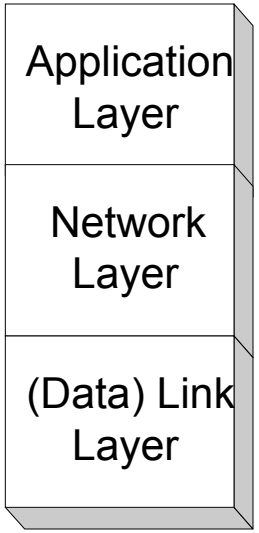
CS4700/CS5700
Fundamentals of Computer Networks

Lecture 14: TCP

Slides used with permissions from Edward W. Knightly,
T. S. Eugene Ng, Ion Stoica, Hui Zhang

What Layers are Needed in a Basic Telephone Network?

- Supports a single application: Telephone
- An end host is a telephone
- Each telephone makes only one voice stream
  - Even with call-waiting and 3-way calling



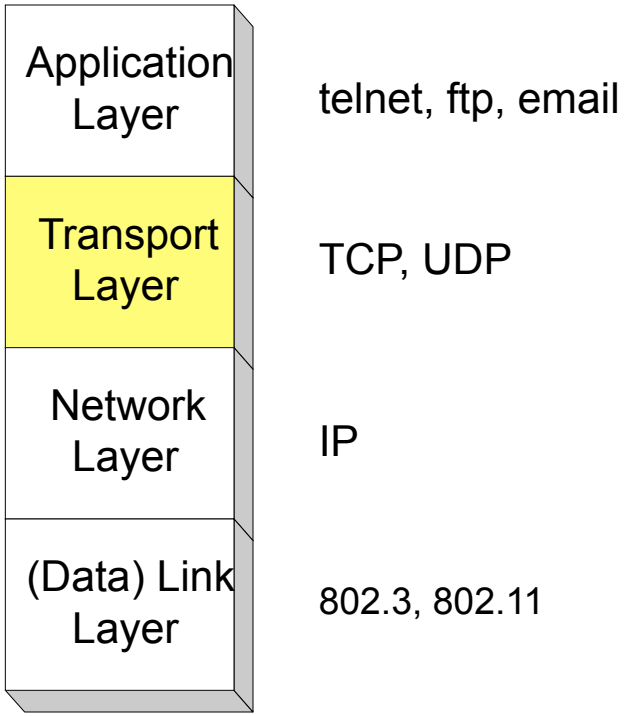
Telephone

Telephone numbering, signaling, routing

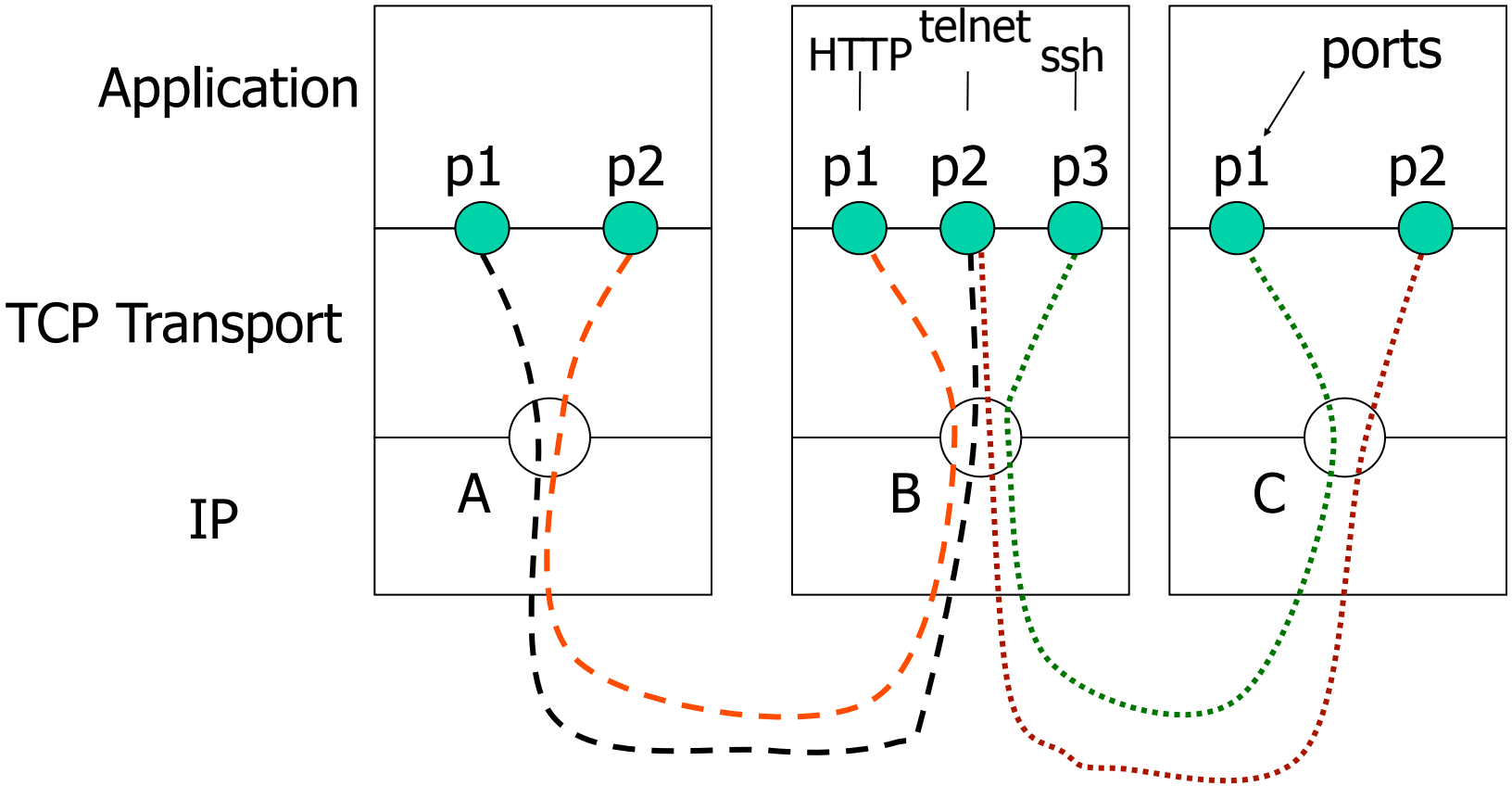
TDMA

Is this Enough for a Datagram Computer Network?

- Supports many applications
- Each end host is usually a general purpose computer
- Each end host can be generating many data streams simultaneously
- In theory, each data stream can be identified as a different “Protocol” in the IP header for demultiplexing
  - At most 256 streams
- Insert Transport Layer to create an interface for different applications
  - Provide (de)multiplexing
  - Provide value-added functions



E.g. Using Transport Layer Port Number to (De) multiplex traffic



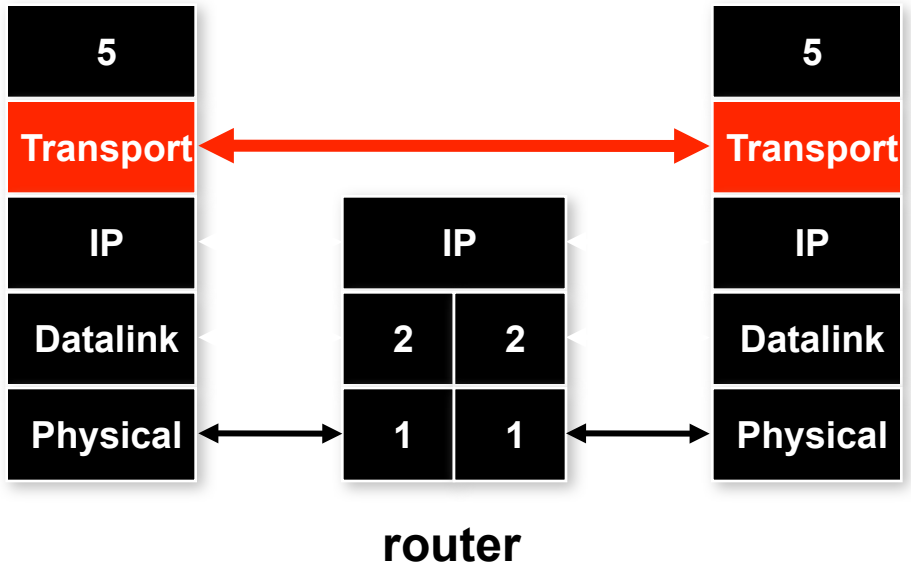
In TCP, a data stream is identified by a set of numbers:
(Source Address, Destination Address, Source Port, Destination Port)

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

Transport Protocols Concern only End Hosts, not Routers

- Lowest level end-to-end protocol.
  - Header generated by sender is interpreted only by the destination
  - Routers view transport header as part of the payload
- Adds functionality to the best-effort packet delivery IP service.
  - Make up for the shortcomings of the core network



(Possible) Transport Protocol Functions

(Possible) Transport Protocol Functions

- Multiplexing/demultiplexing for multiple applications.
  - Port abstraction

(Possible) Transport Protocol Functions

- Multiplexing/demultiplexing for multiple applications.
  - Port abstraction
- Connection establishment.
  - Logical end-to-end connection
  - Connection state to optimize performance

(Possible) Transport Protocol Functions

- Multiplexing/demultiplexing for multiple applications.
  - Port abstraction
- 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.

(Possible) Transport Protocol Functions

- Multiplexing/demultiplexing for multiple applications.
  - Port abstraction
- 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.
  - Avoid flooding the receiver

(Possible) Transport Protocol Functions

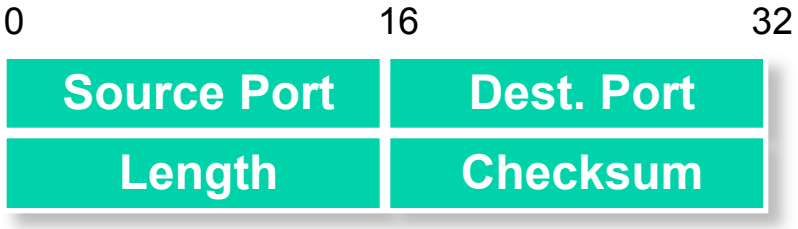
- Multiplexing/demultiplexing for multiple applications.
  - Port abstraction
- Connection establishment.
  - Logical end-to-end connection
  - Connection state to optimize performance
- Error control.
  - Hide unreliability of the network layer from applications
  - Many types of errors: corruption, loss, duplication, reordering.
- End-to-end flow control.
  - Avoid flooding the receiver
- Congestion control.
  - Avoid flooding the network

(Possible) Transport Protocol Functions

- Multiplexing/demultiplexing for multiple applications.
  - Port abstraction
- 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.
  - Avoid flooding the receiver
- Congestion control.
  - Avoid flooding the network
- More….

User Datagram Protocol (UDP)

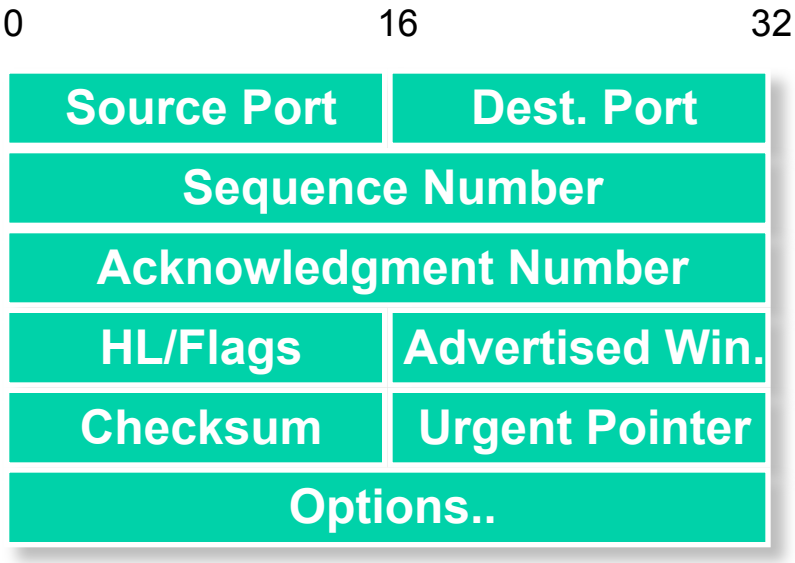
- Connectionless datagram
  - Socket: SOCK_DGRAM
- Port number used for (de)multiplexing
  - port numbers = connection/application endpoint
- Adds end-to-end reliability through optional checksum
  - protects against data corruption errors between source and destination (links, switches/routers, bus)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



Using UDP

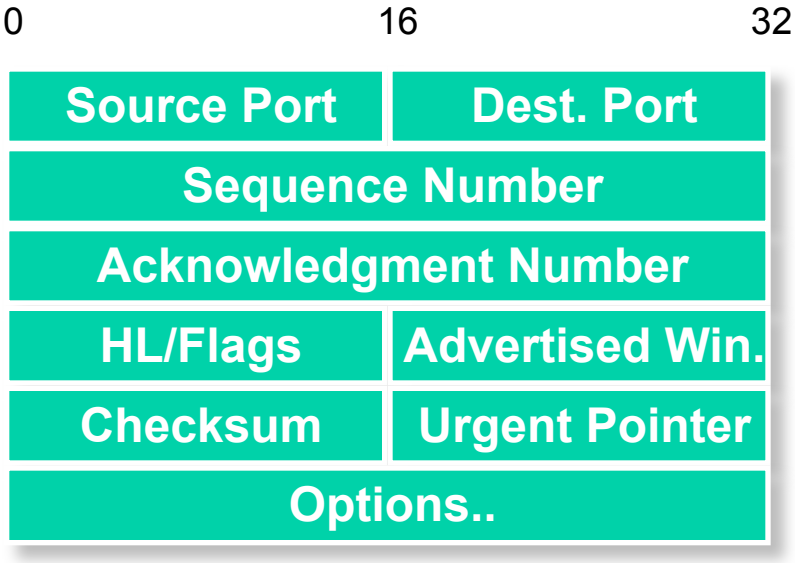
- Custom protocols/applications can be implemented on top of UDP
  - use the port addressing provided by UDP
  - implement own reliability, flow control, ordering, congestion control as it sees fit
- Examples:
  - remote procedure call
  - Multimedia streaming (real time protocol)
  - distributed computing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

Transmission Control Protocol (TCP)



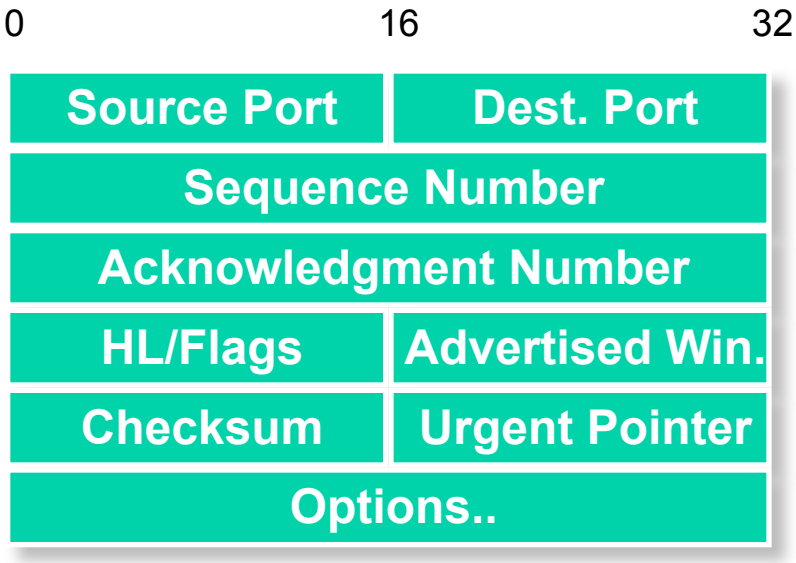
Transmission Control Protocol (TCP)

- Reliable bidirectional in-order byte stream
  - Socket: SOCK_STREAM



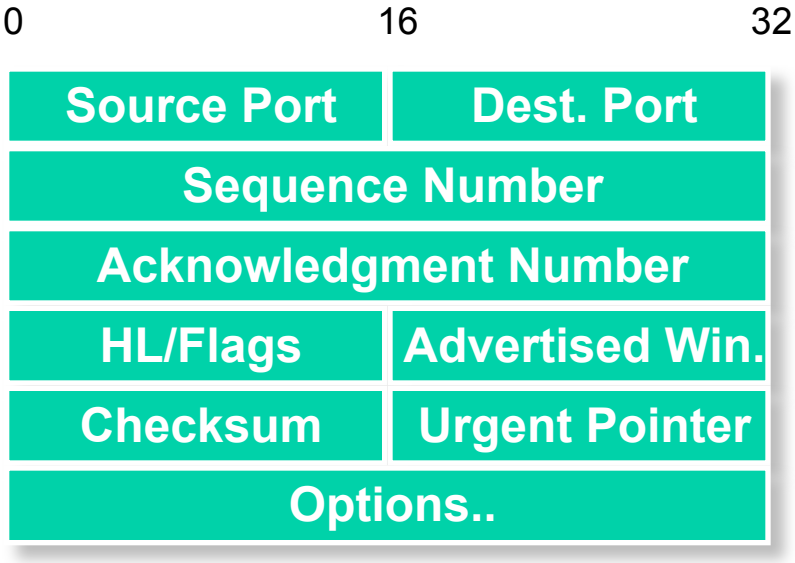
Transmission Control Protocol (TCP)

- Reliable bidirectional in-order byte stream
  - Socket: SOCK_STREAM
- Connections established & torn down



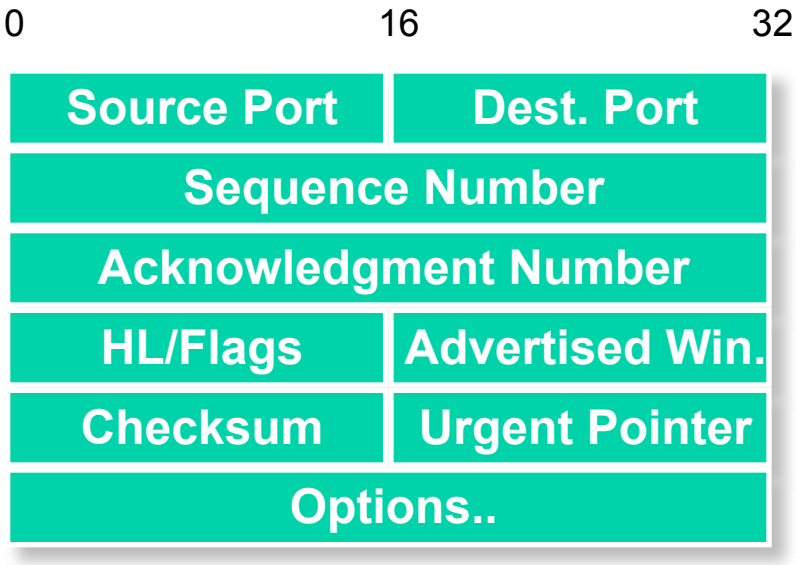
Transmission Control Protocol (TCP)

- Reliable bidirectional in-order byte stream
  - Socket: SOCK_STREAM
- Connections established & torn down
- Multiplexing/ demultiplexing
  - Ports at both ends



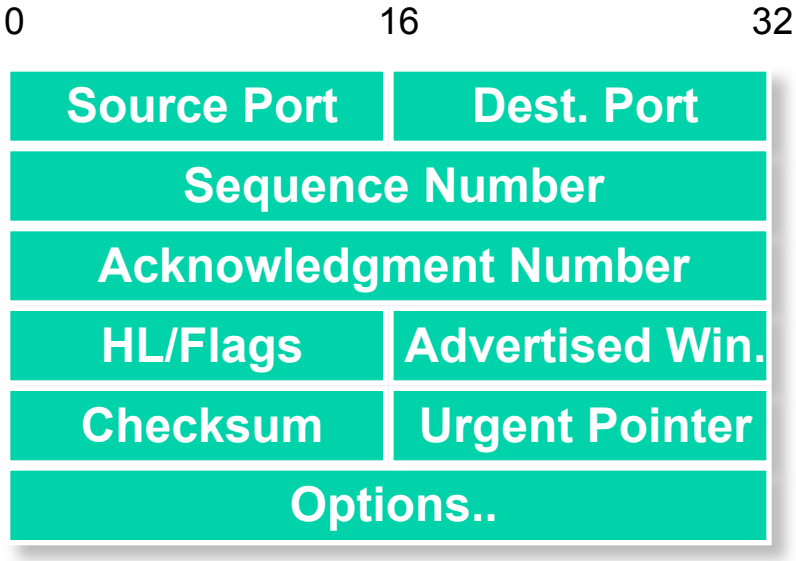
Transmission Control Protocol (TCP)

- Reliable bidirectional in-order byte stream
  - Socket: SOCK_STREAM
- Connections established & torn down
- Multiplexing/ demultiplexing
  - Ports at both ends
- Error control
  - Users see correct, ordered byte sequences



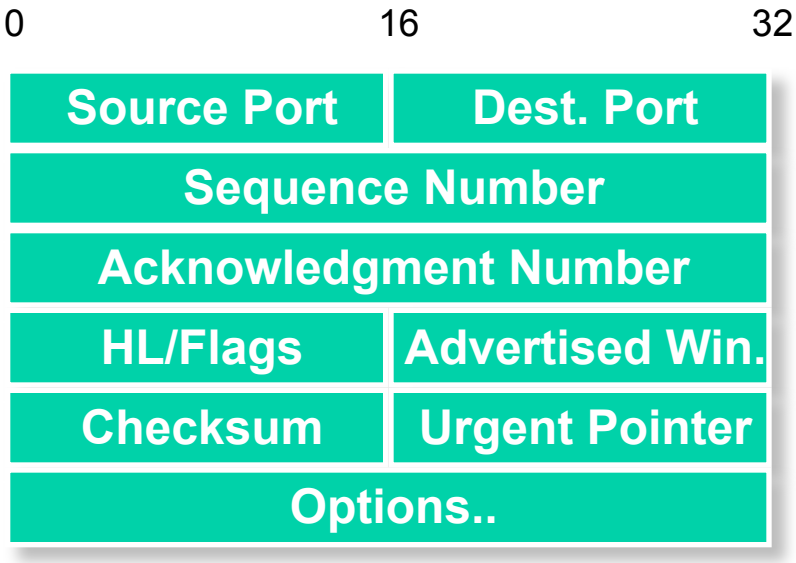
Transmission Control Protocol (TCP)

- Reliable bidirectional in-order byte stream
  - Socket: SOCK_STREAM
- Connections established & torn down
- Multiplexing/ demultiplexing
  - Ports at both ends
- Error control
  - Users see correct, ordered byte sequences
- End-end flow control
  - Avoid overwhelming machines at each end



Transmission Control Protocol (TCP)

- Reliable bidirectional in-order byte stream
  - Socket: SOCK_STREAM
- Connections established & torn down
- Multiplexing/ demultiplexing
  - Ports at both ends
- Error control
  - Users see correct, ordered byte sequences
- End-end flow control
  - Avoid overwhelming machines at each end
- Congestion avoidance
  - Avoid creating traffic jams within network



High Level TCP Features

- Sliding window protocol
  - Use sequence numbers
- Bi-directional
  - Each host can be a receiver and a sender simultaneously
  - For clarity, we will usually discuss only one direction

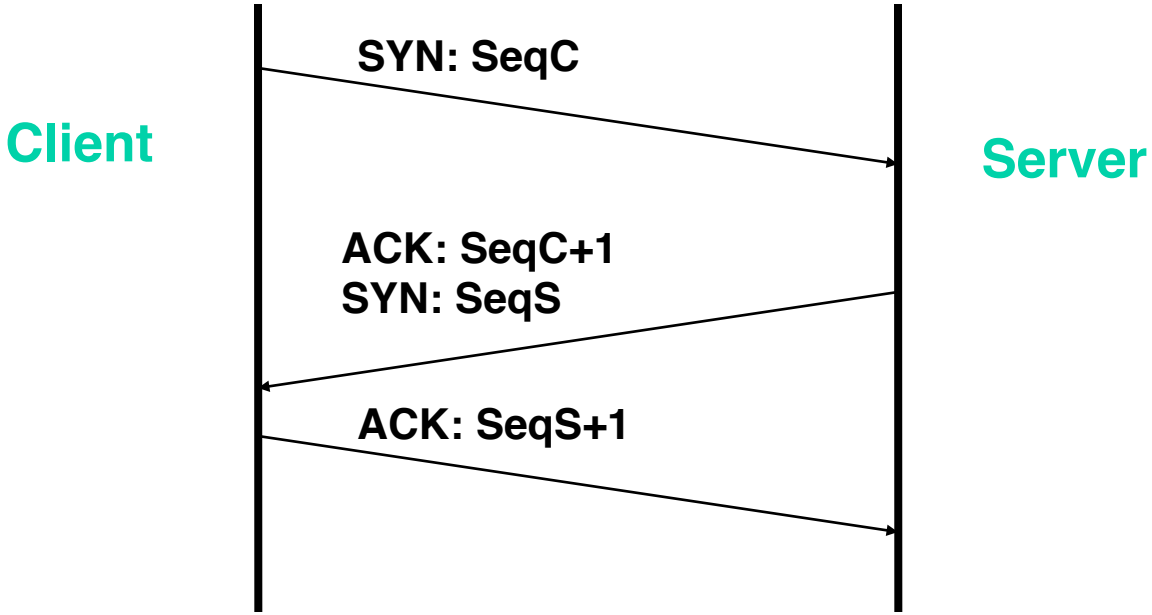
Connection Setup

- 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

Important TCP Flags

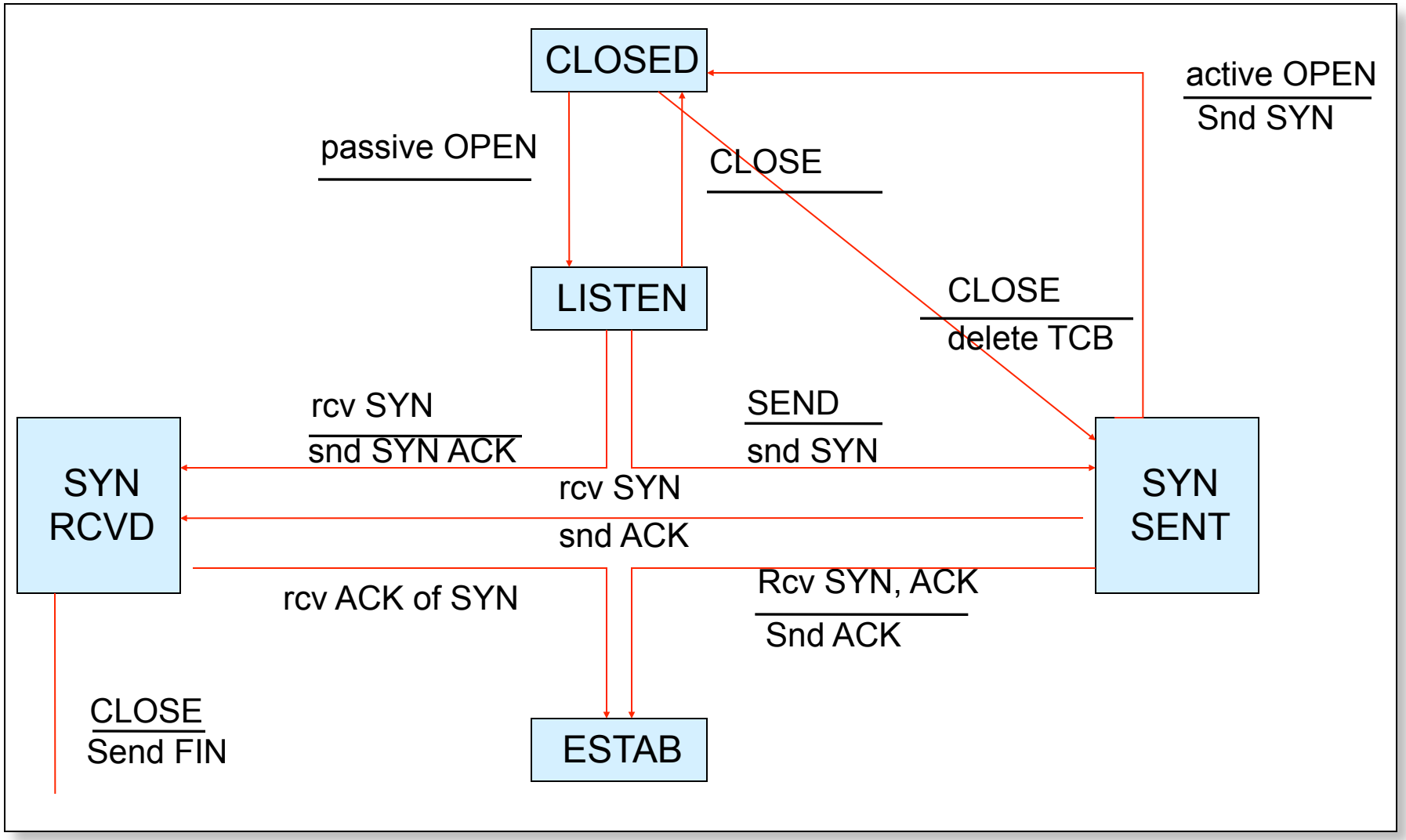
- SYN: Synchronize
  - Used when setting up connection
- FIN: Finish
  - Used when tearing down connection
- ACK
  - Acknowledging received data

Establishing Connection

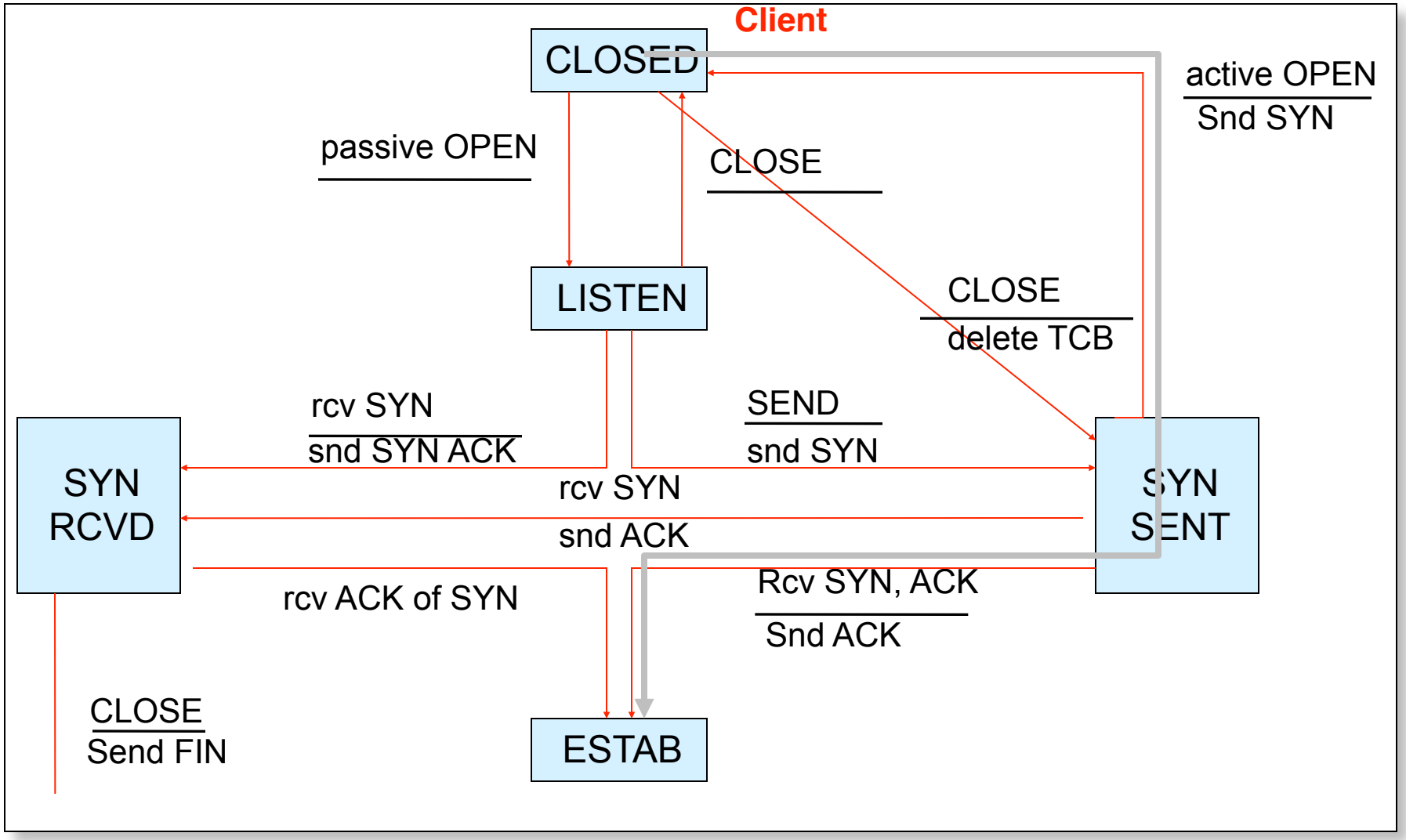


- Three-Way Handshake
  - Each side notifies other of starting sequence number it will use for sending
  - Each side acknowledges other's sequence number
    - SYN-ACK: Acknowledge sequence number + 1
  - Can combine second SYN with first ACK

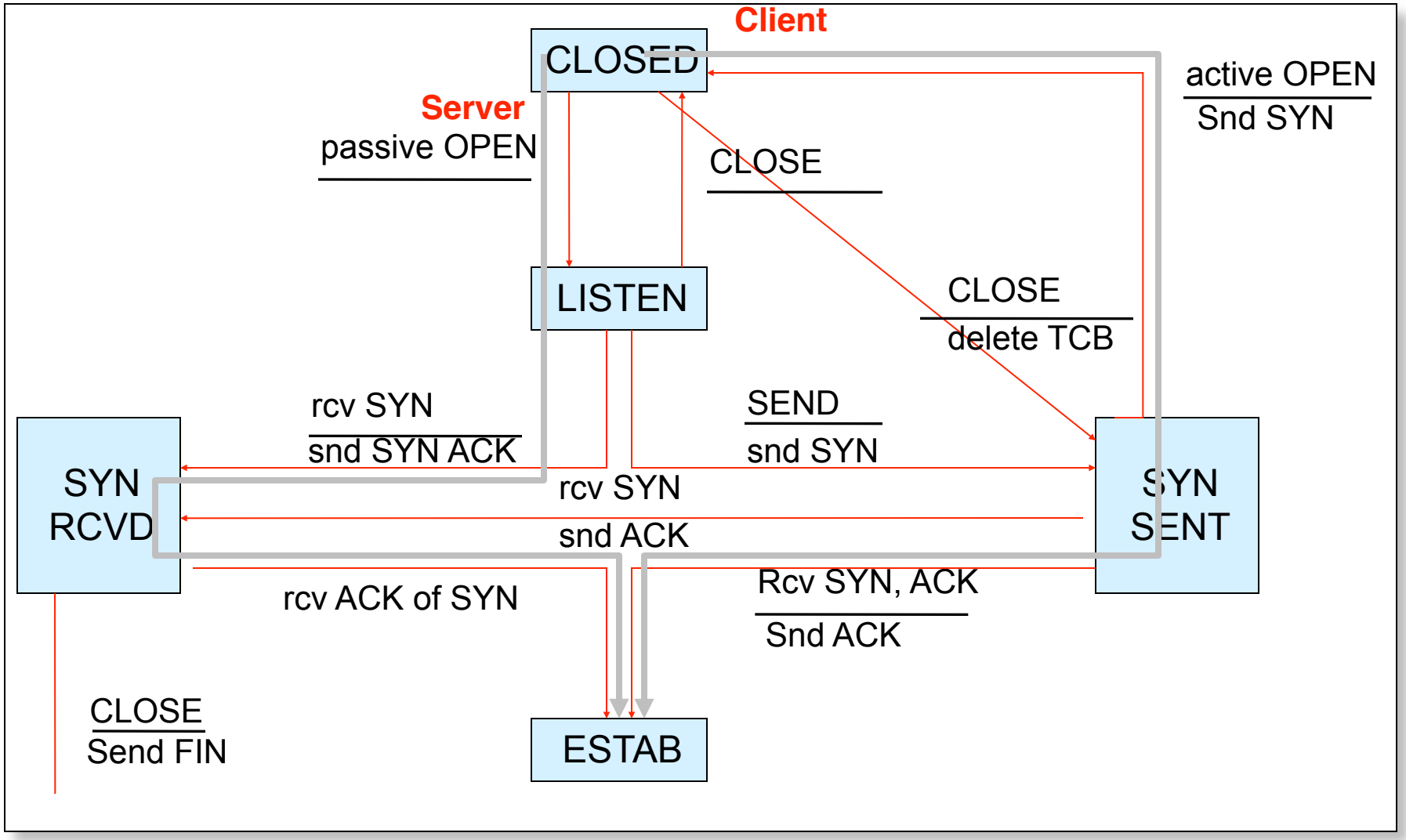
TCP State Diagram: Connection Setup



TCP State Diagram: Connection Setup

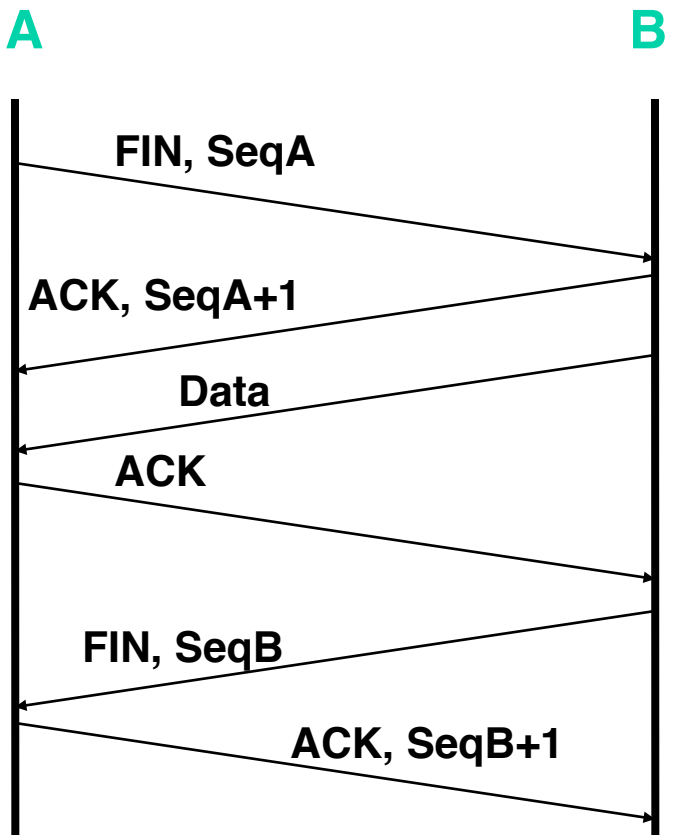


TCP State Diagram: Connection Setup

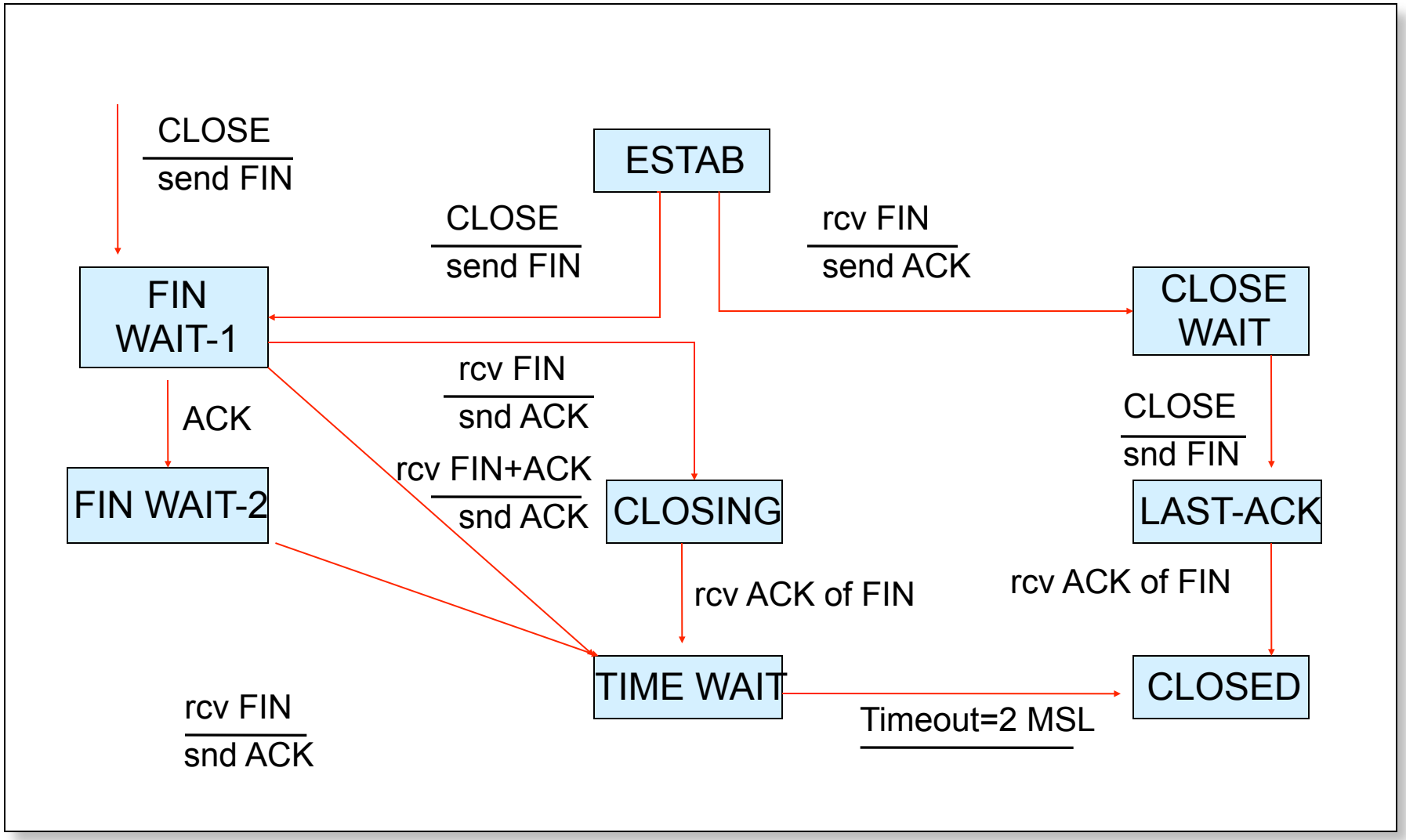


Tearing Down Connection

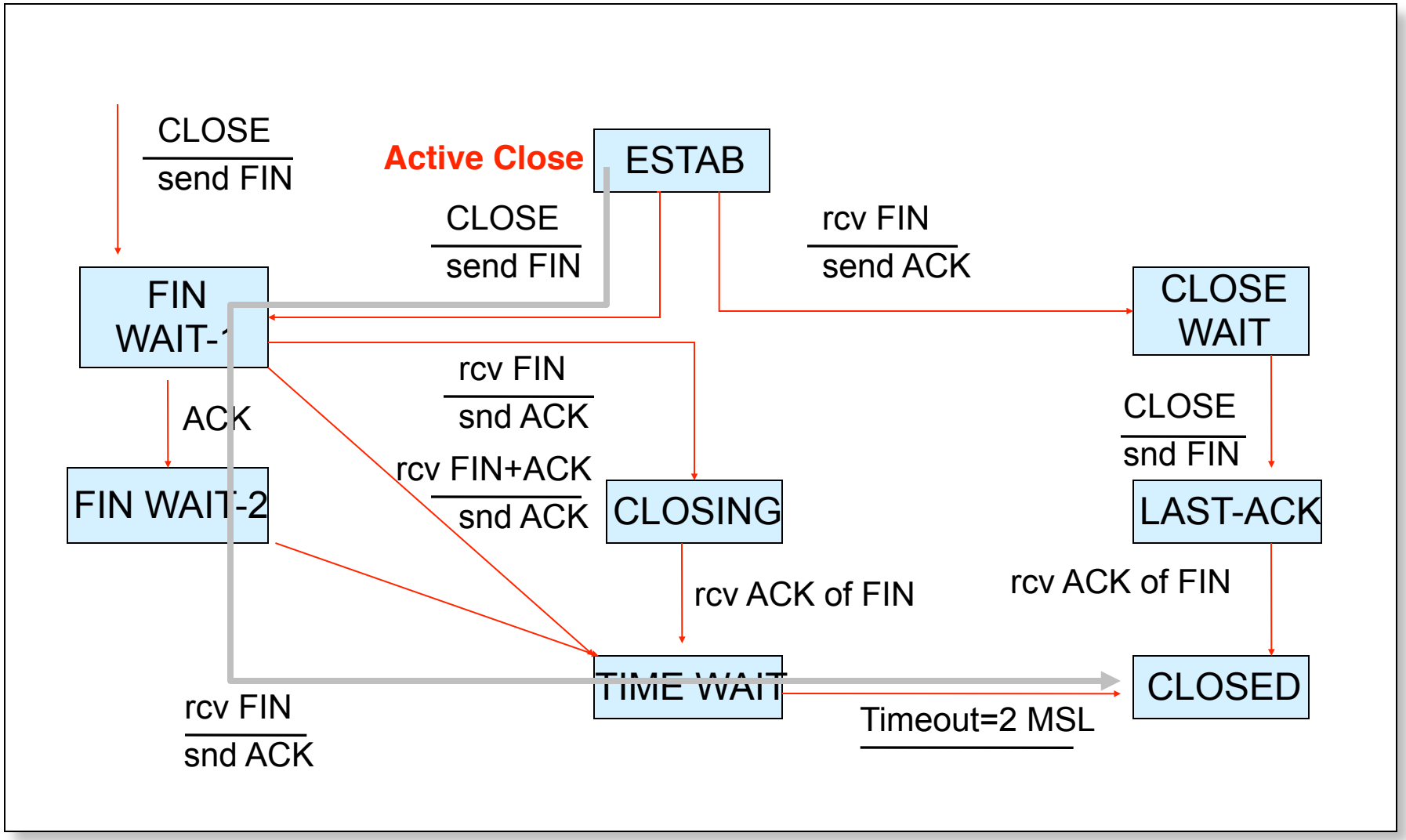
- Either Side Can Initiate Tear Down
  - Send FIN signal
  - "I'm not going to send any more data"
- Other Side Can Continue Sending Data
  - Half open connection
  - Must continue to acknowledge
- Acknowledging FIN
  - Acknowledge last sequence number + 1



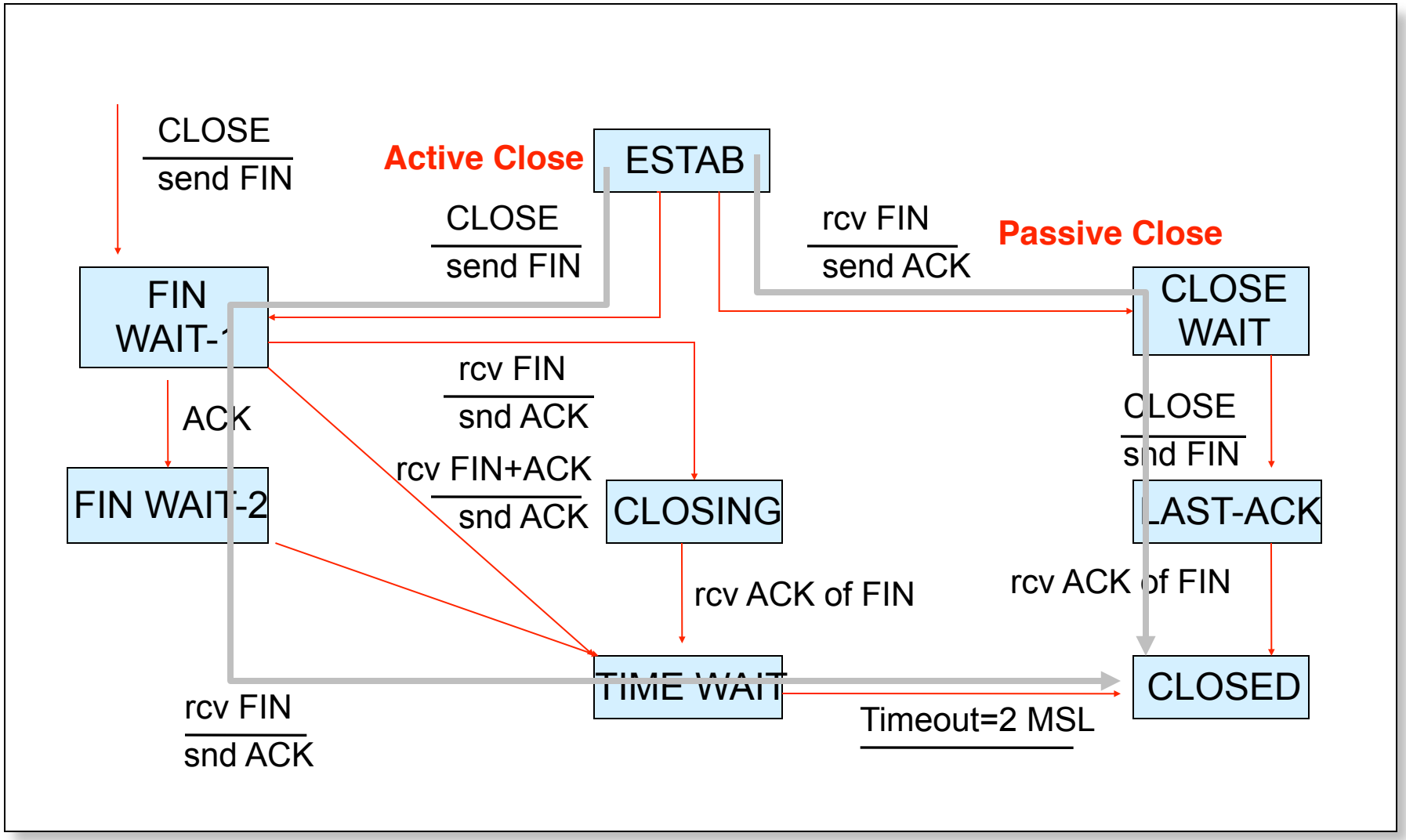
State Diagram: Connection Tear-down



State Diagram: Connection Tear-down

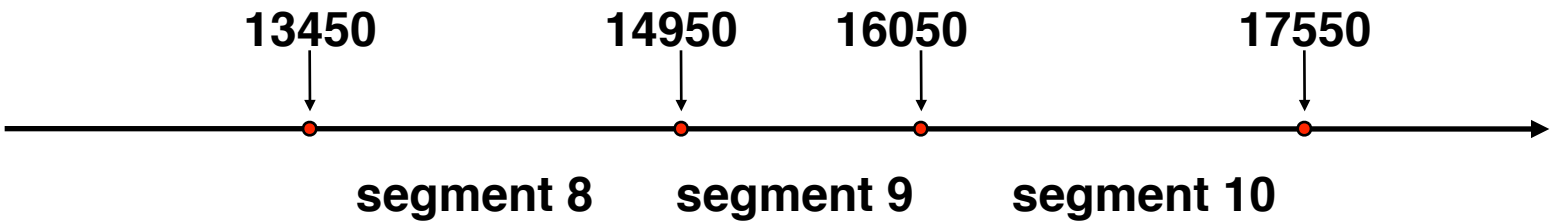


State Diagram: Connection Tear-down



Sequence Number Space

- Each byte in byte stream is numbered.
  - 32 bit value
  - Wraps around
  - Initial values selected at start up time
- TCP breaks up the byte stream in packets ("segments")
  - Packet size is limited to the Maximum Segment Size
  - Set to prevent packet fragmentation
- Each segment has a sequence number.
  - Indicates where it fits in the byte stream



Sequence Numbers

- 32 Bits, Unsigned
- Why So Big?
  - For sliding window, must have

    |Sequence Space| > 2* |Sending Window|
    - 2^32 > 2 * 2^16. No problem
  - Also, want to guard against stray packets
    - With IP, assume packets have maximum segment lifetime (MSL) of 120s
      - i.e. can linger in network for upto 120s
    - Sequence number would wrap around in this time at 286Mbps

Error Control

Error Control

- Checksum (mostly) guarantees end-end data integrity.

Error Control

- Checksum (mostly) guarantees end-end data integrity.
- Sequence numbers detect packet sequencing problems:
  - Duplicate: ignore
  - Reordered: reorder or drop
  - Lost: retransmit

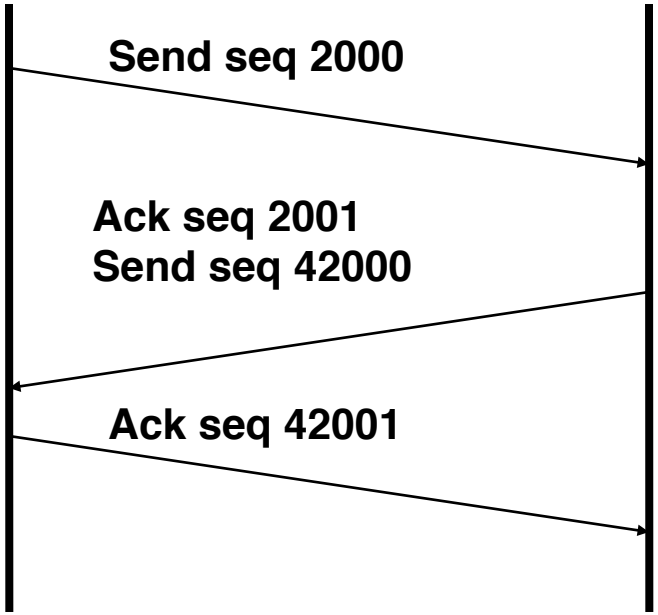
Error Control

- Checksum (mostly) guarantees end-end data integrity.
- Sequence numbers detect packet sequencing problems:
  - Duplicate: ignore
  - Reordered: reorder or drop
  - Lost: retransmit
- Lost segments detected by sender.
  - Use time out to detect lack of acknowledgment
  - Need estimate of the roundtrip time to set timeout

Error Control

- Checksum (mostly) guarantees end-end data integrity.
- Sequence numbers detect packet sequencing problems:
  - Duplicate: ignore
  - Reordered: reorder or drop
  - Lost: retransmit
- Lost segments detected by sender.
  - Use time out to detect lack of acknowledgment
  - Need estimate of the roundtrip time to set timeout
- Retransmission requires that sender keep copy of the 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

TCP Flow Control

- Sliding window protocol
  - For window size *n*, can send up to *n* bytes without receiving an acknowledgement
  - When the data are acknowledged then the window slides forward
- Window size determines
  - How much unacknowledged data can the sender sends
- But there is more detail

Complication!

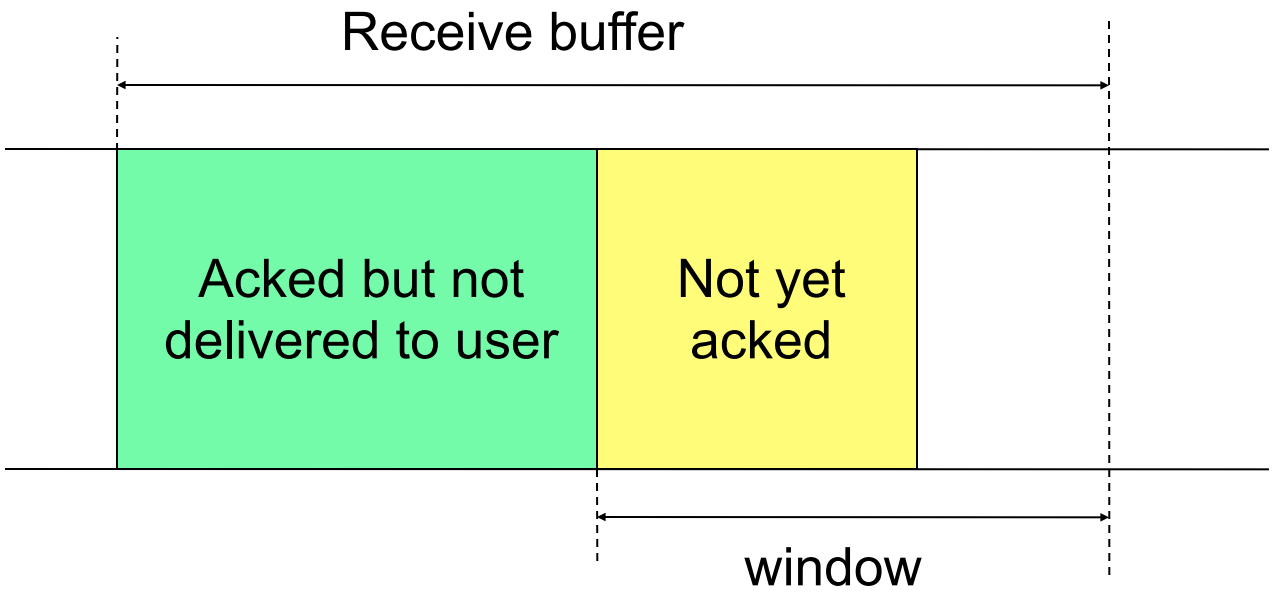
- TCP receiver can delete acknowledged data only after the data has been delivered to the application
- So, depending on how fast the application is reading the data, the receiver's window size may change!!!

Solution

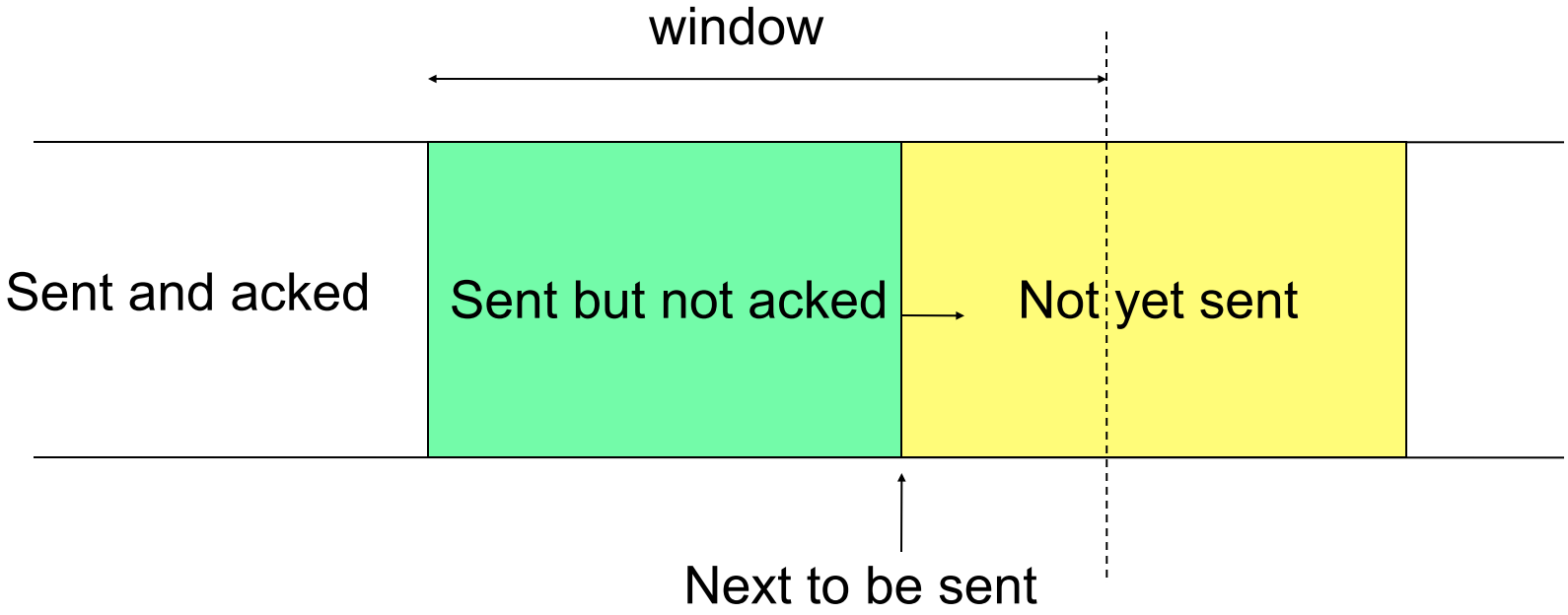
- Receiver tells sender what is the current window size in every packet it transmits to the sender
- Sender uses this current window size instead of a fixed value

- Window size (also called Advertised window) is continuously changing
- Can go to zero!
  - Sender not allowed to send anything!

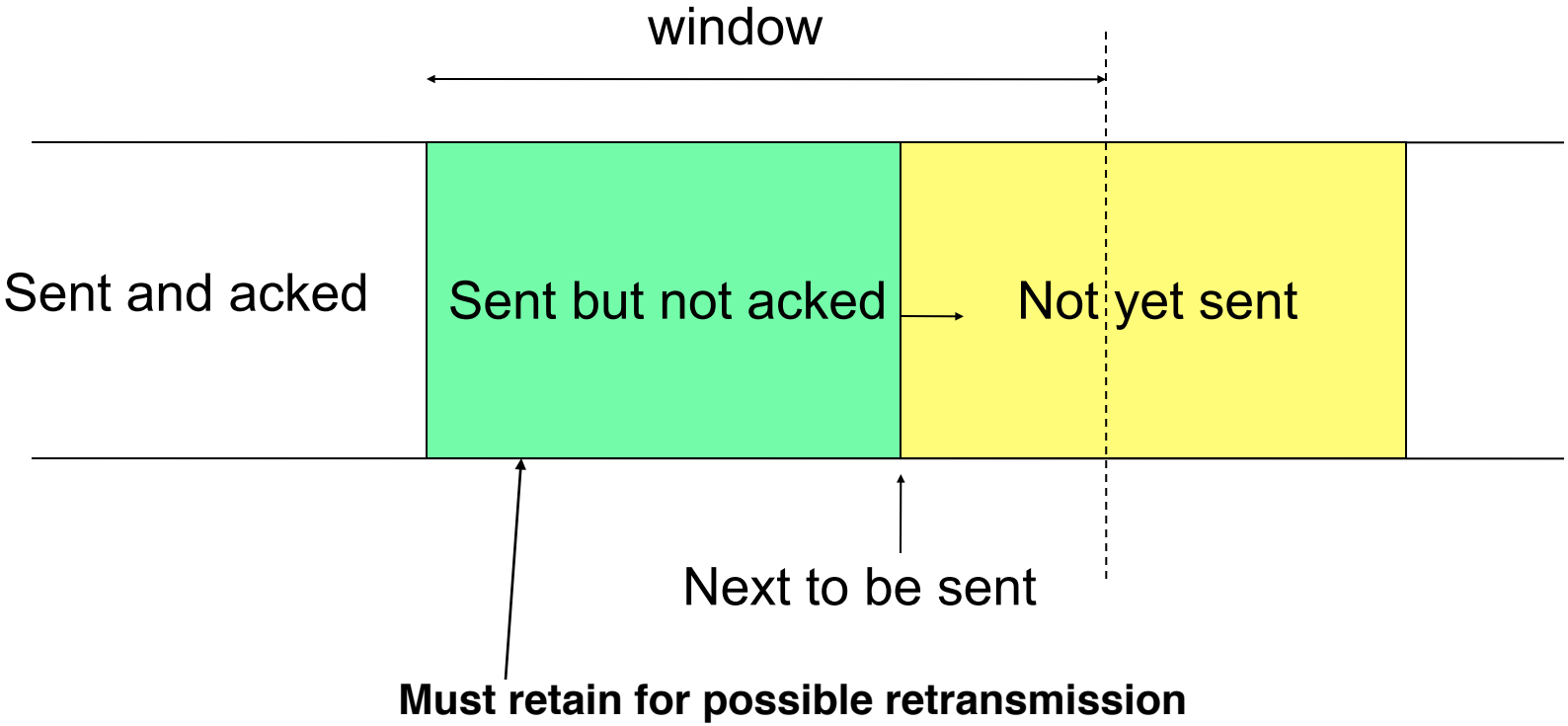
Window Flow Control: Receive Side



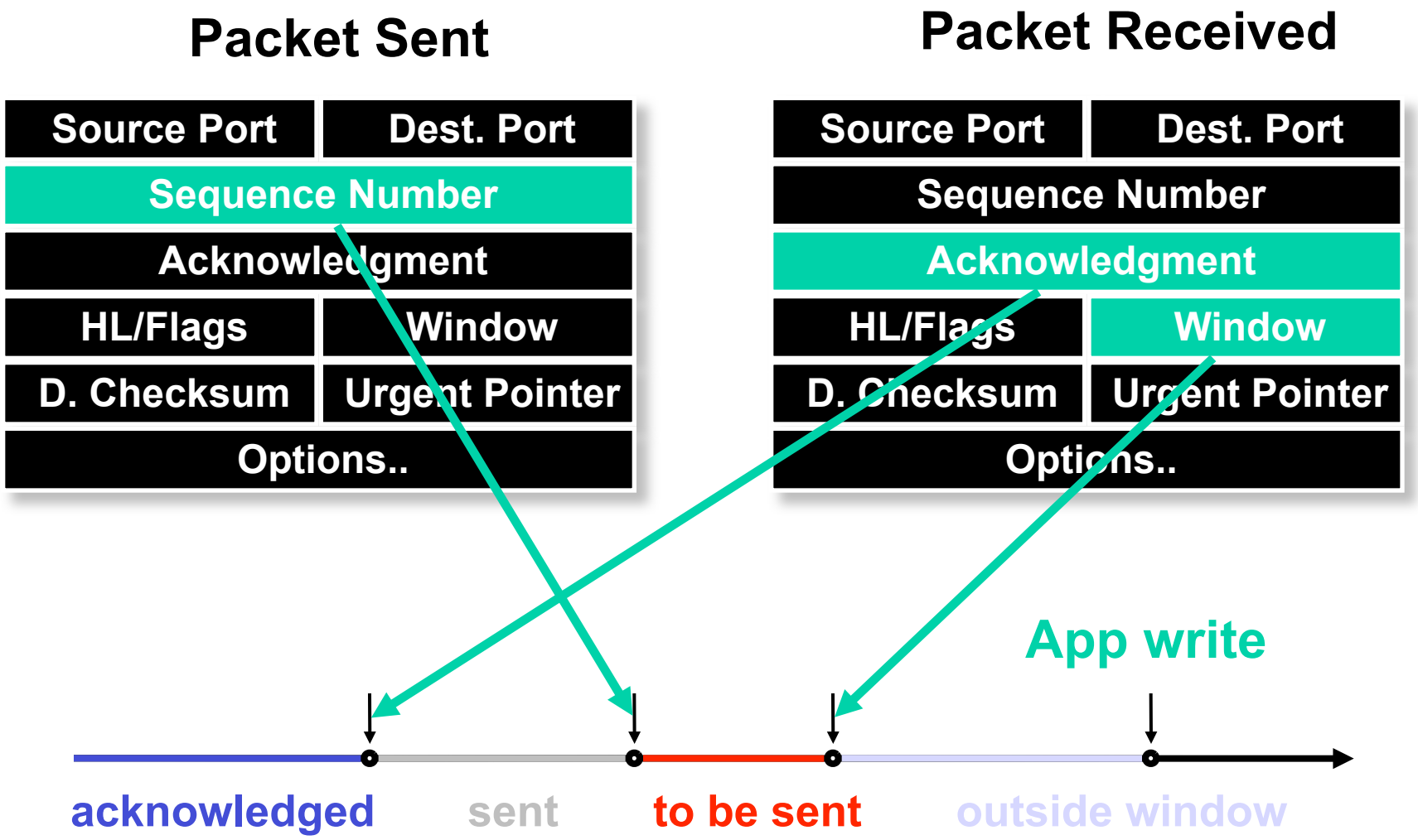
Window Flow Control: Send Side



Window Flow Control: Send Side



Window Flow Control: Send Side



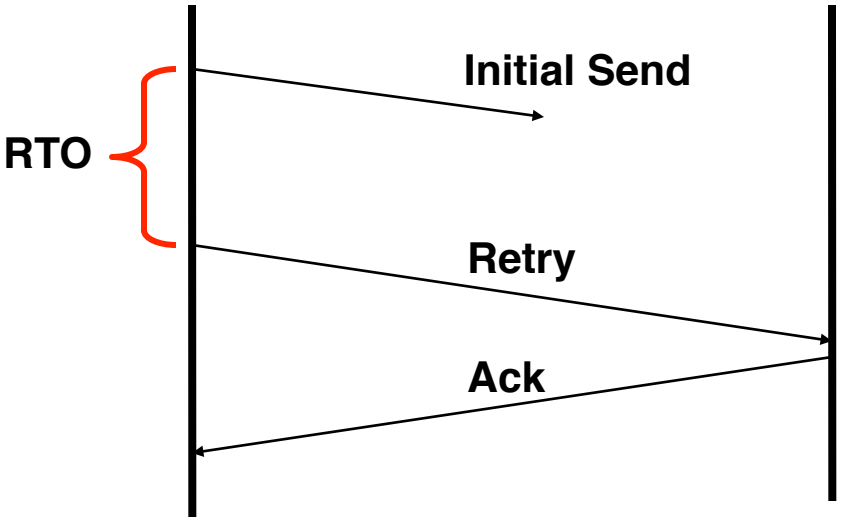
Ongoing Communication

- Bidirectional Communication
  - Each side acts as sender & receiver
  - Every message contains acknowledgement of received sequence
    - Even if no new data have been received
  - Every message advertises window size
    - Size of its receiving window
  - Every message contains sent sequence number
    - Even if no new data being sent
- When Does Sender Actually Send Message?
  - When sending buffer contains at least max. segment size (- header sizes) bytes
  - When application tells it
    - Set PUSH flag for last segment sent
  - When timer expires

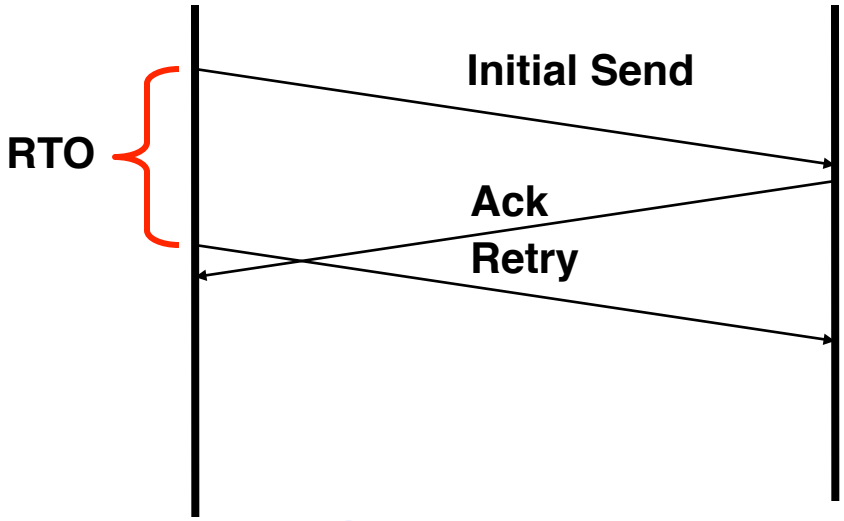
TCP Must Operates Over Any Internet Path

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

Setting Retransmission Timeout (RTO)



Detect dropped packet

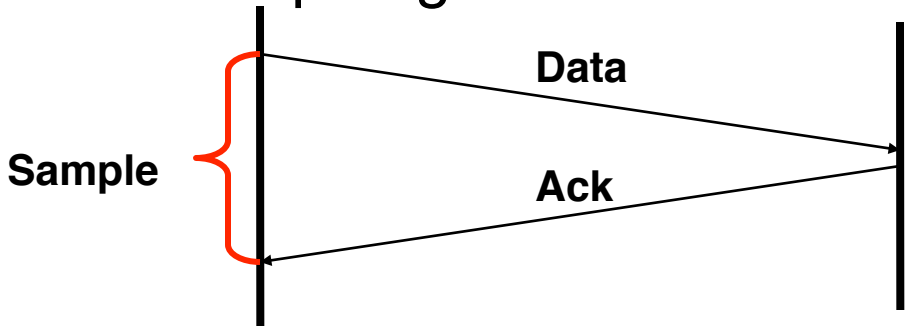


RTO too short

  - 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)

Round-trip Time Estimation

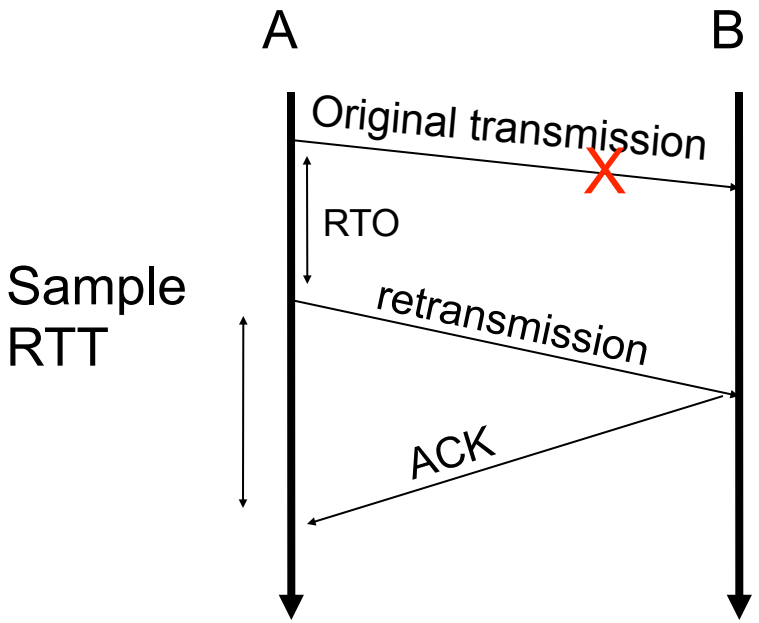
- Every Data/Ack pair gives new RTT estimate



Original TCP Round-trip Estimator

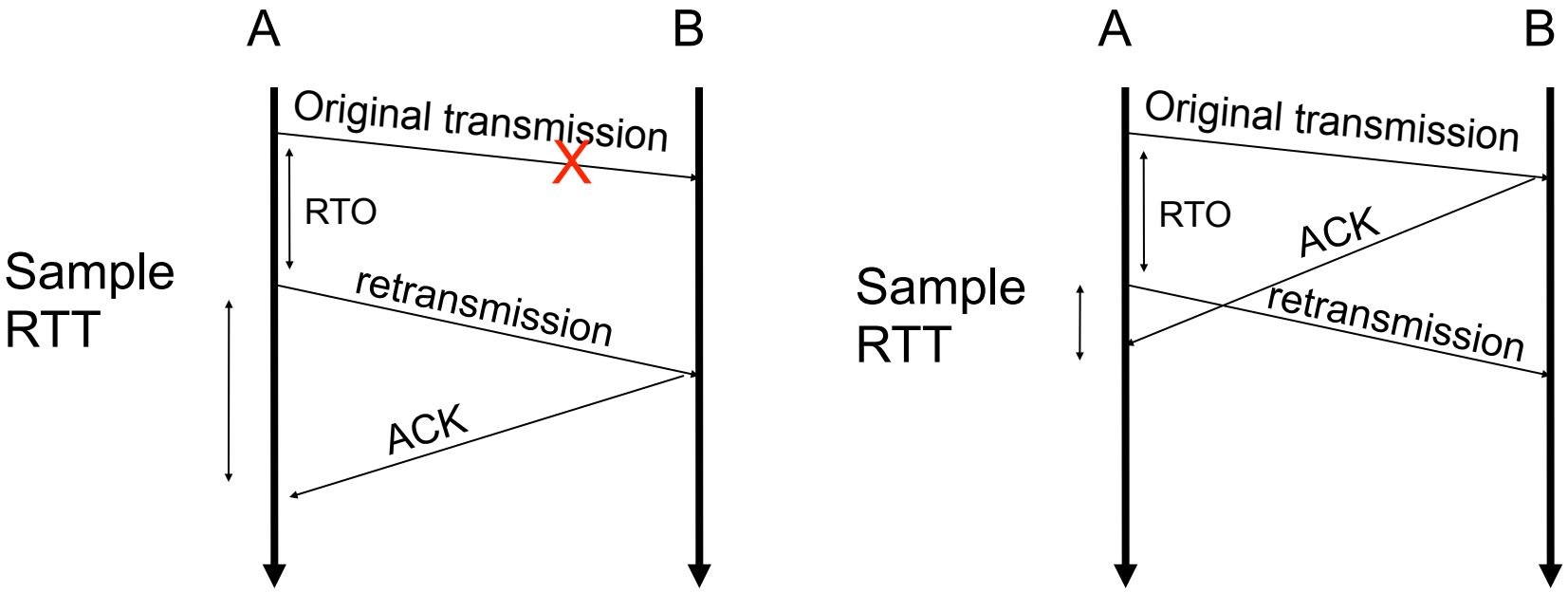
- Round trip times estimated as a moving average:
  - New RTT = α (old RTT) + (1 - α) (new sample)
  - Recommended value for α: 0.8 - 0.9
    - 0.875 for most TCP's
- Retransmit timer set to β RTT, where β = 2
  - Want to be somewhat conservative about retransmitting

RTT Sample Ambiguity



- Ignore sample for segment that has been retransmitted

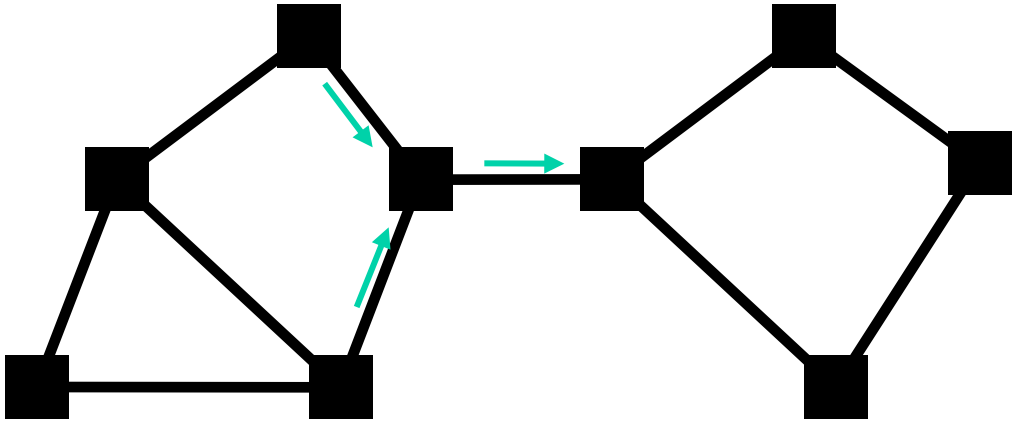
RTT Sample Ambiguity



- Ignore sample for segment that has been retransmitted

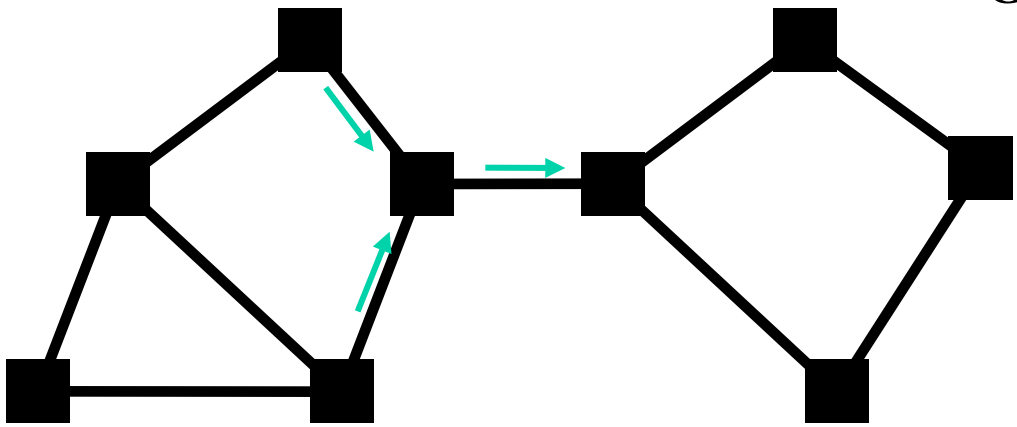
What is Congestion?

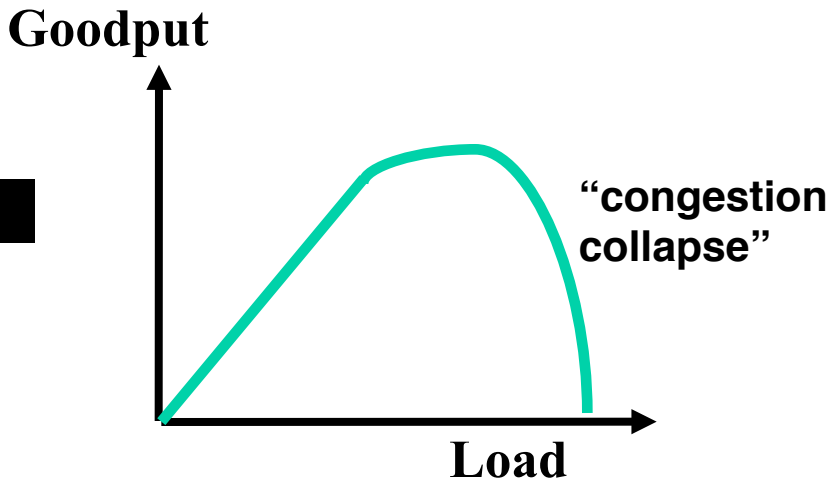
- The load placed on the network is higher than the capacity of the network
  - Not surprising: independent senders place load on network
- Results in packet loss: routers have no choice
  - Can only buffer finite amount of data
  - End-to-end protocol will typically react, e.g. TCP



Why is Congestion Bad?

- Wasted bandwidth: retransmission of dropped packets
- Poor user service : unpredictable delay, low user goodput
- Increased load can even result in lower network goodput
  - Switched nets: packet losses create lots of retransmissions
  - Broadcast Ethernet: high demand -> many collisions





Sending Rate of Sliding Window Protocol

Sending Rate of Sliding Window Protocol

- Suppose A uses a sliding window protocol to transmit a large data file to B

Sending Rate of Sliding Window Protocol

- Suppose A uses a sliding window protocol to transmit a large data file to B
- Window size = 64KB

Sending Rate of Sliding Window Protocol

- Suppose A uses a sliding window protocol to transmit a large data file to B
- Window size = 64KB
- Network round-trip delay is 1 second

Sending Rate of Sliding Window Protocol

- Suppose A uses a sliding window protocol to transmit a large data file to B
- Window size = 64KB
- Network round-trip delay is 1 second

Sending Rate of Sliding Window Protocol

- Suppose A uses a sliding window protocol to transmit a large data file to B
- Window size = 64KB
- Network round-trip delay is 1 second

- What's the expected sending rate?

Sending Rate of Sliding Window Protocol

- Suppose A uses a sliding window protocol to transmit a large data file to B
- Window size = 64KB
- Network round-trip delay is 1 second

- What's the expected sending rate?
- 64KB/second

Sending Rate of Sliding Window Protocol

- Suppose A uses a sliding window protocol to transmit a large data file to B
- Window size = 64KB
- Network round-trip delay is 1 second

- What's the expected sending rate?
- 64KB/second

Sending Rate of Sliding Window Protocol

- Suppose A uses a sliding window protocol to transmit a large data file to B
- Window size = 64KB
- Network round-trip delay is 1 second

- What's the expected sending rate?
- 64KB/second

- What if a network link is only 64KB/second but there are 1000 people who are transferring files over that link using the sliding window protocol?

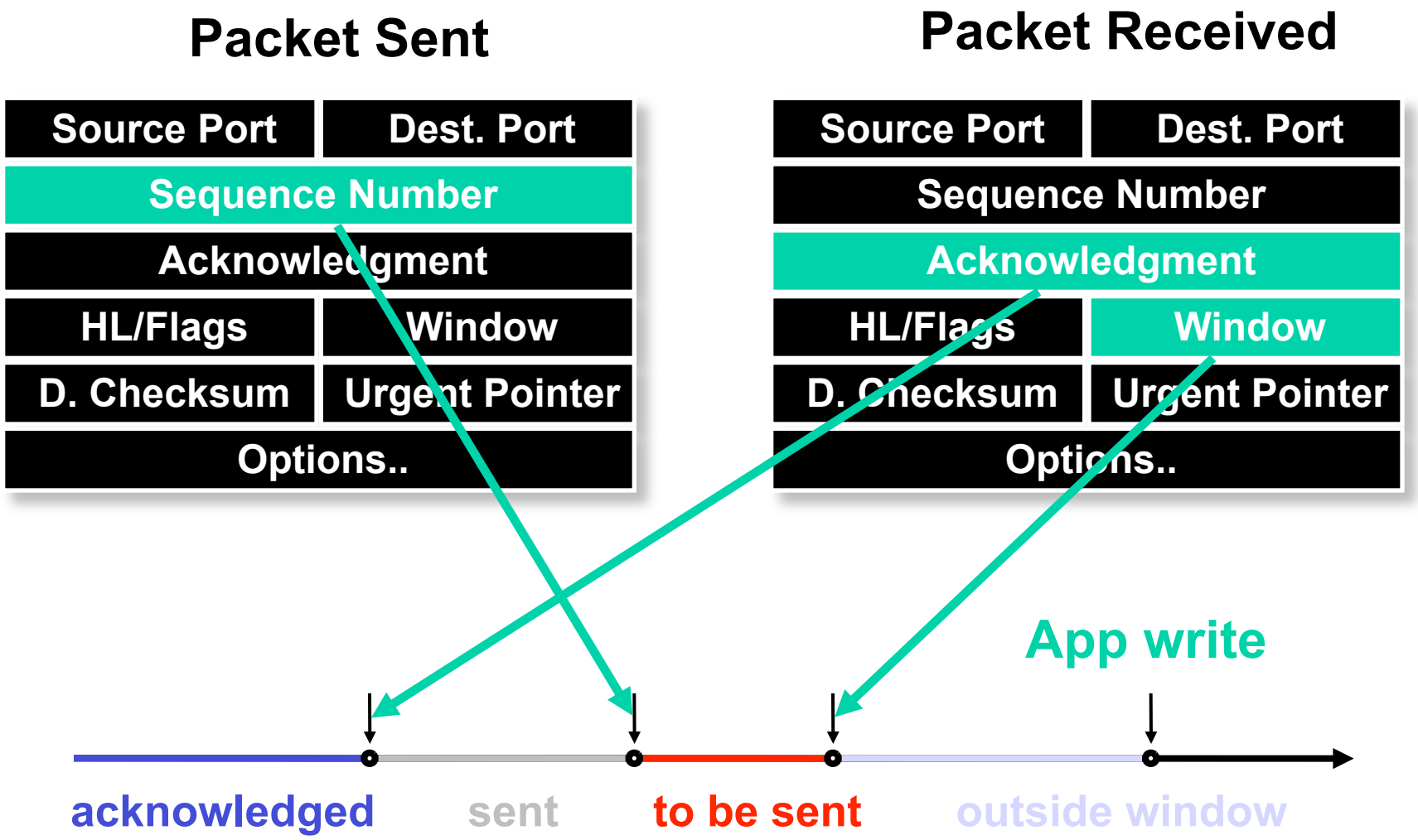
Sending Rate of Sliding Window Protocol

- Suppose A uses a sliding window protocol to transmit a large data file to B
- Window size = 64KB
- Network round-trip delay is 1 second

- What's the expected sending rate?
- 64KB/second

- What if a network link is only 64KB/second but there are 1000 people who are transferring files over that link using the sliding window protocol?
- Packet losses, timeouts, retransmissions, more packet losses… nothing useful gets through, congestion collapse!

TCP Window Flow Control



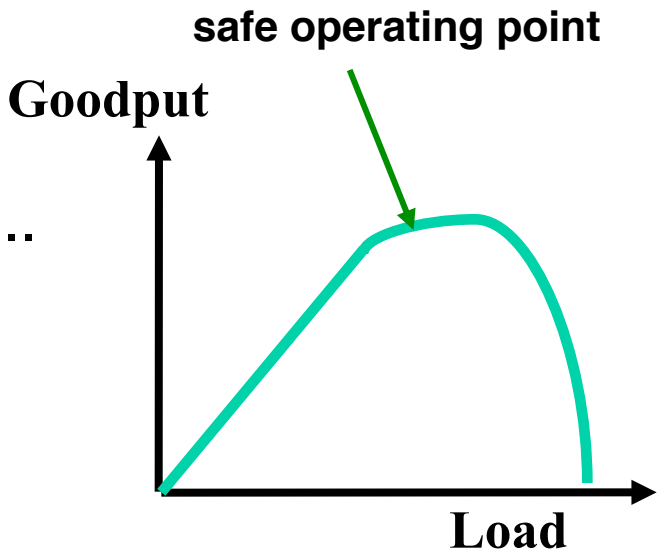
TCP Flow Control Alone Is Not Enough

- We have talked about how TCP's advertised window is used for flow control
  - To keep sender sending faster than the receiver can handle
- If the receiver is sufficiently fast, then the advertised window will be maximized at all time
- But clearly, this will lead to congestion collapse as the previous example if there are too many senders or network is too slow

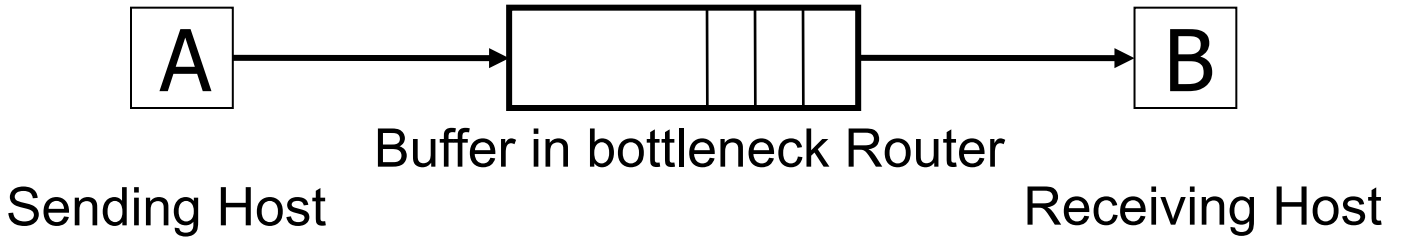
- Key 1: Window size determines sending rate
- Key 2: Window size must be dynamically adjusted to prevent congestion collapse

How Fast to Send? What's at Stake?

- Send too slow: link sits idle
  - wastes time
- Send too fast: link is kept busy but....
  - queue builds up in router buffer (delay)
  - overflow buffers in routers (loss)
  - Many retransmissions, many losses
  - Network goodput goes down



Abstract View



- We ignore internal structure of network and model it as having a single bottleneck link

Three Congestion Control Problems

- Adjusting to bottleneck bandwidth
- Adjusting to variations in bandwidth
- Sharing bandwidth between flows