A REPORT OF ONE MONTH TRAINING

at

SOLITAIRE INFOSYS PVT LTD

GURU NANAK DEV ENGINEERING COLLEGE LUDHIANA

SUBMITTED IN PARTIAL FULFILLMENT OF THE REQUIREMENT FOR THE AWARD OF THE DEGREE OF

BACHELOR OF TECHNOLOGY

(Computer Science & Engineering)

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GURU NANAK DEV ENGINEERING COLLEGE LUDHIANA

(An Autonomous College Under UGC ACT)

GURU NANAK DEV ENGINEERING COLLEGE, LUDHIANA

CANDIDATE'S DECLARATION

I " GAGANDEEP KAUR " hereby declare that I have undertaken one month training at "**SOLITAIRE INFOSYS PVT LTD** " during a period from 11/07/2022 to 10/08/2022 in partial fulfilment of requirements for the award of degree of B.Tech (Computer Science and Engineering) at GURU NANAK DEV ENGINEERING COLLEGE, LUDHIANA. The work which is being presented in the training report submitted to Department of Computer Science and Engineering at GURU NANAK DEV ENGINEERING COLLEGE, LUDHIANA is an authentic record of training work.

Signature of the Student

The one-month industrial training Viva – Voce Examination of__________________
has been held on______________ and accepted.

Signature of Internal Examiner   Signature of Internal Examiner

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CHAPTER – 1 INTRODUCTION

WHAT IS VOIP?

Voip. Voice over internet protocol (VoIP) is a type of phone system that uses an internet connection to make and receive calls, rather than traditional landlines. Most people consider VoIP the alternative to the local telephone company. If you have an internet connection, you can call anyone without the need for traditional, local phone service or physical copper wires. All you need is high-speed internet and a VoIP service provider to handle the calls. The best part is that you aren't bound to a specific desk. You can use a VoIP phone number via a business phone app to turn your computer or any mobile device into a phone. VoIP converts your phone calls into data that is sent over the internet. You can use the Ethernet cables or skip them if you have a strong **Wi-Fi** signal. It does so at a much lower cost than older telephone systems. Voice over IP has many advantages over traditional phone service.

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.

CHARACTERISTICS OF VOIP

Devices or software that use Voice over Internet Protocol provide a lot of additional functionalities that improve voice communications as compared to traditional technologies. Of course, they may vary depending on the brand that provides the service, but some of the most common features include:

- Call Transfers;

- Call Waiting;
- Call Identificator;
- Schedule access;
- Multiple accounts or lines;
- Call Park;

APPLICATIONS OF VOIP

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FaceTime - If you've got an Apple device (iPhone, iPad or iPod Touch) you can make free video calls to other users who also have the app. Apple devices come with the app pre-installed, so you can use it out of the box

Viber - like Skype, widely used by around 200 million subscribers, it lets you make free voice and video calls to friends and family who are also Viber users. You can use it on PCs, Macs, all mobile phones and tablets.

Skype - the biggest name in VoIP with around 600 million users, lets you make free voice or video calls to other Skype subscribers. You can use it on Windows computers and even on some smart TVs.

CHAPTER 2 GET STARTED WITH VOIP

WHAT IS VOIP?

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 bemade 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

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 nvolve a Gatekeeper

-Destination phone generates a ring, the called party picks up the phone, and a two-wayconversation 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.

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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.

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 COMPONENTS

Session Border Controller (SBC): Connects to analog call services with digital voice calls, call records, and provides bandwidth management control. Also, they regulate the flow and balancing of network traffic to maintain superior performance. Behind this VoIP gateway is access to the Public Switched Telephone Network (PSTN).

Media server: Enables features like voicemail and video calling, as well as voice to email, interactive voice response (IVR), and voice-based dialing. These provide useful functions like call recording, call queue messages, and more. These media gateways can handles hundreds, even thousands, of SIP calls at once.

Application server: Enables call forwarding, call waiting, call transfer, phone service to the IP network, and call detail records. These are essential to provide users with core telephony features.

Database services: Stores the registration details for all SIP devices, which locates an endpoint and translates addresses that are potentially different in various networks. These maintain call logs for all internet telephony activities.

SIP services: Session Initiation Protocol (**SIP**) takes care of connecting, disconnecting, and setting the parameters of call sessions. It acts as the foundation for modern voice, messaging, and video technologies.

IP PBX: An Internet Protocol (IP)-ready Private Branch Exchange (PBX) use to provide telephony within a company. It acts as the main control center for legacy phone systems. These rely on SIP trunking for voice service.

Endpoint devices: These are the VoIP phones or softphones (including smartphone apps) that receive VoIP phone service. Examples include desk phones, **VoIP apps**, conference phones, and even fax machines.

IP network: This allows the voice data packets to travel between endpoints over an Internet Protocol network to upstream services accessible via one or more IP addresses.

Codecs: For optimal call quality, codecs convert the analog signals to digital packets with different types of compression. G.722 is the standard for HD voice calls.

USES OF VOIP:-

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 f 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 themethods, 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 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.

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

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.

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 OF 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

CHAPTER – 3 PROJECT VOIP

COMMANDS ON VOIP

telephone-service

max-dn 10

max-ephone 10

auto assign 1 to 10

ip source-address 192.168.1.1 port 2000

do write

exit

ip dhcp pool abc

net 192.168.1.0 255.255.255.0

default-router 192.168.1.1

dns-service 1.1.1.1

option 150 ip 192.168.1.1

#exit

ephone-dn1

number 101

ephone-dn2 number 102

#

switch

inter ra fa0/1-11

switch voice vlan 1

CHAPTER-4 CONCLUSION AND FUTURE SCOPE

FUTURE SCOPE

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

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.

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REFERENCES

- www.google.com
- www.wikipedia.com
- www.studymafia.org