Chapter 1

Reviewing VoIP Basics

In This Chapter

- Seeing what makes up a call
- Separating the fact from fiction
- Gathering the hardware you need
- Sending non-voice transmissions
- Migrating phone numbers or getting new ones

Voice over Internet Protocol (or VoIP, as it's more commonly known) is truly a disruptive technology, bringing in new possibilities while departing from traditional telephony in structure. It packetizes normal voice phone calls and transmits them over the Internet, using the same sort of path on which you send and receive e-mail and Instant Messages, or surf the Web. VoIP has changed how network technicians, engineers, and programmers view telecommunications, how it's transmitted, where it can be delivered, and the lack of flexibility in traditional telephony. It empowers small business to build its own phone systems by using open source software, giving it the power to add features, use multiple carriers to save money, and turn to troubleshooting tools previously reserved only for carriers.

While the technology has evolved, the market has responded. Years after VoIP was rolled out within the networks of long-distance carriers in America, it has gained a foothold in the telecommunications market, but it remains an enigma to many. Many people still hold on to their pre-conceived notions of what VoIP is and how it works.

This chapter explains the basics of VoIP, and it also covers what hardware you need to use it, the challenges of the technology, and some of the added hurdles that it creates.

Parts of a VoIP Call

VoIP isn't one single protocol or software package that converts your analog or digital phone call into something that can run over the Internet. *VoIP* is a group of specialized software elements, each performing a specific task.

Every phone call (whether VoIP or non-VoIP) has two basic components, each using an allotted amount of bandwidth to do its specific job. These call components are

- **The voice portion:** Also called the *media* of the call or the *payload.* The digital, analog, or VoIP representation of your voice is transmitted through the voice portion. Without this part of the call, you can't hear the words, laughs, sighs, or . . . anything from one end of the call to the other.
- **The overhead:** Where everything that isn't the voice portion of the call is transmitted. It handles the housekeeping and maintenance of a call. In this section of bandwidth, the messages to establish the connection of a call are transferred (called *call setup*), as well as the mundane task of clearing a call through a network after both parties hang up (called *tear-down*). This overhead section can also handle a variety of other tasks, such as transmitting codes that translate as ringing, busy signals, or recordings about failed calls (for example, if you dialed a disconnected number). This portion of the call also transmits your Caller ID, as well as the connect and disconnect signals used to begin and end billing on your call.

Figure 1-1 compares the voice and overhead portions of a standard analog and a VoIP phone call.

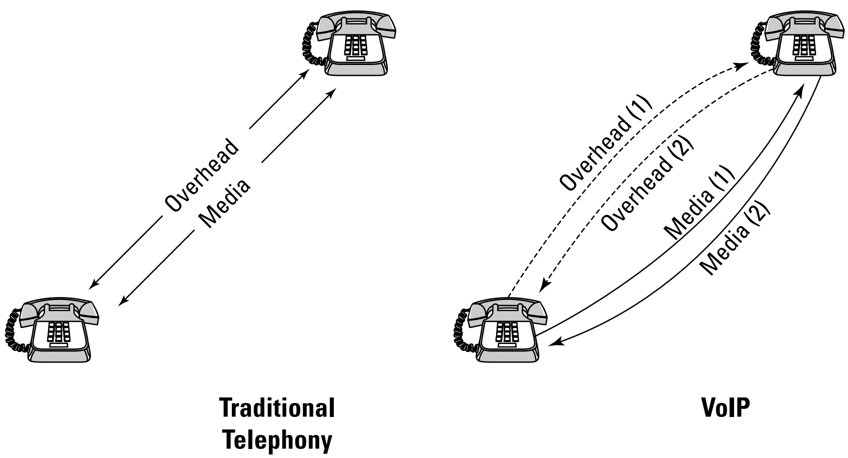

Figure 1-1: Analog call and VoIP call comparison.

Figure 1-1 shows how the overhead and media portion of a traditional call are located in a contiguous section of bandwidth from start to finish. When the individual calls enter the network of your long-distance carrier, your call is then routed into a larger circuit, along with hundreds and thousands of other calls. All the calls have a similar structure, in which the bandwidth is portioned out for each element of the call from point to point along the way. The overhead of the call handles all the call setup and tear-down information from both ends of the call, and the media portion of the bandwidth similarly handles transmission of the speech from both ends. While your call is routed from your local phone carrier to your long-distance carrier, and on to the local carrier that services the number you dialed, both elements of the call continue to be bonded together.

Technically, not every non-VoIP call has the voice and overhead portions of the call sandwiched next to each other from start to finish. Several different protocols are used to send and receive traditional telephony calls:

- In standard loopstart, groundstart, or E&M wink circuits, Figure 1-1 is an accurate depiction of how a call is transmitted. All of these protocols transmit calls while maintaining the conjoined bandwidth of 64 Kbps for the combined overhead and the voice portion. The unique nature of these three protocols are in the Glossary.
- If the call is sent by using ISDN, then the overhead is aggregated on one channel of the circuit, not as a portion of the single channel. The overhead is stripped from the individual calls and given 64 Kbps of bandwidth instead of the minimal room given on a standard telephony call. This allows it to perform all its required duties for many calls, and has additional features that aren't available with loopstart or groundstart calls.
- If the call in Figure 1-1 is sent with SS7 signaling, then it has some of the flexibility of VoIP and a somewhat similar structure, but the overhead is still a continuous stream of data between you and your carrier, not the transmission of overhead packets on an as-needed basis that you see with VoIP.

  If you're interested in finding out more about all these other signaling types, I recommend *Telecom For Dummies* (Wiley Publishing, Inc.), by yours truly.

VoIP is structurally different from a traditional telephony call in three ways:

- The overhead of the call isn't bonded to the media of the call, so the overhead and media can be transmitted via completely different routes.

  Not only can the overhead of the call find its own way from end to end, it can also be transmitted between different endpoints than the media of a call. The detached nature of the overhead allows for much greater flexibility in call transmission as the voice portion of the call can be easily redirected to a new location while still maintaining the overhead between the same two points.

- The overhead of the call (depicted as dashed lines in Figure 1-1) isn't a constant flow of information in a VoIP call — it's sent only when needed.

  Traditional telephone lines are constantly checking in to both ends of the call to ensure that there aren't any problems on the line. VoIP is a much more self-confident protocol — it sends the call and transmits additional information to manage the call only when necessary (such as when the call needs to be re-directed or someone hangs up the phone, ending the call).
- All media and overhead streams can take their own paths between start and end points. Traditional telephony assigns a section of bandwidth for the overhead and the media transmission of the call. Traditional telephony uses 64 Kbps of bandwidth for every phone call. Once the route path is established, both ends of the call use the same 64 Kbps for the bi-directional transmission of overhead and media. VoIP uses a different design that allows each stream of media or overhead to choose its own path to the destination IP. Because of this stream freedom, Figure 1-1 identifies media and overhead paths with individual directional arrows, rather than the two arrows shown on the traditional telephony call.

These three structural differences allow VoIP calls to be managed and routed in ways a traditional telephony call never could be. VoIP allows you to resend a call from its initial start or end point to a new start or end point. Figure 1-2 shows how a VoIP call can be redirected to another destination. In the example, a call between a remote phone and a business's main office is then transferred to a remote office.

This traditional mindset about call forwarding has two main problems:

- **You're using a lot of unnecessary bandwidth.** By accepting the call into your VoIP server at the main office and generating a new call to the remote site, you're using about 168 Kbps of bandwidth for the two sets of media streams and another 62 Kbps for the two sets of overhead streams. So, this one transmission uses a total bandwidth of approximately 230 Kbps. If you forward calls often, they can start to strain your available bandwidth.
- **You're using hardware that can fail or add to the latency of VoIP packet transmissions.** Every server interacting on a VoIP call adds a small amount of *latency* (delay), giving you another variable that you have to consider when troubleshooting calls. Avoid any latency that you can on VoIP calls, such as interaction with unnecessary servers, especially if either network is already over-taxed or you're transmitting faxes or key pad touch tones (which are highly sensitive to latency).

> Problems with intermediary VoIP servers can also kill VoIP calls. VoIP troubleshooting, like all systematic problem-solving in telecom, can quickly degenerate into finger-pointing. Temporarily redirecting the media of a call to completely avoid a network removes that network as a possible source for whatever issue you're experiencing. If the issue is only present when your call crosses the suspect network, open a trouble ticket and have them resolve the problem. If the problem persists either way, you've at least proven that portion of the call to be clean.

Figure 1-3 shows how the standard call from Figure 1-2 should have been transferred. The overhead streams are still spanning between the main office and the customer phone, but the media streams are crossing only from the remote office to the customer, eliminating all latency on the media into and out of the main office and freeing up almost 200 Kbps of bandwidth on the main office IP connection for other calls.

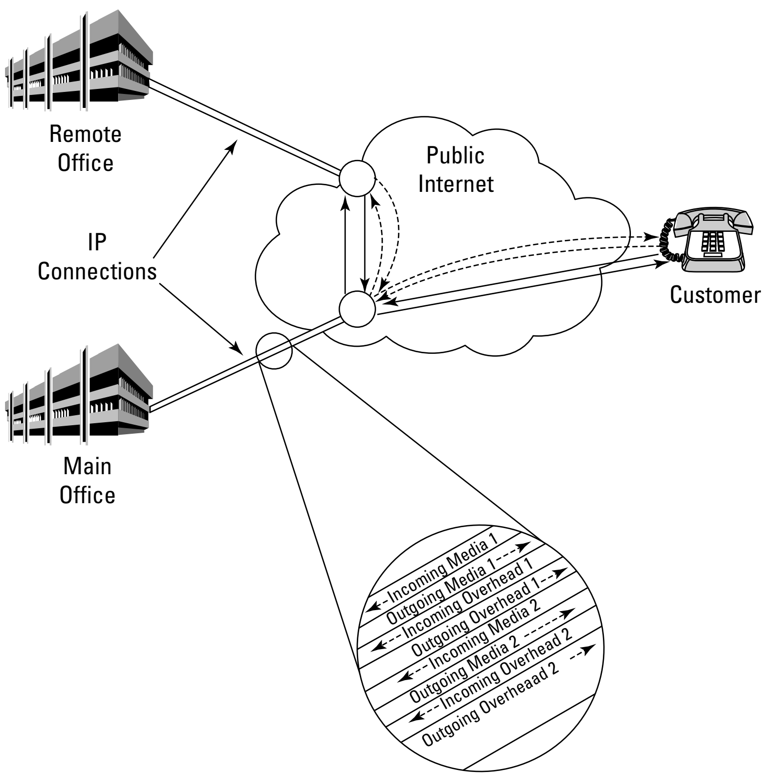

Figure 1-2: Traditional forwarding of VoIP call.

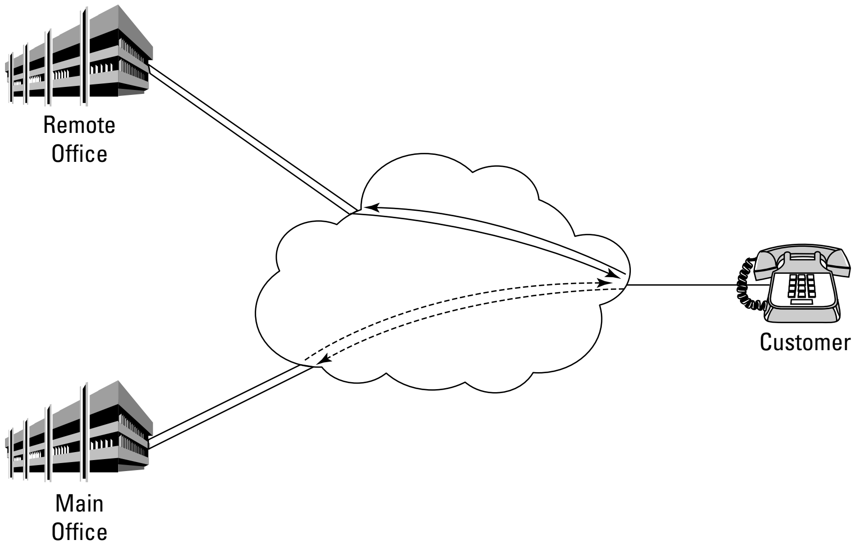

Figure 1-3: A VoIP redirected call.

As useful as this type of redirect is, many carriers don't like it because it requires their servers to work a bit harder. VoIP redirects the media portion of the call by sending a message to the server at the other end to roll the media to the new location. If you're a VoIP provider, this can amount to a lot of messages you're sending to your carrier's server to redirect the calls. Just like VoIP servers work to eliminate latency on the transmission of calls, the carriers try to reduce how hard their servers have to work.

Beware the rogue media stream. As awe inspiring as it is to see a media stream running off into the ether with no overhead in sight, a media stream can always go rogue. If the end IP destination requested in the redirect message is incorrect, some poor innocent IP is bombarded with a media stream. If the IP isn't set up to receive VoIP transmissions, the entire stream is probably rejected, with minimal impact to server receiving the unwanted data. If the receiving server is set up to receive VoIP, it might spin out of control and flood it, preventing it from servicing legitimate VoIP calls in an unintentional *Denial Of Service* (DOS) attack. Even though it wasn't your goal to overload their server, the RTP still prevented their intended customers from reaching them, so they'll call it a DOS attack, until we coin another acronym for involuntary SPAM.

Dispelling VoIP Misperceptions

Some of the confusion surrounding VoIP isn't unique. Fifteen years ago, people had the same fears, concerns, and misperceptions about the hottest technology of the time — ISDN. Today, the marketing machine for VoIP has promised that it will do everything but julienne potatoes, all for free or a low monthly package fee.

VoIP isn't as great or as horrible as anyone portrays it.

Using more bandwidth

Just because VoIP is slick and new doesn't mean that it's entirely more efficient than traditional telephony. VoIP has both uncompressed and compressed call options. Each has its pros and cons (covered in Chapter 2), but they're all contained within the same VoIP structure.

A standard non-VoIP call consumes slightly more than 64 Kbps of bandwidth, and you can place 24 consecutive calls over a normal dedicated 1.5 Mbps circuit. VoIP calls require more bandwidth to handle the additional overhead associated with packetizing it for transmission. If the media portion of the call isn't compressed, the total bandwidth consumption of a VoIP call can exceed 120 Kbps. A full 1.544 Mbps circuit of uncompressed VoIP allows you only about 13 calls, barely more than half the total calls possible if the circuit were traditional telephony.

The good news is that the most common type of VoIP compression allows you to transmit over twice the number of consecutive calls over a 1.544 Mbps circuit. You can save a lot of bandwidth by using VoIP, but how much you can save depends on whether you're compressing the media on the call. (Chapter 3 explains compression.)

Realizing that VoIP isn't free

One of the biggest marketing campaigns surrounding VoIP was the idea that all VoIP calls were free. At one point in time, that may have been true. Before 2007, the U.S. government didn't know how to tax VoIP calls, and so those calls were tax free. Before 2005, most VoIP calls were from one VoIP phone or computer on the Internet to another VoIP phone or computer on the Internet. By avoiding the infrastructure used by traditional telephony calls, it also avoided all the fees. As far as anyone else knew, the transmission wasn't anything other than someone surfing the Web or sending an e-mail.

The business of VoIP has changed since then, and many people simply use VoIP to access a local or long-distance phone carrier. Companies such as Vonage or your local cable TV company (if it also sells local phone service) are typical VoIP providers. These companies set up a VoIP connection between your home and their switch, but if you're calling your aunt in Florida or your grandma in Philadelphia, the call is still passed over the same legacy telephone network that it would if you dialed from a non-VoIP phone. Because the call uses the same switches and systems as a traditional telephony call means that the call is assessed per-minute rates in the same manner.

Any call that you make to a standard telephone number is charged a per-minute rate somewhere along the way. Even if you pay a flat monthly fee for unlimited long distance, your carrier is banking on the fact that it's charging you enough to cover all the minutes it's being billed for your calls.

Accepting that VoIP may not be cheaper than traditional phone service

VoIP's launch marketing hype said that, although it may not be free, at least it's cheaper than using the traditional analog phone lines and digital circuits. But it actually isn't always cheaper. All the long-distance carriers are rolling out VoIP service, but not every small and medium-sized business can save money by using it.

Business customers traditionally purchased dedicated circuits from their long-distance carriers, which allowed them to aggregate traffic and get a lower per-minute rate on their calls. Most carriers have kept the same pricing for the per-minute cost of their calls because the calls' networks and routing still go through the same systems. The main differences in the cost of VoIP, when compared with traditional telephony, are access fees and hardware costs.

Factoring in access fees

The phone carrier providing traditional dedicated circuit charges a monthly fee for the lease of the *local loop,* the cabling that connects your business to your carrier. A VoIP connection requires that you have not only a connection to a carrier, but also a port to the Internet, which usually costs an additional fee. You can generally use an Internet connection from another carrier to reach your long-distance provider, but then you have to worry about latency. Every server you encounter between your own server and your carrier represents a delay that can degrade the quality of your VoIP calls or simply cause your calls to fail. Before jumping into VoIP, be sure to compare how much IP bandwidth you need to match the total quantity of calls and consider all the loop fees and port costs.

The bandwidth used to place VoIP calls and traditional telephony calls can vary. A standard dedicated circuit that has 1.544 Mbps of bandwidth (called a T1 or DS-1 in America) is designed with 24 channels, each capable of processing a call. If your peak calling time has 24 calls going at one time and you don't want to compress your VoIP calls, you need to order two T1 circuits. With two circuits, your cost doubles because you have to pay two local loop fees and two Internet port fees. If you compress your VoIP calls, you can use one T1 circuit for 48 consecutive calls.

Table 1-1 and Table 1-2 show how your choice in compressed or uncompressed VoIP has a direct impact on your bottom line. Contact your Internet provider for the exact local loop and Internet port costs.

Table 1-1 Monthly Charge Comparison for Uncompressed VoIP and Traditional Telephony

Telephony Type	*Maximum Calls*	*QTY of T1s required*	*Local Loop Monthly Charge*	*Internet Port Monthly Charge*	*Total Monthly Charge*
Uncompressed VoIP	24	2	$300 ea	$200 ea	$1,000
Traditional Telephony	24	1	$300 ea	N/A	$300

Table 1-2 Monthly Charge Comparison for Compressed VoIP and Traditional Telephony

Telephony Type	*Maximum Calls*	*QTY of T1s required*	*Local Loop Monthly Charge*	*Internet Port Monthly Charge*	*Total Monthly Charge*
Compressed VoIP	48	1	$300 ea	$200 ea	$500
Traditional Telephony	48	2	$300 ea	N/A	$600

Figuring out the hardware costs

Unless you're creating a brand new company and phone system from scratch, you have to spend some money to either replace existing hardware or augment your network to handle VoIP. You don't have to spend this money if you stay with your existing configuration, so you need to weigh this cost against the financial and business benefits of deploying VoIP. I cover the types of hardware in the section "Identifying the Hardware You Need," later in this chapter.

Worrying about compatibility

Whenever any new technology is released, everyone always worries about compatibility. You may be wondering, "Will I be able to communicate with other VoIP devices?" I have a Fuji digital camera that uses a memory card that works with only Fuji and Olympus cameras — and it doesn't work in the photo printers at the local drugstore, either.

Looking at the compatibility headaches that came with other technologies, international organizations such as IETF (Internet Engineering Task Force) and the ITU (International Telecommunications Union) established guidelines called RFCs (Request For Comments) about how to transmit VoIP calls. This international cooperation allowed everyone to work together to develop systems and logic for VoIP transmissions, instead of everyone making up their own versions and letting the market decide which technology would survive.

Even though these organizations set down the guidelines for the transmission of VoIP calls, programmers still wrote software based on their own interpretations, and the industry quickly realized that those small nuances made all the difference.

VoIP carriers identified this challenge and developed InterOperability (InterOp) testing to ensure that the custom software built for a small business would work with the custom software built for a long-distance carrier. For a period of time, every carrier had an InterOp program with a testing window of a few days to a few weeks. In this testing window, VoIP customers and carriers validated that both ends of a VoIP call could accommodate how that call was being packaged, processed, and managed.

That was then, and this is now. In spite of the fact that you can sit down and create your own version of the VoIP protocol, you don't need to because you can find free software on the Internet that does it for you. You can download Asterisk or AsteriskNOW from `http://asterisknow.com/` and instantly have every bit of software you need to send and receive VoIP calls. The VoIP industry now has a greater level of uniformity in software, and you rarely encounter incompatibility between VoIP devices anymore. Carriers no longer have to worry about InterOp testing (though some still offer it because they've built infrastructure to support it and they have an extensive pre-established set of test cases that need to be accomplished that normally can't be concluded in a normal activation). They instead schedule a normal installation, just like they would if you were activating a traditional telephony circuit.

Many small businesses didn't appreciate the elimination of InterOp testing because they used the InterOp test environment to test and confirm their own internal dial plans or configurations. InterOp was designed to ensure that a business's VoIP server could effectively communicate with the VoIP server of its carrier, but slick technicians extended the testing to confirm that their new find-me-follow-me service was functioning properly or work through some bugs in their internal dial plans.

TIP

VoIP installation (covered in Chapter 8) is a very straightforward process. Incompatibility between VoIP hardware is now as likely as incompatibility between traditional telephony hardware.

Rejoicing in good quality calls

The quality of your VoIP call depends almost entirely on the network over which it's transmitted, from end to end. Long-distance carriers have been using VoIP within their own controlled networks for years. Almost every long-distance call you make is VoIP at some point during its transmission. VoIP is even more prevalent on international calls because the carriers that specialize in this niche market use standard VoIP compression techniques so that they can maximize the profit they get out of their current connections.

TECHNICAL STUFF

One of the first VoIP calls I received was from a programmer in Romania. I used a softphone that was installed on my work PC. The call had a lot of static and sounded like a radio transmission from Mars.

Now, almost every traditional telephony call you make is an example of the call quality you can expect on VoIP. It's no longer a free service with skittish quality; it has established itself as a legitimate form of telephony that's used, and offered by, all major carriers.

Understanding the VoIP Landscape

VoIP is a hybrid of data structure and voice application, so all the flexibility of data programming can now be applied to telephony applications.

TECHNICAL STUFF

Previously, phone systems were filled with proprietary hardware and software. If you wanted to add five more phone lines, you had to hope that your system had the room for an additional card — and the standard upgrade cards probably had more or less ports than you needed. After you finished installing the phone lines, your hardware vendor still had to sit on site to program the lines and give them the standard voicemail box.

The creation and release of VoIP has inspired a generation of programmers who didn't stop at devising a way to send voice calls over the Internet. They took on the office phone systems, as well, and decided to write open source software, such as Asterisk, on which other programmers could build. Now, you can have a fully featured voice telecom application running off a standard server with interface modules that are as easy to install as a new video card.

The largest structural downside to VoIP is the legacy data network over which the calls are transmitted. The Local Area Networks (LANs) and Wide Area Networks (WANs) were built years ago for the smooth transmission of data, and not for a real-time application such as voice transmission.

Here are the challenges that VoIP faces because it has to use existing LAN hardware:

- **Packets Per Second:** Servers, routers and network hardware are rated based on an idea of Packets Per Second (PPS). Data traffic doesn't really concern itself with PPS because, instead of sending a large volume of small packets on the LAN, it solves the problem by sending fewer packets, with each individual packet containing more data. But the real-time demands of VoIP discourage sending large amounts of data in individual packets. Losing a single large packet of data on a voice call might keep the call from connecting or affect the quality of the call. VoIP requires ten times as many packets to send the same volume of information as a data transmission.
- **Retransmission potential:** The primary protocol used to transmit data across the Internet is the Transmission Control Protocol (TCP). It confirms all packets are received by sending a count of how many packets were sent before the transmission is completed. If the receiving end received less than the total number of packets reported to have been sent, the missing packet can be resent without corrupting the transmission. Voice calls aren't that forgiving. They're transmitted with a leaner protocol called User Datagram Protocol (UDP), which doesn't allow packets to be retransmitted. If a packet doesn't arrive, it can't be retransmitted; if it arrives out of sequence, it's discarded.

  The inability to retransmit lost or corrupted data removes the possibility of a simple redundancy tool in VoIP transmissions, and is why every network engineer deploying VoIP is concerned about any delays in the delivery of VoIP packets.
- **Traffic pattern:** Data transfers tend to be long transmissions with a consistent flow of packets from end to end. Uploading or downloading a large file may take 30 or 45 minutes, during which time the packets are diligently being sent and received. Voice traffic has a patchier transmission style — your office may receive a barrage of calls between 9 a.m. and 10 a.m., and then the volume may drop off considerably before another spike hits in the afternoon.

All the issues in the preceding list are a challenge for VoIP to function within an environment that didn't anticipate its arrival. The market is continually responding to new technological needs, and I'm sure that newer, faster equipment will evolve to cater to this growing market.

Identifying the Hardware You Need

You can deploy VoIP as superficially or as deeply into your network as you want. You don't have to replace all your existing phones, your phone system, and the copper wire that connects them together with new VoIP equipment. If you're keeping all your analog phones and infrastructure, at a very minimum, you need a server to act as a gateway.

The term gateway has several different definitions, generally broken down into two categories:

- **Network gateway:** A device owned by a large long-distance or local phone service provider. It has a high call capacity and interacts through ISDN and SS7 signaling with the intermeshed connections of long-distance and local carrier networks referred to as the PSTN (Public Switched Telephone Network) that provide the paths to complete every call that either originates or terminates in the United States.

  ISDN and SS7 signaling are defined in the Glossary.

- **Enterprise gateway:** A device located at a non-carrier (most often, one transmitting at least two million minutes of calls per month) that interacts with the PSTN by using less complex protocols than a network gateway.

In the realm of VoIP, a *gateway* is a device that converts the language of the data received to a different protocol (or even a variant of the same protocol) so that it's compatible with the destination. The most basic example of a VoIP gateway is the Analog Telephone Adapter (ATA), which is a small hardware device that's delivered when you sign up to switch your home phone to VoIP.

Figure 1-4 shows how the ATA connects to your phone and sits between it and your VoIP carrier. It communicates to your carrier by using VoIP, but it converts the signal to analog so that you can use the same telephone you've always had. This type of service is generally sold as an add-on to your cable Internet or Dedicated Subscriber Line (DSL) service, but the standard bandwidth you use surfing the Web generally doesn't degrade the quality of a single phone line.

As long as you keep your voice and data networks separated, you can hold in check the variables that can affect call quality and completion. Figure 1-5 shows the most modest deployment of VoIP in a small business or office.

The small office in Figure 1-5 has existing analog phones and copper cabling that are being reused after adding a server that runs AsteriskNOW and using an Internet connection dedicated to VoIP.

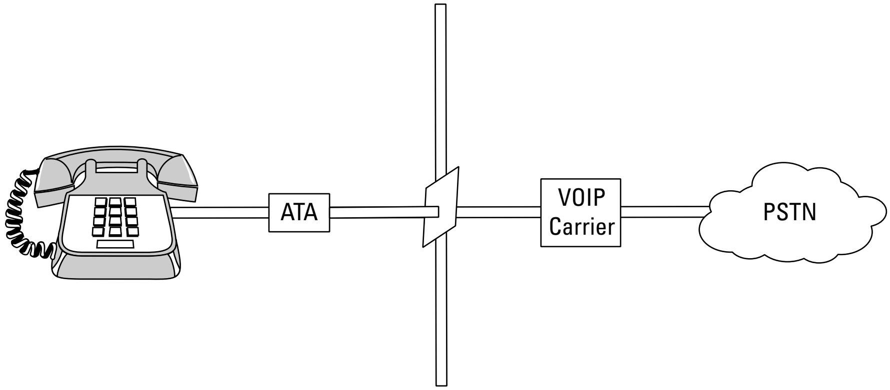

Figure 1-4: Installation of analog telephone adapter.

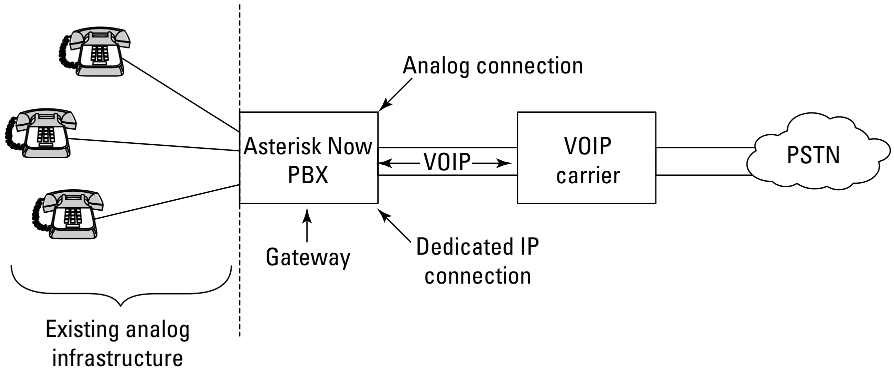

Figure 1-5: Small-business VoIP deployment.

If you are deploying VoIP by replacing your phone system, as shown in Figure 1-5, the main financial expense to absorb is for a strong server and analog cards that have enough ports to support your office requirements. The good news is that you don't have to replace any phones or cabling to make it all work. You can still pick up your old analog phone and dial out because the server converts the call to VoIP and sends it over the Ethernet port. The call runs through the IP connection to your VoIP carrier, which forwards it on through the Public Switched Telephone Network (PSTN) to connect to the phone number you dialed.

AsteriskNOW is my VoIP phone system of choice because the software is feature-rich and easily accessible, and anyone technically skilled enough to install a new motherboard on a computer can install it (including any analog port cards you may have) easily. This solution allows you to use both the traditional analog phone lines you receive from your carrier and an Ethernet port to send and receive VoIP calls.

You can load Asterisk on any server that's running Linux. The interface cards are made by a company called Digium (located at www.digium.com) and aren't cheap, but they work like a charm. The analog cards are optioned with either

- Foreign Exchange Station (FXS) ports that connect to your analog telephone
- Foreign Exchange Office (FXO) ports that connect your server to the your local carrier through the phone jack on the wall.

REMEMBER

When you install the analog cards for Asterisk, don't forget to plug the power into them. FXO ports on the cards receive power from your local phone carrier through the cabling that ends in the jack on the wall. If you want to send calls to phones on the desks of your employees, you need to make the connection from the card to the internal power feed within in the server. If you don't see the green light illuminated to the telephone jack on the back of the card, that port doesn't have power.

Figure 1-6 shows a simplified, but fully integrated VoIP and data network. In this scenario, all the copper phone lines that usually connect telephones to the phone system have been replaced with Ethernet cables, and all analog phones have been replaced with VoIP phones. The same Ethernet that the VoIP phones use to connect to the VoIP proxy server also allows the office PCs to surf the Web, send and receive e-mail and Instant Messages, and access the printers and servers on the LAN.

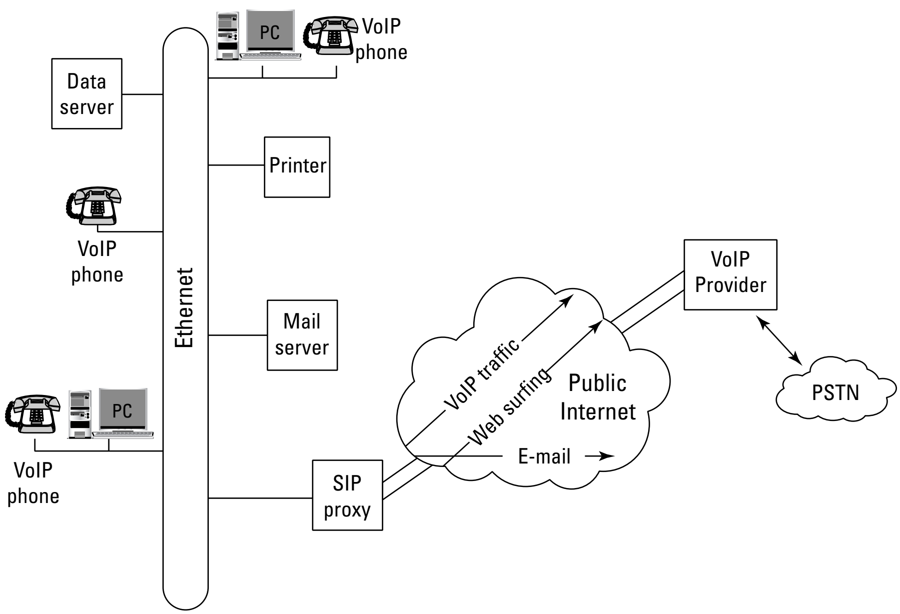

Figure 1-6: Fully integrated VoIP and data network.

This type of deployment requires much more analysis and planning before you take the plunge. All the devices on the LAN can crowd your network with packets, generating delays that can result in failed calls, and poor call quality. Part II of this book covers getting ready for deployment in detail.

The three preceding scenarios show how you can deploy VoIP with as much integration as you desire or require. As long as you have a gateway device to convert VoIP to analog, you can retain legacy hardware and still enjoy the benefits of VoIP. These scenarios are all very basic — you can find many more VoIP hardware devices available than I list in Figures 1-4, 1-5, and 1-6.

VoIP isn't a protocol to transmit voice telephone calls over the Internet, it's more the concept of doing so. Several signaling standards can be used to accomplish VoIP, but at the moment, the market is favoring only two choices — either H.323 or Session Initiation Protocol (SIP). I cover these protocol choices in greater detail in Chapter 2, if you want to know more.

The most popular protocol being used right now is SIP, so I use it as the basis for my hardware and software discussions in this book.

Understanding nodes

A *SIP node* is a generic term used to describe any hardware device that interacts with a SIP call. A SIP node can be the originating SIP hardphone, an intermediary server, or a receiving softphone. Regardless of its place in the call path, each node contains two key software elements:

- **User Agent Client (UAC):** The UAC initiates communication to the next node downstream in the call path and requests information or acknowledgements from it.
- **User Agent Server (UAS):** The UAS receives the communication from the UAC of the node upstream on the call, processes the request, and responds back to it.

Because every SIP node employs these two elements (either in part or in full) during a call, they're also sometimes referred to as SIP User Agents (UA).

Dealing with SIP end points

SIP end points are the first SIP node originating the call and the last SIP node receiving the call. They can include servers that may generate or receive phone calls, but the term *SIP end points* generally refers to the two types of SIP phones:

- **SIP softphone:** *Softphones* are software-based applications that display a small dial pad on your computer screen (similar to the calculator you can find in the Microsoft Windows Accessories folder). You can usually configure a softphone easily, and it allows you to make calls to other IP phones by using the same Internet connection to your PC that you use to surf the Web and send e-mail. The mouthpiece and earpiece of a traditional phone are replaced with the microphone and speakers on your computer.

  I always found the reality of having someone's voice coming through the speakers of my computer a bit disconcerting. Well, only half as odd as screaming into my computer when I couldn't hear them.

- **SIP hardphone:** Any object that you can use to send and receive voice phone calls (that isn't a computer displaying a softphone) is a *hardphone.*

  The traditional candlestick phone is a hardphone. The white and brass princess phone is a hardphone, and the black two-line phone with wireless handset on your desk is a hardphone.

Using the many servers

The downside of SIP is the more phones you have, the less likely you'll be able to deliver or manage everything that SIP can provide within the confines of that SIP hardphone. You can easily fix this problem by deploying servers in your network that have specific functions. You can have one server that provides the functionality of all the following servers, you can pair the functions, or you can distribute them as you see fit in a cluster of servers.

Securing the LAN with a SIP Registrar

When a SIP hardphone is booted up, it normally signs in with a designated database server called a SIP registrar. The server collects information about the SIP phone, identifying the phone's location so that the server can effortlessly send calls to that phone when the server receives an incoming attempt for that extension.

Benefiting from a Feature Server

SIP facilitates the transmission of VoIP, but all the great add-on options are available from associated open source software, such as Asterisk.

Don't try to cram all these fun attributes on the individual PCs that contain soft phones — instead, house them in a centrally located feature server. The services available from a standard server that runs Asterisk include not only normal elements you expect from a traditional phone system, but also some other specialties that you may have had access to from your local phone carrier or a third-party telecom provider. Some of the services that you may have on your feature server are

- Voicemail, with the recorded messages sent to preset e-mail addresses as .WAV files
- Call queues
- Call forwarding
- Conference calling, with call recording
- Call hold
- Call parking (placing the call on hold and transferring it to an extension)
- Least Cost Routing (LCR; see Chapter 4)
- Music on hold
- International call blocking
- Auto-attendant or Interactive Voice Response (IVR) systems
- Blacklisting (call routing based on the call's Caller ID)

Aggregating with a SIP Proxy

A *proxy server* acts as a link between the SIP devices on your LAN and the outside world. It serves as a single focal point to receive calls to distribute throughout your company, thus making your carrier's life easier. It also monitors the outbound calls and can be designed to restrict access for some extensions that may be unsecured (such as a lobby phone). It manages calls between the outside world and the LAN only — SIP phones on the network can contact each other without interaction with the SIP proxy (unless the proxy is also the SIP registrar).

Figure 1-7 shows SIP Phone A and SIP Phone B on a LAN. Because they're both registered with the same SIP register server (which is also the SIP Proxy, in this example), they can call each other without any assistance. If someone calls on a SIP phone that isn't listed with the SIP Registrar or from a traditional phone number that must be routed through the PSTN, the call goes to the SIP Proxy Server. This server helps to secure the telephony network. Without the SIP Proxy acting to receive and manage all communications into and out of the office, each SIP hardphone would have to speak directly through the Internet connection to the VoIP provider. Although this approach isn't a huge technical issue, it's a security issue because now you have multiple IP addresses that can make outbound calls and are vulnerable to being compromised to commit fraud instead of just one.

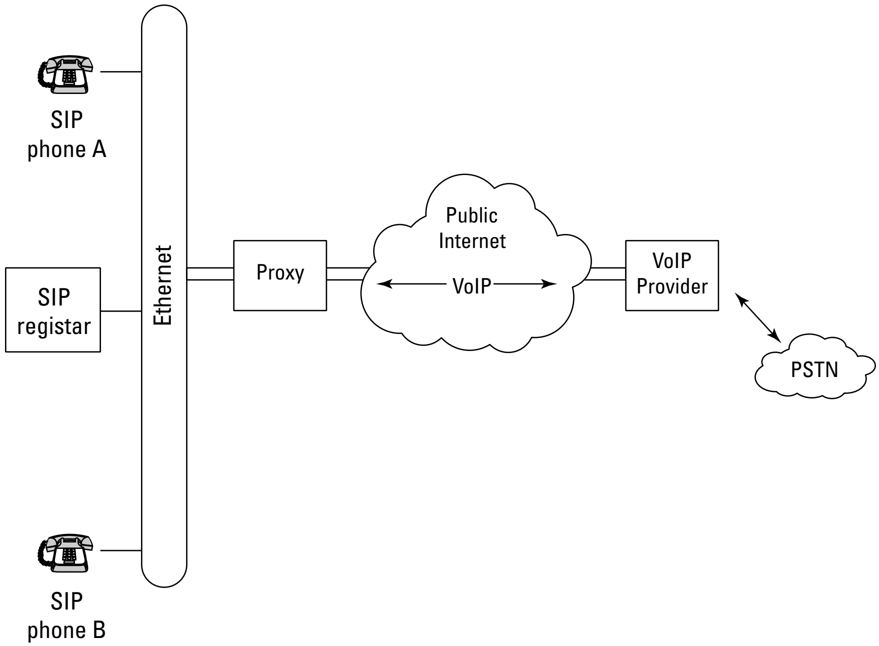

Figure 1-7: SIP Prox Deployment.

The processor speed of each proxy dictates the rate at which you can send new calls to your carrier. Most companies that have a normal work force of 10 to 20 employees don't need to send calls at a rate faster than two or three calls per second (CPS). If your business is telemarketing- or telecom-related, you may need to send up to 70 or 80 calls per second. You can find the calculations to determine your maximum CPS in Chapter 5. If you need more than ten CPS, your carrier will most likely provide multiple Proxy Servers or Session Boarder Controllers (SBCs) on its side to allow you to get the number of calls per second that you need. You may get four SBCs to send your calls to, with 9 CPS per SBC — allowing you to send a maximum of 36 CPS during your peak time without killing your carrier's system.

Overrunning the prescribed CPS from your carrier makes your carrier upset with you. Every carrier SBC services hundreds of customers. If one customer overloads the switch and slams it with 100 CPS when it wasn't supposed to send more than 10 CPS, the system will most likely start to progressively increase the delay in responding to incoming calls, and calls will fail. Not only will calls fail from the one customer who's slamming the switch, but every other law-abiding customer's CPS will also fail. The network security department of your VoIP carrier watch for this variety of Denial Of Service (DOS) attack and quickly resolve the problem. If you're the CPS violator, you may receive a quick call from your carrier's network security department requesting that you cease and desist, or the carrier may just turn off your access to its system. Always watch your output and maintain friendly relations with your carrier.

Transmitting the Non-Voice

VoIP was designed for the transmission of voice conversations over an IP network. It evolved to sample and reproduce the human voice, matching the quality of existing telephony. It has succeeded in that endeavor, but two other essential aspects of telecommunications frequently use non-voice transmissions — touch tones and faxes.

Pushing touch tones

Technically, the sounds you hear when you press the digits on the keypad of your phone are called DTMF (Dual Tone Multi Frequency) tones. They were designed as two tones sent at the same time (hence the "dual tone" part of their name), a feat that can't happen in normal human speech. Because of the complex nature of the sound, VoIP has a difficult time reproducing it when the voice portion (media) of the call is compressed.

Unfortunately, touch tones are commonly used during business calls. Every voicemail system says something like, "Press 1 to page this person." Or you may encounter a complex auto-attendant system that asks you to input your phone number, account number, or extension of the person that you want to speak to. VoIP has found ways to make DTMF tones work, but the solutions vary, depending on whether you're using compressed or uncompressed transmissions.

TIP

I cover the options for DTMF in detail in Chapter 6.

Faxing over VoIP

The squeals, squalls, and hissing that you hear during a fax transmission definitely don't fall in the realm of sounds made by a normal human voice. They're specifically designed to transmit data representing a visual image and, like DTMF tones, provide a challenge for voice-centric VoIP. The VoIP community of engineers and programmers have expended a great deal of effort to establish a viable way for faxes to be sent within a VoIP infrastructure.

TIP

You can get a breakdown of your best options in Chapter 6.

Porting Phone Numbers

VoIP has been helped along in popularity and growth by the widespread availability of high-speed Internet connections and the fact that people can migrate their phone numbers from their current local phone providers. *Local Number Portability* (LNP) is the process by which a phone number is moved from one local phone carrier to another. It's a relatively new process that has a few challenges to be overcome.

TIP

Keep these key points in mind when you move your phone number to a new carrier:

- **You lose all your features.** Every feature that you have on your phone line at this time that *isn't* provided by your phone system is provided by your local phone carrier. Your new local phone carrier needs to provide call forwarding, voicemail, three-way calling, distinctive ringing, Caller ID blocking, and any other features you require. Always ask to ensure your new carrier can provide the features before you sign a 12-month contract and move your phone number.
- **You can't move a virtual phone number.** Some phone numbers don't exist on a physical phone jack on your wall or a piece of copper wire coming into your office. These phone numbers are virtual numbers that exist only to provide a feature.

  A classic example is the distinctive-ringing number you order for your children so that you know when someone's calling to talk to them. The phone number that your local phone carrier provides doesn't physically exist anywhere other than a database in the carrier's network. When a call arrives in the local carrier's central office, it sees the number dialed and sends the distinctive ring to your house.
- **You need to provide a Letter of Authorization.** This letter is a security measure for both you and your carrier. You fill out the Letter of Authorization (LOA), sign and date it (less than 30 days before your request), then submit your phone number for migration. This documentation keeps the carriers honest and provides a paper trail in case someone from a company with 100 phone numbers transposes some digits and mistakenly migrates your home line.
- **The time frame to migrate a number varies by carrier.** Every local phone provider of any size has its own department devoted to processing the incoming and outgoing migration of phone numbers. Although every local phone provider has a porting relationship with every other local phone provider, the nature and structure of that relationship varies. Larger local phone providers may be *e-bonded* (electronically linked) with automated systems that can validate and release a phone number to a new carrier in five days. Smaller carriers may process the requests manually with a short staff and, depending on who's sick in the department and whether it's a holiday, the time to migration may be one month or more.

- **You may not be able to get a directory listing.** You may not get your phone number listed in the white pages of your local phone directory when you migrate that number. Depending on the carrier you select, it (or the carrier it uses to actually receive your phone number) may not want to support the staff required to process, maintain, and manage the service. Ask whether your new carrier provides directory listing.
- **Your carrier can't legally prevent your number from migrating.** The phone number that rings into your office is yours — and with perseverance and diplomacy, it can migrate.

Reviewing the LNP process

The LNP process can be an extremely frustrating experience for everyone involved, especially when the business whose numbers are migrating doesn't know the steps involved. Unfortunately, unlike cell phones, the land-line realm of telecom can't migrate your phone number over in a matter of minutes. It takes days, and the process has three distinct steps.

Say that you're moving your phone number from Pacific Bell to Level 3. After you submit your order to Level 3 (along with your current LOA), Level 3 follows this process:

1. Level 3 requests a Customer Service Record (CSR) from your existing carrier. This process can take as little as a few hours if the releasing carrier (in this example, Pacific Bell) is e-bonded with the carrier making the request (Level 3), or it can take as long as weeks. Pacific Bell returns your CSR, and Level 3 uses the information to match up the LNP request, making sure that someone didn't transpose a phone number or list completely bogus or invalid information.
2. Level 3 issues an Add Service Request (ASR) to Pacific Bell to move the phone number to Level 3. Requesting the CSR from Pacific Bell doesn't tell the company that you intend to move the phone number, and the system to pull the CSR information may not be directly linked to the internal LNP process within Pacific Bell. The issuance of the ASR to Pacific Bell is its first official notification that you are porting your phone number from the company.
3. Pacific Bell issues a Firm Order Commitment (FOC) to Level 3 and identifies the date on which the number will be released.

After the carriers complete all the steps, the number is pre-built within Level 3's network on the night before it's to be released, so the following morning at 9 a.m., it cuts over without a disruption of service. The transition happens seamlessly, and you can't even tell that your number's been moved.

Understanding rejections

Not every porting request flows perfectly smoothly. An order can be rejected by your current local carrier (Pacific Bell in the preceding example) at any point during the migration process for a number of reasons. The rejections are rarely malicious — they're used to protect you and your phone numbers. If the process had no checks and balances, anyone could accidentally migrate your phone number away without your consent. Here are some of the more common rejects you may encounter:

- **Name or address mismatch:** This rejection is generally issued when someone transposes digits on a phone number. When your new local phone carrier matches up your order to migrate your phone number with the company or residence named on the CSR, if the phone number and company name aren't the same, the order is rejected. You can resolve this rejection by
  - Checking to ensure that you wrote the correct phone number down.
  - Providing a copy of your local phone bill, listing your company name and the phone number in question. This documentation should allow you to push through the issue (unless you have a resold account, as explained in the following section).
- **Pending order:** If your phone number has any pending orders on it at the time of migration, the existing local phone carrier rejects the migration request. Ordering new voicemail service on your business line a day before you request to migrate the number sends mixed signals to your carrier.

  As a rule, a carrier rejects any migration request while the phone number in question has any pending orders. You can clear this rejection either by
  - Calling into your existing local carrier and cancelling the pending order
  - Waiting long enough for the pending order to complete and then resubmitting the LNP request.
- **Billing Telephone Number (BTN) doesn't match or is inconsistent with the phone number:** The BTN is the main phone number used in the billing system of the local carrier to identify all phone numbers for your company. Large companies with multiple locations may have unique BTNs for each site to make it easier for their carrier to invoice and manage the account. The BTN mismatch rejection is more complex than a simple name and number mismatch. The phone number and company name may match up, but the address isn't in the same city or state

as the phone number. If you're trying to migrate a phone number in Milwaukee, Wisconsin and you list a BTN for your remote office in Marfa, Texas, the LNP request will probably be rejected. Just like on a name/address mismatch, you can resolve this problem by supplying a copy of your bill and following up on the order daily as it progresses through the LNP process.

- **PIC Freeze:** A Primary Interexchange Carrier (PIC) freeze is a logistical security device that many people have on their phone lines to prevent changing their long-distance carrier without their consent. This process also applies to the migration of local phone numbers. Your local carrier rejects any attempt to migrate your phone number as long as you have the PIC freeze on that number. You can clear this hurdle by calling your existing local carrier and having the PIC freeze removed. Then, you have to resubmit the number for migration.

Grappling with resold accounts

Every VoIP provider that you can receive phone service from doesn't have several millions of dollars in phone switches, cabling, and a batallion of employees to service their networks. A VoIP carrier has to spend massive amounts of time and money to become an established local phone carrier. Such a huge barrier exists to entering the market that many companies bypass becoming a carrier themselves. Instead, they take a short-cut and simply contract with a company that has already gone through the process. The companies that don't own their own networks are technically resellers of VoIP service. There can be several VoIP resellers between the company using the service, and the carrier that's ultimately providing it.

One of the greatest frustrations with the LNP process is the blanket of mystery that covers phone numbers provided through VoIP resellers. This situation leads to a chain of customers and providers that looks similar to Figure 1-8.

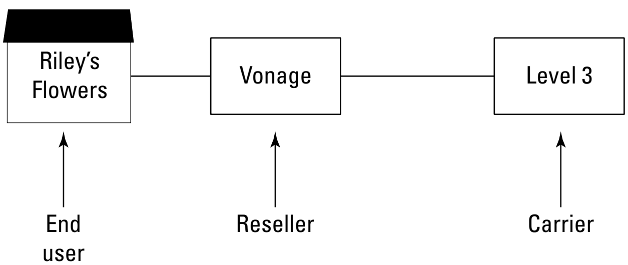

Figure 1-8: Customer - Reseller - Carrier Chain.

An *end user of record* generally identifies the person who receives service on a phone number and is financially responsible for that number. The challenge facing the industry in the situation of resold accounts is that the end user of record changes, depending on who you ask.

The carrier in Figure 1-8 knows only that the phone number in question is on an account for Vonage. When the CSR is received by a carrier attempting to initiate a migration for it, the billing entity on it is generally listed as the reseller. Because Vonage isn't a true local phone provider, it doesn't receive the migration request — so the name, phone number, and address of the end user can't be validated in the same way that a non-resold number can. You can generally clear this reject by submitting a recent copy of your phone bill that shows your company name, address, and the phone number that you're trying to migrate.

But the situation can get even more complex and muddled. Some resellers package VoIP with additional services and private-label it for sub-resellers. Another link appears in this chain of companies, further obscuring the true identity of the legitimate end user who picks up and answers the line when someone calls the number. If a secondary-reseller is in the mix, submitting a bill copy may still not resolve the problem because the address and company name listed won't match anything the VoIP carrier or the first reseller can use to validate the request (because the true end user is only known by the secondary reseller). When all else fails, bring together everyone from the LNP departments of every company in the chain and talk it out. It's amazing how everything begins to work when you bring everyone together.

Only *local* phone carriers own and supply phone numbers, which is why I continually refer only to local phone carriers when discussing the porting process. The line between who's a long-distance and who's a local carrier can often get blurred because of mergers, but a carrier must be licensed in your state to provide local service and have a network of large phone switches built to provide local phone service (categorized as Class Five switches) in order to have numbers to port.

Dealing with the costs

Migrating phone numbers from one local phone carrier to another used to be free in the innocent frontier days of VoIP. Local phone carriers sat one or two people in an office and let them handle the dirty business of LNP. When VoIP grew, and more and more people began migrating their home phone numbers and small business lines to VoIP-enabled local carriers, the job began to occupy more people and more resources.

Then, one day, local phone carriers' accounting departments decided to run a cost-benefit analysis of the process and — not to anyone's surprise — they found that they were losing money. Magically, overnight, the service that used to be free now had all kinds of ancillary fees, such as

- **Installation/migration fee:** A per-number charge that's assessed for the processing, migration, and activation of a phone number from one local phone carrier to a new local phone carrier. The fee can vary, but a $25-per-number charge isn't unheard of.
- **Monthly recurring fee:** This fee is generally under $1 per number per month and helps to offset cost required to maintain the routing information for the phone number in the national routing database.
- **Change/modification fee:** You have to pay this charge if you need to modify information to a number during the migration process. These fees can vary.
- **Snapback:** The Snapback process is the LNP version of dealing with buyer's remorse. If you request to have your phone number migrated, and 24 hours either before the number is to port or after the number has ported to the new carrier, you demand that it be returned to your old carrier, the carriers can do it quickly through a process called Snapback that instantly returns the ownership of a phone line to the prior local carrier. Snapback can only be done up to a few days after the initial release of the number, while the routing and infrastrucuture is still in place at the prior carrier. The downside of this procedure is that it frequently costs $200 or $300 for each phone number returned.

TIP

If you're migrating more than 20 phone numbers from a single carrier, ask your new carrier whether it can process the order as a bulk order for a reduced activation fee. You may be even more successful at reducing the installation fee if you're moving from one reseller to another who both use the same network. Figure 1-8 gives a good example — if you, the end user, move to another Level 3 reseller from Vonage, the numbers don't have to migrate to anywhere. They still remain with Level 3, who merely has to change the reseller account on which they reside. Internal migrations are generally less painful and costly for the carriers, and so those carriers are more inclined to provide financial clemency if you ask.

WARNING!

Many companies don't need to migrate every phone number in their office. The financial cost per number encourages many companies to simply migrate the phone numbers known to the outside world, such as the main office line used for incoming calls and the fax line. In these migrations, your existing carrier may ask whether you want to cancel or migrate the remaining phone numbers. Be very careful to identify any and all numbers that you need to migrate before telling them to cancel the rest. Any number that you cancel can be immediately released to the available pool for anyone to reserve. If you don't correct the situation before the number is fully cancelled, it can easily be gone and lost forever — with no way to recover it. Check your phone numbers twice (or three times) before you cancel any of them.

Identifying who's responsible for what

If you work in a small business that's migrating its phone numbers, you're responsible only for correctly filling out an LOA and any additional paperwork, and possibly supplying a copy of an invoice that shows your company name, address, and the phone number(s) you're porting. If you work in an enhanced VoIP company that has a book of customers on one side of your network and a carrier from which you receive your VoIP service on the other side, more responsibility is resting on your shoulders.

A VoIP reseller must not only provide an LOA and a copy of a bill, but also gather status on all LNP orders in process, identify all LNP rejections, and proactively resolve the rejects by supplying additional documentation or negotiating the release on conference calls. Unless your carrier specifically states in your contract that it is accepting the responsibility to manage the LNP orders and proactively solicit documentation from your end users, and negotiate the release of rejected numbers, it offloads that responsibility to you. Because you have the direct relationship with your end users, and because it would violate the terms of the contract you have with your carrier as a value-added reseller, this complex and unforgiving job rests in your capable hands.

Your carrier is responsible for processing the orders you supply in a timely and accurate manner, and providing you with the means to identify rejections on individual phone numbers. You might get a weekly spreadsheet with this information, or your carrier might provide an interactive Web site. You just need to remember that the reseller is responsible for deciphering the rejections and taking the appropriate action.

If you're a VoIP reseller, you also need to manage all migrated and new phone numbers assigned to you. Because your VoIP Carrier is charging you an activation and monthly fee on these numbers, you need to keep an accurate accounting of those fees to reconcile your invoice every month.

Your carrier should also provide you with a Port Out Notification whenever a phone number is migrated off your account. You have to account for all the ins and outs of your phone number inventory if you want to ensure that the ancillary monthly fees are accurate — especially if your carrier charges for numbers migrated away from you.

Request that your carrier provide your phone number information in a consistent format. Create a process around that format whereby you can simply cut and paste the phone numbers into a database or spreadsheet. Avoid entering phone numbers manually because the potential for human error is far too great — you can end up assigning a phone number that hasn't been reserved for you to an anxious customer who doesn't understand why his or her phone number doesn't actually exist.

Ordering New Phone Numbers

You can often order new phone numbers more easily and at a lower cost than you can migrate existing phone numbers. Your carrier doesn't have to negotiate the release of the phone numbers with another carrier, and it may even have numbers in stock to dole out to you. Remember these two things about new phone numbers:

- **Your new carrier can't guarantee you a specific area code or prefix.** If you live in a city that has an *area code overlay* — where two different area codes service the same geographic area — your carrier can give you either area code, regardless of the one you request. The same goes for the next three digits of your phone number, known as the *prefix.* You may want an 805 area code and 966 prefix for your numbers, but if 805-756 numbers are the only ones available, those are the number that you get.
- **Your new carrier can't guarantee all your phone numbers at the same time.** If you're requesting more than 100 phone numbers from a specific geographic area or market, you may have to wait over a month to receive all of them. New phone numbers are doled out once a month, in lottery fashion, to all local phone carriers. If your carrier can't gather all the phone numbers you want from its inventory or after the first round of new number allocation, you have to wait until at least the following month.