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Voice Over IP Using Distributed Architecture For Packet Transmission And Reception Applications

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ABSTRACT
IP telephony is an alternative over traditional telephony due to the reduction of costs in its calls, so the bits transmitted between extensions using multiVoIP devices (MVP) is 96 bits and the transmission of two extensions (terminal A to B) using Two MVP is 292 bits per call, thus giving a 67.1% saving in data transmission in each call under IP. The telephone communication with IP technologies uses oriented and non-communication oriented protocols, since with TPC / IP 40 packets are transmitted, giving 10.5% of the communication between two extensions; while UDP (non-communication oriented) occupies 357 packets, this is equivalent to 89.5% of the communication, thus asserting that most of the packets used are UDP, so communication in low IP calls does not have a large percentage of security in the transmitted data.

Keywords - Networks, Voice over IP, Network Standards, Protocols, VoIP, H.323.

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I. INTRODUCTION

The evolution of communications, as well as the impact of technology around the world, has allowed to reduce the digital divide in the most vulnerable sectors, achieving substantial changes in society, achieving constant innovation in technological applications and the passage of Traditional telephone communication to a more economical communication and within the reach of the user in general, [1].

According to [2], communication networks are made up of several computers and heterogeneous operating systems which are connected to the internet.

These communication networks have allowed the generation of new innovation processes in telephony and have generated communications called Voice over IP or VoIP, which is a new technology that allows the transmission of audio fragments, digitized through the Internet, so it can be described that the transfer of Voice over the network infrastructure has generated great expectations, due to the saving of resources that this technology represents, which is why when using the data network to generate new applications such as voice, data and video it generates a substantial saving of resources and at the same time facilitates the use of the network infrastructure, thus achieving a better performance of the processes in the network and the use of all its resources and applications, [3].

The standards used in VoIP were determined by the ITU-T to establish a common basis for open communication in all manufacturers, for which VoIP works with standards such as H.323, H.320, etc., which establish a communication mechanism and implementation of voice over IP networks, [4].

It is important to describe that IP networks have allowed the entry of new protocols and communication standards, so it can be said that the solutions of first instance were based on own or proprietary technologies; since these technologies did not allow communication between systems of different manufacturers and was called closed architecture, VoIP therefore allows integrating different telephone platforms that exist in public and private institutions, thus achieving open and scalable communication [5].

For all the above, it is important to emphasize that Voice over IP has four important elements that are client, server, gateway and final equipment.

Client: it is presented as an application through software running on a PC controlled through a graphical interface (GUI); as well as a "virtual" client that resides in the Gateway and have the following characteristics:

- Transmission: Establishes communication, codifies, packages and transmits to the means of propagation.
- Reception: Perform the reverse process. Decodes and reproduces the voice signal through the headphones or speakers.

Server. The servers perform several processes such as complex database operations, domain management, proxy servers, etc., these operations are performed in real time and locally.

Gateways. They are devices that make a bridge of communication between users, which allow joining the VoIP network with the analogue telephone network PSTN (Public Switched Telephone Network).

Telephones: These are final devices called Normal telephones and IP telephones.

In terms of VoIP standards, H.323 allows multimedia communications and guarantees a quality of service (QoS), combined with adapters, gateways and other infrastructure products, allowing universal connectivity within and outside the scope of the same institution and is composed of: Endpoint, Gatekeeper (GK), Gateway (GW), Multipoint Control Unit (MCU), Terminal and Multipoint Processor (MP), [6].

An important point in communications is the quality of service (QoS), which is defined as the capacity of the network to manage the traffic that is generated in the transmission and reception of data, which allows satisfying the needs of the user in the transmit voice over IP, which improves network performance and ensures communication, [7] [8].

II. METHODS

The present article was carried out through research on VoIP technology, local area networks, voice over IP applications, communication standards, etc. This information provided necessary information on the current trends in VoIP technology and its standards, as well as the configuration mode of a Work Station in the form of a router.

For the processes of transmission and communication between the client and the server, as well as a communication with quality of service and security in the data network, two Multitech devices were configured, among which the MVP 400 was used as server and the MVP team 200 as a customer. The Multitech equipment was configured with the H323 Standard for the use of all the applications that the devices allow, since by default they are structured with proprietary standards, which have limitations in IP applications.

A Work Station was configured to have benefits of a router in order to allow the transmission of information in the network. Said station is configured with two Fast Ethernet cards, which allow the passage of two network segments, one server gateway and the other of the client in said station, the Ethereal is also configured that allows the monitoring of packet filtering.

The MVP teams were configured to perform multi-calls and use the fax service using the VOIP network, they were also configured to allow voice transmission under the IP network with normal telephones, these will allow to use all the convergent services of the network and the lower cost of costs for the applications of each team.

III. RESULTS

As a result of the investigation, the different configurations and applications of the MultiVoIP equipment (Multitech) are described below:

1. Configuration of the MultiVOIP MVP400 and MVP200 (Host).

To configure the MultiVOIP equipment, a local network connection was made, which allowed the equipment to be configured through a console (TELNET), it is important to emphasize that the devices can also be configured graphically. To generate communication between the MVP and the team, IP addresses with their respective Mask were assigned to the Multitech devices, as well as to the base station for communication, thus ensuring that all the equipment is in the same network, which will allow access to the configuration of the aforementioned devices.

It is important to configure the communication standard, which is why the H.323 standard is configured, which defines Communications Systems as packet-based multimedia transmission, so that this standard allows having a distributed architecture, which generates applications in a multimedia, as well as VoIP communication.

The configuration of the MVP 200 as host consists of installing the Oem, Coders, H323 Stack, Factory defaults and Firmware. It is important to describe that at the time of adding extensions to this equipment it is necessary to take into account the configuration as Host and activate the Owner Phone Book, which will allow entering the number for communication.

2. Monitoring and filtering of packages using ethereal.

In Fig. 1, a star topology is shown, which will allow the monitoring and filtering of packets, the local network will be in a single Broadcast domain in all the ports and will be connected to the MultiVoIP equipment and the Work Station that it contains the Fedora Core 4.0 Operating System. The server equipment has two interfaces (eth0 and eth1) to perform the function of a router and thus be able to receive and send packets from different networks (LAN, MAN or WAN).

The Work Station should be configured as a router and with software for the monitoring and filtering of packages such as the Ethereal that when using it will obtain the existing traffic in the network when a telephone call is being made from the MultiVoIP equipment.

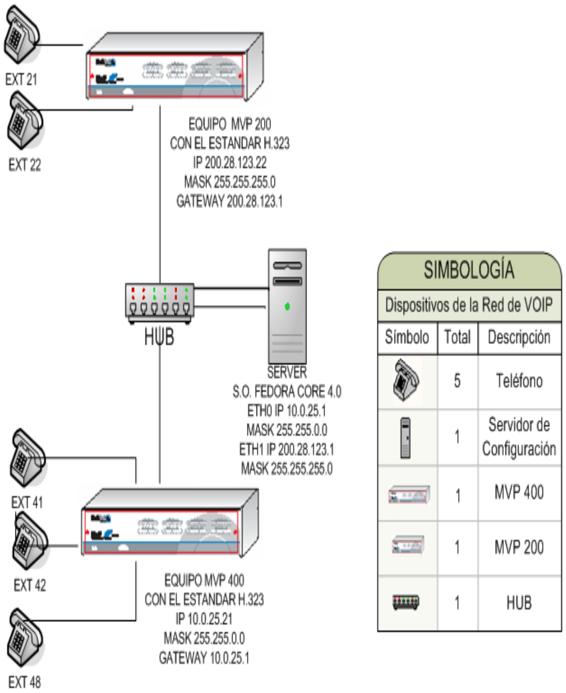

Fig. 1 Network design for monitoring and filtering packages

Once the installation of the Fedora Core 4.0 Operating System and the Windows Operating System to configure the MultiVoIP equipment have been made, the following requirements must be used:
Server.

- eth0 Link encap: Ethernet HWaddr 00: 20: 78: 15: 6B: D6
- inet addr: 10.0.25.1 Bcast: 10.0.255.255 Mask: 255.255.0.0
- eth1 Link encap: Ethernet HWaddr 00: 90: 27: 1B: 09: ED
- inet addr: 200.28.123.1 Bcast: 200.28.123.255 Mask: 255.255.255.0

Multitech equipment.

- MVP 200: IP: 200.28.123.22 Mask: 255.255.255.0, Gateway 200.28.123.1
- MVP 400: IP: 10.0.25.21 Mask: 255.255.0.0, Gateway 10.0.25.1

3. Flames, Multi-calls and faxing with MVP - VoIP equipment.

For calls, communication and configuration with VoIP equipment, 4 analog telephones, 2 faxes and two network cards were used for monitoring. All this implementation has been done in a star network topology; The Fedora Core 4.0 Operating System has also been installed in a Work Station, to obtain TCP / IP packet monitoring and filtering to visualize the information that is being transmitted in VoIP communication.

In the Fig. 2 shows a network topology in which there are analog telephones connected to the equipment (MVP 200 and 400), these in turn connected by a switch or hub to a Work Station to perform the necessary tests in the communication with the H.323 standard.

To perform the tests on calls and check their effectiveness on the equipment, clear communication must be obtained through the MVP teams, for which the following steps are described:
User A calling User B.
• User A dials the required user B extension number.
• The extension called rings.
• The called subscriber answers.
• The connection is established and later to end the call, close the telephones.
User redial (A-B).
• User A to dial the required user B extension number.
• The extension called rings.
• The called subscriber does not answer.
• User A closes the phone.
• To re-call user B, press the radial.
• The extension called rings.
• Finally the connection is established.

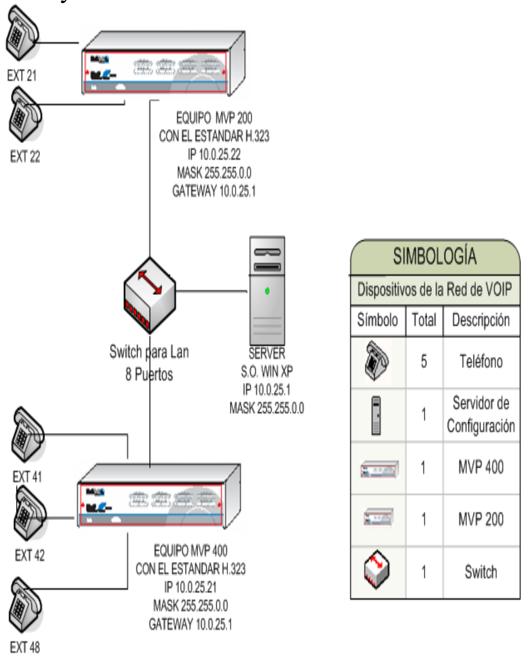

Fig. 2 Network topology with MVP equipment

To perform the multi-calls the teams must be configured in one of the two existing ways:

- If they are in the same network, one will be configured as another HOST as CUSTOMER.
- While, if they are in two different networks, the two teams should be established as individual HOSTs so that each one works with its own telephone directory and allows to correctly route the packets, in order to have an excellent communication.

For multi-calls, the telephones must be connected to all MVP FXS ports, as shown in Figure 2, this will allow calls and Fax to be sent simultaneously, as well as checking their effectiveness.

Fig. 3 shows the connections of a star topology between zone 1 and zone 2 with a series of telephones using all the FXS ports of the MVP equipment.

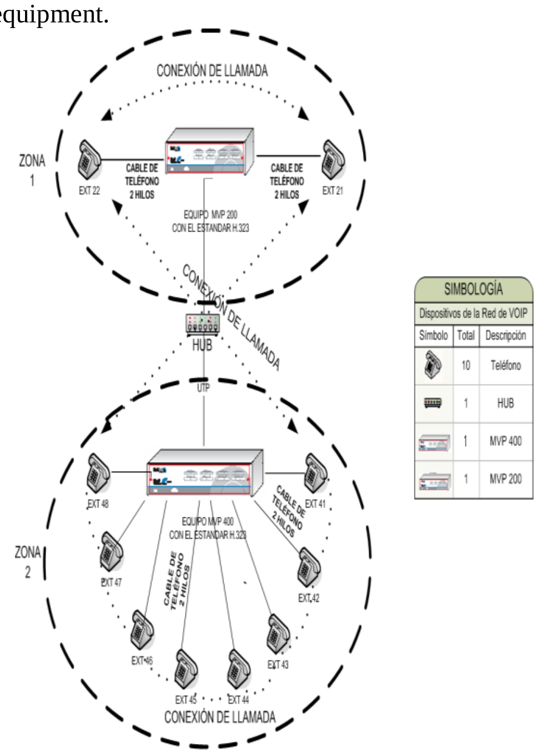

Fig. 3. Multicamera with MVP equipment

It is important to emphasize that the fidelity of communication in calls can be limited by the characteristics of the equipment and the LAN network.

For the Fax Configuration you must enter the parameters described below:

1. Select Channel 1
2. Enter the maximum bandwidth that the equipment supports
3. Maximum duration that the fax will have
4. Connect the Fax equipment in the FXS ports, in the MVP 400 equipment in the channel 1 that is configured as (extension 41), and in the MVP 200 in the channel 1 (extension 21)
5. Make the normal telephone call from extension 40 to extension 21.
6. After having connection, ask for fax tone and close the phones
7. Allow the Fax teams to process their information so that later the MVP teams do their work with the T.38 protocol.
8. Finally wait for the Fax document to reach its destination and receive the delivery acknowledgment in the Fax connected to port 1 of the MVP as Extension 40.

Regarding the sending and receiving of Fax, in Fig. 4, the sending of User Fax A - B is displayed, which are transmitting information using a single equipment (MVP200), as well as in the transmission of the two MVP (200 and 400), which are described below:

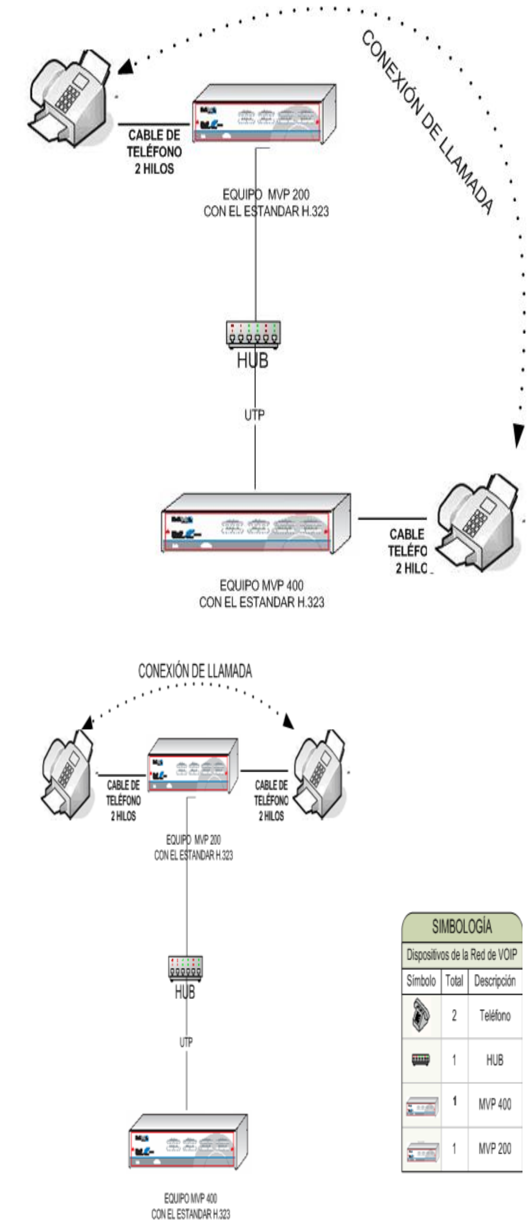

Fig. 4. Faxing with MVP devices

IV. CONCLUSIONS

- Possessing proprietary technologies and standards restricts the field of communications, since there is a danger that the devices will be isolated, this depends directly on customers and suppliers, so that equipment with the H.323 standard allows interoperability between the different manufacturers, in fact, companies like Cisco and other global operators have added to their different marketing products.
- IP telephony is an alternative over traditional telephony due to the reduction of costs involved in calls, as well as the quality of services (QoS) in its communications and the encryption of the information obtained, so that communication under IP 50% decrease in call costs, as well as the infrastructure of conventional telephone networks.
- The bits that are transmitted in the communication between phones using a single MVP is 96 bits; while the transmission from terminal A to terminal B, using the MVP 400 equipment and the MVP 200, is 292 bits per call, thus giving a 67.1% savings in data transmission in each call under IP.
- Regarding the communication protocols in each call, the TPC / IP protocol transmits 40 packets, thus giving 10.5% of the communication; while UDP occupies 357 packets, this is equivalent to 89.5% of the communication, this confirms that most of the packets used are non-communication oriented protocols (UDP), so communication in low IP calls does not have a large percentage of security in the transmitted data.

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