Implementación de Central Telefónica Asterisk sobre Raspberry incluyendo Tarifación y Seguridad para el Usuario

Implementation of Asterisk (PBX) on Raspberry including Pricing and User Security

Implantação de Central Telefonica Asterisk on framboesa incluindo o carregamento e segurança do usuário

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Resumen

En este artículo se encuentra la descripción de un sistema completo de comunicación donde se saca provecho a todas las particularidades que una central telefónica pueda ofrecer, dirigida especialmente para medianas o pequeñas empresas que busquen el mejor sistema de comunicación por VoIP a un menor costo. Se destacan aspectos como la utilización de software gratuito de telefonía disponible, como Asterisk, para la implementación de la central telefónica y un sistema de tarifación con licencia libre llamado Servitux, el cual se utilizó para el control y cobro de cada llamada externa, también se manejó hardware de bajo costo, como es la tarjeta Raspberry Pi, donde se buscó aprovechar las ventajas del sistema operativo dando más seguridad por medio de programación en lenguaje php y html, brindando la posibilidad del cambio de clave para cada una de las diferentes extensiones en la central. Finalmente se realizan pruebas de estrés utilizando el software de licencia libre SIPp, definiendo la capacidad del sistema. También por medio de un análisis MOS queda demostrado que se puede satisfacer las necesidades del cliente en cuanto a la calidad y confiabilidad en el servicio de voz QoE.

Palabras clave: software, telefonía, Servitux, Asterisk, Raspberry, comunicación.

Abstract

This article is the description of a complete communication system where benefits to all the characteristics that a Business Telephone System can offer, especially for medium-sized or small companies who are seeking the best system of VoIP communication at lower cost. We highlights aspects as the use of free software available telephony, as Asterisk, for the implementation of the telephone exchange and a system of charging licensed free called Servitux, which was used for the control and payment of each external call, we also managed low-cost hardware, as the card is Raspberry Pi, where we sought to take advantage of operating system giving more security through programming in php and html language, providing the possibility of the change of key to each of the different extensions in the Private Branch Exchange (PBX). Finally there are stress tests using the SIPp free license software, defining the capacity of the system. Also using a MOS analysis are shown which can satisfy the needs of the customer in terms of quality and reliability in voice Quality of Experience (QoE).

Key words: software, telephony, Servitux, Asterisk, Raspberry, communication.

Resumo

Este artigo é a descrição de um sistema de comunicação completo onde ele tira proveito de todas as características que um call center pode oferecer, destinadas especialmente para as empresas médias ou pequenas que procuram o melhor sistema VoIP de comunicação a um custo menor. aspectos como o uso de software de telefonia livre disponíveis, tais como Asterisk, para a implementação da troca de telefone e sistema de carregamento com licença livre chamado Servitux, que foi usado para o controlo e recolha de cada estande chamada externa, também manipulados hardware de baixo custo, como a placa Raspberry Pi, onde procurou para tirar proveito do sistema operacional dando mais segurança através de programação PHP e HTML, oferecendo a possibilidade de mudar chave para cada um dos diferentes extensões central. Finalmente salientar testes são realizados usando o SIPp licença de software livre, definindo a capacidade do sistema. Também através da análise MOS demonstramos que pode atender às necessidades dos clientes em termos de qualidade e confiabilidade no serviço de voz QoE.

Palavras-chave: software, telefonia, Servitux, Asterisk, framboesa, comunicação.

Fecha Recepción: 2017 **Fecha Aceptación:** 2017

Introduction

The Public Switched Telephone Network (PSTN) is a fairly traditional and complex architecture used in communication systems. The growth of internet use is currently notable, this has provided the possibility to integrate telephony and implement it at the level of software, also called VoIP (Voice Over Internet Protocol), reducing cost and complexity, one of the most important services provided through information networks, even is adaptable to traditional systems of communication as the PSTN (Sinaeepourfard and Mohamed, 2011). Therefore, the different proposals of voice over IP not only are focused on large companies, but also medium-sized, small and different domestic environments.

Currently, there are several applications and programs that are used to implement a PBX. On the market are those that involve a cost of acquisition and use, but there are also those that are available online free of charge.

Within the programs available and no cost for the implementation of a this *Elastix*, configurable through a graphical interface, and *Asterisk PBX*, based on open-source *Linux*. These programs are quite handy, stable, profitable (Qadeer and Imran, 2008) and can be used on different types of *hardware* (Peláez and Tipantuña, 2014), allowing to obtain a variety of benefits to incorporating various alternatives of communication unifying services of telephony, such as e-mail, mail voice, internal communication, all in one platform (Dong, 2011; Masudur, M., and Sarwar, N., 2014).

To determine the quality of communication that develops into a PBX using *Asterisk* as software, testing of analysis or configurations on various characteristics of the plant, usually focusing on the internal communication between extensions (Li, Li, Wang and Nan, 2011).
Existe una variedad de sistemas embebidos, entre ellos la tarjeta *Alix* o *Raspberry Pi*, los cuales pueden ser utilizados como hardware de bajo costo (Villacis, Acosta, y Lara, 2013), para desarrollar y aprovechar al máximo todos los beneficios que *Asterisk* brinda como central telefónica (Estrada, Peláez y Tipantuña, 2015; Murkute & Deshmukh, 2015). Es posible incluir un *Gateway* de voz en el hardware del sistema, que permitirá que exista

comunicación externa permitiendo la entrada y salida de llamadas, desde y hacia la red telefónica PSTN (del inglés *Public Switched Telephone Network*) (Gupta, Agrawal y Qadeer, 2013).

The objective of this article is to show a complete telephone exchange that can be implemented specifically in small and medium enterprises, giving special importance to low cost software and hardware using Asterisk on the Raspberry Pi card, thus obtaining internal communication between extensions where experimentally it will be included For a single time a voice gateway for external communication, and will take into account the security in the use of the system by changing keys for access to each extension whenever it is deemed necessary, also will maintain a billing control And registration for external calls made by staff and customers of the company.

System Design and Implementation

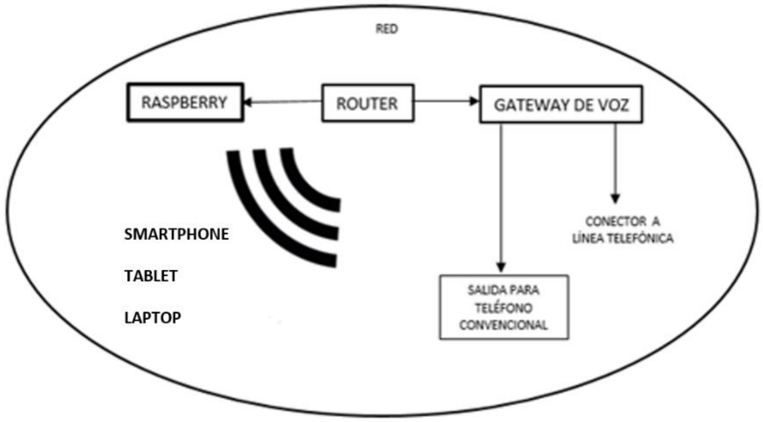
Case Study: Hotel

For the implementation of the system the following hardware was used:

- Computer or reduced board *Raspberry Pi*
- Gateway of voice Ht503
- *Router*

The implementation consists of the configuration of a telephone exchange with free software Asterisk (Asterisk.org, 2017) that uses the same network within the establishment, including telephone services, voicemail and also allows both internal calls between extensions and external To the conventional public network.

Figure 1. Block diagram of the implemented system.

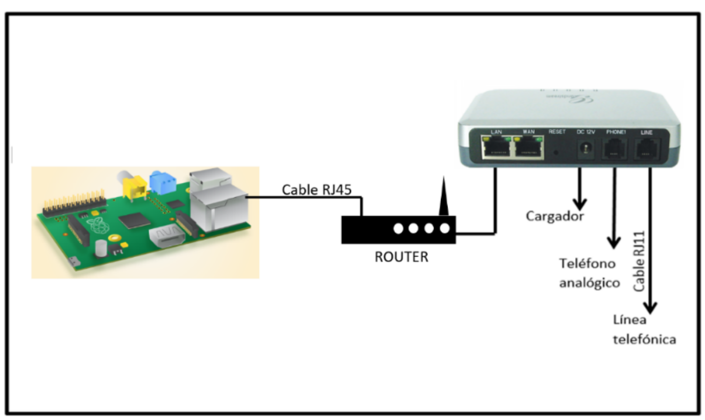


Source: Elaboración propia.

Figure 1 shows the necessary connections of the Raspberry Pi card, the Voice Gateway and the different communication devices of the hotel guests and staff. It will not be necessary to have a telephone in each room since the system allows any smart mobile phone, computer or tablet to communicate internally, making use of the telephone line through different free applications, also called in general softphones, through VoIP technology.

It is important to note that depending on the type of Gateway you can adapt any analog telephone to the system, in this case was adapted for Reception.

Figure 2. Connection diagram. Adapted from (RASPBERRY PI 1 MODEL B, 2017)



Source: (Grandstream, 2016).

Figure 2 shows the connection of the Grandstream HT503 (Grandstream, 2016) voice gateway, which will be connected to the telephone line to allow calls to and from the public network, including the conventional telephone in Reception, .

The different clients who arrive at the hotel and even the owner of the hotel and its employees will be able to connect to an extension with their respective key. In order to have a control and correct use of the telephone service, the implemented system includes the option of change of keys in a way accessible exclusively for the administration of the hotel.

For the control of external calls, a control and billing service has been adapted and configured in the system of the Raspberry card which registers the numbers dialed by each user of the exchange, the duration and the cost for each call, national or international.

Raspberry Configuration

The Raspberry card works specifically with an SD (Secure Digital) card that fulfills the function of hard disk, reason why it is necessary that it has enough memory, in this case 8GB, since it must carry an image mounted. Iso chosen from the publicly licensed Raspberry page. The free license image "raspbx-22-02-2015.iso" (Raspberry, 2017) was used, which will be recorded through the free software application Win32DiskImager, the operating system it handles is Debian and includes the software of Asterisk.

Once the system is loaded on the SD card, the same is placed on the Raspberry Pi, when the device is turned on it will open an interface for command lines, it is necessary to obtain from the repository the different updates of the packages available in the system with Internet access. After updating the system, proceed to the configuration of Asterisk, which is already installed inside the previously downloaded image.

Configuring incoming voicemail to mail

Incoming calls to the Reception of the hotel that have no answer, are the only ones that will have the option to leave voice messages, the same ones that will be sent to the mail indicated in its configuration.

Debian by default will have installed Exim4, which is a service for sending mails. For your configuration the following line will be written in the screen of commands:

```
>>dpkg-reconfigure exim4-config
```

Below is a screen with a series of questions, which will be answered according to the characteristics that the domain of the mail to be used.

In the file with /etc/exim4/passwd.client, the correct mail parameters are configured from where the messages will be sent from the voice mail, in this case the configuration using a Gmail account is the following:

```
gmail-smtp.l.google.com:usuario@gmail.com:contraseña
*.google.com:usuario@gmail.com:contraseña
smtp.gmail.com:usuario@gmail.com:contraseña
```

For the configuration of the mail that will receive the voice messages, the file /etc/Asterisk/voicemail.conf is edited, indicating the number where it is marked to listen to the voice message, the key to be able to access and the mail where the message is received By adding the following line:

```
Numero_para_mensaje_de_voz => clave, Usuario, emailrecepción@gmail.com ;
```

Extension Settings for PBX

Asterisk has its configuration files in the / etc / Asterisk folder which will be modified in order to adapt to the requirements according to the number of extensions needed for the establishment. The files that are modified are sip.conf and extensions.conf, they can be accessed by commands in the console.

The sip.conf file specifies the number of extensions, the channel parameters are programmed for each of the users registered in the file, such as ports, sound codecs, audio, and different connection types. For the configuration of each of the assigned extensions, a context must be taken into account that is configured in the file extensions.conf, in this case called "users" and using the parameters:

```
[Numero extensión]
username=Habitacion_numero_de_habitacion
type=friend
host=dynamic
secret=clavehabitacion
context=users
qualify=yes
nat=yes
dtmfmode=rfc2833
```

For the extension of the Reception is taken into account the configuration to store voice messages that can be addressed in the same way as the phone, to the mail that is assigned, simply following the example above by increasing the line mailbox = num @ default at the end of the configuration.

It is important to configure two extensions, one specifically to allow the outgoing calls to the telephone network and another to receive calls from the telephone network.

In the extensions.conf files the dialing plan is configured, it is where the behavior of the telephone exchange is defined by defining contexts, what actions to take when there is a call to and from the telephone network (PSTN) and when it is carried out A call between internal extensions. The context "users" is the one that is called by each extension in the file sip.conf, in this case will be configured 25 extensions under the same context, of which 13 belong to the rooms:

```
[users]
Exten => Num_de extensión,1,Dial(SIP/ Num_de_extension,20)
```

For the extension assigned to Reception in the configuration it is indicated that the call that goes to Reception and if it does not receive a reply, will be redirected to an extension assigned to the hotel owner or employee of the same, if the call is still not answered Finally it will be sent to a voicemail:

```
exten => Num_extension_recepcion,1,Dial(SIP/ Num_extension_recepcion,20)
exten => Num_extension_recepcion,2,Goto(users, Num_extension_propietario,1)
exten => Num_extension_recepcion,n,VoiceMail(num @default)
```

When an external call is made from any extension, it is indicated in the configuration that the call must start with a specific number for national calls, another for international calls and another for cellular calls followed by any number of numbers between 0 and 9, Depending on the numbers required. The system will send the number without the first digit using the sip extension to the Gateway to connect to the PSTN:

```
exten => _7x. ,1,Dial(SIP/9/${EXTEN:1})
```

Next, the Interactive Voice Response (IVR) will be configured to work when receiving calls from the telephone network (PSTN), the IVR will give options to direct the call and communicate with the extensions of the rooms or with the Reception using the same context:

```
exten => s,n.Background(Grabacion_IVR)
exten => s,n,WaitExten(4)
exten => Num_Habitacion,1,Goto(users,Num_extension_habitacion,1)
```

HT503 Voice Gateway Configuration

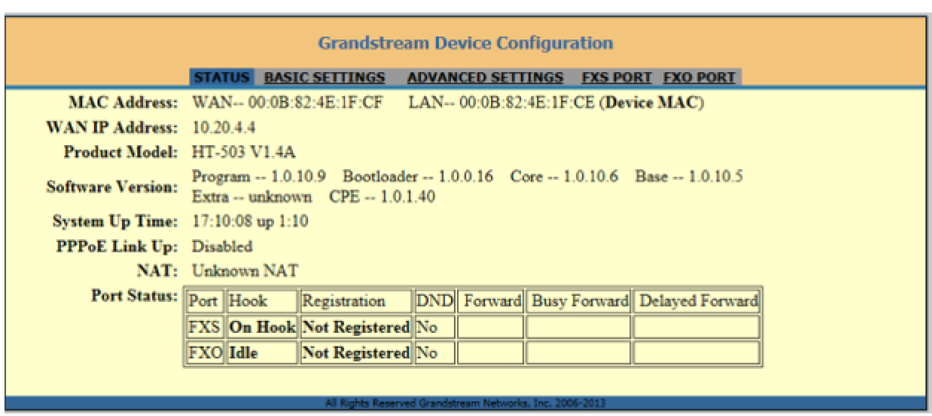
For the implementation of the system at the hotel, the Gateway GrandStream HT503 (Grandstream, 2016) was used, which will allow communication with the PSTN to be established with incoming and outgoing calls. There are different brands and models of Gateway economic that give the possibility of adapting analog phones to the system, depending on how many are used, in this case only one is needed.

Figure 3. Voice Gateway Physical Ports

Source: (Grandstream, 2016).

The Gateway GrandStream HT503 has two analog ports, one FXS represented as PHONE and one FXO represented as LINE, in Figure 3 they are physically represented by the PHONE and LINE inputs, it also has two Ethernet type interfaces.

Figure 4. Configuration tabs for Voice Gateway (Configuring the Grandstream HandyTone 503/HT-503, 2015).



For configuration there are three important tabs as shown in Figure 4, in Basic Settings the network parameters of the gateway will be configured, in FXS PORT the extension is configured where the calls will be received, while in the FXO PORT tab Configures the extension where the calls will be sent, both the FXS and FXO ports are entered the Internet Protocol (IP) address of the Asterisk server.

Control and charging service for calls to the conventional network

Within the implementation of the entire system for the telephone exchange is necessary the control of the external calls of each employee and client.

Following the configuration steps of the Servitux website (Servitux Servicios Informáticos SL., 2016), the OPEN Servitux Tarificador free software application was configured and installed for Asterisk, it does not depend on the telephony infrastructure, it only needs the CDR (From the English Call Detail Record) of Asterisk where all the information related to the calls is stored.

The application is very simple to manage since it allows you to view a record that shows the national, international and cellular external calls of each extension registered in the system, you can control the number of extensions, in case more were created or Eliminate one within the telephone exchange.

The software being configurable has several benefits, such as recording the price of calls, whether national, international or cellular, and the software calculates and shows the total cost of each call per extension.

Key control for extensions

For each extension you have an initial assigned key, but it is necessary that they are in a constant change since the user of each one of them can make a bad use making calls that possibly do not get to be charged.

In order to control the change of keys, the search inside the sip.conf file, which is where the keys are assigned, is programmed in the php and html code, the line "secret = key_asignada" where the key assigned by any different key will be replaced, You can enter the graphical interface from any smart computer or laptop that is inside the network where you will be asked to enter the old key and the new key that you wanted to assign; The owner or principal manager of the hotel will be the only ones who know the address to access the interface.

Load Test

For the load test it is intended to simulate the sending of simultaneous calls with the free software SIPp (SIPp, 2017).

SIPp sends a SIP-Invite message to the Asterisk server, while Asterisk sends another SIP-Invite to the destination, Asterisk forwards call flows to the client, SIPp at the end of the transmission sends a message from Bye to the Asterisk server.

SIPp have a client / server type of operation, allows the creation of custom scenarios, defining call flows, once the PBX is implemented, it must be installed starting with the following commands:

```
apt-get install c++ libncurses5-dev libpcap0.8-dev libnet1-dev
```

```
wget http://surfnet.dl.sourceforge.net/sourceforge/sipp/sipp.3.1.src.tar.gz
```

As required it will be configured both server and client, these scenarios can be found on the SIPp page (SIPp, 2017) where there are files in xml format depending on the scenarios that are required. It is important to include the following line in the sip.conf file to accept all calls without the authentication requirement:

```
Allowguest=yes
```

To execute the load tests the following command will be executed:

```
./sipp -sf UAC.xml -s extension_cliente_asterisk -l num_llamadas_simultaneas -m numllamadas_enviar -r llamadasxsegundo - trace_screen -trace_err -recv_timeout 400000 -t un -nr
```

It uses trace_screen, to create the log file with statistics, trace_shortmsg that is responsible for creating the file that sends and receives SIP messages, trace_err handles the error log file, recv_timeout 400000 is the timeout in milliseconds.

The stress test was performed first by sending 20 calls:

sipp -sf ./UAC.xml -s 333 127.0.0.1:5080 -l 2 -m 20 -r 2 -trace_screen -trace_shortmsg -trace_err -recv_timeout 400000 -t un -nr

Figure 5. SIPp test with 20 calls.

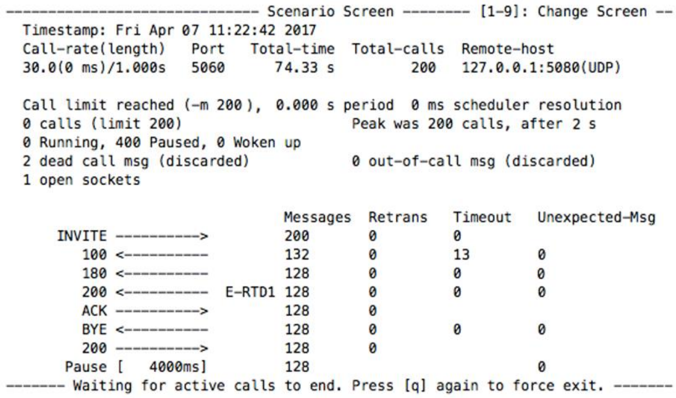
```
------------------------------ Scenario Screen -------- [1-9]: Change Screen -
  Timestamp: Fri Apr 07 18:30:30 2017
  Call-rate(length)   Port   Total-time  Total-calls  Remote-host
  2.0 (0 ms)/1.000s   5060     174.64 s         20    127.0.0.1:5080(UDP)

  Call limit reached (-m 20),  0.000 s period  0 ms scheduler resolution
  0 calls (limit 2)                       Peak was 20  calls, after 6 s
  0 Running, 388 Paused, 0 Woken up
  0 dead call msg (discarded)             0 out-of-call msg (discarded)
  1 open sockets

                               Messages  Retrans   Timeout   Unexpected-Msg
      INVITE ---------->         20        0         0
         100 <----------         20        0         0         0
         180 <----------         20        0         0         0
         200 <----------  E-RTD1 20        0         0         0
         ACK ---------->         20        0
         BYE <----------         20        0         0         0
         200 ---------->         20        0
       Pause [   4000ms]         20                            0
------- Waiting for active calls to end. Press [q] again to force exit. ------
```

Source: Elaboración propia.

It can be seen in figure 5 that the system took 174.64 seconds to process the 20 calls, there was no problem since it completes the traffic of sent calls, now it will be tested with 200 calls:

sipp -sf ./UAC.xml -s 333 127.0.0.1:5080 -l 200 -m 30 -r 200 -trace_screen -trace_shortmsg -trace_err -recv_timeout 400000 -t un -nr

Figure 6. SIPp test with 200 calls.



```
------------------------------ Scenario Screen -------- [1-9]: Change Screen --
  Timestamp: Fri Apr 07 11:22:42 2017
  Call-rate(length)   Port   Total-time  Total-calls  Remote-host
  30.0(0 ms)/1.000s   5060      74.33 s         200   127.0.0.1:5080(UDP)

  Call limit reached (-m 200 ),  0.000 s period  0 ms scheduler resolution
  0 calls (limit 200)                     Peak was 200 calls, after 2 s
  0 Running, 400 Paused, 0 Woken up
  2 dead call msg (discarded)             0 out-of-call msg (discarded)
  1 open sockets

                                 Messages  Retrans   Timeout   Unexpected-Msg
      INVITE ---------->         200       0         0
         100 <----------         132       0         13        0
         180 <----------         128       0         0         0
         200 <----------  E-RTD1 128       0         0         0
         ACK ---------->         128       0
         BYE <----------         128       0         0         0
         200 ---------->         128       0
       Pause [   4000ms]         128                           0
------- Waiting for active calls to end. Press [q] again to force exit. -------
```

Source: Elaboración propia.

It can be observed in Figure 6 that the traffic of 200 calls was not completed, 128 were established simultaneously, which concludes that 128 is the capacity of the system implemented according to the characteristics of the same.

Cost Analysis Central Asterisk Vs. Brand Centers

Telephone exchanges as such prevent the connection of telephones separately to the public telephone network, which means that monthly charges are saved on the telephone line for those who use them.

The implemented Asterisk based exchange gives rise to savings compared to other private telephone exchanges which for your costs depend on your brand.

Equipment Cost

For its reference cost, the equipment used for the proper operation of the system must be taken into account, since stable software and reliable hardware must be integrated, which will guarantee the call traffic, defining the total cost in Table 2. Of the implemented plant Have the following characteristics:

Table 1. Central Cost Implemented.

Hardware	Precio	Caracteristica
Raspberry Pi modelo B	$65.00	Puede manejar hasta 100 extesiones
Gateway Grandstream HT-503	$80.00	1 FXO, 1 FXS
Cable RJ45	$3.00	Incluido conector RJ45
Router	$60.00	TP-LINK
TOTAL	$208	

Source: Elaboración propia.

Its free license does not require the update of the same with additional charge and the information that can be found is very broad, the central will work correctly without wasting resources at a reduced price in the event that the hardware suffers some damage and without No cost in case of damages or software misconfiguration.

The implemented plant has passed connected without the equipment used to suffer from overheating or the system itself tends to slow down.

We have considered the Avaya, CISCO and 3COM brands in Table 2 as the main competitors against the Asterisk hub, in itself, any free software is a threat to these brands. The 3 brands have centrals addressed to medium and small companies but are managed at a certain cost depending on the characteristics of the telephone exchange:

Table 2. Cost of Power Plants PANASONIC, AVAYA y 3CX .

MARCA	PRECIO	CARACTERISTICA
PANASONIC	$800-$1189	Central adaptable a teléfonos analógicos y dispositivos que cuenten con softphone
AVAYA	$480 (Precio depende de País donde se lo requiera)	Es necesario la compra de licencia para cada teléfono IP AVAYA. (AVAYA, 2017)
3CX	$320	Depende de las características de la central que se requiera ya que su precio varía desde una versión gratuita hasta una versión Pro (3CX,2017)

Source: Elaboración propia.

It should be taken into account that they do not handle a free license, so it is necessary to renew it. Depending on the provider you pay for each additional extension that is required.

Results in MOS Analysis

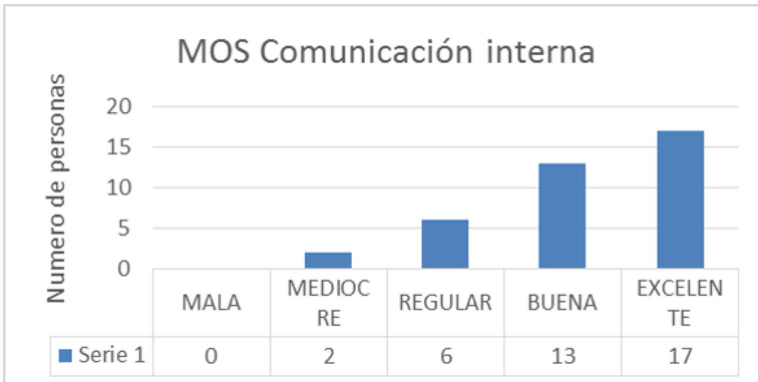
After the implementation of the system, it is necessary to analyze the quality of the service, to understand how customers perceive quality when making their calls, this will be done using the MOS (Mean Opinion Score) analysis for MOSC conversational situations (Mean Opinion Score Conversational), the communication will be evaluated on an Absolute Category Rating defined by ITU-T P.800 (Itu.int., 2016) Excel = 5, Good = 4, Regular = 3, Mediocre = 2, Mala = 1.

It will evaluate the quality of internal communication between extensions and external with the exit of calls to the PSTN in the most realistic situation.

For internal communication, the analysis for being conversational will be 2 participants installed in different rooms maintaining a structured conversation, ie having a beginning, a development and an end. By observation the conversations in the hotel are composed by a greeting, a request by the guest, a response from the hotel employee and to finalize a farewell. Participants at the end of the conversation qualify the communication according to the aforementioned scale and evaluate the communication regarding the degradation of the voice that may exist due to the echo, noise or any interruption that may arise.

In 7 days, of the 13 rooms, a sample of 38 people were taken, regardless of the time they stayed in the hotel, they used the internal communication system, the quality rating had the following results:

Figure 7. Internal communication MOS representation.

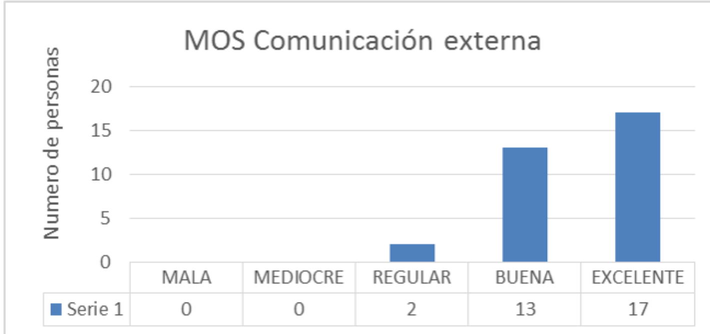


Source: Elaboración propia.

According to Figure 7 there is the possibility that the communication has not been successful in all calls, the IP central will always be exposed to certain interferences, interruptions or distortions, comparing it can be observed that the tendency in the opinion of the people is Between GOOD and EXCELLENT.

For the external communication we tried to evaluate if the users could use the system to make calls from their cellphone or tablet with the softphone application to any telephone number, whether national, international or cellular and thus to have a conversation with their respective start, Development and purpose.

Figure 8. Representation MOS, external communication to the telephone network.



Source: Elaboración propia.

Among the different opinions of the people who participated in the quality tests most stated that, they found a little difficulty with the handling of the softphone but finally managed to establish successful communication, people who failed to establish communication rated the communication as regular and That failed to configure their cell phone.

The system for the most part tends to be GOOD or EXCELLENT, this is how the successful system can be described and it is important to take into account the adaptability of the client depending on the business where the telephone exchange is to be implemented.

Discussion and Conclusions

Within a business it is important to maintain direct contact with customers on a constant basis, so acquiring a communication system has become essential. IP technology offers several advantages, since it optimizes resources through the same Internet network, for voice transmission does not require the dedication or exclusive payment of a specific network as was usually done with traditional telephony, all this allows A reduction of infrastructure, hosting all these services in a virtual "Switchboard" to which you can add new and better functionalities that with traditional telephony were unreachable or presented fairly high costs to achieve this.

With the implementation of the Asterisk-based Telephone System, there is no risk that it will become obsolete due to some lack of functionality or necessity, since they are adaptable to any system, even to old telephony, allowing the installation of the same A problem but rather a solution that in the long run can be updated at any time.

In the different phases of the project, possible solutions were planned in both software and hardware to develop a complete communication system, however its configuration as free open source programming is susceptible to failures, itself vulnerable to human error.

The implemented system not only offers communication between the different network terminals, it also offers value added services such as security for each extension with key management and control of calls both internal and external, depending on the business also offers billing system, with all this Demonstrates that for companies that do not have many resources can access without problems to high quality systems using VoIP systems that end up being equal to or better than those using a traditional fully analog telephone network.

The design of the "Switchboard" has been based on the use of free and free software with a notable decrease in expenses compared to other IP Centrals of recognized brand; Is a great competition for the different services that can be incorporated at no cost, concluding that a resource can be introduced in the market with the same or better characteristics than the usual switchboards of the big suppliers at a cheaper price.

When implementing the system with open source software, it is possible to control the use of different IP telephony services, so that each client is aware and makes rational use of the service.

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