A

Seminar report

On

Voice over Internet Protocol (VoIP)

Submitted in partial fulfillment of the requirement for the award of degree of Bachelor of Technology in Computer Science

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Acknowledgement

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Preface

I have made this report file on the topic **Voice over Internet Protocol**; I have tried my best to elucidate all the relevant detail to the topic to be included in the report. While in the beginning I have tried to give a general view about this topic.

My efforts and wholehearted co-corporation of each and everyone has ended on a successful note. I express my sincere gratitude to …………..who assisting me throughout the preparation of this topic. I thank him for providing me the reinforcement, confidence and most importantly the track for the topic whenever I needed it.

Introduction

Using an ordinary phone for most people is a common daily occurrence as is listening to your favorite CD containing the digitally recorded music. It is only a small extension to these technologies in having your voice transmitted in data packets. The transmission of voice in the phone network was done originally using an analog signal but this has been replaced in much of the world by digital networks. Although many of our phones are still analog, the network that carries that voice has become digital.

In todays phone networks, the analog voice going into our analog phones is digitized as it enters the phone network. This digitization process, shown in Figure 1 below, records a sample of the loudness (voltage) of the signal at fixed intervals of time. These digital voice samples travel through the network one byte at a time.

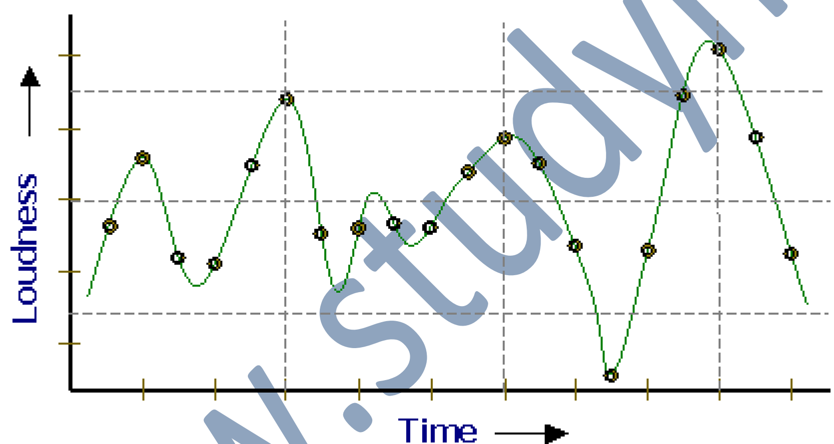


Figure 1. Digital Sampling of an analog voice signal

At the destination phone line, the byte is put into a device that takes the voltage number and produces that voltage for the destination phone. Since the output signal is the same as the input signal, we can understand what was originally spoken. The evolution of that technology is to take numbers that represent the voltage and group them together in a data packet similar to the way computers send and receive information to the Internet. Voice over IP is the technology of taking units of sampled speech data .

So at its most basic level, the concept of VoIP is straightforward. The complexity of VoIP comes in the many ways to represent the data, setting up the connection between the initiator of the call and the receiver of the call, and the types of networks that carry the call.

Using data packets to carry voice is not just done using IP packets. Although it won't be discussed, there is also voice over Frame Relay (VoFR) and Voice over ATM (VoATM) technologies. Many of the issues VoIP being discussed also apply to the other packetized voice technologies.

The increasing multimedia contents in Internet have reduced drastically the objections to putting voice on data networks. Basically, the Internet objections to putting voice on data networks. Basically, the Internet Telephony is to transmit multimedia information in discrete packets like voice or video over Internet or any other IP-based Local Area Network (LAN) or Wide Area Network (WAN).

The commercial Voice Over IP (Internet Protocol) was introduced in early 1995 when VocalTec introduced its Internet telephone software. Because the technologies and the market have gradually reached their maturity, many industry leading companies have developed their products for Voice Over IP applications since 1995.

History of VOIP

Voice-over-Internet Protocol has been a subject of interest almost since the first computer network. By 1973, voice was being transmitted over the early Internet.The technology for transmitting voice conversations over the Internet has been available to end-users since at least the early 1980s.

In 1996, a shrink-wrapped software product called VocalTec Internet Phone (release 4) provided VoIP along with extra features such as voice mail and caller ID. However, it did not offer a gateway to the PSTN, so it was only possible to speak to other Vocaltec Internet Phone users.

In 1997, Level 3 began development of its first softswitch (a term they invented in 1998); softswitches were designed to replace traditional hardware telephone switches by serving as gateways between telephone networks.

Revenue in the total VoIP industry in the US is set to grow by 24.3% in 2008 to $3.19 billion. Subscriber growth will drive revenue in the VoIP sector, with numbers expected to rise by 21.2% in 2008 to 16.6 million. The United States' largest VoIP provider is Vonage.

Voice over Internet Protocol

VoIP, or “Voice over Internet Protocol” refers to sending voice and fax phone calls over data networks, particularly the Internet. This technology offers cost savings by making more efficient use of the existing network.

Traditionally, voice and data were carried over separate networks optimized to suit the differing characteristics of voice and data traffic. With advances in technology, it is now possible to carry voice and data over the same networks whilst still catering for the different characteristics required by voice and data.

Voice-over-Internet-Protocol (VOIP) is an emerging technology that allows telephone calls or faxes to be transported over an IP data network. The IP network could be

- A local area network in an office
- A wide area network linking the sites of a large international organization
- A corporate intranet
- The internet
- Any combination of the above

There can be no doubt that IP is here to stay. The explosive growth of the Internet, making IP the predominate networking protocol globally, presents a huge opportunity to dispense with separate voice and data networks and use IP technology for voice traffic as well as data. As voice and data network technologies merge, massive infrastructure cost savings can be made as the need to provide separate networks for voice and data can be eliminated.

Most traditional phone networks use the Public Switched Telephone Network(PSTN), this system employs circuit-switched technology that requires a dedicated voice channel to be assigned to each particular conversation. Messages are sent in analog format over this network.

Today, phone networks are on a migration path to VoIP. A VoIP system employs a packet-switched network, where the voice signal is digitized, compressed and packetized. This compressed digital message no longer requires a voice channel. Instead, a message can be sent across the same data lines that are used for the Intranet or Internet and a dedicated channels is no longer needed. The message can now share bandwidth with other messages in the network.

Normal data traffic is carried between PC's, servers, printers, and other networked devices through a company's worldwide TCP/IP network. Each device on the network has an IP address, which is attached to every packet for routing. Voice-over-IP packets are no different.

Users may use appliances such as Symbol's NetVision phone to talk to other IP phones or desktop PC-based phones located at company sites worldwide, provided that a voice-enabled network is installed at the site. Installation simply involves assigning an IP address to each wireless handset.

VOIP lets you make toll-free long distance voice and fax calls over existing IP data networks instead of the public switched telephone network (PSTN). Today business that implement their own VOIP solution can dramatically cut long distance costs between two or more locations.

PSTN Versus VoIP: A Feature Comparison

VOIP	PSTN
All channels carried over one Internet connection	Dedicated Lines
Compression can result in 10kbps (in each direction)	Each line is 64kbps (in each direction)
Features such as call waiting, Caller ID and so on are usually included free with service	Features such as call waiting, Caller ID and so on are usually available at an extra cost
Upgrades usually requires only bandwidth and software upgrades	Can be upgraded or expanded with new equipment and line provisioning
Long distance is often included in regular monthly price	Long distance is usually per minute or bundled minute subscription
Lose power, lose phone service without power backup in place	Hardwired landline phones (those without an adapter) usually remain active during power outage
911 emergency calls cannot always be traced to a specific geographic location	When placing a 911 call it can be traced to your location

Working

VoIP stands for Voice over IP (Internet Protocol), a variety of methods for establishing two-way multi-media communications over the Internet or other IP-based packet switched networks. Although VoIP systems are capable of some unique functions (for example: video conferencing, instant messaging, and multicasting), this appendix concentrates on the ways in which VoIP can be used to replicate the voice conversation functionality of the public switched telephone network (PSTN).

There are several competing approaches to implementing VoIP. Each makes use of a variety of protocols to handle signaling, data transfer, and other tasks. To help describe the similarities and differences between these approaches, consider the following simplified description of a telephone call under VoIP:

-Caller picks up the phone (his terminal), hears a dial tone and dials a destination number.

-Destination number is mapped to a destination IP address.

-Call setup routines are invoked, handled by signaling protocols. Depending on the VoIP standard in use, this may involve a device (or function) known as a Gateway, and may also involve a Gatekeeper.

-Destination phone generates a ring, the called party picks up the phone, and a two-way conversation is established.

-Data is moved between the two endpoints using a media protocol, the Real-time Transport Protocol (RTP). A codec (coder/decoder) is used to convert the sound of each caller's voice to digital data, then back to analog audio signals at the other end.

-Conversation ends and the call is torn down. Again, this involves the signaling protocols appropriate to the particular implementation of VoIP, along with any Gateway or Gatekeeper functions.

The instructions governing the call-the call setup and call teardown-are handled separately from the transmission of the actual data content of the call, or the encoding and packetization of voice media.

There are several protocols and methods for VoIP calls – the commonest standards are termed SIP and H.323 – but they all have some basic features in common. To the user phone calls are made and handled in the same way as they always have been except that VoIP phones often have more features available from menus and buttons than regular phones.

When a call is dialed, the system takes the phone number, connects over the local network to whatever system is providing service. That system figures out if the call needs to go into the regular phone network and if so switches it to a gateway that connects the call over the regular phone network. If the call can be completed without going over the regular phone network (the number dialed is also a VoIP system) then the provider system will route the call directly, performing protocol translation (to a different kind of VoIP) if needed.

When traveling on the network, VoIP calls are treated like any other network data – they are broken down into little pieces of digital information (packets) and sent by whatever route the network determines to be fastest. That means different pieces arrive and different times and out of order and then are reassembled back into the proper sequence at the destination.

This is why the 100+ kbps transmission rate is needed – so that the signal can be sent and reassembled quickly enough so that human users on both ends don't notice any delay. It is also one of the weaknesses of VoIP – if the network goes down or has performance issues, so will your VoIP calls.

Since the very early days of distance communication, signals were sent in analog form, in waves. Many years ago, the communication world discovered that sending a signal to a remote destination could have been done also in a digital fashion: before sending it we have to digitalize it with an ADC (analog to digital converter), transmit it, and at the end transform it again in analog format with DAC (digital to analog converter) to use it. VoIP works like that: digitalizing voice in data packets, sending them and reconverting them in voice at destination.

Digital format can be better controlled: we can compress it, route it, convert it to a new and better format, and so on. We also saw that a digital signal is more noise-tolerant than its analog version.

TCP/IP networks are made of IP packets containing a header (to control communication) and a payload to transport data: VoIP uses it to go across the network and come to destination.

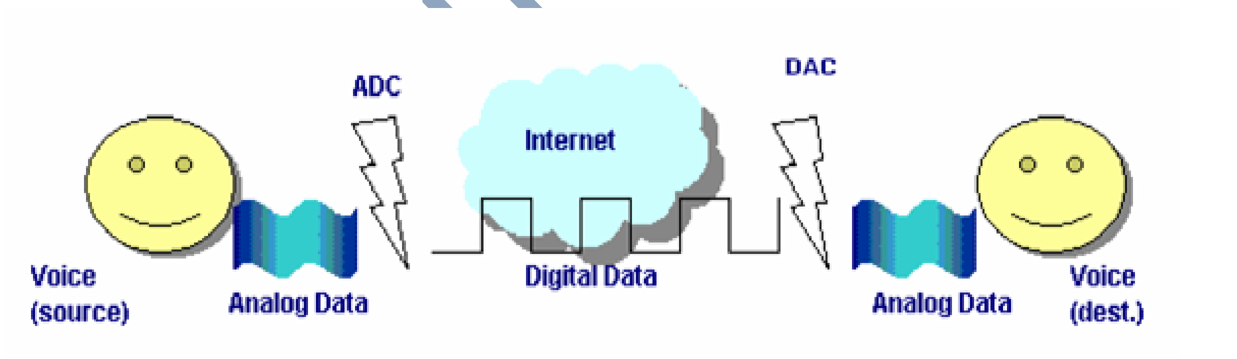


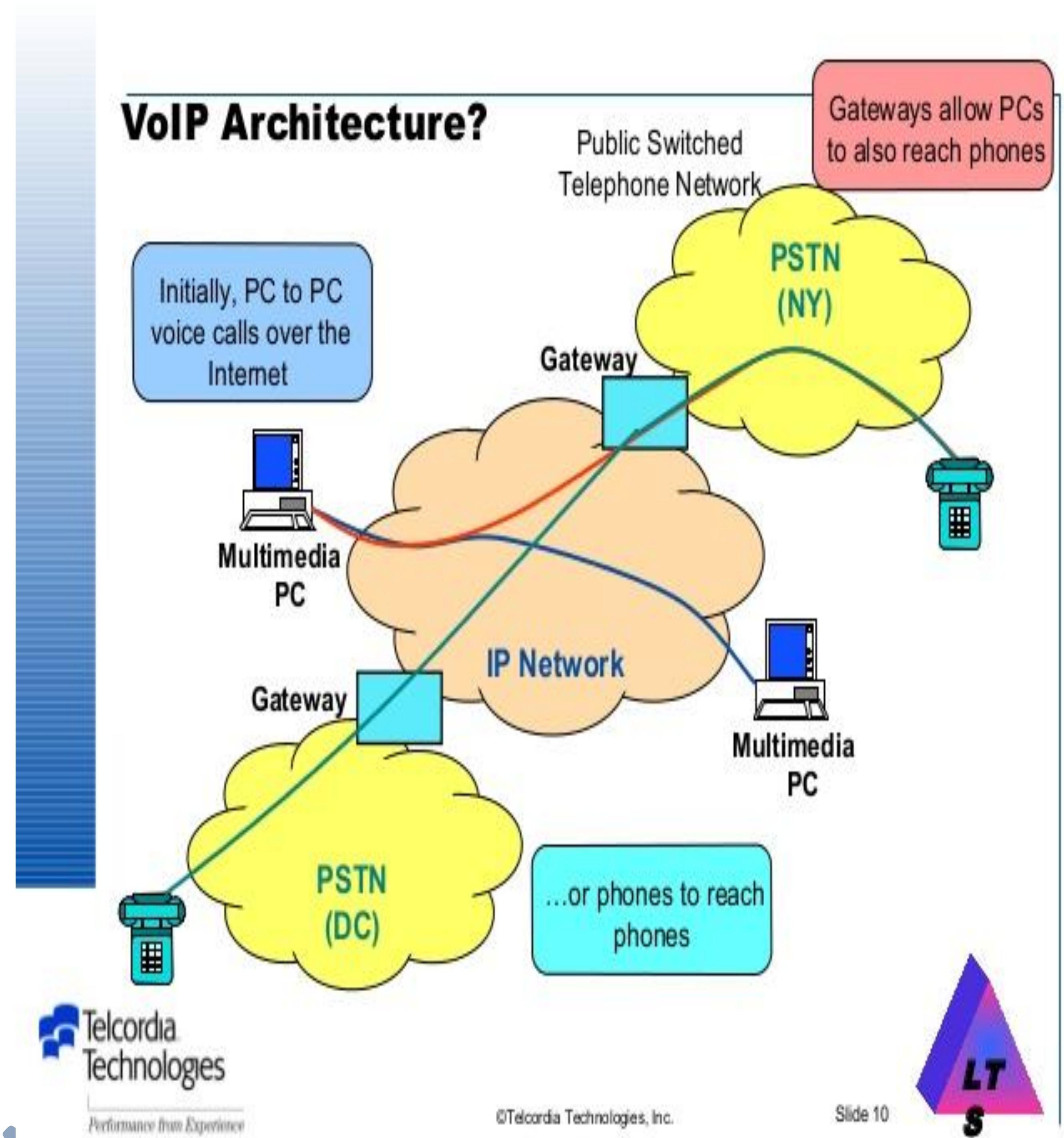
Fig . Basic working of VOIP

VoIP is becoming a key driver in the evolution of voice communications. VoIP technology is useful not only for phones but also as a broad application platform enabling voice interactions on devices such as PCs, mobile handheld, and many vertical-specific application devices where voice communication is an important feature.

VoIP supports two-way transmission of voice traffic over a packet-switched IP (Internet protocol) network. The first widely used VoIP application appeared in the mid-1990s, with services that enabled Internet users to make free voice calls between specially equipped PCs, or between a regular phone and a specially equipped PC. This was a great way to save toll charges on long-distance and international calls. Today, with rapidly advancing technologies, voice quality on managed VoIP networks can match the public voice network.

The primary reason for VoIP was to provide access to voice communication to anyone in any part of the world with minimal or no cost through the Internet backbone. The future of Internet phone would allow an individual to have a personal number which would enable him to communicate from any part of the world without having to pay exorbitant prices.

In addition to IP, VoIP uses the real-time protocol (RTP) to help ensure that packets get delivered in a timely way. Using public networks, it is currently difficult to guarantee Quality of Service (Qos). Better service is possible with private networks managed by an enterprise or by an Internet telephony service provider (ITSP).


VoIP Architecture?
Public Switched
Telephone Network
Gateways allow PCs
to also reach phones
Initially, PC to PC
voice calls over the
Internet
PSTN
(NY)
Gateway
Multimedia
PC
IP Network
Gateway
Multimedia
PC
PSTN
(DC)
...or phones to reach
phones
Telcordia
Technologies
Performance from Experience
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Applications and Uses of Voice over Internet Protocol

THREE WAYS OF MAKING A VOIP CONNECTION

Voice over Internet Protocol, or VoIP, uses your broadband internet connection to place phone calls. By converting your voice (or analog) signal into a digital signal, this makes for a more efficient way to talk on the phone and can save you money.

There are three ways in which you can make a VoIP connection, each way having a different set of requirements and implications. You can either connect using your regular phone and an adapter, a special internet phone, or download software and use your computer. The three ways are differentiated by what you have on each of the two communicating sides.Here are the methods, in greater detail:

1.Computer to Computer:

This mode is the most common, as it is so easy and free. You need to have a computer connected to the Internet, with the necessary hardware to speak and listen (either a headset or speakers and a microphone). You can install voice communication software like Skype ,Express talk and you are ready to talk.

Like any other product voip, PC to PC calling (VoIP telephony) has its advantages and disadvantages. Perhaps the biggest drawback of voice over Internet is the fact that the person you call must be online and must have the same vocation of Internet software as you do! It is a known fact that quality costs. The same principle applies here. Depending on the speed of Internet connectivity, signal quality audio (the sound quality of the call), May vary. So if you or people you call have / has a slow connection or network is busy, May it is almost impossible to have a conversation with VoIP telephony.It's like chatting, but with voice.

This can happen not only on the Internet, but on a Local Area Network (LAN) as well. The network should be IP-enabled, i.e. the Internet Protocol (IP). Should be running and controlling packet transfer on your network. This way, you can communicate with another person on the same network. Whether you are communicating over the Internet or a LAN, you need to have adequate bandwidth. If you have around 50 kbps, it will work, but you won't have great quality. For good quality voice, get at least 100 kbps for a conversation.

2.Phone to Phone:

This mode is very handy, but is not as simple and cheap to set up as the other two. It implies using a phone set on each end to communicate. Thus you can use VoIP and take advantages of its low cost by using a phone set and speak to another person using a phone set as well. There are two ways in which you can use phones to make VoIP calls:

Using IP Phones

An IP Phone looks just like a normal phone. The difference is that instead of working on the normal PSTN network, it is connected to a gateway or router, a device which, simply said, does the necessary mechanisms to get the VoIP communication running. The IP phone therefore does not connect to the RJ-11 socket. Instead, it uses the RJ-45 plug, which is the one we use for wired LANs. If you want to have an idea of what a RJ-11 plug is, have a look at your normal phone or your dial-up modem. It is the plug that connects the wire to the phone or modem. The RJ-45 plug is similar, but bigger.

Fig. Elements of IP Phone

You can of course use wireless technologies like Wi-Fi to connect to a network. In this case, you can either be using a USB or RJ-45 for connection.

Hardware of a stand alone IP phone : Speaker/ear phone and microphone.

- Key pad / touch pad to enter phone number and text (not used for ATAs).
- General purpose processor (GPP) to process application messages.
- Ethernet or wireless network hardware to send and receive messages on data network.

Power source might be a battery or DC source. Some IP phones receive electricity from Power over ethernet.

Common features of IP phones:

- Caller ID.
- Dialing using name/ID: This is different from dialing from your mobile call register as the user does not need to save a number to a sip phone.
- Locally stored and network-based directories
- Conference and multiparty call
- Call park
- Call transfer and call hold

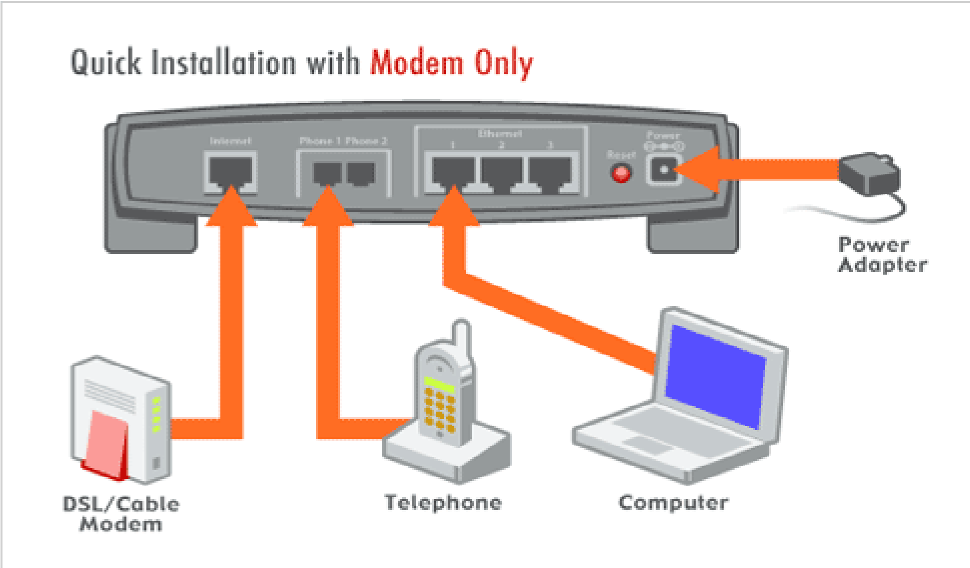
Disadvantages of IP phones:

- Requires internet access to make calls outside the Local Area Network unless a compatible local PBX is available to handle calls to and from outside lines.
- IP Phones and the routers they connect through usually depend on mains electricity unlike PSTN phones which are supplied with power from the telephone Exchange.
- IP networks, particularly residential internet connections are easily congested. This can cause poorer voice quality or the call to be dropped completely.
- IP Phones, like other network devices can be subjected to Denial of service attack as well as other attacks especially if the device is given a public IP address
- Due to the latency induced by protocol overhead they do not work as well on satellite Internet and other high-latency internet connections

Using an ATA:

ATA is short for Analog Telephone Adapter. It is a device that allows you to connect a standard PSTN phone to your computer or directly to the Internet. The ATA converts voice from your normal phone and converts it to digital data ready to be sent over a network or the Internet. ALLO is one of the company manufacturing the ATA box.

Installation of ATA box is shown below .It consist of one port for Power three ports for computer and two port for telephone lines.



If you register for VoIP service, it is common to have an ATA bundled along in the service package, which you can return once you terminate the package. For example, you get an ATA in a package with Vonage and AT&T’s CallVantage. You only have to plug the ATA to your computer or and phone line, install the necessary software, and you are ready to use your phone for VoIP.

VOIP Telephone Service Using ATA

3. Phone to Computer and vice-versa

Now that you understand how you can use your computer, normal phones and IP phones to make VoIP calls, it is easy to figure out that you can call a person using a PSTN phone from your computer. You can also use your PSTN phone to call someone on his computer.

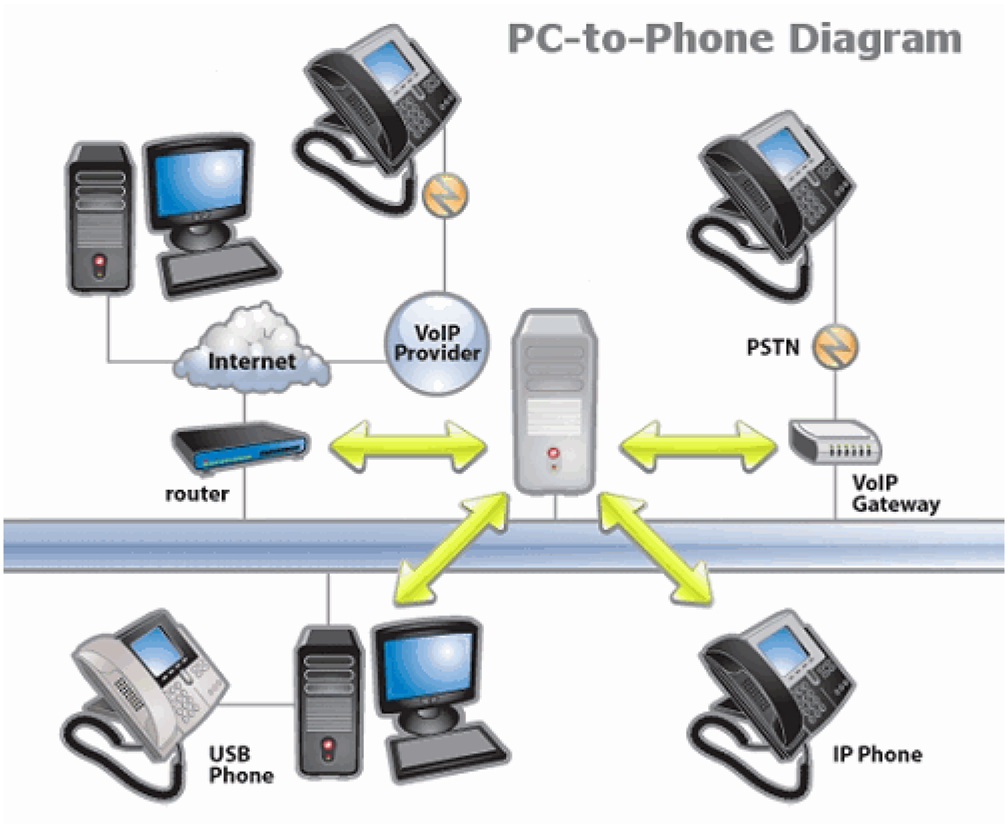


Fig. Phone to Computer Calling in VOIP

You can also have a mixture of VoIP users, using phones and computers to communicate over the same network. The hardware and software are heavier in this case

Advantages VOIP

- Most good quality VOIP software is either cheap or free.
- Free or cheap local/international call rates compared to traditional phone calls.
- VOIP is integrated with features such as chat, whiteboard, audio and video-conferencing.
- Can be used with VOIP adapters, allowing your normal home phone to be turned into a VOIP phone.
- VOIP phone adapters can be carried around with you wherever you travel.
- Computers do not have to be turned on, you can receive VOIP calls on your existing phone.

Disadvantages of VOIP

- Quality of calls across Internet is not assured
- Broadband equivalent connection needed for connecting offsite
- Network switches may need replacement
- Power on Ethernet may need to be established over the LAN
- Phone availability is dependant on network hardware and power
- Some VOIP providers have fees
- Emergency calls 000 do not issue an origin

VOIP Alternatives

- Call cap plans with Telecommunications provider
- Virtual fax lines that get sent to email
- ISDN D channel for EFTPOS
- Advanced forms of Instant messaging
- Use mobile phones and reduce land lines
- Least cost routing with various carrier prefixes

Future Trend

Price is the key driver of the VoIP market today. End-user features such as multimedia conferencing, multicast, call centers, IP call waiting, and message unification are the benefits that will drive the VoIP market well into the future.

The growing competition between ISPs is causing declining margins. ISPs are seeking value-added services to increase revenues per subscriber.

Becoming an ITSP is the solution.The demand for convergent networks is evolving into a requirement for new network/telephone orders and upgrades.

Conclusion

The current Public Switched Telephone Network is a robust and fairly bulletproof system for delivering phone calls. Phones just work, and we've all come to depend on that. On the other hand, computers, e-mail and other related devices are still kind of flaky. Let's face it -- few people really panic when their e-mail goes down for 30 minutes.

It's expected from time to time. On the other hand, a half hour of no dial tone can easily send people into a panic. So what the PSTN may lack in efficiency it more than makes up for in reliability. But the network that makes up the Internet is far more complex and therefore functions within a far greater margin of error. What this all adds up to is one of the major flaws in VoIP: reliability.

First of all, VoIP is dependant on wall power. Your current phone runs on phantom power that is provided over the line from the central office. Even if your power goes out, your phone (unless it is a cordless)still works. With VoIP, no power means no phone. A stable power source must be created for VoIP.

Another consideration is that many other systems in your home may be integrated into the phone line. Digital video recorders, digital subscription TV services and home security systems, all use a standard phone line to do their thing. There is currently no way to integrate these products with VoIP. The related industries are going to have to get together to make this work.

Reference

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