CS3600 — Systems and Networks

NORTHEASTERN UNIVERSITY

Lecture 23: TCP

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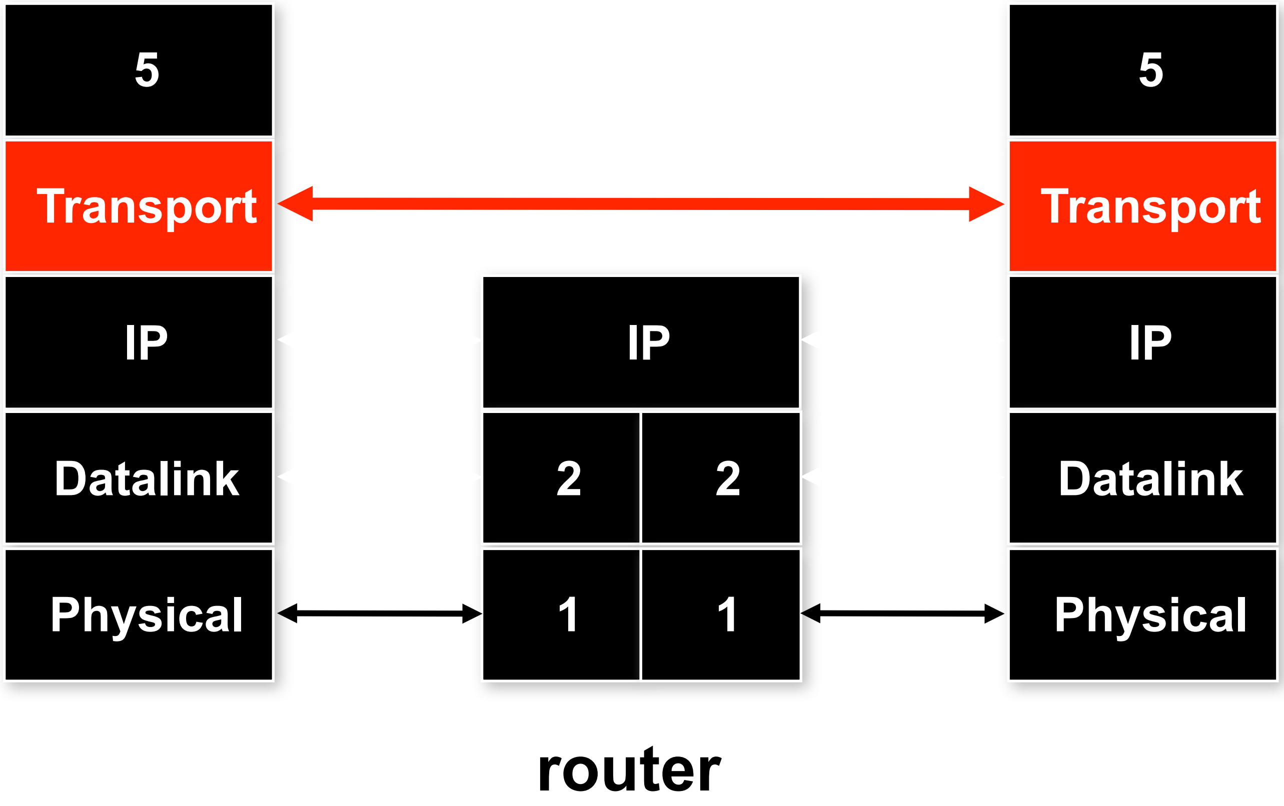
Slides used with permissions from Edward W. Knightly,
T. S. Eugene Ng, Ion Stoica, Hui Zhang

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



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- Congestion control.
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- More….

User Datagram Protocol (UDP)

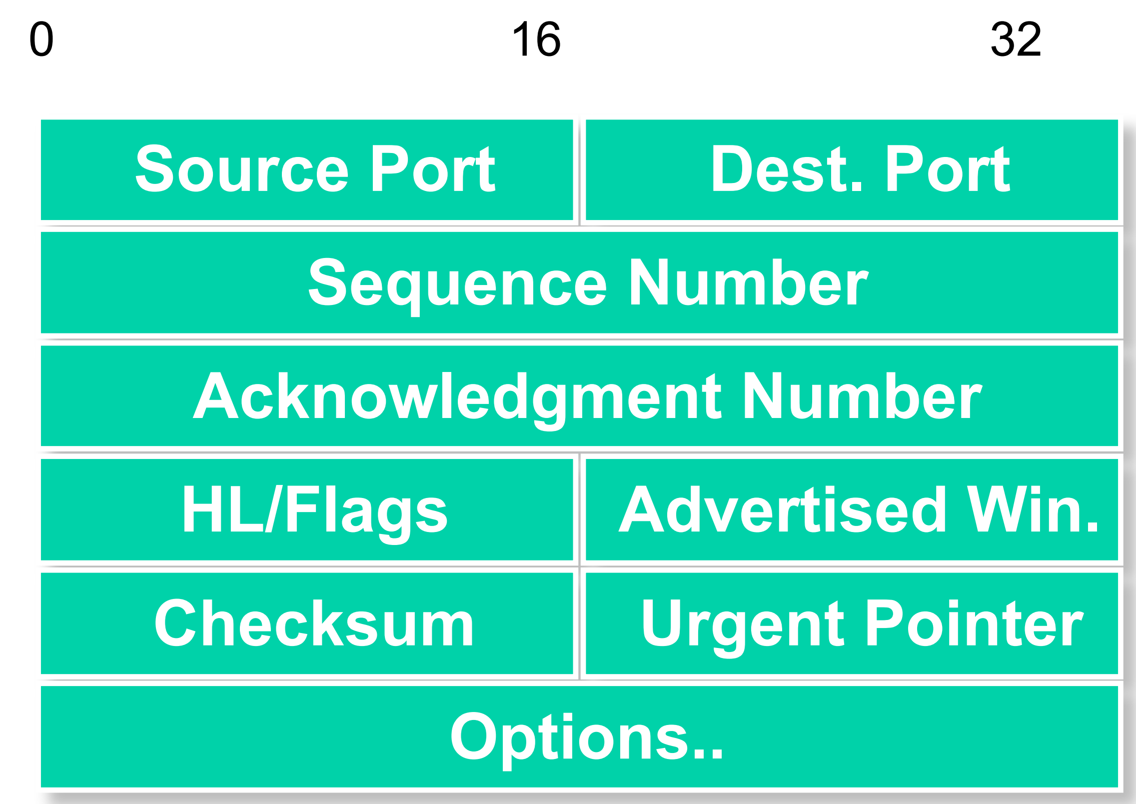
- Connectionless datagram
  - Socket: SOCK_DGRAM
- Port number used for (de)multiplexing
  - port numbers = connection/application endpoint
- Adds end-to-end reliability through optional checksum
  - protects against data corruption errors between source and destination (links, switches/routers, bus)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

0	16	32
Source Port	**Dest. Port**	
Length	**Checksum**	

Using UDP

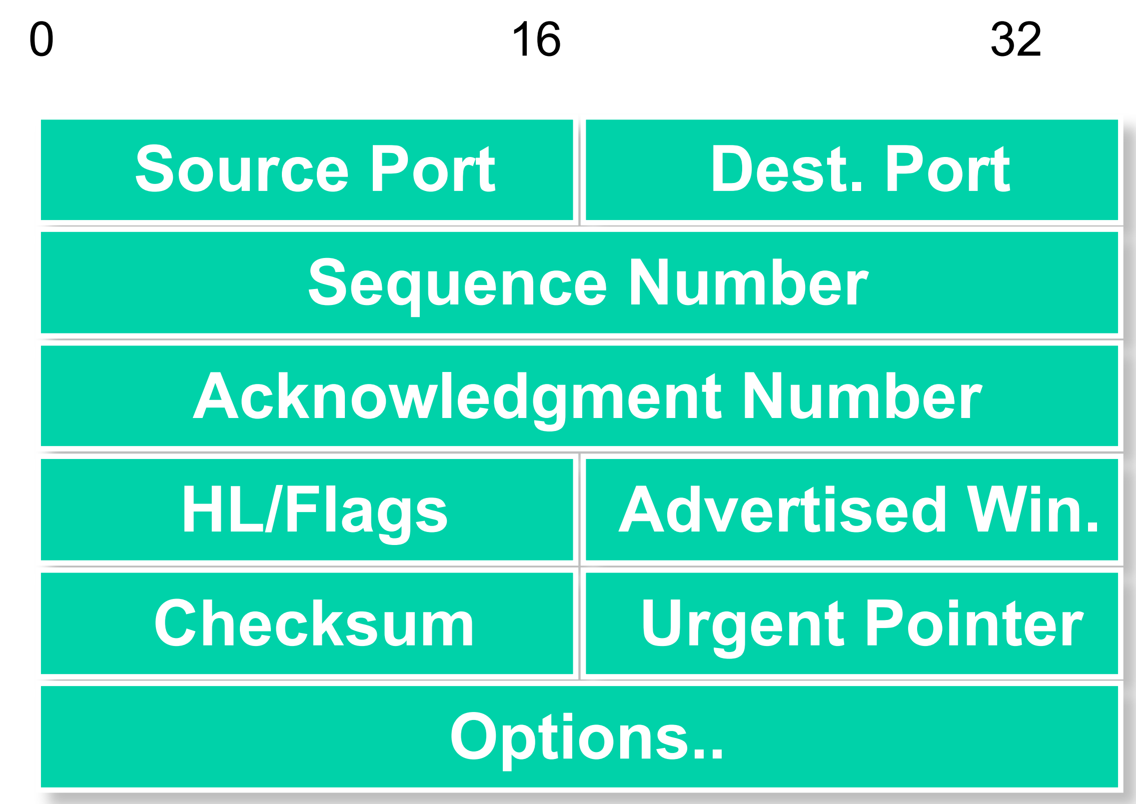
- Custom protocols/applications can be implemented on top of UDP
  - use the port addressing provided by UDP
  - implement own reliability, flow control, ordering, congestion control as it sees fit
- Examples:
  - remote procedure call
  - Multimedia streaming (real time protocol)
  - distributed computing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

Transmission Control Protocol (TCP)



Transmission Control Protocol (TCP)

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



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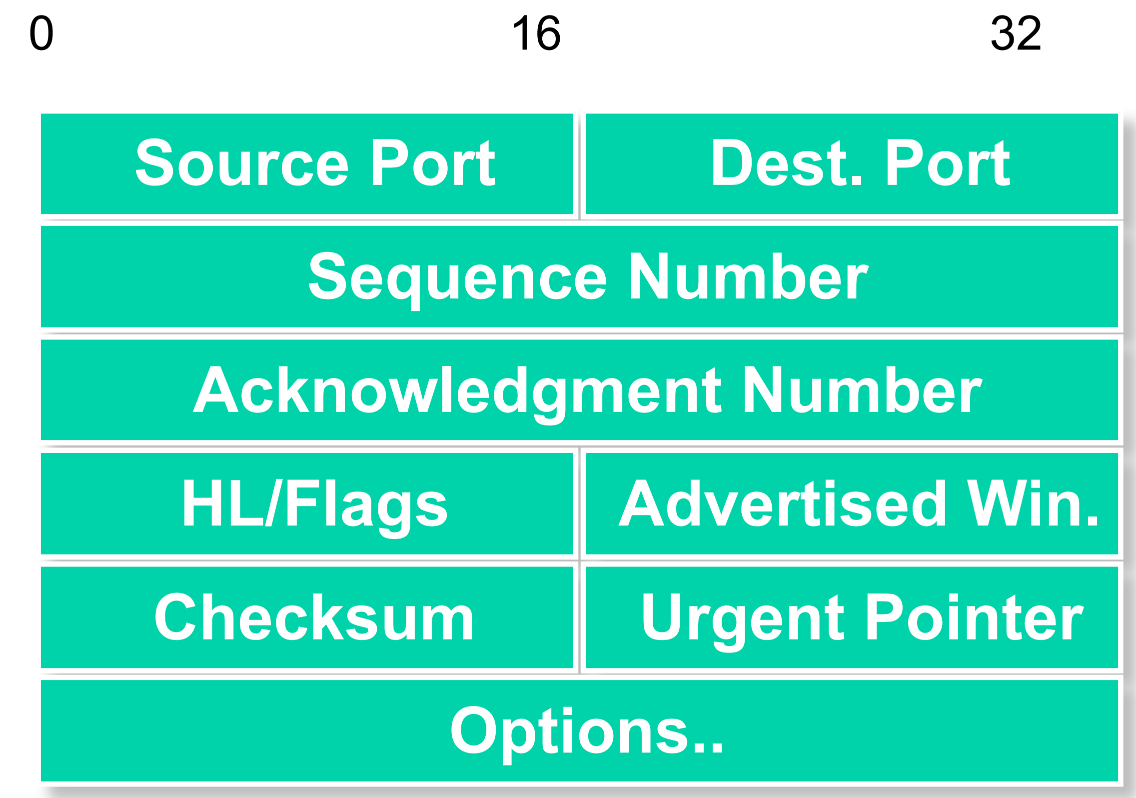

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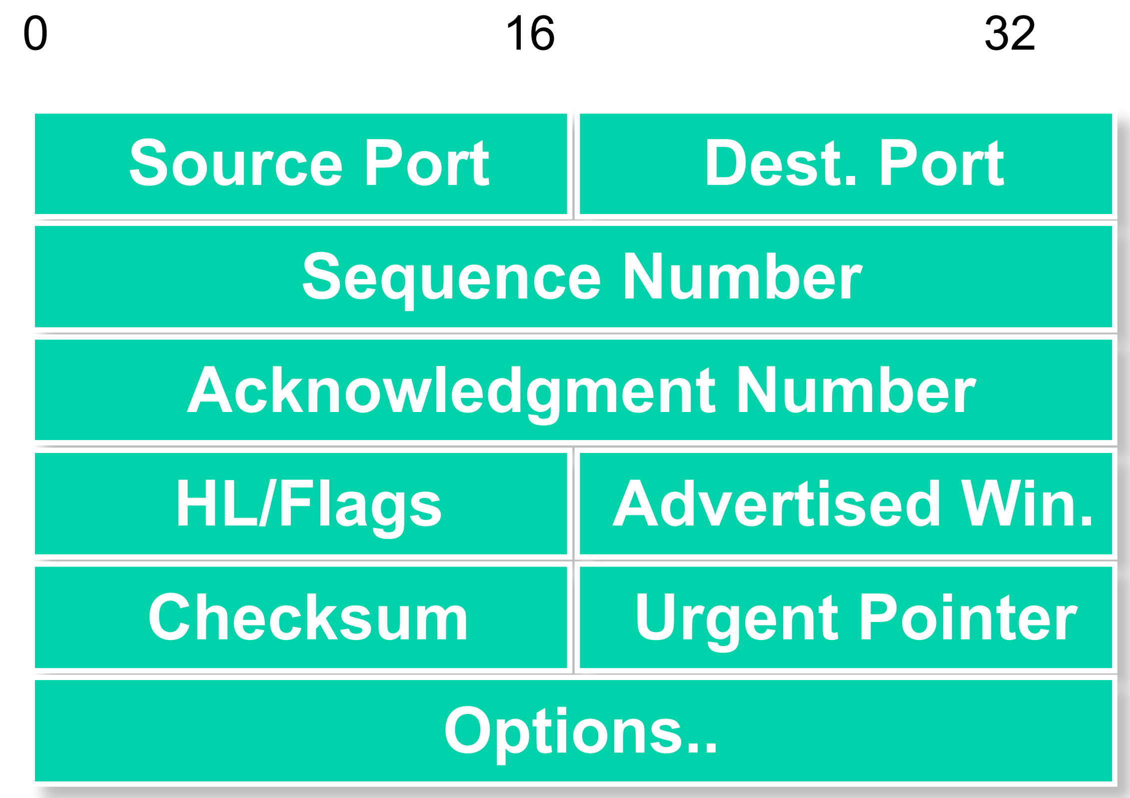

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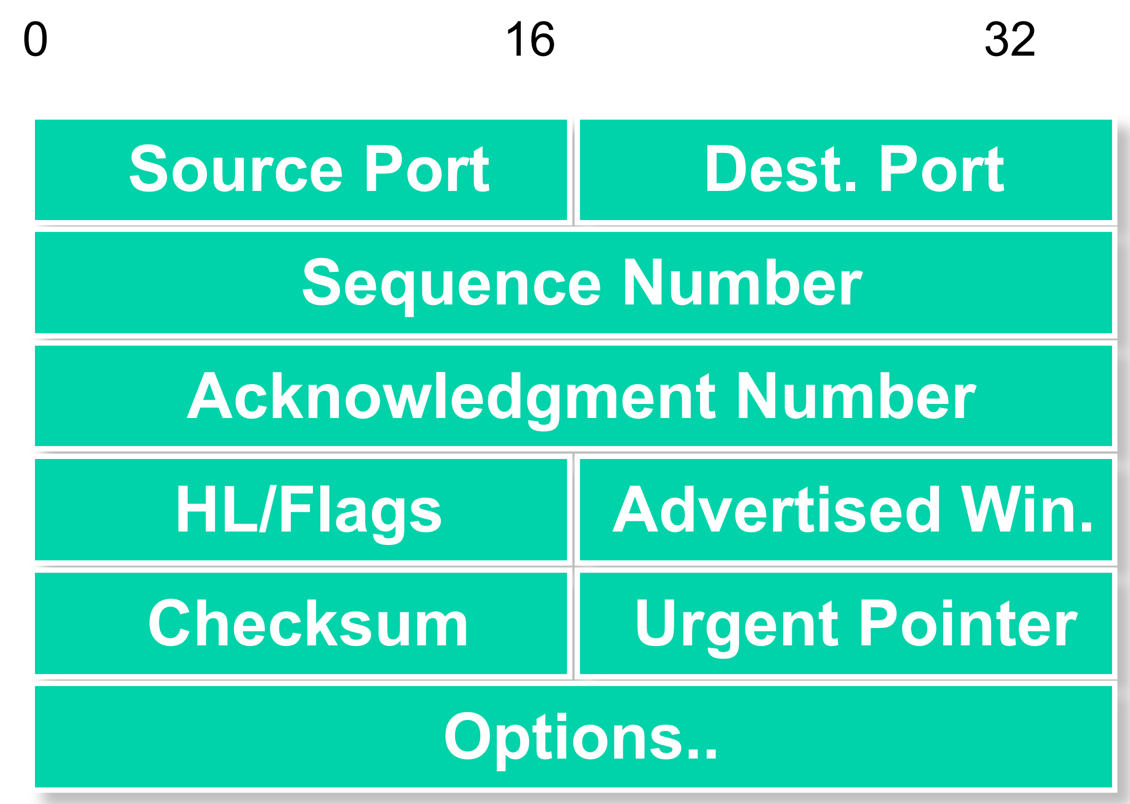

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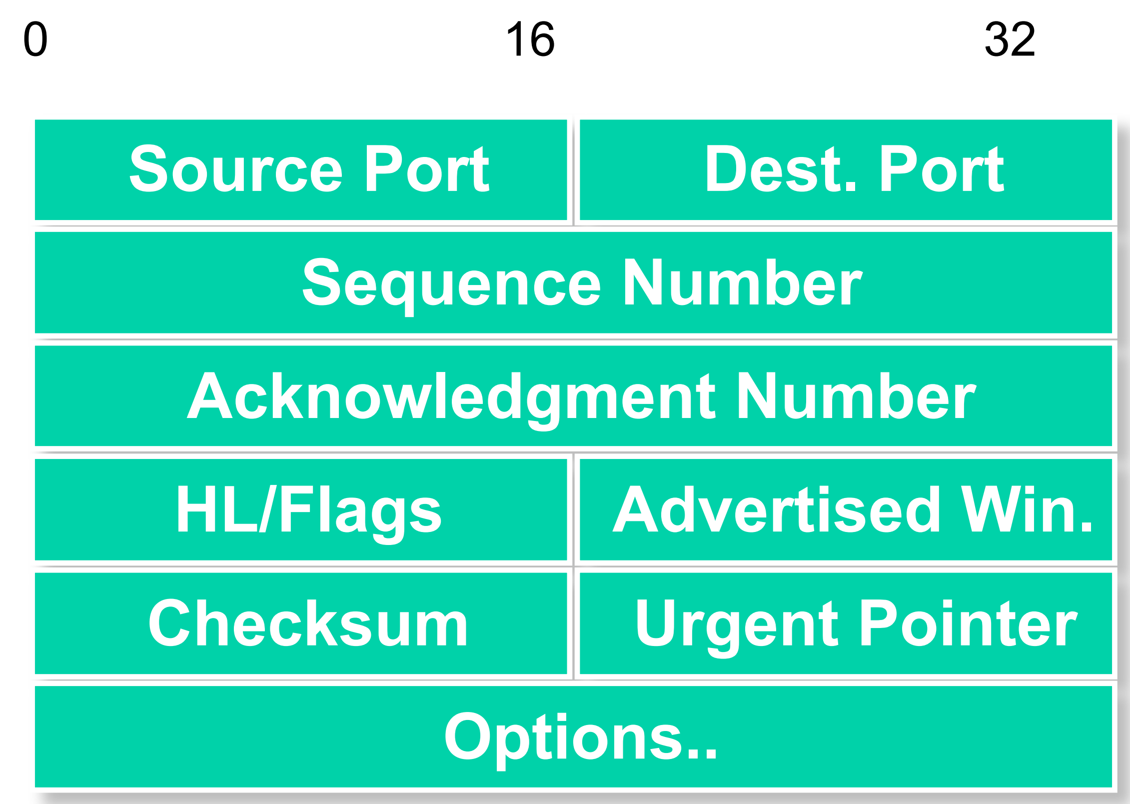

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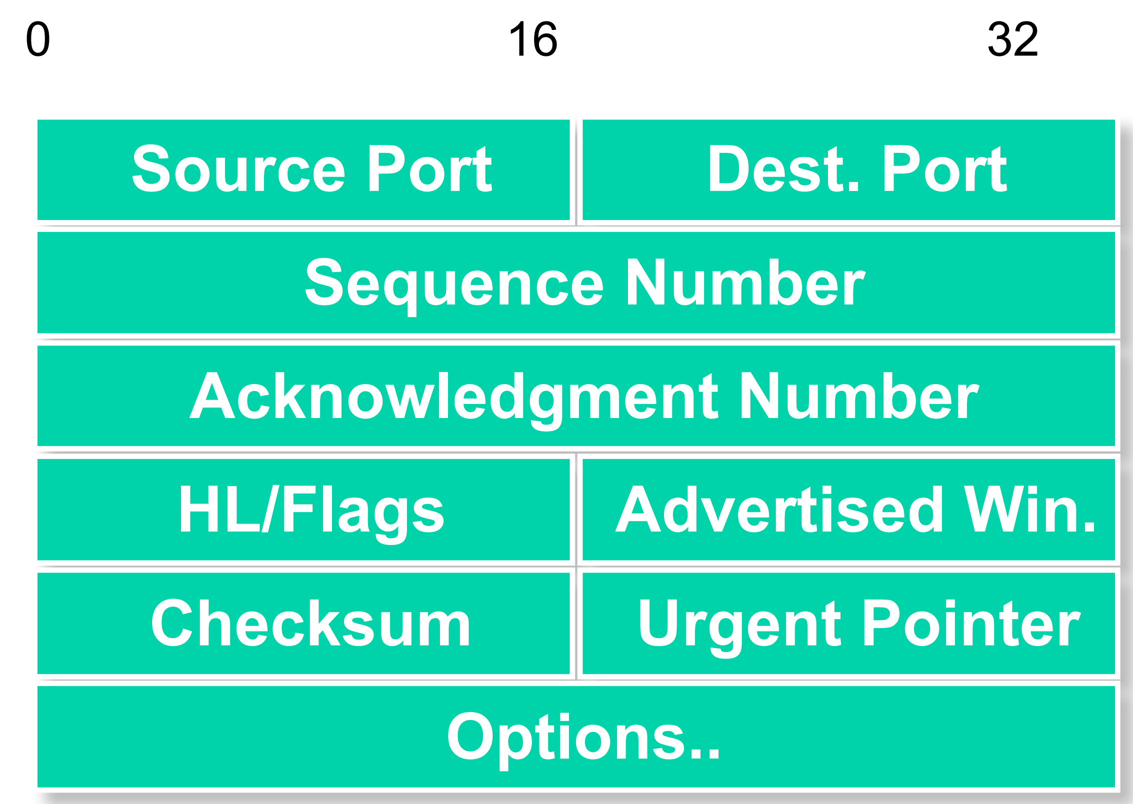
- Congestion avoidance
  - Avoid creating traffic jams within network



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

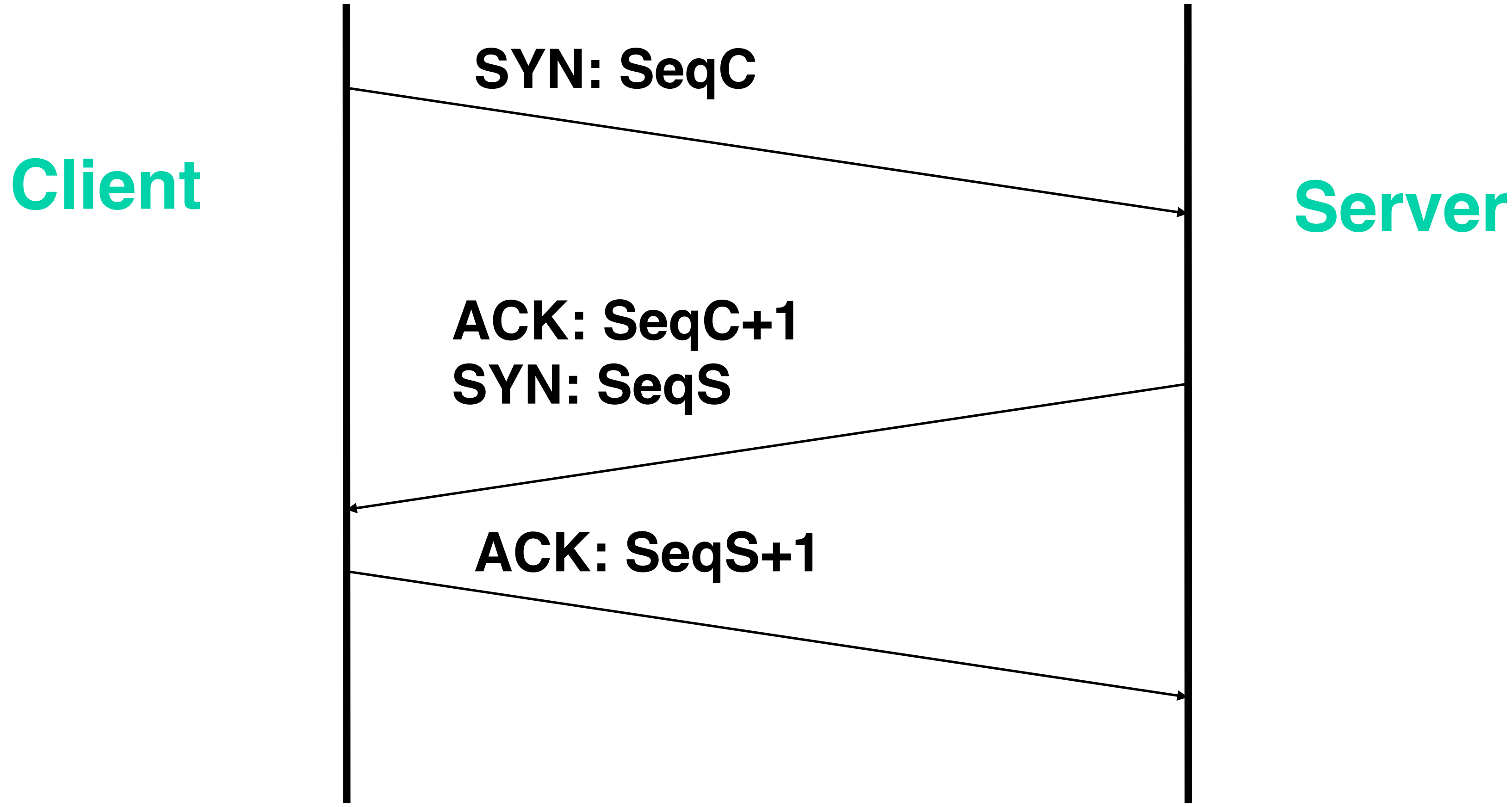
- Why need connection setup?

- 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

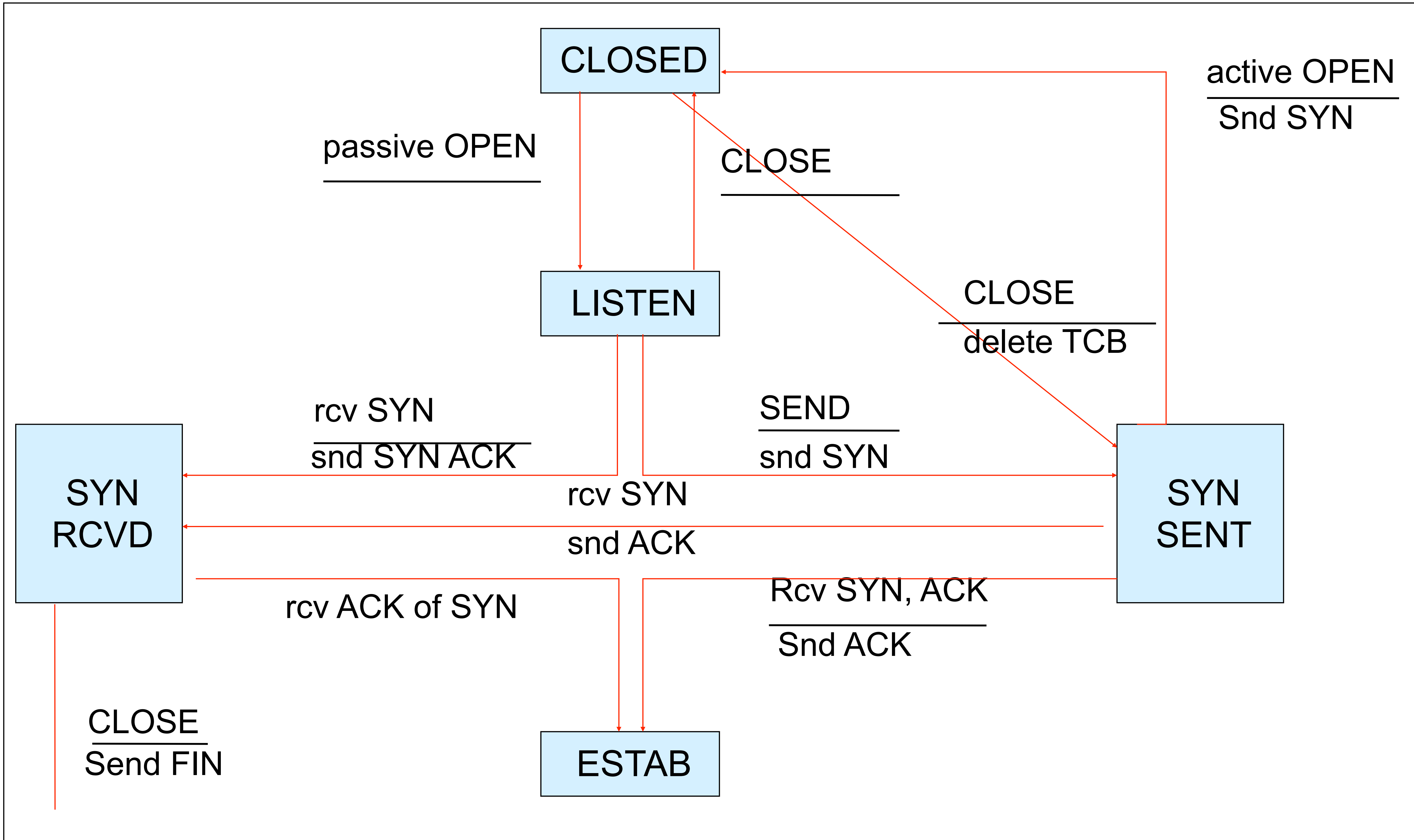
- SYN: Synchronize
  - Used when setting up connection
- FIN: Finish
  - Used when tearing down connection
- ACK
  - Acknowledging received data

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



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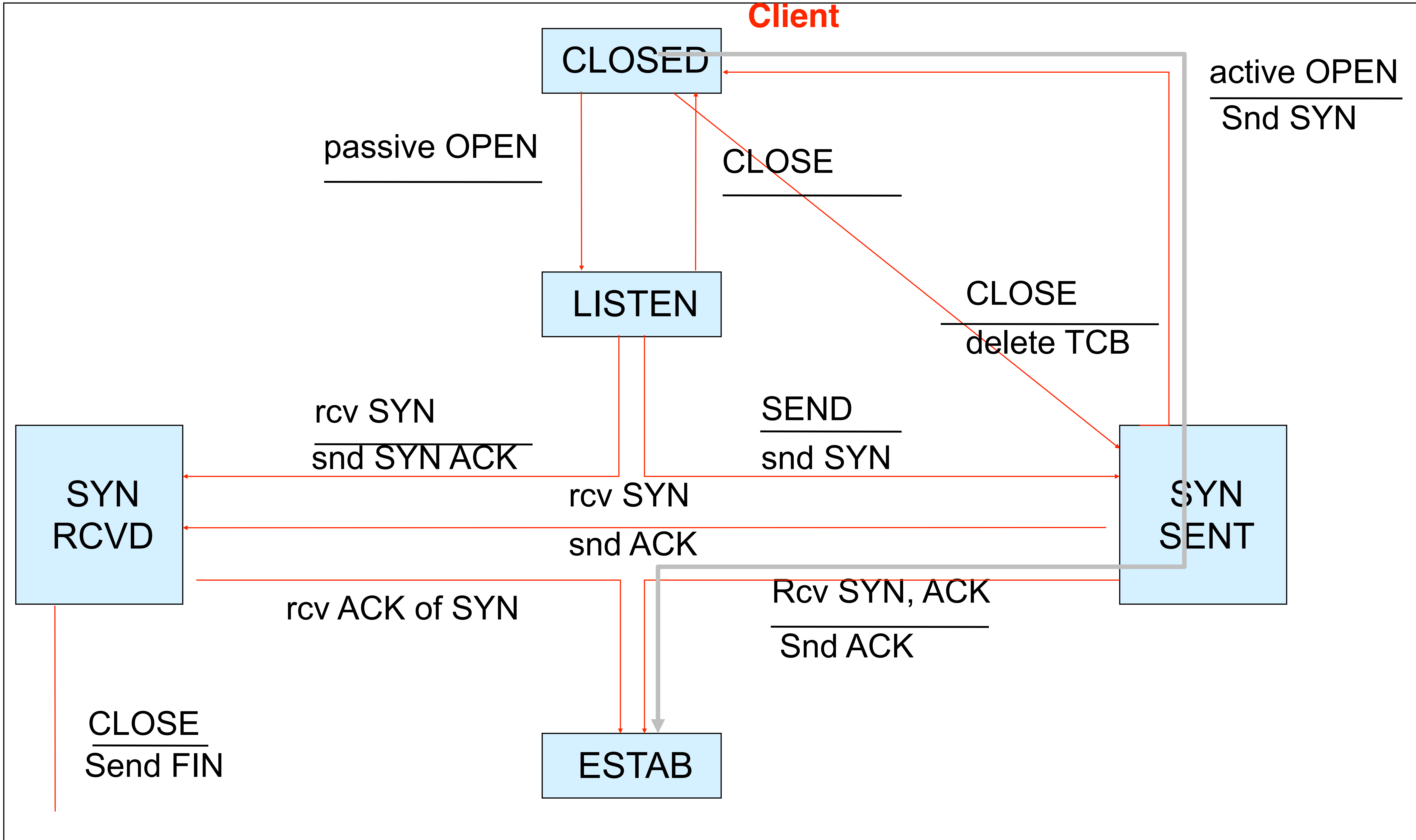

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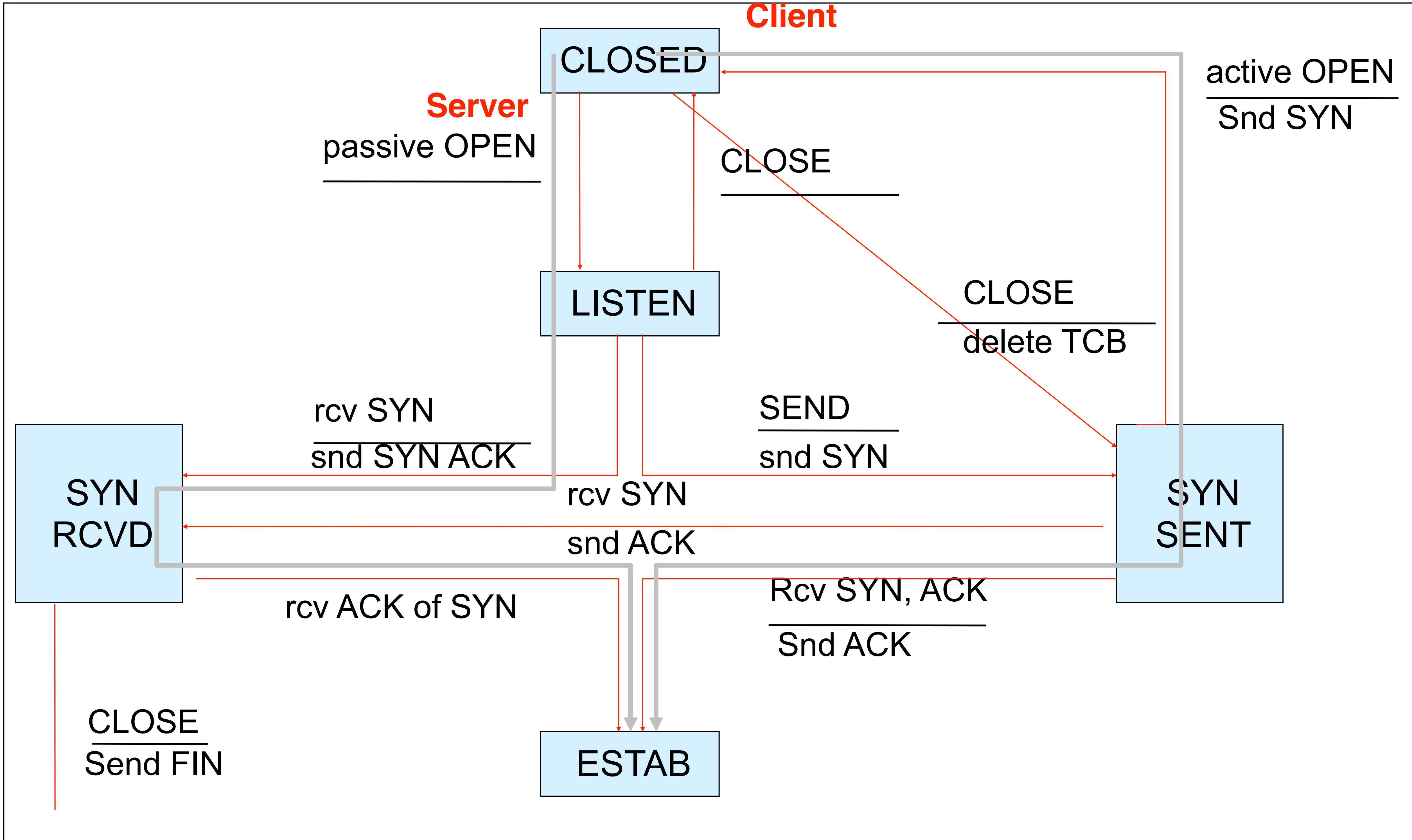
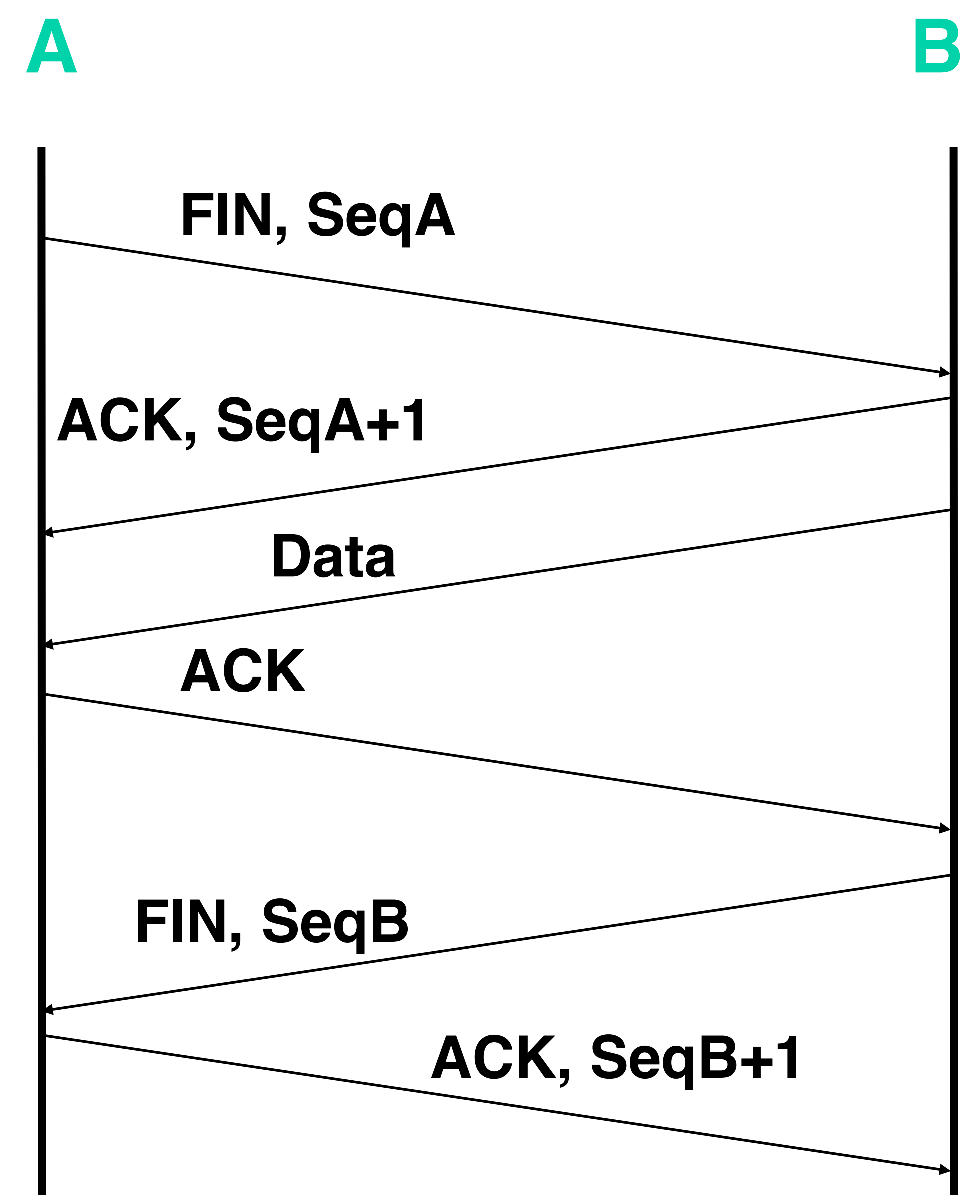


Tearing Down Connection

- 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



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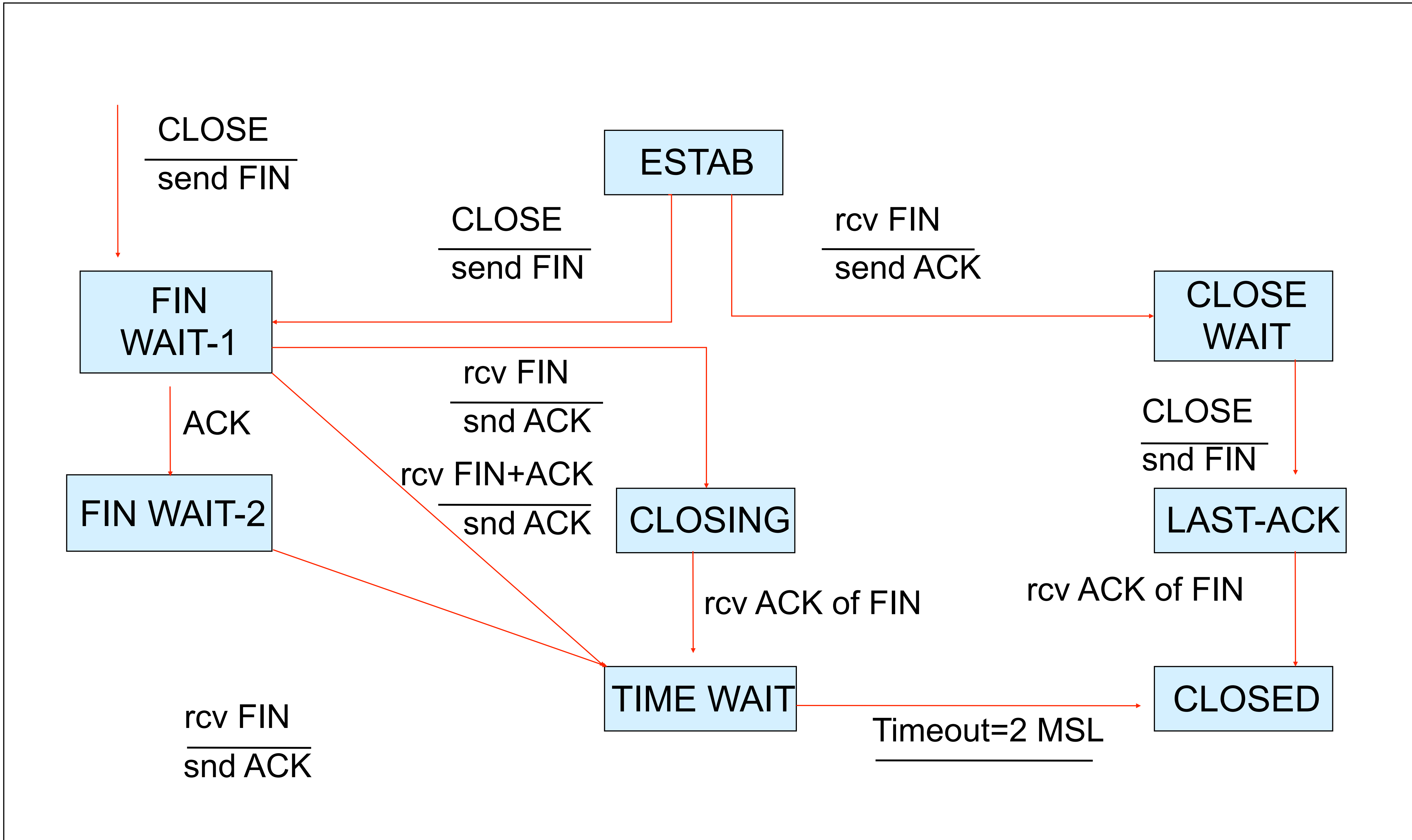

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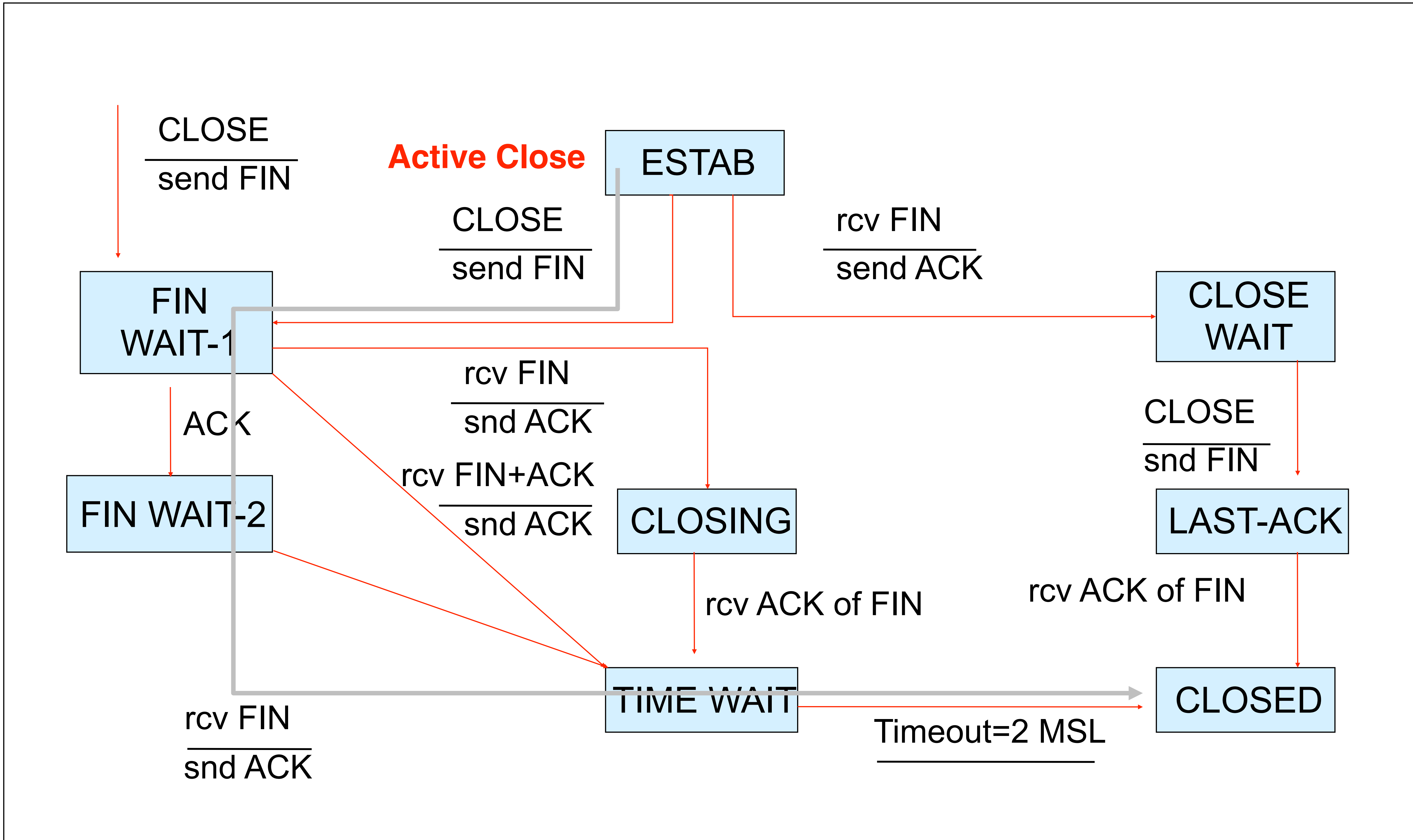

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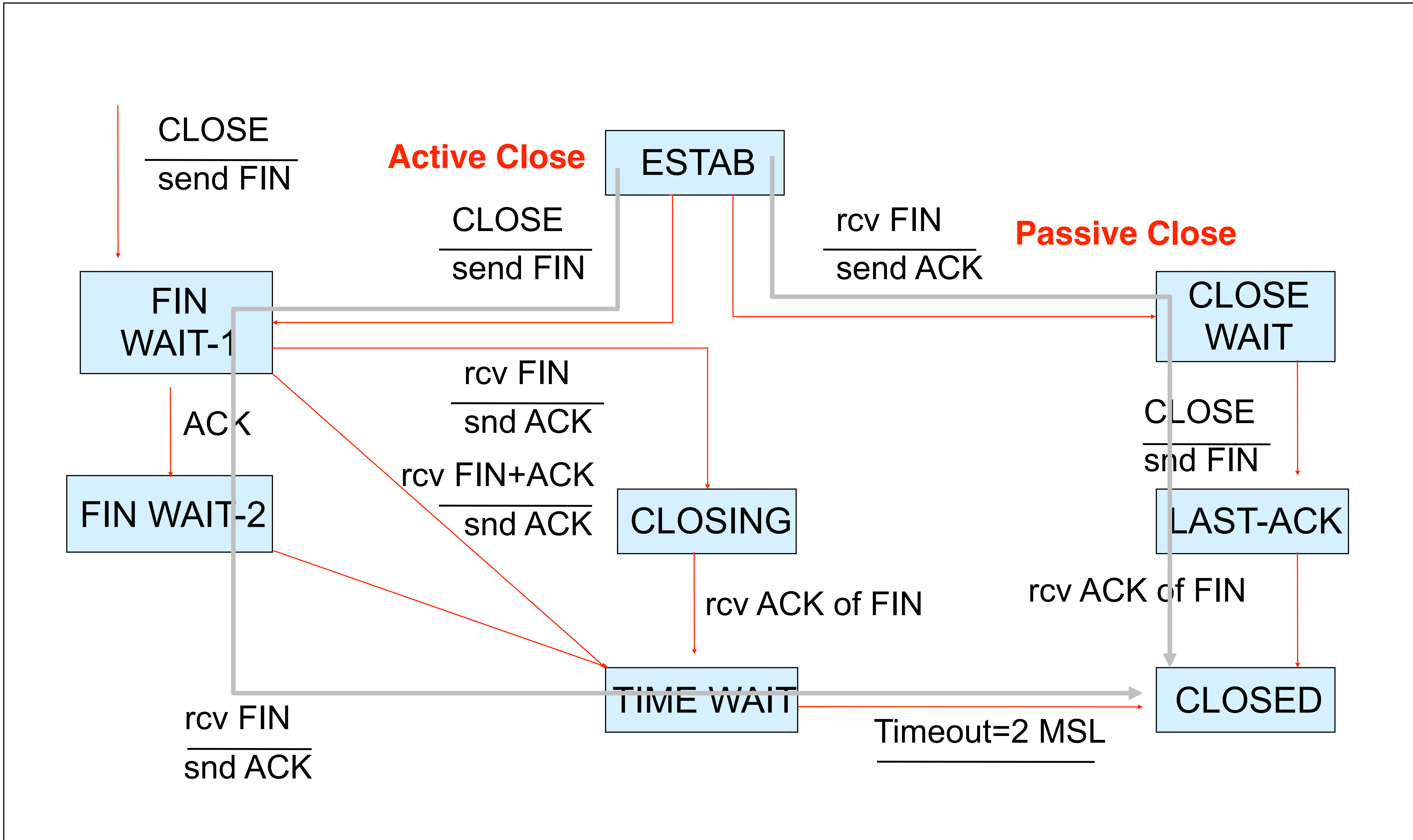
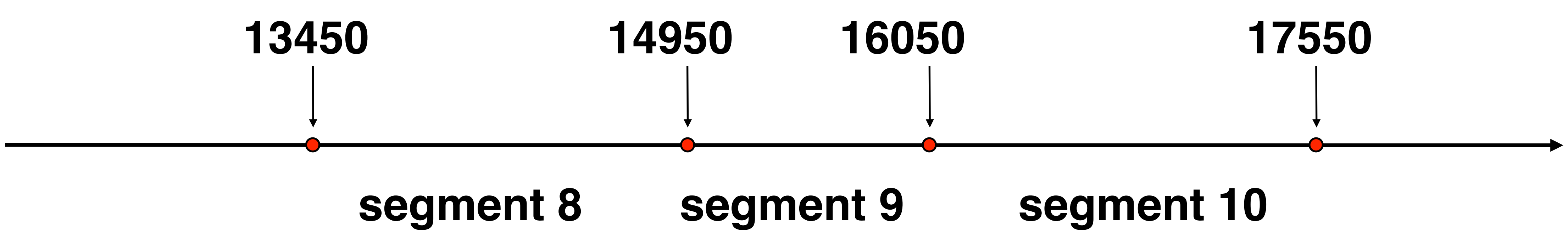


Sequence Number Space

- 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

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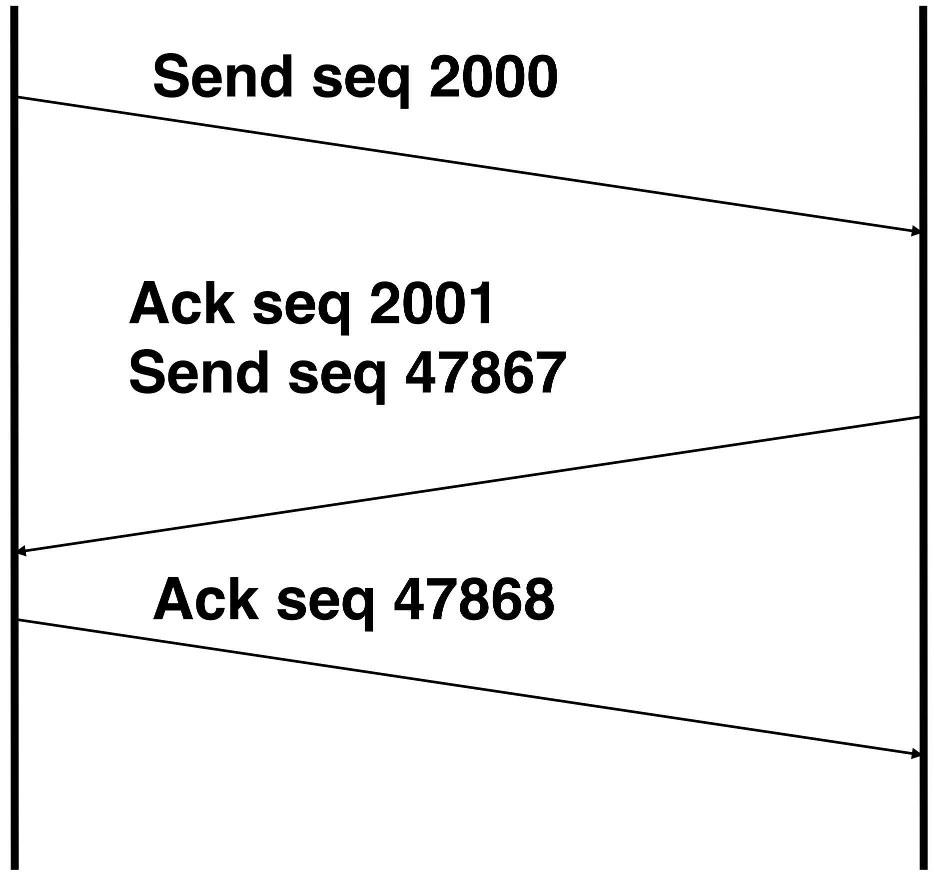
- Retransmission requires that sender keep copy of the data.
  - Copy is discarded when ack is received

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

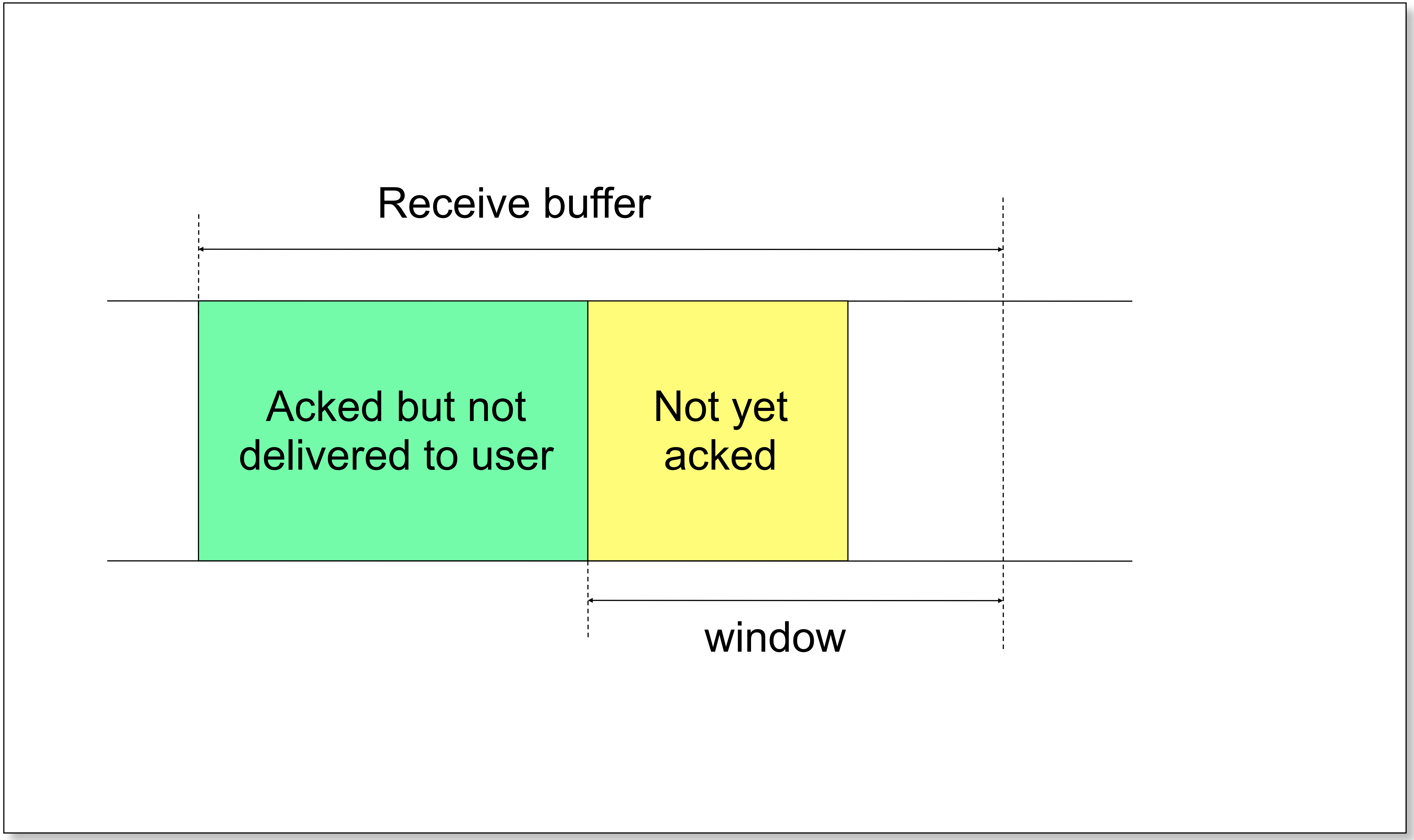
- So, depending on how fast the application is reading the data, the receiver’s window size may change!!!

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



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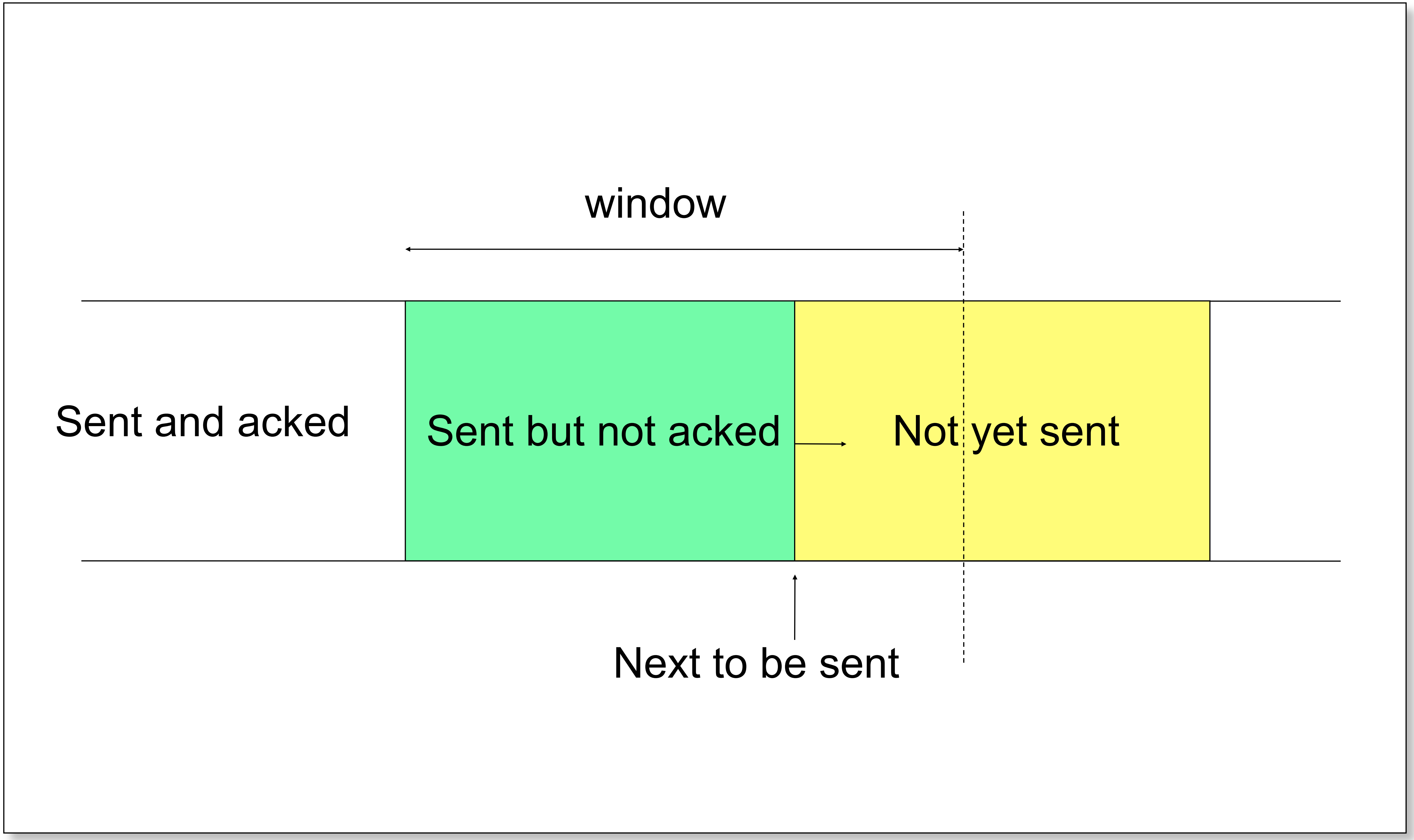

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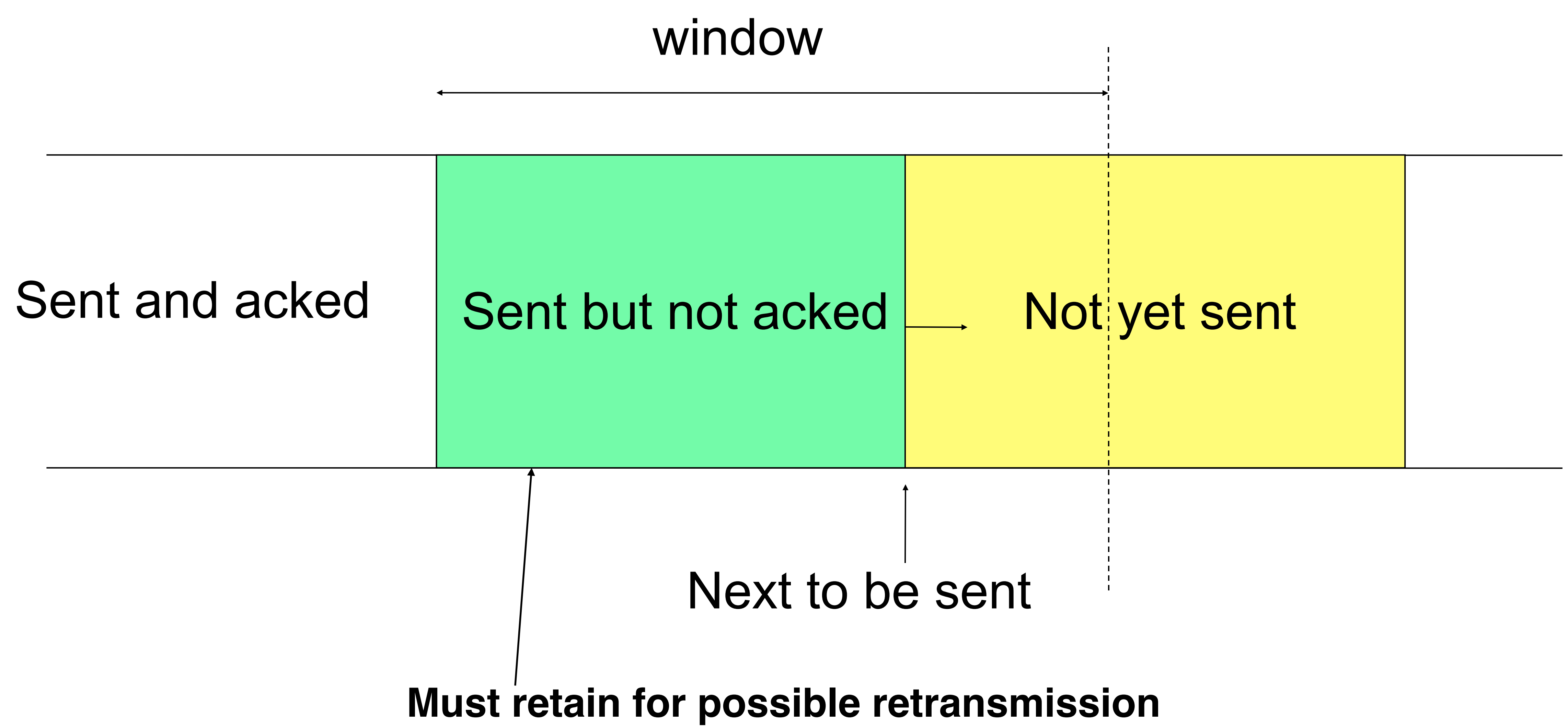

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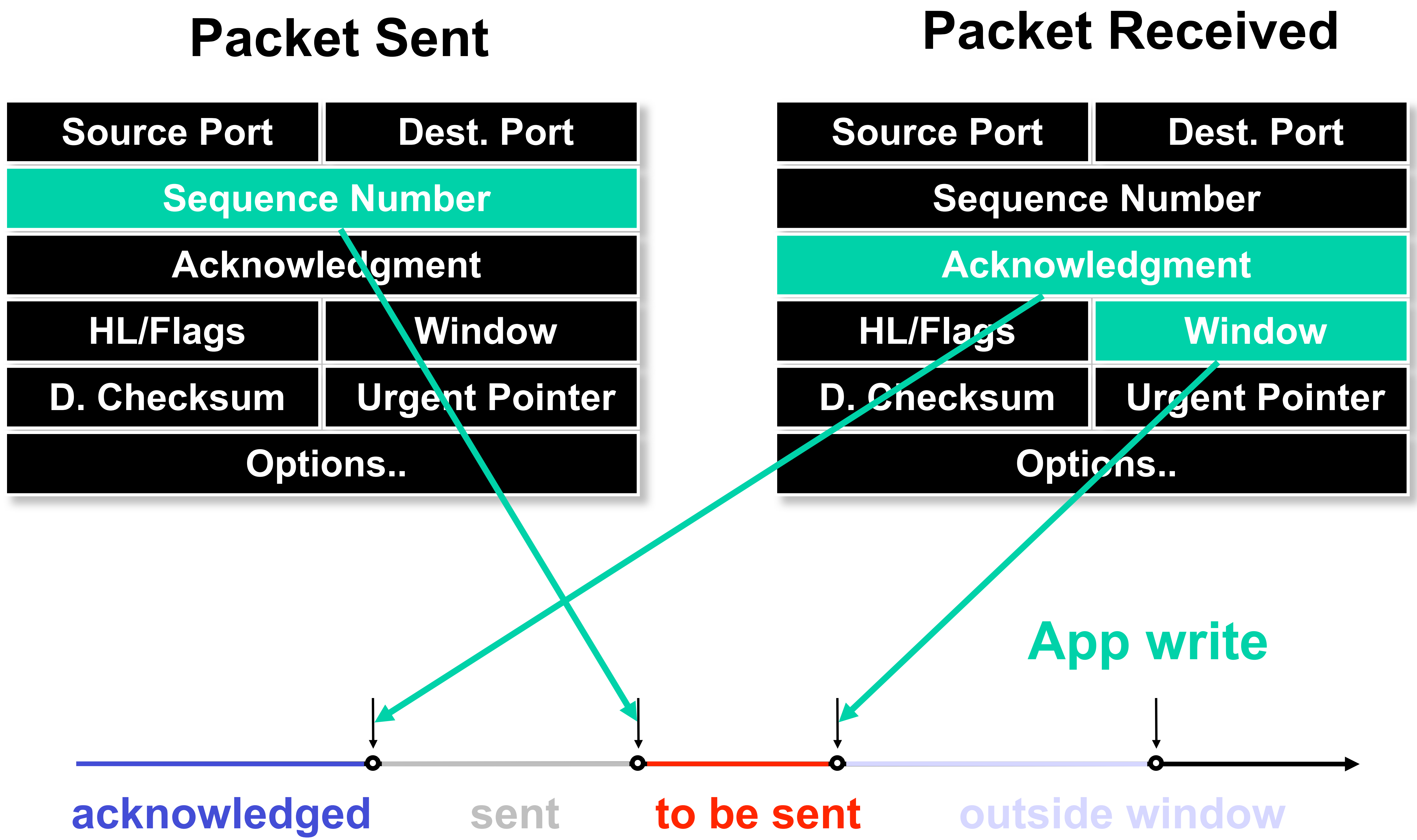
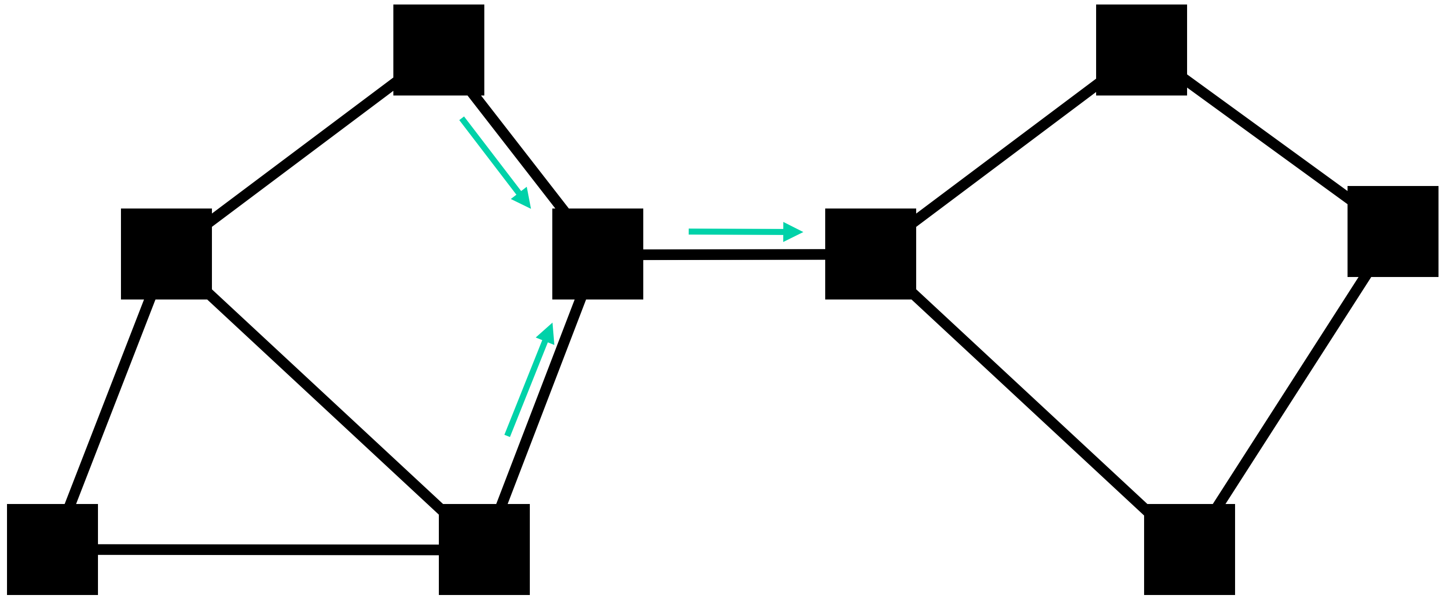


Ongoing Communication

- 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

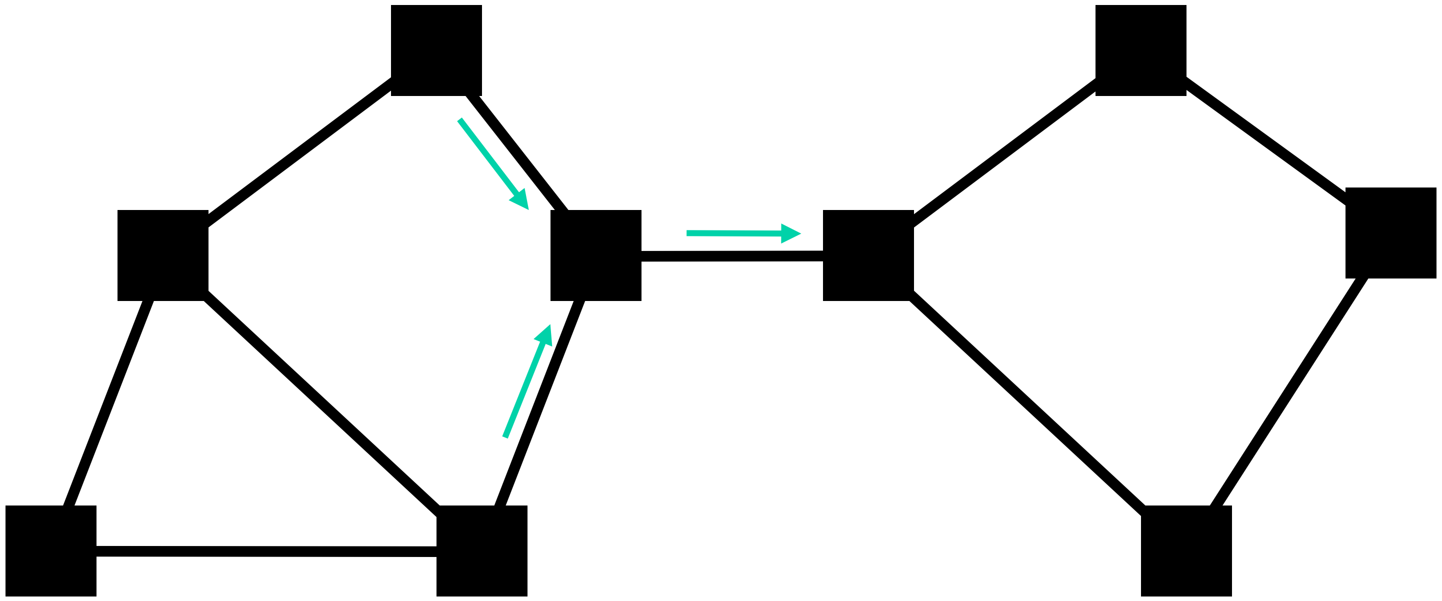
What is Congestion?

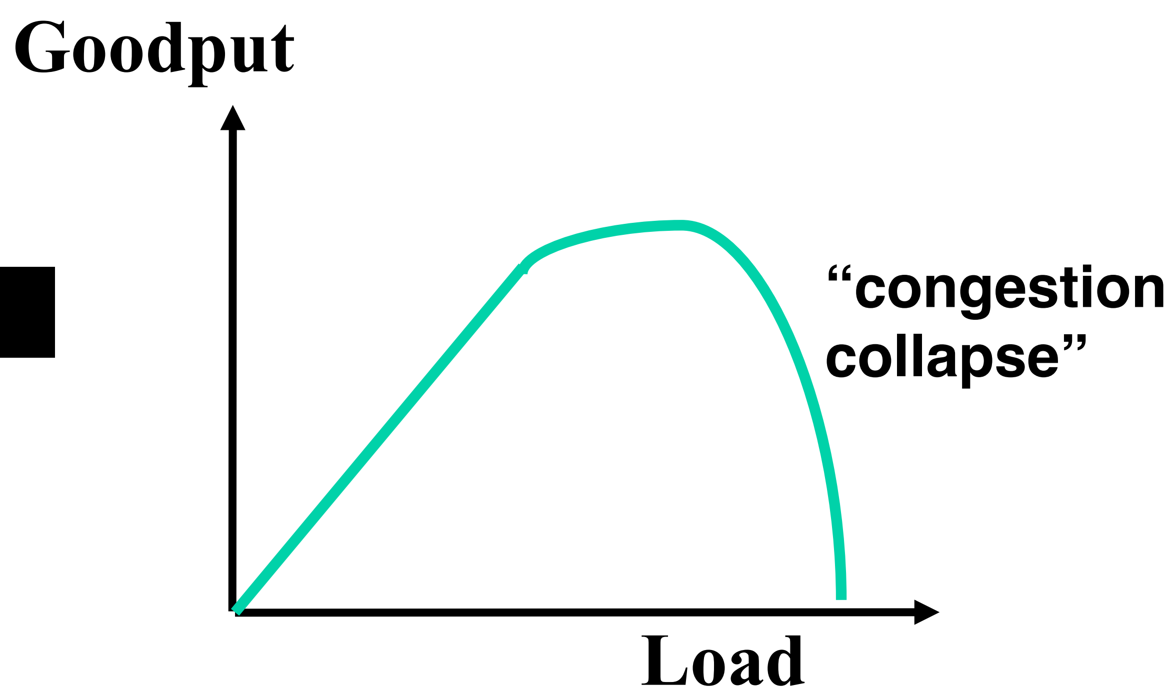
- The load placed on the network is higher than the capacity of the network
  - Not surprising: independent senders place load on network
- Results in packet loss: routers have no choice
  - Can only buffer finite amount of data
  - End-to-end protocol will typically react, e.g. TCP



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





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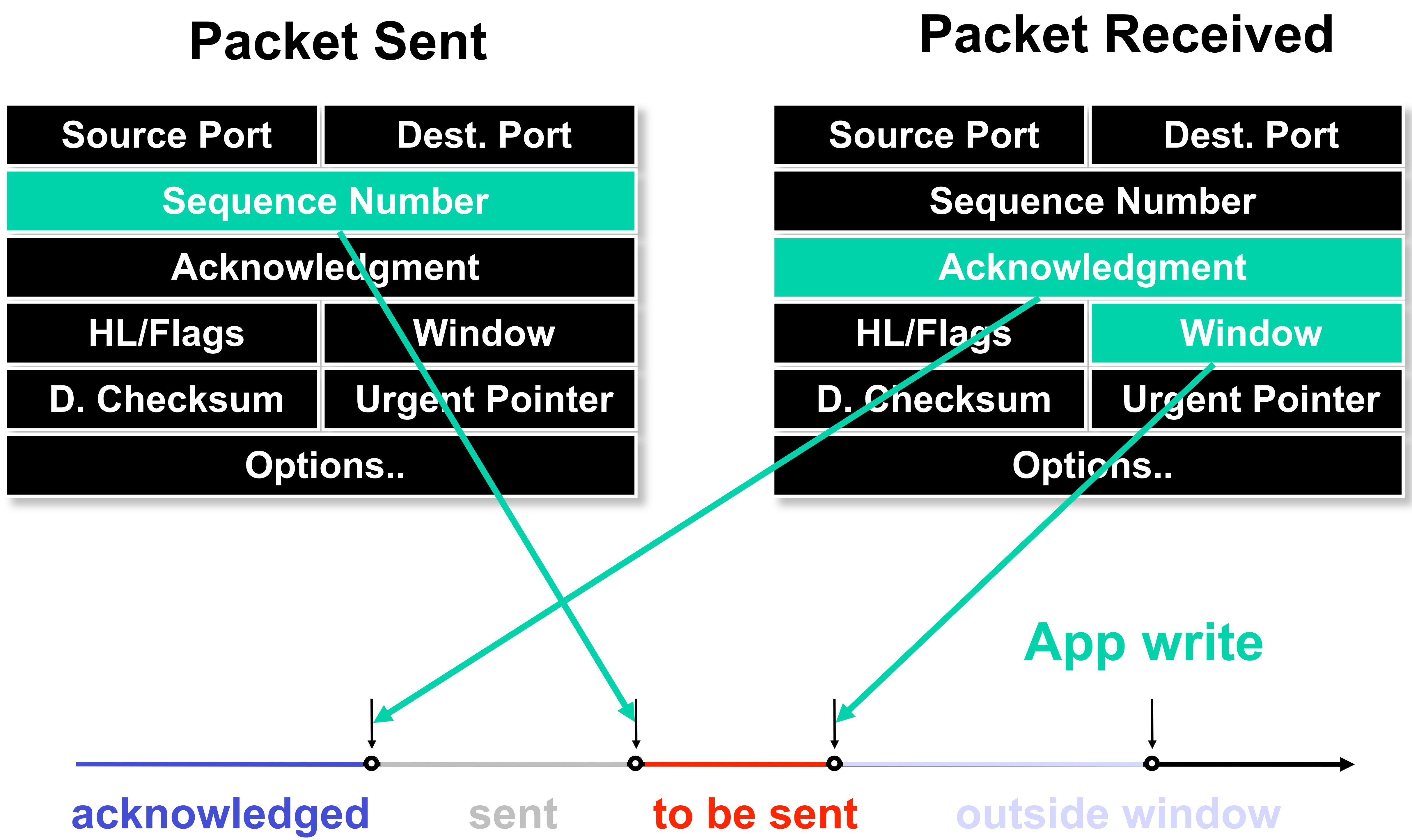
- What if a network link is only 64KB/second but there are 1000 people who are transferring files over that link using the sliding window protocol?
- Packet losses, timeouts, retransmissions, more packet losses… nothing useful gets through, congestion collapse!

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