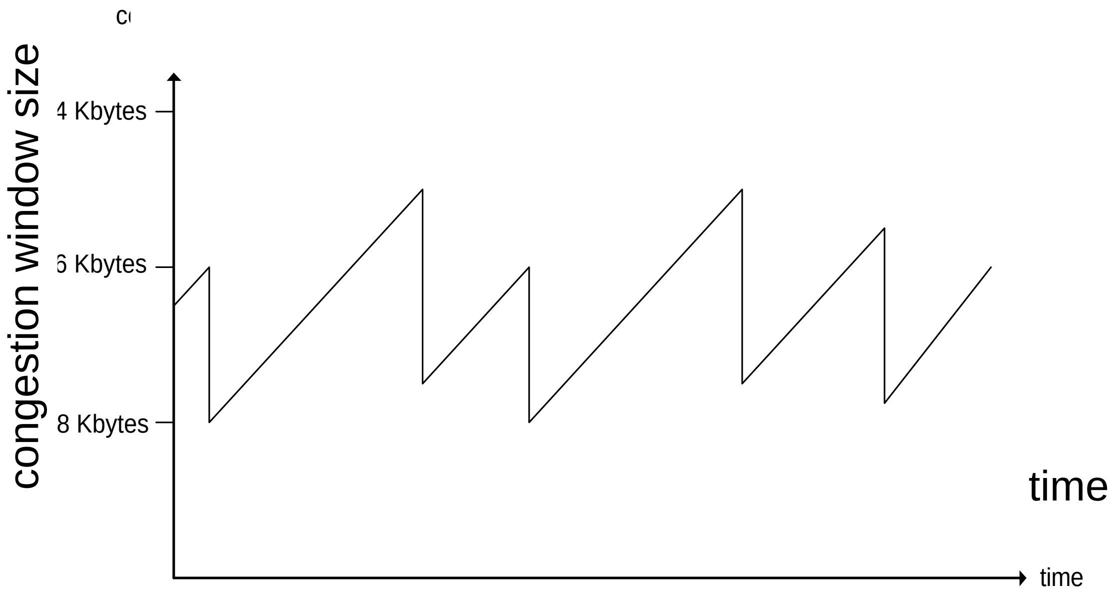
# TCP congestion control:

- "Probing" for usable bandwidth:
  - Ideally: transmit as fast as possible (**cwnd** as large as possible) without loss
  - Increase **cwnd** until loss (congestion)
  - Loss: decrease **cwnd**, then begin probing (increasing) again

# TCP congestion control: Additive increase, multiplicative decrease

- Approach: Increase transmission rate (window size), probing for usable bandwidth, until loss occurs
  - Additive increase: Increase **cwnd** by 1 MSS every RTT until loss detected
  - Multiplicative decrease: Cut **cwnd** in half after loss

Saw tooth behavior: probing for bandwidth



# TCP congestion control: Details

- Sender limits transmission:

  **LastByteSent-LastByteAcked**

  **£ cwnd**

- Roughly,

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

- **Cwnd** is dynamic, function of perceived network congestion

How does sender perceive congestion?

- Loss event = timeout or 3 duplicate acks
- TCP sender reduces rate (**cwnd**) after loss event

Three mechanisms:

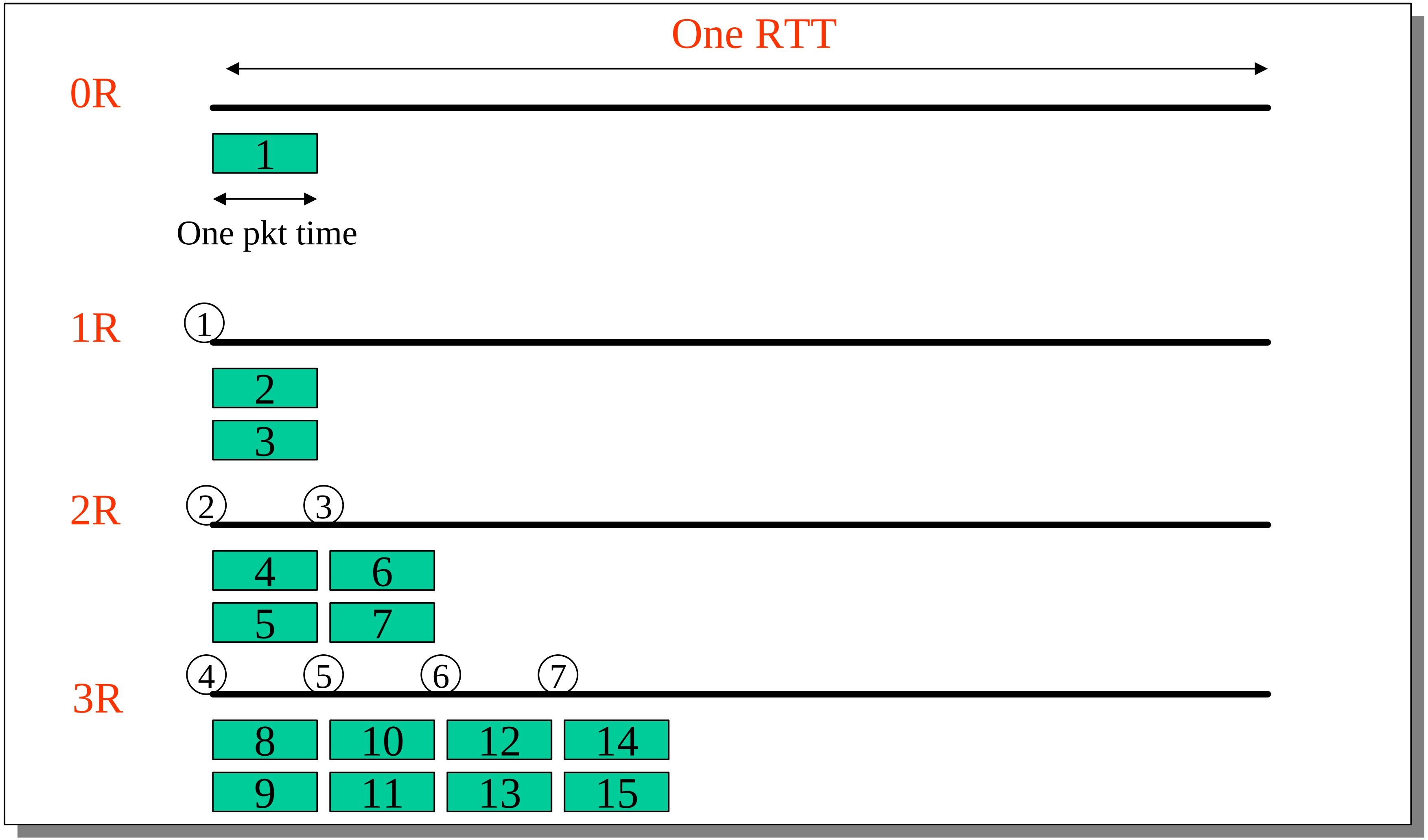
- AIMD
- Slow start
- Conservative after timeout events

# TCP slow start

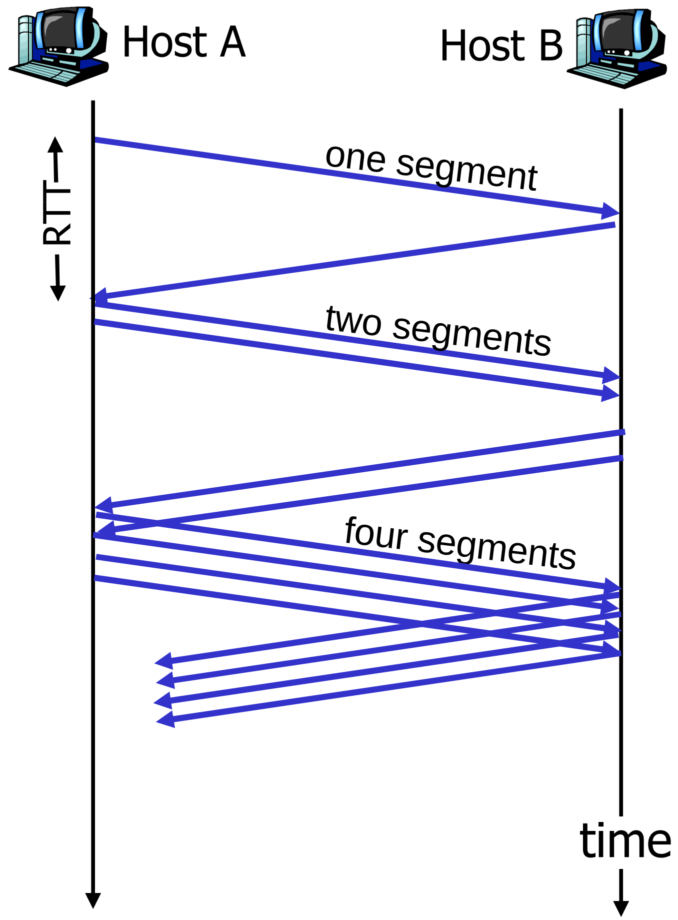
- How do we get the clocking behavior to start?
  - Initialize cwnd = 1 MSS (typically 1460 bytes)
  - Upon receipt of every ack, cwnd = cwnd + 1 MSS
- Implications
  - Window actually increases to W in RTT * $\log_2(W)$
  - Exponential increase up to first loss event
  - Can overshoot window and cause packet loss

- Summary: initial rate is slow but ramps up exponentially fast

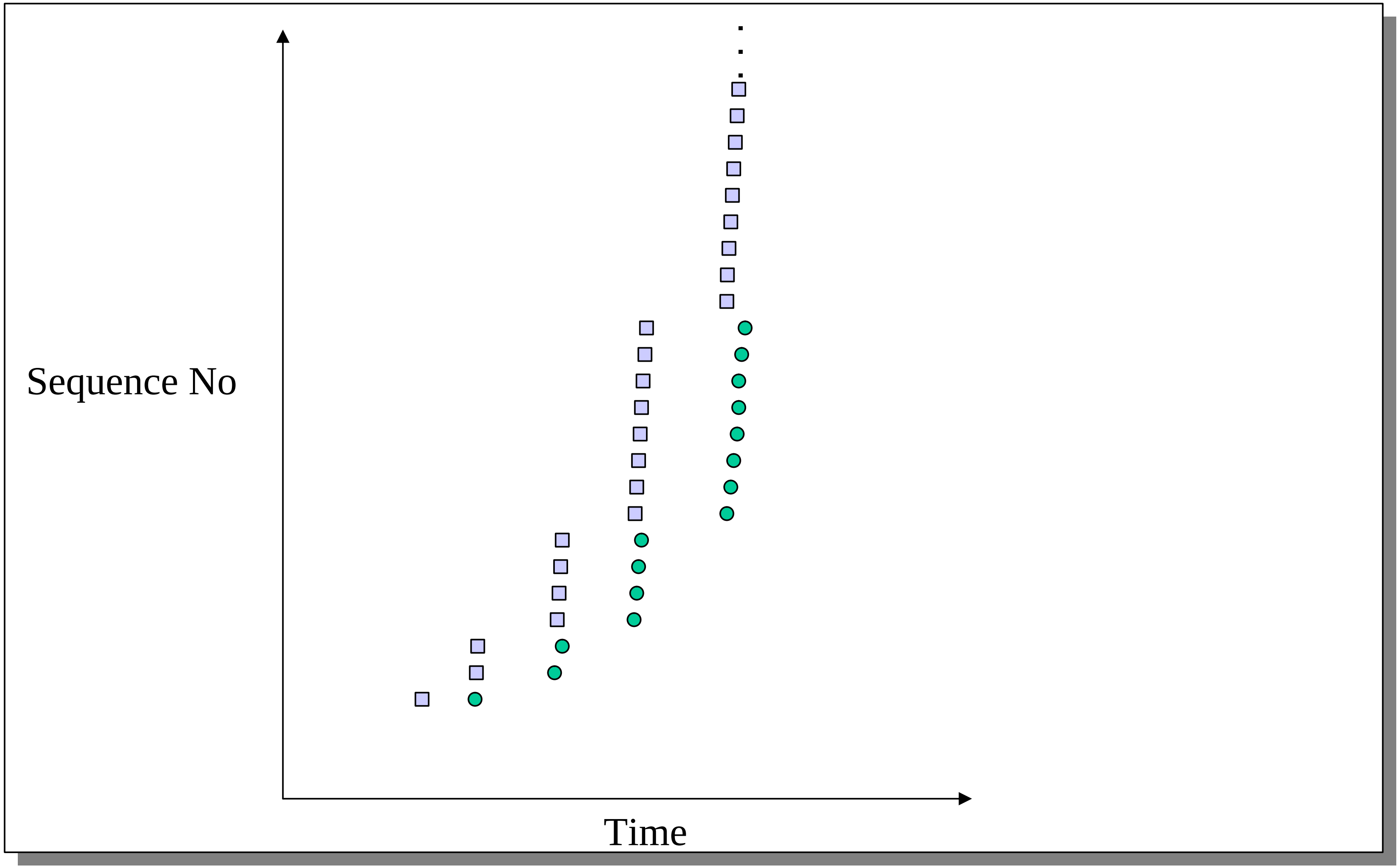
# Slow start example



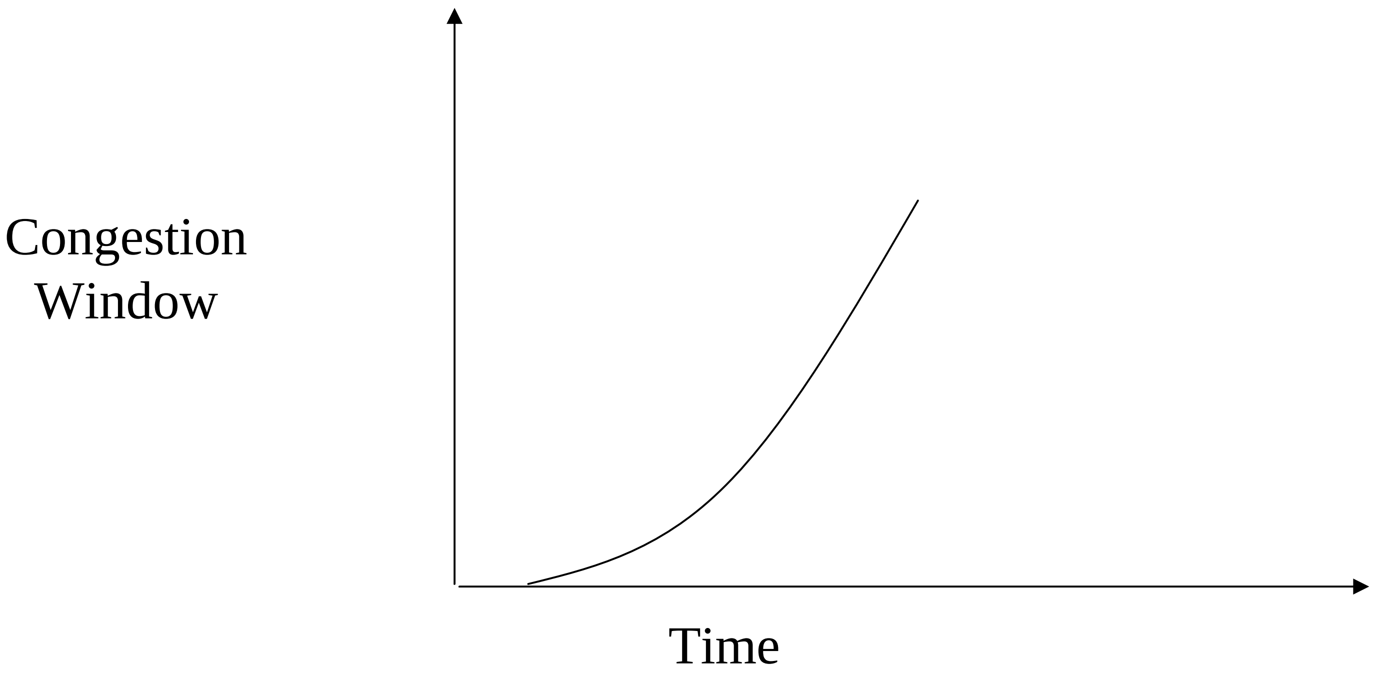
# Slow start example (cont.)



# Slow start sequence number plot



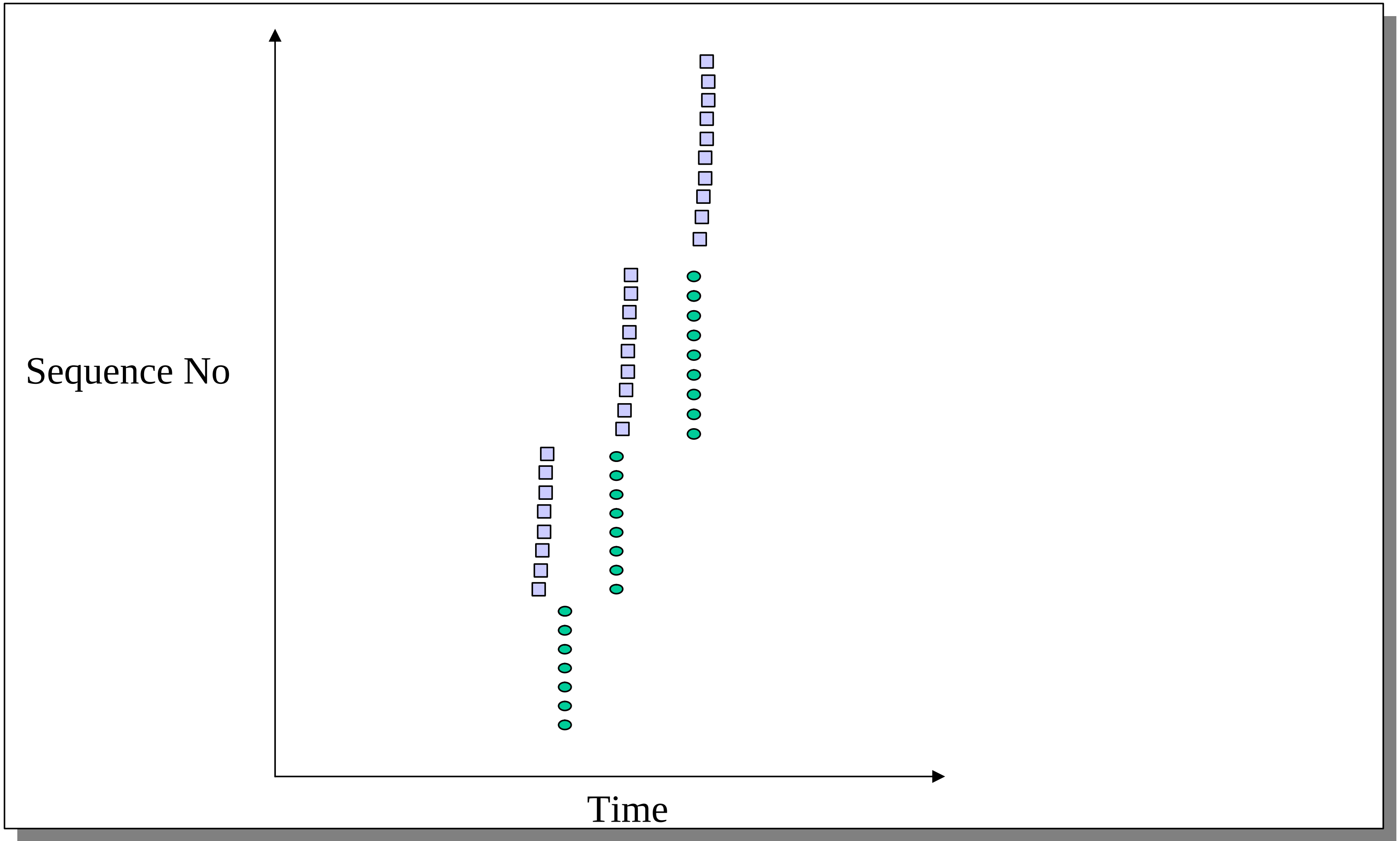
# Congestion window



# Congestion avoidance

- When reaching threshhold (ssthresh) go from slow start to congestion avoidance
- Upon receiving ACK
  - Increase cwnd by MSS/cwnd
  - Results in additive increase
- Loss implies congestion – why?
  - Not necessarily true on all link types

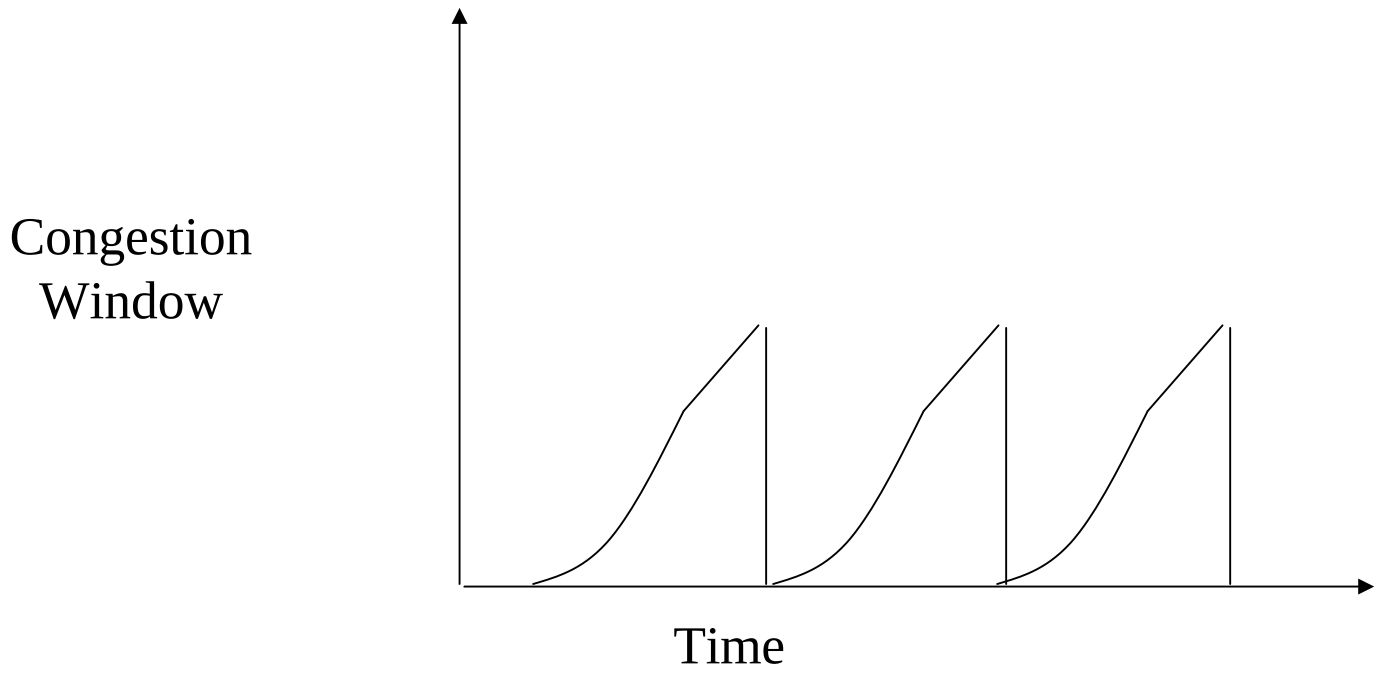
# Congestion avoidance sequence plot



# Return to slow start

- If packet is lost we lose our self clocking as well
  - Need to implement slow-start and congestion avoidance together

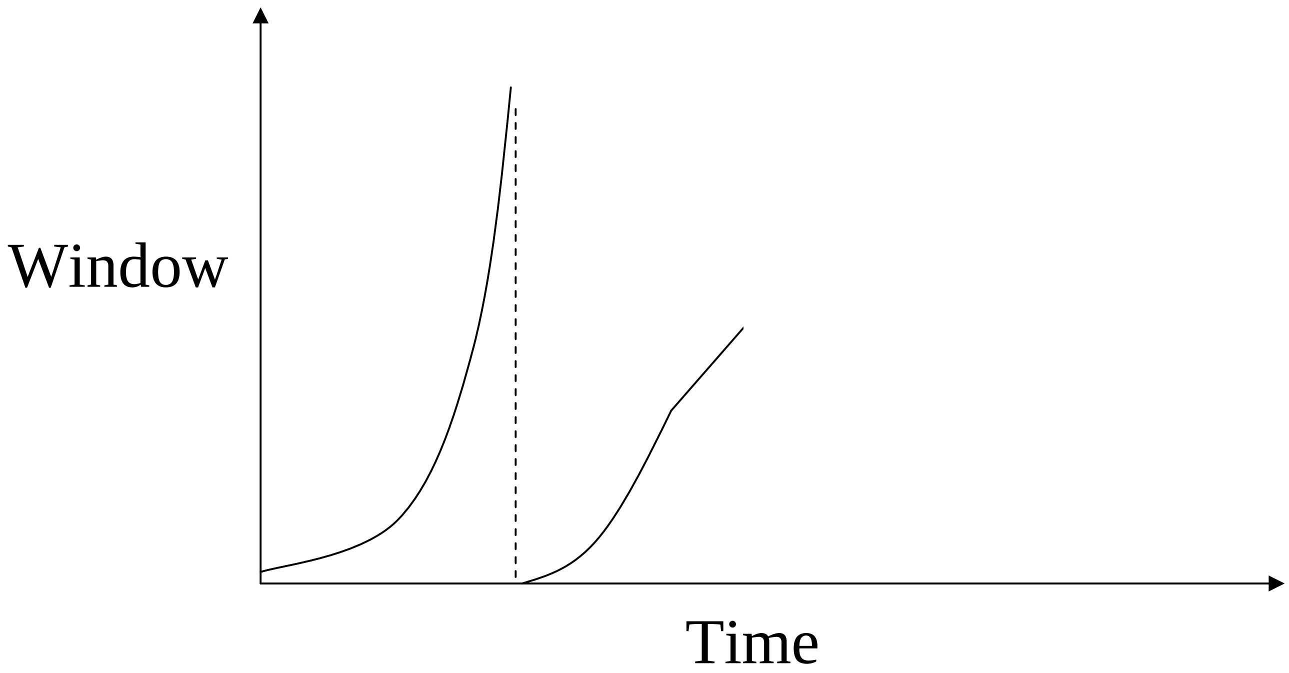
# Congestion window



# Return to slow start (cont.)

- If packet is lost we lose our self clocking as well
  - Need to implement slow-start and congestion avoidance together
- When loss occurs set
  - ssthresh to 0.5 w
  - cwnd to 1 MSS (TCP-Tahoe)

# Overall TCP behavior



# TCP congestion control: Summary

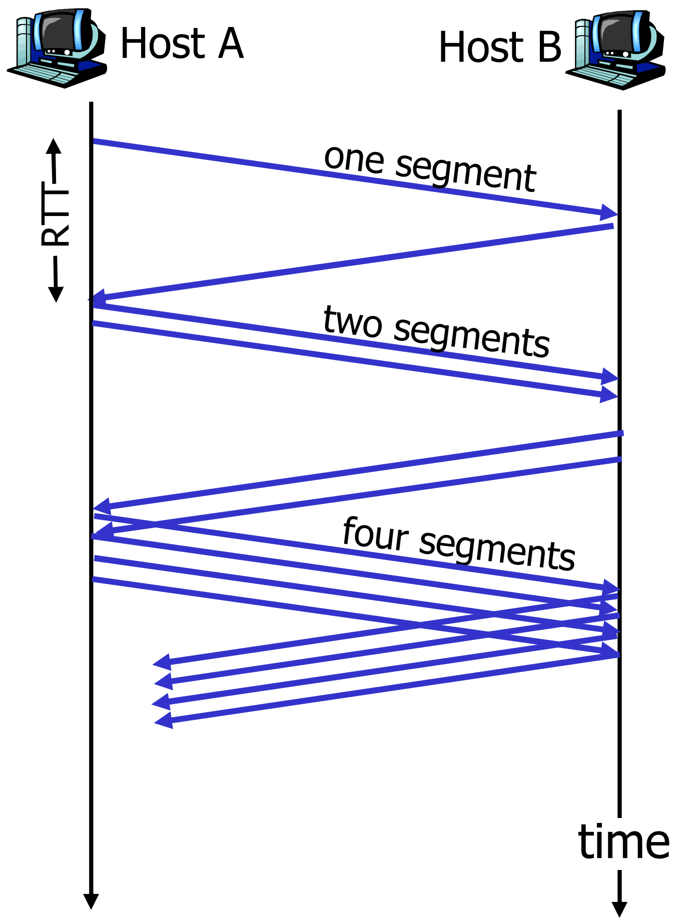
- "Probing" for usable bandwidth:
  - Ideally: transmit as fast as possible (**cwnd** as large as possible) without loss
  - Increase **cwnd** until loss (congestion)
  - Loss: decrease **cwnd**, then begin probing (increasing) again
- Two "phases"
  - Slow start
  - Congestion avoidance
- Important variables:
  - **Cwnd**
  - **Threshold:** defines threshold between two slow start phase, congestion control phase

# TCP slow start

**Slowstart algorithm**

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

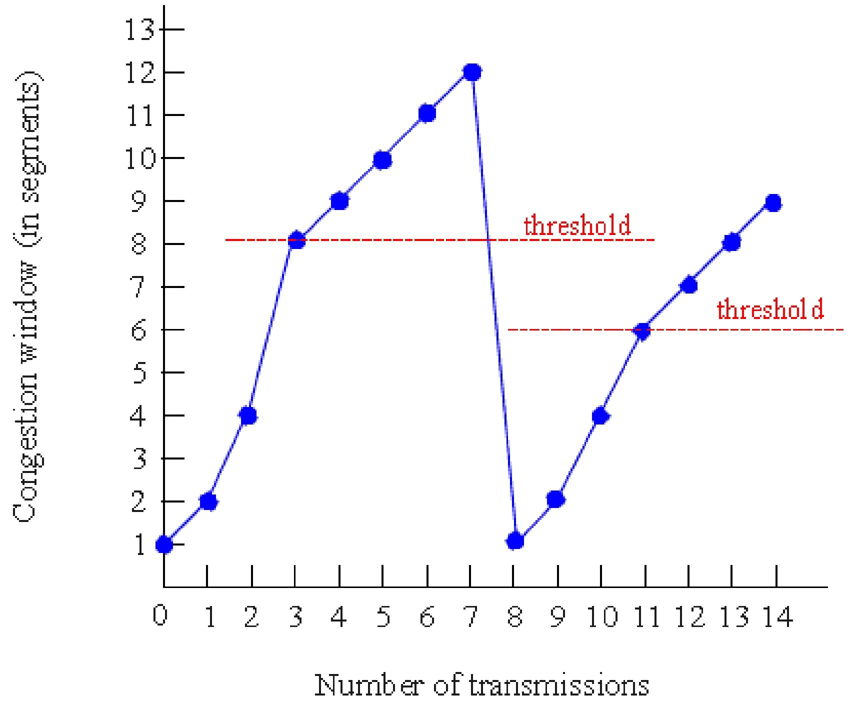
- Exponential increase (per RTT) in window size (not so slow!)
- Loss event: timeout and/or or three duplicate ACKs



# TCP congestion avoidance

Congestion avoidance

```
/* slowstart is over       */
/* cwnd > threshold */
Until (loss event) {
  every w segments ACKed:
 cwnd += 1 MSS
  }
threshold = cwnd /2
cwnd = 1 MSS
perform slow start[1]
```




1: TCP Reno skips slowstart (fast recovery) after three duplicate ACKs

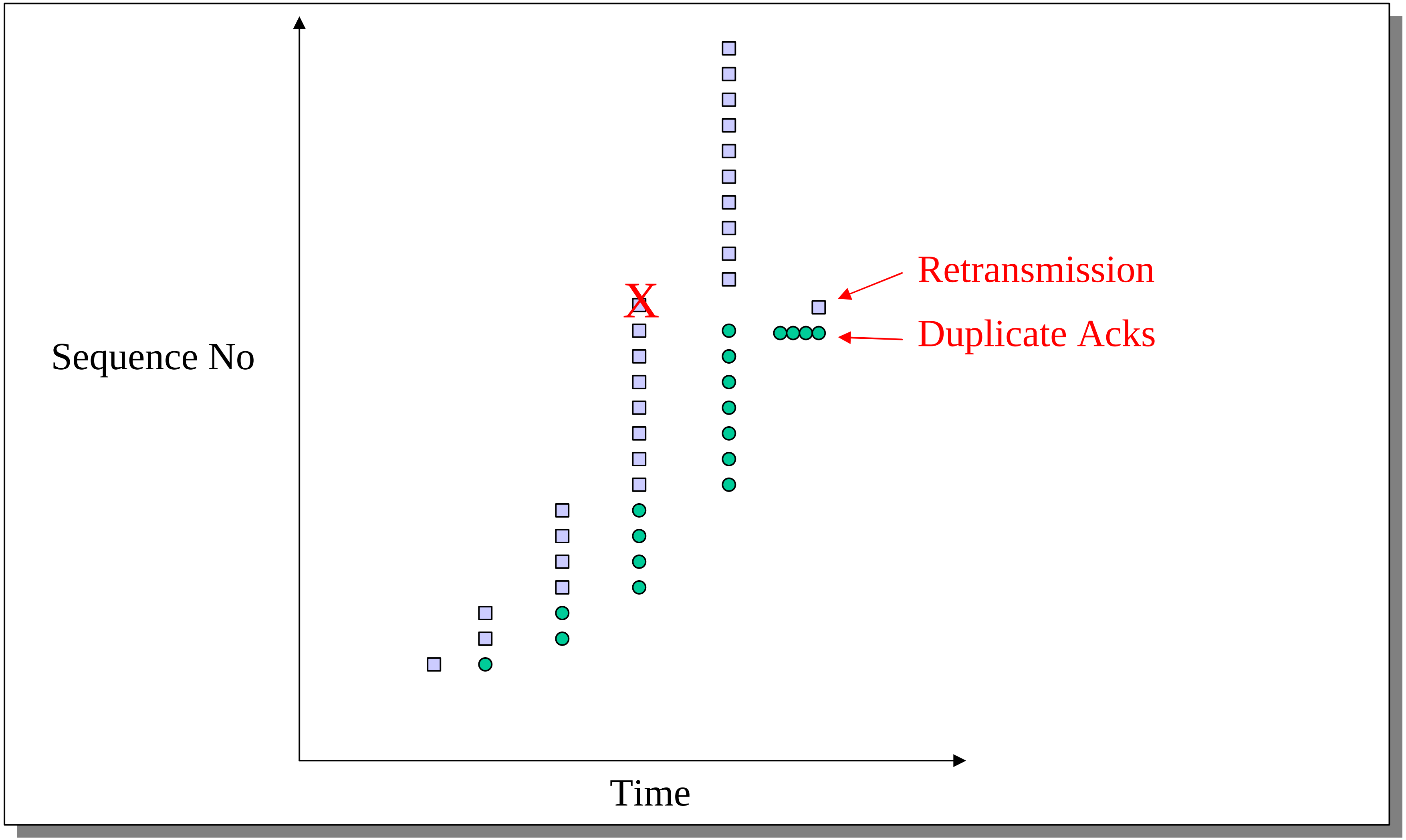
# TCP flavors

- Tahoe, Reno, Vegas, SACK
- TCP Tahoe (distributed with 4.3BSD Unix)
  - Original implementation of Van Jacobson's mechanisms
  - Includes:
    - Slow start
    - Congestion avoidance
    - Fast retransmit

# Fast retransmit

- What are duplicate acks (dupacks)?
  - Repeated acks for the same sequence
- When can duplicate acks occur?
  - Loss
  - Packet re-ordering
  - Window update – advertisement of new flow control window
- Assume re-ordering is infrequent and not of large magnitude
  - Use receipt of 3 or more duplicate acks as indication of loss
  - Don’t wait for timeout to retransmit packet

# Fast retransmit



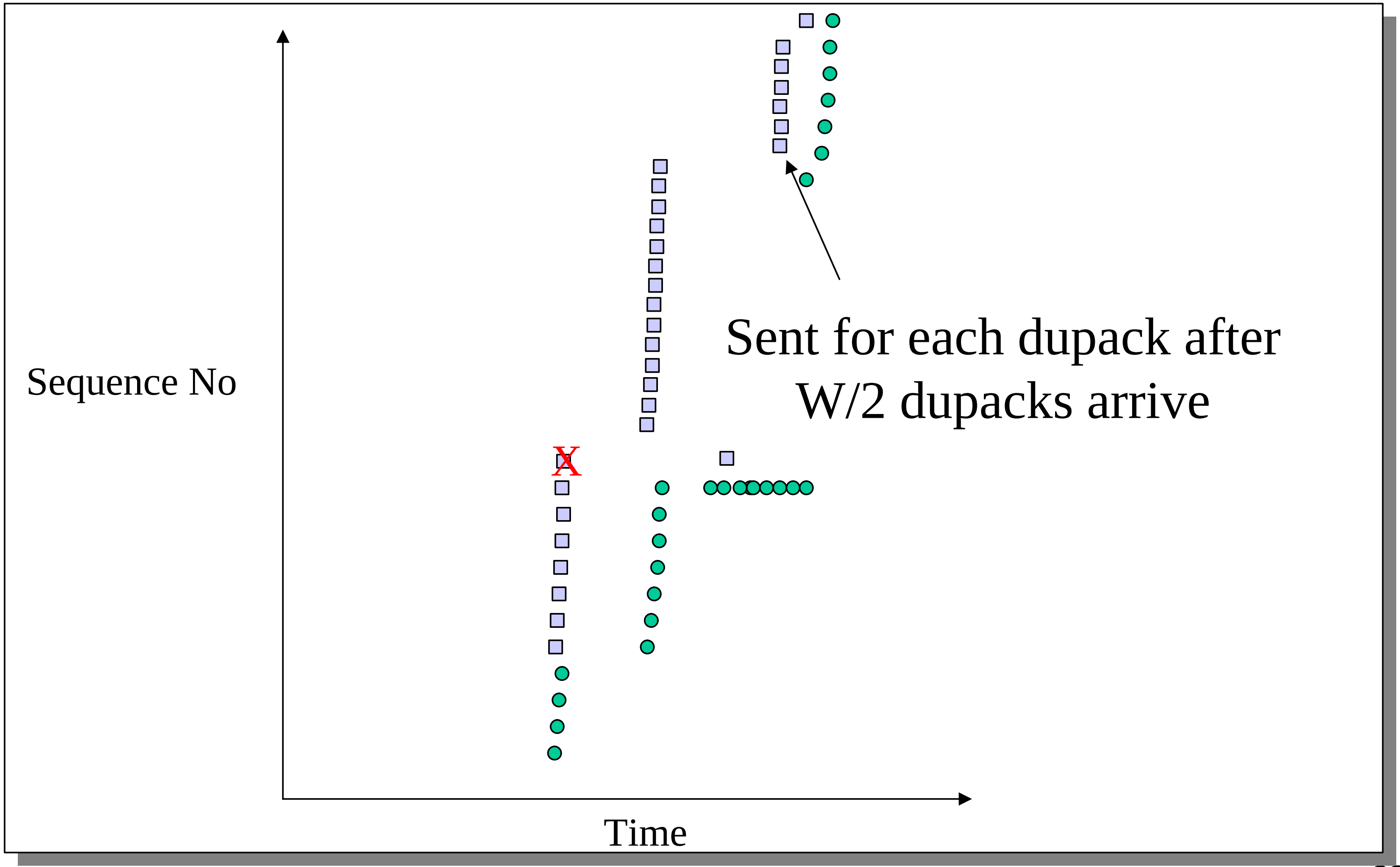
# TCP Reno (1990)

- All mechanisms in Tahoe
- Addition of fast-recovery
  - Opening up congestion window after fast retransmit
- Delayed acks
- Header prediction
  - Implementation designed to improve performance
  - Has common case code inlined
- With multiple losses, Reno typically timeouts because it does not see duplicate acknowledgements

# Fast recovery

- Skip slow start. On 3 dup ack event:
  - ssthresh <- 0.5 cwnd
  - cwdn <- sstresh
- Each duplicate ack notifies sender that single packet has cleared network
- When < cwnd packets are outstanding
  - Allow new packets out with each new duplicate acknowledgement

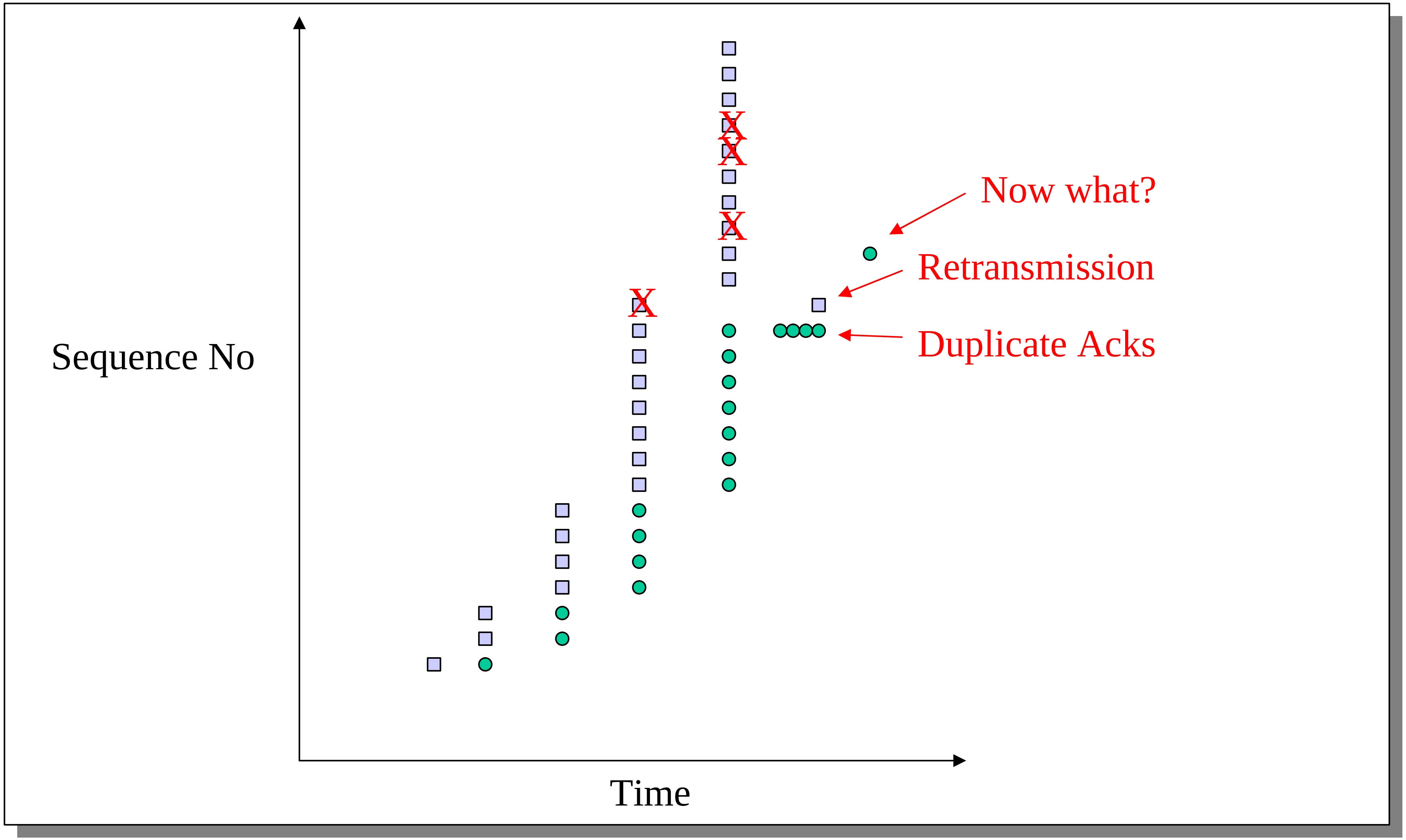
# Fast recovery



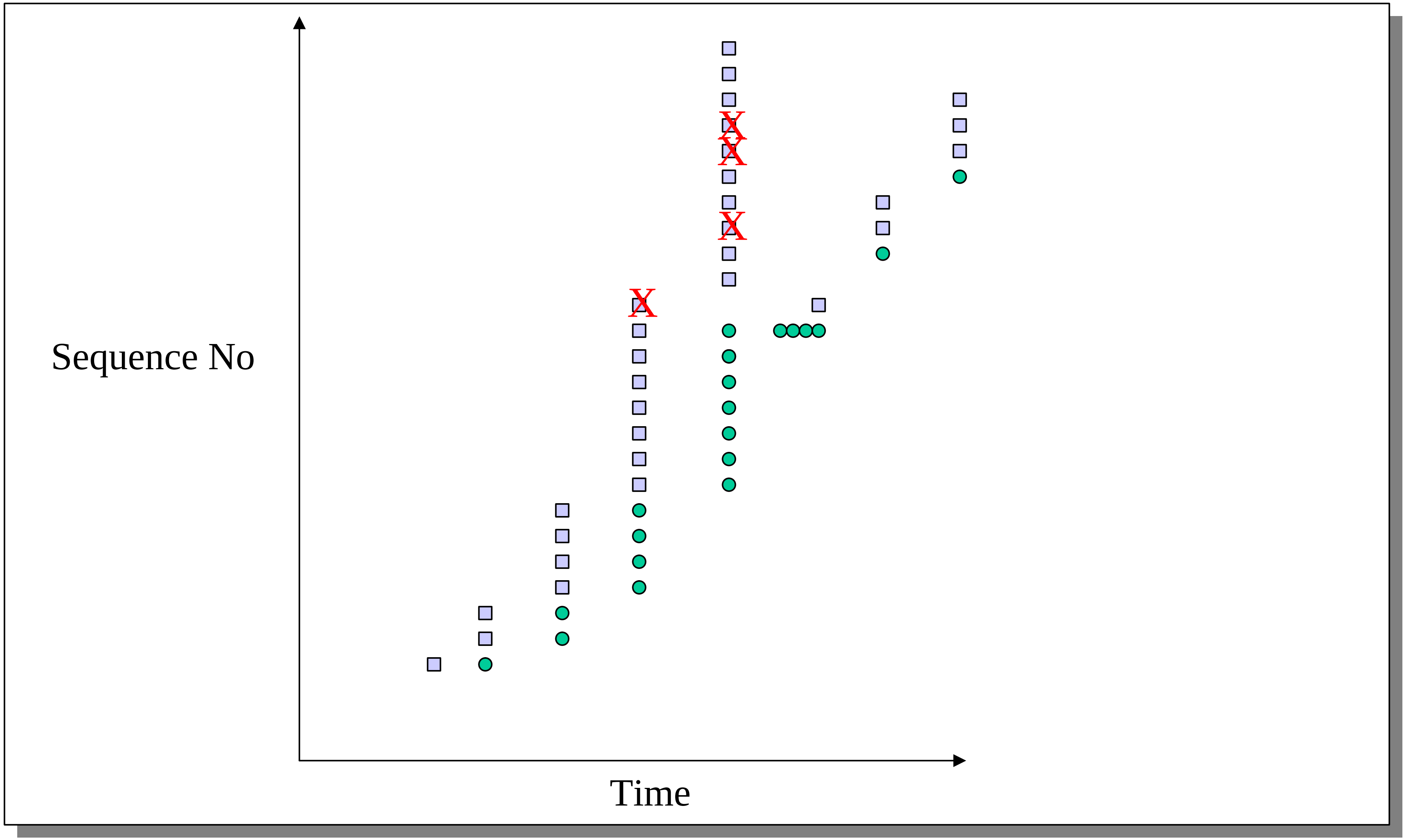
# TCP sender congestion control (Reno)

| State | Event | TCP Sender Action | Commentary |
|---|---|---|---|
| Slow Start (SS) | ACK receipt for previously unacked data | cwnd = cwnd + MSS,<br>If (cwnd > Threshold)<br>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 / (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,<br>cwnd = Threshold,<br>Set state to “Congestion Avoidance” | Fast recovery, implementing multiplicative decrease. cwnd will not drop below 1 MSS. |
| SS or CA | Timeout | Threshold = cwnd /2,<br>cwnd = 1 MSS,<br>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 |

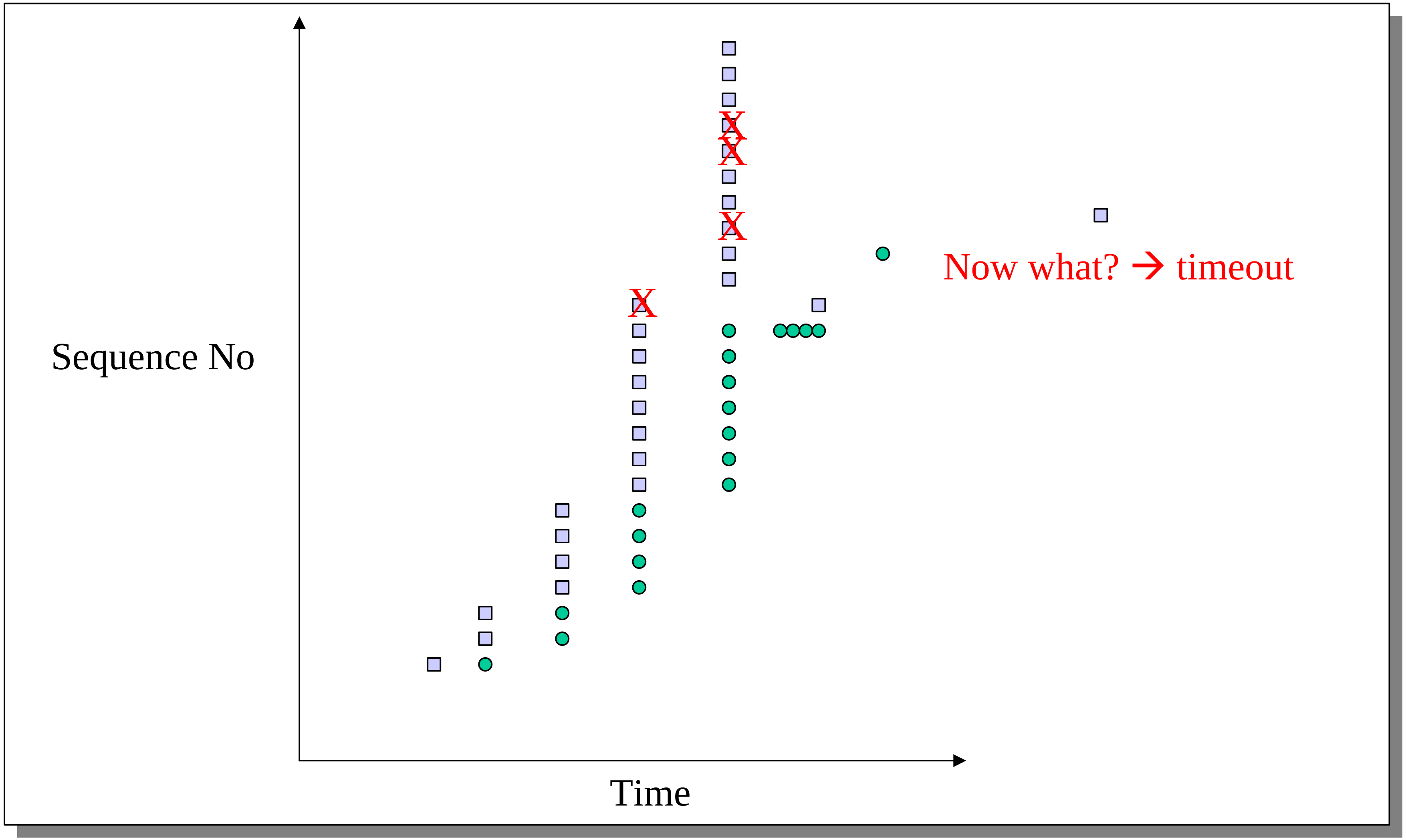
# Multiple losses



# Tahoe



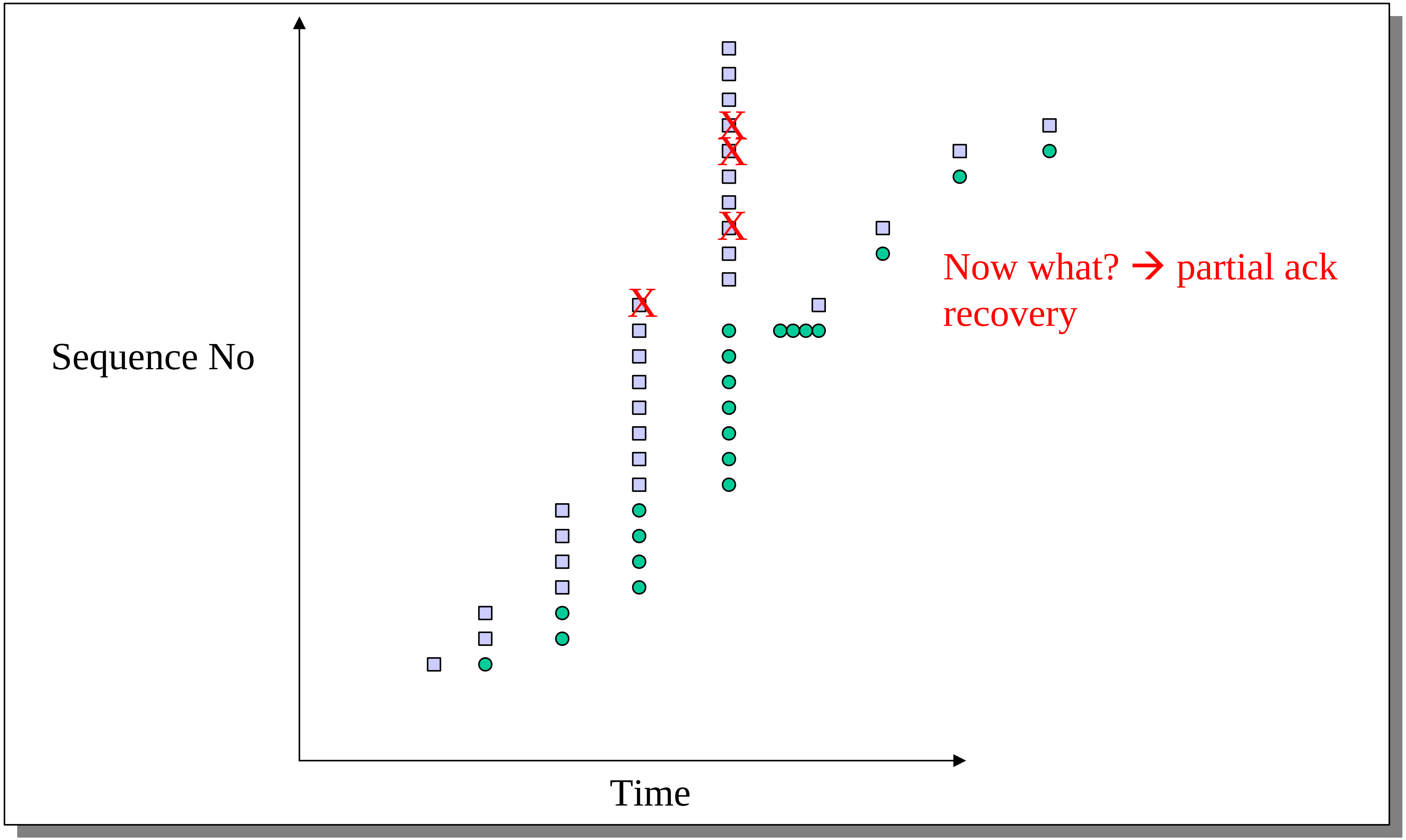
# Reno



# NewReno

- The ack that arrives after retransmission (partial ack) should indicate that a second loss occurred
- When does NewReno timeout?
  - When there are fewer than three dupacks for first loss
  - When partial ack is lost
- How fast does it recover losses?
  - One per RTT

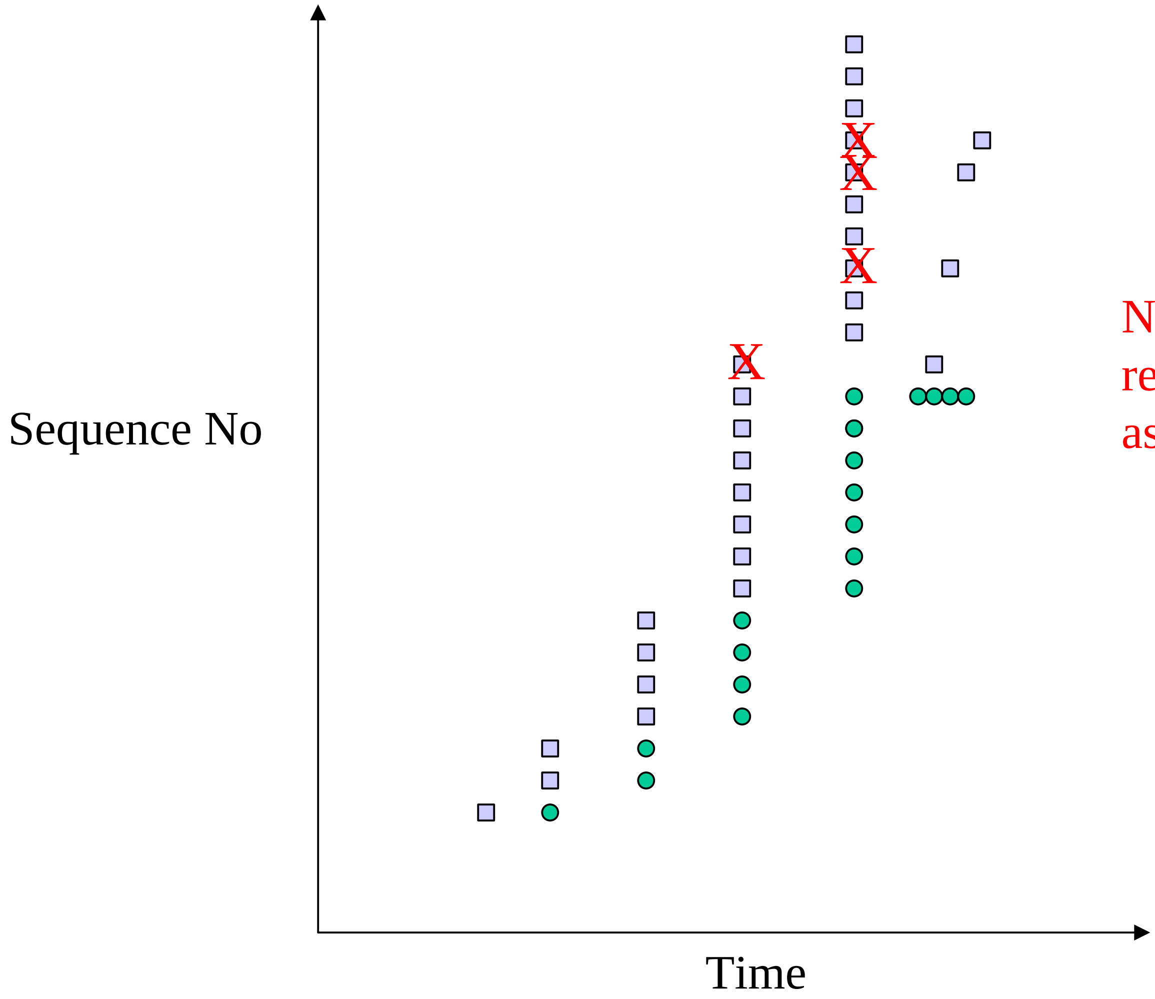
# NewReno



# SACK

- Basic problem is that cumulative acks only provide little information
  - Ack for just the packet received
    - What if acks are lost? → carry cumulative also
    - Not used
  - Bitmask of packets received
    - Selective acknowledgement (SACK)
- How to deal with reordering

# SACK (cont.)




Now what? – send retransmissions as soon as detected

# Performance issues

- Timeout >> fast rexmit
  - Need 3 dupacks/sacks
  - Not great for small transfers
    - Don't have 3 packets outstanding
  - What are real loss patterns like?
- Right edge recovery
  - Allow packets to be sent on arrival of first and second duplicate ack
  - Helps recovery for small windows
- How to deal with reordering?

# TCP extensions

- Implemented using TCP options
  - Timestamp
  - Protection from sequence number wraparound
  - Large windows

# Protection from wraparound

- Wraparound time vs. link speed
  - 1.5 Mbps: 6.4 hours
  - 10 Mbps: 57 minutes
  - 45 Mbps: 13 minutes
  - 100 Mbps: 6 minutes
  - 622 Mbps: 55 seconds → < MSL!
  - 1.2 Gbps: 28 seconds
- Use timestamp to distinguish sequence number wraparound

# Large windows

- Delay-bandwidth product for 100 ms delay
  - 1.5 Mbps: 18 KB
  - 10 Mbps: 122 KB > max 16 bit window
  - 45 Mbps: 549 KB
  - 100 Mbps: 1.2 MB
  - 622 Mbps: 7.4 MB
  - 1.2 Gbps: 14.8 MB
- Scaling factor on advertised window
  - Specifies how many bits window must be shifted to the left
  - Scaling factor exchanged during connection setup

# Maximum segment size (MSS)

- Exchanged at connection setup
  - Typically pick MTU of local link
- What all does this effect?
  - Efficiency
  - Congestion control
  - Retransmission
- Path MTU discovery
  - Why should MTU match MSS?

# Effects of TCP latencies

Q: client latency for object request from WWW server to receipt?

- TCP connection establishment
- Data transfer delay

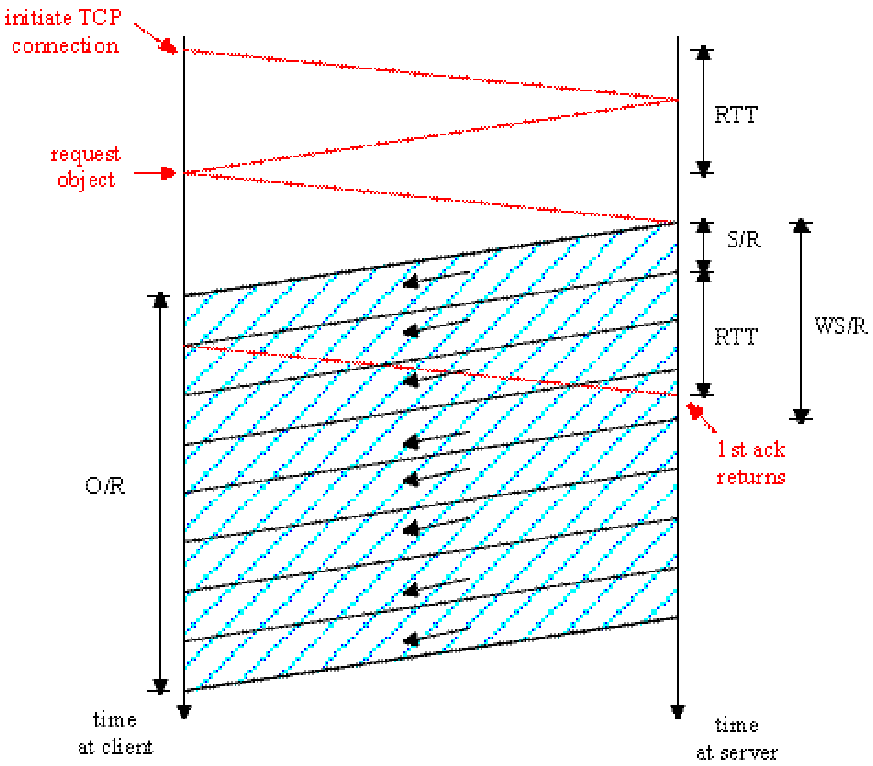
Notation, assumptions:

- Assume: fixed congestion window, W, giving throughput of R bps
- S: MSS (bits)
- O: object/file size (bits)
- No retransmissions (no loss, no corruption)

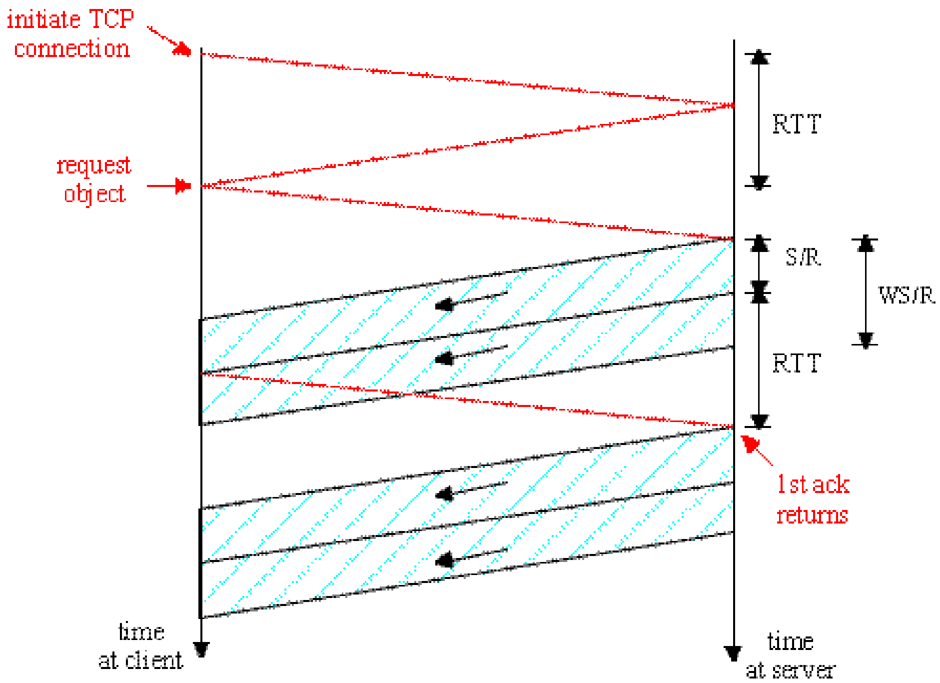
Two cases to consider:

- Case 1: WS/R > RTT + S/R: ACK for first segment in window before window’s worth of data sent
- Case 2: WS/R < RTT + S/R: wait for ACK after sending window’s worth of data sent

# Effects of TCP latencies (cont.)




Case 1: latency = 2RTT + O/R




Case 2: latency = 2RTT + O/R
+ (K-1)[S/R + RTT - WS/R]
K = O/(WS) (number of windows to transmit object)

# Transport layer: Summary

- Principles behind transport layer services:
  - Multiplexing/demultiplexing
  - Reliable data transfer
  - Flow control
  - Congestion control
- Instantiation and implementation in the Internet
  - UDP
  - TCP

Next:

- Leaving the network "edge" (application transport layer)
- Into the network "core"