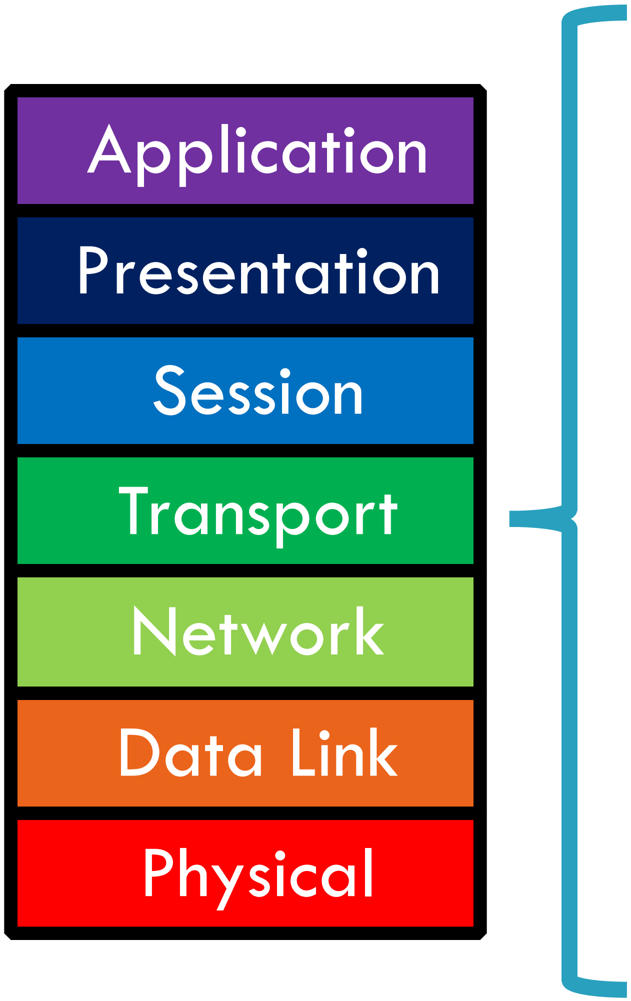
CS 3700
Networks and Distributed Systems

Lecture 9: UDP/TCP

Revised 2/9/2014

Transport Layer



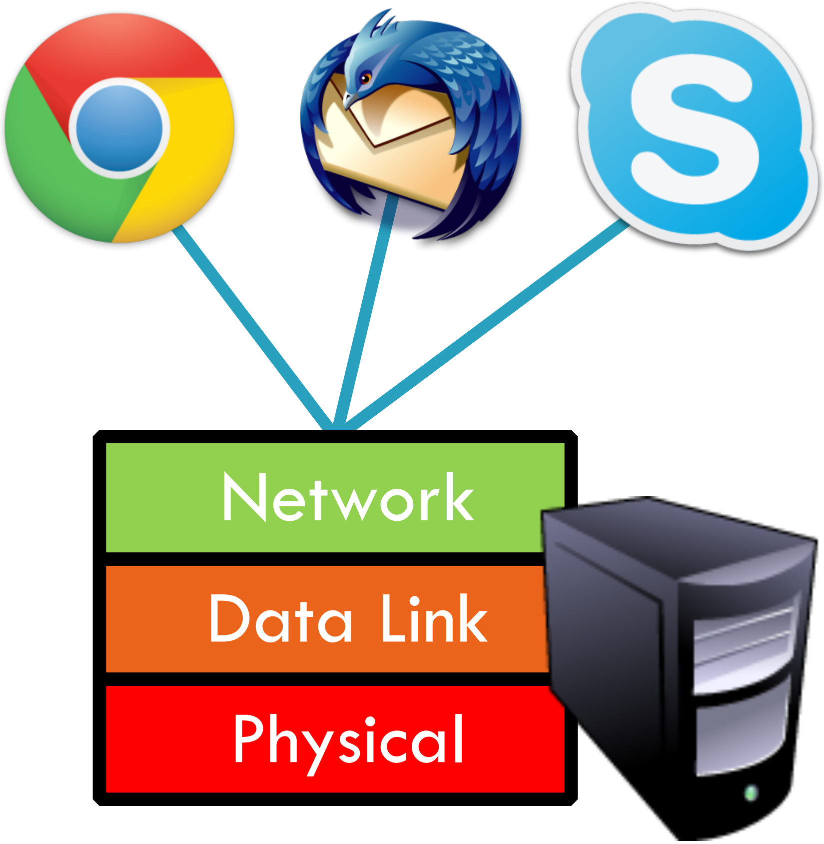
- Function:
  - Demultiplexing of data streams
- Optional functions:
  - Creating long lived connections
  - Reliable, in-order packet delivery
  - Error detection
  - Flow and congestion control
- Key challenges:
  - Detecting and responding to congestion
  - Balancing fairness against high utilization

Outline

- UDP
- TCP

The Case for Multiplexing

- Datagram network
  - No circuits
  - No connections



Network

Data Link

Physical

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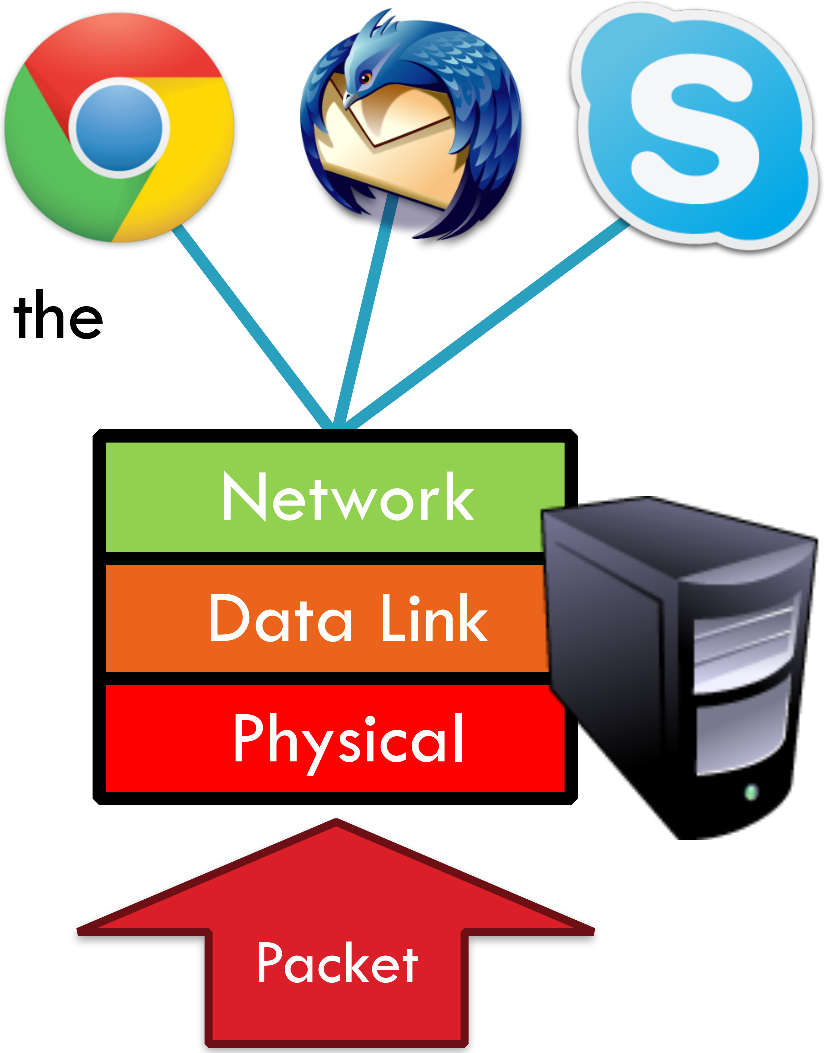
  - Who to deliver packets to?



Network

Data Link

Physical

Packet

The Case for Multiplexing

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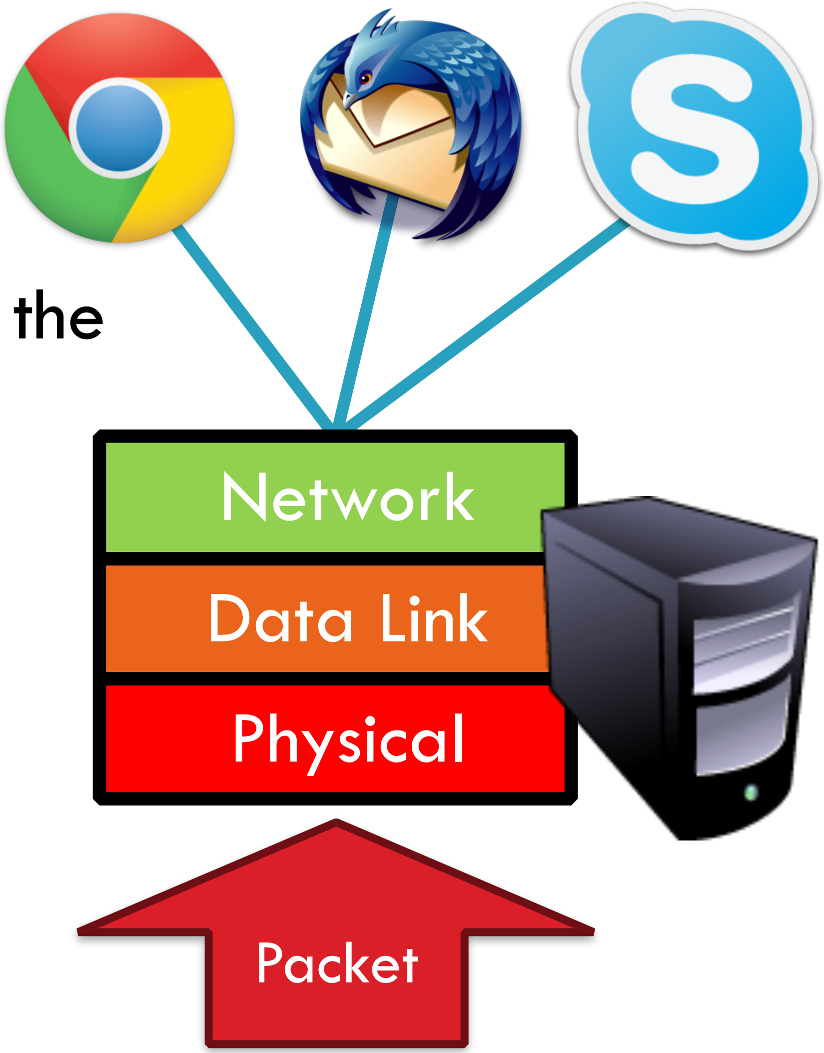
- IP header “protocol” field
  - 8 bits = 256 concurrent streams



The Case for Multiplexing

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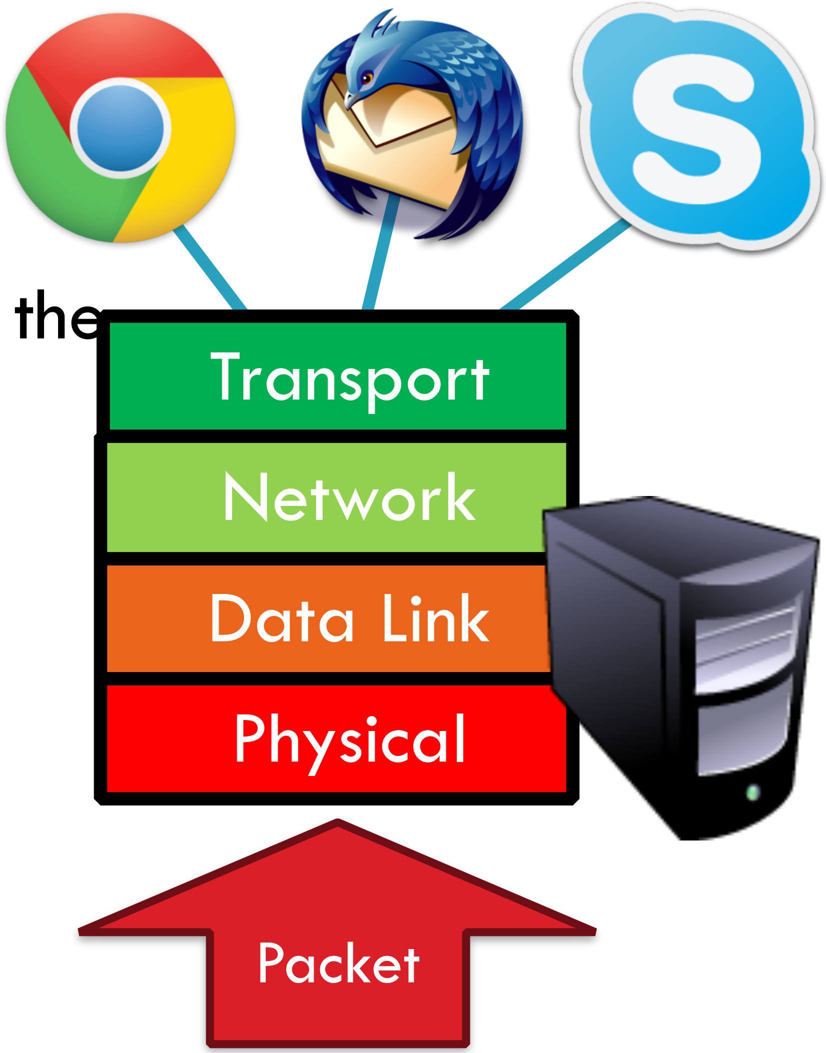
  - Who to deliver packets to?
- IP header "protocol" field
  - 8 bits = 256 concurrent streams
- Insert Transport Layer to handle demultiplexing



Transport

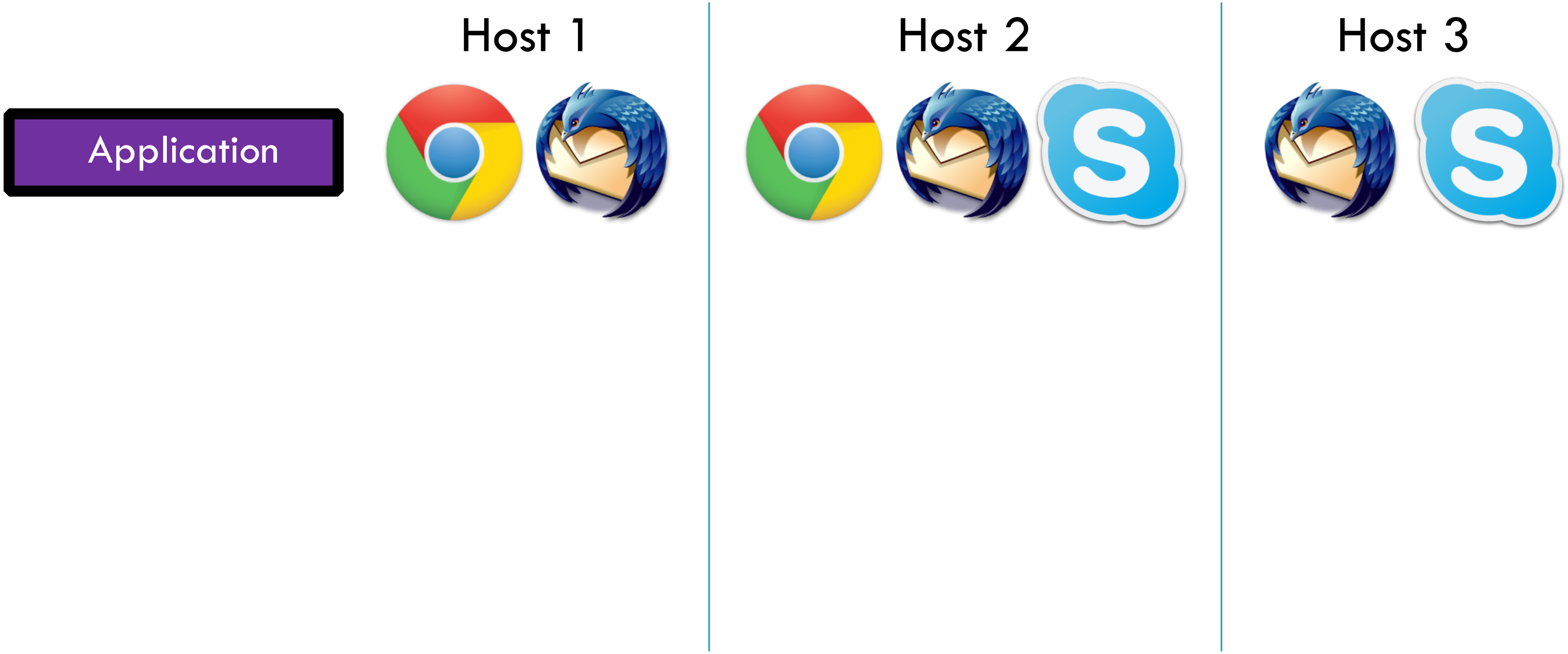
Network

Data Link

Physical

Packet

Demultiplexing Traffic



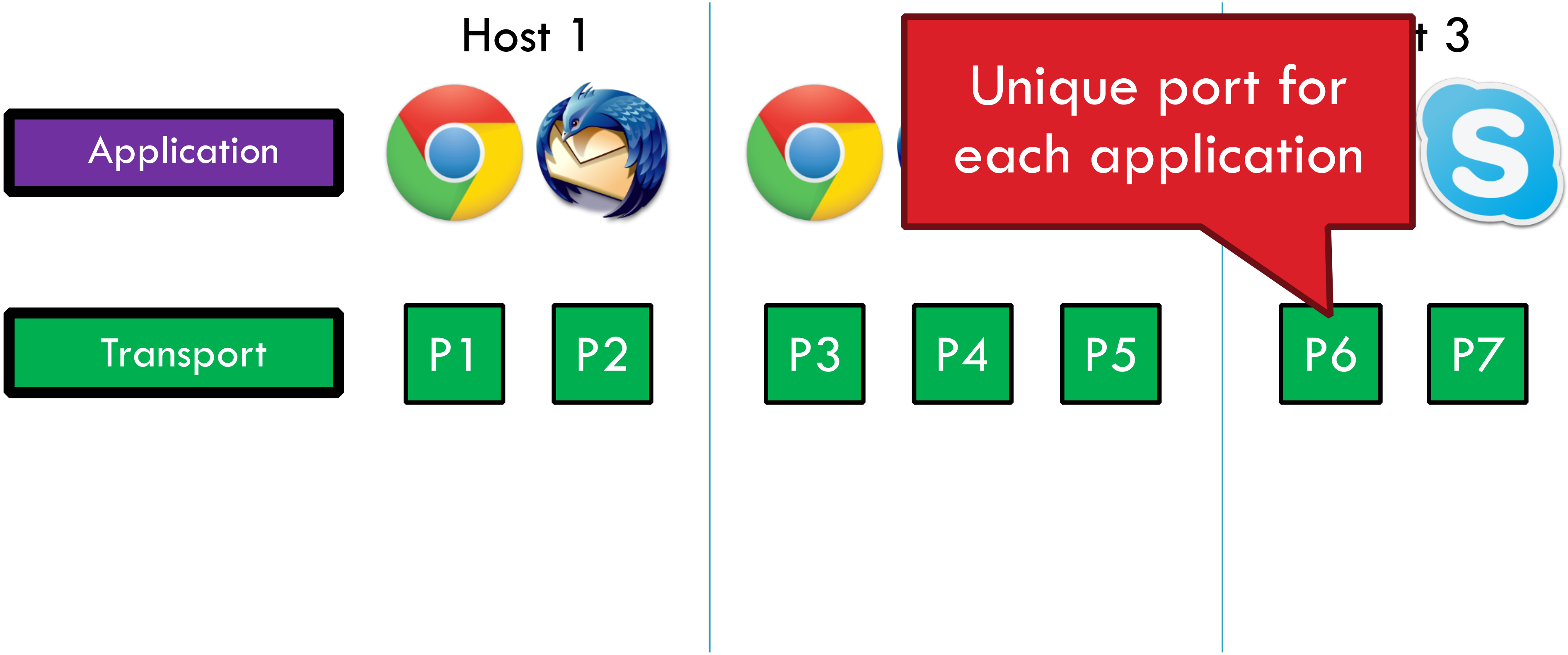
Application

Host 1

Host 2

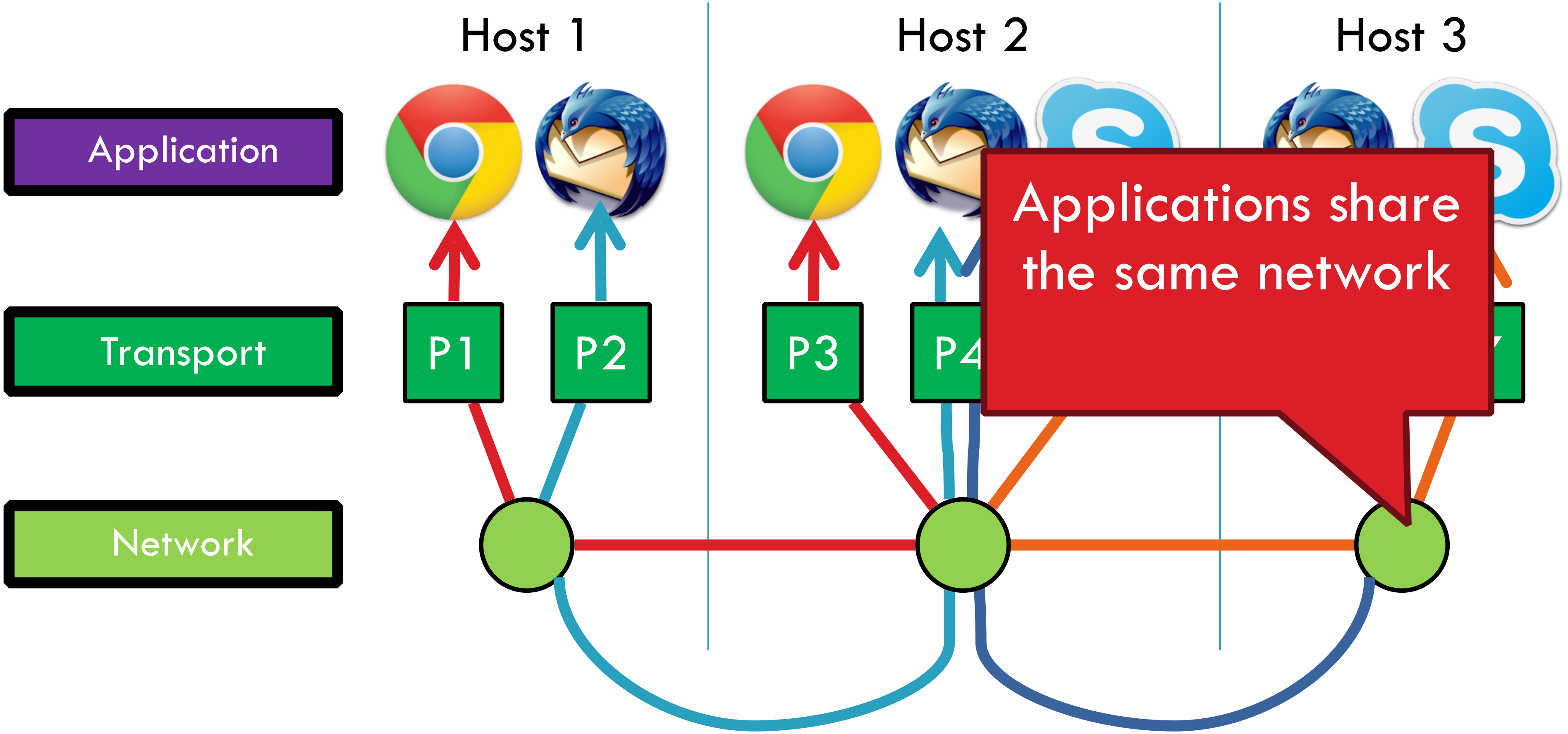
Host 3

Demultiplexing Traffic



Unique port for each application

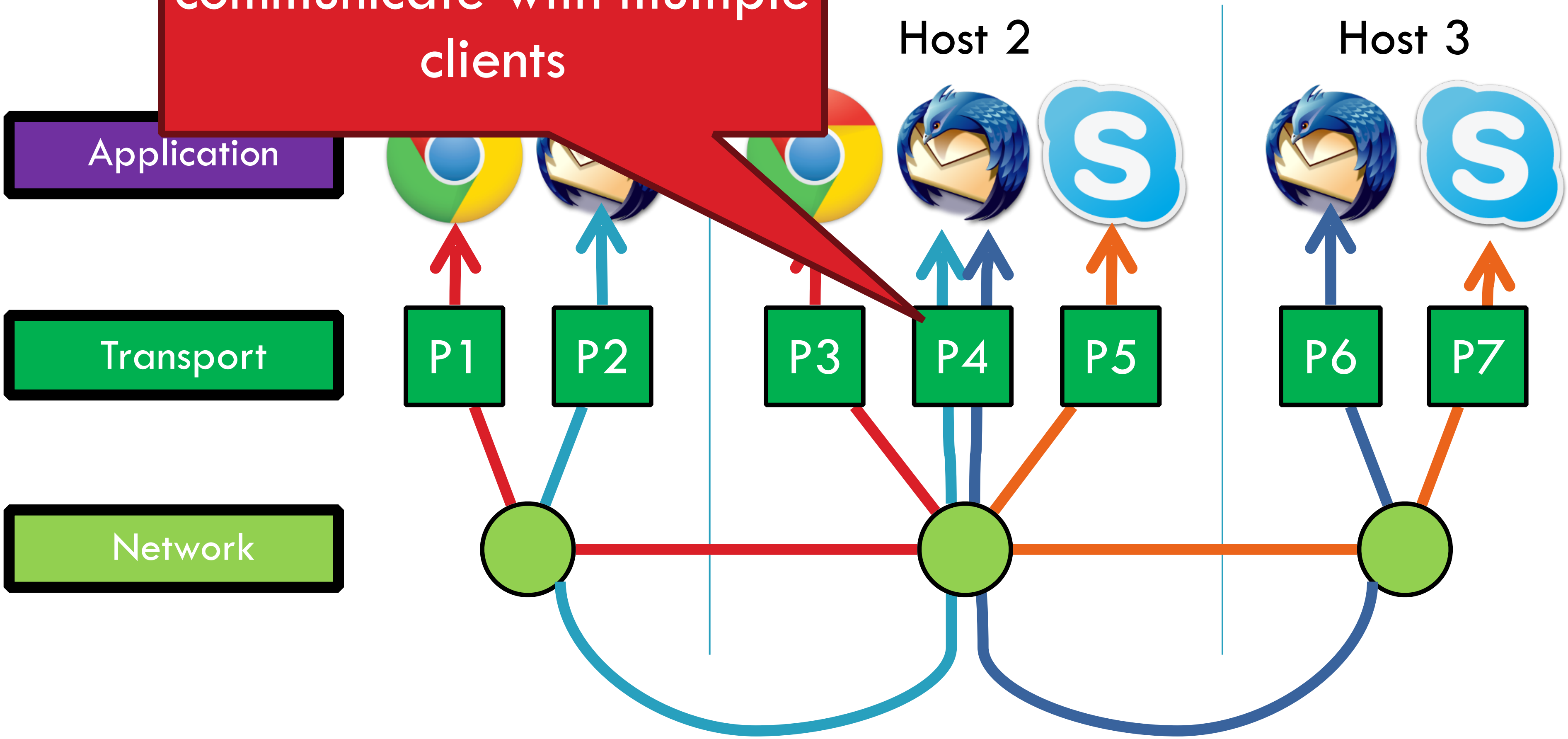
Demultiplexing Traffic



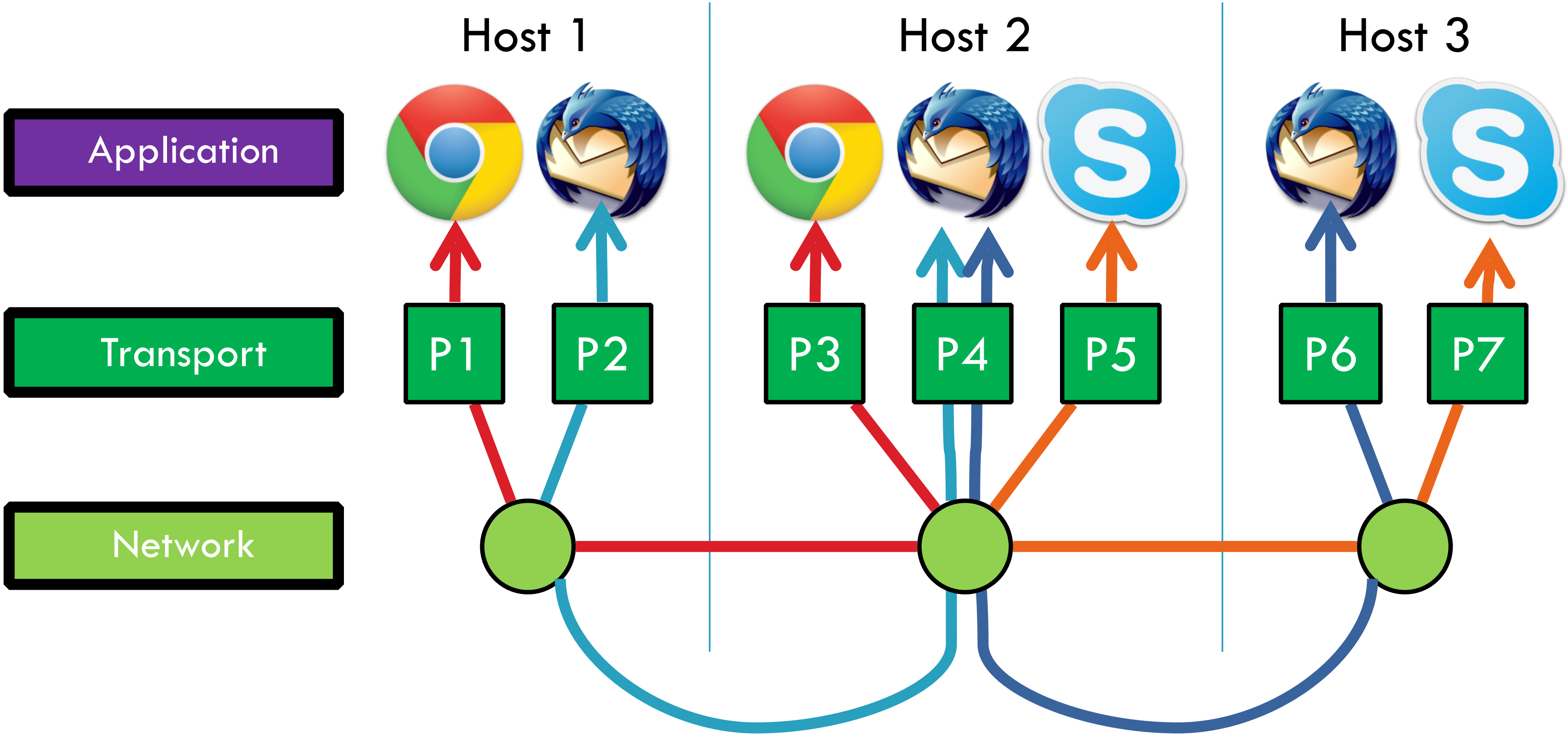
Applications share the same network

Demultiplexing Traffic

Server applications communicate with multiple clients

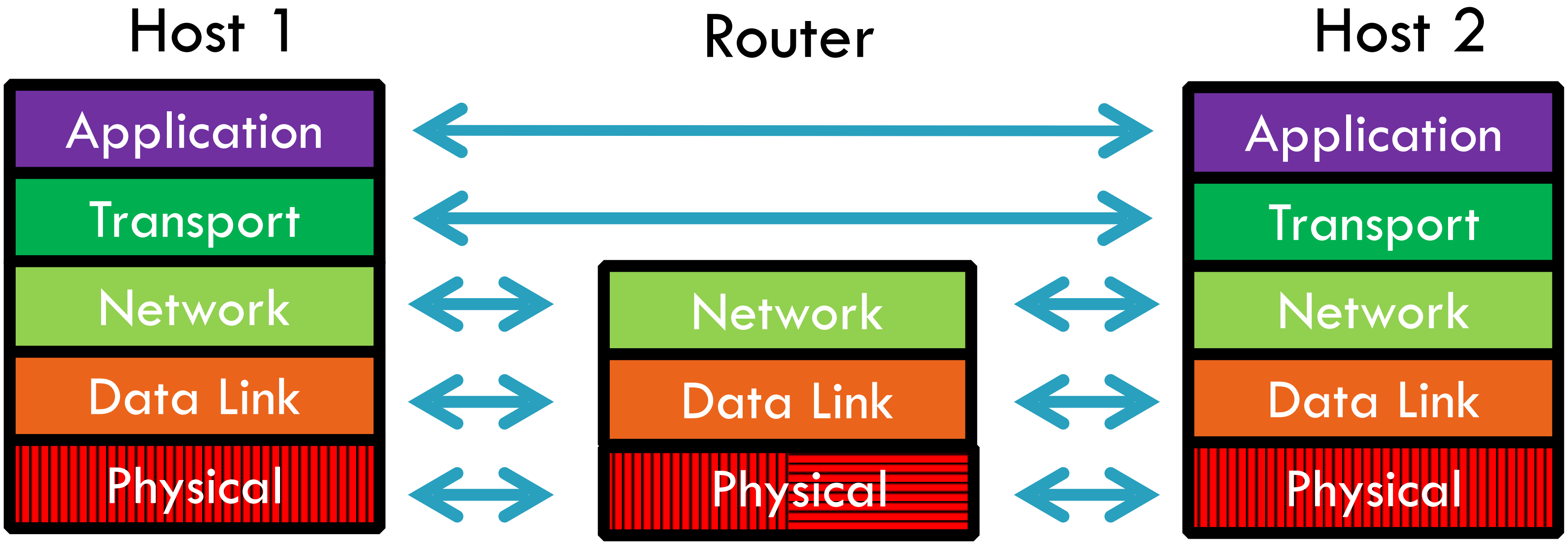


Demultiplexing Traffic



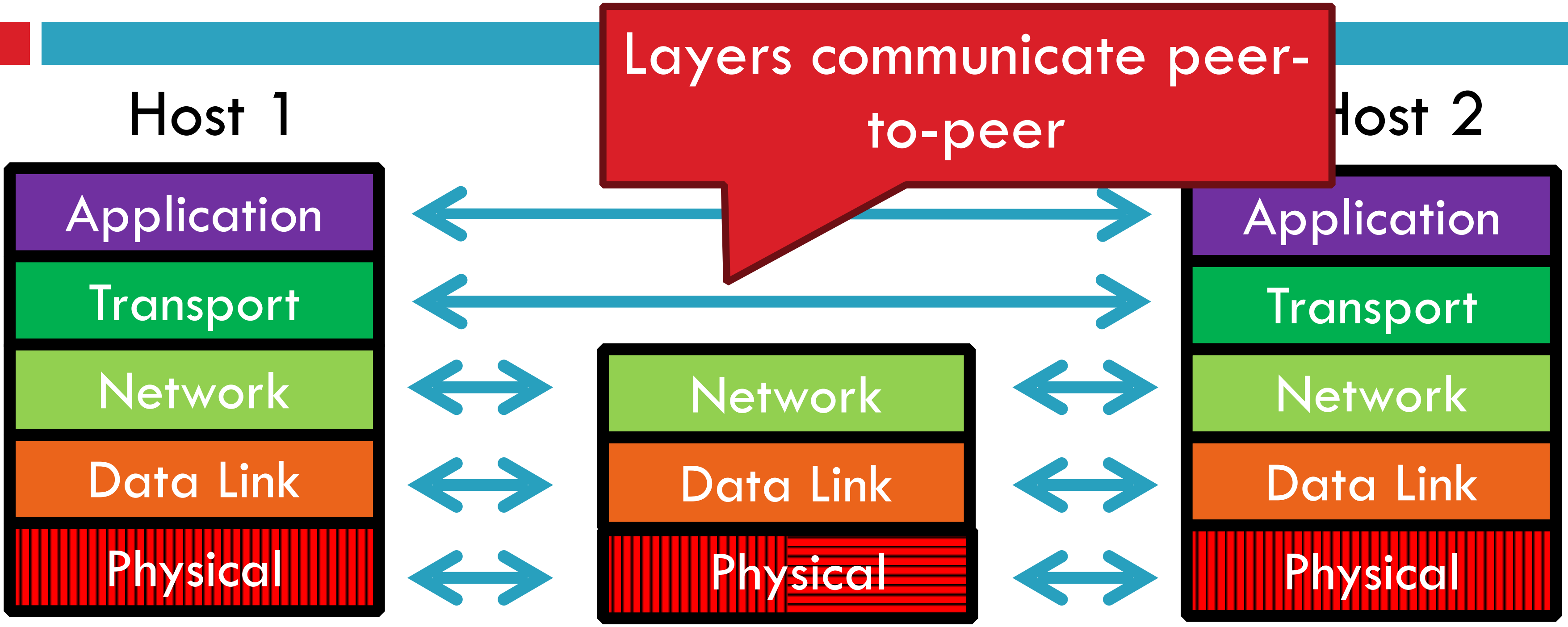
Endpoints identified by <*src_ip, src_port, dest_ip, dest_port*>

Layering, Revisited



- Lowest level end-to-end protocol (in theory)
  - Transport header only read by source and destination
  - Routers view transport header as payload

Layering, Revisited



Layers communicate peer-to-peer

Host 1

Application

Transport

Network

Data Link

Physical

Network

Data Link

Physical

Host 2

Application

Transport

Network

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Physical

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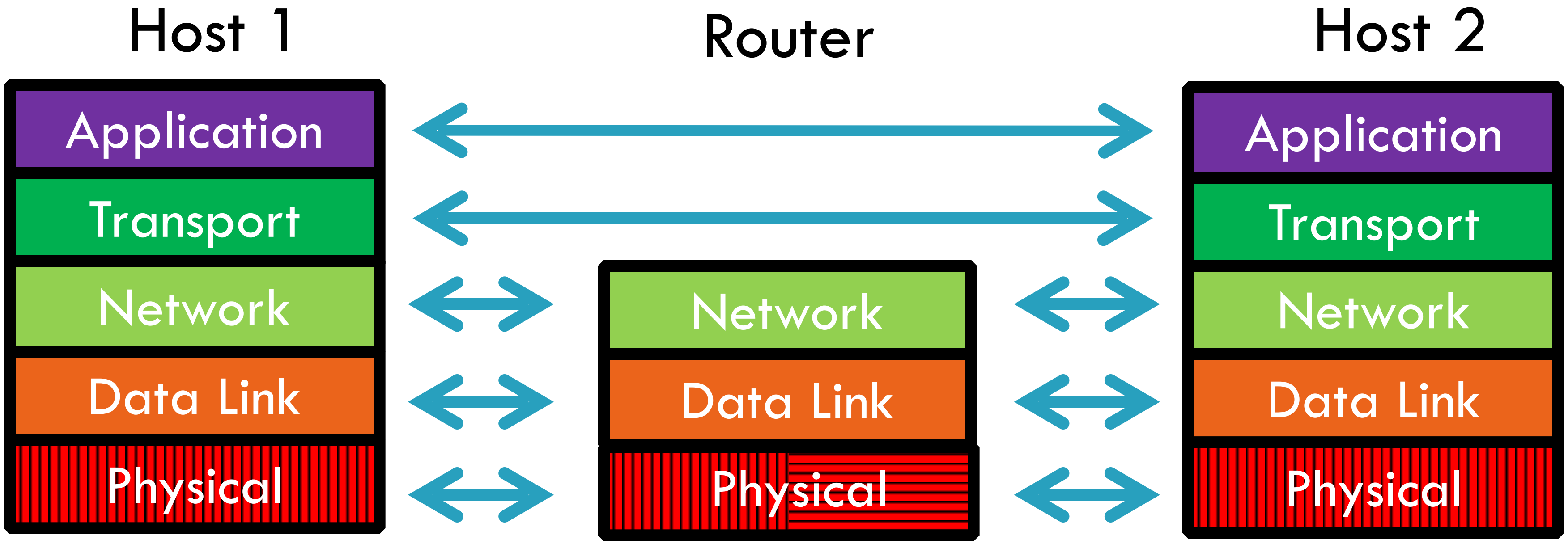

Host 1

Application
Transport
Network
Data Link
Physical

Router

Network
Data Link
Physical

Host 2

Application
Transport
Network
Data Link
Physical

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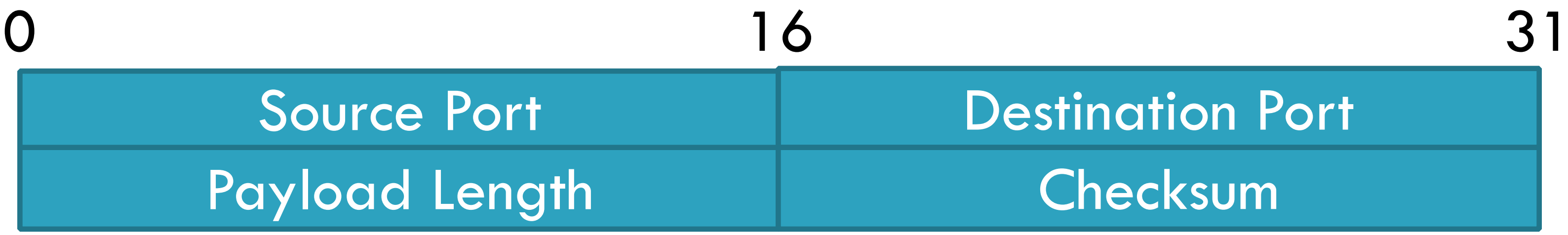
User Datagram Protocol (UDP)



0 16 31

Source Port	Destination Port
Payload Length	Checksum

- Simple, connectionless datagram
  - C sockets: SOCK_DGRAM
- Port numbers enable demultiplexing
  - 16 bits = 65535 possible ports
  - Port 0 is invalid
- Checksum for error detection
  - Detects (some) corrupt packets
  - Does not detect dropped, duplicated, or reordered packets

Uses for UDP

- Invented after TCP
  - Why?

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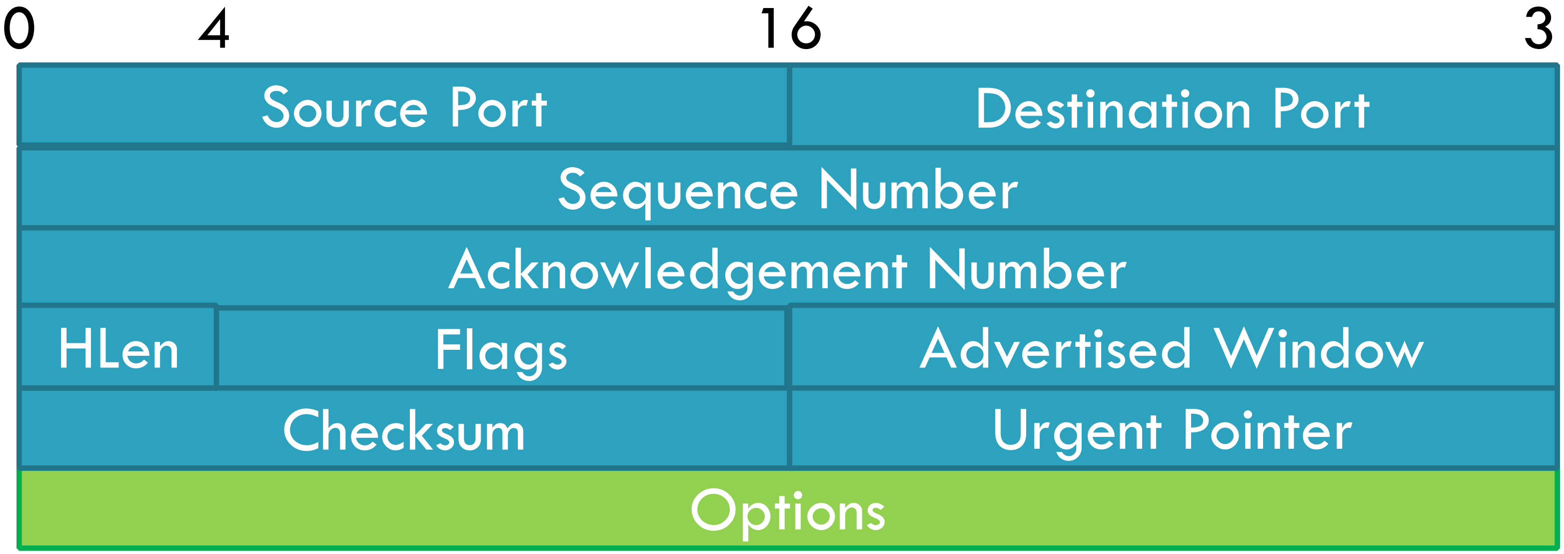
- Invented after TCP
  - Why?
- Not all applications can tolerate TCP
- Custom protocols can be built on top of UDP
  - Reliability? Strict ordering?
  - Flow control? Congestion control?
- Examples
  - RTMP, real-time media streaming (e.g. voice, video)
  - Facebook datacenter protocol

Outline

- UDP
- TCP

Transmission Control Protocol

- Reliable, in-order, bi-directional byte streams
  - Port numbers for demultiplexing
  - Virtual circuits (connections)
  - Flow control
  - Congestion control, approximate fairness



<table>
<tr><th>0</th><th>4</th><th>16</th><th>31</th></tr>
<tr><td colspan="2">Source Port</td><td colspan="2">Destination Port</td></tr>
<tr><td colspan="4">Sequence Number</td></tr>
<tr><td colspan="4">Acknowledgement Number</td></tr>
<tr><td>HLen</td><td>Flags</td><td colspan="2">Advertised Window</td></tr>
<tr><td colspan="2">Checksum</td><td colspan="2">Urgent Pointer</td></tr>
<tr><td colspan="4">Options</td></tr>
</table>

Transmission Control Protocol

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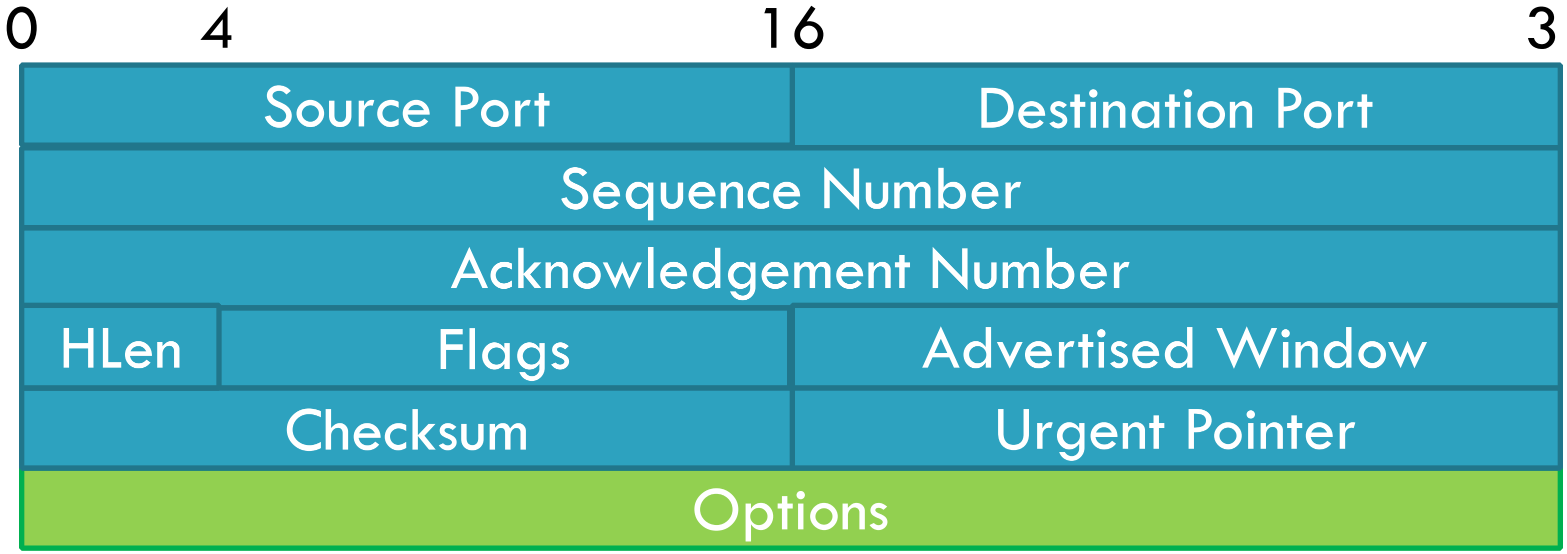
Why these features?



Connection Setup

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    - Why?

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    - Why?
- Important TCP flags (1 bit each)
  - SYN – synchronization, used for connection setup
  - ACK – acknowledge received data
  - FIN – finish, used to tear down connection

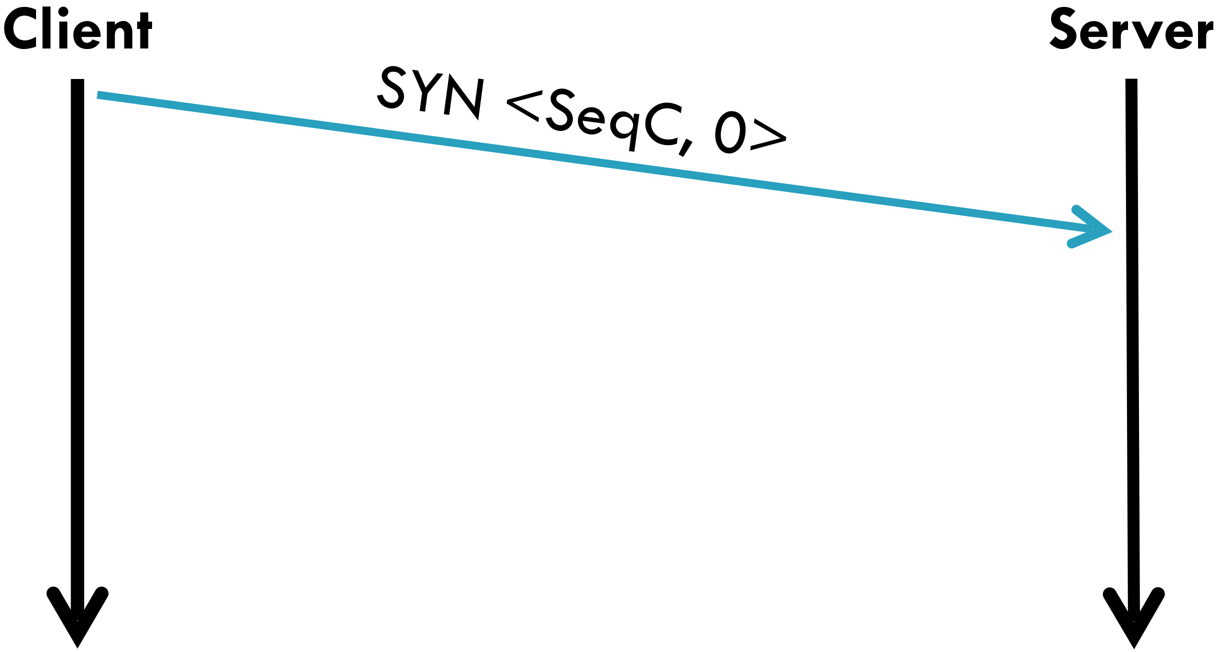
Three Way Handshake

Client

Server

- Each side:
  - Notifies the other of starting sequence number
  - ACKs the other side’s starting sequence number

Three Way Handshake



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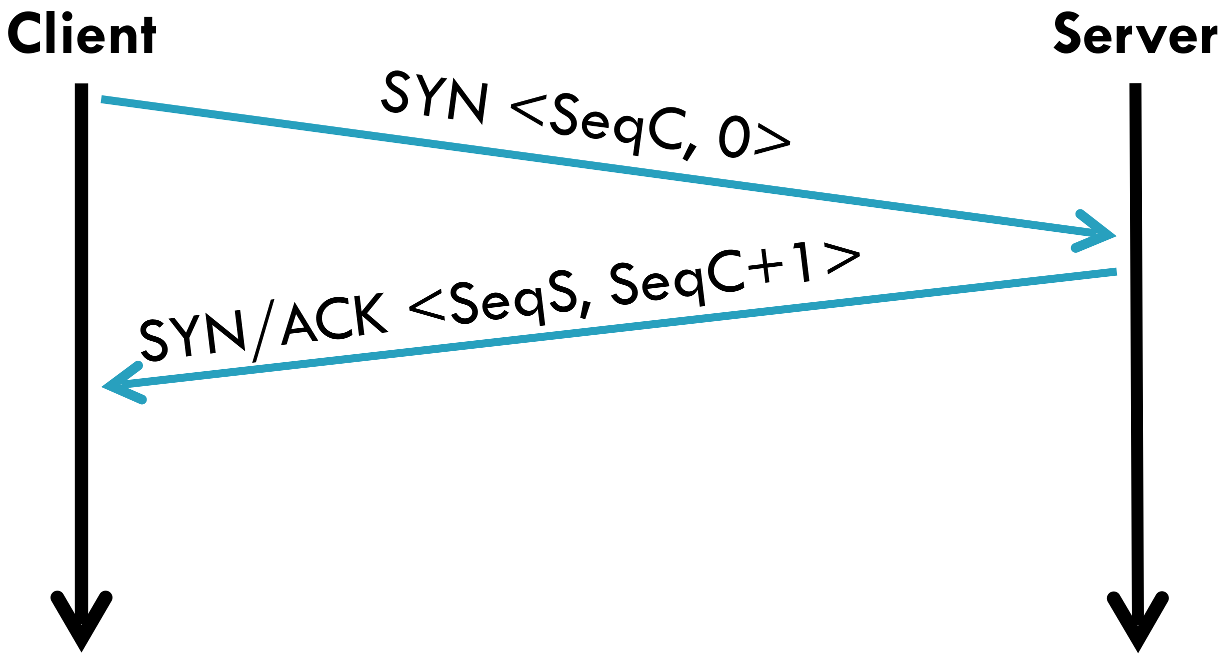

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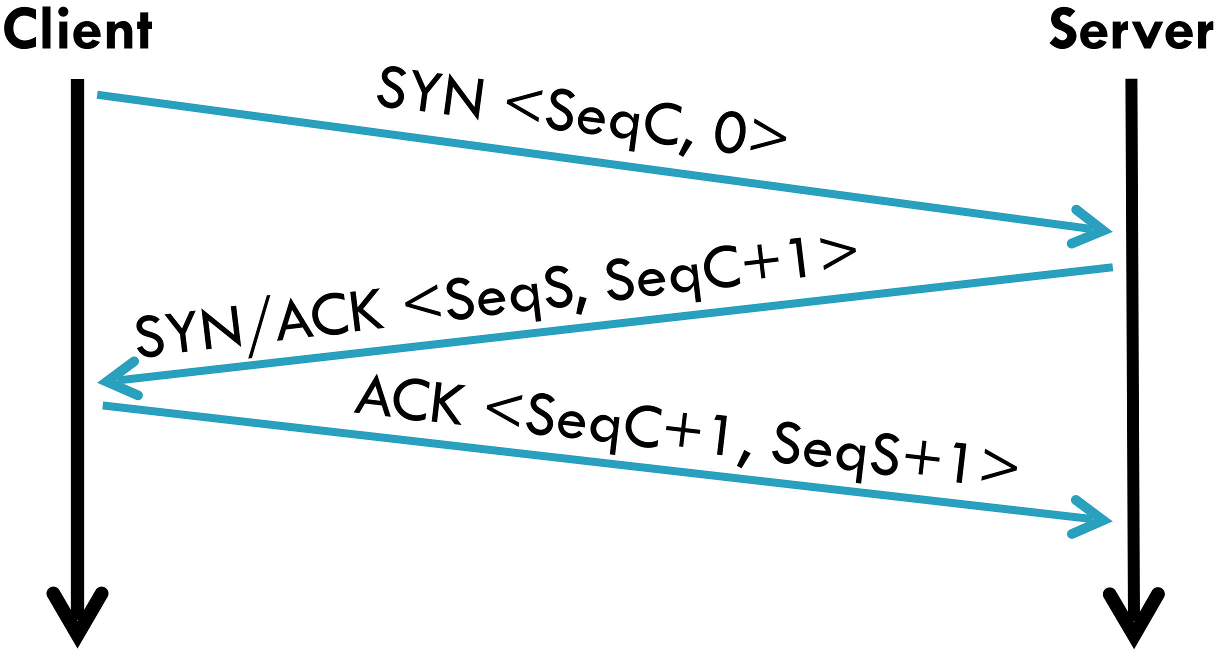

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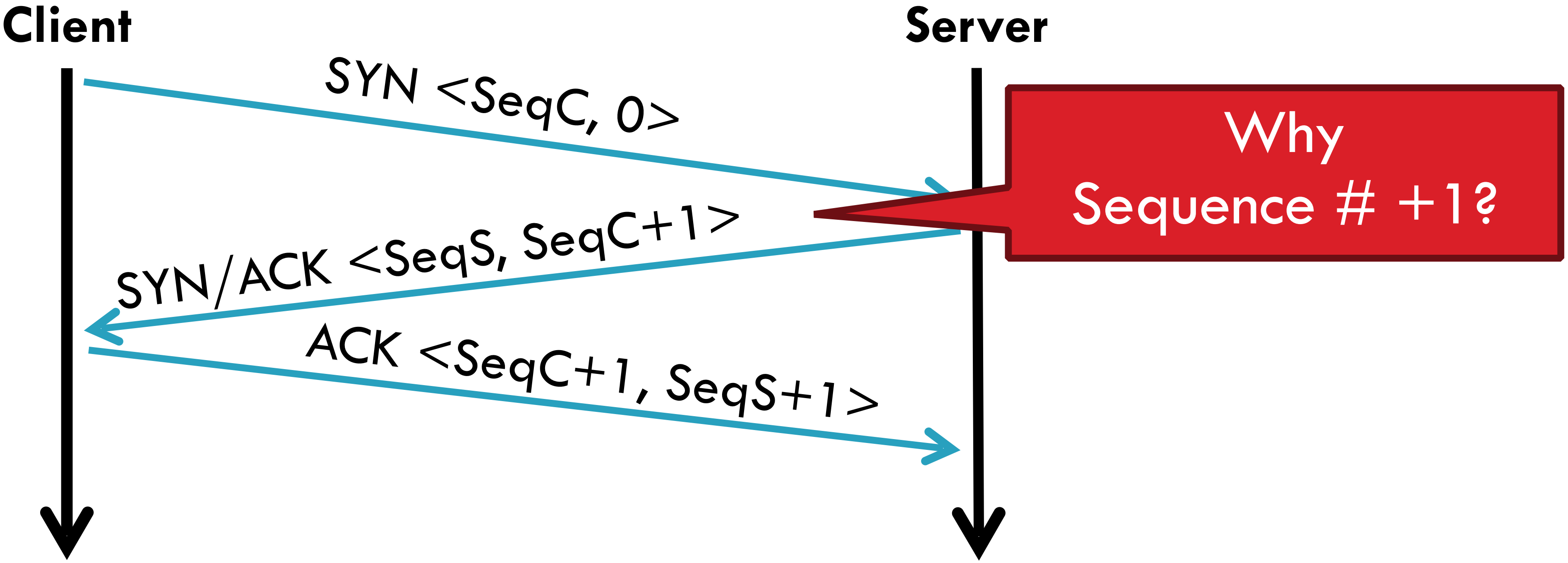

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- Source spoofing
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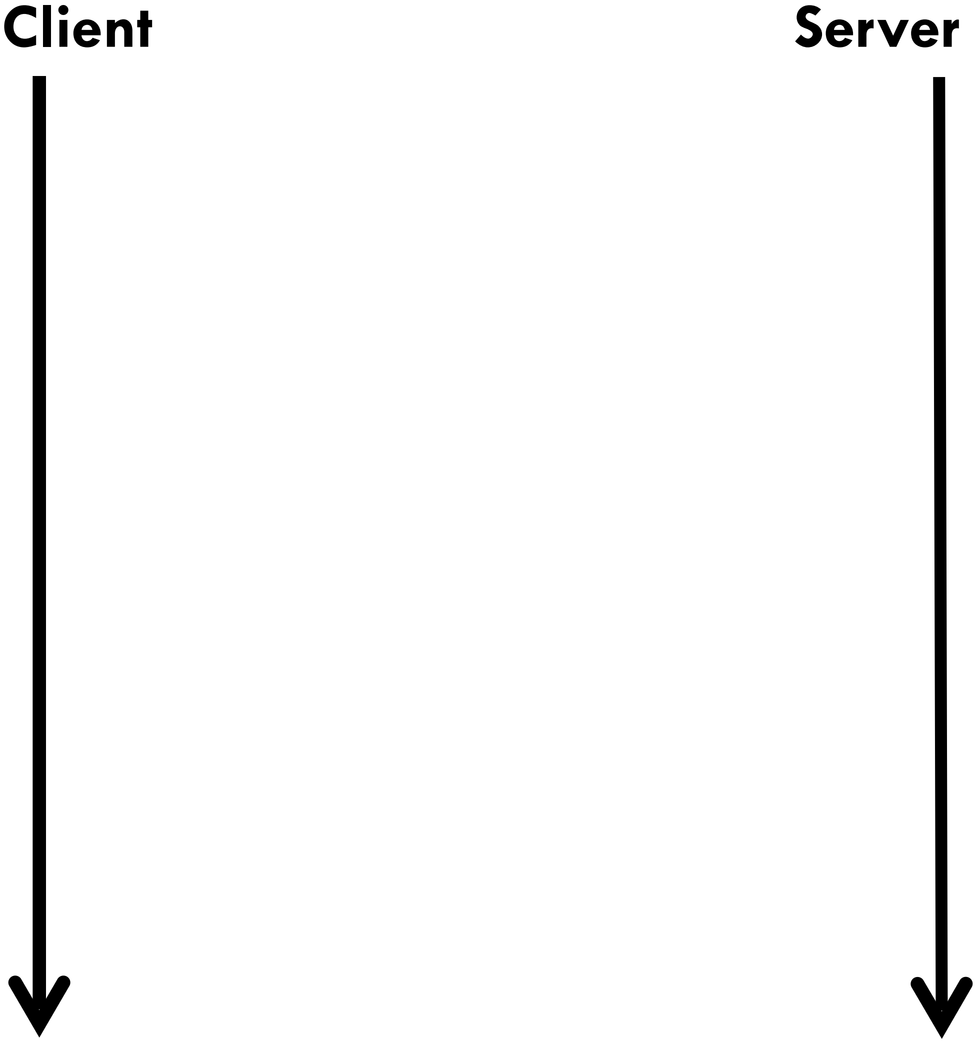
- Connection state management
  - Each SYN allocates state on the server
  - SYN flood = denial of service attack
  - Solution: SYN cookies

Connection Tear Down

- Either side can initiate tear down



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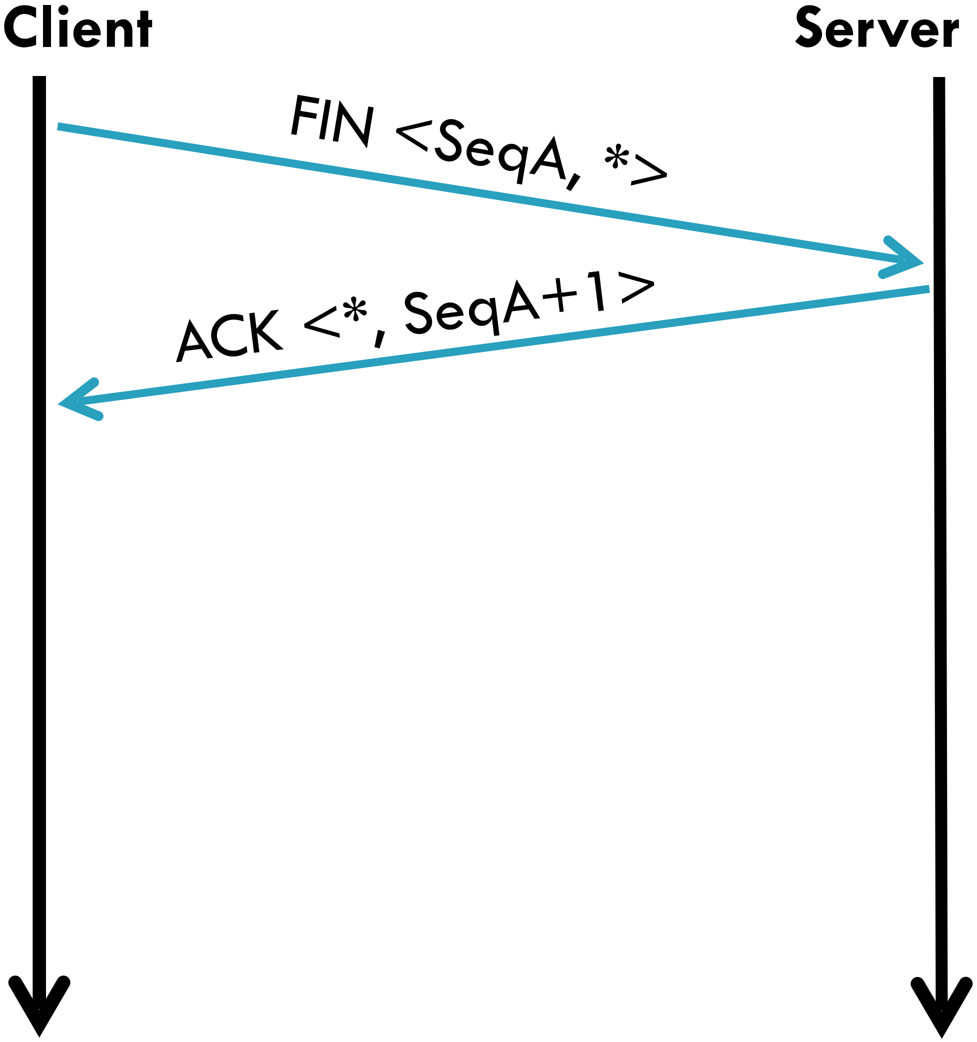

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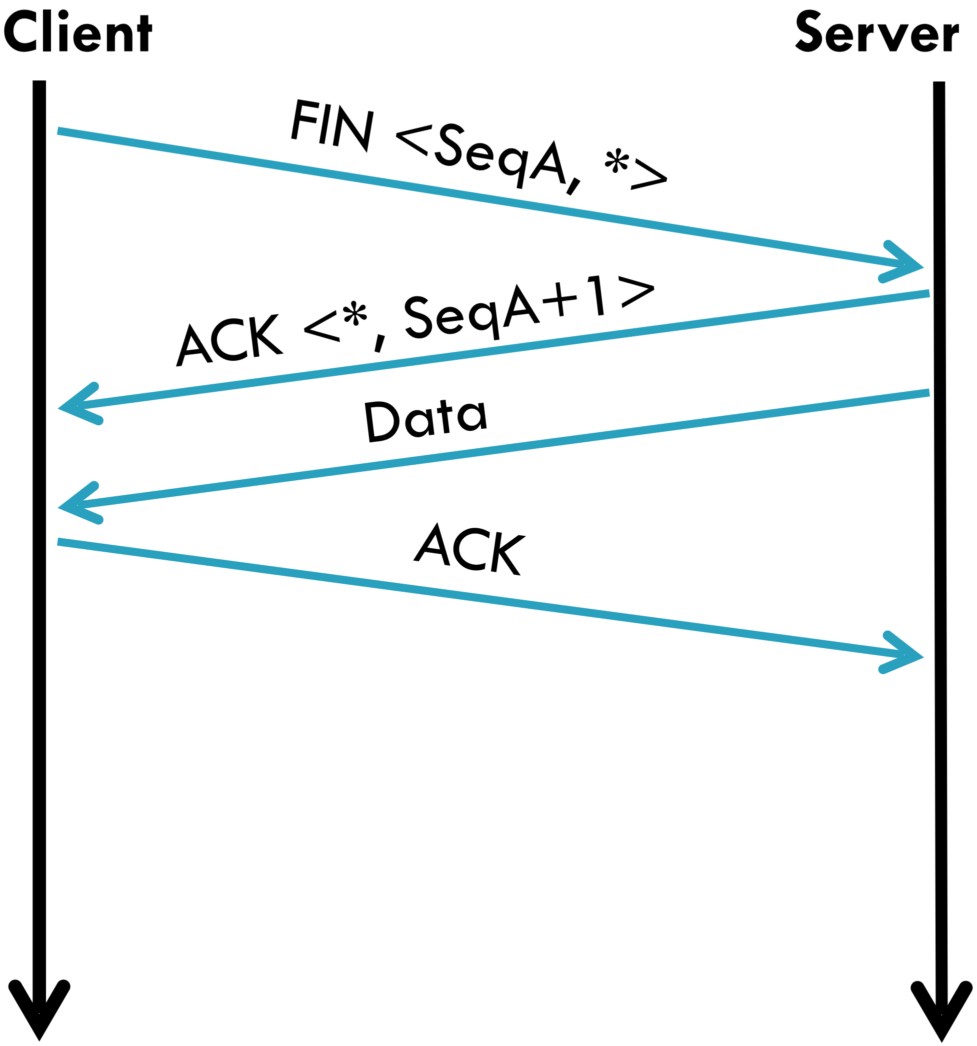
  - Half open connection
  - *shutdown()*



Connection Tear Down

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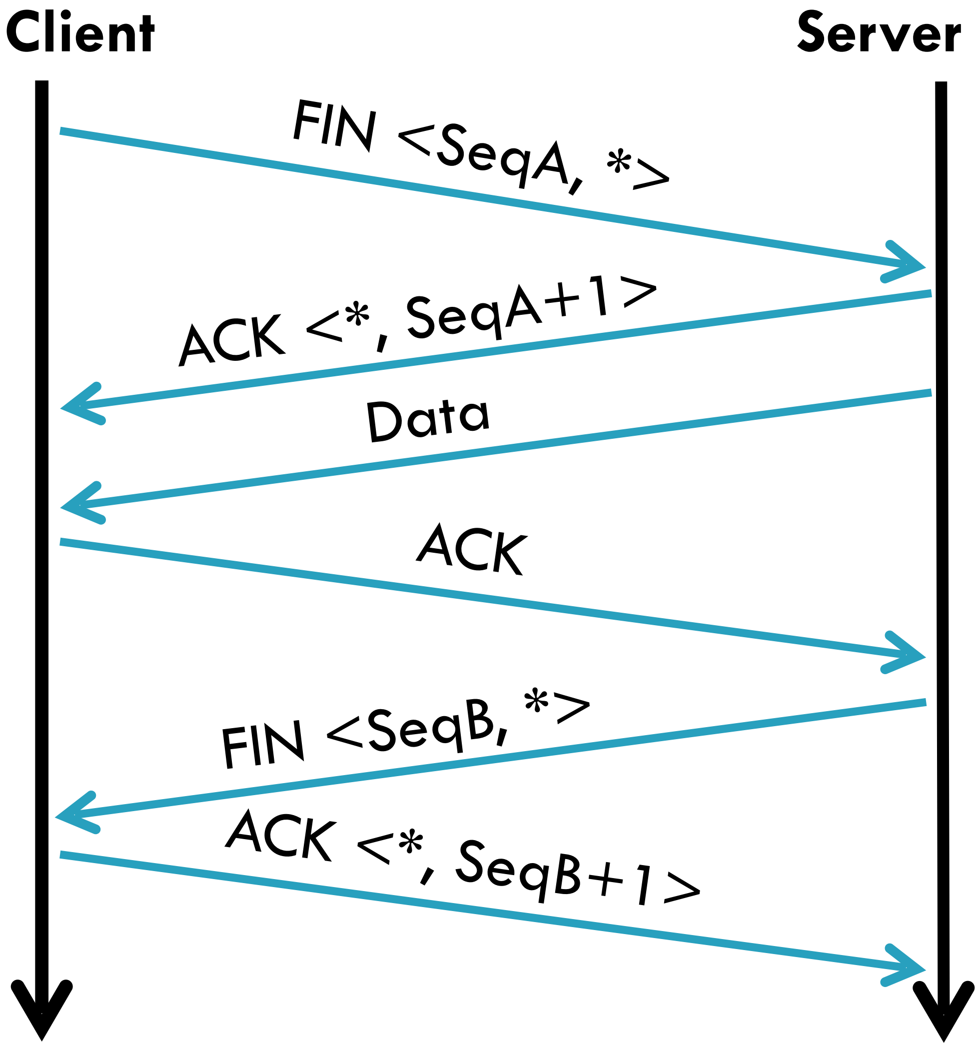
  - *shutdown()*
- Acknowledge the last FIN
  - Sequence number + 1



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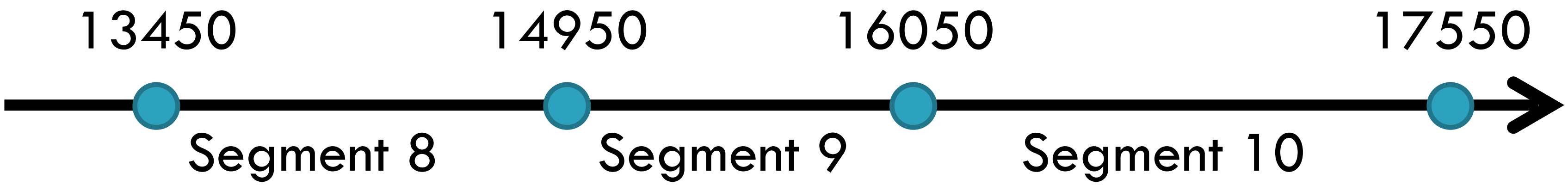
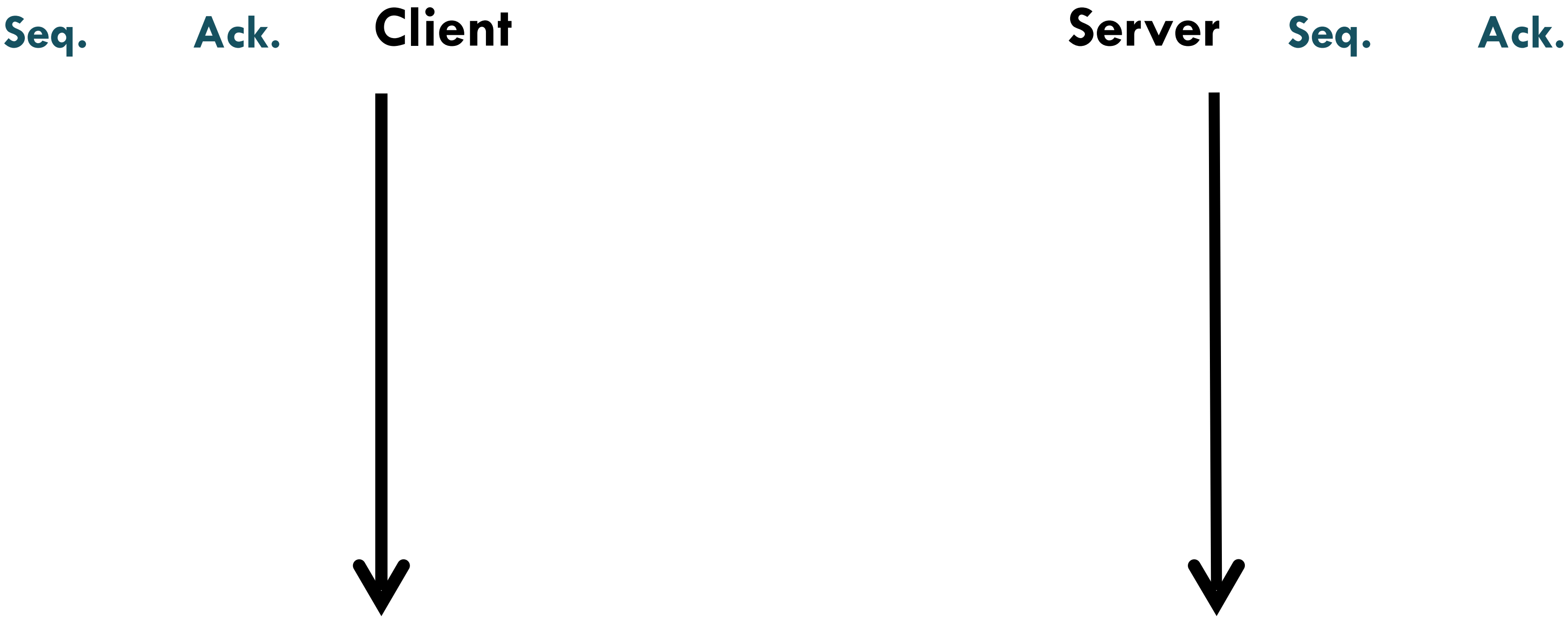


Bidirectional Communication



- Each side of the connection can send and receive
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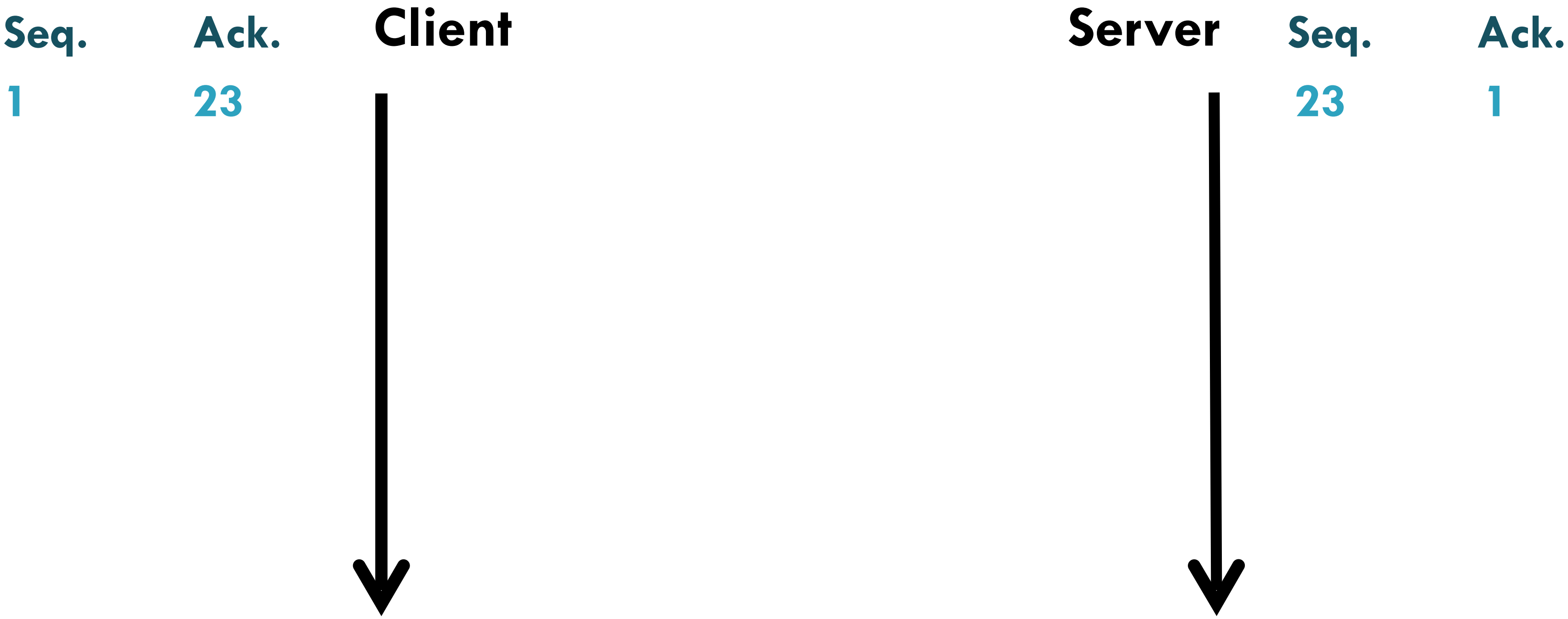
Bidirectional Communication



Seq.	Ack.	Client	Server	Seq.	Ack.
1	23			23	1

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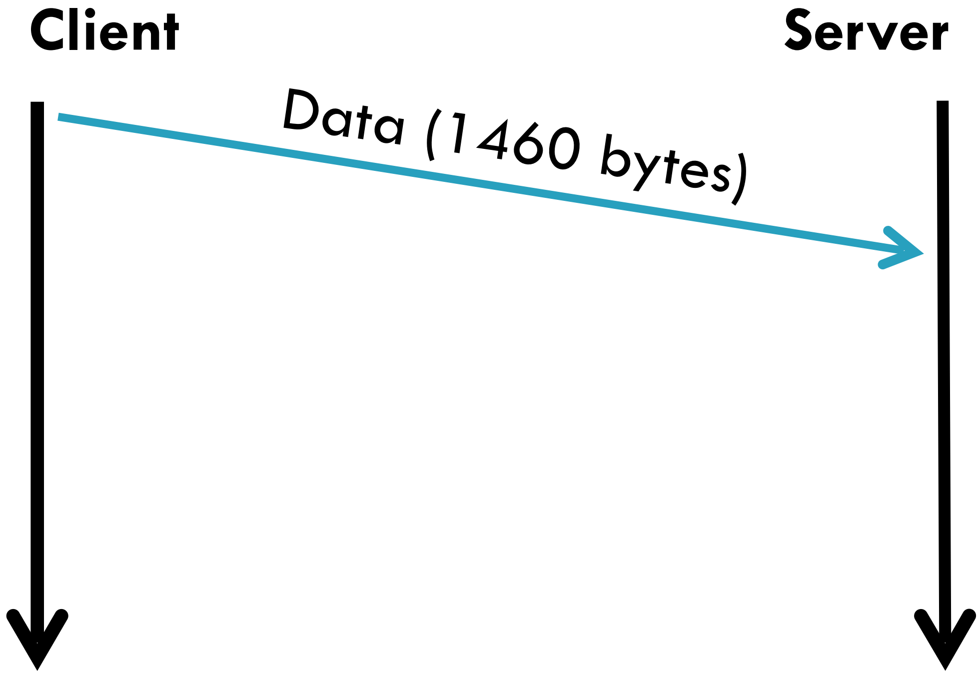

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				23	1461

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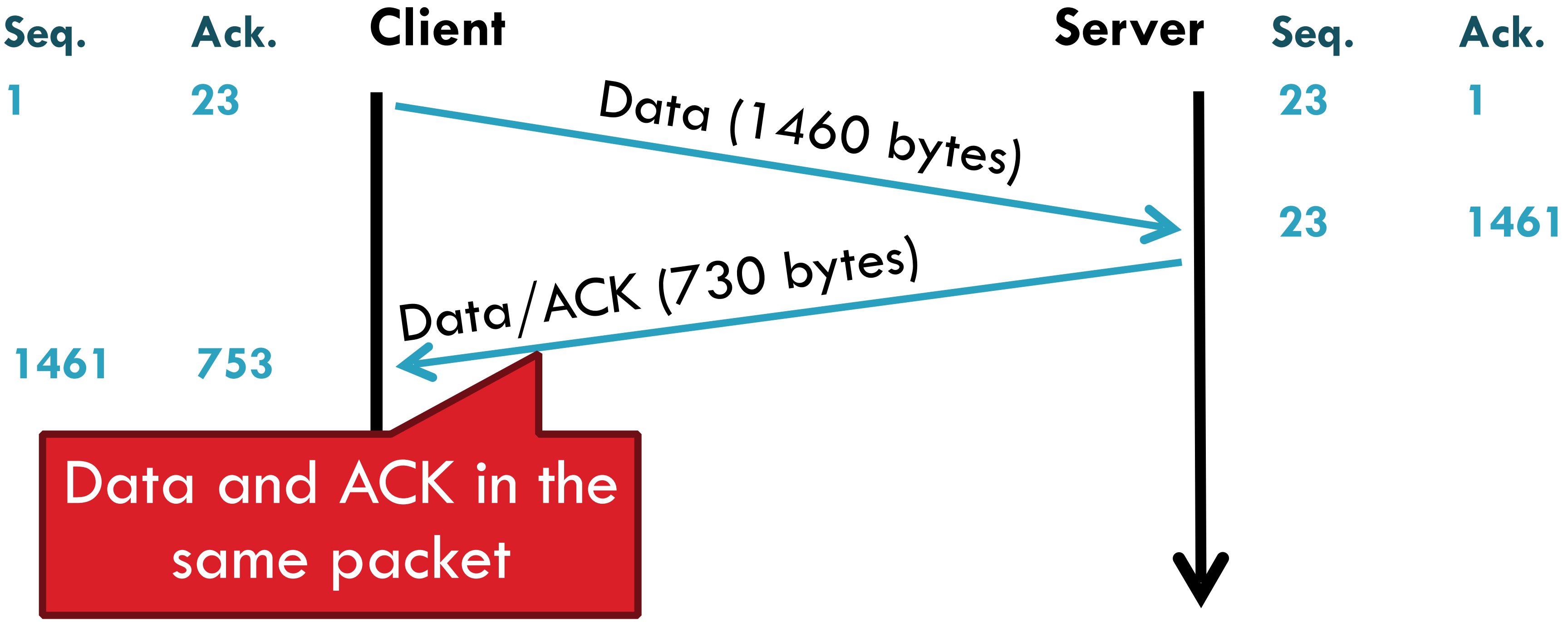
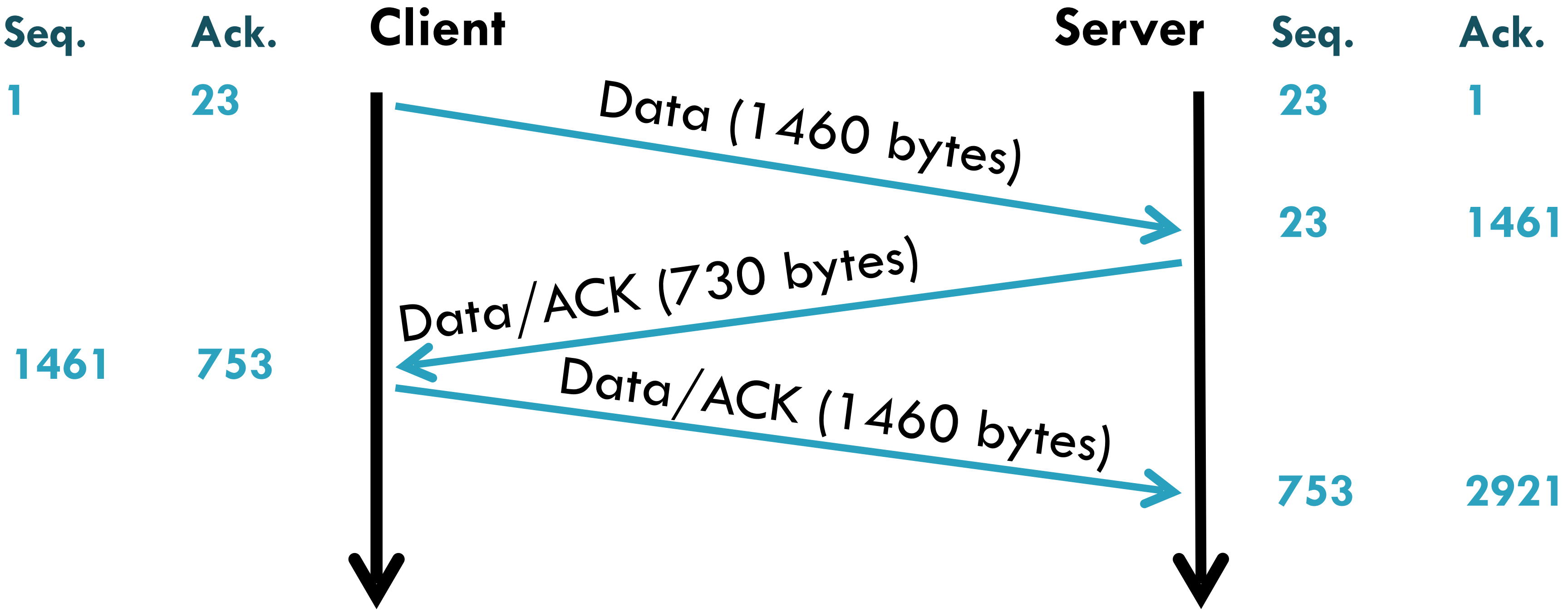


Data and ACK in the same packet

- Each side of the connection can send and receive
  - Different sequence numbers for each direction

Bidirectional Communication



Seq.	Ack.	Client		Server	Seq.	Ack.
1	23		Data (1460 bytes) →		23	1
					23	1461
			← Data/ACK (730 bytes)			
1461	753		Data/ACK (1460 bytes) →			
					753	2921

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  - Different sequence numbers for each direction

Flow Control

- Problem: how many packets should a sender transmit?
  - Too many packets may overwhelm the receiver
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  - For window size *n*, sender may transmit *n* bytes without receiving an ACK
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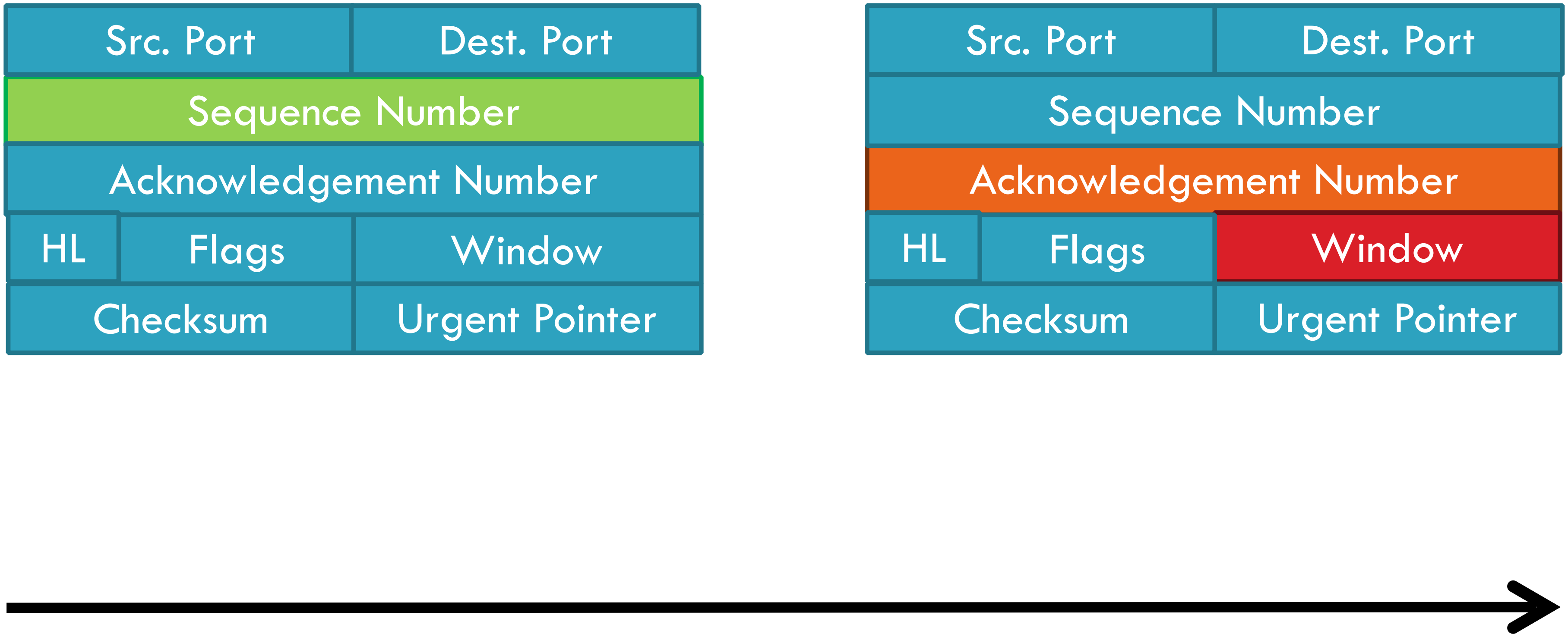
- Window may go to zero!

Flow Control: Sender Side



Packet Sent

Src. Port | Dest. Port

Sequence Number

Acknowledgement Number

HL | Flags | Window

Checksum | Urgent Pointer

Packet Received

Src. Port | Dest. Port

Sequence Number

Acknowledgement Number

HL | Flags | Window

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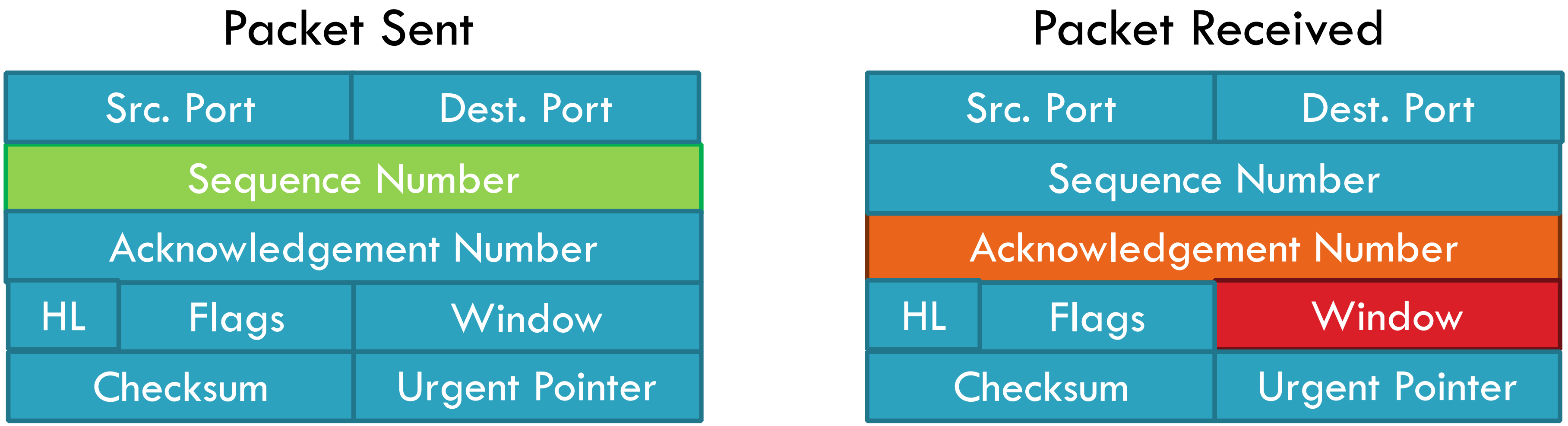

Packet Sent

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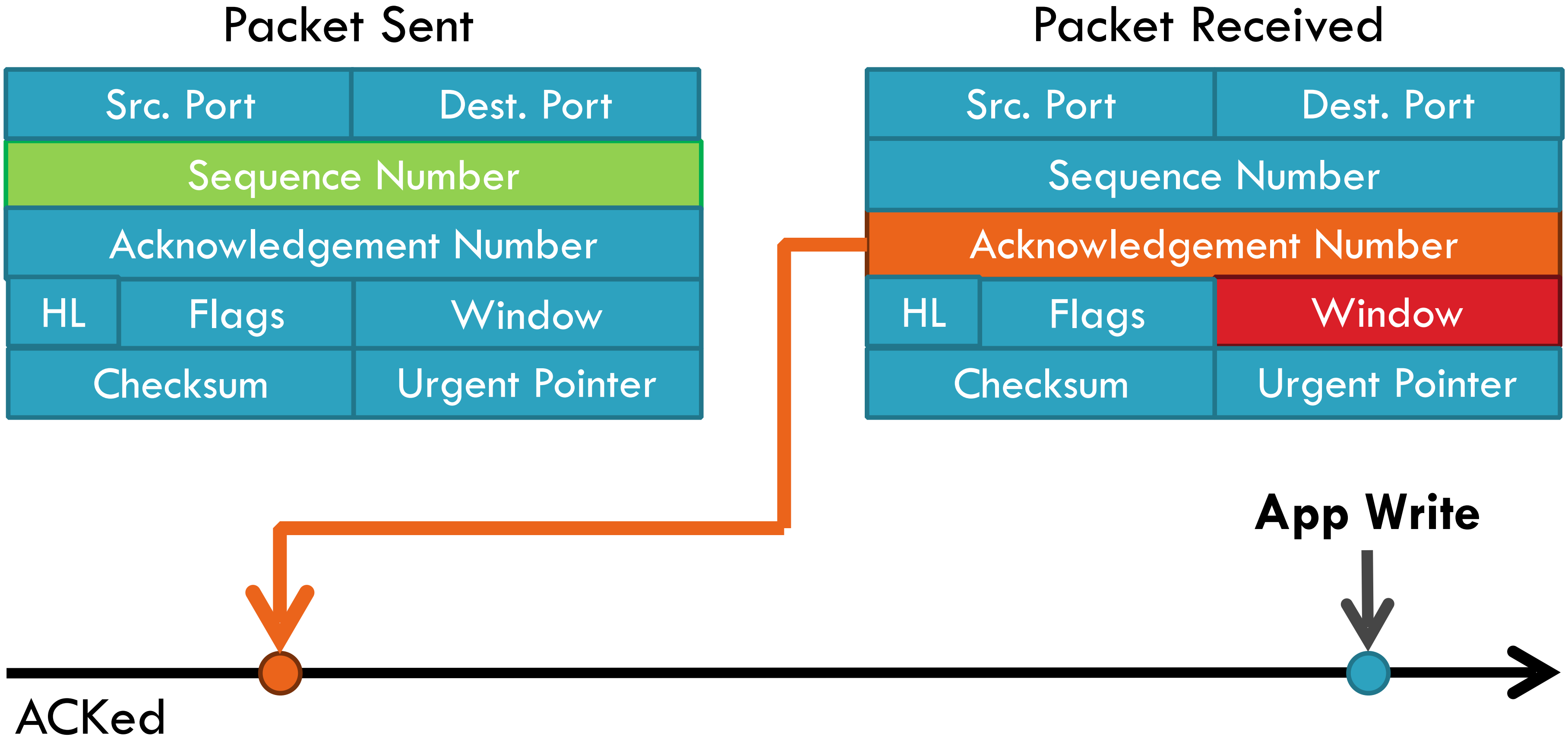

Packet Sent

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Packet Received

Src. Port | Dest. Port

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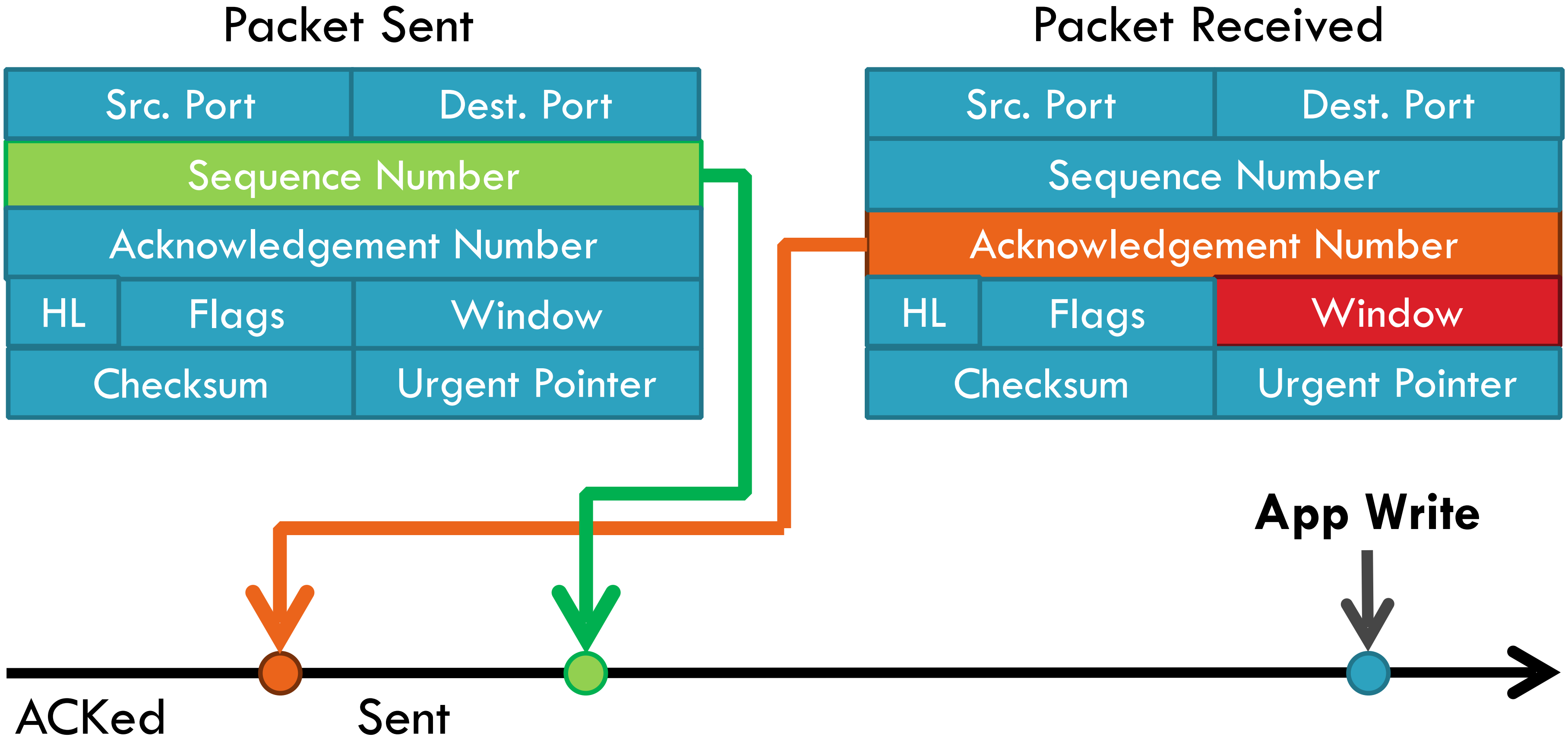
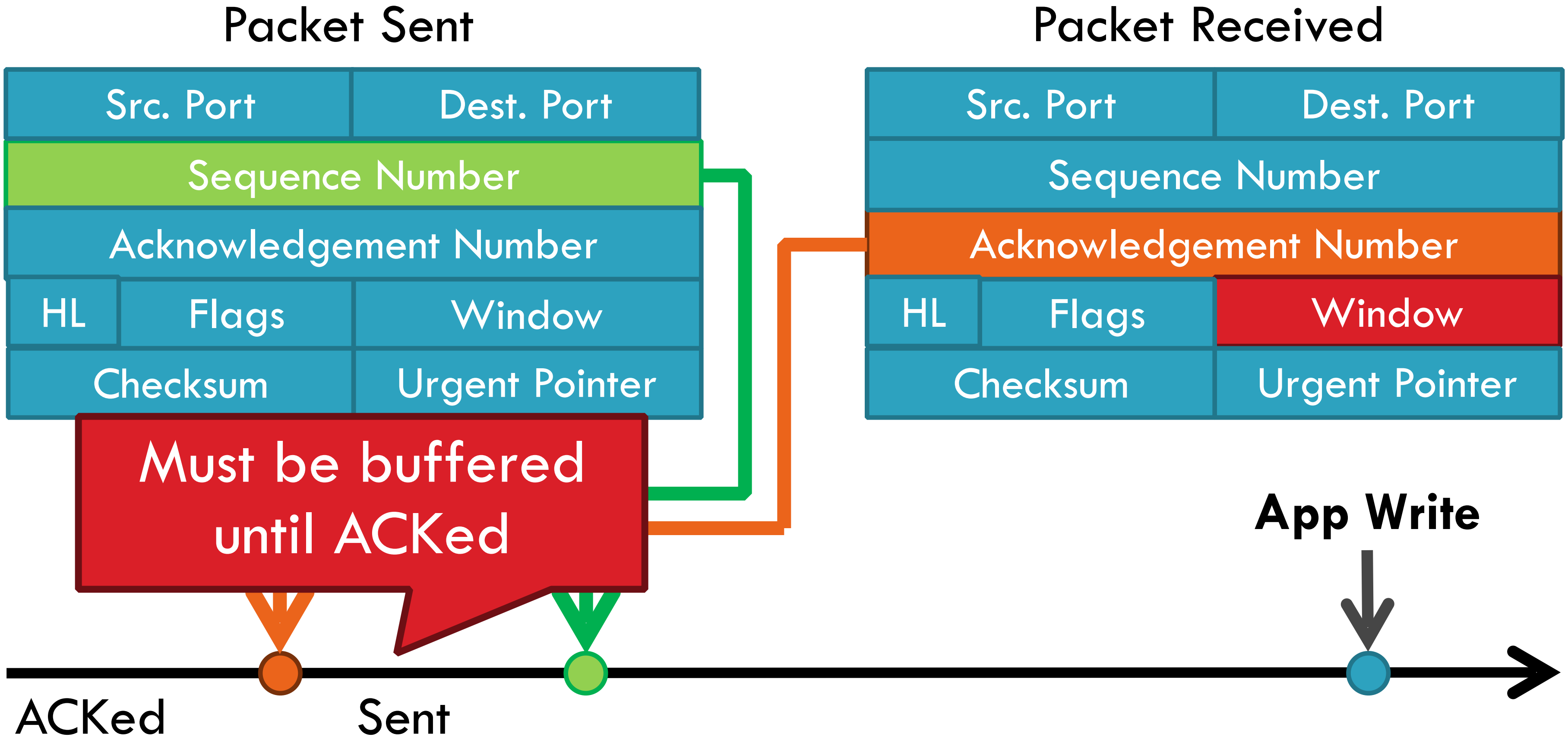
App Write

ACKed

Flow Control: Sender Side



Flow Control: Sender Side



Packet Sent

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Acknowledgement Number

HL | Flags | Window

Checksum | Urgent Pointer

Packet Received

Src. Port | Dest. Port

Sequence Number

Acknowledgement Number

HL | Flags | Window

Checksum | Urgent Pointer

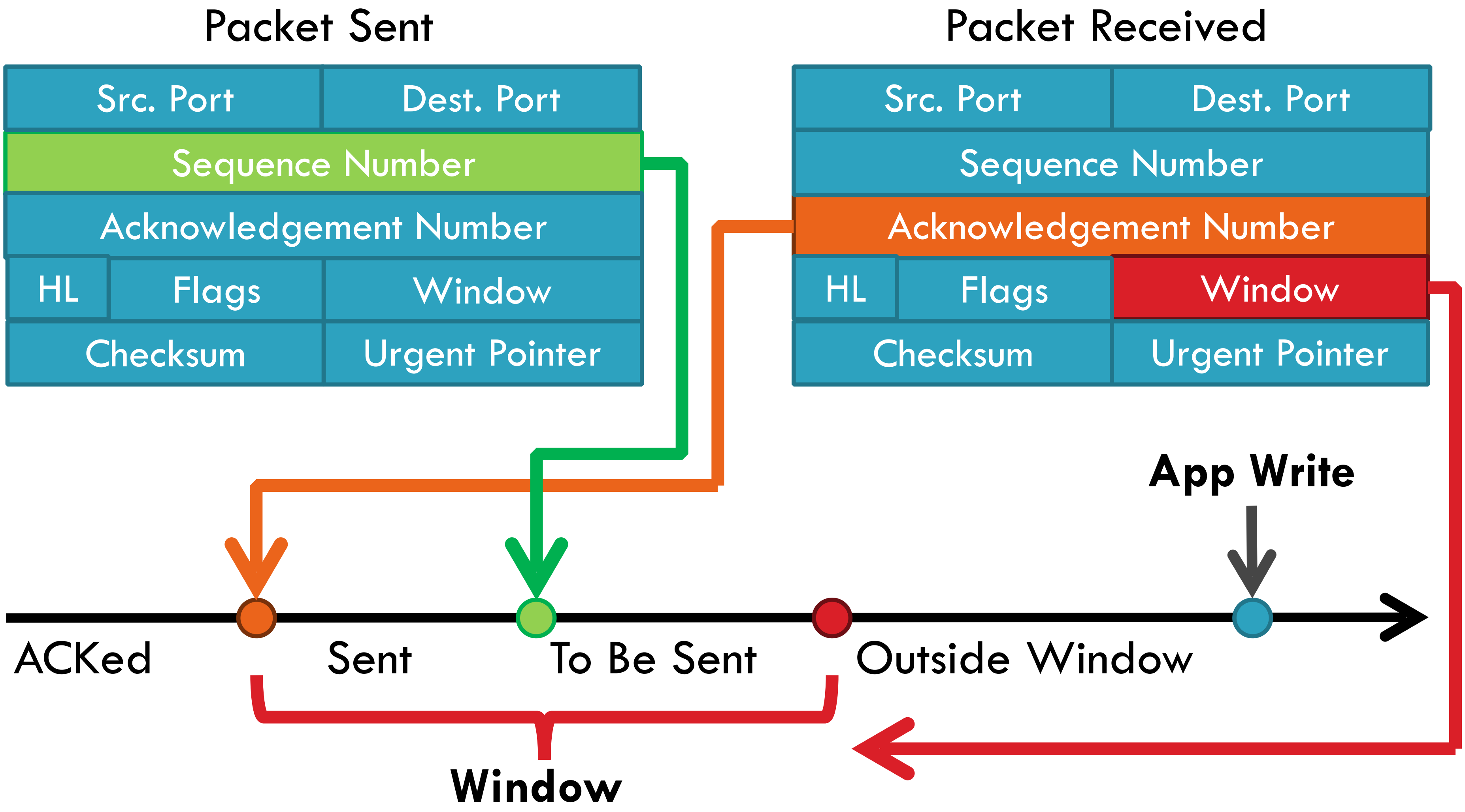
Must be buffered until ACKed

App Write

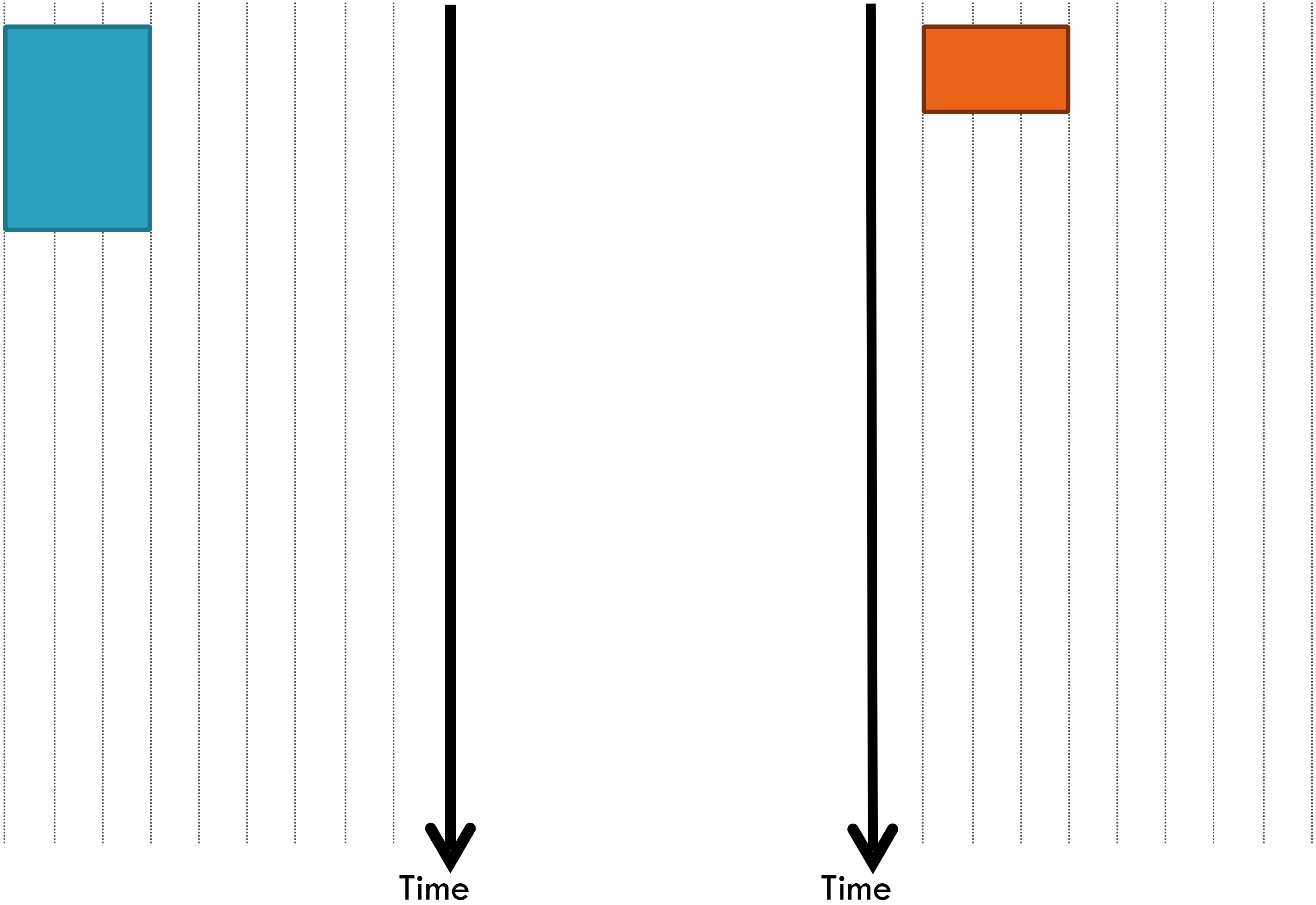
ACKed

Sent

Flow Control: Sender Side



Sliding Window Example



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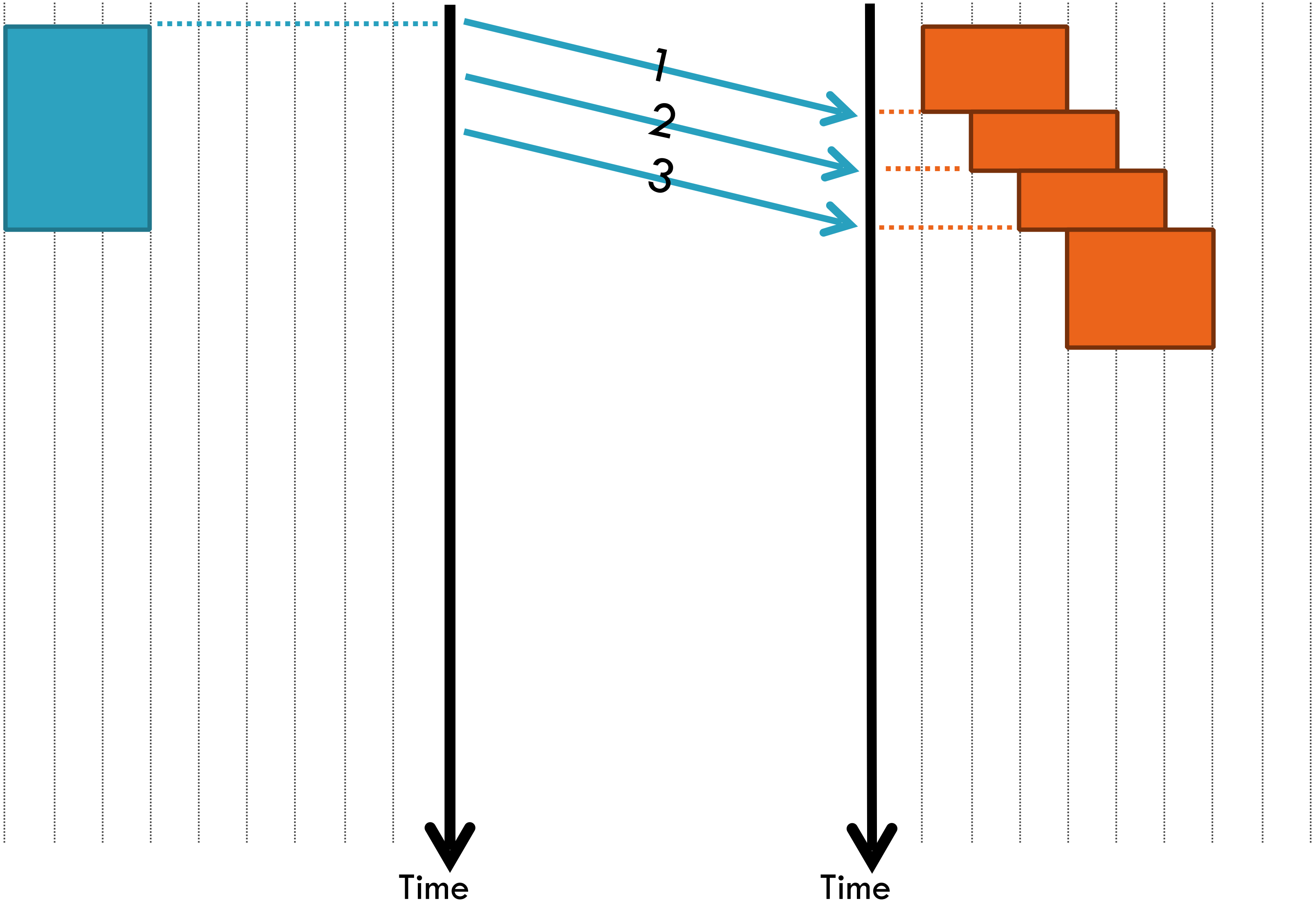

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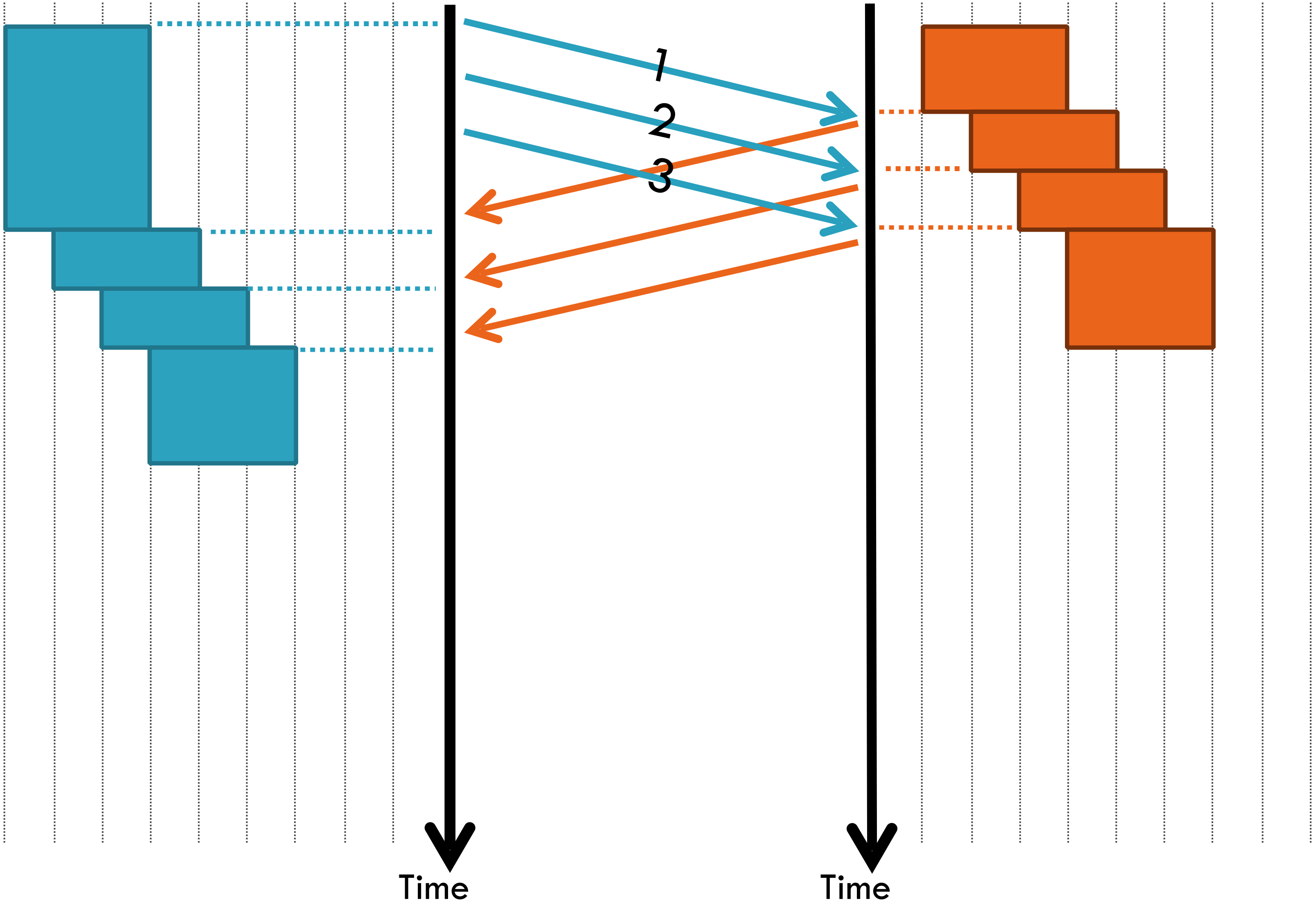

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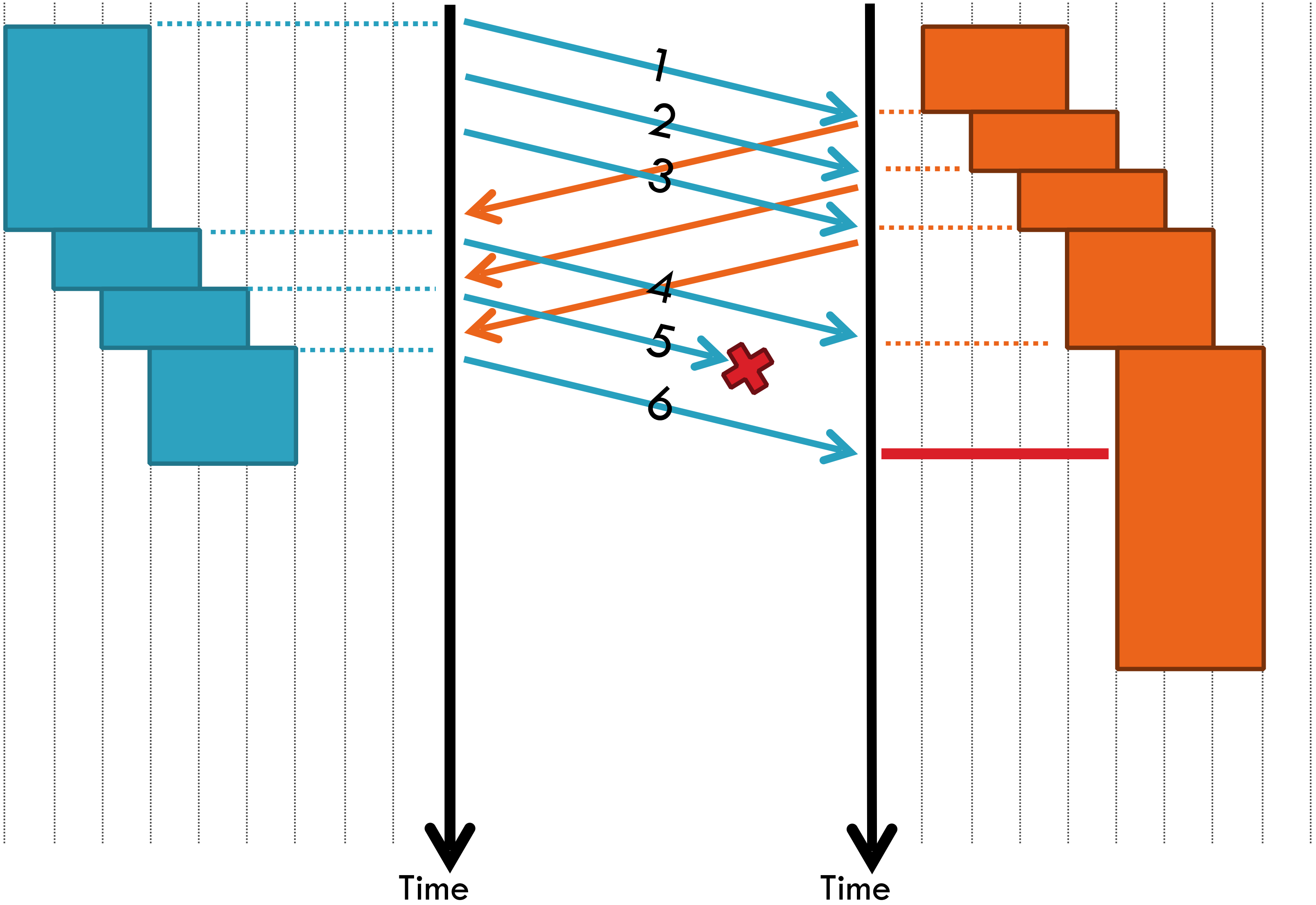

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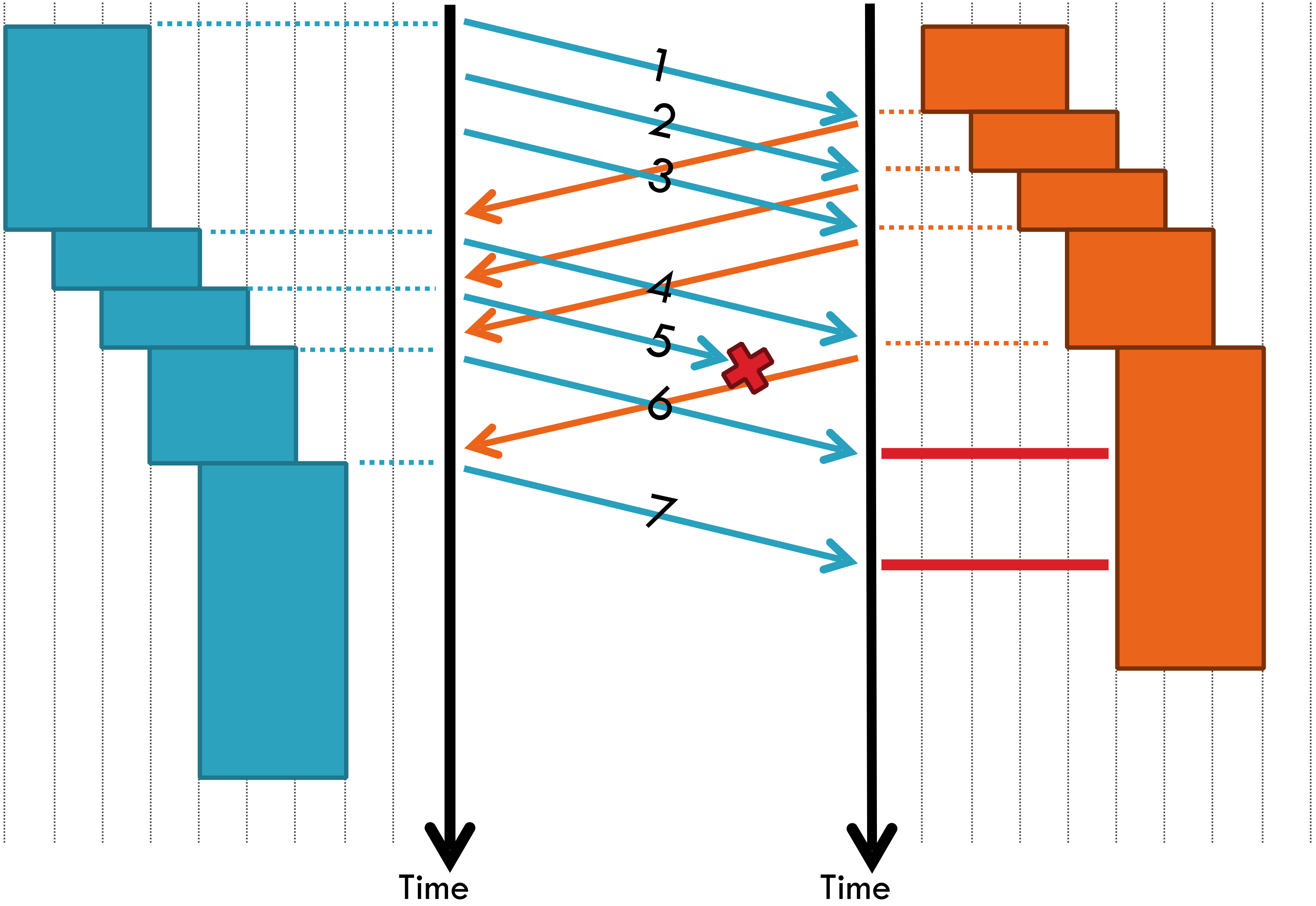

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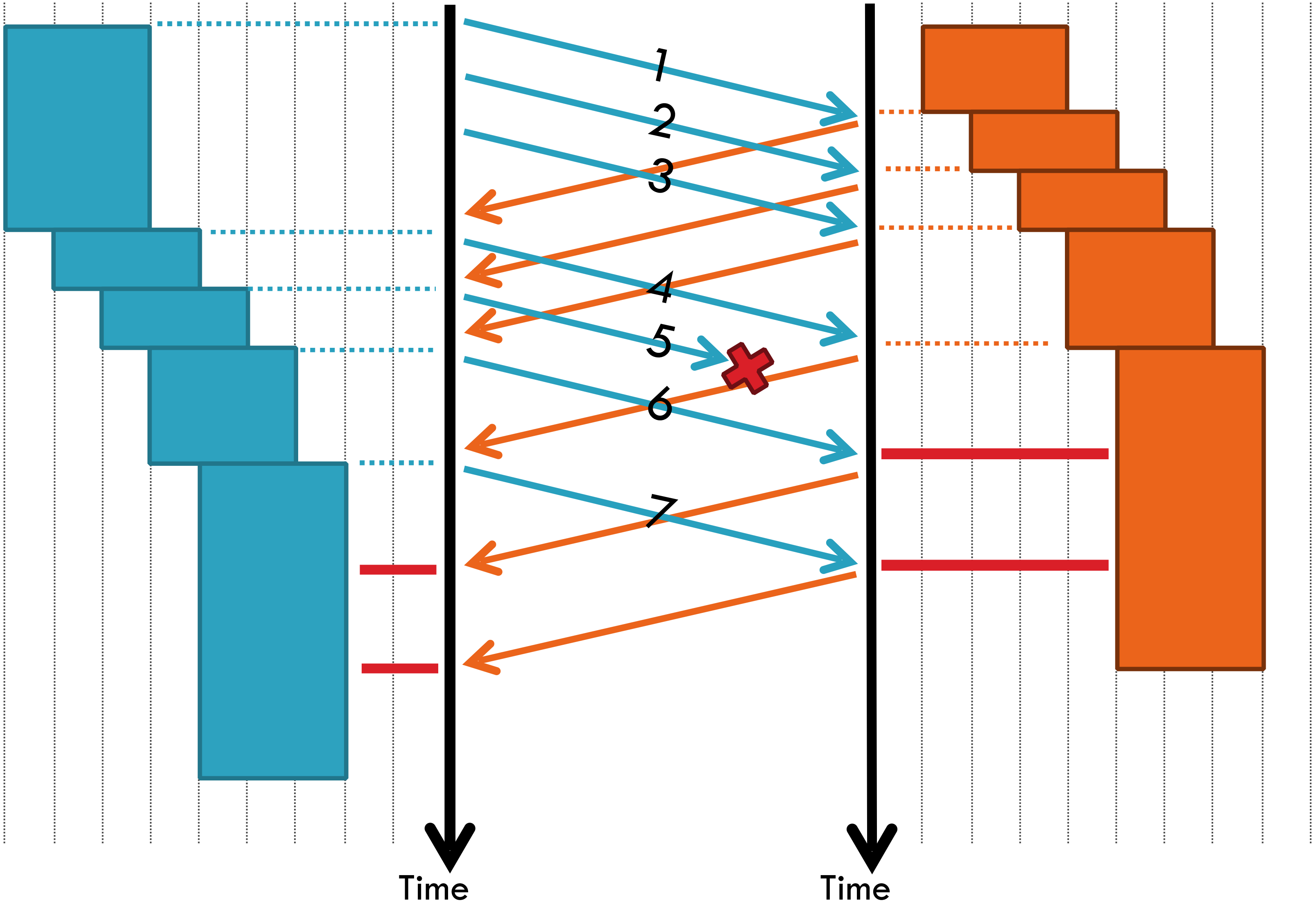

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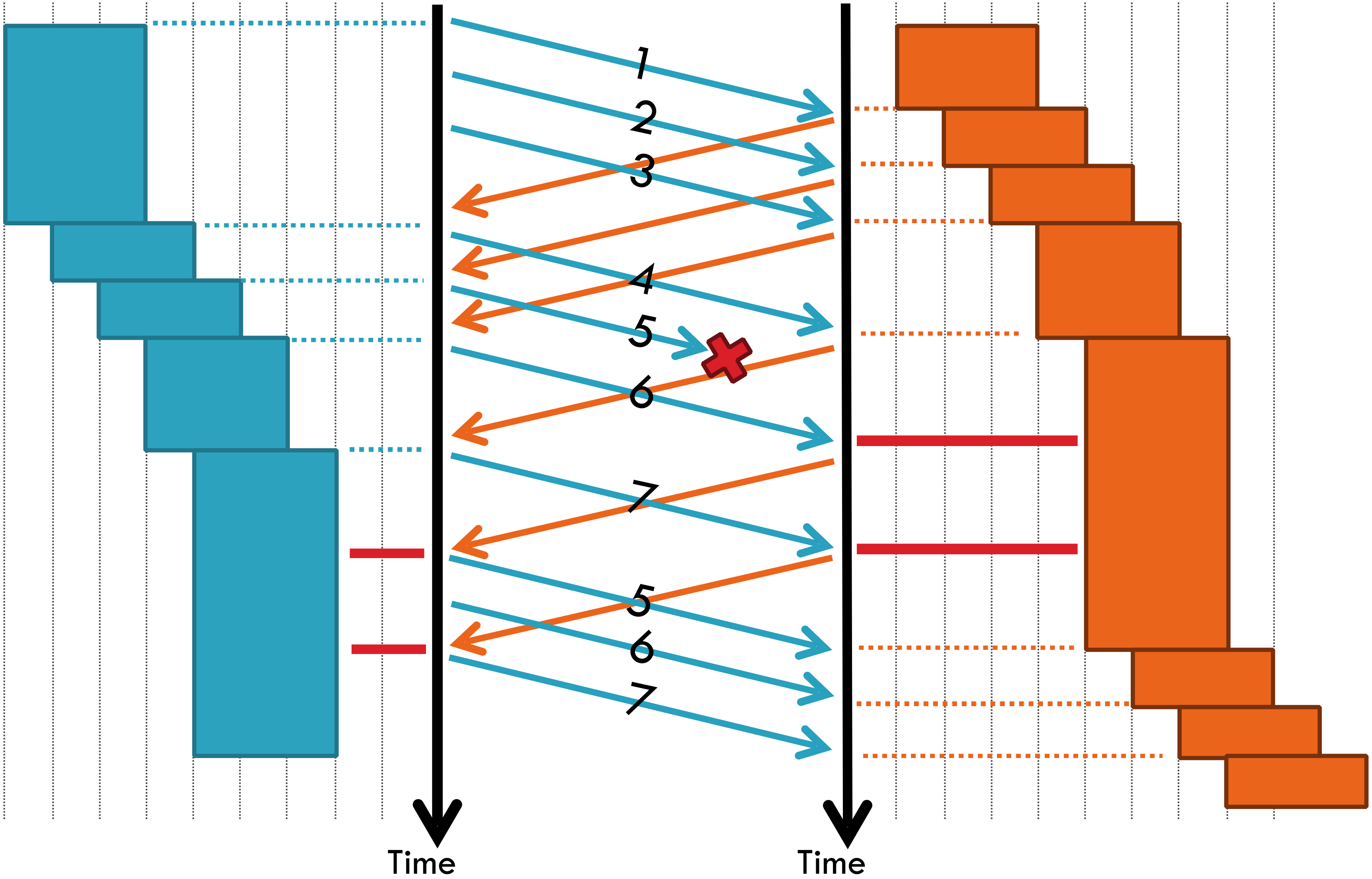

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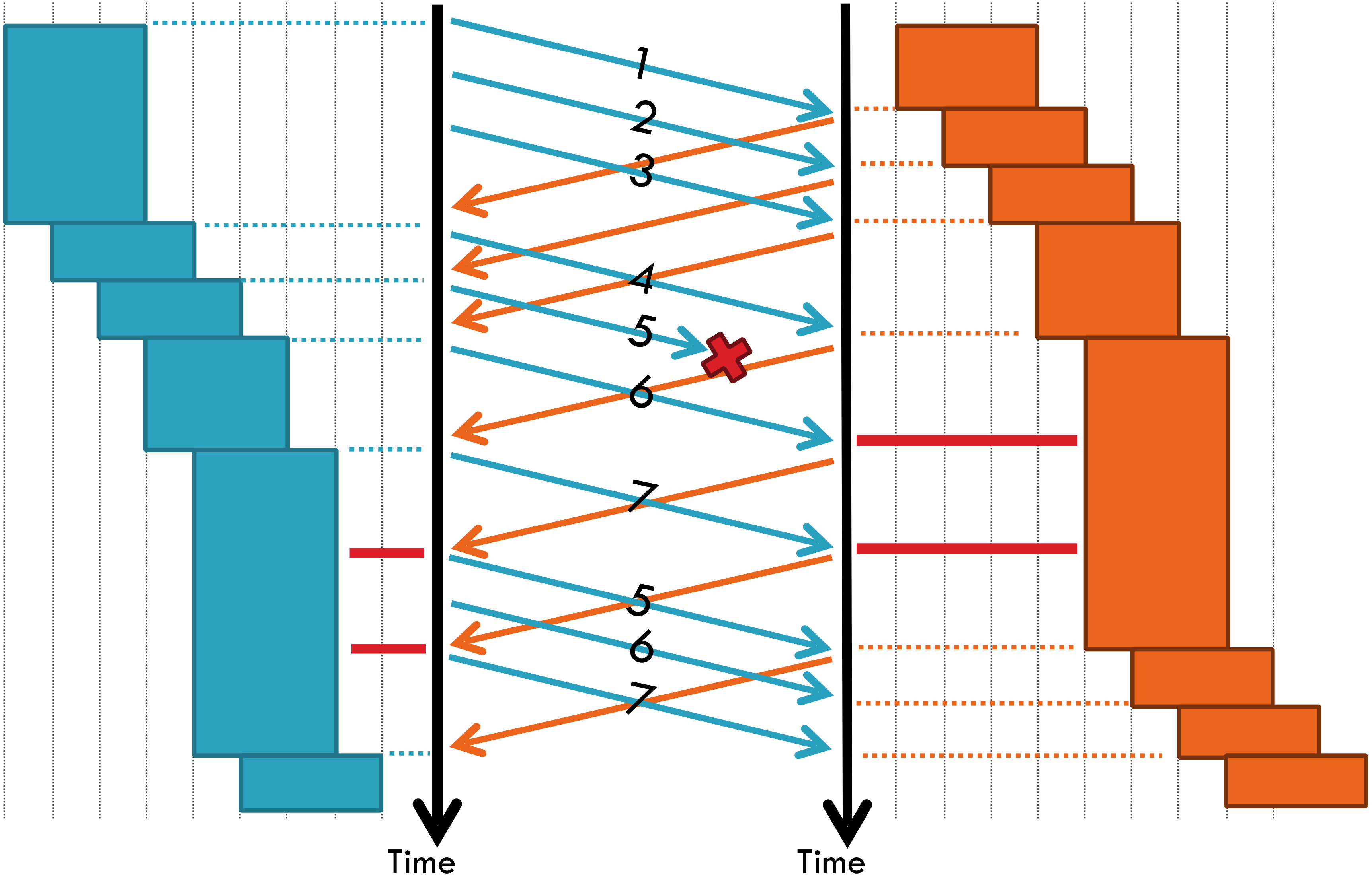

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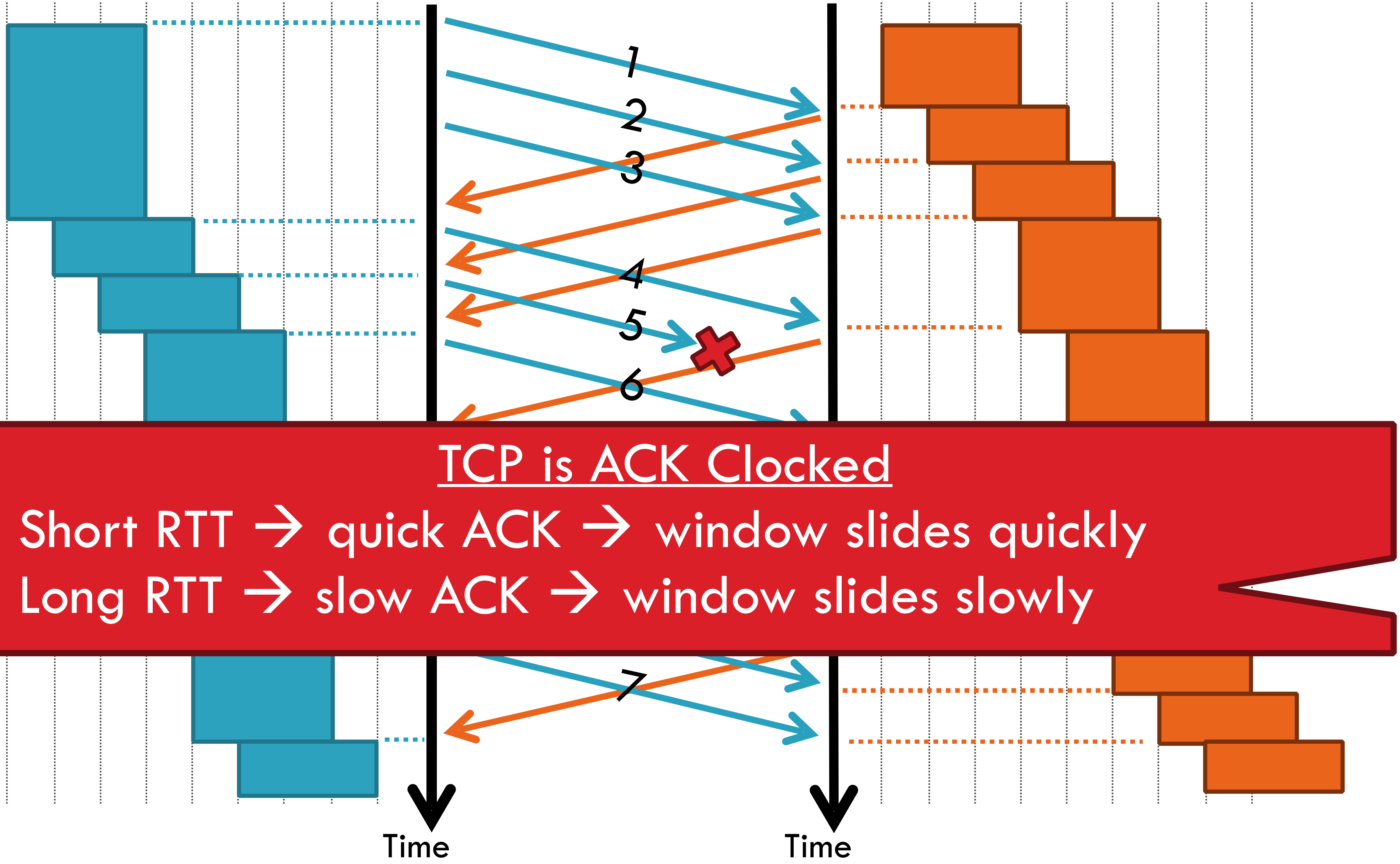

TCP is ACK Clocked

- Short RTT → quick ACK → window slides quickly
- Long RTT → slow ACK → window slides slowly

What Should the Receiver ACK?

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2. Use *cumulative ACK*, where an ACK for sequence n implies ACKS for all $k < n$
3. Use *negative ACKs* (NACKs), indicating which packet did not arrive

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    - What about GigE? PAWS algorithm + TCP options

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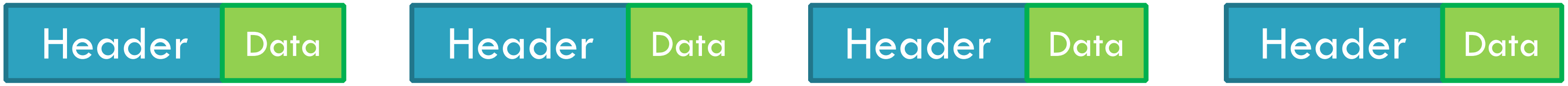

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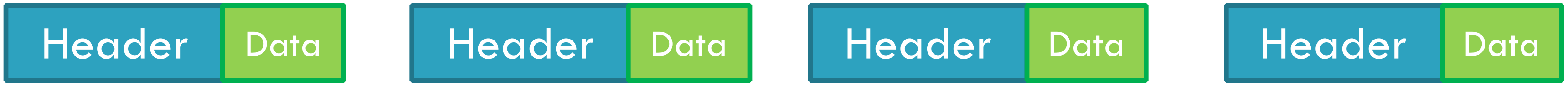

- Equivalent problem: sender transmits packets one byte at a time

```
for (int x = 0; x < strlen(data); ++x)
  write(socket, data + x, 1);
```

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1. If the window >= MSS and available data >= MSS:
    Send the data
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- Problem: Nagle's Algorithm delays transmissions
  - What if you need to send a packet immediately?
  1. int flag = 1;
  2. setsockopt(sock, IPPROTO_TCP, TCP_NODELAY, (char *) &flag, sizeof(int));

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- Lost segments detected by sender
  - Use timeout to detect missing ACKs
  - Need to estimate RTT to calibrate the timeout
  - Sender must keep copies of all data until ACK

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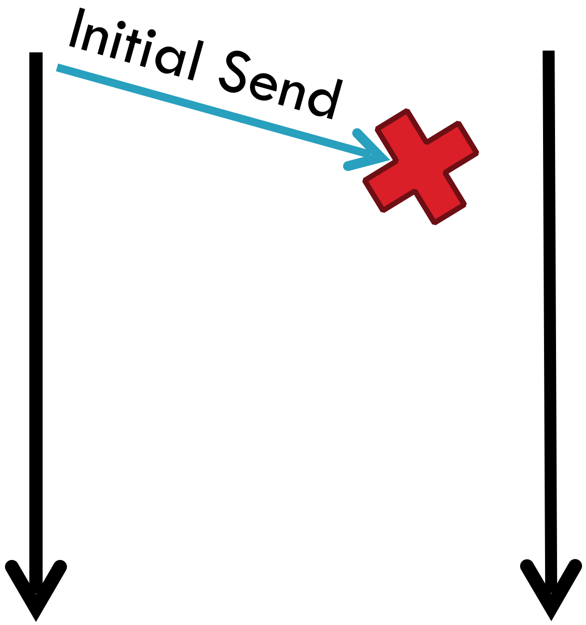

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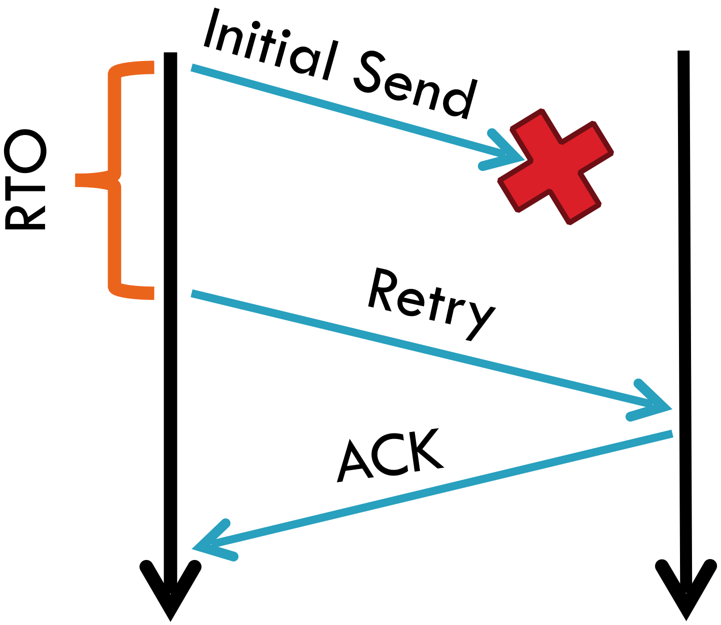

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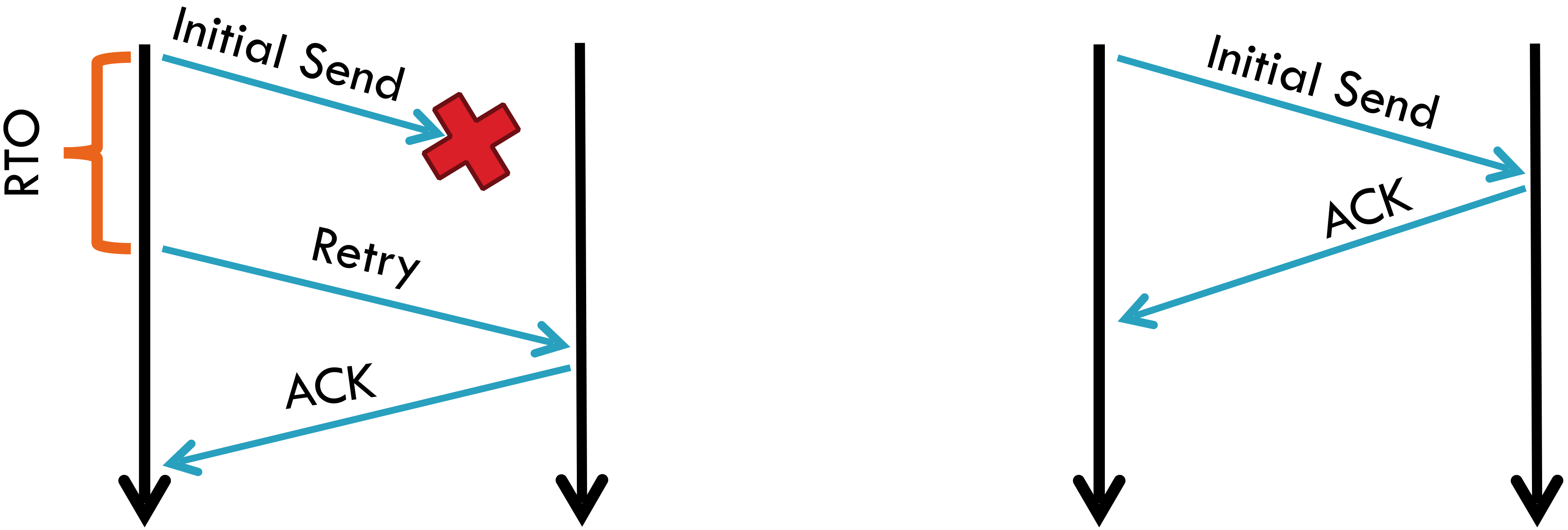

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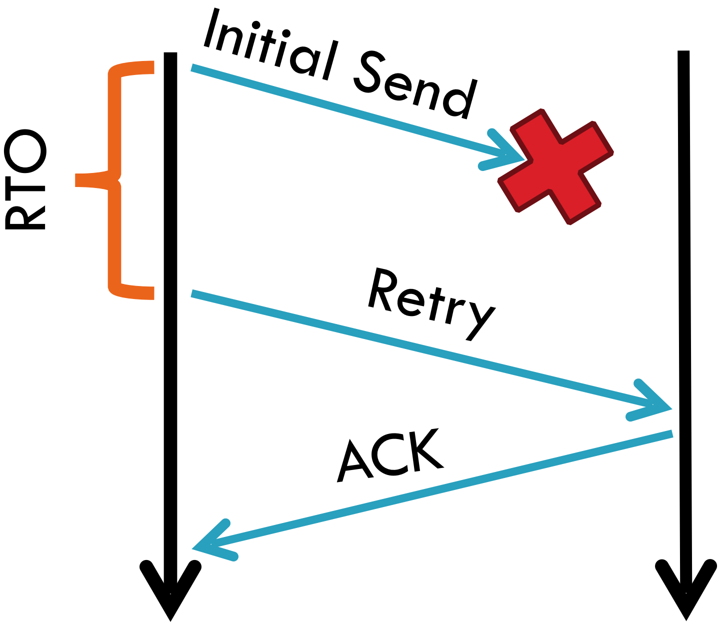

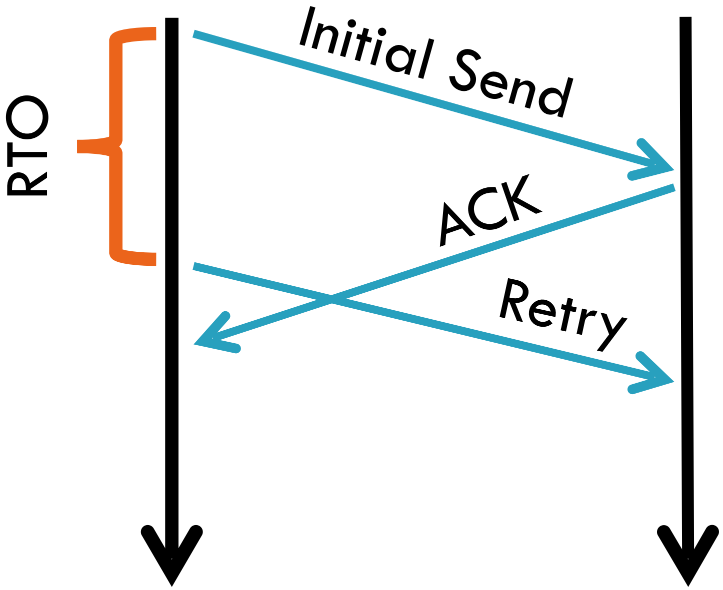

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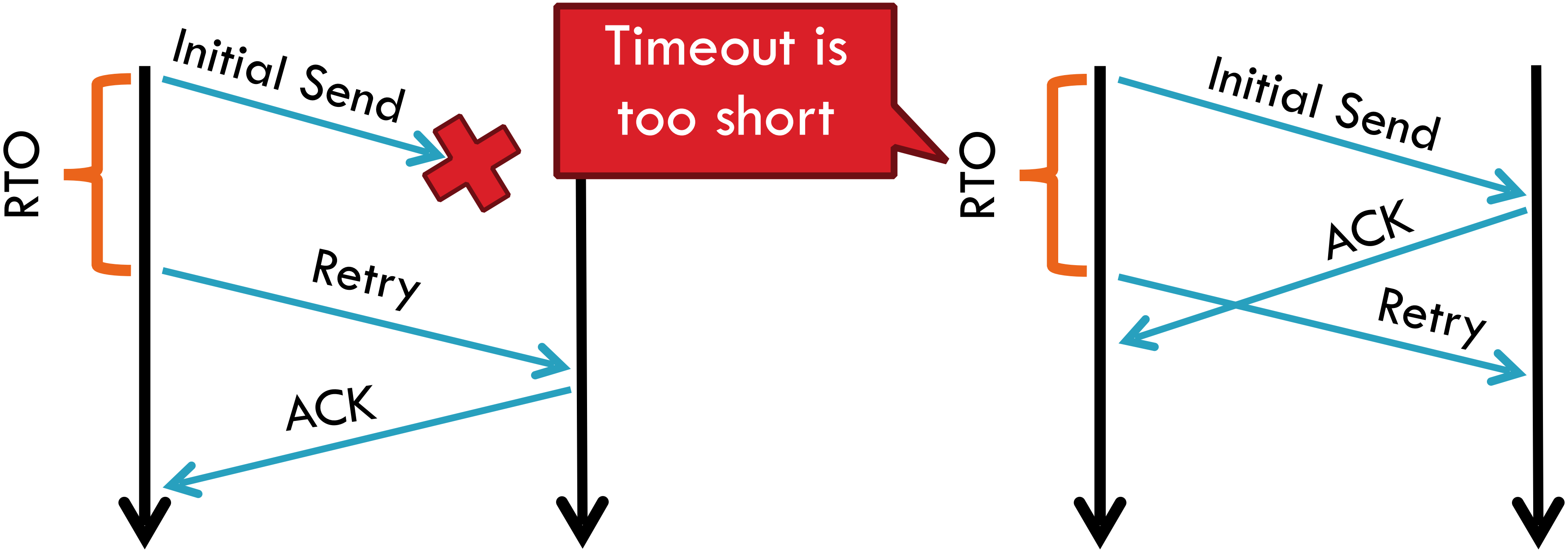

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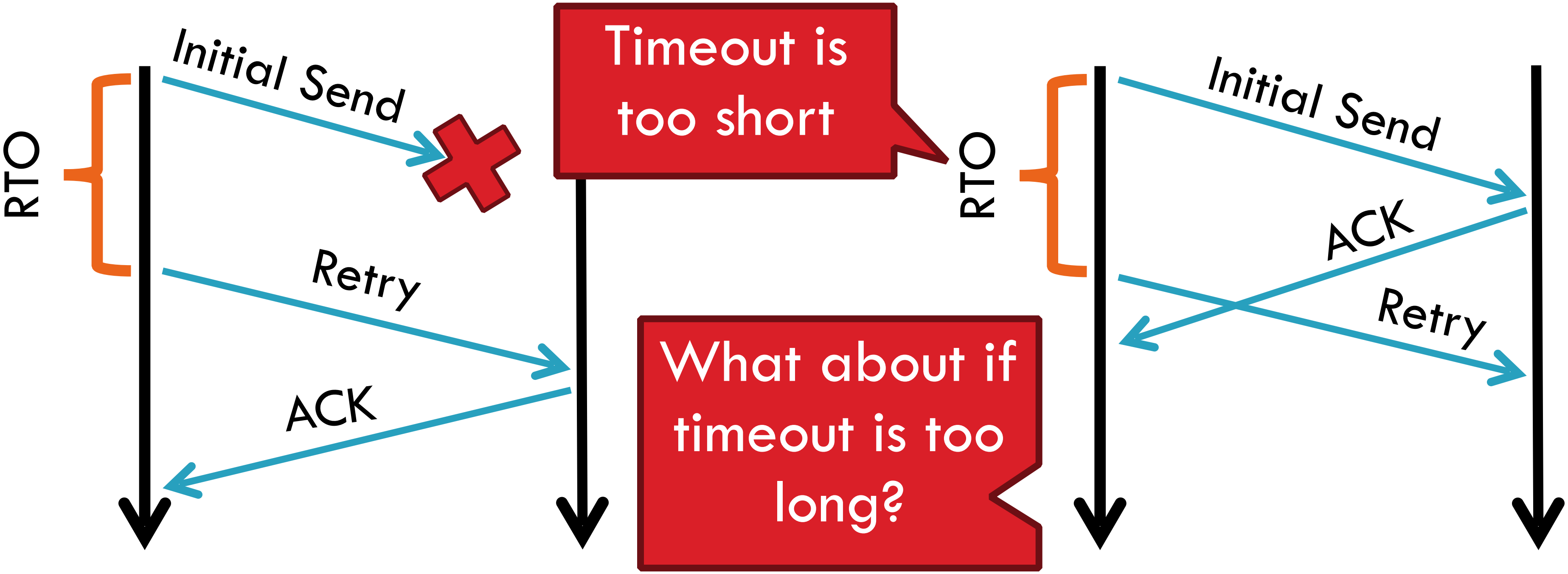
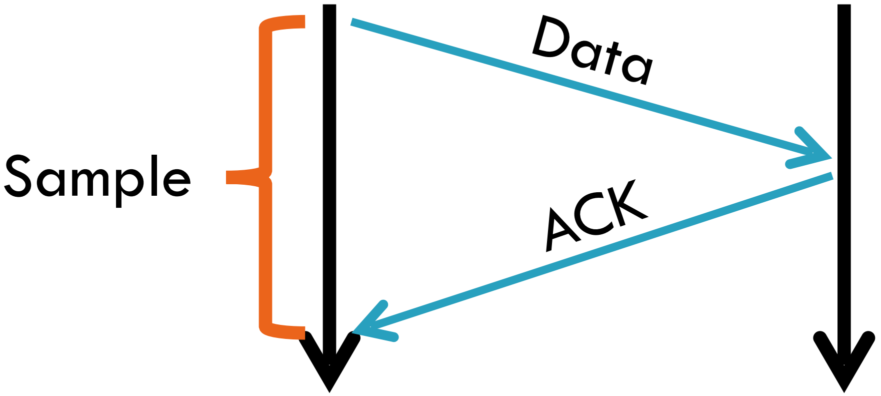


Timeout is too short

What about if timeout is too long?

Round Trip Time Estimation



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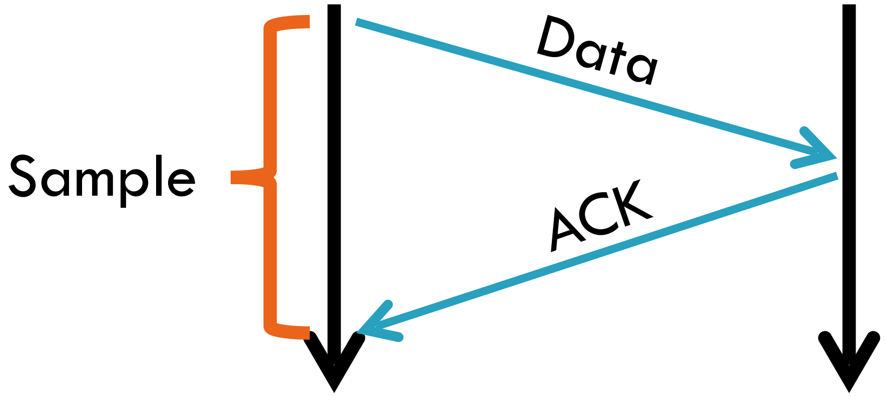
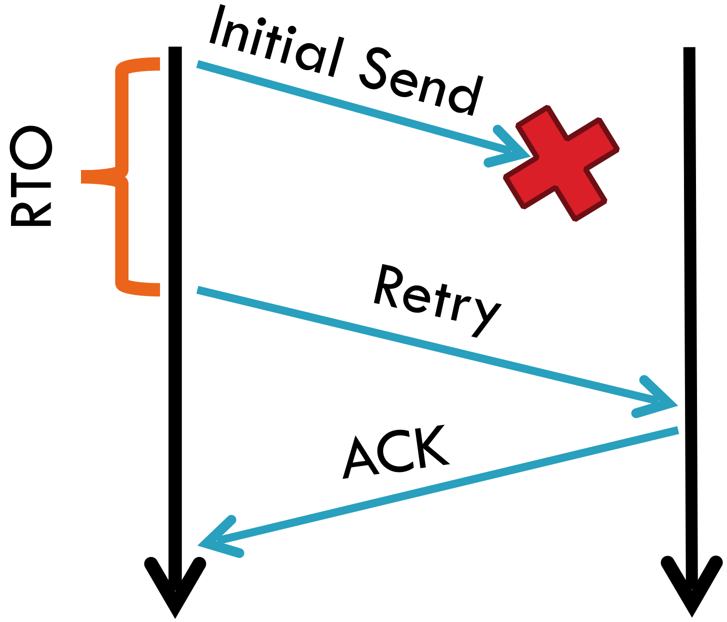
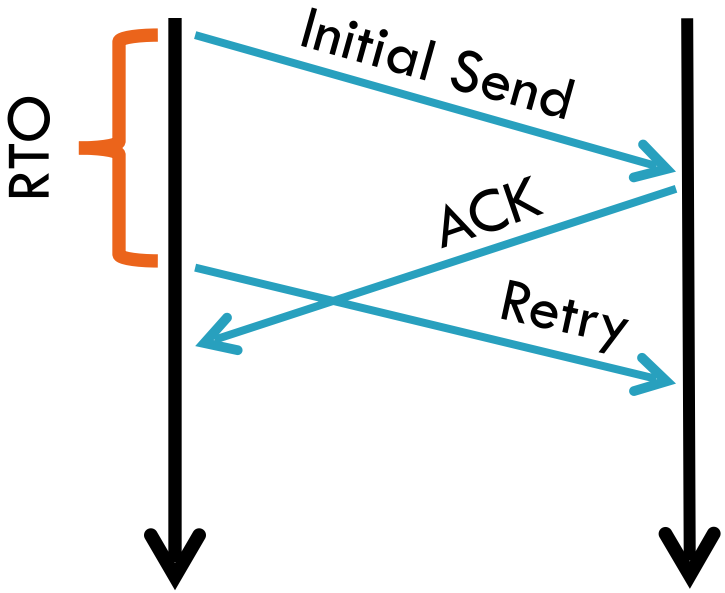


- Original TCP round-trip estimator
  - RTT estimated as a moving average
  - new_rtt = α (old_rtt) + (1 – α)(new_sample)
  - Recommended α: 0.8-0.9 (0.875 for most TCPs)
- RTO = 2 * new_rtt (i.e. TCP is conservative)

RTT Sample Ambiguity





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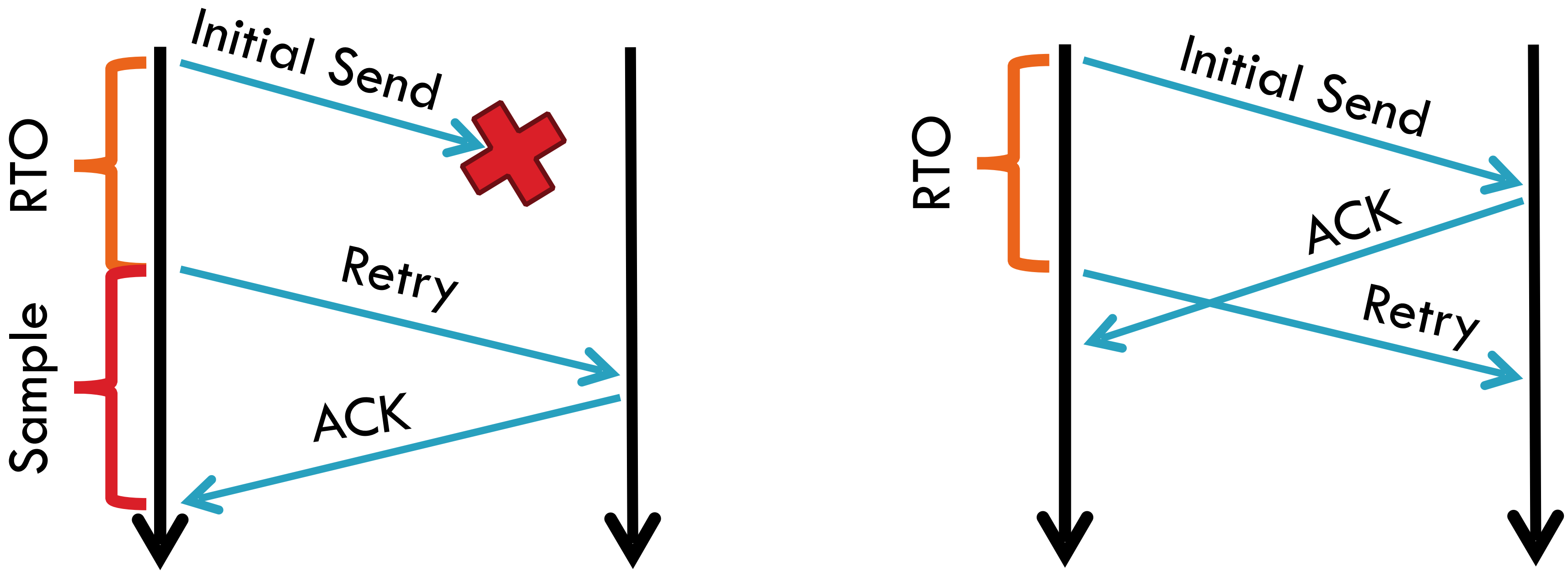

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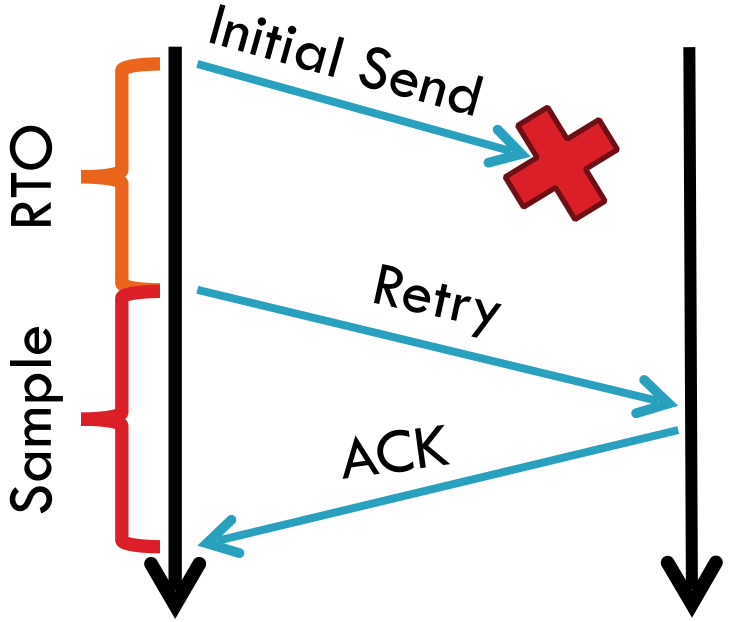

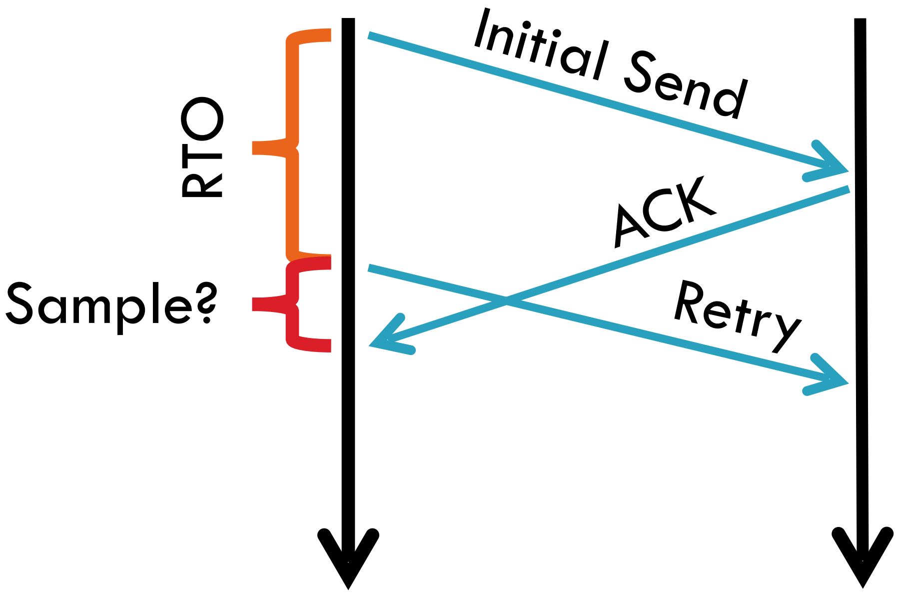

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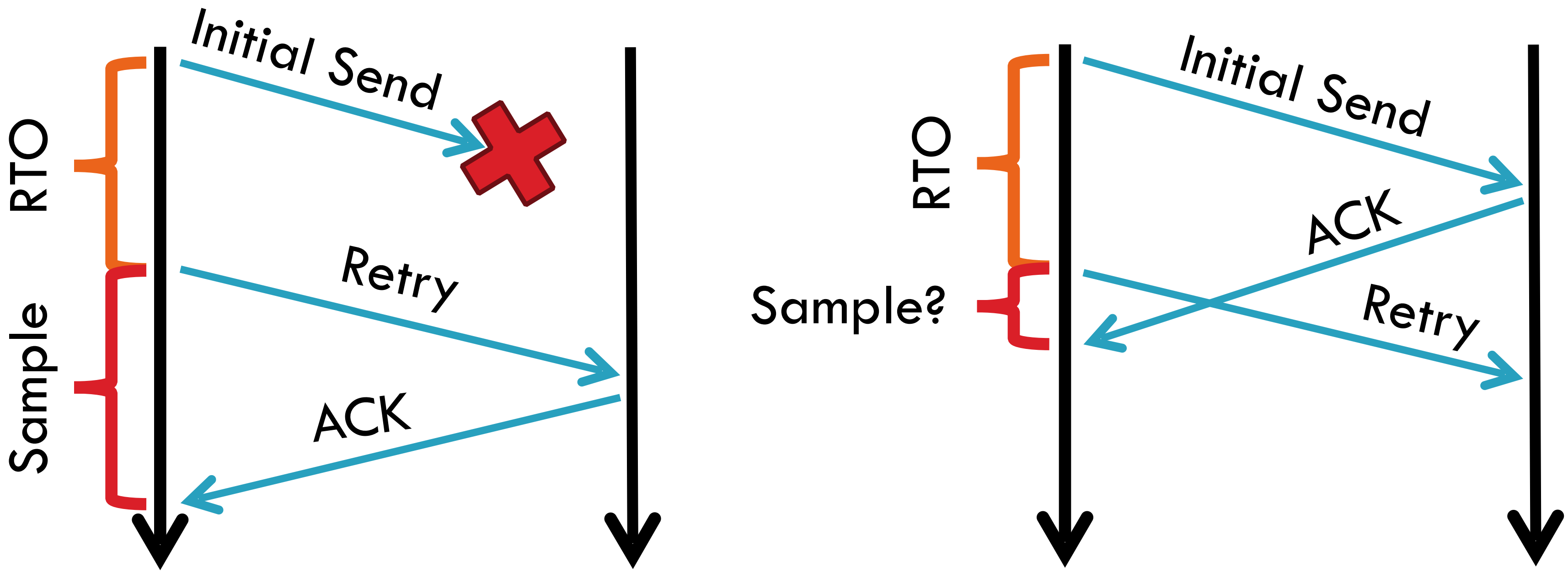

- Karn’s algorithm: ignore samples for retransmitted segments