4TC

PRS

Programmation Réseau et Système

membre de UNIVERSITÉ DE LYON

Département Télécommunications
Services & Usages

INSA LYON

PRS

- Equipe pédagogique
  - Responsable de cours: Razvan Stanica
  - Intervenants TP: Frédéric Le Mouel, Tristan Roussillon, Youssef Badra, Nicolas Loiseau
- Objectif
  - Faire le lien entre « réseaux » et « programmation »
  - Comprendre le rôle et le fonctionnement de la couche transport
  - Première utilisation de l'API Sockets

PRS

- Structure du cours
  - 4h de cours (couche transport avancée)
  - 2h de TD (fonctionnement TCP)
  - 8h de TP « guidé » sur l'API Sockets
  - 16h de TP sur les mécanismes TCP
  - 12h de projet – Implantation d'une couche transport pour un scénario donné
- Evaluation
  - 2 TPs notés (API Sockets)
  - points bonus/malus pour les autres TPs
  - présentation du projet
  - tests du projet

TC

4TC

PRS

Transmission Control Protocol

membre de UNIVERSITÉ DE LYON

Département Télécommunications
Services & Usages

INSA LYON

TCP

- TCP = Transmission Control Protocol
  - Basic concepts already discussed in 3TC IP
- Pre-requisites for PRS
  - TCP header format
  - Connection management
  - TCP state machine
- PRS objective: TCP congestion control

TCP

- TCP Congestion Control
  - Slow Start
  - Congestion Avoidance
  - Fast Retransmit
  - Fast Recovery
  - Selective Acknowledgements
- Standardized mechanisms
  - Original TCP – RFC 793
  - Updated August 2022 – RFC 9293
  - Additional mechanisms – RFC 1122, RFC 2581, RFC 5681

TCP

- Flow Control
  - Manage the data rate at the transmitter in order to not overwhelm a slower receiver
- Congestion Control
  - Manage transmission rate in order to avoid network congestion collapse
- End-to-End Argument
  - Represents the philosophy behind TCP (and behind Internet)
  - Data flow rate is controlled by end hosts
  - The network does not provide any congestion or flow control support

TCP

- TCP Vocabulary
  - Segment = the TCP payload data unit (different from “message”, “packet”, or “frame”)
  - Maximum Segment Size = the size of the largest segment that can be transmitted/received. The result of a negotiation between end hosts
  - Receiver Window (rwnd or awnd) = the number of segments a host can receive at a given moment. Used for flow control purposes
  - Congestion Window (cwnd) = a maximum number of segments that can be transmitted by a host, decided by congestion control mechanisms

TCP

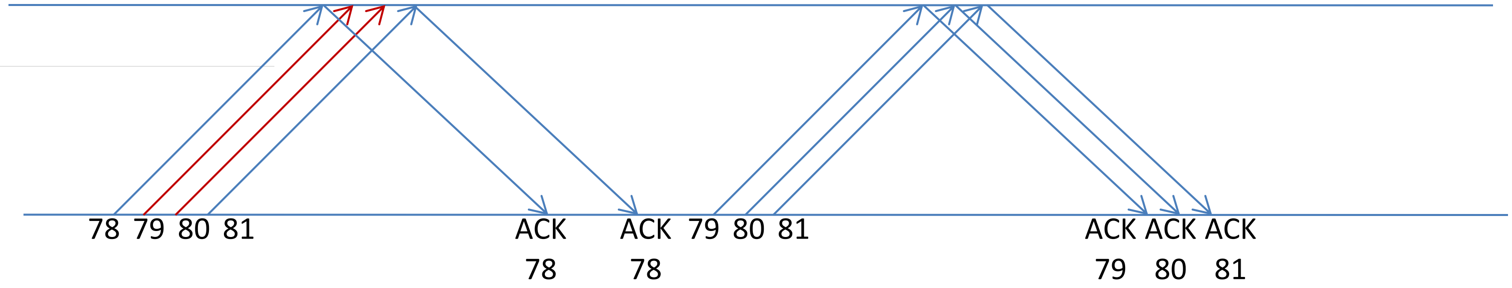
- Basic transmission principle
  - At any given time, a TCP host must not transmit a segment with a sequence number higher than the sum of the highest acknowledged sequence number and the minimum of cwnd and rwnd
- Important metric
  - Round-Trip Time (RTT): the time between the transmission of the segment and the reception of the ACK. RTT can vary significantly during network operation, so TCP keeps an updated estimated value

TCP

- FlightSize
  - The amount of data that has been sent, but not yet acknowledged
  - A common mistake is to consider FlightSize=cwnd
- Congestion detection
  - Based on the assumption that a segment lost in the network is the result of congestion
  - A sender starts a timer (based on its RTT estimate) every time it transmits a segment
  - If an ACK from the destination is not received before the timeout, a loss is detected

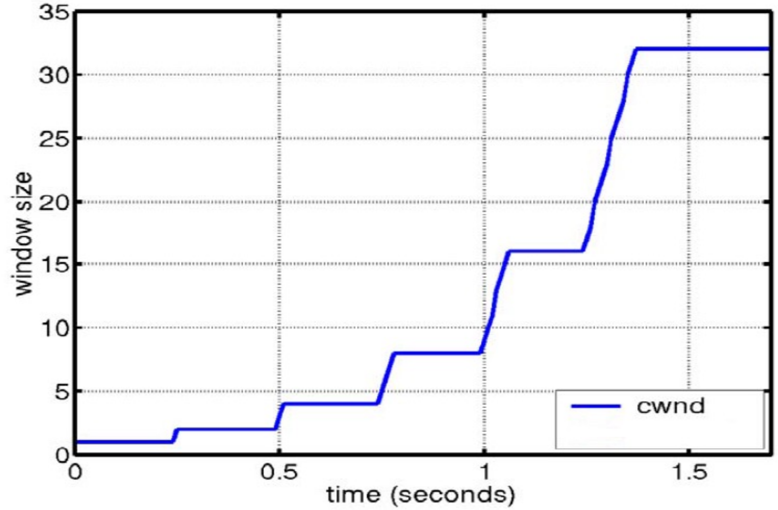
TCP

- Duplicate ACKs
  - In normal operation, a receiver is not allowed to acknowledge discontiguous segments
  - The reception of an out-of-order segment results in the acknowledgement of the last contiguous segment (a duplicate ACK)



TCP

- Slow Start
  - Motivation: the end hosts do not know the state of the network at the beginning of their connection
  - Start with cwnd= 1
  - For every received ACK: cwnd= cwnd+ 1
  - Practically, cwnd doubles during an RTT interval
  - Despite its name, exponential increase of cwnd

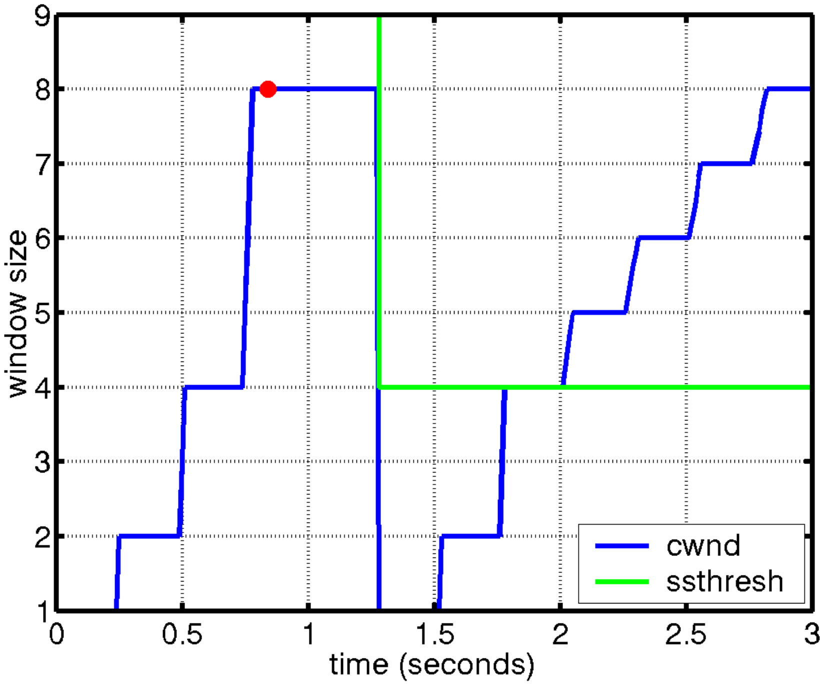


TCP

- Slow Start Threshold (ssthresh)
  - Important TCP parameter
  - Decides the moment when the host goes from Slow Start to Congestion Avoidance
  - Arbitrary initial value (usually very high)
  - ssthresh must follow the congestion level
  - After a lost segment (detected through a timeout or duplicate ACK): ssthresh= FlightSize/2
  - After the retransmission: cwnd= 1

TCP

- Congestion Avoidance
  - The TCP host enters in this mode when cwnd > ssthresh
  - cwnd= cwnd+ 1/cwnd
  - For each RTT: cwnd= cwnd+ 1



TCP

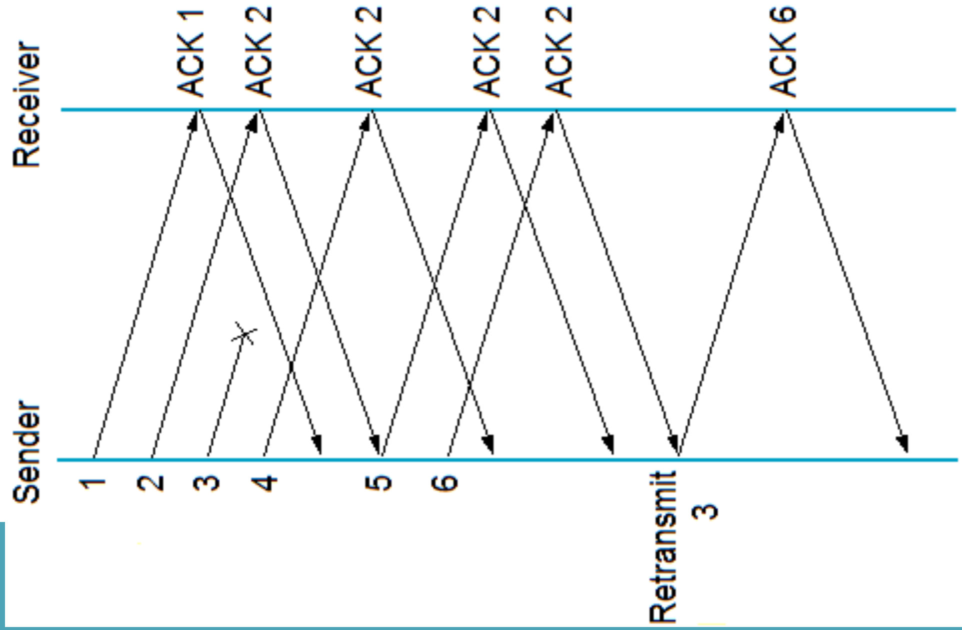
- Congestion Avoidance
  - Motivation: the exponential increase of Slow Start is too aggressive
  - Once a congestion has been detected, the transmitter tries to avoid reaching the congested state once again
  - A static approach can miss the opportunity of an increased throughput
  - The slow cwnd increase can delay the next congestion, while still testing for transmission opportunities

TCP

- Fast Retransmit
  - A timeout is a clear indication of network congestion, but can be very long
  - A duplicate ACK can have different reasons: congestion, segments following different paths, re-ordered ACKs
  - Considering a segment lost after the first duplicate ACK is too aggressive
  - TCP considers a segment lost after 3 duplicate ACKs (that means 4 consecutive ACKs of the same segment)

TCP

- Fast Retransmit
  - The usual operation mode
  - Retransmit lost message
  - Calculate FlightSize= min(rwnd,cwnd)
  - ssthresh= FlightSize/2
  - Enter Slow Start: cwnd= 1



TCP

- Fast Retransmit
  - This mechanism generally eliminates half of the TCP timeouts
  - This yields roughly a 20% increase in throughput
  - It does not work when the transmission window is too small to allow the reception of three duplicate ACKs

TCP

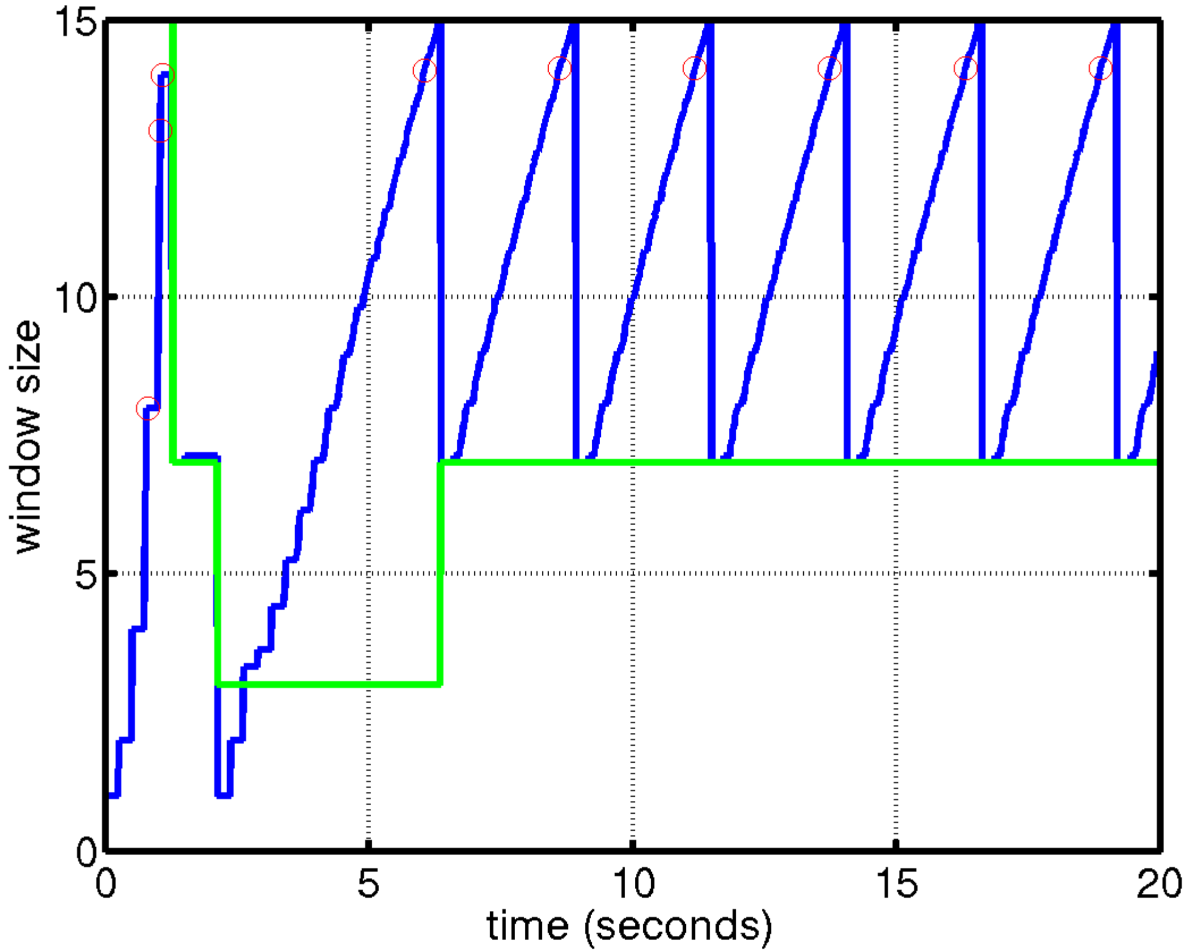
- Fast Recovery
  - The reception of duplicate ACKs also means that network connectivity exists, despite a lost segment
  - Entering Slow Start is not optimal in this case, as the congested state might have disappeared
  - The mechanism allows for higher throughput in case of moderate congestion
  - Complement of Fast Retransmit

TCP

- Fast Recovery
  - Mode entered after 3 duplicate ACKs
  - As usual, set ssthresh= FlightSize/2
  - Retransmit lost packet
  - Window inflation: cwnd= ssthresh+ ndup (number of duplicate ACKs received)
  - This allows the transmission of new segments
  - Window deflation: after the reception of the missing ACK (one RTT later)
  - Skip Slow Start, enter Congestion Avoidance

TCP

- Typical TCP Saw-tooth Pattern



TCP

- Selective Acknowledgements
  - The receiver can only acknowledge contiguous segments
  - No ACK for segments correctly received after a lost segments
  - The sender has no feed-back regarding correctly received segments: retransmit or not?
  - Ideally, the sender should retransmit only the missing segments
  - With SACK, the receiver provides this feed-back to the sender

TCP

- Delayed ACK
  - RFC 1122
  - Reduce overhead by combining multiple ACKs in one segment
  - Delay an ACK by up to 500 ms
  - For a stream of full-sized incoming segments, an ACK is sent every second segment
  - Can be interesting for piggy-backing: data and ACK in the same segment

TCP

- Nagle's algorithm
  - RFC 896
  - Small packet problem
  - Combine small outgoing data and send one single segment
  - If segment with un-received ACK, keep buffering output data until a full size segment can be sent
  - Poor interaction with delayed ACKs

TCP

- TCP Versions
  - TCP Tahoe: Slow Start, Congestion Avoidance, Fast Retransmit
  - TCP Reno: Fast Recovery
  - TCP New Reno: Modified Fast Recovery (window inflation)
  - Many other proposals exist: Vegas, Hybla, BIC, Westwood, …

TCP

- TCP Cubic
  - Current state of the art
  - Window size no longer controlled by received ACKs
  - cwnd computed as a cubic function of time since the last congestion
  - Three phases:
    - aggressive increase until ssthresh (similar to slow start)
    - slow probing for higher window
    - aggressive probing for higher window

TCP

- Beyond the End-to-End Argument
  - The way routers decide to drop packets impacts the functioning of TCP
  - Advanced techniques can be implemented inside the network
- Random Early Detection
  - RED manages router queues and drops packets based on a queue threshold
  - Once the queue is over the threshold, the router drops packets with a certain facility
  - Only the affected TCP senders will enter Slow Start or Congestion Avoidance, slowing the network down before the actual congestion

TCP

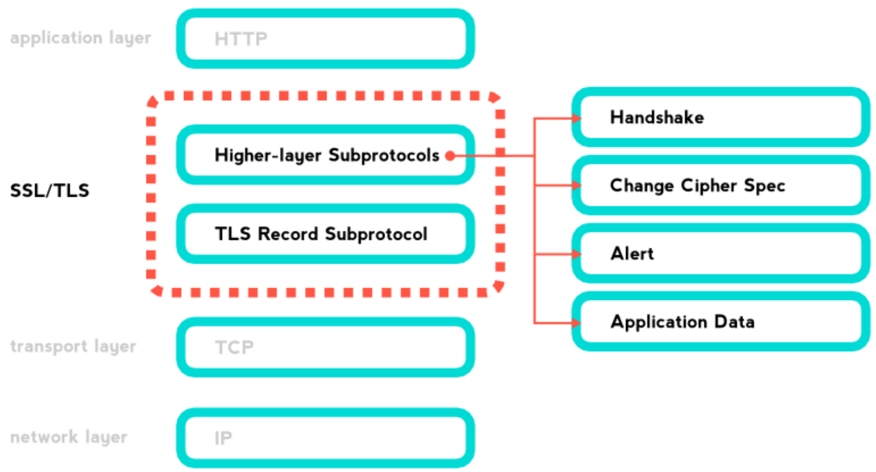
- Explicit Congestion Notification
  - ECN is based on a queue threshold parameter, just as RED
  - As opposed to RED, ECN only marks packets instead of dropping them
  - Routers mark 2 bits in the IP header (Type of Service field) to signal whether congestion is occurring
  - Through cross-layer mechanisms, TCP can learn this information and reduce the congestion window
  - ECN avoids packet drops and reduces the delay created by retransmissions

TCP

- QUIC – Quick UDP Internet Connections
  - User space implementation of a transport protocol
  - Released by Google in 2013
  - From January 2017, implemented in the Chrome browser and the Google Search and You Tube applications
  - Implemented by 8% of websites
  - 30% of Internet traffic in EMEA region (16% in NA)
  - 5% reduction in search time
  - 15% reduction in video rebuffering

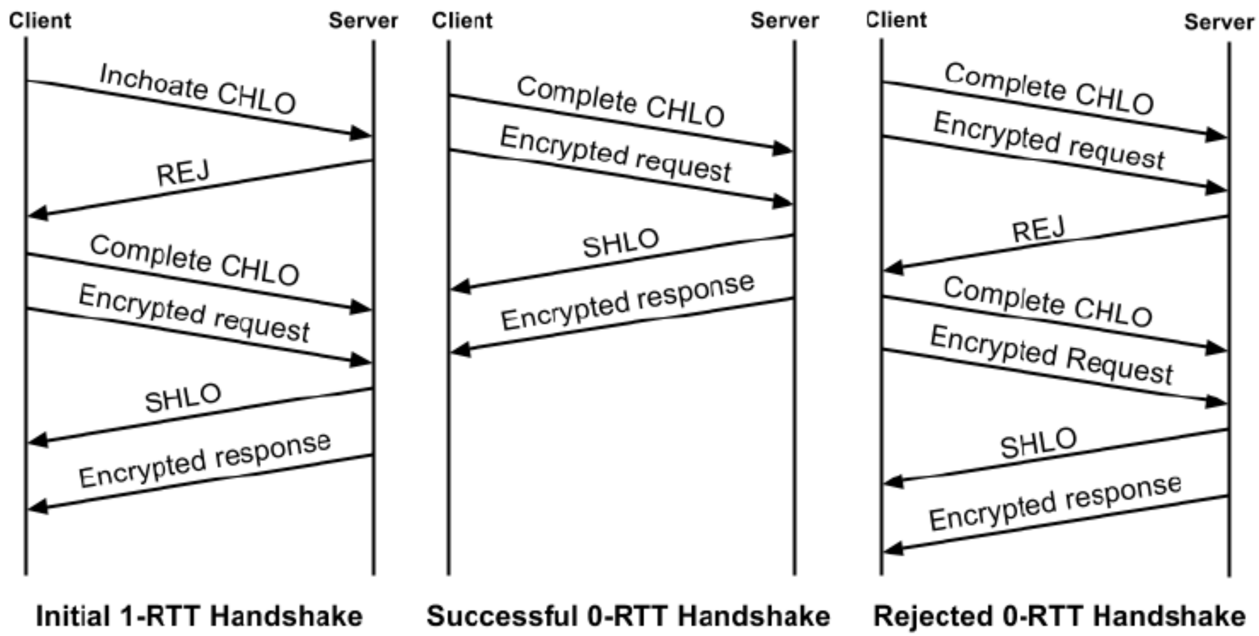
TCP

- QUIC – Motivations
  - Classic functioning: HTTP/2 – TLS – TCP
  - Transport Layer Security (TLS) – extra overhead
  - Data transmitted after 2xRTT



TCP

- QUIC – Principles
  - Save the context of already known servers/clients
  - Results in 0-RTT connection in 85% of the cases



TCP

- QUIC – Principles
  - Multiple streams transmitted over the same connection (similar mechanisms in TCP)
  - Streams are controlled both independently (flow controlled) and per connection
  - A large part of the transport layer information is encrypted
  - Unique identifier, even for retransmissions, easing RTT estimation
  - TCP congestion control mechanisms

TCP

- QUIC – Problems
  - Not always better performance than TCP (e.g. when a lot of packets are delivered in disorder)
  - 2x CPU consumption compared with TCP
  - TCP unfriendly

TCP

- Implementation
  - At the transport layer, an active application is identified by the 5-tuple: (protocol, $@IP_{source}$, $Port_{source}$, $@IP_{dest}$, $Port_{dest}$)
  - A client needs to know $@IP_{server}$ and $Port_{server}$ in order to send a connection request
  - The $Port_{client}$ can be allocated dynamically by the operating system

TCP

- Socket
  - Application programming interface (API) for communication between processes
  - When processes are run on different machines, a socket becomes the basis of network communications
  - Support for both TCP and UDP
  - Bidirectional communication using functions such as *read()/write()* or *send()/recv()*
  - A socket is represented as a file handler in Unix systems

TCP

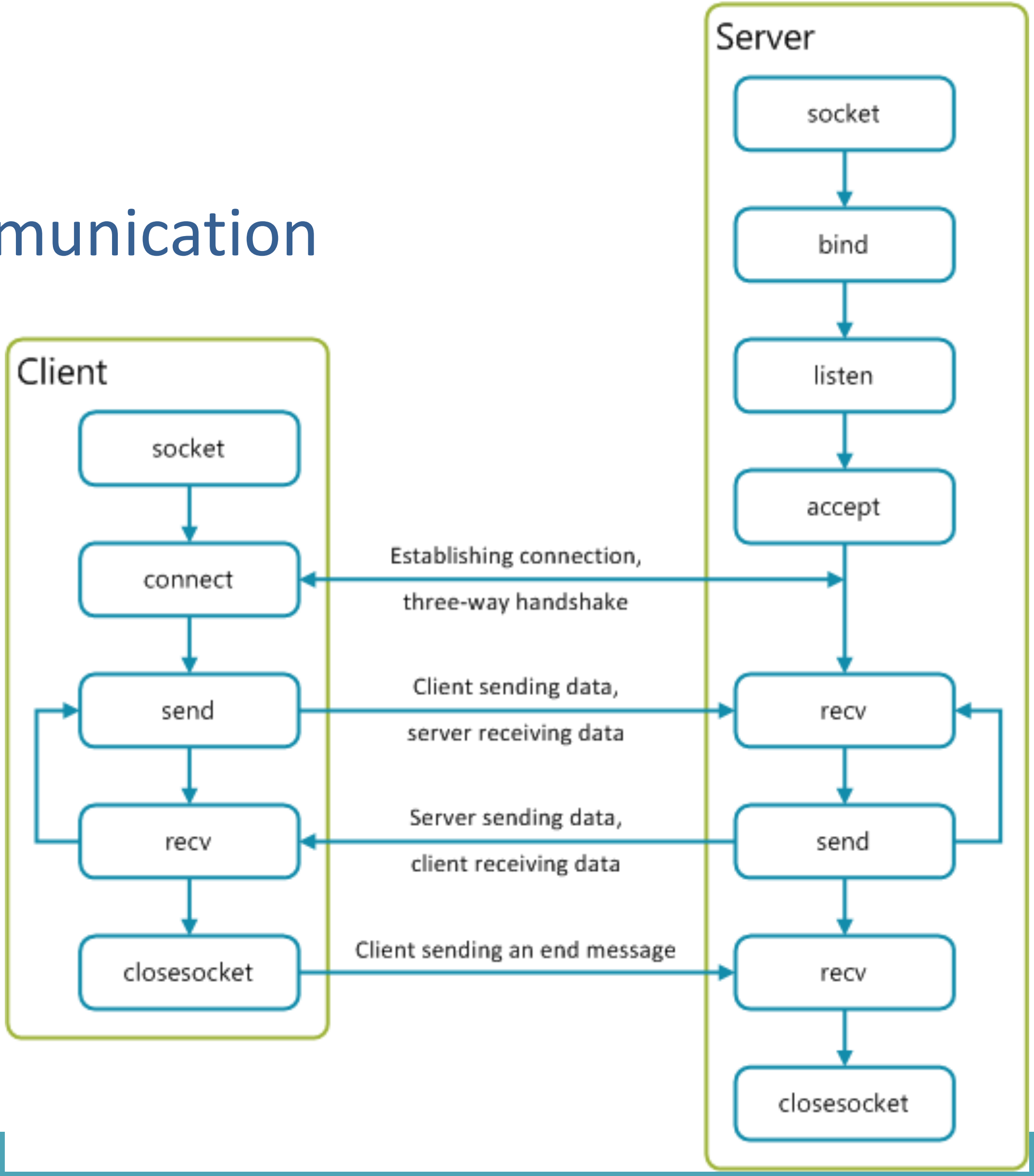
- Socket API
  - A series of libraries
    - sys/socket.h – core socket functions and data structures
    - netinet/in.h – IP, TCP and UDP data structures
    - sys/un.h – data structures for local communications
    - arpa/inet.h – functions for manipulating IP addresses
    - netdb.h – functions for translating protocol and host names

TCP

- Socket API
  - A series of functions
    - socket()
    - bind()
    - listen()
    - connect()
    - accept()
    - send() / write()
    - recv() / read()
    - close()
    - select()
    - poll()

TCP

- TCP communication



TCP

- UDP communication