SYSTEM DESIGN IP TELEPHONY UNDER A FREE SOFTWARE PLATFORM FOR INDUSTRY FLORALP S.A. CITY IBARRA

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***Abstract-** This document discloses the design of the IP telephony system under a free software platform for the floral industry S.A. The design includes the choice of software for telephony, sizing her trunk, dial plan, recommendations telephony equipment and the cost benefit analysis for the system.*

***Index terms.** - IP Telephony, Elastix, Asterisk, IEEE 830.*

I. INTRODUCTION

Traditional phone service has been partially replaced by the use of calls that use the Internet as an economic alternative. FLORALP S.A. industry uses traditional analog phone for communications with other branches within the country, working independently in each of its branches with a data network, sub using resources for communication.

The IP telephony system mainly provides economic benefits because it saves calls to only use the services of Internet and telephony No more; Additional services that provide IP telephony is to improve internal communications as the network becomes flexible and scalable to have instant messaging, call waiting, voicemail between the principals.

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II. BASICS CONCEPTS

A. *Definition VoIP*

Sending voice packets in real time using networks working on IP protocol is called VoIP (Federal Communications Commission, 2009), the same that enables communication between two or more users who are using the same network Internet, without requiring any dedicated single cable, does not require a fixed path and is not limited to the expansion of a network.

B. *Arquitecture*

Major feature of VoIP is handling two types of architecture from the point of view of distribution, these are distributed and centralized architecture. Thus allowing the user to adapt the current network to the VoIP services provided.

Centralized architecture

From some points of view centralized architecture is considered as acceptable, because the administration is focused like call control, simplifying the flow of calls repeating the voice characteristics.

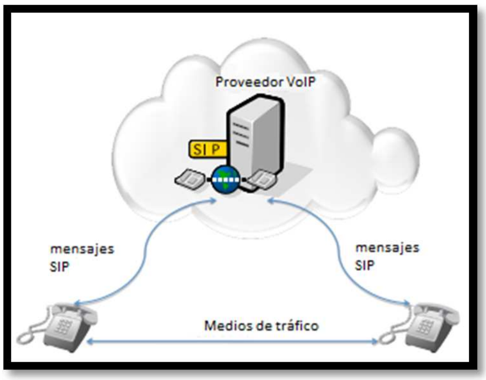

Figure 1. Centralized architecture SIP
Reference: Modified from Distributed Media Server Architecture for SIP using
IP Anycast. Recovered: http://goo.gl/MTTtQ1

Distributed architecture

This type of architecture is associated with H.323 and SIP protocols, the same that allow intelligence to the network through the terminals as IP phones, media servers, any device that can initiate a VoIP call.

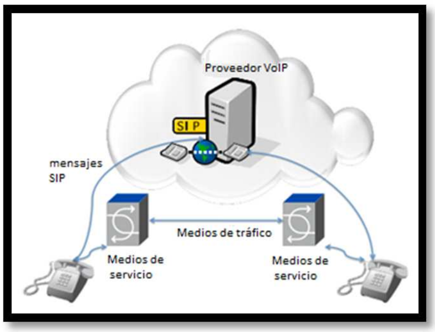

Figure 2. Distributed architecture SIP
Reference: Modified from Distributed Media Server Architecture for SIP using IP Anycast. Recovered: http://goo.gl/MTTtQ1

C. Audio codecs for VoIP

A codec is defined as a set of algorithms responsible for compression and decompression of audio analog to digital bits, in order to compress the signals to occupy less space within a transmission and thus be used in a computer, electronic devices and networks IP getting a good sound quality (EcuRed, 2014)

Table 1. Key features audio codec

Codec	Bit rate (Kbps)	Bit for frame	Compression Type	MOS	Compression delay (ms)
G.711	64	8	PCM	4.1	0.75
G.722	64	14	SAD- PCM	5	0.0625
G.723	5.3	189	ACELP	3.8	30
	5.3	189	ML-MLQ	3.6	30
G.726	32	4	ADPCM	3.85	0.125
G.729	8	80	ADPCM/CS-ACELP	3.92	10
ILBC	13.3	400	LRC	3.8	30
	15.2	308			30
GMS	13.3	260	RPE-LTP	3.6	20

Reference: VoipForo (2014) Codecs. Recovered: http://www.voipforo.com/codec/codecs.php

D. VoIP protocols

As data networks are managed based on a protocol stack and applications working on the VoIP protocols that fit technology and user requirements. There are two main types of VoIP related protocols: transport and data control, they work on the transport layer.

SIP protocol

Session Initiation Protocol (SIP) is a protocol developed by the IETF specified in RFC 3261, developed language ASCII is an open protocol, used for establishing a session in a simple and independent IP network therefore it becomes scalable and it has the ability to couple to different architectures.

SIP methods and responses

Methods

- **REGISTER:** Used by a SIP UA to register your address and IP address on the registration server.
- **INVITE:** used to establish a SIP session between two user agents, contains information on who generates

the call, the recipient and the type of flow to be exchanged.

- **ACK:** used to accept an equally session and confirm that you can start exchanging messages reliably.
- **OPTION:** used to meet capacity characteristics and status of a UA or a server, which can initiate a session between the two.
- **SUBSCRIBE:** used for requesting an update on the status of another UA information, the purpose of all this is to know if a user is online, busy, offline, etc.
- **CANCEL:** Used to order the abandonment of the call it is in progress, of a pending application without determining the session.
- **BYE:** used to end an active session can be generated by the user who initiated the call or who is being called, the BYE command is the only one who can end a session entirely.

Answers

- **Informational (1xx)**: the application has been received and is being processed.
- **Success (2xx)** before the application received is recognized and accepted.
- **Redirection (3xx)** the application cannot be completed and no need for additional steps.
- **Client error (4xx)** account with which you want to log has errors, so the server cannot continue with the application.
- **Server Error (5xx)** the application is received but the server cannot process it, because it issues the server itself.
- **Global Failure (6xx)** the application is received but the server cannot process this type of errors can occur on any server, so that applications will not be forwarded to another server for processing.

E. UDP protocol

User Datagram Protocol (UDP) is a connectionless protocol that does not provide error detection and creation of ACK corresponding to the transport layer of the TCP / IP model.

F. RTP protocol

Transport protocol that works over UDP because it offers features needed in order to gain speed, although they have to sacrifice the reliability of the data, does not guarantee delivery of packets to their final destination or delivery at the right time for them. Use pairs assigned ports from 1025 to 65535 but the best known is the port 5004 (Gil Cabezas, 2009).

G. VoIP encapsulation

While two users want to establish a call, an internal frame encapsulation process and then are transmitted through the IP protocol is performed, this process is performed after a call has been successfully established. The analog signals once converted to digital signals are encapsulated by RTP and subsequently by the UDP protocol.

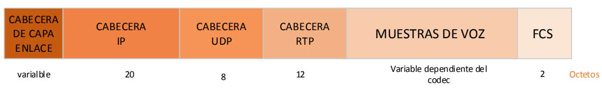

Figure 3. Encapsulation frame of VoIP
Reference: Telefonia VoIP Encapsulation of a frame VoIP. Recovered: http://goo.gl/A5SSDs

H. Call traffic analysis

The main feature that a VoIP network is quality should pay calls, for which the calculation of core and the bandwidth required for the necessary transport of the data is performed so that there is no type of inconvenience to establish a call.

Outgoing traffic analysis

Traffic flow can be calculated with the following equation:

$$T = \frac{C}{A}$$

T: Traffic flow
C: Number of occupations
A: Call duration time
Equation 1. Traffic flow calculation.
Reference: (Culqui, 2013)

In the week of June 6 2- it is disclosed that the day with more flow of calls to the city of Ibarra is Tuesday 3 Likewise the same day for the branch of Quito.

Table 2. Outgoing voice traffic for days in Ibarra

Date	6/2/2014	6/3/2014	6/4/2014	6/5/2014	6/6/2014
Ocupa.	21	18	32	25	25
Duration	1883	1967	3065	2009	2857
T(s/ocupa.)	89.67	109.28	95.78	80.36	114.28

Reference: Sheets CNT EP.

Table 3. Outgoing voice traffic for days in Quito

Date	6/2/2014	6/3/2014	6/4/2014	6/5/2014	6/6/2014
Ocupa.	116	119	77	94	106
Duration	7550	13356	5781	9346	9534
T(s/ocupa.)	65.09	112.24	75.08	99.43	89.94

Reference: Sheets CNT EP.

Calls rush hour

You need to know the time during working hours have increased traffic and well dimensioned network.

$$C = \frac{Number\ of\ occupations}{3600\ s}$$

Equation 2. Calculation of originating calls.
Reference: (Culqui, 2013)

Based on tabulations and calculations in the city of Ibarra must rush hour between the hours of nine o'clock in the morning and in the city of Quito in the time ten o'clock.

Table 4. Rush hour the first week of June in Ibarra

Date	Schedule	Activities	T(s/ocup)	A(Erl)
6/2/2014	12:00:00	2	89.67	0.05
6/3/2014	18:00:00	4	109.28	0.12
6/4/2014	9:00:00	12	95.78	0.32
6/5/2014	8:00:00	6	80.36	0.13
6/6/2014	16:00:00	5	114.28	0.16

Reference: Manual tabulation of returns, based on CNT EP.

Table 5. Rush hour the first week of June in Quito.

Date	Schedule	Activities	T(s/ocup)	A(Erl)
6/2/2014	16:00:00	20	65.09	0.36
6/3/2014	12:00:00	30	112.24	0.94
6/4/2014	9:00:00	17	75.08	0.35
6/5/2014	15:00:00	14	99.43	0.39
6/6/2014	10:00:00	25	89.94	0.62

Reference: Manual tabulation of returns, based on CNT EP

Analysis of incoming traffic

For inbound traffic analysis method of tabulating surveys employees used FLORALP S.A. matrix of Ibarra and Quito offices. According to the data obtained is taken as an average 3-minute call duration.

The calculation of Erlang per day will be carried out by the following equation:

$$A = \frac{number\ of\ calls * duration\ (s)}{3600s}\ (Erlangs)$$

Equation 3. Flow calculation incoming traffic in Erlang.
Reference: (Culqui, 2013)

As a result the following summary table gives:

Table 6. Result of Erlang for a week.

	Ibarra	Quito
Day	Erlang	Erlang
Monday	0.40	0.25
Tuesday	0.30	0.20
Wednesday	0.30	0.20
Thursday	0.40	0.10
Friday	0.35	0.10

Reference: Google docs survey. Based on Excel spreadsheet.

Rush hour incoming calls

According to the survey, incoming calls are busiest on Mondays in the morning hours between 8:30 am and 11:00, less frequently they occur daily from Wednesday to Friday in the same way in the schedule morning and evening comprised between 14:30 and 16:00.

I. *Study of free software platform for ToIP*

The IEEE 830 version 1998 standard and together with its update ISO / IEC / IEEE 29148 of 2011 for Software Requirements Specification allows choosing software for IP-PBX telephone exchange, by evaluating a number of parameters and specific requirements.

REQ01: Administration
REQ02: Interfaces
REQ03: Compatibility
REQ04: RAM
REQ05: Protocol Support
REQ06: Support audio codecs
REQ07: Call Control
REQ08: simultaneous calls
REQ09: Administration and Reporting
REQ10: Report calls
REQ11: Database
REQ12 Version
REQ13: Architecture
REQ14: License
REQ15: Number of users
REQ16: Performance
REQ17: Interoperability
REQ18: Scalability
REQ19: Security

Elastix to presenting new features in areas such as: do not depend on other software to present the user with a graphical interface easy to handle and allows remote administration, have the ability to support various audio codecs to avoid restricting the operation under certain codecs, the number of simultaneous calls is of great importance because the industry is in constant communication and must support the largest number of online conversations without these lose their quality, detailed report calls for usage of the network and improve management, the number of users that can be registered in the database is clearly important as an industry is handled with trends to grow in the course of the coming years; Elastix qualify for these features to be the best option to use.

III. SYSTEM DESIGN OF IP TELEPHONY

A. *Management requirements*

Based on requests by the system administrator area the following features to the IP telephony system requires:

- It must provide the VoIP service to users within each of the branches, so that an internal communication network have.
- Call priorities should be given according to the position of the person who will use it.
- Keep a telephone line for use in case of system failure IP telephony.

- The alternative use of IP phones and softphones for network users.
- To provide high reliable, stable and scalable telephony for future network growth.
- Maintain the privacy of calls made between different branches.

B. Selecting the audio codec

To select the codec calculations are performed with the G.723 codec for voice transmission, which has a payload of voice data 24bytes with 20ms payload in transmission.

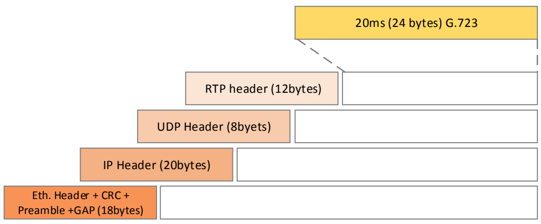

Figure 4. Payload of codec G.723
Reference: (Vasco, 2010). Recovered: http://goo.gl/0PzuPz

Below is a summary table with the data obtained by performing the same calculation for the other audio codecs shown.

Table 7. Calculation results.

Codec	Payload (ms)	Frame for seconds	Bandwidth (Kbits/s)	Simultaneous calls	Bits/frame
G.723	24	28	18.11	1389.74	656
G.711	160	50	87.20	288.60	1744
GSM	32.5	50	36.20	695.19	724
G.729	20	50	31.20	806.60	624

Reference: Excel spreadsheet. Based on calculations obtained.

The GSM audio codec allows the creation of .wav audio files that tablets occupy 16,000 bytes for each 10 seconds of audio. This is a type of codec operates at 13kbits / s. The voice signal is divided into 20ms, each block contains 260 bits.

C. Calculating the bandwidth for the system

When sizing a telephone system should take into account that should occupy a considerable percentage of the total network in a way that allows the transmission of voice and other services that cross the network without any problems.

- 20% capacity corresponds to the voice for a day's work for eight hours.
- 30% for time to have the worst traffic.
- 50% for the worst fifteen minutes a day.

$$Codec\ bandwidth\ of\ GSM = 36.20\ kbps$$

$$Number\ of\ simultaneous\ calls = 12\ calls\ (Ibarra)$$

$$Number\ of\ calls\ with\ growth\ forecast = 16\ calls$$

$$System\ bandwidth = 16\ calls * 36.20Kbps$$

Equation 4: Bandwidth for the system
Recovered: (Vasco, 2010)

$$System\ bandwidth = 579.2Kbps$$

D. Calculation trunk

The calculation of the necessary VoIP traffic based on the table of Erlang B traffic model trunk, as there are complex mathematical calculations available such charts. It is set to a value of GOS, which means the probability of dropped calls, 1% corresponding to the ratio if regenerate 100 calls one of them is lost, just as 100 incoming one call is lost . (Garduño, 2007)

➢ *Trunks for incoming calls*

Taking into account that 20% growth in number of employees, the value of Erlang for the largest number of simultaneous incoming calls is provided initially is 0.4

Erlang for the city of Ibarra and 0.25 Erlang for offices obtained Quito previous sections

Table 8. Number of trunks for incoming traffic

City	Erlang + 20% growth	Number of trunks for incoming
Ibarra	0.48	3
Quito	0.30	3

Reference: Table Erlang, Based on traffic table Erlang B model

- *Trunk for outgoing calls*

Similarly 20% growth in number of employees, the value of Erlang for the largest number of simultaneous outgoing calls is initially 0.32 Erlang for the city of Ibarra and 0.94 Erlang to Quito offices, data from remains Tables 24 and 25 respectively.

Table 9. Number of trunks to outbound traffic

City	Erlang + 20% growth	Number of trunks for incoming
Ibarra	0.384	4
Quito	1.128	5

Reference: Table Erlang, Based on traffic table Erlang B model

E. System Requirements

General requirements

- The phone platform resources must use the latest technologies in terms of service calls is concerned, with the aim of providing a quality service.
- The signaling protocol for communication should be SIP.
- The central IP telephony should provide a hybrid system that handles both analog extensions and SIP extensions.
- Having the ability to work with different platforms and services provide additional future.
- The system must have a growth perspective for the next five years.
- You must additionally have an internal messaging system, typical of FLORALP S.A

Specific requirements

Technical

- Capacity for more than 200 users.
- Able to support 4 analog trunks.
- Support GSM audio codec.
- Support SIP trunks.
- Support for SIP extensions.
- VPN support.

Administrative

- Management system through a completely friendly graphical interface for the administrator.
- SIP extensions configuration an easy way by means of GUI.
- Remote management, can be accessed from anywhere within the LAN and WAN configuration and programming of the telephone exchange network.
- Report consuming phone calls.

Security

- VPN support for encryption of calls over the public network.
- System Access by SSH and Telnet.
- Only the administrator must know the password of the IP PBX.

F. Services to be provided in the network of FLORALP S.A.

Elastix provides a number of services previously listed in the description of the software, and based on the inventory of multimedia equipment owned network

FLORALP S.A. aims to provide the following services:

- **Phone calls:** in including internal calls in offices and calls between different branches.
- **Voicemail:** the service will be integrated to facilitate users to receive, send and listen to voice messages or to receive e-mail account.
- **Call detail:** whereby it may release a report of incoming and outgoing calls, duration of calls, including caller ID.
- **Call waiting:** in order to indicate to a user by means of a short tone that there is an incoming call and you have the opportunity to answer it.

G. *Design dial plan extensions FLORALP S.A.*

To integrate services and improve communication of branches, it is to assign a specific range of extensions for each of the offices for internal calls, then the picture of extensions assigned range shown.

Table 10. Range extensions for FLORALP S.A.

First digit		Second digit		Third digit
Depends of branch		Depends of physical location		Depends on the number of staff
Branch	**Digit**	**Location**	**Digit**	**0-9**
Ibarra	1	Ground floor	0 - 1	0-9
		Top floor	2-3	0-9
		Production	4	0-9
		Homes	5	0-9
Quito	2	Ground floor	0 - 1	0-9
		Top floor	2 - 3	0-9
		Cellar	4	0-9
Guayaquil	3	Administration	0 -1	0-9
		Cellar	2	0-9
Cuenca	4	Administration	0	0-9
		Maintenance	1	0-9
San Gabriel	5	Production	0	0-9
Available	6	New extensions	0-9	0-9

Reference: Department of systems based on distribution of extensions.

H. *Hardware and software for ToIP*

- Based on the characteristics of the current network data has the industry FLORALP S.A. you must acquire the following equipment for possible implementation: An Appliance server on which platform Elastix is mounted in its version 2.5, it must possess features for connecting four analog trunks, FXO and FXS ports, Ethernet interfaces for connection to the network, console port configuration between the main requirement. IP phones the same will be made available to managers and heads of each area; telephone headsets for attendees and mobile users.

- After reviewing the performance and features of different models and manufacturers XORCOM CXR2000 appliance meets the requirements of the current telephone.

- As regards IP phones comparison was made two marks, and Grand stream Yealink, the same equipment that have similar characteristics. The Yealink brand for all ranges, high, medium and low was chosen, because they have best references in the market and are fully compatible with Elastix, using SIP technology and have features of Open VPN, QoS and VLANs.

I. *Configuring VPNs*

A virtual private network is based on the concept of tunneling, this involves establishing and maintaining a logical connection between two extremes where intermediate nodes are involved. The main protocol VPN should provide authentication, data compression, data encryption, dynamic routing and key management.

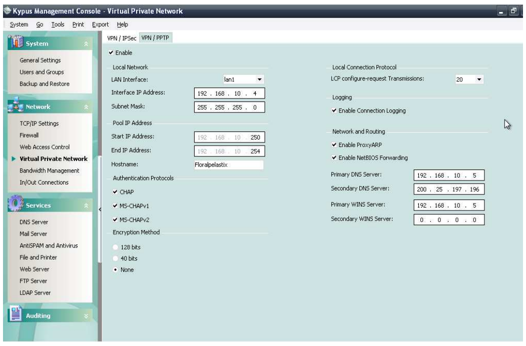

Figure 5. VPN configuration KYPUS
Reference: KYPUS captured image server

Quality of service in the VPN.

The method of allocating a certain bandwidth from the computer on which the VPN is configured, then the bandwidth allocation is shown for implementing VPN in Kypus Proxi.

Figure 6. Allocation of bandwidth for VPN
Reference: KYPUS captured image server

J. Quality of service oriented voice

Process for implementing quality of service

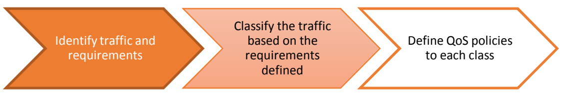

Figure 7. QoS implementation process
Reference: Quality service for TCP / IP networks JFCA jfcalt@gmail.com

Identification of traffic and requirements

In the table below the type of traffic that crosses network data FLORALP S.A. and requirements thereof, according to the criteria of the systems department of the same detailed.

Table 11. Most used protocols in data network.

PROTOCOL	PORT	Percent Usage
smtp	25	7%
http	80	9%
pop3	110	7%
nntp	119	6%
epmap	135	3%
netbios-ssn	139	4%
imap	143	7%
https	443	4%
microsoft-ds	445	4%
urd	465	6%
nntps	563	6%
submission	587	6%
imaps	993	6%
pop3s	995	6%
vnc-http	5800	4%
vnc-server	5900	4%
Otros Protocolos		10%

Reference: test port with Nmap / Zenmap and programs Axence NetTools

Traffic Classification

Once determined applications we proceed to classify traffic, then briefly describes the types of priorities by which the classification is done.

- **Priority criticism:** Considered to applications working in real time, and needs a considerable bandwidth for proper operation and there is no packet loss and jitter. The main applications are critical priority Voice transmission using IP and videoconferencing.
- **High priority:** applications that do not require a high bandwidth, but are used daily by end users, and are important in the performance of the industry. Within this priority are WEB applications and database management.
- **Medium priority:** Applications are available for users, which help to identify them, they can work smoothly despite the delays that may occur in the

network. Under this priority are applications like access to the Web, DNS and DHCP.

- **Low priority:** priority latter includes applications that are useful but note great importance to users, are more resistant to the delay. Applications that they can be categorized as low are: email, downloads, and more.

Defining policies

The policies below are those that are named for a robust network must be met to provide good service quality of voice. In this project will detail what the policies were implemented.

- **Diffserv (Differentiated Service)** flexible architecture, in which certain traffic is treated better than the rest, includes features of speed in traffic, higher average bandwidth availability and lower losses. It is based on the information contained in each packet to meet QoS requirements.
- **QoS MQC (Modular QoS CLI)** is a method of implementation, the classification and QoS policies are performed separately.
- **ACLs (Access Lists)** setting ACLs in order to separate the traffic generated by IP or ports in order to classify traffic that crosses the network.
- **CoS 802.1Q / p and DSCP** (Differentiated Service Code Point): increased flexibility and traffic class, is used to make datagrams and differentiate the service received in the routers. Compatible with IPv4 and IPv6.
- **WRR (Weighted Round Robin)** specified for Queueing and very similar to PQ (Priority Queuig) but with the difference that the service alternating between classes, allows the service never is inactive when there queued packets to be transmitted.
- **CBWFQ (Class Based Weighted Fair Queuing)** queuing mechanism, responsible for ensuring the bandwidth according to priority, working with low latency and reserves the minimum bandwidth and class.
- **CBWRED (Class - Based Weighted Radom Early Detection):** used to prevent congestion, can be used as choose the profile of each class of traffic based on DSCP and IP Precedent.

IV. PERFORMANCE TESTING

The transmission of voice over IP protocol requires control parameters as they are essential for measuring IP telephony service that is delivered to end users. This chapter is intended to verify the correct functioning telephone system, then different tests that were carried out within the network mentioned FLORALP S.A.

Technical tests

Within technical testing two types of tests are performed, the first shall be made without including quality of service in any of the cases and the latter are including QoS parameters set out in Section 4.6 of the fourth.

- Bandwidth used by two simultaneous calls.
- Packet loss, delay and jitter.

Connectivity tests.

Connectivity tests aim to check the correct operation of VPNs configured on the server KYPUS.

- Users Ibarra-server connectivity.
- Users Quito-server connectivity.

V. CONCLUSIONS

Titling this paper is to allow the dairy industry FLORALP S.A, grow through the use of new communication technologies, in addition to this access to your data network to become robust and flexible at the same time; without forgetting that this technology reduces costs by consumption telephony.

The PBX Panasonic Advanced Hybrid System KXTA616 has served its useful life, which causes inconvenience to users; the data network on the other side is flat, no traffic classification, without ACLs and VLANs, the network FLORALP S.A. Ibarra works independently of its other offices.

Free software platforms currently allows to solve applications and services on the same level as paid software can do it, taking into account the benefits of being free is the savings in licensing costs, and that most companies not only public but also private, have such solutions, it is a clear example of the central IP PBX with high demand for its cost and good service.

With the help of IEEE 830 (1998) and ISO / IEC / IEEE 29148 standard (2011), Software Requirements Specification, selection of IP telephony software that fits the needs was made and provide the best solutions users of the network. As a result was obtained that the Elastix platform provides stability, efficiency both in memory and in use of the network, it is easily integrated into any infrastructure in which it is located.

The equipment sizing is based on the choice of PBX software, once defined which will be used, the type of server is dimensioned taking into account that must support a number of users and must have physical characteristics to connect to the network public telephony.

We must remember that there are currently a number of solutions for the transmission of voice over IP, the market can find everything from phone cards FXO / FXS Gateways, appliance dedicated exclusively to work with free software platforms, and even switch to allow reuse almost obsolete Cat5e cabling without worrying about the quality of the voice.

One of the main disadvantages of IP telephony voice quality, compared to an analog line q has no such problems; but these characteristics can be improved by using QoS policies on the data network, allowing the satisfaction of end users.

Thanks to cost-benefit analysis in which calculations of NPV, IRR are included, you can determine whether a project is viable for implementation, provided that the value obtained in the ratio is greater than one; this project despite having a long recovery period is profitable.

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VI. RECOMMENDATIONS.

The design of the phone system some shortcomings in the network of FLORALP S.A. therefore recommended upgrade equipment routing and switching, to allow advanced configurations such as QoS, ACLs, VLANs, etc. is recorded.

Structured cabling has fulfilled its useful life of five years and therefore it is recommended that a certification of ports, to allow the transmission of voice did not have any problems.

It is very important not to forget to perform the service quality settings according to the model presented in the design, and that should provide a call smoothly as echo and delay. We recommend using the echo canceller module that has analog Elastix additional cards or modules.

Elastix now has three stable versions, 2.4, 2.5 and 3.0, blueprint or design for version 2.4 and 2.5 are the most appropriate, they do not require licensing and are available in the www.elastix.com page, version 3.0 includes new management parameters but at the moment it does not have the necessary support and modules are not yet included them as Addons, batch configuration extensions, Easy VPN among others..

The most important data on a network is security, so it is necessary to configure the Firewall module to Elastix brings embedded, disable web access while not the administrator who will use it.

In creating users and in regards to passwords Strong passwords should be used, that is, they must contain numbers, uppercase and lowercase letters and special characters; likewise access keys must be different from those placed Elastix the default; change the known remote access ports, use VPNs for remote administration and even to users outside the network, all this will make the server vulnerable to any attack..

RECOGNITION

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