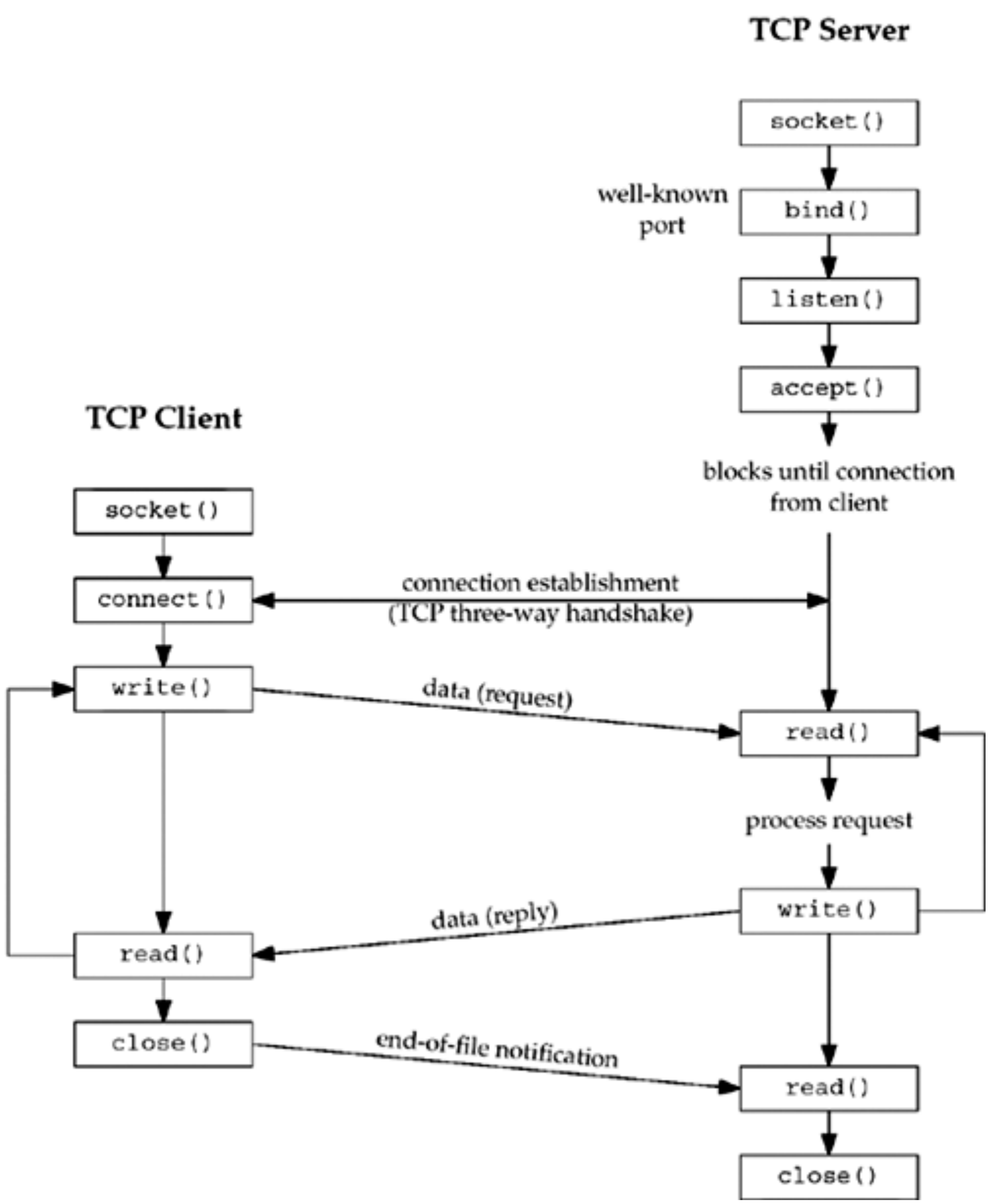
4.1 Introduction

This chapter describes the elementary socket functions required to write a complete TCP client and server. We will first describe all the elementary socket functions that we will be using and then develop the client and server in the next chapter. We will work with this client and server throughout the text, enhancing it many times (Figures 1.12 and 1.13).

We will also describe concurrent servers, a common Unix technique for providing concurrency when numerous clients are connected to the same server at the same time. Each client connection causes the server to `fork` a new process just for that client. In this chapter, we consider only the one-*process*-per-client model using `fork`, but we will consider a different one-*thread*-per-client model when we describe threads in Chapter 26.

Figure 4.1 shows a timeline of the typical scenario that takes place between a TCP client and server. First, the server is started, then sometime later, a client is started that connects to the server. We assume that the client sends a request to the server, the server processes the request, and the server sends a reply back to the client. This continues until the client closes its end of the connection, which sends an end-of-file notification to the server. The server then closes its end of the connection and either terminates or waits for a new client connection.

Figure 4.1. Socket functions for elementary TCP client/server.



4.2 `socket` Function

To perform network I/O, the first thing a process must do is call the `socket` function, specifying the type of communication protocol desired (TCP using IPv4, UDP using IPv6, Unix domain stream protocol, etc.).

```
#include <sys/socket.h>

int socket (int family, int type, int protocol);
```

Returns: non-negative descriptor if OK, -1 on error

family specifies the protocol family and is one of the constants shown in Figure 4.2. This argument is often referred to as *domain* instead of *family*. The socket *type* is one of the constants shown in Figure 4.3. The *protocol* argument to the `socket` function should be set to the specific protocol type found in Figure 4.4, or 0 to select the system's default for the given combination of *family* and *type*.

Figure 4.2. Protocol *family* constants for `socket` function.

family	Description
AF_INET	IPv4 protocols
AF_INET6	IPv6 protocols
AF_LOCAL	Unix domain protocols (Chapter 15)
AF_ROUTE	Routing sockets (Chapter 18)
AF_KEY	Key socket (Chapter 19)

Figure 4.3. *type* of socket for `socket` function.

type	Description
SOCK_STREAM	stream socket
SOCK_DGRAM	datagram socket
SOCK_SEQPACKET	sequenced packet socket
SOCK_RAW	raw socket

Figure 4.4. *protocol* of sockets for `AF_INET` or `AF_INET6`.

Protocol	Description
IPPROTO_TCP	TCP transport protocol
IPPROTO_UDP	UDP transport protocol
IPPROTO_SCTP	SCTP transport protocol

Not all combinations of socket *family* and *type* are valid. Figure 4.5 shows the valid combinations, along with the actual protocols that are valid for each pair. The boxes marked "Yes" are valid but do not have handy acronyms. The blank boxes are not supported.

Figure 4.5. Combinations of *family* and *type* for the `socket` function.

	AF_INET	AF_INET6	AF_LOCAL	AF_ROUTE	AF_KEY
SOCK_STREAM	TCP|SCTP	TCP|SCTP	Yes		
SOCK_DGRAM	UDP	UDP	Yes		
SOCK_SEQPACKET	SCTP	SCTP	Yes		
SOCK_RAW	IPv4	IPv6		Yes	Yes

You may also encounter the corresponding `PF_`*xxx* constant as the first argument to `socket`. We will say more about this at the end of this section.

> We note that you may encounter `AF_UNIX` (the historical Unix name) instead of `AF_LOCAL` (the POSIX name), and we will say more about this in Chapter 15.
>
> There are other values for the *family* and *type* arguments. For example, 4.4BSD supports both `AF_NS` (the Xerox NS protocols, often called XNS) and `AF_ISO` (the OSI protocols). Similarly, the *type* of `SOCK_SEQPACKET`, a sequenced-packet socket, is implemented by both the Xerox NS protocols and the OSI protocols, and we will describe its use with SCTP in Section 9.2. But, TCP is a byte stream protocol, and supports only `SOCK_STREAM` sockets.
>
> Linux supports a new socket type, `SOCK_PACKET`, that provides access to the datalink, similar to BPF and DLPI in Figure 2.1. We will say more about this in Chapter 29.
>
> The key socket, `AF_KEY`, is newer than the others. It provides support for cryptographic security. Similar to the way that a routing socket (`AF_ROUTE`) is an interface to the kernel's routing table, the key socket is an interface into the kernel's key table. See Chapter 19 for details.

On success, the `socket` function returns a small non-negative integer value, similar to a file descriptor. We call this a *socket descriptor*, or a *sockfd*. To obtain this socket descriptor, all we have specified is a protocol family (IPv4, IPv6, or Unix) and the socket type (stream, datagram, or raw). We have not yet specified either the local protocol address or the foreign protocol address.

`AF_`*xxx* Versus `PF_`*xxx*

The "`AF_`" prefix stands for "address family" and the "`PF_`" prefix stands for "protocol family." Historically, the intent was that a single protocol family might support multiple address families and that the `PF_` value was used to create the socket and the `AF_` value was used in socket address structures. But in actuality, a protocol family supporting multiple address families has never been supported and the `<sys/socket.h>` header defines the `PF_` value for a given protocol to be equal to the `AF_` value for that protocol. While there is no guarantee that this equality between the two will always be true, should anyone change this for existing protocols, lots of existing code would break. To conform to existing coding practice, we use only the `AF_` constants in this text, although you may encounter the `PF_` value, mainly in calls to `socket`.

> Looking at 137 programs that call `socket` in the BSD/OS 2.1 release shows 143 calls that specify the `AF_` value and only 8 that specify the `PF_` value.
>
> Historically, the reason for the similar sets of constants with the `AF_` and `PF_` prefixes goes back to 4.1cBSD [Lanciani 1996] and a version of the `socket` function that predates the one we are describing (which appeared with 4.2BSD). The 4.1cBSD version of `socket` took four arguments, one of which was a pointer to a `sockproto` structure. The first member of this structure was named `sp_family` and its value was one of the `PF_` values. The second member, `sp_protocol`, was a protocol number, similar to the third argument to `socket` today. Specifying this structure was the only way to specify the protocol family. Therefore, in this early system, the `PF_` values were used as structure tags to specify the protocol family in the `sockproto` structure, and the `AF_` values were used as structure tags to specify the address family in the socket address structures. The `sockproto` structure is still in 4.4BSD (pp. 626–627 of TCPv2), but is only used internally by the kernel. The original definition had the comment "protocol family" for the `sp_family` member, but this has been changed to "address family" in the 4.4BSD source code.
>
> To confuse this difference between the `AF_` and `PF_` constants even more, the Berkeley kernel data structure that contains the value that is compared to the first argument to `socket` (the `dom_family` member of the `domain` structure, p. 187 of TCPv2) has the comment that it contains an `AF_` value. But, some of the `domain` structures within the kernel are initialized to the corresponding `AF_` value (p. 192 of TCPv2) while others are initialized to the `PF_` value (p. 646 of TCPv2 and p. 229 of TCPv3).
>
> As another historical note, the 4.2BSD man page for `socket`, dated July 1983, calls its first argument *af* and lists the possible values as the `AF_` constants.
>
> Finally, we note that the POSIX standard specifies that the first argument to `socket` be a `PF_` value, and the `AF_` value be used for a socket address structure. But, it then defines only one family value in the `addrinfo` structure (Section 11.6), intended for use in either a call to `socket` or in a socket address structure!

[Team LiB] PREVIOUS NEXT

4.3 `connect` Function

The `connect` function is used by a TCP client to establish a connection with a TCP server.

```
#include <sys/socket.h>

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

Returns: 0 if OK, -1 on error

sockfd is a socket descriptor returned by the `socket` function. The second and third arguments are a pointer to a socket address structure and its size, as described in Section 3.3. The socket address structure must contain the IP address and port number of the server. We saw an example of this function in Figure 1.5.

The client does not have to call `bind` (which we will describe in the next section) before calling `connect:` the kernel will choose both an ephemeral port and the source IP address if necessary.

In the case of a TCP socket, the `connect` function initiates TCP's three-way handshake (Section 2.6). The function returns only when the connection is established or an error occurs. There are several different error returns possible.

1. If the client TCP receives no response to its SYN segment, `ETIMEDOUT` is returned. 4.4BSD, for example, sends one SYN when `connect` is called, another 6 seconds later, and another 24 seconds later (p. 828 of TCPv2). If no response is received after a total of 75 seconds, the error is returned.

   Some systems provide administrative control over this timeout; see Appendix E of TCPv1.

2. If the server's response to the client's SYN is a reset (RST), this indicates that no process is waiting for connections on the server host at the port specified (i.e., the server process is probably not running). This is a *hard error* and the error `ECONNREFUSED` is returned to the client as soon as the RST is received.

   An RST is a type of TCP segment that is sent by TCP when something is wrong. Three conditions that generate an RST are: when a SYN arrives for a port that has no listening server (what we just described), when TCP wants to abort an existing connection, and when TCP receives a segment for a connection that does not exist. (TCPv1 [pp. 246–250] contains additional information.)

3. If the client's SYN elicits an ICMP "destination unreachable" from some intermediate router, this is considered a *soft error*. The client kernel saves the message but keeps sending SYNs with the same time between each SYN as in the first scenario. If no response is received after some fixed amount of time (75 seconds for 4.4BSD), the saved ICMP error is returned to the process as either `EHOSTUNREACH` or `ENETUNREACH`. It is also possible that the remote system is not reachable by any route in the local system's forwarding table, or that the `connect` call returns without waiting at all.

   > Many earlier systems, such as 4.2BSD, incorrectly aborted the connection establishment attempt when the ICMP "destination unreachable" was received. This is wrong because this ICMP error can indicate a transient condition. For example, it could be that the condition is caused by a routing problem that will be corrected.
   >
   > Notice that `ENETUNREACH` is not listed in Figure A.15, even when the error indicates that the destination network is unreachable. Network unreachables are considered obsolete, and applications should just treat `ENETUNREACH` and `EHOSTUNREACH` as the same error.

We can see these different error conditions with our simple client from Figure 1.5. We first specify the local host (127.0.0.1), which is running the daytime server, and see the output.

```
solaris % daytimetcpcli 127.0.0.1
Sun Jul 27 22:01:51 2003
```

To see a different format for the returned reply, we specify a different machine's IP address (in this example, the IP address of the HP-UX machine).

```
solaris % daytimetcpcli 192.6.38.100
Sun Jul 27 22:04:59 PDT 2003
```

Next, we specify an IP address that is on the local subnet (192.168.1/24) but the host ID (100) is nonexistent. That is, there is no host on the subnet with a host ID of 100, so when the client host sends out ARP requests (asking for that host to respond with its hardware address), it will never receive an ARP reply.

```
solaris % daytimetcpcli 192.168.1.100
connect error: Connection timed out
```

We only get the error after the `connect` times out (around four minutes with Solaris 9). Notice that our `err_sys` function prints the humanreadable string associated with the ETIMEDOUT error.

Our next example is to specify a host (a local router) that is not running a daytime server.

```
solaris % daytimetcpcli 192.168.1.5
connect error: Connection refused
```

The server responds immediately with an RST.

Our final example specifies an IP address that is not reachable on the Internet. If we watch the packets with tcpdump, we see that a router six hops away returns an ICMP host unreachable error.

```
solaris % daytimetcpcli 192.3.4.5
connect error: No route to host
```

As with the ETIMEDOUT error, in this example, connect returns the EHOSTUNREACH error only after waiting its specified amount of time.

In terms of the TCP state transition diagram (Figure 2.4), connect moves from the CLOSED state (the state in which a socket begins when it is created by the socket function) to the SYN_SENT state, and then, on success, to the ESTABLISHED state. If connect fails, the socket is no longer usable and must be closed. We cannot call connect again on the socket. In Figure 11.10, we will see that when we call connect in a loop, trying each IP address for a given host until one works, each time connect fails, we must close the socket descriptor and call socket again.

4.4 bind Function

The bind function assigns a local protocol address to a socket. With the Internet protocols, the protocol address is the combination of either a 32-bit IPv4 address or a 128-bit IPv6 address, along with a 16-bit TCP or UDP port number.

```
#include <sys/socket.h>

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

Returns: 0 if OK,-1 on error

> Historically, the man page description of bind has said "bind assigns a name to an unnamed socket." The use of the term "name" is confusing and gives the connotation of domain names (Chapter 11) such as foo.bar.com. The bind function has nothing to do with names. bind assigns a protocol address to a socket, and what that protocol address means depends on the protocol.

The second argument is a pointer to a protocol-specific address, and the third argument is the size of this address structure. With TCP, calling bind lets us specify a port number, an IP address, both, or neither.

- Servers bind their well-known port when they start. We saw this in Figure 1.9. If a TCP client or server does not do this, the kernel chooses an ephemeral port for the socket when either connect or listen is called. It is normal for a TCP client to let the kernel choose an ephemeral port, unless the application requires a reserved port (Figure 2.10), but it is rare for a TCP server to let the kernel choose an ephemeral port, since servers are known by their well-known port.

  > Exceptions to this rule are Remote Procedure Call (RPC) servers. They normally let the kernel choose an ephemeral port for their listening socket since this port is then registered with the RPC port mapper. Clients have to contact the port mapper to obtain the ephemeral port before they can connect to the server. This also applies to RPC servers using UDP.

- A process can bind a specific IP address to its socket. The IP address must belong to an interface on the host. For a TCP client, this assigns the source IP address that will be used for IP datagrams sent on the socket. For a TCP server, this restricts the socket to receive incoming client connections destined only to that IP address.

  Normally, a TCP client does not bind an IP address to its socket. The kernel chooses the source IP address when the socket is connected, based on the outgoing interface that is used, which in turn is based on the route required to reach the server (p. 737 of TCPv2).

  If a TCP server does not bind an IP address to its socket, the kernel uses the destination IP address of the client's SYN as the server's source IP address (p. 943 of TCPv2).

As we said, calling bind lets us specify the IP address, the port, both, or neither. Figure 4.6 summarizes the values to which we set sin_addr and sin_port, or sin6_addr and sin6_port, depending on the desired result.

Figure 4.6. Result when specifying IP address and/or port number to bind.

Process specifies		Result
IP address	port	
Wildcard	0	Kernel chooses IP address and port
Wildcard	nonzero	Kernel chooses IP address, process specifies port
Local IP address	0	Process specifies IP address, kernel chooses port
Local IP address	nonzero	Process specifies IP address and port

If we specify a port number of 0, the kernel chooses an ephemeral port when bind is called. But if we specify a wildcard IP address, the kernel does not choose the local IP address until either the socket is connected (TCP) or a datagram is sent on the socket (UDP).

With IPv4, the *wildcard* address is specified by the constant INADDR_ANY, whose value is normally 0. This tells the kernel to choose the IP address. We saw the use of this in Figure 1.9 with the assignment

```
struct sockaddr_in   servaddr;
servaddr.sin_addr.s_addr = htonl (INADDR_ANY);     /* wildcard */
```

While this works with IPv4, where an IP address is a 32-bit value that can be represented as a simple numeric constant (0 in this case), we cannot use this technique with IPv6, since the 128-bit IPv6 address is stored in a structure. (In C we cannot represent a constant structure on the right-hand side of an assignment.) To solve this problem, we write

```
struct sockaddr_in6    serv;

serv.sin6_addr = in6addr_any;     /* wildcard */
```

The system allocates and initializes the `in6addr_any` variable to the constant `IN6ADDR_ANY_INIT`. The `<netinet/in.h>` header contains the `extern` declaration for `in6addr_any`.

The value of `INADDR_ANY` (0) is the same in either network or host byte order, so the use of `htonl` is not really required. But, since all the `INADDR_` constants defined by the `<netinet/in.h>` header are defined in host byte order, we should use `htonl` with any of these constants.

If we tell the kernel to choose an ephemeral port number for our socket, notice that `bind` does not return the chosen value. Indeed, it cannot return this value since the second argument to `bind` has the `const` qualifier. To obtain the value of the ephemeral port assigned by the kernel, we must call `getsockname` to return the protocol address.

A common example of a process binding a non-wildcard IP address to a socket is a host that provides Web servers to multiple organizations (Section 14.2 of TCPv3). First, each organization has its own domain name, such as `www.`*organization*`.com`. Next, each organization's domain name maps into a different IP address, but typically on the same subnet. For example, if the subnet is 198.69.10, the first organization's IP address could be 198.69.10.128, the next 198.69.10.129, and so on. All these IP addresses are then *aliased* onto a single network interface (using the `alias` option of the `ifconfig` command on 4.4BSD, for example) so that the IP layer will accept incoming datagrams destined for any of the aliased addresses. Finally, one copy of the HTTP server is started for each organization and each copy `binds` only the IP address for that organization.

> An alternative technique is to run a single server that binds the wildcard address. When a connection arrives, the server calls `getsockname` to obtain the destination IP address from the client, which in our discussion above could be 198.69.10.128, 198.69.10.129, and so on. The server then handles the client request based on the IP address to which the connection was issued.
>
> One advantage in binding a non-wildcard IP address is that the demultiplexing of a given destination IP address to a given server process is then done by the kernel.
>
> We must be careful to distinguish between the interface on which a packet arrives versus the destination IP address of that packet. In Section 8.8, we will talk about the weak end system model and the strong end system model. Most implementations employ the former, meaning it is okay for a packet to arrive with a destination IP address that identifies an interface other than the interface on which the packet arrives. (This assumes a multihomed host.) Binding a non-wildcard IP address restricts the datagrams that will be delivered to the socket based only on the destination IP address. It says nothing about the arriving interface, unless the host employs the strong end system model.

A common error from `bind` is `EADDRINUSE` ("Address already in use"). We will say more about this in Section 7.5 when we talk about the `SO_REUSEADDR` and `SO_REUSEPORT` socket options.

4.5 listen Function

The listen function is called only by a TCP server and it performs two actions:

1. When a socket is created by the socket function, it is assumed to be an active socket, that is, a client socket that will issue a connect. The listen function converts an unconnected socket into a passive socket, indicating that the kernel should accept incoming connection requests directed to this socket. In terms of the TCP state transition diagram (Figure 2.4), the call to listen moves the socket from the CLOSED state to the LISTEN state.

2. The second argument to this function specifies the maximum number of connections the kernel should queue for this socket.

```
#include <sys/socket.h>

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

Returns: 0 if OK, -1 on error

This function is normally called after both the socket and bind functions and must be called before calling the accept function.

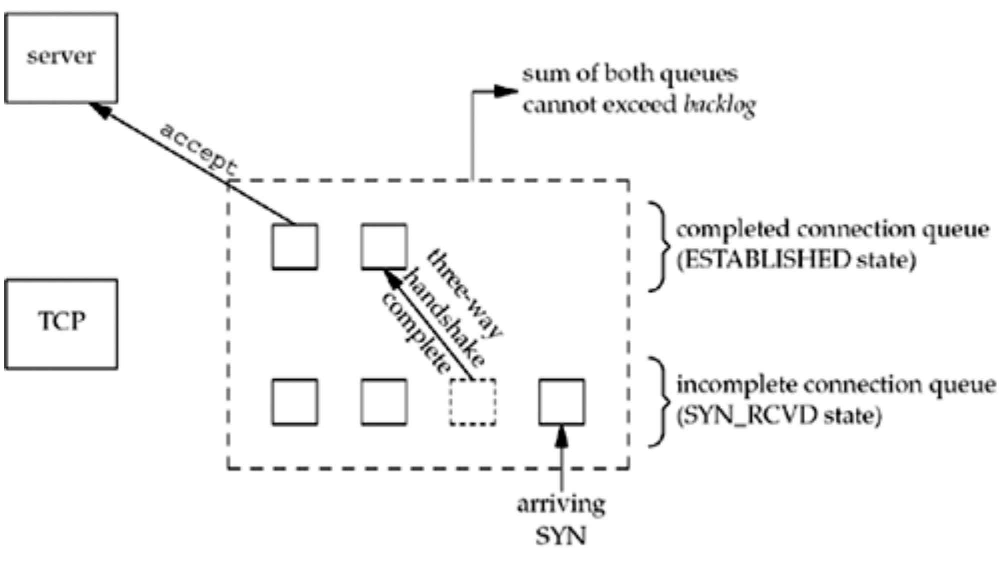
To understand the *backlog* argument, we must realize that for a given listening socket, the kernel maintains two queues:

1. An *incomplete connection queue*, which contains an entry for each SYN that has arrived from a client for which the server is awaiting completion of the TCP three-way handshake. These sockets are in the SYN_RCVD state (Figure 2.4).

2. A *completed connection queue*, which contains an entry for each client with whom the TCP three-way handshake has completed. These sockets are in the ESTABLISHED state (Figure 2.4).

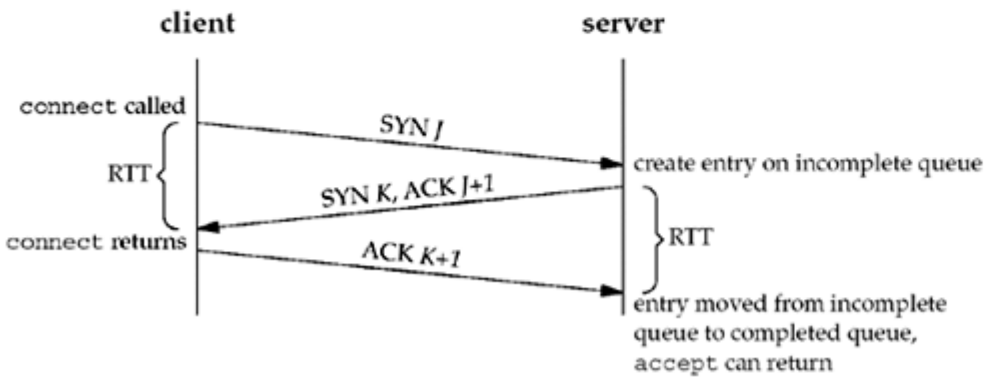
Figure 4.7 depicts these two queues for a given listening socket.

Figure 4.7. The two queues maintained by TCP for a listening socket.



When an entry is created on the incomplete queue, the parameters from the listen socket are copied over to the newly created connection. The connection creation mechanism is completely automatic; the server process is not involved. Figure 4.8 depicts the packets exchanged during the connection establishment with these two queues.

Figure 4.8. TCP three-way handshake and the two queues for a listening socket.



When a SYN arrives from a client, TCP creates a new entry on the incomplete queue and then responds with the second segment of the three-way handshake: the server's SYN with an ACK of the client's SYN (Section 2.6). This entry will remain on the incomplete queue until the third segment of the three-way handshake arrives (the client's ACK of the server's SYN), or until the entry times out. (Berkeley-derived implementations have a timeout of 75 seconds for these incomplete entries.) If the three-way handshake completes normally, the entry moves from the incomplete queue to the end of the completed queue. When the process calls `accept`, which we will describe in the next section, the first entry on the completed queue is returned to the process, or if the queue is empty, the process is put to sleep until an entry is placed onto the completed queue.

There are several points to consider regarding the handling of these two queues.

- The *backlog* argument to the `listen` function has historically specified the maximum value for the sum of both queues.

  There has never been a formal definition of what the *backlog* means. The 4.2BSD man page says that it "defines the maximum length the queue of pending connections may grow to." Many man pages and even the POSIX specification copy this definition verbatim, but this definition does not say whether a pending connection is one in the SYN_RCVD state, one in the ESTABLISHED state that has not yet been accepted, or either. The historical definition in this bullet is the Berkeley implementation, dating back to 4.2BSD, and copied by many others.

- Berkeley-derived implementations add a fudge factor to the *backlog*: It is multiplied by 1.5 (p. 257 of TCPv1 and p. 462 of TCPV2). For example, the commonly specified *backlog* of 5 really allows up to 8 queued entries on these systems, as we show in Figure 4.10.

  The reason for adding this fudge factor appears lost to history [Joy 1994]. But if we consider the *backlog* as specifying the maximum number of completed connections that the kernel will queue for a socket ([Borman 1997b], as discussed shortly), then the reason for the fudge factor is to take into account incomplete connections on the queue.

- Do not specify a *backlog* of 0, as different implementations interpret this differently (Figure 4.10). If you do not want any clients connecting to your listening socket, close the listening socket.

- Assuming the three-way handshake completes normally (i.e., no lost segments and no retransmissions), an entry remains on the incomplete connection queue for one RTT, whatever that value happens to be between a particular client and server. Section 14.4 of TCPv3 shows that for one Web server, the median RTT between many clients and the server was 187 ms. (The median is often used for this statistic, since a few large values can noticeably skew the mean.)

- Historically, sample code always shows a *backlog* of 5, as that was the maximum value supported by 4.2BSD. This was adequate in the 1980s when busy servers would handle only a few hundred connections per day. But with the growth of the World Wide Web (WWW), where busy servers handle millions of connections per day, this small number is completely inadequate (pp. 187–192 of TCPv3). Busy HTTP servers must specify a much larger *backlog*, and newer kernels must support larger values.

  Many current systems allow the administrator to modify the maximum value for the *backlog*.

- A problem is: What value should the application specify for the *backlog*, since 5 is often inadequate? There is no easy answer to this. HTTP servers now specify a larger value, but if the value specified is a constant in the source code, to increase the constant requires recompiling the server. Another method is to assume some default but allow a command-line option or an environment variable to override the default. It is always acceptable to specify a value that is larger than supported by the kernel, as the kernel should silently truncate the value to the maximum value that it supports, without returning an error (p. 456 of TCPv2).

  We can provide a simple solution to this problem by modifying our wrapper function for the `listen` function. Figure 4.9 shows the actual code. We allow the environment variable `LISTENQ` to override the value specified by the caller.

Figure 4.9 Wrapper function for `listen` that allows an environment variable to specify *backlog*.

lib/wrapsock.c

```
137 void
138 Listen (int fd, int backlog)
139 {
```

```
    char    *ptr;

        /* can override 2nd argument with environment variable */
    if ( (ptr = getenv("LISTENQ")) != NULL)
        backlog = atoi (ptr);

    if (listen (fd, backlog) < 0)
        err_sys ("listen error");
}
```

- Manuals and books have historically said that the reason for queuing a fixed number of connections is to handle the case of the server process being busy between successive calls to `accept`. This implies that of the two queues, the completed queue should normally have more entries than the incomplete queue. Again, busy Web servers have shown that this is false. The reason for specifying a large *backlog* is because the incomplete connection queue can grow as client SYNs arrive, waiting for completion of the three-way handshake.

- If the queues are full when a client SYN arrives, TCP ignores the arriving SYN (pp. 930–931 of TCPv2); it does not send an RST. This is because the condition is considered temporary, and the client TCP will retransmit its SYN, hopefully finding room on the queue in the near future. If the server TCP immediately responded with an RST, the client's `connect` would return an error, forcing the application to handle this condition instead of letting TCP's normal retransmission take over. Also, the client could not differentiate between an RST in response to a SYN meaning "there is no server at this port" versus "there is a server at this port but its queues are full."

  Some implementations do send an RST when the queue is full. This behavior is incorrect for the reasons stated above, and unless your client specifically needs to interact with such a server, it's best to ignore this possibility. Coding to handle this case reduces the robustness of the client and puts more load on the network in the normal RST case, where the port really has no server listening on it.

- Data that arrives after the three-way handshake completes, but before the server calls `accept`, should be queued by the server TCP, up to the size of the connected socket's receive buffer.

Figure 4.10 shows the actual number of queued connections provided for different values of the *backlog* argument for the various operating systems in Figure 1.16. For seven different operating systems there are five distinct columns, showing the variety of interpretations about what *backlog* means!

Figure 4.10. Actual number of queued connections for values of *backlog*.

	Maximum actual number of queued connections				
backlog	MacOS 10.2.6 AIX 5.1	Linux 2.4.7	HP-UX 11.11	FreeBSD 4.8 FreeBSD 5.1	Solaris 2.9
0	1	3	1	1	1
1	2	4	1	2	2
2	4	5	3	3	4
3	5	6	4	4	5
4	7	7	6	5	6
5	8	8	7	6	8
6	10	9	9	7	10
7	11	10	10	8	11
8	13	11	12	9	13
9	14	12	13	10	14
10	16	13	15	11	16
11	17	14	16	12	17
12	19	15	18	13	19
13	20	16	19	14	20
14	22	17	21	15	22

AIX and MacOS have the traditional Berkeley algorithm, and Solaris seems very close to that algorithm as well. FreeBSD just adds one to *backlog*.

The program to measure these values is shown in the solution for Exercise 15.4.

As we said, historically the *backlog* has specified the maximum value for the sum of both queues. During 1996, a new type of attack was launched on the Internet called *SYN flooding* [CERT 1996b]. The hacker writes a program to send SYNs at a high rate to the victim, filling the incomplete connection queue for one or more TCP ports. (We use the term *hacker* to mean the attacker, as described in [Cheswick, Bellovin, and Rubin 2003].) Additionally, the source IP address of each SYN is set to a random number (this is called *IP spoofing*) so that the server's SYN/ACK goes nowhere. This also prevents the server from knowing the real IP address of the hacker. By filling the incomplete queue with bogus SYNs, legitimate SYNs are not queued, providing a *denial of service* to legitimate clients. There are two commonly used methods of handling these attacks, summarized in [Borman 1997b]. But what is most interesting in this note is revisiting what the `listen` *backlog* really means. It should specify the maximum number of *completed* connections for a given socket that the kernel will queue. The purpose of having a limit on these completed connections is to stop the kernel from accepting new connection requests for a given socket when the application is not accepting them (for whatever reason). If a system implements this interpretation, as does BSD/OS 3.0, then the application need not specify huge *backlog* values just because the server handles lots of client requests (e.g., a busy Web server) or to provide protection against SYN flooding. The kernel handles lots of incomplete connections, regardless of whether they are legitimate or from a hacker. But even with this interpretation, scenarios do occur where the traditional value of 5 is inadequate.

4.6 `accept` Function

`accept` is called by a TCP server to return the next completed connection from the front of the completed connection queue (Figure 4.7). If the completed connection queue is empty, the process is put to sleep (assuming the default of a blocking socket).

```
#include <sys/socket.h>

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

Returns: non-negative descriptor if OK, -1 on error

The *cliaddr* and *addrlen* arguments are used to return the protocol address of the connected peer process (the client). *addrlen* is a value-result argument (Section 3.3): Before the call, we set the integer value referenced by **addrlen* to the size of the socket address structure pointed to by *cliaddr*; on return, this integer value contains the actual number of bytes stored by the kernel in the socket address structure.

If `accept` is successful, its return value is a brand-new descriptor automatically created by the kernel. This new descriptor refers to the TCP connection with the client. When discussing `accept`, we call the first argument to `accept` the *listening socket* (the descriptor created by `socket` and then used as the first argument to both `bind` and `listen`), and we call the return value from `accept` the *connected socket*. It is important to differentiate between these two sockets. A given server normally creates only one listening socket, which then exists for the lifetime of the server. The kernel creates one connected socket for each client connection that is `accept`ed (i.e., for which the TCP three-way handshake completes). When the server is finished serving a given client, the connected socket is closed.

This function returns up to three values: an integer return code that is either a new socket descriptor or an error indication, the protocol address of the client process (through the *cliaddr* pointer), and the size of this address (through the *addrlen* pointer). If we are not interested in having the protocol address of the client returned, we set both *cliaddr* and *addrlen* to null pointers.

Figure 1.9 shows these points. The connected socket is closed each time through the loop, but the listening socket remains open for the life of the server. We also see that the second and third arguments to `accept` are null pointers, since we were not interested in the identity of the client.

Example: Value-Result Arguments

We will now show how to handle the value-result argument to `accept` by modifying the code from Figure 1.9 to print the IP address and port of the client. We show this in Figure 4.11.

Figure 4.11 Daytime server that prints client IP address and port

intro/daytimetcpsrv1.c

```
#include    "unp.h" 2
#include    <time.h>

int
main(int argc, char **argv)
{
    int     listenfd, connfd;
    socklen_t len;
    struct sockaddr_in servaddr, cliaddr;
    char    buff[MAXLINE];
    time_t  ticks;

    listenfd = Socket(AF_INET, SOCK_STREAM, 0);

    bzero(&servaddr, sizeof(servaddr));
    servaddr.sin_family = AF_INET;
    servaddr.sin_addr.s_addr = htonl(INADDR_ANY);
    servaddr.sin_port = htons(13);  /* daytime server */

    Bind(listenfd, (SA *) &servaddr, sizeof(servaddr));

    Listen(listenfd, LISTENQ);

    for ( ; ; ) {
        len = sizeof(cliaddr);
        connfd = Accept(listenfd, (SA *) &cliaddr, &len);
        printf("connection from %s, port %d\n",
               Inet_ntop(AF_INET, &cliaddr.sin_addr, buff, sizeof(buff)),
               ntohs(cliaddr.sin_port));
```

```
24         ticks = time(NULL);
25         snprintf(buff, sizeof(buff), "%.24s\r\n", ctime(&ticks));
26         Write(connfd, buff, strlen(buff));

27         Close(connfd);
28     }
29 }
```

New declarations

7–8 We define two new variables: `len`, which will be a value-result variable, and `cliaddr`, which will contain the client's protocol address.

Accept connection and print client's address

19–23 We initialize `len` to the size of the socket address structure and pass a pointer to the `cliaddr` structure and a pointer to `len` as the second and third arguments to `accept`. We call `inet_ntop` (Section 3.7) to convert the 32-bit IP address in the socket address structure to a dotted-decimal ASCII string and call `ntohs` (Section 3.4) to convert the 16-bit port number from network byte order to host byte order.

> Calling `sock_ntop` instead of `inet_ntop` would make our server more protocol-independent, but this server is already dependent on IPv4. We will show a protocol-independent version of this server in Figure 11.13.

If we run our new server and then run our client on the same host, connecting to our server twice in a row, we have the following output from the client:

```
solaris % daytimetcpcli 127.0.0.1
Thu Sep 11 12:44:00 2003
solaris % daytimetcpcli 192.168.1.20
Thu Sep 11 12:44:09 2003
```

We first specify the server's IP address as the loopback address (127.0.0.1) and then as its own IP address (192.168.1.20). Here is the corresponding server output:

```
solaris # daytimetcpsrv1
connection from 127.0.0.1, port 43388
connection from 192.168.1.20, port 43389
```

Notice what happens with the client's IP address. Since our daytime client (Figure 1.5) does not call `bind`, we said in Section 4.4 that the kernel chooses the source IP address based on the outgoing interface that is used. In the first case, the kernel sets the source IP address to the loopback address; in the second case, it sets the address to the IP address of the Ethernet interface. We can also see in this example that the ephemeral port chosen by the Solaris kernel is 43388, and then 43389 (recall Figure 2.10).

As a final point, our shell prompt for the server script changes to the pound sign (#), the commonly used prompt for the superuser. Our server must run with superuser privileges to `bind` the reserved port of 13. If we do not have superuser privileges, the call to `bind` will fail:

```
solaris % daytimetcpsrv1
bind error: Permission denied
```


PREVIOUS NEXT

4.7 `fork` and `exec` Functions

Before describing how to write a concurrent server in the next section, we must describe the Unix `fork` function. This function (including the variants of it provided by some systems) is the only way in Unix to create a new process.

```
#include <unistd.h>

pid_t fork(void);
```

Returns: 0 in child, process ID of child in parent, -1 on error

If you have never seen this function before, the hard part in understanding `fork` is that it is called *once* but it returns *twice*. It returns once in the calling process (called the parent) with a return value that is the process ID of the newly created process (the child). It also returns once in the child, with a return value of 0. Hence, the return value tells the process whether it is the parent or the child.

The reason `fork` returns 0 in the child, instead of the parent's process ID, is because a child has only one parent and it can always obtain the parent's process ID by calling `getppid`. A parent, on the other hand, can have any number of children, and there is no way to obtain the process IDs of its children. If a parent wants to keep track of the process IDs of all its children, it must record the return values from `fork`.

All descriptors open in the parent before the call to `fork` are shared with the child after `fork` returns. We will see this feature used by network servers: The parent calls `accept` and then calls `fork`. The connected socket is then shared between the parent and child. Normally, the child then reads and writes the connected socket and the parent closes the connected socket.

There are two typical uses of `fork`:

1. A process makes a copy of itself so that one copy can handle one operation while the other copy does another task. This is typical for network servers. We will see many examples of this later in the text.
2. A process wants to execute another program. Since the only way to create a new process is by calling `fork`, the process first calls `fork` to make a copy of itself, and then one of the copies (typically the child process) calls `exec` (described next) to replace itself with the new program. This is typical for programs such as shells.

The only way in which an executable program file on disk can be executed by Unix is for an existing process to call one of the six `exec` functions. (We will often refer generically to "the `exec` function" when it does not matter which of the six is called.) `exec` replaces the current process image with the new program file, and this new program normally starts at the `main` function. The process ID does not change. We refer to the process that calls `exec` as the *calling process* and the newly executed program as the *new program*.

> Older manuals and books incorrectly refer to the new program as the *new process*, which is wrong, because a new process is not created.

The differences in the six `exec` functions are: (a) whether the program file to execute is specified by a *filename* or a *pathname*; (b) whether the arguments to the new program are listed one by one or referenced through an array of pointers; and (c) whether the environment of the calling process is passed to the new program or whether a new environment is specified.

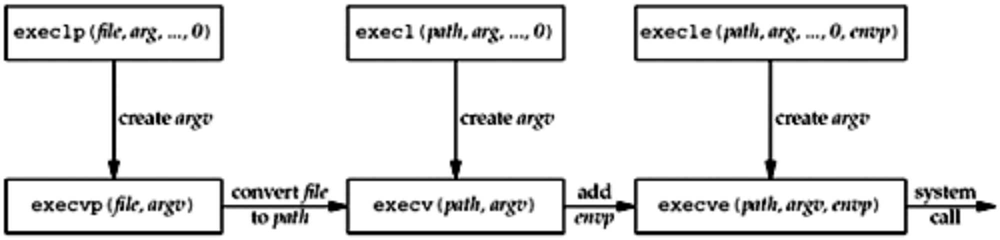
```
#include <unistd.h>

int execl (const char *pathname, const char *arg0, ... /* (char *) 0 */ );

int execv (const char *pathname, char *const argv[]);

int execle (const char *pathname, const char *arg0, ...

                                       /* (char *) 0, char *const envp[] */ );

int execve (const char *pathname, char *const argv[], char *const envp[]);

int execlp (const char *filename, const char *arg0, ... /* (char *) 0 */ );

int execvp (const char *filename, char *const argv[]);
```

All six return: -1 on error, no return on success

These functions return to the caller only if an error occurs. Otherwise, control passes to the start of the new program, normally the `main` function.

The relationship among these six functions is shown in Figure 4.12. Normally, only `execve` is a system call within the kernel and the other five are library functions that call `execve`.

Figure 4.12. Relationship among the six `exec` functions.



Note the following differences among these six functions:

1. The three functions in the top row specify each argument string as a separate argument to the `exec` function, with a null pointer terminating the variable number of arguments. The three functions in the second row have an *argv* array, containing pointers to the argument strings. This *argv* array must contain a null pointer to specify its end, since a count is not specified.
2. The two functions in the left column specify a *filename* argument. This is converted into a *pathname* using the current `PATH` environment variable. If the *filename* argument to `execlp` or `execvp` contains a slash (/) anywhere in the string, the `PATH` variable is not used. The four functions in the right two columns specify a fully qualified *pathname* argument.
3. The four functions in the left two columns do not specify an explicit environment pointer. Instead, the current value of the external variable `environ` is used for building an environment list that is passed to the new program. The two functions in the right column specify an explicit environment list. The *envp* array of pointers must be terminated by a null pointer.

Descriptors open in the process before calling `exec` normally remain open across the `exec`. We use the qualifier "normally" because this can be disabled using `fcntl` to set the `FD_CLOEXEC` descriptor flag. The `inetd` server uses this feature, as we will describe in Section 13.5.

4.8 Concurrent Servers

The server in Figure 4.11 is an *iterative server*. For something as simple as a daytime server, this is fine. But when a client request can take longer to service, we do not want to tie up a single server with one client; we want to handle multiple clients at the same time. The simplest way to write a *concurrent server* under Unix is to `fork` a child process to handle each client. Figure 4.13 shows the outline for a typical concurrent server.

Figure 4.13 Outline for typical concurrent server.

```
pid_t pid;
int   listenfd,  connfd;

listenfd = Socket( ... );

    /* fill in sockaddr_in{} with server's well-known port */
Bind(listenfd, ... );
Listen(listenfd, LISTENQ);

for ( ; ; ) {
    connfd = Accept (listenfd, ... );    /* probably blocks */

    if( (pid = Fork()) == 0) {
       Close(listenfd);    /* child closes listening socket */
       doit(connfd);       /* process the request */
       Close(connfd);      /* done with this client */
       exit(0);            /* child terminates */
    }

    Close(connfd);         /* parent closes connected socket */
}
```

When a connection is established, `accept` returns, the server calls `fork`, and the child process services the client (on `connfd`, the connected socket) and the parent process waits for another connection (on `listenfd`, the listening socket). The parent closes the connected socket since the child handles the new client.

In Figure 4.13, we assume that the function `doit` does whatever is required to service the client. When this function returns, we explicitly `close` the connected socket in the child. This is not required since the next statement calls `exit`, and part of process termination is to close all open descriptors by the kernel. Whether to include this explicit call to `close` or not is a matter of personal programming taste.

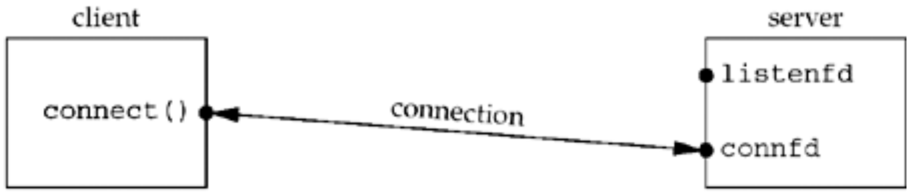
We said in Section 2.6 that calling `close` on a TCP socket causes a FIN to be sent, followed by the normal TCP connection termination sequence. Why doesn't the `close` of `connfd` in Figure 4.13 by the parent terminate its connection with the client? To understand what's happening, we must understand that every file or socket has a reference count. The reference count is maintained in the file table entry (pp. 57–60 of APUE). This is a count of the number of descriptors that are currently open that refer to this file or socket. In Figure 4.13, after `socket` returns, the file table entry associated with `listenfd` has a reference count of 1. After `accept` returns, the file table entry associated with `connfd` has a reference count of 1. But, after `fork` returns, both descriptors are shared (i.e., duplicated) between the parent and child, so the file table entries associated with both sockets now have a reference count of 2. Therefore, when the parent closes `connfd`, it just decrements the reference count from 2 to 1 and that is all. The actual cleanup and de-allocation of the socket does not happen until the reference count reaches 0. This will occur at some time later when the child closes `connfd`.

We can also visualize the sockets and connection that occur in Figure 4.13 as follows. First, Figure 4.14 shows the status of the client and server while the server is blocked in the call to `accept` and the connection request arrives from the client.

Figure 4.14. Status of client/server before call to `accept` returns.

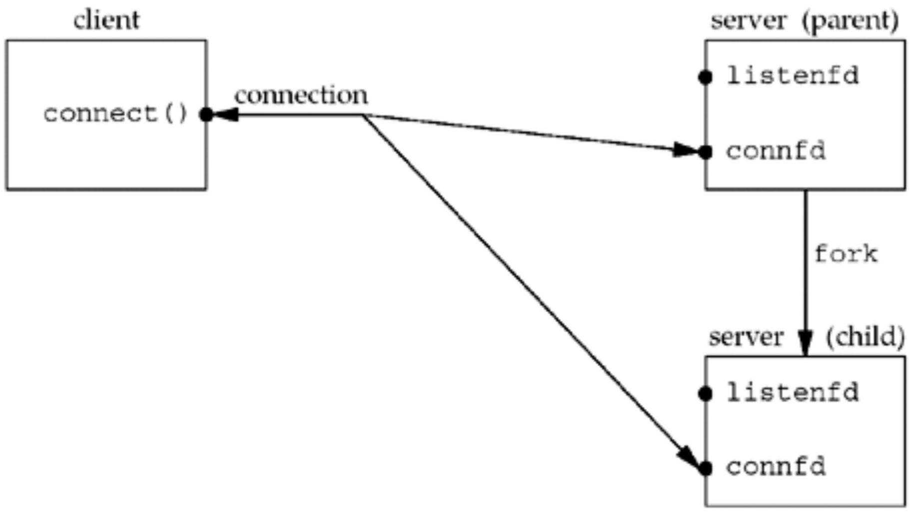
Immediately after `accept` returns, we have the scenario shown in Figure 4.15. The connection is accepted by the kernel and a new socket, `connfd`, is created. This is a connected socket and data can now be read and written across the connection.

Figure 4.15. Status of client/server after return from `accept`.



The next step in the concurrent server is to call `fork`. Figure 4.16 shows the status after `fork` returns.

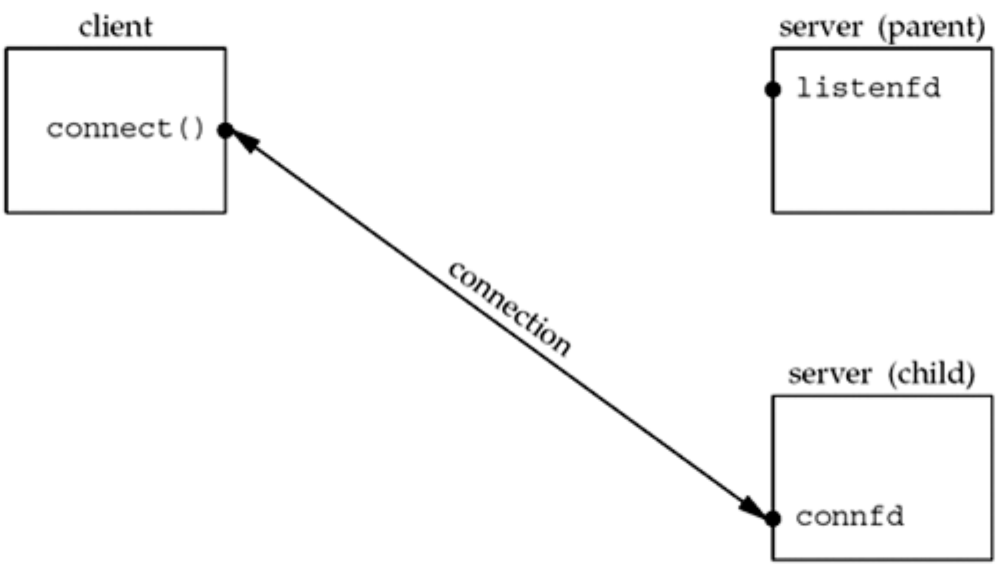
Figure 4.16. Status of client/server after `fork` returns.



Notice that both descriptors, `listenfd` and `connfd`, are shared (duplicated) between the parent and child.

The next step is for the parent to close the connected socket and the child to close the listening socket. This is shown in Figure 4.17.

Figure 4.17. Status of client/server after parent and child close appropriate sockets.



This is the desired final state of the sockets. The child is handling the connection with the client and the parent can call `accept` again on the listening socket, to handle the next client connection.

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4.9 `close` Function

The normal Unix `close` function is also used to close a socket and terminate a TCP connection.

```
#include <unistd.h>

int close (int sockfd);
```

Returns: 0 if OK, -1 on error

The default action of `close` with a TCP socket is to mark the socket as closed and return to the process immediately. The socket descriptor is no longer usable by the process: It cannot be used as an argument to `read` or `write`. But, TCP will try to send any data that is already queued to be sent to the other end, and after this occurs, the normal TCP connection termination sequence takes place (Section 2.6).

In Section 7.5, we will describe the `SO_LINGER` socket option, which lets us change this default action with a TCP socket. In that section, we will also describe what a TCP application must do to be guaranteed that the peer application has received any outstanding data.

Descriptor Reference Counts

At the end of Section 4.8, we mentioned that when the parent process in our concurrent server `close`s the connected socket, this just decrements the reference count for the descriptor. Since the reference count was still greater than 0, this call to `close` did not initiate TCP's four-packet connection termination sequence. This is the behavior we want with our concurrent server with the connected socket that is shared between the parent and child.

If we really want to send a FIN on a TCP connection, the `shutdown` function can be used (Section 6.6) instead of `close`. We will describe the motivation for this in Section 6.5.

We must also be aware of what happens in our concurrent server if the parent does not call `close` for each connected socket returned by `accept`. First, the parent will eventually run out of descriptors, as there is usually a limit to the number of descriptors that any process can have open at any time. But more importantly, none of the client connections will be terminated. When the child closes the connected socket, its reference count will go from 2 to 1 and it will remain at 1 since the parent never `close`s the connected socket. This will prevent TCP's connection termination sequence from occurring, and the connection will remain open.

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4.10 `getsockname` and `getpeername` Functions

These two functions return either the local protocol address associated with a socket (`getsockname`) or the foreign protocol address associated with a socket (`getpeername`).

```
#include <sys/socket.h>

int getsockname(int sockfd, struct sockaddr *localaddr, socklen_t *addrlen);

int getpeername(int sockfd, struct sockaddr *peeraddr, socklen_t *addrlen);
```

Both return: 0 if OK, -1 on error

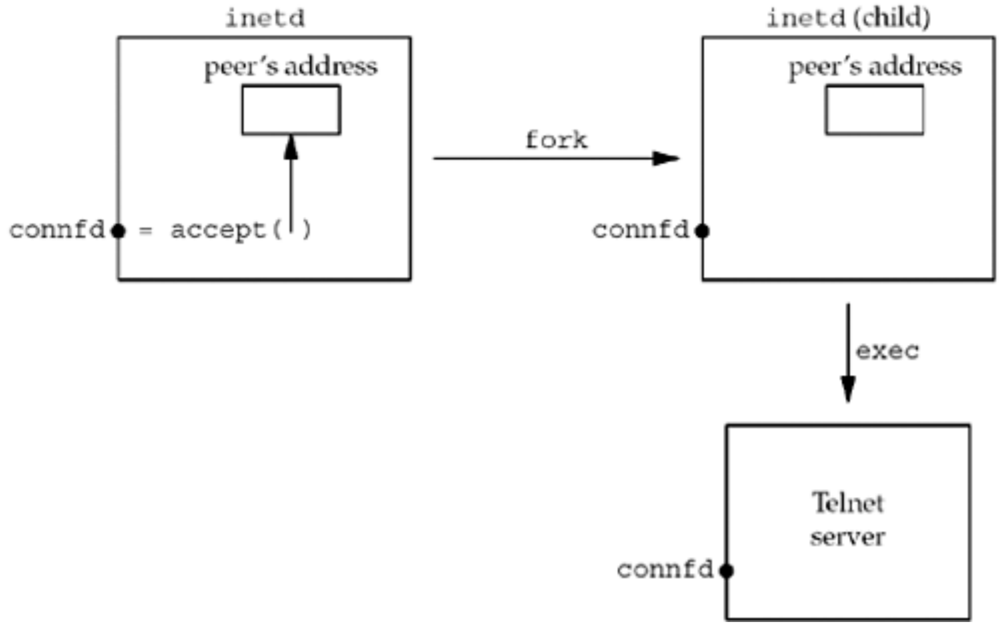
Notice that the final argument for both functions is a value-result argument. That is, both functions fill in the socket address structure pointed to by *localaddr* or *peeraddr*.

> We mentioned in our discussion of `bind` that the term "name" is misleading. These two functions return the protocol address associated with one of the two ends of a network connection, which for IPV4 and IPV6 is the combination of an IP address and port number. These functions have nothing to do with domain names (Chapter 11).

These two functions are required for the following reasons:

- After `connect` successfully returns in a TCP client that does not call `bind`, `getsockname` returns the local IP address and local port number assigned to the connection by the kernel.
- After calling `bind` with a port number of 0 (telling the kernel to choose the local port number), `getsockname` returns the local port number that was assigned.
- `getsockname` can be called to obtain the address family of a socket, as we show in Figure 4.19.
- In a TCP server that `bind`s the wildcard IP address (Figure 1.9), once a connection is established with a client (`accept` returns successfully), the server can call `getsockname` to obtain the local IP address assigned to the connection. The socket descriptor argument in this call must be that of the connected socket, and not the listening socket.
- When a server is `exec`ed by the process that calls `accept`, the only way the server can obtain the identity of the client is to call `getpeername`. This is what happens whenever `inetd` (Section 13.5) `fork`s and `exec`s a TCP server. Figure 4.18 shows this scenario. `inetd` calls `accept` (top left box) and two values are returned: the connected socket descriptor, `connfd`, is the return value of the function, and the small box we label "peer's address" (an Internet socket address structure) contains the IP address and port number of the client. `fork` is called and a child of `inetd` is created. Since the child starts with a copy of the parent's memory image, the socket address structure is available to the child, as is the connected socket descriptor (since the descriptors are shared between the parent and child). But when the child `exec`s the real server (say the Telnet server that we show), the memory image of the child is replaced with the new program file for the Telnet server (i.e., the socket address structure containing the peer's address is lost), and the connected socket descriptor remains open across the `exec`. One of the first function calls performed by the Telnet server is `getpeername` to obtain the IP address and port number of the client.

Figure 4.18. Example of `inetd` spawning a server.



Obviously the Telnet server in this final example must know the value of `connfd` when it starts. There are two common ways to do this. First, the process calling `exec` can format the descriptor number as a character string and pass it as a command-line argument to the newly `exec`ed program. Alternately, a convention can be established that a certain descriptor is always set to the connected socket before calling `exec`. The latter is what `inetd` does, always setting descriptors 0, 1, and 2 to be the connected socket.

Example: Obtaining the Address Family of a Socket

The `sockfd_to_family` function shown in Figure 4.19 returns the address family of a socket.

Figure 4.19 Return the address family of a socket.

lib/sockfd_to_family.c

```
 1 #include    "unp.h"
 2 int
 3 sockfd_to_family(int sockfd)
 4 {
 5     struct sockaddr_storage ss;
 6     socklen_t len;

 7     len = sizeof(ss);
 8     if (getsockname(sockfd, (SA *) &ss, &len) < 0)
 9         return (-1);
10     return (ss.ss_family);
11 }
```

Allocate room for largest socket address structure

5 Since we do not know what type of socket address structure to allocate, we use a `sockaddr_storage` value, since it can hold any socket address structure supported by the system.

Call `getsockname`

7–10 We call `getsockname` and return the address family.

Since the POSIX specification allows a call to `getsockname` on an unbound socket, this function should work for any open socket descriptor.

4.11 Summary

All clients and servers begin with a call to `socket`, returning a socket descriptor. Clients then call `connect`, while servers call `bind`, `listen`, and `accept`. Sockets are normally closed with the standard `close` function, although we will see another way to do this with the `shutdown` function (Section 6.6), and we will also examine the effect of the `SO_LINGER` socket option (Section 7.5).

Most TCP servers are concurrent, with the server calling `fork` for every client connection that it handles. We will see that most UDP servers are iterative. While these two models have been used successfully for many years, in Chapter 30 we will look at other server design options that use threads and processes.