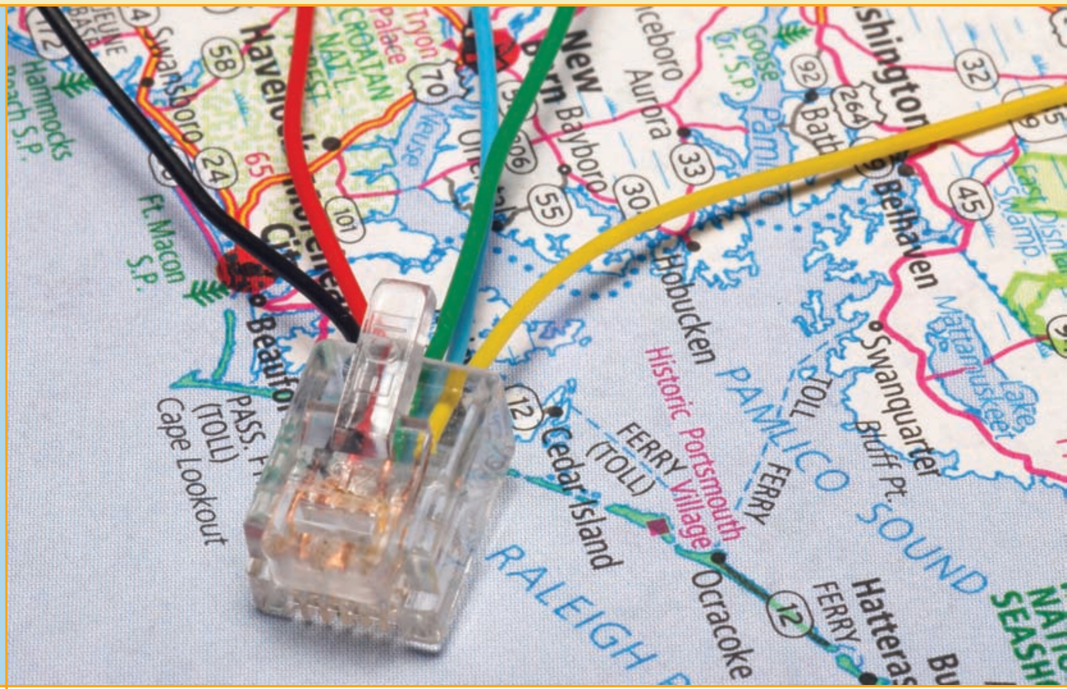
chapter 14

Remote Connectivity

"Gongs and drums, banners and flags, are means whereby the ears and eyes of the host may be focused on one particular point."

—SUN TZU



In this chapter, you will learn how to

- **Describe WAN telephony technologies, such as SONET, T1, and T3**
- **Compare last-mile connections for connecting homes and businesses to the Internet**
- **Discuss and implement various remote access connections**

Simply, the term "remote connection" describes how we get two computers that are too far away to connect using a classic LAN. Broadly, remote connections are much more complex. To break this down, we need to take both a historical and a modern look at how we interconnect distant computers.

Historical/Conceptual

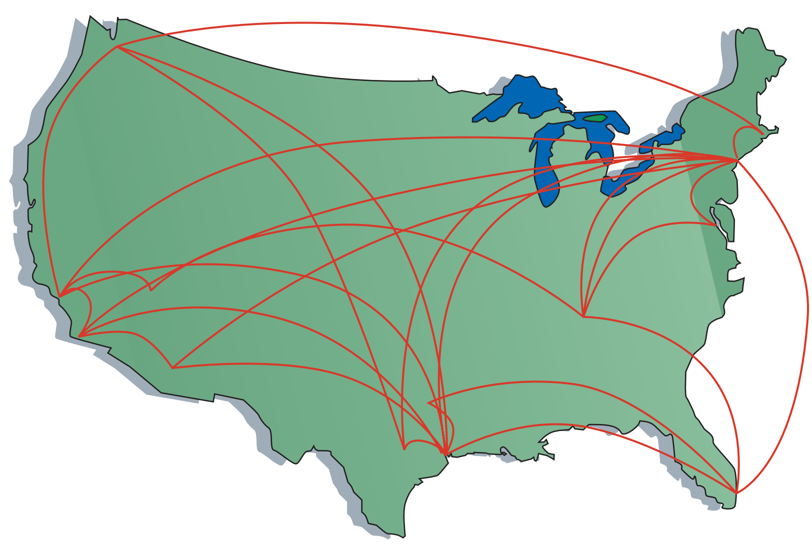
You must appreciate that remote connections have been around for a long time. Before the Internet, network users and developers developed ways to take a single system or network and connect it to another faraway system or network. This wasn't the Internet! These were private interconnections of private networks. These connections were very expensive and, compared to today's options, pretty slow.

As the Internet developed, most of the same technologies used to make the earlier private remote connections became the way the Internet itself interconnects. The days of private interconnections are quickly fading: today the ultimate remote connection is nothing more than you opening your Web browser and accessing someone's faraway Web page. However, there are hundreds of thousands of organizations that still use their own private interconnections (which may or may not also connect to the Internet) for security or speed.

This chapter shows you all the ways you can make remote connections. You'll see every type of remote connection currently in popular use, from good old telephone lines to advanced fiber-optic carriers, and even satellites. There are so many ways to make remote connections that this chapter is broken into three parts. The first part, "Telephony and Beyond," gives you a tour of the technologies that originally existed for long-distance voice connections that now also support data. The next part, "The Last Mile," goes into how we as individual users connect to those long-distance technologies. Last, "Using Remote Access" shows you the many different ways to use these connections to actually connect to another, faraway computer.

Telephony and Beyond

We've already discussed the tier-one ISPs of the Internet but let's look at them once again in a different way. Describing the tier-one Internet is always an interesting topic. Those of us in the instruction business invariably start this description by drawing a picture of the United States (we Americans are so conceited sometimes) and then adding lines connecting big cities, as shown in Figure 14.1.



• Figure 14.1 The tier-one Internet

But what are these lines and where did they come from? If the Internet is just a big TCP/IP network, wouldn't one assume that these lines are Ethernet connections? Maybe copper, maybe fiber, but Ethernet? Well, traditionally they're not (with one exception; see the following Note). The vast majority of the long-distance connections that make up the Internet uses an unique type of signal called SONET. SONET was originally designed to handle special heavy-duty circuits with names like T1. Never heard of SONET or T1? Don't worry—you're about to learn quite a bit.

The majority of the high-speed backbone of the Internet uses technologies designed at least 20 years ago to support telephone calls. We're not talking about your cool, cell phone–type calls here: we're talking the old-school, wire-runs-up-to-the-house, telephone-connected-to-a-phone-jack connections known lovingly as **Plain Old Telephone Service (POTS)**. If you want to understand how the Internet connects, you have to go way back to the 1970s and 1980s, before the Internet really took off, and learn how the U.S. telephone system developed to support networks.

Even as you read this, more and more of the Internet interconnections are moving toward 1-Gigabit and 10-Gigabit Ethernet. However, telephone technologies continue to dominate.

Tech Tip

Telephony in Depth

This section is just the lightest of overviews to get you through the CompTIA Network+ exam. The full history of long-distance communication is an incredible story, full of good guys, bad guys, crazy technology, and huge fortunes won and lost.

The Dawn of Long Distance

Have you ever watched one of those old-time movies in which someone makes a phone call by picking up the phone and saying "Operator, get me Mohawk 4, 3-8-2-5!" Suddenly, the scene changes to some person sitting at a switchboard like the one shown in Figure 14.2.

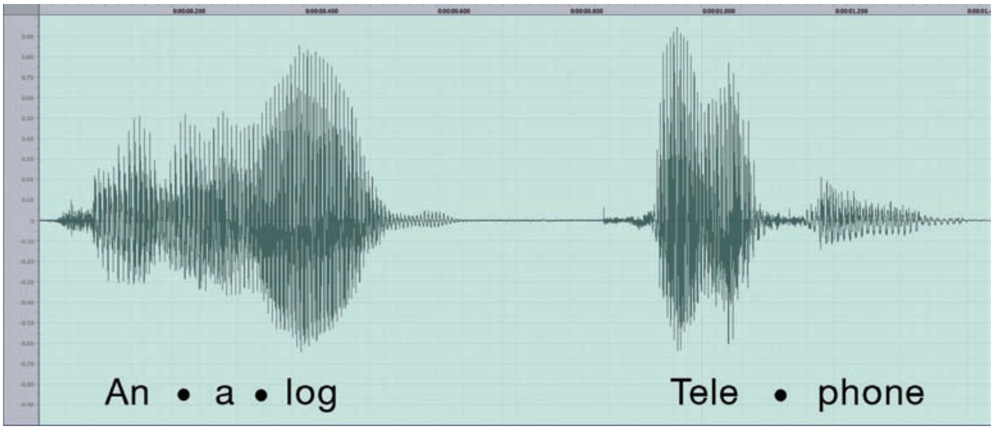
This was the telephone operator. The telephone operator made a physical link between your phone and the other phone, making your connection. This worked pretty well in the first few years of telephones but it quickly became a problem as more and more phone lines began to fill the skies overhead (Figure 14.3).

These first generations of long-distance telephone systems (think 1930s here) used analog signals because that was how your telephone worked—the louder you spoke, the greater the voltage. If you graphed out a voice signal, it looked something like Figure 14.4. This was a problem, because analog signals over long distances, even if you amplified them, lost sound quality very quickly.

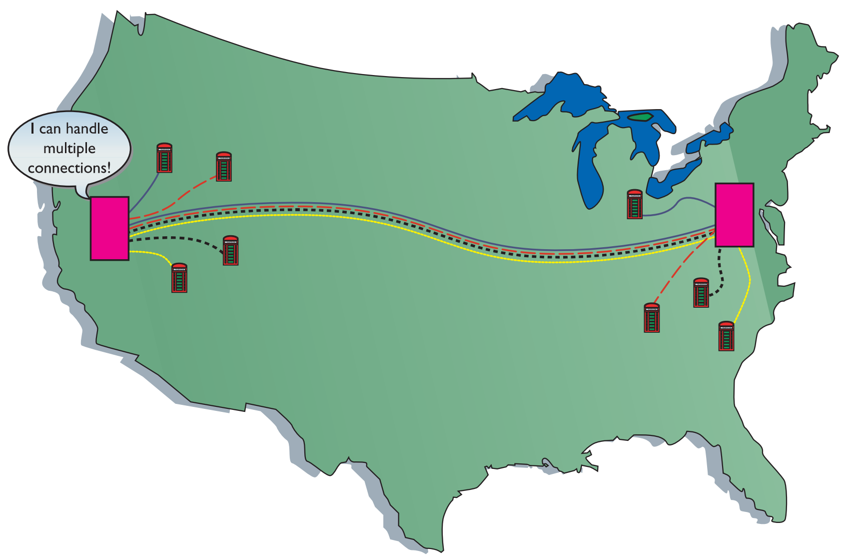
The first problem to take care of was the number of telephone wires. Individual wires were slowly replaced with special boxes called multiplexers. A **multiplexer** took a circuit and combined it with a few hundred other circuits into a single complex circuit on one wire. A second multiplexer on the other end of the connection split the individual connections back out (Figure 14.5).

• **Figure 14.2** Old-time telephone operator (photo courtesy of the Richardson Historical and Genealogical Society)

• **Figure 14.3** Now that's a lot of telephone lines!

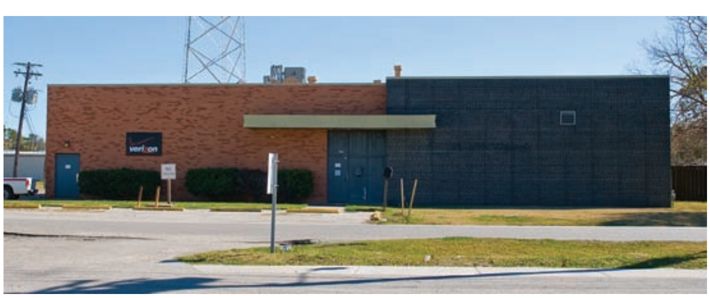


• **Figure 14.4** Another problem of early long-distance telephone systems

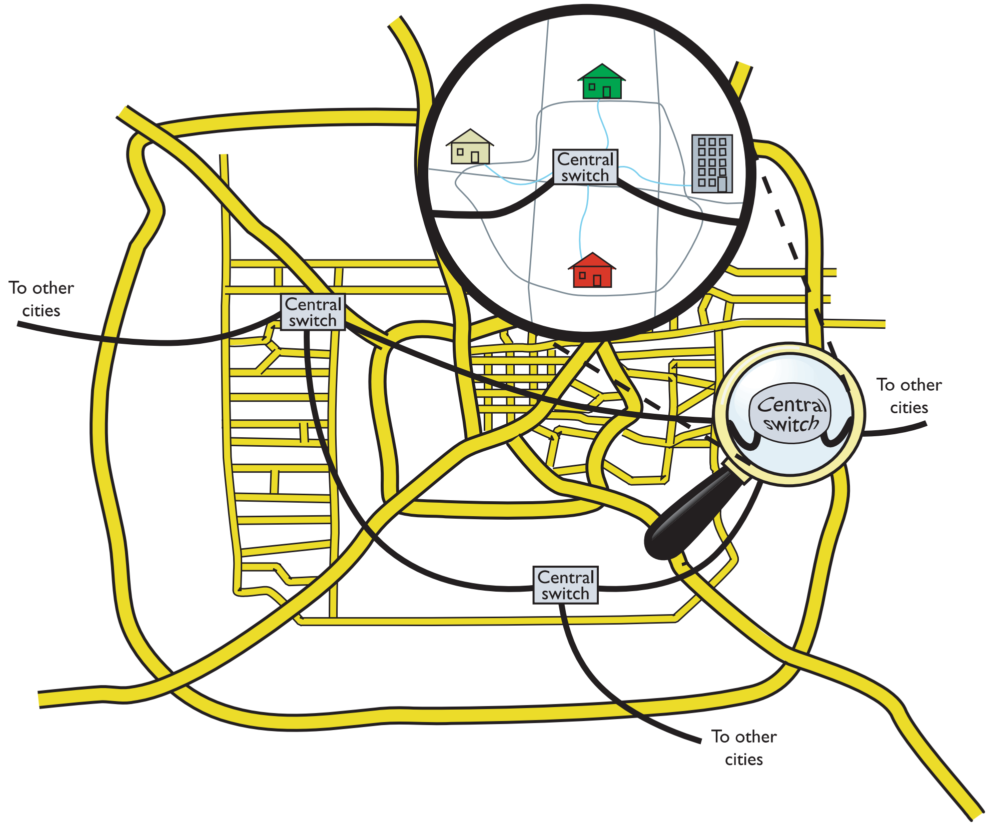


• **Figure 14.5** Multiplexers combine multiple circuits.

Over time, the entire United States was divided into hundreds, eventually thousands, of local exchanges. Local exchanges were a defined grouping of individual phone circuits served by a single multiplexer (calls within the exchange were handled differently). One or more exchanges were (and still are) housed in a physical building called a **central office** (Figure 14.6) where individual voice circuits all came together. Local calls were still manually connected (although dial-up began to appear in earnest by the 1950s, after which a lot of operators lost their jobs) but any connection between exchanges was carried over these special multiplexed trunk lines. Figure 14.7 shows a very stylized example of how this worked.

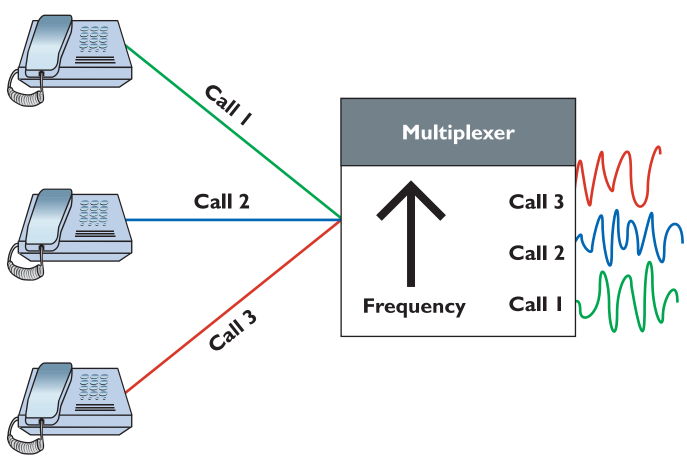


• **Figure 14.6** A central office building



• Figure 14.7 Interconnected central offices

These old-style trunk lines were fascinating technology. How did they put a bunch of voice calls on a single piece of cable yet still somehow keep them separate? The trick was appreciating a little bit about frequency. Your typical telephone only detects a fairly limited frequency range—from around 350 Hz to around 4000 Hz. This range covers enough of the human speech range to make a decent phone call. As the individual calls came into the multiplexer, it added a certain frequency multiplier to each call, keeping every separate call in its own unique frequency range (Figure 14.8). We call this process **Frequency Division Multiplexing (FDM)**.



• Figure 14.8 Multiplexed FDM

This analog network still required a physical connection from one phone to the other, even if those phones were on opposite sides of the country. Long distance was a series of trunk lines, and at each intersection of those lines an operator had to connect the calls. When you physically connect two phones together on one circuit, you are using something called **circuit switching**. As you might imagine, circuit switching isn't that great for long distance, but it's your only option when you use analog.

This analog system worked pretty well through the 1930s to the 1950s but telephones became so common and demand so heavy that the United States needed a new system to handle the heavy load. The folks developing this new system realized that they had to dump analog and replace it with a digital system—and here is where the remote connections that eventually became the Internet were born.

Digital data is much easier to carry long distances than analog data because you can use repeaters (you cannot use repeaters on analog signals). If you remember from earlier chapters, a repeater is not an amplifier. An amplifier just increases the voltage and includes all the pops and hisses created by all kinds of interferences. A repeater takes the entire digital signal and re-creates it out the other end (Figure 14.9).

The downside to digital was that all voice calls were analog. Something, somewhere had to convert the analog voice calls to digital. This was a massive undertaking but luckily virtually the entire U.S. phone system at that time was a monopoly run by a company called AT&T. A single company could make all of its own decisions and its own standards—one of the few times in history where a monopoly was probably a good thing. The AT&T folks had a choice here: completely revamp the entire U.S. phone system, including replacing every single telephone in the United States, or just make the trunk lines digital and let the central offices convert from analog to digital. They chose the latter. Even today, a classic POTS line in your home or small office is analog—the rest of the entire telephone system is digital. This is not unique to the United States; POTS is analog, everything else is digital. The telecommunications industry calls the connection from a central office to individual users the **last mile**. The telephone company's decision to keep the last mile analog has had serious repercussions that still challenge us even in the 21st century (Figure 14.10)

Tech Tip

The Same Lines

The long-distance lines used for voice calls are the same ones that carry the Internet. There is no difference as far as the carriers are concerned.

There were attempts to convert the entire telephone system, including your telephones, to digital but they never really took off. See "ISDN" later in this chapter.

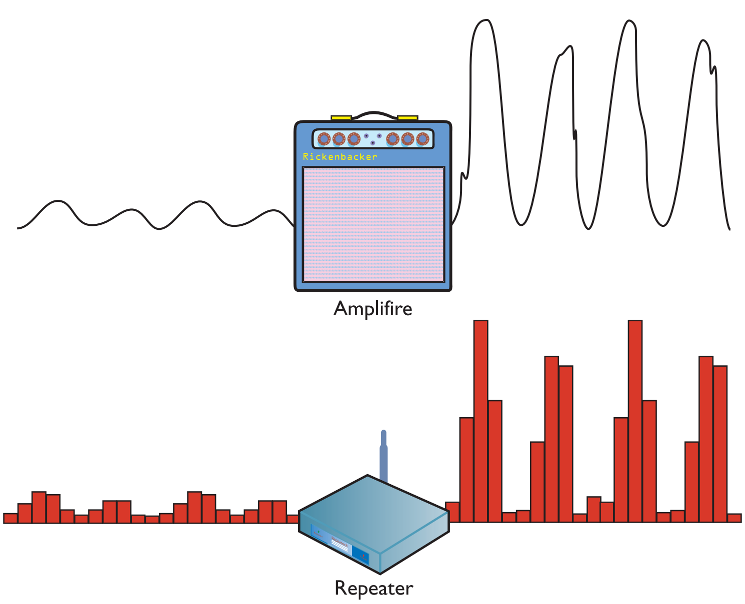
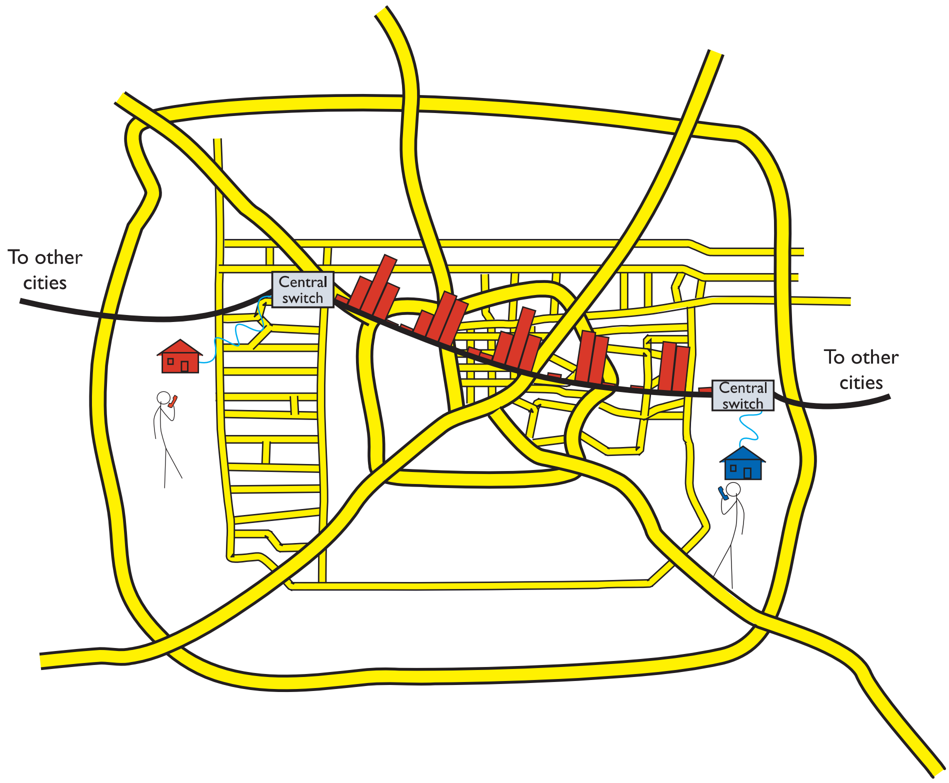


Figure 14.9 Repeater vs. amplifier



• Figure 14.10 Analog and digital

Test Specific

Digital Telephony

The best part about digital telephony is that most of the aspects you've already learned about computer networking work roughly the same way in a telephone network. In fact, most of the concepts that created computer networking came from the telephone industry. For example, the telephone industry was the first technology to heavily adopt the idea of digital packets. It was the first to do what we now call switching. Heck, the telephone industry even made the first working topologies! Let's take advantage of what you already know about how networks work to learn about how the telephone industry invented the idea of digital networks.

When you learned about networks in the first few chapters of this book, you learned about cabling, frame types, speeds, switching, and so on. All of these are important for computer networks. Well, let's do it again (in a much simpler format) to see the cabling, frame types, speed, and switching used in telephone systems. Don't worry, unlike computer networks, in which a certain type of cable might run different types of frames at different speeds, most of the remote connections used in the telephony world tend to have one type of cable that only runs one type of frame at one type of speed.

The only real question is where to start. The best place to begin is with the most basic data chunk you get in the telephone world: DS0.

It All Starts with DS0

When AT&T decided to go digital, it knew all phone calls had to be broken into a digital sample. AT&T chose (some people say "guessed" or "compromised") and decided that if it took an analog signal of a human voice and converted it into 8-bit chunks 8000 times a second, that would be good enough to re-create the sound later. Figure 14.11 shows an example of the analog human voice seen earlier being converted into a digital sample.

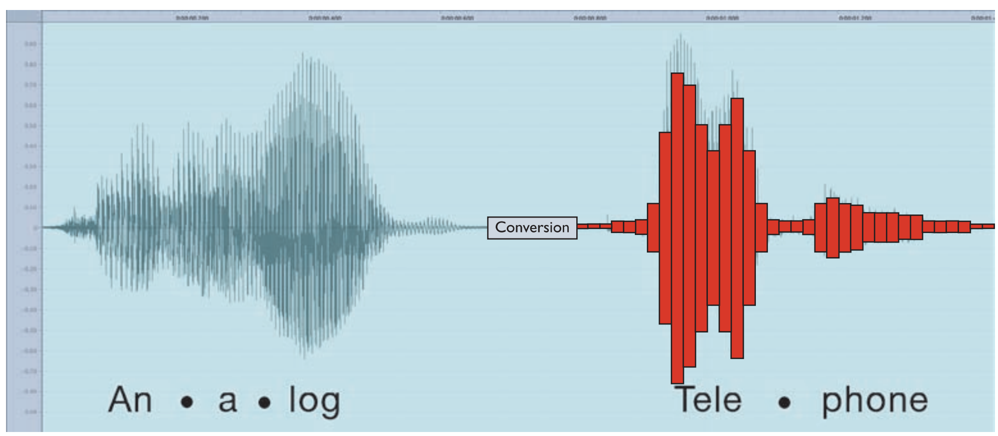
Converting analog sound into 8-bit chunks 8000 times a second creates a data stream (called a digital signal) of 64 kilobits per second (Kbps). This digital signal rate is known as **DS0** and is the simplest data stream (and the slowest rate) of the digital part of the telephone system. Each analog voice call is converted into a DS0 signal at the telephone company's central office. From there they are multiplexed into larger circuits.

Now that we have our voice calls converted to digital data, we need to get them to the right telephone. First, we need network technologies to handle the cabling, frames, and speed. Second, we need to come up with a method to switch the digital voice calls across a network. To handle the former, we need to define the types of interconnections, with names like T1 and OC3. To handle the latter, we no longer connect via multiplexed circuit switching, as we did back with analog, but rather are now switching packets. I'll show you what I mean as we learn about the digital lines in use today.

Tech Tip

Modems

A device that takes an analog signal and converts it into a digital signal is a modulator. A device that takes a digital signal and converts it into an analog signal is a demodulator. A device that does both is called a modulator-demod*ulator, better known as a modem.*



• **Figure 14.11** Analog to digital

Copper Carriers: T1 and T3

The first and still popular digital trunk carriers used by the telephone industry are called T-carriers. There are a number of different versions of T-carriers and the CompTIA Network+ exam expects you to know something about them. Let's begin with the most common and most basic, the venerable T1.

T1 has several meanings. First, it refers to a high-speed digital networking technology called a T1 connection. Second, the term **T1 line** refers to the specific, shielded, two-pair cabling that connects the two ends of a T1 connection (Figure 14.12). Two wires are for sending data and two wires are for receiving data. At either end of a T1 line you'll find an unassuming box called a **Channel Service Unit/Digital Service Unit (CSU/DSU)**. The CSU/DSU

• Figure 14.12 T1 line

Tech Tip

Numbers and Units in WAN Technologies

Those of you with good math skills might immediately question the accuracy of the numbers involved in describing T1 speeds. After all, 193 bits×8000 seconds = 1,544,000 bits per second, or 1.544 million bits per second. You normally wouldn't use the abbreviation Mbps for million *bits per second, right? Sadly, that's exactly the case here.*

When reading the signaling speeds regarding T1 and other WAN technologies (such as SONET, discussed in the following section), think "millions" and "billions" of bits per second when you see Mbps and Gbps!

has a second connection that goes from the phone company (where the boxes reside) to a customer's equipment (usually a router). A T1 connection is point-to-point—you cannot have more than two CSU/DSUs on a single T1 line.

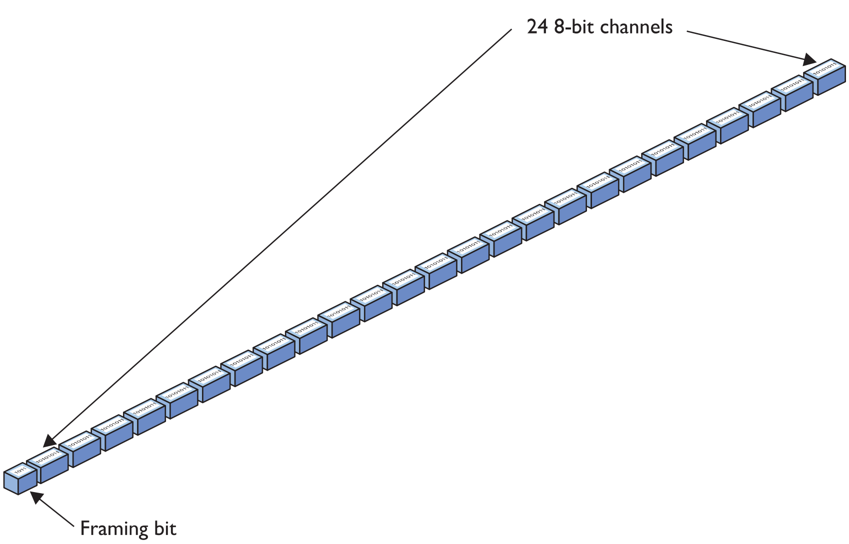
T1 uses a special signaling method called **DS1**. DS1 uses a relatively primitive frame—the frame doesn't need to be complex because with point-to-point there's no addressing necessary. Each DS1 frame has 25 pieces: a framing bit and 24 channels. Each DS1 channel holds a single 8-bit DS0 data sample. The framing bit and data channels combine to make 193 bits per DS1 frame. These frames are transmitted 8000 times/sec, making a total throughput of 1.544 Mbps (Figure 14.13). DS1 defines, therefore, a data transfer speed of 1.544 Mbps, split into 24 64-Kbps DS0 channels. This process of having frames that carry a bit of every channel in every frame sent on a regular interval is called **time division multiplexing**.

An analogy I like to use in class for T1 technology is that of a conveyor belt in a milk-bottling factory. At regular intervals, big crates with 24 bottles come rolling down the belt. When they reach the filling machine, the bottles get filled with milk and the crate keeps rolling down to the other end where two machines take over: the labeling and sorting machines. The labeling machine plucks out the bottles and applies a label to each, appropriate to the contents. The sorting machine sorts the bottles into cases of each type.

This is pretty simple if the filling machine uses only one type of milk. All 24 bottles fill with whole milk; all are labeled as whole milk; and all go into the case marked "Whole Milk." Once enough full bottles of milk arrive, the case gets completed, and you have a product.

That's pretty much how an Ethernet packet works, right? The whole packet is used for a single set of data, and then multiple packets get put together at the end to make your data transfer complete.

The cool thing about the DS1 frame, though, is that you don't have to use the whole frame for a single set of data. With the right CSU/DSU at either end, you can specify which channels go with a specific thread of data. Sloshing back into the analogy . . . the milk company produces four types of milk: whole milk, lowfat milk, chocolate milk, and strawberry milk. The strawberry milk is seasonal; the whole milk sells the most, followed by chocolate, and then lowfat.



• Figure 14.13 DS1 frame

To accommodate the different products, the factory master might designate channels 1–10 for whole milk, 11–18 for chocolate milk, 19–22 for lowfat milk, and 23–24 for strawberry. Now the labeling and sorting machines are going to have to work for a living! When a crate reaches the filling machine, the bottles get filled with the various types of milk, and then the crate trundles on down the belt. The labeling machine knows the numbering system, so it labels bottles 1–10 as whole milk, 11–18 as chocolate, and so on. The sorting machine also knows the system and has four cases at hand, one for each product. As the bottles arrive, it places them into the appropriate cases. Notice that the cases will fill at different rates of speed. It'll take a while for the strawberry milk case to fill, especially compared to the whole milk, because only two channels in each crate carry strawberry.

What happens if the cows temporarily stop producing chocolate milk? Will the whole factory need to be reordered so the filling machine's eight chocolate dispensers can dispense some other kind of milk? The answer at this factory is no. The crates continue to roll down the conveyor belt at regular intervals. The filling machine fills the bottles in channels 1–10 with whole milk, leaves the bottles in channels 11–18 empty, and puts lowfat and strawberry in channels 19–22 and 23–24, respectively.

Tech Tip

DS1 Gets No Respect!

People rarely use the term "DS1." Since T1 lines only carry DS1 signals, we usually just say T1 when describing the signal, even though the term DS1 is more accurate.

DS1/T1 work the same way. The frame just keeps jetting down the line, even if some of the channels contain no data. The CSU/DSU at the other end collects the data streams and keeps them separate. To paraphrase the immortal words of Professor Egon, "Never cross the streams." Otherwise you'd lose data.

To bring the milk bottling factory analogy completely into the realm of networking and T1 connections, keep in mind that there would be two conveyor belts running in opposite directions. Milk flows in; milk flows out. You can both send and receive on T1 connections.

Each 64-Kbps channel in a DS1 signal is a DS0.

A T1 line is a dedicated phone connection that you lease, usually on a monthly basis, from the telephone company. It has no telephone number and it's always connected. An entire T1 bundle can be expensive, so many telephone companies let you buy just some of these individual channels. This is known as **fractional T1 access**.

A **T3 line** is a dedicated telephone connection supporting a data rate of about 43 Mbps. A T3 line consists of 672 individual DS0 channels. T3 lines (sometimes referred to as DS3 lines) are mainly used by regional telephone companies and ISPs connecting to the Internet.

Similar to the North American T1 line, **E1** is the European format for digital transmission. An E1 line carries signals at 2.048 Mbps (32 channels at 64 Kbps), compared to the T1's 1.544 Mbps (24 channels at 64 Kbps). Both E1 and T1 lines may be interconnected for international use. There are also **E3** lines, which carry 16 E1 lines, with a bandwidth of 34.368 Mbps.

A CSU/DSU, as mentioned earlier, connects a leased T1 or T3 line from the telephone company to a customer's equipment. A CSU/DSU has (at least) two connectors, one that goes to the T1/T3 line running out of your demarc and another connection that goes to your router. It performs line encoding and conditioning functions, and often has a loopback function for testing. Many newer routers have CSU/DSUs built into them. Figure 14.14 shows the front of a Juniper Networks router with two T1 interfaces. Two interfaces on one router is quite common, with the dual links providing redundancy if one link goes down.

• **Figure 14.14** CSU/DSU on a Juniper router (photo courtesy of Juniper Networks, Inc.)

The CSU part of a CSU/DSU is designed to protect the T1 or T3 line and the user equipment from lightning strikes and other types of electrical interference. It also stores statistics and has capabilities for loopback testing.

Cross Check

Demarc

You first read about the demarc—the spot where connections from the outside world come into a building—way back in Chapter 6, "Installing a Physical Network," so check your memory and see if you can answer these questions. How does the demarc affect your wallet? What do you call the cable modem or DSL receiver that marks the demarc in many houses and offices?

The DSU part supplies timing to each user port, taking the incoming user data signals and converting the input signal into the specified line code, and then framing the format for transmission over the provided line.

Make sure you know the four T-carriers shown in Table 14.1!

Table 14.1 T-carriers

Carrier	Channels	Speed
T1	24	1.544 Mbps
T3	672	44.736 Mbps
E1	32	2.048 Mbps
E3	512	34.368 Mbps

Fiber Carriers: SONET/SDH and OC

T-carriers were a great start into the digital world, but in the early 1980s, fiber-optic cabling became the primary tool for long-distance communication all over the world. By now AT&T was gone, replaced by a number of competing carriers. Competition was strong and everyone was making their own fiber transmission standards. In an incredible moment of corporate cooperation, in 1987, all of the primary fiber-optic carriers decided to drop their own standards and move to a new international standard called **Synchronous Optical Network (SONET)** in the United States and **Synchronous Digital Hierarchy (SDH)** in Europe.

All of these carriers, all adopting the same standard, created a world of simple interconnections between competing voice and data carriers. This adoption defined the moment that truly made the Internet a universal network. Before SONET, interconnections happened but they were outlandishly expensive, preventing the Internet from reaching many areas of the world.

SONET is the primary standard for long-distance, high-speed, fiber-optic transmission systems. There is a high level of comparison of SONET to network standards like Ethernet because SONET defines interface standards at the Physical and Data Link layers of the OSI seven-layer model. The physical aspect of SONET is partially covered by the Optical Carrier standards, but it also defines a ring-based topology that most SONET adopters now use. SONET does not require a ring, but a SONET ring has extra survivability in case of line loss. As a result, most of the big, long-distance optical pipes for the world's telecommunications networks are SONET rings.

The real beauty of SONET lies in its multiplexing capabilities. A single SONET ring can combine multiple DS1, DS3, even European E1 signals and package them into single, huge SONET frames for transmission. Clearly, for SONET to handle such large data rates it needs high-capacity fiber optics—and that's where the Optical Carrier standards come into play!

The **Optical Carrier (OC)** specification is used to denote the optical data carrying capacity (in Mbps) of fiber-optic cables in networks conforming to the SONET standard. The OC standard is an escalating series of speeds, designed to meet the needs of medium-to-large corporations. SONET establishes OCs from 51.8 Mbps (OC-1) to 39.8 Gbps (OC-768).

Tech Tip

What's in a Name?

Students often wonder why two separate names exist for the same technology. In reality, SONET and SDH vary a little in their signaling and frame type, but routers and other magic boxes on the Internet handle the interoperability between the standards. The American National Standards Institute (ANSI) publishes the standard as SONET; the International Telecommunications Union (ITU) publishes the standard as SDH, but includes SONET signaling. For simplicity's sake and because SONET is the more common term in the United States, this book uses SONET as the generic term for this technology.

SONET is one of the most important standards for making all of our WAN interconnections—and it's also the least likely standard you'll ever see because it's hidden away from all but the biggest networks.

SONET uses the **Synchronous Transport Signal (STS)** signal method. The STS consists of two parts: the **STS payload** (which carries data), and the **STS overhead** (which carries the signaling and protocol information). When we talk about STS, we add a number to the end of "STS" to designate the speed of the signal. For example, STS-1 is the 51.85-Mbps signal that runs on an OC-1 line. STS-3 runs at 155.52 Mbps on OC-3 lines, and so on.

Table 14.2 describes the most common optical carriers.

Table 14.2 Common Optical Carriers

SONET Optical Level	Line Speed	Signal Method
OC-1	51.85 Mbps	STS-1
OC-3	155.52 Mbps	STS-3
OC-12	622.08 Mbps	STS-12
OC-24	1.244 Gbps	STS-24
OC-48	2.488 Gbps	STS-48
OC-192	9.955 Gbps	STS-192
OC-256	13.22 Gbps	STS-256
OC-768	39.82 Gbps	STS-768

Packet Switching

All of these impressive connections that start with *T*s and *O*s are powerful, but they are not in and of themselves a complete WAN solution. These WAN connections with their unique packets (DS-0, STS, and so on) make up the entire mesh of long-range connections we call the Internet, carrying both packetized voice data and TCP/IP packets. All of these connections are point-to-point, so we need to add another level of devices to enable us to connect multiple T1s, T3s, or OC connections together to make that mesh. That's where packet switching comes into play.

Packets, as you know from what you've learned about networking, need some form of addressing scheme to get from one location to another. The telephone industry came up with its own types of packets that run on T-carrier and OC lines to get data from one central office to another. These packet-switching protocols are functionally identical to routable network protocols like TCP/IP. Today's WAN connections predominantly use two different forms of packet switching: Frame Relay and ATM.

Frame Relay

Frame Relay is an extremely efficient packet-switching standard, designed for and used primarily with T-carrier lines. It is especially effective for the off-again/on-again traffic typical of most LAN applications. Frame Relay switches packets quickly, but without any guarantee of data integrity at all. You can't even count on it to deliver all the frames, because it will discard frames whenever there is network congestion. In practice, however, a Frame Relay network delivers data quite reliably because T-carrier digital lines that use Frame Relay have very low error rates.

Frame Relay was extremely popular and is still widely available. If you decide to go with a T1 line in the United States, what you're getting is a T1 line running Frame Relay, although many companies use the newer ATM standard as their packet-switching solution with T-carrier lines.

ATM

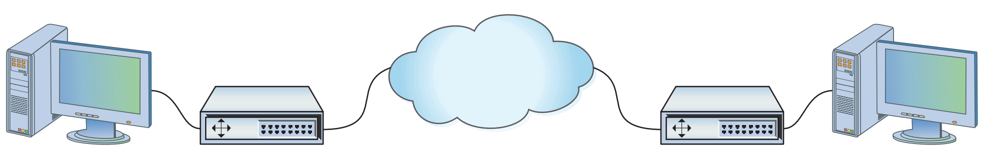
Don't think automatic teller machine here! **Asynchronous Transfer Mode (ATM)** is a network technology originally designed for high-speed LANs in the early 1990s. ATM only saw limited success in the LAN world, but has become extremely popular in the WAN world. In fact, most of the SONET rings that move voice and data all over the world use ATM for packet switching. ATM integrates voice, video, and data on one connection, using short and fixed-length packets called *cells* to transfer information. Every cell sent with the same source and destination travels over the same route, giving ATM the potential to remove the performance bottlenecks that exist in today's LANs and WANs.

The key problem ATM addresses is that data and audio/video transmissions have different transfer requirements. Data can tolerate a delay in transfer, but not signal loss (if it takes a moment for a Web page to appear, you don't care). Audio and video transmissions, on the other hand, can tolerate signal loss but not delay (it would make the phone calls sound delayed). Because ATM transfers information in cells of one set size (53 bytes long), it is scalable and can handle both types of transfers well. ATM transfer speeds range from 155.52 to 622.08 Mbps and beyond. If your location was big enough to order an OC line from your ISP, odds are good you'd be getting an OC line running ATM.

MPLS

Frame Relay and ATM are both fantastic packet-switching technologies but they're designed to support any type of traffic that might come over the network. Today, TCP/IP is the predominant data technology and has a number of issues that neither Frame Relay nor ATM addresses. To address these issues, many ISPs (and large ISP clients) use an additional technology called **Multi-Protocol Label Switching (MPLS)**.

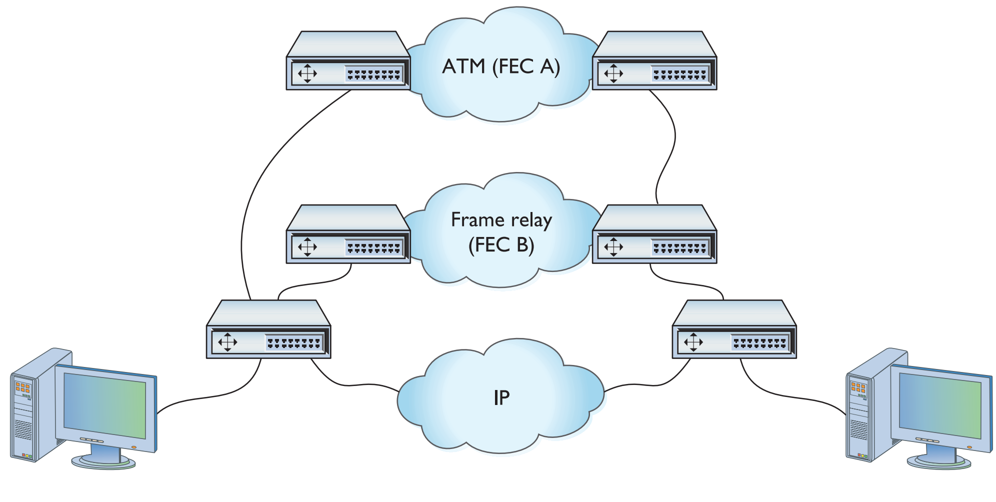
Frame Relay and ATM networks are the property of the telephone networks. Those of us who use these networks (and all of us do) never actually see exactly where the packets really go—nor do we have to, because the telephone companies are very dependable. Since we neither can see nor do anything with these networks, the network industry likes to represent them using a cloud analogy. Figure 14.15 shows an example of a connection between a Web server and a Web client. We only show routers that we actually control—everything else is "in the clouds."



• Figure 14.15 The cloud

In some situations we do want to know what's happening in the clouds. Here's an example: Anyone who can't afford to not be on the Internet has at least two ISP connections. Let's say Company X has two locations. Each location has an ATM OC-1 line with a Frame Relay T-3 line as a backup (Figure 14.16).

Normally, your gateway router decides what to use based on availability and the router table. But what if you have some really critical data (FTP for example) that you only want to send out on the ATM link? There's nothing to force that to happen. The quality of service (QoS) standards that you saw in Chapter 12, "Advanced Networking Devices," only cover the amount of bandwidth to use, not which port to use. MPLS is a router feature that labels certain data to use a desired connection. It works with any type of packet switching (even Ethernet) to force certain types of data to use a certain path. This path in essence is a VPN—just like the VPNs we saw in Chapter 12 (but not encrypted). When you use MPLS, you usually say "MPLS VPN" because it creates a virtualized private tunnel through the cloud of the Internet.



• **Figure 14.16** Company X's network

Real-World WAN

There are two reasons to use a telephony WAN connection: to get your LAN on the Internet or to make a private connection between two or more of your private LANs. How you go about getting one of these lines changes a bit depending on which you want to do. Let's start with connecting to the Internet.

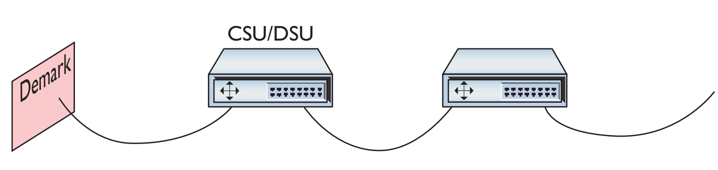
Traditionally, getting a WAN Internet connection was a two-step process: you talked to the telephone company to get the line physically installed and then talked to an ISP to provide you with Internet access. Today, almost every telephone company is also an ISP so this is usually a simple process. Just go online and do a Web search of ISPs in your area and give them a call. You'll get a price quote and, if you sign up, they will do the installation.

There are a few tricks to reduce the price. If you're in an office building, odds are very good that there's already a T1 or better line installed you can use and that an ISP is already serving people in your building. Talk to the building supervisor to find out. If there isn't, you have to pay for a new line. If there's a nearby interconnect, this might be cheap. If you want the telephone company to run an OC line to your house, brace for a quote of thousands of dollars just to get the line.

The telephone company runs your T-carrier (or bigger) line to a demarc. This demarc is important because this is where the phone company's responsibility ends! Everything on "your" side of the demarc is your responsibility. From there, you or your ISP installs a CSU/DSU (for T-carriers) and that device connects to your router.

There's a tremendous amount of variance here depending on who does this for you. The classic example (sticking with T-carrier here) is a demarc, CSU/DSU, and router setup, as shown in Figure 14.17.

T-carriers have been around so long that many of these parts are combined. It's very common to have a single box that combines the CSU/DSU and the router in one handy device, such as the Juniper router shown earlier in Figure 14.14.



• **Figure 14.17** Old-school T-carrier setup

WAN telephony carriers are incredibly dependable, far more dependable than cheaper alternatives (like cable modems), and that's one of the main reasons people still use them. But it's critical that you know how to test your end of the connection if you ever suspect a problem. The single most important test is called the **Bit Error Rate Test (BERT)**. A BERT test is an end-to-end test that verifies the T-carrier connection. Every CSU/DSU has a different way to BERT test—just make sure you know how to do it on yours!

Alternative to Telephony WAN

Telephony WANs were our first big connections. They're still the core of what makes up most of our Internet backbone and private connections but have given way to more advanced technologies. The three biggest newer technologies for Internet connections are Ethernet, DSL, and cable modems. I need to give them a quick mention here.

Many ISPs provide straight Ethernet connections. As you should well appreciate by now, Ethernet has some serious distance limitations, covering no more than a kilometer, except for very expensive single mode fiber solutions. This limits Ethernet to places like large cities where you as a customer are close to your ISP. On the other hand, Ethernet links are extremely easy to set up, because you usually have nothing more than a fiber cable that you connect directly to a router. Many companies are dropping T1 and other slower options for the speed of Ethernet.

DSL and cable have been around for quite a while and deserve some serious discussion. However, you can't install your own DSL or cable modem connection as you can with telephony WANs carriers. For example, you can't have your own private cable modem connection between two of your offices. DSL and cable modems are only for Internet connections and, as a result, are really more of a last-mile issue—let's discuss DSL and cable in the next section.

The Last Mile

Speed is the key to the Internet, but historically there's always been one big challenge: getting data from central offices to individual users. This wasn't a problem for larger companies that could afford their own WAN connections, but what about individuals and small companies that couldn't or wouldn't pay hundreds of dollars a month for a T1? This area, the infamous last mile, was a serious challenge early on for both Internet connections and private connections because the only common medium was POTS lines. A number of last-mile solutions have appeared over the years, and the CompTIA Network+ exam tests you on the most popular ones. Let's begin with the granddaddy of them all, the telephone line that runs to your house.

Telephone

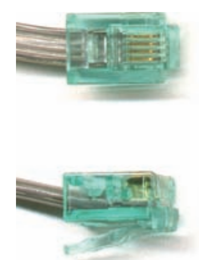
There are many different types of telephone lines available, but all the choices break down into two groups: dedicated and dial-up. **Dedicated lines** are always off the hook (that is, they never hang up on each other).

A dedicated line (like a T1) does not have a phone number. In essence, the telephone company creates a permanent, hard-wired connection between the two locations, rendering a phone number superfluous. **Dial-up lines**, by contrast, have phone numbers; they must dial each other up to make a connection. When they're finished communicating, they hang up. Two technologies make up the overwhelming majority of dial-up connections—PSTN and ISDN.

Public Switched Telephone Network

The oldest, slowest, and most common original phone connection is the **Public Switched Telephone Network (PSTN)**. PSTN is also known as Plain Old Telephone Service. PSTN is just a regular phone line, the same line that used to run into everybody's home telephone jacks from the central office of your local exchange carrier (LEC—the telephone company that provides local connections and usually the one who owns your local central office).

Because PSTN was designed long before computers were common, it was designed to work with only one type of data: sound. Here's how it works. The telephone's microphone takes the sound of your voice and translates it into an electrical analog waveform. The telephone then sends that signal through the PSTN line to the phone on the other end of the connection. That phone translates the signal into sound on the other end using its speaker. The important word here is *analog*. The telephone microphone converts the sounds into electrical waveforms that cycle 2400 times a second. An individual cycle is known as a **baud**. The number of bauds per second is called the **baud rate**. Pretty much all phone companies' PSTN lines have a baud rate of 2400. PSTN uses a connector called RJ-11. It's the classic connector you see on all telephones (Figure 14.18).



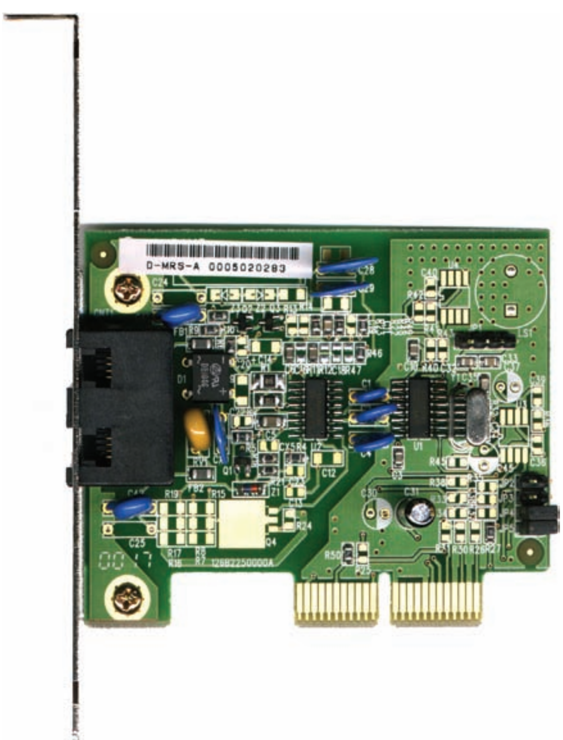
• **Figure 14.18** RJ-11 connectors (top and side views)

When you connect your modem to a phone jack, the line then runs to your **network interface unit (NIU)**, or demarc. The term "network interface unit" is more commonly used to describe the small box on the side of a home that accepts the incoming lines from the telephone company and then splits them to the different wall outlets. Demarc is more commonly used to describe large connections used in businesses. The terms are interchangeable and always describe the interface between the lines the telephone company is responsible for and the lines for which you are responsible (Figure 14.19).

Computers, as you know, don't speak analog—only digital (ones and zeroes) will do. In addition, the people who invented the way PCs communicate decided to divide any digital signal going in and out of your computer into 8 bits at a time. To connect over phone lines, they need two devices: one that converts this 8-bit wide (parallel) digital signal from the computer into serial (1-bit wide) digital data, and then another device to convert (modulate) the data into analog waveforms that can travel across PSTN lines. You already know that the device that converts the analog data to digital and back is called a **modulator-demodulator (modem)**. The modem in your PC (assuming you still have one) also contains a device called a **Universal Asynchronous Receiver/Transmitter (UART)**. The UART takes the 8-bit-wide digital data and converts it into 1-bit-wide digital data and hands it to the modem for conversion to analog data. The process is reversed for incoming data. Even though internal modems are actually both a UART and a modem, we just say the word "modem" (Figure 14.20).

> Internal modems are both a UART and a modem. External modems use a serial or USB port. The serial or USB port contains the UART and the external modem truly is just a modem.

• **Figure 14.19** Typical home demarc



• **Figure 14.20** Internal modem

Baud vs. Bits per Second

Modems use phone lines to transmit data at various speeds. These speeds cause a world of confusion and problems for computer people. This is where a little bit of knowledge becomes dangerous. Standard modems you can buy for your home computer normally transmit data at speeds up to 56 Kbps. That's 56 kilobits per second, *not* 56 kilobaud! Many people confuse the terms *baud* and *bits per second*. This confusion arises because the baud rate and bits per second are the same for modems until the data transfer rate surpasses 2400 bps.

A PSTN phone line takes analog samples of sound 2400 times a second. This standard was determined a long time ago as an acceptable rate for sending voice traffic over phone lines. Although 2400-baud analog signals are in fact fine for voice communication, they are a big problem for computers trying to send data because computers only work with digital signals. The job of the modem is to take the digital signals it receives from the computer and send them out over the phone line in an analog form, using the baud cycles from the phone system. A 2400-bps modem—often erroneously called a 2400-baud modem—uses 1 analog baud to send 1 bit of data.

As technology progressed, modems became faster and faster. To get past the 2400-baud limit, modems would modulate the 2400-baud signal multiple times in each cycle. A 4800-bps modem modulated 2 bits per baud, thereby transmitting 4800 bps. All PSTN modem speeds are always a multiple of 2400, with the latest (and last) generation of modems achieving 2400×24 = 57,600 bps (56 Kbps).

V Standards

For two modems to communicate with each other at their fastest rate, they must modulate signals in the same fashion. The two modems must also negotiate with, or *query*, each other to determine the fastest speed they share. The modem manufacturers themselves originally standardized these processes as a set of proprietary protocols. The downside to these protocols was that unless you had two modems from the same manufacturer, modems often would not work together. In response, a European standards body called the **CCITT** established standards for modems. These standards, known generically as the **V standards**, define the speeds at which modems can modulate. The most common of these speed standards are as follows:

- **V.22** 1200 bps
- **V.22bis** 2400 bps
- **V.32** 9600 bps
- **V.32bis** 14,400 bps
- **V.34** 28,000 bps

- **V.90** 57,600 bps
- **V.92** 57,600 bps

The current modem standard now on the market is the **V.92 standard**. V.92 has the same download speed as the V.90, but upstream rates increase to as much as 48 Kbps. If your modem is having trouble getting 56-Kbps rates with V.90 in your area, you will not notice an improvement. V.92 also offers a Quick Connect feature, which implements faster handshaking to cut connection delays. Finally, the V.92 standard offers a Modem On Hold feature, which enables the modem to stay connected while you take an incoming call-waiting call or even initiate an outgoing voice call. This feature only works if the V.92 server modem is configured to enable it.

Do not memorize these V standards—just know what they do.

In addition to speed standards, the CCITT, now known simply as ITU, has established standards controlling how modems compress data and perform error checking when they communicate. These standards are as follows:

- **V.42** Error checking
- **V.42bis** Data compression
- **V.44** Data compression
- **MNP5** Both error checking and data compression

The beauty of these standards is that you don't need to do anything special to enjoy their benefits. If you want 56-Kbps data transfers, for example, you simply need to ensure that the modems in the local system and the remote system both support the V.90 standard. Assuming you have good line quality, the connections will run at or at least close to 56 Kbps.

ISDN

PSTN lines traditionally just aren't that good. While the digital equipment that connects to a PSTN supports a full 64-Kbps DS0 channel, the combination of the lines themselves and the conversion from analog to digital means that most PSTN lines rarely go faster than at most 33 Kbps—and, yes, that includes the 56-Kbps connections.

There are many pieces to a PSTN telephone connection. First, there's the modem in your computer that converts the digital information to analog. Then there's the phone line that runs from your phone out to your NIU and into the central office. The central office stores the modems that convert the analog signal back to digital and the telephone switches that interconnect multiple individual local connections into the larger telephone network. A central office switch connects to long-distance carriers via high-capacity *trunk lines* (at least a T1) and will also connect to other nearby central offices. The analog last mile was an awful way to send data but it had one huge advantage: most everyone owned a telephone line.

During this upgrade period, customers continued to demand higher throughput from their phone lines. The phone companies were motivated to come up with a way to generate higher capacities. Their answer was fairly straightforward: make the last mile digital. Since everything but the last mile was already digital, by adding special equipment at the central office and the user's location, phone companies felt they could achieve a true, steady, dependable throughput of 64 Kbps per line over the same copper wires already used by PSTN lines. This process of sending telephone transmission across fully digital lines end-to-end is called **Integrated Services Digital Network (ISDN)** service.

ISDN also supports voice, but requires special ISDN telephones.

ISDN service consists of two types of channels: **Bearer channels (B channels)** carry data and voice information using standard DS0 channels (64 Kbps), while **Delta channels (D channels)** carry setup and configuration information at 16 Kbps. Most providers of ISDN let the user choose either one or two B channels. The more common setup is two B/one D, called a **Basic Rate Interface (BRI)** setup. A BRI setup uses only one physical line, but each B channel sends 64 Kbps, doubling the throughput total to 128 Kbps.

There is also a type of ISDN called **Primary Rate Interface (PRI)**. ISDN PRI is actually just a full T-1 line, carrying 23 B channels.

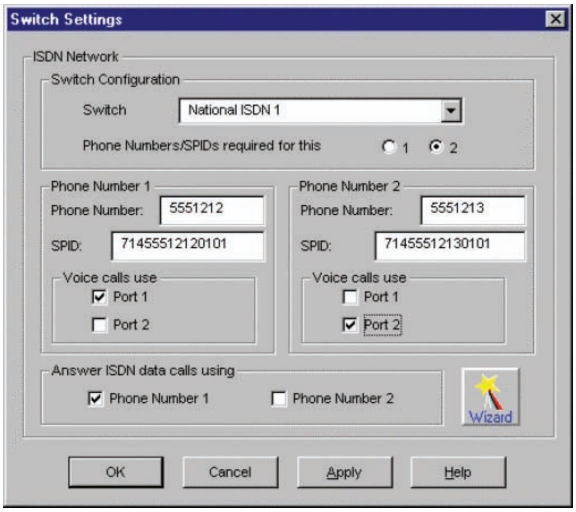
Remember, a B channel is a DS0 channel.

The physical connections for ISDN bear some similarity to PSTN modems. An ISDN wall socket is usually something that looks like a standard RJ-45 network jack. This line runs to your demarc. In home installations most telephone companies will install a second demarc separate from your PSTN demarc. The most common interface for your computer is a device called a **terminal adapter (TA)**. TAs look like regular modems and, like modems, come in external and internal variants. You can even get TAs that also function as hubs, enabling your system to support a direct LAN connection (Figure 14.21).

• Figure 14.21 A TeleWell ISDN terminal adapter

You usually need to be within about 18,000 feet of a central office to use ISDN. When you install an ISDN TA, you must configure the other ISDN telephone number you want to call and a special number called the Service Profile ID (SPID). Your ISP provides the telephone number, and the telephone company gives you the SPID. (In many cases the telephone company is also the ISP.) Figure 14.22 shows a typical installation screen for an internal ISDN TA in an old version of Windows. Note that each channel has a phone number in this case.

ISDN continues to soldier on in today's networking world, but has for the most part been replaced by faster and cheaper methods such as DSL and cable modems. Nevertheless, every



• Figure 14.22 ISDN settings in an old version of Windows major telephone company still provides ISDN. ISDN is often the only option for users in locations where other high-speed connection options don't exist.

DSL

Digital Subscriber Line (DSL) is a fully digital, dedicated (no phone number) connection provided by a number of telephone companies. DSL represented the next great leap forward past ISDN for telephone lines. A DSL connection manifests as just another PSTN connection, using the same telephone lines and RJ-11 jacks as any regular phone line. DSL comes in a number of versions, but the three most important to know for the CompTIA Network+ exam are **Symmetric DSL (SDSL)**, **Asymmetric DSL (ADSL)**, and the newer **Very High Bitrate DSL (VDSL)**. SDSL lines provide the same upload and download speeds, making them excellent for those who send as much data as they receive, although SDSL is relatively expensive (VDSL is a new form of SDSL—see "VDSL" in this section). ADSL uses different upload and download speeds. ADSL download speeds are much faster than the upload speeds. Most small office and home office (SOHO) users are primarily concerned with fast *downloads* for things like Web pages, and can tolerate slower upload speeds. ADSL is always much cheaper than SDSL, and VDSL is usually the most expensive.

SDSL

SDSL provides equal upload and download speed and, in theory, provides speeds up to 15 Mbps, although the vast majority of ISPs provide packages ranging from 192 Kbps to 9 Mbps. A recent tour of some major DSL providers in the author's home town, Houston, Texas, revealed the following SDSL speed options:

- 192 Kbps
- 384 Kbps
- 768 Kbps
- 1.1 Mbps
- 1.5 Mbps

As you might imagine, the pricing for the faster services was higher than for the lower services!

ADSL

ADSL provides theoretical maximum download speeds up to 15 Mbps and upload speeds up to 1 Mbps. However, all ADSL suppliers "throttle" their ADSL speeds and provide different levels of service. Real-world ADSL download speeds vary from 384 Kbps to 15 Mbps, and upload speeds go from as low as 128 Kbps to around 768 Kbps. Touring the same DSL providers in Houston, Texas, here's a few speed options:

- 384 Kbps download/128 Kbps upload
- 1.5 Mbps download/384 Kbps upload
- 6 Mbps download/768 Kbps upload

VDSL

VDSL is the latest version of DSL to appear. Although adoption is fairly low (at least in the United States), its ability to provide speeds up to 100 Mbps in both directions is gaining interest quickly. VDSL achieves these speeds by adding very advanced methods to encode the data. Don't get too excited about these great speed increases. They are very distance dependent: you won't get 100 Mbps unless you're around 300 meters from the DSLAM (see "DSL Features" next). VDSL is designed to run on copper phone lines, but many VDSL suppliers use fiber-optic cabling to increase distances. In the United States, these fiber VDSL services are fiber-to-the-home solutions. The two most popular carriers are AT&T's U-verse and Verizon's Fiber Optic Service (FiOS).

> **Try This!**
>
> **Comparing Options in Your Neighborhood**
>
> So, what do your local providers offer in terms of higher-speed service, if any? Try this! Call up your local phone company or shop them on the Web (www.dslreports.com is an excellent reference). Does the company offer DSL? What about ISDN? What speed options do you have? If you want to compare with other parts of the United States, check one of the national services, such as Speakeasy (www.speakeasy.net).

DSL Features

One nice aspect of DSL is that you don't have to run new phone lines. The same DSL lines you use for data can simultaneously transmit your voice calls.

All versions of DSL have the same central office–to–end user distance restrictions as ISDN—around 18,000 feet from your demarc to the central office. At the central office your DSL provider has a device called a **DSL Access Multiplexer (DSLAM)** that connects multiple customers to the Internet.

> **Tech Tip**
>
> **Speed Guarantees**
>
> *No DSL provider guarantees any particular transmission speed and will only provide service as a "best efforts" contract—a nice way to say that DSL lines are notorious for substantial variations in throughput. This is true even of ISPs that lease the lines from the same telephone service.*

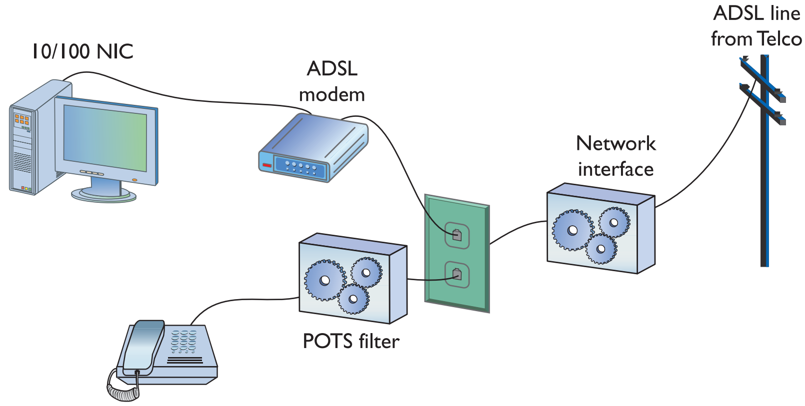
Installing DSL

DSL operates using your pre-existing telephone lines (assuming they are up to specification). This is wonderful, but also presents a technical challenge. For DSL and your run-of-the-mill POTS line to coexist, you need to filter out the DSL signal on the POTS line. A DSL line has three information channels: a high-speed downstream channel, a medium-speed duplex channel, and a POTS channel. Segregating the two DSL channels from the POTS channel guarantees that your POTS line will continue to operate even if the DSL fails. This is accomplished by inserting a filter on each POTS line, or a splitter mechanism that allows all three channels to flow to the DSL modem, but sends only the POTS channel down the POTS line. The DSL company should provide you with a few POTS filters for your telephones. If you need more, most computer/electronics stores stock DSL POTS filters.

> **Tech Tip**
>
> **DSL POTS Filters**
>
> *If you install a telephone onto a line in your home with DSL and you forget to add a filter, don't panic. You won't destroy anything, although you won't get a dial tone either! Just insert a DSL POTS filter and the telephone will work.*

The most common DSL installation consists of a **DSL modem** connected to a telephone wall jack and to a standard NIC in your computer (Figure 14.23). A DSL modem is not an actual modem—it's more like an ISDN terminal adapter—but the term stuck, and even the manufacturers of the devices now call them DSL modems.

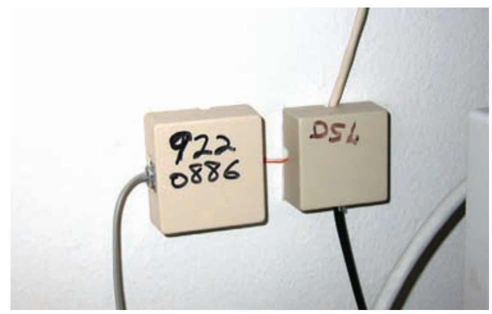
Many offices use DSL. In my office we use a special DSL line (we use a digital phone system, so the DSL must be separate) that runs directly into our equipment room (Figure 14.24).



• **Figure 14.23** An ADSL modem connection to a PC and telco

The one potentially costly aspect of ADSL service is the ISP link. Many ISPs add a significant surcharge to use ADSL. Before you choose ADSL, make sure that your ISP provides ADSL links at a reasonable price. Most telephone companies bundle ISP services with their ADSL service for a relatively low cost.

This DSL line runs into our DSL modem via a standard phone line with RJ-11 connectors. The DSL modem connects to our gateway router with a CAT 5e patch cable, which in turn connects to the company's hub. Figure 14.25 shows an ADSL modem and a router, giving an idea of the configuration in our office.



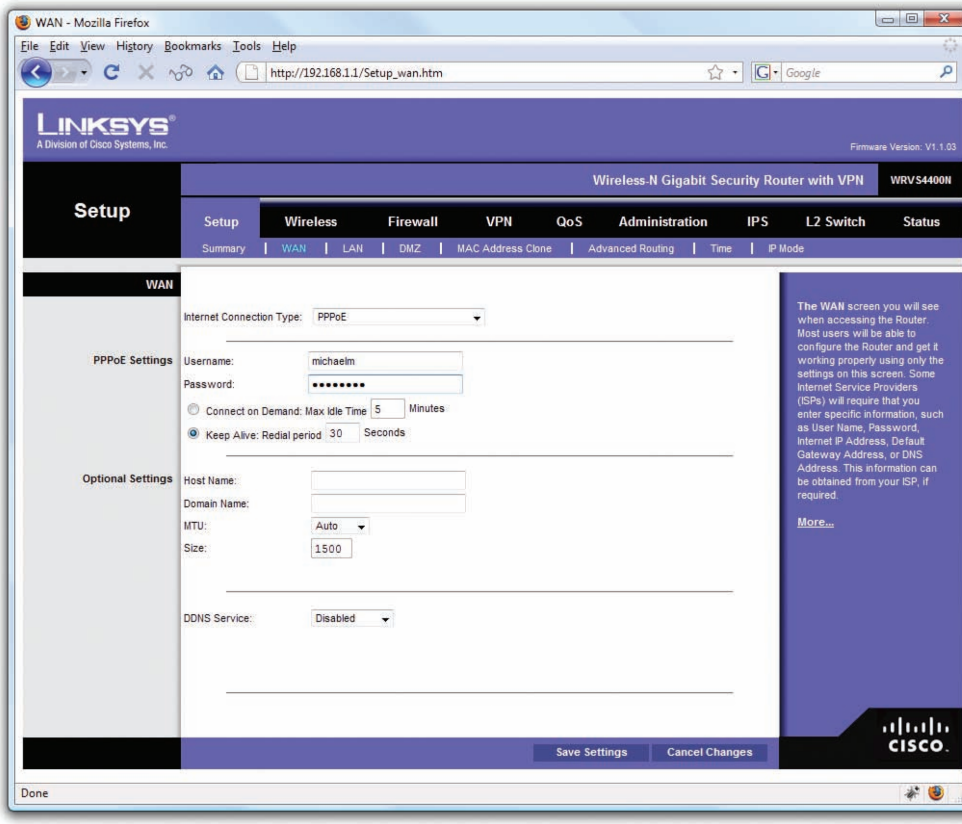
• **Figure 14.24** DSL line into equipment room

Home users often connect the DSL modem directly to their PC's NIC. Either way, there is nothing to do in terms of installing DSL equipment on an individual system—just make sure you have a NIC. The person who installs your DSL will test the DSL line, install the DSL modem, connect it to your system, and verify that it all works. The one issue you may run into with DSL is something called **Point-to-Point Protocol over Ethernet (PPPoE)**.

The first generation of DSL providers used a **bridged connection;** once the DSL line was running it was the same as if you snapped an Ethernet cable into your NIC. You were on the network. Those were good days for DSL. You just plugged your DSL modem into your NIC and, assuming your IP settings were whatever the DSL folks told you to use, you were running.

• **Figure 14.25** DSL connection

The DSL providers didn't like that too much. There was no control—no way to monitor who was using the DSL modem. As a result, the DSL folks started to use PPPoE, a protocol that was originally designed to encapsulate PPP frames into Ethernet frames. The DSL people adopted it to make stronger controls over your DSL connection. In particular, you could no longer simply connect; you now had to log on with an account and a password to make the DSL connection. PPPoE is now predominant on DSL. If you get a DSL line, your operating system has software to enable you to log onto your DSL network. Most SOHO routers come with built-in PPPoE support, enabling you to enter your user name and password into the router itself (Figure 14.26).



• Figure 14.26 PPPoE settings in SOHO router

Cable Modems

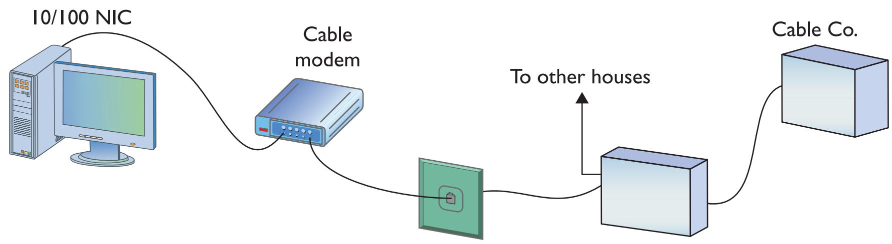
Any high-speed (faster than PSTN) last-mile connection is called a broadband connection.

The first big competition for ADSL came from the cable companies. Almost every house in America has a coax cable running into it for cable TV. In a moment of genius, the cable industry realized that if it could put the Home Shopping Network and the History Channel into every home, why not provide Internet access? The entire infrastructure of the cabling industry had to undergo some major changes to deal with issues like bidirectional communication, but cable modem service is now common in the United States. Cable modems are as common as cable TV boxes.

The single most impressive aspect of cable modems is their phenomenal top speeds. These speeds vary from cable company to cable company, but most advertise speeds in the (are you sitting down?) *10 to 27 megabits per second* range. Most cable modems provide a throughput speed of 1 to 10 Mbps downloading and 500 Kbps to 1 Mbps uploading—there is tremendous variance between different providers.

A cable modem installation consists of a cable modem connected to a cable outlet. The cable modem gets its own cable outlet, separate from the one that goes to the television. It's the same cable line, just split from the main line as if you were adding a second cable outlet for another television. As with ADSL, cable modems connect to PCs using a standard NIC (Figure 14.27).

Cable modems connect using coax cable to a head end, similar to a telephone company's central office. Head ends in turn connect to the cable company's network. This network uses a unique protocol called **Data Over Cable Service Interface Specification (DOCSIS)**. Most recently, the specification was revised (DOCSIS 3.0) to significantly increase transmissions speeds (this time both upstream and downstream) and introduce support for Internet Protocol version 6 (IPv6).



• **Figure 14.27** Cable modem

Many companies sell routers with a built-in cable modem.

It's hard to tell a cable modem from a DSL modem. The only difference, other than the fact that one will have "cable modem" printed on it while the other will say "DSL modem," is that the cable modem has a coax and an RJ-45 connecter while the DSL modem has an RJ-11 and an RJ-45 connector.

Cable modems have proven themselves to be reliable and fast and have surpassed DSL as the broadband connection of choice in homes. Cable companies are also aggressively marketing to business customers with high-speed packages, making cable a viable option for businesses.

Satellite

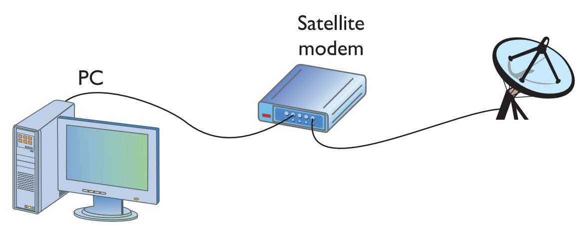
Living in the countryside may have its charms, but getting high-speed Internet access is tough. For those too far away to get anything else, satellite may be your only option. Satellite access comes in two types: one-way and two-way. *One-way* means that you download from satellite but you must use a PSTN connection for uploads. *Two-way* means the satellite service handles both the uploading and downloading.

Satellite isn't as fast as DSL or cable modems, but it's still faster than PSTN. Both one-way and two-way satellite connections provide around 500 Kbps download and 50 Kbps upload. Satellite requires a small satellite antenna, identical to the ones used for satellite television. This antenna connects to a satellite modem, which in turn connects to your PC or your network (Figure 14.28).

Neither cable modems nor satellites use PPP, PPPoE, or anything else that begins with three Ps.

Wireless

Wireless is a big topic for the CompTIA Network+ exam and I've saved an entire chapter (Chapter 16) just to discuss the topic. For now, it's important to appreciate that there are two types of wireless service that people may use for Internet connections: mobile data services (provided by cell phone companies) and 802.11.



• **Figure 14.28** Satellite connection

Mobile data services have names like GSM, GRPS, EDGE, and HSPDA (there are many more standards). These services use the cellular telephone network to provide access. Mostly used with cell phones and smart phones, most mobile data services have wireless NICs you can use on laptops and desktop computers (Figure 14.29).

802.11 is the main wireless standard, the one used in buildings and installed in just about every laptop. 802.11 isn't a common WAN solution, but is sometimes used in areas where alternatives are not available.

Fiber

DSL was the first popular last-mile WAN option but over the years cable modems have taken the lead. In an attempt to regain market share, telephone providers are now rolling out fiber-to-the-home/fiber-to-the-premises options that are giving the cable companies a scare. In the United States, two companies, AT&T (U-verse) and Verizon (FiOS) are offering very attractive ISP, television, and phone services at speeds that will eventually increase past 100 Mbps. These services are quickly gaining in popularity and giving cable companies a run for the money.

• Figure 14.29 Mobile wireless NIC

Which Connection?

With so many connection options for homes and small offices, making a decision is often a challenge. Your first question is availability: which services are available in your area? The second question is, how much bandwidth do you need? This is a question of great argument. Most services will be more than glad to increase service levels if you find that a certain level is too slow. I usually advise clients to start with a relatively slow level, and then increase if necessary. After all, it's hard to go slower once you've tasted the higher speeds, but relatively painless to go faster!

Try This!

Going Connection Shopping

You've already checked on the availability of DSL and ISDN in your neighborhood, but now you have more choices! Try this! Do you have cable or satellite available? A great Web site to start your search is www.dslreports.com. It has a handy search feature that helps you determine the types of service and the costs for DSL, cable, and other services. Which one makes sense for you?

Using Remote Access

Because most businesses are no longer limited to a simple little shop like you would find in a Dickens novel, there is a great need for people to be able to access files and resources over a great distance. Enter remote access. **Remote access** uses WAN and LAN connections to enable a computer user to log onto a network from the other side of a city, a state, or even the globe. As people travel, information has to remain accessible. Remote access enables users to connect a server at the business location and log into the network as if they were in the same building as the company. The only problem with remote access is that there are so many ways to do it! The six most common forms of remote access are as follows:

- **Dial-up to the Internet** Using a dial-up connection to connect to your ISP
- **Private dial-up** Using a dial-up connection to connect to your private network

- **Virtual private network** Using an Internet connection to connect to a private network
- **Dedicated connection** Using a non-dial-up connection to another private network or the Internet
- **Remote terminal** Using a terminal emulation program to connect to another computer
- **VoIP** Voice over IP

In this section we look at the issues related to configuring these six types of connections. After seeing how to configure these types of remote connections, we move into observing some security issues common to every type of remote connections.

Extranet is one of those terms that you'll see more in books than in the day-to-day workings of networks and network techs. So, what is an extranet? Whenever you allow authorized remote users to access some part of your private network, you have created an extranet.

Dial-Up to the Internet

Dialing up to the Internet is the oldest and cheapest method to connect to the Internet and is still somewhat common. Even with broadband and wireless so prevalent, every self-respecting network tech (or maybe just old network techs like me) keep a dial-up account as a backup. You buy a dial-up account from an ISP (many wireless and broadband ISPs give free dial-up—just ask). All operating systems come with dial-up support programs but you'll need to provide:

- A modem (most operating systems check for a modem before setting up a dial-up connection)
- The telephone number to dial (provided to you by the ISP)
- User name and password (provided to you by the ISP)
- Type of connection (dial-up always uses PPP)
- IP information (provided to you by the ISP—usually just DHCP)

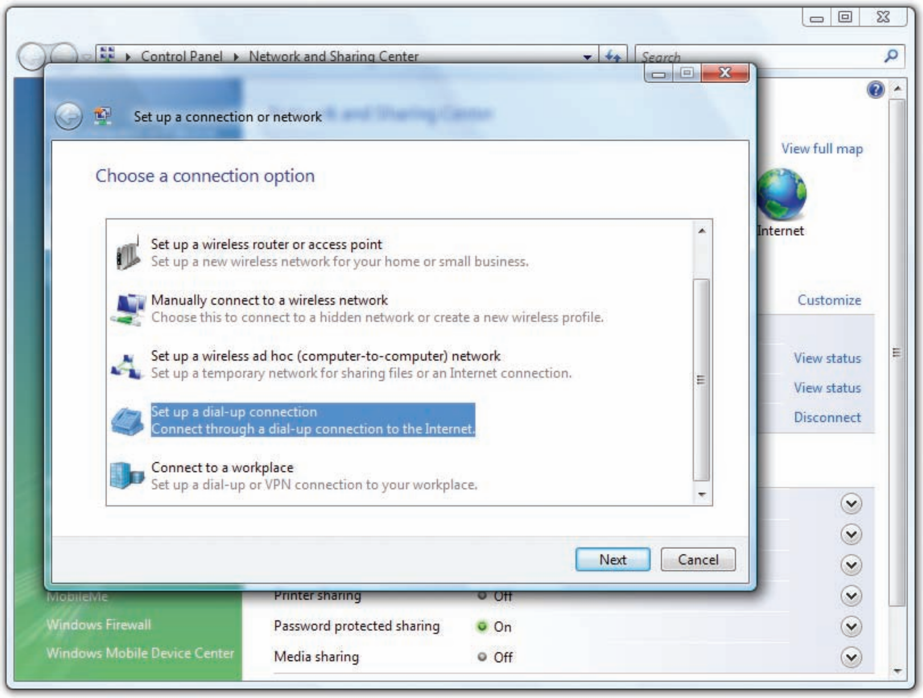
Every operating system comes with the software to help you set up a dial-up connection. In Windows Vista you go to the **Set up a dial-up connection** option in the Network and Sharing Center (Figure 14.30). Whatever the name, this tool is what you use to create dial-up connections.

Private Dial-Up

A private dial-up connection connects a remote system to a private network via a dial-up connection. Private dial-up does not use the Internet! Private dial-up requires two systems. One system acts as a **remote access server (RAS)**. The other system is the client running a connection tool (usually the same tool you just saw in the previous section).

In Windows a RAS is a server running Remote Access Service (RAS) dedicated to handling users who are not directly connected to a LAN but who need to access file and print services on the LAN from a remote location. For example, when a user dials into a network from home using an analog modem connection, she is dialing into a RAS. Once the user is authenticated, she can access shared drives and printers as if her computer were physically connected to the office LAN.

When you run the Microsoft product Remote Access *Service* on a server, you turn that server into remote access *server*.



• **Figure 14.30** Dial-up on Windows Vista

You must set up a server in your LAN as a RAS server. That RAS server, which must have at least one modem, accepts incoming calls and handles password authentication. RAS servers use all the standard authentication methods (PAP, CHAP, EAP, 802.1X, and so on) and have separate sets of permissions for dial-in users and local users. You must also configure the RAS to set the dial-in user's rights and permissions. Configuring a RAS system is outside the scope of this book, because each one is different, but it's something you must do to get RAS to work correctly (Figure 14.31).

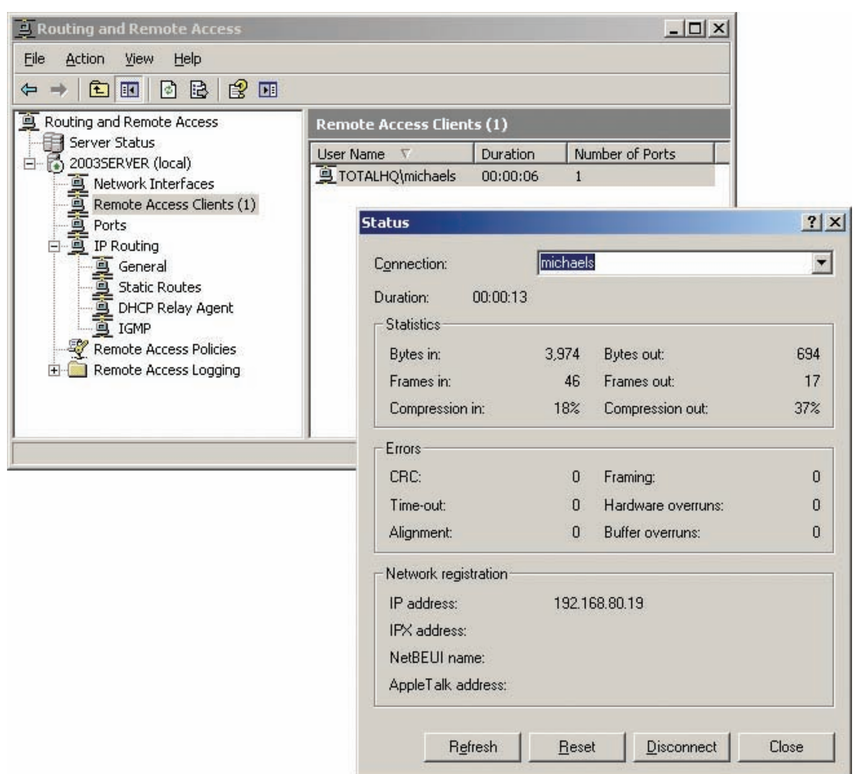
Creating the client side of a private dial-up connection is identical to setting up a dial-up connection to the Internet. The only difference is that instead of

Tech Tip

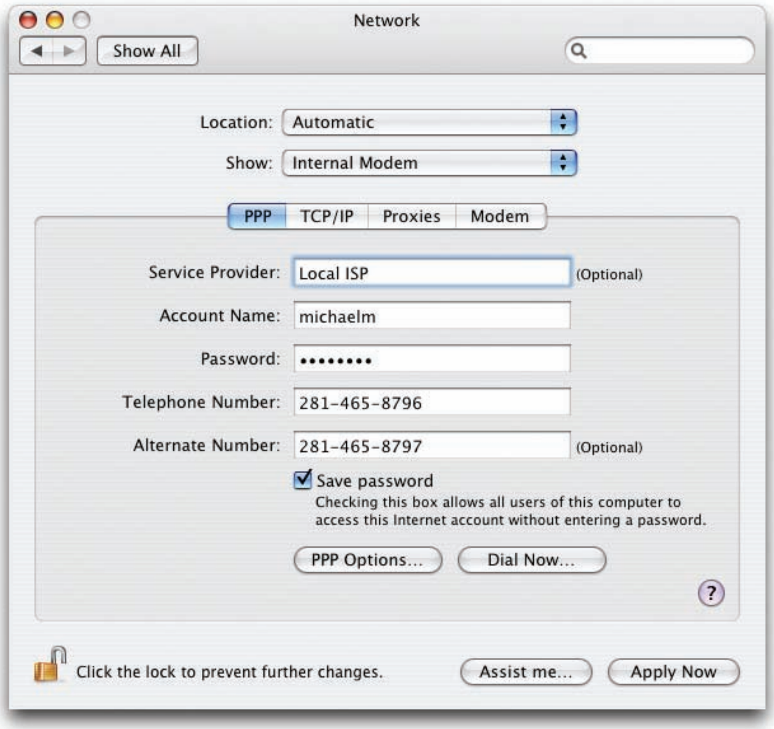
RAS

Remote access server is a catchall phrase. It refers to both the hardware component (servers built to handle the unique stresses of a large number of clients calling in) and the software component of a remote access solution.

Most techs call RAS "razz," rather than use the initials, "R-A-S." This creates a seemingly redundant phrase used to describe a system running RAS: "RAS server." This helps distinguish servers from clients and makes geeks happier.



• **Figure 14.31** Windows RAS in action



• Figure 14.32 Dial-up on Macintosh OS X

having an ISP tell you what IP settings, account name, and password to use, the person who sets up the RAS server tells you this information (Figure 14.32).

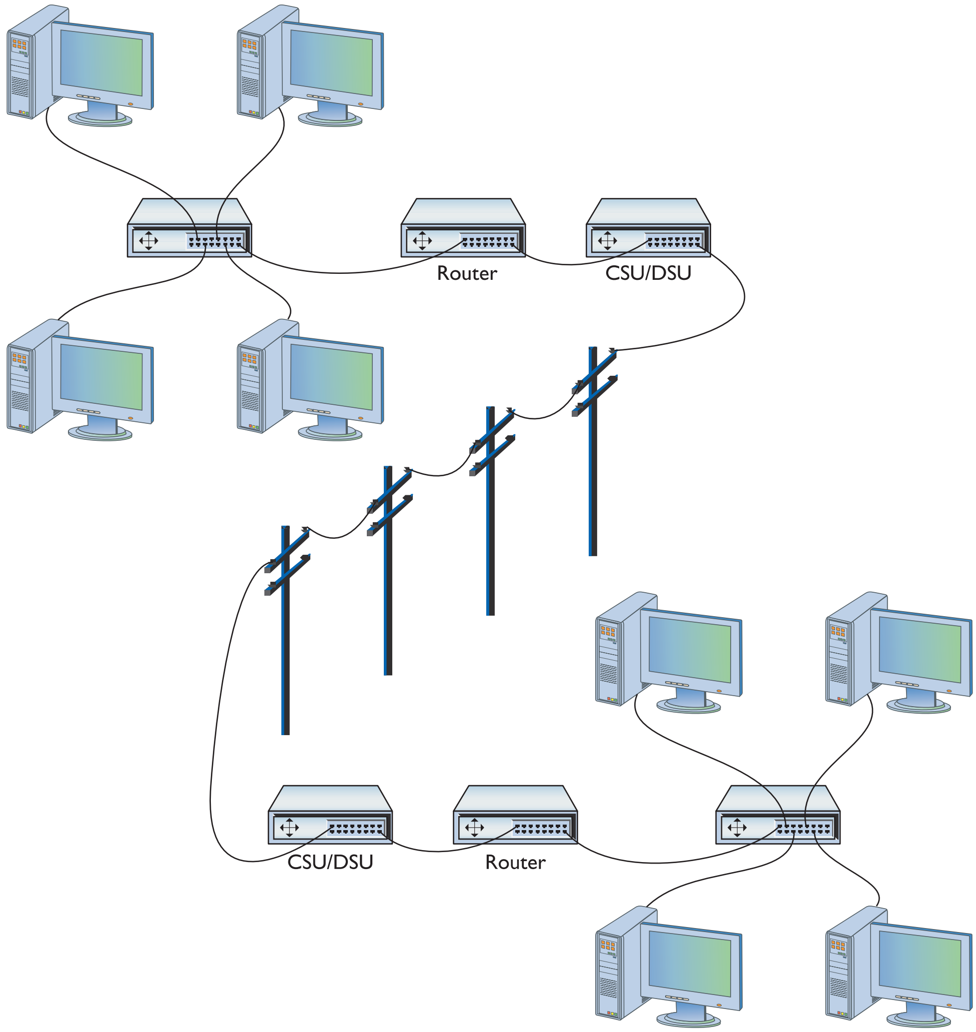
VPNs

We've already covered VPNs in Chapter 12. This might be a good time to review the different types of VLANs available (PPTP, L2TP, IPSec) and remember how each one is used.

Dedicated Connection

Dedicated connections are remote connections that are never disconnected. Dedicated connections can be broken into two groups: dedicated private connections between two locations, and dedicated connections to the Internet. Dedicated private connections manifest themselves as two locations interconnected by a (usually high-speed) connection such as a T1 line (Figure 14.33).

Each end of the T1 line goes into a router (after going through a CSU/DSU, of course!). Note that this connection does not use the Internet in any way—it is not a VPN connection. Dedicated connections of this type are expensive and are only used by organizations that need the high bandwidth and high security these connections provide. These connections are invisible to the individual computers on each network. There is no special remote connection configuration of the individual systems, although there may be



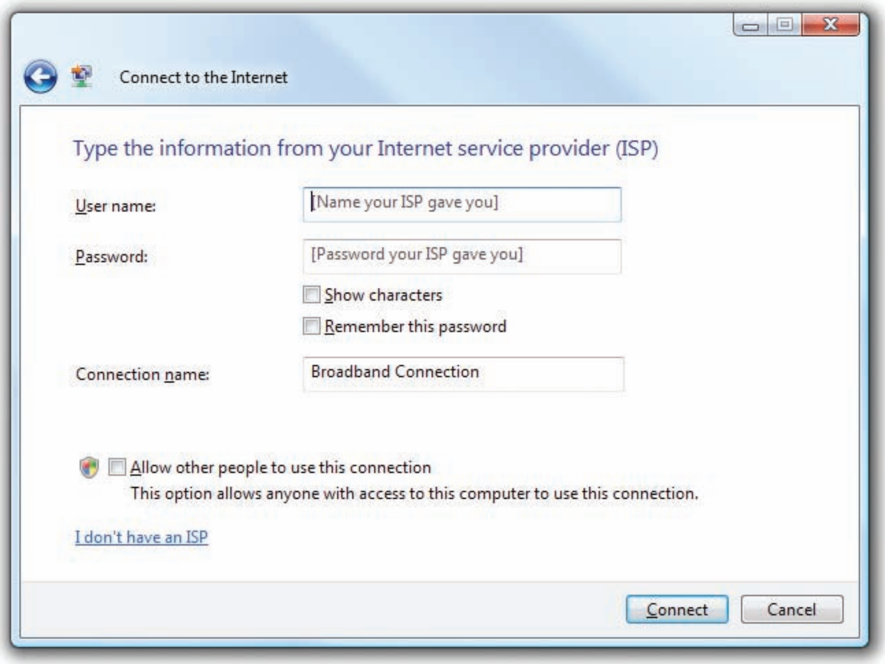
• Figure 14.33 Dedicated private connection

some configuration of DHCP, DNS, and WINS servers to insure that the network runs optimally.

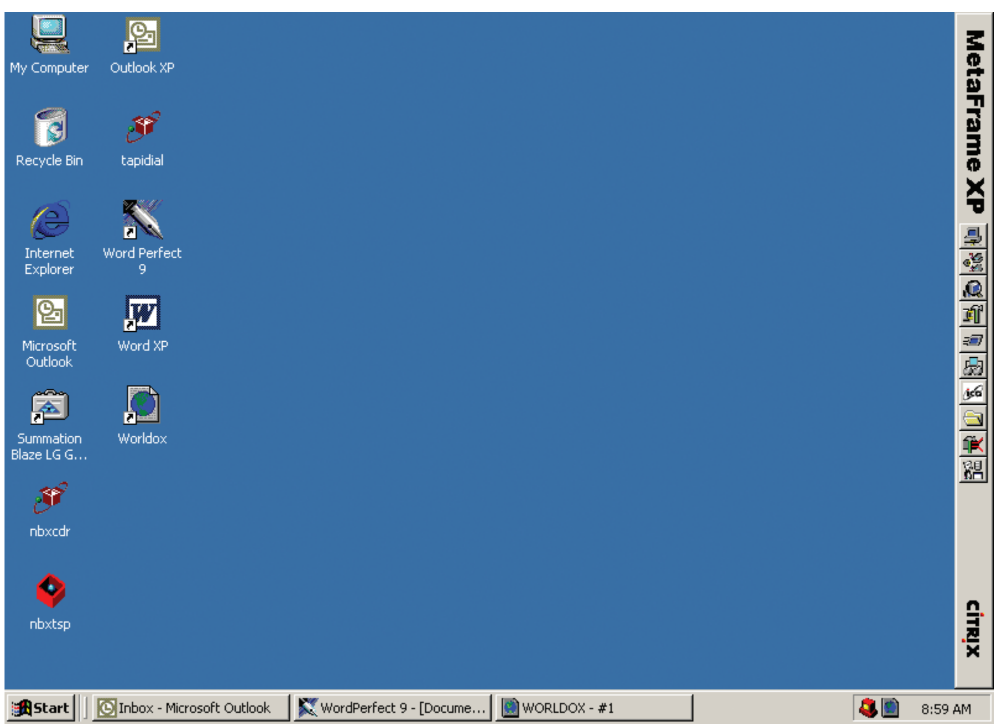
Dedicated connections to the Internet are common today. Cable modems and DSL have made dedicated connections to the Internet inexpensive and very popular. In most cases there is nothing to configure in these dedicated connections but many cable and DSL providers give you a CD-ROM that installs different items such as testing software, PPPoE login support, and little extras such as e-mail clients and software firewalls. Personally, I prefer to not use these (they tend to add a lot of stuff you don't need) and instead use the operating system's tools or a hardware router configured to handle these. Figure 14.34 shows the DSL wizard built into Windows Vista. This program enables you to connect by entering your PPPoE information for your ADSL connection. Once started, these programs usually stay running in the system tray until your next reboot.

Remote Terminal

You can use a terminal emulation program to create a **remote terminal**, a connection on a far away computer that enables you to control that computer as if you were sitting in front of it, logged in. Terminal emulation has been a part of TCP/IP from its earliest days, in the form of good-old Telnet. Because it dates from pre-GUI days, Telnet is a text-based utility; all modern operating systems are graphical, so there was a strong desire to come up with graphical remote terminal tools. Citrix Corporation made the first (arguably) popular (also arguably) terminal emulation product—the *WinFrame/MetaFrame* products (Figure 14.35). Citrix wasn't free, but it ran on any operating system and is a mature and dependable product.



• Figure 14.34 PPPoE connection

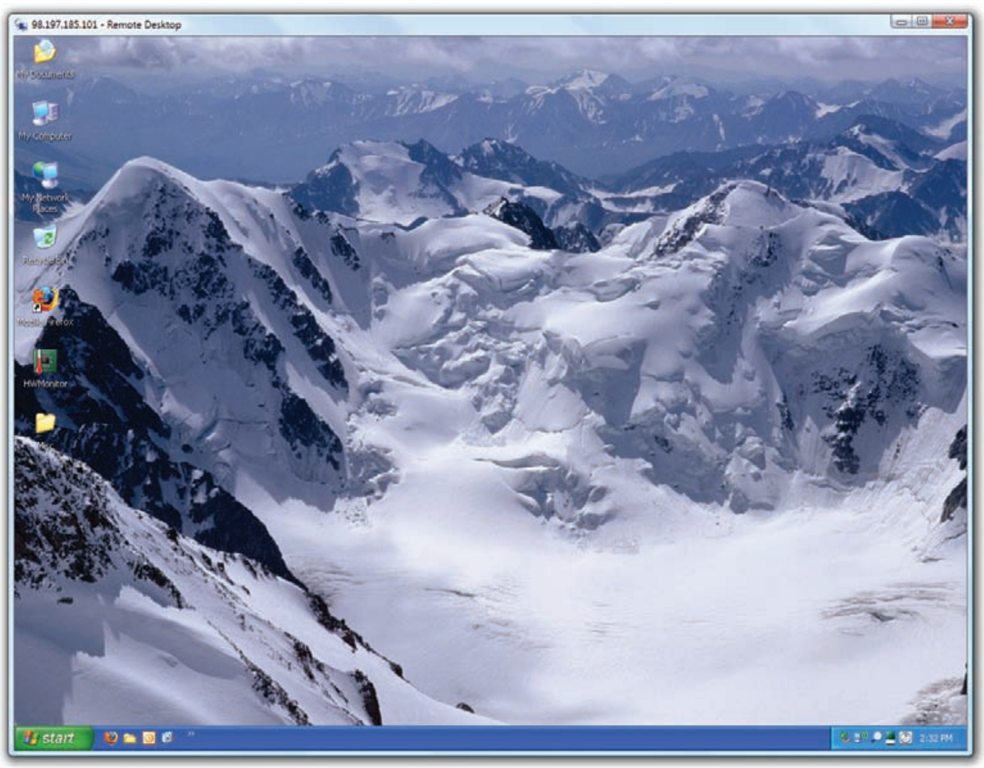


• Figure 14.35 Citrix MetaFrame

All RDP applications run on port 3389 by default.

Remote terminal programs all require a server and a client. The server is the computer to be controlled. The client is the computer from which you do the controlling. Citrix created a standard called Independent Computing Architecture (ICA) that defined how terminal information was passed between the server and the client. Citrix was a breakthrough product—so powerful that Microsoft licensed the Citrix code and created its own product called Windows Terminal Server. Not wanting to pay Citrix any more money, Microsoft then created its own standard called Remote Desktop Protocol (RDP) and unveiled a new remote terminal called Remote Desktop Connection (RDC) starting with Windows XP. Figure 14.36 shows Windows Remote Desktop Connection running on a Windows Vista system, connecting to a Windows 2000 Server.

Unfortunately, Terminal Services only works in the Windows environment; however, a number of third parties make absolutely amazing terminal emulation programs that run on any operating system. The best of these is *VNC*, which stands for virtual network computing (Figure 14.37). VNC doesn't let you share folders or printers, because it is only a terminal emulator. But it runs on every operating system, is solid as a rock, and even runs from a Web browser. It works nicely in Secure Shell (SSH) tunnels for great security, plus it comes by default with every copy of Macintosh OS X and almost every Linux distro. Why bother sharing if you can literally be at the screen? Oh, and did I mention that VNC is free?



• **Figure 14.36** RDC in action

• Figure 14.37 VNC in action

VoIP

Voice over IP (VoIP) is simply using an IP network to transfer voice calls. VoIP's biggest benefit is that is uses an existing network you're already paying for (your Internet connection) to replace another network you're also paying for (PSTN lines). The technology to do VoIP isn't very challenging but making a VoIP system that's standardized so that everyone can use it (and we can still contact those who choose to use PSTN) is quite a bit harder, requiring international standards. VoIP is still a very fractured world but we're getting closer to universally adopted standards so that one day everyone can contact everyone else, no matter what brand of VoIP they use. To do this, there are three important standards you need to know: RTP, SIP, and H.323.

RTP

The **Real-time Transport Protocol (RTP)** is the bedrock of VoIP standards and is heavily adopted. RTP defines the type of packets used on the Internet to move voice or data from a server to clients. The vast majority of VoIP solutions available today use RTP.

SIP and H.323

Session Initiation Protocol (SIP) and **H.323** are competing VoIP standards that handle the initiation, setup, and delivery of VoIP sessions. VoIP takes a lot of special features that are not very common in many other Internet protocols. The biggest one is multicasting. Multicasting isn't in very high demand unless you want to show a number of people a video or want to make a conference call. SIP and H.323 both have methods for handling multicasting.

SIP and H.323 both run on top of RTP. Most VoIP solutions are either SIP/RTP or H.323/RTP

Skype

Almost every VoIP solution available today uses SIP or H.323 running on top of RTP, with one huge exception: the very famous and incredibly popular Skype. Skype was unveiled in 2003 by Niklas Zennström, a Swedish computer guy famous for inventing the Kazaa peer-to-peer file-sharing system. Skype is completely different from and completely incompatible with any other type of VoIP solution: Skype doesn't use servers, instead using a peer-to-peer topology identical to the old Kazaa network. Skype calls are also encrypted using a proprietary encryption method. There isn't a standard method for VoIP encryption at this time, although a lot of smart people are working hard on the issue.

Chapter 14 Review

Chapter Summary

After reading this chapter and completing the exercises, you should understand the following about remote connections.

Describe WAN telephony technologies, such as SONET, T1, and T3

- The majority of long-distance connections that make up the Internet use a unique type of signaling called SONET. The Internet backbone uses technologies designed more than 20 years ago to support telephone calls.
- A multiplexer combines multiple circuits at one end of a connection into a single complex circuit on one wire, then splits the individual connections back out at the other end of the connection.
- A local telephone exchange is a grouping of individual circuits served by a single multiplexer. Exchanges are housed in physical buildings called central offices.
- Multiplexers used Frequency Division Multiplexing to keep individual calls separate.
- Physically connecting two phones on a single circuit is called circuit switching.
- Analog voice calls had to be converted to digital to accommodate travel over long distances. Central offices convert incoming analog calls to digital, transport the digital signal across trunk lines, and then convert the digital signal back to analog for delivery to the destination phone.
- The analog connection from the central office to individual users is called the last mile.
- Converting analog sound into 8-bit chunks 8000 times a second creates a 64-Kbps data stream known as DS0. Every analog voice call is converted to DS0 at the central office, where it is then multiplexed into larger circuits.
- A device that converts an analog signal to digital is a modulator. A device that converts digital signals to analog is a demodulator. A device that does both is a modulator-demodulator, or modem.
- T1 refers to a high-speed digital networking technology while T1 line refers to the physical shielded, two-pair cabling that connect the two ends of a T1 connection.
- A T1 line uses two pairs of wires, one pair to send data and one pair to receive data.
- A T1 line connects to a CSU/DSU at both ends. The CSU/DSU has a second connection connecting the phone company to a customer's equipment. You cannot have more than one CSU/DSU on a single T1 line because a T1 connection is point-to-point.
- Many new routers have a CSU/DSU built into them.
- T1 uses a signaling method called DS1. A DS1 frame is composed of one framing bit and 24 channels. Each DS1 channel holds a single 8-bit DS0, creating 193 bits per DS1 frame. (192 bits from the 24 channels of 8-bit DS0 data samples plus the one framing bit.)
- DS1 frames are transmitted 8000 times/sec for a T1 data transfer speed of 1.544 Mbps. This is split into 24 64-Kbps DS0 channels.
- The process of having frames carry a bit of every channel in every frame sent on a regular interval is called time division multiplexing.
- Because an entire T1 bundle is expensive, many telephone companies allow you to purchase fractional T1 access, or just some of the individual channels.
- A T3 line supports about 43 Mbps and consists of 672 individual DS0 channels. T3 lines are also known as DS3 lines and are used mainly by regional telephone companies and ISPs connecting to the Internet.
- An E1 is the European counterpart to a T1, but carries 32 channels at 64 Kbps for a total of 2.048 Mbps—slightly faster than a T1.
- E1 and T1 lines can be interconnected for international use.
- An E3 carries 16 E1 lines (512 channels), for a total bandwidth of 34.368 Mbps—a little slower than an American T3.

- The CSU part of a CSU/DSU provides protection to the T1 or T3 lines from lightning strikes and other types of electrical interference. It also stores statistics and has loopback testing capability.
- The DSU part of a CSU/DSU supplies timing to each port, converts incoming signals to line code, and frames the format for transmission over the provided line.
- SONET is the primary standard for long-distance, high-speed, fiber-optic transmission in the United States. It is often implemented as a ring for redundancy. SDH is the European equivalent.
- SONET has extensive multiplexing capabilities, such as combining multiple DS1, DS3, and E1 signals into a single huge frame.
- The Optical Carrier (OC) specification defines speeds from 51.8 Mbps (OC-1) to 39.8 Gbps (OC-768) for fiber-optic cables used in networks conforming to the SONET standard.
- SONET uses the STS signal method where the STS payload carries data and the STS overhead carries signaling and protocol information.
- The number at the end of STS, such as STS-1 or STS-3, indicates signal speed. For example, STS-1 runs on an OC-1 line at 51.85 Mbps while STS-3 runs on an OC-3 line at 155.52 Mbps.
- Frame Relay is a packet-switching standard designed for and used primarily with T-carrier lines. Packets are switched quickly, but with no guarantee of data integrity. Frame Relay actually discards frames whenever there is network congestion; however, T-carrier digital lines using Frame Relay have very low error rates.
- Most SONET rings that move voice and data use ATM for packet switching. ATM integrates voice, video, and data on one connection using short, fixed-length cells to transfer information.
- ATM transfer speeds range from 155.52 to 622.08 Mbps and beyond.
- MPLS is a router feature that labels certain data to use a desired connection. For example, a network administrator can specify that all FTP traffic use the ATM connection rather than a secondary link that might be available on the network.
- There are two reasons to use a telephony WAN connection: to get your LAN on the Internet, and to make a private connection between two or more of your private LANs.
- The first step to getting a WAN Internet connection is to have a line physically installed by the telephone company. The second step is to have an ISP provide Internet access via the line.
- The telephone company runs your line to a demarc. The other side of the demarc is where you (or your ISP) installs a CSU/DSU and your router.
- WAN telephony carriers are more dependable than cheaper alternatives, such as cable modem service. A BERT test, available on every CSU/DSU, can verify your T-carrier connection.

Compare last-mile connections for connecting homes and businesses to the Internet

- Dedicated lines are always off the hook, providing a permanent connection, and do not have an associated phone number. Dial-up lines have phone numbers and must dial to make a connection, hanging up when done.
- PSTN, also called POTS, is a regular phone line designed to work only with analog sound and uses an RJ-11 connector.
- Telephone microphones convert analog sounds into electrical waveforms that cycle 2400 times a second. Each individual cycle is a baud. The number of bauds per second is the baud rate.
- PC communications were designed to transmit data in and out of a computer 8 bits at a time. A UART converts 8-bit-wide parallel bits from the computer into 1-bit-wide serial bits to send to a modem for digital-to-analog conversion. The analog signal can then be sent over phone lines. The process is reversed for incoming signals.
- PSTN phone lines sample analog data 2400 times a second. By modulating the 2400-baud signal multiple times each second, faster transmission speeds are reached—up to 57,600 bps (56 Kbps).
- Modems must query each other to determine a common protocol with which to communicate. The European CCITT developed the V standards, which define modem modulation speed and other features. The current standard is V.92.
- The conversion between analog and digital across the last mile resulted in reduced bandwidth. Making the last mile digital overcomes problems introduced by an analog last mile.

- ISDN lines provide a digital connection across the last mile, achieving dependable throughput of 64 Kbps over the same copper wires used by PSTN.
- ISDN consists of two channels: Bearer (B) and Delta (D) channels. B channels carry voice and data using standard DS0 channels. D channels carry setup and configuration information at 16 Kbps.
- A BRI setup includes two B channels and one D channel, providing a total throughput for voice and data of 128 Kbps.
- A PRI setup includes a full T1 line carrying 23 B channels for a total throughput for voice and data of 1472 Kbps (about 1.5 Mbps).
- A terminal adapter acts as the interface between a computer and the ISDN service.
- DSL provides a fully digital dedicated connection. Three versions of DSL are SDSL, ADSL, and VDSL.
- SDSL supports speeds up to 15 Mbps, but most ISPs only provide SDSL up to 9 Mbps. SDSL provides equal upload and download speeds.
- ADSL provides download speeds up to 15 Mbps and upload speeds up to 1 Mbps, though ISPs offer varying combinations of download/upload speeds.
- VDSL is the newest version of DSL and supports both download and upload speeds up to 100 Mbps. VDSL can also run on fiber-optic lines to increase distances.
- All versions of DSL are limited to a maximum distance of around 18,000 feet between a user's demarc and the central office. The central office houses a DSLAM connecting multiple customers to the Internet.
- Because DSL runs over normal POTS lines, it is necessary to filter out the DSL signal on the POTS line. This guarantees your POTS line will continue to work if the DSL fails.
- A DSL modem connects the telephone jack (with the DSL signal) to your computer. A DSL modem isn't a true modem and is more similar to an ISDN TA.
- RJ-11 connectors connect the telephone jack to the DSL modem while RJ-45 connectors connect the DSL modem to the computer's NIC.
- Early DSL providers used bridged connections, but these connections have been replaced by PPPoE so that providers can monitor modem usage and require users to log in with a valid account before they gain Internet access.
- Cable Internet providers offer plans ranging in speeds up to 10 or even 27 Mbps. Download speeds are typically much faster than upload speeds.
- Cable modems use coaxial cable to connect to the head end and use regular CAT 5 or better cabling to connect to the PC. The head end connects to the cable company's network using the DOCSIS protocol.
- Satellite access is available as one-way or two-way. With a one-way connection, you download over the satellite connection but upload over PSTN. Two-way satellite service accommodates both downloads and uploads over the satellite connection.
- Satellite access tops out at around 500 Kbps for download and 50 Kbps for upload, making it slower than both DSL and cable. It is sometimes the only option for remote or geographically challenging areas.
- In an attempt to regain their share of the market from cable providers, some phone companies offering DSL are now offering fiber-to-the-home connections in the form of U-verse (AT&T) or FiOS (Verizon).

Discuss and implement various remote access connections

- Remote access allows users to log onto networks remotely, making files and network resources available to users across the city, state, or globe.
- Dialing into the Internet over an analog phone line and modem is the oldest and least expensive means of connecting to the Internet. Many techs keep a dial-up account as a backup.
- A private dial-up connection, which does not use the Internet, connects a remote system to a private network via a dial-up connection. This requires a remote access server on one end and a client running a connection tool at the other end.
- Dedicated connections are remote connections that never disconnect and can be categorized as either dedicated private connections between two locations or dedicated connections to the Internet.

- Dedicated private connections are usually connected by a high-speed line such as a T1. This direct dedicated connection does not use the Internet.
- Remote terminal emulation allows a user to take over a remote computer as if they were sitting in front of it, as opposed to simply accessing remote resources. WinFrame/MetaFrame (made by Citrix) is a popular terminal emulator.
- Microsoft's terminal emulator is called Remote Desktop Connection, which uses its own Remote Desktop Protocol. VNC is a cross-platform terminal emulator that comes with Macintosh OS X and many Linux distributions.
- VoIP uses an IP network to transfer voice calls. It depends on three standards: RTP, SIP, and H.323.
- RTP defines the type of packets used on the Internet to transfer voice or data between servers and clients. Most VoIP networks use RTP.
- SIP and H.323 both support multicasting on VoIP networks, allowing users to show a video to multiple people or hold conference calls.

Key Terms

Key Term Quiz

Use the Key Terms list to complete the sentences that follow. Not all the terms will be used.

1. A(n) _______________ line has a maximum throughput of 1.544 Mbps.
2. A(n) _______________ is a device that converts signals between analog and digital.
3. It is the job of the _______________ to convert between 8-bit-wide digital data and single-bit-wide digital data.
4. A(n) _______________ signal is defined as digital signal of 64 Kbps.
5. A(n) _______________ combines individual circuits with hundreds of others, creating a complex circuit on a single wire.
6. _______________ is the primary standard in the United States for long-distance, high-speed, fiber-optic transmission systems.
7. In the world of DSL, _______________ provides equal upload and download speeds up to 15 Mbps.
8. _______________ uses an IP network to transfer voice calls.
9. _______________ lines consist of two digital channels over the same copper wire used by regular analog telephones.
10. _______________ is a router feature that can help to optimize network traffic by labeling certain data to use a desired connection.

Multiple-Choice Quiz

1. When an analog sound is converted into 8-bit chunks 8000 times a second, this 64 kilobit per second data stream is created.
   A. DS0
   B. DS1
   C. E1
   D. T1
2. What exists at both ends of a T1 connection?
   A. A frame relay
   B. A CSU/DSU
   C. A digital trunk
   D. A multiplexer
3. This frame type consists of a single framing bit and 24 data channels, each holding an 8-bit DS0 sample, for a total of 193 bits.
   A. DS1
   B. T1
   C. T3
   D. SONET
4. Which line consists of 672 DS0 channels for a total throughput of 43 Mbps?
   A. T1
   B. T3
   C. E1
   D. E3
5. Which standard supports a throughput of up to 39.8 Gbps?
   A. ISDN
   B. VDSL
   C. SONET
   D. MPLS
6. If you purchase a T1 line in the United States, how will packets be switched? (Select two.)
   A. OC
   B. Frame Relay
   C. ATM
   D. BERT
7. What describes the problem with "the last mile"?
   A. The connection from a central office to a user's home is analog whereas the rest of the network is digital.
   B. Users must live within a mile of a central office in order to guarantee quality of service (QoS).

C. SONET connections are limited to a maximum distance of one mile, and connecting central offices via multiplexers is expensive and difficult to maintain.
D. Copper wires that carry analog telephone signals are limited to a maximum distance of one mile.

8. What terms describe a common telephone connection? (Select two.)
   A. ISDN
   B. POTS
   C. Fractional T1
   D. PSTN
9. What marks where the telephone company's responsibility ends and yours begins?
   A. Multiplexer
   B. Demarc
   C. Primary Rate Interface
   D. Bridges connection
10. The CCITT established which set of standards?
    A. Optical Carrier (OC)
    B. DSL (ADSL, SDSL, VDSL)
    C. Data Over Cable Service Interface Specification (DOCSIS)
    D. V standards
11. Which is the fastest ISDN connection?
    A. BERT
    B. BRI
    C. PRI
    D. ATM
12. Sinjay is 200 meters from his ISP's DSLAM. Which DSL version will provide him with up to 100 Mbps of both download and upload speed?
    A. DS3
    B. ADSL
    C. SDSL
    D. VDSL
13. Which protocol is used by cable companies?
    A. MPLS
    B. DOCSIS
    C. PSTN
    D. SIP
14. What is the benefit to using a satellite connection?
    A. It offers speeds faster than both DSL and cable.
    B. The upload and download speeds are always equal.
    C. It is often available in remote locations where DSL and cable are not.
    D. If offers better security than both DSL and cable.
15. Which protocols support multicasting on VoIP networks? (Select two.)
    A. SIP
    B. H.323
    C. RTP
    D. RAS

Essay Quiz

1. Early DSL providers used bridged connections, but now they tend to use PPPoE instead. What is the difference between these connection types and why do you think DSL providers switched?
2. Upon tracing your company's physical T1 line, you find it connected to a box. It appears as though the box has another connection going to your router. What is this box and what does it do?
3. Briefly describe the six types of remote connections that enable users to connect to remote networks.

Lab Projects

• Lab Project 14.1

Many companies, such as Vonage, offer VoIP solutions for home users to replace their analog telephones. Other companies, such as 3Com, offer VoIP solutions for businesses. Research three VoIP solutions and compare them based on the following criteria: Are they targeting home users or businesses? Is long distance included? What is the startup cost? What is the monthly fee? What uptime guarantee is offered? Can emergency calls (911) be made if the network goes down?

• Lab Project 14.2

How much would a T1 cost you? How about a T3? Make a chart listing the provider from whom you could purchase a T1, fractional T1, or T3 and list the services included along with the costs. How do they compare to cable or DSL connections offered in your area?