DESIGN OF IP TELEPHONY SYSTEM BASED ON FREE SOFTWARE BETWEEN SAN ANTONIO CREDIT UNION LTDA AND ITS BRANCHES

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Abstract- This project aims to make the design of an IP telephony system based on free software, which allows the transmission of voice packets over the IP protocol; through the data network of San Antonio credit union LTDA., and their branches, allowing cost savings in telephone lists and optimizing network resources.

I. INTRODUCTION

Networks today support different services and applications such as voice, video and data over a common infrastructure, thus saving resources; it is for this reason that arises in the design of an IP telephony system for the San Antonio Credit Union LTDA.

Currently in the Cooperative are not using the existing network resources as for telephone communication branches with the matrix is carried out through the public switched CNT telephone network, generating monthly expenses, plus they are not using the network resources such as bandwidth of links employed for transmitting databases.

A free software solution for IP telephony system, can integrate voice packets to the IP network, allowing all services have a similar exchange.

II. BASIC CONCEPTS

A. *VoIP Definition*

Voice over IP networks (often referred to by the abbreviation of Voice over IP "VoIP"), is a technology that allows voice communication via any network that accepts the IP protocol used on the Internet.

B. *Building a network for VoIP*

Mainly in a VoIP network three fundamental elements described below are defined.

Terminals: are end users who establish communication through terminal equipment is the software or hardware which should include signal processing, the encoding decoding, sent to the data network.

Server: This is the center of the architecture of VoIP, as it handles call control, routing management system supports call routing through the network, in addition to the management and control of services. The VoIP server is called according to the signaling protocol that is used if the case would be named H323 gatekeepers.

Gateway: The gateway of communication between the traditional telephone network also known as PSTN (Public Switched Telephone Network) and VoIP network which features LAN and FXO ports, acting transparently to the user ports.

C. *Advantages of VoIP.*

The main advantage of VoIP is cost reduction for phone service, since calls are made through the network, existing data, optimizing resources is to transport voice and data on the same infrastructure network.

Thanks to the role that the VoIP Codecs, data packets require low bandwidth, since the silent spaces are filled with data enabling the most efficient use of bandwidth.

VoIP provides an outline of flexible network because it does not need to have a specific network topology, enabling integration with large IP networks. VoIP applications enable monitoring and auditing making it easier to control the use of the application by the managers of the company. It also allows users to be mobile just by moving your phone to another network point and scalability facilitating the incorporation of new users.

D. *Disadvantages of VoIP.*

In the VoIP call quality is slightly inferior to the telephone PSTN because some packets are lost, they are lost or are likely late challenge, this happens with the use of IP protocol does not guarantee that the information reaches the destination which lowers the quality of the calls.

It may occur delay variations or deterioration of communication, produced by network congestion and congestion for a low speed connection.

The network must necessarily have a battery bank and a generator, because if a power failure occurs it would be without telephone service across the enterprise.

E. *IP PBX CENTRAL*

IP PBXs are plants that have the same features as a PBX, but have the ability to connect to the PSTN and a LAN over IP protocol, can be integrated into a single network infrastructure sending voice and data and integrating new services.

The IP core will consist of a VoIP software being centralized connections, which are connected directly to the conventional telephone network via trunk lines and VoIP network, enabling seamless communication for end users.

F. *Protocols used in VoIP*

Protocols are rules set for the communication of two points in this case two IP phones. The protocols to be discussed are those used or directly related to VoIP, to start the IP protocol that is the main Internet protocol network layer, the UDP transport protocol layer protocols real time application layer, specifically signaling protocols and VoIP protocols to end QoS for VoIP.

IP Protocol: a connectionless protocol, does not correct errors leaving this process to higher layers, this protocol has three basic functions which are; addressing, routing and fragmentation.

UDP protocol: is a connectionless protocol, real-time applications used, since it allows better communication, so that the sender knows that the packet has reached the destination for this key feature is the protocol ideal for multimedia transmissions of video and audio transport.

RTP protocol: is part of the application layer function is to multiplex the data stream in real time on your single stream within the UDP segment, its head has the time-stmping field which allows the source to associate a timestamp reducing effects of fluctuating. This protocol is designed to work with the RTCP

.

Signaling Protocol H323: it is generally used when transmitting voice over IP, as it clearly defines the components that are used in a complete system of IP telephony, as they are from the terminals, gateways that are responsible for connecting to external networks functions as a gateway, the control unit and gatepeer functioning as a central, in charge of the establishment management and administration of the connections.

SIP signaling protocol: is based on a client-server used for communication, the exchange of messages, the client sends SIP requests to the server, its main components are two terminals and server architecture

G. *Problem in a net of VoIP*

In the case of the traditional telephony, a call permanent physical circuit so the voice quality is guaranteed is established, on the contrary happens in the case of IP telephony, and which by their nature are not oriented q connection and do not ensure the packets arrive at their destination may be lost or arrive out of order, all these problems degrade the real time signal, which are sensitive to delays. Table 1 VoIP problems agree to model layers TCP / IP.

Table 1. Distortion or noise sources in the various layers of the network

<table>
<tr><td>Aplication</td><td>Background noise</td><td colspan="3">Saturation</td></tr>
<tr><td>Transport</td><td rowspan="2">Distortion amplification</td><td colspan="3">Coding distortion</td></tr>
<tr><td>Net</td><td rowspan="2">Packet loss</td><td rowspan="2">Jitter</td><td rowspan="2">Delays</td></tr>
<tr><td>Data Link</td><td rowspan="2">Signal attenuation</td></tr>
<tr><td>Physical</td><td colspan="3">Electrical interference (static)</td></tr>
</table>

Source: MORO, network infrastructure and telephony systems, 2013

Delay: latency or time slots are being introduced from the beginning of the conversation, through each of the stages of the transmission like; coding, processing produced by network equipment in the encapsulation process and exchange packet header, to the sum of time intervals generated by the system are called delay or latency.

Jitter: the effect of the variation in delay or latency in the backbone of the network due to congestion of it, the loss of synchronization and the rerouting of the packets, ie packets do not arrive in time which it was calculated to come. This effect is perceived by end users as a breathy voice communication.

The echo is caused by the effects of delay, jitter and electronic components of the analog parts of the system to reflect a portion of the processed signal. This effect is visible to the users as they begin to hear what he speaks, in a delayed version.

Packet loss: the percentage of packets lost in the network, either because the packets are timed TTL or network congestion, these packages are not broadcast because real-time distortion causes voice so this should not exceed 1%

H. Quality of Service (QoS)

Quality of service is the ability to provide better service to end users, in the case of IP telephony is the quality of the voice network it provides end users. Prioritize traffic according to their needs, about the different technologies optimizing network resources.

By the nature of real-time data such as voice, video, are sensitive to delay, packet loss, change in delays (jitter), should be applied quality of service, which directly relates the size of queues congestion network and switching speed and bandwidth links. There are three models (QoS) quality of service which are:

The Best-Effort model: that has no policy for quality of service, using its best to send packets to their destination without delay and ensure that there are lost packets, use the FIFO method (First-in fisrt -out), which is a storage technique and sent when there is congestion stores and sends maintaining the order of arrival.

The Integrated Services model: based on the reservation of network resources and signaling the entire path, each router that traverses the network performs the requested reservation. For booking and signaling IP packets flows called the RSVP protocol (Resource Reservation Protocol) is used, the main problem of this protocol is scalability as it has to maintain state information in each router.

The Differentiated Services Model: a model latest QoS, which is based on the use of multiple classes of service, for which it uses different methods such as IP presedence, DSCP (Code Point Differentiated Services The services architecture. differentiated two types of routers, edge routers that are in charge of the process of marking and traffic prioritization and internal routers to avoid congestion.

I. IP Telephony Platforms

The main platforms for IP telephony in free software are:

Asterix: is a software PBX, developed by the company Digium for servers implementing VoIP is distributed under open source license, including basic characteristics of a plant, such as: auto answer, call transfer and parking of calls, voicemail, call queuing, caller ID.

TrixBox: is a distribution of GNU / Linux operating system, is based on Centos and evolved core Asterisk, was developed by Mark Spencer of Digium, the system is ideal for small companies, easy to administer because it has a graphical platform called Free PBX has a fax-to-email Apache web server with PHP and Perl support, database administration, Voicemail and integrating this with email and integration.

Elastix: it is developed and managed by Palo Santo Solutions a company established in Ecuador, is a distribution of Free Software Unified Communications, four software integrates in different packages features like: Asterisk, Hylafax, Openfire and Postfix, providing functions a PBX center as Fax, Instant Messaging and Email, respectively.

III. CURRENT STATUS OF DATA NETWORK AND NETWORK TOPOLOGY

The data network of San Antonio credit union LTDA. Has three LANs, one for each branch and parent Cooperative, for connectivity have hired two fiber optic links to the company Telconet.

The LAN management is centralized in the equipment room of the parent Cooperative, in the department. It has an internet connection 3Mbps provided by the company Telconet, via a fiber optic link that reaches a transceiver and a router, these teams are owned by the service providers, which is shared with the branches through fiber optic links they are also rented to the same company.

The data network is independent of the telephone system which means that telephone costs are high since for communication with the parent branch services conventional PSTN telephony is used, and no resources exploits the existing network, since the cooperative has hired fiber optic links for the transmission of the databases of financial systems and to share Internet service mainly. In Figure 1 the network topology of the matrix is shown.

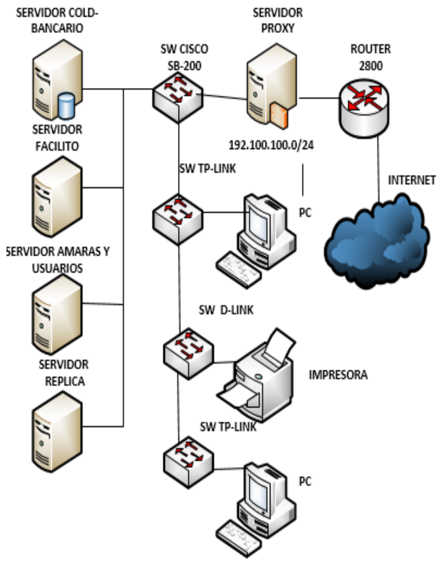

Figure 1: Network topology matrix
Source: Systems Department of the Cooperative

LAN Cooperative has a star topology network with a category 5E structured first floor and 6th on the second floor wiring, uses a logical addressing class C with a range of 192.XXX / 24 addresses, which is not segmented so that management is difficult.

A. Department San Antonio Matrix Systems.

In the matrix of the credit union is where most offices and employees is concentrated because that started 54 years ago in the parish of San Antonio, in this financial institution working 35 employees in different departments, specifically for the IT department in charge two engineers in systems, which handle financial systems, databases, servers, network equipment and within this department is immersed the equipment room.

To determine the current status of the equipment room, it is necessary to inventory the network to determine the existing telecommunications, shown in Table 2.

Table 2 inventory of equipment bathroom equipment

QUANTITY	DISPOSITIVE
4	Servers
4	Switch
1	Firewall
1	Router
1	Transceiver
1	Air conditioner
1	Extinguisher
2	Web cam
1	Telephone exchange
1	Smoke detector
1	UPS. TRIPP-LITE 6Kva

Source. Systems department

B. Equipment Connectivity

In the fourth communications equipment Cooperative besides the own equipment you have two teams, owned by internet service providers and service of optical fiber links. Then the equipment is described.

Router HP A-MSR 900: serves to divide in internet service interfaces and service data link, and that through a single optical fiber reach the two services, this router is configured subnet matrix Cooperative and gateways it.

Transceiver TP-LINK MC112CS: which is responsible for transforming the light signal; an interface for this team reaches the optical fiber, and the other interface will get the category 6 UTP cable which connects to the router HP single source.

Access Switch: The function of these devices within the LAN is to connect the entire network; servers, web cameras and all employees of the cooperative, allowing you to share files, share internet, share printers and IP mainly access the databases of the financial system. In Table 3. The details of the switch showed, makes and models.

Table 3. Equipment connectivity matrix

Quantity	**Model**	**Number of ports**
2	Switch Cisco SG200-26	2 Puertos Gbit 24 Puertos 10/100 Mbit
1	Switch TP-LINK TL-SF1024	24 Puertos 10/100 Mbit
1	Switch D-LINK DES-1024D	24 Puertos 10/100 Mbit

Source. Systems department

C. *Services and Applications Network LAN*

Applications that are used in the San Antonio credit union are mainly financial systems as Facilito system Conexus milenium systems, flexible financial system paid; remote access system Ultra VNC, besides these are applications of Microsoft Office, Internet browsers, Kaspersky antivirus, the website is hosted on a hosting leased from the same company that provides Internet services, ie, the company offers service web server, email and corporate service Sparck chats. Table 4 shows the main applications are shown in the Cooperative.

Table 4. Description of server applications

Server	application
Server Cold Banking	Servidor de Base de datos del sistema financiero
Server Facilito	Servidor de cobros de la red facilito
Server webcams	Servidor de cámaras
User server	Levanta el servicio contable

Source: Systems department

D. *Cabling Voice and Data Network*

The facilities structured matrix wiring is performed in two stages, according to the needs growth has progressed over time of this Cooperative, also due to changes in the infrastructure of the building and in their beginning had one level, and then the entire building renovated ten years ago, also have made the installation of new network points according to the requirements of new staff.

The first part of structured cabling has been carried out in the year 2007, with UTP Category 5E, all network points; Voice and data are certified and labeled, and the cable is routed from the equipment room through PVC pipe, toward the face plates of the various offices on the ground floor of the Cooperative.

The second stage of the wiring was made in 2011, with UTP Category 6 also all points are certified and labeled, they are distributed through tube corrugated plastic, it is distributed to electrical users on the first floor of the different departments.

E. *Current Situation Telephony*

Currently in the parent Cooperative Telephone communication has an analogue exchange PANASONIC KX-TES 824, which provides internal communication between extensions and allowing interconnection with the branches, with the switched telephone network and a cellular output has trunk capacity of eight and twenty-four extensions which provides basic services a central including: Auto Attendant call, call queuing, call forwarding, call recording and printing.

Twenty-four phone extensions that are assigned by the PBX are fully used by employees, as there has been an increase both partners and users in recent years, and extensions are out for new employees. Table 5 numbers with corresponding extensions communicating detailed offices.

Table 5. List of extensions of the matrix.

Departament	Extension number
Información	101
Créditos 1	102
Créditos 2	103
Créditos 3	115
Créditos 4	116
Jefe de Crédito	118
Asistente Contable 1	105
Asistente Contable 2	113
Asistente General	106
Sistemas	107
Cajas	108
Jurídico	117
Riesgos	110
Captaciones 1	111
Captaciones 2	119
Auditoria	120
Gerencia General	121
Jefe de Cajas	124
Jefe de Talento Humano	109

Source: Systems department

F. *Actual Situation Data Network branches*

The data network has a star topology Ethernet type, have the network segment 192.XXX/24 are not segmented, and all transactions that take place throughout the day are stored in databases that are located in the equipment room of the matrix in San Antonio, these data pass through the optical fiber links, also these same databases are stored on the server replies that the branch is located in Ibarra.

The capacity of the links is 1 Mbps, from the branches to the parent San Antonio through dedicated fiber optic through which data and internet courses, have hired backup links in the event of the fall of the service provider links . Branch topology shown in Figure.

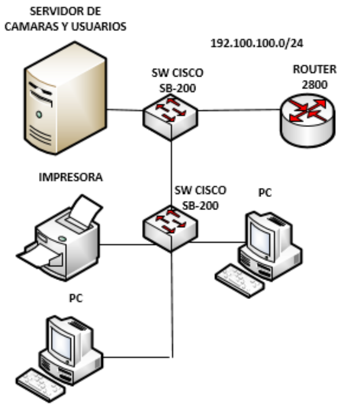

Figure 2. Topology branch
Source: Systems department

G. Current situation Telephony branches

Branches have analogue telephone exchanges PANASONIC KX-TES824, which is connected to two telephone lines or trunks having up to 8 extensions, which are detailed in Table 6, the number with the respective department.

Table 6. Extension List of branches

Branche Ibarra	
Departamento	Número de extensión
Información	101
Créditos	102
Captaciones	103
Jefe de Crédito	105
Branche Atuntaqui	
Información	101
Créditos 1	102
Créditos 2	103
Créditos 3	104
Jefe de Crédito	105

Source: Systems department

H. Analysis of telephone traffic

For analysis it is essential to know how the telephone traffic behaves even phone calls appear at any point in time, so we proceeded to count the number of outgoing and incoming calls that are made of branches into the matrix and between branches Cooperative, this was made based on the call details provided by the National Telecommunications Corporation.

I. Determination of rush hour

The peak time is defined as the period of time where the traffic reaches the maximum values to determine manually counted the call detail, with the following results. Most telephone traffic is generated from five phone numbers to the parent branch. The traffic generated from branch to branch is low. Furthermore the average call duration is two to three minutes. The analysis was performed every two hours with the result that there is a peak time for a total of eighteen calls between the hours of 10:00 to 12:00.

J. Analysis of traffic in and out

Telephone traffic is defined as the accumulation of phone calls in a circuit group is associated with the occupation; it is said that if you are busy studying telephone traffic is find your traffic intensity (A), you must first calculate the average time (T).

$$T = \frac{\text{duración total de llamadas}}{\text{total de ocupaciones}}$$

Equation 1 Average time
Fuente: (GERRERO Julio, 2002)

To find the traffic intensity (A) is defined as the traffic volume that presents a number of circuits and is defined by Eq. Table calculating traffic intensity shown.

$$A = C * T$$

Equation 2: Intensity of traffic
Source: (GERRERO Julio, 2002)

Where C is number of calls per hour and T is the average time, and unity is the Erlang is a dimensionless measure used in telecommunications. By definition is the total occupancy for one hour equals 1 Erlang.

$$1(\text{Erlang}) = \frac{t.n}{60}$$

Equation 3 Equivalence of a Erlang
Source (HUIDOBRO, 2008)

Day more intense telephone traffic is Friday between the hours of 8:00 to 10:00 am with a score of 0.027 Erlang. Similarly outgoing traffic is calculated by determining the Wednesday between the hours of 10:00 to 12:00 the sum of the intensities of traffic yielded a result of 0.024 Erlang

K. Determination bandwidth

The bandwidth needed for the two links of the cooperative, which connects the array with the two branches; we need to know two important factors such as the number of simultaneous calls and compression options for use that is the codec.

To calculate the bandwidth required the following formula is used:

Voice Bandwidth = (payload + L3 + L2)* 8 * pps

Equation 4: Bandwith
Source: (MARCANO, 2012)

Where:
Payload: The load bytes generated by the CODEC.
L3: Headers layer 3 and higher layers in bytes.
L2: link layer header in bytes
8: number of bits having 1 byte
Pps: packets per second rate generated by the CODEC

To calculate the required bandwidth should take the number of simultaneous calls the Voice Bandwitch (bandwidth voice codec applied) into account. Then you have to:

BW (requerido) = simultaneous calls *2*Voice Bandwidth

Equation 5: Bandwith full Duplex
Source: (MARCANO, 2012)

Then the bandwidth required for link San Antonio - Ibarra with a maximum of 18 calls must be 1.9 Mbps is required, and the link San Antonio - Atuntaqui with maximum of 20 calls 2.1Mbps is needed.

Statistics on the percentage of use of transport layer protocols are also determined such as TCP and UDP protocol. The results are shown in Table 7.

Table 7. Summary utilization rate protocols.

Date	Tráfico UDP	Tráfico TCP	Otro tipo de tráfico
7-01-2014	10,77 %	89,09 %	0,15 %
12-01-2014	10,46 %	89,37 %	0,17 %
13-01-2014	11,18 %	88,65 %	0,17 %
14-01-2014	9,94 %	90,14 %	0,12 %
15-01-2014	9,14 %	90,75 %	0,11 %

Source: Based on the program wireshark

TCP traffic is the highest percentage circulating in the network, occupying up to 90% of the busy traffic, as can be seen in the graph and utilization statistics protocol.

IV. SYSTEM DESIGN IP TELEPHONY

For the design dela telephone network is necessary to propose own equipment which are configurable, compatible and scalable, with the equipment already exists, for this design a simulation, which has policies QoS guaranteeing phone service was used IP quality.

A. *Hardware Requirements*

In the hardware requirements is taken into account, the equipment needed in the design of the IP telephony system to propose a design that meets the quality requirements such as: the server, the IP telephones and equipment required network.

B. *Server specifications*

To choose the server is required to review the technical minimum requirements that must be the IP telephony system, according to (Meggelen, 2007), the size of the system is not determined by the number of users, but the number of simultaneous calls or concurrent it expected to support the system. server for small businesses will be chosen, because the number of simultaneous calls does not exceed twenty five calls, since the average number of simultaneous calls generated by the Cooperative is five, of branches of Ibarra and Atuntaqui, to the matrix Cooperative in San Antonio.

C. *Selection and server features*

For server selection three brands that provide, Elastix Xorcom Grandstream IP servers, opting for the Elastix server ElX025 as it meets the needs of the design and can be configured all future applications must be analyzed.

D. *Specifications IP phones*

For the selection of the three major phone manufacturers IP phones will be considered as: Astra, Grandstream, Yearlink, these must comply with the essential requirements such as call waiting, call forwarding, supporting codecs and signaling protocols needed to the present design; lso which it has been chosen Grandstream GXP1450

E. *Choice and characteristics of the card*

The telephone cards serve the function of Gateway allow communication, to the PSTN network, ie to the traditional telephone network for the election of the cards have been taken has few interfaces, of which we have chosen the Open Vox card A1610P

F. *Election and features Layer 3 Switch.*

The cooperative does not have any equipment layer 3, allowing routing network; this is the reason why the network is not subnetted, has no vlans, no QoS policies makes it difficult administration. For the design of an IP telephony system it is necessary to have a Layer 3 switch in each branch and the parent to suggest quality for voice, for this the Layer 3 switch must have QoS characteristics.

For design proposes a Cisco Catalyst Switch 3560G 24 Pts WS, which provides in addition to the characteristics of a switch, acts as Layer 3 routing, has 24 ports that can be used for IP telephony devices, or any end device.

G. Dial Plan

The dial plan determines the number of extensions used in the credit union San Antonio LTDA. For the parent and its subsidiaries. For the dial plan takes into account the same number of digits they had before three digits which are organized according to the functions performed by users and their location.

The first digit identifies if a parent or branch, the second digit identifies the officers of the Cooperative according to the areas of work and finally the third digit identifies the department belonging according to the functions performed within the Cooperative, these extension numbers are used for both internal communication between departments, such as for external communication.

H. IP Addressing Plan

To design the IP telephony system is necessary to separate voice traffic on a subnet, which is exclusively for the use of voice over IP. It is for this reason that three different subnets separated them according to end-user functions of the Cooperative is proposed.

It has been considered to use the same network domains, only been subnetted into subnets, this in addition to providing quality for Voice over IP reduces the broadcast domain that is raised by having a flat network especially in the womb where it has the largest users of the network.

They have been divided into three groups and finance management one which contains them departments management, accounting, reporting, cash, credit, collections, risk, audit, information; other systems is where will all servers, network equipment, IP cameras, IP printers, biometric readers and finally a subnet for all users of IP telephony, as shown in Table 8

Table 8. IP addressing plan for Cooperative Savings and Credit Network San Antonio.

Matriz San Antonio		
Departamento	Direcciones de red	vlan
Telefonía	192.100.100.0/26	10
Gestión	192.100.101.0/26	20
Sistemas	192.100.101.128/27	30
Branche Ibarra		
Telefonía	192.100.100.0/26	10
Gestión	192.100.101.0/26	20
Sistemas	192.100.101.128/27	30
Branche Atuntaqui		
Telefonía	192.100.100.0/26	10
Gestión	192.100.101.0/26	20
Sistemas	192.100.101.128/27	30

Source: Prepared by Ms. Veronica Collahuazo

I. Planning QoS policies for handling voice in IP telephony.

The objective of implementing quality policies telephony service is to prioritize voice traffic and optimize bandwidth, IP telephony uses real-time protocols such as UDP and RTP, the process for the application is filter this traffic through Access List thereby classifying traffic.

Once classified traffic priority levels are assigned to the DiffServ model for real-time applications, to define the boundaries of trust the quality of service is set as close to the source of network traffic and finally proceeded to choose equipment physical layer, those who support PBX as the core service, the analog card, and phones.

J. Election model QoS.

It has chosen to use the Diffserv model, because it is a scalable model and is based on the classification of traffic through the use of PHB (Per Hop Behavior), since the network resources are limited and do not have the bandwidth need to be reserved as suggested by the model.

In the first model Diffserv traffic is classified, it is marked using DSCP values which has 8 bits and can be assigned 7 levels of priority, to voice the assigned level 5, since the level 7 is not defined and level 6 is reserved for future applications, which may take up to 64 combinations.

To apply the Diffserv model, it is configured in the equipment side of the Cooperative, but to ensure that when passing through the voice traffic nodes providers links TELCONET, these teams are in the ability to read headers frames where the priority is assigned to the voice quality policies.

K. Election of traffic classification method

Lists to filter traffic using access control which can be standard or extended, extended ACLs are used because they specify in addition to allowing or denying filtered according to the port. ACLs allow sorting, limit, control traffic, thereby improving network performance and provide a basic level of security.

The ACLs are executed in the order, is allowing the general to the particular, because in the end it has an implicit ACL that denies all traffic that is not permitted, this is how traffic is filtered. Access lists are made based on the classification of TCP or UDP ports.

L. Choosing a method of marking traffic

As mentioned above the packet labeling process is performed through the DiffServ Code Point, once traffic filtering proceeds to dial through IP Precedents, in which seven priority levels set

To determine the DSCP values for each of the classes defined policies must be established, in which the treatment received by each specified. This treatment performs various functions such as dialing, police, queuing or any other function of DiffServ. The marking of packages was made based on some considerations of the baseline configuration CISCO QoS.

For determination of DSCP values, it is done based on operating procedures and based on QoS considerations CISCO. DSCP values shown in Table 9.

Table 9 Values for traffic marking

Priority	Application	Valor DSCP
Crítica	Telefonía ip	EF
	Señalizacion	CS3
Alta	Bases de datos	AF31
	Aplicaciones web	AF33
Default	Cualquier otro	0

Source: (SEVILLA, 2010)pags: 812-813

After defining values for marking different types of traffic in the network, the Class-Based, Packet Marking mechanism whereby an efficient packet marking is provided is used.

V. CONCLUSIONS

Through the analysis of the network infrastructure of the matrix and the branches of the credit union San Antonio LTDA. telephony needs determined by identifying the number of users of telephony, network topology and bandwidth use for the correct dimensioning of network resources. In addition the teams have to determine the use of the same in the new design and propose equipment that may be required and comply with design needs enlisted.

In the analysis of data traffic, it was determined that the use of the bandwidth is low on the links, with averages of 500 kbps, the total capacity of 1Mbps, you have peaks that if they use the full capacity of the channel, but applying quality of service policies, they are reduced since 90% of the generated traffic is TCP; so the links if they are able to transmit voice traffic over IP and you could say that the links are being underutilized.

A comparison of the three major IP telephony platforms was performed; Asterisk, Trixbox and Elastix software free with the IEEE 830 standard, by which it was concluded that Elastix is the platform used for the design of IP telephony system for its features like scalability, additional services such as messaging and advanced Instantly, call center, fax, email, offering and can be implemented later, also has official support plans if necessary which cost $ 70.0 time and online there are plenty of documentation and support forums .

Free software platforms currently are developing similar software solutions paid because they offer the same services of a PBX such as Unified Messaging, Call Recording, Call Center (Call Center), voice mail, fax; so, if Elastix resource poor memory, processor and disk space according to the needs of each company we have a PBX and additional services without purchasing equipment particular make and pay any licensing, permitting saving financial resources in private and public industries.

The proposed system design integrates IP telephony voice and data on a single network infrastructure; this being a great advantage over conventional telephony that currently possess; since the bandwidth data links contracted optimized and economic resources generated by phone calls between the parent and subsidiaries is saved.

To design the IP telephony system has been considered to maintain network equipment having the Cooperative access layer and has proposed new equipment layer three for the design of the VoIP network, as they are indispensable for the administration and configuration of QoS policies. With teams raised and integrate the VoIP network; implicitly with the configuration of VLANs, quality policies on network performance is improved because it has been divided into logical workgroups grouped users with similar to the same VLAN, both the parent and branch requirements;

reducing unnecessary broadcast traffic also improves safety and facilitates administration.

Through simulation we proceeded to test the design of the IP telephony system, the Elastix server was configured in a virtual machine containing the settings for all users telephony branches and die, through the program GNS3 network was simulated , prioritizing voice traffic and a phone call between two extensions was made to carry traffic over the network RTP; where it was found that the quality parameters are within permitted by ITU, such as packet loss is less than 1% and jitter is less than 75ms.

The quality model Diffserv service, divide traffic into different classes and assigns priorities according to applications in architecture two components of edge routers and core routers in the network, in border routers are defined; which in this case would be the switch to layer three of each branch that are proposed for the design should make the classification, labeling and the establishment of policies, core routers are property providers which links through a SLA (Service Lavel agreement), or contract must guarantee and plan the sending of each packet based on the marks they put border routers.

For the engineer who is in charge of the administration of the network, both branches and parent of the Cooperative can configure the IP telephony system; it has been documented configurations of equipment and performed the respective manuals administrator.

With reference budget system IP telephony financial and technical viability of the project, with the calculation of parameters such as internal rate of return where it was found that the value of the investment would be recouped in seven years, and was determined value 1.56 cost-benefit indicator, which indicates that it is feasible and cost-effective implementation of the project design telephony system for future implementation.

VII. RECOMMENDATIONS

To ensure that the branches of the credit union San Antonio LTDA. they can communicate to the parent, it is recommended to hire a redundant link from another communications company because if you drop the communication link is lost.

It is recommended to increase the capacity of the link to 2M bandwidth service providers to satisfy all the needs of the IP telephony and avoid clipping.

For an efficient data network, it is recommended to buy at least one team in each branch layer 3 and matrix; to manage according to the needs of the cooperative and not rely on service providers to any changes in the network and is not convenient to have a flat network because it can’t classify staff according to their functions.

The complete design of the network consists of configuring voice VLANs, and management systems, as well as quality settings, which improves network performance, but for that network management is comprehensive; the use of management software is recommended to monitor the status, availability and uptime of servers, computers, network equipment; review the use of the bandwidth of the parent and its subsidiaries, which provides an overview of the network and helps troubleshoot performance issues.

To ensure IP telephony quality of service is essential to set quality policies in a layer switch three and proceed to delimit the boundaries of trust, that is set equally in the access switch and finally acquire the server and suitable phones that bear the quality protocols.

To train the personnel department of the Cooperative systems for the management server, how it works and modifies, create and update the software extensions Elastix if necessary as it may file further improvements.
For proper operation of the network Voice over IP, you need to divide the network traffic, by subnetting or right through VLANs, as a security measure and to prevent possible errors or failures of the data network and alter the function the voice quality.

The telephony server must be placed the equipment room, where only the administrator can enter the network and to use robust passwords with more than 7 digits, including uppercase and mixed for access to numbers and networking equipment for access to the server.

REFERENCIAS

[1] AMAYA, J. (2010). Sistemas de información gerencial. Bogota: ECOE.
[2] ANDERRUTHY, J. (2009). Telefonía IP. Barcelona: ENI..
[3] BARCELÓ, J. (2008). Protocolosy aplicaciones internet. Barcelona: UOC.
[4] CARBALLAR, J. (2009). VoIP, La telefonía en iinternet. Mexico: Thompson.
[5] ESPAÑA, M. (2008). Servicios avanzados de telecomunicación. Madrid: Diaz de Santos.
[6] GARRISON, K. (2009). Trixbox CE 2.6. Birnimgham: Packt.
[7] GERRERO Julio, R. M. (2002). Telefonistas. Sevilla: MAD, S.L.
[8] LÁZARO, J. (2005). Fundamentos de Telemática. Valencia: Editorial de la Universidad Politécnica de Valencia.
[9] MEGGELEN, J. (2007). ASTERISK. United Stated of American: Miki Loukides.
[10] MORO, M. (2013). Infraestructurasde redes y sistemas de telefonía. Madrid: Paraninfo.
[11] NAGIRREDI, S. (2008). VoIP voice and fax singal processing. New Jersey: WILEY.
[12] SEVILLA, A. Y. (2010). Redes CISCO. Guia de estudio para certificación CCNP. Ra-Ma, 2010.
[13] SIVANES, F. (2010). VoIP, Servicios en red. España: Paraninfo.

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