Linux Networking Basics

A Short Guide for a Short Course

Michael Marsh

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Preface

This book is intended as a companion for the University of Maryland course CMSC389Z, which is a three-week Winter Term course on Network Management and Programming in a Linux Environment. While it may be helpful outside of that context as a general introduction to some computer networking concepts, it is not meant to be anywhere close to complete.

Chapter 1

Introduction

This book was written as the text for a 3-week course *Network Management and Programming in a Linux Environment* at the University of Maryland. The material is very focused on the topics for this course, so there will necessarily be a lot that is omitted, both in breadth and depth.

1.1 Terminology and References

Before we get into any details, let's begin by defining some terms:

- **host** – This is a computer, phone, or any other device connected to a network. We use this as a general term for anything on the network.
- **network interface** – This is something on a host that provides it access to a network. Every host will have *at least* one of these.
- **device** – This is another name for a *network interface*. Often used because it is shorter.

- **traffic** – This refers to data sent over a network. It will consist of many *packets* implementing various *protocols*.
- **packet** – This is a discrete unit of traffic. The network operates by moving individual packets from their sources to their destinations.
- **protocol** – This is a set of messages and rules that hosts use to meaningfully communicate. Protocols tell hosts how to "speak" to other hosts in a way that they will understand, and how to interpret those messages.
- **network edge** – This is the set of hosts that exchange data. Your phone and laptop, as well as the web servers you connect to, are all part of the network edge. There is some network infrastructure that is also considered part of the edge, such as your Wifi router.
- **network core** – This is the set of hosts that provide *service* for the network edge. These are owned and managed by large Internet Service Providers, and consist of routers (which we'll see in Section 3.6), switches (which we won't discuss, but bear some similarity to routers), and some other miscellaneous devices.

We'll introduce other terms as needed, but these are the fundamental building blocks of any discussion of networking. We're going to focus on hosts at the edge of the network, rather than the core.

You may find the following references helpful, both of which are available for free from https://www.cs.umd.edu/∼mmarsh/books.html:

- *Using the Bash Command Line* – including common command-line utilities and scripting
- *A General Systems Handbook* – including numeric representations, git, Linux system administration, and networking commands

If you plan to do more network programming in C, the following references might be worth the investment:

- *TCP/IP Sockets in C*, by Michael J. Donahoo and Kenneth L. Calvert
- *TCP Illustrated: The Protocols, Volume 1, 2nd Edition*, by Kevin R. Fall and W. Richard Stevens
- *UNIX Network Programming: Networking APIs: Sockets and XTI, Volume 1*, by W. Richard Stevens

1.2 Some Linux Basics

You should familiarize yourself with *Using the Bash Command Line*, especially Chapters 1 through 3, 5, 7, and 9. You will likely want to refer back to these chapters frequently until you've internalized most of the information. Chapters 2, 3, and 5 of *A General Systems Handbook* will also be very useful, though we will expand on the material in Chapter 5 (Network Commands) in this book.

Another important thing to note when using Linux is how files encode line endings. There are two common ASCII characters used to indicate the end of a line: *carriage return* (`\r`) and *newline* (`\n`). Newline is also sometimes called *line feed*. You will see these abbreviated to `CR` in `LF` in much documentation. With two different characters (representing different functions on a typewriter), what do you use, and where?

This is where things get complicated. On Linux, line endings use newline (`LF`). Macs also use newline *now*, but used to use carriage return (`CR`). Windows uses a carriage return followed by a newline (`CRLF`). The `CRLF` encoding is also used by ASCII-based network protocols, like HTTP.

Why do we care about this, when this is a Linux-focused course? Because many students use Windows computers, their file editing software will produce files with `CRLF` line endings. Some programs, like the C compiler, handle these different encodings without problems. Others, like the bash shell, insist on `LF` line endings. There is a helpful utility, called `dos2unix`, that you should install in order to convert files. If you are writing a bash script, make sure you use `dos2unix` to ensure that it has the correct line endings.

1.3 Course Structure

For the three-week course, the weeks will cover:

Week 1 Chapter 2: Network Overview, Chapter 3: Connecting Hosts, and Chapter 4: Interactions Between Layers

Week 2 Chapter 5: Network Programming

Week 3 Chapter 6: Traffic

Chapter 2

Network Overview

In this chapter, we will explore the design of the network and the tools used to configure and examine it. We will begin with the high-level network model, then how pairs of hosts are connected, and finally how these are combined into a large distributed system.

2.1 The OSI Model

To understand the design of the network, it is important to familiarize ourselves with the *Open Systems Interconnection* (OSI) Model. In this model, we create multiple *layers*, each of which is reponsible for a particular type of *service*. These layers provide Application Programming Interfaces (APIs) to the layers above and below them.

Within a layer, we define and implement various *protocols* to handle that layer's responsibilities. These protocols must adhere to established APIs, so that one protocol at a layer can be replaced with another without impacting the rest of the layers.

By organizing the protocols in this manner, we gain an *abstraction* of the network. We like abstractions, because they allow us to reason about individual components of a complex system, as well as replace pieces that no longer suit the needs at that layer.

The OSI Model divides the *network stack* into seven layers:

1. Physical

   This is how hosts and other devices are actually connected. Physical layer devices include ethernet cables, RF antennas, fiber optic cables, etc. It also includes how bits (or groups of bits) are *encoded* on those connections (voltages, durations, etc.).

2. Datalink (or just Link)

   This defines how devices group collections of bits into messages, called *frames*, each with a specified source and destination. Ethernet and WiFi are examples of Link layer protocols. A network at this layer is generally referred to as a *subnet* or *local network*.

3. Internetworking (or just Network)

   This defines how local networks are connected in a way that allows for world-wide communications. A single Network layer message, called a *packet*, might be comprised of one or multiple Link layer frames, but a single frame may only contain a single packet's data. Most of the time when we talk about the Network layer, we are talking about version 4 of the Internet Protocol. When looking at a particular portion of the network, we often refer to subsets of it as subnets, even if we are still referring to the Network layer.

4. Transport

   The Network layer only defines host-to-host connections. The Transport layer adds multiplexing through at addition of *ports*, to support process-to-process connections. That is the only thing that the User Datagram

Protocol (UDP) does, but the Transmission Control Protocol (TCP) adds reliability and in-order packet delivery as well.

5. Session

   Some protocols require using multiple Transport layer connections, either on different ports of a single host or to different hosts. When this is required, the Session layer coordinates these.

6. Presentation

   This layer handles data encoding and decoding, so that the data as sent "on the wire" is readable by the application, and vice-versa.

7. Application

   This is where users interact with the network, through an interface such as a web browser. There are many Application layer programs, both graphical and command-line-based.

The Session and Presentation layers are often omitted from networking discussions, because they are used either less frequently or so transparently that they appear to be missing or part of another layer. We will limit our discussion of these two layers by noting (this will make more sense later) that the Session layer often appears in applications such as Voice-over-Internet-Protocol (VoIP) which require coordination between multiple Transport-layer protocols, and the Presentation layer is often captured in functions such as **`htonl`**.

We typically characterize connections between hosts with several quantities:

- **Bandwidth** This refers to the speed at which data can be written to and read from the connection. It is typically given in terms of *bits per second* (**`bps`**), with powers-of-10 prefixes such as *kbps* (10^3bps), *Mbps* (10^6bps), and *Gbps* (10^9bps). Note that the typical *byte* is an octet (8 bits), so you need to divide by 8 to obtain *bytes* per second, and bytes are usually expressed in powers-of-2 prefixes.

- **Latency** This refers to the time it takes for a single bit written at the source to reach the destination. It is often given in milliseconds, but can be considerably longer.

- **Bit-Error Rate** All physical media are imperfect, and this refers to the average fraction (possibly, but not always, given as a percentage) of bits transmitted that are either flipped or undiscernable at the destination. It is often abbreviated *BER*. This does *not* include losses of data due to overwhelming the network with more data than it can handle.

- **Throughput** This refers to how quickly we can move data from its source to its destination. The most we could ever achieve is the connection bandwidth, but this is rarely attainable. It is often helpful to consider throughput for a specific layer, in which case the overhead of the lower layers reduces the throughput relative to the bandwidth by predictable amounts.

- **Goodput** This refers to practical application-layer throughput (the *good throughput*), which is what applications (and users) see.

2.2 Internet Protocol

The Internet Protocol (IP) has two major versions: IPv4 and IPv6. We will focus on IPv4, but where appropriate we will provide analogous IPv6 information. IP is a layer 3 (Internetworking) protocol, as you might guess from its name.

The purpose of IP is to *forward* data across a global network. This requires unique *addresses* for all hosts on the network (though we will see how this requirement can be loosened).

At layer 3, data is divided into *packets*. A packet has a *header* and a *payload*. The payload is the actual data we are interested in delivering, while the header provides sufficient information to move the packet through the network.

IP performs two crucial functions: *routing* and *forwarding*. Routing involves *distributed* protocols that allows participants to create *routing tables* (also called *forwarding tables*). Forwarding is the method by which an IP participant determines what to do with a given packet: Is this the destination? Should the packet be forwarded, and to whom? Should the packet be discarded (*dropped*)?

Hosts that perform routing and forwarding are called *routers* (because they route traffic). These are only required to implement OSI layers 1–3, though many provide extra functionality for remote administration or run on general-purpose computers. *End-hosts* (such as your laptop, phone, smart thermostat, or even some lightbulbs) can only be the source or destination for a packet — they do not normally provide forwarding.

An example of network connection is shown in Figure 2.1. Here we have three networks: a typical home network on the left, an Internet Service Provider (ISP) network in the middle, and a corporate network on the right. Solid lines are layer 1 and layer 2 connections; there might be other layer 1 devices, like *repeaters*,[1] that are not shown. Laptop A is connected via WiFi to a WiFi router B. From the home network's perspective, this router is the *gateway* router (GW). This router is connected via Ethernet to an ISP edge router C. C is connected to switch D, which is in turn connected to another switch E. E is then connected to edge router F, which connects to the corporate gateway router G. In the corporate network, G is connected to a switch H, which connects to the server I in the data center. Dashed lines show the layer 3 connections, which we refer to as an *overlay* on top of layer 2; this only connects the end hosts and the routers. The dotted line is the layer 7 overlay on top of layer 3, where the only hosts are the laptop and server. The ISP will likely refer to B and G as *customer edge* (CE) routers, and C and F as *provider edge* (PE) routers.

[1]A repeater has two ports, and boosts signals from one to the other to overcome attenuation. This is like two people in different rooms, who cannot hear one another, communicating through someone in the hallway who repeats what each is saying for the other.

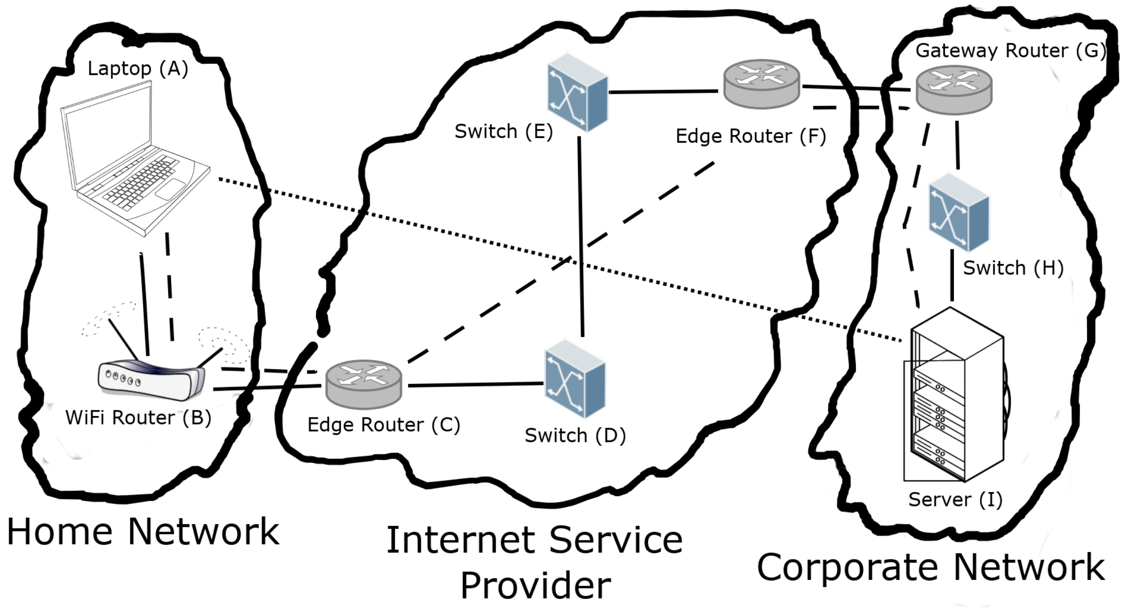


Figure 2.1: Network Diagram, including Layer 3 and 7 Overlays

2.3 Naming

Internet Protocol requires hosts to have unique *names* (with some caveats). These names are 32-bit integers, or four bytes (usually referred to as *octets* in networking, because there are 8 bits in each). We generally write these in *dotted-decimal* notation, so while it is perfectly valid to refer to a host's address as `2148040452`, we would generally write it as `128.8.127.4`.

How do we get the dotted-decimal form? If we write `2148040452` in hexadecimal, it is `0x80087f04`. An octet is two hexadecimal characters, so breaking this down gives us `80 08 7f 04`. If we then write these hexadecimal values in decimal, with a dot between octets, we get `128.8.127.4`.

An IP network is divided into subnets that are managed by different authorities.

We specify these subnets using *Classless Inter-Domain Routing* (CIDR) notation. A *CIDR block* comprises as *base address* and a *prefix length*, separated by a slash. The prefix length tells you how many *bits* in the base address specify the subnet. That is, any address with the same first prefix-length bits as the base address is part of the same network.

To make this clearer, let's take a look at the subnet **`128.8.0.0/16`** (this is University of Maryland's network). Here, the **`/16`** (read "slash-16") tells us that the first two octets of **`128.8.0.0`** specify the subnet. In this case, that's the **`128.8`** part, so any IP address beginning with **`128.8`** is part of this network.

Subnets can be subdivided further, and an individual host is sometimes treated as a **`/32`** CIDR block. Each subdivision of a network typically denotes hosts that are physically closer together, though there are cases where this is not necessarily true (such as with Virtual Private Networks). Forwarding proceeds by moving a packet to the CIDR block containing the destination address with the longest-possible prefix length. The forwarding table tells a router which outgoing connections can receive which CIDR blocks.

Chapter 3

Connecting Hosts

3.1 Transport Layer and Multiplexing

The simplest model for an end host is a computer with a single network connection. This might be an ethernet cable, a WiFi radio, or something else. If only one process on a host could access the network at a single time, we would not have much of the functionality that we currently rely on. This was, in fact, the case for many home computer users who used to use a telephone modem to dial into a bulletin board system (BBS). To get around this limitation, we need some way to *multiplex* connections over a single physical connection.

Layer 4, the Transport Layer, provides us with multiplexing through the use of *ports*. You can think of ports as an additional address within the computer that identifies not just a particular process, but a specific *socket* within a process. We will discuss sockets in more detail in Chapter 5.

For our purposes, we will only consider the two most common Transport Layer protocols for most applications: UDP and TCP. Each of these provides port-based multiplexing. These are not the only Transport Layer protocols, and there are even protocols that do not involve port-based multiplexing.

UDP is the *User Datagram Protocol*. UDP adds a very simple header with 16-bit port numbers on the source and destination side. The destination port enabled the destination host to pass the payload of the UDP packet, the *datagram*, to the appropriate process. While a single datagram *can* (but often does not) contain multiple application-layer messages, a single message *must* be contained within a single datagram. The sender port enables the receiving process to send a reply. Otherwise, UDP adds no additional functionality.

TCP is the *Transmission Control Protocol*. TCP has a much more complex header, which we will not discuss in detail here (see Chapter 6). For the moment, all that we care about is that, like UDP, it contains source and destination ports. In addition, it contains information that allows each end to determine when packets have been dropped, how fast data can be sent, and how to properly order the data received. Along with retransmission of dropped packets, this means TCP provides a *reliable in-order data stream*.

How do we decide which of these to use in an application? If we have reasonably short, self-contained messages, then UDP is often sufficient. The advantage of using UDP for these is that it has a much lower overhead cost than TCP. We can add our own reliability at the application layer, if we need it. If we have more continual data, then TCP is *often* a better choice, since we are guaranteed to have our data streamed to the application layer in the correct order with nothing missing, and taking maximum advantage of the network connection. However, we sometimes want to use UDP for streaming data. This typically occurs in applications where the loss of data is less important than timely delivery. Consider streaming voice: If we lose a packet, we have a very small amount of voice data that is lost, which might not even be noticeable. On the other hand, if we require the lost packet to be detected and retransmitted, we might end up with a very noticeable delay on the receiver's side, which will only grow as additional packets are dropped and retransmitted.

To summarize:

- Simple single-message protocols are often best with UDP.

- Streaming data protocols where reliability is important are often best with TCP.
- Streaming data protocols where latency is important are often best with UDP.

3.2 netcat — Simple Data Exchange

You are hopefully familiar with the **`cat`** command, which reads data from standard input or specified files and *concatenates* it to standard output. There is a network version of this, called *netcat*. While you would not ordinarily use this to, say, browse the web, it can be very helpful when testing a new protocol or implementation.

There are two programs commonly referred to as netcat: **`nc`** and **`ncat`**. Which one you have installed depends on your operating system. **`ncat`** is provided by nmap.org, as source code or binary for most operating systems. The options differ slightly, and even basic usage depends on which version you are using.

Let's begin by using netcat as a simple web browser. Specifically, we are going to use it to do a search using Google for "netcat". Here, the two versions are almost the same. Either of the following will work:

```
nc www.google.com 80
ncat www.google.com 80
```

Both of these connect to the host **`www.google.com`** on port 80 (the standard port for HTTP), and then wait for input. Run one of these, and then type:

```
GET /search?q=netcat
```

and hit Enter or Return. You should see the raw HTML returned by Google. Here we see the first difference: **`nc`** will exit immediately after receiving the response, while **`ncat`** will wait for additional input.

Netcat can operate as a *client* or a *server*. A server is a program that waits for connections requesting its service, and responds to them; a web server is probably the most familiar to people currently. The programs connecting to those servers are clients; think of a web browser. We will start with netcat as a client (as we did above), but first let's look at some general options common to both client and server operation.

Option	**Meaning**	**Notes**
`-4`	Use only IPv4	
`-6`	Use only IPv6	
`-u`	Create UDP connections	The default is TCP
`-v`	Verbose output	

We have seen some basic client operation already. The client behavior can be summarized as:

1. Connect to a server.
2. Send a message.
3. Wait for a response.

The options we provide affect this behavior. We have already seen the **`-u`** option, to make a UDP connection instead of a TCP connection. As a client, you must always supply a hostname and a port (**`www.google.com`** and **`80`**, in our previous example). While there are client-specific options, you are less likely to use them. See the manpage for your particular version of netcat for details.

While it is possible to use netcat as a client generally, it is more useful for debugging servers. Since many protocols are binary, you will often want to store the messages to send in files, and redirect them using the shell's **`<`** operator:

```
nc example.com 1234 < message
```

This is also useful for text-based protocols, to save typing.

Just as running netcat as a client is useful for debugging servers, running netcat as a server can help you debug clients. It is also useful if you just need to capture messages from a client. We will see other ways to capture message in Chapter 6, but if all you need is the application-layer payload, a netcat server could be all that you need.

There are more options for running a server, and this is where the differences between **`nc`** and **`ncat`** are more noticeable.

Option	**Meaning**	**Notes**
`-l`	Listen for connections	Basic server option
`-p <port>`	Listen on port **`<port>`**	Only for **`ncat`**, **`nc`** just takes the port as a regular option
`-k`	Keep the socket open	Don't exit after the initial client disconnects

Here are examples of how you would start a netcat server listening on port 1234:

```
nc -l 1234
ncat -l -p 1234
```

If you run either of these and then run (in a separate terminal)

```
nc 127.0.0.1 1234
```

you can type on one terminal and see it appear on the other. This works in either direction. When you close one side (with **`Ctrl-C`** or **`Ctrl-D`**), both sides will close.

3.3 Standard Ports and Services

There are a lot of common network protocols, such as HTTP, SMTP (email), SSH, and DNS. Whether they use TCP or UDP, these have standardized ports that all applications supporting those protocols should use by default. If you're implementing one of these applications, you need to know these standard ports. Additionally, you might be logging attempted connections to your host, and want to know what possible attackers are attempting to connect to.

On Posix systems, such as Linux or MacOS, these standard ports are enumerated in the file **`/etc/services`**. A typical line might look like

```
http                80/tcp      www www-http # World Wide Web HTTP
```

The first column tells us the protocol, which in this case is HTTP (Hypertext Transfer Protocol). The next column tells us the port (80) and layer 4 protocol (TCP) for this service. The remainder of the line contains common names for the service, with general comments following the **`#`** comment character.

You are not likely to need to refer to this very often, but it is useful to know where to find this information if you need it.

3.4 Finding Active Ports

The typical modern operating system has a lot of services running that receive messages over the network. It also tends to send a lot of messages, both for user-initiated actions as well as background tasks. All of these employ network ports, so it is useful to be able to see what ports are in use.

The standard Linux command to view ports is **`netstat`**. Technically, this shows the *sockets* that are in use, which can include local (non-network) sockets. The simplest usage is

```
root@efcc0d32f10f:/# netstat
Active Internet connections (w/o servers)
Proto Recv-Q Send-Q Local Address     Foreign Address   State
Active UNIX domain sockets (w/o servers)
Proto RefCnt Flags     Type     State       I-Node Path
```

This is running on a docker container with no active connections. It does, however, have services listening:

```
root@efcc0d32f10f:/# netstat -a
Active Internet connections (servers and established)
Proto Recv-Q Send-Q Local Address     Foreign Address   State
tcp        0      0 localhost:mysql   *:*               LISTEN
tcp        0      0 *:http            *:*               LISTEN
Active UNIX domain sockets (servers and established)
Proto RefCnt Flags     Type     State       I-Node Path
unix  2      [ ACC ]   STREAM   LISTENING   53476  /var/run/mysqld/mysqld.sock
```

The **`-a`** flag tells **`netstat`** to list all sockets, whether there are active connections or not. We can see that there are two network sockets running, and one Unix (non-network) socket. The first network socket is listening for connections from localhost only, on the standard TCP port used by MySql. The second network socket is listening for HTTP connections from anywhere over TCP.

If we look in **`/etc/services`**, we can find out what these ports are:

```
root@efcc0d32f10f:/# grep mysql /etc/services
mysql           3306/tcp
mysql           3306/udp
mysql-proxy     6446/tcp                        # MySQL Proxy
mysql-proxy     6446/udp
root@efcc0d32f10f:/# grep http /etc/services
# Updated from http://www.iana.org/assignments/port-numbers and other
# sources like http://www.freebsd.org/cgi/cvsweb.cgi/src/etc/services .
```

```
http            80/tcp          www             # WorldWideWeb HTTP
http            80/udp                          # HyperText Transfer Protocol
https           443/tcp                         # http protocol over TLS/SSL
https           443/udp
http-alt        8080/tcp        webcache        # WWW caching service
http-alt        8080/udp
```

We can identify the relevant lines (exact service match, and TCP) as

```
mysql           3306/tcp
http            80/tcp          www             # WorldWideWeb HTTP
```

This is a bit cumbersome, so we will add the **`-n`** flag, to leave numbers as numbers, and not look them up:

```
root@efcc0d32f10f:/# netstat -an
Active Internet connections (servers and established)
Proto Recv-Q Send-Q Local Address     Foreign Address   State
tcp        0      0 127.0.0.1:3306    0.0.0.0:*         LISTEN
tcp        0      0 0.0.0.0:80        0.0.0.0:*         LISTEN
Active UNIX domain sockets (servers and established)
Proto RefCnt Flags     Type     State        I-Node Path
unix  2      [ ACC ]   STREAM   LISTENING    53476  /var/run/mysqld/mysqld.sock
```

Here we see not only that the ports are given numerically, but so are **`localhost`** (**`127.0.0.1`**) and **`*`** (**`0.0.0.0`**). This will also tend to be faster, especially once we are dealing with remote connections, where we would have to perform a *reverse DNS lookup* to map IP addresses to host names.

We can also limit ourselves to TCP sockets:

```
root@efcc0d32f10f:/# netstat -ant
Active Internet connections (servers and established)
Proto Recv-Q Send-Q Local Address     Foreign Address   State
tcp        0      0 127.0.0.1:3306    0.0.0.0:*         LISTEN
tcp        0      0 0.0.0.0:80        0.0.0.0:*         LISTEN
```

UDP sockets:

```
root@efcc0d32f10f:/# netstat -anu
Active Internet connections (servers and established)
Proto Recv-Q Send-Q Local Address     Foreign Address   State
```

or both:

```
root@efcc0d32f10f:/# netstat -antu
Active Internet connections (servers and established)
Proto Recv-Q Send-Q Local Address     Foreign Address   State
tcp        0      0 127.0.0.1:3306    0.0.0.0:*         LISTEN
tcp        0      0 0.0.0.0:80        0.0.0.0:*         LISTEN
```

We can also request the process name and ID for the sockets:

```
root@efcc0d32f10f:/# netstat -antup
Active Internet connections (servers and established)
Proto Recv-Q Send-Q Local Address     Foreign Address   State   PID/Program name
tcp        0      0 127.0.0.1:3306    0.0.0.0:*         LISTEN  -
tcp        0      0 0.0.0.0:80        0.0.0.0:*         LISTEN  479/apache2
```

We do not always get this information, though.

An important thing to note is that we can list the arguments separately or combined, and the order does not matter:

```
root@efcc0d32f10f:/# netstat -a -n -t -u
Active Internet connections (servers and established)
Proto Recv-Q Send-Q Local Address     Foreign Address   State
tcp        0      0 127.0.0.1:3306    0.0.0.0:*         LISTEN
tcp        0      0 0.0.0.0:80        0.0.0.0:*         LISTEN

root@efcc0d32f10f:/# netstat -aunt
Active Internet connections (servers and established)
Proto Recv-Q Send-Q Local Address     Foreign Address   State
tcp        0      0 127.0.0.1:3306    0.0.0.0:*         LISTEN
tcp        0      0 0.0.0.0:80            0.0.0.0:*             LISTEN

root@efcc0d32f10f:/# netstat -tuna
Active Internet connections (servers and established)
Proto Recv-Q Send-Q Local Address         Foreign Address       State
tcp        0      0 127.0.0.1:3306        0.0.0.0:*             LISTEN
tcp        0      0 0.0.0.0:80            0.0.0.0:*             LISTEN

root@efcc0d32f10f:/# netstat -taun
Active Internet connections (servers and established)
Proto Recv-Q Send-Q Local Address         Foreign Address       State
tcp        0      0 127.0.0.1:3306        0.0.0.0:*             LISTEN
tcp        0      0 0.0.0.0:80            0.0.0.0:*             LISTEN
```

It is also worth noting that some versions of **`netstat`** behave slightly differently. On MacOS, for example, the **`-u`** flag specifies *Unix* sockets, not *UDP*.

We can also see what an active connection looks like:

```
root@efcc0d32f10f:/# nc localhost 80
root@efcc0d32f10f:/# netstat -taun
Active Internet connections (servers and established)
Proto Recv-Q Send-Q Local Address         Foreign Address       State
tcp        0      0 127.0.0.1:3306        0.0.0.0:*             LISTEN
tcp        0      0 0.0.0.0:80            0.0.0.0:*             LISTEN
tcp        0      0 127.0.0.1:80          127.0.0.1:35226       SYN_RECV
tcp        0      0 127.0.0.1:35226       127.0.0.1:80          ESTABLISHED
```

The first command is run in a separate shell. We can see that there are two new sockets, one on port 80 that has received a connection from port 35226 and another on port 35226 that has established a connection to port 80. A normal interaction via a web browser would have both of these sockets in the **`ESTABLISHED`** state, which is part of the TCP protocol.

Argument	**Meaning**
`-a`	Show all sockets
`-t`	Show TCP sockets
`-u`	Show UDP sockets
`-n`	Show numbers, not names
`-p`	Show PID and process name

3.5 Local Networking

Local networking is handled primarily by layer 2, the datalink layer. Linux interacts with layer 2 through *devices*. These might be Ethernet cards, WiFi radios, cellular radios, or even internal virtual devices. In practice, we need to know very little about the layer 2 specifics, since the Linux kernel abstracts this for us. Since applications more directly work with layer 3, we will also see how this interacts with layer 2 at the local level.

3.5.1 Examining Devices with `ifconfig` (The Old Way)

We can examine all of the existing devices on our system with the command **`ifconfig`**. By default, it will only show the active devices. The **`-a`** option will list all devices. Here is an example, from the **`cmsc389z`** docker image:

```
crow@d1b9b82f7557:/$ ifconfig
eth0: flags=4163<UP,BROADCAST,RUNNING,MULTICAST>  mtu 1500
        inet 172.17.0.2  netmask 255.255.0.0  broadcast 172.17.255.255
        ether 02:42:ac:11:00:02  txqueuelen 0  (Ethernet)
        RX packets 15  bytes 1226 (1.2 KB)
        RX errors 0  dropped 0  overruns 0  frame 0
        TX packets 0  bytes 0 (0.0 B)
        TX errors 0  dropped 0 overruns 0  carrier 0  collisions 0

lo: flags=73<UP,LOOPBACK,RUNNING>  mtu 65536
        inet 127.0.0.1  netmask 255.0.0.0
        loop  txqueuelen 1000  (Local Loopback)
        RX packets 0  bytes 0 (0.0 B)
        RX errors 0  dropped 0  overruns 0  frame 0
        TX packets 0  bytes 0 (0.0 B)
        TX errors 0  dropped 0 overruns 0  carrier 0  collisions 0

crow@d1b9b82f7557:/$
crow@d1b9b82f7557:/$ ifconfig -a
eth0: flags=4163<UP,BROADCAST,RUNNING,MULTICAST>  mtu 1500
        inet 172.17.0.2  netmask 255.255.0.0  broadcast 172.17.255.255
        ether 02:42:ac:11:00:02  txqueuelen 0  (Ethernet)
        RX packets 15  bytes 1226 (1.2 KB)
        RX errors 0  dropped 0  overruns 0  frame 0
        TX packets 0  bytes 0 (0.0 B)
        TX errors 0  dropped 0 overruns 0  carrier 0  collisions 0

ip6tnl0: flags=128<NOARP>  mtu 1452
        unspec 00-00-00-00-00-00-00-00-00-00-00-00-00-00-00-00  txqueuelen 1000  (UNSPEC)
        RX packets 0  bytes 0 (0.0 B)
        RX errors 0  dropped 0  overruns 0  frame 0
        TX packets 0  bytes 0 (0.0 B)
        TX errors 0  dropped 0 overruns 0  carrier 0  collisions 0

lo: flags=73<UP,LOOPBACK,RUNNING>  mtu 65536
        inet 127.0.0.1  netmask 255.0.0.0
        loop  txqueuelen 1000  (Local Loopback)
        RX packets 0  bytes 0 (0.0 B)
        RX errors 0  dropped 0  overruns 0  frame 0
        TX packets 0  bytes 0 (0.0 B)
        TX errors 0  dropped 0 overruns 0  carrier 0  collisions 0

tunl0: flags=128<NOARP>  mtu 1480
        tunnel   txqueuelen 1000  (IPIP Tunnel)
        RX packets 0  bytes 0 (0.0 B)
        RX errors 0  dropped 0  overruns 0  frame 0
        TX packets 0  bytes 0 (0.0 B)
        TX errors 0  dropped 0 overruns 0  carrier 0  collisions 0
```

Consider the following lines for **`eth0`** (Ethernet device 0):

```
eth0: flags=4163<UP,BROADCAST,RUNNING,MULTICAST>  mtu 1500
        inet 172.17.0.2  netmask 255.255.0.0  broadcast 172.17.255.255
        ether 02:42:ac:11:00:02  txqueuelen 0  (Ethernet)
```

The first line tells us that the device is “up” and “running”, meaning it is ready to send and receive traffic. It also tells us that the device supports both broadcast (send a message to every host on the local network) and multicast (send a message to a specific set of hosts, possibly on other networks). Further, it tells us that this device supports data frames up to 1500 bytes (the *Maximum Transmission Unit*, or MTU).

The second line tells us the internet address for this device — each device will have its own address. It also specifies a *netmask*, which we can use to identify IP addresses on this same local network by performing a bitwise-AND of the address and netmask. If these values are equal, then they are on the same subnet. The broadcast address is the highest address in the subnet, and messages sent to this address will be delivered to the entire subnet.

The third line tells us the layer 2 (**`ether`**) address for the device. This is often called the MAC (*Media Access Control*) address. While IP addresses are written in dotted-quad format, MAC addresses are written in colon-separated hexadecimal, where the colons separate individual bytes.

3.5.2 Examining Devices with iproute2 (The New Way)

While **`ifconfig`** is a useful command to know, and still commonly referenced, we are not going to use it further. There is a newer networking package called *iproute2*, that was introduced in 2006, and this is what we will use. The primary command in this package is **`ip`**, which has a number of subcommands. We will begin with the **`ip link`** subcommand, which is analogous to **`ifconfig`**:

```
crow@d1b9b82f7557:/$ ip link
1: lo: <LOOPBACK,UP,LOWER_UP> mtu 65536 qdisc noqueue state UNKNOWN mode DEFAULT group default qlen 1000
    link/loopback 00:00:00:00:00:00 brd 00:00:00:00:00:00
2: tunl0@NONE: <NOARP> mtu 1480 qdisc noop state DOWN mode DEFAULT group default qlen 1000
    link/ipip 0.0.0.0 brd 0.0.0.0
3: ip6tnl0@NONE: <NOARP> mtu 1452 qdisc noop state DOWN mode DEFAULT group default qlen 1000
    link/tunnel6 :: brd ::
496: eth0@if497: <BROADCAST,MULTICAST,UP,LOWER_UP> mtu 1500 qdisc noqueue state UP mode DEFAULT group default
    link/ether 02:42:ac:11:00:02 brd ff:ff:ff:ff:ff:ff link-netnsid 0
```

We can look at a specific device with **`ip link show`**:

```
crow@d1b9b82f7557:/$ ip link show eth0
496: eth0@if497: <BROADCAST,MULTICAST,UP,LOWER_UP> mtu 1500 qdisc noqueue state UP mode DEFAULT group default
    link/ether 02:42:ac:11:00:02 brd ff:ff:ff:ff:ff:ff link-netnsid 0
```

We can see most of the information that **`ifconfig eth0`** would show us, with additional information. Some of this will make more sense soon.

What we do not see is the layer 3 information. We can view this with **`ip address show`**:

```
crow@d1b9b82f7557:/$ ip address show eth0
496: eth0@if497: <BROADCAST,MULTICAST,UP,LOWER_UP> mtu 1500 qdisc noqueue state UP group default
    link/ether 02:42:ac:11:00:02 brd ff:ff:ff:ff:ff:ff link-netnsid 0
    inet 172.17.0.2/16 brd 172.17.255.255 scope global eth0
       valid_lft forever preferred_lft forever
```

Now we see the rest of the information we looked at for **`ifconfig`**, but instead of a netmask, we get the IP address in CIDR notation.

We can see additional layer 3 information for our local network with the **`ip route`** subcommand:

```
crow@475283dd6ec2:/$ ip route
default via 172.17.0.1 dev eth0
172.17.0.0/16 dev eth0 proto kernel scope link src 172.17.0.2
```

This tells us that the default route, for anything not specified by an explicit route, is using the gateway router with address **`172.17.0.1`**, and which is reachable via the **`eth0`** device. Further, it tells us that any destination in the **`172.17.0.0/16`** CIDR block is reached via the **`eth0`** device, and our source address for this is **`172.17.0.2`**. Note that this is the same address that appears in **`ip address show eth0`**.

3.5.3 Creating Virtual Devices

At the beginning of this chapter, we mentioned virtual devices at layer 2. The most common virtual device is *loopback*, and you might have noticed this in the output of **`ifconfig`** and **`ip link`**. The device name is usually **`lo`** or **`lo0`**. This is a virtual device for delivering layer 3 traffic between processes on *the same* host, which has the special name **`localhost`**. It has a much simpler frame structure than Ethernet, and is extremely fast (high bandwidth, low latency).

There are other virtual devices often found on a host. *Tunnels* provide virtual devices that *encapsulate* data from one protocol in another, either to translate between protocols (like forwarding IPv6 packets through an IPv4 network) or to make traffic appear to flow through a different local network (such as a Virtual Private Network). Virtual devices are also used to move data between a virtual machine or container's network and the host's network.

We can create our own virtual devices using the **`ip link`** command by adding a new link of type **`veth`** (*virtual ethernet*). This *must* be done as the **`root`** user, as must most of the rest of the commands in this section.

```
root@eaa156967336:/# ip link add v0 type veth peer name v1
root@eaa156967336:/# ip link show
1: lo: <LOOPBACK,UP,LOWER_UP> mtu 65536 qdisc noqueue state UNKNOWN mode DEFAULT group default qlen 1000
    link/loopback 00:00:00:00:00:00 brd 00:00:00:00:00:00
2: tunl0@NONE: <NOARP> mtu 1480 qdisc noop state DOWN mode DEFAULT group default qlen 1000
    link/ipip 0.0.0.0 brd 0.0.0.0
3: ip6tnl0@NONE: <NOARP> mtu 1452 qdisc noop state DOWN mode DEFAULT group default qlen 1000
    link/tunnel6 :: brd ::
6: v1@v0: <BROADCAST,MULTICAST,M-DOWN> mtu 1500 qdisc noop state DOWN mode DEFAULT group default qlen 1000
    link/ether 66:59:d5:2d:62:7b brd ff:ff:ff:ff:ff:ff
7: v0@v1: <BROADCAST,MULTICAST,M-DOWN> mtu 1500 qdisc noop state DOWN mode DEFAULT group default qlen 1000
    link/ether 16:b6:e1:98:ed:b9 brd ff:ff:ff:ff:ff:ff
16: eth0@if17: <BROADCAST,MULTICAST,UP,LOWER_UP> mtu 1500 qdisc noqueue state UP mode DEFAULT group default
    link/ether 02:42:ac:11:00:02 brd ff:ff:ff:ff:ff:ff link-netnsid 0
```

The **`ip link add`** command adds a new device, in this case named **`v0`**. We specify **`veth`** as the type, which then requires us to define a *peer* device locally, which we name **`v1`** with the **`peer name`** argument. Note that we now have two devices, which are shown as **`v0@v1`** and **`v1@v0`**. The actual device name is given before the **`@`**, and the peer device is given after the **`@`**:

```
root@eaa156967336:/# ip link show v0
7: v0@v1: <BROADCAST,MULTICAST,M-DOWN> mtu 1500 qdisc noop state DOWN mode DEFAULT group default qlen 1000
    link/ether 16:b6:e1:98:ed:b9 brd ff:ff:ff:ff:ff:ff
```

What have we actually done here? We have essentially created a virtual ethernet cable (Figure 3.1). Technically, we have actually created a cable with a virtual NIC on either end, but this cable analogy will help us understand the structure we are building.

Figure 3.1: A Network Cable

These devices are, by default, created in the *down* state. That is, they are not ready to send or receive data. In order to do this, we must first add IP addresses to the devices:

```
root@eaa156967336:/# ip addr show v0
7: v0@v1: <BROADCAST,MULTICAST,M-DOWN> mtu 1500 qdisc noop state DOWN group default qlen 1000
    link/ether 16:b6:e1:98:ed:b9 brd ff:ff:ff:ff:ff:ff
root@eaa156967336:/#
root@eaa156967336:/# ip addr add 1.2.3.4/32 dev v0
root@eaa156967336:/# ip addr show v0
7: v0@v1: <BROADCAST,MULTICAST,M-DOWN> mtu 1500 qdisc noop state DOWN group default qlen 1000
    link/ether 16:b6:e1:98:ed:b9 brd ff:ff:ff:ff:ff:ff
    inet 1.2.3.4/32 scope global v0
       valid_lft forever preferred_lft forever
```

We see that initially there is no address set for device **`v0`**. We then run **`ip addr add`**, after which the address has been set. Note that we have to specify the address as a **/32** CIDR block, and the device is given as the last option. Now we can bring the link up:

```
root@eaa156967336:/# ip link set v0 up
root@eaa156967336:/# ip link show v0
7: v0@v1: <NO-CARRIER,BROADCAST,MULTICAST,UP,M-DOWN> mtu 1500 qdisc noqueue state LOWERLAYERDOWN mode DEFAULT group default qlen 1000
    link/ether 16:b6:e1:98:ed:b9 brd ff:ff:ff:ff:ff:ff
```

We cannot send data through this device until the other end of the link, the peer **v1**, has also been configured and brought up:

```
root@eaa156967336:/# ip addr add 1.2.3.5/32 dev v1
root@eaa156967336:/# ip link set v1 up
root@eaa156967336:/# ip addr show v1
6: v1@v0: <BROADCAST,MULTICAST,UP,LOWER_UP> mtu 1500 qdisc noqueue state UP group default qlen 1000
    link/ether 66:59:d5:2d:62:7b brd ff:ff:ff:ff:ff:ff
    inet 1.2.3.5/32 scope global v1
       valid_lft forever preferred_lft forever
```

At this point, we essentially have connected the two devices as shown in Figure 3.2. We will see how to add forwarding routes to these devices and make them behave like physical links in Section 3.7.

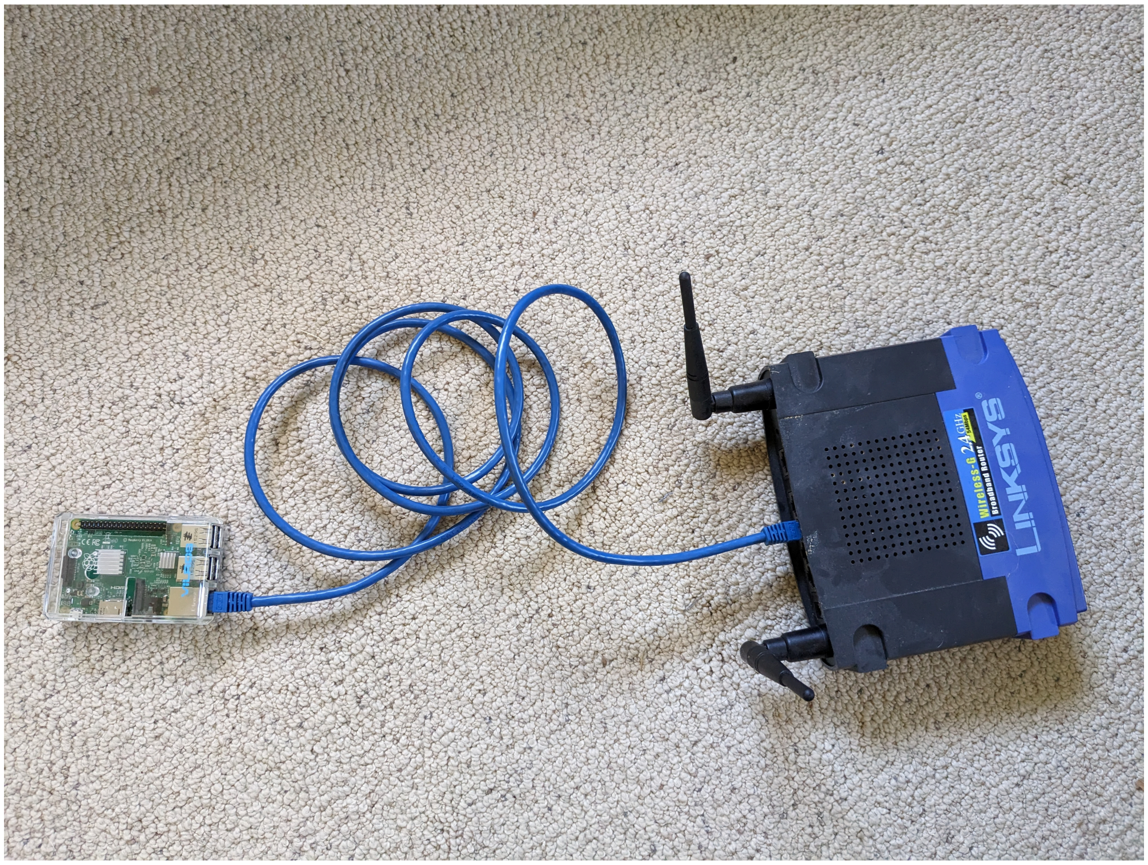


Figure 3.2: Connecting Devices

3.6 End-to-End Connections

We have already seen how we can use netcat to send data across a connection, and this is certainly one way verify that a network connection is active. That requires a specific application to run at each end, however, so it is more a test of a data protocol than connectivity.

If what we want to test is whether it is possible to send *any* data between two hosts, we want something lower-level that is (almost) always present. Fortunately, there is a protocol that can help us with this, called the *Internet Control Message Protocol*, or ICMP.

ICMP has a number of uses. Mostly, it is used by the network to signal problems, such as a subnet or host being unreachable. All hosts implementing IP must also implement at least some of the ICMP protocol. It also has a pair of messages called *Echo Request* and *Echo Response*. We collectively refer to these as *ping*.

We can use the command **`ping`** to send a series of ICMP Echo Requests to a remote host, and wait for the corresponding responses. **`ping`** sends a packet every second, and records the *round-trip-time* (RTT) which is the sum of the one-way latencies. It also keeps track of how many requests do not receive responses, providing a measure of the packet loss on the connection.

```
root@eaa156967336:/# ping 8.8.8.8
PING 8.8.8.8 (8.8.8.8) 56(84) bytes of data.
64 bytes from 8.8.8.8: icmp_seq=1 ttl=37 time=18.8 ms
64 bytes from 8.8.8.8: icmp_seq=2 ttl=37 time=21.5 ms
64 bytes from 8.8.8.8: icmp_seq=3 ttl=37 time=11.5 ms
64 bytes from 8.8.8.8: icmp_seq=4 ttl=37 time=13.4 ms
64 bytes from 8.8.8.8: icmp_seq=5 ttl=37 time=13.8 ms
64 bytes from 8.8.8.8: icmp_seq=6 ttl=37 time=16.1 ms
64 bytes from 8.8.8.8: icmp_seq=7 ttl=37 time=19.9 ms
64 bytes from 8.8.8.8: icmp_seq=8 ttl=37 time=20.6 ms
64 bytes from 8.8.8.8: icmp_seq=9 ttl=37 time=13.0 ms
64 bytes from 8.8.8.8: icmp_seq=10 ttl=37 time=16.2 ms
64 bytes from 8.8.8.8: icmp_seq=11 ttl=37 time=18.4 ms
64 bytes from 8.8.8.8: icmp_seq=12 ttl=37 time=19.3 ms
--- 8.8.8.8 ping statistics ---
12 packets transmitted, 12 received, 0% packet loss, time 11037ms
rtt min/avg/max/mdev = 11.514/16.865/21.451/3.187 ms
```

Since the RTT varies from packet to packet, **`ping`** presents some statistics at the end. Roughly 60% of packets are expected to have RTTs within one standard deviation (**`mdev`**) of the average (**`avg`**). We can also limit the number of ping requests with the **`-c`** option. See the documentation for more details.

Not all hosts will respond to ping requests, so sometimes we will see 100% packet loss:

```
root@eaa156967336:/# ping 2.3.4.5
PING 2.3.4.5 (2.3.4.5) 56(84) bytes of data.
--- 2.3.4.5 ping statistics ---
7 packets transmitted, 0 received, 100% packet loss, time 6162ms
```

In this case, we do not know whether there is no host with address **`2.3.4.5`**, whether it is not responding to pings, or if there is a problem with the connection.

We have another tool that can shed some light on this, called **`traceroute`**. This takes advantage of a field in the IP header called *Time-to-Live*, or TTL. Each router that receives a packet decrements the value of this field, and if it reaches 0 the packet is discarded. The main purpose of this is to prevent packets from propagating indefinitely in the event of a *routing loop*. This occurs when a router *A* forwards a packet to destination *d*, and some router *B* further along the forwarding path sends it back to *A*. When the TTL reaches 0, the router discarding the packet sends an ICMP TTL Exceeded message back to the source.

How does **`traceroute`** use the TTL? It begins by sending a packet to the destination with a TTL of 1. This causes the first router to send a TTL Exceeded message back, which includes the router's IP address as the source. It then increases the initial TTL by 1, in order to learn the IP address of the second router. This continues until either the destination responds or a certain number of failures are observed.

```
root@eaa156967336:/# traceroute 8.8.8.8
traceroute to 8.8.8.8 (8.8.8.8), 30 hops max, 60 byte packets
 1  172.17.0.1 (172.17.0.1)  2.479 ms  0.016 ms  0.006 ms
 2  Fios_Quantum_Gateway.fios-router.home (192.168.1.1)  4.587 ms  4.861 ms  5.249 ms
 3  lo0-100.WASHDC-VFTTP-368.verizon-gni.net (72.83.250.1)  15.181 ms  14.913 ms  14.952 ms
 4  100.41.32.46 (100.41.32.46)  21.881 ms  21.711 ms  23.656 ms
 5  0.ae1.GW16.IAD8.ALTER.NET (140.222.3.87)  24.254 ms 0.ae2.GW16.IAD8.ALTER.NET (140.222.3.89)  15.191 ms
 6  204.148.170.158 (204.148.170.158)  23.847 ms  9.642 ms  9.502 ms
 7  * * *
 8  142.251.70.110 (142.251.70.110)  12.857 ms dns.google (8.8.8.8)  11.014 ms  10.285 ms
```

What we see on each line is the TTL, the IP address (and possibly hostname) of the router at that distance, and then three round-trip times. Why three? Because

traceroute actually repeats each distance probe three times, to show the statistical variation. We also sometimes see multiple router addresses on a line. That is because there are often multiple paths from a source to a destination, so we do not always see the same routers along the way. This also means that just because two routers appear on adjacent lines in the **traceroute** output, that **does not** mean they are directly connected.

Let's see our failed **ping**:

```
root@eaa156967336:/# traceroute 2.3.4.5
traceroute to 2.3.4.5 (2.3.4.5), 30 hops max, 60 byte packets
 1  172.17.0.1 (172.17.0.1)  0.073 ms  0.013 ms  0.009 ms
 2  Fios_Quantum_Gateway.fios-router.home (192.168.1.1)  9.976 ms  10.308 ms  10.273 ms
 3  lo0-100.WASHDC-VFTTP-368.verizon-gni.net (72.83.250.1)  22.969 ms  23.104 ms  23.118 ms
 4  100.41.32.44 (100.41.32.44)  23.456 ms 100.41.32.46 (100.41.32.46)  23.730 ms  23.492 ms
 5  0.ae1.GW12.IAD8.ALTER.NET (140.222.234.27)  24.603 ms 0.ae2.GW12.IAD8.ALTER.NET (140.222.234.29)  23.815
 6  63.88.105.94 (63.88.105.94)  23.912 ms  11.488 ms *
 7  81.52.166.172 (81.52.166.172)  113.875 ms  108.002 ms *
 8  ae324-0.ffttr7.frankfurt.opentransit.net (193.251.240.102)  103.174 ms  104.060 ms  103.430 ms
 9  * * *
10  * * *
29  * * *
30  * * *
```

Each `*` represents a packet sent for which no response was received. The lines with `* * *` received no responses from those TTL values. We have omitted most of these. From this, we can see that the last router to receive our packet was **193.251.240.102**. So, what does this tell us? That depends on what we know about the destination and the router. We will revisit this in Chapter 4.

There is an important consideration when looking at these IP addresses. We noted in Section 3.5 that every device on a host has its own IP address. A router has many network devices, and the IP address we see is for the device that *received* the packet and decremented the TTL to 0. That is, what we are learning are the "near-end" IP addresses of the router. It is entirely possible that the router at **193.251.240.102** has another device in, say, **2.3.0.0/16**.

3.7 Network Namespaces

We have looked at the network stack (there is still a lot left to explore), and seen how each host on the network has its own stack. Linux also allows us to have multiple independent network stacks on the same host. It does this using *network namespaces*. Much like namespaces in programming languages, network namespaces make their devices (objects) visible only to other processes (methods) and devices within the same namespace.

The **ip netns** subcommand manages network namespaces in Linux. You can view the existing namespaces:

```
root@eaa156967336:/# ip netns
root@eaa156967336:/# ip netns show
root@eaa156967336:/#
```

These commands are equivalent, and you can see that, by default, there are no namespaces defined.

We can add a namespace with **ip netns add**:

```
root@eaa156967336:/# ip netns add foo
root@eaa156967336:/# ip netns
foo
```

We can also delete them with **ip netns del**:

```
root@eaa156967336:/# ip netns del foo
root@eaa156967336:/# ip netns
root@eaa156967336:/#
```

Any program can be run in a namespace using **ip netns exec**:

```
root@eaa156967336:/# ip netns add foo
root@eaa156967336:/# ls
bin   dev  home  lib32  libx32  mnt  proc  run   srv  tmp  var
boot  etc  lib   lib64  media   opt  root  sbin  sys  usr
root@eaa156967336:/# ip netns exec foo ls
bin   dev  home  lib32  libx32  mnt  proc  run   srv  tmp  var
boot  etc  lib   lib64  media   opt  root  sbin  sys  usr
```

What does this show us? It shows that we are in the same filesystem, with access to all of the files and programs. Let's look at something that differs:

```
root@eaa156967336:/# ip link
1: lo: <LOOPBACK,UP,LOWER_UP> mtu 65536 qdisc noqueue state UNKNOWN mode DEFAULT group default qlen 1000
    link/loopback 00:00:00:00:00:00 brd 00:00:00:00:00:00
2: tunl0@NONE: <NOARP> mtu 1480 qdisc noop state DOWN mode DEFAULT group default qlen 1000
    link/ipip 0.0.0.0 brd 0.0.0.0
3: ip6tnl0@NONE: <NOARP> mtu 1452 qdisc noop state DOWN mode DEFAULT group default qlen 1000
    link/tunnel6 :: brd ::
6: v1@v0: <BROADCAST,MULTICAST,UP,LOWER_UP> mtu 1500 qdisc noqueue state UP mode DEFAULT group default qlen 1000
    link/ether 66:59:d5:2d:62:7b brd ff:ff:ff:ff:ff:ff
7: v0@v1: <BROADCAST,MULTICAST,UP,LOWER_UP> mtu 1500 qdisc noqueue state UP mode DEFAULT group default qlen 1000
    link/ether 16:b6:e1:98:ed:b9 brd ff:ff:ff:ff:ff:ff
16: eth0@if17: <BROADCAST,MULTICAST,UP,LOWER_UP> mtu 1500 qdisc noqueue state UP mode DEFAULT group default
    link/ether 02:42:ac:11:00:02 brd ff:ff:ff:ff:ff:ff link-netnsid 0
root@eaa156967336:/#
root@eaa156967336:/# ip netns exec foo ip link
1: lo: <LOOPBACK> mtu 65536 qdisc noop state DOWN mode DEFAULT group default qlen 1000
    link/loopback 00:00:00:00:00:00 brd 00:00:00:00:00:00
2: tunl0@NONE: <NOARP> mtu 1480 qdisc noop state DOWN mode DEFAULT group default qlen 1000
    link/ipip 0.0.0.0 brd 0.0.0.0
3: ip6tnl0@NONE: <NOARP> mtu 1452 qdisc noop state DOWN mode DEFAULT group default qlen 1000
    link/tunnel6 :: brd ::
```

If we are not in the namespace, we have all of our normal devices (including the **`veth`** link we created before). In the namespace, we have fewer, all of which are down.

We can run **`ping`** on the loopback address **`127.0.0.1`**:

```
root@eaa156967336:/# ping -c 3 127.0.0.1
PING 127.0.0.1 (127.0.0.1) 56(84) bytes of data.
64 bytes from 127.0.0.1: icmp_seq=1 ttl=64 time=0.045 ms
64 bytes from 127.0.0.1: icmp_seq=2 ttl=64 time=0.091 ms
64 bytes from 127.0.0.1: icmp_seq=3 ttl=64 time=0.094 ms

--- 127.0.0.1 ping statistics ---
3 packets transmitted, 3 received, 0% packet loss, time 2060ms
rtt min/avg/max/mdev = 0.045/0.076/0.094/0.022 ms
root@eaa156967336:/# ip netns exec foo ping -c 3 127.0.0.1
ping: connect: Network is unreachable
```

Here we see that in the **`foo`** namespace, we are unable to ping ourselves, because the **`lo`** device is down.

```
root@eaa156967336:/# ip netns exec foo ip link set lo up
root@eaa156967336:/# ip netns exec foo ping -c 3 127.0.0.1
PING 127.0.0.1 (127.0.0.1) 56(84) bytes of data.
64 bytes from 127.0.0.1: icmp_seq=1 ttl=64 time=0.050 ms
64 bytes from 127.0.0.1: icmp_seq=2 ttl=64 time=0.087 ms
64 bytes from 127.0.0.1: icmp_seq=3 ttl=64 time=0.109 ms

--- 127.0.0.1 ping statistics ---
3 packets transmitted, 3 received, 0% packet loss, time 2027ms
rtt min/avg/max/mdev = 0.050/0.082/0.109/0.024 ms
```

Now that **`lo`** is up, we can ping over loopback.

3.7.1 Using Namespaces to Create Virtual Hosts

We are going to see how to use network namespaces to create a *testbed*. This is an emulated network in a controlled environment, which can be an essential part of debugging and testing a network protocol. To do this, we are going to create a separate namespace for every *virtual host* in our testbed. We will then add links between them, configure routing, and even set link characteristics to control bandwidth, latency, and packet loss.

To start, we want a second namespace, which we will call **`bar`**:

```
root@eaa156967336:/# ip netns add bar
root@eaa156967336:/# ip netns
bar
foo
```

We will need to bring the **`lo`** device up in namespace **`bar`** before we can send any packets. Recall that we have two virtual devices, **`v0`** and **`v1`**, connected as a virtual ethernet link.

```
root@eaa156967336:/# ip link show type veth
6: v1@v0: <BROADCAST,MULTICAST,UP,LOWER_UP> mtu 1500 qdisc noqueue state UP mode DEFAULT group default qlen 1000
    link/ether 66:59:d5:2d:62:7b brd ff:ff:ff:ff:ff:ff
7: v0@v1: <BROADCAST,MULTICAST,UP,LOWER_UP> mtu 1500 qdisc noqueue state UP mode DEFAULT group default qlen 1000
    link/ether 16:b6:e1:98:ed:b9 brd ff:ff:ff:ff:ff:ff
16: eth0@if17: <BROADCAST,MULTICAST,UP,LOWER_UP> mtu 1500 qdisc noqueue state UP mode DEFAULT group default
    link/ether 02:42:ac:11:00:02 brd ff:ff:ff:ff:ff:ff link-netnsid 0
root@eaa156967336:/# ip netns exec foo ip link show type veth
root@eaa156967336:/# ip netns exec bar ip link show type veth
```

Our two namespaces have no virtual devices in them. Going back to our physical analogy, we now think of the Raspberry Pi and router as separate namespaces (virtual hosts). The cable is not yet connecting them (Figure 3.3).

Figure 3.3: Devices Not Yet Connected

We can move devices between namespaces with another option to **ip link set**:

```
root@eaa156967336:/# ip link set v0 netns foo
root@eaa156967336:/# ip link set v1 netns bar
root@eaa156967336:/# ip link show type veth
16: eth0@if17: <BROADCAST,MULTICAST,UP,LOWER_UP> mtu 1500 qdisc noqueue state UP mode DEFAULT group default
    link/ether 02:42:ac:11:00:02 brd ff:ff:ff:ff:ff:ff link-netnsid 0
root@eaa156967336:/# ip netns exec foo ip link show type veth
7: v0@if6: <BROADCAST,MULTICAST> mtu 1500 qdisc noop state DOWN mode DEFAULT group default qlen 1000
    link/ether 16:b6:e1:98:ed:b9 brd ff:ff:ff:ff:ff:ff link-netns bar
root@eaa156967336:/# ip netns exec bar ip link show type veth
6: v1@if7: <BROADCAST,MULTICAST> mtu 1500 qdisc noop state DOWN mode DEFAULT group default qlen 1000
    link/ether 66:59:d5:2d:62:7b brd ff:ff:ff:ff:ff:ff link-netns foo
```

Now we see that **`v0`** is no longer in the default namespace, but is instead in **`foo`**, and similarly **`v1`** is in **`bar`**. This is sometimes called "throwing the device over the wall" into the namespace, because you no longer have access to it from the initial (default) namespace.

In the **`foo`** namespace, the link **`v0`** is unconfigured:

```
root@eaa156967336:/# ip netns exec foo ip addr show v0
7: v0@if6: <BROADCAST,MULTICAST> mtu 1500 qdisc noop state DOWN group default qlen 1000
    link/ether 16:b6:e1:98:ed:b9 brd ff:ff:ff:ff:ff:ff link-netns bar
```

The same is true of **`v1`** in **`bar`**. Let's add addresses to the devices and bring them up:

```
root@eaa156967336:/# ip netns exec foo ip addr add 1.2.3.4/32 dev v0
root@eaa156967336:/# ip netns exec bar ip addr add 1.2.3.5/32 dev v1
root@eaa156967336:/# ip netns exec foo ip link set v0 up
root@eaa156967336:/# ip netns exec bar ip link set v1 up
root@eaa156967336:/# ip netns exec foo ip addr show v0
7: v0@if6: <BROADCAST,MULTICAST,UP,LOWER_UP> mtu 1500 qdisc noqueue state UP group default qlen 1000
    link/ether 16:b6:e1:98:ed:b9 brd ff:ff:ff:ff:ff:ff link-netns bar
    inet 1.2.3.4/32 scope global v0
       valid_lft forever preferred_lft forever
    inet6 fe80::14b6:e1ff:fe98:edb9/64 scope link
       valid_lft forever preferred_lft forever
root@eaa156967336:/# ip netns exec bar ip addr show v1
6: v1@if7: <BROADCAST,MULTICAST,UP,LOWER_UP> mtu 1500 qdisc noqueue state UP group default qlen 1000
    link/ether 66:59:d5:2d:62:7b brd ff:ff:ff:ff:ff:ff link-netns foo
    inet 1.2.3.5/32 scope global v1
       valid_lft forever preferred_lft forever
    inet6 fe80::6459:d5ff:fe2d:627b/64 scope link
       valid_lft forever preferred_lft forever
```

At this point, we once again have the virtual equivalent of Figure 3.2, but now the Pi and router are virtual hosts.

3.7.2 Routing Between Namespaces

At this point, we have devices in **`foo`** and **`bar`** with addresses **`1.2.3.4`** and **`1.2.3.5`**, and the devices are up and available for traffic. What we *do not* yet have is a way for Linux to know how to use these devices. We add this by modifying the *routing tables* for the namespace.

Since **`foo`** and **`bar`** are directly connected by a single **`veth`** link, we can do this fairly simply with the **`ip route add`** command:

```
root@eaa156967336:/# ip netns exec foo ip route add 1.2.3.5/32 dev v0 proto static scope global src 1.2.3.4
root@eaa156967336:/# ip netns exec bar ip route add 1.2.3.4/32 dev v1 proto static scope global src 1.2.3.5
root@eaa156967336:/# ip netns exec foo ip route
1.2.3.5 dev v0 proto static src 1.2.3.4
root@eaa156967336:/# ip netns exec bar ip route
1.2.3.4 dev v1 proto static src 1.2.3.5
```

There is a lot going on here. **`ip route add`** takes a CIDR prefix for the destination, which in this case is just the address of the other end of the link, the device through which to send traffic to this destination, some "standard" options, and then the source address to attach to traffic forwarded through this table entry. After doing this for both namespaces, we can see that we now have routing tables with the routes we just created.

The proof of the routing is in the pinging, so let's give it a try:

```
root@eaa156967336:/# ip netns exec foo ping -c 3 1.2.3.5
PING 1.2.3.5 (1.2.3.5) 56(84) bytes of data.
64 bytes from 1.2.3.5: icmp_seq=1 ttl=64 time=0.023 ms
64 bytes from 1.2.3.5: icmp_seq=2 ttl=64 time=0.075 ms
64 bytes from 1.2.3.5: icmp_seq=3 ttl=64 time=0.107 ms

--- 1.2.3.5 ping statistics ---
3 packets transmitted, 3 received, 0% packet loss, time 2040ms
rtt min/avg/max/mdev = 0.023/0.068/0.107/0.034 ms
```

We have a working network!

Our network is not terribly useful, though. Let's add a third namespace, **`baz`**, and connect it to **`bar`**.

```
ip netns add baz
ip netns exec baz ip link set lo up
ip link add v2 type veth peer name v3
ip link set v2 netns bar
ip link set v3 netns baz
ip netns exec bar ip addr add 1.2.3.5/32 dev v2
ip netns exec baz ip addr add 1.2.3.6/32 dev v3
ip netns exec bar ip link set v2 up
ip netns exec baz ip link set v3 up
ip netns exec bar ip route add 1.2.3.6/32 dev v2 proto static scope global src 1.2.3.5
ip netns exec baz ip route add 1.2.3.5/32 dev v3 proto static scope global src 1.2.3.6
```

What does our network now look like?

```
root@eaa156967336:/# ip netns exec foo ip route
1.2.3.5 dev v0 proto static src 1.2.3.4
root@eaa156967336:/# ip netns exec bar ip route
1.2.3.4 dev v1 proto static src 1.2.3.5
1.2.3.6 dev v2 proto static src 1.2.3.5
root@eaa156967336:/# ip netns exec baz ip route
1.2.3.5 dev v3 proto static src 1.2.3.6
```

What about connectivity?

```
root@eaa156967336:/# ip netns exec foo ping -c 3 1.2.3.5
PING 1.2.3.5 (1.2.3.5) 56(84) bytes of data.
64 bytes from 1.2.3.5: icmp_seq=1 ttl=64 time=0.046 ms
64 bytes from 1.2.3.5: icmp_seq=2 ttl=64 time=0.101 ms
64 bytes from 1.2.3.5: icmp_seq=3 ttl=64 time=0.044 ms

--- 1.2.3.5 ping statistics ---
3 packets transmitted, 3 received, 0% packet loss, time 2041ms
rtt min/avg/max/mdev = 0.044/0.063/0.101/0.026 ms
root@eaa156967336:/# ip netns exec foo ping -c 3 1.2.3.6
ping: connect: Network is unreachable
root@eaa156967336:/# ip netns exec bar ping -c 3 1.2.3.4
PING 1.2.3.4 (1.2.3.4) 56(84) bytes of data.
64 bytes from 1.2.3.4: icmp_seq=1 ttl=64 time=0.022 ms
64 bytes from 1.2.3.4: icmp_seq=2 ttl=64 time=0.271 ms
64 bytes from 1.2.3.4: icmp_seq=3 ttl=64 time=0.362 ms

--- 1.2.3.4 ping statistics ---
3 packets transmitted, 3 received, 0% packet loss, time 2035ms
rtt min/avg/max/mdev = 0.022/0.218/0.362/0.143 ms
root@eaa156967336:/# ip netns exec bar ping -c 3 1.2.3.6
PING 1.2.3.6 (1.2.3.6) 56(84) bytes of data.
64 bytes from 1.2.3.6: icmp_seq=1 ttl=64 time=0.061 ms
64 bytes from 1.2.3.6: icmp_seq=2 ttl=64 time=0.074 ms
64 bytes from 1.2.3.6: icmp_seq=3 ttl=64 time=0.079 ms

--- 1.2.3.6 ping statistics ---
3 packets transmitted, 3 received, 0% packet loss, time 2045ms
rtt min/avg/max/mdev = 0.061/0.071/0.079/0.007 ms
root@eaa156967336:/# ip netns exec baz ping -c 3 1.2.3.4
ping: connect: Network is unreachable
root@eaa156967336:/# ip netns exec baz ping -c 3 1.2.3.5
```

```
PING 1.2.3.5 (1.2.3.5) 56(84) bytes of data.
64 bytes from 1.2.3.5: icmp_seq=1 ttl=64 time=0.039 ms
64 bytes from 1.2.3.5: icmp_seq=2 ttl=64 time=0.073 ms
64 bytes from 1.2.3.5: icmp_seq=3 ttl=64 time=0.099 ms

--- 1.2.3.5 ping statistics ---
3 packets transmitted, 3 received, 0% packet loss, time 2056ms
rtt min/avg/max/mdev = 0.039/0.070/0.099/0.024 ms
```

From this, we can see that **`foo`** and **`bar`** can ping each other, as can **`bar`** and **`baz`**. **`foo`** and **`baz`**, however, cannot.

If we look at our routing tables again, we can see at least part of what the problem is: **`foo`** has no forwarding rule that applies to the address **`1.2.3.6`**, and **`baz`** has no forwarding rule that applies to the address **`1.2.3.4`**. We can add these, and see what changes:

```
ip netns exec foo ip route add 1.2.3.0/24 dev v0 proto static scope global src 1.2.3.4
ip netns exec baz ip route add 1.2.3.0/24 dev v3 proto static scope global src 1.2.3.6
```

Now we examine our routing tables and try to ping:

```
root@eaa156967336:/# ip netns exec foo ip route
1.2.3.0/24 dev v0 proto static src 1.2.3.4
1.2.3.5 dev v0 proto static src 1.2.3.4
root@eaa156967336:/# ip netns exec baz ip route
1.2.3.0/24 dev v3 proto static src 1.2.3.6
1.2.3.5 dev v3 proto static src 1.2.3.6
root@eaa156967336:/# ip netns exec foo ping -c 3 1.2.3.6
PING 1.2.3.6 (1.2.3.6) 56(84) bytes of data.
From 1.2.3.4 icmp_seq=1 Destination Host Unreachable
From 1.2.3.4 icmp_seq=2 Destination Host Unreachable
From 1.2.3.4 icmp_seq=3 Destination Host Unreachable

--- 1.2.3.6 ping statistics ---
3 packets transmitted, 0 received, +3 errors, 100% packet loss, time 2085ms
pipe 3
```

We still cannot connect **`foo`** and **`baz`**, but why not? When **`bar`** receives a packet through device **`v1`** with a destination of **`1.2.3.6`**, it does not actually know that it is supposed to forward it further. That means it just ignores the packet, and it goes no further. Instead, we need **`bar`** to act as a *bridge* between devices **`v1`** and **`v2`**. Without bridging, we would have to have point-to-point links between all hosts in our network (called a *mesh network*). This will work, but it becomes unmanageable, and is not appropriate for all testing situations. We will look at adding bridges, but first we are going to take a look at something useful even in small mesh networks.

3.7.3 Network Emulation

One of the additional programs in iproute2 is **`tc`**, which implements *traffic control*. There is a lot you can do with this, but we are going to limit ourselves to simple *queueing discplines* (**`qdisc`**). We can add a queueing discipline to a device with

```
ip netns exec foo tc qdisc add dev v0 root handle 1:0 netem
```

This adds a new **`qdisc`** to **`foo`**'s **`v0`** device, and configures it for *network emulation* (**`netem`**). We can then update the **`qdisc`** to emulate real network behaviors:

```
ip netns exec foo tc qdisc change dev v0 root netem delay 10ms rate 1Mbit loss random 30
```

Now the **`v0`** device in the **`foo`** namespace will send traffic with a latency of 10ms, a bandwidth of 1Mbps, and will randomly drop 30% of the packets. This is one-way, so if we ping **`bar`**:

```
root@eaa156967336:/# ip netns exec foo ping -c 5 1.2.3.5
PING 1.2.3.5 (1.2.3.5) 56(84) bytes of data.
64 bytes from 1.2.3.5: icmp_seq=2 ttl=64 time=11.4 ms
64 bytes from 1.2.3.5: icmp_seq=3 ttl=64 time=13.5 ms
64 bytes from 1.2.3.5: icmp_seq=5 ttl=64 time=14.2 ms

--- 1.2.3.5 ping statistics ---
5 packets transmitted, 3 received, 40% packet loss, time 4086ms
rtt min/avg/max/mdev = 11.365/13.014/14.210/1.204 ms
```

We see RTTs slightly over 10ms because of the bandwidth limit. If we lower the bandwidth to 100kbps, this becomes more pronounced:

```
root@eaa156967336:/# ip netns exec foo tc qdisc change dev v0 root netem delay 10ms rate 100kbit loss random 30
root@eaa156967336:/# ip netns exec foo ping -c 5 1.2.3.5
PING 1.2.3.5 (1.2.3.5) 56(84) bytes of data.
64 bytes from 1.2.3.5: icmp_seq=2 ttl=64 time=18.2 ms
64 bytes from 1.2.3.5: icmp_seq=3 ttl=64 time=18.4 ms
64 bytes from 1.2.3.5: icmp_seq=4 ttl=64 time=18.4 ms
64 bytes from 1.2.3.5: icmp_seq=5 ttl=64 time=18.0 ms

--- 1.2.3.5 ping statistics ---
5 packets transmitted, 4 received, 20% packet loss, time 4147ms
rtt min/avg/max/mdev = 18.002/18.249/18.442/0.167 ms
```

We are not seeing precisely 30% packet loss, because of the small sample size. If we increase this:

```
root@eaa156967336:/# ip netns exec foo ping -q -c 20 1.2.3.5
PING 1.2.3.5 (1.2.3.5) 56(84) bytes of data.

--- 1.2.3.5 ping statistics ---
20 packets transmitted, 14 received, 30% packet loss, time 19757ms
rtt min/avg/max/mdev = 18.259/19.568/22.042/1.315 ms
```

The **`-q`** flag makes **`ping`** "quiet", printing only the summary at the end.

3.7.4 Adding Bridge Nodes

At this point, we can create mesh networks with realistic network link emulation. We still do not have actual packet forwarding, however, so now it is time to revisit bridging. Within the Linux kernel, a bridge device allows externally facing devices, like our **`veth`** endpoints, to interact through the routing table.[1]

The old way to create a bridge was with the **`brctl`** command. Here are the steps to do this:

[1]Technically, we can configure the system to forward received packets if it is not the destination. This method gives us more control, however, and demonstrates some additional commands.

```
ip netns exec bar brctl addbr br0
ip netns exec bar ip addr add dev br0 local 1.2.3.5
ip netns exec bar ip link set dev br0 up
ip netns exec bar brctl addif br0 v1
ip netns exec bar brctl addif br0 v2
```

After executing these commands, we can communicate between **`foo`** and **`baz`**!

```
root@eaa156967336:/# ip netns exec foo ping -c 3 1.2.3.6
PING 1.2.3.6 (1.2.3.6) 56(84) bytes of data.
64 bytes from 1.2.3.6: icmp_seq=1 ttl=64 time=33.6 ms
64 bytes from 1.2.3.6: icmp_seq=2 ttl=64 time=19.3 ms

--- 1.2.3.6 ping statistics ---
3 packets transmitted, 2 received, 33.3333% packet loss, time 2009ms
rtt min/avg/max/mdev = 19.326/26.444/33.563/7.118 ms
```

The iproute2 package has a **`bridge`** command that replaces **`brctl`**, and has additional functionality. We actually do not even need this to create a simple bridge, as the **`ip`** command can do the basic configuration”

```
ip netns exec bar ip link add name br0 type bridge
ip netns exec bar ip link set dev br0 up
ip netns exec bar ip link set dev v1 master br0
ip netns exec bar ip link set dev v2 master br0
```

This begins by creating a new device (**`br0`**) of **`type bridge`**. Once we have brought the device up, we can set our virtual devices **`v1`** and **`v2`** as subordinate to **`br0`** with the **`master`** property. We can see the bridge configuration with:

```
root@eaa156967336:/# ip netns exec bar bridge link show
6: v1@if7: <BROADCAST,MULTICAST,UP,LOWER_UP> mtu 1500 master br0 state forwarding priority 32 cost 2
9: v2@if8: <BROADCAST,MULTICAST,UP,LOWER_UP> mtu 1500 master br0 state forwarding priority 32 cost 2
```

Now we can create an arbitrary[2] number of virtual hosts, with realistic forwarding tables and link characteristics. This allows us to test protocols in a

[2]If you try to create *too* many, you might crash your system, but you should be able to manage 1000.

wide variety of network configurations, including those that involve network failures or attacks.

Chapter 4

Interactions Between Layers

We have mostly looked at the OSI model's layers independently, but when was the last time you typed an IP address into your web browser, not to mention a sequence of MAC addresses? Instead, we present a reasonable name at an appropriate layer, and rely on additional network services to convert this to the address used by the layer below or above.

4.1 Domain Name System

Let's start with the one you are likely to see many times every day. This book is currently hosted on a server with IP address **`128.8.127.4`**, but you almost certainly did not provide this address directly to your browser. Instead, you would have entered (or followed a link to) **`www.cs.umd.edu`**. This human-friendly name is not routable by the network, so we need to get the associated IP address. How does this work?

The *Domain Name System* (DNS) is a globally distributed database. Its primary (but not only) function is to provide mappings between *fully qualified domain names* (FQDNs, such as **`www.cs.umd.edu`**) and IP addresses (such as

`128.8.127.4`). Domain names are organized hierarchically, beginning with the *top-level domains* (TLDs), and delegating to *registries*. The TLDs are defined and managed by the *Internet Corporation for Assigned Names and Numbers* (ICANN), as are the country-code domains (such as **`.us`**).

Most applications that take FQDNs (ie, *hostnames*) perform DNS lookups automatically. You can do this from the command line, as well, which is often useful. Some systems still use an old program called **`nslookup`** for this, but that has largely been deprecated in favor of the programs **`host`** and **`dig`**.

Most of the time, **`host`** is the program you want to use. Given a FQDN, it will tell you the IP address or addresses associated with it:

```
□  12:58:45 389Z $ host www.cs.umd.edu
www.cs.umd.edu is an alias for www-hlb.cs.umd.edu.
www-hlb.cs.umd.edu has address 128.8.127.4
```

In this case, the FQDN **`www.cs.umd.edu`** is an alias for another FQDN, **`www-hlb.cs.umd.edu`**, so **`host`** retrieves the address record for this as well. You can also perform *reverse lookups*, where you ask for the FQDN matching an IP address:

```
□  13:00:47 389Z $ host 128.8.127.4
4.127.8.128.in-addr.arpa domain name pointer www-hlb.cs.umd.edu.
```

Here we actually perform a special DNS lookup in the **`in-addr.arpa`** *domain*, with the order of the quads reversed. The reason for this is that FQDNs begin with the host-specific part of the name and end with the top-level domain, but IP addresses begin with the largest network segment and end with the host-specific part of the address. Not all IP addresses will have a reverse entry.

When you want to *resolve* a FQDN, you first contact your local *resolver*, also called a *DNS server*. The resolver should be set for you automatically when you join the network if you are using DHCP (*Dynamic Host Configuration Protocol*), which you almost always are. This local resolver maintains local information and a cache of retrieved DNS database records, and will recursively contact other resolvers for you if it does not know the requested information.

We are using "FQDN" rather than "hostname" deliberately here. While a FQDN *is* a hostname, a hostname might not be a FQDN. In particular, a host on your local network could be known simply by its short hostname, such as **`www`** (on the CS department's network):

```
▯  13:00:51 389Z $ host www
www.cs.umd.edu is an alias for www-hlb.cs.umd.edu.
```

In this case, the local host appends its known domain to the requested hostname to obtain the FQDN for the DNS request.

There is another program called **`dig`** that does substantially the same thing. The main difference is in the information displayed:

```
▯  13:43:15 389Z $ dig www.cs.umd.edu

; <<>> DiG 9.10.6 <<>> www.cs.umd.edu
;; global options: +cmd
;; Got answer:
;; ->>HEADER<<- opcode: QUERY, status: NOERROR, id: 12444
;; flags: qr rd ra; QUERY: 1, ANSWER: 2, AUTHORITY: 0, ADDITIONAL: 1

;; OPT PSEUDOSECTION:
; EDNS: version: 0, flags:; udp: 512
;; QUESTION SECTION:
;www.cs.umd.edu.                IN      A

;; ANSWER SECTION:
www.cs.umd.edu.         11373   IN      CNAME   www-hlb.cs.umd.edu.
www-hlb.cs.umd.edu.     83373   IN      A       128.8.127.4

;; Query time: 32 msec
;; SERVER: 10.72.14.49#53(10.72.14.49)
;; WHEN: Tue Nov 08 13:48:48 EST 2022
;; MSG SIZE  rcvd: 81
```

We have been given a lot more information here, including all of the options that were included in the request and a detailed breakdown of the response. Comparing what we saw in the output from **`host`**, an alias is resolved through a **`CNAME`** ("Canonical Name") DNS record, while an address is resolved through an **`A`** ("Address") record. The number that appears after the FQDN (which technically ends in a dot, though you would not provide that as part of a hostname) is the *time to live* for this record. Unlike the TTL of an IP packet, this TTL is measured in seconds, and is how long the record may be cached before it expires and should be retrieved again. In this case, **`www.cs.umd.edu`** should be checked again in a little over 3 hours to make sure it is still an alias for **`www-hlb.cs.umd.edu`**, which itself should be checked in almost 14 hours to see if its IP address has changed.

4.2 Network Blocks

As with the DNS hierarchy, the IP address space is also managed as a hierarchy. The largest network blocks (*netblocks*) are assigned by the *Internet Assigned Numbers Authority* (IANA), and those blocks are then subdivided by their respective owners. Some of this is publicly available, though as networks are divided into ever-smaller subnets, this information becomes purely internal.

We can learn a good deal about both domains and netblocks with the **`whois`** program, which queries yet another distributed database. Here is an example of a domain registration lookup:

```
  14:17:29 389Z $ whois umd.edu
% IANA WHOIS server
% for more information on IANA, visit http://www.iana.org
% This query returned 1 object

refer:        whois.educause.edu

domain:       EDU

organisation: EDUCAUSE
```

```
address:       282 Century Place, Suite 5000
address:       Louisville, CO  80027
address:       United States

contact:       administrative
name:          Information Services Administration
organisation:  EDUCAUSE
address:       4772 Walnut Street, Suite 206
address:       Boulder Colorado 80301
address:       United States
phone:         +1-303-449-4430
fax-no:        +1-303-440-0461
e-mail:        netadmin@educause.edu

contact:       technical
name:          Registry Customer Service
organisation:  VeriSign Global Registry
address:       12061 Bluemont Way
address:       Reston Virginia 20190
address:       United States
phone:         +1-703-925-6999
fax-no:        +1-703-948-3978
e-mail:        info@verisign-grs.com

nserver:       A.EDU-SERVERS.NET 192.5.6.30 2001:503:a83e:0:0:0:2:30
nserver:       B.EDU-SERVERS.NET 192.33.14.30 2001:503:231d:0:0:0:2:30
nserver:       C.EDU-SERVERS.NET 192.26.92.30 2001:503:83eb:0:0:0:0:30
nserver:       D.EDU-SERVERS.NET 192.31.80.30 2001:500:856e:0:0:0:0:30
nserver:       E.EDU-SERVERS.NET 192.12.94.30 2001:502:1ca1:0:0:0:0:30
nserver:       F.EDU-SERVERS.NET 192.35.51.30 2001:503:d414:0:0:0:0:30
nserver:       G.EDU-SERVERS.NET 192.42.93.30 2001:503:eea3:0:0:0:0:30
nserver:       H.EDU-SERVERS.NET 192.54.112.30 2001:502:8cc:0:0:0:0:30
nserver:       I.EDU-SERVERS.NET 192.43.172.30 2001:503:39c1:0:0:0:0:30
nserver:       J.EDU-SERVERS.NET 192.48.79.30 2001:502:7094:0:0:0:0:30
nserver:       K.EDU-SERVERS.NET 192.52.178.30 2001:503:d2d:0:0:0:0:30
nserver:       L.EDU-SERVERS.NET 192.41.162.30 2001:500:d937:0:0:0:0:30
nserver:       M.EDU-SERVERS.NET 192.55.83.30 2001:501:b1f9:0:0:0:0:30
ds-rdata:      28065 8 2 4172496CDE85534E51129040355BD04B1FCFEBAE996DFDDE652006F6F8B2CE76

whois:         whois.educause.edu

status:        ACTIVE
remarks:       Registration information:
remarks:       http://www.educause.edu/edudomain

created:       1985-01-01
changed:       2020-12-10
source:        IANA

# whois.educause.edu
```

```
This Registry database contains ONLY .EDU domains.
The data in the EDUCAUSE Whois database is provided
by EDUCAUSE for information purposes in order to
assist in the process of obtaining information about
or related to .edu domain registration records.

The EDUCAUSE Whois database is authoritative for the
.EDU domain.

A Web interface for the .EDU EDUCAUSE Whois Server is
available at: http://whois.educause.edu

By submitting a Whois query, you agree that this information
will not be used to allow, enable, or otherwise support
the transmission of unsolicited commercial advertising or
solicitations via e-mail.  The use of electronic processes to
harvest information from this server is generally prohibited
except as reasonably necessary to register or modify .edu
domain names.

-------------------------------------------------------------

Domain Name: UMD.EDU

Registrant:
        University of Maryland
        Division of Information Technology
        Bldg 224, Room 3309-C
        College Park, MD 20742
        USA

Administrative Contact:
        Domain Admin
        University of Maryland
        Division of Information Technology
        Bldg 224, Room 3309-C
        College Park, MD 20742-2411
        USA
        +1.3014053003
        dnsadmin@noc.umd.edu

Technical Contact:

        University of Maryland
        Office of Information Technology
        Network Operations Center
        College Park, MD 20742
        USA
        +1.3014053003
        dnstech@noc.umd.edu
```

```
Name Servers:
        NS1.UMD.EDU
        NS2.UMD.EDU
        NS.UMS.EDU

Domain record activated:    31-Jul-1985
Domain record last updated: 26-Sep-2022
Domain expires:             31-Jul-2023
```

This is very long, and includes both the top-level (**`EDU-DOM`**) registration and a recursive lookup in **`EDUCAUSE`**'s system. In both cases, we get DNS resolvers (*name servers*) and points of contact. It has become increasingly common for some of this information to be hidden, particularly for smaller domains.

`whois` can also tell us the CIDR block for an address, and who owns that block:

```
▯  14:26:04 389Z $ whois 128.8.127.4
% IANA WHOIS server
% for more information on IANA, visit http://www.iana.org
% This query returned 1 object

refer:        whois.arin.net

inetnum:      128.0.0.0 - 128.255.255.255
organisation: Administered by ARIN
status:       LEGACY

whois:        whois.arin.net

changed:      1993-05
source:       IANA

# whois.arin.net

NetRange:       128.8.0.0 - 128.8.255.255
CIDR:           128.8.0.0/16
NetName:        UMDNET-1
NetHandle:      NET-128-8-0-0-1
Parent:         NET128 (NET-128-0-0-0-0)
NetType:        Direct Allocation
OriginAS:       AS27
Organization:   University of Maryland (UNIVER-262-Z)
RegDate:        1984-08-01
Updated:        2021-12-14
Ref:            https://rdap.arin.net/registry/ip/128.8.0.0
```

```
OrgName:        University of Maryland
OrgId:          UNIVER-262-Z
Address:        Office of Information Technology
Address:        Patuxent Building
City:           College Park
StateProv:      MD
PostalCode:     20742
Country:        US
RegDate:        2010-01-06
Updated:        2010-01-06
Ref:            https://rdap.arin.net/registry/entity/UNIVER-262-Z


OrgTechHandle: UM-ORG-ARIN
OrgTechName:   UMD DNS Admin Role Account
OrgTechPhone:  +1-301-405-9955
OrgTechEmail:  dnsadmin@noc.net.umd.edu
OrgTechRef:    https://rdap.arin.net/registry/entity/UM-ORG-ARIN

OrgAbuseHandle: UARA-ARIN
OrgAbuseName:   UMD Abuse Role Account
OrgAbusePhone:  +1-301-405-8787
OrgAbuseEmail:  abuse@umd.edu
OrgAbuseRef:    https://rdap.arin.net/registry/entity/UARA-ARIN

RAbuseHandle: UARA-ARIN
RAbuseName:   UMD Abuse Role Account
RAbusePhone:  +1-301-405-8787
RAbuseEmail:  abuse@umd.edu
RAbuseRef:    https://rdap.arin.net/registry/entity/UARA-ARIN

RNOCHandle: UM-ORG-ARIN
RNOCName:   UMD DNS Admin Role Account
RNOCPhone:  +1-301-405-9955
RNOCEmail:  dnsadmin@noc.net.umd.edu
RNOCRef:    https://rdap.arin.net/registry/entity/UM-ORG-ARIN

RTechHandle: UM-ORG-ARIN
RTechName:   UMD DNS Admin Role Account
RTechPhone:  +1-301-405-9955
RTechEmail:  dnsadmin@noc.net.umd.edu
RTechRef:    https://rdap.arin.net/registry/entity/UM-ORG-ARIN
```

4.3 Layers 2 and 3

Consider the routing table we saw before:

```
crow@475283dd6ec2:/$ ip route
default via 172.17.0.1 dev eth0
172.17.0.0/16 dev eth0 proto kernel scope link src 172.17.0.2
```

This tells us that all of our outgoing packets should, by default, be sent through the router with address **`172.17.0.1`**. When we are assembling the ethernet frame, we need to provide a MAC address, but that is not the information we have. How do we get the MAC address for an IP address on our local network?

The *Address Resolution Protocol* (ARP) allows us to learn these mappings, just as DNS allows us to learn the mappings between FQDNs and IP addresses. It works by broadcasting a message that says, “Who has this IP address I’m looking for? Please respond to me.” This is flooded through the local network until it reaches the host with that address, which then responds with its MAC address. The requesting host caches this information so that it does not need to send another ARP request for this IP address until the cache entry expires.

We can examine the current ARP cache with the **`arp`** program:

```
crow@9802b196582a:/$ arp
Address                  HWtype  HWaddress           Flags Mask            Iface
172.17.0.1               ether   02:42:ff:bc:4d:47   C                     eth0
```

This tells us the the address **`172.17.0.1`** corresponds to an ethernet address **`02:42:ff:bc:4d:47`**, and that we can reach this through the **`eth0`** device. There are additional arguments to perform a lookup, manually add entries, or delete entries. The ARP table is maintained in the file **`/proc/net/arp`**, which you can view using **`cat`**, but you should **never modify this file** manually.

4.4 Scanning for Hosts and Ports

If you are going to scan a network, you must have the permission of:

- ***The network owner***
- ***The owners of all hosts connected to the network***
- ***All of the people using the network***

Scanning a network without permission is a serious invasion of privacy, and in many cases might be illegal. If you just want to play with the tools we are looking at in this section, you can create a testbed using the commands from Section 3.7 and run them on that.

The **`nmap`** (https://nmap.org) program scans a host or subnet for open ports. If it is able to connect to the port, it also attempts to identify the host's operating system, service, and version. There are a lot of options, but we will only look at a few.

The simplest scan you can do is:

```
crow@9802b196582a:/$ nmap scanme.nmap.org
Starting Nmap 7.80 ( https://nmap.org ) at 2022-11-10 17:05 UTC
Nmap scan report for scanme.nmap.org (45.33.32.156)
Host is up (0.083s latency).
Other addresses for scanme.nmap.org (not scanned): 2600:3c01::f03c:91ff:fe18:bb2f
Not shown: 996 closed ports
PORT      STATE SERVICE
22/tcp    open  ssh
80/tcp    open  http
9929/tcp  open  nping-echo
31337/tcp open  Elite

Nmap done: 1 IP address (1 host up) scanned in 1.54 seconds
```

Here we see four TCP ports listening, with not much additional information provided.

We can add the **-A** flag gives us considerably more information:

```
crow@9802b196582a:/$ nmap -A scanme.nmap.org
Starting Nmap 7.80 ( https://nmap.org ) at 2022-11-10 17:07 UTC
Nmap scan report for scanme.nmap.org (45.33.32.156)
Host is up (0.094s latency).
Other addresses for scanme.nmap.org (not scanned): 2600:3c01::f03c:91ff:fe18:bb2f
Not shown: 996 closed ports
PORT      STATE SERVICE    VERSION
22/tcp    open  ssh        OpenSSH 6.6.1p1 Ubuntu 2ubuntu2.13 (Ubuntu Linux; protocol 2.0)
| ssh-hostkey:
|   1024 ac:00:a0:1a:82:ff:cc:55:99:dc:67:2b:34:97:6b:75 (DSA)
|   2048 20:3d:2d:44:62:2a:b0:5a:9d:b5:b3:05:14:c2:a6:b2 (RSA)
|   256 96:02:bb:5e:57:54:1c:4e:45:2f:56:4c:4a:24:b2:57 (ECDSA)
|_  256 33:fa:91:0f:e0:e1:7b:1f:6d:05:a2:b0:f1:54:41:56 (ED25519)
80/tcp    open  http       Apache httpd 2.4.7 ((Ubuntu))
|_http-server-header: Apache/2.4.7 (Ubuntu)
|_http-title: Go ahead and ScanMe!
9929/tcp  open  nping-echo Nping echo
31337/tcp open  tcpwrapped
Service Info: OS: Linux; CPE: cpe:/o:linux:linux_kernel

Service detection performed. Please report any incorrect results at https://nmap.org/submit/ .
Nmap done: 1 IP address (1 host up) scanned in 14.65 seconds
```

Now we are getting specifics about the actual programs and versions that **nmap** was able to learn. We also have an inferred operating system.

We can scan multiple hosts, which might be listed as separate command-line options or as CIDR blocks (which might take a while):

```
crow@9802b196582a:/$ nmap -T5 192.168.1.0/24
Starting Nmap 7.80 ( https://nmap.org ) at 2022-11-10 17:46 UTC
Warning: 192.168.1.156 giving up on port because retransmission cap hit (2).
Nmap scan report for 192.168.1.0
Host is up (0.014s latency).
All 1000 scanned ports on 192.168.1.0 are closed

Nmap scan report for Fios_Quantum_Gateway.fios-router.home (192.168.1.1)
Host is up (0.017s latency).
Not shown: 992 closed ports
PORT     STATE    SERVICE
22/tcp   filtered ssh
53/tcp   open     domain
80/tcp   open     http
```

```
443/tcp  open     https
4567/tcp filtered tram
8022/tcp filtered oa-system
8080/tcp open     http-proxy
8443/tcp open     https-alt

Nmap scan report for Mikes-MBP.fios-router.home (192.168.1.156)
Host is up (0.0046s latency).
Not shown: 999 closed ports
PORT     STATE    SERVICE
7937/tcp filtered nsrexecd

Nmap scan report for 192.168.1.184
Host is up (0.023s latency).
Not shown: 997 closed ports
PORT     STATE SERVICE
22/tcp   open  ssh
443/tcp  open  https
5000/tcp open  upnp

Nmap scan report for celephais.fios-router.home (192.168.1.195)
Host is up (0.037s latency).
Not shown: 999 closed ports
PORT     STATE SERVICE
4000/tcp open  remoteanything

Nmap scan report for WyzeCam.fios-router.home (192.168.1.201)
Host is up (0.015s latency).
All 1000 scanned ports on WyzeCam.fios-router.home (192.168.1.201) are closed

Nmap scan report for WyzeCam.fios-router.home (192.168.1.202)
Host is up (0.014s latency).
All 1000 scanned ports on WyzeCam.fios-router.home (192.168.1.202) are closed

Nmap scan report for 192.168.1.205
Host is up (0.0068s latency).
Not shown: 997 closed ports
PORT     STATE SERVICE
53/tcp   open  domain
80/tcp   open  http
7777/tcp open  cbt

Nmap scan report for Pixel-6a.fios-router.home (192.168.1.211)
Host is up (0.017s latency).
All 1000 scanned ports on Pixel-6a.fios-router.home (192.168.1.211) are closed

Nmap scan report for 192.168.1.255
Host is up (0.033s latency).
All 1000 scanned ports on 192.168.1.255 are closed

Nmap done: 256 IP addresses (10 hosts up) scanned in 10.10 seconds
```

This is my home network. We have omitted the **-A** option, but added **-T5**, which sets it to do the fastest scan it can. We see my work laptop, my personal laptop, my router, two webcams, the subnet broadcast address, and some other stuff.

What is this other stuff on my network?

```
crow@9802b196582a:/$ nmap -A 192.168.1.184 192.168.1.205
Starting Nmap 7.80 ( https://nmap.org ) at 2022-11-10 18:02 UTC
Nmap scan report for 192.168.1.184
Host is up (0.011s latency).
Not shown: 997 closed ports
PORT     STATE SERVICE  VERSION
22/tcp   open  ssh      OpenSSH 7.1 (protocol 2.0)
| ssh-hostkey:
|   2048 06:ff:63:a5:ca:87:fe:4b:df:50:19:1d:8a:05:85:66 (RSA)
|   256 3b:24:08:80:33:08:52:66:2b:8f:a8:d0:a1:ee:fb:7c (ECDSA)
|_  256 c6:49:68:73:23:60:aa:e4:57:ab:6e:52:6f:14:2c:98 (ED25519)
443/tcp  open  ssl/http Neato Botvac Connected
|_http-title: Site doesn't have a title.
| ssl-cert: Subject: commonName=Lennox/organizationName=Lennox International Inc./stateOrProvin
| Not valid before: 2016-11-07T12:22:46
|_Not valid after:  2116-10-14T12:22:46
|_ssl-date: TLS randomness does not represent time
| sslv2:
|   SSLv2 supported
|_  ciphers: none
5000/tcp open  upnp?
Service Info: Device: specialized

Nmap scan report for 192.168.1.205
Host is up (0.068s latency).
Not shown: 997 closed ports
PORT     STATE SERVICE VERSION
53/tcp   open  domain  ISC BIND 9.11.35
| dns-nsid:
|_  bind.version: 9.11.35
80/tcp   open  http    Arris TG862G http config
| http-auth:
| HTTP/1.1 401 Unauthorized\x0D
|_  Basic realm=NETGEAR WAC104
|_http-title:  Authorization warning
7777/tcp open  upnp    MiniUPnP 1.6 (Netgear SDK 4.2.0.0; UPnP 1.0)
Service Info: Device: WAP; CPE: cpe:/h:arris:tg862g

Service detection performed. Please report any incorrect results at https://nmap.org/submit/ .
Nmap done: 2 IP addresses (2 hosts up) scanned in 179.92 seconds
```

By adding `-A` for just the two unknown hosts, we see more detail. One of the hosts is my WiFi access point (a layer 2 device), and the other is presumably the controller for my furnace and air conditioner (which allows `ssh` connections, for some reason).

Chapter 5

Network Programming

Most programming languages have some way of interacting with the network, with varying degrees of abstraction. We will focus on using C, because it exposes the network connections with considerably more detail than other languages, so you will become more familiar with these details by writing networking code in C. Once you are familiar with network programming in C, transferring that knowledge to another language that provides a "friendlier" abstraction will be much easier.

5.1 Socket Programming

A socket is an abstraction used by the operating system to treat one end of an inter-process communications link as a file. Like other files, a process can read from or write to a socket. Unlike other files, a socket is specifically bound to a single process, so only that process has access to the socket.

We commonly have three types of sockets: stream, datagram, and raw. Stream sockets instantiate the TCP protocol. Datagram sockets instantiate the UDP protocol. Raw sockets do not have a layer 4 protocol abstraction, so it is up to the application to handle any necessary layer 4 processing. Creating raw sockets is a protected operation, and can only be done by a user with root privileges. You will rarely work with raw sockets, and then only for very low-level operations. As an application developer, you will mostly use TCP and UDP, as discussed in Section 3.1.

5.1.1 Creating a socket

The programming interface for sockets in C is defined in the header file **`sys/socket.h`** (or files included by it). You include it as you would any header file:

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

As mentioned previously, Linux (and many other operating systems) treat a socket as a special kind of file. That means creating a socket is a similar operation to opening a file, with the basic function being

```
int sockfd = socket(PF_INET, SOCK_STREAM, 0);
```

The arguments to **`socket()`** tell the kernel the desired *protocol family* (**`PF_INET`** is IPv4), *socket type* (**`SOCK_STREAM`** is TCP), and *protocol* (**`0`** for TCP, because there is only one protocol supported for this type of socket). For **`SOCK_DGRAM`** (UDP), we would also specify **`0`** for the protocol. There are other, less common, socket types that might require a protocol. See the documentation for details if you are curious.

The value returned by **`socket()`** is a *file descriptor*. This is just an integer, much like the file descriptors for standard input (0), standard output (1), and standard error (2). On failure, **`socket()`** instead returns **`-1`**, so you should **always** check the returned value. If **`socket()`** returns **`-1`**, then it will also set the

`errno` global variable to indicate the cause of the failure. See the documentation for a list of these, but often a failed socket creation is a fatal error for a program.

The documentation might show **`AF_INET`** in the place of **`PF_INET`**, depending on where you look. These constants should have the same value, but technically **`AF_INET`** refers to the IPv4 *address family*, rather than the protocol family. It is largely a matter of style which you use, but we will use **`PF_INET`** when specifying a protocol family and **`AF_INET`** when specifying an address family, just to keep these concepts clear and separate.

5.1.2 Socket options

When created, a socket has some default set of options, depending on the type of socket and protocol. Often, these defaults are appropriate, but sometimes you will need to change them for your program to work as intended. We can change these options with **`setsockopt()`**, as illustrated here:

```
int opt = 1;
setsockopt(sockfd, SOL_SOCKET, SO_REUSEADDR, &opt, sizeof(opt));
```

Here the **`sockfd`** is what was returned by **`socket()`**. **`SOL_SOCKET`** indicates that we are setting a socket-level (as opposed to protocol-level) option. The specific option we are setting is **`SO_REUSEADDR`**, which controls whether we want the operating system to reuse the associated port before the standard timer expires. This is somewhat dangerous in certain circumstances, because you are allowing a new process to claim your port number after you exit, meaning they can potentially receive traffic that was intended for you. The last two arguments are the option itself, and how many bytes it is. Note that we are providing a pointer, rather than just the literal number **`1`**. We have to do this because not all options take an integer. Some might actually take arrays or structures, so it is important to let **`setsockopt()`** know how much data we are providing for the option.

We mentioned that **`SOL_SOCKET`** is for socket-level options. We can also specify options for TCP (**`IPPROTO_TCP`**) or UDP (**`IPPROTO_UDP`**), if we want to adjust the behavior of those. If you are writing an interactive TCP program, for instance, you might want to specify the **`TCP_NODELAY`** option to disable TCP's buffering of small chunks of data. It is *extremely* unlikely that you will want or need to do this, however, and you should definitely take a course on computer networking (such as CMSC417) before using this option, so that you know the implications of disabling the delays.

5.1.3 Socket addresses

At this point, we have created a socket, but we cannot do anything with it yet. We have two choices:

1. We can wait for someone to connect to us, or
2. We can try to connect to someone else.

Either way, we need to add a *socket address* to the socket. For the first case (a *server*), this determines who is able to connect to us. For the second case (a *client*), this specifies the server we are trying to contact.

There are two structures we will look at: **`sockaddr`** and **`sockaddr_in`**. Why do we have two? Actually, we have many more, but these are the only two we will worry about, and the others should be fairly clear from these. In simple terms, **`sockaddr`** is a generic socket address, akin to a base class in an object-oriented language; while **`sockaddr_in`** is a socket address specific to the IPv4 protocol (**`in`** for "internet"), akin to a subclass of **`sockaddr`**. Functions that *take* a socket address generally take a pointer to a **`sockaddr`**, but if we want to *set* or *read* specific fields, we need to use a structure like **`sockaddr_in`**.

Many of the fields in **sockaddr_in** will be 0, so we generally begin by making *everything* 0, and then setting what we need:

```
struct sockaddr_in server_addr;
memset(&server_addr, 0, sizeof(server_addr));
```

The **memset()** function allows you to set all of the bytes in some structure (treated as a simple byte array) to a single value. In this case, that value is **0**. By passing our **server_addr** with an ampersand, **memset()** is receiving it as an address; that is, as a pointer (a **void*** here, more specifically). Because **memset()** knows nothing about the destination other than that it's a pointer, we have to tell it how many bytes to overwrite with **0**, which is why we pass **sizeof(server_addr)** as the last argument. This is the size of the actual structure, in bytes.

Now that we have an empty **server_addr**, it is time to set the fields that matter. There are generally three that we care about:

- **server_addr.sin_family** is the address family. For a **sockaddr_in**, this will usually be **AF_INET** (the IPv4 address family).
- **server_addr.sin_port** is the port on which we will listen for connections. If this were a client, this would be the port on the server to connect to.
- **server_addr.sin_addr.s_addr** is the relevant IP address.

  For a server, this will generally be **INADDR_ANY** (to listen for connections from any host on the network) or **INADDR_LOOPBACK** (to only listen for connections coming from this host over the loopback device). We *can*, however, specify the IP address bound to a specific device to allow connections from outside of this host, but only on that device. This is useful for *multi-homed* servers, which are connected to two or more layer 2 networks.

  For a client, this would be the address of the server we are connecting to.

An important thing to note is that ports and addresses must be in *network byte order*. This is always big-endian, regardless of the operating system's byte ordering. That means your code should always convert numbers to the appropriate ordering, unless you are guaranteed to have received them in network byte order. We will see examples of this later. For a detailed discussion of byte ordering, see the *General Systems Handbook*.

5.1.4 Servers

A *server* is a process that provides some *service* to client processes via the network. This might be serving web pages, providing hostname resolution, uploading or downloading files, or just about anything else. There will be slightly different workflows for a server depending on whether it is using TCP or UDP.

TCP Server

We will start with the workflow for a TCP server, as illustrated by the following code:

```c
#include <sys/socket.h>
#include <stdio.h>
#include <stdint.h>
#include <string.h>
#include <endian.h>
#include <unistd.h>
#include <netinet/in.h>
#include <error.h>
#include <errno.h>

int main(int argc, char** argv) {
   // 1. Create a socket.
   int sockfd;
   sockfd = socket(PF_INET, SOCK_STREAM, 0);
   if ( sockfd < 0 ) {
      fprintf(stderr, "Socket creation failed with error %d\n", errno);
      return 1;
   }
```

```
    int e;

    // 2. Bind a port to the socket.
    uint16_t port = 1234;
    struct sockaddr_in server_addr;
    memset(&server_addr, 0, sizeof(server_addr));
    server_addr.sin_family = AF_INET;
    server_addr.sin_port = htobe16(port);
    server_addr.sin_addr.s_addr = INADDR_ANY;
    e = bind(sockfd, (struct sockaddr*)&server_addr, sizeof(server_addr));
    if ( e < 0 ) {
       fprintf(stderr, "Binding port %d failed with error %d\n", port, errno);
       return 1;
    }

    // 3. Configure the socket to listen for new connections.
    e = listen(sockfd, 5);
    if ( e < 0 ) {
       fprintf(stderr, "Listening failed with error %d\n", errno);
       return 1;
    }


    while ( 1 ) {
       // 4. Accept a new connection.
       struct sockaddr_in client_addr;
       socklen_t client_addr_len = sizeof(client_addr);
       int clientfd = accept(sockfd,
                             (struct sockaddr*)(&client_addr),
                             &client_addr_len);
       if ( clientfd < 0 ) {
          fprintf(stderr, "Accept failed with error %d\n", errno);
          return 1;
       }

       // 5. Exchange data with the client on the new socket.
       unsigned char buf[1024];
       int n = recv(clientfd, buf, sizeof(buf), 0);
       if ( n < 0 ) {
          fprintf(stderr, "Reading from client failed with error %d\n", errno);
          return 1;
       }

       // 6. Close the client's socket.
       close(clientfd);
    }
    return 0;
}
```

Let's go through these steps one-by-one:

1. We first create a socket. If you have socket options to set, you will do that here, as well.

2. Next, we specify how this socket should receive new connections, in a process called *binding*. The **`bind()`** function takes a socket file descriptor, a pointer to a **`struct sockaddr`**, and the size of the actual structure pointed to by the second argument. Note that when we create our **`struct sockaddr_in`**, we convert the port number to network byte order with **`htobe16(port)`**.

3. Now we have to tell the socket that it is ready for new connections, with **`listen()`**. This takes the socket file descriptor, as well as a *backlog*, which is how many new connections will be buffered in a queue while waiting for the next step. Additional connections while the queue is full will fail with **`ECONNREFUSED`** (Error: Connection Refused).

4. When a new connection is available, the **`accept()`** function creates a new socket. It also provides the caller with the socket address of the connecting client by filling in the structure pointed to in the second argument. As before, since there can be different types of sockets with different types of socket addresses, we have to provide a pointer to an integer (which is what **`socklen_t`** is) so that **`accept()`** knows how much space is available and can fill it in with the actual size stored.

5. Now the client socket is ready to exchange data with the client. We will explore this in more detail in Section 5.2.

6. Finally, when we are finished with the client, we call **`close()`** on the client's socket file descriptor.

Note that we have error-handling blocks after each of these function calls (except **`close()`**). These are important: There are many reasons why one of these functions might produce an error, and it is important to understand what these error are. Some are fatal (the program should terminate), and some are not. It is possible for **`accept()`** or data transfer to produce a non-fatal error, though they might require terminating the client connections. Some data transfer errors just need to be attempted again later, as we will see.

As written, this code only handles connections from one client at a time. This is usually not what we want, so we need some way to *multiplex* our client connections. The easiest (but extremely limiting) way to do this is to either fork the process or create a new thread for each client. We will see a much better option in Section 5.1.6.

UDP Server

We now look at the workflow for a UDP server. This will be somewhat similar to our TCP server, but there will be differences, due to the fact that TCP is *connection-oriented*, while UDP is *connectionless*. That is, TCP maintains some *state* (data, status, etc.) for a client connection, while UDP does not.

Here is our UDP server code:

```c
#include <sys/socket.h>
#include <stdio.h>
#include <stdint.h>
#include <string.h>
#include <endian.h>
#include <unistd.h>
#include <netinet/in.h>
#include <errno.h>

int main(int argc, char** argv) {
   // 1. Create a socket.
   int sockfd;
   sockfd = socket(PF_INET, SOCK_DGRAM, 0);
   if ( sockfd < 0 ) {
      fprintf(stderr, "Socket creation failed with error %d\n", errno);
      return 1;
   }
   int e;

   // 2. Bind a port to the socket.
   uint16_t port = 1234;
   struct sockaddr_in server_addr;
   memset(&server_addr, 0, sizeof(server_addr));
   server_addr.sin_family = AF_INET;
   server_addr.sin_port = htobe16(port);
   server_addr.sin_addr.s_addr = INADDR_ANY;
   e = bind(sockfd, (struct sockaddr*)&server_addr, sizeof(server_addr));
   if ( e < 0 ) {
      fprintf(stderr, "Binding port %d failed with error %d\n", port, errno);
      return 1;
   }

   while ( 1 ) {
      // 3. Receive data from a client.
      unsigned char buf[1024];
      struct sockaddr_in client_addr;
      struct sockaddr* caddr = (struct sockaddr*)(&client_addr);
      socklen_t client_addr_len = sizeof(client_addr);
      int n;
      n = recvfrom(sockfd, buf, sizeof(buf), 0, caddr, &client_addr_len);
      if ( n < 0 ) {
         fprintf(stderr, "Receiving from client failed with error %d\n", errno);
         return 1;
      }

      // 4. Send a reply to the client.
      n = sendto(sockfd, buf, n, 0, caddr, client_addr_len);
      if ( n < 0 ) {
         fprintf(stderr, "Sending to client failed with error %d\n", errno);
      }
   }
   return 0;
}
```

Aside from changing the type of socket from **SOCK_STREAM** to **SOCK_DGRAM** in the **socket()** call, the first two steps are the same. The biggest change is that we no longer have to **listen()** for and **accept()** connections. Instead, we use **recvfrom()** to receive a *datagram* (a single UDP packet payload) from a client, storing the address of the client so that we can send a reply, if needed. Essentially, we have combined the **accept()** and **recv()** functions, capturing the client's address when we read data, rather than before we are ready to read data. We also show the **sendto()** function, which uses the client address from **`recvfrom()`** to determine its destination. Again, we will examine these functions in greater detail in Section 5.2. Since each client is using the main socket, we do not close the socket when we are done providing service to a client.

5.1.5 Clients

A *client*, at its core, is a process that requests service from a server. This is easiest to understand by example. Here are some clients you might be familiar with:

- Web browser
- Discord app
- Online game app

All of these connect to some remote host, the server, and interact with it. **`host`**, **`dig`**, and **`whois`** are also clients, and we saw netcat operating as both a server and a client.

The biggest difference between a server and a client is that a server is waiting for connections from other hosts (with **`accept()`**), while a client initiates connections to other hosts. As an additional wrinkle, there are some processes that both initiate connections and receive connections. These might be performing different roles (you connect via a browser client to a web server, which connects as a client to a database server), or they might be acting as *peers*. While we will not go into details about peer-to-peer applications, once you understand how clients and servers work, it is a fairly simple extrapolation (though the protocols themselves can be somewhat complex).

TCP Client

Again, we begin with the TCP version of the client. Here is our code:

```c
#include <sys/socket.h>
#include <stdio.h>
#include <stdint.h>
#include <string.h>
#include <unistd.h>
#include <endian.h>
#include <netdb.h>
#include <errno.h>

int main(int argc, char** argv) {
   // 1. Read the server address and port from the command line.
   if ( argc < 3 ) {
      fprintf(stderr, "Usage: %s <server> <port>\n",argv[0]);
      return 1;
   }

   struct addrinfo hints;
   struct addrinfo* addrlist;
   memset(&hints, 0, sizeof(hints));
   hints.ai_family = PF_UNSPEC;
   hints.ai_socktype = SOCK_STREAM;
   if ( getaddrinfo(argv[1], argv[2], &hints, &addrlist) ) {
      fprintf(stderr, "Error looking up %s:%s\n",argv[1],argv[2]);
      return 1;
   }
   if ( NULL == addrlist ) {
      fprintf(stderr, "No address found for %s:%s\n",argv[1],argv[2]);
      return 1;
   }

   // 2. Create a socket.
   int sockfd;
   sockfd = socket(addrlist->ai_family,
                   addrlist->ai_socktype,
                   addrlist->ai_protocol);
   if ( sockfd < 0 ) {
      fprintf(stderr, "Socket creation failed with error %d\n", errno);
      return 1;
   }

   int e;

   // 3. Connect to the server.
   e = connect(sockfd, addrlist->ai_addr, addrlist->ai_addrlen);
   if ( e < 0 ) {
      fprintf(stderr, "Could not connect\n");
      return 1;
   }

   // 4. Exchange data with the server.
   int n = send(sockfd, "hello", 6, 0);
   if ( n < 0 ) {
      fprintf(stderr, "Sending to server failed with error %d\n", errno);
      return 1;
   }

   // 5. Close the socket.
   close(sockfd);

   return 0;
}
```

Here we see a rather different workflow than we had in our server:

1. First, we take the command-line arguments and use them to obtain the **`struct sockaddr*`** for the server. **`argv[0]`** is always the process name, so here we assume **`argv[1]`** holds the hostname, and **`argv[2]`** holds the port. We are using **`getaddrinfo()`**, and letting it figure out the appropriate address type, limiting ourselves to TCP sockets (**`SOCK_STREAM`**). Note that this code will work whether the server is running IPv4 or IPv6, since we have left the protocol family unspecified (**`PF_UNSPEC`**). This is a good idea in your client code, and will make your code more portable to different network environments. **`getaddrinfo()`** returns 0 on success, and creates a linked list in its last argument, which is why we are passing a pointer to a pointer (this is a very common design).

2. Next, we create a socket to the server. Here we rely on what **`getaddrinfo()`** has returned to us. Since we really only expect one result, and only care about out, we ignore the linked-list nature of **`addrlist`**, and just use the first entry. You could create a loop over the linked list, if you expect multiple results, only some of which might respond.

3. For our network functions, this is the first major difference between the

client and server. Where the server calls **bind()**, **listen()**, and **accept()**, the client just calls **connect()**. This takes a socket address pointer and the length of the structure, and again we can get these directly from **addrlist**.

4. As before, we now are ready to exchange data with the server. Here we see the **send()** function, which is the complement of the **recv()** in the server.

5. When we are done with the socket, we again call **close()** on it.

UDP Client

Now we will take a look at the UDP version of our client:

```
#include <sys/socket.h>
#include <stdio.h>
#include <stdint.h>
#include <string.h>
#include <endian.h>
#include <netdb.h>
#include <errno.h>
#include <unistd.h>

int main(int argc, char** argv) {
   // 1. Read the server address and port from the command line.
   if ( argc < 3 ) {
      fprintf(stderr, "Usage: %s <server> <port>\n",argv[0]);
      return 1;
   }

   struct addrinfo hints;
   struct addrinfo* addrlist;
   memset(&hints, 0, sizeof(hints));
   hints.ai_family = PF_UNSPEC;
   hints.ai_socktype = SOCK_DGRAM;
   if ( getaddrinfo(argv[1], argv[2], &hints, &addrlist) ) {
      fprintf(stderr, "Error looking up %s:%s\n",argv[1],argv[2]);
      return 1;
   }
   if ( NULL == addrlist ) {
      fprintf(stderr, "No address found for %s:%s\n",argv[1],argv[2]);
```

```
        return 1;
    }

    // 2. Create a socket.
    int sockfd;
    sockfd = socket(addrlist->ai_family,
                    addrlist->ai_socktype,
                    addrlist->ai_protocol);
    if ( sockfd < 0 ) {
        fprintf(stderr, "Socket creation failed with error %d\n", errno);
        return 1;
    }

    // 3. Exchange data with the server.
    int n = sendto(sockfd, "hello", 6, 0,
                   addrlist->ai_addr,
                   addrlist->ai_addrlen);
    if ( n < 0 ) {
        fprintf(stderr, "Sending to server failed with error %d\n", errno);
        return 1;
    }

    // 4. Close the socket.
    close(sockfd);

    return 0;
}
```

This is almost identical to the TCP client. Instead of **`SOCK_STREAM`**, we specify **`SOCK_DGRAM`** for the socket type in the hints to **`getaddrinfo()`**. We remove the call to **`connect()`**, since UDP is connectionless. We also replace the call to **`send()`** with **`sendto()`**, which now includes the server's address info.

5.1.6 Efficient Multiplexing

As previously mentioned, we *can* multiplex connections within a server using separate threads or processes for each client. This ends up wasting resources, however, and switching between threads may be fast, but it is not instantaneous.

A better solution is to make our sockets *non-blocking*, and use either **`select()`** or **`poll()`** to tell us when they are ready for reading or writing. We can make a socket non-blocking with:

```
#include <fcntl.h>
fcntl(sockfd, F_SETFL, O_NONBLOCK);
```

This will mostly apply to TCP, but we can also use these techniques if we have multiple UDP ports open, a mix of TCP and UDP, or if we are also accepting input from standard input (file descriptor 0).

When we call **`send()`** or **`recv()`**, and there is not yet enough data to read, the functions will usually block. That is, they will not return until they have completed the request. By making the socket non-blocking, these functions will instead return -1 (indicating an error), and set **`errno`** to **`EAGAIN`** or **`EWOULDBLOCK`** (which might, and often will, be same value).

As we mentioned above, we can use **`select()`** or **`poll()`** to tell us when a socket is ready for a particular operation (typically reading or writing). These functions are similar, but have (for us) essentially the same functionality. They block until:

- a file descriptor is ready, or
- a timeout expires, or
- an error occurs.

The main difference (again, for us), is how the file descriptors and timeout are provided.

In both cases, we will modify the server workflow from what we had in Section 5.1.4:

- If we are reading from the listening socket, we will call **`accept`** and add the client information to the set of file descriptors to watch. (step 4)

- If we are reading from or writing to a client socket, we will exchange as much data as we safely can without blocking. (step 5)
- If a **send** or **recv** on a client socket returns **0**, we will close the socket. (step 6)

Select

For **select()**, we specify three *file descriptor sets*, any of which might be **NULL**, and a pointer to a **timeval** with the timeout information, which might also be **NULL**. Here is some (incomplete) example code:

```
#include <sys/select.h>
#include <sys/time.h>

fd_set read_set, write_set, err_set;
timeval timeout;
timeout.tv_sec = 1;
timeout.tv_usec = 0;

while(1) {
   FD_ZERO(&read_set);
   FD_ZERO(&write_set);
   FD_ZERO(&err_set);
   FD_SET(sockfd, &read_set);

   int nfd = select(sockfd+1, &read_set, &write_set, &err_set, &timeout);
   if ( 0 < nfd ) {
      if ( FD_ISSET(sockfd, &read_set) ) {
         ...
      }
   } else if ( -1 == nfd ) { ... }
}
```

fd_set is basically a bitmap of possible file descriptors, and we operate on it with a set of macros **FD_ZERO**, **FD_SET**, and **FD_ISSET**, all of which expect a pointer to an **fd_set**. We *must* clear each set before calling select, which we do with **FD_ZERO**. Then we specify which file descriptors to watch in a set with **`FD_SET`**. Once **`select()`** returns, we can test if a particular descriptor has been set with **`FD_ISSET`**.

We have to tell **`select()`** how many file descriptors it should be looking at. Since **`fd_set`** is a bitmap, this number is the greatest file descriptor number, plus 1. That is, it is the effective length of the bitmap that includes all of our file descriptors. This is the first argument to **`select()`**, followed by the read set, the write set, the error set, and the timeout. The return value from **`select()`** is the number of file descriptors with something matching from any of the sets, or -1 if there was an error.

`timeval` has two fields, specifying number of seconds (**`tv_sec`**) and microseconds (**`tv_usec`**). If the timeout is 0 (in both fields), **`select()`** will always return immediately. If it is **`NULL`**, then **`select()`** will not return until one of the specified file descriptor sets is ready.

What does “ready” mean? For the read set, it means there is a file descriptor in that set that currently has data we can read. This is the most common case (aside from timeouts). For the write set, “ready” means that a file descriptor in that set is ready to receive more data from us. This might arise when we attempted to write to a socket, but it had no more room in its buffer. Once the buffer has cleared sufficiently, it will return as ready for writing. For the error set, “ready” means there was an out-of-band error that we cannot detect by reading or writing, so we need to check for it. This might be due to the socket being closed.

While it is common for the timeout to be **`NULL`** (often with only the read set non-**`NULL`**), there are times when everything *except* the timeout is **`NULL`**. Why would we do this? The reason is that C provides us with a **`sleep()`** function, which takes an integer number of seconds. If we want to sleep for less than a second, or some other fractional value, we need a different mechanism, and **`select()`** provides us with this. For example, to sleep for half a second we could do:

```
timeval timeout;
timeout.tv_sec = 0;
timeout.tv_usec = 500000;
select(0,NULL,NULL,NULL,&timeout);
```

Because we have not told **`select()`** to monitor any file descriptors, this will wait until the timeout expires, doing nothing else until then.

Poll

The **`poll()`** function scales better than **`select()`**, though for most applications this will not make a substantial difference. Rather than a set of bitmaps, it takes an array of **`struct pollfd`**, and this structure contains the relevant information, including the file descriptor and what events to watch for. Rather than a **`timeval`**, the timeout is specified as an integer holding the number of milliseconds (**`-1`** means to wait indefinitely). The following is the **`poll()`** equivalent of our **`select()`** example:

```
#include <poll.h>
#include <limits.h>

struct pollfd fdarray[FOPEN_MAX];
unsigned long nfds = 0;
int timeout = 1000; // 1 second
fdarray[0].fd = sockfd;
fdarray[0].events = POLLIN | POLLOUT | POLLERR;
nfds++;

while (1) {
   int nfd = poll(fdarray, nfds, timeout);
   if ( 0 < nfd ) {
      for ( int i = 0 ; i < nfds ; ++i ) {
         if ( fdarray[i].revents & POLLIN ) { ... }
      }
   } else if ( -1 == nfd ) { ... }
}
```

`FOPEN_MAX` is a guaranteed minimum to the number of file descriptors that a process can open simultaneously. Usually, the actual limit will be higher than this. `struct polldf` has three fields, the file descriptor (`fd`), a bitmap of flags indicating the events to poll for (`events`), and a bitmap of flags indicating what events were detected (`revents`). Setting `POLLIN` is equivalent to putting the file descriptor in `select()`'s read set, `POLLOUT` is the equivalent of the write set, and `POLLERR` is the equivalent of the error set.

As with `select()`, we have to tell `poll()` how much of the array to consider. Unlike `select()`, this only needs to be the actual number of file descriptors we want to watch, since the array can be densely packed from the beginning. We can also temporarily disable a file descriptor by negating the `fd` field, since negative numbers are ignored. Another nice feature is that, because the `events` fields is unmodified by `poll()`, we do not have to clear and re-set the entries for each call. Instead, we just do a bitwise AND for the flags of interest. As with `select()`, the return value of `poll()` is the number of file descriptors with events returned, or -1 if there is an error.

Additional details

With TCP, you typically have one file descriptor that is listening for new connections. When it is ready for reading, that means a new connection has been received, requiring a call to `accept()`. The resulting client file descriptor should then be added to `read_set` or `fdarray`, depending on whether you are using `select()` or `poll()`. It is important to *only* attempt to read from or write to a client socket.

When a client disconnects, with `select()` you would no longer add its file descriptor to any of the `fd_set`s, while for `poll()` you would set the fields of its `struct pollfd` to 0, or just the `fd` field to a negative number. These entries in the array can then be reused for the next new connection.

You should *always* check to see if a client has disconnected or an error has occurred. Otherwise, you are likely to get strange behavior, and possibly segmentation faults. If you are storing state for clients, make sure you clean this up — free any dynamically allocated memory and make sure that you are no longer looking for socket events from it.

5.2 Data Transfer

In this section, we will explore how to read and write data through sockets, and some ways to format that data.

5.2.1 Reading and Writing

We interact with a socket through a file descriptor, which means the **`read()`** and **`write()`** functions will work. There are much better options, however, which you should use instead, and which we have seen in our example code.

TCP

To read from a TCP socket, we use the **`recv()`** (receive) function; to write to a socket, we use the **`send()`** function. Both of these have almost identical signatures:

1. The first argument is the socket file descriptor.
2. The second argument is a buffer, which is an array of bytes.
3. The third argument is the number of bytes to receive or send.
4. The fourth argument is a bitmap of flags, which will usually be 0.

The return value is the number of bytes received or sent, 0 if the socket has been closed, or -1 if there was an error (in which case `errno` will be set). You should *always* check the return value, and not assume that data was received or sent successfully or completely.

Because TCP is a stream protocol, you do not have to receive or send an entire message at once. Instead, you can work with smaller amounts of data. Note that this is considerably less efficient when sending than when receiving. When receiving, you will read as many bytes as requested from the socket's buffer. When sending, if you write a few bytes, they might be sent immediately, or they might be buffered into a larger packet.

Since the buffer is of type `void*`, we can provide any pointer to `recv()` or `send()`. These might be `char` arrays (strings), `unsigned char` arrays (essentially byte arrays), pointers to integers, pointers to `struct`s, etc. We will take a closer look at some of these options later.

UDP

UDP is a datagram protocol, which means that a message must be contained entirely within a single packet. It is also connectionless, so a single socket can service multiple clients (or servers). That means two things:

1. When we read from a UDP socket, we must read the entire packet — any unread data is **discarded**.
2. When we read from or write to a UDP socket, we must include structures containing information about the other end of the connection. When reading, this structure will be filled in for us.

Since `recv()` and `send()` do not provide the required fields, we instead use two new functions: `recvfrom()` and `sendto()`. These functions are very similar to **`recv()`** and **`send()`**, but they include a pointer to a socket address and an address length as the fifth and sixth arguments. In the case of **`recvfrom()`**, the address length must be a pointer, so that we can store the actual length of the structure. Before calling, it must be the size of the actual structure (such as a **`struct sockaddr_in`**) to which we're passing a pointer.

5.2.2 Formatting Data

There are many ways to format data. We could use a string-based format (like XML or JSON), a flexible binary format (like ASN.1), or other methods. We are going to do something simpler, however.

Integers and Strings

Say we have an integer, **`i`**, and we have defined it as a 4-byte value:

```
#include <stdint.h>

uint32_t i = 12;
```

Since a 32-bit integer is stored in memory as four contiguous bytes, we could send this with something like:

```
int n = send(sockfd,&i,4,0);
```

Here we have passed a pointer to **`i`** as the **`void*`** from which **`send()`** will read, and we are sending four bytes from that initial address. This will work, but it is not *quite* correct.

The problem is that we have not considered *byte-order*. To help us understand what is happening, we will consider a new type that is defined as a **`union`**:

```
typedef union {
   uint32_t i;
   uint8_t  b[4];
} i32b;
```

Because this is a **`union`**, the fields **`i`** and **`b`** *overlap* in memory. That is, the address of the field **`i`** is the same as the address of the element **`b[0]`**. Because the fields have the same lengths, all of the bytes of **`i`** are elements of **`b`**, and vice-versa. We now change our code to:

```
i32b v;
v.i = 12;
n = send(sockfd,&(v.i),4,0);
```

Say we are on a big-endian system. The **`uint32_t`** value **`12`** corresponds to

```
v.b[0] = 0;
v.b[1] = 0;
v.b[2] = 0;
v.b[3] = 12;
```

Now say we are on a little-endian system. That **`12`** now corresponds to

```
v.b[0] = 12;
v.b[1] = 0;
v.b[2] = 0;
v.b[3] = 0;
```

These are *very* different when sent to a remote host.

How do the hosts know what to send and how to interpret what they receive? They have to agree on some standard. Almost always, this is big-endian, and we refer to big-endian as *network byte order*. If we expect our code to be portable (which we absolutely should), then we have to convert between network byte order and *host byte order*:

```
#include <stdint.h>
#include <sys/types.h>

uint32_t i = htobe32(12);
int n = send(sockfd,&i,4,0);
```

Here the **htobe32()** function converts a 32-bit integer in *host* byte order to a 32-bit integer in *big endian* (ie, network) byte order. The reverse of this is **be32toh()**. These are frequently implemented as preprocessor macros, rather than actual C functions, so there is no extra cost to calling them even if your host byte order is already big endian. In this case, you would have (essentially)

```
#define htobe32(x) x
```

We can reverse the process, as well:

```
#include <stdint.h>
#include <sys/types.h>

uint32_t i;
int n = recv(sockfd,&i,4,0);
if ( 4 == n ) {
   i = be32toh(i);
}
```

See *A General Systems Handbook*, Chapter 2, for more details about integer sizes, byte ordering, and byte conversions.

Strings are, in some ways, easier. To send a string, it can be as simple as:

```
char s[] = "hello";
int n = send(sockfd,s,strlen(s)+1,0);
```

This will send the string, including the terminal **NULL**. Receiving it is more difficult, though, because we do not know how long the string is. We could speculatively receive bytes until we see a **NULL**, or we could send the *length* first:

```
char s[] = "hello";
uint8_t len = strlen(s) + 1;
int n = send(sockfd,&len,1,0);
n = send(sockfd,s,len,0);
```

As long as the string length is less than 255, this will work just fine. On the receiving end:

```
char s[256];
uint8_t len;
int n = recv(sockfd,&len,1,0);
n = recv(sockfd,s,len,0);
```

Buffers

As previously mentioned, it can be inefficent to send values one-by-one. We would prefer to fill a *buffer*, and then send that with one function call. To begin, we need to know how long our buffer must be. Consider the string case. We have some string (a literal here, but it does not need to be), and we want to put the string and its length in a single buffer. Again, assuming the length of the string is less than 255, we could have:

```
#include <stdlib.h>

unsigned char* buffer;
char s[] = "hello";
uint8_t len = strlen(s) + 1;
buffer = malloc(len+1);
```

Here **`len`** is the length of the string, with 1 added for the terminating **`NULL`**. Our buffer must accommodate not just this length, but also the additional byte for the length specifier, which is why we allocate **`len+1`** bytes, or **`strlen(s)+2`**.

Now we can fill the buffer using **`memcpy()`**, which copies bytes from one memory location to another:

```
#include <string.h>

buffer[0] = len;
memcpy(buffer+1,s,len);
int n = send(sockfd,buffer,sizeof(buffer),0);
```

Since **`len`** is a single byte, we can assign it directly to the first element of **`buffer`**. The string **`s`** is copied into the buffer starting at the address of **`buffer[1]`**.

You can extend this as necessary. Say we have two 4-byte integers and a string to put into our buffer:

```
uint32_t a = htobe32(1234);
uint32_t b = htobe32(5678);
char s[] = "hello";
uint8_t len = strlen(s) + 1;
unsigned char* buffer = malloc( 4 + 4 + len + 1 );
memcpy(buffer, &a, 4);
memcpy(buffer+4, &b, 4);
buffer[8] = len;
memcpy(buffer+9, s, len);
int n = send(sockfd, buffer, sizeof(buffer), 0);
```

You will often see an additional pointer used to make this look a little nicer, and to make it easier to rearrange things, if needed:

```
uint32_t a = htobe32(1234);
uint32_t b = htobe32(5678);
char s[] = "hello";
uint8_t len = strlen(s) + 1;
unsigned char* buffer = malloc( 4 + 4 + len + 1 );
unsigned char* p = buffer;
memcpy(p, &a, 4);   p += 4
memcpy(p, &b, 4);   p += 4;
p[0] = len;         p += 1;
memcpy(p, s, len);  p += len;
int n = send(sockfd, buffer, sizeof(buffer), 0);
```

We advance our pointer **`p`** (initially pointing to the start of **`buffer`**) by the number of bytes written, so that it is always pointing to the next location where we want to write.

Structures

You might be wondering whether we can just send structures directly, since they are just a collection of fields in memory. We *can*, but only under certain conditions:

- The fields must not include pointers (nested structures are fine).
- The structure must be *packed*.

What is a packed structure? Normally, the compiler will add some padding bytes to a complex structure. These bytes are not used, but they cause the fields' bytes to be aligned in memory in a way that makes accessing them more efficient. This will vary from system to system, however, so we need to ensure that there are *no* padding bytes in a structure that we send across the network. That is what we mean by a packed structure.

We can define a packed structure with:

```
struct __attribute__((__packed__)) msg_struct {
   ...
};
```

This must be done for *every* structure you are going to use this way, including structures that are included as fields within our message structures. You should be **very careful** when using this feature, as it is easy to get things wrong. When in doubt, stick to the earlier methods in this section.

5.2.3 Defining Messages

As a stream protocol, there is no mechanism built into TCP to tell us where a message begins or ends. We will look at one fairly simple way to do this in our application, but it is not the only option. Most of the other methods do some variation on this, but with the details abstracted away from the application developer.

We begin by defining a set of *message types*. These are short fixed-length numbers, often one or two bytes, depending on how many types you need. We start reading data from the socket with this number, and use it to determine how to proceed.

Here is an example:

```
#define INIT_MSG 0
#define REQ_MSG 1
unsigned char msgType;
int n = recv(sockfd, &msgType, 1, 0);

switch(msgType) {
   case INIT_MSG: ...
   case REQ_MSG: ...
}
```

Our message type here is a single byte, which can be 0 or 1 (any other value is undefined). We read a single byte from our socket, and then conditionally execute based on the type of message.

From this point, our protocol should specify what fields are in each type of message, in what order, and what sizes they are. We have already seen how to handle values such as integers and strings, but it is worth considering some other types of fields.

We can handle arrays similarly to strings, in that we begin with a length field, and then read that many values in a loop. Note that these values can themselves be complex, with internal message types.

Another option, if our array elements all have the same size, is to read this amount of data in a loop until we detect an *end-of-field* marker. If this marker is not the same size as the data elements, we can have a linked-list-like structure, where each element contains a **`has_next`** field, or similar.

As long as we specify our messages appropriately, we can use some combination of these techniques to encode and decode anything.

Chapter 6

Traffic

So far, we have looked at how the network is constructed, and how to send messages over that network. Now we will take a look at how we examine traffic *in-flight*. That is, we will attach processes directly to network interfaces to observe the actual packets (and layer 2 frames). This will include both basic (but powerful) packet capturing tools, as well as filtering tools like *firewalls*.

6.1 Packet Structure

To start, we need to take a look at the structure of an IPv4 packet:

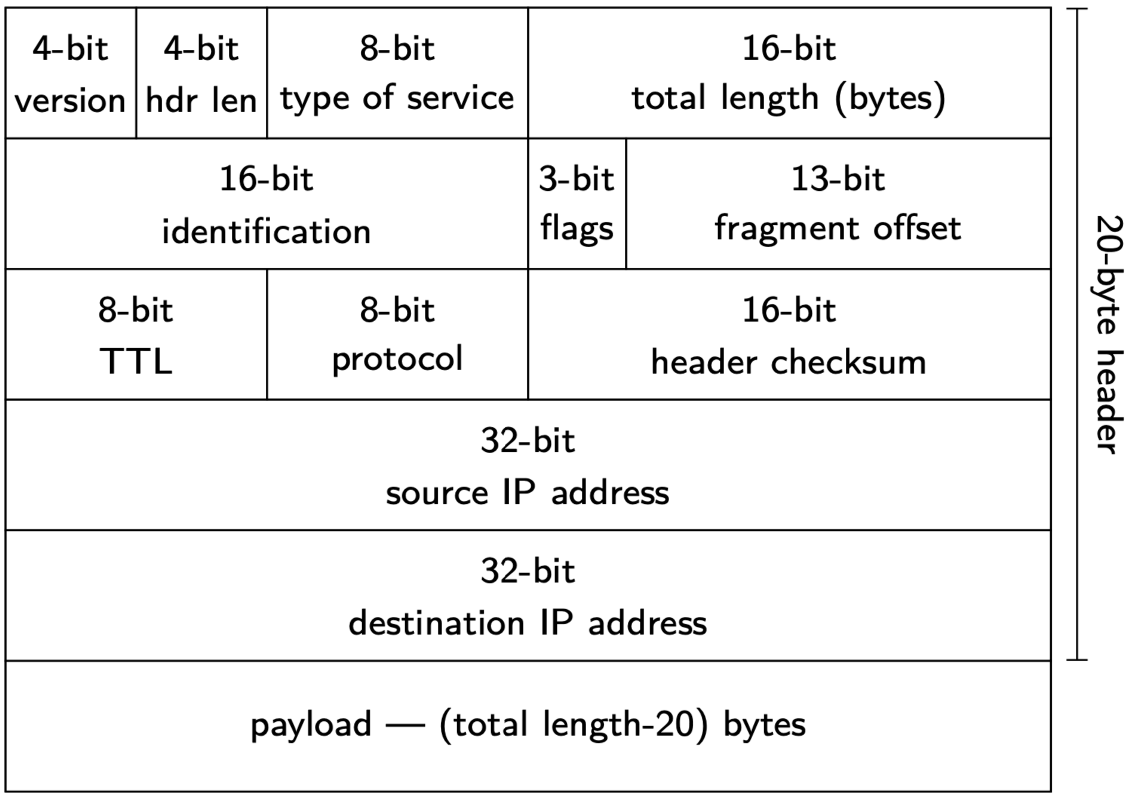


Figure 6.1: Structure of an IPv4 packet.

The first byte is divided between a version (which is `4` for IPv4) and a header length, which is measured in 4-byte *words* (`5` for the minimal 20-byte header). The next byte is the type of service, which is often `0`, and which we will not worry about. The next two bytes are the total length of the packet, in bytes, including the header and payload.

The next four bytes are used to identify and reconstruct packets that might have been *fragmented*. Fragmentation occurs with the packet's total length is greater than the path MTU. The two-byte identifier, when combined with the source and destination, uniquely identifies the entire packet. The next two bytes are split between 3 bits of fragmentation flags, and 13 bits for the fragment offset. The flags might indicate that the packet should not be fragmented, or that there are additional fragments following this one. The offset gives the byte offset in the payload for this fragment, but must be multiplied by 8 to obtain the actual offset (because we have used 3 bits for the flags).

The next four bytes are the TTL (1 byte), the layer 4 protocol specifier (1 byte), and a 2-byte header *checksum*, which allows devices to detect errors. Following those are the 4-byte source address and the 4-byte destination address. If the header length is greater than `5` (that is, more than 20 bytes), the remainder of the header are IPv4 options, which we will not cover.

6.2 Packet Capture

There are a number of tools that we can use for packet capturing. Here, we will focus on `tcpdump` (the classic), as well as Wireshark and its associated programs `tshark` and `dumpcap`. We will also look at how we can extend Wireshark and `tshark`.

Another tool you might want to look at, if you want to go a bit further, is the `scapy` package for python. There is a discussion of this in *A General Systems Handbook*. There are also other, more specialized, tools, which you can find in references more focused on network security.

6.2.1 `tcpdump`

`tcpdump` is a widely-available packet capturing utility. A system with limited capabilities is more likely to have this than the other tools we will look at, so it is worth exploring in a bit of detail. Moreover, many of the options used by `tcpdump` are the same or very close to options you will see in something like `tshark`. It is likely that you will need to run `tcpdump` as the root user.

Despite the name, `tcpdump` captures all network traffic, not merely TCP. You can tell it to capture from specific interfaces, filter on protocols or packet data, and even write to or read from files.

The option you will most often want to specify is **`-i`**, which you use to select an interface to capture. If you do not provide this, **`tcpdump`** will typically connect to the first configured interface, which may or may not be what you want. To capture on *all* interfaces, you would specify **`-i any`** (on some systems this might be **`all`**).

Another common option is **`-n`**, which tells **`tcpdump`** not to perform reverse-lookups to convert numbers to names. That is, the normal behavior is for **`tcpdump`** to take the IP addresses it sees, and attempt to resolve them to hostnames. This can take a while, so it is generally faster to add **`-n`** to your command, and perform any reverse lookups you need manually.

The **`-x`** option tells **`tcpdump`**, which normally only prints header information, to include the packet contents, in hexadecimal format. If you are looking for specific information from the packets, this can be very useful. You can limit the amount that will be printed by setting the *snapshot length* with the **`-s`** option, which takes the number of bytes as an argument. Related options to **`-x`** are **`-xx`** (include the link layer), **`-X`** (include ASCII as well as hex), and **`-XX`** (**`-X`**, but including the link layer).

The **`-v`** option provides more verbose output, and can be specified multiple times for additional details.

If you are viewing TCP traffic, **`tcpdump`** will construct *relative* sequence numbers. The sequence numbers are used by TCP to correctly order data and detect lost packets. They generally start from random offsets, so relative sequence numbers begin both sides' counts from 0. If you need to see the actual, *absolute* sequence numbers, you can do this with the **`-S`** option.

If you want to save the packet capture to a file, you can use the **`-w`** option, which should be followed by a filename. The complement of this is the **`-r`** option, which reads packets from a file rather than capturing from an interface.

Like many programs, **tcpdump** will buffer data before printing it to the screen or writing to a file. You can unbuffer (ie, disable the buffering) with the **-U** flag, which is very useful when you want to be certain that you do not lose any data.

Finally, you can apply a filter to the packets, so that only certain ones are displayed or written to a file. For example, if you only want to see ICMP packets, you would add **icmp** as the filter. If you want to capture HTTPS traffic, you could add **tcp port 443**, which will match the standard HTTPS port as either the source or destination.

Here is an example to capture and display HTTP traffic on the **eth0** interface, including packet data (limited to 128 bytes):

```
$ sudo tcpdump -i eth0 -nx -s 128 tcp port 80
tcpdump: verbose output suppressed, use -v or -vv for full protocol decode
listening on eth0, link-type EN10MB (Ethernet), capture size 128 bytes
19:31:25.623476 IP 172.17.0.2.60522 > 128.8.127.4.80: Flags [S], seq 3656059548, win 64240,
 options [mss 1460,sackOK,TS val 725693382 ecr 0,nop,wscale 7], length 0
        0x0000:  4500 003c a09b 4000 4006 ef00 ac11 0002
        0x0010:  8008 7f04 ec6a 0050 d9eb 0a9c 0000 0000
        0x0020:  a002 faf0 ab4e 0000 0204 05b4 0402 080a
        0x0030:  2b41 33c6 0000 0000 0103 0307
19:31:25.642768 IP 128.8.127.4.80 > 172.17.0.2.60522: Flags [S.], seq 2871939518,
 ack 3656059549, win 65535, options [mss 1460,wscale 2,eol], length 0
        0x0000:  4500 0030 0000 4000 2506 aaa8 8008 7f04
        0x0010:  ac11 0002 0050 ec6a ab2e 51be d9eb 0a9d
        0x0020:  7012 ffff 09bf 0000 0204 05b4 0303 0200
19:31:25.642846 IP 172.17.0.2.60522 > 128.8.127.4.80: Flags [.], ack 1, win 502, length 0
        0x0000:  4500 0028 a09c 4000 4006 ef13 ac11 0002
        0x0010:  8008 7f04 ec6a 0050 d9eb 0a9d ab2e 51bf
        0x0020:  5010 01f6 ab3a 0000
19:31:25.642999 IP 172.17.0.2.60522 > 128.8.127.4.80: Flags [P.], seq 1:142, ack 1, win 502,
 length 141: HTTP: GET / HTTP/1.1
        0x0000:  4500 00b5 a09d 4000 4006 ee85 ac11 0002
        0x0010:  8008 7f04 ec6a 0050 d9eb 0a9d ab2e 51bf
        0x0020:  5018 01f6 abc7 0000 4745 5420 2f20 4854
        0x0030:  5450 2f31 2e31 0d0a 5573 6572 2d41 6765
        0x0040:  6e74 3a20 5767 6574 2f31 2e32 302e 3320
        0x0050:  286c 696e 7578 2d67 6e75 290d 0a41 6363
        0x0060:  6570 743a 202a 2f2a 0d0a 4163 6365 7074
        0x0070:  2d45
19:31:25.643352 IP 128.8.127.4.80 > 172.17.0.2.60522: Flags [.], ack 142, win 65535, length 0
        0x0000:  4500 0028 0000 4000 2506 aab0 8008 7f04
        0x0010:  ac11 0002 0050 ec6a ab2e 51bf d9eb 0b2a
        0x0020:  5010 ffff 35f6 0000
19:31:25.666966 IP 128.8.127.4.80 > 172.17.0.2.60522: Flags [P.], seq 1:592, ack 142,
 win 65535, length 591: HTTP: HTTP/1.1 301 Moved Permanently
        0x0000:  4500 0277 0000 4000 2506 a861 8008 7f04
        0x0010:  ac11 0002 0050 ec6a ab2e 51bf d9eb 0b2a
        0x0020:  5018 ffff a8b8 0000 4854 5450 2f31 2e31
        0x0030:  2033 3031 204d 6f76 6564 2050 6572 6d61
        0x0040:  6e65 6e74 6c79 0d0a 6461 7465 3a20 4672
        0x0050:  692c 2033 3020 4465 6320 3230 3232 2031
        0x0060:  393a 3331 3a32 3520 474d 540d 0a73 6572
        0x0070:  7665
```

As you can see, **`tcpdump`** is able to determine some of the application information, such as the initial request being a **`GET /`**, and the response a **`301 Moved Permanently`**.

Here are some filter examples:

- **`host www.cs.umd.edu`**: To or from **`www.cs.umd.edu`**
- **`src www.cs.umd.edu`**: From **`www.cs.umd.edu`**
- **`dst www.cs.umd.edu`**: To **`www.cs.umd.edu`**
- **`tcp port 443`**: TCP traffic to or from port 443 (HTTPS)
- **`tcp src port 443`**: TCP traffic from port 443
- **`greater 100`**: Packet length greater than 100 bytes
- **`less 100`**: Packet length less than 100 bytes
- **`and`**/**`or`**/**`not`**: Boolean operators (parentheses also supported)

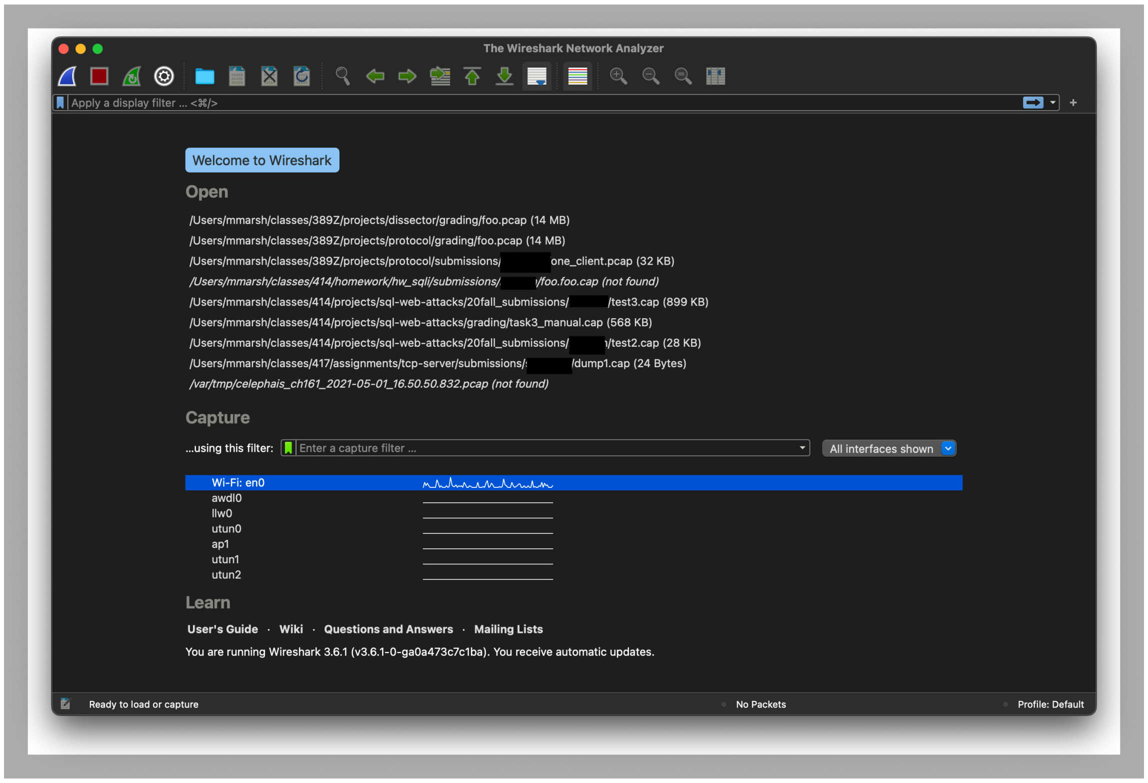
The **`pcap-filter`** manpage (**`libpcap`** is the underlying library that **`tcpdump`** uses) has comprehensive details on the filter language.

6.2.2 Wireshark

Wireshark (`wireshark.org`) is a graphical packet capture and display application. It has all of the features of **`tcpdump`**, but a friendlier display. Like

tcpdump, it can write packet captures and read them in. The output format it uses is slightly different, but it is able to read files produced by **tcpdump**. This can be very convenient if you have **tcpdump** as your only capture option on a system, since you can write the captured packets to a file, and then transfer that file to another machine where Wireshark is available.

When you first start Wireshark, it will ask you to select an interface or an existing capture file. The following is from a Mac, but (aside from interface names) it would look similar under Linux or Windows:


The Wireshark Network Analyzer

Apply a display filter ... <⌘/>

Welcome to Wireshark

Open

```
/Users/mmarsh/classes/389Z/projects/dissector/grading/foo.pcap (14 MB)
/Users/mmarsh/classes/389Z/projects/protocol/grading/foo.pcap (14 MB)
/Users/mmarsh/classes/389Z/projects/protocol/submissions/[redacted]one_client.pcap (32 KB)
/Users/mmarsh/classes/414/homework/hw_sqli/submissions/[redacted]/foo.foo.cap (not found)
/Users/mmarsh/classes/414/projects/sql-web-attacks/20fall_submissions/[redacted]/test3.cap (899 KB)
/Users/mmarsh/classes/414/projects/sql-web-attacks/grading/task3_manual.cap (568 KB)
/Users/mmarsh/classes/414/projects/sql-web-attacks/20fall_submissions/[redacted]/test2.cap (28 KB)
/Users/mmarsh/classes/417/assignments/tcp-server/submissions/[redacted]/dump1.cap (24 Bytes)
/var/tmp/celephais_ch161_2021-05-01_16.50.50.832.pcap (not found)
```

Capture

...using this filter: Enter a capture filter ... All interfaces shown

```
Wi-Fi: en0
awdl0
llw0
utun0
ap1
utun1
utun2
```

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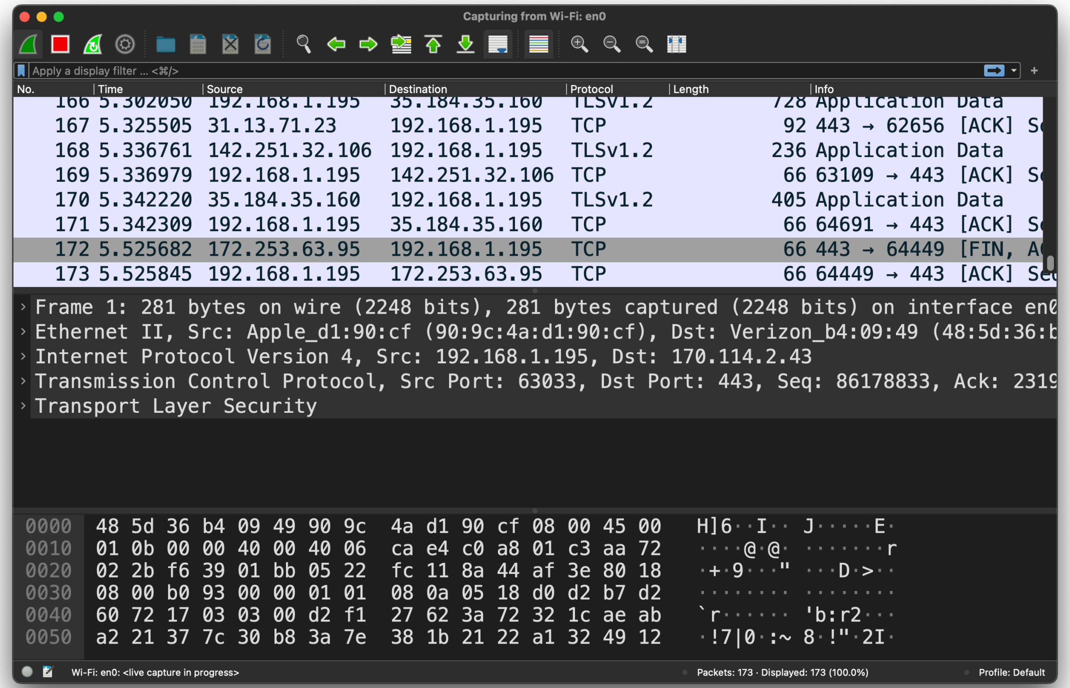
You are running Wireshark 3.6.1 (v3.6.1-0-ga0a473c7c1ba). You receive automatic updates.

Ready to load or capture No Packets Profile: Default

Figure 6.2: Wireshark input selection

The line diagrams next to the interface names show you the amount of traffic on those interfaces. If we select one of these, we get the following:



```
166 5.302050 192.168.1.195  35.184.35.160  TLSv1.2  728 Application Data
167 5.325505 31.13.71.23    192.168.1.195  TCP       92 443 → 62656 [ACK] S
168 5.336761 142.251.32.106 192.168.1.195  TLSv1.2  236 Application Data
169 5.336979 192.168.1.195  142.251.32.106 TCP       66 63109 → 443 [ACK] S
170 5.342220 35.184.35.160  192.168.1.195  TLSv1.2  405 Application Data
171 5.342309 192.168.1.195  35.184.35.160  TCP       66 64691 → 443 [ACK] S
172 5.525682 172.253.63.95  192.168.1.195  TCP       66 443 → 64449 [FIN, A
173 5.525845 192.168.1.195  172.253.63.95  TCP       66 64449 → 443 [ACK] Se
```

```
› Frame 1: 281 bytes on wire (2248 bits), 281 bytes captured (2248 bits) on interface en0
› Ethernet II, Src: Apple_d1:90:cf (90:9c:4a:d1:90:cf), Dst: Verizon_b4:09:49 (48:5d:36:b
› Internet Protocol Version 4, Src: 192.168.1.195, Dst: 170.114.2.43
› Transmission Control Protocol, Src Port: 63033, Dst Port: 443, Seq: 86178833, Ack: 2319
› Transport Layer Security
```

```
0000  48 5d 36 b4 09 49 90 9c  4a d1 90 cf 08 00 45 00   H]6··I·· J·····E·
0010  01 0b 00 00 40 00 40 06  ca e4 c0 a8 01 c3 aa 72   ····@·@· ·······r
0020  02 2b f6 39 01 bb 05 22  fc 11 8a 44 af 3e 80 18   ·+·9···" ···D·>··
0030  08 00 b0 93 00 00 01 01  08 0a 05 18 d0 d2 b7 d2   ········ ········
0040  60 72 17 03 03 00 d2 f1  27 62 3a 72 32 1c ae ab   `r······ 'b:r2···
0050  a2 21 37 7c 30 b8 3a 7e  38 1b 21 22 a1 32 49 12   ·!7|0·:~ 8·!"·2I·
```

```
Wi-Fi: en0: <live capture in progress>    Packets: 173 · Displayed: 173 (100.0%)    Profile: Default
```

Figure 6.3: Wireshark capture

The display is separated into three areas:

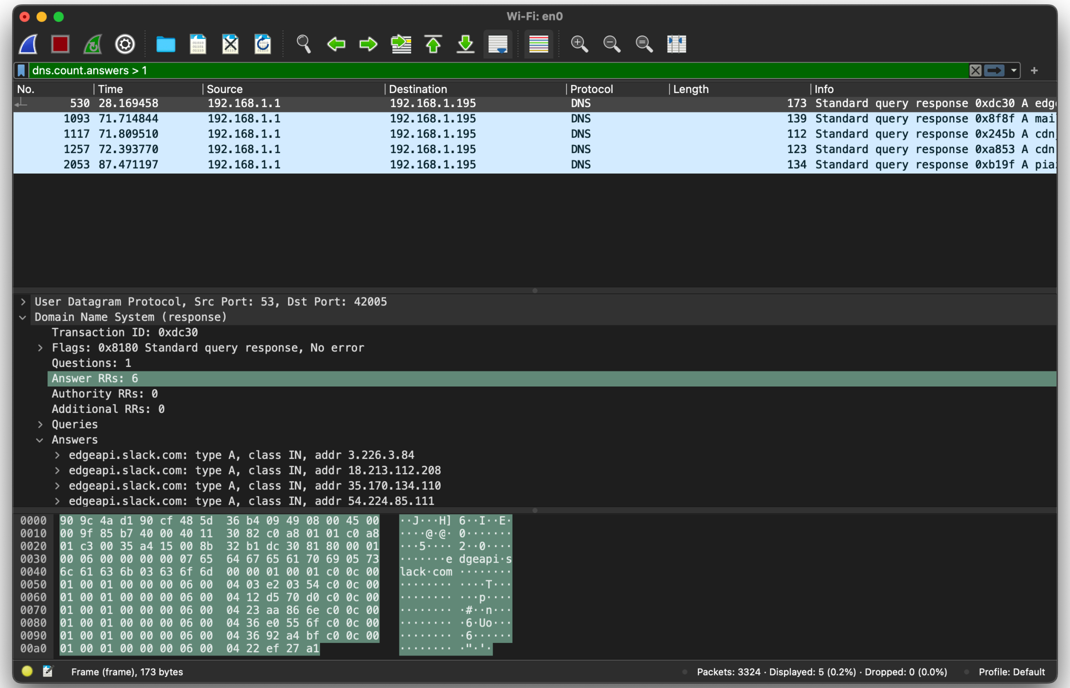
- The packet stream
- Details of the selected packet
- Full packet data

The packet stream shows all of the packets that have been captured and match the current filter (if there is one). It shows much of the header information for each packet, making it fairly easy to pick out packets of interest.

When you select one of the packets, the bottom area shows you the actual bytes, in hexadecimal, and any printable ASCII characters. This includes the layer 2 frame header, so the actual IPv4 header begins with the last two bytes of the first line (`45 00`) at byte 14 (`000e`).

The middle area shows the packet *dissection*, where Wireshark has examined the bytes to determine the protocols and their details. We can see details like the source and destination, the fact that it is a TCP packet, and even that at the application layer it is protected with TLS (that is, it is encrypted).

The filter format for Wireshark is very different than that of **`tcpdump`**. If we want to see DNS responses containing more than one answer, we could specify the filter **`dns.count.answers > 0`**, as shown in Figure 6.4.



```
dns.count.answers > 1
```

```
No.   |Time       |Source        |Destination     |Protocol |Length |Info
  530 28.169458   192.168.1.1    192.168.1.195    DNS          173 Standard query response 0xdc30 A edg
 1093 71.714844   192.168.1.1    192.168.1.195    DNS          139 Standard query response 0x8f8f A mai
 1117 71.809510   192.168.1.1    192.168.1.195    DNS          112 Standard query response 0x245b A cdn
 1257 72.393770   192.168.1.1    192.168.1.195    DNS          123 Standard query response 0xa853 A cdn
 2053 87.471197   192.168.1.1    192.168.1.195    DNS          134 Standard query response 0xb19f A pia
```

```
> User Datagram Protocol, Src Port: 53, Dst Port: 42005
v Domain Name System (response)
    Transaction ID: 0xdc30
  > Flags: 0x8180 Standard query response, No error
    Questions: 1
    Answer RRs: 6
    Authority RRs: 0
    Additional RRs: 0
  > Queries
  v Answers
    > edgeapi.slack.com: type A, class IN, addr 3.226.3.84
    > edgeapi.slack.com: type A, class IN, addr 18.213.112.208
    > edgeapi.slack.com: type A, class IN, addr 35.170.134.110
    > edgeapi.slack.com: type A, class IN, addr 54.224.85.111
```

```
0000  90 9c 4a d1 90 cf 48 5d  36 b4 09 49 08 00 45 00   ··J···H] 6··I··E·
0010  00 9f 85 b7 40 00 40 11  30 82 c0 a8 01 01 c0 a8   ····@·@· 0·······
0020  01 c3 00 35 a4 15 00 8b  32 b1 dc 30 81 80 00 01   ···5···· 2··0····
0030  00 06 00 00 00 00 07 65  64 67 65 61 70 69 05 73   ·······e dgeapi·s
0040  6c 61 63 6b 03 63 6f 6d  00 00 01 00 01 c0 0c 00   lack·com ········
0050  01 00 01 00 00 00 06 00  04 03 e2 03 54 c0 0c 00   ········ ····T···
0060  01 00 01 00 00 00 06 00  04 12 d5 70 d0 c0 0c 00   ········ ···p····
0070  01 00 01 00 00 00 06 00  04 23 aa 86 6e c0 0c 00   ········ ·#··n···
0080  01 00 01 00 00 00 06 00  04 36 e0 55 6f c0 0c 00   ········ ·6·Uo···
0090  01 00 01 00 00 00 06 00  04 36 92 a4 bf c0 0c 00   ········ ·6······
00a0  01 00 01 00 00 00 06 00  04 22 ef 27 a1            ········ ·"·'·
```

```
Frame (frame), 173 bytes          Packets: 3324 · Displayed: 5 (0.2%) · Dropped: 0 (0.0%)          Profile: Default
```

Figure 6.4: Wireshark capture of DNS responses

Since DNS is not encrypted, Wireshark is able to display details of the response, including all of the records that were returned. If you select an item from the dissection, Wireshark will highlight the corresponding bytes. The reverse is also true: clicking on a byte will highlight all of the bytes corresponding to its field in the dissection, as well as the dissection item. You can also right-click on something in the dissection or data areas, and save the corresponding data.

6.2.3 `tshark` and `dumpcap`

Wireshark comes with two command-line tools, which you might have to specify when installing it. The first, `tshark`, is a terminal version of Wireshark. It can do everything that Wireshark can, just with text instead of a graphical display. The second, `dumpcap`, only does packet capturing, and does not display the packet contents.

By default, `tshark` produces output somewhat similar to `tcpdump`. You can change this with options, however:

- `tshark -Tjson`: This produces a JSON string with all of the packets in the capture, including detailed dissections.
- `tshark -Tfields`: This allows you to specify a set of space-delimited columns containing desired fields. This *must* be accompanied by at least one `-e` option, which specifies fields, such as `data` or `dns.count.answers`.

You can also specify stop conditions for `tshark` and `dumpcap`. With the `-c` option, you can tell the program to exit after capturing a certain number of packets. More interestingly, you can use the `-a` option to specify an *autostop* condition, such as `-a duration:30`, which will automatically stop after 30 seconds.

For both `tshark` and `dumpcap`, the `-h` option gives you detailed usage information.

6.3 Extending Wireshark

While Wireshark comes with many built-in dissectors, you can write your own. You can also write your own filters, but we will not consider those. Unless you are going to compile your own version of Wireshark, the recommended way to write a dissector is in the *lua* programming language.

Lua is fairly simple and lightweight, so it has become popular for extensions in a number of applications such as customizable video games. The syntax is not very complex, and if you have programmed in Javascript, a lot of the design principles should look somewhat familiar.

There are two ways to load a dissector into Wireshark or **`tshark`**:

1. Put the file in the directory ~**`/.local/lib/wireshark/plugins/`**. If Wireshark is currently running, you will need to select the “Analyze” menu, and then “Reload Lua Plugins”.

2. When starting the program, add the option **`-X lua_script:<filename>`**, where **`<filename>`** should be replaced with the actual filename.

You can find examples at `https://gitlab.com/wireshark/wireshark/-/wikis/Lua/Examples`, and we will base our discussion on **`fpm.lua`** (©2015 by Hadriel Kaplan, released under the 3-clause BSD license). For general information about lua, see `https://www.lua.org/pil/contents.html`.

TCP dissectors are somewhat complicated, because messages might be split between packets, and a single packet might have parts of multiple messages in it. That means we have to maintain state while dissecting packets, and determine whether we have a complete message yet or not.

6.3.1 Lua Syntax Highlights

Before we take a look at the actual code, we should go over a little bit of Lua syntax.

- Single-line comments begin with `--`, while block comments are delimited by `--[[` and `]]`.
- `~=` is the inequality binary operator (`!=` in most other languages).
- If we have a class `MyClass`, we can call a *class* method (`static` in many languages) `foo` with `MyClass.foo`. If we have an instance `myClass`, we can call an *instance* method `bar` with `myClass:bar()`.
- We define a function with the `function` keyword, and the function definition ends with the `end` keyword. The `return` keyword specifies the value to return.
- By default, variables and functions are created in the global namespace. You can prepend them with the `local` keyword to keep them in the local scope instead.
- Loops have the form `for x in list do ... end` or `for i = start,end,step do ... end`.
- We also have `while` loops and `if-elseif-else` blocks, similarly defined (that is, `do` to begin the body of the block and `end` to end it).

6.3.2 Creating a Protocol and Defining Fields

`fpm.lua` has a lot of comments, which are worth reading for details on why the dissector is written the way it is. It also has some features we will not bother with, such as configuration and debugging with different verbosity levels.

The actual dissector code begins with

```
--------------------------------------------------------------------------------
-- creates a Proto object, but doesn't register it yet
local fpm_proto = Proto("fpm", "FPM Header")
```

This defines a new `Proto` object. The protocol dissection will be nested under this, so all fields will begin with `fpm`, the first argument. Compare this with the filter `dns.count.answers > 0`, where `dns` is the top-level value. The second argument is the verbose name for the protocol. Again comparing with DNS, this is the equivalent of "Domain Name System" in the packet details.

The next thing in the file is a helper function

```
----------------------------------------
-- a function to convert tables of enumerated types to value-string tables
-- i.e., from { "name" = number } to { number = "name" }
local function makeValString(enumTable)
    local t = {}
    for name,num in pairs(enumTable) do
        t[num] = name
    end
    return t
end
```

Given a mapping $A \mapsto B$, this creates the inverse mapping $B \mapsto A$. We use this in

```
local msgtype = {
    NONE     = 0,
    NETLINK  = 1,
}
local msgtype_valstr = makeValString(msgtype)
```

Here we have defined an enumeration `msgtype` that assigns a protocol value (`0` or `1`) to a name (`NONE` or `NETLINK`), and then creates the reverse `msg-type_valstr`. This means that when we extract a message type from a packet, we can use `msgtype_valstr` to recover the name.

The FPM protcol has three header fields, and these are what we define next.

```
----------------------------------------
-- a table of all of our Protocol's fields
local hdr_fields =
{
    version   = ProtoField.uint8 ("fpm.version", "Version", base.DEC),
    msg_type  = ProtoField.uint8 ("fpm.type", "Type", base.DEC, msgtype_valstr),
    msg_len   = ProtoField.uint16("fpm.length", "Length", base.DEC),
}

-- register the ProtoFields
fpm_proto.fields = hdr_fields
```

The last line sets this mapping as the protocol's fields. For each field, we use a class method to construct an explicitly sized object, with the field name (eg, **`fpm.type`**), the text to display for the field (eg, "Type"), the format for the field (eg, **`base.DEC`** for decimal), and optionally a mapping to display a text field corresponding to the numeric value (eg, **`msgtype_valstr`**).

Next we define the header length (which is constant in this case) and provide forward declarations for helper functions.

```
-- this is the size of the FPM message header (4 bytes) and the minimum FPM
-- message size we need to figure out how much the rest of the Netlink message
-- will be
local FPM_MSG_HDR_LEN = 4

-- some forward "declarations" of helper functions we use in the dissector
local createSllTvb, dissectFPM, checkFpmLength
```

If we have a variable header length (like the IPv4 header), then we would want a minimum length.

6.3.3 The Dissector

Now we come to the actual dissector, which is somewhat long. We have omitted the definition of **`tvbs`** and **`dprint2`**. The comments are very descriptive, so we will just note that we have to look at all of the data in the packet, because there might be multiple messages or partial messages. The return value at the end of this should either be the total number of bytes consumed by the dissector (that is, we have converted these bytes into one or more complete messages) or -1 times the number of bytes we need to complete the partial message at the end of the packet. Some of the work has been pushed to the **`dissectFPM`** helper function, which we will look at after this function.

```
--------------------------------------------------------------------------------
-- The following creates the callback function for the dissector.
-- It's the same as doing "fpm_proto.dissector = function (tvbuf,pkt,root)"
-- The 'tvbuf' is a Tvb object, 'pktinfo' is a Pinfo object, and 'root' is a TreeItem object.
-- Whenever Wireshark dissects a packet that our Proto is hooked into, it will call
-- this function and pass it these arguments for the packet it's dissecting.
function fpm_proto.dissector(tvbuf, pktinfo, root)
    dprint2("fpm_proto.dissector called")
    -- reset the save Tvbs
    tvbs = {}

    -- get the length of the packet buffer (Tvb).
    local pktlen = tvbuf:len()

    local bytes_consumed = 0

    -- we do this in a while loop, because there could be multiple FPM messages
    -- inside a single TCP segment, and thus in the same tvbuf - but our
    -- fpm_proto.dissector() will only be called once per TCP segment, so we
    -- need to do this loop to dissect each FPM message in it
    while bytes_consumed < pktlen do

        -- We're going to call our "dissect()" function, which is defined
        -- later in this script file. The dissect() function returns the
        -- length of the FPM message it dissected as a positive number, or if
        -- it's a negative number then it's the number of additional bytes it
        -- needs if the Tvb doesn't have them all. If it returns a 0, it's a
        -- dissection error.
        local result = dissectFPM(tvbuf, pktinfo, root, bytes_consumed)

        if result > 0 then
            -- we successfully processed an FPM message, of 'result' length
            bytes_consumed = bytes_consumed + result
```

```
                -- go again on another while loop
            elseif result == 0 then
                -- If the result is 0, then it means we hit an error of some kind,
                -- so return 0. Returning 0 tells Wireshark this packet is not for
                -- us, and it will try heuristic dissectors or the plain "data"
                -- one, which is what should happen in this case.
                return 0
            else
                -- we need more bytes, so set the desegment_offset to what we
                -- already consumed, and the desegment_len to how many more
                -- are needed
                pktinfo.desegment_offset = bytes_consumed

                -- invert the negative result so it's a positive number
                result = -result

                pktinfo.desegment_len = result

                -- even though we need more bytes, this packet is for us, so we
                -- tell wireshark all of its bytes are for us by returning the
                -- number of Tvb bytes we "successfully processed", namely the
                -- length of the Tvb
                return pktlen
            end
        end

        -- In a TCP dissector, you can either return nothing, or return the number of
        -- bytes of the tvbuf that belong to this protocol, which is what we do here.
        -- Do NOT return the number 0, or else Wireshark will interpret that to mean
        -- this packet did not belong to your protocol, and will try to dissect it
        -- with other protocol dissectors (such as heuristic ones)
        return bytes_consumed
    end
```

6.3.4 Dissecting the Bytes

Most of the real work is being done in the following helper function. It will start by checking the number of bytes we have received so far, and see whether we have a complete message. The **`while`** loop in the dissector ensures we are only dealing with a single protocol message here. If we do not have enough data yet, we return -1 times the number of bytes still to come, as returned by **`checkFpmLength`**. If we have a complete message, we pull apart the bytes within it and build up the protocol's dissection tree. All that you really need to know about the **`Tvb`** type is that it has a **`range`** instance method that takes an offset and number of bytes and returns a **`TvbRange`**, and that **`TvbRange`** can be converted to useable types with its instance methods like **`uint`**. The **`TvbRange`** is what you add to the tree for a given field. This particular protocol has internal data that can either be parsed as the **`NetLink`** protocol or the generic **`Data`** protocol.

```
----------------------------------------
-- The following is a local function used for dissecting our FPM messages
-- inside the TCP segment using the desegment_offset/desegment_len method.
-- It's a separate function because we run over TCP and thus might need to
-- parse multiple messages in a single segment/packet. So we invoke this
-- function only dissects one FPM message and we invoke it in a while loop
-- from the Proto's main disector function.
--
-- This function is passed in the original Tvb, Pinfo, and TreeItem from the Proto's
-- dissector function, as well as the offset in the Tvb that this function should
-- start dissecting from.
--
-- This function returns the length of the FPM message it dissected as a
-- positive number, or as a negative number the number of additional bytes it
-- needs if the Tvb doesn't have them all, or a 0 for error.
--
dissectFPM = function (tvbuf, pktinfo, root, offset)
    dprint2("FPM dissect function called")

    local length_val, length_tvbr = checkFpmLength(tvbuf, offset)

    if length_val <= 0 then
        return length_val
    end

    -- if we got here, then we have a whole message in the Tvb buffer
    -- so let's finish dissecting it...

    -- set the protocol column to show our protocol name
    pktinfo.cols.protocol:set("FPM")

    -- set the INFO column too, but only if we haven't already set it before
    -- for this frame/packet, because this function can be called multiple
    -- times per packet/Tvb
    if string.find(tostring(pktinfo.cols.info), "^FPM") == nil then
        pktinfo.cols.info:set("FPM")
    end

    -- We start by adding our protocol to the dissection display tree.
    local tree = root:add(fpm_proto, tvbuf:range(offset, length_val))
    -- dissect the version field
    local version_tvbr = tvbuf:range(offset, 1)
    local version_val  = version_tvbr:uint()
    tree:add(hdr_fields.version, version_tvbr)

    -- dissect the type field
    local msgtype_tvbr = tvbuf:range(offset + 1, 1)
    local msgtype_val  = msgtype_tvbr:uint()
    tree:add(hdr_fields.msg_type, msgtype_tvbr)

    -- dissect the length field
    tree:add(hdr_fields.msg_len, length_tvbr)

    -- ok now the hard part - try calling a sub-dissector?
    -- only if settings/prefs told us to of course...
    if default_settings.subdissect and (version_val == 1) and (msgtype_val == msgtype.NETLINK)
        -- append the INFO column - this will be overwritten/replaced by the
        -- Netlink dissector, which sadly appears to clear it but not set
        -- anything, so doing this is kind of silly/pointless, but since this
        -- is a tutorial script, this showswhat you might want to do for your
        -- protocol
        if string.find(tostring(pktinfo.cols.info), "^FPM:") == nil then
            pktinfo.cols.info:append(": Netlink")
        else
            pktinfo.cols.info:append(", Netlink")
        end

        -- it carries a Netlink message, so we're going to create a new Tvb
        -- with a a fake Linux SLL header for the built-in Netlink dissector
        -- to use
        local tvb = createSllTvb(tvbuf, offset + FPM_MSG_HDR_LEN, length_val - FPM_MSG_HDR_LEN)

        dprint2("FPM trying sub-dissector for wtap encap type:", default_settings.subdiss_type)

        -- invoke the Netlink dissector (we got the Dissector object earlier,
        -- as variable 'netlink')
        netlink:call(tvb, pktinfo, root)

        dprint2("FPM finished with sub-dissector")
    else
        dprint2("Netlink sub-dissection disabled or not Netlink type, invoking 'data' dissector
        -- append the INFO column
        if string.find(tostring(pktinfo.cols.info), "^FPM:") == nil then
            pktinfo.cols.info:append(": Unknown")
        else
            pktinfo.cols.info:append(", Unknown")
        end

        tvbs[#tvbs+1] = tvbuf(offset + FPM_MSG_HDR_LEN, length_val - FPM_MSG_HDR_LEN):tvb()
        data:call(tvbs[#tvbs], pktinfo, root)
    end
    return length_val
end
```

6.3.5 Determining the Length of a Message

Now we look at the function to determine whether we have a complete message. If all protocol messages are not the same length, this will have to do some parsing without actually consuming bytes. It returns a pair of values. The first element the length of the message, -1 times the number of bytes still needed, or 0 if there was an error. If we have not yet received enough bytes to determine the actual message length, we instead return **`-DESEGMENT_ONE_MORE_SEGMENT`**. The second element is the **`TvbRange`** object containing the message, or **`nil`** if we do not yet have a complete message.

```
----------------------------------------
-- The function to check the length field.
--
-- This returns two things: (1) the length, and (2) the TvbRange object, which
-- might be nil if length <= 0.
checkFpmLength = function (tvbuf, offset)

    -- "msglen" is the number of bytes remaining in the Tvb buffer which we
    -- have available to dissect in this run
    local msglen = tvbuf:len() - offset

    -- check if capture was only capturing partial packet size
    if msglen ~= tvbuf:reported_length_remaining(offset) then
        -- captured packets are being sliced/cut-off, so don't try to desegment/reassemble
        dprint2("Captured packet was shorter than original, can't reassemble")
        return 0
    end

    if msglen < FPM_MSG_HDR_LEN then
        -- we need more bytes, so tell the main dissector function that we
        -- didn't dissect anything, and we need an unknown number of more
        -- bytes (which is what "DESEGMENT_ONE_MORE_SEGMENT" is used for)
        dprint2("Need more bytes to figure out FPM length field")
        -- return as a negative number
        return -DESEGMENT_ONE_MORE_SEGMENT
    end
    -- if we got here, then we know we have enough bytes in the Tvb buffer
    -- to at least figure out the full length of this FPM messsage (the length
    -- is the 16-bit integer in third and fourth bytes)

    -- get the TvbRange of bytes 3+4
    local length_tvbr = tvbuf:range(offset + 2, 2)

    -- get the length as an unsigned integer, in network-order (big endian)
    local length_val  = length_tvbr:uint()

    if length_val > default_settings.max_msg_len then
        -- too many bytes, invalid message
        dprint("FPM message length is too long: ", length_val)
        return 0
    end

    if msglen < length_val then
        -- we need more bytes to get the whole FPM message
        dprint2("Need more bytes to desegment full FPM")
        return -(length_val - msglen)
    end

    return length_val, length_tvbr
end
```

6.3.6 Adding the Dissector to Wireshark

The last thing we need to do is add our dissector to Wireshark's table of dissectors. We get the appropriate table for mapping TCP ports to dissectors with **`DissectorTable.get("tcp.port")`**, and then call the instance method **`add`** with the port to use and the **`Proto`** object.

```
--------------------------------------------------------------------------------
-- We want to have our protocol dissection invoked for a specific TCP port,
-- so get the TCP dissector table and add our protocol to it.
local function enableDissector()
    -- using DissectorTable:set() removes existing dissector(s), whereas the
    -- DissectorTable:add() one adds ours before any existing ones, but
    -- leaves the other ones alone, which is better
    DissectorTable.get("tcp.port"):add(default_settings.port, fpm_proto)
end
-- call it now, because we're enabled by default
enableDissector()
```

```
local function disableDissector()
    DissectorTable.get("tcp.port"):remove(default_settings.port, fpm_proto)
end
```

6.4 Firewalls

Linux generally comes with a firewall package called **`iptables`**. This can actually do more than just filter out traffic that might be harmful. It can also be used to create a *network address translator* (NAT) or set *quality of service* (QoS) bits in a packet.

`iptables` has four default tables:

- **`filter`** does packet filtering, which is what you would typically think of as firewall behavior
- **`nat`** provides network address translation
- **`mangle`** modifies packets, typically for quality of service
- **`raw`** is used for anything that does not fit elsewhere

When you want to operate on a specific table (the default is **`filter`**), you can specify the table with the **`-t`** option.

Each table contains a set of *chains*, and each chain contains a list of *rules*. At different stages of processing a packet, different chains will be invoked. For a particular chain, the rules are checked in order until one matches. The standard chains are

- **`PREROUTING`**: This is applied to packets before routing. That is, before the host consults the routing table for the next hop. This chain exists in the **`nat`**, **`mangle`**, and **`raw`** tables.

- **`INPUT`**: This is applied to packets with this host as the destination. It exists in the **`filter`**, **`nat`**, and **`mangle`** tables.
- **`FORWARD`**: If the current host is neither the source nor destination (that is, it is forwarding the packet), this chain is applied. It exists in the **`filter`** and **`mangle`** tables.
- **`OUTPUT`**: This is applied to packets originating at this host. It exists in all tables.
- **`POSTROUTING`**: This is applies after the outgoing interface has been determined by the routing table, but before the packet is passed to that interface. It exists in the **`nat`** and **`mangle`** tables.

Here is a summary table:

Chain	**`filter`**	**`nat`**	**`mangle`**	**`raw`**
`PREROUTING`	no	yes	yes	yes
`INPUT`	yes	yes	yes	no
`FORWARD`	yes	no	yes	no
`OUTPUT`	yes	yes	yes	yes
`POSTROUTING`	no	yes	yes	no

You can view a chain with the **`--list`** option. You can also define your own chains with **`-N`** (new chain). As noted previously, the chains are ordered lists of rules. Each rule has a *condition* and a *target*. The first rule whose condition is met by the packet is applied, and the rest are ignored. Chains will have a default policy, usually with a target of **`ACCEPT`** or **`DROP`**, for any packets that do not match another rule. This can be set with the **`-P`** option, which takes the chain and target as options. The most common filter targets are:

- **`ACCEPT`**: The packet passes through the chain.
- **`DROP`**: The packet is silently dropped.

- **REJECT**: The packet is dropped, and an error message is returned.

There are many other targets, which are documented in https://www.frozentux.net/iptables-tutorial/iptables-tutorial.html#TARGETS. Some of these allow you to modify the packet, such as changing addresses, ports, or protocol options.

To append a new rule to a chain, you use the **-A** option. To replace a rule, you use the **-R** option. To insert a rule (moving the rest down), you use the **-I** option. To delete a rule, you use the **-D** option. Except for appending, you need to specify the index at which the operation should be applied. These begin at 1. The general format for these operations is, after the initial option, the chain, the index (where appropriate), the conditions, and the target (specified with the **-j** option).

Here are some examples (all operating on the **filter** table):

```
iptables -A INPUT -p tcp --dport 1234 -j ACCEPT
```

This appends a rule to the **INPUT** chain, with the TCP protocol and a destination port of **1234**. These packets should be accepted.

```
iptables -R INPUT 1 -p tcp --dport 4321 -j ACCEPT
```

Assuming we began with an empty chain, this replaces the rule above with the same rule, but the destination port changed to **4321**. The original rule is no longer present.

```
iptables -I INPUT 1 -p udp --dport 1234 -j ACCEPT
```

This inserts a new rule before the one we created, specifying that UDP packets with a destination port of **1234** should be accepted.

```
iptables -D INPUT 2
```

This removes the TCP rule, leaving only the UDP rule.