Design IP telephony through Cloud Computing under a platform of free software GNU / LINUX for Universidad Técnica del Norte

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ABSTRACT

The present study implicates a new solution of telecommunication for IP telephony service which is based on the Cloud Computing, which allows to get an option for communications for the “Universidad Técnica del Norte”. The analysis was performed, based on the IEEE 29148 standard for selecting virtualization platform and IP telephony software and then the IP telephony was designed through cloud computing. In the IP telephony design, the dimensioning of IP telephony was developed through the cloud, it considered the capacity instance, bandwidth, traffic flow and the number of trunks, for the analysis of these parameters the information updated of the University about telephony IP was obtained, then it was proceed to configure in the Cloud platform. Then the service performance tests were made to observe the behavior that this has through this infrastructure.

Keywords: VoIP, PBX, Softphone, Public Swithed Telephone Network, Real-time Transport Protocol, Session Initiation Protocol, Inter-Asterisk eXchange.

1. INTRODUCTION

Nowadays there is a large growth of services and equipment options. In today's competitive environment, companies are looking for ways to reduce expenses. Future technologies that are developing are focusing the cloud, as offering various services such as infrastructure, platform and software in order to reduce costs and execution constantly short term.

IP telephony through the cloud has objectives be a full service which allows corporate communications, who has the support of the applications (voice mail, instant messaging, video conferencing, web applications for PC).

This service in developed countries and is being implemented as an optional solution for cloud-based telephony. According to the company Polycom, dedicated to providing unified communications services. It is expected to generate a return of $ 82 million until 2017. In Ecuador, the company PaloSanto Solutions is now offering the service, with a number of features which was launched in January 2016 with value monthly subscription It lets get service in minimum time.

At the University has the service enabled IP telephony server Elastix ELX5000 but there are some agencies that have not yet enabled IP telephony service, so this could be an option to improve efficiency and reduce costs of implementation. The prospect of this design is to provide a Virtual University for servers providing IP telephony through technological tools that are offered in the cloud which are also under free licensing.

2. MATERIALS AND METHODS

2.1 OPENNEBULA

OpenNebula is a platform belonging to industry focused on open source virtualization data center, which offers a lot of features to build and manage enterprise clouds and virtualized data centers (OpenNebula, 2014). You can see the most important features:

- Security management as RSA key pairs, ssh, and LDAP X509 certificates.
- Advanced control and monitoring of Virtual Infrastructure.
- Resource management as memory size, disks, type of network interface card (NIC), volume management.
- Analyzes traffic coursing in groups connected instances.
- Supports operating systems such as Microsoft Windows and Linux.
- Manage host as create, delete, activate, deactivate.
- Support hypervisor Xen, KVM and VMware-Quemu.
- Network management provides NAT, DHCP, DNS.
- Data storage lvm, vmdk.
- Manages user resources, virtual machine images, templates VM, VM instances, virtual networks, zones, physical hosts, authentication, authorization.

2.1.1 Architecture OpenNebula

The physical architecture that assumes OpenNebula is as classic cluster, virtual machines (VM) are generated in single-server and has at least one physical network so that all these hosts to communicate. It can be seen in Figure 1 OpenNebula architecture.

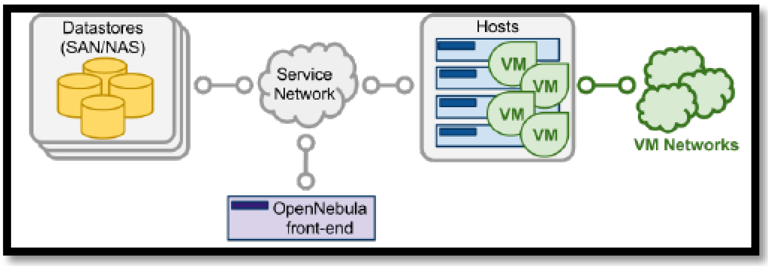

Figure 1. OpenNebula architecture.

The basic components of a system according to the author OpenNebula (Toraldo, 2012) are described below:

✓ **Front-end**

It is the one that is responsible for implementing the OpenNebula services. The machines running on OpenNebula are called frontend. It includes the following components: Daemon management planner virtual machines, server web interface (Sunstone).

✓ **Host**

These are the nodes running hypervisors enabled infrastructure. These are the nodes running hypervisors enabled infrastructure. This are physically virtual machines running on OpenNebula.

✓ **Repositories of images and storage**

This is the base that stores images of virtual machines. OpenNebula has its own repositories to manage image files VMs.

✓ **Network Service**

OpenNebula offers easy adaptability and customization for the network subsystem, to improve the integration of the internal private network platform to the corporate network and also has an internet connection.

2.2 ELASTIX

Elastix is an open source software that has been established for unified communications. The objective is to provide alternatives have established communication for enterprise level as a single solution. According to the author (Landivar, Elastix Unified Communications, 2008), indicates the most relevant features of the VoIP service below:

- ✓ Tool to create lots of extensions which facilitates new facilities.
- ✓ IVR configurable and flexible enough.
- ✓ Call Detail Report (CDRs).
- ✓ Support for call queues.
- ✓ Supports SIP, IAX among others.
- ✓ Codecs supported: G.711 (A-Law & μ-Law), G.726, G.729, GSM.
- ✓ Trunking
- ✓ Incoming and outgoing routes
- ✓ Interactive access from the Web to the Asterisk console.

2.3 System Requirements

System sizing the bandwidth requirements, traffic supports, number of users, creating core and ensure that there is communication between the Engineering Faculty of Applied Science and the Old San Vicente de Paul Hospital.

Setting up a trunk between a server on the cloud one housed in a physical virtual machine, in order to do a test run to see if it is possible to make a call to the PSTN.

2.3.1 Requirements Management

You must provide the IP telephony service in the communication network of the University for testing operation between FICA and the Old Hospital San Vicente de Paul. In addition, the softphone software for computers and smartphones Zoiper for users on the network of the University will be used, because this is the most important characteristics in terms of protocol support IP telephony.

2.4 Dimensioning of IP telephony system through Cloud Computing

In the sizing of the system it will be taken into account certain parameters to be used for system design, is the lists below:

2.4.1 Capacity instance virtualization platform.

The server capacity is for testing operation of the service, but if implemented, the minimum requirements that should be the server to support a total of 1000 extensions and 500 concurrent calls, indicated this analysis it performed on the basis of the characteristics of the current server of the University due to the ELX5000 Quemu-KVM hypervisor can get similar to a real physical performance, in table 1 are the following features:

Table 1. Instance specifications

Funcionality test	Recommendation
4GB RAM	8GB RAM
10GB hard drive	500GB Hard Drive
2 processor core (2.4 Ghz)	Quad core 2.4 Ghz processor (4 cores).

2.2 Calculation of bandwidth for VoIP

To perform calculations bandwidth GSM codec was chosen because it is showing to have the best features for this design, it has a very low compression of 13.2 Kbit / s, the voice signal is divided into blocks of 20 ms, offers good performance regarding the use of the CPU and also enabled on Elastix and requires no license (Landivar, Elastix Unified Communications, 2008).

GSM codec data:

- Baud rate = 13.2 Kbps
- Size payload = 33 bytes
- Payload = 20 ms
- Packets per second (PPS) = 50 PPS

Table 2 shows the calculations is performed for bandwidth with this codec:

Table 2. Calculations bandwidth

Description	Calculations
Calculating the size of the frame of voice	$Frame\ size = Payload + Transport\ layer\ header + network\ layer\ header + data\ link\ layer\ header$ $Frame\ size = 18\text{ bytes} + 20\text{ bytes} + 8\text{bytes} + 12\text{bytes} + 33\text{ bytes}$ $Frame\ size = 91\ bytes$ $Frame\ size\ in\ bits = 91\text{ bytes} * 8 = 728\text{ bits}$
Calculation of the number of packets transmitted per second (PPS).	$PPS(Number\ of\ packagestx/sec) = (transmission\ rate\ codec)/(payload)$ $\text{PPS} = (13.2\text{ kbps})/(33\text{ bytes} * 8\text{ bits}) = 50\text{pps}$
Calculations of the bandwidth	$Bandwidth = (total\ package\ size) * (PPS)$ $\text{Bandwidth} = 728\text{ bits} * 50\text{ pps} = 36.4\text{ kbps}$

Reference: (Salcedo, López, & Hernández, 2011) (CISCO, 2008).

2.2.1 Call traffic statistics

To observe the traffic generated at the Universidad Técnica del Norte was obtained from the report calls Elastix ELX5000. The date from which the data will be analyzed is the February 17, 2016, because at this time more concurrent calls are generated as it was to complete the semester, so that an analysis of the data was made for the calculation of network traffic. The result was at rush hour from 11h00 to 12h00:

- Call Number: 283
- average call time: 68.09 sec

$$\text{Simultaneous calls Bandwidth} = \text{AB} * \text{n (Number of full} - \text{duplex conversations)}{*}2$$

$$\text{Simultaneous calls Bandwidth} = 36.4\ \text{Kbps} * 283 * 2 = 20.60\ \text{Mbps}$$

2.3 Traffic flow

For these calculations the number of simultaneous calls is required, which these were obtained from the report calls Elastix ELX5000 that is 283 calls and the average duration time of 68.09 seconds is called. To calculate this traffic intensity (A) it can be done by the traffic flow equation (Marcano, S. F.):

$$\text{A} = \text{Number of calls for one hour} * \text{call duration (sec)}$$

$$A = \frac{283\ \text{calls}}{3600\text{sec}} * 68.09\ \text{(sec)} = 5.35\ Erlangs$$

2.4 Calculation trunk

These values are done with the traffic flow was obtained from 5.35 Erlangs, the value of acceptable service should be less than 1% lock in rush hour and model of Erlang B. Figure 2 shows an excerpt Erlang B model, giving a result of 12 trunks required for the traffic flow.

No. of Trunks (N)	Traffic (A) in erlangs for P = 0.1%	0.2%	0.5%	1%
11	3.65	4.02	4.61	[illegible]
$\boxed{12}$	~~[illegible]~~	~~[illegible]~~	~~[illegible]~~	5.88
13	4.83	5.27	5.96	6.61
14	5.45	5.92	6.66	7.35
15	6.08	6.58	7.38	8.11

Figure 2. Number of trunk, Erlang B model

2.5 Architecture of IP telephony in the Cloud

The network architecture is proposed for IP telephony through Cloud Computing this is made up of two virtual stations that communicate through a backbone between the Faculty of Engineering of Applied Science (PBX-FICA) and the Old Hospital San Vincent de Paul (PBX-AHSVP). Users must connect to the computer software or mobile Zoiper to the wired or wireless network FICA and authenticate with the extension to be assigned and the IP address of the PBX server-FICA. Similarly, in the Old Hospital wireless network must authenticate users with the extension and IP PBX-AHSVP server, to make calls.

Also hosted virtual panel will communicate in the cloud of the FICA (PBX-FICA) through a trunk to be the server of the University for verification tests with this is to find a way to communicate with the PSTN, and that there are no compatible interfaces. Users FICA be connected with computer software or mobile Zoiper to the wired or wireless network faculty and authenticate with the assigned extension and IP address of the PBX-FICA, server and then make a call with users ELASTIX ELX500 the server that corresponds to the administrative staff of the University. In figure 3 it is observe detail.

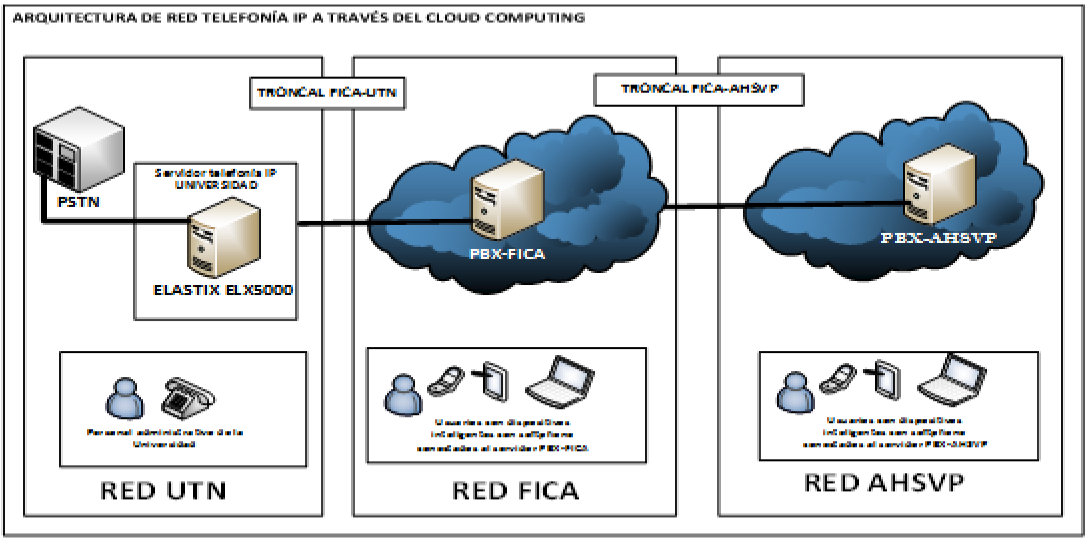

Figure 3. IP telephony architecture in the cloud

2.6 Service network diagram

This network diagram is an adaptation of service currently being held in the University in Figure 4 shows all these teams.

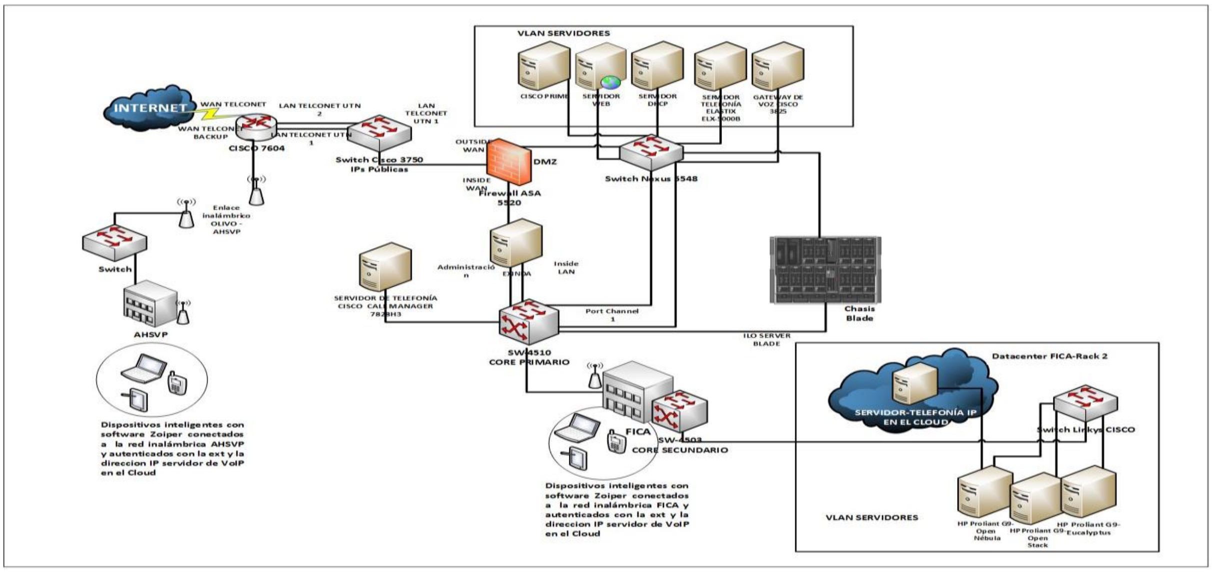

Figure 4. Adaptation of service network diagram Informatics Department

2.7 Elastix configuration on OpenNebula

2.7.1 Uploaded Elastix ISO image to OpenNebula

ISO software image Elastix 2.5 is created on the platform as it is the most stable and compatible with OpenNebula version, name, type and location of ISO are configured. In Figure 5 ISO Elastix 2.5 loaded on the platform OpenNebula is observed.

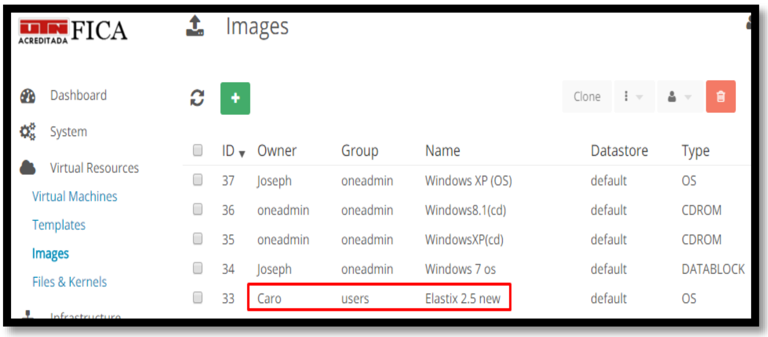

Figure 5. Load ISO image Elastix 2.5 on OpenNebula

2.7.2 Elastix instance deployment on OpenNebula

For instance creation it is done based on the characteristics that were specified in the capacity of the instance on the virtualization platform, which indicates the size of the hard disk to be assigned is 10GB and the memory RAM is 4 GB which is to conduct performance tests. First a template used to create or load the console in a more efficient way in which the CPU resource, type disk space for installation, RAM, network storage among the most important are specified is configured. Subsequently the virtual machine, which is the instance configuration previously created template is created. In Figure 6 the created instance is observed.

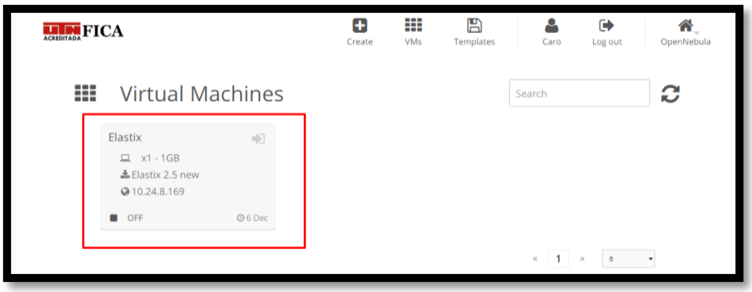

Figure 6. Deployment instance Elastix 2.5 on OpenNebula

2.7.3 Installing Elastix on OpenNebula

To install the software IP telephony, language, keyboard layout, disk partitions, network interface, Gateway, DNS, hostname, passwords root among the most important are configured. After the installation passwords root user access is configured and the web administration the installation disk image is removed, as this installation process is somewhat similar to one carried out on a physical server.

Then you access the root administrator through the server console interface and Web server administration as shown in Figure 7.

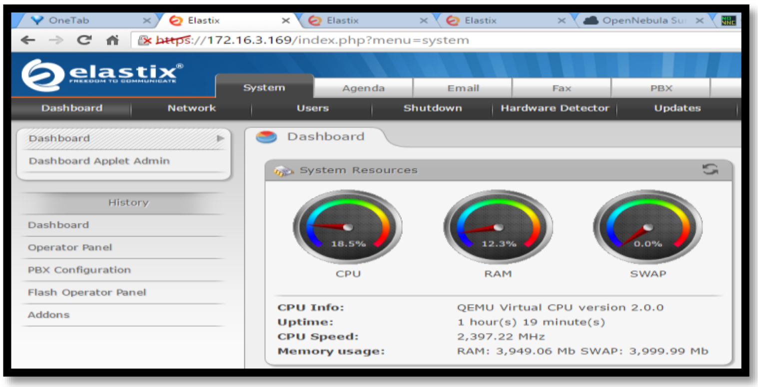

Figure ***7.*** Elastix Web management interface Elastix server

2.7.4 Configuring IP telephony

The network architecture is presented in this design consists of three servers, two virtual server and the University, each server extensions hosted is set to OpenNebula. After the trunk is set to communicate between servers, then perform the respective function tests IP telephony service. The settings that were made were:

- Extensions to SIP and IAX2 protocol to verify the support platform Cloud.
- Troncales to verify communication between servers located in the cloud and also between virtual and physical one.
- Exit routes to communicate between servers through a trunk route and where you want to go.
- IVR to verify support services and this is in the event that a user does not know the extent to which requires calling and interactive voice recognition generates a digital reception and if necessary get help from an operator to find the destination extension.
- Colas and conferences added it as an option of switching the IVR to verify the functionality of this service. A conference was set for a call between multiple users.
- The path will be used in the operation of IVR between trunk to allow the request to enter the trunk.

3. RESULTS

3.1 Technical tests

The data of these tests indicate different behaviors, to collect the results was done with Wireshark tools, Exinda University server and ZoIPer software, in the case of locally samples made are taken in FICA where shows that call quality is poor, has the problem of packet loss and jitter, but are acceptable values as that described in levels permitted and not exceeding these values, can be seen in figure 8 these results that also a compilation in table 3 is performed.

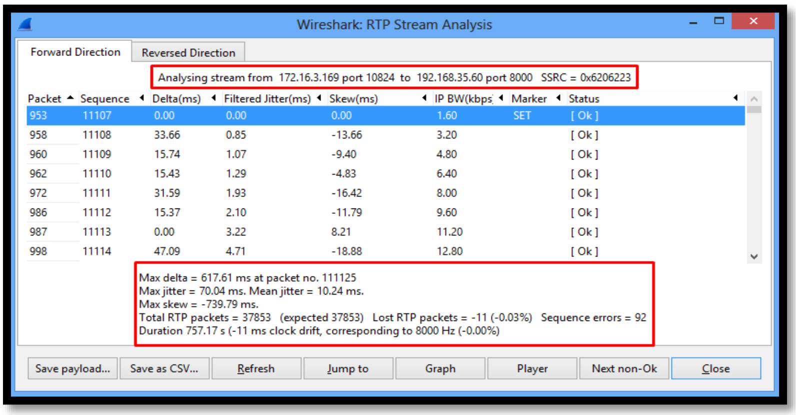

Figure 8. RTP traffic capture with Wireshark local level test

In the case of tests between servers it took place between FICA and AHSVP, another result in which one can conclude that the call is poor is observed, it is not at acceptable levels, has big problems Jitter and packet loss. Figure 9 shows the result of Wireshark observed. Compiling the test is between servers in Table 4, where it is indicated that communication with different communication protocols established.

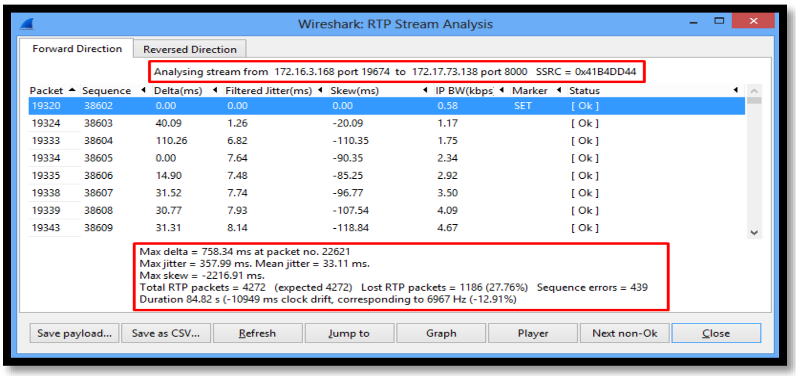

Figure 9. Analysis of an RTP packet FICA captured between PBX and PBX-server-AHSVP

For operation it requires the bandwidth that is mentioned in the sizing is assigned and also implement policies to improve service quality service delivery to reduce the effects of delay, packet loss and jitter. As server performance was verified that the work done is quite efficient since there was saturation after consumption of server processor shown in Table 4, but this server is only for verification testing service behavior if it will implement you must be configured with the requirements specified in the sizing.

Table 3. Technical testing results locally

Parameters	Result
Network	FICA
Jitter	70.04 ms
Packet loss	11 packets -0.03%
Sequence Error	92
Processor	18.50%
RAM	12.30%
Entry Exinda Traffic Report	244.285 Kbps
Traffic report output Exinda	244.285 Kbps
Processor OpenNebula	19%
RAM OpenNebula	2.9 GB

Table 4. Technical test results between servers

Tests	Network	Jitter	Packet loss	Sequence Error	Observations
Tests between PBX and PBX-FICA-RSVP protocol SIP servers	AHSVP	357.99 ms	1186 packets - 27.67%	439	Normal traffic flow with the SIP protocol
Tests between PBX and PBX-FICA-AHSVP MV SIP protocol servers	DDTI	11.96 ms	Loss 0 packets - 0%	0	Normal traffic flow with the SIP protocol
Tests between PBX and PBX-FICA-RSVP protocol IAX servers	DDTI	-	-	-	Normal traffic flow with IAX2 protocol

3.2 Connectivity Tests

These tests allowed verifying the successful operation of the service using an open source solution called OpenNebula with the support of various protocols such as SIP and IAX2, the results in Table 5 summarizes observed.

Table 5. Results connectivity tests

Tests	Functioning
Tests locally	✓
Testing connectivity between PBX and PBX-FICA-RSVP protocol SIP servers	✓
Testing connectivity between PBX and PBX-FICA-AHSVP MV SIP protocol servers	✓
Testing connectivity between PBX and PBX-FICA-AHSVP protocol IAX servers	✓
Testing connectivity between servers and PBX-FICA University Elastix server	✓

Reference: Own elaboration

Figure 10 shows that it has managed to make a call through the server in the cloud successfully with the Zoiper software between two softphone installed on a computer and a mobile device.

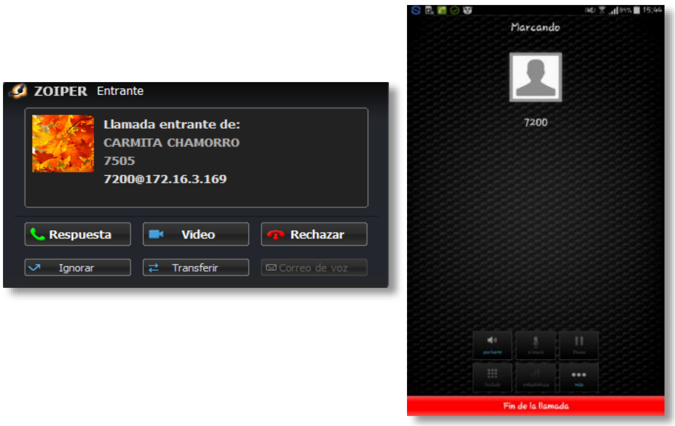

Figure 10. Communication between users through the server OpenNebula

In Figure 11 the successful communication process is observed as it is established the backbone between the server hosted in the cloud and university server in the operator panel Elastix server hosted in the cloud communication indicated and it is between a device with Zoiper softphone and Yealink phone network administrator for the University.

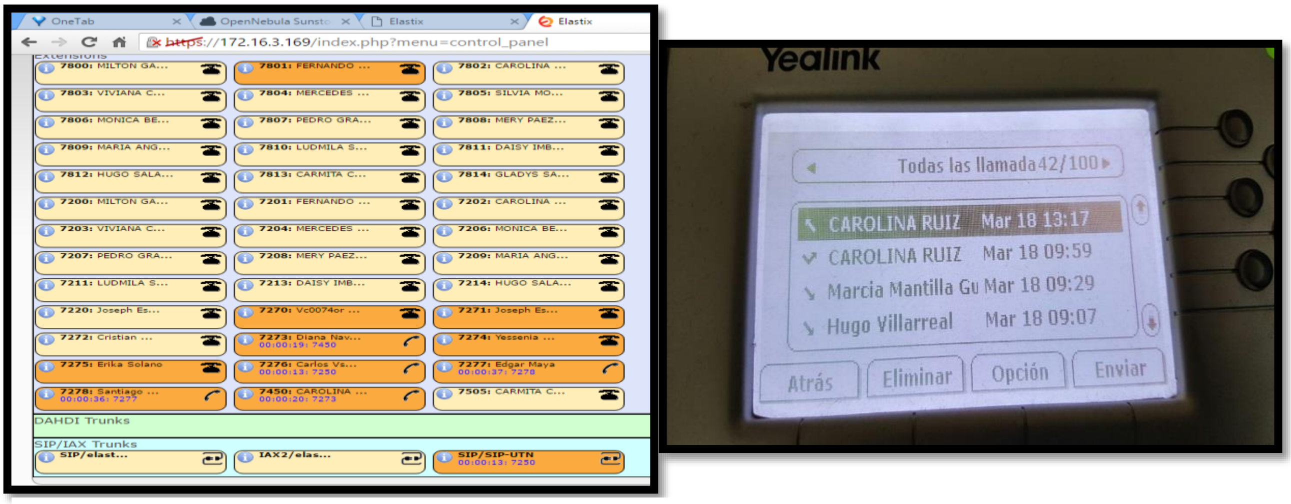

Figure 11. Communication between server user Elastix on OpenNebula and server user to the corresponding University Ing. Vinicio Guerra

4. CONCLUSIONS

This research has defined the design of IP telephony through cloud computing platform which is implemented in the data center of the FICA in the Técnica del Norte University consequently the result raised with the successful verification operation was obtained service where both the platform and software IP telephony solution under free software GNU / LINUX.

The importance of considering the capacity of the instance on the virtualization platform is to avoid problems of saturation with respect to the number of users that is handled as it should allocate resources specified in the sizing.

Taking into account traffic statistics calls that occur in the service at rush hour is necessary to make a good distribution of the resource bandwidth for IP telephony service as for proper operation requires a value of 20.60 Mbps.

In this first phase of implementation of the service, the tests were carried out within the network of the university campus and still not communicating with the external PSTN network but a virtual server with physical one if there is the possibility of troncalizar which if it was successful because the server does not have compatible interfaces FXS and FXO.

The communication was favorably with the two protocols IAX2 SIP signaling and communication with both locally and across trunk was obtained. To observe their behavior certain tools like Wireshark, reports the University Exinda server and software were used Zoiper.

The results of the tests server performance were efficient as no problems arose saturation and did not show any inconvenience, but this is due to the configuration currently maintained, which is devoted solely to performance testing, so if the University will deploy the resources listed in the sizing with what would be achieved to open this service on campus are required.

In tests of traffic flow the results were successful, the service works correctly, but certain drawbacks arise when different networks campus are used because they reflect some problems as permitted levels of jitter, delay and packet loss in it is concluded that the service is poor so you should apply a solution to improve their performance.

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ACKNOWLEDGEMENTS

At The “Universidad Técnica del Norte” for allowing perform this process, MSc. Edgar Maya and thanks to his guidance available was reached successfully obtain the culmination of this research.