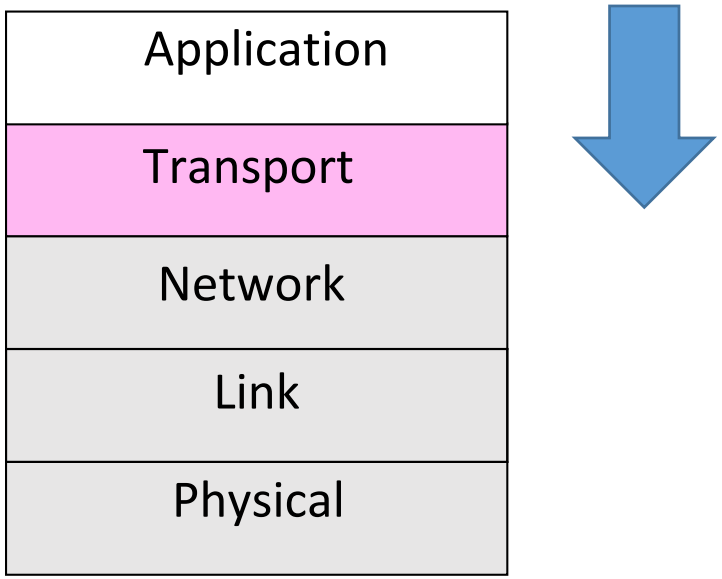
Transport Layer (TCP/UDP)

Part B

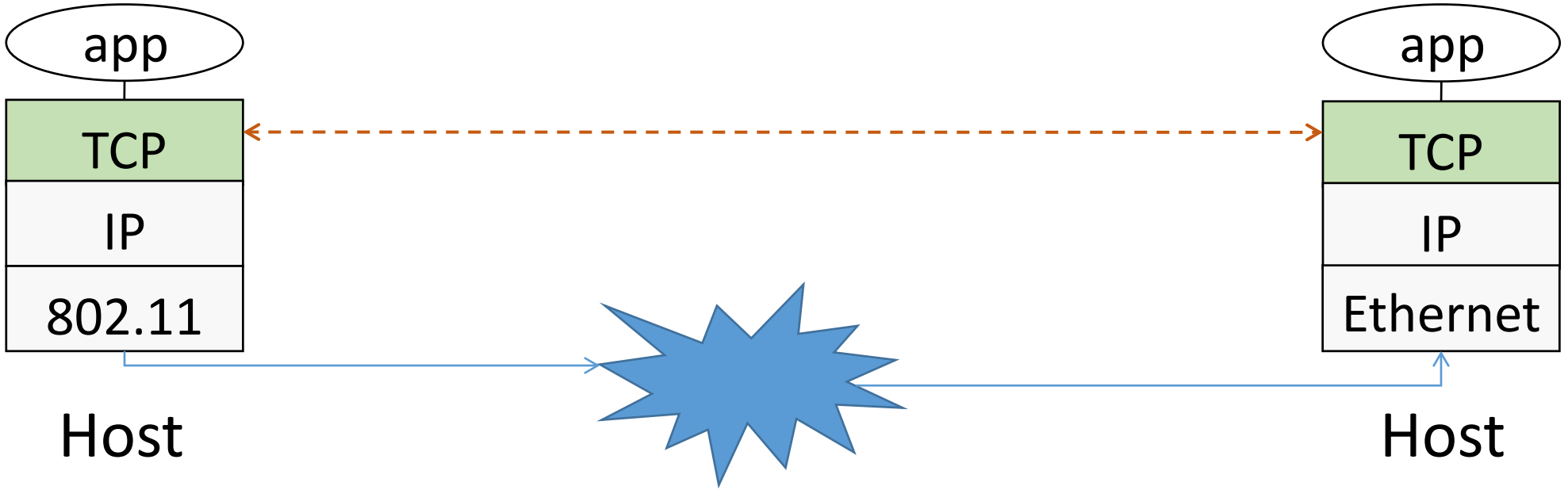
Where we are in the Course

- Moving down to the Transport Layer!

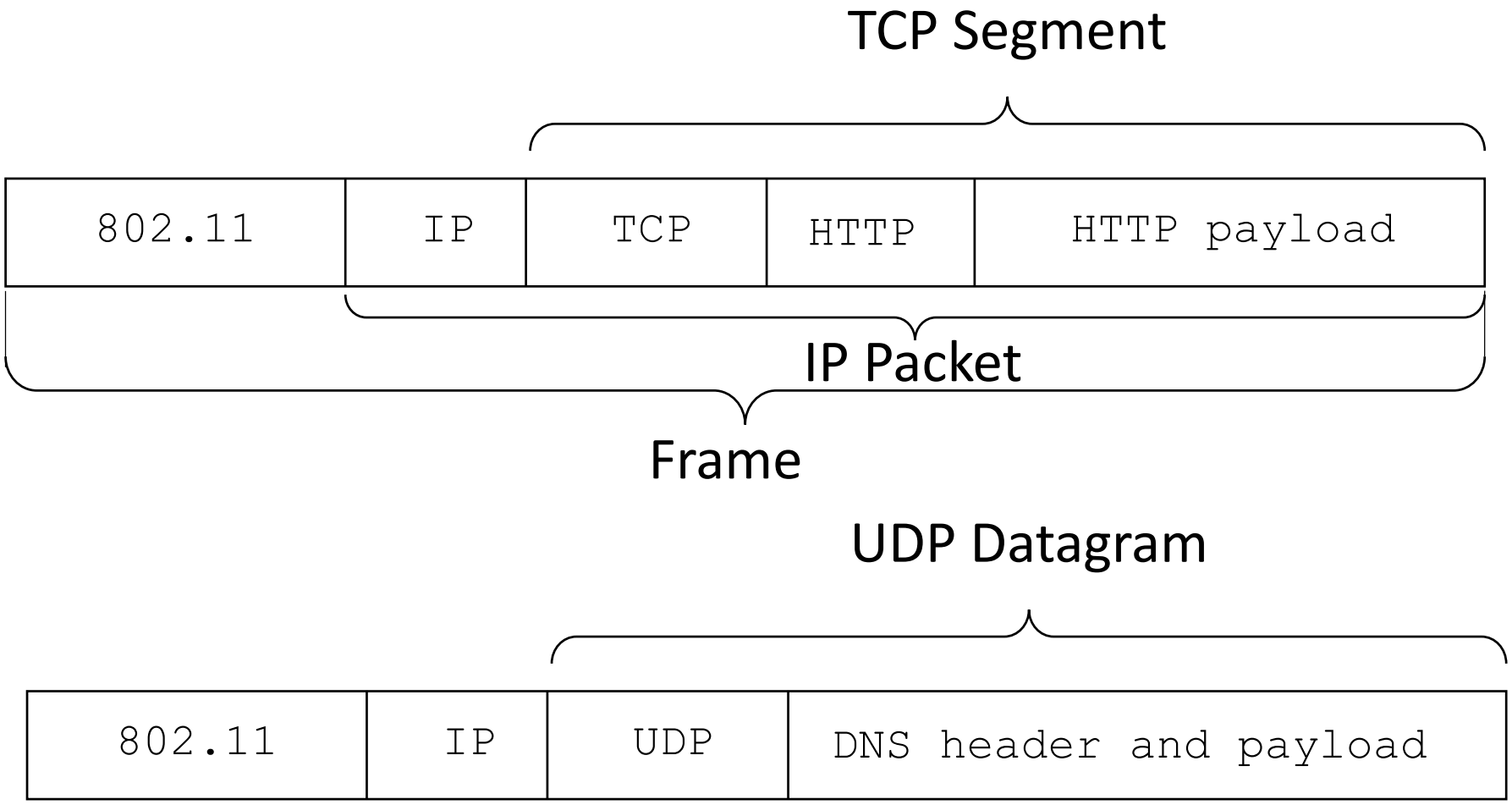


The Transport Layer

- The transport layer provides *end-to-end* connectivity
- To the transport layer, its payload is just bytes



Encapsulation



Transport Layer Services

- Provide different kinds of data delivery across the network to applications

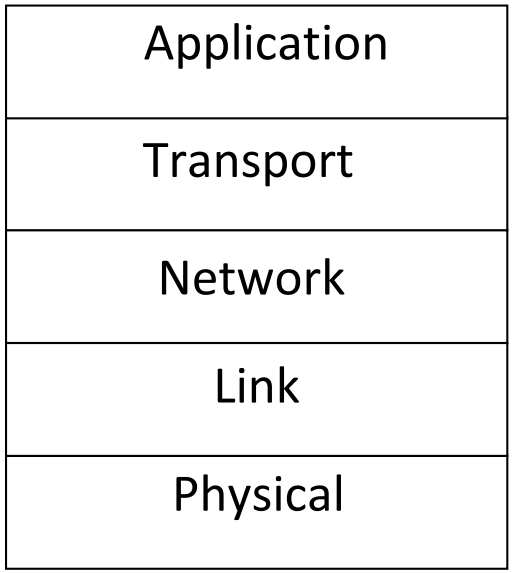
	Unreliable	**Reliable**
Packets	Datagrams (UDP)	
Bytestream		Streams (TCP)

- *Could there be protocols in the two empty boxes?*

(Pre-TCP)
Reliability and Retransmissions

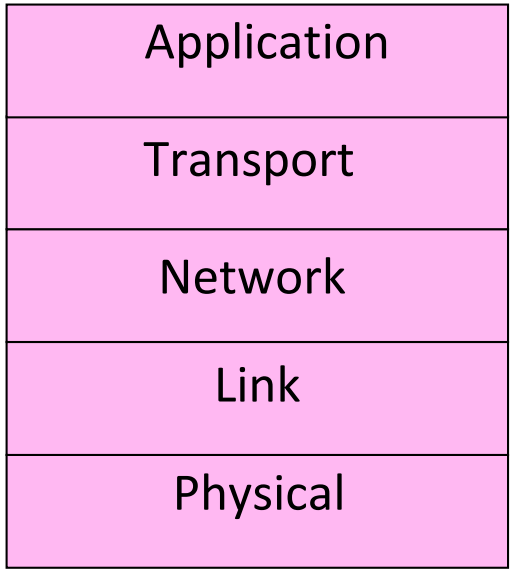
Context on Reliability

- Where in the stack should we place reliability?



Context on Reliability (2)

- Everywhere! It is a key issue
  - Different layers contribute differently



Recover actions
(correctness)

Mask errors
(performance optimization)

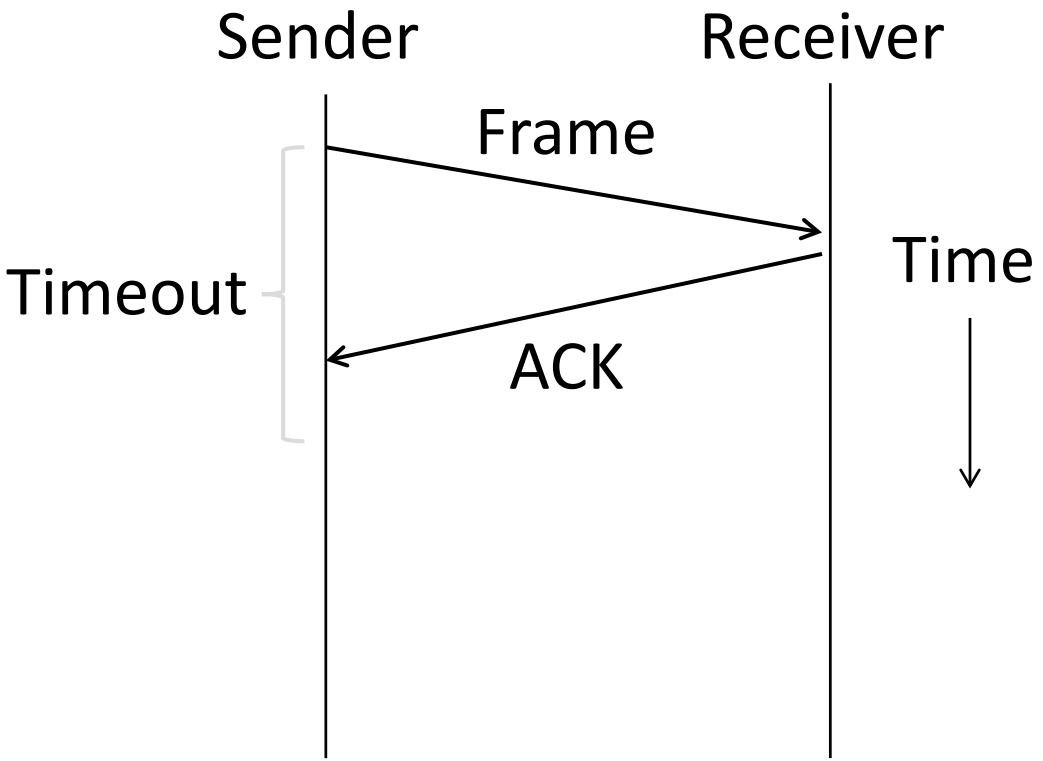
ARQ (Automatic Repeat reQuest)

- ARQ often used when errors are common or must be corrected
  - E.g., WiFi (common) and TCP (must correct)

- Rules at sender and receiver:
  - Receiver automatically acknowledges correct frames with an ACK
    - *positive acknowledgements*
  - Sender automatically resends after a timeout
  - Keep re-sending until an ACK is received

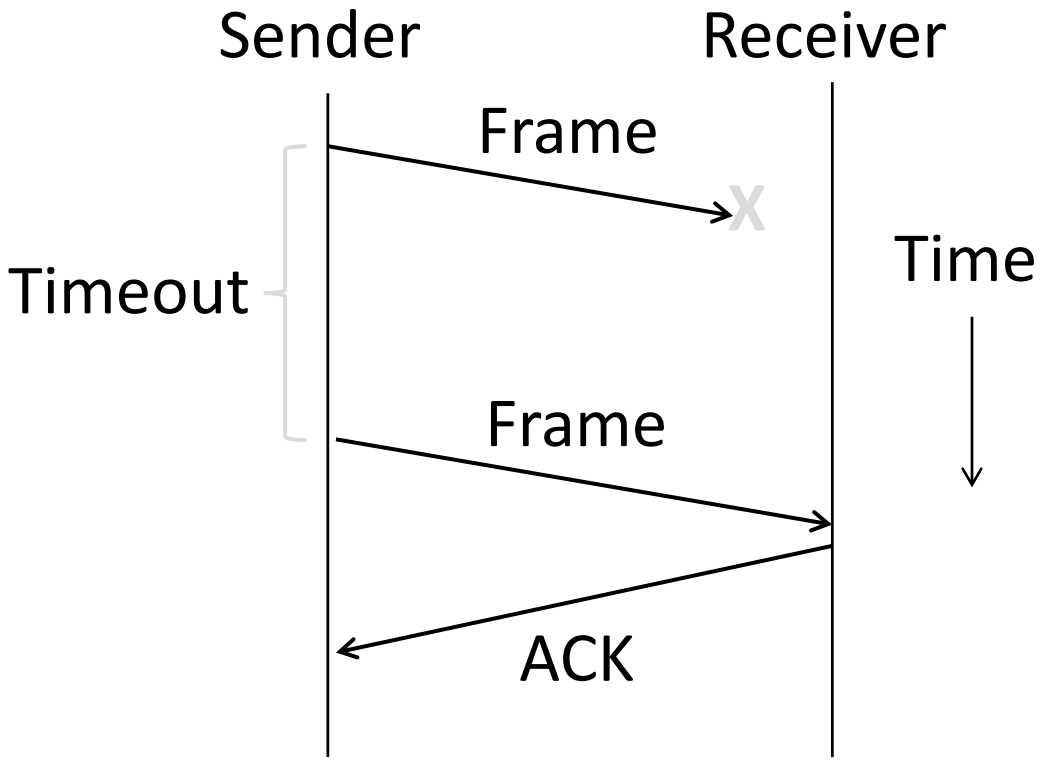
ARQ

- Normal operation (no loss)



ARQ

- Loss and retransmission



So What's Tricky About ARQ?

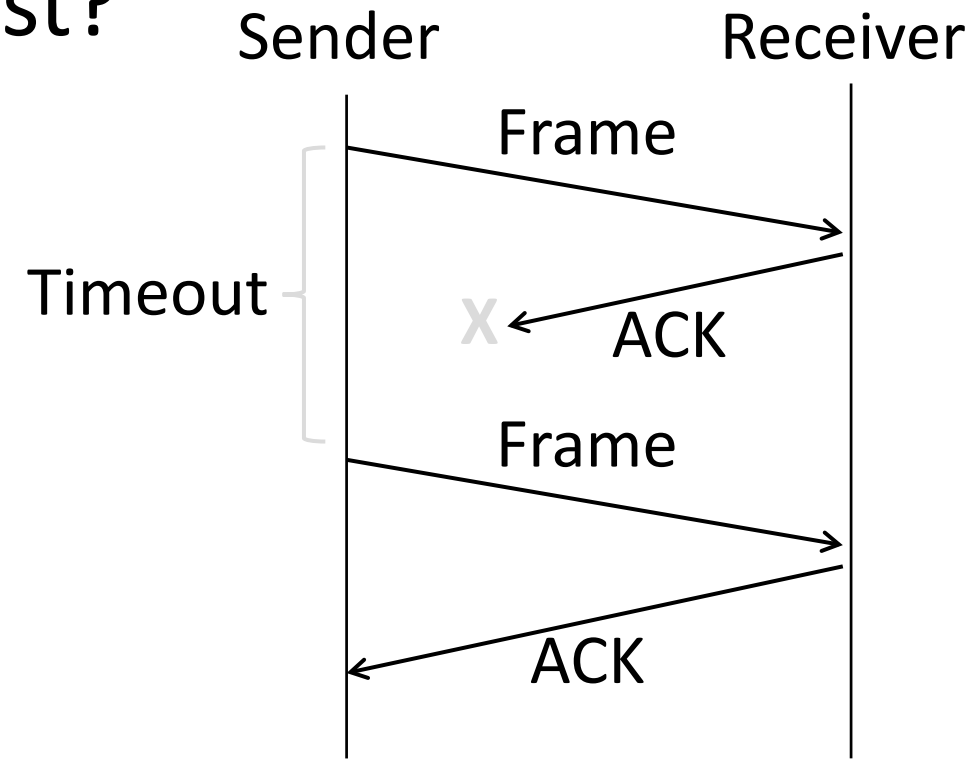
- Two non-trivial issues:
  - How long to set the timeout?
  - How to avoid accepting duplicate frames as new frames
- Want performance in the common case and correctness always

Timeouts

- Timeout should be:
  - Not too big (link goes idle)
  - Not too small (spurious resend)
- Fairly easy on a LAN
  - Clear worst case, little variation
- Fairly difficult over the Internet
  - Much variation, no obvious bound
  - We'll revisit this with TCP (later)

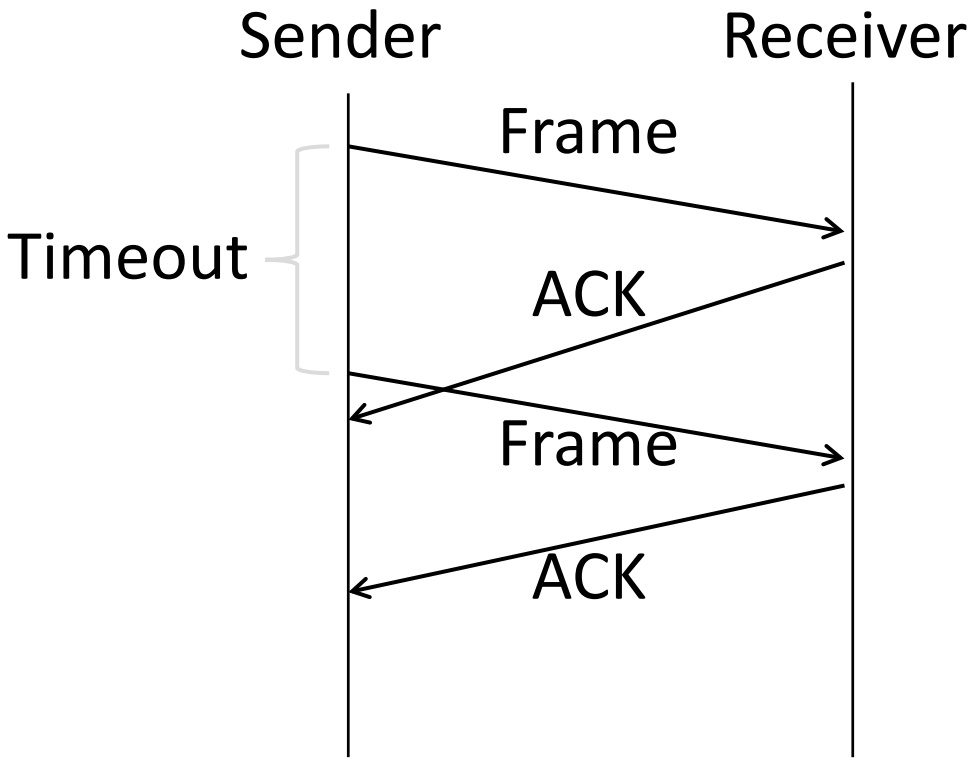
Duplicates

- What happens if an ACK is lost?



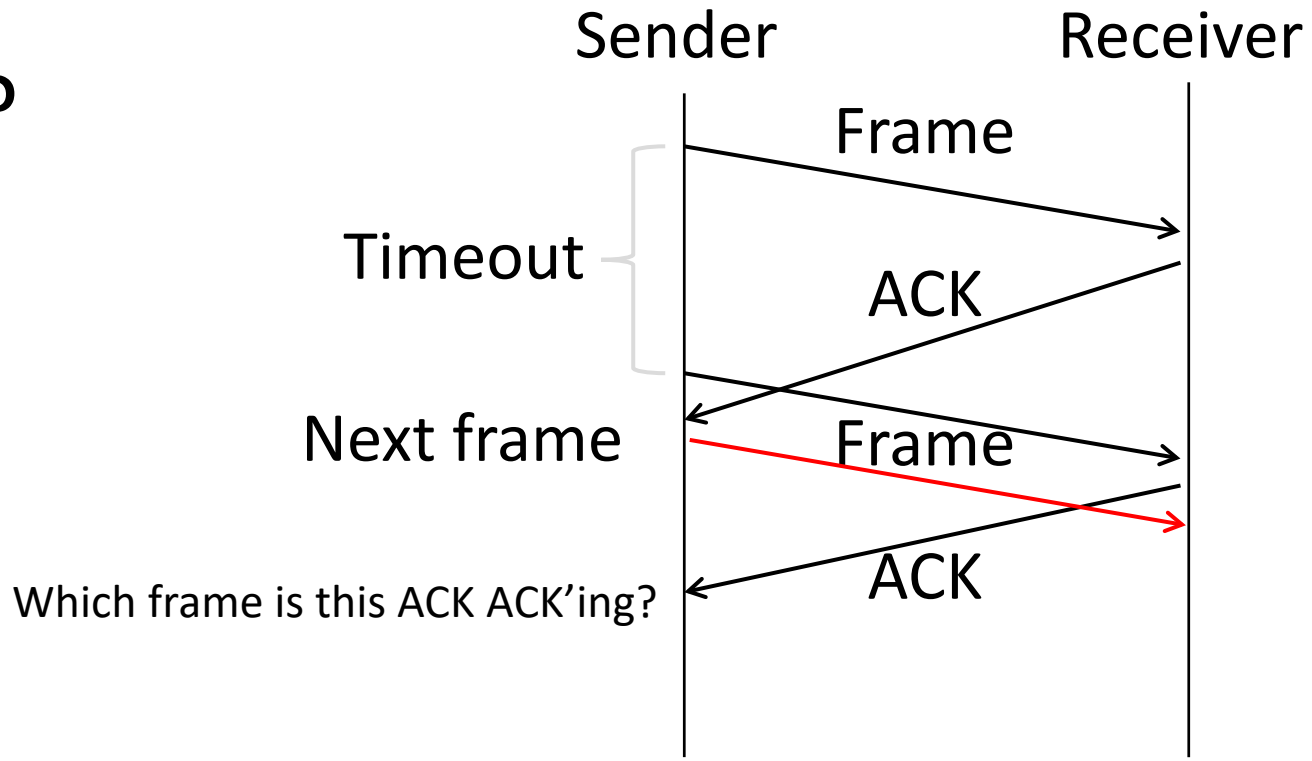
Duplicates

- What happens if the timeout is early?
  - (ACK is delayed)



Duplicates

- What happens if the timeout is early?



Sequence Numbers

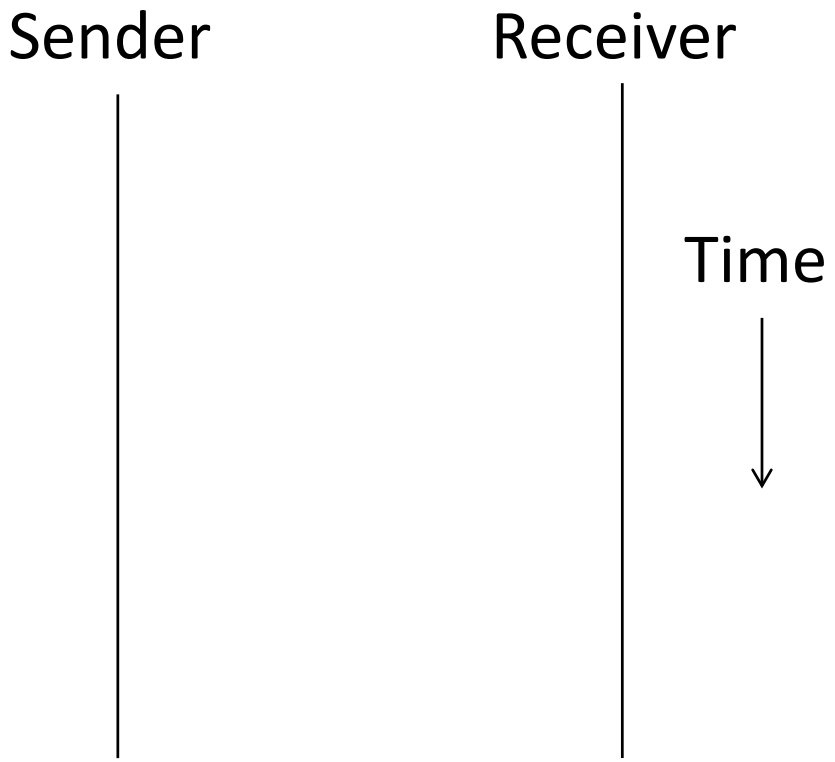
- For correctness, frames and ACKs must both carry names
  - Names must be unique (over some scope) so that we can match transmission and ACK
  - Random numbers over a large range could suffice
- Sequence numbers are a special, useful, case
  - Consecutive integers
  - Allow receiver to detect that a packet was sent but hasn't arrived
    - If I get 1, 2, 3, 5, 6
      - What do I know about the packet with sequence number 4?
      - What do I know about the packet with sequence number 7?

Sequence Numbers

- Sequence number is included in header
- Fixed number of bits allocated => finite number of distinct sequence numbers
  - Why does that matter?
- At an extreme, to distinguish the current frame from the next one, a single bit (two numbers) is sufficient
  - Called Stop-and-Wait protocol
- *In general, the number of packets that can be in flight is limited to half the range of the sequence numbers*

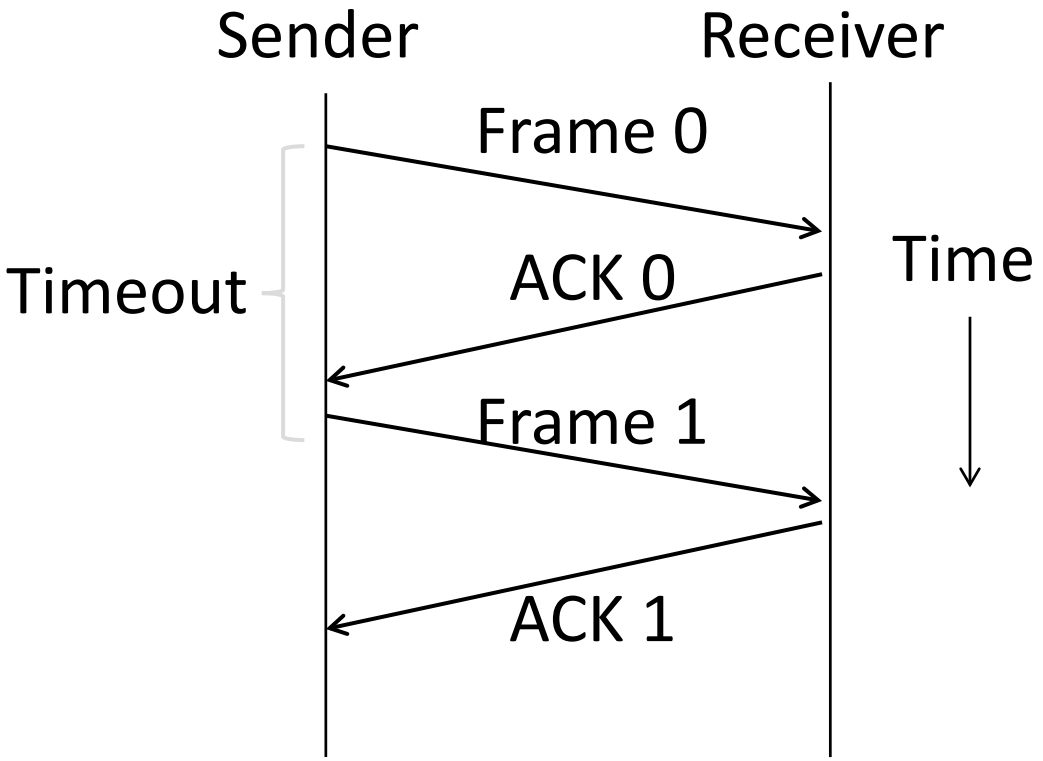
Stop-and-Wait

- In the normal case:



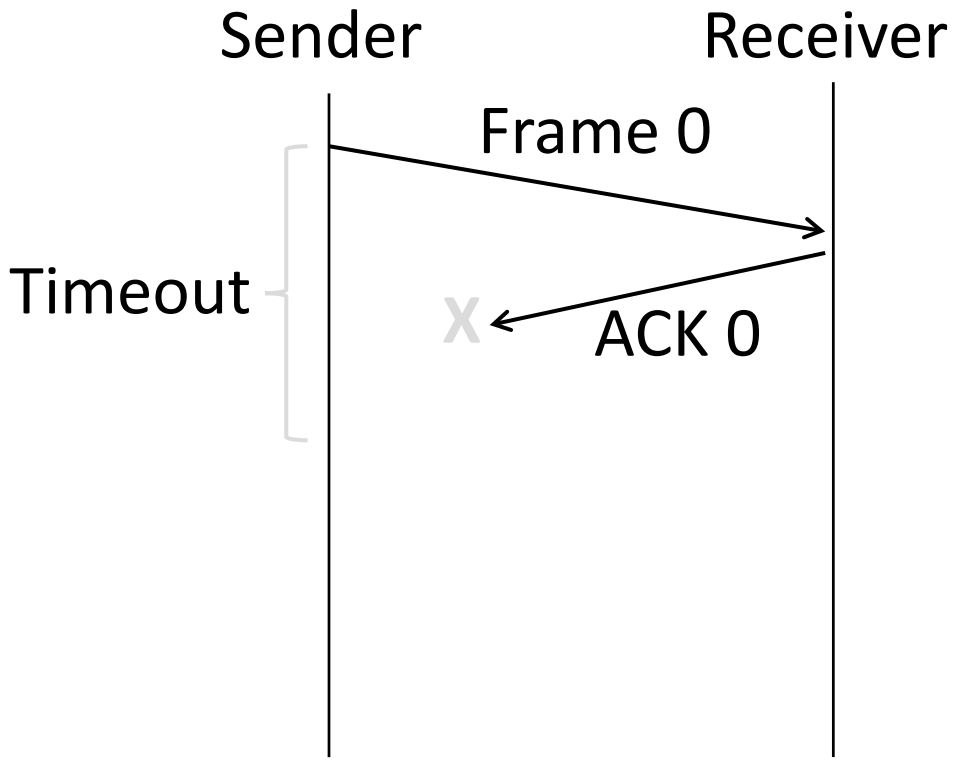
Stop-and-Wait (2)

- In the normal case:



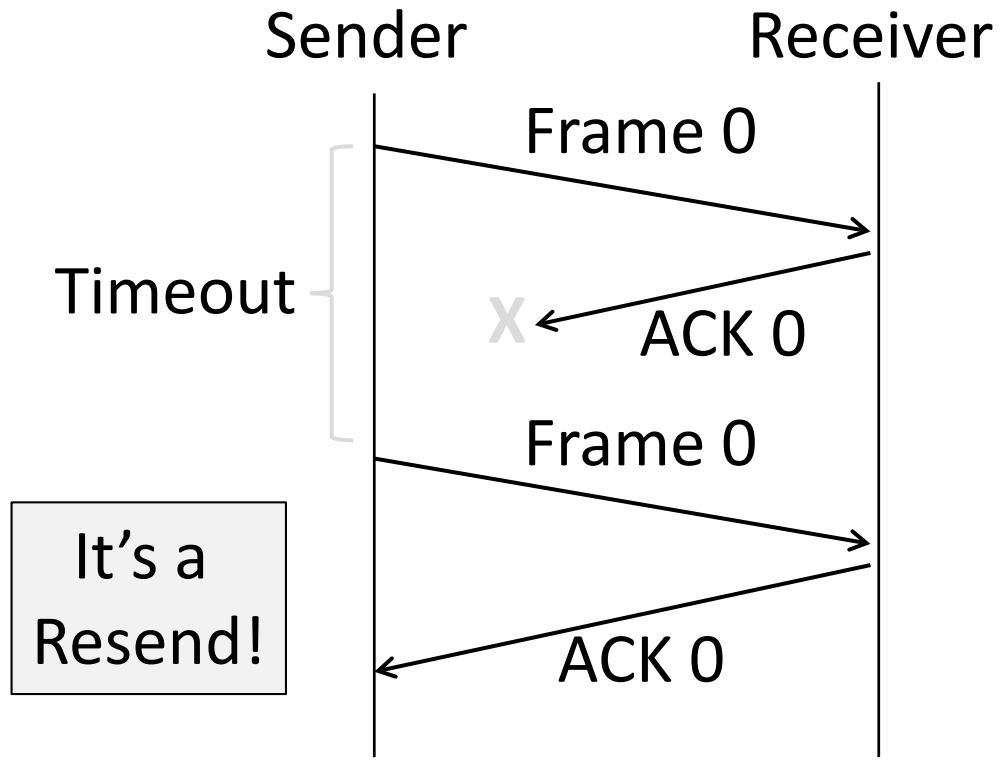
Stop-and-Wait (3)

- With ACK loss:



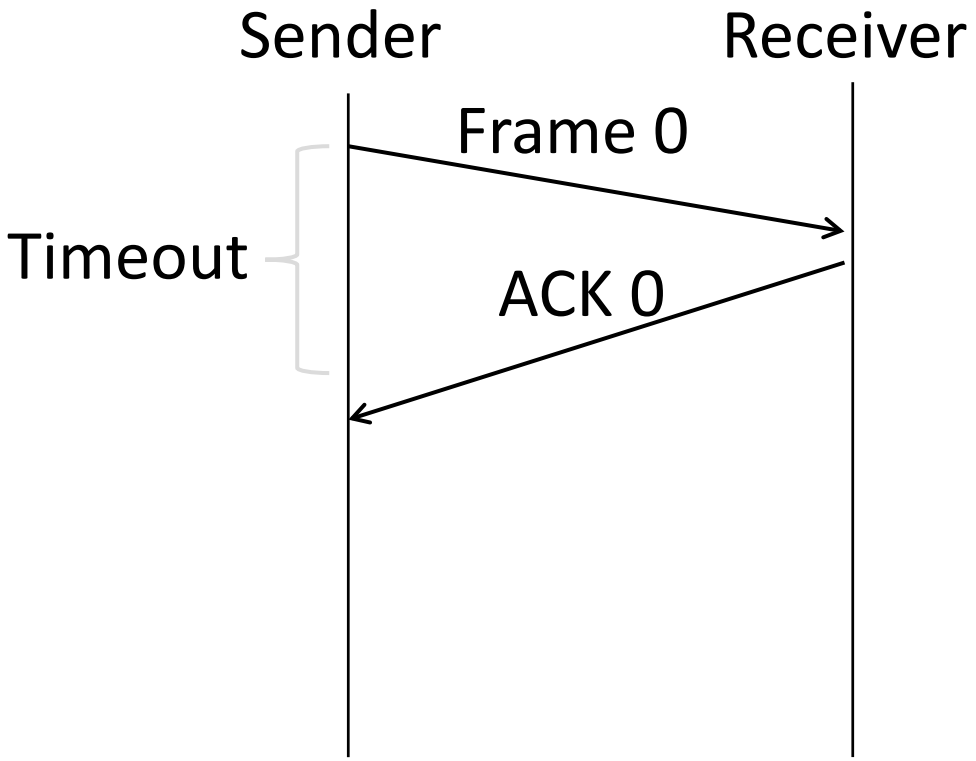
Stop-and-Wait (4)

- With ACK loss:



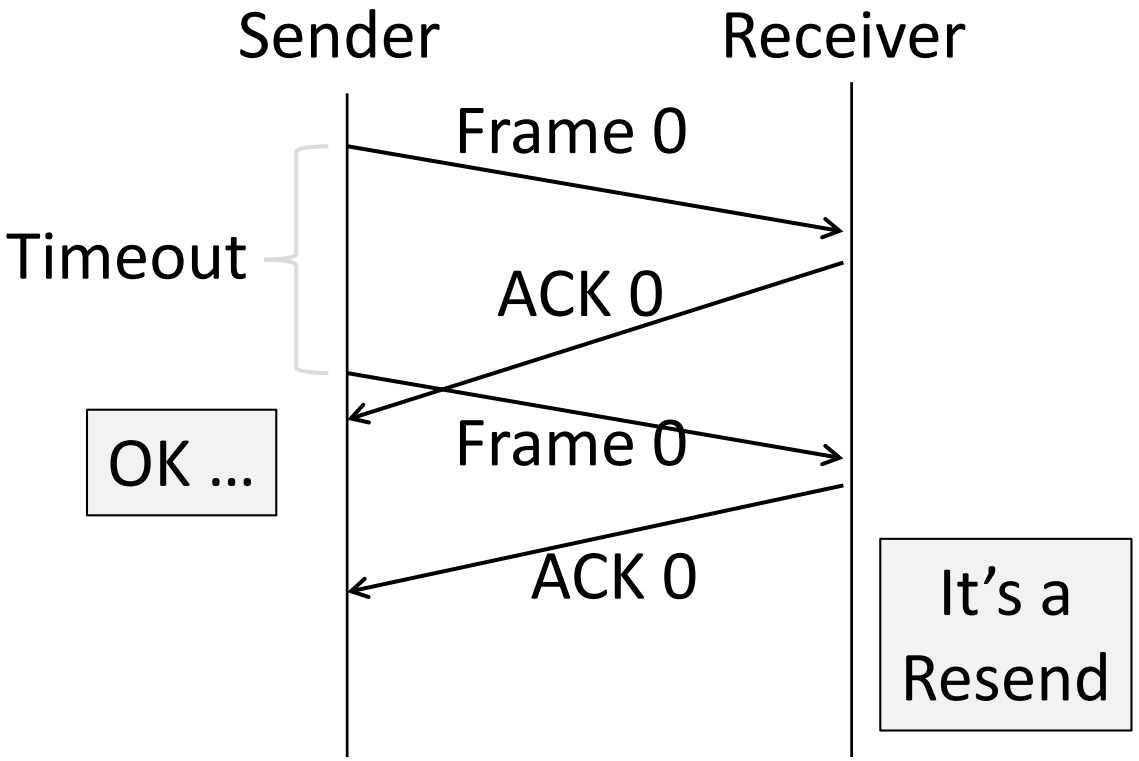
Stop-and-Wait (5)

- With early timeout:



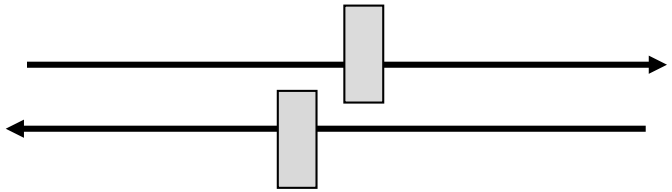
Stop-and-Wait (6)

- With early timeout:



Limitation of Stop-and-Wait

- It allows only a single frame to be outstanding from the sender:
  - Good for LAN, not efficient for high latency communication



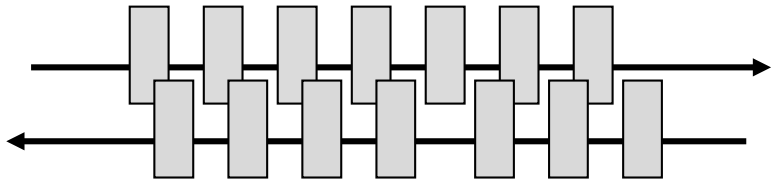
- Ex: R=1 Mbps, D = 50 ms
  - How many frames/sec? If R=1000 Mbps?

Utilization of link

- Let B be the link bandwidth
  - bits/second
- Let D be the link delay (round-trip latency)
  - seconds
- Then BD is the bandwidth-delay product
  - B bits/second * D seconds = BD bits
- BD bits / P bits/packet is the number of unackowledged packets if the sender just sends as fast as possible
  - It's the number of unacknowledged, sent packets required to saturate the link

Sliding Window

- Generalization of stop-and-wait
  - Allows W frames to be outstanding
  - Can send W frames per *round trip time* (=D)

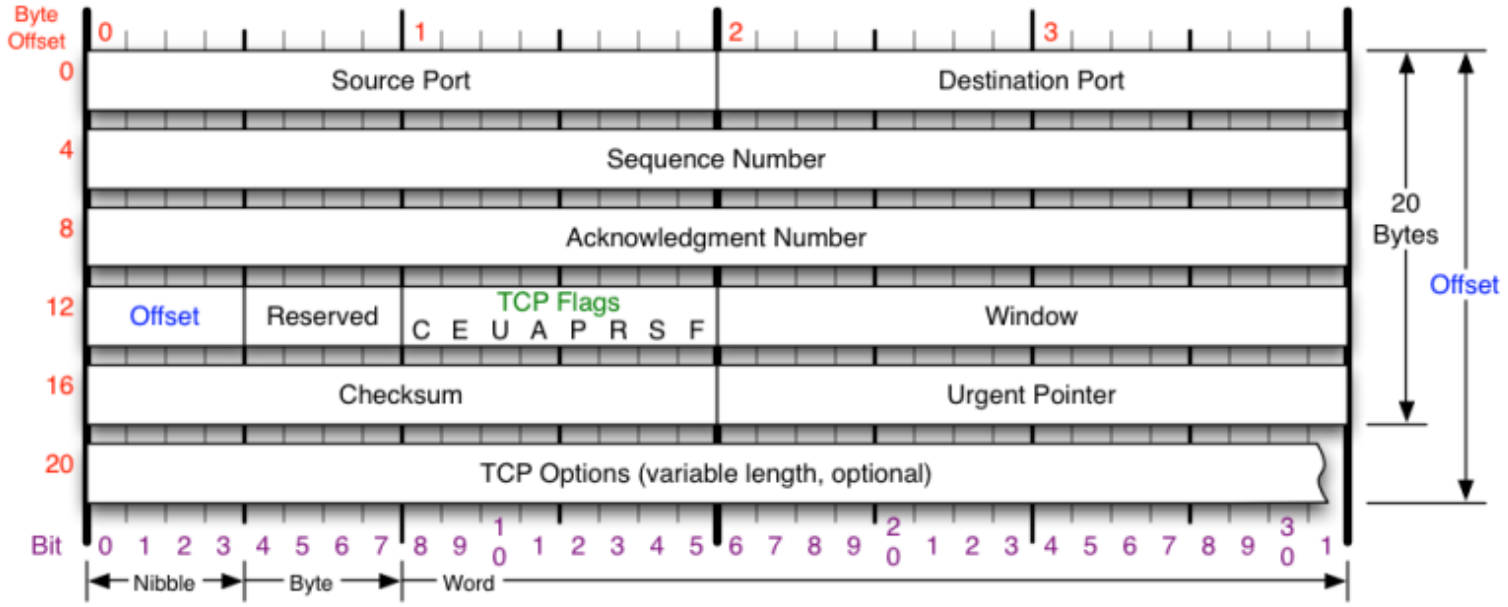


  - Various options for numbering frames/ACKs and handling loss
    - Will look at along with

TCP

From https://nmap.org/book/tcpip-ref.html

TCP Header



TCP Flags

C E U A P R S F

Congestion Window
C 0x80 Reduced (CWR)
E 0x40 ECN Echo (ECE)
U 0x20 Urgent
A 0x10 Ack
P 0x08 Push
R 0x04 Reset
S 0x02 Syn
F 0x01 Fin

Congestion Notification

ECN (Explicit Congestion Notification). See RFC 3168 for full details, valid states below.

Packet State	DSB	ECN bits
Syn	0 0	1 1
Syn-Ack	0 0	0 1
Ack	0 1	0 0
No Congestion	0 1	0 0
No Congestion	1 0	0 0
Congestion	1 1	0 0
Receiver Response	1 1	0 1
Sender Response	1 1	1 1

TCP Options

0 End of Options List
1 No Operation (NOP, Pad)
2 Maximum segment size
3 Window Scale
4 Selective ACK ok
8 Timestamp

Checksum

Checksum of entire TCP segment and pseudo header (parts of IP header)

Offset

Number of 32-bit words in TCP header, minimum value of 5. Multiply by 4 to get byte count.

RFC 793

Please refer to RFC 793 for the complete Transmission Control Protocol (TCP) Specification.

TCP Protocol

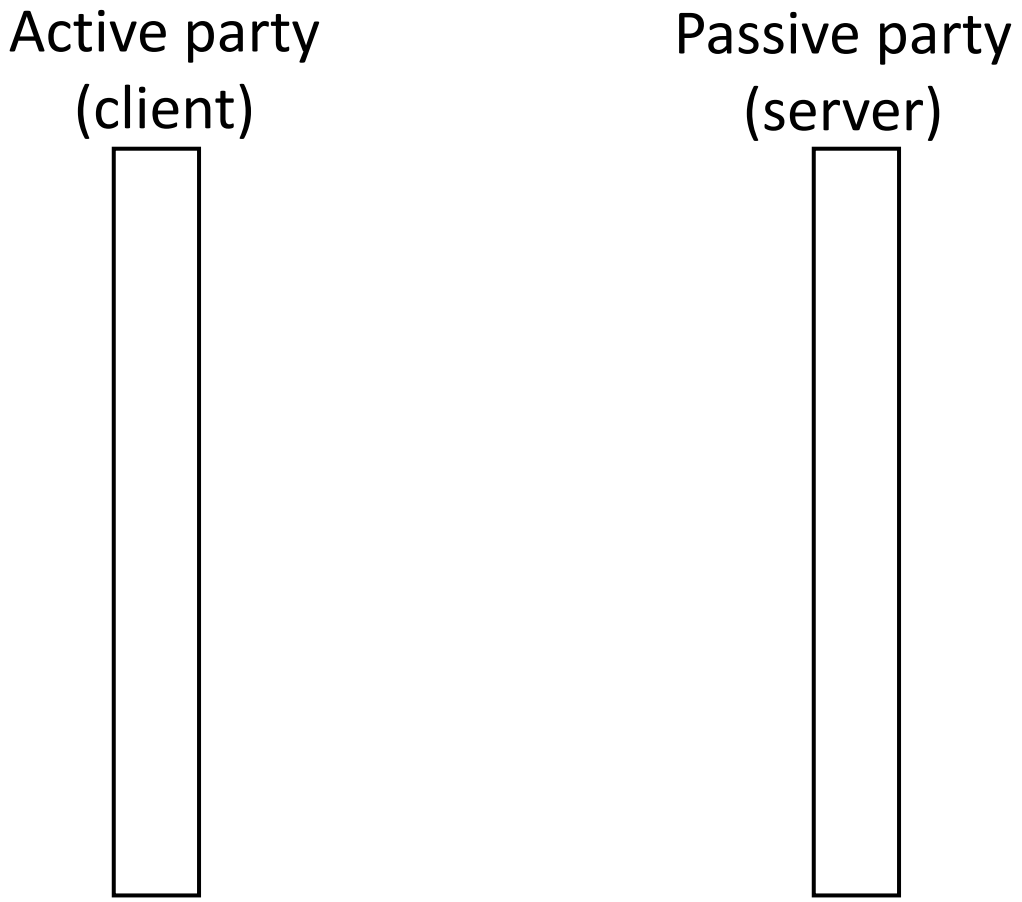
- TCP Consists of 3 primary phases:
  - Connection Establishment (Setup)
  - Sliding Windows/Flow Control
  - Connection Release (Teardown)

Connection Establishment

- Both sender and receiver must be ready before we start the transfer of data
  - Need to agree on a set of parameters
  - e.g., the Maximum Segment Size (MSS)
- This is *signaling*
  - It sets up state at the endpoints
  - Like “dialing” for a telephone call

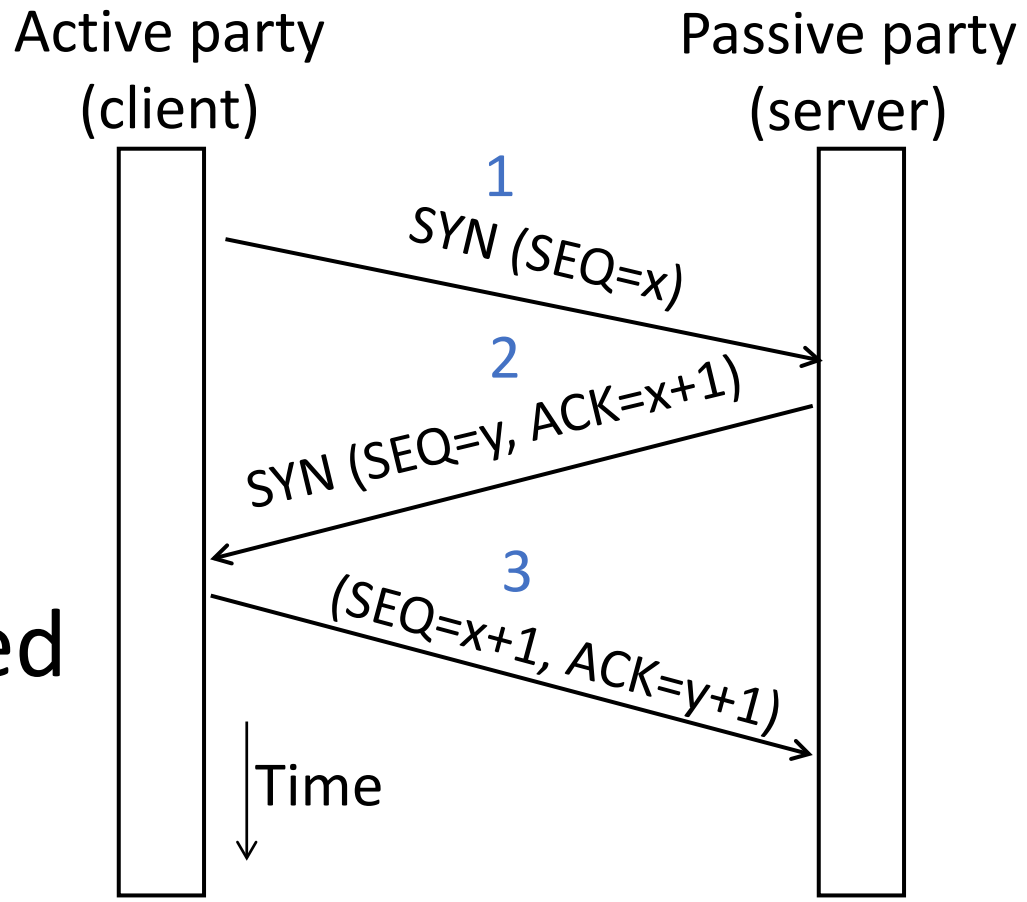
Three-Way Handshake

- Used in TCP; opens connection for data in both directions
- Each side probes the other with a fresh Initial Sequence Number (ISN)
  - Sends on a SYNchronize segment
  - Echo on an ACKnowledge segment
- Chosen to be robust even against delayed duplicates



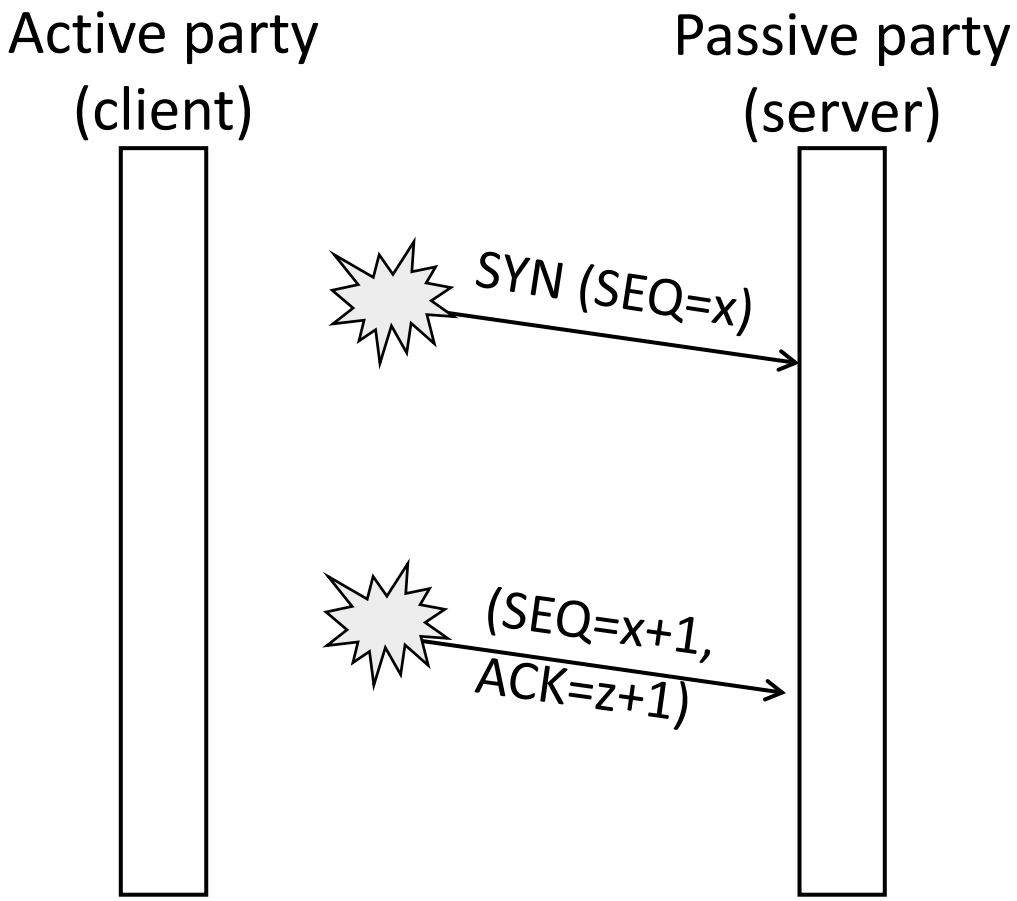
Three-Way Handshake (2)

- Three steps:
  - Client sends SYN(x)
  - Server replies with SYN(y)ACK(x+1)
  - Client replies with ACK(y+1)
  - SYNs are retransmitted if lost
- Sequence and ack numbers carried on further segments



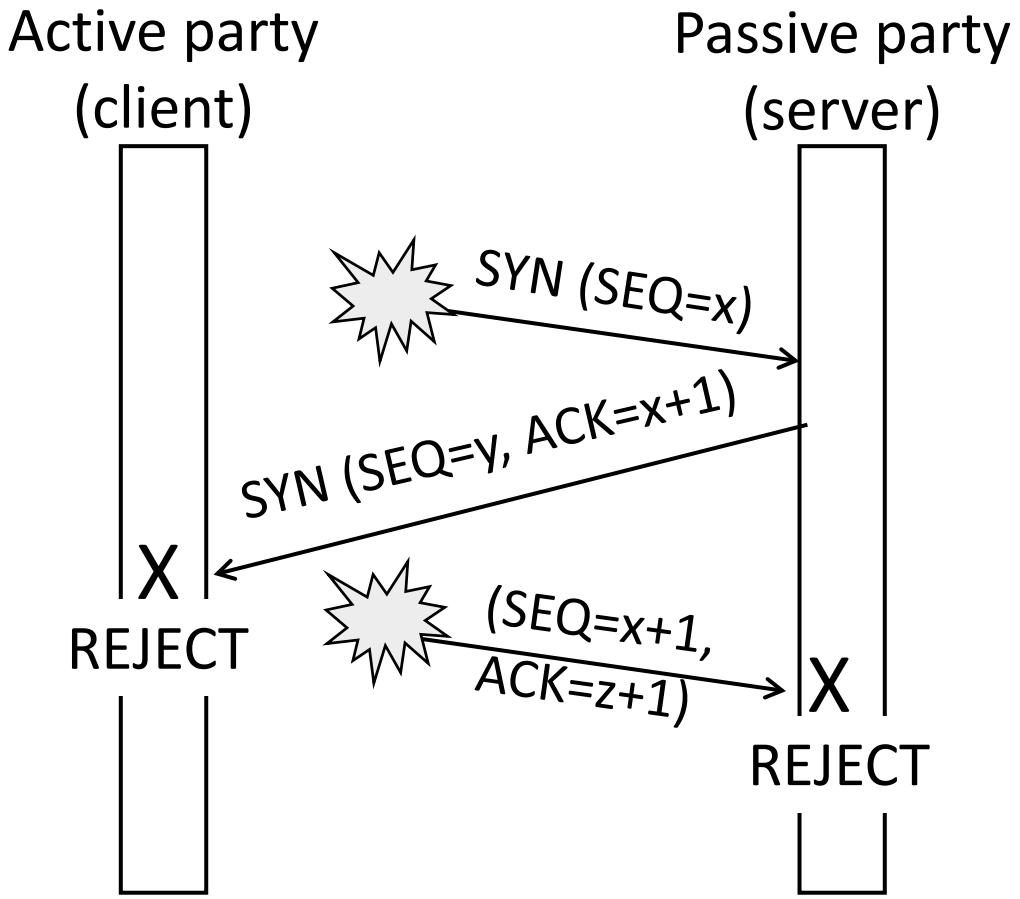
Three-Way Handshake (3)

- Suppose delayed, duplicate copies of the SYN and ACK arrive at the server!
  - Improbable, but anyhow …



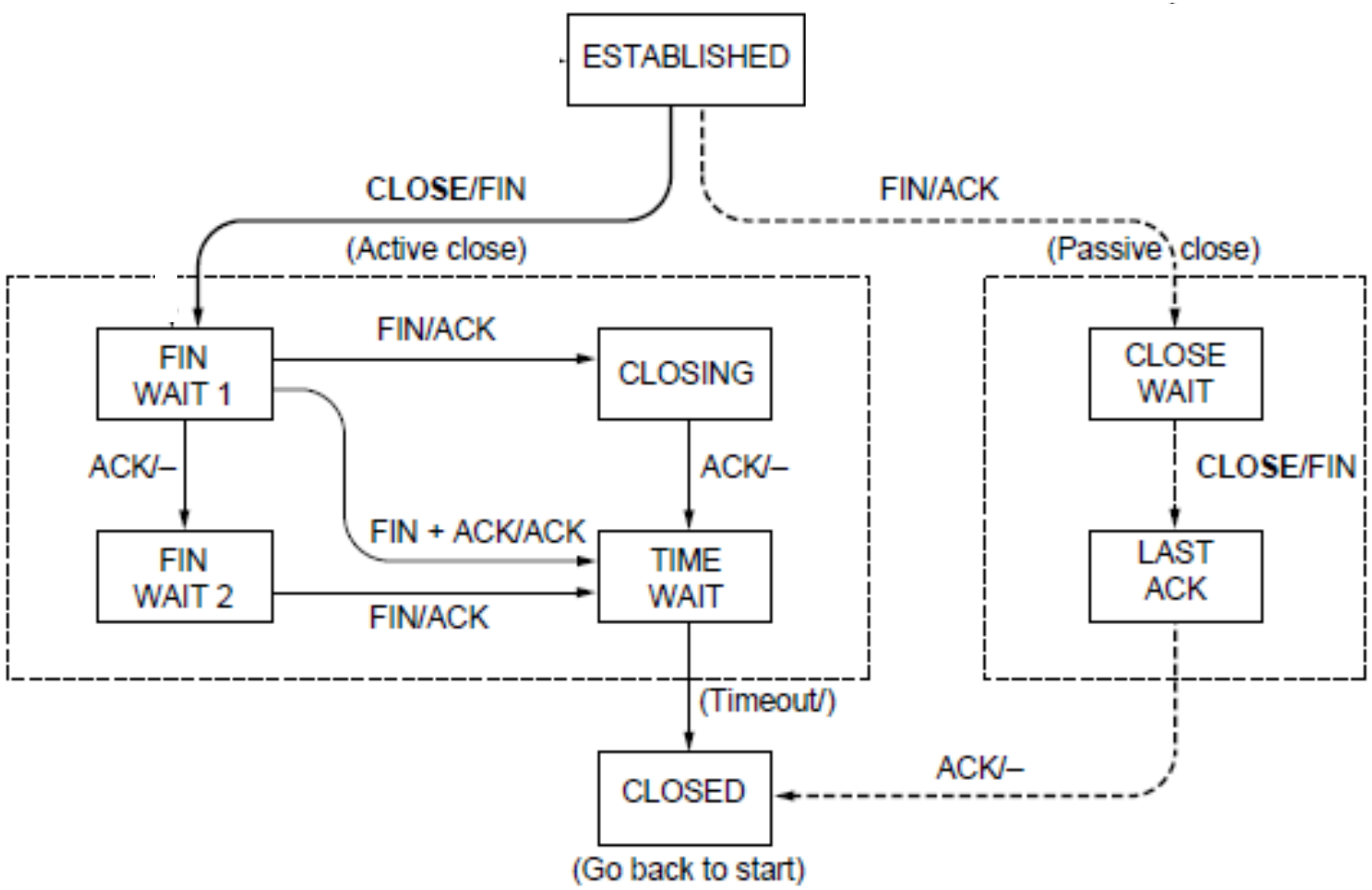
Three-Way Handshake (4)

- Suppose delayed, duplicate copies of the SYN and ACK arrive at the server!
  - Improbable, but anyhow …
- Connection will be cleanly rejected on both sides ☺



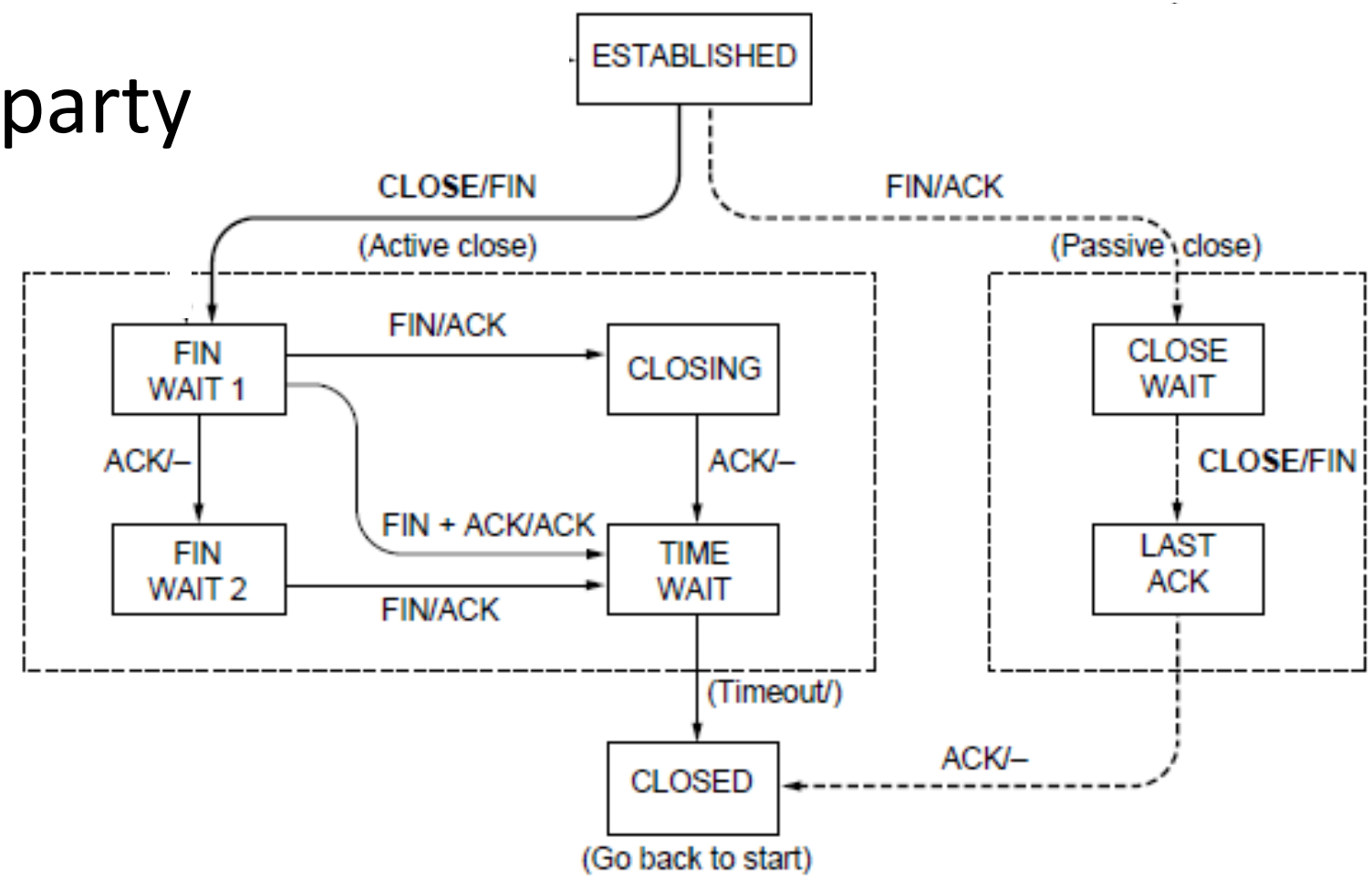
TCP Connection State Machine

Both parties run instances of this state machine



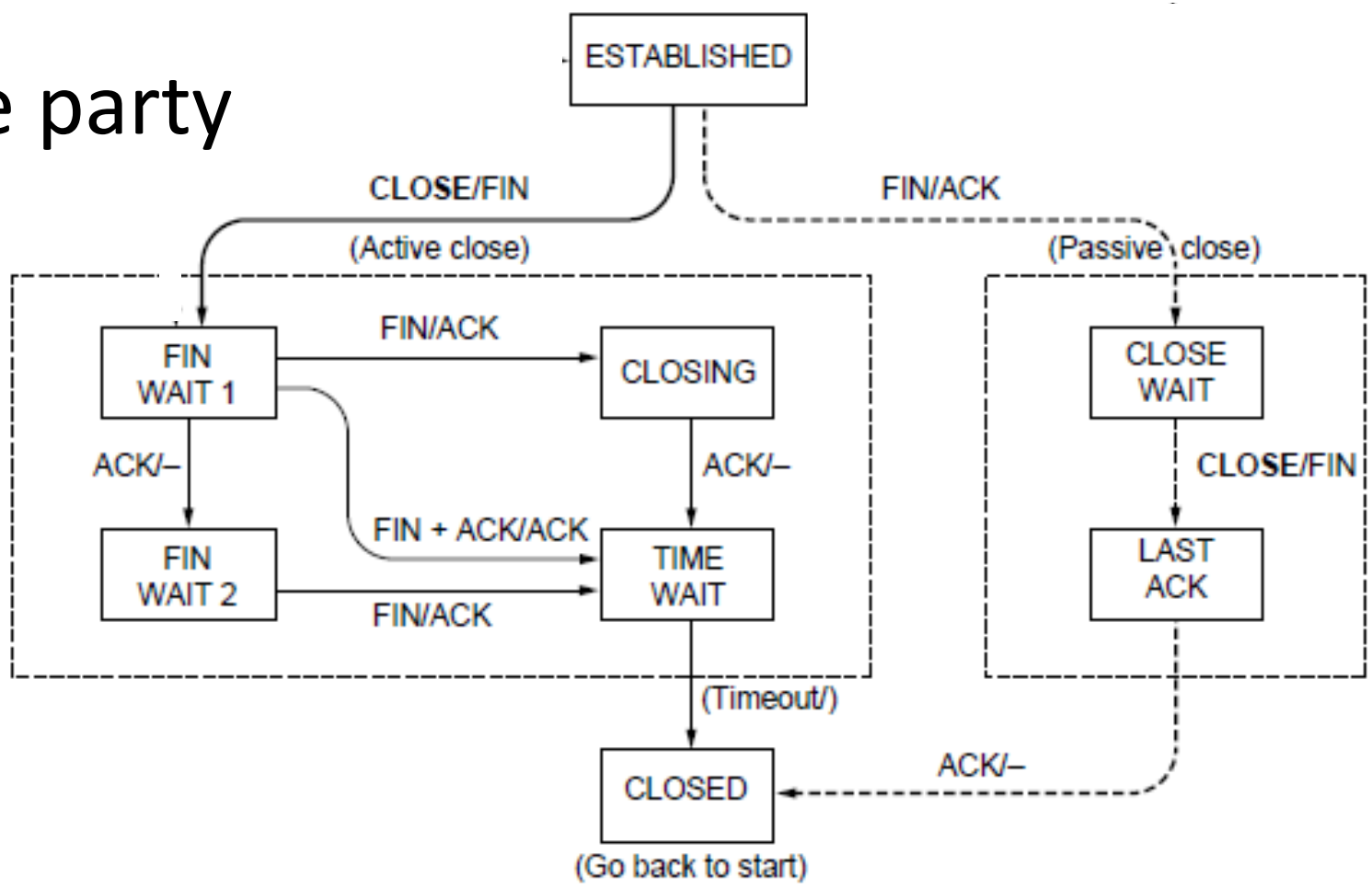
TCP Release

- Follow the active party



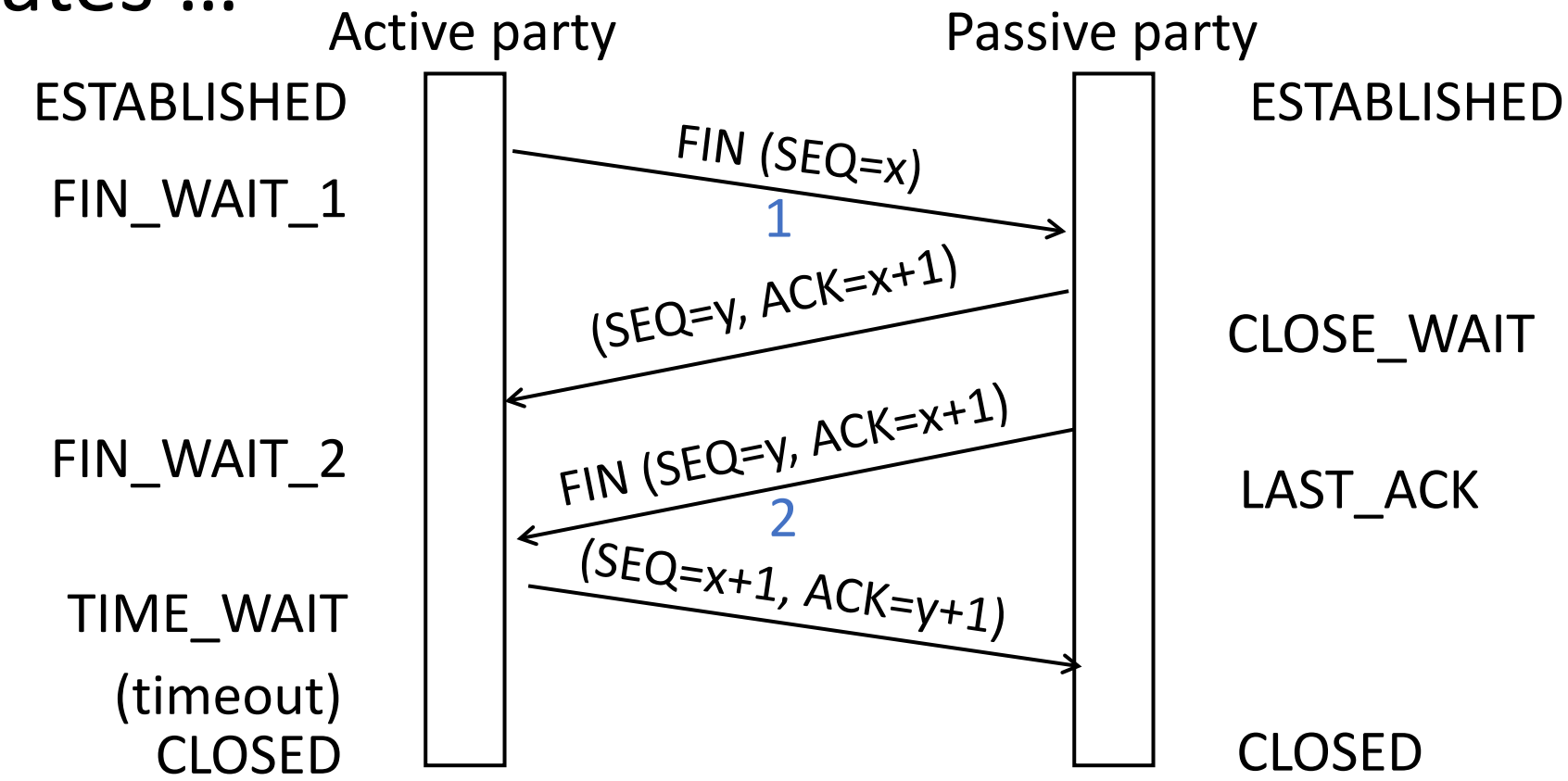
TCP Release

- Follow the passive party



TCP Release

- Again, with states …



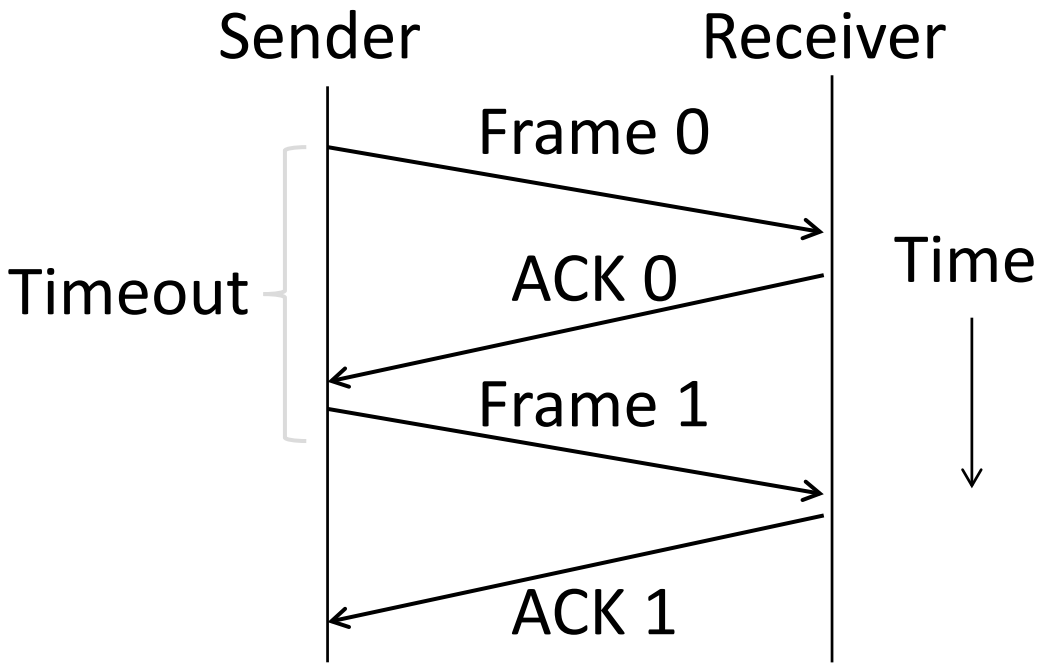
TIME_WAIT State

- Wait a long time after sending all segments and before completing the close
  - Two times the maximum segment lifetime of 60 seconds
- Why?
  - ACK might have been lost, in which case FIN will be resent for an orderly close
  - Could otherwise interfere with a subsequent connection

Flow Control

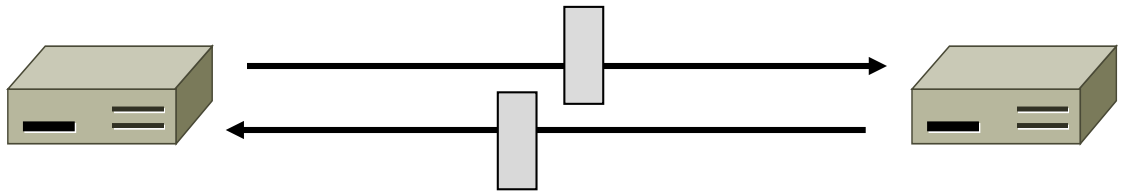
Recall

- ARQ with one message at a time is Stop-and-Wait (normal case below)



Limitation of Stop-and-Wait

- It allows only a single message to be outstanding from the sender:
  - Fine for LAN (only one frame fit)
  - Not efficient for network paths with BD >> 1 packet



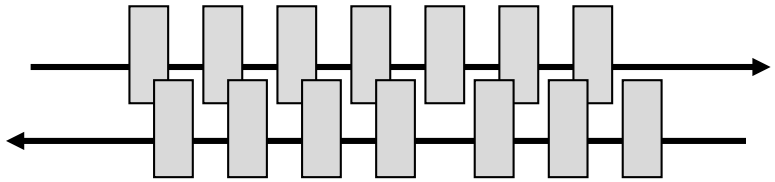
Limitation of Stop-and-Wait (2)

- Example: R=1 Mbps, D = 50 ms
  - RTT (Round Trip Time) = 2D = 100 ms
  - How many packets/sec?

  - What if R=1000 Mbps?

Sliding Window

- Generalization of stop-and-wait
  - Allows W packets to be outstanding
  - Can send W packets per RTT (=2D)



  - Pipelining improves performance
  - Need W=2BD to fill network path

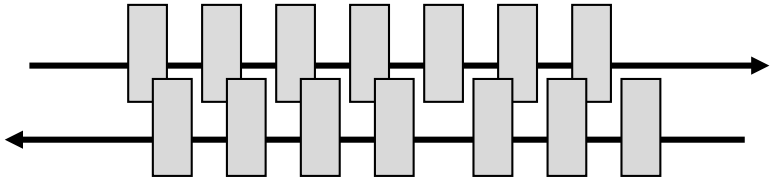
Sliding Window (2)

- What W will use the network capacity?
- Ex: R=1 Mbps, D = 50 ms

- Ex: What if R=10 Mbps?

Sliding Window (3)

- Ex: R=1 Mbps, D = 50 ms
  - 2BD = 10^6 b/sec x 100. 10^{-3} sec = 100 kbit
  - W = 2BD = 10 packets of 1250 bytes



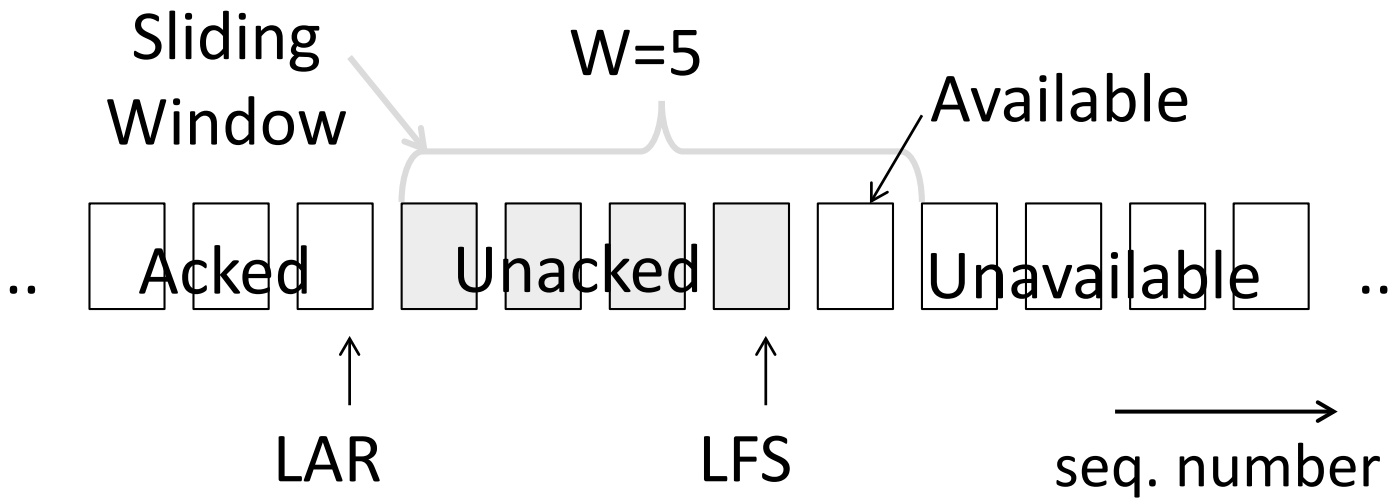
- Ex: What if R=10 Mbps?
  - 2BD = 1000 kbit
  - W = 2BD = 100 packets of 1250 bytes

Sliding Window Protocol

- Many variations, depending on how buffers, acknowledgements, and retransmissions are handled
- Go-Back-N
  - Simplest version, can be inefficient
- Selective Repeat
  - More complex, better performance

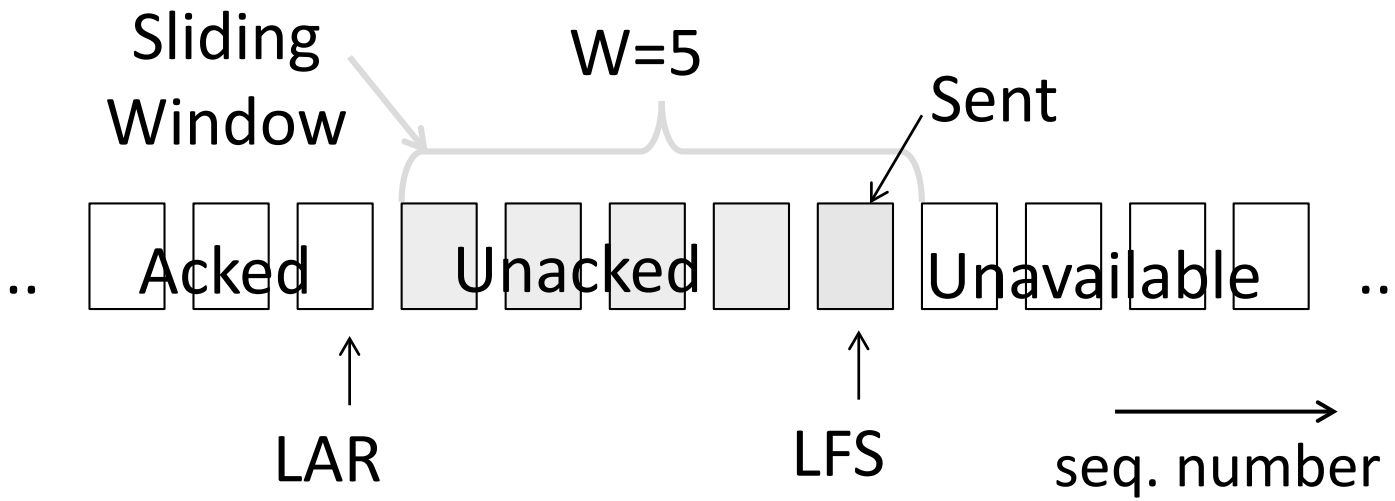
Sliding Window – Sender

- Sender buffers up to W segments until they are acknowledged
  - LFS=LAST FRAME SENT, LAR=LAST ACK REC'D
  - Sends while LFS – LAR ≤ W



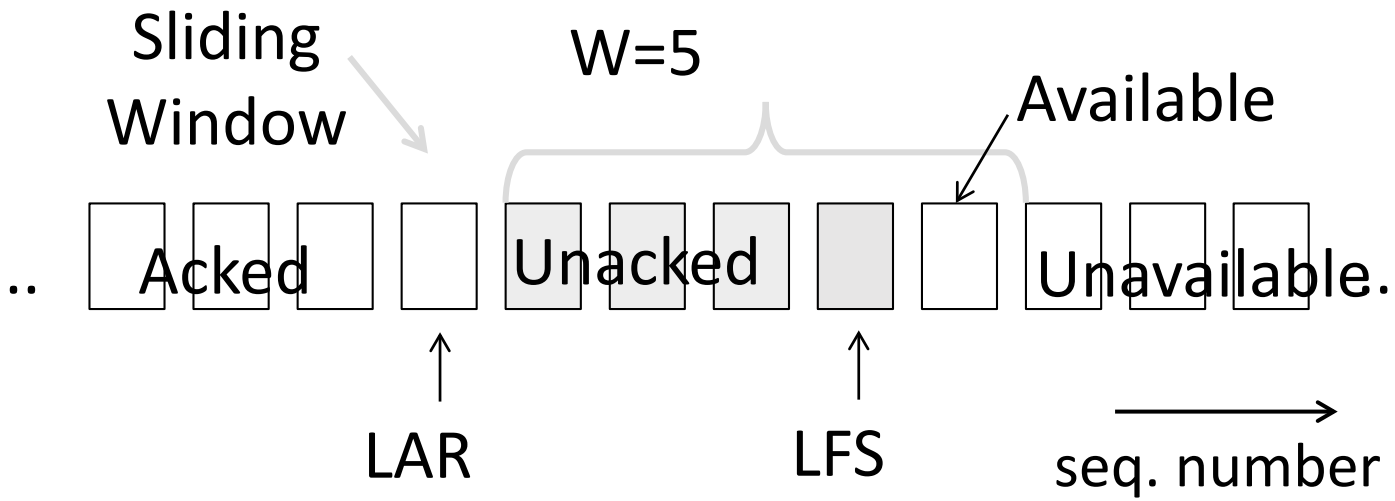
Sliding Window – Sender (2)

- Transport accepts another segment of data from the Application ...
  - Transport sends it (as LFS–LAR → 5)



Sliding Window – Sender (3)

- Next higher ACK arrives from peer…
  - Window advances, buffer is freed
  - LFS–LAR → 4 (can send one more)



Sliding Window – Go-Back-N

- Receiver keeps only a single packet buffer for the next segment
  - State variable, LAS = LAST ACK SENT
- On receive:
  - If seq. number is LAS+1, accept and pass it to app, update LAS, send ACK
  - Otherwise discard (as out of order)

Sliding Window – Selective Repeat

- Receiver passes data to app in order, and buffers out-of-order segments to reduce retransmissions
- ACK conveys highest in-order segment, plus hints about out-of-order segments
- TCP uses a selective repeat design; we'll see the details later

Sliding Window – Selective Repeat (2)

- Buffers W segments, keeps state variable LAS = LAST ACK SENT

- On receive:
  - Buffer segments [LAS+1, LAS+W]
  - Send app in-order segments from LAS+1, and update LAS
  - Send ACK for LAS regardless

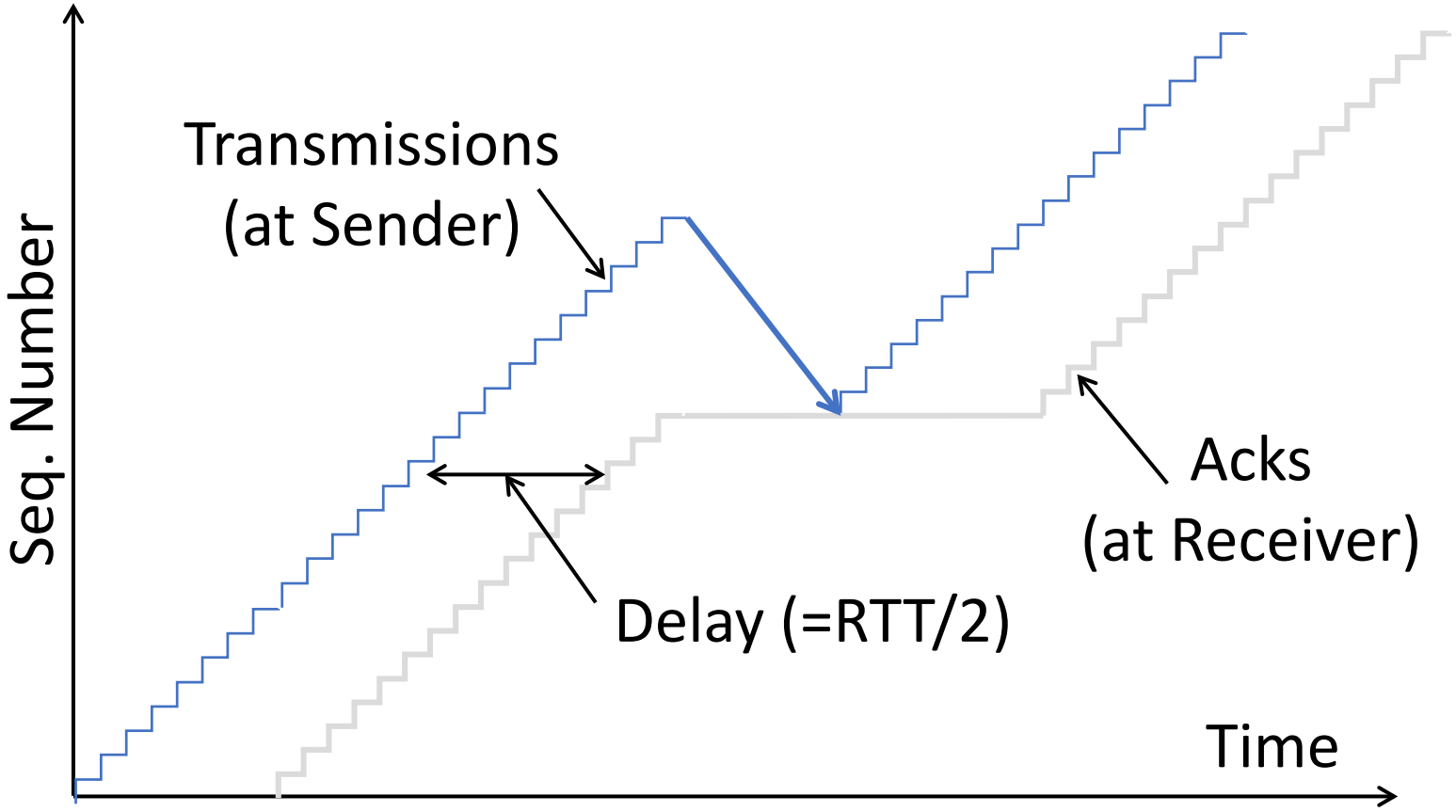
Sliding Window – Retransmissions

- Go-Back-N uses a single timer to detect losses
  - On timeout, resends buffered packets starting at LAR+1
- Selective Repeat uses a timer per unacked segment to detect losses
  - On timeout for segment, resend it
  - Hope to resend fewer segments

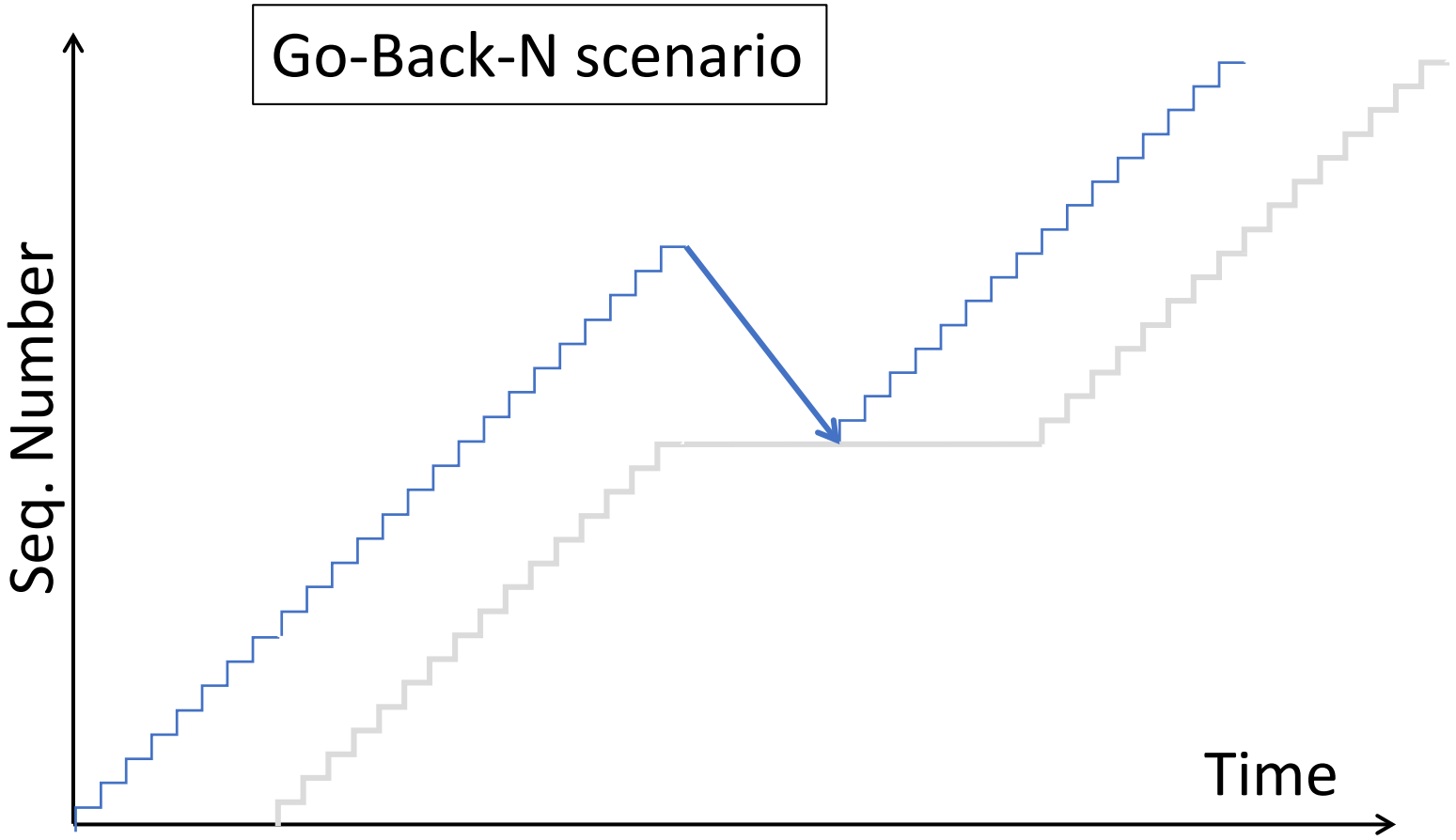
Sequence Numbers

- Need more than 0/1 for Stop-and-Wait …
  - But how many?
- For Selective Repeat, need W numbers for packets, plus W for acks of earlier packets
  - 2W seq. numbers
  - Fewer for Go-Back-N (W+1)
- Typically implement seq. number with an N-bit counter that wraps around at $2^N—1$
  - E.g., N=8: …, 253, 254, 255, 0, 1, 2, 3, …

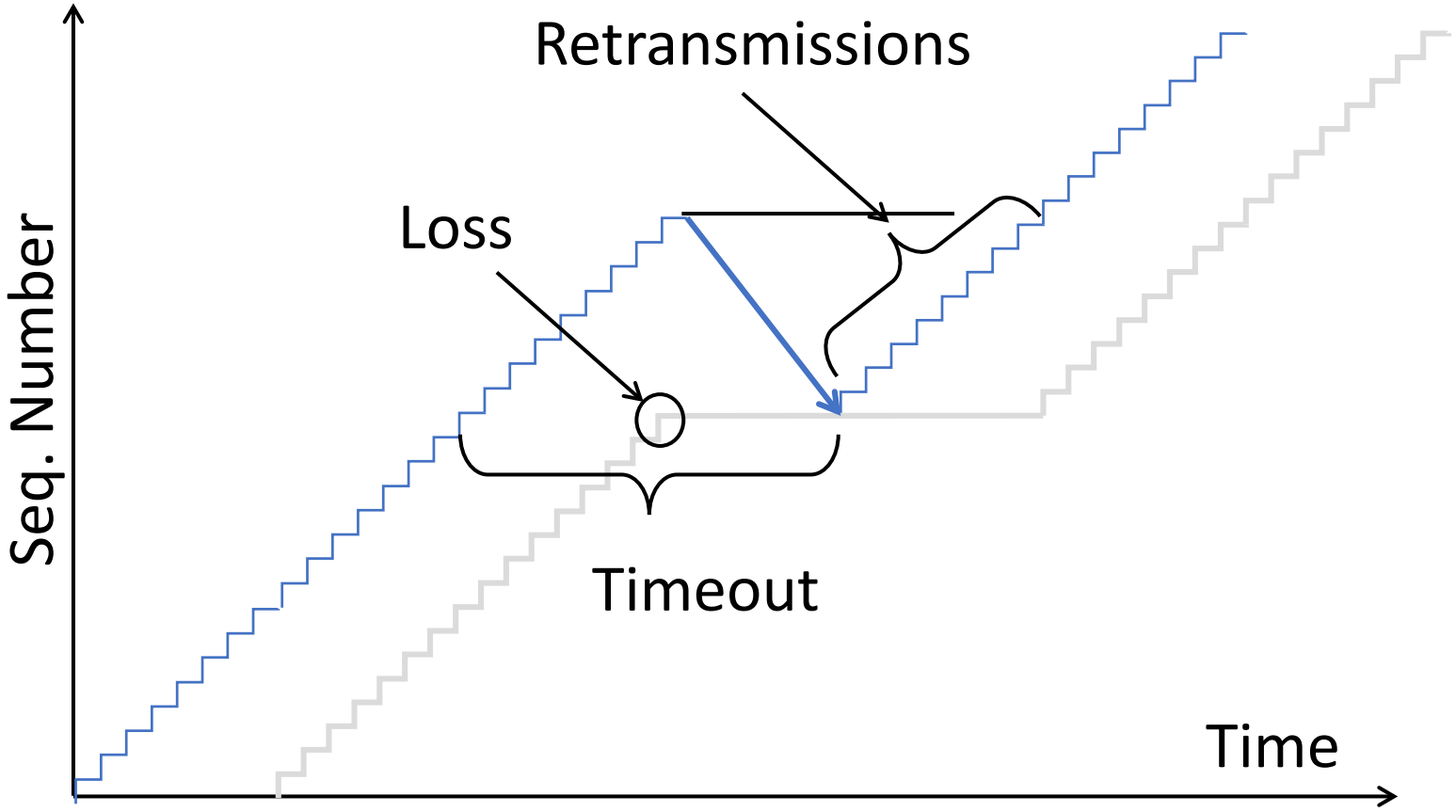
Sequence Time Plot



Sequence Time Plot (2)

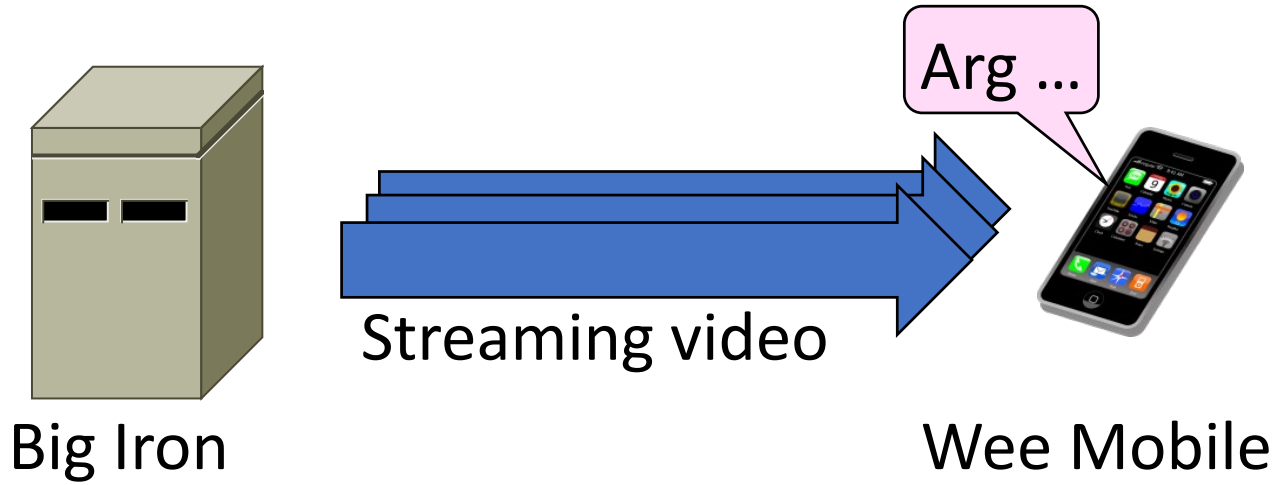


Sequence Time Plot (3)



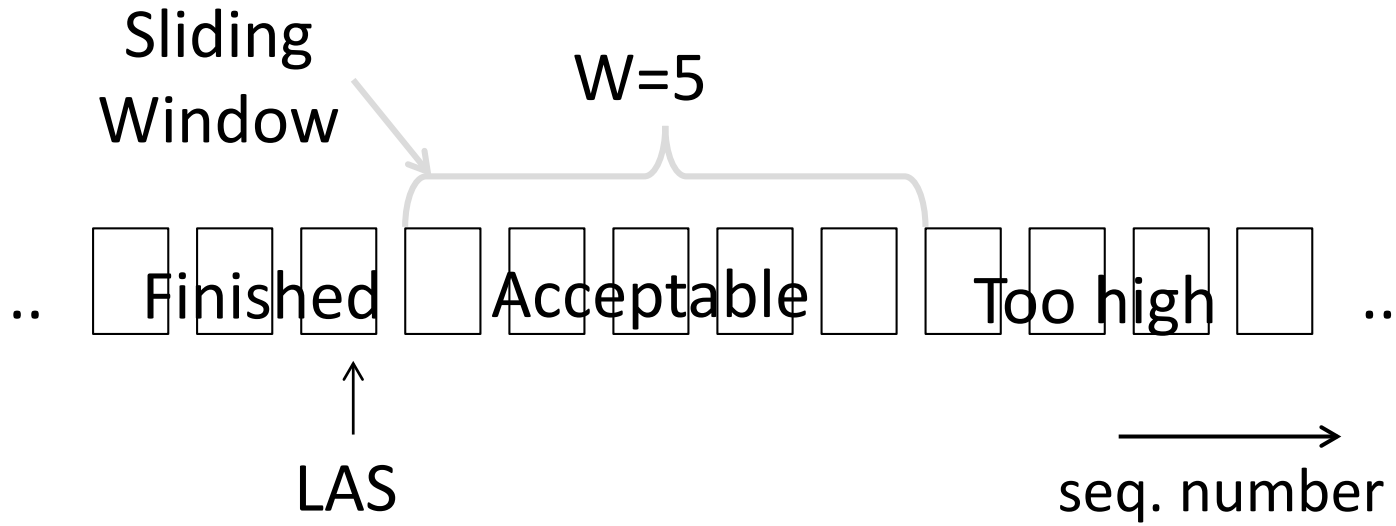
Problem

- Sliding window has pipelining to keep network busy
  - What if the receiver is overloaded?



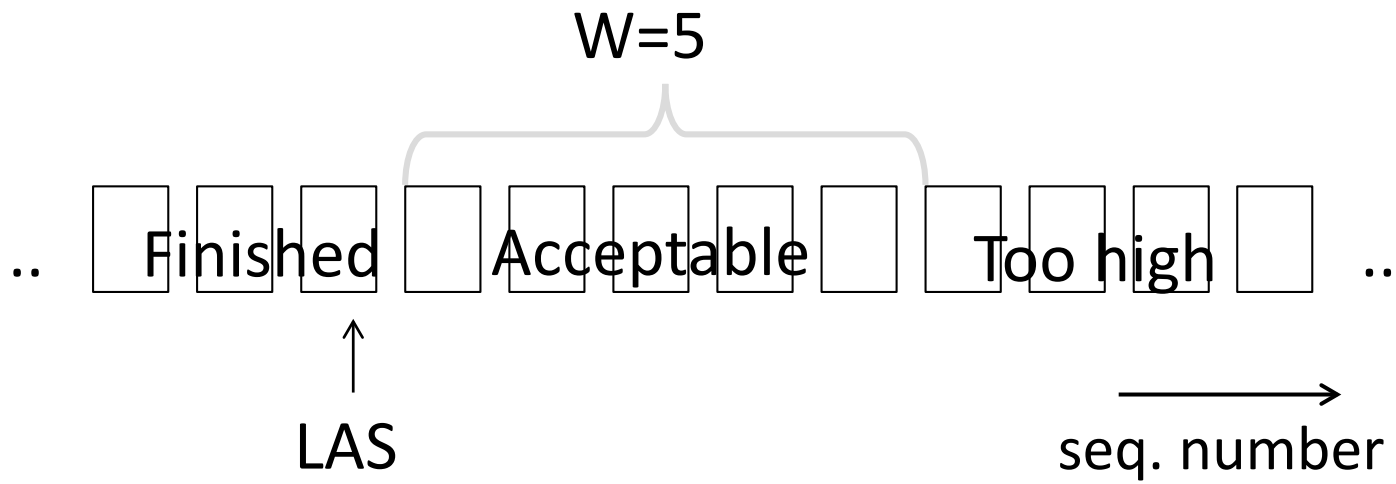
Sliding Window – Receiver

- Consider receiver with W buffers
  - LAS=LAST ACK SENT, app pulls in-order data from buffer with recv() call



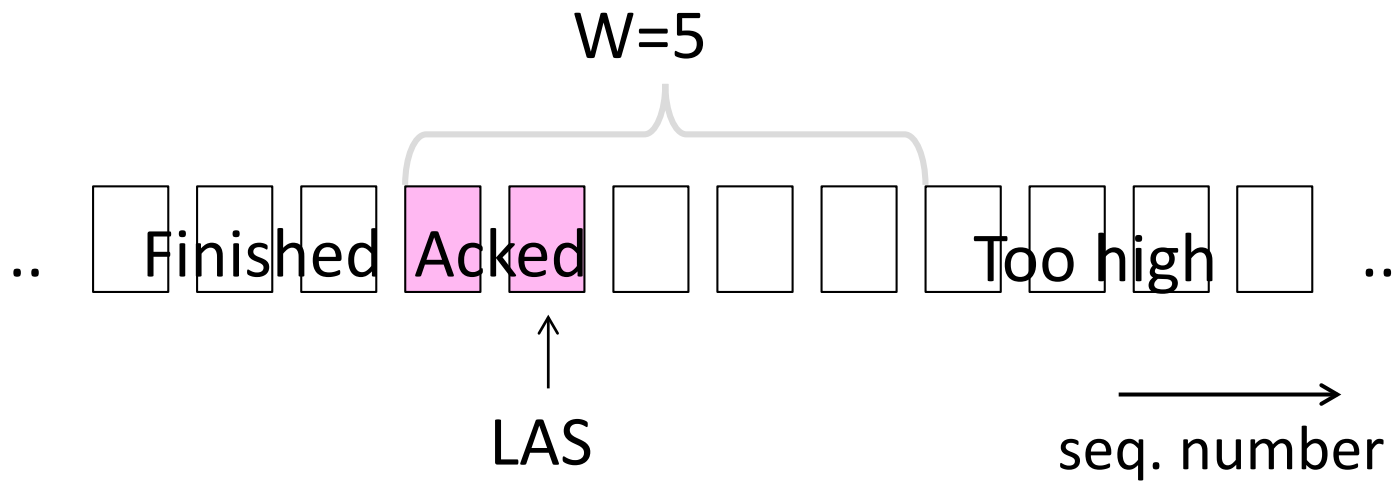
Sliding Window – Receiver (2)

- Suppose the next two segments arrive but app does not call recv()



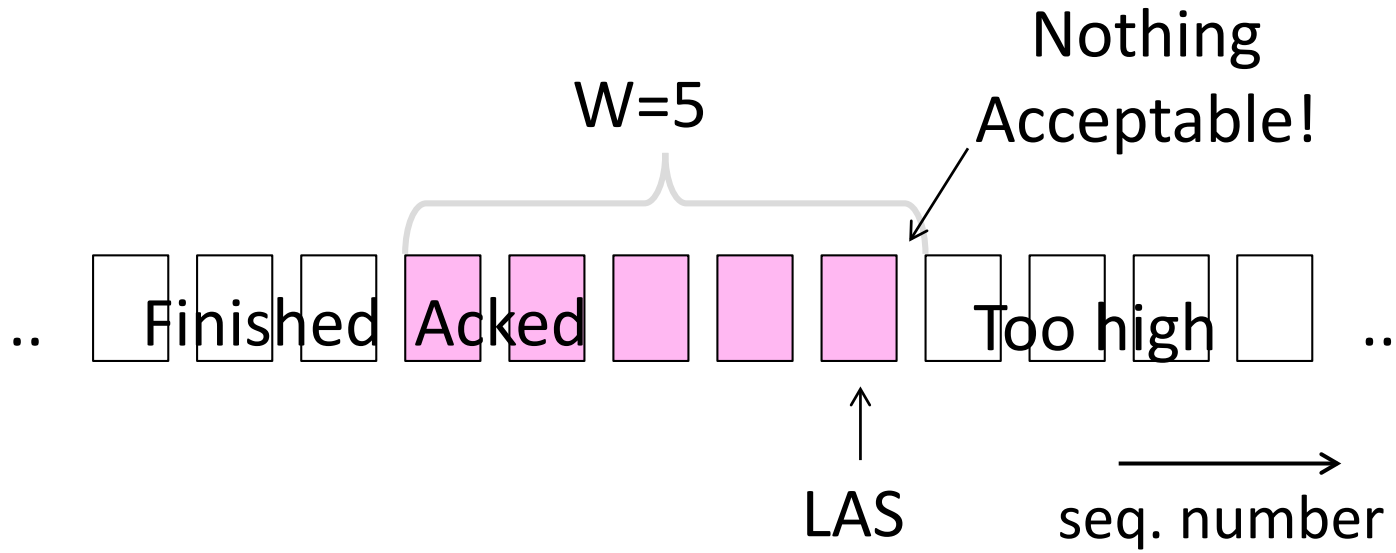
Sliding Window – Receiver (3)

- Suppose the next two segments arrive but app does not call recv()
  - LAS rises, but we can't slide window!



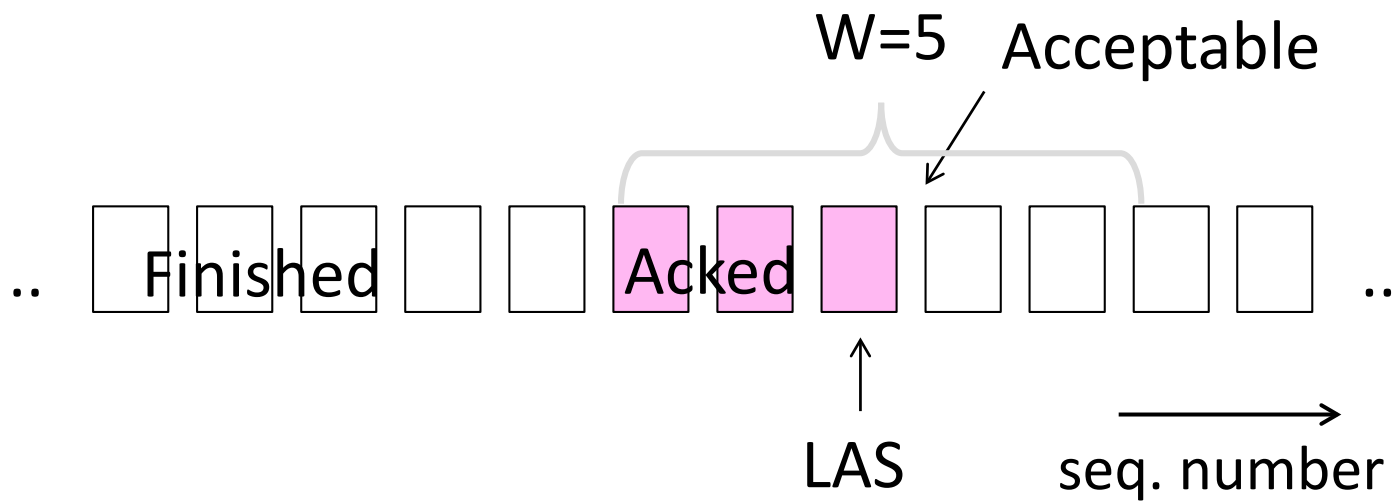
Sliding Window – Receiver (4)

- Further segments arrive (in order) we fill buffer
  - Must drop segments until app recvs!



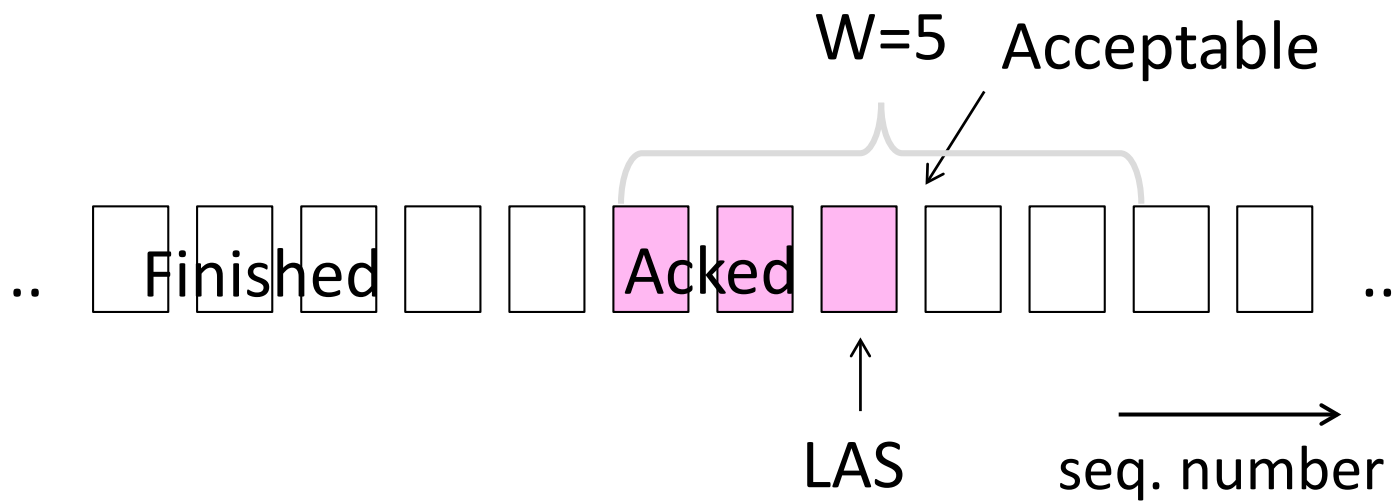
Sliding Window – Receiver (5)

- App recv() takes two segments
  - Window slides (phew)



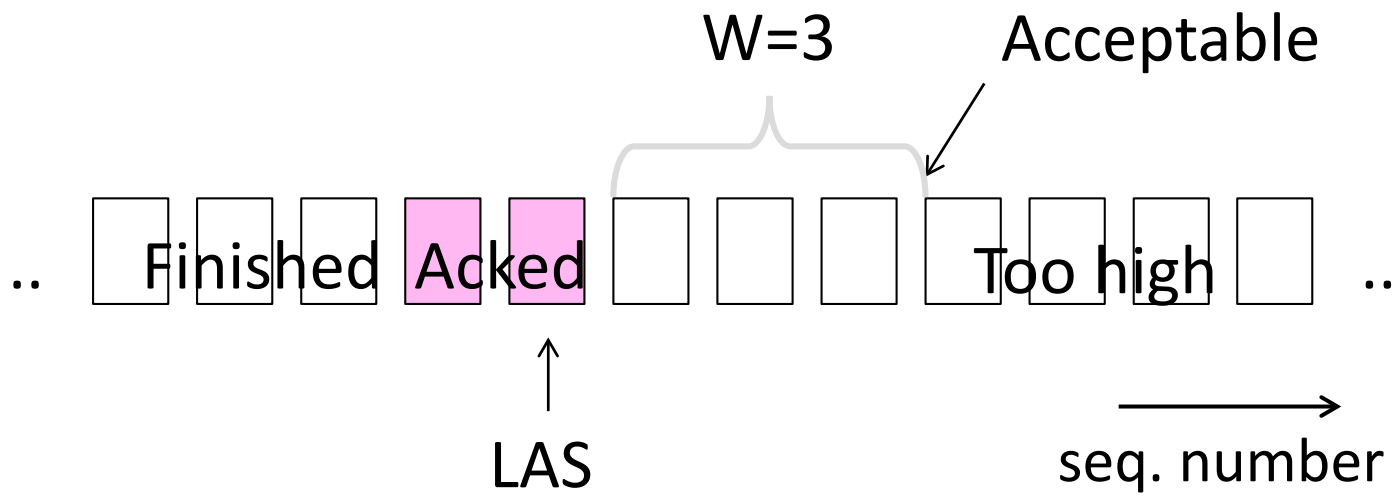
Flow Control

- Avoid loss at receiver by telling sender the available buffer space
  - WIN=#Acceptable, not W (from LAS)



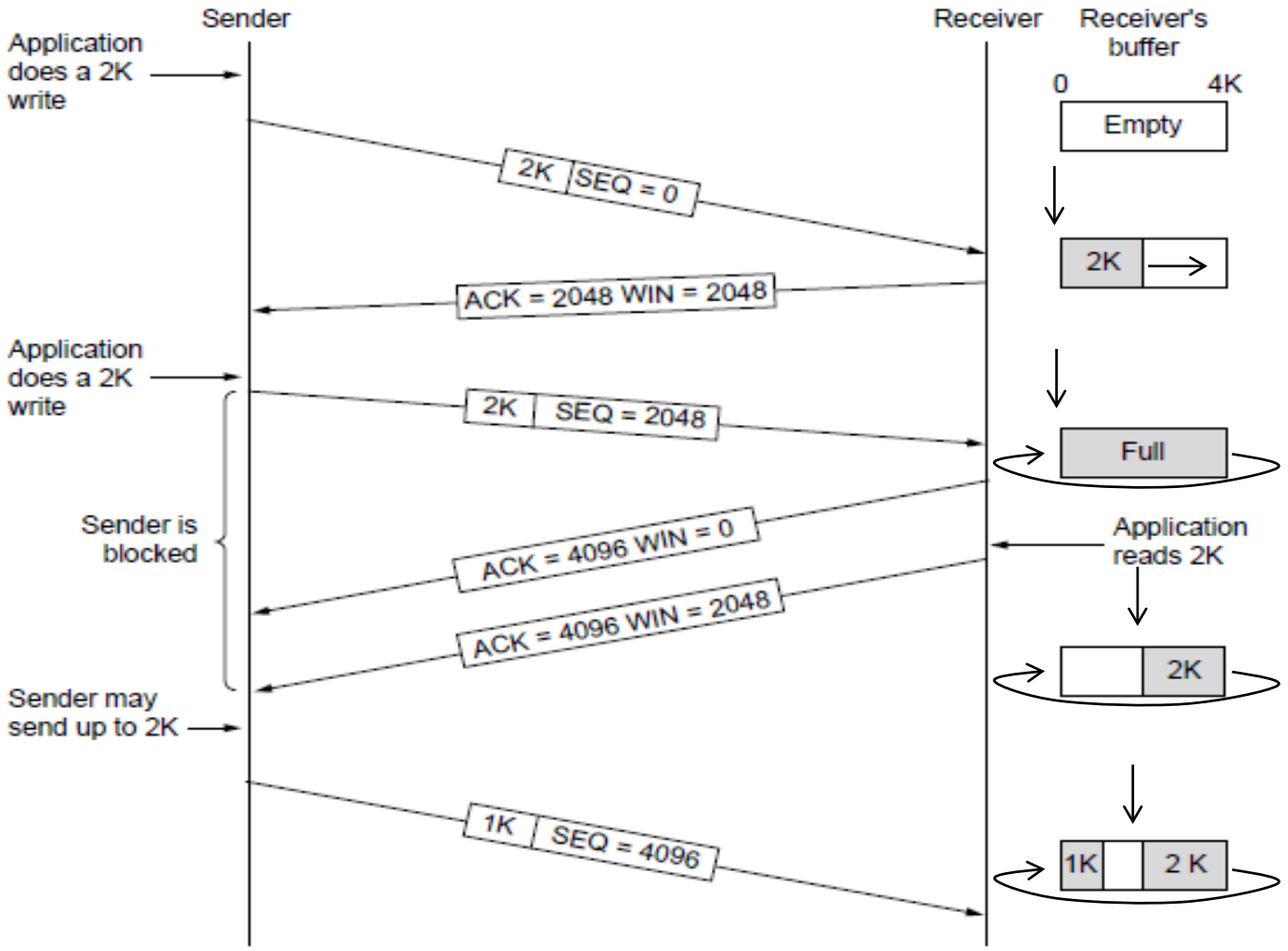
Flow Control (2)

- Sender uses lower of the sliding window and flow control window (WIN) as the effective window size



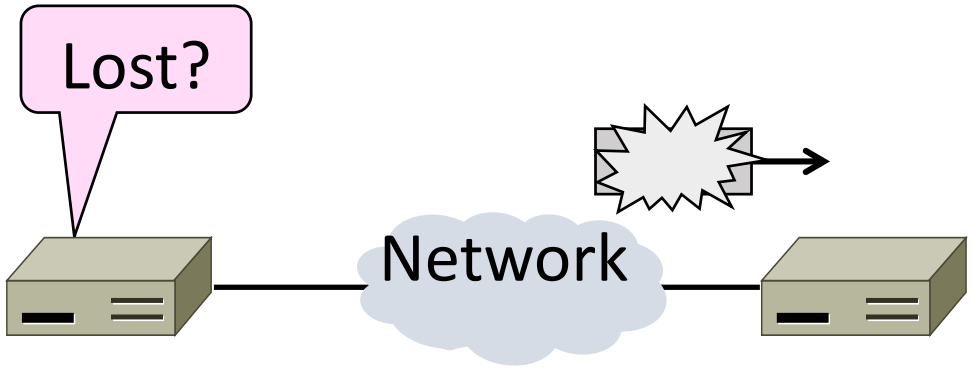
Flow Control (3)

- TCP-style example
  - SEQ/ACK sliding window
  - Flow control with WIN
  - SEQ + length < ACK+WIN
  - 4KB buffer at receiver
  - Circular buffer of bytes



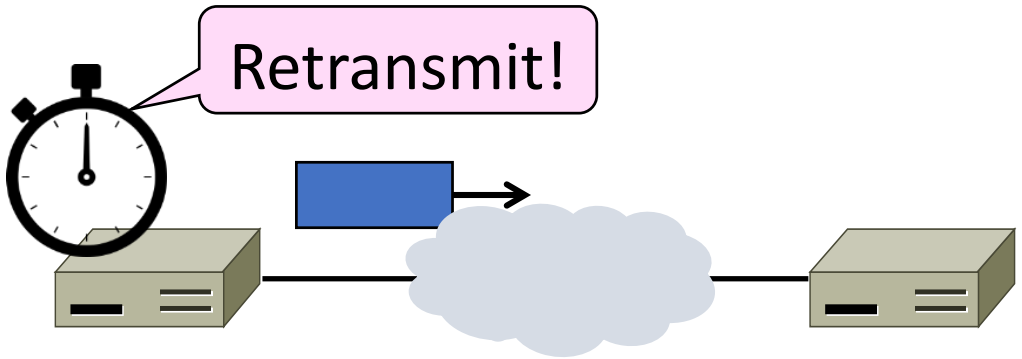
Topic

- How to set the timeout for sending a retransmission
  - Adapting to the network path



Retransmissions

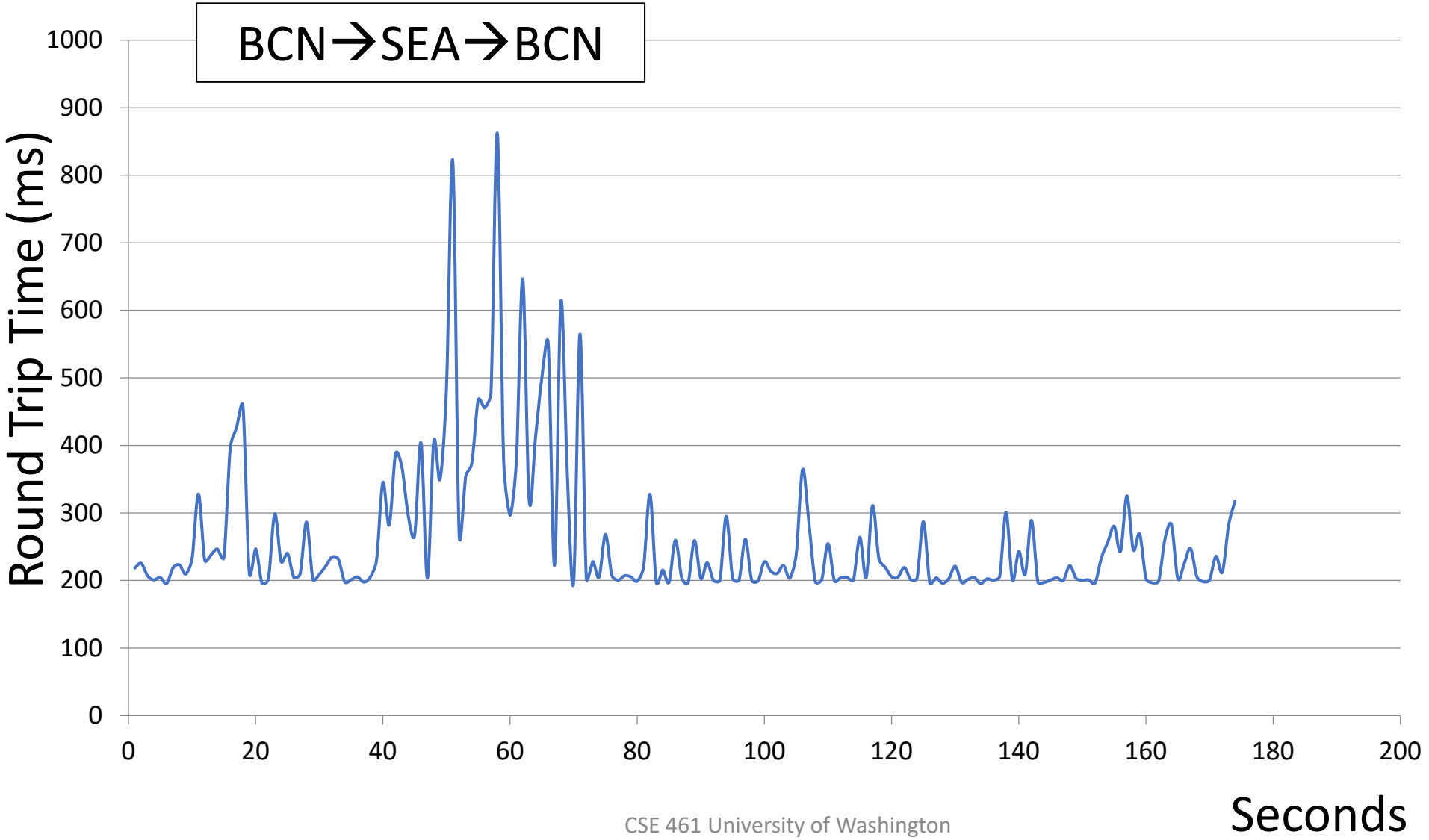
- With sliding window, detecting loss with timeout
  - Set timer when a segment is sent
  - Cancel timer when ack is received
  - If timer fires, retransmit data as lost



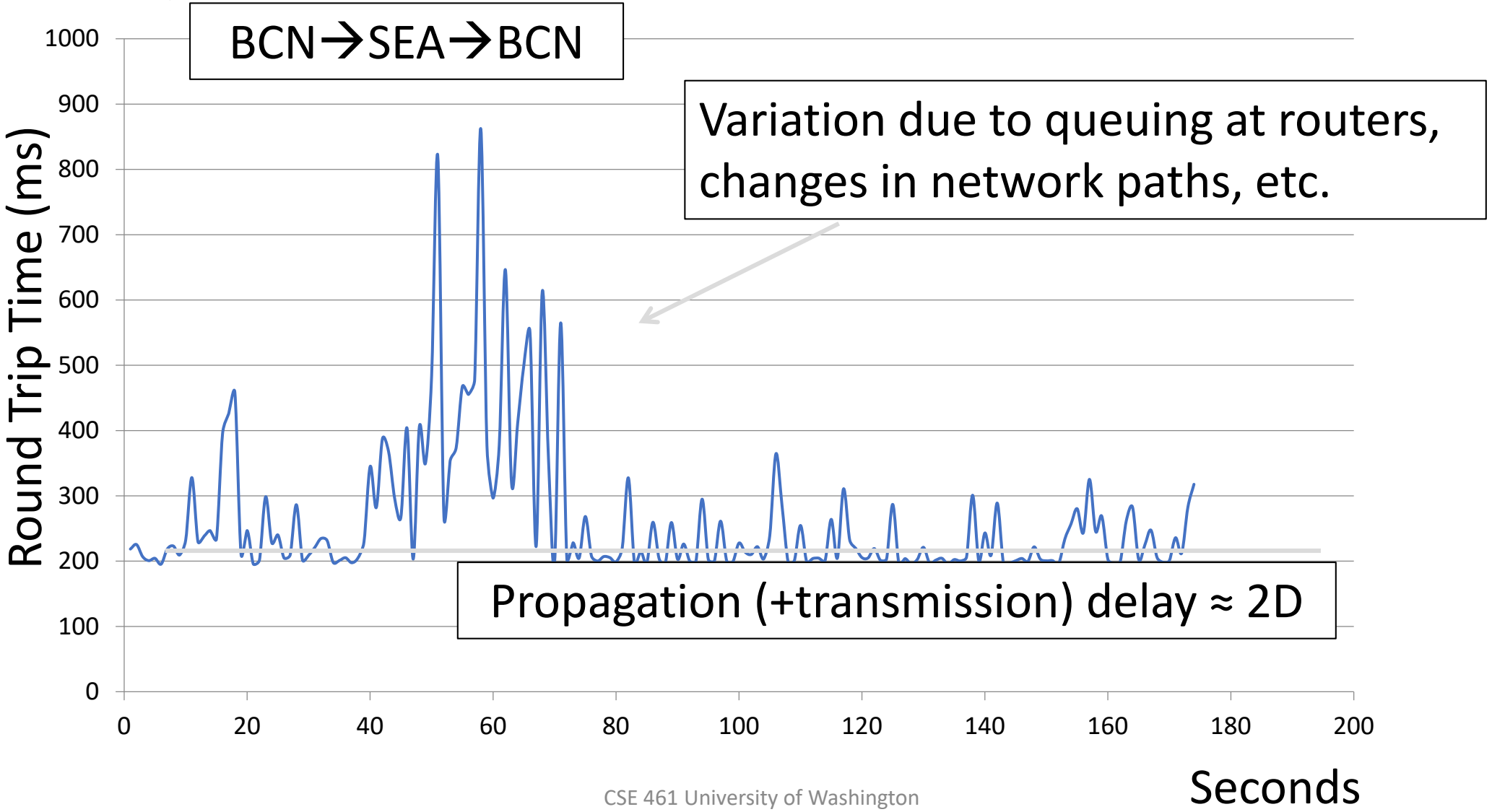
Timeout Problem

- Timeout should be “just right”
  - Too long wastes network capacity
  - Too short leads to spurious resends
  - But what is “just right”?
- Easy to set on a LAN (Link)
  - Short, fixed, predictable RTT
- Hard on the Internet (Transport)
  - Wide range, variable RTT

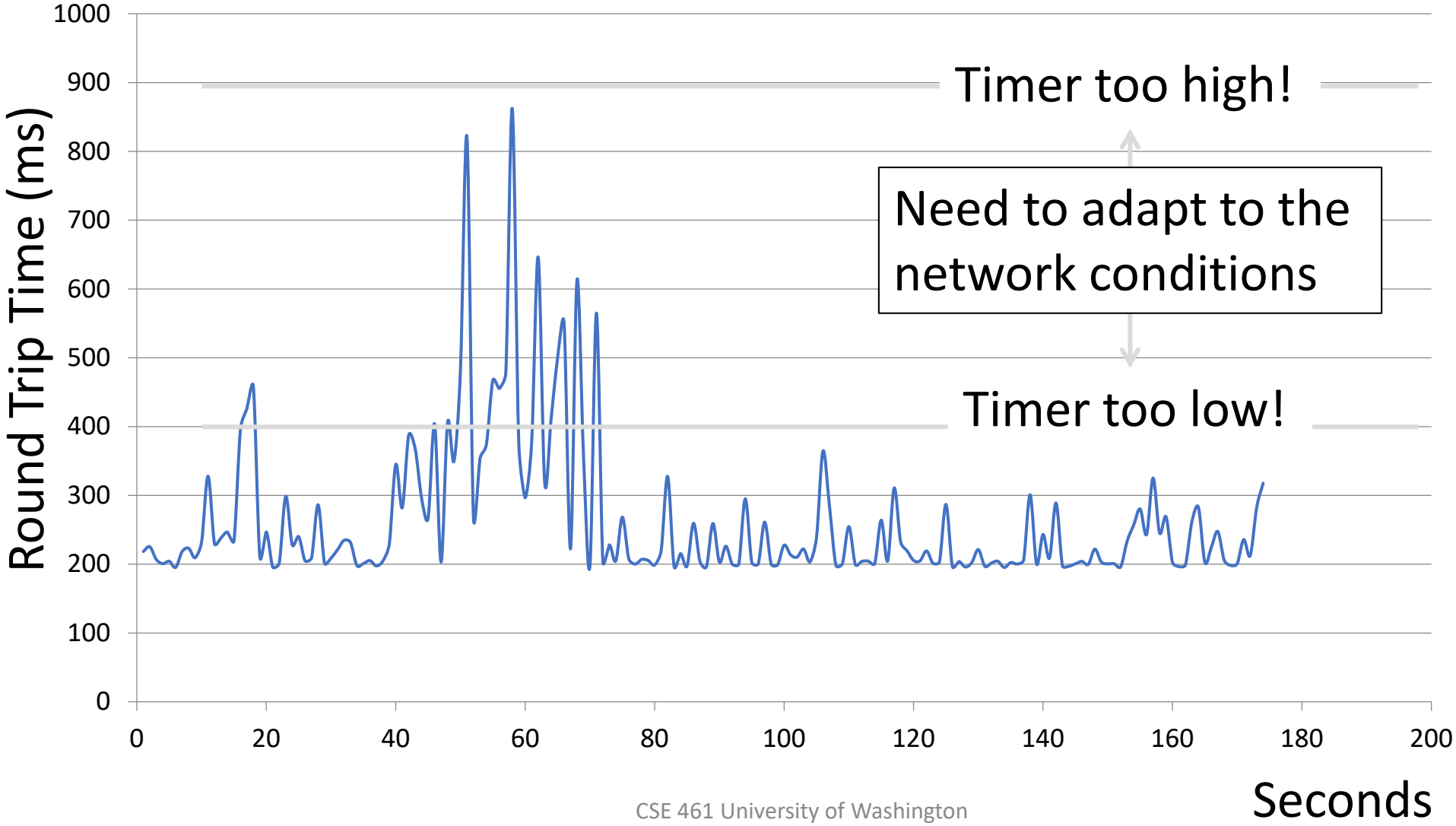
Example of RTTs



Example of RTTs (2)



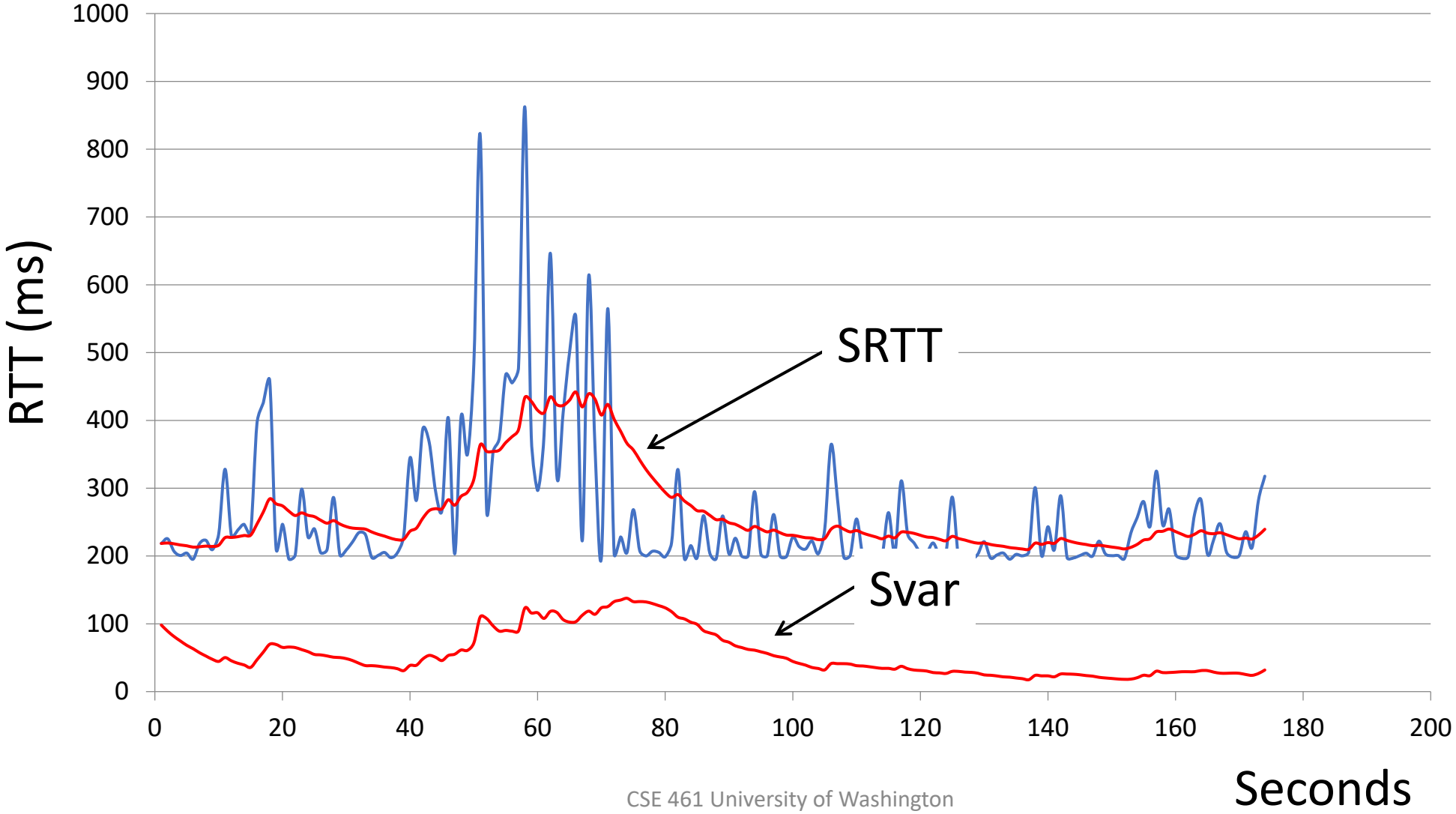
Example of RTTs (3)



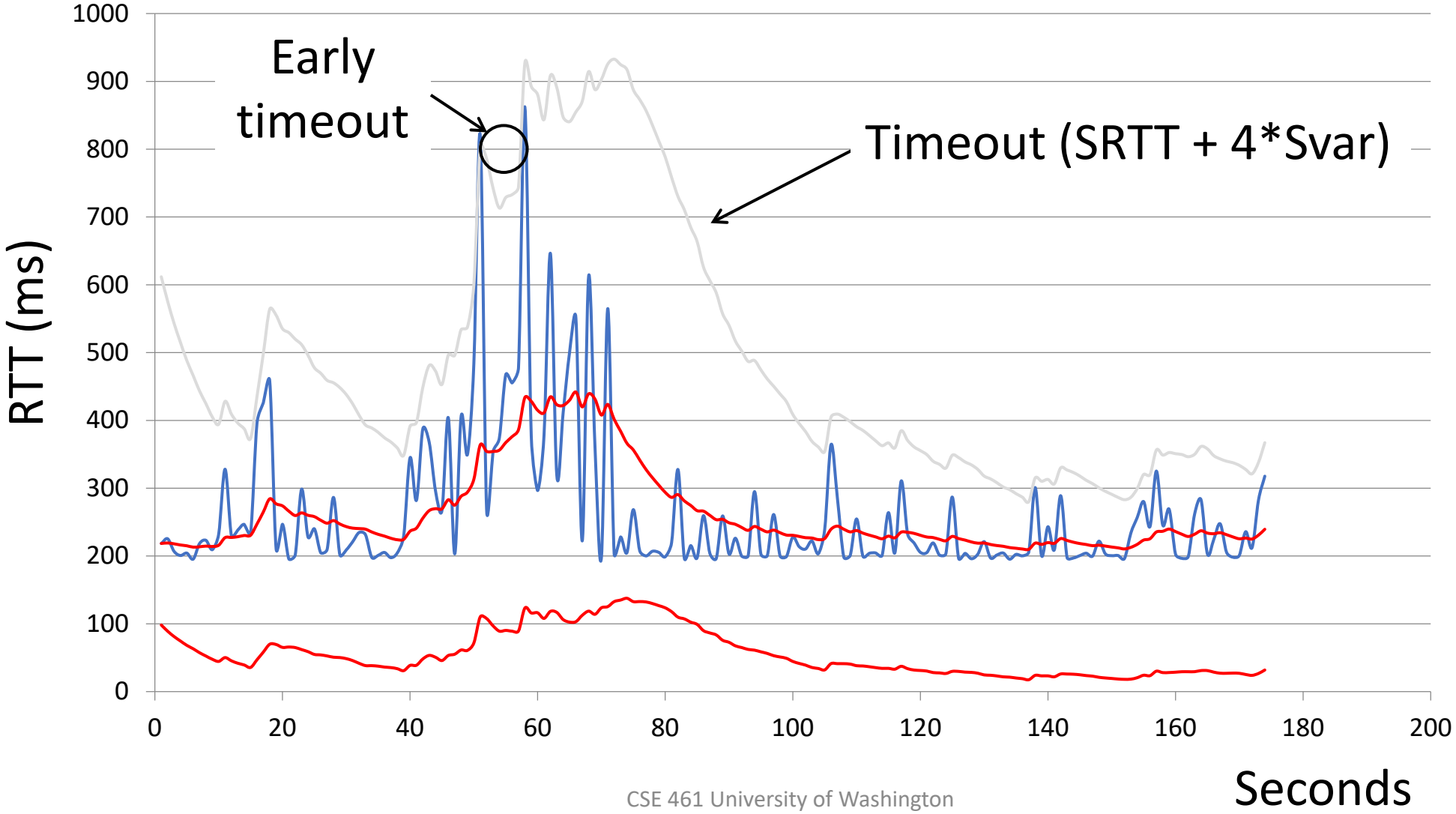
Adaptive Timeout

- Smoothed estimates of the RTT (1) and variance in RTT (2)
  - Update estimates with a moving average
  1. $SRTT_{N+1} = 0.9*SRTT_N + 0.1*RTT_{N+1}$
  2. $Svar_{N+1} = 0.9*Svar_N + 0.1*|RTT_{N+1} - SRTT_{N+1}|$
- Set timeout to a multiple of estimates
  - To estimate the upper RTT in practice
  - TCP $Timeout_N = SRTT_N + 4*Svar_N$

Example of Adaptive Timeout



Example of Adaptive Timeout (2)



Adaptive Timeout (2)

- Simple to compute, does a good job of tracking actual RTT
  - Little “headroom” to lower
  - Yet very few early timeouts
- Turns out to be important for good performance and robustness