18

Broadcasting

18.1 Introduction

In this chapter we describe *broadcasting* and the next chapter describes *multicasting*. All the examples in the text so far have dealt with *unicasting*: a process talking to exactly one other process. Indeed, TCP works with only unicast addresses, although UDP supports other paradigms. Figure 18.1 shows a comparison of the different types of addressing.

Type	IPv4 ?	IPv6 ?	TCP ?	UDP ?	# interfaces identified	# interfaces delivered to
unicast	•	•	•	•	one	one
anycast		•	not yet	•	a set	one in set
multicast	opt.	•		•	a set	all in set
broadcast	•			•	all	all

Figure 18.1 Different forms of addressing.

We have added *anycasting* to this figure because IPv6 plans to support it in the future. But today it is an idea that has not yet been implemented. It is described in RFC 1546 [Partridge, Mendez, and Milliken 1993].

The important points in this figure are:

- Multicasting support is optional in IPv4 but mandatory in IPv6.
- Broadcasting support is not provided in IPv6: any IPv4 application that uses broadcasting must be recoded for IPv6 to use multicasting instead.
- Broadcasting and multicasting require UDP; they do not work with TCP.

One use for broadcasting is to locate a server on the local subnet when the server is assumed to be on the local subnet but its unicast IP address is not known. This is sometimes called *resource discovery*. Another use is to minimize the network traffic on a LAN when there are multiple clients communicating with a single server. There are numerous examples of Internet applications that use broadcasting for this purpose.

- ARP (Address Resolution Protocol). Although this is a fundamental part of IPv4, and not a user application, ARP broadcasts a request on the local subnet that says "will the system with an IP address of a.b.c.d please identify yourself and tell me your hardware address."
- BOOTP (Bootstrap Protocol). The client assumes a server is on the local subnet and sends its request to the broadcast address (often 255.255.255.255, since the client doesn't yet know its IP address, its subnet mask, or the limited broadcast address of the subnet).
- NTP (Network Time Protocol). In one common scenario, an NTP client is configured with the IP address of one or more servers to use, and the client polls the servers at some frequency (every 64 seconds or longer). The client updates its clock using sophisticated algorithms based on the time-of-day returned by the servers and the round-trip time to the servers. But on a broadcast LAN, instead of making each of the clients poll a single server, the server can broadcast the current time every 64 seconds for all the clients on the local subnet, reducing the amount of network traffic.
- Routing daemons. The most commonly used routing daemon, `routed`, broadcasts its routing table on a LAN. This allows all other routers attached to the LAN to receive these routing announcements, without each router having to be configured with the IP addresses of its neighbor routers. This feature is also used (many would say "misused") by hosts on the LAN listening to these routing announcements and updating their routing tables accordingly.

We must note that multicasting can replace both uses of broadcasting (resource discovery and reducing network traffic) and we describe the problems with broadcasting later in this chapter and the next chapter.

> Broadcasting to reduce network traffic on a LAN can have an undesirable interaction with diskless systems. Assume an NTP server broadcasts the current time every 64 seconds. If the NTP daemon on all the diskless clients gets paged out of main memory during this time, then every 64 seconds each of these diskless clients receives an NTP datagram and immediately reads the NTP daemon into main memory from its disk server, also on the LAN. Every 64 seconds there is a surge of LAN activity as each diskless client pages in the NTP daemon. Periodicity of this form can be seen with broadcast applications. Fortunately the decreasing price of disk drives is making diskless systems extinct.

18.2 Broadcast Addresses

If we denote an IPv4 address as {*netid*, *subnetid*, *hostid*}, then we have four types of broadcast addresses. We denote a field containing all one bits as –1.

1. Subnet-directed broadcast address: {*netid*, *subnetid*, –1}. This addresses all the interfaces on the specified subnet. For example, if we use an 8-bit subnet ID with the class B address 128.7, then 128.7.6.255 would be the subnet-directed broadcast address for all interfaces on the 128.7.6 subnet.

   Normally routers do not forward these broadcasts (pp. 226–227 of TCPv2). In Figure 18.2 we show a router connected to the two subnets 128.7.1 and 128.7.6.

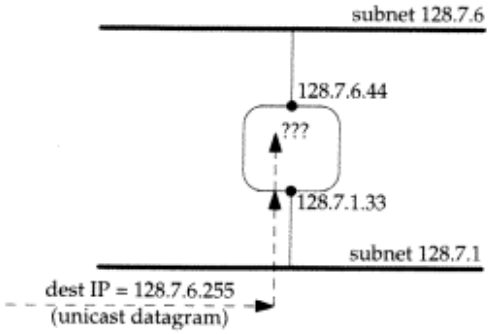


Figure 18.2 Does a router forward a subnet-directed broadcast?

   The router receives a unicast IP datagram on the 128.7.1 subnet with a destination address of 128.7.6.255 (the subnet-directed broadcast address of another interface). The router normally does not forward the datagram onto the 128.7.6 subnet. Some systems have a configuration option that allows subnet-directed broadcasts to be forwarded (Appendix E of TCPv1).

2. All-subnets-directed broadcast address: {*netid*, –1, –1}. This addresses all subnets on the specified network. This type of address is rarely, if ever, used.
3. Network-directed broadcast address: {*netid*, –1}. This type of address is used on a network that does not use subnetting, which is almost nonexistent today.
4. Limited broadcast address: {–1, –1, –1} or 255.255.255.255. Datagrams destined to this address must never be forwarded by a router.

Of the four types of broadcast addresses, the subnet-directed broadcast address is the most common today. But some older systems still send datagrams destined to 255.255.255.255. Also, some older systems do not understand a subnet-directed broadcast address and only interpret datagrams sent to 255.255.255.255 as a broadcast.

> The intent of 255.255.255.255 is to be used as the destination address during the bootstrap process by applications such as TFTP and BOOTP that do not yet know the node's IP address.
>
> The question is: what does a host do when an application sends a UDP datagram to 255.255.255.255? Most hosts allow this (assuming the process has set the `SO_BROADCAST` socket option) and convert the destination address to the subnet-directed broadcast address of the outgoing interface. BSD/OS 3.0 has a new socket option, `IP_ONESBCAST`, and when set, the destination IP address for broadcasts is set to 255.255.255.255 by the kernel, regardless of which broadcast address (subnet-directed or limited) is specified as the destination of the `sendto`.

Another question is: what does a multihomed host do when the application sends a UDP datagram to 255.255.255.255? Some systems send a single broadcast on the primary interface (the first interface that was configured) with the destination IP address set to the subnet-directed broadcast address of that interface (p. 736 of TCPv2). Other systems send one copy of the datagram out from each broadcast-capable interface. Section 3.3.6 of RFC 1122 [Braden 1989] "takes no stand" on this issue. For portability, however, if an application needs to send a broadcast out from all broadcast-capable interfaces, it should obtain the interface configuration (Section 16.6) and do one `sendto` for each broadcast-capable interface with the destination set to that interface's broadcast address.

18.3 Unicast versus Broadcast

Before looking at broadcasting let's make certain we understand the steps that take place when a UDP datagram is sent to a unicast address. Figure 18.3 shows three hosts on an Ethernet.

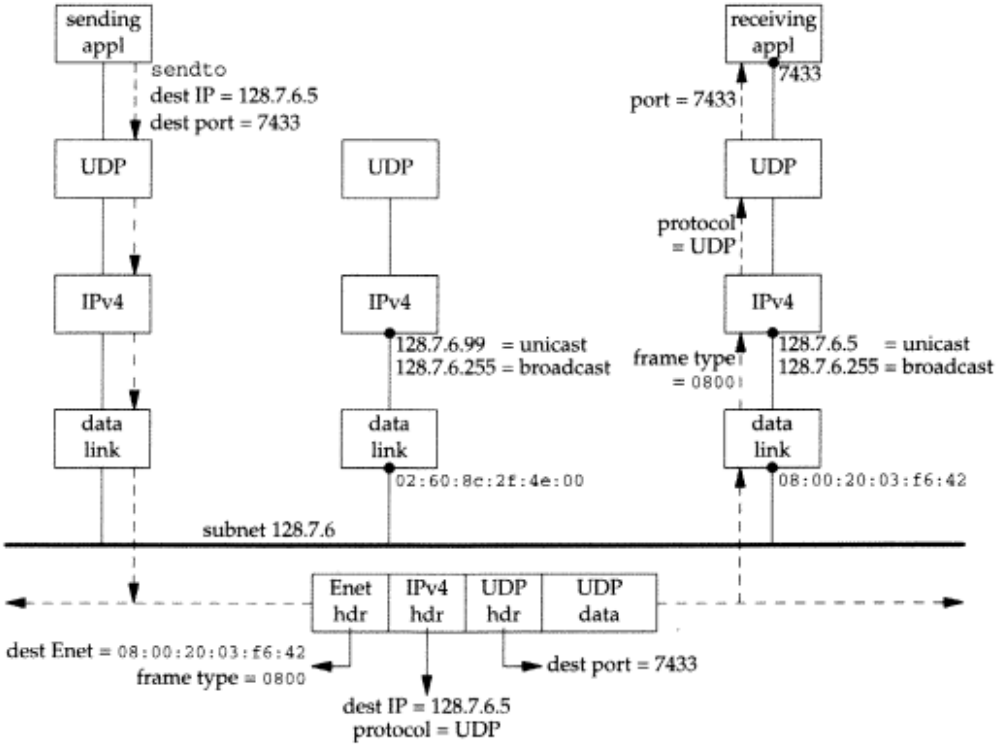


Figure 18.3 Unicast example of a UDP datagram.

The subnet address of the Ethernet is 128.7.6 with 8 bits being used as the subnet ID and 8 bits for the host ID. The application on the left host calls `sendto` on a UDP socket, sending a datagram to 128.7.6.5, port 7433. The UDP layer prepends a UDP header and passes the UDP datagram to the IP layer. IP prepends an IPv4 header, determines the outgoing interface, and in the case of an Ethernet, ARP is invoked to determine the

Ethernet address corresponding to the destination IP address: `08:00:20:03:f6:42`. The packet is then sent as an Ethernet frame with that 48-bit address as the destination Ethernet address. The frame type field of the Ethernet frame will be `0800`, specifying an IPv4 packet. The frame type for an IPv6 packet is `86dd`.

The Ethernet interface on the host in the middle sees the frame pass by and compares the destination Ethernet address with its own Ethernet address (`02:60:8c:2f:4e:00`). Since they are not equal, the interface ignores the frame. With a unicast frame there is no overhead whatsoever to this host. The interface ignores the frame.

The Ethernet interface on the host on the right also sees the frame pass by and when it compares the destination Ethernet address with its own Ethernet address, they are equal. This interface reads in the entire frame, probably generates a hardware interrupt when the frame is complete, and the device driver reads the frame from the interface memory. Since the frame type is `0800`, the packet is placed onto the IP input queue.

When the IP layer processes the packet, it first compares the destination IP address (128.7.6.5) with all of its own IP addresses. (Recall that a host can be multihomed. Also recall our discussion of the strong end system model and the weak end system model in Section 8.8.) Since the destination address is one of the host's own IP addresses, the packet is accepted.

The IP layer then looks at the protocol field in the IPv4 header, and its value will be 17 for UDP. The IP datagram is passed to UDP.

The UDP layer looks at the destination port (and possibly the source port too, if the UDP socket is connected) and in our example places the datagram onto the appropriate socket receive queue. The process is awakened, if necessary, to read the newly received datagram.

The key point in this example is that a unicast IP datagram is received by only the one host specified by the destination IP address. No other hosts on the subnet are affected.

We now consider a similar example, on the same subnet, but with the sending application writing a UDP datagram to the subnet-directed broadcast address: 128.7.6.255. Figure 18.4 shows the arrangement.

When the host on the left sends the datagram, it notices that the destination IP address is the subnet-directed broadcast address and maps this into the Ethernet address of 48 one bits: `ff:ff:ff:ff:ff:ff`. This causes *every* Ethernet interface on the subnet to receive the frame. The two hosts on the right of this figure that are running IPv4 will both receive the frame. Since the Ethernet frame type is `0800`, both hosts pass the packet to the IP layer. Since the destination IP address matches the broadcast address for each of the two hosts, and since the protocol field is 17 (UDP), both hosts pass the packet up to UDP.

The host on the right passes the UDP datagram to the application that has bound UDP port 520. Nothing special need be done by an application to receive a broadcast UDP datagram: it just creates a UDP socket and binds the application's port number to the socket. (We assume the IP address bound is `INADDR_ANY`, as is typical.)

But on the host in the middle, no application has bound UDP port 520. The host's UDP code then discards the received datagram. This host must *not* send an ICMP port unreachable, as doing so could generate a *broadcast storm*: a condition where lots of

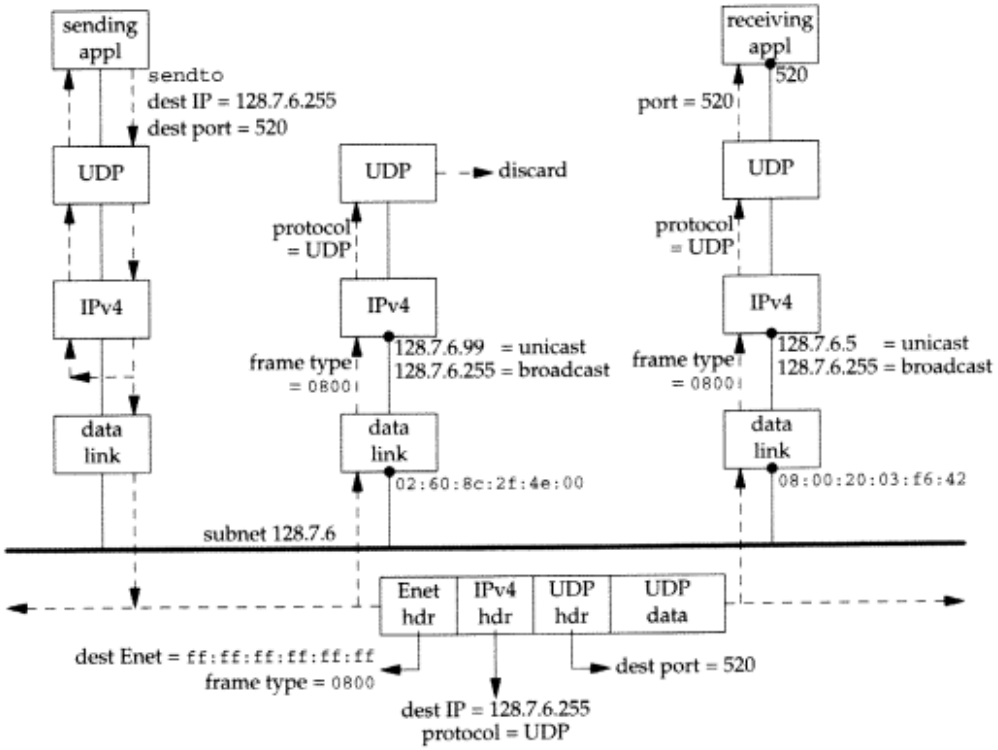


Figure 18.4 Example of a broadcast UDP datagram.

hosts on the subnet generate a response at about the same time, leading to the network being unusable for a few seconds.

In this example we also show the datagram that is output by the host on the left being delivered to itself. This is a property of broadcasts: by definition a broadcast goes to every host on the subnet, which includes the sender (pp. 109–110 of TCPv2). We also assume that the sending application has bound the port that it is sending to (520), so it will receive a copy of each broadcast datagram that it sends. (In general, however, there is no requirement that a process bind a UDP port to which it sends datagrams.)

> In this example we show a logical loopback, performed by either the IP layer or the datalink layer making a copy (pp. 109–110 of TCPv2) and sending the copy up the protocol stack. A network could use a physical loopback, but this can lead to problems in the case of network faults (such as an unterminated Ethernet).

This example shows the fundamental problem with broadcasting: every IPv4 host on the subnet that is not participating in the application must completely process the broadcast UDP datagram all the way up the protocol stack, through and including the UDP layer, before discarding the datagram. (Recall our discussion following Figure 8.21.) Also, every non-IP host on the subnet (say a host running Novell's IPX) must also receive the entire frame at the datalink layer before discarding the frame (assuming

the host does not support the frame type, which would be 0800 for an IPv4 datagram). For applications that generate IP datagrams at a high rate (audio or video, for example) this unnecessary processing can severely affect these other hosts on the subnet. We will see in the next chapter how multicasting gets around this problem.

> Our choice of UDP port 520 in Figure 18.4 is intentional. This is the port used by the routed daemon to exchange RIP (Routing Information Protocol) packets. All routers on a subnet that are using RIP will send a broadcast UDP datagram every 30 seconds. If there are 200 hosts on the subnet, which includes two routers using RIP, 198 hosts will have to process (and discard) these broadcast datagrams every 30 seconds, assuming none of the 198 hosts is running routed.

18.4 dg_cli Function Using Broadcasting

We modify our dg_cli function one more time, this time allowing it to broadcast to the standard UDP daytime server (Figure 2.13), and printing all the replies. The only change we make to the main function (Figure 8.7) is to change the destination port number to 13:

```
servaddr.sin_port = htons(13);
```

We first compile this modified main function with the unmodified dg_cli function from Figure 8.8 and run it on the host bsdi.

```
bsdi % udpcli01 206.62.226.63
hi
sendto error: Permission denied
```

The command-line argument is the subnet-directed broadcast address for the attached Ethernet. We type a line of input, the program calls sendto, and the error EACCES is returned. The reason we receive the error is that we are not allowed to send a datagram to a broadcast destination address unless we explicitly tell the kernel that we will be broadcasting. We do this by setting the SO_BROADCAST socket option (Section 7.5).

> Berkeley-derived implementations implement this sanity check. Solaris 2.5, on the other hand, accepts the datagram destined for the broadcast address even if we do not specify the socket option. Posix.1g says that the kernel "may" return the error.
>
> Broadcasting was a privileged operation with 4.2BSD and the SO_BROADCAST socket option did not exist. This option was added to 4.3BSD and any process was allowed to set the option.

We now modify our dg_cli function as shown in Figure 18.5. This version sets the SO_BROADCAST socket option and prints all the replies received within 5 seconds.

Allocate room for server's address, set socket option

11-13 malloc allocates room for the server's address to be returned by recvfrom. The SO_BROADCAST socket option is set and a signal handler is installed for SIGALRM.

Read line, send to socket, read all replies

14-24 The next two steps, fgets and sendto, are similar to previous versions of this function. But since we are sending a broadcast datagram we can receive multiple

bcast/dgclibcast1.c

```
 1 #include    "unp.h"
 2 static void recvfrom_alarm(int);
 3 void
 4 dg_cli(FILE *fp, int sockfd, const SA *pservaddr, socklen_t servlen)
 5 {
 6     int     n;
 7     const int on = 1;
 8     char    sendline[MAXLINE], recvline[MAXLINE + 1];
 9     socklen_t len;
10     struct sockaddr *preply_addr;
11     preply_addr = Malloc(servlen);
12     Setsockopt(sockfd, SOL_SOCKET, SO_BROADCAST, &on, sizeof(on));
13     Signal(SIGALRM, recvfrom_alarm);
14     while (Fgets(sendline, MAXLINE, fp) != NULL) {
15         Sendto(sockfd, sendline, strlen(sendline), 0, pservaddr, servlen);
16         alarm(5);
17         for ( ; ; ) {
18             len = servlen;
19             n = recvfrom(sockfd, recvline, MAXLINE, 0, preply_addr, &len);
20             if (n < 0) {
21                 if (errno == EINTR)
22                     break;      /* waited long enough for replies */
23                 else
24                     err_sys("recvfrom error");
25             } else {
26                 recvline[n] = 0;    /* null terminate */
27                 printf("from %s: %s",
28                        Sock_ntop_host(preply_addr, len), recvline);
29             }
30         }
31     }
32 }
33 static void
34 recvfrom_alarm(int signo)
35 {
36     return;                     /* just interrupt the recvfrom() */
37 }
```

bcast/dgclibcast1.c

Figure 18.5 `dg_cli` function that broadcasts.

replies. We call `recvfrom` in a loop and print all the replies received within 5 seconds. After 5 seconds, `SIGALRM` is generated, our signal handler is called, and `recvfrom` returns the error `EINTR`.

Print each received reply

25–29 For each reply received we call `sock_ntop_host` which in the case of IPv4 returns

a string containing the dotted-decimal IP address of the server. This is printed along with the server's reply.

If we run the program, specifying the subnet-directed broadcast address of 206.62.226.63, we see the following:

```
bsdi % udpcli01 206.62.226.63
hi
from 206.62.226.35: Sat Jun 14 12:19:36 1997
from 206.62.226.40: Sat Jun 14 12:19:36 1997
from 206.62.226.34: Sat Jun 14 12:19:36 1997
from 206.62.226.43: Sat Jun 14 12:19:36 1997
from 206.62.226.37: Sat Jun 14 12:19:36 1997
from 206.62.226.42: Sat Jun 14 12:19:36 1997
hello
from 206.62.226.35: Sat Jun 14 12:19:43 1997
from 206.62.226.40: Sat Jun 14 12:19:43 1997
from 206.62.226.34: Sat Jun 14 12:19:43 1997
from 206.62.226.43: Sat Jun 14 12:19:43 1997
from 206.62.226.42: Sat Jun 14 12:19:43 1997
from 206.62.226.37: Sat Jun 14 12:19:43 1997
```

Each time we must type a line of input to generate the output UDP datagram. Each time we receive six replies and this includes the sending host. As we said earlier, the destination of a broadcast datagram is *all* the hosts on the attached network, including the sender. Each reply is unicast, because the source address of the request, which is used by each server as the destination address of the reply, is a unicast address.

All the systems report the same time because all run NTP. We see that only six of the nine nodes on this subnet (Figure 1.16) respond. The remaining two hosts and the one router do not respond to the requests, since the requests are sent to a broadcast address.

IP Fragmentation and Broadcasts

Berkeley-derived kernels do not allow a broadcast datagram to be fragmented. If the size of an IP datagram that is being sent to a broadcast address exceeds the outgoing interface MTU, `EMSGSIZE` is returned (pp. 233–234 of TCPv2). This is a policy decision that has existed since 4.2BSD. There is nothing that prevents a kernel from fragmenting a broadcast datagram, but the feeling is that broadcasting puts enough load on the network as it is, so there is no need to multiply this load by the number of fragments.

We can see this scenario with our program in Figure 18.5. We redirect standard input to a file containing a 2000-byte line, which will require fragmentation on an Ethernet.

```
bsdi % udpcli01 206.62.226.63 < 2000line
sendto error: Message too long
```

AIX, BSD/OS, Digital Unix, and UnixWare all implement this limitation, although UnixWare does not send the datagram but does not return an error either. Both Linux and Solaris fragment datagrams sent to a broadcast address. For portability, however, an application that

needs to broadcast should limit its datagrams to 1472 bytes, since the Ethernet MTU is normally the smallest for a LAN.

18.5 Race Conditions

A *race condition* is usually when multiple processes are accessing data that is shared among them but the correct outcome depends on the execution order of the processes. Since the execution order of processes on typical Unix systems is nondeterministic, sometimes the outcome is correct but sometimes the outcome is wrong. The hardest type of race conditions to debug are those in which the outcome is normally correct and only occasionally is the outcome wrong. We talk more about these types of race conditions in Chapter 23, when we discuss mutual exclusion variables and condition variables. Race conditions are always a concern with threads programming since so much data is shared among all the threads (e.g., all the global variables).

Race conditions of a different type often exist when dealing with signals. The problem occurs because a signal can normally be delivered at anytime while our program is executing. Posix.1 allows us to *block* a signal from being delivered, but this is often of little use while we are performing I/O operations.

An example is an easy way to see this problem. A race condition exists in Figure 18.5; take a few minutes and see if you can find it. (*Hint*: Where can we be executing when the signal is delivered?) You can also force the condition to occur as follows: change the argument to `alarm` from 5 to 1, and add `sleep(1)` immediately before the `printf`.

When we make these changes to the function and then type the first line of input, the line is sent as a broadcast and we set the `alarm` for 1 second in the future. We block in the call to `recvfrom` and the first reply then arrives for our socket, probably within a few milliseconds. The reply is returned by `recvfrom` but we then go to sleep for 1 second. Additional replies are received, and they are placed into our socket's receive buffer. But while we are asleep, the `alarm` timer expires and the `SIGALRM` signal is generated: our signal handler is called, and it just returns and interrupts the `sleep` in which we are blocked. We then loop around and read the queued replies, with a 1 second pause each time we print a reply. But when we have read all the replies we block again in the call to `recvfrom` but the timer is not running. We will block forever in `recvfrom`. The fundamental problem is that our intent is for our signal handler to interrupt a blocked `recvfrom`, but the signal can be delivered at any time, and we can be executing anywhere in the infinite `for` loop when the signal is delivered.

We now examine four different solutions to this problem: one incorrect solution and three different correct solutions.

Blocking and Unblocking the Signal

Our first (incorrect) solution reduces the window of error by blocking the signal from being delivered while we are executing the remainder of the `for` loop. Figure 18.6 shows the new version.

bcast/dgclibcast3.c

```
 1 #include    "unp.h"

 2 static void recvfrom_alarm(int);

 3 void
 4 dg_cli(FILE *fp, int sockfd, const SA *pservaddr, socklen_t servlen)
 5 {
 6     int     n;
 7     const int on = 1;
 8     char    sendline[MAXLINE], recvline[MAXLINE + 1];
 9     sigset_t sigset_alrm;
10     socklen_t len;
11     struct sockaddr *preply_addr;

12     preply_addr = Malloc(servlen);

13     Setsockopt(sockfd, SOL_SOCKET, SO_BROADCAST, &on, sizeof(on));

14     Sigemptyset(&sigset_alrm);
15     Sigaddset(&sigset_alrm, SIGALRM);

16     Signal(SIGALRM, recvfrom_alarm);

17     while (Fgets(sendline, MAXLINE, fp) != NULL) {

18         Sendto(sockfd, sendline, strlen(sendline), 0, pservaddr, servlen);

19         alarm(5);
20         for ( ; ; ) {
21             len = servlen;
22             Sigprocmask(SIG_UNBLOCK, &sigset_alrm, NULL);
23             n = recvfrom(sockfd, recvline, MAXLINE, 0, preply_addr, &len);
24             Sigprocmask(SIG_BLOCK, &sigset_alrm, NULL);
25             if (n < 0) {
26                 if (errno == EINTR)
27                     break;      /* waited long enough for replies */
28                 else
29                     err_sys("recvfrom error");
30             } else {
31                 recvline[n] = 0;    /* null terminate */
32                 printf("from %s: %s",
33                        Sock_ntop_host(preply_addr, len), recvline);
34             }
35         }
36     }
37 }

38 static void
39 recvfrom_alarm(int signo)
40 {
41     return;                     /* just interrupt the recvfrom() */
42 }
```

bcast/dgclibcast3.c

Figure 18.6 Block signals while executing within the `for` loop (incorrect solution).

Declare signal set and initialize

14-15 We declare a signal set, initialize it to the empty set (`sigemptyset`), and then turn on the bit corresponding to `SIGALRM` (`sigaddset`).

Unblock and block signal

21-24 Before calling `recvfrom` we unblock the signal (so that it can be delivered while we are blocked) and then block the signal as soon as `recvfrom` returns. If the signal is generated (i.e., the timer expires) while the signal is blocked, the kernel remembers this fact but cannot deliver the signal (i.e., call our signal handler) until the signal is unblocked. This is the fundamental difference between the *generation* of a signal and its *delivery*. Chapter 10 of APUE provides additional details on all these facets of Posix.1 signal handling.

If we compile and run this program, it appears to work fine, but then most programs with a race condition work most of the time! There is still a problem: the unblocking of the signal, the call to `recvfrom`, and the blocking of the signal are all independent system calls. Assume `recvfrom` returns with the final datagram reply and the signal is delivered between the `recvfrom` and the blocking of the signal. The next call to `recvfrom` will block forever. We have reduced the window, but the problem still exists.

A variation of this solution is to have the signal handler set a global flag when the signal is delivered.

```
static void
recvfrom_alarm(int signo)
{
    had_alarm = 1;
    return;
}
```

The flag is initialized to 0 each time `alarm` is called. Our `dg_cli` function checks this flag before calling `recvfrom` and does not call it if the flag is nonzero.

```
for ( ; ; ) {
    len = servlen;
    Sigprocmask(SIG_UNBLOCK, &sigset_alrm, NULL);
    if (had_alarm == 1)
        break;
    n = recvfrom(sockfd, recvline, MAXLINE, 0, preply_addr, &len);
```

If the signal was generated during the time it was blocked (after the previous return from `recvfrom`) and when the signal is unblocked in this piece of code, it will be delivered before `sigprocmask` returns, setting our flag. But there is still a small window of time between the testing of the flag and the call to `recvfrom` when the signal can be generated and delivered, and if this happens, the call to `recvfrom` will block forever (assuming, of course, no additional replies are received).

Blocking and Unblocking the Signal with `pselect`

One correct solution is to use `pselect` (Section 6.9) as shown in Figure 18.7.

bcast/dgclibcast4.c

```
 1 #include    "unp.h"
 2 static void recvfrom_alarm(int);
 3 void
 4 dg_cli(FILE *fp, int sockfd, const SA *pservaddr, socklen_t servlen)
 5 {
 6     int     n;
 7     const int on = 1;
 8     char    sendline[MAXLINE], recvline[MAXLINE + 1];
 9     fd_set  rset;
10     sigset_t sigset_alrm, sigset_empty;
11     socklen_t len;
12     struct sockaddr *preply_addr;
13     preply_addr = Malloc(servlen);
14     Setsockopt(sockfd, SOL_SOCKET, SO_BROADCAST, &on, sizeof(on));
15     FD_ZERO(&rset);
16     Sigemptyset(&sigset_empty);
17     Sigemptyset(&sigset_alrm);
18     Sigaddset(&sigset_alrm, SIGALRM);
19     Signal(SIGALRM, recvfrom_alarm);
20     while (Fgets(sendline, MAXLINE, fp) != NULL) {
21         Sendto(sockfd, sendline, strlen(sendline), 0, pservaddr, servlen);
22         Sigprocmask(SIG_BLOCK, &sigset_alrm, NULL);
23         alarm(5);
24         for ( ; ; ) {
25             FD_SET(sockfd, &rset);
26             n = pselect(sockfd + 1, &rset, NULL, NULL, NULL, &sigset_empty);
27             if (n < 0) {
28                 if (errno == EINTR)
29                     break;
30                 else
31                     err_sys("pselect error");
32             } else if (n != 1)
33                 err_sys("pselect error: returned %d", n);
34             len = servlen;
35             n = Recvfrom(sockfd, recvline, MAXLINE, 0, preply_addr, &len);
36             recvline[n] = 0;    /* null terminate */
37             printf("from %s: %s",
38                    Sock_ntop_host(preply_addr, len), recvline);
39         }
40     }
41 }
42 static void
43 recvfrom_alarm(int signo)
44 {
45     return;                     /* just interrupt the recvfrom() */
46 }
```

bcast/dgclibcast4.c

Figure 18.7 Blocking and unblocking signals with `pselect`.

22-33 We block `SIGALRM` and call `pselect`. The final argument to `pselect` is a pointer to our `sigset_empty` variable, which is a signal set with no signals blocked, that is, all signals unblocked. `pselect` will save the current signal mask (which has `SIGALRM` blocked), test the specified descriptors, block if necessary with the signal mask set to the empty set, but before returning the signal mask of the process is reset to its value when `pselect` was called. The key to `pselect` is that the setting of the signal mask, the testing of the descriptors, and the resetting of the signal mask are atomic operations with regard to the calling process.

34-38 If our socket is readable, we call `recvfrom` knowing it will not block.

As we mentioned in Section 6.9, `pselect` is new with Posix.1g; none of the systems in Figure 1.16 support the function. Nevertheless, Figure 18.8 shows a simple, albeit incorrect, implementation. Our reason for showing this incorrect implementation is to show the three steps involved: setting of the signal mask to the value specified by the caller along with saving the current mask, testing the descriptors, and resetting the signal mask.

lib/pselect.c

```
 9 #include    "unp.h"

10 int
11 pselect(int nfds, fd_set *rset, fd_set *wset, fd_set *xset,
12         const struct timespec *ts, const sigset_t *sigmask)
13 {
14     int     n;
15     struct timeval tv;
16     sigset_t savemask;

17     if (ts != NULL) {
18         tv.tv_sec = ts->tv_sec;
19         tv.tv_usec = ts->tv_nsec / 1000;     /* nanosec -> microsec */
20     }
21     sigprocmask(SIG_SETMASK, sigmask, &savemask);    /* caller's mask */
22     n = select(nfds, rset, wset, xset, (ts == NULL) ? NULL : &tv);
23     sigprocmask(SIG_SETMASK, &savemask, NULL);  /* restore mask */

24     return (n);
25 }
```

lib/pselect.c

Figure 18.8 Simple, incorrect implementation of `pselect`.

Using `sigsetjmp` and `siglongjmp`

Another correct way to solve our problem is not to use the ability of a signal handler to interrupt a blocked system call, but to call `siglongjmp` from the signal handler instead. This is called a *nonlocal goto* because we can use it to jump from one function back to another. Figure 18.9 demonstrates this technique.

Allocate jump buffer

4 We allocate a jump buffer that will be used by our function and its signal handler.

bcast/dgclibcast5.c

```
 1 #include    "unp.h"
 2 #include    <setjmp.h>

 3 static void recvfrom_alarm(int);
 4 static sigjmp_buf jmpbuf;

 5 void
 6 dg_cli(FILE *fp, int sockfd, const SA *pservaddr, socklen_t servlen)
 7 {
 8     int     n;
 9     const int on = 1;
10     char    sendline[MAXLINE], recvline[MAXLINE + 1];
11     socklen_t len;
12     struct sockaddr *preply_addr;

13     preply_addr = Malloc(servlen);

14     Setsockopt(sockfd, SOL_SOCKET, SO_BROADCAST, &on, sizeof(on));

15     Signal(SIGALRM, recvfrom_alarm);

16     while (Fgets(sendline, MAXLINE, fp) != NULL) {

17         Sendto(sockfd, sendline, strlen(sendline), 0, pservaddr, servlen);

18         alarm(5);
19         for ( ; ; ) {
20             if (sigsetjmp(jmpbuf, 1) != 0)
21                 break;
22             len = servlen;
23             n = Recvfrom(sockfd, recvline, MAXLINE, 0, preply_addr, &len);
24             recvline[n] = 0;    /* null terminate */
25             printf("from %s: %s",
26                    Sock_ntop_host(preply_addr, len), recvline);
27         }
28     }
29 }

30 static void
31 recvfrom_alarm(int signo)
32 {
33     siglongjmp(jmpbuf, 1);
34 }
```

bcast/dgclibcast5.c

Figure 18.9 Use of `sigsetjmp` and `siglongjmp` from signal handler.

Call `sigsetjmp`

20–23 When we call `sigsetjmp` directly from our `dg_cli` function, it establishes the jump buffer and returns 0. We proceed on and call `recvfrom`.

Handle `SIGALRM` and call `siglongjmp`

30–34 When the signal is delivered, we call `siglongjmp`. This causes the `sigsetjmp` in the `dg_cli` function to return with a return value equal to the second argument (1), which must be a nonzero value. This will cause the `for` loop in `dg_cli` to terminate.

Using `sigsetjmp` and `siglongjmp` in this fashion guarantees that we will not block forever in `recvfrom` because of a signal delivered at an inopportune time. The only potential for a problem occurs if the signal is delivered while `printf` is in the middle of its output. To prevent this we should combine the signal blocking and unblocking from Figure 18.6 with the nonlocal goto.

Using IPC from Signal Handler to Function

There is yet another correct way to solve our problem. Instead of having the signal handler just return and hopefully interrupt a blocked `recvfrom`, we have the signal handler use IPC (interprocess communication) to notify our `dg_cli` function that the timer has expired. This is somewhat similar to the proposal we made earlier for the signal handler to set the global `had_alarm` when the timer expired, because that global variable was being used as a form of IPC (shared memory between our function and the signal handler). The problem with that solution, however, was our function had to test this variable, and this led to timing problems if the signal was delivered at about the same time.

What we use in Figure 18.10 is a pipe within our process, with the signal handler writing 1 byte to the pipe when the timer expires, and our `dg_cli` function reading that byte to know when to terminate its `for` loop. What makes this such a nice solution is that the testing for the pipe being readable is done using `select`. We test for either the socket being readable or the pipe being readable.

Create pipe

15 We create a normal Unix pipe and two descriptors are returned. `pipefd[0]` is the read end and `pipefd[1]` is the write end.

> We could also use `socketpair` and get a full-duplex pipe. On some systems, notably SVR4, a normal Unix pipe is always full-duplex and we can read from either end and write to either end.

`select` on both socket and read end of pipe

23-30 We `select` on both the socket and the read end of the pipe.

45-50 When `SIGALRM` is delivered, our signal handler writes 1 byte to the pipe, making the read end readable. Our signal handler also returns, possibly interrupting `select`. Therefore if `select` returns `EINTR`, we ignore the error, knowing that the read end of the pipe will also be readable, and that will terminate the `for` loop.

`read` from pipe

38-41 When the read end of the pipe is readable, we `read` the null byte that the signal handler wrote and ignore it. But this tells us that the timer expired, so we `break` out of the infinite `for` loop.

bcast/dgclibcast6.c

```
1 #include    "unp.h"

2 static void recvfrom_alarm(int);
3 static int pipefd[2];
```

```
 4 void
 5 dg_cli(FILE *fp, int sockfd, const SA *pservaddr, socklen_t servlen)
 6 {
 7     int     n, maxfdp1;
 8     const int on = 1;
 9     char    sendline[MAXLINE], recvline[MAXLINE + 1];
10     fd_set  rset;
11     socklen_t len;
12     struct sockaddr *preply_addr;

13     preply_addr = Malloc(servlen);

14     Setsockopt(sockfd, SOL_SOCKET, SO_BROADCAST, &on, sizeof(on));

15     Pipe(pipefd);
16     maxfdp1 = max(sockfd, pipefd[0]) + 1;

17     FD_ZERO(&rset);

18     Signal(SIGALRM, recvfrom_alarm);

19     while (Fgets(sendline, MAXLINE, fp) != NULL) {
20         Sendto(sockfd, sendline, strlen(sendline), 0, pservaddr, servlen);

21         alarm(5);
22         for ( ; ; ) {
23             FD_SET(sockfd, &rset);
24             FD_SET(pipefd[0], &rset);
25             if ( (n = select(maxfdp1, &rset, NULL, NULL, NULL)) < 0) {
26                 if (errno == EINTR)
27                     continue;
28                 else
29                     err_sys("select error");
30             }
31             if (FD_ISSET(sockfd, &rset)) {
32                 len = servlen;
33                 n = Recvfrom(sockfd, recvline, MAXLINE, 0, preply_addr, &len);
34                 recvline[n] = 0;    /* null terminate */
35                 printf("from %s: %s",
36                        Sock_ntop_host(preply_addr, len), recvline);
37             }
38             if (FD_ISSET(pipefd[0], &rset)) {
39                 Read(pipefd[0], &n, 1);     /* timer expired */
40                 break;
41             }
42         }
43     }
44 }

45 static void
46 recvfrom_alarm(int signo)
47 {
48     Write(pipefd[1], "", 1);    /* write 1 null byte to pipe */
49     return;
50 }
```

bcast/dgclibcast6.c

Figure 18.10 Using a pipe as IPC from signal handler to our function.

18.6 Summary

Broadcasting sends a datagram that all hosts on the attached subnet receive. The disadvantage in broadcasting is that every host on the subnet must process the datagram, up through the UDP layer in the case of a UDP datagram, even if the host is not participating in the application. For high data rate applications, such as audio or video, this can place an excessive processing load on these hosts. We will see in the next chapter that multicasting solves this problem because only the hosts that are interested in the application receive the datagram.

Using a version of our UDP echo client that sends a broadcast to the daytime server and then prints all the replies that are received within 5 seconds, lets us look at race conditions with the `SIGALRM` signal. Since the use of the `alarm` function and the `SIGALRM` signal is a common way to place a timeout on a read operation, this subtle error is common in networking applications. We showed one incorrect way to solve the problem, and three correct ways:

- using `pselect`,
- using `sigsetjmp` and `siglongjmp`, and
- using IPC (typically a pipe) from the signal handler to the main loop.

Exercises

18.1 Run the UDP client using the `dg_cli` function that broadcasts: Figure 18.5. How many replies do you receive? Are the replies always in the same order? Do the hosts on your network have synchronized clocks?

18.2 When talking about the fragmentation of broadcast datagrams at the end of Section 18.4, we said that for portability an application should limit its broadcast datagrams to 1472 bytes. Where does this magic number come from?

18.3 Put some `printfs` in Figure 18.10 after `select` returns to see whether it returns an error or readability for one of the two descriptors. When the `alarm` expires, does your system return `EINTR` or readability on the pipe?

18.4 Run a tool such as `tcpdump`, if available, and look for broadcast packets on your LAN; `tcpdump ether broadcast` is the `tcpdump` command. To which protocol suites do the broadcasts belong?

19

Multicasting

19.1 Introduction

As shown in Figure 18.1, a unicast address identifies a *single* interface, a broadcast address identifies *all* interfaces on the subnet, and a multicast address identifies a *set* of interfaces. Unicasting and broadcasting are the endpoints of the addressing spectrum (one or all) and the intent of multicasting is to provide the capability of addressing something in between these endpoints. A multicast datagram should be received only by the interfaces that are interested in the datagram, that is, by the interfaces on the hosts that are running applications that wish to participate in the multicast session. Also, broadcasting is normally limited to a LAN whereas multicasting can be used on a LAN or across a WAN. Indeed, applications on the MBone (Section B.2) multicast across the entire Internet on a daily basis.

The additions to the sockets API to support multicasting are simple: five socket options: three that affect the sending of UDP datagrams to a multicast address, and two that affect the host's reception of multicast datagrams.

19.2 Multicast Addresses

When describing multicast addresses we must distinguish between IPv4 and IPv6.

IPv4 Class D Addresses

Class D addresses, in the range 224.0.0.0 through 239.255.255.255, are the multicast addresses in IPv4 (Figures A.3 and A.4). The low-order 28 bits of the class D address form the multicast *group ID* and the 32-bit address is called the *group address*.

Figure 19.1 shows how multicast addresses are mapped into Ethernet addresses. This mapping for IPv4 multicast addresses is described in RFC 1112 [Deering 1989] for Ethernets, in RFC 1390 [Katz 1993] for FDDI networks, and in RFC 1469 [Pusateri 1993] for token-ring networks. We also show the mapping for IPv6 multicast addresses to allow easy comparison of the resulting Ethernet addresses.

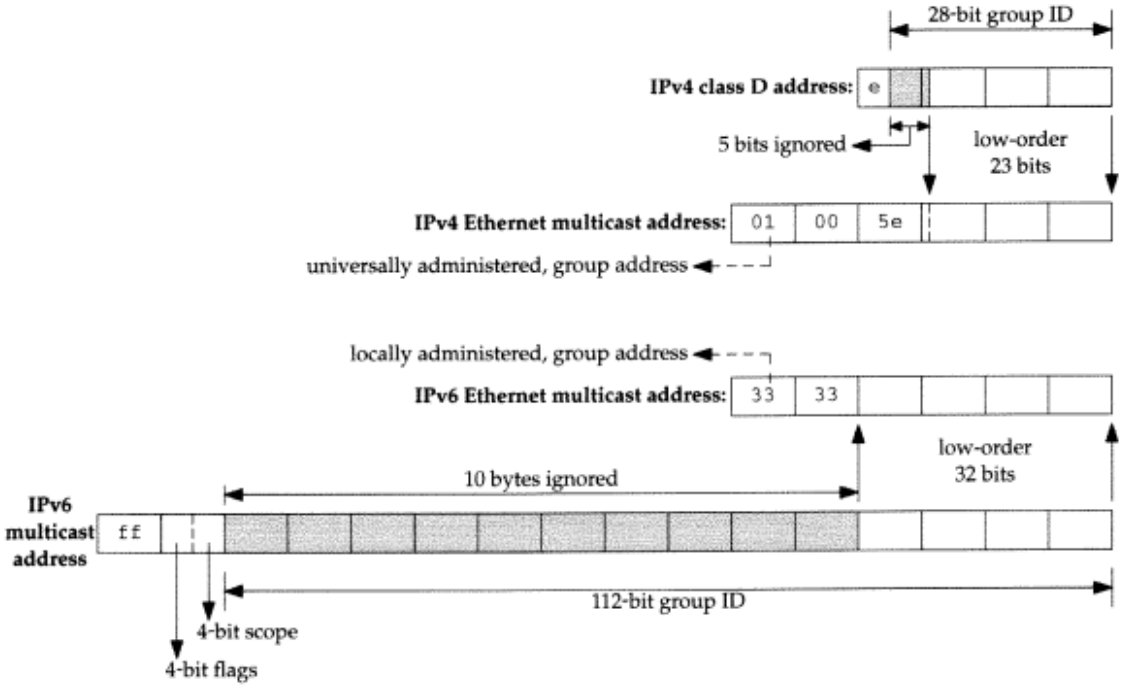


Figure 19.1 Mapping of IPv4 and IPv6 multicast address to Ethernet addresses.

Considering just the IPv4 mapping, the high-order 24 bits of the Ethernet address are always `01:00:5e`. The next bit is always 0 and the low-order 23 bits are copied from the low-order 23 bits of the multicast group address. The high-order 5 bits of the group address are ignored in the mapping. This means that 32 multicast addresses map to a single Ethernet address: the mapping is not one-to-one.

The low-order 2 bits of the first byte of the Ethernet address identify the address as a universally administered group address. Universally administered means the high-order 24 bits have been assigned by the IEEE and group addresses are recognized and handled specially by receiving interfaces.

There are a few special IPv4 multicast addresses.

- 224.0.0.1 is the *all-hosts* group. All multicast-capable hosts on a subnet must join this group on all multicast-capable interfaces. (We talk about what it means to join a multicast group shortly.)
- 224.0.0.2 is the *all-routers* group. All multicast routers on a subnet must join this group on all multicast-capable interfaces.

The range 224.0.0.0 through 224.0.0.255 (which we can also write as 224.0.0.0/24) is called *link local*. These addresses are reserved for low-level topology discovery or maintenance protocols, and datagrams destined to any of these addresses are never forwarded by a multicast router. We say more about the scope of various IPv4 multicast addresses after looking at the IPv6 multicast addresses.

IPv6 Multicast Addresses

The high-order byte of an IPv6 multicast address has the value `ff`. Figure 19.1 shows the mapping from a 16-byte IPv6 multicast address into a 6-byte Ethernet address. The low-order 32 bits of the group address are copied into the low-order 32 bits of the Ethernet address. The high-order 2 bytes of the Ethernet address are `33:33`. This mapping for Ethernets is specified in RFC 1972 [Crawford 1996a], the same mapping for FDDI is in RFC 2019 [Crawford 1996b], and the token ring mapping is in [Thomas 1997].

The low-order 2 bits of the first byte of the Ethernet address specify the address as a locally administered group address. Locally administered means there is no guarantee that the address is unique to IPv6. There could be other protocol suites besides IPv6 sharing the network and using the same high-order 2 bytes of the Ethernet address. As we mentioned earlier, group addresses are recognized and handled specially by receiving interfaces.

The 4-bit IPv6 multicast flags differentiate between a *well-known* multicast group (a value of 0) and a *transient* multicast group (a value of 1). The upper 3 bits of this field are reserved. IPv6 multicast addresses also have a 4-bit *scope* field that we discuss shortly.

There are a few special IPv6 multicast addresses.

- `ff02::1` is the *all-nodes* group. All multicast-capable hosts on a subnet must join this group on all multicast-capable interfaces. This is similar to the IPv4 224.0.0.1 multicast address.
- `ff02::2` is the *all-routers* group. All multicast routers on a subnet must join this group on all multicast-capable interfaces. This is similar to the IPv4 224.0.0.2 multicast address.

Since only the low-order 32 bits of the IPv6 multicast address are used when mapping to the hardware address, [Hinden and Deering 1997] recommend that the 10 bytes to the left of these 32 bits in Figure 19.1 be 0 when assigning new multicast addresses, at least until these 10 bytes are needed in the future.

Scope of Multicast Addresses

IPv6 multicast addresses have an explicit 4-bit *scope* field that specifies how "far" the multicast packet will travel. IPv6 packets also have a hop limit field that limits the number of times the packet is forwarded by a router. The following values have been assigned to the scope field:

1: node-local
2: link-local
5: site-local
8: organization-local
14: global

The remaining values are unassigned or reserved. A node-local datagram must not be output by an interface and a link-local datagram must never be forwarded by a router. What defines a site or an organization is up to the administrators of the multicast routers at that site or organization. IPv6 multicast addresses that differ only in scope represent different groups.

IPv4 does not have a separate scope field for multicast packets. Historically the IPv4 TTL field in the IP header has doubled as a multicast scope field: a TTL of 0 means node-local, 1 means link-local, up through 32 means site-local, up through 64 means region-local, up through 128 means continent-local (whatever that is), and up through 255 are unrestricted in scope (global). This double usage of the TTL field has led to difficulties, as detailed in [Meyer 1997].

Although use of the IPv4 TTL field for scoping is accepted and recommended practice, administrative scoping is preferred, when possible. This defines the range 239.0.0.0 through 239.255.255.255 as the *administratively scoped IPv4 multicast space* ([Meyer 1997]). This is the high end of the multicast address space. Addresses in this range are assigned locally by an organization but are not guaranteed to be unique across organizational boundaries. An organization must configure its boundary routers (multicast routers at the boundary of the organization) not to forward multicast packets destined to any of these addresses.

The administratively scoped IPv4 multicast addresses are then divided into local scope and organization-local scope, the former being similar (but not semantically equivalent) to the IPv6 site-local scope. We summarize the different scoping rules in Figure 19.2.

Scope	IPv6 scope	IPv4	
		TTL scope	administrative scope
node-local	1	0	
link-local	2	1	224.0.0.0 to 224.0.0.255
site-local	5	<32	239.255.0.0 to 239.255.255.255
organization-local	8		239.192.0.0 to 239.195.255.255
global	14	<255	224.0.1.0 to 238.255.255.255

Figure 19.2 Scope of IPv4 and IPv6 multicast addresses.

19.3 Multicasting versus Broadcasting on a LAN

We now return to the examples in Figures 18.3 and 18.4 to show what happens in the case of multicasting. We use IPv4 for the example shown in Figure 19.3, but the steps are similar for IPv6.

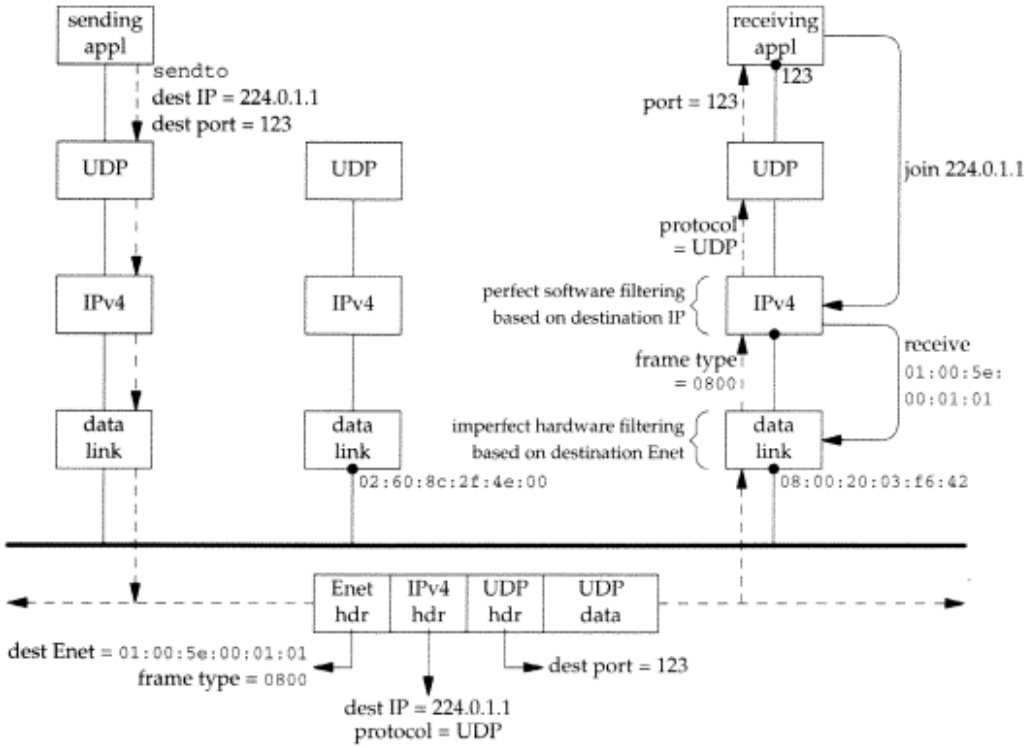


Figure 19.3 Multicast example of a UDP datagram.

The receiving application on the rightmost host starts and creates a UDP socket, binds port 123 to the socket, and then joins the multicast group 224.0.1.1. We will see shortly that this "join" operation is done by calling `setsockopt`. When this happens, the IPv4 layer saves the information internally and then tells the appropriate datalink to receive Ethernet frames destined to `01:00:5e:00:01:01` (Section 12.11 of TCPv2). This is the Ethernet address corresponding to the multicast address that the application has just joined using the mapping that we showed in Figure 19.1.

The next step is for the sending application on the leftmost host to create a UDP socket and send a datagram to 224.0.1.1, port 123. Nothing special is required to send a multicast datagram: the application does not have to join the multicast group. The sending host converts the IP address into the corresponding Ethernet destination address and the frame is sent. Notice that the frame contains both the destination Ethernet address (which is examined by the interfaces) and the destination IP address (which is examined by the IP layers).

We assume that the host in the middle is not IPv4 multicast capable (since support for IPv4 multicasting is optional). This host ignores the frame completely because (1) the destination Ethernet address does not match the address of the interface, (2) the destination Ethernet address is not the Ethernet broadcast address, and (3) the interface has not been told to receive any group addresses (those with the low-order bit of the high-order byte set to 1, as in Figure 19.1).

The frame is received by the datalink on the right, based on what we call *imperfect filtering* done by the interface using the Ethernet destination address. We say this is imperfect because it is normally the case that when the interface is told to receive frames destined to one specific Ethernet group address, it can receive frames destined to other Ethernet group addresses too.

> When told to receive frames destined to a specific Ethernet group address, many current Ethernet interface cards apply a hash function to the address, calculating a value between 0 and 63. One of 64 bits in an array is then turned on. When a frame passes by on the cable that is destined for a group address, the same hash function is applied by the interface to the destination address (which is the first field in the frame), calculating a value between 0 and 63. If the corresponding bit in the array is on, the frame is received; otherwise it is ignored. Newer interface cards increase the size of the bit array from 64 to 512, reducing the probability that an interface will receive frames in which it is not interested. Over time, as more and more applications use multicasting, this size will probably increase even more. Some interface cards today already have perfect filtering. Other interface cards have no multicast filtering at all, and when told to receive a specific group address must receive all multicast frames (sometimes called *multicast promiscuous* mode). One popular interface card does perfect filtering for 16 group addresses as well as having a 512-bit hash table. Even if the interface performs perfect filtering, perfect software filtering at the IP layer is still required because the mapping from the IP multicast address to the hardware address is not one-to-one.

Assuming that the datalink on the right receives the frame, since the Ethernet frame type is IPv4, the packet is passed to the IP layer. Since the received packet was destined to a multicast IP address, the IP layer compares this address against all the multicast addresses that applications on this host have joined. We call this *perfect filtering* since it is based on the entire 32-bit class D address in the IPv4 header. In this example the packet is accepted by the IP layer and passed to the UDP layer, which in turn passes the datagram to the socket that is bound to port 123.

There are three more scenarios that we do not show in Figure 19.3.

1. A host running an application that has joined the multicast address 225.0.1.1. Since the upper 5 bits of the group address are ignored in the mapping to the Ethernet address, this host's interface will also be receiving frames with an Ethernet destination address of `01:00:5e:00:01:01`. In this case the packet will be discarded by the perfect filtering in the IP layer.
2. A host running an application that has joined some multicast group whose corresponding Ethernet address just happens to be one that the interface receives when it is programmed to receive `01:00:5e:00:01:01` (i.e., the interface card performs imperfect filtering). This frame will be discarded either by the datalink layer or by the IP layer.
3. A packet destined to the same group, 224.0.1.1, but a different port, say 4000. The rightmost host in Figure 19.3 still receives the packet, and it is accepted by the IP layer, but assuming a socket does not exist that has bound port 4000, the packet will be discarded by the UDP layer.

> This demonstrates that for a process to receive a multicast datagram the process must join the group and bind the port.

19.4 Multicasting on a WAN

Multicasting on a single LAN, as we discussed in the previous section, is simple. One host sends a multicast packet and any interested host receives the packet. The benefit of multicasting over broadcasting is reducing the load on all the hosts not interested in the multicast packets.

Multicasting is also beneficial on WANs. Consider the WAN shown in Figure 19.4.

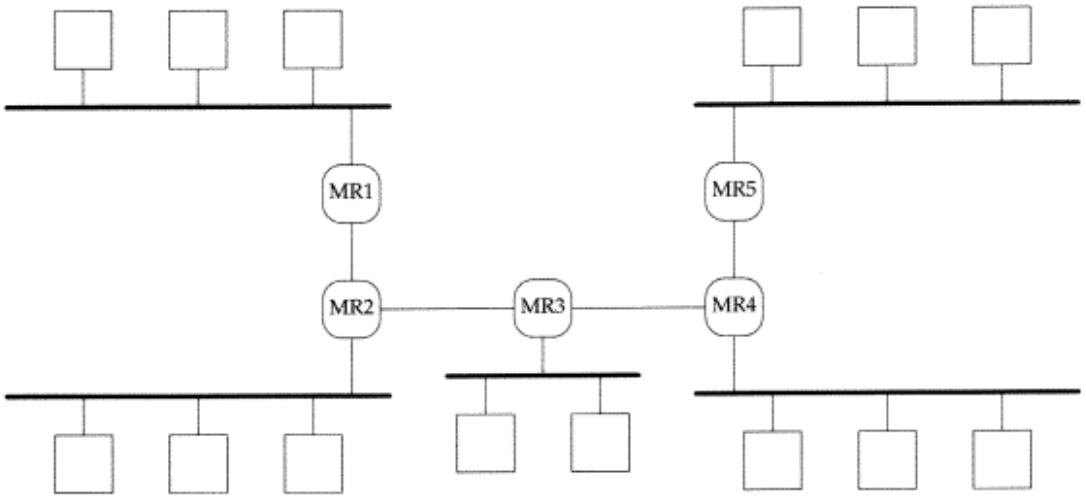


Figure 19.4 Five LANs connected with five multicast routers.

We show five LANs connected with five multicast routers.

Next assume that some program is started on five of the hosts (say a program that listens to a multicast audio session) and those five programs join a given multicast group. Each of the five hosts then joins that multicast group. We also assume that the multicast routers are all communicating with their neighbor multicast router using a *multicast routing protocol*, which we designate as just *MRP*. We show this in Figure 19.5.

When a process on a host joins a multicast group, that host sends an IGMP message to any attached multicast routers telling them that the host has just joined that group. The multicast routers then exchange this information using the multicast routing protocol so that each multicast router knows what to do if it receives a packet destined to the multicast address.

> Multicast routing is still a research topic and could easily consume a book on its own. [Maufer and Semeria 1997] provide an introduction and overview.

We now assume that a process on the host on the top left starts sending packets destined to the multicast address. Say this process is sending the audio packets that the multicast receivers are waiting to receive. We show these packets in Figure 19.6.

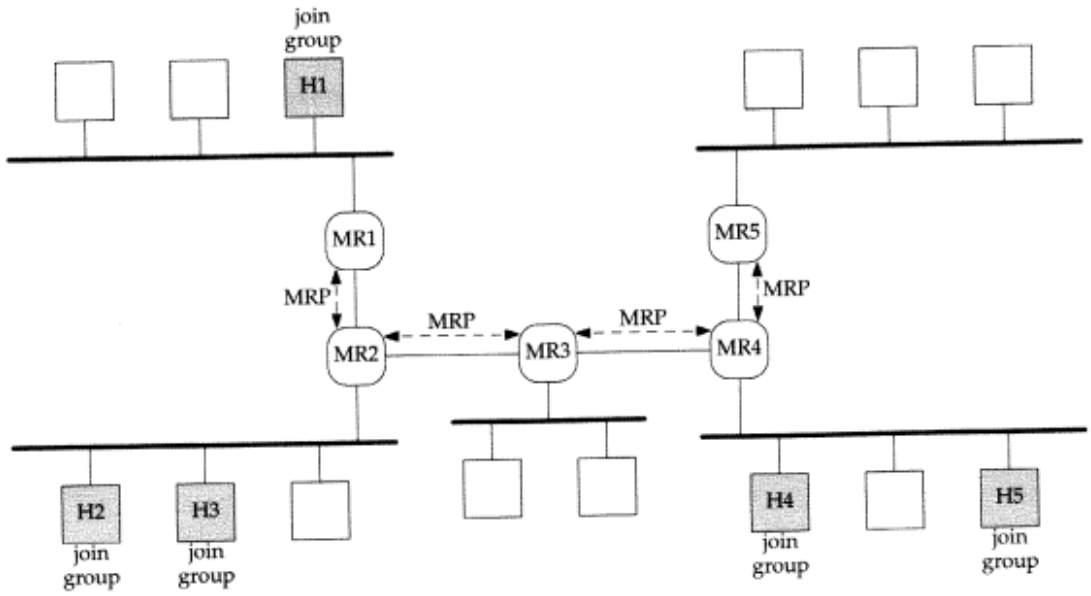


Figure 19.5 Five hosts join a multicast group on a WAN.

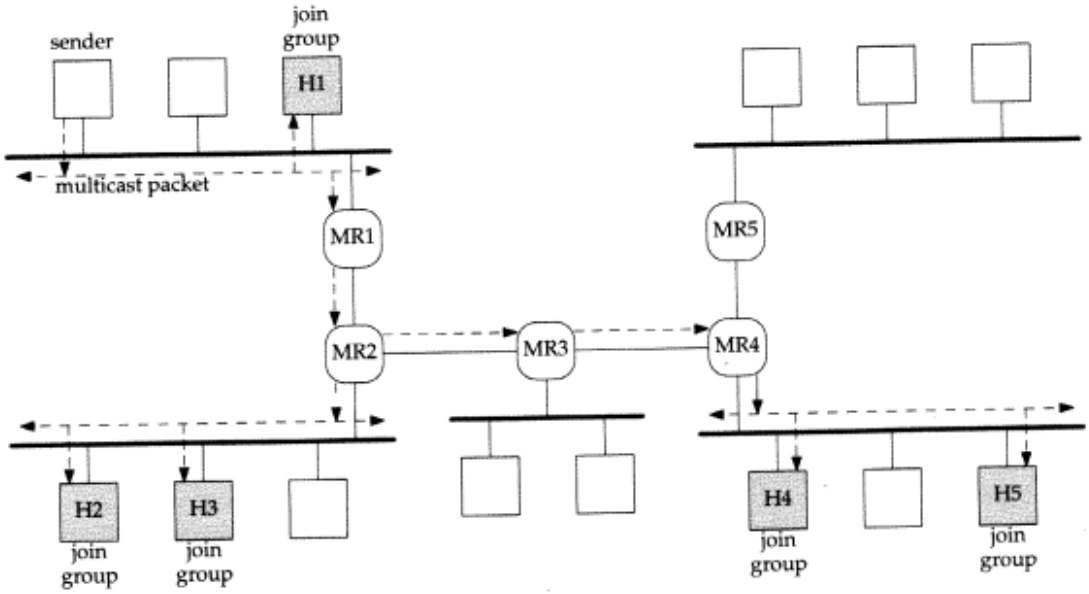


Figure 19.6 Sending multicast packets on a WAN.

We can follow the steps taken as the multicast packets go from the sender to all the receivers.

- The packets are multicast on the top left LAN by the sender. The receiver H1 receives these (since it has joined the group) as does MR1 (since a multicast router must receive all multicast packets).
- MR1 forwards the multicast packet to MR2, because the multicast routing protocol has informed MR1 that MR2 needs to receive packets destined to this group.
- MR2 multicasts the packet onto its attached LAN, since hosts H2 and H3 belong to the group. It also makes a copy of the packet and sends it to MR3.

  Making a copy of the packet, as MR2 does here, is something unique to multicast forwarding. A unicast packet is never duplicated as it is forwarded by routers.
- MR3 sends the multicast packet to MR4 but MR3 does not multicast a copy on its attached LAN, because we assume none of the hosts on the LAN has joined the group.
- MR4 multicasts the packet onto its attached LAN, since hosts H4 and H5 belong to the group. It does not make a copy and send it to MR5, because none of the hosts on MR5's attached LAN belong to the group and MR4 knows this based on the multicast routing information it has exchanged with MR5.

Two less desirable alternatives to multicasting on a WAN are *broadcast flooding* and sending individual copies to each receiver. In the first case the packets would be broadcast by the sender, and each router would broadcast the packet out each of its interfaces, except the arriving interface. It should be obvious that this increases the number of uninterested hosts and routers that must deal with the packet.

In the second case the sender must know the IP address of all the receivers and send each one a copy. With the five receivers that we show in Figure 19.6 this would require five packets on the sender's LAN, four packets going from MR1 to MR2, and two packets going from MR2 to MR3 to MR4.

19.5 Multicast Socket Options

The API support for multicasting requires only five new socket options. Figure 19.7 shows these five socket options, and the datatype of the argument expected in the call to `getsockopt` or `setsockopt` for IPv4 and IPv6. A pointer to a variable of the datatype shown is the fourth argument to `getsockopt` and `setsockopt`. All five of these options are valid with `setsockopt`, but the two that join and leave a multicast group are not allowed with `getsockopt`.

Command	Datatype	Description
IP_ADD_MEMBERSHIP	struct ip_mreq	join a multicast group
IP_DROP_MEMBERSHIP	struct ip_mreq	leave a multicast group
IP_MULTICAST_IF	struct in_addr	specify default interface for outgoing multicasts
IP_MULTICAST_TTL	u_char	specify TTL for outgoing multicasts
IP_MULTICAST_LOOP	u_char	enable or disable loopback of outgoing multicasts
IPV6_ADD_MEMBERSHIP	struct ipv6_mreq	join a multicast group
IPV6_DROP_MEMBERSHIP	struct ipv6_mreq	leave a multicast group
IPV6_MULTICAST_IF	u_int	specify default interface for outgoing multicasts
IPV6_MULTICAST_HOPS	int	specify hop limit for outgoing multicasts
IPV6_MULTICAST_LOOP	u_int	enable or disable loopback of outgoing multicasts

Figure 19.7 Multicast socket options.

The IPv4 TTL and loopback options take a `u_char` argument, while the IPv6 hop limit and loopback options take an `int` and a `u_int` argument, respectively. A common programming error with the IPv4 multicast options is to call `setsockopt` with an `int` argument to specify the TTL or loopback (which is not allowed; pp. 354–355 of TCPv2), since most of the other socket options in Figure 7.1 have integer arguments. The change with IPv6 should be less error prone.

We now describe each of these five socket options in more detail. Notice that the five options are conceptually identical between IPv4 and IPv6; only the name and argument type differs.

`IP_ADD_MEMBERSHIP`, `IPV6_ADD_MEMBERSHIP`

Join a multicast group on a specified local interface. We specify the local interface with one of its unicast addresses for IPv4 or with the interface index for IPv6. The following two structures are used when joining or leaving a group:

```
struct ip_mreq {
  struct in_addr   imr_multiaddr;    /* IPv4 class D multicast addr */
  struct in_addr   imr_interface;    /* IPv4 addr of local interface */
};

struct ipv6_mreq {
  struct in6_addr  ipv6mr_multiaddr; /* IPv6 multicast addr */
  unsigned int     ipv6mr_interface; /* interface index, or 0 */
};
```

If the local interface is specified as the wildcard address (`INADDR_ANY` for IPv4) or an index of 0 for IPv6, then the local interface is chosen by the kernel.

We say that a host belongs to a given multicast group on a given interface if one or more processes currently belongs to that group on that interface.

More than one join is allowed on a given socket but each join must be for a different multicast address, or for the same multicast address but on a different interface from previous joins for that address on this socket. This can be used on a multihomed host where, for example, one socket is created and then for each interface a join is performed for a given multicast address.

Recall from Figure 19.2 that IPv6 multicast addresses have an explicit scope field as part of the address. As we noted, IPv6 multicast addresses that differ only in scope represent different groups. Therefore, if an implementation of the Network Time Protocol wanted to receive all NTP packets, regardless of scope, it would have to join `ff01::101` (node-local), `ff02::101` (link-local), `ff05::101` (site-local), `ff08::101` (organization-local), and `ff0e::101` (global). All the joins could be performed on a single socket, and the `IPV6_PKTINFO` socket option could be set (Section 20.8) to have `recvmsg` return the destination address of each datagram.

Most implementations have a limit on the number of joins that are allowed per socket. This limit is 20 for Berkeley-derived implementations.

When the interface on which to join is not specified, Berkeley-derived kernels look up the multicast address in the normal IP routing table and use the resulting interface (p. 357 of TCPv2). Some systems install a route for all multicast addresses (that is, a route with a destination of 224.0.0.0/8 for IPv4) upon initialization to handle this scenario.

The change was made with IPv6 to use an interface index to specify the interface, instead of the local unicast address that is used with IPv4, to allow joins on unnumbered interfaces and tunnel endpoints. This is why the interface name and index functions that we described in Section 17.6 were introduced with RFC 2133 [Gilligan et al. 1997].

`IP_DROP_MEMBERSHIP`, `IPV6_DROP_MEMBERSHIP`

Leave a multicast group on a specified local interface. The same structures that we just showed for joining a group are used with this socket option. If the local interface is not specified (that is, the value is `INADDR_ANY` for IPv4 or an interface index of 0 for IPv6), the first matching multicasting group membership is dropped.

If a process joins a group but never explicitly leaves the group, when the socket is closed (either explicitly or on process termination), the membership is dropped automatically. It is possible for multiple processes on a host to each join the same group, in which case the host remains a member of that group until the last process leaves the group.

`IP_MULTICAST_IF`, `IPV6_MULTICAST_IF`

Specify the interface for outgoing multicast datagrams sent on this socket. This interface is specified as either an `in_addr` structure for IPv4 or an interface index for IPv6. If the value specified is `INADDR_ANY` for IPv4 or an interface index of 0 for IPv6, this removes any interface previously assigned by this socket option, and the system will choose the interface each time a datagram is sent.

Be careful to distinguish between the local interface specified (or chosen) when a process joins a group (the interface on which arriving multicast datagrams will be received), and the local interface specified (or chosen) when a multicast datagram is output.

Berkeley-derived kernels choose the default interface for an outgoing multicast datagram by searching the normal IP routing table for a route to the destination multicast address, and the corresponding interface is used. This is the same technique used to choose the receiving interface if the process does not specify one when joining a group. The assumption is that if a route exists for a given multicast address (perhaps the default route in the routing table), then the resulting interface should be used for input and output.

`IP_MULTICAST_TTL`, `IPV6_MULTICAST_HOPS`
: Set the IPv4 TTL or the IPv6 hop limit for outgoing multicast datagrams. If this is not specified, both default to 1, which restricts the datagram to the local subnet.

`IP_MULTICAST_LOOP`, `IPV6_MULTICAST_LOOP`
: Enable or disable local loopback of multicast datagrams. By default loopback is enabled: a copy of each multicast datagram sent by a process on the host will also be looped back and processed as a received datagram by that host, if the host belongs to that multicast group on the outgoing interface.

  This is similar to broadcasting, where we saw that broadcasts sent on a host are also processed as a received datagram on that host (Figure 18.4). (But with broadcasting there is no way to disable this loopback.) This means that if a process belongs to the multicast group to which it is sending datagrams, it will receive its own transmissions.

  The loopback that is being described here is an internal loopback performed at the IP layer or higher. Should the interface hear its own transmissions, RFC 1112 [Deering 1989] requires that the driver discard these copies. This RFC also states that the loopback option defaults on as "a performance optimization for upper-layer protocols that restrict the membership of a group to one process per host (such as a routing protocol)."

The first two pairs of socket options (`ADD_MEMBERSHIP` and `DROP_MEMBERSHIP`) affect the *receiving* of multicast datagrams, while the last three pairs affect the *sending* of multicast datagrams (outgoing interface, TTL or hop limit, and loopback). We mentioned earlier that nothing special is required to send a multicast datagram. If none of the multicast socket options is specified before sending a multicast datagram, the interface for the outgoing datagram will be chosen by the kernel, the TTL or hop limit will be 1, and a copy will be looped back.

To receive a multicast datagram a process must join the multicast group and it must also `bind` a UDP socket to the port number that will be used as the destination port number for datagrams sent to the group. The two operations are distinct and both are required. Joining the group tells the host's IP layer and datalink layer to receive multicast datagrams sent to that group. Binding the port is how the application specifies to UDP that it wants to receive datagrams sent to that port. Some applications also `bind` the multicast address to the socket, in addition to the port. This prevents any other datagrams that might be received for that port from being delivered to the socket.

Historically, Berkeley-derived implementations only require that *some* socket on the host join the multicast group, not necessarily the socket that binds the port and then receives the

multicast datagrams. There is the potential, however, with these implementations for multicast datagrams to be delivered to applications that are not multicast aware. Newer multicast kernels now require that the process bind the port and set any multicast socket option for the socket, the latter being an indication that the application is multicast aware. The most common multicast socket option to set is a join of the group. Solaris 2.5 differs slightly and only delivers received multicast datagrams to a socket that has both joined the group and bound the port. For portability, all multicast applications should join the group and bind the port.

Some older multicast-capable hosts do not allow the `bind` of a multicast address to a socket. For portability an application may wish to ignore a `bind` error for a multicast address.

19.6 mcast_join and Related Functions

Although the five multicast socket options for IPv4 are similar to the five multicast socket options for IPv6, there are enough differences that protocol-independent code using multicasting becomes complicated with lots of `#ifdefs`. A better solution is to hide the differences within the following eight functions.

```
#include "unp.h"

int mcast_join(int sockfd, const struct sockaddr *sa, socklen_t salen,
               const char *ifname, u_int ifindex);

int mcast_leave(int sockfd, const struct sockaddr *sa, socklen_t salen);

int mcast_set_if(int sockfd, const char *ifname, u_int ifindex);

int mcast_set_loop(int sockfd, int flag);

int mcast_set_ttl(int sockfd, int ttl);
```

All above return: 0 if OK, –1 on error

```
int mcast_get_if(int sockfd);
```

Returns: nonnegative interface index if OK, –1 on error

```
int mcast_get_loop(int sockfd);
```

Returns: current loopback flag if OK, –1 on error

```
int mcast_get_ttl(int sockfd);
```

Returns: current TTL or hop limit if OK, –1 on error

`mcast_join` joins the multicast group whose IP address is contained within the socket address structure pointed to by *sa*, and whose length is specified by *salen*. We can specify the interface on which to join the group by either the interface name (a nonnull *ifname*) or a nonzero interface index (*ifindex*). If neither is specified, the kernel chooses the interface on which the group is joined. Recall that with IPv6 the interface is

specified to the socket option by its index. If a name is specified for an IPv6 socket, we call `if_nametoindex` to obtain the index. With the IPv4 socket option the interface is specified by its unicast IP address. If a name is specified for an IPv4 socket, we call `ioctl` with a request of `SIOCGIFADDR` to obtain the unicast IP address for the interface. If an index is specified for an IPv4 socket, we first call `if_indextoname` to obtain the name and then process the name as just described.

> An interface name, such as `le0` or `ether0`, is normally the way users specify interfaces, and not with either the IP address or the index. `tcpdump`, for example, is one of the few programs that lets the user specify an interface, and its `-i` option takes an interface name as the argument.

`mcast_leave` leaves the multicast group whose IP address is contained within the socket address structure pointed to by *sa*.

`mcast_set_if` sets the default interface index for outgoing multicast datagrams. If *ifname* is nonnull, then it specifies the interface name; otherwise if *ifindex* is greater than 0, then it specifies the interface index. For IPv6 the name is mapped to an index using `if_nametoindex`. For IPv4 the mapping from either a name or an index into the interface's unicast IP address is done as described for `mcast_join`.

`mcast_set_loop` sets the loopback option to either 0 or 1, and `mcast_set_ttl` sets either the IPv4 TTL or the IPv6 hop limit. The three `mcast_get_XXX` functions return the corresponding value.

Example: `mcast_join` Function

Figure 19.8 shows the first half of our `mcast_join` function. This half handles an IPv4 socket

Handle index

11-19 The IPv4 multicast address in the socket address structure is copied into an `ip_mreq` structure. If an index was specified, `if_indextoname` is called, storing the name into our `ifreq` structure. If this succeeds, we branch ahead to issue the `ioctl`.

Handle name

20-27 The caller's name is copied into an `ifreq` structure and an `ioctl` of `SIOCGIFADDR` returns the unicast address associated with this name. Upon success the IPv4 address is copied into the `imr_interface` member of the `ip_mreq` structure.

Specify default

28-29 If an index was not specified, and a name was not specified, the interface is set to the wildcard address, telling the kernel to choose the interface.

30-31 `setsockopt` performs the join.

lib/mcast_join.c

```
 1 #include    "unp.h"
 2 #include    <net/if.h>

 3 int
 4 mcast_join(int sockfd, const SA *sa, socklen_t salen,
 5            const char *ifname, u_int ifindex)
 6 {
 7     switch (sa->sa_family) {
 8     case AF_INET:{
 9             struct ip_mreq mreq;
10             struct ifreq ifreq;

11             memcpy(&mreq.imr_multiaddr,
12                    &((struct sockaddr_in *) sa)->sin_addr,
13                    sizeof(struct in_addr));

14             if (ifindex > 0) {
15                 if (if_indextoname(ifindex, ifreq.ifr_name) == NULL) {
16                     errno = ENXIO;  /* i/f index not found */
17                     return (-1);
18                 }
19                 goto doioctl;
20             } else if (ifname != NULL) {
21                 strncpy(ifreq.ifr_name, ifname, IFNAMSIZ);
22               doioctl:
23                 if (ioctl(sockfd, SIOCGIFADDR, &ifreq) < 0)
24                     return (-1);
25                 memcpy(&mreq.imr_interface,
26                        &((struct sockaddr_in *) &ifreq.ifr_addr)->sin_addr,
27                        sizeof(struct in_addr));
28             } else
29                 mreq.imr_interface.s_addr = htonl(INADDR_ANY);

30             return (setsockopt(sockfd, IPPROTO_IP, IP_ADD_MEMBERSHIP,
31                                &mreq, sizeof(mreq)));
32         }
```

lib/mcast_join.c

Figure 19.8 Join a multicast group: IPv4 socket.

The second half of the function, which handles IPv6 sockets, is shown in Figure 19.9.

Handle index, name, or default

36–50 First the IPv6 multicast address is copied from the socket address structure into the `ipv6_mreq` structure. If an index was specified, it is stored in the `ipv6mr_interface` member. Else, if a name was specified, the index is obtained by calling `if_nametoindex`. Otherwise the interface index is set to 0 for `setsockopt`, telling the kernel to choose the interface. The group is joined.

lib/mcast_join.c

```
33 #ifdef  IPV6
34     case AF_INET6:{
35             struct ipv6_mreq mreq6;

36             memcpy(&mreq6.ipv6mr_multiaddr,
37                    &((struct sockaddr_in6 *) sa)->sin6_addr,
38                    sizeof(struct in6_addr));

39             if (ifindex > 0) {
40                 mreq6.ipv6mr_interface = ifindex;
41             } else if (ifname != NULL) {
42                 if ( (mreq6.ipv6mr_interface = if_nametoindex(ifname)) == 0) {
43                     errno = ENXIO;  /* i/f name not found */
44                     return (-1);
45                 }
46             } else
47                 mreq6.ipv6mr_interface = 0;

48             return (setsockopt(sockfd, IPPROTO_IPV6, IPV6_ADD_MEMBERSHIP,
49                                &mreq6, sizeof(mreq6)));
50         }
51 #endif

52     default:
53         errno = EPROTONOSUPPORT;
54         return (-1);
55     }
56 }
```

lib/mcast_join.c

Figure 19.9 Join a multicast group: IPv6 socket.

Example: `mcast_set_loop` Function

Figure 19.10 shows our `mcast_set_loop` function.

Since the argument is a socket descriptor, and not a socket address structure, we call our `sockfd_to_family` function to obtain the address family of the socket. The appropriate socket option is set.

We do not show the source code for all remaining `mcast_XXX` functions, but it is freely available (see the Preface).

19.7 `dg_cli` Function Using Multicasting

We modify our `dg_cli` function from Figure 18.5 by just removing the call to `setsockopt`. As we said earlier, none of the multicast socket options needs to be set to send a multicast datagram, if the default settings for the outgoing interface, TTL, and loopback option are OK. We run our program specifying the all-hosts group as the destination address:

lib/mcast_set_loop.c

```
 1 #include    "unp.h"

 2 int
 3 mcast_set_loop(int sockfd, int onoff)
 4 {
 5     switch (sockfd_to_family(sockfd)) {
 6     case AF_INET:{
 7             u_char  flag;

 8             flag = onoff;
 9             return (setsockopt(sockfd, IPPROTO_IP, IP_MULTICAST_LOOP,
10                                &flag, sizeof(flag)));
11         }

12 #ifdef  IPV6
13     case AF_INET6:{
14             u_int   flag;

15             flag = onoff;
16             return (setsockopt(sockfd, IPPROTO_IPV6, IPV6_MULTICAST_LOOP,
17                                &flag, sizeof(flag)));
18         }
19 #endif

20     default:
21         errno = EPROTONOSUPPORT;
22         return (-1);
23     }
24 }
```

lib/mcast_set_loop.c

Figure 19.10 Set the multicast loopback option.

```
solaris % udpcli01 224.0.0.1
hi there
from 206.62.226.40: Fri Jul 18 13:02:41 1997    Linux
from 206.62.226.35: Fri Jul 18 13:02:41 1997    BSD/OS
from 206.62.226.43: Fri Jul 18 13:02:41 1997    AIX
from 206.62.226.34: Fri Jul 18 13:02:41 1997    BSD/OS
from 206.62.226.42: Fri Jul 18 13:02:41 1997    Digital Unix
```

The five hosts respond because each host is multicast capable and has therefore joined the all-hosts group, and because a process has bound the destination UDP port (13, the daytime server). Therefore the arriving multicast datagram is delivered to the socket. This server is normally part of `inetd`. Each reply is unicast, because the source address of the request, which is used by each server as the destination address of the reply, is a unicast address.

> The host `sunos5` is the only multicast-capable host on the subnet that does not respond. We mentioned earlier that Solaris requires that the socket join the group, which is not the case for this example using the standard daytime server.

IP Fragmentation and Multicasts

We mentioned at the end of Section 18.4 that most systems do not allow the fragmentation of a broadcast datagram as a policy decision. Fragmentation is OK with multicasting, as we can easily verify using the same file with a 2000-byte line.

```
bsdi % udpcli01 224.0.0.1 < 2000line
from 206.62.226.35: Fri Jul 18 14:31:50 1997
from 206.62.226.34: Fri Jul 18 14:31:50 1997
from 206.62.226.40: Fri Jul 18 14:31:50 1997
from 206.62.226.43: Fri Jul 18 14:31:50 1997
from 206.62.226.42: Fri Jul 18 14:31:50 1997
```

Many implementations derived from 4.4BSD have a bug in the fragmentation of multicast datagrams: none of the fragments after the first are sent as link-layer multicasts.

19.8 Receiving MBone Session Announcements

To receive a multimedia conference on the MBone (Section B.2) a site needs to know only the multicast address of the conference and the UDP ports for the conference's data streams (audio and video, for example). *SAP*, the Session Announcement Protocol [Handley 1996], describes the way this is done (the packet headers and the frequency with which these announcements are multicast to the MBone) and *SDP*, the Session Description Protocol [Handley and Jacobson 1997], describes the contents of these announcements (how the multicast addresses and UDP port numbers are specified). A site wishing to announce a session on the MBone periodically sends a multicast packet containing a description of the session to a well-known multicast group and UDP port. Sites on the MBone run a program named `sdr` to receive these announcements. This program does a lot: not only does it receive session announcements but it also provides an interactive user interface that displays the information and lets the user send announcements.

In this section we develop a simple program that only receives these session announcements to show an example of a simple multicast receiving program. Our goal is to show the simplicity of a multicast receiver, and not to delve into the details of this one application.

Figure 19.11 shows our `main` program that receives the periodic SAP/SDP announcements.

Well-known name and well-known port

2-3 The multicast address assigned for SAP announcements is 224.2.127.254 and its name is `sap.mcast.net`. All of the well-known multicast addresses (see `ftp://ftp.isi.edu/in-notes/iana/assignments/multicast-addresses`) appear in the DNS under the `mcast.net` hierarchy. The well-known UDP port is 9875.

Create UDP socket

12-17 We call our `udp_client` function to look up the name and port, and it fills in the appropriate socket address structure. We use the defaults if no command-line

mysdr/main.c

```
 1 #include    "unp.h"

 2 #define SAP_NAME    "sap.mcast.net"     /* default group name and port */
 3 #define SAP_PORT    "9875"

 4 void    loop(int, socklen_t);

 5 int
 6 main(int argc, char **argv)
 7 {
 8     int     sockfd;
 9     const int on = 1;
10     socklen_t salen;
11     struct sockaddr *sa;

12     if (argc == 1)
13         sockfd = Udp_client(SAP_NAME, SAP_PORT, (void **) &sa, &salen);
14     else if (argc == 4)
15         sockfd = Udp_client(argv[1], argv[2], (void **) &sa, &salen);
16     else
17         err_quit("usage: mysdr <mcast-addr> <port#> <interface-name>");

18     Setsockopt(sockfd, SOL_SOCKET, SO_REUSEADDR, &on, sizeof(on));
19     Bind(sockfd, sa, salen);

20     Mcast_join(sockfd, sa, salen, (argc == 4) ? argv[3] : NULL, 0);

21     loop(sockfd, salen);        /* receive and print */

22     exit(0);
23 }
```

mysdr/main.c

Figure 19.11 `main` program to receive SAP/SDP announcements.

arguments are specified; otherwise we take the multicast address, port, and interface name from the command-line arguments.

`bind` port

18–19 We set the `SO_REUSEADDR` socket option to allow multiple instances of this program to run on a host, and `bind` the port to the socket. By binding the multicast address to the socket we prevent the socket from receiving any other UDP datagrams that may be received for the port. Binding this multicast address is not required, but it provides filtering by the kernel of packets in which we are not interested.

Join multicast group

20 We call our `mcast_join` function to join the group. If the interface name was specified as a command-line argument, it is passed to our function; otherwise we let the kernel choose the interface on which the group is joined.

21 We call our `loop` function, shown in Figure 19.12, to read and print all the announcements.

mysdr/loop.c

```
 1 #include    "unp.h"

 2 void
 3 loop(int sockfd, socklen_t salen)
 4 {
 5     char    buf[MAXLINE + 1];
 6     socklen_t len;
 7     ssize_t n;
 8     struct sockaddr *sa;
 9     struct sap_packet {
10         uint32_t sap_header;
11         uint32_t sap_src;
12         char    sap_data[1];
13     } *sapptr;

14     sa = Malloc(salen);

15     for ( ; ; ) {
16         len = salen;
17         n = Recvfrom(sockfd, buf, MAXLINE, 0, sa, &len);
18         buf[n] = 0;             /* null terminate */

19         sapptr = (struct sap_packet *) buf;
20         if ((n -= 2 * sizeof(uint32_t)) <= 0)
21             err_quit("n = %d", n);
22         printf("From %s\n%s\n", Sock_ntop(sa, len), sapptr->sap_data);
23     }
24 }
```

mysdr/loop.c

Figure 19.12 Loop that receives and prints SAP/SDP announcements.

Packet format

9–13 `sap_packet` describes the SDP packet: a 32-bit SAP header, followed by a 32-bit source address, followed by the actual announcement. The announcement is simply lines of ISO 8859–1 text and cannot exceed 1024 bytes. Only one session announcement is allowed in each UDP datagram.

Read UDP datagram, print sender and contents

15–23 `recvfrom` waits for the next UDP datagram destined to our socket. When one arrives we place a null byte at the end of the buffer, skip over the two header fields, and print the result. We also print the IP address and port number of the sender of the multicast announcement.

Figure 19.13 shows some typical output from our program.

```
solaris % mysdr
From 128.102.84.134/2840
v=0
o=shuttle 3050400397 3051818822 IN IP4 128.102.84.134
s=NASA - Shuttle STS-79 Mission Coverage
i=Pre-launch and mission coverage of Shuttle Mission STS-79.  Launch expected 9/
16/96 with a mission duration anticipated of 9-10 days.  STS-79 is the 4th in a
```

```
series of joint docking missions between the Shuttle and the MIR Space Station.
This session is being offered by NASA - Ames Research Center as a public service
 to the MBone community.
u=http://www-pao.ksc.nasa.gov/kscpao/kscpao.htm
p=NASA ARC Digital Video Lab (415) 604-6145
e=NASA ARC Digital Video Lab <mallard@mail.arc.nasa.gov>
c=IN IP4 224.2.86.28/127
t=3051608400 3052472400
m=audio 19432 RTP/AVP 0
m=video 61192 RTP/AVP 31
```

Figure 19.13 Typical SAP/SDP announcement.

This announcement describes the NASA coverage on the MBone of a space shuttle mission. The SDP session description consists of numerous lines of the form

type=value

where the *type* is always one character and is case significant. The *value* is a structured text string that depends on the *type*. Spaces are not allowed around the equals sign. `v=0` is the version.

`o=` is the origin. `shuttle` is the username, `3050400397` is the session ID, `3051818822` is the version number for this announcement, `IN` is the network type, `IP4` is the address type, and `128.102.84.134` is the address. The five-tuple consisting of the username, session ID, network type, address type, and address form a globally unique identifier for the session.

`s=` defines the session name, and `i=` is information about the session. We have wrapped the latter every 80 characters. `u=` provides a URI (Uniform Resource Identifier) for more information about the session, and `p=` and `e=` provide a phone number and email address of someone responsible for the conference.

`c=` provides the connection information, which in this example specifies that it is IP based, using IPv4, with a multicast address of 224.2.86.28 and a TTL of 127. `t=` provides the starting time and stopping time, both in NTP units, which are seconds since January 1, 1900, UTC. The `m=` lines are the media announcements. The first of these two lines specifies that the audio is on port 19432 and its format is RTP, the Real-time Transport Protocol, using AVP, the Audio/Video Profile, with payload type 0 (which is a PCM coded signal channel audio, encoded at 8 Khz). The next `m=` line specifies that the video is on port 61192 and the RTP/AVP payload type of 31 is ITU H.261 video format. Notice that the audio and video are both multicast to the same address (224.2.86.28) but to different ports. This means that if a host wants only one of the two, both will be received by the IP layer once the host joins the group and the unwanted data will be discarded by UDP.

19.9 Sending and Receiving

The MBone session announcement program in the previous section received only multicast datagrams. We now develop a simple program that sends and receives multicast

datagrams. Our program consists of two parts. The first part sends a multicast datagram to a specific group every 5 seconds and the datagram contains the sender's hostname and process ID. The second part is an infinite loop that joins the multicast group to which the first part is sending and prints every received datagram (containing the hostname and process ID of the sender). This allows us to start the program on multiple hosts on a LAN and easily see which host is receiving datagrams from which senders.

Figure 19.14 shows the `main` function for our program.

mcast/main.c

```
 1 #include    "unp.h"

 2 void    recv_all(int, socklen_t);
 3 void    send_all(int, SA *, socklen_t);

 4 int
 5 main(int argc, char **argv)
 6 {
 7     int     sendfd, recvfd;
 8     const int on = 1;
 9     socklen_t salen;
10     struct sockaddr *sasend, *sarecv;

11     if (argc != 3)
12         err_quit("usage: sendrecv <IP-multicast-address> <port#>");

13     sendfd = Udp_client(argv[1], argv[2], (void **) &sasend, &salen);

14     recvfd = Socket(sasend->sa_family, SOCK_DGRAM, 0);

15     Setsockopt(recvfd, SOL_SOCKET, SO_REUSEADDR, &on, sizeof(on));

16     sarecv = Malloc(salen);
17     memcpy(sarecv, sasend, salen);
18     Bind(recvfd, sarecv, salen);

19     Mcast_join(recvfd, sasend, salen, NULL, 0);
20     Mcast_set_loop(sendfd, 0);

21     if (Fork() == 0)
22         recv_all(recvfd, salen);    /* child -> receives */

23     send_all(sendfd, sasend, salen);    /* parent -> sends */
24 }
```

mcast/main.c

Figure 19.14 Create sockets, `fork`, and start sender and receiver.

We create two sockets, one for sending and one for receiving. We want the receiving socket to `bind` the multicast group and port, say 239.255.1.2 port 8888. (Recall that we could just `bind` the wildcard IP address and port 8888, but binding the multicast address prevents the socket from receiving any other datagrams that might arrive destined for port 8888.) We then want the receiving socket to join the multicast group. The sending socket will send datagrams to this same multicast address and port, say 239.255.1.2 port 8888. But if we try to use a single socket for sending and receiving, the source protocol address is 239.255.1.2.8888 from the `bind` (using `netstat` notation)

and the destination protocol address for the `sendto` is also 239.255.1.2.8888. But the source protocol address that is bound to the socket becomes the source IP address of the UDP datagram, and RFC 1122 [Braden 1989] forbids an IP datagram from having a source IP address that is a multicast address or a broadcast address. (See Exercise 19.2 also.) Therefore we create two sockets: one for sending and one for receiving.

Create sending socket

13 Our `udp_client` function creates the sending socket, processing the two command-line arguments that specify the multicast address and port number. This function also returns a socket address structure that is ready for calls to `sendto` along with the length of this socket address structure.

Create receiving socket and `bind` multicast address and port

14–18 We create the receiving socket, using the same address family that was used for the sending socket. We set the `SO_REUSEADDR` socket option to allow multiple instances of this program to run at the same time on a host. We then allocate room for a socket address structure for this socket, copy its contents from the sending socket address structure (whose address and port were taken from the command-line arguments), and `bind` the multicast address and port to the receiving socket.

Join multicast group and turn off loopback

19–20 We call our `mcast_join` function to join the multicast group on the receiving socket, and our `mcast_set_loop` function to disable the loopback feature on the sending socket. For the join we specify the interface name as a null pointer and the interface index as 0, telling the kernel to choose the interface.

`fork` and call appropriate functions

21–23 We `fork` and then the child is the receive loop and the parent is the send loop.

Our `send_all` function, which sends one multicast datagram every 5 seconds, is shown in Figure 19.15. The `main` function passes as arguments the socket descriptor, a pointer to a socket address structure containing the multicast destination and the port, and the structure's length.

Obtain hostname and form datagram contents

9–11 We obtain the hostname from the `uname` function and build the output line containing it and the process ID.

Send datagram, then go to sleep

12–15 We send a datagram and then `sleep` for 5 seconds.

The `recv_all` function, which is the infinite receive loop, is shown in Figure 19.16.

Allocate socket address structure

9 A socket address structure is allocated to receive the sender's protocol address for each call to `recvfrom`.

Read and print datagrams

10–15 Each datagram is read by `recvfrom`, null terminated, and printed.

mcast/send.c

```
 1 #include    "unp.h"
 2 #include    <sys/utsname.h>

 3 #define SENDRATE    5           /* send one datagram every 5 seconds */

 4 void
 5 send_all(int sendfd, SA *sadest, socklen_t salen)
 6 {
 7     static char line[MAXLINE];  /* hostname and process ID */
 8     struct utsname myname;

 9     if (uname(&myname) < 0)
10         err_sys("uname error");;
11     snprintf(line, sizeof(line), "%s, %d\n", myname.nodename, getpid());

12     for ( ; ; ) {
13         Sendto(sendfd, line, strlen(line), 0, sadest, salen);

14         sleep(SENDRATE);
15     }
16 }
```

mcast/send.c

Figure 19.15 Send a multicast datagram every 5 seconds.

mcast/recv.c

```
 1 #include    "unp.h"

 2 void
 3 recv_all(int recvfd, socklen_t salen)
 4 {
 5     int     n;
 6     char    line[MAXLINE + 1];
 7     socklen_t len;
 8     struct sockaddr *safrom;

 9     safrom = Malloc(salen);

10     for ( ; ; ) {
11         len = salen;
12         n = Recvfrom(recvfd, line, MAXLINE, 0, safrom, &len);

13         line[n] = 0;            /* null terminate */
14         printf("from %s: %s", Sock_ntop(safrom, len), line);
15     }
16 }
```

mcast/recv.c

Figure 19.16 Receive all multicast datagrams for a group that we have joined.

19.10 SNTP: Simple Network Time Protocol

NTP, the Network Time Protocol, is a sophisticated protocol for synchronizing clocks across a WAN or a LAN, and can often achieve millisecond accuracy. RFC 1305 [Mills

1992] describes the protocol in detail and RFC 2030 [Mills 1996] describes SNTP, a simplified version intended for hosts that do not need the complexity of a complete NTP implementation. It is common for a few hosts on a LAN to synchronize their clocks across the Internet to other NTP hosts, and then redistribute this time on the LAN using either broadcasting or multicasting.

In this section we develop an SNTP client that listens for NTP broadcasts or multicasts on all attached networks and then prints the time difference between the NTP packet and the host's current time-of-day. We do not try to adjust the time-of-day, as that takes superuser privileges, although that would be a trivial addition to the code.

The file `ntp.h`, shown in Figure 19.17, contains some basic definitions of the NTP packet format.

ssntp/ntp.h

```
1  #define JAN_1970     2208988800UL    /* 1970 - 1900 in seconds */

2  struct l_fixedpt {                   /* 64-bit fixed-point */
3      uint32_t int_part;
4      uint32_t fraction;
5  };

6  struct s_fixedpt {                   /* 32-bit fixed-point */
7      u_short int_part;
8      u_short fraction;
9  };

10 struct ntpdata {                     /* NTP header */
11     u_char  status;
12     u_char  stratum;
13     u_char  ppoll;
14     int     precision:8;
15     struct s_fixedpt distance;
16     struct s_fixedpt dispersion;
17     uint32_t refid;
18     struct l_fixedpt reftime;
19     struct l_fixedpt org;
20     struct l_fixedpt rec;
21     struct l_fixedpt xmt;
22 };

23 #define VERSION_MASK    0x38
24 #define MODE_MASK       0x07

25 #define MODE_CLIENT     3
26 #define MODE_SERVER     4
27 #define MODE_BROADCAST  5
```

ssntp/ntp.h

Figure 19.17 `ntp.h` header: NTP packet format and definitions.

2-22 `l_fixedpt` defines the 64-bit fixed point values used by NTP for timestamps and `s_fixedpt` defines the 32-bit fixed point values that are also used by NTP. The `ntpdata` structure is the 48-byte NTP packet format.

Figure 19.18 shows the `main` function.

ssntp/main.c

```
 1 #include    "sntp.h"

 2 int
 3 main(int argc, char **argv)
 4 {
 5     int     sockfd;
 6     char    buf[MAXLINE];
 7     ssize_t n;
 8     socklen_t salen, len;
 9     struct ifi_info *ifi;
10     struct sockaddr *mcastsa, *wild, *from;
11     struct timeval now;

12     if (argc != 2)
13         err_quit("usage: ssntp <IPaddress>");

14     sockfd = Udp_client(argv[1], "ntp", (void **) &mcastsa, &salen);

15     wild = Malloc(salen);
16     memcpy(wild, mcastsa, salen);   /* copy family and port */
17     sock_set_wild(wild, salen);
18     Bind(sockfd, wild, salen);  /* bind wildcard */

19 #ifdef  MCAST
20         /* obtain interface list and process each one */
21     for (ifi = Get_ifi_info(mcastsa->sa_family, 1); ifi != NULL;
22          ifi = ifi->ifi_next) {
23         if (ifi->ifi_flags & IFF_MULTICAST) {
24             Mcast_join(sockfd, mcastsa, salen, ifi->ifi_name, 0);
25             printf("joined %s on %s\n",
26                    Sock_ntop(mcastsa, salen), ifi->ifi_name);
27         }
28     }
29 #endif

30     from = Malloc(salen);
31     for ( ; ; ) {
32         len = salen;
33         n = Recvfrom(sockfd, buf, sizeof(buf), 0, from, &len);
34         Gettimeofday(&now, NULL);
35         sntp_proc(buf, n, &now);
36     }
37 }
```

ssntp/main.c

Figure 19.18 `main` function.

Get multicast IP address

12-14 When the program is executed, the user must specify the multicast address to join as the command-line argument. With IPv4 this would be 224.0.1.1 or the name `ntp.mcast.net`. With IPv6 this would be `ff05::101` for the site-local scope NTP. Our `udp_client` function allocates space for a socket address structure of the correct type (either IPv4 or IPv6) and stores the multicast address and port in that structure. If this program is run on a host that does not support multicasting, any IP address can be specified, as only the address family and port are used from this structure. Note that

our `udp_client` function does not `bind` the address to the socket; it just creates the socket and fills in the socket address structure.

Bind wildcard address to socket

15-18 We allocate space for another socket address structure and fill it in by copying the structure that was filled in by `udp_client`. This sets the address family and port. We call our `sock_set_wild` function to set the IP address to the wildcard and then call `bind`.

Get interface list

20-22 Our `get_ifi_info` function returns information on all the interfaces and addresses. The address family that we ask for is taken from the socket address structure that was filled in by `udp_client` based on the command-line argument.

Join the multicast group

23-27 We call our `mcast_join` function to join the multicast group specified by the command-line argument for each multicast-capable interface. All these joins are done on the one socket that this program uses. As we said earlier, there is normally a limit of 20 joins per socket, but few multihomed hosts have that many interfaces.

Read and process all NTP packets

30-36 Another socket address structure is allocated to hold the address returned by `recvfrom` and the program enters an infinite loop, reading all the NTP packets that the host receives and calling our `sntp_proc` function (described next) to process the packet. Since the socket was bound to the wildcard address, and since the multicast group was joined on all multicast-capable interfaces, the socket should receive any unicast, broadcast, or multicast NTP packet that the host receives. Before calling `sntp_proc` we call `gettimeofday` to fetch the current time, because `sntp_proc` calculates the difference between the time in the packet and the current time.

Our `sntp_proc` function, shown in Figure 19.19, processes the actual NTP packet.

Validate packet

10-21 We first check the size of the packet and then print the version, mode, and server stratum. If the mode is `MODE_CLIENT`, then the packet is a client request, not a server reply, and we ignore it.

Obtain transmit time from NTP packet

22-34 The field in the NTP packet that we are interested in is `xmt`, the transmit timestamp, which is the 64-bit fixed-point time at which the packet was sent by the server. Since NTP timestamps count seconds beginning in 1900 and Unix timestamps count seconds beginning in 1970, we first subtract `JAN_1970` (the number of seconds in these 70 years) from the integer part.

The fractional part is a 32-bit unsigned integer between 0 and 4,294,967,295, inclusive. This is copied from a 32-bit integer (`useci`) to a double precision floating-point variable (`usecf`) and then divided by 4,294,967,296 (2^{32}). The result is greater than or equal to 0.0 and less than 1.0. We multiply this by 1,000,000, the number of microseconds in a second, storing the result as a 32-bit unsigned integer in the variable `useci`.

ssntp/sntp_proc.c

```
 1 #include    "sntp.h"

 2 void
 3 sntp_proc(char *buf, ssize_t n, struct timeval *nowptr)
 4 {
 5     int     version, mode;
 6     uint32_t nsec, useci;
 7     double  usecf;
 8     struct timeval curr, diff;
 9     struct ntpdata *ntp;

10     if (n < sizeof(struct ntpdata)) {
11         printf("\npacket too small: %d bytes\n", n);
12         return;
13     }
14     ntp = (struct ntpdata *) buf;
15     version = (ntp->status & VERSION_MASK) >> 3;
16     mode = ntp->status & MODE_MASK;
17     printf("\nv%d, mode %d, strat %d, ", version, mode, ntp->stratum);
18     if (mode == MODE_CLIENT) {
19         printf("client\n");
20         return;
21     }
22     nsec = ntohl(ntp->xmt.int_part) - JAN_1970;
23     useci = ntohl(ntp->xmt.fraction);    /* 32-bit integer fraction */
24     usecf = useci;                /* integer fraction -> double */
25     usecf /= 4294967296.0;        /* divide by 2**32 -> [0, 1.0) */
26     useci = usecf * 1000000.0;    /* fraction -> parts per million */

27     curr = *nowptr;               /* make a copy as we might modify it below */
28     if ( (diff.tv_usec = curr.tv_usec - useci) < 0) {
29         diff.tv_usec += 1000000;
30         curr.tv_sec--;
31     }
32     diff.tv_sec = curr.tv_sec - nsec;
33     useci = (diff.tv_sec * 1000000) + diff.tv_usec;      /* diff in microsec */
34     printf("clock difference = %d usec\n", useci);
35 }
```

ssntp/sntp_proc.c

Figure 19.19 `sntp_proc` function: process the NTP packet.

This is the number of microseconds and will be between 0 and 999,999 (see Exercise 19.8). We convert to microseconds because the Unix timestamp returned by `gettimeofday` is returned as two integers: the number of seconds since January 1, 1970, UTC, along with the number of microseconds. We then calculate and print the difference between the host's time-of-day and the NTP server's time-of-day, in microseconds.

One thing that our program does not take into account is the network delay between the server and the client. But we assume that the NTP packets are normally received as a broadcast or multicast on a LAN, in which case the network delay should be only a few milliseconds.

If we run this program on our host `solaris` with an NTP server on our host `bsdi` that is multicasting NTP packets to the Ethernet every 64 seconds, we have the following output:

```
solaris # ssntp 224.0.1.1
joined 224.0.1.1.123 on lo0
joined 224.0.1.1.123 on le0

v3, mode 5, strat 3, clock difference = 621 usec

v3, mode 5, strat 3, clock difference = 1205 usec

v3, mode 5, strat 3, clock difference = 1664 usec

v3, mode 5, strat 3, clock difference = 2291 usec

v3, mode 5, strat 3, clock difference = 2942 usec

v3, mode 5, strat 3, clock difference = 3558 usec
```

To run our program we first terminated the normal NTP server running on this host, so when our program starts the time is very close to the server's time. We see this host is losing about 600 microseconds every 64 seconds, or about 810 ms in 24 hours.

19.11 SNTP (Continued)

We now expand our SNTP example with additional features. First, when the program starts it creates one socket per unicast address, one socket per broadcast address, and one socket per interface on which the multicast group is joined, instead of the single socket used in Section 19.10. The purpose of this is to determine the destination address of the packets that we receive. Second, when the program starts it broadcasts and multicasts an SNTP client request out from all attached interfaces, to get an initial estimate of the difference.

Warning: This code should not be taken as the recommended way to code an SNTP client. Our goal is to understand more about broadcasting and multicasting, especially on a multihomed host or router, along with the loopback property of broadcasts and multicasts. We pick SNTP to show these properties, because it is a useful, real-world application. Our client sends an SNTP client request out from all broadcast-capable and multicast-capable interfaces when it starts to show how some programs perform resource discovery when they start. This is not recommended for an SNTP client. A better technique is just to listen for server broadcasts or server multicasts as in Section 19.10.

Figure 19.20 is an overview of the functions that comprise our program. We first call our `get_ifi_info` program from Section 16.6 to obtain the interface list. For each interface we create a UDP socket and `bind` the unicast address, create another UDP socket and `bind` the broadcast address, and create another UDP socket and `bind` the NTP multicast group (224.0.1.1) and join the multicast group on that interface. Our `sntp_send` function sends a broadcast request out from each broadcast-capable socket to any NTP server on the attached subnet for the current time, and sends a multicast

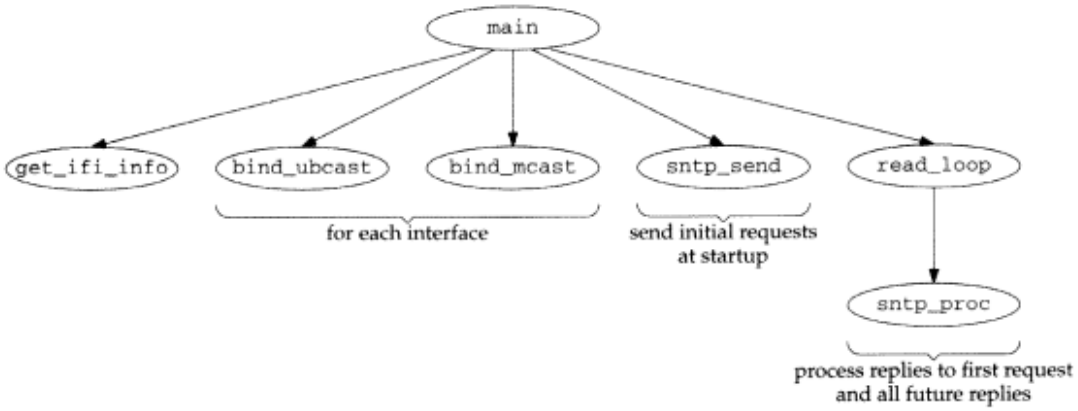


Figure 19.20 Overview of functions in our SNTP client.

request out from all multicast-capable sockets. The purpose of this first batch of sends is to find all the NTP servers on the attached subnets and to get an initial estimate of the current time.

The program then enters an infinite loop, `read_loop`, reading any replies that arrive. We first expect replies from some of the datagrams sent by `sntp_send`, and after that we should just receive the periodic transmissions from all NTP servers on any attached subnet. Most NTP servers that are broadcasting or multicasting send one datagram every 64 seconds. `sntp_proc` is the same function shown in Figure 19.19 that processes a received NTP packet.

We start our code presentation with Figure 19.21, our own `sntp.h` header that is included by all our programs. The `ntp.h` header that it includes is the same one shown in Figure 19.17.

Define `Addrs` structure

3-12 We define a structure of type `Addrs` that contains the information that we need for each address returned for a given interface. Since each `ifi_info` structure returned by our `get_ifi_info` function can have two addresses (a unicast address and a broadcast address, for example), we need to maintain two of our own `Addrs` structures, one for each address. The `addr_sa` member points to the socket address structure that is returned by `get_ifi_info` and `addr_salen` is its length. We save a pointer to the interface name in `addr_ifname`, and this will be used for multicasting. We will create one socket for each address, and the descriptor is saved in `addr_fd`. We also need to know if this socket has been bound to a broadcast address or a multicast address, and this is saved in the `addr_flags` member.

sntp/sntp.h

```
 1 #include    "unpifi.h"
 2 #include    "ntp.h"

 3 #define MAXNADDRS   128         /* max # of addresses to bind() */
 4 typedef struct {
 5     struct sockaddr *addr_sa;   /* ptr to bound address */
 6     socklen_t addr_salen;       /* socket address length */
 7     const char *addr_ifname;    /* interface name, for multicasting */
 8     int     addr_fd;            /* socket descriptor */
 9     int     addr_flags;         /* ADDR_xxx flags (see below) */
10 } Addrs;

11 Addrs   addrs[MAXNADDRS];      /* the actual array of structs */
12 int     naddrs;                /* index into the array */

13 #define ADDR_BCAST  1
14 #define ADDR_MCAST  2

15 const int on;                  /* for setsockopt() */

16                 /* function prototypes */
17 void    bind_mcast(const char *, SA *, socklen_t, int);
18 void    bind_ubcast(SA *, socklen_t, int, int, int);
19 void    read_loop(void);
20 void    sntp_proc(char *, ssize_t nread, struct timeval *);
21 void    sntp_send(void);
```

sntp/sntp.h

Figure 19.21 `sntp.h` header.

Figure 19.22 shows the `main` function.

Get well-known multicast address and well-known port

10–14 The command-line argument is normally the name `ntp.mcast.net`, which maps into the multicast address 224.0.1.1. The service name is `ntp`. Our `udp_client` function allocates space for a socket address structure of the correct type (either IPv4 or IPv6) and stores the multicast address and port in that structure. If this program is run on a host that does not support multicasting, any IP address can be specified, as only the port will be used from this structure. We then close this socket, as the purpose of calling `udp_client` was just to fill in the socket address structure.

Get interface list

15–17 Our `get_ifi_info` function returns information on all the interfaces and addresses. The address family that we ask for is taken from the socket address structure that was filled in by `udp_client` based on the command-line argument. A value of 1 as the second argument to `get_ifi_info` tells it to return information about alias addresses.

Process each unicast and broadcast address

18–22 We call our `bind_ubcast` function one or two times for each address. The first time the final argument is 0, meaning the first argument points to the unicast address,

sntp/main.c

```
 1 #include    "sntp.h"
 2 const int on = 1;                /* for setsockopt() flags */
 3 int
 4 main(int argc, char **argv)
 5 {
 6     int     sockfd, port;
 7     socklen_t salen;
 8     struct ifi_info *ifi;
 9     struct sockaddr *mcastsa, *wild;
10     if (argc != 2)
11         err_quit("usage: sntp <IPaddress>");
12     sockfd = Udp_client(argv[1], "ntp", (void **) &mcastsa, &salen);
13     port = sock_get_port(mcastsa, salen);
14     Close(sockfd);
15         /* obtain interface list and process each one */
16     for (ifi = Get_ifi_info(mcastsa->sa_family, 1); ifi != NULL;
17          ifi = ifi->ifi_next) {
18         bind_ubcast(ifi->ifi_addr, salen, port,
19                     ifi->ifi_myflags & IFI_ALIAS, 0);    /* unicast */
20         if (ifi->ifi_flags & IFF_BROADCAST)
21             bind_ubcast(ifi->ifi_brdaddr, salen, port,
22                         ifi->ifi_myflags & IFI_ALIAS, 1);    /* bcast */
23 #ifdef  MCAST
24         if (ifi->ifi_flags & IFF_MULTICAST)
25             bind_mcast(ifi->ifi_name, mcastsa, salen,
26                        ifi->ifi_myflags & IFI_ALIAS);    /* mcast */
27 #endif
28     }
29     wild = Malloc(salen);        /* socket address struct for wildcard */
30     memcpy(wild, mcastsa, salen);
31     sock_set_wild(wild, salen);
32     bind_ubcast(wild, salen, port, 0, 0);
33     sntp_send();                 /* send first queries */
34     read_loop();                 /* never returns */
35 }
```

sntp/main.c

Figure 19.22 `main` function.

and the next time it is 1, meaning the first argument points to the broadcast address. We then pass the `IFI_ALIAS` flag to our `bind_XXX` functions, letting each function decide how to handle alias addresses, as we will see shortly.

Join multicast group

23–27 If the interface is multicast capable, we call our `bind_mcast` function for each unicast address also, as we want to join the multicast group on each interface.

Bind wildcard address

29–32 After processing all the interface information, we allocate another socket address structure and copy its contents from the one filled in by `udp_client`. We then call our `sock_set_wild` function to store the appropriate wildcard address in the structure. `bind_ubcast` creates a socket and binds the wildcard address to it. This handles datagrams that arrive destined for some other address, such as 255.255.255.255, which we are unable to bind.

Send initial requests and read all subsequent replies

33–34 `sntp_send` broadcasts an SNTP request out from all broadcast-capable interfaces and multicasts an SNTP request out from all multicast-capable interfaces. `read_loop` then reads any replies to these requests, along with any future NTP broadcast or multicast packets that are received.

Figure 19.23 shows the sockets that will be created for our host `bsdi`. Recall from Figure 1.16 that this host is really a router with two Ethernet interfaces. We showed the interfaces and their unicast and broadcast IP addresses in the examples following Figure 16.6, although now we assume that there are no alias addresses. We will create nine sockets and `bind` the same port nine times.

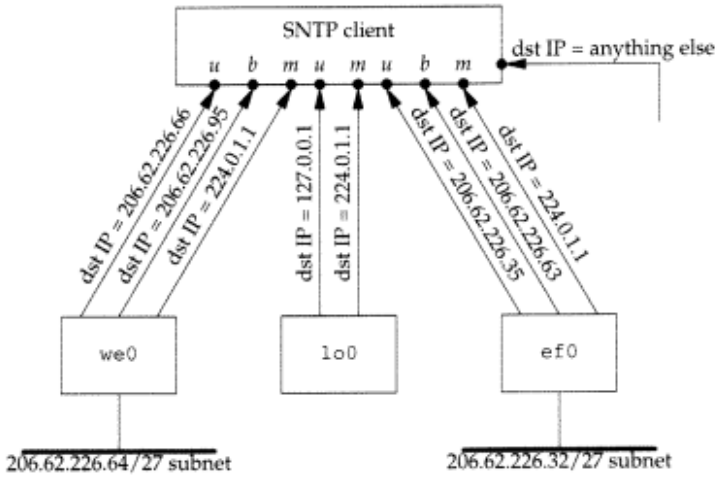


Figure 19.23 Nine sockets created by our SNTP client on host `bsdi`.

We have labeled the bottom eight sockets with *u*, *b*, or *m*, if the socket has bound a unicast address, broadcast address, or multicast address. We show the destination IP address of the packets that will be received on each socket and note that the socket bound to 0.0.0.0 on the right can receive packets destined to any other IP address (often 255.255.255.255) arriving on any interface.

Our `bind_ubcast` function is shown in Figure 19.24. This function is called for all unicast addresses, all broadcast addresses, and the wildcard address. The final argument in the three calls from Figure 19.22 is 1 only for a broadcast address.

sntp/bind_ubcast.c

```
 1 #include    "sntp.h"

 2 void
 3 bind_ubcast(struct sockaddr *sabind, socklen_t salen, int port,
 4             int alias, int bcast)
 5 {
 6     int     i, fd;

 7         /* first see if we've already bound this address */
 8     for (i = 0; i < naddrs; i++) {
 9         if (sock_cmp_addr(addrs[i].addr_sa, sabind, salen) == 0)
10             return;
11     }

12     fd = Socket(sabind->sa_family, SOCK_DGRAM, 0);

13     sock_set_port(sabind, salen, port);

14     Setsockopt(fd, SOL_SOCKET, SO_REUSEADDR, &on, sizeof(on));

15     printf("binding %s\n", Sock_ntop(sabind, salen));
16     if (bind(fd, sabind, salen) < 0) {
17         if (errno == EADDRINUSE) {
18             printf("  (address already in use)\n");
19             close(fd);
20             return;
21         } else
22             err_sys("bind error");
23     }
24     addrs[naddrs].addr_sa = sabind;      /* save ptr to sockaddr{} */
25     addrs[naddrs].addr_salen = salen;
26     addrs[naddrs].addr_fd = fd;
27     if (bcast)
28         addrs[naddrs].addr_flags = ADDR_BCAST;
29     naddrs++;
30 }
```

sntp/bind_ubcast.c

Figure 19.24 `bind_ubcast` function: create socket for unicast or broadcast address.

See if address has already been bound

7-11 We first check if this address has already been bound. While this will not happen for a unicast address, in the case of aliases that share the same broadcast address, we want to bind the broadcast address only one time.

Create socket and bind address

12-23 We create a UDP socket, set the port, and then set the `SO_REUSEADDR` socket option (since we are binding the same port for each address). We `bind` the address to the socket. The socket address structure in which we store the port for the `bind` is the one filled in and returned by `get_ifi_info`. We allow the `bind` to fail, which should happen only if the NTP daemon is itself running on this host.

Save information about this socket

24–29 The information is saved in our `Addrs` structure, including a flag if this address is a broadcast address.

For multicast-capable interfaces we need to create a socket, bind the well-known port, and then join the multicast group on the interface. This is done by our `bind_mcast` function, shown in Figure 19.25.

sntp/bind_mcast.c

```
 1 #include    "sntp.h"

 2 void
 3 bind_mcast(const char *ifname, SA *mcastsa, socklen_t salen, int alias)
 4 {
 5 #ifdef  MCAST
 6     int     fd;
 7     struct sockaddr *msa;

 8     if (alias)
 9         return;                 /* only one mcast join per interface */

10     printf("joining %s on %s\n", Sock_ntop_host(mcastsa, salen), ifname);

11     fd = Socket(mcastsa->sa_family, SOCK_DGRAM, 0);

12     Setsockopt(fd, SOL_SOCKET, SO_REUSEADDR, &on, sizeof(on));
13     Bind(fd, mcastsa, salen);

14     Mcast_join(fd, mcastsa, salen, ifname, 0);

15     addrs[naddrs].addr_sa = mcastsa;
16     addrs[naddrs].addr_salen = salen;
17     addrs[naddrs].addr_ifname = ifname;      /* save pointer, not string copy */
18     addrs[naddrs].addr_fd = fd;
19     addrs[naddrs].addr_flags = ADDR_MCAST;
20     naddrs++;
21 #endif
22 }
```

sntp/bind_mcast.c

Figure 19.25 `bind_mcast` function: join multicast group on interface.

Create socket and bind

8–13 If this address is an alias, we return immediately, as we need only join the multicast group once per interface, regardless of how many unicast addresses are aliased to the interface. The socket is created and the multicast group and well-known port are bound to the socket.

Join multicast group and save information

14–20 The multicast group is joined on the interface. We save the information in our `Addrs` structure. The `mcastsa` pointer that is saved in the `addr_sa` member points to the one socket address structure that was allocated by the call to our `udp_client` function in the `main` function. The `addr_sa` member for each multicast-capable interface points to this same structure, but that is OK, as the structure is never modified. The

addr_ifname pointer points to the interface name string in the ifi_info structure, which is OK because our free_ifi_info function is never called to free this memory.

The next function is sntp_send, shown in Figure 19.26, which is called by main after all the interface information has been processed, and all the sockets have been created.

sntp/sntp_send.c

```
 1 #include    "sntp.h"

 2 void
 3 sntp_send(void)
 4 {
 5     int     fd;
 6     Addrs  *aptr;
 7     struct ntpdata msg;

 8         /* use the socket bound to 0.0.0.0/123 for sending */
 9     fd = addrs[naddrs - 1].addr_fd;
10     Setsockopt(fd, SOL_SOCKET, SO_BROADCAST, &on, sizeof(on));

11     bzero(&msg, sizeof(msg));
12     msg.status = (0 << 6) | (3 << 3) | MODE_CLIENT;     /* see RFC 2030 */

13     for (aptr = &addrs[0]; aptr < &addrs[naddrs]; aptr++) {
14         if (aptr->addr_flags & ADDR_BCAST) {
15             printf("sending broadcast to %s\n",
16                    Sock_ntop(aptr->addr_sa, aptr->addr_salen));
17             Sendto(fd, &msg, sizeof(msg), 0,
18                    aptr->addr_sa, aptr->addr_salen);
19         }
20 #ifdef  MCAST
21         if (aptr->addr_flags & ADDR_MCAST) {
22                 /* must first set outgoing i/f appropriately */
23             Mcast_set_if(fd, aptr->addr_ifname, 0);
24             Mcast_set_loop(fd, 0);  /* disable loopback */

25             printf("sending multicast to %s on %s\n",
26                    Sock_ntop(aptr->addr_sa, aptr->addr_salen),
27                    aptr->addr_ifname);
28             Sendto(fd, &msg, sizeof(msg), 0,
29                    aptr->addr_sa, aptr->addr_salen);
30         }
31 #endif
32     }
33 }
```

sntp/sntp_send.c

Figure 19.26 sntp_send function: broadcast and multicast SNTP requests.

Set SO_BROADCAST socket option and form request

8–12 Instead of creating a socket just for sending, we use the socket bound to the wildcard address for the calls to sendto. Since the IP address is the wildcard, the kernel uses the primary unicast address of the outgoing interface as the source IP address of the UDP datagram. We first set the SO_BROADCAST socket option. We then form the

SNTP request: we set LI (the leap indicator) to 0, the version to 3, and the mode to `MODE_CLIENT`. These are the only fields that need be set in a client request.

Send broadcast request

14–19 If the address is a broadcast address, the request is broadcast.

Send multicast request

21–30 If the address is a multicast address we first specify the outgoing interface for multicasts on the socket (our `mcast_set_if` function) and then disable multicast loopback on this socket (our `mcast_set_loop` function). If we did not disable the loopback feature, we could receive many copies of this packet on all the receiving sockets (see Exercise 19.11). The datagram is sent to the multicast address.

> This code shows a typical paradigm for multicast applications, which we can summarize as
>
> ```
> for (each interface) {
>     mcast_set_if( ... );
>     sendto( ... );
> }
> ```
>
> A datagram is sent out from each interface, so the outgoing interface must be set using the socket option before each `sendto`. This involves an additional system call for each datagram that is output. An alternative is to create one sending socket for each interface and set the outgoing interface for each socket one time after it is created. We will see in Section 20.8 that IPv6 allows the outgoing interface to be specified as ancillary data with a call to `sendmsg`, reducing the number of system calls in this scenario.
>
> There is also a class of multicast applications that does not care about the outgoing interface, sending only one datagram at a time, and letting the kernel choose the outgoing interface.

Using Figure 19.23 as our example, five datagrams would be sent by `sntp_send`: one on each broadcast and multicast socket. Nothing is sent on the unicast sockets.

The next function is `read_loop`, and we show the first half in Figure 19.27. This function is called at the end of the `main` function, and it just reads from all the sockets that were created: any broadcast, multicast, or unicast NTP packets that appear on any of the host's interfaces. We expect most NTP packets to be received as broadcasts or multicasts, but responses to the queries sent by `sntp_send` will be unicast.

Allocate multiple buffers and socket address structures

4–19 We allocate two buffers in which we receive datagrams, two socket address structures in which the corresponding source addresses are stored, and two variables that hold the sizes of the corresponding datagrams. The index corresponding to the current datagram is in `currb`, and the index corresponding to the last datagram is in `lastb`. One index will have the value 0 and the other will have the value 1. The reason we need storage for two datagrams is that we will receive multiple copies of all multicast datagrams and we want to detect the copies and just ignore them. Even on a host with only a single interface, a multicast NTP packet can be received at the socket bound to that interface *and* the socket bound to the wildcard address (see Exercise 19.10).

These multiple copies occur in this example with multicasting because we have bound the same port multiple times: once per interface and again for the wildcard.

sntp/read_loop.c

```
 1 #include    "sntp.h"
 2 static int       check_loop(struct sockaddr *, socklen_t);
 3 static int       check_dup(socklen_t);

 4 static char      buf1[MAXLINE], buf2[MAXLINE];
 5 static char     *buf[2] = { buf1, buf2 };
 6 struct sockaddr *from[2];
 7 static ssize_t   nread[2] = { -1, -1 };
 8 static int       currb = 0, lastb = 1;

 9 void
10 read_loop(void)
11 {
12     int     nsel, maxfd;
13     Addrs  *aptr;
14     fd_set  rset, allrset;
15     socklen_t len;
16     struct timeval now;

17         /* allocate two socket address structures */
18     from[0] = Malloc(addrs[0].addr_salen);
19     from[1] = Malloc(addrs[0].addr_salen);

20     maxfd = -1;
21     for (aptr = &addrs[0]; aptr < &addrs[naddrs]; aptr++) {
22         FD_SET(aptr->addr_fd, &allrset);
23         if (aptr->addr_fd > maxfd)
24             maxfd = aptr->addr_fd;
25     }
```

sntp/read_loop.c

Figure 19.27 `read_loop` function: first half.

Recall from our discussion of the `SO_REUSEADDR` socket option in Chapter 7 that one copy of a multicast or broadcast datagram is delivered to each matching socket, while a unicast datagram is delivered to just one socket. Our socket bound to the wildcard address matches every received multicast packet.

If we didn't care to know which address a given multicast packet was destined to, we could just create one socket and bind the multicast address (224.0.1.1) and the well-known port (123) to it, and always receive just a single copy. This is what we did in our example in Figure 19.11. What complicates this SNTP example is that we want to know the destination address of each datagram.

Prepare descriptor set for `select`

20–25 We prepare a descriptor set for `select` and calculate the maximum of all the descriptors that we can read from.

The second half of our `read_loop` function is shown in Figure 19.28. It calls `select` to wait for one or more of the descriptors to be readable, then reads the datagram, and processes the NTP packet.

sntp/read_loop.c

```
26      for ( ; ; ) {
27          rset = allrset;
28          nsel = Select(maxfd + 1, &rset, NULL, NULL, NULL);

29          Gettimeofday(&now, NULL);   /* get time when select returns */

30          for (aptr = &addrs[0]; aptr < &addrs[naddrs]; aptr++) {
31              if (FD_ISSET(aptr->addr_fd, &rset)) {
32                  len = aptr->addr_salen;
33                  nread[currb] = recvfrom(aptr->addr_fd,
34                                          buf[currb], MAXLINE, 0,
35                                          from[currb], &len);
36                  if (aptr->addr_flags & ADDR_MCAST) {
37                      printf("%d bytes from %s", nread[currb],
38                             Sock_ntop(from[currb], len));
39                      printf(" multicast to %s", aptr->addr_ifname);

40                  } else if (aptr->addr_flags & ADDR_BCAST) {
41                      printf("%d bytes from %s", nread[currb],
42                             Sock_ntop(from[currb], len));
43                      printf(" broadcast to %s",
44                             Sock_ntop(aptr->addr_sa, len));

45                  } else {
46                      printf("%d bytes from %s", nread[currb],
47                             Sock_ntop(from[currb], len));
48                      printf(" to %s",
49                             Sock_ntop(aptr->addr_sa, len));
50                  }

51                  if (check_loop(from[currb], len)) {
52                      printf(" (ignored)\n");
53                      continue;   /* it's one of ours, looped back */
54                  }
55                  if (check_dup(len)) {
56                      printf(" (dup)\n");
57                      continue;   /* it's a duplicate */
58                  }
59                  sntp_proc(buf[lastb], nread[lastb], &now);

60                  if (--nsel <= 0)
61                      break;      /* all done with selectable descriptors */
62              }
63          }
64      }
65  }
```

sntp/read_loop.c

Figure 19.28 `read_loop` function: second half.

`select` and then get current time-of-day

27–29 As soon as `select` returns, we call `gettimeofday` to fetch the current time, which will be used to calculate the difference from the time in the NTP packet.

Determine which descriptor is readable

30-50 We check each descriptor to see which one is readable, call `recvfrom`, print how the datagram was received (broadcast, multicast, or unicast), and on which interface it was received.

Check for loopback

51-54 Our `check_loop` function returns 1 if the packet is one of our own that was looped back. We disabled multicast loopback in Figure 19.26, but there is no way to disable the automatic loopback of broadcast packets that we send.

Check for duplicate

55-58 Since we can receive multiple copies of any received multicast packet and any received broadcast packet, our `check_dup` function checks whether the current packet is an exact duplicate of the previous packet, and if so, returns 1.

Process NTP packet

59 At this point the packet is not a loopback copy and is not a duplicate so we call our `sntp_proc` function (Figure 19.19) to process the NTP packet.

Our `check_loop` function, shown in Figure 19.29, checks whether the packet is a loopback copy of a packet that we sent.

sntp/read_loop.c

```
66 int
67 check_loop(struct sockaddr *sa, socklen_t salen)
68 {
69     Addrs  *aptr;

70     for (aptr = &addrs[0]; aptr < &addrs[naddrs]; aptr++) {
71         if (sock_cmp_addr(sa, aptr->addr_sa, salen) == 0)
72             return (1);         /* it is one of our addresses */
73     }
74     return (0);
75 }
```

sntp/read_loop.c

Figure 19.29 `check_loop` function: return 1 if datagram is one that we sent.

Check sender's address

70-74 To check for a loopback copy we go through all the addresses in our `Addrs` array and compare them against the source IP address of the received datagram.

Our `check_dup` function is shown in Figure 19.30 and it checks whether the received datagram is a complete copy of the previous datagram.

Check length, sender's address, and contents

80-88 We consider the datagram a copy if the lengths of the two datagrams are the same, if the two sender protocol addresses are the same, and if the actual contents of the two datagrams are the same. If the datagram is not a duplicate, the indexes `currb` and `lastb` are swapped. Notice that the call to `sntp_proc` at the end of Figure 19.28

sntp/read_loop.c

```
76 int
77 check_dup(socklen_t salen)
78 {
79     int     temp;

80     if (nread[currb] == nread[lastb] &&
81         memcmp(from[currb], from[lastb], salen) == 0 &&
82         memcmp(buf[currb], buf[lastb], nread[currb]) == 0) {
83         return (1);             /* it is a duplicate */
84     }
85     temp = currb;               /* swap currb and lastb */
86     currb = lastb;
87     lastb = temp;
88     return (0);
89 }
```

sntp/read_loop.c

Figure 19.30 `check_dup` function: return 1 if datagram is a duplicate.

passes the datagram indexed by `lastb` because the call to `check_dup` swaps the indexes for the next call to `recvfrom`.

Recall the nine sockets shown in Figure 19.23. Figure 19.31 shows the output when we run the program on our host `bsdi` with an NTP server running on the host `solaris`. The first nine lines show the sockets that are created, the IP addresses that are bound to the sockets, and the interfaces on which the multicast group is joined.

The next five lines show the packets that are sent by `sntp_send`: one packet to each broadcast address and one packet to each multicast address.

The next six lines show the first five datagrams that are received on the various sockets. These datagrams are all received in response to the queries sent by `sntp_send`. The first datagram is ignored because it is a loopback copy of one of the broadcasts that we sent (look at the source address). The second datagram is from the NTP server (the mode of 4 is `MODE_SERVER` from Figure 19.17) on the host `solaris` (206.62.26.33), which is an NTP version 3 server running at stratum 3. The time difference between the two hosts is about 116 ms. The next three datagrams are ignored: the first is a loopback copy of our broadcast to the other Ethernet, and the next two are loopback copies of our two broadcasts received on the wildcard socket. What has happened here with the two broadcasts that were sent is that one copy of each was looped back, and then one copy of each loopback packet was delivered to the socket that was bound to the corresponding broadcast address, and another copy of each loopback packet was delivered to the wildcard socket. Two broadcasts generated four loopback copies. With multicasting we can turn off these loopback copies, but we are unable to do this with broadcasting.

The final five lines correspond to four received datagrams and these four datagrams correspond to one received NTP packet from the host `solaris`. The first datagram is received as a multicast and is processed to yield a time difference of about 117 ms. The next three datagrams are duplicates of this multicast packet received on the two other

```
bsdi # sntp 224.0.1.1
binding 206.62.226.66.123          we0: Ethernet unicast
binding 206.62.226.95.123          we0: Ethernet broadcast
joining 224.0.1.1 on we0           we0: multicast
binding 206.62.226.35.123          ef0: Ethernet unicast
binding 206.62.226.63.123          ef0: Ethernet broadcast
joining 224.0.1.1 on ef0           ef0: multicast
binding 127.0.0.1.123              lo0: loopback
joining 224.0.1.1 on lo0           lo0: multicast
binding 0.0.0.0.123                wildcard

sending broadcast to 206.62.226.95.123
sending multicast to 224.0.1.1.123 on we0
sending broadcast to 206.62.226.63.123
sending multicast to 224.0.1.1.123 on ef0
sending multicast to 224.0.1.1.123 on lo0

48 bytes from 206.62.226.66.123 broadcast to 206.62.226.95.123 (ignored)
48 bytes from 206.62.226.33.123 to 206.62.226.35.123
v3, mode 4, strat 3, clock difference = -116013 usec
48 bytes from 206.62.226.35.123 broadcast to 206.62.226.63.123 (ignored)
48 bytes from 206.62.226.66.123 to 0.0.0.0.123 (ignored)
48 bytes from 206.62.226.35.123 to 0.0.0.0.123 (ignored)

48 bytes from 206.62.226.33.123 multicast to we0
v3, mode 5, strat 3, clock difference = -117043 usec
48 bytes from 206.62.226.33.123 multicast to ef0 (dup)
48 bytes from 206.62.226.33.123 multicast to lo0 (dup)
48 bytes from 206.62.226.33.123 to 0.0.0.0.123 (dup)
```

Figure 19.31 Output from our `sntp` program.

sockets that are bound to the same IP address and port (224.0.1.1.123), and on the socket bound to just the same port (0.0.0.0.123).

If we continue running the program in this environment, we see that every 64 seconds the NTP server on `solaris` multicasts an NTP packet, and our program receives four copies. The first of the four copies is processed, and the remaining three are duplicates that are ignored.

19.12 Summary

A multicast application starts by joining the multicast group assigned to the application. This tells the IP layer to join the group, which in turns tells the datalink layer to receive multicast frames that are sent to the corresponding hardware layer multicast address. Multicasting takes advantage of the hardware filtering present on most interface cards, and the better the filtering, the fewer the number of undesired packets that are received. Using this hardware filtering reduces the load on all the other hosts that are not participating in the application.

Multicasting on a WAN requires multicast-capable routers and a multicast routing protocol. Until all the routers on the Internet are multicast capable, a virtual network, the MBone (Section B.2) is being used.

Five socket options provide the API for multicasting:

- join a multicast group on an interface,
- leave a multicast group,
- set the default interface for outgoing multicasts,
- set the TTL or hop limit for outgoing multicasts, and
- enable or disable loopback of multicasts.

The first two are for receiving, and the last three are for sending. There is enough difference between the five IPv4 socket options and the five IPv6 socket options that protocol-independent multicasting code becomes littered with `#ifdef`s very quickly. We developed eight functions of our own, all beginning with `mcast_`, that can help in writing multicast applications that work with either IPv4 or IPv6.

Exercises

19.1 Build the program shown in Figure 18.9 and run it specifying an IP address on the command line of 224.0.0.1. What happens?

19.2 Modify the program in the previous example to `bind` 224.0.0.1 and port 0 to its socket. Run it. Are you allowed to `bind` a multicast address to the socket? If you have a tool such as `tcpdump`, watch the packets on the network. What is the source IP address of the datagram you send?

19.3 One way to tell which hosts on your subnet are multicast capable is to `ping` the all-hosts group: 224.0.0.1. Try this.

19.4 If we type `ping 224.0.0.1` on our host `unixware`, which is not multicast capable, we get the following output:

```
unixware % ping 224.0.0.1
PING 224.0.0.1: 56 data bytes
64 bytes from gw.kohala.com (206.62.226.62): icmp_seq=0. time=0. ms
64 bytes from gw.kohala.com (206.62.226.62): icmp_seq=1. time=0. ms
64 bytes from gw.kohala.com (206.62.226.62): icmp_seq=2. time=0. ms
```

What is happening?

19.5 One way to locate any multicast routers on your subnet is to `ping` the all routers group: 224.0.0.2. Try this.

19.6 One way to tell if your host is connected to the MBone is to run our program from Section 19.8, wait a few minutes, and see if any session announcements appear. Try this and see if you receive any announcements.

19.7 The session ID and version in the `o=` line in Figure 19.13 are often NTP timestamps. Do the values shown make sense?

19.8 Go through the calculations in Figure 19.19 when the fractional part of the NTP timestamp is 1,073,741,824 (one-quarter of 2^{32}).

Redo these calculations for the largest possible integer fraction ($2^{32} - 1$).

19.9 In Figures 19.24 and 19.25 we set the `SO_REUSEADDR` socket option to allow the same port to be bound multiple times. But in Figure 19.25 we are performing completely duplicate bindings for each multicast address (224.0.1.1) and port 123, and we see three of these in Figure 19.31. How can we do this on a Berkeley-derived kernel without setting the `SO_REUSEPORT` socket option instead of `SO_REUSEADDR`?

19.10 The final line in Figure 19.31 shows a multicast datagram being received by the socket that was bound to the wildcard address. But if we run our SNTP client under Solaris 2.5, this socket does not receive the multicast datagrams. Why?

19.11 In the example shown in Figure 19.31, if we had not disabled multicast loopback in Figure 19.26, how many additional copies would be received?

19.12 Modify the implementation of `mcast_set_if` for IPv4 to remember each interface name for which it obtains the IP address to prevent calling `ioctl` again for that interface.

20

Advanced UDP Sockets

20.1 Introduction

This chapter is a collection of various topics that affect applications using UDP sockets. First is determining the destination address of a UDP datagram, and the interface on which the datagram was received, because a socket bound to a UDP port and the wildcard address can receive unicast, broadcast, and multicast datagrams, on any interface.

TCP is a byte-stream protocol and it uses a sliding window, so there is no such thing as a record boundary or allowing the sender to overrun the receiver with data. With UDP, however, each input operation corresponds to a UDP datagram (a record) so the problem arises of what happens when the received datagram is larger than the application's input buffer.

UDP is unreliable but there are applications where it makes sense to use UDP instead of TCP. We discuss the factors affecting when UDP can be used instead of TCP. In these UDP applications we must include some features to make up for UDP's unreliability: a timeout and retransmission, to handle lost datagrams, and sequence numbers to match the replies to the requests. We develop a set of functions that we can call from our UDP applications to handle these details.

If the implementation does not support the `IP_RECVDSTADDR` socket option, then one way to determine the destination IP address of a UDP datagram is to bind all the interface addresses and use `select`. We showed one example of this in Section 19.11 and work with this technique some more in this chapter.

Most UDP servers are iterative but there are applications that exchange multiple UDP datagrams between the client and server, requiring some form of concurrency. TFTP is the common example and we discuss how this is done, both with and without `inetd`.

The final topic is the per-packet information that can be specified as ancillary data for an IPv6 datagram: specifying the source IP address, the sending interface, the outgoing hop limit, and the next-hop address. Similar information—the destination IP address, received interface, and received hop limit—can be returned with an IPv6 datagram.

20.2 Receiving Flags, Destination IP Address, and Interface Index

Historically `sendmsg` and `recvmsg` have been used only to pass descriptors across Unix domain sockets (Section 14.7), and even this was rare. But the use of these two functions is increasing for two reasons.

1. The `msg_flags` member, which was added to the `msghdr` structure with 4.3BSD Reno, returns flags to the application. We summarized these flags in Figure 13.7.
2. Ancillary data is being used to pass more and more information between the application and the kernel. We will see in Chapter 24 that IPv6 continues this trend.

As an example of `recvmsg` we will write a function named `recvfrom_flags` that is similar to `recvfrom` but also returns

1. the returned `msg_flags` value,
2. the destination address of the received datagram (from the `IP_RECVDSTADDR` socket option), and
3. the index of the interface on which the datagram was received (the `IP_RECVIF` socket option).

To return the last two items we define the following structure in our `unp.h` header:

```
struct in_pktinfo {
  struct in_addr  ipi_addr;    /* destination IPv4 address */
  int             ipi_ifindex; /* received interface index */
};
```

We have purposely chosen the structure name and member names to be similar to the IPv6 `in6_pktinfo` structure that returns the same two items for an IPv6 socket (Section 20.8). Our `recvfrom_flags` function will take a pointer to an `in_pktinfo` structure as an argument, and if this pointer is nonnull, return the structure through the pointer.

A design problem with this structure is what to return if the `IP_RECVDSTADDR` information is not available (i.e., the implementation does not support the socket option). The interface index is easy to handle because a value of 0 can indicate that the index is not known. But all 32-bit values for an IP address are valid. What we have chosen is to return a value of all zeros (0.0.0.0) as the destination address when the

actual value is not available. While this is a valid IP address, it is never allowed as the destination IP address (RFC 1122 [Braden 1989]); it is valid only as the source IP address when a host is bootstrapping and does not yet know its IP address.

> Unfortunately Berkeley-derived kernels do accept IP datagrams destined to 0.0.0.0 (pp. 218–219 of TCPv2). These are obsolete broadcasts generated by 4.2BSD-derived kernels.

We now show the first half of our `recvfrom_flags` function in Figure 20.1. This function is intended to be used with a UDP socket.

advio/recvfromflags.c

```
 1 #include    "unp.h"
 2 #include    <sys/param.h>       /* ALIGN macro for CMSG_NXTHDR() macro */

 3 #ifdef  HAVE_SOCKADDR_DL_STRUCT
 4 #include    <net/if_dl.h>
 5 #endif

 6 ssize_t
 7 recvfrom_flags(int fd, void *ptr, size_t nbytes, int *flagsp,
 8                SA *sa, socklen_t *salenptr, struct in_pktinfo *pktp)
 9 {
10     struct msghdr msg;
11     struct iovec iov[1];
12     ssize_t n;

13 #ifdef  HAVE_MSGHDR_MSG_CONTROL
14     struct cmsghdr *cmptr;
15     union {
16         struct cmsghdr cm;
17         char    control[CMSG_SPACE(sizeof(struct in_addr)) +
18                         CMSG_SPACE(sizeof(struct in_pktinfo))];
19     } control_un;

20     msg.msg_control = control_un.control;
21     msg.msg_controllen = sizeof(control_un.control);
22     msg.msg_flags = 0;
23 #else
24     bzero(&msg, sizeof(msg));   /* make certain msg_accrightslen = 0 */
25 #endif

26     msg.msg_name = sa;
27     msg.msg_namelen = *salenptr;
28     iov[0].iov_base = ptr;
29     iov[0].iov_len = nbytes;
30     msg.msg_iov = iov;
31     msg.msg_iovlen = 1;

32     if ( (n = recvmsg(fd, &msg, *flagsp)) < 0)
33         return (n);

34     *salenptr = msg.msg_namelen;    /* pass back results */
35     if (pktp)
36         bzero(pktp, sizeof(struct in_pktinfo));     /* 0.0.0.0, i/f = 0 */
```

advio/recvfromflags.c

Figure 20.1 `recvfrom_flags` function: call `recvmsg`.

Include files

2-5 To use the CMSG_NXTHDR macro requires including the <sys/param.h> header. We also need to include the <net/if_dl.h> header, which defines the sockaddr_dl structure, in which the received interface index is returned.

Function arguments

6-8 The function arguments are similar to recvfrom, except the fourth argument is now a pointer to an integer flag (so that we can return the flags returned by recvmsg) and the seventh argument is new: it is a pointer to an in_pktinfo structure that will contain the destination IPv4 address of the received datagram and the interface index on which the datagram was received.

Implementation differences

13-25 When dealing with the msghdr structure, and the various MSG_*xxx* constants, we encounter lots of differences between various implementations. Our way of handling these differences is to use C's conditional inclusion feature (#ifdef). If the implementation supports the msg_control member, space is allocated to hold the values returned by both the IP_RECVDSTADDR and IP_RECVIF socket options, and the appropriate members are initialized.

Fill in msghdr structure and call recvmsg

26-36 A msghdr structure is filled in and recvmsg is called. The values of the msg_namelen and msg_flags members must be passed back to the caller; they are value–result arguments. We also initialize the caller's in_pktinfo structure, setting the IP address to 0.0.0.0 and the interface index to 0.

Figure 20.2 shows the second half of our function.

37-40 If the implementation does not support the msg_control member, we just set the returned flags to 0 and return. The remainder of the function handles the msg_control information.

Return if no control information

41-44 We return the msg_flags value and then return to the caller if (a) there is no control information, (b) the control information was truncated, or (c) the caller does not want an in_pktinfo structure returned.

Process ancillary data

45-46 We process any number of ancillary data objects using the CMSG_FIRSTHDR and CMSG_NXTHDR macros.

Process IP_RECVDSTADDR

47-54 If the destination IP address was returned as control information (Figure 13.9), it is returned to the caller.

Process IP_RECVIF

55-63 If the index of the received interface was returned as control information, it is returned to the caller. Figure 20.3 shows the contents of the ancillary data object that is returned.

advio/recvfromflags.c

```
37 #ifndef HAVE_MSGHDR_MSG_CONTROL
38     *flagsp = 0;                    /* pass back results */
39     return (n);
40 #else

41     *flagsp = msg.msg_flags;        /* pass back results */
42     if (msg.msg_controllen < sizeof(struct cmsghdr) ||
43                 (msg.msg_flags & MSG_CTRUNC) || pktp == NULL)
44                 return (n);

45     for (cmptr = CMSG_FIRSTHDR(&msg); cmptr != NULL;
46          cmptr = CMSG_NXTHDR(&msg, cmptr)) {

47 #ifdef  IP_RECVDSTADDR
48         if (cmptr->cmsg_level == IPPROTO_IP &&
49             cmptr->cmsg_type == IP_RECVDSTADDR) {

50             memcpy(&pktp->ipi_addr, CMSG_DATA(cmptr),
51                    sizeof(struct in_addr));
52             continue;
53         }
54 #endif

55 #ifdef  IP_RECVIF
56         if (cmptr->cmsg_level == IPPROTO_IP &&
57             cmptr->cmsg_type == IP_RECVIF) {
58             struct sockaddr_dl *sdl;

59             sdl = (struct sockaddr_dl *) CMSG_DATA(cmptr);
60             pktp->ipi_ifindex = sdl->sdl_index;
61             continue;
62         }
63 #endif
64         err_quit("unknown ancillary data, len = %d, level = %d, type = %d",
65                  cmptr->cmsg_len, cmptr->cmsg_level, cmptr->cmsg_type);
66     }
67     return (n);
68 #endif  /* HAVE_MSGHDR_MSG_CONTROL */
69 }
```

advio/recvfromflags.c

Figure 20.2 `recvfrom_flags` function: return flags and destination address.

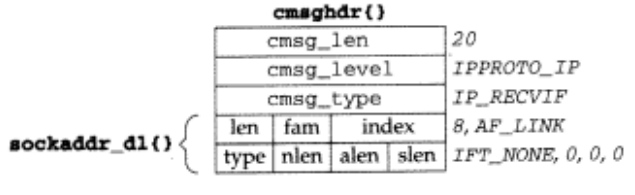


Figure 20.3 Ancillary data object returned for `IP_RECVIF`.

Recall the datalink socket address structure in Figure 17.1. The data returned in the ancillary data object is one of these structures, but the three lengths are 0 (name length, address length, and selector length). Therefore there is no need for any of the data that follows these lengths, so the size of the structure should be 8 bytes, and not the 20 that we show in Figure 17.1. The information that we return is the interface index.

Example: Print Destination IP Address and Datagram-Truncated Flag

To test our function we modify our `dg_echo` function (Figure 8.4) to call `recvfrom_flags` instead of `recvfrom`. We show this new version of `dg_echo` in Figure 20.4.

advio/dgechoaddr.c

```
 1 #include    "unpifi.h"

 2 #undef  MAXLINE
 3 #define MAXLINE 20              /* to see datagram truncation */

 4 void
 5 dg_echo(int sockfd, SA *pcliaddr, socklen_t clilen)
 6 {
 7     int     flags;
 8     const int on = 1;
 9     socklen_t len;
10     ssize_t n;
11     char    mesg[MAXLINE], str[INET6_ADDRSTRLEN], ifname[IFNAMSIZ];
12     struct in_addr in_zero;
13     struct in_pktinfo pktinfo;

14 #ifdef  IP_RECVDSTADDR
15     if (setsockopt(sockfd, IPPROTO_IP, IP_RECVDSTADDR, &on, sizeof(on)) < 0)
16         err_ret("setsockopt of IP_RECVDSTADDR");
17 #endif
18 #ifdef  IP_RECVIF
19     if (setsockopt(sockfd, IPPROTO_IP, IP_RECVIF, &on, sizeof(on)) < 0)
20         err_ret("setsockopt of IP_RECVIF");
21 #endif
22     bzero(&in_zero, sizeof(struct in_addr));    /* all 0 IPv4 address */

23     for ( ; ; ) {
24         len = clilen;
25         flags = 0;
26         n = Recvfrom_flags(sockfd, mesg, MAXLINE, &flags,
27                            pcliaddr, &len, &pktinfo);
28         printf("%d-byte datagram from %s", n, Sock_ntop(pcliaddr, len));
29         if (memcmp(&pktinfo.ipi_addr, &in_zero, sizeof(in_zero)) != 0)
30             printf(", to %s", Inet_ntop(AF_INET, &pktinfo.ipi_addr,
31                                         str, sizeof(str)));
32         if (pktinfo.ipi_ifindex > 0)
33             printf(", recv i/f = %s",
34                    If_indextoname(pktinfo.ipi_ifindex, ifname));
35 #ifdef  MSG_TRUNC
36         if (flags & MSG_TRUNC)
37             printf(" (datagram truncated)");
38 #endif
```

```
39 #ifdef  MSG_CTRUNC
40         if (flags & MSG_CTRUNC)
41             printf(" (control info truncated)");
42 #endif
43 #ifdef  MSG_BCAST
44         if (flags & MSG_BCAST)
45             printf(" (broadcast)");
46 #endif
47 #ifdef  MSG_MCAST
48         if (flags & MSG_MCAST)
49             printf(" (multicast)");
50 #endif
51         printf("\n");

52         Sendto(sockfd, mesg, n, 0, pcliaddr, len);
53     }
54 }
```

advio/dgechoaddr.c

Figure 20.4 `dg_echo` function that calls our `recvfrom_flags` function.

Change `MAXLINE`

2-3 We remove the existing definition of `MAXLINE` that occurs in our `unp.h` header and redefine it to be 20. We do this to see what happens when we receive a UDP datagram that is larger than the buffer that we pass to the input function (`recvmsg` in this case).

Set `IP_RECVDSTADDR` and `IP_RECVIF` socket options

14-21 If the `IP_RECVDSTADDR` socket option is defined, it is turned on. Similarly the `IP_RECVIF` socket option is turned on.

> Unfortunately tests of this form are not adequate because some systems (e.g., Solaris 2.5) define `IP_RECVDSTADDR` even though it is not supported. Therefore we let the call to `setsockopt` fail, and if this happens we just print a message and continue. We cannot even check for a specific error, as different implementations return different errors when a socket option is not implemented. For example, in 4.4BSD `getsockopt` returns `ENOPROTOOPT` for an unknown option, but `setsockopt` returns `EINVAL`. But the multicast socket options in 4.4BSD return `EOPNOTSUPP` for an unknown option.

Read datagram, print source IP address and port

24-28 The datagram is read by calling `recvfrom_flags`. The source IP address and port of the server's reply are converted to presentation format by `sock_ntop`.

Print destination IP address

29-31 If the returned IP address is not 0, it is converted to presentation format by `inet_ntop` and printed.

Print name of received interface

32-34 If the returned interface index is not 0, its name is obtained by calling `if_indextoname` and printed.

Test various flags

35-51 We then test four additional flags and print a message if any are on.

If we run our server under BSD/OS 3.0 on the host `bsdi` (which is multihomed), we can see the various destination IP addresses and the various flags:

```
bsdi % udpserv01
9-byte datagram from 206.62.226.33.41164, to 206.62.226.35, recv i/f = ef0
13-byte datagram from 206.62.226.65.1057, to 206.62.226.95, recv i/f = we0
   (broadcast)
4-byte datagram from 206.62.226.33.41176, to 224.0.0.1, recv i/f = ef0
   (multicast)
20-byte datagram from 127.0.0.1.4632, to 127.0.0.1, recv i/f = lo0
   (datagram truncated)
9-byte datagram from 206.62.226.33.41178, to 206.62.226.66, recv i/f = ef0
```

We have wrapped the lines that print a flag in parentheses, for readability. To generate these five lines of output we ran our `sock` program (Section C.3) on various hosts.

1. The first line is from the host `solaris`, destined to 206.62.226.35, one of the unicast addresses of the server host.
2. The second line is from the host `laptop`, destined to 206.62.226.95, the broadcast address of the Ethernet shared by the client and server (Figure 1.16). The interface changes from the first line, as we expect. Our server receives the broadcast and the `MSG_BCAST` flag is set, indicating that the datagram was received as a link-layer broadcast.
3. The third line is from the host `solaris`, destined to 224.0.0.1, the all-hosts multicast group address. All multicast-capable hosts on the subnet must belong to this group. Our server receives the multicast because the BSD/OS operating system is multicast capable and the destination port matches our server's port. The `MSG_MCAST` flag is set, indicating that the datagram was received as a link-layer multicast.
4. The fourth line is from the host itself, destined to 127.0.0.1, the loopback address. The interface is `lo0`, as we expect. Also, this time we typed in a 41-byte line at the client (which we do not show). The server only receives the first 20 bytes of the datagram and the `MSG_TRUNC` flag is set, indicating that the datagram was truncated.
5. The final line is from the host `solaris` but destined to 206.62.226.66, the unicast address of this server host on its other Ethernet (Figure 1.16). The server still receives the datagram (because the host implements the weak end system model, as described in Section 8.8) but the destination IP address (206.62.226.66) does not match the address of the interface on which the datagram was received.

Earlier Berkeley-derived implementations ignored the `IP_RECVDSTADDR` socket option when the received datagram was a broadcast or a multicast (p. 776 of TCPv2). Newer releases fix this bug.

If the destination address is 255.255.255.255, BSD/OS converts this to the broadcast address of the received interface.

20.3 Datagram Truncation

The example in the previous section shows that under BSD/OS when a UDP datagram arrives that is larger than the application's buffer, `recvmsg` sets the `MSG_TRUNC` flag in the `msg_flags` member of the `msghdr` structure (Figure 13.7). All Berkeley-derived implementations that support the `msghdr` structure with the `msg_flags` member provide this notification.

> This is an example of a flag that must be returned from the kernel to the process. We mentioned in Section 13.3 that one design problem with the `recv` and `recvfrom` functions is that their *flags* argument is an integer, which allows flags to be passed from the process to the kernel but not vice versa.

Unfortunately not all implementations handle a larger-than-expected UDP datagram in this fashion. There are three possible scenarios.

1. Discard the excess bytes and return the `MSG_TRUNC` flag to the application. This requires that the application call `recvmsg` to receive the flag.
2. Discard the excess bytes but do not tell the application.
3. Keep the excess bytes and return them in subsequent read operations on the socket.

We have already seen the first type of behavior under BSD/OS. The second type of behavior is seen under Solaris 2.5: the excess bytes are discarded but since its `msghdr` structure does not support the `msg_flags` member, there is no way to return the error to the application.

> Posix.1g specifies the first type of behavior: discarding the excess bytes and setting the `MSG_TRUNC` flag. Early releases of SVR4 exhibited the third type of behavior.

Since there are such variations in how implementations handle datagrams that are larger than the application's receive buffer, one way to detect the problem is to always allocate an application buffer 1 byte greater than the largest datagram the application should ever receive. If a datagram is ever received whose length equals this buffer, consider it an error.

20.4 When to Use UDP Instead Of TCP

In Sections 2.3 and 2.4 we described the major differences between UDP and TCP. Given that TCP is reliable while UDP is not, the question arises: when should we use UDP instead of TCP, and why? We first list the advantages of UDP.

- As we show in Figure 18.1, UDP supports broadcasting and multicasting. Indeed, UDP *must* be used if the application uses broadcasting or multicasting. We discussed these two addressing modes in Chapters 18 and 19.

- UDP has no connection setup or teardown. With regard to Figure 2.5, UDP requires only two packets to exchange a request and a reply (assuming the size of each is less than the minimum MTU between the two end systems). TCP requires about 10 packets, assuming that a new TCP connection is established for each request–reply exchange.

  Also important in this number-of-packet analysis is the number of packet round trips required to obtain the reply. This becomes important if the latency exceeds the bandwidth, as described in Appendix A of TCPv3. That text shows that the minimum *transaction time* for a UDP request–reply is RTT + SPT where *RTT* is the round-trip time between the client and server, and *SPT* is the server processing time for the request. With TCP, however, if a new TCP connection is used for the request–reply, the minimum transaction time is 2 × RTT + SPT, one RTT greater than the UDP time. TCPv3 and Section 13.9 of this text also describe a modification to TCP, called T/TCP or "TCP for Transactions" that normally obviates the need for the TCP three-way handshake, allowing T/TCP to equal UDP's RTT + SPT transaction time.

It should be obvious with regard to the second point that if a TCP connection is used for multiple request–reply exchanges, then the cost of the connection establishment and teardown is amortized across all the requests and replies, and this is normally a better design than using a new connection for each request–reply. Nevertheless, there are applications that use a new TCP connection for each request–reply (e.g., HTTP) and there are applications in which the client and server exchange one request–reply (e.g., the DNS) and then might not talk to each other for hours or days.

We now list the features of TCP that are not provided by UDP, which means that an application must provide these features itself, if they are necessary to the application. We use the qualifier "necessary" because not all the features are needed by all applications. For example, dropped segments might not need to be retransmitted for a real-time audio application, if the receiver can interpolate the missing data. Also, for simple request–reply transactions, windowed flow control might not be needed if the two ends agree ahead of time on the size of the largest request and reply.

- Positive acknowledgments, retransmission of lost packets, duplicate detection, and sequencing of packets reordered by the network. TCP acknowledges all data, allowing lost packets to be detected. The implementation of these two features requires that every TCP data segment contain a sequence number that can then be acknowledged. It also requires that TCP estimate a retransmission timeout value for the connection and that this value be updated continually as network traffic between the two end systems changes.
- Windowed flow control. A receiving TCP tells the sender how much buffer space it has allocated for receiving data, and the sender cannot exceed this. That is, the amount of unacknowledged data at the sender can never exceed the receiver's advertised window.

- Slow start and congestion avoidance. This is a form of flow control imposed by the sender to determine the current network capacity and to handle periods of congestion. All current TCPs must support these two features and we know from experience (before these algorithms were implemented in the late 1980s) that protocols that do not "back off" in the face of congestion just make the congestion worse (e.g., [Jacobson 1988]).

In summary we can state the following recommendations:

- UDP *must* be used for broadcast or multicast applications. Any form of desired error control must be added to the clients and servers, but applications often use broadcasting or multicasting when some (assumed small) amount of error is OK (such as lost packets for audio or video). Multicast applications requiring reliable delivery have been built (e.g., multicast file transfer) but we must decide whether the performance gain in using multicasting (sending one packet to N destinations, versus sending N copies of the packet across N TCP connections) outweighs the added complexity required within the application to provide reliable communications.
- UDP *can* be used for simple request–reply applications but error detection must then be built into the application. Minimally this involves acknowledgments, timeouts, and retransmission. Flow control is often not an issue for reasonably sized requests and responses. We provide an example of these features in a UDP application in Section 20.5. The factors to consider here are how often the client and server communicate (could a TCP connection be left up between the two?) and how much data is exchanged (if multiple packets are normally required, then the cost of the TCP connection establishment and teardown becomes less of a factor).
- UDP *should not* be used for bulk data transfer (e.g., file transfer). The reason is that windowed flow control, congestion avoidance, and slow start must all be built into the application, along with the features from the previous bullet, which means we are reinventing TCP within the application. We should let the vendors focus on better TCP performance and concentrate our efforts on the application itself.

There are exceptions to these rules, especially in existing applications. TFTP, for example, uses UDP for bulk data transfer. UDP was chosen for TFTP because it is simpler to implement than TCP in bootstrap code (800 lines of C code for UDP versus 4500 lines for TCP in TCPv2, for example), and because TFTP is used only to bootstrap systems on a LAN, and not for bulk data transfer across WANs. But this requires that TFTP include its own sequence number field for acknowledgments, along with a timeout and retransmission capability.

NFS is another exception to the rule: it also uses UDP for bulk data transfer (although some might claim it is really a request–reply application, albeit using large requests and replies). This is partly historical, because in the mid-1980s when it was

designed, UDP implementations were faster than TCP, and NFS was used only on LANs, where packet loss is often orders of magnitude less than on WANs. But as NFS started being used across WANs in the early 1990s, and as TCP implementations passed UDP in terms of bulk data transfer performance, NFS version 3 was designed to support TCP, and most vendors are now providing NFS over both UDP and TCP. Similar reasoning (UDP being faster than TCP in the mid-1980s along with a predominance of LANs over WANs) led the precursor of the DCE remote procedure call (RPC) package (the Apollo NCS package) to also choose UDP over TCP, although current implementations support both UDP and TCP.

We might be tempted to say that UDP usage is decreasing compared to TCP, with good TCP implementations being as fast as the network today, and with fewer application designers wanting to reinvent TCP within their UDP application. But the predicted increase in multimedia applications over the next decade will see an increase in UDP usage, since multimedia usually implies multicasting, which requires UDP.

20.5 Adding Reliability to a UDP Application

If we are going to use UDP for a request–reply application, as mentioned in the previous section, then we *must* add two features to our client:

1. timeout and retransmission to handle datagrams that are discarded, and
2. sequence numbers so the client can verify that a reply is for the appropriate request.

These two features are part of most existing UDP applications that use the simple request–reply paradigm: DNS resolvers, SNMP agents, TFTP, and RPC, for example. We are not trying to use UDP for bulk data transfer—our intent is for an application that sends a request and waits for a reply.

> By definition a datagram is unreliable; therefore we purposely do not call this a "reliable datagram service." Indeed, the term "reliable datagram" is an oxymoron. What we are showing is an application that adds reliability on top of an unreliable datagram service (UDP).

Adding sequence numbers is simple. The client prepends a sequence number to each request and the server must echo this number back to the client in the reply. This lets the client verify that a given reply is for the request that was issued.

The old-fashioned method for handling timeout and retransmission was to send a request and wait for N seconds. If no response was received, retransmit and wait another N seconds. After this has happened some number of times, give up. This is a linear retransmit timer. (Figure 6.8 of TCPv1 shows an example of a TFTP client that uses this technique. Many TFTP clients still use this method.)

The problem with this technique is that the amount of time required for a datagram to make a round trip on an internet can vary from fractions of a second on a LAN to many seconds on a WAN. Factors affecting the round-trip time (RTT) are distance, network speed, and congestion. Additionally, the RTT between a client and server can

change rapidly with time, as network conditions change. We must use a timeout and retransmission algorithm that takes into account the actual RTTs that we measure along with the changes in the RTT over time. Much work has been focused on this area, mostly relating to TCP, but the same ideas apply to any network application.

We want to calculate the retransmission timeout (*RTO*) to use for every packet that we send. To calculate this we measure the RTT: the actual round-trip time for a packet. Every time we measure an RTT we update two statistical estimators: *srtt* is the smoothed RTT estimator and *rttvar* is the smoothed mean deviation estimator. The latter is a good approximation of the standard deviation, but easier to compute since it does not involve a square root. Given these two estimators, the *RTO* to use is *srtt* plus four times *rttvar*. [Jacobson 1988] provides all the details on these calculations, which we can summarize in the following four equations:

$$delta = measuredRTT - srtt$$

$$srtt \leftarrow srtt + g \times delta$$

$$rttvar \leftarrow rttvar + h(|delta| - rttvar)$$

$$RTO = srtt + 4 \times rttvar$$

delta is the difference between the measured RTT and the current smoothed RTT estimator (*srtt*). g is the gain applied to the RTT estimator and equals $\frac{1}{8}$. h is the gain applied to the mean deviation estimator and equals $\frac{1}{4}$.

> The two gains and the multiplier 4 in the *RTO* calculation are purposely powers of 2, so they can be calculated using shift operations instead of multiplying or dividing. Indeed the TCP kernel implementation (Section 25.7 of TCPv2) is normally performed using fixed-point arithmetic for speed, but for simplicity we use floating-point calculations in our code that follows.

Another point made in [Jacobson 1988] is that when the retransmission timer expires, an *exponential backoff* must be used for the next *RTO*. For example, if our first *RTO* is 2 seconds, and the reply is not received in this time, then the next *RTO* is 4 seconds. If there is still no reply, the next *RTO* is 8 seconds, and then 16, and so on.

Jacobson's algorithms tell us how to calculate the *RTO* each time we measure an RTT and how to increase the *RTO* when we retransmit. But a problem arises when we have to retransmit a packet and then receive a reply. This is called the *retransmission ambiguity problem*. Figure 20.5 shows three possible scenarios when our retransmission timer expires.

- The request is lost, or
- the reply is lost, or
- the *RTO* is too small.

When the client receives a reply to a request that was retransmitted, it cannot tell to which request the reply corresponds. In the example on the right the reply corresponds to the original request, while in the two other examples the reply corresponds to the retransmitted request.

Karn's algorithm [Karn and Partridge 1987] handles this scenario with the following rules that apply whenever a reply is received for a request that was retransmitted.

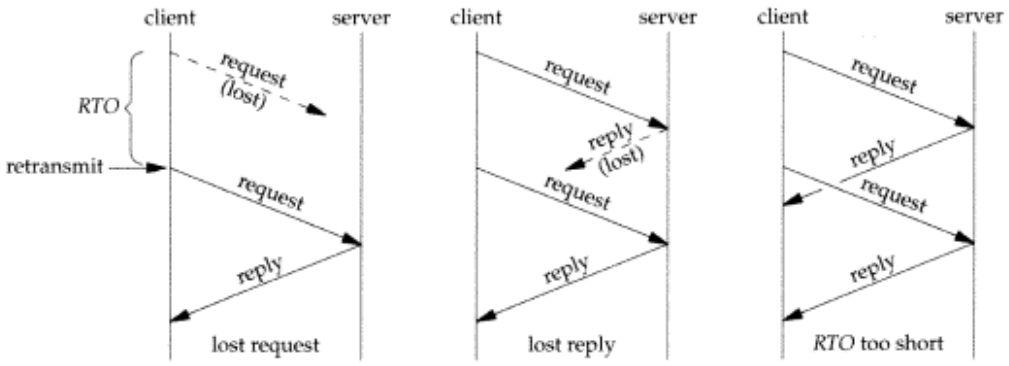


Figure 20.5 Three scenarios when retransmission timer expires.

- If an RTT was measured, do not use it to update the estimators since we do not know to which request the reply corresponds.
- Since this reply arrived before our retransmission timer expired, reuse this *RTO* for the next packet. Only when we receive a reply to a request that is not retransmitted will we update the RTT estimators and recalculate the *RTO*.

It is not hard to take Karn's algorithm into account when coding our RTT functions, but it turns out that an even better and more elegant solution exists. This solution is from the TCP extensions for "long fat pipes" (networks with either a high bandwidth, a long RTT, or both), which are documented in RFC 1323 [Jacobson, Braden, and Borman 1992]. In addition to prepending a sequence number to each request, which the server must echo back, we also prepend a *timestamp* that the server must also echo. Each time we send a request we store the current time in the timestamp. When a reply is received, we calculate the RTT of that packet as the current time minus the timestamp that was echoed by the server in its reply. Since every request carries a timestamp that is echoed by the server, we can calculate the RTT of *every* reply that we receive. There is no longer any ambiguity at all. Furthermore, since all the server does is echo the client's timestamp, the client can use any units desired for the timestamps and there is no requirement at all that the client and server have synchronized clocks.

Example

We now put all of this together in an example. We start with our UDP client `main` function from Figure 8.7 and just change the port number from `SERV_PORT` to 7 (the standard echo server, Figure 2.13).

Figure 20.6 is the `dg_cli` function. The only change from Figure 8.8 is to replace the calls to `sendto` and `recvfrom` with a call to our new function `dg_send_recv`.

Before showing our `dg_send_recv` function and our RTT functions that it calls, Figure 20.7 shows an outline of how we add reliability to a UDP client. All the functions beginning with `rtt_` are shown shortly.

rtt/dg_cli.c

```
 1 #include    "unp.h"
 2 ssize_t Dg_send_recv(int, const void *, size_t, void *, size_t,
 3                       const SA *, socklen_t);
 4 void
 5 dg_cli(FILE *fp, int sockfd, const SA *pservaddr, socklen_t servlen)
 6 {
 7     ssize_t n;
 8     char    sendline[MAXLINE], recvline[MAXLINE + 1];
 9     while (Fgets(sendline, MAXLINE, fp) != NULL) {
10         n = Dg_send_recv(sockfd, sendline, strlen(sendline),
11                          recvline, MAXLINE, pservaddr, servlen);
12         recvline[n] = 0;        /* null terminate */
13         Fputs(recvline, stdout);
14     }
15 }
```

rtt/dg_cli.c

Figure 20.6 `dg_cli` function that calls our `dg_send_recv` function.

```
static sigjmp_buf  jmpbuf;

{
    . . .
    form request

    signal(SIGALRM, sig_alrm); /* establish signal handler */
    rtt_newpack();             /* initialize rexmt counter to 0 */
sendagain:
    sendto();

    alarm(rtt_start());        /* set alarm for RTO seconds */
    if (sigsetjmp(jmpbuf, 1) != 0) {
        if (rtt_timeout())     /* double RTO, retransmitted enough? */
            give up
        goto sendagain;        /* retransmit */
    }
    do {
        recvfrom();
    } while (wrong sequence#);

    alarm(0);                  /* turn off alarm */
    rtt_stop();                /* calculate RTT and update estimators */

    process reply
    . . .
}

void
sig_alrm(int signo)
{
    siglongjmp(jmpbuf, 1);
}
```

Figure 20.7 Outline of RTT functions and when they are called.

When a reply is received but the sequence number is not the one expected, we call `recvfrom` again, but we do not retransmit the request and we do not restart the retransmission timer that is running. Notice in the rightmost example in Figure 20.5 that the final reply from the retransmitted request will be in the socket receive buffer the next time the client sends a new request. That is OK as the client will read this reply, notice that the sequence number is not the one expected, discard the reply, and call `recvfrom` again.

We call `sigsetjmp` and `siglongjmp` to avoid the race condition with the `SIGALRM` signal that we described in Section 18.5.

Figure 20.8 shows the first half of our `dg_send_recv` function.

1-5 We include a new header `unprtt.h`, shown in Figure 20.10, that defines the `rtt_info` structure that maintains the RTT information for a client. We define one of these structures, and numerous other variables.

Define `msghdr` structures and `hdr` structure

6-10 We want to hide the fact from the caller that we prepend a sequence number and a timestamp to each packet. The easiest way to do this is to use `writev`, writing our header (the `hdr` structure) followed by the caller's data, as a single UDP datagram. Recall that the output for `writev` on a datagram socket is a single datagram. This is easier than forcing the caller to allocate room at the front of its buffer for our use and is also faster than copying our header and the caller's data into one buffer (that we would have to allocate) for a single `sendto`. But since we are using UDP and have to specify a destination address, we must use the `iovec` capability of `sendmsg` and `recvmsg`, instead of `sendto` and `recvfrom`. Recall from Section 13.5 that some systems have a newer `msghdr` structure with ancillary data, while older systems still have the access rights members at the end of the structure. To avoid complicating the code with `#ifdefs` to handle these differences, we declare two `msghdr` structures as `static`, forcing their initialization to all zero bits by C and then just ignore the unused members at the end of the structures.

Initialize first time we are called

20-24 The first time we are called we call the `rtt_init` function.

Fill in `msghdr` structures

25-41 We fill in the two `msghdr` structures that are used for output and input. We increment the sending sequence number for this packet but do not set the sending timestamp until we send the packet (since it might be retransmitted, and each retransmission needs the current timestamp).

The second half of the function, along with the `sig_alrm` signal handler, is shown in Figure 20.9 (p. 548).

rtt/dg_send_recv.c

```
 1 #include    "unprtt.h"
 2 #include    <setjmp.h>

 3 #define RTT_DEBUG

 4 static struct rtt_info rttinfo;
 5 static int rttinit = 0;
 6 static struct msghdr msgsend, msgrecv;  /* assumed init to 0 */
 7 static struct hdr {
 8     uint32_t seq;               /* sequence # */
 9     uint32_t ts;                /* timestamp when sent */
10 } sendhdr, recvhdr;

11 static void sig_alrm(int signo);
12 static sigjmp_buf jmpbuf;

13 ssize_t
14 dg_send_recv(int fd, const void *outbuff, size_t outbytes,
15              void *inbuff, size_t inbytes,
16              const SA *destaddr, socklen_t destlen)
17 {
18     ssize_t n;
19     struct iovec iovsend[2], iovrecv[2];

20     if (rttinit == 0) {
21         rtt_init(&rttinfo);     /* first time we're called */
22         rttinit = 1;
23         rtt_d_flag = 1;
24     }
25     sendhdr.seq++;
26     msgsend.msg_name = destaddr;
27     msgsend.msg_namelen = destlen;
28     msgsend.msg_iov = iovsend;
29     msgsend.msg_iovlen = 2;
30     iovsend[0].iov_base = &sendhdr;
31     iovsend[0].iov_len = sizeof(struct hdr);
32     iovsend[1].iov_base = outbuff;
33     iovsend[1].iov_len = outbytes;

34     msgrecv.msg_name = NULL;
35     msgrecv.msg_namelen = 0;
36     msgrecv.msg_iov = iovrecv;
37     msgrecv.msg_iovlen = 2;
38     iovrecv[0].iov_base = &recvhdr;
39     iovrecv[0].iov_len = sizeof(struct hdr);
40     iovrecv[1].iov_base = inbuff;
41     iovrecv[1].iov_len = inbytes;
```

rtt/dg_send_recv.c

Figure 20.8 `dg_send_recv` function: first half.

rtt/dg_send_recv.c

```
42      Signal(SIGALRM, sig_alrm);
43      rtt_newpack(&rttinfo);      /* initialize for this packet */

44    sendagain:
45      sendhdr.ts = rtt_ts(&rttinfo);
46      Sendmsg(fd, &msgsend, 0);

47      alarm(rtt_start(&rttinfo)); /* calc timeout value & start timer */

48      if (sigsetjmp(jmpbuf, 1) != 0) {
49          if (rtt_timeout(&rttinfo) < 0) {
50              err_msg("dg_send_recv: no response from server, giving up");
51              rttinit = 0;        /* reinit in case we're called again */
52              errno = ETIMEDOUT;
53              return (-1);
54          }
55          goto sendagain;
56      }
57      do {
58          n = Recvmsg(fd, &msgrecv, 0);
59      } while (n < sizeof(struct hdr) || recvhdr.seq != sendhdr.seq);

60      alarm(0);                   /* stop SIGALRM timer */
61          /* calculate & store new RTT estimator values */
62      rtt_stop(&rttinfo, rtt_ts(&rttinfo) - recvhdr.ts);

63      return (n - sizeof(struct hdr));    /* return size of received datagram */
64  }

65  static void
66  sig_alrm(int signo)
67  {
68      siglongjmp(jmpbuf, 1);
69  }
```

rtt/dg_send_recv.c

Figure 20.9 `dg_send_recv` function: second half.

Establish signal handler

42–43 A signal handler is established for `SIGALRM` and `rtt_newpack` sets the retransmission counter to 0.

Send datagram

45–47 The current timestamp is obtained by `rtt_ts` and stored in the `hdr` structure that is prepended to the user's data. A single UDP datagram is sent by `sendmsg`. `rtt_start` returns the number of seconds for this timeout and the `SIGALRM` is scheduled by calling `alarm`.

Establish jump buffer

48 We establish a jump buffer for our signal handler with `sigsetjmp`. We wait for the next datagram to arrive by calling `recvmsg`. (We discussed the use of `sigsetjmp` and `siglongjmp` along with `SIGALRM` with Figure 18.9.) If the alarm timer expires, `sigsetjmp` returns 1.

Handle timeout

49-55 When a timeout occurs, `rtt_timeout` calculates the next *RTO* (the exponential backoff) and returns –1 if we should give up, or 0 if we should retransmit. If we give up we set `errno` to `ETIMEDOUT` and return to the caller.

Call `recvmsg`, compare sequence numbers

57-59 We wait for a datagram to arrive by calling `recvmsg`. When it returns, the datagram's length must be at least the size of our `hdr` structure and its sequence number must equal the sequence number that was sent. If either comparison is false, `recvmsg` is called again.

Turn off alarm and update RTT estimators

60-62 When the expected reply is received, the pending `alarm` is turned off and `rtt_stop` updates the RTT estimators. `rtt_ts` returns the current timestamp, and the timestamp from the received datagram is subtracted from this, giving the RTT.

`SIGALRM` handler

65-69 `siglongjmp` is called, causing the `sigsetjmp` in `dg_send_recv` to return 1.

We now look at the various RTT functions that were called by `dg_send_recv`. Figure 20.10 shows the `unprtt.h` header.

lib/unprtt.h

```
 1 #ifndef __unp_rtt_h
 2 #define __unp_rtt_h

 3 #include    "unp.h"

 4 struct rtt_info {
 5     float   rtt_rtt;            /* most recent measured RTT, seconds */
 6     float   rtt_srtt;           /* smoothed RTT estimator, seconds */
 7     float   rtt_rttvar;         /* smoothed mean deviation, seconds */
 8     float   rtt_rto;            /* current RTO to use, seconds */
 9     int     rtt_nrexmt;         /* #times retransmitted: 0, 1, 2, ... */
10     uint32_t rtt_base;          /* #sec since 1/1/1970 at start */
11 };

12 #define RTT_RXTMIN      2       /* min retransmit timeout value, seconds */
13 #define RTT_RXTMAX     60       /* max retransmit timeout value, seconds */
14 #define RTT_MAXNREXMT   3       /* max #times to retransmit */

15                 /* function prototypes */
16 void    rtt_debug(struct rtt_info *);
17 void    rtt_init(struct rtt_info *);
18 void    rtt_newpack(struct rtt_info *);
19 int     rtt_start(struct rtt_info *);
20 void    rtt_stop(struct rtt_info *, uint32_t);
21 int     rtt_timeout(struct rtt_info *);
22 uint32_t rtt_ts(struct rtt_info *);

23 extern int rtt_d_flag;          /* can be set nonzero for addl info */

24 #endif  /* __unp_rtt_h */
```

lib/unprtt.h

Figure 20.10 `unprtt.h` header.

rtt_info structure

4–11 This structure contains the variables necessary to time the packets between a client and server. The first four variables are from the equations given near the beginning of this section.

12–14 These constants define the minimum and maximum retransmission timeouts and the maximum number of times we retransmit.

Figure 20.11 shows a macro and the first two of our RTT functions.

lib/rtt.c

```
1 #include    "unprtt.h"
2 int     rtt_d_flag = 0;         /* debug flag; can be set nonzero by caller */
3 /*
4  * Calculate the RTO value based on current estimators:
5  *       smoothed RTT plus four times the deviation.
6  */
7 #define RTT_RTOCALC(ptr) ((ptr)->rtt_srtt + (4.0 * (ptr)->rtt_rttvar))
8 static float
9 rtt_minmax(float rto)
10 {
11     if (rto < RTT_RXTMIN)
12         rto = RTT_RXTMIN;
13     else if (rto > RTT_RXTMAX)
14         rto = RTT_RXTMAX;
15     return (rto);
16 }
17 void
18 rtt_init(struct rtt_info *ptr)
19 {
20     struct timeval tv;
21     Gettimeofday(&tv, NULL);
22     ptr->rtt_base = tv.tv_sec;  /* #sec since 1/1/1970 at start */
23     ptr->rtt_rtt = 0;
24     ptr->rtt_srtt = 0;
25     ptr->rtt_rttvar = 0.75;
26     ptr->rtt_rto = rtt_minmax(RTT_RTOCALC(ptr));
27     /* first RTO at (srtt + (4 * rttvar)) = 3 seconds */
28 }
```

lib/rtt.c

Figure 20.11 `RTT_RTOCALC` macro, `rtt_minmax` and `rtt_init` functions.

3–7 The `RTT_RTOCALC` macro calculates the *RTO* as the RTT estimator plus four times the mean deviation estimator.

8–16 `rtt_minmax` makes certain that the *RTO* is between the upper and lower limits in the `unprtt.h` header.

17–28 `rtt_init` is called by `dg_send_recv` the first time any packet is sent. `gettimeofday` returns the current time and date, in the same `timeval` structure that we saw with `select` (Section 6.3). We save only the current number of seconds since

the Unix Epoch, which is 00:00:00 January 1, 1970, Coordinated Universal Time (UTC). The measured RTT is set to 0 and the RTT and mean deviation estimators are set to 0 and 0.75, respectively, giving an initial *RTO* of 3 seconds.

> The `gettimeofday` function is not yet part of Posix.1 and might not be supported by some older implementations. It is required by Unix 98. We use it because it is quite common and it provides microsecond resolution on many hosts. An alternate function is the Posix.1 `times` function, but its resolution depends on the "clock tick" used by the kernel, normally 100 ticks per second, for a resolution of 10 ms.

Figure 20.12 shows the next three RTT functions.

lib/rtt.c

```
34 uint32_t
35 rtt_ts(struct rtt_info *ptr)
36 {
37     uint32_t ts;
38     struct timeval tv;

39     Gettimeofday(&tv, NULL);
40     ts = ((tv.tv_sec - ptr->rtt_base) * 1000) + (tv.tv_usec / 1000);
41     return (ts);
42 }

43 void
44 rtt_newpack(struct rtt_info *ptr)
45 {
46     ptr->rtt_nrexmt = 0;
47 }

48 int
49 rtt_start(struct rtt_info *ptr)
50 {
51     return ((int) (ptr->rtt_rto + 0.5));    /* round float to int */
52         /* return value can be used as: alarm(rtt_start(&foo)) */
53 }
```

lib/rtt.c

Figure 20.12 `rtt_ts`, `rtt_newpack`, and `rtt_start` functions.

34-42 `rtt_ts` returns the current timestamp for the caller to store as an unsigned 32-bit integer in the datagram being sent. We obtain the current time and date from `gettimeofday` and then subtract the number of seconds when `rtt_init` was called (the value saved in `rtt_base`). We convert this to milliseconds and also convert the microsecond value returned by `gettimeofday` into milliseconds. The timestamp is then the sum of these two value in milliseconds.

The difference between two calls to `rtt_ts` is the number of milliseconds between the two calls. But we store the millisecond timestamps in an unsigned 32-bit integer, instead of a `timeval` structure.

43-47 `rtt_newpack` just sets the retransmission counter to 0. This function should be called whenever a new packet is sent for the first time.

48-53 `rtt_start` returns the current *RTO* in seconds. The return value can then be used as the argument to `alarm`.

`rtt_stop`, shown in Figure 20.13, is called after a reply is received to update the RTT estimators and calculate the new *RTO*.

lib/rtt.c

```
62 void
63 rtt_stop(struct rtt_info *ptr, uint32_t ms)
64 {
65     double  delta;

66     ptr->rtt_rtt = ms / 1000.0; /* measured RTT in seconds */

67     /*
68      * Update our estimators of RTT and mean deviation of RTT.
69      * See Jacobson's SIGCOMM '88 paper, Appendix A, for the details.
70      * We use floating point here for simplicity.
71      */

72     delta = ptr->rtt_rtt - ptr->rtt_srtt;
73     ptr->rtt_srtt += delta / 8; /* g = 1/8 */

74     if (delta < 0.0)
75         delta = -delta;          /* |delta| */

76     ptr->rtt_rttvar += (delta - ptr->rtt_rttvar) / 4;    /* h = 1/4 */

77     ptr->rtt_rto = rtt_minmax(RTT_RTOCALC(ptr));
78 }
```

lib/rtt.c

Figure 20.13 `rtt_stop` function: update RTT estimators and calculate new *RTO*.

62–78 The second argument is the measured RTT, obtained by the caller by subtracting the received timestamp in the reply from the current timestamp (`rtt_ts`). The equations at the beginning of this section are then applied, storing new values for `rtt_srtt`, `rtt_rttvar`, and `rtt_rto`.

The final function, `rtt_timeout`, is shown in Figure 20.14. This function is called when the retransmission timer expires.

lib/rtt.c

```
83 int
84 rtt_timeout(struct rtt_info *ptr)
85 {
86     ptr->rtt_rto *= 2;           /* next RTO */

87     if (++ptr->rtt_nrexmt > RTT_MAXNREXMT)
88         return (-1);             /* time to give up for this packet */
89     return (0);
90 }
```

lib/rtt.c

Figure 20.14 `rtt_timeout` function: apply exponential backoff.

86 The current *RTO* is doubled: this is the exponential backoff.

87–89 If we have reached the maximum number of retransmissions, –1 is returned to tell the caller to give up; otherwise 0 is returned.

As an example, our client was run twice to two different echo servers across the Internet in the morning on a weekday. Five hundred lines were sent to each server. Eight packets were lost to the first server and 16 packets were lost to the second server. Of the 16 lost to the second server, one packet was lost twice in a row: that is, the packet had to be retransmitted two times before a reply was received. All other lost packets were handled with a single retransmission. We could verify that these packets were really lost by printing the sequence number of each received packet. If a packet is just delayed, and not lost, after the retransmission two replies will be received by the client: one corresponding to the original transmission that was delayed, and one corresponding to the retransmission. Notice we are unable to tell when we retransmit whether it was the client's request or the server's reply that was discarded.

> For the first edition of this book the author wrote a UDP server that randomly discarded packets to test this client. That is no longer needed; all we have to do is run the client to a server across the Internet, and we are almost guaranteed of some packet loss!

20.6 Binding Interface Addresses

One common use for our `get_ifi_info` function is with UDP applications that need to monitor all interfaces on a host to know when a datagram arrives, on which interface it arrives. This allows the receiving program to know the destination address of the UDP datagram, since that address is what determines the socket to which a datagram is delivered, even if the host does not support the `IP_RECVDSTADDR` socket option.

> Recall our discussion at the end of Section 20.2. If the host employs the common weak end system model, the destination IP address may differ from the IP address of the receiving interface. In this case all we can determine is the destination address of the datagram, which need not be an address assigned to the receiving interface. To determine the receiving interface requires either the `IP_RECVIF` or `IPV6_PKTINFO` socket option.
>
> Earlier we showed an example of binding all the interface addresses in Section 19.11 with our SNTP example.

Figure 20.15 is the first part of a simple example of this technique with a UDP server that binds all the unicast addresses, all the broadcast addresses, and finally the wildcard address.

Call `get_ifi_info` to obtain interface information

11-12 `get_ifi_info` obtains all the IPv4 addresses, including aliases, for all interfaces. The program then loops through each returned `ifi_info` structure.

Create UDP socket and `bind` unicast address

13-20 A UDP socket is created and the unicast address is bound to it. We also set the `SO_REUSEADDR` socket option, as we are binding the same port (`SERV_PORT`) for all the IP addresses.

> Not all implementations require that this socket option be set. Berkeley-derived implementations, for example, do not require the option and allow a new `bind` of an already bound port if

advio/udpserv03.c

```
 1 #include    "unpifi.h"
 2 void    mydg_echo(int, SA *, socklen_t, SA *);
 3 int
 4 main(int argc, char **argv)
 5 {
 6     int     sockfd;
 7     const int on = 1;
 8     pid_t   pid;
 9     struct ifi_info *ifi, *ifihead;
10     struct sockaddr_in *sa, cliaddr, wildaddr;
11     for (ifihead = ifi = Get_ifi_info(AF_INET, 1);
12          ifi != NULL; ifi = ifi->ifi_next) {
13             /* bind unicast address */
14         sockfd = Socket(AF_INET, SOCK_DGRAM, 0);
15         Setsockopt(sockfd, SOL_SOCKET, SO_REUSEADDR, &on, sizeof(on));
16         sa = (struct sockaddr_in *) ifi->ifi_addr;
17         sa->sin_family = AF_INET;
18         sa->sin_port = htons(SERV_PORT);
19         Bind(sockfd, (SA *) sa, sizeof(*sa));
20         printf("bound %s\n", Sock_ntop((SA *) sa, sizeof(*sa)));
21         if ( (pid = Fork()) == 0) {  /* child */
22             mydg_echo(sockfd, (SA *) &cliaddr, sizeof(cliaddr), (SA *) sa);
23             exit(0);            /* never executed */
24         }
```

advio/udpserv03.c

Figure 20.15 First part of UDP server that binds all addresses.

the new IP address being bound (a) is not the wildcard, and (b) differs from all the IP addresses that are already bound to the port. Solaris 2.5, however, requires this option for the second bind of a unicast address to the same port to succeed.

fork child for this address

21–24 A child is forked and the function mydg_echo is called for the child. This function waits for any datagram to arrive on this socket and echoes it back to the sender.

Figure 20.16 shows the next part of the main function, which handles broadcast addresses.

Bind broadcast address

25–42 If the interface supports broadcasting, a UDP socket is created and the broadcast address is bound to it. This time we allow the bind to fail with an error of EADDRINUSE because if an interface has multiple addresses (aliases) on the same subnet, then each of the different unicast addresses will have the same broadcast address. We showed an example of this following Figure 16.6. In this scenario we expect only the first bind to succeed.

advio/udpserv03.c

```
25          if (ifi->ifi_flags & IFF_BROADCAST) {
26                  /* try to bind broadcast address */
27              sockfd = Socket(AF_INET, SOCK_DGRAM, 0);
28              Setsockopt(sockfd, SOL_SOCKET, SO_REUSEADDR, &on, sizeof(on));
29              sa = (struct sockaddr_in *) ifi->ifi_brdaddr;
30              sa->sin_family = AF_INET;
31              sa->sin_port = htons(SERV_PORT);
32              if (bind(sockfd, (SA *) sa, sizeof(*sa)) < 0) {
33                  if (errno == EADDRINUSE) {
34                      printf("EADDRINUSE: %s\n",
35                             Sock_ntop((SA *) sa, sizeof(*sa)));
36                      Close(sockfd);
37                      continue;
38                  } else
39                      err_sys("bind error for %s",
40                              Sock_ntop((SA *) sa, sizeof(*sa)));
41              }
42              printf("bound %s\n", Sock_ntop((SA *) sa, sizeof(*sa)));
43              if ( (pid = Fork()) == 0) {  /* child */
44                  mydg_echo(sockfd, (SA *) &cliaddr, sizeof(cliaddr),
45                            (SA *) sa);
46                  exit(0);        /* never executed */
47              }
48          }
49      }
```

advio/udpserv03.c

Figure 20.16 Second part of UDP server that binds all addresses.

`fork` child

43–47 A child is spawned and it calls the function `mydg_echo`.

The final part of the `main` function is shown in Figure 20.17. This code binds the wildcard address to handle any destination addresses other than the unicast and broadcast addresses that we have already bound. The only datagrams that should arrive on this socket should be those destined to the limited broadcast address (255.255.255.255).

Create socket and bind wildcard address

50–62 A UDP socket is created, the `SO_REUSEADDR` socket option is set, and the wildcard IP address is bound. A child is spawned, which calls the `mydg_echo` function.

`main` function terminates

63 The `main` function terminates, and the server continues executing as all the children that were spawned.

The function `mydg_echo`, which is executed by all the children, is shown in Figure 20.18.

advio/udpserv03.c

```
50          /* bind wildcard address */
51      sockfd = Socket(AF_INET, SOCK_DGRAM, 0);
52      Setsockopt(sockfd, SOL_SOCKET, SO_REUSEADDR, &on, sizeof(on));

53      bzero(&wildaddr, sizeof(wildaddr));
54      wildaddr.sin_family = AF_INET;
55      wildaddr.sin_addr.s_addr = htonl(INADDR_ANY);
56      wildaddr.sin_port = htons(SERV_PORT);
57      Bind(sockfd, (SA *) &wildaddr, sizeof(wildaddr));
58      printf("bound %s\n", Sock_ntop((SA *) &wildaddr, sizeof(wildaddr)));

59      if ( (pid = Fork()) == 0) {  /* child */
60          mydg_echo(sockfd, (SA *) &cliaddr, sizeof(cliaddr), (SA *) sa);
61          exit(0);                 /* never executed */
62      }
63      exit(0);
64  }
```

advio/udpserv03.c

Figure 20.17 Final part of UDP server that binds all addresses.

advio/udpserv03.c

```
65  void
66  mydg_echo(int sockfd, SA *pcliaddr, socklen_t clilen, SA *myaddr)
67  {
68      int     n;
69      char    mesg[MAXLINE];
70      socklen_t len;

71      for ( ; ; ) {
72          len = clilen;
73          n = Recvfrom(sockfd, mesg, MAXLINE, 0, pcliaddr, &len);
74          printf("child %d, datagram from %s", getpid(),
75                 Sock_ntop(pcliaddr, len));
76          printf(", to %s\n", Sock_ntop(myaddr, clilen));

77          Sendto(sockfd, mesg, n, 0, pcliaddr, len);
78      }
79  }
```

advio/udpserv03.c

Figure 20.18 mydg_echo function.

New argument

65–66 The fourth argument to this function is the IP address that was bound to the socket. This socket should receive only datagrams destined to that IP address. If the IP address is the wildcard, then the socket should receive only datagrams that are not matched by some other socket bound to the same port.

Read datagram and echo reply

71–78 The datagram is read with recvfrom and sent back to the client with sendto. This function also prints the client's IP address and the IP address that was bound to the socket.

We now run this program on our host `bsdi` after establishing three alias address for the `ef0` Ethernet interface. The three aliases have host IDs of 50, 51, and 52, but all have the same broadcast address of 206.62.226.63.

```
bsdi % udpserv03
bound 206.62.226.66.9877              unicast address of we0 interface
bound 206.62.226.95.9877              broadcast address of we0 interface
bound 206.62.226.35.9877              primary unicast address of ef0 interface
bound 206.62.226.63.9877              broadcast address of ef0 interface
bound 206.62.226.50.9877              first unicast alias
EADDRINUSE: 206.62.226.63.9877
bound 206.62.226.51.9877              second unicast alias
EADDRINUSE: 206.62.226.63.9877
bound 206.62.226.52.9877              third unicast alias
EADDRINUSE: 206.62.226.63.9877
bound 127.0.0.1.9877                  loopback interface
bound 0.0.0.0.9877                    wildcard
```

Note that the three `bind`s of the broadcast addresses for the aliases fail, as we expect. We can check that all these sockets are bound to the indicated IP address and port using `netstat`:

```
bsdi % netstat -na | grep 9877
udp        0      0  *.9877                 *.*
udp        0      0  127.0.0.1.9877         *.*
udp        0      0  206.62.226.52.9877     *.*
udp        0      0  206.62.226.51.9877     *.*
udp        0      0  206.62.226.50.9877     *.*
udp        0      0  206.62.226.63.9877     *.*
udp        0      0  206.62.226.35.9877     *.*
udp        0      0  206.62.226.95.9877     *.*
udp        0      0  206.62.226.66.9877     *.*
```

We should note that our design of one child process per socket is for simplicity, and other designs are possible. For example, to reduce the number of processes the program could manage all the descriptors itself using `select`, never calling `fork`. The problem with this design is the added code complexity. While it is easy to use `select` for all the descriptors, we would have to maintain some type of mapping of each descriptor to its bound IP address (probably an array of structures), so we could print the destination IP address when a datagram was read from a socket. (We used this solution in Section 19.11.) It is often simpler to use a single process or a single thread for one operation or descriptor, instead of having a single process multiplex many different operations or descriptors.

20.7 Concurrent UDP Servers

Most UDP servers are iterative: the server waits for a client request, reads the request, processes the request, sends back the reply, and then waits for the next client request. But when the processing of the client request takes a long time, some form of concurrency is desired.

The definition of a "long time" is whatever is considered too much time for another client to wait while the current client is serviced. For example, if two client requests arrive within 10 ms of each other, and it takes an average of 5 seconds of clock time to service a client, then the second client will have to wait about 10 seconds for its reply, instead of about 5 seconds if the request were handled as soon as it arrived.

With TCP it is simple to just `fork` a new child (or create a new thread as we will see in Chapter 23) and let the child handle the new client. What simplifies this server concurrency when TCP is being used is that every client connection is unique: the TCP socket pair is unique for every connection. But with UDP we must deal with two different types of servers.

1. First is a simple UDP server that reads a client request, sends a reply, and is then finished with the client. In this scenario the server that reads the client request can `fork` a child and let it handle the request. The "request," that is, the contents of the datagram and the socket address structure containing the client's protocol address, are passed to the child in its memory image from `fork`. The child then sends its reply directly to the client.

2. Second is a UDP server that exchanges multiple datagrams with the client. The problem is that the only port number that the client knows for the server is its well-known port. The client sends the first datagram of its request to that port, but how does the server distinguish between subsequent datagrams from that client, and new requests? The typical solution to this problem is for the server to create a new socket for each client, `bind` an ephemeral port to that socket, and use that socket for all its replies. This requires that the client look at the port number of the server's first reply and send subsequent datagrams for this request to that port.

An example of the second type of UDP server is TFTP, the Trivial File Transfer Protocol. To transfer a file using TFTP normally requires many datagrams (hundreds or thousands, depending on the file size), because the protocol sends only 512 bytes per datagram. The client sends a datagram to the server's well-known port (69) specifying the file to send or receive. The server reads the request but sends its reply from another socket that it creates and binds to an ephemeral port. All subsequent datagrams between the client and server for this file use the new socket. This allows the main TFTP server to continue to handle other client requests, which arrive at port 69, while this file transfer takes place (perhaps seconds or even minutes).

If we assume a stand-alone TFTP server (i.e., not invoked by `inetd`), we have the scenario shown in Figure 20.19. We assume that the ephemeral port bound by the child to its new socket is 2134.

If `inetd` is used, the scenario involves one more step. Recall from Figure 12.6 that most UDP servers specify the *wait-flag* as `wait`. In our description following Figure 12.10 we said that this causes `inetd` to stop selecting on the socket until its child terminates, allowing its child to read the datagram that has arrived on the socket. Figure 20.20 shows the steps involved.

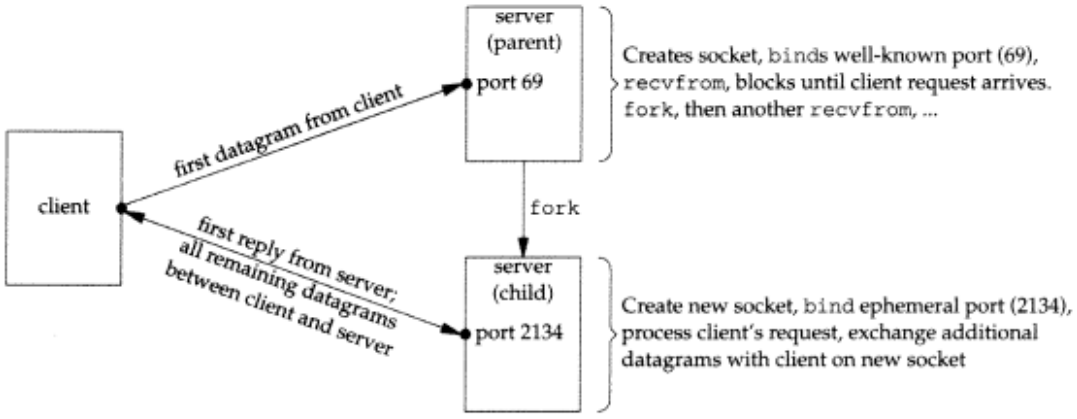


Figure 20.19 Processes involved in stand-alone concurrent UDP server.

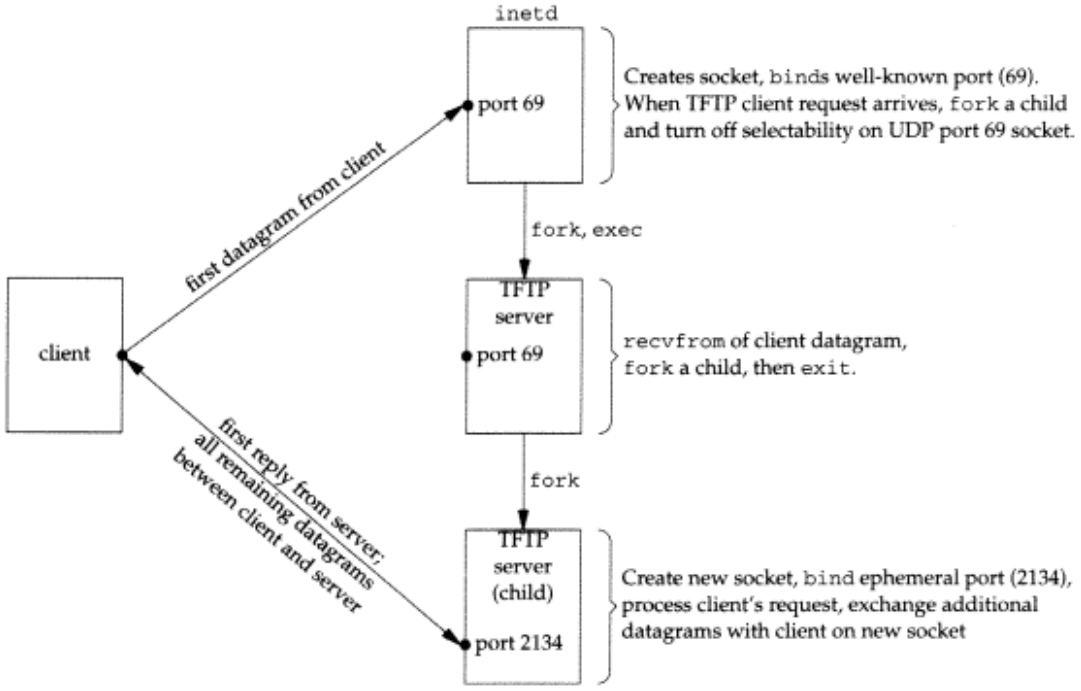


Figure 20.20 UDP concurrent server invoked by `inetd`.

The TFTP server that is the child of `inetd` calls `recvfrom` and reads the client request. It then `fork`s a child of its own, and that child will process the client request. The TFTP server then calls `exit`, sending `SIGCHLD` to `inetd`, which we said tells `inetd` to again `select` on the socket bound to UDP port 69.

20.8 IPv6 Packet Information

IPv6 allows an application to specify up to four pieces of information for an outgoing datagram:

1. the source IPv6 address,
2. the outgoing interface index,
3. the outgoing hop limit, and
4. the next-hop address.

This information is sent as ancillary data with `sendmsg`. Three similar pieces of information can be returned for a received packet, and they are returned as ancillary data with `recvmsg`:

1. the destination IPv6 address,
2. the arriving interface index, and
3. the arriving hop limit.

Figure 20.21 summarizes the contents of the ancillary data, which we discuss shortly.

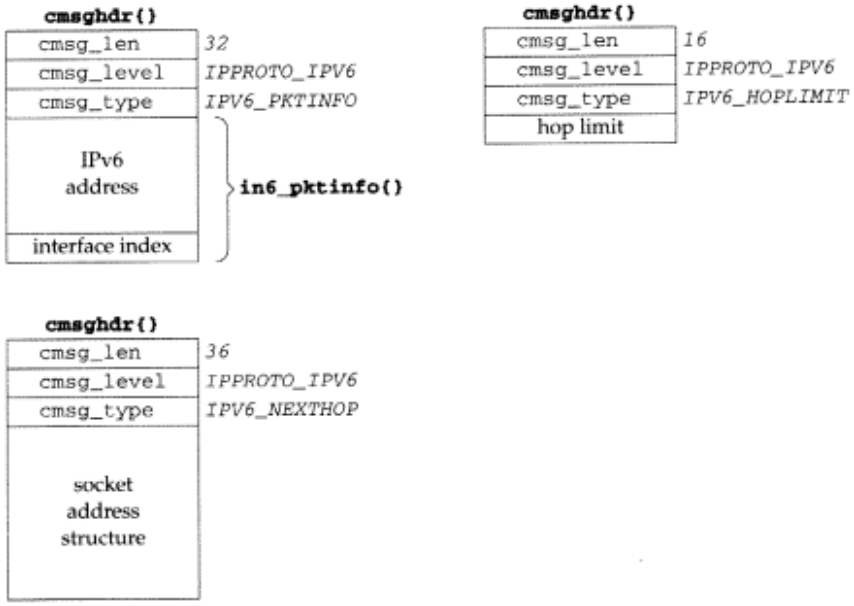


Figure 20.21 Ancillary data for IPv6 packet information.

An `in6_pktinfo` structure contains either the source IPv6 address and the outgoing interface index for an outgoing datagram, or the destination IPv6 address and the arriving interface index for a received datagram:

```
struct in6_pktinfo {
  struct in6_addr ipi6_addr;     /* src/dst IPv6 address */
  int             ipi6_ifindex; /* send/recv interface index */
};
```

This structure is defined by including the <netinet/in.h> header. In the cmsghdr structure containing this ancillary data, the cmsg_level member will be IPPROTO_IPV6, the cmsg_type member will be IPV6_PKTINFO, and the first byte of data will be the first byte of the in6_pktinfo structure. In the example in Figure 20.21 we assume no padding between the cmsghdr structure and the data, and 4 bytes for an integer.

Nothing special need be done to send this information: just specify the control information as ancillary data for sendmsg. But this information is returned as ancillary data by recvmsg only if the application has enabled the IPV6_PKTINFO socket option.

Outgoing and Arriving Interface

Interfaces on an IPv6 node are identified by small positive integers, as we discussed in Section 17.6. Recall that no interface is ever assigned an index of 0. When specifying the outgoing interface, if the ipi6_ifindex value is 0, the kernel will choose the outgoing interface. If the application specifies an outgoing interface for a multicast packet, the interface specified by the ancillary data overrides any interface specified by the IPV6_MULTICAST_IF socket option, for this datagram only.

Source and Destination IPv6 Address

The source IPv6 address is normally specified by calling bind. But supplying the source address together with the data may require less overhead. This option also allows a server to guarantee that the source address of its reply equals the destination address of the client's request, a feature that some clients require and which is hard to accomplish with IPv4 (Exercise 20.4).

When specifying the source IPv6 address as ancillary data, if the ipi6_addr member of the in6_pktinfo structure is IN6ADDR_ANY_INIT, then (a) if an address is currently bound to the socket, it is used as the source address, or (b) if no address is currently bound to the socket, the kernel will choose the source address. If the ipi6_addr member is not the unspecified address, but the socket has already bound a source address, then the ipi6_addr value overrides the already-bound source address for this output operation only. The kernel will verify that the requested source address is indeed a unicast address assigned to the node.

When the in6_pktinfo structure is returned as ancillary data by recvmsg, the ipi6_addr member contains the destination IPv6 address from the received packet. This is similar in concept to the IP_RECVDSTADDR socket option for IPv4.

Specifying and Receiving the Hop Limit

The outgoing hop limit is normally specified with either the IPV6_UNICAST_HOPS socket option for unicast datagrams (Section 7.8) or the IPV6_MULTICAST_HOPS socket option for multicast datagrams (Section 19.5). Specifying the hop limit as ancillary data lets us override either the kernel's default or a previously specified value, for either a unicast destination or a multicast destination, for a single output operation. Returning

the received hop limit is useful for programs such as `traceroute` and for a class of IPv6 applications that need to verify that the received hop limit is 255 (e.g., that the packet has not been forwarded).

The received hop limit is returned as ancillary data by `recvmsg` only if the application has enabled the `IPV6_HOPLIMIT` socket option. In the `cmsghdr` structure containing this ancillary data, the `cmsg_level` member will be `IPPROTO_IPV6`, the `cmsg_type` member will be `IPV6_HOPLIMIT`, and the first byte of data will be the first byte of the integer hop limit. We showed this in Figure 20.21. Realize that the value returned as ancillary data is the actual value from the received datagram, while the value returned by a `getsockopt` of the `IPV6_UNICAST_HOPS` option is the default value that the kernel will use for outgoing datagrams on the socket.

Nothing special need be done to specify the outgoing hop limit: just specify the control information as ancillary data for `sendmsg`. The normal values for the hop limit are between 0 and 255, inclusive, but if the integer value is –1, this tells the kernel to use its default.

> The hop limit is not contained in the `in6_pktinfo` structure for the following reason. Some UDP servers want to respond to client requests by sending their reply out the same interface on which the request was received and with the source IPv6 address of the reply equal to the destination IPv6 address of the request. To do this the application can enable just the `IPV6_PKTINFO` socket option and then use the received control information from `recvmsg` as the outgoing control information for `sendmsg`. The application need not examine or modify the `in6_pktinfo` structure at all. But if the hop limit were contained in this structure, the application would have to parse the received control information and change the hop limit member, since the received hop limit is not the desired value for an outgoing packet.

Specifying the Next Hop Address

The `IPV6_NEXTHOP` ancillary data object specifies the next hop for the datagram as a socket address structure. In the `cmsghdr` structure containing this ancillary data, the `cmsg_level` member is `IPPROTO_IPV6`, the `cmsg_type` member is `IPV6_NEXTHOP`, and the first byte of data is the first byte of the socket address structure.

In Figure 20.21 we show an example of this ancillary data object, assuming that the socket address structure is a 24-byte `sockaddr_in6` structure. In this case the node identified by that address must be a neighbor of the sending host. If that address equals the destination IPv6 address of the datagram, then this is equivalent to the existing `SO_DONTROUTE` socket option. Setting this option requires superuser privileges.

20.9 Summary

There are applications that want to know the destination IP address and the received interface for a UDP datagram. The `IP_RECVDSTADDR` and `IP_RECVIF` socket options can be enabled to return this information as ancillary data with each datagram. Similar information, along with the received hop limit, can be returned for IPv6 sockets by enabling the `IPV6_PKTINFO` socket option.

Despite all the features provided by TCP that are not provided by UDP, there are times to use UDP. UDP must be used for broadcasting or multicasting. UDP can be used for simple request–reply scenarios, but some form of reliability must then be added to the application. UDP should not be used for bulk data transfer.

We added reliability to our UDP client in Section 20.5 by detecting lost packets using a timeout and retransmission. We modified our retransmission timeout dynamically by adding a timestamp to each packet and keeping track of two estimators: the RTT and its mean deviation. We also added a sequence number to verify that a given reply is the one expected. Our client still employed a simple stop-and-wait protocol, but that is the type of application for which UDP can be used.

Exercises

20.1 In Figure 20.18 why are there two calls to `printf`.

20.2 Can `dg_send_recv` (Figures 20.8 and 20.9) ever return 0?

20.3 Recode `dg_send_recv` to use `select` and its timer, instead of using `alarm`, `SIGALRM`, `sigsetjmp`, and `siglongjmp`.

20.4 How can an IPv4 server guarantee that the source address of its reply equals the destination address of the client's request (e.g., similar to the functionality provided by the `IPV6_PKTINFO` socket option)?

20.5 The `main` function in Section 20.6 is protocol dependent on IPv4. Recode it to be protocol independent. Require the user to specify one or two command-line arguments, the first being an optional IP address (e.g., 0.0.0.0 or 0::0), and the second being a required port number. Then call `udp_client` just to obtain the address family, port number, and length of the socket address structure.

What happens if you call `udp_client`, as suggested, without specifying a *hostname* argument, because `udp_client` does not specify the `AI_PASSIVE` hint to `getaddrinfo`?

20.6 Run the client in Figure 20.6 to an echo server across the Internet after modifying the RTT functions to print each RTT. Also modify the `dg_send_recv` function to print each received sequence number. Plot the resulting RTTs along with the estimators for the RTT and its mean deviation.

21

Out-of-Band Data

21.1 Introduction

Many transport layers have the concept of *out-of-band* data, which is sometimes called *expedited data*. The idea is that something important occurs at one end of the connection and that end wants to tell its peer quickly. By "quickly" we mean that this notification should be sent before any "normal" (sometimes called "in-band") data that is already queued to be sent. That is, out-of-band data is considered higher priority than the normal data. But instead of using two connections between the client and server, out-of-band data is mapped onto the existing connection.

Unfortunately once we get beyond the general concepts, and down to the real world, almost every transport layer has a different implementation of out-of-band data. In this chapter we focus on TCP's model of out-of-band data, provide numerous small examples of how it is handled by the sockets API, and then use it to write some simple client–server heartbeat functions that can detect when the peer process either crashes or is unreachable.

21.2 TCP Out-of-Band Data

TCP does not have true *out-of-band data*. Instead, TCP provides an *urgent mode* that we now describe. Assume a process has written N bytes of data to a TCP socket and that data is queued by TCP in the socket send buffer, waiting to be sent to the peer. We show this in Figure 21.1 and have labeled the data bytes 1 through N.

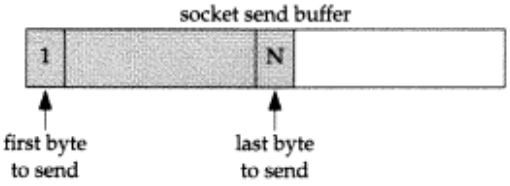


Figure 21.1 Socket send buffer containing data to send.

The process now writes a single byte of out-of-band data, containing the ASCII character `a`, using the `send` function and the `MSG_OOB` flag:

```
send(fd, "a", 1, MSG_OOB);
```

TCP places the data in the next available position in the socket send buffer and sets its *urgent pointer* for this connection to be the next available location. We show this in Figure 21.2 and have labeled the out-of-band byte "OOB."

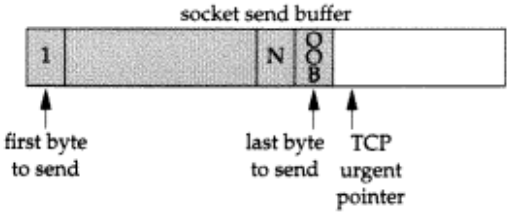


Figure 21.2 Socket send buffer after 1 byte of out-of-band data is written by application.

> TCP's urgent pointer has a sequence number one greater than the byte of data that is written with the `MSG_OOB` flag. As discussed on pp. 292–296 of TCPv1, this is a historical artifact that is now emulated by all implementations. As long as the sending TCP and the receiving TCP agree on the interpretation of TCP's urgent pointer, all is OK.
>
> In Section 7.9 we mentioned the new `TCP_STDURG` socket option that can change this interpretation of which byte the urgent pointer points to. This option should never need to be set.

Given the state of the TCP socket send buffer shown in Figure 21.2, the next segment sent by TCP will have its URG flag set in the TCP header and the urgent offset field in the TCP header will point to the byte following the out-of-band byte. But this segment may or may not contain the byte that we have labeled as OOB. Whether the OOB byte is sent depends on the number of bytes ahead of it in the socket send buffer, the segment size that TCP is sending to the peer, and the current window advertised by the peer.

> We have used the terms *urgent pointer* and *urgent offset*. At the TCP level the two are different. The 16-bit value in the TCP header is called the urgent offset and it must be added to the sequence number field in the header to obtain the 32-bit urgent pointer. TCP looks at the urgent offset only if another bit in the header is set, and this bit is called the *URG flag*. From a programming perspective we need not worry about this detail and just refer to TCP's urgent pointer.

This is an important characteristic of TCP's urgent mode: the TCP header indicates that the sender has entered urgent mode (i.e., the URG flag is set along with the urgent offset), but the actual byte of data referred to by the urgent pointer need not be sent. Indeed, if the sending TCP is flow control stopped (the receiver's socket receive buffer is full, so its TCP has advertised a window of 0 to the sending TCP), the urgent notification is sent without any data (pp. 1016–1017 of TCPv2) as we show in Figures 21.10 and 21.11. This is one reason why applications use TCP's urgent mode (i.e., out-of-band data): the urgent notification is *always* sent to the peer TCP even if the flow of data is stopped by TCP's flow control.

What happens if we send multiple bytes of out-of-band data, as in

```
send(fd, "abc", 3, MSG_OOB);
```

In this example TCP's urgent pointer points one beyond the final byte; that is, the final byte (the `c`) is considered the out-of-band byte.

Now that we have covered the sending of out-of-band data let's look at it from the receiver's side.

1. When TCP receives a segment with the URG flag set, the urgent pointer is examined to see whether this pointer refers to *new* out-of-band data. That is, whether this is the first time TCP's urgent mode has referenced this particular byte in the stream of data from the sender to the receiver. It is common for the sending TCP to send multiple segments (typically over a short period of time) containing the URG flag but with the urgent pointer pointing to the same byte of data. Only the first of these segments causes the receiving process to be notified that new out-of-band data has arrived.

2. The receiving process is notified when a new urgent pointer arrives. First the `SIGURG` signal is sent to the owner of the socket, assuming either `fcntl` or `ioctl` has been called to establish an owner for the socket (Figure 7.15), and assuming the process has established a signal handler for this signal. Second, if the process is blocked in a call to `select` waiting for this socket descriptor to have an exception condition, `select` returns.

   These two potential notifications to the receiving process take place when a new urgent pointer arrives, regardless whether the actual byte of data pointed to by the urgent pointer has arrived at the receiving TCP.

3. When the actual byte of data pointed to by the urgent pointer arrives at the receiving TCP, the data byte can be pulled out-of-band or left inline. By default the `SO_OOBINLINE` socket option is *not* set for a socket so the single byte of data is not placed into the socket receive buffer. Instead, the data byte is placed into a separate 1-byte out-of-band buffer for this connection (pp. 986–988 of TCPv2). The only way for the process to read from this special 1-byte buffer is to call `recv`, `recvfrom`, or `recvmsg` and specify the `MSG_OOB` flag.

   If, however, the process sets the `SO_OOBINLINE` socket option, then the single byte of data referred to by TCP's urgent pointer is left in the normal socket

receive buffer. The process cannot specify the MSG_OOB flag to read the data byte in this case. The process will know when it reaches this byte of data by checking the *out-of-band mark* for this connection, as we describe in Section 21.3.

Some errors are possible.

1. If the process asks for out-of-band data (e.g., specifying the MSG_OOB flag) but the peer has not sent any, EINVAL is returned.
2. If the process has been notified that the peer has sent an out-of-band byte (e.g., by SIGURG or select), and the process tries to read it, but that byte has not yet arrived, EWOULDBLOCK is returned. All the process can do at this point is read from the socket receive buffer (possibly discarding the data if it has no room to store the data), to make space in the buffer so that the peer TCP can send the out-of-band byte.
3. If the process tries to read the same out-of-band byte multiple times, EINVAL is returned.
4. If the process has set the SO_OOBINLINE socket option and then tries to read the out-of-band data by specifying MSG_OOB, EINVAL is returned.

Simple Example Using SIGURG

We now show a trivial example of sending and receiving out-of-band data. Figure 21.3 shows the sending program.

oob/tcpsend01.c

```
 1 #include    "unp.h"

 2 int
 3 main(int argc, char **argv)
 4 {
 5     int     sockfd;

 6     if (argc != 3)
 7         err_quit("usage: tcpsend01 <host> <port#>");

 8     sockfd = Tcp_connect(argv[1], argv[2]);

 9     Write(sockfd, "123", 3);
10     printf("wrote 3 bytes of normal data\n");
11     sleep(1);

12     Send(sockfd, "4", 1, MSG_OOB);
13     printf("wrote 1 byte of OOB data\n");
14     sleep(1);

15     Write(sockfd, "56", 2);
16     printf("wrote 2 bytes of normal data\n");
17     sleep(1);
```

```
18      Send(sockfd, "7", 1, MSG_OOB);
19      printf("wrote 1 byte of OOB data\n");
20      sleep(1);

21      Write(sockfd, "89", 2);
22      printf("wrote 2 bytes of normal data\n");
23      sleep(1);

24      exit(0);
25  }
```

oob/tcpsend01.c

Figure 21.3 Simple out-of-band sending program.

Nine bytes are sent, with a 1-second `sleep` between each output operation. The purpose of the pause is to let each `write` or `send` be transmitted as a single TCP segment and received as such by the other end. We talk later about some of the timing considerations with out-of-band data. When we run this program we see the expected output:

```
solaris % tcpsend01 bsdi 9999
wrote 3 bytes of normal data
wrote 1 byte of OOB data
wrote 2 bytes of normal data
wrote 1 byte of OOB data
wrote 2 bytes of normal data
```

Figure 21.4 is the receiving program.

Establish signal handler and socket owner

16-17 The signal handler for `SIGURG` is established and `fcntl` sets the owner of the connected socket.

> Notice that we do not establish the signal handler until `accept` returns. There is a small probability that out-of-band data can arrive after our TCP completes the three-way handshake but before `accept` returns, which we would miss. But if we established the signal handler before calling `accept` and also set the owner of the listening socket (which carries over to the connected socket), then if out-of-band data arrives before `accept` returns, our signal handler won't yet have a value for `connfd`. If this scenario is important for the application, it should have initialized `connfd` to –1, check for this value in the signal handler, and if true just set a flag for the main loop to check after `accept` returns.

18-25 The process reads from the socket, printing each string that is returned by `read`. When the sender terminates the connection, the receiver then terminates.

`SIGURG` handler

27-36 Our signal handler calls `printf`, reads the out-of-band byte by specifying the `MSG_OOB` flag, and then prints the returned data. Notice that we ask for up to 100 bytes in the call to `recv`, but as we will see shortly, only 1 byte is ever returned as out-of-band data.

> As stated earlier, calling the unsafe `printf` function from a signal handler is not recommended. We do it just to see what's happening with our programs.

oob/tcprecv01.c

```
 1 #include    "unp.h"
 2 int     listenfd, connfd;
 3 void    sig_urg(int);
 4 int
 5 main(int argc, char **argv)
 6 {
 7     int     n;
 8     char    buff[100];
 9     if (argc == 2)
10         listenfd = Tcp_listen(NULL, argv[1], NULL);
11     else if (argc == 3)
12         listenfd = Tcp_listen(argv[1], argv[2], NULL);
13     else
14         err_quit("usage: tcprecv01 [ <host> ] <port#>");
15     connfd = Accept(listenfd, NULL, NULL);
16     Signal(SIGURG, sig_urg);
17     Fcntl(connfd, F_SETOWN, getpid());
18     for ( ; ; ) {
19         if ( (n = Read(connfd, buff, sizeof(buff) - 1)) == 0) {
20             printf("received EOF\n");
21             exit(0);
22         }
23         buff[n] = 0;            /* null terminate */
24         printf("read %d bytes: %s\n", n, buff);
25     }
26 }
27 void
28 sig_urg(int signo)
29 {
30     int     n;
31     char    buff[100];
32     printf("SIGURG received\n");
33     n = Recv(connfd, buff, sizeof(buff) - 1, MSG_OOB);
34     buff[n] = 0;                /* null terminate */
35     printf("read %d OOB byte: %s\n", n, buff);
36 }
```

oob/tcprecv01.c

Figure 21.4 Simple out-of-band receiving program.

Here is the output when we run the receiving program, and then run the sending program from Figure 21.3.

```
bsdi % tcprecv01 9999
read 3 bytes: 123
SIGURG received
read 1 OOB byte: 4
read 2 bytes: 56
```

```
SIGURG received
read 1 OOB byte: 7
read 2 bytes: 89
received EOF
```

The results are as we expect. Each sending of out-of-band data by the sender generates `SIGURG` for the receiver, which then reads the single out-of-band byte.

Simple Example Using `select`

We now redo our out-of-band receiver to use `select` instead of the `SIGURG` signal. Figure 21.5 is the receiving program.

oob/tcprecv02.c

```
 1 #include    "unp.h"
 2 int
 3 main(int argc, char **argv)
 4 {
 5     int     listenfd, connfd, n;
 6     char    buff[100];
 7     fd_set  rset, xset;
 8     if (argc == 2)
 9         listenfd = Tcp_listen(NULL, argv[1], NULL);
10     else if (argc == 3)
11         listenfd = Tcp_listen(argv[1], argv[2], NULL);
12     else
13         err_quit("usage: tcprecv02 [ <host> ] <port#>");
14     connfd = Accept(listenfd, NULL, NULL);
15     FD_ZERO(&rset);
16     FD_ZERO(&xset);
17     for ( ; ; ) {
18         FD_SET(connfd, &rset);
19         FD_SET(connfd, &xset);
20         Select(connfd + 1, &rset, NULL, &xset, NULL);
21         if (FD_ISSET(connfd, &xset)) {
22             n = Recv(connfd, buff, sizeof(buff) - 1, MSG_OOB);
23             buff[n] = 0;        /* null terminate */
24             printf("read %d OOB byte: %s\n", n, buff);
25         }
26         if (FD_ISSET(connfd, &rset)) {
27             if ( (n = Read(connfd, buff, sizeof(buff) - 1)) == 0) {
28                 printf("received EOF\n");
29                 exit(0);
30             }
31             buff[n] = 0;        /* null terminate */
32             printf("read %d bytes: %s\n", n, buff);
33         }
34     }
35 }
```

oob/tcprecv02.c

Figure 21.5 Receiving program that (incorrectly) uses `select` to be notified of out-of-band data.

15-20 The process calls `select` waiting for either normal data (the read set, `rset`) or out-of-band data (the exception set, `xset`). In each case the received data is printed.

When we run this program and then run the same sending program as earlier (Figure 21.3), we encounter the following error:

```
bsdi % tcprecv02 8888
read 3 bytes: 123
read 1 OOB byte: 4
recv error: Invalid argument
```

The problem is that `select` indicates an exception condition until the process reads *beyond* the out-of-band data (pp. 530–531 of TCPv2). We cannot read the out-of-band data more than once because after we read it the first time, the kernel clears the 1-byte out-of-band buffer. When we call `recv` specifying the `MSG_OOB` flag the second time, BSD/OS returns `EINVAL` while Solaris 2.5 returns `EAGAIN`. (Posix.1g specifies `EINVAL` as the error for this condition.)

The solution is to `select` for an exception condition only after reading normal data. Figure 21.6 is a modification of Figure 21.5 that handles this scenario correctly.

5 We declare a new variable named `justreadoob` that indicates whether we just read out-of-band data or not. This flag determines whether or not to `select` for an exception condition.

26-27 When we set the `justreadoob` flag we must also clear the bit for this descriptor in the exception set.

The program now works as expected.

21.3 `sockatmark` Function

Whenever out-of-band data is received, there is an associated *out-of-band mark*. This is the position in the normal stream of data *at the sender* when the sending process sent the out-of-band byte. The receiving process determines whether or not it is at the out-of-band mark by calling the `sockatmark` function while it reads from the socket.

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

int sockatmark(int sockfd);
```

Returns: 1 if at out-of-band mark, 0 if not at mark, –1 on error

This function is an invention of Posix.1g. Posix is replacing all `ioctl`s with functions.

Figure 21.7 (p. 574) shows an implementation of this function using the commonly found `SIOCATMARK` `ioctl`.

oob/tcprecv03.c

```
 1 #include    "unp.h"

 2 int
 3 main(int argc, char **argv)
 4 {
 5     int     listenfd, connfd, n, justreadoob = 0;
 6     char    buff[100];
 7     fd_set  rset, xset;

 8     if (argc == 2)
 9         listenfd = Tcp_listen(NULL, argv[1], NULL);
10     else if (argc == 3)
11         listenfd = Tcp_listen(argv[1], argv[2], NULL);
12     else
13         err_quit("usage: tcprecv03 [ <host> ] <port#>");

14     connfd = Accept(listenfd, NULL, NULL);

15     FD_ZERO(&rset);
16     FD_ZERO(&xset);
17     for ( ; ; ) {
18         FD_SET(connfd, &rset);
19         if (justreadoob == 0)
20             FD_SET(connfd, &xset);

21         Select(connfd + 1, &rset, NULL, &xset, NULL);

22         if (FD_ISSET(connfd, &xset)) {
23             n = Recv(connfd, buff, sizeof(buff) - 1, MSG_OOB);
24             buff[n] = 0;        /* null terminate */
25             printf("read %d OOB byte: %s\n", n, buff);
26             justreadoob = 1;
27             FD_CLR(connfd, &xset);
28         }
29         if (FD_ISSET(connfd, &rset)) {
30             if ( (n = Read(connfd, buff, sizeof(buff) - 1)) == 0) {
31                 printf("received EOF\n");
32                 exit(0);
33             }
34             buff[n] = 0;        /* null terminate */
35             printf("read %d bytes: %s\n", n, buff);
36             justreadoob = 0;
37         }
38     }
39 }
```

oob/tcprecv03.c

Figure 21.6 Modification of Figure 21.5 to `select` for an exception condition correctly.

lib/sockatmark.c

```
1 #include    "unp.h"

2 int
3 sockatmark(int fd)
4 {
5     int     flag;

6     if (ioctl(fd, SIOCATMARK, &flag) < 0)
7         return (-1);
8     return (flag != 0 ? 1 : 0);
9 }
```

lib/sockatmark.c

Figure 21.7 `sockatmark` function implemented using `ioctl`.

The out-of-band mark applies regardless whether the receiving process is receiving the out-of-band data inline (the `SO_OOBINLINE` socket option) or out-of-band (the `MSG_OOB` flag). One common use of the out-of-band mark is for the receiver to treat all of the data specially until the mark is passed.

Example

We now show a simple example to illustrate the following two features of the out-of-band mark.

1. The out-of-band mark always points one beyond the final byte of normal data. This means that, if the out-of-band data is received inline, `sockatmark` returns true if the next byte to be read is the byte that was sent with the `MSG_OOB` flag. Alternately, if the `SO_OOBINLINE` socket option is not enabled, then `sockatmark` returns true if the next byte of data is the first byte that was sent following the out-of-band data.
2. A read operation always stops at the out-of-band mark (pp. 519–520 of TCPv2). That is, if there are 100 bytes in the socket receive buffer but only 5 bytes until the out-of-band mark, and the process performs a `read` asking for 100 bytes, only the 5 bytes up until the mark are returned. This forced stop at the mark is to allow the process to call `sockatmark` to determine if the buffer pointer is at the mark.

Figure 21.8 is our sending program. It sends 3 bytes of normal data, 1 byte of out-of-band data, followed by another byte of normal data. There are no pauses between each output operation.

Figure 21.9 is the receiving program. This program does not use the `SIGURG` signal or `select`. Instead, it calls `sockatmark` to determine when the out-of-band byte is encountered.

oob/tcpsend04.c

```
1 #include    "unp.h"
2 int
3 main(int argc, char **argv)
4 {
5     int     sockfd;
6     if (argc != 3)
7         err_quit("usage: tcpsend04 <host> <port#>");
8     sockfd = Tcp_connect(argv[1], argv[2]);
9     Write(sockfd, "123", 3);
10    printf("wrote 3 bytes of normal data\n");
11    Send(sockfd, "4", 1, MSG_OOB);
12    printf("wrote 1 byte of OOB data\n");
13    Write(sockfd, "5", 1);
14    printf("wrote 1 byte of normal data\n");
15    exit(0);
16 }
```

oob/tcpsend04.c

Figure 21.8 Sending program.

oob/tcprecv04.c

```
1 #include    "unp.h"
2 int
3 main(int argc, char **argv)
4 {
5     int     listenfd, connfd, n, on = 1;
6     char    buff[100];
7     if (argc == 2)
8         listenfd = Tcp_listen(NULL, argv[1], NULL);
9     else if (argc == 3)
10        listenfd = Tcp_listen(argv[1], argv[2], NULL);
11    else
12        err_quit("usage: tcprecv04 [ <host> ] <port#>");
13    Setsockopt(listenfd, SOL_SOCKET, SO_OOBINLINE, &on, sizeof(on));
14    connfd = Accept(listenfd, NULL, NULL);
15    sleep(5);
16    for ( ; ; ) {
17        if (Sockatmark(connfd))
18            printf("at OOB mark\n");
19        if ( (n = Read(connfd, buff, sizeof(buff) - 1)) == 0) {
20            printf("received EOF\n");
21            exit(0);
22        }
23        buff[n] = 0;            /* null terminate */
24        printf("read %d bytes: %s\n", n, buff);
25    }
26 }
```

oob/tcprecv04.c

Figure 21.9 Receiving program that calls `sockatmark`.

Set SO_OOBINLINE socket option

13 We want to receive the out-of-band data inline, so we must set the SO_OOBINLINE socket option. But if we wait until accept returns and set the option on the connected socket, the three-way handshake is complete and out-of-band data may have already arrived. Therefore we must set this option for the listening socket, knowing that all socket options carry over from the listening socket to the connected socket (Section 7.4).

sleep after connection accepted

14-15 The receiver sleeps after the connection is accepted to let all the data from the sender be received. This allows us to demonstrate that a read stops at the out-of-band mark, even though additional data is in the socket receive buffer.

Read all the data from the sender

16-25 The program calls read in a loop, printing the received data. But before calling read, sockatmark checks if the buffer pointer is at the out-of-band mark.

When we run this program we have the following output:

```
bsdi % tcprecv04 6666
read 3 bytes: 123
at OOB mark
read 2 bytes: 45
received EOF
```

Even though all the data has been received by the receiving TCP when read is called the first time (because the receiving process calls sleep), only 3 bytes are returned because the mark is encountered. The next byte read is the out-of-band byte (with a value of 4) because we told the kernel to place the out-of-band data inline.

Example

We now show another simple example to illustrate two additional features of out-of-band data, both of which we mentioned earlier.

1. TCP sends notification of out-of-band data (its urgent pointer) even though it is flow-control stopped from sending data.
2. A receiving process can be notified that the sender has sent out-of-band data (with the SIGURG signal or by select) *before* the out-of-band data arrives. If the process then calls recv specifying MSG_OOB and the out-of-band data has not arrived, an error of EWOULDBLOCK is returned.

Figure 21.10 is the sending program.

9-19 This process sets the size of its socket send buffer to 32768, writes 16384 bytes of normal data, and then sleeps for 5 seconds. We will see shortly that the receiver sets the size of its socket receive buffer to 4096, so these operations by the sender guarantee that the sending TCP fills the receiver's socket receive buffer. The sender then sends 1 byte of out-of-band data, followed by 1024 bytes of normal data, and terminates.

oob/tcpsend05.c

```
 1 #include    "unp.h"
 2 int
 3 main(int argc, char **argv)
 4 {
 5     int     sockfd, size;
 6     char    buff[16384];
 7     if (argc != 3)
 8         err_quit("usage: tcpsend04 <host> <port#>");
 9     sockfd = Tcp_connect(argv[1], argv[2]);
10     size = 32768;
11     Setsockopt(sockfd, SOL_SOCKET, SO_SNDBUF, &size, sizeof(size));
12     Write(sockfd, buff, 16384);
13     printf("wrote 16384 bytes of normal data\n");
14     sleep(5);
15     Send(sockfd, "a", 1, MSG_OOB);
16     printf("wrote 1 byte of OOB data\n");
17     Write(sockfd, buff, 1024);
18     printf("wrote 1024 bytes of normal data\n");
19     exit(0);
20 }
```

oob/tcpsend05.c

Figure 21.10 Sending program.

Figure 21.11 shows the receiving program.

14–20 The receiving process sets the size of the listening socket's receive buffer to 4096. This size will carry over to the connected socket after the connection is established. The process then `accepts` the connection, establishes a signal handler for `SIGURG`, and establishes the owner of the socket. The main loop calls `pause` in an infinite loop.

22–31 The signal handler calls `recv` to read the out-of-band data.

When we start the receiver and then the sender, here is the output from the sender.

```
solaris % tcpsend05 bsdi 5555
wrote 16384 bytes of normal data
wrote 1 byte of OOB data
wrote 1024 bytes of normal data
```

As expected, all the data fits into the sender's socket send buffer, and it terminates. Here is the output from the receiver.

```
bsdi % tcprecv05 5555
SIGURG received
recv error: Resource temporarily unavailable
```

The error string printed by our `err_sys` function corresponds to `EAGAIN`, which is the same as `EWOULDBLOCK` in BSD/OS. TCP sends the out-of-band notification to the receiving TCP, which then generates the `SIGURG` signal for the receiving process. But when `recv` is called specifying the `MSG_OOB` flag, the out-of-band byte cannot be read.

oob/tcprecv05.c

```
 1 #include    "unp.h"
 2 int     listenfd, connfd;
 3 void    sig_urg(int);
 4 int
 5 main(int argc, char **argv)
 6 {
 7     int     size;
 8     if (argc == 2)
 9         listenfd = Tcp_listen(NULL, argv[1], NULL);
10     else if (argc == 3)
11         listenfd = Tcp_listen(argv[1], argv[2], NULL);
12     else
13         err_quit("usage: tcprecv05 [ <host> ] <port#>");
14     size = 4096;
15     Setsockopt(listenfd, SOL_SOCKET, SO_RCVBUF, &size, sizeof(size));
16     connfd = Accept(listenfd, NULL, NULL);
17     Signal(SIGURG, sig_urg);
18     Fcntl(connfd, F_SETOWN, getpid());
19     for ( ; ; )
20         pause();
21 }
22 void
23 sig_urg(int signo)
24 {
25     int     n;
26     char    buff[2048];
27     printf("SIGURG received\n");
28     n = Recv(connfd, buff, sizeof(buff) - 1, MSG_OOB);
29     buff[n] = 0;                /* null terminate */
30     printf("read %d OOB byte\n", n);
31 }
```

oob/tcprecv05.c

Figure 21.11 Receiving program.

The solution is for the receiver to make room in its socket receive buffer by reading the normal data that is available. This will cause its TCP to advertise a nonzero window to the sender, which will eventually let the sender transmit the out-of-band byte.

> We note two related issues in Berkeley-derived implementations (pp. 1016–1017 of TCPv2). First, even if the socket send buffer is full, an out-of-band byte is always accepted by the kernel from the process for sending to the peer. Second, when the process sends an out-of-band byte, a TCP segment is immediately sent that contains the urgent notification. All the normal TCP output checks (Nagle algorithm, silly-window avoidance, etc.) are bypassed.

Example

Our next example demonstrates that there is only a single out-of-band mark for a given TCP connection, and if new out-of-band data arrives before the receiving process reads some existing out-of-band data, the previous mark is lost.

Figure 21.12 is the sending program, which is similar to Figure 21.8 with the addition of another `send` of out-of-band data, followed by one more `write` of normal data.

oob/tcpsend06.c

```
 1 #include    "unp.h"
 2 int
 3 main(int argc, char **argv)
 4 {
 5     int     sockfd;
 6     if (argc != 3)
 7         err_quit("usage: tcpsend04 <host> <port#>");
 8     sockfd = Tcp_connect(argv[1], argv[2]);
 9     Write(sockfd, "123", 3);
10     printf("wrote 3 bytes of normal data\n");
11     Send(sockfd, "4", 1, MSG_OOB);
12     printf("wrote 1 byte of OOB data\n");
13     Write(sockfd, "5", 1);
14     printf("wrote 1 byte of normal data\n");
15     Send(sockfd, "6", 1, MSG_OOB);
16     printf("wrote 1 byte of OOB data\n");
17     Write(sockfd, "7", 1);
18     printf("wrote 1 byte of normal data\n");
19     exit(0);
20 }
```

oob/tcpsend06.c

Figure 21.12 Sending two out-of-band bytes in rapid succession.

There are no pauses in the sending, allowing all the data to be sent to the receiving TCP quickly.

The receiving program is identical to Figure 21.9, which `sleeps` for 5 seconds after accepting the connection to allow the data to arrive at its TCP. Here is the receiving program's output:

```
bsdi % tcprecv06 5555
read 5 bytes: 12345
at OOB mark
read 2 bytes: 67
received EOF
```

The arrival of the second out-of-band byte (the 6) overwrites the mark that was stored when the first out-of-band byte arrived (the 4). As we said, there is only one out-of-band mark per TCP connection.

21.4 TCP Out-of-Band Data Summary

All our examples using out-of-band data so far have been trivial. Unfortunately out-of-band data gets messy when we consider the timing problems that may arise. The first point to consider is that the concept of out-of-band data really conveys three different pieces of information to the receiver.

1. The fact that the sender went into urgent mode. The receiving process can be notified of this with either the `SIGURG` signal or with `select`. This *notification* is transmitted immediately after the sender sends the out-of-band byte, because we saw in Figure 21.11 that TCP sends the notification even if it is flow-control stopped from sending any data to the receiver. This notification might cause the receiver to go into some special mode of processing for any subsequent data that it receives.
2. The *position* of the out-of-band byte; that is, where it was sent with regard to the rest of data from the sender: the out-of-band mark.
3. The actual *value* of the out-of-band byte. Since TCP is a byte-stream protocol that does not interpret the data sent by the application, this can be any 8-bit value.

With TCP's urgent mode we can think of the URG flag as being the notification, the urgent pointer as being the mark, and the byte of data as itself.

The problems with this concept of out-of-band data are that (a) there is only one TCP urgent pointer per connection, (b) there is only one out-of-band mark per connection, and (c) there is only a single 1-byte out-of-band buffer per connection (which is an issue only if the data is not being read inline). We saw with Figure 21.12 that an arriving mark overwrites any previous mark that the process has not yet encountered. If the data is being read inline, previous out-of-band bytes are not lost when new out-of-band data arrives, but the mark is lost.

One common use of out-of-band data is with Rlogin, when the client interrupts the program that it is running on the server (pp. 393–394 of TCPv1). The server needs to tell the client to discard all queued output because up to one window's worth of output may be queued to send from the server to the client. The server sends a special byte to the client, telling it to flush all output, and this byte is sent as out-of-band data. When the client receives the `SIGURG` signal, it just reads from the socket until it encounters the mark, discarding everything up through the mark. (Pp. 398–401 of TCPv1 contain an example of this use of out-of-band data, along with the corresponding `tcpdump` output.) In this scenario, if the server were to send multiple out-of-band bytes in quick succession, it doesn't affect the client, as the client just reads up through the final mark, discarding all the data.

In summary, the usefulness of out-of-band data depends on why it is being used by the application. If the purpose is to tell the peer to discard the normal data, up through the mark, then losing an intermediate out-of-band byte and its corresponding mark is of no consequence. But if it is important that no out-of-band bytes be lost, then the data must be received inline. Furthermore, the data bytes that are sent as out-of-band data

should be differentiated from normal data since intermediate marks can be overwritten when a new mark is received, effectively mixing the out-of-band bytes with the normal data. Telnet, for example, sends its own commands in the normal stream of data between the client and server, prefixing its commands with a byte of 255. (To send this value as data then requires two successive bytes of 255.) This lets it differentiate its commands from the normal user data but requires that the client and server process each byte of data looking for commands.

21.5 Client–Server Heartbeat Functions

We now develop some simple heartbeat functions for our echo client and server. These functions can detect the early failure of either the peer host or the communications path to the peer.

Before showing these functions we must provide some caveats. First, some people want to use the TCP keepalive feature (the `SO_KEEPALIVE` socket option) to provide this functionality, but TCP doesn't send a keepalive probe until the connection has been idle for 2 hours. When people discover this, their next question is how to modify the keepalive parameters down to a much lower value (often on the order of seconds) to detect a failure faster. While it is possible on many systems to shorten TCP's keepalive timer parameters (see Appendix E of TCPv1), these parameters are normally maintained on a per-kernel basis, not a per-socket basis, so changing them affects all sockets that enable this option. Also, the keepalive option was never intended for this purpose (high-frequency polling).

Second, a temporary loss in connectivity between two end systems is not always a bad thing. TCP was designed to cope with this, and Berkeley-derived TCP implementations will retransmit for 8–10 minutes before giving up on a connection. Newer IP routing protocols (e.g., OSPF) can detect the failure of a link and possibly come up with an alternate path in a short time (e.g., on the order of seconds). Therefore one must examine their application to determine whether terminating a connection after not hearing from the peer after 5 or 10 seconds is a good thing or not. Some applications require this type of functionality, but most do not.

We will use TCP's urgent mode to poll the peer on a regular basis; we assume once a second in the description below, along with a 5-second limit, but these can be changed by the application. Figure 21.13 shows the arrangement of the client and server.

In this example the client sends an out-of-band byte to the server once a second, and the receipt of this by the server causes it to send an out-of-band byte back to the client. Each needs to know if the other disappears or is unreachable. The client and server increment their `cnt` variable once a second and the receipt of an out-of-band byte resets this variable to 0. Should the counter reach 5 (that is, the process has not received an out-of-band byte from its peer in 5 seconds), a failure is assumed. Both client and server use the `SIGURG` signal to be notified when the out-of-band byte arrives. We note in the middle of this figure that the data, echoed data, and the out-of-band bytes are all exchanged across a single TCP connection.

Our client `main` function is unchanged from Figure 5.4. Our `str_cli` function (which we do not show) has only three simple changes from the version in Figure 6.13.

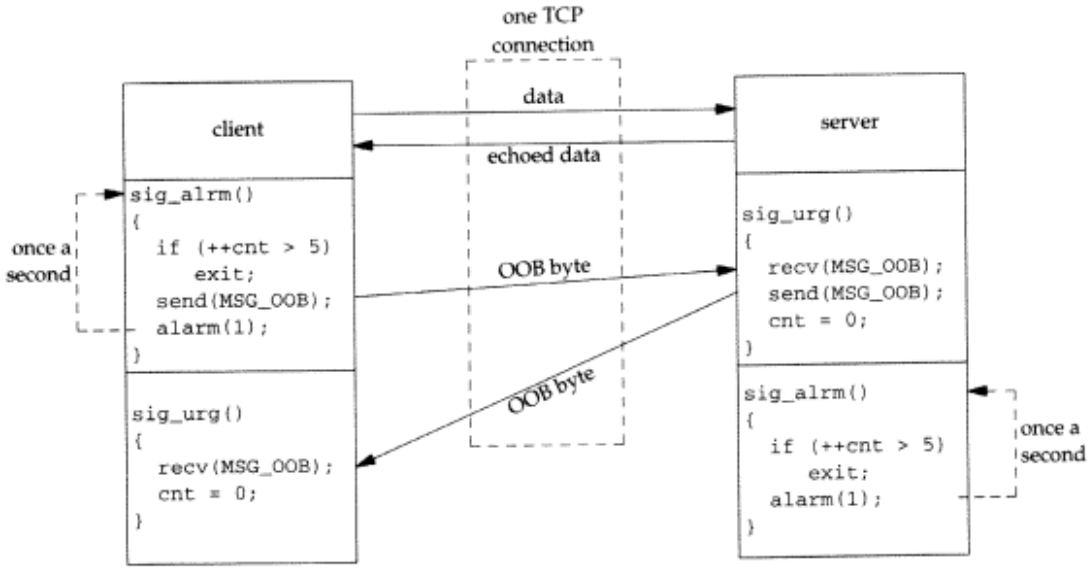


Figure 21.13 Client–server heartbeat using out-of-band data.

1. We call our `heartbeat_cli` function before entering the `for` loop, to set up the client heartbeat feature:

```
heartbeat_cli(sockfd, 1, 5);
```

   The second argument is the polling frequency in seconds, and the third argument is the number of seconds with no response before giving up on the connection.

2. If `select` returns an error of `EINTR`, we `continue` to go around the loop and call `select` again. Note in Figure 21.13 that the client now catches two signals, `SIGALRM` and `SIGURG`, so we must be prepared to handle an interrupted system call.

3. Instead of calling `fputs` to write the echoed line to standard output we call `writen`. We do this because we are catching two signals, which we just said can interrupt slow system calls, and some versions of the standard I/O library do not handle interrupted system calls correctly [Korn and Vo 1991].

Figure 21.14 shows the three functions that provide the client heartbeat functionality.

Global variables

2–5 The first three variables are copies of the arguments to `heartbeat_cli`: the socket descriptor (which the signal handlers need to send and receive the out-of-band data), the frequency of the `SIGALRM`s, and the total number of `SIGALRM`s with no response from the server before the client considers the server or the connection dead. The variable `nprobes` counts the number of `SIGALRM`s since the last reply from the server.

oob/heartbeatcli.c

```
1 #include    "unp.h"

2 static int servfd;
3 static int nsec;                   /* #seconds betweeen each alarm */
4 static int maxnprobes;             /* #probes w/no response before quit */
5 static int nprobes;                /* #probes since last server response */
6 static void sig_urg(int), sig_alrm(int);

7 void
8 heartbeat_cli(int servfd_arg, int nsec_arg, int maxnprobes_arg)
9 {
10      servfd = servfd_arg;        /* set globals for signal handlers */
11      if ( (nsec = nsec_arg) < 1)
12          nsec = 1;
13      if ( (maxnprobes = maxnprobes_arg) < nsec)
14          maxnprobes = nsec;
15      nprobes = 0;

16      Signal(SIGURG, sig_urg);
17      Fcntl(servfd, F_SETOWN, getpid());

18      Signal(SIGALRM, sig_alrm);
19      alarm(nsec);
20 }

21 static void
22 sig_urg(int signo)
23 {
24      int     n;
25      char    c;

26      if ( (n = recv(servfd, &c, 1, MSG_OOB)) < 0) {
27          if (errno != EWOULDBLOCK)
28              err_sys("recv error");
29      }
30      nprobes = 0;                /* reset counter */
31      return;                     /* may interrupt client code */
32 }

33 static void
34 sig_alrm(int signo)
35 {
36      if (++nprobes > maxnprobes) {
37          fprintf(stderr, "server is unreachable\n");
38          exit(0);
39      }
40      Send(servfd, "1", 1, MSG_OOB);
41      alarm(nsec);
42      return;                     /* may interrupt client code */
43 }
```

oob/heartbeatcli.c

Figure 21.14 Client heartbeat functions.

heartbeat_cli function

7-20 The heartbeat_cli function validates and saves the arguments. Signal handlers are established for SIGURG and SIGALRM and the owner of the socket is set to the process ID. alarm schedules the first SIGALRM.

SIGURG handler

21-32 This signal is generated when an out-of-band notification arrives. We try to read the out-of-band byte, but if it has not arrived (EWOULDBLOCK) that is OK. Notice that we are not receiving the out-of-band data inline, which would interfere with the client's reading of its normal data.

Since the server is still alive, nprobes is reset to 0.

SIGALRM handler

33-43 This signal is generated at a regular interval. The counter nprobes is incremented, and if it has reached maxnprobes, we assume the server host has either crashed or is unreachable. In this example we terminate the client process, although other designs could be used: a signal could be sent to the main loop, or another argument to heartbeat_cli could be a client function that is called when it appears that the server is dead.

A byte containing the character 1 is sent as out-of-band data (there is no meaning implied by this value) and alarm schedules the next SIGALRM.

Our server main function is identical to the one in Figure 5.12. The only modification to our str_echo function from Figure 5.3 is to add the line

```
heartbeat_serv(sockfd, 1, 5);
```

before the for loop. This call initializes the heartbeat function for the server.

Figure 21.15 shows the server heartbeat functions.

heartbeat_serv function

7-19 The variable declarations and the function heartbeat_serv are nearly identical to those for the client.

SIGURG handler

20-32 When an out-of-band notification is received by the server, it tries to read the byte. As with the client, if the out-of-band byte has not arrived, that is OK. The out-of-band byte is echoed back to the client as out-of-band data. Notice that if recv returns EWOULDBLOCK, then whatever happens to be in the automatic variable c is echoed back to the client. We do not use the value of the out-of-band byte for any purpose, so this is OK. All that is important is to send 1 byte of out-of-band data, whatever that byte happens to be. nprobes is reset to 0 since we just received notification that the client is alive.

SIGALRM handler

33-42 nprobes is incremented and if it has reached the caller-specified value of maxnalarms, the server process is terminated. Otherwise the next SIGALRM is scheduled.

oob/heartbeatserv.c

```
 1 #include    "unp.h"

 2 static int servfd;
 3 static int nsec;                /* #seconds between each alarm */
 4 static int maxnalarms;          /* #alarms w/no client probe before quit */
 5 static int nprobes;             /* #alarms since last client probe */
 6 static void sig_urg(int), sig_alrm(int);

 7 void
 8 heartbeat_serv(int servfd_arg, int nsec_arg, int maxnalarms_arg)
 9 {
10     servfd = servfd_arg;        /* set globals for signal handlers */
11     if ( (nsec = nsec_arg) < 1)
12         nsec = 1;
13     if ( (maxnalarms = maxnalarms_arg) < nsec)
14         maxnalarms = nsec;

15     Signal(SIGURG, sig_urg);
16     Fcntl(servfd, F_SETOWN, getpid());

17     Signal(SIGALRM, sig_alrm);
18     alarm(nsec);
19 }

20 static void
21 sig_urg(int signo)
22 {
23     int     n;
24     char    c;

25     if ( (n = recv(servfd, &c, 1, MSG_OOB)) < 0) {
26         if (errno != EWOULDBLOCK)
27             err_sys("recv error");
28     }
29     Send(servfd, &c, 1, MSG_OOB);   /* echo back out-of-band byte */

30     nprobes = 0;                /* reset counter */
31     return;                     /* may interrupt server code */
32 }

33 static void
34 sig_alrm(int signo)
35 {
36     if (++nprobes > maxnalarms) {
37         printf("no probes from client\n");
38         exit(0);
39     }
40     alarm(nsec);
41     return;                     /* may interrupt server code */
42 }
```

oob/heartbeatserv.c

Figure 21.15 Server heartbeat functions.

21.6 Summary

TCP does not have true out-of-band data. It provides an urgent pointer that is sent in the TCP header to the peer as soon as the sender goes into urgent mode. The receipt of this pointer by the other end of the connection tells that process that the sender has gone into urgent mode, and the pointer points to the final byte of urgent data. But all the data is still sent subject to TCP's normal flow control.

The sockets API maps TCP's urgent mode into what it calls out-of-band data. The sender goes into urgent mode by specifying the `MSG_OOB` flag in a call to `send`. The final byte of data in this call is considered the out-of-band byte. The receiver is notified when its TCP receives a new urgent pointer by either the `SIGURG` signal, or by `select` indicating that the socket has an exception condition pending. By default TCP takes the out-of-band byte out of the normal stream of data and places it into its own 1-byte out-of-band buffer that the process reads by calling `recv` with the `MSG_OOB` flag. Alternately the receiver can set the `SO_OOBINLINE` socket option, in which case the out-of-band byte is left in the normal stream of data. Regardless of which method is used by the receiver, the socket layer maintains an out-of-band mark in the data stream and will not read through the mark with a single input operation. The receiver determines if it has reached the mark by calling the `sockatmark` function.

Out-of-band data is not heavily used. Telnet and Rlogin use it, as does FTP, but FTP's use is only because early implementations did not provide an I/O multiplexing feature. Out-of-band data was designed at a time when resources were scarce (i.e., processor memory and CPU time). When designing new applications that need a second, high-priority, non-flow-controlled channel between the peers, we should consider a second TCP connection instead of using out-of-band data.

Exercises

21.1 Is there a difference between the single function call

```
send(fd, "ab", 2, MSG_OOB);
```

and the two function calls

```
send(fd, "a", 1, MSG_OOB);
send(fd, "b", 1, MSG_OOB);
```

21.2 Redo Figure 21.6 to use `poll` instead of `select`.

21.3 Modify the `sig_alrm` function in Figures 21.14 and 21.15 to `write` a message to `STDERR_FILENO` each time it is called and the `nprobes` counter is already greater than 0 (e.g., the previous probe was lost). Run the client and server on two hosts on the same LAN and see how often the message is printed. Redirect the client's standard input to a large text file and redirect its standard output to a temporary file. Run the client again, and compare the output file with the input file to make certain no data was lost. Is the message printed more often when lots of data is exchanged? Now run the client and server on two hosts across a WAN and compare the results with the LAN.

21.4 Rewrite the client–server heartbeat functions to use a second TCP connection, instead of using urgent data. Run the same tests from the previous exercise and compare the results.

22

Signal-Driven I/O

22.1 Introduction

Signal-driven I/O is when we tell the kernel to notify us with a signal when something happens on a descriptor. Historically this has been called *asynchronous I/O*, but the signal-driven I/O that we describe is not true asynchronous I/O. The latter is normally defined as the process performing the I/O operation (say a read or write), with the kernel returning immediately after the kernel initiates the I/O operation. The process continues executing while the I/O takes place. Some form of notification is then provided to the process when the operation is complete or encounters an error. We compared the various types of I/O that are normally available in Section 6.2 and showed the difference between signal-driven I/O and asynchronous I/O.

Notice that the nonblocking I/O that we described in Chapter 15 is not asynchronous I/O either. With nonblocking I/O the kernel does not return after initiating the I/O operation; the kernel returns immediately only if the operation cannot be completed without putting the process to sleep.

> Posix.1 provides true asynchronous I/O with its `aio_XXX` functions. These functions let the process specify whether or not a signal is generated when the I/O completes, and which signal to generate.

Berkeley-derived implementations support signal-driven I/O for sockets and terminal devices using the `SIGIO` signal. SVR4 supports signal-driven I/O for streams devices using the `SIGPOLL` signal, which is then equated to `SIGIO`.

22.2 Signal-Driven I/O for Sockets

To use signal-driven I/O with a socket (`SIGIO`) requires the process to perform the following three steps:

1. A signal handler must be established for the `SIGIO` signal.
2. The socket owner must be set, normally with the `F_SETOWN` command of `fcntl` (Figure 7.15).
3. Signal-driven I/O must be enabled for the socket, normally with the `F_SETFL` command of `fcntl` to turn on the `O_ASYNC` flag (Figure 7.15).

> The `O_ASYNC` flag is new with Posix.1g. None of the systems in Figure 1.16 support the flag. In Figure 22.4 we enable signal-driven I/O with the `FIOASYNC` `ioctl` instead. Notice the bad choice of names by Posix.1g: the name `O_SIGIO` would have been a better choice for the new flag.
>
> We should establish the signal handler *before* setting the owner of the socket. Under Berkeley-derived implementations the order of the two function calls does not matter, because the default action of `SIGIO` is to be ignored. Therefore if we were to reverse the order of the two function calls, there is a small chance that a signal could be generated after the call to `fcntl` but before the call to `signal`, but if that happens the signal is just discarded. Under SVR4, however, `SIGIO` is defined to be `SIGPOLL` in the `<sys/signal.h>` header and the default action of `SIGPOLL` is to terminate the process. Therefore under SVR4 we want to be certain the signal handler is installed before setting the owner of the socket.

Although setting a socket for signal-driven I/O is easy, the hard part is determining what conditions cause `SIGIO` to be generated for the socket owner. This depends on the underlying protocol.

`SIGIO` with UDP Sockets

Using signal-driven I/O with UDP is simple. The signal is generated whenever

- a datagram arrives for the socket, or
- an asynchronous error occurs on the socket.

Hence, when we catch `SIGIO` for a UDP socket, we call `recvfrom` to either read the datagram that arrived or to obtain the asynchronous error. We talked about asynchronous errors with regard to UDP sockets in Section 8.9. Recall that these are generated only if the UDP socket is connected.

> `SIGIO` is generated for these two conditions by the calls to `sorwakeup` on pp. 775, 779, and 784 of TCPv2.

`SIGIO` with TCP Sockets

Unfortunately signal-driven I/O is next to useless with a TCP socket. The problem is that the signal is generated too often, and the occurrence of the signal does not tell us

what happened. As noted on p. 439 of TCPv2, the following conditions all cause SIGIO to be generated for a TCP socket (assuming signal-driven I/O is enabled):

- a connection request has completed on a listening socket,
- a disconnect request has been initiated,
- a disconnect request has completed,
- half of a connection has been shut down,
- data has arrived on a socket,
- data has been sent from a socket (i.e., the output buffer has free space), or
- an asynchronous error occurred.

For example, if one is both reading from and writing to a TCP socket, SIGIO is generated when new data arrives and when data previously written is acknowledged, and there is no way to distinguish between the two in the signal handler. If SIGIO is used in this scenario, the TCP socket should be set nonblocking to prevent a read or write from blocking. We should consider using SIGIO only with a listening TCP socket, because the only condition that generates SIGIO for a listening socket is the completion of a new connection.

The only real-world use of signal-driven I/O with sockets that the author was able to find is the NTP (Network Time Protocol) server, which uses UDP. The main loop of the server receives a datagram from a client and sends a response. But there is a non-negligible amount of processing to do for each client's request (more than our trivial echo server). It is important for the server to record accurate timestamps for each received datagram, since that value is returned to the client and then used by the client to calculate the round-trip time to the server. Figure 22.1 shows two ways to build such a UDP server.

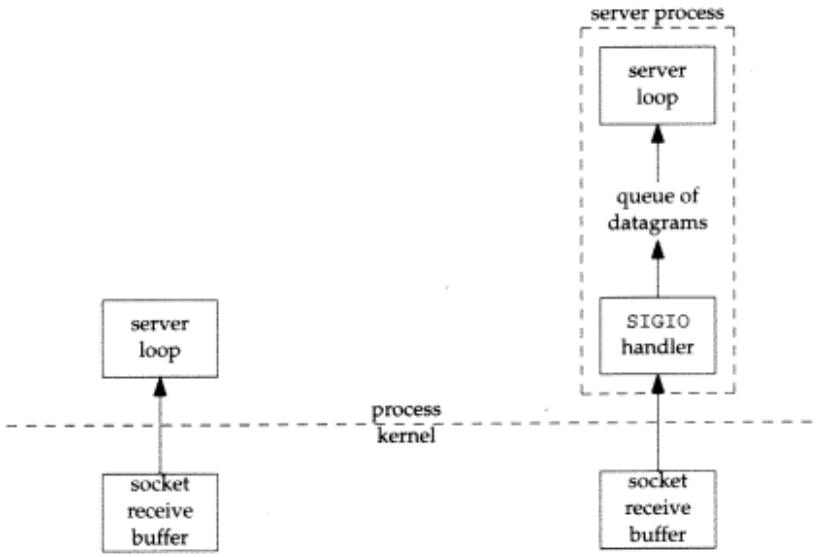


Figure 22.1 Two different ways to build a UDP server.

Most UDP servers (including our echo server from Chapter 8) are designed as shown in the left of this figure. But the NTP server uses the technique shown on the right side: when a new datagram arrives, it is read by the `SIGIO` handler, which also records the time at which the datagram arrived. The datagram is then placed on another queue within the process from which it will be removed by and processed by the main server loop. Although this complicates the server code, it provides accurate timestamps of arriving datagrams.

Recall from Figure 20.4 that the process can set the `IP_RECVDSTADDR` socket option to receive the destination address of a received UDP datagram. One could argue that two additional pieces of information that should also be returned for a received UDP datagram are an indication of the received interface (which can differ from the destination address, if the host employs the common weak end system model), and the time at which the datagram arrived.

For IPv6 the `IPV6_PKTINFO` socket option (Section 20.8) returns the received interface. For IPv4, we discussed the `IP_RECVIF` socket option in Section 20.2.

FreeBSD also provides the `SO_TIMESTAMP` socket option, which returns the time at which the datagram was received as ancillary data in a `timeval` structure. Linux provides an `SIOCGSTAMP` `ioctl` that returns a `timeval` structure containing the time at which the datagram was received.

22.3 UDP Echo Server Using `SIGIO`

We now provide an example similar to the right side of Figure 22.1: a UDP server that uses the `SIGIO` signal to receive arriving datagrams. This example also illustrates the use of Posix reliable signals.

We do not change the client at all from Figures 8.7 and 8.8 and the server `main` function does not change from Figure 8.3. The only changes that we make are to the `dg_echo` function, which we show in the next four figures. Figure 22.2 shows the global declarations.

Queue of received datagrams

3–12 The `SIGIO` signal handler places arriving datagrams onto a queue. This queue is an array of `DG` structures that we treat as a circular buffer. Each structure contains a pointer to the received datagram, its length, a pointer to a socket address structure containing the protocol address of the client, and the size of the protocol address. `QSIZE` of these structures are allocated and we will see in Figure 22.4 that the `dg_echo` function calls `malloc` to allocate memory for all the datagrams and socket address structures. We also allocate a diagnostic counter, `cntread`, that we examine shortly. Figure 22.3 shows the array of structures, assuming that the first entry points to a 150-byte datagram and that the length of its associated socket address structure is 16.

Array indexes

13–15 `iget` is the index of the next array entry for the main loop to process and `iput` is the index of the next array entry for the signal handler to store into. `nqueue` is the total number of datagrams on the queue for the main loop to process.

sigio/dgecho01.c

```
1 #include    "unp.h"

2 static int sockfd;

3 #define QSIZE      8            /* size of input queue */
4 #define MAXDG   4096            /* maximum datagram size */

5 typedef struct {
6     void   *dg_data;            /* ptr to actual datagram */
7     size_t  dg_len;             /* length of datagram */
8     struct sockaddr *dg_sa;     /* ptr to sockaddr{} w/client's address */
9     socklen_t dg_salen;         /* length of sockaddr{} */
10 } DG;
11 static DG dg[QSIZE];           /* the queue of datagrams to process */
12 static long cntread[QSIZE + 1]; /* diagnostic counter */

13 static int iget;               /* next one for main loop to process */
14 static int iput;               /* next one for signal handler to read into */
15 static int nqueue;             /* #on queue for main loop to process */
16 static socklen_t clilen;       /* max length of sockaddr{} */

17 static void sig_io(int);
18 static void sig_hup(int);
```

sigio/dgecho01.c

Figure 22.2 Global declarations.

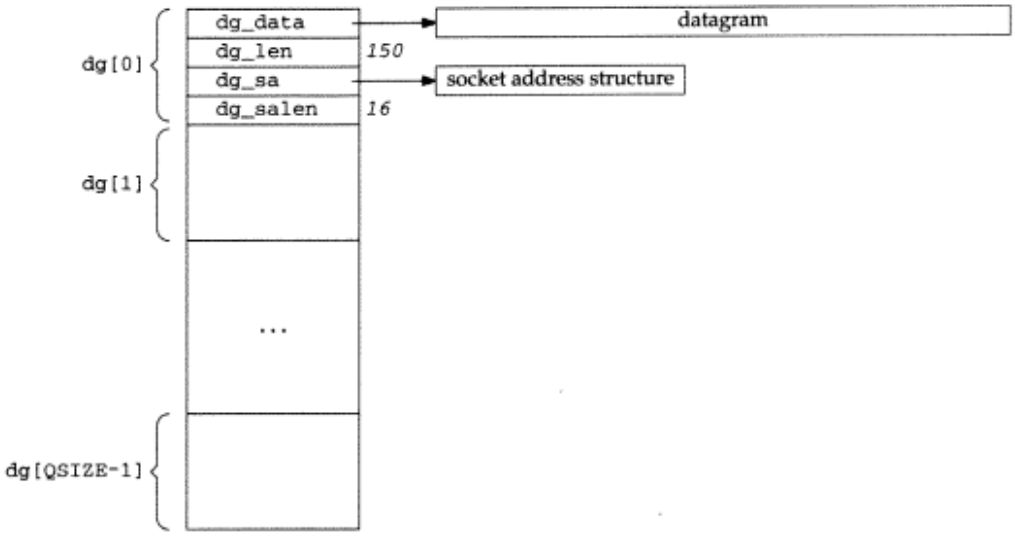


Figure 22.3 Data structures used to hold received datagrams and their socket address structures.

Figure 22.4 shows the main server loop, the dg_echo function.

sigio/dgecho01.c

```
19 void
20 dg_echo(int sockfd_arg, SA *pcliaddr, socklen_t clilen_arg)
21 {
22     int     i;
23     const int on = 1;
24     sigset_t zeromask, newmask, oldmask;

25     sockfd = sockfd_arg;
26     clilen = clilen_arg;

27     for (i = 0; i < QSIZE; i++) {   /* init queue of buffers */
28         dg[i].dg_data = Malloc(MAXDG);
29         dg[i].dg_sa = Malloc(clilen);
30         dg[i].dg_salen = clilen;
31     }
32     iget = iput = nqueue = 0;

33     Signal(SIGHUP, sig_hup);
34     Signal(SIGIO, sig_io);
35     Fcntl(sockfd, F_SETOWN, getpid());
36     Ioctl(sockfd, FIOASYNC, &on);
37     Ioctl(sockfd, FIONBIO, &on);

38     Sigemptyset(&zeromask);      /* init three signal sets */
39     Sigemptyset(&oldmask);
40     Sigemptyset(&newmask);
41     Sigaddset(&newmask, SIGIO); /* the signal we want to block */

42     Sigprocmask(SIG_BLOCK, &newmask, &oldmask);
43     for ( ; ; ) {
44         while (nqueue == 0)
45             sigsuspend(&zeromask);  /* wait for a datagram to process */

46             /* unblock SIGIO */
47         Sigprocmask(SIG_SETMASK, &oldmask, NULL);

48         Sendto(sockfd, dg[iget].dg_data, dg[iget].dg_len, 0,
49                dg[iget].dg_sa, dg[iget].dg_salen);

50         if (++iget >= QSIZE)
51             iget = 0;

52             /* block SIGIO */
53         Sigprocmask(SIG_BLOCK, &newmask, &oldmask);
54         nqueue--;
55     }
56 }
```

sigio/dgecho01.c

Figure 22.4 dg_echo function: server main processing loop.

Initialize queue of received datagrams

27–32 The socket descriptor is saved in a global since the signal handler needs it. The queue of received datagrams is initialized.

Establish signal handlers and set socket flags

33-37 Signal handlers are established for SIGHUP (which we use for diagnostic purposes) and SIGIO. The socket owner is set using fcntl and the signal-driven and nonblocking I/O flags are set using ioctl.

> We mentioned earlier that the O_ASYNC flag with fcntl is the Posix.1g way to specify signal-driven I/O, but since most systems do not yet support it, we use ioctl instead. While most systems do support the O_NONBLOCK flag to set nonblocking, we show the ioctl method here.

Initialize signal sets

38-41 Three signal sets are initialized: zeromask (which never changes), oldmask (which contains the old signal mask when we block SIGIO), and newmask. sigaddset turns on the bit corresponding to SIGIO in newmask.

Block SIGIO and wait for something to do

42-45 sigprocmask stores the current signal mask of the process in oldmask and then logically ORs newmask into the current signal mask. This blocks SIGIO and returns the current signal mask. We then enter the for loop and test the nqueue counter. As long as this counter is 0, there is nothing to do and we call sigsuspend. This Posix function saves the current signal mask internally and then sets the current signal mask to the argument (zeromask). Since zeromask is an empty signal set, this enables all signals. sigsuspend returns after a signal has been caught and the signal handler returns. (It is an unusual function because it *always* returns an error, EINTR.) But before returning, sigsuspend always sets the signal mask to its value when the function was called, which in this case is the value of newmask, so we are guaranteed that when sigsuspend returns, SIGIO is blocked. That is why we can test the counter nqueue, knowing that while we are testing it, a SIGIO signal cannot be delivered.

> Consider what would happen if SIGIO were not blocked while we tested the variable nqueue, which is shared between the main loop and the signal handler. We could test nqueue and find it 0, but immediately after this test, the signal is delivered and nqueue gets set to 1. We then call sigsuspend and go to sleep, effectively missing the signal. We are never awakened from the call to sigsuspend unless another signal occurs. This is similar to the race condition we described in Section 18.5.

Unblock SIGIO and send reply

46-51 We unblock SIGIO by calling sigprocmask to set the signal mask of the process to the value that was saved earlier (oldmask). The reply is then sent by sendto. The iget index is incremented and if its value is the number of elements in the array, its value is set back to 0. We treat the array as a circular buffer. Notice that we do not need SIGIO blocked while modifying iget, because this index is used only by the main loop; it is never modified by the signal handler.

Block SIGIO

52-54 SIGIO is blocked and the value of nqueue is decremented. We must block the signal while modifying this variable, since it is shared between the main loop and the signal handler. Also, we need SIGIO blocked when we test nqueue at the top of the loop.

An alternate technique is to remove both calls to `sigprocmask` that are within the `for` loop, which avoids unblocking the signal and then blocking it later. The problem, however, is that this executes the entire loop with the signal blocked, which decreases the responsiveness of the signal handler. Datagrams should not get lost because of this change (assuming the socket receive buffer is large enough), but the delivery of the signal to the process will be delayed the entire time that the signal is blocked. One goal when coding applications that perform signal handling should be to block the signal for the minimum amount of time.

Figure 22.5 shows the `SIGIO` signal handler.

sigio/dgecho01.c

```
57 static void
58 sig_io(int signo)
59 {
60     ssize_t len;
61     int     nread;
62     DG     *ptr;

63     for (nread = 0;;) {
64         if (nqueue >= QSIZE)
65             err_quit("receive overflow");

66         ptr = &dg[iput];
67         ptr->dg_salen = clilen;
68         len = recvfrom(sockfd, ptr->dg_data, MAXDG, 0,
69                        ptr->dg_sa, &ptr->dg_salen);
70         if (len < 0) {
71             if (errno == EWOULDBLOCK)
72                 break;          /* all done; no more queued to read */
73             else
74                 err_sys("recvfrom error");
75         }
76         ptr->dg_len = len;

77         nread++;
78         nqueue++;
79         if (++iput >= QSIZE)
80             iput = 0;

81     }
82     cntread[nread]++;           /* histogram of #datagrams read per signal */
83 }
```

sigio/dgecho01.c

Figure 22.5 `SIGIO` handler.

The problem that we encounter when coding this signal handler is that Posix signals are normally *not* queued. This means that, if we are in the signal handler, which guarantees that the signal is blocked, and the signal occurs two more times, the signal is delivered only one more time.

> Posix.1 provides some realtime signals that *are* queued, but other signals such as `SIGIO` are normally not queued.

Consider the following scenario. A datagram arrives and the signal is delivered. The signal handler reads the datagram and places it onto the queue for the main loop. But while the signal handler is executing, two more datagrams arrive, causing the signal to be generated two more times. But since the signal is blocked, when the signal handler returns, it is called only one more time. The second time the signal handler executes, it reads the second datagram, but the third datagram is left on the socket receive queue. This third datagram will be read only if and when a fourth datagram arrives. When the fourth datagram arrives, it is the third datagram that is read and placed on the queue for the main loop, not the fourth one.

Because signals are not queued, the descriptor that is set for signal-driven I/O is normally set nonblocking also. We then code our SIGIO handler to read in a loop, terminating only when the read returns EWOULDBLOCK.

Check for queue overflow

64-65 If the queue is full, we terminate. There are other ways to handle this (e.g., additional buffers could be allocated) but for our simple example we just terminate.

Read datagram

66-76 recvfrom is called on the nonblocking socket. The array entry indexed by iput is where the datagram is stored. If there are no datagrams to read, break jumps out of the for loop.

Increment counters and index

77-80 nread is a diagnostic counter of the number of datagrams read per signal. nqueue is the number of datagrams for the main loop to process.

82 Before the signal handler returns, it increments the counter corresponding to the number of datagrams read per signal. We print this array in Figure 22.6 when the SIGHUP signal is delivered, as diagnostic information.

The final function (Figure 22.6) is the SIGHUP signal handler, which prints the cntread array. This counts the number of datagrams read per signal.

sigio/dgecho01.c

```
84 static void
85 sig_hup(int signo)
86 {
87     int     i;

88     for (i = 0; i <= QSIZE; i++)
89         printf("cntread[%d] = %ld\n", i, cntread[i]);
90 }
```

sigio/dgecho01.c

Figure 22.6 SIGHUP handler.

To illustrate that signals are not queued and that we must set the socket nonblocking in addition to setting the signal-driven I/O flag, we will run this server with six clients simultaneously. Each client sends 3645 lines for the server to echo and each client is started from a shell script in the background, so that all clients are started at

about the same time. When all the clients have terminated, we send the SIGHUP signal to the server, causing it to print its cntread array:

```
bsdi % udpserv01
cntread[0] = 2
cntread[1] = 21838
cntread[2] = 12
cntread[3] = 1
cntread[4] = 0
cntread[5] = 1
cntread[6] = 0
cntread[7] = 0
cntread[8] = 0
```

Most of the time the signal handler reads only one datagram, but there are times when more than one is ready. The nonzero counter for cntread[0] is when the signal is generated while the signal handler is executing, but before the signal handler returns, it reads all pending datagrams. When the signal handler is called again, there are no datagrams left to read. Finally, we can verify that the weighted sum of the array elements ($21838 \times 1 + 12 \times 2 + 1 \times 3 + 1 \times 5 = 21870$) equals 6 (the number of clients) times 3645 lines per client.

22.4 Summary

Signal driven I/O has the kernel notify us with the SIGIO signal when "something" happens on a socket.

- With a connected TCP socket there are numerous conditions that can cause this notification, making this feature of little use.
- With a listening TCP socket this notification occurs when a new connection is ready to be accepted.
- With UDP this notification means either a datagram arrived or an asynchronous error arrived, and in both cases we call recvfrom.

We modified our UDP echo server to use signal-driven I/O, using a technique similar to that used by NTP: read the datagram as soon as possible after it arrives, to obtain an accurate timestamp for its arrival and then queue it for later processing.

Exercises

22.1 An alternate design for the loop in Figure 22.4 is the following:

```
for ( ; ; ) {
    Sigprocmask(SIG_BLOCK, &newmask, &oldmask);
    while (nqueue == 0)
        sigsuspend(&zeromask);  /* wait for a datagram to process */
    nqueue--;

        /* unblock SIGGIO */
    Sigprocmask(SIG_SETMASK, &oldmask, NULL);

    Sendto(sockfd, dg[iget].dg_data, dg[iget].dg_len, 0,
           dg[iget].dg_sa, dg[iget].dg_salen);

    if (++iget >= QSIZE)
        iget = 0;
}
```

Is this modification OK?

23

Threads

23.1 Introduction

In the traditional Unix model, when a process needs something performed by another entity, it `fork`s a child process and lets the child perform the processing. Most network servers under Unix are written this way, as we have seen in our concurrent server examples: the parent `accept`s the connection, `fork`s a child, and the child handles the client.

While this paradigm has served well for many years, there are problems with `fork`:

- `fork` is expensive. Memory is copied from the parent to the child, all descriptors are duplicated in the child, and so on. Current implementations use a technique called *copy-on-write*, which avoids a copy of the parent's data space to the child until the child needs its own copy, but regardless of this optimization, `fork` is expensive.
- Interprocess communication (IPC) is required to pass information between the parent and child *after* the `fork`. Information from the parent to the child *before* the `fork` is easy, since the child starts with a copy of the parent's data space and with a copy of all the parent's descriptors. But returning information from the child to the parent takes more work.

Threads help with both problems. Threads are sometimes called *lightweight processes* since a thread is "lighter weight" than a process. That is, thread creation can be 10–100 times faster than process creation.

All threads within a process share the same global memory. This makes the sharing of information easy between the threads, but along with this simplicity comes the problem of *synchronization*. But more than just the global variables are shared. All threads within a process share:

- process instructions,
- most data,
- open files (e.g., descriptors),
- signal handlers and signal dispositions,
- current working directory, and
- user and group IDs.

But each thread has its own:

- thread ID,
- set of registers, including program counter and stack pointer,
- stack (for local variables and return addresses),
- `errno`,
- signal mask, and
- priority.

> One analogy is to think of signal handlers as a type of thread as we discussed in Section 11.14. That is, in the traditional Unix model we have the main flow of execution (one thread) and a signal handler (another thread). If the main flow is in the middle of updating a linked list when a signal occurs, and the signal handler also tries to update the linked list, havoc normally results. The main flow and the signal handler share the same global variables, but each has its own stack.

In this text we cover Posix threads, also called *Pthreads*. These were standardized in 1995 as part of the Posix.1c standard and most versions of Unix will support them in the future. We will see that all the Pthread functions begin with `pthread_`. This chapter is an introduction to threads, so that we can use threads in our network programs. For additional details see [Butenhof 1997].

23.2 Basic Thread Functions: Creation and Termination

In this section we cover five basic thread functions and then use these in the next two sections to recode our TCP client–server using threads instead of `fork`.

`pthread_create` Function

When a program is started by `exec`, a single thread is created, called the *initial thread* or *main thread*. Additional threads are created by `pthread_create`.

```
#include <pthread.h>

int pthread_create(pthread_t *tid, const pthread_attr_t *attr,
                   void *(*func)(void *), void *arg);
```

Returns: 0 if OK, positive `Exxx` value on error

Each thread within a process is identified by a *thread ID*, whose datatype is `pthread_t` (often an `unsigned int`). On successful creation of a new thread, its ID is returned through the pointer *tid*.

Each thread has numerous *attributes*: its priority, its initial stack size, whether it should be a daemon thread or not, and so on. When a thread is created, we can specify these attributes by initializing a `pthread_attr_t` variable that overrides the default. We normally take the default, in which case we specify the *attr* argument as a null pointer.

Finally, when we create a thread, we specify a function for it to execute. The thread starts by calling this function and then terminates either explicitly (by calling `pthread_exit`) or implicitly (by letting this function return). The address of the function is specified as the *func* argument, and this function is called with a single pointer argument, *arg*. If we need multiple arguments to the function, we must package them into a structure and then pass the address of this structure as the single argument to the start function.

Notice the declarations of *func* and *arg*. The function takes one argument, a generic pointer (`void *`), and returns a generic pointer (`void *`). This lets us pass one pointer (to anything we want) to the thread, and lets the thread return one pointer (again, to anything we want).

The return value from the Pthread functions is normally 0 if OK or nonzero on an error. But unlike the socket functions, and most system calls, which return –1 on an error and set `errno` to a positive value, the Pthread functions return the positive error indication as the function's return value. For example, if `pthread_create` cannot create a new thread because we have exceeded some system limit on the number of threads, the function return value is `EAGAIN`. The Pthread functions do not set `errno`. The convention of 0 for OK or nonzero for an error is fine, since all the `Exxx` values in `<sys/errno.h>` are positive. A value of 0 is never assigned to one of the `Exxx` names.

`pthread_join` Function

We can wait for a given thread to terminate by calling `pthread_join`. Comparing threads to Unix processes, `pthread_create` is similar to `fork`, and `pthread_join` is similar to `waitpid`.

```
#include <pthread.h>

int pthread_join(pthread_t tid, void **status);
```

Returns: 0 if OK, positive `Exxx` value on error

We must specify the *tid* of the thread that we wish to wait for. Unfortunately, there is no way to wait for any of our threads (similar to `waitpid` with a process ID argument of –1). We return to this problem when we discuss Figure 23.14.

If the *status* pointer is nonnull, the return value from the thread (a pointer to some object) is stored in the location pointed to by *status*.

`pthread_self` Function

Each thread has an ID that identifies it within a given process. The thread ID is returned by `pthread_create` and we saw it was used by `pthread_join`. A thread fetches this value for itself using `pthread_self`.

```
#include <pthread.h>

pthread_t pthread_self(void);
```

Returns: thread ID of calling thread

Comparing threads to Unix processes, `pthread_self` is similar to `getpid`.

`pthread_detach` Function

A thread is either *joinable* (the default) or *detached*. When a joinable thread terminates, its thread ID and exit status are retained until another thread calls `pthread_join`. But a detached thread is like a daemon process: when it terminates all its resources are released and we cannot wait for it to terminate. If one thread needs to know when another thread terminates, it is best to leave the thread as joinable.

The `pthread_detach` function changes the specified thread so that it is detached.

```
#include <pthread.h>

int pthread_detach(pthread_t tid);
```

Returns: 0 if OK, positive Exxx value on error

This function is commonly called by the thread that wants to detach itself, as in

```
pthread_detach(pthread_self());
```

`pthread_exit` Function

One way for a thread to terminate is to call `pthread_exit`.

```
#include <pthread.h>

void pthread_exit(void *status);
```

Does not return to caller

If the thread is not detached, its thread ID and exit status are retained for a later `pthread_join` by some other thread in the calling process.

The pointer *status* must not point to an object that is local to the calling thread, since that object disappears when the thread terminates.

There are two other ways for a thread to terminate.

- The function that started the thread (the third argument to `pthread_create`) can return. Since this function must be declared as returning a `void` pointer, that return value is the exit status of the thread.
- If the `main` function of the process returns or if any thread calls `exit`, the process terminates, including any threads.

23.3 str_cli Function Using Threads

Our first example using threads is to recode the `str_cli` function from Figure 15.10, which uses `fork`, to use threads. Recall that we have provided numerous other versions of this function: the original in Figure 5.5 used a stop-and-wait protocol, which we showed was far from optimal for batch input, Figure 6.13 used blocking I/O and the `select` function, and the version starting with Figure 15.3 used nonblocking I/O. Figure 23.1 shows the design of our threads version.

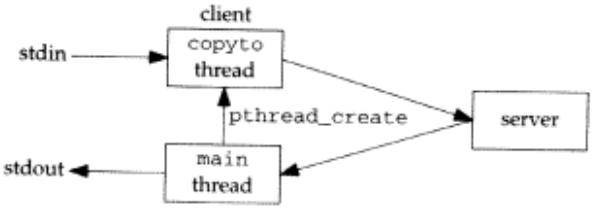


Figure 23.1 Recoding `str_cli` to use threads.

Figure 23.2 shows the `str_cli` function using threads.

unpthread.h header

1 This is the first time we have encountered the `unpthread.h` header. It includes our normal `unp.h` header, followed by the Posix.1 `<pthread.h>` header, and then defines the function prototypes for our wrapper versions of the `pthread_XXX` functions (Section 1.4), which all begin with `Pthread_`.

Save arguments in externals

10-11 The thread that we are about to create needs the values of the two arguments to `str_cli`: `fp`, the standard I/O `FILE` pointer for the input file, and `sockfd`, the TCP socket that is connected to the server. For simplicity we store these two values in external variables. An alternative technique is to put the two values into a structure and then pass a pointer to the structure as the argument to the thread that we are about to create.

Create new thread

12 The thread is created and the new thread ID is saved in `tid`. The function that is executed by the new thread is `copyto`. No arguments are passed to the thread.

threads/strclithread.c

```
 1 #include    "unpthread.h"
 2 void   *copyto(void *);
 3 static int sockfd;              /* global for both threads to access */
 4 static FILE *fp;
 5 void
 6 str_cli(FILE *fp_arg, int sockfd_arg)
 7 {
 8     char    recvline[MAXLINE];
 9     pthread_t tid;
10     sockfd = sockfd_arg;        /* copy arguments to externals */
11     fp = fp_arg;
12     Pthread_create(&tid, NULL, copyto, NULL);
13     while (Readline(sockfd, recvline, MAXLINE) > 0)
14         Fputs(recvline, stdout);
15 }
16 void *
17 copyto(void *arg)
18 {
19     char    sendline[MAXLINE];
20     while (Fgets(sendline, MAXLINE, fp) != NULL)
21         Writen(sockfd, sendline, strlen(sendline));
22     Shutdown(sockfd, SHUT_WR);  /* EOF on stdin, send FIN */
23     return (NULL);
24         /* return (i.e., thread terminates) when end-of-file on stdin */
25 }
```

threads/strclithread.c

Figure 23.2 `str_cli` function using threads.

Main thread loop: copy socket to standard output

13-14 The main thread calls `readline` and `fputs`, copying from the socket to the standard output.

Terminate

15 When the `str_cli` function returns, the `main` function terminates by calling `exit` (Section 5.4). When this happens, *all* threads in the process are terminated. In the normal scenario the other thread has already terminated when it read an end-of-file on standard input. But in case the server terminates prematurely (Section 5.12), calling `exit` terminates the other thread, which is what we want.

`copyto` thread

16-25 This thread just copies from standard input to the socket. When it reads an end-of-file on standard input, a FIN is sent across the socket by `shutdown` and the thread returns. The `return` from this function (which started the thread) terminates the thread.

At the end of Section 15.2 we provided measurements for the five different implementation techniques that we have used with our `str_cli` function. We note that the threads version that we just presented took 8.5 seconds, slightly faster than the version using `fork` (which we expect) but slower than the nonblocking I/O version. Nevertheless, comparing the complexity of the nonblocking I/O version (Section 15.2) versus the simplicity of the threads version, we still recommend using threads instead of nonblocking I/O.

23.4 TCP Echo Server Using Threads

We now redo our TCP echo server from Figure 5.2, using one thread per client, instead of one child process per client. We also make it protocol independent, using our `tcp_listen` function. Figure 23.3 shows the server.

threads/tcpserv01.c

```
 1 #include    "unpthread.h"

 2 static void *doit(void *);      /* each thread executes this function */

 3 int
 4 main(int argc, char **argv)
 5 {
 6     int     listenfd, connfd;
 7     pthread_t tid;
 8     socklen_t addrlen, len;
 9     struct sockaddr *cliaddr;

10     if (argc == 2)
11         listenfd = Tcp_listen(NULL, argv[1], &addrlen);
12     else if (argc == 3)
13         listenfd = Tcp_listen(argv[1], argv[2], &addrlen);
14     else
15         err_quit("usage: tcpserv01 [ <host> ] <service or port>");

16     cliaddr = Malloc(addrlen);

17     for ( ; ; ) {
18         len = addrlen;
19         connfd = Accept(listenfd, cliaddr, &len);
20         Pthread_create(&tid, NULL, &doit, (void *) connfd);
21     }
22 }

23 static void *
24 doit(void *arg)
25 {
26     Pthread_detach(pthread_self());
27     str_echo((int) arg);          /* same function as before */
28     Close((int) arg);             /* we are done with connected socket */
29     return (NULL);
30 }
```

threads/tcpserv01.c

Figure 23.3 TCP echo server using threads (see also Exercise 23.5).

Create a thread

17-21 When `accept` returns, we call `pthread_create` instead of `fork`. The single argument that we pass to the `doit` function is the connected socket descriptor, `connfd`.

> We cast the integer descriptor `connfd` to be a `void` pointer. ANSI C does not guarantee that this works. It works only on systems on which the size of an integer is less than or equal to the size of a pointer. Fortunately most Unix implementations have this property (Figure 1.17). We talk more about this shortly.

Thread function

23-30 `doit` is the function executed by the thread. The thread detaches itself, since there is no reason for the main thread to wait for each thread that it creates. The function `str_echo` does not change from Figure 5.3. When this function returns, we must `close` the connected socket, since the thread shares all descriptors with the main thread. With `fork`, the child did not need to `close` the connected socket because the child then terminated and all open descriptors are closed on process termination. (See Exercise 23.2.)

Also notice that the main thread does not close the connected socket, which we always did with a concurrent server that calls `fork`. This is because all threads within a process share the descriptors, so if the main thread were to call `close`, it would terminate the connection. Creating a new thread does not affect the reference counts for open descriptors, which is different from `fork`.

There is a subtle error in this program, which we describe in detail in Section 23.5. Can you spot the error? (See Exercise 23.5.)

Passing Arguments to New Threads

We mentioned that in Figure 23.3 we cast the integer variable `connfd` to be a `void` pointer, but this is not guaranteed to work on all systems. To handle this correctly requires additional work.

First notice that we cannot just pass the address of `connfd` to the new thread. That is, the following does not work.

```
int
main(int argc, char **argv)
{
    int    listenfd, connfd;
    ...

    for ( ; ; ) {
        len = addrlen;
        connfd = Accept(listenfd, cliaddr, &len);

        Pthread_create(&tid, NULL, &doit, &connfd);
    }
}
```

```
static void *
doit(void *arg)
{
    int  connfd;

    connfd = *((int *) arg);
    Pthread_detach(pthread_self());
    str_echo(connfd);     /* same function as before */
    Close(connfd);        /* we are done with connected socket */
    return(NULL);
}
```

From an ANSI C perspective this is OK: we are guaranteed that we can cast the integer pointer to be a `void *` and then cast this pointer back to an integer pointer. The problem is what this pointer points to.

There is one integer variable `connfd` in the main thread and each call to `accept` overwrites this variable with a new value (the connected descriptor). The following scenario can occur:

- `accept` returns, `connfd` is stored into (say the new descriptor is 5), and the main thread calls `pthread_create`. The pointer to `connfd` (not its contents) is the final argument to `pthread_create`.
- A thread is created and the `doit` function is scheduled to start executing.
- Another connection is ready and the main thread runs again (before the newly created thread). `accept` returns, `connfd` is stored into (say the new descriptor is now 6), and the main thread calls `pthread_create`.

Even though two threads are created, both will operate on the final value stored into `connfd`, which we assume is 6. The problem is that multiple threads are accessing a shared variable (the integer value in `connfd`) with no synchronization. In Figure 23.3 we solved this problem by passing the *value* of `connfd` to `pthread_create`, instead of a pointer to the value. This is fine given the way that C passes integer values to a called function (a copy of the value is pushed onto the stack for the called function).

Figure 23.4 shows a better solution to this problem.

17–22 Each time we call `accept` we first call `malloc` and allocate space for an integer variable, the connected descriptor. This gives each thread its own copy of the connected descriptor.

28–29 The thread fetches the value of the connected descriptor and then calls `free` to release the memory.

Historically the `malloc` and `free` functions have been nonreentrant. That is, calling either function from a signal handler while the main thread is in the middle of one of these two functions has been a recipe for disaster, because of static data structures that are manipulated by these two functions. How can we call these two functions in Figure 23.4? Posix.1 requires that these two functions, along with many others, be *thread-safe*. This is normally done by some form of synchronization performed within the library functions that is transparent to us.

threads/tcpserv02.c

```
 1 #include    "unpthread.h"
 2 static void *doit(void *);      /* each thread executes this function */
 3 int
 4 main(int argc, char **argv)
 5 {
 6     int     listenfd, *iptr;
 7     thread_t tid;
 8     socklen_t addrlen, len;
 9     struct sockaddr *cliaddr;
10     if (argc == 2)
11         listenfd = Tcp_listen(NULL, argv[1], &addrlen);
12     else if (argc == 3)
13         listenfd = Tcp_listen(argv[1], argv[2], &addrlen);
14     else
15         err_quit("usage: tcpserv01 [ <host> ] <service or port>");
16     cliaddr = Malloc(addrlen);
17     for ( ; ; ) {
18         len = addrlen;
19         iptr = Malloc(sizeof(int));
20         *iptr = Accept(listenfd, cliaddr, &len);
21         Pthread_create(&tid, NULL, &doit, iptr);
22     }
23 }
24 static void *
25 doit(void *arg)
26 {
27     int     connfd;
28     connfd = *((int *) arg);
29     free(arg);
30     Pthread_detach(pthread_self());
31     str_echo(connfd);           /* same function as before */
32     Close(connfd);              /* we are done with connected socket */
33     return (NULL);
34 }
```

threads/tcpserv02.c

Figure 23.4 TCP echo server using threads with more portable argument passing.

Thread-Safe Functions

Posix.1 requires that all the functions defined by Posix.1 and by the ANSI C standard be thread-safe, with the exceptions listed in Figure 23.5.

Unfortunately Posix.1g says nothing about thread safety with regard to the networking API functions. The last five lines in this table are from Unix 98. We talked about the nonreentrant property of `gethostbyname` and `gethostbyaddr` in Section 11.14. We mentioned that even though some vendors have defined thread-safe

Need not be thread-safe	Must be thread-safe	Comment
asctime	asctime_r	
	ctermid	thread-safe only if nonnull argument
ctime	ctime_r	
getc_unlocked		
getchar_unlocked		
getgrid	getgrid_r	
getgrnam	getgrnam_r	
getlogin	getlogin_r	
getpwnam	getpwnam_r	
getpwuid	getpwuid_r	
gmtime	gmtime_r	
localtime	localtime_r	
putc_unlocked		
putchar_unlocked		
rand	rand_r	
readdir	readdir_r	
strtok	strtok_r	
	tmpnam	thread-safe only if nonnull argument
ttyname	ttyname_r	
gethostXXX		
getnetXXX		
getprotoXXX		
getservXXX		
inet_ntoa		

Figure 23.5 Thread-safe functions.

versions whose names end in _r, there is no standard for these functions, and they should be avoided. All of the nonreentrant getXXX functions were summarized in Figure 9.9.

We see from this figure that the common technique for making a function thread-safe is to define a new function whose name ends in _r. Two of the functions are thread-safe only if the caller allocates space for the result and passes that pointer as the argument to the function.

23.5 Thread-Specific Data

When writing the code for Chapter 27 the author stumbled over a common programming error when converting a nonthreaded application to use threads. The error was only found when running the server in Figure 27.27, but the error is not in that figure, but in the readline function that is called to handle each client. The same function is called by Figure 23.3.

As with many other thread-related programming errors the failure was nondeterministic. In fact, the programs in Figures 23.3 and 27.27 both worked, but the error was

found when running the timing tests for the threaded version in Figure 27.29. But the version in Figure 27.29 worked when the client and server were on the same host, yet failed at different points with the error "client request for 0 bytes" from our `web_child` function (Figure 27.8) when the client and server were on different hosts. After a few hours of futile debugging, it finally dawned on the author that in the process of speeding up the `readline` function (Figure 3.16) static variables were added (Figure 3.17). This speedup breaks the function when called from different threads within a single process.

This is a common problem when converting existing functions to run in a threads environment and there are various solutions.

1. Use thread-specific data. This is nontrivial and then converts the function into one that works only on systems with threads support. The advantage to this approach is that the calling sequence does not change and all the changes go into the library function and not the applications that call the function. We show a version of `readline` that is thread-safe by using thread-specific data later in this section.

2. Change the calling sequence so that the caller packages all the arguments into a structure, and also store in that structure the static variables from Figure 3.17. This was also done, and Figure 23.6 shows the new structure and the new function prototypes.

```
typedef struct {
  int       read_fd;        /* caller's descriptor to read from */
  char     *read_ptr;       /* caller's buffer to read into */
  size_t    read_maxlen;    /* caller's max #bytes to read */
                /* next three are used internally by the function */
  int       rl_cnt;         /* initialize to 0 */
  char     *rl_bufptr;      /* initialize to rl_buf */
  char      rl_buf[MAXLINE];
} Rline;

void    readline_rinit(int, void *, size_t, Rline *);
ssize_t readline_r(Rline *);
ssize_t Readline_r(Rline *);
```

Figure 23.6 Data structure and function prototype for reentrant version of `readline`.

These new functions can be used on threaded and nonthreaded systems but all the applications that call `readline` must change.

3. Ignore the speedups introduced in Figure 3.17 and go back to the older version in Figure 3.16.

Thread-specific data is a common technique for making an existing function thread-safe. Before describing the Pthread functions that work with thread-specific data, we describe the concept and a *possible* implementation, because the functions appear more complicated than they really are.

Part of the complication in all the threads books that the author has seen is that their descriptions of thread-specific data read like the Pthreads standard, talking about key–value pairs and keys being opaque objects. We describe thread-specific data in terms of *indexes* and *pointers*, because common implementations use a small integer index for the key, and the value associated with the index is just a pointer to a region that the thread `malloc`s.

Each system supports a limited number of thread-specific data items. Posix.1 requires this limit be no less than 128 (per process), and we assume this limit in the following example. The system (probably the threads library) maintains one array of structures per process, which we call `Key` structures, as we show in Figure 23.7.

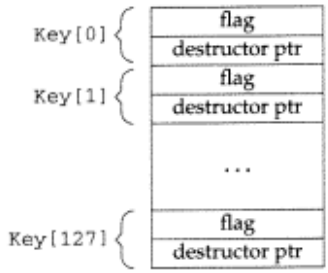


Figure 23.7 Possible implementation of thread-specific data.

The flag in the `Key` structure indicates whether this array element is currently in use, and all the flags are initialized to be "not in use." When a thread calls `pthread_key_create` to create a new thread-specific data item, the system searches through its array of `Key` structures and finds the first one not in use. Its index, 0 through 127, is called the *key* and this index is returned to the calling thread. We talk about the "destructor pointer," the other member of the `Key` structure, shortly.

In addition to the process-wide array of `Key` structures, the system maintains numerous pieces of information about each thread within a process. We call this a `Pthread` structure and part of this information is a 128-element array of pointers, which we call the `pkey` array. We show this in Figure 23.8.

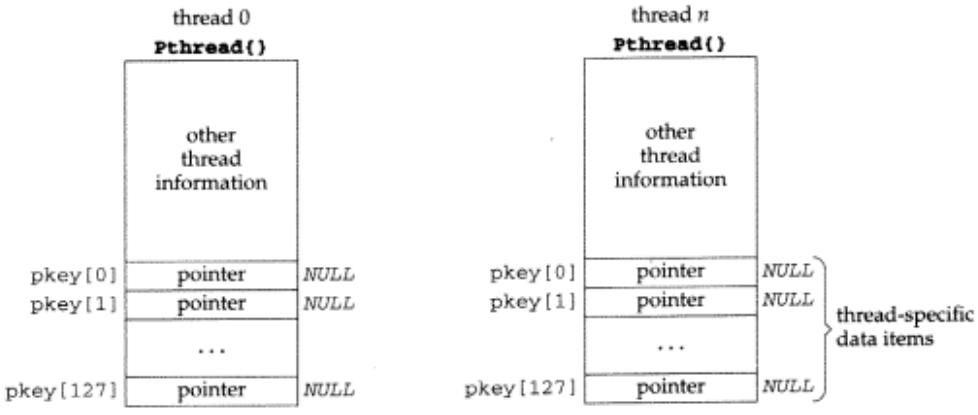


Figure 23.8 Information maintained by the system about each thread.

All entries in the pkey array are initialized to null pointers. These 128 pointers are the "values" associated with each of the possible 128 "keys" in the process.

When we create a key with pthread_key_create, the system tells us its key (index). Each thread can then store a value (pointer) for the key, and each thread normally obtains the pointer by calling malloc. Part of the confusion with thread-specific data is that the pointer is the value in the key–value pair, but the *real* thread-specific data is whatever this pointer points to.

We now go through an example of how thread-specific data is used, assuming that our readline function uses thread-specific data to maintain per-thread state across successive calls to the function. Shortly we will show the code for this, modifying our readline function to follow these steps.

1. A process is started and multiple threads are created.
2. One of the threads will be the first to call readline and it in turn calls pthread_key_create. The system finds the first unused Key structure in Figure 23.7 and returns its index (0–127) to the caller. We assume an index of 1 in this example.

   We will use the pthread_once function to guarantee that pthread_key_create is called only by the first thread to call readline.
3. readline calls pthread_getspecific to get the pkey[1] value (the "pointer" in Figure 23.8 for this key of 1) for this thread, and the returned value is a null pointer. Therefore readline calls malloc to allocate the memory that it needs to keep the per-thread information across successive calls to readline for this thread. readline initializes this memory as needed and calls pthread_setspecific to set the thread-specific data pointer (pkey[1]) for this key to point to the memory that it just allocated. We show this in Figure 23.9, assuming that the calling thread is thread 0 in the process.

   In this figure we note that the Pthread structure is maintained by the system (probably the thread library), but the actual thread-specific data that we malloc is maintained by our function (readline in this case). All that pthread_setspecific does is set the pointer for this key in the Pthread structure to point to our allocated memory. Similarly, all that pthread_getspecific does is return that pointer to us.
4. Another thread, say thread n, calls readline, perhaps while thread 0 is still executing within readline.

   readline calls pthread_once to initialize the key for this thread-specific data item, but since it has already been called, it is not called again.
5. readline calls pthread_getspecific to fetch the pkey[1] pointer for this thread, and a null pointer is returned. This thread then calls malloc followed by pthread_setspecific, just like thread 0, initializing its thread-specific data for this key (1). We show this in Figure 23.10.
6. Thread n continues executing in readline, using and modifying its own thread-specific data.

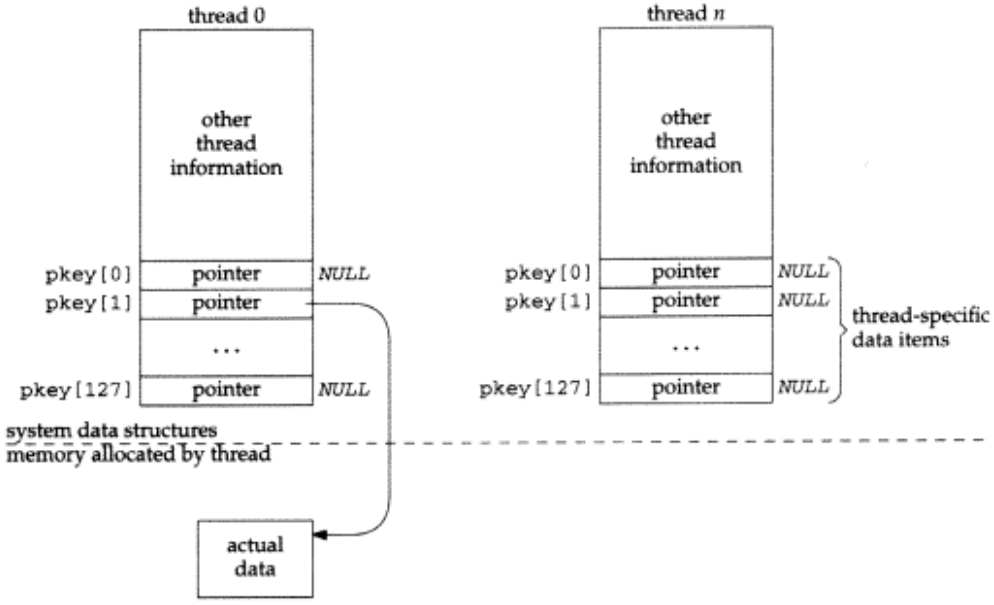


Figure 23.9 Associating `malloc`ed region with thread-specific data pointer.

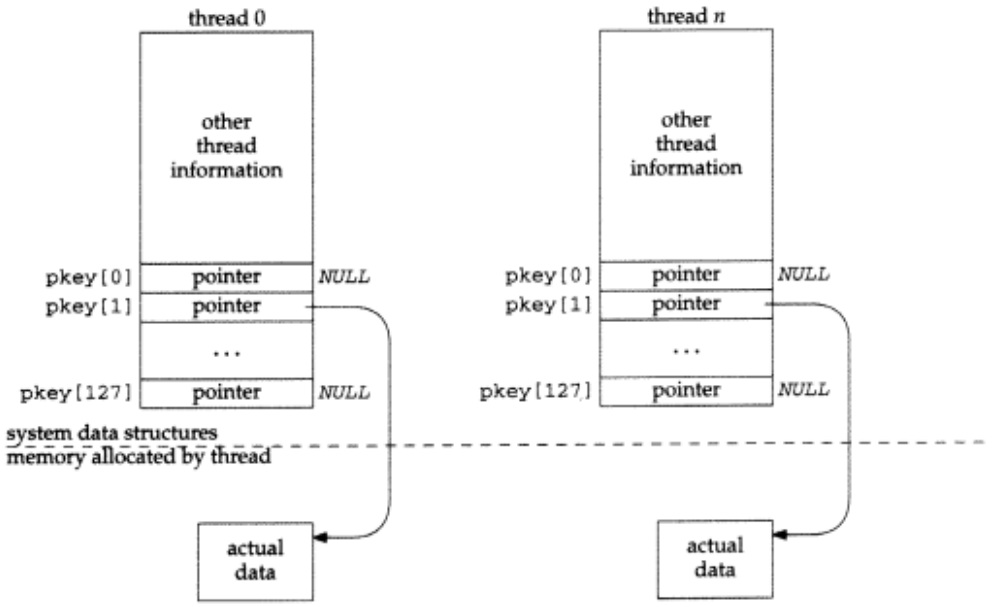


Figure 23.10 Data structures after thread *n* initializes its thread-specific data.

One item we have not addressed is: what happens when a thread terminates? If the thread has called our `readline` function, that function allocated a region of memory that needs to be freed. This is where the "destructor pointer" from Figure 23.7 is used. When the thread that creates the thread-specific data item calls `pthread_key_create`, one argument to this function is a pointer to a *destructor* function. When a thread terminates, the system goes through that thread's `pkey` array, calling the corresponding destructor function for each nonnull `pkey` pointer. What we mean by "corresponding destructor" is the function pointer stored in the `Key` array in Figure 23.7. This is how the thread-specific data is freed when a thread terminates.

The first two functions that are normally called when dealing with thread-specific data are `pthread_once` and `pthread_key_create`.

```
#include <pthread.h>

int pthread_once(pthread_once_t *onceptr, void (*init)(void));

int pthread_key_create(pthread_key_t *keyptr, void (*destructor)(void *value));
```

Both return: 0 if OK, positive `Exxx` value on error

`pthread_once` is normally called every time a function is called that uses thread-specific data, but `pthread_once` uses the value in the variable pointed to by *onceptr* to guarantee that the *init* function is called only one time per process.

`pthread_key_create` must be called only one time for a given key within a process. The key is returned through the *keyptr* pointer, and the *destructor* function, if the argument is a nonnull pointer, will be called by each thread upon termination if that thread has stored a value for this key.

Typical usage of these two functions (ignoring error returns) is as follows:

```
pthread_key_t   rl_key;
pthread_once_t  rl_once = PTHREAD_ONCE_INIT;

void
readline_destructor(void *ptr)
{
    free(ptr);
}

void
readline_once(void)
{
    pthread_key_create(&rl_key, readline_destructor);
}

ssize_t
readline( ... )
{
    ...

    pthread_once(&rl_once, readline_once);
```

```
    if ( (ptr = pthread_getspecific(rl_key)) == NULL) {
        ptr = Malloc( ... );
        pthread_setspecific(rl_key, ptr);
        /* initialize memory pointed to by ptr */
    }
    ...
    /* use the values pointed to by ptr */
}
```

Every time `readline` is called, it calls `pthread_once`. This function uses the value pointed to by its *onceptr* argument (the contents of the variable `rl_once`) to make certain that its *init* function is called only one time. This initialization function, `readline_once`, creates the thread-specific data key that is stored in `rl_key`, and which `readline` then uses in calls to `pthread_getspecific` and `pthread_setspecific`.

The `pthread_getspecific` and `pthread_setspecific` functions are used to fetch and store the value associated with a key. This value is what we called the "pointer" in Figure 23.8. What this pointer points to is up to the application, but normally it points to dynamically allocated memory.

```
#include <pthread.h>

void *pthread_getspecific(pthread_key_t key);
```

Returns: pointer to thread-specific data (possibly a null pointer)

```
int pthread_setspecific(pthread_key_t key, const void *value);
```

Returns: 0 if OK, positive *Exxx* value on error

Notice that the argument to `pthread_key_create` is a pointer to the key (because this function stores the value assigned to the key), while the arguments to the `get` and `set` functions are the key itself (probably a small integer index as we discussed earlier).

Example: `readline` Function Using Thread-Specific Data

We now show a complete example of thread-specific data by converting the optimized version of our `readline` function from Figure 3.17 to be thread-safe, without changing the calling sequence.

Figure 23.11 shows the first part of the function: the `pthread_key_t` variable, the `pthread_once_t` variable, the `readline_destructor` function, the `readline_once` function, and our `Rline` structure that contains all the information we must maintain on a per-thread basis.

Destructor

4–8 Our destructor function just frees the memory that was allocated for this thread.

One-time Function

9–13 We will see that our one-time function is called one time by `pthread_once`, and it just creates the key that is used by `readline`.

threads/readline.c

```
 1 #include    "unpthread.h"

 2 static pthread_key_t rl_key;
 3 static pthread_once_t rl_once = PTHREAD_ONCE_INIT;

 4 static void
 5 readline_destructor(void *ptr)
 6 {
 7     free(ptr);
 8 }

 9 static void
10 readline_once(void)
11 {
12     Pthread_key_create(&rl_key, readline_destructor);
13 }

14 typedef struct {
15     int     rl_cnt;             /* initialize to 0 */
16     char   *rl_bufptr;          /* initialize to rl_buf */
17     char    rl_buf[MAXLINE];
18 } Rline;
```

threads/readline.c

Figure 23.11 First part of thread-safe `readline` function.

`Rline` structure

14–18 Our `Rline` structure contains the three variables that caused the problem by being declared `static` in Figure 3.17. One of these structures will be dynamically allocated per thread, and then released by our destructor function.

Figure 23.12 shows the actual `readline` function and the function `my_read` that it calls. This figure is a modification of Figure 3.17.

`my_read` function

19–35 The first argument to the function is now a pointer to the `Rline` structure that was allocated for this thread (the actual thread-specific data).

Allocate thread-specific data

42 We first call `pthread_once` so that the first thread that calls `readline` in this process calls `readline_once` to create the thread-specific data key.

Fetch thread-specific data pointer

43–46 `pthread_getspecific` returns the pointer to the `Rline` structure for this thread. But if this is the first time this thread has called `readline`, the return value is a null pointer. In this case we allocate space for an `Rline` structure and the `rl_cnt` member is initialized to 0 by `calloc`. We then store this pointer for this thread by calling `pthread_setspecific`. The next time this thread calls `readline`, `pthread_getspecific` will return this pointer that was just stored.

threads/readline.c

```
19 static ssize_t
20 my_read(Rline *tsd, int fd, char *ptr)
21 {
22     if (tsd->rl_cnt <= 0) {
23       again:
24         if ( (tsd->rl_cnt = read(fd, tsd->rl_buf, MAXLINE)) < 0) {
25             if (errno == EINTR)
26                 goto again;
27             return (-1);
28         } else if (tsd->rl_cnt == 0)
29             return (0);
30         tsd->rl_bufptr = tsd->rl_buf;
31     }
32     tsd->rl_cnt--;
33     *ptr = *tsd->rl_bufptr++;
34     return (1);
35 }

36 ssize_t
37 readline(int fd, void *vptr, size_t maxlen)
38 {
39     int     n, rc;
40     char    c, *ptr;
41     Rline  *tsd;

42     Pthread_once(&rl_once, readline_once);
43     if ( (tsd = pthread_getspecific(rl_key)) == NULL) {
44         tsd = Calloc(1, sizeof(Rline));     /* init to 0 */
45         Pthread_setspecific(rl_key, tsd);
46     }
47     ptr = vptr;
48     for (n = 1; n < maxlen; n++) {
49         if ( (rc = my_read(tsd, fd, &c)) == 1) {
50             *ptr++ = c;
51             if (c == '\n')
52                 break;
53         } else if (rc == 0) {
54             if (n == 1)
55                 return (0);     /* EOF, no data read */
56             else
57                 break;          /* EOF, some data was read */
58         } else
59             return (-1);        /* error, errno set by read() */
60     }

61     *ptr = 0;
62     return (n);
63 }
```

threads/readline.c

Figure 23.12 Second part of thread-safe `readline` function.

23.6 Web Client and Simultaneous Connections (Continued)

We now revisit the Web client example from Section 15.5 and recode it using threads instead of nonblocking `connect`s. With threads we can leave the sockets in their default blocking mode and create one thread per connection. It is OK for each thread to block in its call to `connect`, as the kernel will just run some other thread that is ready.

Figure 23.13 shows the first part of the program, the globals and the start of the `main` function.

threads/web01.c

```
 1 #include    "unpthread.h"
 2 #include    <thread.h>          /* Solaris threads */

 3 #define MAXFILES    20
 4 #define SERV        "80"        /* port number or service name */

 5 struct file {
 6     char   *f_name;             /* filename */
 7     char   *f_host;             /* hostname or IP address */
 8     int     f_fd;               /* descriptor */
 9     int     f_flags;            /* F_xxx below */
10     pthread_t f_tid;            /* thread ID */
11 } file[MAXFILES];
12 #define F_CONNECTING    1       /* connect() in progress */
13 #define F_READING       2       /* connect() complete; now reading */
14 #define F_DONE          4       /* all done */

15 #define GET_CMD     "GET %s HTTP/1.0\r\n\r\n"

16 int     nconn, nfiles, nlefttoconn, nlefttoread;

17 void   *do_get_read(void *);
18 void    home_page(const char *, const char *);
19 void    write_get_cmd(struct file *);

20 int
21 main(int argc, char **argv)
22 {
23     int     i, n, maxnconn;
24     pthread_t tid;
25     struct file *fptr;

26     if (argc < 5)
27         err_quit("usage: web <#conns> <IPaddr> <homepage> file1 ...");
28     maxnconn = atoi(argv[1]);

29     nfiles = min(argc - 4, MAXFILES);
30     for (i = 0; i < nfiles; i++) {
31         file[i].f_name = argv[i + 4];
32         file[i].f_host = argv[2];
33         file[i].f_flags = 0;
34     }
35     printf("nfiles = %d\n", nfiles);

36     home_page(argv[2], argv[3]);
```

```
37      nlefttoread = nlefttoconn = nfiles;
38      nconn = 0;
```
threads/web01.c

Figure 23.13 Globals and start of `main` function.

Globals

1-16 We `#include <thread.h>`, in addition to the normal `<pthread.h>` because we need to use Solaris threads in addition to Pthreads, as we describe shortly.

10 We have added one member to the `file` structure: `f_tid`, the thread ID. The remainder of this code is similar to Figure 15.15. With this threads version we do not use `select` and therefore do not need any descriptor sets or the variable `maxfd`.

36 The `home_page` function that is called is unchanged from Figure 15.16.

Figure 23.14 shows the main processing loop of the `main` thread.

threads/web01.c

```
39      while (nlefttoread > 0) {
40          while (nconn < maxnconn && nlefttoconn > 0) {
41                  /* find a file to read */
42              for (i = 0; i < nfiles; i++)
43                  if (file[i].f_flags == 0)
44                      break;
45              if (i == nfiles)
46                  err_quit("nlefttoconn = %d but nothing found", nlefttoconn);
47              file[i].f_flags = F_CONNECTING;
48              Pthread_create(&tid, NULL, &do_get_read, &file[i]);
49              file[i].f_tid = tid;
50              nconn++;
51              nlefttoconn--;
52          }
53          if ( (n = thr_join(0, &tid, (void **) &fptr)) != 0)
54              errno = n, err_sys("thr_join error");
55          nconn--;
56          nlefttoread--;
57          printf("thread id %d for %s done\n", tid, fptr->f_name);
58      }
59      exit(0);
60  }
```
threads/web01.c

Figure 23.14 Main processing loop of `main` function.

If possible, create another thread

40-52 If we are allowed to create another thread (`nconn` is less than `maxnconn`), we do so. The function that each new thread executes is `do_get_read` and the argument is the pointer to the `file` structure.

Wait for any thread to terminate

53-54 We call the Solaris thread function `thr_join` with a first argument of 0 to wait for any one of our threads to terminate. Unfortunately Pthreads does not provide a way to

wait for *any* one of our threads to terminate; the `pthread_join` function makes us specify exactly which thread we wish to wait for. We will see in Section 23.9 that the Pthreads solution for this problem is more complicated, requiring us to use a condition variable for the terminating thread to notify the main thread when it is done.

> The solution that we show, using the Solaris thread `thr_join` function is not portable to all environments. Nevertheless, we want to show this version of our Web client example using threads without having to complicate the discussion with condition variables and mutexes. Fortunately one is able to mix Pthreads with Solaris threads under Solaris.
>
> When the author complained on Usenet about the inability of `pthread_join` to wait for *any* thread to terminate, a few people who had worked on the Pthreads standard justified this design decision, claiming that the `pthread_join` cannot be everything to everybody. The claim is also made that in the process model, there is a parent–child relationship so the ability of `wait` or `waitpid` to wait for any child makes sense. But in a threads environment there is no hierarchical relationship similar to parent–child, so waiting for any thread to terminate does not make sense. The thread whose status is returned by a wait-for-any function need not be one that the calling thread created. They added that if someone really needs to wait for any thread, it can be implemented (nontrivially) using condition variables, as we show shortly. Regardless of these arguments, the author considers the design of `pthread_join` flawed.

Figure 23.15 shows the `do_get_read` function, which is executed by each thread. This function establishes the TCP connection, sends an HTTP `GET` command to the server, and reads the server's reply.

Create TCP socket, establish connection

68–71 A TCP socket is created and a connection is established by our `tcp_connect` function. The socket is a normal blocking socket, so the thread will block in the call to `connect`, until the connection is established.

Write request to server

72 `write_get_cmd` builds the HTTP `GET` command and sends it to the server. We do not show this function again, as the only difference from Figure 15.18 is that the threads version does not call `FD_SET` and does not use `maxfd`.

Read server's reply

73–82 The server's reply is then read. When the connection is closed by the server, the `F_DONE` flag is set, and the function returns, terminating the thread.

We also do not show the `home_page` function, as it is identical to the version shown in Figure 15.16.

We return to this example, replacing the Solaris `thr_join` function with the more portable Pthreads solution, but we must first discuss mutexes and condition variables.

23.7 Mutexes: Mutual Exclusion

Notice in Figure 23.14 that when a thread terminates the main loop decrements both `nconn` and `nlefttoread`. We could have placed these two decrements in the function

threads/web01.c

```
61 void *
62 do_get_read(void *vptr)
63 {
64     int     fd, n;
65     char    line[MAXLINE];
66     struct file *fptr;

67     fptr = (struct file *) vptr;

68     fd = Tcp_connect(fptr->f_host, SERV);
69     fptr->f_fd = fd;
70     printf("do_get_read for %s, fd %d, thread %d\n",
71            fptr->f_name, fd, fptr->f_tid);

72     write_get_cmd(fptr);        /* write() the GET command */

73         /* Read server's reply */
74     for ( ; ; ) {
75         if ( (n = Read(fd, line, MAXLINE)) == 0)
76             break;              /* server closed connection */

77         printf("read %d bytes from %s\n", n, fptr->f_name);
78     }
79     printf("end-of-file on %s\n", fptr->f_name);
80     Close(fd);
81     fptr->f_flags = F_DONE;     /* clears F_READING */

82     return (fptr);              /* terminate thread */
83 }
```

threads/web01.c

Figure 23.15 do_get_read function.

do_get_read, letting each thread decrement these two counters immediately before the thread terminates. But this would be a subtle, yet significant, concurrent programming error.

The problem with placing the code in the function that each thread executes is that these two variables are global, not thread-specific. If one thread is in the middle of decrementing a variable, that thread is suspended, and another thread executes and decrements the same variable, an error can result. For example, assume that the C compiler turns the decrement operator into three instructions: load from memory into a register, decrement the register, and store from the register into memory. Consider the following possible scenario:

1. Thread A is running and it loads the value of nconn (3) into a register.
2. The system switches threads from A to B. A's registers are saved, and B's registers are restored.
3. Thread B executes the three instructions corresponding to the C expression nconn--, storing the new value of 2.

4. Sometime later the system switches threads from B to A. A's registers are restored and A continues where it left off, at the second machine instruction in the three-instructions sequence: the value of the register is decremented from 3 to 2 and the value of 2 is stored in `nconn`.

The end result is that `nconn` is 2 when it should be 1. This is wrong.

These types of concurrent programming errors are hard to find for numerous reasons. First, they occur rarely. Nevertheless, it is an error and it will fail (Murphy's Law). Secondly, the error is hard to duplicate, since it depends on the nondeterministic timing of many events. Lastly, on some systems the hardware instructions might be atomic; that is, there exists a hardware instruction to decrement an integer in memory (instead of the three-instruction sequence that we assumed above) and the hardware cannot be interrupted during this instruction. But this is not guaranteed by all systems, so the code works on one system but not on another.

We call threads programming *concurrent programming* or *parallel programming* since multiple threads can be running concurrently (in parallel), accessing the same variables. While the error scenario that we just discussed assumes a single CPU system, the potential for error also exists if threads A and B are running at the same time, on different CPUs on a multiprocessor system. With normal Unix programming we do not encounter these concurrent programming problems, because with `fork` nothing besides descriptors are shared between the parent and child. We will, however, encounter this same type of problem when we discuss shared memory between processes.

We can easily demonstrate this problem with threads. Figure 23.17 is a simple program that creates two threads and then has each thread increment a global variable 5000 times.

We exacerbate the potential for a problem by fetching the current value of `counter`, printing the new value, and then storing the new value. If we run this program we have the output shown in Figure 23.16.

```
4: 1
4: 2
4: 3
4: 4
                    this continues as thread 4 executes
4: 517
4: 518
5: 518              thread 5 now executes
5: 519
5: 520
                    this continues as thread 5 executes
5: 926
5: 927
4: 519              thread 4 now executes, stored value is wrong
4: 520
```

Figure 23.16 Output from the program in Figure 23.17.

threads/example01.c

```
 1 #include    "unpthread.h"
 2 #define NLOOP 5000
 3 int     counter;                /* this is incremented by the threads */
 4 void   *doit(void *);
 5 int
 6 main(int argc, char **argv)
 7 {
 8     pthread_t tidA, tidB;
 9     Pthread_create(&tidA, NULL, &doit, NULL);
10     Pthread_create(&tidB, NULL, &doit, NULL);
11         /* wait for both threads to terminate */
12     Pthread_join(tidA, NULL);
13     Pthread_join(tidB, NULL);
14     exit(0);
15 }
16 void *
17 doit(void *vptr)
18 {
19     int     i, val;
20     /*
21      * Each thread fetches, prints, and increments the counter NLOOP times.
22      * The value of the counter should increase monotonically.
23      */
24     for (i = 0; i < NLOOP; i++) {
25         val = counter;
26         printf("%d: %d\n", pthread_self(), val + 1);
27         counter = val + 1;
28     }
29     return (NULL);
30 }
```

threads/example01.c

Figure 23.17 Two threads that increment a global variable incorrectly.

Notice the error the first time the system switches from thread 4 to thread 5: the value 518 is stored by each thread. This happens numerous times through the 10,000 lines of output.

The nondeterministic nature of this type of problem is also evident if we run the program a few times: each time the end result is different from the previous run of the program. Also, if we redirect the output to a disk file, sometimes the error does not occur, since the program runs faster, providing fewer opportunities to switch between the threads. The greatest number of errors occurs when we run the program interactively, writing the output to the (slow) terminal, but saving the output in a file using the Unix `script` program (discussed in detail in Chapter 19 of APUE).

The problem that we just discussed, multiple threads updating a shared variable, is the simplest of these problems. The solution is to protect the shared variable with a *mutex* (which stands for "mutual exclusion") and access the variable only when we hold the mutex. In terms of Pthreads, a mutex is a variable of type `pthread_mutex_t`. We lock and unlock the mutex using the following two functions.

```
#include <pthread.h>

int pthread_mutex_lock(pthread_mutex_t *mptr);

int pthread_mutex_unlock(pthread_mutex_t *mptr);
```

Both return: 0 if OK, positive Exxx value on error

If we try to lock a mutex that is already locked by some other thread, we are blocked until the mutex is unlocked.

If a mutex variable is statically allocated, we must initialize it to the constant `PTHREAD_MUTEX_INITIALIZER`. We will see in Section 27.8 that if we allocate a mutex in shared memory we must initialize it at run time by calling the `pthread_mutex_init` function.

> Some systems (e.g., Solaris) define `PTHREAD_MUTEX_INITIALIZER` to be 0, so omitting this initialization is OK, since statically allocated variables are automatically initialized to 0. But there is no guarantee that this is OK and other systems (e.g., Digital Unix) define the initializer to be nonzero.

Figure 23.18 is a corrected version of Figure 23.17 that uses a single mutex to lock the counter between the two threads.

threads/example02.c

```
 1 #include    "unpthread.h"

 2 #define NLOOP 5000

 3 int     counter;                /* this is incremented by the threads */
 4 pthread_mutex_t counter_mutex = PTHREAD_MUTEX_INITIALIZER;

 5 void   *doit(void *);

 6 int
 7 main(int argc, char **argv)
 8 {
 9     pthread_t tidA, tidB;

10     Pthread_create(&tidA, NULL, &doit, NULL);
11     Pthread_create(&tidB, NULL, &doit, NULL);

12         /* wait for both threads to terminate */
13     Pthread_join(tidA, NULL);
14     Pthread_join(tidB, NULL);

15     exit(0);
16 }
```

```
17 void *
18 doit(void *vptr)
19 {
20      int     i, val;

21      /*
22       * Each thread fetches, prints, and increments the counter NLOOP times.
23       * The value of the counter should increase monotonically.
24       */

25      for (i = 0; i < NLOOP; i++) {
26          Pthread_mutex_lock(&counter_mutex);

27          val = counter;
28          printf("%d: %d\n", pthread_self(), val + 1);
29          counter = val + 1;

30          Pthread_mutex_unlock(&counter_mutex);
31      }

32      return (NULL);
33 }
```

threads/example02.c

Figure 23.18 Corrected version of Figure 23.17 using a mutex to protect the shared variable.

We declare a mutex named `counter_mutex` and this mutex must be locked by the thread before the thread manipulates the `counter` variable. When we run this program, the output is always correct: the value is incremented monotonically and the final value printed is always 10,000.

How much overhead is involved with mutex locking? The programs in Figures 23.17 and 23.18 were changed to loop 50,000 times, and timed while the output was directed to `/dev/null`. The difference in CPU time from the incorrect version with no mutual exclusion to the correct version that used a mutex was 10%. This tells us that mutex locking is not a large overhead.

23.8 Condition Variables

A mutex is fine to prevent simultaneous access to a shared variable, but we need something else to let us go to sleep waiting for some condition to occur. Let's demonstrate this with an example. We return to our Web client in Section 23.6 and replace the Solaris `thr_join` with `pthread_join`. But we cannot call the Pthread function until we know that a thread has terminated. We first declare a global that counts the number of terminated threads and protect it with a mutex.

```
int             ndone;          /* number of terminated threads */
pthread_mutex_t ndone_mutex = PTHREAD_MUTEX_INITIALIZER;
```

We then require that each thread increment this counter when it terminates, being careful to use the associated mutex.

```
void *
do_get_read(void *vptr)
{
    ...

    Pthread_mutex_lock(&ndone_mutex);
    ndone++;
    Pthread_mutex_unlock(&ndone_mutex);

    return(fptr);       /* terminate thread */
}
```

This is fine, but how do we code the main loop? It needs to lock the mutex continually and check if any threads have terminated.

```
while (nlefttoread > 0) {
    while (nconn < maxnconn && nlefttoconn > 0) {
            /* find a file to read */
        ...
    }

        /* See if one of the threads is done */
    Pthread_mutex_lock(&ndone_mutex);
    if (ndone > 0) {
        for (i = 0; i < nfiles; i++) {
            if (file[i].f_flags & F_DONE) {
                Pthread_join(file[i].f_tid, (void **) &fptr);

                /* update file[i] for terminated thread */
                ...
            }
        }
    }
    Pthread_mutex_unlock(&ndone_mutex);
}
```

While this is OK, it means the main loop *never* goes to sleep; it just loops, checking ndone every time around the loop. This is called *polling* and is considered a waste of CPU time.

We want a method for the main loop to go to sleep until one of its threads notifies it that something is ready. A *condition variable*, in conjunction with a mutex, provides this facility. The mutex provides mutual exclusion and the condition variable provides a signaling mechanism.

In terms of Pthreads, a condition variable is a variable of type `pthread_cond_t`. They are used with the following two functions.

```
#include <pthread.h>

int pthread_cond_wait(pthread_cond_t *cptr, pthread_mutex_t *mptr);

int pthread_cond_signal(pthread_cond_t *cptr);
```

Both return: 0 if OK, positive Exxx value on error

The term "signal" in the second function's name does not refer to a Unix SIG*xxx* signal.

An example is the easiest way to explain these functions. Returning to our Web client example, the counter ndone is now associated with both a condition variable and a mutex:

```
int             ndone;
pthread_mutex_t ndone_mutex = PTHREAD_MUTEX_INITIALIZER;
pthread_cond_t  ndone_cond  = PTHREAD_COND_INITIALIZER;
```

A thread notifies the main loop that it is terminating by incrementing the counter while its mutex lock is held and by signaling the condition variable:

```
Pthread_mutex_lock(&ndone_mutex);
ndone++;
Pthread_cond_signal(&ndone_cond);
Pthread_mutex_unlock(&ndone_mutex);
```

The main loop then blocks in a call to pthread_cond_wait, waiting to be signaled by a terminating thread:

```
while (nlefttoread > 0) {
    while (nconn < maxnconn && nlefttoconn > 0) {
            /* find a file to read */
        ...
    }

        /* Wait for one of the threads to terminate */
    Pthread_mutex_lock(&ndone_mutex);
    while (ndone == 0)
        Pthread_cond_wait(&ndone_cond, &ndone_mutex);

    for (i = 0; i < nfiles; i++) {
        if (file[i].f_flags & F_DONE) {
            Pthread_join(file[i].f_tid, (void **) &fptr);

            /* update file[i] for terminated thread */
            ...
        }
    }
    Pthread_mutex_unlock(&ndone_mutex);
}
```

Notice that the variable ndone is still checked only while the mutex is held. Then, if there is nothing to do, pthread_cond_wait is called. This puts the calling thread to sleep *and* releases the mutex lock that it holds. Furthermore, when the thread returns from pthread_cond_wait (after some other thread has signaled it), the thread again holds the mutex.

Why is a mutex always associated with a condition variable? The "condition" is normally the value of some variable that is shared between the threads. The mutex is required to allow this variable to be set and tested by the different threads. For example, if we did not have the mutex in the example code just shown, the main loop would test it as follows:

```
        /* Wait for one of the threads to terminate */
    while (ndone == 0)
        Pthread_cond_wait(&ndone_cond, &ndone_mutex);
```

But there is a possibility that the last of the threads increments ndone after the test of ndone == 0, but before the call to pthread_cond_wait. If this happens, this last "signal" is lost, and the main loop would block forever, waiting for something that will never occur again.

This is the same reason that pthread_cond_wait must be called with the associated mutex locked, and why this function unlocks the mutex and puts the calling thread to sleep as a single, atomic operation. If this function did not unlock the mutex, and then lock it again when it returns, the thread would have to do unlock and lock the mutex, and the code would look like:

```
        /* Wait for one of the threads to terminate */
    Pthread_mutex_lock(&ndone_mutex);
    while (ndone == 0) {
        Pthread_mutex_unlock(&ndone_mutex);
        Pthread_cond_wait(&ndone_cond, &ndone_mutex);
        Pthread_mutex_lock(&ndone_mutex);
    }
```

But again, there is a possibility that the final thread could terminate and increment the value of ndone between the call to pthread_mutex_unlock and pthread_cond_wait.

Normally pthread_cond_signal awakens one thread that is waiting on the condition variable. There are instances when a thread knows that multiple threads should be awakened, in which case pthread_cond_broadcast will wake up *all* threads that are blocked on the condition variable.

```
#include <pthread.h>

int pthread_cond_broadcast(pthread_cond_t *cptr);

int pthread_cond_timedwait(pthread_cond_t *cptr, pthread_mutex_t *mptr,
                           const struct timespec *abstime);
```

Both return: 0 if OK, positive Exxx value on error

pthread_cond_timedwait lets a thread place a limit on how long it will block. *abstime* is a timespec structure (as we defined with the pselect function, Section 6.9) that specifies the system time when the function must return, even if the condition variable has not been signaled yet. If this timeout occurs, ETIME is returned.

This time value is an *absolute time*; it is not a *time delta*. That is, *abstime* is the system time—the number of seconds and nanoseconds past January 1, 1970, UTC—when the function should return. This differs from both select and pselect, which specify the number of seconds and microseconds (nanoseconds for pselect) until some time in the future when the function should return. The normal procedure is to call gettimeofday to obtain the current time (as a timeval structure!), copy this into a

timespec structure, adding in the desired time limit. For example,

```
struct timeval  tv;
struct timespec ts;

if (gettimeofday(&tv, NULL) < 0)
    err_sys("gettimeofday error");
ts.tv_sec  = tv.tv_sec + 5;       /* 5 seconds in future */
ts.tv_nsec = tv.tv_usec * 1000;   /* microsec to nanosec */

pthread_cond_timedwait( ... , &ts);
```

The advantage in using an absolute time, instead of a delta time, is if the function prematurely returns (perhaps because of a caught signal). The function can be called again, without having to change the contents of the timespec structure. The disadvantage, however, is having to call gettimeofday before the function can be called the first time.

> Posix.1 defines a new clock_gettime function that returns the current time as a timespec structure.

23.9 Web Client and Simultaneous Connections (Continued)

We now recode our Web client from Section 23.6, removing the call to the Solaris thr_join function and replacing it with a call to pthread_join. As discussed in that section, we now must specify exactly which thread we are waiting for. To do this we will use a condition variable, as described in Section 23.8.

The only change to the globals (Figure 23.13) is to add one new flag and the condition variable:

```
#define   F_JOINED        8   /* main has pthread_join'ed */

int             ndone;        /* number of terminated threads */
pthread_mutex_t ndone_mutex = PTHREAD_MUTEX_INITIALIZER;
pthread_cond_t  ndone_cond  = PTHREAD_COND_INITIALIZER;
```

The only change to the do_get_read function (Figure 23.15) is to increment ndone and signal the main loop before the thread terminates:

```
    printf("end-of-file on %s\n", fptr->f_name);
    Close(fd);

    Pthread_mutex_lock(&ndone_mutex);
    fptr->f_flags = F_DONE;     /* clears F_READING */
    ndone++;
    Pthread_cond_signal(&ndone_cond);
    Pthread_mutex_unlock(&ndone_mutex);

    return(fptr);       /* terminate thread */
}
```

Most changes are in the main loop, Figure 23.14, the new version of which we show in Figure 23.19.

threads/web03.c

```
43      while (nlefttoread > 0) {
44          while (nconn < maxnconn && nlefttoconn > 0) {
45                  /* find a file to read */
46              for (i = 0; i < nfiles; i++)
47                  if (file[i].f_flags == 0)
48                      break;
49              if (i == nfiles)
50                  err_quit("nlefttoconn = %d but nothing found", nlefttoconn);
51              file[i].f_flags = F_CONNECTING;
52              Pthread_create(&tid, NULL, &do_get_read, &file[i]);
53              file[i].f_tid = tid;
54              nconn++;
55              nlefttoconn--;
56          }
57              /* Wait for one of the threads to terminate */
58          Pthread_mutex_lock(&ndone_mutex);
59          while (ndone == 0)
60              Pthread_cond_wait(&ndone_cond, &ndone_mutex);
61          for (i = 0; i < nfiles; i++) {
62              if (file[i].f_flags & F_DONE) {
63                  Pthread_join(file[i].f_tid, (void **) &fptr);
64                  if (&file[i] != fptr)
65                      err_quit("file[i] != fptr");
66                  fptr->f_flags = F_JOINED;   /* clears F_DONE */
67                  ndone--;
68                  nconn--;
69                  nlefttoread--;
70                  printf("thread %d for %s done\n", fptr->f_tid, fptr->f_name);
71              }
72          }
73          Pthread_mutex_unlock(&ndone_mutex);
74      }
75      exit(0);
76  }
```

threads/web03.c

Figure 23.19 Main processing loop of `main` function.

If possible, create another thread

44–56 This code has not changed.

Wait for a thread to terminate

57–60 To wait for one of the threads to terminate we wait for `ndone` to be nonzero. As discussed in Section 23.8, the test must be done while the mutex is locked, and the sleep is performed by `pthread_cond_wait`.

Handle terminated thread

61–73 When a thread has terminated, we go through all the `file` structures to find the appropriate thread, call `pthread_join`, and then set the new `F_JOINED` flag.

Figure 15.20 shows the timing for this version, along with the timing of the version using nonblocking `connect`s.

23.10 Summary

The creation of a new thread is normally faster than the creation of a new process with `fork`. This alone can be an advantage in heavily used network servers. Threads programming, however, is a new paradigm that requires more discipline.

All threads in a process share the global variables and descriptors, allowing this information to be shared between different threads. But this sharing introduces synchronization problems and the Pthread synchronization primitives that we must use are mutexes and condition variables. Synchronization of shared data is a required part of almost every threaded application.

When writing functions that can be called by threaded applications, these functions must be thread-safe. Thread-specific data is one technique that helps with this, and we showed an example with our `readline` function.

We return to the threads model in Chapter 27 with another server design in which the server creates a pool of threads when it starts. An available thread from this pool handles the next client request.

Exercises

23.1 Compare the descriptor usage in a server using `fork` versus a server using a thread, assuming 100 clients are being serviced at the same time.

23.2 What happens in Figure 23.3 if the thread does not `close` the connected socket when `str_echo` returns?

23.3 In Figures 5.5 and 6.13 we print "server terminated prematurely" when we expect an echoed line from the server but receive an end-of-file instead (recall Section 5.12). Modify Figure 23.2 to print this message too, when appropriate.

23.4 Modify Figures 23.11 and 23.12 so that they can compile on a system that does not support threads.

23.5 To see the error with the `readline` function that is used in Figure 23.3, build that program and start the server. Then build the TCP echo client from Figure 6.13 that works in a batch mode correctly. Find a large text file on your system and start the client three times in a batch mode, reading from the large text file and writing its output to a temporary file. If possible, run the clients on a different host from the server. If the three clients terminate correctly (often they hang), look at their temporary output files and compare them to the input file.

Now build a version of the server using the correct `readline` function from Section 23.5. Rerun the test with three clients, and all three clients should now work. You should also put a `printf` in the `readline_destructor` function, the `readline_once` function, and in the call to `malloc` in `readline`. This shows that the key is created only one time, but the memory is allocated for every thread, and that the destructor function is called for every thread.

24

IP Options

24.1 Introduction

IPv4 allows up to 40 bytes of options to follow the fixed 20-byte header. Although 10 different options are defined, the most commonly used is the source route option. Access to these options is through the `IP_OPTIONS` socket option and we demonstrate this with an example that uses source routing.

IPv6 allows extension headers to occur between the fixed 40-byte IPv6 header and the transport layer header (e.g., ICMPv6, TCP, or UDP). Six different extension headers are currently defined. Unlike the IPv4 approach, access to the IPv6 extension headers is through a functional interface instead of forcing the user to understand the actual details of how the headers appear in the IPv6 packet.

24.2 IPv4 Options

In Figure A.1 we show options following the 20-byte IPv4 header. As noted there, the 4-bit header length field limits the total size of the IPv4 header to 15 32-bit words (60 bytes), so the size of the IP options is limited to 40 bytes. Ten different options are defined for IPv4.

1. NOP: no-operation. A 1-byte option typically used for padding to make a later option fall on a 4-byte boundary.
2. EOL: end-of-list. A 1-byte option that terminates option processing. Since the total size of the IP options must be a multiple of 4 bytes, EOL bytes follow the final option.

3. LSRR: loose source and record route (Section 8.5 of TCPv1). We show an example of this shortly.
4. SSRR: strict source and record route (Section 8.5 of TCPv1). We show an example of this shortly.
5. Timestamp (Section 7.4 of TCPv1).
6. Record route (Section 7.3 of TCPv1).
7. Basic security.
8. Extended security.
9. Stream identifier (obsolete).
10. Router alert. This is a new option described in RFC 2113 [Katz 1997]. This option is included in IP datagrams that should be examined by all routers that forward the datagram.

Chapter 9 of TCPv2 provides further details on the kernel processing of the first six options and the indicated sections in TCPv1 provide examples of their use. RFC 1108 [Kent 1991] has additional details on the two security options, which are not widely used.

The `getsockopt` and `setsockopt` functions (with a *level* of `IPPROTO_IP` and an *optname* of `IP_OPTIONS`) fetch and set the IP options. The fourth argument to `getsockopt` and `setsockopt` is a pointer to a buffer (whose size is 44 bytes or less), and the fifth argument is the size of this buffer. The reason that the size of this buffer can be 4 bytes larger than the maximum size of the options is because of the way the source route option is handled, as we describe shortly. Other than the two source route options, the format of what goes into the buffer is the format of the options when placed into the IP datagram.

When the IP options are set using `setsockopt`, the specified options will then be sent on all IP datagrams on that socket. This works for TCP, UDP, and raw IP sockets. To clear these options call `setsockopt` and specify either a null pointer as the fourth argument or a value of 0 as the fifth argument (the length).

> Setting the IP options for a raw IP socket does not work on all implementations if the `IP_HDRINCL` socket option (which we describe in the next chapter) is also set. Many Berkeley-derived implementations do not send the options set with `IP_OPTIONS` when `IP_HDRINCL` is enabled, because the application can set its own IP options in the IP header that it builds (pp. 1056–1057 of TCPv2). Other systems (e.g., FreeBSD) allow the application to specify IP options using either the `IP_OPTIONS` socket option, or setting `IP_HDRINCL` and including them in the IP header that it builds, but not both.

When `getsockopt` is called to fetch the IP options for a connected TCP socket that was created by `accept`, all that is returned is the reversal of the source route option received with the client's SYN for the listening socket (p. 931 of TCPv2). The source route is automatically reversed by TCP because the source route specified by the client was from the client to the server, but the server needs to use the reverse of this route in datagrams it sends to the client. If no source route accompanied the SYN, then the value–result length returned by `getsockopt` through its fifth argument will be 0. For

all other TCP sockets and for all UDP sockets and raw IP sockets, calling `getsockopt` to fetch the IP options just returns a copy of whatever IP options have been set by `setsockopt` for the socket. Note that for a raw IP socket the received IP header, including any IP options, is always returned by the input functions, so the received IP options are always available.

> Berkeley-derived kernels have never returned a received source route, or any other IP options, for a UDP socket. The code shown on p. 775 of TCPv2 to return the IP options has existed since 4.3BSD Reno, but has always been commented out since it does not work. This makes it impossible for a UDP application to use the reverse of a received route for datagrams back to the sender.
>
> Many Berkeley-derived kernels panic (i.e., the system halts) if `getsockopt` or `setsockopt` is called for a raw IP socket. But it takes superuser privileges to create a raw IP socket and there are many more malevolent deeds that someone with superuser privileges can do to a system.

24.3 IPv4 Source Route Options

A *source route* is a list of IP addresses specified by the sender of the IP datagram. If the source route is *strict*, then the datagram must pass through each listed node and only the listed nodes. That is, all the nodes listed in the source route must be neighbors. But if the source route is *loose*, the datagram must pass through each listed node but can also pass through other nodes that do not appear in the source route.

> IPv4 source routing is controversial. [Cheswick and Bellovin 1994, p. 26] advocate disabling the feature on all your routers, and many organizations and service providers do this. One legitimate use for source routing is to detect asymmetric routes using the Traceroute program, as demonstrated on pp. 108–109 of TCPv1, although as more and more routers on the Internet disable source routing, even this use disappears. Nevertheless, specifying and receiving source routes is part of the sockets API and needs to be described.

IPv4 source routes are called *source and record routes*, LSRR for the loose option and SSRR for the strict option, because as the datagram passes through all the listed nodes each one replaces its listed address with the address of the outgoing interface. This allows the receiver to take this new list and reverse it to follow the reverse path back to the sender. Examples of these two source routes, along with the corresponding `tcpdump` output, is in Section 8.5 of TCPv1.

We specify a source route as an array of IPv4 addresses, prefixed by three 1-byte fields, as shown in Figure 24.1. This is the format of the buffer that we will pass to `setsockopt`.

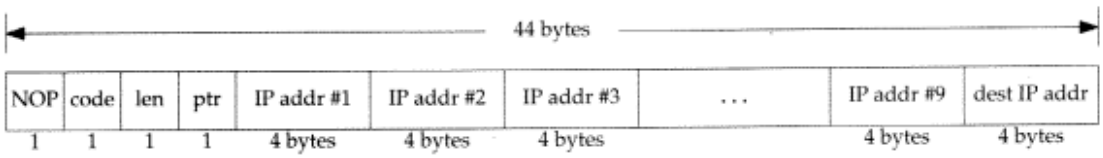


Figure 24.1 Passing a source route to the kernel.

We place a NOP before the source route option, which causes all the IP addresses to be

aligned on a 4-byte boundary. This is not required but takes no additional space (the IP options are always padded to be a multiple of 4 bytes) and aligns the addresses.

In this figure we show up to 10 IP addresses in the route, but the first listed address is removed from the source route option and becomes the destination address of the IP datagram when it leaves the source host. Although there is room for only nine IP addresses in the 40-byte IP option space (do not forget the 3-byte option header that we are about to describe), there are actually 10 IP addresses in an IPv4 header, when the destination address is included.

The *code* is either `0x83` for an LSRR option or `0x89` for an SSRR option. The *len* that we specify is the size of the option in bytes, including the 3-byte header, and including the extra destination address at the end. It will be 11 for a route consisting of one IP address, 15 for a route consisting of two IP addresses, and so on, up to a maximum of 43. The NOP is not part of the option and is not included in the *len* field but is included in the size of the buffer that we specify to `setsockopt`. When the first address in the list is removed from the source route option and placed into the destination address field of the IP header, this *len* value is decremented by 4 (Figures 9.32 and 9.33 of TCPv2). *ptr* is a pointer or offset of the next IP address to be processed in the route, and we initialize it to 4, which points to the first IP address. The value of this field increases by 4 as the datagram is processed by each listed node.

We now develop three functions to initialize, create, and process a source route option. Our functions handle only a source route option. While it is possible to combine a source route with other IP options (such as a timestamp), the options other than the two source route options are rarely used. Figure 24.2 is the first function `inet_srcrt_init` along with some static variables that are used as an option is being built.

ipopts/sourceroute.c

```
 1 #include    "unp.h"
 2 #include    <netinet/in_systm.h>
 3 #include    <netinet/ip.h>

 4 static u_char *optr;           /* pointer into options being formed */
 5 static u_char *lenptr;         /* pointer to length byte in SRR option */
 6 static int ocnt;               /* count of # addresses */

 7 u_char *
 8 inet_srcrt_init(void)
 9 {
10     optr = Malloc(44);        /* NOP, code, len, ptr, up to 10 addresses */
11     bzero(optr, 44);          /* guarantees EOLs at end */
12     ocnt = 0;
13     return (optr);            /* pointer for setsockopt() */
14 }
```

ipopts/sourceroute.c

Figure 24.2 `inet_srcrt_init` function: initialize before storing a source route.

Initialize

10–13 We allocate a maximum sized buffer of 44 bytes and set it to 0. The value of the EOL option is 0, so this initializes the entire option to EOL bytes. The pointer to the

option is returned to the caller and will be passed as the fourth argument to `setsockopt`.

The next function, `inet_srcrt_add`, adds one IPv4 address to the source route being constructed.

ipopts/sourceroute.c

```
15 int
16 inet_srcrt_add(char *hostptr, int type)
17 {
18     int     len;
19     struct addrinfo *ai;
20     struct sockaddr_in *sin;

21     if (ocnt > 9)
22         err_quit("too many source routes with: %s", hostptr);

23     if (ocnt == 0) {
24         *optr++ = IPOPT_NOP;     /* NOP for alignment */
25         *optr++ = type ? IPOPT_SSRR : IPOPT_LSRR;
26         lenptr = optr++;         /* we fill in the length later */
27         *optr++ = 4;             /* offset to first address */
28     }
29     ai = Host_serv(hostptr, "", AF_INET, 0);
30     sin = (struct sockaddr_in *) ai->ai_addr;
31     memcpy(optr, &sin->sin_addr, sizeof(struct in_addr));
32     freeaddrinfo(ai);

33     optr += sizeof(struct in_addr);
34     ocnt++;
35     len = 3 + (ocnt * sizeof(struct in_addr));
36     *lenptr = len;
37     return (len + 1);            /* size for setsockopt() */
38 }
```

ipopts/sourceroute.c

Figure 24.3 `inet_srcrt_add` function: add one IPv4 address to a source route.

Arguments

15-16 The first argument points to either a hostname or a dotted-decimal IP address, and the second argument is 0 for a loose source route or nonzero for a strict source route. We will see that the type of the first address added to the route determines whether the route is loose or strict.

Check for overflow and then initialize

21-28 We check that too many addresses are not specified and then initialize if this is the first address. As we mentioned, we always place a NOP before the source route option. We save a pointer to the *len* field and will store this value as each address is added to the list.

Obtain binary IP address and store in route

29-37 Our `host_serv` function handles either a hostname or a dotted-decimal string and we store the resulting binary address in the list. We update the *len* field and return the total size of the buffer (including the NOP) that the caller must pass to `setsockopt`.

When a received source route is returned to the application by `getsockopt`, the format is different from Figure 24.1. We show the received format in Figure 24.4.

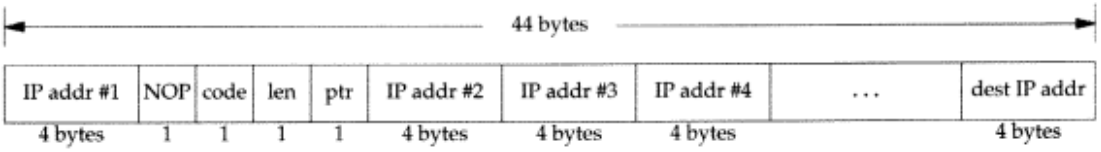


Figure 24.4 Format of source route option returned by `getsockopt`.

First, the order of the addresses has been reversed by the kernel from the ordering in the received source route. What we mean by "reversed" is that if the received source route contains the four addresses A, B, C, and D, in that order, the reverse of this route is D, C, B, and then A. The first 4 bytes contain the first IP address in the list, followed by a 1-byte NOP (for alignment), followed by the 3-byte source route option header, followed by the remaining IP addresses. Up to nine IP addresses can follow the 3-byte header, and the *len* field in the returned header will have a maximum value of 39. Since the NOP is always present, the length returned by `getsockopt` will always be a multiple of 4 bytes.

The format shown in Figure 24.4 is defined in `<netinet/ip_var.h>` as the following structure:

```
#define MAX_IPOPTLEN    40

struct ipoption {
  struct in_addr ipopt_dst;  /* first-hop dst if source routed */
  char           ipopt_list[MAX_IPOPTLEN];   /* options proper */
};
```

In Figure 24.5 we find it just as easy to parse the data ourselves, instead of using this structure.

This returned format differs from the format that we pass to `setsockopt`. If we wanted to convert the format in Figure 24.4 to the format in Figure 24.1 we would have to swap the first 4 bytes with the following 4 bytes and add four to the length field. Fortunately we do not have to do this, as Berkeley-derived implementations automatically use the reverse of a received source route for a TCP socket. That is, the information shown in Figure 24.4 is returned by `getsockopt` for our information only. We do not have to call `setsockopt` to tell the kernel to use this route for IP datagrams sent on the TCP connection—the kernel does that automatically. We will see an example of this shortly with our TCP server.

The next of our source route functions takes a received source route, in the format shown in Figure 24.4, and prints the information. We show our `inet_srcrt_print` function in Figure 24.5.

Save first IP address, skip any NOPs

45-47 The first IP address in the buffer is saved and any NOPs that follow are skipped over.

ipopts/sourceroute.c

```
39 void
40 inet_srcrt_print(u_char *ptr, int len)
41 {
42     u_char  c;
43     char    str[INET_ADDRSTRLEN];
44     struct in_addr hop1;

45     memcpy(&hop1, ptr, sizeof(struct in_addr));
46     ptr += sizeof(struct in_addr);

47     while ( (c = *ptr++) == IPOPT_NOP) ;      /* skip any leading NOPs */

48     if (c == IPOPT_LSRR)
49         printf("received LSRR: ");
50     else if (c == IPOPT_SSRR)
51         printf("received SSRR: ");
52     else {
53         printf("received option type %d\n", c);
54         return;
55     }
56     printf("%s ", Inet_ntop(AF_INET, &hop1, str, sizeof(str)));

57     len = *ptr++ - sizeof(struct in_addr);  /* subtract dest IP addr */
58     ptr++;                          /* skip over pointer */
59     while (len > 0) {
60         printf("%s ", Inet_ntop(AF_INET, ptr, str, sizeof(str)));
61         ptr += sizeof(struct in_addr);
62         len -= sizeof(struct in_addr);
63     }
64     printf("\n");
65 }
```

ipopts/sourceroute.c

Figure 24.5 `inet_srcrt_print` function: print a received source route.

Check for source route option

48–64 We only print the information for a source route, and from the 3-byte header we check the *code*, fetch the *len*, and skip over the *ptr*. We then print all the IP addresses that follow the 3-byte header, except the destination IP address.

Example

We now modify our TCP echo client to specify a source route and our TCP echo server to print a received source route. Figure 24.6 is our client.

Process command-line arguments

8–23 We call our `inet_srcrt_init` function to initialize the source route. Each hop is specified by either a `-g` option (loose) or a `-G` option (strict). The type of the first IP address (loose or strict) specifies the type of the source route. (We do this for simplicity. Clearly we could add code to check that all are of the same type.) Our `inet_srcrt_add` function adds each address to the route.

ipopts/tcpcli01.c

```
 1 #include    "unp.h"
 2 int
 3 main(int argc, char **argv)
 4 {
 5     int     c, sockfd, len = 0;
 6     u_char *ptr;
 7     struct addrinfo *ai;
 8     if (argc < 2)
 9         err_quit("usage: tcpcli01 [ -[gG] <hostname> ... ] <hostname>");
10     ptr = inet_srcrt_init();
11     opterr = 0;                 /* don't want getopt() writing to stderr */
12     while ( (c = getopt(argc, argv, "g:G:")) != -1) {
13         switch (c) {
14         case 'g':               /* loose source route */
15             len = inet_srcrt_add(optarg, 0);
16             break;
17         case 'G':               /* strict source route */
18             len = inet_srcrt_add(optarg, 1);
19             break;
20         case '?':
21             err_quit("unrecognized option: %c", c);
22         }
23     }
24     if (optind != argc - 1)
25         err_quit("missing <hostname>");
26     ai = Host_serv(argv[optind], SERV_PORT_STR, AF_INET, SOCK_STREAM);
27     sockfd = Socket(ai->ai_family, ai->ai_socktype, ai->ai_protocol);
28     if (len > 0) {
29         len = inet_srcrt_add(argv[optind], 0);  /* dest at end */
30         Setsockopt(sockfd, IPPROTO_IP, IP_OPTIONS, ptr, len);
31         free(ptr);
32     }
33     Connect(sockfd, ai->ai_addr, ai->ai_addrlen);
34     str_cli(stdin, sockfd);     /* do it all */
35     exit(0);
36 }
```

ipopts/tcpcli01.c

Figure 24.6 TCP echo client that specifies a source route.

This is our first encounter with the Posix.2 `getopt` function. The third argument is a character string specifying the characters that we allow as command-line arguments, `g` and `G` in this example. Each is followed by a colon, indicating that the option takes an argument. This function works with four global variables that are defined by including `<unistd.h>`:

```
extern char  *optarg;
extern int    optind, opterr, optopt;
```

Before calling `getopt` we set `opterr` to 0 to prevent the function from writing error messages to standard error in the case of an error, because we handle these ourself. Posix.2 states that if the first character of the third argument is a colon, this also prevents the function from writing to standard error, but not all implementations support this.

Handle destination address and create socket

24-27 The final command-line argument is the hostname or dotted-decimal address of the server and our `host_serv` function processes it. We are not able to call our `tcp_connect` function, because we must specify the source route between the calls to `socket` and `connect`. The latter initiates the three-way handshake and we want the initial SYN and all subsequent packets to use this source route.

28-34 If a source route is specified, we must add the server's IP address to the end of the list of IP addresses (Figure 24.1). `setsockopt` installs the source route for this socket. We then call `connect`, followed by our `str_cli` function (Figure 5.5).

Our TCP server is almost identical to the code shown in Figure 5.12, with the following changes. First, we allocate space for the options:

```
int      len;
u_char   *opts;

opts = Malloc(44);
```

We then fetch the IP options after the call to `accept`, but before the call to `fork`:

```
len = 44;
Getsockopt(connfd, IPPROTO_IP, IP_OPTIONS, opts, &len);
if (len > 0) {
    printf("received IP options, len = %d\n", len);
    inet_srcrt_print(opts, len);
}
```

If the received SYN from the client does not contain any IP options, the `len` variable will contain 0 on return from `getsockopt` (it is a value–result argument). As mentioned earlier, we do not have to do anything to cause TCP to use the reverse of the received source route: that is done automatically by TCP (p. 931 of TCPv2). All we are doing by calling `getsockopt` is obtaining a copy of the reversed source route. If we do not want TCP to use this route, we call `setsockopt` after `accept` returns, specifying a fifth argument (the length) of 0, and this removes any IP options currently in use. The source route has already been used by TCP for the second segment of the three-way handshake (Figure 2.5) but if we remove the options, IP will use whatever route it calculates for future packets to this client.

We now show an example of our client–server when we specify a source route. We run our client on the host `solaris` as follows:

```
solaris % tcpcli01 -g gw -g sunos5 bsdi
```

This sends the IP datagrams from `solaris` to the router `gw` (Figure 1.16), to the host `sunos5`, and then to the host `bsdi`, which is running the server. The two intermediate

systems, gw and sunos5, must forward source routed datagrams for this example to work.

When the connection is established at the server, it outputs the following:

```
bsdi % tcpserv01
received IP options, len = 16
received LSRR: 206.62.226.36 206.62.226.62 206.62.226.33
```

The first IP address printed is the first hop of the reverse path (sunos5, as shown in Figure 24.4), and the next two addresses are in the order that is used by the server to send datagrams back to the client. If we watch the client–server exchange using tcpdump, we can see the source route option on every datagram in both directions.

> Unfortunately the operation of the IP_OPTIONS socket option has never been documented, so you may encounter variations on systems that are not derived from the Berkeley source code. For example, under Solaris 2.5 the first address returned in the buffer by getsockopt (Figure 24.4) is not the first-hop address for the return route, but the address of the peer. Nevertheless, the reversed route used by TCP is correct. Also, Solaris 2.5 precedes all source route options with four NOPs, limiting the option to eight IP addresses, instead of the real limit of nine.

Deleting A Received Source Route

Unfortunately source routes present a security hole. Starting with the Net/1 release (1989), the rlogind and rshd servers had code similar to the following:

```
u_char  buf[44];
char    lbuf[BUFSIZ];
int     optsize;

optsize = sizeof(buf);
if (getsockopt(0, IPPROTO_IP, IP_OPTIONS,
               buf, &optsize) == 0 && optsize != 0) {
    /* format the options as hex numbers to print in lbuf[] */
    syslog(LOG_NOTICE,
           "Connection received using IP options (ignored):%s", lbuf);
    setsockopt(0, ipproto, IP_OPTIONS, NULL, 0);
}
```

If a connection arrives with any IP options (the value of optsize returned by getsockopt is nonzero), then a message is logged using syslog and setsockopt is called to clear the options. This prevents any future TCP segments sent on this connection from using the reverse of the received source route. This technique is now known to be inadequate, because by the time the application receives the connection, the TCP three-way handshake is complete, and the second segment (the server's SYN-ACK in Figure 2.5) has already followed the reverse of the source route back to the client (or at least to one of the intermediate hops listed in the source route, which is where the hacker is located). Since the hacker has seen TCP's sequence numbers in both directions, even if no more packets are sent with the source route, the hacker can still send packets to the server, with the correct sequence number.

The only solution for this potential problem is to forbid all TCP connections that arrive with a source route, when you are using the source IP address for some form of validation (as do `rlogind` and `rshd`). Replace the call to `setsockopt` in the code fragment just shown with a closing of the just-accepted connection and a termination of the newly spawned server. The second segment of the three-way handshake has already been sent, but the connection should not be left open.

24.4 IPv6 Extension Headers

We do not show any options with the IPv6 header in Figure A.2 (it is always 40 bytes in length) but an IPv6 header can be followed by optional *extension headers*.

1. Hop-by-hop options must immediately follow the 40-byte IPv6 header. There are no hop-by-hop options currently defined that are usable by an application.
2. Destination options. None are currently defined that are usable by an application.
3. Routing header. This is a source routing option, similar in concept to what we described for IPv4 in Section 24.3.
4. Fragmentation header. This header is automatically generated by a host that fragments an IPv6 datagram and then processed by the final destination when it reassembles the fragments.
5. Authentication header (AH). The use of this header is documented in RFC 1826 [Atkinson 1995a] and [Kent and Atkinson 1997a].
6. Encapsulating security payload (ESP) header. The use of this header is documented in RFC 1827 [Atkinson 1995b] and [Kent and Atkinson 1997b].

We said the fragmentation header is handled entirely by the kernel, and a proposal for socket options to handle the AH and ESP headers is in [McDonald 1997]. This leaves the first three options, which we discuss in the next two sections.

24.5 IPv6 Hop-by-Hop Options and Destination Options

The hop-by-hop and destination options have a similar format, shown in Figure 24.7. The 8-bit *next header* field identifies the next header that follows this extension header. The 8-bit *header extension length* is the length of this extension header, in units of 8 bytes, but not including the first 8 bytes. For example, if this extension header occupies 8 bytes, then its header extension length is 0. If this extension header occupies 16 bytes, then its header extension length is 1, and so on. These two headers are padded to be a multiple of 8 bytes with either the `pad1` option, or with the `padN` option, described shortly.

The hop-by-hop options header and the destination options header each hold any number of individual options, which have the format shown in Figure 24.8.

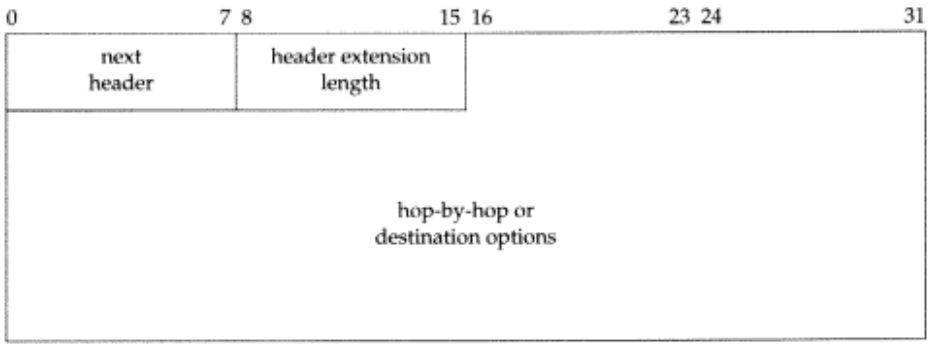


Figure 24.7 Format of hop-by-hop and destinations options.

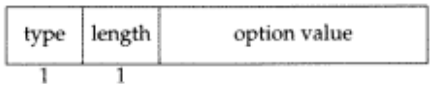


Figure 24.8 Format of individual hop-by-hop and destination options.

This is called *TLV coding* because each option appears with its type, length, and value. The 8-bit *type* field identifies the option type. Additionally the two high-order bits specify what an IPv6 node does with this option if it does not understand the option:

- `00` Skip over this option and continue processing the header.
- `01` Discard the packet.
- `10` Discard the packet and send an ICMP parameter problem type 2 error (Figure A.16) to the sender, regardless of whether or not the packet's destination is a multicast address.
- `11` Discard the packet and send an ICMP parameter problem type 2 error (Figure A.16) to the sender. But the error is sent only if the packet's destination is not a multicast address.

The next high-order bit specifies whether or not the option data changes en route:

- `0` The option data does not change en route.
- `1` The option data may change en route.

The low-order 5 bits then specify the option.

The 8-bit *length* field specifies the length of the option data in bytes. The type field and this length field are not included in this length.

The two pad options are defined in RFC 1883 [Deering and Hinden 1995] and can be used in either the hop-by-hop options header or in the destination options header. The *jumbo payload length*, a hop-by-hop option, is also defined in RFC 1883, and it is generated when needed, and processed when received entirely by the kernel. A new hop-

by-hop option is proposed for IPv6 in [Katz et al. 1997], similar to the IPv4 router alert. We show these in Figure 24.9.

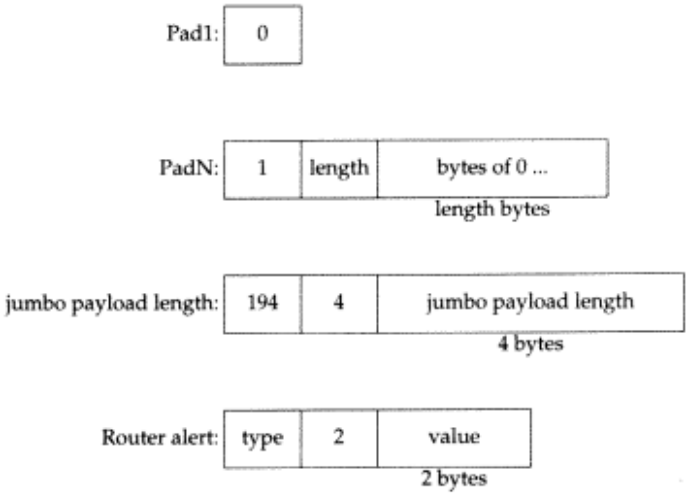


Figure 24.9 IPv6 hop-by-hop options.

The `pad1` byte is the only option without a length and value. It provides 1 byte of padding. The `padN` option is used when 2 or more bytes of padding are required. For 2 bytes of padding, the length of this option would be 0 and the option would consist of just the type field and the length field. For 3 bytes of padding, the length would be 1, and 1 byte of 0 would follow this length. The jumbo payload length option provides a datagram length of 32 bits and is used when the 16-bit payload length field in Figure A.2 is inadequate.

We show these options because each hop-by-hop and destination option also has an associated *alignment requirement*, written as $xn + y$. This means that the option must appear at an integer multiple of x bytes from the start of the header, plus y bytes. For example, the alignment requirement of the jumbo payload option is $4n + 2$, and this is to force the 4-byte option value (the jumbo payload length) to be on a 4-byte boundary. The reason that the y value is 2 for this option is because of the 2 bytes that appear at the beginning of each hop-by-hop and destination options header (Figure 24.8).

The hop-by-hop options and the destination options are normally specified as ancillary data with `sendmsg` and returned as ancillary data by `recvmsg`. Nothing special need be done by the application to send either or both of these options; just specify them in a call to `sendmsg`. But to receive these options, the corresponding socket option must be enabled: `IPV6_HOPOPTS` for the hop-by-hop options, and `IPV6_DSTOPTS` for the destination options. For example, to enable both options to be returned:

```
const int  on = 1;

setsockopt(sockfd, IPPROTO_IPV6, IPV6_HOPOPTS, &on, sizeof(on));
setsockopt(sockfd, IPPROTO_IPV6, IPV6_DSTOPTS, &on, sizeof(on));
```

Figure 24.10 shows the format of the ancillary data objects used to send and receive hop-by-hop and destination options.

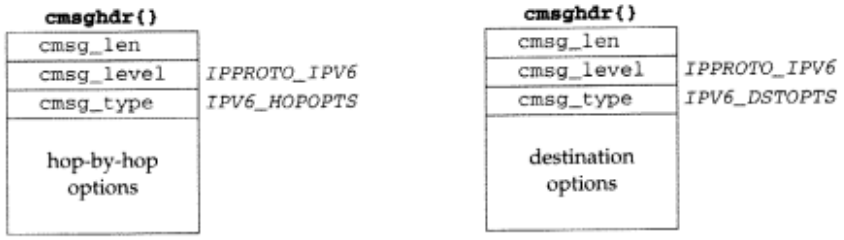


Figure 24.10 Ancillary data objects for hop-by-hop and destination options.

Unlike the other IPv6 ancillary data objects that we have described (Figure 20.21) it is up to each implementation as to what is passed between the user and the kernel as the `cmsg_data` portion of these objects. Instead of defining these contents, six functions are defined to create and process these ancillary data objects. The following four functions build an option to send.

```
#include <netinet/in.h>

int inet6_option_space(int nbytes);
```

Returns: number of bytes required to hold option

```
int inet6_option_init(void *buf, struct cmsghdr **cmsgp, int type);
```

Returns: 0 if OK, –1 on error

```
int inet6_option_append(struct cmsghdr *cmsg, const uint8_t *typep,
                        int multx, int plusy);
```

Returns: 0 if OK, –1 on error

```
uint8_t *inet6_option_alloc(struct cmsghdr *cmsg, int datalen,
                            int multx, int plusy);
```

Returns: pointer to option type field if OK, NULL on error

`inet6_option_space` returns the number of bytes required to hold an option, including the `cmsghdr` structure at the beginning and any pad bytes at the end. The *nbytes* argument is the size of the structure defining the option, which must include any pad bytes at the beginning (the value y in the alignment term $xn + y$), the option type, length, and data.

`inet6_option_init` is called once per ancillary data object that will contain either a hop-by-hop option or a destination option. *buf* points to the buffer that will contain the ancillary data object. *cmsgp* is the address of a pointer to a `cmsghdr` structure, and this function initializes this structure in the buffer pointed to by `buf` and returns a pointer to this structure in **cmsgp*. *type* is either `IPV6_HOPOPTS` or

`IPV6_DSTOPTS` and is stored in the `cmsg_type` member of the `cmsghdr` structure that is built.

`inet6_option_append` appends either a hop-by-hop option or a destination option into an ancillary data object that was initialized by `inet6_option_init`. *cmsg* is a pointer to the `cmsghdr` structure that was initialized by `inet6_option_init`. *typep* is a pointer to the 8-bit option type, which must be followed by the 8-bit option length, and the option data (TLV). The caller sets these values before calling this function. *multx* and *plusy* are the two terms x and y in the alignment requirement for this option.

The previously described function requires the caller to build the option's TLV and then pass a pointer as the *typep* argument and the option is then copied into the ancillary data object. Alternately, the `inet6_option_alloc` function returns a pointer to the ancillary data object, and the caller must then store the options TLV into the ancillary data object. *cmsg* is a pointer to the `cmsghdr` structure that was initialized by `inet6_option_init`. *datalen* is the value of the length byte for this option, and this is required to calculate if any padding must be appended to the option. *multx* and *plusy* are the two terms x and y in the alignment requirement for this option.

The remaining two functions process a received option.

```
#include <netinet/in.h>

int inet6_option_next(const struct cmsghdr *cmsg, uint8_t **tptrp);
```

Returns: 0 if another option to process, –1 on end of options or error

```
int inet6_option_find(const struct cmsghdr *cmsg, uint8_t *tptrp, int type);
```

Returns: 0 if OK, –1 on error

`inet6_option_next` processes the next option in a buffer. *cmsghdr* points to a `cmsghdr` structure of which the `cmsg_level` must be `IPPROTO_IPV6` and the `cmsg_type` must be either `IPV6_HOPOPTS` or `IPV6_DSTOPTS`. The first time this function is called for a given ancillary data object, **tptrp* must be a null pointer and then each time this function returns, **tptrp* points to the 8-bit option type field of the next option to process. The value of **tptrp* is used by the function to remember its place in the ancillary data object from one call to the next. When the last option has been processed, the return value is –1 and **tptrp* is a null pointer. If an error occurs, the return value is –1 and **tptrp* is a nonnull pointer.

`inet6_option_find` is similar to the previous function, but it lets the caller specify the option type to search for (the *type* argument) instead of always returning the next option.

24.6 IPv6 Routing Header

The IPv6 routing header is used for source routing in IPv6. The first 2 bytes of the routing header are the same as we showed in Figure 24.7: a *next header* field followed by a

header extension length. The next 2 bytes specify the *routing type* and the number of *segments left* (i.e., how many listed nodes are still to be visited). Only one type of routing header is specified, type 0, and we show its format in Figure 24.11.

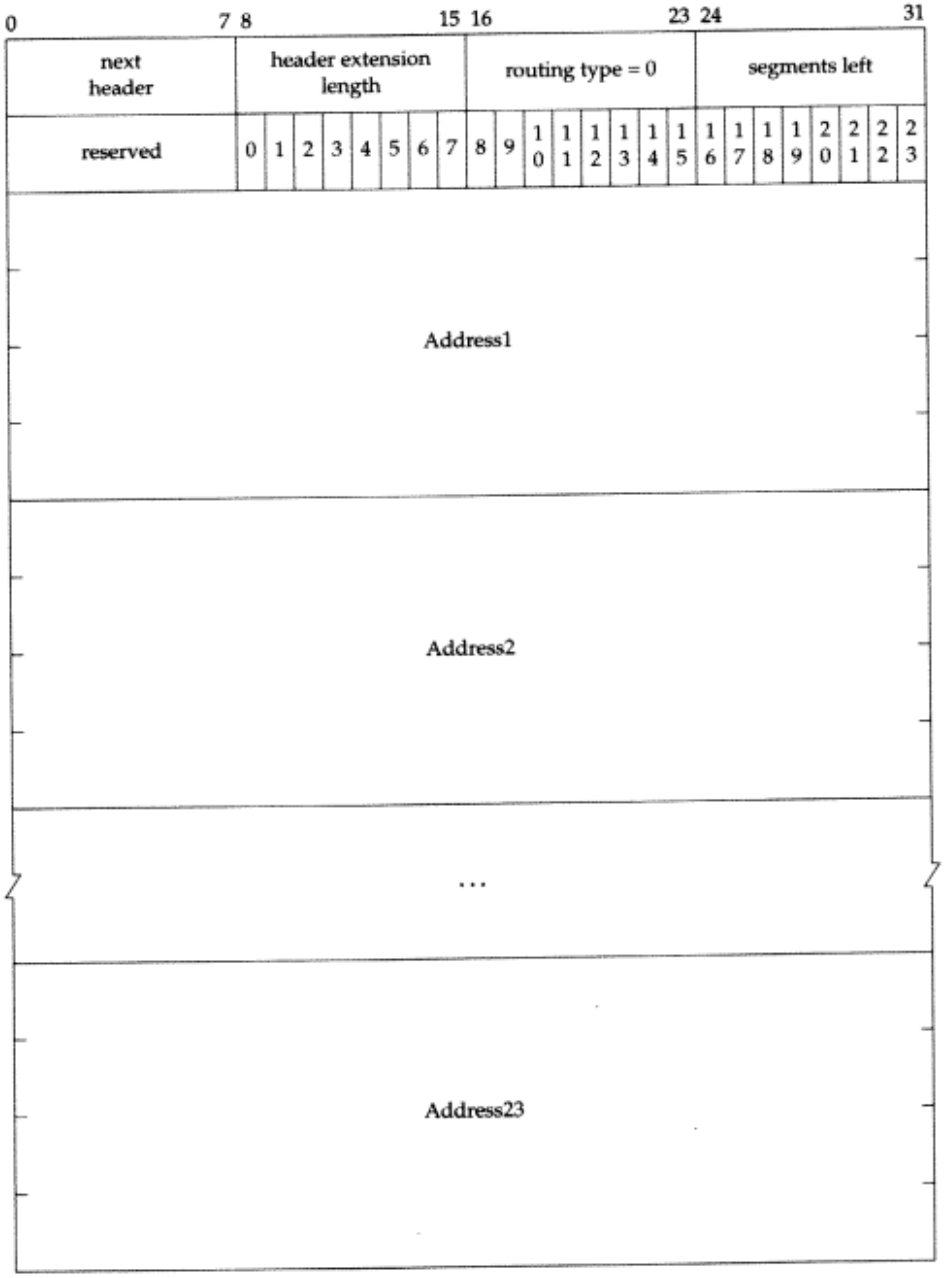


Figure 24.11 IPv6 routing header.

Up to 23 addresses can appear in the routing header and *segments left* is between 1 and 23. The 24 bits that we label from 0 to 23 form the *strict/loose bit map*: a value of 1 means the corresponding address is a *strict* hop (that node must be a neighbor of the previously listed node) and a value of 0 means the corresponding address is a *loose* hop (that node need not be a neighbor). The bit labeled 1 corresponds to Address1, the bit labeled 2 correspond to Address 2, and so on. The bit labeled 0 is for the last hop. RFC 1883 [Deering and Hinden 1995] specifies the details of how the header is processed as the packet travels to the final destination, along with a detailed example.

The routing header is normally specified as ancillary data with `sendmsg` and returned as ancillary data by `recvmsg`. Nothing special need be done by the application to send the header: just specify it in a call to `sendmsg`. But to receive the routing header, the `IPV6_RTHDR` socket option must be enabled, as in

```
const int  on = 1;

setsockopt(sockfd, IPPROTO_IPV6, IPV6_RTHDR, &on, sizeof(on));
```

Figure 24.12 shows the format of the ancillary data object used to send and receive the routing header. Similar to the ancillary data objects for the hop-by-hop and destination options (Figure 24.10), it is up to each implementation as to what is passed between the user and the kernel as the `cmsg_data` portion of this object. Eight functions are defined to create and process the routing header. The following four functions build an option to send.

```
#include <netinet/in.h>

size_t inet6_rthdr_space(int type, int segments);
```

Returns: positive number of bytes if OK, 0 on error

```
struct cmsghdr *inet6_rthdr_init(void *buf, int type);
```

Returns: nonnull pointer if OK, NULL on error

```
int inet6_rthdr_add(struct cmsghdr *cmsg, const struct in6_addr *addr,
                    unsigned int flags);
```

Returns: 0 if OK, −1 on error

```
int inet6_rthdr_lasthop(struct cmsghdr *cmsg, unsigned int flags);
```

Returns: 0 if OK, −1 on error

`inet6_rthdr_space` returns the number of bytes required to hold an ancillary data object containing a routing header of the specified *type* (normally specified as `IPV6_RTHDR_TYPE_0`) with the specified number of *segments*. This size includes the `cmsghdr` structure.

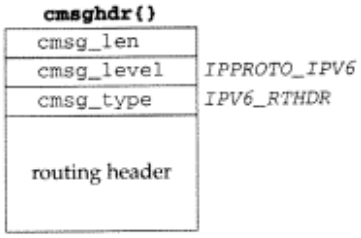


Figure 24.12 Ancillary data object for IPv6 routing header.

`inet6_rthdr_space` and `inet6_option_space` both return the number of bytes required for one type of ancillary data object but neither allocates the memory. This is in case the caller wants to allocate a larger buffer to hold other ancillary data objects too.

`inet6_rthdr_init` initializes the buffer pointed to by *buf* to contain an ancillary data object with a routing header of the specified *type*. The return value is the pointer to the `cmsghdr` structure that is built in the buffer, and this pointer is then used as an argument to the next two functions. The `cmsg_level` and `cmsg_type` members are initialized.

`inet6_rthdr_add` adds the IPv6 address pointed to by `addr` to the end of the routing header being constructed. The *flags* argument is either `IPV6_RTHDR_LOOSE` or `IPV6_RTHDR_STRICT`. Upon success the `cmsg_len` member is updated to account for the new address.

`inet6_rthdr_lasthop` specifies the *flags* (`IPV6_RTHDR_LOOSE` or `IPV6_RTHDR_STRICT`) for the final hop. Realize that a routing header with N addresses has $N + 1$ hops. This requires N calls to `inet6_rthdr_add` and one call to `inet6_rthdr_lasthop`.

The following four functions deal with a received routing header.

```
#include <netinet/in.h>

int inet6_rthdr_reverse(const struct cmsghdr *in, struct cmsghdr *out);
```

Returns: 0 if OK, –1 on error

```
int inet6_rthdr_segments(const struct cmsghdr *cmsg);
```

Returns: number of segments in routing header if OK, –1 on error

```
struct in6_addr *inet6_rthdr_getaddr(struct cmsghdr *cmsg, int index);
```

Returns: nonnull pointer if OK, NULL on error

```
int inet6_rthdr_getflags(const struct cmsghdr *cmsg, int index);
```

Returns: loose/strict flag if OK, –1 on error

`inet6_rthdr_reverse` takes a routing header that was received as ancillary data (pointed to by *in*) and creates a new routing header (in the buffer pointed to by *out*) that sends datagrams along the reverse of that path. The reversal can occur in place; that is, the *in* and *out* pointers can point to the same buffer.

`inet6_rthdr_segments` returns the number of segments in the routing header described by *cmsg*. Upon success, the return value is between 1 and 23, inclusive.

`inet6_rthdr_getaddr` returns a pointer to the IPv6 address specified by *index* in the routing header described by *cmsg*. *index* must have a value between 1 and the value returned by `inet6_rthdr_segments`, inclusive.

`inet6_rthdr_getflags` returns `IPV6_RTHDR_LOOSE` or `IPV6_RTHDR_STRICT` corresponding to *index* (which must have a value between 0 and the value returned by `inet6_rthdr_segments`, inclusive) in the routing header described by *cmsg* for the routing header

> Addresses are indexed starting at 1, and the loose/strict flags are indexed starting at 0, as we show in Figure 24.11. This is consistent with the notation in RFC 1883 [Deering and Hinden 1995].

24.7 IPv6 Sticky Options

We have described the use of ancillary data with `sendmsg` and `recvmsg` to send and receive six different ancillary data objects:

1. IPv6 packet information: the `in6_pktinfo` structure containing either the destination address and outgoing interface index, or the source address and the arriving interface index (Figure 20.21).
2. The outgoing hop limit or the received hop limit (Figure 20.21).
3. The next-hop address (Figure 20.21).
4. Hop-by-hop options (Figure 24.10).
5. Destination options (Figure 24.10).
6. Routing header (Figure 24.12).

We summarized the `cmsg_level` and `cmsg_type` values for these objects, along with the values for the other ancillary data object in Figure 13.11.

Instead of sending these options in every call to `sendmsg`, we can set the `IPV6_PKTOPTIONS` socket option instead. When setting this option the fourth argument points to a buffer containing all the ancillary data objects that should be sent with all packets on this socket. The format of this buffer is exactly as if the buffer were specified as ancillary data to `sendmsg`. These options are called "sticky" because once they are set, they remain set until cleared (setting the socket option with a length of 0). But these sticky options can be overridden on a per-packet basis for a UDP socket or for a raw IPv6 socket by specifying ancillary data in a call to `sendmsg`. If any ancillary data is specified in a call to `sendmsg`, none of the sticky options is sent with that datagram.

The concept of sticky options can also be used with TCP, because ancillary data is never sent or received by `sendmsg` or `recvmsg` on a TCP socket. Instead, a TCP application can set the `IPV6_PKTOPTIONS` socket option and specify any of the six ancillary data objects mentioned at the beginning of this section. These objects then affect all packets sent on this socket.

A TCP application can also call `getsockopt` for the `IPV6_PKTOPTIONS` socket option to retrieve these ancillary data objects. In this case the kernel maintains only the options from the most recently received segment and for those options that the application has explicitly said that it wants (by enabling the corresponding socket option).

24.8 Summary

The most commonly used of the 10 defined IPv4 options is the source route, but its use is dwindling these days because of security concerns. Access to the IPv4 header options is through the `IP_OPTIONS` socket option.

IPv6 defines six extension headers, although as of this writing support for these options is minimal. Access to the IPv6 extension headers is through a functional interface, obviating the need to understand their actual format in the packet. These extension headers are written as ancillary data with `sendmsg` and returned as ancillary data with `recvmsg`.

Exercises

24.1 In our IPv4 source route example at the end of Section 24.3 what changes at the server if instead of specifying the hostname `gw` to the client, we specify the other IP address of this router: 206.85.40.74 from Figure 1.16?

24.2 In our IPv4 source route example at the end of Section 24.3 what changes if we specify each intermediate node to the client with the `-G` option, instead of the `-g` option?

24.3 The length of the buffer specified to `setsockopt` for the `IP_OPTIONS` socket option must be a multiple of 4 bytes. What would we do if we did not place a NOP at the beginning of the buffer, as shown in Figure 24.1?

24.4 How does `ping` receive a source route when the IP Record Route option is used (described in Section 7.3 of TCPv1)?

24.5 In the example code from the `rlogind` server at the end of Section 24.3 that clears a received source route, why is the socket descriptor argument for `getsockopt` and `setsockopt` 0?

24.6 For many years the code that we showed at the end of Section 24.3 that clears a received source route looked like:

```
optsize = 0;
setsockopt(0, ipproto, IP_OPTIONS, NULL, &optsize);
```

What is wrong with this code? Does it matter?

25

Raw Sockets

25.1 Introduction

Raw sockets provide three features not provided by normal TCP and UDP sockets.

1. Raw sockets let us read and write ICMPv4, IGMPv4, and ICMPv6 packets. The Ping program, for example, sends ICMP echo requests and receives ICMP echo replies. (We develop our own version of the Ping program in Section 25.5.) The multicast routing daemon, `mrouted`, sends and receives IGMPv4 packets.

   This capability also allows applications that are built using ICMP or IGMP to be handled entirely as user processes, instead of putting more code into the kernel. The router discovery daemon (`in.rdisc` under Solaris 2.x; Appendix F of TCPv1 describes how to obtain the source code for a publicly available version), for example, is built this way. It processes two ICMP messages (router advertisement and router solicitation) that the kernel knows nothing about.

2. With a raw socket a process can read and write IPv4 datagrams with an IPv4 protocol field that is not processed by the kernel. Recall the 8-bit IPv4 protocol field in Figure A.1. Most kernels only process datagrams containing values of 1 (ICMP), 2 (IGMP), 6 (TCP), and 17 (UDP). But many other values are defined for the protocol field: RFC 1700 [Reynolds and Postel 1994] lists all the values. For example, the OSPF routing protocol does not use TCP or UDP but uses IP directly, setting the protocol field of the IP datagram to 89. The `gated` program that implements OSPF must use a raw socket to read and write these IP datagrams since they contain a protocol field that the kernel knows nothing about. This capability will carry over to IPv6 also.

3. With a raw socket a process can build its own IPv4 header, using the `IP_HDRINCL` socket option. This can be used, for example, to build our own UDP or TCP packets, and we show an example of this in Section 26.6.

This chapter describes raw socket creation, input, and output. We then develop versions of the Ping and Traceroute programs that work with both IPv4 and IPv6.

25.2 Raw Socket Creation

The steps involved in creating a raw socket are as follows:

1. The `socket` function creates a raw socket when the second argument is `SOCK_RAW`. The third argument (the protocol) is normally nonzero. For example, to create an IPv4 raw socket we would write

```
int     sockfd;

sockfd = socket(AF_INET, SOCK_RAW, protocol);
```

   where *protocol* is one of the constants `IPPROTO_`*xxx* defined by including the `<netinet/in.h>` header, such as `IPPROTO_ICMP`. Be aware that just because a protocol has its name defined in this header, such as `IPPROTO_EGP`, does not mean that the kernel supports it.

   Only the superuser can create a raw socket. This prevents normal users from writing their own IP datagrams to the network.

2. The `IP_HDRINCL` socket option can be set:

```
const int  on = 1;

if (setsockopt(sockfd, IPPROTO_IP, IP_HDRINCL, &on, sizeof(on)) < 0)
    error
```

   We describe the effect of this socket option in the next section.

3. `bind` can be called on the raw socket, but this is rare. This function sets only the local address: there is no concept of a port number with a raw socket. With regard to output, calling `bind` sets the source IP address that will be used for datagrams sent on the raw socket (but only if the `IP_HDRINCL` socket option is not set). If `bind` is not called, the kernel sets the source IP address to the primary IP address of the outgoing interface.

4. `connect` can be called on the raw socket, but this is rare. This function sets only the foreign address: again, there is no concept of a port number with a raw socket. With regard to output, calling `connect` lets us call `write` or `send` instead of `sendto`, since the destination IP address is already specified.

25.3 Raw Socket Output

Output on a raw socket is governed by the following rules:

1. Normal output is performed by calling `sendto` or `sendmsg` and specifying the destination IP address. `write`, `writev`, or `send` can also be called if the socket has been connected.
2. If the `IP_HDRINCL` option is not set, the starting address of the data for the kernel to write specifies the first byte following the IP header, because the kernel will build the IP header and prepend it to the data from the process. The kernel sets the protocol field of the IPv4 header that it builds to the third argument from the call to `socket`.
3. If the `IP_HDRINCL` option is set, the starting address of the data for the kernel to write specifies the first byte of the IP header. The amount of data to write must include the size of the caller's IP header. The process builds the entire IP header, except (a) the IPv4 identification field can be set to 0, which tells the kernel to set this value, and (b) the kernel always calculates and stores the IPv4 header checksum.
4. The kernel fragments raw packets that exceed the outgoing interface MTU.

> Unfortunately the `IP_HDRINCL` socket option has never been documented, specifically with regard to the byte ordering of the fields in the IP header. On Berkeley-derived kernels all fields are in network byte order except `ip_len` and `ip_off`, which are in host byte order (pp. 233 and 1057 of TCPv2). On Linux, however, all the fields must be in network byte order.
>
> The `IP_HDRINCL` socket option was introduced with 4.3BSD Reno. Before this the only way for an application to specify its own IP header in packets sent on a raw IP socket was to apply a kernel patch that was introduced in 1988 by Van Jacobson to support Traceroute. This patch required the application to create a raw IP socket specifying a *protocol* of `IPPROTO_RAW`, which has a value of 255 (and is a reserved value and must never appear as the protocol field in an IP header).
>
> The functions that perform input and output on raw sockets are some of the simplest in the kernel. For example, in TCPv2 each function requires about 40 lines of C code (pp. 1054–1057), compared to TCP input at about 2000 lines and TCP output at about 700 lines.

Our description of the `IP_HDRINCL` socket option is for 4.4BSD. Earlier versions, such as Net/2, filled in more fields in the IP header when this option was set.

With IPv4 it is the responsibility of the user process to calculate and set any header checksums contained in whatever follows the IPv4 header. For example, in our Ping program (Figure 25.13) we must calculate the ICMPv4 checksum and store it in the ICMPv4 header before calling `sendto`.

IPv6 Differences

There are a few differences with raw IPv6 sockets [Stevens and Thomas 1997].

- All fields in the protocol headers sent or received on a raw IPv6 socket are in network byte order.
- There is nothing similar to the IPv4 `IP_HDRINCL` socket option with IPv6. Complete IPv6 packets (including extension headers) cannot be read or written on an IPv6 raw socket. Almost all fields in an IPv6 header and all extension headers are available to the application through socket options or ancillary data (see Exercise 25.1). Should an application need to read or write complete IPv6 datagrams, datalink access (described in Chapter 26) must be used.
- Checksums on raw IPv6 sockets are handled differently, as described shortly.

`IPV6_CHECKSUM` Socket Option

For an ICMPv6 raw socket the kernel always calculates and stores the checksum in the ICMPv6 header. This differs from an ICMPv4 raw socket, where the application must do this itself (compare Figures 25.13 and 25.15). While ICMPv4 and ICMPv6 both require the sender to calculate the checksum, ICMPv6 includes a pseudoheader in its checksum (we discuss the concept of a pseudoheader when we calculate the UDP checksum in Figure 26.13). One of the fields in this pseudoheader is the source IPv6 address, and normally the application lets the kernel choose this value. To prevent the application from having to try to choose this address just to calculate the checksum, it is easier to let the kernel calculate the checksum.

For other raw IPv6 sockets (i.e., those created with a third argument to `socket` other than `IPPROTO_ICMPV6`) a socket option tells the kernel whether to calculate and store a checksum in outgoing packets and verify the checksum in received packets. By default this option is disabled, and it is enabled by setting the option value to a nonnegative value, as in

```
int  offset = 2;

if (setsockopt(sockfd, IPPROTO_IPV6, IPV6_CHECKSUM,
               &offset, sizeof(offset)) < 0)
    error
```

This not only enables checksums on this socket, it also tells the kernel the byte offset of the 16-bit checksum: 2 bytes from the start of the application data in this example. To disable the option it must be set to −1. When enabled the kernel will calculate and store the checksum for outgoing packets sent on the socket and also verify the checksums for packets received on the socket.

25.4 Raw Socket Input

The first question that we must answer regarding raw socket input is: which received IP datagrams does the kernel pass to raw sockets? The following rules apply:

1. Received UDP packets and received TCP packets are *never* passed to a raw socket. If a process wants to read IP datagrams containing UDP or TCP packets, the packets must be read at the datalink layer, as described in Chapter 26.
2. *Most* ICMP packets are passed to a raw socket, after the kernel has finished processing the ICMP message. Berkeley-derived implementations pass all received ICMP packets to a raw socket other than echo request, timestamp request, and address mask request (pp. 302–303 of TCPv2). These three ICMP messages are processed entirely by the kernel.
3. *All* IGMP packets are passed to a raw socket, after the kernel has finished processing the IGMP message.
4. *All* IP datagrams with a protocol field that the kernel does not understand are passed to a raw socket. The only kernel processing done on these packets is the minimal verification of some IP header fields: the IP version, IPv4 header checksum, the header length, and the destination IP address (pp. 213–220 of TCPv2).
5. If the datagram arrives in fragments, nothing is passed to a raw socket until all fragments have arrived and have been reassembled.

When the kernel has an IP datagram to pass to the raw sockets, all raw sockets for all processes are examined, looking for all matching sockets. A copy of the IP datagram is delivered to *each* matching socket. The following tests are performed for each raw socket and only if all three tests are true is the datagram delivered to the socket.

1. If a nonzero *protocol* is specified when the raw socket is created (the third argument to `socket`), then the received datagram's protocol field must match this value, or the datagram is not delivered to this socket.
2. If a local IP address is bound to the raw socket by `bind`, then the destination IP address of the received datagram must match this bound address, or the datagram is not delivered to this socket.
3. If a foreign IP address was specified for the raw socket by `connect`, then the source IP address of the received datagram must match this connected address, or the datagram is not delivered to this socket.

Notice that if a raw socket is created with a *protocol* of 0, and neither `bind` nor `connect` is called, then that socket receives a copy of *every* raw datagram that the kernel passes to raw sockets.

Whenever a received datagram is passed to a raw IPv4 socket, the entire datagram, including the IP header, is passed to the process. For a raw IPv6 socket, everything other than any extension headers are passed to the socket (e.g., Figures 25.11 and 25.21).

In the IPv4 header passed to the application, `ip_len`, `ip_off`, and `ip_id` are host byte ordered, but the remaining fields are network byte ordered. Under Linux, all fields are left as network byte ordered.

We mentioned in the previous section that all fields in a datagram received on a raw IPv6 socket are left in their network byte order.

ICMPv6 Type Filtering

A raw ICMPv4 socket receives most ICMPv4 messages received by the kernel. But ICMPv6 is a superset of ICMPv4, including the functionality of ARP and IGMP (Section 2.2). Therefore a raw ICMPv6 socket can potentially receive many more packets compared to a raw ICMPv4 socket. But most applications using a raw socket are interested in only a small subset of all ICMP messages.

To reduce the number of packets passed from the kernel to the application across a raw ICMPv6 socket, an application-specified filter is provided. A filter is declared with a datatype of `struct icmp6_filter`, which is defined by including `<netinet/icmp6.h>`. The current filter for a raw ICMPv6 socket is set and fetched using `setsockopt` and `getsockopt` with a *level* of `IPPROTO_ICMPV6` and an *optname* of `ICMP6_FILTER`.

Six macros operate on the `icmp6_filter` structure.

```
#include <netinet/icmp6.h>

void ICMP6_FILTER_SETPASSALL(struct icmp6_filter *filt);
void ICMP6_FILTER_SETBLOCKALL(struct icmp6_filter *filt);

void ICMP6_FILTER_SETPASS(int msgtype, struct icmp6_filter *filt);
void ICMP6_FILTER_SETBLOCK(int msgtype, struct icmp6_filter *filt);

int  ICMP6_FILTER_WILLPASS(int msgtype, const struct icmp6_filter *filt);
int  ICMP6_FILTER_WILLBLOCK(int msgtype, const struct icmp6_filter *filt);
```

Both return: 1 if filter will pass (block) message type, 0 otherwise

The *filt* argument to all the macros is a pointer to an `icmp6_filter` variable that is modified by the first four macros and examined by the final two macros. The *msgtype* argument is a value between 0 and 255 specifying the ICMP message type.

The `SETPASSALL` macro specifies that all message types are to be passed to the application, while the `SETBLOCKALL` macros specifies that no message types are to be passed. By default, when an ICMPv6 raw socket is created, all ICMPv6 message types are passed to the application.

The `SETPASS` macro enables one specific message type to be passed to the application while the `SETBLOCK` macro blocks one specific message type. The `WILLPASS` macro returns 1 if the specified message type is passed by the filter, or 0 otherwise, while the `WILLBLOCK` macro returns 1 if the specified message type is blocked by the filter, or 0 otherwise.

As an example, consider an application that wants to receive only ICMPv6 router advertisements:

```
struct icmp6_filter  myfilt;

fd = Socket(AF_INET6, SOCK_RAW, IPPROTO_ICMPV6);

ICMP6_FILTER_SETBLOCKALL(&myfilt);
ICMP6_FILTER_SETPASS(ND_ROUTER_ADVERT, &myfilt);
Setsockopt(fd, IPPROTO_ICMPV6, ICMP6_FILTER, &myfilt, sizeof(myfilt));
```

We first block all message types (since the default is to pass all message types) and then pass only router advertisements.

25.5 Ping Program

In this section we develop and present a version of the Ping program that works with both IPv4 and IPv6. We develop our own program, instead of presenting the publicly available source code for two reasons. First, the publicly available Ping program suffers from a common programming disease known as *creeping featurism*: it supports a dozen different options. Our goal in examining a Ping program is to understand the network programming concepts and techniques, without being distracted with all these options. Our version of Ping supports only one option and is about five times smaller than the public version. Second, the public version works with only IPv4 and we want to show a version that also supports IPv6.

The operation of Ping is extremely simple: an ICMP echo request is sent to some IP address and that node responds with an ICMP echo reply. These two ICMP messages are supported under both IPv4 and IPv6. Figure 25.1 shows the format of the ICMP messages.

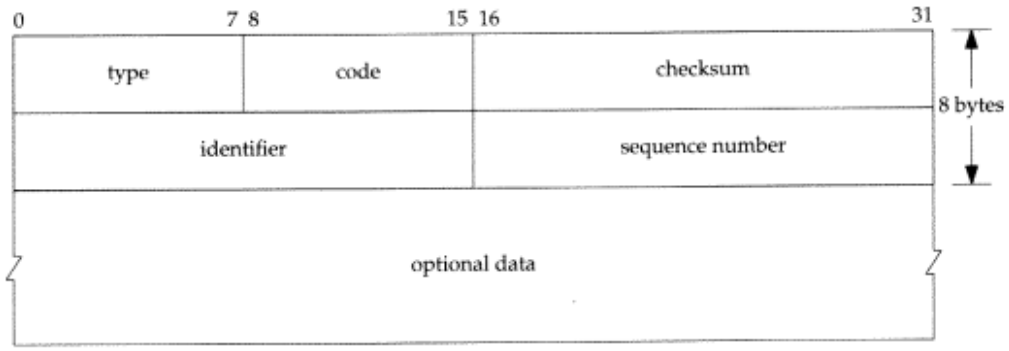


Figure 25.1 Format of ICMPv4 and ICMPv6 echo request and echo reply messages.

Figures A.15 and A.16 show the *type* values for these messages and also show that the *code* is 0. We will see that we set the *identifier* to the process ID of the Ping process and we increment the *sequence number* by one for each packet that we send. We store the 8-byte timestamp of when the packet is sent as the optional data. The rules of ICMP require that the *identifier*, *sequence number*, and any optional data be returned in the echo reply. Storing the timestamp in the packet lets us calculate the RTT when the reply is received.

Figure 25.2 shows some examples of our program. The first uses IPv4 and the second uses IPv6. Note the pound-sign prompt, signifying the superuser, as it takes superuser privileges to create a raw socket.

```
solaris # ping gemini.tuc.noao.edu
PING gemini.tuc.noao.edu (140.252.4.54): 56 data bytes
64 bytes from 140.252.4.54: seq=0, ttl=248, rtt=37.542 ms
64 bytes from 140.252.4.54: seq=1, ttl=248, rtt=34.596 ms
64 bytes from 140.252.4.54: seq=2, ttl=248, rtt=29.204 ms
64 bytes from 140.252.4.54: seq=3, ttl=248, rtt=52.630 ms

solaris # ping 6bone-router.cisco.com
PING 6bone-router.cisco.com (5f00:6d00:c01f:700:1:60:3e11:6770): 56 data bytes
64 bytes from 5f00:6d00:c01f:700:1:60:3e11:6770: seq=0, hlim=255, rtt=116.802 ms
64 bytes from 5f00:6d00:c01f:700:1:60:3e11:6770: seq=1, hlim=255, rtt=129.321 ms
64 bytes from 5f00:6d00:c01f:700:1:60:3e11:6770: seq=2, hlim=255, rtt=109.297 ms
64 bytes from 5f00:6d00:c01f:700:1:60:3e11:6770: seq=3, hlim=255, rtt=78.216 ms
```

Figure 25.2 Sample output from our Ping program.

Figure 25.3 is an overview of the functions that comprise our Ping program.

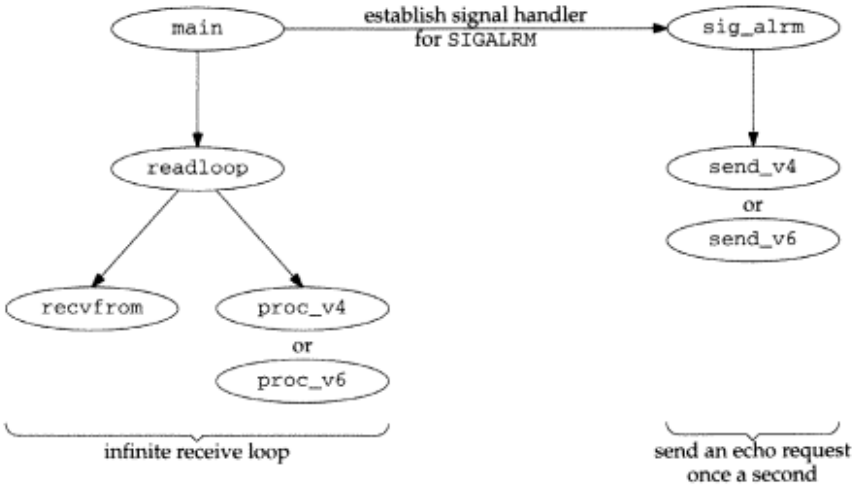


Figure 25.3 Overview of the functions in our Ping program.

The program operates in two parts: one half reads everything received on a raw socket, printing the ICMP echo replies, and the other half sends an ICMP echo request once a second. The second half is driven by a `SIGALRM` signal once a second.

Figure 25.4 shows our `ping.h` header that is included by all our program files.

ping/ping.h

```
1 #include    "unp.h"
2 #include    <netinet/in_systm.h>
3 #include    <netinet/ip.h>
4 #include    <netinet/ip_icmp.h>
```

```
 5 #define BUFSIZE     1500

 6             /* globals */
 7 char    recvbuf[BUFSIZE];
 8 char    sendbuf[BUFSIZE];

 9 int     datalen;                /* #bytes of data, following ICMP header */
10 char   *host;
11 int     nsent;                  /* add 1 for each sendto() */
12 pid_t   pid;                    /* our PID */
13 int     sockfd;
14 int     verbose;

15             /* function prototypes */
16 void    proc_v4(char *, ssize_t, struct timeval *);
17 void    proc_v6(char *, ssize_t, struct timeval *);
18 void    send_v4(void);
19 void    send_v6(void);
20 void    readloop(void);
21 void    sig_alrm(int);
22 void    tv_sub(struct timeval *, struct timeval *);

23 struct proto {
24     void    (*fproc) (char *, ssize_t, struct timeval *);
25     void    (*fsend) (void);
26     struct sockaddr *sasend;    /* sockaddr{} for send, from getaddrinfo */
27     struct sockaddr *sarecv;    /* sockaddr{} for receiving */
28     socklen_t salen;            /* length of sockaddr{}s */
29     int     icmpproto;          /* IPPROTO_xxx value for ICMP */
30 } *pr;

31 #ifdef  IPV6

32 #include    "ip6.h"             /* should be <netinet/ip6.h> */
33 #include    "icmp6.h"           /* should be <netinet/icmp6.h> */

34 #endif
```

ping/ping.h

Figure 25.4 `ping.h` header.

Include IPv4 and ICMPv4 headers

1–22 We include the basic IPv4 and ICMPv4 headers, define some global variables, and our function prototypes.

Define `proto` structure

23–30 We use the `proto` structure to handle the differences between IPv4 and IPv6. This structure contains two function pointers, two pointers to socket address structures, the size of the socket address structures, and the protocol value for ICMP. The global pointer `pr` will point to one of these structures that we will initialize for either IPv4 or IPv6.

Include IPv6 and ICMPv6 headers

31–34 We include two headers that define the IPv6 and ICMPv6 structures and constants ([Stevens and Thomas 1997]).

The `main` function is shown in Figure 25.5.

ping/main.c

```
 1 #include    "ping.h"

 2 struct proto proto_v4 =
 3 {proc_v4, send_v4, NULL, NULL, 0, IPPROTO_ICMP};

 4 #ifdef  IPV6
 5 struct proto proto_v6 =
 6 {proc_v6, send_v6, NULL, NULL, 0, IPPROTO_ICMPV6};
 7 #endif

 8 int     datalen = 56;           /* data that goes with ICMP echo request */

 9 int
10 main(int argc, char **argv)
11 {
12     int     c;
13     struct addrinfo *ai;

14     opterr = 0;                 /* don't want getopt() writing to stderr */
15     while ( (c = getopt(argc, argv, "v")) != -1) {
16         switch (c) {
17         case 'v':
18             verbose++;
19             break;

20         case '?':
21             err_quit("unrecognized option: %c", c);
22         }
23     }

24     if (optind != argc - 1)
25         err_quit("usage: ping [ -v ] <hostname>");
26     host = argv[optind];

27     pid = getpid();
28     Signal(SIGALRM, sig_alrm);

29     ai = Host_serv(host, NULL, 0, 0);

30     printf("PING %s (%s): %d data bytes\n", ai->ai_canonname,
31            Sock_ntop_host(ai->ai_addr, ai->ai_addrlen), datalen);

32         /* initialize according to protocol */
33     if (ai->ai_family == AF_INET) {
34         pr = &proto_v4;
35 #ifdef  IPV6
36     } else if (ai->ai_family == AF_INET6) {
37         pr = &proto_v6;
38         if (IN6_IS_ADDR_V4MAPPED(&(((struct sockaddr_in6 *)
39                                     ai->ai_addr)->sin6_addr)))
40             err_quit("cannot ping IPv4-mapped IPv6 address");
41 #endif
42     } else
43         err_quit("unknown address family %d", ai->ai_family);
```

```
44      pr->sasend = ai->ai_addr;
45      pr->sarecv = Calloc(1, ai->ai_addrlen);
46      pr->salen = ai->ai_addrlen;

47      readloop();

48      exit(0);
49  }
```
ping/main.c

Figure 25.5 `main` function.

Define `proto` structures for IPv4 and IPv6

2-7 We define a `proto` structure for IPv4 and for IPv6. The socket address structure pointers are initialized to null pointers, as we do not yet know whether we will use IPv4 or IPv6.

Length of optional data

8 We set the amount of optional data that gets sent with the ICMP echo request to 56 bytes. This will yield an 84-byte IPv4 datagram (20-byte IPv4 header and 8-byte ICMP header) or a 104-byte IPv6 datagram. Any data that accompanies an echo request must be sent back in the echo reply. We will store the time at which we send an echo request in the first 8 bytes of this data area and then use this to calculate and print the RTT when the echo reply is received.

Handle command-line options

14-28 The only command-line option that we support is `-v`, which will cause us to print most received ICMP messages. (We do not print echo replies belonging to another copy of Ping that is running.) A signal handler is established for `SIGALRM` and we will see that this signal is generated once a second and causes an ICMP echo request to be sent.

Process hostname argument

29-46 A hostname or IP address string is a required argument and it is processed by our `host_serv` function. The returned `addrinfo` structure contains the protocol family, either `AF_INET` or `AF_INET6`. We initialize the `pr` global to the correct `proto` structure. We also make certain that an IPv6 address is not really an IPv4-mapped IPv6 address by calling `IN6_IS_ADDR_V4MAPPED`, because even though the returned address is an IPv6 address, IPv4 packets will be sent to the host. (We could switch and use IPv4 when this happens.) The socket address structure that has already been allocated by the `getaddrinfo` function is used as the one for sending, and another socket address structure of the same size is allocated for receiving.

47 The function `readloop` is where the processing takes place, and we show this in Figure 25.6.

Create socket

10-11 A raw socket of the appropriate protocol is created. The call to `setuid` sets our effective user ID to our real user ID. The program must have superuser privileges to create the raw socket, but now that the socket is created, we can give up the extra privileges. It is always best to give up this extra privilege when it is no longer needed, just in case the program has a latent bug that someone could exploit.

ping/readloop.c

```
 1 #include    "ping.h"

 2 void
 3 readloop(void)
 4 {
 5     int     size;
 6     char    recvbuf[BUFSIZE];
 7     socklen_t len;
 8     ssize_t n;
 9     struct timeval tval;

10     sockfd = Socket(pr->sasend->sa_family, SOCK_RAW, pr->icmpproto);
11     setuid(getuid());           /* don't need special permissions any more */

12     size = 60 * 1024;           /* OK if setsockopt fails */
13     setsockopt(sockfd, SOL_SOCKET, SO_RCVBUF, &size, sizeof(size));

14     sig_alrm(SIGALRM);          /* send first packet */

15     for ( ; ; ) {
16         len = pr->salen;
17         n = recvfrom(sockfd, recvbuf, sizeof(recvbuf), 0, pr->sarecv, &len);
18         if (n < 0) {
19             if (errno == EINTR)
20                 continue;
21             else
22                 err_sys("recvfrom error");
23         }
24         Gettimeofday(&tval, NULL);
25         (*pr->fproc) (recvbuf, n, &tval);
26     }
27 }
```

ping/readloop.c

Figure 25.6 `readloop` function.

Set socket receive buffer size

12-13 We try to set the socket receive buffer size to 61,440 bytes (60×1024), which should be larger than the default. We do this in case the user pings either the IPv4 broadcast address or a multicast address, either of which can generate lots of replies. By making the buffer larger, there is a smaller chance that the socket receive buffer will overflow.

Send first packet

14 We call our signal handler, which we will see sends a packet and schedules a `SIGALRM` for 1 second in the future. It is not common to see a signal handler called directly, as we do here, but it is OK. A signal handler is just a C function, even though it is normally called asynchronously by the kernel.

Infinite loop reading all ICMP messages

15-26 The main loop of the program is an infinite loop that reads all packets returned on the raw ICMP socket. We call `gettimeofday` to record the time that the packet was received and then call the appropriate protocol function (`proc_v4` or `proc_v6`) to process the ICMP message.

Figure 25.8 shows the `proc_v4` function, which processes all received ICMPv4 messages. You may want to refer to Figure A.1, which shows the format of the IPv4 header. Also realize that when the ICMPv4 message is received by the process on the raw socket, the kernel has already verified that the basic fields in the IPv4 header and in the ICMPv4 header are valid (pp. 214 and 311 of TCPv2).

Get pointer to ICMP header

10-14 The IPv4 header length field is multiplied by 4, giving the size of the IPv4 header in bytes. (Remember that an IPv4 header can contain options.) This lets us set `icmp` to point to the beginning of the ICMP header. Figure 25.9 shows the various headers, pointers, and lengths used by the code.

Check for ICMP echo reply

15-19 If the message is an ICMP echo reply, then we must check the identifier field to see if this reply is in response to a request that our process sent. If the Ping program is running multiple times on this host, each process gets a copy of all received ICMP messages.

20-25 We calculate the RTT by subtracting the time the message was sent (contained in the optional data portion of the ICMP reply) from the current time (pointed to by the `tvrecv` function argument). The RTT is converted from microseconds to milliseconds and printed, along with the sequence number field and the received TTL. The sequence number field lets the user see if packets are dropped, reordered, or duplicated, and the TTL gives an indication of the number of hops between the two hosts.

Print all received ICMP messages if verbose option specified

26-30 If the user specified the `-v` command-line option, we print the type and code fields from all other received ICMP messages.

Figure 25.7 shows the `tv_sub` function, which subtracts two `timeval` structures, storing the result in the first structure.

lib/tv_sub.c

```
 1 #include    "unp.h"

 2 void
 3 tv_sub(struct timeval *out, struct timeval *in)
 4 {
 5     if ((out->tv_usec -= in->tv_usec) < 0) {    /* out -= in */
 6         --out->tv_sec;
 7         out->tv_usec += 1000000;
 8     }
 9     out->tv_sec -= in->tv_sec;
10 }
```

lib/tv_sub.c

Figure 25.7 `tv_sub` function: subtract two `timeval` structures.

The processing of ICMPv6 messages is handled by the `proc_v6` function, shown in Figure 25.10 (p. 669). It is similar to the `proc_v4` function.

ping/proc_v4.c

```
 1 #include    "ping.h"

 2 void
 3 proc_v4(char *ptr, ssize_t len, struct timeval *tvrecv)
 4 {
 5     int     hlen1, icmplen;
 6     double  rtt;
 7     struct ip *ip;
 8     struct icmp *icmp;
 9     struct timeval *tvsend;

10     ip = (struct ip *) ptr;     /* start of IP header */
11     hlen1 = ip->ip_hl << 2;     /* length of IP header */

12     icmp = (struct icmp *) (ptr + hlen1);   /* start of ICMP header */
13     if ( (icmplen = len - hlen1) < 8)
14         err_quit("icmplen (%d) < 8", icmplen);

15     if (icmp->icmp_type == ICMP_ECHOREPLY) {
16         if (icmp->icmp_id != pid)
17             return;             /* not a response to our ECHO_REQUEST */
18         if (icmplen < 16)
19             err_quit("icmplen (%d) < 16", icmplen);

20         tvsend = (struct timeval *) icmp->icmp_data;
21         tv_sub(tvrecv, tvsend);
22         rtt = tvrecv->tv_sec * 1000.0 + tvrecv->tv_usec / 1000.0;

23         printf("%d bytes from %s: seq=%u, ttl=%d, rtt=%.3f ms\n",
24                icmplen, Sock_ntop_host(pr->sarecv, pr->salen),
25                icmp->icmp_seq, ip->ip_ttl, rtt);

26     } else if (verbose) {
27         printf("  %d bytes from %s: type = %d, code = %d\n",
28                icmplen, Sock_ntop_host(pr->sarecv, pr->salen),
29                icmp->icmp_type, icmp->icmp_code);
30     }
31 }
```

ping/proc_v4.c

Figure 25.8 `proc_v4` function: process ICMPv4 message.

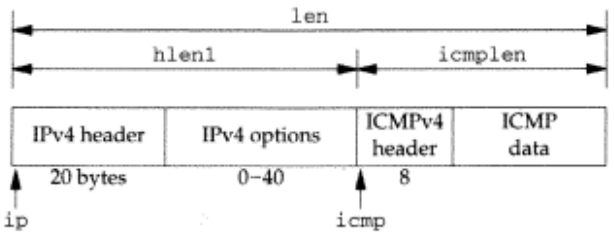


Figure 25.9 Headers, pointers, and lengths in processing ICMPv4 reply.

ping/proc_v6.c

```
 1 #include    "ping.h"

 2 void
 3 proc_v6(char *ptr, ssize_t len, struct timeval *tvrecv)
 4 {
 5 #ifdef  IPV6
 6     int     hlen1, icmp6len;
 7     double  rtt;
 8     struct ip6_hdr *ip6;
 9     struct icmp6_hdr *icmp6;
10     struct timeval *tvsend;

11     ip6 = (struct ip6_hdr *) ptr;   /* start of IPv6 header */
12     hlen1 = sizeof(struct ip6_hdr);
13     if (ip6->ip6_nxt != IPPROTO_ICMPV6)
14         err_quit("next header not IPPROTO_ICMPV6");

15     icmp6 = (struct icmp6_hdr *) (ptr + hlen1);
16     if ( (icmp6len = len - hlen1) < 8)
17         err_quit("icmp6len (%d) < 8", icmp6len);

18     if (icmp6->icmp6_type == ICMP6_ECHO_REPLY) {
19         if (icmp6->icmp6_id != pid)
20             return;             /* not a response to our ECHO_REQUEST */
21         if (icmp6len < 16)
22             err_quit("icmp6len (%d) < 16", icmp6len);

23         tvsend = (struct timeval *) (icmp6 + 1);
24         tv_sub(tvrecv, tvsend);
25         rtt = tvrecv->tv_sec * 1000.0 + tvrecv->tv_usec / 1000.0;

26         printf("%d bytes from %s: seq=%u, hlim=%d, rtt=%.3f ms\n",
27                icmp6len, Sock_ntop_host(pr->sarecv, pr->salen),
28                icmp6->icmp6_seq, ip6->ip6_hlim, rtt);

29     } else if (verbose) {
30         printf("  %d bytes from %s: type = %d, code = %d\n",
31                icmp6len, Sock_ntop_host(pr->sarecv, pr->salen),
32                icmp6->icmp6_type, icmp6->icmp6_code);
33     }
34 #endif  /* IPV6 */
35 }
```

ping/proc_v6.c

Figure 25.10 `proc_v6` function: process received ICMPv6 message.

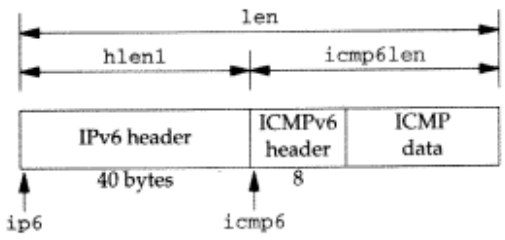


Figure 25.11 Headers, pointers, and lengths in processing ICMPv6 reply.

Get pointer to ICMPv6 header

11-17 The size of the IPv6 header is fixed (40 bytes) and we ensure that the next header is ICMPv6. (Recall that the extension headers, if any, are never returned as normal data, but as ancillary data.) Figure 25.11 shows the various headers, pointers, and lengths used by the code.

Check for ICMP echo reply

18-28 If the ICMP message type is an echo reply, we check the identifier field to see if the reply is for us. If so, we calculate the RTT and then print it along with the sequence number and the IPv6 hop limit.

Print all received ICMP messages if verbose option specified

29-33 If the user specified the `-v` command-line option, we print the type and code fields from all other received ICMP messages.

> If the `-v` option is not specified, we could establish an ICMPv6 filter (the `ICMP6_FILTER` socket option described in Section 25.4) so that only echo replies are returned on our socket by the kernel.

Our signal handler for the `SIGALRM` signal is the `sig_alrm` function, shown in Figure 25.12. We saw in Figure 25.6 that our `readloop` function calls this signal handler once at the beginning to send the first packet. This function just calls the protocol-dependent function to send an ICMP echo request (`send_v4` or `send_v6`) and then schedules another `SIGALRM` for 1 second in the future.

ping/sig_alrm.c

```
1 #include    "ping.h"

2 void
3 sig_alrm(int signo)
4 {
5     (*pr->fsend) ();

6     alarm(1);
7     return;                       /* probably interrupts recvfrom() */
8 }
```

ping/sig_alrm.c

Figure 25.12 `sig_alrm` function: `SIGALRM` signal handler.

The function `send_v4`, shown in Figure 25.13, builds an ICMPv4 echo request message and writes it to the raw socket.

Build ICMPv4 message

7-12 The ICMPv4 message is built. The identifier field is set to our process ID and the sequence number field is set to the global `nsent`, which is then incremented for the next packet. The current time-of-day is stored in the data portion of the ICMP message.

Calculate ICMP checksum

13-15 To calculate the ICMP checksum we set the checksum field to 0 and call the function `in_cksum`, storing the result in the checksum field. The ICMPv4 checksum is calculated from the ICMPv4 header and any data that follows.

ping/send_v4.c

```
 1 #include    "ping.h"

 2 void
 3 send_v4(void)
 4 {
 5     int     len;
 6     struct icmp *icmp;

 7     icmp = (struct icmp *) sendbuf;
 8     icmp->icmp_type = ICMP_ECHO;
 9     icmp->icmp_code = 0;
10     icmp->icmp_id = pid;
11     icmp->icmp_seq = nsent++;
12     Gettimeofday((struct timeval *) icmp->icmp_data, NULL);

13     len = 8 + datalen;          /* checksum ICMP header and data */
14     icmp->icmp_cksum = 0;
15     icmp->icmp_cksum = in_cksum((u_short *) icmp, len);

16     Sendto(sockfd, sendbuf, len, 0, pr->sasend, pr->salen);
17 }
```

ping/send_v4.c

Figure 25.13 `send_v4` function: build an ICMPv4 echo request message and send it.

Send datagram

16 The ICMP message is sent on the raw socket. Since we have not set the `IP_HDRINCL` socket option, the kernel builds the IPv4 header and prepends it to our buffer.

The Internet checksum is the ones-complement sum of the 16-bit values to be checksummed. If the data length is an odd number, then 1 byte of 0 is logically appended to the end of the data, just for the checksum computation. This algorithm is used for the IPv4, ICMPv4, IGMPv4, ICMPv6, UDP, and TCP checksums. RFC 1071 [Braden, Borman, and Partridge 1988] contains additional information and some numerical examples. Section 8.7 of TCPv2 talks about this algorithm in more detail and shows a more efficient implementation. Our `in_cksum` function, shown in Figure 25.14, calculates the checksum.

Internet checksum algorithm

1-27 The first `while` loop calculates the sum of all the 16-bit values. If the length is odd, then the final byte is added in to the sum. The algorithm that we show in Figure 25.14 is the simple algorithm, fine for a Ping program, but inadequate for the high volume of checksum computations performed by the kernel.

This function is taken from the public domain version of Ping.

The final function for our Ping program is `send_v6`, shown in Figure 25.15 (p. 673), which builds and sends an ICMPv6 echo request.

libfree/in_cksum.c

```
 1 unsigned short
 2 in_cksum(unsigned short *addr, int len)
 3 {
 4     int     nleft = len;
 5     int     sum = 0;
 6     unsigned short *w = addr;
 7     unsigned short answer = 0;

 8     /*
 9      * Our algorithm is simple, using a 32 bit accumulator (sum), we add
10      * sequential 16 bit words to it, and at the end, fold back all the
11      * carry bits from the top 16 bits into the lower 16 bits.
12      */
13     while (nleft > 1) {
14         sum += *w++;
15         nleft -= 2;
16     }

17         /* mop up an odd byte, if necessary */
18     if (nleft == 1) {
19         *(unsigned char *) (&answer) = *(unsigned char *) w;
20         sum += answer;
21     }
22         /* add back carry outs from top 16 bits to low 16 bits */
23     sum = (sum >> 16) + (sum & 0xffff);     /* add hi 16 to low 16 */
24     sum += (sum >> 16);             /* add carry */
25     answer = ~sum;                  /* truncate to 16 bits */
26     return (answer);
27 }
```

libfree/in_cksum.c

Figure 25.14 `in_cksum` function: calculate the Internet checksum.

The `send_v6` function is similar to `send_v4`, but notice that it does not compute the ICMPv6 checksum. As we mentioned earlier in the chapter, since the ICMPv6 checksum uses the source address from the IPv6 header in its computation, this checksum is calculated by the kernel for us, after the kernel chooses the source address.

25.6 Traceroute Program

In this section we develop our own version of the Traceroute program. Like the Ping program that we developed in the previous section, we develop and present our own version, instead of presenting the publicly available version. We do this because we need a version that supports both IPv4 and IPv6, and we do not want to be distracted with lots of options that are not germane to our discussion of network programming.

Traceroute lets us determine the path that IP datagrams follow from our host to some other destination. Its operation is simple and Chapter 8 of TCPv1 covers it in

ping/send_v6.c

```
 1 #include    "ping.h"

 2 void
 3 send_v6()
 4 {
 5 #ifdef  IPV6
 6     int     len;
 7     struct icmp6_hdr *icmp6;

 8     icmp6 = (struct icmp6_hdr *) sendbuf;
 9     icmp6->icmp6_type = ICMP6_ECHO_REQUEST;
10     icmp6->icmp6_code = 0;
11     icmp6->icmp6_id = pid;
12     icmp6->icmp6_seq = nsent++;
13     Gettimeofday((struct timeval *) (icmp6 + 1), NULL);

14     len = 8 + datalen;          /* 8-byte ICMPv6 header */

15     Sendto(sockfd, sendbuf, len, 0, pr->sasend, pr->salen);
16         /* kernel calculates and stores checksum for us */
17 #endif  /* IPV6 */
18 }
```

ping/send_v6.c

Figure 25.15 send_v6 function: build and send an ICMPv6 echo request message.

detail with numerous examples of its usage. Traceroute uses the IPv4 TTL field or the IPv6 hop limit field and two ICMP messages. It starts by sending a UDP datagram to the destination with a TTL (or hop limit) of 1. This datagram causes the first hop router to return an ICMP "time exceeded in transit" error. The TTL is then increased by one and another UDP datagram sent, which locates the next router in the path. When the UDP datagram reaches the final destination, the goal is to have that host return an ICMP "port unreachable" error. This is done by sending the UDP datagram to a random port that is (hopefully) not in use on that host.

Early versions of Traceroute were able to set the TTL field in the IPv4 header only by setting the IP_HDRINCL socket option and then building their own IPv4 header. Current systems, however, provide the IP_TTL socket option that lets us specify the TTL to use for outgoing datagrams. (This socket option was introduced with the 4.3BSD Reno release.) It is easier to set this socket option than to build a complete IPv4 header (although we show how to build our own IPv4 and UDP headers in Section 26.6). The IPv6 IPV6_UNICAST_HOPS socket option lets us control the hop limit field for IPv6 datagrams.

Figure 25.16 shows our trace.h header, which all of our program files include.

1-11 We include the standard IPv4 headers that define the IPv4, ICMPv4, and UDP structures and constants. The rec structure defines the data portion of the UDP datagram that we send, but we will see that we never need to examine this data. It is sent mainly for debugging purposes.

traceroute/trace.h

```
 1 #include    "unp.h"
 2 #include    <netinet/in_systm.h>
 3 #include    <netinet/ip.h>
 4 #include    <netinet/ip_icmp.h>
 5 #include    <netinet/udp.h>

 6 #define BUFSIZE     1500

 7 struct rec {                      /* of outgoing UDP data */
 8     u_short rec_seq;             /* sequence number */
 9     u_short rec_ttl;             /* TTL packet left with */
10     struct timeval rec_tv;       /* time packet left */
11 };

12             /* globals */
13 char    recvbuf[BUFSIZE];
14 char    sendbuf[BUFSIZE];

15 int     datalen;                 /* #bytes of data, following ICMP header */
16 char   *host;
17 u_short sport, dport;
18 int     nsent;                   /* add 1 for each sendto() */
19 pid_t   pid;                     /* our PID */
20 int     probe, nprobes;
21 int     sendfd, recvfd;          /* send on UDP sock, read on raw ICMP sock */
22 int     ttl, max_ttl;
23 int     verbose;

24             /* function prototypes */
25 char   *icmpcode_v4(int);
26 char   *icmpcode_v6(int);
27 int     recv_v4(int, struct timeval *);
28 int     recv_v6(int, struct timeval *);
29 void    sig_alrm(int);
30 void    traceloop(void);
31 void    tv_sub(struct timeval *, struct timeval *);

32 struct proto {
33     char   *(*icmpcode) (int);
34     int     (*recv) (int, struct timeval *);
35     struct sockaddr *sasend;     /* sockaddr{} for send, from getaddrinfo */
36     struct sockaddr *sarecv;     /* sockaddr{} for receiving */
37     struct sockaddr *salast;     /* last sockaddr{} for receiving */
38     struct sockaddr *sabind;     /* sockaddr{} for binding source port */
39     socklen_t salen;             /* length of sockaddr{}s */
40     int     icmpproto;           /* IPPROTO_xxx value for ICMP */
41     int     ttllevel;            /* setsockopt() level to set TTL */
42     int     ttloptname;          /* setsockopt() name to set TTL */
43 } *pr;

44 #ifdef  IPV6

45 #include    "ip6.h"              /* should be <netinet/ip6.h> */
46 #include    "icmp6.h"            /* should be <netinet/icmp6.h> */

47 #endif
```

traceroute/trace.h

Figure 25.16 `trace.h` header.

Define `proto` structure

32-43 As with our Ping program in the previous section, we handle the protocol differences between IPv4 and IPv6 by defining a `proto` structure that contains function pointers, pointers to socket address structures, and other constants that differ between the two IP versions. The global `pr` will be set to point to one of these structures that is initialized for either IPv4 or IPv6, after the destination address is processed by the `main` function (since the destination address is what specifies whether we use IPv4 or IPv6).

Include IPv6 headers

44-47 We include the headers that define the IPv6 and ICMPv6 structures and constants.

The `main` function is shown in Figure 25.17. It processes the command-line arguments, initializes the `pr` pointer for either IPv4 or IPv6, and calls our `traceloop` function.

traceroute/main.c

```
 1 #include    "trace.h"

 2 struct proto proto_v4 =
 3 {icmpcode_v4, recv_v4, NULL, NULL, NULL, NULL, 0,
 4  IPPROTO_ICMP, IPPROTO_IP, IP_TTL};

 5 #ifdef  IPV6
 6 struct proto proto_v6 =
 7 {icmpcode_v6, recv_v6, NULL, NULL, NULL, NULL, 0,
 8  IPPROTO_ICMPV6, IPPROTO_IPV6, IPV6_UNICAST_HOPS};
 9 #endif

10 int     datalen = sizeof(struct rec);   /* defaults */
11 int     max_ttl = 30;
12 int     nprobes = 3;
13 u_short dport = 32768 + 666;

14 int
15 main(int argc, char **argv)
16 {
17     int     c;
18     struct addrinfo *ai;

19     opterr = 0;                 /* don't want getopt() writing to stderr */
20     while ( (c = getopt(argc, argv, "m:v")) != -1) {
21         switch (c) {
22         case 'm':
23             if ( (max_ttl = atoi(optarg)) <= 1)
24                 err_quit("invalid -m value");
25             break;

26         case 'v':
27             verbose++;
28             break;

29         case '?':
30             err_quit("unrecognized option: %c", c);
31         }
32     }
```

```
33      if (optind != argc - 1)
34          err_quit("usage: traceroute [ -m <maxttl> -v ] <hostname>");
35      host = argv[optind];

36      pid = getpid();
37      Signal(SIGALRM, sig_alrm);

38      ai = Host_serv(host, NULL, 0, 0);

39      printf("traceroute to %s (%s): %d hops max, %d data bytes\n",
40             ai->ai_canonname,
41             Sock_ntop_host(ai->ai_addr, ai->ai_addrlen),
42             max_ttl, datalen);

43      /* initialize according to protocol */
44      if (ai->ai_family == AF_INET) {
45          pr = &proto_v4;
46  #ifdef  IPV6
47      } else if (ai->ai_family == AF_INET6) {
48          pr = &proto_v6;
49          if (IN6_IS_ADDR_V4MAPPED(&(((struct sockaddr_in6 *) ai->ai_addr)->sin6_addr)))
50              err_quit("cannot traceroute IPv4-mapped IPv6 address");
51  #endif
52      } else
53          err_quit("unknown address family %d", ai->ai_family);

54      pr->sasend = ai->ai_addr;   /* contains destination address */
55      pr->sarecv = Calloc(1, ai->ai_addrlen);
56      pr->salast = Calloc(1, ai->ai_addrlen);
57      pr->sabind = Calloc(1, ai->ai_addrlen);
58      pr->salen = ai->ai_addrlen;

59      traceloop();

60      exit(0);
61  }
```

traceroute/main.c

Figure 25.17 `main` function for Traceroute program.

Define `proto` structures

2-9 We define the two `proto` structures, one for IPv4 and one for IPv6, although the pointers to the socket address structures are not allocated until the end of this function.

Set defaults

10-13 The maximum TTL or hop limit that the program uses defaults to 30, although we provide the `-m` command-line option to let the user change this. For each TTL we send three probe packets, but this could be changed with another command-line option. The initial destination port is 32768 + 666 and this will be incremented by one each time we send a UDP datagram. We hope that these ports are not in use on the destination host when the datagrams finally reach the destination, but there is no guarantee.

Process command-line arguments

19-37 The `-v` command-line option causes most received ICMP messages to be printed.

Process hostname or IP address argument and finish initialization

38-58 The destination hostname or IP address is processed by our `host_serv` function, returning a pointer to an `addrinfo` structure. Depending on the type of returned address, IPv4 or IPv6, we finish initializing the `proto` structure, store the pointer in the `pr` global, and allocate additional socket address structures of the correct size.

59 The function `traceloop`, shown in Figure 25.18, sends the datagrams and reads the returned ICMP messages. This is the main loop of the program.

Create two sockets

9-11 We need two sockets: a raw socket on which we read all returned ICMP messages and a UDP socket on which we send the probe packets with the increasing TTLs. After creating the raw socket, we reset our effective user ID to our real user ID, since we no longer require superuser privileges.

Bind source port of UDP socket

12-15 We `bind` a source port to the UDP socket that is used for sending, using the low-order 16 bits of our process ID with the high-order bit set to 1. Since it is possible for multiple copies of the Traceroute program to be running at any given time, we need a way to determine if a received ICMP message was generated in response to one of our datagrams, or in response to a datagram sent by another copy of the program. We use the source port in the UDP header to identify the sending process because the returned ICMP message always returns the UDP header from the datagram that caused the ICMP error.

Establish signal handler for `SIGALRM`

16 We establish our function `sig_alrm` as the signal handler for `SIGALRM` because each time we send a UDP datagram we wait 3 seconds for an ICMP message before sending the next probe.

traceroute/traceloop.c

```
 1 #include    "trace.h"

 2 void
 3 traceloop(void)
 4 {
 5     int     seq, code, done;
 6     double  rtt;
 7     struct rec *rec;
 8     struct timeval tvrecv;

 9     recvfd = Socket(pr->sasend->sa_family, SOCK_RAW, pr->icmpproto);
10     setuid(getuid());           /* don't need special permissions any more */

11     sendfd = Socket(pr->sasend->sa_family, SOCK_DGRAM, 0);

12     pr->sabind->sa_family = pr->sasend->sa_family;
13     sport = (getpid() & 0xffff) | 0x8000;   /* our source UDP port# */
14     sock_set_port(pr->sabind, pr->salen, htons(sport));
15     Bind(sendfd, pr->sabind, pr->salen);

16     sig_alrm(SIGALRM);
```

```
17      seq = 0;
18      done = 0;
19      for (ttl = 1; ttl <= max_ttl && done == 0; ttl++) {
20          Setsockopt(sendfd, pr->ttllevel, pr->ttloptname, &ttl, sizeof(int));
21          bzero(pr->salast, pr->salen);
22          printf("%2d  ", ttl);
23          fflush(stdout);
24          for (probe = 0; probe < nprobes; probe++) {
25              rec = (struct rec *) sendbuf;
26              rec->rec_seq = ++seq;
27              rec->rec_ttl = ttl;
28              Gettimeofday(&rec->rec_tv, NULL);
29              sock_set_port(pr->sasend, pr->salen, htons(dport + seq));
30              Sendto(sendfd, sendbuf, datalen, 0, pr->sasend, pr->salen);
31              if ( (code = (*pr->recv) (seq, &tvrecv)) == -3)
32                  printf(" *");   /* timeout, no reply */
33              else {
34                  char    str[NI_MAXHOST];
35                  if (sock_cmp_addr(pr->sarecv, pr->salast, pr->salen) != 0) {
36                      if (getnameinfo(pr->sarecv, pr->salen, str, sizeof(str),
37                                      NULL, 0, 0) == 0)
38                          printf(" %s (%s)", str,
39                                  Sock_ntop_host(pr->sarecv, pr->salen));
40                      else
41                          printf(" %s",
42                                  Sock_ntop_host(pr->sarecv, pr->salen));
43                      memcpy(pr->salast, pr->sarecv, pr->salen);
44                  }
45                  tv_sub(&tvrecv, &rec->rec_tv);
46                  rtt = tvrecv.tv_sec * 1000.0 + tvrecv.tv_usec / 1000.0;
47                  printf("  %.3f ms", rtt);
48                  if (code == -1) /* port unreachable; at destination */
49                      done++;
50                  else if (code >= 0)
51                      printf(" (ICMP %s)", (*pr->icmpcode) (code));
52              }
53              fflush(stdout);
54          }
55          printf("\n");
56      }
57  }
```

traceroute/traceloop.c

Figure 25.18 `traceloop` function: main processing loop.

Main loop; set TTL or hop limit and send three probes

17-28 The main loop of the function is a double nested `for` loop. The outer loop starts the TTL or hop limit at 1, and increases it by 1, while the inner loop sends three probes (UDP datagrams) to the destination. Each time the TTL changes, we call `setsockopt` to set the new value using either the `IP_TTL` or `IPV6_UNICAST_HOPS` socket option.

Each time around the outer loop we initialize the socket address structure pointed to by `salast` to 0. This structure will be compared to the socket address structure returned by `recvfrom` when the ICMP message is read, and if the two structures are different, the IP address from the new structure is printed. Using this technique the IP address corresponding to the first probe for each TTL is printed, and should the IP address change for a given value of the TTL (say a route changes while we are running the program) the new IP address is then printed.

Set destination port and send UDP datagram

29-30 Each time a probe packet is sent, the destination port in the `sasend` socket address structure is changed by calling our `sock_set_port` function. The reason for changing the port for each probe is that when we reach the final destination, all three probes are sent to a different port, and hopefully at least one of the ports is not in use. `sendto` sends the UDP datagram.

Read ICMP message

31-53 One of our functions `recv_v4` or `recv_v6` calls `recvfrom` to read and process the ICMP messages that are returned. These two functions return –3 if a timeout occurs (telling us to send another probe if we haven't sent three for this TTL), –2 if an ICMP "time exceeded in transit" error is received, –1 if an ICMP "port unreachable" error is received (which means we have reached the final destination), or the nonnegative ICMP code if some other ICMP destination unreachable error is received.

Print reply

33-53 As we mentioned earlier, if this is the first reply for a given TTL, or if the IP address of the node sending the ICMP message has changed for this TTL, we print the hostname and IP address, or just the IP address (if the call to `getnameinfo` doesn't return the hostname). The RTT is calculated as the time difference from when we sent the probe to the time the ICMP message is returned and printed.

Our `recv_v4` function is shown in Figure 25.19.

traceroute/recv_v4.c

```
 1 #include    "trace.h"

 2 /*
 3  * Return: -3 on timeout
 4  *         -2 on ICMP time exceeded in transit (caller keeps going)
 5  *         -1 on ICMP port unreachable (caller is done)
 6  *       >= 0 return value is some other ICMP unreachable code
 7  */

 8 int
 9 recv_v4(int seq, struct timeval *tv)
10 {
11     int     hlen1, hlen2, icmplen;
12     socklen_t len;
13     ssize_t n;
14     struct ip *ip, *hip;
15     struct icmp *icmp;
16     struct udphdr *udp;
```

```
17      alarm(3);
18      for ( ; ; ) {
19          len = pr->salen;
20          n = recvfrom(recvfd, recvbuf, sizeof(recvbuf), 0, pr->sarecv, &len);
21          if (n < 0) {
22              if (errno == EINTR)
23                  return (-3);    /* alarm expired */
24              else
25                  err_sys("recvfrom error");
26          }
27          Gettimeofday(tv, NULL); /* get time of packet arrival */
28          ip = (struct ip *) recvbuf;     /* start of IP header */
29          hlen1 = ip->ip_hl << 2; /* length of IP header */
30          icmp = (struct icmp *) (recvbuf + hlen1);   /* start of ICMP header */
31          if ( (icmplen = n - hlen1) < 8)
32              err_quit("icmplen (%d) < 8", icmplen);
33          if (icmp->icmp_type == ICMP_TIMXCEED &&
34              icmp->icmp_code == ICMP_TIMXCEED_INTRANS) {
35              if (icmplen < 8 + 20 + 8)
36                  err_quit("icmplen (%d) < 8 + 20 + 8", icmplen);
37              hip = (struct ip *) (recvbuf + hlen1 + 8);
38              hlen2 = hip->ip_hl << 2;
39              udp = (struct udphdr *) (recvbuf + hlen1 + 8 + hlen2);
40              if (hip->ip_p == IPPROTO_UDP &&
41                  udp->uh_sport == htons(sport) &&
42                  udp->uh_dport == htons(dport + seq))
43                  return (-2);    /* we hit an intermediate router */
44          } else if (icmp->icmp_type == ICMP_UNREACH) {
45              if (icmplen < 8 + 20 + 8)
46                  err_quit("icmplen (%d) < 8 + 20 + 8", icmplen);
47              hip = (struct ip *) (recvbuf + hlen1 + 8);
48              hlen2 = hip->ip_hl << 2;
49              udp = (struct udphdr *) (recvbuf + hlen1 + 8 + hlen2);
50              if (hip->ip_p == IPPROTO_UDP &&
51                  udp->uh_sport == htons(sport) &&
52                  udp->uh_dport == htons(dport + seq)) {
53                  if (icmp->icmp_code == ICMP_UNREACH_PORT)
54                      return (-1);    /* have reached destination */
55                  else
56                      return (icmp->icmp_code);   /* 0, 1, 2, ... */
57              }
58          } else if (verbose) {
59              printf(" (from %s: type = %d, code = %d)\n",
60                     Sock_ntop_host(pr->sarecv, pr->salen),
61                     icmp->icmp_type, icmp->icmp_code);
62          }
63          /* Some other ICMP error, recvfrom() again */
64      }
65  }
```

traceroute/recv_v4.c

Figure 25.19 recv_v4 function: read and process ICMPv4 messages.

Set alarm and read each ICMP message

17-27 An alarm is set for 3 seconds in the future and the function enters a loop that calls `recvfrom`, reading each ICMPv4 message returned on the raw socket.

> This function contains the same race condition that we described in Section 18.5 with regard to a `SIGALRM` interrupting a read operation.

Get pointer to ICMP header

28-32 `ip` points to the beginning of the IPv4 header (recall that a read on a raw socket always returns the IP header), and `icmp` points to the beginning of the ICMP header. Figure 25.20 shows the various headers, pointers, and lengths used by the code.

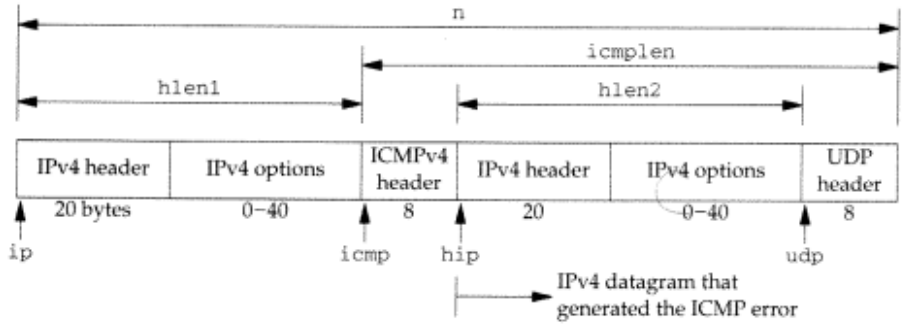


Figure 25.20 Headers, pointers, and lengths in processing ICMPv4 error.

Process ICMP time exceeded in transit message

33-43 If the ICMP message is a "time exceeded in transit" message, it is possibly a reply to one of our probes. `hip` points to the IPv4 header that is returned in the ICMP message, following the 8-byte ICMP header. `udp` points to the UDP header that follows. If the ICMP message was generated by a UDP datagram and if the source and destination ports of that datagram are the values that we sent, then this is a reply to our probe from an intermediate router.

Process ICMP port unreachable message

44-57 If the ICMP message is a "destination unreachable," then we look at the UDP header returned in the ICMP message to see if the message is a response to our probe. If so, and if the ICMP code is "port unreachable" we return –1 as we have reached the final destination. If the ICMP message is from one of our probes, but it is not a "port unreachable," then that ICMP code value is returned. A common example of this is a firewall returning some other unreachable code for the destination host that we are probing.

Handle other ICMP messages

58-62 All other ICMP messages are printed if the `-v` flag was specified.

The next function, `recv_v6`, is shown in Figure 25.22 and is the IPv6 equivalent to the previously described function. This function is nearly identical to `recv_v4`, except

for the different constant names and the different structure member names. Also, the size of the IPv6 header is a fixed 40 bytes, while with IPv4 we had to fetch the header length field and multiply it by 4 to account for any IP options. Figure 25.21 shows the various headers, pointers, and lengths used by the code.

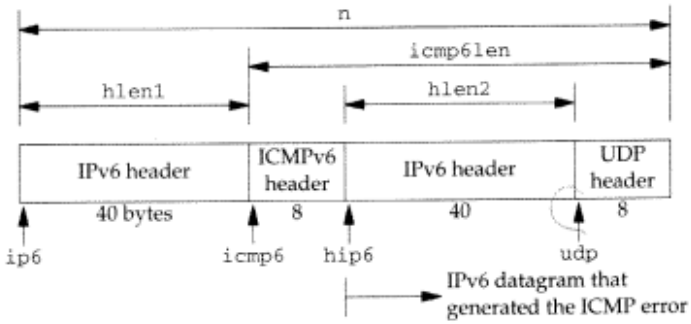


Figure 25.21 Headers, pointers, and lengths in processing ICMPv6 error.

We define two functions, `icmpcode_v4` and `icmpcode_v6`, that can be called from the bottom of the `traceloop` function to print a description string corresponding to an ICMP destination unreachable error. Figure 25.23 (p. 684) shows just the IPv6 function. The IPv4 function is similar, albeit longer, as there are more ICMPv4 destination unreachable codes (Figure A.15).

The final function in our Traceroute program is our `SIGALRM` handler, the `sig_alrm` function shown in Figure 25.24 (p. 684). All this function does is return, causing an error return of `EINTR` from the `recvfrom` in either `recv_v4` or `recv_v6`.

traceroute/recv_v6.c

```
 1 #include    "trace.h"

 2 /*
 3  * Return: -3 on timeout
 4  *         -2 on ICMP time exceeded in transit (caller keeps going)
 5  *         -1 on ICMP port unreachable (caller is done)
 6  *       >= 0 return value is some other ICMP unreachable code
 7  */

 8 int
 9 recv_v6(int seq, struct timeval *tv)
10 {
11 #ifdef  IPV6
12     int     hlen1, hlen2, icmp6len;
13     ssize_t n;
14     socklen_t len;
15     struct ip6_hdr *ip6, *hip6;
16     struct icmp6_hdr *icmp6;
17     struct udphdr *udp;

18     alarm(3);
```

```
19      for ( ; ; ) {
20          len = pr->salen;
21          n = recvfrom(recvfd, recvbuf, sizeof(recvbuf), 0, pr->sarecv, &len);
22          if (n < 0) {
23              if (errno == EINTR)
24                  return (-3);    /* alarm expired */
25              else
26                  err_sys("recvfrom error");
27          }
28          Gettimeofday(tv, NULL); /* get time of packet arrival */
29          ip6 = (struct ip6_hdr *) recvbuf;   /* start of IPv6 header */
30          hlen1 = sizeof(struct ip6_hdr);
31          icmp6 = (struct icmp6_hdr *) (recvbuf + hlen1);     /* ICMP hdr */
32          if ( (icmp6len = n - hlen1) < 8)
33              err_quit("icmp6len (%d) < 8", icmp6len);
34          if (icmp6->icmp6_type == ICMP6_TIME_EXCEEDED &&
35              icmp6->icmp6_code == ICMP6_TIME_EXCEED_TRANSIT) {
36              if (icmp6len < 8 + 40 + 8)
37                  err_quit("icmp6len (%d) < 8 + 40 + 8", icmp6len);
38              hip6 = (struct ip6_hdr *) (recvbuf + hlen1 + 8);
39              hlen2 = sizeof(struct ip6_hdr);
40              udp = (struct udphdr *) (recvbuf + hlen1 + 8 + hlen2);
41              if (hip6->ip6_nxt == IPPROTO_UDP &&
42                  udp->uh_sport == htons(sport) &&
43                  udp->uh_dport == htons(dport + seq))
44                  return (-2);    /* we hit an intermediate router */
45          } else if (icmp6->icmp6_type == ICMP6_DST_UNREACH) {
46              if (icmp6len < 8 + 40 + 8)
47                  err_quit("icmp6len (%d) < 8 + 40 + 8", icmp6len);
48              hip6 = (struct ip6_hdr *) (recvbuf + hlen1 + 8);
49              hlen2 = 40;
50              udp = (struct udphdr *) (recvbuf + hlen1 + 8 + hlen2);
51              if (hip6->ip6_nxt == IPPROTO_UDP &&
52                  udp->uh_sport == htons(sport) &&
53                  udp->uh_dport == htons(dport + seq)) {
54                  if (icmp6->icmp6_code == ICMP6_DST_UNREACH_NOPORT)
55                      return (-1);    /* have reached destination */
56                  else
57                      return (icmp6->icmp6_code);     /* 0, 1, 2, ... */
58              }
59          } else if (verbose) {
60              printf(" (from %s: type = %d, code = %d)\n",
61                     Sock_ntop_host(pr->sarecv, pr->salen),
62                     icmp6->icmp6_type, icmp6->icmp6_code);
63          }
64          /* Some other ICMP error, recvfrom() again */
65      }
66  #endif
67  }
```

traceroute/recv_v6.c

Figure 25.22 recv_v6 function: read and process ICMPv6 messages.

traceroute/icmpcode_v6.c

```
1 #include    "trace.h"

2 char *
3 icmpcode_v6(int code)
4 {
5     switch (code) {
6     case ICMP6_DST_UNREACH_NOROUTE:
7         return ("no route to host");
8     case ICMP6_DST_UNREACH_ADMIN:
9         return ("administratively prohibited");
10    case ICMP6_DST_UNREACH_NOTNEIGHBOR:
11        return ("not a neighbor");
12    case ICMP6_DST_UNREACH_ADDR:
13        return ("address unreachable");
14    case ICMP6_DST_UNREACH_NOPORT:
15        return ("port unreachable");
16    default:
17        return ("[unknown code]");
18    }
19 }
```

traceroute/icmpcode_v6.c

Figure 25.23 Return the string corresponding to an ICMPv6 unreachable code.

traceroute/sig_alrm.c

```
1 #include    "trace.h"

2 void
3 sig_alrm(int signo)
4 {
5     return;                    /* just interrupt the recvfrom() */
6 }
```

traceroute/sig_alrm.c

Figure 25.24 `sig_alrm` function.

Example

We first show an example using IPv4:

```
solaris # traceroute gemini.tuc.noao.edu
traceroute to gemini.tuc.noao.edu (140.252.3.54): 30 hops max, 12 data bytes
 1 gw.kohala.com (206.62.226.62)  3.839 ms  3.595 ms  3.722 ms
 2 tuc-1-s1-9.rtd.net (206.85.40.73)  42.014 ms  21.078 ms  18.826 ms
 3 frame-gw.ttn.ep.net (198.32.152.9)  39.283 ms  24.598 ms  50.037 ms
 4 tucson-nap-1.arizona.edu (198.32.152.248)  44.350 ms  78.109 ms  47.003 ms
 5 Butch-ENET-BONE.Telcom.Arizona.EDU (128.196.11.5)  29.849 ms  46.664 ms  83.571 ms
 6 gateway.tuc.noao.edu (140.252.104.1)  37.376 ms  36.430 ms  30.555 ms
 7 gemini.tuc.noao.edu (140.252.3.54)  70.476 ms  43.555 ms  88.716 ms
```

Here is an example using IPv6. We have wrapped the long lines for a more readable output.

```
solaris # traceroute ipng9.ipng.nist.gov
traceroute to ipng9.ipng.nist.gov (5f00:3100:8106:3300:0:c0:3302:5a):
30 hops max, 12 data bytes
 1   6bone-router.cisco.inner.net (5f00:6d00:c01f:700:1:60:3e11:6770)
           185.869 ms *  127.082 ms
 2   buzzcut.ipv6.nrl.navy.mil (5f00:3000:84fa:5a00::5)
           187.736 ms  199.455 ms  172.839 ms
 3   ipng9.ipng.nist.gov (5f00:3100:8106:3300:0:c0:3302:5a)
           206.762 ms *  441.081 ms
```

In this example the second probe with a hop limit of 1 timed out, as did the second probe with a hop limit of 3.

25.7 An ICMP Message Daemon

Receiving asynchronous ICMP errors on a UDP socket has been, and continues to be, a problem. The ICMP errors are received by the kernel but are rarely delivered to the application that needs to know. In the sockets API we have seen that it requires connecting the UDP socket to one IP address to receive these errors (Section 8.11). The reason for this limitation is that the only error return from `recvfrom` is an integer `errno` code, and if the application sends datagrams to multiple destinations and then calls `recvfrom`, this function cannot tell the application which datagram encountered an error.

We will see in Section 31.4 that XTI improves on this (slightly) by returning an error from its equivalent of `recvfrom` and then the application must call another function (`t_rcvuderr`) to obtain the actual error and the destination address and port number from the datagram that caused the error. The problem with this solution, however, is that the kernel probably keeps information on only one of these asynchronous errors at a time. If the application sends (say) three datagrams, and two elicit ICMP errors, only one is returned to the application.

In this section we provide a different solution that does not require any kernel changes. We provide an ICMP message daemon, `icmpd`, that creates a raw ICMPv4 socket and a raw ICMPv6 socket and receives all ICMP messages that the kernel passes these two raw sockets. It also creates a Unix domain stream socket, `bind`s it to the pathname `/tmp/icmpd`, and listens for incoming client `connect`s to this pathname. We show this in Figure 25.25.

A UDP application (which is a client to the daemon) first creates its UDP socket, the socket for which it wants to receive asynchronous errors. The application must `bind` an ephemeral port to this socket, for reasons we discuss later. It then creates a Unix domain socket and `connect`s to this daemon's well-known pathname. We show this in Figure 25.26.

The application then "passes" its UDP socket to the daemon across the Unix domain connection using *descriptor passing*, as we described in Section 14.7. This gives the daemon a copy of the socket so that it can call `getsockname` and obtain the port number bound to the socket. We show this passing of the socket in Figure 25.27.

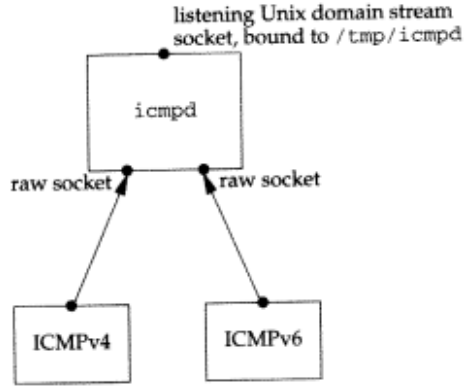


Figure 25.25 `icmpd` daemon: initial sockets created.

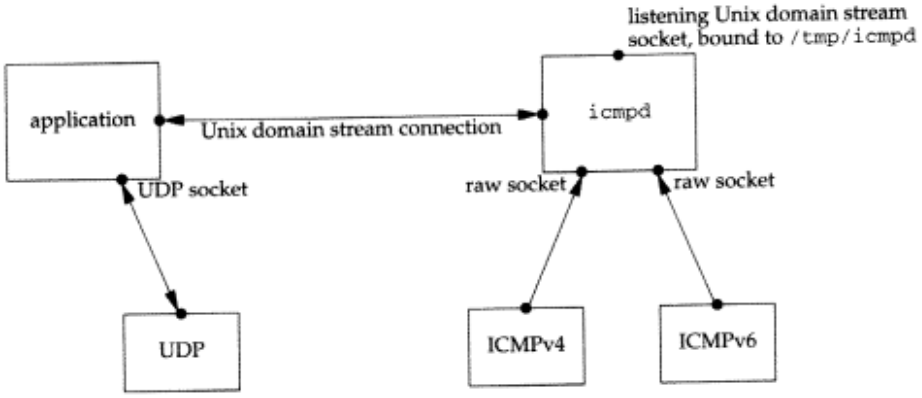


Figure 25.26 Application creates its UDP socket and a Unix domain connection to the daemon.

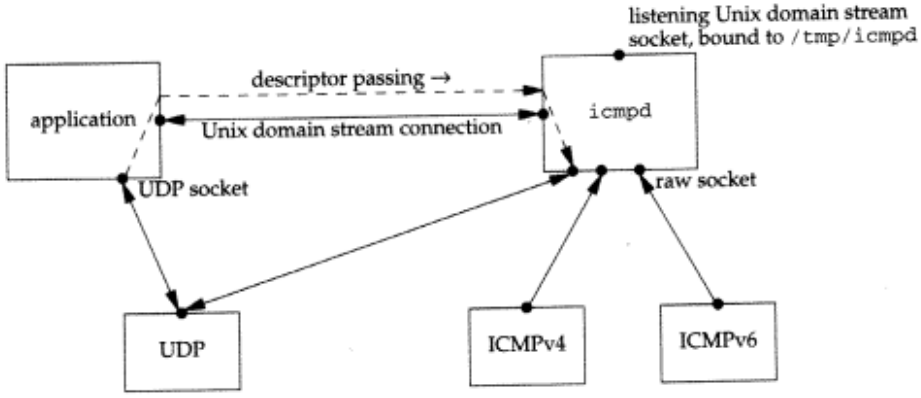


Figure 25.27 Passing UDP socket to daemon across Unix domain connection.

After the daemon obtains the port number bound to the UDP socket, it closes its copy of the socket, taking us back to the arrangement shown in Figure 25.26.

> If the host supports credential passing (Section 14.8), the application could also send its credentials to the daemon. The daemon could then check whether this user should be allowed access to this facility.

From this point on any ICMP errors that the daemon receives that are in response to UDP datagrams sent from the port bound to the application's UDP socket cause the daemon to send a message (which we describe shortly) across the Unix domain socket to the application. The application must therefore use `select` or `poll` awaiting data on either the UDP socket or the Unix domain socket.

We now look at the source code for an application using this daemon, and then the daemon itself. We start with Figure 25.28, our header that is included by both the application and the daemon.

icmpd/unpicmpd.h

```
 1 #ifndef __unpicmp_h
 2 #define __unpicmp_h

 3 #include    "unp.h"

 4 #define ICMPD_PATH      "/tmp/icmpd"    /* server's well-known pathname */

 5 struct icmpd_err {
 6     int     icmpd_errno;        /* EHOSTUNREACH, EMSGSIZE, ECONNREFUSED */
 7     char    icmpd_type;         /* actual ICMPv[46] type */
 8     char    icmpd_code;         /* actual ICMPv[46] code */
 9     socklen_t icmpd_len;        /* length of sockaddr{} that follows */
10     struct sockaddr icmpd_dest; /* may be bigger */
11     char    icmpd_fill[MAXSOCKADDR - sizeof(struct sockaddr)];
12 };

13 #endif  /* __unpicmp_h */
```

icmpd/unpicmpd.h

Figure 25.28 `unpicmpd.h` header.

4-12 We define the server's well-known pathname and the `icmpd_err` structure that is passed from the server to the application whenever an ICMP message is received that should be passed to this application.

6-8 A problem is that the ICMPv4 message types differ numerically (and sometimes conceptually) from the ICMPv6 message types (Figures A.15 and A.16). The actual ICMP *type* and *code* values are returned, but we also map these into an `errno` value (`icmpd_errno`), similar to the final column in Figures A.15 and A.16. The application can deal with this value, instead of the protocol-dependent ICMPv4 or ICMPv6 values. Figure 25.29 shows the ICMP messages that are handled, and their mapping into an `errno` value. The daemon returns five types of ICMP errors.

1. Port unreachable, indicating that no socket is bound to the destination port at the destination IP address.

icmpd_errno	ICMPv4 error	ICMPv6 error
ECONNREFUSED	port unreachable	port unreachable
EMSGSIZE	fragmentation needed but DF set	packet too big
EHOSTUNREACH	time exceeded	time exceeded
EHOSTUNREACH	source quench	
EHOSTUNREACH	all other destination unreachables	all other destination unreachables

Figure 25.29 icmpd_errno mapping from ICMPv4 and ICMPv6 errors.

2. Packet too big, which is used with path MTU discovery. Currently there is no API defined to allow a UDP application to perform path MTU discovery. What often happens on kernels that support path MTU discovery for UDP is the receipt of this ICMP error causes the kernel to record the new path MTU value in the kernel's routing table, but the UDP application that sent the datagram that got discarded is not notified. Instead, the application must time out and retransmit the datagram, in which case the kernel will find the new (and smaller) MTU in its routing table, and the kernel will then fragment the datagram. Passing this error back to the application lets the application retransmit sooner, and perhaps the application can reduce the size of the datagrams it sends.
3. The time exceeded error is normally seen with a code of 0, indicating that either the IPv4 TTL or IPv6 hop limit reached 0. This often indicates a routing loop, which might be a transient error.
4. ICMPv4 source quenches, while deprecated by RFC 1812 [Baker 1995], may be sent by routers (or by misconfigured hosts acting as routers). They indicate that the packet has been discarded, and we therefore treat them like a destination unreachable. Note that IPv6 does not have a source quench error.
5. All other destination unreachables indicate that the packet has been discarded.

10 The icmpd_dest member is a socket address structure containing the destination IP address and port of the datagram that generated the ICMP error. This member will be either a sockaddr_in structure for IPv4 or a sockaddr_in6 structure for IPv6. If the application is sending datagrams to multiple destinations, it probably has one socket address structure per destination. By returning this information in a socket address structure, the application can compare it against its own structures to find the one that caused the error.

11 The icmpd_fill member pads out the icmpd_err structure to accommodate the largest possible socket address structure.

UDP Echo Client That Uses Our icmpd Daemon

We now modify our UDP echo client, the dg_cli function to use our icmpd daemon. Figure 25.30 shows the first half of the function.

icmpd/dgcli01.c

```
 1 #include    "unpicmpd.h"
 2 void
 3 dg_cli(FILE *fp, int sockfd, const SA *pservaddr, socklen_t servlen)
 4 {
 5     int     icmpfd, maxfdp1;
 6     char    sendline[MAXLINE], recvline[MAXLINE + 1];
 7     fd_set  rset;
 8     ssize_t n;
 9     struct timeval tv;
10     struct icmpd_err icmpd_err;
11     Sock_bind_wild(sockfd, pservaddr->sa_family);
12     icmpfd = Tcp_connect("/unix", ICMPD_PATH);
13     Write_fd(icmpfd, "1", 1, sockfd);
14     n = Read(icmpfd, recvline, 1);
15     if (n != 1 || recvline[0] != '1')
16         err_quit("error creating icmp socket, n = %d, char = %c",
17                  n, recvline[0]);
18     FD_ZERO(&rset);
19     maxfdp1 = max(sockfd, icmpfd) + 1;
```

icmpd/dgcli01.c

Figure 25.30 First half of `dg_cli` application.

2-3 The function arguments are the same as all previous versions of this function.

`bind` wildcard address and ephemeral port

11 We call our `sock_bind_wild` function to `bind` the wildcard IP address and an ephemeral port to the UDP socket. We do this so that the copy of this socket that we pass to the daemon has bound a port, as the daemon needs to know this port.

> The daemon could also do this `bind`, if a local port has not already been bound to the socket that it receives, but this does not work in all environments. Under SVR4 implementations, such as Solaris 2.5, in which sockets are not part of the kernel, when one process `bind`s a port to a shared socket, the other process with a copy of that socket gets strange errors when it tries to use the socket. The easiest solution is to require the application to `bind` the local port before passing the socket to the daemon.

Establish Unix domain connection to daemon

12 We call our `tcp_connect` function to create a Unix domain stream socket and `connect` to the daemon's well-known pathname. (Recall from Section 11.5 that our implementation of `getaddrinfo` supports Unix domain sockets.)

Send UDP socket to daemon, await daemon's reply

13-17 We call our `write_fd` function from Figure 14.13 to send a copy of our UDP socket to the daemon. We also send a single byte of data, the character "1," because some implementations do not like passing a descriptor without any data. The daemon sends back a single byte of data, consisting of the character "1" to indicate success. Any other reply indicates an error.

18-19 We initialize a descriptor set and calculate the first argument for `select` (the maximum of the two descriptors, plus one).

The last half of our client is shown in Figure 25.31. This is the loop that reads a line from standard input, sends the line to the server, reads back the server's reply, and writes the reply to standard output.

icmpd/dgcli01.c

```
20      while (Fgets(sendline, MAXLINE, fp) != NULL) {
21          Sendto(sockfd, sendline, strlen(sendline), 0, pservaddr, servlen);
22          tv.tv_sec = 5;
23          tv.tv_usec = 0;
24          FD_SET(sockfd, &rset);
25          FD_SET(icmpfd, &rset);
26          if ( (n = Select(maxfdp1, &rset, NULL, NULL, &tv)) == 0) {
27              fprintf(stderr, "socket timeout\n");
28              continue;
29          }
30          if (FD_ISSET(sockfd, &rset)) {
31              n = Recvfrom(sockfd, recvline, MAXLINE, 0, NULL, NULL);
32              recvline[n] = 0;    /* null terminate */
33              Fputs(recvline, stdout);
34          }
35          if (FD_ISSET(icmpfd, &rset)) {
36              if ( (n = Read(icmpfd, &icmpd_err, sizeof(icmpd_err))) == 0)
37                  err_quit("ICMP daemon terminated");
38              else if (n != sizeof(icmpd_err))
39                  err_quit("n = %d, expected %d", n, sizeof(icmpd_err));
40              printf("ICMP error: dest = %s, %s, type = %d, code = %d\n",
41                      Sock_ntop(&icmpd_err.icmpd_dest, icmpd_err.icmpd_len),
42                      strerror(icmpd_err.icmpd_errno),
43                      icmpd_err.icmpd_type, icmpd_err.icmpd_code);
44          }
45      }
46  }
```

icmpd/dgcli01.c

Figure 25.31 Last half of `dg_cli` application.

Call `select`

22–29 Since we are calling `select` we can easily place a timeout on our wait for the echo server's reply. We set this to 5 seconds, enable both descriptors for readability, and call `select`. If a timeout occurs, we print a message and go back to the top of the loop.

Print server's reply

30–34 If a datagram is returned by the server, we print it to standard output.

Handle ICMP error

35–44 If our Unix domain connection to the `icmpd` daemon is readable, we try to read an `icmpd_err` structure. If this succeeds, we print the relevant information that the daemon returns.

> `strerror` is an example of a simple, almost trivial, function that should be more portable than it is. First, ANSI C and Posix.1 say nothing about an error return from the function. The Solaris 2.5 manual page says that the function returns a null pointer if the argument is out of range. But this means code like

```
printf("%s", strerror(arg));
```

is incorrect, because `strerror` can return a null pointer. But the UnixWare 2.1 implementation, along with all the source code implementations that the author could find, handle an invalid argument by returning a pointer to a string such as "Unknown error." This makes sense and means the code above is fine. But Unix 98 changes this and says that because no return value is reserved to indicate an error, if the argument is out of range, the function sets `errno` to `EINVAL`. (They do not say anything about the returned pointer in the case of an error.) This means that completely conforming code must set `errno` to 0, call `strerror`, test whether `errno` equals `EINVAL`, and print some other message in case of an error.

UDP Echo Client Examples

We now show some examples of this client, before looking at the daemon source code. We first send datagrams to an IP address that is not connected to the Internet.

```
solaris % udpcli01 192.3.4.5 echo
hi there
socket timeout
and hello
socket timeout
```

We assume `icmpd` is running and expect ICMP host unreachables to be returned by some router, but none are received. Instead, our application times out. We show this to reiterate that a timeout is still required, and the generation of ICMP messages such as "host unreachable" may not occur. We then ran this example about 30 seconds later and did receive the expected ICMP error:

```
solaris % udpcli01 192.3.4.5 echo
hello
ICMP error: dest = 192.3.4.5.7, No route to host, type = 3, code = 1
```

Our next example sends a datagram to the standard echo server on a host that is not running the server. We receive an ICMPv4 port unreachable, as expected.

```
solaris % udpcli01 gemini.tuc.noao.edu echo
hello, world
ICMP error: dest = 140.252.4.54.7, Connection refused, type = 3, code = 3
```

`icmpd` Daemon

We start the description of our `icmpd` daemon with the `icmpd.h` header, shown in Figure 25.32.

`client` array

2-17 Since the daemon can handle any number of clients, we use an array of `client` structures to keep the information about each client. This is similar to the data structures we used in Section 6.8. In addition to the descriptor for the Unix domain connection to the client, we also store the address family of the client's UDP socket `AF_INET` or `AF_INET6`) and the port number bound to this socket. We also declare the function prototypes and the globals shared by these functions.

Figure 25.33 shows the first half of the `main` function.

icmpd/icmpd.h

```
1 #include    "unpicmpd.h"

2 struct client {
3     int     connfd;             /* Unix domain stream socket to client */
4     int     family;             /* AF_INET or AF_INET6 */
5     int     lport;              /* local port bound to client's UDP socket */
6     /* network byte ordered */
7 } client[FD_SETSIZE];

8                           /* globals */
9 int     fd4, fd6, listenfd, maxi, maxfd, nready;
10 fd_set  rset, allset;
11 socklen_t addrlen;
12 struct sockaddr *cliaddr;

13                  /* function prototypes */
14 int     readable_conn(int);
15 int     readable_listen(void);
16 int     readable_v4(void);
17 int     readable_v6(void);
```

icmpd/icmpd.h

Figure 25.32 `icmpd.h` header for `icmpd` daemon.

icmpd/icmpd.c

```
1 #include    "icmpd.h"

2 int
3 main(int argc, char **argv)
4 {
5     int     i, sockfd;

6     if (argc != 1)
7         err_quit("usage: icmpd");

8     maxi = -1;                  /* index into client[] array */
9     for (i = 0; i < FD_SETSIZE; i++)
10        client[i].connfd = -1;  /* -1 indicates available entry */
11    FD_ZERO(&allset);

12    fd4 = Socket(AF_INET, SOCK_RAW, IPPROTO_ICMP);
13    FD_SET(fd4, &allset);
14    maxfd = fd4;

15 #ifdef  IPV6
16    fd6 = Socket(AF_INET6, SOCK_RAW, IPPROTO_ICMPV6);
17    FD_SET(fd6, &allset);
18    maxfd = max(maxfd, fd6);
19 #endif

20    listenfd = Tcp_listen("/unix", ICMPD_PATH, &addrlen);
21    FD_SET(listenfd, &allset);
22    maxfd = max(maxfd, listenfd);
23    cliaddr = Malloc(addrlen);
```

icmpd/icmpd.c

Figure 25.33 First half of `main` function: create sockets.

Initialize `client` array

9-10 The `client` array is initialized by setting the connected socket member to –1.

Create sockets

12-23 Three sockets are created: a raw ICMPv4 socket, a raw ICMPv6 socket, and a Unix domain stream socket. We call our `tcp_listen` function to create the latter, which also `bind`s its well-known pathname to the socket and calls `listen`. This is the socket to which clients `connect`. The maximum descriptor is also calculated for `select` and a socket address structure is allocated for calls to `accept`.

Figure 25.34 shows the second half of the `main` function, which is an infinite loop that calls `select`, waiting for any of the daemon's descriptors to be readable.

icmpd/icmpd.c

```
24      for ( ; ; ) {
25          rset = allset;
26          nready = Select(maxfd + 1, &rset, NULL, NULL, NULL);

27          if (FD_ISSET(listenfd, &rset))
28              if (readable_listen() <= 0)
29                  continue;

30          if (FD_ISSET(fd4, &rset))
31              if (readable_v4() <= 0)
32                  continue;

33  #ifdef  IPV6
34          if (FD_ISSET(fd6, &rset))
35              if (readable_v6() <= 0)
36                  continue;
37  #endif

38          for (i = 0; i <= maxi; i++) {   /* check all clients for data */
39              if ( (sockfd = client[i].connfd) < 0)
40                  continue;
41              if (FD_ISSET(sockfd, &rset))
42                  if (readable_conn(i) <= 0)
43                      break;      /* no more readable descriptors */
44          }
45      }
46      exit(0);
47  }
```

icmpd/icmpd.c

Figure 25.34 Second half of `main` function: handle readable descriptor.

Check listening Unix domain socket

27-29 The listening Unix domain socket is tested first and if ready, `readable_listen` is called. The variable `nready`, the number of descriptors that `select` returns as readable, is a global. Each of our `readable_XXX` function decrements this variable and returns its new value as the return value of the function. When this value reaches 0, all the readable descriptors have been processed and `select` is called again.

Check raw ICMP sockets

30–37 The raw ICMPv4 socket is tested, and then the raw ICMPv6 socket.

Check connected Unix domain sockets

38–44 We then check whether any of the connected Unix domain sockets are readable. Readability on any of these sockets means that the client has sent a descriptor, or that the client has terminated.

Figure 25.35 shows the `readable_listen` function, called when the daemon's listening socket is readable. This indicates a new client connection.

icmpd/readable_listen.c

```
 1 #include    "icmpd.h"

 2 int
 3 readable_listen(void)
 4 {
 5     int     i, connfd;
 6     socklen_t clilen;

 7     clilen = addrlen;
 8     connfd = Accept(listenfd, cliaddr, &clilen);

 9         /* find first available client[] structure */
10     for (i = 0; i < FD_SETSIZE; i++)
11         if (client[i].connfd < 0) {
12             client[i].connfd = connfd;  /* save descriptor */
13             break;
14         }
15     if (i == FD_SETSIZE)
16         err_quit("too many clients");
17     printf("new connection, i = %d, connfd = %d\n", i, connfd);

18     FD_SET(connfd, &allset);    /* add new descriptor to set */
19     if (connfd > maxfd)
20         maxfd = connfd;         /* for select() */
21     if (i > maxi)
22         maxi = i;               /* max index in client[] array */

23     return (--nready);
24 }
```

icmpd/readable_listen.c

Figure 25.35 Handle new client connections.

7–23 The connection is accepted and the first available entry in the `client` array is used. The code in this function was copied from the beginning of Figure 6.22.

When a connected socket is readable, our `readable_conn` function is called (Figure 25.36), and its argument is the index of this client in the `client` array.

Read client data and possibly a descriptor

13–18 We call our `read_fd` function from Figure 14.11 to read the data and possibly a descriptor. If the return value is 0, the client has closed its end of the connection, possibly by terminating.

icmpd/readable_conn.c

```
1 #include    "icmpd.h"

2 int
3 readable_conn(int i)
4 {
5     int     unixfd, recvfd;
6     char    c;
7     ssize_t n;
8     socklen_t len;
9     union {
10        char    buf[MAXSOCKADDR];
11        struct sockaddr sock;
12    } un;

13    unixfd = client[i].connfd;
14    recvfd = -1;
15    if ( (n = Read_fd(unixfd, &c, 1, &recvfd)) == 0) {
16        err_msg("client %d terminated, recvfd = %d", i, recvfd);
17        goto clientdone;        /* client probably terminated */
18    }
19        /* data from client; should be descriptor */
20    if (recvfd < 0) {
21        err_msg("read_fd did not return descriptor");
22        goto clienterr;
23    }
```

icmpd/readable_conn.c

Figure 25.36 Read data and possible descriptor from client.

One design decision was whether to use a Unix domain stream socket between the application and the daemon, or a Unix domain datagram socket. The application's UDP socket can be passed over either type of Unix domain socket. The reason we use a stream socket is to detect when a client terminates. All its descriptors are automatically closed when it terminates, including its Unix domain connection to the daemon, which tells the daemon to remove this client from the `client` array. Had we used a datagram socket, we would not know when the client terminates.

19–23 If the client has not closed the connection, then we expect a descriptor.

The second half of our `readable_conn` function is shown in Figure 25.37.

Get port number bound to UDP socket

24–28 `getsockname` is called so the daemon can obtain the port number bound to the socket. Since we do not know what size buffer to allocate for the socket address structure, we declare a `union` of a character array and a generic socket address structure. This guarantees that the character array is suitably aligned for a socket address structure, something that we are not guaranteed if we just declared a character array by itself. We discussed this problem with Figure 11.7, and in that program we called `malloc` to guarantee the alignment.

29–36 The address family of the socket is stored in the `client` structure, along with the port number. If the port number is 0, we call our `sock_bind_wild` function to `bind` the wildcard address and an ephemeral port to the socket, but as we mentioned earlier, this does not work on SVR4 implementations.

icmpd/readable_conn.c

```
24      len = sizeof(un.buf);
25      if (getsockname(recvfd, (SA *) un.buf, &len) < 0) {
26          err_ret("getsockname error");
27          goto clienterr;
28      }
29      client[i].family = un.sock.sa_family;
30      if ((client[i].lport = sock_get_port(&un.sock, len)) == 0) {
31          client[i].lport = sock_bind_wild(recvfd, client[i].family);
32          if (client[i].lport <= 0) {
33              err_ret("error binding ephemeral port");
34              goto clienterr;
35          }
36      }
37      Write(unixfd, "1", 1);       /* tell client all OK */
38      FD_SET(unixfd, &allset);
39      if (unixfd > maxfd)
40          maxfd = unixfd;
41      if (i > maxi)
42          maxi = i;
43      Close(recvfd);              /* all done with client's UDP socket */
44      return (--nready);
45    clienterr:
46      Write(unixfd, "0", 1);       /* tell client error occurred */
47    clientdone:
48      Close(unixfd);
49      if (recvfd >= 0)
50          Close(recvfd);
51      FD_CLR(unixfd, &allset);
52      client[i].connfd = -1;
53      return (--nready);
54  }
```

icmpd/readable_conn.c

Figure 25.37 Get port number that client has bound to its UDP socket.

Tell client OK

37–42 One byte consisting of the character "1" is sent back to the client. The new descriptor is added to the set of descriptors for `select` and `maxfd` and `maxi` are updated if necessary.

`close` client's UDP socket

43 We are finished with the client's UDP socket and `close` it. This descriptor was passed to us by the client and is therefore a copy; hence the UDP socket is still open in the client.

Handle errors and termination of client

45–53 If an error occurs, a byte of "0" is written back to the client. When the client terminates, our end of the Unix domain connection is closed, and that descriptor is removed from the set of descriptors for `select`. The `connfd` member of the `client` structure is set to –1, indicating it is available.

Our `readable_v4` function is called when the raw ICMPv4 socket is readable. We show the first half in Figure 25.38. This code is similar to the ICMPv4 code shown earlier in Figures 25.8 and 25.19.

icmpd/readable_v4.c

```
 1 #include    "icmpd.h"
 2 #include    <netinet/in_systm.h>
 3 #include    <netinet/ip.h>
 4 #include    <netinet/ip_icmp.h>
 5 #include    <netinet/udp.h>

 6 int
 7 readable_v4(void)
 8 {
 9     int     i, hlen1, hlen2, icmplen, sport;
10     char    buf[MAXLINE];
11     char    srcstr[INET_ADDRSTRLEN], dststr[INET_ADDRSTRLEN];
12     ssize_t n;
13     socklen_t len;
14     struct ip *ip, *hip;
15     struct icmp *icmp;
16     struct udphdr *udp;
17     struct sockaddr_in from, dest;
18     struct icmpd_err icmpd_err;

19     len = sizeof(from);
20     n = Recvfrom(fd4, buf, MAXLINE, 0, (SA *) &from, &len);

21     printf("%d bytes ICMPv4 from %s:",
22            n, Sock_ntop_host((SA *) &from, len));

23     ip = (struct ip *) buf;      /* start of IP header */
24     hlen1 = ip->ip_hl << 2;      /* length of IP header */

25     icmp = (struct icmp *) (buf + hlen1);   /* start of ICMP header */
26     if ( (icmplen = n - hlen1) < 8)
27         err_quit("icmplen (%d) < 8", icmplen);

28     printf(" type = %d, code = %d\n", icmp->icmp_type, icmp->icmp_code);
```

icmpd/readable_v4.c

Figure 25.38 Process received ICMPv4 datagram, first half.

This function prints some information about every received ICMPv4 message. This was done for debugging when developing this daemon and could be output based on a command-line argument.

Figure 25.39 shows the last half of our `readable_v4` function.

icmpd/readable_v4.c

```
29      if (icmp->icmp_type == ICMP_UNREACH ||
30          icmp->icmp_type == ICMP_TIMXCEED ||
31          icmp->icmp_type == ICMP_SOURCEQUENCH) {
32          if (icmplen < 8 + 20 + 8)
33              err_quit("icmplen (%d) < 8 + 20 + 8", icmplen);

34          hip = (struct ip *) (buf + hlen1 + 8);
35          hlen2 = hip->ip_hl << 2;
36          printf("\tsrcip = %s, dstip = %s, proto = %d\n",
37                 Inet_ntop(AF_INET, &hip->ip_src, srcstr, sizeof(srcstr)),
38                 Inet_ntop(AF_INET, &hip->ip_dst, dststr, sizeof(dststr)),
39                 hip->ip_p);
40          if (hip->ip_p == IPPROTO_UDP) {
41              udp = (struct udphdr *) (buf + hlen1 + 8 + hlen2);
42              sport = udp->uh_sport;

43                  /* find client's Unix domain socket, send headers */
44              for (i = 0; i <= maxi; i++) {
45                  if (client[i].connfd >= 0 &&
46                      client[i].family == AF_INET &&
47                      client[i].lport == sport) {

48                      bzero(&dest, sizeof(dest));
49                      dest.sin_family = AF_INET;
50  #ifdef  HAVE_SOCKADDR_SA_LEN
51                      dest.sin_len = sizeof(dest);
52  #endif
53                      memcpy(&dest.sin_addr, &hip->ip_dst,
54                             sizeof(struct in_addr));
55                      dest.sin_port = udp->uh_dport;

56                      icmpd_err.icmpd_type = icmp->icmp_type;
57                      icmpd_err.icmpd_code = icmp->icmp_code;
58                      icmpd_err.icmpd_len = sizeof(struct sockaddr_in);
59                      memcpy(&icmpd_err.icmpd_dest, &dest, sizeof(dest));

60                          /* convert type & code to reasonable errno value */
61                      icmpd_err.icmpd_errno = EHOSTUNREACH;   /* default */
62                      if (icmp->icmp_type == ICMP_UNREACH) {
63                          if (icmp->icmp_code == ICMP_UNREACH_PORT)
64                              icmpd_err.icmpd_errno = ECONNREFUSED;
65                          else if (icmp->icmp_code == ICMP_UNREACH_NEEDFRAG)
66                              icmpd_err.icmpd_errno = EMSGSIZE;
67                      }
68                      Write(client[i].connfd, &icmpd_err, sizeof(icmpd_err));
69                  }
70              }
71          }
72      }
73      return (--nready);
74  }
```

icmpd/readable_v4.c

Figure 25.39 Process received ICMPv4 datagram, second half.

Check message type, notify application

29-31 The only ICMPv4 messages that we pass to the application are destination unreachable, time exceeded, and source quench (Figure 25.29).

Check for UDP error, find client

34-42 `hip` points to the IP header that is returned following the ICMP header. This is the IP header of the datagram that elicited the ICMP error. We verify that this IP datagram is a UDP datagram and then fetch the source UDP port number from the UDP header following the IP header.

43-55 A search is made of all the `client` structures for a matching address family and port. If a match is found, an IPv4 socket address structure is built containing the destination IP address and port from the UDP datagram that caused the error.

Build `icmpd_err` structure

56-70 An `icmpd_err` structure is built that is sent to the client across the Unix domain connection to this client. The ICMPv4 message type and code are first mapped into an `errno` value, as described with Figure 25.29.

ICMPv6 errors are handled by our `readable_v6` function, the first half of which is shown in Figure 25.40. The ICMPv6 handling is similar to the code in Figures 25.10 and 25.22.

The second half of our `readable_v6` function is shown in Figure 25.41 (p. 701). This code is similar to Figure 25.39: it checks the type of ICMP error, checks that the datagram that caused the error was a UDP datagram, and then builds the `icmpd_err` structure that is sent to the client.

icmpd/readable_v6.c

```
 1 #include    "icmpd.h"
 2 #include    <netinet/in_systm.h>
 3 #include    <netinet/ip.h>
 4 #include    <netinet/ip_icmp.h>
 5 #include    <netinet/udp.h>

 6 #ifdef  IPV6
 7 #include    "ip6.h"             /* should be <netinet/ip6.h> */
 8 #include    "icmp6.h"           /* should be <netinet/icmp6.h> */
 9 #endif

10 int
11 readable_v6(void)
12 {
13 #ifdef  IPV6
14     int     i, hlen1, hlen2, icmp6len, sport;
15     char    buf[MAXLINE];
16     char    srcstr[INET6_ADDRSTRLEN], dststr[INET6_ADDRSTRLEN];
17     ssize_t n;
18     socklen_t len;
19     struct ip6_hdr *ip6, *hip6;
20     struct icmp6_hdr *icmp6;
21     struct udphdr *udp;
22     struct sockaddr_in6 from, dest;
23     struct icmpd_err icmpd_err;

24     len = sizeof(from);
25     n = Recvfrom(fd6, buf, MAXLINE, 0, (SA *) &from, &len);

26     printf("%d bytes ICMPv6 from %s:",
27            n, Sock_ntop_host((SA *) &from, len));

28     ip6 = (struct ip6_hdr *) buf;   /* start of IPv6 header */
29     hlen1 = sizeof(struct ip6_hdr);
30     if (ip6->ip6_nxt != IPPROTO_ICMPV6)
31         err_quit("next header not IPPROTO_ICMPV6");

32     icmp6 = (struct icmp6_hdr *) (buf + hlen1);
33     if ( (icmp6len = n - hlen1) < 8)
34         err_quit("icmp6len (%d) < 8", icmp6len);

35     printf(" type = %d, code = %d\n", icmp6->icmp6_type, icmp6->icmp6_code);
```

icmpd/readable_v6.c

Figure 25.40 Process received ICMPv6 datagram, first half.

icmpd/readable_v6.c

```
36      if (icmp6->icmp6_type == ICMP6_DST_UNREACH ||
37          icmp6->icmp6_type == ICMP6_PACKET_TOO_BIG ||
38          icmp6->icmp6_type == ICMP6_TIME_EXCEEDED) {
39          if (icmp6len < 8 + 40 + 8)
40              err_quit("icmp6len (%d) < 8 + 40 + 8", icmp6len);

41          hip6 = (struct ip6_hdr *) (buf + hlen1 + 8);
42          hlen2 = sizeof(struct ip6_hdr);
43          printf("\tsrcip = %s, dstip = %s, next hdr = %d\n",
44                 Inet_ntop(AF_INET6, &hip6->ip6_src, srcstr, sizeof(srcstr)),
45                 Inet_ntop(AF_INET6, &hip6->ip6_dst, dststr, sizeof(dststr)),
46                 hip6->ip6_nxt);
47          if (hip6->ip6_nxt == IPPROTO_UDP) {
48              udp = (struct udphdr *) (buf + hlen1 + 8 + hlen2);
49              sport = udp->uh_sport;

50                  /* find client's Unix domain socket, send headers */
51              for (i = 0; i <= maxi; i++) {
52                  if (client[i].connfd >= 0 &&
53                      client[i].family == AF_INET6 &&
54                      client[i].lport == sport) {

55                      bzero(&dest, sizeof(dest));
56                      dest.sin6_family = AF_INET6;
57  #ifdef  HAVE_SOCKADDR_SA_LEN
58                      dest.sin6_len = sizeof(dest);
59  #endif
60                      memcpy(&dest.sin6_addr, &hip6->ip6_dst,
61                             sizeof(struct in6_addr));
62                      dest.sin6_port = udp->uh_dport;

63                      icmpd_err.icmpd_type = icmp6->icmp6_type;
64                      icmpd_err.icmpd_code = icmp6->icmp6_code;
65                      icmpd_err.icmpd_len = sizeof(struct sockaddr_in6);
66                      memcpy(&icmpd_err.icmpd_dest, &dest, sizeof(dest));

67                          /* convert type & code to reasonable errno value */
68                      icmpd_err.icmpd_errno = EHOSTUNREACH;   /* default */
69                      if (icmp6->icmp6_type == ICMP6_DST_UNREACH) {
70                          if (icmp6->icmp6_code == ICMP_UNREACH_PORT)
71                              icmpd_err.icmpd_errno = ECONNREFUSED;
72                          else if (icmp6->icmp6_code == ICMP_UNREACH_NEEDFRAG)
73                              icmpd_err.icmpd_errno = EMSGSIZE;
74                      }
75                      Write(client[i].connfd, &icmpd_err, sizeof(icmpd_err));
76                  }
77              }
78          }
79      }
80      return (--nready);
81  #endif
82  }
```

icmpd/readable_v6.c

Figure 25.41 Process received ICMPv6 datagram, second half.

25.8 Summary

Raw sockets provide three capabilities:

1. We can read and write ICMPv4, IGMPv4, and ICMPv6 packets.
2. We can read and write IP datagrams with a protocol field that the kernel does not handle.
3. We can build our own IPv4 header, normally used for diagnostic purposes (or by hackers, unfortunately).

Two commonly used diagnostic tools, Ping and Traceroute, use raw sockets, and we have developed our own versions of both that support IPv4 and IPv6. We also developed our own `icmpd` daemon that provides access to ICMP errors for a UDP socket. This example also provides an example of descriptor passing across a Unix domain socket between a client and server that are unrelated.

Exercises

25.1 We said that almost all fields in an IPv6 header and all extension headers are available to the application through socket options or ancillary data. What information in an IPv6 datagram is *not* available to an application?

25.2 What happens in Figure 25.39 if for some reason the client stops reading from its Unix domain connection to the `icmpd` daemon, and lots of ICMP errors arrive for the client? What is the easiest solution?

25.3 If we specify the subnet-directed broadcast address to our Ping program, it works. That is, a broadcast ICMP echo request is sent as a link-layer broadcast, even though we do not set the `SO_BROADCAST` socket option. Why?

25.4 What happens with our Ping program if we ping the all-hosts multicast group, 224.0.0.1 on a multihomed host?

26

Datalink Access

26.1 Introduction

Providing access to the datalink layer for an application is a powerful feature that is available with most current operating systems. This provides the following capabilities:

- The ability to watch the packets received by the datalink layer, allowing programs such as `tcpdump` to be run on normal computer systems (as opposed to dedicated hardware devices to watch packets). When combined with the capability of the network interface to go into a *promiscuous mode*, this allows an application to watch all the packets on the local cable, not just the packets destined for the host on which the program is running.
- The ability to run certain programs as normal applications instead of as part of the kernel. For example, most Unix versions of an RARP server are normal applications that read RARP requests from the datalink (RARP requests are not IP datagrams) and then write the reply back to the datalink.

The three common methods to access the datalink layer under Unix are the BSD Packet Filter (BPF), the SVR4 Data Link Provider Interface (DLPI), and the Linux `SOCK_PACKET` interface. We present an overview of these three but then describe `libpcap`, the publicly available packet capture library. This library works with all three and using this library makes our programs independent of the actual datalink access provided by the operating system. We describe this library by developing a program that sends DNS queries to a name server (we build our own UDP datagrams and write them to a raw socket) and reading the reply using `libpcap` to determine if the name server enables UDP checksums.

26.2 BPF: BSD Packet Filter

4.4BSD and many other Berkeley-derived implementations support BPF, the BSD packet filter. The implementation of BPF is described in Chapter 31 of TCPv2. The history of BPF, a description of the BPF pseudomachine, and a comparison with the SunOS 4.1.x NIT packet filter is provided in [McCanne and Jacobson 1993].

Each datalink driver calls BPF right before a packet is transmitted and right after a packet is received, as shown in Figure 26.1.

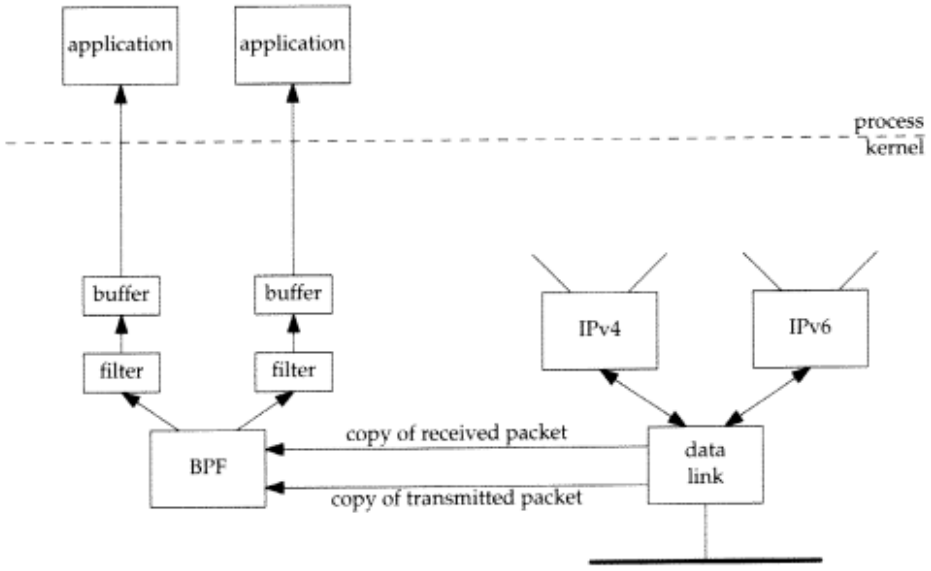


Figure 26.1 Packet capture using BPF.

Examples of these calls are in Figures 4.11 and 4.19 of TCPv2 for an Ethernet interface. The reason for calling BPF as soon as possible after reception and as late as possible before transmission is to provide accurate timestamps.

While it is not hard to provide a tap into the datalink to catch all packets, the power of BPF is in its filtering capability. Each application that opens a BPF device can load its own filter that is then applied by BPF to each packet. While some filters are simple (`udp or tcp` receives only UDP or TCP packets), others can examine fields in the packet headers for certain values. For example,

```
tcp and port 80 and tcp[13:1] & 0x7 != 0
```

was used in Chapter 14 of TCPv3 to collect only TCP segments to or from port 80 that had either the SYN, FIN, or RST flags on. The expression `tcp[13:1]` refers to the 1-byte value starting at byte offset 13 from the start of the TCP header.

BPF implements a register-based filter machine that applies the application-specific filters to each received packet. While one can write their own filter programs in the machine language of this pseudomachine (which is described on the BPF manual page), the simplest interface is to compile ASCII strings (such as the one beginning with `tcp`

that we just showed) into this machine language using the `pcap_compile` function that we describe in Section 26.6.

Three techniques are used by BPF to reduce its overhead.

1. The BPF filtering is within the kernel, which minimizes the amount of data copied from BPF to the application. This copy, from kernel space to user space, is expensive. If every packet were copied, BPF could have trouble keeping up with fast datalinks.
2. Only a portion of each packet is passed by BPF to the application. This is called the *capture length*. Most applications need only the packet headers, not the packet data. This also reduces the amount of data that is copied by BPF to the application. `tcpdump`, for example, defaults this value to 68, which allows room for a 14-byte Ethernet header, a 20-byte IP header, a 20-byte TCP header, and 14 bytes of data. But to print additional information for other protocols (e.g., DNS and NFS) requires the user to increase this value when `tcpdump` is run.
3. BPF buffers the data destined for an application and this buffer is copied to the application only when the buffer is full, or when the *read timeout* expires. This timeout value can be specified by the application. `tcpdump`, for example, sets the timeout to 1000 ms, while the RARP daemon sets it to 0 (since there are few RARP packets, and the RARP server needs to send a response as soon as it receives the request). The purpose of the buffering is to reduce the number of system calls. The same number of packets are still copied between BPF and the application, but each system call has an overhead, and reducing the number of system calls always reduces the overhead. (Figure 3.1 of APUE compares the overhead of the `read` system call, for example, when reading a given file in different chunk sizes varying between 1 byte and 131,072 bytes.)

   Although we show only a single buffer in Figure 26.1, BPF maintains two buffers for each application and fills one while the other is being copied to the application. This is the standard *double buffering* technique.

In Figure 26.1 we show only the BPF reception of packets: packets received by the datalink from below (the network) and packets received by the datalink from above (IP). The application can also write to BPF, causing packets to be sent out the datalink, but most applications only read from BPF. There is no reason to write to BPF to send IP datagrams, because the `IP_HDRINCL` socket option allows us to write any type of IP datagram desired, including the IP header. (We show an example of this in Section 26.6.) The only reason to write to BPF is to send our own network packets that are not IP datagrams. The RARP daemon does this, for example, to send its RARP replies, which are not IP datagrams.

To access BPF one must `open` a BPF device that is not currently open. For example, one would try `/dev/bpf0` and if the error return is `EBUSY`, then try `/dev/bpf1`, and so on. Once the device is opened, about a dozen `ioctl` commands set the characteristics of the device: load the filter, set the read timeout, set the buffer size, attach a

datalink to the BPF device, enable promiscuous mode, and so on. I/O is then performed using `read` and `write`.

26.3 DLPI: Data Link Provider Interface

SVR4 provides datalink access through DLPI, the Data Link Provider Interface. DLPI is a protocol-independent interface designed by AT&T that interfaces to the service provided by the datalink layer [Unix International 1991]. Access to DLPI is by sending and receiving streams messages.

To tap into the datalink layer the application simply `open`s the device (e.g., `le0`) and attaches it using the DLPI `DL_ATTACH_REQ` request. But for efficient operation two additional streams modules are normally pushed onto the stream: `pfmod`, which performs packet filtering within the kernel, and `bufmod`, which buffers the data destined for the application. We show this in Figure 26.2.

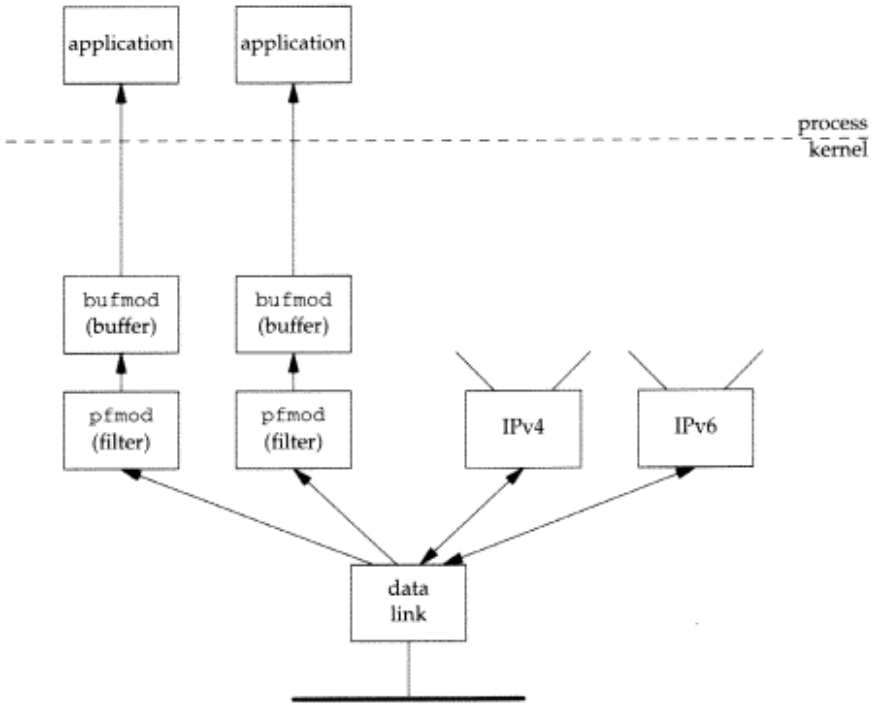


Figure 26.2 Packet capture using DLPI, `pfmod`, and `bufmod`.

Conceptually this is similar to what we described in the previous section for BPF: `pfmod` supports filtering within the kernel using a pseudomachine and `bufmod` reduces the amount of data and the number of system calls by supporting a capture length and a read timeout.

One interesting difference, however, is the type of pseudomachine supported by the BPF and `pfmod` filters. The BPF filter is a directed acyclic control flow graph (CFG)

while pfmod uses a boolean expression tree. The former maps naturally into code for a register machine while the latter maps naturally into code for a stack machine [McCanne and Jacobson 1993]. This paper shows that the CFG implementation used by BPF is normally 3 to 20 times faster than the boolean expression tree, depending on the complexity of the filter.

26.4 Linux: SOCK_PACKET

To receive packets from the datalink layer under Linux we create a socket of type SOCK_PACKET. To do this we must have superuser privileges (similar to creating a raw socket), and the third argument to socket must be a nonzero value specifying the Ethernet frame type. For example, to receive all frames from the datalink, we write

```
fd = socket(AF_INET, SOCK_PACKET, htons(ETH_P_ALL));
```

This would return frames for all protocols that the datalink receives. If we wanted only IPv4 frames, the call would be

```
fd = socket(AF_INET, SOCK_PACKET, htons(ETH_P_IP));
```

Other constants for the final argument are ETH_P_ARP and ETH_P_IPV6, for example.

Specifying a protocol of ETH_P_*xxx* tells the datalink which frame types to pass to the socket for the frames that the datalink receives. If the datalink supports a promiscuous mode (e.g., an Ethernet) then the device must also be put into a promiscuous mode, if desired. This is done by an ioctl of SIOCGIFFLAGS to fetch the flags, setting the IFF_PROMISC flag, and then storing the flags with SIOCSIFFLAGS.

Comparing this Linux feature to BPF and DLPI there are some differences.

1. The Linux feature provides no kernel buffering and no kernel filtering. There is a normal socket receive buffer, but multiple frames cannot be buffered together and passed to the application with a single read. This increases the overhead involved in copying the potentially voluminous amounts of data from the kernel to the application.
2. The Linux feature provides no filtering by device. If ETH_P_IP is specified in the call to socket, then all IPv4 packets from all devices (Ethernets, PPP links, SLIP links, and the loopback device, for example) are passed to the socket. A generic socket address structure is returned by recvfrom and the sa_data member contains the device name (e.g., eth0). The application must then discard data from any device in which it is not interested. The problem again is too much data can be returned to the application, which can be a problem when monitoring a high-speed network.

26.5 libpcap: Packet Capture Library

The packet capture library, libpcap, provides implementation-independent access to the underlying packet capture facility provided by the operating system. Currently it

supports only the reading of packets (although adding a few lines of code to the library lets one write datalink packets too).

Support currently exists for BPF under Berkeley-derived kernels, DLPI under Solaris 2.x, NIT under SunOS 4.1.x, the Linux `SOCK_PACKET` socket, and a few other operating systems. This library is used by `tcpdump`. About 25 functions comprise the library but rather than just describe the functions, we will show the actual use of the common functions in a complete example in the following section. All the library functions begin with the `pcap_` prefix. The `pcap` manual page describes these functions in more detail.

The library is publicly available from `ftp://ftp.ee.lbl.gov/libpcap.tar.Z`.

26.6 Examining the UDP Checksum Field

We now develop an example that sends a UDP datagram containing a DNS query to a name server and reads the reply using the packet capture library. The goal of the example is to determine whether the name server computes a UDP checksum or not. With IPv4 the computation of a UDP checksum is optional. Most current systems enable these checksums by default but unfortunately older systems, notably SunOS 4.1.x, disable these checksums by default. All systems today, and especially a system running a name server, should *always* run with UDP checksums enabled, as corrupted datagrams can corrupt the server's database.

Enabling or disabling UDP checksums is normally done on a systemwide basis, as described in Appendix E of TCPv1.

We build our own UDP datagram (the DNS query) and write it to a raw socket. We could use a normal UDP socket to send the query, but we want to show how to use the `IP_HDRINCL` socket option to build a complete IP datagram. But we can never obtain the UDP checksum when reading from a normal UDP socket and we can never read UDP or TCP packets using a raw socket (Section 25.4). Therefore we must use the packet capture facility to obtain the entire UDP datagram containing the name server's reply. We then examine the UDP checksum field in the UDP header and if it is 0, the server does not have UDP checksums enabled.

Figure 26.3 summarizes the operation of our program. We write our own UDP datagrams to the raw socket and read back the replies using `libpcap`. Notice that UDP also receives the name server reply, and it will respond with an ICMP port unreachable, because it knows nothing about the source port number that our application chooses. The name server will ignore this ICMP error. We also note that it is harder to write a test program of this form that uses TCP, even though we are easily able to write our own TCP segments, because any reply to the TCP segments that we generate will normally cause our TCP to respond with an RST to whomever we sent the segment.

One way around this is to send the TCP segments with a source IP address that belongs to the attached subnet but is not currently assigned to some other node. Add an ARP entry to the

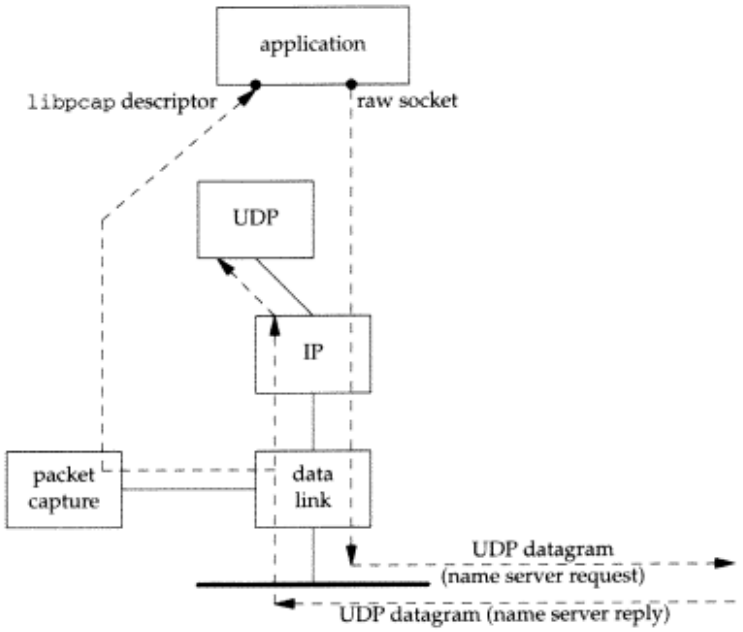


Figure 26.3 Our application to check if a name server has UDP checksums enabled.

sending host for this new IP address so that the sending host will answer ARP requests for this new address. But do not configure the new IP address as an alias. This will cause the IP stack on the sending host to discard packets received for this new IP address, assuming that the sending host is not acting as a router.

Figure 26.4 is a summary of the functions that comprise our program.

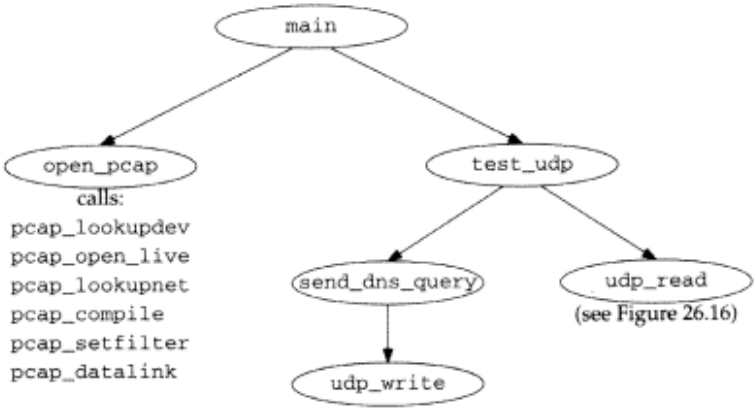


Figure 26.4 Summary of functions for our `udpcksum` program.

Figure 26.5 shows the header `udpcksum.h`, which includes our basic `unp.h` header along with various system headers that are needed to access the structure definitions for the IP and UDP packet headers.

udpcksum/udpcksum.h

```
 1 #include    "unp.h"
 2 #include    <pcap.h>

 3 #include    <netinet/in_systm.h>    /* required for ip.h */
 4 #include    <netinet/in.h>
 5 #include    <netinet/ip.h>
 6 #include    <netinet/ip_var.h>
 7 #include    <netinet/udp.h>
 8 #include    <netinet/udp_var.h>
 9 #include    <net/if.h>
10 #include    <netinet/if_ether.h>

11 #define LOCALPORT   "39123"     /* source port (default) */
12 #define TTL_OUT     64          /* outgoing TTL */

13                     /* declare global variables */
14 extern struct sockaddr *dest, *local;
15 extern socklen_t destlen, locallen;
16 extern int datalink;
17 extern char *device;
18 extern pcap_t *pd;
19 extern int rawfd;
20 extern int snaplen;
21 extern int verbose;
22 extern int zerosum;

23                     /* function prototypes */
24 void    cleanup(int);
25 char   *next_pcap(int *);
26 void    open_pcap(void);
27 void    test_udp(void);
28 void    udp_write(char *, int);
29 struct udpiphdr *udp_read(void);
```

udpcksum/udpcksum.h

Figure 26.5 `udpcksum.h` header.

3-10 Additional Internet headers are required to deal with the IP and UDP header fields.

11-29 We define some global variables and prototypes for our own functions that we show shortly.

The first part of the `main` function is shown in Figure 26.6.

Check number of command-line arguments

20-21 The program requires at least two arguments: the hostname or IP address that is running the DNS server and the service name (`domain`) or port number (53) of the server. We do not show the `usage` function; it just prints a summary of the command format and terminates.

udpcksum/main.c

```
 1 #include    "udpcksum.h"

 2             /* define global variables */
 3 struct sockaddr *dest, *local;
 4 socklen_t destlen, locallen;

 5 int     datalink;               /* from pcap_datalink(), in <net/bpf.h> */
 6 char   *device;                  /* pcap device */
 7 int     fddipad;                /* HACK; for libpcap if FDDI defined */
 8 pcap_t *pd;                     /* packet capture struct pointer */
 9 int     rawfd;                  /* raw socket to write on */
10 int     snaplen = 200;          /* amount of data to capture */
11 int     verbose;
12 int     zerosum;                /* send UDP query with no checksum */

13 static void usage(const char *);

14 int
15 main(int argc, char *argv[])
16 {
17     int     c, on = 1;
18     char   *ptr, localname[1024], *localport;
19     struct addrinfo *aip;

20     if (argc < 2)
21         usage("");

22     /*
23      * Need local IP address for source IP address for UDP datagrams.
24      * Can't specify 0 and let IP choose, as we need to know it for
25      * the pseudo-header to calculate the UDP checksum.
26      * Both localname and localport can be overridden by -l option.
27      */

28     if (gethostname(localname, sizeof(localname)) < 0)
29         err_sys("gethostname error");
30     localport = LOCALPORT;
```

udpcksum/main.c

Figure 26.6 `main` function: definitions.

Obtain local hostname

28–30 Since we will be building our own IP and UDP headers, we must know the source IP address when we write the UDP datagram. We cannot leave it as 0 and let IP choose the address, because the address is part of the UDP pseudoheader (which we describe shortly) that we must use for the UDP checksum computation. Therefore we call `gethostname` to obtain the host's name. We also default the source port to the value in `udpcksum.h`.

The next part of the `main` function, shown in Figure 26.7, processes the command-line arguments.

udpcksum/main.c

```
31      opterr = 0;                 /* don't want getopt() writing to stderr */
32      while ( (c = getopt(argc, argv, "0i:l:v")) != -1) {
33          switch (c) {

34          case '0':
35              zerosum = 1;
36              break;

37          case 'i':
38              device = optarg;    /* pcap device */
39              break;

40          case 'l':               /* local IP address and port#: a.b.c.d.p */
41              if ( (ptr = strrchr(optarg, '.')) == NULL)
42                  usage("invalid -l option");

43              *ptr++ = 0;         /* null replaces final period */
44              localport = ptr;    /* service name or port number */
45              strncpy(localname, optarg, sizeof(localname));
46              break;

47          case 'v':
48              verbose = 1;
49              break;

50          case '?':
51              usage("unrecognized option");
52          }
53      }
```

udpcksum/main.c

Figure 26.7 `main` function: process command-line arguments.

Process command-line options

31-36 We call `getopt` to process the command-line arguments. The `-0` option lets us send our UDP query without a UDP checksum to see if the server handles this differently from a datagram with a checksum.

37-39 The `-i` option lets us specify the interface on which to receive the server's reply. If this is not specified, the packet capture library chooses one, which might not be correct on a multihomed host. This is one way reading from a packet capture device differs from reading from a normal socket: with a socket we can wildcard the local address, allowing us to receive packets arriving on any interface. But with a packet capture device we receive arriving packets on only one interface.

> We note that the Linux `SOCK_PACKET` feature does not limit its datalink capture to a single device. Nevertheless, `libpcap` provides this filtering based on either its default or on our `-i` option.

40-46 The `-l` option lets us specify the source IP address and port number. The port (or a service name) is taken as the string following the final period, and the source IP address is taken as everything before the final period.

The last part of the `main` function is shown in Figure 26.8.

udpcksum/main.c

```
54      if (optind != argc - 2)
55          usage("missing <host> and/or <serv>");

56          /* convert destination name and service */
57      aip = host_serv(argv[optind], argv[optind + 1], AF_INET, SOCK_DGRAM);
58      dest = aip->ai_addr;        /* don't freeaddrinfo() */
59      destlen = aip->ai_addrlen;

60          /* convert local name and service */
61      aip = host_serv(localname, localport, AF_INET, SOCK_DGRAM);
62      local = aip->ai_addr;       /* don't freeaddrinfo() */
63      locallen = aip->ai_addrlen;

64      /*
65       * Need a raw socket to write our own IP datagrams to.
66       * Process must have superuser privileges to create this socket.
67       * Also must set IP_HDRINCL so we can write our own IP headers.
68       */

69      rawfd = Socket(dest->sa_family, SOCK_RAW, 0);

70      Setsockopt(rawfd, IPPROTO_IP, IP_HDRINCL, &on, sizeof(on));

71      open_pcap();                /* open packet capture device */

72      setuid(getuid());           /* don't need superuser privileges any more */

73      Signal(SIGTERM, cleanup);
74      Signal(SIGINT, cleanup);
75      Signal(SIGHUP, cleanup);

76      test_udp();

77      cleanup(0);
78  }
```

udpcksum/main.c

Figure 26.8 `main` function: convert hostnames and service names, create socket.

Process destination name and port, then local name and port

54–63 We verify that exactly two command-line arguments remain: the destination hostname and service name. We call `host_serv` to convert these into a socket address structure, the pointer to which we save in `dest`. We then do the same conversion of the local hostname and port, saving the pointer to the socket address structure in `local`.

Create raw socket and open packet capture device

64–71 We create a raw socket and enable the `IP_HDRINCL` socket option. This option lets us write complete IP datagrams, including the IP header. The function `open_pcap` opens the packet capture device, and we show this function next.

Change permissions and establish signal handlers

72–75 We need superuser privileges to create a raw socket. We normally need superuser privileges to open the packet capture device, but this depends on the implementation. For example, with BPF the administrator can set the permissions of the `/dev/bpf` devices to whatever is desired for that system. We now give up these additional

permissions, assuming the program file is set-user-ID. If the process has superuser privileges, calling `setuid` sets our real user ID, effective user ID, and saved set-user-ID to our real user ID (`getuid`). We establish signal handlers in case the user terminates the program before it is done.

Perform test and cleanup

76-77 The function `test_udp` (Figure 26.10) performs the test and then returns. `cleanup` (Figure 26.18) prints summary statistics from the packet capture library and terminates the process.

Figure 26.9 shows the `open_pcap` function, which we called from the `main` function to open the packet capture device.

Choose packet capture device

10-14 If the packet capture device was not specified (the `-i` command-line option), then `pcap_lookupdev` chooses a device. It issues the `SIOCGIFCONF ioctl` and chooses the lowest numbered device that is up, but not the loopback. Many of the `pcap` library functions fill in an error string if an error occurs. The sole argument to this function is an array that is filled in with an error string.

Open device

15-17 `pcap_open_live` opens the device. The term "live" refers to an actual device being opened, instead of a save file containing previously saved packets. The first argument is the device name, the second is the number of bytes to save per packet (`snaplen`, which we initialized to 200 in Figure 26.6), the third is a promiscuous flag, the fourth is a timeout value in milliseconds, and the fifth is a pointer to an error message array.

If the promiscuous flag is set, the interface is placed into promiscuous mode, causing it to receive all packets passing by on the wire. This is the normal mode for `tcpdump`. For our example, however, the DNS server replies will be sent to our host.

The timeout argument is a read timeout. Instead of having the device return a packet to the process every time a packet is received (which could be inefficient, invoking lots of copies of individual packets from the kernel to the process), a packet is returned only when either the device's read buffer is full, or when the read timeout expires. If the read timeout is set to 0, every packet is returned as soon as it is received.

Obtain network address and subnet mask

18-23 `pcap_lookupnet` returns the network address and subnet mask for the packet capture device. We must specify the subnet mask in the call to `pcap_compile` that follows, because the packet filter needs this to determine if an IP address is a subnet-directed broadcast address.

Compile packet filter

24-30 `pcap_compile` takes a filter string (which we build in the `cmd` array) and compiles it into a filter program (stored in `fcode`). This will select the packets that we wish to receive.

udpcksum/pcap.c

```
 1 #include    "udpcksum.h"

 2 #define CMD     "udp and src host %s and src port %d"

 3 void
 4 open_pcap(void)
 5 {
 6     uint32_t localnet, netmask;
 7     char    cmd[MAXLINE], errbuf[PCAP_ERRBUF_SIZE], str1[INET_ADDRSTRLEN],
 8             str2[INET_ADDRSTRLEN];
 9     struct bpf_program fcode;

10     if (device == NULL) {
11         if ( (device = pcap_lookupdev(errbuf)) == NULL)
12             err_quit("pcap_lookup: %s", errbuf);
13     }
14     printf("device = %s\n", device);

15         /* hardcode: promisc=0, to_ms=500 */
16     if ( (pd = pcap_open_live(device, snaplen, 0, 500, errbuf)) == NULL)
17         err_quit("pcap_open_live: %s", errbuf);

18     if (pcap_lookupnet(device, &localnet, &netmask, errbuf) < 0)
19         err_quit("pcap_lookupnet: %s", errbuf);
20     if (verbose)
21         printf("localnet = %s, netmask = %s\n",
22                Inet_ntop(AF_INET, &localnet, str1, sizeof(str1)),
23                Inet_ntop(AF_INET, &netmask, str2, sizeof(str2)));

24     snprintf(cmd, sizeof(cmd), CMD,
25              Sock_ntop_host(dest, destlen),
26              ntohs(sock_get_port(dest, destlen)));
27     if (verbose)
28         printf("cmd = %s\n", cmd);
29     if (pcap_compile(pd, &fcode, cmd, 0, netmask) < 0)
30         err_quit("pcap_compile: %s", pcap_geterr(pd));

31     if (pcap_setfilter(pd, &fcode) < 0)
32         err_quit("pcap_setfilter: %s", pcap_geterr(pd));

33     if ( (datalink = pcap_datalink(pd)) < 0)
34         err_quit("pcap_datalink: %s", pcap_geterr(pd));
35     if (verbose)
36         printf("datalink = %d\n", datalink);
37 }
```

udpcksum/pcap.c

Figure 26.9 open_pcap function: open and initialize packet capture device.

Load filter program

31-32 `pcap_setfilter` takes the filter program that we just compiled and loads it into the packet capture device. This initiates the capturing of the packets that we selected with the filter.

Determine datalink type

33-36 `pcap_datalink` returns the type of datalink for the packet capture device. We need this when receiving packets to determine the size of the datalink header that will be at the beginning of each packet that we read (Figure 26.14).

After calling `open_pcap`, the `main` function calls `test_udp`, which we show in Figure 26.10. This function sends a DNS query and reads the server's reply.

udpcksum/udpcksum.c

```
47 void
48 test_udp(void)
49 {
50     volatile int nsent = 0, timeout = 3;
51     struct udpiphdr *ui;

52     Signal(SIGALRM, sig_alrm);

53     if (sigsetjmp(jmpbuf, 1)) {
54         if (nsent >= 3)
55             err_quit("no response");
56         printf("timeout\n");
57         timeout *= 2;            /* exponential backoff: 3, 6, 12 */
58     }
59     canjump = 1;                 /* siglongjmp is now OK */

60     send_dns_query();
61     nsent++;

62     alarm(timeout);
63     ui = udp_read();
64     canjump = 0;
65     alarm(0);

66     if (ui->ui_sum == 0)
67         printf("UDP checksums off\n");
68     else
69         printf("UDP checksums on\n");
70     if (verbose)
71         printf("recevied UDP checksum = %x\n", ntohs(ui->ui_sum));
72 }
```

udpcksum/udpcksum.c

Figure 26.1 `test_udp` function: send queries and read responses.

`volatile` variables

50 We want the two automatic variables `nsent` and `timeout` to retain their value after a `siglongjmp` from the signal handler back to this function. An implementation is allowed to restore automatic variables back to their value when `sigsetjmp` was called (p. 178 of APUE), but adding the `volatile` qualifier prevents this from happening.

Establish signal handler and jump buffer

52-53 A signal handler is established for `SIGALRM` and `sigsetjmp` establishes a jump buffer for `siglongjmp`. (These two functions are described in detail in Section 10.15 of

APUE.) The second argument of 1 to `sigsetjmp` tells it to save the current signal mask, since we will call `siglongjmp` from our signal handler.

Handle `siglongjmp`

54–58 This code is executed only when `siglongjmp` is called from our signal handler. This indicates that a timeout occurred: we sent a request and never received a reply. If we have sent three requests, we terminate. Otherwise we print a message and multiply the timeout value by 2. This is an *exponential backoff*, which we also described in Section 20.5. The first timeout will be for 3 seconds, then 6, and then 12.

The reason we use `sigsetjmp` and `siglongjmp` in this example, rather than just catching `EINTR` (as in Figure 13.1), is because the packet capture library reading functions (which are called by our `udp_read` function) restart a read operation when `EINTR` is returned. Since we do not want to modify the library functions to return this error, our only solution is to catch the `SIGALRM` signal and perform a nonlocal goto, returning control to our code instead of the library code.

Send DNS query and read reply

60–65 `send_dns_query` (Figure 26.12) sends a DNS query to a name server. `udp_read` (Figure 26.14) reads the reply. We call `alarm` to prevent the read from blocking forever. If the specified timeout period (in seconds) expires, `SIGALRM` is generated and our signal handler calls `siglongjmp`.

Examine received UDP checksum

66–71 If the received UDP checksum is 0, the server did not calculate and send a checksum.

Figure 26.11 shows our signal handler, `sig_alrm`, which handles the `SIGALRM` signal.

udpcksum/udpcksum.c

```
 1 #include    "udpcksum.h"
 2 #include    <setjmp.h>

 3 static sigjmp_buf jmpbuf;
 4 static int canjump;

 5 void
 6 sig_alrm(int signo)
 7 {
 8     if (canjump == 0)
 9         return;
10     siglongjmp(jmpbuf, 1);
11 }
```

udpcksum/udpcksum.c

Figure 26.11 `sig_alrm` function: handle `SIGALRM` signal.

8–10 The flag `canjump` was set in Figure 26.10 after the jump buffer was initialized by `sigsetjmp`. If the flag has been set, `siglongjmp` causes the flow of control to act as if the `sigsetjmp` in Figure 26.10 had returned with the return value of 1.

Figure 26.12 shows the `send_dns_query` function that sends a UDP query to a DNS server. This function builds the application data, a DNS query.

udpcksum/udpcksum.c

```
16 void
17 send_dns_query(void)
18 {
19     size_t  nbytes;
20     char    buf[sizeof(struct udpiphdr) + 100], *ptr;
21     short   one;

22     ptr = buf + sizeof(struct udpiphdr);    /* leave room for IP/UDP headers */

23     *((u_short *) ptr) = htons(1234);   /* identification */
24     ptr += 2;
25     *((u_short *) ptr) = htons(0x0);    /* flags */
26     ptr += 2;
27     *((u_short *) ptr) = htons(1);  /* #questions */
28     ptr += 2;
29     *((u_short *) ptr) = 0;     /* #answer RRs */
30     ptr += 2;
31     *((u_short *) ptr) = 0;     /* #authority RRs */
32     ptr += 2;
33     *((u_short *) ptr) = 0;     /* #additional RRs */
34     ptr += 2;

35     memcpy(ptr, "\001a\014root-servers\003net\000", 20);
36     ptr += 20;
37     one = htons(1);
38     memcpy(ptr, &one, 2);       /* query type = A */
39     ptr += 2;
40     memcpy(ptr, &one, 2);       /* query class = 1 (IP addr) */
41     ptr += 2;

42     nbytes = 36;
43     udp_write(buf, nbytes);
44     if (verbose)
45         printf("sent: %d bytes of data\n", nbytes);
46 }
```

udpcksum/udpcksum.c

Figure 26.12 `send_dns_query` function: send a query to a DNS server.

Initialize buffer pointer

20–22 `buf` has room for a 20-byte IP header, an 8-byte UDP header, and 100 bytes of user data. `ptr` is set to point to the first byte of user data.

Build DNS query

23–34 To understand the details of the UDP datagram built by this function requires an understanding of the DNS message format. This is found in Section 14.3 of TCPv1. We set the identification field to 1234, the flags to 0, the number of questions to 1, and then the number of answer resource records (RRs), the number of authority RRs, and the number of additional RRs to 0.

The <arpa/nameser.h> header defines a HEADER datatype for the first 12 bytes of a query header. We find it just as easy to store these 12 bytes ourselves for the simple query we are building.

35-41 We then form the single question that follows in the message: an A query for the IP addresses of the host a.root-servers.net. This domain name is stored in 20 bytes and consists of 4 labels: the 1-byte label a, the 12-byte label root-servers (remember that \014 is an octal character constant), the 3-byte label net, and the root label whose length is 0. The query type is 1 (called an A query) and the query class is also 1.

Write UDP datagram

42-45 This message consists of 36 bytes of user data (eight 2-byte fields and the 20-byte domain name). We call our function udp_write to build the UDP and IP headers, and then to write the IP datagram to our raw socket.

Figure 26.13 shows our function udp_write, which builds the IP and UDP headers and then writes the datagram to the raw socket.

Initialize packet header pointers

11-13 ip points to the beginning of the IP header (an ip structure) and ui points to the same location, but the structure udpiphdr is the combined IP and UDP headers.

Update lengths

14-17 ui_len is the UDP length: the number of bytes of user data plus the size of the UDP header (8 bytes). userlen (the number of bytes of user data that follows the UDP header) is incremented by 28 (20 bytes for the IP header and 8 bytes for the UDP header) to reflect the total size of the IP datagram.

Fill in UDP header and calculate UDP checksum

18-35 When the UDP checksum is calculated, it includes not only the UDP header and the UDP data, but also fields from the IP header. These additional fields from the IP header form what is called the *pseudoheader*. The inclusion of the pseudoheader provides additional verification that if the checksum is correct then the datagram was delivered to the correct host and to the correct protocol code. These statements initialize the fields in the IP header that form the pseudoheader. The code is somewhat obtuse but is explained in Section 23.6 of TCPv2. The end result is storing the UDP checksum in the ui_sum member if the zerosum flag (the -0 command-line argument) is not set.

If the calculated checksum is 0, the value 0xffff is stored instead. In ones-complement arithmetic the two values are the same, but UDP sets the checksum to 0 to indicate that the sender did not store a UDP checksum. Notice that we did not check for a calculated checksum of 0 in Figure 25.13 because the ICMPv4 checksum is required: the value of 0 does not indicate the absence of a checksum.

We note that Solaris 2.x, for x < 6, has a bug with regard to checksums for TCP segments or UDP datagrams sent on a raw socket with the IP_HDRINCL socket option set. The kernel calculates the checksum and we must set the ui_sum field to the UDP length.

Fill in IP header

36-49 Since we have set the IP_HDRINCL socket option, we must fill in most fields in the

IP header. (Section 25.3 discusses these writes to a raw socket when this socket option is set.) We set the identification field to 0 (`ip_id`), which tells IP to set this field. IP also calculates the IP header checksum. `sendto` writes the IP datagram.

udpcksum/udpwrite.c

```
 6 void
 7 udp_write(char *buf, int userlen)
 8 {
 9     struct udpiphdr *ui;
10     struct ip *ip;

11         /* Fill in and checksum UDP header */
12     ip = (struct ip *) buf;
13     ui = (struct udpiphdr *) buf;
14             /* add 8 to userlen for pseudo-header length */
15     ui->ui_len = htons((u_short) (sizeof(struct udphdr) + userlen));
16             /* then add 28 for IP datagram length */
17     userlen += sizeof(struct udpiphdr);

18     ui->ui_next = 0;
19     ui->ui_prev = 0;
20     ui->ui_x1 = 0;
21     ui->ui_pr = IPPROTO_UDP;
22     ui->ui_src.s_addr = ((struct sockaddr_in *) local)->sin_addr.s_addr;
23     ui->ui_dst.s_addr = ((struct sockaddr_in *) dest)->sin_addr.s_addr;
24     ui->ui_sport = ((struct sockaddr_in *) local)->sin_port;
25     ui->ui_dport = ((struct sockaddr_in *) dest)->sin_port;
26     ui->ui_ulen = ui->ui_len;
27     ui->ui_sum = 0;
28     if (zerosum == 0) {
29 #ifdef notdef                    /* change to ifndef for Solaris 2.x, x < 6 */
30         if ( (ui->ui_sum = in_cksum((u_short *) ui, userlen)) == 0)
31             ui->ui_sum = 0xffff;
32 #else
33         ui->ui_sum = ui->ui_len;
34 #endif
35     }
36         /* Fill in rest of IP header; */
37         /* ip_output() calcuates & stores IP header checksum */
38     ip->ip_v = IPVERSION;
39     ip->ip_hl = sizeof(struct ip) >> 2;
40     ip->ip_tos = 0;
41 #ifdef  linux
42     ip->ip_len = htons(userlen);     /* network byte order */
43 #else
44     ip->ip_len = userlen;        /* host byte order */
45 #endif
46     ip->ip_id = 0;               /* let IP set this */
47     ip->ip_off = 0;              /* frag offset, MF and DF flags */
48     ip->ip_ttl = TTL_OUT;

49     Sendto(rawfd, buf, userlen, 0, dest, destlen);
50 }
```

udpcksum/udpwrite.c

Figure 26.13 `udp_write` function: build UDP and IP headers and write IP datagram to raw socket.

The next function is udp_read, shown in Figure 26.14, which was called from Figure 26.10.

udpcksum/udpread.c

```
 7 struct udpiphdr *
 8 udp_read(void)
 9 {
10     int     len;
11     char   *ptr;
12     struct ether_header *eptr;

13     for ( ; ; ) {
14         ptr = next_pcap(&len);

15         switch (datalink) {
16         case DLT_NULL:          /* loopback header = 4 bytes */
17             return (udp_check(ptr + 4, len - 4));

18         case DLT_EN10MB:
19             eptr = (struct ether_header *) ptr;
20             if (ntohs(eptr->ether_type) != ETHERTYPE_IP)
21                 err_quit("Ethernet type %x not IP", ntohs(eptr->ether_type));
22             return (udp_check(ptr + 14, len - 14));

23         case DLT_SLIP:          /* SLIP header = 24 bytes */
24             return (udp_check(ptr + 24, len - 24));

25         case DLT_PPP:           /* PPP header = 24 bytes */
26             return (udp_check(ptr + 24, len - 24));

27         default:
28             err_quit("unsupported datalink (%d)", datalink);
29         }
30     }
31 }
```

udpcksum/udpread.c

Figure 26.14 udp_read function: read next packet from packet capture device.

14–29 Our function next_pcap (Figure 26.15) returns the next packet from the packet capture device. Since the datalink headers differ depending on the actual device type, we branch based on the value returned by the pcap_datalink function.

These magic offsets of 4, 14, and 24 are shown in Figure 31.9 of TCPv2. The 24-byte offsets shown for SLIP and PPP are for BSD/OS 2.1.

Despite having the qualifier "10MB" in the name DLT_EN10MB, this datalink type is also used for 100 Mbit/sec Ethernet.

Our function udp_check (Figure 26.17) examines the packet and verifies fields in the IP and UDP headers.

Figure 26.15 shows the next_pcap function, which returns the next packet from the packet capture device.

udpcksum/pcap.c

```
38 char *
39 next_pcap(int *len)
40 {
41     char   *ptr;
42     struct pcap_pkthdr hdr;

43         /* keep looping until packet ready */
44     while ( (ptr = (char *) pcap_next(pd, &hdr)) == NULL) ;

45     *len = hdr.caplen;          /* captured length */
46     return (ptr);
47 }
```

udpcksum/pcap.c

Figure 26.15 next_pcap function: return next packet.

43-44 We call the library function pcap_next, which returns the next packet. A pointer to the packet is returned as the return value of the function and the second argument points to a pcap_pkthdr structure, which is also filled in on return:

```
struct pcap_pkthdr {
  struct timeval  ts;     /* timestamp */
  bpf_u_int32     caplen; /* length of portion captured */
  bpf_u_int32     len;    /* length this packet (off wire) */
};
```

The timestamp is when the packet capture device read the packet, as opposed to the actual delivery of the packet to the process, which could be some time later. caplen is the amount of data that was captured (recall that we set our variable snaplen to 200 in Figure 26.6, and then this was the second argument to pcap_open_live in Figure 26.9). The purpose of the packet capture facility is to capture the packet headers, and not all the data in each packet. len is the full length of the packet on the wire. caplen will always be less than or equal to len.

45-46 The captured length is returned through the pointer argument and the return value of the function is the pointer to the packet. Keep in mind that the "pointer to the packet" points to the datalink header, which is the 14-byte Ethernet header in the case of an Ethernet frame, or a 4-byte pseudo-link header in the case of the loopback interface.

If we look at the implementation of pcap_next in the library, it shows the division of labor between the different functions. We show this in Figure 26.16. Our application calls the pcap_ functions and some of these functions are device independent, while others are dependent on the type of packet capture device. For example, we show that the BPF implementation calls read, while the DLPI implementation calls getmsg and the Linux implementation calls recvfrom.

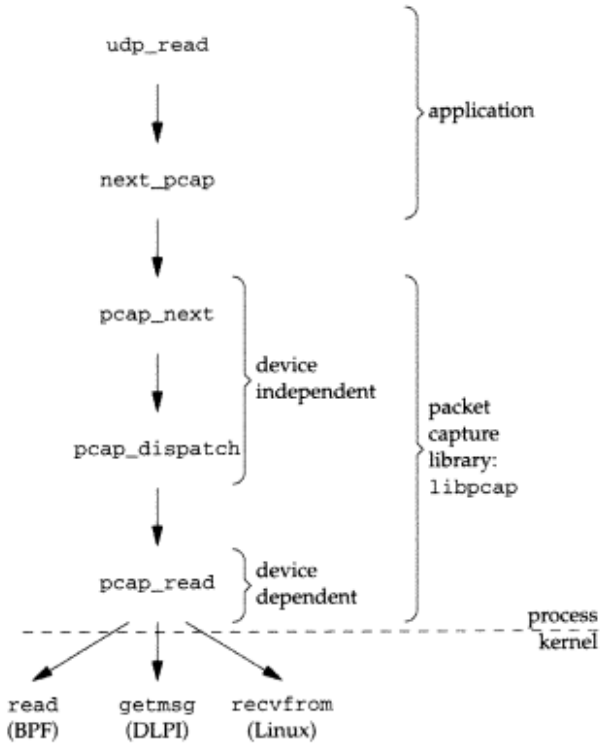


Figure 26.16 Arrangement of function calls to read from packet capture library.

Our function `udp_check` verifies numerous fields in the IP and UDP headers. It is shown in Figure 26.17. We must do these verifications because when the packet is passed to us by the packet capture device, the IP layer has not yet seen the packet. This differs from a raw socket.

44-61 The packet length must include at least the IP and UDP headers. The IP version is verified along with the IP header length and the IP header checksum. If the protocol field indicates a UDP datagram, the function returns the pointer to the combined IP/UDP header. Otherwise the program terminates, since the packet capture filter that we specified in our call to `pcap_setfilter` in Figure 26.9 should not return any other type of packet.

udpcksum/udpread.c

```
38 struct udpiphdr *
39 udp_check(char *ptr, int len)
40 {
41     int     hlen;
42     struct ip *ip;
43     struct udpiphdr *ui;

44     if (len < sizeof(struct ip) + sizeof(struct udphdr))
45         err_quit("len = %d", len);

46         /* minimal verification of IP header */
47     ip = (struct ip *) ptr;
48     if (ip->ip_v != IPVERSION)
49         err_quit("ip_v = %d", ip->ip_v);
50     hlen = ip->ip_hl << 2;
51     if (hlen < sizeof(struct ip))
52         err_quit("ip_hl = %d", ip->ip_hl);
53     if (len < hlen + sizeof(struct udphdr))
54         err_quit("len = %d, hlen = %d", len, hlen);

55     if ( (ip->ip_sum = in_cksum((u_short *) ip, hlen)) != 0)
56         err_quit("ip checksum error");

57     if (ip->ip_p == IPPROTO_UDP) {
58         ui = (struct udpiphdr *) ip;
59         return (ui);
60     } else
61         err_quit("not a UDP packet");
62 }
```

udpcksum/udpread.c

Figure 26.17 udp_check function: verify fields in IP and UDP headers.

udpcksum/cleanup.c

```
 2 void
 3 cleanup(int signo)
 4 {
 5     struct pcap_stat stat;

 6     fflush(stdout);
 7     putc('\n', stdout);

 8     if (verbose) {
 9         if (pcap_stats(pd, &stat) < 0)
10             err_quit("pcap_stats: %s\n", pcap_geterr(pd));
11         printf("%d packets received by filter\n", stat.ps_recv);
12         printf("%d packets dropped by kernel\n", stat.ps_drop);
13     }
14     exit(0);
15 }
```

udpcksum/cleanup.c

Figure 26.18 cleanup function.

The `cleanup` function shown in Figure 26.18 is called by the `main` function immediately before the program terminates, and also as the signal handler if the user aborts the program (Figure 26.8).

Fetch and print packet capture statistics

8-13 `pcap_stats` fetches the packet capture statistics: the total number of packets received by the filter and the number of packets dropped by the kernel.

Example

We first run our program with the `-0` command-line option to verify that the name server responds to datagrams that arrive with no checksum. We also specify the `-v` flag.

```
solaris # udpcksum -0 -v connix.com domain
device = le0
localnet = 206.62.226.32, netmask = 255.255.255.224
cmd = udp and src host 198.69.10.4 and src port 53
datalink = 1
sent: 36 bytes of data
UDP checksums on
recevied UDP checksum = ad39

2 packets received by filter
0 packets dropped by kernel
```

Next we run our program to a name server that does not have UDP checksums enabled.

```
solaris # udpcksum -v gw.pacbell.com domain
device = le0
localnet = 206.62.226.32, netmask = 255.255.255.224
cmd = udp and src host 192.150.170.2 and src port 53
datalink = 1
sent: 36 bytes of data
UDP checksums off
recevied UDP checksum = 0

1 packets received by filter
0 packets dropped by kernel
```

26.7 Summary

With raw sockets we have the capability to read and write IP datagrams that the kernel does not understand, and with access to the datalink layer we can extend that capability to read and write *any* type of datalink frame, not just IP datagrams. `tcpdump` is probably the most commonly used program that accesses the datalink layer directly.

Different operating systems have different ways of accessing the datalink layer. We looked at the Berkeley-derived BPF, SVR4's DLPI, and the Linux `SOCK_PACKET`. But we can ignore all their differences and still write portable code using the freely available packet capture library, `libpcap`.

Exercises

26.1 What is the purpose of the `canjump` flag in Figure 26.11?

26.2 In our `udpcksum` program, common error replies are an ICMP port unreachable (the destination is not running a name server) or an ICMP host unreachable. In either case we need not wait for a timeout of our `udp_read` in Figure 26.10 because the ICMP error is essentially a reply to our DNS query. Modify the program to catch these ICMP errors.

27

Client–Server Design Alternatives

27.1 Introduction

We have several choices for the type of process control to use when writing a Unix server.

- Our first server, Figure 1.9, was an *iterative server*, but there are a limited number of scenarios where this is recommended, because the server cannot process a pending client until it has completely serviced the current client.
- Figure 5.2 was our first *concurrent server* and it called `fork` to spawn a child process for every client. Traditionally most Unix servers fall into this category.
- In Section 6.8 we developed a different version of our TCP server consisting of a single process using `select` to handle any number of clients.
- In Figure 23.3 we modified our concurrent server to create one thread per client, instead of one process per client.

There are two other modifications to the concurrent server design that we look at in this chapter.

- *Preforking* has the server call `fork` when it starts, creating a pool of child processes. One process from the currently available pool handles each client request.
- *Prethreading* has the server create a pool of available threads when it starts, and one thread from the pool handles each client.

There are numerous details with preforking and prethreading that we examine in this chapter: what if there are not enough processes or threads in the pool? what if there are too many processes or threads in the pool? how can the parent and its children or threads synchronize with each other?

Clients are typically easier to write than servers because there is less process control in a client. Nevertheless, we have already examined various ways to write our simple echo client, and we summarize these in Section 27.2.

In this chapter we look at nine different server designs and we run each server against the same client. Our client–server scenario is typical of the Web: the client sends a small request to the server and the server responds with data back to the client. Some of the servers we have already discussed in detail (e.g., the concurrent server with one `fork` per client) while the preforked and prethreaded servers are new and therefore discussed in detail in this chapter.

We run multiple instances of a client against each server, measuring the CPU time required to service a fixed number of client requests. Instead of scattering all our CPU timings throughout the chapter, we summarize them in Figure 27.1 and refer to this figure throughout the chapter. We note that the times in this figure measure the CPU time required *only for process control* and the iterative server is our baseline that we subtract from the actual CPU time, because an iterative server has no process control overhead. We include the baseline times of 0.0 in this figure to reiterate this point. We use the term *process control CPU time* in this chapter to denote this difference from the baseline for a given system.

Row	Server description	Process control CPU time, seconds (difference from baseline)		
		Solaris	DUnix	BSD/OS
0	Iterative server (baseline measurement; no process control)	0.0	0.0	0.0
1	Concurrent server, one `fork` per client request	504.2	168.9	29.6
2	Prefork with each child calling `accept`		6.2	1.8
3	Prefork with file locking to protect `accept`	25.2	10.0	2.7
4	Prefork with thread mutex locking to protect `accept`	21.5		
5	Prefork with parent passing socket descriptor to child	36.7	10.9	6.1
6	Concurrent server, create one thread per client request	18.7	4.7	
7	Prethreaded with mutex locking to protect `accept`	8.6	3.5	
8	Prethreaded with main thread calling `accept`	14.5	5.0	

Figure 27.1 Timing comparisons of the various servers discussed in this chapter.

We ran the various servers on three hosts: `sunos5` (Solaris 2.5.1), `alpha` (Digital Unix 4.0b), and `bsdi` (BSD/OS 3.0). Notice that not all of the servers can be run on all three hosts. For example, row 2 cannot be run on most SVR4 hosts (as we discuss in Section 27.7) and none of the threaded servers can be run under BSD/OS (since the kernel does not support threads). The three server hosts are of different architectures, so we are not able to compare the timing between the three server hosts. The intent of these timings is to see how the different server designs compare on a given host, and not to compare different hardware architectures and operating systems. For example,

row 7, the prethreaded server with mutex locking to protect the `accept` is the fastest server under both Solaris and Digital Unix, while row 2 is the fastest under BSD/OS.

All of these server timings were obtained by running the client shown in Figure 27.4 on two different hosts on the same subnet as the server. For all tests both clients spawned five children to create five simultaneous connections to the server, for a maximum of 10 simultaneous connections at the server at any time. Each client requested 4000 bytes from the server across the connection. For those tests involving a preforked or a prethreaded server, the server created 15 children or 15 threads when it started.

# children or # threads	Process control CPU time, seconds (difference from baseline)					
	prefork, no locking around `accept` (row 2)		prefork, file locking around `accept` (row 3)			prethreaded, mutex locking around `accept` (row 7)
	DUnix	BSD/OS	Solaris	DUnix	BSD/OS	Solaris
15	6.2	1.8	25.2	10.0	2.7	8.6
30	7.8	3.5	27.3	11.2	5.6	10.0
45	8.9	5.5	29.7	13.1	8.7	19.6
60	10.1	6.9	34.2	14.3	11.2	28.6
75	11.4	8.7	39.8	16.0	13.7	29.3
90	12.6	10.9	130.1	17.6	15.5	28.6
105	13.2	12.0		19.7	17.6	30.4
120	15.7	13.5		22.0	19.2	29.4

Figure 27.2 Effect of extraneous children or threads on server CPU time.

child # or thread #	#clients serviced									
	prefork, no locking around `accept` (row 2)		prefork, file locking around `accept` (row 3)			prefork, descriptor passing (row 5)			prethreaded, thread locking around `accept` (row 7)	
	DUnix	BSD/OS	Solaris	DUnix	BSD/OS	Solaris	DUnix	BSD/OS	Solaris	DUnix
0	318	333	347	335	335	1006	718	530	333	335
1	343	340	328	334	335	950	647	529	323	337
2	326	335	332	334	332	720	589	509	333	338
3	317	335	335	333	333	582	554	502	328	311
4	309	332	338	333	331	485	526	501	329	345
5	344	331	340	335	335	457	501	495	322	332
6	340	333	335	330	332	385	447	488	324	355
7	337	333	343	334	333	250	389	484	360	322
8	340	332	324	333	334	105	314	460	341	336
9	309	331	315	333	336	32	208	443	348	337
10	356	334	326	333	331	14	62	59	358	334
11	354	333	340	334	338	9	18	0	331	340
12	356	334	330	333	333	4	14	0	321	317
13	302	332	331	333	331	1	12	0	329	326
14	349	332	336	333	331	0	1	0	320	335
	5000	5000	5000	5000	5000	5000	5000	5000	5000	5000

Figure 27.3 Number of clients or threads serviced by each of the 15 children or threads.

Some server designs involve creating a pool of child processes or a pool of threads. Another item that we look at is the effect of too many children or too many threads. Figure 27.2 summarizes these results and we discuss these numbers in the appropriate sections.

Another topic that we consider when we have a collection of child processes or threads to service the clients is the distribution of the client requests to the available pool. Figure 27.3 summarizes these distributions and we discuss each column in the appropriate section.

27.2 TCP Client Alternatives

We have already examined various client designs, but it is worth summarizing their strengths and weaknesses.

1. Figure 5.5 was the basic TCP client. There were two problems with this program. First, while it is blocked awaiting user input, it does not see network events, such as the peer closing the connection. Additionally it operates in a stop-and-wait mode, making it inefficient for batch processing.
2. Figure 6.9 was the next iteration and by using `select` the client was notified of network events while waiting for user input. The problem, however, is that this program did not handle batch mode correctly. Figure 6.13 corrected this problem by using the `shutdown` function.
3. Figure 15.3 began the presentation of our client using nonblocking I/O.
4. The first of our clients that went beyond the single-process, single-thread design was Figure 15.9, which used `fork` with one process handling the client-to-server data, and the other process handling the server-to-client data.
5. Figure 23.2 used two threads, instead of two processes.

At the end of Section 15.2 we summarized the timing differences between these various versions. As we noted there, although the nonblocking I/O version was the fastest, the code was more complex, and using either two processes or two threads simplifies the code.

27.3 TCP Test Client

Figure 27.4 shows the client that we will use to test all the variations of our server.

10-12 Each time we run the client we specify the hostname or IP address of the server, the server's port, the number of children for the client to `fork` (allowing us to initiate multiple connections to the same server concurrently), the number of requests each child should send to the server, and the number of bytes to request the server to return each time.

server/client.c

```
 1 #include    "unp.h"
 2 #define MAXN    16384           /* max #bytes to request from server */
 3 int
 4 main(int argc, char **argv)
 5 {
 6     int     i, j, fd, nchildren, nloops, nbytes;
 7     pid_t   pid;
 8     ssize_t n;
 9     char    request[MAXLINE], reply[MAXN];
10     if (argc != 6)
11         err_quit("usage: client <hostname or IPaddr> <port> <#children> "
12                  "<#loops/child> <#bytes/request>");
13     nchildren = atoi(argv[3]);
14     nloops = atoi(argv[4]);
15     nbytes = atoi(argv[5]);
16     snprintf(request, sizeof(request), "%d\n", nbytes);     /* newline at end */
17     for (i = 0; i < nchildren; i++) {
18         if ( (pid = Fork()) == 0) {  /* child */
19             for (j = 0; j < nloops; j++) {
20                 fd = Tcp_connect(argv[1], argv[2]);
21                 Write(fd, request, strlen(request));
22                 if ( (n = Readn(fd, reply, nbytes)) != nbytes)
23                     err_quit("server returned %d bytes", n);
24                 Close(fd);      /* TIME_WAIT on client, not server */
25             }
26             printf("child %d done\n", i);
27             exit(0);
28         }
29         /* parent loops around to fork() again */
30     }
31     while (wait(NULL) > 0)      /* now parent waits for all children */
32         ;
33     if (errno != ECHILD)
34         err_sys("wait error");
35     exit(0);
36 }
```

server/client.c

Figure 27.4 TCP client program for testing our various servers.

17-30 The parent calls `fork` for each child, and each child establishes the specified number of connections with the server. On each connection the child sends a line specifying the number of bytes for the server to return, and then the child reads that amount of data from the server. The parent just `wait`s for all the children to terminate. Notice that the client closes each TCP connection, so TCP's TIME_WAIT state occurs on the

client, not on the server. This is a difference between our client–server and normal HTTP connections.

When we measure the various servers in this chapter we execute the client as

```
% client 206.62.226.36 8888 5 500 4000
```

This creates 2500 TCP connections to the server: 500 connections from each of five children. On each connection 5 bytes are sent from the client to the server ("4000\n") and 4000 bytes are transferred from the server back to the client. We run the client from two different hosts to the same server, providing a total of 5000 TCP connections, with a maximum of 10 simultaneous connections at the server at any given time.

> Sophisticated benchmarks exist for testing various Web servers. One is called WebStone and is available from `http://www.mindcraft.com/webstone`. We do not need anything this sophisticated to make some general comparisons of the various server design alternatives that we examine in this chapter.

We now present the nine different server designs.

27.4 TCP Iterative Server

An iterative TCP server processes each client's request completely before moving on to the next client. Iterative TCP servers are rare but we showed one in Figure 1.9: a simple daytime server.

We do, however, have a use for an iterative server in comparing the various servers in this chapter. If we run the client as

```
% client 206.62.226.36 8888 1 5000 4000
```

to an iterative server, we get the same number of TCP connections (5000) and the same amount of data transferred across each connection. But since the server is iterative, there is *no process control whatsoever* performed by the server. This gives us a baseline measurement of the CPU time required to handle this number of clients that we can then subtract from all the other server measurements. From a process control perspective the iterative server is the fastest possible, because it performs no process control. We then compare the *differences* from this baseline in Figure 27.1.

We do not show our iterative server, as it is a trivial modification to the concurrent server that we present in the next section.

27.5 TCP Concurrent Server, One Child per Client

Traditionally a concurrent TCP server calls `fork` to spawn a child to handle each client. This allows the server to handle numerous clients at the same time, one client per process. The only limit on the number of clients is the operating system limit on the number of child processes for the user ID under which the server is running. Figure 5.12 is an example of a concurrent server and most TCP servers are written in this fashion.

The problem with these concurrent servers is the amount of CPU time that it takes to `fork` a child for each client. Years ago (the late 1980s), when a busy server handled

hundreds or perhaps even a few thousand clients per day, this was OK. But the explosion of the World Wide Web (WWW) has changed this attitude. Busy Web servers measure the number of TCP connections per day in the millions. This is for an individual host, and the busiest sites run multiple hosts, distributing the load among the hosts. (Section 14.2 of TCPv3 talks about a common way to distribute this load using what is called "DNS round robin.") Later sections describe various techniques that avoid the per-client `fork` incurred by a concurrent server but concurrent servers are still common.

Figure 27.5 shows the `main` function for our concurrent TCP server.

server/serv01.c

```
 1 #include    "unp.h"

 2 int
 3 main(int argc, char **argv)
 4 {
 5     int     listenfd, connfd;
 6     pid_t   childpid;
 7     void    sig_chld(int), sig_int(int), web_child(int);
 8     socklen_t clilen, addrlen;
 9     struct sockaddr *cliaddr;

10     if (argc == 2)
11         listenfd = Tcp_listen(NULL, argv[1], &addrlen);
12     else if (argc == 3)
13         listenfd = Tcp_listen(argv[1], argv[2], &addrlen);
14     else
15         err_quit("usage: serv01 [ <host> ] <port#>");
16     cliaddr = Malloc(addrlen);

17     Signal(SIGCHLD, sig_chld);
18     Signal(SIGINT, sig_int);

19     for ( ; ; ) {
20         clilen = addrlen;
21         if ( (connfd = accept(listenfd, cliaddr, &clilen)) < 0) {
22             if (errno == EINTR)
23                 continue;       /* back to for() */
24             else
25                 err_sys("accept error");
26         }
27         if ( (childpid = Fork()) == 0) {    /* child process */
28             Close(listenfd);    /* close listening socket */
29             web_child(connfd);  /* process the request */
30             exit(0);
31         }
32         Close(connfd);          /* parent closes connected socket */
33     }
34 }
```

server/serv01.c

Figure 27.5 `main` function for TCP concurrent server.

This function is similar to Figure 5.12: it calls `fork` for each client connection and handles the `SIGCHLD` signals from the terminating children. This function, however,

we have made protocol independent by calling our `tcp_listen` function. We do not show the `sig_chld` signal handler: it is the same as Figure 5.11, with the `printf` removed.

We also catch the `SIGINT` signal, generated when we type our terminal interrupt key. We type this key after the client completes, to print the CPU time required for the program. Figure 27.6 shows the signal handler. This is an example of a signal handler that does not return.

server/serv01.c

```
35 void
36 sig_int(int signo)
37 {
38     void    pr_cpu_time(void);

39     pr_cpu_time();
40     exit(0);
41 }
```

server/serv01.c

Figure 27.6 Signal handler for `SIGINT`.

Figure 27.7 shows the `pr_cpu_time` function that is called by the signal handler.

server/pr_cpu_time.c

```
 1 #include    "unp.h"
 2 #include    <sys/resource.h>

 3 #ifndef HAVE_GETRUSAGE_PROTO
 4 int     getrusage(int, struct rusage *);
 5 #endif

 6 void
 7 pr_cpu_time(void)
 8 {
 9     double  user, sys;
10     struct rusage myusage, childusage;

11     if (getrusage(RUSAGE_SELF, &myusage) < 0)
12         err_sys("getrusage error");
13     if (getrusage(RUSAGE_CHILDREN, &childusage) < 0)
14         err_sys("getrusage error");

15     user = (double) myusage.ru_utime.tv_sec +
16         myusage.ru_utime.tv_usec / 1000000.0;
17     user += (double) childusage.ru_utime.tv_sec +
18         childusage.ru_utime.tv_usec / 1000000.0;
19     sys = (double) myusage.ru_stime.tv_sec +
20         myusage.ru_stime.tv_usec / 1000000.0;
21     sys += (double) childusage.ru_stime.tv_sec +
22         childusage.ru_stime.tv_usec / 1000000.0;

23     printf("\nuser time = %g, sys time = %g\n", user, sys);
24 }
```

server/pr_cpu_time.c

Figure 27.7 `pr_cpu_time` function: print total CPU time.

The `getrusage` function is called twice to return the resource utilization of both the calling process (`RUSAGE_SELF`) and of all the terminated children of the calling process (`RUSAGE_CHILDREN`). The values printed are the total user time (CPU time spent in the user process) and the total system time (CPU time spent within the kernel executing on behalf of the calling process).

Returning to Figure 27.5, it calls the function `web_child` to handle each client request. Figure 27.8 shows this function.

server/web_child.c

```
 1 #include    "unp.h"
 2 #define MAXN    16384           /* max #bytes that a client can request */
 3 void
 4 web_child(int sockfd)
 5 {
 6     int     ntowrite;
 7     ssize_t nread;
 8     char    line[MAXLINE], result[MAXN];
 9     for ( ; ; ) {
10         if ( (nread = Readline(sockfd, line, MAXLINE)) == 0)
11             return;             /* connection closed by other end */
12             /* line from client specifies #bytes to write back */
13         ntowrite = atol(line);
14         if ((ntowrite <= 0) || (ntowrite > MAXN))
15             err_quit("client request for %d bytes", ntowrite);
16         Writen(sockfd, result, ntowrite);
17     }
18 }
```

server/web_child.c

Figure 27.8 `web_child` function to handle each client's request.

After the client establishes the connection with the server, the client writes a single line specifying the number of bytes the server must return to the client. This is somewhat similar to HTTP: the client sends a small request and the server responds with the desired information (often an HTML file or a GIF image, for example). In the case of HTTP the server normally closes the connection after sending back the requested data, although newer versions are using *persistent connections*, holding the TCP connection open for additional client requests. In our `web_child` function the server allows additional requests from the client, but we saw in Figure 27.4 that our client sends only one request per connection and the client then closes the connection.

Row 1 of Figure 27.1 shows the timing result for this concurrent server. When compared to the subsequent lines in this figure, we see that the concurrent server requires the most amount of CPU time, which is what we expect with one `fork` per client.

One server design that we do not measure in this chapter is one invoked by `inetd`, which we covered in Section 12.5. From a process control perspective a server invoked by `inetd`

involves a `fork` and an `exec`, so the CPU time will be even greater than the times shown in row 1 of Figure 27.1.

27.6 TCP Preforked Server, No Locking around `accept`

Our first of the "enhanced" TCP servers uses a technique called *preforking*. Instead of doing one `fork` per client, the server preforks some number of children when it starts, and then the children are ready to service the clients as each client connection arrives. Figure 27.9 shows a scenario where the parent has preforked N children and two clients are currently connected.

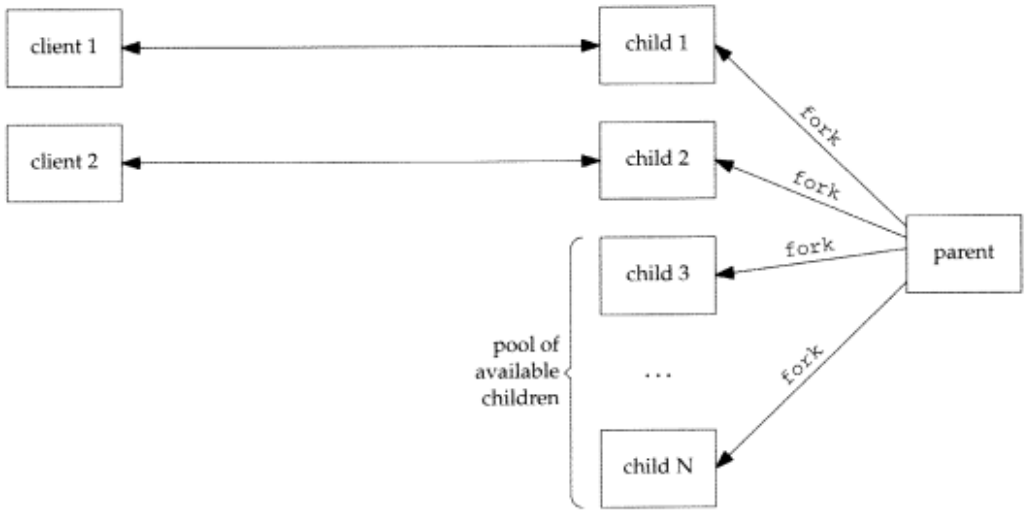


Figure 27.9 Preforking of children by server.

The advantage of this technique is that new clients can be handled without the cost of a `fork` by the parent. The disadvantage is that the parent must guess when it starts how many children to prefork. If the number of clients at any time ever equals the number of children, additional clients are ignored until a child is available. But recall from Section 4.5 that the clients are not completely ignored. The kernel will complete the three-way handshake for any additional clients, up to the `listen` backlog for this socket, and then pass the completed connections to the server when it calls `accept`. But the client application can notice a degradation in response time because even though its `connect` might return immediately, its first request might not be handled by the server for some time.

With some extra coding the server can always handle the client load. What the parent must do is continually monitor the number of available children, and if this value drops below some threshold, the parent must `fork` additional children. Also, if the number of available children exceeds another threshold, the parent can terminate some of the excess children, because when we discuss Figure 27.2 we will see that having too many available children can degrade performance too.

But before worrying about these enhancements, let's examine the basic structure of this type of server. Figure 27.10 shows the main function for the first version of our preforked server.

server/serv02.c

```
 1 #include    "unp.h"
 2 static int nchildren;
 3 static pid_t *pids;
 4 int
 5 main(int argc, char **argv)
 6 {
 7     int     listenfd, i;
 8     socklen_t addrlen;
 9     void    sig_int(int);
10     pid_t   child_make(int, int, int);
11     if (argc == 3)
12         listenfd = Tcp_listen(NULL, argv[1], &addrlen);
13     else if (argc == 4)
14         listenfd = Tcp_listen(argv[1], argv[2], &addrlen);
15     else
16         err_quit("usage: serv02 [ <host> ] <port#> <#children>");
17     nchildren = atoi(argv[argc - 1]);
18     pids = Calloc(nchildren, sizeof(pid_t));
19     for (i = 0; i < nchildren; i++)
20         pids[i] = child_make(i, listenfd, addrlen);     /* parent returns */
21     Signal(SIGINT, sig_int);
22     for ( ; ; )
23         pause();                /* everything done by children */
24 }
```

server/serv02.c

Figure 27.10 main function for preforked server.

11-18 An additional command-line argument is the number of children to prefork. An array is allocated to hold the process IDs of the children, which we need when the program terminates, to allow the main function to terminate all the children.

19-20 Each child is created by child_make, which we examine in Figure 27.12.

Our signal handler for SIGINT, which we show in Figure 27.11, differs from Figure 27.6.

30-34 getrusage reports on the resource utilization of *terminated* children, so we must terminate all the children before calling pr_cpu_time. We do this by sending SIGTERM to each child, and then we wait for all the children.

Figure 27.12 shows the child_make function, which is called by main to create each child.

7-9 fork creates each child and only the parent returns. The child calls the function child_main, which we show in Figure 27.13 and which is an infinite loop.

server/serv02.c

```
25 void
26 sig_int(int signo)
27 {
28     int     i;
29     void    pr_cpu_time(void);

30         /* terminate all children */
31     for (i = 0; i < nchildren; i++)
32         kill(pids[i], SIGTERM);
33     while (wait(NULL) > 0)      /* wait for all children */
34         ;
35     if (errno != ECHILD)
36         err_sys("wait error");

37     pr_cpu_time();
38     exit(0);
39 }
```

server/serv02.c

Figure 27.11 Signal handler for `SIGINT`.

server/child02.c

```
 1 #include    "unp.h"

 2 pid_t
 3 child_make(int i, int listenfd, int addrlen)
 4 {
 5     pid_t   pid;
 6     void    child_main(int, int, int);

 7     if ( (pid = Fork()) > 0)
 8         return (pid);           /* parent */

 9     child_main(i, listenfd, addrlen);   /* never returns */
10 }
```

server/child02.c

Figure 27.12 `child_make` function: create each child.

server/child02.c

```
11 void
12 child_main(int i, int listenfd, int addrlen)
13 {
14     int     connfd;
15     void    web_child(int);
16     socklen_t clilen;
17     struct sockaddr *cliaddr;

18     cliaddr = Malloc(addrlen);

19     printf("child %ld starting\n", (long) getpid());
20     for ( ; ; ) {
21         clilen = addrlen;
22         connfd = Accept(listenfd, cliaddr, &clilen);

23         web_child(connfd);      /* process the request */
24         Close(connfd);
25     }
26 }
```

server/child02.c

Figure 27.13 `child_main` function: infinite loop executed by each child.

20-25 Each child calls accept and when this returns, the function web_child (Figure 27.8) handles the client request. The child continues in this loop until terminated by the parent.

4.4BSD Implementation

If you have never seen this type of arrangement (multiple processes calling accept on the same listening descriptor), you probably wonder how it can even work. It's worth a short digression on how this is implemented in Berkeley-derived kernels (e.g., as presented in TCPv2).

The parent creates the listening socket before spawning any children and recall that all descriptors are duplicated in each child each time fork is called. Figure 27.14 shows the arrangement of the proc structures (one per process), the one file structure for the listening descriptor and the one socket structure.

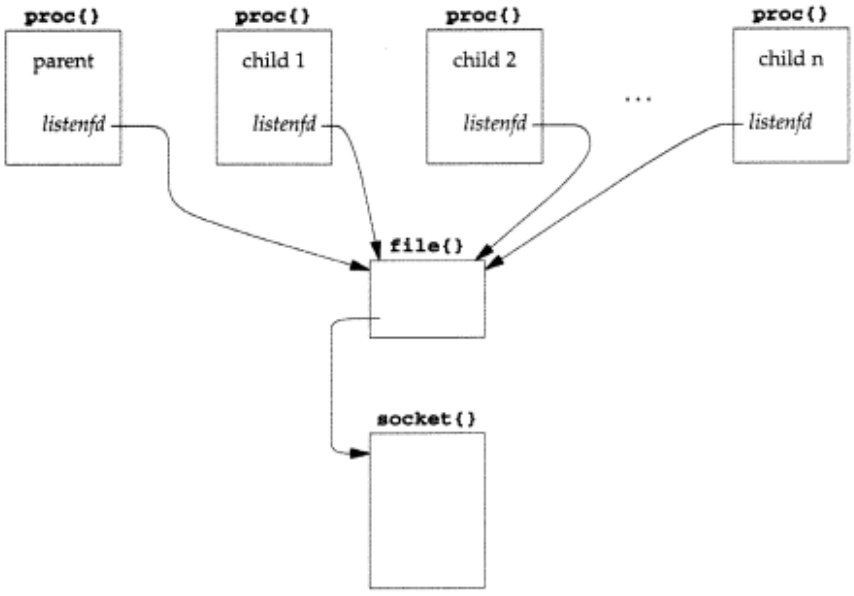


Figure 27.14 Arrangement of proc, file, and socket structures.

Descriptors are just an index in an array in the proc structure that reference a file structure. One of the properties of the duplication of descriptors in the child that occurs with fork is that a given descriptor in the child references the same file structure as that same descriptor in the parent. Each file structure has a reference count which starts at one when the file or socket is opened and is incremented by one each time fork is called or each time the descriptor is duped. In our example with N children, the reference count in the file structure would be N + 1 (don't forget the parent that still has the listening descriptor open, even though the parent never calls accept).

When the program starts, N children are created, and all N call accept and all are put to sleep by the kernel (line 140, p. 458 of TCPv2). When the first client connection

arrives, all N children are awakened. This is because all N have gone to sleep on the same "wait channel," the `so_timeo` member of the `socket` structure, because all N share the same listening descriptor, which points to the same `socket` structure. Even though all N are awakened, the first of the N to run will obtain the connection and the remaining N – 1 will all go back to sleep, because when each of the remaining N – 1 execute the statement on line 135 of p. 458 of TCPv2, the queue length will be 0 since the first child to run already took the connection.

This is sometimes called the *thundering herd* problem because all N are awakened even though only one will obtain the connection. Nevertheless, the code works, with the performance side effect of waking up too many processes each time a connection is ready to be `accept`ed. We now measure this performance effect.

Effect of Too Many children

The CPU time of 1.8 for the BSD/OS server in row 2 of Figure 27.1 is for 15 children and a maximum of 10 simultaneous clients. We can measure the effect of the thundering herd problem by just increasing the number of children for the same maximum number of clients (10). We show these CPU times in Figure 27.2, for this example, and for two other examples that we discuss in future sections. Here we discuss only the `accept` blocking and save discussion of the remaining four columns for later sections.

We see an increase in the CPU time every time we add another 15 (unneeded) children. To avoid the thundering herd problem we do not want too many extra children hanging around.

> Some Unix kernels have a function, often named `wakeup_one`, that wakes up only one process that is waiting for some event, instead of waking up all processes waiting for the event [Schimmel 1994]. The BSD/OS kernel does not have such a function.

Distribution of Connections to the Children

The next thing to examine is the distribution of the client connections to the pool of available children that are blocked in the call to `accept`. To collect this information we modify the `main` function to allocate an array of long integer counters in shared memory, one counter per child. This is done with

```
long   *cptr, *meter(int);   /* for counting #clients/child */

cptr = meter(nchildren);     /* before spawning children */
```

Figure 27.15 shows the `meter` function.

We use anonymous memory mapping, if supported (e.g., 4.4BSD), or the mapping of `/dev/zero` (e.g., SVR4). Since the array is created by `mmap` before the children are spawned, the array is then shared between this process (the parent) and all of its children that are created later by `fork`.

We then modify our `child_main` function (Figure 27.13) so that each child increments its counter when `accept` returns and our `SIGINT` handler prints this array after all the children are terminated.

server/meter.c

```
 1 #include    "unp.h"
 2 #include    <sys/mman.h>

 3 /*
 4  * Allocate an array of "nchildren" longs in shared memory that can
 5  * be used as a counter by each child of how many clients it services.
 6  * See pp. 467-470 of "Advanced Programming in the Unix Environment".
 7  */

 8 long *
 9 meter(int nchildren)
10 {
11     int     fd;
12     long   *ptr;

13 #ifdef  MAP_ANON
14     ptr = Mmap(0, nchildren * sizeof(long), PROT_READ | PROT_WRITE,
15                MAP_ANON | MAP_SHARED, -1, 0);
16 #else
17     fd = Open("/dev/zero", O_RDWR, 0);

18     ptr = Mmap(0, nchildren * sizeof(long), PROT_READ | PROT_WRITE,
19                MAP_SHARED, fd, 0);
20     Close(fd);
21 #endif

22     return (ptr);
23 }
```

server/meter.c

Figure 27.15 `meter` function to allocate an array in shared memory.

Figure 27.3 shows the distribution. When the available children are blocked in the call to `accept`, the kernel's scheduling algorithm distributes the connections uniformly to all the children.

`select` Collisions

While looking at this example under 4.4BSD we can also examine another poorly understood, but rare phenomenon. Section 16.13 of TCPv2 talks about *collisions* with the `select` function and how the kernel handles this possibility. A collision occurs when multiple processes call `select` on the same descriptor, because room is allocated in the `socket` structure for only one process ID to be awakened when the descriptor is ready. If multiple processes are waiting for the same descriptor, the kernel must wake up *all* processes that are blocked in a call to `select`, since it doesn't know which processes are affected by the descriptor that just became ready.

We can force `select` collisions with our example by preceding the call to `accept` in Figure 27.13 with a call to `select`, waiting for readability on the listening socket. The children will spend their time blocked in this call to `select` instead of in the call to `accept`. Figure 27.16 shows the portion of the `child_main` function that changes, using plus signs to note the lines that have changed from Figure 27.13.

```
    printf("child %ld starting\n", (long) getpid());
+   FD_ZERO(&rset);
    for ( ; ; ) {
+       FD_SET(listenfd, &rset);
+       Select(listenfd+1, &rset, NULL, NULL, NULL);
+       if (FD_ISSET(listenfd, &rset) == 0)
+           err_quit("listenfd readable");
+
        clilen = addrlen;
        connfd = Accept(listenfd, cliaddr, &clilen);

        web_child(connfd);      /* process the request */
        Close(connfd);
    }
```

Figure 27.16 Modification to Figure 27.13 to block in `select` instead of `accept`.

If we make this change and then examine the BSD/OS kernel's `nselcoll` counter before and after, we see 1814 collisions one time we run the sever, and 2045 collisions the next time. Since the two clients create a total of 5000 connections for each run of the server, this corresponds to about 35–40% of the calls to `select` invoking a collision.

If we compare the BSD/OS server's CPU time for this example, the value of 1.8 in Figure 27.1 increases to 2.9 when we add the call to `select`. Part of this increase is probably because of the additional system call (since we are calling `select` and `accept`, instead of just `accept`) and another part is probably because of the kernel overhead in handling the collisions.

The lesson to be learned from this discussion is when multiple processes are blocking on the same descriptor, it is better to block in a function such as `accept`, instead of blocking in `select`.

27.7 TCP Preforked Server, File Locking around `accept`

The implementation that we just described for 4.4BSD that allows multiple processes to call `accept` on the same listening descriptor works only with Berkeley-derived kernels that implement `accept` within the kernel. System V kernels, which implement `accept` as a library function, do not allow this. Indeed, if we run the server from the previous section on Solaris 2.5 (an SVR4-based kernel) with multiple children, soon after the clients start connecting to the server a call to `accept` in one of the children returns `EPROTO`, which means a protocol error.

> The reasons for this problem with the SVR4 library version of `accept` arise from the streams implementation (Chapter 33) and the fact that the library `accept` is not an atomic operation. Solaris 2.6 fixes this problem but the problem still exists in most other SVR4 implementations.

The solution is for the application to place a *lock* of some form around the call to `accept`, so that only one process at a time is blocked in the call to `accept`. The remaining children will be blocked trying to obtain the lock.

There are various ways to provide this locking around the call to accept, as we describe in the second volume of this series. In this section we use Posix file locking with the fcntl function.

The only change to the main function (Figure 27.10) is adding a call to our my_lock_init function before the loop that creates the children:

```
+   my_lock_init("/tmp/lock.XXXXXX");  /* one lock file for all children */
    for (i = 0; i < nchildren; i++)
        pids[i] = child_make(i, listenfd, addrlen); /* parent returns */
```

The child_make function remains the same as Figure 27.12. The only change to our child_main function (Figure 27.13) is to obtain a lock before calling accept and release the lock after accept returns:

```
    for ( ; ; ) {
        clilen = addrlen;
+       my_lock_wait();
        connfd = Accept(listenfd, cliaddr, &clilen);
+       my_lock_release();

        web_child(connfd);          /* process the request */
        Close(connfd);
    }
```

Figure 27.17 shows our my_lock_init function that uses Posix file locking.

server/lock_fcntl.c

```
 1 #include    "unp.h"

 2 static struct flock lock_it, unlock_it;
 3 static int lock_fd = -1;
 4                     /* fcntl() will fail if my_lock_init() not called */

 5 void
 6 my_lock_init(char *pathname)
 7 {
 8     char    lock_file[1024];

 9         /* must copy caller's string, in case it's a constant */
10     strncpy(lock_file, pathname, sizeof(lock_file));
11     Mktemp(lock_file);

12     lock_fd = Open(lock_file, O_CREAT | O_WRONLY, FILE_MODE);
13     Unlink(lock_file);          /* but lock_fd remains open */

14     lock_it.l_type = F_WRLCK;
15     lock_it.l_whence = SEEK_SET;
16     lock_it.l_start = 0;
17     lock_it.l_len = 0;

18     unlock_it.l_type = F_UNLCK;
19     unlock_it.l_whence = SEEK_SET;
20     unlock_it.l_start = 0;
21     unlock_it.l_len = 0;
22 }
```

server/lock_fcntl.c

Figure 27.17 my_lock_init function using Posix.1 file locking.

9–13 The caller specifies a pathname template as the argument to `my_lock_init` and the `mktemp` function creates a unique pathname based on this template. A file is then created with this pathname and immediately `unlink`ed. By removing the pathname from the directory, if the program crashes, the file completely disappears. But as long as one or more processes have the file open (i.e., the file's reference count is greater than 0), the file itself is not removed. (This is the fundamental difference between removing a pathname from a directory and closing an open file.)

14–21 Two `flock` structures are initialized: one to lock the file and one to unlock the file. The range of the file that is locked starts at byte offset 0 (a `l_whence` of `SEEK_SET` with `l_start` set to 0). Since `l_len` is set to 0, this specifies that the entire file is locked. We never write anything to the file (its length is always 0) but that is OK: the advisory lock is still handled correctly by the kernel.

> The author first initialized these structures when they were declared, using
>
> ```
> static struct flock  lock_it = { F_WRLCK, 0, 0, 0, 0 };
> static struct flock  unlock_it = { F_UNLCK, 0, 0, 0, 0 };
> ```
>
> but there are two problems. First, there is no guarantee that the constant `SEEK_SET` is 0. But more importantly, there is no guarantee by Posix as to the order of the members in the structure. On Solaris and Digital Unix the `l_type` member is the first one in the structure, but on BSD/OS it is not. All Posix guarantees is that the members that Posix requires are present in the structure. Posix does not guarantee the order of the members, and Posix also allows additional, non-Posix members, to be in the structure. Therefore, initializing a structure to anything other than all zeros should always be done by actual C code, and not by an initializer when the structure is allocated.
>
> An exception to this rule is when the structure initializer is provided by the implementation. For example, when initializing a Pthread mutex lock in Chapter 23 we wrote
>
> ```
> pthread_mutex_t  mlock = PTHREAD_MUTEX_INITIALIZER;
> ```
>
> The `pthread_mutex_t` datatype is often a structure, but the initializer is provided by the implementation and can differ from one implementation to the next.

Figure 27.18 shows the two functions that lock and unlock the file. These are just calls to `fcntl`, using the structures that were initialized in Figure 27.17.

This new version of our preforked server now works on SVR4 systems by assuring that only one child process at a time is blocked in the call to `accept`. Comparing rows 2 and 3 in Figure 27.1 for the Digital Unix and BSD/OS servers shows that this type of locking adds to the server's process control CPU time.

> Release 1.1 of the Apache Web server, `http://www.apache.org`, preforks its children and then uses either the technique in the previous section (all children blocked in the call to `accept`), if the implementation allows this, or uses file locking around the `accept`.

Effect of Too Many children

We can check this version to see if the same thundering herd problem exists, which we described in the previous section. Figure 27.2 shows the results when we increase the number of unneeded children. With the Solaris column that uses file locking around the

server/lock_fcntl.c

```
23 void
24 my_lock_wait()
25 {
26     int     rc;

27     while ( (rc = fcntl(lock_fd, F_SETLKW, &lock_it)) < 0) {
28         if (errno == EINTR)
29             continue;
30         else
31             err_sys("fcntl error for my_lock_wait");
32     }
33 }

34 void
35 my_lock_release()
36 {
37     if (fcntl(lock_fd, F_SETLKW, &unlock_it) < 0)
38         err_sys("fcntl error for my_lock_release");
39 }
```

server/lock_fcntl.c

Figure 27.18 `my_lock_wait` and `my_lock_release` functions using `fcntl`.

`accept` we are only able to measure up through 75 children, as the next step (90) does something that causes the CPU time to increase a lot. One possible reason is that the system ran out of memory with all the processes and started swapping.

Distribution of Connections to the Children

We can examine the distribution of the clients to the pool of available children by using the function that we described with Figure 27.15. Figure 27.3 shows the result. All three operating systems distribute the file locks uniformly to the waiting processes.

27.8 TCP Preforked Server, Thread Locking around `accept`

As we mentioned there are various ways to implement locking between processes. The Posix file locking in the previous section is portable to all Posix-compliant systems, but it involves filesystem operations, which can take time. In this section we use thread locking, taking advantage of the fact that this can be used not only for locking between the threads within a given process, but also for locking between different processes.

Our `main` function remains the same as in the previous section, as do our `child_make` and `child_main` functions. The only thing that changes is our three locking functions. To use thread locking between different processes requires that (1) the mutex variable be stored in memory that is shared between all the processes, and (2) the thread library must be told that the mutex is shared among different processes.

This also requires that the thread library support the `PTHREAD_PROCESS_SHARED` attribute. Digital Unix 4.0b does not, so we cannot run this server under this operating system.

There are various ways to share memory between different processes, as we describe in the second volume of this series. In our example we will use the `mmap` function with the `/dev/zero` device, which works under Solaris and other SVR4 kernels. Figure 27.19 shows our `my_lock_init` function.

server/lock_pthread.c

```
 1 #include    "unpthread.h"
 2 #include    <sys/mman.h>
 3 static pthread_mutex_t *mptr;   /* actual mutex will be in shared memory */
 4 void
 5 my_lock_init(char *pathname)
 6 {
 7     int     fd;
 8     pthread_mutexattr_t mattr;
 9     fd = Open("/dev/zero", O_RDWR, 0);
10     mptr = Mmap(0, sizeof(pthread_mutex_t), PROT_READ | PROT_WRITE,
11                 MAP_SHARED, fd, 0);
12     Close(fd);
13     Pthread_mutexattr_init(&mattr);
14     Pthread_mutexattr_setpshared(&mattr, PTHREAD_PROCESS_SHARED);
15     Pthread_mutex_init(mptr, &mattr);
16 }
```

server/lock_pthread.c

Figure 27.19 `my_lock_init` function using Pthread locking between processes.

9–12 We `open` `/dev/zero` and then call `mmap`. The number of bytes that are mapped is the size of a `pthread_mutex_t` variable. The descriptor is then closed, which is OK, because the descriptor has been memory mapped.

13–15 In our previous Pthread mutex examples we initialized the global or static mutex variable using the constant `PTHREAD_MUTEX_INITIALIZER` (e.g., Figure 23.18). But with a mutex in shared memory we must call some Pthread library functions to tell the library that the mutex is in shared memory and that it will be used for locking between different processes. We first initialize a `pthread_mutexattr_t` structure with the default attributes for a mutex and then set the `PTHREAD_PROCESS_SHARED` attribute. (The default for this attribute is `PTHREAD_PROCESS_PRIVATE`, allowing use only within a single process.) `pthread_mutex_init` then initializes the mutex with these attributes.

Figure 27.20 shows our `my_lock_wait` and `my_lock_release` functions. Each is now just a call to a Pthread function to lock or unlock the mutex.

Comparing rows 3 and 4 in Figure 27.1 for the Solaris server shows that thread mutex locking is faster than file locking.

27.9 TCP Preforked Server, Descriptor Passing

The final modification to our preforked server is to have only the parent call `accept` and then "pass" the connected socket to one child. This gets around the possible need

server/lock_pthread.c

```
17 void
18 my_lock_wait()
19 {
20     Pthread_mutex_lock(mptr);
21 }

22 void
23 my_lock_release()
24 {
25     Pthread_mutex_unlock(mptr);
26 }
```

server/lock_pthread.c

Figure 27.20 `my_lock_wait` and `my_lock_release` functions using Pthread locking.

for locking around the call to `accept` in all the children but requires some form of descriptor passing from the parent to the children. This technique also complicates the code somewhat because the parent must keep track of which children are busy and which are free to pass a new socket to a free child.

In the previous preforked examples the process never cared which child received a client connection. The operating system handled this detail, giving one of the children the first call to `accept` or giving one of the children the file lock or the mutex lock. The first five columns of Figure 27.3 also show that the three operating systems that we are measuring do this in a fair, round-robin fashion.

With this example we need to maintain a structure of information about each child. We show our `child.h` header in Figure 27.21 that defines our `Child` structure.

server/child.h

```
1 typedef struct {
2     pid_t   child_pid;          /* process ID */
3     int     child_pipefd;       /* parent's stream pipe to/from child */
4     int     child_status;       /* 0 = ready */
5     long    child_count;        /* #connections handled */
6 } Child;

7 Child  *cptr;                   /* array of Child structures; calloc'ed */
```

server/child.h

Figure 27.21 `Child` structure.

We store the child's process ID, the parent's stream pipe descriptor that is connected to the child, the child's status, and a count of the number of clients that the child has handled. We will print this counter in our `SIGINT` handler to see the distribution of the client requests among the children.

Let us first look at the `child_make` function, which we show in Figure 27.22. We create a stream pipe, a Unix domain stream socket (Chapter 14), before calling `fork`. After the child is created, the parent closes one descriptor (`sockfd[1]`) and the child closes the other descriptor (`sockfd[0]`). Furthermore, the child duplicates its end of the stream pipe (`sockfd[1]`) onto standard error, so that each child just reads and writes to standard error to communicate with the parent. This gives us the arrangement shown in Figure 27.23.

server/child05.c

```
 1 #include    "unp.h"
 2 #include    "child.h"

 3 pid_t
 4 child_make(int i, int listenfd, int addrlen)
 5 {
 6     int     sockfd[2];
 7     pid_t   pid;
 8     void    child_main(int, int, int);

 9     Socketpair(AF_LOCAL, SOCK_STREAM, 0, sockfd);

10     if ( (pid = Fork()) > 0) {
11         Close(sockfd[1]);
12         cptr[i].child_pid = pid;
13         cptr[i].child_pipefd = sockfd[0];
14         cptr[i].child_status = 0;
15         return (pid);           /* parent */
16     }
17     Dup2(sockfd[1], STDERR_FILENO);     /* child's stream pipe to parent */
18     Close(sockfd[0]);
19     Close(sockfd[1]);
20     Close(listenfd);            /* child does not need this open */
21     child_main(i, listenfd, addrlen);   /* never returns */
22 }
```

server/child05.c

Figure 27.22 `child_make` function descriptor passing preforked server.

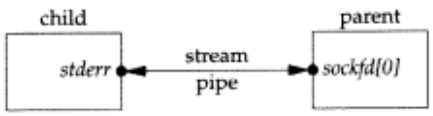


Figure 27.23 Stream pipe after parent and child both close one end.

After all the children are created, we have the arrangement shown in Figure 27.24. We close the listening socket in each child, as only the parent calls `accept`. We show that the parent must handle the listening socket along with all the stream sockets. As you might guess, the parent uses `select` to multiplex all these descriptors.

Figure 27.25 shows the `main` function. The changes from previous versions of this function are that descriptor sets are allocated and the bits corresponding to the listening socket along with the stream pipe to each child are turned on in the set. The maximum descriptor value is also calculated. We allocate memory for the array of `Child` structures. The main loop is driven by a call to `select`.

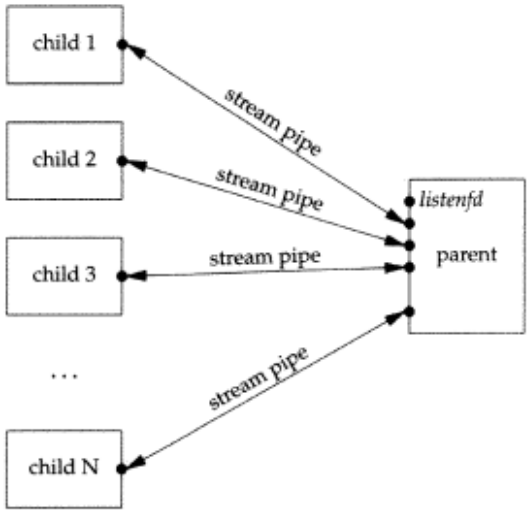


Figure 27.24 Stream pipes after all children have been created.

server/serv05.c

```
 1 #include    "unp.h"
 2 #include    "child.h"

 3 static int nchildren;

 4 int
 5 main(int argc, char **argv)
 6 {
 7     int     listenfd, i, navail, maxfd, nsel, connfd, rc;
 8     void    sig_int(int);
 9     pid_t   child_make(int, int, int);
10     ssize_t n;
11     fd_set  rset, masterset;
12     socklen_t addrlen, clilen;
13     struct sockaddr *cliaddr;

14     if (argc == 3)
15         listenfd = Tcp_listen(NULL, argv[1], &addrlen);
16     else if (argc == 4)
17         listenfd = Tcp_listen(argv[1], argv[2], &addrlen);
18     else
19         err_quit("usage: serv05 [ <host> ] <port#> <#children>");

20     FD_ZERO(&masterset);
21     FD_SET(listenfd, &masterset);
22     maxfd = listenfd;
23     cliaddr = Malloc(addrlen);

24     nchildren = atoi(argv[argc - 1]);
25     navail = nchildren;
26     cptr = Calloc(nchildren, sizeof(Child));
```

```
27          /* prefork all the children */
28      for (i = 0; i < nchildren; i++) {
29          child_make(i, listenfd, addrlen);    /* parent returns */
30          FD_SET(cptr[i].child_pipefd, &masterset);
31          maxfd = max(maxfd, cptr[i].child_pipefd);
32      }
33      Signal(SIGINT, sig_int);
34      for ( ; ; ) {
35          rset = masterset;
36          if (navail <= 0)
37              FD_CLR(listenfd, &rset);     /* turn off if no available children */
38          nsel = Select(maxfd + 1, &rset, NULL, NULL, NULL);
39              /* check for new connections */
40          if (FD_ISSET(listenfd, &rset)) {
41              clilen = addrlen;
42              connfd = Accept(listenfd, cliaddr, &clilen);
43              for (i = 0; i < nchildren; i++)
44                  if (cptr[i].child_status == 0)
45                      break;       /* available */
46              if (i == nchildren)
47                  err_quit("no available children");
48              cptr[i].child_status = 1;    /* mark child as busy */
49              cptr[i].child_count++;
50              navail--;
51              n = Write_fd(cptr[i].child_pipefd, "", 1, connfd);
52              Close(connfd);
53              if (--nsel == 0)
54                  continue;        /* all done with select() results */
55          }
56              /* find any newly-available children */
57          for (i = 0; i < nchildren; i++) {
58              if (FD_ISSET(cptr[i].child_pipefd, &rset)) {
59                  if ( (n = Read(cptr[i].child_pipefd, &rc, 1)) == 0)
60                      err_quit("child %d terminated unexpectedly", i);
61                  cptr[i].child_status = 0;
62                  navail++;
63                  if (--nsel == 0)
64                      break;       /* all done with select() results */
65              }
66          }
67      }
68  }
```

server/serv05.c

Figure 27.25 `main` function that uses descriptor passing.

Turn off listening socket if no available children

36–37 The counter `navail` keeps track of the number of available children. If this counter is 0, the listening socket is turned off in the descriptor set for `select`. This prevents us

from accepting a new connection for which there is no available child. The kernel still queues these incoming connections, up to the listen backlog, but we do not want to accept them until we have a child ready to process the client.

accept new connection

39-55 If the listening socket is readable, a new connection is ready to accept. We find the first available child and pass the connected socket to the child using our write_fd function from Figure 14.13. We write 1 byte along with the descriptor, but the recipient does not look at the contents of this byte. The parent closes the connected socket.

We always start looking for an available child with the first entry in the array of Child structures. This means the first children in the array always receive new connections to process before later elements in the array. We will verify this when we discuss Figure 27.3 and look at the child_count counters after the server terminates. If we didn't want this bias toward earlier children, we could remember which child received the most previous connection and start our search one element past that one each time, circling back to the first element when we reach the end. There is no advantage in doing this (it really doesn't matter which child handles a client request if multiple children are available), unless the operating system scheduling algorithm penalizes processes with longer total CPU times. Spreading the load more evenly among all the children would tend to average out their total CPU times.

Handle any newly available children

56-66 We will see that our child_main function writes a single byte back to the parent across the stream pipe when the child has finished with a client. That makes the parent's end of the stream pipe readable. We read the single byte (ignoring its value) and then mark the child as available. Should the child terminate unexpectedly, its end of the stream pipe will be closed, and the read returns 0. We catch this and terminate, but a better approach is to log the error and spawn a new child to replace the one that terminated.

Our child_main function is shown in Figure 27.26.

Wait for descriptor from parent

32-33 This function differs from the ones in the previous two sections, because our child no longer calls accept. Instead the child blocks in a call to read_fd waiting for the parent to pass it a connected socket descriptor to process.

Tell parent we are ready

38 When we have finished with the client we write 1 byte across the stream pipe to tell the parent we are available.

In Figure 27.1 comparing rows 4 and 5 for our Solaris server we see that this server is slower than the version in the previous section that used thread locking between the children. Comparing rows 3 and 5 for our Digital Unix and BSD/OS servers leads to a similar conclusion: passing a descriptor across the stream pipe to each child, and writing a byte back across the stream pipe from the child takes more time than locking and unlocking either a mutex in shared memory or a file lock.

server/child05.c

```
23 void
24 child_main(int i, int listenfd, int addrlen)
25 {
26     char    c;
27     int     connfd;
28     ssize_t n;
29     void    web_child(int);

30     printf("child %ld starting\n", (long) getpid());
31     for ( ; ; ) {
32         if ( (n = Read_fd(STDERR_FILENO, &c, 1, &connfd)) == 0)
33             err_quit("read_fd returned 0");
34         if (connfd < 0)
35             err_quit("no descriptor from read_fd");

36         web_child(connfd);      /* process the request */
37         Close(connfd);

38         Write(STDERR_FILENO, "", 1);    /* tell parent we're ready again */
39     }
40 }
```

server/child05.c

Figure 27.26 `child_main` function: descriptor passing, preforked server.

Figure 27.3 shows the distribution of the `child_count` counters in the `Child` structure, which we print in the `SIGINT` handler when the server is terminated. The earlier children do handle more clients, as we discussed with Figure 27.25.

27.10 TCP Concurrent Server, One Thread per Client

The last five sections have focused on one process per client, both one `fork` per client and preforking some number of children. If the server supports threads, we can use threads instead of child processes.

Our first threaded version is shown in Figure 27.27. It is a modification of Figure 27.5 that creates one thread per client, instead of one process per client. This version is very similar to Figure 23.3.

Main thread loop

19–23 The main thread blocks in a call to `accept` and each time a client connection is returned a new thread is created by `pthread_create`. The function executed by the new thread is `doit` and its argument is the connected socket.

Per-thread function

25–33 The `doit` function detaches itself so the main thread does not have to wait for it and calls our `web_client` function (Figure 27.4). When that function returns the connected socket is closed.

server/serv06.c

```
 1 #include    "unpthread.h"
 2 int
 3 main(int argc, char **argv)
 4 {
 5     int     listenfd, connfd;
 6     void    sig_int(int);
 7     void   *doit(void *);
 8     pthread_t tid;
 9     socklen_t clilen, addrlen;
10     struct sockaddr *cliaddr;
11     if (argc == 2)
12         listenfd = Tcp_listen(NULL, argv[1], &addrlen);
13     else if (argc == 3)
14         listenfd = Tcp_listen(argv[1], argv[2], &addrlen);
15     else
16         err_quit("usage: serv06 [ <host> ] <port#>");
17     cliaddr = Malloc(addrlen);
18     Signal(SIGINT, sig_int);
19     for ( ; ; ) {
20         clilen = addrlen;
21         connfd = Accept(listenfd, cliaddr, &clilen);
22         Pthread_create(&tid, NULL, &doit, (void *) connfd);
23     }
24 }
25 void *
26 doit(void *arg)
27 {
28     void    web_child(int);
29     Pthread_detach(pthread_self());
30     web_child((int) arg);
31     Close((int) arg);
32     return (NULL);
33 }
```

server/serv06.c

Figure 27.27 `main` function for TCP threaded server.

We note from Figure 27.1 that this simple threaded version is faster on both Solaris and Digital Unix than even the fastest of the preforked versions. This one-thread-per-client version is also many times faster than the one-child-per-client version (row 1).

> In Section 23.5 we noted three alternatives for converting a function that is not thread-safe into one that is thread-safe. Our `web_child` function calls our `readline` function, and the version shown in Figure 3.17 is not thread-safe. Alternatives 2 and 3 from Section 23.5 were timed with the example in Figure 27.27. The speedup from alternative 3 to alternative 2 was less than one percent, probably because `readline` is used only to read the 5-character count from the client. Therefore, for simplicity we use the less efficient version from Figure 3.16 for the threaded server examples in this chapter.

27.11 TCP Prethreaded Server, per-Thread `accept`

We found earlier in this chapter that it is faster to prefork a pool of children than to create one child for every client. On a system that supports threads it is reasonable to expect a similar speedup by creating a pool of threads when the server starts, instead of creating a new thread for every client. The basic design of this server is to create a pool of threads and then let each thread call `accept`. Instead of having each thread block in the call to `accept`, we will use a mutex lock (similar to Section 27.8) that allows only one thread at a time to call `accept`. There is no reason to use file locking to protect the call to `accept` from all the threads, because with multiple threads in a single process we know that a mutex lock can be used.

Figure 27.28 shows the `pthread07.h` header that defines a `Thread` structure that maintains some information about each thread.

server/pthread07.h

```
1 typedef struct {
2     pthread_t thread_tid;        /* thread ID */
3     long    thread_count;        /* #connections handled */
4 } Thread;
5 Thread *tptr;                    /* array of Thread structures; calloc'ed */

6 int     listenfd, nthreads;
7 socklen_t addrlen;
8 pthread_mutex_t mlock;
```

server/pthread07.h

Figure 27.28 `pthread07.h` header.

We also declare a few globals, such as the listening socket descriptor and a mutex variable that all the threads need to share.

Figure 27.29 shows the `main` function.

server/serv07.c

```
1 #include    "unpthread.h"
2 #include    "pthread07.h"

3 pthread_mutex_t mlock = PTHREAD_MUTEX_INITIALIZER;

4 int
5 main(int argc, char **argv)
6 {
7     int     i;
8     void    sig_int(int), thread_make(int);

9     if (argc == 3)
10        listenfd = Tcp_listen(NULL, argv[1], &addrlen);
11    else if (argc == 4)
12        listenfd = Tcp_listen(argv[1], argv[2], &addrlen);
13    else
14        err_quit("usage: serv07 [ <host> ] <port#> <#threads>");
15    nthreads = atoi(argv[argc - 1]);
16    tptr = Calloc(nthreads, sizeof(Thread));
```

```
17      for (i = 0; i < nthreads; i++)
18          thread_make(i);         /* only main thread returns */

19      Signal(SIGINT, sig_int);

20      for ( ; ; )
21          pause();                /* everything done by threads */
22  }
```
server/serv07.c

Figure 27.29 `main` function for prethreaded TCP server.

The `thread_make` and `thread_main` functions are shown in Figure 27.30.

server/pthread07.c

```
 1 #include    "unpthread.h"
 2 #include    "pthread07.h"

 3 void
 4 thread_make(int i)
 5 {
 6     void   *thread_main(void *);

 7     Pthread_create(&tptr[i].thread_tid, NULL, &thread_main, (void *) i);
 8     return;                     /* main thread returns */
 9 }

10 void *
11 thread_main(void *arg)
12 {
13     int     connfd;
14     void    web_child(int);
15     socklen_t clilen;
16     struct sockaddr *cliaddr;

17     cliaddr = Malloc(addrlen);

18     printf("thread %d starting\n", (int) arg);
19     for ( ; ; ) {
20         clilen = addrlen;
21         Pthread_mutex_lock(&mlock);
22         connfd = Accept(listenfd, cliaddr, &clilen);
23         Pthread_mutex_unlock(&mlock);
24         tptr[(int) arg].thread_count++;

25         web_child(connfd);      /* process the request */
26         Close(connfd);
27     }
28 }
```
server/pthread07.c

Figure 27.30 `thread_make` and `thread_main` functions.

Create thread

7 Each thread is created and executes the `thread_main` function. The only argument is the index number of the thread.

21-23 The `thread_main` function calls the functions `pthread_mutex_lock` and `pthread_mutex_unlock` around the call to `accept`.

In Figure 27.1, comparing rows 6 and 7, we see that this latest version of our server is faster than the create-one-thread-per-client version under both Solaris and Digital Unix. We expect this, since we create the pool of threads only once, when the server starts, instead of creating one thread per client. Indeed, this version of our server is the fastest on these two hosts.

Figure 27.3 shows the distribution of the `thread_count` counters in the `Thread` structure, which we print in the `SIGINT` handler when the server is terminated. The uniformity of this distribution is caused by the thread scheduling algorithm that appears to cycle through all the threads in order, when choosing which thread receives the mutex lock.

> On a Berkeley-derived kernel such as Digital Unix we do not need any locking around the call to `accept` and can make a version of Figure 27.30 without any mutex locking and unlocking. Doing so, however, increases the process control CPU time from 3.5 seconds for row 7 in Figure 27.1 to 3.9 seconds. If we look at the two components of the CPU time, the user time and the system time, without any locking the user time decreases (because the locking is done in the threads library which executes in user space) but the system time increases (the kernel's thundering herd as all threads blocked in `accept` are awakened when a connection arrives). Since some form of mutual exclusion is required, to return each connection to a single thread, it is faster for the threads to do this themselves than for the kernel.

27.12 TCP Prethreaded Server, Main Thread `accept`

Our final server design using threads has the main thread create a pool of threads when it starts, and then only the main thread calls `accept` and passes each client connection to one of the available threads in the pool. This is similar to the descriptor passing version in Section 27.9.

The design problem is how does the main thread "pass" the connected socket to one of the available threads in the pool. There are various ways to implement this. We could use descriptor passing, as we did earlier, but there's no need to pass a descriptor from one thread to another, since all the threads and all the descriptors are in the same process. All the receiving thread needs to know is the descriptor number. Figure 27.31 shows the `pthread08.h` header that defines a `Thread` structure, identical to Figure 27.28.

server/pthread08.h

```
1 typedef struct {
2     pthread_t thread_tid;        /* thread ID */
3     long    thread_count;        /* #connections handled */
4 } Thread;
5 Thread *tptr;                   /* array of Thread structures; calloc'ed */

6 #define MAXNCLI 32
7 int     clifd[MAXNCLI], iget, iput;
8 pthread_mutex_t clifd_mutex;
9 pthread_cond_t clifd_cond;
```

server/pthread08.h

Figure 27.31 `pthread08.h` header.

Define shared array to hold connected sockets

6-9 We also define a `clifd` array in which the main thread will store the connected socket descriptors. The available threads in the pool take one of these connected sockets and service the corresponding client. `iput` is the index into this array of the next entry to be stored into by the main thread and `iget` is the index of the next entry to be fetched by one of the threads in the pool. Naturally this data structure that is shared between all the threads must be protected and we use a mutex along with a condition variable.

Figure 27.32 is the `main` function.

Create the pool of threads

23-25 `thread_make` creates each of the threads.

Wait for each client connection

27-38 The main thread blocks in the call to `accept`, waiting for each client connection to arrive. When one arrives, the connected socket is stored in the next entry in the `clifd` array, after obtaining the mutex lock on the array. We also check that the `iput` index has not caught up with the `iget` index, which indicates that our array is not big enough. The condition variable is signaled and the mutex is released, allowing one of the threads in the pool to service this client.

The `thread_make` and `thread_main` functions are shown in Figure 27.33. The former is identical to the version in Figure 27.30.

Wait for client descriptor to service

17-26 Each thread in the pool tries to obtain a lock on the mutex that protects the `clifd` array. When the lock is obtained, there is nothing to do if the `iget` and `iput` indexes are equal. In that case the thread goes to sleep by calling `pthread_cond_wait`. It will be awakened by the call to `pthread_cond_signal` in the main thread after a connection is accepted. When the thread obtains a connection, it calls `web_child`.

The times in Figure 27.1 show that this server is slower than the one in the previous section, in which each thread called `accept` after obtaining a mutex lock. The reason is that this section's example requires both a mutex and a condition variable, compared to just a mutex in Figure 27.30.

If we examine the histogram of the number of clients serviced by each thread in the pool, it is similar to the final column in Figure 27.3. This means the threads library cycles through all the available threads when doing the wakeup based on the condition variable when the main thread calls `pthread_cond_signal`.

server/serv08.c

```
 1 #include    "unpthread.h"
 2 #include    "pthread08.h"

 3 static int nthreads;
 4 pthread_mutex_t clifd_mutex = PTHREAD_MUTEX_INITIALIZER;
 5 pthread_cond_t clifd_cond = PTHREAD_COND_INITIALIZER;

 6 int
 7 main(int argc, char **argv)
 8 {
 9     int     i, listenfd, connfd;
10     void    sig_int(int), thread_make(int);
11     socklen_t addrlen, clilen;
12     struct sockaddr *cliaddr;

13     if (argc == 3)
14         listenfd = Tcp_listen(NULL, argv[1], &addrlen);
15     else if (argc == 4)
16         listenfd = Tcp_listen(argv[1], argv[2], &addrlen);
17     else
18         err_quit("usage: serv08 [ <host> ] <port#> <#threads>");
19     cliaddr = Malloc(addrlen);

20     nthreads = atoi(argv[argc - 1]);
21     tptr = Calloc(nthreads, sizeof(Thread));
22     iget = iput = 0;

23         /* create all the threads */
24     for (i = 0; i < nthreads; i++)
25         thread_make(i);         /* only main thread returns */

26     Signal(SIGINT, sig_int);

27     for ( ; ; ) {
28         clilen = addrlen;
29         connfd = Accept(listenfd, cliaddr, &clilen);

30         Pthread_mutex_lock(&clifd_mutex);
31         clifd[iput] = connfd;
32         if (++iput == MAXNCLI)
33             iput = 0;
34         if (iput == iget)
35             err_quit("iput = iget = %d", iput);
36         Pthread_cond_signal(&clifd_cond);
37         Pthread_mutex_unlock(&clifd_mutex);
38     }
39 }
```

server/serv08.c

Figure 27.32 `main` function for prethreaded server.

server/pthread08.c

```
 1 #include    "unpthread.h"
 2 #include    "pthread08.h"

 3 void
 4 thread_make(int i)
 5 {
 6     void   *thread_main(void *);

 7     Pthread_create(&tptr[i].thread_tid, NULL, &thread_main, (void *) i);
 8     return;                     /* main thread returns */
 9 }

10 void *
11 thread_main(void *arg)
12 {
13     int     connfd;
14     void    web_child(int);

15     printf("thread %d starting\n", (int) arg);
16     for ( ; ; ) {
17         Pthread_mutex_lock(&clifd_mutex);
18         while (iget == iput)
19             Pthread_cond_wait(&clifd_cond, &clifd_mutex);
20         connfd = clifd[iget];   /* connected socket to service */
21         if (++iget == MAXNCLI)
22             iget = 0;
23         Pthread_mutex_unlock(&clifd_mutex);
24         tptr[(int) arg].thread_count++;

25         web_child(connfd);      /* process the request */
26         Close(connfd);
27     }
28 }
```

server/pthread08.c

Figure 27.33 `thread_make` and `thread_main` functions.

27.13 Summary

In this chapter we have looked at nine different server designs and run them all against the same Web-style client, comparing the amount of CPU time spent performing process control:

0. iterative server (baseline measurement; no process control),
1. concurrent server, one `fork` per client,
2. prefork with each child calling `accept`,
3. prefork with file locking to protect `accept`,
4. prefork with thread mutex locking to protect `accept`,
5. prefork with parent passing socket descriptor to child,
6. concurrent server, create one thread per client request,
7. prethreaded with mutex locking to protect `accept`, and
8. prethreaded with main thread calling `accept`.

We can make a few summary comments.

- First, if the server is not heavily used, the traditional concurrent server model, with one `fork` per client is fine. This can even be combined with `inetd`, letting it handle the accepting of each connection. The remainder of our comments are meant for heavily used servers, such as Web servers.
- Creating a pool of children or a pool of threads reduces the process control CPU time compared to the traditional one-`fork`-per-client design, by a factor of 10 or more. The coding is not complicated but what is required, above and beyond the examples that we have shown, is monitoring the number of free children and increasing or decreasing this number as the number of clients being served changes dynamically.
- Some implementations allow multiple children or threads to block in a call to `accept` while on other implementations we must place some type of lock around the call to `accept`. Either file locking or Pthread mutex locking can be used.
- Having all the children or threads call `accept` is normally simpler and faster than having the main thread call `accept` and then pass the descriptor to the child or thread.
- Having all the children or threads block in a call to `accept` is preferable over blocking in a call to `select`, because of the potential for `select` collisions.
- Using threads is normally faster than using processes. But the choice of one-child-per-client or one-thread-per-client depends on what the operating system provides and can also depend on what other programs, if any, are invoked to service each client. For example, if the server that `accept`s the client's connection calls `fork` and `exec`, it can be faster to `fork` a single threaded process than to `fork` a multithreaded process.

Exercises

27.1 In Figure 27.14 why does the parent keep the listening socket open, instead of closing it after all the children are created?

27.2 Can you recode the server in Section 27.9 to use a Unix domain datagram socket instead of a Unix domain stream socket? What changes?

27.3 Run the client and as many of the servers as your environment supports, and compare your results with those reported in this chapter.

Part 4

XTI: X/Open Transport Interface

28

XTI: TCP Clients

28.1 Introduction

Figure 1.15 showed that the sockets API was introduced in 1983 with 4.2BSD and initially worked with the TCP/IP protocol suite and the Unix domain protocols. During the mid-1980s, before Posix.1 was complete, there was still a rift within the Unix community between "Berkeley Unix" and "AT&T Unix." In the networking world, the claim was that TCP/IP would be replaced "shortly" with the OSI protocols.

In 1986 AT&T introduced a different networking API called TLI (Transport Layer Interface) with Release 3.0 of System V (SVR3). Although there are numerous similarities between TLI and sockets, TLI was modeled after the OSI Transport Service Definition. SVR3 also provided the first commercial release of the streams subsystem, which we say more about in Chapter 33. Unfortunately SVR3 did not include any networking protocols such as TCP/IP: it included only the streams and TLI building blocks. This led to a few companies providing third-party networking protocols for System V, usually TCP/IP and some preliminary implementations of the OSI protocols. System V Release 4 (SVR4) in 1990 finally provided the TCP/IP protocols as part of the basic operating system.

We mentioned X/Open in Section 1.10. In 1988 they released a modification of TLI called XTI: the *X/Open Transport Interface*. XTI is basically a superset of TLI and has gone through several versions. We describe XTI instead of TLI in this text because the Posix.1g standard started with XTI, not TLI. What we describe in the following chapters is XTI as specified for Unix 98 [Open Group 1997], which is nearly identical to the Posix.1g XTI.

XTI uses the term *communications provider* to describe the protocol implementation. The commonly available communications providers are for the Internet protocols, that is, TCP and UDP. The term *communications endpoint* refers to an object that is created

and maintained by a communications provider and then used by an application. These endpoints are referred to by file descriptors. We will often shorten these two terms to just *provider* and *endpoint*.

TLI referred to these as the *transport provider* and the *transport endpoint*.

All the XTI functions begin with `t_`. The header that the application includes to obtain all the XTI definitions is `<xti.h>`. Some Internet-specific definitions are obtained by including `<xti_inet.h>`.

We discuss XTI in the following order:

- TCP clients,
- name and address functions,
- TCP servers,
- UDP clients and servers,
- options,
- streams, and
- additional functions.

Our discussion of XTI is shorter than our discussion of sockets because the network programming *techniques* are the same. What changes are the function names, function arguments, and some of the nitty-gritty details (e.g., accepting TCP connections), but there is no need to duplicate every one of the sockets examples using XTI.

28.2 `t_open` Function

The first step in establishing a communications endpoint is to open the Unix device that identifies the particular communications provider. This function returns a descriptor (a small integer) that is used by the other XTI functions.

```
#include <xti.h>
#include <fcntl.h>

int t_open(const char *pathname, int oflag, struct t_info *info);
```

Returns: 0 if OK, –1 on error

The actual *pathname* to use depends on the implementation, but typical values for TCP/IP endpoints are `/dev/tcp`, `/dev/udp`, or `/dev/icmp`. Typical values for loopback endpoints are `/dev/ticots`, `/dev/ticotsord`, and `/dev/ticlts`.

The *oflag* argument specifies the open flags. Its value is `O_RDWR`. For a nonblocking endpoint the flag `O_NONBLOCK` is logically ORed with `O_RDWR`.

This XTI function is similar to the `socket` function. Both return a file descriptor that is associated with a user-specified protocol.

The `t_info` structure is a collection of integer values that describe the protocol-dependent features of the provider. This structure is returned through the *info* pointer,

if this argument is not a null pointer. This is our first encounter with one of the XTI structures that begins with t_. There are seven of these structures, which we say more about in Section 28.4.

```
struct t_info {
  t_scalar_t  addr;     /* max #bytes of communications protocol address */
  t_scalar_t  options;  /* max #bytes of protocol-specific options */
  t_scalar_t  tsdu;     /* max #bytes of transport service data unit (TSDU) */
  t_scalar_t  etsdu;    /* max #bytes of expedited TSDU (ETSDU) */
  t_scalar_t  connect;  /* max #bytes of data on conn. establishment */
  t_scalar_t  discon;   /* max #bytes of data on t_XXXdis() & t_XXXreldata() */
  t_scalar_t  servtype; /* service type supported */
  t_scalar_t  flags;    /* other information (new with XTI) */
};
```

This is our first encounter with the t_scalar_t datatype, which is new with Unix 98. Older implementations use a long integer for all these members, but this presents a problem on 64-bit architectures as we discussed in Section 1.11. t_scalar_t and t_uscalar_t are therefore defined to be int32_t and uint32_t, respectively.

Before describing each of the members of the t_info structure, we show some typical values for TCP and UDP, in Figures 28.1 and 28.2, which we explain shortly.

	AIX 4.2	DUnix 4.0B	HP-UX 10.30	Solaris 2.6	UnixWare 2.1.2
addr	16	16	16	16	16
options	512	4096	1024	504	360
tsdu	0	0	0	0	0
etsdu	−1	−1	−1	−1	−1
connect	−2	−2	−2	−2	−2
discon	−2	−2	−2	−2	−2
servtype	T_COTS_ORD	T_COTS_ORD	T_COTS_ORD	T_COTS_ORD	T_COTS_ORD

Figure 28.1 t_info values for TCP.

	AIX 4.2	DUnix 4.0B	HP-UX 10.30	Solaris 2.6	UnixWare 2.1.2
addr	16	16	16	16	16
options	512	768	256	468	328
tsdu	8192	9216	65508	65508	65508
etsdu	−2	−2	−2	−2	−2
connect	−2	−2	−2	−2	−2
discon	−2	−2	−2	−2	−2
servtype	T_CLTS	T_CLTS	T_CLTS	T_CLTS	T_CLTS

Figure 28.2 t_info values for UDP.

We are interested in three cases for each of the first six variables in the t_info structure: ≥0, −1 (also called T_INFINITE), and −2 (also called T_INVALID).

addr This specifies the maximum size in bytes of a protocol-specific address. A value of −1 indicates there is no limit to the size. A value of −2 indicates there is no user access to the protocol addresses.

The value of 16 shown for TCP and UDP is the size of a `sockaddr_in` structure. For an IPv6 endpoint this value will probably be the size of a `sockaddr_in6` structure.

`options` This specifies the size in bytes of the protocol-specific options. A value of –1 indicates there is no limit to the size. A value of –2 indicates there is no user access to the options. We talk more about XTI options in Chapter 32.

As we can see in the examples, there is little commonality amongst the various implementations, with the size ranging from 256 to 1024 bytes.

`tsdu` TSDU stands for "transport service data unit." This variable specifies the maximum size in bytes of a record whose boundaries are preserved from one endpoint to the other. A value of zero indicates that the communications provider does not support the concept of a TSDU, although it supports a byte stream of data (i.e., no record boundaries). A value of –1 indicates there is no limit to the size. A value of –2 indicates that the transport of normal data is not supported (a rare condition).

For TCP the value is always 0, since TCP provides a byte-stream service without any record boundaries. The predominant value for UDP is 65508, which is wrong. The maximum size of an IP datagram is 65535 bytes (the 16-bit total length field in Figure A.1), so the maximum size of a UDP datagram is 65535 minus 20 (for the IP header) minus 8 (for the UDP header), or 65507.

`etsdu` ETSDU stands for "expedited transport service data unit" and this variable specifies the maximum size in bytes of an ETSDU. This is what we called out-of-band data in Chapter 21. A value of zero indicates that the communications provider does not support the concept of ETSDU, although it supports a byte stream of out-of-band data (i.e., record boundaries are not preserved in the out-of-band data). A value of –1 indicates there is no limit to the size. A value of –2 indicates that the transport of expedited data is not supported.

As we expect, UDP does not support any form of out-of-band data. TCP supports the concept, but there is no limit to the amount of out-of-band data that the application can send. (Recall our discussion of TCP's urgent mode in Section 21.2.)

`connect` Some connection-oriented protocols support the transfer of user data along with a connection request. This variable specifies the maximum amount of this data. A value of –1 indicates there is no limit to the size. A value of –2 indicates that the communications provider does not support this feature.

TCP does not support this feature, so its value is always –2, and since UDP is not a connection-oriented protocol, its value is also –2. The connection-oriented OSI transport layer supports this feature.

Note that TCP allows sending data with a SYN, as described on pp. 14–16 of TCPv3. Sockets and XTI, however, provide no way to cause TCP to send data with a SYN.

Nevertheless, what this member of the t_info structure is referring to is something different (e.g., the capability provided by the OSI transport layer).

discon Some connection-oriented protocols support the transfer of user data along with a disconnection request. We will see the possibility of this when we discuss the t_snddis and t_rcvdis functions later in this chapter. This variable specifies the maximum amount of this data. A value of −1 indicates there is no limit to the size. A value of −2 indicates that the communications provider does not support this feature. This variable also specifies the amount of user data that can be sent with an orderly release, using the t_sndreldata and t_rcvreldata functions, which we describe in Section 34.10.

TCP does not support this feature, but it is supported by the OSI transport layer.

servtype This specifies the type of service provided by the communications provider. There are three possibilities which we show in Figure 28.3.

servtype	Description
T_COTS	connection-oriented service, without orderly release
T_COTS_ORD	connection-oriented service, with orderly release
T_CLTS	connectionless service

Figure 28.3 Types of service provided by communications providers.

TCP is connection oriented with orderly release and UDP is connectionless.

flags This member, which is new with XTI, specifies additional flags for the communications provider. The two constants shown in Figure 28.4 are defined by including the <xti.h> header that can be returned in this member.

flag	Description
T_SENDZERO	provider supports 0-length writes
T_ORDRELDATA	provider supports orderly release data (t_sndreldata and t_rcvreldata)

Figure 28.4 Values for flags member of t_info structure.

TCP does not support 0-length writes, but UDP does (resulting in a 28-byte IP datagram, with just an IP header and a UDP header, but no data). TCP does not support the T_ORDRELDATA flag either.

28.3 t_error and t_strerror Functions

Recall that most of the socket functions (e.g., socket, bind, connect, and so on) return −1 when they encounter an error and set the variable errno to provide

additional information about the error. The XTI functions normally return –1 on an error and set the variable `t_errno` to provide additional information about the error. (Recall our discussion of `errno` in Section 23.1 and how it is a per-thread variable. In a threads environment `t_errno` must also be a per-thread variable.) `t_errno` is similar to `errno` in that it is set only when an error occurs and it is not cleared on successful calls.

All the XTI error codes are defined as a result of including `<xti.h>` and begin with `T`, as in `TBADADDR` (incorrect address format), `TBADF` (illegal transport descriptor), and so on.

One special error value is `TSYSERR` and when it is returned in `t_errno`, it tells the application to look at the value in `errno` for the system error indication.

The two functions `t_error` and `t_strerror` are provided to help format error messages resulting from XTI functions.

```
#include <xti.h>

int t_error(const char *msg);
                                                        Returns: 0

const char *t_strerror(int errnum);
                                                Returns: pointer to message
```

`t_error` produces a message on the standard error output. This message consists of the string pointed to by *msg* (assuming this pointer is nonnull) followed by a colon and a space, followed by a message string corresponding to the current value of `t_errno`. If `t_errno` equals `TSYSERR`, then a message string is also output corresponding to the current value of `errno`. Finally a newline is output.

`t_strerror` returns a string describing the value of *errnum*, which is assumed to be one of the possible `t_errno` values. Unlike `t_error`, `t_strerror` does nothing special if this value is `TSYSERR`.

The program in Figure 28.5 shows the use of these two XTI error functions, along with our `err_xti` function. (We describe the latter in Section D.4.)

xtiintro/strerror.c

```
 1 #include    "unpxti.h"
 2 int
 3 main(int argc, char **argv)
 4 {
 5     printf("%s\n", t_strerror(TPROTO));
 6     errno = ETIMEDOUT;
 7     printf("%s\n", t_strerror(TSYSERR));
 8     t_errno = TSYSERR;
 9     errno = ETIMEDOUT;
10     t_error("t_error says");
```

```
11      t_errno = TSYSERR;
12      errno = ETIMEDOUT;
13      err_xti("err_xti says");
14      exit(0);
15  }
```

xtiintro/strerror.c

Figure 28.5 Example of `t_error` and `t_strerror` functions.

The output from this program is

```
aix % strerror
XTI protocol error
system error
t_error says: system error, Connection timed out
err_xti says: system error: Connection timed out
```

28.4 `netbuf` Structures and XTI Structures

XTI defines seven structures that are used to pass information between the application and the XTI functions. One of these, the `t_info` structure that we described in Section 28.2, is just a collection of integer values that describe protocol-dependent features of the provider. The remaining six structures each contain between one and three `netbuf` structures. The `netbuf` structure defines a buffer that is used to pass data from the application to the XTI function or vice versa.

```
struct netbuf {
  unsigned int  maxlen;  /* maximum size of buf */
  unsigned int  len;     /* actual amount of data in buf */
  void         *buf;     /* data (char* before Posix.1g) */
};
```

Figure 28.6 shows the six XTI structures that contain one or more `netbuf` structures, and the various other members of the XTI structure.

Datatype	XTI structure					
	t_bind	t_call	t_discon	t_optmgmt	t_uderr	t_unitdata
struct netbuf	addr	addr			addr	addr
struct netbuf		opt		opt	opt	opt
struct netbuf		udata	udata			udata
t_scalar_t					error	
t_scalar_t				flags		
unsigned int	qlen					
int			reason			
int		sequence	sequence			

Figure 28.6 Six XTI structures and their members.

These six XTI structures that contain the `netbuf` structures are always passed by reference between the application and the XTI function. That is, we pass the address of

the XTI structure as an argument to an XTI function. Therefore the XTI function can always read and update any of the three members of the `netbuf` structure (although none of the functions change the `maxlen` member).

The use of the three members of the `netbuf` structure depends on which direction the structure is being passed: from the application to the XTI function, or vice versa, as shown in Figure 28.7. We also note whether the XTI function reads the value of the member or writes the value of the member.

Member	Data from application to XTI	Data from XTI to application
`maxlen`	Ignored.	Read-only. Size of buffer pointed to by `buf`. XTI function will not store more than this amount of data in `buf`. If 0, then nothing is returned and `len` and `buf` are ignored.
`len`	Read-only. Application sets this to amount of data pointed to by `buf`.	Write-only. XTI function sets this member to the actual amount of data stored in `buf`, and this value will always be less than or equal to the value of `maxlen`.
`buf`	Pointer to data stored by application and then processed by XTI function.	Pointer to data stored by XTI function and then processed by application.

Figure 28.7 Processing of three members of `netbuf` structure.

If XTI has more data to return than `maxlen` allows, the XTI call fails with `t_errno` set to `TBUFOVFLW`.

Since the address of the `netbuf` structure is always passed to an XTI function, and since the structure contains both the size of the buffer (`maxlen`) and the amount of data actually stored in the buffer (`len`), there is no need in XTI for all the value–result arguments used with sockets.

28.5 `t_bind` Function

This function assigns the local address to an endpoint and activates the endpoint. In the case of TCP or UDP the local address is an IP address and a port.

```
#include <xti.h>

int t_bind(int fd, const struct t_bind *request, struct t_bind *return);
```

Returns: 0 if OK, –1 on error

The second and third arguments point to `t_bind` structures:

```
struct t_bind {
  struct netbuf  addr;    /* protocol-specific address */
  unsigned int   qlen;    /* max# of outstanding connections (if server) */
};
```

The endpoint is specified by *fd*. There are three cases to consider for the *request* argument.

request == NULL
: The caller does not care what local address gets assigned to the endpoint. The provider selects an address. The value of the qlen element is assumed to be zero (see below).

request != NULL, but *request->addr.len* == 0
: The caller does not care what local address gets assigned to the endpoint, and again the provider selects an address. Unlike the previous case, however, the caller can now specify a nonzero value for the qlen member of the *request* structure.

request != NULL, and *request->addr.len* > 0
: The caller specifies a local address for the communications provider to assign to the communications endpoint.

Whether the application specifies the address or whether the provider selects an address, the provider returns the address that it assigns to the endpoint in the *return* structure. If the *return* argument is a null pointer, the provider does not return the actual address.

The value of qlen has meaning only for a connection-oriented server: it specifies the maximum number of connections to queue for this endpoint. It is possible for this value to be changed by the provider, in which case the qlen element of the *return* structure indicates the actual value supported by the provider. We say more about this value and measure the actual number of connections queued for various values of qlen with Figure 30.14.

Notice that the addr member of the t_bind structure is an actual netbuf structure, and not a pointer to one of these structures. We will see that this is common to these XTI structures: most contain one or more netbuf structures within the t_*XXX* structure.

If XTI cannot bind the requested address, the error TADDRBUSY is returned. If TLI encountered this problem, it could bind another local address to the endpoint, requiring the caller to then compare the assigned address to the requested address.

The XTI method for the caller telling the provider to select an appropriate address is more generic than the method used by bind. For example, with TCP and UDP over IPv4 we must specify an Internet address of INADDR_ANY and a port of zero for the provider to select the local address. This is IPv4-specific and not generic to bind.

The qlen value corresponds to the backlog argument specified to listen. For a connection-oriented server, the t_bind function does the same work as the bind and listen functions.

A connection-oriented XTI client must call t_bind before calling t_connect (which we describe next). This differs from connect, which calls bind internally, if the socket has not been bound.

28.6 t_connect Function

A connection-oriented client initiates a connection with a server by calling t_connect. The client specifies the server's protocol address (e.g., IP address and port for a TCP server).

```
#include <xti.h>

int t_connect(int fd, const struct t_call *sendcall, struct t_call *recvcall);
```

Returns: 0 if OK, –1 on error

The second and third arguments point to a t_call structure:

```
struct t_call {
  struct netbuf  addr;      /* protocol-specific address */
  struct netbuf  opt;       /* protocol-specific options */
  struct netbuf  udata;     /* user data to accompany connection request */
  int            sequence;  /* for t_listen() & t_accept() functions */
};
```

The t_call structure pointed to by the *sendcall* argument specifies the information needed by the transport provider to establish the connection: the addr structure specifies the server's address, opt specifies any protocol-specific options desired by the caller, and udata contains any user data to be transferred to the server during connection establishment. (Recall from Figure 28.1 that TCP does not support any user data being sent with the connection request.) The sequence member has no significance for this function but is used when this structure is used with the t_accept function.

On return from this function, the t_call structure pointed to by the *recvcall* argument contains information associated with the connection that is returned by the communications provider to the caller: the addr structure contains the address of the peer process, opt contains any protocol-dependent optional data associated with the connection, and udata contains any user data returned by the peer's transport provider during connection establishment. Again, the sequence member has no meaning.

The contents of the opt structure are protocol dependent. The caller can set the len field of this structure to 0, telling the communications provider to use default values for any connection options. We talk more about XTI options in Chapter 32.

The caller can specify a null pointer for the *recvcall* argument, if the return information about the connection is not desired.

By default, this function does not return until the connection is completed, or an error occurs. We discuss how to perform a nonblocking connect in Section 34.3.

We saw in Section 4.3 that common errors when establishing a TCP connection are receiving an RST, receiving an ICMP destination unreachable, and timing out. Unfortunately, when one of these common errors occurs, t_connect returns –1, but t_errno is set to TLOOK, requiring more code to determine the exact reason. We discuss this problem in Sections 28.9 and 28.10 and show an example in Figure 28.13.

The t_connect function is similar to the connect function.

28.7 t_rcv and t_snd Functions

By default, XTI applications cannot call the normal `read` and `write` functions (unless the `tirdwr` module is pushed onto the stream, as we describe in Section 28.12). Instead XTI applications must call `t_rcv` and `t_snd`.

```
#include <xti.h>

int t_rcv(int fd, void *buff, unsigned int nbytes, int *flagsp);

int t_snd(int fd, const void *buff, unsigned int nbytes, int flags);
```

Both return: number of bytes read or written if OK, –1 on error

The first three arguments are similar to the first three arguments to `read` and `write`: descriptor, buffer pointer, and number of bytes to read or write.

> The input and output functions in the sockets API all use `size_t` for the buffer size, and `ssize_t` for the return value. The XTI functions use `unsigned int` and `int`.

The *flags* argument to `t_snd` is either zero, or some combination of the constants shown in Figure 28.8.

flag	Description
`T_EXPEDITED`	send or receive expedited (out-of-band) data
`T_MORE`	there is more data to send or receive

Figure 28.8 *flags* for `t_rcv` and `t_snd`.

`T_EXPEDITED` is used with `t_snd` to send out-of-band data (Section 34.12). This flag is set on return from `t_rcv` when out-of-band data is received.

`T_MORE` is provided so that multiple `t_rcv` or `t_snd` function calls can read or write what the protocol considers a logical record. This feature applies only to those protocols that support the concept of records. We show an example of this flag with the `t_rcvudata` function and the record-oriented UDP protocol in Figure 31.7. This flag is also used with TCP when reading out-of-band data, as we describe in Section 34.12, but is never used with normal TCP data.

> XTI defines a `T_PUSH` flag that tells the provider to send all the data that is has accumulated but not yet sent. This flag is used with XTI over SNA (IBM's Systems Network Architecture) but should not be used with TCP, and more specifically does *not* cause TCP's PUSH flag to be set.

Note that the *flags* argument to `t_snd` is an integer value, while the corresponding argument for `t_rcv` is a pointer to an integer. But the value pointed to by *flagsp* for `t_rcv` is not a true value–result argument, because its value is not examined by the function; it is set only on return.

Both of these functions return the actual number of bytes read or written. The return value from `t_snd` can be less than *nbytes* if the endpoint is nonblocking or if a signal is caught by the process.

These two functions correspond to the `send` and `recv` functions. The XTI `T_EXPEDITED` flag corresponds to `MSG_OOB`, although with XTI we cannot specify this flag to `t_rcv`.

Recall with a TCP socket that the receipt of a FIN causes `read` to return 0 and the receipt of an RST causes `read` to return –1, with `errno` set to `ECONNRESET`. `t_rcv` behaves differently when either of these conditions occur on an XTI endpoint:

- When a TCP FIN is received for an XTI endpoint, `t_rcv` returns –1 with `t_errno` set to `TLOOK`. The XTI function `t_look` must then be called, and it returns `T_ORDREL`. This is called an *orderly release indication*.
- When a TCP RST is received for an XTI endpoint, `t_rcv` returns –1 with `t_errno` set to `TLOOK`. The XTI function `t_look` must then be called, and it returns `T_DISCONNECT`. This is called a *disconnect* or an *abortive release*.

We first discuss the `t_look` function, followed by the orderly release and abortive release functions.

28.8 `t_look` Function

Various *events* can occur for an XTI endpoint and these events can occur *asynchronously*. By that we mean that the application can be performing some task when an unrelated event occurs on the endpoint. Some events indicate an error condition (`T_UDERR`, an error in a previously sent datagram) while other events are not an error (`T_EXDATA`, the arrival of expedited data).

For example, assume the application calls `t_snd` to send data to the peer, but right before this something happens at the peer and the peer process sends an RST and terminates. This unexpected event (having received an RST when the application calls `t_snd`) is passed to the application by having `t_snd` return –1 with `t_errno` set to `TLOOK`. The application then calls `t_look` to determine what happened (e.g., which event occurred) on the endpoint. The event in this case will be `T_DISCONNECT`, the receipt of a disconnect (an RST).

```
#include <xti.h>

int t_look(int fd);
```

Returns: event (Figure 28.9) if OK, –1 on error

The integer value returned by this function corresponds to one of the nine events shown in Figure 28.9.

When an event occurs on an XTI endpoint, it is considered *outstanding* until it is *consumed*. Figure 28.10 shows which XTI functions consume the XTI events and also shows that two events are consumed by calling `t_look`.

Event	Description
T_CONNECT	connection confirmation received
T_DATA	normal data received
T_DISCONNECT	disconnect received
T_EXDATA	expedited data received
T_GODATA	flow control restrictions on normal data lifted
T_GOEXDATA	flow control restrictions on expedited data lifted
T_LISTEN	connect indication received
T_ORDREL	orderly release indication received
T_UDERR	error in previously sent datagram

Figure 28.9 Events for an XTI endpoint.

Event	Cleared by t_look?	Consuming function
T_CONNECT		t_connect, t_rcvconnect
T_DATA		t_rcv, t_rcvv, t_rcvudata, t_rcvvudata
T_DISCONNECT		t_rcvdis
T_EXDATA		t_rcv, t_rcvv
T_GODATA	yes	t_snd, t_sndv, t_sndudata, t_sndvudata
T_GOEXDATA	yes	t_snd, t_sndv
T_LISTEN		t_listen
T_ORDREL		t_rcvrel
T_UDERR		t_rcvuderr

Figure 28.10 XTI events and which functions consume the event.

For the t_connect function on a blocking endpoint (the default), the T_CONNECT event is handled by the function itself and not seen by the application. In the example we were considering (the receipt of a FIN when we call t_snd) this figure shows that we must call t_rcvrel to clear the event.

> At the beginning of this section we described how the receipt of an RST generates a T_DISCONNECT event for the endpoint. Until Unix 98 the receipt of a FIN would generate a T_ORDREL event for the endpoint. Unix 98 makes this optional.

28.9 t_sndrel and t_rcvrel Functions

XTI supports two ways of releasing a connection: an *orderly release* and an *abortive release*. The differences are that an abortive release does not guarantee the delivery of any outstanding data, while the orderly release guarantees this. All communications providers must support an abortive release, while the support of an orderly release is optional. Recall, however, from Figure 28.1 that TCP provides an orderly release.

We can send and receive an orderly release with the following functions.

```
#include <xti.h>

int t_sndrel(int fd);

int t_rcvrel(int fd);
```

Both return: 0 if OK, −1 on error

To understand the semantics of an orderly release, we must remember that a connection-oriented protocol is usually a full-duplex connection between the two processes. The data transfer in one direction is independent of the data being transferred in the other direction. Figure 28.11 shows one use of these functions with TCP, to take advantage of TCP's half-close.

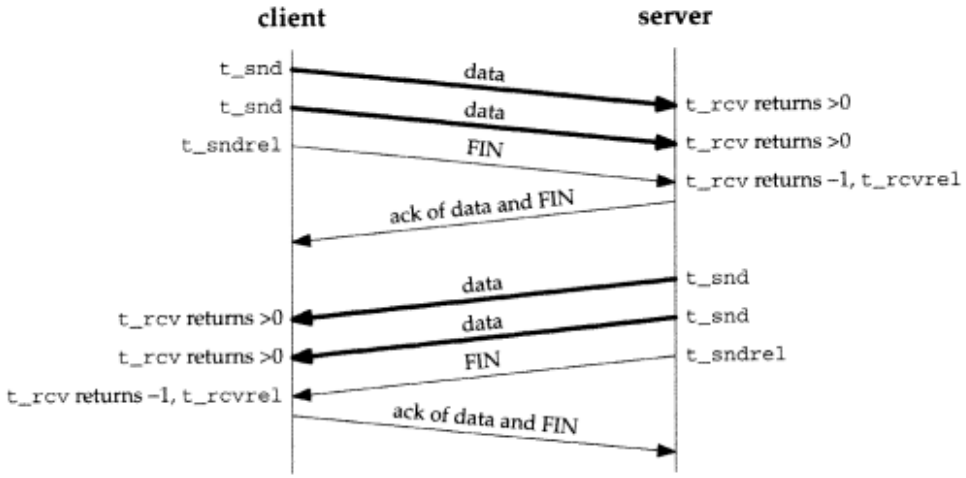


Figure 28.11 TCP's half-close using XTI.

A process issues an orderly release by calling `t_sndrel`. This tells the provider that the application has no more data to send on this endpoint. For a TCP endpoint, TCP sends a FIN to the peer (after any data that is already queued to be sent to the peer). The process that calls `t_sndrel` can continue to receive data, it can still read from the descriptor, but it can no longer write to the descriptor.

> This function performs the same action as `shutdown` with a second argument of `SHUT_WR` (1) on a TCP socket.

A process acknowledges the receipt of a connection release by calling the `t_rcvrel` function. This process can still write to the descriptor but it can no longer read from the descriptor.

> There is nothing in the sockets API comparable to `t_rcvrel`. The receipt of a FIN is delivered to the process as an end-of-file (e.g., `read` returns 0).
>
> This feature of XTI forces the application to deal with the full-duplex orderly release, even if the application is not interested in using this feature, as we will see in Figure 28.13.

28.10 t_snddis and t_rcvdis Functions

The following two functions handle an abortive release (a disconnect).

```
#include <xti.h>

int t_snddis(int fd, const struct t_call *call);

int t_rcvdis(int fd, struct t_discon *discon);
```

Both return: 0 if OK, −1 on error

The `t_snddis` function is used for two different purposes:

- to perform an abortive release of an existing connection, which in terms of TCP causes an RST to be sent, and
- to reject a connection request.

For an abortive release of an existing connection, the *call* argument can be a null pointer, in which case no information is sent to the peer process. Otherwise, the interpretation of the fields in the `t_call` structure is shown in Figure 28.12.

Member	Disconnection of existing connection	Rejection of new connection
addr	ignored	ignored
opt	ignored	ignored
udata	optional	optional
sequence	ignored	required

Figure 28.12 `t_call` structure used with `t_snddis`.

The optional `udata` member specifies user data to accompany the disconnection, but we saw in Figure 28.1 (the `discon` member of the `t_info` structure) that this is not supported by TCP.

> An abortive release is generated by a sockets application by setting the `SO_LINGER` socket option, setting `l_onoff` to a nonzero value and `l_linger` to 0, and then closing the socket (Chapter 7).

When a `T_DISCONNECT` event occurs on an XTI endpoint (e.g., an RST is received by TCP), the application must receive the abortive release by calling `t_rcvdis`. If the *discon* argument is a nonnull pointer, a `t_discon` structure is filled in with the reason for the abortive release.

```
struct t_discon {
  struct netbuf  udata;     /* user data */
  int            reason;    /* protocol-specific reason code */
  int            sequence;
};
```

The udata member contains the optional user data that accompanied the disconnect, reason is a protocol-dependent reason for the disconnect, and sequence is applicable only for servers that are receiving connections.

There is nothing in the sockets API comparable to t_rcvdis. The receipt of an RST is delivered to the process as an input error (e.g., read returns −1) with errno set to ECONNRESET. Writing to a socket that has received an RST generates SIGPIPE.

28.11 XTI TCP Daytime Client

We now recode our TCP daytime client from Figure 1.5 using XTI. Figure 28.13 shows the function.

xtiintro/daytimecli01.c

```
 1 #include    "unpxti.h"
 2 int
 3 main(int argc, char **argv)
 4 {
 5     int     tfd, n, flags;
 6     char    recvline[MAXLINE + 1];
 7     struct sockaddr_in servaddr;
 8     struct t_call tcall;
 9     struct t_discon tdiscon;
10     if (argc != 2)
11         err_quit("usage: daytimecli01 <IPaddress>");
12     tfd = T_open(XTI_TCP, O_RDWR, NULL);
13     T_bind(tfd, NULL, NULL);
14     bzero(&servaddr, sizeof(servaddr));
15     servaddr.sin_family = AF_INET;
16     servaddr.sin_port = htons(13);  /* daytime server */
17     Inet_pton(AF_INET, argv[1], &servaddr.sin_addr);
18     tcall.addr.maxlen = sizeof(servaddr);
19     tcall.addr.len = sizeof(servaddr);
20     tcall.addr.buf = &servaddr;
21     tcall.opt.len = 0;          /* no options with connect */
22     tcall.udata.len = 0;        /* no user data with connect */
23     if (t_connect(tfd, &tcall, NULL) < 0) {
24         if (t_errno == TLOOK) {
25             if ( (n = T_look(tfd)) == T_DISCONNECT) {
26                 tdiscon.udata.maxlen = 0;
27                 T_rcvdis(tfd, &tdiscon);
28                 errno = tdiscon.reason;
29                 err_sys("t_connect error");
30             } else
31                 err_quit("unexpected event after t_connect: %d", n);
32         } else
33             err_xti("t_connect error");
34     }
```

```
35      for ( ; ; ) {
36          if ( (n = t_rcv(tfd, recvline, MAXLINE, &flags)) < 0) {
37              if (t_errno == TLOOK) {
38                  if ( (n = T_look(tfd)) == T_ORDREL) {
39                      T_rcvrel(tfd);
40                      break;
41                  } else if (n == T_DISCONNECT) {
42                      tdiscon.udata.maxlen = 0;
43                      T_rcvdis(tfd, &tdiscon);
44                      errno = tdiscon.reason;     /* probably ECONNRESET */
45                      err_sys("server terminated prematurely");
46                  } else
47                      err_quit("unexpected event after t_rcv: %d", n);
48              } else
49                  err_xti("t_rcv error");
50          }
51          recvline[n] = 0;        /* null terminate */
52          fputs(recvline, stdout);
53      }
54      exit(0);
55  }
```

xtiintro/daytimecli01.c

Figure 28.13 Daytime client using XTI.

unpxti.h header

1 We define our own `unpxti.h` header that we `#include` in all our XTI programs. We show this header in Section D.3.

Create endpoint, bind any local address

12-13 `t_open` creates the XTI endpoint and we let the system choose its local protocol address by calling `t_bind` with a null second argument.

Specify server's address and port

14-22 We fill in an Internet socket address structure with the server's IP address and port, identical to Figure 1.5. We then fill in a `t_call` structure to point to this socket address structure and we also set the `len` members of the `opt` and `udata` structure to 0, indicating no options and no user data.

> There is nothing in XTI that requires the `t_call` structure to point to a `sockaddr_in` structure for IPv4. Nevertheless, almost all Unix implementations implement XTI with the Internet protocols using the `sockaddr_in` structure to pass the protocol address between the application and the provider. In protocol-independent code the use of this structure should be hidden from the application. We show how to do this in the next chapter.

Establish connection

23-34 `t_connect` establishes the connection, which in this case performs TCP's three-way handshake. As we mentioned earlier, if the connection establishment fails with one of the common errors, `t_connect` returns `TLOOK` and we then call `t_look` to find the event, calling `t_rcvdis` if the event is `T_DISCONNECT`. In this case we also store the reason for the disconnect in `errno` and call our `err_sys` function to print the appropriate error message.

Read from server, copy to standard output

35-53 We read the data from the server with `t_rcv`, printing the data on the standard output, until we hit the end of the connection.

Handle orderly release and disconnect

37-47 When `t_rcv` returns an error, if `t_errno` is `TLOOK` we call `t_look` to obtain the current event for the endpoint. If that event is `T_ORDREL`, we call `t_rcvrel`; otherwise if it is `T_DISCONNECT`, we call `t_rcvdis`.

If we run this program to the same set of hosts as in Section 4.3, we see the following output. First we connect to a host that is running the daytime server.

```
unixware % daytimecli01 206.62.226.35
Tue Feb  4 15:00:26 1997
```

Next we specify a nonexistent host on the local subnet.

```
unixware % daytimecli01 206.62.226.55
t_connect error: Connection timed out
```

Next we specify a router that is not running a daytime server, receiving an RST in response to our SYN.

```
unixware % daytimecli01 140.252.1.4
t_connect error: Connection refused
```

Finally we specify an IP address that is not connected to the Internet, receiving an ICMP host unreachable in response to our SYNs.

```
unixware % daytimecli01 192.3.4.5
t_connect error: No route to host
```

XTI and Sockets Interoperability

We note in the first example shown with our XTI daytime client, connecting to the server on the host `bsdi` (206.62.226.35), the server is written using sockets but our client is written using XTI. Nevertheless, the client communicates fine with the server. Similarly we could write a daytime server using XTI and it would communicate fine with our client from Figure 1.5 that uses sockets.

This interoperability is provided by the Internet protocol suite and has nothing to do with sockets or XTI. A client written using TCP or UDP interoperates with a server using the same transport protocol if the client and server speak the same *application protocol*, regardless of what API is used to write either the client or the server. It is the application protocol (e.g., HTTP, FTP, Telnet, and so on) and the transport layer (TCP or UDP) that determine the interoperability. The API we use to write either the client or server makes no difference.

28.12 xti_rdwr Function

As shown in the example in the previous section, by default we cannot use `read` and `write` on a descriptor that references an XTI endpoint. To see what happens, if we modify Figure 28.13 to use `read` and `write` instead of `t_rcv` and `t_snd`, copying the `for` loop from Figure 1.5, we encounter the following error after the connection is established:

```
unixware % daytimecli03 206.62.226.35
read error: Not a data message
```

We get the same results under AIX, but under HP-UX and Solaris the server's response is read and then an error results:

```
hpux % daytimecli03 198.69.10.4
Wed Apr  2 18:59:40 1997
read error: Bad message
```

We will explain the difference in these two scenarios when discussing Figure 33.11.

Fortunately there is often a way around this problem. If we have a streams-based implementation of XTI (which is common), we can push the streams module `tirdwr` onto the stream and then we can use `read` and `write`. Figure 33.3 shows the streams module involved. But since this capability is implementation dependent, instead of putting the actual `ioctl` command in our program, we define a simple function of our own. This way, if other implementations are encountered, only this library function needs to be changed.

```
#include "unpxti.h"

int xti_rdwr(int fd);
```

Returns: 0 if OK, –1 on error

In Figure 28.14 we show the trivial implementation of this function that just pushes the streams module `tirdwr` onto the stream.

libxti/xti_rdwr.c

```
1 #include    "unpxti.h"

2 int
3 xti_rdwr(int fd)
4 {
5     return (ioctl(fd, I_PUSH, "tirdwr"));
6 }
```

libxti/xti_rdwr.c

Figure 28.14 `xti_rdwr`: push the streams module `tirdwr` onto the stream.

There are a few caveats when using this feature.

- This streams module can be used only when the endpoint is in the data transfer phase. In our example in Figure 28.13, we place our call to `xti_rdwr` after `t_connect` returns.

- Once this module has been pushed onto the stream, none of the XTI functions (those beginning with `t_`) can be called.
- This module cannot be used by applications that use out-of-band data, as the arrival of out-of-band data cannot be handled by `read`.
- If an orderly release is received, it causes `read` to return 0 (i.e., end-of-file).
- Unfortunately, if a disconnect is received, it also causes `read` to return 0. This makes it impossible to distinguish between the (normal) receipt of a FIN and the (exceptional) receipt of an RST. The receipt of an RST also causes any future calls to `write` to fail.

Recall that with sockets the receipt of an RST causes `read` to return –1 of `ECONNRESET`.

28.13 Summary

XTI clients are similar to sockets clients, calling `t_open` instead of `socket`, and `t_connect` instead of `connect`. The same Internet socket address structure is used by both to specify the protocol address of the server, although with XTI this structure is described by another `netbuf` structure. By default `t_rcv` and `t_snd` are used by XTI instead of `read` and `write`, although the latter two can be used, depending on the environment, and we saw that in a streams environment a different streams module must be pushed onto the stream to use these two functions.

XTI defines nine events that can occur on an endpoint. When one of these occurs the XTI function returns an error of `TLOOK` and we must call the `t_look` function to determine what has happened and then handle it as desired. Handling these events often increases the amount of code that we must write, compared to the same scenario with sockets.

Exercises

28.1 In the second scenario for `t_bind` we mentioned that the *request* argument is nonnull but the `addr.len` member of that structure is 0, allowing the caller to specify a nonzero value for the `qlen` member. Is this scenario useful?

28.2 When we ran the program in Figure 28.13 to the unreachable host 192.3.4.5 we said that we received an ICMP host unreachable in response to our SYNs. Why is SYNs plural?

28.3 At the beginning of Section 28.8 we described the scenario where an application calls `t_snd` but an RST has been received on the connection. Compare this to how the sockets API handles a `write` when an RST has been received on the connection.

28.4 Write a function named `xti_read` that has the same arguments as `read` but calls `t_rcv` and handles the two scenarios from Figure 28.13: (a) if an orderly release is received, return 0, and (b) if a disconnect is received, return –1 with `errno` set to the reason.

29

XTI: Name and Address Functions

29.1 Introduction

XTI itself says nothing about name and address translation. The only functions required by Unix 98 in this area are the ones we covered in Chapter 9: `gethostbyname`, `gethostbyaddr`, `getservbyname`, and the like. Nevertheless, since many implementations of XTI are on SVR4-derived systems, most of these provide the name and address functions derived from SVR4: what we call the `netconfig` and `netdir` functions. These functions are called the "Network selection and name-to-address mapping facility" in SVR4.

In our client in Section 28.11 we filled in an Internet socket address structure with an IP address and a port number. This is protocol dependent and our goal in this chapter is to avoid knowing the contents of the `netbuf` structure, handling it instead as an opaque structure. We should start with a hostname and a service name, call some functions, and the end result should be a `netbuf` structure, ready for a call to `t_connect`, for example, in a TCP client. This is similar to our use of the `getaddrinfo` function in Section 11.2.

> One problem with the omission of the functions that we will describe from any standard is the lack of a definitive description as to how they operate. For example, most implementations of `netdir_getbyname` accept either a name or a decimal port number for a TCP or UDP service name, but other implementations accept only a name and not a port number.

29.2 /etc/netconfig File and netconfig Functions

The starting point for XTI name and address mapping is the /etc/netconfig file. This is a text file with one line for each supported protocol. Some typical values for the fields for each protocol are shown in Figure 29.1.

Network ID	Semantics	Flags	Protocol family	Protocol name	Device
tcp	tpi_cots_ord	v	inet	tcp	/dev/tcp
udp	tpi_clts	v	inet	udp	/dev/udp
icmp	tpi_raw	-	inet	icmp	/dev/icmp
rawip	tpi_raw	-	inet	-	/dev/rawip
ticlts	tpi_clts	v	loopback	-	/dev/ticlts
ticots	tpi_cots	v	loopback	-	/dev/ticots
ticotsord	tpi_cots_ord	v	loopback	-	/dev/ticotsord
spx	tpi_cots_ord	v	netware	spx	/dev/nspx2
ipx	tpi_clts	v	netware	ipx	/dev/ipx

Figure 29.1 Typical entries in the /etc/netconfig file.

There are actually seven fields for each line of the file but we do not show the final field, which specifies one or more libraries for directory lookups for that network. Typical values for this final field for the Internet protocols are /usr/lib/tcpip.so or /usr/lib/resolv.so. These are normally dynamically loadable libraries that provide the network-specific portion of the name-to-address translations.

The network IDs for the four Internet protocols are as we expect. The next three rows are loopback entries (ti stands for "transport independent"), and the final two rows are for the Novell Netware protocols (which we do not discuss in this text).

The values shown for the network semantics for the Internet protocols correspond to the service types shown in Figure 28.1, with the exception of tpi_raw, which is used for ICMP and raw IP. Note that TCP provides a connection-oriented service with an orderly release, as we saw in Figure 28.1.

The only flag currently defined is v, which means the entry is visible to the NETPATH library routines (described shortly).

The device name is used as the argument to t_open.

The network services library provides numerous functions to read the netconfig file. The function setnetconfig opens the file and the function getnetconfig then reads the next entry in the file. endnetconfig closes the file and releases any memory that was allocated.

> The term *network services library* is from System V and usually refers to the library that is specified to the linker as -lnsl. This library, say /usr/lib/libnsl.so, contains all the XTI library functions as well as the functions we are about to describe.

```
#include <netconfig.h>

void *setnetconfig(void);
```

Returns: nonnull pointer if OK, NULL on error

```
struct netconfig *getnetconfig(void *handle);
```

Returns: nonnull pointer if OK, NULL on end-of-file

```
int endnetconfig(void *handle);
```

Returns: 0 if OK, –1 on error

The pointer returned by `setnetconfig` (called a *handle*) is then used as the argument to the remaining two functions. Each entry in the file is returned as a `netconfig` structure:

```
struct netconfig {
  char           *nc_netid;     /* "tcp", "udp", etc. */
  unsigned long   nc_semantics; /* NC_TPI_CLTS, etc. */
  unsigned long   nc_flag;      /* NC_VISIBLE, etc. */
  char           *nc_protofmly; /* "inet", "loopback", etc. */
  char           *nc_proto;     /* "tcp", "udp", etc. */
  char           *nc_device;    /* device name for network id */
  unsigned long   nc_nlookups;  /* # of entries in nc_lookups */
  char          **nc_lookups;   /* list of lookup libraries */
  unsigned long   nc_unused[8];
};
```

The first six members in this structure correspond to the six columns in Figure 29.1. If we wrote a program that looked like the following outline

```
void  *handle;
struct netconfig  *nc;

handle = setnetconfig();
while ( (nc = getnetconfig(handle)) != NULL) {
    /* print netconfig structure */
}
endnetconfig(handle);
```

and assuming the `/etc/netconfig` file were as shown in Figure 29.1, nine `netconfig` structures would be printed, one per line of the figure, and in that order.

29.3 NETPATH Variable and netpath Functions

The `getnetconfig` function returns the next entry in the file, letting us go through the entire file, line by line. But for interactive programs (typically clients) we want the searching of the file limited only to the protocols that the user is interested in. This is done by allowing the user to set an environment variable named `NETPATH` and then using the following functions instead of the `netconfig` functions described in the previous section.

```
#include <netconfig.h>

void *setnetpath(void);
```

Returns: nonnull pointer if OK, NULL on error

```
struct netconfig *getnetpath(void *handle);
```

Returns: nonnull pointer if OK, NULL on end-of-file

```
int endnetpath(void *handle);
```

Returns: 0 if OK, −1 on error

For example, we could set the environment variable with the KornShell as

```
export NETPATH=udp:tcp
```

Using this setting, if we coded a program as shown in the following outline

```
void  *handle;
struct netconfig  *nc;

handle = setnetpath();
while ( (nc = getnetpath(handle)) != NULL) {
    /* print netconfig structure */
}
endnetpath(handle);
```

only two entries would be printed, one for UDP followed by one for TCP. The order of the two structures returned now corresponds to the order of the protocols in the environment variable, and not to the order in the `netconfig` file.

If the `NETPATH` environment variable is not set, all visible entries are returned, in their order in the `netconfig` file.

29.4 `netdir` Functions

The `netconfig` and `netpath` functions let us find a desired protocol. We also need to look up a hostname and a service name, based on the protocol that we choose with the `netconfig` or `netpath` functions. This is provided by the `netdir_getbyname` function.

```
#include <netdir.h>

int netdir_getbyname(const struct netconfig *ncp,
                     const struct nd_hostserv *hsp,
                     struct nd_addrlist **alpp);
```

Returns: 0 if OK, nonzero on error

```
void netdir_free(void *ptr, int type);
```

The first function converts a hostname and service name into an address. *ncp* points to a `netconfig` structure that was returned by `getnetconfig` or `getnetpath`. We must also fill in an `nd_hostserv` structure with the hostname and service name and pass a pointer to this structure as the second argument.

```
struct nd_hostserv {
  char  *h_host;          /* hostname */
  char  *h_serv;          /* service name */
};
```

The third argument points to a pointer to an `nd_addrlist` structure, and on success **alpp* contains a pointer to one of these structures:

```
struct nd_addrlist {
  int             n_cnt;   /* number of netbufs */
  struct netbuf  *n_addrs; /* array of netbufs containing the addrs */
};
```

Notice that this `nd_addrlist` structure points to an array of one or more `netbuf` structures, each of which contains one of the host's addresses. (Recall that a host can be multihomed.)

For example, using an example similar to Figure 11.1, where the hostname is `bsdi` (which has two IP addresses) and the service name is `domain` (TCP and UDP ports 53), then Figure 29.2 shows the information returned by `netdir_getbyname`, assuming that the `netconfig` structure used as the first argument to this function was for TCP.

We again assume that the format used by the provider to represent an Internet address is the `sockaddr_in` structure. While this is common, it is not required.

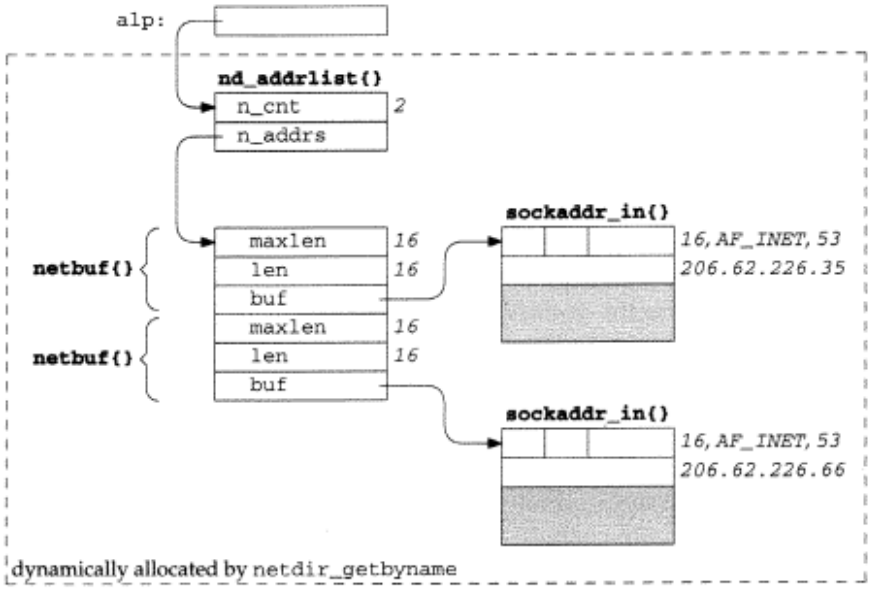


Figure 29.2 Data structures returned by `netdir_getbyname`.

The final argument to netdir_getbyname for this example would be a pointer to our alp variable.

When we have finished with these dynamically allocated structures, we call netdir_free with *ptr* pointing to the nd_addrlist structure, and *type* set to ND_ADDRLIST.

The reverse conversion—given a netbuf structure containing an address, return the hostname and service name—is provided by netdir_getbyaddr.

```
#include <netdir.h>

int netdir_getbyaddr(const struct netconfig *ncp,
                     struct nd_hostservlist **hslpp,
                     const struct netbuf *addr);
```

Returns: 0 if OK, nonzero on error

The first and third arguments provide the input: a pointer to a netconfig structure and a pointer to a netbuf structure. The result is a pointer to an nd_hostservlist structure, and this pointer is stored in **hslpp*.

```
struct nd_hostservlist {
  int                 h_cnt;       /* number of nd_hostservs */
  struct nd_hostserv  *h_hostservs; /* the hostname/service-name pairs */
};
```

This structure in turn points to an array of one or more nd_hostserv structures. The memory for the nd_hostservlist structure, the array of nd_hostserv structures that it points to, and the hostname and service name strings that this last structure points to are all dynamically allocated. This space is freed by calling netdir_free with a *type* of ND_HOSTSERVLIST.

29.5 t_alloc and t_free Functions

One of the requirements for protocol independence with an API is knowing the size of the protocol's addresses without having to know the exact format of the address. With sockets, this size is provided by the ai_addrlen member of the addrinfo structure that is returned by the getaddrinfo function (Section 11.2). With XTI, this size is provided by the addr member of the t_info structure that is returned by the t_open function.

Given this size, the next step is to dynamically allocate the required structures. With sockets we only had to worry about socket address structures and we just called malloc when necessary (e.g., Figure 27.5). But with XTI there are six structures (Figure 28.6), each of which contains one or more netbuf structures. These netbuf structures point to a buffer whose size depends on the size of the protocol address (such as the addr member of the t_call structure in Section 28.6). To simplify the dynamic allocation of these XTI structures and the netbuf structures that they contain, the t_alloc and t_free functions are provided.

```
#include <xti.h>

void *t_alloc(int fd, int structtype, int fields);
                                        Returns: nonnull pointer if OK, NULL on error

int t_free(void *ptr, int structtype);
                                        Returns: 0 if OK, –1 on error
```

The *structtype* argument specifies which of the seven XTI structures is to be allocated or freed and must be one of the constants shown in Figure 29.3.

The *fields* argument lets us specify that space for one or more `netbuf` structures should also be allocated and initialized appropriately. *fields* is the bitwise-OR of the constants shown in Figure 29.4. Recall from Figure 28.6 that the `netbuf` structure is always named `addr`, `opt`, or `udata`.

structtype	Type of structure
T_BIND	struct t_bind
T_CALL	struct t_call
T_DIS	struct t_discon
T_INFO	struct t_info
T_OPTMGMT	struct t_optmgmt
T_UDERROR	struct t_uderr
T_UNITDATA	struct t_unitdata

Figure 29.3 *structtype* argument for `t_alloc` and `t_free`.

fields	Allocate and initialize
T_ALL	all relevant fields of the given structure
T_ADDR	addr field of t_bind, t_call, t_uderr, or t_unitdata
T_OPT	opt field of t_optmgmt, t_call, t_uderr, or t_unitdata
T_UDATA	udata field of t_call, t_discon, or t_unitdata

Figure 29.4 *fields* argument for `t_alloc`.

The reason for these different values of the *fields* argument is that some of the XTI structures contain more than one `netbuf` structure, and we might not want to allocate space for all the buffers. For example, the `t_call` structure that we showed in Section 28.6, contains three `netbuf` structures.

```
struct t_call {
  struct netbuf  addr;      /* protocol-specific address */
  struct netbuf  opt;       /* protocol-specific options */
  struct netbuf  udata;     /* user data */
  int            sequence;  /* applies only to t_listen() func */
};
```

Specifying some combination of `T_ADDR`, `T_OPT`, and `T_UDATA` gives us complete control of the allocation. We will normally use `T_ALL` for our examples, because this is the simplest. (See also Exercise 29.2.)

Given the values in Figure 28.1 for AIX 4.2, if we call `t_alloc` with an *fd* that refers to a TCP endpoint, *structtype* of `T_CALL`, and *fields* of `T_ALL`, we get the picture shown in Figure 29.5 on return.

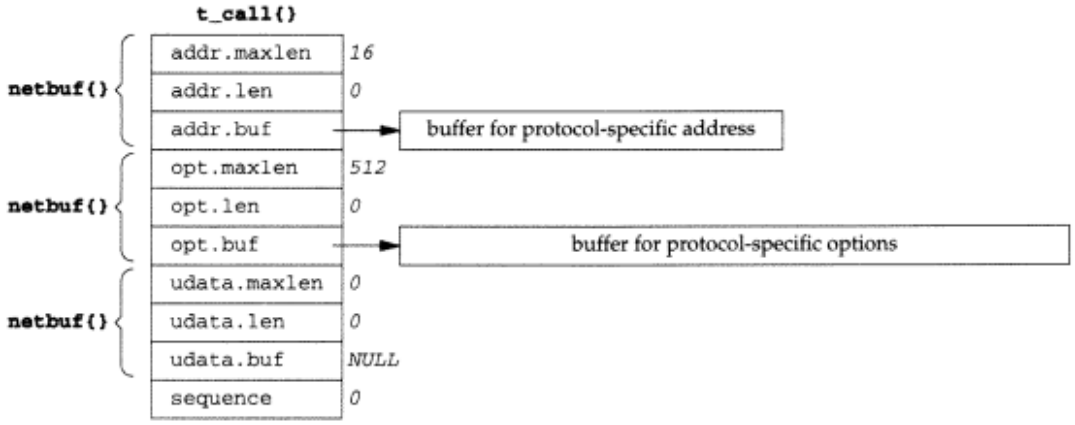


Figure 29.5 Allocation of structures and buffers by `t_alloc`.

This call to `t_alloc` allocates space for the `t_call` structure, which contains three `netbuf` structures. One buffer is allocated for the protocol-specific address (the `addr` member) and another for the protocol-specific options (the `opt` member). The two `buf` pointers are initialized along with the two `maxlen` members, and the two `len` members are set to 0. The third `netbuf` structure is not used for TCP (the user data to accompany the connection request), so the two lengths in the `udata` member are set to 0 and the buffer pointer is set to a null pointer.

The `t_free` function frees a structure that was previously allocated by `t_alloc`. The *structtype* argument specifies the type of the structure, and the constants shown in Figure 29.3 are used for this. `t_free` not only frees the memory that was allocated for the structure specified by *structtype*, but it also first checks any `netbuf` structures contained therein and releases the memory used by those buffers. In our example in Figure 29.5, `t_free` would first release the two buffers, and then the `t_call` structure.

29.6 `t_getprotaddr` Functions

The `t_getprotaddr` function returns both the local and foreign protocol addresses associated with an endpoint.

```
#include <xti.h>

int t_getprotaddr(int fd, struct t_bind *localaddr, struct t_bind *peeraddr);
```

Returns: 0 if OK, –1 on error

The addr member (a netbuf structure) of the two t_bind structures is used by this function. When the function is called, the maxlen and buf members of the netbuf structures specify where the result is to be stored. A maxlen of 0 indicates that the corresponding address should not be returned. On return the len members of the netbuf structures contain the size of the address that was stored in buf. This value will be 0 for the local address, if it has not yet been bound, and will be 0 for the peer address, if the endpoint has not yet been connected.

> TLI had an undocumented function named t_getname that could return both the local protocol address and the foreign protocol address.
>
> The t_getprotaddr function is a combination of both getsockname and getpeername.
>
> If we are interested in only one of the two addresses, we must still allocate a t_bind structure for the unwanted address and set its maxlen to 0. A simpler design would allow us to specify a null pointer for either of the two arguments when we do not want that address returned.

29.7 xti_ntop Function

We want a simple way to print an XTI protocol address (simpler than netdir_getbyaddr) so we write our own function xti_ntop to do this, similar to our sock_ntop function in Section 3.8. Most XTI implementations provide two functions named taddr2uaddr and uaddr2taddr. The term taddr refers to a *transport address*, contained in a netbuf structure, and the term uaddr refers to a *universal address*, a human-readable text string, stored as a null-terminated C string. These implementations print IPv4 universal addresses as six decimal numbers separated by five decimal points, with the first four numbers being the dotted-decimal IPv4 address and the final two numbers being the 2 bytes of the TCP or UDP port number.

The problem with these two functions, however, is that they require an argument to a netconfig structure, which gives information about the protocol whose address is being converted. But XTI addresses for IPv4 and IPv6 are self-defining. For example, when an IPv4 addresses is stored within a netbuf structure, the address is really a socket address structure and the first member is the address family, AF_INET. The length of this netbuf structure is 16 bytes (e.g., the addr row in Figures 28.1 and 28.2). We expect IPv6 addresses to be stored as sockaddr_in6 structures, with an address family of AF_INET6 and a length of 24 bytes. We are not guaranteed that all XTI addresses stored in netbuf structures are self-defining like this, but IPv4 and IPv6 addresses will be.

> Self-defining may be too strong a term. It is possible for some other protocol suite to use a 16-byte address whose first 2 bytes just happen to equal the constant AF_INET. But practically speaking, this should not be a problem.

We must then choose how to pass the protocol address to our function. Our first choice would be one of the XTI t_*XXX* structures. But for clients the protocol address of the server is in the addr member of a t_call structure (Section 28.6), for servers the

protocol address of the client is in the `addr` member of a `t_bind` structure (Section 30.2), and for any endpoint the local and foreign addresses returned by `t_getprotaddr` are in a `t_bind` structure. Since there is no consistency here (if we passed a pointer to one of these structures, we would also have to pass a flag indicating the type of structure), we will skip the XTI structures and pass a pointer to a `netbuf` structure to our function instead.

```
#include "unpxti.h"

char *xti_ntop(const struct netbuf *np);
```

Returns: nonnull pointer if OK, NULL on error

The argument is a pointer to a `netbuf` structure containing the address. The result is stored in static storage within the function. On success the return value is a pointer to the string containing the presentation format of the address.

We will use another function named `xti_ntop_host`, with the same calling sequence, that formats only the IP address, ignoring the port number.

These two functions are similar to the code shown in Figure 3.13. We do not show the source code, but it is freely available (see the Preface).

29.8 `tcp_connect` Function

We can now combine the `getnetpath` function, which returns information about one or more protocols with the `netdir_getbyname` function, which looks up a hostname and service and redo our `tcp_connect` function from Section 11.8 to use XTI instead of sockets and `getaddrinfo`. We show this function in Figure 29.6.

Initialize

13-15 `setnetpath` opens the `netconfig` file. The `nd_hostserv` structure is initialized with pointers to the hostname and service name.

Get next entry from `netconfig` file

16-18 `getnetpath` searches the `netconfig` file for the next protocol in the `NETPATH` variable. If the protocol is not TCP, we ignore the entry. Since we are looking for only the entry for TCP, we could call

```
ncp = getnetconfigent("tcp");
```

to locate just this entry. The call

```
freenetconfigent(ncp);
```

would then free the memory allocated by `getnetconfigent`. But since we would also like this code to work with IPv6, we go through the loop looking at each `netconfig` structure. Currently it is not known what the entry in the `netconfig` file will look like for TCP over IPv6, and how the XTI name functions will handle IPv6.

libxti/tcp_connect.c

```
 1 #include    "unpxti.h"

 2 int
 3 tcp_connect(const char *host, const char *serv)
 4 {
 5     int     tfd, i;
 6     void   *handle;
 7     struct t_call tcall;
 8     struct t_discon tdiscon;
 9     struct netconfig *ncp;
10     struct nd_hostserv hs;
11     struct nd_addrlist *alp;
12     struct netbuf *np;

13     handle = Setnetpath();

14     hs.h_host = (char *) host;
15     hs.h_serv = (char *) serv;

16     while ( (ncp = getnetpath(handle)) != NULL) {
17         if (strcmp(ncp->nc_proto, "tcp") != 0)
18             continue;

19         if (netdir_getbyname(ncp, &hs, &alp) != 0)
20             continue;

21             /* try each server address */
22         for (i = 0, np = alp->n_addrs; i < alp->n_cnt; i++, np++) {
23             tfd = T_open(ncp->nc_device, O_RDWR, NULL);

24             T_bind(tfd, NULL, NULL);

25             tcall.addr.len = np->len;
26             tcall.addr.buf = np->buf;   /* pointer copy */
27             tcall.opt.len = 0;  /* no options */
28             tcall.udata.len = 0;    /* no user data with connect */

29             if (t_connect(tfd, &tcall, NULL) == 0) {
30                 endnetpath(handle);     /* success, connected to server */
31                 netdir_free(alp, ND_ADDRLIST);
32                 return (tfd);
33             }
34             if (t_errno == TLOOK && t_look(tfd) == T_DISCONNECT) {
35                 t_rcvdis(tfd, &tdiscon);
36                 errno = tdiscon.reason;
37             }
38             t_close(tfd);
39         }
40         netdir_free(alp, ND_ADDRLIST);
41     }
42     endnetpath(handle);
43     return (-1);
44 }
```

libxti/tcp_connect.c

Figure 29.6 tcp_connect function for XTI

Search for hostname and service name

19-20 `netdir_getbyname` looks up the hostname and service name, using the `netconfig` structure returned by `getnetpath`.

Go through all server addresses

21-28 This loop tries each returned address for the server, calling `t_open`, `t_bind`, and `t_connect` for each address, until a connection is established, or until all the addresses have been tried. The `t_call` structure is initialized from the `netbuf` structure returned by `netdir_getbyname`.

Connection succeeds

29-33 If the connection succeeds, we clean up and return the connected descriptor. `endnetpath` frees the memory allocated for the `netconfig` structure and closes the `netconfig` file, and `netdir_free` frees all the memory starting with the `nd_addrlist` structure (Figure 29.2).

Handle `t_connect` errors

34-38 If `t_connect` fails, we check for `TLOOK` and call `t_rcvdis` if the connection was refused. We set `errno` to the protocol-dependent error code for the caller to examine. The endpoint is closed.

Finished with all addresses

40-41 After all the addresses have been tried, the `nd_addrlist` structure and the array of `netbuf` structures pointed to by it are freed by `netdir_free`. The `while` loop will keep going through the `netconfig` file, possibly returning additional protocols to try.

`getaddrinfo` combines the call to `getnetpath` with the testing for the correct protocol or semantics, with the call to `netdir_getbyname`.

An XTI endpoint that fails connection establishment can still be used in another call to `t_connect`. That is, we could move the calls to `t_open` and `t_bind` outside the `for` loop, calling these two functions once for each time through the `while` loop. Naturally, we would also move the call to `t_close` outside the `for` loop. With sockets, however, when a call to `connect` fails, the socket is no longer usable and must be closed (see Figure 11.6, for example).

But there is a subtle problem with this approach with XTI. The problem appears when the host to which we are trying to connect has multiple addresses, and the connection establishment fails. In this scenario the local port never changes and each call to `t_connect` for the next address is delayed by an exponential backoff from the previous call to `t_connect`, because all these connection establishments are from the same local endpoint. That is, if the first call to `t_connect` fails, the next call to `t_connect` might be delayed by 1 second, and if this fails the next call to `t_connect` might be delayed by 2 seconds, and so on. To avoid this problem we `t_close` the endpoint when `t_connect` fails and then create a new endpoint for the next call to `t_connect`.

Example

We now use our `tcp_connect` function and redo our protocol independent daytime client from Figure 11.7 using XTI instead of sockets. Our XTI version is in Figure 29.7.

xtiintro/daytimecli02.c

```
 1 #include    "unpxti.h"

 2 int
 3 main(int argc, char **argv)
 4 {
 5     int     tfd, n, flags;
 6     char    recvline[MAXLINE + 1];
 7     struct t_bind *bound, *peer;
 8     struct t_discon tdiscon;

 9     if (argc != 3)
10         err_quit("usage: daytimecli02 <hostname/IPaddress> <service/port#>");

11     tfd = Tcp_connect(argv[1], argv[2]);

12     bound = T_alloc(tfd, T_BIND, T_ALL);
13     peer = T_alloc(tfd, T_BIND, T_ALL);
14     T_getprotaddr(tfd, bound, peer);
15     printf("connected to %s\n", Xti_ntop(&peer->addr));

16     for ( ; ; ) {
17         if ( (n = t_rcv(tfd, recvline, MAXLINE, &flags)) < 0) {
18             if (t_errno == TLOOK) {
19                 if ( (n = T_look(tfd)) == T_ORDREL) {
20                     T_rcvrel(tfd);
21                     break;
22                 } else if (n == T_DISCONNECT) {
23                     T_rcvdis(tfd, &tdiscon);
24                     errno = tdiscon.reason;     /* probably ECONNRESET */
25                     err_sys("server terminated prematurely");
26                 } else
27                     err_quit("unexpected event after t_rcv: %d", n);
28             } else
29                 err_xti("t_rcv error");
30         }
31         recvline[n] = 0;        /* null terminate */
32         fputs(recvline, stdout);
33     }
34     exit(0);
35 }
```

xtiintro/daytimecli02.c

Figure 29.7 Protocol independent daytime client.

Establish connection

11 We call our `tcp_connect` function from Figure 29.6 to look up the hostname and service name and establish the connection.

Print peer's protocol address

12–15 We allocate two `t_bind` structures and call `t_getprotaddr` to obtain the local protocol address and the peer's protocol address. We print the peer's address by calling our `xti_ntop` function.

Read data from server until EOF

16-33 The reading of the data from the server is identical to the code in Section 28.11.

We can run the program as follows:

```
unixware % daytimecli02 aix daytime
connected to 206.62.226.43.13
Fri Feb  7 13:28:24 1997
```

29.9 Summary

In SVR4 implementations of XTI network selection is normally done using the `/etc/netconfig` file and the function `netdir_getbyname` then looks up the hostname and service name, returning an array of `netbuf` structures, one per address and service. This is similar to the `getaddrinfo` function in Chapter 11. The reverse mapping, from the protocol address to the presentation form, is done by `netdir_getbyaddr`, which is similar to `getnameinfo`.

Since so many structures are used by XTI, the seven `t_XXX` structures, and the `netbuf` structures contained therein, two functions are provided to dynamically allocate and free these structures: `t_alloc` and `t_free`.

Exercises

29.1 `getnetconfig` returns a pointer to a structure that it fills in, similar to `gethostbyname`. But we said the latter function was not thread-safe. Is `getnetconfig` thread-safe, and if so, how does it do this?

29.2 Write a program that calls `t_alloc` twice for a `t_call` structure for a TCP endpoint. The first time specify the third argument as `T_ALL` and the second time specify the third argument as `T_ADDR|T_OPT|T_UDATA`. What happens?

29.3 Why does `t_free` require a *structtype* argument?

29.4 In Figure 29.6 why don't we initialize the `nd_hostserv` structure as

```
struct nd_hostserv hs = { host, serv };
```

30

XTI: TCP Servers

30.1 Introduction

Undoubtedly the most confusing aspect of XTI is the handling of incoming connections by a connection-oriented server. With sockets we just call `accept` and all the details are handled by the kernel or by the sockets library. In the case of TCP, arriving SYNs from clients are placed onto an incomplete connection queue for that endpoint (Figure 4.6). When the three-way handshake completes, `accept` returns (Figure 2.5). If multiple connections are on the completed connection queue, they are returned in a FIFO order by `accept`. In the real world (Figure 4.9) the number of completed connections is normally 0 while the number of incomplete connections is often nonzero.

The *intent* of the XTI model (which is based on the design of the OSI transport service) is to allow the transport layer to tell the server process when a SYN arrives from a client (called a *connect indication*), passing the client's protocol address (IP address and port) to the server. The server process is then allowed to either accept or reject the connection request. The server's TCP, in this model, would not send its SYN/ACK or its RST until the server process tells it what to do. This model is shown in Figure 30.1.

Notice the server's function calls: the first call to `t_bind` (with a nonzero `qlen`) indicates that the endpoint will be accepting incoming connections, `t_listen` returns when the connection is "available" (we say more about this shortly), and the server must then call `t_open`, `t_bind`, and `t_accept` to accept the connection. LISTEN indicates the endpoint's TCP state (Figure 2.4).

When the server process receives notification of the connection request it can also choose not to accept the connection, calling `t_snddis` to reject the request. We show this in Figure 30.2. The *intent* here is that the server is notified when the SYN arrives (the connect indication) and chooses not to accept the connection (perhaps based on the client's IP address or port number, or on the user data sent with the connection request,

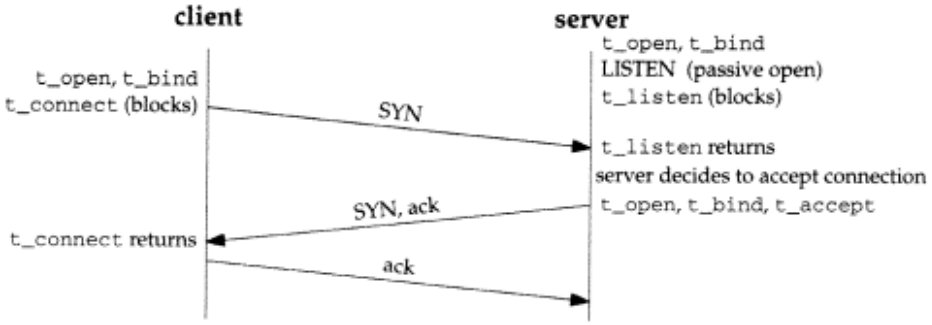


Figure 30.1 Intended model when XTI server accepts connection request.

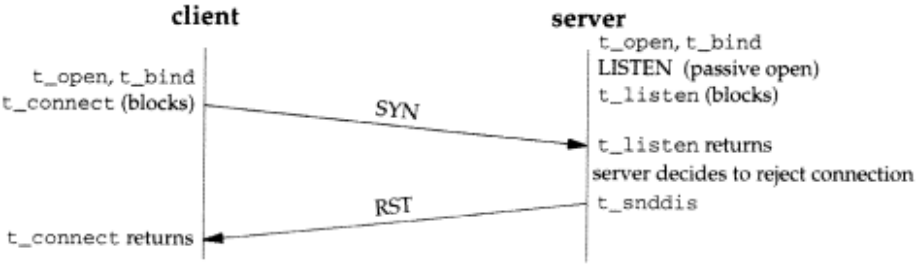


Figure 30.2 Intended model when XTI server rejects connection request.

if supported by the protocol). The application then calls `t_snddis`, causing an RST to be sent instead of completing the three-way handshake. This would make the client's call to `t_connect` return an error.

> Recall from our sockets discussion and the time line in Figure 2.5 that a sockets server can never cause a client's `connect` to fail, because `accept` returns when the three-way handshake completes, one-half of an RTT after `connect` returns. If a sockets server doesn't like a client (perhaps based on the client's IP address or port returned by `accept`), all the server can do is terminate the connection, either with a normal `close` (sending a FIN) or by first setting the `SO_LINGER` socket option and then calling `close` (sending an RST).

We have italicized the word *intent* when describing the XTI scenario because this is not what really happens. This scenario was the intent of the OSI protocols, but most existing TCP implementations automatically accept incoming connection requests (as long as the complete and incomplete connection queues are not full) and do not notify the server process until the three-way handshake is complete.

> The technique of notifying the application when the SYN arrives and then not completing the three-way handshake until the application indicates whether it wants to accept or reject the connection request is sometimes called a *lazy accept*. At least two historical implementations of TLI, from The Wollongong Group and Sequent Computer Systems, performed lazy accepts.

Both have changed to the de facto "standard" method of returning from t_listen when the three-way handshake completes. One reason for the change is that the lazy accept breaks most implementations of FTP.

Posix.1g also requires that when t_listen returns successfully for a TCP endpoint, this indicates a completed connection, and not a connect indication.

[Jacobson 1994] notes that 4.4BSD was supposed to provide a per-socket option to allow a lazy accept with the sockets API for TCP, but this was never implemented. 4.4BSD supports the lazy accept for the OSI protocols.

30.2 t_listen Function

The normal scenario for a connection-oriented XTI server is to call the following functions.

```
listenfd = t_open( ... );          /* create listening endpoint */
t_bind(listenfd, ... );            /* t_bind.qlen > 0 */

for ( ; ; ) {
    t_listen(listenfd, ... );      /* blocks awaiting connection */
    connfd = t_open( ... );        /* create new fd for connected endpoint */
    t_bind(connfd, NULL, NULL); /* any local addr */
    t_accept(listenfd, connfd, ... );  /* accept on new fd */

    ...          /* process connected endpoint */

    t_close(connfd);
}
```

t_listen is the function that normally blocks, waiting for a connection from a client.

```
#include <xti.h>

int t_listen(int fd, struct t_call *call);
```

Returns: 0 if OK, –1 on error

We described the t_call structure when we described the t_connect function but show it again here:

```
struct t_call {
  struct netbuf  addr;      /* protocol-specific address */
  struct netbuf  opt;       /* protocol-specific options */
  struct netbuf  udata;     /* user data to accompany connection request */
  int            sequence;  /* for t_listen() & t_accept() functions */
};
```

The structure returned through the *call* pointer contains relevant parameters of the connection: addr contains the protocol address of the client, opt contains any protocol-specific options, and udata contains any user data that was sent along with the connection request (which is not supported by TCP). The sequence variable contains a unique value that identifies this connection request. This value will be used when we call t_accept (or t_snddis) to identify which connection to accept (or reject).

Although this function appears similar to the `accept` function, it is different, as `t_listen` waits only for a connection to arrive; it does not accept the connection. To do that, the XTI user has to call the `t_accept` function.

Although *sequence* is an integer, some implementations store an address in this member. Do not assume it is a small integer like a descriptor.

30.3 `tcp_listen` Function

We now write our own function that creates a listening endpoint on which incoming connections can be accepted. The calling sequence is identical to the function of the same name shown in Figure 11.8.

Initialize

16-18 `setnetconfig` opens the `/etc/netconfig` file. If the `host` argument is a null pointer, we pass the special string `HOST_SELF` to `netdir_getbyname`. This causes the listening socket to be bound to the wildcard address (0.0.0.0 for IPv4).

Find matching protocol

19-22 We process each line of the `/etc/netconfig` file looking for the TCP protocol. Notice that we use `getnetconfig` for a server, while in Figure 29.6 we called `getnetpath` for a client. This is because a server should not assume that the `NETPATH` environment variable is set to anything meaningful, since the server might be started by an initialization script or from the command line. Clients, on the other hand, are normally started from an interactive shell on behalf of a user and can assume the variable might be set by the user.

Look up hostname and service name

23-27 `netdir_getbyname` looks up the hostname and service name, using the pointer to the `netconfig` structure for the desired protocol.

Open device

28-29 `t_open` opens the appropriate device (such as `/dev/tcp`) and we save a copy of this device name in the external `xti_serv_dev`. We do this because the caller of `tcp_listen` will need to call `t_open` again, once per connection, and needs this device name to maintain protocol independence. With the sockets version of `tcp_listen` we didn't need to save anything like this, because `accept` (not the process) automatically creates the new socket for each connection.

This technique is not thread-safe. This is an unfortunate side effect of requiring the application to maintain state information (the name of the device) between the call to `t_open` for the listening descriptor, and the later calls to `t_open` for each connected descriptor. One way to make this operation thread-safe is to call `strdup` to copy the device name into dynamically allocated storage and then return the pointer through another argument to `tcp_listen`, for the caller to `free`.

libxti/tcp_listen.c

```
 1 #include    "unpxti.h"
 2 #include    <limits.h>          /* PATH_MAX */

 3 char    xti_serv_dev[PATH_MAX + 1];

 4 int
 5 tcp_listen(const char *host, const char *serv, socklen_t *addrlenp)
 6 {
 7     int     listenfd;
 8     void   *handle;
 9     char   *ptr;
10     struct t_bind tbind;
11     struct t_info tinfo;
12     struct netconfig *ncp;
13     struct nd_hostserv hs;
14     struct nd_addrlist *alp;
15     struct netbuf *np;

16     handle = Setnetconfig();

17     hs.h_host = (host == NULL) ? HOST_SELF : (char *) host;
18     hs.h_serv = (char *) serv;

19     while ( (ncp = getnetconfig(handle)) != NULL &&
20             strcmp(ncp->nc_proto, "tcp") != 0) ;

21     if (ncp == NULL)
22         return (-1);

23     if (netdir_getbyname(ncp, &hs, &alp) != 0) {
24         endnetconfig(handle);
25         return (-2);
26     }
27     np = alp->n_addrs;          /* use first address */

28     listenfd = T_open(ncp->nc_device, O_RDWR, &tinfo);
29     strncpy(xti_serv_dev, ncp->nc_device, sizeof(xti_serv_dev));

30     tbind.addr = *np;           /* copy entire netbuf{} */
31         /* can override LISTENQ constant with environment variable */
32     if ( (ptr = getenv("LISTENQ")) != NULL)
33         tbind.qlen = atoi(ptr);
34     else
35         tbind.qlen = LISTENQ;
36     T_bind(listenfd, &tbind, NULL);

37     netdir_free(alp, ND_ADDRLIST);
38     endnetconfig(handle);

39     if (addrlenp)
40         *addrlenp = tinfo.addr; /* size of protocol addresses */
41     return (listenfd);
42 }
```

libxti/tcp_listen.c

Figure 30.3 XTI `tcp_listen` function: create listening endpoint.

Enter TCP's LISTEN state

30-36 We call `t_bind`, binding the address returned by `netdir_getbyname` to the endpoint. By setting the `qlen` member of the `t_bind` structure nonzero, this indicates a listening endpoint, and in the case of TCP the endpoint enters the LISTEN state. (We are talking about TCP's LISTEN state here. The XTI state is called `T_IDLE`.) Incoming connections will now be accepted by the transport provider. We let the environment variable `LISTENQ` override the default value of this constant from our `unp.h` header. We have similar code in our `Listen` wrapper function for the sockets `listen` function (Figure 4.8).

Free memory, return values

37-41 We call `netdir_free` and `endnetconfig` to free the allocated memory. We return the size of the protocol addresses (if requested) and the return value of the function is the listening endpoint.

Notice that we do not call `t_listen` as that is where the server blocks awaiting the incoming connection.

30.4 `t_accept` Function

Once the `t_listen` function indicates that a connection has arrived, we choose whether to accept the request or not. To accept the request the `t_accept` function is called.

```
#include <xti.h>

int t_accept(int listenfd, int connfd, struct t_call *call);
```

Returns: 0 if OK, –1 on error

listenfd specifies the endpoint where the connection arrived; that is, this is the endpoint that was the argument to `t_listen`. The *connfd* argument specifies the endpoint where the connection is to be established. Normally the server creates a new endpoint, *connfd*, to receive the connection.

The *call* argument identifies which connection is being accepted (in case multiple connections are pending, which we talk about shortly), and its value is whatever was returned by `t_listen`.

Notice that it is the server's responsibility to create the new endpoint for a server. This is usually done by calling `t_open` between the calls to `t_listen` and `t_accept`.

> We also have the option of specifying the same descriptor for *listenfd* and *connfd*; that is, we accept the new connection on the listening endpoint. But if we do this, no further connections are accepted by the provider until we have finished with this connection (e.g., this is an iterative server). It only makes sense in this scenario to set the `qlen` to one. Given the limitations of this scenario, and the need of most real-world servers to handle multiple connections at the same time, we won't show any examples of this.

30.5 xti_accept Function

We now write a simple function named xti_accept to perform the steps required to accept a connection using XTI. In the common case we should write something like the following for our XTI server applications:

```
listenfd = Tcp_listen( ... );  /* create listening endpoint */

for ( ; ; ) {
    connfd = Xti_accept(listenfd, ... );  /* block, then accept */

    ...    /* process connfd */

    t_close(connfd);
}
```

This is similar to the sockets code, just replacing accept with xti_accept.

```
#include "unpxti.h"

int xti_accept(int listenfd, struct netbuf *cliaddr, int rdwr);
```

Returns: nonnegative descriptor if OK, –1 on error

On success, the return value is the new connected descriptor, the client's address is returned in the netbuf structure pointed to by *cliaddr*, and if the *rdwr* argument is nonzero, our xti_rdwr function is called for the connected endpoint.

We show a simple version of our xti_accept function in Figure 30.4. We say "simple" because we will see shortly that it can fail when multiple connections are ready at the same time. We will fix this in Section 30.8.

Wait for connection

8–9 A t_call structure is allocated to hold the information about the client's connection. t_listen blocks, waiting for a connection.

Create new endpoint and bind any local address

10–12 A new endpoint is created by t_open, using the pathname that was saved in the external variable xti_serv_dev by tcp_listen. Any local address is bound to the endpoint. This call to t_bind is optional. If we do not bind something to the endpoint, it will be unbound when t_accept is called, and the communications provider will bind it automatically to some address that is appropriate.

Accept the connection

13 t_accept accepts the connection. t_accept knows which connection to accept by looking at the sequence member of the t_call structure, which was filled in by t_listen to identify this particular connection.

Allow read and write, if desired

14–15 If the caller specifies a nonzero *rdwr* argument, our xti_rdwr function pushes the tirdwr module onto the stream, allowing read and write to be used instead of t_rcv and t_snd.

libxti/xti_accept_simple.c

```
 1 #include    "unpxti.h"

 2 int
 3 xti_accept(int listenfd, struct netbuf *cliaddr, int rdwr)
 4 {
 5     int     connfd;
 6     u_int   n;
 7     struct t_call *tcallp;

 8     tcallp = T_alloc(listenfd, T_CALL, T_ALL);

 9     T_listen(listenfd, tcallp); /* blocks */

10         /* following assumes caller called tcp_listen() */
11     connfd = T_open(xti_serv_dev, O_RDWR, NULL);
12     T_bind(connfd, NULL, NULL);
13     T_accept(listenfd, connfd, tcallp);

14     if (rdwr)
15         Xti_rdwr(connfd);

16     if (cliaddr) {              /* return client's protocol address */
17         n = min(cliaddr->maxlen, tcallp->addr.len);
18         memcpy(cliaddr->buf, tcallp->addr.buf, n);
19         cliaddr->len = n;
20     }
21     T_free(tcallp, T_CALL);
22     return (connfd);
23 }
```

libxti/xti_accept_simple.c

Figure 30.4 Simple version of `xti_accept` function.

Return client's protocol address

16–20 The caller can specify a nonnull *cliaddr* argument that points to a `netbuf` structure. That structure must be initialized by the caller to point to a buffer in which the client's protocol address is returned. We ensure we do not overflow the caller's buffer and then set the `len` member to the size of the address that we return.

Clean up and return

21–22 The `t_call` structure is freed and the connected descriptor is returned.

30.6 Simple Daytime Server

We now rewrite our simple daytime server from Figure 11.10 using XTI, calling our `tcp_listen` and `xti_accept` functions.

Create endpoint

10–17 `tcp_listen` creates the listening endpoint. We allocate memory for the client's protocol address and initialize our `netbuf` structure for this purpose.

xtiintro/daytimesrv01.c

```
 1 #include    "unpxti.h"

 2 int
 3 main(int argc, char **argv)
 4 {
 5     int     listenfd, connfd;
 6     char    buff[MAXLINE];
 7     time_t  ticks;
 8     socklen_t addrlen;
 9     struct netbuf cliaddr;

10     if (argc == 2)
11         listenfd = Tcp_listen(NULL, argv[1], &addrlen);
12     else if (argc == 3)
13         listenfd = Tcp_listen(argv[1], argv[2], &addrlen);
14     else
15         err_quit("usage: daytimetcpsrv01 [ <host> ] <service or port>");
16     cliaddr.buf = Malloc(addrlen);
17     cliaddr.maxlen = addrlen;

18     for ( ; ; ) {
19         connfd = Xti_accept(listenfd, &cliaddr, 0);
20         printf("connection from %s\n", Xti_ntop(&cliaddr));

21         ticks = time(NULL);
22         snprintf(buff, sizeof(buff), "%.24s\r\n", ctime(&ticks));
23         T_snd(connfd, buff, strlen(buff), 0);

24         T_close(connfd);
25     }
26 }
```

xtiintro/daytimesrv01.c

Figure 30.5 Daytime server using XTI.

Wait for connection and accept it

19–20 Our `xti_accept` function waits for the connection, creates a new endpoint, returns the connected descriptor, and returns the client's IP address and port number. We print the client's protocol address using our `xti_ntop` function.

Generate daytime output

21–24 Calling `time` and then `ctime` generates the current time and date in a human-readable format, and `t_snd` sends this back to the client across the connection. The endpoint is closed with `t_close`.

Notice that we simply call `t_close` when have finished sending data. Since TCP provides an orderly release, this sends a FIN and goes through the normal four-packet connection termination sequence (Figure 2.5), but `t_close` returns immediately.

This is identical to calling `close` on a TCP socket.

If we want to wait until the peer TCP receives our data and sends its FIN, we must call `t_sndrel` to send our FIN and then wait for the FIN from the peer with `t_rcvrel`. We would replace the call to `T_close` at the end of Figure 30.5 with the following:

```
T_sndrel(connfd);
while ( (n = t_rcv(connfd, buff, MAXLINE, &flags)) >= 0)
    ;
if (t_errno == TLOOK) {
    if ( (n = T_look(connfd)) == T_ORDREL) {
        T_rcvrel(connfd);
    } else if (n == T_DISCONNECT) {
        T_rcvdis(connfd, NULL);
    } else
        err_quit("unexpected event after t_rcv: %d", n);
} else
    err_xti("t_rcv error");
T_close(connfd);
```

`t_sndrel` sends the FIN and we must then wait for an orderly release indication at which time we call `t_rcvrel`. To do this we call `t_rcv`, ignoring any data that may arrive.

> This scenario is similar to calling `shutdown` for a socket and then waiting until `read` returns an end-of-file (Figure 7.8).
>
> With XTI we can also cause a `t_close` or `close` to linger, if there is still data queued to send to the peer, instead of returning immediately. This is done by setting the `XTI_LINGER` option, which we describe in Section 32.3. This is similar to the `SO_LINGER` socket option.

30.7 Multiple Pending Connections

We have alluded to the complexity involved when multiple connections arrive at nearly the same time for a listening endpoint. To demonstrate this problem we return to our TCP server in Figure 27.5. We used this server to measure the process control time required for various types of servers. We can run the client that we wrote for this server (Figure 27.4) and specify the number of children to `fork`, establishing multiple connections with the server.

Figure 30.6 shows the server, which is just Figure 27.5 coded to use XTI instead of sockets.

10-22 We call our `tcp_listen` and `xti_accept` functions that we developed earlier in this chapter.

`SIGINT` handler

31-37 Our signal handler calls our internal `xti_accept_dump` function, and we use this to collect the counters shown in Figure 30.13. This function prints the `count` member of each `cli` structure (Figure 30.7).

xtiserver/serv01.c

```
 1 #include    "unpxti.h"

 2 int
 3 main(int argc, char **argv)
 4 {
 5     int     listenfd, connfd;
 6     pid_t   childpid;
 7     void    sig_chld(int), sig_int(int), web_child(int);
 8     socklen_t addrlen;
 9     struct netbuf cliaddr;

10     if (argc == 2)
11         listenfd = Tcp_listen(NULL, argv[1], &addrlen);
12     else if (argc == 3)
13         listenfd = Tcp_listen(argv[1], argv[2], &addrlen);
14     else
15         err_quit("usage: serv01 [ <host> ] <port#>");
16     cliaddr.buf = Malloc(addrlen);
17     cliaddr.maxlen = addrlen;

18     Signal(SIGCHLD, sig_chld);
19     Signal(SIGINT, sig_int);

20     for ( ; ; ) {
21         connfd = Xti_accept(listenfd, &cliaddr, 1);
22         printf("connection from %s\n", Xti_ntop(&cliaddr));

23         if ( (childpid = Fork()) == 0) {      /* child process */
24             Close(listenfd);    /* close listening endpoint */
25             web_child(connfd);  /* process the request */
26             exit(0);
27         }
28         Close(connfd);          /* parent closes connected endpoint */
29     }
30 }

31 void
32 sig_int(int signo)
33 {
34     void    xti_accept_dump(void);

35     xti_accept_dump();
36     exit(0);
37 }
```

xtiserver/serv01.c

Figure 30.6 TCP concurrent server to demonstrate multiple connection problem.

If we start this server to listen on TCP port 9999

```
unixware % serv01 9999
```

and run the client from another host as

```
solaris % client unixware 9999 1 600 4000
```

everything works fine. (1 is the number of children to `fork`, 600 is the number of connections per child, and 4000 is the number of bytes per connection.) Our server works because we tell the client to spawn only one child, so the connections arrive serially at the server from this one client.

If we change the third command-line argument from 1 to 2, however, we get the following error almost immediately from our server:

```
t_accept error: event requires attention
```

The problem is that two connections arrive at the server at about the same time—one connection from each of the two children. TCP's three-way handshake takes place for both connections because the server's TCP establishes the connections.

While the server TCP is establishing the connections, our server process is blocked in a call to `t_listen` from our server's call to `xti_accept`. When the first connection completes the three-way handshake, `t_listen` returns, and then `t_open` and `t_accept` are called. But when `t_accept` is called for this first connection, the second connection has completed the three-way handshake and is also ready to be accepted. The rules of XTI now dictate that instead of `t_accept` completing the first connection, it returns an error with `t_errno` set to `TLOOK` ("event requires attention"). The event pending is `T_LISTEN` (a connect indication is pending) because there is another connection pending (the second connection).

What is happening here is that `t_accept` always returns an error if there is another connection ready. What we have to do is call `t_listen` to receive all the connect indications, saving the `t_call` structure for each connection, and then call `t_accept` for each connection.

> Why does XTI accept connections in this bizarre manner? If `t_listen` really returned when the client's SYN arrived (Figure 30.1), then forcing the server to call `t_listen` on all SYNs that have arrived, before calling `t_accept` for any single connection, gives the server process the opportunity to choose the order in which it accepts the pending connections. The server might prioritize them, for example, based on the IP address or port number returned by `t_listen` or based on the user data that might accompany the connection request (which is not supported by TCP). But given that `t_listen` does not return for TCP until the three-way handshake completes, all this feature does is add (needless) complexity to the server.
>
> What we just described corresponds to XTI in Unix 95. Posix.1g and Unix 98 change the description of `t_accept` to say that it *may* fail with `t_errno` set to `TLOOK`. Nevertheless we must be prepared for `t_accept` to fail in this fashion.

30.8 `xti_accept` Function (Revisited)

We now redo our simple `xti_accept` function from Figure 30.4 into one that is more robust. There are two scenarios that we must handle.

- `t_accept` failing because there is another connection pending (`t_look` will return `T_LISTEN`), and
- `t_accept` failing because a pending connection has received an RST (`t_look` will return `T_DISCONNECT`).

To handle these semantics we must maintain a queue of pending connections. The potential size of this queue is the value of `qlim` when we called `t_bind` for the listening endpoint. There are lots of possible data structures that we can use to keep track of the pending connections. For simplicity we will use a simple stack (array) of `cli` structures. Each structure contains the connected descriptor, a diagnostic counter of how often the structure is used, and a pointer to a `t_call` structure.

Let's assume that three clients establish connections with our server at about the same time. The server TCP will complete all three of the three-way handshakes and when the first one is returned by `t_listen` we use the `cli[0]` structure in our array, as shown in Figure 30.7.

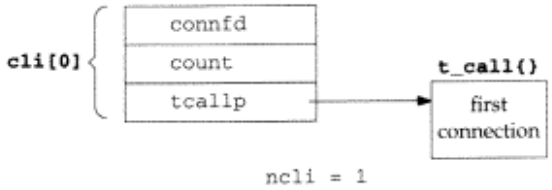


Figure 30.7 Data structures after first client connection returned by `t_listen`.

`connfd` is the new descriptor that we create by calling `t_open`, on which the connection will be accepted. `count` is a diagnostic counter that we examine shortly with our test program. `tcallp` is a pointer to a `t_call` structure that is filled in by `t_listen` and then passed to `t_accept`. It contains the client's protocol address (the `addr` member) and the connection identifier (the `sequence` member). We also keep a counter (`ncli`) of the number of entries in our array of `cli` structures, and its value would be 1.

Assume that we call `t_accept` to accept this first connection, but `t_accept` returns an error of `TLOOK` and `t_look` returns `T_LISTEN`. We must then call `t_listen` again to fetch the next connection. We just add another entry to our `cli` array and increment `ncli`. We show this in Figure 30.8.

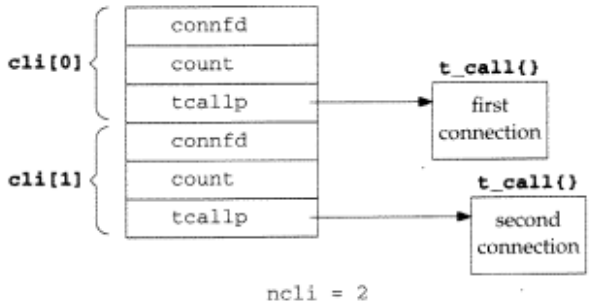


Figure 30.8 Data structures after second client connection returned by `t_listen`.

We always call `t_accept` for the "last" entry in the array, the one whose array index is `ncli-1`. This causes us to process the connections on a last-in, first-out basis (LIFO), instead of the first-in, first-out (FIFO) that one might expect. It is not hard to change this, if desired, but adds complexity.

Figure 4.9 shows that even on a moderately busy Web server, it is rare for more than one completed connection to be ready for the application to accept at any given time. Therefore the simplicity of our design is practical.

At this point we call `t_accept` for `cli[1]` but assume it also returns an error of `TLOOK` and `t_look` then returns `T_LISTEN`. We must add another entry to our array, `cli[2]`, as shown in Figure 30.9.

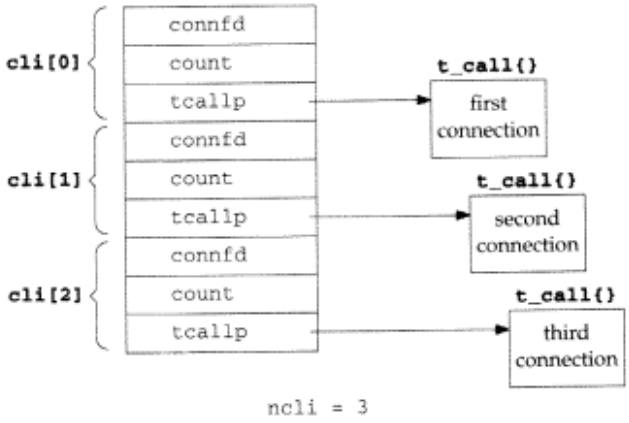


Figure 30.9 Data structures after third client connection returned by `t_listen`.

At this point we have three connections pending and we now call `t_accept` for `cli[2]`. But now assume that the first client (`cli[0]`) has just aborted its connection by sending an RST. Once again the call to `t_accept` fails with an error of `TLOOK`, but this time `t_look` returns `T_DISCONNECT`. We must now call `t_rcvdis` to receive the disconnect. Recall that one entry in the `t_discon` structure that this function fills in is the `sequence` identifier of the connection that was aborted (Section 28.10). We must use this to search our array of `cli` structures, looking for the entry whose `t_call` structure has a matching `sequence`. We then move the entries "up" by one, overwriting `cli[0]` with `cli[1]`, and then overwriting `cli[1]` with `cli[2]`. We also decrement `ncli` and have the data structures shown in Figure 30.10.

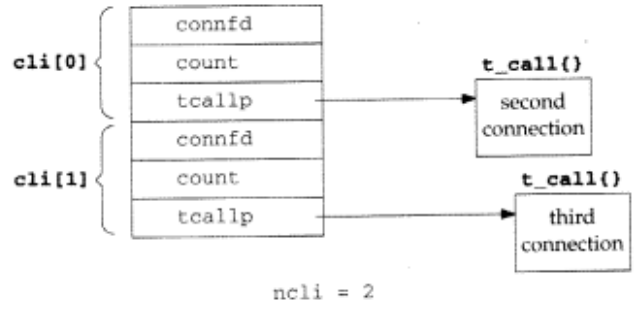


Figure 30.10 Data structures after first connection is aborted.

We call t_accept (again) for cli[1] but assume this time it succeeds. We then remove the cli[1] entry from the array, decrement ncli to become 1, and pass the third connection back to the caller of xti_accept. Remember that everything described so far in this example, starting with the first call to t_listen, has taken place in our xti_accept function. This is now the first time this function has returned a connected descriptor to the caller.

The next time xti_accept is called, ncli will be 1, and t_accept is called for cli[0]. Assuming it succeeds, the second connection is returned to the caller.

We now show the source code for our xti_accept function, the first half of which is in Figure 30.11.

libxti/xti_accept.c

```
 1 #include    "unpxti.h"

 2 static int ncli = -1, ndisconn;
 3 static struct cli {
 4     int     connfd;             /* connected fd or -1 if disconnected */
 5     int     count;
 6     struct t_call *tcallp;      /* ptr to t_alloc'ed structure */
 7 } *cli;                    /* cli[0], cli[1], ..., cli[ncli-1] are in use */

 8 int
 9 xti_accept(int listenfd, struct netbuf *cliaddr, int rdwr)
10 {
11     int     i, event;
12     u_int   n;
13     char   *ptr;
14     struct t_discon tdiscon;

15     if (ncli == -1) {           /* initialize first time through */
16         if (cli != NULL)
17             err_quit("already initialized");
18         if ( (ptr = getenv("LISTENQ")) != NULL)
19             n = atoi(ptr);
20         else
21             n = LISTENQ;
22         cli = Calloc(n, sizeof(struct cli));
23         for (i = 0; i < n; i++)
24             cli[i].tcallp = T_alloc(listenfd, T_CALL, T_ALL);
25         ncli = 0;
26     }
```

libxti/xti_accept.c

Figure 30.11 xti_accept function: first part.

Declare static variables

2–7 The counter ncli, another diagnostic counter of aborted connections ndisconn, and a pointer to our array of cli structures are declared as static.

Initialize the first time we are called

15–26 The first time we are called we allocate an array of cli structures, the number of entries in the array being the constant LISTENQ or the value of the environment variable of the same name. This will be the same value as tcp_listen used in the call to

t_bind. For every element in the array we then call t_alloc to allocate a t_call structure and store the returned pointer in our cli structure.

> In a threads environment this one copy of the cli array and its ncli counter must be protected to allow multiple threads to call xti_accept at the same time.

The second half of the function is shown in Figure 30.12.

Wait for a connection

28-35 If our array is empty, we must call t_listen and wait for a connection. This call to t_listen is where the listening server spends most of its time. When t_listen returns, the client's protocol address and the connection identifier have been saved in the t_call structure by t_listen. We call t_open to create a new endpoint on which the connection will be accepted and bind this endpoint to any local address.

Call t_accept; return on success

36-46 We call t_accept to accept the connection specified by cli[ncli-1], and if it succeeds we return this connected descriptor to the caller. We also call our xti_rdwr function, if the caller want to use read and write, and optionally return the client's protocol address.

Handle additional pending connections

47-53 If t_accept returns TLOOK and t_look returns T_LISTEN, another connection is pending and we must call t_listen to receive the information about this new connection. We note that this call will not block, since the endpoint has a T_LISTEN event pending. We also call t_open and t_bind and save the information in the array entry cli[ncli].

Handle disconnection of pending connection

54-66 If t_accept returns TLOOK and t_look returns T_DISCONNECT, one of the connections that has already been returned by t_listen has been aborted by the client. We call t_rcvdis to obtain the information about the aborted connection (i.e., its sequence) and then search our array for the matching entry. When we find the matching entry we calculate the number of entries beyond the one being aborted (n) and call memmove to move the remaining entries. We call memmove, instead of memcpy, because the former correctly handles overlapping fields, which we will have in this scenario (see Exercise 30.3).

We can test our server from Figure 30.6 using this new version of xti_accept and it works as expected with multiple clients. We also want to see how often t_accept returns an error because of an additional pending connection. To see this we write a simple function that prints the count member of the cli structure, and the ndisconn counter (which we describe shortly), and then call this function from the server parent's SIGINT signal handler (Figure 30.6). Figure 30.13 (p. 814) shows the results with the server on UnixWare 2.1.2 and the client on Solaris 2.5.1. We vary the number of client children from 1 to 4, always issuing a total of 600 connections from all the children.

libxti/xti_accept.c

```
27      for ( ; ; ) {
28          if (ncli == 0) {        /* need to wait for a connection */
29              T_listen(listenfd, cli[ncli].tcallp);   /* block here */

30                  /* following assumes caller called tcp_listen() */
31              cli[ncli].connfd = T_open(xti_serv_dev, O_RDWR, NULL);
32              T_bind(cli[ncli].connfd, NULL, NULL);
33              cli[ncli].count++;
34              ncli++;
35          }
36          if (t_accept(listenfd, cli[ncli - 1].connfd,
37                       cli[ncli - 1].tcallp) == 0) {
38              ncli--;             /* success */
39              if (rdwr)
40                  Xti_rdwr(cli[ncli].connfd);

41              if (cliaddr) {      /* return client's protocol address */
42                  n = min(cliaddr->maxlen, cli[ncli].tcallp->addr.len);
43                  memcpy(cliaddr->buf, cli[ncli].tcallp->addr.buf, n);
44                  cliaddr->len = n;
45              }
46              return (cli[ncli].connfd);

47          } else if (t_errno == TLOOK) {
48              if ( (event = T_look(listenfd)) == T_LISTEN) {
49                  T_listen(listenfd, cli[ncli].tcallp);   /* won't block */
50                  cli[ncli].connfd = T_open(xti_serv_dev, O_RDWR, NULL);
51                  T_bind(cli[ncli].connfd, NULL, NULL);
52                  cli[ncli].count++;
53                  ncli++;

54              } else if (event == T_DISCONNECT) {
55                  T_rcvdis(listenfd, &tdiscon);
56                  for (i = 0; i < ncli; i++) {
57                      if (cli[i].tcallp->sequence == tdiscon.sequence) {
58                          T_close(cli[i].connfd);
59                          ndisconn++;
60                          ncli--;
61                          if ( (n = ncli - i) > 0)
62                              memmove(&cli[i], &cli[i + 1],
63                                      n * sizeof(struct cli));
64                          break;
65                      }
66                  }
67              } else
68                  err_quit("unexpected t_look event %d", event);
69          } else
70              err_xti("unexpected t_accept error");
71      }
72  }
```

libxti/xti_accept.c

Figure 30.12 xti_accept function: second half.

Server counter	#client children			
	1	2	3	4
cli[0].count	600	309	95	102
cli[1].count		291	286	121
cli[2].count			219	235
cli[3].count				142
Total	600	600	600	600

Figure 30.13 Counter of how often `t_accept` returns `T_LISTEN`.

Even with only two clients, each establishing 300 connections, one after the other, at about the same time, `t_accept` returns `TLOOK` with `T_LISTEN` about half the time.

> Why do we see multiple completed connections ready for the server to accept so often in this scenario, when we showed earlier (Figure 4.9) that on a busy Web server this is a rare occurrence? One reason is that we are forcing this scenario in Figure 30.13 with all the connections coming from a client on the same LAN. Second, the rate of the connections, 600 in about 12 seconds, corresponds to more than 4 million connections per day. Lastly, we purposely ran this example on a slow server (a 75-Mhz Pentium CPU) to test the handling of multiple pending connections with our `xti_accept` function. We can summarize this scenario as being infrequent enough in the real world so that handling them in a LIFO order by our `xti_accept` function is fine, but since the scenario can and does occur, the server must handle it.

To test this server with a client that aborts just-established connections we modify our client from Figure 27.4 by changing the innermost loop to the following:

```
    for (j = 0; j < nloops; j++) {
        fd = Tcp_connect(argv[1], argv[2]);

+       if (i == 2 && (j % 3) == 0) {
+           struct linger   ling;
+
+           ling.l_onoff = 1;
+           ling.l_linger = 0;
+           Setsockopt(fd, SOL_SOCKET, SO_LINGER, &ling, sizeof(ling));
+           Close(fd);
+
+           /* and just continue on for this client connection ... */
+           fd = Tcp_connect(argv[1], argv[2]);
+       }

        Write(fd, request, strlen(request));

        if ( (n = Readn(fd, reply, nbytes)) != nbytes)
            err_quit("server returned %d bytes", n);

        Close(fd);          /* TIME_WAIT on client, not server */
    }
```

The lines preceded by a plus sign are new. This modification causes the third child (`i` equals 2) to abort every third just-completed connection. To send the RST we set the `SO_LINGER` socket option accordingly and `close` the socket. We then create another

connection and continue with the loop. The effect on the server depends on the timing: some RSTs may arrive between the server's call to `t_listen` and its call to `t_accept` (and we count these with our `ndisconn` counter to verify that the code is exercised). Others may arrive before `t_listen` notifies the server of the connection, and others may arrive after the connection is accepted.

XTI Queue Length versus Listen Backlog

The XTI queue length and the `listen` backlog are similar but not identical. First, there has never been an exact specification of what the `listen` backlog means. We saw in Figure 4.10 that current systems differ in their interpretation.

Posix.1g states that the XTI queue length value specifies the number of "outstanding connect indications" that the provider should support for the endpoint. An outstanding connect indication is one that has been passed to the application by the provider but not yet accepted or rejected. The provider *may* queue more connect indications than specified but must ensure that there are never more than `qlen` delivered to the application that are still outstanding at any given time.

If the implementation passed connect indications to the application when they arrived (i.e., when the client SYN arrives at the server), then this form of application queueing might make sense. But given that a TCP connection is completely established before the application is notified, there is no real need for the application to queue these connections.

As usual, the way to make any sense out of standards is to see what the implementations really provide when we specify different values for the `qlen`. We modified Figure E.15 to work with XTI instead of sockets and ran the program on our five systems that support XTI. The results are shown in Figure 30.14.

	AIX 4.2		DUnix 4.0B		HP-UX 10.30		Solaris 2.6		UWare 2.1.2	
request `qlen`	return `qlen`	actual #conns	return `qlen`	actual #conns	return `qlen`	actual #conns	return `qlen`	actual #conns	return `qlen`	actual #conns
0	0	0	0	0	0	0	0	0		0
1	1	3	1	2	1	1	1	1		1
2	2	6	2	4	2	2	2	2		2
3	3	8	3	6	3	3	3	3		3
4	4	11	4	8	4	4	4	4		4
5	5	13	5	10	5	5	5	5		5
6	5	13	6	12	6	6	6	6		6
7	5	13	7	14	7	7	7	7		7
8	5	13	8	16	8	8	8	8		8
9	5	13	9	18	9	9	9	9		9
10	5	13	10	20	10	10	10	10		10
11	5	13	11	22	11	11	11	11		11
12	5	13	12	24	12	12	12	12		12
13	5	13	13	26	13	13	13	13		13
14	5	13	14	28	14	13	14	14		14

Figure 30.14 Actual number of queued connections for values of XTI `qlen`.

Recall from Section 28.5 that the third argument to `t_bind` is a pointer to a `t_bind` structure that is filled in upon return by the provider. By looking at the `qlen` member of this structure we can see what the provider sets this to. (XTI calls this a "negotiated" value.) We note that most systems return the value that was specified, unless a smaller value is supported (AIX). One system (UnixWare) does not return the value.

None of the systems allow any connections when the `qlen` is 0 (which differs from a `listen` backlog of 0 in Figure 4.10), and two of the systems returned an error from `t_connect` (HP-UX and Solaris). Some of the implementations queue more connections than specified by `qlen` (AIX and Digital Unix), but the remaining three do not.

> Here we are measuring the number of connections queued by the provider, not by the application, but it is this queueing by the provider in which we are most interested.

Setting the Server's Listening Queue Length to 1

One way to avoid the complication involved in accepting XTI connections is to set the `qlen` member of the `t_bind` structure to 1. But the problem with this solution is that many implementations will then queue only one client connection (Figure 30.14) and then ignore all other arriving SYNs until this connection is accepted.

We can test this feature with our server from this section. We again run the server on UnixWare 2.1.2, which queues only one connection when the specified `qlen` is 1. Like the numbers in Figure 30.13, we vary the number of client children that are issuing connections between 1 and 4, but this time we measure the clock time (in seconds) required for the fixed number of connections (600).

Queue length	#client children			
	1	2	3	4
1	10.6	12.2	15.6	13.2
1024	10.6	10.2	10.3	10.4

Figure 30.15 Clock time for 600 total connections, varying number of children and listen queue.

For a queue length of 1, the clock time increases as the number of children increases. This is caused by many of the client SYNs being ignored by the server, because one connection has filled the queue, and the client must retransmit the SYN. But when the queue length is greater than the number of simultaneous connections, the clock time decreases for the few number of children that we are testing here.

These numbers verify our previous statement: a queue length of 1 is unrealistic for a real-world server.

30.9 Summary

Accepting client connections with XTI is much harder than the same operation with sockets. As we described, the reason for the added complexity is to allow protocols to provide a lazy accept, where the application is notified when the connect request

arrives, and not when the connection establishment is complete. TCP implementations do not provide a lazy accept, and Unix 98 no longer requires `t_accept` to return an event of `T_LISTEN` when another connection is pending, but for backward compatibility XTI servers must handle this scenario.

Exercises

30.1 We mentioned in Section 30.2 that some implementations store a pointer in the `sequence` member of the `t_call` structure that is filled in by `t_listen`. What happens on a 64-bit architecture?

30.2 Why do we add one to `PATH_MAX` in the declaration of `xti_serv_dev` in Figure 30.3?

30.3 In Figure 30.12 we call `memmove` and mention that this is needed since the source and destination overlap. Assume a 4-byte array, with elements `x[0]` through `x[3]` (drawn from left to right) and assume that we want to delete `x[1]`, moving the next two elements "left" by 1 byte, leaving three elements. Draw a picture of the source field and destination field. Then describe what happens if the copy operation starts from the beginning of the source field to the beginning of the destination (copying from right to left). Then describe what happens if the copy operation starts from the end of the source field to the end of the destination (copying from left to right). Does `memcpy` guarantee in which direction the copy takes place?

30.4 Recode Figure E.15 to use XTI instead of sockets.

30.5 Recode Figures 30.11 and 30.12 to use a linked list of `cli` structures instead of the fixed-size array that we used for simplicity. Allocate the structures dynamically.

31

XTI: UDP Clients and Servers

31.1 Introduction

XTI provides three functions for connectionless clients and servers: `t_sndudata` to send a datagram, `t_rcvudata` to receive a datagram, and `t_rcvuderr` to obtain information about an asynchronous error. With sockets we had the option of calling `connect` for a UDP application, but XTI does not provide that choice.

31.2 `t_rcvudata` and `t_sndudata` Functions

These two functions are used with connectionless protocols (such as UDP) to receive and send datagrams.

```
#include <xti.h>

int t_rcvudata(int fd, struct t_unitdata *unitdata, int *flagsp);

int t_sndudata(int fd, struct t_unitdata *unitdata);
```

Both return: 0 if OK, −1 on error

For the `t_sndudata` function the `t_unitdata` structure specifies the destination address, any options, and the actual data to send.

```
struct t_unitdata {
  struct netbuf  addr;    /* protocol-specific address */
  struct netbuf  opt;     /* protocol-specific options */
  struct netbuf  udata;   /* user data */
};
```

For the t_rcvudata function this structure specifies where to store the sender's protocol address, any received options, and the actual data.

Both of these functions return 0 if everything is OK, or −1 on an error. This differs from most read and write functions that usually return the number of bytes transferred. With t_rcvudata the size of the received datagram is returned as the udata.len member of the t_unitdata structure. t_sndudata does not return the number of bytes written; it just returns 0 on success (i.e., the entire datagram has been copied into the kernel's buffers).

The integer pointed to by *flagsp* is similar to the final argument to t_rcv: it is not a value–result argument because its value is not examined by the function; it is set only on return. The T_MORE flag is returned if another call to t_rcvudata is required to read more of the datagram (i.e., what remains of the datagram exceeds the length of the receive buffer). We show an example of this in Section 31.6.

These two XTI functions correspond to the sendto and recvfrom functions.

31.3 udp_client Function

Before showing a UDP example using XTI we will write a function named udp_client, with the same calling sequence as shown in Section 11.10, that creates an XTI endpoint for a UDP client. This function, shown in Figure 31.1, handles the hostname and service name conversions described in Chapter 29.

Look up hostname and service name

12-19 The calls to getnetpath and netdir_getbyname are similar to the calls described in Figure 29.6.

Open device, bind any local address

20-21 t_open opens the appropriate device and t_bind binds any local address to the endpoint.

Allocate t_unitdata structure

22 t_alloc allocates a t_unitdata structure but only the addr structure within, not the opt or udata structures. We do not allocate the opt structure because per-datagram options are rare with UDP (Chapter 32). We do not allocate the udata structure because the range of UDP datagram sizes is large, up to 65507 bytes as shown in Figure 28.2, but few applications deal with these maximum-sized datagrams. Since most applications deal with smaller datagrams (a few thousand bytes at the most), it makes more sense for the application to allocate the data buffer itself, based on its needs.

Use first returned address for server

23-25 The addr structure is filled in with the first address returned for the server. Figure 31.2 (p. 822) shows the data structures involved, assuming that two netbuf structures are returned for the specified hostname and service name, and assuming that the implementation uses the sockaddr_in structure to represent IPv4 addresses.

libxti/udp_client.c

```
 1 #include    "unpxti.h"

 2 int
 3 udp_client(const char *host, const char *serv, void **vptr, socklen_t *lenp)
 4 {
 5     int     tfd;
 6     void   *handle;
 7     struct netconfig *ncp;
 8     struct nd_hostserv hs;
 9     struct nd_addrlist *alp;
10     struct netbuf *np;
11     struct t_unitdata *tudptr;

12     handle = Setnetpath();

13     hs.h_host = (char *) host;
14     hs.h_serv = (char *) serv;

15     while ( (ncp = getnetpath(handle)) != NULL) {
16         if (strcmp(ncp->nc_proto, "udp") != 0)
17             continue;

18         if (netdir_getbyname(ncp, &hs, &alp) != 0)
19             continue;

20         tfd = T_open(ncp->nc_device, O_RDWR, NULL);

21         T_bind(tfd, NULL, NULL);

22         tudptr = T_alloc(tfd, T_UNITDATA, T_ADDR);

23         np = alp->n_addrs;      /* use first server address */
24         tudptr->addr.len = min(tudptr->addr.maxlen, np->len);
25         memcpy(tudptr->addr.buf, np->buf, tudptr->addr.len);

26         endnetpath(handle);
27         netdir_free(alp, ND_ADDRLIST);

28         *vptr = tudptr;         /* return pointer to t_unitdata{} */
29         *lenp = tudptr->addr.maxlen;    /* and size of addresses */
30         return (tfd);
31     }
32     endnetpath(handle);
33     return (-1);
34 }
```

libxti/udp_client.c

Figure 31.1 udp_client function for XTI.

The addr.maxlen value should be the same as the maxlen values in the structure returned by netdir_getbyname (16 for IPv4), but we use the min macro to be certain we do not overflow the destination of the memcpy. We show four lengths of 0 and two null pointers for the opt and udata structures, since t_alloc initializes these members to these values because we told it to allocate and initialize only the addr structure with the T_ADDR argument.

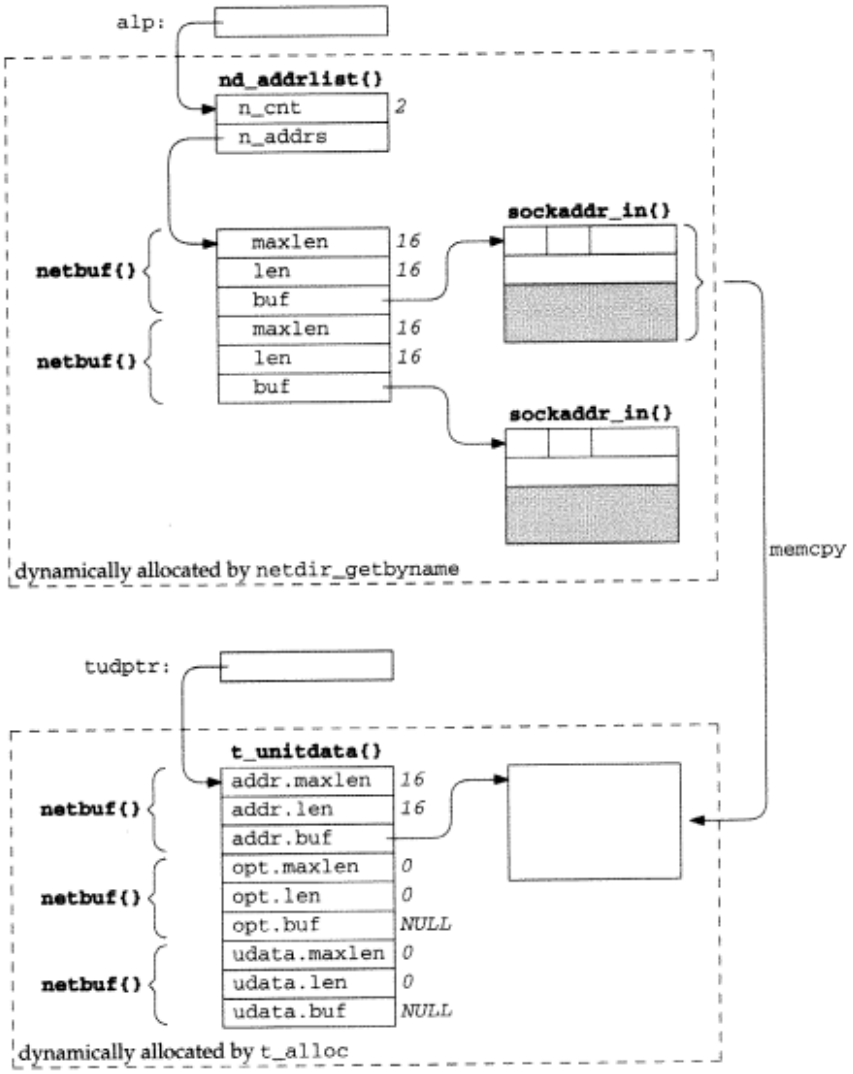


Figure 31.2 Data structures during call to `udp_client`.

Free memory and return

26-33 `endnetpath` frees the memory allocated for the `netconfig` structure and `netdir_free` releases the memory that `netdir_getbyname` allocated (Figure 31.2). The pointer to the `t_unitdata` structure is returned to the caller, along with the size of the protocol addresses and the descriptor for the endpoint. We return the size of the protocol addresses as `addr.maxlen` instead of `addr.len`, because this value is returned for the caller to use in a call to `malloc`. Should the addresses be variable-length, we should return the maximum size, and not just the size of this address.

Example: Daytime Client

We now use our udp_client function to recode our protocol-independent daytime client from Figure 11.12 to use XTI, which we show in Figure 31.3.

xtiudp/daytimeudpcli1.c

```
 1 #include    "unpxti.h"

 2 int
 3 main(int argc, char **argv)
 4 {
 5     int     tfd, flags;
 6     char    recvline[MAXLINE + 1];
 7     socklen_t addrlen;
 8     struct t_unitdata *sndptr, *rcvptr;

 9     if (argc != 3)
10         err_quit("usage: daytimeudpcli <hostname> <service>");

11     tfd = Udp_client(argv[1], argv[2], (void **) &sndptr, &addrlen);

12     rcvptr = T_alloc(tfd, T_UNITDATA, T_ADDR);

13     printf("sending to %s\n", Xti_ntop_host(&sndptr->addr));

14     sndptr->udata.maxlen = MAXLINE;
15     sndptr->udata.len = 1;
16     sndptr->udata.buf = recvline;
17     recvline[0] = 0;            /* 1-byte datagram containing null byte */
18     T_sndudata(tfd, sndptr);

19     rcvptr->udata.maxlen = MAXLINE;
20     rcvptr->udata.buf = recvline;
21     T_rcvudata(tfd, rcvptr, &flags);
22     recvline[rcvptr->udata.len] = 0;    /* null terminate */
23     printf("from %s: %s", Xti_ntop_host(&rcvptr->addr), recvline);

24     exit(0);
25 }
```

xtiudp/daytimeudpcli1.c

Figure 31.3 UDP daytime client using XTI and our udp_client function.

Create endpoint

11–13 We call our udp_client function to create the XTI endpoint and to allocate a t_unitdata structure that we will use for sending datagrams. We allocate another t_unitdata structure that we will use for receiving replies. We call our xti_ntop_host function to print the server's IP address.

Send datagram

14–18 We initialize the udata structure within the t_unitdata structure to point to our recvline buffer and to contain one byte of 0. t_sndudata sends the datagram to the server.

We should be able to send a 0-byte UDP datagram, given our discussion with Figure 28.4, but many implementations of XTI do not allow this.

Read reply

19-23 We initialize the `udata` structure for the receiving `t_unitdata` structure and call `t_rcvudata` to read the server's reply. The reply is null terminated and printed to standard output, along with the server's address.

This example shares all the unreliable UDP properties of our sockets client in Figure 11.12: the client will block forever in the call to `t_rcvudata` if there is no response. If we run the client to a host that is running the server, the output is as we expect.

```
unixware % daytimeudpcli1 bsdi daytime
sending to 206.62.226.35
from 206.62.226.35: Fri Feb 28 17:23:40 1997
```

What if we send a datagram to this same host, but to a UDP port that no process has bound? We expect an ICMP port unreachable error to be returned. Recall with our sockets client, if the client does not call `connect`, this error is not returned to the client. There is nothing similar to `connect` for a UDP endpoint with XTI, but we see that the error is returned to our client and our `T_rcvudata` wrapper function prints an error:

```
unixware % daytimeudpcli1 bsdi 9999
sending to 206.62.226.35
t_rcvudata error: event requires attention
```

When an asynchronous error is received for a UDP endpoint, `t_rcvudata` returns an error of `TLOOK` and we must call `t_rcvuderr` to determine the actual error. We discuss this in the next section.

31.4 `t_rcvuderr` Function: Asynchronous Errors

For a connectionless protocol, errors can be returned asynchronously. That is, a datagram can be correctly transmitted by the protocol stack, only to have an error in it detected somewhere else in the network. Common errors with UDP datagrams are to elicit an ICMP port unreachable from the destination host or an ICMP host unreachable from some intermediate router. When this ICMP error is received at some later time by the provider, it requires some form of notification from the provider to the process and some means for the process to determine the actual error. As shown at the end of the previous section, XTI provides this notification by setting `t_errno` to `TLOOK` when we call `t_rcvudata` to indicate that an error occurred on a previously sent datagram. We can then call the `t_rcvuderr` function to determine what happened and to clear the error status.

```
#include <xti.h>

int t_rcvuderr(int fd, struct t_uderr *uderr);
```

Returns: 0 if OK, −1 on error

If the *uderr* pointer is nonnull, a t_uderr structure is filled in with information about the error.

```
struct t_uderr {
  struct netbuf  addr;     /* protocol-specific address */
  struct netbuf  opt;      /* protocol-specific options */
  t_scalar_t     error;    /* protocol-specific error */
};
```

The addr structure contains the destination address of the datagram that caused the error, the opt structure contains any protocol-specific options from the datagram that caused the error, and error contains a protocol-specific error code. For UDP, error is normally one of the errno values from <sys/errno.h>.

If *uderr* is a null pointer, this clears the error status without returning any information.

Example: ICMP Port Unreachable

We now recode our client from Figure 31.3 to handle asynchronous errors. This is shown in Figure 31.4.

Handle asynchronous errors

23-33 What has changed from Figure 11.12 is the call to our wrapper function T_rcvudata is replaced with a call to t_rcvudata, and we handle an asynchronous error by calling t_rcvuderr, printing the returned error code.

If we run this program and send a datagram to a host that does not support the daytime protocol, we receive the ICMP port unreachable error from t_rcvuderr:

```
unixware % daytimeudpcli2 gateway.tuc.noao.edu daytime
sending to 140.252.104.1
error 146 for datagram sent to 140.252.104.1

unixware % grep 146 /usr/include/sys/errno.h
#define ECONNREFUSED    146       /* Connection refused */
```

We see that the error value returned in the t_uderr structure is the errno value corresponding to the ICMP error (Figure A.15).

Unfortunately, as nice as this design feature appears (returning ICMP errors for XTI UDP endpoints), there are still problems. First, there is no requirement that the provider notify the application when these errors occur. With UnixWare 2.1.2, for example, ICMP port unreachables are returned to the application, but ICMP host unreachables are not. Also, if we modify our client to send three datagrams to three different servers and then read all the replies, but two of the datagrams elicit an ICMP port unreachable error, only the first of these two errors is returned to the application by t_rcvuderr. This is because the provider maintains only one error per endpoint. All of these problems are what led us to develop an independent way of notifying a datagram application of asynchronous errors: our icmpd daemon in Section 25.7.

Notice that all we receive is the error code and the destination address of the datagram that caused the error. Another piece of information that is not returned is the source address of who returned the error (e.g., the source address of the ICMP error).

xtiudp/daytimeudpcli2.c

```
 1 #include    "unpxti.h"

 2 int
 3 main(int argc, char **argv)
 4 {
 5     int     tfd, flags;
 6     char    recvline[MAXLINE + 1];
 7     socklen_t addrlen;
 8     struct t_unitdata *sndptr, *rcvptr;
 9     struct t_uderr *uderr;

10     if (argc != 3)
11         err_quit("usage: a.out <hostname or IPaddress> <service or port#>");

12     tfd = Udp_client(argv[1], argv[2], (void **) &sndptr, &addrlen);

13     rcvptr = T_alloc(tfd, T_UNITDATA, T_ADDR);
14     uderr = T_alloc(tfd, T_UDERROR, T_ADDR);

15     printf("sending to %s\n", Xti_ntop_host(&sndptr->addr));

16     sndptr->udata.maxlen = MAXLINE;
17     sndptr->udata.len = 1;
18     sndptr->udata.buf = recvline;
19     recvline[0] = 0;            /* 1-byte datagram containing null byte */
20     T_sndudata(tfd, sndptr);

21     rcvptr->udata.maxlen = MAXLINE;
22     rcvptr->udata.buf = recvline;
23     if (t_rcvudata(tfd, rcvptr, &flags) == 0) {
24         recvline[rcvptr->udata.len] = 0;     /* null terminate */
25         printf("from %s: %s", Xti_ntop_host(&rcvptr->addr), recvline);
26     } else {
27         if (t_errno == TLOOK) {
28             T_rcvuderr(tfd, uderr);
29             printf("error %ld for datagram sent to %s\n",
30                    uderr->error, Xti_ntop_host(&uderr->addr));
31         } else
32             err_xti("t_rcvudata error");
33     }
34     exit(0);
35 }
```

xtiudp/daytimeudpcli2.c

Figure 31.4 UDP client using XTI that handles asynchronous errors.

31.5 `udp_server` Function

We can also recode our `udp_server` function from Figure 11.14 to use XTI, and we show this in Figure 31.5.

libxti/udp_server.c

```
 1 #include    "unpxti.h"

 2 int
 3 udp_server(const char *host, const char *serv, socklen_t *addrlenp)
 4 {
 5     int     tfd;
 6     void   *handle;
 7     struct t_bind tbind;
 8     struct t_info tinfo;
 9     struct netconfig *ncp;
10     struct nd_hostserv hs;
11     struct nd_addrlist *alp;
12     struct netbuf *np;

13     handle = Setnetconfig();

14     hs.h_host = (host == NULL) ? HOST_SELF : (char *) host;
15     hs.h_serv = (char *) serv;

16     while ( (ncp = getnetconfig(handle)) != NULL &&
17             strcmp(ncp->nc_proto, "udp") != 0) ;

18     if (ncp == NULL)
19         return (-1);

20     if (netdir_getbyname(ncp, &hs, &alp) != 0)
21         return (-2);
22     np = alp->n_addrs;          /* use first address */

23     tfd = T_open(ncp->nc_device, O_RDWR, &tinfo);

24     tbind.addr = *np;           /* copy entire netbuf{} */
25     tbind.qlen = 0;             /* not used for connectionless server */
26     T_bind(tfd, &tbind, NULL);

27     endnetconfig(handle);
28     netdir_free(alp, ND_ADDRLIST);

29     if (addrlenp)
30         *addrlenp = tinfo.addr; /* size of protocol addresses */
31     return (tfd);
32 }
```

libxti/udp_server.c

Figure 31.5 udp_server function using XTI.

Lookup protocol, host, and service

13-22 The beginning of the function is similar to our tcp_listen function (Figure 30.3), finding the protocol by calling getnetconfig and then calling netdir_getbyname to look up the hostname and service name.

Open device, bind server's IP address and port

23-28 t_open opens the correct device and t_bind binds the server's IP address (the wildcard address if the host argument is a null pointer) and port. The memory allocated by netdir_getbyname is returned by netdir_free.

Return address length and descriptor

29–31 The size of the protocol's addresses is returned if the final argument is a nonnull pointer, and the descriptor for the endpoint is the return value of the function.

Example: Daytime Server

We can now recode our simple daytime server from Figure 11.15 using XTI. We show this in Figure 31.6.

xtiudp/daytimeudpsrv2.c

```
 1 #include    "unpxti.h"
 2 #include    <time.h>

 3 int
 4 main(int argc, char **argv)
 5 {
 6     int     tfd, flags;
 7     char    buff[MAXLINE];
 8     time_t  ticks;
 9     struct t_unitdata *tud;

10     if (argc == 2)
11         tfd = Udp_server(NULL, argv[1], NULL);
12     else if (argc == 3)
13         tfd = Udp_server(argv[1], argv[2], NULL);
14     else
15         err_quit("usage: daytimeudpsrv [ <host> ] <service or port>");

16     tud = T_alloc(tfd, T_UNITDATA, T_ADDR);

17     for ( ; ; ) {
18         tud->udata.maxlen = MAXLINE;
19         tud->udata.buf = buff;
20         if (t_rcvudata(tfd, tud, &flags) == 0) {
21             printf("datagram from %s\n", Xti_ntop(&tud->addr));
22             ticks = time(NULL);
23             snprintf(buff, sizeof(buff), "%.24s\r\n", ctime(&ticks));
24             tud->udata.len = strlen(buff);
25             T_sndudata(tfd, tud);

26         } else if (t_errno == TLOOK)
27             T_rcvuderr(tfd, NULL);  /* just clear error */
28         else
29             err_xti("t_rcvudata error");
30     }
31 }
```

xtiudp/daytimeudpsrv2.c

Figure 31.6 UDP daytime server using XTI.

Create XTI endpoint

10–16 Our `udp_server` function creates the endpoint and binds the local IP address and port. We allocate a `t_unitdata` structure by calling `t_alloc`, specifying the `T_ADDR` argument, so that it allocates a buffer for only the protocol address.

Read request, send reply

17-30 The program then loops, reading a client request with `t_rcvudata` and sending a reply with `t_sndudata`. If one of our replies generates an asynchronous error, `t_rcvudata` will return an error with `t_errno` set to `TLOOK` and we handle the error by calling `t_rcvuderr`. Notice that the final argument to `t_rcvuderr` is a null pointer to clear the error without returning any information (since there is nothing for us to do when these errors occur). If we did not handle these errors as shown, but aborted if `t_rcvudata` returned an error, then any client could crash our server by sending a datagram to the server and then immediately terminating. When the reply was received by the client's host, it would respond with an ICMP port unreachable, causing the server's `t_rcvudata` to return an error. Therefore the handling of these asynchronous errors is mandatory for a UDP server written using XTI.

31.6 Reading a Datagram in Pieces

Recall our discussion of datagram truncation in Section 20.3 and the different scenarios when a datagram is read on a UDP socket, but the datagram length exceeds the number of bytes requested by the application. XTI handles this scenario differently.

Recall the final *flagsp* argument for `t_rcvudata`. If the application's buffer is not large enough to hold the next datagram on the queue, the number of bytes returned will be `udata.maxlen` and the `T_MORE` bit in the integer pointed to by the *flagsp* argument will be turned on. This flag tells the application to call `t_rcvudata` again to read the remainder of this datagram. The sender's address and options are returned by only the first call to `t_rcvudata` for a given datagram. For all subsequent calls to this function that read the remainder of the datagram, `addr.len` and `opt.len` members will be 0 on return.

We can show the use of this feature by modifying our client from Figure 31.4, as shown in Figure 31.7.

Redefine `MAXLINE`

2-3 We redefine the constant `MAXLINE` from our `unp.h` header to be 2 bytes, and this is the size of our `recvline` buffer.

Create endpoint, send datagram to server

12-22 This code has not changed from Figure 31.4.

Read reply, 2 bytes at a time

23-41 We now call `t_rvcudata` in a loop, until our `flags` variable does not have the `T_MORE` bit set. We print the server's IP address for only the first piece of the datagram, when the `addr.len` member is nonzero.

xtiudp/daytimeudpcli4.c

```
 1 #include    "unpxti.h"

 2 #undef  MAXLINE
 3 #define MAXLINE 2

 4 int
 5 main(int argc, char **argv)
 6 {
 7     int     tfd, flags;
 8     char    recvline[MAXLINE + 1];
 9     socklen_t addrlen;
10     struct t_unitdata *sndptr, *rcvptr;
11     struct t_uderr *uderr;

12     if (argc != 3)
13         err_quit("usage: a.out <hostname or IPaddress> <service or port#>");

14     tfd = Udp_client(argv[1], argv[2], (void **) &sndptr, &addrlen);

15     rcvptr = T_alloc(tfd, T_UNITDATA, T_ADDR);
16     uderr = T_alloc(tfd, T_UDERROR, T_ADDR);

17     printf("sending to %s\n", Xti_ntop_host(&sndptr->addr));

18     sndptr->udata.maxlen = MAXLINE;
19     sndptr->udata.len = 1;
20     sndptr->udata.buf = recvline;
21     recvline[0] = 0;            /* 1-byte datagram containing null byte */
22     T_sndudata(tfd, sndptr);

23     do {
24         rcvptr->udata.maxlen = MAXLINE;
25         rcvptr->udata.buf = recvline;
26         flags = 0;
27         if (t_rcvudata(tfd, rcvptr, &flags) == 0) {
28             recvline[rcvptr->udata.len] = 0;    /* null terminate */
29             if (rcvptr->addr.len > 0)
30                 printf("from %s: ", Xti_ntop_host(&rcvptr->addr));
31             printf("%s\n", recvline);
32         } else {
33             if (t_errno == TLOOK) {
34                 T_rcvuderr(tfd, uderr);
35                 printf("error %ld from %s\n",
36                         uderr->error, Xti_ntop_host(&uderr->addr));
37             } else
38                 err_xti("t_rcvudata error");
39             flags = 0;
40         }
41     } while (flags & T_MORE);
42     exit(0);
43 }
```

xtiudp/daytimeudpcli4.c

Figure 31.7 UDP client using XTI that reads returned datagram in pieces.

We now run this client to a daytime server.

```
unixware % daytimeudpcli4 bsdi daytime
sending to 206.62.226.35
from 206.62.226.35: Su
n
Ma
r
 2
 1
1:
53
:5
0
19
97
```

If we remove the newline from the `printf` format string for `recvline` (line 31, which we used only to show how much data was returned by `t_rcvudata`), we get the more familiar output:

```
unixware % daytimeudpcli4 bsdi daytime
sending to 206.62.226.35
from 206.62.226.35: Sun Mar  2 12:04:48 1997
```

31.7 Summary

The two XTI functions `t_rcvudata` and `t_sndudata` are similar to `recvfrom` and `sendto`. One new feature with XTI, which is not provided with sockets, is reading a datagram in pieces, having the `T_MORE` flag returned when there is more to read.

Asynchronous errors are returned with XTI by having `t_rcvudata` and `t_sndudata` return an error of `TLOOK`. We then call `t_rcvuderr` to obtain more protocol-dependent information about the error. This is better than the sockets approach (returning an asynchronous error only if the socket is connected), but even with the XTI approach asynchronous errors can be lost, and our application is still depending on the protocol stack to decide which ICMP errors to return. A better solution is to use a daemon like `icmpd` (Section 25.7) and return all the errors on a separate channel.

32

XTI Options

32.1 Introduction

Another of the mystery areas of XTI has been option processing. The standards and most manuals spend page after page describing the intricacies of option processing and option negotiation, providing no examples, and ending with a statement of the form "the details are protocol dependent."

The term *negotiation* is used heavily with XTI options. An option is not "set"; it is negotiated, meaning the provider may not set the option to exactly what we ask for. When an XTI option is negotiated, the actual value used by the provider is returned, so we can see what that value is.

Figure 32.1 shows all of the standard XTI options, both the generic options (those beginning with `XTI_`) and those for IPv4.

> Unix 98 prepended `T_` to all the `INET_`, `IP_`, `TCP_`, and `UDP_` names, but Posix.1g does not do this. For example, the UDP option is called `UDP_CHECKSUM` by Posix.1g. Unix 98 accepts these Posix.1g names as legacy names, but we will use the newer names in this text.

XTI classifies options as either *end-to-end* or *local*. End-to-end options normally cause some type of information to be sent to the peer across the network. An example is the IPv4 type-of-service field. It can be set by one endpoint (for either TCP or UDP), is carried in the IPv4 header, and is available to the other endpoint. The IPv4 header options and the UDP checksum are the two other end-to-end options in Figure 32.1. An example of a local option is `T_IP_REUSEADDR`, as this option affects the ability of the calling process to bind a port number that is already in use to its endpoint but has no effect on the data that is sent to the other endpoint.

Level	Name	Datatype	End-to-end	Absolute	Description
XTI_GENERIC	XTI_DEBUG	t_uscalar_t[]		•	enable debug tracing
	XTI_LINGER	t_linger{}		•	linger on close if data to send
	XTI_RCVBUF	t_uscalar_t			receive buffer size
	XTI_RCVLOWAT	t_uscalar_t			receive buffer low-water mark
	XTI_SNDBUF	t_uscalar_t			send buffer size
	XTI_SNDLOWAT	t_uscalar_t			send buffer low-water mark
T_INET_IP	T_IP_BROADCAST	u_int		•	permit sending of broadcast mesg
	T_IP_DONTROUTE	u_int		•	bypass routing table lookup
	T_IP_OPTIONS	u_char[]	•	•	IP header options
	T_IP_REUSEADDR	u_int		•	allow local address reuse
	T_IP_TOS	u_char	•	•	type-of-service and precedence
	T_IP_TTL	u_char		•	time-to-live
T_INET_TCP	T_TCP_KEEPALIVE	t_kpalive{}		•	periodically test if connection alive
	T_TCP_MAXSEG	t_uscalar_t			TCP MSS (read-only)
	T_TCP_NODELAY	t_uscalar_t		•	disable Nagle algorithm
T_INET_UDP	T_UDP_CHECKSUM	t_uscalar_t	•	•	enable UDP checksum

Figure 32.1 XTI options.

Some XTI options are classified as an *absolute requirement*, which we also show in Figure 32.1. When setting the value of an option with this property, if the requested value cannot be assigned to the option, failure is returned. If an option does not have this property, and we try to set the option to some value that is not within the range of supported values, the provider will change the requested value to be within the acceptable range. An example of the latter is the receive buffer size, `XTI_RCVBUF`, as most systems enforce a lower limit and an upper limit on this value. If we request a value less than the lower limit or greater than the upper limit, the value will be changed to the appropriate limit and the return is then "partial success."

XTI options are specified and received in the following ways:

1. Calling the `t_optmgmt` function lets us specify any options (end-to-end and local) that we desire. We can also call this function to obtain the current value or the default value of an option.
2. For a UDP endpoint we can specify our desired options (end-to-end and local) with each call to `t_sndudata`, using the `opt` member of the `t_unitdata` structure.
3. For a UDP endpoint any end-to-end options that arrive with the datagram are returned by `t_rcvudata` through the `opt` member of the `t_unitdata` structure.
4. For a TCP client we can specify our desired options (end-to-end and local) when calling `t_connect`, as the `opt` member of the `t_call` structure.
5. For a TCP server any end-to-end options that arrive with the connection are returned by `t_listen` through the `opt` member of the `t_call` structure.

The t_optmgmt function is a combination of the getsockopt and setsockopt functions. In the sockets API, however, there is no way to specify options when sending or receiving UDP datagrams, or when initiating or accepting TCP connections. The sendmsg and recvmsg functions provide the capability to specify and receive ancillary data, and this is used for IPv6 options.

Figure 32.2 summarizes the sending and receiving of options by the XTI functions.

Endpoint	Function	Return end-to-end only	Return end-to-end and local	Specify end-to-end and local
any endpoint	t_optmgmt		•	•
TCP endpoint	t_accept			•
	t_connect		•	•
	t_listen	•		
	t_rcvconnect		•	
UDP endpoint	t_rcvudata	•		
	t_rcvvudata	•		
	t_rcvuderr		•	
	t_sndudata			•
	t_sndvudata			•

Figure 32.2 XTI functions that can specify and return options.

Figure 32.2 indicates that we can specify options with t_accept. With TCP this is not possible, because the connection is already established when t_listen returns. Hence, any desired end-to-end options that we want to effect the three-way handshake must be specified for the listening endpoint.

32.2 t_opthdr Structure

XTI options are always specified and returned through a netbuf structure named opt that is a member of the t_call, t_optmgmt, t_uderr, and t_unitdata structures (Figure 28.6). The contents of the options buffer is one or more t_opthdr structures, each followed by an optional value.

```
struct t_opthdr {
  t_uscalar_t  len;     /* total length of option:
                           sizeof(struct t_opthdr) + length of value */
  t_uscalar_t  level;   /* protocol affected */
  t_uscalar_t  name;    /* option name */
  t_uscalar_t  status;  /* status value */
          /* followed by the option value, and then possible padding */
};
```

One difference from TLI to XTI is that TLI said nothing about the format of the options buffer other than it was implementation dependent. Many implementations of TLI used a structure named opthdr, which had only three elements: level, name, and len.

We show two of these XTI structures, pointed to by a netbuf structure that is part of a t_unitdata structure, in Figure 32.3.

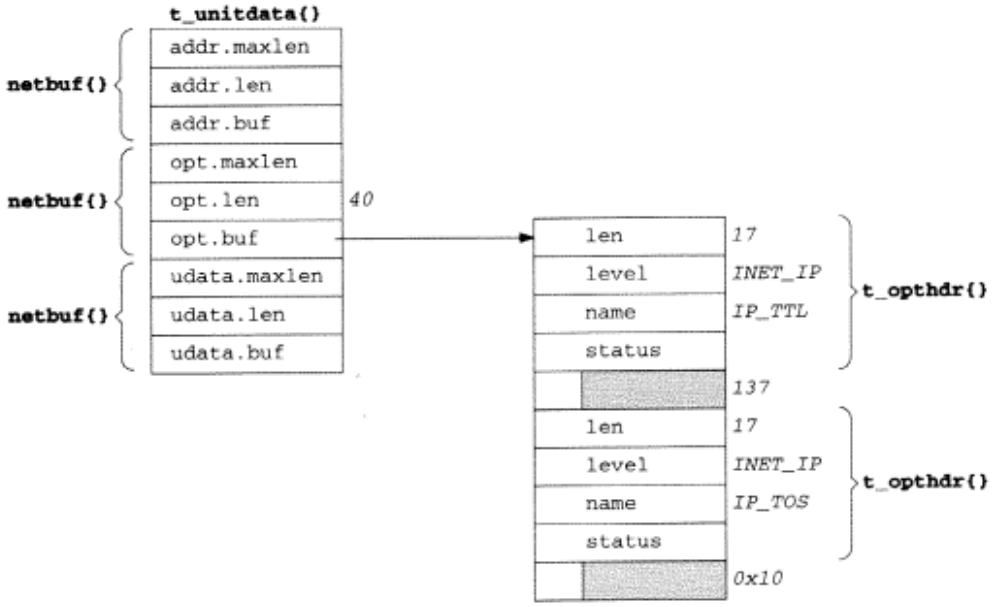


Figure 32.3 Example of two options pointed to by a `netbuf` structure.

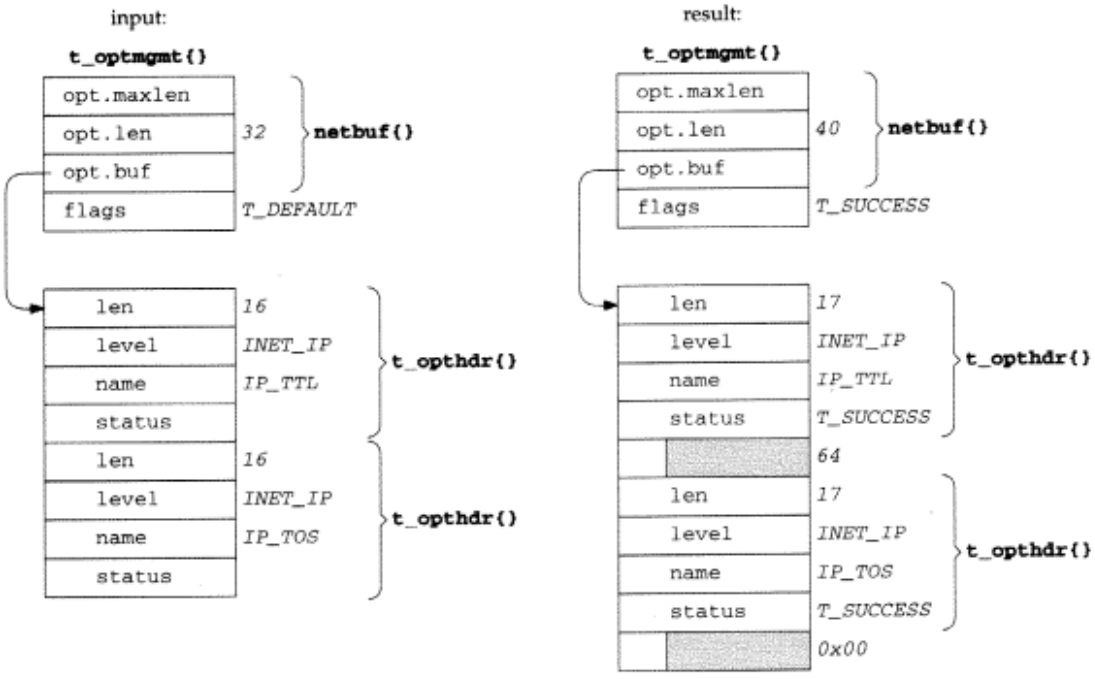


Figure 32.4 Requesting the default value of two options from `t_optmgmt`.

In Figure 32.3 we are specifying the IP TTL as 137 and the IP type-of-service as `0x10` (routine precedence and low delay). Each option value is a 1-byte `u_char` (Figure 32.1) and we show 3 bytes of padding after each value. We also assume here that a `t_uscalar_t` occupies 4 bytes; hence the total size of the option buffer is 40 bytes.

We show another example in Figure 32.4, this time a call to the `t_optmgmt` function, which we describe in Section 32.4. This function takes pointers to two `t_optmgmt` structures: one is the input and the other is the result.

In this example we are asking for the default values of the IP TTL and TOS options (the `flags` of `T_DEFAULT`) so we specify only a `t_opthdr` structure for each option without any value. The result is a copy of the input, with the default values returned after each of the `t_opthdr` structures. In the result structures the `status` member is also filled in.

> Unix 98, but not Posix.1g, defines three macros that can be used when processing a `t_opthdr` structure and the data that follows: `T_OPT_FIRSTHDR`, `T_OPT_NEXTHDR`, and `T_OPT_DATA`. These are similar to the three macros `CMSG_FIRSTHDR`, `CMSG_NXTHDR`, and `CMSG_DATA` used to process ancillary data with sockets, which we described in Section 13.6.

32.3 XTI Options

Most XTI options can be mapped directly to one of the socket options that we described in Chapter 7. Therefore our description of them here is brief. We also note that the header `<xti_inet.h>` must be included to define the constants for all the IP, TCP, and UDP options.

> Note that XTI does not define any way to multicast.

`XTI_DEBUG` Option

This option is similar to the `SO_DEBUG` socket option and normally supported by only TCP. This option is disabled by specifying an option header with no value. By this we mean that the `len` member of the `t_opthdr` structure is just the size of this structure (16 bytes, for example, in Figure 32.4).

`XTI_LINGER` Option

This option is similar to the `SO_LINGER` socket option and is supported by TCP. It specifies what to do when an endpoint is closed. The `t_linger` structure is

```
struct t_linger {
  t_scalar_t  l_onoff;   /* T_NO, T_YES */
  t_scalar_t  l_linger;  /* T_UNSPEC (use default), T_INFINITE,
                            or linger time in seconds */
};
```

We specified in Figure 32.1 that this option is an absolute value, but only the value of

l_onoff is an absolute requirement; the value of l_linger is not an absolute requirement. That is, the implementation can place lower and upper limits on the linger time itself.

Unlike the SO_LINGER socket option, XTI_LINGER is not used to send an RST. t_snddis sends an RST.

XTI_RCVBUF and XTI_RCVLOWAT Options

These two options are similar to the SO_RCVBUF and SO_RCVLOWAT socket options. The first option specifies the size of the endpoint's receive buffer and the second the receive buffer low-water mark used with poll or select.

Figure 32.1 does not consider the XTI_RCVBUF option end-to-end, but with TCP's long fat pipe support (RFC 1323 [Jacobson, Braden, and Borman 1992]), this option does indeed have end-to-end significance since it affects TCP's window scale option that is negotiated with the three-way handshake.

XTI_SNDBUF and XTI_SNDLOWAT Options

These two options are similar to the SO_SNDBUF and SO_SNDLOWAT socket options. The first option specifies the size of the endpoint's send buffer and the second the send buffer low-water mark that is used with poll or select.

T_IP_BROADCAST Option

This option is similar to the SO_BROADCAST socket option. The value of this option is either T_YES or T_NO.

T_IP_DONTROUTE Option

This option is similar to the SO_DONTROUTE socket option. The value of this option is either T_YES or T_NO.

T_IP_OPTIONS Option

This option is similar to the IP_OPTIONS socket option. The value of this option is used as the IPv4 header options. An example of these options is provided in Chapter 24. The option is disabled if specified with no value (i.e., only a t_opthdr structure).

Calling t_optmgmt with a request of T_CURRENT returns the current value of the IP options that will be used in outgoing datagrams.

T_IP_REUSEADDR Option

This option is similar to the SO_REUSEADDR socket option. The value of this option is either T_YES or T_NO.

T_IP_TOS Option

This option is similar to the IP_TOS socket option. The option value is a combination of the IPv4 precedence field, from the values shown in Figure 32.5, with the IPv4 type-of-service field, from the values shown in Figure 32.6.

Constant	Value
T_ROUTINE	0
T_PRIORITY	1
T_IMMEDIATE	2
T_FLASH	3
T_OVERRIDEFLASH	4
T_CRITIC_ECP	5
T_INETCONTROL	6
T_NETCONTROL	7

Figure 32.5 IPv4 precedence values used with T_IP_TOS option.

Constant	Description
T_NOTOS	normal
T_LDELAY	minimize delay
T_HITHRPT	maximize throughput
T_HIREL	maximize reliability
T_LOCOST	minimize cost

Figure 32.6 IPv4 type-of-service values used with T_IP_TOS option.

The macro SET_TOS (defined by including the <xti.h> header) combines its first argument, a precedence value from Figure 32.5, with its second argument, a type-of-service value from Figure 32.6, and the result should be used with this XTI option.

Calling t_optmgmt with a request of T_CURRENT returns the current value of the option that will be used in outgoing datagrams.

T_IP_TTL Option

This option is similar to the IP_TTL socket option. The value of the option is the IPv4 time-to-live field. This option may be set to specify the value used in outgoing datagrams. There is no way, however, to obtain the TTL from a received datagram.

T_TCP_KEEPALIVE Option

This option is similar to the SO_KEEPALIVE socket option and controls the sending of keepalive packets on a TCP connection. This XTI option uses the following structure:

```
struct t_kpalive {
  t_scalar_t  kp_onoff;   /* T_NO (disable), T_YES (enable), or
                             T_YES|T_GARBAGE (enable & send garbage byte) */
  t_scalar_t  kp_timeout; /* timeout in minutes; T_UNSPEC means default */
};
```

This option is similar to the XTI_LINGER option in that the value of kp_onoff is an absolute requirement but the value of kp_timeout is not an absolute requirement.

> Sending a garbage byte should not be required and in fact T_GARBAGE was removed from Unix 98. The use of the garbage byte is discussed on p. 335 of TCPv1.

T_TCP_MAXSEG Option

This option is similar to the TCP_MAXSEG socket option. This option is read-only and returns the maximum segment size (MSS) for a TCP endpoint. Since this option is read-only, its value cannot be an absolute requirement.

T_TCP_NODELAY Option

This option is similar to the TCP_NODELAY socket option. The value of this option is either T_YES (disable the Nagle algorithm) or T_NO (the default, the Nagle algorithm is enabled). We say more about the Nagle algorithm in Section 7.9.

T_UDP_CHECKSUM Option

This XTI option is one of the end-to-end options; therefore it is always returned by t_rcvudata if received options are requested (i.e., if opt.maxlen is nonzero). The value of this option is either T_YES or T_NO.

> This option should *never* be enabled and providing the ability for an application to disable the sending of UDP checksums for an endpoint is a mistake. Examples exist of data corruption when UDP checksums are disabled and there is no reason to ever disable UDP checksums. The only use of this option should be to detect whether a peer has UDP checksums enabled.

32.4 t_optmgmt Function

The t_optmgmt function lets us perform the following operations with regard to XTI options:

- check whether one or more options are supported,
- obtain the default value of one or more options,
- obtain the current value of one or more options, and
- negotiate values for one or more options.

```
#include <xti.h>

int t_optmgmt(int fd, const struct t_optmgmt *request, struct t_optmgmt *result);
```

Returns: 0 if OK, –1 on error

We specify our request as a `t_optmgmt` structure, and one of these structures is returned as the result. If we are not interested in the result, we set the `maxlen` member of the structure pointed to by *result* to 0. We showed an example of these two structures in Figure 32.4.

```
struct t_optmgmt {
  struct netbuf  opt;     /* one or more t_opthdr structures */
  t_scalar_t     flags;   /* action on input, result on output */
};
```

The `flags` member of the *request* structure specifies the action desired by the caller:

`T_CHECK`	check whether the options are supported,
`T_DEFAULT`	return the default values of the options,
`T_CURRENT`	return the current values of the options, and
`T_NEGOTIATE`	negotiate values for the options.

We will examine each of these four operations in the following sections.

We are able to specify multiple options in a single call to `t_optmgmt` as shown in Figure 32.4. But if we do this all options must specify the same `level`. This is OK in Figure 32.4 because the `level` of both options is `T_INET_IP`. There is another complication when negotiating new values for multiple options in a single call to this function: the returned `flags` contains the worst single result, even though each option contains its own `status` return. To avoid these complications, it is simplest to manipulate just one option at a time in each call to `t_optmgmt`.

This XTI function corresponds to the `getsockopt` and `setsockopt` functions.

32.5 Checking If an Option Is Supported and Obtaining the Default

Our first example of XTI options is to check which of the options listed in Figure 32.1 are supported on our system, and for each supported option, to print its default value. Figure 32.7 shows our program.

xtiopt/checkopts.c

```
 1 #include    "unpxti.h"

 2 struct xti_opts {
 3   char          *opt_str;
 4   t_uscalar_t    opt_level;
 5   t_uscalar_t    opt_name;
 6   char        *(*opt_val_str)(struct t_opthdr *);
 7 } xti_opts[] = {
 8     "XTI_DEBUG",         XTI_GENERIC,    XTI_DEBUG,      xti_str_uscalard,
 9     "XTI_LINGER",        XTI_GENERIC,    XTI_LINGER,     xti_str_linger,
10     "XTI_RCVBUF",        XTI_GENERIC,    XTI_RCVBUF,     xti_str_uscalard,
11     "XTI_RCVLOWAT",      XTI_GENERIC,    XTI_RCVLOWAT,   xti_str_uscalard,
12     "XTI_SNDBUF",        XTI_GENERIC,    XTI_SNDBUF,     xti_str_uscalard,
13     "XTI_SNDLOWAT",      XTI_GENERIC,    XTI_SNDLOWAT,   xti_str_uscalard,
14     "T_IP_BROADCAST",    T_INET_IP,      T_IP_BROADCAST, xti_str_uiyn,
15     "T_IP_DONTROUTE",    T_INET_IP,      T_IP_DONTROUTE, xti_str_uiyn,
16     "T_IP_OPTIONS",      T_INET_IP,      T_IP_OPTIONS,   xti_str_uchard,
```

```
17      "T_IP_REUSEADDR",   T_INET_IP,      T_IP_REUSEADDR, xti_str_uiyn,
18      "T_IP_TOS",         T_INET_IP,      T_IP_TOS,       xti_str_ucharx,
19      "T_IP_TTL",         T_INET_IP,      T_IP_TTL,       xti_str_uchard,
20      "T_TCP_KEEPALIVE",  T_INET_TCP,     T_TCP_KEEPALIVE,xti_str_kpalive,
21      "T_TCP_MAXSEG",     T_INET_TCP,     T_TCP_MAXSEG,   xti_str_uscalard,
22      "T_TCP_NODELAY",    T_INET_TCP,     T_TCP_NODELAY,  xti_str_usyn,
23      "T_UDP_CHECKSUM",   T_INET_UDP,     T_UDP_CHECKSUM, xti_str_usyn,
24      NULL,               0,              0,              NULL
25  };
26  int
27  main(int argc, char **argv)
28  {
29      int     fd;
30      struct t_opthdr *topt;
31      struct t_optmgmt *req, *ret;
32      struct xti_opts *ptr;
33      if (argc != 2)
34          err_quit("usage: checkopts <device>");
35      fd = T_open(argv[1], O_RDWR, NULL);
36      T_bind(fd, NULL, NULL);
37      req = T_alloc(fd, T_OPTMGMT, T_ALL);
38      ret = T_alloc(fd, T_OPTMGMT, T_ALL);
39      for (ptr = xti_opts; ptr->opt_str != NULL; ptr++) {
40          topt = (struct t_opthdr *) req->opt.buf;
41          topt->level = ptr->opt_level;
42          topt->name = ptr->opt_name;
43          topt->len = sizeof(struct t_opthdr);
44          req->opt.len = topt->len;
45          req->flags = T_CHECK;
46          printf("%s: ", ptr->opt_str);
47          if (t_optmgmt(fd, req, ret) < 0) {
48              err_xti_ret("t_optmgmt error");
49          } else {
50              topt = (struct t_opthdr *) ret->opt.buf;
51              printf("%s", xti_str_flags(topt->status));
52              if (topt->status == T_SUCCESS || topt->status == T_READONLY) {
53                  req->flags = T_DEFAULT;
54                  if (t_optmgmt(fd, req, ret) < 0) {
55                      err_xti_ret("t_optmgmt error for T_DEFAULT");
56                  } else {
57                      topt = (struct t_opthdr *) ret->opt.buf;
58                      printf(", default = %s", (*ptr->opt_val_str) (topt));
59                  }
60              }
61              printf("\n");
62          }
63      }
64      exit(0);
65  }
```

xtiopt/checkopts.c

Figure 32.7 Check which XTI options are supported.

2-25 We define and initialize a structure defining all the XTI options from Figure 32.1. The final member of each array element is a pointer to a function that prints the value of the option. We need one function for each of the different option types. We do not show the source code for all these functions here.

Open device

35-38 We take the device name as a command-line argument and open the device. This lets us run the program twice, once for `/dev/tcp` and once for `/dev/udp`, as we expect different options to be supported by each provider. We bind any local address to the endpoint, because most calls to `t_optmgmt` require that the endpoint be bound. We also allocate two `t_optmgmt` structures, one for our request and one for the function's reply.

Call `t_optmgmt` for request of `T_CHECK`

39-48 We call `t_optmgmt` for each option in our `xti_opts` array, with a request flag of `T_CHECK`. We fill in our `req` structure, building a single `t_opthdr` structure in the `opt` buffer (Section 32.2). This structure contains just a `t_opthdr` structure, without any data (e.g., similar to the left side of Figure 32.4).

Call `t_optmgmt` for request of `T_DEFAULT`

49-62 If the first call to `t_optmgmt` succeeds, we print the `status` of the option. If the `status` is `T_SUCCESS` or `T_READONLY`, we call `t_optmgmt` again, this time with a request of `T_DEFAULT`. If this second call succeeds, we call the function pointed to by the `opt_val_str` member of our structure in Figure 32.7 to print the default value. When we call `t_optmgmt` the second time, we change only the `flags` member of our request structure. Since the pointer to this structure in the function prototype for `t_optmgmt` has the `const` qualifier, we know the structure was not changed by the first call.

We now run the program two times under AIX 4.2: first for TCP and then for UDP. Notice that AIX uses the device names `/dev/xti/tcp` and `/dev/xti/udp`.

```
aix % checkopts /dev/xti/tcp
XTI_DEBUG: T_SUCCESS, default = 0
XTI_LINGER: T_SUCCESS, default = T_NO, 0 sec
XTI_RCVBUF: T_SUCCESS, default = 16384
XTI_RCVLOWAT: T_SUCCESS, default = 1
XTI_SNDBUF: T_SUCCESS, default = 16384
XTI_SNDLOWAT: T_SUCCESS, default = 4096
T_IP_BROADCAST: T_SUCCESS, default = T_NO
T_IP_DONTROUTE: T_SUCCESS, default = T_NO
T_IP_OPTIONS: T_SUCCESS, default = 0 (length of value)
T_IP_REUSEADDR: T_SUCCESS, default = T_NO
T_IP_TOS: T_SUCCESS, default = 0x00
T_IP_TTL: T_SUCCESS, default = 0
T_TCP_KEEPALIVE: T_SUCCESS, default = T_NO, T_UNSPEC
T_TCP_MAXSEG: T_READONLY, default = 512
T_TCP_NODELAY: T_SUCCESS, default = T_NO
T_UDP_CHECKSUM: t_optmgmt error: incorrect option format
```

```
aix % checkopts /dev/xti/udp
XTI_DEBUG: T_SUCCESS, default = 0
XTI_LINGER: T_SUCCESS, default = T_NO, 0 sec
XTI_RCVBUF: T_SUCCESS, default = 41600
XTI_RCVLOWAT: T_SUCCESS, default = 1
XTI_SNDBUF: T_SUCCESS, default = 9216
XTI_SNDLOWAT: T_SUCCESS, default = 4096
T_IP_BROADCAST: T_SUCCESS, default = T_NO
T_IP_DONTROUTE: T_SUCCESS, default = T_NO
T_IP_OPTIONS: T_SUCCESS, default = 0 (length of value)
T_IP_REUSEADDR: T_SUCCESS, default = T_NO
T_IP_TOS: T_SUCCESS, default = 0x00
T_IP_TTL: T_SUCCESS, default = 0
T_TCP_KEEPALIVE: t_optmgmt error: incorrect option format
T_TCP_MAXSEG: t_optmgmt error: incorrect option format
T_TCP_NODELAY: t_optmgmt error: incorrect option format
T_UDP_CHECKSUM: T_NOTSUPPORT
```

The supported values are as we expect, other than the `T_IP_TTL` value. The `T_UDP_CHECKSUM` option that is not supported by TCP, and the three TCP options that are not supported by UDP cause `t_optmgmt` to return an error of `TBADOPT`. The UDP provider understands the `T_UDP_CHECKSUM` option but returns `T_NOTSUPPORT` since it is not supported. The string "(length of value)" that is printed for `T_IP_OPTIONS` indicates that the `len` member that was returned was 0, so there is no value to output.

32.6 Getting and Setting XTI Options

We now show examples of getting and setting XTI options. We define two functions of our own, `xti_getopt` and `xti_setopt`, that have identical calling sequences to `getsockopt` and `setsockopt` (Section 7.2).

```
#include "unpxti.h"

int xti_getopt(int fd, int level, int name, void *optval, socklen_t *optlen);

int xti_setopt(int fd, int level, int name, const void *optval, socklen_t optlen);
```

Both return: 0 if OK, −1 on error

These functions can simplify our XTI programs, since each comprises about 20−30 lines of C code.

`xti_getopt` Function

To fetch the current value of an XTI option we call `t_optmgmt` with the `flags` member of the request structure set to `T_CURRENT`. Figure 32.8 shows our `xti_getopt` function.

libxti/xti_getopt.c

```
 1 #include    "unpxti.h"

 2 int
 3 xti_getopt(int fd, int level, int name, void *optval, socklen_t *optlenp)
 4 {
 5     int     rc, len;
 6     struct t_optmgmt *req, *ret;
 7     struct t_opthdr *topt;

 8     req = T_alloc(fd, T_OPTMGMT, T_ALL);
 9     ret = T_alloc(fd, T_OPTMGMT, T_ALL);
10     if (req->opt.maxlen == 0)
11         err_quit("xti_getopt: opt.maxlen == 0");

12     topt = (struct t_opthdr *) req->opt.buf;
13     topt->level = level;
14     topt->name = name;
15     topt->len = sizeof(struct t_opthdr);     /* just a t_opthdr() */
16     req->opt.len = topt->len;

17     req->flags = T_CURRENT;
18     if (t_optmgmt(fd, req, ret) < 0) {
19         T_free(req, T_OPTMGMT);
20         T_free(ret, T_OPTMGMT);
21         return (-1);
22     }
23     rc = ret->flags;

24     if (rc == T_SUCCESS || rc == T_READONLY) {
25             /* copy back value and length */
26         topt = (struct t_opthdr *) ret->opt.buf;
27         len = topt->len - sizeof(struct t_opthdr);
28         len = min(len, *optlenp);
29         memcpy(optval, topt + 1, len);
30         *optlenp = len;
31     }
32     T_free(req, T_OPTMGMT);
33     T_free(ret, T_OPTMGMT);

34     if (rc == T_SUCCESS || rc == T_READONLY)
35         return (0);
36     return (-1);                /* T_NOTSUPPORT */
37 }
```

libxti/xti_getopt.c

Figure 32.8 `xti_getopt` function: get the current value of an XTI option.

Allocate request and reply structures

8–11 We call `t_alloc` to allocate room for a request structure and a reply structure. We also verify that the size of the options buffer is nonzero.

> Older implementations of TLI often used a value of 0 for the size of the TCP options, meaning the application had to allocate its own buffer.

Fill in `t_opthdr` structure

12-16 We fill in a `t_opthdr` structure with the option's *level* and *name*. We do not specify any value in the request structure because this is not required when fetching the current value of an option.

Call `t_optmgmt` and return option value

17-31 We call `t_optmgmt` and then save the `flags` member in the returned structure. If the return value was `T_SUCCESS` or `T_READONLY`, we copy back the value of the option and its size. (The pointer expression `topt+1` points to the returned option value, which immediately follows the `t_opthdr` structure.) The final argument to our function is a value–result argument and we are careful not to overflow the caller's buffer (in case it is too small).

Free memory and return

32-36 We free the memory allocated by `t_alloc` and return 0 on success or −1 on an error.

`xti_setopt` Function

To set the value of an XTI option we call `t_optmgmt` with the `flags` member of the request structure set to `T_NEGOTIATE`. Figure 32.9 shows our `xti_setopt` function. This function is similar to the `xti_getopt` function in Figure 32.8 with a few exceptions.

Copy caller's value

12-19 We copy the caller's option value into the buffer that we build, placing it immediately following the `t_opthdr` structure.

Call `t_opmgmt`

20-26 The request `flags` is now `T_NEGOTIATE` for `t_optmgmt`.

Free memory and return

27-31 If the option value is not an absolute requirement the return value is `T_PARTSUCCESS`, which is OK.

XTI lets us set an option and fetch its value in a single call to `t_optmgmt`. This might be useful for an option whose value is not an absolute requirement (e.g., the send and receive buffer sizes). Using our functions requires a call to `xti_setopt` followed by a call to `xti_getopt`. We could have defined a function that does both, but the extra call to `xti_getopt` would rarely be the bottleneck of an application.

libxti/xti_setopt.c

```
1 #include    "unpxti.h"

2 int
3 xti_setopt(int fd, int level, int name, void *optval, socklen_t optlen)
4 {
5     int     rc;
6     struct t_optmgmt *req, *ret;
7     struct t_opthdr *topt;

8     req = T_alloc(fd, T_OPTMGMT, T_ALL);
9     ret = T_alloc(fd, T_OPTMGMT, T_ALL);
10    if (req->opt.maxlen == 0)
11        err_quit("xti_setopt: req.opt.maxlen == 0");

12    topt = (struct t_opthdr *) req->opt.buf;
13    topt->level = level;
14    topt->name = name;
15    topt->len = sizeof(struct t_opthdr) + optlen;
16    if (topt->len > req->opt.maxlen)
17        err_quit("optlen too big");
18    req->opt.len = topt->len;
19    memcpy(topt + 1, optval, optlen);   /* copy option value */

20    req->flags = T_NEGOTIATE;
21    if (t_optmgmt(fd, req, ret) < 0) {
22        T_free(req, T_OPTMGMT);
23        T_free(ret, T_OPTMGMT);
24        return (-1);
25    }
26    rc = ret->flags;

27    T_free(req, T_OPTMGMT);
28    T_free(ret, T_OPTMGMT);

29    if (rc == T_SUCCESS || rc == T_PARTSUCCESS)
30        return (0);
31    return (-1);                /* T_FAILURE, T_NOTSUPPORT, T_READONLY */
32 }
```

libxti/xti_setopt.c

Figure 32.9 `xti_setopt` function: set the value of an XTI option.

Example

We now use the two functions that were just shown. The program in Figure 32.10 fetches the current value of TCP's maximum segment size, sets the size of the send buffer to 65536, and then fetches and prints the size of the send buffer. If we compile and execute this program, its output is

```
aix % getsetopt
TCP mss = 512
send buffer size = 65536
```

xtiopt/getsetopt.c

```
 1 #include    "unpxti.h"

 2 int
 3 main(int argc, char **argv)
 4 {
 5     int     fd;
 6     socklen_t optlen;
 7     t_uscalar_t mss, sendbuff;

 8     fd = T_open(XTI_TCP, O_RDWR, NULL);
 9     T_bind(fd, NULL, NULL);

10     optlen = sizeof(mss);
11     Xti_getopt(fd, T_INET_TCP, T_TCP_MAXSEG, &mss, &optlen);
12     printf("TCP mss = %d\n", mss);

13     sendbuff = 65536;
14     Xti_setopt(fd, XTI_GENERIC, XTI_SNDBUF, &sendbuff, sizeof(sendbuff));

15     optlen = sizeof(sendbuff);
16     Xti_getopt(fd, XTI_GENERIC, XTI_SNDBUF, &sendbuff, &optlen);
17     printf("send buffer size = %d\n", sendbuff);

18     exit(0);
19 }
```

xtiopt/getsetopt.c

Figure 32.10 Example of our `xti_getopt` and `xti_setopt` functions.

32.7 Summary

XTI options are negotiated with the possibility that the provider returns a different value than we asked for. Although XTI option processing is very general, the simplest approach is to define two basic functions that look like `getsockopt` and `setsockopt` and call them from our application.

33

Streams

33.1 Introduction

Before describing some of the additional features of XTI, such as signal-driven I/O and out-of-band data, we need to understand some implementation details. XTI and the networking protocols are normally implemented using the streams system, as is the terminal I/O system on most SVR4-derived kernels.

In this chapter we provide an overview of the streams system and the functions used by an application to access a stream. Our goal is to understand the implementation of networking protocols within the streams framework. We also develop a simple TCP client using TPI, the interface into the transport layer that both XTI and sockets normally use on a system based on streams. Additional information on streams, including information on writing kernel routines that utilize streams, can be found in [Rago 1993].

Streams were designed by Dennis Ritchie [Ritchie 1984] and first made widely available with SVR3 in 1986. They have never been standardized by Posix. The basic streams functions are required by Unix 98: `getmsg`, `getpmsg`, `putmsg`, `putpmsg`, `fattach`, and all of the streams `ioctl` commands. XTI is often implemented using streams. Any system derived from System V should provide streams, but the various 4.xBSD releases do not provide streams.

The streams system is often written as STREAMS, but it is not even an acronym, so we write it as just *streams*.

Be careful to distinguish between the stream I/O system that we are describing in this chapter, versus "standard I/O streams." The latter term is used when talking about the standard I/O library (e.g., functions such as `fopen`, `fgets`, `printf`, and the like).

33.2 Overview

Streams provide a full-duplex connection between a process and a *driver*, as shown in Figure 33.1. Although we describe the bottom box as a driver, this need not be associated with a hardware device; it can also be a pseudo-device driver (e.g., a software driver).

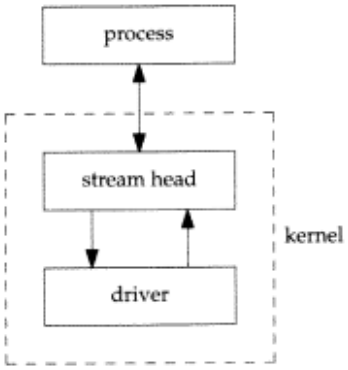


Figure 33.1 A stream shown between a process and a driver.

The *stream head* consists of the kernel routines that are invoked when the application makes a system call for a streams descriptor (e.g., `read`, `putmsg`, `ioctl`, and the like).

A process can dynamically add and remove intermediate processing *modules* between the stream head and the driver. A module performs some type of filtering on the messages going up and down a stream. We show this in Figure 33.2.

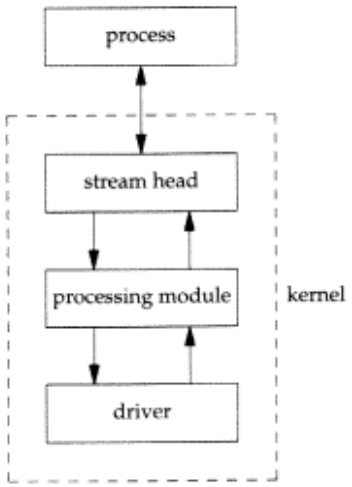


Figure 33.2 A stream with a processing module.

Any number of modules can be pushed onto a stream. When we say *push*, we mean that each new module gets inserted just below the stream head.

A special type of pseudo-device driver is a *multiplexor*, which accepts data from multiple sources. A streams-based implementation of the TCP/IP protocol suite, as found on SVR4 for example, could be as shown in Figure 33.3.

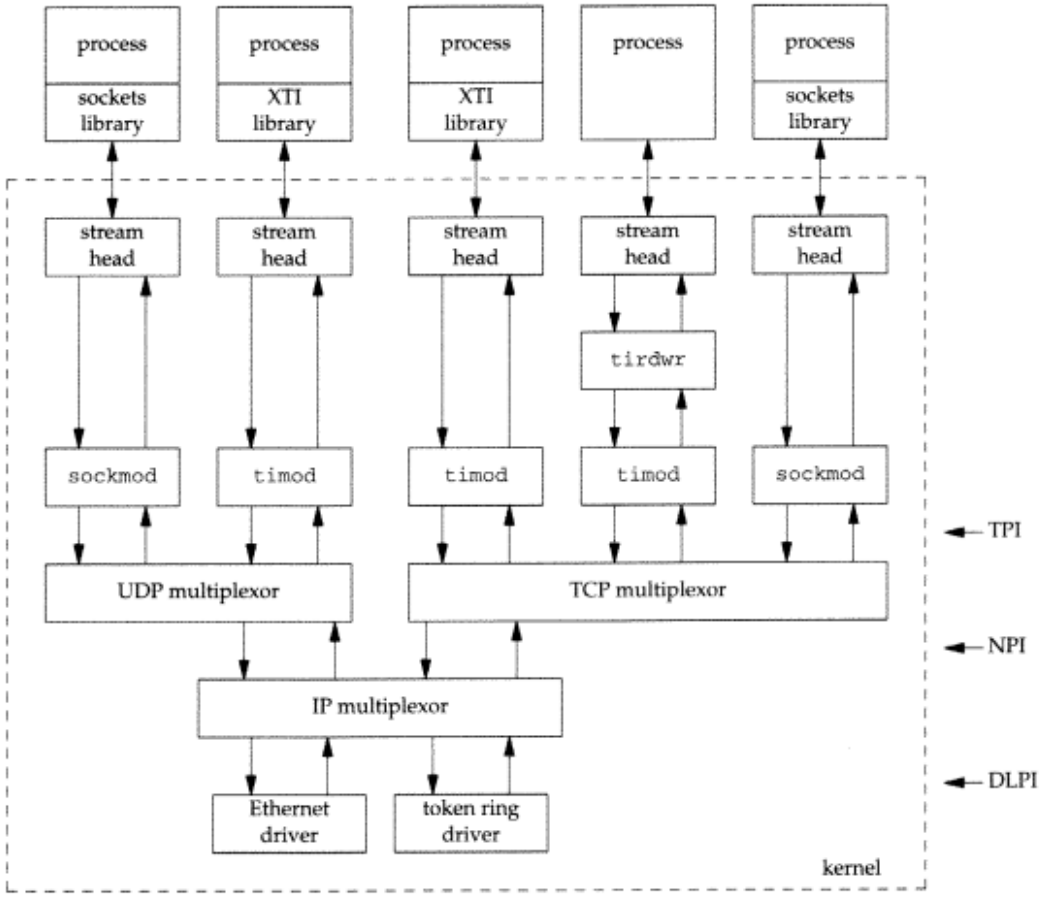


Figure 33.3 Simplified streams implementations of TCP/IP using streams.

- When a socket is created, the module `sockmod` is pushed onto the stream by the sockets library. It is the combination of the sockets library and the `sockmod` streams module that provides the sockets API to the process.
- When an XTI endpoint is created, the module `timod` is pushed onto the stream by the XTI library. It is the combination of the XTI library and the `timod` streams module that provides the XTI API to the process.
- We mentioned in Section 28.12 that the streams module `tirdwr` must normally be pushed onto a stream to use `read` and `write` with an XTI endpoint. The middle process using TCP in Figure 33.3 has done this. This process has probably abandoned the use of XTI by doing this, so we have removed the XTI library.

- Three service interfaces define the format of the networking messages exchanged up and down a stream. TPI, the *Transport Provider Interface* [Unix International 1992b], defines the interface provided by a transport-layer provider (e.g., TCP and UDP) to the modules above it. NPI, the *Network Provider Interface* [Unix International 1992a], defines the interface provided by a network-layer provider (e.g., IP). DLPI is the *Data Link Provider Interface* [Unix International 1991]. An alternate reference for TPI and DLPI, which contains sample C code, is [Rago 1993].

  The claim is regularly made to Usenet that "in a streams environment sockets are implemented on top of TLI (XTI)." This is false. As we can see in Figure 33.3, both sockets and XTI are implemented on top of TPI. This claim is often followed with "therefore TLI (XTI) is faster than sockets." This is also false. The TCP, UDP, and IP layers are the same, regardless of whether XTI or sockets are used. What changes is the user library and whether `timod` or `sockmod` is on the stream. But the author is not aware of any numbers comparing these libraries and modules, For the bottleneck of most applications (data transfer), the code path is probably similar for XTI and sockets, unless special optimizations have been applied to one and not the other.

Each component in a stream—the stream head, all processing modules, and the driver—contain at least one pair of *queues*: a write queue and a read queue. We show this in Figure 33.4.

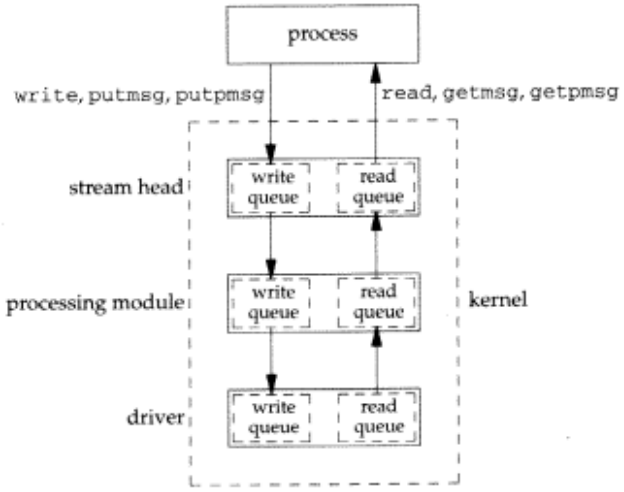


Figure 33.4 Each component in a stream has at least one pair of queues.

Message Types

Streams messages can be categorized as *high priority*, *priority band*, or *normal*. There are 256 different priority bands, between 0 and 255, with normal messages in band 0. The priority of a streams message is used for both queueing and flow control. By convention, high-priority messages are unaffected by flow control.

Figure 33.5 shows the ordering of the messages on a given queue.

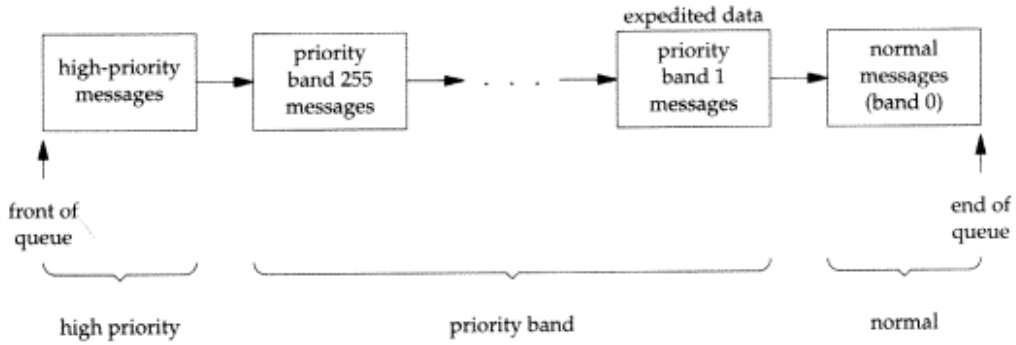


Figure 33.5 Ordering of streams messages on a queue, based on priority.

Although the streams system supports 256 different priority bands, networking protocols often use band 1 for expedited data and band 0 for normal data.

> TCP's out-of-band data is not considered true expedited data by TPI. Indeed, TCP uses band 0 for both normal data and its out-of-band data (as we will verify in Figure C.4). The use of band 1 for expedited data is for protocols in which the expedited data (not just the urgent pointer, as in TCP) is sent ahead of normal data.
>
> Beware of the term *normal*. In releases before SVR4 there were no priority bands; there were just normal messages and priority messages. SVR4 implemented priority bands, requiring the `getpmsg` and `putpmsg` functions, which we describe shortly. The older priority messages were renamed high priority. The question is what to call the new messages, with priority bands between 1 and 255. Common terminology [Rago 1993] is to refer to everything other than high-priority messages as normal-priority messages and then subdivide these normal-priority messages into priority bands. The term *normal message* should always refer to a message with a band of 0.

Although we talk about normal-priority messages and high-priority messages, there are about a dozen normal-priority message types and around 18 high-priority message types. From an application's perspective, and the `getmsg` and `putmsg` functions that we are about to describe, we are interested in only three different types of messages: `M_DATA`, `M_PROTO`, and `M_PCPROTO` (`PC` stands for "priority control" and implies a high-priority message). Figure 33.6 shows how these three different message types are generated by the `write` and `putmsg` functions.

Function	Control?	Data?	Flags	Message type generated
`write`		yes		`M_DATA`
`putmsg`	no	yes	0	`M_DATA`
`putmsg`	yes	don't care	0	`M_PROTO`
`putmsg`	yes	don't care	`MSG_HIPRI`	`M_PCPROTO`

Figure 33.6 Streams message types generated by `write` and `putmsg`.

We will see what we mean by control, data, and flags in our description of the `putmsg` function in the next section.

33.3 `getmsg` and `putmsg` Functions

The data transferred up and down a stream consists of messages and each message contains *control*, *data*, or both. If we use `read` and `write` on a stream, these transfer only data. To allow a process to read and write both data and control information, two new functions were added.

```
#include <stropts.h>

int getmsg(int fd, struct strbuf *ctlptr, struct strbuf *dataptr, int *flagsp);

int putmsg(int fd, const struct strbuf *ctlptr,
           const struct strbuf *dataptr, int flags);
```

Both return: nonnegative value if OK (see text), –1 on error

Both the control and data portions of the message are described by a `strbuf` structure:

```
struct strbuf {
  int    maxlen;     /* maximum size of buf */
  int    len;        /* actual amount of data in buf */
  char  *buf;        /* data */
};
```

Note the similarity between the `strbuf` structure and the `netbuf` structure. The names of the three elements in each structure are identical.

But the two lengths in the `netbuf` structure are unsigned integers, while the two lengths in the `strbuf` structure are signed integers. The reason is that some of the streams functions use a `len` or `maxlen` value of –1 to indicate something special.

We can send only control information, only data, or both using `putmsg`. To indicate the absence of control information we can either specify *ctlptr* as a null pointer, or set *ctlptr->len* to –1. The same technique is used to indicate no data.

If there is no control information, an `M_DATA` message is generated by `putmsg` (Figure 33.6); otherwise either an `M_PROTO` or an `M_PCPROTO` message is generated, depending on the *flags*. The *flags* argument to `putmsg` is 0 for a normal message or `RS_HIPRI` for a high-priority message.

The final argument to `getmsg` is a value–result argument. If the integer pointed to by *flagsp* is 0 when the function is called, the first message on the stream is returned (which can be normal or high priority). If the integer value is `RS_HIPRI` when the function is called, the function waits for a high-priority message to arrive at the stream head. In both cases the value stored in the integer pointed to by *flagsp* will be 0 or `RS_HIPRI`, depending on the type of message returned.

Assuming we pass nonnull *ctlptr* and *dataptr* values to `getmsg`, if there is no control information to return (i.e., an `M_DATA` message is being returned), this is indicated by

setting *ctlptr->len* to –1 on return. Similarly, *dataptr->len* is set to –1 if there is no data to return.

The return value from putmsg is 0 if all is OK, or –1 on an error. But getmsg returns 0 only if the entire message was returned to the caller. If the control buffer is too small for all the control information, the return value is MORECTL (which is guaranteed to be nonnegative). Similarly if the data buffer is too small, MOREDATA can be returned. If both are too small, the logical OR of these two flags is returned.

33.4 getpmsg and putpmsg Functions

When support for different priority bands was added to streams with SVR4, the following two variants of getmsg and putmsg were added.

```
#include <stropts.h>

int getpmsg(int fd, struct strbuf *ctlptr,
            struct strbuf *dataptr, int *bandp, int *flagsp);

int putpmsg(int fd, const struct strbuf *ctlptr,
            const struct strbuf *dataptr, int band, int flags);
```

Both return: nonnegative value if OK, –1 on error

The *band* argument to putpmsg must be between 0 and 255, inclusive. If the *flags* argument is MSG_BAND, then a message is generated in the specified priority band. Setting *flags* to MSG_BAND and specifying a band of 0 is equivalent to calling putmsg. If *flags* is MSG_HIPRI, *band* must be 0, and a high-priority message is generated. (Note that this flag is named differently from the RS_HIPRI flag for putmsg.)

The two integers pointed to by *bandp* and *flagsp* are value–result arguments for getpmsg. The integer pointed to by *flagsp* for getpmsg can be MSG_HIPRI (to read a high-priority message), MSG_BAND (to read a message whose priority band is at least equal to the integer pointed to by *bandp*), or MSG_ANY (to read any message). On return the integer pointed to by *bandp* contains the band of the message that was read, and the integer pointed to by *flagsp* contains MSG_HIPRI (if a high-priority message was read) or MSG_BAND (if some other message was read).

33.5 ioctl Function

With streams we again encounter the ioctl function that we described in Chapter 16.

```
#include <stropts.h>

int ioctl(int fd, int request, ... /* void *arg */ );
```

Returns: 0 if OK, –1 on error

The only change from the function prototype shown in Section 16.2 is the headers that must be included when dealing with streams.

There are about 30 requests that affect a stream head. Each request begins with `I_` and they are normally documented on the `streamio` manual page. We showed the `I_PUSH` request in Figure 28.14 when we pushed the `tirdwr` module onto a stream.

When we discuss signal-driven I/O with XTI in Section 34.11 we discuss the `I_SETSIG` request.

33.6 TPI: Transport Provider Interface

In Figure 33.3 we showed that TPI is the service interface into the transport layer from above. Both sockets and XTI use this interface in a streams environment. In Figure 33.3 it is a combination of the sockets library and `sockmod`, along with a combination of the XTI library and `timod` that exchange TPI messages with TCP and UDP.

TPI is a *message-based* interface. It defines the messages that are exchanged up and down a stream between the application (e.g., the XTI or sockets library) and the transport layer: the format of these messages and what operation each message performs. In many instances the application sends a request to the provider (such as "bind this local address") and the provider sends back a response ("OK" or "error"). Some events occur asynchronously at the provider (the arrival of a connection request for a server), causing a message or a signal to be sent up the stream.

We are able to bypass both XTI and sockets and use TPI directly. In this section we rewrite our simple daytime client using TPI, instead of sockets (Figure 1.5) or XTI (Figure 28.13). Using programming languages as an analogy, using sockets or XTI is like programming in a high-level language such as C or Pascal, while using TPI directly is like programming in assembler. We are not advocating the use of TPI directly in real applications. But examining how TPI works and developing this example gives us a better understanding of how the sockets library and the XTI library work in a streams environment.

Figure 33.7 is our `tpi_daytime.h` header.

streams/tpi_daytime.h

```
1 #include    "unpxti.h"
2 #include    <sys/stream.h>
3 #include    <sys/tihdr.h>

4 void    tpi_bind(int, const void *, size_t);
5 void    tpi_connect(int, const void *, size_t);
6 ssize_t tpi_read(int, void *, size_t);
7 void    tpi_close(int);
```

streams/tpi_daytime.h

Figure 33.7 Our `tpi_daytime.h` header.

We need to include one additional streams header along with `<sys/tihdr.h>`, which contains the definitions of the structures for all the TPI messages.

Figure 33.8 is the `main` function for our daytime client.

streams/tpi_daytime.c

```
 1 #include    "tpi_daytime.h"

 2 int
 3 main(int argc, char **argv)
 4 {
 5     int     fd, n;
 6     char    recvline[MAXLINE + 1];
 7     struct sockaddr_in myaddr, servaddr;

 8     if (argc != 2)
 9         err_quit("usage: tpi_daytime <IPaddress>");

10     fd = Open(XTI_TCP, O_RDWR, 0);

11         /* bind any local address */
12     bzero(&myaddr, sizeof(myaddr));
13     myaddr.sin_family = AF_INET;
14     myaddr.sin_addr.s_addr = htonl(INADDR_ANY);
15     myaddr.sin_port = htons(0);

16     tpi_bind(fd, &myaddr, sizeof(struct sockaddr_in));

17         /* fill in server's address */
18     bzero(&servaddr, sizeof(servaddr));
19     servaddr.sin_family = AF_INET;
20     servaddr.sin_port = htons(13);  /* daytime server */
21     Inet_pton(AF_INET, argv[1], &servaddr.sin_addr);

22     tpi_connect(fd, &servaddr, sizeof(struct sockaddr_in));

23     for ( ; ; ) {
24         if ( (n = tpi_read(fd, recvline, MAXLINE)) <= 0) {
25             if (n == 0)
26                 break;
27             else
28                 err_sys("tpi_read error");
29         }
30         recvline[n] = 0;        /* null terminate */
31         fputs(recvline, stdout);
32     }
33     tpi_close(fd);
34     exit(0);
35 }
```

streams/tpi_daytime.c

Figure 33.8 `main` function for our daytime client written to TPI.

Open transport provider, bind local address

10-16 We open the device corresponding to the transport provider (normally `/dev/tcp`). We fill in an Internet socket address structure with `INADDR_ANY` and a port of 0, telling TCP to bind any local address to our endpoint. We call our own function `tpi_bind` (shown shortly) to do the bind.

Fill in server's address, establish connection

17-22 We fill in another Internet socket address structure with the server's IP address (taken from the command line) and port (13). We call our `tpi_connect` function to establish the connection.

Read data from server, copy to standard output

23-33 As in our other daytime clients, we just copy data from the connection to standard output, stopping when we receive the end-of-file from the server (e.g., the FIN). We have written this loop to look like our sockets client (Figure 1.5) instead of our XTI client (Figure 28.13), because our `tpi_read` function will convert an orderly release from the server into a return of 0. We then call our `tpi_close` function to close our endpoint.

Our `tpi_bind` function is shown in Figure 33.9.

Fill in `T_bind_req` structure

16-20 The `<sys/tihdr.h>` header defines the `T_bind_req` structure:

```
struct T_bind_req {
  long          PRIM_type;      /* T_BIND_REQ */
  long          ADDR_length;    /* address length */
  long          ADDR_offset;    /* address offset */
  unsigned long CONIND_number;  /* connect indications requested */
      /* followed by the protocol address for bind */
};
```

All TPI requests are defined as a structure that begins with a long integer type field. We define our own `bind_req` structure that begins with the `T_bind_req` structure, followed by a buffer containing the local address to be bound. TPI says nothing about the contents of this buffer; it is defined by the provider. TCP providers expect this buffer to contain a `sockaddr_in` structure.

We fill in the `T_bind_req` structure, setting the `ADDR_length` member to the size of the address (16 bytes for an Internet socket address structure) and `ADDR_offset` to the byte offset of the address (it immediately follows the `T_bind_req` structure). We are not guaranteed that this location is suitably aligned for the `sockaddr_in` structure that is stored there, so we call `memcpy` to copy the caller's structure into our `bind_req` structure. We set `CONIND_number` to 0, because we are a client, not a server.

Call `putmsg`

21-23 TPI requires that the structure that we just built be passed to the provider as one `M_PROTO` message. We therefore call `putmsg` specifying our `bind_req` structure as the control information, with no data and with a flag of 0.

Call `getmsg` to read high-priority message

24-30 The response to our `T_BIND_REQ` request will be either a `T_BIND_ACK` message or a `T_ERROR_ACK` message. These acknowledgment messages are sent as high-priority messages (`M_PCPROTO`) so we read them using `getmsg` with a flag of `RS_HIPRI`. Since the reply is a high-priority message, it will bypass any normal-priority messages on the stream.

streams/tpi_bind.c

```
 1 #include    "tpi_daytime.h"

 2 void
 3 tpi_bind(int fd, const void *addr, size_t addrlen)
 4 {
 5     struct {
 6         struct T_bind_req msg_hdr;
 7         char    addr[128];
 8     } bind_req;
 9     struct {
10         struct T_bind_ack msg_hdr;
11         char    addr[128];
12     } bind_ack;
13     struct strbuf ctlbuf;
14     struct T_error_ack *error_ack;
15     int     flags;

16     bind_req.msg_hdr.PRIM_type = T_BIND_REQ;
17     bind_req.msg_hdr.ADDR_length = addrlen;
18     bind_req.msg_hdr.ADDR_offset = sizeof(struct T_bind_req);
19     bind_req.msg_hdr.CONIND_number = 0;
20     memcpy(bind_req.addr, addr, addrlen);   /* sockaddr_in{} */

21     ctlbuf.len = sizeof(struct T_bind_req) + addrlen;
22     ctlbuf.buf = (char *) &bind_req;
23     Putmsg(fd, &ctlbuf, NULL, 0);

24     ctlbuf.maxlen = sizeof(bind_ack);
25     ctlbuf.len = 0;
26     ctlbuf.buf = (char *) &bind_ack;
27     flags = RS_HIPRI;
28     Getmsg(fd, &ctlbuf, NULL, &flags);
29     if (ctlbuf.len < (int) sizeof(long))
30         err_quit("bad length from getmsg");

31     switch (bind_ack.msg_hdr.PRIM_type) {
32     case T_BIND_ACK:
33         return;

34     case T_ERROR_ACK:
35         if (ctlbuf.len < (int) sizeof(struct T_error_ack))
36             err_quit("bad length for T_ERROR_ACK");
37         error_ack = (struct T_error_ack *) &bind_ack.msg_hdr;
38         err_quit("T_ERROR_ACK from bind (%d, %d)",
39                  error_ack->TLI_error, error_ack->UNIX_error);

40     default:
41         err_quit("unexpected message type: %d", bind_ack.msg_hdr.PRIM_type);
42     }
43 }
```

streams/tpi_bind.c

Figure 33.9 `tpi_bind` function: bind a local address to an endpoint.

These two messages are

```
struct T_bind_ack {
  long           PRIM_type;      /* T_BIND_ACK */
  long           ADDR_length;    /* address length */
  long           ADDR_offset;    /* address offset */
  unsigned long  CONIND_number; /* connect ind to be queued */
      /* followed by the bound address */
};

struct T_error_ack {
  long           PRIM_type;      /* T_ERROR_ACK */
  long           ERROR_prim      /* primitive in error */
  long           TLI_error;      /* TLI error code */
  long           UNIX_error;     /* UNIX error code */
};
```

All these messages begin with the type, so we can read the reply assuming it is a `T_BIND_ACK` message, look at the type, and process the message accordingly. We do not expect any data from the provider, so we specify a null pointer as the third argument to `getmsg`.

> When we verify that the amount of control information returned is at least the size of a long integer, we must be careful to cast the `sizeof` value to an integer. The `sizeof` operator returns an unsigned integer value but it is possible for the returned `len` field to be –1. But since the less-than comparison is comparing a signed value on the left to an unsigned value on the right, the compiler casts the signed value to an unsigned value. On a twos-complement architecture, –1 considered as an unsigned value is very large, causing –1 to be greater than 4 (if we assume a long integer occupies 4 bytes).

Process reply

31-33 If the reply is `T_BIND_ACK`, the bind was successful, and we return. The actual address that was bound to the endpoint is returned in the `addr` member of our `bind_ack` structure, which we ignore.

34-39 If the reply is `T_ERROR_ACK`, we verify that the entire message was received and then print the three return values in the structure. In this simple program we terminate when an error occurs; we do not return to the caller.

We can see these errors from the bind request by changing our `main` function to bind some port other than 0. For example, if we try to bind port 1 (which requires superuser privileges, since it is a port less than 1024) we get

```
aix % tpi_daytime 206.62.226.33
T_ERROR_ACK from bind (3, 0)
```

The error `EACCES` has the value of 3 on this system. If we change the port to a value greater than 1023, but one that is currently in use by another TCP endpoint, we get

```
aix % tpi_daytime 206.62.226.33
T_ERROR_ACK from bind (23, 0)
```

The error `EADDRBUSY` has a value of 23 on this system.

This error is new with the TPI to support XTI. Older versions of TPI that support TLI would bind another unused port if the requested one was busy. This meant that a server binding a well-known port would have to compare the returned address (from the `T_bind_ack` message, which is returned by `t_bind` if the third argument is a nonnull pointer) to the requested address, and abort if they were not equal.

The next function, shown in Figure 33.10, is `tpi_connect`, which establishes the connection with the server.

Fill in request structure and send to provider

18-26 TPI defines a `T_conn_req` structure that contains the protocol address and options for the connection:

```
struct T_conn_req {
  long    PRIM_type;    /* T_CONN_REQ */
  long    DEST_length;  /* destination address length */
  long    DEST_offset;  /* destination address offset */
  long    OPT_length;   /* options length */
  long    OPT_offset;   /* options offset */
      /* followed by the protocol address and options for connection */
};
```

As in our `tpi_bind` function, we define our own structure named `conn_req` that includes a `T_conn_req` structure along with room for the protocol address. We fill in our `conn_req` structure, setting the two members dealing with options to 0. We call `putmsg` with only control information and a flag of 0 to send an `M_PROTO` message down the stream.

streams/tpi_connect.c

```
 1 #include    "tpi_daytime.h"

 2 void
 3 tpi_connect(int fd, const void *addr, size_t addrlen)
 4 {
 5     struct {
 6         struct T_conn_req msg_hdr;
 7         char    addr[128];
 8     } conn_req;
 9     struct {
10         struct T_conn_con msg_hdr;
11         char    addr[128];
12     } conn_con;
13     struct strbuf ctlbuf;
14     union T_primitives rcvbuf;
15     struct T_error_ack *error_ack;
16     struct T_discon_ind *discon_ind;
17     int     flags;

18     conn_req.msg_hdr.PRIM_type = T_CONN_REQ;
19     conn_req.msg_hdr.DEST_length = addrlen;
20     conn_req.msg_hdr.DEST_offset = sizeof(struct T_conn_req);
21     conn_req.msg_hdr.OPT_length = 0;
22     conn_req.msg_hdr.OPT_offset = 0;
23     memcpy(conn_req.addr, addr, addrlen);   /* sockaddr_in{} */
```

```
24      ctlbuf.len = sizeof(struct T_conn_req) + addrlen;
25      ctlbuf.buf = (char *) &conn_req;
26      Putmsg(fd, &ctlbuf, NULL, 0);

27      ctlbuf.maxlen = sizeof(union T_primitives);
28      ctlbuf.len = 0;
29      ctlbuf.buf = (char *) &rcvbuf;
30      flags = RS_HIPRI;
31      Getmsg(fd, &ctlbuf, NULL, &flags);
32      if (ctlbuf.len < (int) sizeof(long))
33          err_quit("tpi_connect: bad length from getmsg");

34      switch (rcvbuf.type) {
35      case T_OK_ACK:
36          break;

37      case T_ERROR_ACK:
38          if (ctlbuf.len < (int) sizeof(struct T_error_ack))
39              err_quit("tpi_connect: bad length for T_ERROR_ACK");
40          error_ack = (struct T_error_ack *) &rcvbuf;
41          err_quit("tpi_connect: T_ERROR_ACK from conn (%d, %d)",
42                   error_ack->TLI_error, error_ack->UNIX_error);

43      default:
44          err_quit("tpi_connect: unexpected message type: %d", rcvbuf.type);
45      }

46      ctlbuf.maxlen = sizeof(conn_con);
47      ctlbuf.len = 0;
48      ctlbuf.buf = (char *) &conn_con;
49      flags = 0;
50      Getmsg(fd, &ctlbuf, NULL, &flags);
51      if (ctlbuf.len < (int) sizeof(long))
52          err_quit("tpi_connect2: bad length from getmsg");

53      switch (conn_con.msg_hdr.PRIM_type) {
54      case T_CONN_CON:
55          break;

56      case T_DISCON_IND:
57          if (ctlbuf.len < (int) sizeof(struct T_discon_ind))
58              err_quit("tpi_connect2: bad length for T_DISCON_IND");
59          discon_ind = (struct T_discon_ind *) &conn_con.msg_hdr;
60          err_quit("tpi_connect2: T_DISCON_IND from conn (%d)",
61                   discon_ind->DISCON_reason);

62      default:
63          err_quit("tpi_connect2: unexpected message type: %d",
64                   conn_con.msg_hdr.PRIM_type);
65      }
66  }
```

streams/tpi_connect.c

Figure 33.10 `tpi_connect` function: establish connection with server.

Read response

27-45 We call `getmsg` expecting to receive either a `T_OK_ACK` message

```
struct T_ok_ack {
  long    PRIM_type;      /* T_OK_ACK   */
  long    CORRECT_prim;   /* correct primitive */
};
```

if the connection establishment was started, or a T_ERROR_ACK message (which we showed earlier). In the case of an error, we terminate. Since we do not know what type of message we will receive, a union named T_primitives is defined as the union of all the possible requests and replies, and we allocate one of these that we use as the input buffer for the control information when we call getmsg.

Wait for connection establishment to complete

46-65 The successful T_OK_ACK message that was just received only tells us that the connection establishment was started. We must now wait for a T_CONN_CON message to tell us that the other end has confirmed the connection request.

```
struct T_conn_con {
  long    PRIM_type;      /* T_CONN_CON */
  long    RES_length;     /* responding address length */
  long    RES_offset;     /* responding address offset */
  long    OPT_length;     /* option length */
  long    OPT_offset;     /* option offset */
      /* followed by peer's protocol address and options */
};
```

We call getmsg again, but the expected message is sent as an M_PROTO message, not an M_PCPROTO message, so we set the flags to 0. If we receive the T_CONN_CON message, the connection is established, and we return, but if the connection was not established (either the peer process was not running, a timeout, or whatever), a T_DISCON_IND message is sent up the stream instead:

```
struct T_discon_ind {
  long    PRIM_type;      /* T_DISCON_IND */
  long    DISCON_reason;  /* disconnect reason */
  long    SEQ_number;     /* sequence number */
};
```

We can see the different errors that are returned by the provider. We first specify the IP address of a host that is not running the daytime server:

```
solaris26 % tpi_daytime 140.252.1.4
tpi_connect2: T_DISCON_IND from conn (146)
```

The error of 146 corresponds to ECONNREFUSED. Next we specify an IP address that is not connected to the Internet:

```
solaris26 % tpi_daytime 192.3.4.5
tpi_connect2: T_DISCON_IND from conn (145)
```

The error this time is ETIMEDOUT. But if we run our program again, specifying the same IP address, we get a different error:

```
solaris26 % tpi_daytime 192.3.4.5
tpi_connect2: T_DISCON_IND from conn (148)
```

The error this time is EHOSTUNREACH. The difference in the last two results is that the first time no ICMP host unreachable errors were returned, while the next time this error was returned.

The next function is tpi_read, shown in Figure 33.11. It reads data from a stream.

streams/tpi_read.c

```
 1 #include    "tpi_daytime.h"

 2 ssize_t
 3 tpi_read(int fd, void *buf, size_t len)
 4 {
 5     struct strbuf ctlbuf;
 6     struct strbuf datbuf;
 7     union T_primitives rcvbuf;
 8     int     flags;

 9     ctlbuf.maxlen = sizeof(union T_primitives);
10     ctlbuf.buf = (char *) &rcvbuf;

11     datbuf.maxlen = len;
12     datbuf.buf = buf;
13     datbuf.len = 0;

14     flags = 0;
15     Getmsg(fd, &ctlbuf, &datbuf, &flags);

16     if (ctlbuf.len >= (int) sizeof(long)) {
17         if (rcvbuf.type == T_DATA_IND)
18             return (datbuf.len);
19         else if (rcvbuf.type == T_ORDREL_IND)
20             return (0);
21         else
22             err_quit("tpi_read: unexpected type %d", rcvbuf.type);
23     } else if (ctlbuf.len == -1)
24         return (datbuf.len);
25     else
26         err_quit("tpi_read: bad length from getmsg");
27 }
```

streams/tpi_read.c

Figure 33.11 tpi_read function: read data from a stream.

Read control and data; process reply

9–26 This time we call getmsg to read both control information and data. The strbuf structure for the data points to the caller's buffer. Four different scenarios can occur on the stream.

- The data can arrive as an M_DATA message, and this is indicated by the returned control length set to –1. The data was copied into the caller's buffer by getmsg, and we just return the length of this data as the return value of the function.
- The data can arrive as a T_DATA_IND message, in which case the control information will be a T_data_ind structure:

```
struct T_data_ind {
  long    PRIM_type;  /* T_DATA_IND */
  long    MORE_flag;  /* more data */
};
```

If this message is returned, we ignore the `MORE_flag` member (it will never be set for a stream protocol such as TCP) and just return the length of the data that was copied into the caller's buffer by `getmsg`.

- A `T_ORDREL_IND` message is returned if all the data has been consumed and the next item is a FIN:

```
struct T_ordrel_ind {
  long    PRIM_type;      /* T_ORDREL_IND */
};
```

This is the orderly release that we described in Section 28.9. We just return 0, indicating to the caller that the end-of-file has been encountered on the connection.

- A `T_DISCON_IND` message is returned if a disconnect has been received. We discussed this in Section 28.10 and said this occurs in the case of TCP if an RST is received on an existing connection. We do not handle this scenario in this simple example, but we did handle it in Figure 28.13.

We can now explain the two different scenarios that we saw in Section 28.12 when we called `read` but had not pushed the `tirdwr` module onto the stream. In the first example, which generated the error "read error: Not a data message," the provider had sent a `T_DATA_IND` message up the stream as an `M_PROTO` message (since it had control and data). But `read` handles only `M_DATA` messages, hence the error.

In the second example the error was "read error: Bad message" but this appeared after the server's expected reply was received and printed. On this implementation the provider sent the data up the stream as an `M_DATA` message, so it was handled by `read` correctly. But the next message up the stream was a `T_ORDREL_IND` message, which `read` cannot handle.

Our final function is `tpi_close`, shown in Figure 33.12.

Send orderly release to peer

7-10 We build a `T_ordrel_req` structure

```
struct T_ordrel_req {
  long  PRIM_type;   /* T_ORDREL_REQ */
};
```

and send it as an `M_PROTO` message using `putmsg`. This corresponds to the XTI `t_sndrel` function.

This example has given us a flavor for TPI. The application sends messages down a stream to the provider (requests) and the provider sends messages up the stream (replies). Some exchanges are a simple request–reply scenario (binding a local address) while others may take a while (establishing a connection), allowing us to do something

streams/tpi_close.c

```
 1 #include    "tpi_daytime.h"
 2 void
 3 tpi_close(int fd)
 4 {
 5     struct T_ordrel_req ordrel_req;
 6     struct strbuf ctlbuf;
 7     ordrel_req.PRIM_type = T_ORDREL_REQ;
 8     ctlbuf.len = sizeof(struct T_ordrel_req);
 9     ctlbuf.buf = (char *) &ordrel_req;
10     Putmsg(fd, &ctlbuf, NULL, 0);
11     Close(fd);
12 }
```

streams/tpi_close.c

Figure 33.12 `tpi_close` function: send an orderly release to peer.

while we wait for the reply. Our choice of writing a TCP client using TPI was done for simplicity; writing a TCP server and handling connections as we described in Section 30.7 becomes much harder.

It should be obvious that the mapping from the XTI functions to TPI is very close. On the other hand, the mapping from sockets to TPI is not as close. Nevertheless, both the XTI and socket libraries handle lots of the details required by TPI, simplifying our applications.

> We can compare the number of system calls required for the network operations that we have seen in this chapter, when using TPI versus a kernel that implements sockets within the kernel. Binding a local address takes two system calls with TPI, but only one with kernel sockets (TCPv2, p. 454). To establish a connection on a blocking descriptor takes three system calls with TPI, but only one with kernel sockets (TCPv2, p. 466).

33.7 Summary

XTI is often implemented using streams. Four new functions are provided to access the streams subsystem, `getmsg`, `getpmsg`, `putmsg`, and `putpmsg`, and the existing `ioctl` function is heavily used by the streams subsystem also.

TPI is the SVR4 streams interface from the upper layers into the transport layer. It is used by both XTI and sockets, as shown in Figure 33.3. We developed a version of our daytime client using TPI directly, as an example to show the message-based interface that TPI uses.

Exercises

33.1 In Figure 33.12 we call `putmsg` to send the orderly release request down the stream and then immediately `close` the stream. What happens if our orderly release request is lost by the streams subsystem when the stream is closed?

34

XTI: Additional Functions

34.1 Introduction

In the previous chapters we have covered the XTI functions for

- TCP clients,
- hostname and service name lookups,
- TCP servers,
- UDP clients and servers,
- options, and
- the common streams implementation.

This chapter covers the remaining XTI functions.

34.2 Nonblocking I/O

An endpoint can be put into a nonblocking mode. This is done by specifying the `O_NONBLOCK` flag in the call to `t_open` when the endpoint is created, or at a later time with the `fcntl` function (as shown in Section 7.10).

The operation of some of the XTI functions changes when the endpoint is nonblocking.

- `t_connect` returns immediately with a return of −1 and `t_errno` set to `TNODATA`. With TCP this call initiates the three-way handshake, and we must call `t_rcvconnect` (Section 34.3) to wait for the connection establishment to complete.

- `t_rcvconnect` returns –1 with `t_errno` set to `TNODATA` if a connection is in progress but has not yet completed.
- `t_listen` returns immediately with a return of –1 and `t_errno` set to `TNODATA` when there are no connections ready for the application to call `t_accept`.
- The four receive functions, `t_rcv`, `t_rcvudata`, `t_rcvv`, and `t_rcvvudata`, return –1 with `t_errno` set to `TNODATA` if there is no data available. If some data is available, that data is returned, even though it may be less than asked for by the application. (The last two functions mentioned are new; we describe them in Section 34.8.)
- The four send functions, `t_snd`, `t_sndudata`, `t_sndv`, and `t_sndvudata`, return –1 with `t_errno` set to `TFLOW` if the provider is not able to accept any data. If some data can be accepted, then the return value might be less than the amount requested for `t_snd` and `t_sndv`. The two datagram functions write a complete datagram, or they return an error. (The last two of the four functions listed are new; we describe them in Section 34.9.)

34.3 `t_rcvconnect` Function

In the previous section we mentioned initiating a connection in the nonblocking mode and then waiting for the connection to complete by calling `t_rcvconnect`.

```
#include <xti.h>

int t_rcvconnect(int fd, struct t_call *recvcall);
```

Returns: 0 if OK, –1 on error

The sequence of steps typically used with this function are as follows:

1. An endpoint is created using `t_open` and set nonblocking.
2. `t_connect` initiates the connection establishment. Since the endpoint is in the nonblocking mode, this function returns immediately with a value of –1 and `t_errno` set to `TNODATA`.
3. At some later time the process calls `t_rcvconnect` to determine if the connection has completed. If the endpoint is no longer in a nonblocking mode (the process has turned off the nonblocking flag since calling `t_connect` in step 2), then `t_rcvconnect` blocks until the connection is established. If the endpoint is still in a nonblocking mode, then this call to `t_rcvconnect` either (a) returns immediately with a return value of 0 if the connection is established, or (b) returns a value of –1 with `t_errno` set to `TNODATA` if the connection is not yet established.

Note in the case of a blocking t_connect (the default), the provider returns the information in the t_call structure that is pointed to by the third argument to t_connect. But with a nonblocking t_connect, this information is returned in the t_call structure that is pointed to by the second argument to t_rcvconnect.

Unless the application converts the endpoint from nonblocking to blocking between the calls to t_connect and t_rcvconnect (steps 2 and 3 above), calling t_rcvconnect to determine when a nonblocking connection establishment completes is a waste of time, because the application must call t_rcvconnect in a loop of some form, waiting for the connection to complete (or an error to be returned). This is called *polling*. Better techniques for waiting for a nonblocking connection establishment to complete are to call either select or poll (Chapter 6), or to use signal-driven I/O (Section 34.11).

Recall our discussion of an interrupted connection establishment at the end of Section 15.4. With XTI, if a call to t_connect on a blocking endpoint is interrupted, we just call t_rcvconnect to wait for the connection establishment to complete.

34.4 t_getinfo Function

Recall the t_info structure that is returned by the t_open function (Section 28.2). The following function returns the same information to the caller.

```
#include <xti.h>

int t_getinfo(int fd, struct t_info *info);
```

Returns: 0 if OK, –1 on error

This function is called, for example, by t_alloc, to obtain the information about an endpoint that is already open for t_alloc to obtain the required buffer sizes.

34.5 t_getstate Function

Every transport endpoint has a *current state* associated with it. The following function returns the current state (an integer value) to the caller.

```
#include <xti.h>

int t_getstate(int fd);
```

Returns: current state if OK, –1 on error

The current state is specified by one of the constants shown in Figure 34.1. The final three columns indicate which states are valid for the different service types (Figure 28.3).

State	Description	T_COTS	T_COTS_ORD	T_CLTS
T_DATAXFER	data transfer	•	•	
T_IDLE	bound, but idle	•	•	•
T_INCON	incoming connection pending for passive endpoint	•	•	
T_INREL	incoming orderly release		•	
T_OUTCON	outgoing connection pending for active endpoint	•	•	
T_OUTREL	outgoing orderly release		•	
T_UNBND	unbound	•	•	•
T_UNINIT	uninitialized: starting and final state	•	•	•

Figure 34.1 Possible states of an XTI endpoint.

A state transition diagram can be developed to show exactly how the state of a transport endpoint changes as different XTI functions are called and as different events occur at the endpoint. This diagram would also show which XTI functions are allowed in the different states. For example, the only function call allowed in the T_UNINIT state is t_open and the new state becomes T_UNBND. Four events can occur in the T_UNBND state:

1. A successful return from t_close changes the state to T_UNINIT.
2. Calling t_optmgmt is allowed but does not change the state. (What this state transition diagram cannot show, however, is that the option processing might change depending on the state. For example, the T_UDP_CHECKSUM option behaves differently in the T_UNBND state, versus other states.)
3. A successful return from t_bind changes the state to T_IDLE.
4. Passing a connection to the endpoint (by t_accept) is allowed, and changes the state to T_DATAXFER.

Once we get past these first two states, however, the diagram becomes unwieldy, so we will not attempt to show it.

34.6 t_sync Function

Historically TLI was implemented as a library of functions in SVR3. Consider a program using TLI that calls exec as shown in Figure 34.2. Perhaps the program on the left is a listening server that waits for a connection to arrive and be accepted and then execs the program on the right to handle the client. (Remember that the process ID does not change across an exec, but the caller's memory is replaced with the new program that then begins execution at its main function.)

The problem encountered with this scenario in SVR3 was that state information is maintained in both the TLI library within the process and in the provider within the kernel. After an exec all of the state information in the library is discarded, and the library in the new program starts off fresh. The purpose of the t_sync function was to allow the new program (on the right in Figure 34.2) to synchronize the state of its library with the provider in the kernel.

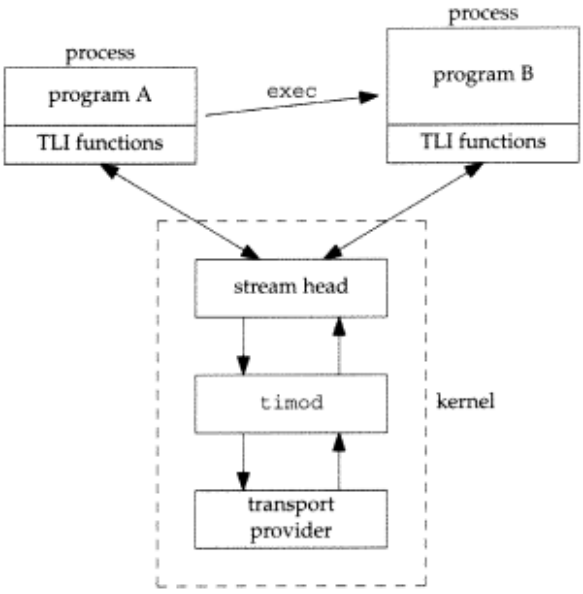


Figure 34.2 TLI implementation when a process calls exec.

With SVR4, however, the need to call t_sync for the scenario that we show in Figure 34.2 disappeared. The TLI library functions could detect the need for performing a synchronization themselves. For example, if the library has a variable declared as

```
static int  synced;     /* initialized to 0 when program starts */
```

then each function can begin with the sequence similar to the following:

```
int
t_connect(fd, ... )
{
    if (synced == 0)
        t_sync(fd);  /* also sets synced = 1 */

    ...
}
```

While this handles the case of a process calling exec, there is still one scenario, albeit rare, where t_sync is required: when multiple processes are sharing an XTI endpoint. In this scenario it is assumed that the processes are cooperating with each other and that each calls t_sync when it deems necessary (which depends, of course, on the specifics of the application). One example where t_sync might be necessary is in a parent–child relationship, if the parent calls t_listen and then the child calls t_accept. The parent would need to call t_sync to update its library copy of the endpoint state, which might change after the child calls t_accept.

```
#include <xti.h>

int t_sync(int fd);
```

Returns: current state if OK, –1 on error

The successful return from this function is one of the states shown in Figure 34.1.

There is no operation similar to this function in the sockets API.

34.7 t_unbind Function

The effect of the `t_bind` function is undone by `t_unbind`.

```
#include <xti.h>

int t_unbind(int fd);
```

Returns: 0 if OK, –1 on error

This function disables the transport endpoint specified by *fd*. No further data will be accepted for this endpoint. `t_bind` may be called, however, to bind another local address to the endpoint.

With sockets this operation can be performed only on a connected UDP socket by calling `bind` with an invalid address.

34.8 t_rcvv and t_rcvvudata Functions

These two functions extend the `t_rcv` and `t_rcvudata` functions to operate on a *vector* of buffers, instead of just a single buffer. They provide a *scatter read* capability.

These two functions and the two described in the next section were introduced with Posix.1g.

The concept of operating on a vector of buffers comes from the `readv` and `writev` functions, along with the `recvmsg` and `sendmsg` functions.

```
#include <xti.h>

int t_rcvv(int fd, struct t_iovec *iov, unsigned int iovcnt, int *flags);

int t_rcvvudata(int fd, struct t_unitdata *unitdata,
                struct t_iovec *iov, unsigned int iovcnt, int *flags);
```

Both return: number of bytes read or written if OK, –1 on error

The *iov* argument to both functions is a pointer to an array of `t_iovec` structures:

```
struct t_iovec {
  void   *iov_base;  /* starting address of buffer */
  size_t  iov_len;   /* length of buffer in bytes */
}
```

The number of entries in the array is specified by the *iovcnt* argument. The limit on the number of entries in the array is given by the constant `T_IOV_MAX`, defined by including the `<xti.h>` header, whose value must be at least 16.

Comparing these new functions to their earlier counterparts we see the following:

- The buffer pointer and its length are the middle two arguments to `t_rcv`.
- The buffer pointers and their lengths are in an array of `t_iovec` structures for `t_rcvv`, and the middle two arguments for this function point to this array of structures and specify the number of entries in the array.
- The buffer pointer and its length are in the `udata` member of the `t_unitdata` structure for `t_rcvudata`. Also, this function returns 0 upon success, with the actual length of the received datagram in the `udata.len` member of the `t_unitdata` structure.
- The buffer pointers and their lengths are in an array of `t_iovec` structures for `t_rcvvudata`. The third argument to this function is a pointer to this array of structures and the fourth argument is the number of entries in the array. A pointer to a `t_unitdata` structure is the second argument to this function and the `addr` and `opt` member are still used (for the sender's protocol address and any received options), but the `udata` member is ignored. This function returns the number of bytes in the datagram as its return value, not 0.

34.9 t_sndv and t_sndvudata Functions

These two functions extend the `t_snd` and `t_sndudata` functions to operate on a *vector* of buffers, instead of just a single buffer. They are the send counterparts of the two functions described in the previous section and provide a *gather write* capability.

```
#include <xti.h>

int t_sndv(int fd, struct t_iovec *iov, unsigned int iovcnt, int flags);
```

Returns: number of bytes read or written if OK, –1 on error

```
int t_sndvudata(int fd, struct t_unitdata *unitdata,
                struct t_iovec *iov, unsigned int iovcnt);
```

Returns: 0 if OK, –1 on error

The *iov* argument to both functions is a pointer to an array of `t_iovec` structures, which we showed in the previous section. The number of entries in the array is specified by the *iovcnt* argument.

The output buffers are specified by the two middle arguments to `t_sndv`, replacing the two middle arguments to `t_snd`. For the datagram functions, the output buffer is specified by the `udata` member of the `t_unitdata` structure with `t_sndudata` but by the *iov* vector with `t_sndvudata`. The `udata` member of the `t_unitdata` structure is ignored by `t_sndvudata`.

34.10 `t_rcvreldata` and `t_sndreldata` Functions

If we send an orderly release with `t_sndrel` (Section 28.9) we cannot send data with the orderly release notification (the only argument to the function is a descriptor), but if we send a disconnect with `t_snddis` (Section 28.10), we can send data (the `udata` member of the `t_call` structure). We find the same limitation for `t_rcvrel`, compared to `t_rcvdis`. To get around this limitation XTI invented two new functions that send and receive data with an orderly release.

```
#include <xti.h>

int t_sndreldata(int fd, const struct t_discon *discon);

int t_rcvreldata(int fd, struct t_discon *discon);
```

Both return: 0 if OK, –1 on error

The difference between these two functions and `t_sndrel` and `t_rcvrel` is the addition of the second argument (a pointer to a `t_discon` structure).

These functions are useful only when the provider supports the sending of data with an orderly release, as indicated by the `T_ORDRELDATA` flag in the `flag` member of the `t_info` structure (Figure 28.4). If supported, the amount of orderly release data is limited to the value of the `discon` member of the `t_info` structure.

TCP does not support this optional feature.

34.11 Signal-Driven I/O

Signal-driven I/O is provided by the streams system, not XTI. The signal name is `SIGPOLL` and the signal is not delivered just because the process installs a signal handling function for the signal. The process must also tell the kernel that it wants to receive the signal by issuing the `I_SETSIG` streams `ioctl` request, specifying which conditions should generate the signal. This is similar to what we must do to receive the `SIGIO` and `SIGURG` signals that we described for the sockets API.

The third argument to `ioctl` is an integer value that specifies the conditions for which a `SIGPOLL` signal should be generated. If this value is 0, the process will no longer receive the `SIGPOLL` signal for the stream. This value can also be formed as the logical OR of the following constants:

S_BANDURG	If this flag is specified in conjunction with S_RDBAND, the SIGURG signal will be generated instead of SIGPOLL when a message in a priority band greater than 0 can be read.
S_ERROR	The stream is in error.
S_HANGUP	A hangup message has reached the stream head.
S_HIPRI	A high-priority message can be read.
S_INPUT	This is equivalent to S_RDNORM | S_RDBAND and means that a message with any band (including 0) can be read.
S_OUTPUT	The write queue just below the stream head is no longer flow controlled for normal messages (band 0).
S_MSG	A streams signal message is at the front of the stream's read queue.
S_RDNORM	A normal message (band 0) can be read.
S_RDBAND	A message in a priority band greater than 0 can be read.
S_WRNORM	Equivalent to S_OUTPUT.
S_WRBAND	The write queue is no longer flow controlled for messages in a priority band greater than 0.

The S_BANDURG flag is used by the sockets API when it is implemented using streams.

There is no output equivalent for S_HIPRI. This is because putmsg and putpmsg do not block when sending a high-priority message: these messages are not flow controlled.

The streams signal SIGPOLL is used for both signal-driven I/O and for the notification of the arrival of out-of-band data. This corresponds to the two signals SIGIO and SIGURG that we described with the sockets API.

The default action for SIGPOLL is to terminate the process, so when using this signal we must establish the signal handler and then call ioctl to enable the signal.

There is a conflict between the default action of SIGPOLL, which we just said terminates the process, and SIGIO. Posix.1g specifies that the default action of SIGIO is to be ignored. Since SVR4 systems define these two signals to be the same, these systems will have to change the default action for SIGPOLL to be ignored, to be Posix.1g compliant.

Even though the SIGPOLL signal can be generated for numerous conditions, typical applications that do not deal with out-of-band data are interested in only S_RDNORM and S_WRNORM.

34.12 Out-of-Band Data

Out-of-band data is called *expedited data* by XTI. Support for this feature is provided by the transport provider and the streams system. We mentioned in Chapter 33 that

out-of-band data is often implemented as normal-priority data in priority band 1. Normal data (i.e., not out-of-band data) is in priority band 0.

> We also mentioned in Chapter 33 that since TCP's out-of-band data is not true expedited data (in the XTI sense), it is actually implemented in band 0, not band 1.

Everything that we said in Chapter 21 about the support for out-of-band data with TCP, and the mapping of TCP's urgent mode into out-of-band data, applies to XTI just like sockets.

Out-of-band data is sent with the XTI `t_snd` function by specifying a *flags* argument of `T_EXPEDITED`. This flag value is also returned to the caller by the `t_rcv` function.

We cannot use the `read` and `write` functions when writing an application that deals with out-of-band data (recall our `xti_rdwr` function in Section 28.12). We must use `t_snd` and `t_rcv`.

Since TCP's out-of-band data corresponds to normal-priority messages in band 0, to receive `SIGPOLL` when out-of-band data arrives requires that we specify `S_RDNORM` when we call `ioctl` with a request of `I_SETSIG` (Section 34.11). Since this also generates the signal when normal data arrives, if we want to differentiate between normal data and out-of-band data, we must call `t_look` from our signal handler and check for either `T_DATA` or `T_EXDATA` (Figure 28.9). XTI sets the event `T_EXDATA` as soon as a TCP segment with an urgent pointer is received, and this event remains set until all data up through the urgent pointer has been received.

> `SIGPOLL` corresponds to the `SIGURG` signal for sockets.

If using `poll` to await the arrival of out-of-band data (Section 6.10), the `events` member of the `pollfd` structure must be set to `POLLRDNORM` since it appears as normal data in band 0. We will verify that TCP's out-of-band data appears as normal data to `poll` in Figures 34.4 and C.5. Also note that this treatment of out-of-band data with XTI differs from sockets, which considers out-of-band data as belonging to a priority band.

> Using `poll` corresponds to calling `select` and waiting for an exception condition, but `poll` does not tell us what type of data arrived: we must call `t_look` and `t_rcv`.
>
> Recall that by default the sockets API removes a received out-of-band byte from the normal stream of data, placing it into its own special 1-byte buffer that the application reads using `recv` with the `MSG_OOB` flag. There is nothing similar to this mode with XTI: TCP's out-of-band data is always received inline, something we have to enable with sockets using the `SO_OOBINLINE` socket option.

We now look at a few examples to see how signal-driven I/O and `poll` work with XTI's out-of-band data.

Example Using `SIGPOLL`

Figure 34.3 is a program that uses `SIGPOLL` to be notified when data is available on an XTI endpoint.

xtioob/tcprecv01.c

```
 1 #include    "unpxti.h"
 2 #define NREAD   100
 3 int     listenfd, connfd;
 4 void    sig_poll(int);
 5 int
 6 main(int argc, char **argv)
 7 {
 8     int     n, flags;
 9     char    buff[NREAD + 1];    /* +1 for null at end */
10     if (argc == 2)
11         listenfd = Tcp_listen(NULL, argv[1], NULL);
12     else if (argc == 3)
13         listenfd = Tcp_listen(argv[1], argv[2], NULL);
14     else
15         err_quit("usage: tcprecv01 [ <host> ] <port#>");
16     connfd = Xti_accept(listenfd, NULL, NULL);
17     Signal(SIGPOLL, sig_poll);
18     Ioctl(connfd, I_SETSIG, S_RDNORM);
19     for ( ; ; ) {
20         flags = 0;
21         if ( (n = t_rcv(connfd, buff, NREAD, &flags)) < 0) {
22             if (t_errno == TLOOK) {
23                 if ( (n = T_look(connfd)) == T_ORDREL) {
24                     printf("received T_ORDREL\n");
25                     exit(0);
26                 } else
27                     err_quit("unexpected event after t_rcv: %d", n);
28             }
29             err_xti("t_rcv error");
30         }
31         buff[n] = 0;            /* null terminate */
32         printf("read %d bytes: %s, flags = %s\n",
33                n, buff, Xti_flags_str(flags));
34     }
35 }
36 void
37 sig_poll(int signo)
38 {
39     printf("SIGPOLL received, event = %s\n", Xti_tlook_str(connfd));
40 }
```

xtioob/tcprecv01.c

Figure 34.3 Receive normal and out-of-band data using `SIGPOLL` on an XTI endpoint.

Create listening endpoint and wait for connection

10-16 We call our `tcp_listen` function to create a listening endpoint and then our `xti_accept` function waits for a connection to arrive and accepts it.

Establish signal handler

17-18 We call `signal` to establish a signal handler for `SIGPOLL` and then call `ioctl` to enable the signal to be generated when normal data arrives for the endpoint.

Loop, reading data

19-34 We call `t_rcv` to receive the data, handling an orderly release when the peer closes the connection. We print the data bytes that are received, along with the `flags` returned by `t_rcv`. Our function `xti_flags_str` returns a pointer to a message describing the flags that are passed as an argument.

Signal handler

36-40 Our signal handler just prints a message that includes the current event for the endpoint. Our function `xti_tlook_str` calls `t_look` and returns a pointer to a message describing the current event for the endpoint.

We start this program and then run the program from Figure 21.3 as the client. Here is the output from our server:

```
unixware % tcprecv01 9999
read 3 bytes: 123, flags = 0
SIGPOLL received, event = T_EXDATA
read 1 bytes: 4, flags = T_EXPEDITED
SIGPOLL received, event = T_DATA
read 2 bytes: 56, flags = 0
SIGPOLL received, event = T_EXDATA
read 1 bytes: 7, flags = T_EXPEDITED
SIGPOLL received, event = T_DATA
read 2 bytes: 89, flags = 0
SIGPOLL received, event = T_ORDREL
received T_ORDREL
```

The first 3 bytes are received as normal data but `SIGPOLL` is not generated. This is a timing issue. Recall from Figure 2.5 that the client's `connect` (or `t_connect` if XTI is being used) returns one-half an RTT before the server's `accept` (or `t_accept`), given how the TCP three-way handshake operates. This gives the client a head start in sending its first segment of data, and we see in this example that the first 3 bytes arrive before we establish our signal handler.

We then receive `SIGPOLL` and the event is `T_EXDATA`. `t_rcv` returns a flag of `T_EXPEDITED`. We can see from the remaining lines of output that each time a TCP segment arrives, `SIGPOLL` is generated, and we must call `t_look` to see what event has occurred.

When we have read all the data and are notified that the client closed its end of the connection, the signal is generated and the event is `T_ORDREL`, as expected.

Example Using `poll`

Our next example, shown in Figure 34.4, uses the `poll` function.

xtioob/tcprecv03.c

```
 1 #include    "unpxti.h"
 2 #define NREAD   100
 3 int     listenfd, connfd;
 4 int
 5 main(int argc, char **argv)
 6 {
 7     int     n, flags;
 8     char    buff[NREAD + 1];    /* +1 for null at end */
 9     struct pollfd pollfd[1];
10     if (argc == 2)
11         listenfd = Tcp_listen(NULL, argv[1], NULL);
12     else if (argc == 3)
13         listenfd = Tcp_listen(argv[1], argv[2], NULL);
14     else
15         err_quit("usage: tcprecv03 [ <host> ] <port#>");
16     connfd = Xti_accept(listenfd, NULL, NULL);
17     pollfd[0].fd = connfd;
18     pollfd[0].events = POLLIN;
19     for ( ; ; ) {
20         Poll(pollfd, 1, INFTIM);
21         printf("revents = %x\n", pollfd[0].revents);
22         if (pollfd[0].revents & POLLIN) {
23             flags = 0;
24             if ( (n = t_rcv(connfd, buff, NREAD, &flags)) < 0) {
25                 if (t_errno == TLOOK) {
26                     if ( (n = T_look(connfd)) == T_ORDREL) {
27                         printf("received T_ORDREL\n");
28                         exit(0);
29                     } else
30                         err_quit("unexpected event after t_rcv: %d", n);
31                 }
32                 err_xti("t_rcv error");
33             }
34             buff[n] = 0;        /* null terminate */
35             printf("read %d bytes: %s, flags = %s\n",
36                    n, buff, Xti_flags_str(flags));
37         }
38     }
39 }
```

xtioob/tcprecv03.c

Figure 34.4 Receive normal and out-of-band data using `poll` on an XTI endpoint.

Wait for client connection

10–16 The creation of the listening endpoint and accepting the client's connection have not changed from Figure 34.3.

Prepare for `poll`

17-18 We allocate a 1-element `pollfd` array and initialize it to tell us when normal or priority data arrives for our connected endpoint.

Call `poll`

19-38 We call `poll` with an infinite time limit. When it returns, we print the `revents` member of our `pollfd` structure to see what type of data has arrived. If the event is `POLLIN`, we call `t_rcv` to read the data and print the data and the returned flags.

We run this program with Figure 21.3 as the client (the same client as in the previous example).

```
unixware % tcprecv03 7777
revents = 1
read 3 bytes: 123, flags = 0
revents = 1
read 1 bytes: 4, flags = T_EXPEDITED
revents = 1
read 2 bytes: 56, flags = 0
revents = 1
read 1 bytes: 7, flags = T_EXPEDITED
revents = 1
read 2 bytes: 89, flags = 0
revents = 1
received T_ORDREL
```

Each time `poll` returns, the event is 1, which is `POLLIN` on this system. `t_rcv` tells us the type of data being returned through the returned flags.

34.13 Loopback Transport Providers

Many implementations of XTI provide a loopback transport provider. The names in the `netconfig` file are normally `ticlts`, `ticots`, and `ticotsord` for the three types of XTI providers (Figure 28.3). The `ti` prefix stands for "transport independent." These three names are also the filenames in the `/dev` directory for `t_open`.

Unix domain sockets are often implemented on streams-based systems using two of these three providers: `ticlts` is used for `SOCK_DGRAM` sockets, and either `ticots` or `ticotsord` is used for `SOCK_STREAM` sockets, depending which of the two appears first in the `netconfig` file.

One point to be aware of, when using these providers directly with XTI, is that the addresses used are called *flex addresses*, which are just arbitrary strings of one or more bytes. These addresses are not null terminated; their length is specified by the `len` member of the `netbuf` structure containing the address. But the `addr` member of the `t_info` structure can be returned as –1 (`T_INFINITE`), causing `t_alloc` not to allocate a buffer for the address.

> One of the differences between TLI and XTI is the handling of `T_INFINITE` by `t_alloc`. TLI would allocate a 1024-byte buffer by default, while XTI does not allocate a buffer.

34.14 Summary

Nonblocking I/O is enabled for an XTI endpoint by specifying the `O_NONBLOCK` in the call to `t_open`, or anytime later with `fcntl`. The changes are similar to the changes with sockets when an endpoint is made nonblocking, with the exception of a nonblocking `t_connect`. We can wait for this to complete with the `t_rcvconnect` function.

Four new I/O functions are defined by XTI to operate on vectors of buffers: `t_rcvv`, `t_rcvvudata`, `t_sndv`, and `t_sndvudata`.

Signal-driven I/O is enabled for an XTI endpoint using `ioctl`, and we also must specify all the conditions under which the signal is to be generated: one of the `S_`*xxx* flags. Out-of-band data is sent by `t_snd` when the `T_EXPEDITED` flag is set. Nothing special need be done to receive out-of-band data: `t_rcv` returns a flag of `T_EXPEDITED`. A signal can also be generated when out-of-band data arrives.

Appendix A

IPv4, IPv6, ICMPv4, and ICMPv6

A.1 Introduction

This appendix is an overview of IPv4, IPv6, ICMPv4, and ICMPv6. This material provides additional background that may be helpful in understanding the discussion of TCP and UDP in Chapter 2. Some features of IP and ICMP are also used in some of the advanced chapters: IP options (Chapter 24), along with the Ping and Traceroute programs (Chapter 25), for example.

A.2 IPv4 Header

The IP layer provides a connectionless and unreliable datagram delivery service (RFC 791 [Postel 1981a]). IP makes its best effort to deliver an IP datagram to the specified destination, but there is no guarantee that the datagram arrives. Any desired reliability must be added by the upper layers. In the case of a TCP application, this is performed by TCP itself. In the case of a UDP application, this must be done by the application itself, since UDP is unreliable, and we show an example of this in Section 20.5.

One of the most important functions of the IP layer is *routing*. Every IP datagram contains the source address and the destination address. Figure A.1 shows the format of an IPv4 header.

- The 4-bit *version* is 4. This has been the version of IP in use since the early 1980s.
- The *header length* is the length of the entire IP header, including any options, in 32-bit words. The maximum value for this 4-bit field is 15, giving a maximum IP

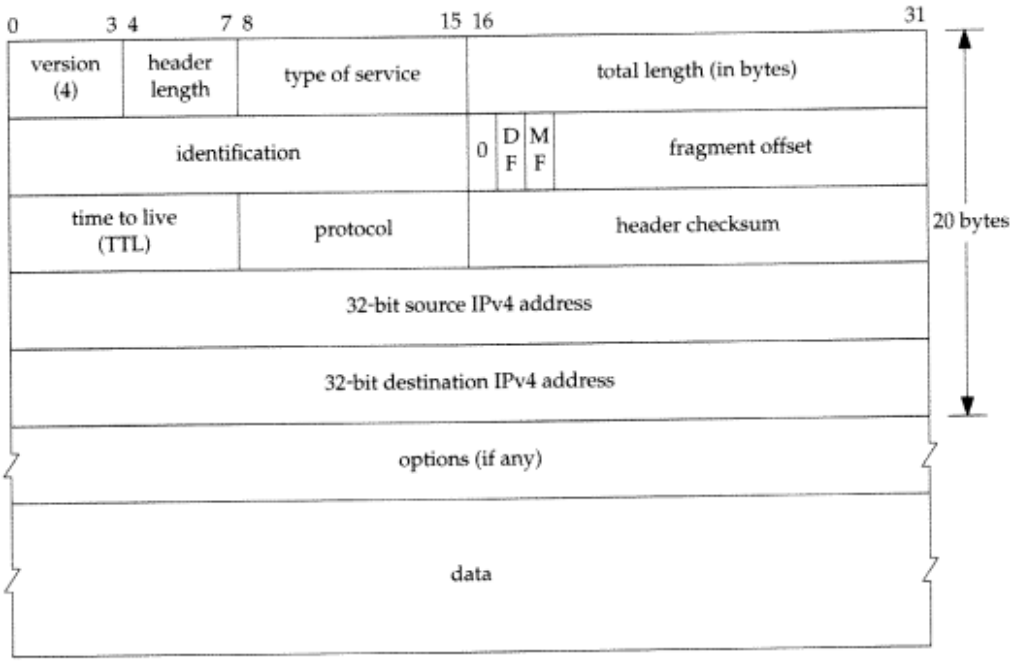


Figure A.1 Format of the IPv4 header.

header length of 60 bytes. Therefore, with the fixed portion of the header occupying 20 bytes, this allows for up to 40 bytes of options.

- The 8-bit *type-of-service* field (TOS) is composed of a 3-bit precedence field (which is ignored), 4 bits specifying the type-of-service, and an unused bit that must be 0. We can set this field with the `IP_TOS` socket option (Figure 7.12).
- The 16-bit *total length* is the total length in bytes of the IP datagram including the IPv4 header. The amount of data in the datagram is this field minus four times the header length. This field is required because some datalinks pad the frame to some minimum length (e.g., Ethernet) and it is possible for the size of a valid IP datagram to be less than the datalink minimum.
- The 16-bit *identification* field is set to a different value for each IP datagram and is used with fragmentation and reassembly (Section 2.9).
- The *DF* bit (don't fragment), the *MF* bit (more fragments), and the 13-bit *fragment offset* field are also used with fragmentation and reassembly.
- The 8-bit *time-to-live* field (TTL) is set by the sender and then decremented by one by each router that forwards the datagram. The datagram is discarded by any router that decrements the value to 0. This limits the lifetime of any IP datagram to 255 hops. A common default for this field is 64 but we can query and change this default with the `IP_TTL` and `IP_MULTICAST_TTL` socket options (Section 7.6).

- The 8-bit *protocol* field specifies the type of data contained in the IP datagram. Typical values are 1 (ICMPv4), 2 (IGMPv4), 6 (TCP), and 17 (UDP). These values are specified in RFC 1700 [Reynolds and Postel 1994].
- The 16-bit *header checksum* is calculated over just the IP header (including any options). The algorithm is the standard Internet checksum algorithm, a simple 16-bit ones-complement addition, which we show in Figure 25.14.
- The *source IPv4 address* and the *destination IPv4 address* are both 32-bit fields.
- We describe the *options* field in Section 24.2 and show an example of the IPv4 source route option in Section 24.3.

A.3 IPv6 Header

Figure A.2 shows the format of an IPv6 header (RFC 1883 [Deering and Hinden 1995]).

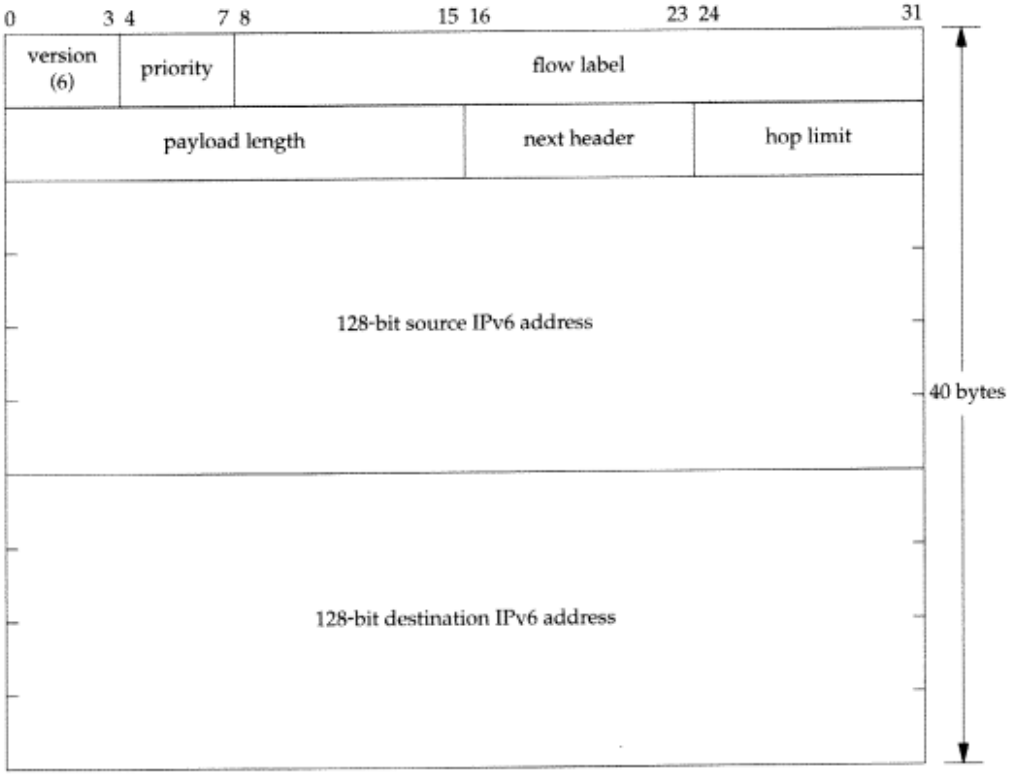


Figure A.2 Format of the IPv6 header.

- The 4-bit *version* is 6. Since this field occupies the first four bits of the first byte of the header (similar to the IPv4 version, Figure A.1), it allows a receiving IP stack that supports both versions to differentiate between the two versions.

During the development of IPv6 in the early 1990s, before the version number of 6 was assigned, the protocol was called *IPng*, for "IP next generation." You may still encounter references to IPng.

- The 4-bit priority field is set by the sender.

  The usefulness of this field is still a research topic. RFC 1883 [Deering and Hinden 1995] defines two sets of values for this field: values 0 through 7 identify traffic that backs off in response to congestion (e.g., TCP data) and values 8 through 15 identify traffic that does not back off in response to congestion (e.g., real-time packets sent at a constant rate). As of mid-1997 the proposal is that these 4 bits have no significance to receivers but the sender can set the low-order bit to indicate that the traffic is "interactive" (i.e., delay is more important than throughput). These 4 bits can also be rewritten by routers for private purposes.

- The 24-bit *flow label* can be chosen randomly by the application for a given socket. (The use of this field is still experimental.) A *flow* is a sequence of packets from a particular source to a particular destination for which the source desires special handling by intervening routers. For a given flow, once the flow label is chosen by the source, it does not change. A flow label of 0 (the default) identifies packets that do not belong to a flow.

  The combination of the priority and flow label fields is called the *flow information*. Both fields are contained in the `sin6_flowinfo` member of the `sockaddr_in6` socket address structure (Figure 3.4).

- The 16-bit *payload length* is the length in bytes of everything following the 40-byte IPv6 header. A value of 0 means the length requires more than 16 bits and is contained in a jumbo payload option (Figure 24.9). This is called a *jumbogram*.

- The 8-bit *next header* field is similar to the IPv4 protocol field. Indeed, when the upper layer is basically unchanged, the same values are used, such as 6 for TCP and 17 for UDP. There were so many changes from ICMPv4 to ICMPv6 that the latter was assigned a new value of 58.

- The 8-bit *hop limit* is similar to the IPv4 TTL field. The hop limit is decremented by one by each router that forwards the packet and the packet is discarded by any router that decrements the value to 0. The default value for this field can be set and fetched with the `IPV6_UNICAST_HOPS` and `IPV6_MULTICAST_HOPS` (Sections 7.8 and 19.5) socket options. The `IPV6_HOPLIMIT` socket option also lets us set this field and obtain its value from a received datagram.

- The *source IPv6 address* and the *destination IPv6 address* are both 128-bit fields.

  An IPv6 datagram can have numerous headers following the 40-byte IPv6 header. That is why the field is called the "next header" and not the "protocol." An IPv4 datagram has only a single protocol header following the IPv4 header.

  Early specifications of IPv4 had routers decrement the TTL by either one or the number of seconds that the router held the packet, whichever was greater. Hence the name "time-to-live."

In reality, however, the field was always decremented by one. IPv6 calls for its hop limit field to always be decremented by one, hence the name change from IPv4.

The most significant change from IPv4 to IPv6 is, of course, the larger IPv6 address fields. Another change is simplifying the IPv6 header because the simpler the header, the faster the header can be processed by a router. We can note other changes between the headers.

- There is no IPv6 header length field since there are no options in the header. There are optional headers that follow the fixed 40-byte IPv6 header, but each of these has its own length field.
- The two IPv6 addresses are aligned on a 64-bit boundary if the header itself is 64-bit aligned. This can speed up processing on 64-bit architectures.
- There are no fragmentation fields in the IPv6 header because there is a separate fragmentation header for this purpose. This design decision was made because fragmentation is the exception, and exceptions should not slow down normal processing.
- The IPv6 header does not include its own checksum. This is because all the upper layers—TCP, UDP, and ICMPv6—have their own checksum that includes the upper-layer header, the upper-layer data, and the following fields from the IPv6 header: IPv6 source address, IPv6 destination address, payload length, and next header. By omitting the checksum from the header, routers that forward the packet need not recalculate a header checksum after they modify the hop limit. Again, speed of forwarding by routers is the key point.

In case this is your first encounter with IPv6, we also note the following major differences from IPv4 to IPv6:

- There is no broadcasting with IPv6 (Chapter 18). Multicasting (Chapter 19), which is optional with IPv4, is mandatory with IPv6.
- IPv6 routers do not fragment packets that they forward. Fragmentation with IPv6 is performed only by the originating host.
- IPv6 requires support for authentication and security options.
- IPv6 requires support for path MTU discovery (Section 2.9). Technically this support is optional and could be omitted from minimal implementations such as bootstrap loaders, but if a node does not implement this feature, it cannot send datagrams larger than the IPv6 minimum link MTU (576 bytes; Section 2.9).

A.4 IPv4 Addresses

A 32-bit IPv4 address has one of the five formats shown in Figure A.3. Historically, an organization was assigned either a class A, B, or C network ID and it could do whatever

it wanted with the host ID portion of the address. But that changed in the mid-1990s with the advent of *classless* addresses, which we talk about shortly.

IPv4 addresses are usually written as four decimal numbers, separated by decimal points. This is called *dotted-decimal notation* and each decimal number represents one of the 4 bytes of the 32-bit address. The first of the four decimal numbers identifies the address class, as shown in Figure A.4.

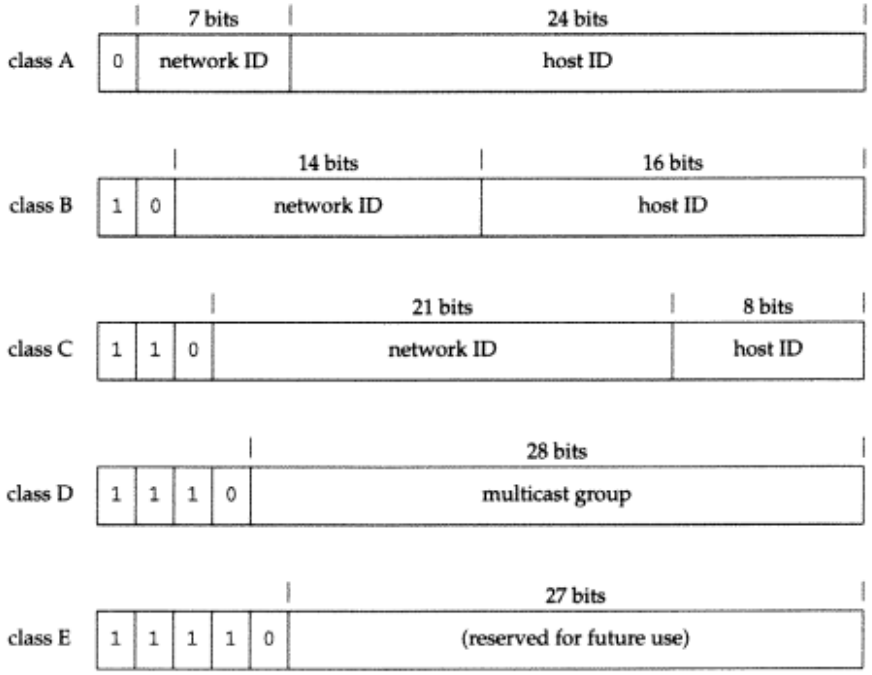


Figure A.3 IPv4 address formats.

Class	Range
A	**0**.0.0.0 to **127**.255.255.255
B	**128**.0.0.0 to **191**.255.255.255
C	**192**.0.0.0 to **223**.255.255.255
D	**224**.0.0.0 to **239**.255.255.255
E	**240**.0.0.0 to **247**.255.255.255

Figure A.4 Ranges for the five different classes of IPv4 addresses.

Classless Addresses and CIDR

IPv4 addresses are now considered classless. That is, we can ignore the distinction between class A, B, and C addresses and the implied boundaries between the network ID and the host ID shown in Figure A.3 for class A, B, and C addresses. Instead,

whenever an IPv4 network address is assigned to an organization, what is assigned is a 32-bit network address and a corresponding 32-bit mask. Bits of 1 in the mask cover the network address and bits of 0 in the mask cover the host. Since the bits of 1 in the mask are always contiguous from the leftmost bit, and the bits of 0 in the mask are always contiguous from the rightmost bit, this address mask can also be specified as a *prefix length* that denotes the number of contiguous bits of 1 starting from the left. For example, class A addresses have an implied mask of 255.0.0.0 or a prefix length of 8, class B addresses have an implied mask of 255.255.0.0 or a prefix length of 16, and class C addresses have an implied mask of 255.255.255.0 or a prefix length of 24.

But the advantage of classless addresses is that we are no longer restricted to just these three fixed prefix lengths of 8, 16, and 24 for class A, B, and C addresses. Instead, addresses can be assigned with different prefix lengths. For example, using classless addresses an Internet Service Provider (ISP) can take one class C address and assign it to four different customers, each with a mask of 255.255.255.192, which is a prefix length of 26. Each of the four customers then has 6 bits (instead of 8) to play with, in terms of choosing a subnet boundary (if any) and then assigning subnet IDs and host IDs. (We talk about subnetting shortly.)

All IPv4 addresses assigned today for the Internet are classless. The same concept is used with IPv6 addresses. IPv4 network addresses are normally written as a dotted-decimal number, followed by a slash, followed by the prefix length. Figure 1.16 showed examples of this.

Using classless addresses requires classless routing, and this is normally called CIDR (RFC 1519 [Fuller et al. 1993]). The goals of CIDR usage are to decrease the size of the Internet backbone routing tables and to reduce the rate of IPv4 address depletion. Section 10.8 of TCPv1 talks more about CIDR.

Subnet Addresses

IPv4 addresses are normally *subnetted* (RFC 950 [Mogul and Postel 1985]). This adds another level to the address hierarchy:

- network ID (assigned to site),
- subnet ID (chosen by site), and
- host ID (chosen by site).

The boundary between the network ID and the subnet ID is fixed by the prefix length of the assigned network address. This prefix length is normally assigned by the organization's ISP. But the boundary between the subnet ID and the host ID is chosen by the site. All the hosts on a given subnet share a common *subnet mask* and this mask specifies the boundary between the subnet ID and the host ID. Bits of 1 in the subnet mask cover the network ID and subnet ID, and bits of 0 cover the host ID.

For example, consider the author's subnets in Figure 1.16. The assigned network address from the ISP is 206.62.226.0/24, which is an entire class C network. The author then divides the remaining 8 bits into a 3-bit subnet ID and a 5-bit host ID. Figure A.5 shows this. The subnet mask for these addresses is `0xffffffe0` or 255.255.255.224.

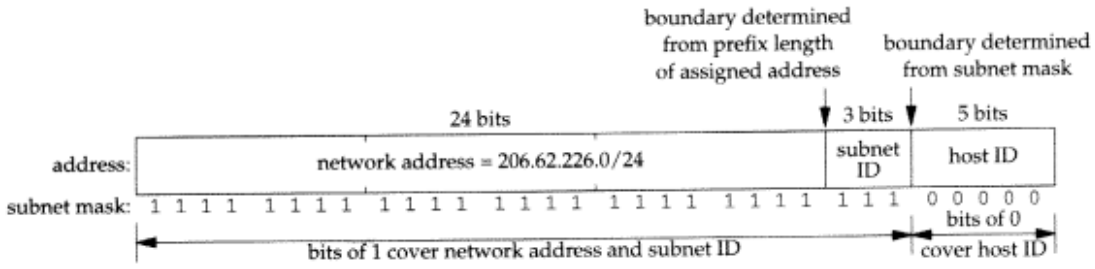


Figure A.5 24-bit network address with 3-bit subnet ID and 5-bit host ID.

The top subnet in Figure 1.16 has the three subnet bits set to `001` and we designate this subnet as 206.62.226.32/27. The "/27" notation indicates that the subnet mask comprises the leftmost 27 bits. We are using this prefix notation for both the overall network address, 206.62.226.0/24, and for the subnet addresses, such as 206.62.226.32/27. The hosts on this subnet will have addresses between 206.62.226.33 and 206.62.226.62, and the address with a host ID of all one bits, 206.62.226.63, is the subnet broadcast address (Section 18.2). The other subnet in Figure 1.16 has the three subnet bits set to `010` and we designate this as 206.62.226.64/27.

> When the subnetting of IP addresses started in the mid-1980s a noncontiguous subnet mask was allowed but not recommended (RFC 950 [Mogul and Postel, 1985]). But with the current use of classless addresses, noncontiguous subnet masks are no longer allowed. IPv6 also requires that all address masks be contiguous starting at the leftmost bit.
>
> RFC 950 recommends not using the two subnets with a subnet ID of all zero bits or all one bits. Some software today supports these two forms of subnet IDs.

As another example of subnetting from Figure 1.16, consider the bottom subnet. The assigned network address for `noao.edu` is 140.252.0.0/16, which is an entire class B network. NOAO then divides the remaining 16 bits into an 8-bit subnet ID and an 8-bit host ID, which is typical for organizations with class B addresses. We show this in Figure A.6.

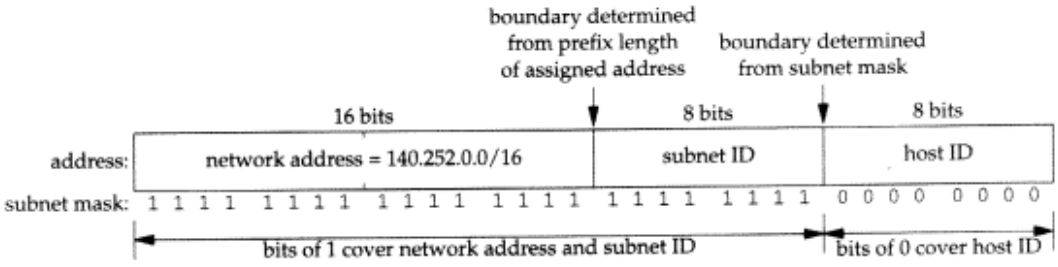


Figure A.6 16-bit network address with 8-bit subnet ID and 8-bit host ID.

The subnet mask is `0xffffff00` or 255.255.255.0. The subnet that we show has a subnet ID of 1 and we designate this subnet as 140.252.1.0/24.

Loopback Addresses

By convention the address 127.0.0.1 is assigned to the loopback interface. Anything sent to this IP address loops around and becomes IP input. We often use this address when testing a client and server on the same host. This address is normally known by the name `INADDR_LOOPBACK`.

> Any address on the network 127/8 can be assigned to the loopback interface, but 127.0.0.1 is common.

Unspecified Address

The address consisting of 32 zero bits is the IPv4 unspecified address. In an IPv4 packet it can appear only as the source address in packets sent by a node that is bootstrapping before the node learns its IP address. In the sockets API this address is called the wildcard address and is normally known by the name `INADDR_ANY`.

Multihoming and Address Aliases

Traditionally the definition of a *multihomed* host has been a host with multiple interfaces: two Ethernets, for example, or an Ethernet and a point-to-point link. Each interface must have a unique IPv4 address. When counting interfaces to determine if a host is multihomed, the loopback interface does not count.

A router, by definition, is multihomed since it forwards packets that arrive on one interface out another interface. But a multihomed host is not a router unless it forwards packets. Indeed, a multihomed host must not assume it is a router just because the host has multiple interfaces; it must not act as a router unless it has been configured to do so (typically by the administrator enabling a configuration option).

The term multihoming, however, is more general and covers two different scenarios (Section 3.3.4 of RFC 1122 [Braden 1989]).

1. A host with multiple interfaces is multihomed and each interface must have its own IP address. This is the traditional definition.
2. Newer hosts have the capability of assigning multiple IP addresses to a given physical interface. Each additional IP address, after the first (primary), is called an *alias* or a *logical interface*. Often the aliased IP addresses share the same subnet address as the primary address but have different host IDs. But it is also possible for the aliases to have a completely different network address or subnet addresses from the primary. We show an example of aliased addresses in Section 16.6.

Hence the definition of a multihomed host is one with multiple interfaces, regardless of whether those interfaces are physical or logical.

> Multihoming is also used in another context. A network that has multiple connections to the Internet is also called multihomed. For example, some sites have two connections to the Internet, instead of one, providing a backup capability.

A.5 IPv6 Addresses

IPv6 addresses comprise 128 bits and are usually written as eight 16-bit hexadecimal numbers. There are no address classes, per se, as we have with IPv4; instead the high-order bits of the 128-bit address imply the type of address ([Hinden and Deering 1997]). Figure A.7 shows the different values of the high-order bits and what type of address these bits imply.

Allocation	Format prefix
reserved	0000 0000
unassigned	0000 0001
reserved for NSAP	0000 001
reserved for IPX	0000 010
unassigned	0000 011
unassigned	0000 1
unassigned	0001
aggregatable global unicast addresses	001
unassigned	010
unassigned	011
unassigned	100
unassigned	101
unassigned	110
unassigned	1110
unassigned	1111 0
unassigned	1111 10
unassigned	1111 110
unassigned	1111 1110 0
link-local unicast address	1111 1110 10
site-local unicast address	1111 1110 11
multicast addresses	1111 1111

Figure A.7 Meaning of high-order bits of IPv6 addresses.

These high-order bits are called the *format prefix*. For example, if the high-order 3 bits are `001`, the address is called an *aggregatable global unicast address*. If the high-order 8 bits are `11111111` (`0xff`), it is a multicast address. If the high-order 8 bits are `00000000`, the address is reserved, and we will see some examples of these addresses.

Aggregatable Global Unicast Addresses

Probably the most common form of IPv6 address will be the aggregatable global unicast address, which in Figure A.7 begins with a 3-bit prefix of `001`. These addresses will replace the IPv4 class A, B, and C addresses.

> The original IPv6 address specification, RFC 1884 [Hinden and Deering 1995], called for *provider-based unicast addresses* with the 3-bit prefix of `010`. These are described in RFC 2073 [Rekhter et al. 1997]. At the IETF (Internet Engineering Task Force) meeting in March 1997, however, the decision was made to proceed with a different form of unicast address.

The format of aggregation-based unicast addresses are defined in [Hinden, O'Dell, and Deering 1997] and contain the following fields, starting at the leftmost bit and going right:

- format prefix (001),
- TLA ID (top-level aggregation identifier),
- NLA ID (next-level aggregation identifier),
- SLA ID (site-level aggregation identifier; e.g., subnet ID), and
- interface identifier.

Figure A.8 shows an example of a aggregatable global unicast address.

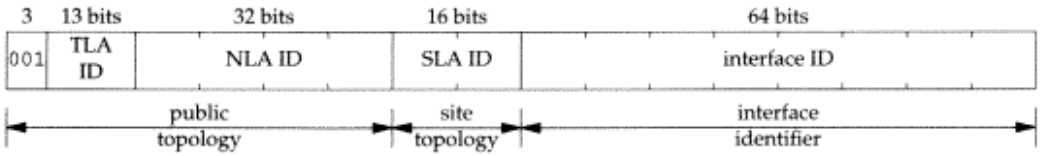


Figure A.8 IPv6 aggregatable global unicast addresses.

The interface ID must be constructed in IEEE *EUI-64* format [IEEE 1997b]. This is a superset of the 48-bit IEEE 802 MAC addresses that are assigned to most LAN interface cards. This identifier should be automatically assigned for an interface, based on its hardware MAC address when possible. Details for constructing EUI-64 based interface identifiers are in Appendix A of [Hinden and Deering 1997].

6bone Test Addresses

The 6bone is a virtual network used for early testing of the IPv6 protocols (Section B.3). As of this writing the aggregatable global unicast addresses are not yet being assigned, although there are plans to use a special format of these addresses on the 6bone ([Hinden, Fink, and Postel 1997]). Instead, the address format shown in Figure A.9, which is documented in RFC 1897 [Hinden and Postel 1996], is being used for all nodes on the 6bone.

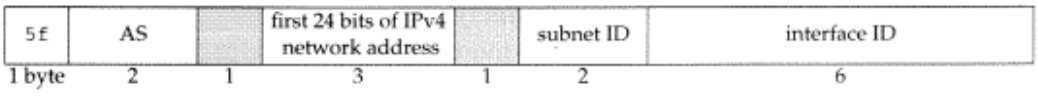


Figure A.9 IPv6 test addresses for 6bone.

These addresses are considered temporary, and nodes using these addresses will have to renumber when aggregatable global unicast addresses are assigned.

The high-order byte is 0x5f. The 16-bit *AS* field is the *autonomous system number* assigned to the organization or to its ISP. These are used with IPv4 to identify routing domains. The next field is the high-order 24 bits of the node's current IPv4 address. The *subnet ID* is whatever the organization chooses and the *interface ID* is normally the 48-bit IEEE 802 MAC address.

In Section 9.2 we showed the IPv6 address for the host `solaris` in Figure 1.16 as `5f1b:df00:ce3e:e200:0020:0800:2078:e3e3`. The AS is 7135 (`0x1bdf`) and 206.62.226 is `0xce3ee2`. The subnet ID is `0x0020` and the low-order 48 bits are the MAC address of the host's Ethernet card.

IPv4-Mapped IPv6 Addresses

IPv4-mapped IPv6 addresses allow IPv6 applications on hosts supporting both IPv4 and IPv6 to communicate with IPv4-only hosts during the transition of the Internet to IPv6. These addresses are automatically created by DNS resolvers (Figure 9.5) when a query is made by an IPv6 application for the IPv6 addresses of a host, but that host has only IPv4 addresses.

We saw in Figure 10.4 that using this type of address with an IPv6 socket causes an IPv4 datagram to be sent to the IPv4 host. These addresses are not stored in any DNS data files—they are created when needed by a resolver.

Figure A.10 shows the format of these addresses The low-order 32 bits contain an IPv4 address.

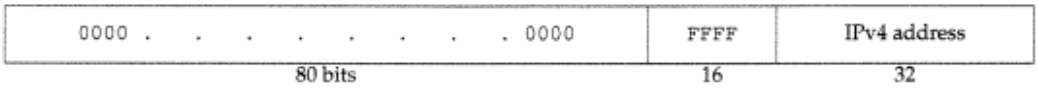


Figure A.10 IPv4-mapped IPv6 address.

When writing an IPv6 address, a consecutive string of zeros can be abbreviated with two colons. Also, the embedded IPv4 address is written using dotted-decimal notation. For example, we can abbreviate the IPv4-mapped IPv6 address `0:0:0:0:0:FFFF:206.62.226.33` as `::FFFF:206.62.226.33`.

IPv4-Compatible IPv6 Addresses

IPv4-compatible IPv6 addresses are also used during the transition from IPv4 to IPv6. The administrator for a host supporting both IPv4 and IPv6 that does not have a neighbor IPv6 router should create a DNS AAAA record containing an IPv4-compatible IPv6 address. Any other IPv6 host with an IPv6 datagram to send to an IPv4-compatible IPv6 address will then *encapsulate* the IPv6 datagram with an IPv4 header and this is called an *automatic tunnel*. We talk more about tunneling in Section B.3 and show an example of this type of IPv6 datagram encapsulated within an IPv4 header in Figure B.2. Each tunnel on the 6bone, however, is *configured* (e.g., set up by an administrator in a startup file), whereas with IPv4-compatible IPv6 addresses only the address is configured by hand (e.g., placed into a DNS data file as a AAAA record) and the tunneling is then automatic.

Figure A.11 shows the format of an IPv4-compatible IPv6 address.

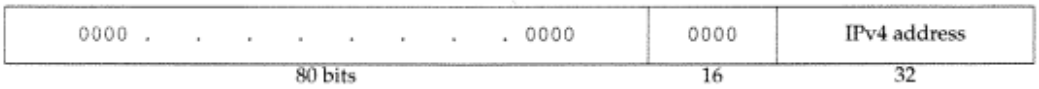


Figure A.11 IPv4-compatible IPv6 address.

An example of this type of address is `::206.62.226.33`.

Loopback Address

An IPv6 address consisting of 127 zero bits and a single one bit, written as `::1`, is the IPv6 loopback address. In the sockets API it is referenced as `in6addr_loopback` or `IN6ADDR_LOOPBACK_INIT`.

Unspecified Address

An IPv6 address consisting of 128 zero bits, written as `0::0` or just `::`, is the IPv6 unspecified address. In an IPv6 packet it can appear only as the source address in packets sent by a node that is bootstrapping, before the node learns its IPv6 address.

In the sockets API this address is called the wildcard address and when specifying it, for example, to `bind` for a listening TCP socket, indicates that the socket will accept client connections destined to any of the node's addresses. It is referenced as `in6addr_any` or `IN6ADDR_ANY_INIT`.

Link-Local Address

A link-local address is used on a single link, when it is known that the datagram will not be forwarded. Example uses are automatic address configuration at bootstrap time, and neighbor discovery (similar to IPv4's ARP). Figure A.12 shows the format of these addresses.

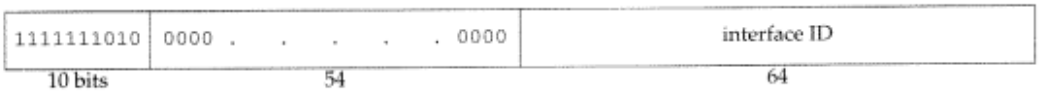


Figure A.12 IPv6 link-local address.

These addresses always begin with `fe80`. An IPv6 router must not forward a datagram with a link-local source or destination address to another link. In Section 9.2 we show the link-local address associated with the name `aix-6ll`.

Site-Local Address

Site-local addresses can be used for addressing within a site without the need for a global prefix. Figure A.13 shows the format of these addresses.

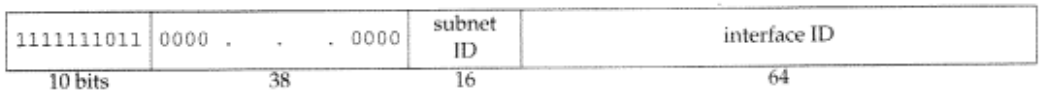


Figure A.13 IPv6 site-local address.

These addresses always begin with `fec0`. An IPv6 router must not forward a datagram with a site-local source or destination address outside of that site.

A.6 ICMPv4 and ICMPv6: Internet Control Message Protocol

ICMP is a required and integral part of any IPv4 or IPv6 implementation. ICMP is normally used to communicate error messages between IP nodes, both routers and hosts, but it is occasionally used by applications. The Ping and Traceroute applications (Chapter 25), for example, both use ICMP.

The first 32 bits of both ICMPv4 and ICMPv6 messages are the same and are shown in Figure A.14. RFC 792 [Postel 1981b] documents ICMPv4 and RFC 1885 [Conta and Deering 1995] documents ICMPv6.

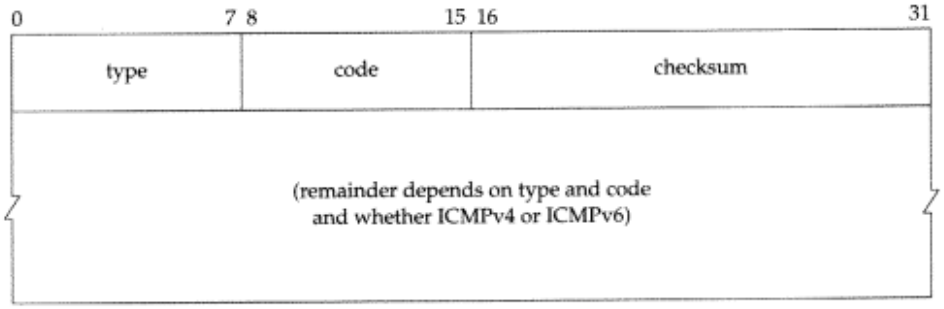


Figure A.14 Format of ICMPv4 and ICMPv6 messages.

The 8-bit *type* is the type of the ICMPv4 or ICMPv6 message and some types have an 8-bit *code* with additional information. The *checksum* is the standard Internet checksum although there is a difference in which fields are used for the ICMPv4 checksum versus the ICMPv6 checksum.

From a network programming perspective we need to understand which ICMP messages can be returned to an application, what causes the error, and how that error is returned to the application. Figure A.15 lists all the ICMPv4 messages and how they are handled by 4.4BSD. The final column indicates the `errno` value returned by those messages that return an error to the application. Figure A.16 lists the ICMPv6 messages.

type	*code*	Description	Handled by or `errno`
0	0	echo reply	user process (Ping)
3		destination unreachable:	
	0	network unreachable	`EHOSTUNREACH`
	1	host unreachable	`EHOSTUNREACH`
	2	protocol unreachable	`ECONNREFUSED`
	3	port unreachable	`ECONNREFUSED`
	4	fragmentation needed but DF bit set	`EMSGSIZE`
	5	source route failed	`EHOSTUNREACH`
	6	destination network unknown	`EHOSTUNREACH`
	7	destination host unknown	`EHOSTUNREACH`
	8	source host isolated (obsolete)	`EHOSTUNREACH`
	9	destination network administratively prohibited	`EHOSTUNREACH`
	10	destination host administratively prohibited	`EHOSTUNREACH`
	11	network unreachable for TOS	`EHOSTUNREACH`
	12	host unreachable for TOS	`EHOSTUNREACH`
	13	communication administratively prohibited	(ignored)
	14	host precedence violation	(ignored)
	15	precedence cutoff in effect	(ignored)
4	0	source quench	kernel for TCP, ignored by UDP
5		redirect:	
	0	redirect for network	kernel updates routing table
	1	redirect for host	kernel updates routing table
	2	redirect for type-of-service and network	kernel updates routing table
	3	redirect for type-of-service and host	kernel updates routing table
8	0	echo request	kernel generates reply
9	0	router advertisement	user process
10	0	router solicitation	user process
11		time exceeded:	
	0	TTL equals 0 during transit	user process
	1	TTL equals 0 during reassembly	user process
12		parameter problem:	
	0	IP header bad (catchall error)	`ENOPROTOOPT`
	1	required option missing	`ENOPROTOOPT`
13	0	timestamp request	kernel generates reply
14	0	timestamp reply	user process
15	0	information request (obsolete)	(ignored)
16	0	information reply (obsolete)	user process
17	0	address mask request	kernel generates reply
18	0	address mask reply	user process

Figure A.15 Handling of the ICMP message types by 4.4BSD.

type	*code*	Description	Handled by or `errno`
1		destination unreachable:	
	0	no route to destination	EHOSTUNREACH
	1	administratively prohibited (firewall filter)	EHOSTUNREACH
	2	not a neighbor (incorrect strict source route)	EHOSTUNREACH
	3	address unreachable (any other reason)	EHOSTDOWN
	4	port unreachable (UDP)	ECONNREFUSED
2	0	packet too big	kernel does PMTU discovery
3		time exceeded:	
	0	hop limit exceeded in transit	user process
	1	fragment reassembly time exceeded	user process
4		parameter problem:	
	0	erroneous header field	ENOPROTOOPT
	1	unrecognized next header	ENOPROTOOPT
	2	unrecognized option	ENOPROTOOPT
128	0	echo request (Ping)	kernel generates reply
129	0	echo reply (Ping)	user process (Ping)
130	0	group membership query	user process
131	0	group membership report	user process
132	0	group membership reduction	user process
133	0	router solicitation	user process
134	0	router advertisement	user process
135	0	neighbor solicitation	user process
136	0	neighbor advertisement	user process
137	0	redirect	kernel updates routing table

Figure A.16 ICMPv6 messages.

The notation "user process" means that the kernel does not process the message and it is up to a user process with a raw socket to handle the message. We must also note that different implementations may handle certain messages differently. For example, although Unix systems normally handle router solicitations and router advertisements in a user process, other implementations might handle these messages in the kernel.

ICMPv6 clears the high-order bit of the *type* field for the error messages (*types* 1–4) and sets this bit for the informational messages (*types* 128–137).

Appendix B

Virtual Networks

B.1 Introduction

When a new feature is added to TCP, such as the long fat pipe support defined in RFC 1323, support is required only in the hosts using TCP; no changes are required in the routers. These RFC 1323 changes, for example, are slowly appearing in host implementations of TCP and when a new TCP connection is established each end can determine if the other end supports the new feature. If both hosts support the feature, it can be used.

This differs from changes being made to the IP layer, such as multicasting at the end of the 1980s and IPv6 in the mid-1990s, because these new features require changes in all the hosts *and* all the routers. But people want to start using the new features without having to wait for all the systems to be upgraded. To do this a *virtual network* is established on top of the existing IPv4 Internet using *tunnels*.

B.2 The MBone

Our first example of a virtual network that is built using tunnels is the MBone, which started around 1992 [Eriksson 1994]. If two or more hosts on a LAN support multicasting, multicast applications can be run on all these hosts and communicate with each other. To connect this LAN to some other LAN that also has multicast-capable hosts a tunnel is configured between one host on each of the LANs, as shown in Figure B.1. We show the following numbered steps in this figure.

1. An application on the source host, MH1, sends a multicast datagram to a class D address.

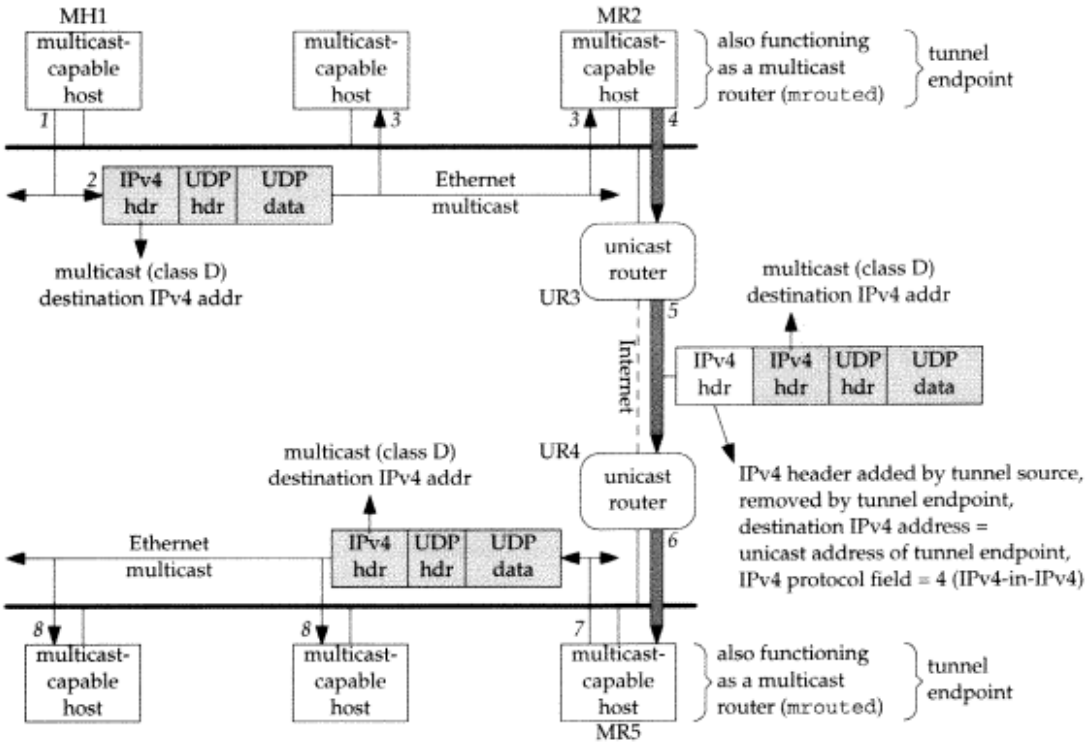


Figure B.1 IPv4-in-IPv4 encapsulation used on MBone.

2. We show this as a UDP datagram, since most multicast applications use UDP. We talk more about multicasting and how to send and receive multicast datagrams in Chapter 19.
3. The datagram is received by all the multicast-capable hosts on the LAN, including MR2. We note that MR2 is also functioning as a multicast router, running the `mrouted` program, which performs multicast routing.
4. MR2 prepends another IPv4 header at the front of the datagram with the destination IPv4 address of this new header set to the unicast address of the tunnel endpoint, MR5. This unicast address is configured by the administrator of MR2 and is read by the `mrouted` program when it starts up. Similarly the unicast address of MR2 is configured for MR5, the other end of the tunnel. The protocol field in the new IPv4 header is set to 4, which is the value for IPv4-in-IPv4 encapsulation. The datagram is sent to the next-hop router, UR3, which we explicitly denote as a unicast router. That is, UR3 does not understand multicasting, which is the whole reason we are using a tunnel. The shaded portion of the IPv4 datagram has not changed from what was sent in step 1, other than the decrementing of the TTL field in the shaded IPv4 header.
5. UR3 looks at the destination IPv4 address in the outermost IPv4 header and forwards the datagram to the next-hop router, UR4, another unicast router.

6. UR4 delivers the datagram to its destination, MR5, the tunnel endpoint.
7. MR5 receives the datagram and since the protocol field indicates IPv4-in-IPv4 encapsulation, it removes the first IPv4 header and then outputs the remainder of the datagram (a copy of what was multicast on the top LAN) as a multicast datagram on its LAN.
8. All the multicast-capable hosts on the lower LAN receive the multicast datagram.

The end result is that the multicast datagram sent on the top LAN also gets transmitted as a multicast datagram on the lower LAN. This occurs even though the two routers that we show attached to these two LANs, and all the Internet routers between these two routers, are not multicast capable.

In this example we show the multicast routing function being performed by the `mrouted` program running on one host on each LAN. This is how the MBone started. But around 1996 multicast routing functionality started appearing in the routers from most major router vendors. If the two unicast routers UR3 and UR4 in Figure B.1 were multicast capable, then we would not need to run `mrouted` at all, and UR3 and UR4 would function as the multicast routers. But if there still exist other routers between UR3 and UR4 that are not multicast capable, then a tunnel is still required. But the tunnel endpoints would then be MR3 (a multicast-capable replacement for UR3) and MR4 (a multicast-capable replacement for UR4), not MR2 and MR5.

> In the scenario that we show in Figure B.1 every multicast packet appears twice on the top LAN and twice on the bottom LAN: once as a multicast packet, and again as a unicast packet within the tunnel as the packet goes between the host running `mrouted` and the next-hop unicast router (e.g., between MR2 and UR3, and between UR4 and MR5). This extra copy is the cost of tunneling. The advantage in replacing the two unicast routers UR3 and UR4 in Figure B.1 with multicast-capable routers (what we called MR3 and MR4) is to avoid this extra copy of every multicast packet from appearing on the LANs. Even if MR3 and MR4 must then establish a tunnel between themselves, because some intermediate routers between them (that we do not show) are not multicast capable, this is still advantageous since it avoids the duplicate copies on each LAN.

B.3 The 6bone

The 6bone is a virtual network that was created in 1996 for reasons similar to the MBone: users with islands of IPv6-capable hosts wanted to connect them together using a virtual network without waiting for all the intermediate routers to become IPv6-capable. Figure B.2 shows an example of two IPv6-capable LANs connected with a tunnel across IPv4-only routers. We show the following numbered steps in this figure.

1. Host H1 on the top LAN sends an IPv6 datagram containing a TCP segment to host H4 on the bottom LAN. We designate these two hosts as "IPv6 hosts" but both probably run IPv4 also. The IPv6 routing table on H1 specifies that host HR2 is the next-hop router and an IPv6 datagram is sent to this host

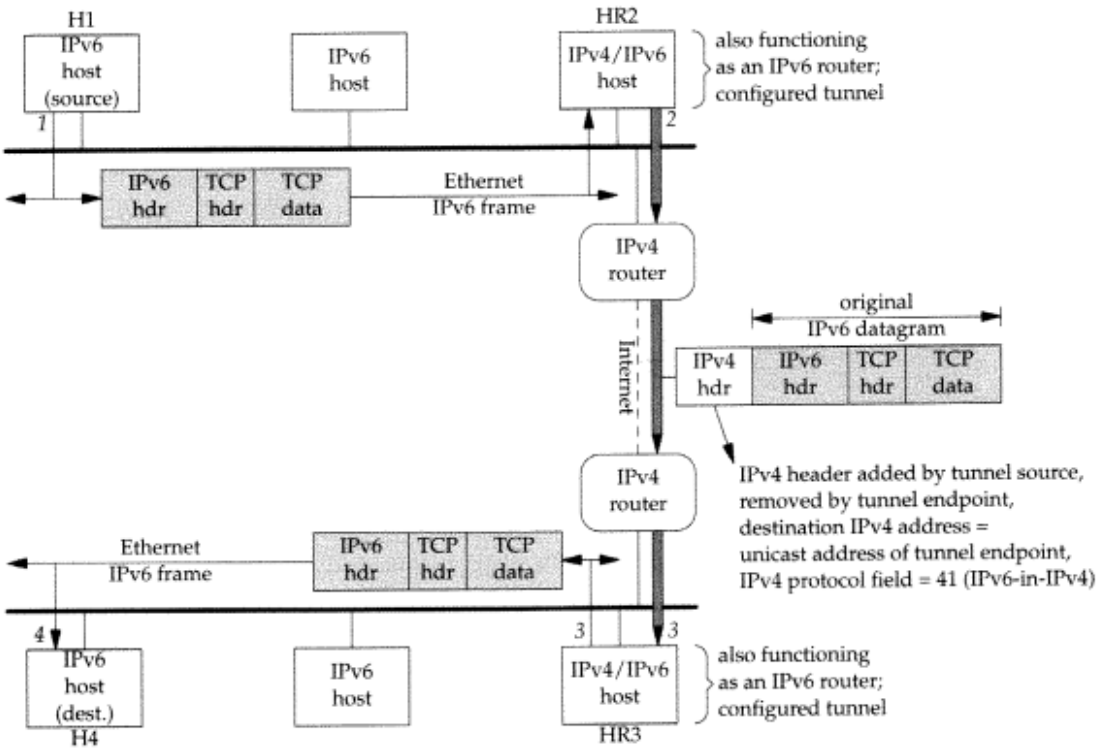


Figure B.2 IPv6-in-IPv4 encapsulation on 6bone.

2. Host HR2 has a configured tunnel to host HR3. This configured tunnel allows IPv6 datagrams to be sent between the two tunnel endpoints across an IPv4 internet by encapsulating the IPv6 datagram in an IPv4 datagram (called "IPv6-in-IPv4 encapsulation"). The IPv4 protocol field has a value of 41. We note that the two IPv4/IPv6 hosts at the ends of the tunnel, HR2 and HR3, are both acting as IPv6 routers since they are forwarding IPv6 datagrams that they receive on one interface out another interface. The configured tunnel counts as an interface, even though it is a virtual interface, and not a physical interface.
3. The tunnel endpoint, HR3, receives the encapsulated datagram, strips off the IPv4 header, and sends the IPv6 datagram onto its LAN.
4. The destination, H4, receives the IPv6 datagram.

The goal for these virtual networks is that over time, as the intermediate routers gain the required functionality (multicast routing in terms of the MBone and IPv6 routing in terms of the 6bone), the virtual networks disappear. We describe both of these virtual networks since some of the examples in the text use the MBone and the 6bone.

Appendix C

Debugging Techniques

This appendix contains some hints and techniques for debugging network applications. No one technique is the answer for everyone; instead there are various tools that we should be familiar with, and then use whatever works in our environment.

C.1 System Call Tracing

Many versions of Unix provide a system call tracing facility. This can often provide a valuable debugging technique.

Working at this level we need to differentiate between a *system call* and a *function*. The former is an entry point into the kernel and that is what we are able to trace with the tools we will look at in this section. Posix, and most other standards, use the term function to describe what appear to the users to be functions, even though on some implementations they may be system calls. For example, on a Berkeley-derived kernel `socket` is a system call even though it appears to be a normal C function to the application programmer. But under SVR4 we will see shortly that it is a library function in the sockets library that issues calls to `putmsg` and `getmsg`, these latter two being the actual system calls.

In this section we examine the system calls involved in running our daytime client. We showed the sockets version in Figure 1.5 and the XTI version in Figure 28.13.

SVR4 Streams-Based Sockets Library

We start with a streams-based implementation of sockets, as found under SVR4. The `truss` program is provided by SVR4 to run a program and trace the system calls that are executed. We run this program as

```
unixware % truss  -o truss.out  -v getmsg,putmsg,ioctl \
daytimetcpcli  140.252.1.54
```

(We have wrapped this long command onto two lines.) The `-o` option directs the output to a file (there is often lots of output from this technique), and the `-v` option turns on verbose tracing for the three specified system calls. This will show us additional information about the arguments for these system calls.

The beginning of the output (about 40 lines) deals with linking the program with dynamic libraries using memory-mapped I/O. This is done using `open` and `mmap`. (The latter system call is described in Section 12.9 of APUE and is not relevant to our discussion of networking APIs.) We omit these lines of output.

Next is the opening of the `/etc/netconfig` file, followed by a `read` of the entire file (806 bytes).

```
open("/etc/netconfig", O_RDONLY, 0666)          = 3
 ...
read(3, " t c p\t t p i _ c o t s"..., 8192)    = 806
read(3, 0x0804A6B8, 8192)                       = 0
 ...
close(3)                                        = 0
```

We have omitted some calls to `ioctl` and `lseek` that are not relevant to our discussion. Whenever we do this we show a line with three dots. The value of 3 at the right side of the equals sign for the call to `open` is the returned descriptor. The call to `read` asks for 8192 bytes but the return value is 806 (the size of the file). The next call to `read` returns 0 (end-of-file). `truss` also shows the first 12 bytes returned by `read` (beginning with `tcp`) and this is the start of the first line of the file. The file is then closed. We would guess that this reading of the `netconfig` file is finished when the sockets library starts, with our first call to `socket`.

The next system call is an `open` of `/dev/tcp` and the returned descriptor is again 3.

```
open("/dev/tcp", O_RDWR, 027776333624)          = 3
ioctl(3, I_FIND, "sockmod")                     = 0
ioctl(3, I_PUSH, "sockmod")                     = 0
 ...
ioctl(3, I_STR, 0x080467E4)                     = 0
        cmd=TI_BIND timout=-1 len=32 dp=0x0804ABA8
 ...
```

The `ioctl` that follows the `open` checks whether the module `sockmod` is already on the stream. The return value is 0 indicating that it is not on the stream, so an `ioctl` of `I_PUSH` pushes it onto the stream. This is followed by numerous `ioctl`s for the stream and lots of signal handling (which we omit). The `ioctl` of `I_STR` sends an internal streams `ioctl` message and the command is `TI_BIND`. The length of 32 probably indicates a `T_BIND_REQ`, which consists of four 4-byte fields in the `T_bind_req` structure (which we discussed with Figure 33.9), followed by a 16-byte `sockaddr_in` structure. This is probably a bind of any local address being performed by the sockets library when we call `connect` on an unbound socket.

We would expect to see a `putmsg` followed by a `getmsg` here, as in our `tpi_bind` function.

We then see the first call to `putmsg` and it is for control information only, with a flag of 0.

```
putmsg(3, 0x08046958, 0x00000000, 0)            = 0
        ctl:  maxlen=428  len=36   buf=0x0804ABA8
getmsg(3, 0x08046924, 0x08046918, 0x08046934)   = 0
        ctl:  maxlen=428  len=8    buf=0x0804ABA8
        dat:  maxlen=128  len=-1   buf=0x08046898
getmsg(3, 0x08046964, 0x08046958, 0x0804697C)   = 0
        ctl:  maxlen=428  len=56   buf=0x0804ABA8
        dat:  maxlen=128  len=-1   buf=0x080468D8
```

The length of 36 is probably a `T_CONN_REQ` request (Figure 33.10): five 4-byte values followed by a 16-byte `sockaddr_in` structure. The next `getmsg` returns 8 bytes of control information without any data, and this is probably a `T_OK_ACK` message (Figure 33.10). The next `getmsg` returns 56 bytes of control information without any data, and this is probably a `T_CONN_CON` message (Figure 33.10): five 4-byte fields, a 16-byte `sockaddr_in` structure, and 20 bytes of options. We can only guess that the final 20 bytes are options, based on the size of a `t_opthdr` structure (four 4-byte fields, Section 32.2), followed by the 1-byte `IP_TOS` option (the only end-to-end option from Figure 32.1 that we would expect to have returned at this point), followed by 3 bytes of padding (see also Figure 32.3).

The next system call is `getmsg`, and it returns 8 bytes of control information and 26 bytes of data. This is probably a `T_DATA_IND` message.

```
getmsg(3, 0x08046AEC, 0x08046AE0, 0x08046B14)   = 0
        ctl:  maxlen=428  len=8    buf=0x0804ABA8
        dat:  maxlen=4096 len=26   buf=0x08046BB0
write(1, " F r i   A p r     4   0".., 26)       = 26
getmsg(3, 0x08046AEC, 0x08046AE0, 0x08046B14)   = 0
        ctl:  maxlen=428  len=-1   buf=0x0804ABA8
        dat:  maxlen=4096 len=0    buf=0x08046BB0
_exit(0)
```

Our client then calls `write` to standard output (descriptor 1) for the 26 bytes of data. The next call to `getmsg` returns no control information and 0 bytes of data, probably an indication from the provider that the end-of-file has been encountered.

We would expect to receive a `T_ORDREL_IND` message at this point.

SVR4 Streams-Based XTI Library

Our next example is a streams-based XTI library, Solaris 2.6. We expect the system calls to be similar to our calls in Section 33.6.

After the library mapping with `mmap`, we find the call to `open` for the TCP transport provider, followed by a check for `timod` and then a push of this module onto the stream.

```
 ...
open("/dev/tcp", O_RDWR)                        = 3
ioctl(3, I_FIND, "timod")                       = 0
ioctl(3, I_PUSH, "timod")                       = 0
```

We then find numerous calls to `ioctl` along with signal handling (which we omit). One of these calls to `ioctl` appears to be in response to our call to `t_bind`, and it appears that the XTI library does the bind by issuing this `ioctl` to `timod`. (This is similar to what we saw with the sockets-over-streams example earlier with UnixWare.)

The first call to `putmsg` sends 36 bytes of control information and no data. This is probably a `T_CONN_REQ` request from our call to `t_connect`.

```
 ...
putmsg(3, 0xEFFFE7F4, 0x00000000, 0)            = 0
        ctl:  maxlen=912  len=36   buf=0x0002BAB8: "\0\0\0\0\0\0\010"..
getmsg(3, 0xEFFFE710, 0xEFFFE700, 0xEFFFE71C)   = 0
        ctl:  maxlen=912  len=8    buf=0x0002C1E8: "\0\0\013\0\0\0\0"
        dat:  maxlen=0    len=-1   buf=0x00000000
        flags:  0x0001
getmsg(3, 0xEFFFE7F4, 0xEFFFE770, 0xEFFFE77C)   = 0
        ctl:  maxlen=912  len=36   buf=0x0002BAB8: "\0\0\0\f\0\0\010"..
        dat:  maxlen=0    len=-1   buf=0x00000000
        flags:  0x0000
```

We also see that Solaris prints the first 8 bytes of the buffer in hexadecimal with C escapes. For the call to `putmsg` we have 7 bytes of 0 followed by a byte of `0x10` (16). But this is really two 4-byte fields: the first 4 bytes are the `T_CONN_REQ` request (which has a value of 0), followed by the length of the destination address (16).

The first call to `getmsg` in the previous output returns a `T_OK_ACK` message, and the next returns a `T_CONN_CON` message. We can tell the type of the first returned message because the value of `T_OK_ACK` is 19 (`0x13`) and the next 4 bytes indicate the primitive that is being acknowledged (the `T_CONN_REQ`, which we said has a value of 0). We can tell the type of the second returned message because the value of `T_CONN_CON` is 12 (which is printed as the C escape for the formfeed character, `\f`), followed by the length of the peer's address (16, printed as `0x10`).

Solaris prints the flags variable that is pointed to by the final argument to `getmsg` and we see the first call returns a value of 1 (which is `MSG_HIPRI`, as we expect since the `T_OK_ACK` message is an `M_PCPROTO` message) and the second call returns a value of 0 (a normal message, as expected). We also notice that the Solaris `T_CONN_CON` does not appear to return any options (the length of 36), whereas our UnixWare example earlier appeared to return 20 bytes of options.

The next system call of interest is another call to `getmsg`, probably in response to our call to `t_rcv`. This call returns 26 bytes of data and no control information, probably an `M_DATA` message with the server's response.

```
getmsg(3, 0xEFFFE7EC, 0xEFFFE7DC, 0xEFFFE81C)   = 0
        ctl:  maxlen=912  len=-1   buf=0x0002BAB8
        dat:  maxlen=4096 len=26   buf=0xEFFFE8D0: " F r i   A p r  "..
        flags:  0x0000
write(1, " F r i   A p r     4   1"..., 26)      = 26
getmsg(3, 0xEFFFE7EC, 0xEFFFE7DC, 0xEFFFE81C)   = 0
        ctl:  maxlen=912  len=4    buf=0x0002BAB8: "\0\0\017"
        dat:  maxlen=4096 len=-1   buf=0xEFFFE8D0
        flags:  0x0000
 ...
_exit(0)
```

Our client calls `write` and the next call to `getmsg` returns a `T_ORDREL_IND` message (4 bytes of control with no data). The value of `T_ORDREL_IND` is 23, which is printed as `0x17`.

BSD Kernel Sockets

Our next example is BSD/OS, a Berkeley-derived kernel in which all the socket functions are system calls. The system call tracing program is `ktrace`. This writes the trace information to a file (whose default name is `ktrace.out`), which we print with `kdump`. We execute our sockets client as

```
bsdi % ktrace daytimetcpcli 206.62.226.43
Fri Apr  4 17:24:30 1997
```

We then execute `kdump` to output the trace information to standard output.

```
13187 daytimetcpcli CALL  socket(0x2,0x1,0)
13187 daytimetcpcli RET   socket 3

13187 daytimetcpcli CALL  connect(0x3,0xefbfc9a0,0x10)
13187 daytimetcpcli RET   connect 0

13187 daytimetcpcli CALL  read(0x3,0xefbfc9b0,0x1000)
13187 daytimetcpcli GIO   fd 3 read 26 bytes
      "Fri Apr  4 17:24:30 1997\r\n"
13187 daytimetcpcli RET   read 26/0x1a

...

13187 daytimetcpcli CALL  write(0x1,0x9000,0x1a)
13187 daytimetcpcli GIO   fd 1 wrote 26 bytes
      "Fri Apr  4 17:24:30 1997\r\n"
13187 daytimetcpcli RET   write 26/0x1a

13187 daytimetcpcli CALL  read(0x3,0xefbfc9b0,0x1000)
13187 daytimetcpcli GIO   fd 3 read 0 bytes
      ""
13187 daytimetcpcli RET   read 0

13187 daytimetcpcli CALL  exit(0)
```

13187 is the process ID. `CALL` identifies a system call, `RET` is the return, and `GIO` stands for generic process I/O. We see the calls to `socket` and `connect`, followed by the call to `read` that returns 26 bytes. Our client writes these bytes to standard output and then the next call to `read` returns 0 (end-of-file).

Solaris 2.6 Kernel Sockets

Solaris 2.x is based on SVR4 and all the releases before 2.6 have implemented sockets as shown in Figure 33.3. One problem, however, with all SVR4 implementations that implement sockets in this fashion, is that they rarely provide 100% compatibility with Berkeley-derived kernel sockets. To provide additional compatibility, Solaris 2.6 changes the implementation technique and implements sockets using a `sockfs` filesystem. This provides kernel sockets, as we can verify using `truss` on our sockets client.

```
solaris26 % truss -v connect daytimetcpcli 198.69.10.4
Sat Apr  5 11:32:07 1997
```

After the normal library linking, the first system call we see is to `so_socket`, a system call invoked by our call to `socket`.

```
so_socket(2, 2, 0, "", 1)                       = 3
connect(3, 0xEFFFE8C8, 16)                      = 0
        name = 198.69.10.4/13
read(3, " S a t   A p r     5   1"..., 4096)     = 26
 ...
Sat Apr  5 11:32:07 1997
write(1, " S a t   A p r     5   1"..., 26)      = 26
read(3, 0xEFFFE8D8, 4096)                       = 0
 ...
_exit(0)
```

The first three arguments to `so_socket` are our three arguments to `socket`.

We then see that `connect` is a system call, and `truss`, when invoked with the `-v connect` flag, prints the contents of the socket address structure pointed to by the second argument (the IP address and port number). The only system calls that we have replaced with ellipses are a few dealing with standard input and standard output.

One side effect of this new implementation is the addition of 18 system calls to the kernel.

C.2 Standard Internet Services

Be familiar with the standard Internet services described in Figure 2.13. We have used the daytime service many times for testing our clients. The discard service is convenient port to which we can send data. The echo service is similar to the echo server that we have used throughout the text.

Many sites now prevent access to these services through their firewalls, because of some denial-of-service attacks using these services in 1996 (Exercise 12.3). Nevertheless, you can hopefully use these services within your own network.

C.3 `sock` Program

The author's `sock` program first appeared in TCPv1, where it was frequently used to generate special case conditions, most of which were then examined in the text using `tcpdump`. The handy thing about the program is that it generates so many different scenarios, saving us from having to write special test programs.

We do not show the source code for the program in this text (it is over 2000 lines of C), but the source code is freely available (see the Preface).

The program operates in one of four modes, and each mode can use either TCP or UDP.

1. Standard input, standard output client (Figure C.1).

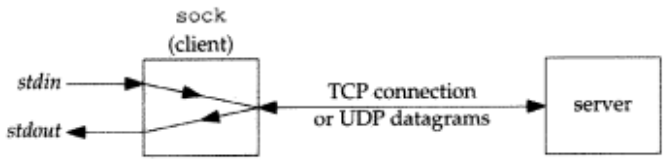


Figure C.1 sock client, standard input, standard output.

In the client mode everything read from standard input is written to the network, and everything received from the network is written to standard output. The server's IP address and port must be specified and in the case of TCP an active open is performed.

2. Standard input, standard output server. This mode is similar to the previous mode, except the program binds a well-known port to its socket and in the case of TCP performs a passive open.
3. Source client (Figure C.2).

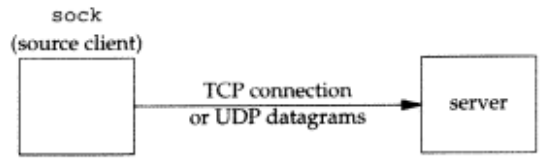


Figure C.2 sock program as source client.

The program performs a fixed number of writes to the network of some specified size.

4. Sink server (Figure C.3).

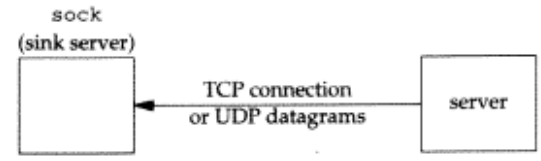


Figure C.3 sock program as sink server.

The program performs a fixed number of reads from the network.

These four operating modes correspond to the following four commands:

```
sock [ options ] hostname service
sock [ options ] -s [ hostname ] service
sock [ options ] -i hostname service
sock [ options ] -is [ hostname ] service
```

where *hostname* is a hostname or IP address and *service* is a service name or port number. In the two server modes the wildcard address is bound, unless the optional *hostname* is specified.

About 40 command-line options can also be specified, and these drive the optional features of the program. We will not detail these options here, but almost every socket option described in Chapter 7 can be set. Executing the program without any arguments prints a summary of the options:

```
-b n  bind n as client's local port number
-c    convert newline to CR/LF & vice versa
-f a.b.c.d.p  foreign IP address = a.b.c.d, foreign port# = p
-g a.b.c.d  loose source route
-h    issue TCP half-close on standard input EOF
-i    "source" data to socket, "sink" data from socket (w/-s)
-j a.b.c.d  join multicast group
-k    write or writev in chunks
-l a.b.c.d.p  client's local IP address = a.b.c.d, local port# = p
-n n  #buffers to write for "source" client (default 1024)
-o    do NOT connect UDP client
-p n  #ms to pause before each read or write (source/sink)
-q n  size of listen queue for TCP server (default 5)
-r n  #bytes per read() for "sink" server (default 1024)
-s    operate as server instead of client
-u    use UDP instead of TCP
-v    verbose
-w n  #bytes per write() for "source" client (default 1024)
-x n  #ms for SO_RCVTIMEO (receive timeout)
-y n  #ms for SO_SNDTIMEO (send timeout)
-A    SO_REUSEADDR option
-B    SO_BROADCAST option
-D    SO_DEBUG option
-E    IP_RECVDSTADDR option
-F    fork after connection accepted (TCP concurrent server)
-G a.b.c.d  strict source route
-H n  IP_TOS option (16=min del, 8=max thru, 4=max rel, 2=min cost)
-I    SIGIO signal
-J n  IP_TTL option
-K    SO_KEEPALIVE option
-L n  SO_LINGER option, n = linger time
-N    TCP_NODELAY option
-O n  #ms to pause after listen, but before first accept
-P n  #ms to pause before first read or write (source/sink)
-Q n  #ms to pause after receiving FIN, but before close
-R n  SO_RCVBUF option
-S n  SO_SNDBUF option
-T    SO_REUSEPORT option
-U n  enter urgent mode before write number n (source only)
-V    use writev() instead of write(); enables -k too
-W    ignore write errors for sink client
-X n  TCP_MAXSEG option (set MSS)
-Y    SO_DONTROUTE option
-Z    MSG_PEEK
-2    IP_ONESBCAST option (255.255.255.255) for broadcast
```

C.4 Small Test Programs

Another useful debugging technique, one that the author uses all the time when writing books, is writing small test programs to see how one specific feature works in a carefully constructed test case. It helps when writing small test programs to have a set of library wrapper functions and some simple error functions, such as the ones we have used throughout this text. This reduces the amount of code that we have to write but still provides the required testing for errors.

Example: XTI Out-of-Band Data, Which Band?

As an example of this technique, combined with system call tracing, we will answer the question: "how does XTI send out-of-band data in TCP?" We establish a TCP connection with a server, followed by a call to `t_snd`, sending 1 byte with the `T_EXPEDITED` flag set. Figure C.4 shows our simple test program.

debug/test01.c

```
 1 #include    "unpxti.h"
 2 int
 3 main(int argc, char **argv)
 4 {
 5     int     tfd;
 6     if (argc != 3)
 7         err_quit("usage: test01 <hostname/IPaddress> <service/port#>");
 8     tfd = Tcp_connect(argv[1], argv[2]);
 9     t_snd(tfd, "", 1, T_EXPEDITED);
10     exit(0);
11 }
```

debug/test01.c

Figure C.4 Simple test program to see how XTI sends TCP out-of-band data.

We then run this program under Solaris 2.6 using `truss` to trace the system calls.

```
solaris26 % truss -v putmsg,putpmsg test01 198.69.10.4 discard
```

The final lines of output are

```
putpmsg(3, 0xEFFFF7D4, 0xEFFFF7C0, 0, 0x0004)    = 0
        ctl:  maxlen=8    len=8    buf=0xEFFFF7CC: "\0\0\004\0\0\0\0"
        dat:  maxlen=1    len=1    buf=0x00015318: "\0"
```

which answers our question. The third argument to `putpmsg` is 0, the band number. The value of 4 in the first 4 bytes of the control buffer is `T_EXDATA_REQ` (expedited data request), indicating that the XTI library sends this message to the provider as a normal message in band 0.

Example: XTI Out-of-Band Data, Which `poll` Event?

Our next question regarding TCP out-of-band data and XTI is which of the possible input events from Figure 6.23 should we poll for when awaiting out-of-band data:

POLLIN, POLLRDNORM, POLLRDBAND, or POLLPRI? This time we start with our simple XTI server Figure 30.5, producing the program shown in Figure C.5.

debug/test03.c

```
 1 #include    "unpxti.h"

 2 int
 3 main(int argc, char **argv)
 4 {
 5     int     listenfd, connfd, n, flags;
 6     char    buff[MAXLINE];
 7     struct pollfd fds[1];

 8     if (argc == 2)
 9         listenfd = Tcp_listen(NULL, argv[1], NULL);
10     else if (argc == 3)
11         listenfd = Tcp_listen(argv[1], argv[2], NULL);
12     else
13         err_quit("usage: daytimetcpsrv01 [ <host> ] <service or port>");

14     connfd = Xti_accept(listenfd, NULL, 0);

15     fds[0].fd = connfd;
16     fds[0].events = POLLIN | POLLRDNORM | POLLRDBAND | POLLPRI;
17     for ( ; ; ) {
18         n = poll(fds, 1, INFTIM);
19         printf("poll returned %d, revents = 0x%x\n", n, fds[0].revents);

20         n = T_rcv(connfd, buff, sizeof(buff), &flags);
21         printf("received %d bytes, flags = %d\n", n, flags);
22     }
23 }
```

debug/test03.c

Figure C.5 Simple test program to check how to poll for TCP out-of-band data.

We set all four conditions in our input events, and print the returned events. We then call t_rcv and print the number of bytes returned along with the returned flags.

We want to send this program normal data and out-of-band data, and to do this we run our sock program on another host as the client:

```
solaris % sock -v -i -w 1 -n 3 -U 2 -p 4000 192.9.5.9 8888
connected on 206.62.226.33.34560 to 192.9.5.9.8888
TCP_MAXSEG = 1460
wrote 1 bytes
wrote 1 byte of urgent data
wrote 1 bytes
wrote 1 bytes
```

The -v flag turns on the verbose flag, -i causes the program to write data to the network (the source client mode), -w 1 says to write 1 byte at a time, -n 3 says to perform three writes, -U 2 says to write 1 byte of urgent data immediately before the second write, and -p 4000 causes a pause of 4000 ms (4 sec) after every write. We start our test program and then start our sock program, and here is the output from our test program:

```
solaris26 % test03 8888
poll returned 1, revents = 0x41
received 1 bytes, flags = 0
poll returned 1, revents = 0x41
received 1 bytes, flags = 2
poll returned 1, revents = 0x41
received 1 bytes, flags = 0
poll returned 1, revents = 0x41
received 1 bytes, flags = 0
poll returned 1, revents = 0x41
t_rcv error: An event requires attention
```

Each time the returned events are POLLIN and POLLRDNORM, telling us that the arrival of TCP's out-of-band data is not handled specially by poll. (We look up the values for the 2 bits in the returned event of 0x41 in the <sys/poll.h> header. When writing small test programs like this, it is normally easier and faster to print these values numerically and then look up the values in the appropriate headers.) But t_rcv returns a flag of 2 (T_EXPEDITED) when it returns the out-of-band byte.

C.5 tcpdump Program

An invaluable tool when dealing with network programming is a tool such as tcpdump. This program reads the packets from the network and prints lots of information about the packets. It also has the capability of printing only those packets that match some criteria that we specify. For example,

```
% tcpdump '(udp and port daytime) or icmp'
```

prints only the UDP datagrams with a source or destination port of 13 (the daytime server), or ICMP packets. The command

```
% tcpdump 'tcp and port 80 and tcp[13:1] & 2 != 0'
```

prints only the TCP segments with a source or destination port of 80 (the HTTP server) that have the SYN flag set. The SYN flag has a value of 2 in the byte with an offset of 13 from the start of the TCP header. The command

```
% tcpdump 'tcp and tcp[0:2] > 7000 and tcp[0:2] <= 7005'
```

prints only TCP segments with a source port between 7001 and 7005. The source port starts at byte offset 0 in the TCP header and occupies 2 bytes.

Appendix A of TCPv1 details the operation of this program in more detail.

> This program is available from ftp.ee.lbl.gov and works under many different flavors of Unix. It was written by Van Jacobson, Craig Leres, and Steven McCanne.
>
> Some vendors supply a program of their own with similar functionality. For example, Solaris 2.x provides the snoop program. The advantage in tcpdump is that it works under so many versions of Unix, and using a single tool in a heterogeneous environment, instead of a different tool for each environment, is a big advantage.

C.6 `netstat` Program

We have used the `netstat` program many times throughout the text. This program serves multiple purposes.

- It shows the status of networking endpoints. We showed this in Section 5.6, when we followed the status of our endpoint as we started our client and server.
- It shows the multicast groups that a host belongs to on each interface. The `-ia` flags are the normal way to show this, or the `-g` flags under Solaris 2.x.
- It shows the per-protocol statistics with the `-s` option. We showed this in Section 8.13, when looking at the lack of flow control with UDP.
- It displays the routing table with the `-r` option and the interface information with the `-i` option. We showed this in Section 1.9, where we used `netstat` to discover the topology of our network.

There are other uses of `netstat` and most vendors have added their own features. Check the manual page on your system.

C.7 `lsof` Program

The name `lsof` stands for "list open files." Like `tcpdump`, it is a publicly available tool that is handy for debugging and has been ported to many versions of Unix.

One common use for `lsof` with networking is to find which process has a socket open on a specified IP address or port. `netstat` tells us which IP addresses and ports are in use, and the state of the TCP connections but it does not identify the process. For example, to find which process provides the daytime server, we execute

```
solaris % lsof -i TCP:daytime
COMMAND  PID  USER  FD   TYPE      DEVICE  SIZE/OFF   INODE NAME
inetd    222  root  15u  inet  0xf5a801f8       0t0     TCP *:daytime
```

This tells us the command (this service is provided by the `inetd` server), its process ID, the owner, descriptor (15 and the `u` means it is open for read–write), the type of socket, the address of its protocol control block, the size or offset of the file (not meaningful for a socket), the protocol type, and the name.

One common use for this program is when we start a server that binds its well-known port, and get the error that the address is already in use. We then use `lsof` to find the process that is using the port.

Since `lsof` reports on open files, it cannot report on network endpoints that are not associated with an open file: TCP endpoints in the TIME_WAIT state.

> `ftp://vic.cc.purdue.edu/pub/tools/unix/lsof` is the location for this program. It was written by Vic Abell.
>
> Some vendors supply their own utility that does similar things. For example, BSD/OS supplies the `fstat` program. The advantage in `lsof` is that it works under so many versions of Unix, and using a single tool in a heterogeneous environment, instead of a different tool for each environment, is a big advantage.

Appendix D

Miscellaneous Source Code

D.1 unp.h Header

Almost every program in the text includes our `unp.h` header, shown in Figure D.1. This header includes all the standard system headers that most network programs need, along with some general system headers. It also defines constants such as `MAXLINE` and ANSI C function prototypes for the functions that we define in the text (e.g., `readline`) and all the wrapper functions that we use. We do not show these prototypes.

lib/unp.h

```
 1 /* Our own header.  Tabs are set for 4 spaces, not 8 */

 2 #ifndef __unp_h
 3 #define __unp_h

 4 #include    "../config.h"       /* configuration options for current OS */
 5                                 /* "../config.h" is generated by configure */

 6 /* If anything changes in the following list of #includes, must change
 7    acsite.m4 also, for configure's tests. */

 8 #include    <sys/types.h>       /* basic system data types */
 9 #include    <sys/socket.h>      /* basic socket definitions */
10 #include    <sys/time.h>        /* timeval{} for select() */
11 #include    <time.h>            /* timespec{} for pselect() */
12 #include    <netinet/in.h>      /* sockaddr_in{} and other Internet defns */
13 #include    <arpa/inet.h>       /* inet(3) functions */
14 #include    <errno.h>
15 #include    <fcntl.h>           /* for nonblocking */
16 #include    <netdb.h>
17 #include    <signal.h>
```

```
18 #include    <stdio.h>
19 #include    <stdlib.h>
20 #include    <string.h>
21 #include    <sys/stat.h>        /* for S_xxx file mode constants */
22 #include    <sys/uio.h>         /* for iovec{} and readv/writev */
23 #include    <unistd.h>
24 #include    <sys/wait.h>
25 #include    <sys/un.h>          /* for Unix domain sockets */

26 #ifdef  HAVE_SYS_SELECT_H
27 #include    <sys/select.h>      /* for convenience */
28 #endif

29 #ifdef  HAVE_POLL_H
30 #include    <poll.h>            /* for convenience */
31 #endif

32 #ifdef  HAVE_STRINGS_H
33 #include    <strings.h>         /* for convenience */
34 #endif

35 /* Three headers are normally needed for socket/file ioctl's:
36  * <sys/ioctl.h>, <sys/filio.h>, and <sys/sockio.h>.
37  */
38 #ifdef  HAVE_SYS_IOCTL_H
39 #include    <sys/ioctl.h>
40 #endif
41 #ifdef  HAVE_SYS_FILIO_H
42 #include    <sys/filio.h>
43 #endif
44 #ifdef  HAVE_SYS_SOCKIO_H
45 #include    <sys/sockio.h>
46 #endif

47 #ifdef  HAVE_PTHREAD_H
48 #include    <pthread.h>
49 #endif

50 /* OSF/1 actually disables recv() and send() in <sys/socket.h> */
51 #ifdef  __osf__
52 #undef  recv
53 #undef  send
54 #define recv(a,b,c,d)   recvfrom(a,b,c,d,0,0)
55 #define send(a,b,c,d)   sendto(a,b,c,d,0,0)
56 #endif

57 #ifndef INADDR_NONE
58 #define INADDR_NONE 0xffffffff  /* should have been in <netinet/in.h> */
59 #endif

60 #ifndef SHUT_RD                 /* these three Posix.1g names are quite new */
61 #define SHUT_RD     0           /* shutdown for reading */
62 #define SHUT_WR     1           /* shutdown for writing */
63 #define SHUT_RDWR   2           /* shutdown for reading and writing */
64 #endif
65 #ifndef INET_ADDRSTRLEN
66 #define INET_ADDRSTRLEN     16  /* "ddd.ddd.ddd.ddd\0"
```

```
67                                      1234567890123456 */
68 #endif

69 /* Define following even if IPv6 not supported, so we can always allocate
70    an adequately-sized buffer, without #ifdefs in the code. */
71 #ifndef INET6_ADDRSTRLEN
72 #define INET6_ADDRSTRLEN    46  /* max size of IPv6 address string:
73                     "xxxx:xxxx:xxxx:xxxx:xxxx:xxxx:xxxx:xxxx" or
74                     "xxxx:xxxx:xxxx:xxxx:xxxx:xxxx:ddd.ddd.ddd.ddd\0"
75                      1234567890123456789012345678901234567890123456 */
76 #endif

77 /* Define bzero() as a macro if it's not in standard C library. */
78 #ifndef HAVE_BZERO
79 #define bzero(ptr,n)        memset(ptr, 0, n)
80 #endif

81 /* Older resolvers do not have gethostbyname2() */
82 #ifndef HAVE_GETHOSTBYNAME2
83 #define gethostbyname2(host,family)     gethostbyname((host))
84 #endif

85 /* The structure returned by recvfrom_flags() */
86 struct in_pktinfo {
87     struct in_addr ipi_addr;     /* dst IPv4 address */
88     int     ipi_ifindex;         /* received interface index */
89 };

90 /* We need the newer CMSG_LEN() and CMSG_SPACE() macros, but few
91    implementations support them today.  These two macros really need
92    an ALIGN() macro, but each implementation does this differently. */
93 #ifndef CMSG_LEN
94 #define CMSG_LEN(size)      (sizeof(struct cmsghdr) + (size))
95 #endif
96 #ifndef CMSG_SPACE
97 #define CMSG_SPACE(size)    (sizeof(struct cmsghdr) + (size))
98 #endif

99 /* Posix.1g requires the SUN_LEN() macro but not all implementations define
100    it (yet).  Note that this 4.4BSD macro works regardless whether there is
101    a length field or not. */
102 #ifndef SUN_LEN
103 #define SUN_LEN(su) \
104     (sizeof(*(su)) - sizeof((su)->sun_path) + strlen((su)->sun_path))
105 #endif

106 /* Posix.1g renames "Unix domain" as "local IPC".
107    But not all systems define AF_LOCAL and PF_LOCAL (yet). */
108 #ifndef AF_LOCAL
109 #define AF_LOCAL    AF_UNIX
110 #endif
111 #ifndef PF_LOCAL
112 #define PF_LOCAL    PF_UNIX
113 #endif

114 /* Posix.1g requires that an #include of <poll.h> define INFTIM, but many
115    systems still define it in <sys/stropts.h>.  We don't want to include
```

```
116     all the streams stuff if it's not needed, so we just define INFTIM here.
117     This is the standard value, but there's no guarantee it is -1. */
118 #ifndef INFTIM
119 #define INFTIM          (-1)    /* infinite poll timeout */
120 #ifdef  HAVE_POLL_H
121 #define INFTIM_UNPH             /* tell unpxti.h we defined it */
122 #endif
123 #endif

124 /* Following could be derived from SOMAXCONN in <sys/socket.h>, but many
125    kernels still #define it as 5, while actually supporting many more */
126 #define LISTENQ     1024        /* 2nd argument to listen() */

127 /* Miscellaneous constants */
128 #define MAXLINE     4096        /* max text line length */
129 #define MAXSOCKADDR  128        /* max socket address structure size */
130 #define BUFFSIZE    8192        /* buffer size for reads and writes */

131 /* Define some port number that can be used for client-servers */
132 #define SERV_PORT        9877   /* TCP and UDP client-servers */
133 #define SERV_PORT_STR   "9877"  /* TCP and UDP client-servers */
134 #define UNIXSTR_PATH    "/tmp/unix.str"     /* Unix domain stream cli-serv */
135 #define UNIXDG_PATH     "/tmp/unix.dg"  /* Unix domain datagram cli-serv */

136 /* Following shortens all the type casts of pointer arguments */
137 #define SA  struct sockaddr

138 #define FILE_MODE   (S_IRUSR | S_IWUSR | S_IRGRP | S_IROTH)
139                     /* default file access permissions for new files */
140 #define DIR_MODE    (FILE_MODE | S_IXUSR | S_IXGRP | S_IXOTH)
141                     /* default permissions for new directories */

142 typedef void Sigfunc (int);     /* for signal handlers */

143 #define min(a,b)    ((a) < (b) ? (a) : (b))
144 #define max(a,b)    ((a) > (b) ? (a) : (b))

145 #ifndef HAVE_ADDRINFO_STRUCT
146 #include    "../lib/addrinfo.h"
147 #endif

148 #ifndef HAVE_IF_NAMEINDEX_STRUCT
149 struct if_nameindex {
150     unsigned int if_index;      /* 1, 2, ... */
151     char   *if_name;            /* null terminated name: "le0", ... */
152 };
153 #endif

154 #ifndef HAVE_TIMESPEC_STRUCT
155 struct timespec {
156     time_t  tv_sec;             /* seconds */
157     long    tv_nsec;            /* and nanoseconds */
158 };
159 #endif
```

lib/unp.h

Figure D.1 Our header `unp.h`.

D.2 config.h Header

The GNU autoconf tool was used to aid in the portability of all the source code in this text. It is available from ftp://prep.ai.mit.edu/pub/gnu/. This tool generates a shell script named configure that you must run after downloading the software onto your system. This script determines the features provided by your Unix system: do socket address structures have a length field? is multicasting supported? are datalink socket address structures supported? and so on, generating a header named config.h. This header is the first header included by our unp.h header in the previous section. Figure D.2 shows the config.h header for BSD/OS 3.0.

The lines beginning with #define in column 1 are for features that the system provides. The lines that are commented out and contain #undef are features that the system does not provide.

i386-pc-bsdi3.0/config.h

```
 1 /* config.h.  Generated automatically by configure.  */
 2 /* Define the following if you have the corresponding header */
 3 #define CPU_VENDOR_OS "i386-pc-bsdi3.0"
 4 /* #undef   HAVE_NETCONFIG_H */      /* <netconfig.h> */
 5 /* #undef   HAVE_NETDIR_H */         /* <netdir.h> */
 6 #define HAVE_PTHREAD_H 1             /* <pthread.h> */
 7 #define HAVE_STRINGS_H 1             /* <strings.h> */
 8 /* #undef   HAVE_XTI_INET_H */       /* <xti_inet.h> */
 9 #define HAVE_SYS_FILIO_H 1           /* <sys/filio.h> */
10 #define HAVE_SYS_IOCTL_H 1           /* <sys/ioctl.h> */
11 #define HAVE_SYS_SELECT_H 1          /* <sys/select.h> */
12 #define HAVE_SYS_SOCKIO_H 1          /* <sys/sockio.h> */
13 #define HAVE_SYS_SYSCTL_H 1          /* <sys/sysctl.h> */
14 #define HAVE_SYS_TIME_H 1            /* <sys/time.h> */

15 /* Define if we can include <time.h> with <sys/time.h> */
16 #define TIME_WITH_SYS_TIME 1

17 /* Define the following if the function is provided */
18 #define HAVE_BZERO 1
19 #define HAVE_GETHOSTBYNAME2 1
20 /* #undef   HAVE_PSELECT */
21 #define HAVE_VSNPRINTF 1

22 /* Define the following if the function prototype is in a header */
23 /* #undef   HAVE_GETADDRINFO_PROTO */       /* <netdb.h> */
24 /* #undef   HAVE_GETNAMEINFO_PROTO */       /* <netdb.h> */
25 #define HAVE_GETHOSTNAME_PROTO 1            /* <unistd.h> */
26 #define HAVE_GETRUSAGE_PROTO 1              /* <sys/resource.h> */
27 #define HAVE_HSTRERROR_PROTO 1              /* <netdb.h> */
28 /* #undef   HAVE_IF_NAMETOINDEX_PROTO */    /* <net/if.h> */
29 #define HAVE_INET_ATON_PROTO 1              /* <arpa/inet.h> */
30 #define HAVE_INET_PTON_PROTO 1              /* <arpa/inet.h> */
31 /* #undef   HAVE_ISFDTYPE_PROTO */          /* <sys/stat.h> */
32 /* #undef   HAVE_PSELECT_PROTO */           /* <sys/select.h> */
33 #define HAVE_SNPRINTF_PROTO 1               /* <stdio.h> */
34 /* #undef   HAVE_SOCKATMARK_PROTO */        /* <sys/socket.h> */

35 /* Define the following if the structure is defined. */
```

```
36 /* #undef   HAVE_ADDRINFO_STRUCT */          /* <netdb.h> */
37 /* #undef   HAVE_IF_NAMEINDEX_STRUCT */      /* <net/if.h> */
38 #define HAVE_SOCKADDR_DL_STRUCT 1            /* <net/if_dl.h> */
39 #define HAVE_TIMESPEC_STRUCT 1               /* <time.h> */

40 /* Define the following if feature is provided. */
41 #define HAVE_SOCKADDR_SA_LEN 1      /* sockaddr{} has sa_len member */
42 #define HAVE_MSGHDR_MSG_CONTROL 1   /* msghdr{} has msg_control member */

43 /* Names of XTI devices for TCP and UDP */
44 /* #undef   HAVE_DEV_TCP */                  /* most XTI have devices here */
45 /* #undef   HAVE_DEV_XTI_TCP */              /* AIX has them here */
46 /* #undef   HAVE_DEV_STREAMS_XTISO_TCP */    /* OSF 3.2 has them here */

47 /* Define the following to the appropriate datatype, if necessary */
48 /* #undef   int8_t */                    /* <sys/types.h> */
49 /* #undef   int16_t */                   /* <sys/types.h> */
50 /* #undef   int32_t */                   /* <sys/types.h> */
51 #define uint8_t unsigned char           /* <sys/types.h> */
52 #define uint16_t unsigned short         /* <sys/types.h> */
53 #define uint32_t unsigned int           /* <sys/types.h> */
54 /* #undef   size_t */                    /* <sys/types.h> */
55 /* #undef   ssize_t */                   /* <sys/types.h> */
56 /* socklen_t should be typedef'd as uint32_t, but configure defines it
57    to be an unsigned int, as it is needed early in the compile process,
58    sometimes before some implementations define uint32_t. */
59 #define socklen_t unsigned int          /* <sys/socket.h> */
60 #define sa_family_t SA_FAMILY_T         /* <sys/socket.h> */
61 #define SA_FAMILY_T uint8_t

62 #define t_scalar_t int32_t              /* <xti.h> */
63 #define t_uscalar_t uint32_t            /* <xti.h> */

64 /* Define the following, if system suports the feature */
65 #define IPV4 1              /* IPv4, uppercase V name */
66 #define IPv4 1              /* IPv4, lowercase v name, just in case */
67 /* #undef   IPV6 */         /* IPv6, uppercase V name */
68 /* #undef   IPv6 */         /* IPv6, lowercase v name, just in case */
69 #define UNIXDOMAIN 1        /* Unix domain sockets */
70 #define UNIXdomain 1        /* Unix domain sockets */
71 #define MCAST 1             /* multicasting support */
```

i386-pc-bsdi3.0/config.h

Figure D.2 Our `config.h` header for BSD/OS 3.0.

D.3 `unpxti.h` Header

All our XTI programs include a header named `unpxti.h`, which is shown in Figure D.3. As with the listing of our `unp.h` header in Section D.1, we omit all the function prototypes. We also omit the definitions of the `T_` names between `T_INET_TCP` and `T_IP_BROADCAST`, as they are nearly identical.

libxti/unpxti.h

```
 1 #ifndef __unp_xti_h
 2 #define __unp_xti_h

 3 #include    "unp.h"

 4 #include    <xti.h>
 5 #ifdef  HAVE_XTI_INET_H
 6 #include    <xti_inet.h>
 7 #endif
 8 #ifdef  HAVE_NETCONFIG_H
 9 #include    <netconfig.h>
10 #endif
11 #ifdef  HAVE_NETDIR_H
12 #include    <netdir.h>
13 #endif

14 #ifdef  INFTIM_UNPH
15 #undef  INFTIM                  /* was not in <poll.h>, undef for <stropts.h> */
16 #endif

17 #include    <stropts.h>

18 /* Provide compatibility with the new names prepended with T_
19    in XNS Issue 5, which are not in Posix.1g. */

20 #ifndef T_INET_TCP
21 #define T_INET_TCP      INET_TCP
22 #endif
   ...
56 #ifndef T_IP_BROADCAST
57 #define T_IP_BROADCAST  IP_BROADCAST
58 #endif

59 /* Define the appropriate devices for t_open(). */
60 #ifdef  HAVE_DEV_TCP
61 #define XTI_TCP     "/dev/tcp"
62 #define XTI_UDP     "/dev/udp"
63 #endif
64 #ifdef  HAVE_DEV_XTI_TCP
65 #define XTI_TCP     "/dev/xti/tcp"
66 #define XTI_UDP     "/dev/xti/udp"
67 #endif
68 #ifdef  HAVE_DEV_STREAMS_XTISO_TCP
69 #define XTI_TCP     "/dev/streams/xtiso/tcp+"   /* + for XPG4 */
70 #define XTI_UDP     "/dev/streams/xtiso/udp+"   /* + for XPG4 */
71 #endif

72         /* device to t_open() for t_accept(); set by tcp_listen() */
73 extern char xti_serv_dev[];
```

libxti/unpxti.h

Figure D.3 Our header `unpxti.h` for XTI programs.

D.4 Standard Error Functions

We define our own set of error functions that are used throughout the text to handle error conditions. The reason for our own error functions is to let us write our error handling with a single line of C code, as in

```
if (error condition)
    err_sys(printf format with any number of arguments);
```

instead of

```
if (error condition) {
    char  buff[200];
    snprintf(buff, sizeof(buff), printf format with any number of arguments);
    perror(buff);
    exit(1);
}
```

Our error functions use the variable-length argument list facility from ANSI C. See Section 7.3 of [Kernighan and Ritchie 1988] for additional details.

Figure D.4 lists the differences between the various error functions. If the global integer `daemon_proc` is nonzero, the message is passed to `syslog` with the indicated level; otherwise the error is output to standard error.

Function	strerror (errno) ?	Terminate ?	syslog level
err_dump	yes	abort();	LOG_ERR
err_msg	no	return;	LOG_INFO
err_quit	no	exit(1);	LOG_ERR
err_ret	yes	return;	LOG_INFO
err_sys	yes	exit(1);	LOG_ERR
err_xti	yes	exit(1);	LOG_ERR
err_xti_ret	yes	return;	LOG_INFO

Figure D.4 Summary of our standard error functions.

Figure D.5 shows the first five functions from Figure D.4.

lib/error.c

```
 1 #include    "unp.h"

 2 #include    <stdarg.h>          /* ANSI C header file */
 3 #include    <syslog.h>          /* for syslog() */

 4 int     daemon_proc;            /* set nonzero by daemon_init() */

 5 static void err_doit(int, int, const char *, va_list);

 6 /* Nonfatal error related to a system call.
 7  * Print a message and return. */

 8 void
 9 err_ret(const char *fmt,...)
10 {
```

```
11      va_list ap;

12      va_start(ap, fmt);
13      err_doit(1, LOG_INFO, fmt, ap);
14      va_end(ap);
15      return;
16  }

17  /* Fatal error related to a system call.
18   * Print a message and terminate. */

19  void
20  err_sys(const char *fmt,...)
21  {
22      va_list ap;

23      va_start(ap, fmt);
24      err_doit(1, LOG_ERR, fmt, ap);
25      va_end(ap);
26      exit(1);
27  }

28  /* Fatal error related to a system call.
29   * Print a message, dump core, and terminate. */

30  void
31  err_dump(const char *fmt,...)
32  {
33      va_list ap;

34      va_start(ap, fmt);
35      err_doit(1, LOG_ERR, fmt, ap);
36      va_end(ap);
37      abort();                    /* dump core and terminate */
38      exit(1);                    /* shouldn't get here */
39  }

40  /* Nonfatal error unrelated to a system call.
41   * Print a message and return. */

42  void
43  err_msg(const char *fmt,...)
44  {
45      va_list ap;

46      va_start(ap, fmt);
47      err_doit(0, LOG_INFO, fmt, ap);
48      va_end(ap);
49      return;
50  }

51  /* Fatal error unrelated to a system call.
52   * Print a message and terminate. */

53  void
54  err_quit(const char *fmt,...)
55  {
56      va_list ap;
```

```
57      va_start(ap, fmt);
58      err_doit(0, LOG_ERR, fmt, ap);
59      va_end(ap);
60      exit(1);
61  }

62  /* Print a message and return to caller.
63   * Caller specifies "errnoflag" and "level". */

64  static void
65  err_doit(int errnoflag, int level, const char *fmt, va_list ap)
66  {
67      int     errno_save, n;
68      char    buf[MAXLINE + 1];

69      errno_save = errno;         /* value caller might want printed */
70  #ifdef  HAVE_VSNPRINTF
71      vsnprintf(buf, MAXLINE, fmt, ap);   /* this is safe */
72  #else
73      vsprintf(buf, fmt, ap);     /* this is not safe */
74  #endif
75      n = strlen(buf);
76      if (errnoflag)
77          snprintf(buf + n, MAXLINE - n, ": %s", strerror(errno_save));
78      strcat(buf, "\n");

79      if (daemon_proc) {
80          syslog(level, buf);
81      } else {
82          fflush(stdout);         /* in case stdout and stderr are the same */
83          fputs(buf, stderr);
84          fflush(stderr);
85      }
86      return;
87  }
```

lib/error.c

Figure D.5 Our standard error functions.

Appendix E

Solutions to Selected Exercises

Chapter 1

1.3 Under AIX we get

```
aix % daytimetcpcli 206.62.226.33
socket error: Addr family not supported by protocol
```

To find more information on this error we first use `grep` to search for the string `Addr` in the `<sys/errno.h>` header.

```
aix % grep Addr /usr/include/sys/errno.h
#define EAFNOSUPPORT 66 /* Address family not supported by protocol family */
#define EADDRINUSE   67 /* Address already in use */
```

The first is the `errno` returned by `socket`. We then look at the manual page:

```
aix % man socket
```

Most manual pages give additional, albeit terse, information toward the end under a heading of the form "Errors."

1.4 We change the first declaration to be

```
int   sockfd, n, counter = 0;
```

We add the statement

```
counter++;
```

as the first statement of the `while` loop. Finally we execute

```
printf("counter = %d\n", counter);
```

before terminating. The value printed is always 1.

1.5 We declare an `int` named `i` and change the call to `write` to be

```
for (i = 0; i < strlen(buff); i++)
    Write(connfd, &buff[i], 1);
```

The results vary, depending on the client host and the server host. If the client and server are on the same host, the counter is normally 1, which means even though the server does 26 `write`s, the data is returned by a single `read`. But if the client runs under Solaris 2.5.1 and the server runs under BSD/OS 3.0, the counter is normally 2. If we watch the packets on the Ethernet, we see the first character is sent in a packet by itself, but the next packet contains the remaining 25 characters. (Our discussion of the Nagle algorithm in Section 7.9 explains the reason for this behavior.) If the client runs under BSD/OS 3.0 and the server runs under Solaris 2.5.1, the counter is now 26. If we watch the packets we see each character sent in its own packet.

The purpose of this example is to reiterate that different TCPs do different things with the data and our application must be prepared to read the data as a stream of bytes, until the end of the data stream is encountered.

Chapter 2

2.1 All RFCs are available at no charge through electronic mail, anonymous FTP, or the World Wide Web. A starting point is `http://www.ietf.org`. The directory `ftp://ftp.isi.edu/in-notes` is one location for RFCs. The starting point is to fetch the current RFC index, normally the file `rfc-index.txt`. Look at the entry for RFC 1340, with a title of "Assigned Numbers," and note that it has been made obsolete by RFC 1700. Although RFC 1700 is current at the time of writing, it will probably be obsolete by the time you read this. Go forward from these obsolete RFCs and find the current (i.e., highest numbered) of the "Assigned Numbers" RFC.

The section of this RFC titled "Version Numbers" identifies the various IP version numbers. Version 0 is reserved, version 1–3 are unassigned, and version 5 is the Internet Stream protocol.

2.2 If we search the RFC index (see the solution to the previous exercise) with an editor of some form, looking for the term "Stream," we find that RFC 1819 defines Version 2 of the Internet Stream Protocol. Whenever looking for information that might be covered by an RFC, the RFC index should be searched.

2.3 With IPv4 this generates a 576-byte IP datagram (20 bytes for the IPv4 header and 20 bytes for the TCP header), the minimum reassembly buffer size with IPv4.

2.4 In this example the server performs the active close, not the client.

2.5 The host on the token ring cannot send packets with more than 1460 bytes of data, because the MSS it received was 1460. The host on the Ethernet can send packets with up to 4096 bytes of data, but it will not exceed the MTU of the outgoing interface (the Ethernet) to avoid fragmentation. TCP cannot exceed the MSS announced by the other end, but it can always send less than this amount.

2.6 The "Protocol Numbers" section of the Assigned Numbers RFC shows a value of 89 for OSPF.

Chapter 3

3.1 In C a function cannot change the value of an argument that is passed by value. For a called function to modify a value passed by the caller requires that the caller pass a pointer to the value to be modified.

3.2 The pointer must be incremented by the number of bytes read or written, but C does not allow a `void` pointer to be incremented (since the compiler does not know the datatype pointed to).

Chapter 4

4.1 Look at the definitions for the constants beginning with `INADDR_` other than `INADDR_ANY` (which is all zero bits) and `INADDR_NONE` (which is all one bits). For example, the class D multicast address `INADDR_MAX_LOCAL_GROUP` is defined as `0xe00000ff` with the comment "224.0.0.255," which is clearly in host byte order.

4.2 Here are the new lines added after the call to `connect`:

```
len = sizeof(cliaddr);
Getsockname(sockfd, (SA *) &cliaddr, &len);
printf("local addr: %s\n",
       Sock_ntop((SA *) &cliaddr, len));
```

This requires a declaration of `len` as a `socklen_t` and a declaration of `cliaddr` as a `struct sockaddr_in`. Notice that the value–result argument for `getsockname` (`len`) must be initialized before the call to the size of the variable pointed to by the second argument. The most common programming error with value–result arguments is to forget this initialization.

4.3 When the child calls `close`, the reference count is decremented from 2 to 1, so a FIN is not sent to the client. Later, when the parent calls `close`, the reference count is decremented to 0 and the FIN is sent.

4.4 `accept` returns `EINVAL`, since the first argument is not a listening socket.

4.5 Without a call to `bind` the call to `listen` assigns an ephemeral port to the listening socket.

Chapter 5

5.1 The duration of the TIME_WAIT state should be between 1 and 4 minutes, giving an MSL between 30 seconds and 2 minutes.

5.2 Our client–server programs do not work with a binary file. Assume the first 3 bytes in the file are binary 1, binary 0, and a newline. The call to `fgets` in

Figure 5.5 reads up to `MAXLINE-1` characters, or until a newline is encountered, or up through the end-of-file. In this example it will read the first three characters and then terminate the string with a null byte. But our call to `strlen` in Figure 5.5 returns 1, since it stops at the first null byte. One byte is sent to the server, but the server blocks in its call to `readline`, waiting for a newline character. The client blocks waiting for the server's reply. This is called a *deadlock*: both processes are blocked waiting for something that will never arrive from the other one. The problem here is that `fgets` signifies the end of the data that it returns with a null byte, so the data that it reads cannot contain any null bytes.

5.3 Telnet converts the input lines into NVT ASCII (Section 26.4 of TCPv1), which means terminating every line with the 2-character sequence of a CR (carriage return) followed by an LF (linefeed). Our client adds only a newline, which is actually a linefeed character. Nevertheless, we can use the Telnet client to communicate with our server as our server echoes back every character, including the CR that precedes each newline.

5.4 No, the final two segments of the connection termination sequence are not sent. When the client sends the data to the server, after we kill the server child (the "another line"), the server TCP responds with an RST. The RST aborts the connection and also prevents the server end of the connection (the end that did the active close) from passing through the TIME_WAIT state.

5.5 Nothing changes because the server process that is started on the server host creates a listening socket and is waiting for new connection requests to arrive. What we send in step 3 is a data segment destined for an ESTABLISHED TCP connection. Our server with the listening socket never sees this data segment, and the server TCP still responds to it with an RST.

5.6 Figure E.1 shows the program. Running this program under AIX generates

```
aix % tsigpipe 206.62.226.34
SIGPIPE received
write error: Broken pipe
```

The initial `sleep` of 2 seconds is to let the daytime server send its reply and close its end of the connection. Our first `write` sends a data segment to the server, which responds with an RST (since the daytime server has completely closed its socket). Note that our TCP allows us to write to a socket that has received a FIN. The second `sleep` lets the server's RST be received, and our second `write` generates `SIGPIPE`. Since our signal handler returns, `write` returns an error of `EPIPE`.

5.7 Assuming the server host supports the *weak end system model* (which we describe in Section 8.8) everything works. That is, the server host will accept an incoming IP datagram (which contains a TCP segment in this case) arriving on the leftmost datalink even though the destination IP address is the address of the rightmost datalink. We can test this by running our server on our host `bsdi` (Figure 1.16) and then starting the client on our host `solaris` but specifying the other IP address of the server (206.62.226.66) to the client. After the connection is

tcpcliserv/tsigpipe.c

```
 1 #include    "unp.h"

 2 void
 3 sig_pipe(int signo)
 4 {
 5     printf("SIGPIPE received\n");
 6     return;
 7 }

 8 int
 9 main(int argc, char **argv)
10 {
11     int     sockfd;
12     struct sockaddr_in servaddr;

13     if (argc != 2)
14         err_quit("usage: tcpcli <IPaddress>");

15     sockfd = Socket(AF_INET, SOCK_STREAM, 0);

16     bzero(&servaddr, sizeof(servaddr));
17     servaddr.sin_family = AF_INET;
18     servaddr.sin_port = htons(13);  /* daytime server */
19     Inet_pton(AF_INET, argv[1], &servaddr.sin_addr);

20     Signal(SIGPIPE, sig_pipe);

21     Connect(sockfd, (SA *) &servaddr, sizeof(servaddr));

22     sleep(2);
23     Write(sockfd, "hello", 5);
24     sleep(2);
25     Write(sockfd, "world", 5);

26     exit(0);
27 }
```

tcpcliserv/tsigpipe.c

Figure E.1 Generate `SIGPIPE`.

established, if we run `netstat` on the server we see that the local IP address is the destination IP address from the client's SYN, not the IP address of the datalink on which the SYN arrived (as we mentioned in Section 4.4).

5.8 Our client was on a little-endian Intel system, where the 32-bit integer with a value of 1 is stored as shown in Figure E.2.

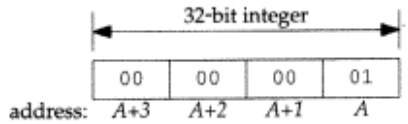


Figure E.2 Representation of the 32-bit integer 1 in little-endian format.

The 4 bytes are sent across the socket in the order *A*, *A+1*, *A+2*, and *A+3* where they are stored in the big-endian format, as shown in Figure E.3.

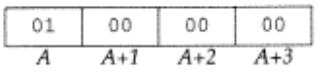


Figure E.3 Representation of the 32-bit integer from Figure E.2 in big-endian format.

This value of `0x01000000` is interpreted as 16,777,216. Similarly the integer 2 sent by the client will be interpreted at the server as `0x02000000`, or 33,554,432. The sum of these two integers is 50,331,648, or `0x03000000`. When this big-endian value on the server is sent to the client, it is interpreted on the client as the integer value 3.

But the 32-bit integer value of –22 is represented on the little-endian system as shown in Figure E.4, assuming a two's complement representation of negative numbers.

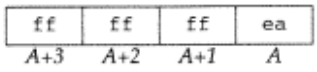


Figure E.4 Representation of the 32-bit integer –22 in little-endian format.

This is interpreted on the big-endian server as `0xeaffffff`, or –352,321,537. Similarly the little-endian representation of –77 is `0xffffffb3` but this is represented on the big-endian server as `0xb3ffffff`, or –1,275,068,417. The addition on the server yields a binary result of `0x9effffe`, or –1,627,389,954. This big-endian value is sent across the socket to the client where it is interpreted as the little-endian value `0xfefff9e`, or –16,777,314, which is the value printed in our example.

5.9 The technique is correct (converting the binary values to network byte order) but the two functions `htonl` and `ntohl` cannot be used. Even though the `l` in these functions once meant "long," these functions operate on 32-bit integers (Section 3.4). On a 64-bit system a `long` will probably occupy 64 bits, and these two functions will not work correctly. One might define two new functions, `hton64` and `ntoh64`, to solve this problem, but this will not work on systems that represent `long`s using 32 bits.

5.10 In the first scenario the server blocks forever in the call to `readn` in Figure 5.20 because the client sends two 32-bit values but the server is waiting for two 64-bit values. Swapping the client and server between the two hosts causes the client to send two 64-bit values, but the server reads only the first 64 bits, interpreting it as two 32-bit values. The second 64-bit value remains in the server's socket receive buffer. The server writes back one 32-bit value and the client will block forever in its call to `readn` in Figure 5.19, waiting to read one 64-bit value.

5.11 IP's routing function looks at the destination IP address (the server's IP address) and searches the routing table to determine the outgoing interface and the next hop (Chapter 9 of TCPv1). The primary IP address of the outgoing interface is used as the source IP address, assuming the socket has not already bound a local IP address.

Chapter 6

6.1 The array of integers is contained within a structure and C allows structures to be assigned across an equals sign.

6.2 If `select` tells us that the socket is writable, the socket send buffer has room for 8192 bytes, but we call `write` for this blocking socket with a buffer length of 8193 bytes, `write` can block, waiting for room for the final byte. Read operations on a blocking socket will always return a short count if some data is available but write operations on a blocking socket will block until all the data can be accepted by the kernel. Therefore when using `select` to test for writability, we must set the socket nonblocking to avoid blocking.

6.3 If both descriptors are readable, only the first test is performed, the test of the socket. But this does not break the client; it just makes it less efficient. That is, if `select` returns with both descriptors readable, the first `if` is true, causing a `readline` from the socket followed by an `fputs` to standard output. The next `if` is skipped (because of the `else` that we prepended) but `select` is then called again and immediately finds standard input readable and returns immediately. The key concept here is that what clears the condition of "standard input being readable" is not `select` returning, but reading from the descriptor.

6.4 Use the `getrlimit` function to fetch the values for the `RLIMIT_NOFILE` resource and then call `setrlimit` to set the current soft limit (`rlim_cur`) to the hard limit (`rlim_max`). For example, under Solaris 2.5 the soft limit is 64 but any process can increase this to the default hard limit of 1024.

`getrlimit` and `setrlimit` are not part of Posix.1 but are required by Unix 98.

6.5 The server application continually sends data to the client, which the client TCP acknowledges and throws away.

6.6 `shutdown` with `SHUT_WR` or `SHUT_RDWR` always sends a FIN while `close` sends a FIN only if the descriptor reference count is 1 when `close` is called.

6.7 `readline` returns an error, and our `Readline` wrapper function terminates the server. Server's must be more robust than this. Notice that we handle this condition in Figure 6.26, although even that code is inadequate. Consider what happens if connectivity is lost between the client and server and one of the server's responses eventually times out. The error returned could be `ETIMEDOUT`.

In general, a server should not abort for errors such as these. It should log the error, close the socket, and continue servicing other clients. Realize that handling an error of this type by aborting is unacceptable in a server such as this one, where one process is handling all clients. But if the server were a child handling just one client, then having that one child abort would not affect the parent (which we assume handles all new connections and spawns the children) or any of the other children that are servicing other clients.

Chapter 7

7.2 Figure E.5 shows one solution to this exercise. We have removed the printing of the data string returned by the server as that value is not needed.

sockopt/rcvbuf.c

```
 1 #include    "unp.h"
 2 #include    <netinet/tcp.h>     /* for TCP_MAXSEG */

 3 int
 4 main(int argc, char **argv)
 5 {
 6     int     sockfd, rcvbuf, mss;
 7     socklen_t len;
 8     struct sockaddr_in servaddr;

 9     if (argc != 2)
10         err_quit("usage: rcvbuf <IPaddress>");

11     sockfd = Socket(AF_INET, SOCK_STREAM, 0);

12     len = sizeof(rcvbuf);
13     Getsockopt(sockfd, SOL_SOCKET, SO_RCVBUF, &rcvbuf, &len);
14     len = sizeof(mss);
15     Getsockopt(sockfd, IPPROTO_TCP, TCP_MAXSEG, &mss, &len);
16     printf("defaults: SO_RCVBUF = %d, MSS = %d\n", rcvbuf, mss);

17     bzero(&servaddr, sizeof(servaddr));
18     servaddr.sin_family = AF_INET;
19     servaddr.sin_port = htons(13);  /* daytime server */
20     Inet_pton(AF_INET, argv[1], &servaddr.sin_addr);

21     Connect(sockfd, (SA *) &servaddr, sizeof(servaddr));

22     len = sizeof(rcvbuf);
23     Getsockopt(sockfd, SOL_SOCKET, SO_RCVBUF, &rcvbuf, &len);
24     len = sizeof(mss);
25     Getsockopt(sockfd, IPPROTO_TCP, TCP_MAXSEG, &mss, &len);
26     printf("after connect: SO_RCVBUF = %d, MSS = %d\n", rcvbuf, mss);

27     exit(0);
28 }
```

sockopt/rcvbuf.c

Figure E.5 Print socket receive buffer size and MSS before and after connection establishment.

First, there is no "correct" output from this program. The results vary from system to system. Some systems (notably Solaris 2.5.1 and earlier) always return 0 for the socket buffer sizes, preventing us from seeing what happens with this value across the connection.

With regard to the MSS, the value printed before `connect` is the implementation default (often 536 or 512) while the value printed after `connect` depends on a possible MSS option from the peer. On a local Ethernet, for example, the value after `connect` could be 1460. After a `connect` to a server on a remote network, however, the MSS may be similar to the default, unless your system supports path

MTU discovery. If possible, run a tool like `tcpdump` (Section C.5) while the program is running to see the actual MSS option on the SYN segment from the peer.

With regard to the socket receive buffer size, many implementations round this value up after the connection is established to a multiple of the MSS. Another way to see the socket receive buffer size after the connection establishment is to watch the packets using a tool like `tcpdump` and look at TCP's advertised window.

7.3 Allocate a `linger` structure named `ling` and initialize it as

```
str_cli(stdin, sockfd);

ling.l_onoff = 1;
ling.l_linger = 0;
Setsockopt(sockfd, SOL_SOCKET, SO_LINGER, &ling, sizeof(ling));

exit(0);
```

This should cause the client TCP to terminate the connection with an RST instead of the normal four segment exchange. The server child's call to `readline` returns an error of `ECONNRESET` and the message printed is

```
readline error: Connection reset by peer
```

The client socket should not go through the TIME_WAIT state, even though the client did the active close.

7.4 The first client calls `setsockopt`, `bind`, and `connect`. But between the first client's calls to `bind` and `connect` if the second client calls `bind`, `EADDRINUSE` is returned. But as soon as the first client connects to the peer, the second client's `bind` will work, since the first client's socket is then connected. The only way to handle this is for the second client to try calling `bind` multiple times if `EADDRINUSE` is returned and not give up the first time the error is returned. (This race condition was pointed out in the Posix.1g standard.)

7.5 We run the program on a host without multicast support (UnixWare 2.1.2).

```
unixware % sock -s 9999 &                          start first server with wildcard
[1]     29697
unixware % sock -s 206.62.226.37 9999              try second server but without -A
can't bind local address: Address already in use
unixware % sock -s -A 206.62.226.37 9999 &         try again with -A; works
[2]     29699
unixware % sock -s -A 127.0.0.1 9999 &             third server, with -A; works
[3]     29700
unixware % netstat -na | grep 9999
tcp        0      0  127.0.0.1.9999         *.*                    LISTEN
tcp        0      0  206.62.226.37.9999     *.*                    LISTEN
tcp        0      0  *.9999                 *.*                    LISTEN
```

7.6 We first try on a host without multicast support (UnixWare 2.1.2).

```
unixware % sock -s -u -A 206.62.226.37 8888 &      first one starts
[4]     29707
```

```
unixware % sock -s -u -A 206.62.226.37 8888
can't bind local address: Address already in use   cannot start another
```

We specify the SO_REUSEADDR option for both instances, but it does not work.

We now try this on a host that supports multicasting but does not support the SO_REUSEPORT option (Solaris 2.6).

```
solaris26 % sock -s -u 8888 &                    first one starts
[1]     1135
solaris26 % sock -s -u 8888
can't bind local address: Address already in use
solaris26 % sock -s -u -A 8888 &                 try second again with -A; works
solaris26 % netstat -na | grep 8888              and we can see the duplicate bindings
      *.8888                         Idle
      *.8888                         Idle
```

On this system we do not need to specify SO_REUSEADDR for the first bind, only for the second.

Finally we run this scenario under BSD/OS 3.0, which supports multicasting and the SO_REUSEPORT option. We first try SO_REUSEADDR for both servers, but this does not work.

```
bsdi % sock -u -s -A 7777 &
[1]     17610
bsdi % sock -u -s -A 7777
can't bind local address: Address already in use
```

Next we try SO_REUSEPORT but only for the second server, not for the first. This does not work since a completely duplicate binding requires the option for all sockets that share the binding.

```
bsdi % sock -u -s 8888 &
[1]     17612
bsdi % sock -u -s -T 8888
can't bind local address: Address already in use
```

Finally we specify SO_REUSEPORT for both servers, and this works.

```
bsdi % sock -u -s -T 9999 &
[1]     17614
bsdi % sock -u -s -T 9999 &
[2]     17615
bsdi % netstat -na | grep 9999
udp        0      0  *.9999                 *.*
udp        0      0  *.9999                 *.*
```

7.7 This does nothing because Ping uses an ICMP socket and the SO_DEBUG socket option affects only TCP sockets. The description for the SO_DEBUG socket option has always been something generic such as "this option enables debugging in the respective protocol layer" but the only protocol layer to implement the option has been TCP.

7.8 Figure E.6 shows the time line.

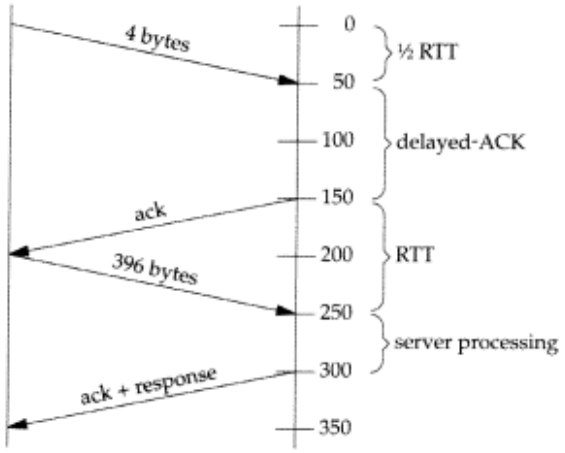


Figure E.6 Interaction of Nagle algorithm with delayed-ACK.

7.9 Setting the `TCP_NODELAY` socket option causes the data from the second `write` to be sent immediately, even though the connection has a small packet outstanding. We show this in Figure E.7. The total time in this example is just over 150 ms.

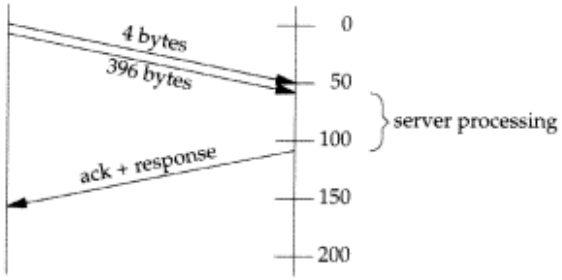


Figure E.7 Avoidance of Nagle algorithm by setting `TCP_NODELAY` socket option.

7.10 The advantage to this solution is reducing the number of packets, as we show in Figure E.8.

7.11 Section 4.2.3.2 states "the delay MUST be less than 0.5 seconds, and in a stream of full-sized segments there SHOULD be an ACK for at least every second segment." Berkeley-derived implementations delay an ACK by at most 200 ms (p. 821 of TCPv2).

7.12 The server parent in Figure 5.2 spends most of its time blocked in the call to `accept` and the child in Figure 5.3 spends most of its time blocked in the call to `read`, which is called by `readline`. The keepalive option has no effect on a listening socket so the parent is not affected should the client host crash. The child's

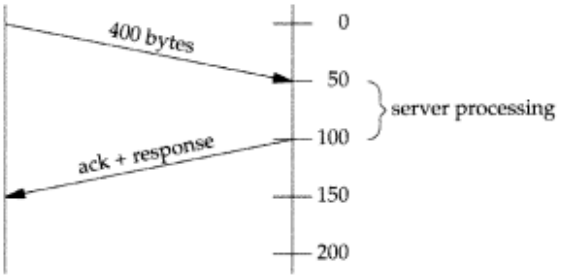


Figure E.8 Using `writev` instead of setting the `TCP_NODELAY` socket option.

`read` will return an error of `ETIMEDOUT`, sometime around 2 hours after the last data exchange across the connection.

7.13 The client in Figure 5.5 spends most of its time blocked in the call to `fgets`, which in turn is blocked in some type of read operation on standard input within the standard I/O library. When the keepalive timer expires around 2 hours after the last data exchange across the connection, and all the keepalive probes fail to elicit a response from the server, the socket's pending error is set to `ETIMEDOUT`. But the client is blocked in the call to `fgets` on standard input and will not see this error until it performs a read or write on the socket. This is one reason why we modified Figure 5.5 to use `select` in Chapter 6.

7.14 This client spends most of its time blocked in the call to `select`, which will return the socket as readable as soon as the pending error is set to `ETIMEDOUT` (as we described in the previous solution).

7.15 Only two segments, not four. There is a very low probability that the two systems will have timers that are exactly synchronized; hence one end's keepalive timer will expire shortly before the other's. The first one to expire sends the keepalive probe, causing the other end to ACK this probe. But the receipt of the keepalive probe causes the keepalive timer on the host with the (slightly) slower clock to be reset for 2 hours in the future.

7.16 The original sockets API did not have a `listen` function. Instead, the fourth argument to `socket` contained socket options, and `SO_ACCEPTCON` was used to specify a listening socket. When `listen` was added, the flag stayed around, but is now set only by the kernel (p. 456 of TCPv2).

Chapter 8

8.1 Yes. The `read` returns 4096 bytes of data but the `recvfrom` returns 2048 (the first of the two datagrams). A `recvfrom` on a datagram socket never returns more than one datagram, regardless of how much the application asks for.

8.2 If the protocol uses variable-length socket address structures, `clilen` could be too large. We will see in Chapter 14 that this is OK with Unix domain socket

address structures, but the correct way to code the function is to use the actual length returned by `recvfrom` as the length for `sendto`.

8.4 Running `ping` like this is an easy way to see ICMP messages that are received by the host on which `ping` is being run. We reduce the number of packets sent from the normal one per second just to reduce the output. If we run our UDP client on our host `solaris`, specifying the server's IP address as 206.62.226.42, and also run the `ping` program, we get the following output:

```
solaris % ping -v -I 60 127.0.0.1
PING 127.0.0.1: 56 data bytes
64 bytes from localhost (127.0.0.1): icmp_seq=0. time=2. ms
ICMP Port Unreachable from gateway alpha.kohala.com (206.62.226.42)
 for udp from solaris.kohala.com (206.62.226.33)
           to alpha.kohala.com (206.62.226.42) port 9877
```

8.5 It probably has a socket receive buffer size, but data is never accepted for a listening TCP socket. Most implementations do not preallocate memory for socket send buffers or socket receive buffers. The socket buffer sizes specified with the `SO_SNDBUF` and `SO_RCVBUF` socket options are just upper limits for that socket.

8.6 We run the `sock` program on the multihomed host `bsdi` specifying the `-u` option (use UDP) and the `-l` option (specifying the local IP address and port).

```
bsdi % sock -u -l 206.62.226.66.4444 206.62.226.42 8888
hello
recv error: Connection refused
```

The local IP address is the lower Ethernet in Figure 1.16, but the datagram must go out the upper Ethernet to get to the destination. The "Connection refused" error that is returned is because the `sock` program calls `connect` and the server host returns an ICMP port unreachable. Watching the network with `tcpdump` shows that the source IP address is the one that was bound by the client, not the outgoing interface address.

```
14:39:46.211130 206.62.226.66.4444 > 206.62.226.42.8888: udp 6
14:39:46.211656 206.62.226.42 > 206.62.226.66: icmp: 206.62.226.42
                                                udp port 8888 unreachable
```

8.7 Putting a `printf` in the client should introduce a delay between each datagram, allowing the server to receive more datagrams. Putting a `printf` in the server should cause the server to lose more datagrams.

8.8 The largest IPv4 datagram is 65535 bytes, limited by the 16-bit total length field in Figure A.1. The IP header requires 20 bytes and the UDP header requires 8 bytes, leaving a maximum of 65507 bytes for user data. With IPv6, without jumbogram support, the size of the IPv6 header is 40 bytes, leaving a maximum of 65487 bytes for user data.

Figure E.9 shows the new version of `dg_cli`. If you forget to set the send buffer size, Berkeley-derived kernels return an error of `EMSGSIZE` from `sendto`, since the size of the socket send buffer is normally less than required for a maximum sized UDP datagram (be sure to do Exercise 7.1). But if we set the client's socket

udpcliserv/dgclibig.c

```
 1 #include    "unp.h"
 2 #undef  MAXLINE
 3 #define MAXLINE 65507
 4 void
 5 dg_cli(FILE *fp, int sockfd, const SA *pservaddr, socklen_t servlen)
 6 {
 7     int     size;
 8     char    sendline[MAXLINE], recvline[MAXLINE + 1];
 9     ssize_t n;
10     size = 70000;
11     Setsockopt(sockfd, SOL_SOCKET, SO_SNDBUF, &size, sizeof(size));
12     Setsockopt(sockfd, SOL_SOCKET, SO_RCVBUF, &size, sizeof(size));
13     Sendto(sockfd, sendline, MAXLINE, 0, pservaddr, servlen);
14     n = Recvfrom(sockfd, recvline, MAXLINE, 0, NULL, NULL);
15     printf("received %d bytes\n", n);
16 }
```

udpcliserv/dgclibig.c

Figure E.9 Writing the maximum size UDP/IPv4 datagram.

buffer sizes as shown in Figure E.9 and run the client program, nothing is returned by the server. We can verify that the client's datagram is sent to the server by running `tcpdump`, but if we put a `printf` in the server, its call to `recvfrom` does not return the datagram. The problem is that the server's UDP socket receive buffer is smaller than the datagram that we are sending, so the datagram is discarded and not delivered to the socket. On a BSD/OS system we can verify this by running `netstat -s` and looking at the "dropped due to full socket buffers" counter, before and after our big datagram is received. The final solution is to modify the server, setting its socket send and receive buffer sizes.

On most networks a 65535-byte IP datagram is fragmented. Recall from Section 2.9 that an IP layer must support a reassembly buffer size of only 576 bytes. Therefore you may encounter hosts that will not receive the maximum sized datagrams sent in this exercise. Also, many Berkeley-derived implementations, including 4.4BSD-Lite2, have a sign bug that prevents UDP from accepting a datagram larger than 32767 bytes (line 95 of p. 770 of TCPv2).

Chapter 9

9.1 Figure E.10 shows our program that calls `gethostbyaddr`.

names/hostent2.c

```
 1 #include    "unp.h"
 2 int
 3 main(int argc, char **argv)
 4 {
 5     char   *ptr, **pptr;
```

```
 6      char    str[INET6_ADDRSTRLEN];
 7      struct hostent *hptr;

 8      while (--argc > 0) {
 9          ptr = *++argv;
10          if ( (hptr = gethostbyname(ptr)) == NULL) {
11              err_msg("gethostbyname error for host: %s: %s",
12                      ptr, hstrerror(h_errno));
13              continue;
14          }
15          printf("official hostname: %s\n", hptr->h_name);

16          for (pptr = hptr->h_aliases; *pptr != NULL; pptr++)
17              printf("    alias: %s\n", *pptr);

18          switch (hptr->h_addrtype) {
19          case AF_INET:
20  #ifdef  AF_INET6
21          case AF_INET6:
22  #endif
23              pptr = hptr->h_addr_list;
24              for ( ; *pptr != NULL; pptr++) {
25                  printf("\taddress: %s\n",
26                       Inet_ntop(hptr->h_addrtype, *pptr, str, sizeof(str)));

27                  if ( (hptr = gethostbyaddr(*pptr, hptr->h_length,
28                                             hptr->h_addrtype)) == NULL)
29                      printf("\t(gethostbyaddr failed)\n");
30                  else if (hptr->h_name != NULL)
31                      printf("\tname = %s\n", hptr->h_name);
32                  else
33                      printf("\t(no hostname returned by gethostbyaddr)\n");
34              }
35              break;

36          default:
37              err_ret("unknown address type");
38              break;
39          }
40      }
41      exit(0);
42  }
```

names/hostent2.c

Figure E.10 Modification to Figure 9.4 to call `gethostbyaddr`.

This program works fine for a host with a single IP address. If we run the program in Figure 9.4 for a host with four IP addresses we get

```
solaris % hostent gemini.tuc.noao.edu
official hostname: gemini.tuc.noao.edu
        address: 140.252.8.54
        address: 140.252.4.54
        address: 140.252.3.54
        address: 140.252.1.11
```

But if we run the program in Figure E.10 for the same host only the first IP address is output:

```
solaris % hostent2 gemini.tuc.noao.edu
official hostname: gemini.tuc.noao.edu
        address: 140.252.8.54
        name = gemini.tuc.noao.edu
```

The problem is that the two functions `gethostbyname` and `gethostbyaddr` share the same `hostent` structure, as we show at the beginning of Section 11.14. When our new program calls `gethostbyaddr`, it reuses this structure along with the storage that the structure points to (i.e., the `h_addr_list` array of pointers), wiping out the remaining three IP addresses returned by `gethostbyname`.

9.2 If your system does not supply the reentrant version of `gethostbyaddr` (which we describe in Section 11.15), then you must make a copy of the array of pointers returned by `gethostbyname`, along with the data pointed to by this array, before calling `gethostbyaddr`.

9.3 The `my_addrs` function is shown in Figure E.11 and the `main` function in Figure E.12.

names/myaddrs1.c

```
 1 #include    "unp.h"
 2 #include    <sys/param.h>

 3 char **
 4 my_addrs(int *addrtype)
 5 {
 6     struct hostent *hptr;
 7     char    myname[MAXHOSTNAMELEN];

 8     if (gethostname(myname, sizeof(myname)) < 0)
 9         return (NULL);

10     if ( (hptr = gethostbyname(myname)) == NULL)
11         return (NULL);

12     *addrtype = hptr->h_addrtype;
13     return (hptr->h_addr_list);
14 }
```

names/myaddrs1.c

Figure E.11 Version of Figure 9.7 that calls `gethostname`.

9.4 If `gethostbyname` returns a `hostent` structure specifying one or more IPv6 addresses, `h_length` will be 16. This will overflow the `sockaddr_in` structure, writing over whatever happens to follow it in memory. This resolver option should be set *only* if the program is prepared to deal with IPv6 addresses, which our program is not. This example also shows why the length argument to `memcpy` should be the size of the destination, `sizeof(struct in_addr)` in this example, and not the size of the source, `hp->h_length`, even if the two should be the same.

9.5 The `chargen` server sends data to the client until the client closes the connection (i.e., until you abort the client).

names/prmyaddrs1.c

```
1 #include    "unp.h"
2 char  **my_addrs(int *);
3 int
4 main(int argc, char **argv)
5 {
6     int     addrtype;
7     char  **pptr, buf[INET6_ADDRSTRLEN];
8     if ( (pptr = my_addrs(&addrtype)) == NULL)
9         err_quit("my_addrs error");
10    for ( ; *pptr != NULL; pptr++)
11        printf("\taddress: %s\n",
12               Inet_ntop(addrtype, *pptr, buf, sizeof(buf)));
13    exit(0);
14 }
```

names/prmyaddrs1.c

Figure E.12 Test program for Figures 9.7 and E.11.

9.6 As mentioned with Figure 9.3, this is a feature of newer releases of BIND. Figure E.13 shows the modified version. The order of the tests on the hostname string is important. We call `inet_pton` first, as it is a fast, in-memory test for whether or not the string is a valid dotted-decimal IP address. Only if this fails do we call `gethostbyname`, which typically involves some network resources and some time.

If the string is a valid dotted-decimal IP address, we make our own array of pointers (`addrs`) to the single IP address, allowing the loop using `pptr` to remain the same.

Since the address has already been converted to binary in the socket address structure, we change the call to `memcpy` in Figure 9.8 to call `memmove` instead, because when a dotted-decimal IP address is entered, the source and destination fields are the same in this call. Exercise 30.3 talks about overlapping fields with `memcpy` and `memmove`.

names/daytimetcpcli2.c

```
1 #include    "unp.h"
2 int
3 main(int argc, char **argv)
4 {
5     int     sockfd, n;
6     char    recvline[MAXLINE + 1];
7     struct sockaddr_in servaddr;
8     struct in_addr **pptr, *addrs[2];
9     struct hostent *hp;
10    struct servent *sp;
11    if (argc != 3)
12        err_quit("usage: daytimetcpcli2 <hostname> <service>");
```

```
13      bzero(&servaddr, sizeof(servaddr));
14      servaddr.sin_family = AF_INET;
15      if (inet_pton(AF_INET, argv[1], &servaddr.sin_addr) == 1) {
16          addrs[0] = &servaddr.sin_addr;
17          addrs[1] = NULL;
18          pptr = &addrs[0];
19      } else if ((hp = gethostbyname(argv[1])) != NULL) {
20          pptr = (struct in_addr **) hp->h_addr_list;
21      } else
22          err_quit("hostname error for %s: %s", argv[1], hstrerror(h_errno));
23      if ( (n = atoi(argv[2])) > 0)
24          servaddr.sin_port = htons(n);
25      else if ((sp = getservbyname(argv[2], "tcp")) != NULL)
26          servaddr.sin_port = sp->s_port;
27      else
28          err_quit("getservbyname error for %s", argv[2]);
29      for ( ; *pptr != NULL; pptr++) {
30          sockfd = Socket(AF_INET, SOCK_STREAM, 0);
31          memmove(&servaddr.sin_addr, *pptr, sizeof(struct in_addr));
32          printf("trying %s\n",
33                 Sock_ntop((SA *) &servaddr, sizeof(servaddr)));
34          if (connect(sockfd, (SA *) &servaddr, sizeof(servaddr)) == 0)
35              break;              /* success */
36          err_ret("connect error");
37          close(sockfd);
38      }
39      if (*pptr == NULL)
40          err_quit("unable to connect");
41      while ( (n = Read(sockfd, recvline, MAXLINE)) > 0) {
42          recvline[n] = 0;        /* null terminate */
43          Fputs(recvline, stdout);
44      }
45      exit(0);
46  }
```

names/daytimetcpcli2.c

Figure E.13 Allow dotted-decimal IP address or hostname, port number or service name.

9.7 Figure E.14 shows the program.

names/daytimetcpcli3.c

```
1  #include    "unp.h"
2  int
3  main(int argc, char **argv)
4  {
5      int     sockfd, n;
6      char    recvline[MAXLINE + 1];
7      struct sockaddr_in servaddr;
8      struct sockaddr_in6 servaddr6;
9      struct sockaddr *sa;
10     socklen_t salen;
11     struct in_addr **pptr;
```

```
12      struct hostent *hp;
13      struct servent *sp;
14      if (argc != 3)
15          err_quit("usage: daytimetcpcli3 <hostname> <service>");
16      if ( (hp = gethostbyname(argv[1])) == NULL)
17          err_quit("hostname error for %s: %s", argv[1], hstrerror(h_errno));
18      if ( (sp = getservbyname(argv[2], "tcp")) == NULL)
19          err_quit("getservbyname error for %s", argv[2]);
20      pptr = (struct in_addr **) hp->h_addr_list;
21      for ( ; *pptr != NULL; pptr++) {
22          sockfd = Socket(hp->h_addrtype, SOCK_STREAM, 0);
23          if (hp->h_addrtype == AF_INET) {
24              sa = (SA *) &servaddr;
25              salen = sizeof(servaddr);
26          } else if (hp->h_addrtype == AF_INET6) {
27              sa = (SA *) &servaddr6;
28              salen = sizeof(servaddr6);
29          } else
30              err_quit("unknown addrtype %d", hp->h_addrtype);
31          bzero(sa, salen);
32          sa->sa_family = hp->h_addrtype;
33          sock_set_port(sa, salen, sp->s_port);
34          sock_set_addr(sa, salen, *pptr);
35          printf("trying %s\n", Sock_ntop(sa, salen));
36          if (connect(sockfd, sa, salen) == 0)
37              break;              /* success */
38          err_ret("connect error");
39          close(sockfd);
40      }
41      if (*pptr == NULL)
42          err_quit("unable to connect");
43      while ( (n = Read(sockfd, recvline, MAXLINE)) > 0) {
44          recvline[n] = 0;        /* null terminate */
45          Fputs(recvline, stdout);
46      }
47      exit(0);
48  }
```

names/daytimetcpcli3.c

Figure E.14 Modification of Figure 9.8 to work with IPv4 and IPv6.

We use the `h_addrtype` value returned by `gethostbyname` to determine the type of address. We also use our `sock_set_port` and `sock_set_addr` functions (Section 3.8) to set these two fields in the appropriate socket address structure.

Although this program works, it has two limitations. First, we must handle all the differences, looking at `h_addrtype` and then setting `sa` and `salen` appropriately. A better solution is to have a library function that not only looks up the hostname and the service name but also fills in the entire socket address structure

(e.g., `getaddrinfo` in Section 11.2). Second, this program compiles only on hosts that support IPv6. To make this compile on an IPv4-only host would add numerous `#ifdefs` to the code, complicating it.

We return to the concept of protocol independence in Chapter 11 and see better ways to accomplish it.

Chapter 10

10.1 Here are the relevant excepts (e.g., with the login and directory listings omitted).

```
solaris % ftp bsdi
Connected to bsdi.kohala.com.
220 bsdi.kohala.com FTP server ...
...
230 Guest login ok, access restrictions apply.
ftp> debug
Debugging on (debug=1).
ftp> dir
---> PORT 206,62,226,33,129,145
200 PORT command successful.
---> LIST
150 Opening ASCII mode data connection for /bin/ls.

...

solaris % ftp sunos5
Connected to sunos5.kohala.com.
220 sunos5.kohala.com FTP server ...
...
230 Guest login ok, access restrictions apply.
ftp> debug
Debugging on (debug=1).
ftp> dir
---> LPRT 6,16,95,27,223,0,206,62,226,0,0,32,8,0,32,120,227,227,2,129,148
200 LPRT command successful.
---> LIST
150 ASCII data connection for /bin/ls (5f1b:df00:ce3e:e200:20:800:2078:e3e3,
3172) (0 bytes).
```

Chapter 11

11.1 Allocate a big buffer (larger than any socket address structure) and call `getsockname`. The third argument is a value–result argument that returns the actual size of the protocol's addresses. Unfortunately, this works only for protocols with fixed-length socket address structures (e.g., IPv4 and IPv6) but is not guaranteed to work with protocols that can return variable-length socket address structures (e.g., Unix domain sockets, Chapter 14).

11.2 We first allocate arrays to hold the hostname and service name:

```
char   host[NI_MAXHOST], serv[NI_MAXSERV];
```

After `accept` returns, we call `getnameinfo` instead of `sock_ntop`:

```
if (getnameinfo(cliaddr, len, host, NI_MAXHOST, serv, NI_MAXSERV,
                NI_NUMERICHOST | NI_NUMERICSERV) == 0)
    printf("connection from %s.%s\n", host, serv);
```

Since this is a server, we specify the `NI_NUMERICHOST` and `NI_NUMERICSERV` flags, to avoid a DNS look and a lookup of `/etc/services`.

11.3 The first problem is that the second server cannot `bind` the same port as the first server because the `SO_REUSEADDR` socket option is not set. The easiest way to handle this is to make a copy of the `udp_server` function, rename it `udp_server_reuseaddr`, have it set the socket option, and call this new function from the server.

11.4 If the variable were global, `getaddrinfo` would not be thread-safe.

11.5 Yes. `ftp://ftp.isi.edu/in-notes/iana/assignments/port-numbers` shows that the service names `c1/1` and `914c/g` both contain slashes.

11.6 When the client outputs "Trying 206.62.226.35..." `gethostbyname` has returned the IP address. Any client pause before this is the time taken by the resolver to look up the hostname. The output "Connected to bsdi.kohala.com." means `connect` has returned. Any pause between these two lines of output is the time taken by `connect` to establish the connection.

Chapter 12

12.1 The `close` of the descriptors in `daemon_init` closes the listening TCP socket that was created by `tcp_listen`. Since programs that we write as a daemon might be executed from one of the system startup scripts, we should not assume that any error message can be written to a terminal. All error messages, even a startup error such as an invalid command-line argument, should be logged using `syslog`.

12.2 The TCP versions of the `echo`, `discard`, and `chargen` servers all run as a child process after being `fork`ed by `inetd`, because these three run until the client terminates the connection. The other two TCP servers, `time` and `daytime`, do not require a `fork` because their service is trivial to implement (get the current time and date, format it, write it, and close the connection), so these two are handled directly by `inetd`. All five UDP services are handled without a `fork` because each generates at most a single datagram in response to the client datagram that triggers the service. These five are therefore handled directly by `inetd`.

12.3 This is a well-known denial of service attack ([CERT 1996a]). The first datagram from port 7 causes the `chargen` server to send a datagram back to port 7. This is echoed and sends another datagram to the `chargen` server. This loop continues. One solution, implemented in BSD/OS, is to refuse datagrams to any of the internal servers if the source port of the incoming datagram belongs to any of the internal servers. Another solution is to disable these internal services, either through `inetd` on each host, or at an organizations's router to the Internet.

12.4 The client's IP address and port are obtained from the socket address structure filled in by `accept`.

The reason `inetd` does not do this for a UDP socket is because the `recvfrom` to read the datagram is performed by the actual server that is `exec`ed, and not by `inetd` itself.

`inetd` could read the datagram specifying the `MSG_PEEK` flag (Section 13.7), just to obtain the client's IP address and port, but leaving the datagram in place for the actual server to read.

Chapter 13

13.1 If no handler had been set, the return from the first call to `signal` will be `SIG_DFL` and the call to `signal` to reset the handler just sets it back to its default.

13.3 Here is just the `for` loop:

```
for ( ; ; ) {
    if ( (n = Recv(sockfd, recvline, MAXLINE, MSG_PEEK)) == 0)
        break;      /* server closed connection */

    Ioctl(sockfd, FIONREAD, &npend);
    printf("%d bytes from PEEK, %d bytes pending\n", n, npend);

    n = Read(sockfd, recvline, MAXLINE);
    recvline[n] = 0;    /* null terminate */
    Fputs(recvline, stdout);
}
```

13.4 The data is still output because falling off the end of the `main` function is the same as returning from this function, and the `main` function is called by the C startup routine as

```
exit(main(argc, argv));
```

Hence `exit` is called, and the standard I/O cleanup routine is called.

Chapter 14

14.1 The `unlink` removes the pathname from the filesystem and when the client calls `connect` at a later time, the `connect` will fail. The server's listening socket is not affected, but no clients will be able to `connect` after the `unlink`.

14.2 The client cannot `connect` to the server even if the pathname still exists, because for the `connect` to succeed a Unix domain socket must be currently open and bound to that pathname (Section 14.4).

14.3 When the server prints the client's protocol address by calling `sock_ntop`, the output is "datagram from (no pathname bound)" because no pathname is bound to the client's socket by default.

One solution is to specifically check for a Unix domain socket in `udp_client` and `udp_connect` and `bind` a temporary pathname to the socket. This puts the protocol dependency in the library function, where it belongs, and not in our application.

14.4 Even though we force 1-byte `writes` by the server for its 26-byte reply, putting the `sleep` in the client guarantees that all 26 segments are received before `read` is called, causing `read` to return the entire reply. This is just to confirm (again) that TCP is a byte stream with no inherent record markers.

To use the Unix domain protocols we start the client and server with the two command-line arguments `/local` (or `/unix`) and `/tmp/daytime` (or any other temporary pathname that you wish to use). Nothing changes: 26 bytes are returned by `read` each time the client runs.

Since the server specifies the `MSG_EOR` flag for each `send`, each byte is considered a logical record, and `read` returns 1 byte each time it is called. What is happening here is that Berkeley-derived implementations support the `MSG_EOR` flag, by default. This is undocumented, however, and should not be used in production code. We use it here as an example of the difference between a byte stream and a record-oriented protocol. From an implementation perspective, each output operation goes into an mbuf (memory buffer) and the `MSG_EOR` flag is retained by the kernel with the mbuf as the mbuf goes from the sending socket to the receiving socket's receive buffer. When `read` is called, the `MSG_EOR` flag is still attached to each mbuf, so the generic kernel `read` routine (which supports the `MSG_EOR` flag since some protocols use the flag) returns each byte by itself. Had we used `recvmsg` instead of `read`, the `MSG_EOR` flag would be returned in the `msg_flags` member each time `recvmsg` returned 1 byte. This does not work with TCP because the sending TCP never looks at the `MSG_EOR` flag in the mbuf that it is sending, and even if it did, there is no way to pass this flag to the receiving TCP in the TCP header. (Thanks to Matt Thomas for pointing out this undocumented "feature.")

14.5 Figure E.15 shows an implementation of this program. We show an XTI version in Figure E.21.

debug/backlog.c

```
 1 #include    "unp.h"

 2 #define PORT        9999
 3 #define ADDR        "127.0.0.1"
 4 #define MAXBACKLOG  100

 5             /* globals */
 6 struct sockaddr_in serv;
 7 pid_t   pid;                    /* of child */

 8 int     pipefd[2];
 9 #define pfd pipefd[1]           /* parent's end */
10 #define cfd pipefd[0]           /* child's end */

11             /* function prototypes */
12 void    do_parent(void);
```

```
13 void    do_child(void);

14 int
15 main(int argc, char **argv)
16 {
17     if (argc != 1)
18         err_quit("usage: backlog");

19     Socketpair(AF_UNIX, SOCK_STREAM, 0, pipefd);

20     bzero(&serv, sizeof(serv));
21     serv.sin_family = AF_INET;
22     serv.sin_port = htons(PORT);
23     Inet_pton(AF_INET, ADDR, &serv.sin_addr);

24     if ( (pid = Fork()) == 0)
25         do_child();
26     else
27         do_parent();

28     exit(0);
29 }

30 void
31 parent_alrm(int signo)
32 {
33     return;                      /* just interrupt blocked connect() */
34 }

35 void
36 do_parent(void)
37 {
38     int     backlog, j, k, junk, fd[MAXBACKLOG + 1];

39     Close(cfd);
40     Signal(SIGALRM, parent_alrm);

41     for (backlog = 0; backlog <= 14; backlog++) {
42         printf("backlog = %d: ", backlog);
43         Write(pfd, &backlog, sizeof(int));  /* tell child value */
44         Read(pfd, &junk, sizeof(int));  /* wait for child */

45         for (j = 1; j <= MAXBACKLOG; j++) {
46             fd[j] = Socket(AF_INET, SOCK_STREAM, 0);
47             alarm(2);
48             if (connect(fd[j], (SA *) &serv, sizeof(serv)) < 0) {
49                 if (errno != EINTR)
50                     err_sys("connect error, j = %d", j);
51                 printf("timeout, %d connections completed\n", j - 1);
52                 for (k = 1; k <= j; k++)
53                     Close(fd[k]);
54                 break;          /* next value of backlog */
55             }
56             alarm(0);
57         }
58         if (j > MAXBACKLOG)
59             printf("%d connections?\n", MAXBACKLOG);
60     }
```

```
61      backlog = -1;               /* tell child we're all done */
62      Write(pfd, &backlog, sizeof(int));
63  }
64  void
65  do_child(void)
66  {
67      int     listenfd, backlog, junk;
68      const int on = 1;
69      Close(pfd);
70      Read(cfd, &backlog, sizeof(int));   /* wait for parent */
71      while (backlog >= 0) {
72          listenfd = Socket(AF_INET, SOCK_STREAM, 0);
73          Setsockopt(listenfd, SOL_SOCKET, SO_REUSEADDR, &on, sizeof(on));
74          Bind(listenfd, (SA *) &serv, sizeof(serv));
75          Listen(listenfd, backlog);  /* start the listen */
76          Write(cfd, &junk, sizeof(int));     /* tell parent */
77          Read(cfd, &backlog, sizeof(int));   /* just wait for parent */
78          Close(listenfd);        /* closes all queued connections too */
79      }
80  }
```

debug/backlog.c

Figure E.15 Determine actual number of queued connections for different *backlog* values.

Chapter 15

15.1 The descriptor is shared between the parent and child, so it has a reference count of 2. If the parent calls `close`, this just decrements the reference count from 2 to 1 and since it is still greater than 0, a FIN is not sent. This is another reason for the `shutdown` function: to force a FIN to be sent even if the descriptor's reference count is greater than 0.

15.2 The parent will keep writing to the socket that has received a FIN, and the first segment sent to the server will elicit an RST in response. The next `write` after this will send `SIGPIPE` to the parent as we discussed in Section 5.12.

15.3 When the child calls `getppid` to send `SIGTERM` to the parent, the returned process ID will be 1, the `init` process, which inherits all children whose parents terminate while their children are still running. The child will try to send the signal to the `init` process, but will not have adequate permission. But if there is a chance that this client could run with superuser privileges, allowing it to send this signal to `init`, then the return value of `getppid` should be tested before sending the signal.

15.4 If these two lines are removed, `select` is called. But `select` will return immediately because with the connection established the socket is writable. This test and `goto` are to avoid the unnecessary call to `select`.

15.5 This can happen if the server immediately sends data when its `accept` returns, and if the client host is busy when the second packet of the three-way handshake

arrives to complete the connection at the client end (Figure 2.5). SMTP servers, for example, immediately write to a new connection, before reading from it, to send a greeting message to the client.

Chapter 16

16.1 No, it does not matter because the first three members of the `union` in Figure 16.2 are socket address structures.

Chapter 17

17.1 The `sdl_nlen` member will be 5 and the `sdl_alen` member will be 8. This requires 21 bytes, so the size is rounded up to 24 bytes (p. 89 of TCPv2), assuming a 32-bit architecture.

17.2 The kernel's response is never sent to this socket. This socket option determines whether the kernel sends its reply to the sending process, as discussed on pp. 649–650 of TCPv2. It defaults to on, since most processes want the replies. But disabling the option prevents the replies from being sent to the sender.

Chapter 18

18.1 If you get more than a few replies, they should not be in the same order each time. The sending host, however, is normally the first reply, since the datagrams to and from it do not appear on the actual network.

18.2 1472 is the Ethernet MTU (1500) minus 20 bytes for the IPv4 header and minus 8 bytes for the UDP header.

18.3 Under BSD/OS when the signal handler writes the byte to the pipe and then returns, `select` returns `EINTR`. It is called again and returns readability on the pipe.

Chapter 19

19.1 When we run the program the output is:

```
solaris % udpcli05 224.0.0.1
hi
from 206.62.226.34: Thu Jun 19 17:28:32 1997
from 206.62.226.43: Thu Jun 19 17:28:32 1997
from 206.62.226.42: Thu Jun 19 17:28:32 1997
from 206.62.226.40: Thu Jun 19 17:28:32 1997
from 206.62.226.35: Thu Jun 19 17:28:32 1997
```

The five responding hosts are running AIX, BSD/OS, Digital Unix, and Linux. The only multicast-capable nodes not responding are those running Solaris 2.5 and the Cisco router.

What is happening here is that the destination address of the UDP datagram is 224.0.0.1, the all-hosts group that all multicast-capable nodes must join. The UDP datagram is sent as a multicast Ethernet frame and all the multicast-capable nodes receive the datagram, since they all belong to the group. The responding hosts all pass the received datagram to the UDP daytime server (normally part of `inetd`) even though that socket has not joined the group. The Solaris implementation, however, requires that the destination socket must join the group to receive the datagram.

This example demonstrates that a UDP program that was never designed to respond to multicast datagrams can receive these datagrams. We saw the same thing happen with this daytime example in Chapter 18: a UDP program that was never designed to respond to broadcast datagrams can receive these datagrams.

19.2 Figure E.16 shows a simple modification to the `main` function to `bind` the multicast address and port 0.

mcast/udpcli06.c

```
 1 #include    "unp.h"

 2 int
 3 main(int argc, char **argv)
 4 {
 5     int     sockfd;
 6     socklen_t salen;
 7     struct sockaddr *cli, *serv;

 8     if (argc != 2)
 9         err_quit("usage: udpcli06 <IPaddress>");

10     sockfd = Udp_client(argv[1], "daytime", (void **) &serv, &salen);

11     cli = Malloc(salen);
12     memcpy(cli, serv, salen);    /* copy socket address struct */
13     sock_set_port(cli, salen, 0);    /* and set port to 0 */
14     Bind(sockfd, cli, salen);

15     dg_cli(stdin, sockfd, serv, salen);

16     exit(0);
17 }
```

mcast/udpcli06.c

Figure E.16 UDP client `main` function that `bind`s a multicast address.

Unfortunately on the three systems that this was tried on, BSD/OS, Digital Unix, and Solaris 2.5, all allow the `bind` and then send the UDP datagrams with a multicast source IP address. The five responding systems (the same ones as in the previous exercise) all swap the source and destination IP addresses in the reply, so all five replies are multicast! Nothing happens on the multicast-capable client hosts with the replies that they receive, because the destination port of the replies is the ephemeral port that the client kernel chose when the multicast address was bound, and there were no sockets bound to that port. ICMP port unreachables are not generated in response to a UDP datagram that is multicast.

19.3 If we do this from our host `solaris`, which is multicast capable, we get:

```
solaris % ping 224.0.0.1
PING 224.0.0.1: 56 data bytes
64 bytes from solaris.kohala.com (206.62.226.33): icmp_seq=0. time=4. ms
64 bytes from linux.kohala.com (206.62.226.40): icmp_seq=0. time=9. ms
64 bytes from aix.kohala.com (206.62.226.43): icmp_seq=0. time=11. ms
64 bytes from bsdi.kohala.com (206.62.226.35): icmp_seq=0. time=13. ms
64 bytes from alpha.kohala.com (206.62.226.42): icmp_seq=0. time=15. ms
64 bytes from sunos5.kohala.com (206.62.226.36): icmp_seq=0. time=17. ms
64 bytes from bsdi2.kohala.com (206.62.226.34): icmp_seq=0. time=54. ms
64 bytes from gw.kohala.com (206.62.226.62): icmp_seq=0. time=75. ms
^?
----224.0.0.1 PING Statistics----
1 packets transmitted, 8 packets received, 8.00 times amplification
round-trip (ms)  min/avg/max = 4/24/75
```

Every host from the top Ethernet in Figure 1.16 responds (including the sender, of course) except `unixware`, which is not multicast capable.

19.4 Since the kernel is not multicast capable it treats the destination as a normal IP address. It looks up 224.0.0.1 in the normal IP routing table and matches the default route, which points to the router `gw` (recall Figure 1.16). A unicast ICMP echo request is sent to this router with the destination IP address of 224.0.0.1 and the hardware address of that router's Ethernet interface (that is, the hardware address is not a multicast address). The router accepts the received packet because it is addressed to its interface and the destination IP address is a multicast group that the router belongs to.

19.5 If we do this from our host `solaris` we get:

```
solaris % ping 224.0.0.2
PING 224.0.0.2: 56 data bytes
64 bytes from bsdi.kohala.com (206.62.226.35): icmp_seq=0. time=3. ms
64 bytes from gw.kohala.com (206.62.226.62): icmp_seq=0. time=24. ms
^?
----224.0.0.2 PING Statistics----
1 packets transmitted, 2 packets received, 2.00 times amplification
round-trip (ms)  min/avg/max = 3/13/24
```

We expect `bsdi` to respond, as it is the multicast router on the subnet with a tunnel to the MBone (Section B.2) and is running `mrouted`. The router `gw` also responds, but it is not acting as a multicast router.

19.7 In Figure 19.17 we see that NTP timestamps are the number of seconds since January 1, 1900. There are 31,536,000 seconds in a year ($365 \times 24 \times 60 \times 60$) so the two values are about 96.7 years (ignoring leap years), which makes sense. Also, the version number of this announcement is greater than the session ID (which we expect was assigned when the session was first announced), which also makes sense.

19.8 The value 1,073,741,824 is converted to a floating-point number and divided by 4,294,967,296, yielding 0.250. This is multiplied by 1,000,000, yielding 250,000, which in microseconds is one-quarter of a second.

The largest fraction is 4,294,967,295, which divided by 4,294,967,296 yields 0.99999999976716935634. Multiplying this by 1,000,000 and truncating to an integer yields 999,999, the largest value for the number of microseconds.

19.9 In our discussion of these two socket options in Section 7.5 we noted that `SO_REUSEADDR` is considered equivalent to `SO_REUSEPORT` if the IP address being bound is a multicast address. This simplifies the portability of our code, because if this were not the case we would have to write:

```
#ifdef SO_REUSEPORT
    Setsockopt(fd, SOL_SOCKET, SO_REUSEPORT, &on, sizeof(on));
#else
    Setsockopt(fd, SOL_SOCKET, SO_REUSEADDR, &on, sizeof(on));
#endif
```

19.10 Some systems will not deliver a multicast datagram to a socket unless that socket has joined the multicast group. Just binding the port is inadequate on these systems. Solaris 2.5 operates in this fashion. Berkeley-derived implementations, on the other hand, deliver the datagram to all matching sockets, regardless of whether that particular socket has joined the group. Recall that our `bind_ubcast` function is called from Figure 19.22 to `bind` the socket to the wildcard address, so the multicast group is not joined on this socket.

19.11 Twelve additional datagrams would be received: three multicasts are sent in Figure 19.31 and each one is then delivered to four matching sockets (three multicast sockets plus the wildcard socket).

Chapter 20

20.1 Recall that `sock_ntop` uses its own static buffer to hold the result. If we call it twice as arguments in a call to `printf`, the second call overwrites the result of the first call.

20.2 Yes, if the reply contains 0 bytes of user data (i.e., just an `hdr` structure).

20.3 Since `select` does not modify the `timeval` structure that specifies its time limit, you need to note the time when the first packet is sent (this is already returned in units of milliseconds by `rtt_ts`). If `select` returns with the socket being readable, note the current time, and if `recvmsg` is called again, calculate the new timeout for `select`.

20.4 The common technique is to create one socket per interface address, as we did in Sections 19.11 and 20.6, and send the reply from the same socket on which the request arrived.

20.5 Calling `getaddrinfo` without a hostname argument and without the `AI_PASSIVE` flag set causes it to assume the localhost: 0::1 (IPv6) and 127.0.0.1 (IPv4). Recall that an IPv6 socket address structure is returned before an IPv4 socket address structure by `getaddrinfo`, assuming IPv6 is supported. If both protocols are supported on the host, the call to `socket` in `udp_client` will succeed with the family equal to `AF_INET6`.

Figure E.17 is the protocol-independent version of this program.

advio/udpserv04.c

```
 1 #include    "unpifi.h"
 2 void    mydg_echo(int, SA *, socklen_t);
 3 int
 4 main(int argc, char **argv)
 5 {
 6     int     sockfd, family, port;
 7     const int on = 1;
 8     pid_t   pid;
 9     socklen_t salen;
10     struct sockaddr *sa, *wild;
11     struct ifi_info *ifi, *ifihead;
12     if (argc == 2)
13         sockfd = Udp_client(NULL, argv[1], (void **) &sa, &salen);
14     else if (argc == 3)
15         sockfd = Udp_client(argv[1], argv[2], (void **) &sa, &salen);
16     else
17         err_quit("usage: udpserv04 [ <host> ] <service or port>");
18     family = sa->sa_family;
19     port = sock_get_port(sa, salen);
20     Close(sockfd);              /* we just want family, port, salen */
21     for (ifihead = ifi = Get_ifi_info(family, 1);
22          ifi != NULL; ifi = ifi->ifi_next) {
23             /* bind unicast address */
24         sockfd = Socket(family, SOCK_DGRAM, 0);
25         Setsockopt(sockfd, SOL_SOCKET, SO_REUSEADDR, &on, sizeof(on));
26         sock_set_port(ifi->ifi_addr, salen, port);
27         Bind(sockfd, ifi->ifi_addr, salen);
28         printf("bound %s\n", Sock_ntop(ifi->ifi_addr, salen));
29         if ( (pid = Fork()) == 0) {  /* child */
30             mydg_echo(sockfd, ifi->ifi_addr, salen);
31             exit(0);            /* never executed */
32         }
33         if (ifi->ifi_flags & IFF_BROADCAST) {
34                 /* try to bind broadcast address */
35             sockfd = Socket(family, SOCK_DGRAM, 0);
36             Setsockopt(sockfd, SOL_SOCKET, SO_REUSEADDR, &on, sizeof(on));
37             sock_set_port(ifi->ifi_brdaddr, salen, port);
38             if (bind(sockfd, ifi->ifi_brdaddr, salen) < 0) {
39                 if (errno == EADDRINUSE) {
40                     printf("EADDRINUSE: %s\n",
41                            Sock_ntop(ifi->ifi_brdaddr, salen));
42                     Close(sockfd);
43                     continue;
44                 } else
45                     err_sys("bind error for %s",
46                             Sock_ntop(ifi->ifi_brdaddr, salen));
47             }
48             printf("bound %s\n", Sock_ntop(ifi->ifi_brdaddr, salen));
```

```
49              if ( (pid = Fork()) == 0) {  /* child */
50                  mydg_echo(sockfd, ifi->ifi_brdaddr, salen);
51                  exit(0);        /* never executed */
52              }
53          }
54      }
55          /* bind wildcard address */
56      sockfd = Socket(family, SOCK_DGRAM, 0);
57      Setsockopt(sockfd, SOL_SOCKET, SO_REUSEADDR, &on, sizeof(on));
58      wild = Malloc(salen);
59      memcpy(wild, sa, salen);    /* copy family and port */
60      sock_set_wild(wild, salen);
61      Bind(sockfd, wild, salen);
62      printf("bound %s\n", Sock_ntop(wild, salen));
63      if ( (pid = Fork()) == 0) {  /* child */
64          mydg_echo(sockfd, wild, salen);
65          exit(0);                /* never executed */
66      }
67      exit(0);
68  }
69  void
70  mydg_echo(int sockfd, SA *myaddr, socklen_t salen)
71  {
72      int     n;
73      char    mesg[MAXLINE];
74      socklen_t len;
75      struct sockaddr *cli;
76      cli = Malloc(salen);
77      for ( ; ; ) {
78          len = salen;
79          n = Recvfrom(sockfd, mesg, MAXLINE, 0, cli, &len);
80          printf("child %d, datagram from %s",
81                 getpid(), Sock_ntop(cli, len));
82          printf(", to %s\n", Sock_ntop(myaddr, salen));
83          Sendto(sockfd, mesg, n, 0, cli, len);
84      }
85  }
```

advio/udpserv04.c

Figure E.17 Protocol-independent version of program from Section 20.6.

Chapter 21

21.1 Yes. In the first example 2 bytes are sent with a single urgent pointer that points to the byte following the `b`. But in the second example (the two function calls), first the `a` is sent with an urgent pointer that points just beyond it, and this is followed by another TCP segment containing the `b` with a different urgent pointer that points just beyond it.

21.2 Figure E.18 shows the version using `poll`.

oob/tcprecv03p.c

```
 1 #include    "unp.h"
 2 int
 3 main(int argc, char **argv)
 4 {
 5     int     listenfd, connfd, n, justreadoob = 0;
 6     char    buff[100];
 7     struct pollfd pollfd[1];
 8     if (argc == 2)
 9         listenfd = Tcp_listen(NULL, argv[1], NULL);
10     else if (argc == 3)
11         listenfd = Tcp_listen(argv[1], argv[2], NULL);
12     else
13         err_quit("usage: tcprecv03p [ <host> ] <port#>");
14     connfd = Accept(listenfd, NULL, NULL);
15     pollfd[0].fd = connfd;
16     pollfd[0].events = POLLRDNORM;
17     for ( ; ; ) {
18         if (justreadoob == 0)
19             pollfd[0].events |= POLLRDBAND;
20         Poll(pollfd, 1, INFTIM);
21         if (pollfd[0].revents & POLLRDBAND) {
22             n = Recv(connfd, buff, sizeof(buff) - 1, MSG_OOB);
23             buff[n] = 0;        /* null terminate */
24             printf("read %d OOB byte: %s\n", n, buff);
25             justreadoob = 1;
26             pollfd[0].events &= ~POLLRDBAND;    /* turn bit off */
27         }
28         if (pollfd[0].revents & POLLRDNORM) {
29             if ( (n = Read(connfd, buff, sizeof(buff) - 1)) == 0) {
30                 printf("received EOF\n");
31                 exit(0);
32             }
33             buff[n] = 0;        /* null terminate */
34             printf("read %d bytes: %s\n", n, buff);
35             justreadoob = 0;
36         }
37     }
38 }
```

oob/tcprecv03p.c

Figure E.18 Version of Figure 21.6 using `poll` instead of `select`.

Chapter 22

22.1 No, the modification introduces an error. The problem is that `nqueue` is decremented before the array entry `dg[iget]` is processed, allowing the signal handler to read a new datagram into this array element.

Chapter 23

23.1 In the `fork` example there will be 101 descriptors in use, one listening socket and 100 connected sockets. But each of the 101 processes (one parent, 100 children) has just one descriptor open (ignoring any others, such as standard input, if the server is not daemonized). In the threaded server, however, there are 101 descriptors in the single process. Each thread (including the main thread) is handling one descriptor.

23.2 The final two segments of the TCP connection termination—the server's FIN and the client's ACK of this FIN—will not be exchanged. This leaves the client's end of the connection in the FIN_WAIT_2 state (Figure 2.4). Berkeley-derived implementations will time out the client's end when it remains in this state for just over 11 minutes (pp. 825–827 of TCPv2). The server will also run out of descriptors (eventually).

23.3 This message should be printed by the main thread when it reads an end-of-file from the socket *and* the other thread is still running. A simple way to do this is to declare another external named `done` that is initialized to 0. Before the thread `copyto` returns, it sets this variable to 1. The main thread checks this variable, and if 0, prints the error message. Since only one thread sets the variable, there is no need for any synchronization.

Chapter 24

24.1 Nothing changes at the server. First, that router employs the common weak end system model, so it accepts the incoming datagrams from the Ethernet even though the destination address is that of its other interface. Next, with an IPv4 source route the forwarding host replaces its address in the list with the address of the outgoing interface. That outgoing interface is the Ethernet interface (206.62.226.62), regardless of which address the packet is sent to.

24.2 Nothing changes. All the systems are neighbors, so a strict source route is identical to a loose source route.

24.3 We would place an EOL (a byte of 0) at the end of the buffer.

24.4 Since `ping` creates a raw socket (Chapter 25), it receives the complete IP header, including any IP options, on every datagram that it reads with `recvfrom`.

24.5 Because `rlogind` is invoked by `inetd` (Section 12.5).

24.6 The problem is that the fifth argument to `setsockopt` is the pointer to the length, instead of the length. This bug was probably fixed when ANSI C prototypes were first used.

As it turns out the bug is harmless, because as we mentioned, to clear the `IP_OPTIONS` socket option we can specify either a null pointer as the fourth argument or a fifth argument (the length) of 0 (p. 269 of TCPv2).

Chapter 25

25.1 The version number field and the next header field in the IPv6 header are not available. The payload length field is available as either an argument to one of the output functions or as the return value from one of the input functions, but if a jumbo payload option is required, that actual option itself is not available to an application. The fragment header is also not available to an application.

25.2 Eventually the client's socket receive buffer will fill, causing the daemon's `write` to block. We do not want this to happen, as that stops the daemon from handling any more data on any of its sockets. The easiest solution is for the daemon to set its end of the Unix domain connection to the client nonblocking. The daemon must then call `write` instead of the wrapper function `Write` and just ignore an error of `EWOULDBLOCK`.

25.3 Berkeley-derived kernels, by default, allow broadcasting on a raw socket (p. 1057 of TCPv2). The `SO_BROADCAST` socket option needs to be specified only for UDP sockets.

25.4 Our program does not check for a multicast address and does not set the `IP_MULTICAST_IF` socket option. Therefore the kernel chooses the outgoing interface, probably by searching the routing table for 224.0.0.1. We also do not set the `IP_MULTICAST_TTL` field, so it defaults to 1, which is OK.

Chapter 26

26.1 This flag indicates that the jump buffer has been set by `sigsetjmp` (Figure 26.10). While the flag may seem superfluous, there is a chance that the signal can be delivered after the signal handler is established, but before the call to `sigsetjmp`. Even if the program doesn't cause the signal to be generated, signals can also be generated in other ways (such as with the `kill` command).

Chapter 27

27.1 The parent keeps the listening socket open in case it needs to `fork` additional children at some later time (which would be an enhancement to our code).

27.2 Yes, a datagram socket can be used to pass a descriptor instead of using a stream socket. With a datagram socket the parent does not receive an end-of-file on its end of the stream pipe when a child terminates prematurely, but the parent could use `SIGCHLD` for this purpose. Realize that one difference in this scenario, where `SIGCHLD` can be used versus our `icmpd` daemon in Section 25.7, is that in the latter there was not a parent–child relationship between the client and server, so the end-of-file on the stream pipe was the only way for the server to detect the disappearance of a client.

Chapter 28

28.1 Not usually, because the only type of application that sets the `qlen` member nonzero is a connection-oriented server (e.g., TCP). But servers normally bind their well-known port, instead of letting the system choose an ephemeral port. One exception is an RPC (remote procedure call) server, which binds an ephemeral port and then registers the port with the RPC port mapper.

28.2 The receipt of an ICMP destination unreachable in response to a SYN is not a fatal error. TCP should retransmit the SYN some number of times, or until its retransmission timer expires.

28.3 A `write` to a socket that has received an RST generates `SIGPIPE`. If the process does nothing with this signal, its default action terminates the process. If the process ignores the signal, `write` returns an error of `EPIPE`.

Chapter 29

29.1 Yes, the function is thread-safe. `setnetconfig` dynamically allocates the memory used to hold the `netconfig` structure, and the arrays that it points to. This memory is released by `endnetconfig`. Be careful not to reference the `netconfig` structure returned by `getnetconfig` after calling `endnetconfig`, as that memory has been freed.

29.2 Figure E.19 shows the program.

debug/test06.c

```
 1 #include    "unpxti.h"

 2 int
 3 main(int argc, char **argv)
 4 {
 5     int     fd;
 6     struct t_call *tcall;

 7     fd = T_open(XTI_TCP, O_RDWR, NULL);

 8     tcall = T_alloc(fd, T_CALL, T_ALL);
 9     printf("first t_alloc OK\n");

10     tcall = T_alloc(fd, T_CALL, T_ADDR | T_OPT | T_UDATA);
11     printf("second t_alloc OK\n");

12     exit(0);
13 }
```

debug/test06.c

Figure E.19 Compare `T_ALL` versus specifying each `netbuf` structure.

When we execute this program the output is

```
alpha % test06
first t_alloc OK
t_alloc error: system error: Invalid argument
```

A TCP endpoint does not support user data with a connection request (the value of –2 for the `connect` row in Figure 28.1). When `T_ALL` is specified, these unsupported structures are skipped by `t_alloc` and we saw in Figure 29.5 that `udata.len` is set to 0 and the `udata.buf` is set to a null pointer. But if we specify each of the three `netbuf` structures as the third argument to `t_alloc`, instead of `T_ALL`, an error is returned if any of the specified fields are unsupported. This is another reason to always use `T_ALL`.

29.3 The function cannot tell the type of structure given just its pointer. If `t_free` released only the structure, this argument would not be required. But since it goes through the structure and releases any buffers pointed to by `netbuf` structures within the structure, it must know the type of structure.

29.4 We are not guaranteed of the order of the elements with the structure.

Chapter 30

30.1 This technique does not work because an integer is normally stored in 32 bits, while pointers require 64 bits (Figure 1.17).

30.2 The Posix.1 `PATH_MAX` constant does not include the terminating null byte.

30.3 In Figure E.20 we first draw the 4-byte array and the two 2-byte fields, which shows the overlap of the source and destination.

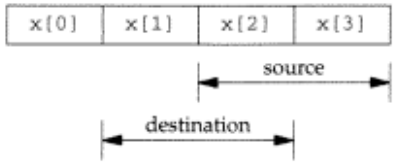


Figure E.20 Source and destination of copy operation overlap.

If the copy is done from the beginning of the source to the beginning of the destination, as in the C fragment

```
while (nbytes--)
    *dst++ = *src++;
```

then two assignment statements are executed:

```
x[1] = x[2];
x[2] = x[3];
```

which is correct. But if the copy is done in the other direction, as in the C fragment

```
src += nbytes;
dst += nbytes;
while (nbytes--)
    *--dst = *--src;
```

then the two assignment statements are

```
x[2] = x[3];
x[1] = x[2];
```

which is wrong (why?). There is no guarantee in which direction `memcpy` copies; hence when the fields overlap, `memmove` must be called. `memmove` handles the overlap by copying in the "correct" direction, depending on the relationship between the source and destination.

30.4 Figure E.21 shows the two functions `do_parent` and `do_child`. The only change prior to this from Figure E.15 is to `#include unpxti.h` instead of `unp.h`.

debug/qlen.c

```
35 void
36 do_parent(void)
37 {
38     int     qlen, j, k, junk, fd[MAXBACKLOG + 1];
39     struct t_call tcall;

40     Close(cfd);
41     Signal(SIGALRM, parent_alrm);

42     for (qlen = 0; qlen <= 14; qlen++) {
43         printf("qlen = %d: ", qlen);
44         Write(pfd, &qlen, sizeof(int));     /* tell child value */
45         Read(pfd, &junk, sizeof(int));  /* wait for child */

46         for (j = 0; j <= MAXBACKLOG; j++) {
47             fd[j] = T_open(XTI_TCP, O_RDWR, NULL);
48             T_bind(fd[j], NULL, NULL);

49             tcall.addr.maxlen = sizeof(serv);
50             tcall.addr.len = sizeof(serv);
51             tcall.addr.buf = &serv;
52             tcall.opt.len = 0;
53             tcall.udata.len = 0;

54             alarm(2);
55             if (t_connect(fd[j], &tcall, NULL) < 0) {
56                 if (errno != EINTR)
57                     err_xti("t_connect error, j = %d", j);
58                 printf("timeout, %d connections completed\n", j - 1);
59                 for (k = 1; k < j; k++)
60                     T_close(fd[k]);
61                 break;          /* next value of qlen */
62             }
63             alarm(0);
64         }
65         if (j > MAXBACKLOG)
66             printf("%d connections?\n", MAXBACKLOG);
67     }
68     qlen = -1;                  /* tell child we're all done */
69     Write(pfd, &qlen, sizeof(int));
70 }
```

```
71 void
72 do_child(void)
73 {
74     int     listenfd, qlen, junk;
75     struct t_bind tbind, tbindret;

76     Close(pipefd[1]);

77     Read(cfd, &qlen, sizeof(int));  /* wait for parent */
78     while (qlen >= 0) {
79         listenfd = T_open(XTI_TCP, O_RDWR, NULL);

80         tbind.addr.maxlen = sizeof(serv);
81         tbind.addr.len = sizeof(serv);
82         tbind.addr.buf = &serv;
83         tbind.qlen = qlen;

84         tbindret.addr.maxlen = 0;
85         tbindret.addr.len = 0;

86         T_bind(listenfd, &tbind, &tbindret);
87         printf("returned qlen = %d, ", tbindret.qlen);
88         fflush(stdout);

89         Write(cfd, &junk, sizeof(int));     /* tell parent */

90         Read(cfd, &qlen, sizeof(int));  /* just wait for parent */
91         T_close(listenfd);      /* closes all queued connections too */
92     }
93 }
```

debug/qlen.c

Figure E.21 Determine actual number of queued connections for different `qlen` values.

Chapter 33

33.1 We are assuming here that the default for the protocol is an orderly release when the stream is closed, which is true for TCP.

Bibliography

All RFCs are available at no charge through electronic mail, anonymous FTP, or the World Wide Web. A starting point is `http://www.ietf.org`. The directory `ftp://ftp.isi.edu/in-notes` is one location for RFCs. URLs are not specified for the RFCs.

Items marked "Internet Draft" are works in progress of the IETF (Internet Engineering Task Force). These drafts expire six months after publication. The appropriate version of the draft may change after this book is published, or the draft may be published as an RFC. They are available at no charge across the Internet, similar to the RFCs. `http://www.ietf.org` is a major repository for Internet Drafts. The filename portion of the URL for each Internet Draft is included, since the filename contains the version number.

Whenever an electronic copy was found of a paper or report referenced in this bibliography, its URL is included. Be aware that these URLs can change over time, and readers are encouraged to check the Errata for this text on the author's home page for any changes: `http://www.kohala.com/~rstevens`.

Albitz, P., and Liu, C. 1997. *DNS and BIND, Second Edition.* O'Reilly & Associates, Sebastopol, Calif.

Almquist, P. 1992. "Type of Service in the Internet Protocol Suite," RFC 1349, 28 pages (July).
How to use the type-of-service field in the IPv4 header.

Atkinson, R. J. 1995a. "IP Authentication Header," RFC 1826, 13 pages (Aug.).

Atkinson, R. J. 1995b. "IP Encapsulating Security Payload (ESP)," RFC 1827, 12 pages (Aug.).

Baker, F., ed. 1995. "Requirements for IP Version 4 Routers," RFC 1812, 175 pages (June).

Borman, D. A. 1997a. "Re: Frequency of RST Terminated Connections," January 30, 1997, end2end-interest mailing list.

http://www.kohala.com/~rstevens/borman.97jan30.txt

Borman, D. A. 1997b. "TCP and UDP over IPv6 Jumbograms," RFC 2147, 3 pages (May).

Borman, D. A. 1997c. "Re: SYN/RST cookies," June 6, 1997, tcp-impl mailing list.

http://www.kohala.com/~rstevens/borman.97jun06.txt

Braden, R. T., ed. 1989. "Requirements for Internet Hosts—Communication Layers," RFC 1122, 116 pages (Oct.).

The first half of the Host Requirements RFC. This half covers the link layer, IPv4, ICMPv4, IGMPv4, ARP, TCP, and UDP.

Braden, R. T. 1992a. "TIME-WAIT Assassination Hazards in TCP," RFC 1337, 11 pages (May).

Braden, R. T. 1992b. "Extending TCP for Transactions—Concepts," RFC 1379, 38 pages (Nov.).

Braden, R. T. 1993. "TCP Extensions for High Performance: An Update," Internet Draft, 10 pages (June).

http://www.kohala.com/~rstevens/tcplw-extensions.txt
This is an update to RFC 1323 [Jacobson, Braden, and Borman 1992] that never got published as an RFC, but an update to RFC 1323 should appear someday.

Braden, R. T. 1994. "T/TCP—TCP Extensions for Transactions, Functional Specification," RFC 1644, 38 pages (July).

Braden, R. T., Borman, D. A., and Partridge, C. 1988. "Computing the Internet Checksum," RFC 1071, 24 pages (Sept.).

Bradner, S. 1996. "The Internet Standards Process—Revision 3," RFC 2026, 36 pages (Oct.).

Butenhof, D. R. 1997. *Programming with POSIX Threads.* Addison-Wesley, Reading, Mass.

CERT. 1996a. "UDP Port Denial-of-Service Attack," Advisory CA-96.01, Computer Emergency Response Team, Pittsburgh, Pa. (Feb.).

ftp://info.cert.org/pub/cert_advisories/CA-96.01.UDP_service_denial

CERT. 1996b. "TCP SYN Flooding and IP Spoofing Attacks," Advisory CA-96.21, Computer Emergency Response Team, Pittsburgh, Pa. (Sept.).

ftp://info.cert.org/pub/cert_advisories/CA-96.21.tcp_syn_flooding

Cheswick, W. R., and Bellovin, S. M. 1994. *Firewalls and Internet Security: Repelling the Wily Hacker.* Addison-Wesley, Reading, Mass.

Comer, D. E., and Lin, J. C. 1994. "TCP Buffering and Performance Over an ATM Network," Purdue Technical Report CSD-TR 94-026, Purdue University, West Lafayette, Ind. (Mar.).

ftp://gwen.cs.purdue.edu/pub/lin/TCP.atm.ps.Z

Conta, A., and Deering, S. E. 1995. "Internet Control Message Protocol (ICMPv6) for the Internet Protocol Version 6 (IPv6) Specification," RFC 1885, 20 pages (Dec.).

Crawford, M. 1996a. "A Method for the Transmission of IPv6 Packets over Ethernet Networks," RFC 1972, 4 pages (Aug.).

Crawford, M. 1996b. "A Method for the Transmission of IPv6 Packets over FDDI Networks," RFC 2019, 6 pages (Oct.).

Deering, S. E. 1989. "Host Extensions for IP Multicasting," RFC 1112, 17 pages (Aug.).

Deering, S. E., and Hinden, R. 1995. "Internet Protocol, Version 6 (IPv6) Specification," RFC 1883, 37 pages (Dec.).

Dewar, R. B. K., and Smosna, M. 1990. *Microprocessors: A Programmer's View.* McGraw-Hill, New York.

Eriksson, H. 1994. "MBONE: The Multicast Backbone," *Communications of the ACM*, vol. 37, no. 8, pp. 54–60 (Aug.).

Fenner, W. C. 1997. Private Communication.

Fuller, V., Li, T., Yu, J. Y., and Varadhan, K. 1993. "Classless Inter-Domain Routing (CIDR): An Address Assignment and Aggregation Strategy," RFC 1519, 24 pages (Sept.).

Garfinkel, S. L., and Spafford, E. H. 1996. *Practical UNIX and Internet Security, Second Edition.* O'Reilly & Associates, Sebastopol, Calif.

Gierth, A. 1996. Private Communication.

Gilligan, R. E., Thomson, S., Bound, J., and Stevens, W. R. 1997. "Basic Socket Interface Extensions for IPv6," RFC 2133, 32 pages (Apr.).

Handley, M. 1996. "SAP: Session Announcement Protocol," Internet Draft, 14 pages (Nov.).
`draft-ietf-mmusic-sap-00.txt`

Handley, M., and Jacobson, V. 1997. "SDP: Session Description Protocol," Internet Draft (Mar.).
`draft-ietf-mmusic-sdp-03.txt`

Hinden, R., and Deering, S. E. 1995. "IP Version 6 Addressing Architecture," RFC 1884, 18 pages (Dec.).

Hinden, R., and Deering, S. E. 1997. "IP Version 6 Addressing Architecture," Internet Draft, 25 pages (July).
`draft-ietf-ipngwg-addr-arch-v2-02.txt`
This document should replace RFC 1884 [Hinden and Deering 1995] when it becomes an RFC.

Hinden, R., Fink, R., and Postel, J. B. 1997. "IPv6 Testing Address Allocation," Internet Draft, 4 pages (July).
`draft-ietf-ipngwg-testv2-addralloc-01.txt`
This document should replace RFC 1897 [Hinden and Postel 1996] when it becomes an RFC.

Hinden, R., O'Dell, M., and Deering, S. E. 1997. "An IPv6 Aggregatable Global Unicast Address Format," Internet Draft, 9 pages (July).
`draft-ietf-ipngwg-unicast-aggr-02.txt`
This document should replace RFC 2073 [Rekhter et al. 1997] when it becomes an RFC.

Hinden, R., and Postel, J. B. 1996. "IPv6 Testing Address Allocation," RFC 1897, 4 pages (Jan.).

IEEE. 1996. "Information Technology—Portable Operating System Interface (POSIX)—Part 1: System Application Program Interface (API) [C Language]," IEEE Std 1003.1, 1996 Edition, Institute of Electrical and Electronics Engineers, Piscataway, N. J. (July).

This version of Posix.1 contains the 1990 base API, the 1003.1b realtime extensions (1993), the 1003.1c pthreads (1995), and the 1003-1i technical corrections (1995). This is also International Standard ISO/IEC 9945-1: 1996 (E). Ordering information on IEEE standards and draft standards is available at `http://www.ieee.org`.

IEEE. 1997a. "Information Technology—Portable Operating System Interface (POSIX)—Part xx: Protocol Independent Interfaces (PII)," P1003.1g/D6.6, Institute of Electrical and Electronics Engineers, Piscataway, N. J. (Mar.).

This should be the final draft of Posix.1g. Unfortunately, the IEEE does not make these drafts available online.

IEEE. 1997b. *Guidelines for 64-bit Global Identifier (EUI-64) Registration Authority.* Institute of Electrical and Electronics Engineers, Piscataway, N. J.

`http://standards.ieee.org/db/oui/tutorials/EUI64.html`

Jacobson, V. 1988. "Congestion Avoidance and Control," *Computer Communication Review*, vol. 18, no. 4, pp. 314–329 (Aug.).

`ftp://ftp.ee.lbl.gov/papers/congavoid.ps.Z`
A classic paper describing the slow start and congestion avoidance algorithms for TCP.

Jacobson, V. 1994. "Re: half baked anycastoff idea...," June 27, 1994, end2end-interest mailing list.

`http://www.kohala.com/~rstevens/vanj.94jun27.txt`

Jacobson, V., Braden, R. T., and Borman, D. A. 1992. "TCP Extensions for High Performance," RFC 1323, 37 pages (May).

Describes the window scale option, the timestamp option, and the PAWS algorithm, along with the reasons these modifications are needed. [Braden 1993] updates this RFC.

Jacobson, V., Braden, R. T., and Zhang, L. 1990. "TCP Extensions for High-Speed Paths," RFC 1185, 21 pages (Oct.).

Josey, A., ed. 1997. *Go Solo 2: The Authorized Guide to Version 2 of the Single UNIX Specification.* Prentice Hall, Upper Saddle River, N.J.

Joy, W. N. 1994. Private Communication.

Karn, P., and Partridge, C. 1987. "Improving Round-Trip Time Estimates in Reliable Transport Protocols," *Computer Communication Review*, vol. 17, no. 5, pp. 2–7 (Aug.).

`ftp://sics.se/users/craig/karn-partridge.ps`

Katz, D. 1993. "Transmission of IP and ARP over FDDI Network," RFC 1390, 11 pages (Jan.).

Katz, D. 1997. "IP Router Alert Option," RFC 2113, 4 pages (Feb.).

Katz, D., Atkinson, R. J., Partridge, C., and Jackson, A. 1997. "IPv6 Router Alert Option," Internet Draft, 5 pages (June).

`draft-ietf-ipngwg-ipv6-router-alert-02.txt`

Kent, S. T. 1991. "U.S. Department of Defense Security Options for the Internet Protocol," RFC 1108, 17 pages (Nov.).

Kent, S. T., and Atkinson, R. J. 1997a. "IP Authentication Header," Internet Draft, 22 pages (July).
draft-ietf-ipsec-auth-header-01.txt

Kent, S. T., and Atkinson, R. J. 1997b. "IP Encapsulating Security Payload (ESP)," Internet Draft, 19 pages (July).
draft-ietf-ipsec-esp-v2-00.txt

Kernighan, B. W., and Pike, R. 1984. *The UNIX Programming Environment*. Prentice Hall, Englewood Cliffs, N.J.

Kernighan, B. W., and Ritchie, D. M. 1988. *The C Programming Language, Second Edition*. Prentice Hall, Englewood Cliffs, N.J.

Korn, D. G., and Vo, K. P. 1991. "SFIO: Safe/Fast String/File IO," *Proceedings of the 1991 Summer USENIX Conference*, pp. 235–255, Nashville, Tenn.
A description of an alternative to the standard I/O library. The source code is available from http://www.research.att.com/sw/tools/reuse.

Lanciani, D. 1996. "Re: sockets: AF_INET vs. PF_INET," Message-ID: <3561@news.IPSWITCH.COM>, Usenet, comp.protocols.tcp-ip Newsgroup (Apr.).
http://www.kohala.com/~rstevens/lanciani.96apr10.txt

Maslen, T. M. 1997. "Re: gethostbyXXXX() and Threads," Message-ID <maslen.862463530@shellx>, Usenet, comp.programming.threads Newsgroup (May).
http://www.kohala.com/~rstevens/maslen.97may01.txt

Maufer, T., and Semeria, C. 1997. "Introduction to IP Multicast Routing," Internet Draft (Mar.).
draft-ietf-mboned-intro-multicast-02.txt

McCann, J., Deering, S. E., and Mogul, J. C. 1996. "Path MTU Discovery for IP version 6," RFC 1981, 15 pages (Aug.).

McCanne, S., and Jacobson, V. 1993. "The BSD Packet Filter: A New Architecture for User-Level Packet Capture," *Proceedings of the 1993 Winter USENIX Conference*, pp. 259–269, San Diego, Calif.
ftp://ftp.ee.lbl.gov/papers/bpf-usenix93.ps.Z

McDonald, D. L. 1997. "A Simple IP Security API Extension to BSD Sockets," Internet Draft, 12 pages (Mar.).
draft-mcdonald-simple-ipsec-api-01.txt

McDonald, D. L., Phan, B. G., and Atkinson, R. J. 1996. "A Socket-Based Key Management API (and surrounding infrastructure)," *Proceedings of the INET'96 Conference*, pp. 53–63 (June), Montreal, Quebec.
http://www.cs.hut.fi/ssh/crypto/pf-key.ps

McDonald, D. L., Metz, C. W., and Phan, B. G. 1997. "PF_KEY Key Management API, Version 2," Internet Draft, 67 pages (July).
draft-mcdonald-pf-key-v2-03.txt

McKusick, M. K., Bostic, K., Karels, M. J., and Quarterman, J. S. 1996. *The Design and Implementation of the 4.4BSD Operating System*. Addison-Wesley, Reading, Mass.

Meyer, D. 1997. "Administratively Scoped IP Multicast," Internet Draft, 7 pages (June).

`draft-ietf-mboned-admin-ip-space-03.txt`

Mills, D. L. 1992. "Network Time Protocol (Version 3): Specification, Implementation, and Analysis," RFC 1305, 113 pages (Mar.).

Mills, D. L. 1996. "Simple Network Time Protocol (SNTP) Version 4 for IPv4, IPv6 and OSI," RFC 2030, 18 pages (Oct.).

Mogul, J. C., and Deering, S. E. 1990. "Path MTU Discovery," RFC 1191, 19 pages (Apr.).

Mogul, J. C., and Postel, J. B. 1985. "Internet Standard Subnetting Procedure," RFC 950, 18 pages (Aug.).

Nemeth, E. 1997. Private Communication.

Open Group, The. 1997. *CAE Specification, Networking Services (XNS), Issue 5*. The Open Group, Reading, Berkshire, U.K.

This is the specification for sockets and XTI in Unix 98. This manual also has appendices describing the use of XTI with NetBIOS, the OSI protocols, SNA, and the Netware IPX and SPX protocols. Three appendices cover the use of both sockets and XTI with ATM.

Partridge, C., Mendez, T., and Milliken, W. 1993. "Host Anycasting Service," RFC 1546, 9 pages (Nov.).

Partridge, C., and Pink, S. 1993. "A Faster UDP," *IEEE/ACM Transactions on Networking*, vol. 1, no. 4, pp. 429–440 (Aug.).

Paxson, V. 1996. "End-to-End Routing Behavior in the Internet," *Computer Communication Review*, vol. 26, no. 4, pp. 25–38 (Oct.).

`ftp://ftp.ee.lbl.gov/papers/routing.SIGCOMM.ps.Z`

Piscitello, D. M. 1994. "FTP Operation Over Big Address Records (FOOBAR)," RFC 1639, 5 pages (June).

Plauger, P. J. 1992. *The Standard C Library*. Prentice Hall, Englewood Cliffs, N.J.

Postel, J. B. 1980. "User Datagram Protocol," RFC 768, 3 pages (Aug.).

Postel, J. B., ed. 1981a. "Internet Protocol," RFC 791, 45 pages (Sept.).

Postel, J. B. 1981b. "Internet Control Message Protocol," RFC 792, 21 pages (Sept.).

Postel, J. B., ed. 1981c. "Transmission Control Protocol," RFC 793, 85 pages (Sept.).

Pusateri, T. 1993. "IP Multicast Over Token-Ring Local Area Networks," RFC 1469, 4 pages (June).

Rago, S. A. 1993. *UNIX System V Network Programming*. Addison-Wesley, Reading, Mass.

Rekhter, Y., Lothberg, P., Hinden, R., Deering, S. E., and Postel, J. B. 1997. "An IPv6 Provider-Based Unicast Address Format," RFC 2073, 7 pages (Jan.).

Reynolds, J. K., and Postel, J. B. 1994. "Assigned Numbers," RFC 1700, 230 pages (Oct.).

Changes to some of the information contained in this RFC can occur before the RFC is updated. All the tables of information in the RFC are taken from files in the directory `ftp://ftp.isi.edu/in-notes/iana/assignments`, and these files are updated as changes occur. The RFC contains the URLs for the files in this directory, and these files should be consulted for the latest information.

Ritchie, D. M. 1984. "A Stream Input-Output System," *AT&T Bell Laboratories Technical Journal*, vol. 63, no. 8, pp. 1897–1910 (Oct.).

Salus, P. H. 1994. *A Quarter Century of Unix*. Addison-Wesley, Reading, Mass.

Salus, P. H. 1995. *Casting the Net: From ARPANET to Internet and Beyond*. Addison-Wesley, Reading, Mass.

Schimmel, C. 1994. *UNIX Systems for Modern Architectures: Symmetric Multiprocessing and Caching for Kernel Programmers*. Addison-Wesley, Reading, Mass.

Srinivasan, R. 1995. "XDR: External Data Representation Standard," RFC 1832, 24 pages (Aug.).

Stevens, W. R. 1992. *Advanced Programming in the UNIX Environment*. Addison-Wesley, Reading, Mass.

All the details of Unix programming. Referred to through this text as APUE.

Stevens, W. R. 1994. *TCP/IP Illustrated, Volume 1: The Protocols*. Addison-Wesley, Reading, Mass.

A complete introduction to the Internet protocols. Referred to through this text as TCPv1.

Stevens, W. R. 1996. *TCP/IP Illustrated, Volume 3: TCP for Transactions, HTTP, NNTP, and the UNIX Domain Protocols*. Addison-Wesley, Reading, Mass.

Referred to through this text as TCPv3.

Stevens, W. R., and Thomas, M. 1997. "Advanced Sockets API for IPv6," Internet Draft (July).

`draft-stevens-advanced-api-04.txt`

Tanenbaum, A. S. 1987. *Operating Systems Design and Implementation*. Prentice Hall, Englewood Cliffs, N.J.

Thomas, S. 1997. "Transmission of IPv6 Packets over Token Ring Networks," Internet Draft, 10 pages (June).

`draft-ietf-ipngwg-trans-tokenring-00.txt`

Thomson, S., and Huitema, C. 1995. "DNS Extensions to Support IP version 6," RFC 1886, 5 pages (Dec.).

Torek, C. 1994. "Re: Delay in re-using TCP/IP port," Message-ID <199501010028.QAA16863 @elf.bsdi.com>, Usenet, comp.unix.wizards Newsgroup (Dec.).

`http://www.kohala.com/~rstevens/torek.94dec31.txt`

Unix International. 1991. "Data Link Provider Interface Specification," Revision 2.0.0, Unix International, Parsippany, N. J. (Aug.).

`http://www.kohala.com/~rstevens/dlpi.2.0.0.ps`
A newer version of this specification is available online from The Open Group at `http://www.rdg.opengroup.org/pubs/catalog/web.htm`.

Unix International. 1992a. "Network Provider Interface Specification," Revision 2.0.0, Unix International, Parsippany, N. J. (Aug.).

`http://www.kohala.com/~rstevens/npi.2.0.0.ps`

Unix International. 1992b. "Transport Provider Interface Specification," Revision 1.5, Unix International, Parsippany, N. J. (Dec.).

`http://www.kohala.com/~rstevens/tpi.1.5.ps`
A newer version of this specification is available online from The Open Group at `http://www.rdg.opengroup.org/pubs/catalog/web.htm`.

Vixie, P. A. 1996. Private Communication.

Wright, G. R., and Stevens, W. R. 1995. *TCP/IP Illustrated, Volume 2: The Implementation*. Addison-Wesley, Reading, Mass.

The implementation of the Internet protocols in the 4.4BSD-Lite operating system. Referred to through this text as TCPv2.

Index

Networking is a field that is pockmarked with acronyms. Rather than provide a separate glossary (with most of the entries being acronyms), this index also serves as a glossary for all the acronyms used in the book. The primary entry for the acronym appears under the acronym name. For example, all references to the Internet Control Message Protocol appear under ICMP. The entry under the compound term "Internet Control Message Protocol" refers back to the main entry under ICMP.

The notation "definition of" appearing with a C function refers to the boxed function prototype for that function, its primary description. The "definition of" notation for a structure refers to its primary definition. Some functions also contain the notation "source code" if the source code implementation for that function appears in the text.

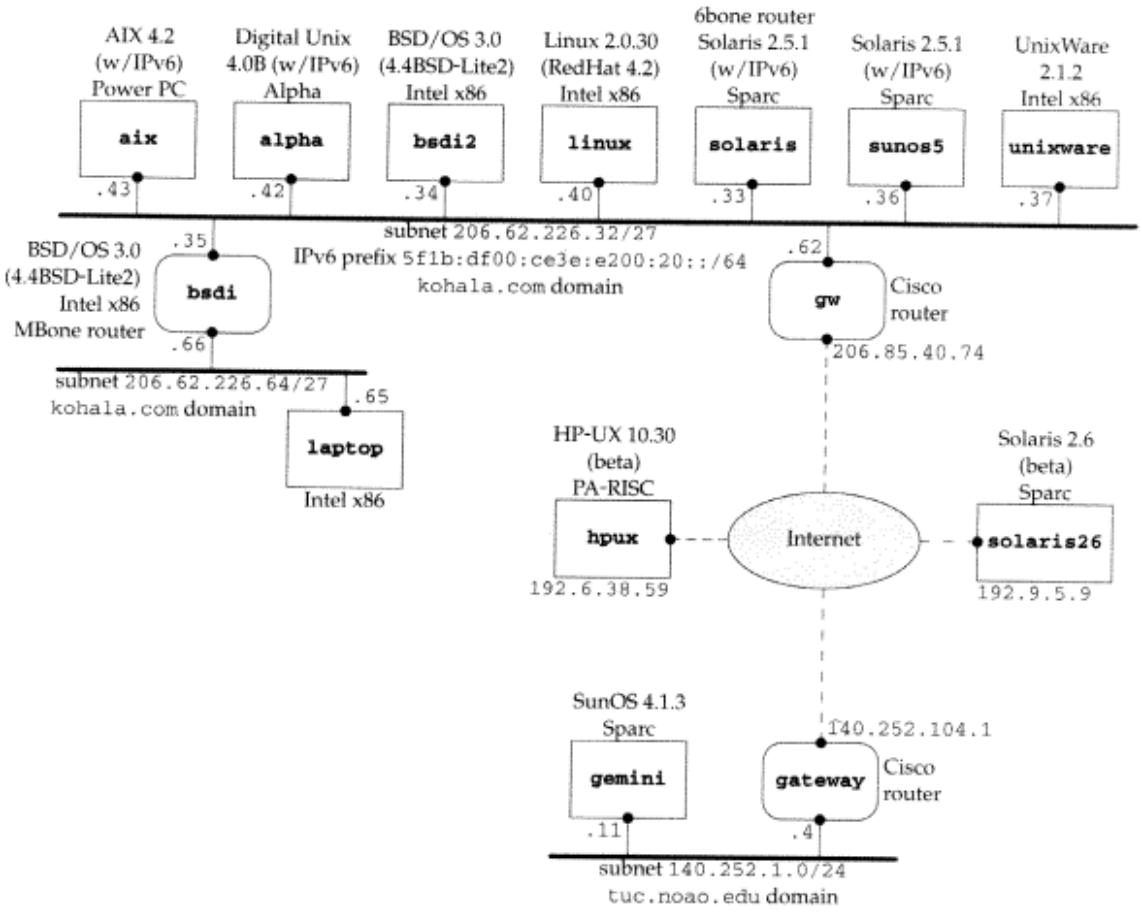
Function and Macro Definitions

(**Bold** page numbers indicate source code implementation)

Function and Macro Definitions

(**Bold** page numbers indicate source code implementation)

Structure Definitions



Hosts and networks used for most examples in the text.