Chapter 1

Internetworking

DESCRIBE THE OPERATION OF DATA NETWORKS

- ✓ Describe the purpose and functions of various network devices
- ✓ Select the components required to meet a given network specification
- ✓ Use the OSI and TCP/IP models and their associated protocols to explain how data flows in a network
- ✓ Describe common networking applications including web applications
- ✓ Describe the purpose and basic operation of the protocols in the OSI and TCP models
- ✓ Describe the components required for network and Internet communications
- ✓ Identify and correct common network problems at layers 1, 2, 3 and 7 using a layered model approach

IMPLEMENT A SMALL SWITCHED NETWORK

- ✓ Select the appropriate media, cables, ports, and connectors to connect switches to other network devices and hosts
- ✓ Explain the technology and media access control method for Ethernet technologies
- ✓ Explain network segmentation and basic traffic management concepts
- ✓ Identify, prescribe, and resolve common switched network media issues, configuration issues, autonegotiation, and switch hardware failures

Welcome to the exciting world of internetworking. This first chapter will really help you understand the basics of internetworking by focusing on how to connect networks using Cisco routers and switches. First, you need to know exactly what an internetwork is, right? You create an internetwork when you connect two or more LANs or WANs through a router and configure a logical network addressing scheme with a protocol, such as IP. I'll be covering these four topics in this chapter:

- Internetworking basics
- Network segmentation
- How bridges, switches, and routers are used to physically segment a network
- How routers are employed to create an internetwork

I'm also going to dissect the Open Systems Interconnection (OSI) model and describe each part in detail because you really need a good grasp of OSI for a solid foundation to build your networking knowledge upon. The OSI model has seven hierarchical layers that were developed to enable networks to communicate reliably between disparate systems. Since this book centers on all things CCENT, it's crucial for you to understand the OSI model as Cisco sees it, so that's how I'll be presenting the seven layers to you.

Since there are a bunch of different types of devices specified in the different layers of the OSI model, it's also very important to understand the many types of cables and connectors used for connecting all those devices to a network. We'll go over cabling Cisco devices, discussing how to connect to a router or switch (along with Ethernet LAN technologies) and even how to connect a router or switch with a console connection.

We'll finish the chapter by discussing the three-layer hierarchical model that was developed by Cisco to help you design, implement, and troubleshoot internetworks.

After you finish reading this chapter, take the time to answer the review questions and complete the written labs. These are given to you to really lock the information from this chapter into your memory. So don't skip them!

To find up-to-the-minute updates for this chapter, please see `www.lammle.com` and/or `www.sybex.com/go/ccent`.

Internetworking Basics

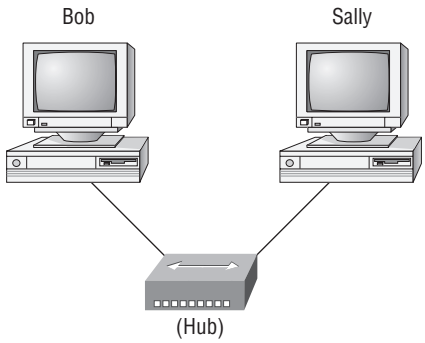
Before we explore internetworking models and the specifications of the OSI reference model, you've got to understand the big picture and learn the answer to the key question: Why is it so important to learn Cisco internetworking?

Networks and networking have grown exponentially over the last 15 years—understandably so. They've had to evolve at light speed just to keep up with huge increases in basic, mission-critical user needs, such as sharing data and printers, as well as more advanced demands, such as video-conferencing. Unless everyone who needs to share network resources is located in the same office area (an increasingly uncommon situation), the challenge is to connect the relevant networks together so all users can share the networks' wealth.

Addressing

Starting with a look at Figure 1.1, you get a picture of a basic local area network (LAN) that's connected using a hub. This network is actually one collision domain and one broadcast domain—but no worries if you have no idea what this means because I'm going to talk about both collision and broadcast domains so much throughout this whole chapter that you'll probably even dream about them!

FIGURE 1.1 The basic network



The basic network allows devices to share information.
The term computer language refers to binary code (0s or 1s).
The two hosts above communicate using hardware or MAC addresses.

Okay, about Figure 1.1… How would you say the PC named Bob communicates with the PC named Sally? Well, they're both on the same LAN connected with a multiport repeater (a hub). So, does Bob just send out a data message, "Hey Sally, you there?" or does Bob use Sally's Internet Protocol (IP) address and put things more like, "Hey 192.168.0.3, are you there?" Hopefully, you picked the IP address option, but even if you did, the news is still bad—both answers are wrong! Why? Because Bob is actually going to use Sally's *Media Access*

Control (MAC) address (known as a hardware address), which is burned right into the network card of Sally's PC, to get ahold of her.

Great, but how does Bob get Sally's MAC address since Bob knows only Sally's name and doesn't even have her IP address yet? Bob is going to start with name resolution (hostname to IP address resolution), something that's usually accomplished using Domain Name Service (DNS). And of note, if these two are on the same LAN, Bob can just broadcast to Sally asking her for the information (no DNS needed)—welcome to Microsoft Windows (Vista included)!

Here's an output from a network analyzer depicting a simple name resolution process from Bob to Sally:

```
   Time      Source     Destination   Protocol   Info
53.892794  192.168.0.2  192.168.0.255  NBNS  Name query NB SALLY<00>
   Time      Source     Destination   Protocol   Info
53.892794  192.168.0.2  192.168.0.255  NBNS  Name query NB SALLY<00>
```

As I already mentioned, since the two hosts are on a local LAN, Windows (Bob) will just broadcast to resolve the name Sally (the destination 192.168.0.255 is a broadcast address). Let's take a look at the rest of the information:

```
EthernetII,Src:192.168.0.2(00:14:22:be:18:3b),Dst:Broadcast (ff:ff:ff:ff:ff:ff)
```

What this output shows is that Bob knows his own MAC address and source IP address but not Sally's IP address or MAC address, so Bob sends a broadcast address of all *f*s for the MAC address (a Data Link layer broadcast) and an IP LAN broadcast of 192.168.0.255. Again, don't freak—you're going to learn all about broadcasts in Chapter 3, "IP Subnetting, Variable Length Subnet Masks (VLSMs), Troubleshooting IP, and Introduction to NAT."

Before the name is resolved, the first thing Bob has to do is broadcast on the LAN to get Sally's MAC address so he can communicate to her PC and resolve her name to an IP address:

```
Time      Source     Destination Protocol Info
5.153054  192.168.0.2 Broadcast  ARP Who has 192.168.0.3? Tell 192.168.0.2
```

Next, check out Sally's response:

```
Time      Source     Destination   Protocol  Info
5.153403 192.168.0.3 192.168.0.2  ARP  192.168.0.3 is at 00:0b:db:99:d3:5e
5.53.89317  192.168.0.3 192.168.0.2 NBNS Name query response NB 192.168.0.3
```

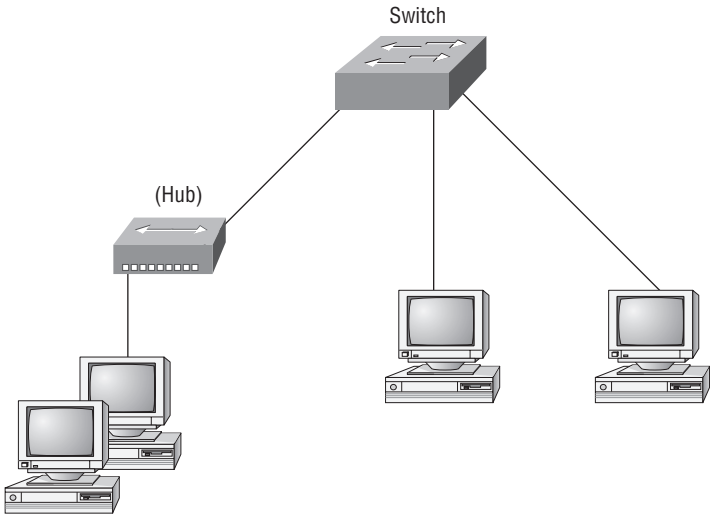
Okay, sweet—Bob now has both Sally's IP address and her MAC address! Both are listed as the source address at this point because this information was sent from Sally back to Bob. So, *finally*, Bob has all the goods he needs to communicate with Sally. And just so you know, I'm going to tell you all about ARP and show you exactly how Sally's IP address was resolved to a MAC address a little later in Chapter 6, "IP Routing."

By the way, I want you to understand that Sally still had to go through the same resolution processes to communicate back to Bob—sounds crazy, huh? Consider this a welcome to IPv4 and basic networking with Windows (and we haven't even added a router yet!).

Hubs, Bridges, Routers, and Switches

To complicate things further, it's also likely that at some point you'll have to break up one large network into a bunch of smaller ones because user response times will have dwindled to a slow crawl as the network grew and grew. And with all that growth, your LAN's traffic congestion will have reached epic proportions. The answer to this problem is breaking up that really big network into a number of smaller ones—something called *network segmentation*. You do this by using devices like *routers*, *switches*, and *bridges*. Figure 1.2 shows a network that's been segmented with a switch so each network segment connected to the switch is now a separate collision domain. But make note of the fact that this network is still one broadcast domain.

FIGURE 1.2 A switch can replace the hub, breaking up collision domains.



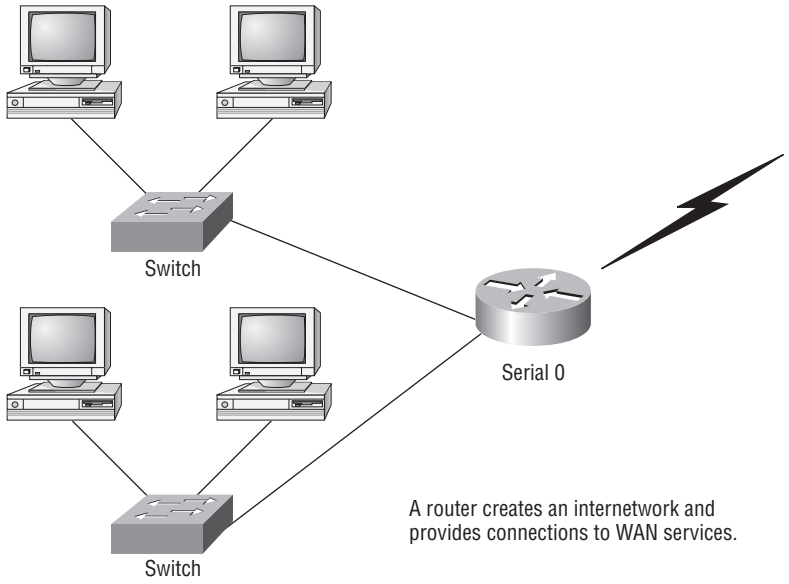
Keep in mind that the hub used in Figure 1.2 just extended the one collision domain from the switch port. Here's a list of some of the things that commonly cause LAN traffic congestion:

- Too many hosts in a broadcast domain
- Broadcast storms
- Multicasting
- Low bandwidth
- Adding hubs for connectivity to the network
- A bunch of ARP or IPX traffic (IPX is a Novell routed protocol that is like IP, but really, really chatty. Typically, IPX is not used in today's networks.)

Take another look at Figure 1.2. Did you notice that I replaced the main hub from Figure 1.1 with a switch? Whether you did or didn't, I did that because hubs don't segment a network; they just connect network segments together. So basically, it's an inexpensive way to connect a couple of PCs together, which is great for home use and troubleshooting, but that's about it!

Now, routers are used to connect networks and route packets of data from one network to another. Cisco became the de facto standard of routers because of its high-quality router products, great selection, and fantastic service. Routers, by default, break up a *broadcast domain*—the set of all devices on a network segment that hear all the broadcasts sent on that segment. Figure 1.3 shows a router in our little network that creates an internetwork and breaks up broadcast domains.

FIGURE 1.3 Routers create an internetwork.



The network in Figure 1.3 is a pretty cool network. Each host is connected to its own collision domain, and the router has created two broadcast domains. And don't forget that the router provides connections to wide area network (WAN) services as well! The router uses something called a serial interface for WAN connections, specifically, a V.35 physical interface on a Cisco router.

Breaking up a broadcast domain is important because when a host or server sends a network broadcast, every device on the network must read and process that broadcast—unless you've got a router. When the router's interface receives this broadcast, it can respond by basically saying, "Thanks, but no thanks," and discard the broadcast without forwarding it on to other networks. Even though routers are known for breaking up broadcast domains by default, it's important to remember that they break up collision domains as well.

There are two advantages of using routers in your network:

- They don't forward broadcasts by default.
- They can filter the network based on layer 3 (Network layer) information (like the IP address).

Routers can be used for the following four functions in your network:

- Packet switching
- Packet filtering
- Internetwork communication
- Path selection

Remember that routers are really switches; they're actually what we call layer 3 switches (we'll talk about layers later in this chapter). Unlike layer 2 switches, which forward or filter frames, routers (layer 3 switches) use logical addressing and provide what is called packet switching. Routers can also provide packet filtering by using access lists, and when routers connect two or more networks together and use logical addressing (IP or IPv6), you have what is called an internetwork. Finally, routers use a routing table (map of the internetwork) to make path selections and to forward packets to remote networks.

In this book, I'll just talk about IP addressing. If you'd like to know more about IPv6, pick up a copy of Sybex's *CCNA: Cisco Certified Network Associate Study Guide*. There's a whole chapter on IPv6.

Conversely, layer 2 switches, the ones we usually call just plain switches, aren't used to create internetworks because they do not break up broadcast domains by default; they're employed to add functionality to a network LAN. The main purpose of these switches is to make a LAN work better—to optimize its performance—providing more bandwidth for the LAN's users. And these switches don't forward packets to other networks, as routers do. Instead, they only "switch" frames from one port to another within the switched network. Okay, you may be thinking, "Wait a minute, what are frames and packets?" I'll tell you all about them later in this chapter, I promise!

By default, switches break up collision domains. *Collision domain* is an Ethernet term used to describe a network scenario in which one device sends a packet on a network segment and every other device on the same segment is forced to pay attention to it. If, at the same time, a different device tries to transmit, a collision occurs and both devices must retransmit—one at a time. Not very efficient! This situation is typically found in a hub environment, where each host segment connects to a hub that represents only one collision domain and only one broadcast domain. By contrast, each and every port on a switch represents its own collision domain.

Switches create separate collision domains within a single broadcast domain. Routers provide a separate broadcast domain for each interface.

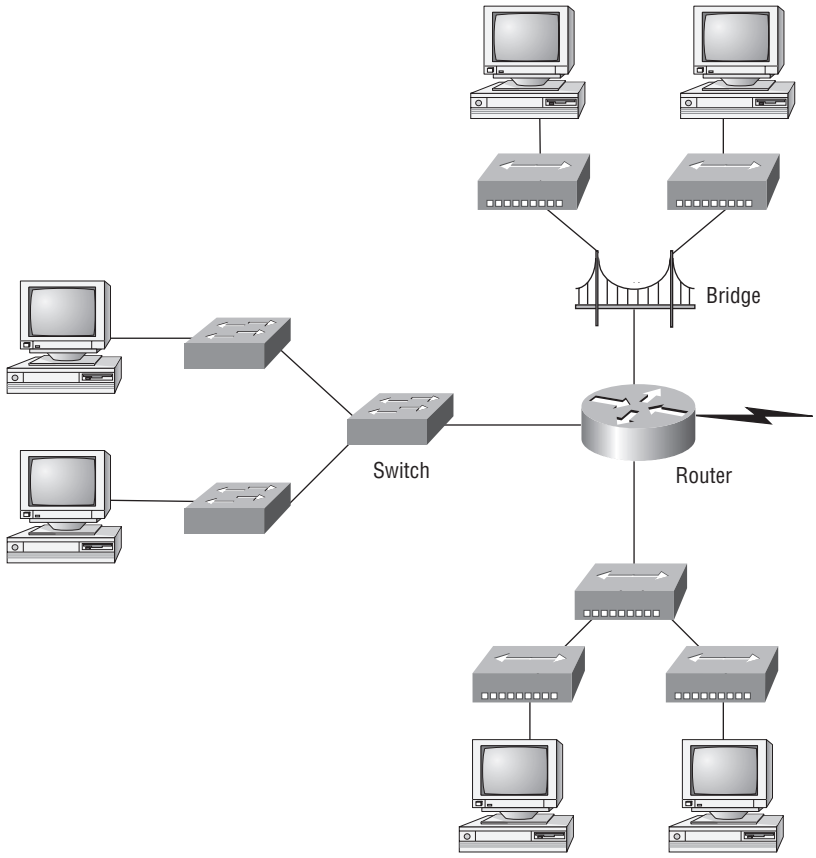
The term *bridging* was introduced before routers and hubs were implemented, so it's pretty common to hear people referring to bridges as switches. That's because bridges and switches basically do the same thing—break up collision domains on a LAN (in reality, you cannot buy a physical bridge these days, only LAN switches, but they use bridging technologies, so Cisco still calls them multiport bridges).

So, what this means is that a switch is basically just a multiple-port bridge with more brainpower, right? Well, pretty much, but there are differences. Switches do provide a bridging function, but they do so with greatly enhanced management ability and features. Plus, most of the time, bridges had only 2 or 4 ports. Yes, you could get your hands on a bridge with up to 16 ports, but that's nothing compared to the hundreds of ports available on some switches!

You would use a bridge in a network to reduce collisions within broadcast domains and to increase the number of collision domains in your network. Doing this provides more bandwidth for users. And keep in mind that using hubs in your Ethernet network can contribute to congestion. As always, plan your network design carefully!

Figure 1.4 shows how a network would look with all these internetwork devices in place. Remember that the router will not only break up broadcast domains for every LAN interface, it will break up collision domains as well.

FIGURE 1.4 Internetworking devices



When you looked at Figure 1.4, did you notice that the router is found at center stage and that it connects each physical network together? We have to use this layout because of the older technologies involved—bridges and hubs.

On the top internetwork in Figure 1.4, you'll notice that a bridge was used to connect the hubs to a router. The bridge breaks up collision domains, but all the hosts connected to both hubs are still crammed into the same broadcast domain. Also, the bridge created only two collision domains, so each device connected to a hub is in the same collision domain as every other device connected to that same hub. This is actually pretty lame, but it's still better than having one collision domain for all hosts.

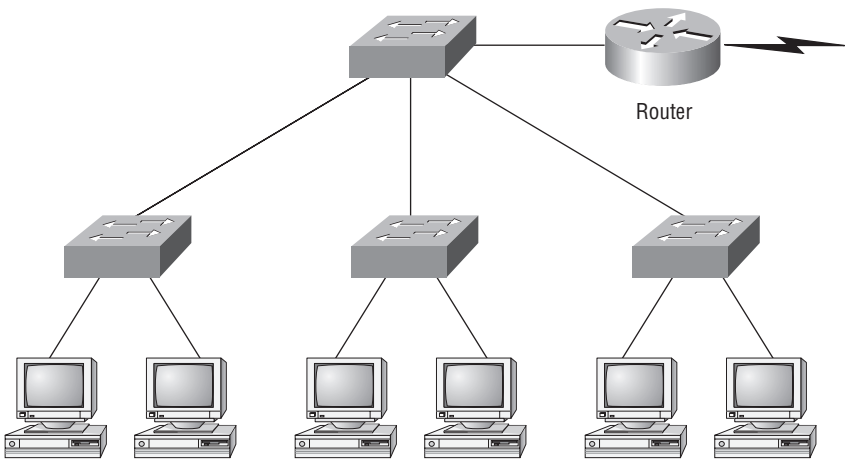
Notice something else: The three interconnected hubs at the bottom of the figure also connect to the router. This setup creates one collision domain and one broadcast domain and makes the bridged network, with its two collision domains, look much better indeed!

Although bridges/switches are used to segment networks, they will not isolate broadcast or multicast packets.

The best network connected to the router is the LAN switch network on the left. Why? Because each port on that switch breaks up collision domains. But it's not all good—all devices are still in the same broadcast domain. Do you remember why this can be a really bad thing? Because all devices must listen to all broadcasts transmitted, that's why. And if your broadcast domains are too large, the users have less bandwidth and are required to process more broadcasts. Network response time eventually will slow to a level that could cause office riots.

Once we have only switches in our network, things change a lot! Figure 1.5 shows the network that is typically found today.

FIGURE 1.5 Switched networks creating an internetwork



Okay, here I've placed the LAN switches at the center of the network world so the routers are connecting only logical networks together. If I implemented this kind of setup, I've created virtual LANs (VLANs), something that you don't have to worry about in the ICND1 objectives. VLANs are covered in depth in the Sybex CCNA Study Guide. But it is really important to understand that even in a switched network, you still need a router to provide your inter-VLAN communication, or internetworking. Don't forget that!

Obviously, the best network is one that's correctly configured to meet the business requirements of the company it serves. The best network design is one in which LAN switches with routers are used, and correctly placed in the network. This book will help you understand the basics of routers and switches so you can make tight, informed decisions on a case-by-case basis.

Let's go back to Figure 1.4 again. Look at the figure. How many collision domains and broadcast domains are in this internetwork? Hopefully, you answered nine collision domains and three broadcast domains! The broadcast domains are definitely the easiest to see because only routers break up broadcast domains by default. And since there are three connections, that gives you three broadcast domains. But do you see the nine collision domains? Just in case that's a no, I'll explain. The all-hub network is one collision domain; the bridge network equals three collision domains. Add in the switch network of five collision domains—one for each switch port—and you've got a total of nine.

Now, in Figure 1.5, each port on the switch is a separate collision domain and each VLAN is a separate broadcast domain. But you still need a router for routing between VLANs. How many collision domains do you see here? I'm counting 10—remember that connections between the switches are considered a collision domain!

So now that you've gotten an introduction to internetworking and the various devices that live in an internetwork, it's time to head into internetworking models.

Real World Scenario

Should I Just Replace All My Hubs with Switches?

You're a network administrator at a large company in San Jose. The boss comes to you and says that he got your requisition to buy a switch and he's not sure about approving the expense; do you really need it?

Well, if you can replace all the hubs with switches, sure—why not? Switches really add a lot of functionality to a network that hubs just don't have. But most of us don't have an unlimited budget. Hubs still can create a nice network—that is, of course, if you design and implement the network correctly.

Let's say that you have 40 users plugged into four hubs, 10 users each. At this point, the hubs are all connected together so that you have one large collision domain and one large broadcast domain. If you can afford to buy just one switch and plug each hub into a switch port, and plug the servers into the switch, you'll have four collision domains and one broadcast domain. Not great, but for the price of one switch, your network is a much better thing. So, go ahead! Put that requisition in to buy all new switches. What do you have to lose?

Internetworking Models

When networks first came into being, computers could typically communicate only with computers from the same manufacturer. For example, companies ran either a complete DECnet solution or an IBM solution—not both together. In the late 1970s, the *Open Systems Interconnection (OSI) reference model* was created by the International Organization for Standardization (ISO) to break this barrier.

The OSI model was meant to help vendors create interoperable network devices and software in the form of protocols so that different vendor networks could work with each other. Like world peace, it'll probably never happen completely, but it's still a great goal.

The OSI model is the primary architectural model for networks. It describes how data and network information are communicated from an application on one computer through the network media to an application on another computer. The OSI reference model breaks this approach into layers.

In the following section, I am going to explain the layered approach and how we can use this approach to help us troubleshoot our internetworks.

The Layered Approach

A *reference model* is a conceptual blueprint of how communications should take place. It addresses all the processes required for effective communication and divides these processes into logical groupings called *layers*. When a communication system is designed in this manner, it's known as *layered architecture*.

Think of it like this: You and some friends want to start a company. One of the first things you'll do is sit down and think through what tasks must be done, who will do them, the order in which they will be done, and how they relate to each other. Ultimately, you might group these tasks into departments. Let's say you decide to have an order-taking department, an inventory department, and a shipping department. Each of your departments has its own unique tasks, keeping its staff members busy and requiring them to focus on only their own duties.

In this scenario, I'm using departments as a metaphor for the layers in a communication system. For things to run smoothly, the staff of each department will have to trust and rely heavily upon the others to do their jobs and competently handle their unique responsibilities. In your planning sessions, you would probably take notes, recording the entire process to facilitate later discussions about standards of operation that will serve as your business blueprint, or reference model.

Once your business is launched, your department heads, each armed with the part of the blueprint relating to their own department, will need to develop practical methods to implement their assigned tasks. These practical methods, or protocols, will need to be compiled into a standard operating procedures manual and followed closely. Each of the various procedures in your manual will have been included for different reasons and have varying degrees of importance and implementation. If you form a partnership or acquire another company, it will be imperative that its business protocols—its business blueprint—match yours (or at least be compatible).

Similarly, software developers can use a reference model to understand computer communication processes and see what types of functions need to be accomplished on any one layer. If they are developing a protocol for a certain layer, all they need to concern themselves with is that specific layer's functions, not those of any other layer. Another layer and protocol will handle the other functions. The technical term for this idea is *binding*. The communication processes that are related to each other are bound, or grouped together, at a particular layer.

Advantages of Reference Models

One of the greatest functions of the reference model specifications is to assist in data transfer between disparate hosts—meaning, for example, that they enable us to transfer data between a Unix host and a PC or a Mac. The reference models aren't physical models, though. Rather, they're sets of guidelines that application developers can use to create and implement applications that run on a network. They also provide a framework for creating and implementing networking standards, devices, and internetworking schemes.

The Open Systems Interconnection (OSI) model is a hierarchical model, and the same benefits and advantages you gain from implementing OSI standards can apply to any layered model. The primary purpose of all such models, especially the OSI model, is to allow different vendors' networks to interoperate.

Advantages of using the OSI layered model include, but are not limited to, the following:

- It divides the network communication process into smaller and simpler components, thus aiding component development, design, and troubleshooting.
- It allows multiple-vendor development through standardization of network components.
- It encourages industry standardization by defining what functions occur at each layer of the model.
- It allows various types of network hardware and software to communicate.
- It prevents changes in one layer from affecting other layers, so it does not hamper hardware or software development.

The OSI Reference Model

Basically, the ISO is pretty much the Emily Post of the network protocol world. Just as Ms. Post wrote the book setting the standards—or protocols—for human social interaction, the ISO developed the OSI reference model as the precedent and guide for an open network protocol set. Defining the etiquette of communication models, it remains today the most popular means of comparison for protocol suites.

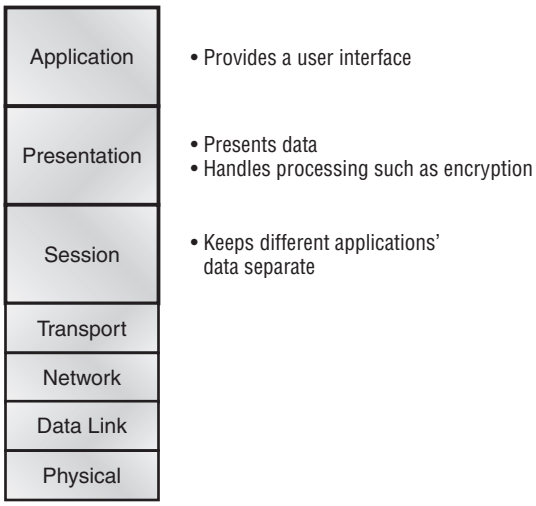
The OSI reference model has seven layers:

- Application layer (layer 7)
- Presentation layer (layer 6)
- Session layer (layer 5)
- Transport layer (layer 4)

- Network layer (layer 3)
- Data Link layer (layer 2)
- Physical layer (layer 1)

The OSI layers are divided into two groups. The top three layers define how the applications within the end stations communicate with each other and with users. The bottom four layers define how data is transmitted end to end. Figure 1.6 shows the three upper layers and their functions, and Figure 1.7 shows the four lower layers and their functions.

FIGURE 1.6 The upper layers



When you study Figure 1.6, understand that the user interfaces with the computer at the Application layer and also that the upper layers are responsible for applications communicating between hosts. Notice that none of the upper layers knows anything about networking or network addresses. That's the responsibility of the four bottom layers.

In Figure 1.7, you can see that it's the four bottom layers that define how data is transferred through a physical wire or through switches and routers. These bottom layers also determine how to rebuild a data stream from a transmitting host to a destination host's application.

In this chapter I've discussed hubs, bridges, switches and routers, but where are these devices in relation to the OSI model? Hubs/repeaters are Physical layer devices, Bridges/Switches work at the Data Link layer, and Routers work at the Network layer. None of the network devices work at all seven of the OSI model's layer. The following network devices operate on all seven layers of the OSI model:

- Network management stations (NMSs)
- Web and application servers
- Gateways (not default gateways)
- Network hosts

Figure 1.8 shows a summary of the functions defined at each layer of the OSI model. With this in hand, you're now ready to explore each layer's function in detail.

FIGURE 1.7 The lower layers

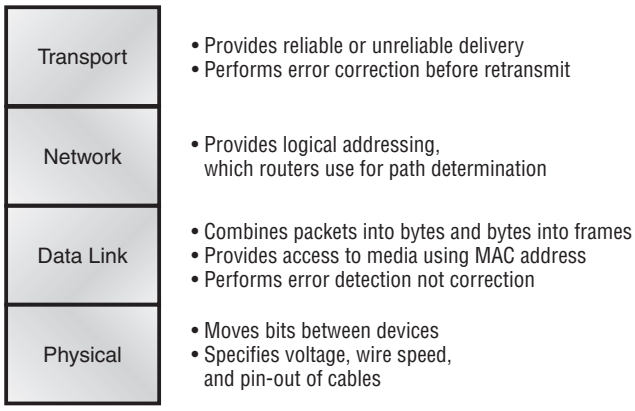
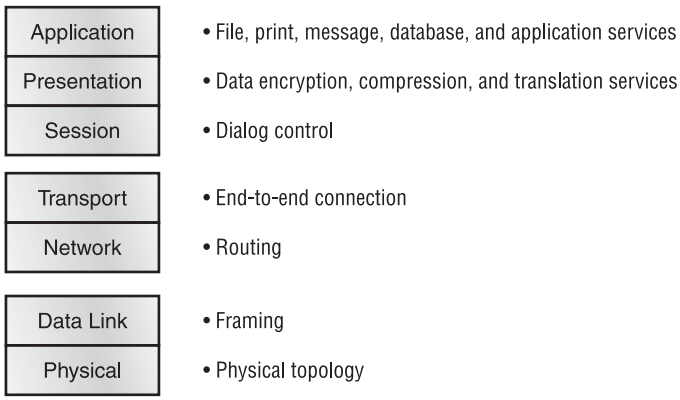


FIGURE 1.8 Layer functions



The Application Layer

The *Application layer* of the OSI model marks the spot where users actually communicate to the computer. This layer only comes into play when it's apparent that access to the network is going to be needed soon. Take the case of Internet Explorer (IE). You could uninstall every trace of networking components from a system, such as TCP/IP, the network interface card (NIC), and so on, and you could still use IE to view a local HTML document—no problem. But things would definitely get messy if you tried to do something like view an HTML document that must be

retrieved using Hypertext Transfer Protocol (HTTP) or nab a file with File Transfer Protocol (FTP) or Trivial File Transfer Protocol (TFTP). That's because IE will respond to requests such as those by attempting to access the Application layer. And what's happening is that the Application layer is acting as an interface between the actual application program—which isn't at all a part of the layered structure—and the next layer down by providing ways for the application to send information down through the protocol stack. In other words, IE doesn't truly reside within the Application layer—it interfaces with Application layer protocols when it needs to deal with remote resources.

The Application layer is also responsible for identifying and establishing the availability of the intended communication partner and determining whether sufficient resources for the intended communication exist.

These tasks are important because computer applications sometimes require more than only desktop resources. Often, they'll unite communicating components from more than one network application. Prime examples are file transfers and email, as well as enabling remote access, network management activities, client/server processes, and information location. Many network applications provide services for communication over enterprise networks, but for present and future internetworking, the need is fast developing to reach beyond the limits of current physical networking.

It's important to remember that the Application layer is acting as an interface between the actual application programs. This means that Microsoft Word, for example, does not reside at the Application layer but instead interfaces with the Application layer protocols. Chapter 2 will present some programs that actually reside at the Application layer—for example, FTP and TFTP.

The Presentation Layer

The *Presentation layer* gets its name from its purpose: It presents data to the Application layer and is responsible for data translation and code formatting.

This layer is essentially a translator and provides coding and conversion functions. A successful data-transfer technique is to adapt the data into a standard format before transmission. Computers are configured to receive this generically formatted data and then convert the data back into its native format for actual reading—for example, from Extended Binary Coded Decimal Interchange Code (EBCDIC) to American Standard Code for Information Interchange (ASCII). By providing translation services, the Presentation layer ensures that data transferred from the Application layer of one system can be read by the Application layer of another one.

The OSI has protocol standards that define how standard data should be formatted. Tasks like data compression, decompression, encryption, and decryption are associated with this layer. Some Presentation layer standards are involved in multimedia operations too.

The Session Layer

The *Session layer* is responsible for setting up, managing, and then tearing down sessions between Presentation layer entities. This layer also provides dialogue control between devices, or nodes. It coordinates communication between systems and serves to organize their communication by offering three different modes: *simplex*, *half duplex*, and *full duplex*. To sum up, the Session layer basically keeps one application's data separate from other applications' data.

The Transport Layer

The *Transport layer* segments and reassembles data into a data stream. Services located in the Transport layer segment and reassemble data from upper-layer applications and unite it onto the same data stream. They provide end-to-end data transport services and can establish a logical connection between the sending host and destination host on an internetwork.

Some of you are probably familiar with Transmission Control Protocol (TCP) and User Datagram Protocol (UDP) already. (But if you're not, no worries—I'll tell you all about them in Chapter 2.) If so, you know that both work at the Transport layer and that TCP is a reliable service and UDP is not. This means that application developers have more options because they have a choice between the two protocols when working with TCP/IP protocols.

The Transport layer is responsible for providing mechanisms for multiplexing upper-layer applications, establishing sessions, and tearing down virtual circuits. It also hides details of any network-dependent information from the higher layers by providing transparent data transfer.

The term *reliable networking* can be used at the Transport layer. It means that acknowledgments, sequencing, and flow control will be used.

The Transport layer can be connectionless or connection-oriented. However, Cisco is mostly concerned with you understanding the connection-oriented portion of the Transport layer. The following sections provide the skinny on the connection-oriented (reliable) protocol of the Transport layer.

Flow Control

Data integrity is ensured at the Transport layer by maintaining *flow control* and by allowing users to request reliable data transport between systems. Flow control prevents a sending host on one side of the connection from overflowing the buffers in the receiving host—an event that can result in lost data. Reliable data transport employs a connection-oriented communications session between systems, and the protocols involved ensure that the following will be achieved:

- The segments delivered are acknowledged back to the sender upon their reception.
- Any segments not acknowledged are retransmitted.
- Segments are sequenced back into their proper order upon arrival at their destination.
- A manageable data flow is maintained in order to avoid congestion, overloading, and data loss.

Some types of flow control are buffering, congestion avoidance, and windowing.

The purpose of flow control is to provide a means for the receiver to govern the amount of data sent by the sender.

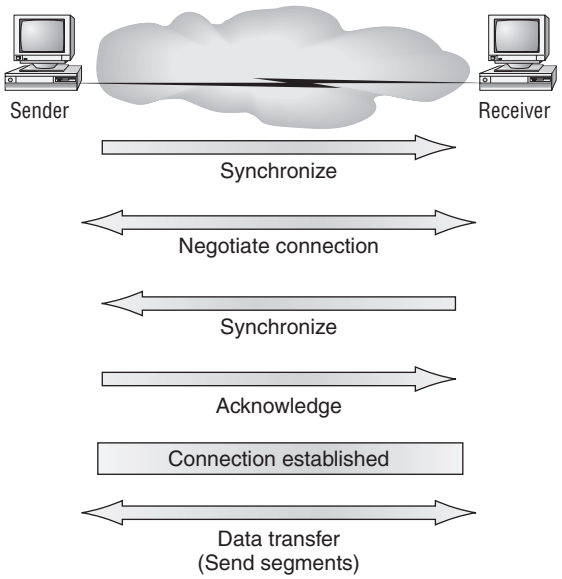
Connection-Oriented Communication

In reliable transport operation, a device that wants to transmit sets up a connection-oriented communication with a remote device by creating a session. The transmitting device first establishes a connection-oriented session, called a *call setup* or a *three-way handshake*, with its peer system. Data is then transferred; when the transfer is finished, a call termination takes place to tear down the virtual circuit.

Figure 1.9 depicts a typical reliable session taking place between sending and receiving systems. Looking at it, you can see that both hosts' application programs begin by notifying their individual operating systems that a connection is about to be initiated. The two operating systems communicate by sending messages over the network confirming that the transfer is approved and that both sides are ready for it to take place. After all of this required synchronization takes place, a connection is fully established and the data transfer begins (this virtual circuit setup is called overhead!).

While the information is being transferred between hosts, the two machines periodically check in with each other, communicating through their protocol software to ensure that all is going well and that the data is being received properly.

FIGURE 1.9 Establishing a connection-oriented session



Let me sum up the steps in the connection-oriented session—the three-way handshake—pictured in Figure 1.9:

- The first "connection agreement" segment is a request for synchronization.
- The second and third segments acknowledge the request and establish connection parameters—the rules—between hosts. These segments request that the receiver's sequencing is synchronized here as well so that a bidirectional connection is formed.
- The final segment also is an acknowledgment. It notifies the destination host that the connection agreement has been accepted and that the actual connection has been established. Data transfer can now begin.

Sounds pretty simple, but things don't always flow so smoothly. Sometimes during a transfer, congestion can occur because a high-speed computer is generating data traffic a lot faster than the network can transfer. A bunch of computers simultaneously sending datagrams through a single gateway or destination can also botch things up nicely. In the latter case, a gateway or destination can become congested even though no single source caused the problem. In either case, the problem is basically akin to a freeway bottleneck—too much traffic for too small a capacity. It's not usually one car that's the problem; there are simply too many cars on that freeway.

Okay, so what happens when a machine receives a flood of datagrams too quickly for it to process? It stores them in a memory section called a *buffer*. But this buffering action can solve the problem only if the datagrams are part of a small burst. If not, and the datagram deluge continues, a device's memory will eventually be exhausted, its flood capacity will be exceeded, and it will react by discarding any additional datagrams that arrive.

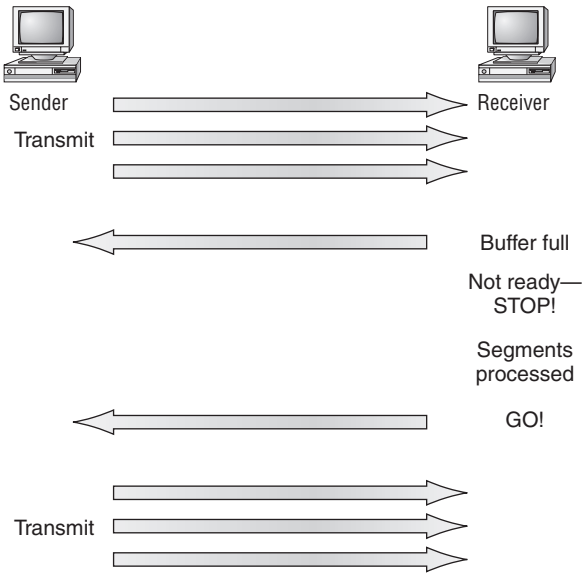
No huge worries here, though. Because of the transport function, network flood control systems really work quite well. Instead of dumping resources and allowing data to be lost, the transport can issue a "not ready" indicator to the sender, or source, of the flood (as shown in Figure 1.10). This mechanism works kind of like a stoplight, signaling the sending device to stop transmitting segment traffic to its overwhelmed peer. After the peer receiver processes the segments already in its memory reservoir—its buffer—it sends out a "ready" transport indicator. When the machine waiting to transmit the rest of its datagrams receives this "go" indictor, it resumes its transmission.

In fundamental, reliable, connection-oriented data transfer, datagrams are delivered to the receiving host in exactly the same sequence they're transmitted—and the transmission fails if this order is breached! If any data segments are lost, duplicated, or damaged along the way, a failure will transmit. This problem is solved by having the receiving host acknowledge that it has received each and every data segment.

A service is considered connection-oriented if it has the following characteristics:

- A virtual circuit is set up (e.g., a three-way handshake).
- It uses sequencing.
- It uses acknowledgments.
- It uses flow control.

FIGURE 1.10 Transmitting segments with flow control



The types of flow control are buffering, windowing, and congestion avoidance.

Windowing

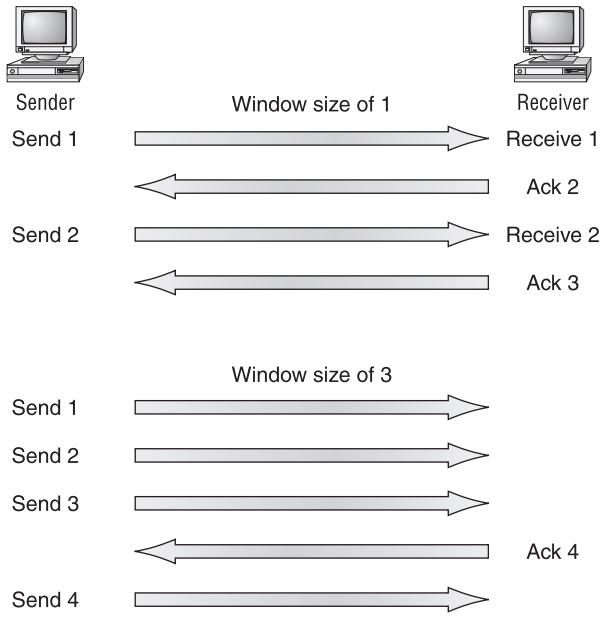
Ideally, data throughput happens quickly and efficiently. And as you can imagine, it would be slow if the transmitting machine had to wait for an acknowledgment after sending each segment. But because there's time available *after* the sender transmits the data segment and *before* it finishes processing acknowledgments from the receiving machine, the sender uses the break as an opportunity to transmit more data. The quantity of data segments (measured in bytes) that the transmitting machine is allowed to send without receiving an acknowledgment for them is called a *window*.

Windows are used to control the amount of outstanding, unacknowledged data segments.

So the size of the window controls how much information is transferred from one end to the other. While some protocols quantify information by observing the number of packets, TCP/IP measures it by counting the number of bytes.

As you can see in Figure 1.11, there are two window sizes—one set to 1 and one set to 3.

FIGURE 1.11 Windowing



When you've configured a window size of 1, the sending machine waits for an acknowledgment for each data segment it transmits before transmitting another. If you've configured a window size of 3, it's allowed to transmit three data segments before an acknowledgment is received.

In our simplified example, both the sending and receiving machines are workstations. In reality this is not done in simple numbers but in the amount of bytes that can be sent.

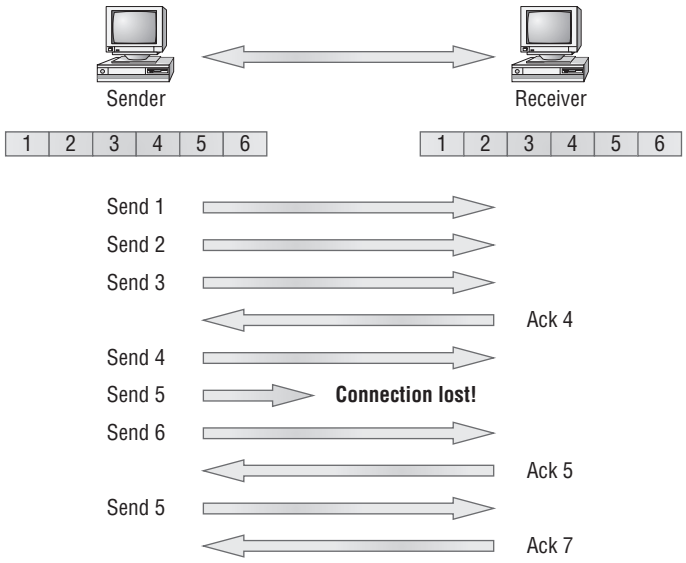
If a receiving host fails to receive all the segments that it should acknowledge, the host can improve the communication session by decreasing the window size.

Acknowledgments

Reliable data delivery ensures the integrity of a stream of data sent from one machine to the other through a fully functional data link. It guarantees that the data won't be duplicated or lost. This is achieved through something called *positive acknowledgment with retransmission*—a technique that requires a receiving machine to communicate with the transmitting source by sending an acknowledgment message back to the sender when it receives data. The sender documents each segment it sends and waits for this acknowledgment before sending the next segment. When it sends a segment, the transmitting machine starts a timer and retransmits if it expires before an acknowledgment is returned from the receiving end.

In Figure 1.12, the sending machine transmits segments 1, 2, and 3. The receiving node acknowledges it has received them by requesting segment 4. When it receives the acknowledgment, the sender then transmits segments 4, 5, and 6. If segment 5 doesn't make it to the destination, the receiving node acknowledges that event with a request for the segment to be resent. The sending machine will then resend the lost segment and wait for an acknowledgment, which it must receive in order to move on to the transmission of segment 7.

FIGURE 1.12 Transport layer reliable delivery



The Network Layer

The *Network layer* (also called layer 3) manages device addressing, tracks the location of devices on the network, and determines the best way to move data, which means that the Network layer must transport traffic between devices that aren't locally attached. Routers (layer 3 devices) are specified at the Network layer and provide the routing services within an internetwork.

It happens like this: First, when a packet is received on a router interface, the destination IP address is checked. If the packet isn't destined for that particular router, it will look up the destination network address in the routing table. Once the router chooses an exit interface, the packet will be sent to that interface to be framed and sent out on the local network. If the router can't find an entry for the packet's destination network in the routing table, the router drops the packet.

Two types of packets are used at the Network layer: data and route updates.

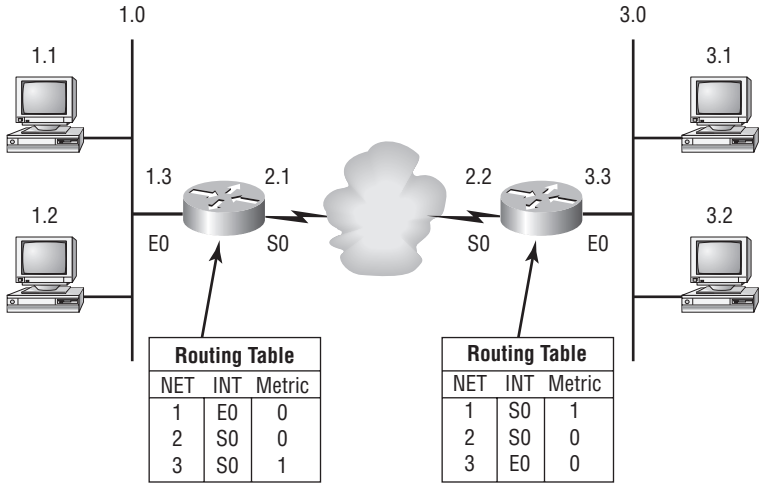
Data packets Data packets are used to transport user data through the internetwork. Protocols used to support data traffic are called *routed protocols*; examples of routed protocols are IP and IPv6. You'll learn about IP addressing in Chapters 2 and 3. (IPv6 is beyond the scope of this book. It is explained in *CCNA: Cisco Certified Network Associate Study Guide*.).

Route update packets Route update packets are used to update neighboring routers about the networks connected to all routers within the internetwork. Protocols that send route update packets are called routing protocols; examples of some common ones are Routing Information Protocol (RIP), RIP version 2 (RIPv2), Enhanced Interior Gateway Routing Protocol (EIGRP), and Open Shortest Path First (OSPF). Route update packets are used to help build and maintain routing tables on each router.

In Figure 1.13, I've given you an example of a routing table. The routing table used in a router includes the following information:

Network addresses Protocol-specific network addresses. A router must maintain a routing table for individual routing protocols because each routing protocol keeps track of a network with a different addressing scheme (IP, IPv6, and Internetwork Packet Exchange [IPX], for example). Think of it as a street sign in each of the different languages spoken by the residents that live on a particular street. So, if there were American, Spanish, and French folks on a street named Cat, the sign would read Cat/Gato/Chat.

FIGURE 1.13 Routing table used in a router



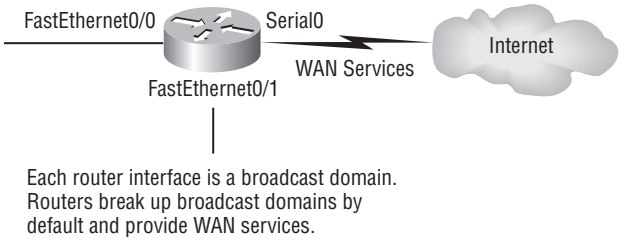
Interface The exit interface a packet will take when destined for a specific network.

Metric The distance to the remote network. Different routing protocols use different ways of computing this distance. I'm going to cover routing protocols in Chapter 6, but for now,

know that some routing protocols (namely RIP) use something called a *hop count* (the number of routers a packet passes through en route to a remote network), while others use bandwidth, delay of the line, or even tick count ($^1/_{18}$ of a second).

And as I mentioned earlier, routers break up broadcast domains, which means that by default, broadcasts aren't forwarded through a router. Do you remember why this is a good thing? Routers also break up collision domains, but you can also do that using layer 2 (Data Link layer) switches. Because each interface in a router represents a separate network, it must be assigned unique network identification numbers, and each host on the network connected to that router must use the same network number. Figure 1.14 shows how a router works in an internetwork.

FIGURE 1.14 A router in an internetwork



Here are some points about routers that you should really commit to memory:

- Routers, by default, will not forward any broadcast or multicast packets.
- Routers use the logical address in a Network layer header to determine the next hop router to forward the packet to.
- Routers can use access lists, created by an administrator, to control security on the types of packets that are allowed to enter or exit an interface.
- Routers can provide layer 2 bridging functions, if needed, and can simultaneously route through the same interface.
- Layer 3 devices (routers in this case) provide connections between VLANs.
- Routers can provide quality of service (QoS) for specific types of network traffic.

Switching is covered in Chapter 7, "LAN Switching."

The Data Link Layer

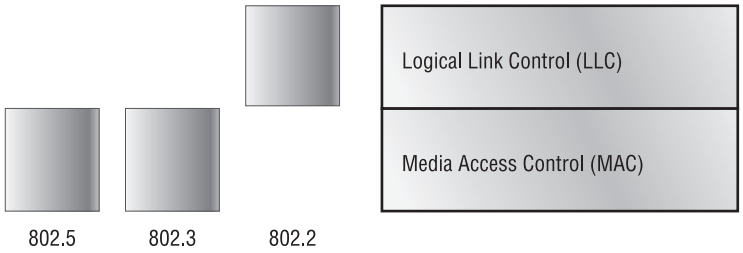
The *Data Link layer* provides the physical transmission of the data and handles error notification, network topology, and flow control. This means that the Data Link layer will ensure

that messages are delivered to the proper device on a LAN using hardware addresses and will translate messages from the Network layer into bits for the Physical layer to transmit.

The Data Link layer formats the message into pieces, each called a *data frame*, and adds a customized header containing the hardware destination and source address. This added information forms a sort of capsule that surrounds the original message in much the same way that engines, navigational devices, and other tools were attached to the lunar modules of the Apollo project. These various pieces of equipment were useful only during certain stages of space flight and were stripped off the module and discarded when their designated stage was complete. Data traveling through networks is similar.

Figure 1.15 shows the Data Link layer with the Ethernet and Institute of Electrical and Electronics Engineers (IEEE) specifications. When you check it out, notice that the IEEE 802.2 standard is used in conjunction with and adds functionality to the other IEEE standards.

FIGURE 1.15 Data Link layer



It's important for you to understand that routers, which work at the Network layer, don't care at all about where a particular host is located. They're only concerned about where networks are located and the best way to reach them—including remote ones. Routers are totally obsessive when it comes to networks. And for once, this is a good thing! It's the Data Link layer that's responsible for the actual unique identification of each device that resides on a local network.

To allow a host to send packets to individual hosts on a local network as well as transmit packets between routers, the Data Link layer uses hardware addressing. Each time a packet is sent between routers, it's framed with control information at the Data Link layer, but that information is stripped off at the receiving router and only the original packet is left completely intact. This framing of the packet continues for each hop until the packet is finally delivered to the correct receiving host. It's really important to understand that the packet itself is never altered along the route; it's only encapsulated with the type of control information required for it to be properly passed on to the different media types.

The IEEE Ethernet Data Link layer has two sublayers:

Media Access Control (MAC) 802.3 Defines how packets are placed on the media. Contention media access is "first come/first served" access where everyone shares the same bandwidth—hence the name. Physical addressing is defined here, as well as logical topologies. What's a logical topology? It's the signal path through a physical topology. Line discipline, error notification (not correction), ordered delivery of frames, and optional flow control can also be used at this sublayer.

Logical Link Control (LLC) 802.2 Responsible for identifying Network layer protocols and then encapsulating them. An LLC header tells the Data Link layer what to do with a packet once a frame is received. It works like this: A host will receive a frame and look in the LLC header to find out where the packet is destined—say, the IP protocol at the Network layer. The LLC can also provide flow control and sequencing of control bits.

The switches and bridges I talked about near the beginning of the chapter both work at the Data Link layer and filter the network using hardware (MAC) addresses. We will look at these in the following section.

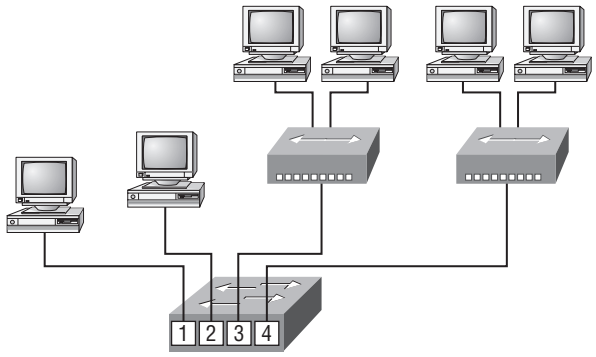
Switches and Bridges at the Data Link Layer

Layer 2 switching is considered hardware-based bridging because it uses specialized hardware called an *application-specific integrated circuit (ASIC)*. ASICs can run up to gigabit speeds with very low latency rates.

Latency is the time measured from when a frame enters a port to the time it exits a port.

Bridges and switches read each frame as it passes through the network. The layer 2 device then puts the source hardware address in a filter table and keeps track of which port the frame was received on. This information (logged in the bridge's or switch's filter table) is what helps the machine determine the location of the specific sending device. Figure 1.16 shows a switch in an internetwork.

FIGURE 1.16 A switch in an internetwork



Each segment has its own collision domain.
All segments are in the same broadcast domain.

The real estate business is all about location, location, location, and it's the same way for both layer 2 and layer 3 devices. Though both need to be able to negotiate the network, it's crucial to remember that they're concerned with very different parts of it. Primarily, layer 3 machines (such as routers) need to locate specific networks, whereas layer 2 machines (switches and bridges) need to eventually locate specific devices. So, networks are to routers what individual devices are to switches and bridges. And routing tables that "map" the internetwork are for routers what filter tables that "map" individual devices are for switches and bridges.

After a filter table is built on the layer 2 device, it will forward frames only to the segment where the destination hardware address is located. If the destination device is on the same segment as the frame, the layer 2 device will block the frame from going to any other segments. If the destination is on a different segment, the frame can be transmitted only to that segment. This is called *transparent bridging*.

When a switch interface receives a frame with a destination hardware address that isn't found in the device's filter table, it forwards the frame to all connected segments. If the unknown device that was sent the "mystery frame" replies to this forwarding action, the switch updates its filter table regarding that device's location. But in the event the destination address of the transmitting frame is a broadcast address, the switch will forward all broadcasts to every connected segment by default.

All devices that the broadcast is forwarded to are considered to be in the same broadcast domain. This can be a problem; layer 2 devices propagate layer 2 broadcast storms that choke performance, and the only way to stop a broadcast storm from propagating through an internetwork is with a layer 3 device—a router.

The biggest benefit of using switches instead of hubs in your internetwork is that each switch port is actually its own collision domain. (Conversely, a hub creates one large collision domain.) But even armed with a switch, you still can't break up broadcast domains. Neither switches nor bridges will do that. They'll typically simply forward all broadcasts instead.

Another benefit of LAN switching over hub-centered implementations is that each device on every segment plugged into a switch can transmit simultaneously—at least, they can as long as there is only one host on each port and a hub isn't plugged into a switch port. As you might have guessed, hubs allow only one device per network segment to communicate at a time.

Binary to Decimal and Hexadecimal Conversion

Before we finish this chapter and move on to discussing the TCP/IP protocol stack and IP addressing in Chapter 2, 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), and each digit is called 1 bit (short for *bi*nary digi*t*). Typically, you count either 4 or 8 bits together, with these being referred to as a nibble and 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 1.1 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 1.1 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 1. And if a zero (0) is placed in a bit spot, 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; that 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, which demonstrates the maximum value of a byte:

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

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 1.2 is a table you should memorize before braving the IP sections in Chapters 2 and 3.

TABLE 1.2 Binary to Decimal Memorization Chart

Binary Value	Decimal Value
10000000	128
11000000	192

TABLE 1.2 Binary to Decimal Memorization Chart *(continued)*

Binary Value	Decimal Value
11100000	224
11110000	240
11111000	248
11111100	252
11111110	254
11111111	255

Hexadecimal addressing is completely different from 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, and so on can't be used (because they are two-digit numbers), the letters *A*, *B*, *C*, *D*, *E*, and *F* are used to represent 10, 11, 12, 13, 14, and 15, respectively.

Hex is short for *hexadecimal,* which is a numbering system that uses the first 6 letters of the alphabet (*A* through *F*) to extend beyond the available 10 digits in the decimal system. Hexadecimal has a total of 16 digits.

Table 1.3 shows both the binary value and the decimal value for each hexadecimal digit.

TABLE 1.3 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

TABLE 1.3 Hex to Binary to Decimal Chart *(continued)*

Hexadecimal Value	Binary Value	Decimal Value
6	0110	6
7	0111	7
8	1000	8
9	1001	9
A	1010	10
B	1011	11
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. It doesn'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 and then put them together into a byte. 6 = 0110 and A (which is 10 in hex) = 1010, 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 0x55. And in decimal format, the binary number is 01010101, which converts to 64 + 16 + 4 + 1 = 85.

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 need to get working on the Physical layer. Suppose you had the following binary number:

10110101

The hex answer would be 0xB5, since 1011 converts to B and 0101 converts to 5 in hex value. The decimal equivalent is 128 + 32 + 16 + 4 + 1 = 181.

See Written Lab 1.4 for more practice with binary/hex/decimal conversion.

The Physical Layer

Finally arriving at the bottom, we find that the *Physical layer* does two things: It sends bits and receives bits. Bits come only in values of 1 or 0—a Morse code with numerical values. The Physical layer communicates directly with the various types of actual communication media. Different kinds of media represent these bit values in different ways. Some use audio tones, while others employ *state transitions*—changes in voltage from high to low and low to high. Specific protocols are needed for each type of media to describe the proper bit patterns to be used, how data is encoded into media signals, and the various qualities of the physical media's attachment interface.

The Physical layer specifies the electrical, mechanical, procedural, and functional requirements for activating, maintaining, and deactivating a physical link between end systems. This layer is also where you identify the interface between the *data terminal equipment (DTE)* and the *data communication equipment (DCE)*. (Some old phone-company employees still call DCE data circuit-terminating equipment.) The DCE is usually located at the service provider, while the DTE is the attached device. The services available to the DTE are most often accessed via a modem or *channel service unit/data service unit (CSU/DSU)*.

The Physical layer's connectors and different physical topologies are defined by the OSI as standards, allowing disparate systems to communicate. The CCENT objectives are only interested in the IEEE Ethernet standards.

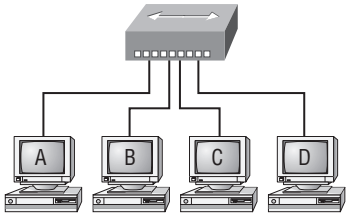
Hubs at the Physical Layer

A *hub* is really a multiple-port repeater. A repeater receives a digital signal and reamplifies or regenerates that signal and then forwards it out all active ports without looking at any data. An active hub does the same thing. Any digital signal received from a segment on a hub port is regenerated or reamplified and transmitted out all ports on the hub. This means all devices plugged into a hub are in the same collision domain as well as in the same broadcast domain. Figure 1.17 shows a hub in a network.

Hubs, like repeaters, don't examine any of the traffic as it enters and is then transmitted out to the other parts of the physical media. Every device connected to the hub, or hubs, must listen if a device transmits. A physical star network—where the hub is a central device and cables extend in all directions out from it—is the type of topology a hub creates. Visually, the design really does resemble a star, whereas Ethernet networks run a logical bus topology, meaning that the signal has to run through the network from end to end.

Hubs and repeaters can be used to enlarge the area covered by a single LAN segment, although I do not recommend this. LAN switches are affordable for almost every situation.

FIGURE 1.17 A hub in a network



All devices in the same collision domain.
All devices in the same broadcast domain.
Devices share the same bandwidth.

Ethernet Networking

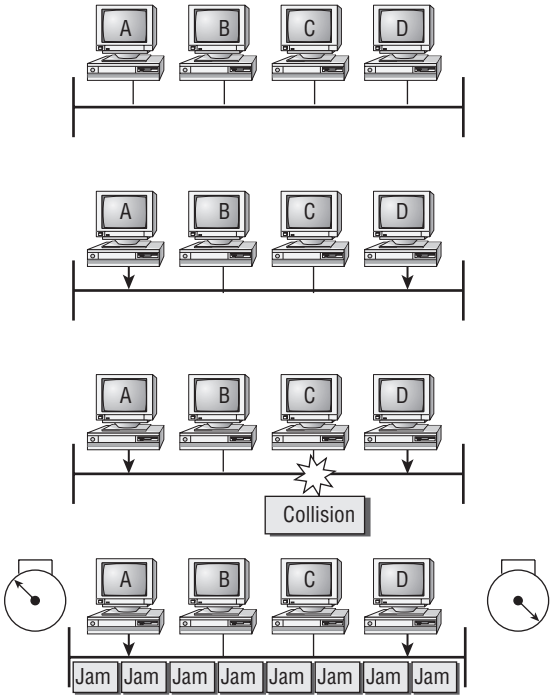
Ethernet is a contention media access method that allows all hosts on a network to share the same bandwidth of a link. Ethernet is popular because it's readily scalable, meaning that it's comparatively easy to integrate new technologies, such as Fast Ethernet and Gigabit Ethernet, into an existing network infrastructure. It's also relatively simple to implement in the first place, and with it, troubleshooting is reasonably straightforward. Ethernet uses both Data Link and Physical layer specifications, and this section of the chapter will give you both the Data Link layer and Physical layer information you need to effectively implement, troubleshoot, and maintain an Ethernet network.

Ethernet networking uses *Carrier Sense Multiple Access with Collision Detection (CSMA/CD)*, a protocol that helps devices share the bandwidth evenly without having two devices transmit at the same time on the network medium. CSMA/CD was created to overcome the problem of those collisions that occur when packets are transmitted simultaneously from different nodes. And trust me—good collision management is crucial, because when a node transmits in a CSMA/CD network, all the other nodes on the network receive and examine that transmission. Only bridges and routers can effectively prevent a transmission from propagating throughout the entire network!

So, how does the CSMA/CD protocol work? Let's start by taking a look at Figure 1.18.

When a host wants to transmit over the network, it first checks for the presence of a digital signal on the wire. If all is clear (no other host is transmitting), the host will then proceed with its transmission. But it doesn't stop there. The transmitting host constantly monitors the wire to make sure no other hosts begin transmitting. If the host detects another signal on the wire, it sends out an extended jam signal that causes all nodes on the segment to stop sending data (think busy signal). The nodes respond to that jam signal by waiting a while before attempting to transmit again. Backoff algorithms determine when the colliding stations can retransmit. If collisions keep occurring after 15 tries, the nodes attempting to transmit will then timeout. Pretty clean!

FIGURE 1.18 CSMA/CD



Carrier Sense Multiple Access with Collision Detection (CSMA/CD)

When a collision occurs on an Ethernet LAN, the following happens:

- A jam signal informs all devices that a collision occurred.
- The collision invokes a random backoff algorithm.
- Each device on the Ethernet segment stops transmitting for a short time until the timers expire.
- All hosts have equal priority to transmit after the timers have expired.

The following are the effects of having a CSMA/CD network sustaining heavy collisions:

- Delay
- Low throughput
- Congestion

Backoff on an 802.3 network is the retransmission delay that's enforced when a collision occurs. When a collision occurs, a host will resume transmission after the forced time delay has expired. After this backoff delay period has expired, all stations have equal priority to transmit data.

In the following sections, I am going to cover Ethernet in detail at both the Data Link layer (layer 2) and the Physical layer (layer 1).

Half- and Full-Duplex Ethernet

Half-duplex Ethernet is defined in the original 802.3 Ethernet; Cisco says it uses only one wire pair with a digital signal running in both directions on the wire. Certainly, the IEEE specifications discuss the process of half duplex somewhat differently, but what Cisco is talking about is a general sense of what is happening here with Ethernet.

It also uses the CSMA/CD protocol to help prevent collisions and to permit retransmitting if a collision does occur. If a hub is attached to a switch, it must operate in half-duplex mode because the end stations must be able to detect collisions. Half-duplex Ethernet—typically 10BaseT—is only about 30 to 40 percent efficient as Cisco sees it because a large 10BaseT network will usually only give you 3 to 4Mbps, at most.

But full-duplex Ethernet uses two pairs of wires instead of one wire pair like half duplex. And full duplex uses a point-to-point connection between the transmitter of the transmitting device and the receiver of the receiving device. This means that with full-duplex data transfer, you get a faster data transfer compared to half duplex. And because the transmitted data is sent on a different set of wires than the received data, no collisions will occur.

The reason you don't need to worry about collisions is because now it's like a freeway with multiple lanes instead of the single-lane road provided by half duplex. Full-duplex Ethernet is supposed to offer 100 percent efficiency in both directions—for example, you can get 20Mbps with a 10Mbps Ethernet running full duplex or 200Mbps for Fast Ethernet. But this rate is something known as an aggregate rate, which translates as "you're supposed to get" 100 percent efficiency. No guarantees, in networking as in life.

Full-duplex Ethernet can be used in three situations:

- With a connection from a switch to a host
- With a connection from a switch to a switch
- With a connection from a host to a host using a crossover cable

Full-duplex Ethernet requires a point-to-point connection when only two nodes are present. You can run full duplex with just about any device except a hub.

Now, if it's capable of all that speed, why wouldn't it deliver? Well, when a full-duplex Ethernet port is powered on, it first connects to the remote end and then negotiates with the other end of the Fast Ethernet link. This is called an *auto-detect mechanism*. This mechanism first decides on the exchange capability, which means it checks to see if it can run at 10 or 100Mbps. It then checks to see if it can run full duplex, and if it can't, it will run half duplex.

Remember that half-duplex Ethernet shares a collision domain and provides a lower effective throughput than full-duplex Ethernet, which typically has a private collision domain and a higher effective throughput.

Last, remember these important points:

- There are no collisions in full-duplex mode.
- A dedicated switch port is required for each full-duplex node.
- Both the host network card and the switch port must be capable of operating in full-duplex mode.

Now let's take a look at how Ethernet works at the Data Link layer.

Ethernet at the Data Link Layer

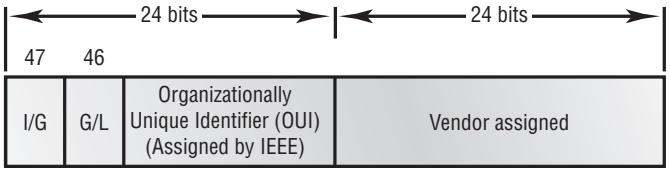
Ethernet at the Data Link layer is responsible for Ethernet addressing, commonly referred to as hardware addressing or MAC addressing. Ethernet is also responsible for framing packets received from the Network layer and preparing them for transmission on the local network through the Ethernet contention media access method.

Ethernet Addressing

Here's where we get into how Ethernet addressing works. It uses the MAC address burned into each and every Ethernet NIC. The MAC, or hardware, address is a 48-bit (6-byte) address written in a hexadecimal format.

Figure 1.19 shows the 48-bit MAC addresses and how the bits are divided.

FIGURE 1.19 Ethernet addressing using MAC addresses



The *organizationally unique identifier (OUI)* is assigned by the IEEE to an organization. It's composed of 24 bits, or 3 bytes. The organization, in turn, assigns a globally administered address (24 bits, or 3 bytes) that is unique (supposedly, again—no guarantees) to each and every adapter it manufactures. Look closely at the figure. The high-order bit is the Individual/Group (I/G) bit. When it has a value of 0, we can assume that the address is the MAC address of a device and may well appear in the source portion of the MAC header. When it is a 1, we can assume that the address represents either a broadcast or multicast address in Ethernet or a broadcast or functional address in Token Ring (TR) and Fiber Distributed Data Interface (FDDI). And who really knows about FDDI?

The next bit is the global/local bit, or just G/L bit (also known as U/L, where *U* means *universal*). When set to 0, this bit represents a globally administered address (as in administered by the IEEE). When the bit is a 1, it represents a locally governed and administered address (as in what DECnet used to do). The low-order 24 bits of an Ethernet address represent a locally administered or manufacturer-assigned code. This portion commonly starts with 24 0s for the first card made

and continues in order until there are 24 1s for the last (16,777,216th) card made. You'll find that many manufacturers use these same 6 hex digits as the last 6 characters of their serial number on the same card.

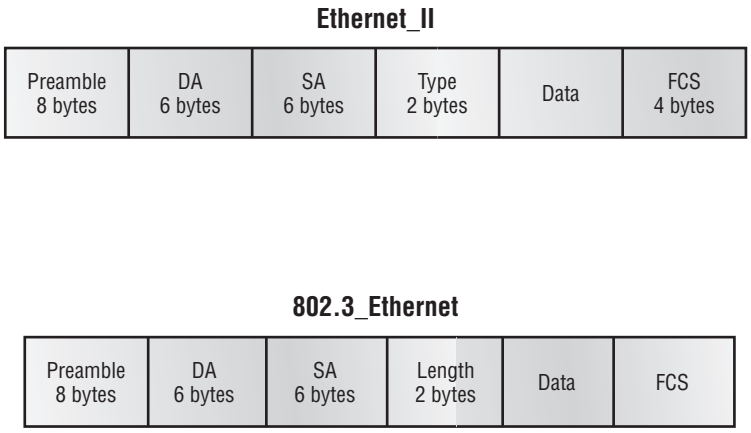
Ethernet Frames

The Data Link layer is responsible for combining bits into bytes and bytes into frames. Frames are used at the Data Link layer to encapsulate packets handed down from the Network layer for transmission on a type of media access.

The function of Ethernet stations is to pass data frames between each other using a group of bits known as a MAC frame format. This provides error detection from a cyclic redundancy check (CRC). But remember—this is error detection, not error correction. An 802.3 frame and Ethernet_II frame are shown in Figure 1.20.

Encapsulating a frame within a different type of frame is called *tunneling*.

FIGURE 1.20 802.3 and Ethernet frame formats



Following are the details of the different fields in the 802.3 and Ethernet frame types:

Preamble An alternating 1,0 pattern provides a 5MHz clock at the start of each packet, which allows the receiving devices to lock the incoming bit stream. The preamble is seven octets.

Start Frame Delimiter (SFD)/Synch The SFD is one octet (synch). The SFD is 10101011, where the last pair of 1s allows the receiver to come into the alternating 1,0 pattern somewhere in the middle and still sync up and detect the beginning of the data.

Destination Address (DA) This transmits a 48-bit value using the least significant bit (LSB) first. The DA is used by receiving stations to determine whether an incoming packet is

addressed to a particular node. The destination address can be an individual address or a broadcast or multicast MAC address. Remember that a broadcast is all 1s (or *F*s in hex) and is sent to all devices but a multicast is sent only to a similar subset of nodes on a network.

Source Address (SA) The SA is a 48-bit MAC address used to identify the transmitting device, and it is transmitted LSB first. Broadcast and multicast address formats are illegal within the SA field.

Length or Type 802.3 uses a Length field, but the Ethernet frame uses a Type field to identify the Network layer protocol. 802.3 cannot identify the upper-layer protocol and must be used with a proprietary LAN—IPX, for example.

Data This is a packet sent down to the Data Link layer from the Network layer. The size can vary from 64 to 1500 bytes.

Frame Check Sequence (FCS) FCS is a field at the end of the frame that's used to store the CRC.

Let's pause here for a minute and take a look at some frames caught on our trusty OmniPeek network analyzer. You can see that the frame below has only three fields: Destination, Source, and Type (shown as Protocol Type on this analyzer):

```
Destination:   00:60:f5:00:1f:27
Source:        00:60:f5:00:1f:2c
Protocol Type: 08-00 IP
```

This is an Ethernet_II frame. Notice that the type field is IP, or 08-00 (mostly just referred to as 0x800) in hexadecimal.

The next frame has the same fields, so it must be an Ethernet_II frame too:

```
Destination:   ff:ff:ff:ff:ff:ff Ethernet Broadcast
Source:        02:07:01:22:de:a4
Protocol Type: 08-00 IP
```

Did you notice that this frame was a broadcast? You can tell because the destination hardware address is all 1s in binary, or all *F*s in hexadecimal.

Let's take a look at one more Ethernet_II frame. You can see that the Ethernet frame is the same Ethernet_II frame we use with the IPv4 routed protocol but the type field has 0x86dd when we are carrying IPv6 data, and when we have IPv4 data, we use 0x0800 in the protocol field:

```
Destination: IPv6-Neighbor-Discovery_00:01:00:03 (33:33:00:01:00:03)
Source: Aopen_3e:7f:dd (00:01:80:3e:7f:dd)
Type: IPv6 (0x86dd)
```

This is the beauty of the Ethernet_II frame. Because of the protocol field, we can run any Network layer routed protocol and it will carry the data because it can identify the Network layer protocol.

Ethernet at the Physical Layer

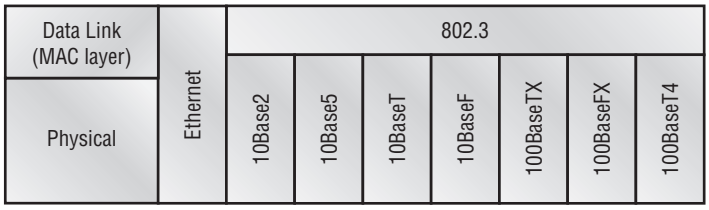
Ethernet was first implemented by a group called DIX (Digital, Intel, and Xerox). They created and implemented the first Ethernet LAN specification, which the IEEE used to create the IEEE 802.3 Committee. This was a 10Mbps network that ran on coax and then eventually twisted-pair and fiber physical media.

The IEEE extended the 802.3 Committee to two new committees known as 802.3u (Fast Ethernet) and 802.3ab (Gigabit Ethernet on category 5) and then finally 802.3ae (10Gbps over fiber and coax).

Figure 1.21 shows the IEEE 802.3 and original Ethernet Physical layer specifications.

When designing your LAN, it's really important to understand the different types of Ethernet media available to you. Sure, it would be great to run Gigabit Ethernet to each desktop and 10Gbps between switches, and although this might happen one day, justifying the cost of that network today would be pretty difficult. But if you mix and match the different types of Ethernet media methods currently available, you can come up with a cost-effective network solution that works great.

FIGURE 1.21 Ethernet Physical layer specifications



The Electronic Industries Association and the newer Telecommunications Industry Alliance (EIA/TIA) is the standards body that creates the Physical layer specifications for Ethernet. The EIA/TIA specifies that Ethernet use a *registered jack (RJ) connector* with a 4 5 wiring sequence on *unshielded twisted-pair (UTP)* cabling (RJ45). However, the industry is moving toward calling this just an 8-pin modular connector.

Each Ethernet cable type that is specified by the EIA/TIA has inherent attenuation, which is defined as the loss of signal strength as it travels the length of a cable and is measured in decibels (dB). The cabling used in corporate and home markets is measured in categories. A higher-quality cable will have a higher-rated category and lower attenuation. For example, category 5 is better than category 3 because category 5 cables have more wire twists per foot and therefore less crosstalk. Crosstalk is the unwanted signal interference from adjacent pairs in the cable.

Here are the original IEEE 802.3 standards:

10Base2 10Mbps, baseband technology, up to 185 meters in length. Known as *thinnet* and can support up to 30 workstations on a single segment. Uses a physical and logical bus with Attachment Unit Interface (AUI) connectors. The 10 means 10Mbps, *Base* means baseband technology (which is a signaling method for communication on the network), and the 2 means

almost 200 meters. 10Base2 Ethernet cards use use BNC and T-connectors to connect to a network. (BNC stands for British Naval Connector, Bayonet Neill Concelman, or Bayonet Nut.)

10Base5 10Mbps, baseband technology, up to 500 meters in length. Known as *thicknet*. Uses a physical and logical bus with AUI connectors. Up to 2,500 meters with repeaters and 1,024 users for all segments.

10BaseT 10Mbps using category 3 UTP wiring. Unlike with the 10Base2 and 10Base5 networks, each device must connect into a hub or switch, and you can have only one host per segment or wire. Uses an RJ45 connector (8-pin modular connector) with a physical star topology and a logical bus.

Each of the 802.3 standards defines an AUI, which allows a one-bit-at-a-time transfer to the Physical layer from the Data Link media access method. This allows the MAC to remain constant but means the Physical layer can support any existing and new technologies. The original AUI interface was a 15-pin connector, which allowed a transceiver (transmitter/receiver) that provided a 15-pin-to-twisted-pair conversion.

The thing is, the AUI interface cannot support 100Mbps Ethernet because of the high frequencies involved. So 100BaseT needed a new interface, and the 802.3u specifications created one called the Media Independent Interface (MII), which provides 100Mbps throughput. The MII uses a *nibble*, defined as 4 bits. Gigabit Ethernet uses a Gigabit Media Independent Interface (GMII) and transmits 8 bits at a time.

802.3u (Fast Ethernet) is compatible with 802.3 Ethernet because they share the same physical characteristics. Fast Ethernet and Ethernet use the same maximum transmission unit (MTU), use the same MAC mechanisms, and preserve the frame format that is used by 10BaseT Ethernet. Basically, Fast Ethernet is just based on an extension to the IEEE 802.3 specification, except that it offers a speed increase of 10 times that of 10BaseT.

Here are the expanded IEEE Ethernet 802.3 standards:

100BaseTX (IEEE 802.3u) EIA/TIA category 5, 6, or 7 UTP two-pair wiring. One user per segment; up to 100 meters long. It uses an RJ45 connector with a physical star topology and a logical bus.

100BaseFX (IEEE 802.3u) Uses fiber cabling 62.5/125-micron multimode fiber. Point-to-point topology; up to 412 meters long. It uses an ST or SC connector, which are media-interface connectors.

1000BaseCX (IEEE 802.3z) Copper twisted-pair called twinax (a balanced coaxial pair) that can only run up to 25 meters.

1000BaseT (IEEE 802.3ab) Category 5, four-pair UTP wiring up to 100 meters long.

1000BaseSX (IEEE 802.3z) MMF using 62.5- and 50-micron core; uses an 850 nano-meter laser and can go up to 220 meters with 62.5-micron, 550 meters with 50-micron.

1000BaseLX (IEEE 802.3z) Single-mode fiber that uses a 9-micron core and 1300 nano-meter laser and can go from 3 kilometers up to 10 kilometers.

If you want to implement an Ethernet network medium that is not susceptible to electromagnetic interference (EMI) and voltage potential differences, fiber-optic cable provides a more secure, long-distance cable that is not susceptible to EMI at high speeds.

Ethernet Cabling

Ethernet cabling is an important discussion, especially if you are planning on taking the Cisco exams. Three types of Ethernet cables are available:

- Straight-through cable
- Crossover cable
- Rolled cable

We will look at each in the following sections.

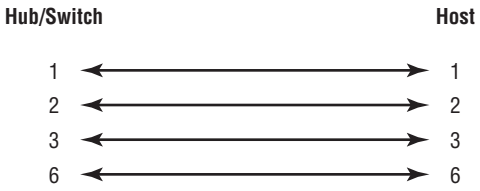
Straight-Through Cable

The *straight-through cable* is used to connect the following:

- Host to switch or hub
- Router to switch or hub

Four wires are used in straight-through cable to connect Ethernet devices. It is relatively simple to create this type; Figure 1.22 shows the four wires used in a straight-through Ethernet cable.

FIGURE 1.22 Straight-through Ethernet cable



Notice that only pins 1, 2, 3, and 6 are used. Just connect 1 to 1, 2 to 2, 3 to 3, and 6 to 6 and you'll be up and networking in no time. However, remember that this would be an Ethernet-only cable and wouldn't work with voice, Token Ring, Integrated Services Digital Network (ISDN), and so on.

Crossover Cable

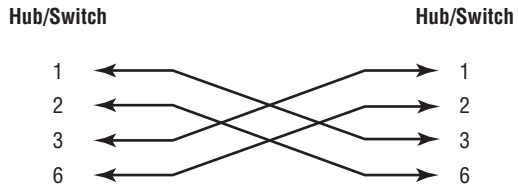
The *crossover cable* can be used to connect the following:

- Switch to switch
- Hub to hub
- Host to host
- Hub to switch
- Router direct to host

The same four wires are used in this cable as in the straight-through cable; we just connect different pins together. Figure 1.23 shows how the four wires are used in a crossover Ethernet cable.

Notice that instead of connecting 1 to 1, 2 to 2, and so on, here we connect pins 1 to 3 and 2 to 6 on each side of the cable.

FIGURE 1.23 Crossover Ethernet cable

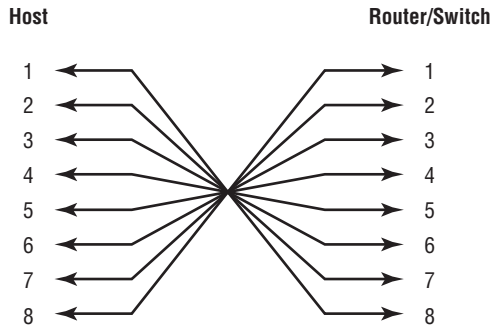


Rolled Cable

Although *rolled cable* isn't used to connect any Ethernet connections together, you can use a rolled Ethernet cable to connect a host to a router console serial communication (com) port.

If you have a Cisco router or switch, you would use this cable to connect your PC running HyperTerminal to the Cisco hardware. Eight wires are used in this cable to connect serial devices, although not all eight are used to send information, just as in Ethernet networking. Figure 1.24 shows the eight wires used in a rolled cable.

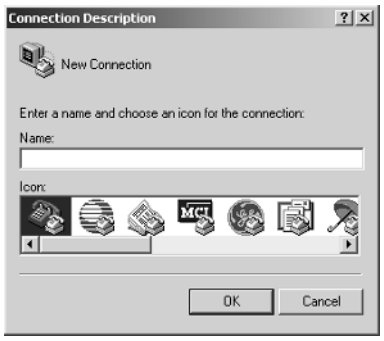
FIGURE 1.24 Rolled Ethernet cable



These are probably the easiest cables to make because you just cut the end off on one side of a straight-through cable, turn it over, and put it back on (with a new connector, of course).

Once you have the correct cable connected from your PC to the Cisco router or switch, you can start HyperTerminal to create a console connection and configure the device. Set the configuration as follows:

1. Open HyperTerminal and enter a name for the connection. It is irrelevant what you name it, but I always just use Cisco. Then click OK.



2. Choose the communications port—either COM1 or COM2, whichever is open on your PC.

3. Now set the port settings. The default values (2400bps and no flow control hardware) will not work; you must set the port settings as shown in Figure 1.25.

Notice that the bit rate is now set to 9600 and the flow control is set to None. At this point, you can click OK and press the Enter key and you should be connected to your Cisco device console port.

We've taken a look at the various RJ45 unshielded twisted-pair (UTP) cables. Keeping this in mind, what cable is used between the switches in Figure 1.26?

In order for host A to ping host B, you need a crossover cable to connect the two switches together. But what types of cables are used in the network shown in Figure 1.27?

In Figure 1.27, there are a variety of cables in use. For the connection between the switches, we'd obviously use a crossover cable like we saw in Figure 1.23. The trouble is, we have a console connection that uses a rolled cable. Plus, the connection from the router to the switch is a straight-through cable, as is true for the hosts to the switches. Keep in mind that if we had a serial connection (which we don't), it would be a V.35 that we'd use to connect us to a WAN.

FIGURE 1.25 Port settings for a rolled cable connection

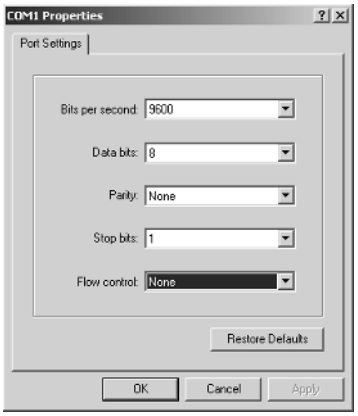


FIGURE 1.26 RJ45 UTP cable question #1

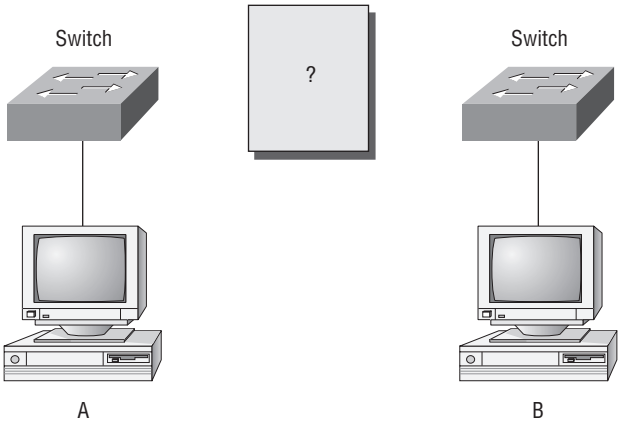
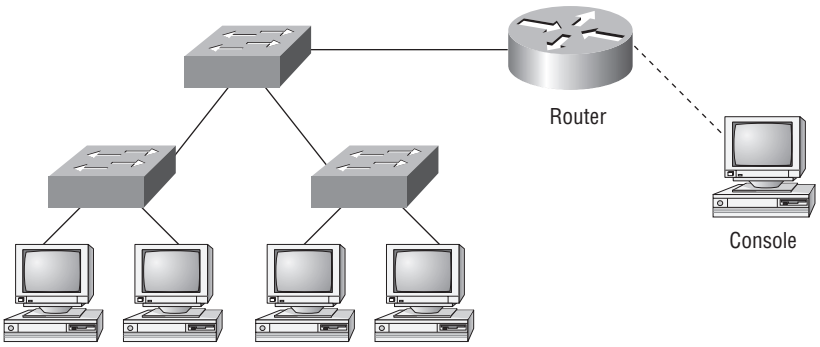


FIGURE 1.27 RJ45 UTP cable question #2



Data Encapsulation

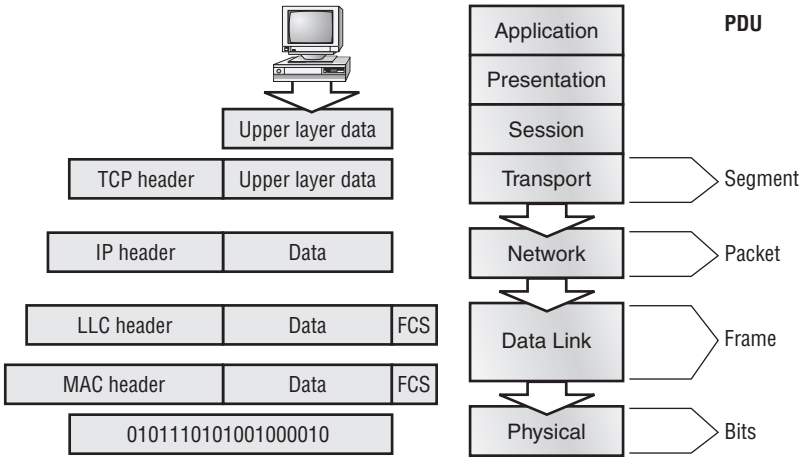
When a host transmits data across a network to another device, the data goes through *encapsulation*: It is wrapped with protocol information at each layer of the OSI model. Each layer communicates only with its peer layer on the receiving device.

To communicate and exchange information, each layer uses *Protocol Data Units (PDUs)*. These hold the control information attached to the data at each layer of the model. They are usually attached to the header in front of the data field but can also be in the trailer, or end, of it.

Each PDU attaches to the data by encapsulating it at each layer of the OSI model, and each has a specific name depending on the information provided in each header. This PDU information is read only by the peer layer on the receiving device. After it's read, it's stripped off and the data is then handed to the next layer up.

Figure 1.28 shows the PDUs and how they attach control information to each layer. This figure demonstrates how the upper-layer user data is converted for transmission on the network. The data stream is then handed down to the Transport layer, which sets up a virtual circuit to the receiving device by sending over a synch packet. Next, the data stream is broken up into smaller pieces, and a Transport layer header (a PDU) is created and attached to the header of the data field; now the piece of data is called a segment. Each segment is sequenced so the data stream can be put back together on the receiving side exactly as it was transmitted.

FIGURE 1.28 Data encapsulation



Each segment is then handed to the Network layer for network addressing and routing through the internetwork. Logical addressing (for example, IP) is used to get each segment to the correct network. The Network layer protocol adds a control header to the segment handed down from the Transport layer, and what we have now is called a *packet* or *datagram*. Remember that the Transport and Network layers work together to rebuild a data stream on a receiving host, but it's not part of their work to place their PDUs on a local network segment—which is the only way to get the information to a router or host.

It's the Data Link layer that's responsible for taking packets from the Network layer and placing them on the network medium (cable or wireless). The Data Link layer encapsulates each packet in a *frame*, and the frame's header carries the hardware address of the source and destination hosts. If the destination device is on a remote network, then the frame is sent to a router to be routed through an internetwork. Once it gets to the destination network, a new frame is used to get the packet to the destination host.

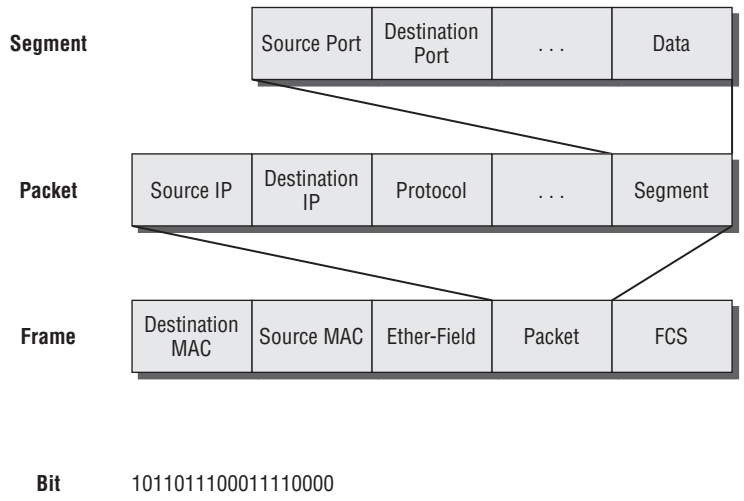
To put this frame on the network, it must first be put into a digital signal. Since a frame is really a logical group of 1s and 0s, the Physical layer is responsible for encoding these digits into a digital signal, which is read by devices on the same local network. The receiving devices will synchronize on the digital signal and extract (decode) the 1s and 0s from the digital signal. At this point, the devices build the frames, run a CRC, and then check their answer against the answer in the frame's FCS field. If it matches, the packet is pulled from the frame and what's left of the frame is discarded. This process is called *de-encapsulation*. The packet is handed to the Network layer, where the address is checked. If the address matches, the segment is pulled from the packet and what's left of the packet is discarded. The segment is processed at the Transport layer, which rebuilds the data stream and acknowledges to the transmitting station that it received each piece. It then happily hands the data stream to the upper-layer application.

At a transmitting device, the data encapsulation method works like this:

1. User information is converted to data for transmission on the network.
2. Data is converted to segments and a reliable connection is set up between the transmitting and receiving hosts.
3. Segments are converted to packets or datagrams, and a logical address is placed in the header so each packet can be routed through an internetwork.
4. Packets or datagrams are converted to frames for transmission on the local network. Hardware (Ethernet) addresses are used to uniquely identify hosts on a local network segment.
5. Frames are converted to bits, and a digital encoding and clocking scheme is used.

To explain this in more detail using the layer addressing, I'll use Figure 1.29.

FIGURE 1.29 PDU and layer addressing

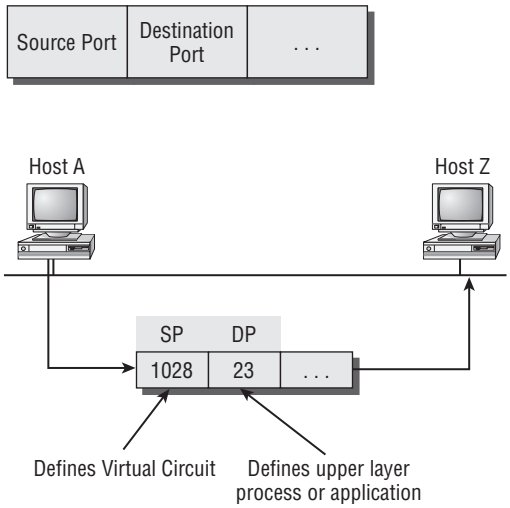


Remember that a data stream is handed down from the upper layer to the Transport layer. As technicians, we really don't care who the data stream comes from because that's really a programmer's problem. Our job is to rebuild the data stream reliably and hand it to the upper layers on the receiving device.

When the receiving host receives the data, a de-encapsulation will occur—meaning we'll start from the Physical layer and go up to the Application layer, removing headers as we go. For example, if a router receives a frame on an interface, the router will take the packet from the frame and then search the routing table to determine where to forward the packet, i.e., the exit interface.

Before we go further in our discussion of Figure 1.29, let's discuss port numbers and make sure we understand them. The Transport layer uses port numbers to define both the virtual circuit and the upper-layer process, as you can see from Figure 1.30.

FIGURE 1.30 Port numbers at the Transport layer



The Transport layer takes the data stream, makes segments out of it, and establishes a reliable session by creating a virtual circuit. It then sequences (numbers) each segment and uses acknowledgments and flow control. If you're using TCP, the virtual circuit is defined by the source port number. Remember, the host just makes this up starting at port number 1024 (0 through 1023 are reserved for use as well-known port numbers). The destination port number defines the upper-layer process (application) that the data stream is handed to when the data stream is reliably rebuilt on the receiving host.

Now that you understand port numbers and how they are used at the Transport layer, let's go back to Figure 1.29. Once the Transport layer header information is added to the piece of data, it becomes a segment and is handed down to the Network layer along with

the destination IP address. (The destination IP address was handed down from the upper layers to the Transport layer with the data stream, and it was discovered through a name resolution method at the upper layers—probably DNS.)

The Network layer adds a header, and adds the logical addressing (IP addresses), to the front of each segment. Once the header is added to the segment, the PDU is called a packet. The packet has a protocol field that describes where the segment came from (either UDP or TCP) so it can hand the segment to the correct protocol at the Transport layer when it reaches the receiving host.

The Network layer is responsible for finding the destination hardware address that dictates where the packet should be sent on the local network. It does this by using the Address Resolution Protocol (ARP)—something I'll talk about more in Chapter 2. IP at the Network layer looks at the destination IP address and compares that address to its own source IP address and subnet mask. If it turns out to be a local network request, the hardware address of the local host is requested via an ARP request. If the packet is destined for a remote host, IP will look for the IP address of the default gateway (router) instead.

The packet, along with the destination hardware address of either the local host or default gateway, is then handed down to the Data Link layer. The Data Link layer will add a header to the front of the packet and the piece of data then becomes a frame. (We call it a frame because both a header and a trailer are added to the packet, which makes the data resemble bookends or a frame, if you will.) This is shown in Figure 1.29. The frame uses an Ether-Type field to describe which protocol the packet came from at the Network layer. Now a CRC is run on the frame, and the answer to the CRC is placed in the Frame Check Sequence field found in the trailer of the frame.

The frame is now ready to be handed down, one bit at a time, to the Physical layer, which will use bit timing rules to encode the data in a digital signal. Every device on the network segment will synchronize with the clock and extract the 1s and 0s from the digital signal and build a frame. After the frame is rebuilt, a CRC is run to make sure the frame is okay. If everything turns out to be all good, the hosts will check the destination address to see if the frame is for them.

If all this is making your eyes cross and your brain freeze, don't freak. I'll be going over exactly how data is encapsulated and routed through an internetwork in Chapter 6.

The Cisco Three-Layer Hierarchical Model

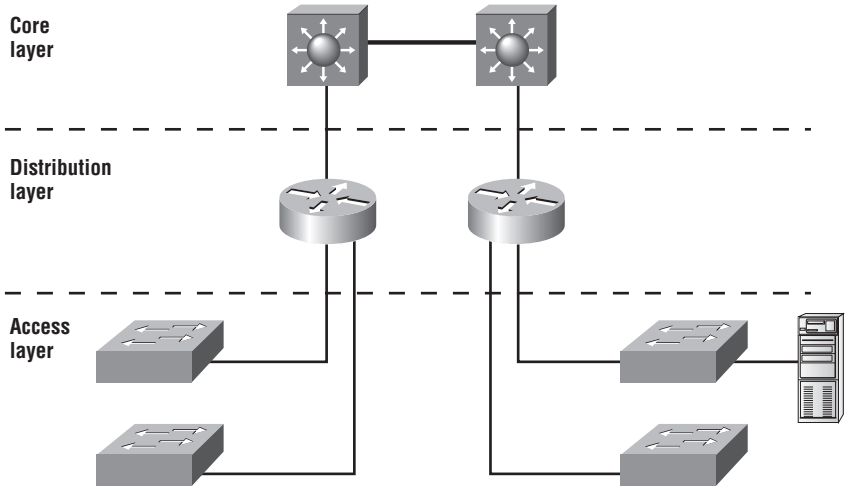
Most of us were exposed to hierarchy early in life. Anyone with older siblings learned what it was like to be at the bottom of the hierarchy. Regardless of where you first discovered hierarchy, today most of us experience it in many aspects of our lives. It is *hierarchy* that helps us understand where things belong, how things fit together, and what functions go where. It brings order and understandability to otherwise complex models. If you want a pay raise, for instance, hierarchy dictates that you ask your boss, not your subordinate. That is the person whose role it is to grant (or deny) your request. So basically, understanding hierarchy helps us discern where we should go to get what we need.

Hierarchy has many of the same benefits in network design that it does in other areas of life. When used properly, it makes networks more predictable. It helps us define which areas should perform certain functions. Likewise, you can use tools such as access lists at certain levels in hierarchical networks and avoid them at others.

Let's face it: Large networks can be extremely complicated, with multiple protocols, detailed configurations, and diverse technologies. Hierarchy helps us summarize a complex collection of details into an understandable model. Then, as specific configurations are needed, the model dictates the appropriate manner in which to apply them.

The Cisco hierarchical model can help you design, implement, and maintain a scalable, reliable, cost-effective hierarchical internetwork. Cisco defines three layers of hierarchy, as shown in Figure 1.31, each with specific functions.

FIGURE 1.31 The Cisco hierarchical model



The following are the three layers and their typical functions:

- The core layer: backbone
- The distribution layer: routing
- The access layer: switching

Each layer has specific responsibilities. Remember, however, that the three layers are logical and are not necessarily physical devices. Consider the OSI model, another logical hierarchy. The seven layers describe functions but not necessarily protocols, right? Sometimes a protocol maps to more than one layer of the OSI model, and sometimes multiple protocols communicate within a single layer. In the same way, when we build physical implementations of hierarchical networks, we may have many devices in a single layer, or we might have a single device performing functions at two layers. The definition of the layers is logical, not physical.

Now, let's take a closer look at each of the layers.

The Core Layer

The *core layer* is literally the core of the network. At the top of the hierarchy, the core layer is responsible for transporting large amounts of traffic both reliably and quickly. The only purpose of the network's core layer is to switch traffic as fast as possible. The traffic transported across the core is common to a majority of users. However, remember that user data is processed at the distribution layer, which forwards the requests to the core if needed.

If there is a failure in the core, *every single user* can be affected. Therefore, fault tolerance at this layer is an issue. The core is likely to see large volumes of traffic, so speed and latency are driving concerns here. Given the function of the core, we can now consider some design specifics. Let's start with some things we don't want to do:

- Don't do anything to slow down traffic. This includes using access lists, routing between virtual local area networks (VLANs), and implementing packet filtering.
- Don't support workgroup access here.
- Avoid expanding the core (i.e., adding routers) when the internetwork grows. If performance becomes an issue in the core, give preference to upgrades over expansion.

Now, there are a few things that we want to do as we design the core:

- Design the core for high reliability. Consider data-link technologies that facilitate both speed and redundancy, such as FDDI, Fast Ethernet (with redundant links), or even Asynchronous Transfer Mode (ATM).
- Design with speed in mind. The core should have very little latency.
- Select routing protocols with lower convergence times. Fast and redundant data-link connectivity is no help if your routing tables are shot!

The Distribution Layer

The *distribution layer* is sometimes referred to as the *workgroup layer* and is the communication point between the access layer and the core. The primary functions of the distribution layer are to provide routing, filtering, and WAN access and to determine how packets can access the core, if needed. The distribution layer must determine the fastest way that network service requests are handled—for example, how a file request is forwarded to a server. After the distribution layer determines the best path, it forwards the request to the core layer if necessary. The core layer then quickly transports the request to the correct service.

The distribution layer is the place to implement policies for the network. Here you can exercise considerable flexibility in defining network operation. There are several actions that generally should be done at the distribution layer:

- Routing
- Implementing tools (such as access lists), packet filtering, and queuing
- Implementing security and network policies, including address translation and firewalls
- Redistributing between routing protocols, including static routing

- Routing between VLANs and other workgroup support functions
- Defining broadcast and multicast domains

Things to avoid at the distribution layer are limited to those functions that exclusively belong to one of the other layers.

The Access Layer

The *access layer* controls user and workgroup access to internetwork resources. The access layer is sometimes referred to as the *desktop layer*. The network resources most users need will be available locally. The distribution layer handles any traffic for remote services. The following are some of the functions to be included at the access layer:

- Continued (from distribution layer) use of access control and policies
- Creation of separate collision domains (segmentation)
- Workgroup connectivity into the distribution layer

Technologies such as DDR and Ethernet switching are frequently seen in the access layer. Static routing (instead of dynamic routing protocols) is seen here as well.

As already noted, three separate levels does not imply three separate routers. There could be fewer, or there could be more. Remember, this is a *layered* approach.

Summary

Whew! I know this seemed like the chapter that wouldn't end, but it did—and you made it through! You're now armed with a ton of fundamental information; you're ready to build upon it and are well on your way to certification.

I started by discussing simple, basic networking and the differences between collision and broadcast domains. I also discussed the various devices used in an internetwork.

I then discussed the OSI model—the seven-layer model used to help application developers design applications that can run on any type of system or network. Each layer has its special jobs and select responsibilities within the model to ensure that solid, effective communications do, in fact, occur. I provided you with complete details of each layer and discussed how Cisco views the specifications of the OSI model.

In addition, each layer in the OSI model specifies different types of devices. I described the different devices, cables, and connectors used at each layer. Remember that hubs are Physical layer devices and repeat the digital signal to all segments except the one from which it was received. Switches segment the network using hardware addresses and break up collision domains. Routers break up broadcast domains (and collision domains) and use logical addressing to send packets through an internetwork.

Last, this chapter covered the Cisco three-layer hierarchical model. I described in detail the three layers and how each is used to help design and implement a Cisco internetwork. We are now going to move on to IP addressing in the next chapter.

Exam Essentials

Remember the possible causes of LAN traffic congestion. Too many hosts in a broadcast domain, broadcast storms, multicasting, and low bandwidth are all possible causes of LAN traffic congestion.

Understand the difference between a collision domain and a broadcast domain. Collision domain is an Ethernet term used to describe a network collection of devices in which one particular device sends a packet on a network segment, forcing every other device on that same segment to pay attention to it. On a broadcast domain, a set of all devices on a network segment hear all broadcasts sent on that segment.

Understand the difference between a hub, a bridge, a switch, and a router. Hubs create one collision domain and one broadcast domain. Bridges break up collision domains but create one large broadcast domain. They use hardware addresses to filter the network. Switches are really just multiple port bridges with more intelligence. They break up collision domains but create one large broadcast domain by default. Switches use hardware addresses to filter the network. Routers break up broadcast domains (and collision domains) and use logical addressing to filter the network.

Remember the difference between connection-oriented and connectionless network services. Connection-oriented services use acknowledgments and flow control to create a reliable session. More overhead is used than in a connectionless network service. Connectionless services are used to send data with no acknowledgments or flow control. This is considered unreliable.

Remember the OSI layers. You must remember the seven layers of the OSI model and what function each layer provides. The Application, Presentation, and Session layers are upper layers and are responsible for communicating from a user interface to an application. The Transport layer provides segmentation, sequencing, and virtual circuits. The Network layer provides logical network addressing and routing through an internetwork. The Data Link layer provides framing and placing of data on the network medium. The Physical layer is responsible for taking 1s and 0s and encoding them into a digital signal for transmission on the network segment.

Understand how a three-way handshake creates a virtual circuit. When a host starts a communication session to another host/server (when using TCP), a virtual circuit is created using three packets (hence the name three-way handshake). The transmitting host makes up a source port number from 1024 to 65535. The destination port number will be that of the process or application that data is destined for, like port 80 for example (HTTP).

Remember the types of Ethernet cabling and when you would use them. The three types of cables that can be created from an Ethernet cable are straight-through (to connect a PC's or a router's Ethernet interface to a hub or switch), crossover (to connect hub to hub, hub to switch, switch to switch, or PC to PC), and rolled (for a console connection from a PC to a router or switch).

Understand how to connect a console cable from a PC to a router and start HyperTerminal. Take a rolled cable and connect it from the COM port of the host to the console port of a router. Start HyperTerminal and set the BPS to 9600 and flow control to None.

Remember the three layers in the Cisco three-layer model. The three layers in the Cisco hierarchical model are the core, distribution, and access layers.

Written Lab 1

In this section, you'll complete the following labs to make sure you've got the information and concepts contained within them fully dialed in:

- Lab 1.1: OSI Questions
- Lab 1.2: Defining the OSI Layers and Devices
- Lab 1.3: Identifying Collision and Broadcast Domains
- Lab 1.4: Binary/Decimal/Hexadecimal Conversion

(The answers to the written labs can be found following the answers to the review questions for this chapter.)

Written Lab 1.1: OSI Questions

Answer the following questions about the OSI model:

1. Which layer chooses and determines the availability of communicating partners along with the resources necessary to make the connection, coordinates partnering applications, and forms a consensus on procedures for controlling data integrity and error recovery?
2. Which layer is responsible for converting data packets from the Data Link layer into electrical signals?
3. At which layer is routing implemented, enabling connections and path selection between two end systems?
4. Which layer defines how data is formatted, presented, encoded, and converted for use on the network?
5. Which layer is responsible for creating, managing, and terminating sessions between applications?
6. Which layer ensures the trustworthy transmission of data across a physical link and is primarily concerned with physical addressing, line discipline, network topology, error notification, ordered delivery of frames, and flow control?
7. Which layer is used for reliable communication between end nodes over the network and provides mechanisms for establishing, maintaining, and terminating virtual circuits; transport-fault detection and recovery; and controlling the flow of information?

8. Which layer provides logical addressing that routers will use for path determination?
9. Which layer specifies voltage, wire speed, and pinout cables and moves bits between devices?
10. Which layer combines bits into bytes and bytes into frames, uses MAC addressing, and provides error detection?
11. A Cisco router has received a frame on an interface that is connected to a local network segment. The router has de-encapsulated the frame. What step is next in processing the packet?
12. Which layer is represented by frames?
13. Which layer is represented by segments?
14. Which layer is represented by packets?
15. Which layer is represented by bits?
16. Put the following in order of encapsulation:
    - Packets
    - Frames
    - Bits
    - Segments
17. Which layer segments and reassembles data into a data stream?
18. Which layer provides the physical transmission of the data and handles error notification, network topology, and flow control?
19. Which layer manages device addressing, tracks the location of devices on the network, and determines the best way to move data?
20. What is the bit length and expression form of a MAC address?

Written Lab 1.2: Defining the OSI Layers and Devices

Fill in the blanks with the appropriate layer of the OSI or hub, switch, or router device.

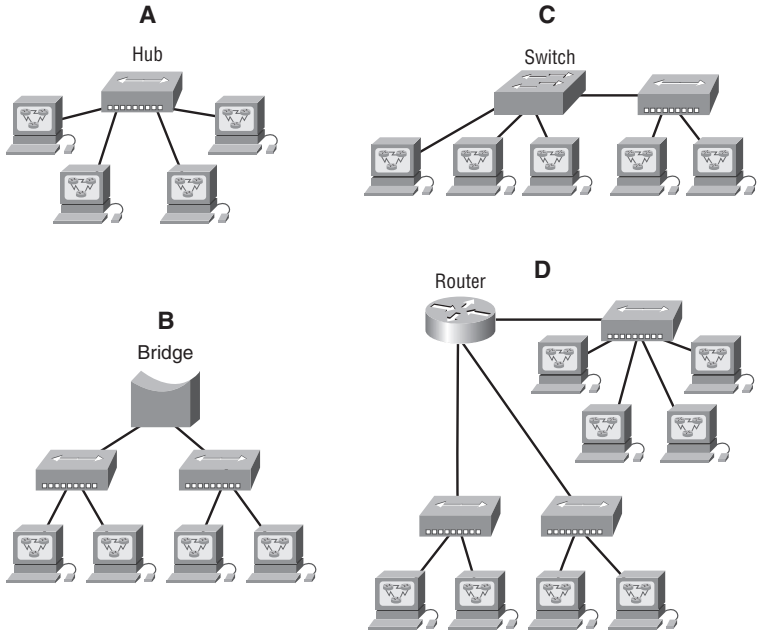
Description	Device or OSI Layer
This device sends and receives information about the Network layer.	
This layer creates a virtual circuit before transmitting between two end stations.	
This layer uses service access points.	
This device uses hardware addresses to filter a network.	
Ethernet is defined at these layers.	
This layer supports flow control and sequencing.	
This device can measure the distance to a remote network.	
Logical addressing is used at this layer.	

Description	Device or OSI Layer
Hardware addresses are defined at this layer.	
This device creates one big collision domain and one large broadcast domain.	
This device creates many smaller collision domains, but the network is still one large broadcast domain.	
This device can never run full duplex.	
This device breaks up collision domains and broadcast domains.	

Written Lab 1.3: Identifying Collision and Broadcast Domains

Using the information shown in the following illustration, identify the number of collision domains and broadcast domains in each specified device. Each device is represented by a letter:

1. Hub
2. Bridge
3. Switch
4. Router



Written Lab 1.4: Binary/Decimal/Hexadecimal Conversion

1. Convert from decimal IP address to binary format.

Complete the following table to express 192.168.10.15 in binary format.

128	64	32	16	8	4	2	1	Binary

Complete the following table to express 172.16.20.55 in binary format.

128	64	32	16	8	4	2	1	Binary

Complete the following table to express 10.11.12.99 in binary format.

128	64	32	16	8	4	2	1	Binary

2. Convert the following from binary format to decimal IP address.

Complete the following table to express 11001100.00110011.10101010.01010101 in decimal IP address format.

128	64	32	16	8	4	2	1	Decimal

Complete the following table to express 11000110.11010011.00111001.11010001 in decimal IP address format.

128	64	32	16	8	4	2	1	Decimal

Complete the following table to express 10000100.11010010.10111000.10100110 in decimal IP address format.

128	64	32	16	8	4	2	1	Decimal

3. Convert the following from binary format to hexadecimal.

Complete the following table to express 11011000.00011011.00111101.01110110 in hexadecimal.

128	64	32	16	8	4	2	1	Hexadecimal

Complete the following table to express 11001010.11110101.10000011.11101011 in hexadecimal.

128	64	32	16	8	4	2	1	Hexadecimal

Complete the following table to express 10000100.11010010.01000011.10110011 in hexadecimal.

128	64	32	16	8	4	2	1	Hexadecimal

Review Questions

The following questions are designed to test your understanding of this chapter's material. For more information on how to get additional questions, please see this book's Introduction.

1. A receiving host has failed to receive all of the segments that it should acknowledge. What can the host do to improve the reliability of this communication session?
   - **A.** Send a different source port number.
   - **B.** Restart the virtual circuit.
   - **C.** Decrease the sequence number.
   - **D.** Decrease the window size.
2. Which fields are contained within an IEEE Ethernet frame header? (Choose two.)
   - **A.** Source and destination MAC address
   - **B.** Source and destination network address
   - **C.** Source and destination MAC address and source and destination network address
   - **D.** FCS field
3. Which layer 1 devices can be used to enlarge the area covered by a single LAN segment? (Choose two.)
   - **A.** Switch
   - **B.** NIC
   - **C.** Hub
   - **D.** Repeater
   - **E.** RJ45 transceiver
4. Segmentation of a data stream happens at which layer of the OSI model?
   - **A.** Physical
   - **B.** Data Link
   - **C.** Network
   - **D.** Transport

5. Which of the following describe router functions? (Choose four.)
   - **A.** Packet switching
   - **B.** Collision prevention
   - **C.** Packet filtering
   - **D.** Broadcast domain enlargement
   - **E.** Internetwork communication
   - **F.** Broadcast forwarding
   - **G.** Path selection

6. Routers operate at layer __. LAN switches operate at layer __. Ethernet hubs operate at layer __. Word processing operates at layer __.
   - **A.** 3, 3, 1, 7
   - **B.** 3, 2, 1, none
   - **C.** 3, 2, 1, 7
   - **D.** 2, 3, 1, 7
   - **E.** 3, 3, 2, none

7. When data is encapsulated, which is the correct order?
   - **A.** Data, frame, packet, segment, bit
   - **B.** Segment, data, packet, frame, bit
   - **C.** Data, segment, packet, frame, bit
   - **D.** Data, segment, frame, packet, bit

8. Why does the data communication industry use the layered OSI reference model? (Choose two.)
   - **A.** It divides the network communication process into smaller and simpler components, thus aiding component development, design, and troubleshooting.
   - **B.** It enables equipment from different vendors to use the same electronic components, thus saving research and development funds.
   - **C.** It supports the evolution of multiple competing standards and thus provides business opportunities for equipment manufacturers.
   - **D.** It encourages industry standardization by defining what functions occur at each layer of the model.
   - **E.** It provides a framework by which changes in functionality in one layer require changes in other layers.

9. What are two purposes for segmentation with a bridge?
   - **A.** To add more broadcast domains
   - **B.** To create more collision domains
   - **C.** To add more bandwidth for users
   - **D.** To allow more broadcasts for users

10. Which of the following are unique characteristics of half-duplex Ethernet when compared to full-duplex Ethernet? (Choose two.)

A. Half-duplex Ethernet operates in a shared collision domain.

B. Half-duplex Ethernet operates in a private collision domain.

C. Half-duplex Ethernet has higher effective throughput.

D. Half-duplex Ethernet has lower effective throughput.

E. Half-duplex Ethernet operates in a private broadcast domain.

11. You want to implement an Ethernet network medium that is not susceptible to EMI or voltage potential differences between two buildings. Which type of cabling should you use?

A. Thicknet coax

B. Thinnet coax

C. Category 5 UTP cable

D. Fiber-optic cable

12. Acknowledgments, sequencing, and flow control are characteristic of which OSI layer?

A. Layer 2

B. Layer 3

C. Layer 4

D. Layer 7

13. Which of the following are types of flow control? (Choose all that apply.)

A. Buffering

B. Cut-through

C. Windowing

D. Congestion avoidance

E. VLANs

14. Which of the following types of connections can use full duplex? (Choose three.)

A. Hub to hub

B. Switch to switch

C. Host to host

D. Switch to hub

E. Switch to host

15. What is the purpose of flow control?

A. To ensure that data is retransmitted if an acknowledgment is not received

B. To reassemble segments in the correct order at the destination device

C. To provide a means for the receiver to govern the amount of data sent by the sender

D. To regulate the size of each segment

16. Which three statements are true about the operation of a full-duplex Ethernet network?

A. There are no collisions in full-duplex mode.

B. A dedicated switch port is required for each full-duplex node.

C. Ethernet hub ports are preconfigured for full-duplex mode.

D. In a full-duplex environment, the host network card must check for the availability of the network media before transmitting.

E. The host network card and the switch port must be capable of operating in full-duplex mode.

17. What type of RJ45 UTP cable is used between switches?

A. Straight-through

B. Crossover cable

C. Crossover with a CSU/DSU

D. Crossover with a router in between the two switches

18. How does a host on an Ethernet LAN know when to transmit after a collision has occurred? (Choose two.)

A. In a CSMA/CD collision domain, multiple stations can successfully transmit data simultaneously.

B. In a CSMA/CD collision domain, stations must wait until the media is not in use before transmitting.

C. You can improve the CSMA/CD network by adding more hubs.

D. After a collision, the station that detected the collision has first priority to resend the lost data.

E. After a collision, all stations run a random backoff algorithm. When the backoff delay period has expired, all stations have equal priority to transmit data.

F. After a collision, all stations involved run an identical backoff algorithm and then synchronize with each other prior to transmitting data.

19. What type of RJ45 UTP cable do you use to connect a PC's COM port to a router or switch console port?

A. Straight-through

B. Crossover cable

C. Crossover with a CSU/DSU

D. Rolled

20. You have the following binary number:

10110111

What are the decimal and hexadecimal equivalents?

A. 69/0x2102

B. 183/B7

C. 173/A6

D. 83/0xC5

Answers to Review Questions

1. D. A receiving host can control the transmitter by using flow control (TCP uses Windowing by default). By decreasing the window size, the receiving host can slow down the transmitting host so the receiving host does not overflow its buffers.

2. A, D. An Ethernet frame has source and destination MAC addresses, an Ether-Type field to identify the Network layer protocol, the data, and the FCS field that holds the answer to the CRC.

3. C, D. Not that you really want to enlarge a single collision domain, but a hub (multiport repeater) will provide this for you.

4. D. The Transport layer receives large data streams from the upper layers and breaks them up into smaller pieces called segments.

5. A, C, E, G. Routers provide packet switching, packet filtering, internetwork communication, and path selection.

6. B. Routers operate at layer 3. LAN switches operate at layer 2. Ethernet hubs operate at layer 1. Word processing applications communicate to the Application layer interface, but do not operate at layer 7, so the answer would be none.

7. C. The encapsulation method is data, segment, packet, frame, bit.

8. A, D. The main advantage of a layered model is that it can allow application developers to change aspects of a program in just one layer of the layer model's specifications. Advantages of using the OSI layered model include, but are not limited to, the following: It divides the network communication process into smaller and simpler components, thus aiding component development, design, and troubleshooting; it allows multiple-vendor development through standardization of network components; it encourages industry standardization by defining what functions occur at each layer of the model; it allows various types of network hardware and software to communicate; and it prevents changes in one layer from affecting other layers, so it does not hamper development.

9. B, C. Bridges break up collision domains, which allow more bandwidth for users.

10. A, D. Unlike full duplex, half-duplex Ethernet operates in a shared collision domain, and it has a lower effective throughput than full duplex.

11. D. Fiber-optic cable provides a more secure, long-distance cable that is not susceptible to EMI interference at high speeds.

12. C. A reliable Transport layer connection uses acknowledgments to make sure all data is transmitted and received reliably. A reliable connection is defined by a virtual circuit that uses acknowledgments, sequencing, and flow control, which are characteristics of the Transport layer (layer 4).

13. A, C, D. The common types of flow control are buffering, windowing, and congestion avoidance.

14. B, C, E. Hubs cannot run full-duplex Ethernet. Full duplex must be used on a point-to-point connection between two devices capable of running full duplex. Switches and hosts can run full duplex between each other, but a hub can never run full duplex.

15. C. Flow control allows the receiving device to control the transmitter so the receiving device's buffer does not overflow.

16. A, B, E. Full-duplex means you are using both wire pairs simultaneously to send and receive data. You must have a dedicated switch port for each node, which means you will not have collisions. Both the host network card and the switch port must be capable and set to work in full-duplex mode.

17. B. To connect two switches together, you would use a RJ45 UTP crossover cable.

18. B, E. Once transmitting stations on an Ethernet segment hear a collision, they send an extended jam signal to ensure that all stations recognize the collision. After the jamming is complete, each sender waits a predetermined amount of time, plus a random time. After both timers expire, the senders are free to transmit, but they must make sure the media is clear before transmitting and that they all have equal priority.

19. D. To connect to a router or switch console port, you would use an RJ45 UTP rolled cable.

20. B. You must be able to take a binary number and convert it into both decimal and hexadecimal. To convert to decimal, just add up the 1s using their values. The values that are turned on with the binary number of 10110111 are 128 + 32 + 16 + 4 + 2 + 1 = 183. To get the hexadecimal equivalent, you need to break the eight binary digits into nibbles (4 bits), 1011 and 0111. By adding up these values, you get 11 and 7. In hexadecimal, 11 is *B*, so the answer is 0xB7.

Answers to Written Lab 1

1. The Application layer is responsible for finding the network resources broadcast from a server and adding flow control and error control (if the application developer chooses).
2. The Physical layer takes frames from the Data Link layer and encodes the 1s and 0s into a digital signal for transmission on the network medium.
3. The Network layer provides routing through an internetwork and logical addressing.
4. The Presentation layer makes sure that data is in a readable format for the Application layer.
5. The Session layer sets up, maintains, and terminates sessions between applications.
6. PDUs at the Data Link layer are called frames. As soon as you see *frame* in a question, you know the answer.
7. The Transport layer uses virtual circuits to create a reliable connection between two hosts.
8. The Network layer provides logical addressing, typically IP addressing and routing.
9. The Physical layer is responsible for the electrical and mechanical connections between devices.
10. The Data Link layer is responsible for the framing of data packets.
11. The router searches the routing table to determine where to forward the packet.
12. The Data Link layer frames packets received from the Network layer.
13. The Transport layer segments user data.
14. The Network layer creates packets out of segments handed down from the Transport layer.
15. The Physical layer is responsible for transporting 1s and 0s in a digital signal.
16. Segments, packets, frames, bits
17. Transport
18. Data Link
19. Network
20. 48 bits (6 bytes) expressed as a hexadecimal number

Answer to Written Lab 1.2

Description	Device or OSI Layer
This device sends and receives information about the Network layer.	Router
This layer creates a virtual circuit before transmitting between two end stations.	Transport
This layer uses service access points.	Data Link (LLC sublayer)
This device uses hardware addresses to filter a network.	Bridge or switch
Ethernet is defined at these layers.	Data Link and Physical
This layer supports flow control and sequencing.	Transport
This device can measure the distance to a remote network.	Router
Logical addressing is used at this layer.	Network
Hardware addresses are defined at this layer.	Data Link (MAC sublayer)
This device creates one big collision domain and one large broadcast domain.	Hub
This device creates many smaller collision domains, but the network is still one large broadcast domain.	Switch or bridge
This device can never run full duplex.	Hub
This device breaks up collision domains and broadcast domains.	Router

Answers to Written Lab 1.3

1. Hub: One collision domain, one broadcast domain
2. Bridge: Two collision domains, one broadcast domain
3. Switch: Four collision domains, one broadcast domain
4. Router: Three collision domains, three broadcast domains

Answers to Written Lab 1.4

1. Convert from decimal IP address to binary format.

Complete the following table to express 192.168.10.15 in binary format.

Decimal	128	64	32	16	8	4	2	1	Binary
192	1	1	0	0	0	0	0	0	11000000
168	1	0	1	0	1	0	0	0	10101000
10	0	0	0	0	1	0	1	0	00001010
15	0	0	0	0	1	1	1	1	00001111

Complete the following table to express 172.16.20.55 in binary format.

Decimal	128	64	32	16	8	4	2	1	Binary
172	1	0	1	0	1	1	0	0	10101100
16	0	0	0	1	0	0	0	0	00010000
20	0	0	0	1	0	1	0	0	00010100
55	0	0	1	1	0	1	1	1	00110111

Complete the following table to express 10.11.12.99 in binary format.

Decimal	128	64	32	16	8	4	2	1	Binary
10	0	0	0	0	1	0	1	0	00001010
11	0	0	0	0	1	0	1	1	00001011
12	0	0	0	0	1	1	0	0	00001100
99	0	1	1	0	0	0	1	1	01100011

2. Convert the following from binary format to decimal IP address.

Complete the following table to express 11001100.00110011.10101010.01010101 in decimal IP address format.

Binary	128	64	32	16	8	4	2	1	Decimal
11001100	1	1	0	0	1	1	0	0	204
00110011	0	0	1	1	0	0	1	1	51
10101010	1	0	1	0	1	0	1	0	170
01010101	0	1	0	1	0	1	0	1	85

Complete the following table to express 11000110.11010011.00111001.11010001 in decimal IP address format.

Binary	128	64	32	16	8	4	2	1	Decimal
11000110	1	1	0	0	0	1	1	0	198
11010011	1	1	0	1	0	0	1	1	211
00111001	0	0	1	1	1	0	0	1	57
11010001	1	1	0	1	0	0	0	1	209

Complete the following table to express 10000100.11010010.10111000.10100110 in decimal IP address format.

Binary	128	64	32	16	8	4	2	1	Decimal
10000100	1	0	0	0	0	1	0	0	132
11010010	1	1	0	1	0	0	1	0	210
10111000	1	0	1	1	1	0	0	0	184
10100110	1	0	1	0	0	1	1	0	166

3. Convert the following from binary format to hexadecimal.

Complete the following table to express 11011000.00011011.00111101.01110110 in hexadecimal.

Binary	128	64	32	16	8	4	2	1	Hexadecimal
11011000	1	1	0	1	1	0	0	0	D8
00011011	0	0	0	1	1	0	1	1	1B
00111101	0	0	1	1	1	1	0	1	3D
01110110	0	1	1	1	0	1	1	0	76

Complete the following table to express 11001010.11110101.10000011.11101011 in hexadecimal.

Binary	128	64	32	16	8	4	2	1	Hexadecimal
11001010	1	1	0	0	1	0	1	0	CA
11110101	1	1	1	1	0	1	0	1	F5

10000011	1	0	0	0	0	0	1	1	83
11101011	1	1	1	0	1	0	1	1	EB

Complete the following table to express 10000100.11010010.01000011.10110011 in hexadecimal.

Binary	128	64	32	16	8	4	2	1	Hexadecimal
10000100	1	0	0	0	0	1	0	0	84
11010010	1	1	0	1	0	1	1	0	D2
01000011	0	1	0	0	0	0	1	1	43
10110011	1	0	1	1	0	0	1	1	B3