Chapter 1

Evaluating the Possibilities with Asterisk

In This Chapter

- Working with standard voice calls
- Harnessing the power of VoIP
- Tying together standard voice and VoIP
- Contemplating the wireless potential
- Determining your Asterisk hardware requirements

The sophistication of telecom hardware available to the average company has increased tremendously. It was amazing in the 1980s when you could use an automated dialer to direct calls to your long-distance carrier instead of your local carrier. In the 1990s, the technology and market had evolved to where you could identify calls based on where you were dialing and match them to the carriers you may have had at your disposal for the best rate. In 2000, all the buzz was to use Internet circuits to transmit calls. At the time, the quality was a bit suspect, but most of those issues have been worked out, and people now send Voice over Internet Protocol (VoIP) calls around the world every day.

This evolution created the environment that spawned Asterisk. A division in telecom has always existed between the engineers that transfer data and the engineers that work on voice service. In general, these are different types of people, and they don't hang around with each other. Asterisk brings them both together for the first time. It allows you to handle voice calls with the same mind-set of routing data.

This chapter introduces you to the depth and breadth of what you can do with Asterisk. We cover the general types of calls and features as well as the varieties of telephony platforms you can use to interact with Asterisk. We also cover the hardware necessary to make and receive calls, and then we give you the parameters for the server that you need to run your Asterisk for optimum performance.

Finding Out What You Can Do with Asterisk

Asterisk bridges the methodical, planned open-source architecture previously the sole domain of data transmissions and takes it into the voice world dominated by rigid proprietary phone systems. The intelligence and flexibility of telecom hardware available to the person on the street has grown at a very fast pace over the past 20 years.

REMEMBER

In the future, Asterisk has the power of bringing together people from around the world for free. VoIP connections handled entirely across the public IP network don't incur per-minute charges like a normal phone call. Only when the proprietary networks of the long-distance and local phone companies handle a call do per-minute charges apply. Otherwise, the transmission travels across your office or across the world for free, just like an e-mail.

In the sections that follow, we show you what you can do with your Asterisk.

Using Asterisk for your phone system

In the beginning, Asterisk was rumored to have been designed as a voice-mail system. That is by no means the limit of its potential; voice mail is simply the beginning of its capabilities. The software now functions as a platform for receiving and transferring calls with all the standard features you want from a phone system, such as

- Voice mail, which allows callers to record a voice message when you are on your phone line or away from the office
- Conference calling, which provides the ability for multiple people to call into your Asterisk and talk together just as if they were all physically sitting in the same conference room
- Dial-by-name directory, which allows callers to reach an extension by spelling out the first or last name of the person of the person they are calling on the keypad of their phone
- Call parking, which sends a call someplace to wait on hold
- Music on hold, which gives callers something to listen to while you have them on hold

These same features are available with any phone system. You need to purchase hardware to allow you to use either dedicated digital (T-1) lines or regular, analog, plain old telephone service (POTS) lines.

You can use Asterisk as your phone system without ordering any standard analog or digital phone lines from your local carrier. As long as you have a dedicated Internet connection to your home or office, you can purchase incoming and outgoing phone service from your Internet Service Provider (ISP) and never have to worry about standard phone lines.

The main benefit of Asterisk is its flexibility. You aren't locked in to any preset parameters of the phone system. Some of the great features you have complete control over with Asterisk are as follows:

- How you receive calls
- Whether a call is sent directly to voice mail
- Whether a call is sent directly to a conferencing room, where several people can speak on the same conversation
- How calls are routed within your phone system and your company
- How to forward calls (such as first to your landline, then to your cellphone, and then to voice mail)
- How much time you allot before you rescind the call and transfer it to the next destination (such as a cellphone or voice mail)
- The extensions and features available to every phone and piece of hardware

Jumping into VoIP with Asterisk

One of the best aspects of Asterisk is the ease with which you can integrate the ability to send and receive calls over your dedicated Internet connection. *Voice over Internet Protocol,* more commonly referred to as *VoIP,* is the greatest revolution to hit telephony since the dial tone. All the benefits of VoIP have yet to be realized, but Asterisk comes standard with the ability to convert your voice call into VoIP packets.

One of the fastest growing applications of Asterisk is its use with VoIP. Many companies have sprung up for the sole purpose of providing advanced features to their customers via VoIP. The "find-me-follow-me" type of service that allows one call to attempt to reach you on your office line and then your cellphone, before the call is rescinded and sent to your voice mail, is just one of the great features of VoIP. You can even e-mail your voice mail as a WAV file to your BlackBerry or Palm device. Talk about never missing a call again. If you are breathing and available by any means of technology, Asterisk can find you.

The future of telecom is in the transmission of voice, fax, and video over Internet lines. The ability to packetize all these transmissions into the IP realm is a huge step forward in telecom. It allows for growth at an accelerated pace as the transmissions are handled together, maximizing the bandwidth of every connection. By breaking the mindset of traditional telephony where every voice call is locked into a 64Kbps channel, we lose the hardware requirements preventing us from using the Internet over our regular residential phone lines.

Bridging technologies of VoIP and non-VoIP

VoIP is great, but not every piece of hardware you may need to use is VoIP-based. You may use VoIP service within your office but have a better rate for out-of-state calls on a carrier that doesn't have a VoIP interface. In this case, you have to send the call with the standard, non-VoIP method using traditional time-division multiplexing (TDM) connections.

Not every long-distance carrier has the VoIP service you want. Some carriers only provide inbound VoIP service to avoid providing directory assistance, 411, and 911 services. Other carriers may avoid these requirements entirely by boycotting VoIP connections.

Your local or long-distance carrier may charge you differently for VoIP than for TDM service. The per-minute rate you are charged may be different, the area of coverage provided may be restricted, or additional monthly recurring charges or installation fees may apply.

Just because your carrier may not accept VoIP calls doesn't mean that you can't use VoIP somewhere in your system. You simply need to convert your calls from VoIP to TDM, or TDM to VoIP, when you enter or leave your carrier's network. The ability of Asterisk to act as a *gateway* (converting calls from VoIP to TDM or vice versa) gives you the freedom to use every telephony option at your disposal.

Bringing wireless into the equation

You aren't confined to traditional landlines with Asterisk, so don't get trapped by that old private branch exchange (PBX) mind-set. In the land of wireless technology, Asterisk is a good friend. It can easily interface with any of the following devices:

- Regular cordless phones
- Cordless VoIP phones
- Wireless headsets connected directly to your computer via Bluetooth

You can use most Bluetooth wireless headsets that you use with your cellphone. You use your computer as the audio device and pair it with any VoIP software phone (*softphone*) for your PC. This configuration allows you to make calls from your softphone via your PC with VoIP. Asterisk implements its own Bluetooth channel driver, allowing Asterisk to function as your softphone on your PC.

Running your telephony business with Asterisk

Using Asterisk for your internal company calls is only the starting point of its potential. Many companies, especially VoIP-based resellers, may run their entire business on Asterisk servers. If you run a telephony business, you'll find the following useful features in Asterisk:

- Flexibility in programming
- Reporting features
- Call Detail Record (CDR) logs
- The ability to use TDM, VoIP, and wireless connections

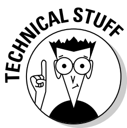
But the fun doesn't stop there. Asterisk is easily partitioned for multiple companies to use all the available features, but still confine their calls to within their company. You don't have to worry about pressing 0 to speak to an operator at Company A and getting the receptionist for Company B. This requires a bit more effort because you must partition all the calls by company in your CDR.

Realizing the benefits of VoIP to big businesses

Multilocation corporations that have offices in different states or even countries can use Asterisk to virtually eliminate long-distance calls between offices. Installing an Asterisk at each location allows you to send and receive VoIP calls, taking your standard long-distance calls from your existing carrier. Just as e-mail eliminates the need for sending letters through the post office, VoIP now allows you to bypass your local phone carrier.

If you are using the Asterisk internally, a basic configuration should cover most of the scenarios you might encounter (see Figure 1-1, later in this chapter). If your company is a telecom reseller that provides phone service to customers, possibly as a VoIP value-added reseller, your configuration needs to be a bit different. In spite of the fact that you might have a few phones in your office attached to the Asterisk server, Asterisk primarily processes your end users' calls.

Asterisk is the best platform for VoIP resellers because you can expand and build on it as your business grows and becomes more complex. You can realize a configuration as grand as your company can become. Your setup can range from a single server that provides central call processing for multiple offices using VoIP phones to multiple servers that process calls from multiple offices that use (a) VoIP phones directly and (b) VoIP PBXs such as Asterisk to fulfill all their office needs.



A PBX is a common phone system that you would find in an office environment. It generally provides voice mail, call transfer, call hold, and the general routing of calls handled by an office phone system.

Getting Acquainted with AsteriskNOW

AsteriskNOW is a complete software package, which allows you to load Asterisk software and a distribution of Linux operating software on a server. After you complete the installation, you can configure it using an Internet GUI. All the regular features provided in Asterisk, such as voicemail, conference call setup, call queuing and dialplans, are also available through the AsteriskNOW release.

AsteriskNOW makes Asterisk available to a much broader range of companies and individuals. The idea of working in a Linux operating environment and building the routing rules with manual programming might be too much for some. For those who want to the power of Asterisk, but don't have the time to build out the programming, AsteriskNOW is your ticket.

AsteriskNOW is just like the standard version of Asterisk in its hardware requirements. It's a real-time program requiring the full and undivided attention of your server. Whether you are using the software as a phone system for your business or as a basis for the value-added telecom service you are providing, you must have it on the best hardware possible. Regardless of the size of your company or server, the Asterisk program must be given top priority in all tasks to reduce the chance of network delays degrading the quality of your calls. A modest amount of delay caused by other network activity can result in static or sections of calls being dropped. At the other end of the spectrum, large delays could result in static or the failure to connect both inbound and outbound calls.

Introducing the Supporting Hardware

Asterisk is wonderful, but it is only software diligently working inside a server. It is fully capable of converting a call from VoIP to TDM, but you still need some kind of hardware interface to connect your server to the phones in your office or the outside world. Even when you install the required hardware to make these connections, you still may need additional software drivers to bring them to life. Table 1-1 lists the hardware and software you need to make connections with Asterisk.

Table 1-1 Asterisk Interface Hardware and Drivers

Application	*Hardware Available*	*Software Driver Required*
Analog connection	Digium FXS and FXO cards	Zapata
Digital T-1/E-1 connection	Digium T1 card for 1, 2, or 4 ports	Zapata
Digital T-1/E-1 connection	Zapata single T1 card	Zapata
Digital T-1/E-1 connection	Sangoma single T1 card	Zapata
VoIP connection	Network interface card (NIC)	ztdummy*
All above applications	No additional hardware	libpri**

**The ztdummy driver acts as a timing source whenever you aren't using a POTS/T1/E1/J1 card. Applications such as conferencing and calls transferred within the Asterisk server require this timing source. The 2.6 Linux kernel is configured with an internal clock source to avoid this issue. Make sure that you download the latest driver.*

***The libpri software is a required part of the Zapata modules.*

Determining your analog hardware needs

Servers don't have an unlimited supply of expansion slots available on them. Before you download your Asterisk software, you need to develop at least a two-year plan; we recommend a five-year plan. Analog cards take up the most room and are truly limiting. A four-port analog card can only handle four connections. This is in contrast to a T-1 Internet port that could previously handle 20 consecutive VoIP calls and can now manage 50 or more because of compression. If you will need more than four analog ports in the next year, make sure that you have a second four-port card available. If you need more than eight analog lines and are running out of expansion slots, we recommend purchasing the 24-port card now and leaving the ports open until you need them.

Using analog interfaces

If you aren't versed in telephony jargon, we recommend using the Foreign Exchange Station (FXS) and Foreign Exchange Office (FXO) cards for service using regular phone lines, just as you do in your home. These lines come from your carrier and terminate in a small jack on the wall that are plugged into one phone. FXS interfaces connect directly to a handheld telephone or a dialer. The FXS cards provide the dial tone, caller ID, and ring voltage to the phone so that you understand that your call is being processed.

The FXO cards connect to the phone lines from your local carrier and transmit your call to the carrier for processing. These cards detect dial tone and ringing from the far end so that your FXS card can then forward this information to you.

The Digium analog cards allow you to either use four individual phone lines through standard RJ-11 jacks (the same jacks your home phone connects to) or an interface where you can use up to 24 phone lines through a specialized plug called an *amphenol connector* that separates every channel into 24 pairs of wires.

REMEMBER

Analog cards from Digium are modular and can support FXO and FXS ports in any configuration. Your four-port card can have one, two, or three FXO cards with one, two, or three FXS ports. It's great to have options!

If you want 24 phone lines, you need special cabling with the amphenol connector and at least one more piece of hardware. You need a *break-out box* or *punch-down rack* to allow you to wire into the individual channels (in the case of the punch-down rack) or to plug your phones into one of the 24 standard RJ-11 jacks (in a break-out box).

You can purchase analog and digital cards that handle 1, 2, 4, and 24 lines as well as digital T1/E1/J1 cards for 1, 2, or 4 ports from the following companies:

- Voipsupply: `www.voipsupply.com`
- Sangoma: `www.sangoma.com`
- Digium: `www.digium.com`

Using external analog or digital cards with Asterisk allows you to connect to your existing phone lines. The larger T1/E1/J1 interface cards with two and four ports are only cost-effective in specials setups where you have an internal need for these phone lines or if you are reselling the lines to your customers.

We suggest that you expand your service with VoIP. Everyone in your company can access phone lines, without requiring you to purchase a single phone line for each employee. One dedicated Internet circuit can easily handle the voice requirements for 10 or 20 people with the correct configuration. Turn to the section "Sending calls out VoIP" later in this chapter if you decide to take your phone service to the next level.

Going digital and dedicated

Digium, Sangoma, and Zapata manufacture digital cards for Asterisk. Digium has the widest selection of digital cards that offer a two- and four-port model. Sangoma has a similar offering; its four-port cards are currently less expensive than the four-port Digium cards.

Sangoma offers a clear channel DS-3 card (capable of handling IP bandwidth equal to 28 T-1 lines) at this time. The company is also planning to release a channelized version of the card that can handle 672 voice channels (28 T-1s of 24 channels each).

The Zapata drivers are necessary and work well with any of the cards you choose. You have to love software that works with everything.

T-1 is an industry term for a circuit with 1.544 Mbps of bandwidth that is broken into 24 individual channels that can process one call each. This is the standard building block of dedicated digital telephony in the United States and Canada. Europe and much of the rest of the world use circuits called E-1s that are broken into 32 individual channels, giving you the capacity to handle eight more calls than the U.S. T-1 circuits.

All the cards are solid performers, but we have always experienced very good performance with the Digium cards. We also try to support our local Asterisk vendors, and because Digium resides in Huntsville, AL, where one of the authors runs his company, it seems the neighborly thing to do. If we take out the geographic bias, we recommend either the Digium or Sangoma T1/E1 cards.

Sending calls out VoIP

Asterisk comes standard with all the drivers you need to make VoIP calls. As long as you have a dedicated Internet connection and a NIC (network interface card) in your computer, you are ready to go. The only remaining piece of the puzzle you need is the ztdummy driver to help maintain the clocking on the calls, and that is only if you are transferring the calls internally within the Asterisk (or across to another Asterisk server) or if you are setting up conference calls. If you are dialing from a VoIP phone to a VoIP provider, you don't need the ztdummy driver.

Using VoIP to its fullest requires some research on your part. If you want to send and receive VoIP calls, you may need to have two VoIP carriers. Some carriers only provide *inbound* service (where people call you). This restriction allows the VoIP carrier to dodge the current federal requirement of providing 911 service. Other VoIP carriers specialize in *outbound* service (where you dial out) but may have limitations on some of the services provided. You may incur an additional charge for 411 and 911 services, as well as for a directory listing in the white pages. Even if you do get your name in the white pages, you may have to jump another hurdle trying to get your name in the 411 directories. Research these features with your carriers to be sure that you choose the correct carrier.

Communicating with your phones or dialers

So far, we have been speaking about hardware required to connect your Asterisk to a telecom carrier. The connection may be to a single analog telephone line or as many as four individual dedicated digital T-1 or E-1 lines. This is a vital link in sending calls to, or receiving calls from, the outside world, but it isn't everything you need. Unless you have a recorded message to a list of phone numbers in the same server that is running your Asterisk software, you need some type of interface to individual phones or another server that has a dialing program built on it (if you are a telemarketer).

Figure 1-1 depicts the possible setup of a server running Asterisk with an analog, digital, and dedicated Internet connection to individual carriers and wired into phones in the office on the other side. Its connections to telephones within your office are on the left, and the connections to a local carrier, long-distance carrier, or Internet Service Provider are on the right.

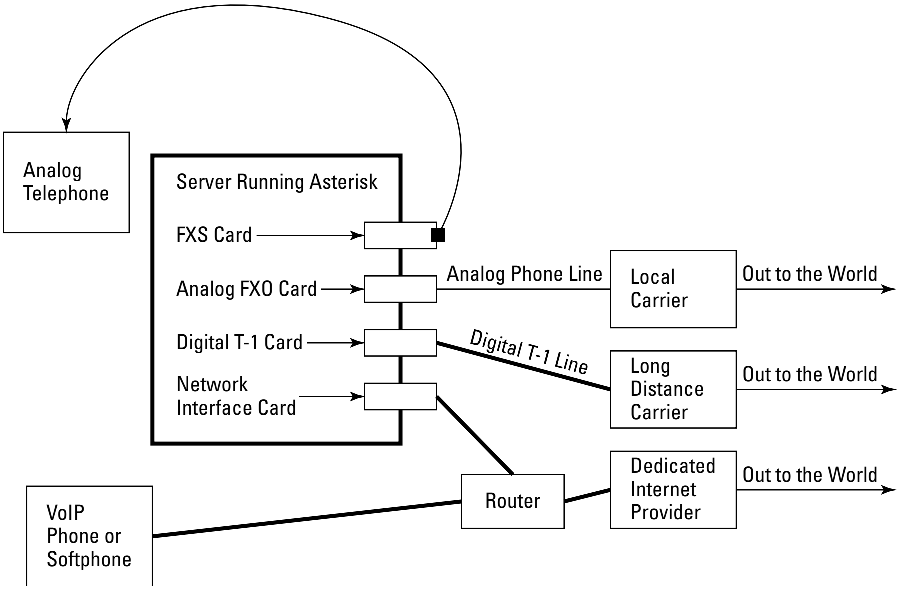


Figure 1-1: Hardware configuration for Asterisk from phone to carriers.

REMEMBER

The telephones you connect to your Asterisk server must be compatible with the type of telephony you are using. You can't use an Integrated Services Digital Network (ISDN) phone on a connection sending VoIP, just as you can't use a TDM phone on a digital circuit.

Several manufacturers produce VoIP phones that range in price from $50 to $600 per phone. Your specific application and budget dictate which phone is the best for you. Check out the following Web sites for VoIP phone manufacturers:

- VoIP Supply: `www.voipsupply.com`
- Cisco: `www.cisco.com`
- Linksys: `www.linksys.com`
- Polycom: `www.polycom.com`
- D-link: `www.dlink.com`
- Grandstream: `www.grandstream.com`
- Sipura: `www.sipura.com`