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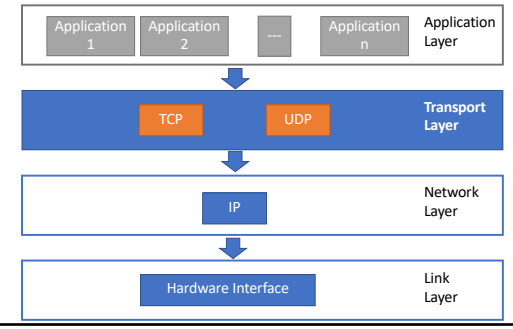
COMP2009J
Computer Networks

# Transport Layer

## Last Week

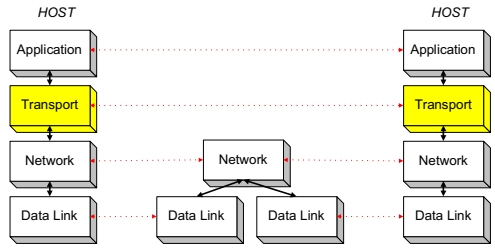
- IP Addressing:
  - an **end to end** addressing
- Two approaches for routing
  1. Static: hand-crafted
  2. Dynamic:
     - Vector Distance
     - Link State

## 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

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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

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Port 3550

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

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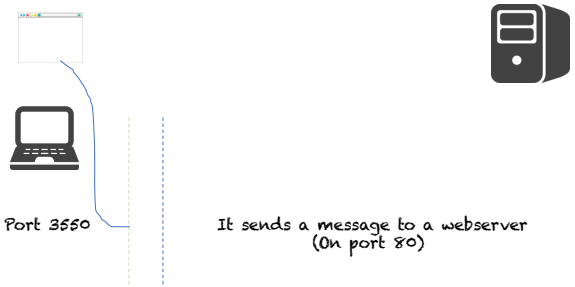


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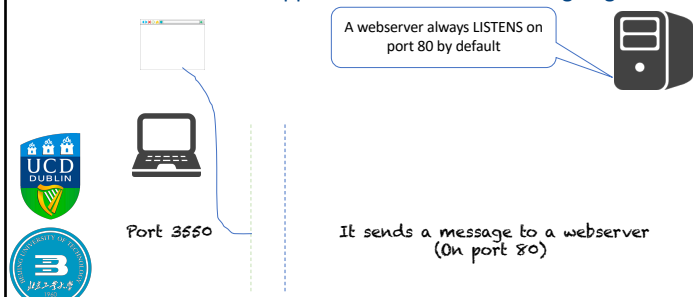


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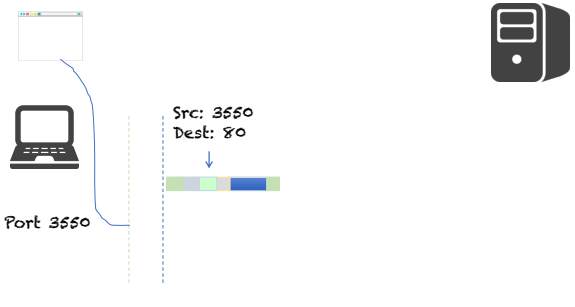


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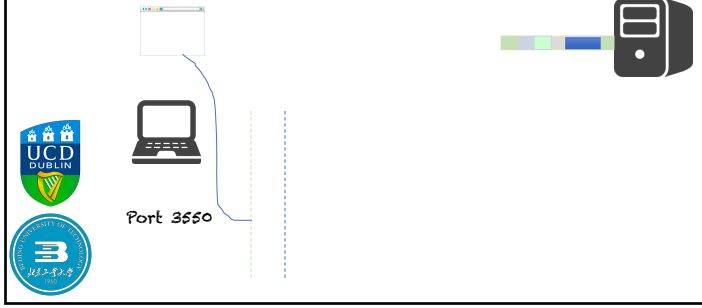

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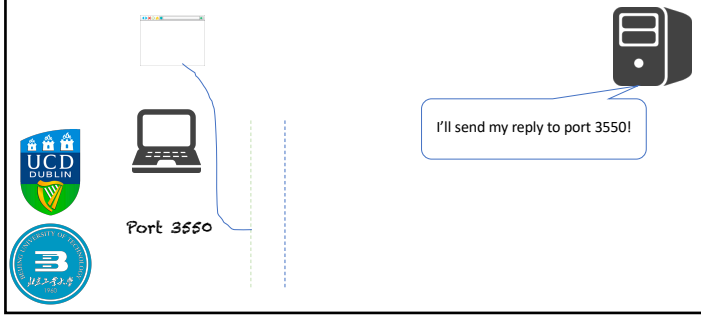


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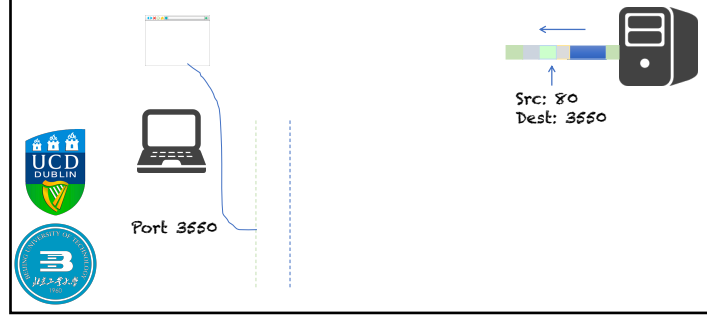


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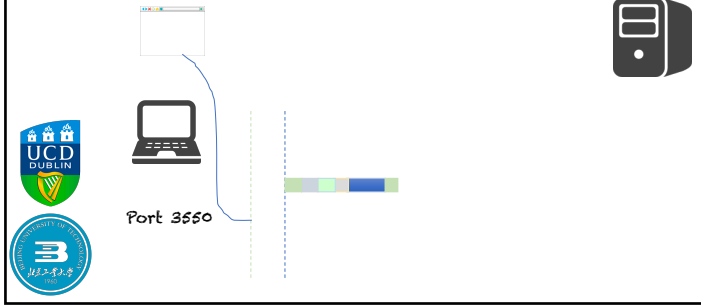


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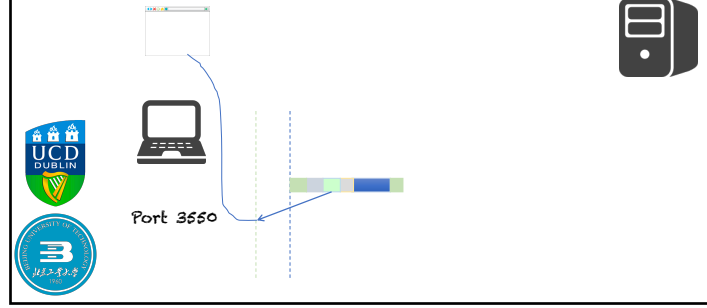

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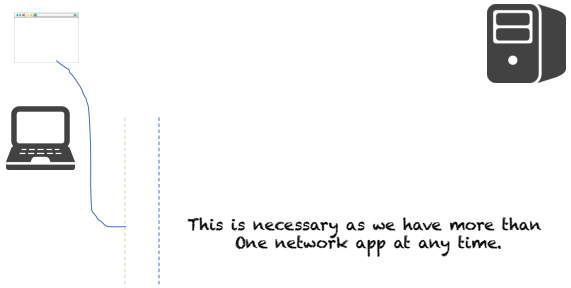

This is necessary as we have more than One network app at any time.

## Port Addressing

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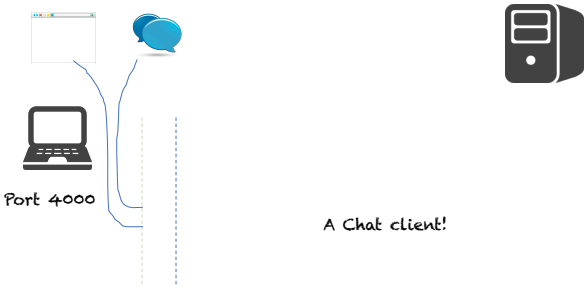


A Chat client!

## Port Addressing

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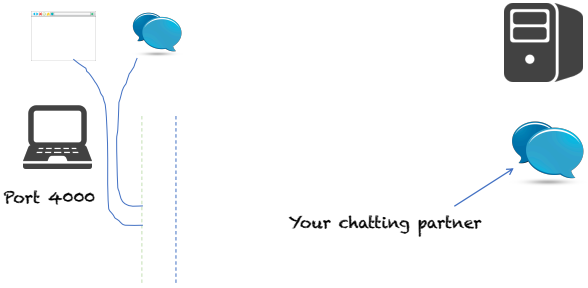


Your chatting partner

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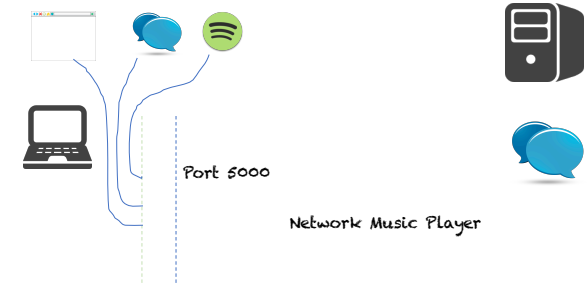


Network Music Player

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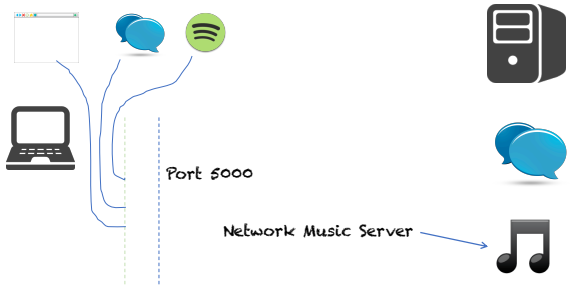


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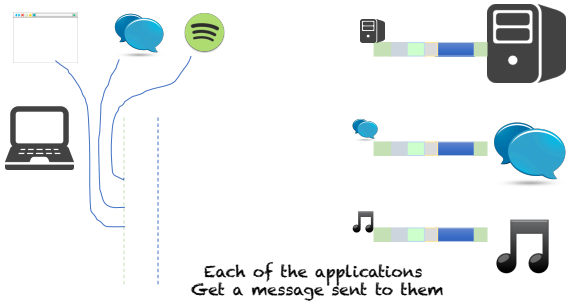


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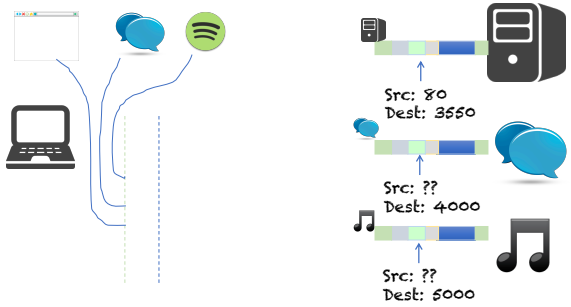


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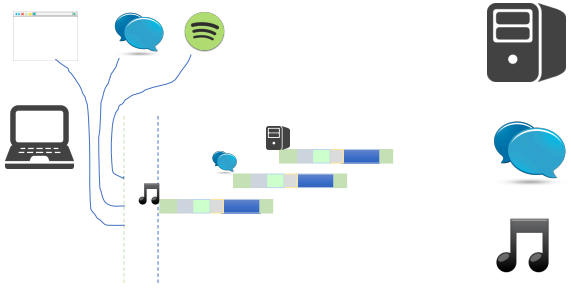

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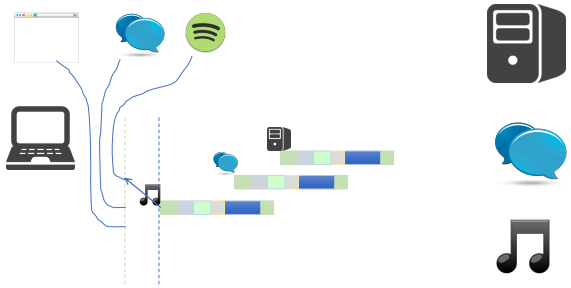

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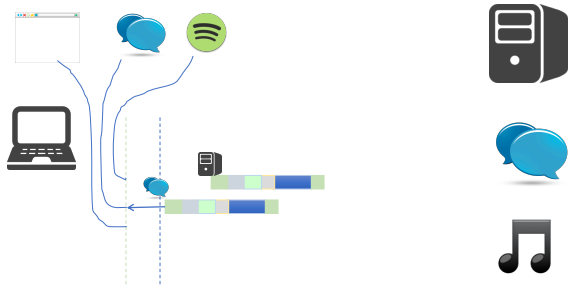

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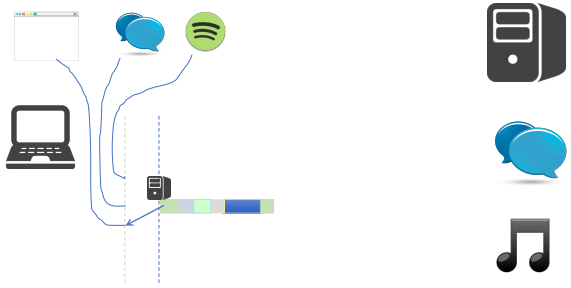


## Port Addressing

- 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

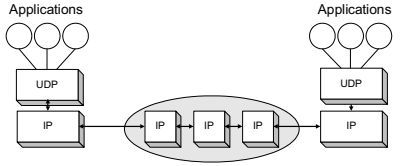
| | UDP - User Datagram Protocol | TCP - Transmission Control Protocol |
|---|---|---|
| Characteristics | • End-To-End Transport<br>• Unreliable | • End-To-End Transport<br>• Reliable |
| Protocol Details | • Simple<br>• Connectionless<br>• Datagram oriented | • Complex<br>• Connection-oriented<br>• Stream oriented |
| Use cases | • Unicast and Multicast<br>• Useful only for few applications multimedia applications<br>  • E.g., Live video streaming<br>• Used a lot for network services<br>  • E.g., Routing (RIP), Naming (DNS) | • Unicast only<br>• Used for most Internet applications:<br>  • E.g., Web (HTTP), Email (SMTP), File Transfer (FTP) |

## Port Numbers

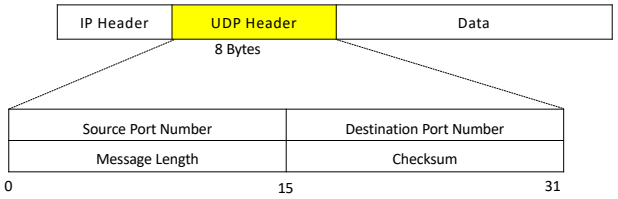
- UDP and TCP use a tuple **<IP address, port number>**
  - port numbers are used to identify applications
  - There are 65,535 ports per host
- There are many standard port numbers always used for particular applications (mostly)
  - 80 – HTTP service (web servers)
  - 443 – HTTPS (secure HTTP)
  - 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-to-host delivery service of IP datagram to an application-to-application service



## UDP Format

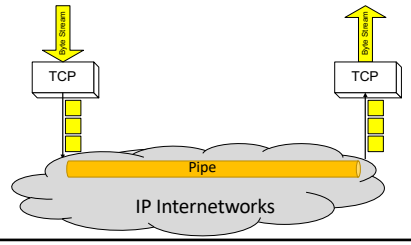



- **Port numbers** identify sending and receiving applications (processes). Maximum port number is $2^{16}$-1= 65,535
- **Message Length** is at least 8 bytes (I.e., Data field can be empty) and at most 65,535
- **Checksum** is for header (of UDP and some of the IP header fields)

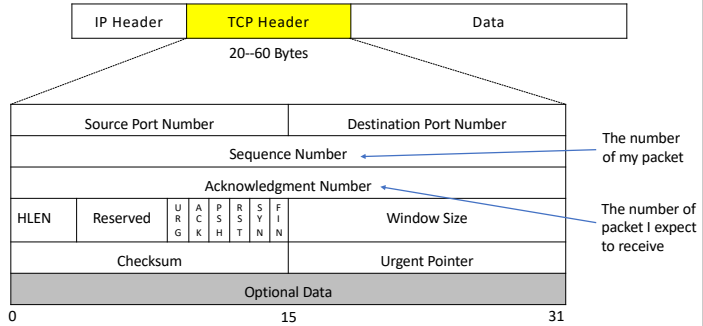
## TCP Overview

**TCP = Transmission Control Protocol**

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




## TCP Format




## 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

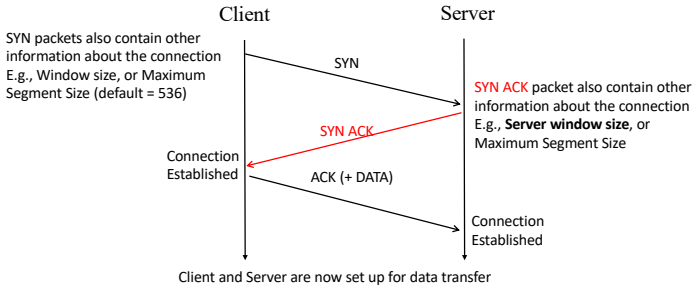
## Set Up/Close a Connection

- 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

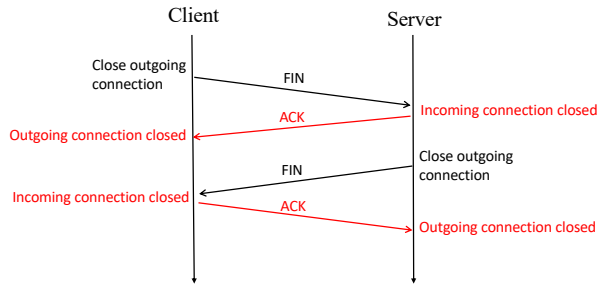
## Set Up a Connection: 3-Way Handshake

- 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

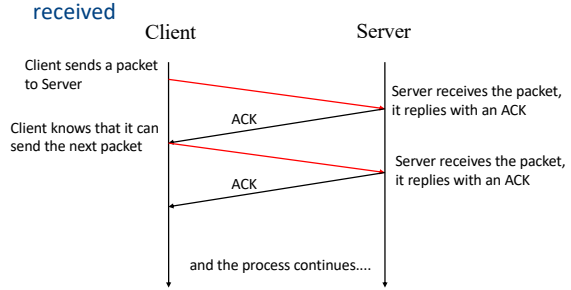


## Sequence Numbers

- 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 is packets have been lost over the network
- Received packets are acknowledged with the next expected byte

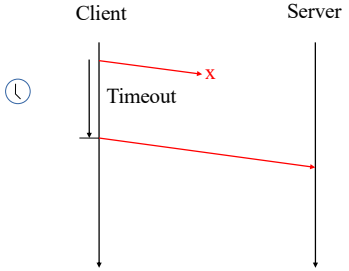
## Acknowledging packets

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



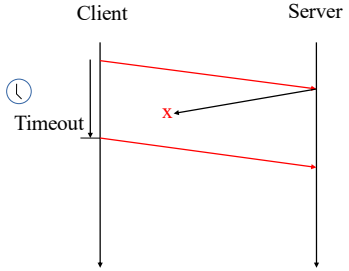

## Packet Timeout

- 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

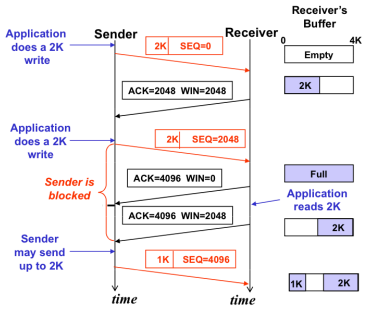
- Stop and Wait
  - Time consuming, does not use the full size of the pipe
- Sliding Window
  - Uses the pipe size (i.e., the minimum bandwidth over the path) to determine how many packets can be sent at a time
  - This is different from Layer 2 where window is fixed
  - Layer 4 windowing starts small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

Why does TCP do this?

## TCP Buffering and Windowing

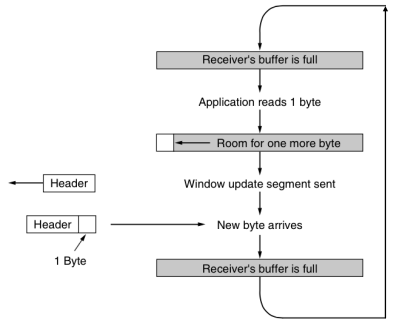
- Let's assume that an application sends large amounts of data between hosts at once
  - What if one host is not fast enough at handling the data?
  - What if there is a bottleneck at one point in the network?
- The data sent through the network is dropped as the host/network may not have enough resources 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



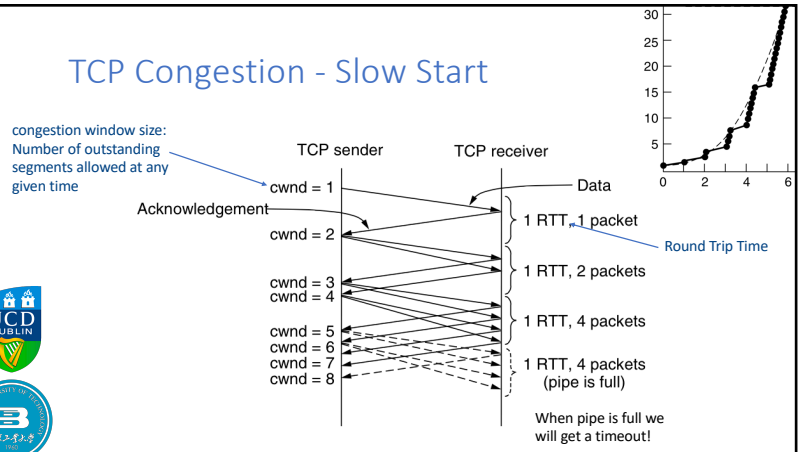
## TCP Congestion

- Congestion happens when 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

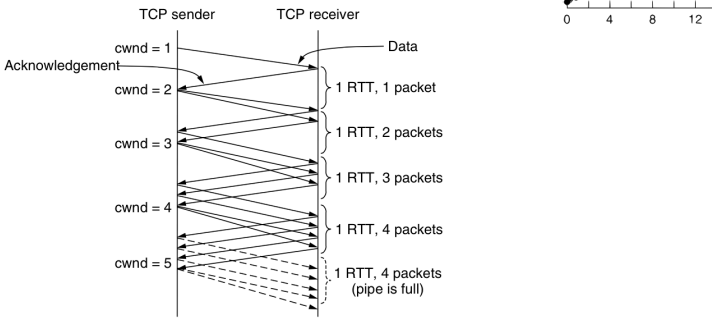
## Avoiding Congestion

- 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
- However, sending one packet per acknowledgement may underutilize the network
- TCP implements some schemes to reduce congestion, while using the network capacity
  - Slow Start: start with a small window and increase fast
  - Additive increase: a small linear increase

## TCP Congestion - Slow Start




## 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 by half 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 ½ the size and continue with additive increase

## TCP Congestion

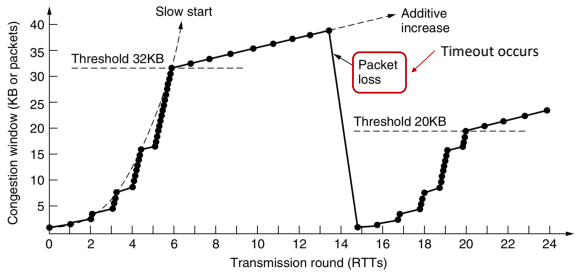
**Tahoe:**

- Start With Slow Start until:
  - SS Threshold is met
    - ➢ Go to Congenstion 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
  - SSthreshold = ½ CWND
  - CWND = initial state
  - ➢ Go to Slow Start

**Reno:**

- Start With Slow Start until:
  - SS Threshold is met
    - ➢ Go to Congenstion 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 = ½ current cwnd
  - ➢ Go to Congestion Avoidance

## TCP - Tahoe




## TCP - Reno