Chapter 5

Getting Switched

In This Chapter

- Supporting VoIP over the PSTN
- Controlling millions of calls
- Understanding quality of service issues

In previous chapters, I discuss the beginnings of the public switched telephone network, or the PSTN. I also outline some history of the PSTN. Perhaps the most significant piece of history was the development and deployment of digital telephony services in the early 1960s. Since then, the PSTN has experienced other significant transport line and service developments, including ISDN and DSL. In the consumer market, these newer transports and services are provided over the existing POTS line, providing greater bandwidth.

Today the PSTN is also known simply as the *switched network*. This chapter provides you with the details you need to understand the other transports in the switched network. Also covered are VoIP services available on the PSTN and how calls are controlled on the PSTN.

Understanding How the PSTN Supports VoIP

No other network in the world can compare to the reliability of the U.S. switched network. (Granted, a handful of disasters have disrupted PSTN services in specific regions, but these are the exceptions, not the rule.) Such reliability, however, comes at a high price: The cost of the switched network, particularly recurring (per-minute) charges, is the highest in the world. Regional toll and international calling using the PSTN are the most highly regulated switched network services. This means high regulatory fees in addition to recurring usage charges. But VoIP greatly reduces and may eliminate these types of charges.

The PSTN-VoIP baseline

The quality of service and high performance of the switched network have rarely been in question in the past fifty years. It is only natural that this quality, which we've come to accept and expect, would be considered a baseline, or standard, that VoIP needs to live up to.

Living up to the quality standards of the PSTN presents a problem for VoIP. Remember that VoIP is unregulated, which means it has no enforceable quality standards. Quite frankly, VoIP can't meet the level of quality set by the PSTN in each and every network design, and therefore VoIP is not for everybody. This will change as VoIP replaces the traditional telephony services and customers demand acceptable quality standards.

As it is now, VoIP runs best when implemented on a private, dedicated network. With this in place, any company can utilize any of the other transports to place and receive telephony calls at low or no cost. (The dedicated network options are covered in detail in Chapter 7.) In this chapter, I clarify the three switched transports (POTS, ISDN, and DSL) that may be used to deliver reasonably good quality VoIP to the consumer market, to smaller companies, and to those in the home seeking to connect through their company's larger corporate network.

The POTS transport

As you already know, POTS is a transport that runs through the circuit-switched PSTN. All transport lines in the PSTN have a circuit-identification number, which is either all numeric or alphanumeric. For example, a POTS telephone number has an area code, a prefix, and a suffix that correspond to the physical circuit and the lines that make up that circuit.

Although POTS does not run VoIP directly, POTS is required for the later digital transport, DSL. Because of the need for a POTS line to have a circuit ID, you must have a POTS line established before you can order broadband DSL.

DSL runs on the same line as your POTS telephone service. This raises an interesting question. If you are looking to get broadband DSL so you can run VoIP, do you need to have the added cost of the POTS service? For now, you do. I expect this will change as competition heats up and POTS carriers continue to lose consumers to the broadband cable carriers. (More about this dilemma in a moment, in "The DSL transport.")

The ISDN transport

Work on developing ISDN began in the 1970s but would not be sold to the bandwidth-hungry customer until the early to mid 1990s. Many said it was too little too late, and the consumer market for ISDN never took off. After the news of the first VoIP telephony call over the Internet spread in 1995, a renewed interest in ISDN emerged for a short while. But by this time our attention was turned to the emerging DSL technology first deployed in 1998.

The eventual ISDN standard provided for two flavors of ISDN: Basic Rate Interface (BRI) and Primary Rate Interface (PRI). The ISDN standard defines the basic unit of bandwidth as a B channel, which provides 64 Kbps of bandwidth. *B* stands for *bearer* channel, which is another name for the channel that carries POTS calls over the PSTN.

In the digital world, all transport lines provide one or more channels, just like your cable television provides different channels to carry various programs. Unlike POTS calls, ISDN calls originate in digital form and travel over the switched network to the destination being called.

BRI

By the time ISDN rolled out to the public, other transports and services had evolved that provided more bandwidth without the complexities and cost factors associated with the BRI flavor of ISDN. Some BRI customers are still out there, but they are usually in the process of converting to DSL, cable modem, or some variation of wireless technology. The monthly recurring charges for BRI transport services are considered exorbitant — and are even higher than POTS recurring charges. As a result, BRI has generally dropped out of the VoIP picture.

PRI

The PRI implementation of ISDN has proven to be a cost-effective transport option for companies with a single location seeking to run IP telephony on their LAN. PRI supports local POTS calls into the PSTN.

PRI is the one ISDN transport that has remained useful for supporting VoIP. It provides a customer with 23 B channels of digital bandwidth. In addition, the carrier configures the PRI to have one 64 Kbps D channel, which is used to manage the line. The industry summarizes the PRI aggregate digital bandwidth as “23 B + 1 D.”

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In general, a VoIP call is made over a PRI transport as follows:

1. **A caller uses their IP-enabled phone to dial a number.**
2. **The number initiates a packet on the LAN. For inside calls, the packet stays on the LAN. For outside calls, the packet is switched to the PSTN gateway.**

   A *PSTN gateway* is a hardware device whose basic function is to sit between two dissimilar networks and translate the packets that pass through into the format required by the destination side. Many different levels of gateways include other functions, such as routing and network management.
3. **The PSTN gateway may use a PRI transport to connect directly to the PSTN. When the gateway gets the call request, it allocates a B channel on the PRI transport, initiates the call, and passes the call to the PRI transport.**
4. **After the call is on the PRI, it is translated for operation over the PSTN as a circuit-switched call.**
5. **When the call reaches the destination telephone, a circuit is established between the caller and the receiver for the life of the call.**
6. **When the call is complete and either party hangs up, the PRI B channel is returned to the channel pool controlled by the PSTN gateway.**

Figure 5-1 provides an illustration of a network layout that could support this call scenario using a PSTN gateway.

Because PRI is a switched transport, it easily connects to the PSTN, which is also switched. PRIs are compatible with the call control used on the PSTN to manage POTS-related calls. (Call control is discussed at some length later in this chapter.)

In effect, the PRI is capable of handling twenty-three individual telephone calls simultaneously, delivering an aggregate bandwidth capacity of 1.472 Mbps over the PSTN. In addition, the PRI transport can be used for computer data as well as videoconferencing. The PRI transport continues to be employed because it is effective, compatible with the PSTN, and cheap, averaging about $275 to $425 per month. Local recurring telephony charges still apply, as they do with POTS or any off-net VoIP call.

The PRI is not a good fit for every VoIP network environment. But for a single-location LAN running IP telephony (VoIP on the LAN) with separate Internet access, the PRI is an effective solution for gaining access to the PSTN for your network.

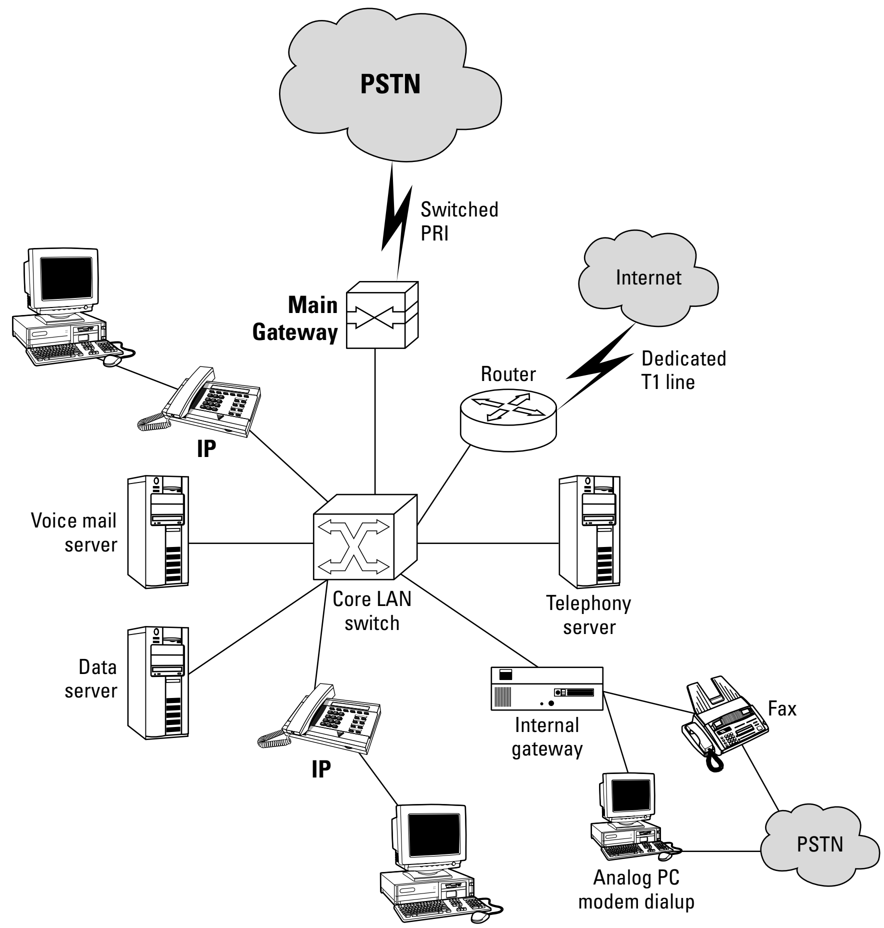

Figure 5-1: Making VoIP calls over a PRI transport.

The DSL transport

DSL stands for *digital subscriber line,* a form of broadband service primarily used by the consumer market and those who telecommute from home. (DSL is also used by some companies for data-only Internet services.) DSL has become one of the most popular transports for running VoIP in the home.

TIP

Many criticize the fact that DSL customers must have a POTS line with an assigned phone number. A counter argument is safety — and backup phone service. If you keep your POTS phone plugged into your broadband service, you can call 911 directly. (Consumer VoIP carriers are only now being required to deliver real-time local 911 services, but it will take at least two to three years for the service be uniform and operational.) In addition, if your VoIP provider's network should go down or lose power, you can still make telephone calls using your POTS phone. The main benefit of having VoIP in the home at this time is to eliminate recurring toll charges, not to replace 911 or local calling.

So what can a user expect to pay for getting the benefits of VoIP in their home? Table 5-1 details the typical costs of home VoIP service, when running over a DSL line.

Table 5-1 Costs for VoIP over a DSL Transport

Item	*One-Time Charges ($)*	*Recurring Charges ($)*
POTS line	65	25
DSL service	100 (waived)	50
VoIP adapter	225 (waived)	0
VoIP service plan	35	50
Total	100	125

As you can see, it doesn't take much to financially justify having VoIP service in your home. This is particularly true if you are already using DSL as your Internet access method. Thus, the monthly cost that is strictly VoIP is the one-time service-plan activation charge of $35 — and the monthly recurring charge of $50. If your monthly toll usage charges for intralata, intrastate, interstate or international run higher than this amount (or more than $600 per year), you are money-ahead by getting VoIP.

Controlling Calls

You might wonder how the PSTN manages to control millions of circuit-switched telephone calls each day. Most people know that their telephone connects to the public telephone network, but they don't know what happens to make it work beyond that point.

A critical part of the PSTN infrastructure is making each telephone call successful. To do this, there needs to be some mechanism by which a call is initiated (sometimes referred to as *call setup*), maintained, and terminated (sometimes referred to as *call tear down*). Call setup establishes a channel over which communication can occur, and call tear down releases that channel so it can be used by a different call. These steps — setup, maintenance, and tear down — are collectively referred to as *call control,* or *CC* for short.

Call control does not apply to VoIP on-net calls. When you are operating within the PSTN or are initiating a call on a VoIP network and the destination is on the PSTN, call control becomes relevant. In this section, I discuss various aspects of call control, with particular emphasis on how it affects VoIP.

Signaling system 7 (SS7)

We have come to expect a certain degree of quality with POTS telephony. That quality is ensured by the use of signaling system 7 (SS7), the call-control protocol. SS7 assigns a separate channel in parallel to each PSTN call and provides call control information through this separate channel. That information is responsible for maintaining the call so that the line doesn't "go dead" while you are talking with someone and so you don't get other distractions such as line fade or crosstalk.

SS7 is also responsible for the high-quality accounting services that support the billings and usage involved with every PSTN telephone call. In summary form, customers are able to manage their aggregate monthly telephony usage costs, assess their monthly recurring line access costs, find out what regulatory fees are being applied monthly, and even determine who is and isn't using the telephony system and what they are using it for. SS7 makes all these reports possible.

A good analogy for SS7 is the air traffic control network in the United States. To control the thousands of hourly flights, a parallel network provides real-time information that the air controller staff uses to control the flights in their respective regions. Figure 5-2 illustrates the parallel nature of SS7 in relation to regular phone calls.

Call control and VoIP

Call control on the PSTN is one of the reasons why today's quality of service is so good — and one of the historical impediments to VoIP. The obstacles presented by call control were overcome through improved technology such as PSTN gateways and newer call-control methods that support dedicated packet-switching calling into the PSTN.

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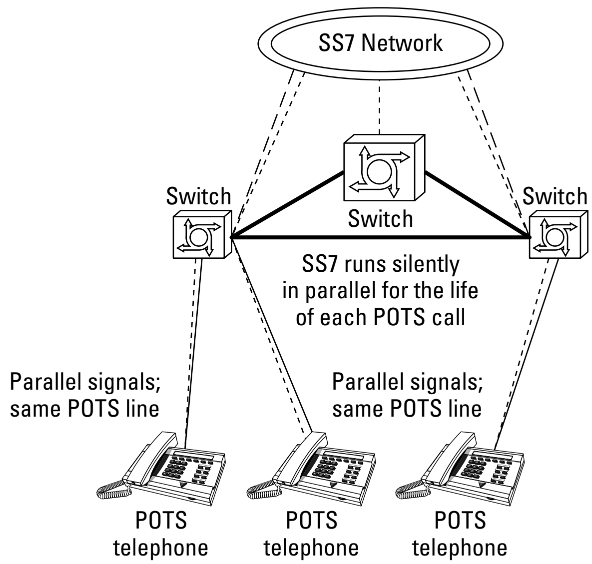

Figure 5-2: SS7 operates in parallel to regular phone calls on the PSTN.

Achieving these newer forms of call control, however, is both complex and costly. For this reason, many companies are opting to design their networks to use their private dedicated transports to support on-net VoIP telephone calls as much as possible. Companies with large multilocation networks that cover the entire country (or even just one or two regions of the country) can design their VoIP networks to route calls destined for other calling areas as far as possible over their private network before going off-net to the PSTN. This type of design optimizes private network use, reduces or eliminates costs incurred with the PSTN, and still provides the QoS benefits of the PSTN.

Only in cases where the company has exorbitant local recurring PSTN charges does it make sense to consider changing how calls into the PSTN are controlled. When a call must go off-net, converting from the packetized VoIP network to the public-switched network is at the very heart of converging the calls.

Delays and errors

When converting to VoIP, another call-control consideration pertains to controlling network errors and delay. In data networks, delay is not a big deal because the network can compensate for it by reassembling packets at the destination. If that fails, a well-designed network can request that the data be retransmitted.

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Even though all the error correction and retransmission can be performed at top network speeds, data packet speed transmission requirements are much slower than those required for transmitting VoIP packets. If I don't get an e-mail message for a couple of minutes, this is no major disruption. But if there is a delay in a real-time voice transmission, it messes up the quality of the conversation.

To work at the same QoS levels as POTS, VoIP demands real-time speed that exceeds the requirements of traditional data networking. That is why you can't adequately operate VoIP on networks that run at less than broadband speed. Unlike data packets, VoIP does not retransmit packets. What you hear (and see, when transmitting video) is what you get.

A big part of the need for minimal broadband speeds pertains to the underlying requirements for network design. As covered in Chapter 1, the computer network uses the TCP/UDP/IP networking model and its related protocols to support both data packets and VoIP packets that travel the network. Keep in mind that any network connection can and usually is several network connections strung together to ultimately provide the end-to-end connection. As a result, the design of the network used to support VoIP is critical to the QoS that you have with VoIP telephone calls.

Quality and VoIP

When we talk about various forms of telephone quality of service (QoS), it is understood in the industry that *toll quality* means the highest form of telephony service quality controlled by SS7. One of the largest problems with early VoIP was QoS. Connections could be made over the PSTN and into the Internet, but what callers heard was far from what we've become accustomed to with the toll quality of POTS. Since then, we've learned that VoIP QoS comes down to controlling specific errors that are commonplace on VoIP networks. Today, all of those errors can be controlled through network design.

REMEMBER

VoIP telephony service using DSL is said to be near toll quality. With a private, dedicated approach to VoIP, your company can achieve toll quality (or better) VoIP service.

VoIP hardware and software is improving to the point where error rates will soon be a thing of the past on all network types. Even so, it is important to understand the three factors that can affect QoS on a VoIP call:

- Network delay
- Poor compression
- Signal attenuation

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Network delay

In 1995, the delay for a VoIP call using the Internet ranged from 400 to 4000 milliseconds (ms), or 4/10 second to 4 seconds. This delay was quite noticeable in a conversation and was typical when using VoIP over POTS lines. Fortunately, since 1995, network options have improved, which has reduced delay and thereby improved quality.

Delay continues to be a factor in VoIP network design and management. It is something that network professionals watch for continuously. Actual delay depends on the type of network access, the overall distance between the caller and the receiver, the total number of users on the network, the network type involved in the connection, and even the equipment used.

Leading IP telephone manufacturers and VoIP carriers use a benchmark of 150 ms as the maximum acceptable delay. That delay metric, by the way, was established long ago as the delay benchmark for POTS.

Closely related to delay is a factor known as *jitter*. Whenever VoIP packets are received outside the expected window of time (delayed), jitter occurs. Jitter may be caused by timing delays introduced by equipment or transport failures, increased network traffic, or changes in the configuration of the network. A *jitter buffer* can be used to store packets as they arrive at the receiving end; the packets are then distributed to the destination VoIP telephone in the correct order. In this way, delayed VoIP packets don't disrupt the conversation.

Poor compression

VoIP converts the sound of your voice into packets of data, sends them across the network being used as the transport, and then reconstructs them into sound at the receiver's end of the network. The sound information, often called the *payload,* is put together with overhead information identifying where the packet should go to create the final packet transmitted over the network.

VoIP often uses compression techniques to reduce the total size of each packet. The benefit of compression is that it increases speed and optimizes available bandwidth. As a result, there is less delay and higher QoS.

VoIP is not regulated and is still rather new, so compression techniques used by consumer-market VoIP carriers are not standardized and are still being developed.

VoIP carrier service providers can be differentiated in terms of service quality relating to compression and network speeds. How much of a commitment will the carrier make to ensure a high quality of VoIP service? You want to read in your service level agreement (SLA) how much speed and bandwidth you can expect. Reputable carriers usually list 256 Kbps as the minimum speed and as high as 1.536 Mbps as the maximum speed. If they state that it is "best effort" within this range, that is not so bad. If they state nothing or give no range, move on to a different carrier.

Signal attenuation

Signal attenuation is the degradation of a signal over distance and time. Did you ever drive out to the country or into the mountains with a radio on? As you get farther and farther from the source of the signal, the radio fades out or you hear more static.

This is similar to signal attenuation over the VoIP network. VoIP packets are represented on many transports as a change in voltage, and that voltage can degrade over time and distance. By the time the packets arrive at their destination, the signal inside the packets is no longer in its original form. If attenuation is very poor, the person you are calling may not even be able to distinguish who you are or what you are saying.

Attenuation can be reduced or eliminated by improving the speed at which packets are delivered on the network. One way to fight attenuation is with compression, discussed in the preceding section. The better the compression, the better the attenuation.

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