Mohamed Khider University of Biskra
Faculty of Sciences and Technology
Department of Electrical engineering

MASTER THESIS

ElectricalEngineering
Telecommunication
Networks and Telecommunication

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On : Monday, 25 June 2018

Design and configuration of VoIPConception et configuration d'un environnement de communication téléphonique sur Internet VoIP

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Academic Year : 2017 – 2018

الجمهورية الجزائرية الديمقراطية الشعبية
République Algérienne Démocratique et Populaire
وزارة التعليم العالي و البحث العلمي
Ministère de l'enseignement Supérieur et de la recherche scientifique

Université Mohamed Khider Biskra

Faculté des Sciences et de la Technologie
Département de Génie Electrique
Filière : Télécommunications
Option : Réseaux et Télécommunications

Mémoire de Fin d'Etudes
En vue de l'obtention du diplôme:

MASTER

Thème

Design and configuration of VoIP
Conception et configuration d'un environnement de communication téléphonique sur Internet VoIP

Présenté par :

MEFTAH Younes

Avis favorable de l'encadreur :

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Avis favorable du Président du Jury

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Cachet et signature

Dedication

I dedicate this work

The sake of Allah,my Creator and my Master,

My great Mom and messenger, Mohamed(May Allah bless

and grand him),who taught us the purpose of life

My beloved brothers and sister

My friends who encourage and support me

All the people in my life who touch my heart

Acknowledgment

As one journeys through life it is soon recognized that very little of what we accomplish as individuals is a solitary act. Cooperation and collaboration are what underlie our actions as human beings and it is through our inter-actions with others that we invariablycreate our art.

As such, I would like to acknowledge those colleagues, friends, and loved who made my master journey possible. I would like to thank the professors supervisor Dr. Z. HAMAIZIA Laboratory of Metallic and Semi conducting Materials (LMSM) Electrical Engineering Department, Faculty of Sciences and Technology University of Biskra for her unfailing good humour and insightful contributions to my overall purpose; my committee members for incredible vitality, love of prose, and abiding fidelity , enriching perspective.

Abstract:

VoIP is a set of protocols for voice transport over IP networks, several tools have emerged around this protocol,systems and software that are capable of managing communications with digital telephones, analog telephones and also provide the PBX hardware PBX functionalities, gateways with suppliers of VoIP services and other advanced functionalities.

The project will consist of the installation (software and hardware necessary) and advanced configuration (call forwarding, communication of groupse) of a system based on the asterisk free software and VoIP protocols, as well as the virtualization of service, for easy portability in different systems.

The objectives of this project are: to know the VoIP protocols, and become familiar with the existing software and hardware for this set of protocols, in addition to exploring their possibilities of implementation in networks corporate.

Key words: VoIP, Asterisk, PBX, IP telephony, PSTN

Résumé:

VoIP est un ensemble de protocoles pour le transport de la voix sur IP, plusieurs outils ont émergé autour de ce protocole, systèmes et logiciels capables de gérer les communications avec les téléphones numériques, les téléphones analogiques et les PBX matériels PBX, les passerelles avec les fournisseurs de VoIP, services et autres fonctionnalités avancées.

Le projet comprendra l'installation (logiciel et matériel nécessaire) et la configuration avancée (renvoi d'appel, communication de groupes) d'un système basé sur le logiciel sans astérisque et les protocoles VoIP, ainsi que la virtualisation de service, pour la portabilité facile dans différents systèmes.

Les objectifs de ce projet sont: connaître les protocoles VoIP, et se familiariser avec les logiciels et matériels existants pour cet ensemble de protocoles, en plus d'explorer leurs possibilités de mise en œuvre dans les réseaux d'entreprise.

Mots clés: RTCP,VOIP , Astérisque, IP,PBX

ملخص

الصوت عبر بروتكول الانترنت عبارة عن مجموعة من بروتوكولات النقل الصوتي عبر شبكات الانترنت ، وقد ظهرت أدوات عديدة حول هذا البروتوكول والأنظمة والبرمجيات القادرة على إدارة الاتصالات مع الهواتف الرقمية والهواتف

التناظرية وكذلك توفير وظائف PBX للأجهزة PBX والبوابات مع موردي VoIP الخدمات والوظائف المتقدمة الأخرى.

سيتألف المشروع من التركيب (البرامج والأجهزة اللازمة) والتكوين المتقدم (إعادة توجيه المكالمة ، واتصالات المجموعات) لنظام قائم على برمجيات النجمة الحرة وبروتوكولات VoIP ، بالإضافة إلى المحاكاة الافتراضية الخدمة ، لسهولة الحمل في أنظمة مختلفة.

تتمثل أهداف هذا المشروع في: معرفة بروتوكولات الاتصال الصوتي عبر الإنترنت (VoIP)، والتعرف على البرامج والأجهزة الموجودة لهذه المجموعة من البروتوكولات ، بالإضافة إلى استكشاف إمكانيات تنفيذها في الشبكات المشتركة.

كلمات مفتاحية: الصوت عبر بروتوكول الانترنت, بروتوكول الانترنت , PBX, PSTN

TABLE OF FIGURES

TABLE OF TABLES

Abbreviation

ABREVIATIONS LIST

A

ATA: Analog Telephone Adaptor

D

DHCP: Dynamic Host Control Protocol

DNS: Domain Name Server

DTMF: Dual Tone Multi Frequency

F

FDM: Frequency Division Multiplexing

FTP: Foiled Twisted Paired

FXO: Foreign eXchange Office

FXS: Foreign eXchange Subscriber

I

IAX: Inter-Asterisk eXchange

IP: Internet Protocol

IP-PBX: Internet Protocol Private Branch eXchange

IPV4: Internet Protocol Version 4

IPV6: Internet Protocol Version 6

ISDN: Integrated Service Digital Network

IVR: Interactive Voice Response

L

LAN: Local Area Network

M

MCU: Multipoint Control Unit

N

NAT: Network Address TranslatorO

P

PBX: Private Branch eXchange

R

RTP: Real time Transport Protocol

RTCP : Real time Control Protocol

S

SDES: Source Description

SDP: Session Description Protocol

SIP: Session Initiation Protocol

SMTP: Simple Mail Transfer Protocol

SR: Sender Report

SSH: Secure Shell

STP: Shielded Twisted Pair

T

TCP/IP: Transmission Control Protocol/Internet Protocol

TCP: Transmission Control Protocol

TDM: Time Division Multiplex

U

UAC: User Agent Client

V

VoIP: Voice over IP VoIP

W

WAN: Wide Area Network

WIFI: WIreless FIdelity

WLAN: Wireless Local Area Network

WWW: World Wide Web

TABLE OF CONTENTS

CHAPTER II

TELEPHONY NETWORK

CHAPTER III

VOIP FUNDAMENTALS

General Introduction

Vocal communication is one of the most important parts of human development. The Man's need to communicate at great distances has influenced his development to all levels and at all times, from personal levels, to levels economic, from local developments to national or continental developments, from prehistory to the present. Technology has always tried to facilitate this need for remote communication from the beginning of evolution, from the first communications with signals from smoke, going through the telegraph, to the current communications through telephony mobile that allow us to communicate from almost any location on the planet, and even today, where communications have been made from thespace.

Therefore, one of the most widespread, used, and common technologies are the related to voice communications. In a current society, that many they call "information society" in which information is crucial for the development of any activity and in which the Internet is increasingly important, and more part of our everyday world, it is obvious that communications are a vital importance for the development of any business activity.

In addition the incessant and increasingly strong role of the Internet in life of the people and the current lines of technological development of the telecommunications, in which there is a strong tendency towards the so-called "all IP", do logical development of IP-based technologies that allow these communications and voice services, integrated within the Internet that are so demanded and necessary They are in today's society.

but we also talk about video communications, voice messaging services, automatic response voice systems, etc

Basically, VoIP (Voice over IP) or VoIP (Voice over IP) is a set of protocols for voice transport over IP networks, and we should not only understand the use of VoIP for use on the Internet, but we have to include any network that works under this protocol, although as is obvious the Internet is the most important. From this simple definition it is difficult to understand that VoIP covers a large number of technologies, since the services that it provides us and the technologies involved are many and very varied.

When we talk about VoIP we must also talk about their environment, because, around this protocol, and driven by this, have emerged various software utilities and hardware devices, which allow their development and growth.

Chapter I

I.1. INTRODUCTION

Each of the past three centuries has been dominated by a single technology, the 18th century was the era of the great mechanical systems accompanying the industrial revolution. The 19th century was the age of the steam engine, During the 20th century, the key technology was information gathering, processing, and distribution.
Among other developments, we saw the installation of world wide telephone networks, the invention of radio and television, the birth and unprecedented growth of the computer industry, and the launching of communication satellites.

As a result of rapid technological progress, these areas are rapidly converging and the differences between collecting, transporting, storing, and processing information are quickly disappearing, organizations with hundreds of offices spread over a wide geographical area routinely expect to be able to examine the current status of even their most remote outpost at the push of a button.

In this chapter, we begin to provide an overview of basic networking concepts, including network architecture, design, then describe cabling and technologies used to link network devices, finally we describe the communications protocols associated with networks.

I.2. WHAT IS NETWORK ?

- A network consists of two or more computers that are linked in order to share resources (such as printers and CDs), exchange files, or allow electronic communications.
- The computers on a network may be linked through cables, telephone lines, radio waves, satellites, or infrared light beams (1).

I.3. TYPES OF NETWORK TOPOLOGY

I.3.1. Bus Topology

- All nodes (file server, workstations, and peripherals) are connected to the linear cable.
- Popular on LANs because they are inexpensive and easy to install.
- All the nodes (file server, peripherals) are connected by one single cable (1).

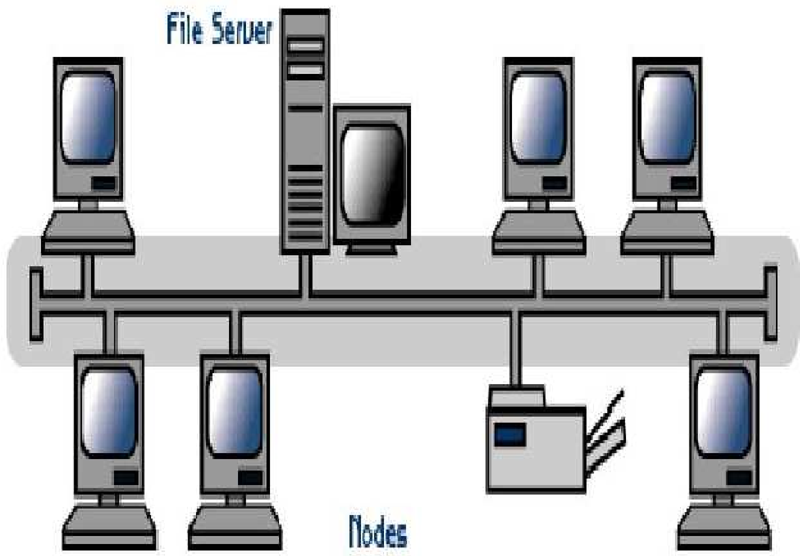

Figure.I.1:Bus topology.

I.3.2. Ring Topology

- In a ring network, every device has exactly two neighbours for communication purposes.
- All messages travel through a ring in the same direction.
- A failure in any cable or device breaks the loop and can take down the entire network.
- To implement a ring network we use the token ring technology.
- A token, or small data packet, is continuously passed around the network, when device needs to transmit, it reserves the token for the next trip around, then it attaches its data packet to it (1).

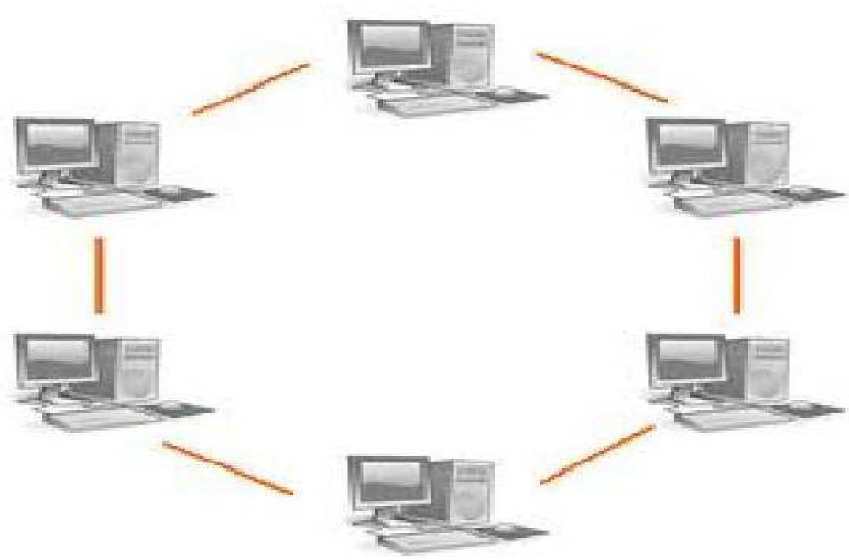

Figure.I.1:Ring topology.

I.3.3. Star Topology

- In a star network, each node (file server, workstations, and peripherals) is connected to a central device called a hub.
- The hub takes a signal that comes from any node and passes it along to all the other nodes in the network.
- Data on a star network passes through the hub, switch, or concentrator before continuing to its destination.
- The hub, switch, or concentrator manages and controls all functions of the network.
- The star topology reduces the chance of network failure by connecting all of the systems to a central node (1).

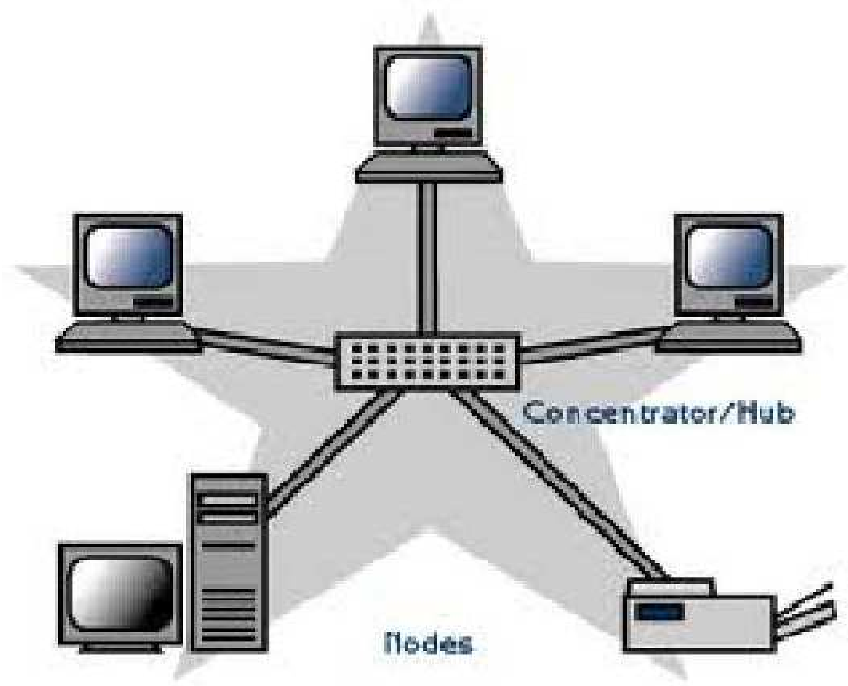

Figure.I.2:Star topology.

I.3.4. Tree Topology

- A tree topology (hierarchical topology) can be viewed as a collection of star networks arranged in a hierarchy.
- This tree has individual peripheral nodes which are required to transmit to and receive from one other only and are not required to act as repeate regenerators.
- The tree topology arranges links and nodes into distinct hierarchies in order to allow greater control and easier troubleshooting.
- This is particularly helpful for colleges, universities and schools so that each of the connect to the big network in some way (1).

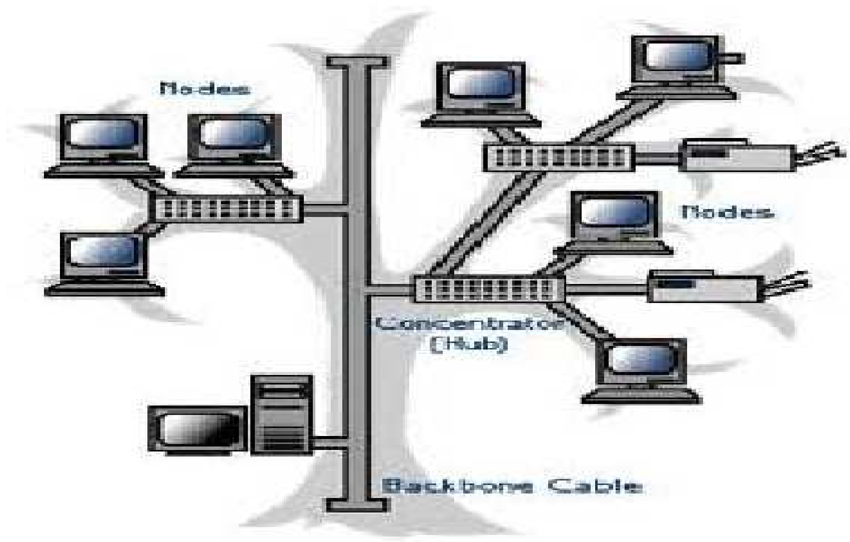

Figure.I.3:Tree topology.

I.3.5. Mesh Topology

- In this topology, each node is connected to every other node in the network.
- Implementing the mesh topology is expensive and difficult.
- In this type of network, each node may send message to destination through multiple paths.
- While the data is travelling on the Mesh Network it is automatically configured to reach the destination by taking the shortest route which means the least number of hops.
- Mesh has n(n-1)/2 physical channels to link n devices (1).

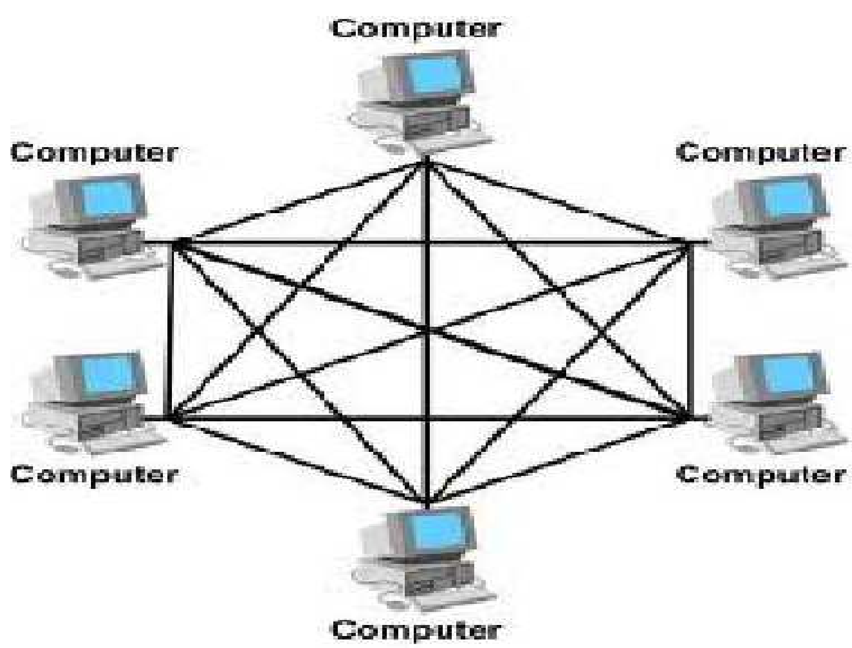

Figure.I.4:Mesh topology.

I.3.6. Hybrid Topology

- A combination of any two or more network topologies.
- A hybrid topology always accrues when two different basic network topologies are connected.
- It is a mixture of above mentioned topologies. Usually, a central computer is attached with sub controllers which in turn participate in a variety of topologies (1).

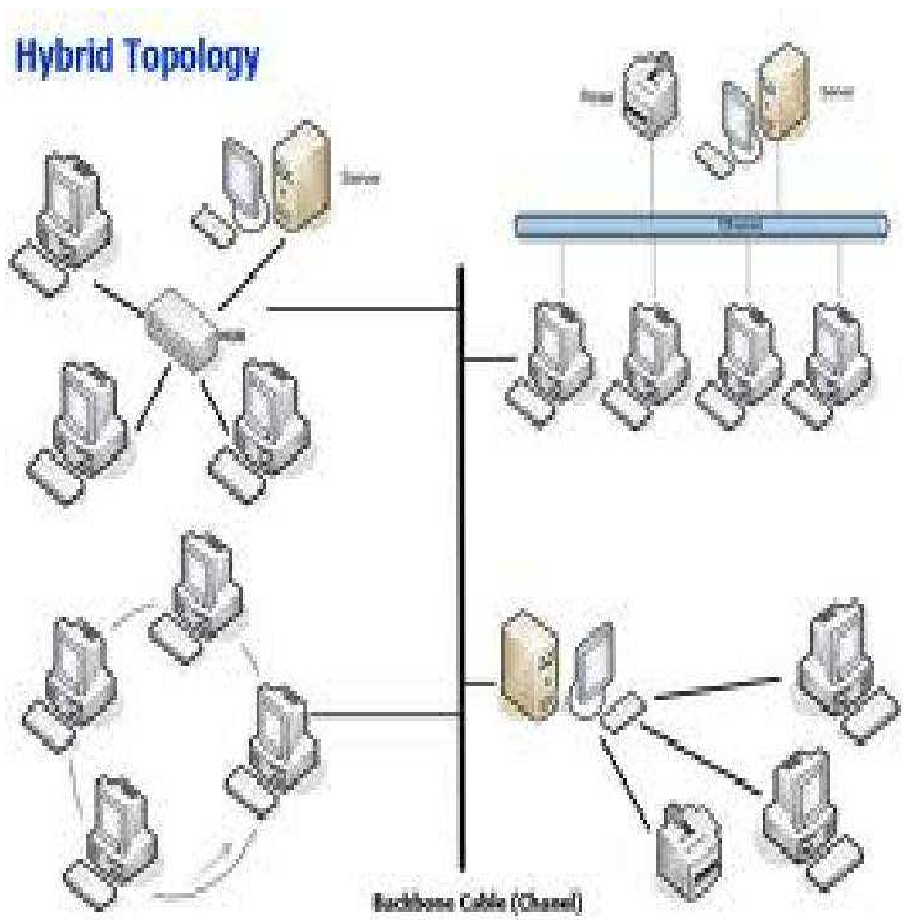

Figure.I.5:Hybrid topology.

I.4. TYPE OF COMMUNICATION NETWORKS

Communication Networks can be of following 5 types:

1) Local Area Network (LAN).
2) Metropolitan Area Network (MAN).
3) Wide Area Network (WAN).
4) Wireles.
5) Inter Network (Internet) (2).

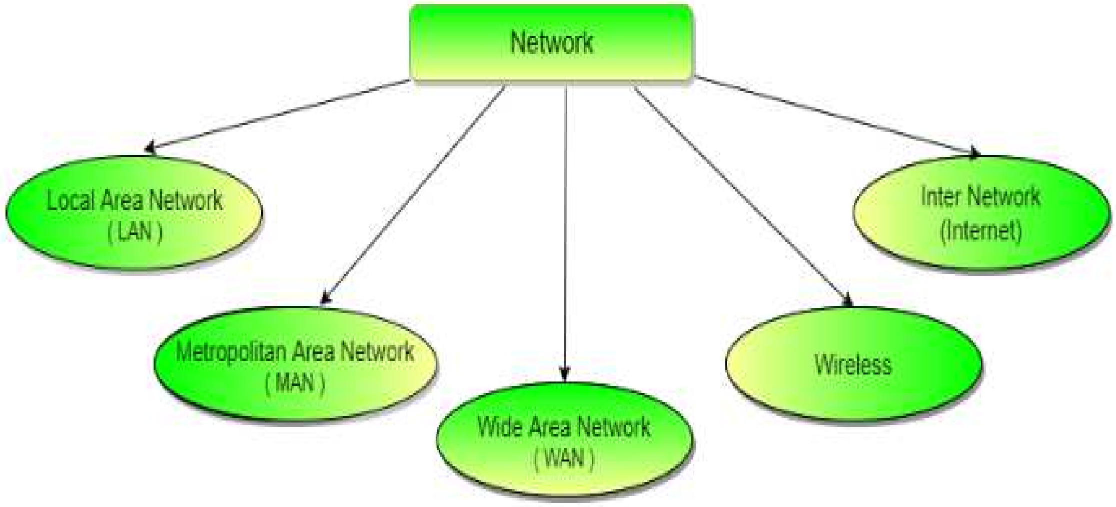

Figure.I.6:Type of communication.

I.5. TRANSMISSION MEDIA

The means through which data are transmitted and received, transmission media may be physical, such as wire or cable, or atmospheric (wireless), such as radio waves.

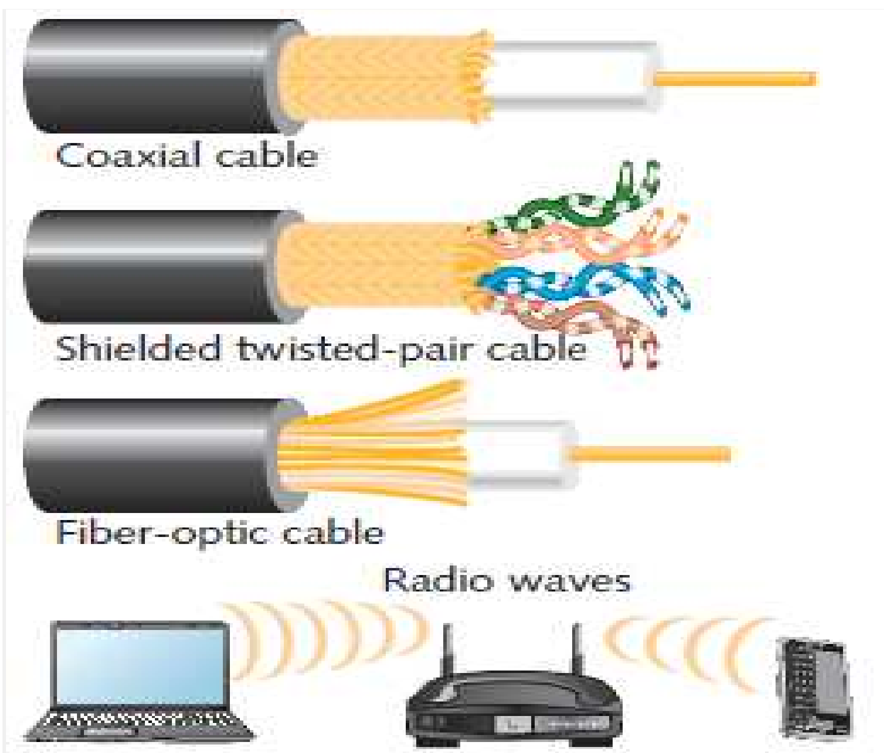

Figure.I.7:Examples of transmission media.

Each of these media has vastly different physical properties and uses different methods to encode messages :

Media	Example	Encoding
Copper	Twisted-pair cable usually used as LAN media.	Electrical pulses
Fiber-optic	Glass or plastic fibers in a vinyl coating usually used for long runs in a LAN and as a trunk.	Light pulses
Wireless	Connects local users through the air.	Electromagnetic waves

Table.I.1:Networking media.

The difference in the media make each one ideal for different roles in networking situations. When choosing network media, administrators must consider the following:

- The distance the media can carry the signal.
- The environment in which the media works.
- The bandwidth requirements for users.
- The cost of installation.
- The cost of connectors and compatible equipment (3).

I.6. TRANSMISSION MODES IN COMPUTER NETWORKS

Transmission mode refers to the mechanism of transferring of data between two devices connected over a network.

These modes direct the direction of flow of information, there are three types of transmission modes:

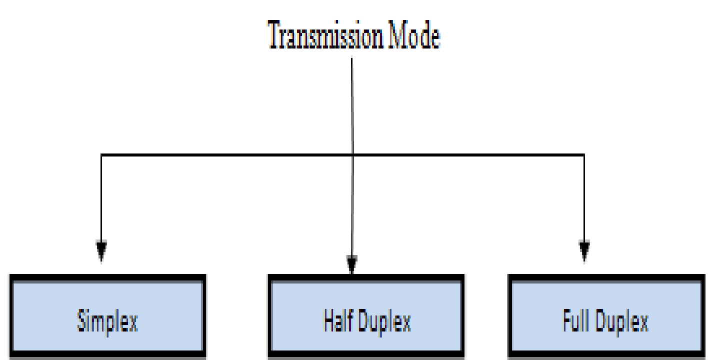

Figure.I. 8 :Transmission modes.

I.6.1. Simplex Mode

Signal is transmitted in one direction only, example radio and TV and paging systems (4).

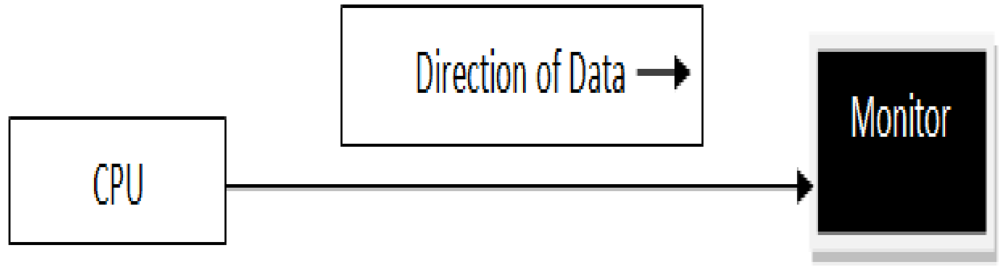

Figure.I.9:Simplex mode.

I.6.2. Half duplex Mode

Signals are transmitted in one direction at a time, example radio systems (4).

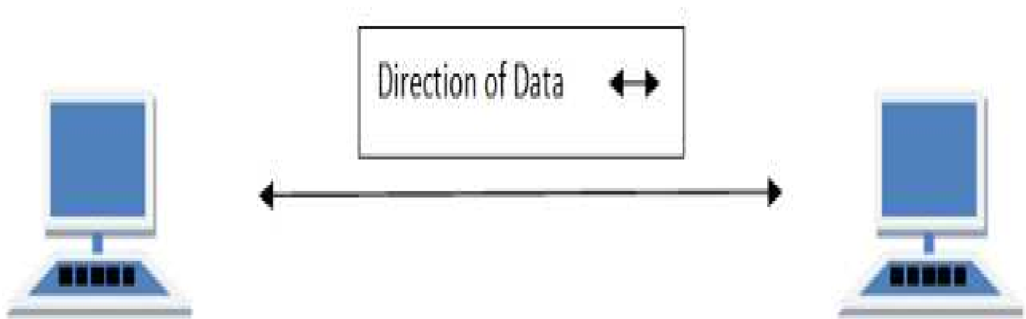

Figure.I.10:Half duplex mode.

I.6.3. Full duplex Mode

Signals are transmitted in both directions at the same time, examples : Conventional telephone, cellular or mobile telephone systems and ISD (4).

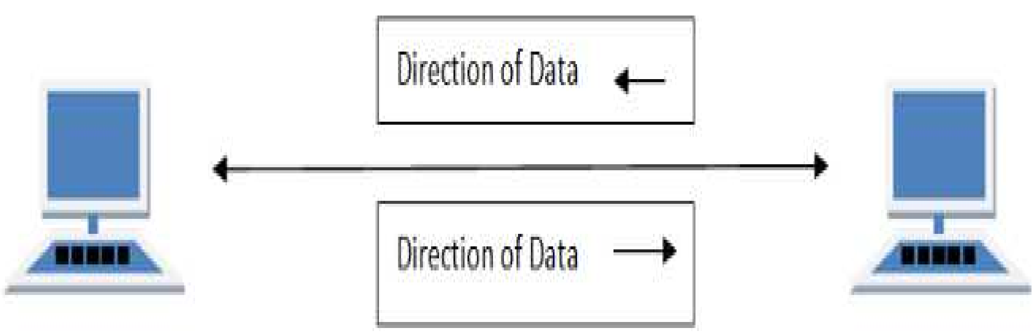

Figure.I.11:Full duplex mode.

I.7. THE OPEN SYSTEMS INTERCONNECTION (OSI) MODEL

Known as the OSI model, provides an abstract description of the network communication process, developed by the International Organization for Standardization (ISO) to provide a road map for non proprietary protocol development.

The OSI model is just a reference model, so manufacturers have been free to create protocols and products that combine functions of one or more layers (3).

No	Layer Name	Description
7	Application	Performs services for the applications used by the end users.
6	Presentation	Provides data format information to the application, For example, the presentation layer tells the application layer whether there isencryption or whether it is a .jpg picture.
5	Session	Manages sessions between users. For example, the session layer.
4	Transport	Defines data segments and numbers them at the source, transfers the data, and reassembles the data at the destination.
3	Network	Creates and addresses packets for end-to-end delivery through intermediary devices in other networks.
2	Data Link	Creates and addresses frames for host-to-host delivery on the local LANs and between WAN devices.
1	Physical	Transmits binary data over media between devices, physical layer protocols define media specifications.

Table.I.2:OSI model.

I.8. TCP/IP MODEL

The TCP/IP model defines the four communication functions that protocols perform. TCP/IP is an open standard, which means that one company does not control it, the rules and implementations of the TCP/IP model were cooperatively developed by members of the industry using Request for Comments (RFC) documents, RFC documents are publicly accessible documents that define specifications and policies of the protocols and of the Internet in general, solicitation and maintenance of RFC are the responsibility of the IETF (3).

Layer	Description
Application	Represents application data to the user, Forexample, the HTTP presents data to the user in a web browser application like InternetExplorer.
Transport	Supports communication between devices and performs error correction.
Internet	Finds the best path through the network.
Network access	Controls hardware devices and media.

Table.I.3:The functions of each layer of the TCP/IP model.

I.9. MOVING DATA IN THE NETWORK

I.9.1. Protocol Data Units (PDUs)

As application data is passed down the protocol stack on its way to be transmitted across the network media, various protocols add information to it at each level , this is commonly known as the encapsulation process.

The form that a piece of data takes at any layer is called a protocol data unit (PDU), during encapsulation, each succeeding layer encapsulates the PDU that it receives from the layer above in accordance with the protocol being used, at each stage of the process, a PDU has a different name to reflect its new functions.

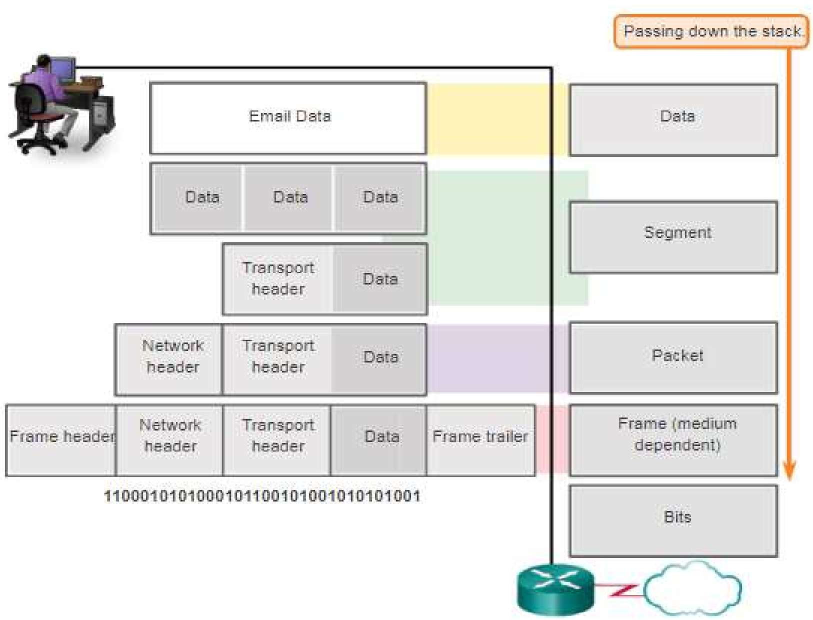

Figure.I.12:Protocol data unit naming conventions.

I.9.2. End-to-End Data Flow

The flow of data from the application layer to the physical layer is called encapsulation encapsulation, the flow of data in the opposite direction, from the physical layer to the application layer, is called decapsulation.

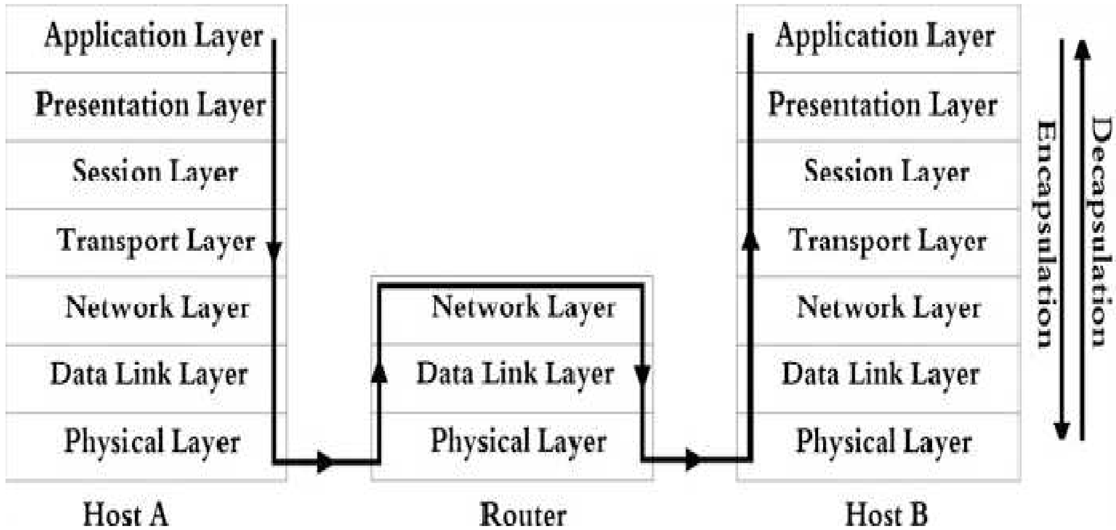

Figure.I.13:The process of transferring data.

I.9.3. Encapsulation

Data encapsulation is the process that adds additional protocol header information to the data before transmission, in most forms of data communications, the original data is encapsulated or wrapped in several protocols before being transmitted.

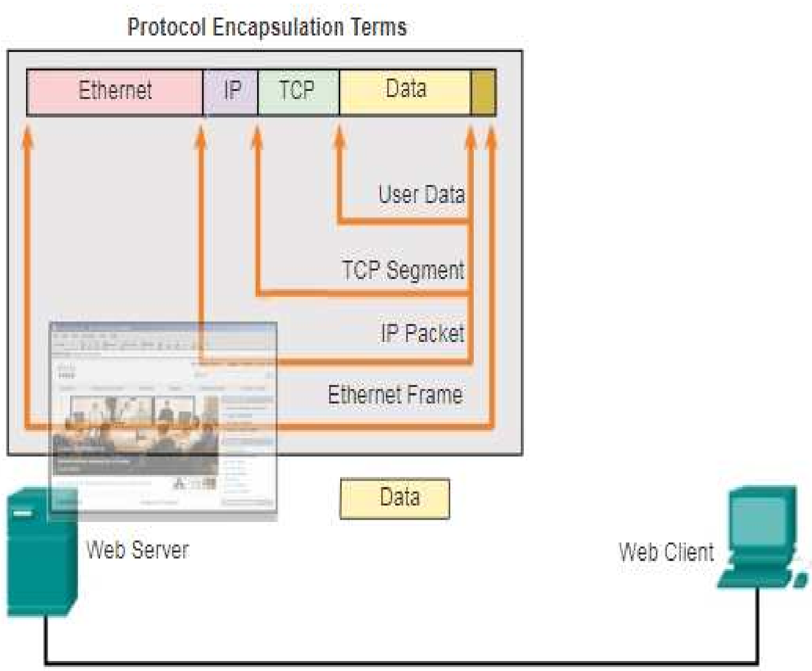

Figure.I.14:The encapsulation process.

I.9.4. De-encapsulation

This process is reversed at the receiving host, and is known as de-encapsulation. De-encapsulation is the process used by a receiving device to remove one or more of the protocol headers.

The data is de-encapsulated as it moves up the stack toward the end-user application.

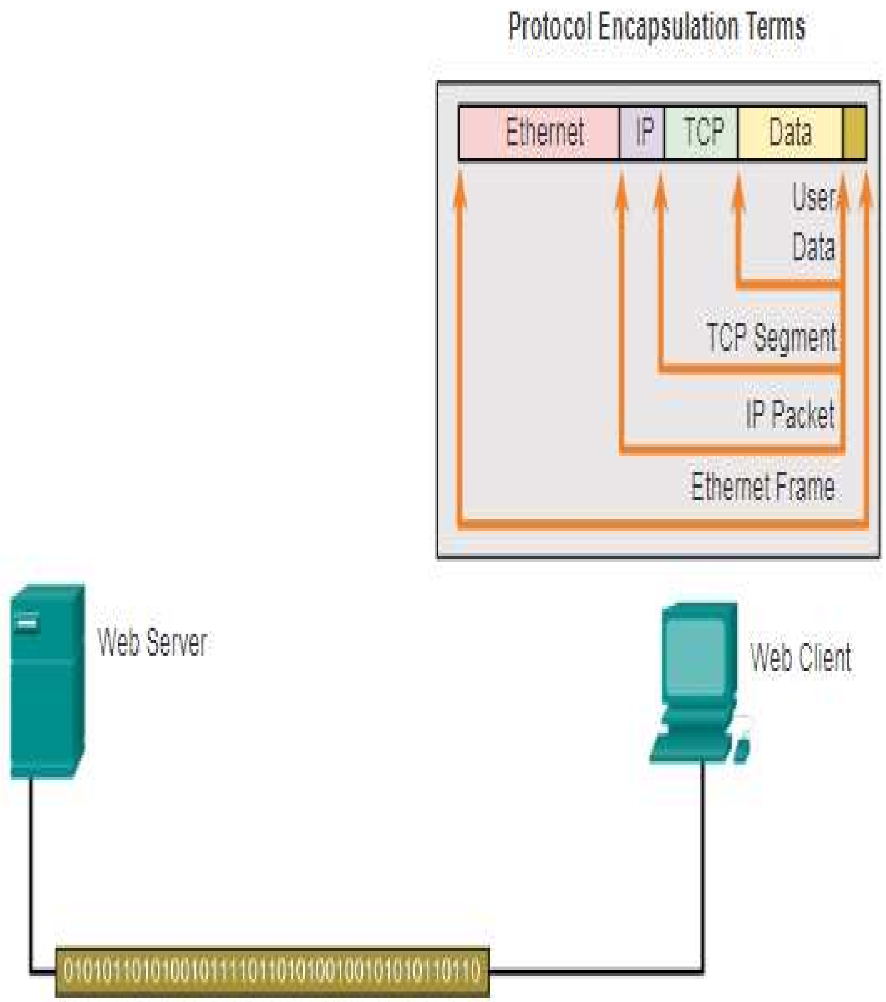

Figure.I.15:The de-encapsulation process.

I.10. IP ADDRESSING

An IP address is a numeric identifier assigned to each machine on an IP network, it designates the specific location of a device on the network.

An IP address is a software address, not a hardware address the latter is hard-coded on a Network Interface Card (NIC) and used for finding hosts on a local network, IP addressing was designed to allow a host on one network to communicate with a host on a different network, regardless of the type of LANs the hosts are participating in (5).

I.10.1. Class Addresses

The actual sizes of the network and host part of the addresses in a network can be easily predicted.

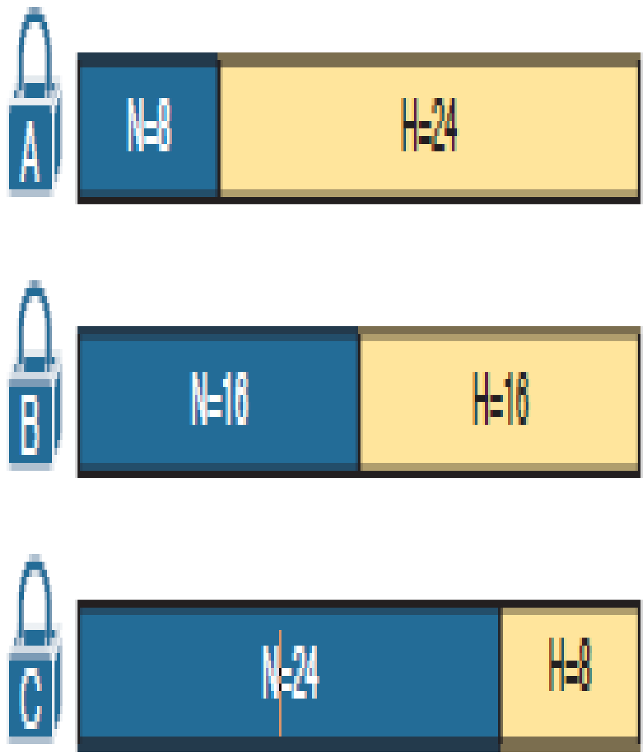

Figure.I.16:Addresses sizes.

The size of an unsubnetted Class A, B, or C network is as follows:

- Class A**:** $2^{24} - 2 = 16{,}777{,}214$
- Class B**:** $2^{16} - 2 = 65{,}534$
- Class C**:** $2^{8} - 2 = 254$ (6)

Class addresses	Range decimal	Range binary
A	1-126	00000001-01111111
B	128-191	10000000-10111111
C	192-223	11000000-11011111
D	224-239	11100000-11101111

Table.I.4:Network numbers and classes of addresses (7).

I.10.2. Types of Addresses

Many different types of IP addresses exist, offers a brief description of these types (8).

IP addresses Type	Definition
Reserved	Addresses that cannot be assigned to a network component, such as a network address, a broadcast, or Class D and E addresses.
Network ID	The network number assigned to a segment or VLAN.
Host ID	The host component of an IP address.
Directed Broadcast	The broadcast address of a network ID these can be routed by routers.
Local Broadcast	The all hosts address for everyone on the same segment: 255.255.255.255 (routers will not route this type of address).
Loopback	The internal address of a device used for testing functions (127.0.0.1).
Autoconfigured	An address automatically assigned to a network component (DHCP is an example of this type of addressing).
Public	An address used to access devices across the Internet or other public networks.
Private	An address to access devices in a local network, which cannot be used to access public networks.

Table.I.5:Types of IP addresses.

I.10.3. Subnet masks

Subnet masking is used to recognize or reveal the subnet contained in the IP address, every node on the same subnet must have the same subnet mask.

The subnet mask helps you to decode the IP address and decipher the individual network bits, subnet bits, and host or interface bits, network devices use the subnet mask to determine which part of the IP is a network address and which part is the interface, by using the IP address and subnet mask, networking devices can calculate which devices are located on the same subnet, on a different subnet on the same organizational network, or on a completely different network (9).

I.10.4. Variable-length subnet masking :

Is the more realistic way of subnetting a network to make for the most efficient use of all of the bits. VLSM is the process of subnetting a subnet and using different subnet masks for different networks in your IP plan (10).

I.11. Network Address Translation (NAT)

I.11.1. When Do We Use NAT?

NAT, at times, decreases the overwhelming amount of Public IP addresses required in your networking environment, and NAT comes in really handy when two companies that have duplicate internal addressing schemes merge, NAT is also great to have around when an organization changes its Internet Service Provider (ISP) and the networking manager doesn't want to hassle with changing the internal address scheme.

List of situations when it's best to have NAT on your side:

- You need to connect to the Internet and your hosts don't have globally unique IP addresses.
- You change to a new ISP that requires you to renumber your network.
- You require two intranets with duplicate addresses to merge (11).

1.11.2. Types of Network Address Translation

- **Static NAT :**

This type of NAT is designed to allow one-to-one mapping between local and global addresses, keep in mind that the static flavor requires that you have one real internet IP address for every host on your network.

- **Dynamic NAT :**

This version gives you the ability to map an unregistered IP address to a registered IP address from out of a pool of registered IP addresses, we don't have to statically configure your router to map an inside to an outside address like you would using static NAT, but we do have to have enough real, bona-fide IP addresses for everyone who's going to be sending packets to and from the Internet (6).

I.12. DYNAMIC HOST CONFIGURATION PROTOCOL

The DHCP enables clients on a network to obtain IP addresses and other information from a DHCP server, the protocol automates the assignment of IP addresses, subnet masks, gateway,and other IP networking parameters.

DHCP allows a host to obtain an IP address dynamically when it connects to the network.

The DHCP server is contacted by sending a request, and an IP address is requested.

The DHCP server chooses an address from a configured range of addresses called a pool and assigns it to the host client for a set period. (3)

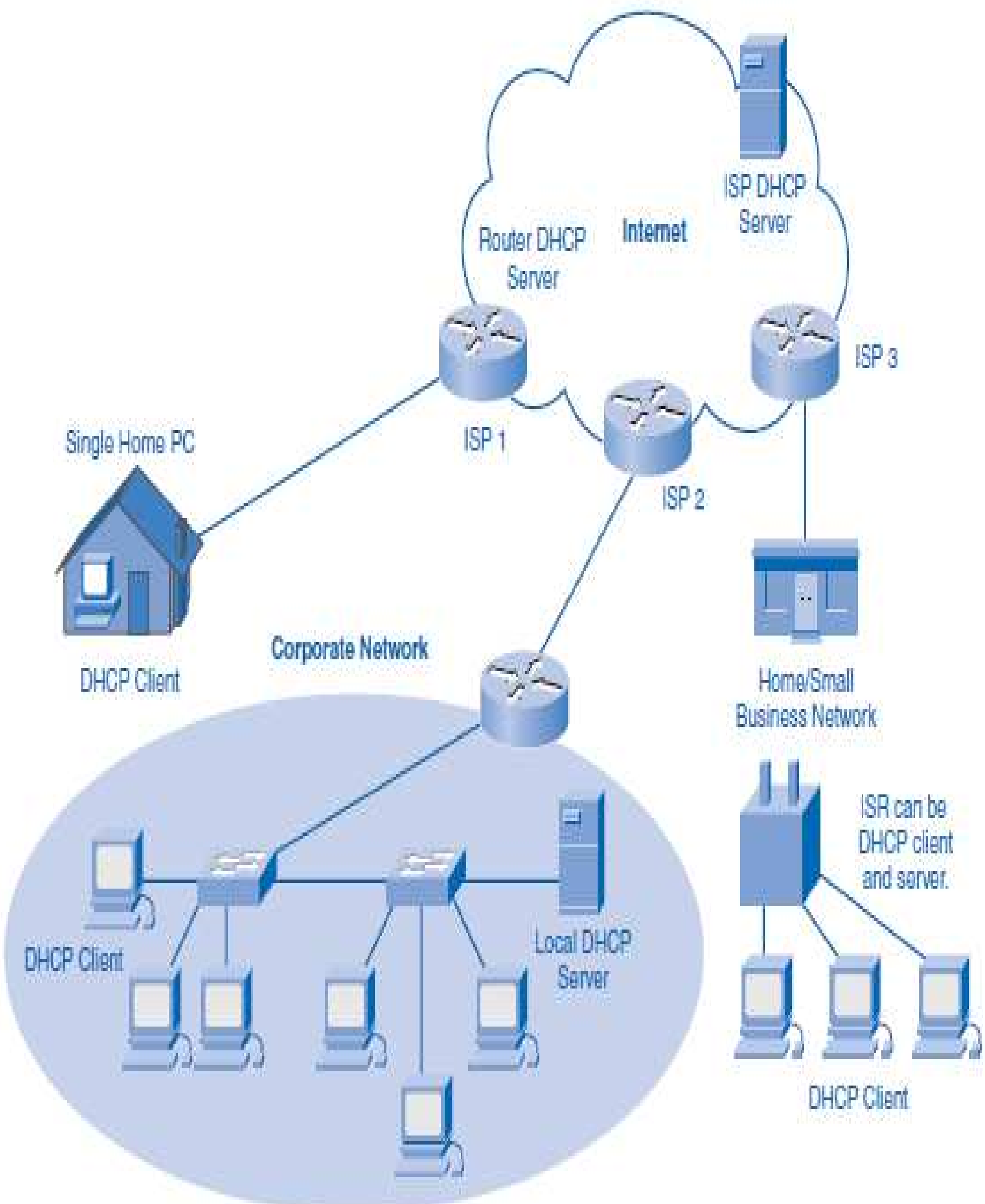

Figure.I.17:The different DHCP servers.

DHCP clients obtain a DHCP lease for an IP address, a subnet mask, and various. DHCP options from DHCP servers in a four-step process:

1. **DHCP Discover :**

The client broadcasts a request for a DHCP server.

2. **DHCP Offer :**

DHCP servers on the network offer an address to the client.

3. **DHCP Rerquest :**

The client broadcasts a request to lease an address from one of the offering DHCP servers.

4. **DHCP Acknowledges :**

The DHCP server that the client responds to acknowledges the client, assigns it any configured DHCP options, and updates its DHCP database.

The client then initializes and binds its TCP/IP protocol stack and can begin network communication. (12)

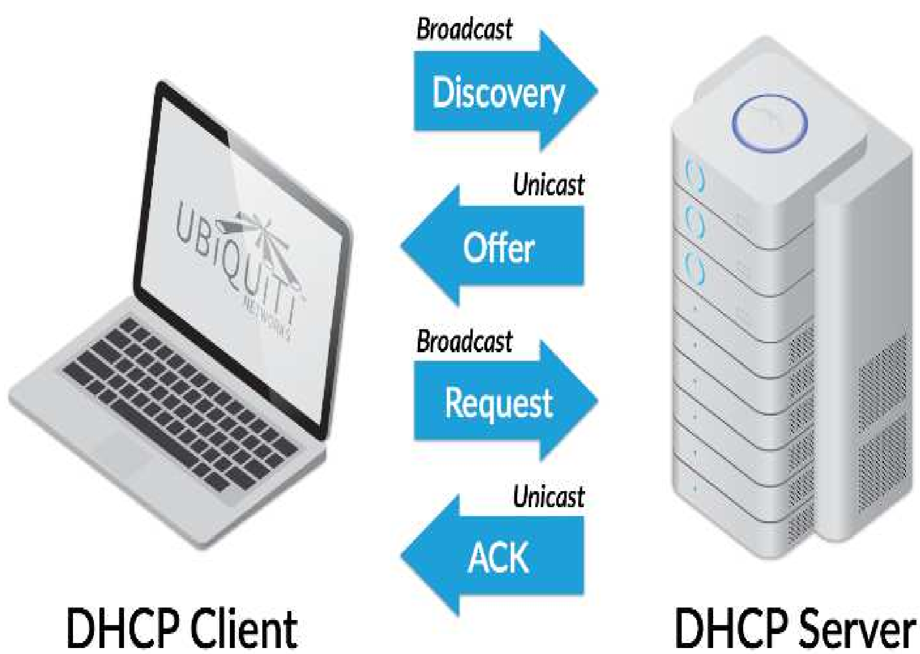

Figure.I. 18 : DHCP servers step process. (13)

I.13. DOMAIN NAME SYSTEM (DNS)

I.13.1. How DNS Works

The DNS protocol defines an automated service that matches resource names with the required numeric network address, it includes the format for queries, responses, and data formats.

DNS protocol communications use a single format called a message, this message format is used for all types of client queries and server responses, error messages, and the transfer of resource record information between servers.

When configuring a network device, you generally provide one or more DNS server addresses that the DNS client can use for name resolution, usually the Internet service provider (ISP) gives you the addresses to use for the DNS servers, when a user's application requests to connect to a remote device by name, the requesting DNS client queries one of these DNS servers to resolve the name to a numeric address (3).

I.13.2. DNS Hierarchy

DNS uses a hierarchical system to create a name database to provide name resolution. The hierarchy looks like an inverted tree with the root at the top and branches below.

At the top of the hierarchy, the root servers maintain records about how to reach the toplevel domain servers, which in turn have records that point to the secondary-level domain servers and so on.

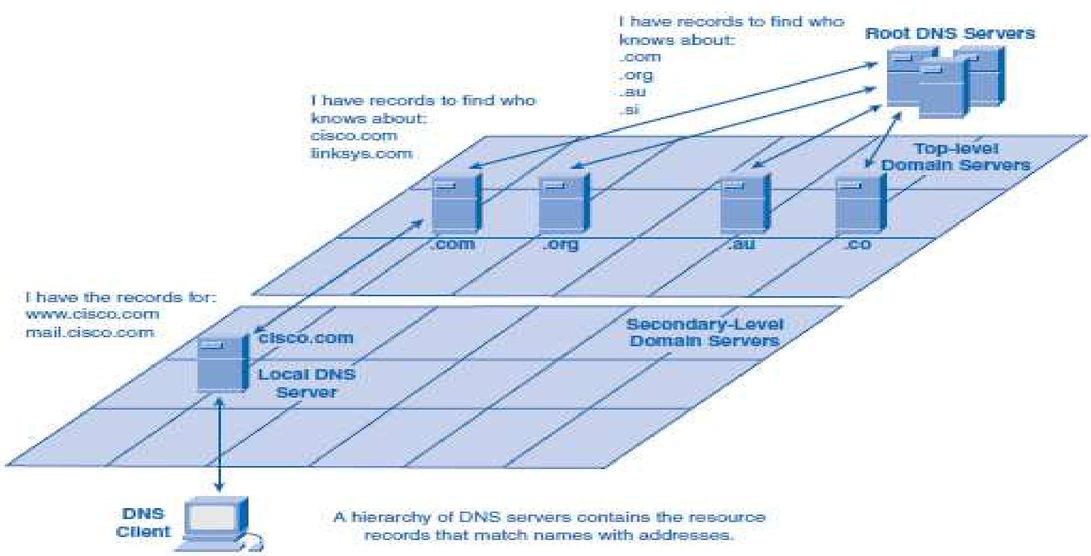

Figure.I.2:Hierarchy of DNS servers.

The different top-level domains represent either the type of organization or the country of origin, the following are examples of top-level domains are:

- **.au:** Australia
- **.co:** Colombia
- **.com:** A business or industry
- **.jp:** Japan
- **.org:** A nonprofit organization

After top-level domains are second-level domain names, and below them are other lower level domains (3).

I.14. SUMMARY

A computer network is a system for connecting two or more devices using one of the systems technologies Communication for the exchange of information, resources and data among them available to the network such as the machine Printer or application software of any kind and also allow for direct communication between users.

In general, the study of computer networks is one of the branches of communication science, it is possible that the computers in the network are very close to each other.
The network is made up of a set of devices in as far away as inter-urban networks countries and even continents and such networks are often connected to the internet or satellite.

The next chapter provides with the background covering traditional telephony, Topics such as analog network signaling, analog interface types, the analog to digital conversion process, and numbering plans are detailed to give you a firm foundation in traditional voice terminology and processes.

Chapter II

II.1. INTRODUCTION

Since its inception in 1876, the telephone has captured the imagination of people around the world, from its simple origins, the telephone has evolved from the humble device it once was to the modern cell phone or satellite phone.

The telephone has fundamentally and instantaneously changed the human race's ability to communicate over long distances, today not only can we communicate via the human voice but we can rapidly send scientific and health data, pictures, and information between any two people or organizations on earth in just a few seconds by picking up the phone and dialing the party at the other end, the development of the telephone has progressed at such a rapid rate in the last 50 years alone that it is truly amazing what can be done with the telephone in our modern lives; and it is now mostly taken for granted.

In this chapter, we discuss what would be known as the traditional telephony world. It begins where the telephone system originally started analog connectivity, it then moves into digital connections and considerations and concludes the traditional voice discussion with the primary pieces need to know from the public switched telephone network (PSTN).

II.2. THE CIRCUIT SWITCHED NETWORK

The most common telephone system on the planet today is still analog, especially at the edge of the network, analog telephony uses the modulation of electric signals along a wire to transport voice.

Although it is a very old technology, analog transmission has many advantages:

- It is simple and keeps the end-to-end delay of voice transmission very low because the signal propagates along the wire almost at the speed of light.
- It is also inexpensive when there are relatively few users talking at the same time and when they are not too far apart, but the most basic analogue technology requires one pair of wires per active conversation, which becomes rapidly unpractical and expensive analog telephony has many drawbacks:
- Unless we use manual switchboards, analog switches require a lot of electromechanical gear, which is expensive to buy and maintain.
- Parasitic noise adds up at all stages of the transmission because there is no way to differentiate the signal from the noise and the signal cannot be cleaned (13).

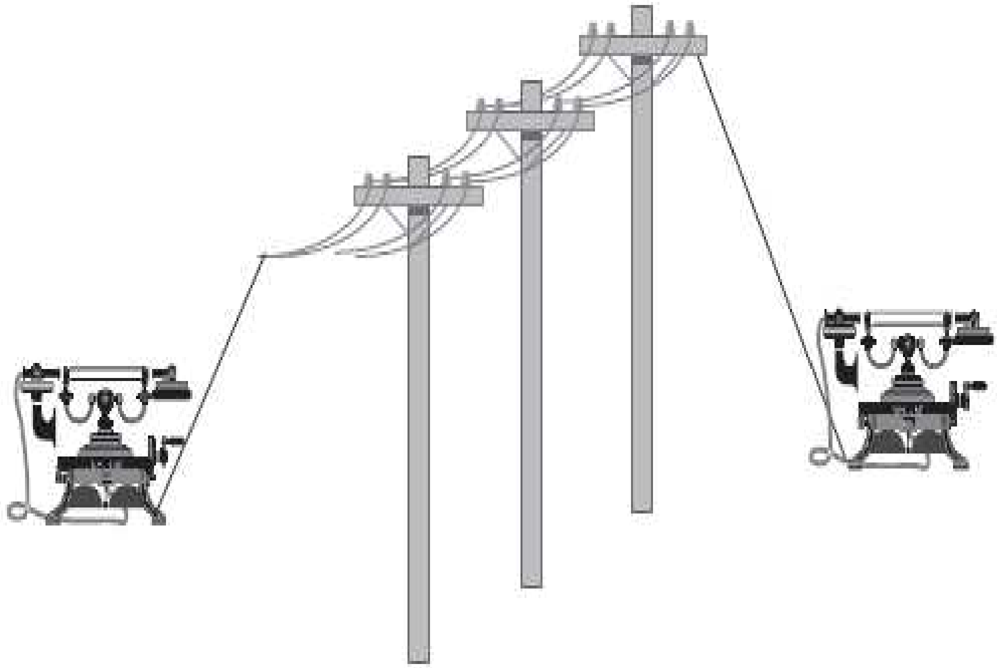

Figure.II.1:Analog telephony.

II.3. THE PUBLIC SWITCHED TELEPHONE NETWORK

Telephones are so simple to use that they hide the complexity inside the network that provides the many features we enjoy, in designing a UC deployment,it's good to understand what UC will replace and extend; that is, what we have used to date.

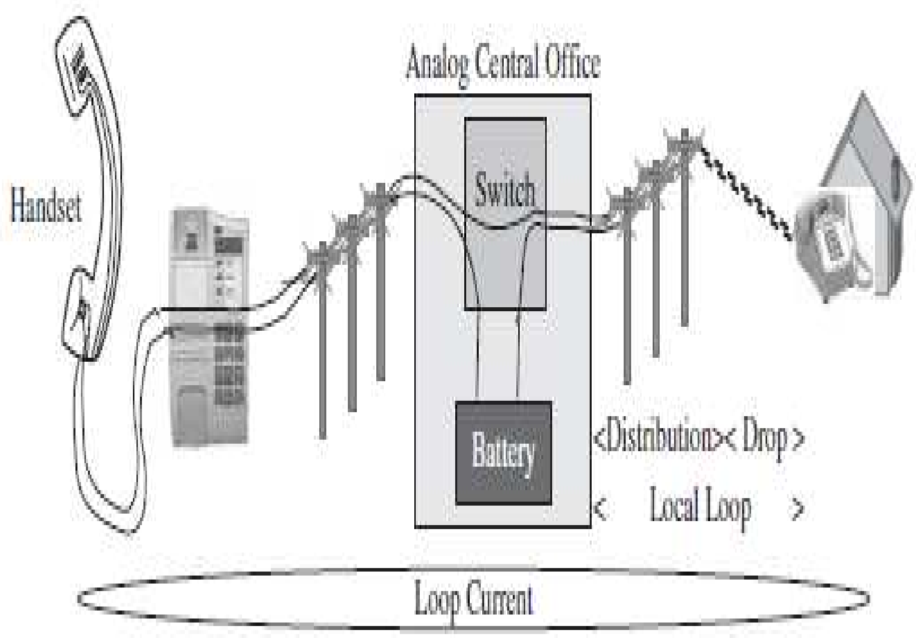

Figure.II.2:Current loop from CO battery to phone.

Describes the original telephone technology, the analog phone or pots line Bell's great invention, the phone at the house or office connects to the telco's central office over a 2-wire copper line, the copper wires are twisted to reduce interference from external sources, such as AM radio stations and large electrical motors, but are not shielded by an external metal wrap hence the term unshielded twisted pair (UTP), Electrical current to operate the phone comes from the battery in the central office, the phone needs no other power supply, power from the CO was necessary when the first phones were installed because at that time lighting was by gas, not many homes (and not all offices) had electricity.

Electrical current flows in a loop from end to end, through both phones, the portion of the connection between the customer and the CO came to be called the local loop, the transmitter in the mouth piece varies the rate of current flow in response to the sound waves from a talker's mouth, since the current flows in a loop, the same changes occur at the receiver where the miniature audio speaker in the earpiece reproduces the talker's voice (14).

All the signaling standards and communication methods discussed in the previous section typically focus around the connection to one, massive voice network known as the public switched telephone network (PSTN), if we have ever made a call from a home telephone, we have experienced the results of the traditional telephony network, this network is not unlike many of the data networks of today.

II.4. PIECES OF THE PUBLIC SWITCHED TELEPHONE NETWORK

When the phone system was originally created, individual phones were wired together to allow people to communicate, The modern PSTN is now a worldwide network (much like the Internet), built from the following pieces :

- Analog telephone : Able to connect directly to the PSTN and is the most common device on the PSTN. Converts audio into electrical signals.
- Local loop : The link between the customer premises (such as a home or business) and the telecommunications service provider.
- CO switch : Provides services to the devices on the local loop, these services include signaling, digit collection, call routing, setup, and teardown.
- Trunk : Provides a connection between switches, these switches could be CO or private.
- Private switch : Allows a business to operate a miniature PSTN inside its company. This provides efficiency and cost savings because each phone in the company does not require a direct connection to the CO switch.

Digital telephone : Typically connects to a PBX system. Converts audio into binary 1s and 0s, which allows more efficient communication than analog.

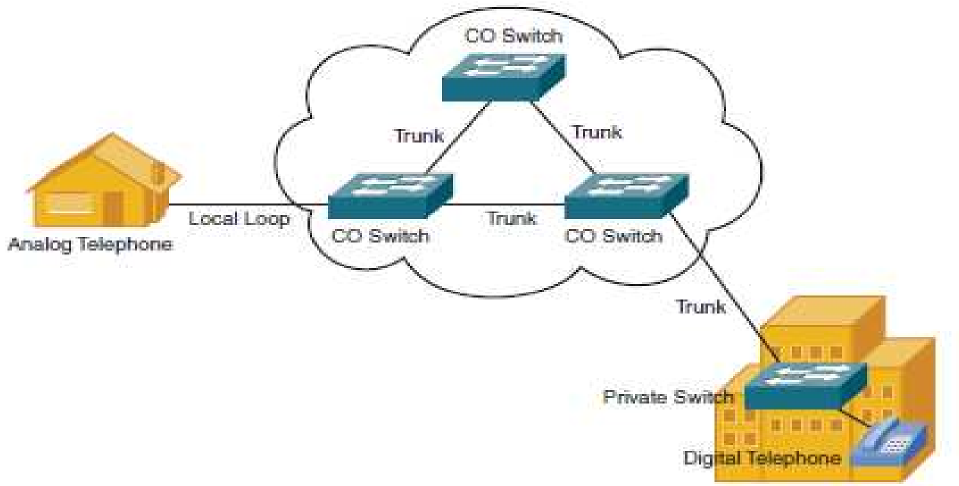

Figure.II.3:PSTN components.

II.5. CONNECTIONS TO AND BETWEEN THE PSTN

When we want to connect to the PSTN, we have a variety of options.
Home users and small offices can connect using analog ports.
Each two-wire analog connection has the capability to support a single call. For home users, a single, analog connection to the PSTN may be sufficient.
For small offices, the number of incoming analog connections directly relates to the office size and average call volume, as businesses grow, we can consolidate the multiple analog connections into one or more digital T1 or E1 connections.

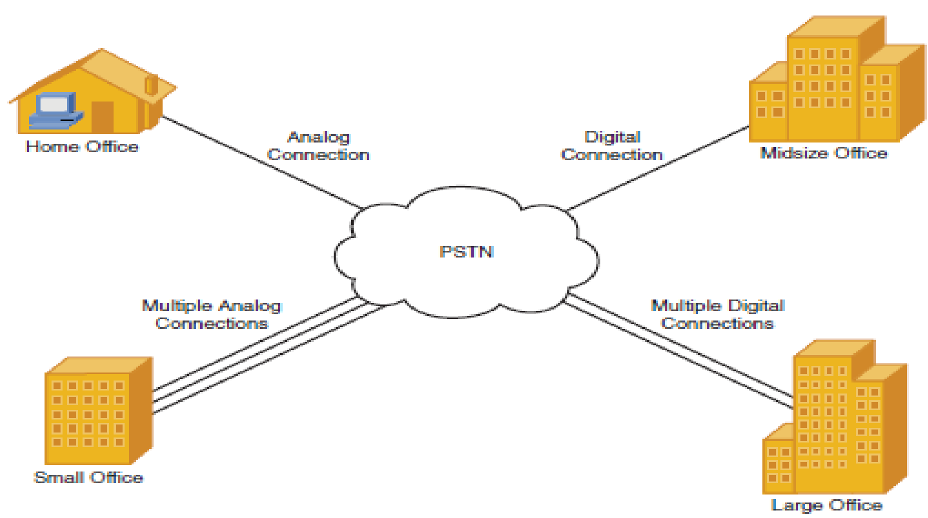

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Figure.II.4:Connections to the PSTN.

In the PSTN lies a network of networks, similar to the internet, which connects offices from multiple telephony providers together into a massive world wide network.

For all the telephony providers of the world to communicate together, a common signaling protocol must be used, similar to the way TCP/IP operates in the data realm.

The voice signaling protocol used around the world is SS7.

SS7 is an out-of-band signaling method used to communicate call setup, routing, billing, and informational messages between telephone company CO around the world. When a user makes a call, the first CO to receive the call performs an SS7 lookup to locate the number, once the destination is found, SS7 is responsible for routing the call through the voice network to the destination and providing all informational signaling (such as ring back) to the calling device.

II.6. ANALOG SYSTEMS

Analog refers to transmission of electronic information achieved by adding signals of varying frequency or amplitude to a carrier wave of a given frequency.

Traditional broadcast media such as radio, television, and PSTN use analog technology, typically represented as a series of varying sine waves.

The term analog can be traced to the similarity between the actual fluctuations of the human voice and the analogous, or comparable, modulation of a carrier wave.

The human voice occupies the 20Hz to 20KHz range, with most energy in the 300–3300Hz range, Viewed graphically both the human voice and a modulated carrier wave display periods of little or no activity followed by periods of activity (15).

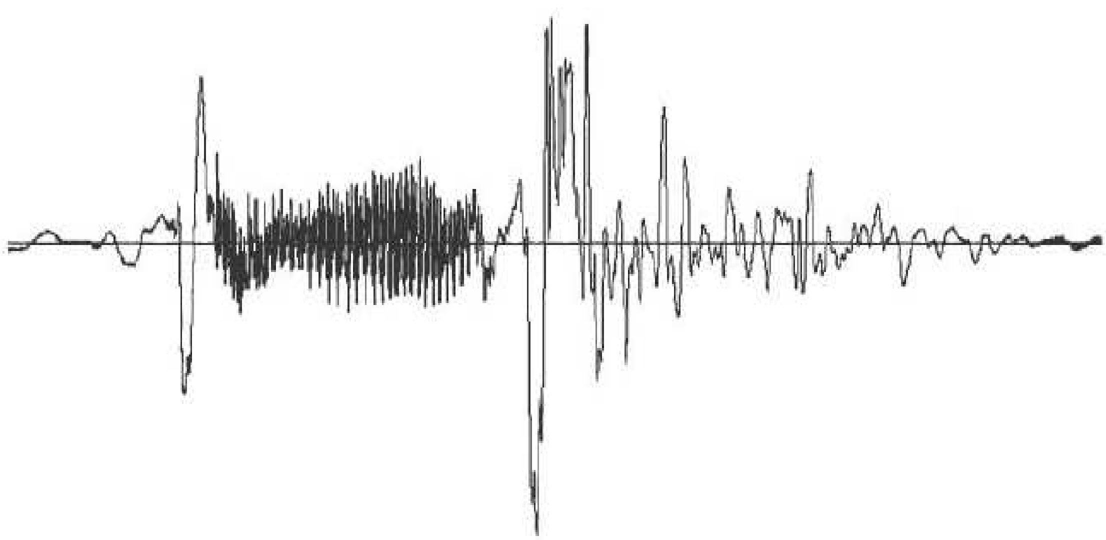

Figure.II.5:Electrical analog waveform of human speech. (16)

II.6.1. Loop Start and Ground Start Signaling

Each analog circuit is composed of a pair of wires, one wire is the ground, or positive side of the connection (this is often called the tip), the other wire is the battery, or negative side of the connection (often called the ring).

These two wires are what power the analog phone and allow it to function, just like the wires that connect our car battery to the car (17).

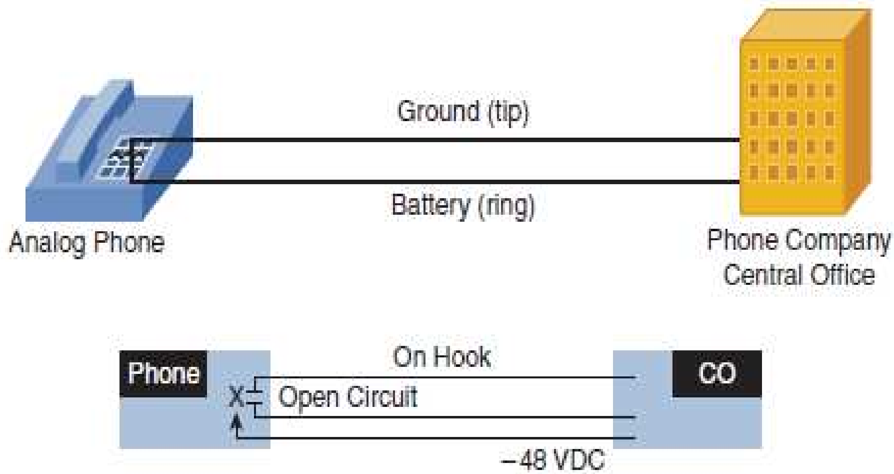

Figure.II.6:Connections of the ground and battery wires to an analog phone.

The jagged line over the wires in the analog phone represents a broken circuit. Anytime the phone is on hook, the phone separates the two wires, preventing electric signal from flowing through the phone.

When the phone is lifted off hook, the phone connects the two wires, causing an electrical signal (48V DC voltage) to flow from the phone company central office (CO) into the phone. This is known as loop start signaling, loop start signaling is the typical signaling type used in home environments, loop start signaling is susceptible to a problem known as glare.

Glare occurs when we pick up the phone to make an outgoing call at the same time as a call comes in on the phone line before the phone has a chance to ring.

In home environments, this is not usually a problem for a couple reasons, first, the chances of having a simultaneous outgoing and incoming call are slim, second, if we do happen to have an incoming call, it's always meant for our house (unless the caller dialed the wrong number).

In business environments, glare can become a significant problem because of the large number of employees and high call volume, for example, a corporation may have a key system (which allows it to run its own, internal phone system) with five analog trunks to the PSTN.

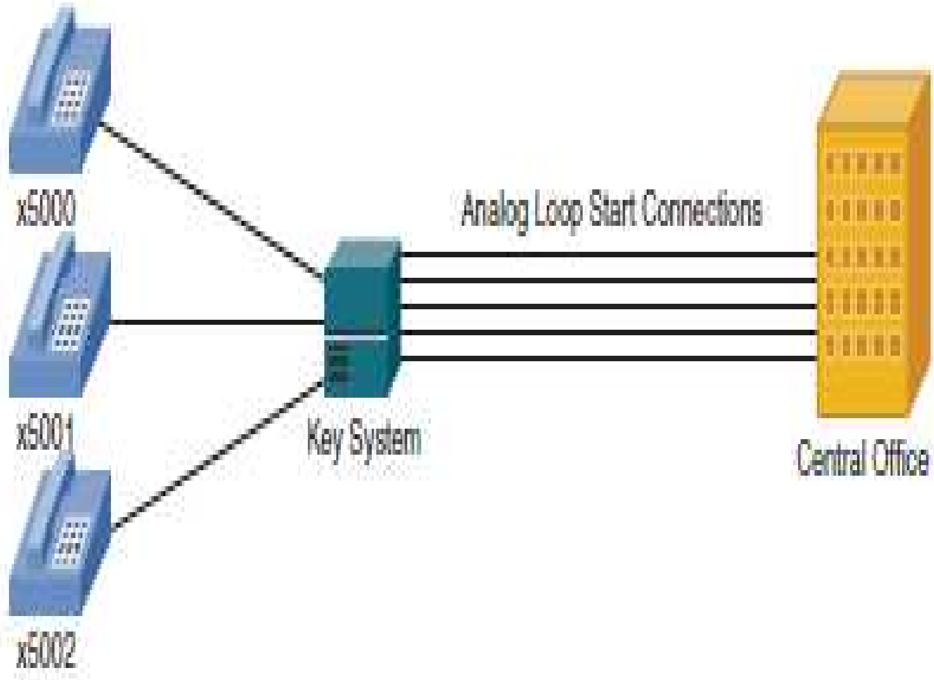

Figure.II.7:Illustration of glare.

If a call comes in for x5002 at the same time as x5000 picks up the phone, the key system will connect the two signals, causing x5000 to receive the call for x5002,this happens because the loop start signal from x5000 seizes the outgoing PSTN line at the same time as the key system receives the incoming call on the same PSTN line, This is an instance of glare. Because of glare, most modern PBX systems designed for larger, corporate environments use ground start signaling, ground start signaling originated from its implementation in pay phone systems.

The problem of glare, to receive a dial tone from the conter office, the PBX must send a ground; signal on the wires, this intentionally signals to the telephone CO that an outgoing call is going to happen, whereas using the loop start method of signaling just connects the wires to receive an incoming call or place an outgoing call (17).

II.6.2. Analog Voice Interfaces

Literally dozens of different analog interfaces FXS, FXO, understand the situations where each interface type is used.

II.6.2.1. Foreign Exchange Station Interface

The Foreign Exchange Station (FXS) is an interface that connects directly to analog endpoint such as an analog phone or fax machine, the connection handoff is a standard RJ - 11 port.

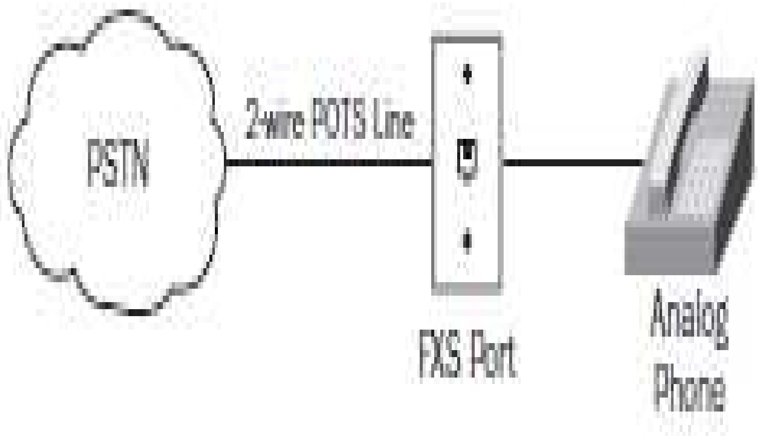

Figure.II.8:An FXS interface.

FXS ports are commonly found in residential homes that require very few analog lines. The interface provides voltage and signaling to analog devices.

II.6.2.2. Foreign Exchange Office Interface

Instead of plugging directly into an analog phone like the FXS port does, a Foreign Exchange Office (FXO) port connects to a PBX.

The FXO interface assumes that all dial tones, ring indicators, and other call progress signaling are provided locally by the equipment attached to it such as a key system or PBX. Contrast that with an FXS connection where the device that plugs into the port does not provide any form of signaling. Instead, it relies on the backend equipment to provide it. Typically, a business will provision a handful of FXO ports from the PSTN and then connect the phones to the PBX.

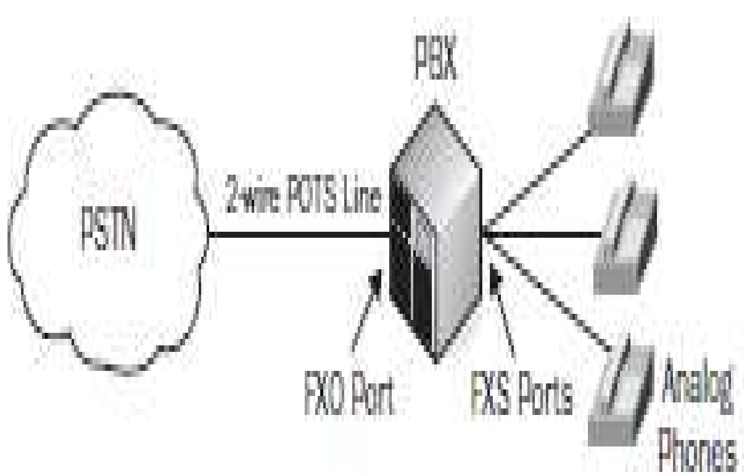

Figure.II.9:An FXO interface.

With this type of setup, there will be only one PSTN number per line, as with the FXS port, the connection handoff is RJ-11.

FXS interfaces connect analog telephones, while FXO interfaces are used to connect to the PSTN, the first hop out to the PSTN cloud is called the central office.

In this scenario, either the single number would ring all phones on the PBX when dialed, or it would ring a single phone handled by a live operator.

II.6.3. Setup and Release of a Call :

Each telephone has a switch that indicates an on or off hook condition.

- When the hook is raised, the switch is closed and an approximately 50 mA of current starts flowing, this is detected by a relay giving information to the control unit in the exchange, the control unit is an efficient and reliable computer in the telephone exchange, it activates signaling circuits.
- which then receive dialed digits from subscriber A,the control unit in the telephone exchange controls the switching matrix that connects the speech circuit through to the called subscriber B.
- Connection is made according to the numbers dialed by subscriber A,when the call is being routed to subscriber B, the telephone exchange supplies to the subscriber loop a ringing voltage and the bell of subscriber B telephone starts ringing.

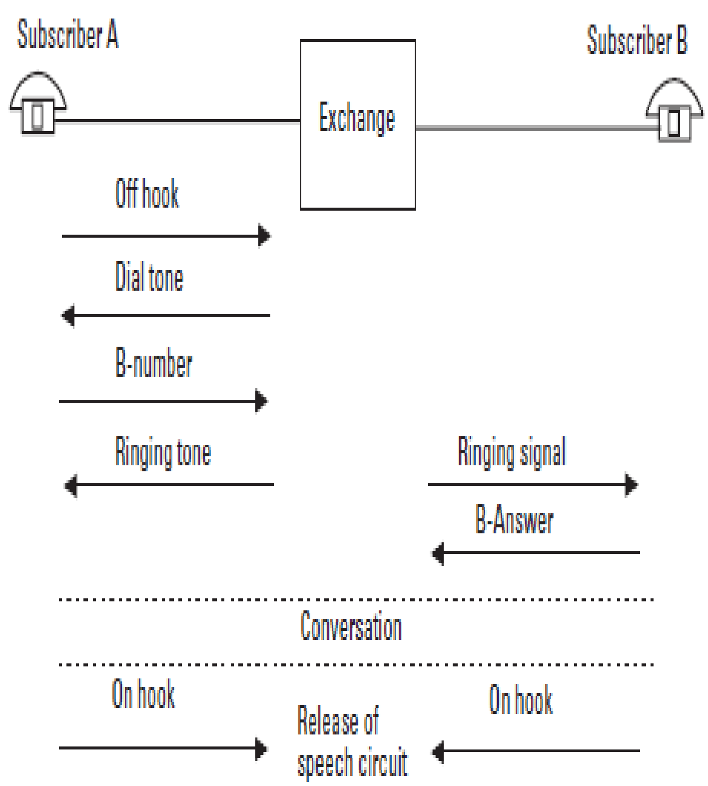

Figure.II.10:Subscriber signaling.

II.6.4. Analog Network Event Signaling :

All analog phone calls need a way to signal events on the phone network in order to establish communication between end devices, there are three distinct types of network signaling signaling in a voice network :

- Address signaling.
- Supervisory signaling.
- Informational signaling.

II.6.4.1 Address Signaling :

Once the phone company has used informational signaling to generate a dial tone signal,the user can dial digits. there are two types of address signaling in use worldwide : Once the phone company has used informational signaling to generate a dial tone signal, the user can dial digits.

1. **Dual-tone multifrequency (DTMF):**

The buttons on a telephone keypad use a pair of high and low electrical frequencies (dual-ton) to generate a signal each time a caller presses a digit (17).

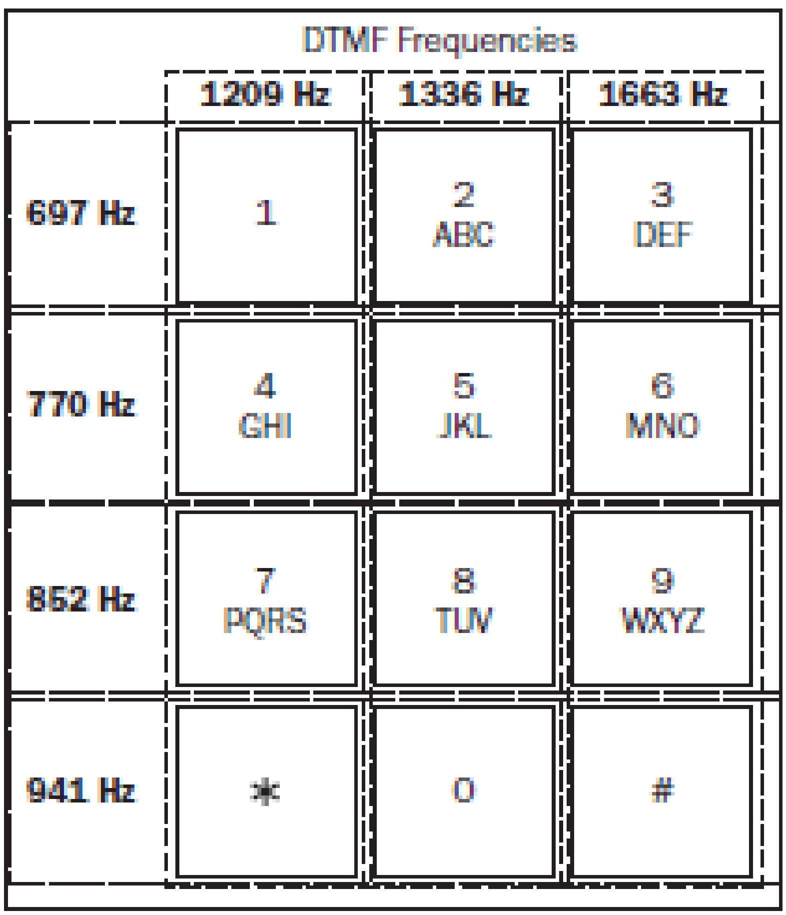

Figure.II.11:DTMF frequency creation.

Figure.II.12:Modern telephones use push buttons.

2. **Pulse:**

The rotary-dial wheel of a phone connects and disconnects the local loop circuit as it rotates around to signal specific digits.

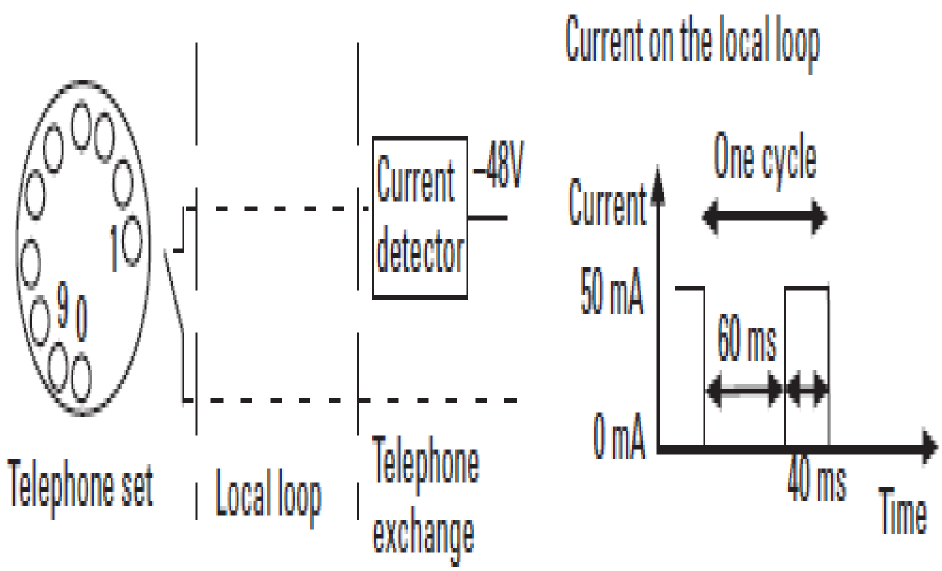

Figure.II. 13 : Rotary, or pulse, dialing.

Figure.II.14:Rotary dial telephone.

II.6.4.2. Supervisory Signaling

Supervisory signaling deals with the behind the scenes part of call setup and teardown. There are many different types of supervisory signaling, depending on the types of circuits being used and the type of phone equipment making the signals.

This signaling is done to ensure that the phone system properly interprets user input and that user input is properly handled.

When we pick up our phone, a seizure supervisory signal is first sent to the telephone switch to ensure that we have control over the analog circuit.

As soon as the line is seized, we receive informational signal in the form of a dial tone (17).

Supervisory Signal Type	Signal Meaning
Seizure	Signals the phone system to change the line/trunk state from idle to active.
Wink/hook flash	Indicates that the phone system is ready to receive address information in the form of DTMF or pulse digits.
Answer	Indicates when the remote-side phone is answered and two-way communication is established.
Disconnect	Indicates that either phone in the two-way communication goes on-hook. The call is torn down and the circuit returned to an idle state.
Robbed-bit	In-band bits are used to signal the start and end of address information.

Table.II.1:Supervisory signals.

II.6.4.3. Informational Signaling

Informational signaling is all about letting the calling party know what is going on with the phone system and the attempted call,as soon as we pick up the receiver of a phone, we hear a dial tone, this tone informs we that the phone is operational and talking to the phone switch, people commonly listen to make sure they hear the dial tone before dialing a number (17).

Informational Signal Type	Signal Meaning
Dial tone	Phone is in an off-hook state and ready to accept user input with the keypad.
Busy	Called number phone is currently in use.
Number not in service	Called number is not available on the phone network.
Call waiting	An incoming call is being made to line 2 on the phone; line 1 is in use.
Ring-back	The phone company is attempting to establish the connection to the called party.
Reorder	All local circuits are busy; thus the call cannot be completed.This is also known as a "fast busy" signal.
Congestion	The long-distance company is unable to complete the call.
Handset off-hook	Someone has picked up the handset of a phone from the cradle.

Table.II.2:Informational signals.

II.7. DIGITAL CONNECTION

Analog signaling was a massive improvement over tin cans and string, but still posed plenty of problems of its own, an analog electrical signal experiences degradation (signal fading) over long distances, to increase the distance the analog signal could travel, the phone company had to install repeaters to regenerate the signal as it became weak.

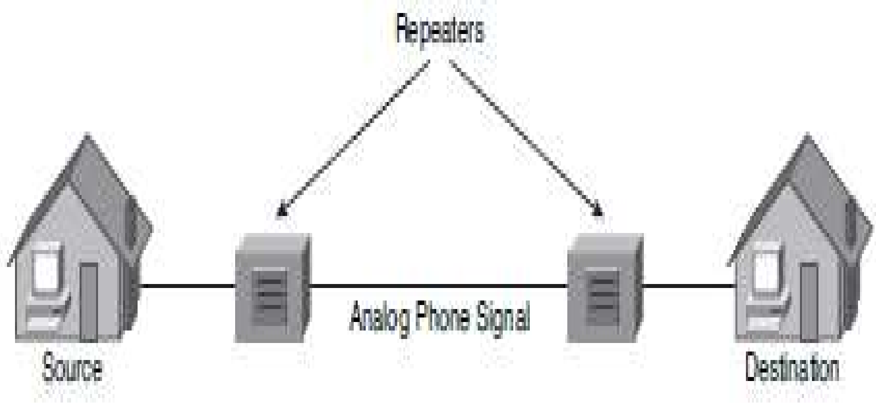

Figure.II.15:Analog signal repeaters.

Unfortunately, as the analog signal was regenerated, the repeater device was unable to differentiate between the voice traveling over the wire and line noise, each time the repeater regenerated the voice, it amplified the line noise as well.

The second difficulty encountered with analog connections was the sheer number of wires the phone company had to run to support a large geographical area or a business with a large number of phones, because each phone required two wires, the bundles of wire became massive and difficult to maintain (imagine the hassle of a single pair of wires in the bundle breaking), a solution to send multiple calls over a single wire was digital connection is that solution (17).

II.7.1. Converting Analog to Digital Signals

Simply put, digital signals use numbers to represent levels of voice instead of using a combination of electrical signals.

When someone talks about digitizing voice, they are speaking about the process of changing analog voice signals into a series of numbers which we can use to put the voice back together at the other end of the line.

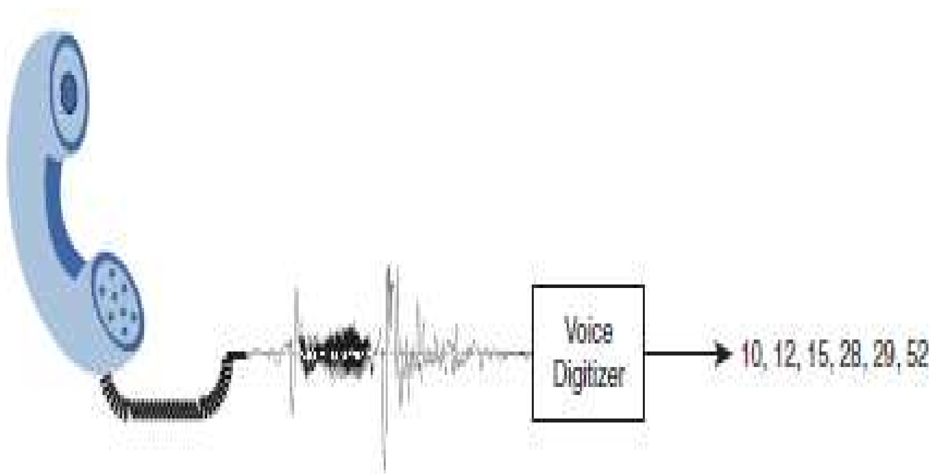

Figure.II.16:Converting Analog to Digital Signals.

To convert an analog signal into digital format, the converting device goes through a four-step process:

1. Sample the signal.

2. Quantize the signal.

3. Encode the quantized value into binary format.

4. Optionally compress the sample to save bandwidth (6)

II.7. 2. Sending Multiple Calls over a Single Line

The original problems of analog connections:

- The signal degrades over long distances.
- We can't send multiple calls over a single line (resulting in massive cabling requirements).

Digitizing voice solves the first problem because we can easily transmit a numeric value any distance a cable can run without any degradation or line noise, Time-division multiplexing (TDM) solves the second problem.

TDM allows voice networks to carry multiple conversations at the same time over a single, four-wire path, because the multiple conversations have been digitized, the numeric values are transmitted in specific time slots that differentiate the separate conversations.

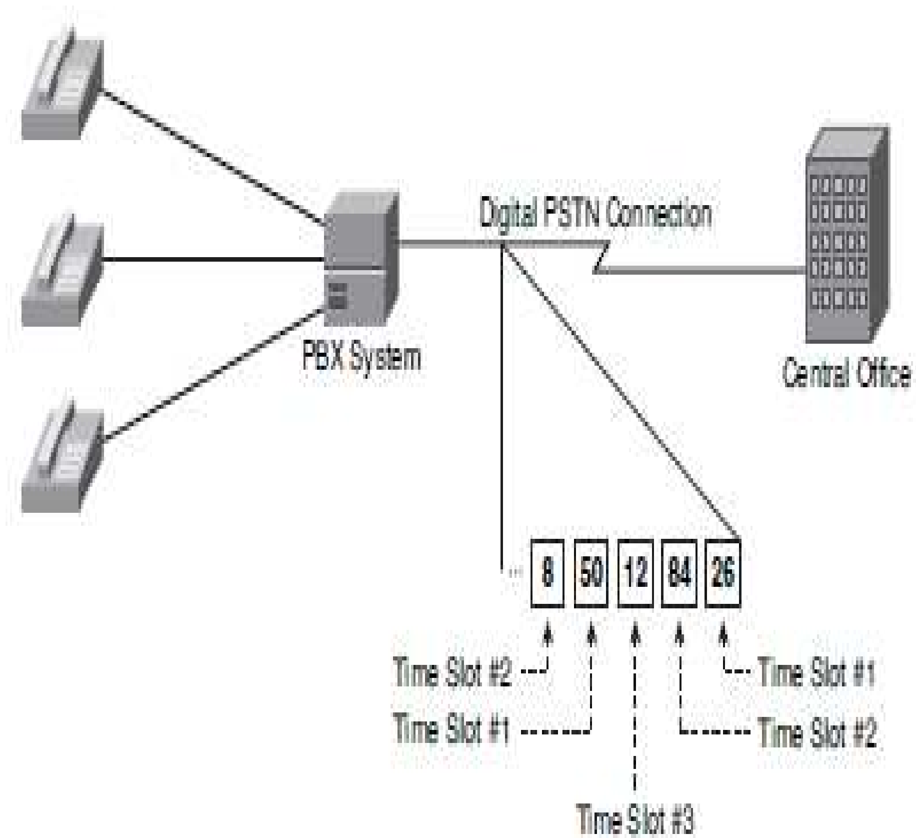

Figure.II. 17:Time-Division Multiplexing Voice Channels.

II.7. 3. Digital Voice Interfaces

Analog circuits are fine if we require only a few PSTN lines into our business, if we need approximately 10 or more external lines, it is typically more cost effective to look into a digital trunk circuit such as a T1 or E1, from a physical point of view, T1 and E1 circuits are typically terminated at the customer site in the form of copper wiring, usually category 5 cabling, this same cabling is used for Ethernet LAN connections.

The circuits are terminated using a standard RJ - 45 connector, looking at the eight pinouts on the RJ - 45, we can see that a T1 uses pins 1 and 2 for transmit and 4 and 5 for receive (17).

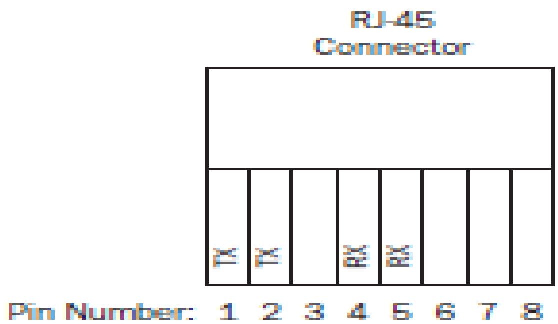

Figure.II.18:T1 and E1 RJ-45 pinouts.

II.8. PSTN NUMBERING PLANS

When connecting to the PSTN, we must use a valid, E.164 standard address for our telephone system. E.164 is an international numbering plan created by the International Telecommunication Union (ITU).

II.8.1. The International Numbering Plan

The International Numbering Plan is commonly known as the International Telecommunications Union (ITU) E.164 standard,a globally recognized organization, the ITU is responsible for creating inter border communications standards.The E.164 standard defines the format of PSTN numbers on a global scale, the structure consists of three distinct categories.

Structure	Format	Description
Country code (CC)	1–3 digits	Defines the country of origin
National destination code (NDC)	0–15 digits	Optional country/region-specific code
Subscriber code (SC)	1–15 digits	Central office significant code

Table.II.3:ITU E.164 structure.

Within a given country code, the national destination and subscriber codes are primarily governed by the local country or region and can be in any format.
The only caveat is that the ITU E.164 numbering plan stipulates that the maximum number of digits for an international call must be less than or equal to 15 and must use the assigned country code at the beginning of the dial string.

Country or Region	E.164 Country Code
North America	1
Mexico	52
United Kingdom	44
France	33
Germany	49
India	91
Hong Kong	852
Spain	53

Table.II.4:ITU country code sampling.

II.9. UNDERSTANDING PBX AND KEY SYSTEMS

Many businesses have hundreds or even thousands of phones they support in the organization. If the company purchases a direct PSTN connection for each one of these phones, the cost would be astronomical. Instead, most organizations choose to use a PBX or key system internally to manage in-house phones.
These systems allow internal users to make phone calls inside the office without using any PSTN resources. Calls to the PSTN forward out the company's PSTN trunk link.
When we first look at a PBX system, it looks like a large box full of cards, each card has a specific function:

- Line cards : Provide the connection between telephone handsets and the PBX system.
- Trunk cards : Provide connections from the PBX system to the PSTN or other PBX systems.
- Control complex : Provides the intelligence behind the PBX system, all call setup, routing, and management functions are contained in the control complex.

If we look at a PBX from a network equipment mindset, single point of failure might be one of the first thoughts that jump into our mind.

Key systems are geared around small business environments (typically less than 50 users). As technology has advanced, the line between key systems and PBXs has begun to blur. However, key systems typically support fewer features and have a shared line feel.

II.10. SUMMARY

The telephone changed the lives of humans all around the world, in the beginning the telephone was heavy, large, and too much too afford.

As time went on the telephone got smaller, lighter, and able to afford. Later on, cell phones were invented which made communication a lot easier and better than the telephone invention because it was mobile, chargeable, and smaller.

In the next chapter, we introduce to Voice over Internet Protocol , covers voice gateway purpose and components, and voice gateway and endpoint communication protocols signaling and transporting.

Chapter III

III.1. INTRODUCTION

Voice over Internet Protocol (VoIP) is going to do is start to weaken the foundation of the way we've done things for 100 years, congress already should be discussing the next telecom bill, Since the introduction of the VocalTec's VocalChat PC-to-PC phone in March of 1995, many articles in the trade press frequently claimed that, in the near future, telephone traffic would be just another application running over the Internetn, such statements gloss over many engineering, regulatory and economic details that preclude voice from being just another Internet application.

The appearance of VoIP comes at a juncture when telecommunications system has already turned into a large-scale, complex system with multiple, competing infrastructures. VoIP,however, greatly augments the nested complexity by affording a technology that enables multiple architectures and business models for delivering the same voice (and often converged voice and data) service, while remaining agnostic to the underlying infrastructure. The VoIP-enabled architectures have very different capabilities and costs from one another. Many do not or cannot support social regulations such as emergency, wiretapping and disability access, most exploit the economic arbitrage opportunities by evading access charges and universal service contributions, added to this is the combination of reduced asset specificity due to VoIP layered architecture and a global standard based ubiquitous IP technology that frees the service providers of the need to own the delivery infrastructure, and enables them to offer service from anywhere globally, such a misalignment between regulatory obligations and technical capabilities, has the potential to incubate large-scale systemic failures due to lack of coordination between the local optimization focused private markets and the highly compartmentalized public institutions.

In this chapter gives a basic overview about VoIP technology and the related terms and shows the processes that go behind the scene while communicating via VoIP.

III.2. VOIP DEFINITION

Stands for Voice Over Internet Protocol, and is often pronounced voip .
VoIP is basically a telephone connection over the Internet.
The data is sent digitally, using the Internet Protocol (IP) instead of analog telephone lines, this allows people to talk to one another long distance and around the world without having to pay long distance or international phone charges (18).

III.3. COMPONEMENTS OF A VOIP NETWORK

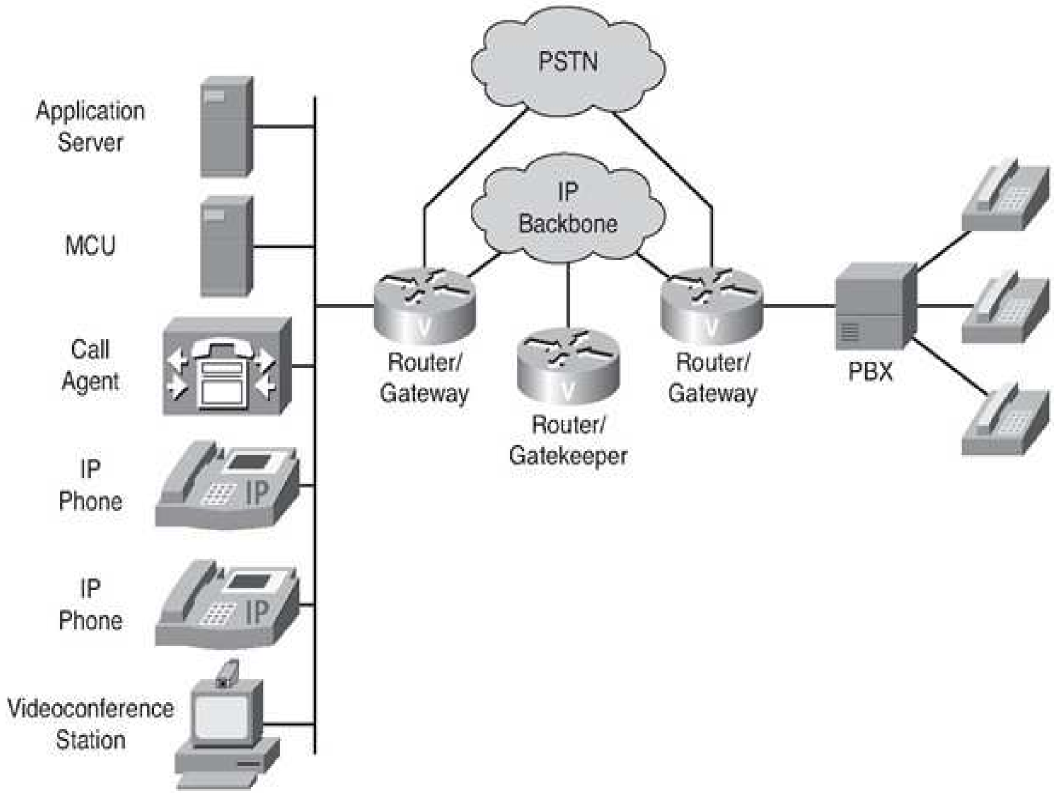

Figure.III.1:Components of a packet voice network.

The following is a description of these basic components:

- IP Phones: IP Phones provide IP endpoints for voice communication.
- Gatekeeper: A gatekeeper provides Call Admission Control (CAC), bandwidth control and management, and address translation.
- Gateway: The gateway provides translation between VoIP and non VoIP networks, such as the PSTN, gateways also provide physical access for local analog and digital voice devices, such as telephones, fax machines, key sets, and private branch exchanges (PBX).
- Multipoint Control Unit (MCU): An MCU provides real time connectivity for participants in multiple locations to attend the same videoconference or meeting.
- Call agent: A call agent provides call control for IP phones, bandwidth control and management, and address translation.
- Application servers: Application servers provide services such as voice mail, unified messaging, and Cisco Communications Manager Attendant Console.
- Video conference station: A videoconference station provides access for end user participation in video conferencing, the video conference station contains a video

capture device for video input and a microphone for audio input, a user can view video streams and hear audio that originates at a remote user station.

- Other components: such as software voice applications, interactive voice response (IVR) systems, and soft phones, provide additional services to meet the needs of an enterprise site (19).

III.4. ADVANTAGES AND DISADVANTAGE OF VOIP

VoIP has become a force to reckon with in the telecom sector with many people and businesses saving considerable time and resources by opting for a relatively inexpensive voice over internet protocol, VoIP has many advantages and disadvantages.

III.4.1. Advantages of VOIP

There are many benefits to using VoIP for business. For example:

- We can integrate it with an existing phone connection.
- With VoIP PC-to-PC, calls are free no matter the distance and PC to Phone charges are nominal.
- For a monthly fee we may make unlimited free calls within a geographic area.
- A virtual number enables we to make calls from anywhere as long as a broadband connection is available.
- We may purchase a number in a geography area of we choice, which works out very cheap.
- We may access our VoIP account just like our email Id from anywhere in the world as long as we have an internet phone, This makes it easy for those who travel frequently to make calls frequently to those back at home at local call rates, no matter where they are.
- We may call or message or do both at the same time with VoIP services.
- VOIP cost about half the cost of traditional phone services and it seems that the taxes and surcharges are much lower, also our bill is easier to understand and it can be viewed via the Internet.

III.4.2. Disadvantages of VOIP

Every product has its own demerits, So VOIP also has it,they are

- Loss of service during outages.
- Without power VoIP phones are useless, so in case of emergencies during power cuts it can be a major disadvantage.
- With VoIP emergency calls, it is hard to locate we and send help in time.

- Some times during calls, there may be periods of silence when data is lost while it is being unscrambled.
- Latency and traffic.
- No standard protocol is applicable (20).

III.5. H.323 PROTOCOL

III.5.1. Definition

Is a standard that specifies the components, protocols and procedures that provide multimedia communication services real time audio, video, and data communications over packet networks, including Internet protocol (IP) based networks.

H.323 is part of a family of ITU–T recommendations called H.32x that provides multimedia communication services over a variety of networks (18).

III.5.2. What Is H.323 ?

The H.323 standard is a cornerstone technology for the transmission of real time audio, video, and data communications over packet-based networks.

It specifies the components, protocols, and procedures providing multimedia communication over packet-based networks.

Packet based networks include IP based (including the Internet) or Internet packet exchange (IPX) based local-area networks (LANs), enterprise networks (ENs), metropolitan-area networks (MANs), and wide-area networks (WANs).

H.323 can be applied in a variety of mechanisms audio only (IP telephony), audio and video (videotelephony), audio and data, and audio, video and data.

H.323 can also be applied to multipoint-multimedia communications.

H.323 provides myriad services and therefore, can be applied in a wide variety of areas consumer, business, and entertainment applications (21).

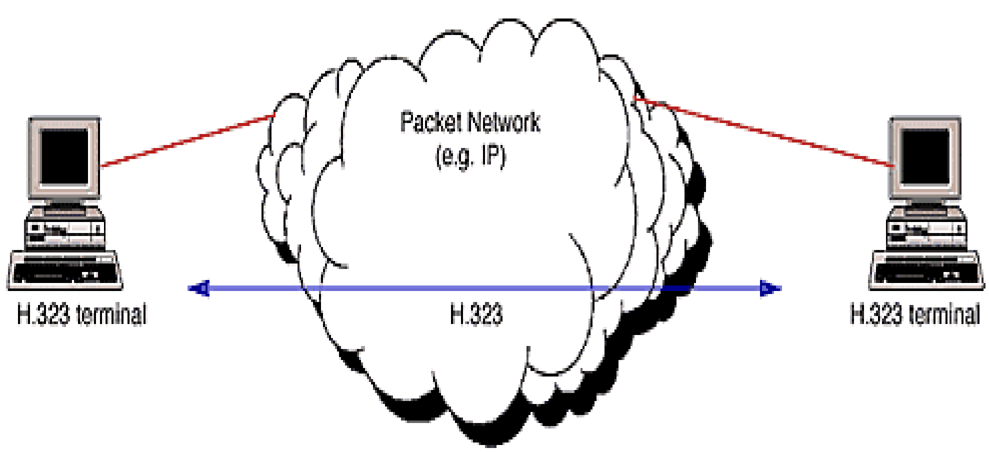

Figure.III.2:H.323 Terminals on a Packet Network.

III.5.3. Components of H.323

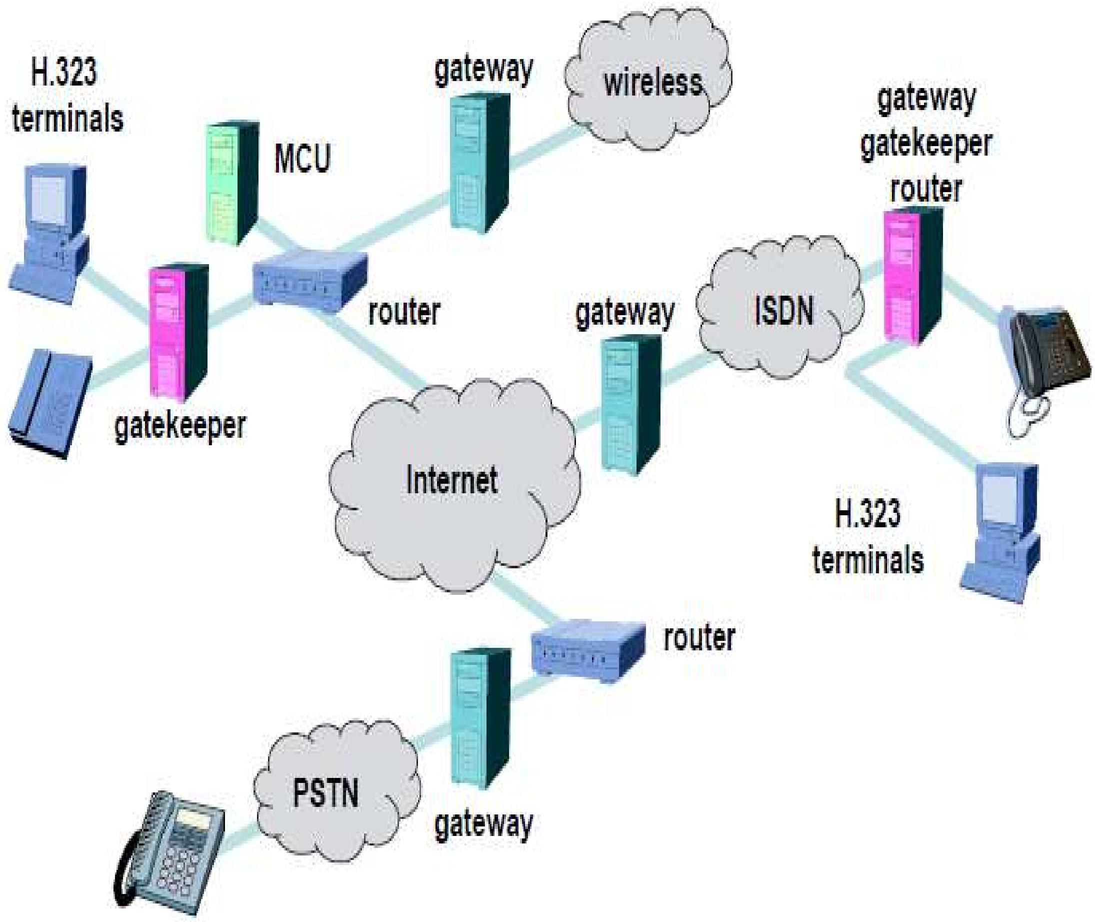

Figure.III.3:Components of H.323.

- Terminals : LAN based communication end-points
- Gateway (media and/or signaling) : Interface between packet- and circuit-switched networks.
  - Media gateway : voice transcoding, protocol conversion.
  - Media gateway controller : call handling, call state.
  - Signaling gateway : signaling mediation.
- Gatekeeper : Admission control, SNMP services, address translation.
- MCU (Multipoint Control Unit) : Handling of broadcasts / conference calls (22).

III.5.4. H.323 Protocol Stack

The audio, video and registration packets use the unreliable User Datagram Protocol (UDP) while the data and control application packets use the reliable Transmission Control Protocol (TCP) as the transport protocol.

Except for the T.120 protocol, the other protocols are described in the paper (23).

Data	Control and Signaling		Audio/Video	Registration
T.120	H.225.0 Call Signaling	H.245 Conference Control	RTP/RTCP	H.225.0 RAS
TCP			UDP	
Network Layer				
Data link Layer				
Physical Layer				

Table.III.1:The protocol stack of H.323.

III.5.5. H.323: Example of communication

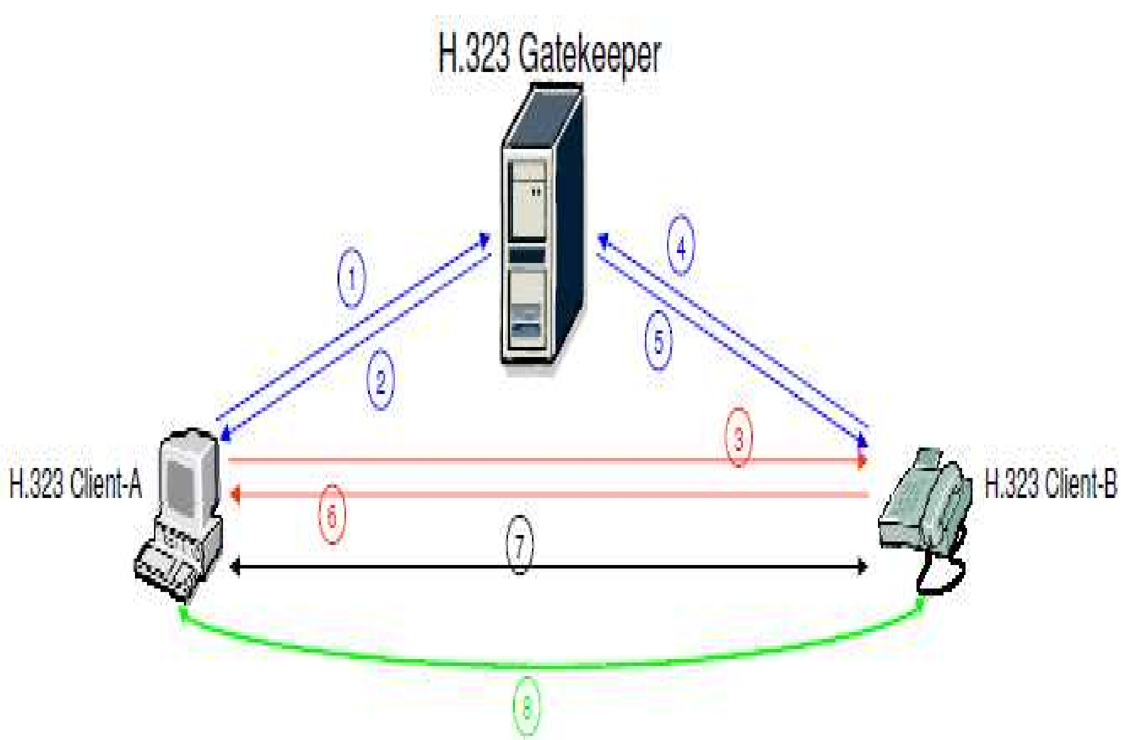

Figure.III.4:Communication H.323.

1. Client-A sends ARQ (Admission ReQuest) to Gatekeeper : it asks to be connected to Client-B alias.
2. Gatekeeper confirms or rejects the call (ACF, Admission ConFirm / ARJ, Admission ReJect) : it returns the Client-B call signaling address (IP address and port of the call signaling address).
3. Client-A sends a SETUP to Client-B call signaling Address.
4. Client-A sends ARQ (Admission ReQuest) to Gatekeeper : it asks to be connected to Client-A (he knows already Alias and call signaling address).
5. Gatekeeper confirms or rejects the call (ACF, Admission ConFirm / ARJ, Admission ReJect) : it returns the Client-A call signaling address (IP address and port) if it was asked.
6. Client-B sends a CONNECT to Client-A call signaling address.
7. Capabilities about the call are exchanged among the terminals.
8. The media is sent end-to-end,the IP addresses and ports negotiated during the signaling (24).

III.6. SESSION INITIATION PROTOCOL (SIP)

III.6.1. Definition

SIP is a simple signaling protocol used for Internet conferencing and telephony.
SIP is fully defined in RFC 2543, based on the Simple Mail Transport Protocol (SMTP) and the Hypertext Transfer Protocol (HTTP), SIP was developed within the IETF Multiparty Multimedia Session Control (MMUSIC) working group.
SIP specifies procedures for telephony and multimedia conferencing over the Internet.
SIP is an application-layer protocol independent of the underlying packet protocol (TCP, UDP,ATM, X.25).
SIP is based on a client/server architecture in which the client initiates the calls and the servers answer the calls, by conforming to these existing text based Internet standards (SMTP and HTTP), troubleshooting and network debugging are facilitated .
SIP is widely supported and is not dependent on a single vendors equipment or implementation.
SIP is a newer protocol than H.323 and does not have maturity and industry support at this time.
However, because of its simplicity, scalability, modularity, and ease with which it integrates with other applications, this protocol is attractive for use in packetized voice architectures.
Some of the key features that SIP offers are :

- Address resolution, name mapping, and call redirection.
- Dynamic discovery of endpoint media capabilities by use of the Session Description Protocol (SDP).
- Dynamic discovery of endpoint availability.
- Session origination and management between host and endpoints (25).

III.6.2. Basic SIP Components

When we are ready to enhance our enterprise communications with SIP, understand the basic building blocks that form the foundation of our new SIP enabled enterprise.

III.6.2. 1. User agents

User agents (UAs) are applications installed on SIP endpoints,such as an IP phone, mobile phone, or a laptop or desktop PC, that interface between the user and the SIP network. A UA can act as either a client or a server, when sending SIP requests, the UA acts as a user agent client (UAC), and when servicing a request, it acts as a user agent server (UAS).
A back-to-back user agent (B2BUA) is an application that acts as an intermediary between two parties,but appears as an endpoint to both parties like a middleman, it serves as both UAS and UAC simultaneously to process session requests.

Figure.III.5:Some typical SIP user agents.

SIP devices can communicate directly if they know each others URI (Uniform Resource Identifier) or IP address, but more commonly.

SIP servers are used in an enterprise network to provide an infrastructure for routing, registration, and authentication and authorization services.

IP based devices can identify and communicate with one another using IP addressing alone. However, in most cases, our network uses the Domain Name System (DNS) to establish sessions with device names, which DNS translates into IP addresses.

Similarly,SIP devices frequently consult directory servers (often by name), which provide endpoint addresses that the devices then contact to set up a call.

III.6.2. 2. SIP servers

SIP servers provide centralized information and enablement services in a SIP ecosystem.

The core SIP servers and an overview of their basic functions are described here :

- Registrar Server
- Location Service
- Redirect Server
- Proxy Server
- Presence Server (26)

III.6.3. A Basic SIP Call Example

The communication then follows these steps:

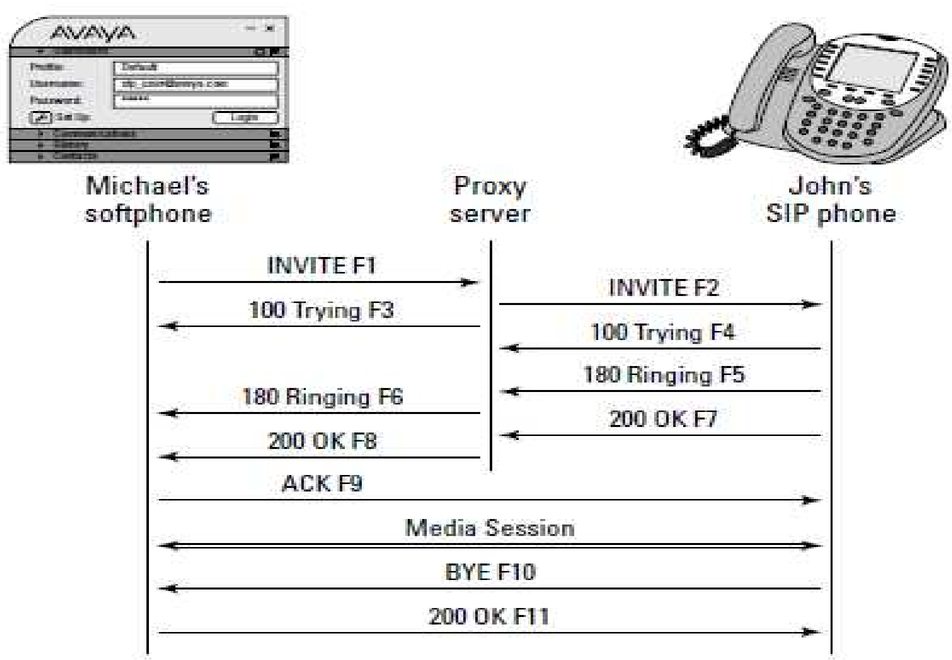

Figure.III.6:A typical SIP sessions ladder diagram.

1. Michelle@smallcompany.com (the UAC) initiates a session by inviting Tony@bigcompany.com and sending this request to the proxy server at small company.com. John's
   Michelle's UA generates an INVITE request, which is sent to the proxy at smallcompany.com, INVITE message contains Session Description Protocol(SDP) parameters that define the types of media she is capable of accepting and where she wants the media to be sent.
2. The proxy at smallcompany.com performs a DNS SRV record lookup for SIP services at bigcompany.com since bigcompany.com is a foreign domain. This record lookup returns proxy.bigcompany.com, which is then resolved to a physical IP address by DNS.
   Michelle's INVITE request is then forwarded to the proxy server at bigcompany.com.
3. The bigcompany.com proxy server receives and processes the invitation, and looks up Tony's contact in the location database of the Registrar (physical IP address of the UA).
4. The location database of the Registrar returns host@ 192.168.1.100 where Tony is currently located.
5. The bigcompany.com proxy server forwards the INVITE request to Tony UA at host@192.168.1.100.
6. The UAS at host@192.168.1.100 asks Tony whether he wants to accept the call, Tony's may hear a ring, see a text message, or see a blinking LED.
7. Tony's acceptance is sent back through the big company.com proxy, which forwards it to the small company.com proxy, which forwards it to Michelle's UA. The body of Tony's acceptance includes SDP parameters defining the selected media chosen from what Michelle had originally offered and where Tony wants the media to be sent.
8. Michelle's UA responds to the acceptance with an ACK (acknowledgement) directly to Tony's UA, which tells Tony's UA that Michelle is ready to start the call.
9. At the end of the conversation, Tony hangs up his phone. His UAC sends a BYE message directly to Michelle's UA.
10. Michelle's UAC responds with a 200-OK message directly to Tony's UA, which ends the session. Though this call flow describes the initiation of a basic phone call, that simple call flow would be the same for establishing video conferencing or other media sessions using SIP (26).

III.6.4. Key Benefits of Session Initiation Protocol :

- Simplicity SIP is a very simple protocol, software development time is very short compared with that of traditional telephony products.
- Extensibility SIP has learned from HTTP and SMTP and has built a rich set of extensibility and compatibility functions.
- Modularity SIP was designed to be highly modular.A key feature is its independent use of protocols. For example, SIP issues invitations to called parties.
- Integration SIP has the capability to integrate with the Web, e-mail,streaming media applications, and other protocols.
- Interoperability Because it is an open, RFC-based standard, SIP can offer interoperability between different vendors platforms seamlessly (25).

III.7. ROLE OF THE TRANSPORT LAYER

The transport layer is responsible for establishing a temporary communication session between two applications and delivering data between them.

An application generates data that is sent from an application on a source host to an application on a destination host, without regard to the destination host type, the type of media over which the data must travel, the path taken by the data, the congestion on a link, or the size of the network.

The transport layer is the link between the application layer and the lower layers that are responsible for network transmission.

The transport layer provides a method of delivering data across the network in a way that ensures the data can be properly put back together on the receiving end.

The transport layer provides for the segmentation of data, and the controls necessary to reassemble these segments into the various communication streams.

In TCP/IP, these segmentation and reassembly processes can be achieved using two very different transport layer protocols: Transmission Control Protocol (TCP) and User Datagram Protocol (UDP).

The primary responsibilities of transport layer protocols are:

- Tracking the individual communication between applications on the source and destination hosts.
- Segmenting data for manageability and reassembling segmented data into streams of application data at the destination.
- Identifying the proper application for each communication stream.

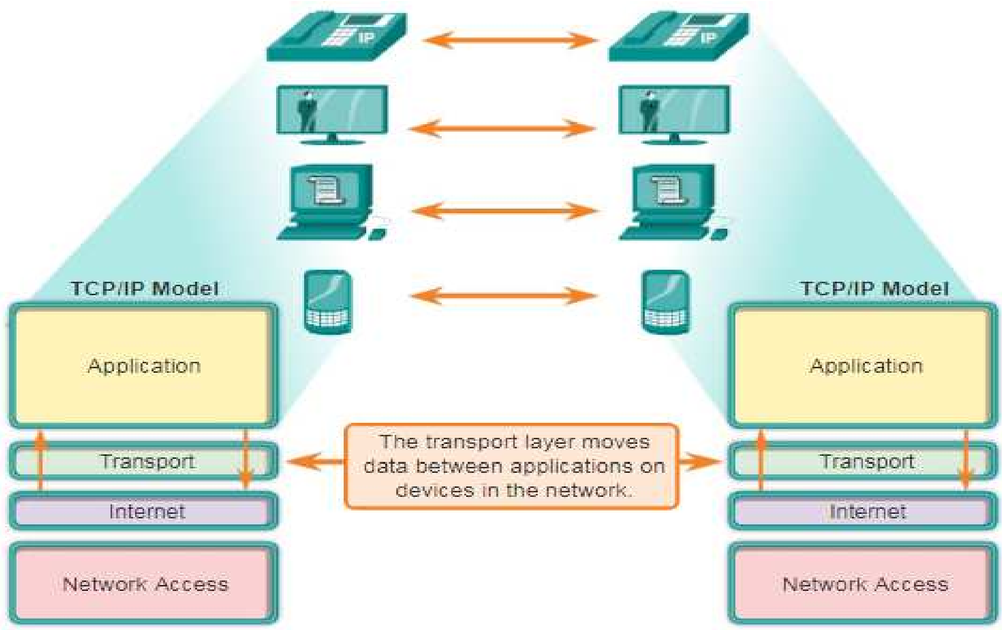

Figure.III.7:Enabling applications on devices to communicte.

III.8. Transmission Control Protocol (TCP)

III.8.1. Definition

TCP was initially described in RFC 793, in addition to supporting the basic functions of data segmentation and reassembly :

- Connection-oriented conversations by establishing sessions.
- Reliable delivery.
- Ordered data reconstruction.
- Flow control.

Figure.III.8:TCP services.

III.8.2. Role of TCP

Once TCP establishes a session, it is then able to keep track of the conversation within that session. Because of the ability of TCP to track actual conversations, it is considered a stateful protocol.

A stateful protocol is a protocol that keeps track of the state of the communication session. TCP tracks which information it has sent and which information has been acknowledged, if the data is not acknowledged, the sender assumes the data did not arrive and resends it, the stateful session begins with the session establishment and ends when the session is closed with session termination.

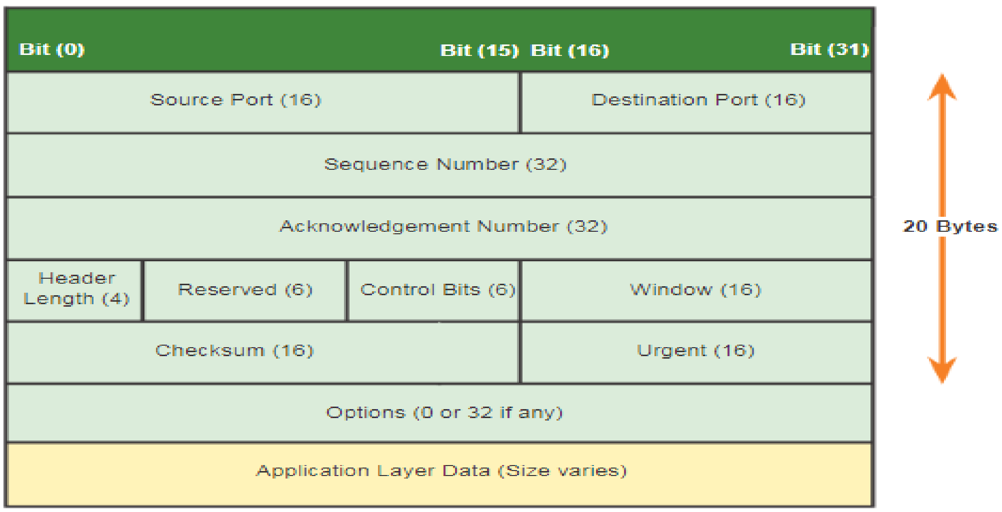

Table.III.2:TCP segment.

TCP incurs additional overhead to gain these functions each TCP segment has 20 bytes of overhead in the header encapsulating the application layer data. This is considerably more than a UDP segment, which only has 8 bytes of overhead, extra overhead includes:

- Sequence number (32 bits) : Used for data reassembly purposes.
- Acknowledgement number (32 bits) : Indicates the data that has been received.
- Header length (4 bits) : Known as data offset, indicates the length of the TCP segment header.
- Reserved (6 bits) : This field is reserved for the future.
- Control bits (6 bits) : Includes bit codes, or flags, that indicate the purpose and function of the TCP segment.
- Window size (16 bits) : Indicates the number of segments that can be accepted at one time.
- Checksum (16 bits) : Used for error checking of the segment header and data.
- Urgent (16 bits) : Indicates if data is urgent.

III.9. USER DATAGRAM PROTOCOL (UDP)

III.9.1. Definition

UDP is considered a best-effort transport protocol, described in RFC 768.
UDP is a lightweight transport protocol that offers the same data segmentation and reassembly as TCP, but without TCP reliability and flow control. UDP is such a simple protocol, that it is usually described in terms of what it does not do compared to TCP.

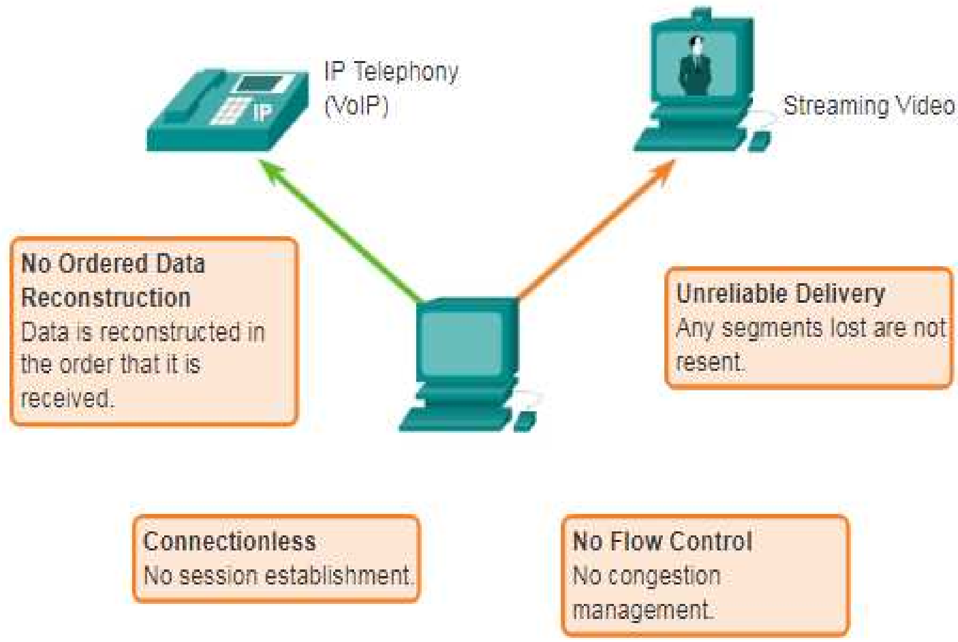

Figure.III.9:UDP.

III.9. 2. Role of UDP

Although UDP does not include the reliability and flow control mechanisms of TCP, as shown in the figure, UDP's low overhead data delivery makes it an ideal transport protocol for applications that can tolerate some data loss.
The pieces of communication in UDP are called datagrams. These datagrams are sent as best effort by the transport layer protocol, a few applications that use UDP are Domain Name System (DNS), video streaming, and Voice over IP (VoIP).
One of the most important requirements for delivering live video and voice over the network is that the data continues to flow quickly, video and voice applications can tolerate some data loss with minimal or no noticeable effect, and are perfectly suited to UDP.
UDP is a stateless protocol, meaning neither the client, nor the server, is obligated to keep track of the state of the communication session,UDP is not concerned with reliability or flow control.
Data may be lost or received out of sequence without any UDP mechanisms to recover or

reorder the data.

Bit (0) Bit (15)	Bit (16) Bit (31)	
Source Port (16)	Destination Port (16)	8 Bytes
Length (16)	Checksum (16)	
Application Layer Data (Size varies)		

Table.III.3:UDP Datagram.

III.10. REAL-TIME TRANSPORT PROTOCOL (RTP)

III.10.1. Definition

Short for Real-time Transport Protocol defines a standard packet format for delivering audio and video over the Internet. It is defined in RFC 1889, it was developed by the Audio Video Transport Working group and was first published in 1996.

RTP is used extensively in communication and entertainment systems that involve streaming media, such as telephony, video teleconference applications, television services and web-based push-to-talk features (27).

III.10.2. The RTP header structure

```
 0                   1                   2                   3
 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|V=2|P|X|  CC   |M|     PT      |       sequence number         |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|                           timestamp                           |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|           synchronization source (SSRC) identifier            |
+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+
|            contributing source (CSRC) identifiers             |
|                             ....                              |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
```

Figure.III.10:RTP header format.

- Version (V) : 2 bits

  This field identifies the version of RTP,the version is 2 upto RFC 1889.
- Padding (P) : 1 bit

  If the padding bit is set, the packet contains one or more additional padding octets at the end which are not part of the payload, the last octet of the padding contains a count of how many padding octets should be ignored.
- Extension (X) : 1 bit

  If the extension bit is set, the fixed header is followed by exactly one header extension.
- CSRC count (CC) : 4 bits

  The CSRC count contains the number of CSRC identifiers that follow the fixed header.
- Marker (M): 1 bit

  Marker bit is used by specific applications to serve a purpose of its own. We will discuss this in more detail when we study Application Level Framing.
- Payload Type (PT) : 7 bits

  This field identifies the format of the RTP payload and determines its interpretation by the application, this field is not intended for multiplexing separate media.
- Sequence Number: 16 bits

  The sequence number increments by one for each RTP data packet sent, and may be used by the receiver to detect packet loss and to restore packet sequence, the initial value of the sequence number is random (unpredictable).
- Timestamp : 32 bits

  The timestamp reflects the sampling instant of the first octet in the RTP data packet. The sampling instant must be derived from a clock that increments monotonically and linearly in time to allow synchronization and jitter calculations.
- SSRC: 32 bits

  The SSRC field identifies the synchronization source,this identifier is chosen randomly with the intent that no two synchronization sources within the same RTP session will have the same SSRC identifier.
- CSRC list : 0 to 15 items, 32 bits each

  The CSRC list identifies the contributing sources for the payload contained in this packet, the number of identifiers is given by the CC field.

III.11. REAL-TIME CONTROL PROTOCOL (RTCP)

III.11.1. Definition

The RTP control protocol (RTCP) it is used to monitor the transmission and quality-of-service (QoS) of streaming media, is based on the periodic transmission of control packets to all participants in the session, using the same distribution mechanism as the data packets, this protocol enables the periodic exchange of control information between session participants, with the main goal of providing qualityrelated feedback, this feedback can be used to detect and potentially correct distribution problems (28).

III.11.2. Types of RTCP packets :

- Sender Report (SR) : is used by active session participants to relay transmission and reception statistics.
- Receiver Report (RR) : is used to send reception statistics from those participants that receive but do not send media.
- Source Description (SDES) : contains one or more descriptions related to a particular session participant.
- BYE : indicates the end of participation in a session.
- APP : stands for application-specific functions, the APP packet enables RTCP to send packets that convey information specific to a particular media type or application (29)

III.11.3. RTCP message formats :

Each RTCP packet carries in its header one of the following packet type codes:

- 200 = SR Sender Report packet
- 201 = RR Receiver Report packet
- 202 = SDES Source Description packet
- 203 = BYE Goodbye packet
- 204 = APP Application-defined packet

III.11.4. The RTCP header structure :

<table>
<tr><th>Offsets</th><th>Octet</th><th colspan="8">0</th><th colspan="8">1</th><th colspan="8">2</th><th colspan="8">3</th></tr>
<tr><th>Octet</th><th>Bit [a]</th><th>0</th><th>1</th><th>2</th><th>3</th><th>4</th><th>5</th><th>6</th><th>7</th><th>8</th><th>9</th><th>10</th><th>11</th><th>12</th><th>13</th><th>14</th><th>15</th><th>16</th><th>17</th><th>18</th><th>19</th><th>20</th><th>21</th><th>22</th><th>23</th><th>24</th><th>25</th><th>26</th><th>27</th><th>28</th><th>29</th><th>30</th><th>31</th></tr>
<tr><td></td><td>0</td><td colspan="2">Version</td><td>P</td><td colspan="5">RC</td><td colspan="8">PT</td><td colspan="16">length</td></tr>
<tr><td></td><td>32</td><td colspan="32">SSRC</td></tr>
</table>

Table.III.4:RTCP packet header.

- Version : (2 bits) Identifies the version of RTP, which is the same in RTCP packets as in RTP data packets. The version defined by this specification is two (2).
- P (Padding) : (1 bits) Used to indicate if there are extra padding bytes at the end of the RTP packet. A padding might be used to fill up a block of certain size, for example as required by an encryption algorithm. The last byte of the padding contains the number of padding bytes that were added (including itself).
- RC (Reception report count) : (5 bits) The number of reception report blocks contained in this packet. A value of zero is valid.
- PT (Packet type) : (8 bits) Contains a constant to identify RTCP packet type.
- Length : (16 bits) Indicates the length of this RTCP packet.
- SSRC : (32 bits) Synchronization source identifier uniquely identifies the source of a stream (30).

III.12. SUMMARY

Voice over IP traffic does not necessarily have to travel over the global internet, it may also be deployed on private IP networks for example on a LAN inside a single building.

This new technology will for the most part, lead our communications into the next century, Consumers are thirsting for both a cheaper and more reliable telephone service, and VoIP might just be the answer, in the near future, telephony will be much more competitive in pricing and hopefully, the monopolies of the telephone industry shall no longer be able to control their individual sectors, with some fine-tuning and additional research,VoIP can become the new wave in communication.

The next chapter , We recognize the operation of the signaling and communication protocols and apply application protocol tracking with wireshark, and the simulation work process using packet tracer.

Chapter IV

IV.1.INTRODUCTION

For some time, those responsible for communications of companies have in mind the possibility of using its data infrastructure to transport the internal voice traffic of the company. However, it is the appearance of new standards, as well as the improvement and cheapening of voice compression technologies, which is finally causing its implementation and the implementation of VoIP, after having verified that from a PC with multimedia elements it is possible to perform telephone calls over the Internet, we can think that IP telephony is little more than a toy, because the quality of voice that we obtain through Internet is very poor. However, if in our company has a data network that has a very large bandwidth, we can also think about the use of this network for voice traffic between the different delegations or departments of the organization.

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Actually the integration of voice and data in the same network is an old idea, since time solutions have emerged from different manufacturers that, through the use of multiplexers, allow to use the WAN data networks of companies (typically point-to-point connections and Frame-Relay) for the transmission of voice traffic, the lack of standards, as well as the long term of amortization of this type of solutions has not allowed a wide implementation of them.

IV.2 ASTERISK VOIP PRIVATE BRANCH EXCHANGE

IV.2.1. What is Asterisk ?

Asterisk is a software implementation of a telephone private branch exchange (PBX), it allows attached telephones to make calls to one another, and to connect to other telephone services, such as the public switched telephone network (PSTN) and Voice over Internet Protocol (VoIP) services.

In simpler terms it's server software that helps you to create a small telephony network on VoIP platform, it utilises IPV4/IPV6 network, it does not requires analog lines.

IV.2.2. History of Asterisk

Asterisk is designed in 1999, created by Mark Spencer, then a student at Auburn University (United States - Alabama), in search of a private telephone switch to create a technical support center on Linux, he is dissuaded by the high tariffs of existing solutions, and decides to create his own call router in Linux.

IV.2.3. Project requirements

- Personal Computer (PC)
- Virtual Machine (VMware)
- AsteriskNOW-3.0.0-x86_64
- Windows 7 32 bit
- X-Lite Phone for Windows
- 3CX Phone for Windows
- PuTTY
- Wireshark

IV.2.4. Asterisk step-by-step installation

First, VMware (virtual operating system) was installed on computer, After that creat new virtual machine with advanced options.

Figure.IV.1:Create virtual machine.

Select operating system will be installed on this virtual machine Linux version CentOS 6 64-bit

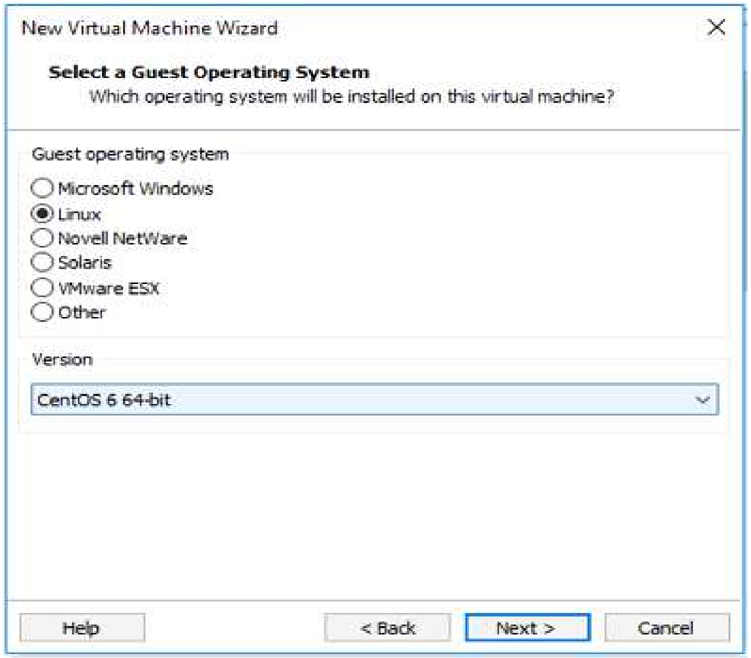


Figure.IV.2:Select operation system.

Now ,on click power on this virtual machine for start installing server asterisk

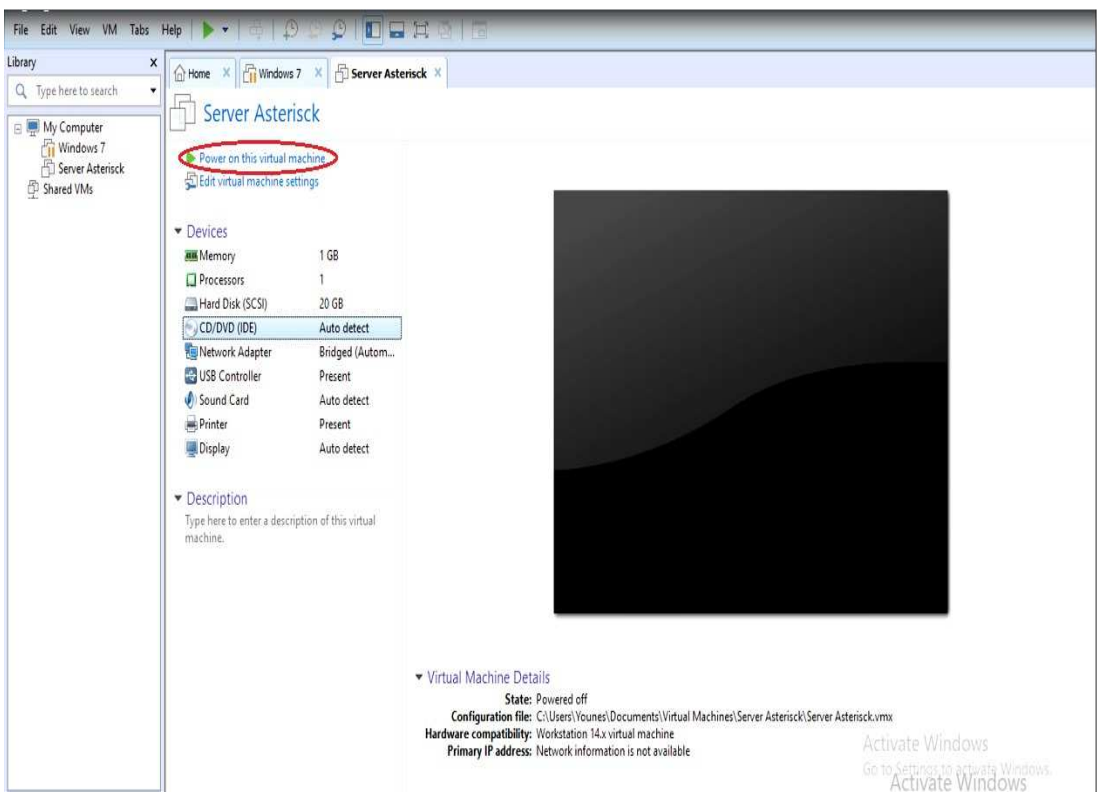


Figure.IV.3 : Power on this virtual machine.

Choose to install with Asterisk 11 and FreePBX ,on type 1 then click ENTER

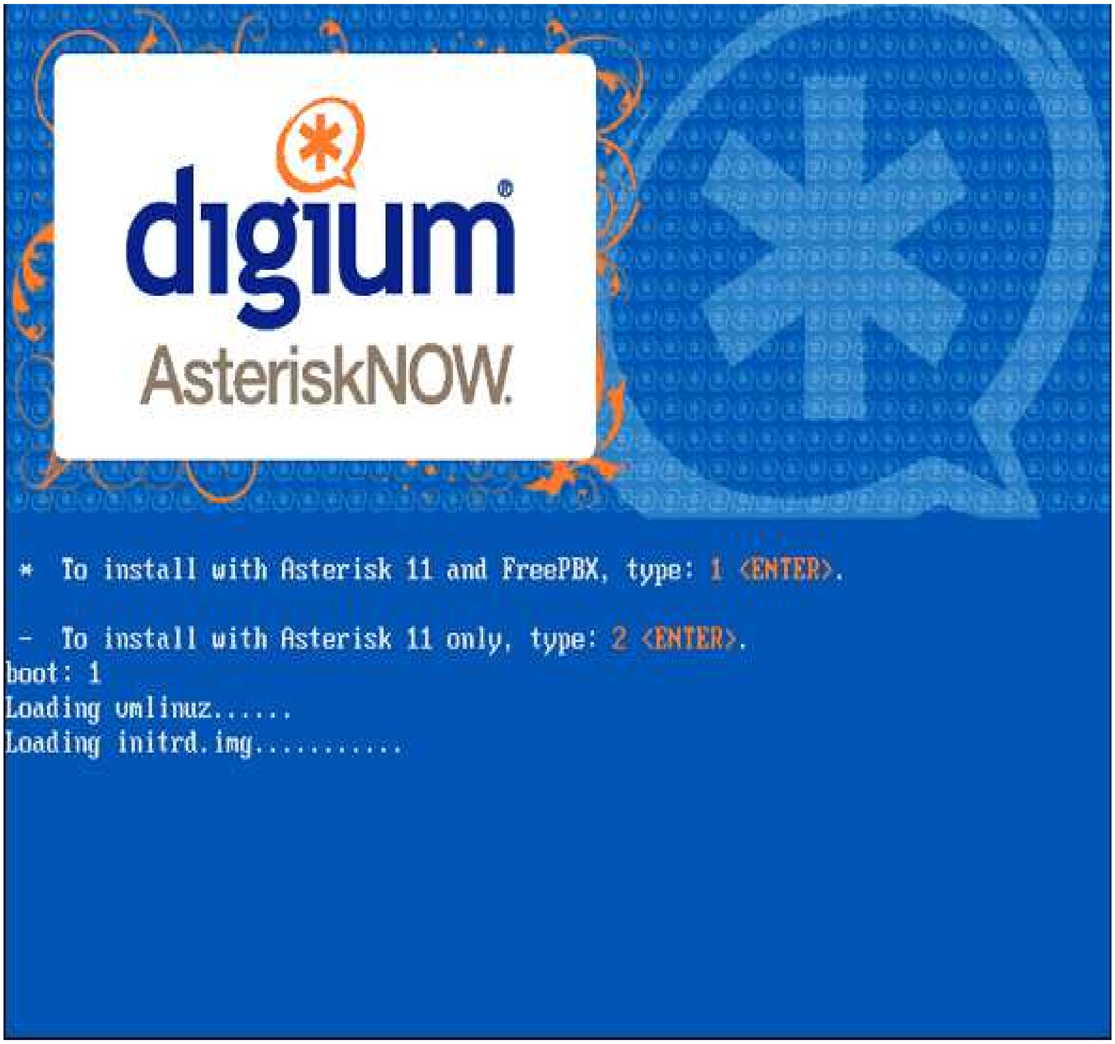


Figure.IV. 4 : Install asterisk.

Enter a password for root user then click NEXT

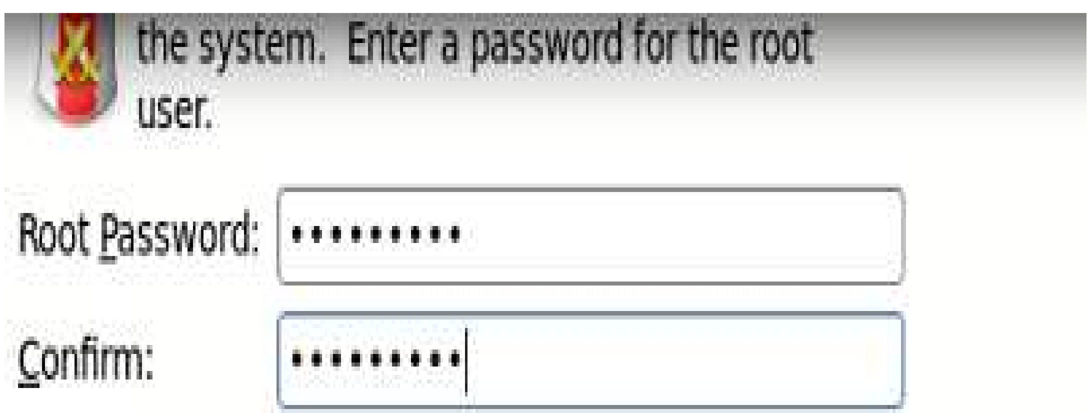


Figure.IV. 5 : Password for root.

Finally, CentOS installation is complete,now we reboot to use the installed system

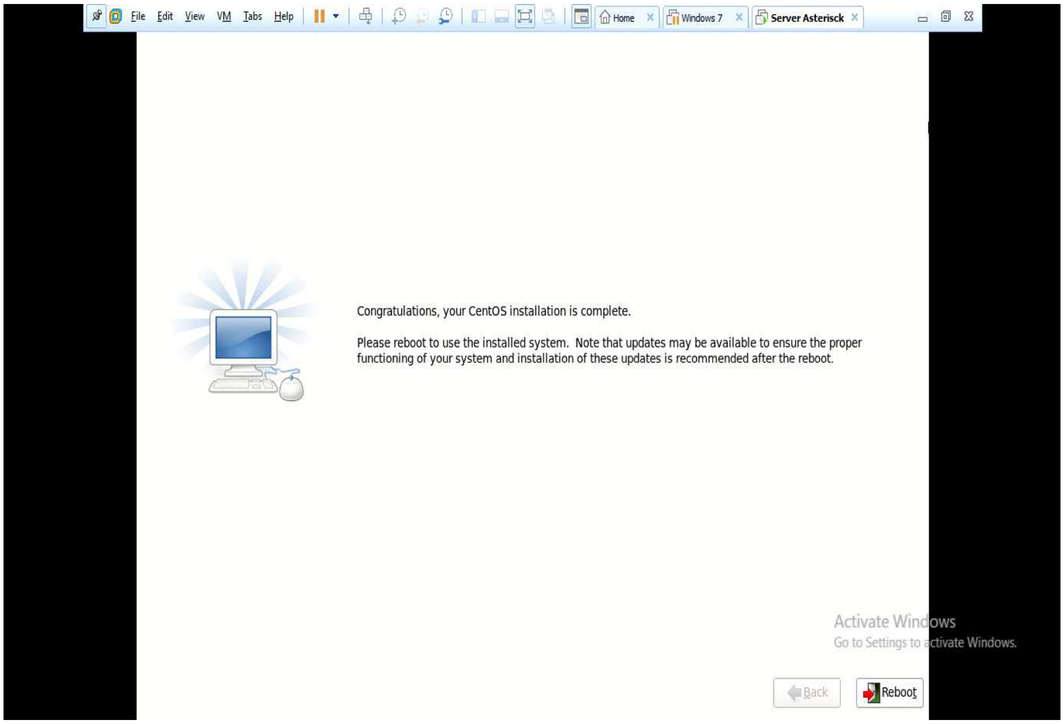


Figure.IV.6:CentOS installation complete.

After reboot,we can enter to Asterisk with two methods, the first my web browser to http:192.168.1.9/,we see asterisk use DHCP for give ip address.

The second method is CLI (Command Line Interface),we can enter to CLI, after type login name is (root) and password (*********)

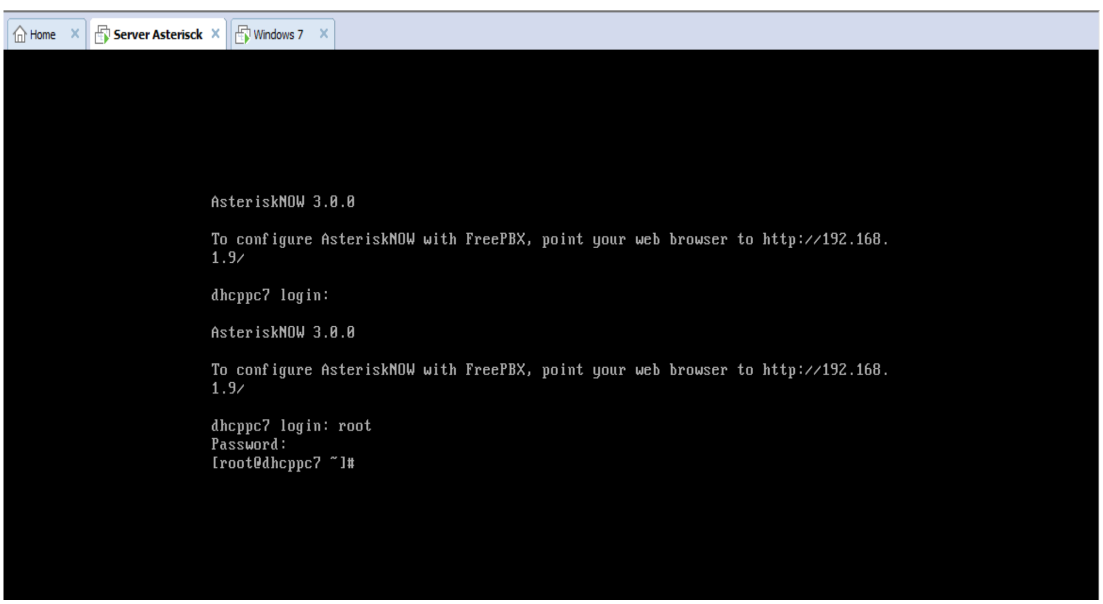


Figure.IV.7:Starting AsterisckNOW server.

IV.2.5. Configuration asterisk

To configure asterisk,enter login name and password used in the installation,type the command (setup) to enter configuration

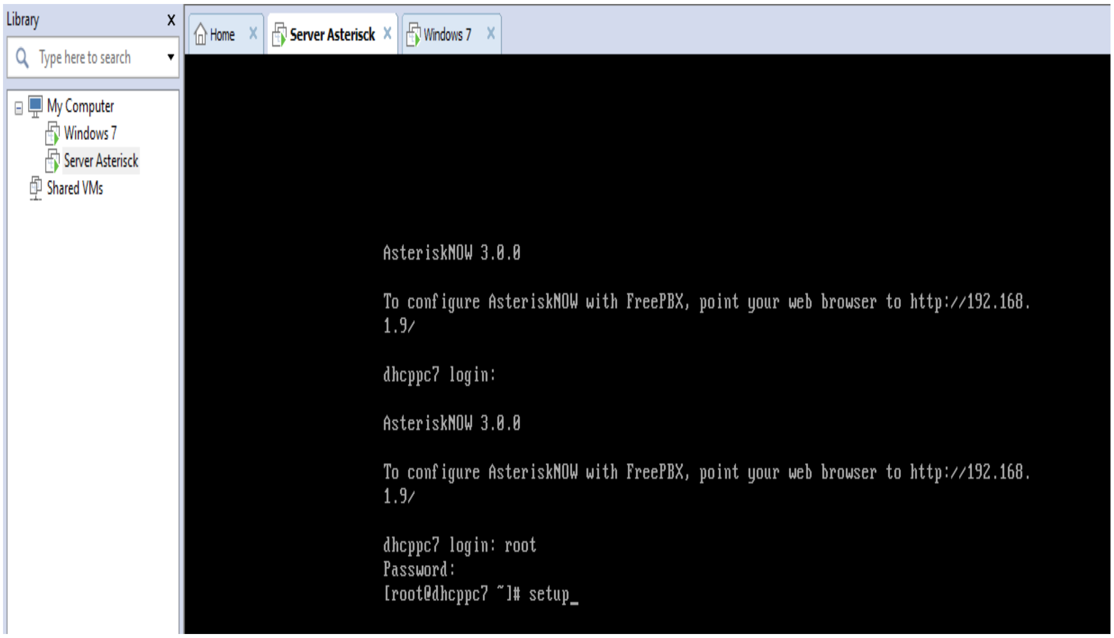


Figure.IV. 8 : Enter Configure AsteriskNOW.

We go to network configuration

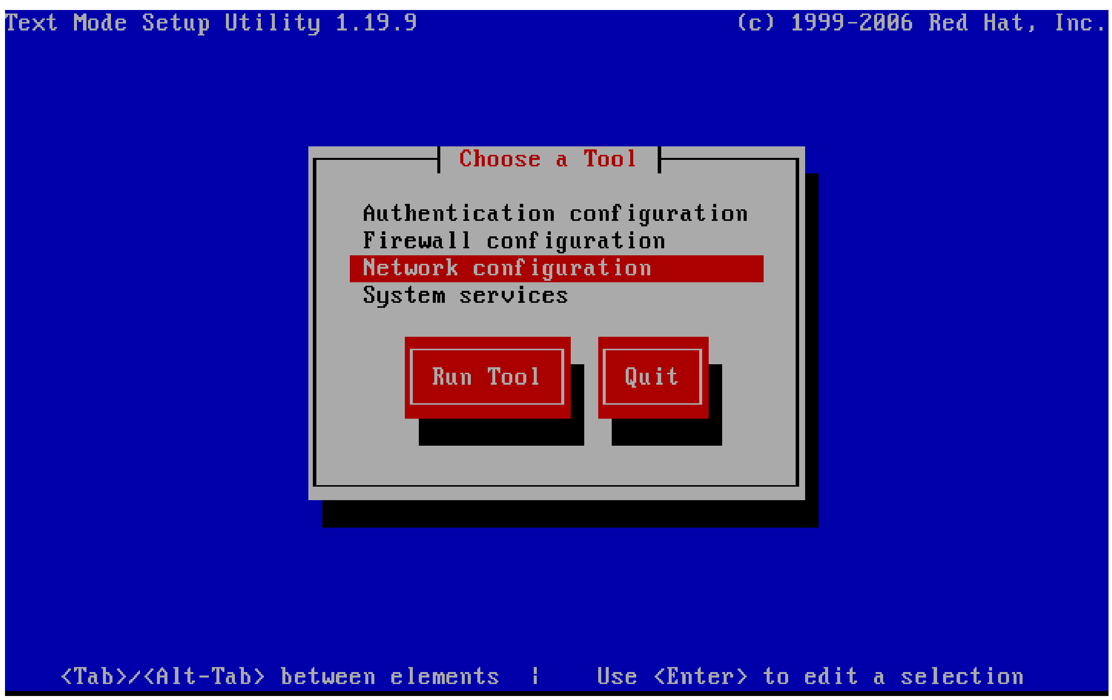


Figure.IV. 9 : Mode setup.

Its shows network configuration and DNS configuration ,add new configuration: Name,static IP Netmask, Default gateway, IP Primary DNS Server,Hostname,Primary DNS,DNS search path.

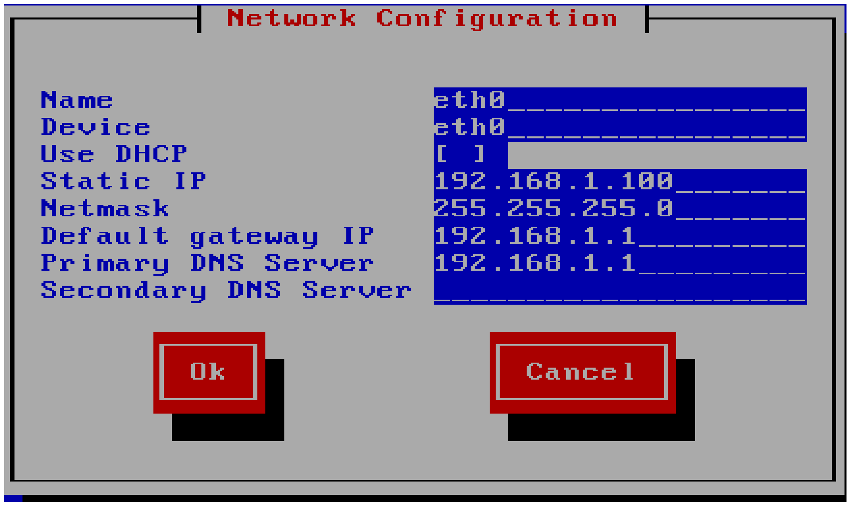


Figure.IV.10:Network configuration.

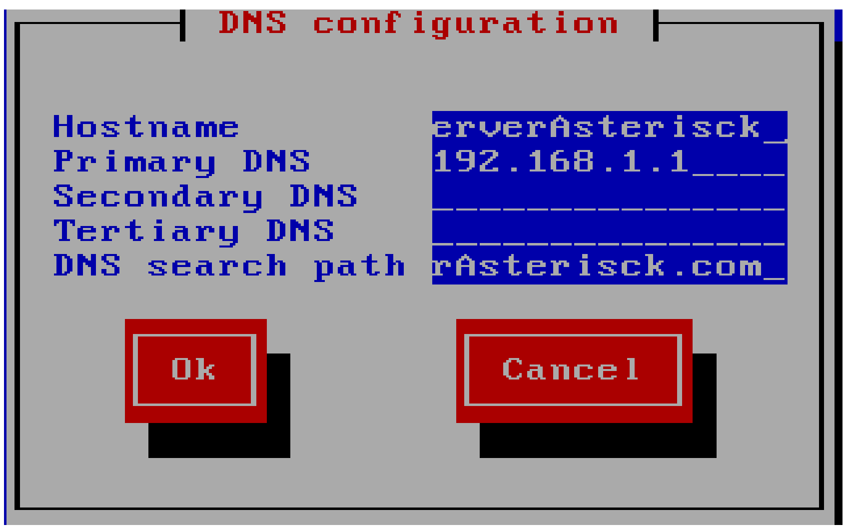


Figure.IV.11:DNS configuration.

Save all configuration ,the go out to CLI and type (reboot) then click enter

When start again have new static IP 192.168.1.100 and Host name ServerAsterisck

IV.2.6. Create extensions

Write this IP address 192.168.1.100 in my browser, following page will be open,on choose FreePBX Administration then enter credentials Login (admin) and Password (admin)

This is default username and password, there is also an option available to change username and password according to your desire.

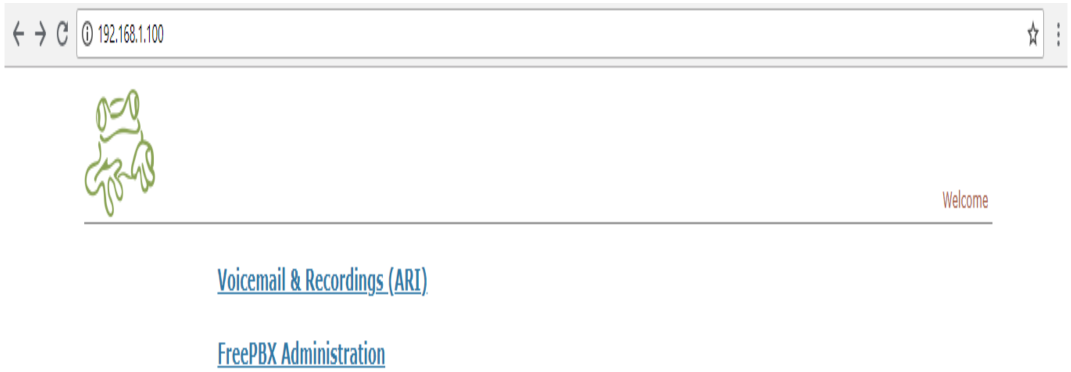


Figure.IV.12:Enter IP address in browser.

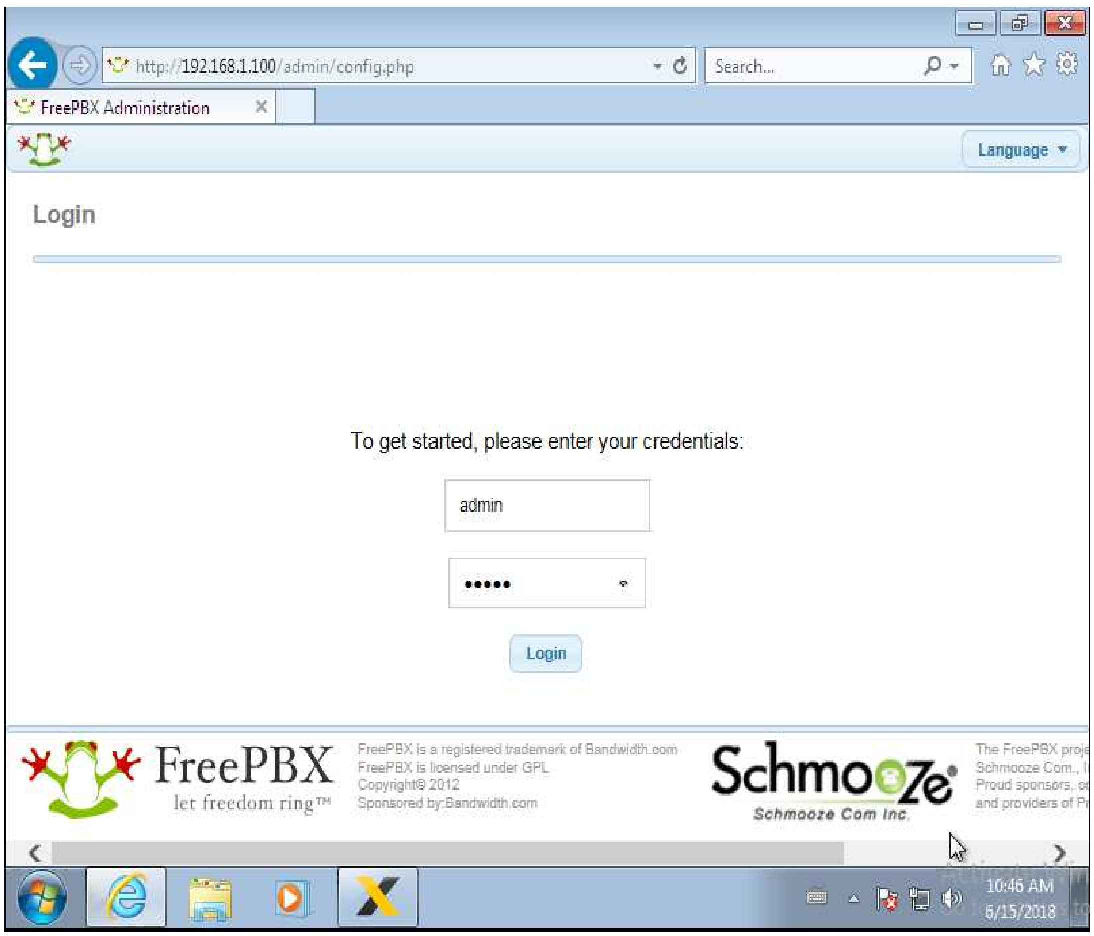


Figure.IV.13:Online access of FreePBX.

To create an extension, after logged in on PBX server:

Click PBX >> PBX setting >> click on extension on left hand side.

“Generic sip device” is showing in front of “Device” in the box.

Click on “Submit”

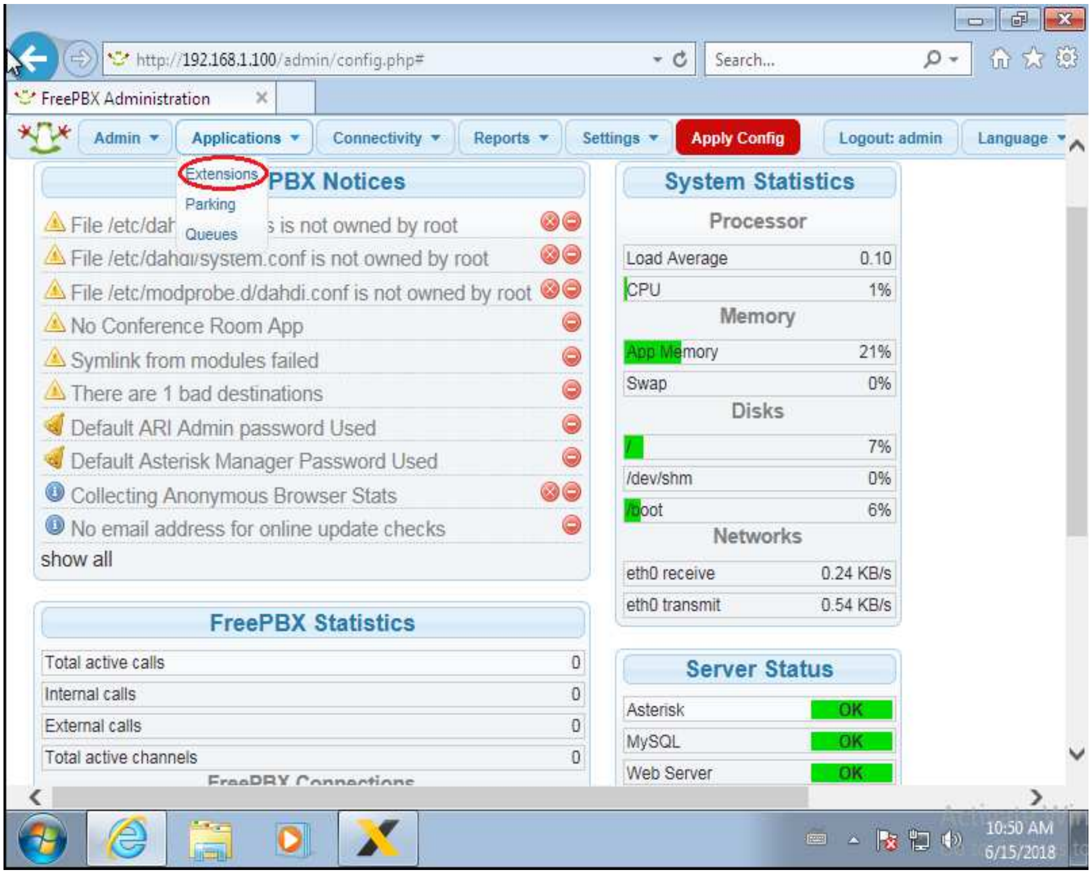


Figure.IV. 14:Create extensions

Figure.IV.15:Select device

Display Name	SIP Alias	Secret
Client 1	100	youben198
Client 2	200	youben198
Client 3	300	youben198
Client 4	400	youben198

Table.IV.1:Create four extension

We see is four extension create

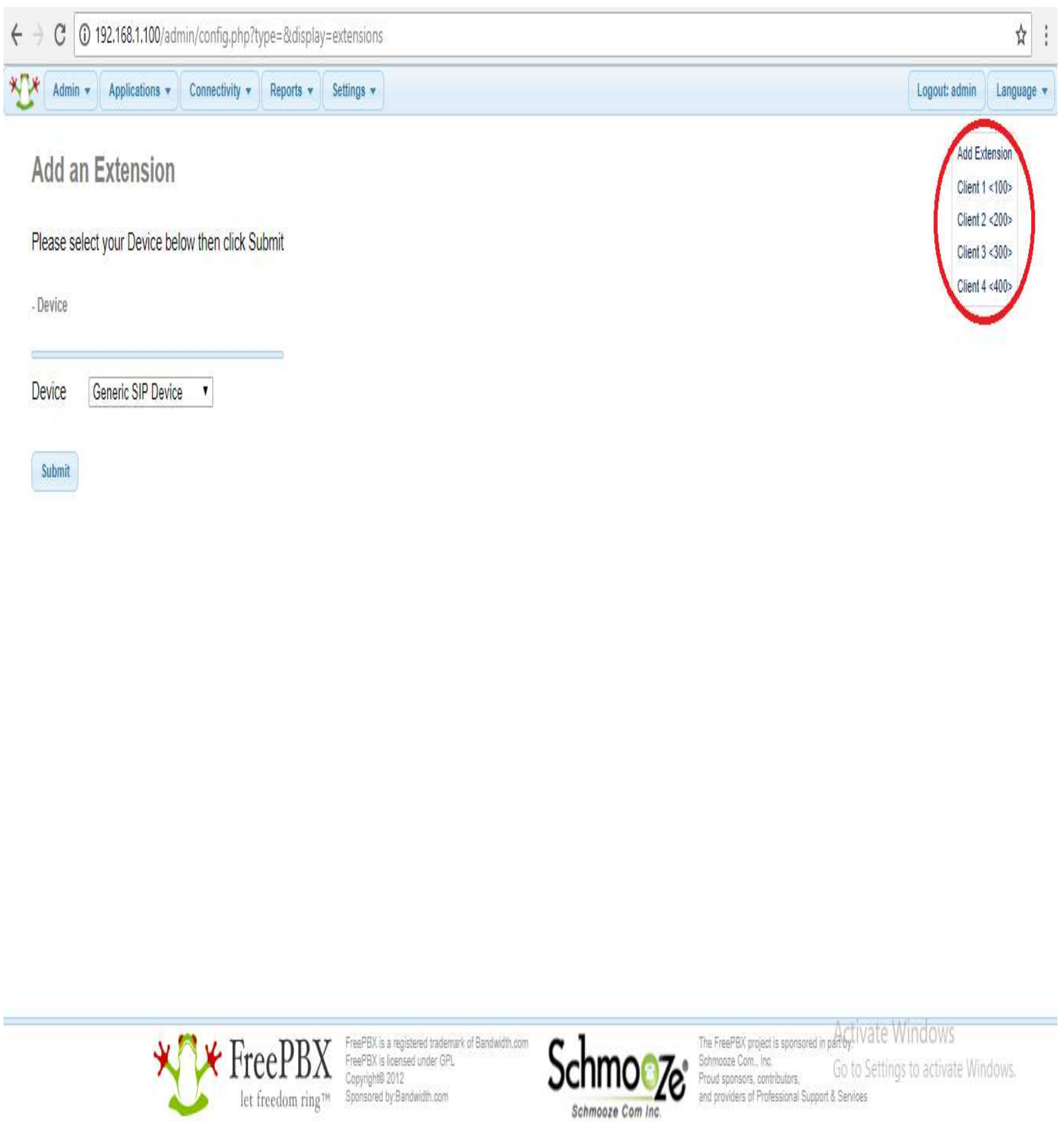


Figure.IV.16:Four extensions created

IV.2.7. Soft phone

first, test accessibility from all Clients to Server Asterisk enter to cmd and type ping 192.168.1.100,this ping is successful (0% loss)

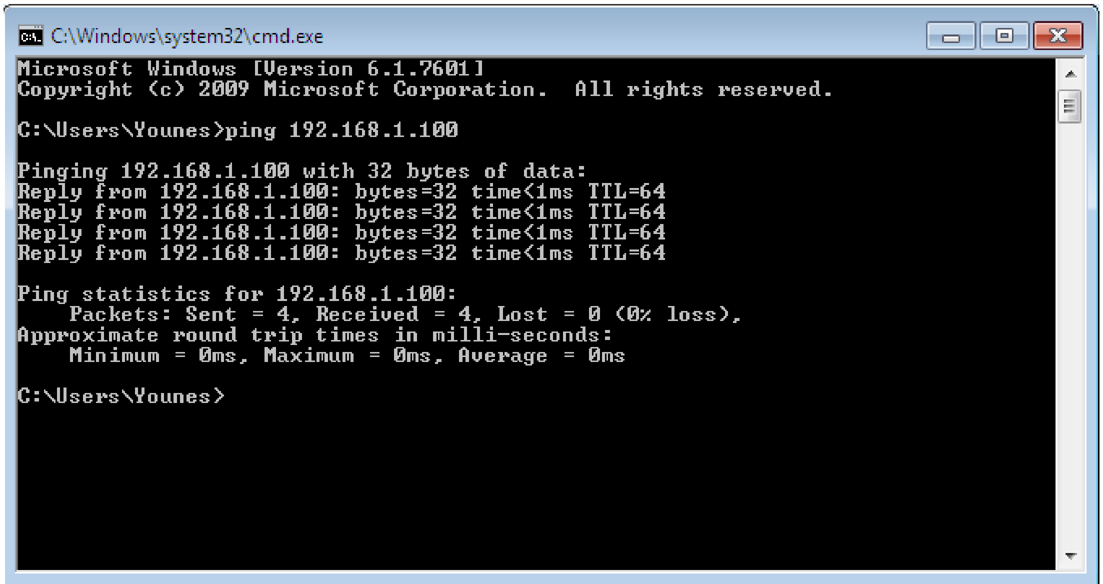


Figure.IV.17:Test accessibility

Next step is soft phone installation and their configuration. Such as X-lite and 3CX.

IV.2.7.1. Configuration des SOFTPhones X Lite

Install X Lite softphone on Client 1 et Client 3

Click on softphone option >> account setting

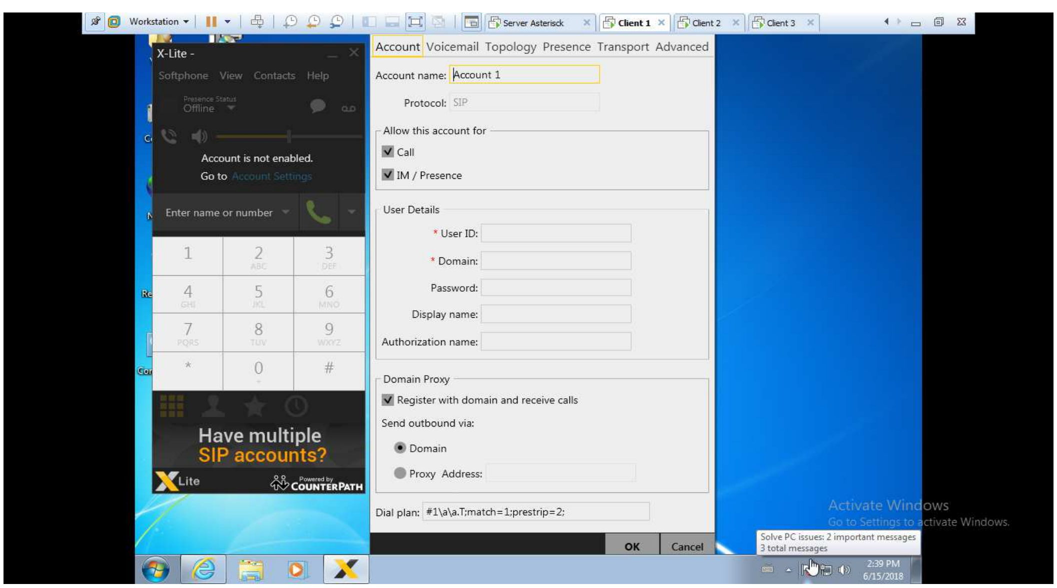


Figure.IV.18:Configuration X Lite

Account name	User ID	DOMAIN	Password
Client 1	100	192.168.1.100	********
Client 2	200	192.168.1.100	********

IV.2.7.2. Configuration des SOFTPhones 3CX

Install 3CX softphone on client 3 and Client 4

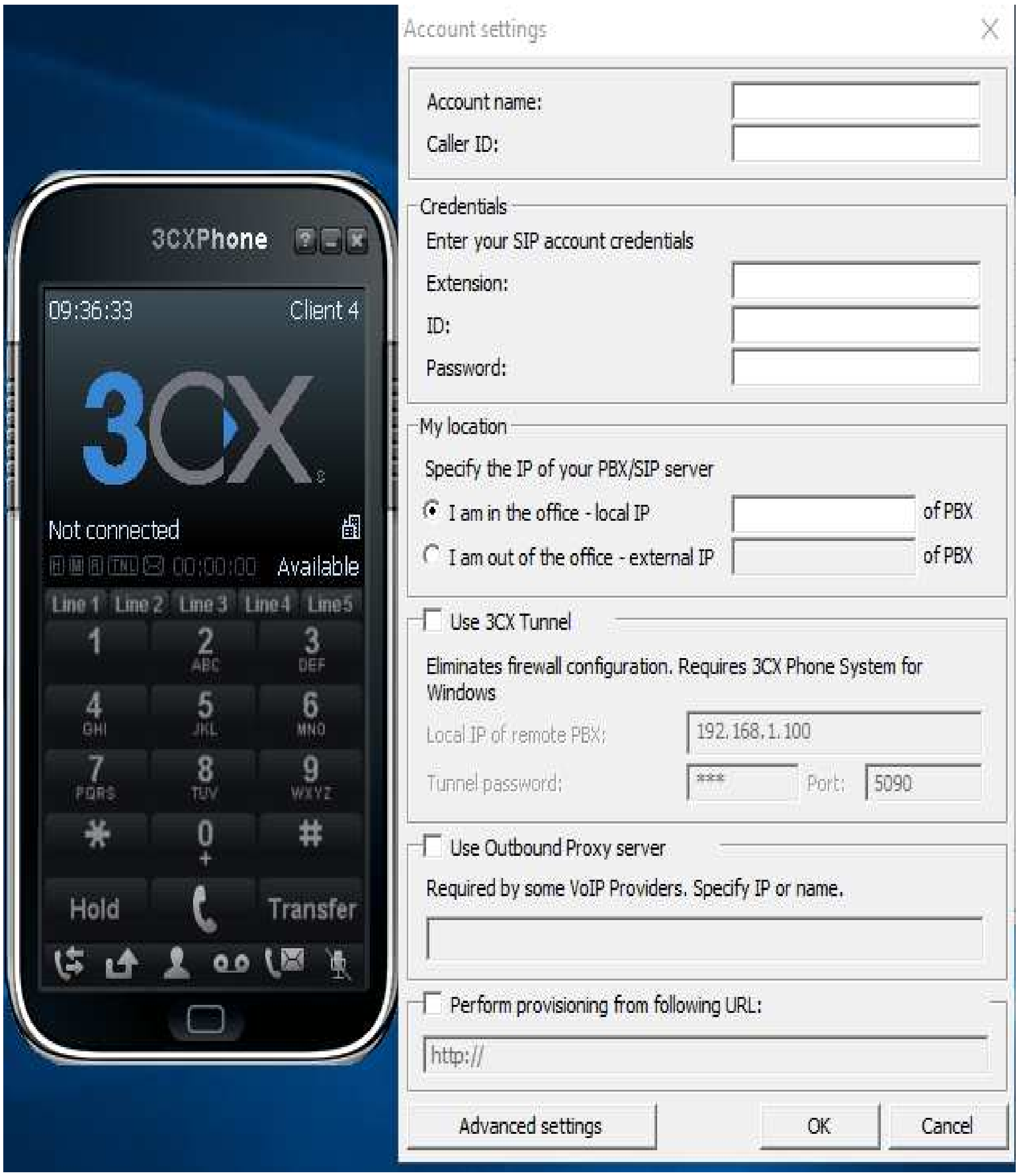


Figure.IV.19:Configuration 3CX

Account name	Extension	ID	Local IP	Password
Client 3	300	300	192.168.1.100	********
Client 4	400	400	192.168.1.100	********

IV.2.7.3. Test Call

Following image is the result of successful VoIP call between two extensions <100> and <400>.

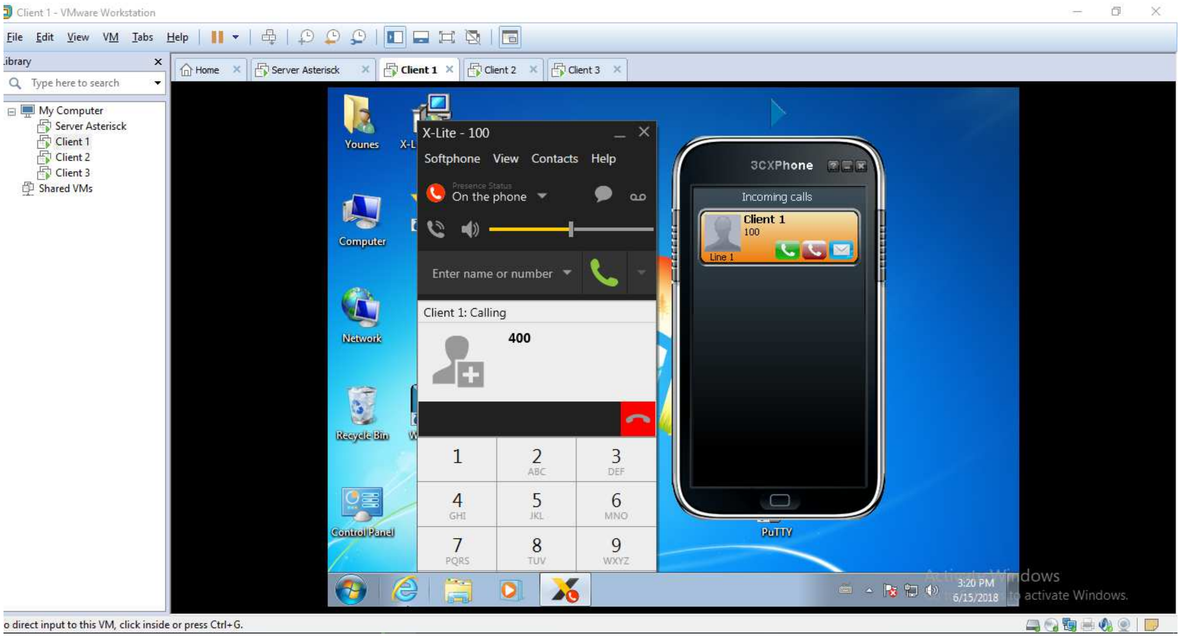


Figure.IV.20:Successful test call among extensions.

IV.2.7. Voice-mail setting

To setup a voice-mail service, go to PBX setting >> extensions >> get in extension <scroll down to voice-mail and directory:

Select enable >> voicemail password. Use something you can type on a phone keypad.

Enter an e-mail address where you would like to receive your voice messages sent and click submit. Then click on the red apply bar at the top of the screen.

We can enable setting of voice-mail setup on any extension according to our desire and need.

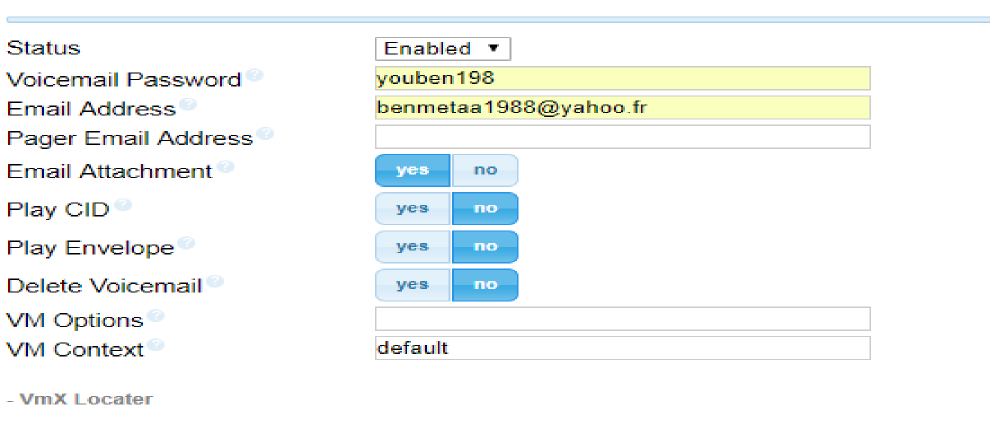


Figure.IV.21:Voice mail configuration.

IV.2.7. 1. Voicemail Result

Following result shows, voice mail sent from extension to email ID. (from <400> to Benmetaa1988@yahoo.fr

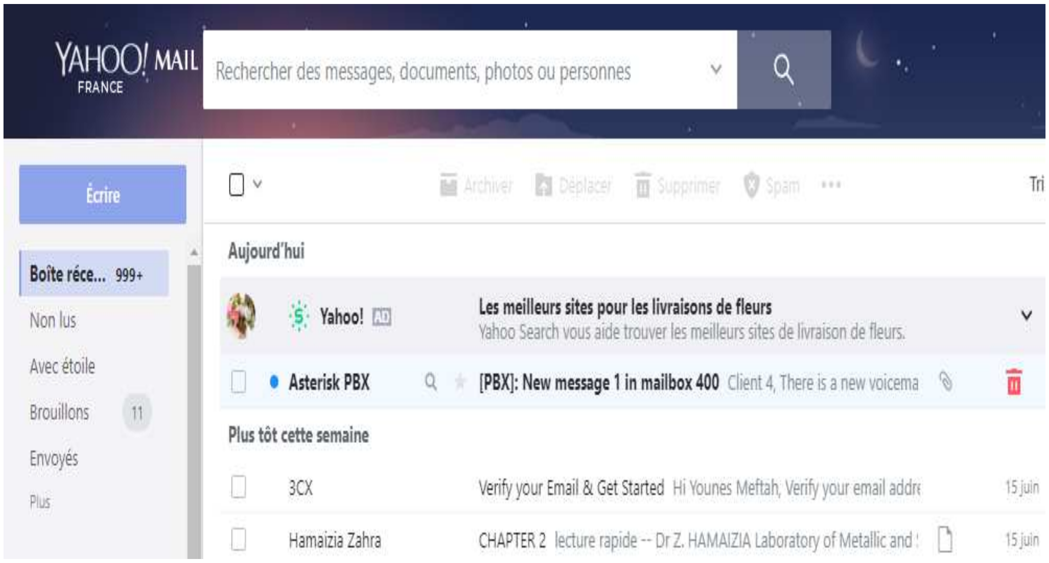


Figure.IV. 22 : Result of voice mail in e-mail ID.

IV.2.8. Management and configuration asterisk with SSH protocol

Open PuTTY and select SSH protocol,enter IP address (192.168.1.100) or Host Name, Port 22

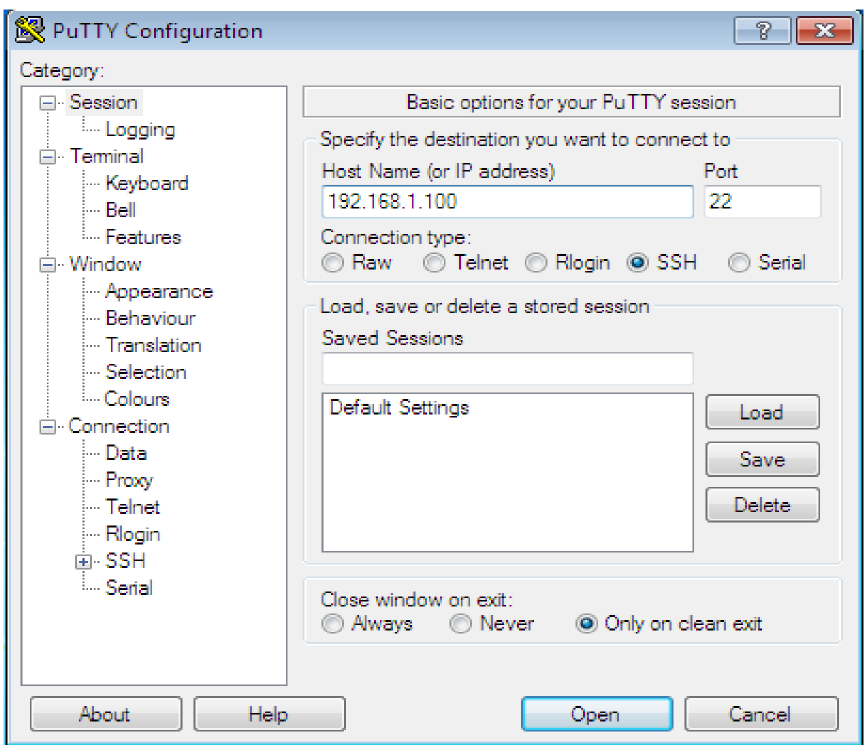
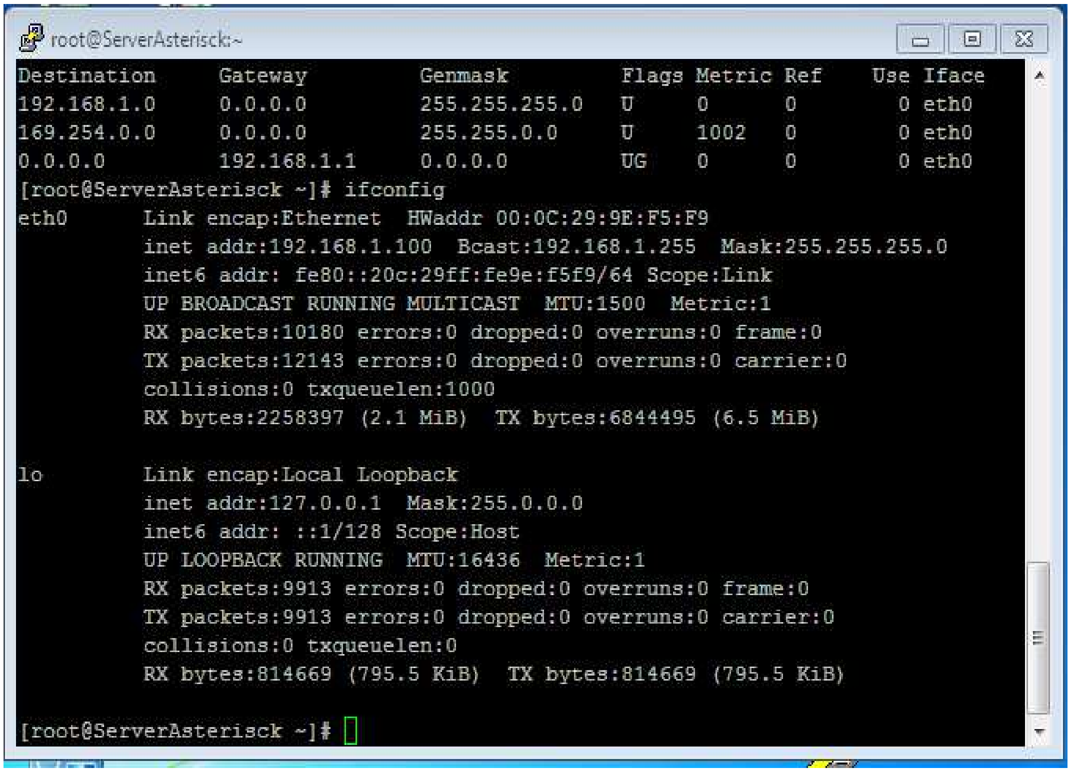


Figure.IV.23:Access server asterisk.

The Asterisk command line interface (CLI) is reached by using the Linux shell command

ifconfig: show inforamation network card

asterisk –vvvvvvvr: debug protocol sip

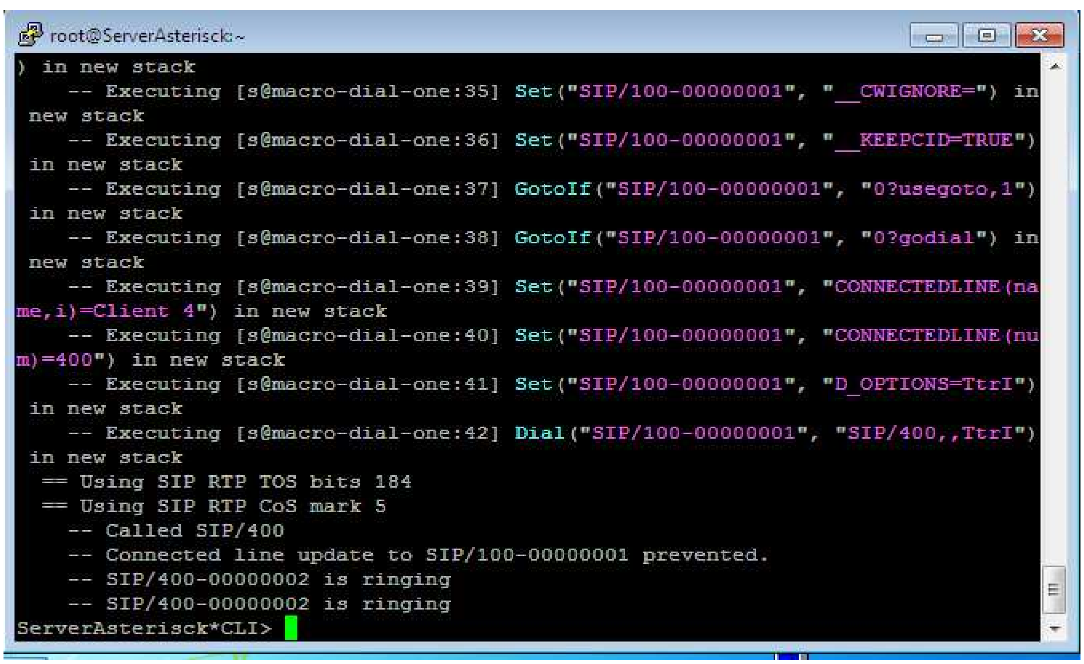


```
root@ServerAsterisck:~
Destination     Gateway         Genmask         Flags Metric Ref    Use Iface
192.168.1.0     0.0.0.0         255.255.255.0   U     0      0        0 eth0
169.254.0.0     0.0.0.0         255.255.0.0     U     1002   0        0 eth0
0.0.0.0         192.168.1.1     0.0.0.0         UG    0      0        0 eth0
[root@ServerAsterisck ~]# ifconfig
eth0      Link encap:Ethernet  HWaddr 00:0C:29:9E:F5:F9
          inet addr:192.168.1.100  Bcast:192.168.1.255  Mask:255.255.255.0
          inet6 addr: fe80::20c:29ff:fe9e:f5f9/64 Scope:Link
          UP BROADCAST RUNNING MULTICAST  MTU:1500  Metric:1
          RX packets:10180 errors:0 dropped:0 overruns:0 frame:0
          TX packets:12143 errors:0 dropped:0 overruns:0 carrier:0
          collisions:0 txqueuelen:1000
          RX bytes:2258397 (2.1 MiB)  TX bytes:6844495 (6.5 MiB)

lo        Link encap:Local Loopback
          inet addr:127.0.0.1  Mask:255.0.0.0
          inet6 addr: ::1/128 Scope:Host
          UP LOOPBACK RUNNING  MTU:16436  Metric:1
          RX packets:9913 errors:0 dropped:0 overruns:0 frame:0
          TX packets:9913 errors:0 dropped:0 overruns:0 carrier:0
          collisions:0 txqueuelen:0
          RX bytes:814669 (795.5 KiB)  TX bytes:814669 (795.5 KiB)

[root@ServerAsterisck ~]#
```

Figure.IV. 24 Command ifconfig.



```
root@ServerAsterisck:~
) in new stack
    -- Executing [s@macro-dial-one:35] Set("SIP/100-00000001", "__CWIGNORE=") in
 new stack
    -- Executing [s@macro-dial-one:36] Set("SIP/100-00000001", "__KEEPCID=TRUE")
 in new stack
    -- Executing [s@macro-dial-one:37] GotoIf("SIP/100-00000001", "0?usegoto,1")
 in new stack
    -- Executing [s@macro-dial-one:38] GotoIf("SIP/100-00000001", "0?godial") in
 new stack
    -- Executing [s@macro-dial-one:39] Set("SIP/100-00000001", "CONNECTEDLINE(na
me,i)=Client 4") in new stack
    -- Executing [s@macro-dial-one:40] Set("SIP/100-00000001", "CONNECTEDLINE(nu
m)=400") in new stack
    -- Executing [s@macro-dial-one:41] Set("SIP/100-00000001", "D_OPTIONS=TtrI")
 in new stack
    -- Executing [s@macro-dial-one:42] Dial("SIP/100-00000001", "SIP/400,,TtrI")
 in new stack
  == Using SIP RTP TOS bits 184
  == Using SIP RTP CoS mark 5
    -- Called SIP/400
    -- Connected line update to SIP/100-00000001 prevented.
    -- SIP/400-00000002 is ringing
    -- SIP/400-00000002 is ringing
ServerAsterisck*CLI>
```

Figure.IV. 25 : Command debug sip.

IV.2.9. VoIP Wireshark Capture

An effective way of capturing traffic in a LAN is to install Wireshark on the main VoIP server. After Wireshark is installed on the VoIP server, open it and select the interfaces that the capture will occur on. Select the Start button, and captured packets should begin flying across the screen.

We make sure that a minimum of two end devices are registered with the SIP server, and place a call from one end device to another.

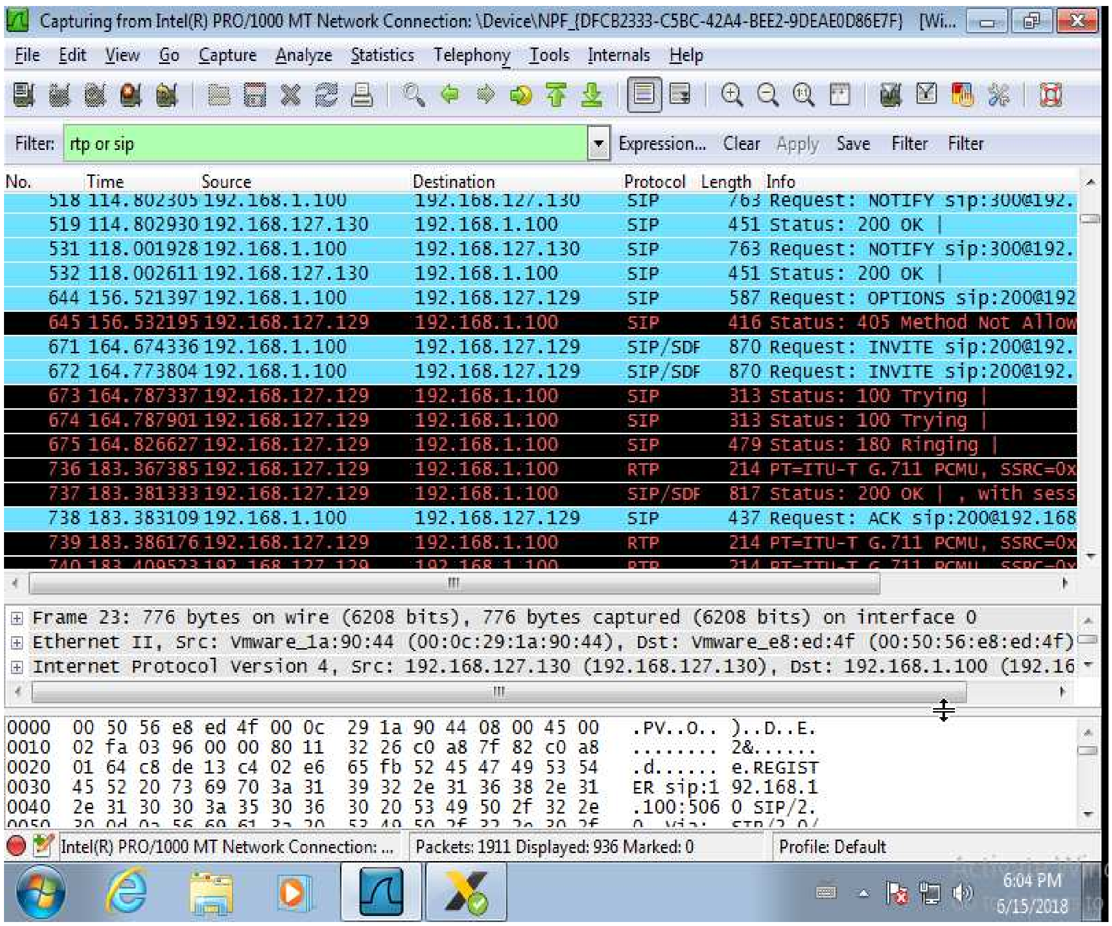


Figure.IV.26: List VoIP calls.

The VoIP calls list shows the following information per call:

- Start Time: Start time of the call.
- Stop Time: Stop time of the call.
- Initial Speaker: The IP source of the packet that initiated the call.
- For SIP calls, it is the "From" field of the INVITE

- Protocol: Any of the protocols listed above
- Packets: Number of packets involved in the call.
- State: The current call state. The possible values are
  - CALL SETUP: call in setup state (Setup, Proceeding, Progress or Alerting).
  - IN CALL: call is still connected.
  - CANCELLED: call was released before connect from the originated caller.
  - COMPLETED: call was connected and then released.
  - REJECTED: call was released before connect by the destination side.
  - UNKNOWN: call in unknown state.

For H323 calls it shows if the call uses Fast Start or/and H245 Tunneling.

IV.2.9.1. Playing VoIP calls

We can now see all RTP streams available for the calls

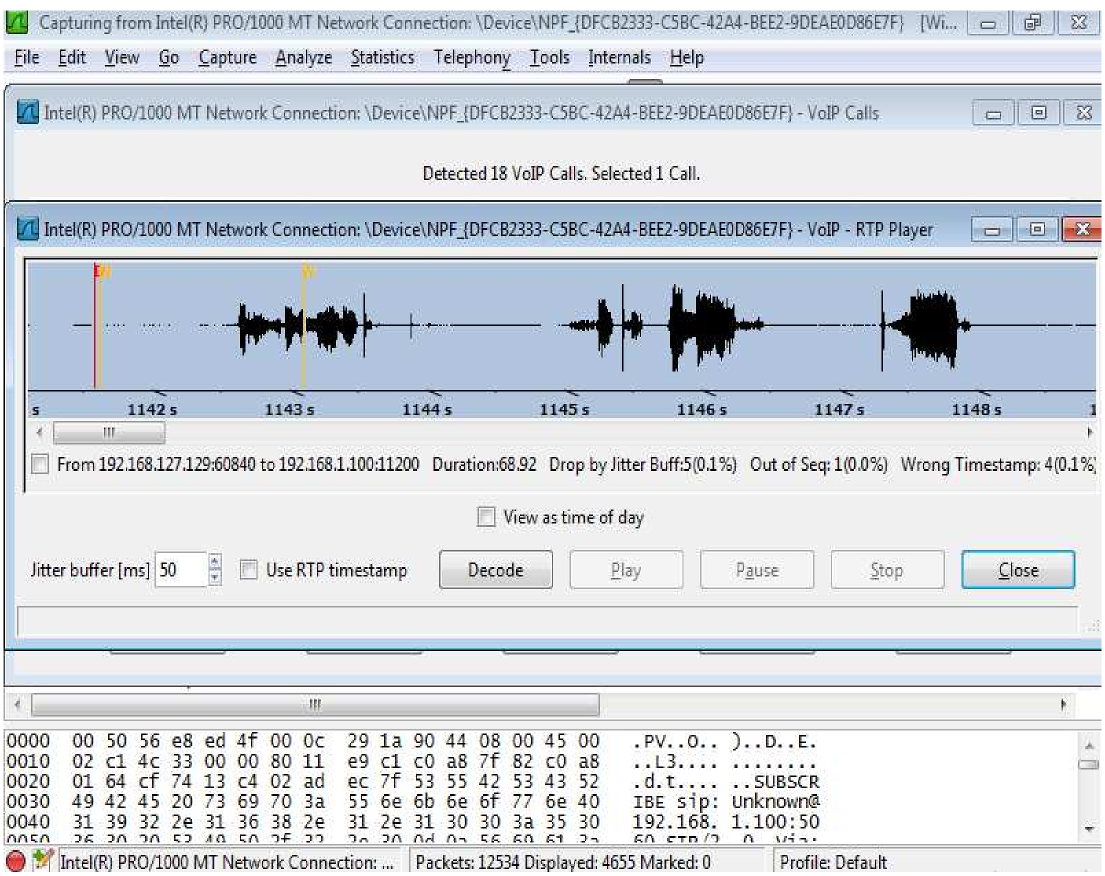


Figure.IV.27:RTP player.

RTP packets that are dropped because of the jitter buffer are reported (Drop by Jitter Buff), as well as the packets that are out of sequence (Out of Seq).

IV.2.10. Firewall Setup for Maximum Security and Usability

Using protocols Secure SHell (SSH) and Hypertext Transfer Protocol (HTTP) constitute risk,we stop this protocols by suspended with firewall.

We go to firewall configuration and difine services are trusted ,the http and ssh is untrusted

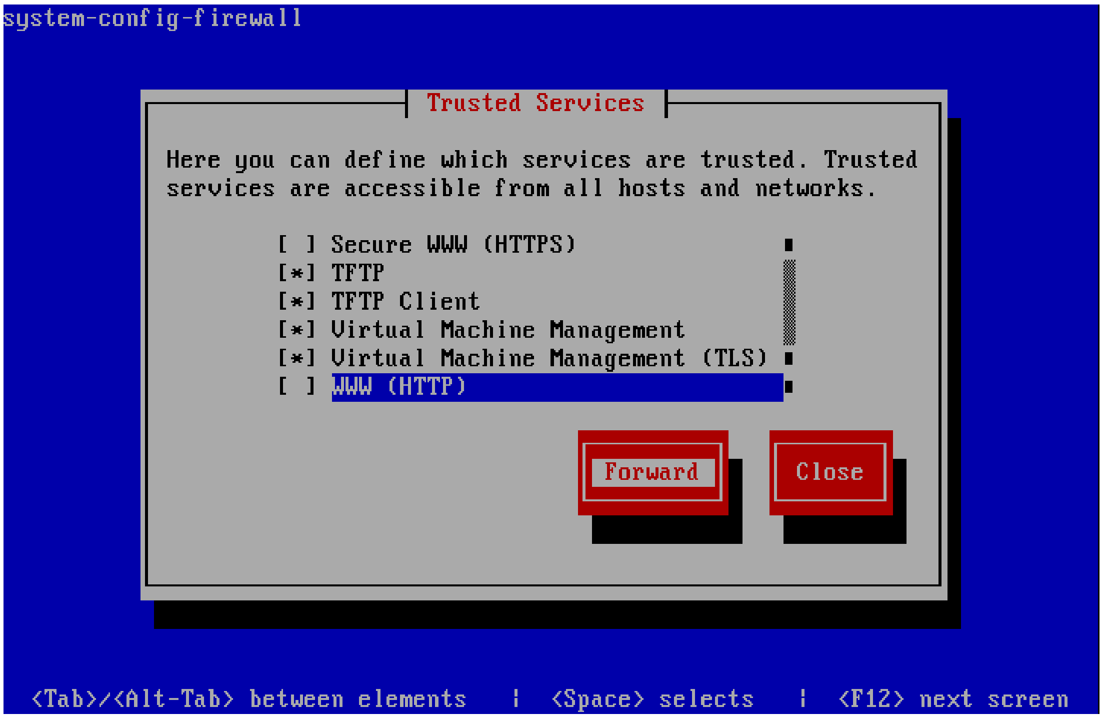


Figure.IV.28:Deselect HTTP and SSH.

IV.2.10. 1.Result deselect HTTP and SSH

Go to browser and enter 192.168.1.100

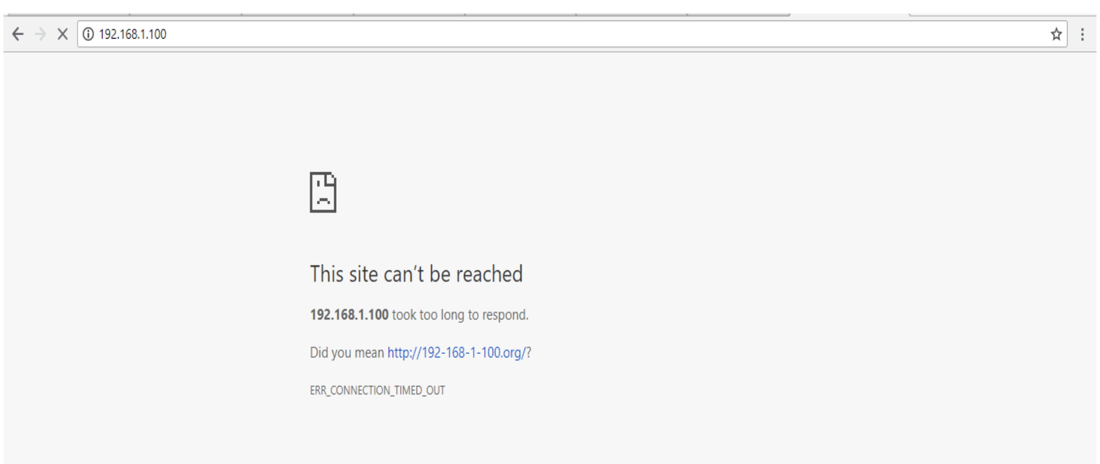


Figure.IV.29:Error connetion.

Go to putty and choose SSH protocol, port 22 IP address 192.168.1.100

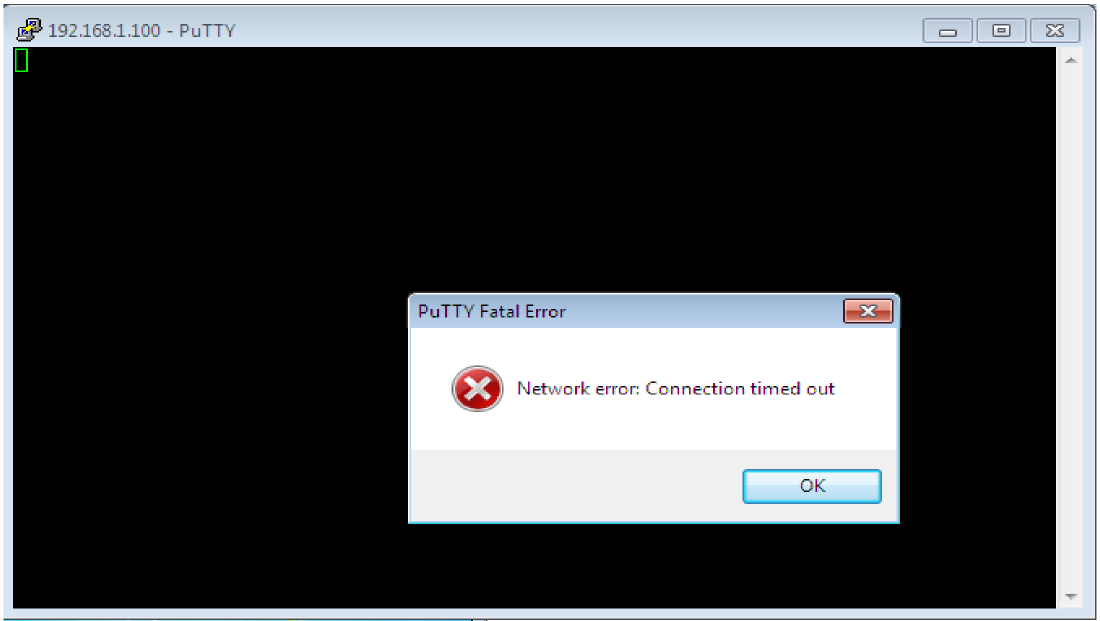


Figure.IV.30:Network error

IV.3. VoIP SIMULATION WITH PACKET TRACER

IV.3.1. Network Topology

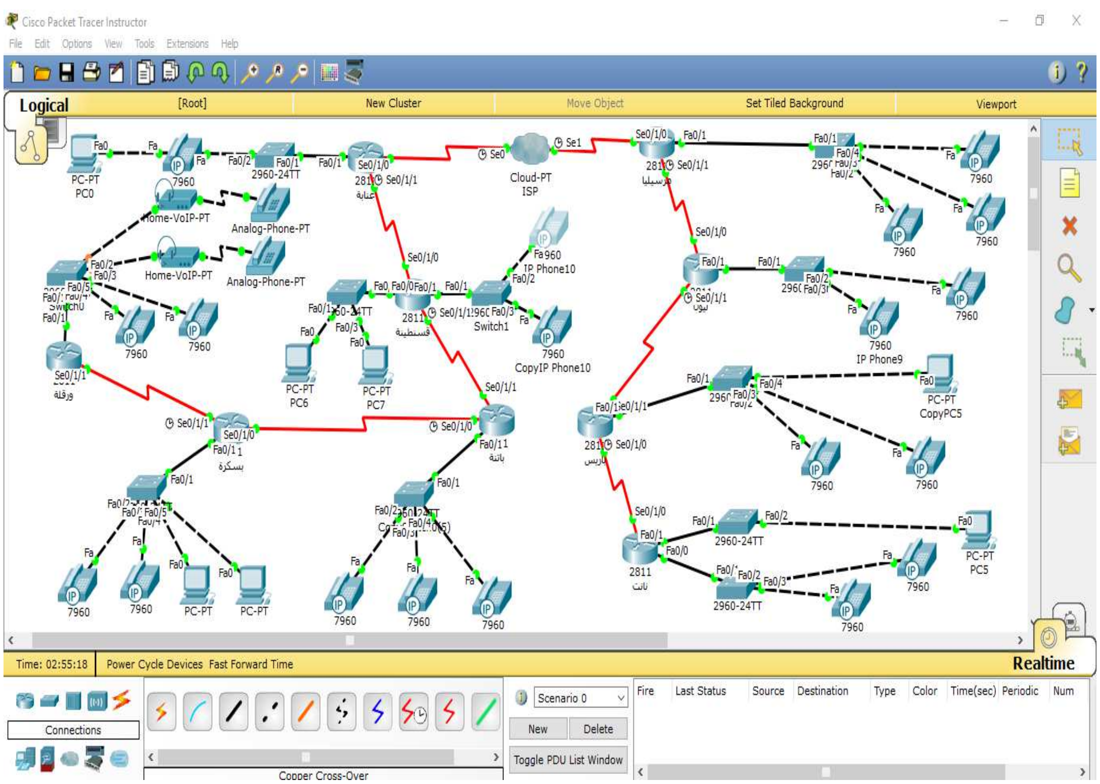


Figure.IV.31:Network Topology.

IV.2.2.Scenario

To deploy well-designed computer networks supporting converged applications of data, audio, and video.

- To support better performance, resilience, security, maintainability.
- This is to propose a network model that can be implemented in different network environments and also be used to implement MAN, WAN.
- Use of Voice over Internet Protocol technology allows to make a phone call using an Internet connection or a dedicated network that uses the IP protocol, rather than go through the normal telephone line.
- This save bandwidth used. Data packets, containing voice information are routed over the network, encoded in digital form.
- To filters & check unauthorized traffic that travels across the network.
- The major objective of this network design is to allow only the genuine users to access the network and prevent the intruders from accessing it.

IV.2.3.Network description

- The routers used are 2811
- Also have opted for 2960 switches
- Have added PC's as voice functionality can be added to those if desired and to check whether both the network could ping each other or not and for various operational functionalities.
- Have kept IPV4 addressing sceme to the various networks.
- The router Annaba and the router Marseille are connected via leased line (Frame Relay).

IV.3.3.Network configuration

1. Router configuration

Router>enable

Router#conf terminal

Router(config)#interface fastethernet or serial

Router(config-if)#ip address network mask

Router(config-if)#clock rate 64000

Router(config-if)#no shut

2. Frame-Relay configuration

Frame relay is a telecommunication service designed for cost-efficient data transmission for intermittent traffic between local area networks (LANs) and between end-points in a wide area network (WAN). Frame relay puts data in a variable-size unit called a frame and leaves any necessary error correction (retransmission of data) up to the end- points, which speeds up overall data transmission.

Router (frame-relay)

Router(config)#int s0/2/0

Router(config-if)#encapsulation frame-relay

Router(config-if)#frame-relay lmi-type cisco

Router(config-if)#frame-relay interface-dlci 102

Router(config-if)#no shut

3. DHCP configuration

Router(config)#ip dhcp pool

Router(dhcp-config)#(network ID) (Networkmask)

Router(dhcp-config)#default-router (address IP)

Router(dhcp-config)#exit

4. RIP (ROUTING INFORMATION PROTOCOL)

It is a dynamic routing protocol technique

RIP prevents routing loops by implementing a limit on the number of hops allowed in a path from the source to a destination,the maximum number of hops allowed for RIP is 15.

Router(config)#router rip

Router(config-router)#network ID

5. VLAN

In computer networking, a single layer-2 network may be partitioned to create multiple distinct broadcast domains, which are mutually isolated so that packets can only pass between them via one or more routers; such a domain is referred to as a Virtual Local Area Network, Virtual LAN or VLAN.

Switch>enable

Switch#conf terminal

Switch(config)#vlan (number)

Switch(config-vlan)#name

Switch(config-vlan)#exit

Switch(config-vlan)#int fastethernet

6. TELNET

Telnet is a user command and an underlying TCP/IP protocol for accessing remote computers. Through Telnet, an administrator or another user can access someone else's rerrererrr rerremotely.

Router(config)#line vty 0 4

Router(config-line)#password

Router(config-line)#login

Router(config-line)#exit

7. Voice over IP

Voice over IP (voice over Internet Protocol, VoIP) is a methodology and group of technologies for the delivery of voice communications and multimedia sessions over Internet Protocol (IP) networks, such as the Internet.

8. Call Manager Express

Router(config)#telephony-service

Router(config-telephony)#max-dn 5

Router(config-telephony)#max-ephones 5

Router(config-telephony)#ip source-address port 2000

Router(config-telephony)#auto assign 4 to 6

Router(config-telephony)#auto assign 1 to 5

Router(config-telephony)#exit

9. Phone Directory config

Router(config)#ephone-dn 1

Router(config-ephone-dn)#number 1101

Router(config-ephone-dn)#exit

10. Dial-Peer config

Router(config)#dial-peer voice 1 voip

Router(config-dial-peer)#destination-pattern 2.

Router(config-dial-peer)#session target ipv4:192.168.0.2

Figure.IV.32:IP phone ring out.

Figure.IV. 33: Call came from out.

IV.4. SUMMARY

Voice over IP (VolP) protocol is used to carry voice signal or the IP network. This allows us to use IP Telephone instead of the dedicated voice transmission telephone lines. This paper is focused in the Implementation of Voice over Internet Protocol (VolP) using open source (Asterisk). There are some characteristics for system in this chapter such as types of protocol used to establish the connection. The platform for this system design is by using Linux as an Operating System (Debian). The open source software is used to implement the proposed solution.

The problem is necessary for the IP telephony stream to be converted by a gateway to another format, either for interoperation with a different IP based multimedia scheme or because you are placing a call to the traditional Public Telephone Network (PSTN). The overll technology requirements of an IP telephony solution can therefore be split into four categories signaling, encoding, transport and gateway control.

The objective of this chapter is to setup a VolP server by using Asterisk open source which is implemented on a Local Area Network (LAN) and also the main characteristics of VoIP will be explained, to make more use of the internet line rather than its usage for only surfing and chatting.

CONCLUSION AND FUTURE WORK

Based on the work done throughout this thesis and the proposed methodology to deploy a network with VoIP capability, it can be used as a guide or reference for a more exhaustive evaluation of the various parameters that impact on the performance of the network . You could also evaluate different scenarios either by applying more parameters of quality of service in the devices or introducing some different parameter or some new technology.From this work and the results involved can continue to develop experiments with more complex network scenarios including more equipment, with this the evaluation of the network implemented under new parameters.

Some of the companies that offer solutions of this nature are: Cisco, Alcatel, etc. Alcatel offers PBX which can serve up to 200 users, one of them is the OMNITOUCH Call Center Office, Cisco offers "Cisco AVVID" that provides solutions for multiservice networks capable of transmitting data, video and voice mainly in IP networks, HP also presents options to allow access to voice services. Open call is one of the solutions that allows access to the IP network to transmit voice and data. It is important to mention that in the telecommunications department there is no VoIP network, so this work would also be useful as a reference and future deployment of a VoIP network in that department, that the basic configuration is already presented in this investigation.

In conclusion, we can ensure that the best way to obtain a VoIP network with the desired performance is based on a good planning and implementation above all of the quality of service requirements taking as reference the values of the parameters such as throughput, delay and jitter that impact the performance of a network of these characteristics.

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