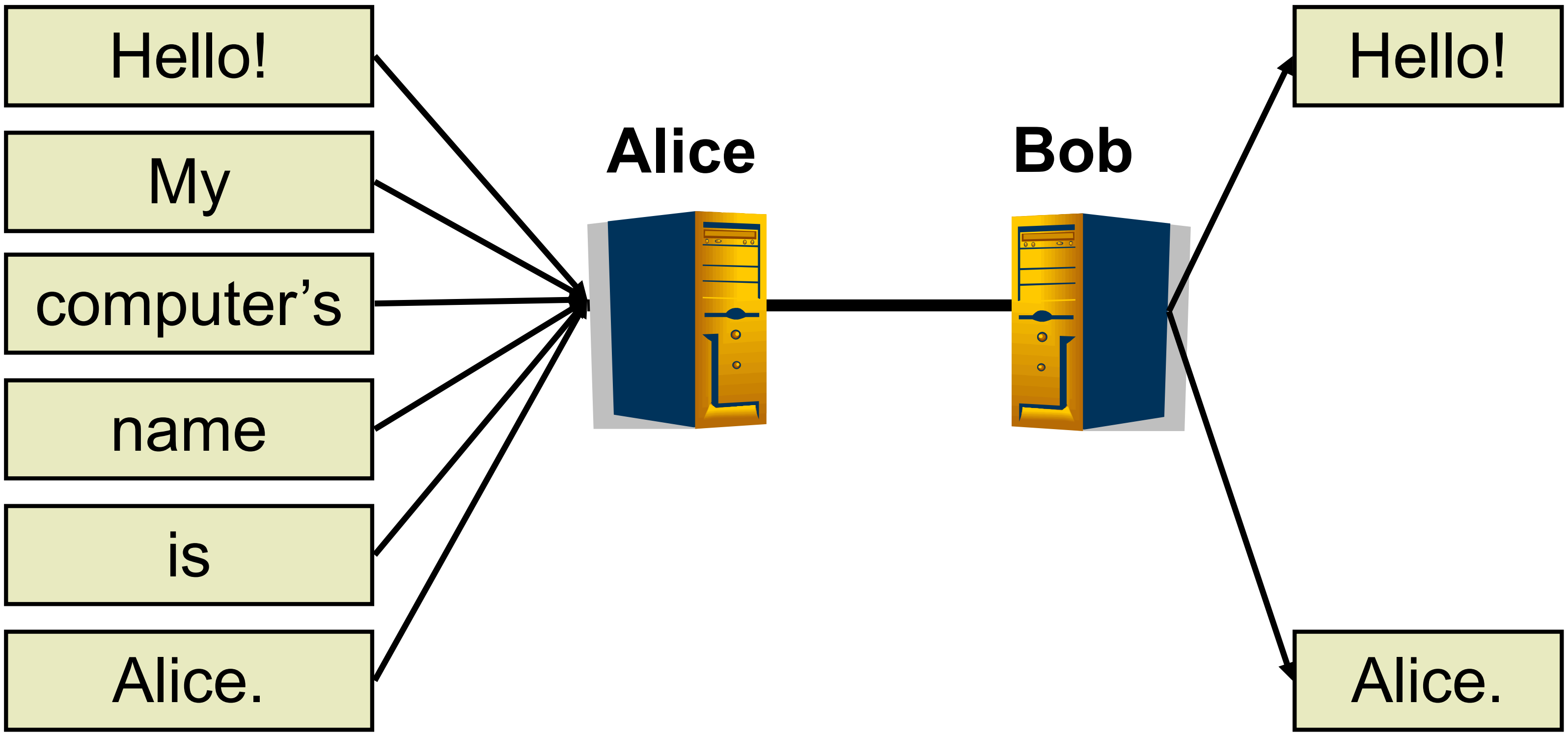
CS/ECE 439: Wireless Networking

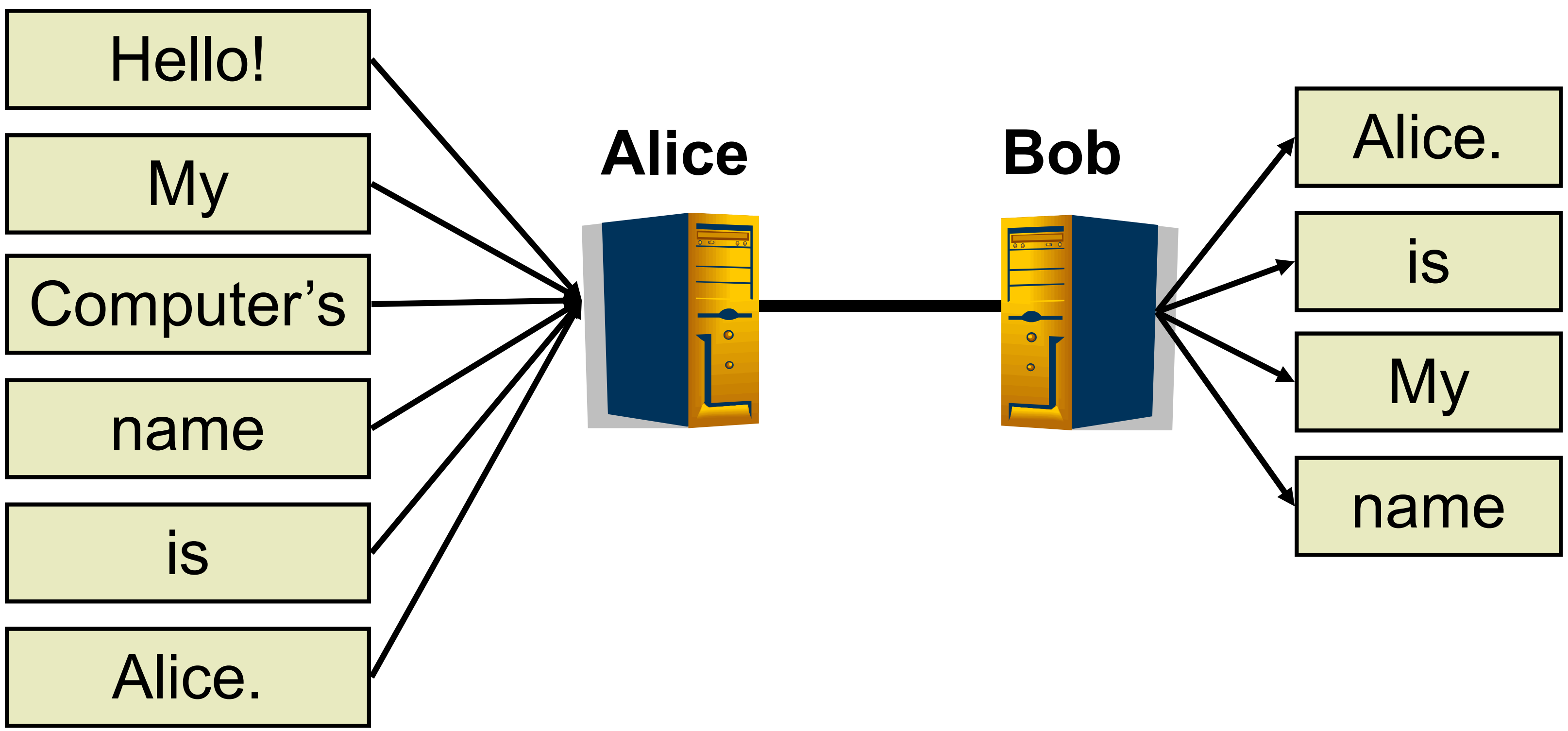
Transport Layer – dealing with errors and unreliability

 Fall 2020

Reliable Transmission



Reliable Transmission



Reliable Transmission

- Suppose error protection identifies valid and invalid packets
  - How?
- Can we make the channel appear reliable?
  - Insure packet delivery
  - Maintain packet order
  - Provide reliability at full link capacity

Reliable Transmission Outline

- Fundamentals of Automatic Repeat reQuest (ARQ) algorithms
  - A family of algorithms that provide reliability through retransmission
- ARQ algorithms (simple to complex)
  - stop-and-wait
  - sliding window
    - go-back-n
    - selective repeat

Terminology

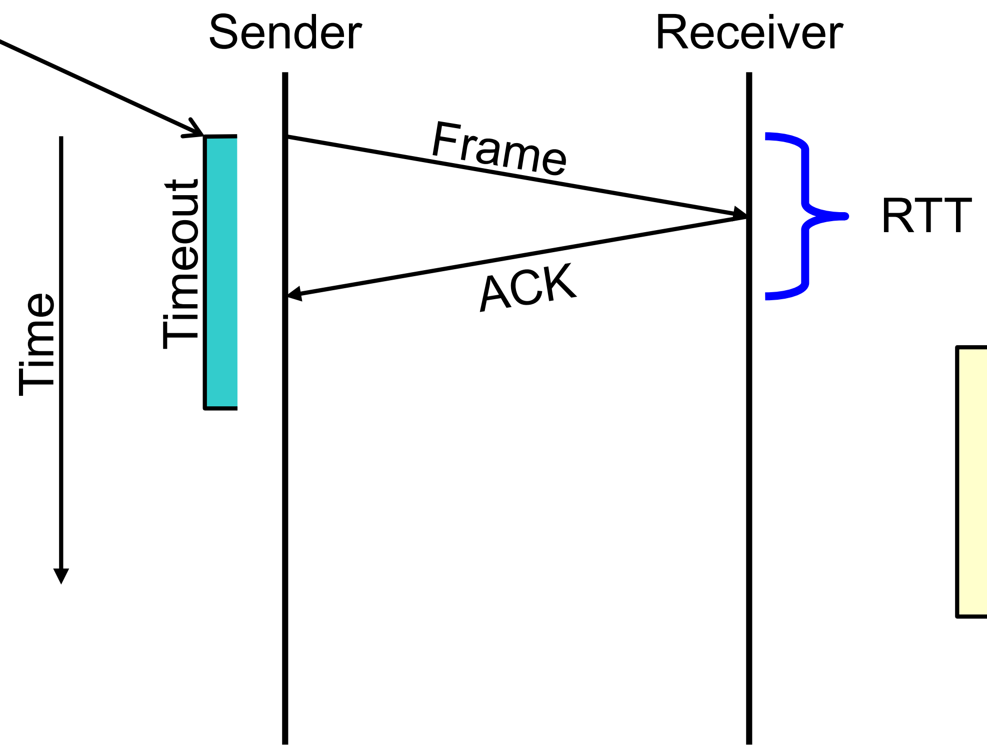
- Acknowledgement (ACK)
  - Receiver tells the sender when a frame is received
    - Selective acknowledgement (SACK)
      - Specifies set of frames received
    - Cumulative acknowledgement (ACK)
      - Have received specified frame and all previous
- Timeout (TO)
  - Sender decides the frame (or ACK) was lost
  - Sender can try again

Stop-and-Wait

- Basic idea
  1. Send a frame
  2. Wait for an ACK or TO
  3. If TO, go to 1
  4. If ACK, get new frame, go to 1

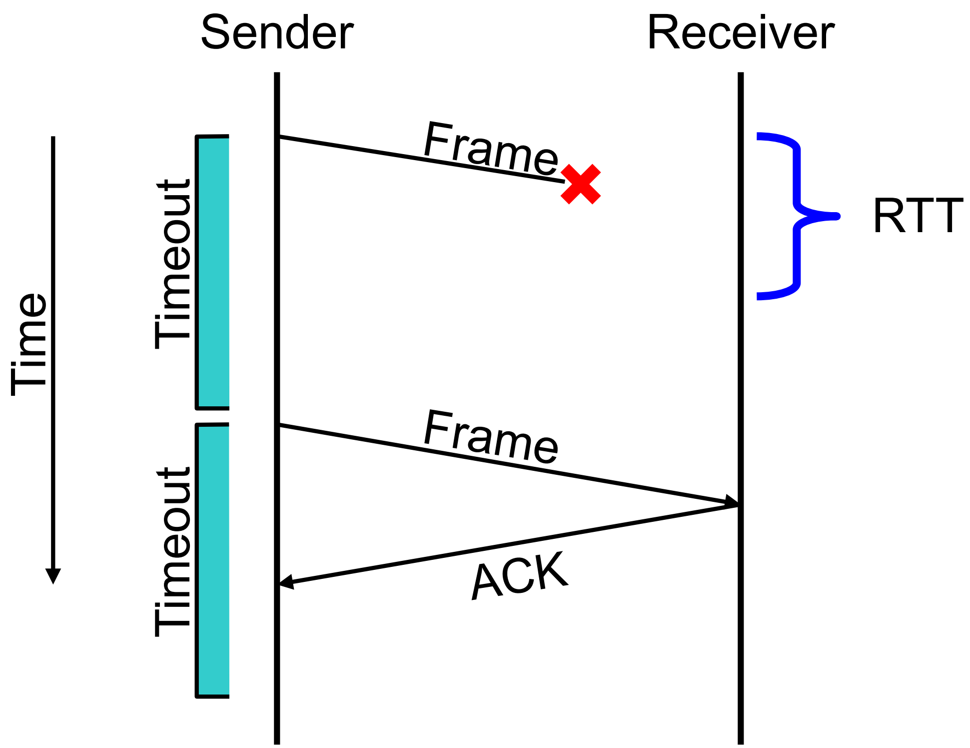
Stop-and-Wait: Success

How long should the timeout be?

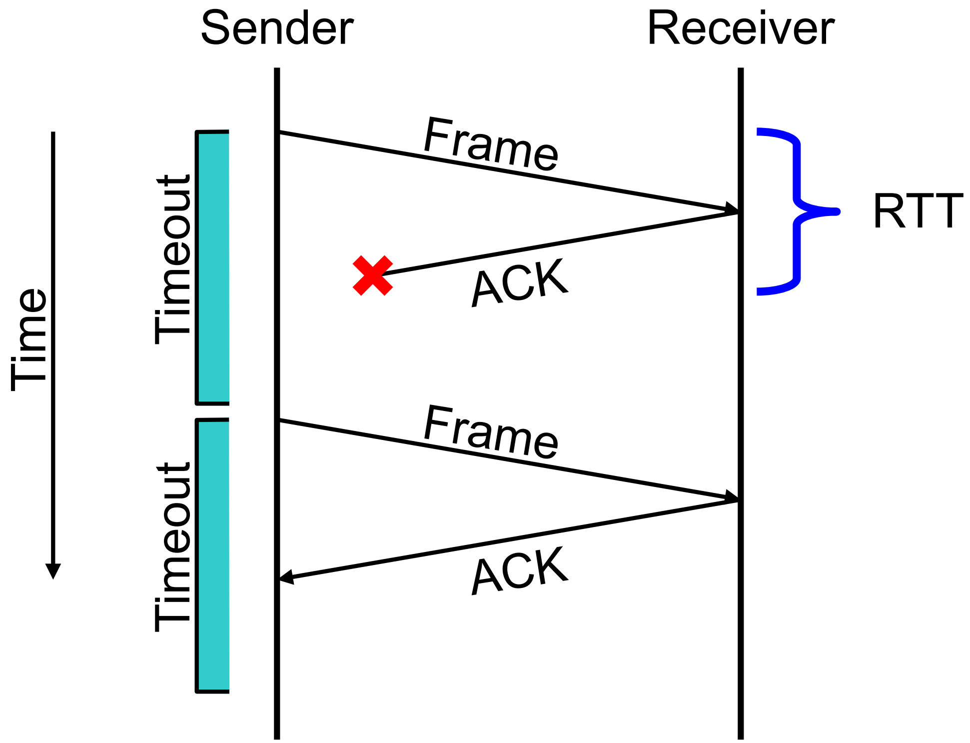


What can go wrong?
How will it affect our protocol?

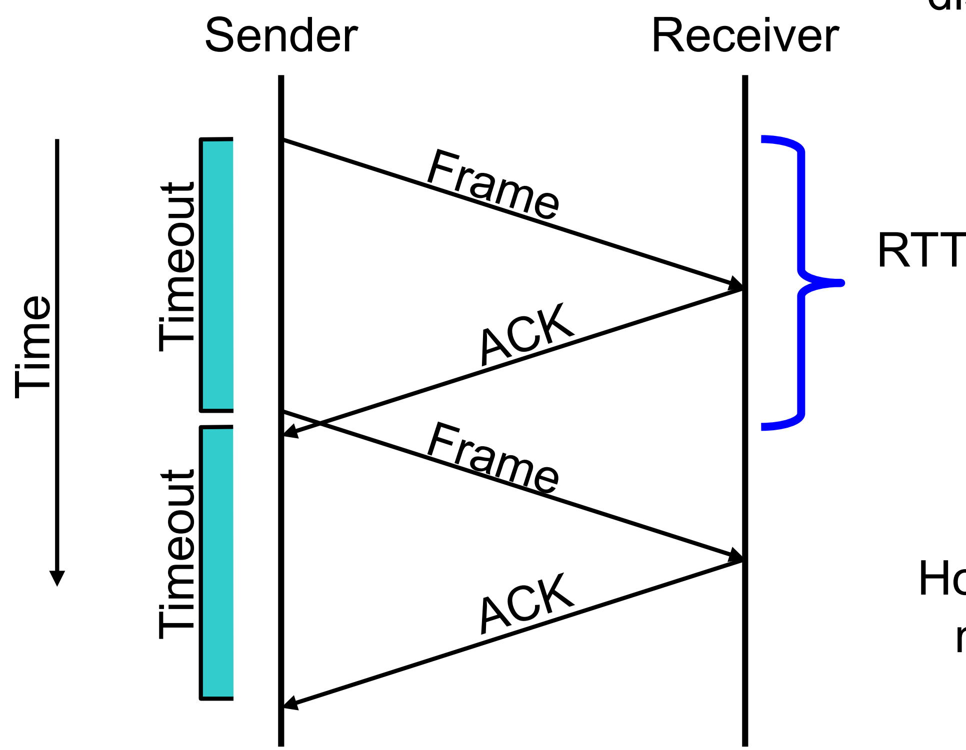
Stop-and-Wait: Lost Frame



Stop-and-Wait: Lost ACK



Stop-and-Wait: Delayed Frame



How can receiver distinguish between two frames?

How many bits do you need for sequence numbers?

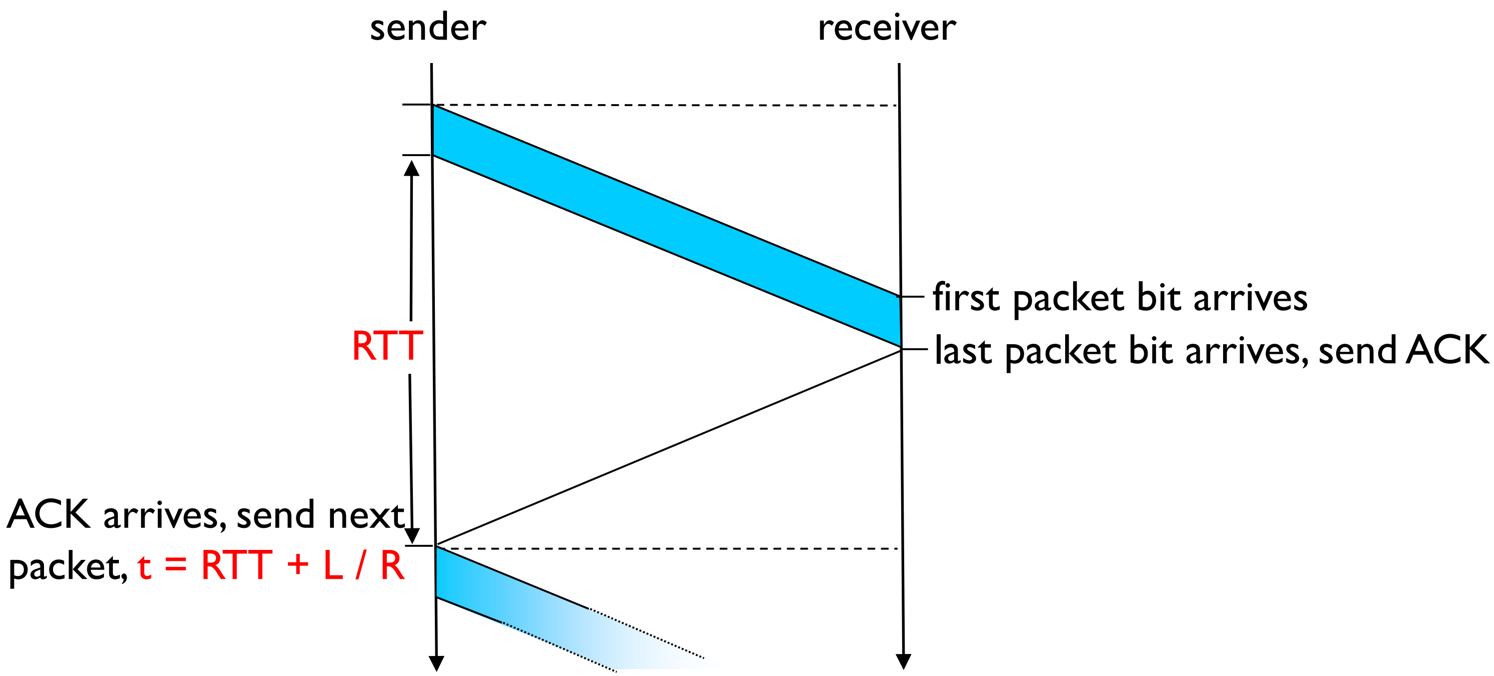
Stop-and-Wait

- Goal
  - Guaranteed, at-most-once delivery
- Protocol Challenges
  - Dropped frame/ACK
  - Duplicate frame/ACK
- Requirements
  - 1-bit sequence numbers (if physical network maintains order)
    - sender tracks frame ID to send
    - receiver tracks next frame ID expected

Stop-and-Wait

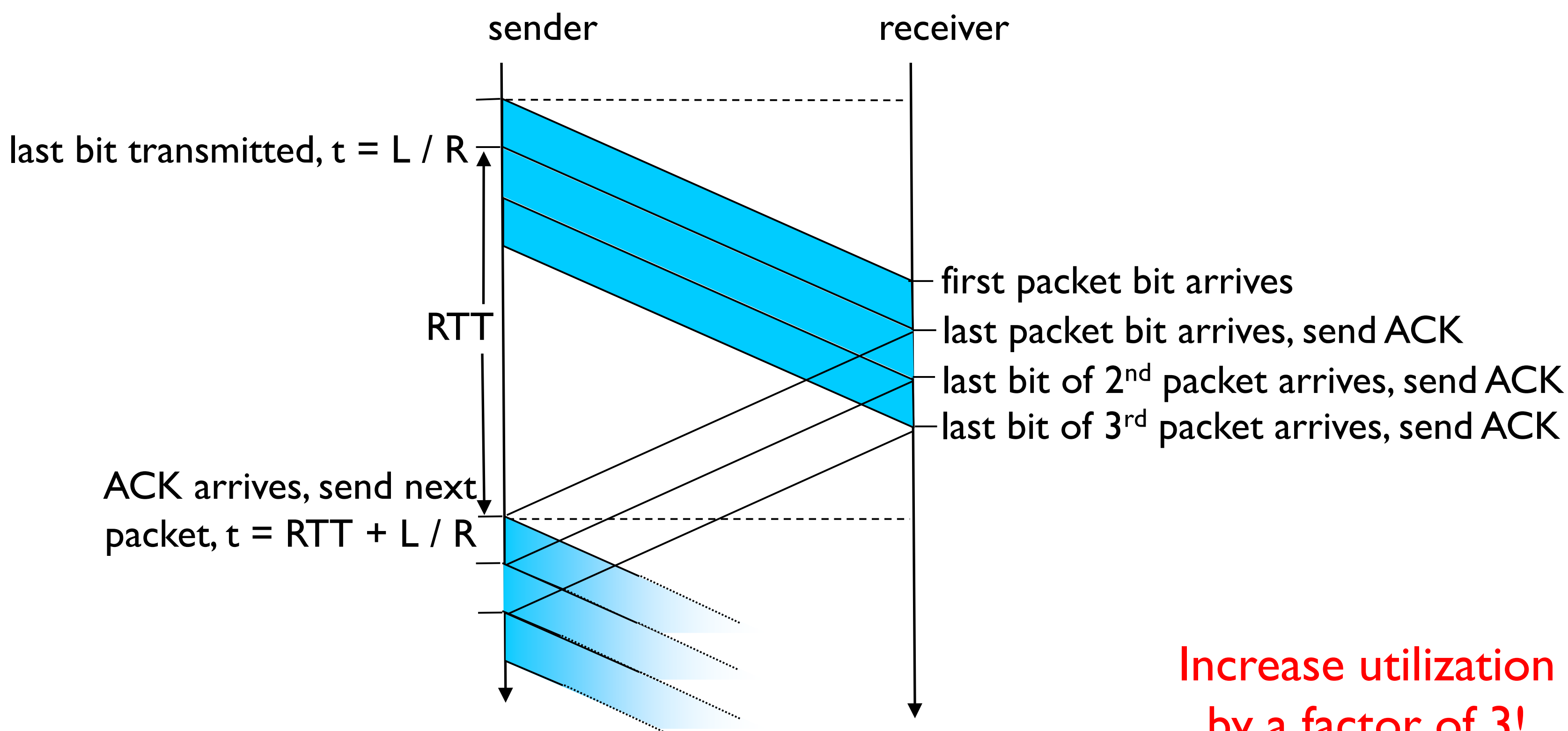
- We have achieved
  - Frames delivered reliably and in order
  - Is that enough?
- Problem
  - Only allows one outstanding frame
    - Does not keep the pipe full
  - Example
    - 100ms RTT
    - One frame per RTT = 1KB
    - 1024x8x10 = 81920 kbps
    - Regardless of link bandwidth!

Stop-and-Wait



$$U_{sender} = \frac{L / R}{RTT + L / R}$$

Keeping the Pipe Full

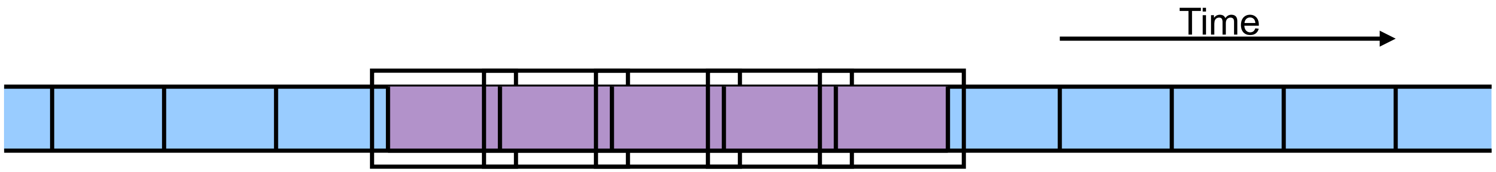


Increase utilization by a factor of 3!

$$U_{sender} = \frac{3 * L / R}{RTT + L / R}$$

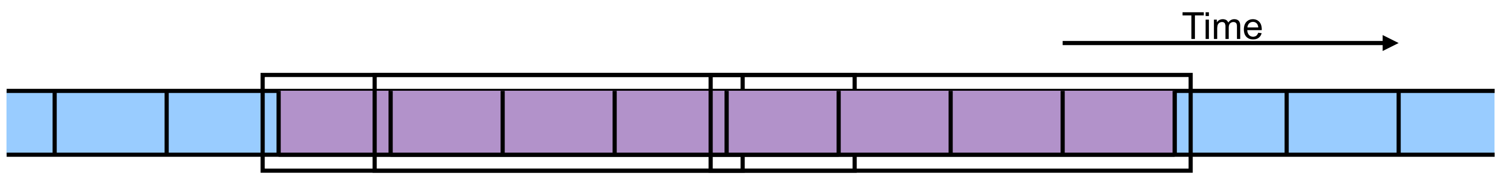
Concepts

- Consider an ordered stream of data frames
- Stop-and-Wait
  - Window of one frame
  - Slides along stream over time



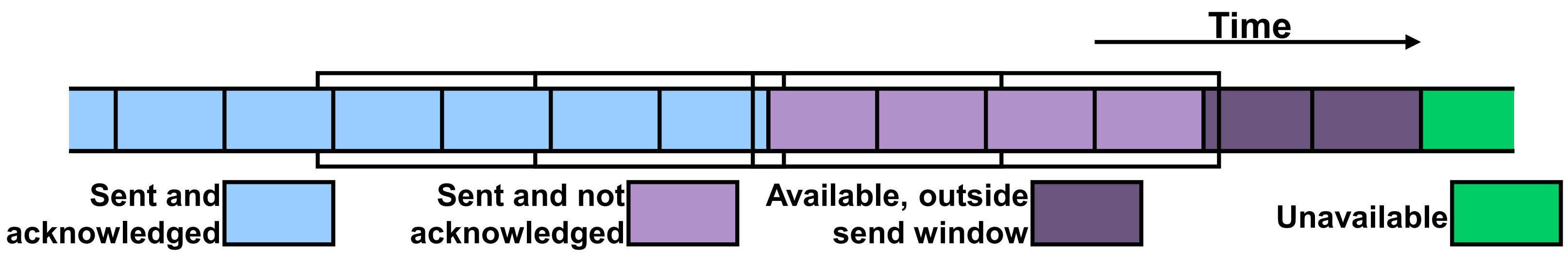
Concepts

- Sliding Window Protocol
  - Multiple-frame send window
  - Multiple frame receive window



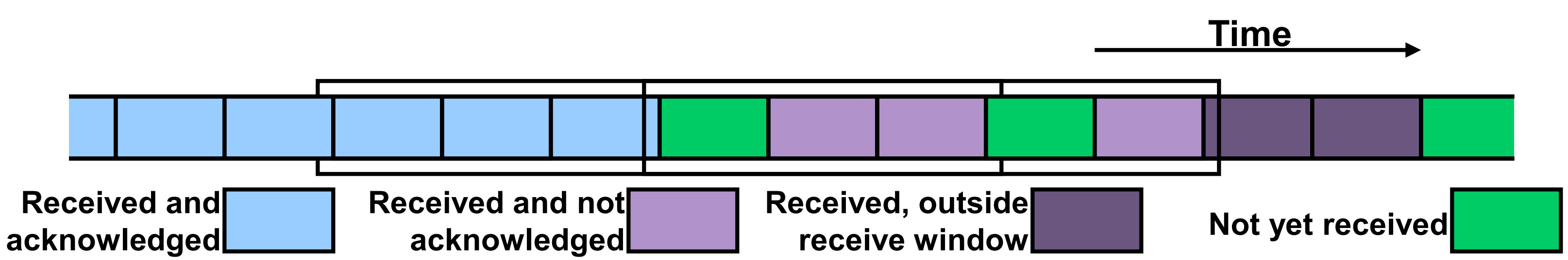
Sliding Window

- Send Window
  - Fixed length
  - Starts at earliest unacknowledged frame
  - Only frames in window are active



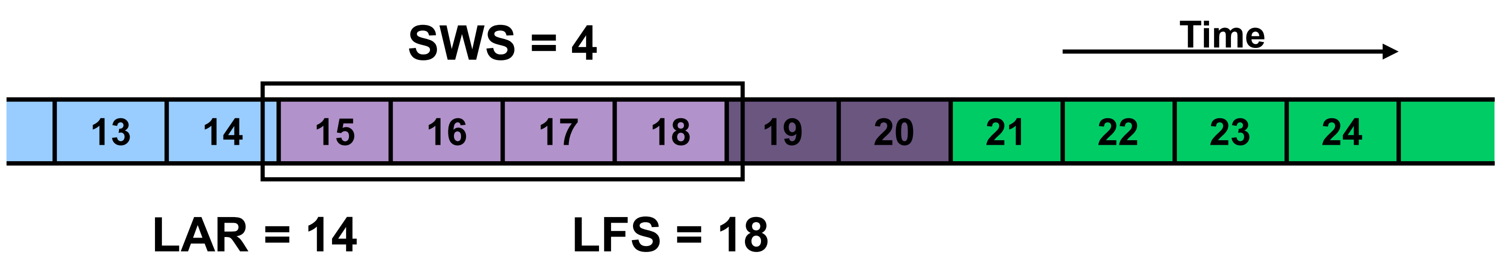
Sliding Window

- Receive Window
  - Fixed length (unrelated to send window)
  - Starts at earliest frame not received
  - Only frames in window accepted



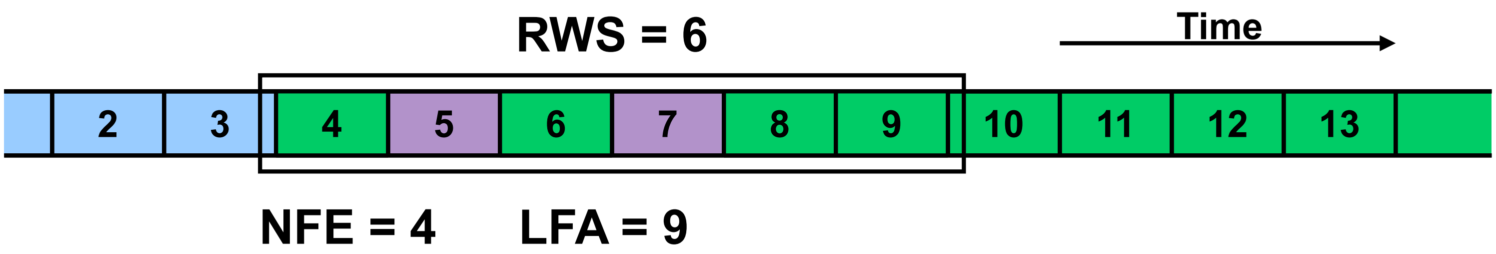
Sliding Window Terminology

- Sender Parameters
  - Send Window Size (SWS)
  - Last Acknowledgement Received (LAR)
  - Last Frame Sent (LFS)



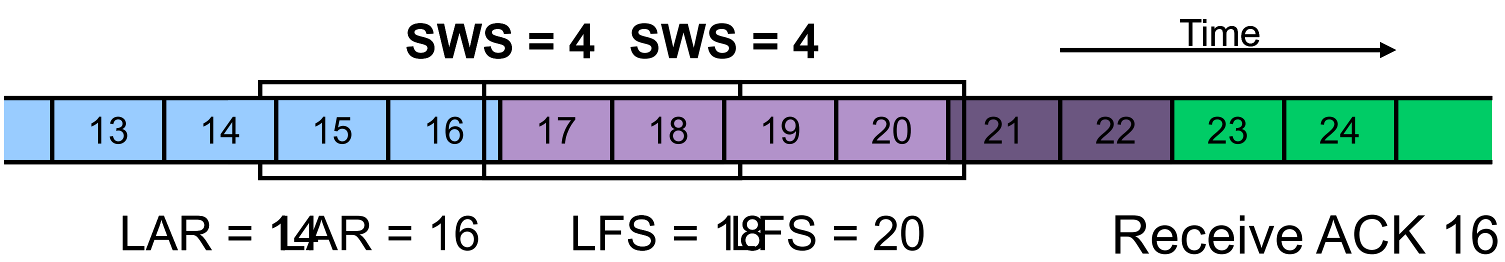
Sliding Window Terminology

- Receiver Parameters
  - Receive Window Size (RWS)
  - Next Frame Expected (NFE)
  - Last Frame Acceptable (LFA)



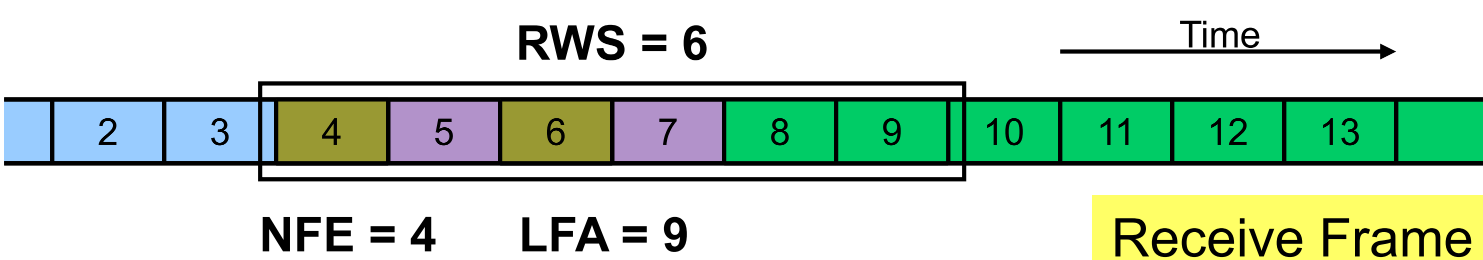
Sliding Window Details

- Sender Tasks
  - Assign sequence numbers
  - On ACK Arrival
    - Advance LAR
    - Slide window



Sliding Window Details

- Receiver Tasks
  - On Frame Arrival (N)
    - Silently discard if outside of window
      - N < NFE (NACK possible, too)
      - N >= NFE + RWS
    - Send cumulative ACK if within window

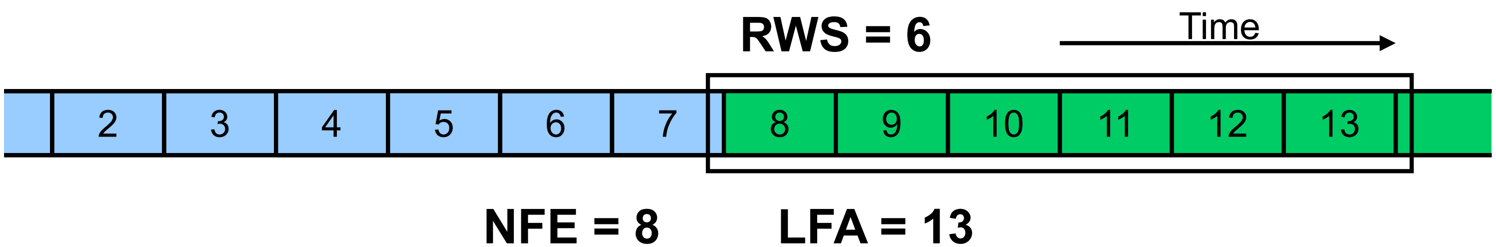


Receive Frame 4

Send ACK 7

Sliding Window Details

- Receiver Tasks
  - On Frame Arrival (N)
    - Silently discard if outside of window
      - N < NFE (NACK possible, too)
      - N >= NFE + RWS
    - Send cumulative ACK if within window



Sliding Window Details

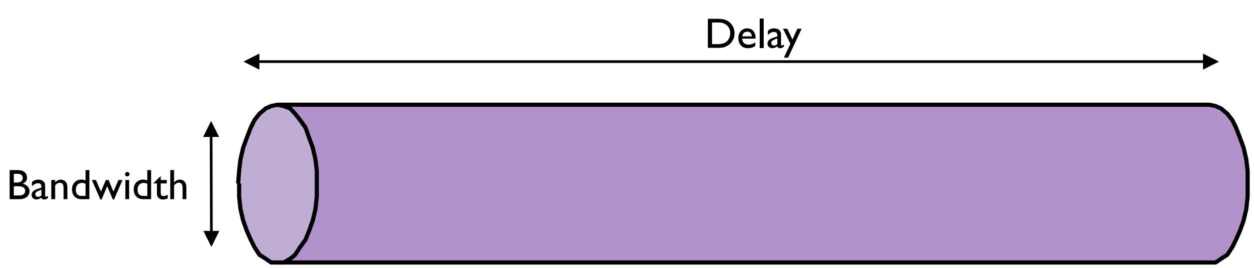
- Sequence number space
  - Finite number, so wrap around
  - Need space larger than SWS (outstanding frames)
    - In fact, need twice as large

Window Sizes

- How big should we make SWS?
  - Compute from delay x bandwidth
- How big should we make RWS?
  - Depends on buffer capacity of receiver

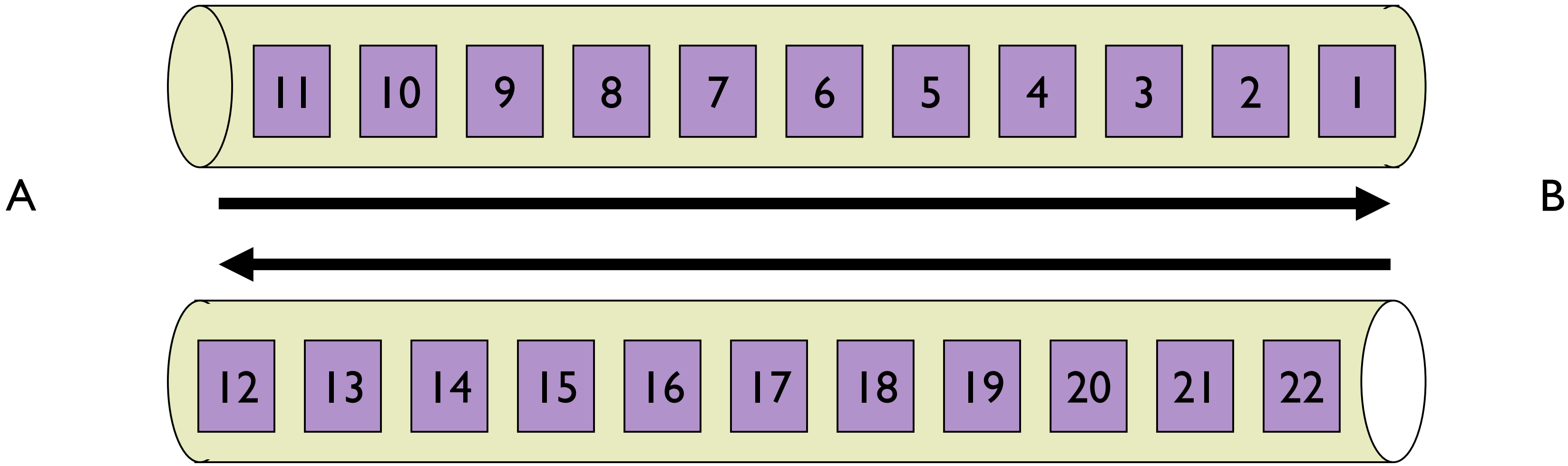
Delay x Bandwidth Product - Revisited

- Amount of data in "pipe"
  - channel = pipe
  - delay = length
  - bandwidth = area of a cross section
  - bandwidth x delay product = volume



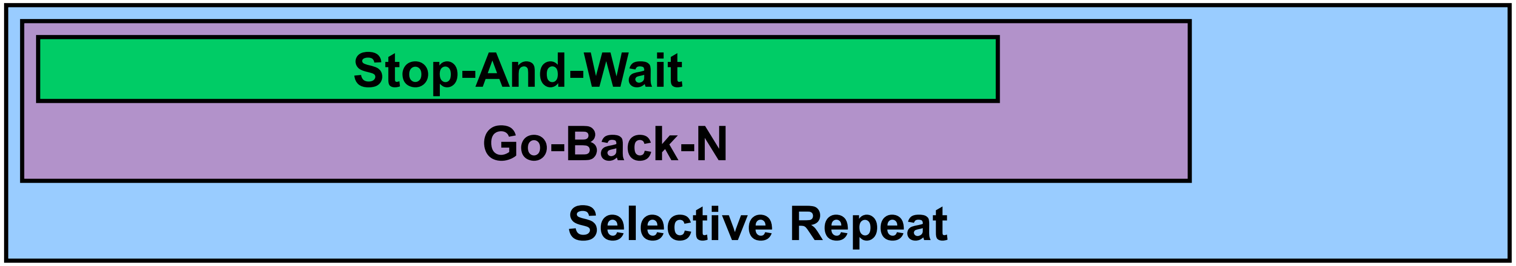
Delay x Bandwidth Product

- Bandwidth x delay product
  - How many bits the sender must transmit before the first bit arrives at the receiver if the sender keeps the pipe full
  - Takes another one-way latency to receive a response from the receiver



ARQ Algorithm Classification

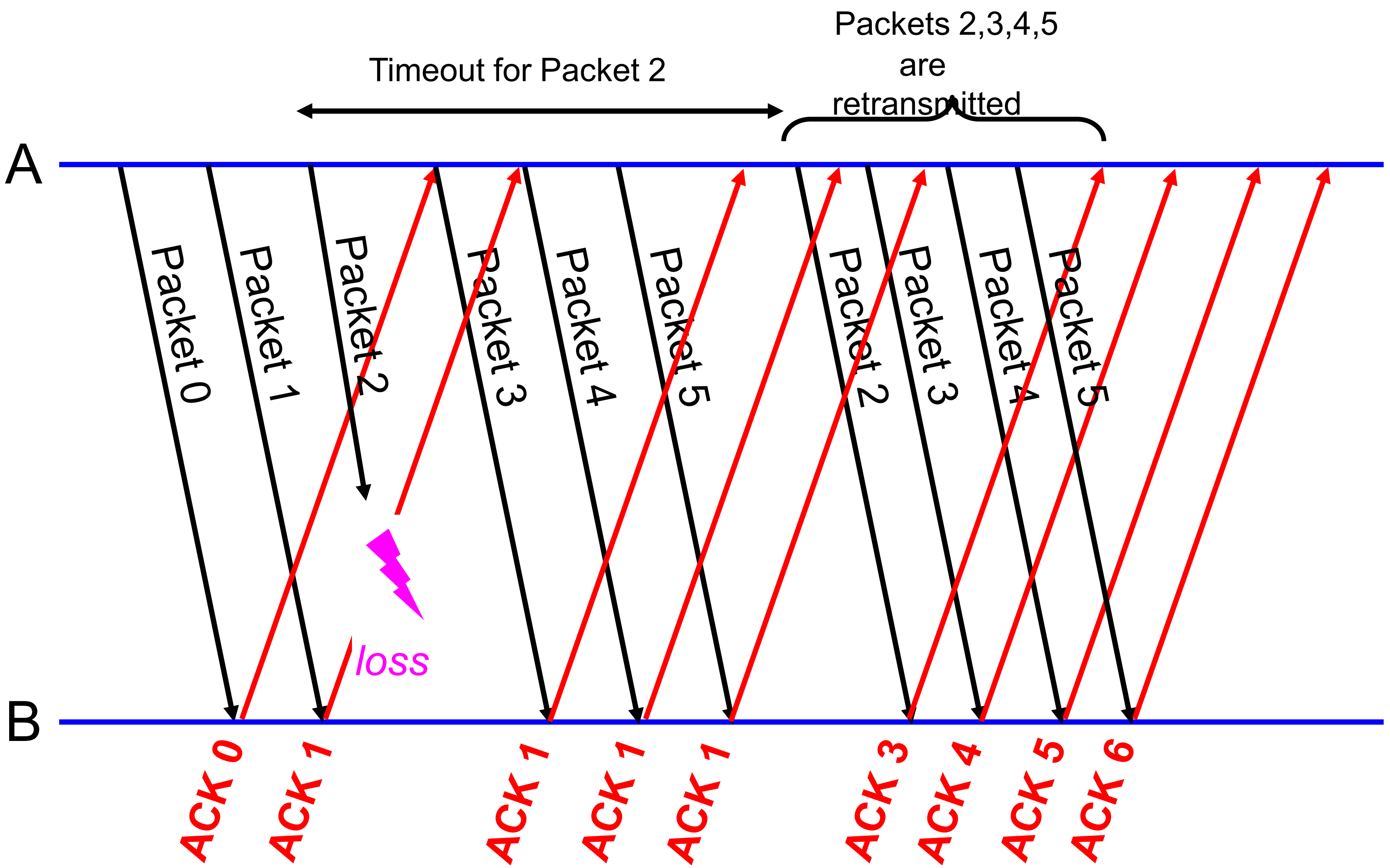
- Three Types:
  - Stop-and-Wait: SWS = 1 RWS = 1
  - Go-Back-N: SWS = N RWS = 1
  - Selective Repeat: SWS = N RWS = M
    - Usually M = N



Sliding Window Variations: Go-Back-N

- SWS = N, RWS = 1
- Receiver only buffers one frame
- If a frame is lost, the sender may need to retransmit up to N frames
  - i.e., sender "goes back" N frames
- Variations
  - How long is the frame timeout?
  - Does receiver send NACK for out-of-sequence frame?

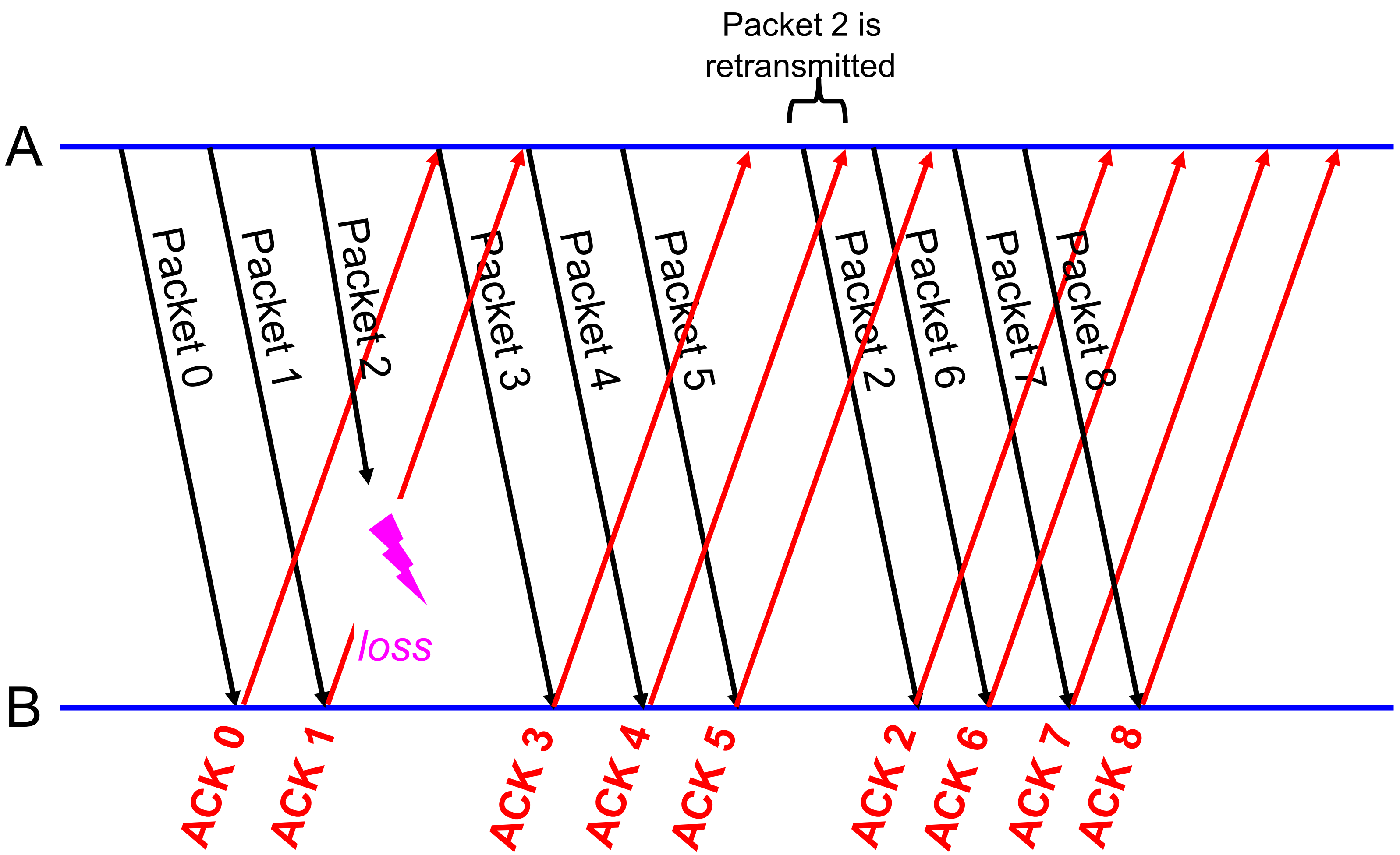
Go-Back-N: Cumulative ACKs



Sliding Window Variations: Selective Repeat

- SWS = N, RWS = M
- Receiver individually acknowledges all correctly received frames
  - Buffers up to M frames, as needed, for eventual in-order delivery to upper layer
- If a frame is lost, sender must only resend
  - Frames lost within the receive window
- Variations
  - How long is the frame timeout?
  - Use cumulative or per-frame ACK?
  - Does protocol adapt timeouts?
  - Does protocol adapt SWS and/or RWS?

Selective Repeat



Roles of a Sliding Window Protocol

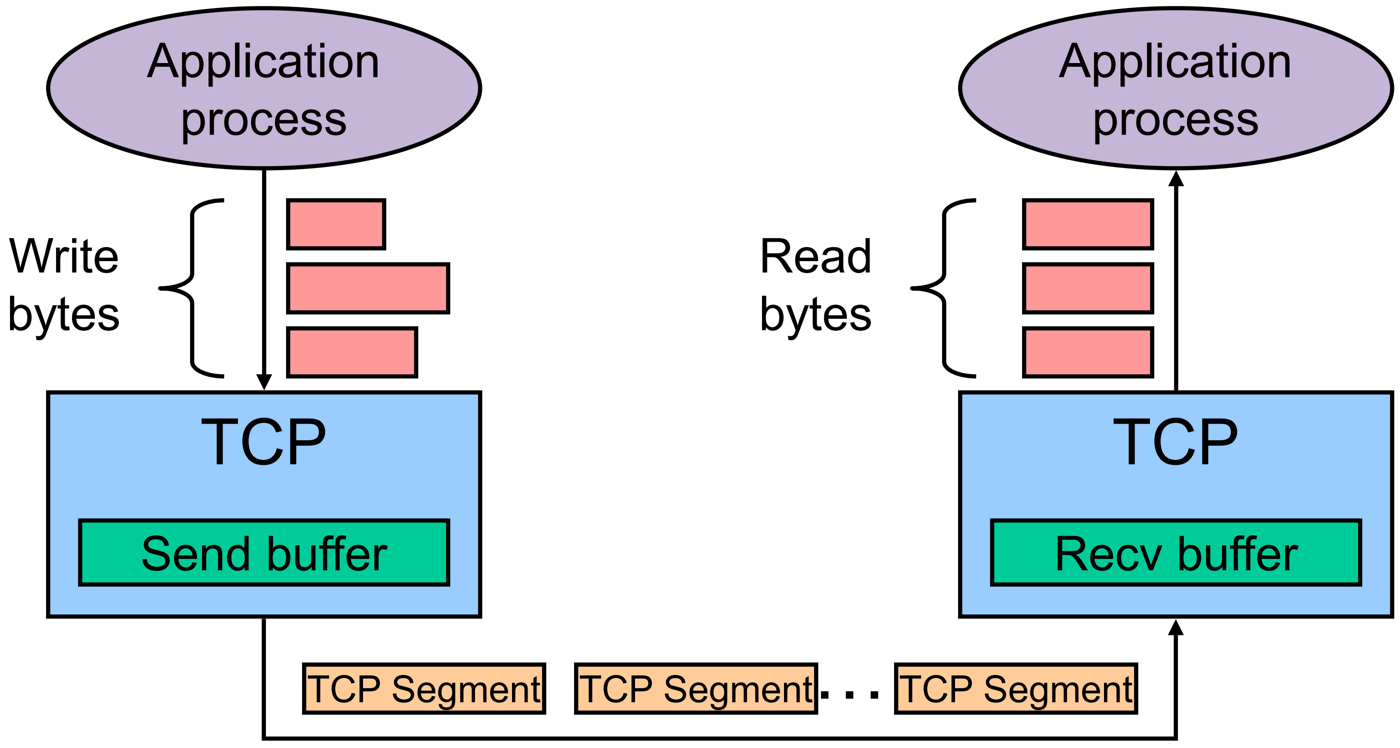
- Reliable delivery on an unreliable link
  - Core function
- Preserve delivery order
  - Controlled by the receiver
- Flow control
  - Allow receiver to throttle sender

- Separation of Concerns
  - Must be able to distinguish between different functions that are sometimes rolled into one mechanism

TCP Data Transport

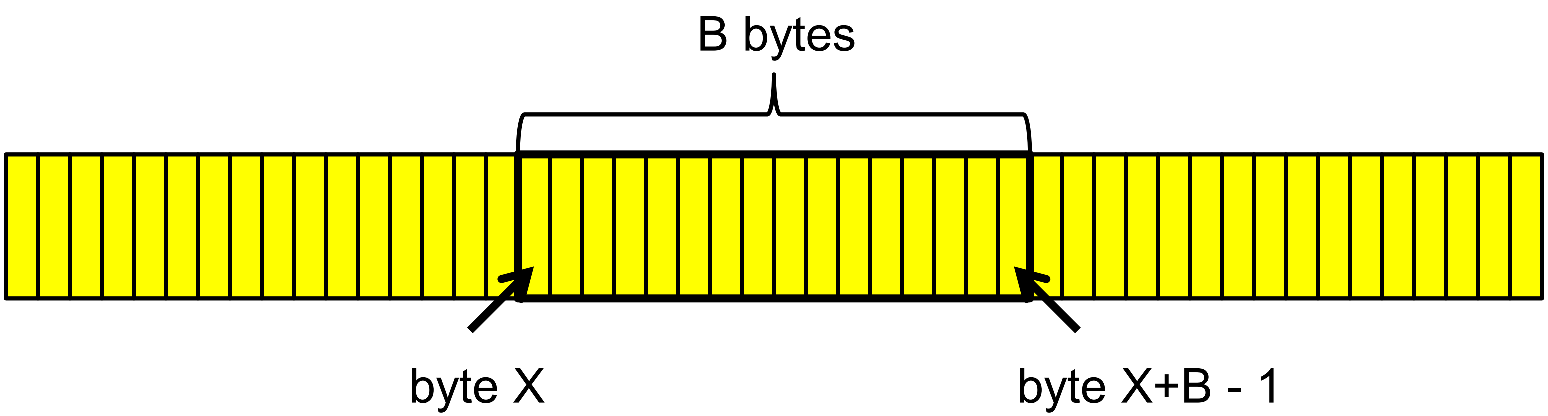
- Data broken into segments
  - Limited by maximum segment size (MSS)
  - Defaults to 352 bytes
  - Negotiable during connection setup
  - Typically set to
    - MTU of directly connected network – size of TCP and IP headers
- Three events cause a segment to be sent
  - ≥ MSS bytes of data ready to be sent
  - Explicit PUSH operation by application
  - Periodic timeout

TCP Byte Stream



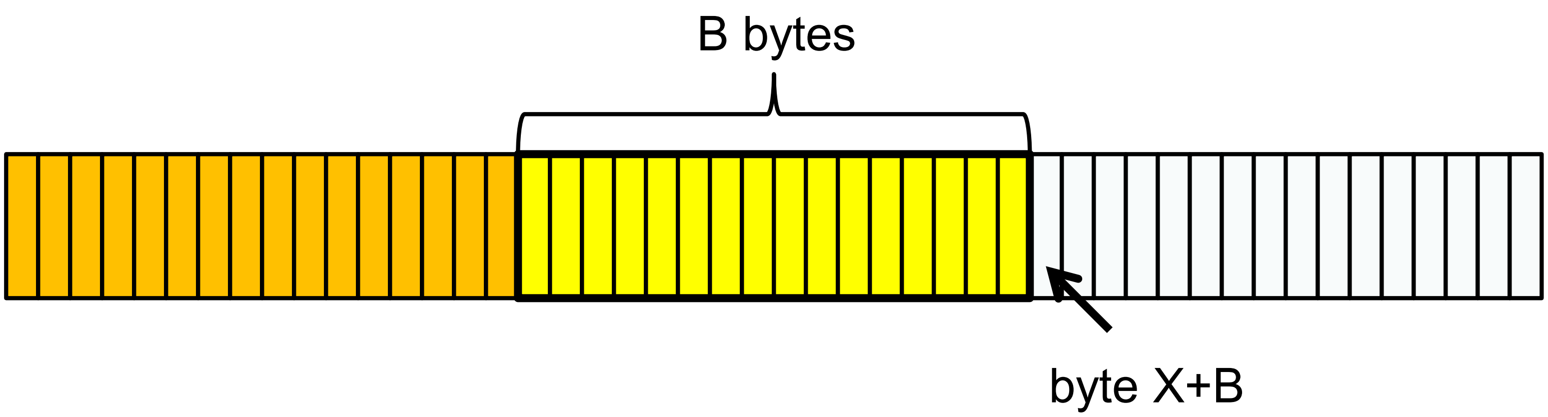
ACKing and Sequence Numbers

- Sender sends packet
  - Data starts with sequence number X
  - Packet contains B bytes
    - X, X+1, X+2, ….X+B-1



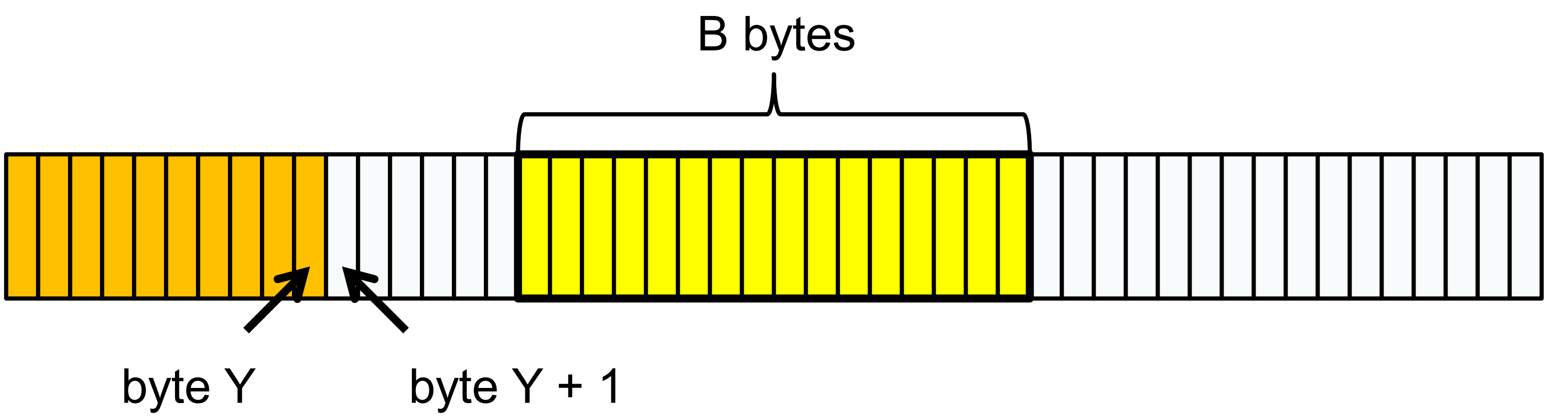
ACKing and Sequence Numbers

- Upon receipt of packet, receiver sends an ACK
  - If all data prior to X already received:
    - ACK acknowledges X+B (because that is next expected byte)



ACKing and Sequence Numbers

- Upon receipt of packet, receiver sends an ACK
  - If highest byte already received is some smaller value Y
    - ACK acknowledges Y+1
    - Even if this has been ACKed before



TCP Sliding Window Protocol

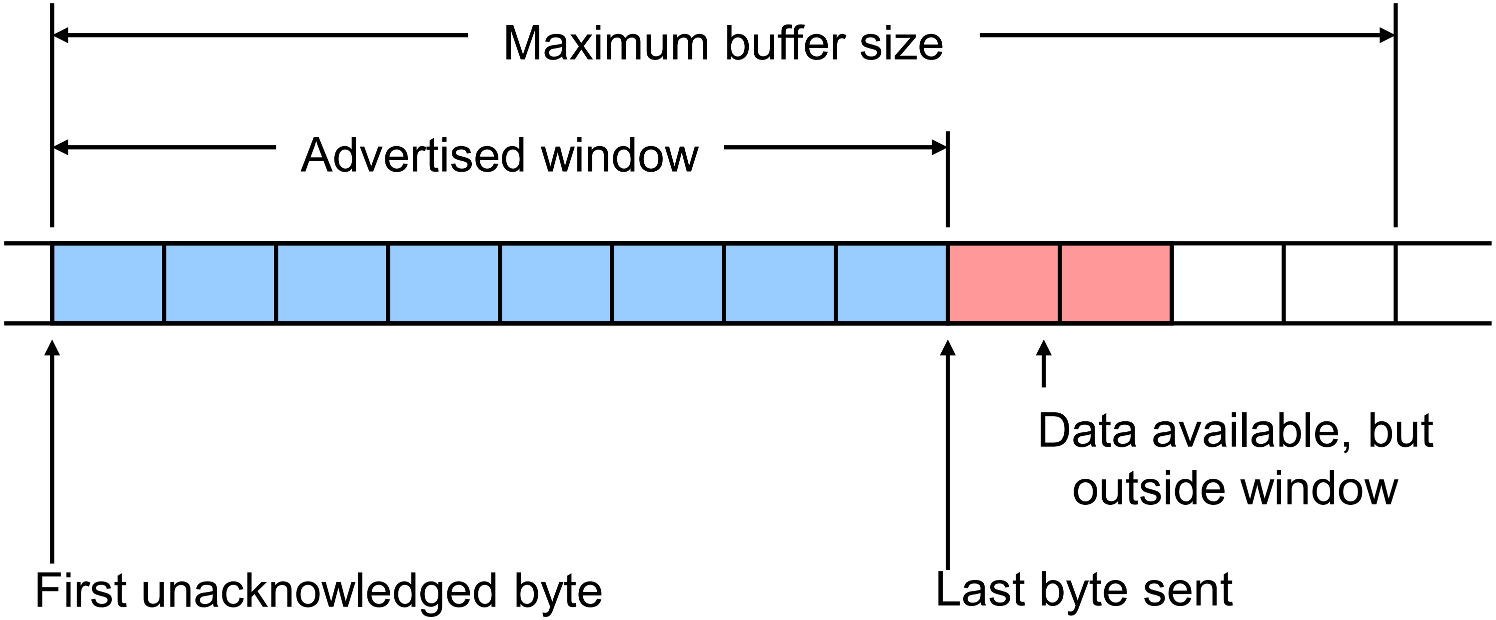
- Sequence numbers
  - Indices into byte stream
- ACK sequence number
  - Actually next byte expected as opposed to last byte received

TCP Sliding Window Protocol

- Advertised window
  - Enables dynamic receive window size
- Receive buffers
  - Data ready for delivery to application until requested
  - Out-of-order data to maximum buffer capacity
- Sender buffers
  - Unacknowledged data
  - Unsent data out to maximum buffer capacity

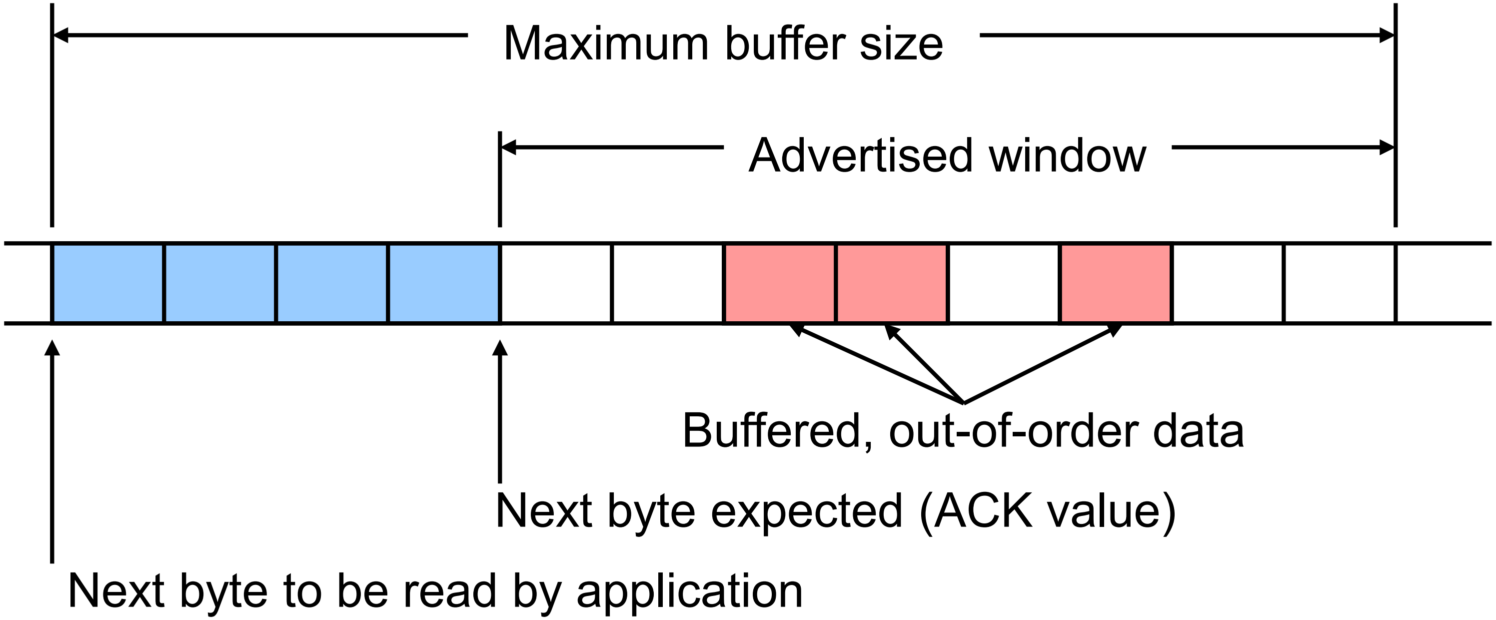
TCP Sliding Window Protocol – Sender Side

- **`LastByteAcked <= LastByteSent`**
- **`LastByteSent <= LastByteWritten`**
- Buffer bytes between **`LastByteAcked`** and **`LastByteWritten`**



TCP Sliding Window Protocol – Receiver Side

- **`LastByteRead < NextByteExpected`**
- **`NextByteExpected <= LastByteRcvd + 1`**
- Buffer bytes between **`NextByteRead`** and **`LastByteRcvd`**



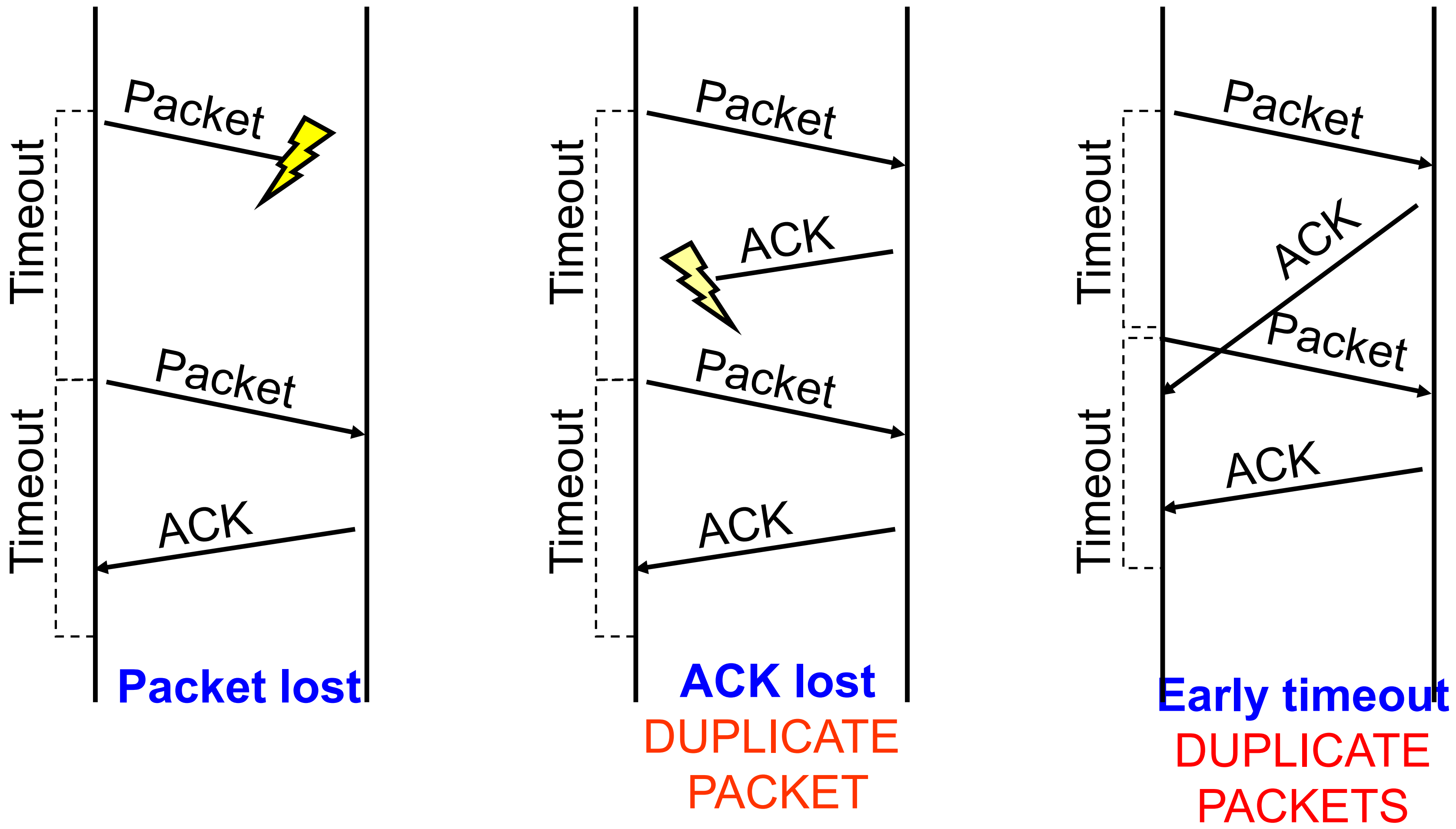
Flow Control vs. Congestion Control

- Flow control
  - Preventing senders from overrunning the capacity of the receivers
- Congestion control
  - Preventing too much data from being injected into the network, causing switches or links to become overloaded
- Which one does TCP provide?
- TCP provides both
  - Flow control based on advertised window
  - Congestion control discussed later in class

Advertised Window Limits Rate

- W = window size
  - Sender can send no faster than W/RTT bytes/sec
  - Receiver implicitly limits sender to rate that receiver can sustain
  - If sender is going too fast, window advertisements get smaller & smaller

Reasons for Retransmission



How Long Should Sender Wait?

- Sender sets a timeout to wait for an ACK
  - Too short
    - wasted retransmissions
  - Too long
    - excessive delays when packet lost

TCP Round Trip Time and Timeout

- How should TCP set its timeout value?
  - Longer than RTT
    - But RTT varies
  - Too short
    - Premature timeout
    - Unnecessary retransmissions
  - Too long
    - Slow reaction to segment loss

- Estimating RTT
  - SampleRTT
    - Measured time from segment transmission until ACK receipt
    - Will vary
    - Want smoother estimated RTT
  - Average several recent measurements
    - Not just current SampleRTT

TCP Congestion Control

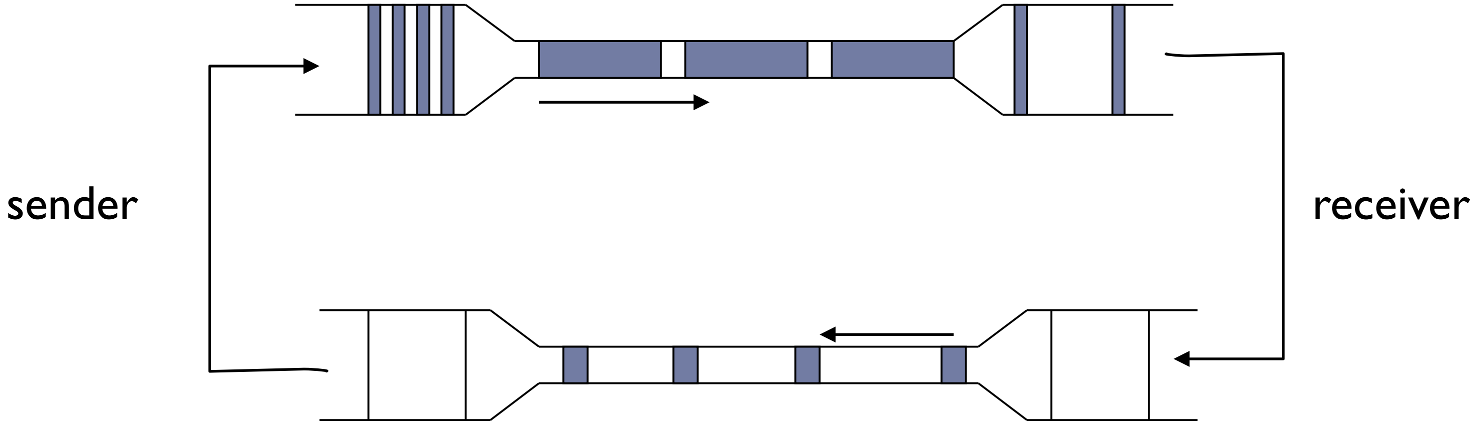
- Idea
  - Assumes best-effort network
  - Each source determines network capacity for itself
  - Implicit feedback
  - ACKs pace transmission (self-clocking)
- Challenge
  - Determining initial available capacity
  - Adjusting to changes in capacity in a timely manner

TCP Congestion Control

- Basic idea
  - Add notion of congestion window
  - Effective window is smaller of
    - Advertised window (flow control)
    - Congestion window (congestion control)
  - Changes in congestion window size
    - Slow increases to absorb new bandwidth
    - Quick decreases to eliminate congestion

TCP Congestion Control

- Specific strategy
  - Self-clocking
    - Send data only when outstanding data ACK'd
    - Equivalent to send window limitation mentioned



TCP Congestion Control

- Specific strategy
  - Self-clocking
    - Send data only when outstanding data ACK'd
    - Equivalent to send window limitation mentioned
  - Growth
    - Add one maximum segment size (MSS) per congestion window of data ACK'd
    - It's really done this way, at least in Linux:
      - see tcp_cong_avoid in tcp_input.c.
      - Actually, every ack for new data is treated as an MSS ACK'd
    - Known as additive increase

TCP Congestion Control

- Specific strategy (continued)
  - Decrease
    - Cut window in half when timeout occurs
    - In practice, set window = window /2
    - Known as multiplicative decrease
  - Additive increase, multiplicative decrease (AIMD)

Additive Increase/ Multiplicative Decrease

- Tools
  - React to observance of congestion
  - Probe channel to detect more resources
- Observation
  - On notice of congestion
    - Decreasing too slowly will not be reactive enough
  - On probe of network
    - Increasing too quickly will overshoot limits

Additive Increase/ Multiplicative Decrease

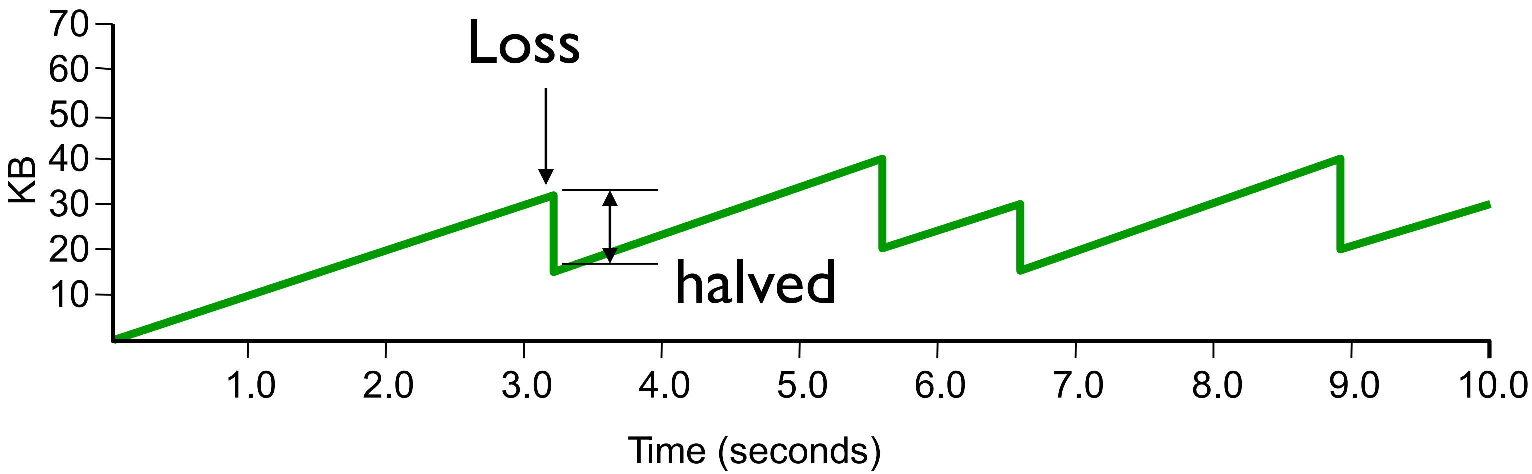
- New TCP state variable
  - `CongestionWindow`
    - Similar to **`AdvertisedWindow`** for flow control
  - Limits how much data source can have in transit
    - **`MaxWin = MIN(CongestionWindow, AdvertisedWindow)`**
    - **`EffWin = MaxWin - (LastByteSent - LastByteAcked)`**
    - TCP can send no faster then the slowest component, network or destination
- Idea
  - Increase **`CongestionWindow`** when congestion goes down
  - Decrease **`CongestionWindow`** when congestion goes up

Additive Increase/ Multiplicative Decrease

- Question
  - How does the source determine whether or not the network is congested?
- Answer
  - Timeout signals packet loss
  - Packet loss is rarely due to transmission error (on wired lines)
  - Lost packet implies congestion!

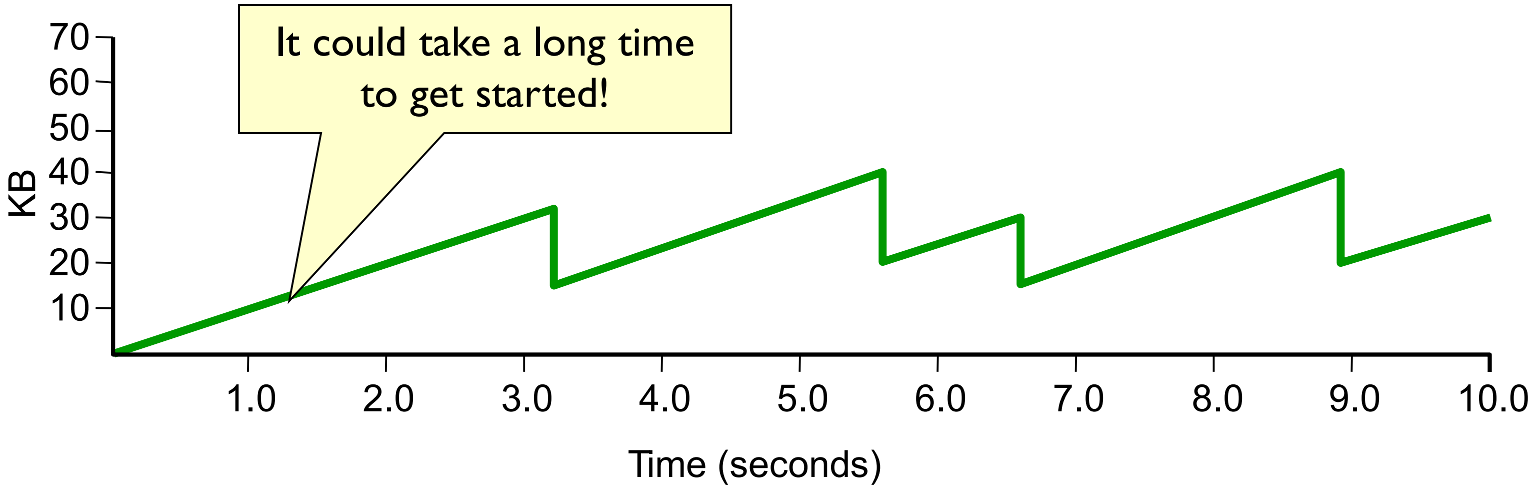
AIMD – Sawtooth Trace

- Packet loss is seen as sign of congestion and results in a multiplicative rate decrease
  - Factor of 2
- TCP periodically probes for available bandwidth by increasing its rate



TCP Start Up Behavior

- How should TCP start sending data?
  - AIMD is good for channels operating at capacity
  - AIMD can take a long time to ramp up to full capacity from scratch



TCP Start Up Behavior

- How should TCP start sending data?
  - AIMD is good for channels operating at capacity
  - AIMD can take a long time to ramp up to full capacity from scratch
  - Use Slow Start to increase window rapidly from a cold start

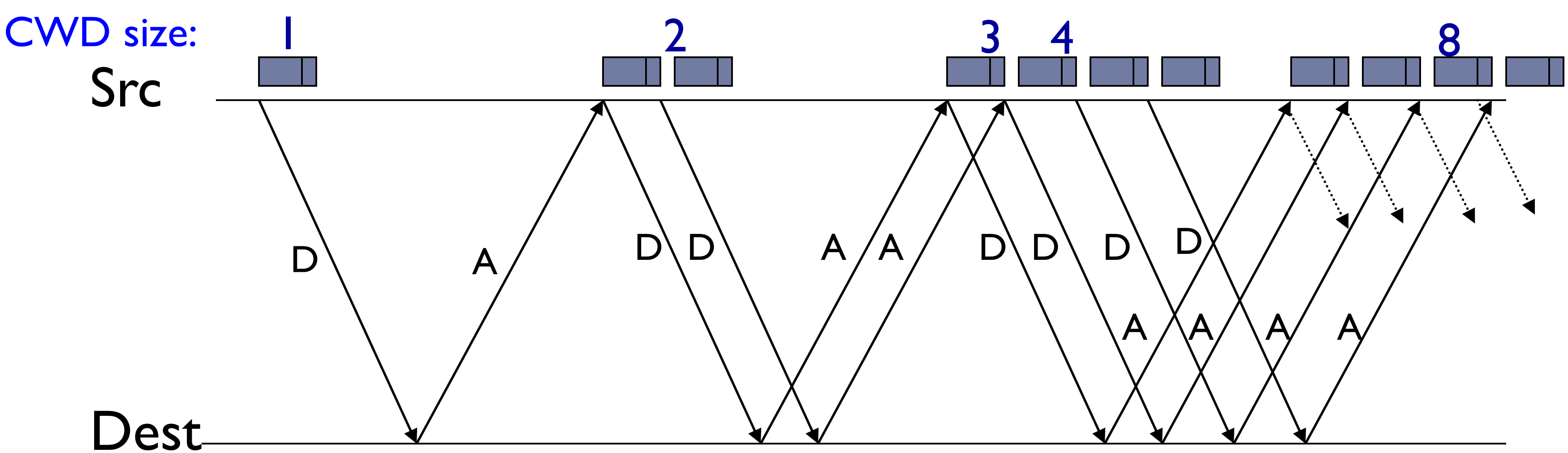
TCP Start Up Behavior: Slow Start

- Initialization of the congestion window
  - Congestion window should start small
    - Avoid congestion due to new connections
  - Start at 1 MSS,
    - Initially, CWND is 1 MSS
    - Initial sending rate is MSS/RTT
  - Reset to 1 MSS with each timeout
    - timeouts are coarse-grained, ~1/2 sec

TCP Start Up Behavior: Slow Start

- Growth of the congestion window
- Linear growth could be pretty wasteful
  - Might be much less than the actual bandwidth
  - Linear increase takes a long time to accelerate
- Start slow but then grow fast
  - Sender starts at a slow rate
  - Increase the rate exponentially
  - Until the first loss event

Slow Start Example



Slow Start

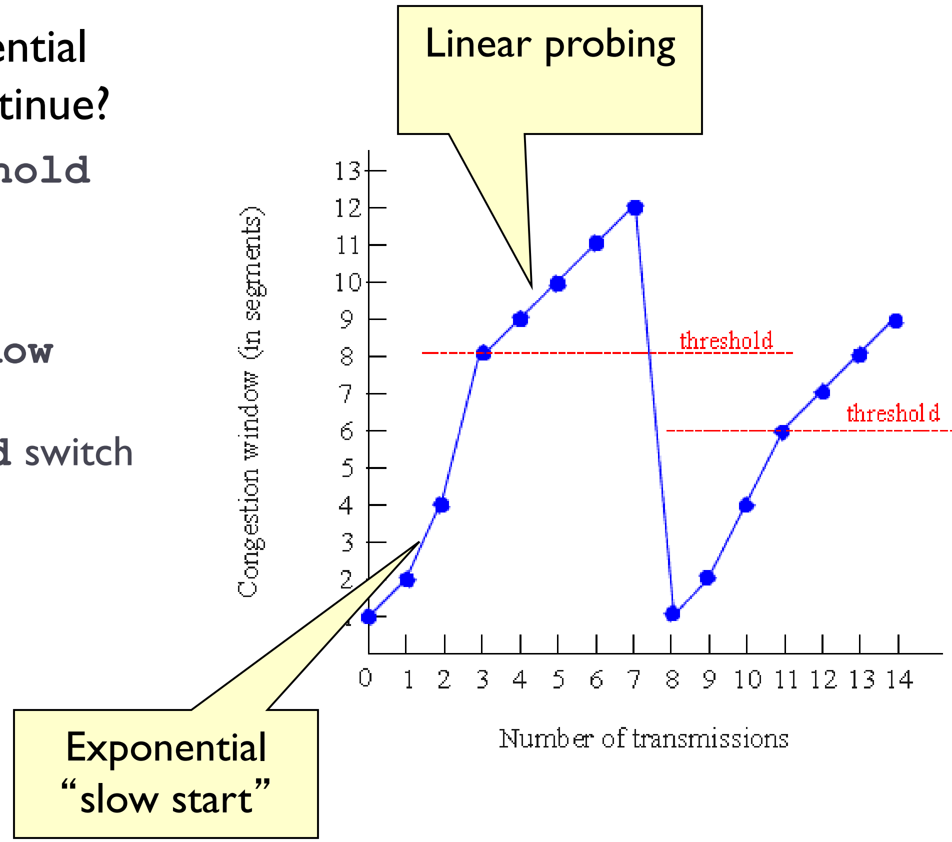
- Used
  - When first starting connection
  - When connection times out
- Why is it called slow-start?
  - Because TCP originally had no congestion control mechanism
  - The source would just start by sending a whole window's worth of data

TCP Congestion Control

- Maintain threshold window size
  - Threshold value
    - Initially set to maximum window size
    - Set to 1/2 of current window on timeout
  - Use multiplicative increase
    - When congestion window smaller than threshold
    - Double window for each window ACK'd
- In practice
  - Increase congestion window by one MSS for each ACK of new data (or N bytes for N bytes)

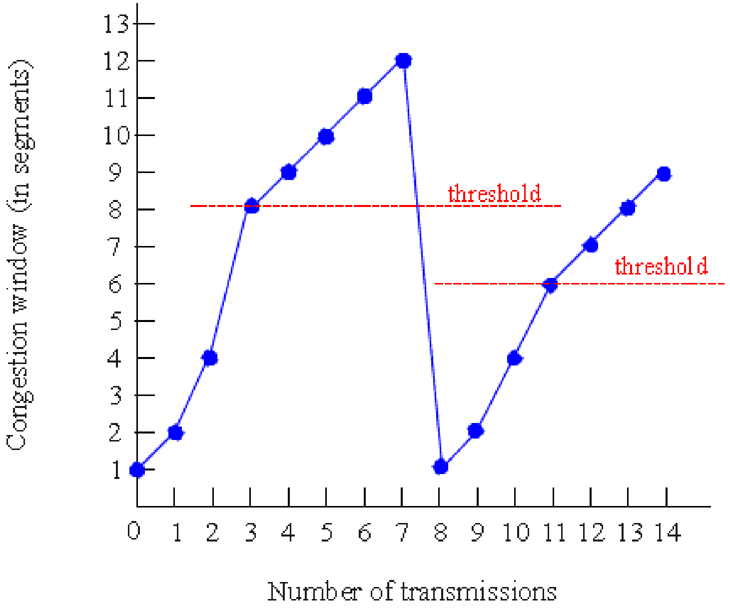
Slow Start

- How long should the exponential increase from slow start continue?
  - Use **`CongestionThreshold`** as target window size
  - Estimates network capacity
  - When **`CongestionWindow`** reaches **`CongestionThreshold`** switch to additive increase



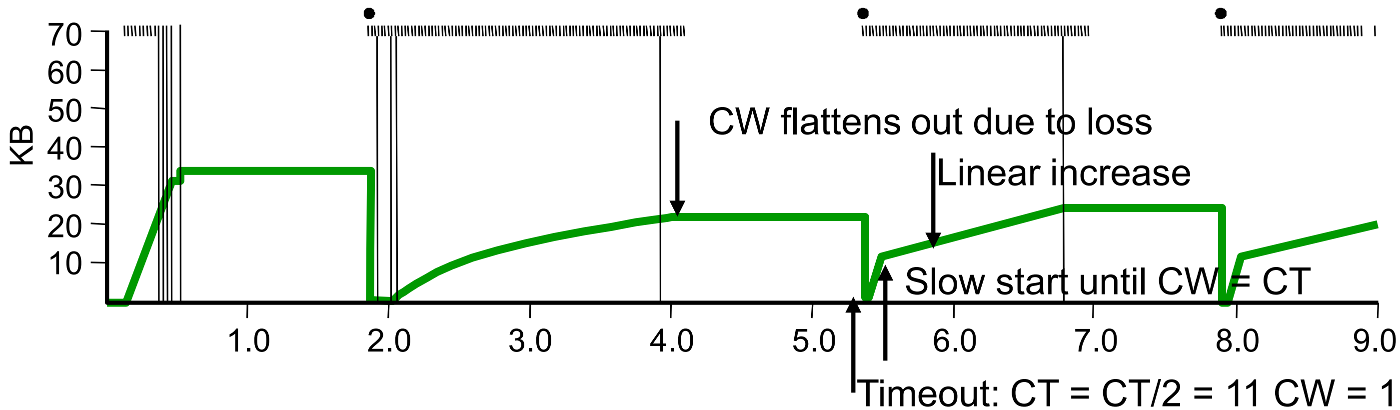
Slow Start

- Initial values
  - `CongestionThreshold = 8`
  - `CongestionWindow = 1`
- Loss after transmission 7
  - `CongestionWindow` currently 12
  - Set `Congestionthreshold = CongestionWindow/2`
  - Set `CongestionWindow = 1`



Slow Start

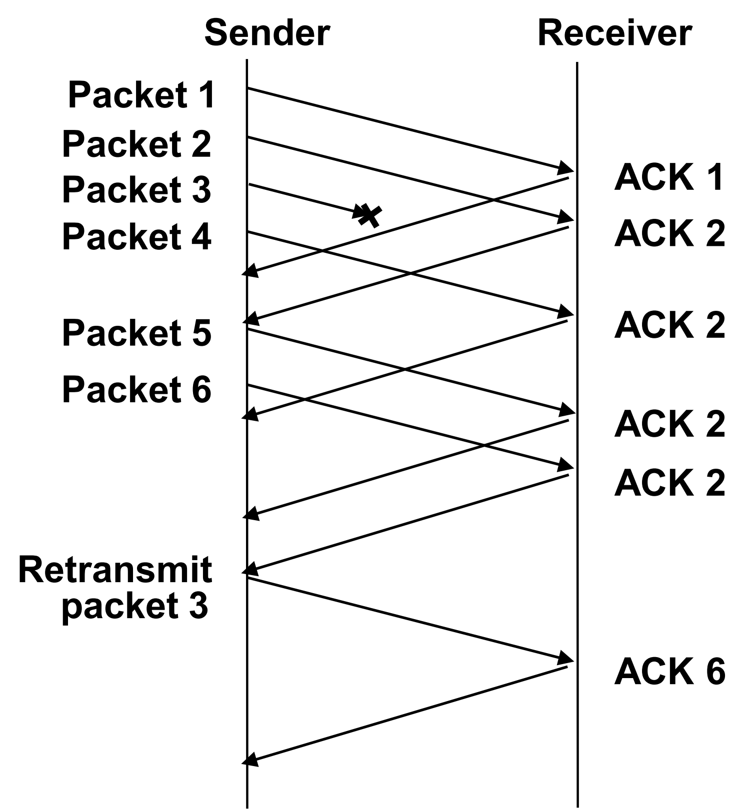
- Example trace of `CongestionWindow`



- Problem
  - Have to wait for timeout
  - Can lose half `CongestionWindow` of data

Fast Retransmit and Fast Recovery

- Problem
  - Coarse-grain TCP timeouts lead to idle periods
- Solution
  - Fast retransmit: use duplicate ACKs to trigger retransmission

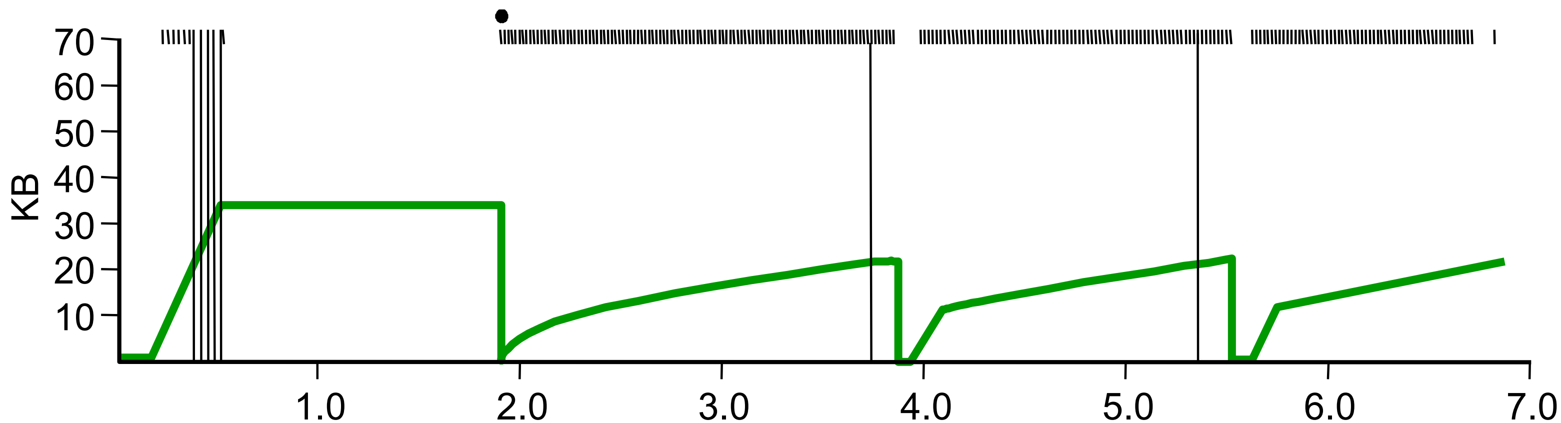


Fast Retransmit and Fast Recovery

- Send ACK for each segment received
- When duplicate ACK's received
  - Resend lost segment immediately
  - Do not wait for timeout
  - In practice, retransmit on 3rd duplicate
- Fast recovery
  - When fast retransmission occurs, skip slow start
  - Congestion window becomes 1/2 previous
  - Start additive increase immediately

Fast Retransmit and Fast Recovery

- Results



- Fast Recovery
  - Bypass slow start phase
  - Increase immediately to one half last successful **`CongestionWindow`** **`(ssthresh)`**

TCP Congestion Window Trace