Chapter 1

Getting Down to Business with VoIP

In This Chapter

- Getting over POTS
- Packetizing plain old telephone service
- Beginning with IPT on the LAN
- Reaching out with VoIP
- Uncovering the TCP/IP model

Technological innovation is hurling itself upon us once again. This time, it's coming in the form of improving the way we communicate, bringing with it new capabilities that change the meaning of the phrase *telephone call.* VoIP (often pronounced "voyp") is the name of this new communications technology.

VoIP, which stands for *voice over Internet protocol,* basically means voice transmitted over a digital network. Well, that isn't technically accurate because the Internet isn't strictly necessary for VoIP, although it was at first. What is necessary for VoIP technology is the use of the same protocols that the Internet uses. (A *protocol* is a set of rules used to allow orderly communication.) Thus, *voice over Internet protocol* means voice that travels by way of the same protocols used on the Internet.

VoIP is often referred to as *IP telephony* (IPT) because it uses Internet protocols to make enhanced voice communications possible. The Internet protocols are the basis of IP networking, which supports corporate, private, public, cable, and even wireless networks. VoIP unites an organization's many locations — including mobile workers — into a single converged communications network and provides a range of support services and features unequalled in the world of telephony.

Technically, IPT refers to telephone calls carried over the organization's local area network (LAN) such as a single building location, a campus-like network, or even a LAN within your home. When IPT crosses from the LAN to the WAN or any other external network, including other LANs operated by the same company at distant locations or the Internet, it becomes VoIP.

In the Beginning, There Was POTS

Before digital networking took off, everyone had to use the one and only *POTS,* which stands for *plain old telephone service* (honestly, it does). POTS runs over a network called the *PSTN,* or *public switched telephone network.* These POTS telephone systems use the tried-and-true method of telephone service known as *circuit-switched.* (See Chapter 2 for more about the history of POTS, the PSTN, and the operation of circuit-switched telephony.)

For customers, the costs related to the regulated circuit-switched PSTN remain much higher than they need to be. Consumers as well as companies that must rely on POTS on a daily basis know what the POTS way of telephony means to their bottom line. The good news is that VoIP is an alternative that can greatly reduce or eliminate POTS-related costs. (Chapter 3 fully details the recurring charges of the POTS way of doing telephony.) VoIP also enhances productivity, leaving more money in the budget to do other things besides pay telephone bills.

From POTS to Packets

VoIP technology enables traditional telephony services to operate over computer networks using packet-switched protocols. *Packet-switched VoIP* puts voice signals into packets, similar to an electronic envelope. Along with the voice signals, the *VoIP packet* includes both the caller's and the receiver's network addresses. VoIP packets can traverse any VoIP-compatible network. Because VoIP uses packets, much more information can be carried over the network to support and enhance your communication needs when compared to traditional telephony methods.

In a circuit-switched network such as POTS, routing is less dynamic than with a packet-switched network. In the POTS world, if a line is down, the call can't go through. In a packet-switched network, multiple routes can be established, and packets can travel any of the available routes. If one of the lines supporting the network is down, the packet can switch to another working route to keep the call up.

With VoIP, voice signals can travel the same packet-switched network infrastructure that companies already use for their computer data. Chapter 7 goes into more detail about dedicated packet-switched networks that support VoIP.

Eye for IP Telephony

VoIP also makes possible other services that older telephony systems can't provide. VoIP telephony services are *interoperable,* meaning that they work well over all kinds of networks. They are also highly *portable,* which means they will work with any IP-enabled device such as an IP telephone, a computer, or even a personal digital assistant (PDA).

IP telephony works by taking traditional voice signals and converting them to a form that can be easily transmitted over a local area network. Thus, the heart of IP telephony is the same as traditional data networking with computers. IP-enabled phones handle the voice-to-data conversion well, but don't be misled — implementing VoIP doesn't mean that everyone has to use IP-enabled phones. The best VoIP providers implement IP telephony in a manner that protects your investment in existing telephone equipment, even if you have analog telephone stations. (You'll find more on this topic in Chapter 10.)

All IP phones have one important thing in common: a built-in network interface card (NIC), just like a computer uses. The NIC is critical for any network device because it provides the device with a physical address and a way to communicate over the network.

The physical address supplied by a NIC is called a *MAC address.* MAC stands for *media access control.* The MAC address uses a standardized address and is usually represented by six hexadecimal numbers separated by dashes. For example, the following is a valid MAC address: 00-0A-E4-02-7B-99.

To support IP telephony, a server is typically dedicated to run the software used to manage calls. Servers are just like personal computers, except they have more memory, speed, and capacity. The server stores the database that contains all the MAC addresses corresponding to all the IP telephone extensions assigned to users. Depending on the size of the LAN and the number of users, you may use more than one server. For example, some LANs running IP telephony dedicate a server just for handling voice mail.

Depending on the size of the LAN, one or more devices known as switches are installed. These *switches* are boxes that have a series of ports into which all LAN-addressable devices ultimately connect. (Examples of LAN-addressable devices include computers, printers, wireless access devices, gateways, and storage devices.) Usually the switches are set up in the communications closets around the LAN, and they operate 24/7. All the switches are interconnected, often with fiber-optic cable.

In a nutshell, all network devices, including your IP telephone, must physically connect to the LAN through a port on a switch.

Calling over a computer network

Voice over Internet protocol is often taken to mean basically what it states: Voice traveling over the Internet. When VoIP was developed, it worked only with the Internet. Today, VoIP works on all other major network types, including those used throughout the corporate sector.

Making internal calls

When you want to call a coworker at your same location, you dial the phone number corresponding to the person's name. The signals are packetized and sent to the managing server, where the packet picks up the MAC address of the person you're calling. Next, the packet is forwarded to the switch, then to a particular port on that switch, and finally to the IP telephone connected to the port. The coworker's telephone rings. When the coworker picks up the receiver or answers the call, a virtual connection is established between the coworker and yourself for the life of the call. IP telephony does all this at lightning speed.

Making external calls

The process of calling a coworker at an offsite location varies only a little. The call is still initiated in the same way. But because the coworker is connected to a different LAN, the local server sends the call not to a switch located on your LAN but through the company's WAN (wide area network). This is where IP telephony technically becomes VoIP.

Each LAN in a multilocation network is connected to the larger WAN. If you're located at the company's headquarters in Pittsburgh, and you call a coworker located at the office in Los Angeles, your call begins as an IP telephony call on your LAN. It then travels from your LAN through a gateway, switch, or router that is programmed to re-packetize your call and encode the VoIP packet with additional information, such as the address for the destination LAN.

TECHNICAL STUFF

Network gurus refer to the process of packetizing your voice telephone call as *encapsulation*. A good analogy for this fancy techno-term is putting a letter into an envelope for mailing. The difference is that these encapsulated packets contain the content of the telephone conversation in digitized form.

To participate in the company's VoIP WAN, each LAN needs at least one edge device, such as a router, a switch, or a gateway. An *edge device* is just that — a device that sits on the boundary, or edge, of your local network and provides a connection to external networks. Depending on the company's network design, these edge devices can even have multiple interfaces that connect them to more than one outside network. The edge devices take care of all the IP telephony traffic going off-LAN by encapsulating the signals into packets, encoding the packets with the correct addressing information, and forwarding the packets out onto the WAN, where they make their way in a packet-switched manner to their respective destinations.

When the packets arrive at the destination LAN, the edge device on that LAN breaks down the VoIP packets and forwards them internally to the server that manages IP services. From this point, the rest of the process is similar to IP telephony services described in the preceding section: The phone rings, the person being called answers, and a virtual circuit is established between the caller and the receiver.

Gaining Flexibility with VoIP

VoIP is not just about making and receiving telephone calls; it's about a whole new way of communicating. Sure, it includes telephone calls, but there is so much more to the VoIP telephony picture. VoIP integrates most if not all other forms of communication. You can even run videoconferencing to your desktop.

With VoIP, your company enjoys increased productivity and customer satisfaction. These improvements are typically realized through the flexibility offered by enhanced calling features. A few calling features, such as voice mail and call transfer, have been around in the POTS world for quite some time. On the other hand, integrating data, voice, and video applications to run over a single network and work with wireless phones are more recent innovations made possible by IP telephony.

Following are some enhanced calling features made possible by IP telephony:

- **Vemail:** Before IP telephony and VoIP, you accessed voice mail through a telephone and accessed e-mail through a computer. With VoIP, you can read your voice mail on your computer screen and listen to your e-mail through an IP-enabled telephone. The new term for this converged feature is *vemail* (pronounced "v-e-mail").
- **Web surfing:** Because VoIP operates with the same set of IP rules and protocols that support Web-based applications, it is possible to access the Web with an IP-enabled telephone. If you have an IP telephone with a large enough screen, it can display Web pages or a list of your favorite Web links. For instance, you could use your phone to view your stock exchange trading status or the current weather forecast.

In an IP telephony world, these calling features (and many more) are available with no monthly recurring charges. VoIP, with all of its many benefits, is quickly replacing traditional POTS-based technologies. VoIP is even becoming a superior replacement for many former computer-only applications.

One of the big stories with VoIP is the many new and exciting features that increase your ability to be agile and mobile. You no longer have to say "I've got to get to a phone!" VoIP can be on your desk, computer, mobile phone, or PDA. It can be hardwired or have no wires at all. This flexibility is astounding to those familiar with traditional telephony.

TIP

If you have a mobile user base, be sure to check out IP soft phones. A *soft phone* is software that works on a laptop computer or pocket PC and provides most of the functionality of a traditional desk phone. If a user can connect to a network, the soft phone provides a way to reap the benefits of IP telephony regardless of location.

Looking at the TCP/IP Model

Many people marvel at the very thought that the POTS method of placing telephone calls can be replaced by a technology that essentially runs on the computer network. They are also startled by the many new and exciting features that come with VoIP. However, people also question how VoIP can possibly work and are a bit suspicious about whether VoIP can really live up to all the claims.

The answer can be found in the very same model that has been supporting data-only networking since the inception of the Internet more than twenty-five years ago: the TCP/IP model.

Pronounced "t, c, p, i, p," the model uses a five-layer approach to networking. TCP/IP is adapted to enable it to also support VoIP. TCP/IP has proven to be just as effective with packetizing telephony as it has been for many years with packetizing computer data.

To fully understand VoIP, it pays to know a little about the technical underpinnings that make it work over the network of your choice. In this section, I describe the layers of the TCP/IP model in relation to computer networks. Then I insert into this content the parts that change when TCP/IP supports VoIP.

TCP/IP layers

TCP/IP is first and foremost a group of networking protocols. *Protocols* are the rules that govern how network traffic gets packaged electronically for transmission over a network. Some TCP/IP protocols are used strictly for data networking, some are used strictly for VoIP telephony, and some are used by both data and VoIP. Each protocol corresponds to one of five possible layers that make up the TCP/IP model:

- **Application:** Special protocols at this layer ensure the quality and deliverability of VoIP packets.
- **Transport:** The user datagram protocol (UDP) at this layer transports the VoIP packets from start to finish, which in this case means from caller to receiver and vice versa.
- **Internetwork:** At this layer, IP addressing is added to the packet. Every VoIP phone or computer acting as a VoIP phone gets a unique IP address that routes delivery of VoIP packets to and from the caller and receiver during the life of the call.
- **Network interface:** At this layer, MAC addressing is added to the packet. (The MAC address is supplied by the NIC required for all network devices.)
- **Physical:** This layer converts all packets to electro or electro-optical signals to be carried over the local or external network.

Each layer is associated with one or more protocols. A packet must traverse all five layers: once when the packet is sent and again when it is received.

Basically, the VoIP packet originates with the caller. The packet travels down all five layers on the caller's side of the network and gets packaged with the correct protocols at each layer. After the packet reaches the lowest layer, the physical layer, it is sent over the network to its destination. When the packet reaches its destination, it makes its way up through the layers and gets unpackaged. When it reaches the application layer of the receiver, the packet is translated into a voice signal that the receiver hears.

TCP/IP differences

TCP/IP protocols are applied a little differently depending on whether you have a traditional data packet or a VoIP packet. Figure 1-1 illustrates the packet breakdown and corresponding layers involved in a TCP/IP network connection for a standard Web application, which uses a traditional data packet.

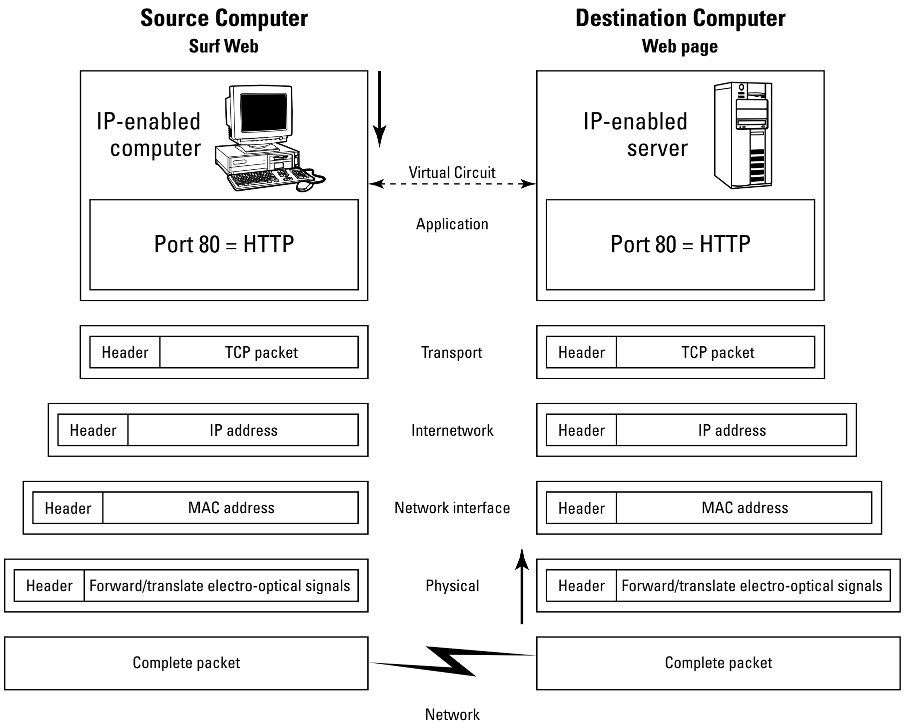

Figure 1-1: Passing a data packet through the TCP/IP protocol layers.

Note that the transport layer of the packet uses the familiar TCP protocol to construct the packet exchanges between the source computer and the destination computer (in this case, the Web server).

Figure 1-2 illustrates the layered protocol stack shown in Figure 1-1, but this time applied to a VoIP call. Note the *header* at each layer's version of the packet (except the application layer). The various headers identify what layer the packet is on during its travel from caller to receiver.

If you compare Figures 1-1 and 1-2, you notice two protocol differences, at the application and transport layers of the TCP/IP model. Other than these differences, a great deal of symmetry exists between voice and data using TCP/IP. That's why VoIP can behave like a telephone while delivering many computer-related functions.

Application layer differences

The first difference between the VoIP implementation of TCP/IP and the traditional data implementation is in the application layer. In a VoIP call, the application layer utilizes the following three protocols:

- **NTP:** Network time protocol. This protocol enables timing, which helps ensure that the signals are transmitted and received within the proper timeframe to assure quality.
- **RTP:** Real-time transport protocol. This protocol provides end-to-end network transport functions for digital voice signals encapsulated in the VoIP packet.
- **RTCP:** Real-time transport control protocol. This protocol monitors voice signal delivery and provides minimal control functions to ensure the delivery of packets.

All three of the application layer protocols combine, at nanosecond speeds, to deliver VoIP voice packets.

Transport layer differences

The second difference between the traditional data implementation of TCP/IP and the VoIP implementation is in the transport layer. The lion's share of computer data networking uses the TCP protocol at the transport layer. For VoIP, the transport layer uses UDP, user datagram protocol. (UDP is used also for real-time videoconferencing networks.)

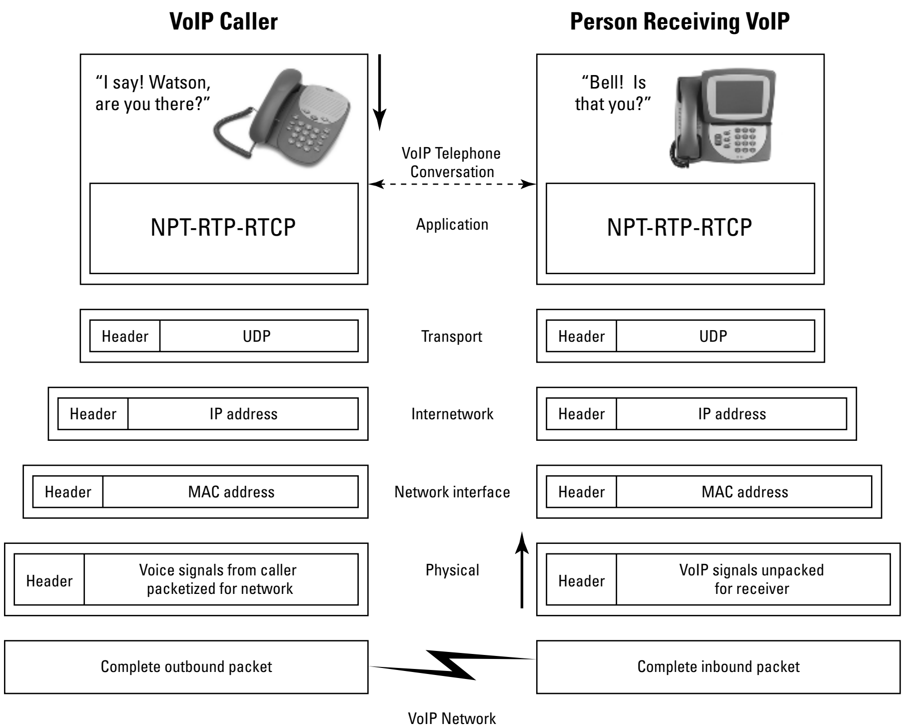

Figure 1-2: Passing a VoIP call through the TCP/IP protocol layers.

TCP is slower than UDP, but it provides guaranteed delivery of its computer data packets. Keep in mind that we are measuring speed here in nanoseconds. Even if it takes a long time for the packets to reach their destination computer, eventually TCP ensures delivery.

Because voice is a real-time application, it is more important that the voice packets get to the receiver as quickly as possible. That is why UDP is by far the hands-down favorite to provide the transport layer for VoIP networks.