VOIP::About

SIP Text

Mizutech

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Contents

Three internet phones

are being used, namely: PC softphone , VoIP handset, and the ATA or the analog telephone adapter. The first one is a software that makes use of the computer's sound card, speakers, and microphone. The VoIP handset relies on your personal computer and is more complex than the first one. The ATA will not require you to have a personal computer because the high-speed internet connection is directly connected to it. These internet phones also have different softwares.

Voice over Internet Protocol (VoIP) is the latest technology allowing you to deliver and receive voice calls using an Internet connection, particularly broadband, as a substitute to an analog or an ordinary phone line. Also, it is referred to as the Broadband telephony, Voice over Broadband, IP Telephony Broadband Phone, and Internet telephony.

The best way to appeal to upper management is to focus on the costeffectiveness of convergence. Map out your current expenses and contrast those numbers with the expense of VoIP convergence. The numbers speak for themselves: VoIP convergence reduces operating expenses enough to pay

for itself in the near term, and it can save the company a whole lot of money going forward

VoIP services may allow you to call other people by using the same service; however, there are some which allow you to call anyone having a telephone number. This includes international numbers, mobile phones, local numbers, and long distance. Some VoIP services just work over your computer or a special VoIP phone, while other services let you use a traditional phone connected to the VoIP adapter.

The telephone is very important for all people. It allows you to keep in touch with your family and friends and it also allows you to communicate with your business acquaintances. Today, with the advancement in communication technology, it allowed people to communicate more clearly at a very cheap rate. Some communication mediums are also provided for free, such as emails.

With all the different kinds of communication technology available today, you would definitely want the best kind of communication system available in order for you to communicate with your family, friends and business acquaintances more efficiently and also at a much cheaper rate.

This kind of technology is called VoIP or Voice over Internet Protocol. This is the latest technology available to all consumers all over the world to provide cheaper and more efficient calls. Unlike conventional phone systems, the VoIP uses the internet to transmit and receive voice and video signals. This means that the data transmitted is in digital format. Because it is in digital format, it will mean faster transmission rate at a much clearer and sharper audio quality.

Great advantage of VoIP

is that most VoIP service providers enable free calls within their subscribers. This means that if you are calling a person who is subscribed to the same VoIP service provider you are subscribed in wherever they are in the world you will be able to call them for free. There will be no hidden charges, and not a single cent will be charged to you. Not only will you be able to make free calls within the subscribers of the same VoIP service provider, but the best part is, the calls are unlimited.

Most people are familiar in using the internet when conducting meetings. VoIP has a similarity with this process; however it uses a telephone being connected to a modem. This modem can be requested or purchased from the ISP (Internet Service Provider). The installation is very simple: VoIP modem is plug into the current modem and the phone and computer is plug into the VoIP modem.

Just imagine, if you are in the United States, and you are calling someone in the United Kingdom who is also subscribed to the same service provider you are subscribed in, you can talk for hours without worrying about the phone bill.

Not only are VoIP phones able to provide you with free calls, but they also charge far lower long distance fees and rates. For example, if you are calling from your VoIP phone to a landline phone in another state or in another country, the overseas or long distance call charges will be far lower than charges that you will have in conventional landline phones.

All of these things are just the beginning of the advantages of VoIP. One other advantage is that VoIP offers free features that are usually expensive in landline phones. Features such as three-way calling, video conferencing, caller ID, return call, call waiting and others are considered to be standard features of VoIP.

Because of these features, VoIP is considered to be the phone of the future.

However, you will need a broadband Internet connection for you to be able to maximize your usage for VoIP. You need a broadband or high speed Internet connection in order to transmit and receive data signals from the Internet at a faster rate. Without a broadband connection the calls can result to poor quality.

But, you should consider that most households and offices today uses a broadband internet connection and is readily available in almost every part of the United States, especially with the wireless internet capability that is widely available today. Broadband connection today is cheap and widely available, so getting VoIP in your home can be easy.

All you need to do is download a VoIP software program from the Internet from your preferred VoIP service provider website, register and start making those free and low cost calls.

The only disadvantage of VoIP is that when the Internet connection is down, you will not be able to make calls. Also, if you are using an IP phone used for VoIP and looks like a conventional phone, you can never use it if the power is out. This is because IP phones are dependent on home electricity to operate, unlike traditional phones where it has an independent source of power in their lines.

However, if you want to have a low cost phone call, VoIP is your best choice. Besides, electricity supply and internet connectivity is now getting more stable than ever before. With VoIP, you can be sure that you will be able to save a lot of money and also let you consider getting rid of your traditional landline phone.

Today, free things are hard to find and everything may seem to be all for sale. From getting your food to making calls, all these things are offered with a price. However, because of the advancement of technology, there is now a way to call your friends and family who lives in different countries for free.

It is now possible for you to call your family and friends cost free, even if it is a long distance call. And, the best thing about it is that the calls are unlimited. With this kind of feature, who wouldn't want to get it? Another great thing about this is that it offers great features that will usually be expensive when you get it in a conventional landline phone.

If you are wondering what kind of service this is, it is called VoIP. First of all, one company that offers these services is called the VoipBuster. This company offers VoIP and also has great VoIP offers for its subscriber.

However, before you subscribe the services offered by VoipBuster, you first need to know all about VoIP first. Since VoIP is relatively a new technology, you need to know about it first in order to know what to expect with VoipBuster.

VoIP is very much like your conventional telephone where you make calls and receive calls. However, unlike your conventional landline phones, VoIP doesn't use telephone lines to transmit and receive signals. VoIP uses the internet to transmit and receive data. And, for this reason, VoIP will be able to provide you with better voice quality and faster transmission rate. This is because your voice is converted into digital format, and transmitted in the internet. You will also receive digital signals from the person you are speaking to in digital signal. This is one of the main advantages of VoIP.

Another great thing about this technology is that it is able to offer its user's great additional telephone features for free as a standard package where it can tend to be an expensive add on in conventional landline phones. Features such as three-way calling, caller ID, return call, audio conferencing, call waiting, video conferencing and others are offered by VoIP as a standard feature. This means more value for your money.

Long distance calls

also tend to be cheaper in VoIP than on conventional landline phones. Some companies even offer free calls. However, this only happens when the person on the other end of the line is also subscribed to the same VoIP service provider you are subscribed in.

However, VoipBuster offers free VoIP to landline, VoIP to VoIP, and VoIP to mobile phone calls absolutely for free. Here are the countries that you can call for free if you use the VoipBuster:

- United States
- Taiwan
- Switzerland
- Spain
- South Korea
- Singapore
- Andorra
- Austria

- Australia
- Canada
- Chile
- Cyprus
- Belgium
- Denmark
- Finland
- Estonia
- France
- Hong Kong
- Ireland
- Iceland
- Italy
- Japan
- Liechtenstein
- Luxemburg
- Monaco
- Malaysia
- Netherlands
- New Zealand
- Norway
- Panama
- Portugal
- Puerto Rico

Although you need to top up your account to make these calls, it is definitely a great and cheap way to get in touch with your family and friends. By purchasing credits, you will be entitled to 120 Freedays. This means that you can call people in the mentioned countries at no cost at all. If the 120 days expires, the normal rates for calls will apply. However, because it uses the VoIP technology, you can expect that the normal rates will be far lower than the normal rates in conventional landline phones.

If you are interested in getting the VoipBuster services, all you need to do is have an active high speed internet connection, download the VoipBuster program in your computer, register and start making free calls. If you need to call other countries besides the mentioned countries, VoipBuster normal rates will apply.

With the available 120 days of free calls to the mentioned countries, just imagine how many of your family and friends you can call at an unlimited time. With VoipBuster, you'll never worry about large phone bills when making long distance or overseas calls again.

In a family with several children, it's not very good to compare them with each other. Oftentimes this will result to a feeling of enviousness and rivalry. They would also try to compete for their parent's attention. Comparing can be very harmful to the kid's relationship towards each other.

Comparing things can be useful if you're trying to compare a VoIP service provider. And this goes true even to different businesses offering products and services in order to attract customers; they want to compare their products/services against a competitor. Whichever provider that comes out best will get the fairer share of the market.

This is one reason why all business owners have developed ways to attract prospects, including all VoIP providers. But do you know what VoIP is?

The main technology responsible for internet phones is the voice over IP, more popularly called VoIP. As its name implies, this new technology allows any person to make voice conversations over the IP network.

Through VoIP, a P2P session is established by optimizing protocols as a group. This will result to an uninterrupted stream of digital data carried through the net. The two VoIP protocols which are now widely accepted are the open standards and the session initiation protocol or SIP. Voice signals are continuous in nature, and so it requires this session.

VoIP providers

use any of the three phones and software. So if you have preferences especially with the hardware, you have to compare different VoIP providers. By comparing them, you will be able to choose one that has almost all your needs.

The VoIP provider is responsible for the initiation of P2P sessions. Aside from that, they also associate the telephone numbers to different IP addresses, and route VoIP data to other personal computers, to the mobile number or landline number that you're trying to call.

Although VoIP providers differ in their hardware and software being used, the process of making a call is still the same. They only differ on the hardware's processing.

Not all people are ready for this new technology. They still compare it with their long-trusted conventional telephone. In comparing these two, the biggest difference lies on the costs. With VoIP, you can surely cut down your monthly phone bills. VoIP is often associated with 'free' calls.

VoIP service is well-known for free calls for promotional or trial basis. However, when you sign up for their service, you will be asked to pay low-cost plans to enjoy the service. But this cost is lower than that of traditional phones.

Since VoIP is still new in the market, other people want to rely on their traditional phones for reliable connections and exceptional sound quality.

In doing a comparison between a traditional phone and voice over IP, you can see their advantages and disadvantages over the other. If you try count each of their good and bad aspects, you can come up with a good decision. It's either you settle for VoIP or remain loyal to your traditional phone.

You alone can compare these two services. If you have a regular phone, why not settle first for a free trial period? Look for a VoIP provider which offers such service, you can find a lot of them on the internet. After the trial period expires, make the decision. Trying it yourself can give you a clearer view of how VoIP works.

If for some reason, you're not satisfied with the result, then continue with your traditional phone service.

Comparing VoIP

and the regular phone service will only be effective if you know what you need in a phone service. Most people who go for VoIP are the ones which want added features on their phone, one that only VoIP providers can provide.

The Internet is immensely popular these days. Many individuals even end up marrying their chat-mates and there are also those who are able to find work or a career on the net. Most people are familiar with the internet as a helpful tool in doing researches, browsing through different websites, instant messaging, and music streaming. But did you know that it is also slowly entering a different industry?

Yes, you've heard it right. Telecommunications is one big industry, and the internet is making its way to this market. Now, it is possible to call a friend or a loved one anywhere in the world. You can easily make international calls at less cost. You might be wondering how, but if you're an avid internet user, you probably know that by now.

FlyFone provides products like the FlyFone router. This is used in case you want to avail VoIP service. For your information, VoIP or voice over internet protocol is a new technology introduced in the market. It allows you to make international calls or receive calls from friends and family overseas. Local calls cal also be made using VoIP. As compared to landlines, VoIP is much cheaper, and you will be able to save money on your phone bills.

Getting VoIP service is much easier and faster if you already have a broadband connection or a fast internet connection. This is needed in order to plug your FlyFone adapter or other FlyFone products and enjoy unlimited calls.

SIP was originally implemented for voice over internet protocol (VoIP) applications, initially for residential and now for businesses. SIP has become the accepted signaling for VoIP, although SIP clients and Proxies can contain certain liberties of the standards that might create issues when using different SIP implementations. This first phase of SIP for VoIP was a solid design, stable in requirements, but fairly simple in nature.
Today, there is a new game in town that has thrust SIP into warp speed. That technology is IMS. SIP was always designed to be more then just a vehicle for VoIP, and IMS will put that to the test. Standards bodies are moving very fast to modify and extend standards for IMS. Since IMS is simply an architecture, the real winners are the applications running over the IMS architecture. Progress will move quickly as new applications will drive new requirements on SIP clients. This paper will examine what new requirements will be placed on a SIP client. Because of the nature of IMS, SIP client changes will go beyond SIP. This paper will review the areas of changes that will affect the SIP client and will discuss:

You don't have to be hesitant about getting a FlyFone VoIP. Although it is a new technology, you're guaranteed a good service for less cost. Many people are looking for ways to save money and this is one step to do just that. If your friends, business associates, relatives, and family have FlyFone, then you will be able to call each other on the internet for free.

FlyFone VoIP also has features like caller ID, call forwarding, redial, and account management online. These features are made available to you depending on the plan that you choose.

Try to research first what VoIP is on the internet. You can also talk to other people who finally decided to switch to VoIP. If you think the service will work best for you, then you're ready to make a final decision.

The nest step is to choose a plan. Before jumping at any plan that comes your way, you must first make a list of all your wants and needs with regards to using a phone. Since there are many VoIP services and providers, choosing the best service will be much easier if you know what you want in a VoIP service or plan. You have to choose between residential or commercial/business.

After choosing the plan, you also have to ensure that the computer system in your home or office is compatible with the service. Don't settle for anything less than a high-speed connection. Residential VoIP service is easy to install; in fact, you can do it yourself. Commercial VoIP on the other hand requires professional installers.

With FlyFone VoIP, you can choose your phone number. Other providers even let you choose area codes. Check with your provider first and know their policies. Don't forget to install the accessories and features that come along with your VoIP service.

If you want to keep your old phone, you can get a FlyFone router. It is a small device that will be attached to your old phone so that you can enjoy using VoIP. This device figures out datagrams, maintains/controls bandwidth, control traffic, and ensures voice quality to all of your phone calls.

With technology always improving things by the second, time will come when routers will not be used. Digital phones are great examples for this; but not all people are welcoming the change with an open hand. So if you want to keep your old phone because it has a sentimental value of some sort, keep your phone and get a router.

FlyFone offers different VoIP products. So check with your VoIP provider for these products. Getting this new service may mean shelling out a small amount of money, but after that, you can save a great deal, not to mention calling your loved ones anytime, anywhere.

Today, the use of computers has spread worldwide. And this was even pushed forward with the introduction of the Internet. Even little children are now familiar with the Internet.

If you're an avid user of the Internet, then you've probably heard of what VoIP is and its many uses. VoIP or voice over Internet protocol is a new application on the net that allows a person to make phone calls. Somehow, traditional telephone lines are not enough. And this is the very reason how VoIP came into existence.

VOIP Telephone networks

are threatened because of the emergence of VoIP. More and more businesses, and people, are opting to replace their existing phone lines.

VoIp was not really intended to replace traditional telephone lines. In fact before, it was widely used for voice communication among computer users from different locations. Today, this application is still in use but is now more developed and acts like a telephone network. With the use of VoIP, anyone can call or receive calls anywhere, regardless of the location. You can receive calls as long as you have an Internet connection or your telephone set is connected to a network in your local area or LAN.

How did VoIP started? The Israeli's were the very first ones to make a voice connection over the computer, and that all happened way back in the year 1995. Within that year, the new technology became a software, the Internet Phone Software. Before, they used a modem, speakers, sound card, and microphone.

The computers which can use this new technology were only those having installed the software. With regards to the quality of sound, it was rather poor and can't even come close to the traditional phones.

By the year 1998, that same technology was further developed. The gateway to computer-to-telephone connections was opened. The Internet was also used by people having telephone-to-telephone connections; and with the aid of the computer in initiating and establishing the phone call connection, the regular phone can be used.

At present, both commercial and residential sectors are finding VoIP quite useful. Different services also offer different connections like PC-to-telephone or the traditional phone to another phone.

Internet telephones are plugged into the USB port or sound card of the computer. The phones have ringers and number pads just like the ordinary phone set. But now, it is already possible not to have a computer in using VoIP. You simply connect your telephone set directly to the broadband modem (cable or DSL).

When you use VoIP, your voice is converted to digital data. The data is compressed after your voice has been digitized. Around 1500 bytes 'packets' of digital data is transferred through the net. These packets also contain information on their origin, destination, and timestamp. The timestamp allows correct order-reconstruction. Upon reaching their destination, the packets are again re-assembled and are converted back to analog. The receiving party will be able to hear your voice.

Delay is one thing that hinders good communication. And if you want to use VoIP, you should first have a broadband connection. This makes it a lot easier and simple to add VoIP.

If you have second thoughts about using VoIP, you can always try it out for free. There are many VoIP packages that you can download on the net like Net2Phone, Gizmo, Skype, and other VoIP providers. This will be a lot easier if you and your friends all try it out together. Once you have the software downloaded, give your friends a call. Make sure that you have a headset and attach it to your computer's sound card, and there you have it.

If you want to call your friend, just click the name you want to contact. It also has other features like call forwarding, conference calls, and voicemail without extra charge if the call is made using computers.

If you're satisfied with the results, then perhaps you want to replace your existing phone with VoIP. But before making any decision, make sure that everything is to your advantage by using VoIP over the traditional phone. Don't be confused between an Internet phone and an IP phone. Internet phones are connected to the USB or the sound card while an IP phone is plugged to the modem.

If you want to enjoy VoIP now, then it's all up to you.

The telephone

is one of the most useful inventions man has ever made. It is something that made the world smaller and it is also something that modern society is continuing to use in businesses and also in homes. Since the telephone was invented, services have since improved and additional phone features are also available, such as three-way calling, or call waiting.

The telephone line is now also being used to connect to the internet. High speed internet connection through DSL is now available in most telephone companies. Today, the telecommunications industry is now taking it one step forward and developed a new kind of telephone system that is starting to gain popularity in today's society. Imagine, with this kind of communication system, you will be able to talk to your loved ones abroad for a fraction of the cost that you will usually get in conventional landline phones. Also, features, such as call waiting, call forwarding, return call, video conferencing capabilities, caller ID and others are integrated with this particular communication system as a standard package, which means that it's entirely free.

Free calls are also integrated with this new kind of phone system. If you are calling someone from halfway around the world, who is also subscribed to the service provider you are subscribed in, the calls will be free. It would be great if you have this kind of phone system. Today, it is now possible because of the advancement in the communications technology and this communication system is called VoIP or Voice over Internet Protocol or sometimes referred to as Internet Voice.

The reason why calls are so cheap with VoIP is that it is connected to the internet. As most people know, the internet provides a free and open communication system. Another great thing about VoIP is because the data is transmitted in a digital format in the internet, transmission and reception of voice data is faster and clearer than ever before.

The fact that VoIP has so many benefits, many people are now considering getting rid of their conventional landline phones and getting hooked up with VoIP. As you can see, VoIP is one of the latest solutions in communications technology that will allow people to make much cheaper long distance calls than conventional landline phones.

However, you have to consider that there will sometimes be hiccups in your internet connection. Because of this, it can significantly affect your VoIP phone. It will result in garbled communication or it can result in delays. This is why QoS or Quality of Service is now being integrated into VoIP to provide a solution for garbled communication.

The two main benefits of QoS for VoIP are:

- It protects VoIP on shared media
- It can prioritize VoIP

Bursty data applications will result in lower quality VoIP as well as losing 2 VoIP packets.

You also have to consider that VoIP is sensitive to delays and jitters because of the occasional internet hiccups. With QoS prioritization, it will able to minimize its effects.

QoS can maximize the quality of VoIP by controlling bursts of excess bandwidth and traffic. QoS can avoid congestion of bandwidth thus, making quality better when you use VoIP.

The quality of service

should be integrated in VoIP, especially for business VoIP to avoid miscommunication or delay in communication. Today, VoIP is still considered to be in its infancy. So, with all the bugs fixed, you can expect that VoIP will be integrated with QoS.

You have to consider the fact that VoIP is still not really perfect. For this reason, you have to have QoS with your VoIP. This will definitely improve the quality of communication and also decrease the effects of delays and garbled voice signals.

VoIP has so many benefits to offer people. You just have to consider that VoIP is still a relatively new technology. So, you should expect some bugs or some hiccups when making calls. However, if you want VoIP in your business, you should consider getting QoS to improve the quality of communication.

With QoS, you can certainly make sure that your VoIP will generate higher quality calls. So, if you are frequently experiencing garbled communication when you are using VoIP, you should consider getting QoS. This particular tool can definitely improve data transmission and reception.

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Your application for a job overseas has been approved. You and your family are celebrating, and after a few days you're on your way to earning a good sum of money. You're just one of the many people looking for a good fortune in a place you're not even familiar with; and to think that you have to bear with the loneliness of being away from your family.

In this particular situation, the telephone is your best friend. It's the easiest way to get in touch with your family. But sometimes, making too many calls can get very expensive. And so you try to cut back in your calling costs. And this means that you have to call your family fewer times. But there is no need to be lonely; VoIP can answer your problem.

Voice over Internet protocol will allow you to make calls to your family free if both of you have PCs. Almost every household have personal computers, and so it is much easier to get an internet connection. If you want to make VoIP calls, you should have a high-speed internet connection; go for cable or DSL. This will allow you to receive calls fast and clear.

The main reason why many people want to have VoIP in the residential homes is because of the cost. Compared to land-based phone connections, VoIP is relatively cheaper. And not only that, there are other features being offered that landline phones can't provide. Using VoIP is not that complicated, and it's just like a traditional phone. The main difference lies in the cost which many people enjoy.

VoIP is not only for commercial purposes, but also for residential. Residential VoIP offers different rates as compared to commercial VoIP. You can check out the internet for a list of residential VoIP providers. They cater to the residential calling market or home-based office business. They usually sell or provide an adaptor box (ATA). The ATA hooks between a broadband connection and a telephone. Before using VoIP, you must familiarize yourself first with the language and terms that concerns the use of VoIP. This makes it easier for you understand how it works.

In using traditional phones, it usually entails additional fee charges on your monthly bill. But with VoIP, you can make or receive calls with almost the same quality and reliability without the added fees.

Most land-based phone companies have hidden charges that increase your monthly bills. With VoIP, you can ask the provider about their different packages to fit your budget and your needs. Some providers offer unlimited local calling or long distance calls, and other perks like voicemail, e911, call waiting, caller ID, 3-way calling, call forwarding, etc. You can't get all these services if you have a regular phone at home.

You have to evaluate first all your needs, including the budget. And that is the only time for you to look for the right VoIP provider on the net. You can even ask your friends and officemates if they know of an excellent VoIP provider.

With all the advantages that VoIP has to offer, you should also be aware that there are also some drawbacks in using VoIP.

A typical example is when you want to use your fax machine. VoIP is not compatible with fax machines so you can't use the two together. Another problem with VoIP is when there's a blackout. If you don't have any generators or battery backups, you can't make or receive a call.

If you think that the advantages outweigh the disadvantages, then it's probably time to get a VoIP. If you live and work abroad away from your family, it's good to have a broadband connection in both your homes. This way, you can keep in touch with your family more often for less cost.

VoIP calls can also be made from your personal computer to regular phones or a cellular phone, but it all depends on what you and your provider agreed upon. But it usually comes with an extra fee, but if you try to compare it with your current landline fees, it's still a lot cheaper.

Do you want a residential VoIP? Get one now, and enjoy making international calls together with your family and friends.

The state of Michigan is one of the most preferred locations by most Americans to retire or to raise a family. In this state, you can expect finding the job you want, finding good schools and educational benefits for your children, and it is also a place where everyone can enjoy.

Michigan is famous for the Great Lakes. Because the state is situated next to it, you can enjoy great boating experience with your family. This state is also famous for fishing. So, if you enjoy boating and fishing, the State of Michigan is the place you would probably want to live in.

In Michigan, VoIP is now readily available. If you live in Michigan, and you already have an active high speed internet connection, you too can avail of the VoIP services. First of all, you need to understand what VoIP is and how you can benefit from it.

VoIP stands for Voice over Internet Protocol and is sometimes referred to as Internet Voice. This particular tool is very much like your conventional landline telephone. However, unlike your conventional landline telephone, VoIP will be able to provide better quality and cheaper communication with your family, friends and your business associates.

One great feature of VoIP is that long distance and overseas calls are much cheaper than conventional landline phones. In fact, when you compare a 5 minute overseas call from a conventional landline phones to a VoIP phone, you will see that the overall cost in the call made from VoIP is only a fraction of what you will spend on conventional landline phones.

Another feature of VoIP is that it offers features, which you would consider as additional features in conventional landline phones, as standard features. Features, such as three-way calling, caller ID, return call, call waiting, audio conferencing, and video conferencing are provided as standard features in VoIP. This means that you will never pay a single cent to use these services unlike in conventional landline phones where they will charge extra for each of the features mentioned.

As you can see, VoIP can definitely give you a lot of value for your money. In fact, VoIP offers are so affordable that some people consider it as too good to be true. If you want to take advantage of VoIP and have one in your own home, you can consider calling your nearest VoIP service provider in your area and inquire about their services.

Today, there are quite a few VoIP service providers existing in Michigan today. If you live in the State of Michigan, you should consider subscribing to one of these service providers in order to use VoIP right in the comforts of your own home or in your business in Michigan.

Here are the different VoIP service providers existing in Michigan today:

- Packet8 VOIP Michigan
- Sunrocket Michigan
- Iconnecthere Michigan
- MyPhoneCompany Michigan
- Vonage Michigan
- Verizon VoiceWing VOIP

All these companies offer great VoIP services and offer it at a very low monthly fee. Some of these companies offer an IP phone for free or you can purchase one at their company if you need additional IP phones.

Michigan also takes advantage of the E911 feature for VoIP. The Enhanced 911 feature is the solution for emergency calls made through VoIP phones. The state of Michigan is now pressuring VoIP service providers to have this feature integrated in their VoIP system.

Depending on your DSL connection in Michigan, VoIP will work very effectively and efficiently. So, if you plan on getting VoIP for your home or for your business, you should consider checking your DSL connection first and determine if it is able to provide quality service to avoid low quality transmission and reception of VoIP data signals through the internet.

So, if you want a phone that is able to offer great value for your money, or if you want a phone system that offers very low rates for long distance and overseas calls, VoIP will be able to provide you with this service. In Michigan, VoIP is now readily available. All you have to do is call the company that appeals to you most and also the company where a lot of people are subscribed in. A VoIP service provider with a lot of subscriber means free calls and also means that it is trusted by a lot of people.

Voice over Internet Protocol (VOIP)

, a technological breakthrough on communication, allows you to route voice conversations using a broadband internet connection or any Internet Protocol-based network instead of the typical phone line. VoIP is also known by the names: IP telephony, broadband telephony, internet telephony, voice over broadband or broadband phone.

There are numerous advantages on using Voip over the standard phone line. Primarily, you can automatically receive incoming phone calls whenever you are connected to the internet. This is regardless of the network you use. While talking on the phone, you can simultaneously use the other programs in your computer. And even if you are on a trip, you just need a mobile phone that connects to the internet and it's as good as you're home. Incoming calls due to your home will be treated as local calls wherever you connect to the internet.

Another advantage is that VoIP to VoIP call is usually free of charge. However, VoIP to PSTN call is charged to the VoIP user.

In addition to VoIP service advantages especially to those who have stable internet connection is that it offers faster access and delivers quality voice data.

To top it all, there are added features integrated in VoIP phone services which are not available with the traditional analog phones. These include video conversation, conferencing, sending or exchanging files parallel to the conversation, managing your address book and more.

How does VoIP work? The service converts your voice to digital signals which will travel through the internet to its destination. VoIP services allow you to make calls by a computer or a VoIP phone. A traditional phone can also be used to make and receive calls on cases when the computer is turned off.

The rudimentary requirement of course is a high speed broadband internet connection. A special type of VoIP phone is plugged directly into the broadband connection and thus, it can largely operate just like the traditional phone. However, in case you want to use your traditional telephone, you will need to connect it to a VoIP adapter. The service provider will provide the dial tone. When using the computer, you will need the VoIP software and microphone to make calls.

How does the VoIP system work for long distance or local calls? Normally, you will be asked to choose an area code which maybe different from the area code of where you live. This will be the determinant of the long distance or local calls. If the person calling you is outside the area or the service covered in your VoIP, he will be charged for a long distance call. Nevertheless, there are variations depending upon the terms of your VoIP provider. In fact, there are VoIP providers that offer a flat rate for a certain time limit anywhere you call.

So who can you call with your VoIP service? This will again depend on the terms and services of the VoIP provider. Basically, you will be able to call anyone with a telephone or mobile number. This includes local, long distance or international numbers. The person you are calling will no longer need any special device other than his phone to receive your VoIP call. There are VoIP providers that allow simultaneous calls or voice conferences.

The disadvantage of VoIP phones is that some services don't work when there is power outage. For this reason, VoIP providers usually provide the backup power. Another disadvantage is that you will need to be constantly connected to the internet in order to receive or make calls. But if you want to continuously receive calls, you can divert the call to the analog phone or mobile phone. Another course is to activate the voice mail box where your messages will be stored when you can't attend to them.

Since not all VoIP providers offer the directory assistance or white page listing, choose the one that does. Make sure also that your VoIP provider can directly connect you to the emergency service 911. Nonetheless, calling 911 through your VoIP phone will be treated differently from that using the traditional phone. In any manner, what counts is that you can readily seek for help for instances when you need them.

With a VoIP phone, you are connected everywhere in the globe.

Using the VoIP or Voice over Internet Protocol, people can place their voice call on the internet through IP (Internet Protocol) Packets. Packets are data bits that are routed or passed from one node to another until the final node receives it. The packets are returned back to the analog voice pathways via final node that is why voices are heard on traditional phones.

You can have two phone service options, the VoIP and POTS (Plain Old Telephone Service). These switches telephone calls on the speediest connection. Telephone calls done through POTS create a dedicated, static connection which travels on similar exact routes for both directions. The connection remains open unless it is broken by the caller. Telephone calls done through VoIP systems via packet switches can carry the complete address of the call's destination. Different routes and time allocation can be taken by each packet while traveling on its pathways.

The user needs to have an internet connection of 256 kbs (high-speed) or higher and a phone broadband adapter to make telephone calls through the VoIP systems. Connect the telephone adapter on the existing system via modem or router. New broadband adapters have built-in routers. Connect this to the high-speed modem then plug the router or computer on the broadband telephone adapter. The installation is very easy.

In most cases, it is favorable to use the broadband telephone adapter with built-in routers. The user can make a phone call on the computer while the router is giving the needed bandwidth on the phone call reducing lag times as well dropped calls frequency. VoIP systems today are much better compared to the technology before. Dropping of calls can still be done when the network is very slow.

VoIP phone

services are practical for small business and home users. However, it will not be an ideal choice for large businesses since stability is still lacking. Keep in mind that every time the internet services go down, the phone services also does.

There are many fuss and buzz concerning VoIP services and it might be difficult for you to understand everything. Enhancements have been made on the system to solve clarity conflicts during phone calls. Some users said that online telephone calls are noisy and have lesser quality compared to traditional phone lines. VoIP providers ensure that their new systems gave the users an amazing experience similar to using normal telephone lines or surpassed it even better. Calls from one continent to another also offer that same voice quality.

Another concern is the VoIP reliability in case of power failures. It was a great problem faced by most VoIP users before. But now smart systems are used to redirect calls to the secondary telephone number once the system was detected to be offline. This advancement can transfer your directed call on the required alternative numbers like your personal phone number.

VoIP system will enable the users to save money on different services. Traditional phone companies offer limited packages only. It includes prepaid and postpaid for business and residential accounts. However VoIP service providers offer plenty of packages providing you great options to choose from. Most often, companies are supplying plans including the lowest and basic plans, unlimited plans, business plans, best value plans, and others. You can choose according to what type of reference calls you make. Several packages also supply affordable rates on long distance calls compared to traditional phone regular rate. You can go around to your phone company to make long distance phone calls without any charge.

Previously, you are charged for services including three way calling, caller ID, call waiting, last call return, and repeat dial. But with VoIP, you are no longer required to pay for these services. Besides, with the filtering options offered by some providers, you can control several calls according to its numbers. It includes forwarding calls to a particular number, sending call to a voicemail, giving a busy sign, playing a message that you're not in service, or sending the caller to a rejection but funny hotline.

This breakthrough is the best way to a better future of telephone systems. It really makes sense when talking about ROI, both on the economic and infrastructure viewpoints. VoIP is truly a great phone service.

When you are subscribing VoIP service, you can have access to the world for a lesser cost without compromising the call quality. Moreover, you can also enjoy the features offered by VoIP phone companies which are not possible with the regular analog phone.

There are few things to know about the VoIP system. The operation is simple and largely similar to the traditional telephone.

VoIP or Voice over Internet Protocol is also referred as the broadband phone. It uses the vast power of high speed internet to transmit your voice by converting it to digital signals which will be processed by the computer software. The software will be provided by the VoIP phone company.

VoIP services can be classified as either phone-based or computer-based. When using the traditional phone, it must be connected to an adapter to make or receive calls. On the other hand, you will also need a microphone and the software to make calls with your computer.

Contrary to the traditional telephone, when you make a call through your VoIP phone, your call is manipulated through the high speed internet connection and not on copper wires. In this manner, you get faster access and better call quality.

The advantage of availing VoIP services is that you can take it with you when even when traveling as long as you have the adapter with you and there is an active internet connection. In this manner, you can make and receive calls as if you are in your business place or at home. Plus, you won't be charged for an additional fee for this convenience.

To make a call, simply dial the number through your VoIP phone just like when placing a call with the traditional landline. As with your computer, you make a call through your headset and microphone which are plugged into the computer. Most VoIP phone companies allow you to make calls to any local, international, long distance or mobile number.

When you have calls, the VoIP phone rings or the computer software will alert you.

The quality of voice when using the VoIP service can be abreast or even exceed the level of quality of the traditional landline. This will depend greatly on the quality and speed of your broadband connection so make a good choice of a partner company.

You might wonder, will it necessitate you to own a computer to avail of VoIP services? Well, you won't have to. The VoIP provider provides the adapter, which is often free of charge and is used to connect the phone to the broadband. However, the computer is a helpful tool for you to utilize the additional features of your VoIP phone company.

There are differences in terms and services for every VoIP phone company so it is wise to read credible reviews. You will learn more about the company's pros and cons straight from their subscribers.

When choosing a VoIP phone company, make sure that it will allow you to call even a regular phone network. Not all providers offer this. Also, make sure that you can enjoy the conference calling, a VoIP feature that allows you to speak to more than one person at a time.

Surely, your primary consideration is the call rate which varies again with each provider. Anyhow, VoIP has comparatively low long distance rates compared to the same call placed through the analog phone. Additionally, there are also VoIP providers that offer free and unlimited long distance calls.

To help you decide on which provider you will subscribe, VoIP companies offer a trial service for a month. This means that you will not be tied up to the conditions within 30 days plus the guarantee to return your money if they don't meet your needs.

There are basically six plans to choose from. They are as follow:

- Home
- Business
- International calls
- Computer to Phone
- Computer to Computer
- Cable phone service

Each plan has additional features to offer. Depending upon your preference and needs, choose the plan that suits you.

Always keep in mind that when choosing the provider to work with, your topmost concern should be the reliability and quality. Do not be hesitant to ask for information before signing up.

The telephone is considered as the primary tool used for communication. Almost every homes and businesses in the United States has a telephone installed. The convenience it offers is unsurpassed and many people consider it as one of the most important piece of technology ever invented by man. Besides, with a phone, you will be able to constantly keep in touch with your family and friends at a much efficient and faster way.

Another tool used for communication is the internet. Like the telephone, this is also considered as a breakthrough in communications technology. The internet allows people to communicate freely through emails and instant messenger programs. Another great thing about the internet is you will be able to share your digital files from your computer to another user.

The internet today is also used for other purposes. Your kids can use it to research for their term papers, you can use it to purchase products and services and you can even use it for entertainment by playing online games, listen to streaming music or watch streaming videos.

However, the primary use for the internet is for communication.

Today, another kind of technology is available in the market. This too is considered as a breakthrough in the communications technology. In fact, with all the benefits it can offer, many people are now subscribing to this service and are canceling their subscription on their conventional landline phones.

This communication tool is called VoIP or Voice over Internet Protocol or sometimes referred to as internet voice. Many people ask why more and more people are now subscribing from VoIP service providers for this service and are getting rid of their conventional landline phones. Some even asked what is so special about VoIP.

First of all, VoIP is very much like your conventional telephone. However, VoIP uses the internet to transmit and receive audio data. Also, your voice is converted to digital audio data. What this means for you is that it will enable you to transmit and receive digital audio data at a much faster rate with clearer quality.

The fact that VoIP uses the internet to transmit digital data, long distance and overseas call is much cheaper than conventional landline phones. Imagine calling overseas using a conventional landline phone, you will see that a simple 5-minute call can cost a lot of money. With VoIP, calling the same destination at the same duration, you will see that it only cost a fraction of what landline phones will charge you. Some VoIP providers offer free long distance and overseas call.

Another great benefit that VoIP can offer you is that it will throw in extra phone features, such as three-way calling, caller ID, audio conferencing, video conferencing, call waiting, return call and other features as standard. In conventional landline phones, you will see that each of these features are considered as extra feature and they will charge extra monthly fees for using a specific service. With VoIP, all these features are free.

So, if you are interested in getting VoIP installed in your home, you should consider getting in touch with your nearest VoIP service provider. As you will notice, there are quite a number of VoIP service providers out in the market today offering their services and claiming to be the best.

However, you can never really be sure if a particular VoIP service provider is really capable of providing great quality service unless you try out their services.

This is why many VoIP service providers today are now offering a trial period, usually 30 days, for clients to test the quality of their services. If you are not satisfied with their services, you can simply inform your service provider that you are no longer interested in using their services. If you do this within the trial period, you will be refunded.

Look for a VoIP service provider that is able to offer maximum benefits for their clients. Some VoIP service providers also offer free ATA (Analog Telephone Adapter), free or low cost IP phones and other great offers.

It is recommended that you should apply for a service provider that is already trusted by the community and offers a broad range of great incentives for their clients, such as free long distance calls, unlimited calls to specific countries and other things like that.

By getting the best VoIP service provider, you can be sure that you can definitely benefit a lot and get great value for your money.

With the advancement of communications technology, people are now able to communicate at a much reliable rate. Garbled conversation or delayed data transmission and reception are now considered as a thing of the past and state-of-the-art telecommunications technology are now being used to provide quality services for consumers all over the United States.

Today, one of the most widely used tools for communication is the internet. Through the internet, you will be able to communicate with your loved ones or your business associates through email. The internet is also used for shopping, for conducting business and for sharing computer files.

However, communication is the primary tool that the internet is used for. Because of this, another kind of communication tool is used and is now widely available for people today. In fact, many people are now getting this communication tool and are now replacing their conventional landline phones. This new kind of communication tool is called VoIP or Voice over Internet Protocol.

VoIP offers the latest technology in communication. You may ask what's so great about VoIP that many people are now replacing their conventional landline phones with it. First of all, VoIP works just like your conventional landline phone. However, unlike conventional landline phone, VoIP uses the internet to transmit and receive audio data. This means that by using VoIP, the phone instantly converts your voice to digital data and transmits it over the internet.

Because VoIP converts your voice to digital data and is transmitted over the internet, communication will be much faster and also much more reliable than other forms of telecommunications.

Another thing about VoIP is that because it uses the internet for transmitting receiving digital data, calls are much cheaper than conventional landline calls. If you compare a 5 minute overseas call made between VoIP and a conventional landline phone, you will see that the call made from VoIP is much cheaper than the same call made using the conventional landline phone.

These are some of the reasons why VoIP is now gaining a foothold in the communications industry. In fact, many people are now considering getting rid of their old conventional landline phones and subscribe with VoIP service providers.

VoIP phones are also unique from conventional landline phones. VoIP phones are also called IP phones. Although it looks a lot like your conventional landline phone, it uses the RJ-45 connector instead of the RJ-11 that conventional phones use. The RJ-45 connector is what you use in your Ethernet. With this connector, it can be connected to your computer with an active internet connection or connected directly to your internet connection.

The IP phone is also integrated with a power plug to plug it in your power outlet to provide electricity.

Another way to get a VoIP phone is through the ATA. With this kind of tool, you will be able to convert your analog phone into an IP phone. The ATA stands for Analog Telephone Adapter. This particular tool is an analog-to-digital converter and is connected to the router which provides the internet connection.

Your computer can also be a VoIP phone. All you need to do is purchase a low-cost software program or download it for free over the internet, a microphone and a speaker to communicate.

As you can see, there are a lot of ways you can get a VoIP phone in your house. You have to consider that you should have an existing high speed internet connection in order to effectively use VoIP. It is important that you should have an active high speed connection because VoIP relies on high speed internet to efficiently transmit and receive digital data.

One downfall of VoIP is that it is dependent on outside power for it to work. Unlike your conventional phone line that has an independent power supply inside the telephone lines, VoIP phones will not work in case of power failure or a black out.

However, if you can live with this kind of disadvantage, and you still want to have access to free or low-cost long distance and overseas calls, you should consider getting VoIP.

In the future, you will also be able to see wireless IP phones that will soon come out of the market. If you are in a hotspot Wi-Fi signal, you will be able to place a call just like what you would do in a mobile phone.

VoIP is definitely one of best kind of communication tools ever developed.

Communications has been revolutionized by technology to fit with today's lifestyle. The number one status was attained by VoIP in terms of phone systems giving services to homes as well as businesses. Many people preferred the VoIP systems instead of the traditional telephony. VoIP or Voice over Internet Protocol is a type of protocol carrying voice signals through IP networks. It was an experimental protocol created in 1973 for ARPANET.ce providers.

If you are going to try this breakthrough then you should be convinced that VoIP will truly give you benefits through weighing the advantages and disadvantages. The next thing you will do is to choose your provider. There are small and big VoIP providers which are highly competitive offering attractive schemes and umpteen options. It is recommended that as a potential customer, you need to learn various calling plans and make the necessary comparisons. Moreover, know major aspects such as services, features, prices, quality, and others.

VoIP can provide you with two options. First is using free services like SIPphone and Skype and second is using services that will charge you monthly. The first option is the best if your purpose is to communicate with your friends and family. However, the second option is perfect if you are going to use VoIP as a substitute to your landline system for business and home working purposes. Paid services can offer great package of VoIP quality services without spending too much.

Reference point is important in making a sensible choice of your provider. Create notes specifying the VoIP services that you need. This will serve as your guide when reviewing various packages. Things like conference calls, call waiting, or forwarding are most common on these packages. Another is giving you the opportunity to make free phone calls to Australia and Europe.

There are several factors that you must consider when choosing your ideal provider.

1. Ask if you can make a call to other countries without additional costs. Take for example, you are residing in California but you can call someone in Paris. This will not cost you anything and the person whom you are calling will be charged only the amount for local calls.

2. Asses the reliability of every provider. Some offers might appear to be too good to be true. Make a personal survey by reading reviews, complaints, and blogs on different VoIP provider's websites. Too many unresolved disputes and negative reviews only mean that the particular provider is not a good choice. Carefully check the good and bad features and be always open-minded.

If you've got broadband, you're already using the Internet for data communication. Wouldn't it be great to use it for telephone calls, too?

Internet telephony service providers

(TSPs) get your voice onto the Net, allow you to make and receive phone calls just like traditional phone companies, and tend to shrink your phone bill to boot. Some of these service providers give you a basic, free service that enables you to call other users over the Internet. Others allow you to make toll-free calls free of charge, but charge for local and long-distance calls.

TSPs that allow you to call traditional telephone service subscribers do so by connecting your standard home phone to the Net. Some TSPs also let you use a special piece of software called a softphone to place calls with your PC. To get connected to a TSP, you need a broadband Internet router configured as a DHCP server, a spare Ethernet port either on your router or on a nearby switch, and a good old-fashioned analog telephone.

TSPs are data centers with telephony servers that route calls to and from your home network or broadband VoIP device. The real-time packets that carry each call's sound over your broadband link use IP and User Datagram Protocol (UDP) protocols, and the TSP communicates key moments in the calllike dialing, connecting, and hanging upusing signaling protocols that are similar in some ways to the ones your browser uses to surf the Web.

The VoIP device that most TSPs provide to connect your home phone is known as an analog telephone adapter, or ATA. These little boxes allow you to connect a residential-style analog phone to your broadband Internet connection, and they are normally supplied by your VoIP TSP when you sign up for their service.

In addition to an ATA, some TSPs permit you to place VoIP calls using the following:

An IP phone

These telephones connect directly to an Ethernet network using a patch cable or wireless link. They have an IP address as a PC would, and they communicate with the VoIP TSP's data center over your Internet broadband link.

A softphone

These are software programs that run on a PC and permit telephone-style communication using your broadband link. They appear on your Windows, Linux, or Mac desktop with graphical user interfaces that often resemble a telephone, and they require that your PC have a microphone and speakers.

For this hack, I'll concentrate on connecting to a TSP that provides an ATA, allowing you to use an analog phone to place and receive calls via the Internet. Table 1-1 lists domestic (U.S. and Canada) TSPs that provide broadband VoIP calling.

"Bring your own device" means the TSP allows you to make phone calls across its VoIP network using your choice of equipment, such as an IP phone, a PC, or your own ATA. TSPs that don't allow you to bring your own device will provide an ATA to make the connection

The competition of numerous VoIP providers over traditional phone companies is very tight. Each provider is offering several benefits such as lower costs, improved technology, and quality sounds. There are five leading providers of VoIP services.

- Vonage VoIP offers excellent reliability and innovative solutions. The services are convenient wherever you are. It keeps your number and receives calls with a good reception and quality sounds at minimized rates.

- Packet 8 Service and AT+T can be the best choice if you are looking for internet telephony on voice quality. They have an improved geographic variability minimizing conversational disruptions.

- Verizon's Voicewing and Skype Technologies offer more options and increasingly becoming popular to most internet surfers.

3. Choose a VoIP provider which offers a guarantee of returning your money in case its services fails. There are many established providers having service records that are proven and offer "money back" agreement.

4. Check for a technical support program on 24/7 calls. Know if their services and equipments have upgrading routines and good maintenance.

5. Make a list on what you need with regards to computer specs, adapters, net connection, and other related systems. Give attention to these technical needs as well as on the VoIP system installations. Read posted information and FAQs posted on the VoIP provider's websites.

Choosing the right VoIP service provider

will let you experience their excellent quality in making a call and saving your money.

Whenever you work with telephony, be it desktop telephony apps or full-fledged IP phone systems, you're bound to encounter prerecorded sounds; things like on-hold messages, voicemail greetings, and even elevator music are often generated by computerized telephony applications. You might even need to create sounds that can be used with these apps. Generating your own telephony-ready sounds is a snap using desktop recording software. You can even resample your recordings so that you'll know exactly how they'll sound in a VoIP environment. This way, you can "preview" them.

Sound-effects producers who need to make somebody's voice sound as though it's being heard through a telephone employ a technique called downsampling. This gives recordings that tinny telephone flavor. For a perfect phone-sound simulation, you'd also need to chop the high and low frequencies of the sound using an equalizer tool, but downsampling alone produces a pretty convincing "phone sound." Here's how it's done.

The easiest way to downsample a sound is by using a simple sound editing tool such as Richard Bannister's Cacophony for Mac OS X, or Windows Sound Recorder, which comes with Windows. In essence, you open the sound file, change its sampling resolution to 8 bits per sample and 8,000 samples per second, and then save the file. (On a standard telephone call, there are 8 bits per sample and 8,000 samples per second in the media stream.) This matches your prerecorded sound to the sampling resolution of a typical phone call.

world. However, to discuss their analog in SIP, we will use generic names, which may or may not exactly map to the PSTN or PBX features. Although the IETF does not standardize features or services, many of these services implemented using SIP are described in Johnston [7].

Standardizing the key PBX functions across the Internet may herald a significant disruption in the PBX market, where all products are vendor-proprietary and interoperability (as with the ITU QSIG standard) is difficult to achieve. In addition, PBX phones from different vendors are not interchangeable. IP PBXs based on SIP have the potential of basic standards-based interoperability and also the potential of interchangeable SIP phones for baseline PBX features. We will take a closer look at baseline PBX features.

PBX or CLASS features generally include the following:

¦¦ Call transfer—There are three types of call transfer services (blind, unattended, and attended) that can be implemented using the REFER method [8]. In a blind transfer, the transferor sends a REFER and then immediately sends a BYE and terminates the existing session without waiting for the outcome of the transfer. In it is an unattended transfer, the transferor may keep the transferee on hold pending the outcome of the REFER request. Once the transferor receives notification that the transfer has succeeded, a BYE is sent to tear down the existing session. Finally, the attended transfer involves a temporary conference call between the three parties, in which the transferor knows the exact progress of the transfer. Once the transfer is complete, the transferor can then drop out of the call. The types of call transfer are described in Table 11.3 and in [9].

¦¦ Call waiting—This is a service implemented on single-line telephones. Since there is no such thing as a “line” in a SIP network, this feature does not have an exact analog. However, a SIP phone that offers multiple

“line” behavior would return a 180 Ringing response and initiate alerting even when there is an active session established. The called party can then either place the session on hold and answer the second call, or ignore the second caller.

¦¦ Call hold—This feature has many forms in the PSTN, from a button on a telephone set that simply cuts the speaker and microphone to advanced features in PBX or ISDN systems. In a SIP network, a call is placed on hold by sending a re-INVITE, changing the media stream from bidirectional (sendrecv) to unidirectional (sendonly). Note that older SIP implementations may implement hold with a re-INVITE with a connection IP address of 0.0.0.0 in the SDP. The call is taken off hold when either party sends a re-INVITE with bidirectional media or a nonzero connection IP address.

¦¦ Call park and pickup—In this feature, a call is placed on hold at one location and then retrieved (picked up) at another location. There are a number of ways to implement these features in SIP. Some of them use third-party call control and a re-INVITE, while others use a REFER and then a redirect.

¦¦ Call forwarding—There are three options with this feature: forward on busy, don't answer, and unconditional. Forwarding can be done in SIP either in a proxy or in a user agent, as shown in Chapter 7, "SIP Service Creation." A proxy can translate one URI for another, resulting in a forward that is transparent to the calling user agent. Alternatively, a user agent or a proxy can issue a redirection response (302 Moved). A proxy receiving a 486 Busy Here response can invoke a call-forward-on-busy feature by generating an ACK and then forwarding the INVITE to another URI. A proxy can also start a ring timer upon receipt of a 180 Ringing response, and then send a CANCEL and proxy the INVITE to another URI to implement a call-forward, don't-answer service.

¦¦ Calling line identification—The ability to display calling line identification is a useful feature in the PSTN to aid the caller during alerting in deciding whether to answer a call or to implement automated screening services. For example, a feature could be implemented to block incoming SIP calls in which the calling party has not been identified. The basic functionality is built into SIP to accomplish this, using the From header. However, since the From header is populated by the calling user agent and not by a trusted source such as a carrier, this calling line identification is not verified or guaranteed to be accurate. The use of the Identity header field to provide a way to check the validity of the From header field is discussed in Chapter 9, "SIP Security."

¦¦ Incoming and outgoing call screening—Incoming and outgoing call screening can be implemented in either a proxy or a user agent. ARequest-URI or From header is compared to a list of allowed or blocked URIs, and an appropriate response generated, such as 403 Forbidden, in the event of a call being blocked. The outgoing call screening feature can only be implemented in a proxy if the user agent is configured to always use that proxy as an outgoing proxy. The incoming call-screening feature can only be implemented in a proxy if the user agent is configured to accept only requests from an incoming proxy, redirecting all other requests with a 305 Use Proxy response.

¦¦ Automatic callback and recall—Automatic recall allows a PSTN user to return a missed call based on calling line identification. This is easily implemented in a user agent by caching the From header from the previous failed INVITE request. Automatic callback allows a PSTN user whose call fails because of a busy signal to have the call automatically placed, as soon as the called party becomes free. This can be implemented in a SIP network using a simple presence service, in which a SUBSCRIBE is sent to request notification when the called user agent is no longer busy. The NOTIFY response would then automatically generate a new INVITE to complete the call. This was shown as an example in Figure 6.11 in Chapter 6, "SIP Overview."

¦¦ Speed dial—Speed dial allows a user to place a call by dialing a shorter digit string, often stored in the network or in the telephone set. A SIP user agent can use any speed dial method. Alternatively, a mapping from a "nickname" to a full URI is possible to allow easier "dialing" in a similar way that nicknames are useful in e-mail.

¦¦ Conference calling—Conference calling is described in Chapter 14, "SIP Conferencing."

¦¦ Voicemail—This important service is described in Chapter 12, "Voicemail and Unified Messaging."

As the bandwidth in the Internet core increases, and broadband Internet access is delivering ever higher speeds, so does the Internet performance increase

with regard to packet loss and delay.
Figure 18.3 illustrates how the monthly average packet loss over the Internet has decreased over 10 years by an order of magnitude and is at present in the 0.1 percent to 1.00 percent range except in a very few regions of the world.
Most global ISPs guarantee average packet loss lower than 0.5 percent but actually deliver average packet loss in the 0.1 percent to 0.2 percent range over their networks. Going back to the codec performance in the presence of packet loss in Figure 18.2, it is obvious that average packet loss in the 1 percent range will not adversely affect codec performance in any way. Packet loss bursts may degrade quality for voice, but we have not seen any published data for Internet burst packet loss.
Figure 18.4 shows the global average delay measured over periods of 24 hours and 30 days, respectively.
These measurements prove that because of the very high-speed routers deployed on the Internet, the average delays are close to the delay of the speed of light in fiber-optic cables.

Watching Steve Jobs pitching the digital lifestyle at Macworld Expo is a favorite pastime of Mac enthusiasts. In fact, there's little that Mac users love more than watching the leader of the Mac world tout new developments and cool little tweaks in Apple's flagship iLife applications: iTunes, iPhoto, iDVD, iMovie, and GarageBand. But for all the pomp and circumstance surrounding these ravishing rollouts, Apple seems to have missed a critical component of the digital lifestyle, one that was around long before DVDs or MP3stelephony.

Fortunately, an Italian company called Ovolab has created a really cool application that serves as the missing telephony link for iLife. Phlink is a hardware-software combination that answers calls with a voicemail greeting, logs them, and even allows you to set up AppleScripts that you can control remotely from a touch-tone phone. The hardware piece of the Phlink setup is a USB device with two RJ11-type ports, one for your standard phone line and another for your analog legacy telephone. The software component (available at http://www.ovolab.com/phlink/) consists of an application that looks like iTunes (see Figure 2-10). You get all of this for less than the cost of dinner (at a really nice restaurant).

VoIP applications tend, like email, to have a few servers facilitating interaction on behalf of many clients. In the case of email, those clients are applications like Microsoft Outlook, Eudora, and Apple Mail. But in Voice over IP, clients can be standalone devices, like IP phones and interface boxes (ATAs like those described in Chapter 1), or desktop applications like softphones or instant-messaging apps. The information in this hack will help you decide which VoIP client is right for you.

Some VoIP clients use well-known standards such as the Session Initiation Protocol (SIP) and are designed for use with your choice of VoIP service providers. Others are designed specifically to attach only to a certain service such as AOL Instant Messenger (AIM). Still others are built using open standards but are hard-wired to work with only certain services; Yahoo! Messenger uses SIP but works only with the Yahoo! service. That is, you can't use the VoIP features of Yahoo! Messenger with your own choice of VoIP service providers.

Some VoIP clients are quite functional "out of the box," such as Skype, which provides a user-friendly wizard to sign you up for Skype service and get you logged in. With others, such as X-Lite and GnoPhonewhich are designed for use with your choice of service providers, or even with your own VoIP serveryou really need to know what you're doing to get much use out of them. Since X-Lite and GnoPhone aren't officially sanctioned for use with a particular provider, you've got to know how to configure them yourself.

The telecommunications industry

is facing a lot of challenges. Their landline telephone services are in extreme competition with today's voice over IP or VoIP. VoIP is telephony's future.

Through voice over IP, you can make unlimited calls at very low cost as compared with traditional phone lines. If you have a fast internet connection, you can enjoy the benefits that VoIP offers.

Both the commercial and residential owners are slowly recognizing the many advantages that VoIP has to offer. That is why the telecommunications industry is trying to attract their present customers with attractive offers. This competition brought about big improvements in VoIP making it the most promising tool to keep in touch with friends/family and to save on monthly phone bills.

Now, with the many improvements that took place, VoIP offers reliability and call quality, and if you're a business owner, you're company can benefit a lot from Voice over IP. But in terms of scale, are you quite sure that VoIP providers can meet with your demands?

Residential VoIP services are consumer-oriented; and to attract more customers, they offer low-cost price packages. There are VoIP providers like Packet8, Vonage, and Lingo, and they offer $20 per month on unlimited calling. Residential telephone lines are more expensive than VoIP so more consumers prefer the services of the latter. Competition is increasing even more because cable companies are also trying to enter the scene. Because of this situation, the prices are getting lower; and this is to the advantage of the consumers.

Even if you're in the commercial sector, you can still make use of consumer-grade voice over IP especially if you only have a few employees. You will only need a few VoIP lines; and to distribute these lines, you can use extensions all throughout the building. The question is how you are going to do it; it's quite simple. There are two ways. One sure way is using a cordless telephone system with multi-line and handsets. So each of your key employees can have two phone lines (landline and a VoIP line). The VoIP line should be plugged into your base station.

Another way is to totally disconnect the landline phone service and use the existing phone wiring in your office. You will need to connect the analog phones into jacks in order to use the VoIP service.

As you're business grows, having a consumer-oriented VoIP line is no longer feasible. Only small businesses can benefit from it and their packages are quite limited. If you were thinking of expansion, perhaps it would be best to look for the right commercial VoIP provider. You must be able to find a dedicated line that offer transfer rates and guaranteed uptime.

Now it's time to look into commercial VoIP providers. With little investment, you can start with a service under the consumer-level. Long term contracts are not needed, nor are expensive VoIP equipments. When you finally decide to upgrade, it doesn't really matter because you're initial venture into the commercial VoIP service will not be costly.

Some of the most famous commercial providers are Primus, Avaya, Covad, and Verizon. If you're business opted for a commercial VoIP connection, there are also advantages.

Almost all VoIP providers in the commercial sector give failover to PSTN, especially if the quality of VoIP service falls below the standards. They also provide multiple lines for VoIP that can be managed centrally. Some business owners' use the modular approach so that when you're business grows, it's easier to increase the phone lines.

Most medium to large businesses have PBX infrastructures; and the equipment of commercial VoIP providers can be easily integrated. You don't have to re-train your employees in using the voice over IP because the switch is transparent.

As the business grows, another issue of much concern is security. What if in the future your business will venture into other regulated fields or in government jobs? With a commercial VoIP provider, security is not a problem because they are now using encryption technology in their service which prevents hackers to intercept packets.

When you finally decide to select the right VoIP provider, service features and price should not be the only consideration. You should make sure that the commercial provider is scaleable enough to meet the demands of your expanding business. Consider your future needs; after all, the goal of businesses is to expand.

If you're used to having a computer with an internet connection every day, it's very difficult to lose such a helpful gadget. Having an internet connection provides you greater access to certain things like online shopping, online chat-rooms, dating, e-mails, and instant messaging.

Not so long ago, another new technology was introduced into the internet world. This technology can make a great change in the telecommunications industry. But this should not be taken negatively by the industry; instead, this should serve as an avenue for them to make improvements in order to stay competitive in the world market.

Through the internet, many people are benefited; especially those people who live far apart. You can even make friends all over the world. This new technology is called voice over internet protocol or VoIP. Now, voice information is delivered effectively on the net.

Major VoIP providers like Cisco, 3Com, Netspeak, and VocalTec made efforts to promote VoIP, directory service usage, and the use of signals like touch-tone for automatic voice mail and call distribution.

If you frequently make calls, local or international, then you might consider getting a VoIP service. It can cut down the cost of your telephone bills for you to save money. What you need is broadband connection, and a VoIP service provider to start enjoying its benefits.

But in choosing a particular provider, one must gather information about the different VoIP providers available on the internet or in your locality. Major VoIP providers include:

- Vonage
- Packet8
- SunRocket
- Verizon
- Viatalk
- Lingo

There are still other providers, and with a lot of them offering their services, choosing the perfect one can be quite a difficult task. This is where VoIP reviews will come in handy. Reviews can help you see the up and down sides of a certain provider. There are reviews conducted by the provider in their websites, but it would be much better to look into independent reviews as they are not biased compared to the other one.

In looking into the reviews, you must ensure that the VoIP provider offers call quality at a reasonable price and other features as well.

Different providers also have different call quality. Since VoIP is new, it has its downside. So you should make sure that the provider can meet this standard.

Most VoIP providers have a range of price plans. Then there are also basic plans, so don't forget to look into these things before you sign-up. You will not be confused with the plans because it is somewhat similar to the plans offered in cellular phones.

VoIP provider

offers added features that you're traditional phone can't give like voice mail, call forwarding, account management, and other features. Different providers have different features, and sometimes it would depend on the plan you choose.

Here are some reviews of VoIP providers:

1. ViaTalk - provides inexpensive service with dependable call quality; they offer plans like other leading VoIP providers at a cheaper price; US-based call center and very few complaints
2. Vonage - their rates are slightly higher than other providers because of their best call quality and more features
3. Packet8 - much cheaper than Vonage; you can add video on your calls; call quality is very reliable
4. SunPacket - a lot of features for a low price; can give two phone numbers for free; you can cancel your account anytime you like and they have free trial periods

VoIP is a new technology that is still undergoing improvements. And so, in choosing the right provider, you have a lot of things to consider. There is no one best VoIP provider. These companies are in the market because they have proven their competitiveness. Your job is to identify your calling needs and your budget for making/receiving calls and choose the appropriate provider that suits you best.

If you're through identifying your needs and wants in a VoIP provider, then you're ready to look into VoIP reviews on the internet; there are even websites where you can find reviews on almost every provider available. A little research can go a long way.

If you're tired with your old phone and its gigantic bills, then start using VoIP in calling your family and friends.

Voice over Internet Protocol, or VoIP, has been a subject of boiling issues for the past years. Many businessmen from large corporations have turned to VoIP and have made savings of at least 50 percent with their telephone bills.

VoIP has grown in demand because it can assist users with their tasks. This is one thing which may not be available from traditional networks. Some of such tasks include the following:

* In UK, the US, and other member countries of organizations like VoIP User, phone numbers are available for free use.
* Incoming calls can be controlled to be automatically forwarded to a VoIP phone, wherever you may be connected to the network. So, a traveling businessman can still receive incoming calls by just connecting to the internet with the use of a VoIP phone.
* With the use of VoIP phones, call center agents can do their work from any location with a stable and fast internet connection.
* Various VoIP packages offer free features such as Caller ID, Call Forwarding, 3-way Calling, and Automatic Redial.

VoIP solutions entail VoIP problems. For the past years, the problems that have stirred up on the use of VoIP focused on the voice signal quality. If IP packets are delayed or lost at any point in the VoIP network, users will experience a brief drop-out of voice transfer. This case is obvious where there are long interworking and distances between endpoints and in congested networks. But this has changed. Many companies have sought for VoIP solutions for a clearer signal that has even surpassed that of the present analog landline service.

Another VoIP drawback is that sending faxes is difficult because of networking restraints and software limitations. However, a VoIP solution is in progress to outline an IP-based alternative for Fax-over-IP delivery. This is called T.38 protocol. A substitute solution to fix this drawback is by treating the faxing method as a message switching mechanism that does not require real time data broadcast. Examples are sending fax as a remote printout or as an email attachment. The incoming fax data can be completely buffered by the end system before being displayed or printed.

However, there are three main issues that concern VoIP solutions before VoIP can dominate the world. These are mobile VoIP phone service, power outages, and Emergency 911 services.

The first VoIP solution that needs to be directed is a way to substitute voip phone service to cellular phone service. Presently, wireless VoIP, or wVoIP, is reliant on the reach and location of Wi-Fi hotspots. Unfortunately, Wi-Fi hotspots have lower reach compared to the present microwave systems of cell phones and may not be practical on a larger range.

A VoIP solution for the problem of replacing a cell phone service can possibly be an advanced cellular-VoIP service. Presently, there a few manufacturers that can produce dual-purpose phones. Such cell phones work as standard cell phones; but when the user is within a hotspot, the cell phone can get connected with VoIP. This type of solution will help the users save money, particularly when they are in hotels, airports, cafés, and other hotspot areas.

Another drawback of VoIP is the irregularity of the Emergency 911 service. Not all current service providers of VoIP provide full Emergency 911 service. But, this has changed as the Federal Communications Commission, or FCC, has taken an action on this dilemma. It has directed all the phone service companies to provide the standard 911 service, even if the customers have not specifically requested to avail the service.

Another VoIP dilemma that has to be prevailed over with VoIP solutions is the power outage concerns. During a power outage, the local phones are still operational. But VoIP requires high-speed internet connection. Without power, the internet access is lost, and so with the VoIP connection. The use of cell phones is the current alternative workaround during residential power outage. Many VoIP companies and providers have started to install power backups in the VoIP hardware sets to attend to this issue.

But these VoIP shortcomings would not stop the VoIP technology in its widespread deployment. VoIP solutions are coming. Don't get left behind! Do you have VoIP already?

The continuous revolution of technology is steadfastly converting our world into a better, more comfortable place to live in. Before, people only communicate through mails delivered by hand from one place to another. Years of innovation paved way to the use of telephones, beepers, and cellular phones. Communication technology's significant breakthroughs are still spewing out from the brains of the experts, and so today, talking to anybody anywhere around the world is made cheaper with the use of a computer and an internet connection.

Ccomplex modern VoIP phone systems are made of several interlinked components. For most businesses, it is as complex and ambitious as they desire to let every employee have a phone with an extension. Furthermore, it requires voicemail for transferring calls and so forth. Unfortunately, this is not the way VoIP companies provide phone service. Modern business phone system is complicated and difficult as it is made up of three components or more.

The main components for having a VoIP phone system are the phones, the service, and the connection. The phones are the real units you hold for talking. The service refers to the company that makes the whole system work and connects phones to the universal telephone network. The connection are the wires or servers, gateways, switches, and routers that are running full tilt and connected to the Internet and to the PSTN.

A broadband or a high speed Internet connection is one main requirement for having VoIP. Such connection can be managed with a cable modem or high speed services, such as a local area network or a DSL. Your computer is an important equipment needed. The adaptor or specialized phone is also required.

The three simple components can cover a big deal of complicated situation. Such complexity is dealt with by having time and money invested in understanding and maintaining the phone system. You may probably need to pay someone else to do it for you.

Today, VoIP phone systems provide telephone calls, where a fraction of the call routes over an IP network. Part of the call would travel over the Internet by a public or a private route. Once you are calling a regular phone, a part of the call will as well travel over the regular public telephone system or PSTN.

The IMS is a layered communications architecture that leverages a range of pre-existing industrystandard protocols such as Session Initiation Protocol (SIP), H.248 and SIGTRAN. Of these, SIP is the most important, being the signaling protocol used throughout the IMS for all call/session control. It is very important to note that the IMS only details multiple functional entities (or service functions) that are the building blocks of the infrastructure. Each of these functional entities has well defined open interfaces associated with them in order to make it easier for service providers to deploy solutions that provide the necessary function and/or interface into the various parts of the system, as well as for vendors to develop “best of breed” solutions that make up the architecture (at least in theory). Signaling and media paths have been separated, similar to the SS7 telephone network, and many functions have both a signaling and media component associated with them.
The IMS standard includes specifications for interfaces with both IP networks and “legacy” fixed line and wireless networks. For IP networks, IMS supports SIP calls from endpoints across either public or private networks. Calls across the public network use a border gateway function (typically implemented today in integrated Session Border Controller products) to provide the control needed to route IP calls through the user’s and/or provider’s firewall.
Calls from the legacy PSTN enter the IMS network through the normal SS7/TDM channels. Inside the IMS cloud these calls are mapped down to the common SIP and Real Time Protocols (RTP) that are the underlying signaling and transport mechanisms of the core IMS network.

Not all regions in the world enjoy broadband Internet access, and in some countries there is just not enough bandwidth to support acceptable VoIP. The lack of bandwidth has actually led some Internet experts to be concerned about the collapse of traffic if VoIP becomes more widespread in such bandwidth-starved networks [11].
We have not seen, however, any reports of such congestion-based collapse, and the proliferation of broadband and video in most parts of the world makes any noticeable large-scale congestion collapse because of VoIP highly unlikely.

A peer is a platform-specifi c implementation of a Java API. In the case of JAIN SIP, the peer corresponds to a particular vendor's SIP protocol stack—that is, the actual software that implements SIP. We have represented it with some pieces of machinery to illustrate this fact.
A provider provides functions using the platform-specifi c capabilities of the peer with which it is associated. In the case of JAIN SIP, the provider allows applications to send and receive SIP messages. It is represented by a Java interface called SipProvider.
In addition to SipProvider, there are other two interfaces that also provide access to some aspects of the underlying SIP implementation:
¦ the SipStack interface
¦ the ListeningPoint interface

Conference Control
The conference control protocol provides for data manipulation and state retrieval from the conference object. It allows us to create/delete/modify conferences, add/delete users, add/delete/modify media, put participants on mute, alter the gain media streams, assign roles to participants, create sidebars, and so forth.
The XCON framework does not specify a concrete conference control protocol. At the time of writing, there is not yet an IETF standard conference control protocol. An attempt to specify such a protocol was done in [draft-levin-xcon-cccp], which is now expired.
Much of the fl exibility in the XCON architecture comes from the existence of a specifi c protocol for conference control. In the architecture for basic media services and in the SIP conferencing framework, there was no such protocol, and the conference management functions were performed by SIP. For instance, a conference was created when the fi rst user joined; also, a participant was able to request that another user is joined to the conference by sending a REFER request to the focus. These capabilities are very limited in nature. Thanks to the utilization

As we saw in previous sections, enhanced conferencing applications introduce a quite broad set of new requirements that cannot be met by the simple architectural approach used for basic services. In the previous two sections, we saw two frameworks to cope with these stringent requirements. These frameworks defi ne a number of functional entities and the interfaces between them. There are different ways to group these entities into physical elements. In some cases, all the elements (focus, notifi cation server, mixer, and so on) are implemented in the same box, whereas, in other approaches, all the entities except for the mixer sit at an application server, and the mixer is part of a separate media server. A possible physical instantiation of the XCON architecture following this second approach is shown in Figure 19.17. In this picture, we see an interface between the application server and the media server.

DoS attacks, whether they are intentional or unintended, are the most difficult VoIPrelated threat to defend against.The packet switching nature of data networks allows multiple connections to share the same transport medium.Therefore, unlike telephones in circuit-switched networks, an IP terminal endpoint can receive and potentially participate in multiple calls at once.Thus, an endpoint can be used to amplify attacks. On VoIP networks, resources such as bandwidth must be allocated efficiently and fairly to accommodate the maximum number of callers.This property can be violated by attackers who aggressively and abusively obtain an unnecessarily large amount of resources. Alternatively, the attacker simply can flood the network with large number of packets so that resources are unavailable to all other callers.

In addition, viruses and worms create DoS conditions due to the network traffic generated by these agents as they replicate and seek out other hosts to infect.These agents are proven to wreak havoc with even relatively well-secured data networks. VoIP networks, by their nature, are exquisitely sensitive to these types of attacks. Remedies for DoS include logical network partitioning at layers 2 and 3, stateful firewalls with application inspection capabilities, policy enforcement to limit flooded packets, and out-of-band management. Out-of-band management is required so that in the event of a DoS event, system administrators are still able to monitor the network and respond to additional events.

Theft of services and information is also problematic on VoIP networks.These threats are almost always due to active attack. Many of these attacks can be thwarted by implementing additional security controls at layer 2.This includes layer 2 security features such as DHCP Snooping, Dynamic ARP Inspection, IP Source Guard, Port Security, and VLAN ACLs.The fundamental basis for this class of attacks is that the identity of one or more of the devices that participate is not legitimate.

Endpoints must be authenticated, and end users must be validated in order to ensure legitimacy. Hijacking and call interception revolves around the concept of fooling and manipulating weak or nonexistent authentication measures.We are all familiar with different forms of authentication, from the password used to login to your computer to the key that unlocks the front door.The conceptual framework for authentication is made up of three factors:"something you have" (a key or token),"something you know" (a password or secret handshake), or "something you are" (fingerprint or iris pattern). Authentication mechanisms validate users by one or a combination of these.Any type of unauthenticated access, particularly to key infrastructure components such as the IP PBX or DNS server, for example, can result in disagreeable consequences for both users and administrators.

Eavesdropping or snooping

Eavesdropping or snooping is a passive attack where the adversary monitors network traffic and tries to get information about the system. In applications that do not use encryption such as telnet and ftp, the account names and passwords are transmitted in plain text. There an attacker with access to the network obtains the tokens that grants admission to the service. Although it is difficult to get access to the network, encrypted authentication schemes are an essential in the Internet.

Impersonation

Impersonation is the attack where an adversary is pretending to be somebody else to get access to a service or a resource. Depending on the kind of authentication that is implemented, the attacker has to show knowledge of some kind of shared secret.

A particular case of impersonation is IP spoofing: Many applications rely on source address authentication. IP spoofing is the technique where this source address is forged. The attacker has to change the behaviour of his TCP/IP stack to carry out this attack. This requires knowledge about networking and good programming skills.

This technique is also used to overcome firewalls. Here, the adversary replaces his source IP address with the IP address of a host of the inner network. The firewall, the system protecting a network connected to the Internet, interprets the packet as coming from the inside network and forwards it to the destination. This attack and other attacks that exploit TCP/IP are described in more detail in section 4.5.

Denial of service attack

An Internet service is a program running on a host computer awaiting connections from clients. A DoS attack prevents the accessibility of such a service. It is an active and direct attack. The attacker does not aim to steal anything. He simply wants to put the service out of order. But not every time a service is not accessible has to be caused by a DoS attack. This can also be caused by misconfiguration or misuse.

Depending on the nature of the enterprise offering the service this can have rather severe consequences, for instance for an online shop. A DoS attack can be of one of the three following types:

• Physical destruction or alteration of network components
• Destruction or alteration of configuration information
• Consumption of scarce, limited, or non-renewable resources

VoIP Protocols

As of this writing, the main VoIP call-control protocols are H.323, Simple Gateway Control Protocol (SGCP), Internet Protocol Device Control (IPDC), MGCP, and SIP. They are defined as follows:
• H.323 is the ITU-T recommendation with the largest installed base, simply because it has been around the longest and no other protocol choices existed before H.323. Chapter 10, "H.323," discusses this protocol in detail.
• SGCP was developed starting in 1998 to reduce the cost of endpoints (gateways) by having the intelligent call-control occur in a centralized platform (or gateway controller). Chapter 12, "Gateway Control Protocols," covers this in more detail.
• IPDC is very similar to SGCP, but it has many other mechanisms for operations, administration, management, and provisioning (OAM&P) than SGCP. OAM&P is crucial to carrier networks because it covers how they are maintained and deployed.
• In late 1998, the IETF put IPDC and SGCP in a room and out popped MGCP. MGCP is basically SGCP with a few additions for OAM&P. MGCP is covered in more detail in Chapter 12.
• SIP is being developed as a media-based protocol that will enable end devices (endpoints or gateways) to be more intelligent, and enable enhanced services down at the call-control layer. Chapter 11, "Session Initiation Protocol," covers SIP in detail.
To briefly explain the various differences between these call-control protocols, let's take a look at how they signal endpoints.
H.323
H.323 is an ITU-T recommendation that specifies how multimedia traffic is carried over packet networks. H.323 utilizes existing standards (Q.931, for example) to accomplish its goals. H.323 is a rather complex protocol that was not created for simple development of applications. Rather, it was created to enable multimedia applications to run over "unreliable" data networks. Voice traffic is only one of the applications for H.323. Most of the initial work in this area focused on multimedia applications, with video and data-sharing a major part of the protocol.
Applications require significant work if they are to be scalable with H.323; for example, to accomplish a call transfer requires a separate specification (H.450.2). SGCP and MGCP, on the other hand, can accomplish a call transfer with a simple command, known as a modify connection (MDCX), to the gateway or endpoint. This simple example represents the different approaches built into the protocol design itself—one tailored to large deployment for simple applications (MGCP), and the other tailored to more complicated applications but showing limitations in its scalability (H.323).

CLASS Features
CLASS is a popular suite of features available to subscribers. CLASS features provide subscribers with a powerful and convenient tool to control incoming and outgoing calls. Telecordia, formerly known as Bell Communications Research (Bellcore), defined the CLASS standard, which added to the custom calling feature foundation. With CLASS, users interact with the switch software from their own telephone sets and give instructions on which services they want. SS7 messages and functions are then invoked and sent within the network to perform the requested operations.
The following list describes some common CLASS features:
• Customer-originated trace—Enables the subscriber to dial a code after he or she receives a harassing call, which notifies the local law enforcement agency.
• Automatic callback—Used when the subscriber receives a busy signal. This feature notifies the subscriber when the called party line is free by placing the call.
• Automatic recall—Enables the subscriber to easily return a missed call.
• Display features—Requires a display telephone to display the calling name and calling number.
• Calling number blocking—Enables the called party to hide his or her identity when dialing subscribers who have CLASS display capabilities.
• Call screening—Enables subscribers to accept, reject, or forward calls based on a list of received calling numbers.

the home, voice will simply be another application in everyone's home. The available features listed in the previous list are just the tip of the iceberg. This chapter discussed enhanced services, where the PSTN operators make a hefty portion of their revenue.
The PSTN offers many valuable services to subscribers and is critical to the operation of small, medium, and large businesses. These subscribers and businesses are, however, increasingly relying on the power and value of data networking and the Internet. To this end,

the PSTN services discussed in this chapter as well as new voice services will, over time, be delivered over data networks and the Internet.

drastically cut the actual costs of renting a building, putting a phone at each desk, and purchasing the required infrastructure (call-routing technology, PCs, and so on) by using a Packet Telephony Call Center (PTCC).

Each call center is different, but for many call centers, the ability to grow the business as needed (perhaps as discretely as one station at a time) is a great benefit. Currently, call centers must grow in chunks. The size of these chunks depends on how many ports the call centers can purchase for their Private Branch eXchange (PBX) at a time. This is a great disadvantage because call centers usually need to be flexible and be able to grow and shrink as the number of required stations changes.

Many call centers are unable to grow in smaller chunks because the hardware necessary to provide desktop phone services is sold only in larger units (such as growing one to several T1s or E1s at a time, instead of a phone at a time). This prevents the call centers from being able to grow quickly based on seasonal or natural growth.

Circuit-Switching Call Centers (CSCCs) enable users to work from home and still take calls, but this equipment is expensive. With PTCCs, users can log in to a phone no matter where they are and have access to the exact same features as if they were at their desk, and the costs are much lower.

A CSCC currently uses a device known as a PBX Extender, a remote piece of equipment that extends the features of the PBX to the user's premises. A PBX Extender can run upward of $1000 per user, and that's just

for the equipment itself. You also have to purchase software that must be added to the central site; the circuit to the worker's residence; and Customer Premise Equipment (CPE) gear, such as the router, for the remote site.

When you use a VoIP network, however, you don't need additional equipment for the remote site. You can take the same phone you use at work and have exactly the same functionality. Of course, the company still has to purchase the circuit to the worker's residence, as well as the CPE equipment.

Nevertheless, VoIP lowers the costs of locating stations anywhere geographically. In doing so, VoIP gives call-center operators a great advantage in terms of hiring skilled or unskilled workers, as well as growing and shrinking the number of stations needed at any given point in time.

In a packet telephony infrastructure, you can have a group of distributed virtual agents that you can locate anywhere, and you can still offer them the same tools that a traditional call center offers. Figure 6-1 shows ways you can use a common IP infrastructure to unite various methods, and it showcases one possibility of telecommuters as virtual agents.

Taking a deeper look into Acme, you need to understand its current network. Acme's headquarters are located in Austin, Texas. Acme has several remote sales and development offices across the United States, as well as in Tokyo and London, where its two largest offices are located. The remaining offices in the U.S. concentrate mainly on sales. Two of Acme's main goals were to cut costs while preparing to deploy a more cost-effective voice network, and to increase bandwidth between sites.

Acme has two intercontinental T1 circuits connected to both London and Tokyo. Multiplexers are used on these circuits to separate 12 channels of each T1 to voice and 12 channels of each T1 to data. The U.S. sites run across a Frame Relay network. The Atlanta site houses a small sales office where from two to five people work at any given time. The Raleigh and San Diego sites have slightly larger regional offices employing both sales people and development staff. Atlanta has a committed information rate (CIR) of 0 and can burst up to 56 K. Raleigh and San Diego both have a 64 K CIR and can burst up to 128 K.

The IS department conducted a study and determined that both data and voice bandwidth needs were growing. The IS department decided to research methods for compressing voice and taking advantage of unused time-division multiplexing (TDM) bandwidth currently utilized by the multiplexing configuration.

The IS department also conducted a study to determine calling patterns. It found that most long-distance calls from all sites are clustered around the various regions in which the corporation has branches.

Acme asked itself several questions to determine whether a combined voice and data network would provide the expected savings.

IP itself is a connectionless protocol that resides at Layer 3 (the network layer), which means that no reliability mechanisms, flow control, sequencing, or acknowledgments are present. Other protocols, such as TCP, can sit on top of IP (Layer 4, session) and can add flow control, sequencing, and other features.

Given IP's relative position in the OSI reference model, it doesn't have to deal with common data link issues such as Ethernet, Asynchronous Transfer Mode (ATM), Frame Relay, and Token Ring, or with physical issues such as Synchronous Optical Network (SONET), copper, and fiber. This makes IP virtually ubiquitous.

You can run IP into a home or business through any means necessary (for instance, wireless, broadband, or baseband). This doesn't mean that when you design a network you can ignore the lower two layers. It only means that they are independent of any applications you put on IP.
IP is considered a bursty protocol, which means that the applications residing above IP experience long periods of silence, followed by a need for a large portion of bandwidth. A good example of this is e-mail. If you set your mail package to download e-mail every 20 minutes, about 20 minutes of silence exist during which no bandwidth is needed.
One of the major benefits of IP is the ability to write an application once and have it delivered through an assorted type of media anywhere, regardless of whether this occurs through a digital subscriber line (DSL) connection in your home or a T1 line in your business.
You can address an IP packet in three general ways: through unicast, multicast, or broadcast mechanisms. Briefly explained, these three mechanisms provide the means for every IP packet to be labeled with a destination address, each in its unique way:

Once we define the requirements and baseline the current network, we can move from the planning phase to the design phase. For convenience, we'll divide this wireless design project into three smaller projects, compartmentalizing the large project into smaller ones to make it easier to provide a solution for each project on a smaller scale.The first project provides wireless access in the satellite buildings.The second project simply adds wireless LAN connectivity to the conference room in the main hospital building.This enables the employees with wireless connectivity through the wireless interface cards in their laptops.The third project adds the wireless links from the satellite buildings to the main hospital and then adds the redundant links between each pair of satellite buildings. Let's review the design requirements:
 Provide wireless access for laptops in all satellite buildings.
 Provide wireless connectivity in conference rooms in the main hospital building.

Telephony is the communication of spoken information between two or more participants, by means of signals carried over electric wires or radio waves. Ever since Alexander Graham Bell invented the telephone circuit and first envisioned the public telephone system, consumers and businesses have relied on telephony as a staple of human interaction.

With the advent of Internet technologies and high-speed data connectivity in the enterprise, a new family of telephony technologies began taking hold. Voice over IP, or VoIP, has significant appeal for the enterprise, for service providers, and for end users, because it allows the Internet and commonplace data networks, like those at offices, factories, and campuses, to become carriers for voice calls, video conferencing, and other real-time media applications. VoIP-savvy organizations are discovering that they can apply the paradigm of distributed, software-based networking to voice applications and enable a new generation of telecommunications features, cost-savings, and productivity enhancements.

VoIP can replace business telephone systems, or it can add value to existing traditional telephony devices. For instance, long-distance connectivity between two offices with traditional telephone systems can often be accomplished with a lower cost per call when VoIP is employed.

VoIP network protocols can serve as a platform for other communication media like text messaging and video conferencing. In fact, you've probably used a flavor of VoIP for such an application by now; they've been popular as an Internet pastime for several years. Yahoo! offers a "party line"-style service that features Voice over IP chat rooms (http://chat.yahoo.com). Apple's iChat and Microsoft's NetMeeting applications also offer text, voice, and video calling delivered through VoIP protocols.

Dozens of standards define how Voice over IP works, but little documentation exists on best practices for implementing and maintaining the technology in the enterprise. There's not much introductory instruction for VoIP, so beginners may have a hard time taking their first steps with it. There have been several high-profile implementation failures among large enterprise adopters, and this may be why IP telephony has such an intimidating reputation.

Nonetheless, if it's done right, Voice over IP can transform the cost model of telecommunications by combining the overhead of voice and data expertise and infrastructure. It can also enhance productivity for end users by introducing new features and for telecom administrators by centralizing management functions. Voice over IP can decrease the expense of future computer-telephony integration projects, while making it easier to link voice systems with web servers and database applications.

This book will give you practical guidance on switching to Voice over IP from traditional telephony systems. It gives a brief introduction to traditional telecom systems, and correlates their features and fundamentals to those of IP telephony systems, while showing ways of integrating traditional telephony assets into an IP-based voice network. It will describe the standards involved, so you can make educated choices among the large selection of components and vendors. It will also help you conquer some of the most problematic issues that people face when building telephony systems with Voice over IP.

Limits of Traditional Telephony

The PSTN's capabilities are largely proportional to its physical connections, because every call must have a circuit, or loop, set up at the beginning of the call and torn down at the end. While the PSTN's switching equipment does a great job of this, some hard limitations are associated with its "circuit-switched" nature.

New features can take a while for the phone company to roll out. It took many years to upgrade central office switches to support features like call-waiting and three-way calling. Even now, some parts of the PSTN still don't support caller ID.

Capacity limits are another engineering challenge on traditional telephony networks. The fidelity of a call's sound reproduction is limited to the available bandwidth between the caller and the recipient, and the maximum number of calls between two offices is limited to the availability of voice circuits that exist between them. The problem posed to the enterprise is one of cost: every PSTN circuit used by the enterprise, be it a POTS line or a T1, adds to its telecommunications expenses.

The telephone companies and phone equipment vendors have made great strides to identify and resolve capacity and cost problems. High-density digital circuits like T1s and T3s have brought the cost of high-density telephony down, and PBX features such as least cost routing (LCR) allow the enterprise to minimize its long-distance calling expenditures. Long-distance calling has become cheaper, and the cost of on-premises PBX equipment and feature-rich business telephones has dropped over time, too.

At one time, telephony features were considered a competitive advantage. As businesses adopted them, they became part of the cost of doing business, and users began seeking a new telephony paradigmone that could inspire big competitive advances again. The question the telecom industry sought to answer was, "Where do we go from here?"

Enterprise telephony innovators began looking to the Internet for the answer. Because of core differences in engineering philosophy and many years of additional discourse on the matter, the Internet is superior in many ways to traditional voice networks.

On the Internet (and IP networks in general), communications protocols are in a constant state of improvement, so more and more features can be delivered while bandwidth efficiency steadily improves and the cost of the network shrinks. On the Internet, capacity is tied closely to the efficiency of software, rather than to the physical capacity of a telephone switch, as with a PBX and the PSTN. As software improves, IP networks grow in capacity, but traditional switches need additional (often expensive) hardware to add capacity.

IP networks have always had another advantage over the traditional public voice network: their software uses standardized hardware components like low-cost PCs. This means that even when hardware upgrades are necessary, they can be procured less expensively. Unlike traditional PBXs, hardware upgrades on IP networks intrinsically improve software productivity and enable more and more capacity. Generally, capacity is easier to scale on IP networks than it is on circuit-switched networks like the PSTN.

While the PSTN is quite reliable, it is far less disaster-proof than IP networks. The Internet Protocol permits redundancy and failover capabilities that are inexpensive and relatively easy to implement. Geographic diversity, a technique used on data networks to circumvent local connectivity interruptions, is very easy for the enterprise to achieve with the Internet, but more difficult on the PSTN. For example, you can connect to two Internet service providers and use the same set of IP addresses with both, thanks to the BGP[*] standard, but it's nearly impossible to use the same set of phone numbers with two telephone companies.

A number of companies offer VoIP calling services that can be used in the home, more or less replacing conventional PSTN service. They deliver telephone calling capabilities using a broadband Internet connection. Not all of them permit placing calls to or receiving calls from the PSTN, but almost all allow you to call other users of the same service using the Internet instead of the PSTN. Some providers even have "peering" arrangements that allow you to call subscribers to other providers' services using the Internet.

Some of these services work only with proprietary telephone-calling software and don't allow you to use a hardphone. Certain providers can support the use of a special hardphone that connects to your PC's USB port and uses the PC as a gateway mechanism for accessing the network. Others provide an ATA device so that you can use one or more analog phones to place and receive calls using the service. Still others offer the ability to use IP phones.

Many of these services offer competitive calling rates, decent sound quality, and features that are close to that of the traditional phone company. There are solutions for adding more features and interesting hacks to a home-based VoIP network, too. Some of them are covered in this book.

Many vendors are producing cost-effective VoIP server devices for small Ethernet LANs. These devices connect together endpoints in a small office like a PBX, either through the use of conventional analog and digital phones (more on this later) or new-generation IP phones. In either case, connecting calls to the PSTN and between local phones is usually the responsibility of these server deviceslike a PBX or KTSand there are several ways to make that PSTN connection.

Using VoIP in small business environments is easier when there's some network savvy around the office. Some traditional phone vendors are now implementing VoIP systems, so the availability of third-party networking expertise is accelerating the adoption of VoIP in small businesses.

There are potential pitfalls along the path to Voice over IP. Implementing VoIP is like any infrastructural investmentit has hard costs and implications for the enterprise user community. How you deal with these costs and social issues is largely defined by how much VoIP you implement at a time. That's why the concept of migrating is important.

Challenges also arise from using a relatively young technology for a task that has been reliably delivered "the old way" for decades. If the data network hosting the VoIP system isn't provisioned correctly, the results can be disastrous. Security, stability, and call quality in a VoIP system are all tied to their counterparts on the underlying data network. If your network is insecure, unstable, or lossy, your IP-based voice system will be, too.

A leading cause of failed VoIP implementations is poor perceived call quality, which usually stems from administrator misunderstanding of VoIP's requirements. VoIP is more than just call management and voice conversations; it is also a comprehensive set of methods to deal with quality of service. Lack of attention to these aspects of VoIP will doom even the most well-intentioned implementer.

These issues have contributed to IP telephony's reputation as difficult to manage, inferior in quality, and even damaging to corporate image. These perceptions can be avoided, and the opposite outcomes achieved, if VoIP is done right.

This book will help you implement and understand VoIP networking, call management, telephony features, and call accounting within the context of an enterprise data network. Along the way, you'll build a useful, real-world call-management system, a voice mail server, and more. You'll employ next-generation VoIP hardware and software, use open source telephony tools, and leverage traditional telephony components. You'll even be able to use that old residential-style analog phone for VoIP calls across the Internet.

You'll understand the differences between old-school and next-generation telephony and be able to implement a software-based PBX, maximize quality of service, and know many of the standards that govern the world of converged networks. You'll be able to identify situations in which traditional telephony can be integrated with IP telephony, and you'll learn how to provision emergency calling services that have always relied on the public telephone system. This way, your switch to VoIP will be a success.

Conversations are the basis of human communication. Conversations can be spoken, written, or gestured. Conversations can even be one directional, such as a coach bawling out his star quarterback after an uncharacteristic interception. Conversations may be "one-to-many" (such as a political candidate giving a stump speech) or "many-to-one" (such as a constituency lobbying that candidate after she's in office). Conversations are more than just an analogy for networksthey literally are modern networking.

The underpinnings of enterprise networks are also conversations. IP data networks run on protocols that use a conversational approach to data exchange. The most common protocols for web browsing (HTTP) and email (SMTP) use a two-way "data conversation" in order to communicate. The process is simple: a client host sends an inquiry to a server host or a peer host, and then the server or peer sends a response back to the client.

Conversations between hosts on an Internet Protocol (IP) network are similar to those between people, except that instead of using words, the messages are communicated across the networks using units called datagrams. A datagram is like a letter in an envelope. Once it has the proper markings, namely the recipient's address and return address, and a stamp, the entire letter can be delivered by the postal service. A datagram's markings are called headers, and they contain delivery information, like postal letters: instead of postal addresses, datagrams use something called host addresses. Different networking technologies have different names for datagrams, including cells, frames, and packets. Having a good understanding of IP networks is crucial to your success with Voice over IP. An excellent reference on the subject is TCP/IP Network Administration (O'Reilly).

When voice sounds are transmitted using datagrams on the IP network, telephony gains all the same characteristics as the data network itself. Just like applications for file sharing and printing via the network, software can be made to perform useful tasks using the datagrams of voice streams and signalstasks like conference calling and voice mail. These tasks are the applications of Voice over IP. Voice applications delivered using IP datagrams is the essence of VoIP.

VoIP, like the network that carries it, is therefore not an application by itself, but a way to build applications using myriad software tools and devices. These building blocks can be specialized VoIP server hardware like an analog telephone adapter (ATA), or they can be highly programmable servers that do the job of a PBX. Regardless, all VoIP components must participate in the protocol conversations that make the audible, human phone conversations possible. That means that all VoIP components must be speaking the same language.

In human conversation, people can speak many different languages. Even among different dialects of the same language, people can have a hard time understanding each othera Bostonian and a Texan sound about as different as a Canadian and an Australian, even though they all speak English. Unfortunately, telephony standards have had similar challenges.

Many standards govern the world of Voice over IP, and some have interoperability problems, just as people with local accents sometimes confuse each other. One such annoyance lies in the definition of the word VoIP itself.

Are "VoIP" and "IP telephony" two different technologies, or do they both describe the same thing? Well, it really depends on whom you ask. Some vendors prefer IP telephony when referring to their IP-based voice offerings, arguing that VoIP refers to the specific act of transmitting digitized sound data on an IP network and IP telephony refers to the overall technology family. Others give VoIP the broader definition, identifying it as inclusive of IP telephony, and referring to IP telephony only as the act of mimicking traditional telephony applications.

For the purposes of this book, we'll take the latter tack: VoIP refers to the overall technology family, while IP telephony means specific application functions such as call signaling and voice mail. So when we talk about conference calling, we might call it telephony, but when we talk about conference calling, call-waiting, and voice encoding, we will refer to them collectively as VoIP. In general conversation, though, VoIP and IP telephony can be used interchangeably.

VoIP certainly has a few disadvantages when compared to old-school phone hardware. High-utilization service guarantees are harder to deliver with VoIP than with an old-fashioned PBX. The same scalability characteristics that attract people to VoIP can ultimately be the reasons their implementations fail: a VoIP network can be so extensible that service-level guarantees are hard to make, whereas a traditional circuit-switched voice network has hard capacity limits, around which levels of service tolerance can be guaranteed easily. Certain broadcast audio applications, like overhead paging, can be difficult with VoIP, too.

The gains VoIP brings to the table far exceed the few difficulties it imposes, though. There's nothing that old PBX can do that a VoIP telephony system can't, even if VoIP makes a few things tougher.

One thing VoIP makes easier is physical provisioning. While a PBX requires a network of electrical, usually copper wire, loops, VoIP requires an IP network. Since IP networks are a staple of every modern business, the logistics of building a network for voice is largely simplified because the required physical elements are already in place for other common business applications: databases, messaging, Internet access, and so on. VoIP is carried on the network the same way those are.

If you're an Internet user (and who isn't these days), then you know TCP/IP is the core protocol that defines the architecture of the Internet. In most organizations, and even in many homes, a TCP/IP local area network is an important interpersonal communications tool, used for email, web surfing, and instant messaging. When VoIP replaces the traditional telephone using TCP/IP, the local area network becomes the key piece of telecommunications infrastructure.

Once that key piece is standardized within the enterprise, VoIP administrators have only one network to maintainthe one that supports TCP/IP. This means supporting a single network cabling system, rather than separate ones for voice and data. If you use wireless Ethernet, you don't need local area cabling at allVoIP will still work. Meanwhile, old-school PBX administrators still have to maintain a separate local area cabling plant that serves only the PBX system.

But that key piece of telecom can be a key failure point, too. When the voice and data networks are separated, as they are in traditional telephony, their physical paths lie separately, protecting the voice system from failures isolated on the data network, and vice versa.

But with VoIP, these paths converge. When the path is broken by an equipment failure, a power failure, or a construction crew accidentally slicing underground cables, the data network fails. When a computer virus swamps your data network, VoIP phone calls may no longer be possible. When data fails, voice fails, too.

Even in the home, where you might rely on a cable or DSL Internet connection, your VoIP calling capability will swiftly disappear when your broadband provider's service fails or your power goes out.

Historically, the quality of phone calls' audio has been measured using the mean opinion score (MOS) from a group of listeners. These listeners hear sound samples from calls of varying quality, recorded during different sets of network conditions. A sound sample from each set of conditions is played for the opinion group, and each rates the sample's quality on a scale of 1 to 5, with 5 being the best quality. The conditions that can be used to alter the quality of a sound sample are choice of codecs, transcoding combinations, packet interval, and packet loss rate.
MOS can aid you in rating the perceived quality of your legacy system. Not all system builders will have the time or inclination to do this. But, if you support more than a few-dozen users or if call quality can vary depending on the call path, you should determine the MOS rating of all call paths on your current system. Ask a group of users to place calls across each call path and then record their MOS opinions of each call.

Do this before you replace your legacy links with VoIP. This way, you can use the scores as a guide when selecting standards and equipment for your new VoIP system. You can also tell whether you've succeeded in replicating the call quality you had before you started to replace legacy equipment.

After the implementation, particularly in large corporate or carrier-class networks, establish an SLA (service-level agreement) between you and your users that provides an MOS expectation for every call path that meets or exceeds that of your legacy system.

Telephony software is everywhere, and a majority of it is free. Much of it is open source. It's a good idea to experiment with as much of this stuff as you can, because the VoIP family of technologies is evolving continuously. The capabilities and efficiency of these tools rise daily.

Whether you use Linux, Solaris, Windows, or Macintosh, plenty of fully developed softphones, servers, troubleshooting tools, APIs, and sample code are available. What follows is a partial list of some excellent VoIP software and service providers. Many of them have been used in projects throughout this book.

We'll also briefly cover the growing list of vendors that manufacture modular and turnkey telephony hardware.

If you listen to all of the media hype, VoIP is the next killer application for broadband. In fact, people are turning to VoIP at an incredible rate. So what's the biggest driver in this movement to VoIP? Price. VoIP technology has managed to do what deregulation of the telecommunications industry could not—it's created competition between telecommunications companies and other players in the industry.

Early predictions that VoIP would completely undermine the $200 billion traditional telephone industry were a little ambitious. Those early predications called for VoIP to be a major means of communications by 2005. The VoIP industry is growing, but it hasn't overtaken other means of communications, yet. Even if those predictions were ambitious and overly optimistic, they appear to be only a few years off track. More recent predictions combined with the current growth rate of VoIP indicate that by 2010 VoIP will be a major factor in the telecommunications industry, and one of the main ways that people communicate.

These updated predictions are much more likely to be accurate, even if no one knows for sure what the ultimate outcome of current shifts in the landscape of the telecommunications industry will be. And no one can know for sure. Far too many factors influence the industry at this time.

For example, the number of service providers in the VoIP industry will most certainly change. The large telecommunications companies are getting involved now, and it's likely they will consume all but the strongest of the smaller companies.

Note

Currently there are about 2000 VoIP companies in the market. Most of those companies are very small with a limited number of subscribers. It's likely that over the next few years the majority of those companies will either fail or be absorbed by larger companies.

What is certain is that VoIP is here to stay. And it will forever alter the landscape of the telecommunications industry. Already telecommunications companies—traditional phone companies, cable companies, and Internet service providers—are feeling the pressure of the VoIP industry. Those that offer VoIP services are seeing growth, those that do not have stagnated. The companies will need to adapt to the new technology or they will find themselves alone in the dust of the Information Age.

One of the first elements of the industry that will change is standards. Numerous standards are currently used within the VoIP industry. But with time, all of those standards will begin to combine and eventually a single set of standards will become the foundation of the industry. There's more to come on standards later in this chapter.

Another factor that will affect the VoIP industry is politics. The current political climate is very favorable for the VoIP industry. However, as the popularity and adoption of VoIP increases, that too will change—and the industry is already beginning to feel some of that change.

The one clear fact about the VoIP industry at this time is that it's here to stay. The VoIP fad of the early '90s has been replaced by the more mature and reliable VoIP of today. Advances in the industry have finally pulled VoIP to the forefront, putting it right in front of consumers whose interest is peaked by the savings, advanced features, and freedom of choice that VoIP offers. The hype surrounding the technology is quickly turning to fact and as reality sets in, the VoIP industry will continue to grow and replace traditional telephone services

Industry standards present one of the toughest challenges for any industry and VoIP is no exception. Industry standards are the benchmark by which companies measure the viability of their service. In the VoIP industry, standards haven't become standardized. In other words, there is no single set of standards that all of the service providers in the industry must adhere to.

Industry standards give an industry or market, like VoIP, a solid foundation upon which all services are built, making it possible for all consumers to access the service being offered without really thinking about it. For example, in the telephone industry, standards are what makes it possible for you to plug a phone into a jack and make a call without considering how the call is initiated, routed, or terminated, or whether the phone will work with the service that you have. Standards also drive some functionality such as 911 or Enhanced 911 services and wiretap capabilities. Unfortunately, the VoIP industry doesn't have a single set of standards, so consumers must know what standards are governing the industry and how those standards affect services.

The lack of cross-industry standards reduces the interoperability of VoIP equipment and services. In other words, some equipment works with some VoIP services, but not with others. This is especially evident in PC-to-PC VoIP and in some instances of VoIP services provided by one of the many VoIP service providers.

In the case of the VoIP industry, standards start with a standard protocol or set of guidelines for how the technology of VoIP should behave. These protocols point to how voice should be converted to digital data; how it should be transported across the network, which in the case of VoIP is usually the Internet; and how the digital data should be converted back to voice. And those are just a few of the elements of the process that are covered by standards.

However, without a single set of standards that every VoIP service provider can rely on, many of the major service providers in the VoIP industry are scrambling to bring the "best" set of standards to the table first. As a leader in industry standard, whichever of these companies succeeds stands to gain the highest number of customers the fastest. So you can see, players in the industry are vying for the frontrunner position by pursing industry standard protocols.

Consumers are pushing standards, too. You want to have VoIP service, but at the same time you want it work like the traditional telephone service that you're accustomed to. Without standards, that's not possible.

In addition, there's unofficial pressure from government agencies, such as the Federal Bureau of Investigations (FBI), for VoIP companies to standardize on specific issues—the wiretap issue being the FBI concern. It's this pressure in combination with inter-industry pressure that is driving the standards that you'll see in the future. And it all starts with the protocols.

VoIP Protocols

The pressure to develop industry standards does have one benefit. It means that service providers are continually working to improve their services to meet expectations and to set the bar for expected service higher than all the other companies providing the same service. The result is a fierce competition that drives a set of protocols that govern how your VoIP service is provided in five areas:

Integration: At least for now, VoIP must integrate with existing telephone service (which is called the Public Switched Telephone Network, or PSTN). VoIP calls are transported via the Internet; however, unless the call is conducted from VoIP subscriber to VoIP subscriber, the call is delivered using the PSTN. The integration of VoIP and PSTN is essential to growth in the VoIP industry. Protocols govern how calls are handed from VoIP networks to PSTN and vice versa.

Interoperability: You'd think integration and interoperability are the same. They're not. Each company that offers VoIP service also offers different types of equipment. Without some guidelines for how that equipment should interact with the equipment that other vendors provide, you could be stuck using only one vendor without the possibility of calling someone else who uses equipment from another vendor. This interoperability of equipment makes it possible for the VoIP industry to expand and will eventually lead to a single device that all manufacturers and service providers rely on. It's like the telephone that you use today. It works regardless of who made it or who provides your phone service. Protocols and standards will ensure that VoIP phones and equipment will eventually reach that point.

Scalability: Scalability is the ability of an application or network to grow and shrink according to the amount of traffic using that application or traveling over that network. Some protocols are designed to govern the scalability of VoIP services. Past predictions of how fast the VoIP industry will grow were overly enthusiastic, calling for growth to happen much faster than it has. Even though the industry and public adoption of VoIP hasn't developed as quickly as those early estimates, it's still happening much faster than the adoption of telephone service did. Revised estimates call for 75 percent of the population to be using VoIP by 2010, a much more realistic view of how quickly VoIP will catch on. Still, without some guidelines governing scalability, the VoIP industry would falter under that kind of growth. These protocols help to ensure that VoIP infrastructure will continue to grow and meet the demands that consumers place on it.

Quality: Quality has been the biggest sticking point for the VoIP industry. Early VoIP services were plagued with poor quality. However, the protocols that govern VoIP quality have come a long way since those first services. It is these protocols that provide guidelines for how voice should be carried over the Internet, which was originally designed to carry data. Protocols that address quality provide guidance for turning voice to digital data and then back to voice, as well as guidance for how that data is transported from one place to another once the change has been made.

Security: The Internet is plagued with security risks. And voice conversations that travel over the Internet are not exempt from that challenge even though VoIP services have not been hard hit by security threats as of yet. It's only a matter of time before it is and VoIP is as easy a target as any other type of information technology service. Once converted to packets that are transported via the Internet, your conversation is subject to the same security threats that any other data that travels over the Internet is subject to. Hackers that capture other types of information can (and do) capture voice data, and viruses that target cell phones or e-mail can (and do) target VoIP. Even denial of service (DoS) attacks are conducted against VoIP services. Without security protocols to govern how VoIP is protected, your security risks could be much higher. Currently there's no consensus on what the best method of security is for VoIP—secure socket layers (SSL), authentication, encryption, tunneling—all of these types of security play a role, and security protocols ensure that some form of security is protecting your conversation.

In the simplest terms these standards have to work together because they dictate how computers find each other on the Internet and how information is exchanged between the computers to allow VoIP packets to flow between destinations. In addition there has to be an agreed-upon payload format so that the contents of the VoIP packet are recognized and properly decoded on both ends of the conversation. This is no easy process. It requires that the technologies surrounding VoIP be flexible enough to serve the many needs of many companies and many consumers.

As you can see, VoIP protocols and standards are needed for numerous reasons. It's not necessary for you to understand the inner workings of VoIP to be aware of some of the most frequently used protocols. You do, however, need to know what those protocols govern and how they affect you because over time they will mature and change, and as they change, your service will change as well.

Two types of protocols affect the VoIP industry: call signaling protocols and device control protocols. Call signaling protocols set up communications between two endpoints or an endpoint and gateway. Device control protocols control the functions of the device itself. Following are some of the most common call signaling and device control protocols in use in the industry.

H.323

One of the oldest of the sets of VoIP protocols in use, H.323 is a call signaling protocol that was recommended by the International Telecommunications Union (ITU) and originally adopted for use in 1996. H.323 is really a set of protocols that provides guidelines for any type of audio-visual packet communications. For example, if you attend a Web conference or video conference, chances are H.323 guides how the conference is delivered to you. The set includes protocols for VoIP, video conferencing, and other methods of data sharing.

The H.323 standard was originally developed to govern video conferencing, but it has been extended to include VoIP. The current version of H.323 came into effect in early 1998, and it governs both point-to-point communications, such as telephone calls, and multipoint communications, such as video or Web conferences.

This standard ensures interoperability between vendors by governing the endpoints of communications—terminals, gateways, gatekeepers, and multipoint control units:

Terminals: The terminal is the endpoint that provides two-way, real-time communications—it's the ATA in VoIP. Under this protocol, all H.323-compliant terminals must support the use of channels, call signaling, real-time transport protocol (RTP), and registration administration status (RAS).

Gateways: The gateway is the interface between the PSTN and the Internet. At the gateway, data is converted from analog to digital or vice versa and then delivered to the receiver, and the H.323 protocol governs how that conversion takes place.

Gatekeepers: The gatekeeper is like the administrator for the gateway. It controls authorizations, addresses, signaling, and bandwidth management, among other things. H.323 outlines the parameters under which all of this administration takes place. Each gatekeeper has numerous endpoints registered with it and it performs these administrative tasks for all of those endpoints in what is called a zone.

Multipoint Control Units: A multipoint control unit (MPU) is an endpoint that allows three or more terminals and gateways to participate in a multipoint conference. Using other elements of H.323, the MPU determines the common capabilities of the endpoints.

H.323 also works in conjunction with numerous other protocols, such as H.225, which describes call signaling and packetization; H.235, which provides guidelines for security; and H.245, which governs opening and closing logical channels for data control and exchange. This type of multi-protocol governance is called stacking. Protocols are stacked together to address numerous issues.

SIP

Sessions Initiation Protocol (SIP) is also a call signaling protocol that operates at the application layer for creating, modifying, and terminating VoIP connections. It was first introduced about the same time that H.323 was introduced, but wasn't recognized as a standard until 1999. Since that time it's gone through several revisions and was republished in its current form in 2002.

SIP establishes a common ground on which two separate connections can meet. So, for example, if you make a call using VoIP to someone who does not have VoIP, SIP is the protocol that finds the middle ground, or compatibilities between your equipment and connection and the receiver's equipment and connection.

SIP helps create a connection by directing the following activities:

Determines the connection point that will be used for communication

Determines how calls will be transferred and terminated

Determines call parameters at both ends of the call and rings the receiver

Determines the availability of the user receiving the call

Determines the capabilities of both the caller and the receiver

The protocol achieves these standards by using two components: user agents and network systems. The user agents are end systems along the network that receive requests and return responses for the user. So, for example, if you're making a call to someone, the user agent gets a small piece of information that tells it you would like to place a call. The agent then processes that information and sends it out across the Internet. It then gets a response that tells you if the phone you're calling is available. If it is, the agent puts the call through. If the phone is not available, the agent returns a response that the call cannot be completed—that response could be a busy signal or a recording of some type that tells you the number is not available.

The second part of SIP is the network systems. SIP determines the communication between each of the three types of servers that make up a network system. Those servers are a registration server that logs a user's current location, a proxy server that forwards requests to the next server down the line, and the redirect server that sends information about the next server in line to the proxy server. Together, these three types of servers make up the path through which data travels from one location to another. Although there are only three types of servers, dozens of those servers might be involved in the transport of your call from your location to the receiver's location. SIP ensures that the call is routed properly and arrives at the destination.

In a lot of ways, H.323 and SIP are similar in function. In fact, fundamentally the two protocols do the same thing. Where they differ is in nature. H.323 was designed for use on a single local area network (LAN) for video conferencing, whereas SIP was designed for use on the Internet for VoIP. H.323 borrows from legacy communication systems and SIP does not. Also, H.323 is a binary protocol whereas SIP is ASCII-based. Because of these differences, H.323 works, but can be cumbersome. SIP, which is built much like the HTTP protocol that is the foundation of the Internet, is much simpler and designed to work with the attributes of the Internet.

Like H.323, SIP works in conjunction with a number of other protocols:

Real-time Transport Protocol (RTP)

Real-time Streaming Protocol (RTSP)

Session Announcement Protocol (SAP)

Resource Reservation Protocol (RSVP)

Others exist as well. And it's not necessary for you to know that RTP is for transporting real-time data and virtually every device uses RTP for transmitting audio and video packets, or that SAP is for advertising multimedia sessions. What's important for you to understand is that SIP is a protocol that was designed with VoIP and the Internet in mind. That makes it more effective and efficient for guiding VoIP services.

Note

RTP is one of the protocols that addresses issues like packet order so that when your voice arrives at its destination it is replayed to the receiver just as it was sent, rather than in whatever order the packets were received (which would result in a strange, garbled conversation, indeed). RTP also provides mechanisms that help address packet delay and jitter, which is the result of that delay.

Many debates exist over which of the call signaling protocols is better. The results of those debates are mixed, and depending on who you speak with you'll hear varying opinions on the matter. Both protocols are endpoint protocols, which means they both provide all the information necessary to locate a remote endpoint and to establish media streams between the sending and receiving devices.

H.323 is superior in some ways—it provides for better interoperability with the public telephone network, it provides better support for video, and allows better interoperability with legacy video systems. SIP, on the other hand, isn't designed to address the problems that can be encountered in legacy communications systems. However, SIP is easier to develop and troubleshoot. SIP also seems to be more efficient for VoIP, and many equipment manufacturers seem to be turning to SIP over H.323.

Of course, the fact that SIP is more popular doesn't mean that all service providers use SIP. Nor does it mean that all equipment will work with SIP. To be sure that your equipment works with the protocol that your service provider supports, ask your service provider about compatibility. Even better, get your equipment from your chosen service provider instead of a thirdparty vendor.

Other Protocols

Dozens of other protocols affect VoIP, but they're parts of the larger picture. For example, the Media Gateway Controller (MEGACO), which is also called H.248, is a device control protocol; the Media Gateway Control Protocol (MGCP) is also a device control protocol; and the Skinny Client Control Protocol (SCCP) is a proprietary VoIP protocol that was developed by Selsius Corporation.

Each of these different protocols is used in concert with other protocols, each of them directing one aspect or facet of VoIP. Currently, no single protocol directs all VoIP activity. H.323 and SIP are the two most broad-reaching protocols, and in truth it would be more accurate to call them protocol stacks.

As mentioned earlier, H.323 stacks H.225, H.235, and H.245 (among others) together to address VoIP as a whole. If you were to envision the protocol stack, it would look like what's shown in Figure 4.1.

The same is true for SIP. Because it combines numerous other, more specific protocols, it's a stack and would look very much like Figure 4.2 if you envisioned it.

You might be wondering why the government doesn't provide regulations or legislation to govern the VoIP industry and at the very least decrease the growing pains. Actually, it has. Until recently, the government looked at VoIP as an information service and under the guise of encouraging the growth of information technology, decided to take a hands-off approach to VoIP.

As an alternative, the government in general and the Federal Communications Commission (FCC) in particular declared VoIP a no-regulation zone and decided to leave it to the states to govern VoIP, if they so desired. The theory was that if the FCC and the rest of the government stayed out of it, VoIP would grow, reducing the cost of telecommunications and increasing the competition in the industry. The theory proved accurate—VoIP has grown at a steady rate, and is predicted to continue to grow at an everincreasing pace largely because of its price appeal. But the FCC's hands-off policy has opened a whole new debate.

VoIP is the most disruptive technology to hit the telecommunications industry in more than a century. Now, established telecommunications companies are up in arms because the decision by the FCC to make VoIP an information service means that it's exempt from the regulatory guidelines and federal taxes that other telecommunications companies must pay. Those telecommunications companies are screaming mad. Not only are they paying these taxes, but they also must allow VoIP companies to access (for a fee) certain pieces of the public switched telephone network.

For example, if you use VoIP to make a call to your aunt in Poughkeepsie who still uses a rotary phone and wouldn't know VoIP if it smacked her on the forehead, your call is transported via the Internet to a switch where it's converted to analog and delivered to your technologically challenged aunt. From the switch to your aunt's old-fashioned phone, the lines that are used are part of the PSTN; in other words, they're copper wire, and one of the four major telecommunications companies in the U.S. owns those lines. But the VoIP service provider you use is given rights to access those lines to complete the delivery of calls, regardless of who owns them. Yes, the VoIP service provider has to pay a fee, but the fee is only a fraction of what it costs to maintain those lines and telecommunications companies aren't one bit happy about the situation.

The whole debate seems to have no logical answer. If the government were to include VoIP as a telecommunications company rather than an information services company, then VoIP companies could move offshore, freeing themselves of the regulatory constraints of being stateside while still maintaining their customer bases because VoIP has no walls and physical location is of no consequence.

The technology doesn't make the determination any easier. VoIP is a technology that rides on or is hidden inside another technology and that makes it hard to monitor, much less regulate. It's nearly impossible to determine where a call is going or where a user is, and this makes all VoIP traffic interstate by nature. When you add international call routing to the mix, it simply muddies the waters further.

All of this adds up to an atmosphere of animosity. Telecommunications companies are furious because it appears that VoIP companies are getting special treatment. And to some extent, they are. VoIP companies have it easy by comparison and it strikes chords of unfair competition. In fact, telecommunications companies are pushing for legislation that makes it clear that those telecommunications companies feel as if they're being unfairly penalized. In the end, it's likely that VoIP will be subject to all of the same legislation and regulations that other telecommunications companies are subject to, but when that's going to happen isn't clear. It's likely to be sooner rather than later, however.

Much like the early days of telephone service, VoIP service providers usually provide the equipment needed to use their service. The problem with this equipment is that it's often proprietary and cannot be used with another provider's VoIP service. So, for example, if you go to an electronics store and purchase an ATA, it will probably be labeled for a specific service provider. If, when you get the ATA home you decide you want to use a different service provider, you'll need to return the ATA and get a different one because the ATA probably won't work with the other service provider.

It would seem that all of the equipment should work together, regardless of the service provider that you choose. And some equipment does work with multiple vendors, but most vendors lock their equipment so that it can't be used with another company's service. Those that don't may not use the same protocols and standards that other service providers use, and the equipment is usually designed to work with specific protocols and standards.

That's one of the reasons that the standards and protocols that were discussed previously are so desperately needed within the VoIP industry. Currently, equipment might or might not work across a variety of service providers. But as standards and standard protocols are established and adopted, more and more ATAs will become standard and usable across all service providers. It's also likely that those same pieces of equipment will become multi-functional, giving you more than just VoIP telephone service.

So, if you're planning to use a VoIP service that is provided by a service provider and allows you to call anyone, anywhere, using your own analog telephone, you most definitely need an ATA and a router. The router will provide your Internet access (which should always be high-speed if you're planning to use VoIP). The ATA is a separate box that connects to the Internet router and handles the phone calls.

Number portability, or the ability to take your phone number with you from one service provider to another, was a move by the FCC to make the telecommunications industry more competitive. The legislation driving number portability applies to landline phone companies and wireless phone companies. It provides regulations and guidelines that require phone companies to allow users to take their phone number with them should they decide to change service providers either from landline to landline, landline to wireless, wireless to wireless, or wireless to landline.

The most important feature of number portability to understand is that the owner of record for a telephone number is the person who can port that number from one provider to another. So, for example, if you have a landline phone that you intend to disconnect and you want to take the phone to a wireless service provider, in order to port your existing number from one service provider to another, both companies must have the same name and address listed on the account. Your name and address establish you as the owner of the number.

The service provider that you are using before the switch or the service provider that you're switching to—whether they are landline service providers or wireless service providers—is legally entitled to charge you for porting your number. That charge can be in the form of a one-time payment or a monthly fee to cover the costs of porting your existing number.

The exception to the legislation is that number portability regulations and requirements don't apply to VoIP service providers. Those companies are not required to provide the service because technically they are not viewed as telephone service providers but instead are viewed as data service providers. VoIP service providers still fall under the category of information service providers, so they are not subject to the same regulations and legislation as communications companies.

However, some VoIP service providers do offer number portability. Those VoIP service providers that do may charge an additional fee for the service, just like wireless service providers and traditional phone service providers. But if your VoIP service provider doesn't offer number portability, there's no legal remedy because they aren't required by law to do so. If your number is that important to you, look for a service provider that offers this feature.

Session Initiation Protocol (SIP)

was developed by the Internet Engineering Task Force as a way of signaling multiuser distributed telephony and messaging applications on an IP network. SIP has garnered much praise from IT professionals, while suffering some criticism from traditional telecommunications people. The main reason for its less-than-perfect repute with telecom pros is its origin outside the telecom world. But many telecom guys have had to forgive this, because they're learning that SIP has almost no shortcomings when compared to its ITU-inspired cousin.

The essential duties and formulaic pieces of SIP are the same as H.323. That is, there are VoIP endpoints of varying capabilities, and there are servers that participate in the signaling process and establish policy for the voice network. Unlike H.323, however, SIP is far more extensible. It is more than just a set of voice and video telephony protocols. Rather, it's a packaging framework for all types of message-based applications, from intercom calling to instant messaging and AV services.

Companies like Broadvox, Voicepulse, Broadvoice, Packet8, and others have emerged as frontier providers for dial-tone-style services delivered over the Internet, using SIP as the signaling system. Under these service offerings, consumers can purchase telephone calling capabilities that use the Internet, rather than a POTS line, as the transport for their phone service.

Avaya, Cisco, Siemens, Alcatel, and the major telephony hardware vendors have indicated a strong support attitude about SIP, while some have even backed away from H.323 investments. This bodes well for SIP's future, and there are already more SIP IP phones installed worldwide than there are H.323 ones.

This makes SIP both an easy decision and a challenging one. SIP's extensibility comes by way of a non-telephony mindset.

Traditional telecom engineers have balked about the wordiness, or bulkiness, of SIP's message structure. Instead of using compact, machine-friendly message packets like H.323, SIP uses lengthy, human-readable headers like SMTP or HTTP. Proponents of SIP counter that this human readability makes SIP easier to troubleshoot, and I tend to agree.

SIP is currently in Version 2.0. Its definition is found in RFCs 3261 through 3265. The defined purpose of SIP is to coordinate and facilitate monitoring of media sessions on the network. It supports a variety of addressing schemes and can be designed as a centralized or distributed topology.

7.3.1. SIP Nodes

SIP endpoints and servers are called nodes. A SIP phone is a node. SIP phones can communicate directly with each other in order to establish media sessions, just as H.323 terminals can establish direct channels. But more often than not, especially in an enterprise setting, SIP is used with a SIP server. SIP phones normally report to a dedicated SIP server node called a registrar upon boot-up.

7.3.1.1 SIP registrar

The SIP registrar is a database server that communicates with SIP nodes in order to collect, store, and disperse information about the whereabouts of SIP users. When a SIP node registers with a registrar, it tells that registrar how to get hold of the user, specifically what IP address and port to use for future SIP communication. You could think of the registrar as a router, because its main purpose is to give advice on how to reach SIP users, just as a TCP/IP router's purpose is to give advice on how to reach other networks.

7.3.1.2 URIs

SIP endpoints can be referenced using Uniform Resource Indicators, but so can SIP users. Consider this URI:

```
sip:lerxt@sip.bytor.com
```

This convention indicates both the user to be contacted and the server that is expected to know the address of that user's SIP endpoint. In this case, the user is lerxt and the server is sip.bytor.com. Secure SIP URIs, that is, those that indicate an encrypted signaling connection, use the sips: prefix instead of sip:. Encryption of SIP signals, if desired, occurs by way of Transport Layer Security, defined in RFC 2246.

A SIP URI doesn't always correspond to a single phone. If a user is available at one of several phones, then all of those phones can ring simultaneously, or in a specific sequence, based on the handling server's configuration. Most SIP registrars support simultaneous registration of the same user at multiple phones, and the most common way of handling this situation is to ring them all when a call is received for that user.

7.3.1.3 SIP methods and responses
SIP signals fall into 10 categories called methods. Each method accomplishes a different function for SIP:

INVITE

This method is used to start sessions and advertise endpoint capabilities.

ACK

This method is used to acknowledge to the called SIP peer that an INVITE has succeeded.

BYE

This method occurs when the call is completed, that is, one user at a minimum wishes to end the call.

CANCEL

This method is used during attempts to override a prior request that hasn't yet been completed.

OPTIONS

This method is used to query a SIP peer for its capabilities information, without actually establishing a media channel.

REGISTER

This method notifies the SIP server at which endpoints a particular user can be reached.

INFO

This method is used to transmit telephony application signals through the SIP signaling path; these signals can include dialed digits.

PRACK

This method (Provisional ACK) is used to notify an endpoint of intent to set up a complex call without actually providing an ACK. PRACK is the SIP equivalent of "all is well."

SUBSCRIBE

This method provides a way of establishing event handlers within SIP telephony applicationsi.e., "Tell me when Bob misses a call" or "Tell Bob when I am registered with the server."

NOTIFY

This method delivers messages between endpoints as events occuri.e., "Bob missed a call."

When a call must be started, ended, or altered, a SIP method is employed. The SIP methods in the preceding list are similar in concept to the HTTP methods GET and POST, and like HTTP, SIP expects response codes when it sends a method. SIP's numeric response codes are three digits long and break down into six categories:

A complete list of SIP responses is found in Appendix A.

Project 11.2 shows how to use a packet capture tool to observe SIP methods and responses.

Typically, a SIP caller initiates a method directed to a SIP callee, and that SIP callee initiates a response, according to the success or failure of the caller's method.

In Figure 7-9, you can see that a (highly simplified) call from an Internet host (A), in the form of a SIP INVITE, to 5150@oreilly.com, would ordinarily result in a 200 OK response, clearing the way for the call to begin. Now, if the INVITE method specified a SIP peer whom the SIP server didn't know how to reach, a 404 Not Found response would be in order, as in Figure 7-10.

Figure 7-9. A call to 5150@oreilly.com would normally result in a 200 OK response, if 5150@oreilly.com were registered on the SIP registrar labeled B

A SIP INVITE header looks like this:

```
INVITE sip:5150@oreilly.com SIP/2.0
Via: SIP/2.0/UDP oreilly.com:5060;branch=9889gg1424
Max-forwards: 7
To: 5150 <5150@oreilly.com>
From: 1984 <1984@vh.com>
```

Figure 7-10. In the same setup as Figure 7-9, a call to 1138@oreilly.com, which is not registered in the registrar labeled B, will result in a 404 response

SIP proxy

Calls from one SIP endpoint to another can be considered local if they are both registered with the same SIP registrar, even if the physical endpoints are on different continents. The point is, they are on the same domain and are therefore peers on a local network of sorts. Nonlocal calls, however, are routed through specialized SIP server software known as a SIP proxy.

A SIP proxy is a server that routes or redirects SIP INVITE methods on behalf of one or more domains, just as a web server provides responses to HTTP methods for certain domains. So, when an incoming call from a foreign network is recognized, the SIP proxy's job is to connect it to the called user's endpoint, if possible.

Outbound SIP proxies serve the task of connecting calls, but on behalf of a local network of SIP users. Many users may share the same SIP proxy because they work in the same office or perhaps because they subscribe to the same SIP dial-tone service provider. Outbound SIP proxies are often used to overcome network communications problems posed by NAT firewalls.

Some local calls may benefit from being routed through a SIP proxy, too. Forcing even local SIP endpoints to use a SIP proxy allows for easy enforcement of a dial-plan, greater administrative control over the voice network, and the ability to do centralized telephony applications such as call recording.

7.3.1.5 SIP user agent elements
All SIP proxies and endpoints are comprised of two key software elements, the user agent client and user agent server (UAC and UAS). All SIP devicesbe they softphones, hardphones, voice mail servers, or full-blown PBX serversmust be able to speak the SIP protocol, and the UAC and UAS elements are their mouthpieces. The UAC sends methods and receives responses, so its logical equivalent in HTTP is the web browser. The UAS receives SIP methods, processes them, and returns responses, so it's more like a web server. In varying degrees of completeness, all SIP endpoints and servers have both a UAC and a UAS.

In Figure 7-11, you can see the signaling process for a nonlocal call. This particular example uses an inbound call, which is fielded by a proxy server. By the time the SIP signaling begins, the calling endpoint already knows what host to contact for calls destined for the receiving endpoint's domain. This occurs by way of a DNS lookup for the hostname in the form sip.domain.com. The calling endpoint can then contact the proxy server for the domain in question, and send it an INVITE method.

Figure 7-11. The SIP signaling process for a call from endpoint A to endpoint C through proxy B

The proxy server may immediately respond with an informational Trying response, or it may save the Trying response until after it has forwarded the proxied INVITE method to the appropriate endpoint on the local network. Incidentally, the way the proxy knows which local endpoint to contact is by performing a database lookup with the SIP registrar. If the user being called doesn't exist in the registrar, then the proxy will return a 404 Not Found response.

In this example, though, the user, jake@oreilly.com, does exist and is able to respond to the proxied INVITE method. His endpoint's responses are proxied back to the caller, and ultimately, the 200 OK response is sent, indicating the call is clear to proceed. One of the most important pieces of this startup signaling process occurs during the INVITE methods and 200 responses: SDP capabilities negotiation.

SIP redirect

When a SIP server responds to a calling endpoint's INVITE method with a 3xx response, that SIP server is redirecting the calling endpoint to a different SIP server. The calling endpoint should then contact that server with an INVITE method for further assistance in connecting the media stream. This feature is not implemented on all systems that support SIP. In fact, where complex, signaling-neutral dial-plan programming is available (like in Asterisk), SIP redirection isn't always necessary. The use of SIP redirects is more common in large, SIP-only networks with multiple servers, such as those that span the Internet.

Session Description Protocol

SDP is the de facto session capabilities protocol of SIP, similar to the H.245 protocol in H.323. It is defined in RFC 2327. When a call is placed from one SIP endpoint to another, an SDP capabilities construct is sent as text payload in a SIP INVITE, following the SIP packet header Content-type that indicates an SDP message is to follow. Here's a sample SDP payload:

```
v=0
o=HanSolo 7575 440 IN IP4 10.1.1.103
c=IN IP4 10.1.1.103:16385
m=audio 16385 RTP/AVP 1
a=rtpmap:1 G726/8000
```

This particular construct is requesting a G.726 media channel originating from user HanSolo at 10.1.1.203 using RTP for media packaging on UDP port 16385.

Using SDP, the calling endpoint can request certain codecs, sampling rates, or even a packaging protocol other than RTP (this is very rare, though). In the previous SDP block, the v token indicates the version of SDP being used, though SIP doesn't care, as SIP could theoretically use any version. The o token is a string of identifiers that uniquely name this SDP request, often including an NTP timestamp and the IP address and protocol designation of the sender. The c token tells which IP address to use for the media channel.

The m token describes the UDP port number and media framing protocol (RTP in this example), followed by a numeric identifier for this framing capability, called an RTP profile. More than one m token definition may be sent if the requesting SIP endpoint is advertising more than one set of RTP capabilities. The a token tells the RTP profile identifier, codec, and sample rate to use for the media channel. For an exploration of SDP messages captured by the Ethereal packet sniffer, see Chapter 11.

In Asterisk, the status of SIP peers can be displayed at the Asterisk command line, using sip show peers and sip show users. Calls in progress can be tracked using Asterisk Manager (astman).

7.3.1.8 Real-Time Streaming Protocol

Unlike H.323, the SIP protocol family provides a built-in recommendation for streaming prerecorded audio and video, in this case the RTSP protocol. This is the same protocol used by RealOne and similar media player applications. It's defined by RFC 2326.

SIP packet encoding

While H.323 uses the obscure ASN.1 (abstract syntax notation) encoding, SIP uses plain text. More specifically, SIP uses a text-based conversational approach for its signaling messages, whereas H.323 uses the Q.931 ISDN approachone that is, by all accounts, meagerly understood outside telecommunications engineering circles.

Although SIP has many functional equivalents in H.323, it's fair to say that SIP's approach is more distributed, and certainly more Internet-like, while H.323's is more PSTN-like or mainframe-like. Table 7-2 illustrates this concept.

Table 7-2. Comparison of SIP, H.323, and IAX protocol families Function/Characteristic
SIP
H.323
IAX

Endpoint discovery and admission
SIP REGISTER methods
RAS Protocol
IAX REG Control Frames

Call setup and teardown
SIP INVITE methods
H.225 Protocol
IAX NEW and HANGUP Control Frames

Capabilities negotiation, codec selection, and media session port selection
Session Definition Protocol
H.245 Protocol
IAX Capability Information Meta Frame

Packetization and sound sample transmission
RTP/RTCP Protocol
RTP/RTCP Protocol
IAX Voice/Data Full and Mini Frame

Streaming of recorded audio and video
RTSP
None recommended
None recommended

Frame encoding
Text (ASCII & similar)
ASN.1
Binary/proprietary

Messaging approach
HTTP-like
ISDN-like / Q.931
Proprietary

SoftPBX call path is called a…
Proxy
Gatekeeper-routed
SoftPBX

Call-routing reference device is called a…
Registrar
Gatekeeper
Server

Independent call path is called…
Redirect
Directed signaling
Direct signaling

PSTN interface approach
None recommended
H.323 gateway
None recommended

Encryption of signaling messages
TLS/TCP
None recommended
A work in progress

Endpoint identification
SIP URI, email address, E.164 address, or alias

E.164 address
Email address, E.164 address, or alias

Connections through firewalls
Gatekeeper/softPBX call path
Proxy/softPBX call path
No proxy needed

Multiplexed trunks
Yes
Yes
Yes

UDP port number
5060/5061
1503,1720,1731
5036 (v1)

SIP versus H.323: the great debate
H.323 is older than SIP, and more considerate of legacy infrastructure, too. It defines standard procedures and best practices for interfacing with old-school telephony technologypractices like the use of gateways with built-in support of legacy protocols like ISDN and FXO/FXS.

SIP makes no such provisions, but is still the favorite among those schooled in Internet thought: it's extensible and reusable and does a whole lot more for telephony apps, ultimately, than H.323. SIP allows multiple endpoints to register the same alias, allows freedom from E.164, and enables other, non-voice applications like instant messaging and presence. For these reasons, some in the industry have pronounced SIP the winner of the battle for telephony signaling.

SIP doesn't define gateways for interaction with the PSTN, so a legacy-aware signaling system like MGCP, MEGACO, or H.323 is often used alongside SIP in order to facilitate legacy gateways.

The fact is, both protocol suites are very much necessary. Parts of the H.323 recommendationRTP and PSTN gateways in particularare in use by SIP networks so the importance of H.323's features is obvious.

One day, SIP may contain some legacy interfacing of its own, but for now, H.323 fills that role quite aptly. All the big VoIP equipment vendors use its recommendations in order to interface with legacy systems like the PSTN. SIP, conversely, was built for the new networkthe Internet network. Some mix of H.323 or MEGACO/H.248 (covered later) is in order if your SIP system plans on talking to a traditional telephony system.

Even companies that develop and use proprietary VoIP protocols use more than one. What's more, those companies may use one proprietary protocol and several other common protocols to achieve the best-quality VoIP service. There's still no single "best practice," but the intent to find one is there. The race to be the discoverer is what makes these different protocols interesting. It's also what's pushing the VoIP industry to continually improve the quality of VoIP services and equipment.

As the telephone became more important in the business world, innovators extended its capabilities and made it more convenient. They did so using enterprise telephony devicesgateways that connect privately owned phones together into a private voice network with self-managed calling features. When the enterprise gained ownership over its own voice network, it set about building telephony applications specific to its business.

One such device is a KTS, or key telephone system. In many small businesses, telephones can share a group of telephone company POTS lines through the use of a KTS. Each phone in a key system has direct access to one or more of the telephone company's lines, just as a simple residential phone has access to a single line. Unlike a single-line phone setup, KTSs allow a group of phones to use more than one telephone line at a time. This allows a single operator to place a call on hold while answering a call on another line, among other things, without using any phone company calling features. Generally, KTSs are not referred to as switches, because they rely upon the circuit-switching abilities of the central office in order to connect calls.

In many larger offices, telephones connect to a private, on-premises switch that interfaces with the telephone company's lines. This switch is called a PBX, or private branch exchange. PBXs are smaller, enterprise-friendly versions of the heavy-duty switches used by the telephone company, and they allow businesses to run their own telephony applications in-house. Unlike with a key system, PBX phones in the office can call other phones in the office without tying up an external telephone line. So several simultaneous conversations between parties in the same office can occur without making use of the PSTN at all. One job of the PBX is to determine how to "route" callsthat is, how to ascertain whether the calling party is trying to reach another person within the same office or trying to reach somebody via the PSTN. Most PBX vendors refer to the call-routing scheme as the dial-plan.

Single-line phones, key systems, and PBX systems all connect to the PSTNbut for various reasons. Single-line phones and key systems connect calls to the PSTN even if they are from one phone to another in the same office, but a PBX connects calls to the PSTN only if they are bound for an outside organization.

Despite their various capabilities, POTS, KTSs, and PBXs are all based on the same circuit-switching, electromagnetic-signaling technologies. Even POTS' higher-capacity digital cousins, ISDN and T1, which are able to squeeze many simultaneous phone calls onto a single copper loop, are members of the same technology family. All the standards that govern these traditional telephony systems stem from the International Telecommunications Union (ITU), and many have been unchanged for decades because they are incredibly reliable. They'd have to be reliable in order to run the global telephone system, wouldn't they?

Let us assume in our previous example that, once the call is established, John decides to put the audio stream on hold. In our example, media is fl owing in both directions, which is the default value if no specifi c direction attribute is present in the SDP (sendrecv, sendonly, recvonly, or inactive). Therefore, in order to put the call on hold, John just needs to send a reINVITE to Alice that includes an SDP with the attribute sendonly for the audio stream. If, later on, he wants to retrieve the call, he needs to just send a new reINVITE and change the SDP attribute to sendrecv or simply not add any attribute.

Your VoIP phone bill is probably lower than that of your friends who still use traditional calling plans. But a lower phone bill isn't the only luxury that comes with converting your service to VoIP. Because your call uses the Internet rather than the public telephone network to route your call, you have access to several cool dialing shortcuts when you call subscribers of other VoIP services. When an IP network alone provides the pathway between caller and receiver, it's said to be pure (or native) Voice over IP.

This can actually save you money, especially if you make a lot of international calls. If you're a Free World Dialup (FWD) subscriber and you talk frequently with your buddy in Mexico, who uses Vonage, using dialing shortcuts will keep your calls pure VoIP and allow you to circumvent any related long-distance calling charges that would be assessed if your calls were to traverse the Public Switched Telephone Network (PSTN).

To make pure VoIP calls using your TSP's service, you have to be aware of the dialing shortcuts your TSP provides to route calls to other TSP networks using the Internetinstead of the PSTNas the carrier network. Most VoIP TSPs will assume your call is destined for the PSTNjust because it's an 11-digit phone number. So these shortcuts tell the TSP that you don't want to route your call to the PSTN. Instead, you want to route it over the Internet to another VoIP TSP.

Why do this? If you have an unlimited calling plan, it won't really save you any money. The call probably won't sound any better either. But this technique does conserve your TSP's public telephone network capacity when you use pure VoIP rather than VoIP-to-PSTN calling. If your VoIP TSP bills you by the minute, it might not charge for calls that don't use its PSTN capacity. Plus, it's just cool to let the Internet replace the Bell System for your phone calls. Here's how.

VoIP services such as FWD, Vonage, IAXTel, VoicePulse, and Packet8 offer dialing shortcuts to allow calls between their customers. If you're a Packet8 subscriber, you can reach any FWD subscriber by dialing 0451 and the six-digit FWD number assigned to that subscriber (FWD subscribers don't have traditional 11-digit phone numbers because the service doesn't provide PSTN calling).Consult Table 1-2 for a rundown of the VoIP dialing shortcuts that you can use to route calls between the various VoIP services.

Your business will need to have an internal Ethernet network because of the IP network utilization for VoIP. Once you are using computers in your business, you certainly have it, so you do not have to worry. However, in case you realize that you do not have one, you can purchase an affordable one then have it easily set up.

We may consider a forecast that in the next two decades, each home will be utilizing a VoIP phone system. This will represent a considerable revolutionized communication technology. However, the experts must always consider the density of calls per second or the possible problems encumbered by traffic. Anyway, with all the innovations brought about each year, we must expect a highly transformed communication technology ahead.

SIP is the key signaling protocol that is used in real-time IP communication sessions such as voice, video, and IM communications. For example, in VoIP, SIP is used as the call establishment protocol and converts the various PSTN phone mechanisms (off-hook, digit dialing, hold, etc.) into packetized signaling messages to establish voice calls across a network. SIP has established itself as the primary signaling protocol for VoIP and has left the competing H.323 signaling protocol in a distant second place. Figure 3.1 shows SIP relative to the TCP/IP stack as well as the other related protocols such as SDP, RTP, and DNS. Each of these protocols, together with SIP, is required to conduct real-time,multi-media calls over an IP network (private network or the public internet).

Voice over Internet Protocol (VoIP) is the latest technology allowing you to deliver and receive voice calls using an Internet connection, particularly broadband, as a substitute to an analog or an ordinary phone line. Also, it is referred to as the Broadband telephony, Voice over Broadband, IP Telephony Broadband Phone, and Internet telephony.

Your voice is translated into a digital signal that travels over the Internet through the VoIP services. Once calling a regular telephone number, the signal is coverts to a regular telephone signal just before it arrives at the destination. VoIP can let you make a call either directly from a special VoIP phone, a computer, or a traditional phone linked to one special adapter. Wireless "hot spots" such as airports and parks allow you to connect to the Internet enabling you to use VoIP service wirelessly.