IP PBX PHONE SYSTEM

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DEDICATION

Optional dedication page.

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CHAPTER 1

INTRODUCTION

Communication is the activity of conveying information through the exchange of thoughts, messages or information, as by speech, visuals, signals, writing, or behavior.

Telecommunication is a general term for a vast array of technologies that sends information over distances. Mobile phones, land lines and voice over internet protocol (VOIP) are all telephony technologies.

Communication is interaction among people or sharing information. There are two types of as analog communications and digital communications.

Digital communications mean transferring data from one place to another. It is done by physical path or physical connection. In digital communication digital values are taken as discrete set.

The internet is a global system of interconnected computer networks that use the standard internet protocol (TCP/IP) to serve billions of users worldwide.

VOIP is a technology that allows telephone calls to be made over computer networks like that the Internet. VOIP converts analog voice signals into digital data packets and supports real time and two ways transmission of conversations using Internet Protocol (IP).

A PBX is called "Private Branch Exchange" which is a private telephone network used within a company.

The aim of the thesis is the research the IP PBX system and to realize an IP PBX system which is chosen. It is chosen IP PBX that is 3CX Phone System.

This thesis consists of four sections and last section implementations.

In chapter 2; TCP/IP Protocol and Data Communication, and in chapter 3; the definition of VOIP and their kinds of using. Also it is mentioned the relationship with IP PBX and VOIP, and in chapter4; PBX Systems. The definition of PBX System and IP PBX is the new version of PBX .It has been explained the kinds of IP PBX and using of their features. It has been explained using of benefits IP PBX Phone System. And the implementation section, it is explained 3CX Phone System.

In chapter 2: TCP/IP Protocol and Data Communication. This chapter introduces Transmission Control Protocol/Internet Protocol (TCP/IP) operating systems. For the TCP/IP Architecture; TCP/IP Layers and TCP/IP protocol suite, network administrators must understand the current standards process, and the common terms used to describe network devices and portions of a network. There are two layered communication protocols for using.

TCP (Transfer Control Protocol) and IP (Internet Protocol) are working together TCP (Transfer Control Protocol) is top layer. TCP (Transfer Control Protocol) is used for transmission of data from an application to the network. IP (Internet Protocol) is lower layer. IP (Internet Protocol) deals the communication with other computers.

The User Datagram Protocol is the one of core members of the Internet Protocol Suite, the set of Network Protocols used for the Internet. [2]

TCP/IP Protocol Layers: Application Layer, Transport Layer, Internetwork Layer, Network layer. Protocols which are defined for application layer serve the top programmers. SMTP is the standard protocol used to exchange Internet mail between TCP/IP hosts. Electronic mails on the Internet standard protocol that allows the purchase and shipping. [3]

FTP or File Transfer Protocol is used to transfer data from one computer to another over the internet or through a network. [5]

The Network News Transfer Protocol (NNTP) is an application protocol used for transporting Usenet news articles (net news) between new servers and is used for reading and posting articles by end user client applications. [6]

OSI Layer Model is a creation defined by international organization for standards and OSI stands for Open Systems Interconnection.OSI Model includes seven different layers. [1]

This section; defines IP Protocol and explains of IP network and their elements. And then explains their figures. IP Protocol is primary network used on the internet. It is explained in this section; the definition of IP networks and their features of them. Networks provide communication between computing devices. To communicate properly, all computers (hosts) on a network need to use the same communication protocols. [9]

This section introduces IP address and the importance of using it. It has been explained the kinds of IP address and using of their features. An IP address is an identifier for computer or device on TCP/IP network. [12]

In chapter 3: VOIP This section is beginning the definition of VOIP and it is explained about advantages and disadvantages of VOIP. VOIP is a technology that allows telephone calls to be made over computer networks like that the Internet. VOIP converts analog voice signals into digital data packets and supports real time and two ways transmission of conversations using Internet

Protocol (IP). VOIP telephone also known as a SIP phone or a soft phone allows the user to make phone calls to any soft phone mobile or landline by using VOIP. This way the voice is carried through the internet instead of the traditional PSTN system. [17] It is explained the communication between PSTN and IP Networks and It has variation of communication. These are following: from phone to phone, from computer to phone, from computer to computer, Mobil VOIP, and Wireless VOIP. Also it is explained using of their features. It has been explained about the VOIP relationship of IP PBX.

In chapter 4: PBX SYSTEMS: This section is beginning the definition of PBX System. IP PBX is the new version of PBX .It is explained the kinds of IP PBX and using of their features. It is explained using of benefits IP PBX Phone System. The kinds of IP PBX Phone System; Asterisk (Linux based IP PBX), SIPX (Another Linux based IP PBX) and 3CX (Windows based IP PBX).

It is about the research the IP PBX system and to perform an IP PBX system which is chosen. A PBX is called "Private Branch Exchange" which is a private telephone network used within a company. A PBX (Private Branch Exchange) is a switch station for telephone systems. [19]

The users of PBX phone system share a number of outside lines for making external phone calls. A virtual PBX system is a network of telecommunication channels that functions without physical connections. [20]

A VOIP Phone System / IP PBX system consists of one or more SIP phones / VOIP phones, an IP PBX server and optionally includes a VOIP Gateway.

The IP PBX server is similar to a proxy server: SIP clients, being either soft phones or hardware based phones, register with the IP PBX server, and when they wish to make a call they ask the IP PBX to establish the connection. [22]

The IP PBX Phone System has some features of keys.

SIP (Session Initiation Protocol): There are two types of SIP Phones.

The first type is hardware SIP phone, which resembles the common telephone but can receive and make calls using the Internet instead of the traditional PSTN System. [24] They allow any computer to be used as a telephone by means of a headset with a microphone and a sound card. A broadband connection and connection of VOIP provider or a SIP server is also required. SIP also defines server network elements. The IP PBX Phone System is explained how many kinds based of IP PBX Phone systems and their features.

Asterisk is Linux based IP PBX: Asterisk is an open framework for building communications applications. Asterisk turns an ordinary computer into a communications server. Asterisk powers IP systems, VOIP gateways, conference servers and other custom solutions. Asterisk is free and open source. Asterisk is sponsored by Diguim. [23]

SIPXECS (Enterprise Communication Server) is another Linux based IP PBX. SIPXECS (Enterprise Communication Server) is an open source voice over IP telephony server. [25]The main feature is a software implementation of the SIP (Session Initiation Protocol) which makes IP based communication system (IP PBX). SIPXECS is not like Asterisk and it is very popular open source PBX. There are difference between SIPXECS and Asterisk.

3CX is Windows based IP PBX: 3CX Phone System is windows based IP PBX. It is traditional software based IP PBX which is replaced hardware of PBX. Evolve your communications 3CX Phone System for Windows, an IP Phone System that completely replaces your proprietary PBX. [27]

2) TCP/IP PROTOCOL AND DATA COMMUNICATION

2.1) TCP/IP Protocol

TCP/IP Protocol and OSI (Open System Interconnection) have different operating systems machines which provide communication with each other in the model creates the transmission layer that communicates between applications. There are two layered communication protocols for using. TCP (Transfer Control Protocol) and IP (Internet Protocol) are working together.

- TCP(Transfer Control Protocol)

TCP (Transfer Control Protocol) is top layer. TCP (Transfer Control Protocol) is used for transmission of data from an application to the network.TCP is responsible for breaking data into IP packets before they sent and they assembling the packets when they arrive.

- IP (Internet Protocol)

IP (Internet Protocol) is lower layer. IP (Internet Protocol) deals the communication with other computers. IP is responsible for sending the receiving data packets over the Internet.

2.2) TCP/IP Protocol Architecture

TCP/IP is a Protocol set, it is a common used protocol set that independent computer systems can be run systematically at the internet, TCP/IP protocol set is used so that reason, the usage of TCP/IP became common. That is; the institutions which are used different protocol sets without TCP/IP in their LAN, for Internet connection; they installed TCP/IP protocol sets or added transit systems for TCP/IP.

Although the TCP/IP protocols are a specific Transport Layer protocol (TCP) that is running on top of a Network Layer protocol (IP), the TCP/IP actually is used to describe a large number of protocols that includes the following set of protocols. [1]

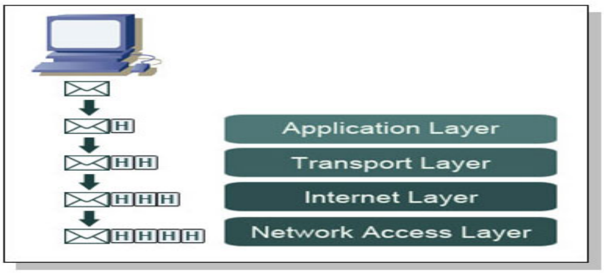


Figure2.1: TCP/IP Layers

The different layers in the TCP/IP protocol are not very well structured as it is the case in the OSI reference model where different layers may interact with other layers skipping layers in between. This gives the TCP/IP protocol suite more flexibility. The following figure shows a mapping between the layers of the TCP/IP and the OSI model. Note that this mapping is not fully agreed on where different textbooks and different people use slightly different mappings.[1]

OSI reference model is built on having seven layers that the architecture of the TCP/IP protocol Model and OSI Model are as shown in the following figure.

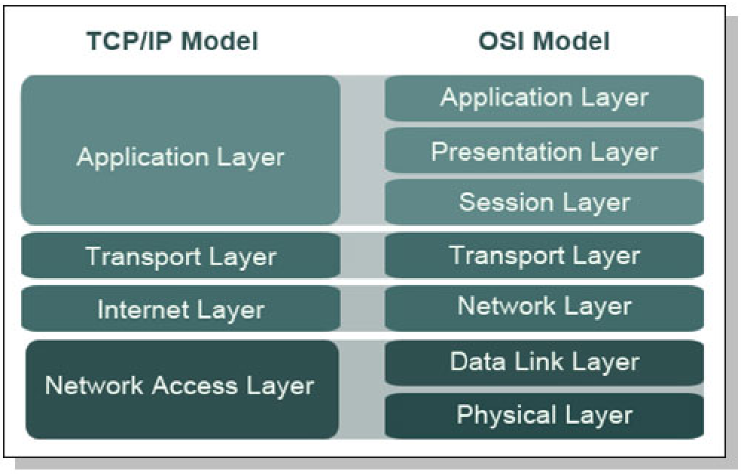


Figure2.2:TCP/IP Model and OSI Model

We see that some of the protocols of the:

Application Layer

Hyper Text Transfer Protocol (HTTP), Simple Mail Transfer Protocol (SMTP), File Transfer Protocol (FTP), Domain Name System (DNS) Protocol and Real-time Transfer Protocol (RTP).

Transport Layer

Transport Control Protocol (TCP) which is used by HTTP, SMTP, and FTP, and User Datagram Protocol (UDP) which is used by DNS and RTP.

Internet Layer

Internet Control Message Protocol (ICMP), Address Resolution Protocol(ARP) Reverse Address Resolution Protocol (RARP) and Internet Protocol (IP) which is used by TCP, UDP, and ICMP.

Network Access Layer

Many systems exist in this layer including LAN, Token Ring and Asynchronous Transfer Protocol (ATM).[1]

2.2.1) TCP/IP Protocol Layers

1. **Network Access Layer**

Network Access Layer is the first layer of four layers TCP/IP model. Network Access Layer defines details of how data is physically sent through the network, including how bits are electrically or optically signaled by hardware devices that interface directly with a network medium, such as coaxial cable, optical fiber, or twisted pair copper wire. The protocol also includes in Network Access Layer Ethernet, Token Ring, FDDI, X.25, Frame Relay e.g.

The most popular LAN architecture among those listed above is Ethernet. Ethernet uses an Access Method called CSMA/CD (Carrier Sense Multiple Access/Collision Detection) to access the media.

TCP/IP Model and the comparison between four layered TCP/IP model and seven layered OSI Model. [2]

2. **Internet Layer**

Internet Layer is the second layer of the four layers TCP/IP model. The position of Internet Layer is between Network Access Layer and Transport Layer. Internet layer pack data into data packets known as IP datagram, which contain source and destination address (logical address or IP address) information that is used to forward the datagram between hosts and across networks. The Internet Layer is also responsible for routing of IP datagram. The main protocols included at Internet layer are IP (Internet Protocol), ICMP (Internet Control Message Protocol), ARP (Address Resolution Protocol), RARP (Reverse Address Resolution Protocol) and IGMP (Internet Group Management Protocol).[2]

3. **Transport Layer**

Transport Layer is the third layer of the four layers TCP/IP model. The position of Transport Layer is between Application Layer and Internet Layer. The purpose of Transport Layer is to allow devices on the source and destination hosts to carry on a conversation. Transport Layer defines the level of service and status of the connection used when transporting data.

The main protocols included at Transport layer are TCP (Transmission Control Protocol) and UDP (User Datagram Protocol). [2]

4. Application Layer

Application layer is the top most layer four layers TCP/IP model. Application Layer is present on the top of Transport Layer. Application Layer defines TCP/IP Application Protocols and how hosts programs interface with Transport Layer services to use the network.Application layer includes all the higher level protocols that is like DNS (Domain Naming System, HHTP(Hypertext Transfer Protocol),TELNET, FTP(File Transfer Protocol), TFTP (Trivial File Transfer Protocol), SNMP (Simple Network Management Protocol),SMTP (Simple Mail Transfer Protocol) , DHCP (Dynamic Host Configuration Protocol), X Windows RDP (Remote Desktop Protocol) etc...[2]

2.3) TCP/IP Protocol Suite

Protocols which are defined for application layer serve the top programmers. Above these; there are programmers that the user interact directly or the programmers that provide to reach the computer's source to the other users.

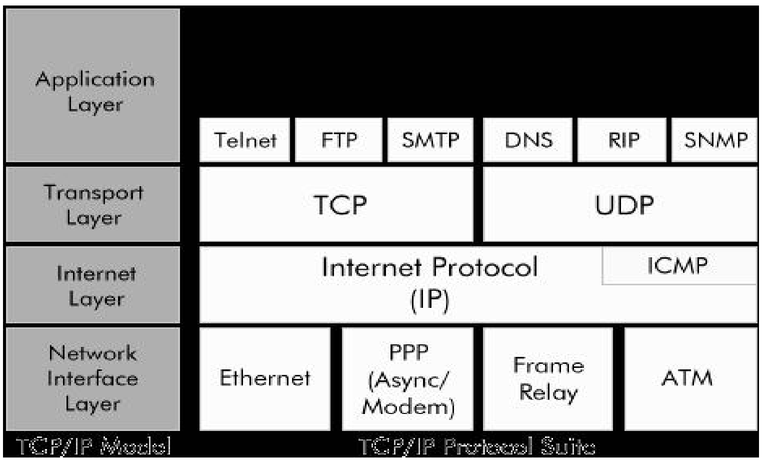


Figure2.3: TCP/IP Model and TCP/IP Protocol Suite

2.3.1) TCP (Transmission Control Protocol)

TCP (Transfer Control Protocol) is top layer. TCP (Transfer Control Protocol) is used for transmission of data from an application to the network.TCP is responsible for breaking data into IP packets before they sent and they assembling the packets when they arrive**.**

SMTP (Simple Mail Transport Protocol): SMTP is the standard protocol used to exchange Internet mail between TCP/IP hosts. Electronic mails on the Internet standard protocol that allows the purchase and shipping. Between the SMTP e-mail servers on the Internet from any computer and access to the e-mail server provides mail.

The SMTP design is based on the following model of communication as the result of user mail request the sender SMTP established a two way transmission channel to a receiver –SMTP.

The receiver-SMTP may be either the ultimate destination or an intermediate. SMTP commands are generated by the sender-SMTP and sent to the receiver-SMTP. SMTP replies are sent from the receiver-SMTP to the sender-SMTP in response to the commands. [3]

Once the transmission channel is established, the SMTP-sender sends a mail command indicating the sender of the mail. If the SMTP-receiver can accept mail it responds with an OK reply. The SMTP-sender then sends a RCPT command identifying a recipient of the mail. If the SMTP-receiver can accept mail for that recipient it responds with an OK reply; if not, it responds with a reply rejecting that recipient but not the whole mail transaction). The SMTP-sender and SMTP-receiver may negotiate several recipients.

When the recipients have been negotiated the STMP-sender sends the mail data, terminating with a special sequence. If the SMTP-receiver successfully processes the mail data it responds with an OK reply. [3]

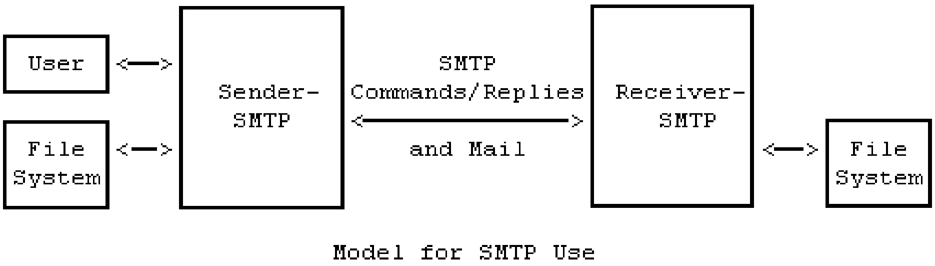


Figure2.4 : SMTP Model

TELNET: A TELNET is a form of remote connection. The user is on a system which connects the other system, it also provides to connect, as if like its own terminal. A TELNET connection is a Transmission Control Protocol (TCP) connection used to transmit data with interspersed TELNET control information. The TELNET Protocol is built three main ideas: first, the concept of a Network Virtual Terminal"; second, the principle of negotiated options and the third a symmetric view of terminals and process. [4]

Also Telnet means to establish a connection with the Telnet protocol, either with command line client or with a programmatic interface. When a TELNET connection is first established, each end supposed to originate and terminate at a "Network Virtual Terminal" (NVT). An NVT is an imaginary device which provides a standard, network wide. This eliminates the need for "server" and "user" hosts to keep information about the characteristics of each other terminals and terminal handling conventions. [4] For example; if you want to change your password, Telnet to the server log in and run the password command.

FTP (File Transfer Protocol): FTP provides to send file transfer from one computer to other computer. It is a basic protocol which is used for file transfer.FTP or File Transfer Protocol is used to transfer data from one computer to another over the internet or through a network. Specifically, FTP is a commonly used protocol for exchanging files over any network that supports the TCP/IP Protocol. There are two computers involved in a FTP transfer: a server and a client. [5]

The FTP server, running FTP server software, listens on the network for connection requests from other computers. The client computer, running FTP client software, initiates a connection to the server. Once connected, the client can do a number of file manipulation operations such as uploading files to the server, download files from the server, rename or delete files on the server and so on. [5]

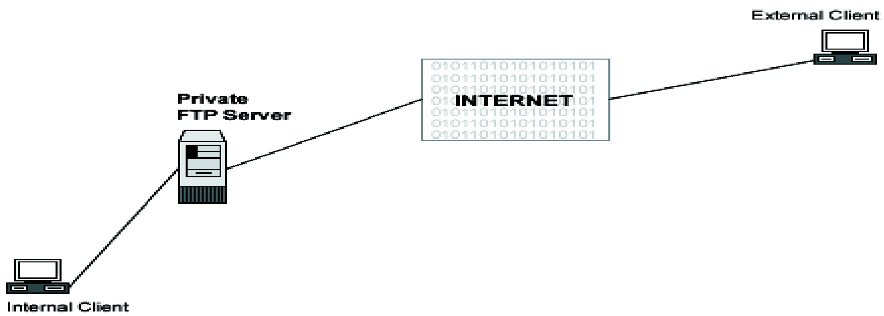


Figure2.5. FTP SHEMA

NNTP (Network News Transport Protocol): The Network News Transfer Protocol (NNTP) is an application protocol used for transporting Usenet news articles (net news) between new servers and is used for reading and posting articles by end user client applications. As local area networks and Internet participation proliferated, it became desirable to allow newsreaders to be run on personal computers connected to local networks. Because distributed file systems were not yet widely available, a new protocol was developed based on the client-server model. It resembles the Simple Mail Transfer Protocol (SMTP). [6]

NNTP operates over any reliable bi-directional 8-bit-wide data stream channel. When the connection is establish, the NNTP server host must send a greeting. The client host and server host then exchange commands and responses until the connection is closed or aborted. If the connection used is TCP, then the server host starts the NNTP service by listening on a TCP port. When a client host wishes to make use of the service, it must establish a TCP connection with the server host by connecting to that host on the same port on which the server is listening. [6]

HHTP (Hypertext Transfer Protocol): Short for Hyper Text Transfer Protocol, the underlying protocol used by the World Wide Web. HTTP defines how messages are formatted and transmitted, and what actions Web servers and browsers should take in response to various commands.

For example, when you enter a URL in your browser, this actually sends an HTTP command to the Web server directing it to fetch and transmit the requested Web page. The other main standard that controls how the World Wide Web works is HTML, which covers how Web pages are formatted and displayed. HTTP is called a stateless protocol because each command is executed independently, without any knowledge of the commands that came before it. This is the main reason that it is difficult to implement Web sites that react intelligently to user input.

2.3.2. (UDP) User Datagram Protocol

The User Datagram Protocol is the one of core members of the Internet Protocol Suite, the set of Network Protocols used for the Internet. Computer applications with UDP can send messages in this case referred to as data grams to other hosts on the Internet Protocol.

UDP uses a simple transmission model with a minimum of Protocol mechanism. It also provides checksums for data integrity and port numbers for addressing different function at the source and destination of the datagram.

UDP is suitable for purposes where error checking and correction is either not necessary or performed in the application, avoiding the overhead of such processing at the network interface level. Time-sensitive applications often use UDP because dropping packets is preferable to waiting for delayed packets, which may not be an option in a real-time system.

If error correction facilities are needed at the network interface level, an application may use the Transmission Control Protocol (TCP) or Stream Control Transmission Protocol (SCTP) which are designed for this purpose.

SNMP (Simple Network Management Protocol): SNMP is used for the devices which are inside the net; router, key, HUB etc…Network devices that have the SNMP supporting can be directed by SNMP messages from a distance**.** For this, there must be a SNMP port (SNMP Agent) in the devices.

DNS (Domain Name System: The Domain Name System (DNS) is a distributed naming system for computers, services, or any resource connected to the Internet or a private network.DNS allows a domain name to be used as a pseudonym for a specific IP address.

Most prominently, it translates domain names meaningful for users to the numerical IP addresses needed for the purpose of locating computer services and devices worldwide. [1]

2.4. OSI (Open System Interconnection)

OSI Layer Model is a creation defined by international organization for standards and OSI stands for Open Systems Interconnection.

OSI Model includes seven different layers. A layer is an assortment of theoretically comparable functions that offer services to the layer over it and obtains services from the layer below it. [1]

OSI Layer Model offering a framework for networking which employ protocols in seven layers. The processing control exceed from one layer to next layer and this process continue till the end .The processing start from bottom layer and then over the channel to further station and backing the hierarchy.

The OSI model layer consists of seven layers and each layer interacts with each other. The layer one and two called media layer and layer 3, 4, 5, 6, and 7 called host layers. OSI layer model is classified into seven categories discussed in detail under. [1]

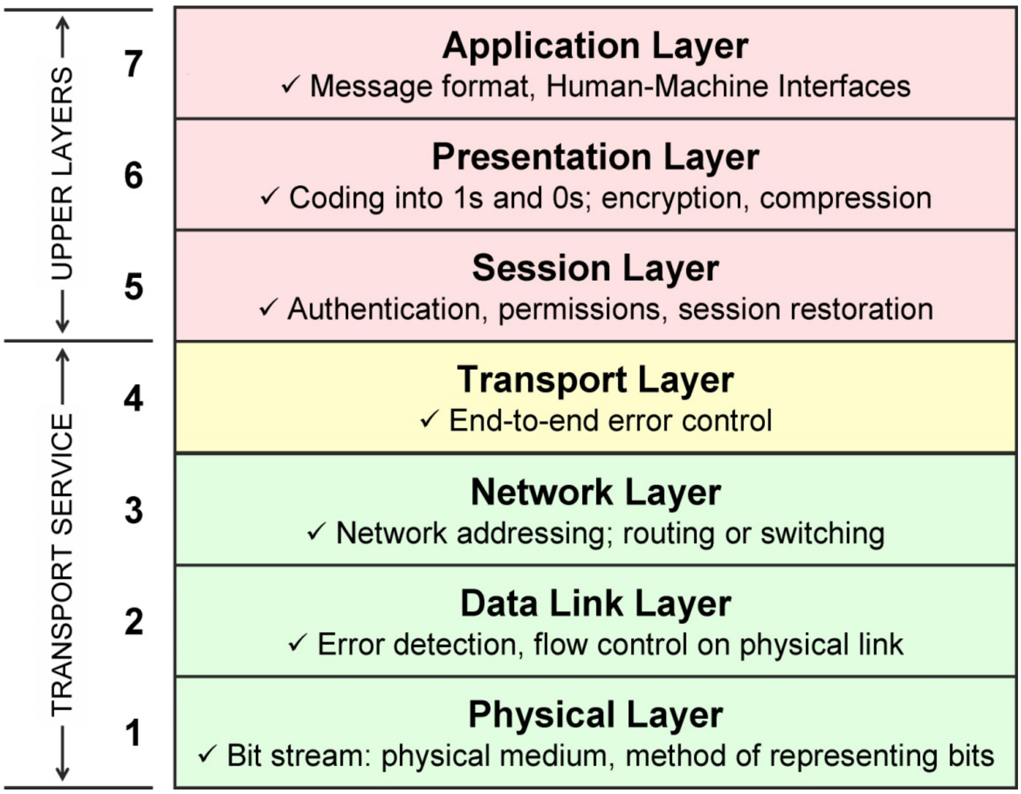


Figure2.6: OSI Model

1. **OSI Physical Layer:** OSI Physical Layer is responsible for media, signal and binary communication.OSI Physical Layer describes the physical and electrical stipulations for devices in depth it identify the relationship among physical medium and devices such as bus adopters, repeaters, hubs, cables, pins, voltages and network adapters etc...[10] The functionality of OSI Physical Layer contrast with the OSI Data Link Layer as physical layer is a primarily with the communication of a particular device with a standard while data link layer deals at last two or multiple devices. OSI Physical layer contains cables, cards, and various physical features for data carrier such as protocol, ATM, RS232, and Ethernet. [2]

2. **Data Link Layer:** OSI Data Link Layer provides Physical addressing. OSI Data Link Layer gives procedural and functional resources for broadcasting of data among networks. It also identifies errors of physical layer and tries to correct them. The main purpose of Data Link Layer is handled point to multi point and point to point media.
3. **OSI Network Layer:** OSI Network Layer is used for logical addressing as virtual circuits which are used to transmit data from node to node and determination of Path. OSI Network Layer is also offering routing and switching technologies. The error handling, packet sequencing, internetworking, addressing, and congestion control are the main functionality of Network layer. It also provides best quality of service on the request of transport layer.[2]
4. **OSI Transport Layer:** OSI Transport Layer provides connections from end to end flow control data, and reliability of transmit data. OSI Transport Layer can maintain path of the section and resend those that fail. The most common example of Transport layer is Transmission Control Protocol (TCP) and User Datagram Protocol (UDP).The working of OSI Transport Layer is just like a post office which deals lots of mail. Transport layer. The SPX, TCP/IP's, DNS are examples of implemented protocols on this layer. [2]
5. **OSI Session Layer:** OSI Session Layer deals with Inter host communication. It is responsible to manage, establish and conclude the link among applications. Through OSI Session layer the setting up of new connection can be handled, if needed conversation terminated, and exchanging of dialogue between the applications at every end.

OSI Session Layer also administers session and link coordination. The SAP, and TCP/IP remote procedure call are the examples of implemented protocols.

6. **OSI Presentation Layer:** OSI Presentation Layer is providing data representation, convert plain text into code as encryption and decoding of data. OSI Presentation Layer offering liberty from compatibility troubles therefore it is also called syntax layer. It also set up a perspective among application layer entities. OSI Presentation Layer decoded data demonstration from application to network format and vice versa.
7. **OSI Application Layer:** OSI Application Layer is responsible for network process to application. It supports various types of applications and end user procedures. OSI Application Layer identifies the communication associate, Excellency of service, user verification, privacy, and restraint of data syntax. Application Layer also offering various services such as file transformation, e-mail, and network software services. It contains Telnet and FTP and also includes Tiered application architectures. The well known examples of OSI model layers are web browsing, SAP, SMTP, TCP/IP, and NFS. [2]

2.5 IP Protocol

IP (Internet Protocol) is the primary network protocol used on the internet and it is developed in 1970. IP is often used together the Transport Control Protocol (TCP) on the internet and many other networks. The Internet Protocol contains a set of related and among the most widely used network protocol.

Besides Internet Protocol (IP) itself, higher level protocols like that TCP, UDP, HTTP, and FTP all integrate with IP to provide additional capabilities. The lower level Internet Protocols like that ARP and ICMP. These higher level protocols interact more closely with applications like Web browsers while lower level protocols interact with network adapters and other computer hardware. IP specific the format of packets are called data grams, and the addressing scheme. Most networks combine IP with a higher level protocol is called Transmission Control Protocol (TCP), which establishes a virtual connection between a destination and source. [9]

IP (Internet Protocol) is something like the postal system by itself. It allows you to address a package and drop it in the system, but there is not direct link between you and the recipient. TCP/IP, on the other hand, establishes a connection between two hosts so that they can send messages back and forth for a period of time. [9] The current version of IP is IP4 and a new version is called IPV6. The data has been on the Internet Protocol network and is organized into packets. Each IP packet includes both a header so that specifies source, destination, and other information about the data and message data itself. IP functions at layer 3 of OSI model. It can therefore top of different data link interface including Ethernet and WI-FI.

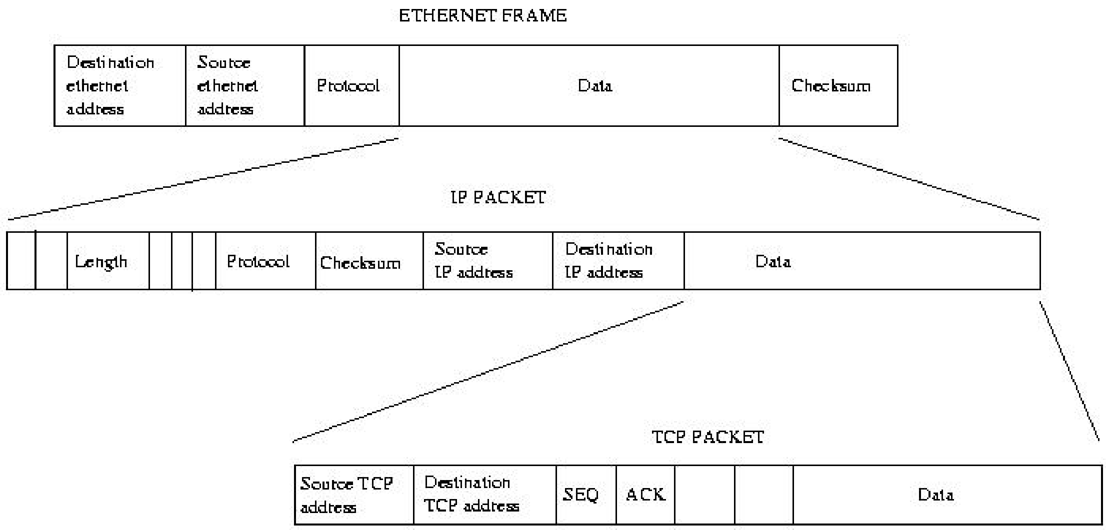


Figure 2.7: IP PROTOCOL LAYERS

a. Ethernet

A connection-oriented network is X.25, which was the first public data network. It was deployed in the 1970s at a time when telephone service was a monopoly everywhere and the telephone company in each country expected there to be one data network per country theirs. To use X.25, a computer first established a connection to the remote computer, that is, placed a telephone call. This connection was given a connection number to be used in data transfer packets (because multiple connections could be open at the same time). Data packets were very simple, consisting of a 3-byte header and up to 128 bytes of data. The header consisted of a 12-bit connection number, a packet sequence number, an acknowledgement number, and a few miscellaneous bits. X.25 networks operated for about a decade with mixed success. [10]

b. Frame Relay

Frame Relay is a standardized wide area network technology that specifies the physical and logical link layers of digital telecommunications channels using a packet switching methodology. Originally designed for transport across Integrated Services Digital Network (ISDN) infrastructure, it may be used today in the context of many other network interfaces. Network providers commonly implement Frame Relay for voice (VOFR) and data as an encapsulation technique, used between local area networks (LANs) over a wide area network (WAN). Each end-user gets a private line (or leased line) to a Frame Relay node. The Frame Relay network handles the transmission over a frequently-changing path transparent to all end-user extensively-used WAN protocols. It is less expensive than leased lines and that is one reason for its popularity. The extreme simplicity of configuring user equipment in a Frame Relay network offers another reason for Frame Relay's popularity. [11]

c. ATM (Asynchronous Transmission Model)

ATM was designed in the early 1990’s.ATM was going to solve all the world’s networking telecommunication problems by merging voice ,data, cable television ,telegraph everything else into a single integrated system that could do everything for everyone. It did not happen. In large part, the problems were similar to those we described earlier concerning OSI, that is, bad timing, technology, implementation.[11] ATM was more successful than OSI, and it is now widely used within the telephone system, often for moving IP Packets.

2.6. IP Network

Networks provide communication between computing devices.

To communicate properly, all computers (hosts) on a network need to use the same communication protocols. An Internet Protocol network is network of computer using Internet Protocol for their communication Protocol. All computers within an IP network must have an IP address that uniquely identifies that individual host. An Internet Protocol-based network (an IP Network) is a group of hosts that share a common physical connection and that use Internet Protocol for network layer communication. [11]

At IP based Networks while the packet switching networks are used but at traditional phone systems circuit switching is used. A circuit switching network, for providing communication forms closed at two crucial points, between networks. Setting connection is separated for communication between two crucial points. The most important problem here is the capacity becomes free because the separated circuit is not used in whole time.

Also; during the communications if there is an error in circuit than all connection cancelled and re establish the new one. For IP based network, the packet switching network technologies are used for using available capacity faster and efficient and for minimizing the connection risks.

The messages which are sent on a packet switching network firstly divided according to their sending address. Then each packet is sent by stating to the location on the network.

A packet does not need to direct on the same links like the other direct on the same links like the other packets.

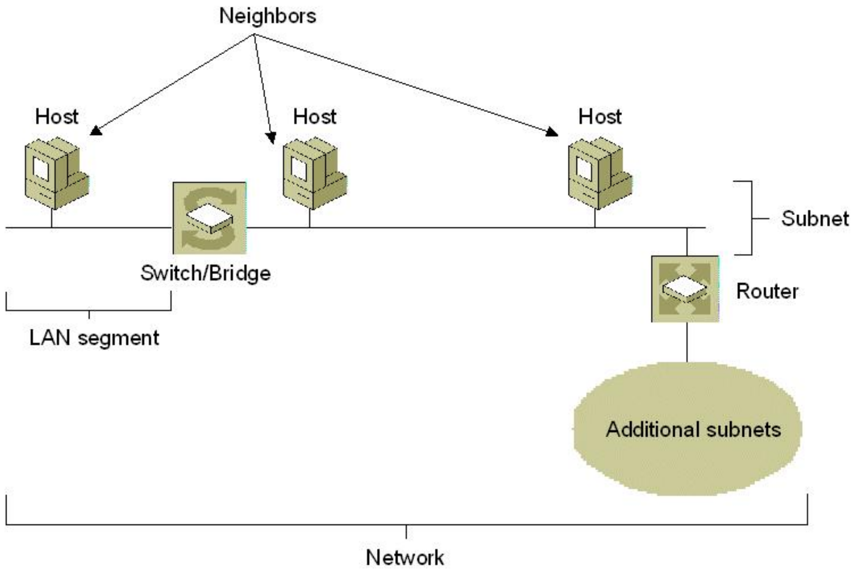


Figure 2.8: IP Network

Node: Any device, including routers and hosts, which run an implementation of IP.

Router: A node that can forward IP packets not explicitly addressed to itself. On an IPv6 network, a router also typically advertises its presence and host configuration information.

Host: A node that cannot forward IP packets are not explicitly addressed to itself (a non-router). A host is typically the source and the destination of IP traffic. A host silently discards traffic that it receives but that is not explicitly addressed itself.

Upper-layer protocol: A protocol uses IP as its transport. Examples include Internet layer protocols such as the Internet Control Message Protocol (ICMP) and Transport layer protocols such as the Transmission Control Protocol (TCP) and User Datagram Protocol (UDP). However, Application layer protocols that use TCP and UDP as their transports are not considered upper-layer protocols. File Transfer Protocol [FTP] and Domain Name System [DNS]

LAN segment: A portion of a subnet consisting of a single medium that is bounded by bridges or Layer 2 switches.

Subnet: One or more LAN segments that are bounded by routers and use the same IP address prefix. Other terms for subnet are network segment and link.

Network: Two or more subnets connected by routers. Another term for network is internetwork.

Neighbor: A node connected to the same subnet as another node.

Interface: The representation of a physical or logical attachment of a node to a subnet. An example of a physical interface is a network adapter. An example of a logical interface is a tunnel interface that is used to send IPv6 packets across an IPv4 network.

Address: An identifier that can be used as the source or destination of IP packets and that is assigned at the Internet layer to an interface or set of interfaces.

Packet: The protocol data unit (PDU) that exists at the Internet layer and comprises an IP header and payload. [11]

2.7 IP Address

An IP address is an identifier for computer or device on TCP/IP network. Networks using TCP/IP Protocol route message based on the IP Address of the destination. In other words; an IP address is a private number online devices use to identify and communicate with each through computer network. The format of an IP address is a 32-bit numeric address written as four numbers separated by periods. Each number can be zero to 255. For example, 1.160.10.240 could be an IP address. [12]

You can assign IP Address at random as long as each one is unique with an isolated network. However, connecting to private network to the internet requires using registered IP address. An IP address can be static or dynamic. A dynamic address is a temporary address that is assigned each time a computer or device accesses the internet.

2.7.1 IP Number

An IP address is a unique global address for a network interface. Dynamically assigned IP addresses (DHCP) and IP addresses in private networks (NAT).

An IP address is a 32 bit long .The network prefix identifies a network and the host number identifies a specific host (actually, interface on the network).

IP addresses are written in a so-called dotted decimal notation, and also each byte is identified by a decimal number in the range (0-255).[13]

2.7.1.1. Finding Special IP Address

Reserved or (by convention) special addresses: Loopback interfaces is all addresses 127.0.0.1-127.0.0.255 are reserved for loopback interfaces. Most systems use 127.0.0.1 as loopback address loopback interface is associated with name “local host” IP address of a network is host number is set to all zeros, e.g., 128.143.0.0 Broadcast address is host number is all ones, e.g., 128.143.255.255. Broadcast goes to all hosts on the network it is often ignored due to security concerns. Test and Experimental addresses is certain address ranges are reserved for “experimental use”. [13]

2.7.2 IP Datagram

IP datagram is the "envelopes" that carry data across IP networks. Datagram is assembled by the source computer and sent out on the network. Routers transfer the datagram from one network to another. To traverse a particular network, datagram is encapsulated within the frames of that network. [14]

Bit 0 16 31

<table>
<tr><td>Version</td><td>IHL</td><td>Type of service</td><td colspan="2">Total length</td></tr>
<tr><td colspan="3">Identification</td><td>Flags</td><td>Fragment offset</td></tr>
<tr><td colspan="2">Time to live</td><td>Protocol</td><td colspan="2">Header checksum</td></tr>
<tr><td colspan="5">Source address</td></tr>
<tr><td colspan="5">Destination address</td></tr>
<tr><td colspan="5">Options/padding</td></tr>
</table>

Table 2.1: IP DATAGRAM

Explain of IP Datagram:

Version: The version number of the protocol**.**

IHL (Internet header length): Length of the header**.**

Total length: The total length of the datagram, including the header**.**

Identification: If a datagram is fragmented, this field contains a value that identifies a fragment as belonging to a particular datagram.

Flags: DF (Don't Fragment) or MF (More Fragments). DF indicates that the datagram should not be fragmented and is used when attempting to discover the maximum packet size for networks. MF indicates that this is not the last fragment.

Fragment offset: Where the datagram fragment belongs in the set of fragments. Time of live a counter that is decremented with every pass through a router. When 0, the datagram is assumed to be in a loop and is discarded.

Protocol: Identifies the transport layer process to receive the datagram.

Header checksum: An error detection feature that indicates to the receiver whether a packet has been corrupted**.**

Source address: The IP address of the host sending the datagram.

Destination address: The IP address of the host to receive the datagram.

Options/padding: Optional information and filler to ensure the header is a multiple of 32 bits. [14]

2.7.3. IP Addressing and Host Name

The previous discussion describes how IP delivers datagram over router-connected networks. This section describes the other important component of IP: the addressing scheme. In reality, there are multiple addressing and naming schemes in use on a typical IP network at any one time. For example, there are host naming schemes (as opposed to numbering schemes) that allow humans to refer to computers with easy-to-remember names. The Internet's DNS (Domain Name Service) provides a service that translates names into IP addresses. Refer to "DNS (Domain Name Service)" for more information about Internet naming schemes. There is also the IP addressing scheme, which consists of both a network identifier and a host identifier. [15]

2.7.4 DNS (Domain Name System)

The Domain Name System (DNS) is a distributed naming system for computers, services, or any resource connected to the Internet or a private network.DNS allows a domain name to be used as a pseudonym for a specific IP address. Most prominently, it translates domain name meaningful for users to the numerical IP addresses needed for the purpose of locating computer services and devices worldwide. By providing a worldwide, distributed keyword-based redirection service, the Domain Name System is an essential component of the functionality of the Internet. [2] When you type in a web site name, your system looks up the name on an assigned DNS server and resolves it to its IP address. It can then access the web site.

2.8. IP Address Structure

IP stands for Internet protocol, and its primary purpose is to enable communications between networks. As a result, a 32-bit IP address actually consists of two parts:

The network ID (or network address): Identifies the network on which a host computer can be found

The host ID (or host address): Identifies a specific device on the network indicated by the network ID. Most of the complexity of working with IP addresses has to do with figuring out which part of the complete 32-bit IP address is the network ID and which part is the host ID, as described in the following sections. [12]

IP addresses are usually represented in a format known as dotted decimal notation. In dotted-decimal notation, each group of eight bits an octet is represented by its decimal equivalent. For example, consider the following binary IP address:

11000000101010001000100000011100

To convert this value to dotted-decimal notation, first divide it into four octets, as follows:

11000000 10101000 10001000 00011100

Then, convert each of the octets to its decimal equivalent:

11000000 10101000 10001000 00011100

192 168 136 28

Then, use periods to separate the four decimal numbers, like this:

192.168.136.28 [13]

Table 2.3 uses x, y, z to designate four octets values in any given IP address.

The table used to show the following:

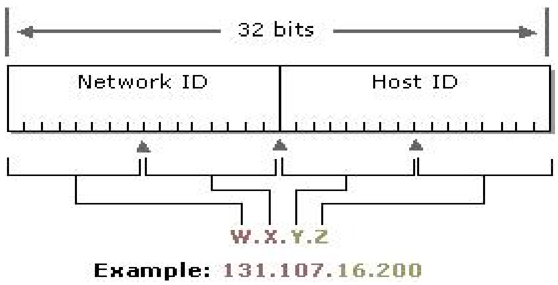


TABLE 2.2: IP ADRESS STRUCTURE

2.8.1. IP Address Class

In the early days of the Internet, the 32-bit IP address space was allocated into three address classes: class A, class B, and class C. As discussed later, the class system would be all but phased out by now except that so many organizations "own" class-based blocks of addresses and many will not voluntarily give them up. Also, the changeover has been difficult.

Rule	Minimums and maximums	Decimal range
Class A: First bit is always 0.	**0**0000000 = 0 **0**1111111 = 127	1 - 126* * 0 and 127 are reserved.
Class B: First two bits are always 10.	**10**000000 = 128 **10**111111 = 191	128 - 191
Class C: First three bits are always 110.	**110**00000 = 192 **110**11111 = 223	192 - 223
Class D: First four bits are always 1110.	**1110**0000 = 224 **1110**1111 = 239	224 - 239

Table 2.3: IP ADDRESS CLASSES

Class A : Identified by the first bit set as 0. The next 7 bits define the network address, and the remaining 24 bits identify hosts. Network number 127 is reserved for loopback testing. The 24-bit host address space identifies 16,777,214 hosts per each of the 126 networks. Most class A network schemes were assigned to U.S. government agencies, educational institutions, research organizations, and large companies in the early days of the Internet.

Class B : Identified by the first 2 bits set as 10. The next 14 bits define the network address, and the remaining 16 bits identify hosts. This scheme defines 16,384 networks and 65,534 hosts per network.

Class C : Identified by the first 3 bits set as 110. The next 21 bits define the network address, and the remaining 8 bits identify hosts. This scheme defines 2,097,152 networks and 254 hosts per network.

2.8.2. Private IP Address Classes

A private IP addressing scheme allows an organization to use any IP internal addressing scheme (class and subnet scheme) that fits it requirements. Any devices connected directly to the Internet (Web servers, e-mail servers, etc.) require a public IP network address, which can be obtained from a network registrar.[12]

A proxy server or NAT (network address translation) server separates the internal and external networks and acts as a "gateway" between them. What these servers do is intercept outgoing packets and change the private IP address to a public IP address. When a response to the packets comes back, the servers convert the public IP address back to the appropriate private IP address.

- Class A: 10.0.0.0 to 10.255.255.255
- Class B: 172.16.0.0 to 172.31.255.255
- Class C: 192.168.0.0 to 192.168.255.255

2.8.3. The Subnet Mark

A subnet mask is an IP address feature that serves as a sort of template to indicate which bits in the IP address define the network and which bits define the host. All devices on the same IP network must use the same subnet mask. The subnet mask became necessary when subnet procedures (described next) were developed for IP addresses. The standard subnet masks used for the class A, B, and C networks are shown in the following table, along with the binary equivalent. [15]

Class	**Subnet Mask (Decimal)**	**Subnet Mask (Binary)**
Class A	255.0.0.0	11111111 00000000 00000000 00000000
Class B	255.255.0.0	11111111 11111111 00000000 00000000
Class C	255.255.255.0	11111111 11111111 11111111 00000000

Table 2.4: IP CLASSES AND THE SUBNET MARK

Note how the binary 1s indicate the bits that are used for the network address portion of the IP address. They essentially "mask out" the network address to reveal the host address. As an example, a class B address of 128.10.50.25 and a class B subnet mask of 255.255.0.0 are shown in the following table. The mask indicates that the first two bytes are the network address, so the last two bytes are the host address.[15]

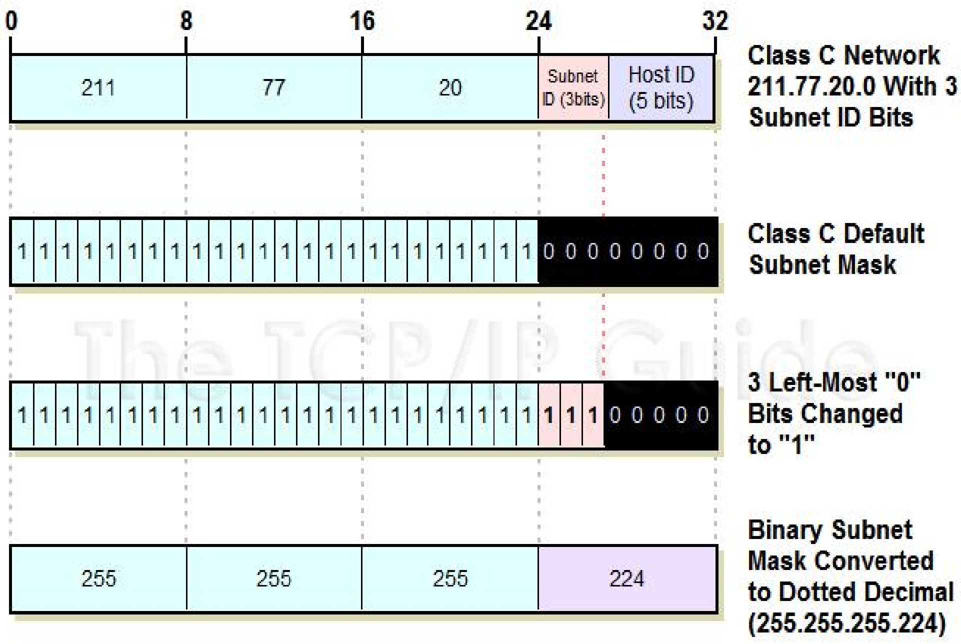


Table 2.5: THE SUBNET MARK

2.8.4. IP Address Version

IPV1: The first version for IP address.

IPV2: The second version for IP address

IPV3: The third version for IP address

IPV1-3 is used for defined and replaced.

IPV4: The fourth version for IP address.

IP v4 is current version

IPV5: The fifth version for IP address

IP v5 is uses to stream protocol.

IPV6: The sixth version for IP address

IP v6 is replacement for IP v4

IPV6 is during development.

2.8.5. IPV4 ADDRESS

The octets have an important role in the IP address and they divide the IP address into classes. They are split into two sections net and host. The net section always starts with the first octet and is used to define the network the machine belongs. The host section defines the actual machine in the network and always contains the last octet.

Internet Protocol Version 4 (IPv4) was the first publicly used version of the Internet Protocol. IPv4 addresses are typically displayed as four numbers, each in the range 0 to 255, or 8 bits per number, for a total of 32 bits. Thus IPv4 provides an addressing capability of 2^{32} or approximately 4.3 billion addresses.[15]

2.8.5.1. CLASSES OF IPV4

Class A

The class A IP address has a first or net octet in the range of 1 and 126. The other three octets define the hosts. The A class network allows for a total of 2,147,483,648 unique IP addresses and is mainly used for the network of a very large corporation.[15]

Class B

The class B IP address uses the first two octets as network identifiers, and the last two as host identifiers. The first octet in the class B IP addresses is in the 128 to 191 range. IPs from this class is most commonly used by mid-sized networks, such as college campuses. [15]

Class C

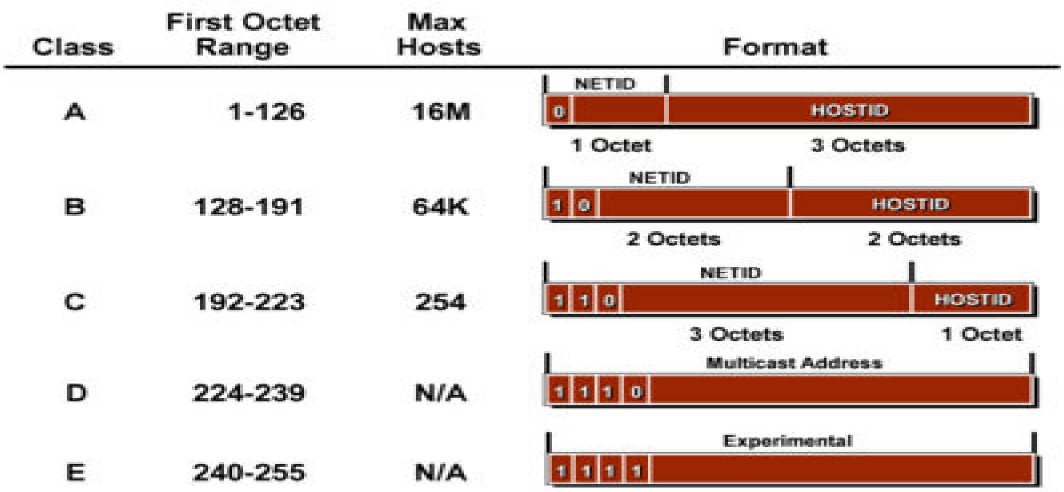
For this class, the Net identifier is composed of the first 3 octets and the first octet is in the 192 to 223 range. The last octet is used to identify the host. With a limited number of hosts, this IP class is suitable for small to mid-sized networks.[15]

Class D

IP addresses from this class are mostly used for multicasting. They are in the 224.0.0.0 to 239.255.255.255 range. The 224.0.0.0 to 224.0.0.255 range is used only for local area network (LAN) multicasting. [15]

Class E

IP addresses in this class have the first octet in the 240-255 range. They are reserved for experimental usage and computers, trying to use them will not be able to communicate properly online.[15]



Class	First Octet Range	Max Hosts	Format
A	1-126	16M	0 NETID (1 Octet) HOSTID (3 Octets)
B	128-191	64K	1 0 NETID (2 Octets) HOSTID (2 Octets)
C	192-223	254	1 1 0 NETID (3 Octets) HOSTID (1 Octet)
D	224-239	N/A	1 1 1 0 Multicast Address
E	240-255	N/A	1 1 1 1 Experimental

TABLE2.6: IPv4 classes

2.8.6. IPV6 ADDRESS

Internet Protocol version 6 (IPv6) is the latest version of the Internet Protocol (IP), the communications. IPv6 was developed by the Internet Engineering Task Force (IETF) to deal with the long-anticipated problem of IPv4 address exhaustion. [15]

2.8.6.1. THE KINDS OF IPV6

Unique IPv6 Addresses identify a single network interface. Multicast (Multicast) Addresses define a group Packets sent to all interfaces that are included in the group (Any cast) Addresses define a group created different interfaces. Packet is transmitted only members of the group nearest.

IPv6 uses a 128-bit address, allowing for 2^{128}, or approximately 3.4×10^{38} addresses, or more than 7.9×10^{28} times as many as IPv4, which uses 32-bit addresses. IPv4 allows for only approximately 4.3 billion addresses. The two protocols are not designed to be interoperable, complicating the transition to IPv6. [15]

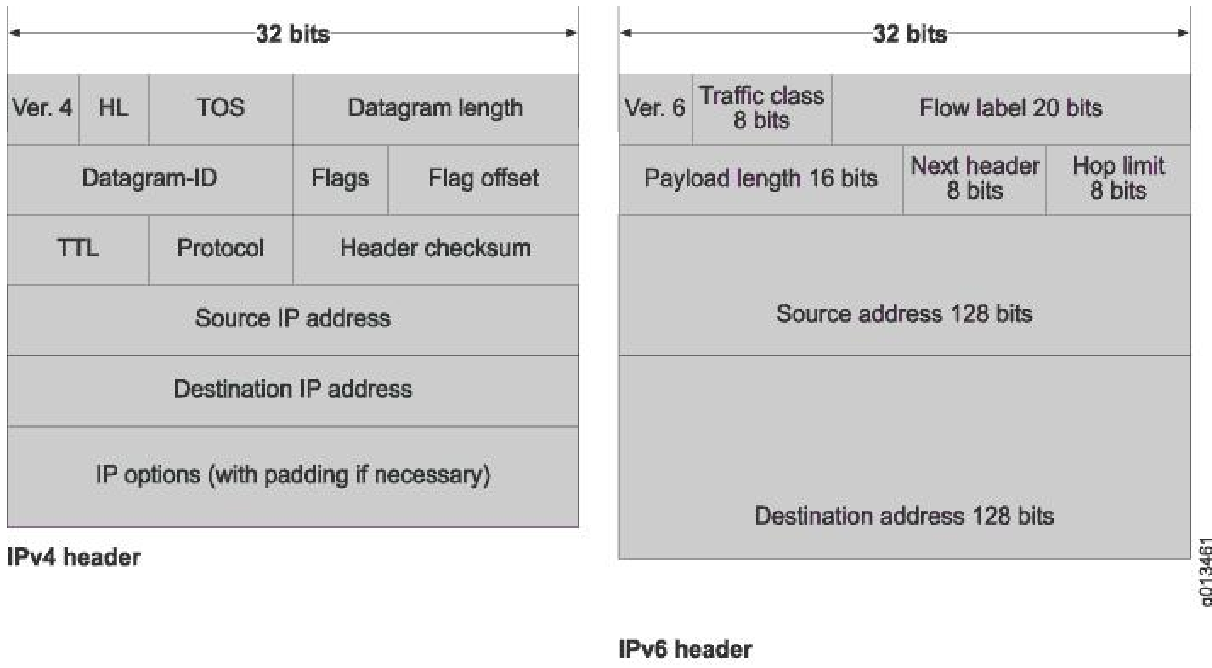


Table 2.7:IPV6

3. VOIP (VOICE OVER INTERNET PROTOCOL)

3.1 VOIP (Voice over Internet Protocol)

VoIP is a technology that allows telephone calls to be made over computer networks such as the Internet. VoIP converts analog voice signals into digital data packets and supports real-time, two ways transmission of conversations using Internet Protocol (IP). VoIP calls can be made on the Internet using a VoIP service providers and standard computer audio systems.

Alternatively, some service providers support VoIP through ordinary telephones that use special adapters to connect to a home computer network. Many VoIP implementations are based on the H.323 technology standard. [16]

VoIP PBX (Voice over Internet Protocol Private Branch Exchange) phone systems are communication systems that use the Internet instead of telephone lines to facilitate calls, messages, voice mails and facsimiles. It is able to transform all communication data into a digital format so that it can be sent and received through the Internet. The data conversion that Internet based phone systems will allow you to exchange information with customers that are still utilizing traditional phone systems. Communication that is transmitted over the Internet reaches the intended person faster than one that is sent via telephone lines and phone networks. [16]

VOIP telephone also known as a SIP phone or a soft phone allows the user to make phone calls to any soft phone mobile or landline by using VOIP.

This way the voice is carried through the internet instead of the traditional PSTN system. [17]

Voice over IP features some additional advantages including:

- If you want, you can talk to many people at once .
- It is used to send other types of data other than voice (such as files) .
- Innovations in data transfer and Internet speeds come about, it can only get better.

There are a variety of devices that can take VoIP calls. Some aren't even hardware at all, but software programs you can install on your computer.
VoIP offers a substantial cost savings over traditional long distance telephone calls. The main disadvantage of VoIP is a greater potential for dropped calls and degraded voice quality when the underlying network links are under heavy load.

3.2.COMPONENTS OF VOIP

The mechanism of VOIP requires basic components to be configured.

These components are categorized as follows:

- Codec
- Transmission Control Protocol/ Internet Protocol (TCP/IP) or VOIP Protocols
- IP Telephony server or PBXs
- VOIP gateways or soft-phones

a. Codec

A Codec can either mean compressor-de compressor or coder-decoder. This could be hardware or software with a purpose of performing transformations on data streams or signal from analog to digital and vice versa so that it can be transmitted over a networked interconnection. [18]

b. TCP/IP or VOIP Protocol

Protocols that are used to transmit voice signals over IP network are generally referred as VOIP or VOIP protocols. When making a call on the VOIP terminal application programs that are based at the higher level are used. These programs have to interact with lower levels of the TCP/IP stack. For the purpose of proving telephony services, there is a need that a number of different standards and protocols come together. [18]

When initiating and completing a call on a VOIP terminals into the network, protocol are required to facilitate call setup and streaming of voice.

These protocols are classified in two categories namely call setup protocols and voice streaming protocols.[18]

c. IP Telephony Servers and PBXs

A server is usually a computer running an application that manages the setup or connection of telephone calls between terminals.

It registers terminal’s IP addresses and stores them for the purpose of connecting calls. The server will receive call setup request messages, determine the status of destination devices, check the authorization of users to originate and/or receive calls, and create and send the necessary messages to process the call requests. The VoIP network requires a client - server topology where in this case IP PBX server is the main telephony server. An IP PBX is a private branch exchange (telephony switching system within an enterprise) that switches calls between VoIP users on local lines while allowing all users to share a certain number of external phone lines. The typical IP PBX can also switch calls between a VoIP user and a traditional telephone user, or between two traditional telephone users in the same way that a conventional PBX does. The abbreviation may appear in various texts as IP PBX with a conventional PBX, separate networks are necessary for voice and data communications.

One of the main advantages of an IP PBX is the fact that it employs converged data and voice networks. [18]

This provides flexibility as an enterprise grows, and canal so reduce long-term operation and maintenance costs. Like a traditional PBX, an IP PBX is owned by the enterprise. In VoIP systems, IP PBXs are normally built on a PC platform running on any operating system.

An example of an IP PBX is the Asterisk which is built and runs on Linux operating system. These IP PBXs provide functions and features equivalent to the traditional PBXs of the PSTN. These IP telephony servers can be clustered in a group and managed as a unit in order to increase scalability, reliability and redundancy.

H323 protocol uses the Gatekeeper to provide all admission (CAC) and other management functions such as address look up for multimedia services.[18]

d. VoIP Gateways, Routers And Switches

PSTN gateway interfaces between networks and IP networks or working as transition elements interworking with an expression that performs the functions of other modules.

A gateway, the packet H.323-compliant terminals on a switched network is a circuit-switched network other H.323 terminals or other real-time two-way traffic between a gateway a network that provides "end point" works. Other ITU H.310 terminals (B-ISDN), H.320 (ISDN), H.321 (ATM), H.322 (GQoS-LAN), H.324 (PSTN), H.324 (Mobile) or POTS may be terminals. Gateways are end points that make it possible to connect call between end points that would normally not inter operate. They usually translate from one signaling protocol to another such as from (SIP) Session Initiation Protocol and also translating of network addresses between different network addressing schemes.

The gateways make it possible to interface VOIP and traditional PBX. In order to move RTC voice datagram, you need to have VOIP gateways set. [18]

VOIP gateways provide a link between the VOIP network and the traditional PSTN network making it possible to make a call to telecommunication lines. The VOIP gateways use SS7 protocol to signal switches in the PSTN network.[17]

e. IP Phones and Soft phones

This is the end point of communication which is usually in form of hard phone or a soft phone. There are referred to as answering machines and they are referred to as answering machines and they are identified by an IP address which is capable of handling many terminals for the same purpose.[20] The one that is enabled first completes the call and others become disabled. From inception of VoIP, computers have been used as terminals although currently telephone adaptors and or VoIP telephones are available. [18]

3.3.PSTN (Public Switched Telephone Network)

The Public Switched Telephone Network (PSTN) has been evolving ever since Alexander Graham Bell made the first voice transmission over wire in 1876. [20] PSTN (Public Switched Telephone Network) is called general transmission telephone network. PSTN is used to make circuit transfer in the world wide. In the beginning, it is established as constant analog telephone network but nowadays it is completely digital. It includes constant telephone also includes mobile telephone line. PSTN usually has been served according to standards has been prepared by ITU-T.

a. Analog and Digital Signaling

Everything you hear, including human speech, is in analog form. Until several decades ago, the telephony network was based on an analog infrastructure as well. Although analog communication is ideal for human interaction, it is neither robust nor efficient at recovering from line noise. (Line noise is normally caused by the introduction of static into a voice network.) In the early telephony network, analog transmission was passed through amplifiers to boost the signal. [18]

b. Digital Voice Signals

PCM is the most common method of encoding an analog voice signal into a digital stream of 1s and 0s. All sampling techniques use the Nyquist theorem , which basically states that if you sample at twice the highest frequency on a voice line, you achieve good-quality voice transmission.

The PCM process is as follows:

• Analog waveforms are put through a voice frequency filter to filter out anything greater than 4000 Hz.

These frequencies are filtered to 4000 Hz to limit the amount of crosstalk in the voice network. Using the Nyquist theorem, you need to sample at 8000 samples per second to achieve good-quality voice transmission.

• The filtered analog signal is then sampled at a rate of 8000 times per second.

• After the waveform is sampled, it is converted into a discrete digital form.

This sample is represented by a code that indicates the amplitude of the waveform at the instant the sample was taken. The telephony form of PCM uses eight bits for the code and a logarithm compression method that assigns more bits to lower-amplitude signals.If you multiply the eight-bit words by 8000 times per second, you get 64,000 bits per second (bps). The basis for the telephone infrastructure is 64,000 bps (or 64 kbps).

Two basic variations of 64 kbps PCM are commonly used: µ-law, the standard used in North America; and a law, the standard used in Europe. The methods are similar in that both use logarithmic compression to achieve from 12 to 13 bits of linear PCM quality in only eight-bit words, but they differ in relatively minor details.[18]

3.3.1 VOIP/ PSTN

As with almost every industry, it is usually better and easier to acquire additional business from current customers than it is to go out and get new customers. [19]The PSTN is not different. Local Exchange Carriers (LECs) have been increasing the features. These services come in two common flavors: custom calling features and CLASS features. Custom calling features rely upon the end office switch, not the entire PSTN, to carry information from circuit switch to circuit-switch. CLASS features, however, require SS7 Connectivity to carry these features from end to end in the PSTN. [17]

The following list includes a few of the popular custom calling features commonly found in the PSTN today:

- Call waiting: Notifies customers who already placed a call that they are receiving an incoming call.
- Call forwarding: Enables a subscriber to forward incoming calls to a different destination.
- Three-way calling: Enables conference calling. [17]

3.4.The Communication Between PSTN And IP Network

The sound services are provided by PSTN and ISDN circuit switching networks until now. Circuit Switching networks during the period of calling are provided connection which allocated for users, but IP networks the sound turned to the data movement when the calling is made and it is delivered by any possible way which is on Internet or special networks, similar to e-mail. The packets are recollected by the receiver.

If the last users started and received the callings with a wide band, a computer or a telephone which connected to the networks and the callings can be directed to the other wide band subscribers who use software applications.

It is thought that the system is more active than the networks of circuit switching. There are kinds of forms in VOIP services. The way of classifying VOIP is according to the terminal structure which is connected with networks.

3.4.1. From phone to phone

A PSTN subscriber calls to another PSTN subscriber calls is considered. This is the same PSTN subscribers as well as network PSTN networks may be different. Traditional phones, which convert the phone signals into IP on the contrary they can be connected to IP Networks by Routers.

This kind of usage abolished the need of use with a computer.

3.4.2From computer to computer

The users can use VOIP with their computers, if the VOIP suitable communication software is set up in two computers. The two users must be online before setting up any connection. This kind of usage is occurred in traditional internet. Some special consumer equipments in this sense for classifying VOIP services are more similar than common phone.

3.4.3From phone to computer

Traditional phones for providing a calling on Internet with gateway that abolished the needs of computer connected to IP Network gives its users to meet with the users who connected to IP Network. Gateway compresses the sound traffic that gets from PSTN, and sends this, on IP Network and on the other hand sums and solves this traffic.

We can also classify the services according to the address forms. These are; PSTN phone numbers, internet address and some private number plans.

3.4.4 Mobile VOIP

At past, several VOIP solutions do not work with mobile phone networks. Last technological news permit the users either on mobile networks or IP W-LAN technology which makes sound calling. 2G Mobile Systems use basic circuit switching networks for transmitting sound services , already these systems replace with packet switching and IP directing 3G Systems. One of the main 3G standards code Division Multiple Access 2000, users Mobile IP which develops from kernel network architect. One of the other 3G standard W-CDMA its kernel architect contains both VOIP and IP multimedia systems that support wide band audio-lingual services. In some of countries the using of VOIP technology has been increased .For example; In U.S.A. The push to talk service starts with the technology of Verizon Wireless and Sprint PCS VOIP. One of the standards of 3G ; the TD-CDMA's Mobile VOIP Service's developing becomes FAT. By the time, some countries like Japan starts the open speech discussions about I MT-2000 technologies included TDD technology.

3.4.5 Wireless VOIP

There are developments in wireless VOIP area .For example; The IP technology which is used for sound transmitting united with wireless LAN (W-LAN).The IP phone which uses wireless technology and generally called WI-FI Telephone gets to develop recently but market is still small. For example; by VOIP provider VONAGE users portable WI-FI Phones enable to standard receive phone callings in the W-LAN access points. In recent years; the dual WI-FI/Mobile hand type devices are developed that they can transmit the sound on WI-FI and it can be used as an alternative of sound service. For example; Motorola and Texas instruments in past years study about on a dual mode sound transmitting and achieve to try the device that will provide this.

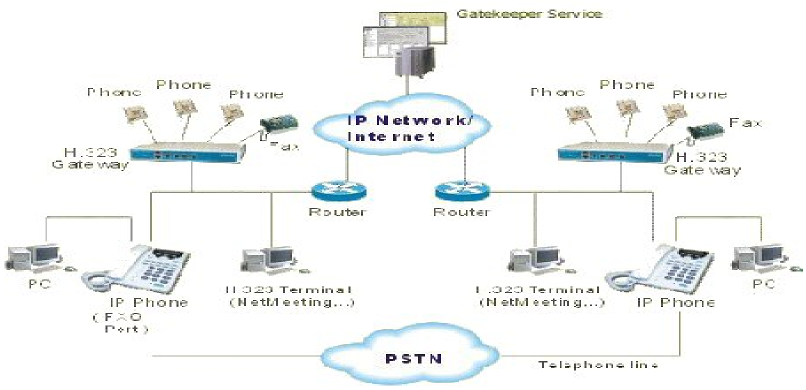


Figure3.1. VOIP diagram

3.5 BENEFITS OF VOIP (Voice over Internet Protocol)

Many of the benefits of VoIP (Voice over Internet Protocol) are derived from the use of Internet Protocol (IP) as the transport mechanism.

- Benefits of VOIP (Voice over Internet Protocol) include cost saving and single infrastructure savings and new applications.

- Using a packet telephony calls center versus a circuit-switched call center.

- Service provider prepaid calling card application**.**

4. IP PBX PHONE SYSTEM

4.1. IP PBX PHONE SYSTEM

A PBX is called "Private Branch Exchange" which is a private telephone network used within a company. A PBX (Private Branch Exchange) is a switch station for telephone systems. It consists mainly of several branches of telephone systems and it switches connections to and from them, thereby linking phone lines. [19]

The users of PBX phone system share a number of outside lines for making external phone calls. These phone systems are able to facilitate communication processes without encountering any delays that are prevalent in traditional phone systems. Without any interruptions, a call transfer reaches the intended destination faster. Callers will not have to wait long on the other end of the line for their calls to connect because these phone systems do not experience busy or loss of signals. [19]

A traditional PBX is made up of two key elements; these are lines and stations. The lines are connections to be global public switched telephony network by the way telephone of company. Stations are telephones or other endpoint devices like as fax machines, modems, credit card and terminals. It also supports traditional analog and digital telephones, allowing enterprises to migrate slowly to an all-IP telephony environment.

Nowadays there are four different PBX phone system options

- PBX
- Hosted/ Virtual PBX
- IP PBX
- Hosted/ Virtual IP PBX

IP PBX is added "IP" (Internet Protocol) so it is called IP PBX. The internal phones that are in the company as in your house you can connect as you have been your office, you may open phone and you may answer coming calls on the phone.

The IP PBX phone system will also provide your company with telecommuting feature. To telecommute means to communicate on the go. It will permit you to send and receive calls from your clients regardless of your location. [19]

IP PBX is a software based PBX phone system solution which helps accomplish certain takes and delivers services that can be difficult and costly to implement when using a traditional property IP PBX.

Also, an IP PBX is Unified Communication System or Business System.

A versatile business phone extension that is able to add mobile and home phone numbers will also be supplied to your business by IP PBX phone systems. [19]

This will further enhance the mobility your business has in communication.

This telecommunication service along with telecommuting keeps you connected to your business. [19]

Telecommunication costs of these phone systems are also cheaper than traditional phones.

The distance between the parties involved and the duration of the call will not affect the price that you will incur. Its telecommunication fees will be based on the quantity of data that is processed. Aside from lower communication rates, these phone systems also have cheaper upgrade costs. This is because all upgrades are done using software updates instead of purchasing new hardware or communication devices.[19]

4.1.1. VIRTUAL PBX SYSTEM

A virtual PBX system is a network of telecommunication channels that functions without physical connections. It runs via Voice over Internet Protocol, allowing users to connect their existing phone devices to one main number. [20]

A virtual PBX system is a few business telecommunication systems that provide competent communication service at a cost that can easily be afforded. This makes virtual PBX a phone system that any business should have, especially companies with limited funds and resources. [20]

Virtual PBX Phone System is one of the best business phone systems. Virtual Phone Systems include internet fax, voice mail, voicemail inbox, automated attendant, call forwarding, virtual PBX extensions, virtual offices and call screening.

4.1.2. How Does An IP PBX / VOIP Phone System Work?

A VOIP Phone System / IP PBX system consists of one or more SIP phones / VOIP phones, an IP PBX server and optionally includes a VOIP Gateway. The IP PBX server is similar to a proxy server: SIP clients, being either soft phones or hardware based phones, register with the IP PBX server, and when they wish to make a call they ask the IP PBX to establish the connection. The IP PBX has a directory of all phones/users and their corresponding SIP address and thus is able to connect an internal call or route an external call via either a VOIP gateway or a VOIP service provider.[21]

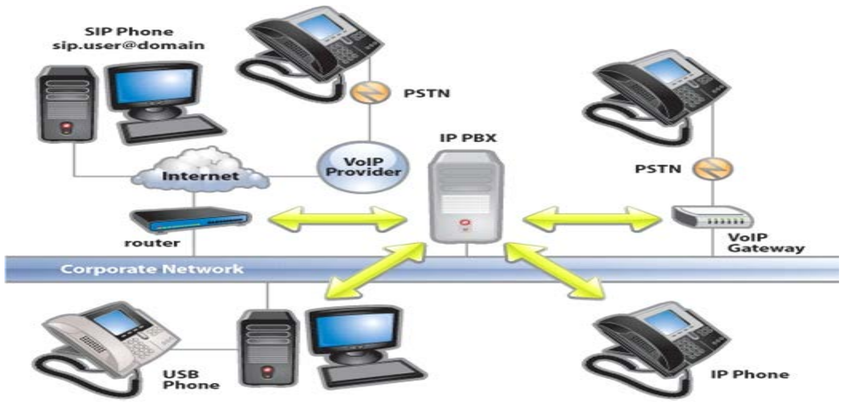


FIGURE 4.1: **How Does An IP PBX / VOIP Phone System Work?**

4.2 THE FEATURES OF IP PBX PHONE SYSTEM

The IP PBX Phone System has some features of keys. If you are looking for an IP PBX , here are some of features, you should be sure are included.

a. Virtual PBX Server provides access platform using IP Soft/ USB Phone
b. Call recording System
c. Call attendant System
d. Call on Hold player

1. Routing/Distribution/ Call Forwarding - This is a phone system feature that guarantees that all incoming calls will be sent to the person or department best equipped to address the caller’s questions. If the phone system has problems performing this very basic task, it is vital that it be replaced with one that can.[22]

2. Auto-Attendant - In conjunction with call forwarding, this phone system feature serves to answer inbound calls and carry out the call routing protocols; it connects calls from one extension fast and competently. If the system either has no option for an automated attendant, or if the feature is slow to answer, one must be very cautious.[22]
3. Music and/or Message on Hold - This phone system feature guarantees that the callers will not break off the call even if they are not immediately connected to the person or department they wish to speak to. The music and messages serve to inform and entertain. If this option is neither flexible nor available, the phone system can be useless for business.[22]
4. Voicemail - Not all calls can be answered by a department or person; this feature ensures that even the calls that were not picked up within the day can still be returned. With voicemail, callers can leave messages that can be reviewed at a later time. If the phone system does not have good voicemail function, it is not a good fit for business. [25]

5. Caller ID - This particular phone feature makes call management and call returns much more fitting for busy professionals. Without this feature, or if this feature does not translate properly through the whole system, there is a risk that important phone calls and messages will be either missed or disregarded. [22]

There are many other phone system features that a businessman may find useful in the context of his particular organization, but the above-mentioned five are the big ones that must-haves in any business phone system.

4.3. THE BENEFITS IP PBX PHONE SYSTEM

An IP PBX is a complete telephony system that provides calls over IP data networks. All conversations are sent as data packets. The technology includes advances communication features and provides worry-free scalability and robustness. The system consists of one or more SIP phones, an IP-PBX server and optionally a VoIP gateway to connect to existing PSTN lines.

Much easier to install & configure than a proprietary phone system**:** An IP PBX runs as software on a computer and can leverage the advanced processing power of the computer and user interface as well as Windows' features. Anyone proficient in networking and computers can install and maintain an IP PBX. By contrast a proprietary phone system often requires an installer trained on that particular proprietary system.[23]

Easier to manage because of web/GUI based configuration interface: An IP PBX can be managed via a web-based configuration interface or a GUI, allowing you to easily maintain and fine tune your phone system. Proprietary phone systems have difficult-to-use interfaces which are often designed to be used only by the phone technicians.[23]

Significant cost savings using VOIP providers: With an IP PBX you can easily use a VOIP service provider for long distance and international calls. The monthly savings are significant. If you have branch offices, you can easily connect phone systems between branches and make free phone calls.[23]

Eliminate phone wiring: An IP Telephone system allows you to connect hardware phones directly to a standard computer network port (which it can share with the adjacent computer). Software phones can be installed directly onto the PC. You can now eliminate the phone wiring and make adding or moving of extensions much easier. In new offices you can completely eliminate the extra ports to be used by the office phone system.[23]

i. Eliminate vendor lock in: IP PBXs are based on the open SIP standard. You can now mix and match any SIP hardware or software phone with any SIP-based IP PBX, PSTN Gateway or VOIP provider. In contrast, a proprietary phone system often requires proprietary phones to use advanced features, and proprietary extension modules to add features.[23]

ii. Scalable Proprietary systems are easy to outgrow: Adding more phone lines or extensions often requires expensive hardware modules. In some cases you need an entirely new phone system. Not so with an IP PBX: a standard computer can easily handle a large number of phone lines and extensions – just add more phones to your network to expand.

    Better customer service & productivity: With an IP PBX you can deliver better customer service and better productivity: Since the IP telephone system is now computer-based you can integrate phone functions with business applications. For example: Bring up the customer record of the caller automatically when you receive his/her call, dramatically improving customer service and cutting cost by reducing time spent on each caller. Outbound calls can be placed directly from Outlook, removing the need for the user to type in the phone number.

Twice the phone system features for half the price: Since an IP PABX is software-based, it is easier for developers to add and improve feature sets. Most VOIP phone systems come with a rich feature set, including auto attendant, voice mail, ring groups, advanced reporting and more. These options are often very expensive in proprietary systems.

The process of being able to easily move offices/desks based on the task at hand, has become very popular. Unfortunately traditional PBXs require extensions to be re-patched to the new location. With an IP PBX the user simply takes his phone to his new desk. Users can roam too if an employee has to work from home, he/she can simply fire up their SIP software phone and are able to answer calls to their extension, just as they would in the office. Calls can be diverted anywhere in the world because of the SIP protocol characteristics.

Better phone usability: SIP phones are easier to use Employees often struggle using advanced phone features: Setting up a conference, transferring a call – On an old PBX it all requires instruction.

Not so with an IP PBX – all features are easily performed from a user friendly Windows GUI.

In addition, users get a better overview of the status of other extensions and of inbound lines and call queues via the IP PBX Windows client.

Proprietary systems often require expensive ‘system’ phones to get an idea what is going on your phone system. Even then, status information is cryptic at best. [23]

4.4. IP SIGNALLING PROTOCOL

- H 323
- SIP (SESSION INITIATION PROTOCOL)
- GATEWAY CONTROL PROTOCOLS
- VIRTUAL SWITCHED PROTOCOLS

4.4.1. H 323

H.323 is an International Telecommunication Union telecommunication Standardization Sector (ITU-T) specification for transmitting audio, video, and data across an Internet Protocol (IP) network including Internet. [17]

The H.323 standard addresses call signaling and control multimedia transport and control and bandwidth control for point to point and multipoint conferences.[20]

4.4.2. SIP (SESSION INITIATION PROTOCOL)

IP phones are the same things as VOIP Phones or soft phones. These are telephones that allow phone calls to be made using VOIP technology. The first type is the hardware SIP phone, which resembles the common telephone but can receive and make calls using the internet instead of the traditional PSTN System. SIP Phones can also be software-based.

These allow any computer to be used as a telephone by means of a headset with a microphone and a sound card.

A broadband connection and connection of VOIP provider and a SIP server is also required.[18]

Session Initiation Protocol (SIP) is an IETF-defined signaling protocol widely used[citation needed] for controlling communication sessions such as voice and video calls over Internet Protocol (IP). The protocol can be used for creating, modifying and terminating two-party (unicast) or multiparty (multicast) sessions. Sessions may consist of one or several media streams. Other SIP applications include video conferencing, streaming multimedia distribution, instant messaging, presence information, file transfer and online games [citation needed].SIP is an application layer protocol designed to be independent of the underlying transport layer; it can run on Transmission Control Protocol (TCP), User Datagram Protocol (UDP), or Stream Control Transmission Protocol (SCTP). [25]

4.4.3 NETWORK ELEMENTS

SIP also defines server network elements. Although two SIP endpoints can communicate without any intervening SIP infrastructure, which is why the protocol is described as peer to peer this approach is often impractical for a public service.

a. USER AGENT

A SIP that end device is called a SIP user agent. User agent client (UAC) end system applications that contain both a user agent client and user agent server

(UAS) otherwise known as client and server respectively.[17]

A SIP user agent is a logical network end point used to create or receive SIP messages and manages a SIP session. A SIP user agent can perform the role of user agent client which sends SIP requests, and User Agent Server which receives the requests and returns a SIP response.

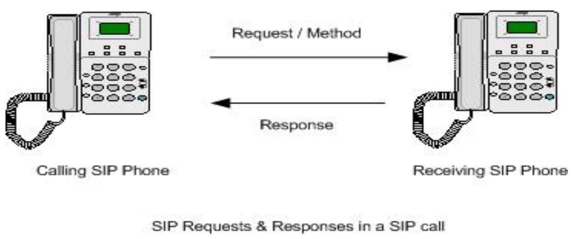


Figure 4.2.SIP Requests and Responses in a SIP Call

Client: SIP requests and acts as the user’s calling agent.

Server: Receives requests and returns responses on behalf of the user acts as the user called.

SIP phones may be implemented as a hardware device or as a soft phone.SIP is a standard telephony platform and it is often driven by 4G efforts, the distinction between hardware based and software based.

b. PROXY SERVER

A SIP proxy server receives A SIP request from a user agent or another proxy server and acts of the user agent in forwarding or responding to the request. A router forwards IP packets at the IP layer, a SIP proxy forward SIP messages at the application layer.[19]

A proxy server firstly plays the role of routing which means its job is to ensure that a request is sent to another entity "closer" to the targeted user. Proxies are also useful for enforcing policy for example; making sure a user is allowed to make a call.

A proxy server has access to a database or a location service to aid it in processing the request. The interface between the proxy and the location service is not defined by SIP protocol.[20]

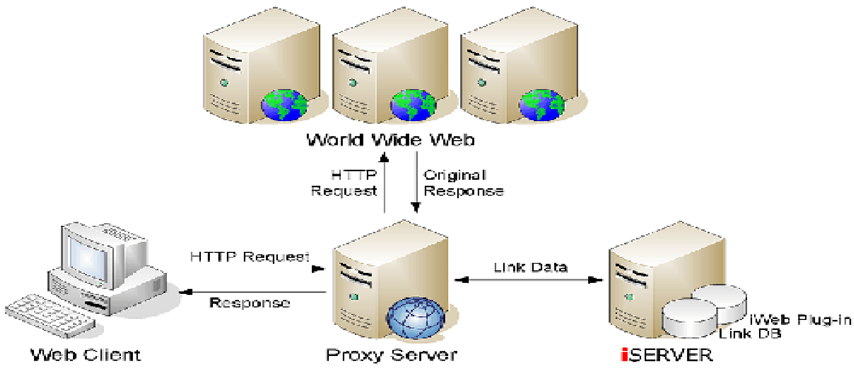


Figure4.3: Proxy-server

c. REGISTRAR

A server that accepts Register requests and places the information, it receives in these requests into the location service for the domain it handles which registers one or more IP addresses to a certain SIP URI, indicated by the sip: scheme, although other protocol schemes are possible.

SIP registrars are logical elements and are commonly co located with SIP proxies. But it is also possible and it often good for network scalability to place this location service with a redirect server.[17]

d. REDIRECT SERVER

A user agent server that generates 3xx (Redirection) responses to requests it receives, directing the client to contact an alternate set of URI. [17]

The redirect server allows proxy servers to direct SIP Session Invitations to external domains.

e. GATEWAY

Gateways can be used to interface a SIP network to other networks such as PSTN (Public Switched Network) which uses different protocols or technologies.

The H.323 gateway reflects the characteristics of Switched Circuit Network (SCN) endpoint and H.323 endpoint. It translates between audio, video, and data transmission formats as well as communication systems and protocol. [17]

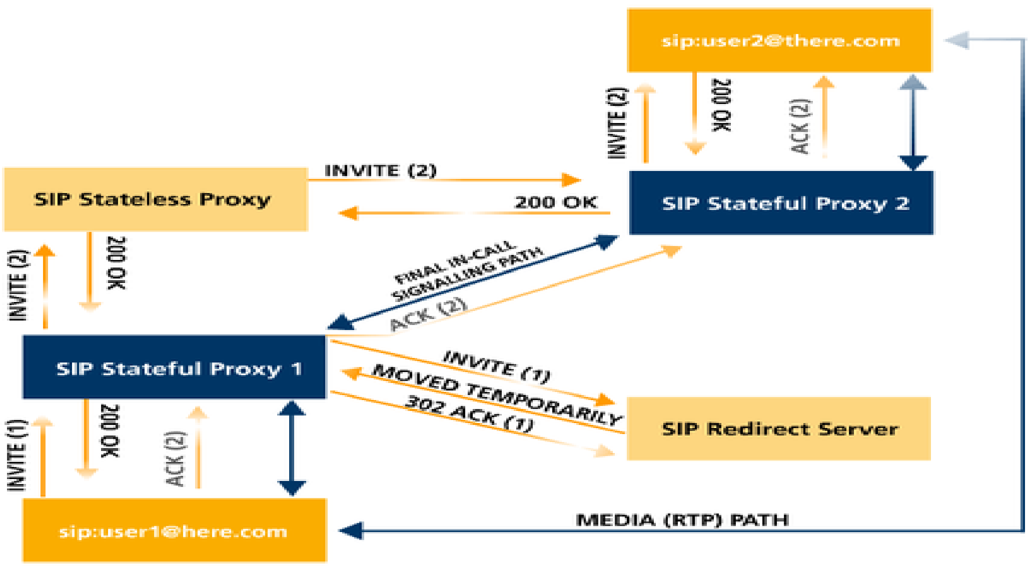


Figure4.4: SIP Diagram

4.4.4. GATEWAY CONTROL PROTOL

Gateway Control Protocol enables call control elements to control connections between trunking, residential, and access type VOIP gateways. [17]

Although these gateways target different segments all of the convert time division multiplexing voice to packet voice.

Gateway Control Protocol is used to establish, maintain and disconnect calls across an Internet Protocol (IP) network. The required connections between desired and corresponding endpoints.[17]

4.4.5. VIRTUAL SWITCH CONTROLLER

At a high level, the virtual switch controller (VSC) provides the following:

- Call signal processing includes Integrated Services Digital Network (ISDN)
- Address resolution, call routing, resource management, connection control, and call detail record generation.
- Service access functions for accessing services executing on external server platforms
- Management interfaces using Simple Network Management Protocol performance and configuration. Web based configuration tool and element management system. [17]

4.5) THE KINDS OF IP PBX PHONE SYSTEM

4.5.1) ASTERISK

Asterisk is an open source framework for building communication application. Asterisk turns an ordinary computer into a communication server.
Asterisk is technology and protocol which means that you can connect it to the outside world using VOIP or traditional telephone technologies.
Asterisk powers are IP PBX systems, VOIP gateways, conference servers and other custom solutions. It is used by small businesses, large businesses, call centers, carriers and government agencies, worldwide. Asterisk is free and open source. The Asterisk project started in 1999 when Mark Spencer released the initial code under the GPL open source license. Since that time it has been enhanced and tested by a global community of thousands. Today Asterisk is maintained by the combined efforts Diguim and Asterisk community.[26]
Linux distribution that installs the operating system, Asterisk, drivers for Digium and phones and an open source administrative user interface called Free PBX. The installation process if fully automated and takes roughly 20 minutes to convert a computer into a working phone system.[26]

The asterisk software includes many features availed PBX system.

- Voice mail
- Interactive voice response
- Conference calling
- Automatic call distribution

4.5.2) THE FEATURES OF ASTERISK

Asterisk creates a PBX that rivals the features and functionality of traditional telephony switches. Asterisk is cost-effective, low-maintenance, and flexible enough to handle all voice and data networking. With Asterisk software, Telephony hardware, and a common PC, anyone can replace an existing switch or complement a PBX by adding VOIP, voicemail, conferencing and many other capabilities. Asterisk integrates with analog phones and most standards-based IP telephone handsets and software. Asterisk reduces the cost of traditional telecommunication technology and operation, and moves voice over IP (VOIP) to mainstream. Asterisk integrates a pre- existing analog telephone network with the benefits of IP technology, greatly reducing costs. [26]

4.5.3. SIPXecs (Enterprise Communication Server)

SIPXecs (Enterprise Communication Server) is an open source voice over IP telephony server. [25] The SIPXes IP PBX is an open source alternative to private branch exchange (PBX) Systems from vendors such as Avaya, Nortel, Cisco, Siemens, NEC and others.

The main feature is a software implementation of the (SIP) session Initiation Protocol which makes IP based communication system (IP PBX).SIPX is not like Asterisk and it is very popular open source PBX, but design of SIPXecs deviates from Asterisk in many ways.

There are many features of SIPXecs and these are following: private branch exchange (PBX) like voice mail, voice response system, auto attendants e.g.. The main components of the system are designed around Freeswitch a media router. The SIP standard can be used to build a fully featured solution.

SIPXes IP PBX offers a long list of features all based on standard SIP signaling.

4.5.4) SIPXECS AND ASTERISK

Asterisk is best described as a platform where the SIPXeces IP PBX is turn-key solution with pros and cons to both. Asterisk is a common line based application with several open source and closed source Web UI applications available. SIPXecs is complete solution with the web administration application built in. [24]

Asterisk supports SIP, H.323, Cisco, SCCP, NORTEL and SS7. SIPXecs only supports SIP. SIPXecs IP PBX is only solution that offers plug and play management for phones and gateways. There are two critical difference between Asterisk and SIPXecs that significantly affect performance:

SIPXecs IP PBX uses external gateways. It supports as many external gateways you need without limit and offers in case a gateway is unavailable or busy. It also offers least cost routing where gateways can be deployed anywhere you need them. [24]

Asterisk uses PCI gateway cards where the number of trunk port is limited by the number of PCI slots available in a server. [24]

SIPXecs IP PBX does not route calls (media) through the server because it separates signaling from media. [24]

Therefore SIPXecs can support as many same calls as your LAN/WAN bandwidth permits. Asterisk has a hard limit because calls go through the Asterisk server.

Programming language: Asterisk is written in C. The SIPXecs communication server is written C++.The SIPXecs IP PBX Configuration server is a Java application. SIPXecs relies on XML for internal data structure and a set of related modern languages and protocols.[24]

4.5.5) 3CX PHONE SYSTEM

3CX Phone System is windows based IP PBX. It is a traditional software based IP PBX which is replaced hardware of PBX.

Evolve your communications with 3CX Phone System for Windows and an IP Phone System that completely replaces proprietary PBX, supports standard SIP software or hardware phones, VoIP services and traditional PSTN phone lines.

The VOIP is a special astral that is developed by Microsoft Windows so, it is easy to used 3CX and any SIP phone is imposable to used software and hardware. [27]

a. Key Features

- There is no requirement a different phone cable. Phones and Computers are used the same cables.
- It is easier to establish and use owing to Web based Configuration Surface
- A software based IP PBX is cheaper than a hardware based PBX.
- They can be moved their offices without getting any required difference and at VOIP central configuration.
- Instead of connecting only one seller, you can make selection among the lots of SIP based hardware phones.
- By using VOIP devices, you can call with standard PSTN devices.
- The prices of calling costs can be decreased, if you use any of VOIP (Voice Over Internet Protocol) or WAN (Wide Area Network)
- Complete phone system - Provides call switching, routing & queuing.
- Unified Communications - Receive voice mail via e-mail& see user presence
- Auto Attendant (e.g. 1 for sales, 2 for support, etc.)
- Reduce long distance and inter office call costs.

- No more expensive proprietary system phones - Use standard SIP phones
- Eliminate the phone wiring and make moving offices easier
- Easy call control, presence and extension management
- Click to Dial & Call Pop-up for Microsoft Outlook
- Receive & Make calls via the standard PSTN using VoIP Gateways or cards .
- Save on monthly call costs using SIP trunks, VoIP providers or Skype Connect.[30]

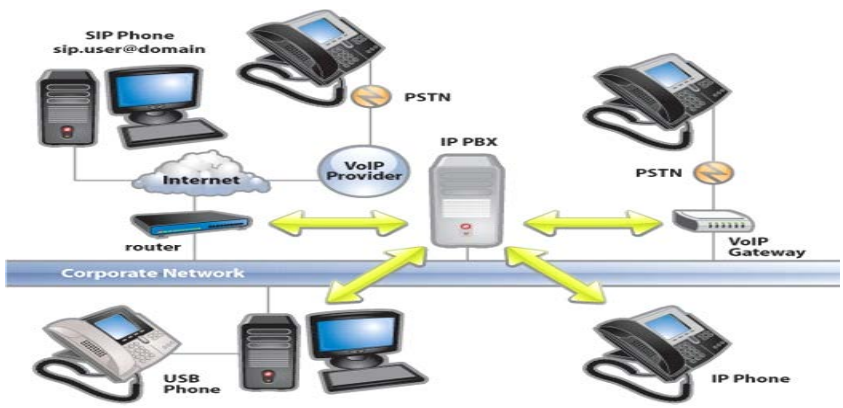


Figure 4.5: 3CX PHONE SYSTEM

b. The Advantages of a Software Based VoIP/ IP PBX

3CX Phone System for windows is a software based IP PBX that replaces a hardware PBX. IP PBX has been developed s specifically for Microsoft Windows and is based on the SIP standard, making it easier to manage and allowing you to use any SIP phone (software or hardware).A software-based IP PBX offer many benefits:

Easier to install & manage via web-based configuration interface

Far less expensive to purchase and expand than a hardware based PBX

Improve productivity with presence, desktop based call control and extension management. No need for separate phone wiring phones use computer network. Deliver mobility by allowing employees to work from home using a remote extension.

Choose between popular IP hardware phones or soft phones no vendor lock in Receive and make calls via the standard PSTN using VoIP Gateways or cards. [27]

4.6.DID (DIRECT INWARD DIAL)

DID(DIRECT INWARD DIAL) is also called DDI in Europe and it is a feature offered by telephone companies for use with their customers PAB x System, where the telephone company (telecommunication) allocates a range of numbers associated with one or more phone lines. DID requires that you purchase an ISDN or Digital line and ask the telephone company to you assign a range of numbers. You then need DID capable equipment at your premises which consists of BRI, E1 or T1 cards or gateways. [28]

4.7. STUN (Simple Traversal of User Datagram Protocol)

A STUN (User Datagram Protocol) (UDP) of Net address converters (NATs) on simple transition) server, NAT client's (for example the computers which are behind firewall) gives opportunity to VOIP service provider to direct a phone call without local network STUN communication server, clients' own IP address , the type of NAT'S that they support and gives oppurtinionity to find side internet port which is related with a certain local port. This information is used for setting the UDP relation between client and VOIP service provide and it is used for starting phone call. STUN protocol is defined in RFC 3489 with Stun communication server can be communicated at UDP 3478 port but communication server will say that the other IP port numbers can be tried by clients. RFC, the usage of IP and port usage is clarified voluntary/optional.[29]

CHAPTER5

5. IMPLEMENTIONS

5.1) 3CX Phone System

Using 3CX Phone for 3CX Phone System to truly take your business mobile , Android, IOS and Windows client integrates seamlessly with 3CX Phone System and up to allow you to make and receive office calls from anywhere in the world via WI-FI or 3G. [30]

By using your mobile phone as your extension you can make free calls to your connected offices and make savings on telecommunications costs.

5.2) Setup 3CX IP PBX/VOIP For Windows

3CX Phone System is a software based IP PBX for windows.

What you will need to setup 3CX Phone System:

- 3CX soft phone
- Purchased VOIP number
- Win XP computer
- IIS 6 on your XPOS
- The 3CX soft phone
- A static IP FQDN(countable on the internet) or a dynamic IP
- Time to install, configure and test.

5.3) 3CX CONFIGURATIONS

5.3.1) 3CX Server

1. The firstly; go to the web address of the login page. The web address is determined by the customer.
2. On the login page, type the user name and password into the User name and password fields. We write “admin” as user name.
3. And then; click on the login button to go to the main form page.

Figure5.1:3CX Login

4. Add Extension: On the main form page, there are two ways to add an extension:

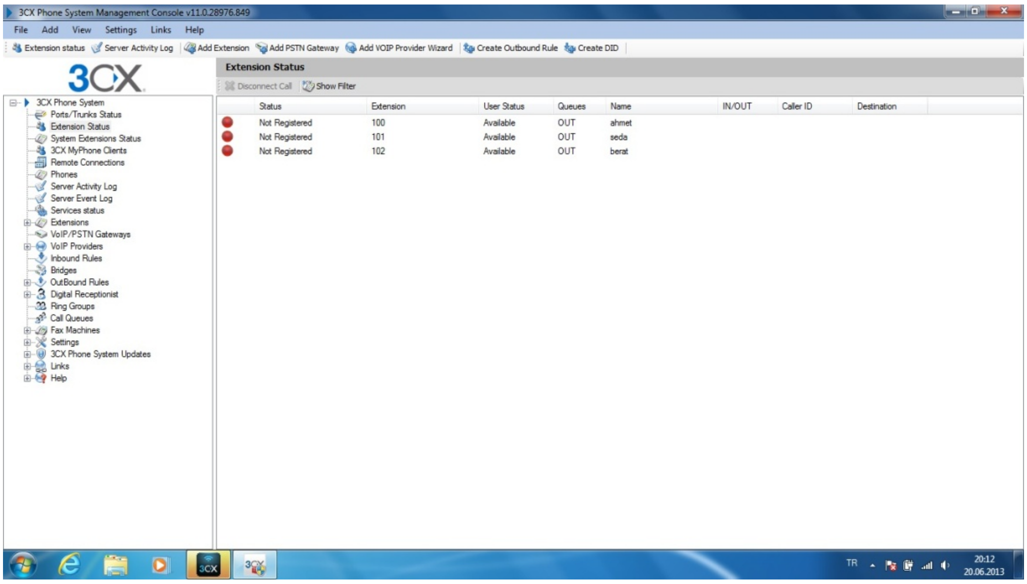


Figure5.2.Extension Configuration

5.Extension Configuration

On the Extension page, enter the following information:

- Extension Number
- First Name
- Last Name
- Authentication ID
- Authentication Password

5.3.2)3CX Soft-phone Register Settings

It is run 3CX soft-phone. Click right on soft-phone. And click on “account”. We write those are;

Account name: Seda.

Caller ID:101

Extension:101

ID (SIP user):101

Password:101

My location: I am in office 192.168.1.2. with register local IP.

If the computer and SIP server are location the same network

or these are location on the same switch, we write local IP.

I am not office: If there is register on the internet to SIP Sever,

we write external (wan) IP .

5.3.3) 3CX Phone System in the same LAN as the PBX

3CX Phone can be configured to connect to 3CX Phone System from the same LAN, and from a remote location. If 3CX Phone will be used from a remote location, you can take advantage of the built in tunnel functionality to overcome NAT traversal.

3CX Phone has STUN settings pre- configured, but it is known that it is only needs to use STUN if it setups as a remote extension in Direct Mode. The STUN Setting will not be employed if the phone is configured to work as a Local extension or as a Remote Extension with Tunnel Protocol.[31]

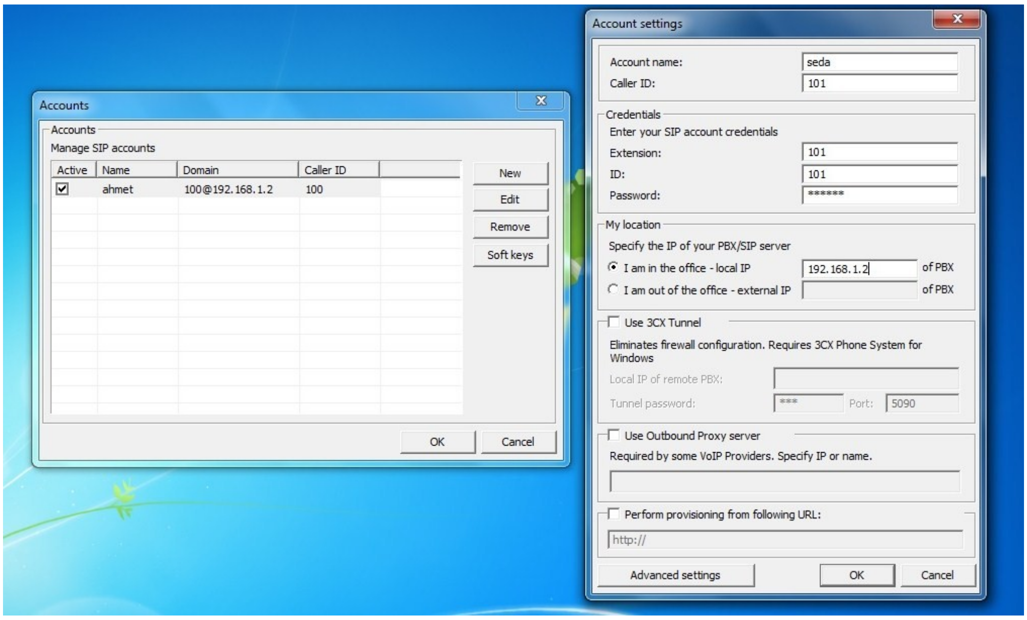


FİGURE5.3: 3CX Phone System in the same LAN as the PBX

5.3.4)3CX Phone from a Remote Location and Tunnel Mode

3CX Phone provide a built in tunnel client which connect to tunnel server implemented directly within 3CX Phone System. The Tunnel Server of connections on the 3CX Phone System machine on port 5090 in both UDP and TCP. Configuration on the server side is straightforward and you need implement port forwarding on the WAN to LAN device between 3CX Phone System and the internet so that any traffic received by the WAN to LAN device on the WAN interface to the public IP Address to port 5090 will be forward inside the LAN to the 3CX Phone System machine's Local IP Address.[31]

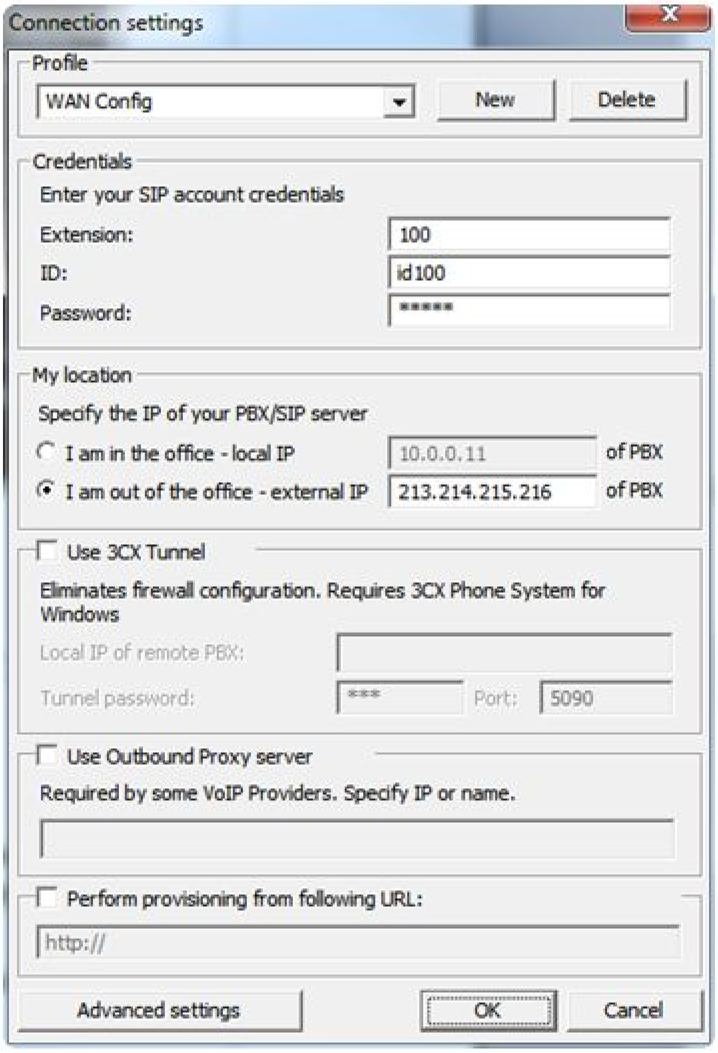


Figure 5.4: 3CX Phone from a Remote Location and Tunnel Mode

Configuring 3CX Phone System to work from a Remote Location in Direct Mode (without using the built in tunnel) is also straightforward. Select the radio button labeled "I am out the office -external IP and enter the public IP Address of 3CX Phone System machine.[31]

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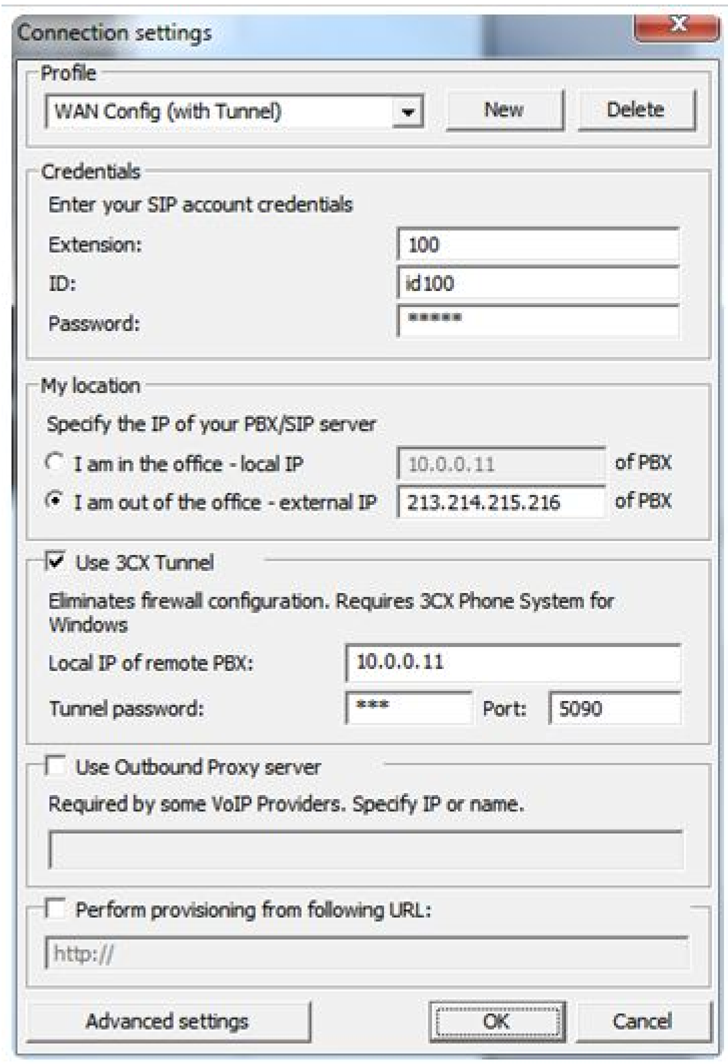

Figure5.5.3CX PHONE SYSTEM TUNNEL MODE

5.4) **3CX Phone For IOS**

3CX Phone for 3CX phone System is an IOS VOIP client that has been specifically designed to work seamlessly with 3CX phone System and later.

You may use your I-phone or I-pad as your office extension, meaning you may see the presence and status of your colleagues and employees and calls between you will be free, and you save money, on your telecommunications bills. You may make and receive office calls with your I-phone or I-pad from anywhere in the world.[31]

FIGURE 5.6.3CX PHONE SYSTEM CALLS

5.5)3CX Phone For Android

The 3CX Phone for Android app integrates seamlessly with 3CX Phone System and allows up to make and receive office calls on your Android smart-phone and tablet from anywhere using 3G and WI-FI.[31]

5.6) 3CX Phone System Trunks

There are number of ways you can implement the 3CX system, that is including on the local office LAN, or in a data center.

While it is recommended use is in on the local LAN, with the implementation of 3CX is the developer of 3CX Phone System. 3CX is an open standard unified communications platform for Windows that works with standard SIP phones and replaces a proprietary PBX.

We need to connect 3CX outside, we have our internal network. This connection is called a trunk.[31]

- PSTN trunks
- SIP trunks
- Introduction to dial plans
- Hardware needed for analog lines.

5.6.1) PSTN Trunks

A Public Switched Telephone Network (PSTN) trunk is an old fashioned analog Basic Rate Interface (BRI) ISDN or Primary Rate Interface (PRI) phone lines.

3CX can use any of these with the correct analog to SIP gateway.

For using an analog PSTN line, you will need an FXO gateway.[34]

5.6.2) SIP Trunks

A SIP trunk is a call that is routed by IP over the Internet through an Internet Telephony Service Provider (ITSP).

IP PBXs and communicate over IP not only within the enterprise, but also outside the enterprise, a SIP trunk provided by an ITSP that connects to the traditional PSTN. You can see that, you have a local area network containing your desktop, servers, phones, and your 3CX phone system. To reach the outside world using a SIP trunk, you have to go through your router. Depending on your network, you could be using a private IP address (10.x.x.x.,172.16.x.x. or 192.168.x.x) which is not allowed on the public Internet, so it has to get translated to the public IP address. This translation process is called Network Address Translation (NAT).[31]

There are three components necessary to successfully deploy SIP trunks:

- A PBX with a SIP-enabled trunk side.
- An enterprise edge device understanding SIP
- An Internet Telephony or SIP trucking service provider

SIP trunk usually uses in the real world, the company is a single IP connection to provide services on which multiple PSTN connection. It is known that of PSTN VOIP service offers cheaper and better quality audio connection. The many big companies VOIP communication within the company reverted to it via the IP-PBX. In connection with the PSTN into the fabric ITSP (Internet telephony service provider - Internet phone service provider) given by enterprises are benefiting from PSTN service. [30]

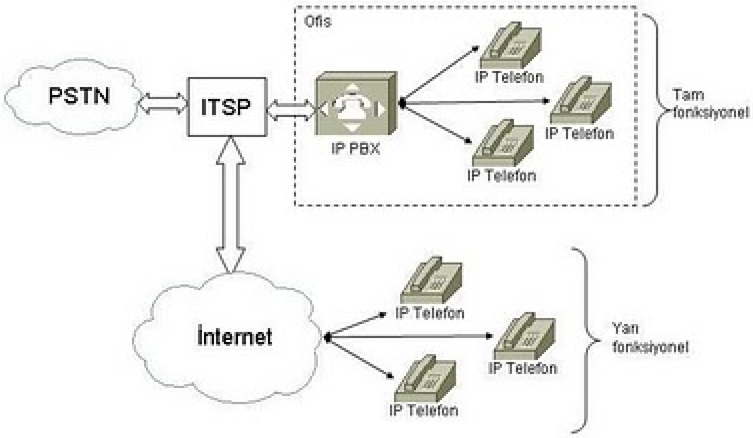

Figure5.7. 3CX TRUNKS

It can access over the Internet to the PSTN as well as outside the office can establish a connection with the IP phones. ITSP firm is obliged to provide the required QoS. [30]

A point to be aware of the IP over the Internet can be reached by all the services offer to telephones office during the IP PBX.Another alternative is referred to as the hosted PBX trunking. This method is cheaper than the previous one.

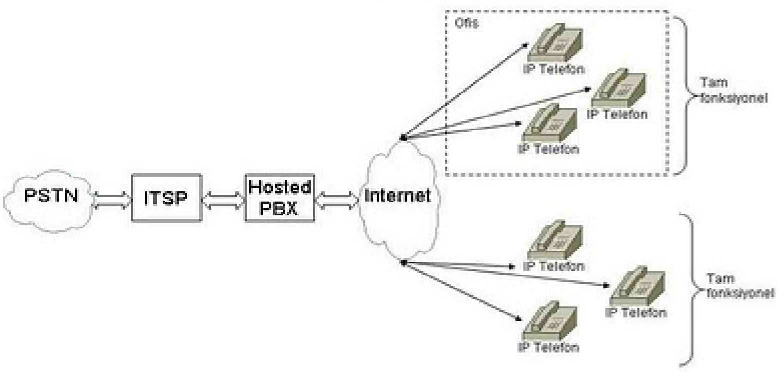

Figure 5.8.3CX TRUNKS

5.6.3) Connecting 3CX Your Trunk

The setup of 3CX connects to a PSTN line using an analog gateway and it has then connecting 3CX to a SIP/ITSP line.

The first thing ; you need to know is that every line or port in 3CX is assigned and it is very own number like the Ring Groups, Digital Receptionists, and Call Queues have their own account number assigned. This makes it easier to route calls using a number.

1. The first way is to click Add and then PSTN Gateway:

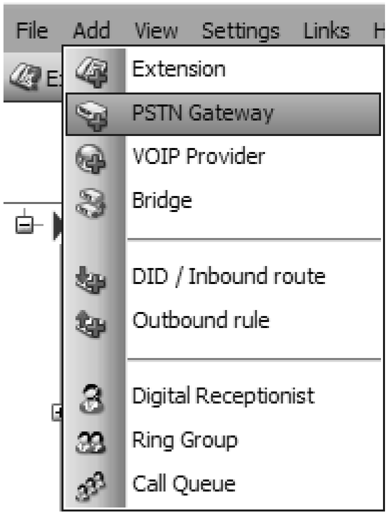

Figure:5.9: PSTN Gateways

2. The second way is to click add PSTN Gateway on the main 3CX toolbar .

Figure 5.10: Add PSTN Gateways

3. The third method is to use the navigation pane on the left hand side; click PSTN devices, and then click add Gateway on the right-hand side.

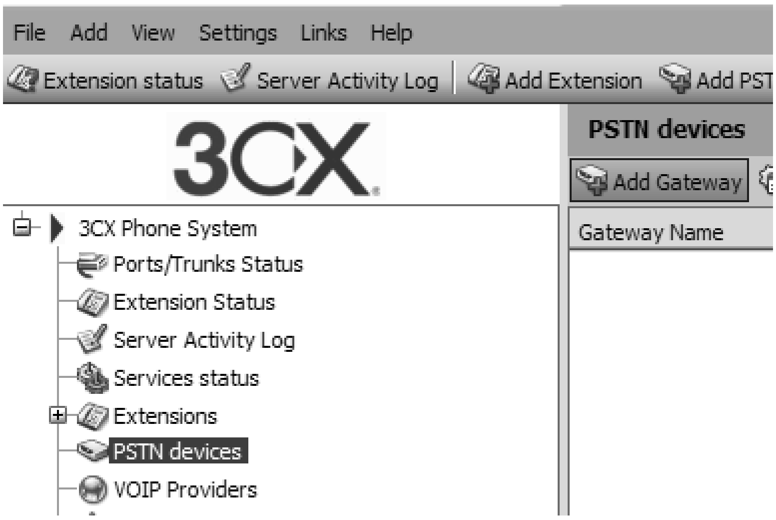

Figure5.11:3CX PSTN Devices

Now, we have started the PSTN Gateway wizard and we can run through the steps. In firstly; it has a name like an A or 1.

Using this method you expand easily and keep the naming conventions the same for all devices. I was using VOIP Gateway, then I used the same name and used a B at the end. Using a label marker, I also label with a name IP Address and depending on the password and username.

Figure 5.12: Vendor Supported Gateways

The next step is to pick which supported gateway.

There are more gateways that work with 3CX that are listed on their website, but they do not work with the gateway wizard.

After you can select your gateway, click next:

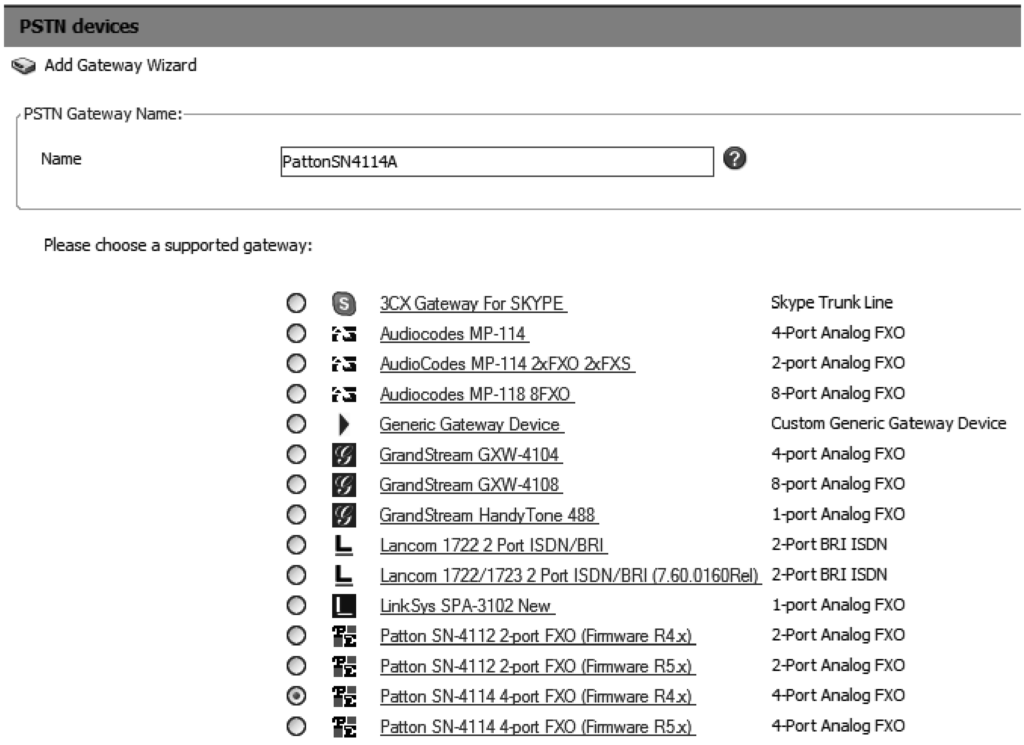

Figure 5.13:Gateways Wizard

On the next screen; you will be select a few options. The first one is Tone set Selection to select which country this gateway is going to be installed in.

The next section is for incoming Caller ID info. It will wait another ring (depending on which country and your phone company information).[30]

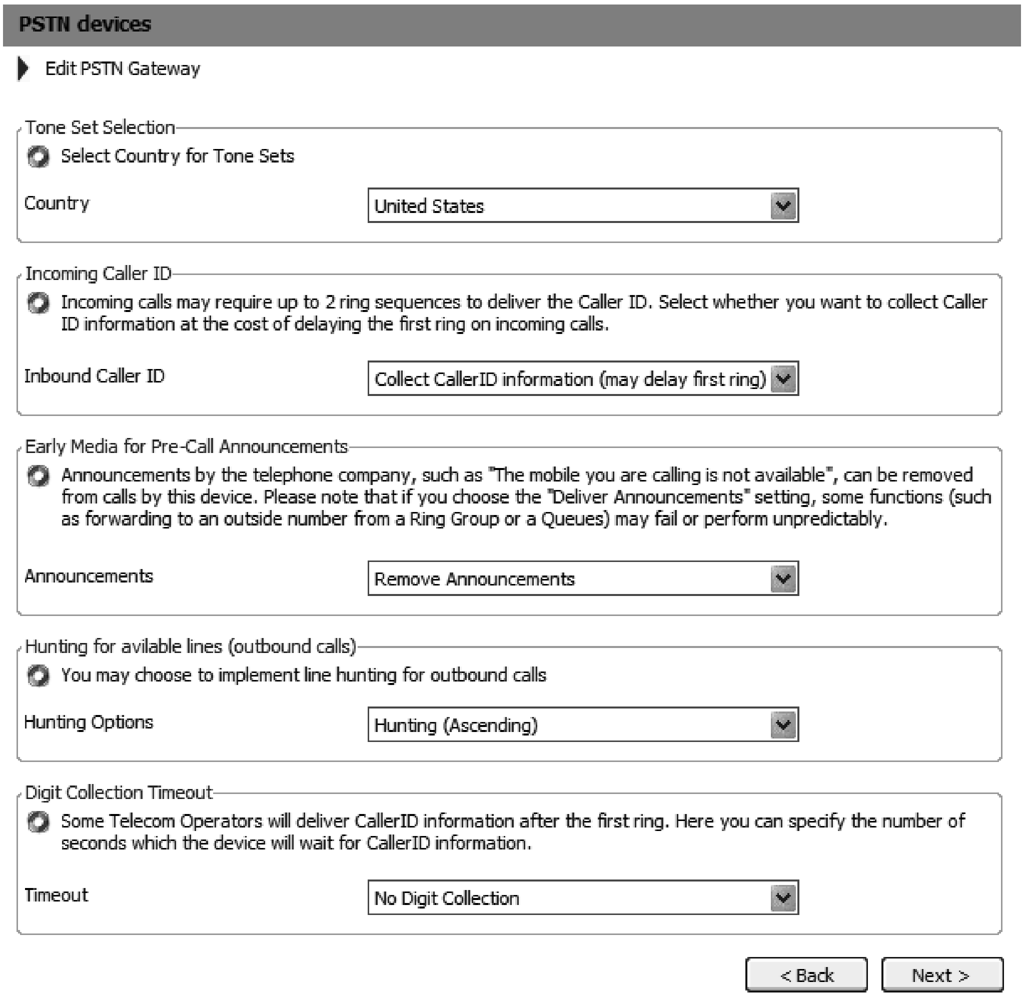

Figure 5.14: Edit PSTN Gateways

Our next wizard screen is device specific. We start to give it a Gateway or Hostname or IP address. If you do not have your own DNS sever or you are using WINS or host files, you will want to use an IP address. Even if you have a DNS server , you would use an IP address. You do not want to lose your connection to the gateway if your DNS server is down. So we need to specify Gateway Port to use. Unless you have a reason to use something different stick with the well known default SIP port 5060.[30]

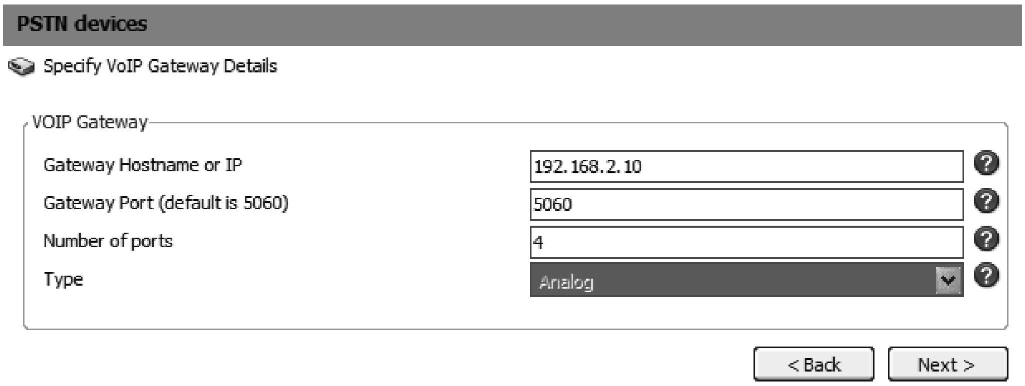

Figure 5.15:Specify VoIP Gateway Details

Now; we get to create the port numbers, names, passwords and some rules for call processing. So that we have a Virtual Extension number ,Authentication ID, and Authentication Password.

When we are using analog single call lines we need to leave the Channels section to 1.The only real thing we may want to change is the Inbound Route Day and Inbound Route Night. We explain that 3CX Phone system what to do with an incoming calls.[30]

5.6.4) 3CX SIP Trunk And VoIP Providers

3CX Phone System for windows is an award winning software based IP PBX that replaces traditional proprietary hardware PBX/ PABX. It is entirely SIP standard based and therefore interoperates with most popular SIP phones, SIP VOIP Gateways and SIP VOIP providers.

In the 3CX Phone Management Console click on “VOIP Providers” from the left menu. Click on “Add VoIP Provider” button on the top of the page and “Add VoIP Provider Wizard” page appears. The name of the provider, choose "Generic SIP Trunk" or "Generic VoIP Provider" and click "Next" button.

1. Start 3CX Windows Management Console

Figure5.16: 3CX Windows Management Console

2. Under VOIP Providers, Add Provider

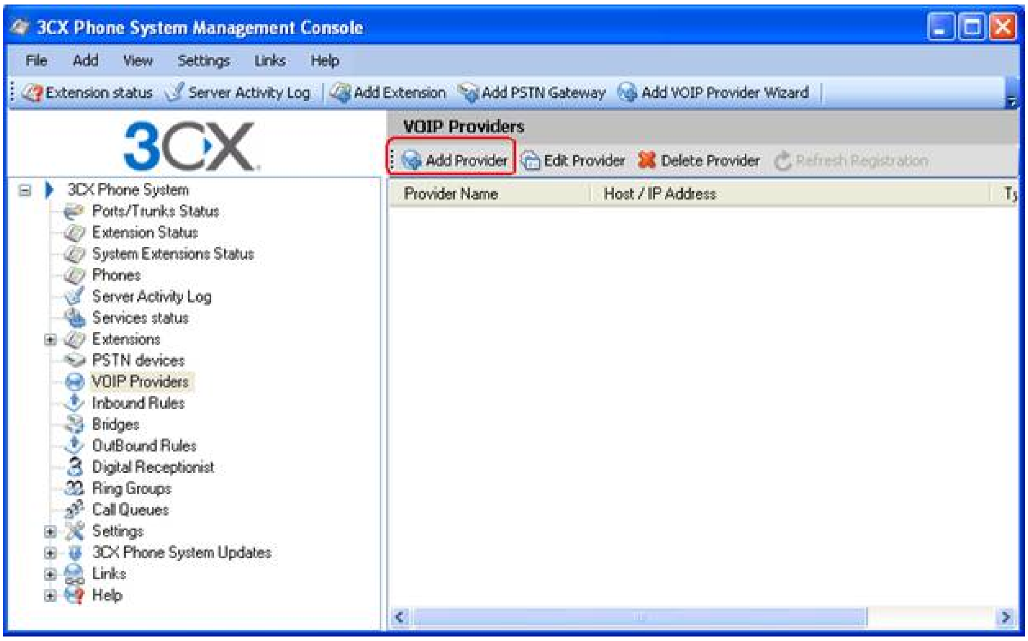

Figure5.17: Add Provider

1. Under name of Provider select “Generic SIP Trunk.

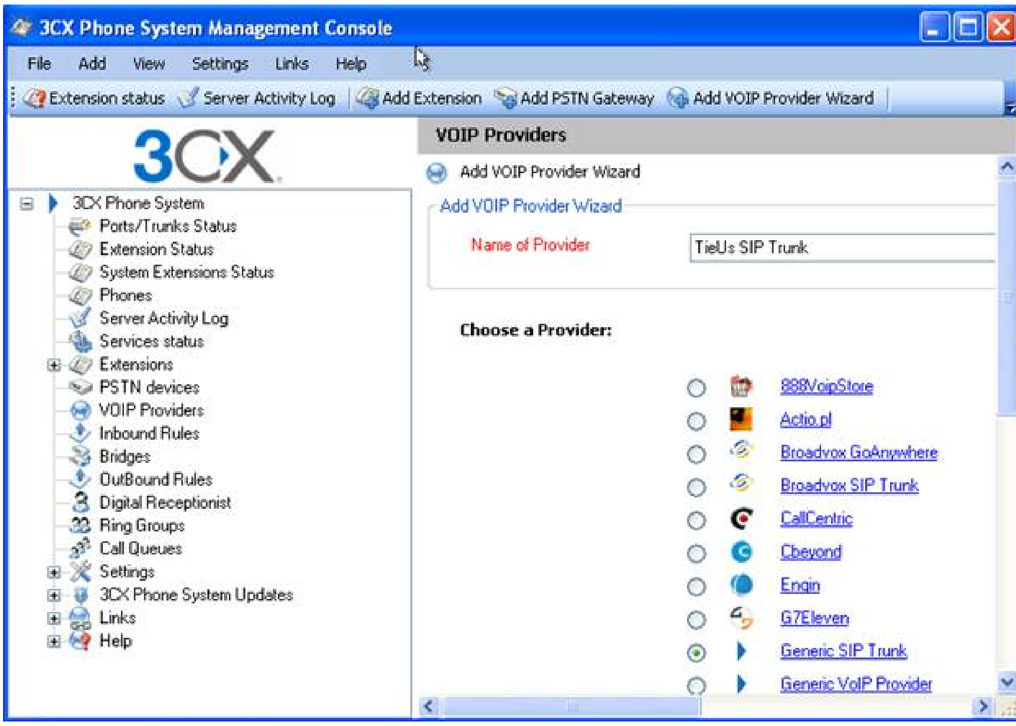

Figure 5.18: select “Generic SIP Trunk.

2. Click Next Under VOIP Provider ,enter the SIP server IP Address. Please check the SIP account information, we send you , the SIP Server or IP Address will be different from the IP Address below.

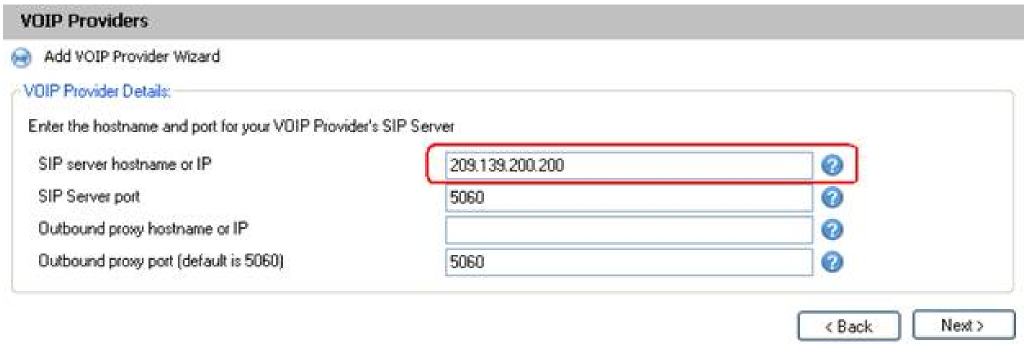

Figure5.19:VoIP Providers

3. Click next, enter your SIP account information here. Enter 10 digit as External Number enter 14 digit authentication ID, enter 5 digit Authentication Password.

   Enter the maximum simultaneous call. The number should be matching our system setting.

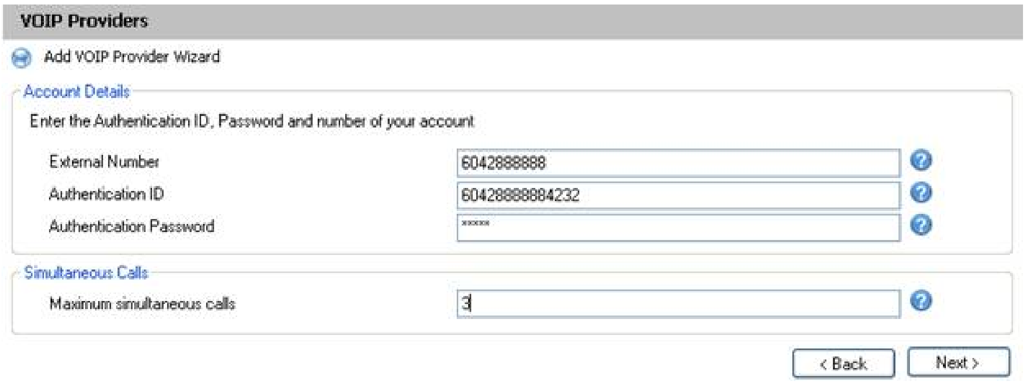

Figure5.20: Add VoIP Providers Wizard

4. Click next .You are required to setup the behavior of 3CX when receiving SIP Trunk incoming call.

You can connect the call to certain extension or you can connect to digital Receptionist (Auto Attendant), provided that you already have recorded the voice message. For initial testing purpose we recommend you to connect the call to extension, so you can test the incoming call after setup.

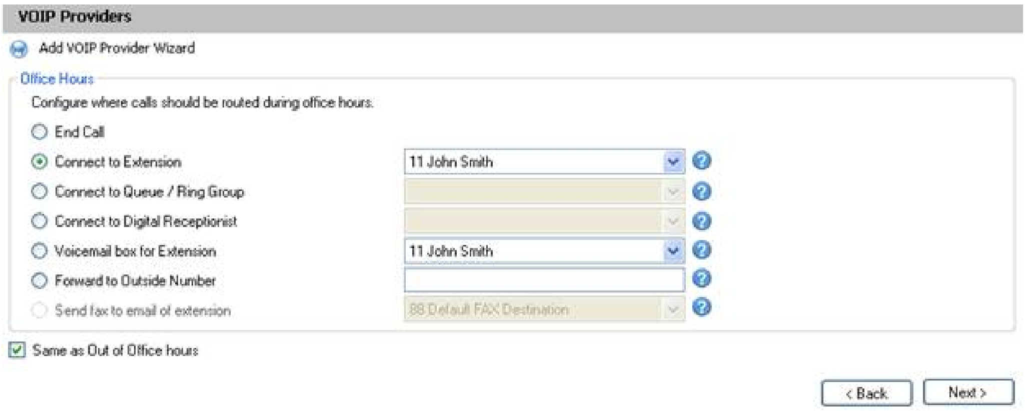

Figure5.21: VoIP Providers2

5. You need to setup the outgoing call behavior. In general, to distinguish the internal call between extension and outgoing call to outside number, you can setup a prefix so 3CX know how to route the call through SIP trunk. For example; you can add calls to numbers starting with Prefix with “9”. When you want to dial out from extension, simple dial 9+10 digital number you want to dial.

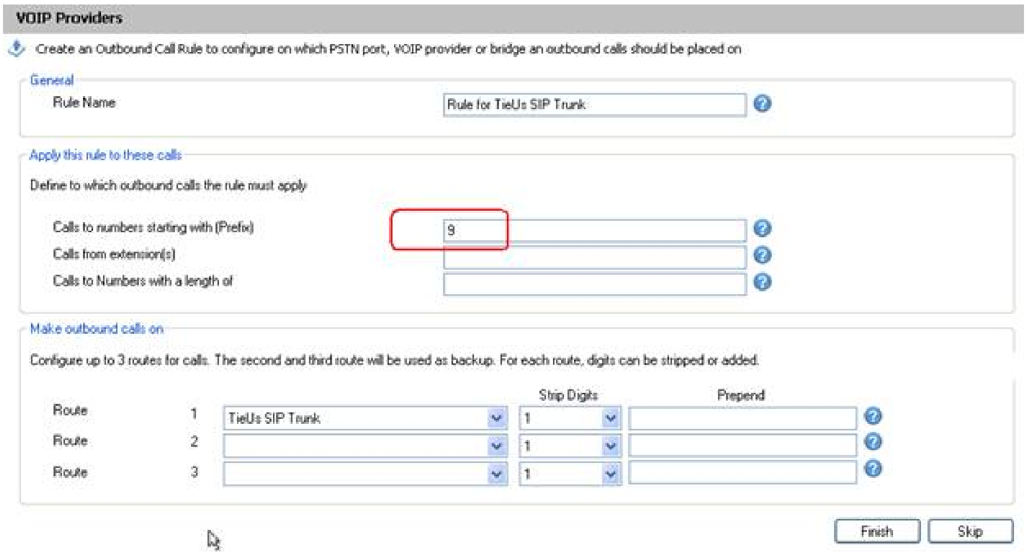

Figure5.22:VoIP Providers3

6. Click on “Finish to compete the initial setup. You can observe if the trunk or extension is registering correctly by “looking at Port/ Trunks Status or Extension Status

Figure5.23:Port/trunks Status

7. To make incoming and out coming call: Only if encounter problem or one way voice, check Firewall and Router Setting port 5060 UDP should open for SIP Trunk signaling. Port 5480-5486 need to open according to 3CX Specs.

In general setup a static map or forward of ports: 5060-5100 (TCP and UDP) for SIP related signal 9000-9015(TCP and UDP) for RTP related signal, and 3400-3499 (TCP and UDP) for tunnel related.[32]

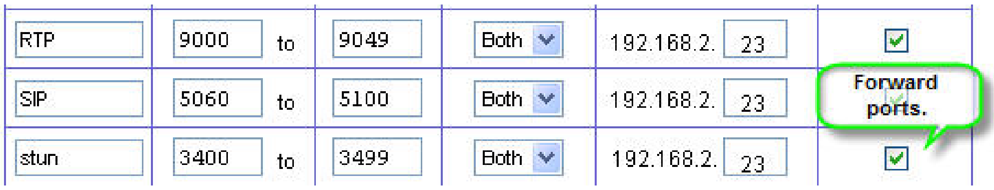

Figure5.24:Forward ports

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