MIGRATION OF TELEPHONY IP FROM A PROPRIETARY PLATFORM TO A PLATFORM UNDER FREE SOFTWARE GNU/LINUX FOR TECHNICAL NORTH UNIVERSITY

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Summary.- Telephony IP is an application of VoIP technology, which is part of multi-service networks that integrates voice and data via the IP protocol. This application has achieved cost reduction, integration of services across networks and additional features.

This article aims to make an analysis of the requirements for Telephony IP migration of a proprietary platform under a free software platform.

I. INTRODUCTION

The IP telephony system at the Technical North University (UTN), is currently operating under a Cisco platform that is handled in a closed manner; that is, using protocols and proprietary standards and is subject to licensing costs.

Telephony Migration to a platform based on Free Software provides independence and guarantee future; signaling protocols and interoperable standards, possible regardless of vendor proprietary protocols support and integration capability with pre-existing, including higher standard codecs, provided the UTN a VoIP system, robust, scalable, flexible and high environments performance, which has the same features and some additional current service, which can be easily extended depending on the characteristics of structural and organizational growth.

II. BASIC CONCEPTS

A. VoIP Definition

VoIP is a technology that allows the transmission of voice over IP protocol, sending the same voice without using the traditional analog circuitry PSTN telephony packets. Voice over Internet protocol uses its own computer language and is subject to certain service policies.

B. VoIP Operation

"The VoIP is the result of the following process; first the voice of a continuous signal in time and amplitude discrete transforms, this signal is sampled and then be quantified (evaluated at the time), then this signal is encoded and compressed last. Once this process has been performed, the voice is in binary form, making it possible to form packets to be sent through the data network. "The following figure (see Figure 1), the process shown digitized voice and sent by a data channel:

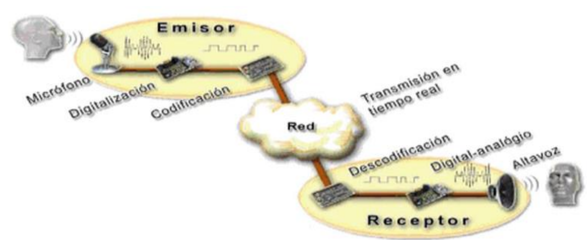


Figure 1. Voice transmission in digital networks.
Source: Carballar Falcón, J. A. (2007). *VoIP. La tecnología de Internet.*

C. IP Telephony

IP telephony is the immediate implementation of VoIP. It is the provision of telephone services where the transport network for voice is a data network under the IP protocol. Allows ordinary telephone calls using the Internet, from a PC and / or terminals that support the protocol. To make a phone call via the IP protocol, voice must be digitized, compressed and sent with IP datagram format.

The simplification of the communications infrastructure in a unified telephony system via the data network, IP telephony makes an application with many advantages, including: centralized management, free internal calls, numbering plan and access to functionalities such as IVR, voicemail, integration manager agendas with emails and many applications that can adapt to the needs of companies.

D. *H.323 Protocol*

This protocol was created to transmit voice, video and data over packet switched networks in real time. Standards for compression and decompression of audio and video set and specifies interoperability mechanisms equipment manufacturers. H.323 has the ability to seamlessly integrate with the PSTN and simultaneously send communications over the Internet. Calls are sent over the TCP protocol over port 1720.

Architecture H.323

It consists of a set of features and functions, required and optional. It has four components:

Terminals: They are at the ends of the network and provide two-way communications in real time. H.323 terminals must support mandatory communications with the codec G.711.

Gateway: They are responsible for providing real-time interface terminals with terminal H.323y the switched network gateways.

Gatekeeper: Its primary function is the address translation and access control to the network.
It is not a mandatory requirement in an H.323 system, but for large networks is the need to centralize the management of call control. An important feature of a gatekeeper is call routing.

Multipoint Control Unit: Designed to withstand the conference between three or more H.323 terminals. It consists of three major components:

- Multipoint Controller (MU)
- Multipoint Processor (MP)
- H.323 proxy

E. *SIP Protocol*

Is a signaling protocol for creating, modifying and terminating multimedia sessions with one or more participants in an IP network that is transmitting voice, video, instant messaging, network games or any application in real time, SIP works on TCP port 5060 and UDP 5061.

- It is based on text, is open and flexible, similar to HTTP and SMTP.
- The message protocol is request / response and the model that handles is Client / Server.
- SIP is extensible and fits comfortably in different architectures and deployment scenarios.
- It is connection-oriented end-to-end.
- SIP is not a general purpose protocol; ie only helps establish and terminate communication.

SIP Components

They are elements involved in a SIP session and are 2 main: user agents and servers.

User Agent (UA): These are the endpoints of the SIP protocol. They are responsible for submitting and processing protocol messages. An agents do not care user interface, but their messages and behavior. Exist between the UA:

- User Agent Client (UAC): Those that generate SIP requests and receive responses.
- User Agent Server (UAS): Those that generate answers to the SIP client requests.

Servers: SIP servers are classified as:

- Registration Server: Satisfy SIP REGISTER requests, update the database location. Help with logical addresses of users in the format user @ domain.
-

- Proxy Server: Routed meeting invitations and send them to user agents. Also revised SIP queries and processed and shipped to other SIP servers.
- Redirect Server: They generate answers to who generates the communication destination address or the server that approaches the destination. It also listens for requests and responses issued location.

F. IP-PBX

An IP-PBX is a PBX that works on the Internet protocol, using the data infrastructure of a company. It is fully digital, accept voice and video.

Types of Links

Links are (core) internal connections of the plant, these devices connect IP PBX.

Analog links: Are links that work like a traditional analog phone line and connect. Do not support high speed transmission of data and allow a simultaneous single line connection.

Digital links: known as T1 or E1 links, have a bandwidth of 2 Mbps, currently installed on fiber using three voice channels without multiplexing. Links between Central: are links connecting stations together.

Currently we are developing a QSIG standard, which allows interfacing with equipment from different manufacturers.

VoIP links: Are links on any form of IP transport. They behave like analog lines: between the lines and the central elements that convert voice and this goes to an analog channel and then are inserted into the IP network are installed.

GSM links: As in the case of VoIP, there are devices that make calls using the GSM network.

G. Traffic Theory

In networks with Voice over IP, traffic theory is the tool used to size the telephone system and allocate resources, eg bandwidth; with a minimum Grade of Service (GoS).

H. Traffic Flow

If the amount of traffic generated is known and required GoS is possible to calculate the number of trunks required for the system. Traffic flow can be calculated using:

$$A = C * T$$

Equation 1. Formula for calculating the flow of traffic.

Where:
A: traffic offered
C: number of calls originated in an hour
T: time average duration of a call

III. IP TELEPHONY TRAFFIC

A. Setting the Peak Time

The time traffic are 60 minutes a day, in which telephony traffic recorded a significant increase, ie, it is the time in which telephone calls have a maximum traffic.

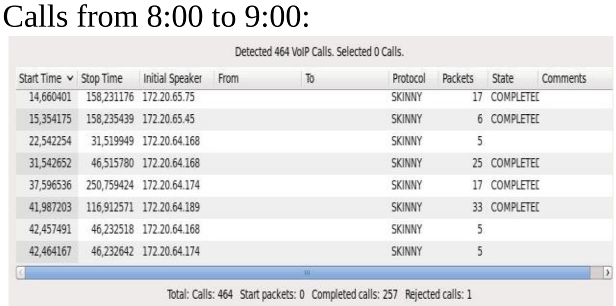
To calculate the hour traffic quickly, take into account one day weekday of the month and that traffic multiplied by 15% or 17%, as it is considered as a general rule the more traffic that occurs during one day.

B. Analysis hour traffic at UTN.

To determine when traffic on the UTN, with Wireshark tool, network monitoring determines the rush hour (peak hour) is between 12:00 and 13:00.

Below is shown on pictures telephony traffic at rush hour, taken the day June 9, 2014.

Calls from 8:00 to 9:00:



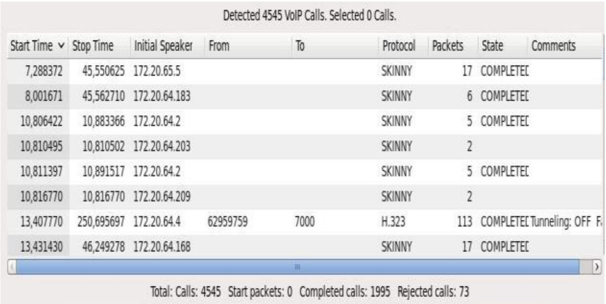
Detected 464 VoIP Calls. Selected 0 Calls.

Start Time	Stop Time	Initial Speaker	From	To	Protocol	Packets	State	Comments
14,660401	158,231176	172.20.65.75			SKINNY	17	COMPLETEI	
15,354175	158,235439	172.20.65.45			SKINNY	6	COMPLETEI	
22,542254	31,519949	172.20.64.168			SKINNY	5		
31,542652	46,515780	172.20.64.168			SKINNY	25	COMPLETEI	
37,596536	250,759424	172.20.64.174			SKINNY	17	COMPLETEI	
41,987203	116,912571	172.20.64.189			SKINNY	33	COMPLETEI	
42,457491	46,232518	172.20.64.168			SKINNY	5		
42,464167	46,232642	172.20.64.174			SKINNY	5		

Total: Calls: 464 Start packets: 0 Completed calls: 257 Rejected calls: 1

Figure 2. Analysis of rush hour on the UTN 8:00 to 9:00.
Source: Wireshark, UTN, 2014.

Calls from 9:00 to 12:00:



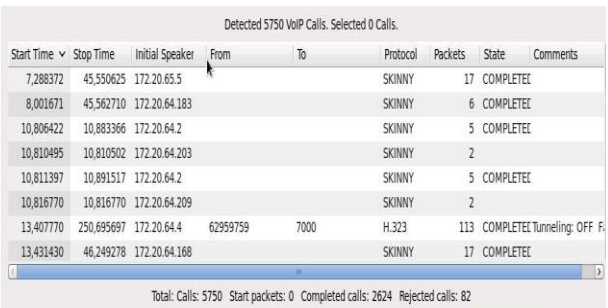
Detected 4545 VoIP Calls. Selected 0 Calls.

Start Time	Stop Time	Initial Speaker	From	To	Protocol	Packets	State	Comments
7,288372	45,550625	172.20.65.5			SKINNY	17	COMPLETED	
8,001671	45,562710	172.20.64.183			SKINNY	6	COMPLETED	
10,806422	10,883366	172.20.64.2			SKINNY	5	COMPLETED	
10,810495	10,810502	172.20.64.203			SKINNY	2		
10,811397	10,891517	172.20.64.2			SKINNY	5	COMPLETED	
10,816770	10,816770	172.20.64.209			SKINNY	2		
13,407770	250,695697	172.20.64.4	62959759	7000	H.323	113	COMPLETED	Tunneling: OFF F.
13,431430	46,249278	172.20.64.168			SKINNY	17	COMPLETED	

Total: Calls: 4545 Start packets: 0 Completed calls: 1995 Rejected calls: 73

Figure 3. Analysis of rush hour on the UTN 9:00 a 12:00.
Source: Wireshark, UTN, 2014.

Calls from 12:00 to 13:00:



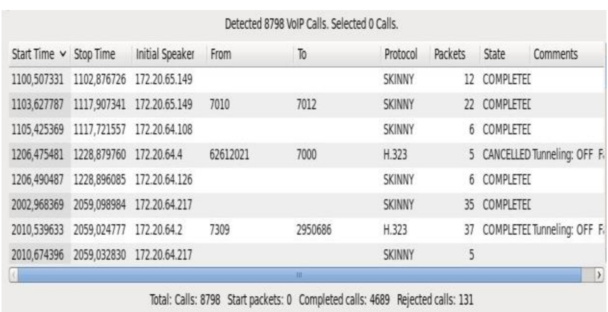
Detected 5750 VoIP Calls. Selected 0 Calls.

Start Time	Stop Time	Initial Speaker	From	To	Protocol	Packets	State	Comments
7,288372	45,550625	172.20.65.5			SKINNY	17	COMPLETED	
8,001671	45,562710	172.20.64.183			SKINNY	6	COMPLETED	
10,806422	10,883366	172.20.64.2			SKINNY	5	COMPLETED	
10,810495	10,810502	172.20.64.203			SKINNY	2		
10,811397	10,891517	172.20.64.2			SKINNY	5	COMPLETED	
10,816770	10,816770	172.20.64.209			SKINNY	2		
13,407770	250,695697	172.20.64.4	62959759	7000	H.323	113	COMPLETED	Tunneling: OFF F.
13,431430	46,249278	172.20.64.168			SKINNY	17	COMPLETED	

Total: Calls: 5750 Start packets: 0 Completed calls: 2624 Rejected calls: 82

Figure 4. Analysis of rush hour on the UTN 12:00 a 13:00.
Source: Wireshark, UTN, 2014.

Calls from 14:00 to 17:00:



Detected 8798 VoIP Calls. Selected 0 Calls.

Start Time	Stop Time	Initial Speaker	From	To	Protocol	Packets	State	Comments
1100,507331	1102,876726	172.20.65.149			SKINNY	12	COMPLETED	
1103,627787	1117,907341	172.20.65.149	7010	7012	SKINNY	22	COMPLETED	
1105,425369	1117,721557	172.20.64.108			SKINNY	6	COMPLETED	
1206,475481	1228,879760	172.20.64.4	62612021	7000	H.323	5	CANCELLED	Tunneling: OFF F.
1206,490487	1228,896085	172.20.64.126			SKINNY	6	COMPLETED	
2002,968369	2059,098984	172.20.64.217			SKINNY	35	COMPLETED	
2010,539633	2059,024777	172.20.64.2	7309	2950686	H.323	37	COMPLETED	Tunneling: OFF F.
2010,674396	2059,032830	172.20.64.217			SKINNY	5		

Total: Calls: 8798 Start packets: 0 Completed calls: 4689 Rejected calls: 131

Figure 5. Analysis of rush hour on the UTN 14:00 a 15:00.
Source: Wireshark, UTN, 2014.

In short, calls originating on June 09, are shown in Table 1:

Table 1. Data obtained from call traffic analyzer

HORA DE LLAMADA	NÚMERO DE LLAMADAS
08:00 a 09:00	464
09:00 a 12:00	4545
12:00 a 13:00	5057
14:00 a 17:00	8798

Source: Wireshark, UTN, 2014.

To perform the necessary calculations the following call data is referenced from 12:00 until 13:00.

Number of calls in 1 hour:

$$Número\ de\ llamadas = 1205$$

$$Tiempo\ promedio\ de\ cada\ llamada = 43.25\ segundos$$

Equation 2. Total calls in one hour
Source: Wireshark, UTN, 2014

C. *Traffic Offered Calculation*

It is known as traffic flow, its unit is the Erlang and is calculated using the formula:

$$A = C * T$$

Equation 3. Traffic Flow
Source: Traffic in Telecommunications Networks. Professor Diogenes Marcano

Now, in the case of the UTN, we have:

$$A = C * T$$

$$A = \frac{1205\ llamadas * 47.12\ segundos}{3600\ segundos}$$

$$A = 15.77\ Erlangs$$

Equation 4. Traffic Flow
Source: Wireshark, UTN, 2014

Then, the approximate value of 15.77 Erlangs is the amount of traffic that courses on the current telephone network Technical University North. This requirement serves as a reference for projecting future growth in the migration of service.

D. Grade of Service (GoS)

The level of service is a condition of QoS that is used in telephony, and is defined as the probability of blocking a call to the first attempt at rush hour.

For the GoS is acceptable, the probability value must be less than 1% blocking rush hour.

The GoS simplified formula is:

$$GoS = \frac{Número\ de\ llamadas\ rechazadas}{Número\ de\ llamadas\ realizadas}$$

$$GoS = \frac{22}{1205}$$

$$GoS = 0.0182$$

Equation 5. Grade of Service.
Source: Traffic in Telecommunications Networks. Professor Diogenes Marcano

By Erlang B table and C, were used for this case since the two tables must find values for two cases: Erlang B for when you want to find the GoS without any queues; and Erlang C, when taking into account the queues at the GoS.

Knowing the number of channels and the average call time can find the amount of traffic:

$$Número\ de\ canales = 30$$

$$GoS = 0.0182$$

Equation 6. Information channels and GoS
Source: Data from UTN

IV. CHOICE OF SOFTWARE

The choice of IP telephony software for use in the UTN, is based on the IEEE-830 SRS, after the development of this standard of comparison, once established parameters for selection, which is evaluated software that has the best characteristics.

The following requirements are taken into account:

1. Architecture
2. Current Status
3. Interfaces
4. Hardware for IP Telephony
5. Administration
6. Number of Users Supported
7. Number of Concurrent Calls
8. Commercial Level
9. Supported Protocols
10. Supported Codecs
11. Report Calls
12. IVR - Call Manager
13. Flexibility
14. Scalability
15. Interoperability

A. Rating Software for Use

After qualifying the three software, Elastix is the application of IP telephony suitable for their features and functionality it provides.

- Elastix is a free software platform, with a high commercial level, applied to small, medium and large enterprise solutions for user support (from 2-1000). As it has a high level of trade integration, there is much information on the internet and books that are accessible to users, so that the system can be updated at any time and in case of a failure the solution can be easily found.
- It has a modular architecture, ie, that within the platform several services that operate independently grouped: Centos Linux operating system, Asterisk IP-PBX, web servers and database, SMTP mail services and Hylafax for work with fax. Thus forming a robust software for IP voice communications.
- As Elastix is intended for dedicated voice over IP solutions, has support analog and digital interfaces.
- Elastix is designed natively for SIP and IAX communications, can work with IP addresses from end to end, thus reducing the processing time networking equipment to eliminate analog-digital conversion. However, they can join this platform protocols: MGCP, H.323, SCCP. Thus provides flexibility and interoperability features.
- Elastix is software that adapts to changes without much difficulty, and is easily scalable. Through a trunk link between Elastix and Cisco, the two systems can coexist until the migration is complete.
- For the above features it is concluded that the best software is Elastix, for services, for ease of administration. And by integrating in a single software distributions. The proposed with

Elastix, system benefits the institution that will feature an updated and improved service, making way for future projects to unified communications. It should also be emphasized that Elastix is a product that has been developed in Ecuador, becoming a robust and innovative application for the benefit of the world of TIC's.

V. DESIGN IP TELEPHONY MIGRATION UNDER THE PLATFORM GNU / LINUX

To measure the VoIP system, it draws on the ISO / OSI model, VoIP parameters for each layer are studied, based on requirements necessary.

1. **Physical Layer:** The internal network UTN has a structured twisted pair UTP, Category 5e and 6. For IP telephony wiring, the institution has an analog access it provides an E1 with CNT EP. The phone line with which it communicates to the PSTN, is 2997800.

2. **Data Link Layer:** Found an Ethernet architecture. This layer is logically distributed networking equipment (switches) that are connected in each building of the UTN. And are configured 200 extensions.

3. **Network Layer:** Use the Internet Protocol, IP addresses are distributed 172.X.X.X network, this network is subdivided logically VLANs that are configured on a switch
In addition to designing the new IP telephony system, it is necessary to know the amount of bandwidth to be allocated to provide a quality service.

4. **Transport Layer:** For internal data network UTN, provide the IP telephony service should use a transport protocol is connectionless UDP, since it makes no flow control and error and does not allow broadcasts, reliability by changing a gain of speed, typical of VoIP applications. In addition to the UDP protocol, working with the RTP protocol, its function is to multiplex multiple data streams in real time in a single UDP packet.

5. **Session Layer**: This layer is responsible for establishing and terminating the communication link between the sending and receiving devices, managing the session to IP telephony in the UTN, SIP, signaling set is used, configure, customize and ends multimedia sessions in this case works between central and voice terminals.

6. **Presentation Layer:** Responsible for encrypting data, also compressed to reduce its size. So in this layer the appropriate codec for IP telephony UTN is determined.
The following table (See Table 2), shows a summary of the codecs that are used:

Table 2. Comparison of Codecs

Nombre	Bit rate (kb/s)	Sampling rate (kHz)	Período de Empaquetamiento (ms)	Licencia	MOS[36] (Mean Opinion Score)
G.711	64	8	20-30	No	4.4
G.722	64	16	20-30	No	3 a 4
G.726	32	8	20-30	No	3.85
G.729	8	8	20	Si	4
GSM	13	8	20	No	3.7
Speex	8, 16, 32	8/16/32	30	No	3
iLBC	8	13.3	20-30	No	2

Source: Summary of codecs. Web: http://www.voipforo.com/codec/codecs.php

As shown in Table codecs packaging have a period of 20 ms, so it is not possible to determine the best codec according to this parameter. For telephone networks, the MOS helps to consider the quality of service, it is for this reason that this parameter as the best codec used is G.711.

7. **Application Layer**: provides the interface and services that support applications. This is what is at the user's view.
Migration IP telephony software has been chosen, according to the IEEE 830 standard SRS software that meets the requirements is Elastix.

In summary, the design of migration will:

Figure 6. IP Telephony in the UTN within the OSI model.
Source: OSI Reference Model. Web: http://belarmino.galeon.com.

A. Calculating Bandwidth for VoIP call

Now, the bandwidth required for a VoIP call is calculated, the following steps are followed:

Step 1: Obtain the period codec packaging and bandwidth codec.
The information required codec, speed, bandwidth, packaging period is:

Nombre	Bit rate (kb/s)	Sampling rate (kHz)	Período de Empaquetamiento (ms)	Licencia	MOS[1] (Mean Opinion Score)
G.711	64	8	20-30	No	4.4

Figure 7. Features G.711
Fuente: Comparison of Codecs

Then packaging the period will be:

$$Período\ de\ empaquetamiento = 20ms$$
$$Ancho\ de\ Banda = 64kbps$$

Step 2: Get the necessary information from the link; if cRTP or IPSec is used and the size of the protocol header link layer is determined.

In the case of the UTN, no control of transport protocol used and not on the network has a configured tunnel protocol type. Then, is to determine the size of the header link layer protocol:

Table 3. Parameters of the link layer header.

Parámetro	Valor
Cabecera de capa 2	6 a 18 bytes
Cabeceras IP+UDP+RTP	40 bytes
Tamaño del payload	160 bytes para G.711

Source: Based on data link frame layer and the codec used.

Step 3: Calculate the period or packaging size.

To calculate the size of packaging in bytes per packet use the following formula.

$$Tamaño\ de\ empaquetamiento = \left(\frac{Período\ de\ empaquetamiento}{1000}\right) x \left(\frac{AB\ códec\ x\ 1000}{8}\right)$$
$$Tamaño\ de\ empaquetamiento = \left(\frac{20ms}{1000}\right) x \left(\frac{64kbps * 1000}{8}\right)$$
$$Tamaño\ del\ empaquetamiento = 160\ bytes$$

Equation 7. Formula for calculating the period or size of packaging
Source: Sizing IP telephone systems.

Step 4: Add the packaging size with the size of all headers and trailers.
To obtain total packet size is taken into account the link layer header and IP header but does not take into account the tunnel header, as the UTN not have IPsec or MPLS protocol, replacing values have:

$$Total\ del\ paquete = Cabecera\ Capa\ Enlace + (IP + UDP + RTP) + Payload$$

Equation 8. Sum of headers that determine the size of the package
Source: Sizing IP telephone systems.

$$Total\ del\ paquete = 6 + 40 + 160$$
$$Total\ del\ paquete = 206\ bytes$$

Now, the packet size calculated using the 802.1Q header:
802.1Q Ethernet overhead Trunk: 22 bytes

$$Tamaño\ del\ paquete = 22 + 40 + 160$$
$$Tamaño\ del\ paquete = 222\ bytes$$

The size of packaging G.711 codec is a high value but represents better voice quality, but remember that higher compression lower voice quality.

Step 5: Determine the speed of the pack in pps (packets per second), we use the following formula:

$$Velocidad\ del\ paquete\ [pps] = \frac{1}{(Período\ del\ empaquetamiento/1000)}$$

Equation 9. Speed packaging equation
Source: Sizing IP telephone systems.

Then the package speed is G.711:

$$Velocidad\ del\ paquete\ [pps] = \frac{1}{\frac{20ms}{1000}}$$
$$Velocidad\ del\ paquete = 50[pps]$$

Step 6: Calculate the total bandwidth
With data calculated above, the value of bandwidth is:

$$\frac{Tamaño\ total\ del\ paquete}{Tamaño\ del\ payload} = \frac{Requerimiento\ total\ de\ ancho\ de\ banda}{Requerimiento\ nominal\ de\ ancho\ de\ banda}$$

Equation 10. Requirement Formula AB
Source: Sizing IP telephone systems.

AB requirement for IP telephony system to be implemented on an Ethernet network with a header equal to 18 bytes:

$$Requerimiento\ de\ AB = \frac{206\ bytes * 64\ kbps}{160\ bytes}$$
$$\boldsymbol{Requerimiento\ de\ AB = 82.4\ kbps}$$

As the network of UTN, we have a configuration of VLANs, is also calculated requirement Ethernet header AB with 802.1Q Trunk equal to 22 bytes:

$$Requerimiento\ de\ AB = \frac{222\ bytes * 64\ kbps}{160\ bytes}$$

Step 7: Apply VAD (Voice Activity Detection) to detect silences:
Approximately one third of the average voice call is silent; for this reason should be applied VAD method to eliminate patterns of silence; which reduces the AB to 35% of what is actually required.

AB requirement for an Ethernet network using VAD:

$$Requerimiento\ de\ AB = 82.4\ kbps\ x\ 35\%$$
$$\boldsymbol{Requerimiento\ de\ AB = 53,56\ kbps}$$

AB requirement for a network with VLANs using VAD:

$$Requerimiento\ de\ AB = 88.8\ kbps\ x\ 35\%$$
$$\boldsymbol{Requerimiento\ de\ AB = 57,72\ kbps}$$

To size channels for VoIP, the VAD is not taken into account, especially in channels carrying less than 24 channels simultaneously, in the case of the UTN which has a capacity of 1 E1.

Now, it is also necessary to calculate the number of trunks required for a minimum GoS. Since this calculation is complex to do it mathematically using tables Erlang B and Erlang C is made, is commonly used Erlang B is blocking a call if all channels are busy.

With the values of the flow of traffic and the GoS.

$$A = 15{,}77\ Erlangs$$
$$GoS = 0{,}0182$$

The Grade of Service GoS has a value of 1%, meaning that for every 100 calls made, there is the probability of a crash and that of every 100 calls made, one is rejected when all channels are busy.

B. Calculating the number of trunks

For the number of trunks required for a VoIP system, we must consider the flow of traffic and the GoS.
Then:

$$A = 15{,}77\ Erlangs$$
$$GoS = 0{,}0182$$

Therefore, according to the table of Erlang loss probability B (See Figure 59.), with a GOS of 1% for a traffic flow of 15.77 Erl, 24 trunks are needed for sizing VoIP system.

Ch	*1%*	*2%*	*5%*	*Ch*	*1%*	*2%*	*5%*
7	2.50	2.94	3.74	**29**	19.49	21.04	23.83
8	3.12	3.63	4.54	**30**	20.34	21.93	24.80
9	3.78	4.34	5.37	**31**	21.19	22.83	25.77
10	4.46	5.08	6.21	**32**	22.04	23.73	26.75
11	5.16	5.84	7.07	**33**	22.90	24.63	27.72
12	5.87	6.61	7.95	**34**	23.77	25.53	28.70
13	6.60	7.40	8.83	**35**	24.63	26.44	29.68
14	7.35	8.20	9.72	**36**	25.51	27.34	30.66
15	8.10	9.00	10.63	**37**	26.38	28.25	31.64
16	8.87	9.82	11.54	**38**	27.25	29.16	32.62
17	9.65	10.65	12.46	**39**	28.13	30.08	33.61
18	10.43	11.49	13.39	**40**	29.01	31.00	34.60
19	11.23	12.33	14.31	**41**	29.89	31.92	35.58
20	12.03	13.18	15.25	**42**	30.77	32.83	36.57
21	12.83	14.03	16.19	**43**	31.66	33.76	37.56
22	13.65	14.90	17.13	**44**	32.54	34.68	38.56
23	14.47	15.76	18.08	**45**	33.43	35.60	39.55
24	15.29	16.63	19.03	**46**	34.32	36.53	40.55
25	16.12	17.50	19.99	**47**	35.21	37.46	41.54
26	16.95	18.38	20.94	**48**	36.11	38.39	42.54
27	17.97	19.26	21.90	**49**	37.00	39.32	43.53
28	18.64	20.15	22.86	**50**	37.90	40.25	44.53

Figure 8. Number of trunk.
Source: Network Planning. Web: welcome2igor.chat.ru/gsm_network_planning/gsm_network_planning.htm

C. General Technical Requirements for IP Telephony Network

GENERAL

- The core of IP voice must be under free software.
- The new VoIP system should be similar to the system currently operating.
- Should handle SIP protocol.
- Should initially be compatible with the H.323 protocol, so it can coexist with the current system of telephony in the UTN.
- The proposed solution must support calls to the PSTN and easy connection to the internal network.
- The central voice, must be capable of engaging in future redundancy various functionalities.

TECHNICAL

- Must initially support 200 extensions, with capacity for future growth.
- Requires a total of 24 SIP trunks.
- Support G.711 A-law.
- A DHCP server to assign IP addresses dynamically.

D. Hardware Requirements

Hardware requirements for IP telephony in the UTN, is based on the physical and network infrastructure that exists in the institution. Considering the use of the following equipment to perform the migration, which depends on the existing parameters in addition to future growth.

Given that migration must have the same characteristics of the system is in operation, is considered to comply with the following, so that there is harmony between what is working and what is proposed:

- The main authorities of the University, ie the Rector and Vice-Chancellors Administrative and Academic respectively, shall have a terminal computer support features video and conference rooms.
- For managers of each department, deans and faculty subdecanos each, a total of 24 phones is required, with fewer than phones leading authority features.
- For other extensions a SIP phone with basic features, that allow communication is recommended.
- For the operator, operator type requires a phone so that you can manage incoming calls to this extension.

Below is a table (see Table 4), a summary of IP telephony hardware required is as follows:

Table 4. Hardware IP Telephony.

HARDWARE PARA TELEFONÍA IP			
Nro.	**COMPONENTES**	**CANTIDAD**	**OBSERVACIONES**
1	Servidor VoIP Elastix	1	Servidor para comunicaciones
2	Teléfono Ejecutivo Tipo I	3	Equipo para autoridades principales
3	Teléfono Ejecutivo Tipo II	24	Equipo para directores de departamento, decanos y subdecanos
4	Teléfono Operadora	1	Teléfono para la extensión 7000
5	Teléfono Básico	150	Equipo para los usuarios

Source: Based on the current IP telephony system UTN.

E. Sizing Applications

To exploit the resources of Elastix, at the Technical University INSTALL applications focused on streamlining communications.

Voicemail

The voicemail is the voicemail service, it is used in case the extension you are calling is not answered, the caller can leave a message to the person I call. The voicemail must be configured on all extensions and each person may consult the messages that have dialing * 97.

Call Group

They are a set of extensions that are in the same group, configured so that when a call ringing at an extension, can be answered in another. And answer the call.

The UTN has been taken into account two agencies who need this service:

Library: In this dependence extensions that integrate the same group "Group 1", will;

Table 5. Call Group 1

DEPENDENCIA	EXTENSIÓN
JEFA DE BIBLIOTECA	7701
ANALISTA SISTEMAS BIBLIOTECA	7702
BIBLIOTECARIO	7703
PROCESOS TÉCNICOS BIBLIOTECA	7704

Source: Telephone Directory UTN. DDTI-UTN

Center for Continuing Education: The two extensions belonging to the "Group 2" will be;

Table 6. Call Group 2

DEPENDENCIA	EXTENSIÓN
COORDINACIÓN GENERAL CEC	7724
ASISTENTE CEC	7725

Source: Telephone Directory UTN. DDTI-UTN

- CID
  Call parking, call pickup
- Automatic transfer of calls, calls are routed to an extension without transfer.
- Music on Hold .wav or .mp3
- Queues
- Conferences

  Minimum 1 conference room, is considered a future growth determined by the network administrator.
  Minimal handling of 15 users per conference room.

F. Numbering Plan

For the numbering plan for the migration has taken into account the same numbering plan that exists in the UTN, since the directory keeps an established order and above users are already familiar with it.

Therefore, the directory is configured:

It consists of 4 digits;

The first is a general level: 7;

The second will be from dependence;

[0-1]: Administrative Central Plant
[2]: Faculty FICA
[3]: Faculty FICAYA
[4]: Faculty FFCCSS
[5]: Faculty FECYT
[6]: Faculty FACAE
[7-8]: Administrative buildings CAI Programs, Library, College.

The third and fourth numbers, is set in numerical order for each unit.

G. Dial Plan

In the dial plan, there are restrictions for each group of officials from the UTN.

The permissions are set so that entities can save resources.

According to Table 7. Dial Plan UTN, calls are restricted as follows:

- Operator: Make internal, local and national calls.
- Authorities: They have call barring.
- Department Directors: access to internal, local and national calls.
- Secretaries and Officers: Have access to internal calls.

Table 7. Dial Plan UTN

	GRUPOS				
LLAMADAS	Operador	Autoridades	Directores de Departamento	Secretarias	Funcionarios
Internas	✓	✓	✓	✓	✓
Llamadas 1700	✓				
Llamadas 1800	✓	✓	✓	✓	✓
Locales	✓	✓	✓	✓	✓
Nacionales	✓	✓	✓		
Internacionales		✓			
Celulares		✓			
APLICACIONES					
Interceptar llamadas	✓	✓	✓	✓	✓
Voicemail	✓	✓	✓	✓	✓
Grupos de llamada					✓
Salas de Conferencia		✓	✓	✓	✓
IVR	✓	✓	✓	✓	✓

Source: Telephone Directory UTN. DDTI-UTN

VI. CONCLUSIONS

With the migration of IP telephony in a Cisco proprietary platform at a low Free Software, the University will have a voice service, updated and, from it, may develop other projects that are aimed at the integration of services focused unified communications.

With the implementation of an IP telephony has no licensing cost, the UTN be reducing acquisition costs licenses restricting its growth and service flexibility.

When working with the H.323 protocol, a complex and rigid model, especially used by equipment manufacturers (owners), charge is generated in the processor, thus the time will also increase, sometimes causing data network consume more resources which usually does.

A protocol fully digital signage as it is SIP, is based on IP addresses, which when using the Data not have to make an analog-digital conversion, giving the network a saving in processing times and ease of scalability and interoperability.

To determine operating parameters, such as number of concurrent calls and rush hour in a voice server, a software monitoring traffic is used.

The implementation of a new system of IP telephony allows the institution to think of new communication services, unifying software and services that are handled independently, but together can form a unified communications platform and very robust applicability to workers administrative, faculty and students of the UTN.

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