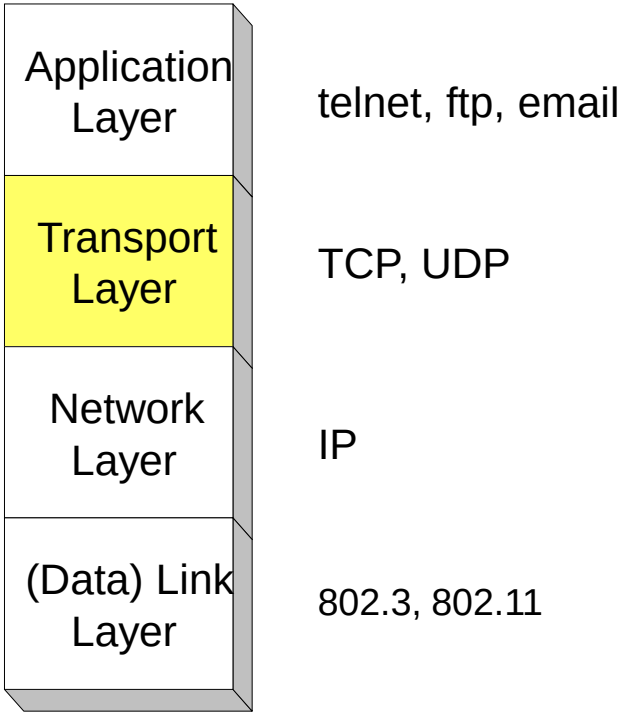
ECE 463

TCP

Reliable Transmission

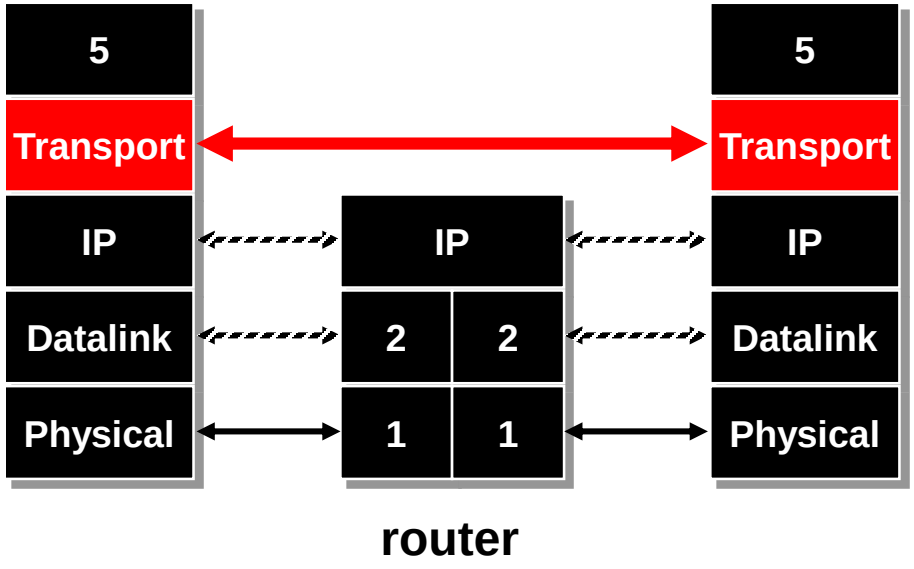
Transport Layer



Transport Protocols Concern only End Hosts, not Routers

•Header generated by sender is interpreted only by the destination

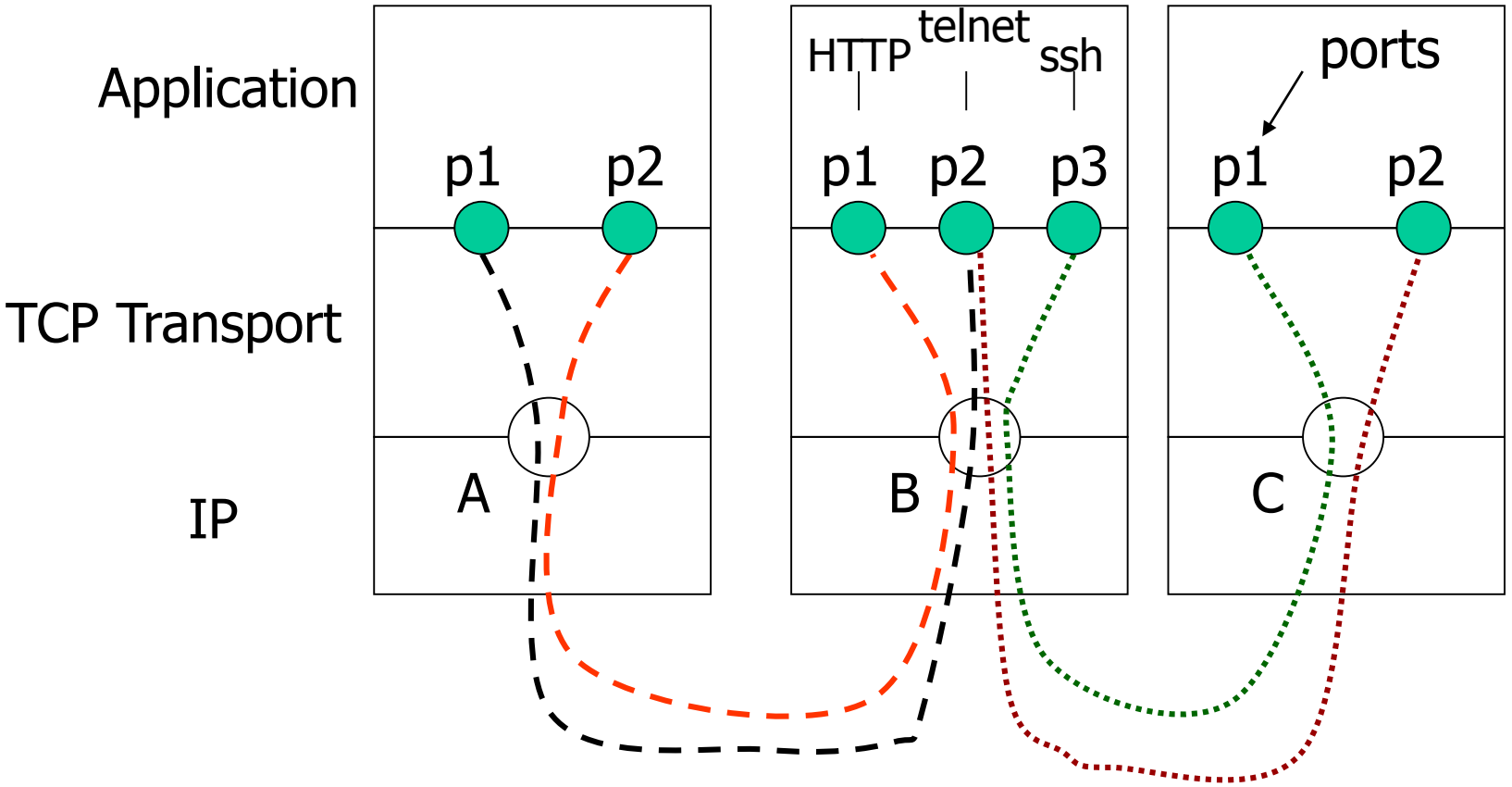
•Routers view transport header as part of the payload



Transport Layer in Internet

- Purpose 1: (De)multiplexing of data streams to different application processes
- Purpose 2: Provide value-added services that many applications want
  - Recall network layer in Internet provides a “Best-effort” service only, transport layer can add value to that
    - Application may want reliability, etc
  - No need to reinvent the wheel each time you write a new application

Using Transport Layer Port Number to (De)multiplex traffic



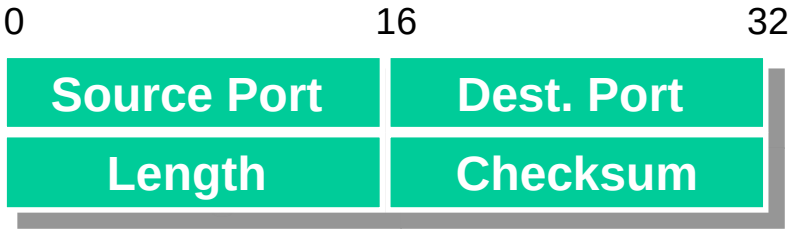
In TCP, a data stream is identified by a set of numbers:
(Source Address, Destination Address, Source Port, Destination Port)

Popular Transport Protocols

- UDP:
  - Barebones,minimal
  - Does not provide much functionality besides multiplexing
- TCP:
  - Elaborate, lots of additional functionality provided.

User Datagram Protocol (UDP)

- Connectionless datagram
  - Socket: SOCK_DGRAM
- Port number used for (de)multiplexing
  - port numbers = connection/application endpoint
- Adds end-to-end reliability through optional checksum
  - protects against data corruption errors between source and destination (links, switches/routers, bus)
  - does not protect against packet loss, duplication or reordering



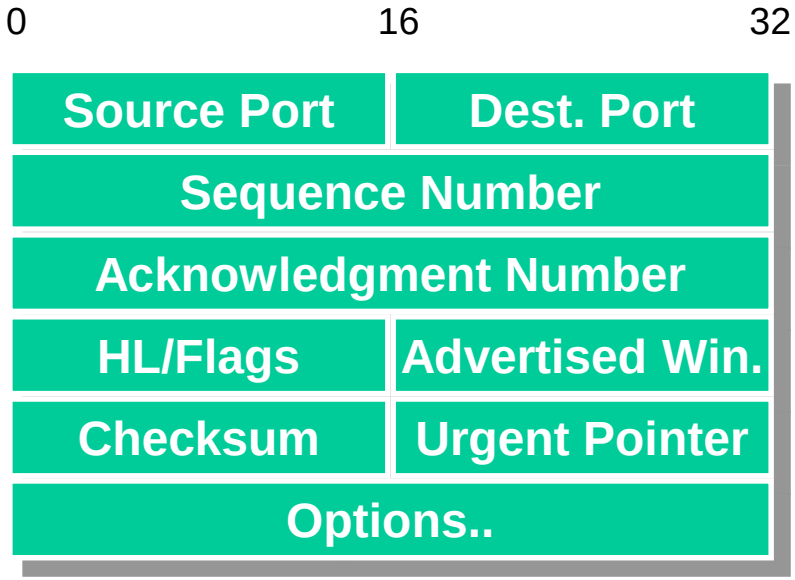
Using UDP

- Custom protocols/applications can be implemented on top of UDP
  - use the port addressing provided by UDP
  - implement own reliability, flow control, ordering, congestion control as it sees fit
- Examples:
  - remote procedure call
  - Multimedia streaming (real time protocol)
  - distributed computing communication libraries

Transmission Control Protocol (TCP)

- Reliable bidirectional in-order byte stream
  - Socket: SOCK_STREAM
- Lots of functionality
- Connection establishment.
  - Logical end-to-end connection, connection state to optimize performance
- Error control
  - Hide unreliability of the network layer from applications
  - Many types of errors: corruption, loss, duplication, reordering.
- End-to-end flow control and congestion control
  - Avoid flooding the receiver and network.

TCP Header

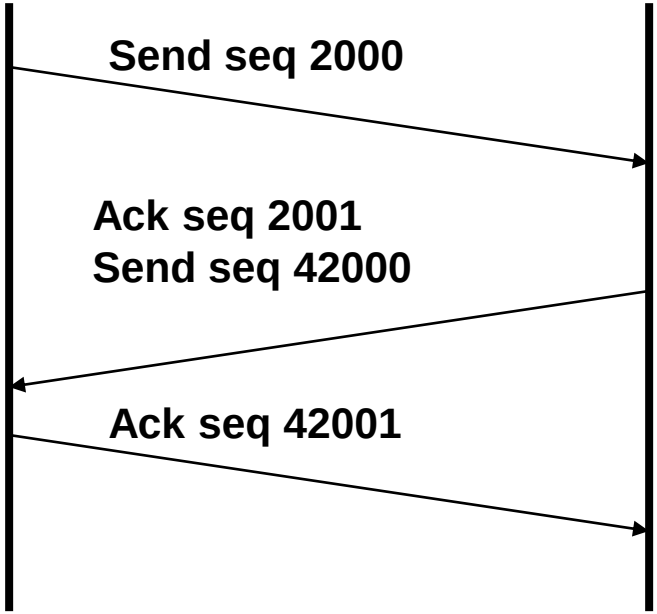


- 20 bytes total
- Sequence Number, ACK, Advertised window: relate to TCP functionality for achieving reliable delivery.
- Sequence Number is for forward direction, ACK for reverse direction.
- HL: Specifies Header Length
- Flags: 6 flags in all
- Urgent pointer: not common TCP usage, used to signal certain data is “out-of-band” and must be processed immediately

Important TCP Flags

- SYN: Synchronize
  - Used when setting up connection
- FIN: Finish
  - Used when tearing down connection
- ACK
  - Acknowledging received data
- RESET:
  - Receiver wants to abort connection, as it received unexpected segment.
- Push and Urgent flags:
  - Not as commonly used.
  - Signify receiving process must be notified, or out-of-band data

Bidirectional Communication



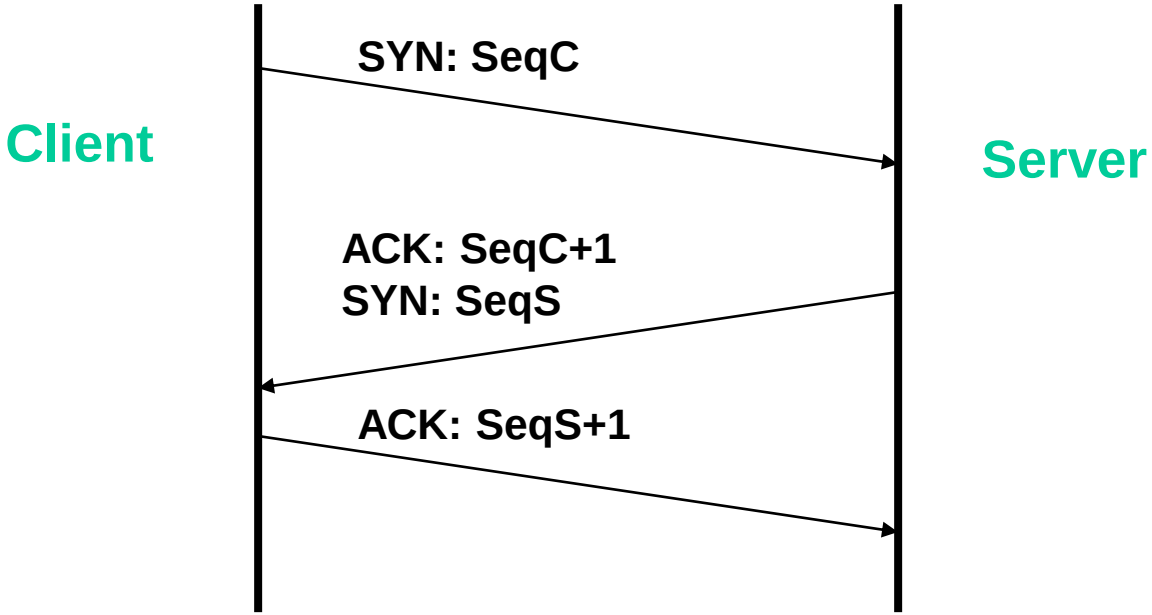
- Each Side of Connection can Send *and* Receive
- What this Means
  - Maintain different sequence numbers for each direction
  - Single segment can contain new data for one direction, plus acknowledgement for other
    - But some contain only data & others only acknowledgement

Connection Setup

- Why need connection setup?

- Mainly to agree on starting sequence numbers
  - Starting sequence number is randomly chosen
  - Reason, to reduce the chance that sequence numbers of old and new connections from overlapping

Establishing Connection



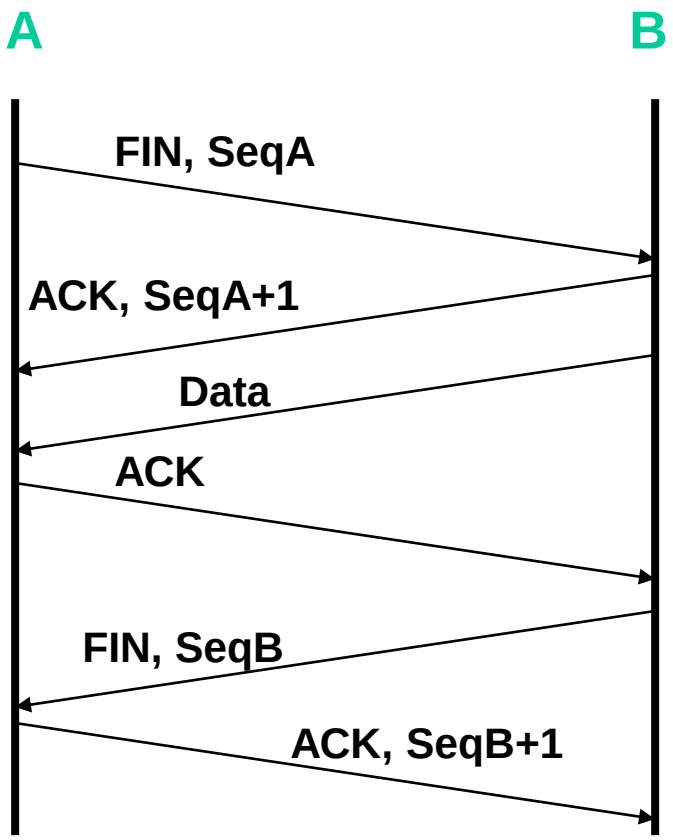
- Three-Way Handshake
  - Each side notifies other of starting sequence number it will use for sending
  - Each side acknowledges other's sequence number
    - SYN-ACK: Acknowledge sequence number + 1
  - Can combine second SYN with first ACK

SYN-Flood Attacks

- Exploit 3-way handshake
- Attacker spoofs IP addresses, sends a large number of SYNs to the destination.
- Creates “partial unopened” state at the server
- Server sends SYN-ACK to (spoofed) addresses, to which no response is received.
- Rate at which SYNs sent much faster than rate at which server eliminates partial state.

Tearing Down Connection

- Either Side Can Initiate Tear Down
  - Send FIN signal
  - "I'm not going to send any more data"
- Other Side Can Continue Sending Data
  - Half open connection
  - Must continue to acknowledge
- Acknowledging FIN
  - Acknowledge last sequence number + 1

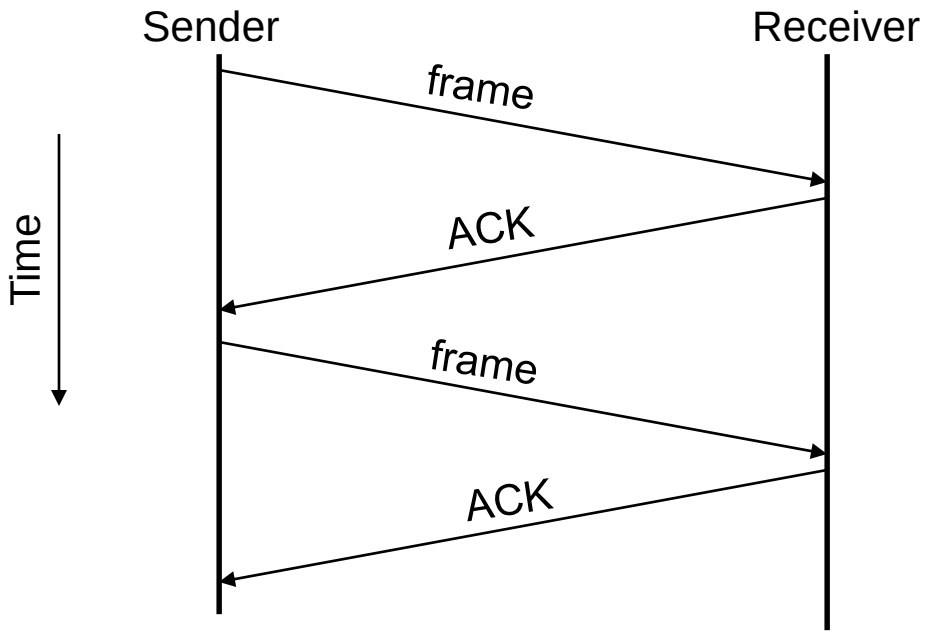


Reliable Transmission

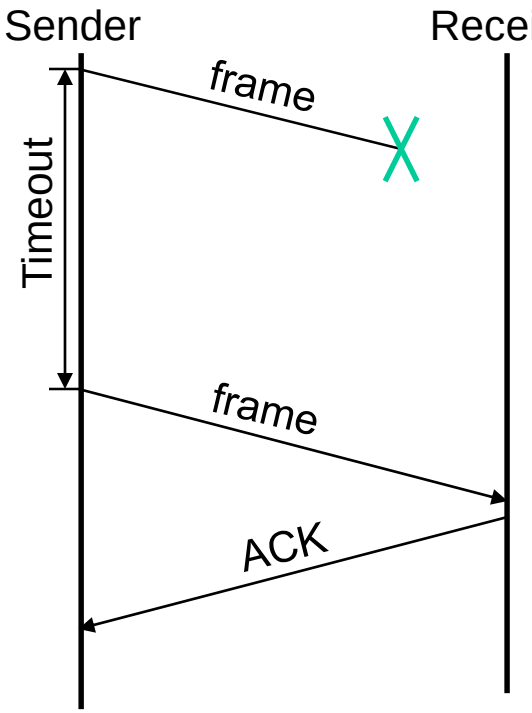
- Use two basic techniques:
  - Acknowledgements (ACKs)
  - Timeouts
- Two examples:
  - Stop-and-go
  - Sliding window

Stop-and-Go

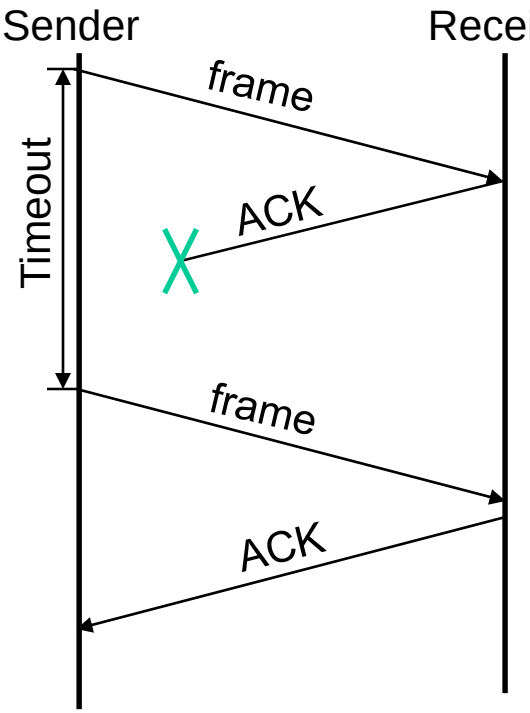
- Receiver: send an acknowledge (ACK) back to the sender upon receiving a packet (frame)
- Sender: excepting first packet, send a packet only upon receiving the ACK for the previous packet



What Can Go Wrong?

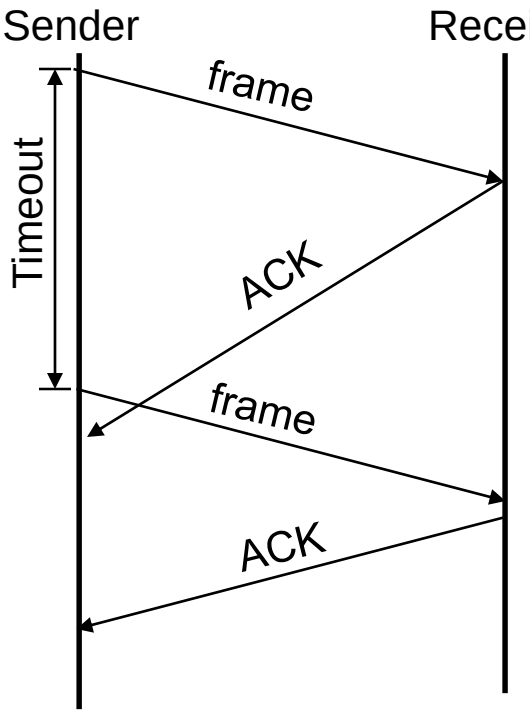


Frame lost → resent it on Timeout



ACK lost → resent packet

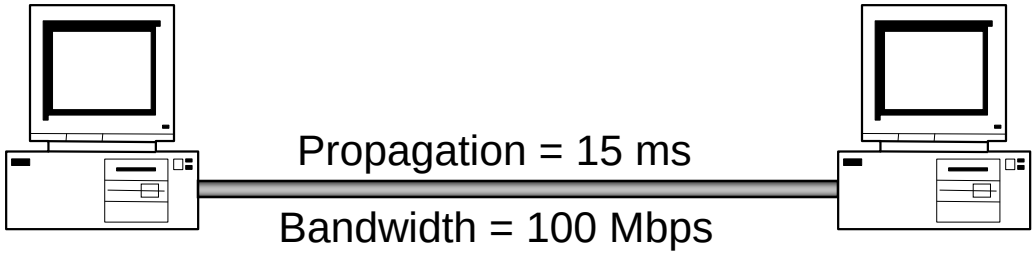
Need a mechanisms to detect duplicate packet



ACK delayed → resent packet

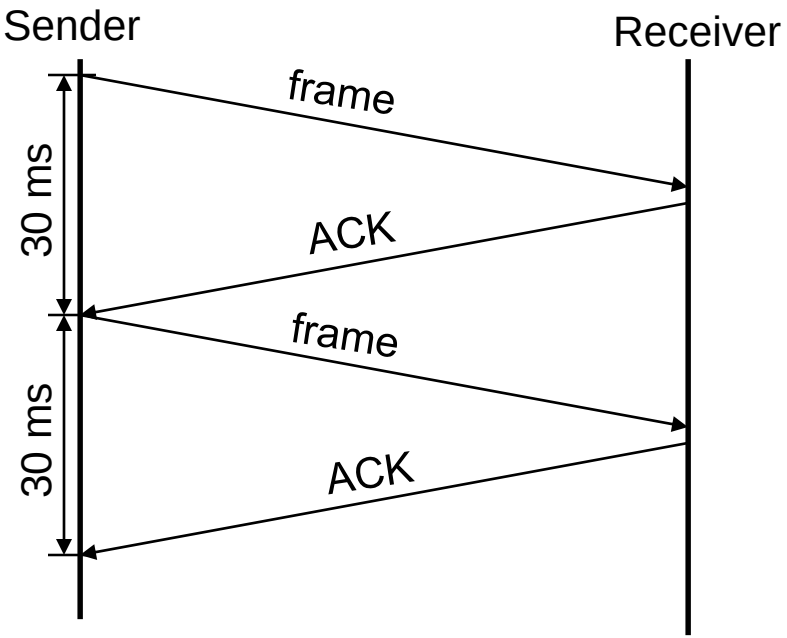
Stop-and-Go Disadvantage

- May lead to inefficient link utilization
- Example: assume
  - One-way propagation = 15 ms
  - Bandwidth = 100 Mbps
  - Packet size = 1000 bytes → transmit = $(8*1000)/10^8$ = 0.08ms
  - Neglect queue delay → Latency = approx. 15 ms; RTT = 30 ms

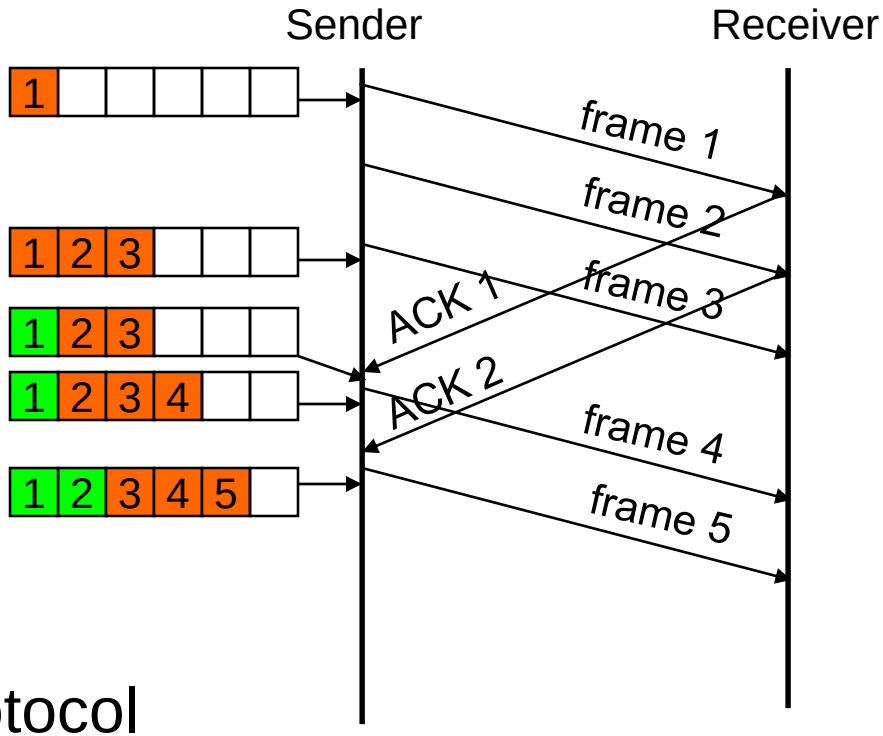


Stop-and-Go Disadvantage (cont'd)

- Send a message every 30 ms → Throughput = (8*1000)/0.03 = 0.2666 Mbps
- Thus, the protocol uses less than 0.3% of the link capacity!



Sliding Window Approach



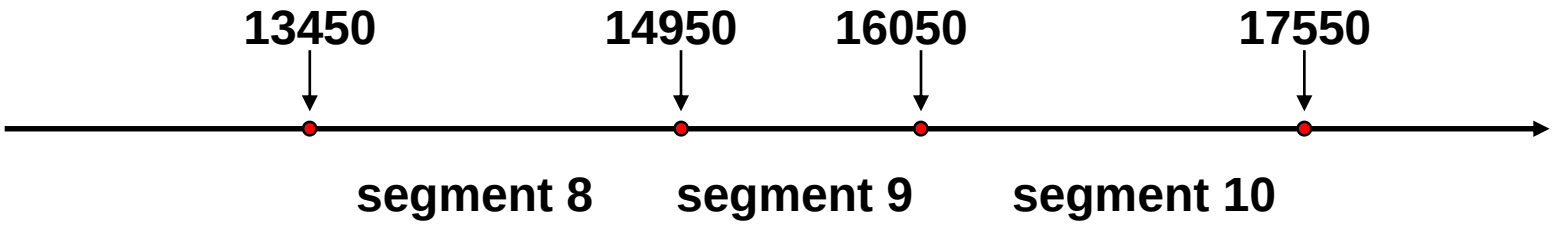
Sliding window protocol

Send multiple packets without waiting for ACK.

Sliding window size: Max number of packets that can be sent without ACK being received (3 in figure)

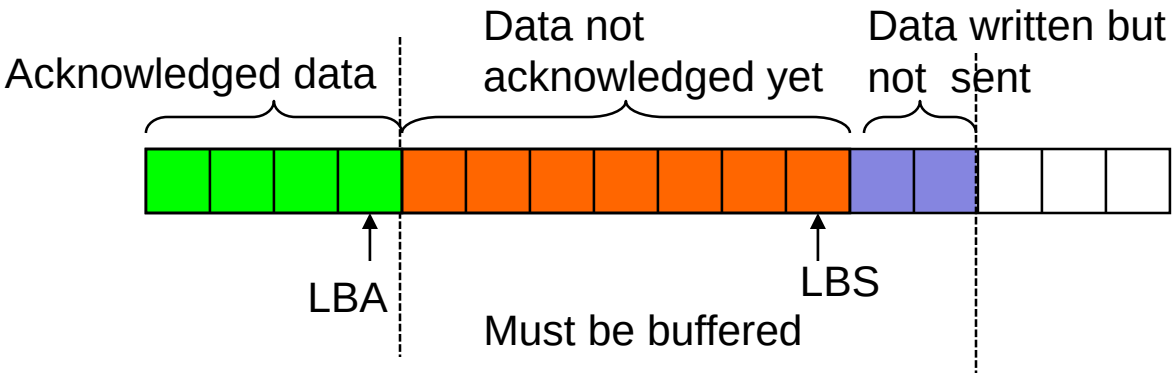
Sequence Numbers in TCP

- Each byte in byte stream is numbered.
  - 32 bit value
- TCP breaks up the byte stream in segments
- Each segment has a sequence number.
  - Indicates where it fits in the byte stream



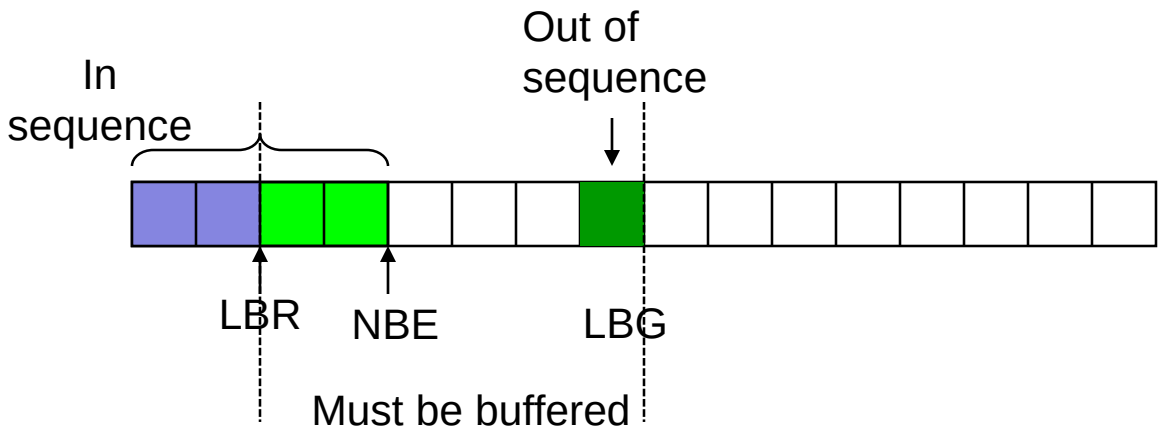
Sliding Window Protocol: Sender

- TCP: operates at a byte stream level (rather than packet level)
- Sender maintains a window of sequence numbers
  - SWS (sender window size) – maximum data that can be sent without receiving an ACK
  - LBA (last byte acknowledged)
  - LBS (last byte sent)
- TCP sender side socket buffer:
  - Data sent but not acknowledged
  - Data written by application but not sent yet



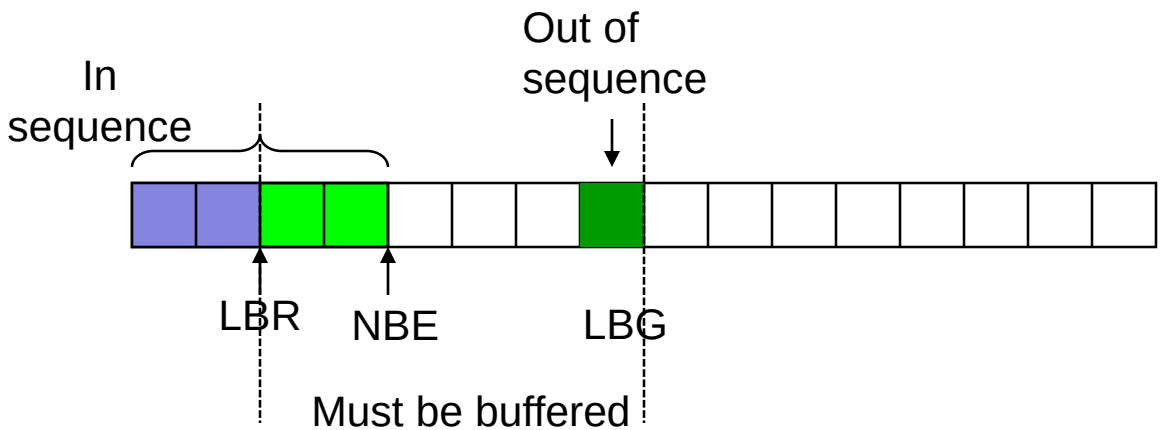
Sliding Window Protocol: Receiver

- TCP maintains a receive socket buffer
  - Stores data that arrived out-of-order (cannot be given to application yet)
  - Also stores data that arrived in-order but not yet read by application (slow process)
- Receiver maintains a window of sequence numbers
  - NBE (next byte expected – all previous bytes received in sequence
  - LBR (last byte read by application)
  - LBG (last byte got by receiver)



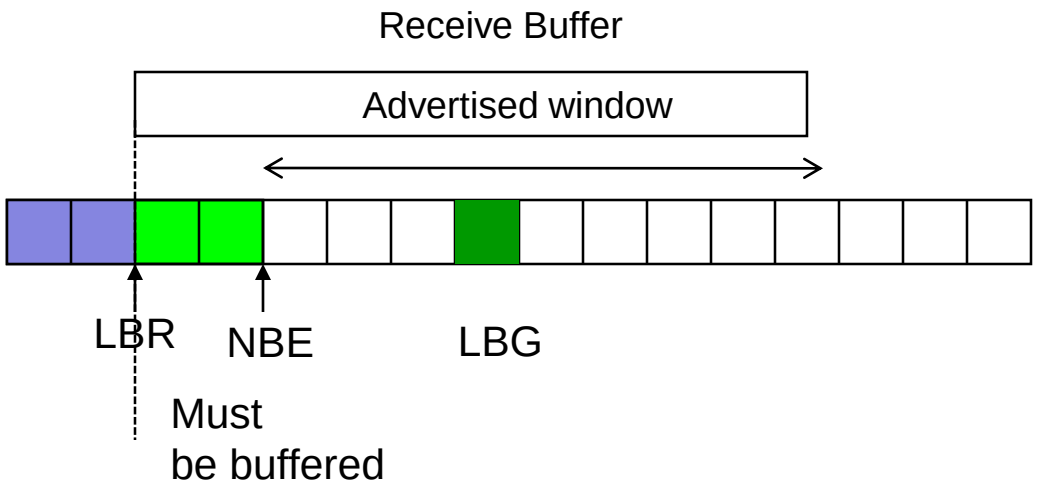
Sliding Window Protocol: Receiver

- If incoming byte < NBE
  - Discard packet
- Else
  - Accept packet (provided it fits in the buffer)
  - ACK largest byte that all previous bytes were received



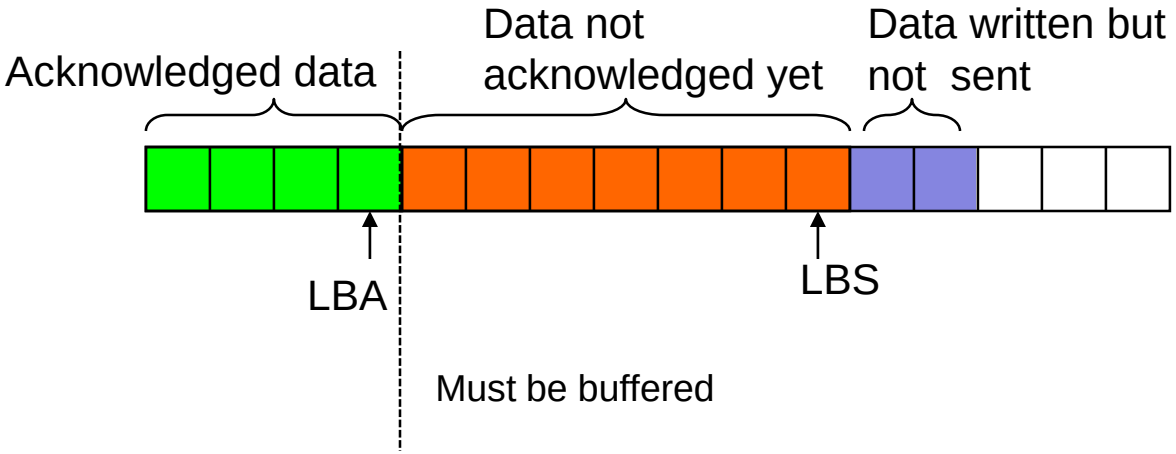
Flow Control

- Receiver throttles the sender by advertising a window no larger than the amount of data it can buffer
- Advertised Window depends on
  - Size of receive buffer
  - How fast receiver process can read the bytes from the buffer
- TCP sending window impacted by advertised window

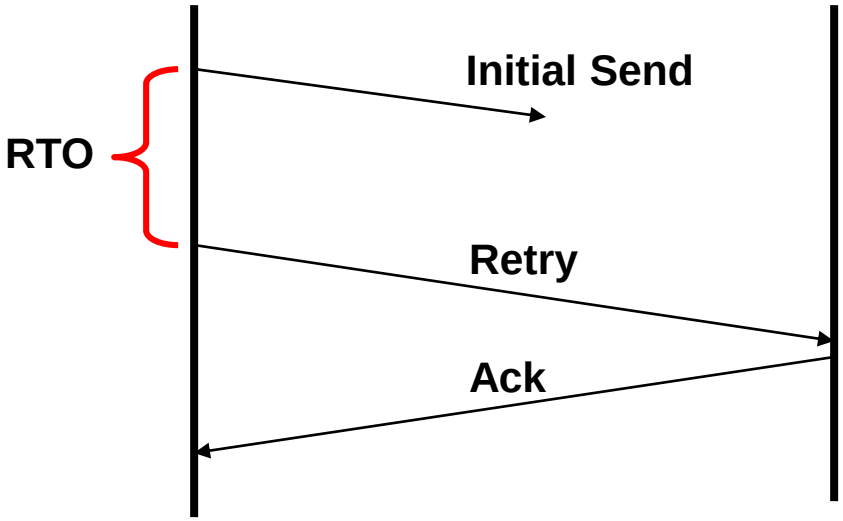


Flow Control (sender side)

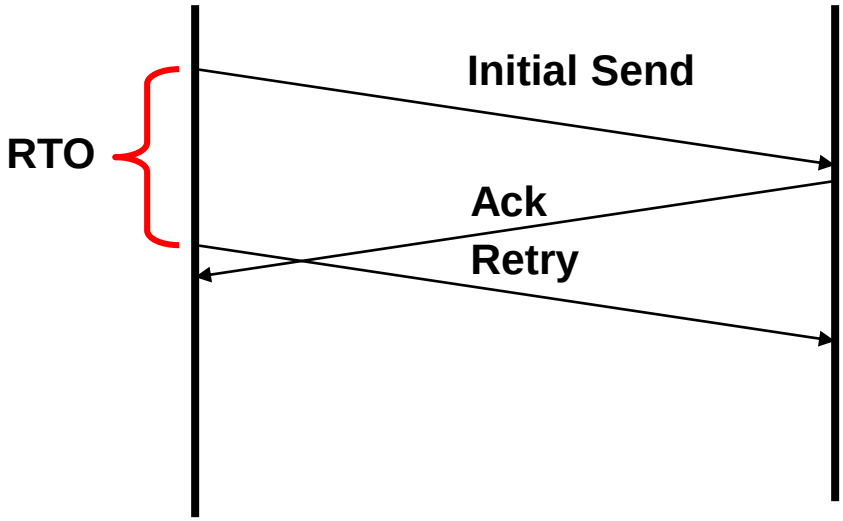
- Receiver throttles the sender by advertising a window no larger than the amount of data it can buffer
- Effective window at sender (limits how much it can send)
  Advertised window – (outstanding unacknowledged data)
- Sending process can write to send-side TCP buffer only if space in sender side buffer – otherwise will block



Setting Retransmission Timeout (RTO)



Detect dropped packet



RTO too short

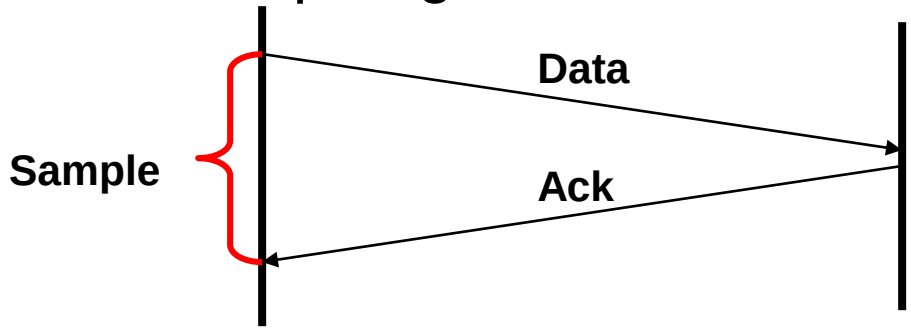
  - Time between sending & resending segment
- Challenge
  - Too long: Add latency to communication when packets dropped
  - Too short: Send too many duplicate packets
  - General principle: Must be > 1 Round Trip Time (RTT)

Setting Timeouts

- Retransmission time-out should be set based on round-trip delay
- But round-trip delay different for each path!
- Must estimate RTT dynamically

Round-trip Time Estimation

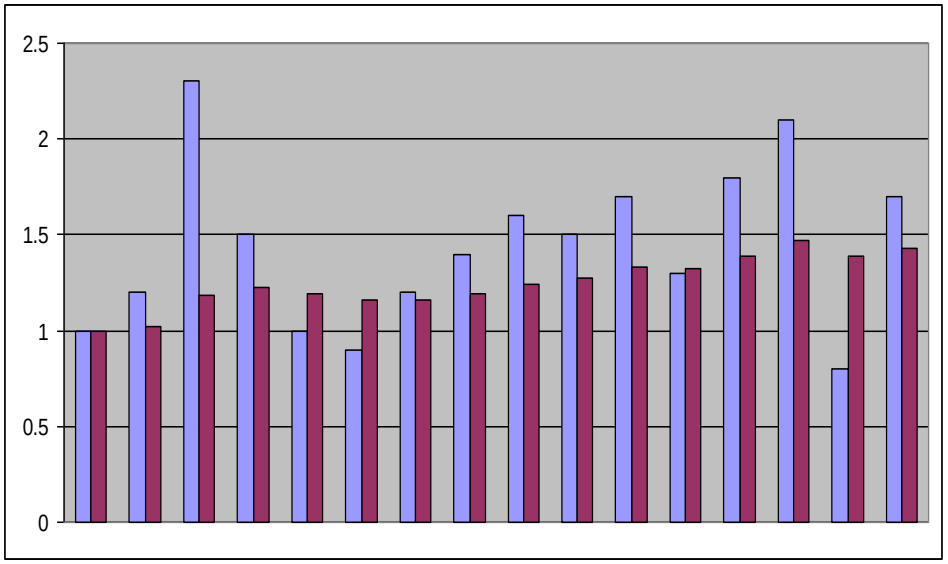
- Every Data/Ack pair gives new RTT estimate



- Can Get Lots of Short-Term Fluctuations

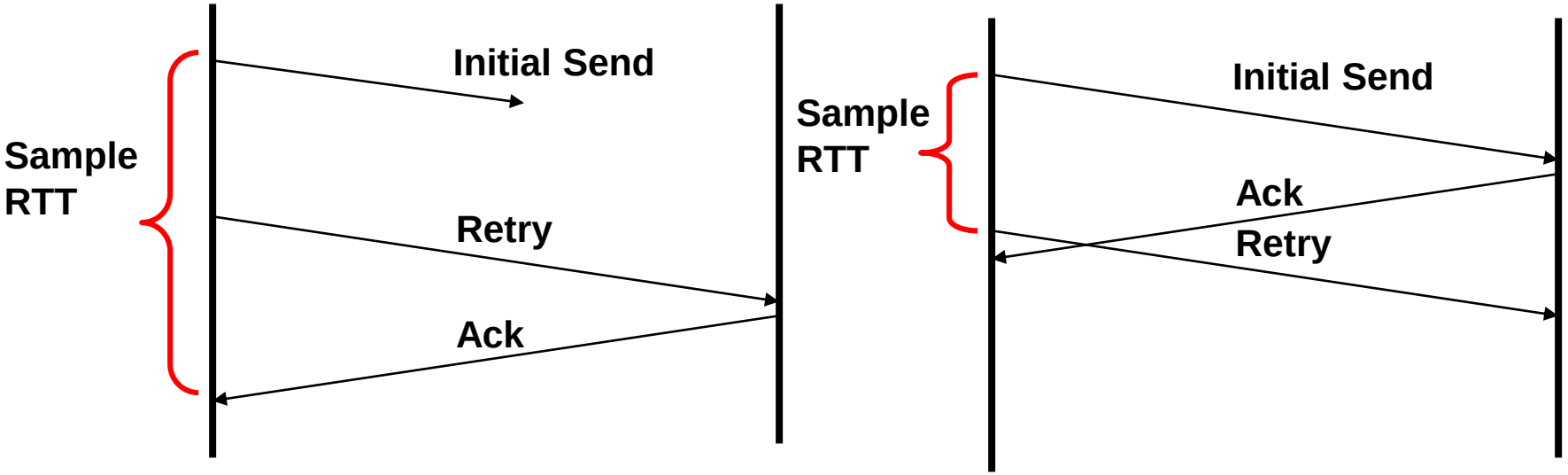
Original TCP Round-trip Estimator

- Round trip times exponentially averaged:
  - New RTT = α (old RTT) + (1 - α) (new sample)
  - Recommended value for α: 0.8 - 0.9
    - 0.875 for most TCP's



- Retransmit timer set to (b * RTT), where b = 2
  - Every time timer expires, RTO exponentially backed-off
- Enhanced algorithm used in practice
  - [Jacobson/Karels: see textbook if interested]

Issue



What value of Sample RTT to use?

Solution: TCP simply neglects SampleRTT in such cases.
[Karn/Partridge algorithm]

Extra

Buffer sizes

- Size of TCP socket buffers at sender and receiver side critical
- Would like this to be Bandwidth * Delay product of network, to ensure the pipe can be full.
- Must be tuned carefully for individual path.
- Traditionally, OSs used default socket buffer sizes that must be overriden by application programmer/end-user
  - "Tuning" in manual fashion tricky/hard.
- Research efforts at "auto-tuning" TCP buffers:
  - **Automatic TCP Buffer Tuning**
    Jeffrey Semke, Matthew Mathis, and Jamshid Mahdavi, SIGCOMM 1998
- More recently finding its way into some Operating Systems.