

Communications Systems Engineering

Moritz Hüllmann

March 4, 2022

1 Network Programming

There are five socket types, but only three are relevant.

- SOCK_RAW
- SOCK_STREAM
- SOCK_DGRAM
- SOCK_RDM
- SOCK_SEQPACKET

Combined with **Protocol Family**, communication is defined completely. Focus is on **IPv4/v6**.
Relevant protocol families

- PF_INET – IPv4
- PF_INET6 – IPv6

Sockets give access to **transport layer**. One may also use AF instead of PF prefixes.

1.1 C-Implementation of Sockets

Definition: socket(·)

This function creates a new reference to a socket in the OS.

```
int socket(int socket_family, int socket_type, int protocol);
```

protocol – optional, if there is only one possibility. Set to 0 if not wanted

Returns: 0 on success, -1 on error

Definition: bind(·)

This socket binds a socket to a specific address and port.

```
int bind(int sock, const struct sockaddr* addr, socklen_t addrlen);
```

sock – a file descriptor, i.e. the return value of socket(·)

Returns: 0 on success, -1 on error

bind(·) – commonly used by servers. `struct sockaddr* addr` has all information regarding IPv4/v6.

One has to be careful regarding byte order. Sockets usually require **network byte order**.

htonl(.) Host to network long

ntohl(.) Network to host long

... and many more

Definition: getaddrinfo(.)

This function gets all possible based on the information passed to it. Can be used to determine IP addresses, ports and so on.

```
int getaddrinfo(const char *node, const char *service, const struct addrinfo *hints,
               struct addrinfo **res);
```

hints – used to tell the function what it should do specifically.

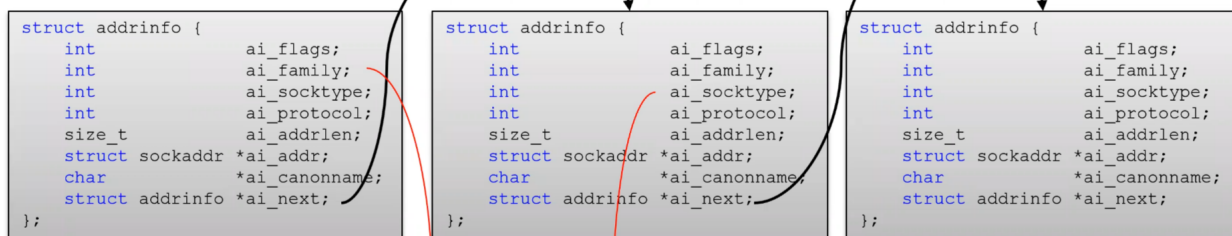
Returns: 0 on success, -1 on error. On success, **res** contains a pointer to a linked list of whatever was requested

This function should be used everytime, one implements server/client applications, as it can handle DNS resolution, handle both IPv4 and v6 and so on.

Suggested call order:

1. getaddrinfo(.)
2. socket(.)
3. bind(.)

• Returns:



```
int socket(int socket_family, int socket_type, int protocol);
```

```
int bind(int sockfd, const struct sockaddr* addr, socklen_t addrlen);
```

Figure 1: How to use getaddrinfo(.). Note: All information used should originate in one block, not two as shown above.

1.1.1 Server-side programming

Up until now, only communication setup was discussed, we need to communicate now.

Definition: listen(·)

This function listens for incoming connection requests.

```
int listen(int sockfd, int backlog);
```

sockfd – Socket to listen on.

backlog – Queue length for pending requests.

Note: This is only valid with SOCK_STREAM and SOCK_SEQPACKET.

If the queue is full, the server will not answer. listen(·) is a blocking function.

Definition: accept(·)

This function is used to accept requests received by listen(·).

```
int accept(int sockfd, struct sockaddr *addr, socklen_t *addrlen);
```

addr – holds the address that should be accepted

addrlen – length of addr. One can pass a structure that is big enough for IPv4 and IPv6.

Returns: File descriptor for new socket, that handles the accepted connection from now on.

1.1.2 Client-side programming

Definition: connect(·)

This function connects the socket to a server. It may perform a TCP handshake and similar things in the process.

```
int connect(int sockfd, struct sockaddr *addr, socklen_t addrlen);
```

addr – address to connect to

addrlen – length of the address

Returns: 0 on success, -1 on failure

1.1.3 Sending and receiving

One can *write* to and *read* from sockets, with the following functions:

- write(·) and read(·)
- send(·) and recv(·) – TCP
- sendto(·) and recvfrom(·) – UDP

Definition: Stream

A stream is like reading from a file. TCP just decides to cut the stream at certain points and transmits each fragment as packets. We do not know the context of a single packet.

Definition: Datagram

Messages transmitted via datagrams are not cut up by the kernel, they are sent as is. If the message is too large, there either is an error, or the package gets fragmented. Each datagram is self-contained, so every received packet can be seen as a unit.

Definition: send(·)/write(·)

These functions can be used to write to a socket. As in Linux everything is a file, one can use `write(·)` instead of `send(·)`.

```
ssize_t send(int sockfd, const void* buf, size_t len, int flags);  
ssize_t write(int fd, const void* buf, size_t count);
```

`flags` – instruct kernel to handle send request in a certain way

`buf` – Buffer with data to send

Returns: Bytes actually written. -1 on error, non-negative otherwise **Note:** `write(·)` is equal to `send(·)` with `flags = 0`

Definition: recv(·)/read(·)

These functions are designed to read a certain amount of bytes from a socket (or file).

```
ssize_t recv(int sockfd, void* buf, size_t len, int flags);  
ssize_t read(int fd, void* buf, size_t count);
```

Parameters are analogous to `send(·)`.

Actually using the return value of the reading functions is really important. It may be unknown, how many bytes the applications is going to receive in a single read operation. If *retval* \neq *length*, then not enough was read.

Definition: sendto(·) for connection-less sockets

This function sends a certain amount of bytes from a buffer to the specified address and port.

```
ssize_t sendto(int sockfd, const void* buf, size_t len, int flags, const struct  
sockaddr *dest_addr, socklen_t addrlen);
```

`dest_addr` – Address and port to send to

Note: There is no equivalent `write` function.

Note: Parameters up to `flags` same as for `send(·)`. Exclusively for sending datagrams.

Definition: `recvfrom(·)` for connection-less sockets

This functions receives a certain amount of bytes from a given address.

```
ssize_t recvfrom(int sockfd, void* buf, size_t len, int flags, const struct sockaddr
                *src_addr, socklen_t *addrlen);
```

Note: `src_addr` will be replaced with the address, that really was received from. `addrlen` will then hold the correct length of the structure.

Note: Parameters up to `flags` same as `recv(·)`.

Again, if it is not known which version of IP is going to be used, use `struct sockaddr_storage`, to allow for both versions.

1.1.4 Closing connections

There are several methods for closing a connection, each to be used in a different scenario.

Definition: `close(·)`

Drops all packets queued for receiving and sending, and shuts down everything else related to the socket. Also frees all previously allocated space.

```
int close(int sockfd);
```

Returns: 0 on success, -1 on error

For more conservative shutdown operations, the following function can be used.

Definition: `shutdown(·)`

More conservative `close(·)` variant.

```
int shutdown(int sockfd, int flags);
```

`flags` – 0: no further receives, 1: no further sends, 2: both

Note: Return values same as for `close(·)`

Note: Sends `FIN` and/or `FIN_ACK`, other party might get `RST` (reset) when trying to read from the shut down socket

Note: This allows all queued data to be send.

`shutdown(·)` cannot stand alone. `close(·)` must be called after a call to `shutdown(·)`, as it releases all hugged structs.

1.1.5 Connection overview

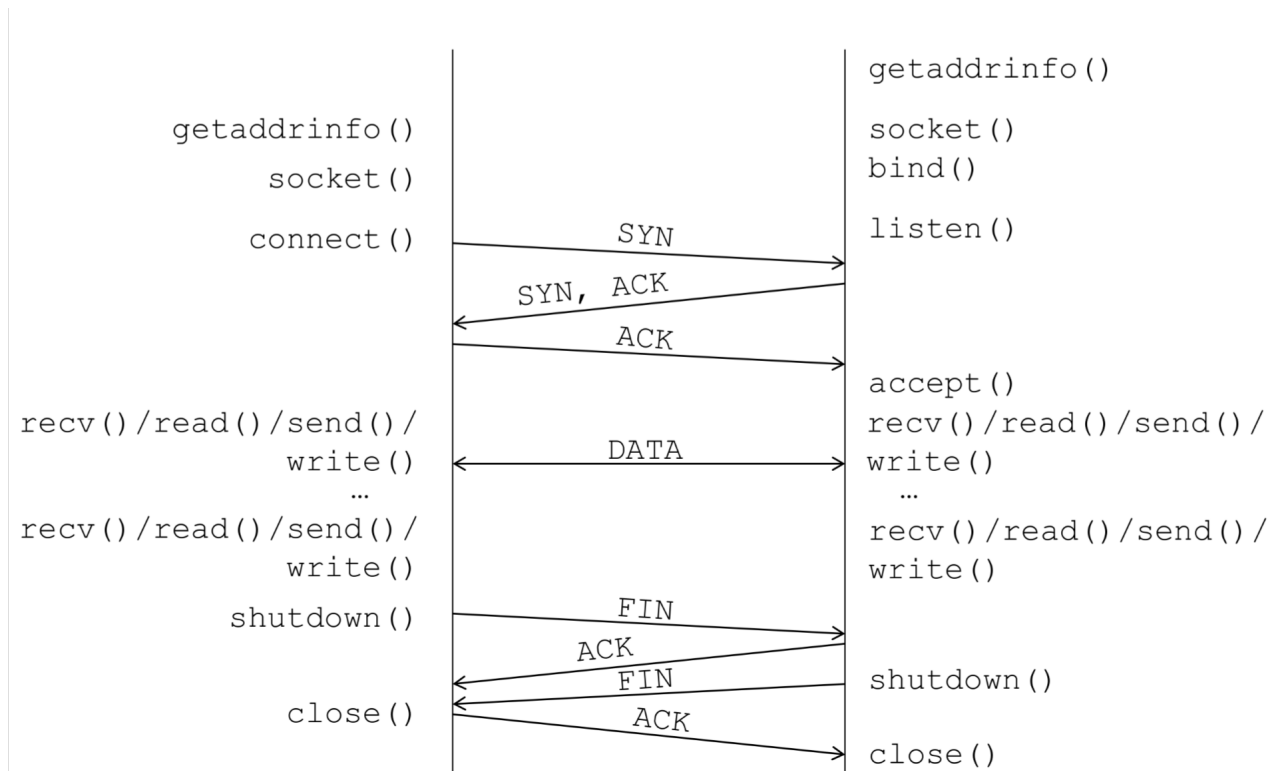


Figure 2: Overview over the course of a connection-aware socket

1.1.6 Configuring a socket

Definition: `setsockopt(·)`

Apply options to sockets.

```
int setsockopt(int sockfd, int level, int optname, const void* optval, socklen_t
    optlen);
```

`level` – basically ISO/OSI level the option should be applied to

Returns: 0 on success, -1 on error

Getting options set for a sockets uses a functions with more or less the same signature as `setsockopt(·)`, but is called `getsockopt`.

I/O-controls can also be used to set and read properties of a socket. Commonly:

- Timestamp of last received packet
- How many bytes are unsend (TCP)
- How many bytes are unread (TCP)
- ...

1.1.7 Non-blocking sockets

There are two methods that one could use: `select()` and `epoll`.

Non-blocking sockets using `select()` All network calls are blocking by default. This is not optimal in a server setting, as this introduces great overhead and occupies many resources of the server, that could be used elsewhere. **Threads** are not an option for highly scalable systems, as they introduce great computation overhead. `fcntl()` can be used to set flags to sockets, specifically `O_NONBLOCK`. Callbacks are called, to notify a program, if a socket is write-/readable. Polling is not an option, this is busy waiting.

Definition: `select()`

```
int select(int nfds, fd_set *readfds, fd_set *writefds, fd_set *exceptfds, struct
           timeval *timeout);
```

`nfds` – highest file descriptor number in all sets + 1

`readfds` – list of sockets to monitor readability for

`writefds` – list of sockets to monitor writeability for

`exceptfds` – list of sockets to monitor exceptions for

`timeout` – maximum wait time

Returns: 0 on timeout, -1 on error, else number of fd's with event `read/write/exceptfds` is then set to the sockets with events. Manual checking which sockets have events is required.

Non-blocking sockets using `epoll` It knows two modes:

1. level triggered, which is just as `select()`
2. and edge triggered.

Edge triggered `epoll` informs on every *change* of states for read and write queues, as well as excepts.

If there are 5 new bytes received, calling `epoll` returns that new data is readable. However, if just 1 byte is read and `epoll` called again, it will block, as there is no **new** data available.

An `epoll` file descriptor is attached to the sockets queue, directly in the kernel. Each `epoll fd` has a queue of its own, and notifies for every socket listed there.

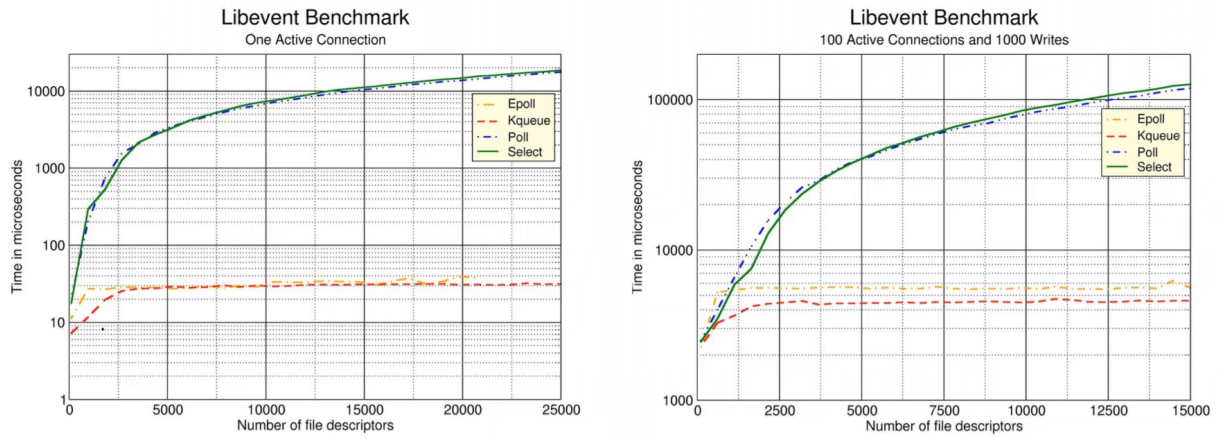


Figure 3: Performance of `epoll` vs. `select()`

1.2 TCP options and packetization

TCP needs to decide important questions:

1. When to wait for more data from user space
2. When to send queued data

If not done so, every `send()` results in a packet. If only one byte is send, that would result in packages with 40 bytes of overhead, but only one byte of payload.

1.2.1 Nagle's Algorithm

Algorithm: Nagle's Algorithm

This algorithm is default TCP socket behaviour.

Small chunks are accumulated and send, when previous data was ACKed.

```
if (there is new data to send) {
    if (window size >= max(segment size) && available data >= max(segment size)) {
        complete max(segment size) and send now;
        queue remaining data;
    }
    else if (exists(data in flight waiting to be ACKed)) {
        queue data and send when ACK is received;
    }
    else {
        send data immediately;
    }
}
```

Note: This algorithm is not suited for every application, as it may wait, when an immediate send might be necessary.

Disable with `TCP_NODELAY` with `setsockopt(·)`.

1.2.2 Delayed ACKs

Basic idea: when data is received, we will want to send data as a response. Piggy back ACKs for previous data on data you want to send. The delay is $\leq 0.5s$. At least every second packet with MSS is ACKed immediately.

This also is TCP default behaviour. However, don't combine with section 1.2.1, this may lead to unnecessary waiting times of up to 0.5 s.

Disable with `TCP_QUICKACK` with `setsockopt(·)`.

Also possible: packetize TCP yourself, using `TCP_CORK`, which works best with `TCP_NODELAY`. It will still send out MSS packets immediately. Allows for application level flushing.

Can also be achieved with `send(·)` and its flags, by adding `MSG_MORE` to it.

1.2.3 TCP fast open

Normal TCP has 1 RTT delay due to 3-way handshake.

This option adds the TCP request to TCP's SYN message. Leads to 5-7% speed-up.

Problems:

Don't process request before handshake is complete: risk to security, if handshake would not be completed. Possible DoS scenarios:

- Resource exhaustion attack – Leave connection half open with SYN-flood
- Reflection attack – Spoof live IP addresses, so those get spamed

Also problem with duplicated data. This also makes the above attacks easier.

Avoiding those problems:

What to do to tackle the problems

- Keep verified hosts on a *secure* whitelist
- Only trust peers that completed a handshake – this needs a proof
- Application must tolerate duplicated SYN data

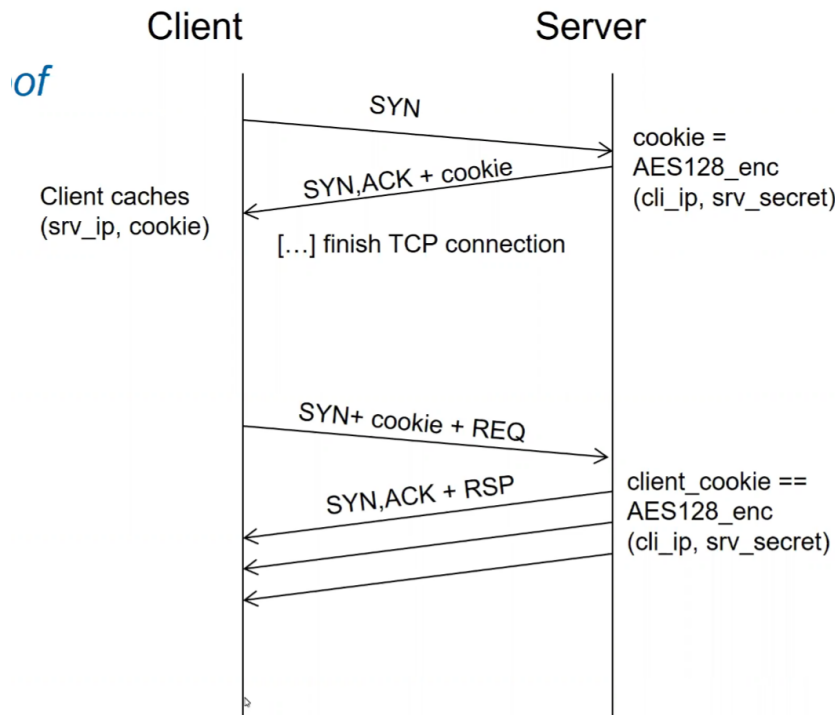


Figure 4: Using cookie as proof of validity of connection

Cookies are not valid forever. If cookie is invalid continue with normal TCP handshake.

Cookies may not reach destination. Can be dropped by middleboxes, also timeouts can occur. Cookies can be stolen → Redirection attacks Cookies are dependent on the IP address of the client → change in networks invalidates the cookie, so mobile usage is not optimal

TCP fast open is not used today, as it leads to more problems than it solves. It also has significant problems with middleboxes.

Activating TCP fast open on servers using `setsockopt()` with `TCP_FASTOPEN` together with a maximum count of connections.

Activating TCP fast open for clients by piggybacking data to the connect call. Also need to use `MSG_FASTOPEN`.

1.2.4 Multipass TCP

Enable multiple connections between clients and servers. This enables for some backup paths, if one connection suddenly fails. However, TCP is *single-path* only, meaning there can only be one connection per socket. This leads to poor performance for mobile users, if, for example, if you change from WiFi to 4G networks.

How to use multi-path TCP:

1. Send options `MP_CAPABLE` with `SYN` message to signal multi-path capability.
2. If servers sends the same option back, both parties know, that the other party supports
3. Send specific `JOIN` with another `SYN` message to server, signalling to which connection to join the incoming one to.

Again, there are problems with middle boxes. Middleboxes (esp. NAT) may **change** the following field in the IP header.

- IP source address
- IP destination address
- Source port
- Destination port
- Sequence number
- ACK number

They especially can remove the `MP_CAPABLE` flag, which leads to unsuccessful connection establishment → normal TCP connection established.

Also, non-sequential ACKs (when only looking at one path) may be dropped.

Middleboxes may also change all other possible fields, but those changes are not common.

2 Design Patterns

This chapter describes, how to design networking protocols properly, such that they can be extended and worked with nicely.

General design principles are

- Simplicity
- Modularity
- Well-foremdness
- Robustness
- Consistency

These lead to the 10 rules of design.

Definition: 10 rules of design

There are 10 rules, that every desing process regarding protocols must follow.

1. Define the problem well
2. Define the service first
3. Design external functionality first, then internal
4. Keep it simple
5. Do not connect what's independent
6. Don't impose irrelevant restrictions
7. Build high level prototyp and validate first
8. Implement, evaluate and **optimize** the design
9. Check equivalence of prototyp and implementation
10. Don't skip 1-7

2.1 Layering

Layering describes a modular structure of a protocol, each layer being responsible for a distinct task.

Advantages

- Smaller subproblems to handle on each layer
- Implementation as modules
- Exchangeable modules
- Reusable modules

Disadvantages

- Information is hidden → performance loss
- Redundantly implemented functionality on different layers

Cross-layer communication can help with redundancy, but is not common, as it is hard to change.