

Implementation of VoIP services through the Integration of technologies Call Manager Express and Asterisk Server

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This paper presents the VoIP (Voice Over Internet Protocol) services implementation applied to an organization. It is a proposal for improving the voice transmission quality at low bit rate available to all users. To this end, are integrated with VoIP technology. Infrastructure for internet access and simultaneously has wireless IEEE 802.11n/g/b standards. Given a network communication over voice over internet using SIP protocol for communication between a server Call Manager Express (CME) and an Asterisk server. Managing calls to the needs of users from one server to another customer and arriving at the other end. Furthermore, the implementation is suitable for a voice VLAN and a VLAN for data. To convert analog voice to digital voice, the codec G7.11 is used.

Implementation, communication, voice, protocol, internet, integration, service, organization, customer.

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Introduction

The Implementation of VoIP services is one in which voice communication is established over internet protocol (IP). This term describes the essential concept of making a phone call and putting it on an IP packet network.

A technology that is considered the basis of the convergence of *voice, data and video* over a single network and under the same protocol. That is, in essence implementation of technology that promises reliability, accessibility, and above all low costs [5].

This work is founded on actual experience Integrating VoIP with two different technologies trying to analyze the current situation and previous research studies to develop options for the future and its application to the reality of communication in organizations whether they are: educational institutions, government agencies and companies [17].

VoIP users will grow exponentially in the coming years and it is anticipated that they will replace existing technologies currently existing. It can be seen by referring to applications: such as Skype, Google Talk, Fring, and Rebtel This Type of software is the Most Popular VoIP technology to search. The companies have chosen to compete In this area due to its popularity, usefulness, and its perceived future in a few years when a link to the Internet is more common, when will be more convenient to use the data network than the current connections provided: such as GPRS (General Packet Radio Service) , GSM (Global System for Mobile) [6,7].

It should encompass the selection of technologies, including hardware, software, and necessary platforms, along with installation and configuration of both basic functions and other more advanced IVR (Interactive Voice Response), group communication, gateways to the PSTN (Public Switched Telephone Network), etc., of a VoIP based system that can meet the basic needs of telephony services in an organization.

It uses free software for the Implementation of the telephone to make its easy integration into the infrastructure [6, 7]. Simultaneously managing two very different technologies is Cisco (Call Manager Express) and free software (Asterisk) respectively. Cisco is known as a leader in networking for the Internet with many of its standard or proprietary protocols in other words the same technology would only work with Cisco. The other side has the solution under GPL (Generic Public License) That is cheaper and not having to pay license fees to use the service and a wide range of VoIP protocols like SIP, IAX (Inter Asterisk eXchange), H.323, MGCP (Media Gateway Control Protocol). Having a more likely possibility to Establish Communication between different technologies [1, 4, 8, and 10].

Communication modes

Internet telephony can take various forms, but the most popular is the verbal conversation between two people.

These forms of Voice over IP, are established according to the devices used by the participants in the communication:

Softphone to Softphone

- Softphone to Phone (Fixed or Cellular).
- Telephone (fixed or cellular) to telephone (fixed or cellular).

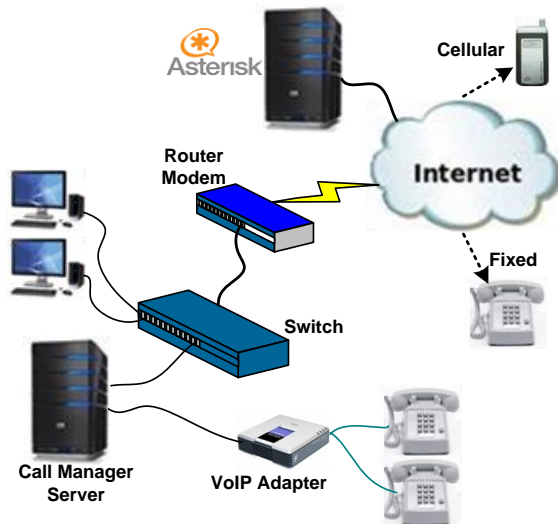


Figure 1

Figure 1 shows the arrangement in equipment and services that are going to work together. You can see that it takes a server on the Asterisk which is mounted to switch either calls to the PSTN or to an internet service that may still end up in a traditional analog phone, mobile, or softphone (combination of *software* and telephone).

There is a device that works like Linksys analog telephone adapter and also has a very important role as it is the router that directs calls to the VoIP service provider for them to be redirected once again to traditional or cellular phone.

Depending on the type of VoIP service , you can make phone calls and IP (Internet Protocol) from a softphone (computer application), special VoIP phone or a traditional phone with or without an adapter.

Moreover, the existence of new access points to high speed internet or "hot spots" in public places: such as airports, parks, and cafes allow you to connect to Internet and use VoIP service wirelessly.

If the VoIP service provider you have regularly assigned a phone number, then may you regularly receive calls from phones that do not need any special equipment and can surely be marked as is usually we done.

Justification (traditional telephony vs IP Telephony)

Drawing on the IP telephony capabilities, you can perform the same functions or features of traditional telephony, but also has a number of new features, among which we can mention are: [7, 8, and 10]

- Call transfer.
- Monitoring of calls.
- Reporting service (call history).
- Recording calls.
- Identify users.
- Videoconferencing.
- Music on hold.
- Volume Control.
- Emergency calls.
- Calls on hold.
- Answering calls automatically.
- Blocking caller.
- Web mail interface to check.
- Call simulator
- Reduction in installation and maintenance costs.
- Conference call.

Maximum mobility: The availability of your telephone extension anywhere in the world, provided you have an internet connection [7].

Quality of Service (QoS): This consists in assigning priorities to the transmitted packets over the IP Network. For example, setting a higher priority to voice packets are time sensitive during transmission [9, 11].

Integration: Provides the integration of telecommunications services: such as voice, data, video and Internet on the same network, in an efficient, fast and effective way.

It's worth mentioning that in recent years hundreds of companies have changed their traditional telephony services to VoIP. Since 1995 when VoIP started to be used with small applications in the last 10 years we have had a technology revolution until today so this is nothing new and is well established to offer great quality service [7,9,10,13].

Methodology

Cisco Lifecycle Services

The methodology of Cisco Lifecycle Services was considered because it permits the use of a business plan, an assessment of the network, and the documentation on the Implementation of the system. With this you get to make calls from extension to extension assigned by VoIP servers, to conventional phones with a lower rate than what is currently available and it has an infrastructure that is easy to manage, control, and scale that adapts to future needs [7].

Infrastructure used

- Server Call Manager Express 7,
- Cisco Catalyst 2960 Switch,
- Cisco ATA 186(Analog Telephone Adapter)
- Cisco IP Phones 7940 and 7906
- Server Asterisk,

- 3CX Softphone, IP Communicator and Linphone
- AP router D-LINK to access the Internet
- Wireless service with IEEE802.11n/g/b [6].

Cisco 2821 router features of Call Manager Express (CME)

- Offers call processing for Cisco IP phones to branch offices or small office environments.
- Permits a wide range of integrated service routers to offer Cisco IP telephony's commonly used features by business users to meet the requirements of voice and video communications for small to medium sized offices.
- Enables the deployment of a communications system that is economical and very reliable through a single integrated services router [2,3,7,and 10].

Features of the Cisco Catalyst Switch 2960

- Supports voice, data, video, and secure access.
- Offers scalable management that adjusts to a company's needs [3].
- *Features of the 186 ATA (Analog Telephone Adapter)*
- Connects POTS (plain old telephone system) to the Internet.
- It's a phone jack for VoIP [7].

Features of the Router D-LINK DIR - 615 Wireless.

- Provides excellent performance in transfer rate up to 300Mbps and wireless coverage up to 5 times the standard 802.11g signal.
- Enables Internet connection sharing within your network wirelessly and wired as well as video , music, photos, and documents.
- Uses smart antenna technology to transmit multiple streams of data which enable you to make and receive wireless signals to the farthest corners [7].

Features of the 3CX Softphone

- It is a phone program that allows:
- Making voice conversations
- Having an extension and connecting to have free communication.
- Performing cheap calls to traditional telephony system [7].

VoIP Protocols

H.323 is Commonly used for Voice over IP and IP-based videoconferencing.

MGCP implements the control interface gateway media as a set of transactions. The transactions are composed of a command and a mandatory response.

SCCP (Skinny Client Control Protocol) terminal control protocol owner. It is defined as a set of messages between a thin client and the Call Manager [10].

SIP (Session Initiation Protocol), designed according to the model of the Internet and whose purpose is the communication between multimedia devices. SIP makes this communication possible by two protocols that are RTP / RTCP (Real Time Protocol/Control Protocol in Real-Time) and SDP.

It is based on request and response messages. This protocol uses a port(5060) for signaling and RTP audio 2 ports for each connection (at least 3 ports). If you have 100 simultaneous SIP calls 200 (RTP) ports would be used along with port 5060 for signaling [5,7,18].

IAX (*Inter-Asterisk eXchange protocol*) protocol connections between Asterisk VoIP servers are used for connections between clients and servers that support the protocol [5,7].

Codec

Voice communication is analog, while the network is digital data. The process of converting analog waveforms to digital information is made with a coder - decoder (CODEC). There are many ways to transform an analog signal, each Governed by various standards, Table 1 shows some of them [7].

Standard	Application	is used
G.711	Designed composite audio, used mainly in telephony. Operates to 64Kb / s.	Yes
G.728	Conversely adaptive uses previous voice samples for adapting the coefficients filter.	No
G.729	Used mainly with VoIP applications Requires low bandwidth. Opera at 8 Kb / s	Yes

Table 1

Asterisk

Asterisk is an application to monitor and manage communications of any kind, analog, digital, or VoIP by all VoIP protocols it implements.

It is an environment of open source work, created to design communication applications. Also power systems, PBXs, IP gateways, VoIPconference servers, and much more [1].

Asterisk is based on separate modules that can be uploaded and downloaded at will, Depending on the needs you want to provide the system. Each module has a specific function, that can handle all aspects of the system, via the channel types (IAXSIP) DAHDI or connections to other systems to interact with Asterisk (mail, databases, web, etc. ..) [1, 7.13].

Applications of Asterisk

Asterisk applications are management actions that apply to calls within the dial plan. For example, the most popular in all plans would be the application, Dial which simply aims to launch a call to a channel based on the properties identified during implementation.

There are other common applications like Voice Mail (responsible for voice mail management), Record (to record the sound of a channel). Some common features of the applications are as follows [1,7,13]:

- The actions are exclusively focused by and for the channels.
- Are loaded dynamically.
- Are run synchronously.
- The classic format of this type of modules is `app_ <name>. so`

Resources for Asterisk

The specific function of the resources is to integrate Asterisk with external systems such as: databases, web servers, calendars and more.

To have the ability to use their own applications on the system, but one of the differences with respect to these, is statically loaded, and can operate on multiple ongoing channels simultaneously, rather than dynamically created for each channel [1, 7.13]. The classic form of this type of module is `res_ <name>. so`

Functions DialPlan

The basic notion behind the function is the ability to obtain or add certain specific information to each channel.

They use to be complementary to the applications and are able to offer improvements to certain aspects of the system which in itself could be limited. Therefore the most common way they are used is by the application Set.

For example, a typical function is able to collect the call identifier of a channel (CALLERID) to handle within the dialing plan at will [1,7,and 13]. The classic form of this type of module is `func_ <name>. so`

Drivers Channel

They are specific drivers for each channel type available now or in the future for Asterisk. These are the ones who specifically provide the possibility of returning the system completely independent of them in order to treat them fully homogeneous. It is exactly a kind of interface Between the core of Asterisk and the "logic" in the operating system. All typical channels, have a corresponding module for the driver [7].

The classic form of this type of module is *chan_ <name>. so*

Translators Codecs and formats

They are the representation for digital, audio, and video transmission systems (codecs) and storage (formats). That is, are responsible for converting via software, from one type to another format or codec simultaneously within the course of the call. For example, if a call comes on the DAHDI channel, and wants to switch to a SIP extension in Asterisk system with G.711 the codec translator will be responsible for this conversion in real time. On the other hand, if you are dealing with files, the responsibility for interpreting the contents to pass through the corresponding audio channel would be translator formats. In this area the most popular are the standards, GSM and WAV in which most of the default system sounds of Asterisk are (prerecorded messages) [7]. The classic format codecs modules is: *codec_ <name>. so* and for formats: *format_ <name>. so*

Trunks or gateways

It's a link That connects external calls of a telephone, concentrating and unifying multiple simultaneous communications on a single signal for transmission and efficient transmission distance (usually digital) and to establish communications with another exchange or a whole network of them [1,3,4,7].

Service provider VoIP

It's a company dedicated to providing VoIP service. It is dedicated to connect by phone VoIP users with users of conventional mobile phone.

If you want to call from the softphone or through an Internet connection to a traditional line telephone or mobile phone, you need someone to manage the voice traffic and send it through everyday conventional lines . That is where the VoIP provider's work comes in [14,15]. Such calls are not usually free,

You have to create an account with a provider and buy some credits that will be used to call landlines and mobile phones [14, 15].A PBX type telephone system uses a trunk line to make the center part of the network of other plants and maintain communications. Generally the digital PBX trunk lines are links That support up to 30 channels (lines) and voice intercom. If communication of the trunk is interrupted, there would be no communication between plants by any of its 30 channels [14,15].

IAX is a protocol used by Asterisk PBX server (telephone) open which is source and sponsored Digium. It is used to manage connections Between VoIP Asterisk servers, and between servers and clientes that also use IAX protocol [1].

Implementation of Infrastructure

The Asterisk server was installed and configured and later the extensions were added that users would use with the protocol, SIP along with adding a trunk to a service provider which performs the routing of calls to what is Known as public switched telephone network PSTN via Call Manager Express(CME) its protocol is taken advantage of by Skinny Client Control Protocol(SCCP) to interact With The IP Phone 7940 and 7906, it has a network over voice communication over the Internet with the ability to work.

With The SIP protocol to establish a communication between the CME and Asterisk Server redirecting calls to the needs of users from one server to another customer and arriving at the other end. In both the catalyst switch is suitable for Implemented a voice VLAN and data VLAN [1,7]. 'SCALL Manager acts as a proxy signaling for initiated calls through other protocols: such as H.323, SIP, ISDN or MGCP [7.10].

Subnet Design

Step 1: Identify the broadcast address for the network address [7]. Table 2 shows the data of the private network.

Parametersnetwork	Data
network address	172.16.0.6/16
Subnet Mask	255.255.0.0
Broadcast Address	172.255.255.0
Number of host's	65,234

Table 2

Step 2: Determine the number of subnets.

The number of subnets is determined by the amount of bits that are in the subnet counting range (for this network, 3 bits.)

$$2^n = 2^3 = 8 \text{ subnets.}$$

Step 3: Identification of the number of hosts per subnet.

The amount of hosts per subnet is determined to by the number of host bits (In this case, 13 bits counting from right to left of the host 172.16.00000000.00000000) minus 2 (1 for the subnet address and 1 broadcast address subnet).

$$2^n - 2 = 2^{13} - 2 = 8190 \text{ hosts per subnet.}$$

Step 4 : Identify the range of host addresses.

To get the range there are several ways, the simplest is to subtract from 256 the number of the subnet mask adapted and which takes 3 bits of the host portion to form subnets (128 + 64 + 32 = 224).

In this case is: $256 - 224 = 32$.

So, the range of each subnet is 32, Table 3 shows the information related to the network 172.16.0/19.

172.16.0.0 /19	Usable IP's			
No. Subnet	Subred	IP start	IP Final	Broadcast
1	172.16.0.0	172.16.0.1	172.16.31.254	172.16.31.255
2	172.16.32.0	172.16.32.1	172.16.63.254	172.16.63.255
3	172.16.64.0	172.16.64.1	172.16.95.254	172.16.95.255
4	172.16.96.0	172.16.96.1	172.16.127.54	172.16.127.255
5	172.16.128.0	172.16.128.1	172.16.159.54	172.16.159.255
6	172.16.160.0	172.16.160.1	172.16.191.54	172.16.191.255
7	172.16.192.0	172.16.192.1	172.16.223.54	172.16.223.255
8	172.16.224.0	172.16.224.1	172.16.255.54	172.16.255.255

Table 3

Step 5: Results.

In this case only the subnet number 1 is considered. And table 4 shows the results obtained for the design of this subnet.

Parámetros de red	Datos
SubnetAddress	172.16.0/19
MaskSubnet	255.255.224.0
Subnet Bits	$2^3 = 8$ subredes
host bits per subnet	$2^{13} - 2 = 8,190$ hosts por subred
IP for this subnet start	172.16.0.1
IP for this subnet finish	172.16.31.254
Broadcast address	172.16.31.255

Table 4

Subnets avoid unnecessary broadcasts, simplify management and monitor growth.

Dial Plan

According to the dial plan, extension mappings were made only considering those used in connection with testing each forms of communication. These are shown in Table 5. While Table 6 presents all allocations made.

Extensions						
	Special	IP-Comunicator	Mobil	ATA	IP-Phone	PC
Ext. Start	2000	3000	6000	7000	8000	9000
Ext. Finish	2999	3999	6999	7999	8999	9999

Table 5

Server	Extension	Name	MAC
Asterisk	9000	ZAGA	2C:81:58:E2:7B:97
Asterisk	9001	CRISTHIAN	00:22:5F:B2:A5:95
Asterisk	6000	IPHONEMACIEL	24:AB:81:13:5D:BB
CME	8000	M.C.C. CANO	00:19:55:FB:32:4E
CME	8001	M.I. MASCOTTE	00:12:00:45:82:A0
CME	3000	TSU. JAIME	08:00:27:A9:62:AB
CME	8002	M.I. HERRERA	00:22:90:BC:21:70
CME	8003	DR. BARRON	00:23:33:9D:45:36
CME	7000	TIC Secretary	00:1D:45:95:64:24
CME	7001	DEM Secretary	1D:45:95:64:24:01
CME	2000	2000	44:E4:D9:E6:AD:41
Asterisk	2001	Conference	N/A

Table 6

Dial Patterns:

Local: 01 + area code + Phone:

National 01 + area code + Phone

Cell: 044 and 045 + area code+ phone

Implementation of Elastix server

Elastix is a free distribution of Unified Communications Server that Integrates into one package [3,4]:

- VoIP PBX
- Instant Messaging
- Email
- Collaboration

It implements much of its functionality on four important software such as: Asterisk, Hylafax, Openfire and Postfix.

These functions Provide PBX, Fax, IM and email respectively. Furthermore, it runs on CentOS operating system and the version currently installed is Elastix 2 [4,7].

Installing Elastix

Elastix is distributed as an ISO file can be burned to a CD from any CD burning software.

Having Elastix ISO properly recorded on a CD, once burned the CD is inserted into the PC when starting.

But we must ensure that the boot is the optical drive (CD-ROM), this is enabled in the BIOS of the PC. As soon as it's started the installation window will appear.

Note: At installation time the hard disk is formatted during installation, thereby ensuring the information is important.

1. At the opening screen of the initial installation hit ENTER to start the installation settings.
2. Now proceed to choose the type of keyboard according to the language. If the keyboard is Spanish language, select "es."
3. Then select the time zone according to the region.
4. Was enter the password That will be used by the administrator of Elastix.
5. The password that the administrator for Elastix will be using is entered.
6. Once the installation is complete, proceed to reboot.(Note: You must remove the disc and preferably modify the BIOS to boot from the hard drive as the first option.)

7. After restarting, the system will Automatically start the Following kernel: Elastix-base (2.6.18-53.1.19.e15)
8. Then you enter as root user and the password entered at the time of installation.
9. Upon completion of the installation, the screen displays information about joining the Elastix kernel. This is shown below.

CentOS release 5.9 (end)
 Kernel 2.6.18-i686
 348.1.1.e15on an

Asterisk login: root
 Password:

Network Configuration

Once accessed the configuration is done by assigning a network IP address, for access via HTTP. The screen displays the Following information:

IPv4 Configuration for eth0

Accton Technology Corporation
 SMC2-1211TX
 00: E0: 29:9 D: 19: A2
 () Configuration of dynamic IP (DHCP)
 (*) Configuration Manual TCP /IP

IPAddress Prefix (Network
 Mask)
 172.16.0.2_ _____ / __ 255.255.224.0_

OK Previous

This information Indicate s the IP address assignment.To enter via http, then open the web browser and enter the IP address in the address bar, to open the server you enter the username and password Assigned [4,7,13].

And then, the screen is displayed in the main interface elastix server, which is shown in Figure 2.

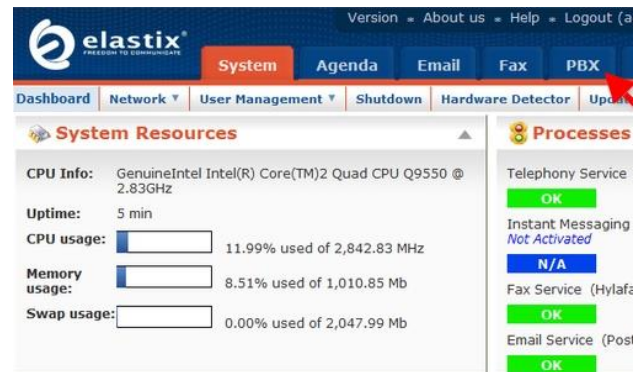


Figure 2

Adding extensions

The extensions are those numeric identifiers that enable communication between hosts on the local network and external telephone numbers, this number is then assigned to what can be a IP softphone or a phone.

To add it, find the tab with the legend of PBX submenu and Extensions then choose between the protocols you want to use in Which there is: SIP, IAX2, ZAP,DAHDI, etc. Once you have selected it, press the Submit button [4,7,and 13].

Next is a new page to add a basic extension, the field User Extension must be filled. this option will contain the number or caller ID to send and receive calls for it to work, consider the following: special numbers of functionalities that codes give access to various functions of the PBX based on this your Dial Plan is established.

Likewise, in the field Display Name the user name is show when making a call to another extension, for example M.I. MASCOTTE and most important is to add a password so that not everyone can use the account and only one person has access to it, this is defined in Secret.

When finished press Submit and a box appears box asking reload the Asterisk server to apply the settings that Have been established [4,7].The first section is the Edit Extension which has the following options:

- Display name: This is the Caller ID That displays the user when calling another user.
- CID Num Alias: This parameter is optional and what it does is replace the user caller id for the one that is Indicated here. It only affects internal calls.
- SIP Alias: If you want to allow direct calls by SIP, for example by an unregistered terminal, you can put a simple and short name instead of the extension number [4,7].

Add trunk

Trunks are useful additional servers to interface VoIP service, they may have a different extension range or can work with a technology or different communication protocols extending more options. One of the main stem used in this system is the one that communicates with the VoIP service provider (vozia) and a secondary to integrate Cisco and ATA phones to also add analog terminals. To add a trunk to the Elastix server, go to the PBX tab and submenu trunks [4,7].

Among the protocols That Can be selected are: SIP, DAHDI, Zap, IAX2, ENUM, DUNDi. Choose the most adequate.

Having selected the protocol to use for the trunk, a new page opens to modify the configuration. In the part of Outgoing Settings the main parameters are introduced. Trunk Name is the name of the trunk to be assigned. PEER Details or USER Details depending on the context.

If you type =user: can only receive calls as "user".

If you type =friend: you make calls as "peer" and receive calls as "user" (used for extensions.)

If you type =peer: to make calls as "peer" and receive calls as "peer" (used for trunks) [4,7].

Overall parameters are specified as: VoIP provider IP address, user name, account, password, account, etc.

The "Register String " is a parameter that is used because you must be registered with the server When the IP address is dynamic, in the case of Internet service in most companies. This string tells the remote server where to find the user forever, even though its public IP address changes [4,7]. The format is:

```
Username: Password @ proveedorIP or
Username: Password @ proveedorIP
/User
```

Outgoing routes Setup

Outgoing routes are extremely important in conjunction with dialing patterns when making decisions to use different trunks. To configure an outbound route, the first thing is to go to the menu where it says "Outbound Routes "and a menu appears where in "RouteName" you will put a descriptive name. RoutePassword: Here a code is assigned that it will always you to input to use this route projection [4,7].

Configuring inbound routes

Outgoing calls must be configured for when you want to make out calls, but when you want to receive calls you must procure a virtual number service or better called DID (Direct Inward Dialling or Direct Inbound Dialing).

Having the account of this service you can direct to where it says "inbound routes" at the bottom of the page where it says "Set Destination", where it says "Extensions" if the extension is located to which you wish to receive the call, also worth noting you can receive an IVR system (Interactive Voice Response or Interactive Voice Response) that amounts to a virtual operator that describes a menu to find the right destination extension or correct extension [4,7].

Additional Services

Additional services are those features that give a bonus to VoIP communication.

Leaving aside the ability to send and receive calls, you should also monitor and manage the service and provide alternatives in different situations [4,7].

Were configured the following services:

- Email
- Voicemail
- IVR
- Call conference
- clock control.

Email

The function of email in Elastix server has the main objective to provide the right conditions to turn voice mail enabled, since in this case the voice mail works via email where call recording is stored [4,7].

Voice Mail

Having created the email account domain you can proceed to create the users linked to existing extensions, for that you must go to the tab System and the submenu Users to press the button. Add New User. To activate the service begin of voicemail you must set the time it will ring before being redirected to another extension or in this case it is sent to voice mail, the configuration is called Ring Time and standard telephone companies is set to 20 seconds [4,7].

Conference Call

The conference call is the synchronization of multiple simultaneous calls that through a password allows or denies access to configure this service go to the tab PBX and submenu PBX Configuration in the left side menu there is a group called Internal Options & Configuration click on conferences [4,7].

Security

Elastix comes with an integrated firewall which can block ports that are not going to use and increase safety to prevent attack on certain vulnerable ports or in special cases on certain IP addresses.

To configure the service go to the Security tab and the Firewall submenu, now find the port you want to close [4,7,18].

Equipment Setup

1.9.1 Cisco Router 2821 Call Manager Express (CME)

Network Configuration

Enable DHCP service to act as a server address, which may provide IP addresses to devices that are within the same network, so exclude some addresses which are manually implemented [7.10].

The information displayed on the screen corresponds to the telephony service and are shown below:

```

sip-ua
credentials username 3000 password 7
1240044F165C0A522E7F2F realm asterisk
authentication username 3000 password 7
135C164A0F5B027C2F7023
retry invite 4
retry response 3
retry bye 2
retry register 10
timers register 250
registrar ipv4:172.16.0.2 expires 3600
sip-server ipv4:172.16.0.2:5060
!
!
    
```

Add client SIP

Enable telephone service, you create the ephone-dn which contains the user name and extension, then you create the ephone which relates the MAC address of the device the codec that will be used, the type of device (ATA, IP Cisco 7940, Cisco IP 7906, ICPC) and assigned the line button on the phone to address specified MAC [2,3,7].

Dial plan CME

To configure, assign the rules for voice dialing assigning a name to the rule, a description of the rule, the translation, the destination pattern, the session destination, protocol and codec used.

Switch Cisco Catalyst 2960

VLANs are created in which they will work, in this case voice and data.

Then you enter a specific port range which will interact in the voice VLAN [2,3,7].

The screen displays the information for the verification of assigning ports to VLANs active. Shown in Table 7.

Switch-VoIP#show vlan					
VLAN	Name	Status	Ports		
1	default	active			
2	Voice	active	Fa0/1, Fa0/2, ----- , Fa0/12 Gi0/1, Gi0/2		
3	Data	active	Fa0/13, ----- , Fa0/24		
150	Super voice	active			
1001	fddi-default	act/unsup			
1002	token-ring-default	act/unsup			
1003	fddinet-dafault	act/unsup			
1004	trnet-default	act/unsup			
1005		act/unsup			
VLAN	Type	SAID	MTU	Parent	RingNo
1	enet	100001	1500	-	-
2	enet	100002	1500	-	-
3	enet	100003	1500	-	-
150	enet	100004	1500	-	-
VLAN	BridgeNo	Stp	BrdgMode	Trans1	Trans2
1	-	-	-	0	0
2	-	-	-	0	0
3	-	-	-	0	0
150	-	-	-	0	0

Table 7

With this configuration, the VLAN are implemented in the switch, the fragmentation of large broadcast domain into several smaller parts reduces broadcast traffic and improves network performance. The fragmentation of VLAN domains also allows greater confidentiality of information [7].

ATA 186 Analog Telephone Adapter

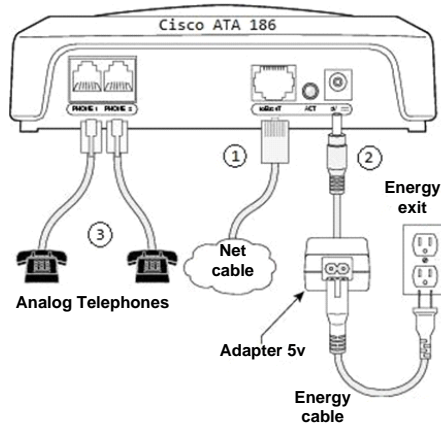


Figure 3

Considering Figure 3, after making the connections you pick up the phone and press the red flashing button located on the top of ATA 186.

Upon hearing the recording dial 20 followed by the pound key (#), which enables DHCP mode. Then restart the ATA 186 by unplugging the power cord for 10 seconds and reconnect. Now it detects the IP address that contains the ATA picking up the phone and pressing the red flashing button located on the top of the ATA 186 [7].

Upon hearing the recording dial 80 followed by the pound key (#) and listen to the IP address that was assigned by DHCP as this 172.16.31.254.

Access the web browser and enter the IP address in the address bar on set of / dev, so it is entered as follows: 172.16.31.254 / dev.

Configuration of Router D-LINK DIR-615Wireless

In this device, the following settings are made:

WAN Interface

To configure the device enter the IP address you entered in the browser address bar and open the home screen D-Link in which data is entered from the installation [7].

LAN interface

When adding the IP address of the internal network which was used as a gateway, you enter the corresponding IP address and save the changes by clicking the save settings button [7].

WLAN Interface

To configure the WAN network, enter setup and within the WIRELESS SETTINGS tab click Internet Connection Manual [7].

Enablement of web filter

To enable network filter select ADVANCED NETWORK FILTER tab, being inside you enter the MAC addresses of the devices that have access to the network, allowing you to block unauthorized access. When entering addresses all devices that will access the information is saved by clicking the save settings button and wait for the D-Link device to restart [7].

Softphone Configuration (Cisco IP Communicator)

Accessing address:
www.fiberdownload.com/Download/19120/Cisco-IP-Communicator. It downloads and runs the installer with extension .Msi [7].

3CX Configuration

Access the following address:
<http://www.3cx.com/VOIP/voip-phone.html>
download and run the installer with extension .msi.

Then open the application. Which appears in the active state and the name of the account you use (see Figure 4a). Now once active, the only thing left to do is try to making a call to any active extension (see Figure 4b).



Figure 4

Supplier evaluation

Figure 5 shows the main information for contract fees and costs of each provider by means of these graphs that differ both in prices and volume, increases or decreases the competitive advantages of each in a fast and efficient way [7].

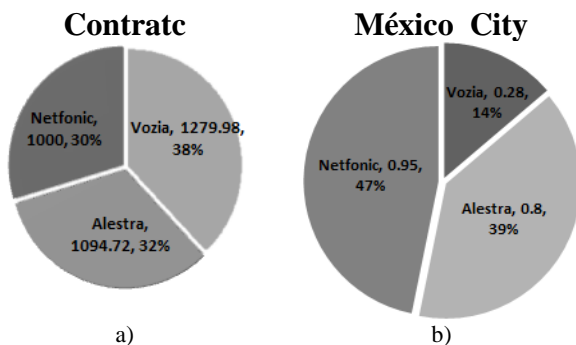


Figure 5

In Table 8. It shows which VoIP service provider is more economical to contract.

Provider	Payment / Serv. Tel.	%
Vozia	2200	17%
Netfonic	2450	19%
Alestra	2604.16	21%
Telmex	5539.62	43%
Price of dollar in Mexico		
Sell	12.8793	
Buy	12.8738	

Table 8

After study, it was decided to make the contract with the supplier Vozia it turned out to be the most convenient.

Operation



Figure 6

Figure 6 is considered to describe the steps of communication service operation:

Step 1. The two participants in the conversation are connected to the VoIP server with their phones.

Step 2. Issuer Team asks for the equipment and receiver using a specific protocol (SIP, H.323, IAX / 2).

Step 3. The VoIP server returns the sender contact information (eg IP number.)

Step 4. Phones establish a connection and agree on a code (G.711, G.729, and GSM).

Step 5. Voice data is compressed and sent by the RTP protocol.

Step 6. Receiver receives RTP packets, decode voice data.

Step 7. Listen voice.

Results

When operating the VoIP service communication tests were deployment between the various forms of telephone communication over the Internet.

Of course, having made a contract for VoIP telephone service provider *vozia* for the purpose of verifying savings. Information was obtained for a comparative costs and calls between Telmex and Vozia suppliers. This is shown in Table 9.

Call	Minutes	Telmex	Vozia	TPM US/MX
Local	100 Calls	\$70MX	\$67.56*MX (5.25 US)	0.035US/0.45MX
Cell phone	150 Minutes	\$119.48M X	\$115.83MX (9 US)	0.060US/0.77MX
National	200 Minutes	\$119.48M X	\$41.18MX (3.2 US)	0.016US/0.20MX
US	100 Minutes	\$179	\$23.16MX (1.18 US)	0.18US/0.23MX

*Equivalent to 150 minutes

Table 9.

As it can be seen in Table 9 the difference is remarkable. From local calls there are already savings and is more significant in national calls.

Implementation of VoIP services is a foundation on which calls can be made internal extension to extension, local, mobile, national and international with the lowest market economic MX ranging from 0.25 cents to 0.77 cents MX this means great savings on a comparison against traditional telephony, following costs are provided to certain types of calls:

Internal Call: From extension to extension. The call is free extension to extension has no cost and can be performed even without having access to the Internet, you only need access to the local area network (LAN).

Local Call: In extension to the local town of Valley Santiago. A local call to the VoIP service provider Vozia considered in this case as OFF NET (off-grid), meaning that it does not belong to the list to make inexpensive calls since it is one of the main cities of Mexico. Its estimated cost in dollars is \$ 0.45 perminute.

National call: In extension to the city of Mexico is considered ON NET (In-Network) in major cities so there are already so many surcharges of what is known as interconnection . The approximate cost in pesos is \$ 0.206perminute.

Cellular call: From extension to a mobile phone. In what is a call to a phone company is the rate that is considered more expensive because it is the interface even for phone companies still conventional lines is very cheap. Its estimated cost in dollars is \$ 0.771 per minute.

Calling abroad: The cheapest calls can be made precisely to other countries since the servers that route calls are mainly in the united states.

That is the reason that the call to Mexico is more expensive to call a foreign country as is USA. Approximate cost in pesos to call Washington, USA is \$ 0.231 per minute.

The server was changed to a local network that is a private IP. When calls are generated problems arose since the NAT was interfering with communication with the SIP protocol.

Therefore, we chose to use the protocol IAX2 as the server *vozia* is Asterisk could be used without problems, the advantage of this protocol is that it is lighter than SIP to transport themselves on the network because it eliminated some packet headers, supports NAT no problem which is what I mainly wanted to get, in other words quality voice transfer at low cost, but its drawback is that it is standardized and so far only devices that work with Asterisk can use so there is not much hardware or software to work with IAX.

Conclusions

Two different technologies have integrated, such as Call Manager Express and Asterisk. And so, getting calls from both extensions and out and have the services of call forwarding, call waiting, etc.

Also managing the service without relying on a vendor that you can continue adding extensions, manage voicemail service, answering machine settings, see the call states as records, accounts, who else called extensions, reporting for period of time determined by the user, the place that was called and the caller.

It's a proposal to improve the state of networking in an organization and provide better service and more secure Internet, scalable and focused on Quality of Service (QoS) where they can be properly integrated converged services.

The SIP protocol security IAX was changed to keep unauthorized people from using service outside the organization so that they could make calls. And although it is lighter, has the disadvantage that it is not standardized. Making this change could keep the server on a local network with private IP respective communicate without problem VOZIA VoIP server even with some improvements in the network traffic.

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