# TCP

**Transmission Control Protocol** 

# OSI Network Model



# Transport Layer Protocols

#### User Datagram Protocol (UDP)

- Unreliable, unordered delivery
- Connectionless
- Best-effort, segments might be lost, delivered out-of-order, duplicated
- Reliability (if required) is the responsibility of the app

#### Transmission Control Protocol (TCP)

- Reliable, inorder delivery
- Connection setup
- Flow control

Congestion control

# Sockets Interface



# Sockets Interface: socket

• Clients and servers use the socket function to create a socket descriptor:

int socket(int domain, int type, int protocol)

• Example:



Protocol specific! Best practice is to use getaddrinfo to generate the parameters automatically, so that code is protocol independent.

getaddrinfo("www.example.com", "http", &hints, &res); int s = socket(res->ai\_family, res->ai\_socktype, res->ai\_protocol);

# Sockets Interface: bind

• A server uses bind to ask the kernel to associate the server's socket address with a socket descriptor:

int bind(int sockfd, const struct sockaddr \*addr, socklen\_t addrlen);

- The process can read bytes that arrive on the connection whose endpoint is addr by reading from descriptor sockfd.
- Similarly, writes to sockfd are transferred along connection whose endpoint is addr.
- Protocol specific! Best practice is to use getaddrinfo to generate the parameters automatically, so that code is protocol independent.

# Sockets Interface: listen

- By default, kernel assumes that descriptor from socket function is an active socket that will be on the client end of a connection.
- A server calls the listen function to tell the kernel that a descriptor will be used by a server rather than a client:

int listen(int sockfd, int backlog);

- Converts sockfd from an active socket to a listening socket that can accept connection requests from clients.
- backlog is a hint about the number of outstanding connection requests that the kernel should queue up before starting to refuse requests.

# Sockets Interface: accept

• Servers wait for connection requests from clients by calling accept:

int accept(int sockfd, struct sockaddr \*addr, socklen\_t \*addrlen);

- Waits for connection request to arrive on the connection bound to listenfd, then fills in client's socket address in addr and size of the socket address in addrlen.
- Returns a connected file descriptor that can be used to communicate with the client via Unix I/O routines (fwrite, etc.).

# Connected vs. Listening Descriptors

- Listening descriptor
  - End point for client connection requests
  - Created once and exists for lifetime of the server
- <u>Connected</u> descriptor
  - End point of the connection between client and server
  - A new descriptor is created each time the server accepts a connection request from a client
  - Exists only as long as it takes to service client
- Why the distinction?
  - Allows for concurrent servers that can communicate over many client connections simultaneously
    - E.g., Each time we receive a new request, we fork a child to handle the request

# Sockets Interface



# Sockets Interface: connect

• A client establishes a connection with a server by calling connect:

int connect(int sockfd, const struct sockaddr \*addr, socklen t addrlen);

- Attempts to establish a connection with server at socket address addr
  - If successful, then clientfd (returned value) is now ready for reading and writing.
  - Resulting connection is characterized by socket pair
    - (x:y, addr.sin\_addr:addr.sin\_port)
    - x is client address
    - y is ephemeral port that uniquely identifies client process on client host

Best practice is to use getaddrinfo to supply the arguments addr and addrlen.

# accept Illustrated



1. Server blocks in accept, waiting for connection request on listening descriptor listenfd



2. Client makes connection request by calling and blocking in connect



3. Server returns connfd from accept. Client returns from connect. Connection is now established between clientfd and connfd

# **TCP** Connections

- TCP is connection-oriented
- A connection is initiated with a three-way handshake
- Server will typically create a new socket to handle the new connection



# Reliable Transport

- Each SYN segment will include a randomly chosen sequence number
- Sequence number of each segment is incremented by data length
- Receiver sends ACK segments acknowledging latest sequence number received
- Sender maintains copy of all sent but unacknowledged segments; resends if ACK does not arrive within timeout
- Timeout is dynamically adjusted to account for round-trip delay



### Transport-Layer Segment Formats

UDP

Source Port #	Dest. Port #
application message (payload)	

#### TCP



application message (payload)

# **Pipelined Protocols**

- Pipelining allows sender to send multiple "in-flight", yet-to-beacknowledged packets
  - Increases throughput
  - Needs buffering at sender and receiver
- How big should the window be?



#### what if a packet in the middle goes missing?



# Example

- Window Size = 4
- Sender can have up to 4 unacknowledged messages
- When ACK for first message is received, it can send another message



### TCP Fast Retransmit

 Receiver always ACKs the last id it successfully received

 Sender detects loss without waiting for timeout, resends missing packet



### Practice with TCP Sequence Numbers

- Consider the sequence of transmitted messages shown on the right
- What will be the next ACK number sent by the server?
- What will be the next Seq number sent by the client?



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# **TCP** Congestion Control

TCP operates under a principle of

- additive increase
  - window size++ every RTT if no packets lost
- multiplicative decrease
  - window size/2 if a packet is dropped

Bandwidth

Time

# **TCP Congestion Control**

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#### **TCP** Fairness

• Goal: if k TCP sessions share some bottleneck link of bandwidth R, each should have average throughput of R/k



Loss: decreases throughput proportional to current bandwidth

Congestion avoidance: increases throughput linearly (evenly)

# TCP Slow Start

 Problem: linear increase takes a long time to build up a decent window size, and most transactions are small

**Bandwidth** 

 Solution: allow window size to increase exponentially until first loss

Time

# TCP Slow Start

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# Practice with TCP Window Size

Assume someone changes the code of their TCP client by modifying the congestion avoidance as follows:

- Instead of increasing the window size by 1 each time an ACK is received,
- They double the window size each time an ACK is received (like in the slow-start phase).

What would be the pros and cons of this modification?

# **TCP** Connections

- TCP is connection-oriented
- A connection is initiated with a three-way handshake
- Recall: server will typically create a new socket to handle the new connection
- FIN works (mostly) like SYN but to teardown a connection



# TCP Summary

- Reliable, in-order message delivery
- Connection-oriented, three-way handshake
- Transmission window for better throughput
  - timeouts based on link parameters (e.g., RTT, variance)
- Congestion control
  - Linear increase, exponential backoff
- Fast adaptation
  - Exponential increase in the initial phase

# Network Model

#### Web up to HTTP/2



# HTTP/3

- HTTP/1.1: TCP (+ optional TLS or SSL) transmitted on IPv4 or IPv6
- HTTP/2: TCP + TLS transmitted on IPv4 or IPv6
- HTTP/3: QUIC (built on UDP) transmitted on IPv4 or IPv6
- HTTP/3 Availability
  - Chrome since April 2020
  - Edge since April 2020
  - Firefox since April 2021
  - Safari (not yet enabled by default)



#### TCP vs UDP



### TCP vs QUIC





### TCP + TLS vs QUIC



34

# HTTP/3 Layers

