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Chapter 2: Internet Protocols

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Chapter 2

Internet Protocols

THE CCNA EXAM TOPICS COVERED IN THIS CHAPTER INCLUDE THE FOLLOWING:

- ✓ **TECHNOLOGY**
  - Evaluate TCP/IP communication process and its associated protocols

The *Transmission Control Protocol/Internet Protocol (TCP/IP)* suite was created by the Department of Defense (DoD) to ensure and preserve data integrity, as well as maintain communications in the event of catastrophic war. So it follows that if designed and implemented correctly, a TCP/IP network can be a truly dependable and resilient one. In this chapter, I'll cover the protocols of TCP/IP, and throughout this book, you'll learn how to create a marvelous TCP/IP network—using Cisco routers, of course.

We'll begin by taking a look at the DoD's version of TCP/IP and then compare this version and its protocols with the OSI reference model discussed in Chapter 1, "Internetworking."

Once you understand the protocols used at the various levels of the DoD model, you'll learn how to convert between a binary number, hexadecimal number, and decimal number. Then I'll cover IP addressing and the different classes of addresses used in networks today.

Subnetting will be covered in Chapter 3, "IP Subnetting and Variable Length Subnet Masks (VLSMs)."

Because broadcast addresses are so important to understanding IP addressing, subnetting, and VLSM, an understanding of the various flavors of broadcast addresses is critical. I'll cover the various types of broadcast addresses that you just must know. Lastly, I'll provide an introduction to Network Address Translation (NAT) and how Cisco uses NAT.

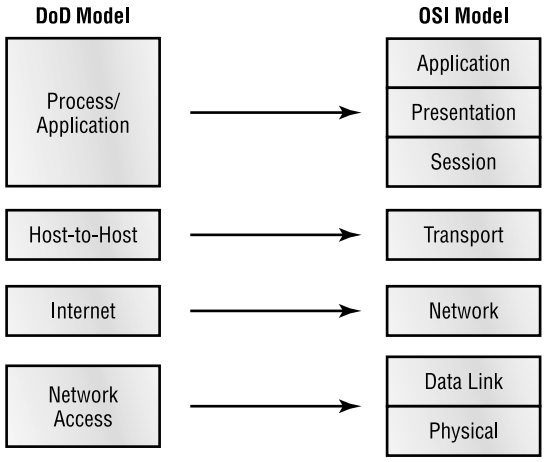
TCP/IP and the DoD Model

The DoD model is basically a condensed version of the OSI model—it's composed of four, instead of seven, layers:

- Process/Application layer
- Host-to-Host layer
- Internet layer
- Network Access layer

Figure 2.1 shows a comparison of the DoD model and the OSI reference model. As you can see, the two are similar in concept, but each has a different number of layers with different names.

FIGURE 2.1 The DoD and OSI models



When talking about the different protocols in the IP stack, the layers of the OSI and DoD models are interchangeable. In other words, the Internet layer and the Network layer describe the same thing, as do the Host-to-Host layer and the Transport layer.

A vast array of protocols combine at the DoD model's *Process/Application layer* to integrate the various activities and duties spanning the focus of the OSI's corresponding top three layers (Application, Presentation, and Session). We'll be looking closely at those protocols in the next part of this chapter. The Process/Application layer defines protocols for node-to-node application communication and also controls user-interface specifications.

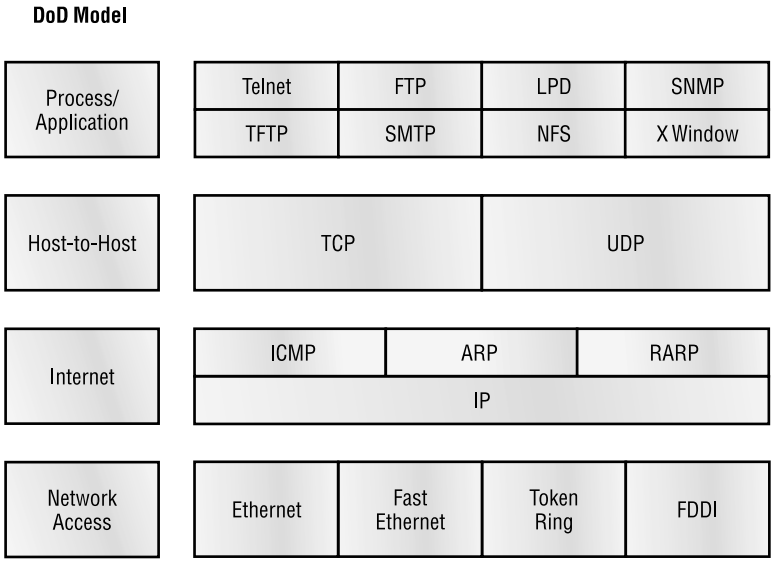
The *Host-to-Host layer* parallels the functions of the OSI's Transport layer, defining protocols for setting up the level of transmission service for applications. It tackles issues such as creating reliable end-to-end communication and ensuring the error-free delivery of data. It handles packet sequencing and maintains data integrity.

The *Internet layer* corresponds to the OSI's Network layer, designating the protocols relating to the logical transmission of packets over the entire network. It takes care of the addressing of hosts by giving them an IP (Internet Protocol) address, and it handles the routing of packets among multiple networks.

At the bottom of the DoD model, the *Network Access layer* monitors the data exchange between the host and the network. The equivalent of the Data Link and Physical layers of the OSI model, the Network Access layer oversees hardware addressing and defines protocols for the physical transmission of data.

The DoD and OSI models are alike in design and concept and have similar functions in similar layers. Figure 2.2 shows the TCP/IP protocol suite and how its protocols relate to the DoD model layers.

FIGURE 2.2 The TCP/IP protocol suite



In the following sections, we will look at the different protocols in more detail, starting with the Process/Application layer protocols.

The Process/Application Layer Protocols

In this section, I'll describe the different applications and services typically used in IP networks. The different protocols and applications covered in this section include the following:

- Telnet
- FTP
- TFTP
- NFS
- SMTP
- LPD
- X Window
- SNMP
- DNS
- DHCP/BootP

Telnet

Telnet is the chameleon of protocols—its specialty is terminal emulation. It allows a user on a remote client machine, called the Telnet client, to access the resources of another machine, the Telnet server. Telnet achieves this by pulling a fast one on the Telnet server and making the client machine appear as though it were a terminal directly attached to the local network. This projection is actually a software image—a virtual terminal that can interact with the chosen remote host.

These emulated terminals are of the text-mode type and can execute refined procedures like displaying menus that give users the opportunity to choose options from them and access the applications on the duped server. Users begin a Telnet session by running the Telnet client software and then logging into the Telnet server.

The name *Telnet* comes from "telephone network," which is how most Telnet sessions used to occur.

File Transfer Protocol (FTP)

File Transfer Protocol (FTP) is the protocol that actually lets us transfer files, and it can accomplish this between any two machines using it. But FTP isn't just a protocol; it's also a program. Operating as a protocol, FTP is used by applications. As a program, it's employed by users to perform file tasks by hand. FTP also allows for access to both directories and files and can accomplish certain types of directory operations, such as relocating into different ones. FTP teams up with Telnet to transparently log you into the FTP server and then provides for the transfer of files.

Accessing a host through FTP is only the first step, though. Users must then be subjected to an authentication login that's probably secured with passwords and usernames implemented by system administrators to restrict access. But you can get around this somewhat by adopting the username "anonymous"—though what you'll gain access to will be limited.

Even when employed by users manually as a program, FTP's functions are limited to listing and manipulating directories, typing file contents, and copying files between hosts. It can't execute remote files as programs.

Trivial File Transfer Protocol (TFTP)

Trivial File Transfer Protocol (TFTP) is the stripped-down, stock version of FTP, but it's the protocol of choice if you know exactly what you want and where to find it, plus it's so easy to use and it's fast too! It doesn't give you the abundance of functions that FTP does, though. TFTP has no directory-browsing abilities; it can do nothing but send and receive files. This compact little protocol also skimps in the data department, sending much smaller blocks of data than FTP, and there's no authentication as with FTP, so it's insecure. Few sites support it because of the inherent security risks.

Real World Scenario

When should you use FTP?

Your San Francisco office needs a 50MB file e-mailed to them right away. What do you do? Most e-mail servers would reject the e-mail because they have size limits. Even if there's no size limit on the server, it still would take a while to send this big file to SF. FTP to the rescue!

If you need to give someone a large file or you need to get a large file from someone, FTP is a nice choice. Smaller files (less than 5MB) can just be sent via e-mail if you have the bandwidth of DSL or a cable modem. However, most ISPs don't allow files larger than 5MB to be e-mailed, so FTP is an option you should consider if you are in need of sending and receiving large files (who isn't these days?). To do this, you will need to set up an FTP server on the Internet so that the files can be shared. Besides, FTP is faster than e-mail, which is another reason to use FTP for sending or receiving large files. In addition, because it uses TCP and is connection-oriented, if the session dies, FTP can start up where it left off. Try that with your e-mail client!

Network File System (NFS)

Network File System (NFS) is a jewel of a protocol specializing in file sharing. It allows two different types of file systems to interoperate. It works like this: Suppose the NFS server software is running on an NT server, and the NFS client software is running on a Unix host. NFS allows for a portion of the RAM on the NT server to transparently store Unix files, which can, in turn, be used by Unix users. Even though the NT file system and Unix file system are unlike—they have different case sensitivity, filename lengths, security, and so on—both Unix users and NT users can access that same file with their normal file systems, in their normal way.

Simple Mail Transfer Protocol (SMTP)

Simple Mail Transfer Protocol (SMTP), answering our ubiquitous call to e-mail, uses a spooled, or queued, method of mail delivery. Once a message has been sent to a destination, the message is spooled to a device—usually a disk. The server software at the destination posts a vigil, regularly checking this queue for messages. When it detects them, it proceeds to deliver them to their destination. SMTP is used to send mail; POP3 is used to receive mail.

Line Printer Daemon (LPD)

The Line Printer Daemon (LPD) protocol is designed for printer sharing. The LPD, along with the LPR (Line Printer) program, allows print jobs to be spooled and sent to the network's printers using TCP/IP.

X Window

Designed for client-server operations, *X Window* defines a protocol for writing client/server applications based on a graphical user interface (GUI). The idea is to allow a program, called a client, to run on one computer and have it display things through a window server on another computer.

Simple Network Management Protocol (SNMP)

Simple Network Management Protocol (SNMP) collects and manipulates this valuable network information. It gathers data by polling the devices on the network from a management station at fixed or random intervals, requiring them to disclose certain information. When all is well, SNMP receives something called a *baseline*—a report delimiting the operational traits of a healthy network. This protocol can also stand as a watchdog over the network, quickly notifying managers of any sudden turn of events. These network watchdogs are called *agents*, and when aberrations occur, agents send an alert called a *trap* to the management station.

Domain Name Service (DNS)

Domain Name Service (DNS) resolves hostnames—specifically, Internet names, such as `www.routersim.com`. You don't have to use DNS; you can just type in the IP address of any device you want to communicate with. An IP address identifies hosts on a network and the Internet as well. However, DNS was designed to make our lives easier. Think about this: What would happen if you wanted to move your web page to a different service provider? The IP address would change and no one would know what the new one was. DNS allows you to use a domain name to specify an IP address. You can change the IP address as often as you want, and no one will know the difference.

DNS is used to resolve a *fully qualified domain name (FQDN)*—for example, `www.lammle.com` or `todd.lammle.com`. An FQDN is a hierarchy that can logically locate a system based on its domain identifier.

If you want to resolve the name "todd," you either must type in the FQDN of `todd.lammle.com` or have a device such as a PC or router add the suffix for you. For example, on a Cisco router, you can use the command `ip domain-name lammle.com` to append each request with the `lammle.com` domain. If you don't do that, you'll have to type in the FQDN to get DNS to resolve the name.

An important thing to remember about DNS is that if you can ping a device with an IP address but cannot use its FQDN, then you might have some type of DNS configuration failure.

Dynamic Host Configuration Protocol (DHCP)/BootP (Bootstrap Protocol)

Dynamic Host Configuration Protocol (DHCP) gives IP addresses to hosts. It allows easier administration and works well in small-to-even-very-large network environments. All types of hardware can be used as a DHCP server, including a Cisco router.

DHCP differs from BootP in that BootP gives an IP address to a host, but the host's hardware address must be entered manually in a BootP table. You can think of DHCP as a dynamic BootP. But remember that BootP is also used to send an operating system that a host can boot from. DHCP can't do that.

But there is a lot of information a DHCP server can provide to a host when the host is requesting an IP address from the DHCP server. Here's a list of the information a DHCP server can provide:

- IP address
- Subnet mask
- Domain name
- Default gateway (routers)
- DNS
- WINS information

A DHCP server can give us even more information than this, but the items in that list are the most common.

A client that sends out a DHCP Discover message in order to receive an IP address sends out a broadcast at both layer 2 and layer 3. The layer 2 broadcast is all "Fs" in hex, which looks like this: FF:FF:FF:FF:FF:FF. The layer 3 broadcast is 255.255.255.255, which means all networks and all hosts. DHCP is connectionless, which means it uses User Datagram Protocol (UDP) at the Transport layer, also known as the Host-to-Host layer—the layer that we'll talk about next.

The Host-to-Host Layer Protocols

The main purpose of the Host-to-Host layer is to shield the upper-layer applications from the complexities of the network. This layer says to the upper layer, "Just give me your data stream, with any instructions, and I'll begin the process of getting your information ready to send."

The following sections describe the two protocols at this layer:

- Transmission Control Protocol (TCP)
- User Datagram Protocol (UDP)

In addition, we'll look at some of the key host-to-host protocol concepts, as well as the port numbers.

Remember: this is still considered layer 4, and Cisco really likes the way it can use acknowledgments, sequencing, and flow control.

Transmission Control Protocol (TCP)

Transmission Control Protocol (TCP) takes large blocks of information from an application and breaks them into segments. It numbers and sequences each segment so that the destination's TCP protocol can put the segments back into the order the application intended. After these segments are sent, TCP (on the transmitting host) waits for an acknowledgment of the receiving end's TCP virtual circuit session, retransmitting those that aren't acknowledged.

Before a transmitting host starts to send segments down the model, the sender's TCP protocol contacts the destination's TCP protocol to establish a connection. What is created is known as a *virtual circuit*. This type of communication is called *connection-oriented*. During this initial handshake, the two TCP layers also agree on the amount of information that's going to be sent before the recipient's TCP sends back an acknowledgment. With everything agreed upon in advance, the path is paved for reliable communication to take place.

TCP is a full-duplex, connection-oriented, reliable, and accurate protocol, but establishing all these terms and conditions, in addition to error checking, is no small task. TCP is very complicated and, not surprisingly, costly in terms of network overhead. And since today's networks are much more reliable than those of yore, this added reliability is often unnecessary.

TCP Segment Format

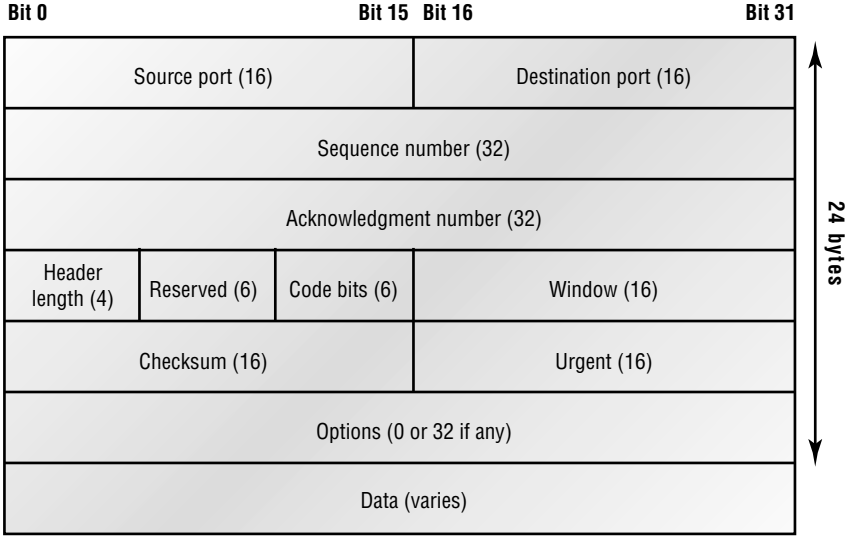
Since the upper layers just send a data stream to the protocols in the Transport layers, I'll demonstrate how TCP segments a data stream and prepares it for the Internet layer. The Internet layer then routes the segments as packets through an internetwork. The segments are handed to the receiving host's Host-to-Host layer protocol, which rebuilds the data stream to hand to the upper-layer applications or protocols.

Figure 2.3 shows the TCP segment format. The figure shows the different fields within the TCP header.

The TCP header is 20 bytes long, or up to 24 bytes with options. You need to understand what each field in the TCP segment is. The TCP segment contains the following fields:

Source port The port number of the application on the host sending the data. (Port numbers will be explained a little later in this section.)

FIGURE 2.3 TCP segment format



Destination port The port number of the application requested on the destination host.

Sequence number Puts the data back in the correct order or retransmits missing or damaged data, a process called *sequencing*.

Acknowledgment number Defines which TCP octet is expected next.

Header length The number of 32-bit words in the TCP header. This indicates where the data begins. The TCP header (even one including options) is an integral number of 32 bits in length.

Reserved Always set to zero.

Code bits Control functions used to set up and terminate a session.

Window The window size the sender is willing to accept, in octets.

Checksum The cyclic redundancy check (CRC), because TCP doesn't trust the lower layers and checks everything. The CRC checks the header and data fields.

Urgent A valid field only if the Urgent pointer in the code bits is set. If so, this value indicates the offset from the current sequence number, in octets, where the first segment of non-urgent data begins.

Options May be 0 or a multiple of 32 bits, if any. What this means is that no options have to be present (option size of 0). However, if any options are used that do not cause the option field to total a multiple of 32 bits, padding of 0s must be used to make sure the data begins on a 32-bit boundary.

Data Handed down to the TCP protocol at the Transport layer, which includes the upper-layer headers.

Let's take a look at a TCP segment copied from a network analyzer:

```
TCP - Transport Control Protocol
 Source Port:      5973
 Destination Port: 23
 Sequence Number:  1456389907
 Ack Number:       1242056456
 Offset:           5
 Reserved:         %000000
 Code:             %011000
      Ack is valid
      Push Request
 Window:           61320
 Checksum:         0x61a6
 Urgent Pointer:   0
 No TCP Options
 TCP Data Area:
 vL.5.+.5.+.5.+.5  76 4c 19 35 11 2b 19 35 11 2b 19 35 11
  2b 19 35 +. 11 2b 19
Frame Check Sequence: 0x0d00000f
```

Did you notice that everything I talked about above is in the segment? As you can see from the number of fields in the header, TCP creates a lot of overhead. Application developers may opt for efficiency over reliability to save overhead, so User Datagram Protocol was also defined at the Transport layer as an alternative.

User Datagram Protocol (UDP)

If you were to compare *User Datagram Protocol (UDP)* with TCP, the former is basically the scaled-down economy model that's sometimes referred to as a thin protocol. Like a thin person on a park bench, a thin protocol doesn't take up a lot of room—or in this case, much bandwidth on a network.

UDP doesn't offer all the bells and whistles of TCP either, but it does do a fabulous job of transporting information that doesn't require reliable delivery—and it does so using far fewer network resources. (UDP is covered thoroughly in Request for Comments 768.)

The Requests for Comments (RFCs) form a series of notes, started in 1969, about the Internet (originally the ARPAnet). The notes discuss many aspects of computer communication, focusing on networking protocols, procedures, programs, and concepts but also including meeting notes, opinion, and sometimes humor.

There are some situations where it would definitely be wise for developers to opt for UDP rather than TCP. Remember the watchdog SNMP up there at the Process/Application layer? SNMP monitors the network, sending intermittent messages and a fairly steady flow of status updates and alerts, especially when running on a large network. The cost in overhead to establish, maintain, and close a TCP connection for each one of those little messages would reduce what would be an otherwise healthy, efficient network to a dammed-up bog in no time!

Another circumstance calling for UDP over TCP is when reliability is already handled at the Process/Application layer. Network File System (NFS) handles its own reliability issues, making the use of TCP both impractical and redundant. But ultimately, it's up to the application developer who decides whether to use UDP or TCP, not the user who wants to transfer data faster.

UDP does *not* sequence the segments and does not care in which order the segments arrive at the destination. But after that, UDP sends the segments off and forgets about them. It doesn't follow through, check up on them, or even allow for an acknowledgment of safe arrival—complete abandonment. Because of this, it's referred to as an unreliable protocol. This does not mean that UDP is ineffective, only that it doesn't handle issues of reliability.

Further, UDP doesn't create a virtual circuit, nor does it contact the destination before delivering information to it. Because of this, it's also considered a *connectionless* protocol. Since UDP assumes that the application will use its own reliability method, it doesn't use any. This gives an application developer a choice when running the Internet Protocol stack: TCP for reliability or UDP for faster transfers.

So if you're using Voice over IP (VoIP), for example, you really don't want to use UDP, because if the segments arrive out of order (very common in IP networks), they'll just pass the segments up to the next OSI (DoD) layer in whatever order they're received, resulting in some

seriously garbled data. On the other hand, TCP sequences the segments so they get put back together in exactly the right order—something UDP just can't do.

UDP Segment Format

Figure 2.4 clearly illustrates UDP's markedly low overhead as compared to TCP's hungry usage. Look at the figure carefully—can you see that UDP doesn't use windowing or provide for acknowledgments in the UDP header?

It's important for you to understand what each field in the UDP segment is. The UDP segment contains the following fields:

Source port Port number of the application on the host sending the data.

Destination port Port number of the application requested on the destination host.

Length Length of UDP header and UDP data.

Checksum Checksum of both the UDP header and UDP data fields.

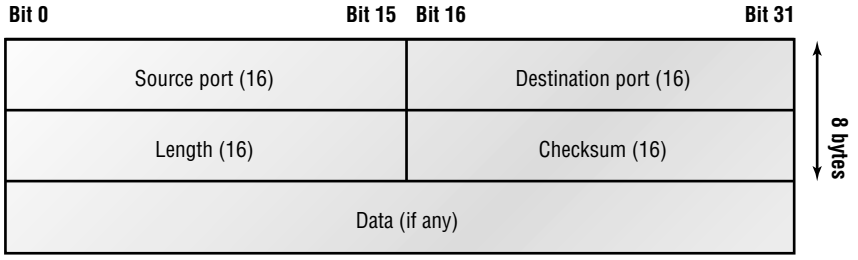
Data Upper-layer data.

UDP, like TCP, doesn't trust the lower layers and runs its own CRC. Remember that the Frame Check Sequence (FCS) is the field that houses the CRC, which is why you can see the FCS information.

The following shows a UDP segment caught on a network analyzer:

```
UDP - User Datagram Protocol
 Source Port:      1085
 Destination Port: 5136
 Length:           41
 Checksum:         0x7a3c
 UDP Data Area:
 ..Z.............  00 01 5a 96 00 01 00 00 00 00 00 11 00
  00 00
...C..2...._C._C  2e 03 00 43 02 1e 32 0a 00 0a 00 80 43
  00 80
Frame Check Sequence: 0x00000000
```

FIGURE 2.4 UDP segment



Notice that low overhead! Try to find the sequence number, ack number, and window size in the UDP segment. You can't (I hope) because they just aren't there!

Key Concepts of Host-to-Host Protocols

Since you've seen both a connection-oriented (TCP) and connectionless (UDP) protocol in action, it would be good to summarize the two here. Table 2.1 highlights some of the key concepts that you should keep in mind regarding these two protocols. You should memorize this table.

TABLE 2.1 Key Features of TCP and UDP

TCP	UDP
Sequenced	Unsequenced
Reliable	Unreliable
Connection-oriented	Connectionless
Virtual circuit	Low overhead
Acknowledgments	No acknowledgment
Windowing flow control	No windowing or flow control

A telephone analogy could really help you understand how TCP works. Most of us know that before you speak to someone on a phone, you must first establish a connection with that other person—wherever they are. This is like a virtual circuit with the TCP protocol. If you were giving someone important information during your conversation, you might say, "You know?" or ask, "Did you get that?" Saying something like this is a lot like a TCP acknowledgment—it's designed to get you verification. From time to time (especially on cell phones), people also ask, "Are you still there?" They end their conversations with a "Goodbye" of some kind, putting closure on the phone call. TCP also performs these types of functions.

Alternately, using UDP is like sending a postcard. To do that, you don't need to contact the other party first. You simply write your message, address the postcard, and mail it. This is analogous to UDP's connectionless orientation. Since the message on the postcard is probably not a matter of life or death, you don't need an acknowledgment of its receipt. Similarly, UDP does not involve acknowledgments.

Port Numbers

TCP and UDP must use *port numbers* to communicate with the upper layers, because they're what keeps track of different conversations crossing the network simultaneously. Originating-source port numbers are dynamically assigned by the source host and will equal some number starting at 1024. 1023 and below are defined in RFC 3232 (or just see `www.iana.org`), which discusses what are called well-known port numbers.

Virtual circuits that don't use an application with a well-known port number are assigned port numbers randomly from a specific range instead. These port numbers identify the source and destination application or process in the TCP segment.

Figure 2.5 illustrates how both TCP and UDP use port numbers.

The different port numbers that can be used are explained next:

- Numbers below 1024 are considered well-known port numbers and are defined in RFC 3232.
- Numbers 1024 and above are used by the upper layers to set up sessions with other hosts, and by TCP to use as source and destination addresses in the TCP segment.

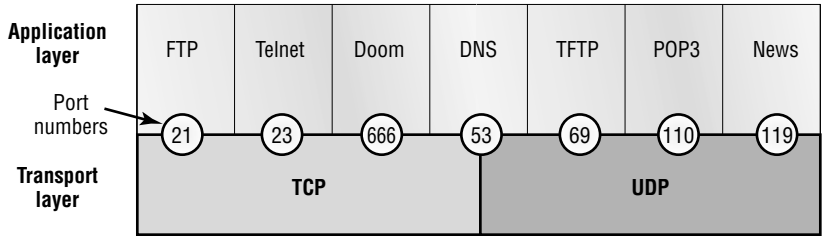
In the following sections we'll take a look at an analyzer output showing a TCP session.

TCP Session: Source Port

The following listing shows a TCP session captured with Etherpeek analyzer software:

```
TCP - Transport Control Protocol
 Source Port:      5973
 Destination Port: 23
 Sequence Number:  1456389907
 Ack Number:       1242056456
 Offset:           5
 Reserved:         %000000
 Code:             %011000
      Ack is valid
      Push Request
 Window:           61320
 Checksum:         0x61a6
 Urgent Pointer:   0
 No TCP Options
 TCP Data Area:
 vL.5.+.5.+.5.+.5  76 4c 19 35 11 2b 19 35 11 2b 19 35 11
  2b 19 35 +. 11 2b 19
Frame Check Sequence: 0x0d00000f
```

FIGURE 2.5 Port numbers for TCP and UDP



Notice that the source host makes up the source port and, in this case, is 5973. The destination port is 23, which is used to tell the receiving host the purpose of the intended connection (Telnet).

By looking at this session, you can see that the source host makes up the source port. But why does the source make up a port number? To differentiate between sessions with different hosts, my friend. How else would a server know where information is coming from if it didn't have a different number from a sending host? TCP and the upper layers don't use hardware and logical addresses to understand the sending host's address as the Data Link and Network layer protocols do. Instead, they use port numbers. And it's easy to imagine the receiving host getting thoroughly confused if all the hosts used the same port number to get to FTP!

TCP Session: Destination Port

You'll sometimes look at an analyzer and see that only the source port is above 1024 and the destination port is a well-known port, as shown in the following Etherpeek trace:

```
TCP - Transport Control Protocol
 Source Port:      1144
 Destination Port: 80 World Wide Web HTTP
 Sequence Number:  9356570
 Ack Number:       0
 Offset:           7
 Reserved:         %000000
 Code:             %000010
      Synch Sequence
 Window:           8192
 Checksum:         0x57E7
 Urgent Pointer:   0
 TCP Options:
  Option Type: 2 Maximum Segment Size
    Length:    4
    MSS:       536
  Option Type: 1 No Operation
  Option Type: 1 No Operation
  Option Type: 4
    Length:    2
    Opt Value:
  No More HTTP Data
Frame Check Sequence: 0x43697363
```

And sure enough, the source port is over 1024, but the destination port is 80, or HTTP service. The server, or receiving host, will change the destination port if it needs to.

In the preceding trace, a "syn" packet is sent to the destination device. The syn sequence is what's telling the remote destination device that it wants to create a session.

TCP Session: Syn Packet Acknowledgment

The next trace shows an acknowledgment to the syn packet:

```
TCP - Transport Control Protocol
 Source Port:      80 World Wide Web HTTP
 Destination Port: 1144
 Sequence Number:  2873580788
 Ack Number:       9356571
 Offset:           6
 Reserved:         %000000
 Code:             %010010
      Ack is valid
      Synch Sequence
 Window:           8576
 Checksum:         0x5F85
 Urgent Pointer:   0
 TCP Options:
  Option Type: 2 Maximum Segment Size
    Length:    4
    MSS:       1460
  No More HTTP Data
Frame Check Sequence: 0x6E203132
```

Notice the *Ack is valid*, which means that the source port was accepted and the device agreed to create a virtual circuit with the originating host.

And here again, you can see that the response from the server shows the source is 80 and the destination is the 1144 sent from the originating host—all's well.

Table 2.2 gives you a list of the typical applications used in the TCP/IP suite, their well-known port numbers, and the Transport layer protocols used by each application or process.

TABLE 2.2 Key Protocols That Use TCP and UDP

TCP	UDP
Telnet 23	SNMP 161
SMTP 25	TFTP 69
HTTP 80	DNS 53
FTP 21	
DNS 53	
HTTPS 443	

Notice that DNS uses both TCP and UDP. Whether it opts for one or the other depends on what it's trying to do. Even though it's not the only application that can use both protocols, it's certainly one that you should remember in your studies.

The Internet Layer Protocols

In the DoD model, there are two main reasons for the Internet layer's existence: routing, and providing a single network interface to the upper layers.

None of the other upper- or lower-layer protocols have any functions relating to routing—that complex and important task belongs entirely to the Internet layer. The Internet layer's second duty is to provide a single network interface to the upper-layer protocols. Without this layer, application programmers would need to write "hooks" into every one of their applications for each different Network Access protocol. This would not only be a pain in the neck, but it would lead to different versions of each application—one for Ethernet, another one for Token Ring, and so on. To prevent this, IP provides one single network interface for the upper-layer protocols. That accomplished, it's then the job of IP and the various Network Access protocols to get along and work together.

All network roads don't lead to Rome—they lead to IP. And all the other protocols at this layer, as well as all those at the upper layers, use it. Never forget that. All paths through the DoD model go through IP. The following sections describe the protocols at the Internet layer:

- Internet Protocol (IP)
- Internet Control Message Protocol (ICMP)
- Address Resolution Protocol (ARP)
- Reverse Address Resolution Protocol (RARP)
- Proxy ARP

Internet Protocol (IP)

Internet Protocol (IP) essentially is the Internet layer. The other protocols found here merely exist to support it. IP holds the big picture and could be said to "see all," in that it's aware of all the interconnected networks. It can do this because all the machines on the network have a software, or logical, address called an IP address, which I'll cover more thoroughly later in this chapter.

IP looks at each packet's address. Then, using a routing table, it decides where a packet is to be sent next, choosing the best path. The protocols of the Network Access layer at the bottom of the DoD model don't possess IP's enlightened scope of the entire network; they deal only with physical links (local networks).

Identifying devices on networks requires answering these two questions: Which network is it on? And what is its ID on that network? The first answer is the *software address*, or *logical address* (the correct street). The second answer is the hardware address (the correct mailbox). All hosts on a network have a logical ID called an IP address. This is the software, or logical, address and contains valuable encoded information, greatly simplifying the complex task of routing. (IP is discussed in RFC 791.)

IP receives segments from the Host-to-Host layer and fragments them into datagrams (packets) if necessary. IP then reassembles datagrams back into segments on the receiving side. Each datagram is assigned the IP address of the sender and of the recipient. Each router (layer 3 device) that receives a datagram makes routing decisions based on the packet's destination IP address.

Figure 2.6 shows an IP header. This will give you an idea of what the IP protocol has to go through every time user data is sent from the upper layers and is to be sent to a remote network.

The following fields make up the IP header:

Version IP version number.

Header Length Header length (HLEN) in 32-bit words.

ToS with IP Precedence Bits Type of Service tells how the datagram should be handled. The first 3 bits are the priority bits.

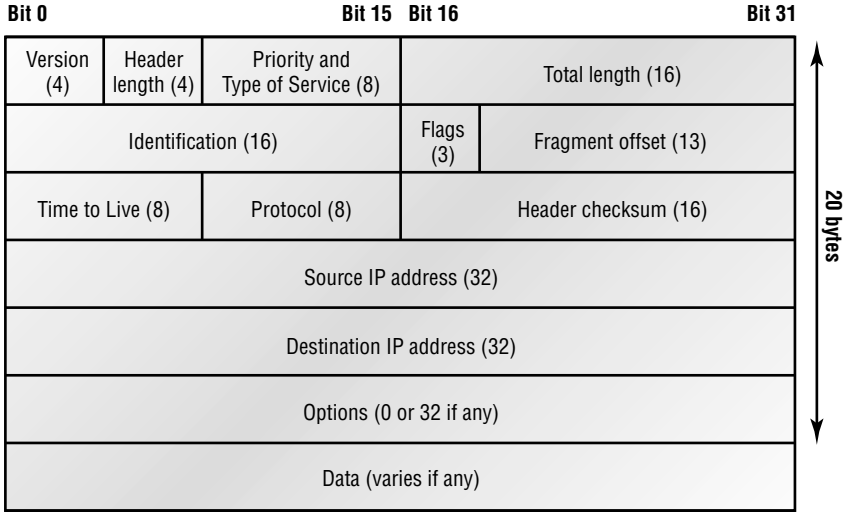
Total length Length of the packet including header and data.

Identifier Unique IP-packet value.

Flags Specifies whether fragmentation should occur.

Frag offset Provides fragmentation and reassembly if the packet is too large to put in a frame. It also allows different maximum transmission units (MTUs) on the Internet.

FIGURE 2.6 IP header



TTL The time to live (TTL) is set into a packet when it is originally generated. If it doesn't get to where it wants to go before the TTL expires, boom—it's gone. This stops IP packets from continuously circling the network looking for a home.

Protocol Port of upper-layer protocol (TCP is port 6 or UDP is port 17 [hex]). Also supports Network layer protocols.

Header checksum Cyclic redundancy check (CRC) on header only.

Source IP address 32-bit IP address of sending station.

Destination IP address 32-bit IP address of the station this packet is destined for.

IP options Used for network testing, debugging, security, and more.

Data After the IP option field will be the upper-layer data.

Here's a snapshot of an IP packet caught on a network analyzer (notice that all the header information discussed above appears here):

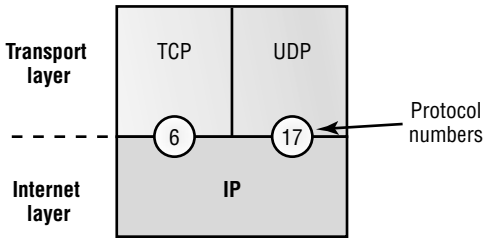
```
IP Header - Internet Protocol Datagram
 Version:              4
 Header Length:        5
 Precedence:           0
 Type of Service:      %000
 Unused:               %00
 Total Length:         187
 Identifier:           22486
 Fragmentation Flags: %010 Do Not Fragment
 Fragment Offset:      0
 Time To Live:         60
 IP Type:              0x06 TCP
 Header Checksum:      0xd031
 Source IP Address:    10.7.1.30
 Dest. IP Address:     10.7.1.10
 No Internet Datagram Options
```

Can you distinguish the logical, or IP, addresses in this header?

The Type field—it's typically a Protocol field, but this analyzer sees it as an IP Type field—is important. If the header didn't carry the protocol information for the next layer, IP wouldn't know what to do with the data carried in the packet. The example above tells IP to hand the segment to TCP.

Figure 2.7 demonstrates how the Network layer sees the protocols at the Transport layer when it needs to hand a packet to the upper-layer protocols.

FIGURE 2.7 The Protocol field in an IP header



In this example, the Protocol field tells IP to send the data to either TCP port 6 or UDP port 17 (both hex addresses). But it will only be UDP or TCP if the data is part of a data stream headed for an upper-layer service or application. It could just as easily be destined for Internet Control Message Protocol (ICMP), Address Resolution Protocol (ARP), or some other type of Network layer protocol.

Table 2.3 is a list of some other popular protocols that can be specified in the Protocol field.

TABLE 2.3 Possible Protocols Found in the Protocol Field of an IP Header

Protocol	Protocol Number
ICMP	1
IGRP	9
EIGRP	88
OSPF	89
IPv6	41
GRE	47
IPX in IP	111
Layer 2 tunnel (L2TP)	115

Internet Control Message Protocol (ICMP)

Internet Control Message Protocol (ICMP) works at the Network layer and is used by IP for many different services. ICMP is a management protocol and messaging service provider for IP. Its messages are carried as IP datagrams. RFC 1256 is an annex to ICMP, which affords hosts' extended capability in discovering routes to gateways.

Periodically, router advertisements are announced over the network, reporting IP addresses for the router's network interfaces. Hosts listen for these network infomercials to acquire route information. A router solicitation is a request for immediate advertisements and may be sent by a host when it starts up.

RFC 792 references ICMP and describes how ICMP must be implemented by all TCP/IP hosts.

The following are some common events and messages that ICMP relates to:

Destination Unreachable If a router can't send an IP datagram any further, it uses ICMP to send a message back to the sender, advising it of the situation. For example, if a router receives a packet destined for a network that the router doesn't know about, it will send an ICMP Destination Unreachable message back to the sending station.

Buffer Full If a router's memory buffer for receiving incoming datagrams is full, it will use ICMP to send out this message until the congestion abates.

Hops Each IP datagram is allotted a certain number of routers, called hops, to pass through. If it reaches its limit of hops before arriving at its destination, the last router to receive that datagram deletes it. The executioner router then uses ICMP to send an obituary message, informing the sending machine of the demise of its datagram.

Ping Ping (Packet Internet Groper) uses ICMP echo messages to check the physical and logical connectivity of machines on an internetwork.

Traceroute Using ICMP timeouts, Traceroute is used to discover the path a packet takes as it traverses an internetwork.

Both Ping and Traceroute (also just called Trace; Microsoft Windows uses tracert) allow you to verify address configurations in your internetwork.

The following data is from a network analyzer catching an ICMP echo request:

```
Flags:          0x00
 Status:         0x00
 Packet Length: 78
 Timestamp:     14:04:25.967000 12/20/03
Ethernet Header
 Destination: 00:a0:24:6e:0f:a8
 Source:      00:80:c7:a8:f0:3d
 Ether-Type:  08-00 IP
IP Header - Internet Protocol Datagram
 Version:             4
```

```
 Header Length:       5
 Precedence:          0
 Type of Service:     %000
 Unused:              %00
 Total Length:        60
 Identifier:          56325
 Fragmentation Flags: %000
 Fragment Offset:     0
 Time To Live:        32
 IP Type:             0x01 ICMP
 Header Checksum:     0x2df0
 Source IP Address:   100.100.100.2
 Dest. IP Address:    100.100.100.1
 No Internet Datagram Options
ICMP - Internet Control Messages Protocol
 ICMP Type:       8 Echo Request
 Code:            0
 Checksum:        0x395c
 Identifier:      0x0300
 Sequence Number: 4352
 ICMP Data Area:
 abcdefghijklmnop  61 62 63 64 65 66 67 68 69 6a 6b 6c 6d
 qrstuvwabcdefghi  71 72 73 74 75 76 77 61 62 63 64 65 66
Frame Check Sequence: 0x00000000
```

Notice anything unusual? Did you catch the fact that even though ICMP works at the Internet (Network) layer, it still uses IP to do the Ping request? The Type field in the IP header is 0x01, which specifies the ICMP protocol.

The Ping program just uses the alphabet in the data portion of the packet as a payload, 100 bytes by default.

If you remember reading about the Data Link layer and the different frame types in Chapter 1, you should be able to look at the preceding trace and tell what type of Ethernet frame this is. The only fields are destination hardware address, source hardware address, and Ether-Type. The only frame that uses an Ether-Type field exclusively is an Ethernet_II frame. (SNAP uses an Ether-Type field also, but only within an 802.2 LLC field, which isn't present in the frame.)

Address Resolution Protocol (ARP)

Address Resolution Protocol (ARP) finds the hardware address of a host from a known IP address. Here's how it works: When IP has a datagram to send, it must inform a Network

Access protocol, such as Ethernet or Token Ring, of the destination's hardware address on the local network. (It has already been informed by upper-layer protocols of the destination's IP address.) If IP doesn't find the destination host's hardware address in the ARP cache, it uses ARP to find this information.

As IP's detective, ARP interrogates the local network by sending out a broadcast asking the machine with the specified IP address to reply with its hardware address. So basically, ARP translates the software (IP) address into a hardware address—for example, the destination machine's Ethernet board address—and from it, deduces its whereabouts on LAN by broadcasting for this address. Figure 2.8 shows how an ARP looks to a local network.

ARP resolves IP addresses to Ethernet (MAC) addresses.

The following trace shows an ARP broadcast—notice that the destination hardware address is unknown, and is all Fs in hex (all 1s in binary)—and is a hardware address broadcast:

```
 Flags:          0x00
 Status:         0x00
 Packet Length: 64
 Timestamp:      09:17:29.574000 12/06/03
Ethernet Header
 Destination:    FF:FF:FF:FF:FF:FF Ethernet Broadcast
 Source:         00:A0:24:48:60:A5
 Protocol Type: 0x0806 IP ARP
ARP - Address Resolution Protocol
 Hardware:                 1 Ethernet (10Mb)
 Protocol:                 0x0800 IP
 Hardware Address Length: 6
 Protocol Address Length: 4
 Operation:                1 ARP Request
 Sender Hardware Address: 00:A0:24:48:60:A5
 Sender Internet Address: 172.16.10.3
 Target Hardware Address: 00:00:00:00:00:00 (ignored)
 Target Internet Address: 172.16.10.10
Extra bytes (Padding):
 ................ 0A 0A 0A 0A 0A 0A 0A 0A 0A 0A 0A 0A 0A
  0A 0A 0A 0A 0A
Frame Check Sequence: 0x00000000
```

Reverse Address Resolution Protocol (RARP)

When an IP machine happens to be a diskless machine, it has no way of initially knowing its IP address. But it does know its MAC address. *Reverse Address Resolution Protocol (RARP)*

discovers the identity of the IP address for diskless machines by sending out a packet that includes its MAC address and a request for the IP address assigned to that MAC address. A designated machine, called a *RARP server*, responds with the answer, and the identity crisis is over. RARP uses the information it does know about the machine's MAC address to learn its IP address and complete the machine's ID portrait.

RARP resolves Ethernet (MAC) addresses to IP addresses.

Figure 2.9 shows a diskless workstation asking for its IP address with a RARP broadcast:

FIGURE 2.8 Local ARP broadcast

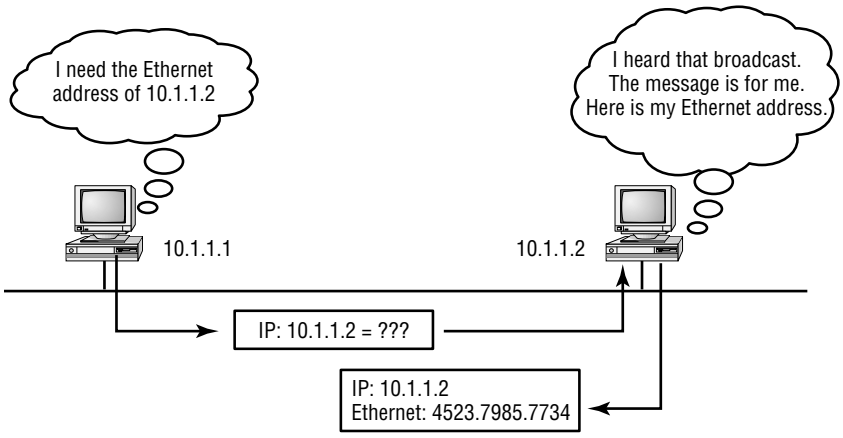
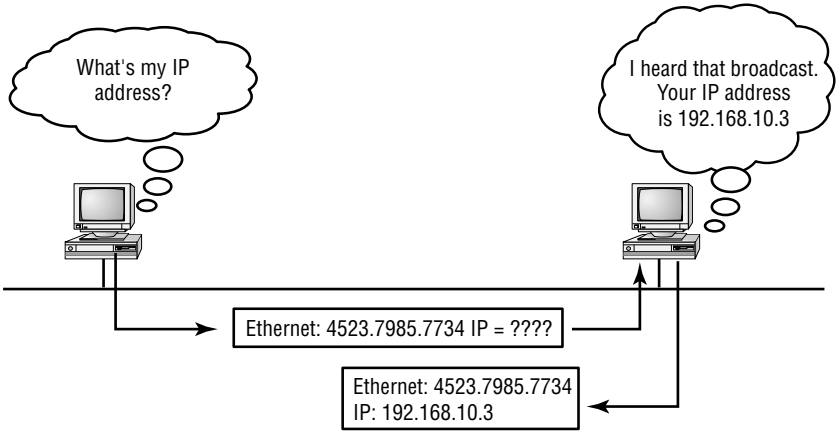


FIGURE 2.9 RARP broadcast example



Proxy Address Resolution Protocol (Proxy ARP)

On a network, your hosts can't have more then one default-gateway configured. Think about this…What if the default-gateway (router) happens to go down? The host won't just start sending to another router automatically—you've got to reconfigure that host. But Proxy ARP can actually help machines on a subnet reach remote subnets without configuring routing or even a default gateway.

One advantage of using Proxy ARP is that is can be added to a single router on a network without disturbing the routing tables of all the other routers that live there too. But there's a serious downside to using Proxy ARP. Using Proxy ARP will definitely increase the amount of traffic on your network segment, and hosts will have a larger ARP table than usual in order to handle all the IP-to-MAC address mappings. And Proxy ARP is configured on all Cisco routers by default—you should disable it if you don't think you're going to use it.

One last thought on Proxy ARP: Proxy ARP really isn't really a separate protocol. It is a service run by routers on behalf of other devices that are separated from their query by a router, although they think they share the subnet with the other device.

Binary to Decimal and Hexadecimal Conversion

Before we continue discussing the TCP/IP protocol stack and IP addressing, it's really important for you to truly understand the differences between binary, decimal, and hexadecimal numbers, and how to convert one format into the other.

So we'll start with binary numbering. It's pretty simple, really. The digits used are limited to either a 1 (one) or a 0 (zero), with each digit being called one bit (short for *bi*nary digi*t*). Typically, you count either 4 or 8 bits together, with these being referred to as a nibble or a byte, respectively.

What interests us in binary numbering is the value represented in a decimal format—the typical decimal format being the base 10 number scheme that we've all used since kindergarten. The binary numbers are placed in a value spot: starting at the right and moving left, with each spot having double the value of the previous spot.

Table 2.4 shows the decimal values of each bit location in a nibble and a byte. Remember, a nibble is 4 bits and a byte is 8 bits.

TABLE 2.4 Binary Values

Nibble values	Byte values
8 4 2 1	128 64 32 16 8 4 2 1

What all this means is that if a one digit (1) is placed in a value spot, then the nibble or byte takes on that decimal value, and adds it to any other value spots that have a one. And if a zero (0) is placed in a bit spot, then you don't count that value.

Let me clarify things. If we have a 1 placed in each spot of our nibble, we would then add up 8 + 4 + 2 + 1, to give us a maximum value of 15. Another example for our nibble values would be 1010, which means that the 8 bit and the 2 bit are turned on, which equals a decimal value of 10. If we have a nibble binary value of 0110, then our decimal value would be 6, because the 4 and 2 bits are turned on.

But the byte values can add up to a value that's significantly higher than 15. This is how: If we counted every bit as a one (1), then the byte binary value would look like this (remember, 8 bits equal a byte):

11111111

We would then count up every bit spot because each is turned on. It would look like this:

128 + 64 + 32 + 16 + 8 + 4 + 2 + 1 = 255

which demonstrates the maximum value of a byte.

There are plenty of other decimal values that a binary number can equal. Let's work through a few examples:

10010110

Which bits are on? The 128, 16, 4, and 2 bits are on, so we'll just add them up: 128 + 16 + 4 + 2 = 150.

01101100

Which bits are on? The 64, 32, 8, and 4 bits are on, so we just need to add them up: 64 + 32 + 8 + 4 = 108.

11101000

Which bits are on? The 128, 64, 32 and 8 bits are on, so just add the values up: 128 + 64 + 32 + 8 = 232

Table 2.5 is a table you should memorize before braving the subnetting section in Chapter 3.

TABLE 2.5 Binary to Decimal Memorization Chart

Binary Value	Decimal Value
10000000	128
11000000	192
11100000	224
11110000	240
11111000	248
11111100	252

TABLE 2.5 Binary to Decimal Memorization Chart *(continued)*

Binary Value	Decimal Value
11111110	254
11111111	255

Perhaps it is needless to say (but just in case): You need to understand binary-to-decimal conversion before moving on to Chapter 3.

Hexadecimal addressing is completely different than binary or decimal—it's converted by reading nibbles, not bytes. By using a nibble, we can convert these bits to hex pretty simply. First, understand that the hexadecimal addressing scheme uses only the numbers 0 through 9. And since the numbers 10, 11, 12, etc. can't be used (because they are two digits), the letters A, B, C, D, E, and F are used to represent 10, 11, 12, 13, 14, and 15, respectively.

Table 2.6 shows both the binary value and the decimal value for each hexadecimal digit.

TABLE 2.6 Hex to Binary to Decimal Chart

Hexadecimal Value	Binary Value	Decimal Value
0	0000	0
1	0001	1
2	0010	2
3	0011	3
4	0100	4
5	0101	5
6	0110	6
7	0111	7
8	1000	8
9	1001	9
A	1010	10
B	1011	11

TABLE 2.6 Hex to Binary to Decimal Chart *(continued)*

Hexadecimal Value	Binary Value	Decimal Value
C	1100	12
D	1101	13
E	1110	14
F	1111	15

Did you notice that the first 10 hexadecimal digits (0–9) are the same value as the decimal values? If not, look again. This handy fact makes those values super easy to convert.

So suppose you have something like this: 0x6A. (Sometimes Cisco likes to put 0x in front of characters so you know that they are a hex value. They don't have any other special meaning.) What are the binary and decimal values? All you have to remember is that each hex character is one nibble and two hex characters together make a byte. To figure out the binary value, we need to put the hex characters into two nibbles, then put them together into a byte. 6 = 0110 and A (which is 10 in hex) = 0101, so the complete byte would be 01101010.

To convert from binary to hex, just take the byte and break it into nibbles. Here's what I mean:

Say you have the binary number 01010101. First, break it into nibbles—0101 and 0101—with the value of each nibble being 5, since the 1 and 4 bits are on. This makes the hex answer 55. Why? Because we're converting a whole byte. And in decimal format, the binary number is 01101010, which converts to 64 + 32 + 8 + 2 = 106.

Here's another binary number:

11001100

Your answer would be 1100 = 12 and 1100 = 12 (therefore, it's converted to CC in hex). The decimal conversion answer would be 128 + 64 + 8 + 4 = 204.

One more example, then we've got to move on into IP addressing. Suppose you had the following binary number:

10110101

The hex answer would be B5, since 1011 converts to B and 1010 converts to 5 in hex value. The decimal equivalent is 128 + 32 + 16 + 4 + 1 = 181.

IP Addressing

One of the most important topics in any discussion of TCP/IP is IP addressing. An *IP address* is a numeric identifier assigned to each machine on an IP network. It designates the specific location of a device on the network.

An IP address is a software address, not a hardware address—the latter is hard-coded on a Network Interface Card (NIC) and used for finding hosts on a local network. IP addressing was

designed to allow a host on one network to communicate with a host on a different network, regardless of the type of LANs the hosts are participating in.

Before we get into the more complicated aspects of IP addressing, you need to understand some of the basics. First I'm going to explain some of the fundamentals of IP addressing and its terminology. Then you'll learn about the hierarchical IP addressing scheme and private IP addresses.

IP Terminology

Throughout this chapter you'll learn several important terms vital to your understanding of the Internet Protocol. Here are a few to get you started:

Bit A *bit* is one digit, either a 1 or a 0.

Byte A *byte* is 7 or 8 bits, depending on whether parity is used. For the rest of this chapter, always assume a byte is 8 bits.

Octet An octet, made up of 8 bits, is just an ordinary 8-bit binary number. In this chapter, the terms byte and octet are completely interchangeable.

Network address This is the designation used in routing to send packets to a remote network—for example, 10.0.0.0, 172.16.0.0, and 192.168.10.0.

Broadcast address The address used by applications and hosts to send information to all nodes on a network is called the *broadcast address*. Examples include 255.255.255.255, which is all networks, all nodes; 172.16.255.255, which is all subnets and hosts on network 172.16.0.0; and 10.255.255.255, which broadcasts to all subnets and hosts on network 10.0.0.0.

The Hierarchical IP Addressing Scheme

An IP address consists of 32 bits of information. These bits are divided into four sections, referred to as *octets* or bytes, each containing 1 byte (8 bits). You can depict an IP address using one of three methods:

- Dotted-decimal, as in 172.16.30.56
- Binary, as in 10101100.00010000.00011110.00111000
- Hexadecimal, as in AC.10.1E.38

All these examples truly represent the same IP address. Hexadecimal isn't used as often as dotted-decimal or binary when IP addressing is discussed, but you still might find an IP address stored in hexadecimal in some programs. The Windows Registry is a good example of a program that stores a machine's IP address in hex.

The 32-bit IP address is a structured or hierarchical address, as opposed to a flat or nonhierarchical address. Although either type of addressing scheme could have been used, *hierarchical addressing* was chosen for a good reason. The advantage of this scheme is that it can handle a large number of addresses, namely 4.3 billion (a 32-bit address space with two possible values for each position—either 0 or 1—gives you 2^{32}, or 4,294,967,296). The disadvantage of the flat addressing scheme, and the reason it's not used for IP addressing, relates to routing. If every address were unique, all routers on the Internet would need to store the address of each and every machine on

the Internet. This would make efficient routing impossible, even if only a fraction of the possible addresses were used.

The solution to this problem is to use a two- or three-level, hierarchical addressing scheme that is structured by network and host, or network, subnet, and host.

This two- or three-level scheme is comparable to a telephone number. The first section, the area code, designates a very large area. The second section, the prefix, narrows the scope to a local calling area. The final segment, the customer number, zooms in on the specific connection. IP addresses use the same type of layered structure. Rather than all 32 bits being treated as a unique identifier, as in flat addressing, a part of the address is designated as the network address, and the other part is designated as either the subnet and host or just the node address.

In the following sections, I'm going to discuss IP network addressing and the different classes of address we can use to address our networks with.

Network Addressing

The *network address* (which can also be called the network number) uniquely identifies each network. Every machine on the same network shares that network address as part of its IP address. In the IP address 172.16.30.56, for example, 172.16 is the network address.

The *node address* is assigned to, and uniquely identifies, each machine on a network. This part of the address must be unique because it identifies a particular machine—an individual—as opposed to a network, which is a group. This number can also be referred to as a *host address*. In the sample IP address 172.16.30.56, the 30.56 is the node address.

The designers of the Internet decided to create classes of networks based on network size. For the small number of networks possessing a very large number of nodes, they created the rank *Class A network*. At the other extreme is the *Class C network*, which is reserved for the numerous networks with a small number of nodes. The class distinction for networks between very large and very small is predictably called the *Class B network*.

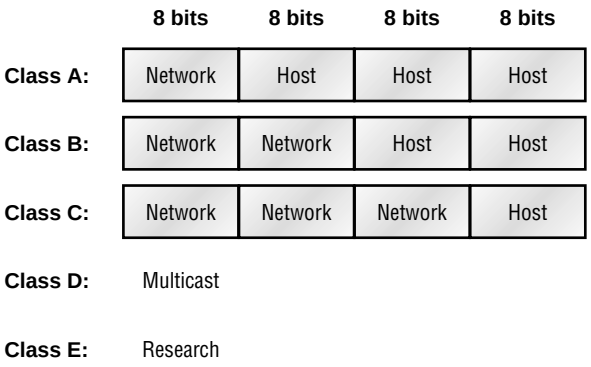
Subdividing an IP address into a network and node address is determined by the class designation of one's network. Figure 2.10 summarizes the three classes of networks—a subject I'll explain in much greater detail throughout this chapter.

To ensure efficient routing, Internet designers defined a mandate for the leading-bits section of the address for each different network class. For example, since a router knows that a Class A network address always starts with a 0, the router might be able to speed a packet on its way after reading only the first bit of its address. This is where the address schemes define the difference between a Class A, a Class B, and a Class C address. In the next sections, I'll discuss the differences between these three classes, followed by a discussion of the Class D and Class E addresses (class A, B and C are the only ranges that are used to address hosts in our networks).

Network Address Range: Class A

The designers of the IP address scheme said that the first bit of the first byte in a Class A network address must always be off, or 0. This means a Class A address must be between 0 and 127, inclusive.

FIGURE 2.10 Summary of the three classes of networks



Consider the following network address:

0xxxxxxx

If we turn the other 7 bits all off and then turn them all on, we'll find the Class A range of network addresses:

00000000 = 0
01111111 = 127

So, a Class A network is defined in the first octet between 0 and 127, and it can't be less or more. (yes, I know 0 and 127 are not valid in a class A network—I'll talk about illegal addresses in a minute.)

Network Address Range: Class B

In a Class B network, the RFCs state that the first bit of the first byte must always be turned on, but the second bit must always be turned off. If you turn the other 6 bits all off and then all on, you will find the range for a Class B network:

10000000 = 128
10111111 = 191

As you can see, a Class B network is defined when the first byte is configured from 128 to 191.

Network Address Range: Class C

For Class C networks, the RFCs define the first 2 bits of the first octet as always turned on, but the third bit can never be on. Following the same process as the previous classes, convert from binary to decimal to find the range. Here's the range for a Class C network:

11000000 = 192
11011111 = 223

So, if you see an IP address that starts at 192 and goes to 223, you'll know it is a Class C IP address.

Network Address Ranges: Classes D and E

The addresses between 224 and 255 are reserved for Class D and E networks. Class D (224–239) is used for multicast addresses and Class E (240–255) for scientific purposes, but I'm not going into these types of addresses in this book (and you don't need to know them).

Network Addresses: Special Purpose

Some IP addresses are reserved for special purposes, so network administrators can't ever assign these addresses to nodes. Table 2.7 lists the members of this exclusive little club and the reasons why they're included in it.

TABLE 2.7 Reserved IP Addresses

Address	Function
Network address of all 0s	Interpreted to mean "this network or segment."
Network address of all 1s	Interpreted to mean "all networks."
Network 127.0.0.1	Reserved for loopback tests. Designates the local node and allows that node to send a test packet to itself without generating network traffic.
Node address of all 0s	Interpreted to mean "network address" or any host on specified network.
Node address of all 1s	Interpreted to mean "all nodes" on the specified network; for example, 128.2.255.255 means "all nodes" on network 128.2 (Class B address).
Entire IP address set to all 0s	Used by Cisco routers to designate the default route. Could also mean "any network."
Entire IP address set to all 1s (same as 255.255.255.255)	Broadcast to all nodes on the current network; sometimes called an "all 1s broadcast" or limited broadcast.

Class A Addresses

In a Class A network address, the first byte is assigned to the network address, and the three remaining bytes are used for the node addresses. The Class A format is:

network.node.node.node

For example, in the IP address 49.22.102.70, the 49 is the network address, and 22.102.70 is the node address. Every machine on this particular network would have the distinctive network address of 49.

Class A network addresses are one byte long, with the first bit of that byte reserved and the 7 remaining bits available for manipulation (addressing). As a result, the maximum number of Class A networks that can be created is 128. Why? Because each of the 7 bit positions can be either a 0 or a 1, thus 2^7 or 128.

To complicate matters further, the network address of all 0s (0000 0000) is reserved to designate the default route (see Table 2.7 in the previous section). Additionally, the address 127, which is reserved for diagnostics, can't be used either, which means that you can really only use the numbers 1 to 126 to designate Class A network addresses. This means the actual number of usable Class A network addresses is 128 minus 2, or 126.

Each Class A address has three bytes (24-bit positions) for the node address of a machine. This means there are 2^{24}—or 16,777,216—unique combinations and, therefore, precisely that many possible unique node addresses for each Class A network. Because node addresses with the two patterns of all 0s and all 1s are reserved, the actual maximum usable number of nodes for a Class A network is 2^{24} minus 2, which equals 16,777,214. Either way, that's a huge amount of hosts on a network segment!

Class A Valid Host IDs

Here's an example of how to figure out the valid host IDs in a Class A network address:

- All host bits off is the network address: 10.0.0.0.
- All host bits on is the broadcast address: 10.255.255.255.

The valid hosts are the numbers in between the network address and the broadcast address: 10.0.0.1 through 10.255.255.254. Notice that 0s and 255s can be valid host IDs. All you need to remember when trying to find valid host addresses is that the host bits can't all be turned off or all be on at the same time.

Class B Addresses

In a Class B network address, the first two bytes are assigned to the network address and the remaining two bytes are used for node addresses. The format is:

```
network.network.node.node
```

For example, in the IP address 172.16.30.56, the network address is 172.16, and the node address is 30.56.

With a network address being two bytes (8 bits each), there would be 2^{16} unique combinations. But the Internet designers decided that all Class B network addresses should start with the binary digit 1, then 0. This leaves 14 bit positions to manipulate, therefore 16,384 (that is, 2^{14}) unique Class B network addresses.

A Class B address uses two bytes for node addresses. This is 2^{16} minus the two reserved patterns (all 0s and all 1s), for a total of 65,534 possible node addresses for each Class B network.

Class B Valid Host IDs

Here's an example of how to find the valid hosts in a Class B network:

- All host bits turned off is the network address: 172.16.0.0.
- All host bits turned on is the broadcast address: 172.16.255.255.

The valid hosts would be the numbers in between the network address and the broadcast address: 172.16.0.1 through 172.16.255.254.

Class C Addresses

The first three bytes of a Class C network address are dedicated to the network portion of the address, with only one measly byte remaining for the node address. The format is

network.network.network.node

Using the example IP address 192.168.100.102, the network address is 192.168.100, and the node address is 102.

In a Class C network address, the first three bit positions are always the binary 110. The calculation is: 3 bytes, or 24 bits, minus 3 reserved positions, leaves 21 positions. Hence, there are 2^{21}, or 2,097,152, possible Class C networks.

Each unique Class C network has one byte to use for node addresses. This leads to 2^8 or 256, minus the two reserved patterns of all 0s and all 1s, for a total of 254 node addresses for each Class C network.

Class C Valid Host IDs

Here's an example of how to find a valid host ID in a Class C network:

- All host bits turned off is the network ID: 192.168.100.0.
- All host bits turned on is the broadcast address: 192.168.100.255.

The valid hosts would be the numbers in between the network address and the broadcast address: 192.168.100.1 through 192.168.100.254.

Private IP Addresses

The people who created the IP addressing scheme also created what we call private IP addresses. These addresses can be used on a private network, but they're not routable through the Internet. This is designed for the purpose of creating a measure of well-needed security, but it also conveniently saves valuable IP address space.

If every host on every network had to have real routable IP addresses, we would have run out of IP addresses to hand out years ago. But by using private IP addresses, ISPs, corporations, and home users only need a relatively tiny group of bona fide IP addresses to connect their networks to the Internet. This is economical because they can use private IP addresses on their inside networks and get along just fine.

To accomplish this task, the ISP and the corporation—the end user, no matter who they are—need to use something called a *Network Address Translation (NAT)*, which basically takes

a private IP address and converts it for use on the Internet. Many people can use the same real IP address to transmit out onto the Internet. Doing things this way saves megatons of address space—good for us all!

I'll provide an introduction to what NAT is in the section "Introduction to Network Address Translation (NAT)."

The reserved private addresses are listed in Table 2.8.

TABLE 2.8 Reserved IP Address Space

Address Class	Reserved address space
Class A	10.0.0.0 through 10.255.255.255
Class B	172.16.0.0 through 172.31.255.255
Class C	192.168.0.0 through 192.168.255.255

Real World Scenario

So, what private IP address should I use?

That's a really great question: Should you use Class A, Class B, or even Class C private addressing when setting up your network? Let's take Acme Corporation in San Francisco as an example. This company is moving into a new building and needs a whole new network (what a treat this is!). They have 14 departments, with about 70 users in each. You could probably squeeze one or two Class C addresses to use, or maybe you could use a Class B, or even a Class A, just for fun.

The rule of thumb in the consulting world is, when you're setting up a corporate network—regardless of how small it is—you should use a Class A network address because it gives you the most flexibility and growth options. For example, if you used the 10.0.0.0 network address with a /24 mask, then you'd have 65,536 networks, each with 254 hosts. Lots of room for growth with that network!

But if you're setting up a home network, you'd opt for a Class C address because it is the easiest for people to understand and configure. Using the default Class C mask gives you one network with 254 hosts—plenty for a home network.

With the Acme Corporation, a nice 10.1.x.0 with a /24 mask (the x is the subnet for each department) makes this easy to design, install, and troubleshoot.

Broadcast Addresses

Even though I've referred to broadcast addresses throughout Chapters 1 and 2, I really haven't gone into their different terms and uses. Here are the four different types I'd like to define:

Layer 2 broadcasts These are sent to all nodes on a LAN.

Broadcasts (layer 3) These are sent to all nodes on the network.

Unicast These are sent to a single destination host.

Multicast These are packets sent from a single source, and transmitted to many devices on different networks.

First, understand that layer 2 broadcasts are also known as hardware broadcasts—they only go out on a LAN, and they usually don't go past the LAN boundary (router) unless they become a unicast packet (discussed in a minute). The typical hardware address is 6 bytes (48 bits) and looks something like 0c.43.a4.f3.12.c2. The broadcast would be all 1s in binary and all Fs in hexadecimal, as in FF.FF.FF.FF.FF.FF.

Then there's the plain old broadcast addresses at layer 3. Broadcast messages are meant to reach all hosts on a broadcast domain. These are the network broadcasts that have all host bits on. Here's an example that you're already familiar with: The network address of 172.16.0.0 255.255.0.0 would have a broadcast address of 172.16.255.255—all host bits on. Broadcasts can also be "all networks and all hosts," as indicated by 255.255.255.255. A good example of a broadcast message is an Address Resolution Protocol (ARP) request. When a host has a packet, it knows the logical address (IP) of the destination. To get the packet to the destination, the host needs to forward the packet to a default gateway if the destination resides on a different IP network. If the destination is on the local network, the source will forward the packet directly to the destination. Because the source doesn't have the MAC address it needs to forward the frame to, it sends out a broadcast, something that every device in the local broadcast domain will listen to. This broadcast says, in essence, "If you are the owner of IP address 192.168.2.3, please forward your MAC address to me," with the source giving the appropriate information.

A unicast is different because it's a broadcast that becomes directed to an actual destination IP address—in other words, it's directed to a specific host, and a DHCP client request is a good example of how a unicast works. Here's an example: Your host on a LAN sends out an FF.FF.FF.FF.FF.FF layer 2 broadcast and 255.255.255.255 layer 3 destination broadcast looking for a DHCP server on the LAN. The router will see that this is a broadcast meant for the DHCP server because it has a destination port number of 67 (BootP server), and will forward the request to the IP address of the DHCP server on another LAN. So, basically, if your DHCP server IP address is 172.16.10.1, your host just sends out a 255.255.255.255 DHCP client broadcast request, and the router changes that broadcast to the specific destination address of 172.16.10.1.

Be careful here—a host address can really look like a broadcast address! For example, a Unicast is a broadcast that's forwarded to a specific host. Is the IP address 172.16.128.255/18 a valid host or a broadcast address? Sure looks like a broadcast address!

First, /18 is 255.255.192.0. This makes the valid networks 64.0 and 128.0, where the broadcast address for the 172.16.128.0 subnet is 172.16.191.255, so the valid hosts are 128.1 through 191.254. 172.16.128.255 is a valid host and is a unicast! Don't worry—I'll show you how to subnet this in the next chapter. For now, understand that just because an IP address has a 0 or 255 in the address doesn't always make it a broadcast.

Multicast is a different beast entirely. At first glance, it appears to be a hybrid of unicast and broadcast communication, but that isn't quite the case. Multicast does allow point-to-multipoint communication, which is similar to broadcasts, but it happens in a different manner. The crux of *multicast* is that it enables multiple recipients to receive messages without flooding the messages to all hosts on a broadcast domain.

Multicast works by sending messages or data to IP *multicast group* addresses. Routers then forward copies of the packet out every interface that has hosts *subscribed* to that group address. This is where multicast differs from broadcast messages—with multicast communication, copies of packets, in theory, are sent only to subscribed hosts.

There are several different groups that users or applications can subscribe to. The range of multicast addresses starts with 224.0.0.0, and goes through 239.255.255.255. As you can see, this range of addresses falls within IP Class D address space based on classful IP assignment.

Introduction to Network Address Translation (NAT)

Whether your network is the home or the corporate type, if it uses the private IP addresses that I just talked about, you have to translate your private inside addresses to a global outside address by using NAT. The main idea is to conserve Internet global address space, but it also increases network security by hiding internal IP addresses from external networks. In NAT terminology, the *inside network* is the set of networks that are subject to translation. The *outside network* refers to all other addresses—usually those located on the Internet. However, just to help confuse you, it's important to understand that you can translate packets coming into the private network as well.

NAT operates on a Cisco router—generally only connecting two networks together—and translates your private (inside local) addresses within the internal network, into public (inside global) addresses before any packets are forwarded to another network. This functionality gives you the option to configure NAT so that it will advertise only a single address for your entire network to the outside world. Doing this effectively hides the internal network from the whole world really well, giving you some much-needed additional security.

There are different flavors of NAT:

Static NAT Designed to allow one-to-one mapping between local and global addresses. This flavor requires you to have one real Internet IP address for every host on your network.

Dynamic NAT Designed to map an unregistered IP address to a registered IP address from out of a pool of registered IP addresses. You don't have to statically configure your router to map an inside to an outside address as in static NAT, but you do have to have enough real IP addresses for everyone who wants to send packets to and from the Internet.

Overloading This is the most popular type of NAT configuration. Overloading is a form of dynamic NAT that maps multiple unregistered IP addresses to a single registered IP address (many-to-one) by using different ports. Therefore, it's also known as *port address translation (PAT)*. By using PAT (NAT Overload), you can have thousands of users connect to the Internet using only one real global IP address—pretty slick! NAT Overload is the reason we have not run out of valid IP address on the Internet.

An inside global address is a registered address that represents an inside host to an outside host.

Summary

If you made it this far and understood everything the first time through, you should be proud of yourself. We really covered a lot of ground in this chapter, but understand that the information in this chapter is key to being able to navigate through the rest of this book. And even if you didn't get a complete understanding the first time around, don't stress. It really wouldn't hurt you to read this chapter more than once. There is still a lot of ground to cover, so make sure you've got it all down, and get ready for more

In this chapter, you learned about the Internet Protocol stack and the various protocols used at each layer of the DoD model. You then learned about translating binary to decimal to hex. I can't stress how important it is for you to truly understand the difference between the three types of addresses and how to translate each one.

After you learned about the DoD model and binary, decimal, and hex addressing, you learned the all-so-important IP addressing. I discussed in detail the difference between each class of address and how to find a network address, broadcast address, and valid host range, which is critical information to understand before going on to Chapter 3, "IP Subnetting and Variable Length Subnet Masks (VLSMs)."

Lastly, I covered the private IP address range and Network Address Translation (NAT).

Since you've already come this far, there's no reason to stop now and waste all those brainwaves and new neurons. So don't stop—go through the written and review questions at the end of this chapter and make sure you understand each answer's explanation. The best is yet to come!

Exam Essentials

Remember the Process/Application layer protocols. Telnet is a terminal emulation program that allows you to log into a remote host and run programs. File Transfer Protocol (FTP) is a connection-oriented service that allows you to transfer files. Trivial FTP (TFTP) is a connectionless file transfer program. Simple Mail Transfer Protocol (SMTP) is a send-mail program.

Remember the Host-to-Host layer protocols. Transmission Control Protocol (TCP) is a connection-oriented protocol that provides reliable network service by using acknowledgments and flow control. User Datagram Protocol (UDP) is a connectionless protocol that provides low overhead and is considered unreliable.

Remember the Internet layer protocols. Internet Protocol (IP) is a connectionless protocol that provides network address and routing through an internetwork. Address Resolution Protocol (ARP) finds a hardware address from a known IP address. Reverse ARP (RARP) finds an IP address from a known hardware address. Internet Control Message Protocol (ICMP) provides diagnostics and unreachable messages.

Remember the Class A range. The IP range for a Class A network is 1–126. This provides 8 bits of network addressing and 24 bits of host addressing by default.

Remember the Class B range. The IP range for a Class B network is 128–191. Class B addressing provides 16 bits of network addressing and 16 bits of host addressing by default.

Remember the Class C range. The IP range for a Class C network is 192–223. Class C addressing provides 24 bits of network addressing and 8 bits of host addressing by default.

Written Lab 2

1. What is the Class C address range in decimal and in binary?
2. What layer of the DoD model is equivalent to the Transport layer of the OSI model?
3. What is the valid range of a Class A network address?
4. What is the 127.0.0.1 address used for?
5. How do you find the network address from a listed IP address?
6. How do you find the broadcast address from a listed IP address?
7. What is the Class A private IP address space?
8. What is the Class B private IP address space?
9. What is the Class C private IP address space?
10. What are all the available characters that you can use in hexadecimal addressing?

(The answers to Written Lab 2 can be found following the answers to the Review Questions for this chapter.)

Review Questions

1. What is the decimal and hexadecimal equivalent of the binary number 10011101? (Choose two.)
   - **A.** 159
   - **B.** 157
   - **C.** 185
   - **D.** 0x9D
   - **E.** 0xD9
   - **F.** 0x159
2. Which of the following allows a router to respond to an ARP request that is intended for a remote host?
   - **A.** Gateway DP
   - **B.** Reverse ARP (RARP)
   - **C.** Proxy ARP
   - **D.** Inverse ARP (IARP)
   - **E.** Address Resolution Protocol (ARP)
3. You want to implement a mechanism that automates the IP configuration, including IP address, subnet mask, default gateway, and DNS information. Which protocol will you use to accomplish this?
   - **A.** SMTP
   - **B.** SNMP
   - **C.** DHCP
   - **D.** ARP
4. What protocol is used to find the hardware address of a local device?
   - **A.** RARP
   - **B.** ARP
   - **C.** IP
   - **D.** ICMP
   - **E.** BootP
5. Which of the following are layers in the TCP/IP model? (Choose three.)
   - **A.** Application
   - **B.** Session
   - **C.** Transport
   - **D.** Internet
   - **E.** Data Link
   - **F.** Physical

6. Which class of IP address provides a maximum of only 254 host addresses per network ID?
   - **A.** Class A
   - **B.** Class B
   - **C.** Class C
   - **D.** Class D
   - **E.** Class E

7. Which of the following describe the DHCP Discover message? (Choose two.)
   - **A.** It uses FF-FF-FF-FF-FF-FF as a layer 2 broadcast.
   - **B.** It uses UDP as the Transport layer protocol.
   - **C.** It uses TCP as the Transport layer protocol.
   - **D.** It does not use a layer 2 destination address.

8. What does the "Inside Global" address represent in the configuration of NAT?
   - **A.** The summarized address for all of the internal subnetted addresses
   - **B.** The MAC address of the router used by inside hosts to connect to the Internet
   - **C.** A globally unique, private IP address assigned to a host on the inside network
   - **D.** A registered address that represents an inside host to an outside network

9. Which of the following protocols uses TCP port 443?
   - **A.** HTML
   - **B.** HTTPS
   - **C.** TFTP
   - **D.** Telnet
   - **E.** SMTP

10. Which of the following services use TCP? (Choose three.)
    - **A.** DHCP
    - **B.** SMTP
    - **C.** SNMP
    - **D.** FTP
    - **E.** HTTP
    - **F.** TFTP

11. Which of the following services use UDP? (Choose three.)
    - **A.** DHCP
    - **B.** SMTP
    - **C.** SNMP
    - **D.** FTP
    - **E.** HTTP
    - **F.** TFTP

12. Which of the following are TCP/IP protocols used at the Application layer of the OSI model? (Choose three.)

A. IP

B. TCP

C. Telnet

D. FTP

E. TFTP

13. You have the binary number 10110011. Which two of the following are equivalent to this?

A. 128

B. 179

C. 184

D. 0xAC

E. 0x3C

F. 0xB3

14. If you use either Telnet or FTP, which is the highest layer you are using to transmit data?

A. Application

B. Presentation

C. Session

D. Transport

15. The DoD model (also called the TCP/IP stack) has four layers. Which layer of the DoD model is equivalent to the Network layer of the OSI model?

A. Application

B. Host-to-Host

C. Internet

D. Network Access

16. You have the binary number 11000111. Which two of the following are equivalent to this?

A. 228

B. 179

C. 199

D. 0xC7

E. 0x3C

F. 0xB7

17. What layer in the TCP/IP stack is equivalent to the Transport layer of the OSI model?

A. Application

B. Host-to-Host

C. Internet

D. Network Access

18. Your company uses Voice over IP (VoIP). The system sends UDP datagrams containing the voice data between communicating hosts. When areas of the network become busy, some of the datagrams arrive at their destination out of order. What happens when this occurs?

A. UDP will send an ICMP Information request message to the source host.

B. UDP will pass the information in the datagrams up to the next OSI layer in the order in which they arrive.

C. UDP will drop the datagrams that arrive out of order.

D. UDP will use the sequence numbers in the datagram headers to reassemble the data into the order in which it was transmitted.

19. What is the address range of a Class B network address in binary?

A. 01*xxxxxx*

B. 0*xxxxxxx*

C. 10*xxxxxx*

D. 110*xxxxx*

20. Which of the following protocols use both TCP and UDP?

A. FTP

B. SMTP

C. Telnet

D. DNS

Answers to Review Questions

1. B, D. To turn a binary number into decimal, you just have to add the values of each bit that is a 1. The values of 10011101 are 128, 16, 8, 4, and 1. 128 + 16 + 8 + 4 + 1 = 157. Hexadecimal is a base 16 number system. The values of hexadecimal are 0, 1, 2, 3, 4, 5, 6, 7, 8, 9, A, B, C, D, E, F—sixteen characters total, from which to create all the numbers you'll ever need. So, if 1001 in binary is 9, then the hexadecimal equivalent is 9. Since we then have 1101, which is 13 in binary, the hexadecimal answer is D and the complete hexadecimal answer is 0x9D.

2. C. Proxy ARP can help machines on a subnet reach remote subnets without configuring routing or a default gateway.

3. C. Dynamic Host Configuration Protocol (DHCP) is used to provide IP information to hosts on your network. The information provided by DHCP is IP address, subnet mask, default gateway, and DNS information, plus others.

4. B. Address Resolution Protocol (ARP) is used to find the hardware address from a known IP address.

5. A, C, D. This seems like a hard question at first because it doesn't make sense. The listed answers are from the OSI model and the question asked about the TCP/IP protocol stack (DoD model). However, let's just look for what is wrong. First, the Session layer is not in the TCP/IP model; neither are the Data Link or Physical layers. This leaves us with the Transport layer (Host-to-Host in the DoD model), Internet layer (Network layer in the OSI), and Application layer (Application/Process in the DoD).

6. C. A Class C network address has only 8 bits for defining hosts: $2^8 - 2 = 254$.

7. A, B. A client that sends out a DHCP Discover message in order to receive an IP address sends out a broadcast at both layer 2 and layer 3. The layer 2 broadcast is all Fs in hex, or FF:FF:FF:FF:FF:FF. The layer 3 broadcast is 255.255.255.255, which means all networks and all hosts. DHCP is connectionless, which means it uses User Datagram Protocol (UDP) at the Transport layer, also called the Host-to-Host layer.

8. D. The inside local IP addresses are the untranslated, private IP addresses and the inside global IP addresses are what the inside local IP addresses get changed to when NAT occurs.

9. C. HTTPS (HTTP Secure) uses TCP port 443. TFTP uses port 69, Telnet uses 23, and SMTP uses port 25.

10. B, D, E. SMTP, FTP, and HTTP use TCP.

11. A, C, F. DHCP, SNMP and TFTP use UDP. SMTP, FTP, and HTTP use TCP.

12. C, D, E. Telnet, File Transfer Protocol (FTP), and Trivial FTP are all Application layer protocols. IP is a Network layer protocol. Transmission Control Protocol (TCP) is a Transport layer protocol.

13. B, F. You need to find the decimal and hexadecimal equivalent to the listed binary number of 10110011. First, add the byte digits up: 128 + 32 + 16 + 2 + 1 = 179. Then, break the byte into nibbles and add those up as your hexadecimal answer: 1011 = 11, which is B in hex, and 0011, which is 3 in decimal and hex. So the answer is B3. However, Cisco likes to try to confuse you by adding the 0x in front, which just means that the following characters are hex characters.

14. A. Both FTP and Telnet use TCP at the Transport layer; however, they both are Application layer protocols, so the Application layer is the best answer for this question.

15. C. The four layers of the DoD model are Application/Process, Host-to-Host, Internet, and Network Access. The Internet layer is equivalent to the Network layer of the OSI model.

16. C, D. First, convert the binary number of 11000111 to a decimal address by adding up the bits that are ones (1s): 128 + 64 + 4 + 2 + 1 = 199. Now, to get our hexadecimal address, break the byte into nibbles: 1100 = 12, or C in hex, and 0111 = 7 in both decimal and hex. However, Cisco likes to try to confuse you by adding the 0x in front, which just means that the following characters are hex characters.

17. B. The four layers of the TCP/IP stack (also called the DoD model) are Application/Process, Host-to-Host, Internet, and Network Access. The Host-to-Host layer is equivalent to the Transport layer of the OSI model.

18. B. UDP has no sequencing field and has no idea in which order a segment arrives. It will just pass any segment to the upper layers as they arrive.

19. C. The range of a Class B network address is 128–191. This makes our binary range 10*xxxxxx*.

20. D. DNS uses TCP for zone exchanges between server, and UDP is used when a client is trying to resolve a hostname to an IP address.

Answers to Written Lab 2

1. 192–223, 110*xxxxx*
2. Host-to-Host
3. 1–126
4. Loopback or diagnostics
5. Turn all host bits off.
6. Turn all host bits on.
7. 10.0.0.0 through 10.255.255.255
8. 172.16.0.0 through 172.31.255.255
9. 192.168.0.0 through 192.168.255.255
10. 0–9 and A, B, C, D, E, and F