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CA169
Networks & Internet

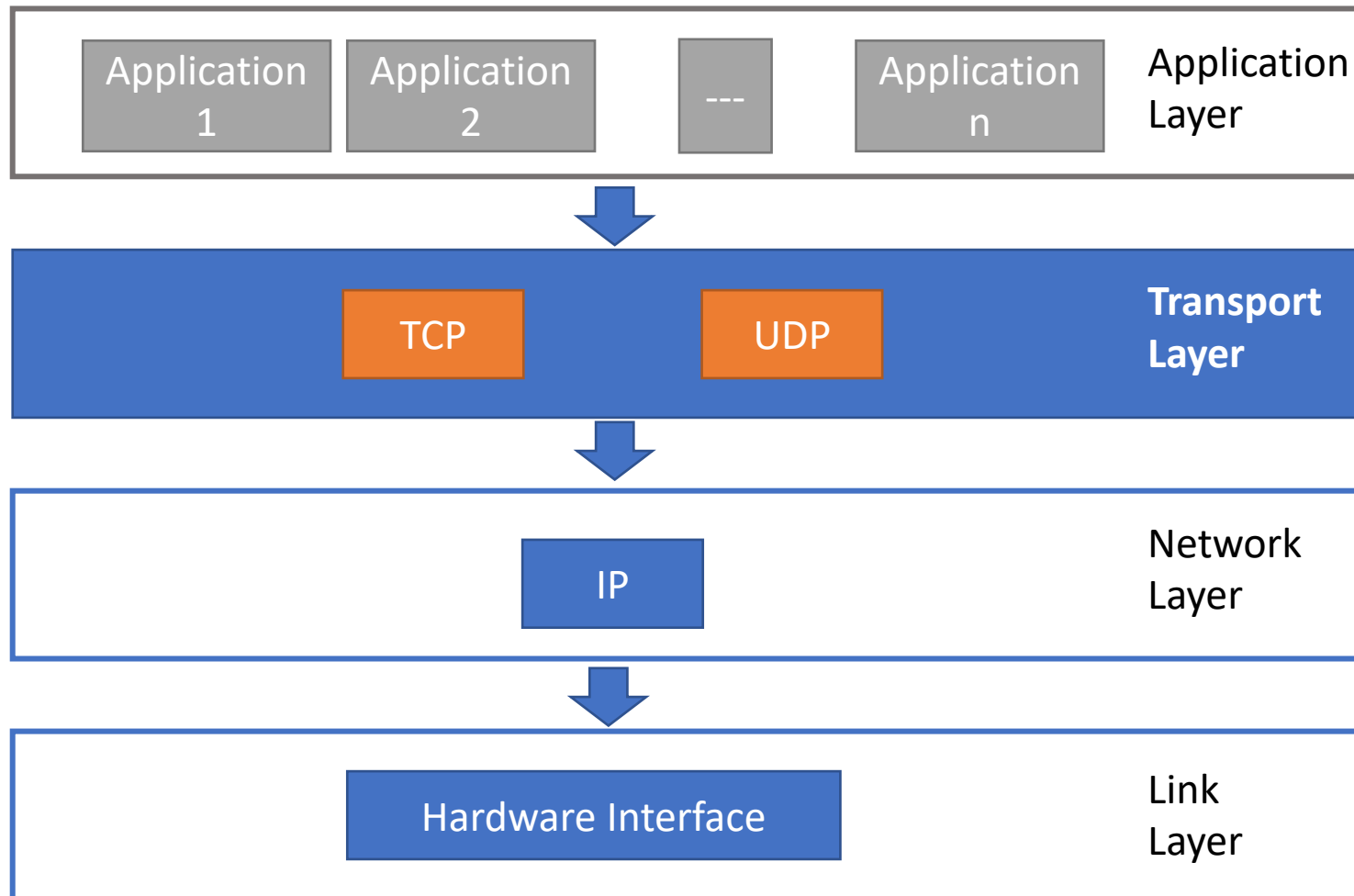
Transport Layer



Last Week

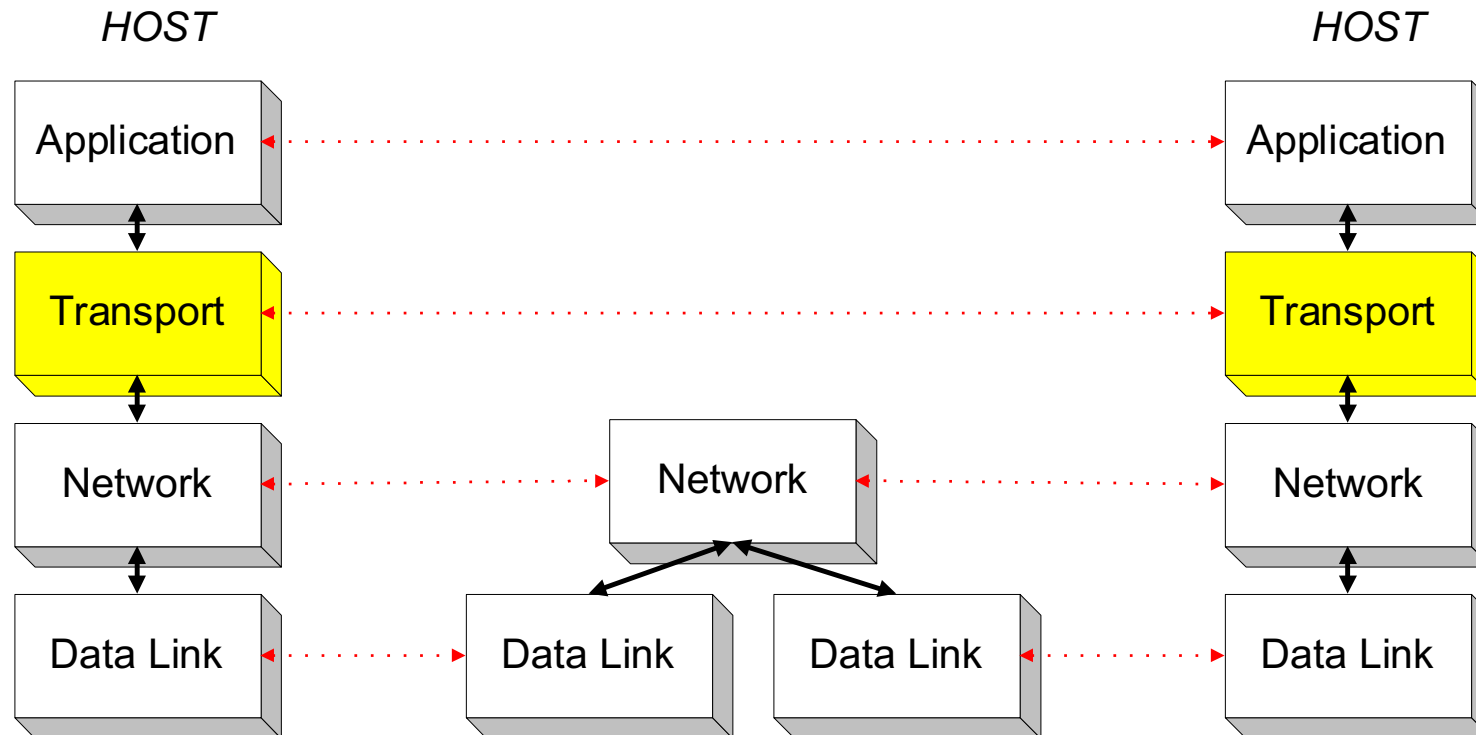
- IP Addressing:
 - an **end to end** addressing
- Two approaches for routing
 1. Static: hand-crafted
 2. Dynamic:
 - Vector Distance
 - Link State

Where do we stand?



Orientation

- Similarly to Network layer protocols, Transport layer protocols are also end-to-end protocols
- They are only implemented at the hosts



Today

- Port Addressing
- UDP: End-to-End Unreliable Transmission
- TCP: End-to-End Reliable Transmission
 - Connection Management
 - Packet Re-transmission
 - Window & Congestion Control
 - Tahoe
 - Reno

Port Addressing

- To address the application where the data is going to

Port Addressing

- To address the application where the data is going to.



Your Computer

Port Addressing

- To address the application where the data is going to.



Network interface

Port Addressing

- To address the application where the data is going to.



Software Transport Layer

Port Addressing

- To address the application where the data is going to.



You open an application that requires
network access!

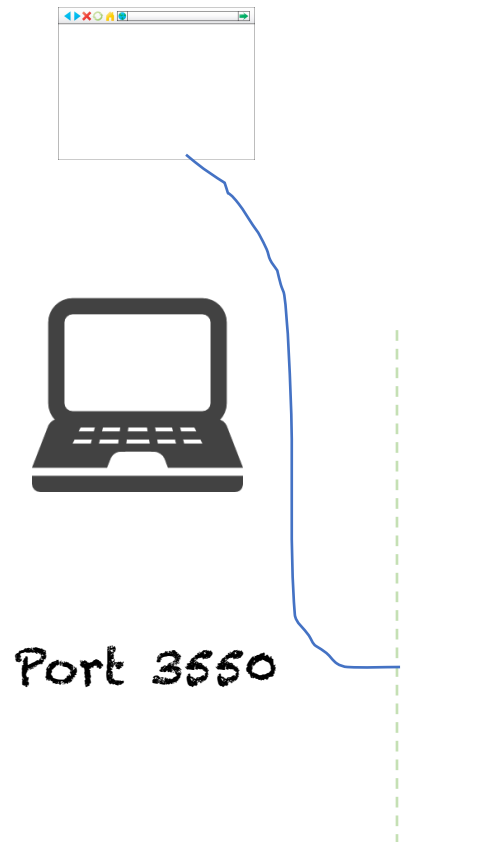
Port Addressing

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Port Addressing

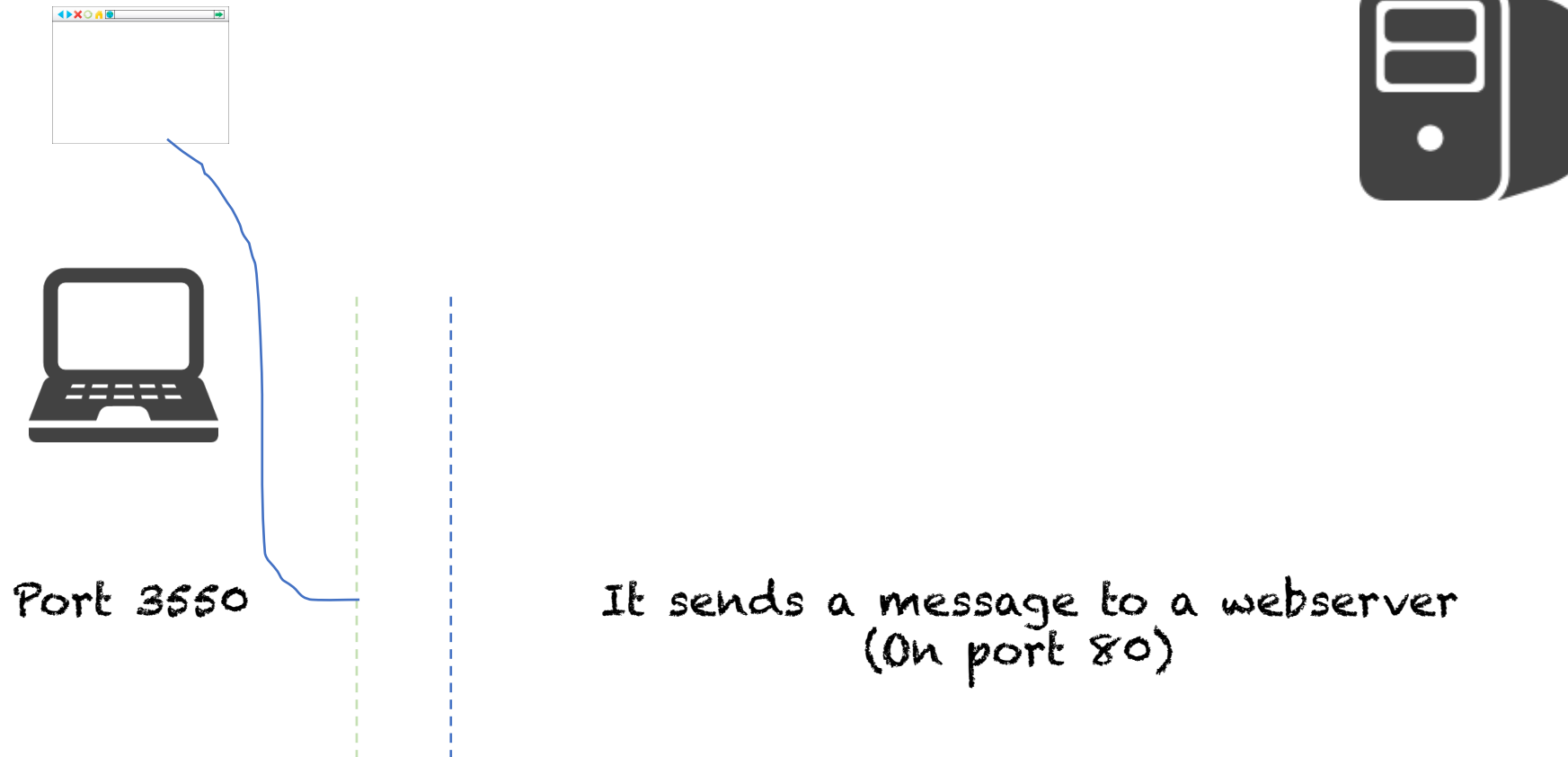
- To address the application where the data is going to.



It chooses a network port to use!
(programmed or user defined)

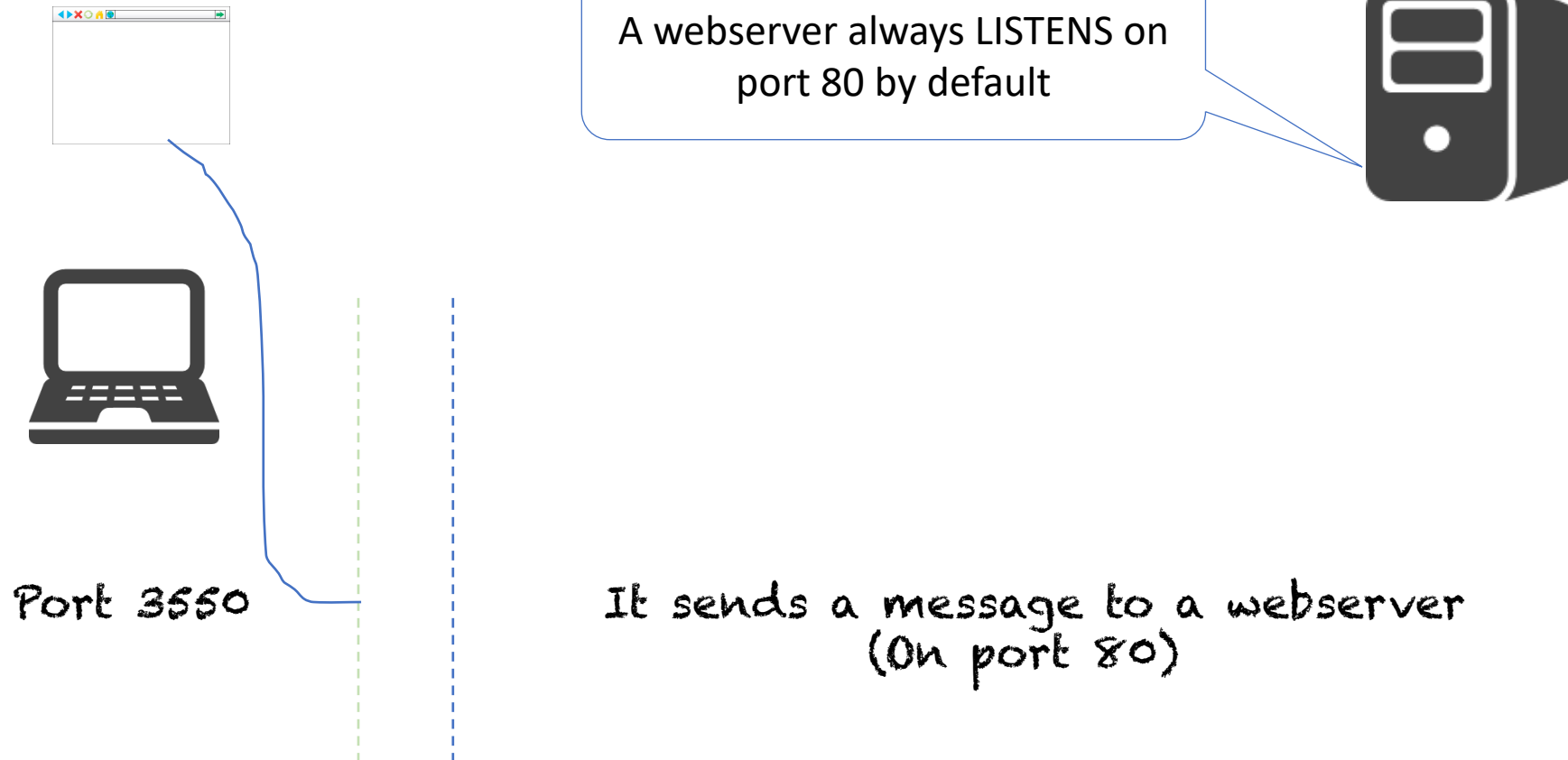
Port Addressing

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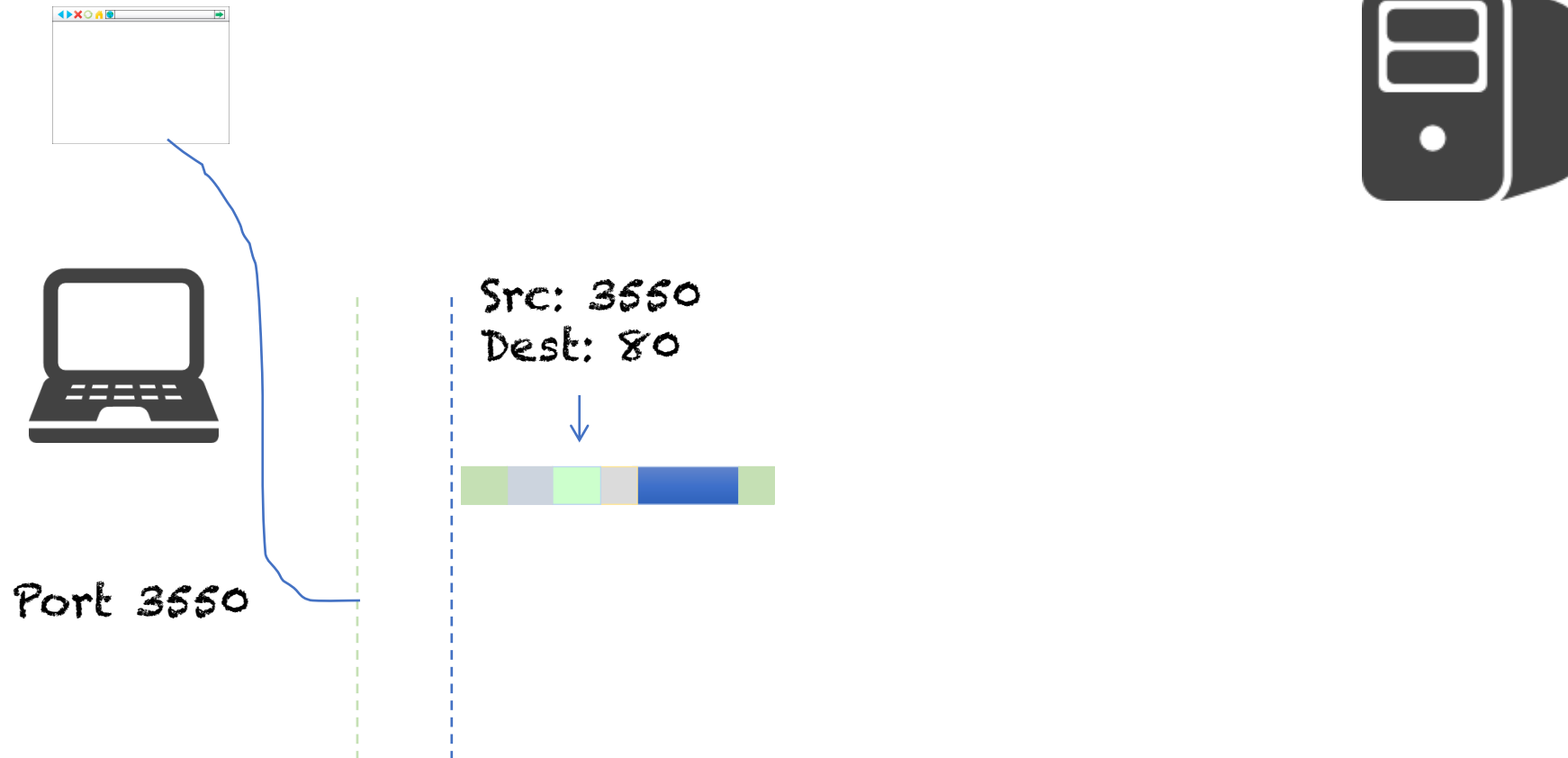
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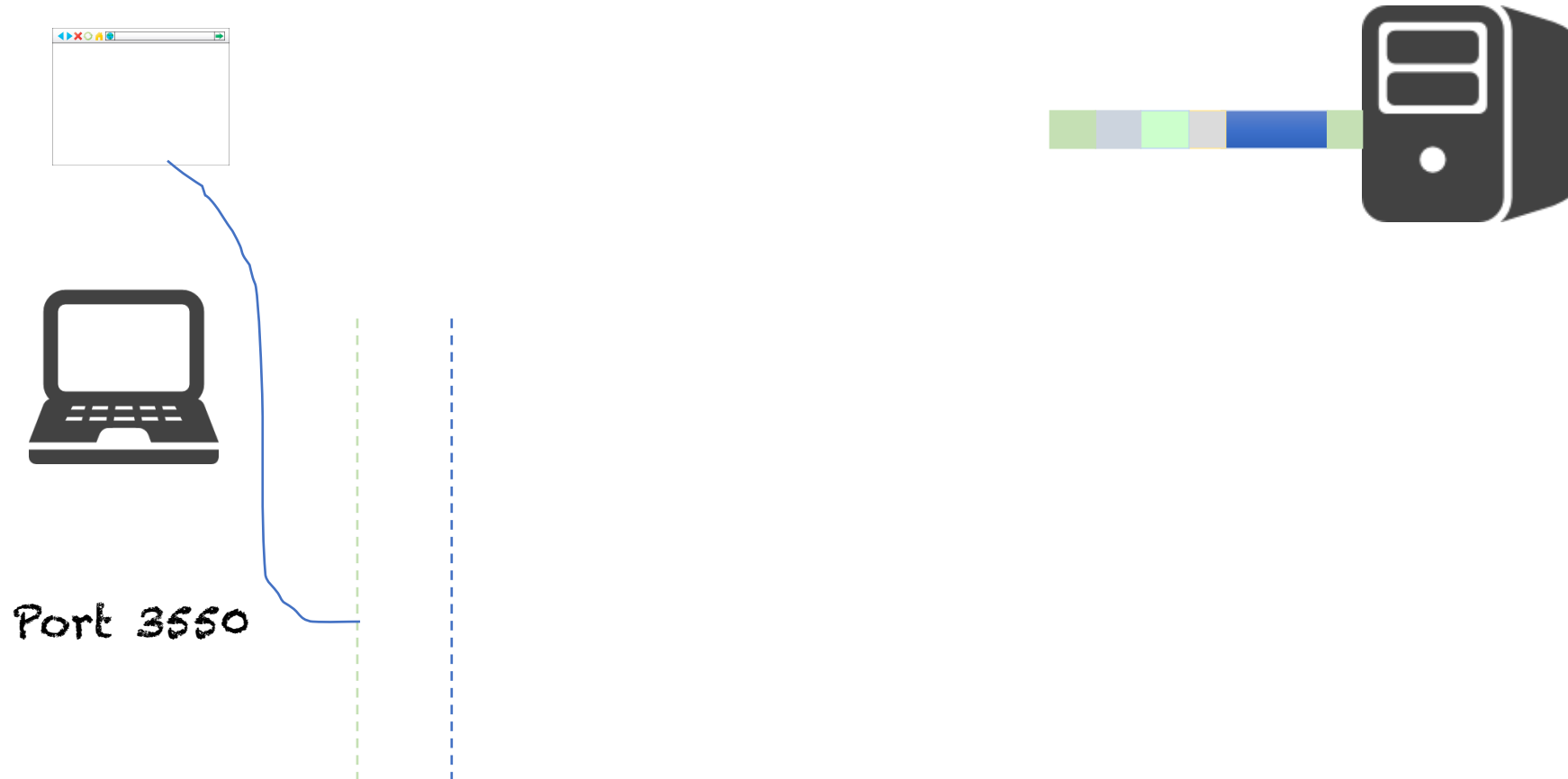
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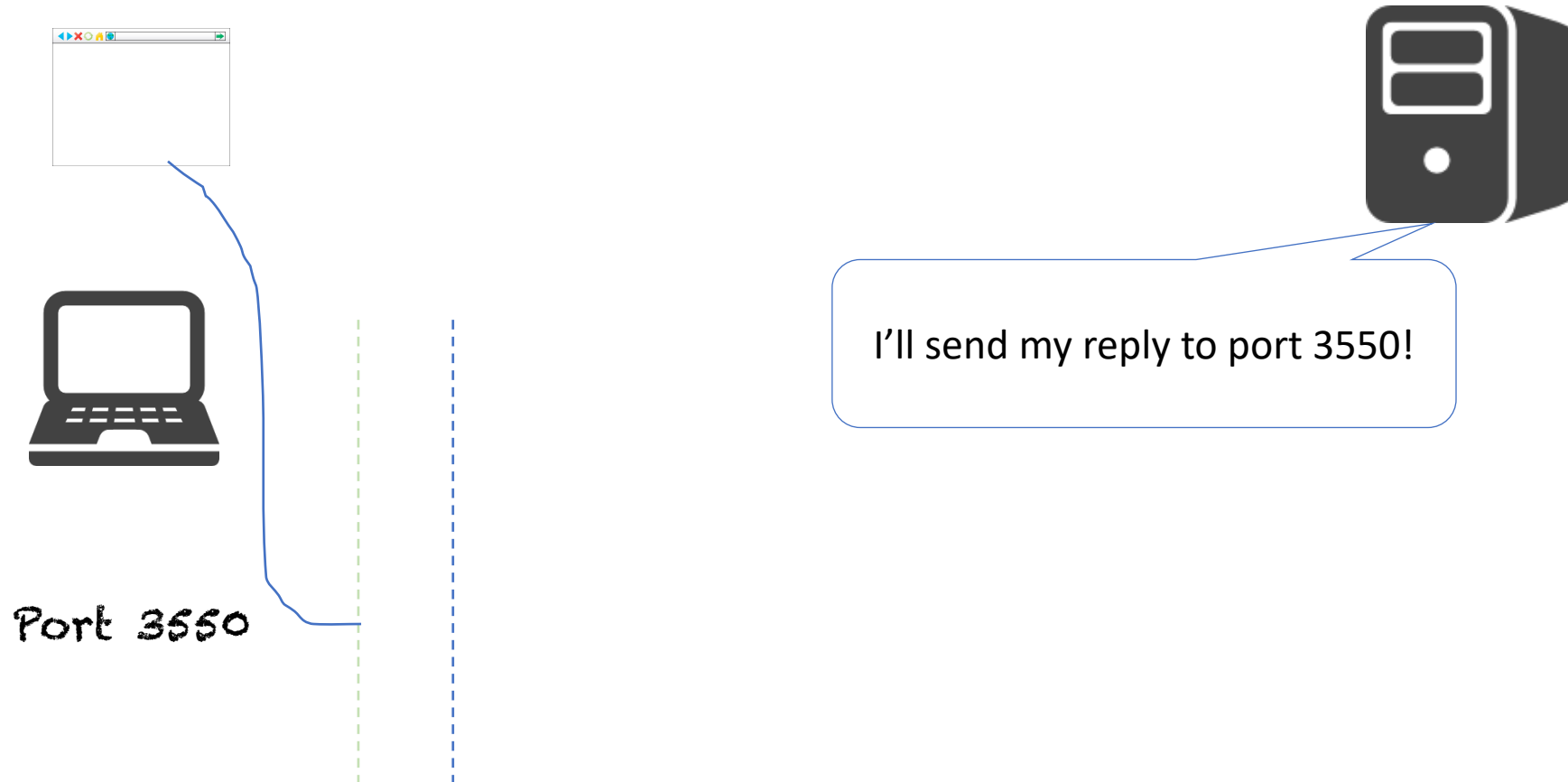
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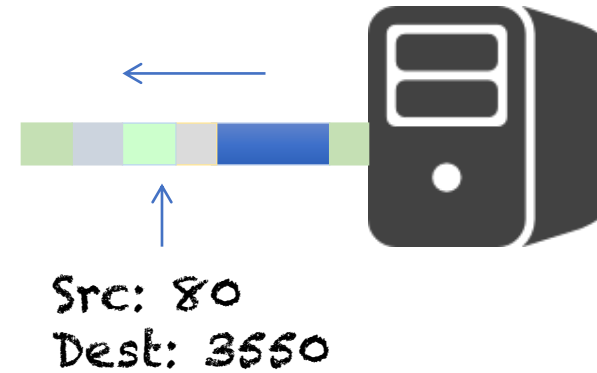
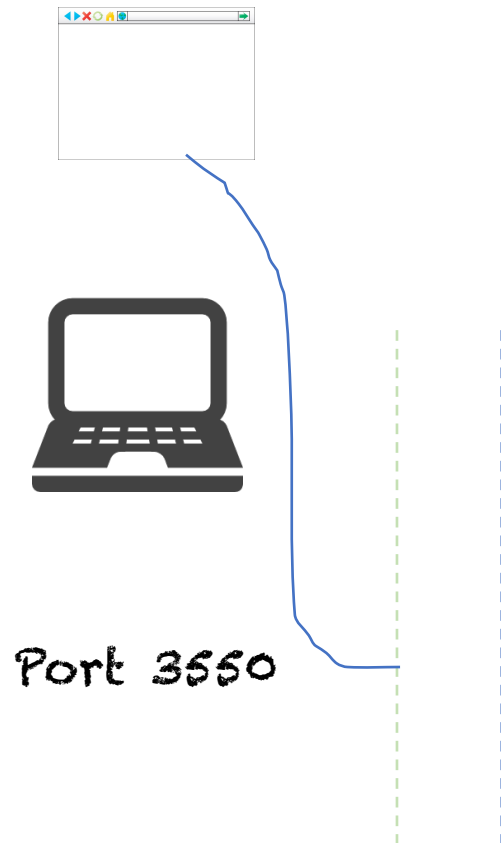
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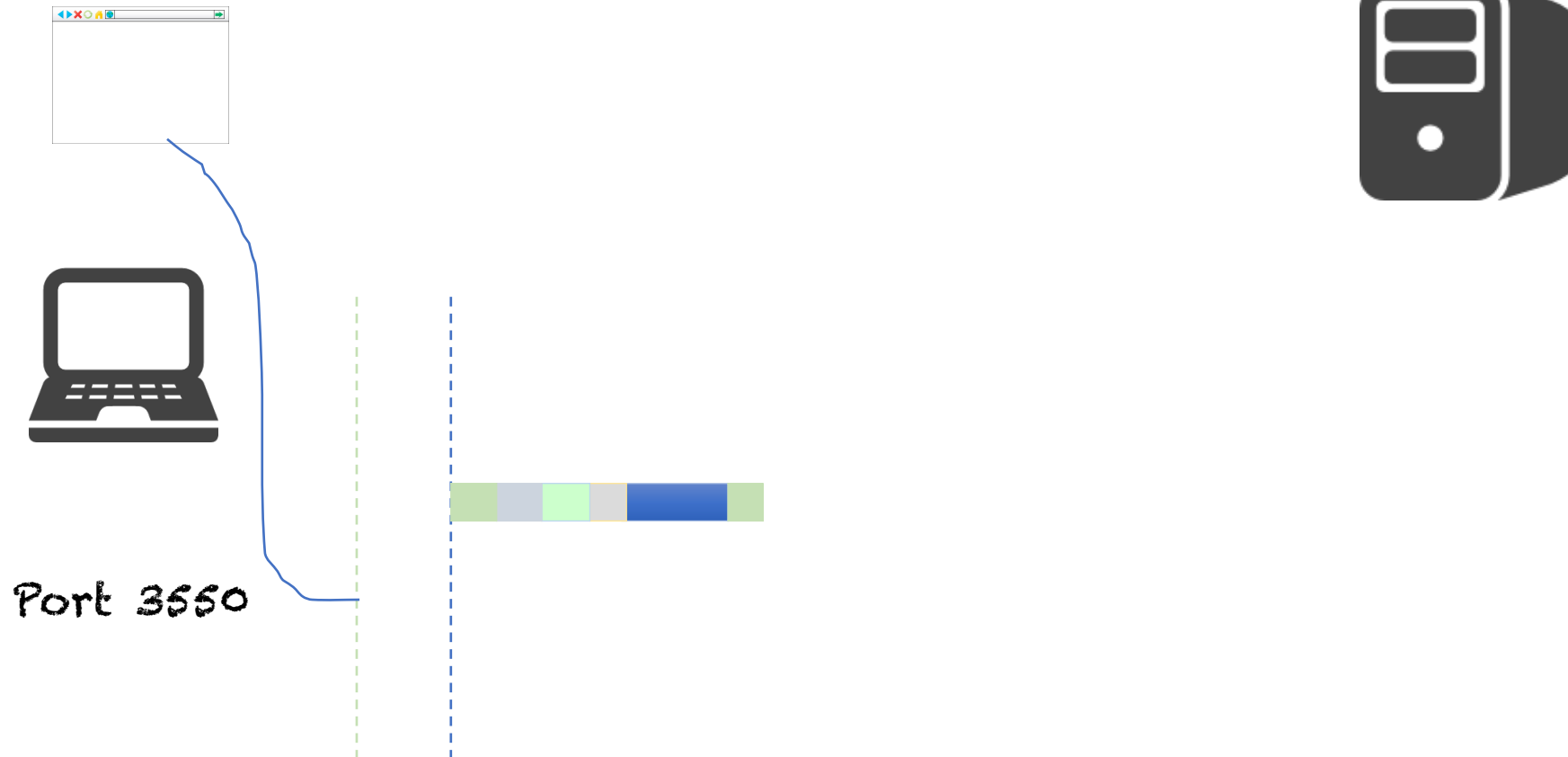
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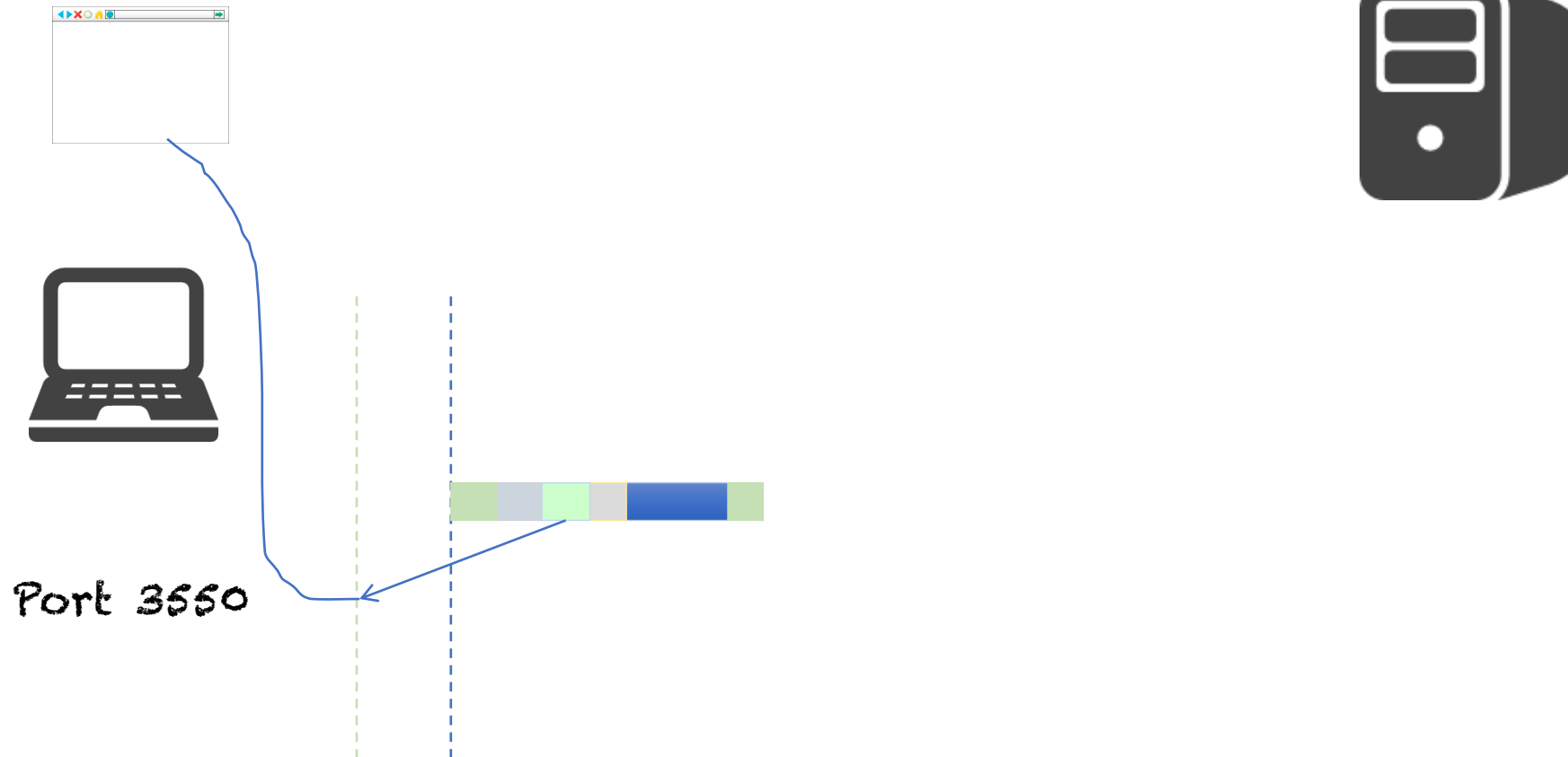
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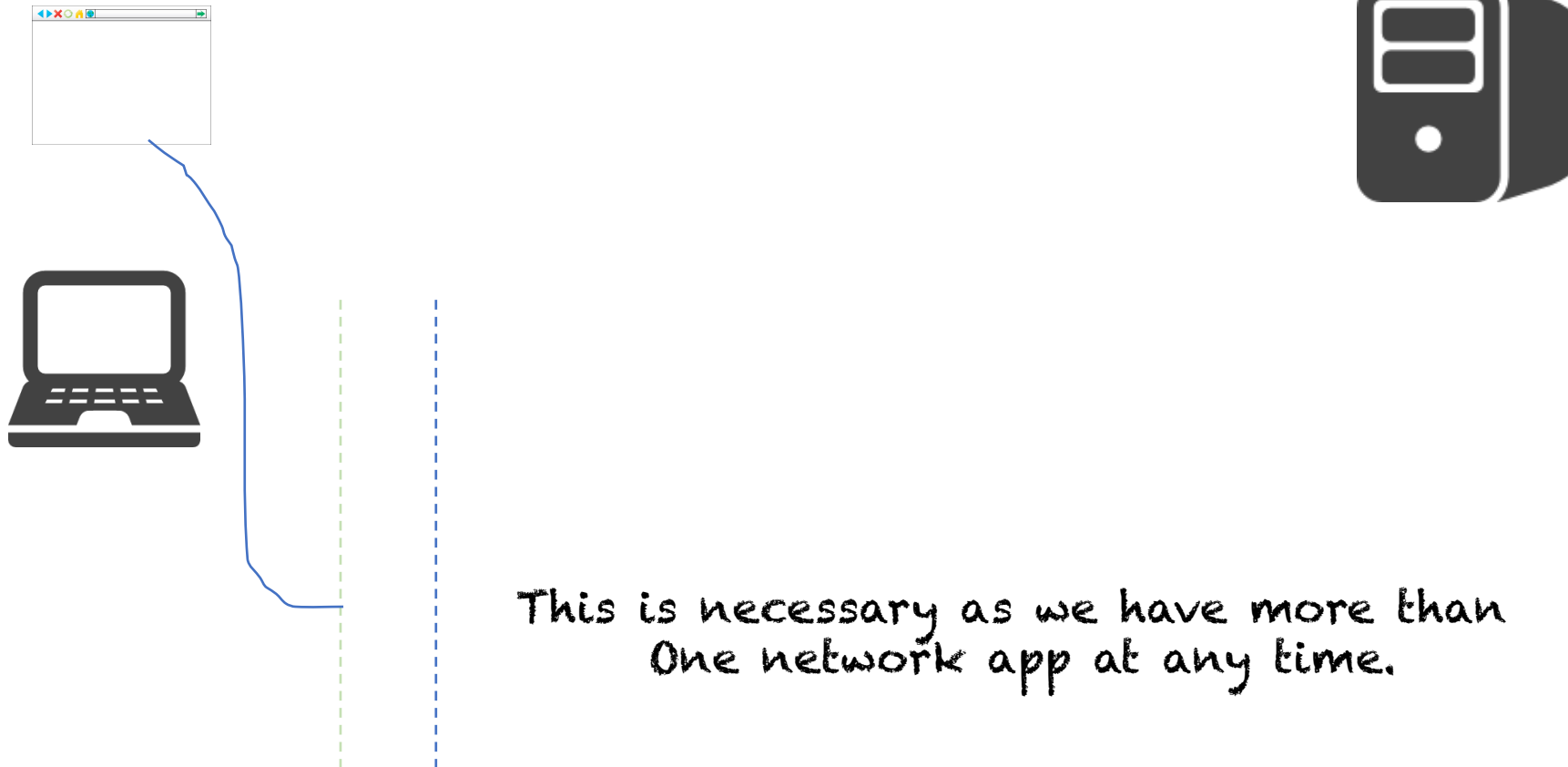
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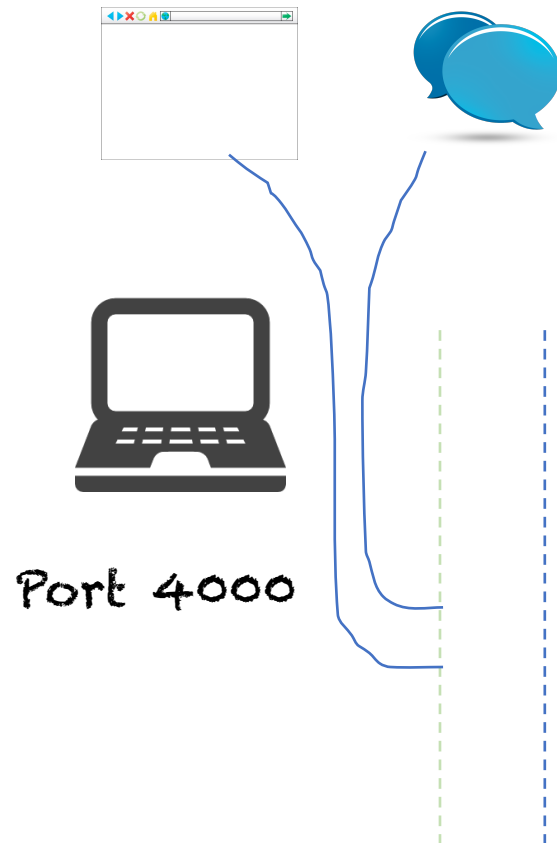
Port Addressing

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Port Addressing

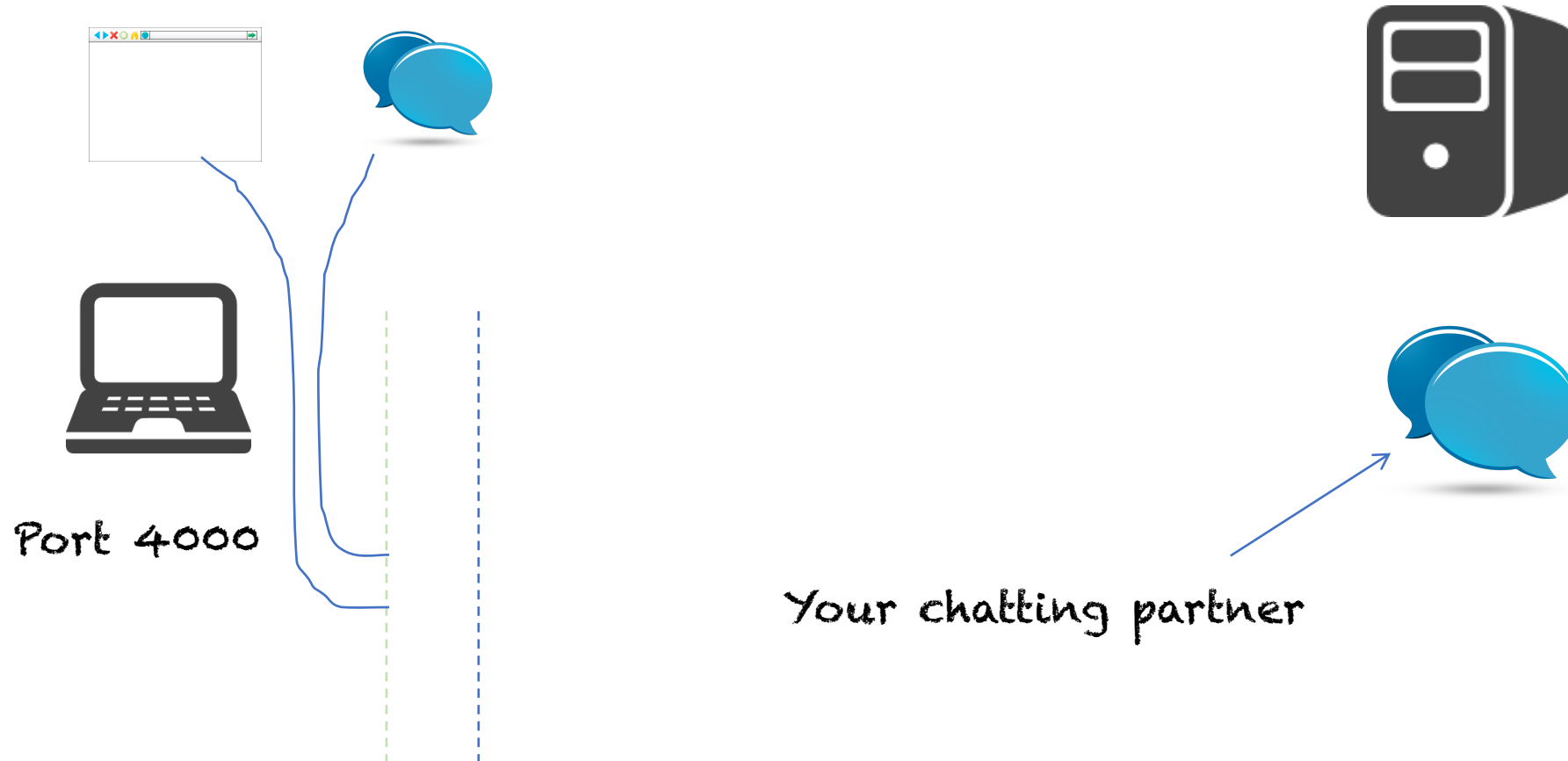
- To address the application where the data is going to.



A Chat client!

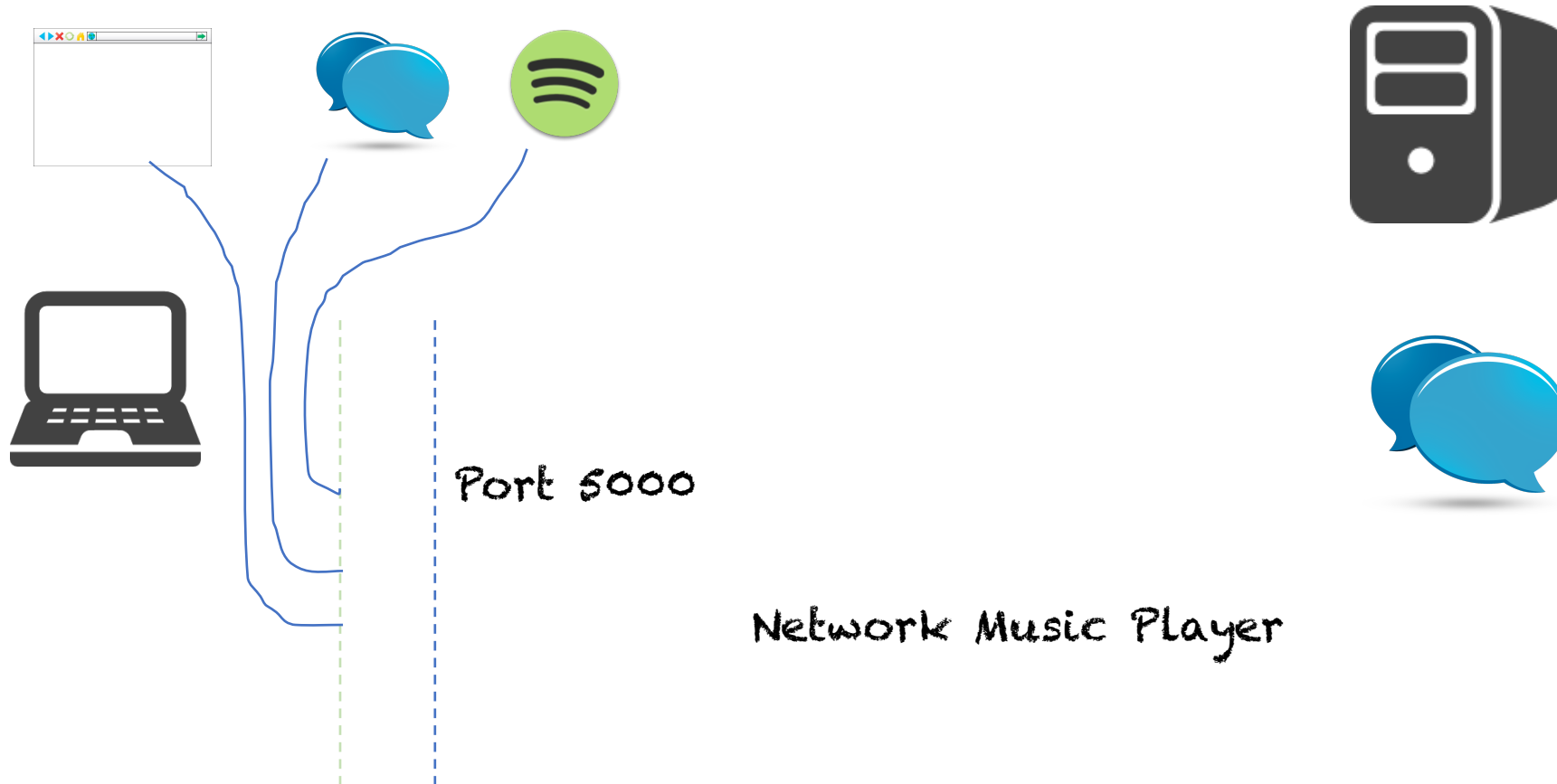
Port Addressing

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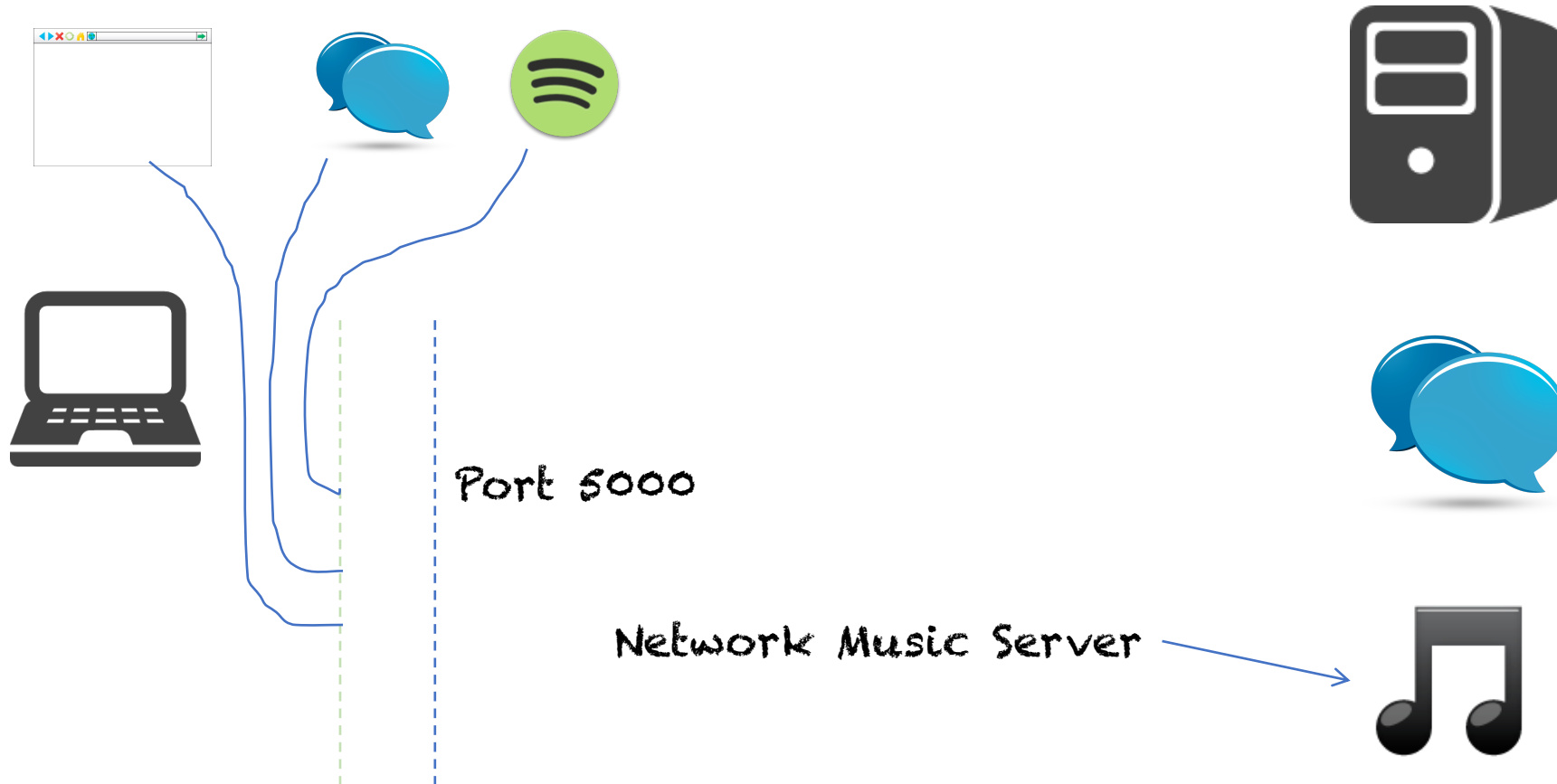
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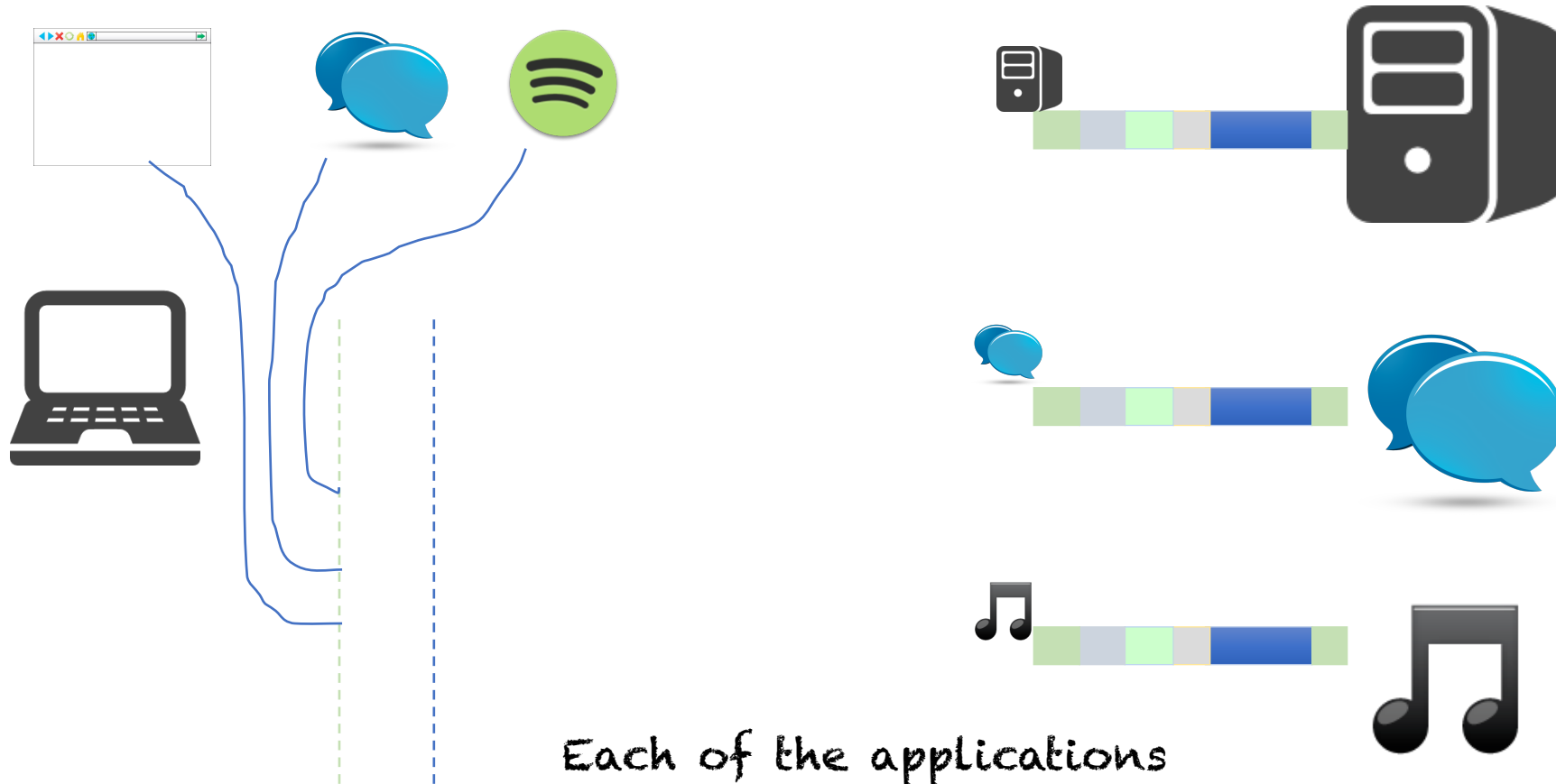
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Port Addressing

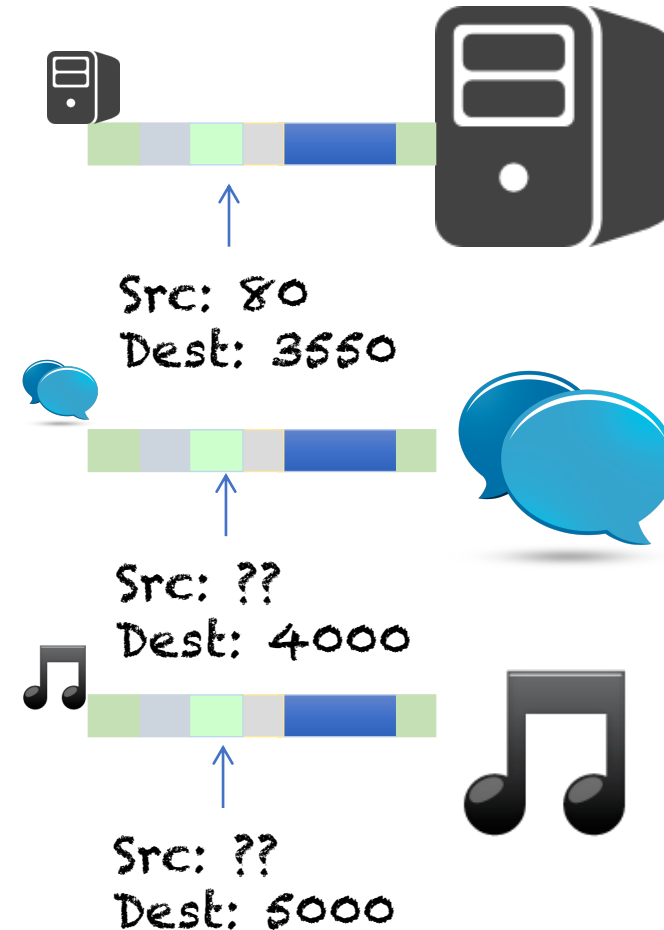
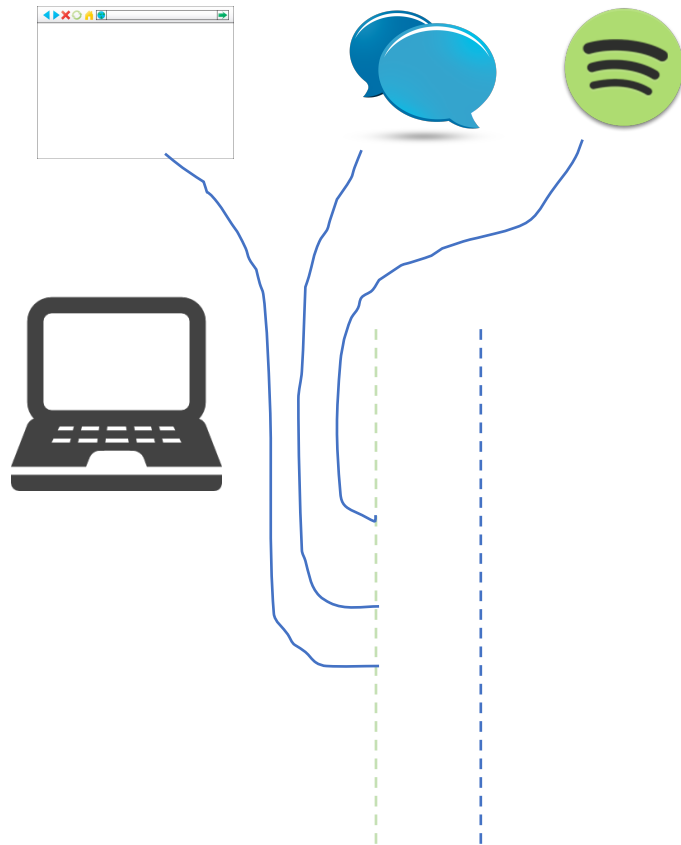
- To address the application where the data is going to.



Each of the applications
Get a message sent to them

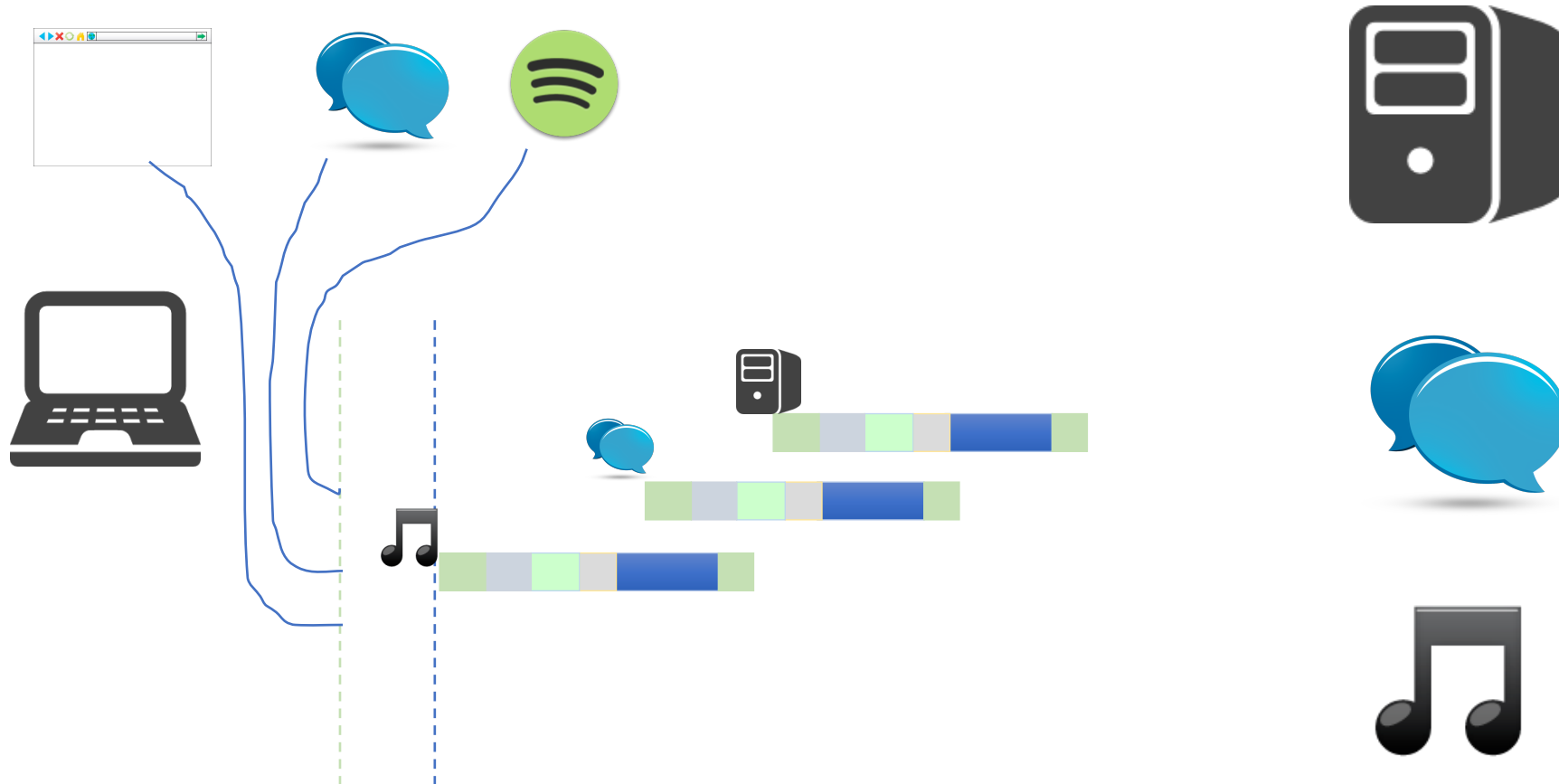
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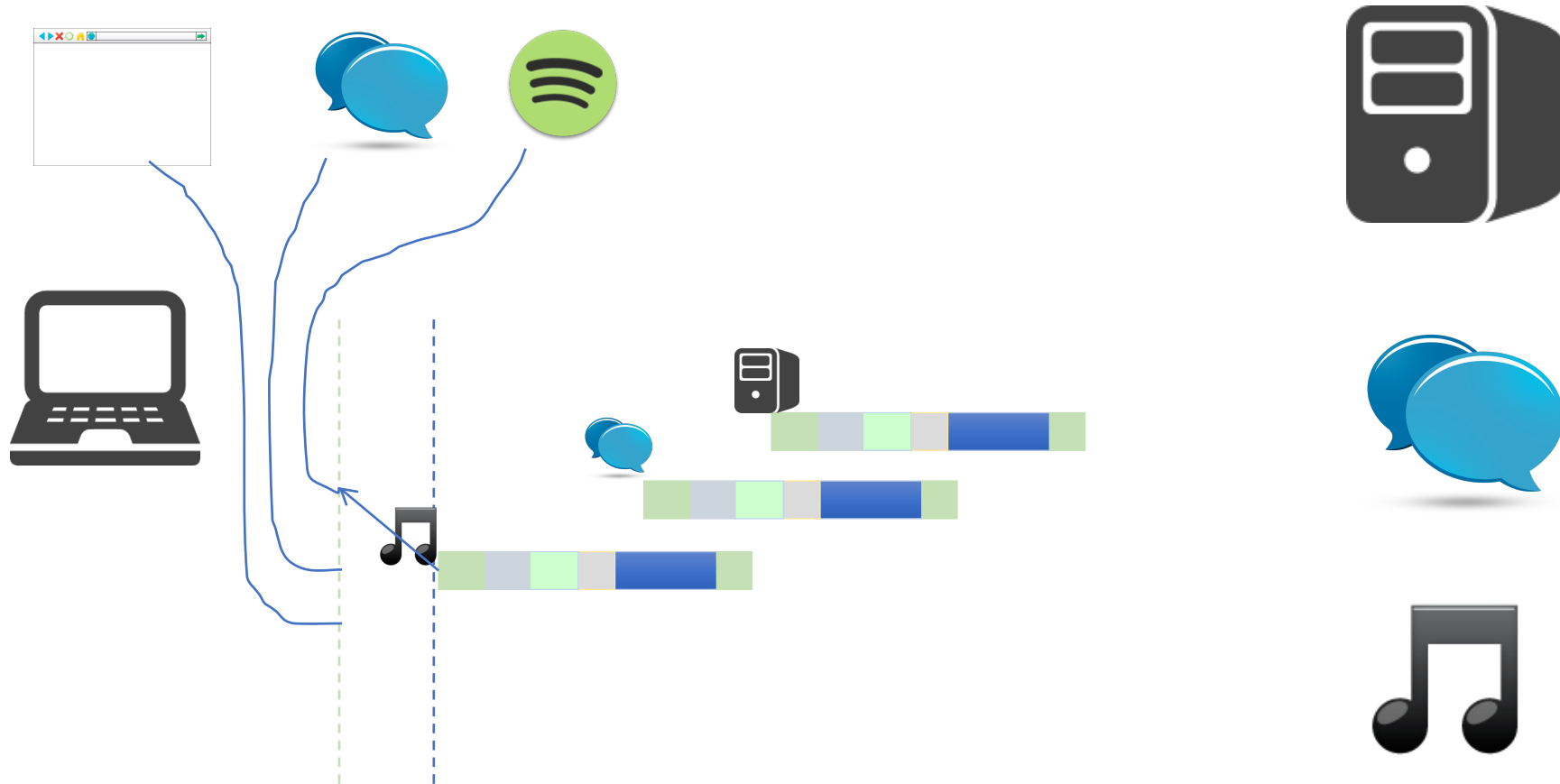
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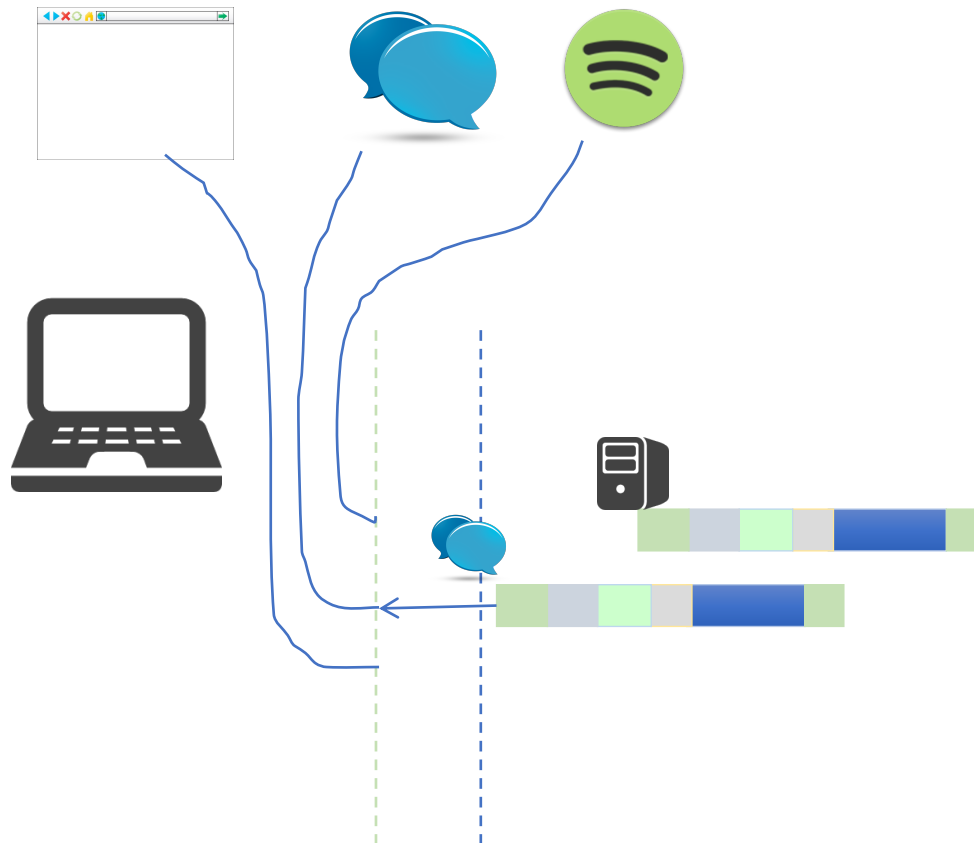
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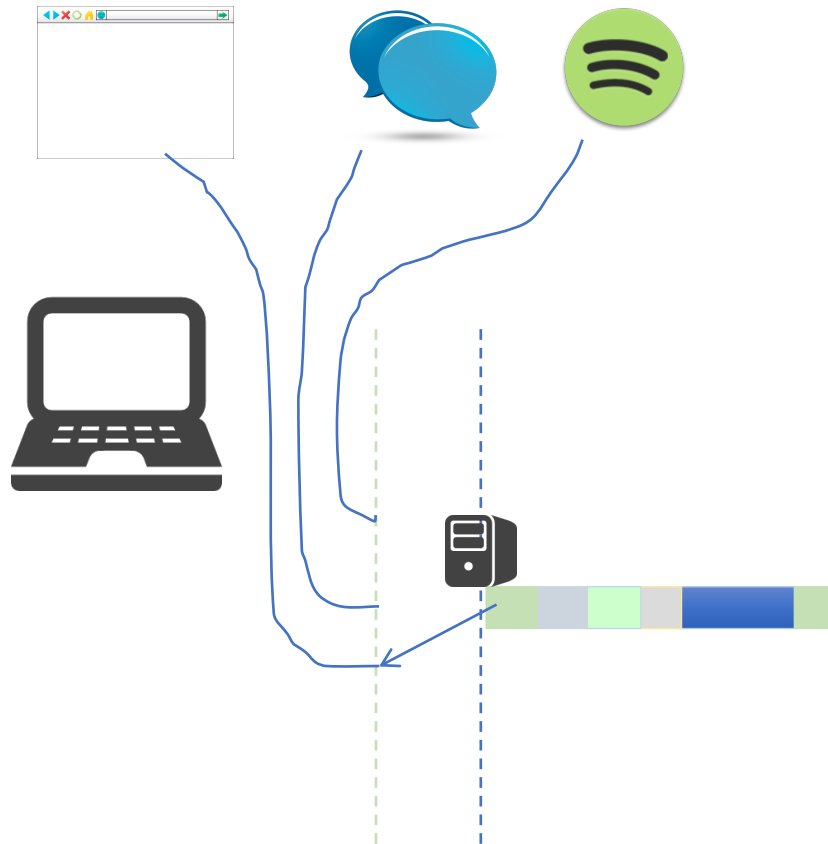
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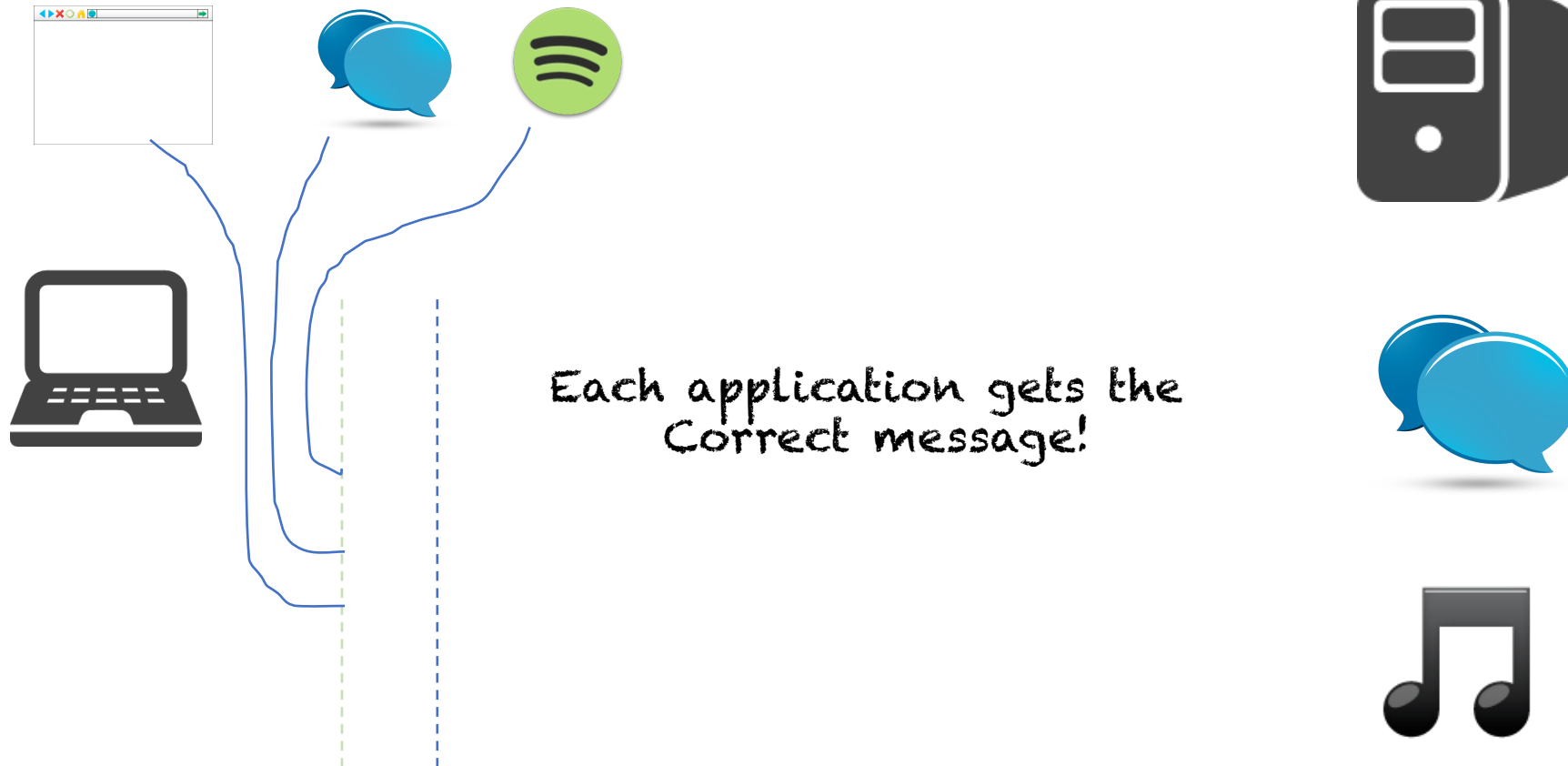
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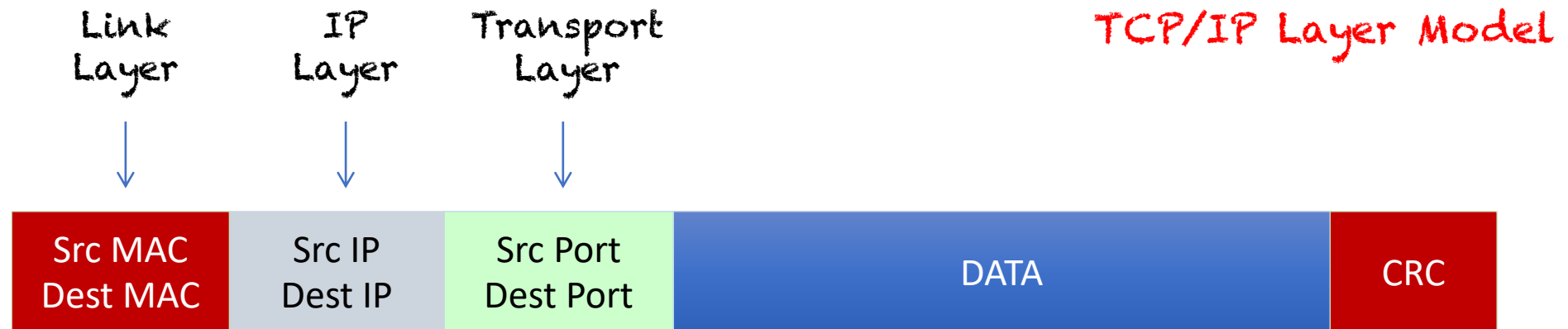
Port Addressing

- To address the application where the data is going to.



Addressing in headers

- The addresses are split over different layers



Transport Protocols in the Internet

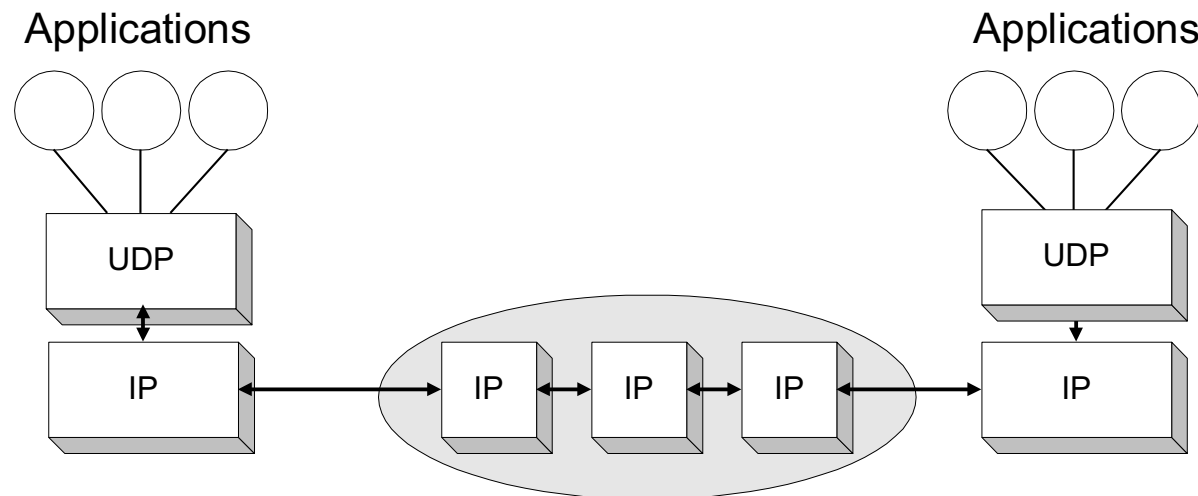
	UDP - User Datagram Protocol	TCP - Transmission Control Protocol
Characteristics	<ul style="list-style-type: none">• End-To-End Transport• Unreliable	<ul style="list-style-type: none">• End-To-End Transport• Reliable
Protocol Details	<ul style="list-style-type: none">• Simple• Connectionless• Datagram oriented	<ul style="list-style-type: none">• Complex• Connection-oriented• Stream oriented
Use cases	<ul style="list-style-type: none">• Unicast and Multicast• Useful only for few applications multimedia applications<ul style="list-style-type: none">• E.g., Live video streaming• Used a lot for network services<ul style="list-style-type: none">• E.g., Routing (RIP), Naming (DNS)	<ul style="list-style-type: none">• Unicast only• Used for most Internet applications:<ul style="list-style-type: none">• E.g., Web (HTTP), Email (SMTP), File Transfer (FTP)

Port Numbers

- UDP and TCP use a tuple **<IP address, port number>**
 - port numbers are used to identify applications
 - There are 65,535 ports per host
- There are many standard port numbers always used for particular applications (mostly)
 - 80 – HTTP service (web servers)
 - 443 – HTTPS (secure HTTP)
 - 25 – SMTP (Simple Mail Transfer Protocol)
 - 3306 – MySQL Database
- Many other port numbers are free, and allow applications to set their own port numbers

UDP - User Datagram Protocol

- UDP supports **unreliable end-to-end** transmissions of datagrams
- UDP merely extends the host-to-host delivery service of IP datagram to an application-to-application service



UDP Format



8 Bytes

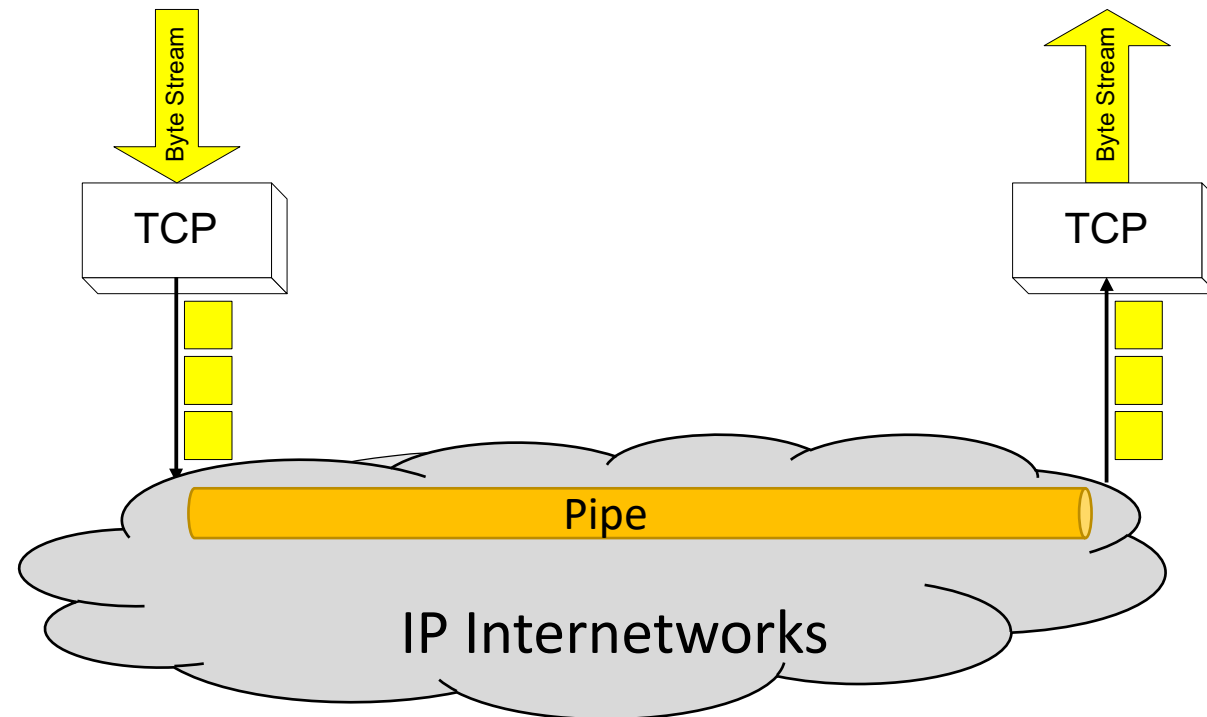
Source Port Number	Destination Port Number
Message Length	Checksum
0	15
	31

- **Port numbers** identify sending and receiving applications (processes).
Maximum port number is $2^{16}-1= 65,535$
- **Message Length** is at least 8 bytes (I.e., Data field can be empty) and at most 65,535
- **Checksum** is for header (of UDP and some of the IP header fields)

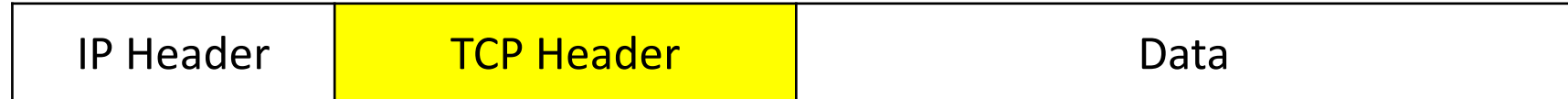
TCP Overview

TCP = Transmission Control Protocol

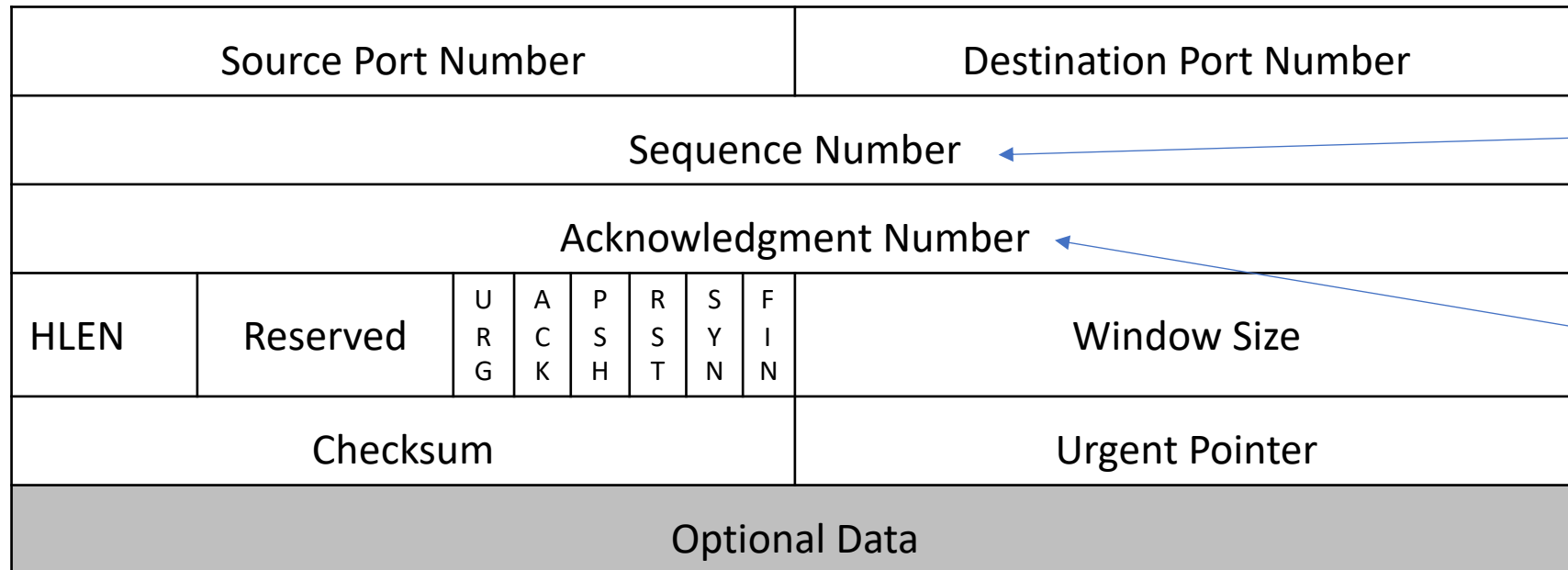
- Connection-oriented protocol
- Provides a reliable, end-to-end byte stream over an unreliable internetwork



TCP Format



20--60 Bytes



The number of my packet

The number of packet I expect to receive

0

15

31

Challenges for Reliable Transfer

- Over a perfectly reliable channel
 - All the data arrives in order, just as it was sent
 - Simple: Sender sends data, and Receiver receives data
- Over a channel with bit errors
 - All the data arrives in order, but some bits corrupted
 - We need **checksums** so that the Receiver detects errors and asks for a retransmission
- Over a lossy and congested channels
 - Some data are missing, and others arrive with delay
 - Receiver needs to detect a loss or a disorder in packets
 - We need **sequence numbers** to detect missing data and put data back in order

TCP Uses 3 Mechanisms to Achieve a Reliable Transfer

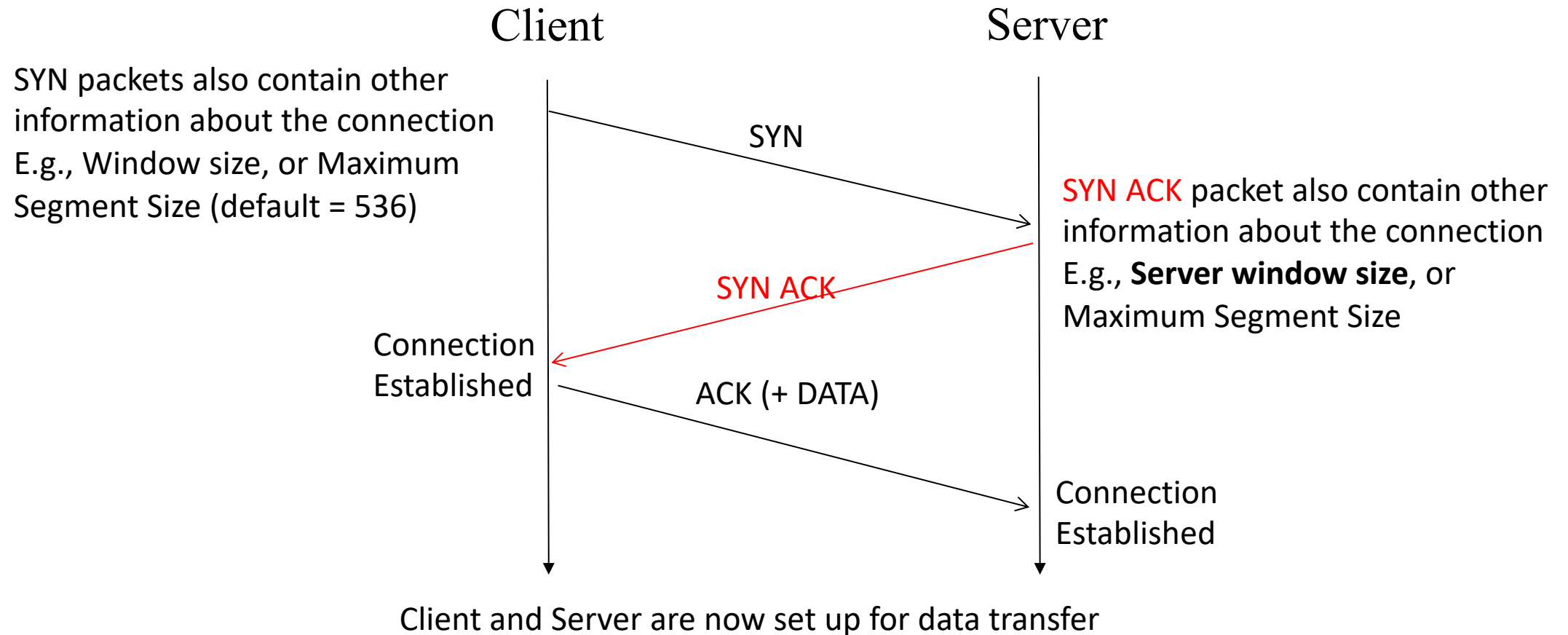
1. Connection: establish and release a connection
2. Re-transmission: retransmit lost or corrupted data
3. Window and Congestion Control: manage the amount of sent data for an efficient resource usage

Set Up/Close a Connection

- TCP must first set up a connection with the end point before sending information
 - Make sure the endpoint is accessible/available
 - Make sure the end point has the facilities to handle the data
- TCP must close the connection after finishing the transmission
- TCP uses:
 - 3-Way Handshake to set-up the connection
 - 4-Way Handwave to close the connection

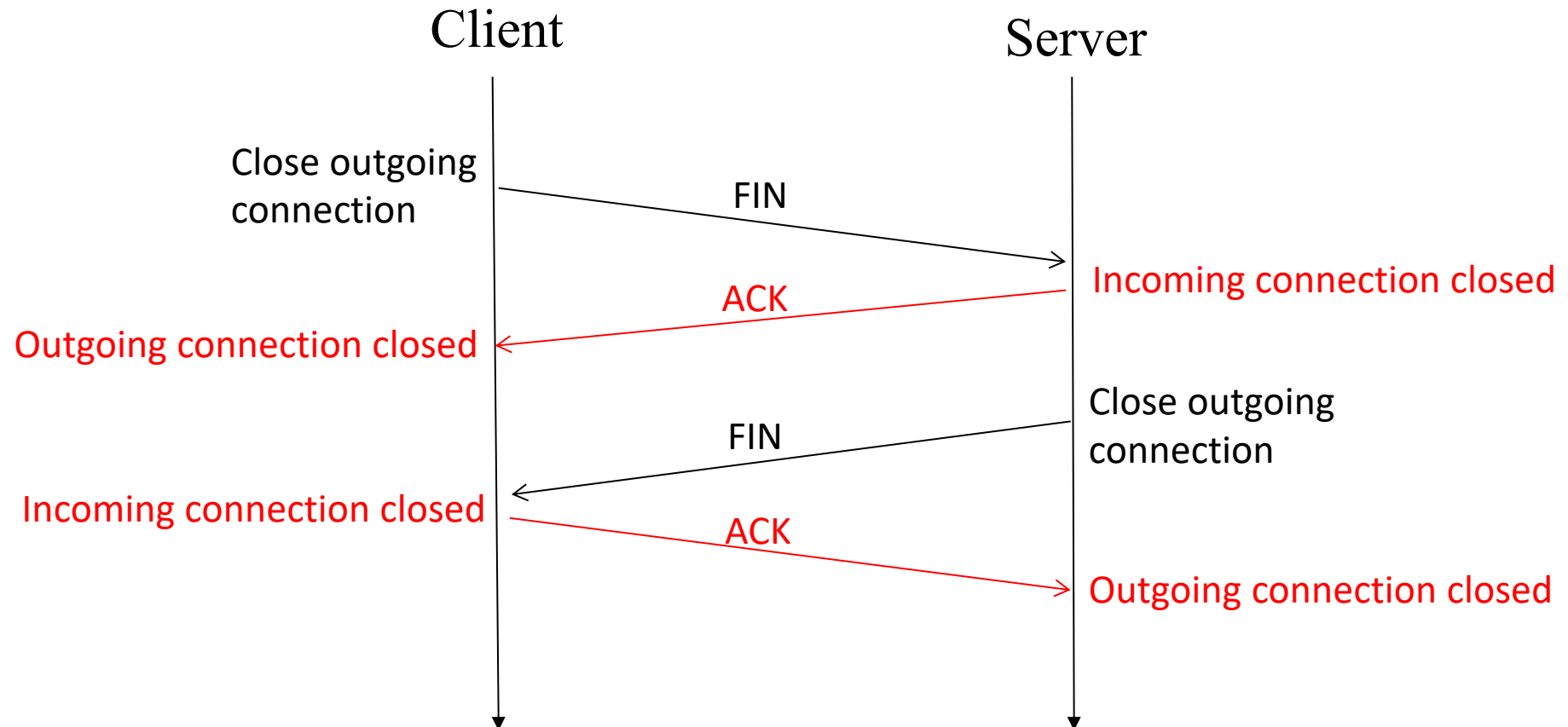
Set Up a Connection: 3-Way Handshake

- We have a Client/Server relationship



Close a Connection: 4-Way Handwave

- Each of the Client and Server have to close both their outgoing and ingoing connections

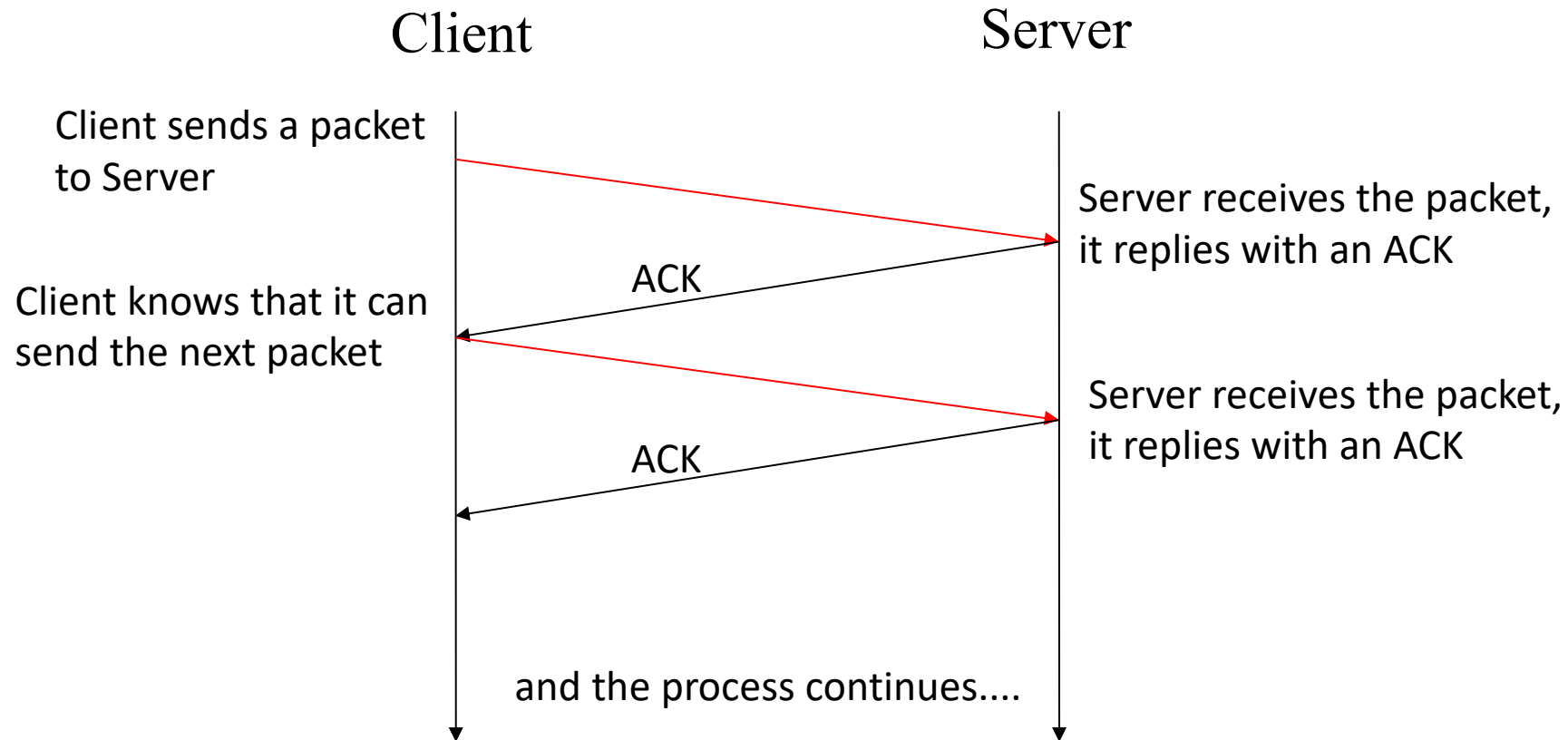


Sequence Numbers

- Once a connection has been established, each subsequent packet contains a sequence number
- This sequence number is used to reorder data in the correct order at the end point. The sequence number is the byte count of the transmission
- It can also be used to determine if packets have been lost over the network
- Received packets are acknowledged with the next expected byte

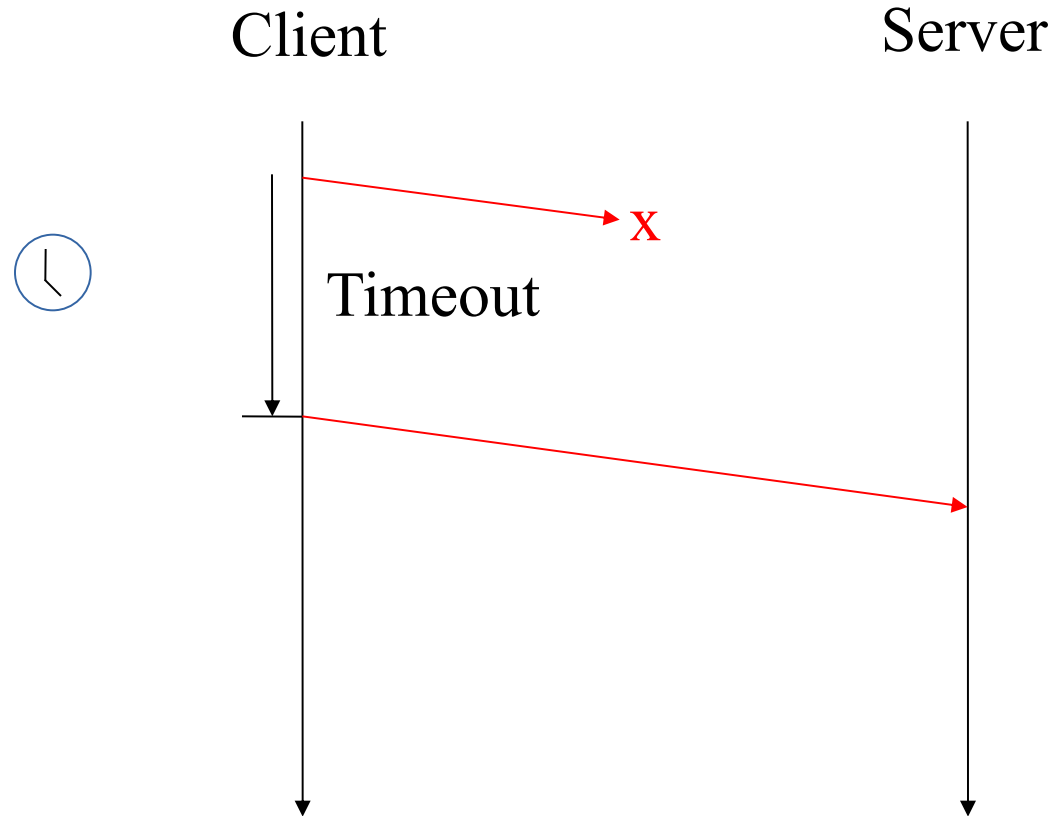
Acknowledging packets

- Acknowledged packets is an important concept for TCP
- It confirms to the Client that his sent data have been received



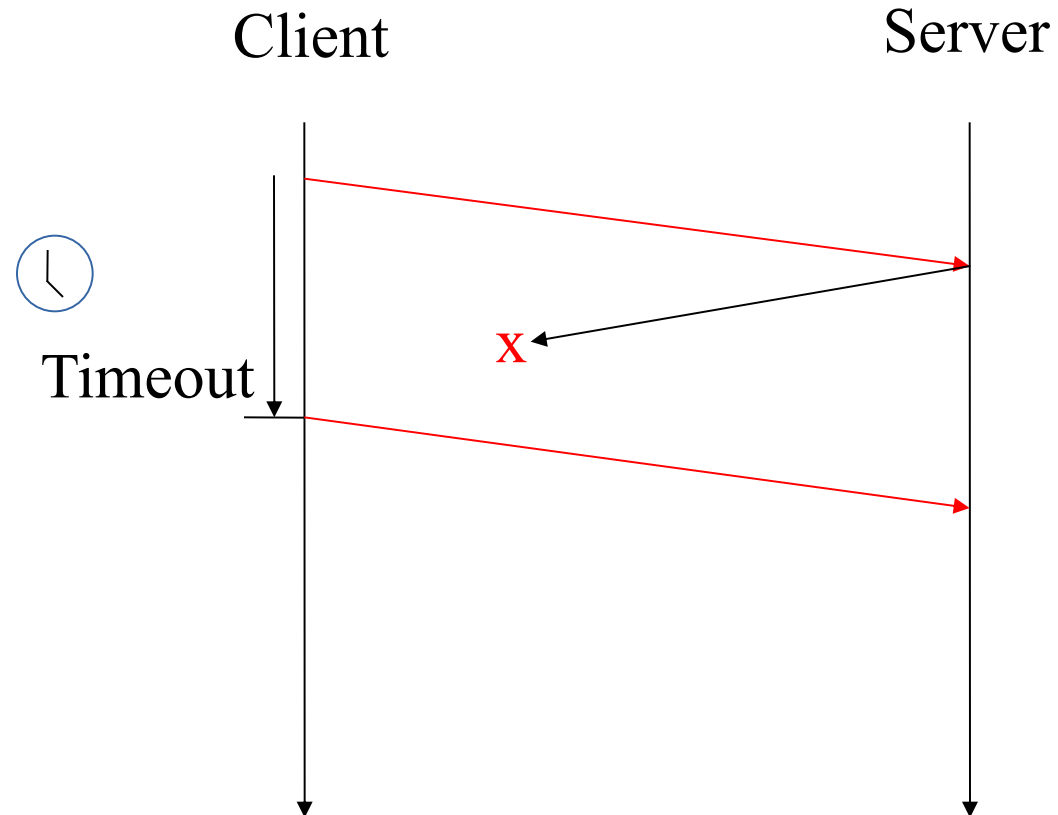
Packet Timeout

- What if the Packet is not received by the Server.
- Client sends a packet, but it is lost



Packet Timeout

- What if the Packet is not received by the Server.
- Server sends an ACK, but the ACK is lost



Acknowledge Schemes

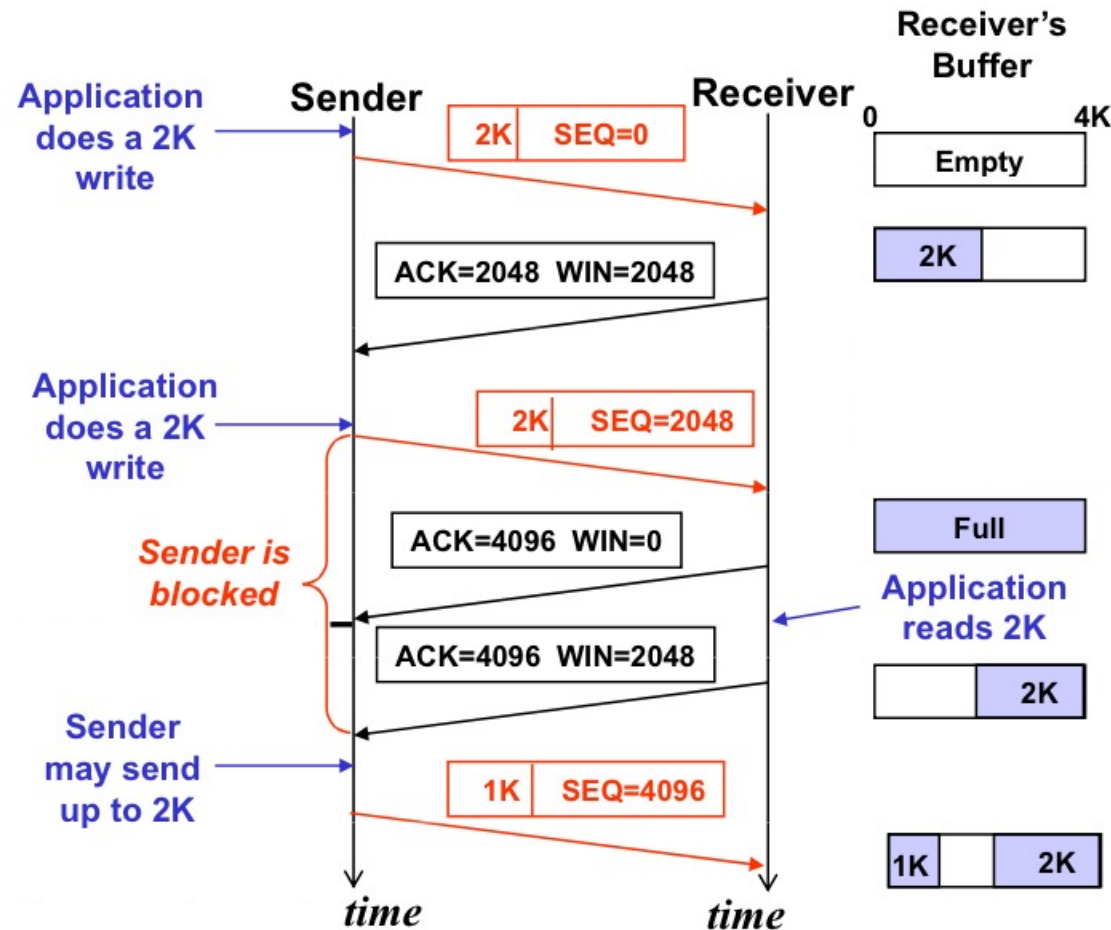
- Stop and Wait
 - Time consuming, does not use the full size of the pipe
- Sliding Window
 - Uses the pipe size (i.e., the minimum bandwidth over the path) to determine how many packets can be sent at a time
 - This is different from Layer 2 where window is fixed
 - Layer 4 windowing starts small then expands to fill the space in the pipe

Why does TCP do this?

TCP Buffering and Windowing

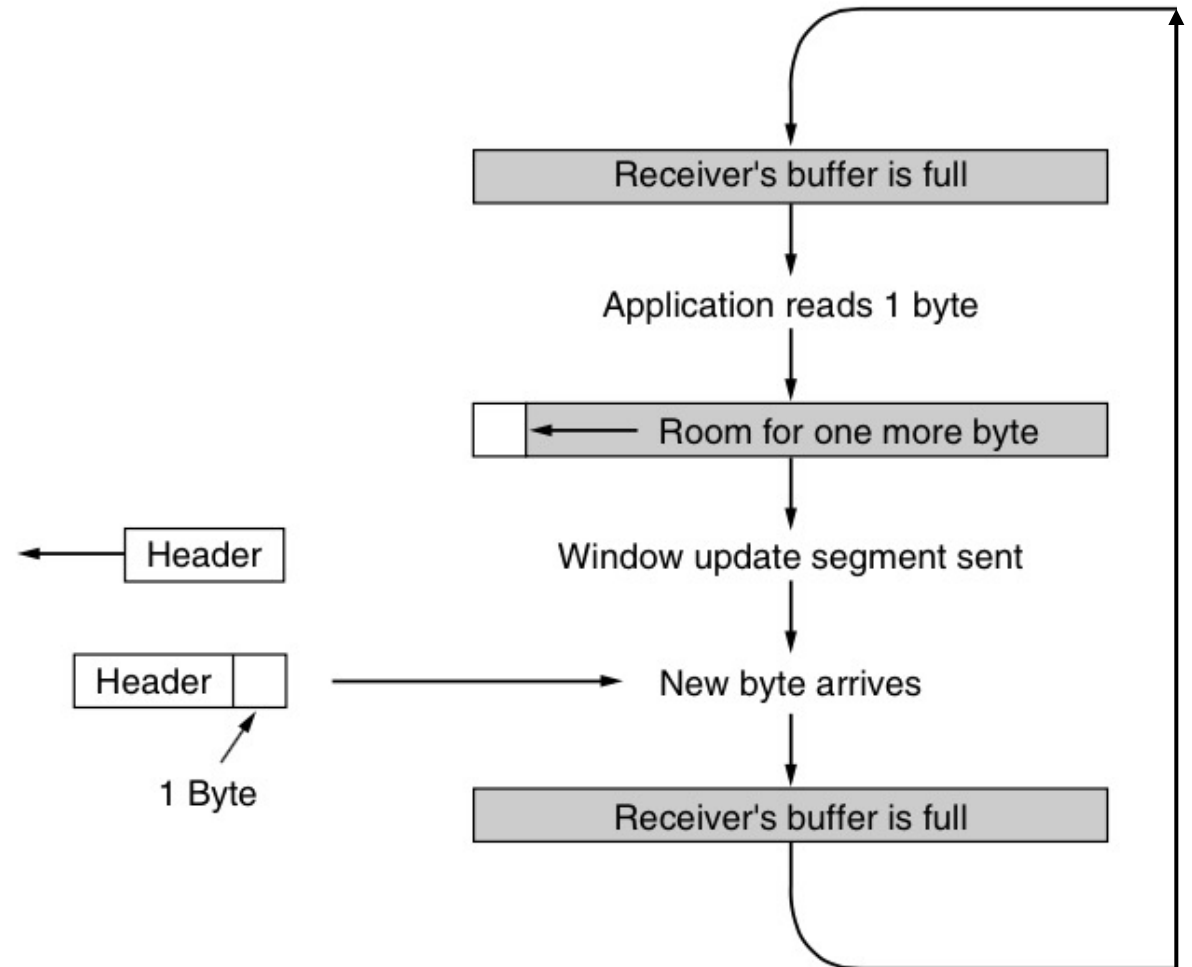
- Let's assume that an application sends large amounts of data between hosts at once
 - What if one host is not fast enough at handling the data?
 - What if there is a bottleneck at one point in the network?
- The data sent through the network is dropped as the host/network may not have enough resources to deal with the data
- This is a waste of network resources
- Solution: TCP uses **Windowing** and **Congestion control**

TCP Buffering and Windowing



TCP Buffering and Windowing

- Imagine if a receiver sends a window update every time there is space for one byte in the buffer
 - Silly Window!
- Solution: receiver only sends a window update if:
 - buffer is half empty,
 - or has more space than maximum-size segment

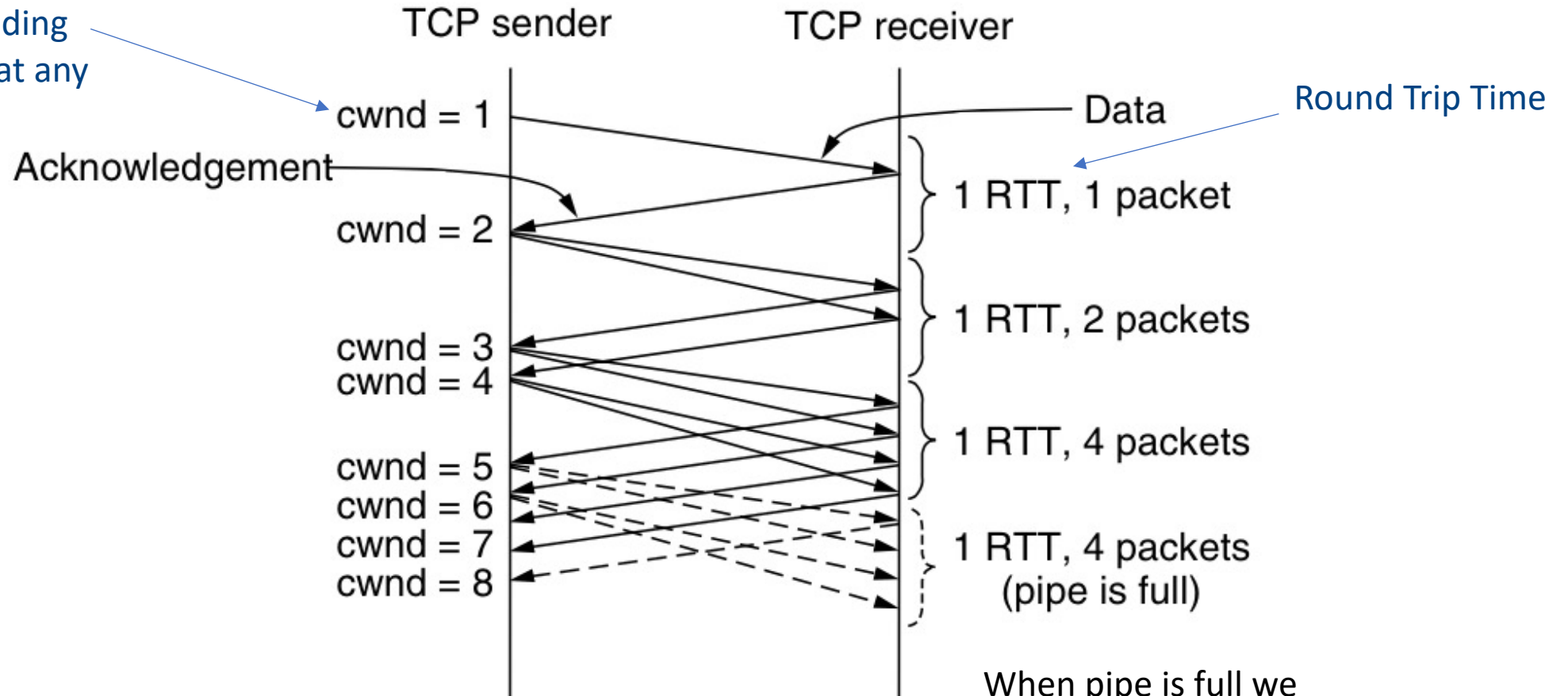


TCP Congestion

- Sending more packets than the network or receiver can handle
- We make an assumption in TCP
 - Timeout = packet loss due to congestion
 - Packets rarely lost through other means
- To avoid congestion, a sender will think:
 - If I receive an ACK for a packet, there is enough space in the network for me to send another!
 - Using ACK's to pace the transmission of new packets
- Sending one packet per acknowledgement may underutilize the network however
- TCP implements some schemes to reduce congestion, while using the network capacity

TCP Congestion - Slow Start

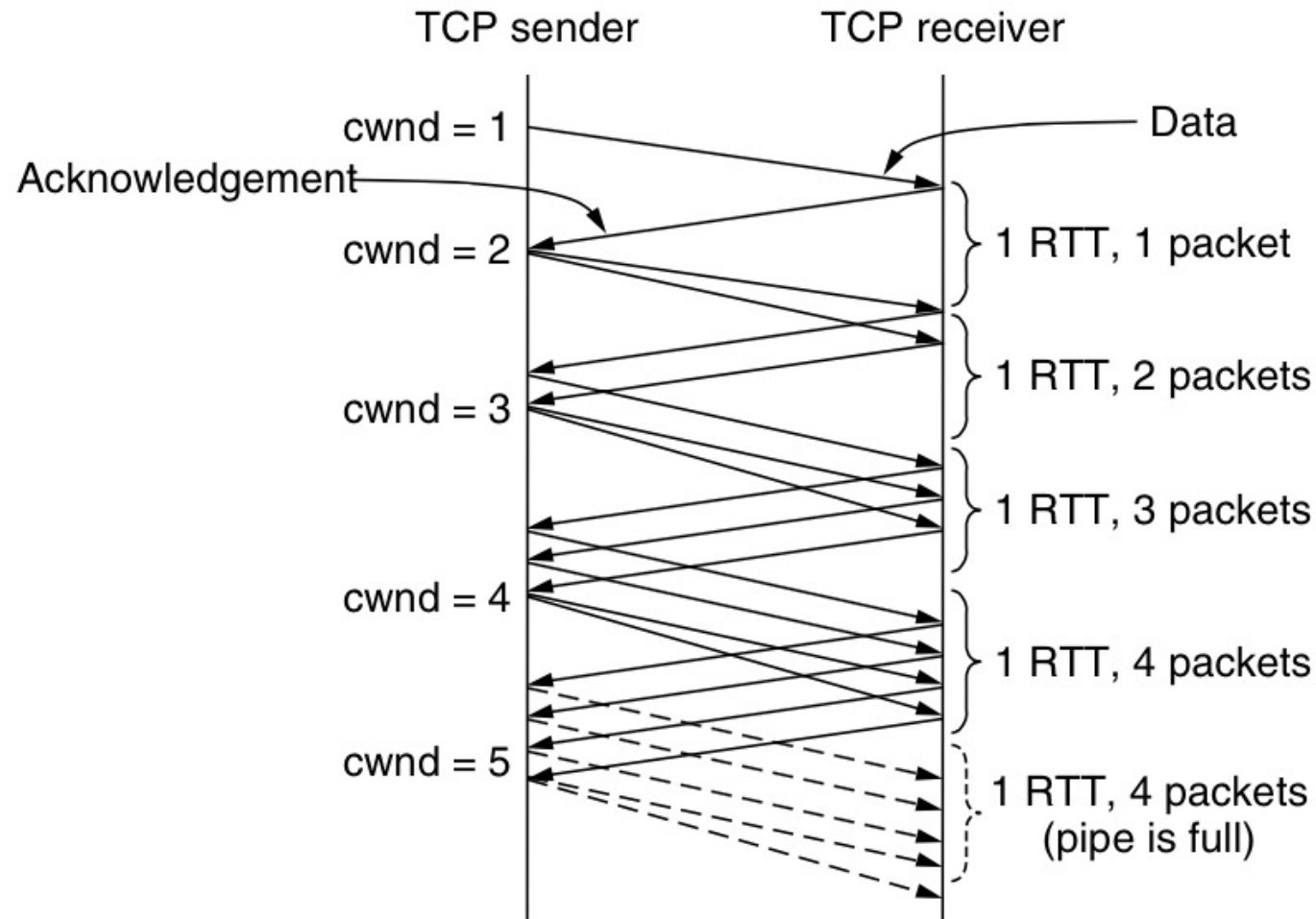
congestion window size:
Number of outstanding
segments allowed at any
given time



Exponential growth of packets

When pipe is full we
will get a timeout!

TCP Congestion – Additive Increase



TCP Congestion

- **Slow Start:** Using Slow Start algorithm
- **SS Threshold:** Slow Start threshold
- **Congestion Avoidance:** Using Additive increase algorithm
- **Multiplicative Decrease:** Reducing the window when congestion is noticed
- **Fast Retransmit:** Using repeat ACKs to determine if a packet has been lost (don't wait for Timeout, send lost packet!)
- **Fast Recovery:** on packet loss set the cwnd to $\frac{1}{2}$ the size and continue with additive increase

TCP Congestion

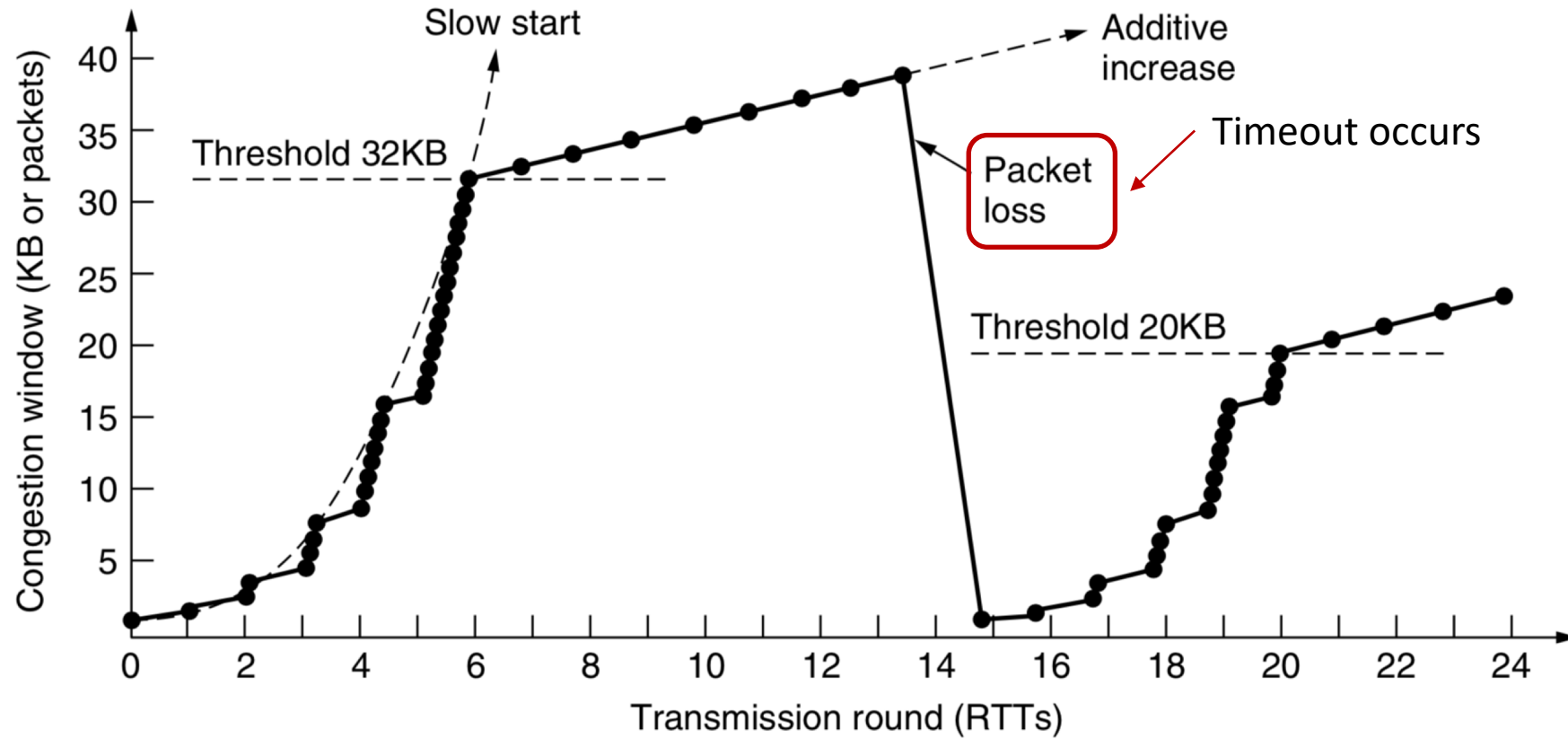
Tahoe:

- Start With Slow Start until:
 - SS Threshold is met
 - Go to Congestion Avoidance
 - Packet lost (3 resubmitted Acks)
 - Go to packet loss!
- Start Congestion Avoidance until:
 - Packet loss (3 resubmitted Acks)
 - Go to packet loss!
- On Packet Loss:
 - Fast retransmit
 - SStreshold = $\frac{1}{2}$ CWND
 - CWND = initial state
 - Go to Slow Start
- Time Out:
 - CWND = 1
 - Go to Slow Start

Reno:

- Start With Slow Start until:
 - SS Threshold is met
 - Go to Congestion Avoidance
 - Packet lost (3 resubmitted Acks)
 - Go to packet loss!
- Start Congestion Avoidance until:
 - Packet loss (3 resubmitted Acks)
 - Go to packet loss!
- On Packet Loss:
 - Fast retransmit
 - CWND = $\frac{1}{2}$ Current CWND
 - Go to Congestion Avoidance
- Time Out:
 - CWND = 1
 - Go to Slow Start

TCP - Tahoe



TCP - Reno

