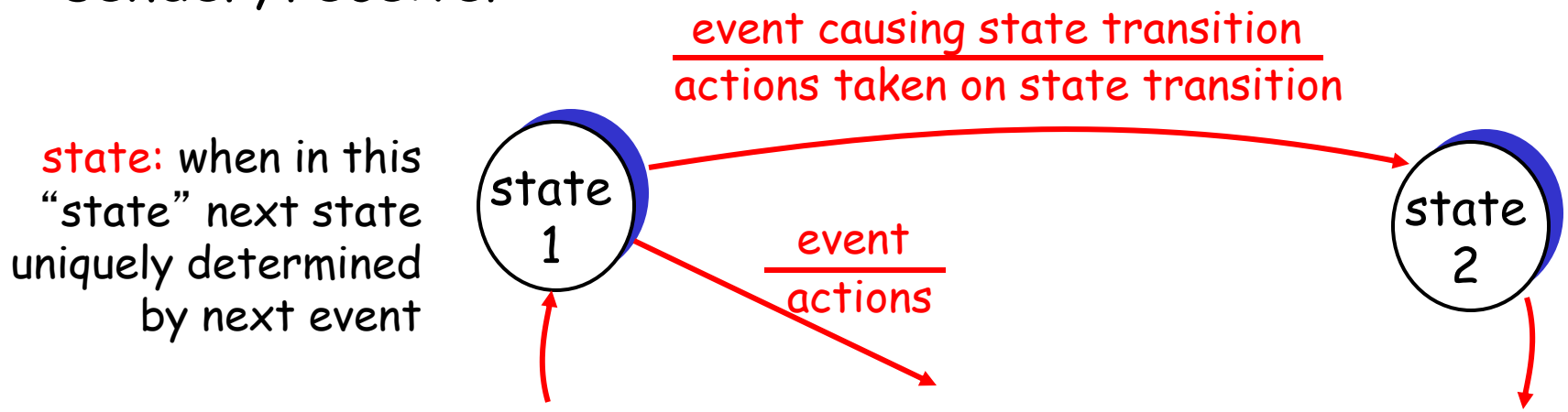
udt_send(): called by rdt, to transfer packet over unreliable channel to receiver

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

Recap: Reliable Data Transfer Setting

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



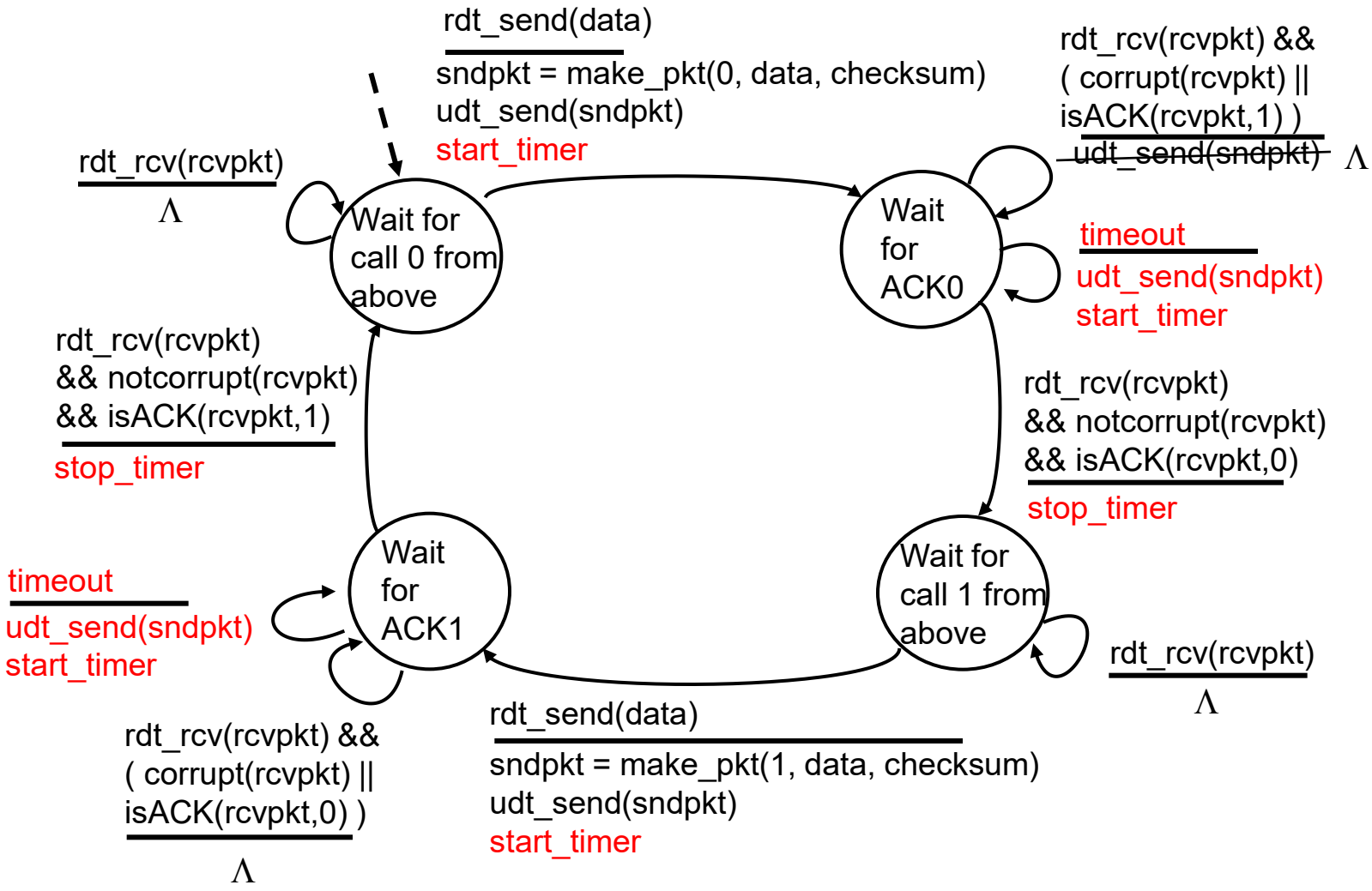
rdt3.0: Channels with Errors *and* Loss

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

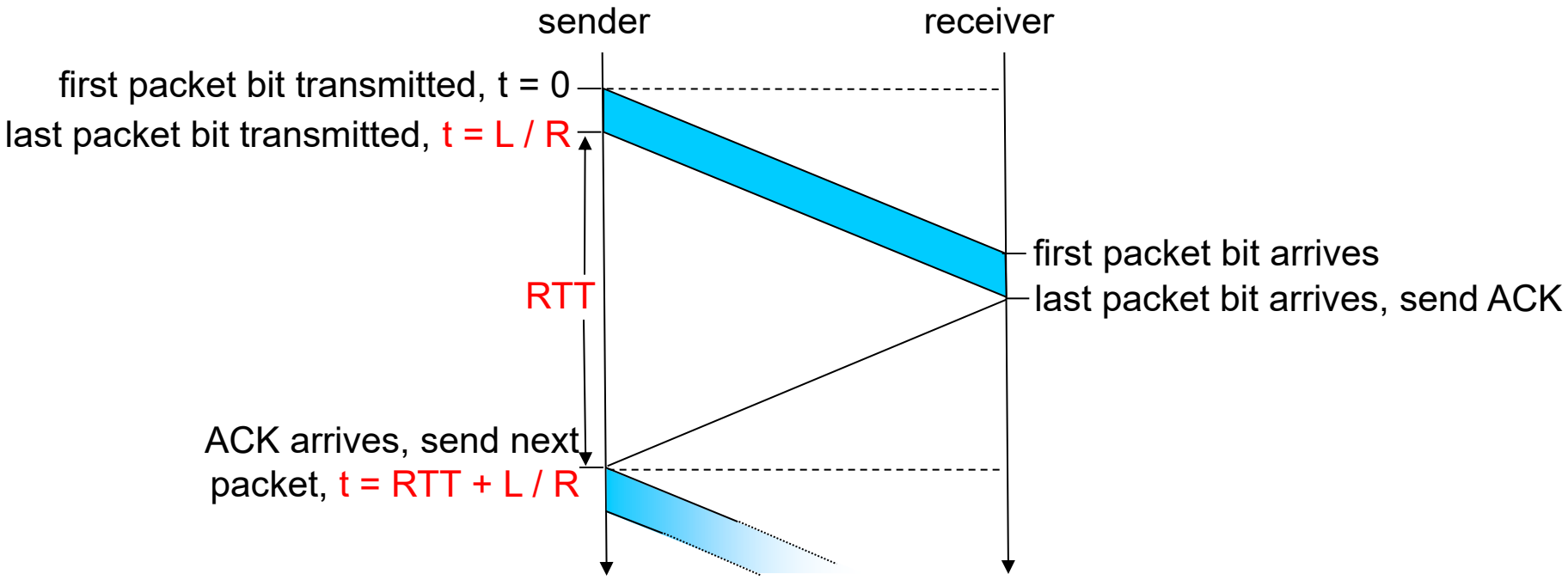
Q: Does rdt2.2 work under losses?

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 Sender



rdt3.0: Stop-and-Wait Performance



What is U_{sender}: utilization – fraction of time link busy sending?

Assume: 1 Gbps link, 15 ms e-e prop. delay, 1KB packet

Performance of rdt3.0

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

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

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

- 1KB pkt every 30 msec -> 33kB/sec throughput over 1 Gbps link
- network protocol limits use of physical resources !

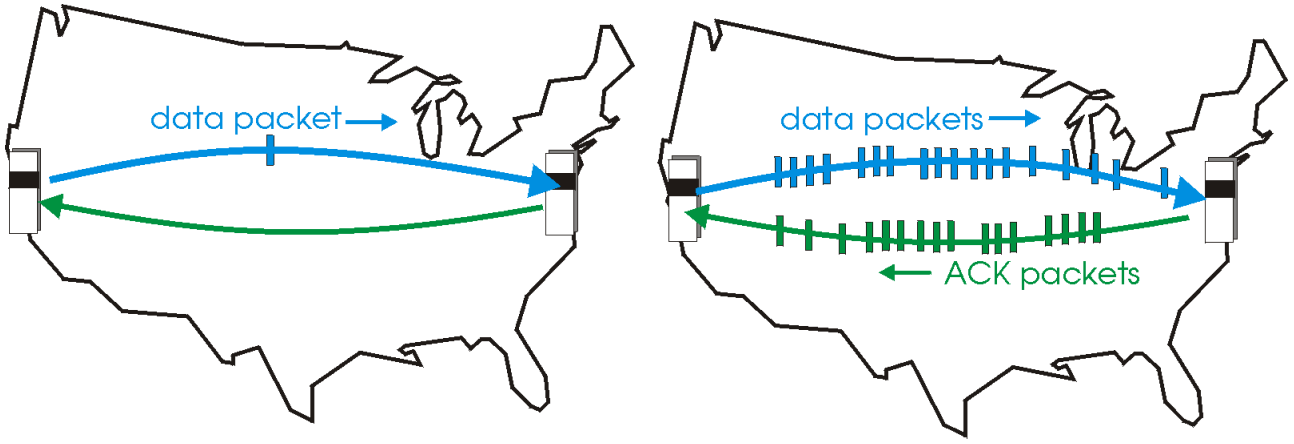
A Summary of Questions

- How to improve the performance of rdt3.0?
- What if there are reordering and duplication?
- How to determine the "right" timeout value?

Sliding Window Protocols: Pipelining

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

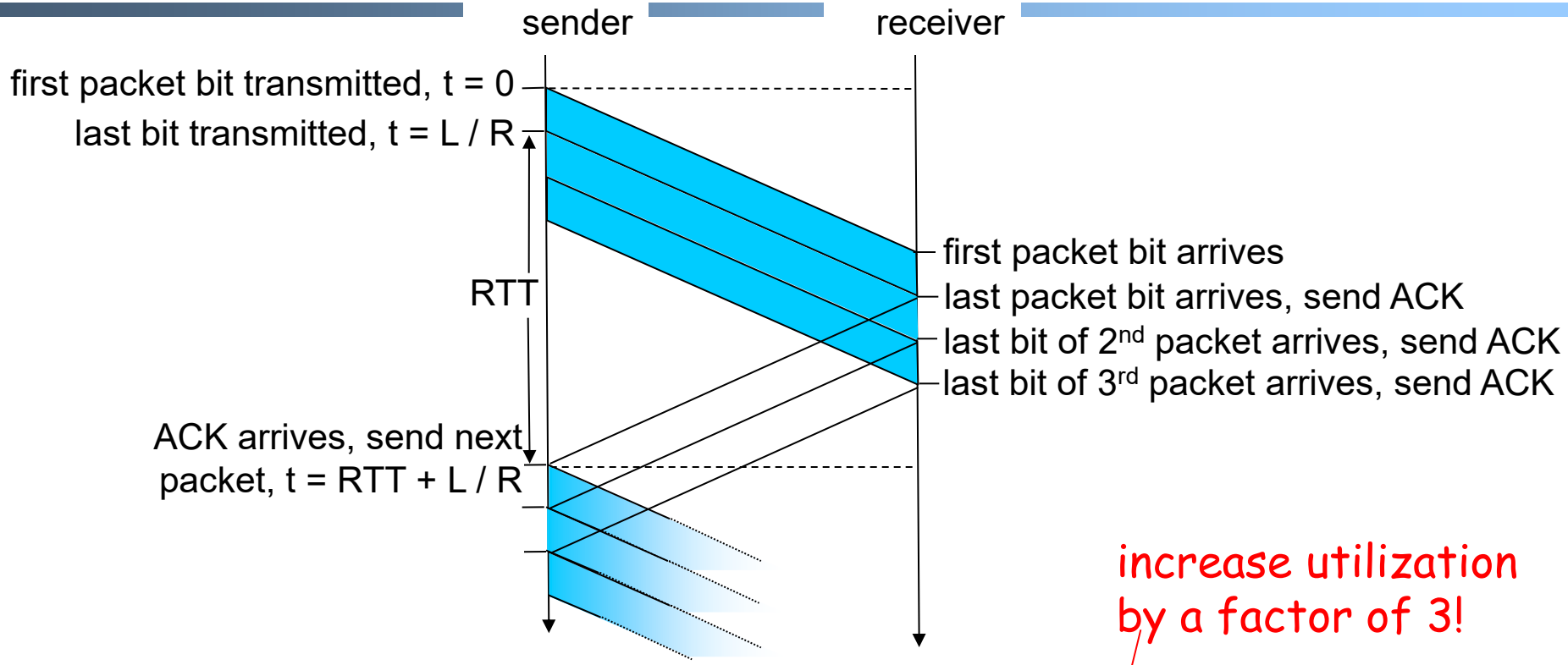
- range of sequence numbers must be increased
- buffering at sender and/or receiver



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

(b) a pipelined protocol in operation

Pipelining: Increased Utilization



increase utilization by a factor of 3!

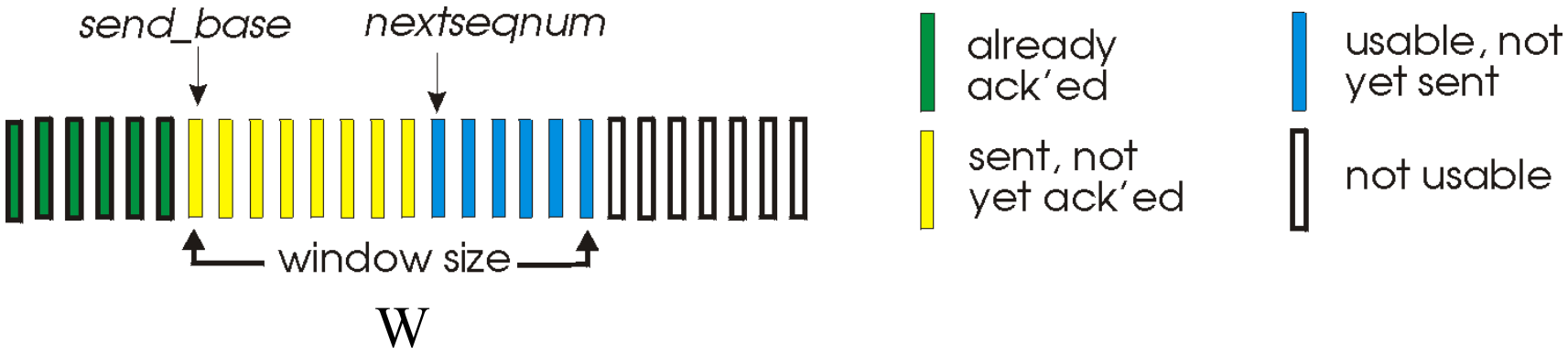
$$U_{sender} = \frac{3 * L / R}{RTT + L / R} = \frac{.024}{30.008} = 0.0008$$

Question: a rule-of-thumb window size?

Realizing Sliding Window: Go-Back-n

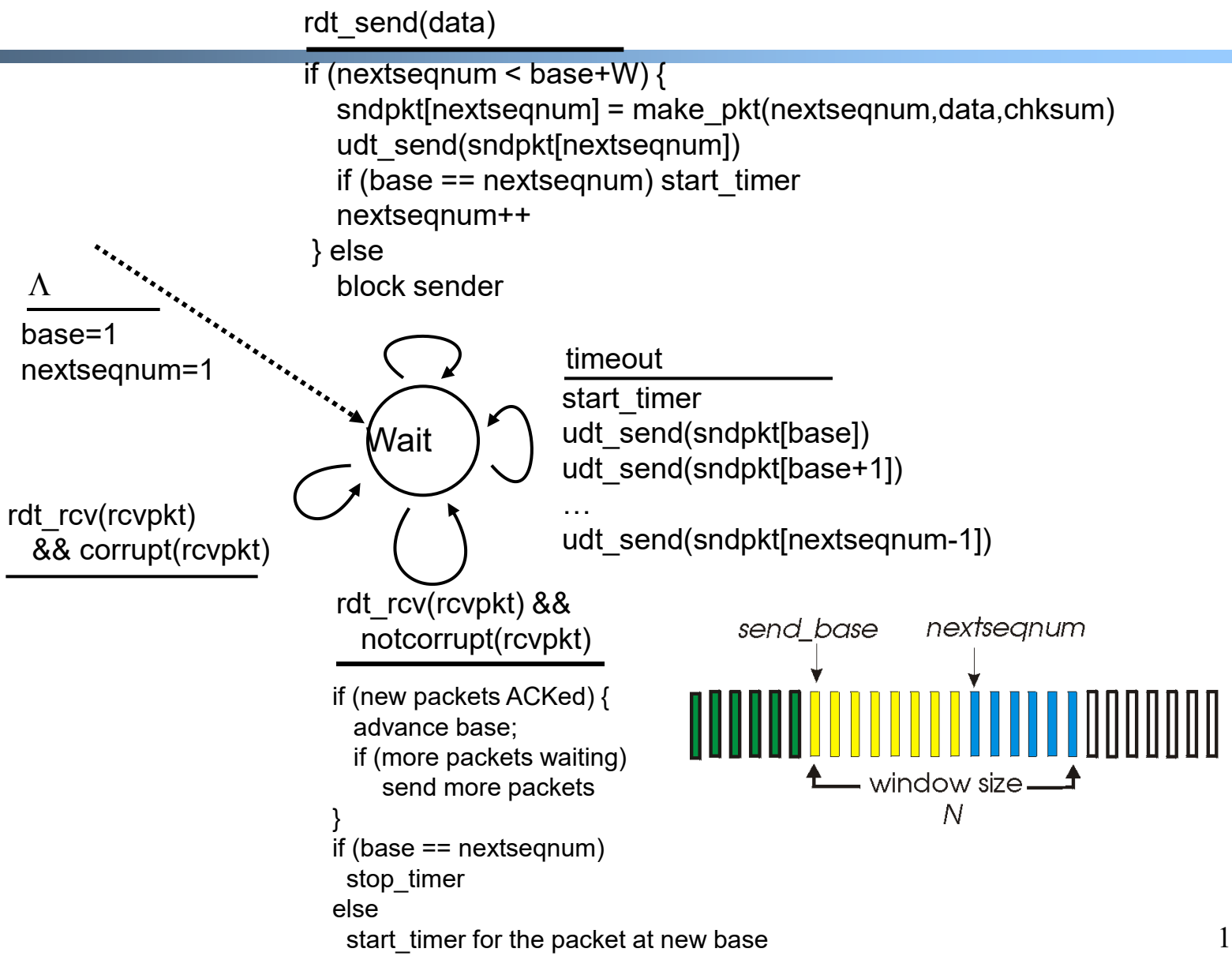
Sender:

- k-bit seq # in pkt header
- "window" of up to W, consecutive unack'ed pkts allowed

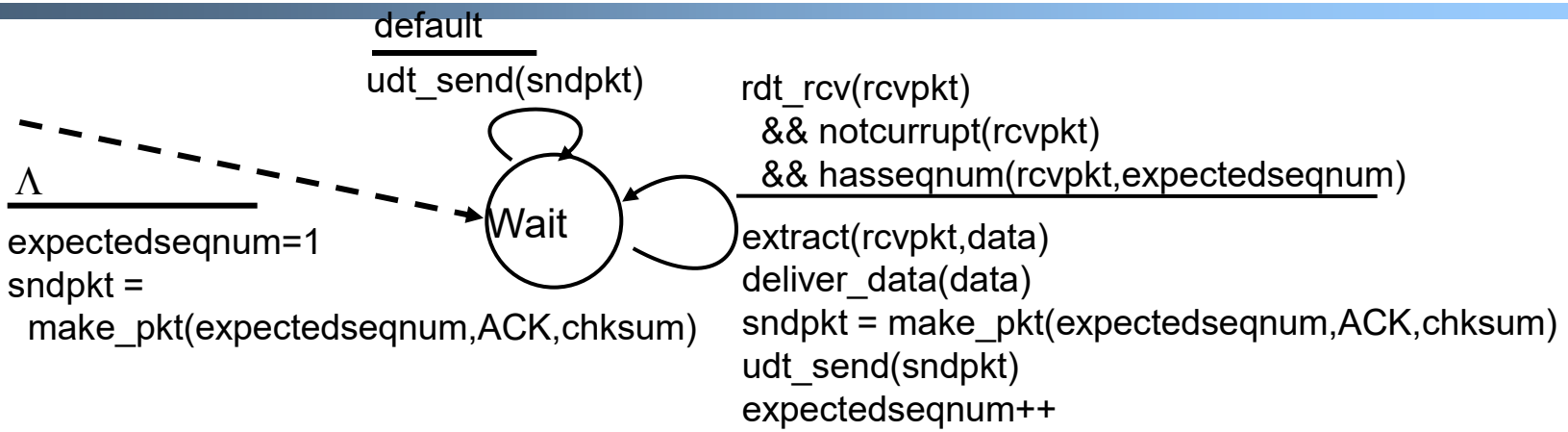


- ACK(n): ACKs all pkts up to, including seq # n - "cumulative ACK"
  - note: ACK(n) could mean two things: I have received upto and include n, or I am waiting for n
- timer for the packet at base
- *timeout(n)*: retransmit pkt n and all higher seq # pkts in window

GBN: Sender FSM



GBN: Receiver FSM

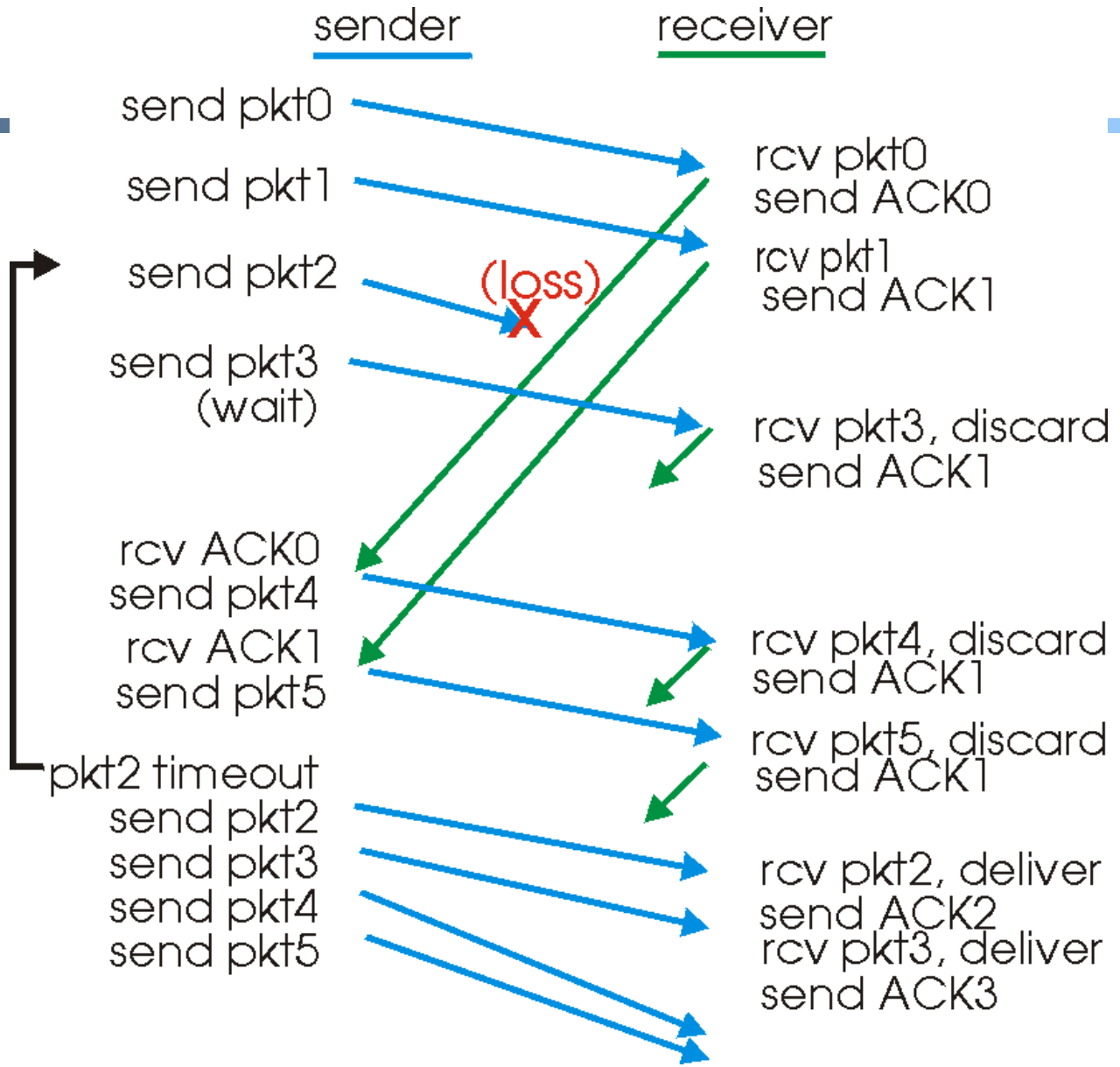


Only state: **`expectedseqnum`**

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

GBN in Action

window size = 4



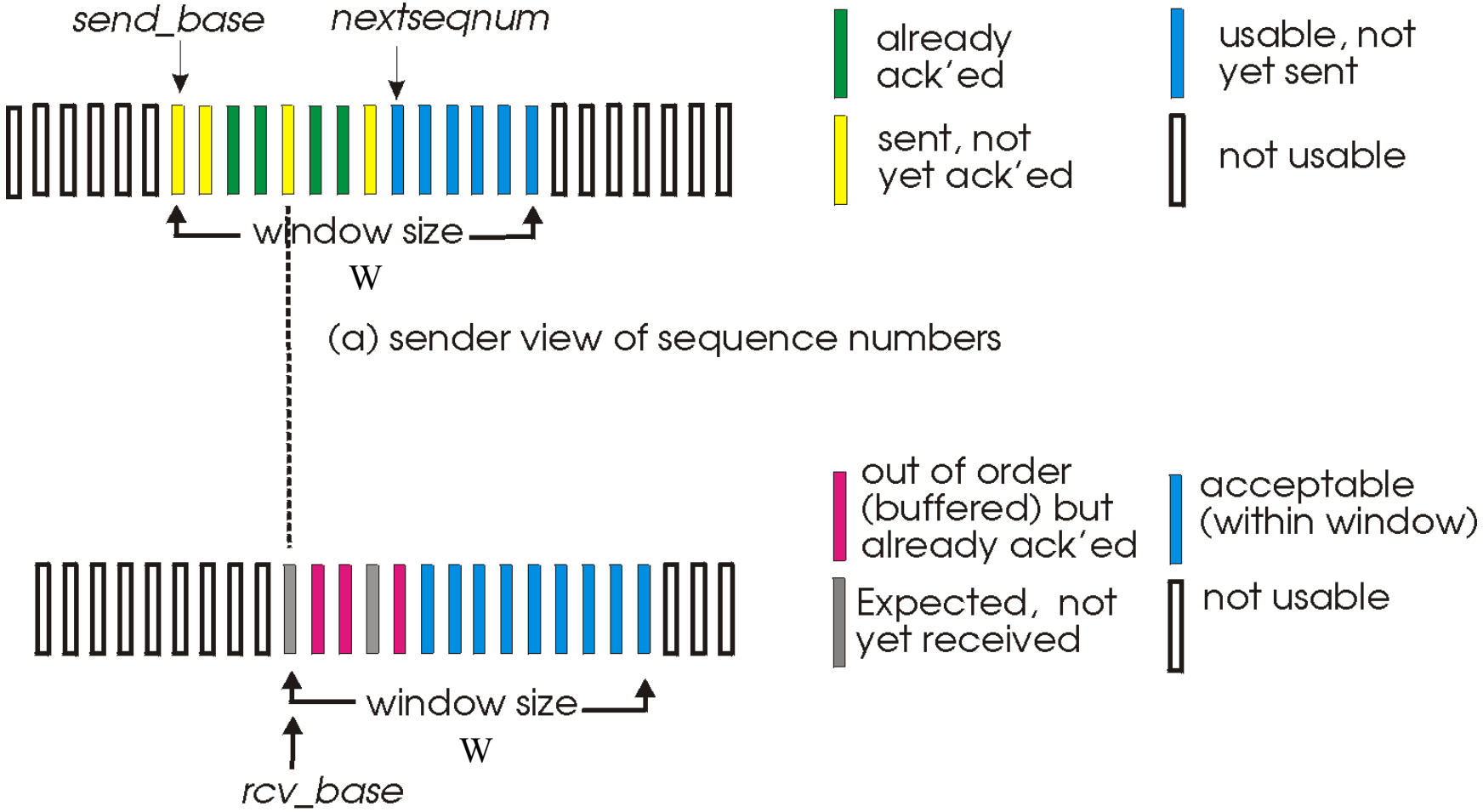
Analysis: Efficiency of Go-Back-n

- Assume window size W
- Assume each packet is lost with probability p
- On average, how many packets do we send for each data packet received?

Selective Repeat

- Sender window
  - Window size W: W consecutive unACKed seq #'s
- Receiver *individually* acknowledges correctly received pkts
  - buffers out-of-order pkts, for eventual in-order delivery to upper layer
  - ACK(n) means received packet with seq# n only
  - buffer size at receiver: window size
- Sender only resends pkts for which ACK not received
  - sender timer for each unACKed pkt

Selective Repeat: Sender, Receiver Windows



Selective Repeat

sender

data from above :

- unACKed packets is less than window size W, send; otherwise block app.

timeout(n):

- resend pkt n, restart timer

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

- mark pkt n as received
- update sendbase to the first packet unACKed

receiver

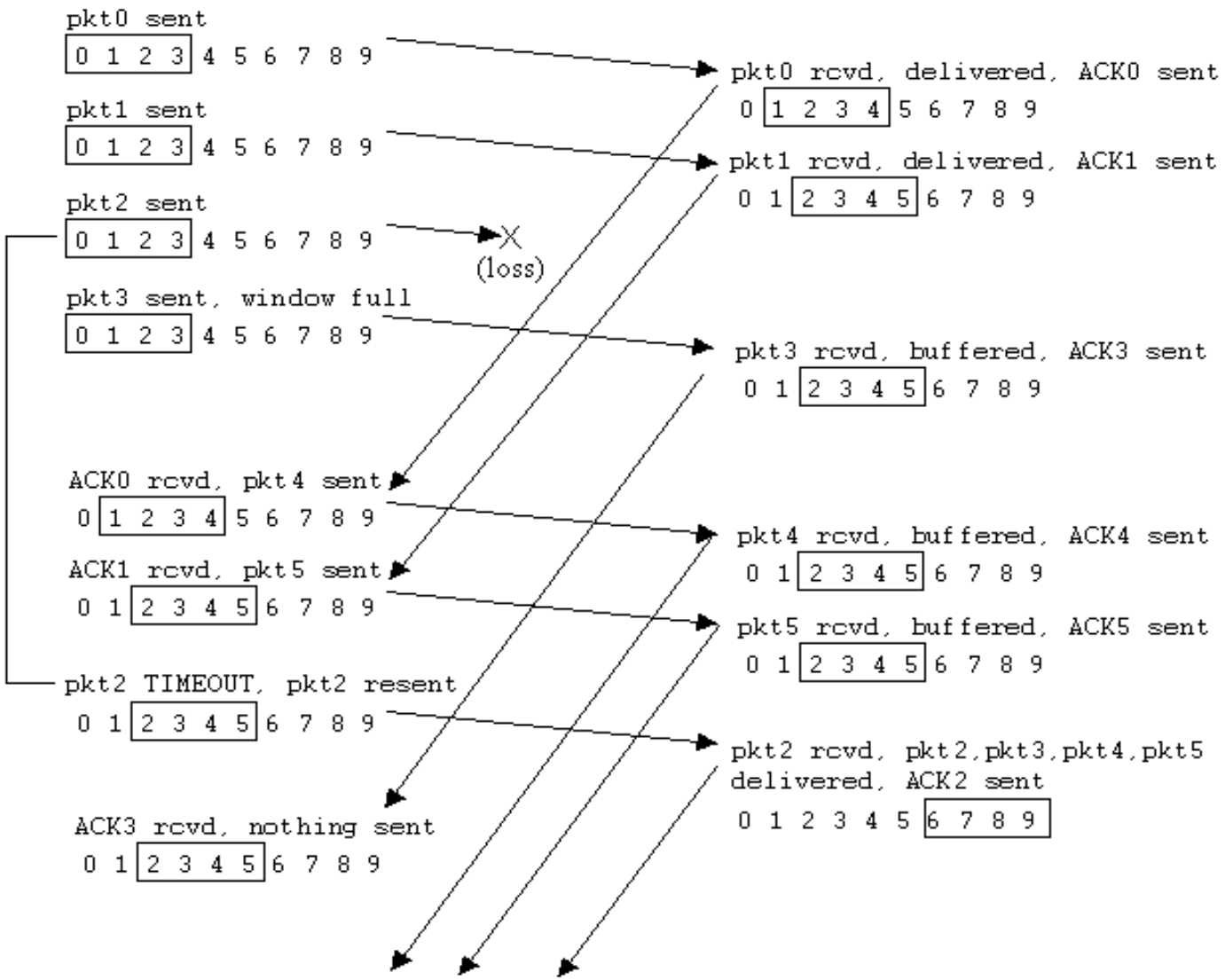
pkt n in [rcvbase, rcvbase+W-1]

- send ACK(n)
- if (out-of-order)
  mark and buffer pkt n
  else /*in-order*/
  deliver any in-order packets

otherwise:

- ignore

Selective Repeat in Action



Discussion: Efficiency of Selective Repeat

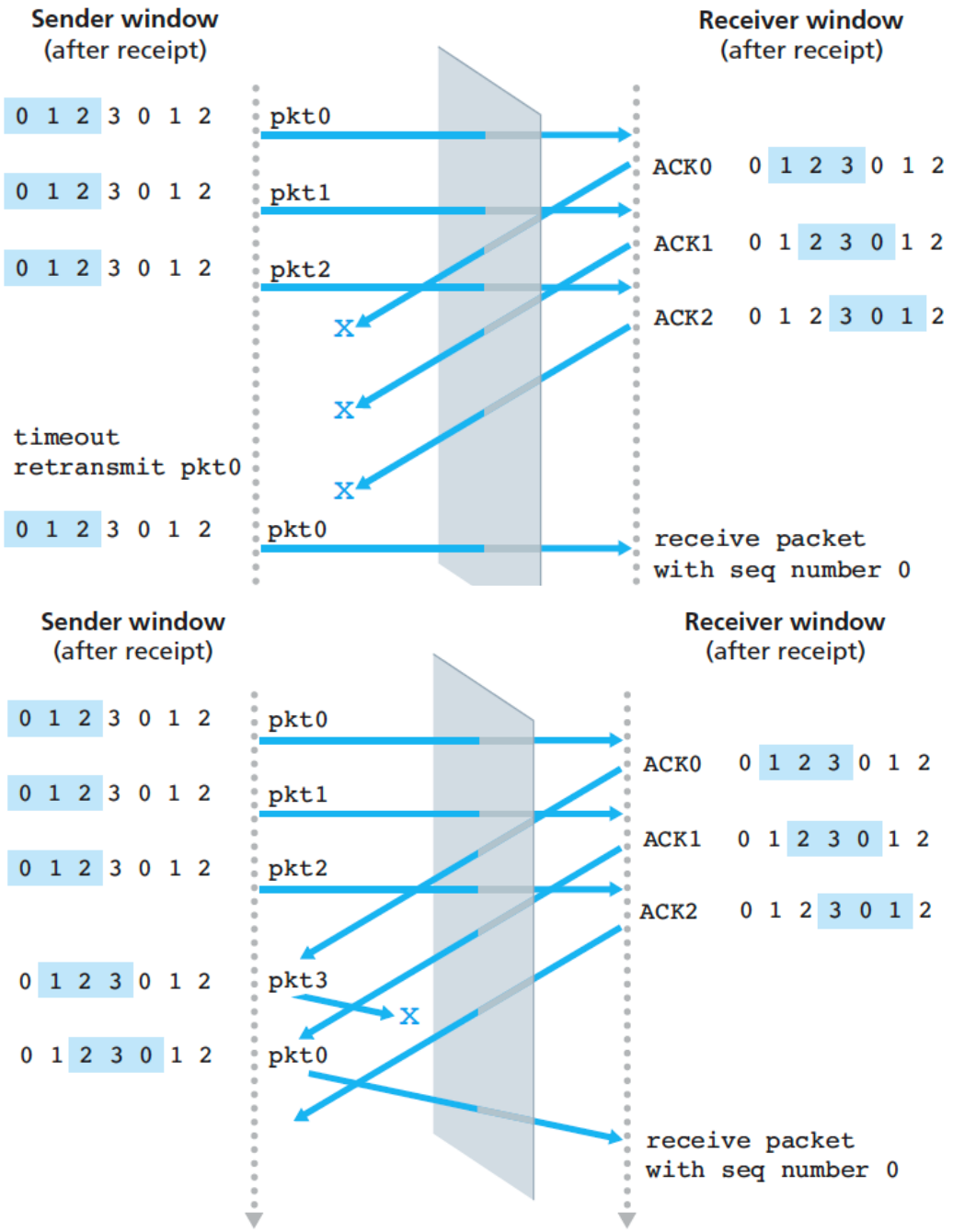
- Assume window size W
- Assume each packet is lost with probability p
- On average, how many packets do we send for each data packet received?

Selective Repeat: Seq# Ambiguity

Example:

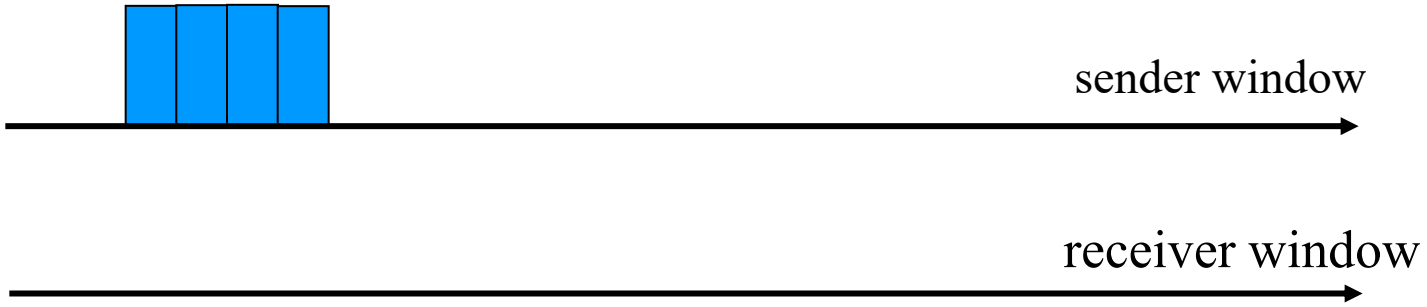
- seq #'s: 0, 1, 2, 3
- window size=3

- Error: incorrectly passes duplicate data as new.

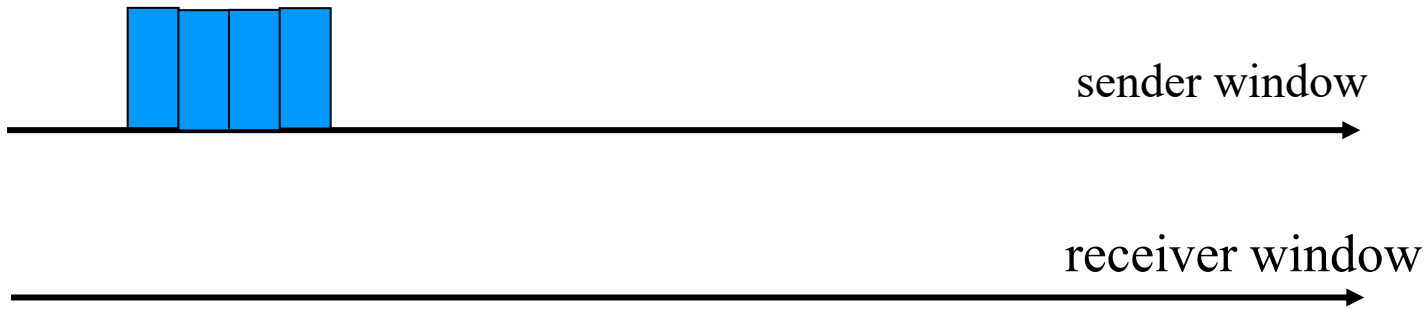


State Invariant: Window Location

- Go-back-n (GBN)



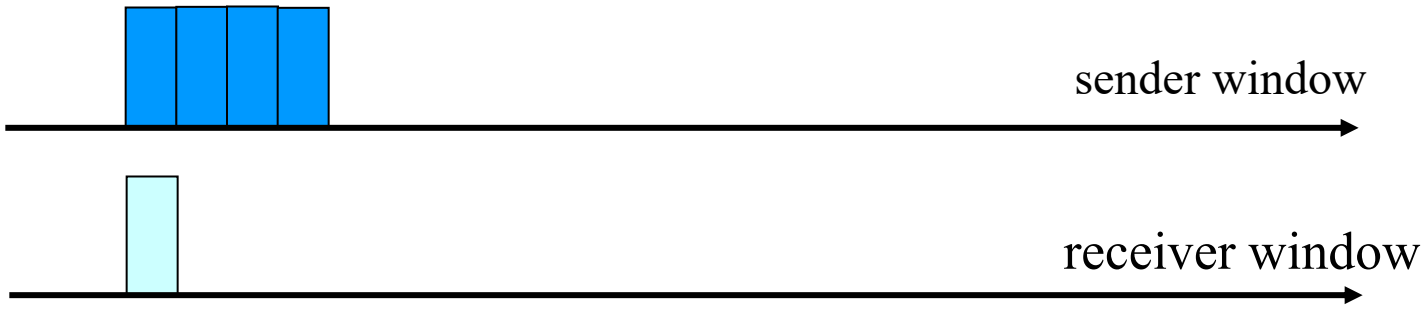
- Selective repeat (SR)



Window Location

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

- Go-back-n (GBN)



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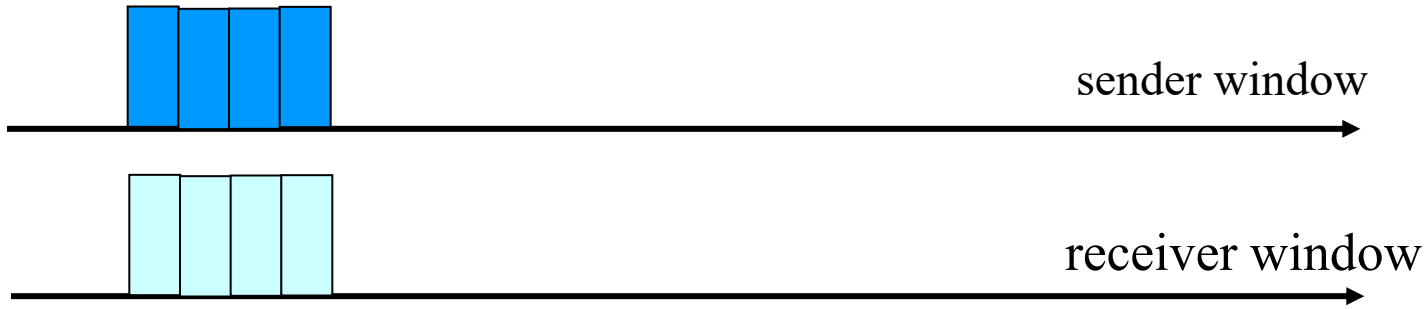

Selective Repeat

sender

data from above :

- unACKed packets is less than window size W, send; otherwise block app.

timeout(n):

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

- mark pkt n as received
- update sendbase to the first packet unACKed

receiver

pkt n in [rcvbase, rcvbase+W-1]

- send ACK(n)
- if (out-of-order)
  mark and buffer pkt n
  else /*in-order*/
  deliver any in-order packets

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

- send ACK(n)

otherwise:

- ignore

Sliding Window Protocols: Go-back-n and Selective Repeat

	Go-back-n	Selective Repeat
data bandwidth: sender to receiver (avg. number of times a pkt is transmitted)	Less efficient $\frac{1-p+pw}{1-p}$	More efficient $\frac{1}{1-p}$
ACK bandwidth (receiver to sender)	More efficient	Less efficient
Relationship between M (the number of seq#) and W (window size)	M > W	M ≥ 2W
Buffer size at receiver	1	W
Complexity	Simpler	More complex

p: the loss rate of a packet; M: number of seq# (e.g., 3 bit M = 8); W: window size

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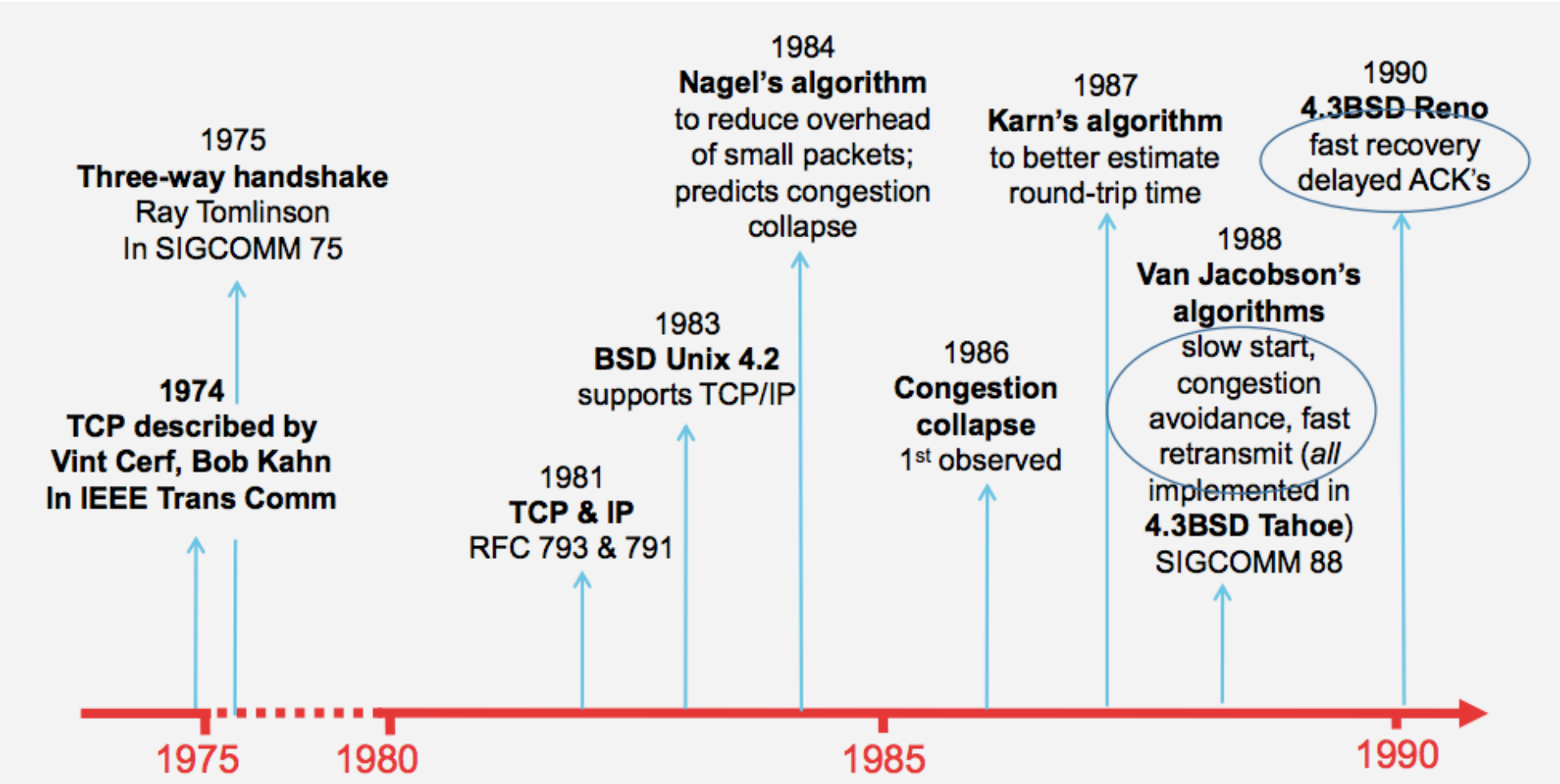
- Admin and Recap
- Reliable data transfer
  - perfect channel
  - channel with bit errors
  - channel with bit errors and losses
  - sliding window: reliability with throughput
- *TCP reliability*

TCP: Overview

RFCs: 793, 1122, 1323, 2018, 2581

- Point-to-point reliability: one sender, one receiver
- Flow controlled and congestion controlled

Evolution of TCP



Source: http://webcourse.cs.technion.ac.il/236341/Winter2015-2016/ho/WCFiles/Tutorial10.pdf

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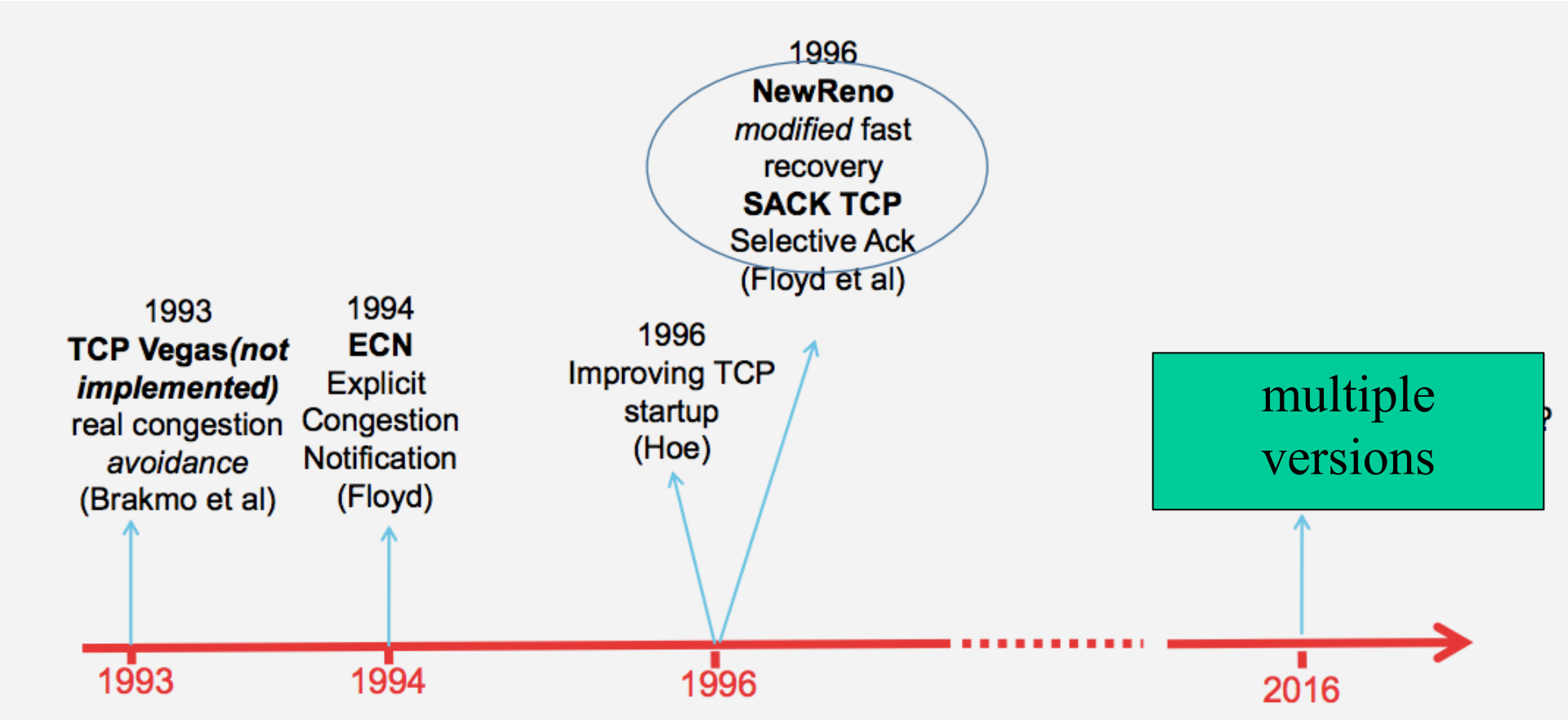
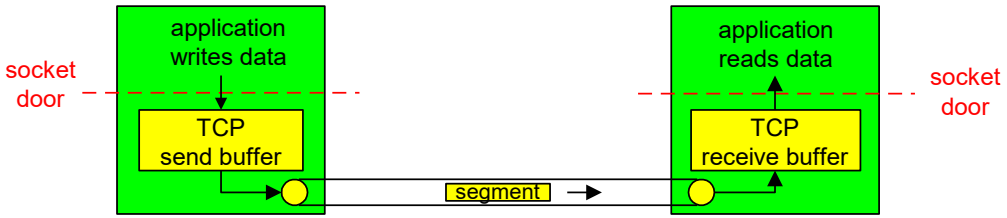


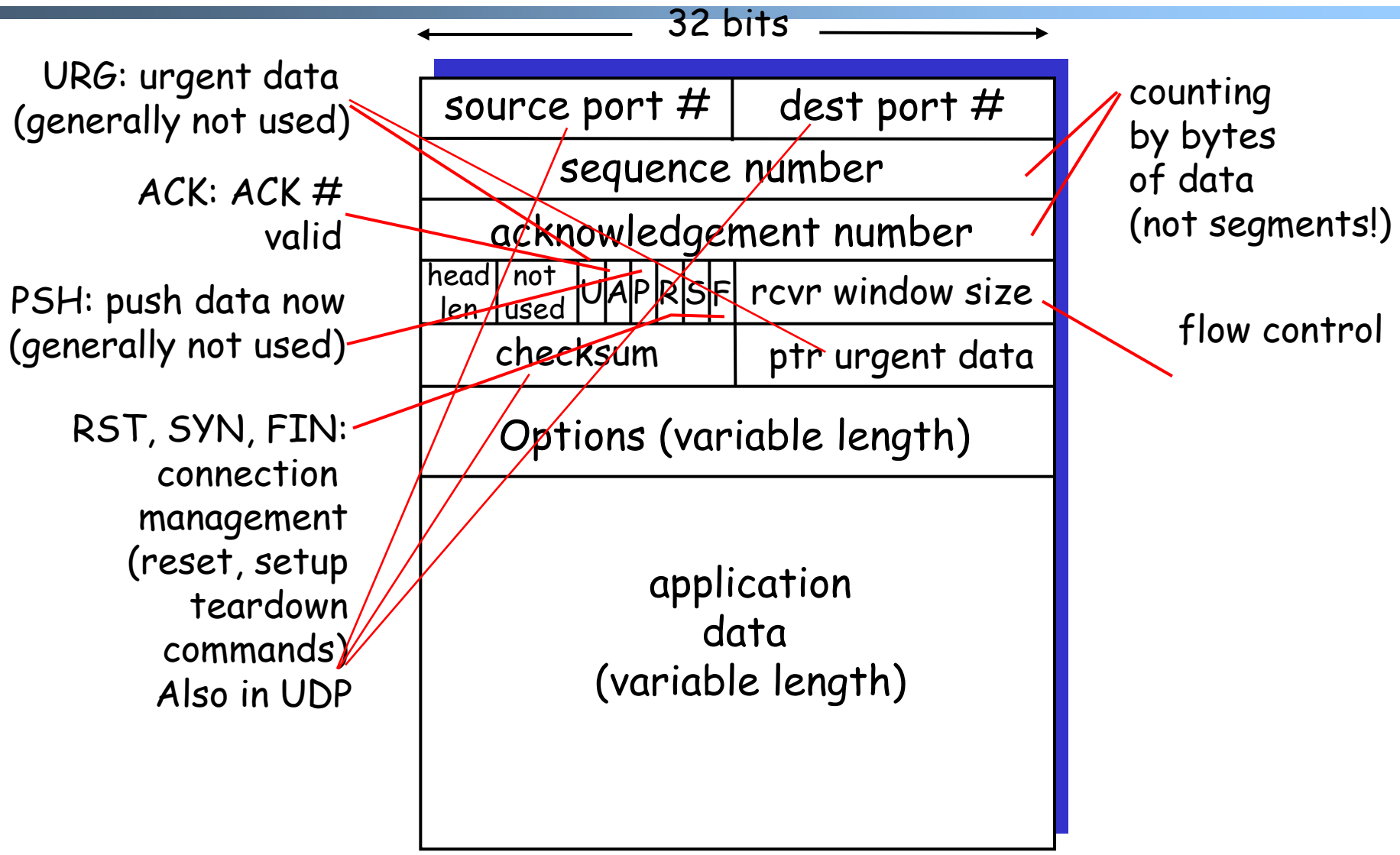
Source: http://webcourse.cs.technion.ac.il/236341/Winter2015-2016/ho/WCFiles/Tutorial10.pdf

TCP Reliable Data Transfer

- Connection-oriented:
  - connection management
    - setup (exchange of control msgs) init's sender, receiver state before data exchange
    - close
- Full duplex data:
  - bi-directional data flow in same connection
- A sliding window protocol
  - a combination of go-back-n and selective repeat:
    - send & receive buffers
    - cumulative acks
    - TCP uses a single retransmission timer
    - do not retransmit all packets upon timeout



TCP Segment Structure



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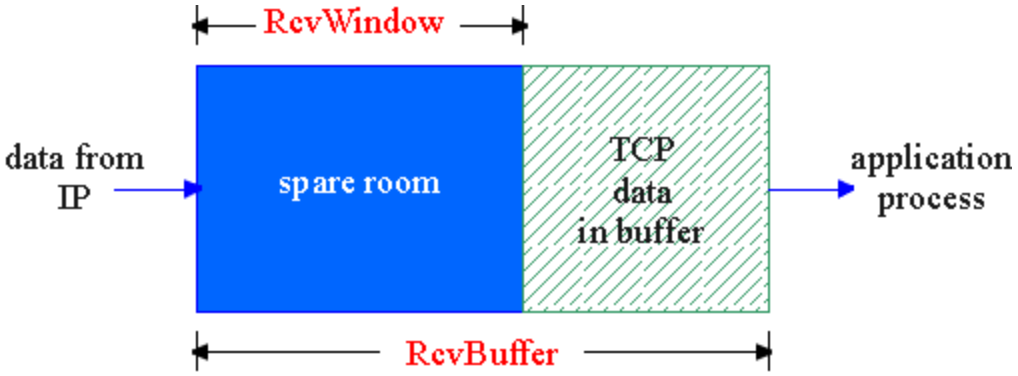
- TCP reliability
  - *data seq#, ack, buffering*

Flow Control

- receive side of a connection has a receive buffer:



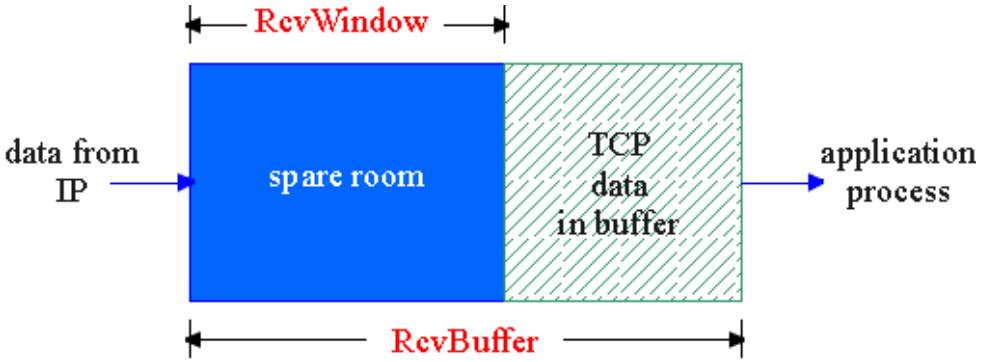
- app process may be slow at reading from buffer

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



- spare room in buffer

= **RcvWindow**

source port #	dest port #
sequence number	
acknowledgement number	
head len / not used / U A P R S F	rcvr window size
checksum	ptr urgent data
Options (variable length)	
application data (variable length)	

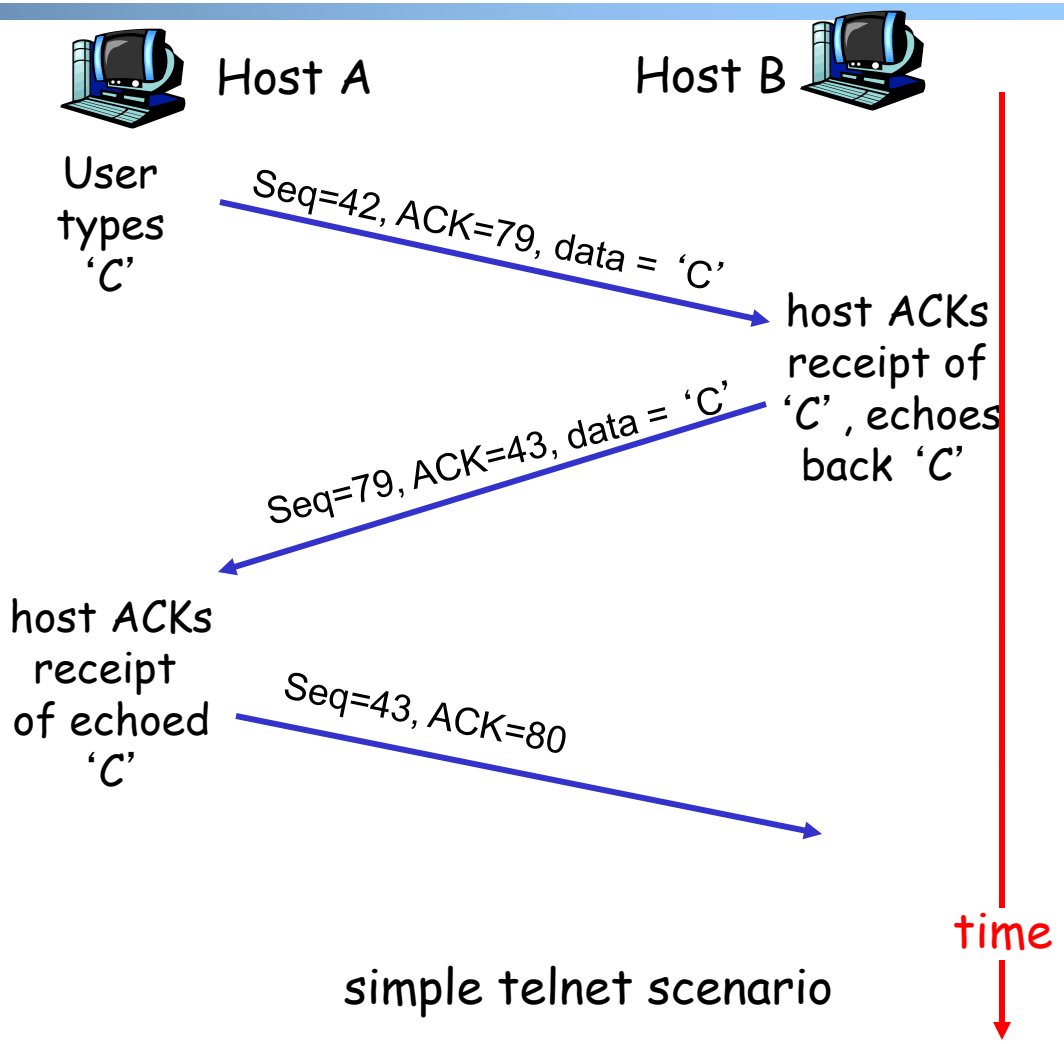
TCP Seq. #'s and ACKs

Seq. #'s:

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

ACKs:

- seq # of next byte expected from other side
- cumulative ACK in standard header
- selective ACK in options



simple telnet scenario

TCP Send/Ack Optimizations

- TCP includes many tune/optimizations, e.g.,
  - the "small-packet problem": sender sends a lot of small packets (e.g., telnet one char at a time)
    - Nagle's algorithm: do not send data if there is small amount of data in send buffer and there is an unack'd segment
  - the "ack inefficiency" problem: receiver sends too many ACKs, no chance of combing ACK with data
    - Delayed ack to reduce # of ACKs/combine ACK with reply

TCP Receiver 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

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  - sliding window: reliability with throughput
- TCP reliability
  - data seq#, ack, buffering
  - *timeout realization*

TCP Reliable Data Transfer

- Basic structure: sliding window protocol
- Remaining issue: How to determine the "right" parameters?
  - timeout value?
  - sliding window size?

History

- Key parameters for TCP in mid-1980s
  - fixed window size W
  - timeout value = 2 RTT
- Network collapse in the mid-1980s
  - UCB ←→ LBL throughput dropped by 1000X !
- The intuition was that the collapse was caused by wrong parameters…

Timeout: Cost of Timeout Param

Why is good timeout value important?

- too short
  - premature timeout
  - unnecessary retransmissions; many duplicates
- too long
  - slow reaction to segment loss

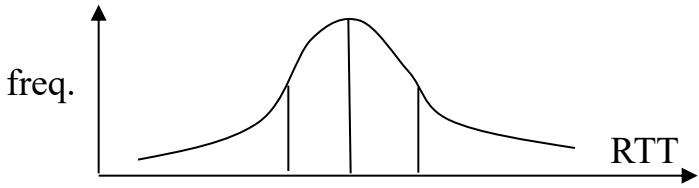
Q: Is it possible to set Timeout as a constant?

Q: Any problem w/ the early approach: Timeout = 2 RTT

Setting Timeout

Problem:

- Ideally, we set timeout = RTT,
  but RTT is not a fixed value
  =>
  using the average of **RTT** will generate many timeouts due to network variations
- Possibility: using the average/median of RTT
- Issue: this will generate many timeouts due to network variations



Solution:

- **Set Timeout RTO = avg +** "safety margin" based on variation

TCP approach:

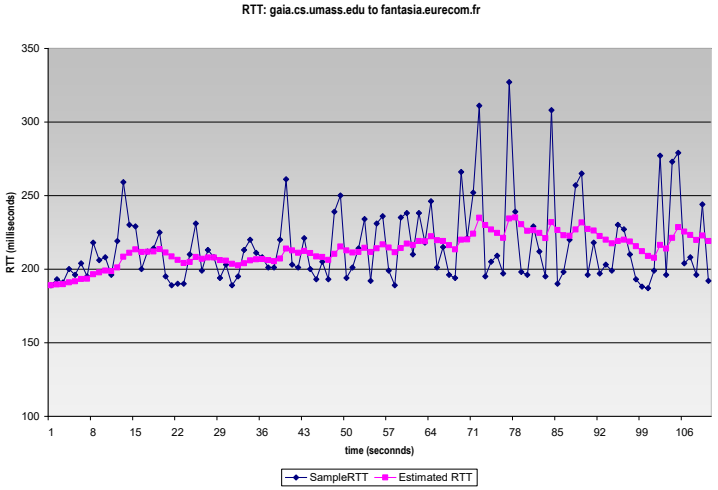
Timeout = EstRTT + 4 * DevRTT

Compute EstRTT and DevRTT

- Exponential weighted moving average (EWMA)
  - influence of past sample decreases exponentially fast

```
EstRTT = (1-alpha)*EstRTT + alpha*SampleRTT
```

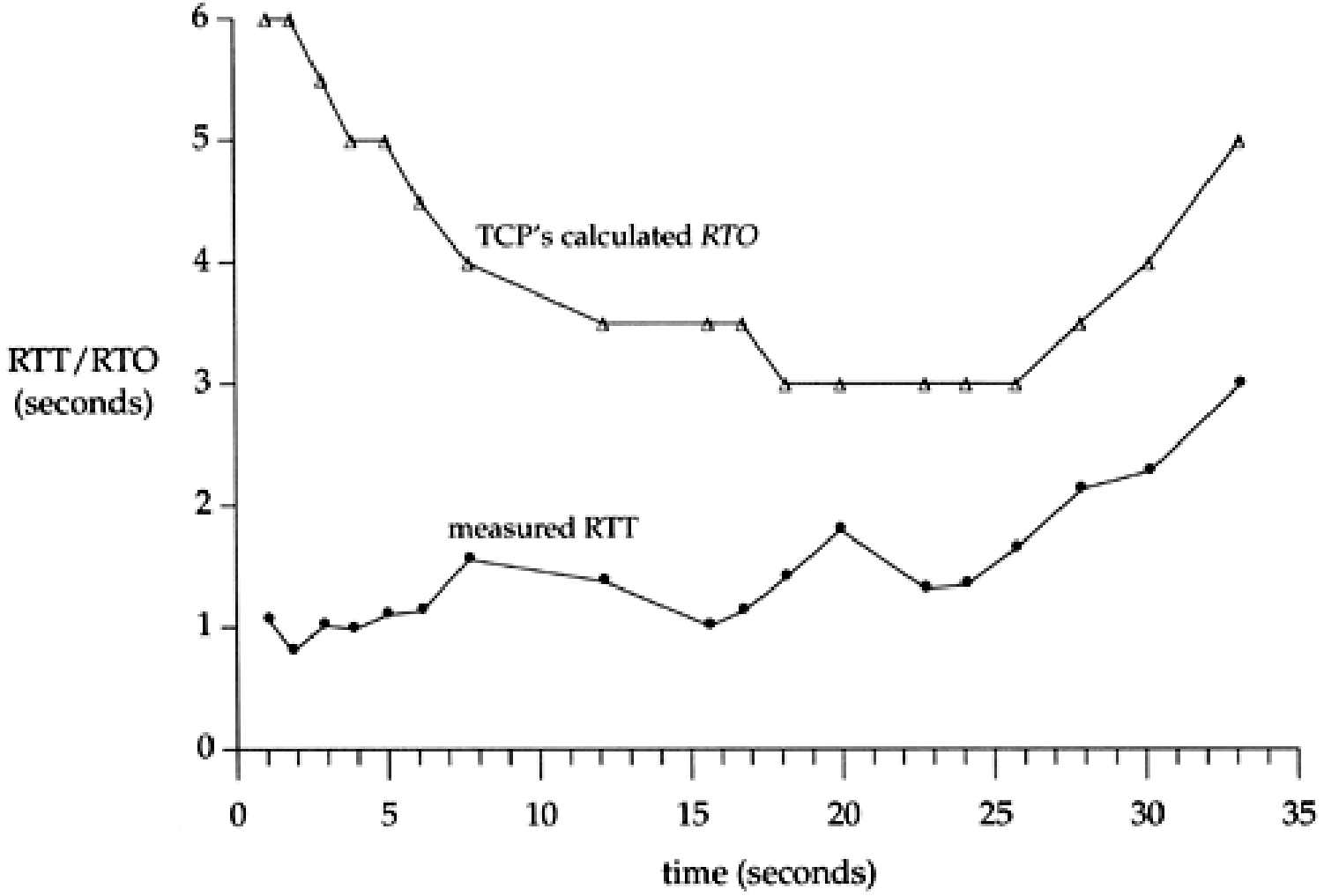
- **SampleRTT**: measured time from segment transmission until ACK receipt
- typical value: **alpha** = 0.125




```
DevRTT = (1-beta)*DevRTT + beta|SampleRTT-EstRTT|

(typically, beta = 0.25)
```

An Example TCP Session



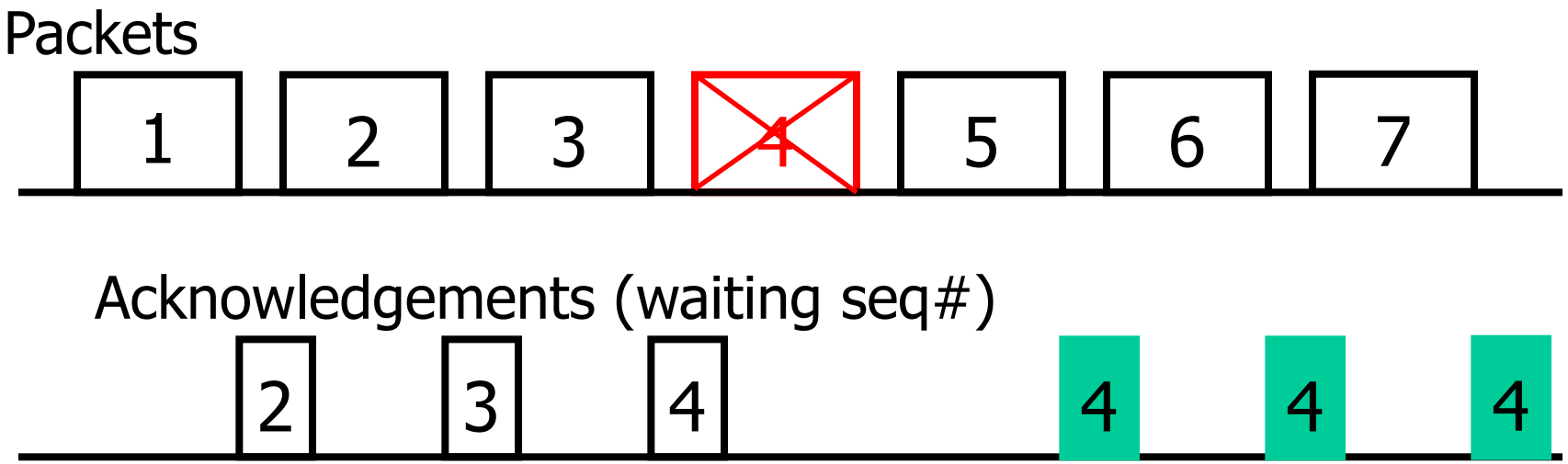
Fast Retransmit

- Issue: Timeout period often relatively long:
  - long delay before resending lost packet
- Question: Can we detect loss faster than RTT?

- 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 ACKs for the same data, it supposes that segment after ACKed data was lost:
  - resend segment before timer expires

Triple Duplicate Ack



Fast Retransmit:

```
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

TCP: reliable data transfer

Simplified TCP sender

```
00  sendbase = initial_sequence number agreed by TWH
01   nextseqnum = initial_sequence number by TWH
02    loop (forever) {
03      switch(event)
04      event: data received from application above
05                if (window allows send)
06                   create TCP segment with sequence number nextseqnum
06                   if (no timer) start timer
07                   pass segment to IP
08                   nextseqnum = nextseqnum + length(data)
                  else put packet in buffer
09       event: timer timeout for sendbase
10            retransmit segment
11            compute new timeout interval
12            restart timer
13       event: ACK received, with ACK field value of y
14            if (y > sendbase) { /* cumulative ACK of all data up to y */
15                 cancel the timer for sendbase
16                 sendbase = y
17                 if (no timer and packet pending) start timer for new sendbase
17                 while (there are segments and window allow)
18                      sent a segment;
18            }
19            else { /* y==sendbase, duplicate ACK for already ACKed segment */
20                 increment number of duplicate ACKs received for y
21                 if (number of duplicate ACKS received for y == 3) {
22                      /* TCP fast retransmit */
23                    resend segment with sequence number y
24                    restart timer for segment y
25                 }
26      }  /* end of loop forever */
```

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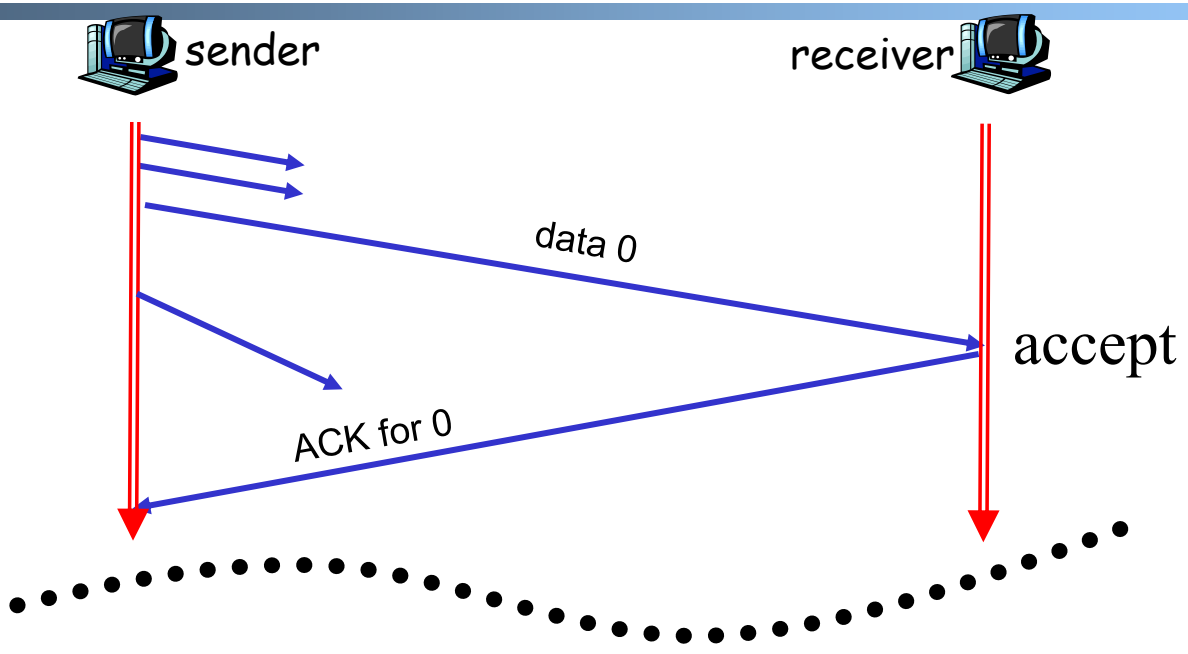
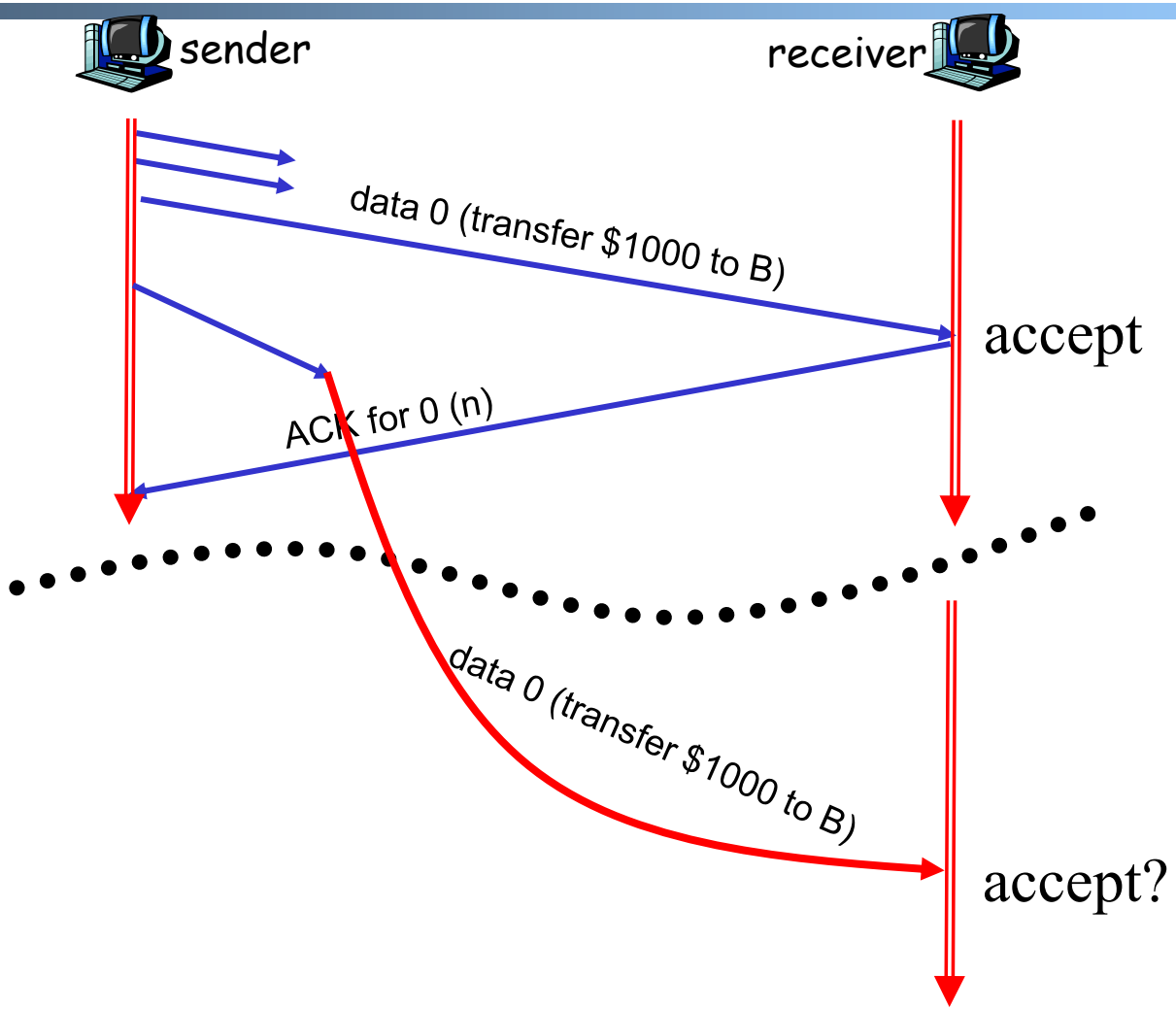
- TCP reliability
  - data seq#, ack, buffering
  - timeout realization
  - *connection management*

Why Connection Setup/When to Accept (Safely Deliver) First Packet?



Why Connection Setup/When to Accept (Safely Deliver) First Packet?



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Transport "Safe-Setup" Principle

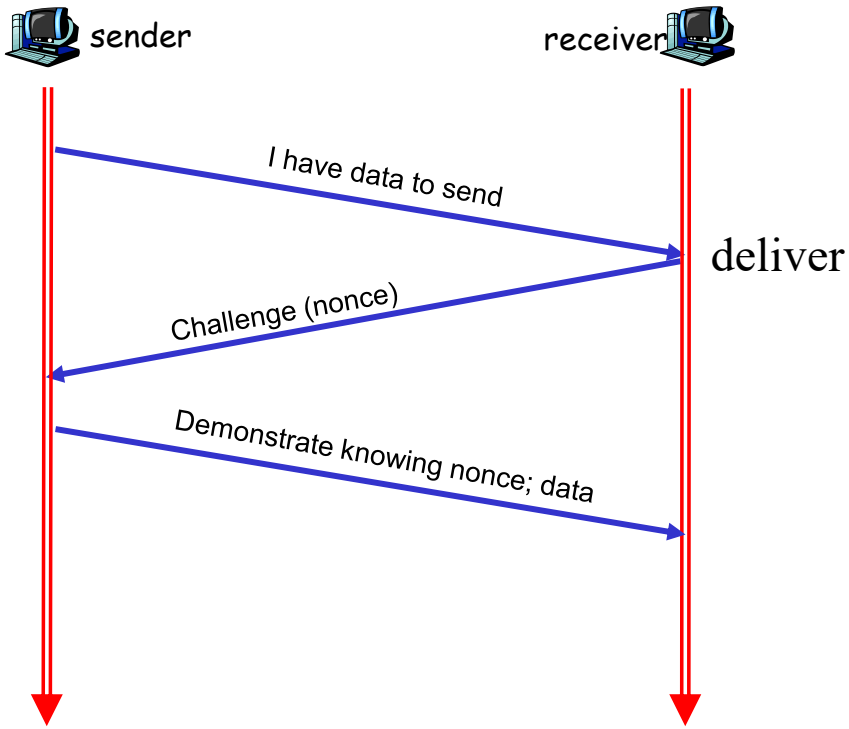
- A general safety principle for a receiver R to accept a message from a sender S is the general "authentication" principle, which consists of two conditions:

Transport authentication principle:
- [p1] Receiver can be sure that what Sender says is **fresh**
- [p2] Receiver receives something that ***only*** Sender can say

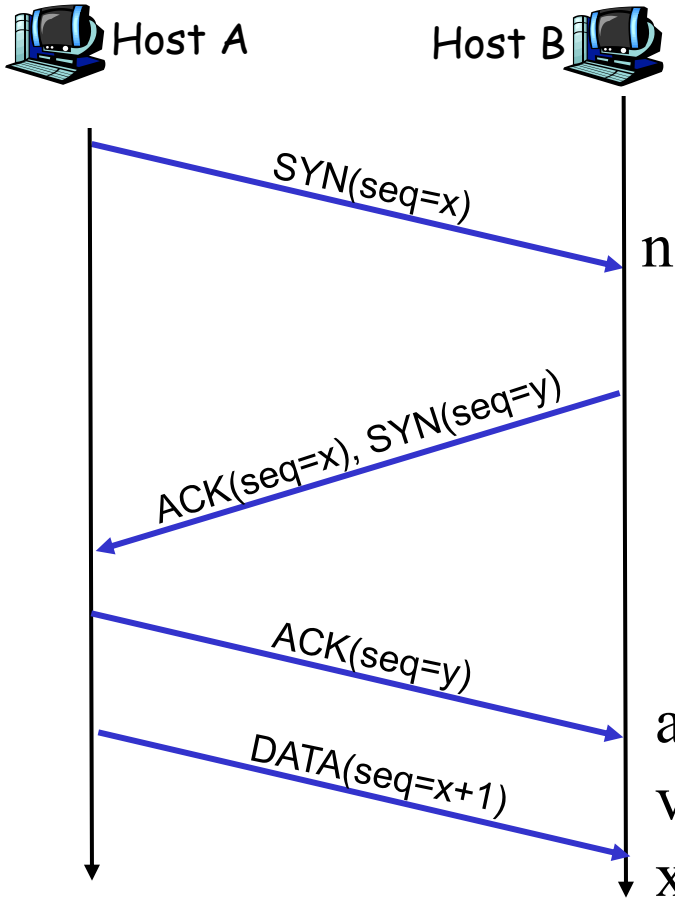
We first assume a secure setting: no malicious attacks.

Exercise: Techniques to allow a receiver to check for freshness (e.g., add a time stamp)?

Generic Challenge-Response Structure Checking Freshness



Three Way Handshake (TWH) [Tomlinson 1975]



notify initial seq#. Accept?

think of y as a challenge

accept data only after
verified y is bounced back
x is the init. seq

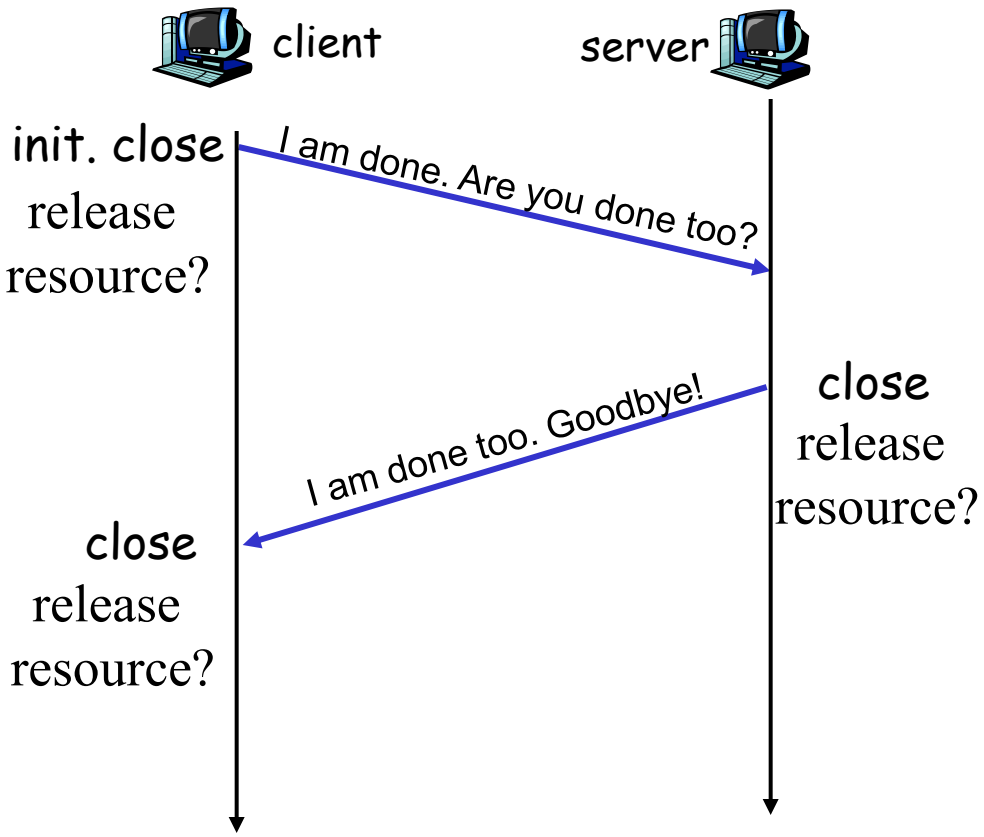
SYN: indicates connection setup

Make "Challenge y" Robust

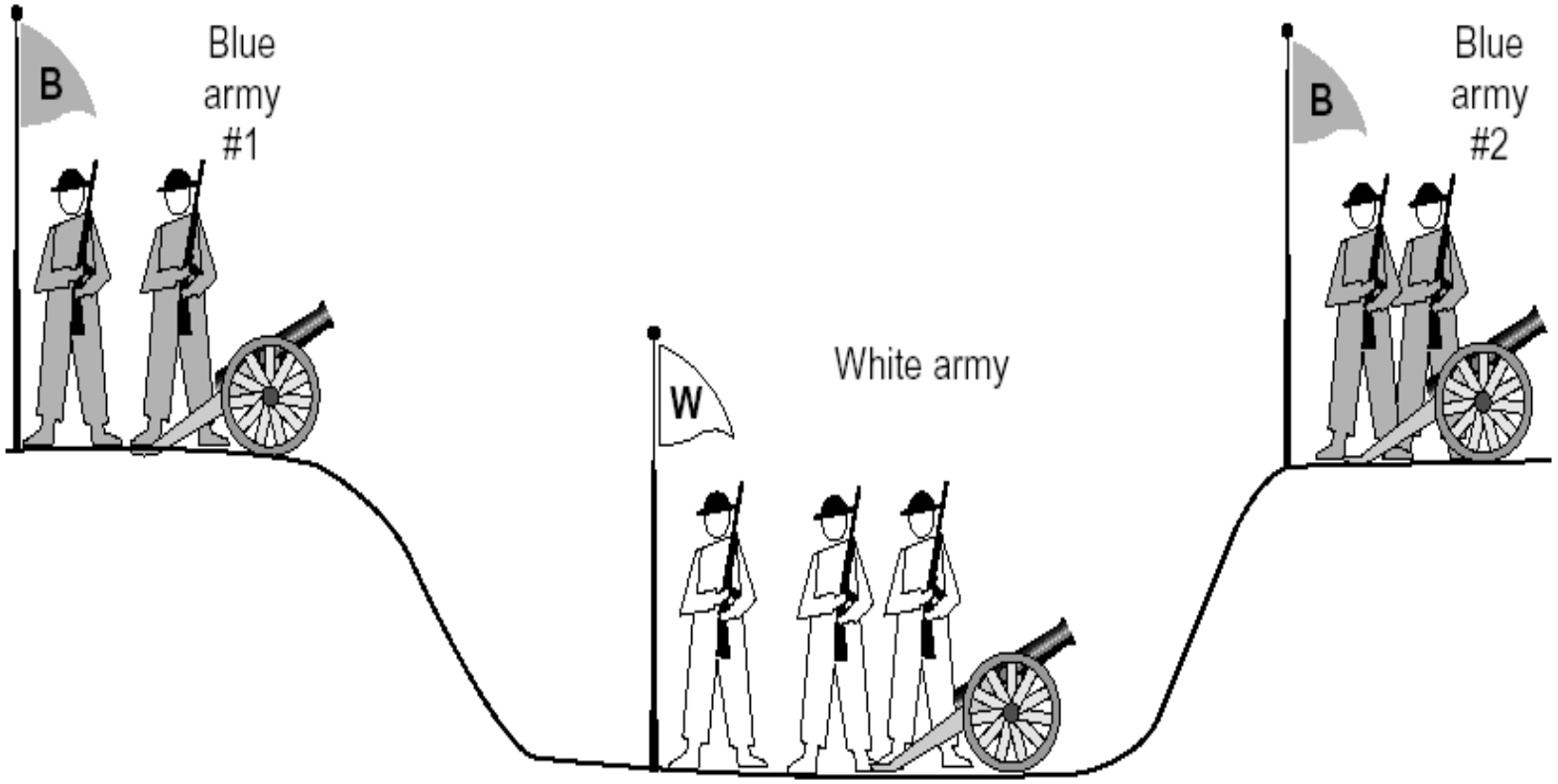
- To avoid that "SYNC ACK y" comes from reordering and duplication
  - for each connection (sender-receiver pair), ensuring that two identically numbered packets are never outstanding at the same time
    - network bounds the life time of each packet
    - a sender will not reuse a seq# before it is sure that all packets with the seq# are purged from the network
    - seq. number space should be large enough to not limit transmission rate
- Increasingly move to cryptographic challenge and response

Connection Close

- Why connection close?
  - so that each side can release resource and remove state about the connection (do not want dangling socket)



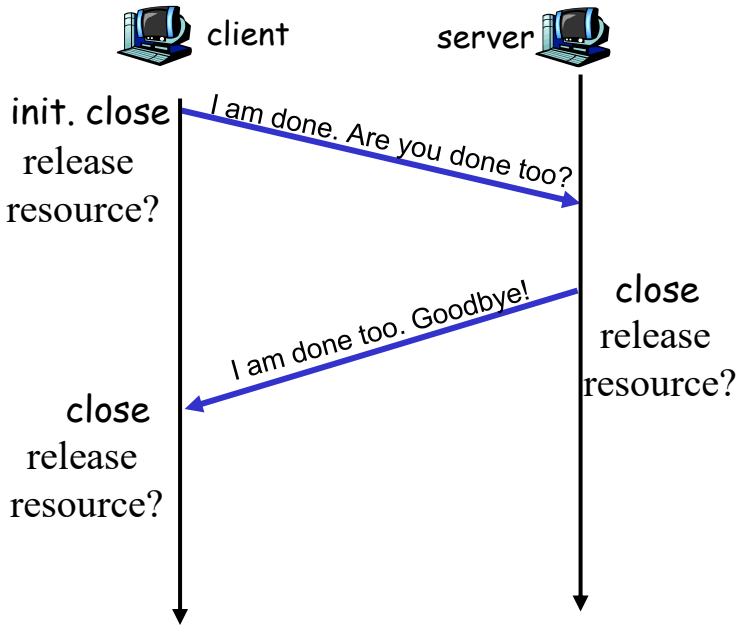
General Case: The Two-Army Problem

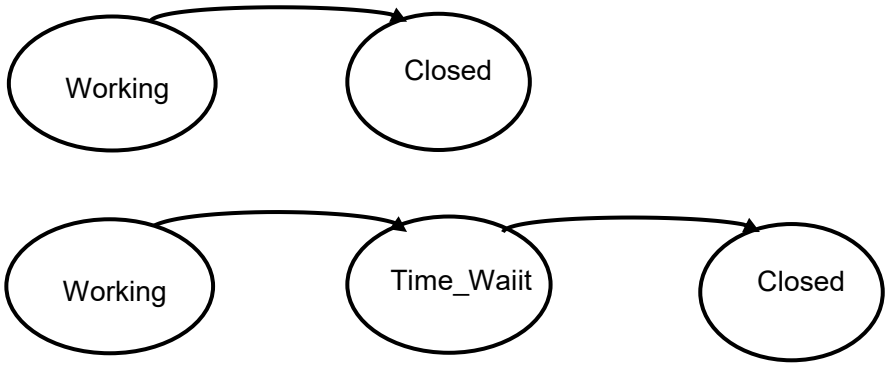


The gray (blue) armies need to agree on whether or not they will attack the white army. They achieve agreement by sending messengers to the other side. If they both agree, attack; otherwise, no. Note that a messenger can be captured!

Time_Wait

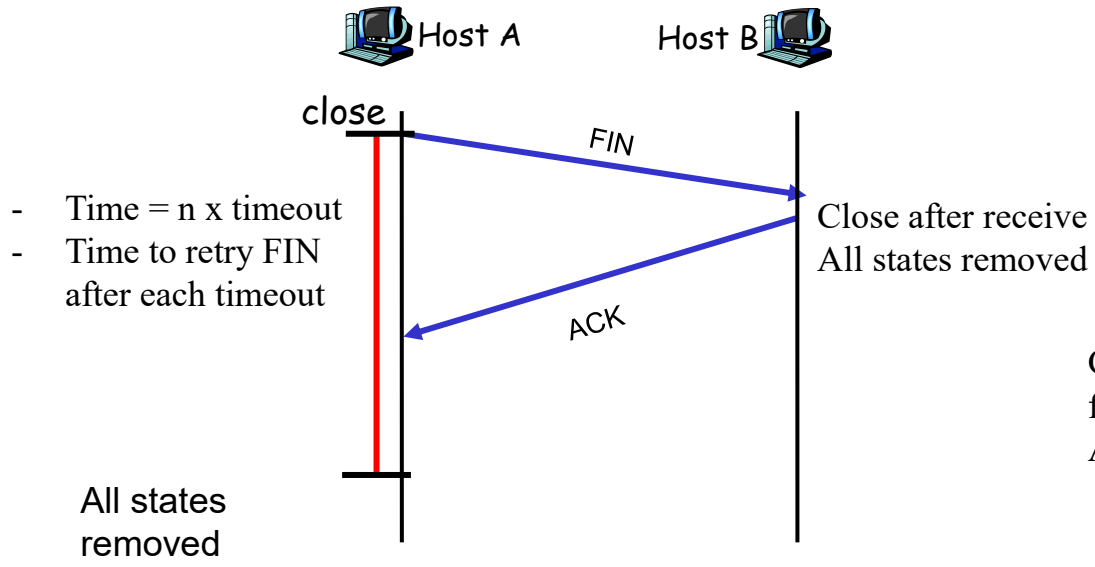
- Generic technique: Timeout to "solve" infeasible problem
  - Instead of message-driven state transition, use a timeout based transition; use timeout to handle error cases



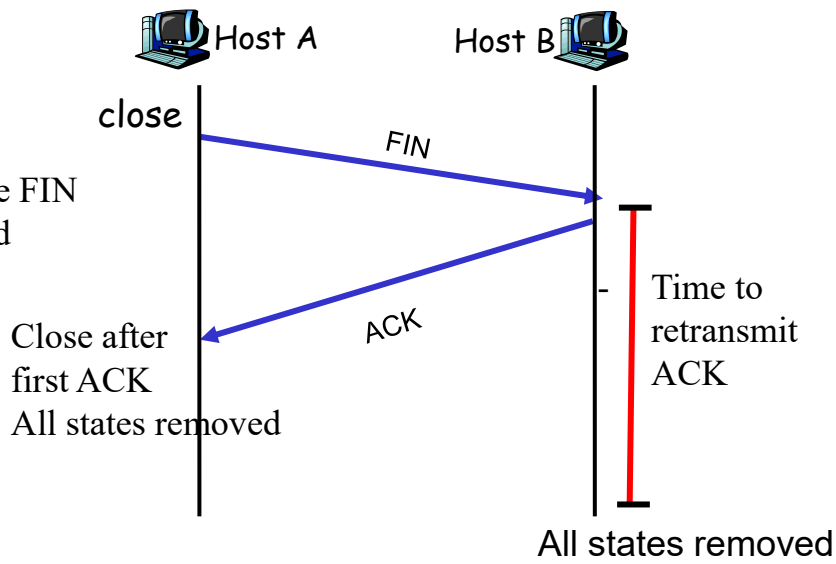


Time_Wait Design Options

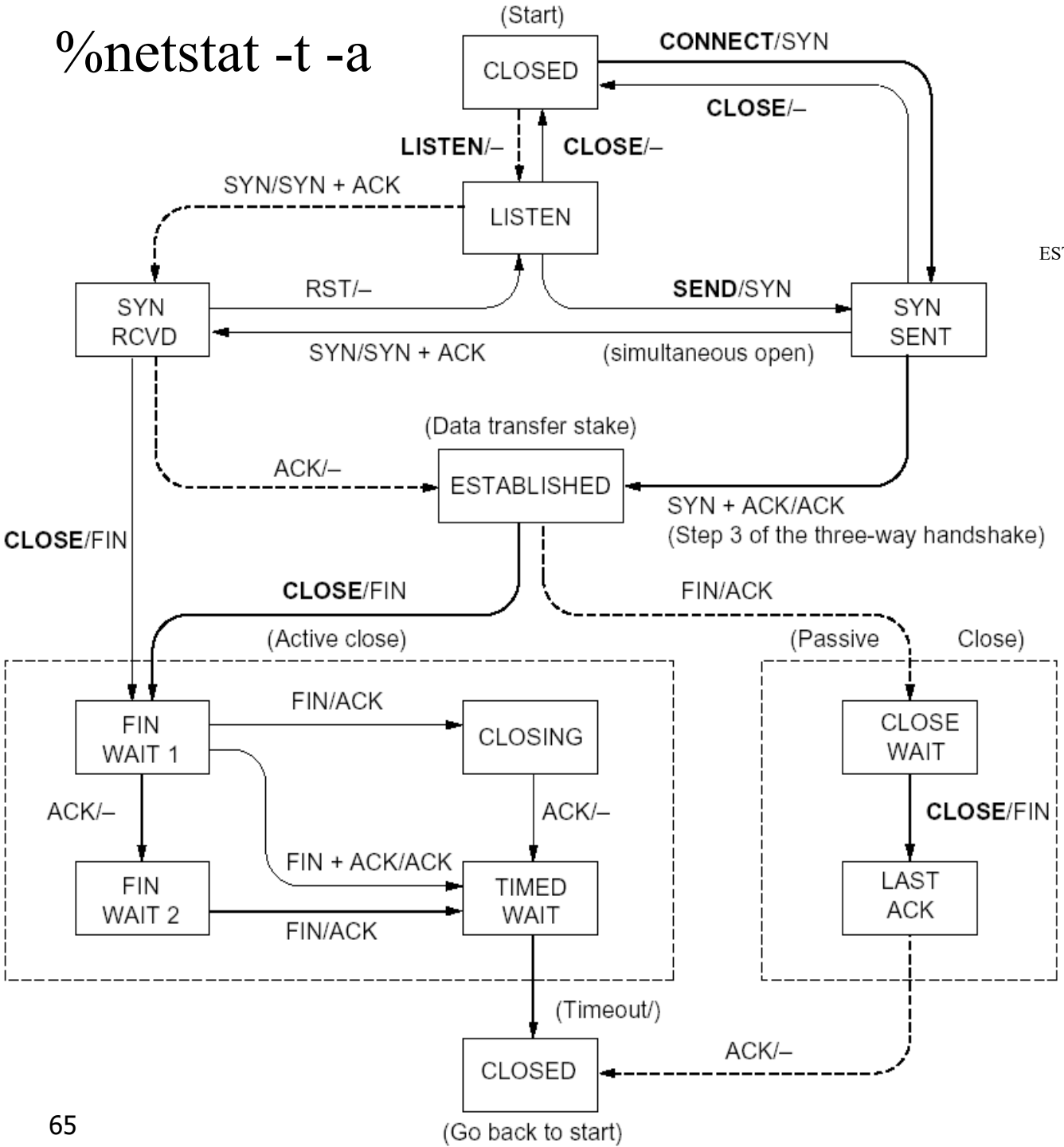
Design 1 (initiator time wait)

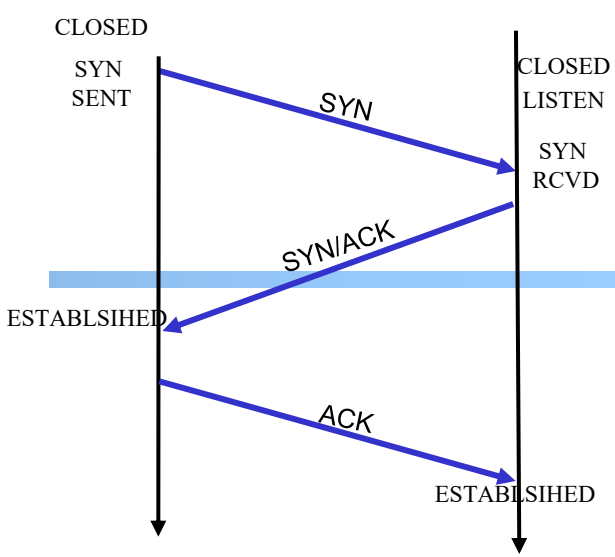


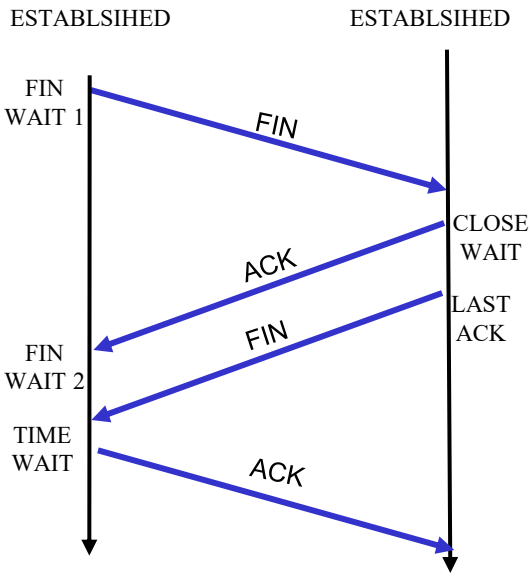
Design 2 (receiver time wait)



%netstat -t -a

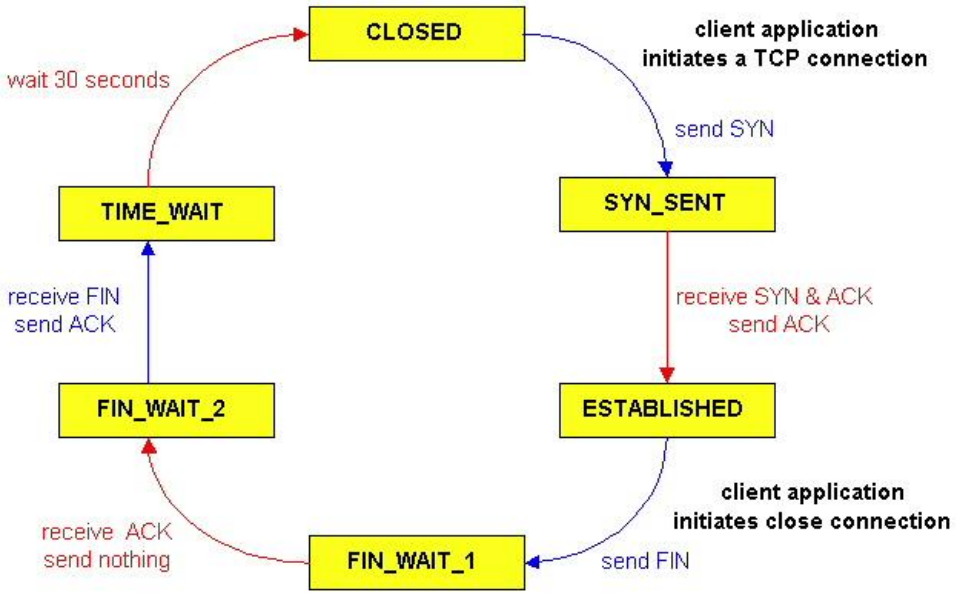

(Start)
CLOSED
CONNECT/SYN
CLOSE/–
LISTEN/–
CLOSE/–
SYN/SYN + ACK
LISTEN
SYN RCVD
RST/–
SEND/SYN
SYN SENT
SYN/SYN + ACK
(simultaneous open)
(Data transfer stake)
ACK/–
ESTABLISHED
SYN + ACK/ACK
(Step 3 of the three-way handshake)
CLOSE/FIN
CLOSE/FIN
FIN/ACK
(Active close)
(Passive Close)
FIN WAIT 1
FIN/ACK
CLOSING
ACK/–
ACK/–
FIN + ACK/ACK
FIN WAIT 2
TIMED WAIT
FIN/ACK
CLOSE WAIT
CLOSE/FIN
LAST ACK
(Timeout/)
CLOSED
ACK/–
(Go back to start)

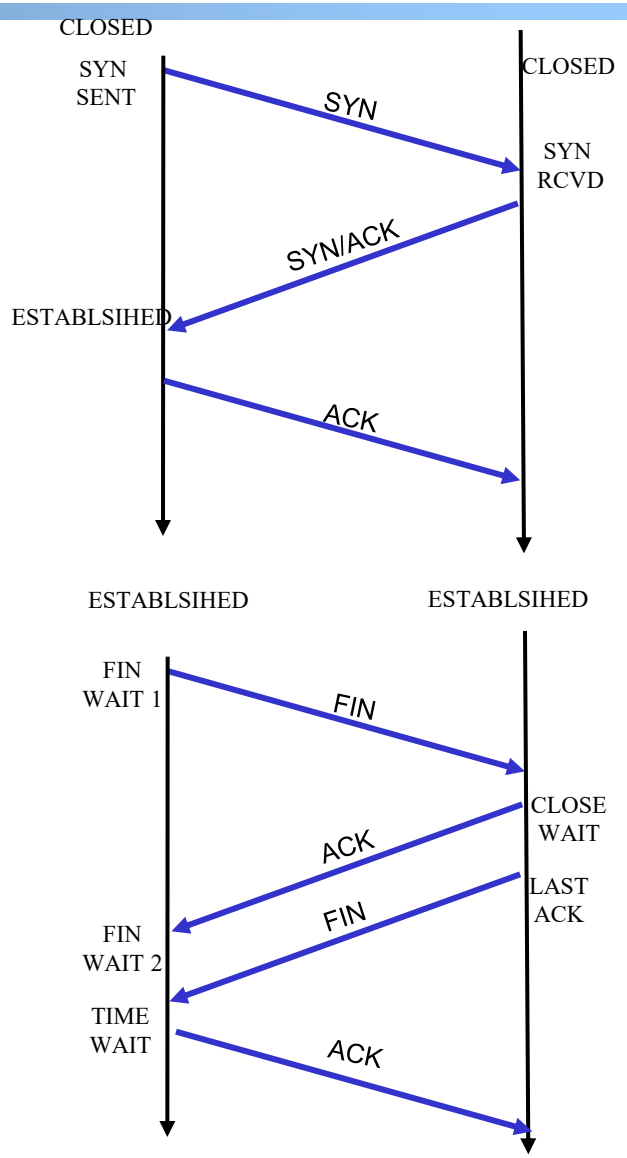

CLOSED
SYN SENT
SYN
CLOSED LISTEN
SYN RCVD
SYN/ACK
ESTABLSIHED
ACK
ESTABLSIHED


ESTABLSIHED
ESTABLSIHED
FIN WAIT 1
FIN
CLOSE WAIT
ACK
LAST ACK
FIN
FIN WAIT 2
TIME WAIT
ACK

TCP Connection Management

TCP lifecycle: init SYN/FIN

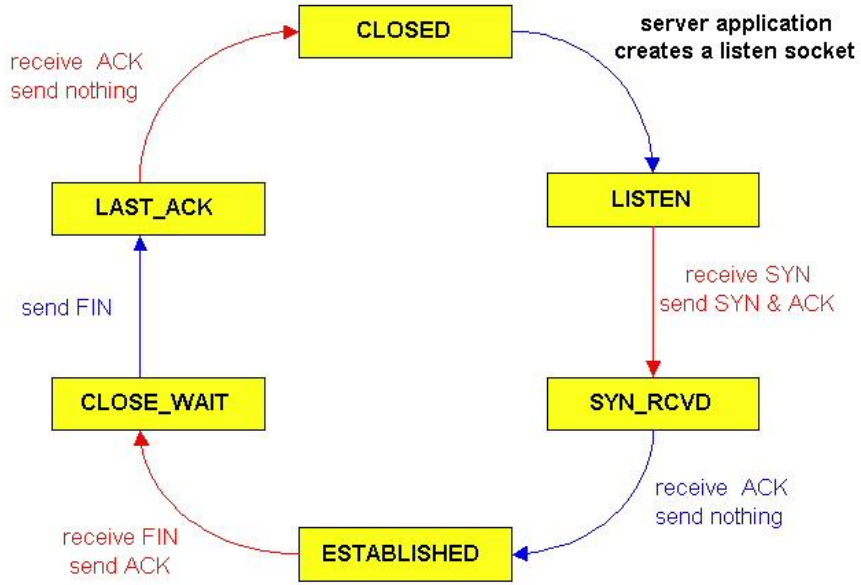


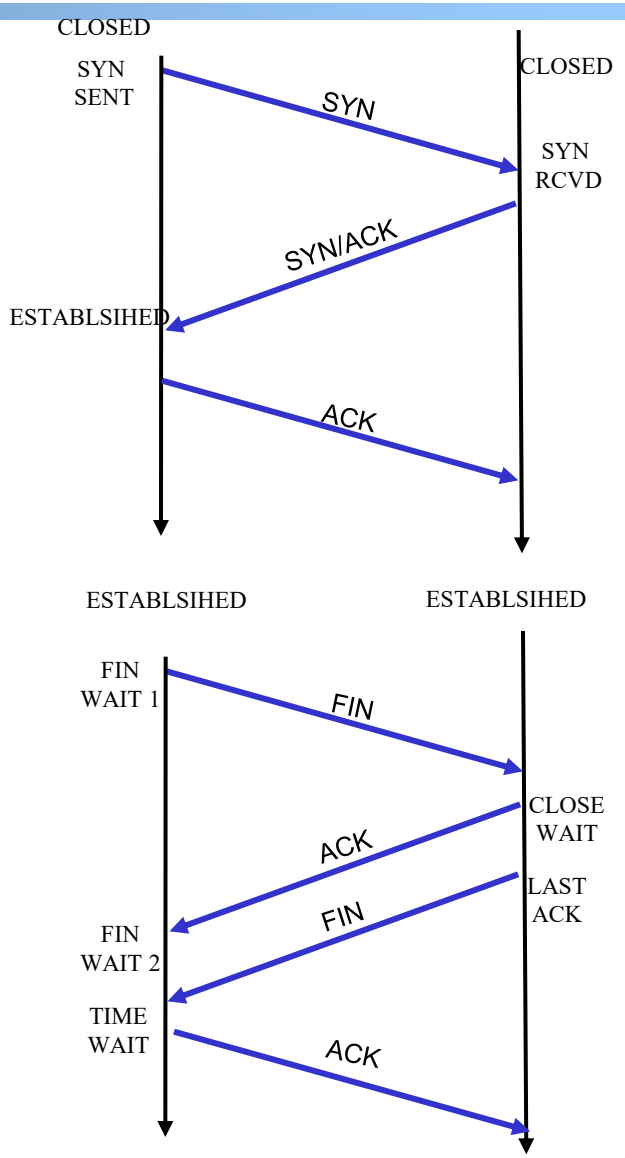


http://dsd.lbl.gov/TCP-tuning/ip-sysctl-2.6.txt

TCP Connection Management

TCP lifecycle: wait for SYN/FIN





A Summary of Questions

- Basic structure: sliding window protocols
- How to determine the "right" parameters?
  - ✓ timeout: mean + variation
  - ○ sliding window size?