STUDY, DESIGN AND IMPLEMENTATION OF A PROTOTYPE OF A CENTRAL IP VIDEO TELEPHONY (INTERNET PROTOCOL), UNDER A GNU / LINUX PLATFORM FOR THE ORGANIZATION "LIEBENZELL MISSION OF ECUADOR"

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Abstract **-** The objective of this project is to study, design and implement a central video telephony with free software for the organization Liebenzell Mission of Ecuador, to meet the need for internal communication. The study is conducted of the main characteristics of digital transmission, telephone networks, the TCP / IP, VoIP and video telephony, with an introduction to the operating system GNU / Linux, and your phone working tool called Asterisk is the main basis for the design and subsequent implementation of the core IP telephony and video telephony.

Indices of new words **-** GNU, Linux, IP.

I. INTRODUCTION

The constant technological advances especially in the field of communication networks forced to remain updated regarding the various options that can meet the needs of communication. However to focus on the latest technology, it is necessary to conduct a study of the protocols involved in IP telephony.

But first a brief detail of traditional telephony, the first networks were known as the voice circuit-switched networks, the operating principle is that a circuit is formed between the two ends of the communication and the elements of pass circuit connected across the duration of the connection, in Figure 1, we can observe typical circuit switched telephony end to end.

Figure 1. Switched telephone network circuits.

Switched networks are divided into circuit switched networks, packet switched networks and photonic switching.

A. CIRCUIT SWITCHED NETWORK.

The circuit switched network establishes a path through the nodes of the network dedicated to the interconnection of two stations. At each link, a logical channel is dedicated to each connection. Data is transmitted as fast as possible, at each node, incoming data is routed through the dedicated channel without undergoing delays, the result is basically equivalent to physically connect a pair of cables one end to the other as shown in Figure 2.

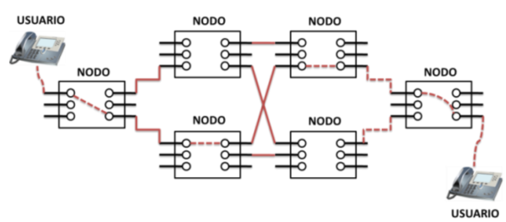


Figure 2. Representation of a circuit switched network between two users.

The circuit networks are very expensive because they require a dedicated circuit for each of the subscribers. Circuit switching should not be confused with analogue networks, because, in circuit networks, can transmit data in digital and analogue is not possible.

B. PACKET SWITCHED NETWORK.

The data transmission is now paramount, the circuit switched network is not suitable for data transmission due to their regular transmission lines are busy, even when no information traveling within the network. The solution to these problems is packet switched networks, as the name implies, data is transmitted in packets, in addition to the transmission of larger data sets, the sender divides the data into smaller groups and additional packages a number of control bits in each of the intermediate nodes packets are received and stored for some time and re-transmitted up to the place of destination, in Figure 3, there is shown a packet network with different nodes and a communication between two users.

Figure 3. Packet switching network between users.

Packet networks have the advantage of optimizing resources, because the same medium can be used to send multiple packets.

II. STUDY

A. Voice over Internet Protocol (VoIP)

IP telephony, which is better known as Voice over IP, VoIP and VoIP (for its acronym in English), integrates voice and data, enabling a voice signal to travel through the Internet using an IP protocol (Protocol Internet).

DIFFERENCES BETWEEN TRADITIONAL TELEPHONY AND IP

Telephony in traditional telephony, the telephone establishes a connection between the two participants indestructible, this connection is used to carry voice signals.

While the IP telephony, the digitized voice is compressed and conveyed through the IP network to a recipient that contains an elementary process.

History and evolution:
• VoIP started in Israel in 1995. (Communication from PC to PC).
• Vocaltec, Inc. Launches First softphone ("Internet Phone Software").
• In 1997, Jeff Pulver at VON together (fair / congress).
• In 1998 appeared the first gave VoIP ATA / gateways.
• During 1998, manufactures switches with Layer 3 QoS.
• In 1999 sold his first Cisco VoIP platforms, with the H323 signaling protocol.
• VoIP in 2000 represented more than 3% of voice traffic. And make Asterisk.
• In 2002, the SIP protocol begins moving to H323.
• In 2003, Skype appears.
• Skype is bought in 2005 at $ 2.6 billion by eBay.

D. IP PROTOCOL.

The Internet Protocol (IP) is the method or protocol by which data is sent from one computer to another over the Internet. Each computer (known as a host) on the Internet has at least one IP address that identifies it from other computers on the Internet.

When you send or receive data (for example, a note by email or a Web page), the message is divided into small sections called packets. Each of these packages contains the Internet address of the sender and receiver address. Any packet that is sent first to a gateway computer comprising a small part of the Internet. The gateway computer reads the destination address and forwards the packet to an adjacent gateway that in turn reads the destination address and so forth across the Internet until one gateway recognizes the packet as belonging to a computer within your immediate neighborhood or domain. That gateway then forwards the packet directly to the computer whose address is specified.

E. SIP.

The Session Initiation Protocol (SIP: Session Initiation Protocol), is developed by the IETF, protocol is a client-server. This means that the client generates requests sent to the server that processes them, for the particular case of IP telephony and video telephony, may include protocols such as SIP, TCP / UDP, RTP and SDP (Landivar, 2009).

The media information (audio, video or data) and reception capacities are transported in the cargo (payload) of SIP messages. SDP serves this purpose, establishing a list of audio and video capabilities and indicating where to send the data (Landivar, 2009).

SIP defines several methods:

- **TO INVITE**: It's used to invite a user to a call and establish a new connection.
- **TO SAY GOOD BYE**: It finishes a connection between two users on a call.
- **OPTIONS**: It requests information about the capabilities of users, without establishing a connection.
- **STATE**: It informs another server about the progress of actions signaling required.
- **ACK**: It's used for reliable exchange of requests / responses.
- **CANCEL**: It finishes the search for a user.
- **REGISTRATION**: It delivers to a SIP server information about a user's location.

Below in Figure 4 shows the message exchange process.

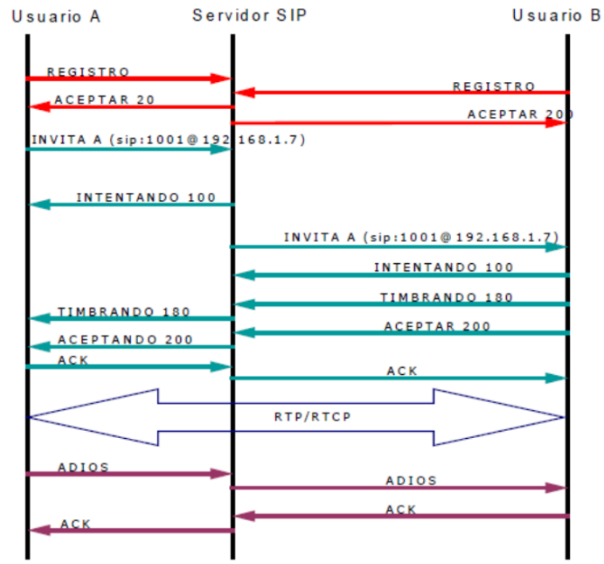


Figure 4. Exchange process SIP messages

F. PROTOCOL RTP.

RTP known as point-to-point protocol, the network transport functions suitable for applications transmitting real-time data such as audio and video data. RTP does not address resource reservation and does not guarantee quality in addition to real-time service. To ensure data transport or to be a control protocol (RTCP) to allow monitoring of the data delivery in a manner scalable to large multicast networks, and to provide minimal control and identification functionality. RTP and RTCP are designed to be independent of the underlying transport and network layers. The protocol supports the use of RTP-level translators and mixers.

G. VIDEO OVER IP.

Video over IP can be used for a variety of purposes, including downloading, streaming, surveillance via closed-circuit television, video conferencing and video telephony. Transferring video over IP networks is a challenge in terms of bandwidth and quality of service, but the modern compression technologies and bandwidth increasingly more feasible the use of video over IP networks.

H. VIDEO AND AUDIO CODECS.

The different codecs detonate in Tables 1 and 2 below:

Información del Codec			Cálculos de ancho de Banda			
Codec & Bit Rate (Kbps)	Códec Tamaño de la Muestra (Bytes)	Códec Intervalo de la Muestra (ms)	Tamaño de la carga Útil (Bytes)	Tamaño de la carga Útil de voz (ms)	Paquetes por Segundo (PPS)	Ancho de banda Ethernet (Kbps)
G.711 (64 Kbps)	80 Bytes	10 ms	160 Bytes	20 ms	50	87.2 Kbps
G.729 (8 Kbps)	10 Bytes	10 ms	20 Bytes	20 ms	50	31.2 Kbps
G.723.1 (6.3 Kbps)	24 Bytes	30 ms	24 Bytes	30 ms	33.3	21.9 Kbps
G.723.1 (5.3 Kbps)	20 Bytes	30 ms	20 Bytes	30 ms	33.3	20.8 Kbps
G.726 (32 Kbps)	20 Bytes	5 ms	80 Bytes	20 ms	50	55.2 Kbps
G.726 (24 Kbps)	15 Bytes	5 ms	60 Bytes	20 ms	50	47.2 Kbps
G.728 (16 Kbps)	10 Bytes	5 ms	60 Bytes	30 ms	33.3	31.5 Kbps
G722_64k(64 Kbps)	80 Bytes	10 ms	160 Bytes	20 ms	50	87.2 Kbps
ilbc_mode_20(15.2Kbps)	38 Bytes	20 ms	38 Bytes	20 ms	50	38.4Kbps
ilbc_mode_30(13.33Kbps)	50 Bytes	30 ms	50 Bytes	30 ms	33.3	28.8 Kbps
GSM (13.3 Kbps)	33 Bytes	20	33 Bytes	20	50	36.4 Kbps

Table 1. Different audio codecs used for VoIP.

Códec de Información			Los cálculos de ancho de banda		
Códec & Bit Rate (Kbps)	Códec Tamaño de la muestra (Bytes)	Códec intervalo de muestreo (ms)	Tamaño de carga útil de video (Bytes)	Tamaño de carga útil video (PPS)	Ancho de banda de Ethernet (Kbps)
H.261 (64 Kbps)	De 40 Bytes hasta 2 Kbps	20ms	200 Bytes	50	103 Kbps
H.263 (8 Kbps)	Menor a 64 hasta 512 Kbps	20ms	230 Bytes	50	115.2 Kbps
H.264 (64 Kbps)	Menor a 64 Kbps hasta 960 Kbps	20ms	**262 Bytes**	50	**128 Kbps**

Table 2. Different video codecs used for VoIP.

I. ASTERISK.

Asterisk allows real-time connectivity between the PSTN and VoIP networks, but to connect to the PSTN must add the corresponding dedicated peripheral Asterisk includes several features that were previously available only in proprietary systems, which were often expensive. Some of the applications that allow Asterisk, they're creating extensions, voice messaging to email, conference calling and video conferencing, interactive voice menus, automatic call distribution, among others.

Additionally you can create new functions by the language of Asterisk modules written in C or by correctly programming in Perl and other languages. Some software applications that provides Asterisk, when connecting incoming call with outgoing calls or other asterisk users are voicemail, "Meet me" (conference), among others. To view the applications available in Asterisk use the command "show applications" at the interface of the command line asterisk.

III. DESIGN

A. LIEBENZELL MISSION ORGANIZATION OF ECUADOR.

The organization "LME", was founded on August 7, 1992, in Ibarra city, it is part of the Liebenzell International Mission from Germany, it is a nonprofit organization. It is located in "Luis Felipe Borja" and "Fray Vacas Galindo" 9-111 streets.

Figure 13 shows the physical infrastructure that the organization currently has LME, as shown in Figure 5.

Figure 5. Building of the organization "MLE", located in Ibarra city.

The organization "LME" currently does not have an internal communication system, and this, coupled with its high propensity to growth is essential to the implementation of a communication system that is sufficiently flexible, scalable, reliable and easy to use, in order to solve the user's needs. Also because it is a non-profit organization is difficult to access a telephone because of its price, payments implementation among others aspects. It also will make the analysis of telephone service available to the organization "LME". This analysis will solve the users' needs and establishes whether those needs will be addressed with IP video telephony.

B. DESIGN OF A PROTOTYPE OF A CENTRAL IP VIDEO TELEPHONY, UNDER GNU LINUX PLATFORM.

To be able to design the video-telephony central prototype, the researcher considered the need for communication within the organization "LME", the study of the current situation and network components are in the organization, because it is a prototype of a central part of video-telephony under free software, that's why it is given the name of "SERVIMAATEL".

In "SERVIMAATEL" the main part is free software because you can modify it according to the requirements and needs, enabling the design of a flexible central video-telephony modifications, updates and the ability to add new functions that may appear in the future. The installation, implementation and configuration of the software on the prototype of the central IP video telephony, should allow to manage all software, interfaces and applications. The operating system will be free software only, users must allow the use of other applications that must necessarily be free. The software must work with whatever type of processor, whether this Intel or AMD, Figure 6 shows a block diagram of how the video-telephone system is composed.

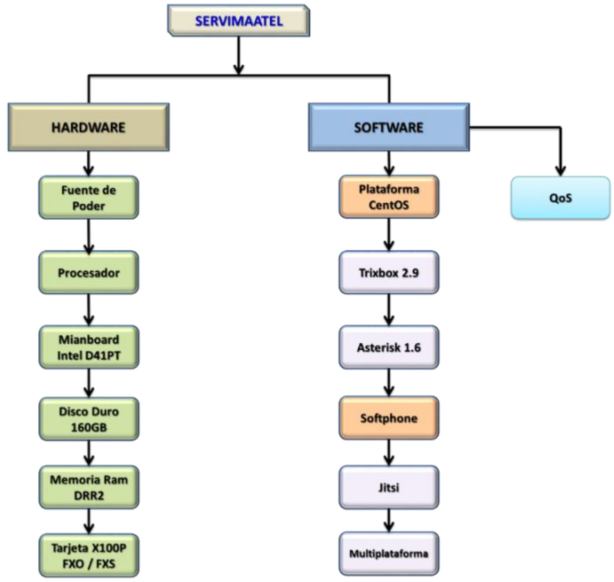


Figure 6. Block diagram of the power of video telephony for "MLE"

B. TRIXBOX.

It is an open software that has similar characteristics to the Elastix platform with several benefits which are below (FONALTY, 2011):

- Answering Machine (IVR).
- Multi-language support.
- Messaging support.
- Integration with Outlook.
- Voicemail.
- Voice Mail to e-mail.
- Analogic phones and IP.
- Web Control Panel.
- Reporting and monitoring.
- Conference bridges.
- Video support.
- "Openfire" instant messaging server.
- Support external messaging, Hotmail, Yahoo, Gmail and other features.

C. CODEC FOR SERVIMAATEL.

For this case the codecs used are GSM and H264, to use each of the protocols for IP telephony. The protocols permit to implement a telephone with low consumption of network resources when transporting audio and video.

E. QOS IN SERVIMAATEL.

QoS is the ability to treat packets that are transmitted differently from one network device based on the contents of the package. The QoS configuration performs different tasks based on traffic direction and the device location. QoS works in the access layer, wherein the IP packet first enters the network, the QoS policy is to classify and mark each package. This kind of policy is applied in the incoming direction of the access layer in the user interface.

To implement the QoS becomes SERVIMAATEL use zeroshell Linux implementation.

Zeroshell QoS (Quality of Service) supports the traffic management to control traffic in a congested network. It has the ability to guarantee minimum bandwidth, limit the max bandwidth and assign a priority to a traffic class, this will be very useful for the video telephony central. This module allows the setting interfaces Ethernet, Virtual Private Nets, bridges and VPN links. It also provides the ability to classify traffic using filters allowing L7 deep packet inspection, usefully managing applications and P2P VoIP, a call that is guaranteed by means of packets prioritization, as shown in Table 3.

Clases QoS	Protocolos	Prioridad	Garantizado	Máximo
VOIP	SIP	Alta	192 Kbps	256Kbps
P2P	eMule, EDonkey, KaZaA, Gnutella, BitTorrent, Direct Connect, Youtobe, entre ootros.	Baja	128 Kbps	192Kbips
SHELL	ssh, telnet	Alta		
BULK	ftp, smtp	Baja		
DEFAULT	No clasificado	Media		

Table 3. Packet prioritization using Zeroshell.

To properly implement the QoS policies it is necessary to note that when a class of QoS is applied to a network card, you control the outgoing traffic of the card and also its entry as shown in Figure 7.

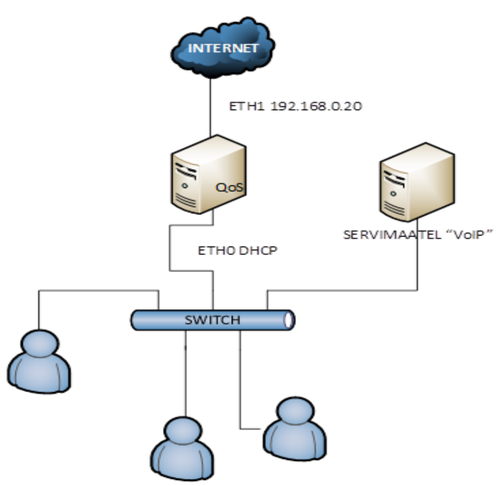
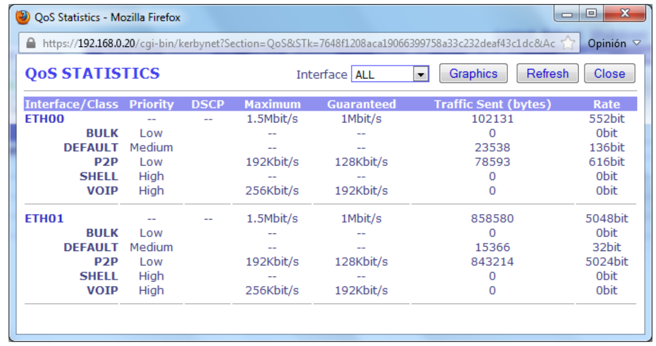


Figure 7. Diagram of how traffic is handled with two Ethernet cards.

For each network card on which QoS is enabled, you can view the associated QoS classes and each class is setup (priority, maximum bandwidth and guaranteed bandwidth) and the number of bytes that are sent out class and type, in other words, the number of bits that are transmitted per second, as it is seen in Figure 8.


QoS Statistics - Mozilla Firefox

https://192.168.0.20/cgi-bin/kerbynet?Section=QoS&STk=7648f1208aca19066399758a33c232deaf43c1dc&Ac

QoS STATISTICS — Interface ALL — Graphics — Refresh — Close

Interface/Class	Priority	DSCP	Maximum	Guaranteed	Traffic Sent (bytes)	Rate
ETH00	--	--	1.5Mbit/s	1Mbit/s	102131	552bit
BULK	Low		--	--	0	0bit
DEFAULT	Medium		--	--	23538	136bit
P2P	Low		192Kbit/s	128Kbit/s	78593	616bit
SHELL	High		--	--	0	0bit
VOIP	High		256Kbit/s	192Kbit/s	0	0bit
ETH01	--	--	1.5Mbit/s	1Mbit/s	858580	5048bit
BULK	Low		--	--	0	0bit
DEFAULT	Medium		--	--	15366	32bit
P2P	Low		192Kbit/s	128Kbit/s	843214	5024bit
SHELL	High		--	--	0	0bit
VOIP	High		256Kbit/s	192Kbit/s	0	0bit

Figure 8. Number of bits transmitted in a class that apply QoS.

IV. IMPLEMENTATION

A. IMPLEMENTATION

After completing the design of the prototype, and the necessary configuration, appropriate and optimal operation. The "SERVIMAATEL" is ready to be deployed in the organization "LME". At the time of implementation there were drawbacks because it took into account the technical parameters available to the organization, in Figure 48, we observe the physical space in which stood the central prototype video telephony, and service being installed users of the organization Liebenzell Mission of Ecuador, as shown in Figure 9.

Figure 9. Implementation of central video-telephony in the "LME"

With the implementation of the prototype, the next chapter will be tested for operation and correct any errors that may occur.

B. CONSIDERATIONS.

Throughout the development of this project, it was proved several times the software versions, because sometimes presented problems such as recognition of hardware devices for testing various settings, modifications to files several times, damaging the operating system Linux. But at the end of this project was achieved successfully.

To measure the quality of service from the central video telephony, we used the wireshark program, consisting of practical tools to measure the traffic carried by a data network. Also checked the SIP message sequence, in establishing a successful call, you can appreciate the codecs that are proposed for GSM audio codec and video codec H264.

V. CONCLUSIONS

The realization of this project was the need to study, design and implement a central video telephony under a Linux platform for the organization "LME", using H.323, SIP and Video4Linux support, with this you can have a robust and current, providing most VoIP services licensed systems.

The study and design made enables convergence of services for new and existing telecommunications to the organization "MLE" of Ecuador to benefit from the new technologies of communication using IP protocol, this protocol can unify voice applications, video and data to ensure the transmission service used to transport SIP protocol which can work on IP-based networks.

The use of free software, free does not mean it is freely distributed. This means that open source is in the main pages of free software available on the web, open source allows different users to use this code and modify it to the needs and enhance it for future applications in this case for telephony and video telephony for organization "MLE".

Using the platform Trixbox 2.9 for IP telephony is the basis for plant design video telephony, is software that contains as core Asterisk 1.6, which does the work of a PBX that allows its administration so friendly to the user, supports various audio and video codecs, and is used for softphone calls Jitsi telephony and IP video telephony, allowing easy handling, also use the module to zeroshell applied to QoS.

According to the requirements of the organization "MLE" and the current situation, the central video telephony is configured to provide different services and be compatible with IPv6, which is the protocol that is being implemented.

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He is a professional in Electronic Engineering and Telecommunications. He is currently Professor of the Faculty of Engineering in Applied Sciences at "Universidad Técnica del Norte" in areas such as: Electronic circuits, Physics, and others related fields.

He has experience in areas such as: Preventive and corrective maintenance of data transmission equipment. Installation and maintenance of WLAN networks.

Through his service, he has worked consistently and unconditionally in the development of electronics and telecommunications projects.

M. Aguas, Author

He is a graduate student of the School of Electrical Engineering and Communication Networks at "Universidad Técnica del Norte". He is currently manager of SERVIMAAT company, he is a project designer of data network and he's head director of technical support area.

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Miguel Aguas has provided technical support to several companies in Ibarra city, he has being recognized as one of the best professionals.