MIGRATION SYSTEM PROTOCOL TELEPHONY IP IPv4 to IPv6 IN THE UNIVERSIDAD TÉCNICA DEL NORTE

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Abstract— The objective of this project is to migrate the IP telephony system from IPv4 to Ipv6 protocol at the Universidad Técnica del Norte. The justification of the project was carried out and the need for the implementation of IPv6 data networks, also some benefits of the implementation of this protocol. The theoretical foundation was developed by gathering information from books, journals, theses, articles and websites related to IPv6 and VoIPv6. Gathering information on the current state of the network was held, equipment and devices involved in the system, configurations, physical and logical topologies. For the design, ipv6 routing table was made, to then be configured in the system elements. Dual Stack was used as a method of coexistence between IPv4 and IPv6 on the system equipment. After testing operation, the final implementation was done in the system, the telephone system migrating to IPv6 protocol. Finally, conclusions and recommendations product research and work done are written..

Index of terms—Dual Stack, IPv4, IPv6, SIP, VTP, VLAN, VoIP, PSTN.

I. INTRODUCTION

The depletion of IPv4 addresses, the emergence of new devices that require an IP addressing to interact with the internet, the arrival of the internet of things globally, the growing transition from IPv4 to IPv6 protocol in major telecommunications systems worldwide, they are a reality that they see the importance and necessity of IPv6 today.

Most colleges in Ecuador, as well as investigation centers have migrated or are in a process of transition from IPv4 to IPv6 in their services. The Universidad Técnica del Norte (UTN), has extensive telecommunications infrastructure which is governed by the IPv4 protocol, also, has assigned an IPv6 address range by the Consorcio Ecuatoriano para el Desarrollo de Internet Avanzado (CEDIA), to be implemented in their systems and thus the university can make the transition from their services from IPv4 to IPv6.

To meet the objective of migrating telecommunications services at the University of progressive and properly, it arises migrate the IP telephone system of the University of IPv4 to IPv6, thus take part in the transition to IPv6 throughout the university. This will allow the university to be counted among the institutions that are in the process of transition to the new IPv6 protocol and comply with the commitment to innovate and be at the forefront of new technologies.

II. THEORETICAL ARGUMENT

A. Internet Protocol IP

In general, the Internet protocol IP is which allows communication between devices belonging to a network based on address data.

There are two versions of the protocol operating on the Internet: 4 and 6. Since version 4 the most used.

CHARACTERISTICS:

- addressing.
- routing.
- encapsulation.
- Best effort

B. Internet Protocol version 4 (IPv4)

It is the fourth version of Internet protocol, is based on a 32bit address, limiting it to $2 \land 32 = 4 \ 294 \ 967 \ 296$ unique addresses to be distributed worldwide.

In principle, the number of addresses was enough, but over the years with the evolution of technology, the growth of internet and ways to access it, it was necessary to find methods and ways to optimize the use of addresses, and which they tended to dwindle [1].

C. Internet Protocol version 6 (IPv6)

The problem of IP address shortage is resolved and has a large number of addresses to be used a total of 2 ^ 128 addresses [1].

It offers a more efficient administration of addresses. IPv6 being a hierarchical protocol allows an orderly register of addressing. Methods that allowed extend the life of IPv4 protocol, such as CIDR and NAT, also, in certain real-time

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applications, cause loss of information packets are removed [2]. Easier management of TCP / IP protocol.

D. Dual Stack

Dual Stack It is one of the mechanisms of transition and coexistence between IPv4 and IPv6 protocols most commonly used, because this use of a dual node IPv6 / IPv4 stack is made, which allows communication as if it were an IPv4 or IPv6 node at a time, for this to happen it is necessary that each node is configured with two types of IPv4 and IPv6 [3].

The implementation of this mechanism to enable or disable any of the stack, for this reason an IPv4 / IPv6 node can have three modes of operation [3]:

• A node with the IPv4 stack enabled and disabled IPv6 stack functions as an IPv4 node.

• A node with IPv6 stack IPv4 stack enabled and disabled, functions as an IPv6 node

• A node with both stacks enabled can use and run both protocols at the same time independently.

E. VoIP and IP telephony

VoIP means Voice over IP, which also can be known as Voice over IP and IP Voice; is the group of resources that enable the voice signal can be transmitted by the Internet using the IP protocol. This means that the voice is sent digitally, in data packets.

IP telephony is the direct application that offers VoIP is a service that is offered to the public and which makes use of VoIP technology [4].

F. Functionality test.

The tests performed here can determine the functionality and operation of the project and show objective raised. For which use is made of the topology of Fig. 1 and Wireshark packet capture, which allows VoIP traffic observe IPv6 generated when making phone calls is used



Fig. 1 Dual Stack Topology Demonstrating Project

IPV4 CALL TO IPV4.

The In Fig. 2 the origin of the call extension 7053 is configured in IPv4 and destination of the call extension 7475 is also configured in IPv4 is observed. One can observe the initiation and connection setup, then the packet exchange and finally the termination of the call is, thus showing that the call is made and that there is communication



Fig. 2 Establishing a connection and exchange of packets called IPv4

CALL FROM 7053 IPv4 TO 7043 IPv6.

Similarly, in Fig. 3, the source of the call, which is the extension 7053, registered with an IPv4 address and destination of the call, which is the extension 7043 that is registered with an address in IPv6 is shown; Furthermore, the process of starting, establishment, exchange and communication completion, thus verified that the call is made without any problem and that there is communication in both IPv4 and IPv6 simultaneously shown.

	Realtek PCIe GB	E Family Controller - Graph Analysis						
Time	172.16. IP SERVIDOR IP EXTENSIÓN 172.16.	Comment						
169,092	INVITE SDP (opusRTPType-96 SILKRT	SIP From: 7053 <sip:7053@172.16. 7043@[2800:68:19:=""]<="" th="" to:<sip=""></sip:7053@172.16.>						
169,092	401 Unauthorized	SIP Status Origen Destino						
169,093	ACK	SIP Request						
169,094	INVITE SDP (opusRTPType-96 SILKRT	SIP From: "7053" <sip:7053@172.16.]<="" th="" to:<sip:7043@[2800:68:19:=""></sip:7053@172.16.>						
169,096	100 Trying	SIP Status						
169,136	180 Ringing	SIP Status						
169,320	180 Ringing	SIP Status						
184,140	200 OK SDP (GSM g711U g711A telepl	SIP Status						
184,149	(SOED) ACK	SIP Request						
184,300	(5201) RTP (GSM)	RTP Num packets:114 Duration:2.262s SSRC:0x5D828766						
184,301	RTP (GSM)	RTP Num packets:110 Duration:2.179s SSRC:0x7B263E7C						
186,544	(SOED) BYE (SOED)	SIP Request						
186,555	(5060) 200 OK	SIP Status						

Fig. 3 Establishing a connection and exchange of packets called IPv4 to IPv6

G. IPv6 function tests

To demonstrate the correct operation of IP telephony system the Wireshark tool, which allows you to capture the kind of traffic that is generated within a network is used, you can also see the process of interconnection between two devices within the network. It is a very valuable tool to effectively see how the IPv6 protocol within the network

INTERNAL CALLS BETWEEN EXTENSIONS.

To check the operation a call between extensions configured in the terminal clients is performed (see Fig. 4) and Wireshark proceed to check traffic in IPv6 that is generated, the process of establishing connection to the packet exchange and flow of traffic. The call is made between 7053 and 7048 extensions.

Realtek PCIe GBE Family Controller - VolP	Calls	+								
Detected 3 VolP Calls. Selected 0 Calls.										
 From 	 To 	Protocol Packets								
"7053" <sip:7053@[2800:68:19: 0]<="" td=""><td><sip:7048@[2800:68:19: 0]<="" td=""><td>SIP</td></sip:7048@[2800:68:19:></td></sip:7053@[2800:68:19:>	<sip:7048@[2800:68:19: 0]<="" td=""><td>SIP</td></sip:7048@[2800:68:19:>	SIP								
"7053" <sip:7053@[2800:68:19: 0]<="" td=""><td><sip:7048@[2800:68:19: 0<="" td=""><td>SIP</td></sip:7048@[2800:68:19:></td></sip:7053@[2800:68:19:>	<sip:7048@[2800:68:19: 0<="" td=""><td>SIP</td></sip:7048@[2800:68:19:>	SIP								
"Juana " <sip:7048@[2800:68:19: 0<="" td=""><td><sip:7053@[2800:68:19: :="" cb<="" td=""><td>SIP</td></sip:7053@[2800:68:19:></td></sip:7048@[2800:68:19:>	<sip:7053@[2800:68:19: :="" cb<="" td=""><td>SIP</td></sip:7053@[2800:68:19:>	SIP								
ORIGEN DE LLAMADA	DESTINO DE LLAMADA									
Total: Calls: 3 Start packets: 0 Completed calls: 2 Rejected calls: 2										
Flow Player	Select <u>A</u> ll	Close								

Fig. 4 VoIP call made between extensions: 7048-7053 in IPv6.

In Fig. 4 it can be seen that the call was successful; Now in Fig. 5 can be seen the establishment of the call, the exchange of packets, traffic flow and completion of communication.

Time	2800:68:19: : 2800:68:19: cb 2800:68:19: 0	Comment
570,221	INVITE SDP (g711U SSM g711A telephone-ev	SIP From: "Juana " <sip:7048@[2800:68:19: 0]="" <sip:7053@[2800:68:19:="" cb<="" td="" to=""></sip:7048@[2800:68:19:>
570,240	50401 180 Ringing	SIP Status ORIGEN DESTINO
570,742	(5060) 180 Ringing	SIP Status
571,741	180 Ringing	SIP Status
572,602	100 OK S <u>DP (g711U GSM g71</u> 1A telephone-oventRTP.	SIP Status
572,603	(SG60)	SIP Request
572,684	RTP (q711U) (5030)	RTP Num packets:151 Duration:3.008s SSRC:0x1D86788A
572,737	RTP (g711U)	RTP Num packets:148 Duration:2:940s SSRC:0xAC1A4D2
575,682	(5060) BYE (5060)	SIP Request
575,683	(5060) 200 OK	SIP Status FINALIZACION DE LA CX

Fig. 5 Establishment of a SIP session in IPv6 between 7048 and 7053 extensions

In the verification test internal calls between extensions configured on the phone or terminal equipment has met the target, this allows you to see the settings made in the IP phone, such as static address and IPv6 account were appropriate. And so communication between the offices of the various departments with the UTN.

EXTERNAL CALLS TO PSTN.

In the same way the Wireshark tool which can be seen making a call between an extension of the internal network with a number of the PSTN or external, with which you want to communicate is used; in this case the test is done on a call from extension 7054 to a cell phone number 0989332767, which belongs to the network administrator, this can be seen in Fig. 6.

	Detected 5 Vo	IP Calls. Selected 0 Calls.						
٠	From •	То		٠	Protocol	Packets	•	State
1	"7054" <sip:7054@[2800:6< td=""><td><pre><sip:7053@[2800:68:19:< pre=""></sip:7053@[2800:68:19:<></pre></td><td>1</td><td></td><td>SIP</td><td></td><td>10</td><td>COMPLETED</td></sip:7054@[2800:6<>	<pre><sip:7053@[2800:68:19:< pre=""></sip:7053@[2800:68:19:<></pre>	1		SIP		10	COMPLETED
1	"7054" <sip:7054@[2800:6< td=""><td><pre><sip:7053@[2800:68:19:< pre=""></sip:7053@[2800:68:19:<></pre></td><td>1</td><td></td><td>SIP</td><td></td><td>10</td><td>CANCELLED</td></sip:7054@[2800:6<>	<pre><sip:7053@[2800:68:19:< pre=""></sip:7053@[2800:68:19:<></pre>	1		SIP		10	CANCELLED
1	"7054" <sip:7054@[2800:6< td=""><td><pre><sip:7053@[2800:68:19:< pre=""></sip:7053@[2800:68:19:<></pre></td><td>1</td><td></td><td>SIP</td><td></td><td>10</td><td>CANCELLED</td></sip:7054@[2800:6<>	<pre><sip:7053@[2800:68:19:< pre=""></sip:7053@[2800:68:19:<></pre>	1		SIP		10	CANCELLED
	"Paul Espinel" <sip:8000@< td=""><td><sip:7054@[2800:68:19:1-< td=""><td></td><td></td><td>SIP</td><td></td><td>8</td><td>CANCELLED</td></sip:7054@[2800:68:19:1-<></td></sip:8000@<>	<sip:7054@[2800:68:19:1-< td=""><td></td><td></td><td>SIP</td><td></td><td>8</td><td>CANCELLED</td></sip:7054@[2800:68:19:1-<>			SIP		8	CANCELLED
1	'7054" <sip:7054@[2800:6< td=""><td><sip:90989332767@[2800:6< td=""><td>8:19:]</td><td></td><td>SIP</td><td></td><td>9</td><td>COMPLETED</td></sip:90989332767@[2800:6<></td></sip:7054@[2800:6<>	<sip:90989332767@[2800:6< td=""><td>8:19:]</td><td></td><td>SIP</td><td></td><td>9</td><td>COMPLETED</td></sip:90989332767@[2800:6<>	8:19:]		SIP		9	COMPLETED
	ORIGEN DE LA LLAMADA	DESTINO DE LA LLAMADA						

Fig. 6 Process external call

Then in Fig. 7 the process of establishing the SIP session between the extension and the external phone number is displayed. Thus the operation of calls is verified in IPv6 outward of the university network.

2800:68:19:1 2800:68:19: cb 2800:68:19: 0	Comment								
NVITE SDP (opusRTPType-96 SILKRTPType-97 SILK	SIP From "7054" <sip:7054@[2800:68:19: 0]<="" sip:90989332767@[2800:68:19:="" th="" to:=""></sip:7054@[2800:68:19:>								
401 Unauthorized	SIP Status ORIGEN DE DESTINO DE LA LLAMADA								
ACK	LA LLAMADA								
NVITE SDP (opusRTPType-96 SILKRTPType-97 SILK	NUMERO CELULAR SID From 170541 4 sin 7054/01/2000/68-10 To 4 sin 20080332767 01/2000/68-10 01								
(5060) 100 Trying									
200 OK SBP (GSM g7111L g711A telephone-eventBTP	INICIO DE LA CX								
(SOGO)	SIP Status								
	SIP Request								
(15750) (5022)	RTP Num packets:121 Buration:2.397s SSRC:0x1602E889								
(15750) RTP (GSM)	RTP Num packets:1251 Duration:25.1175 35RC:0x3864045								
RTP (telephone-event) DTMF One 1	RTP Num packets:5 Duration:0.075s SSRC:0x1602E889 ESTABLECIMIENTO								
RTP (GSM)	RTP Num packets27 Duration:0.521s SSRC:0x1602E889 RTP Num packets5 Duration:0.081s SSRC:0x1602E889 RTP Num packets-5 Duration:0.081s SSRC:0x1602E889								
RTP (telephone-event) DTMF Nine 9									
(15750) RTP (GSM)									
BTP (teléphone-event) DTME Nine 9	RIPNium packet: 5 Duration:0.076r SSRC:0x16025880								
(15750) PTD (CSM) (5022)	KIP Hum packets:5 Duration:0.0765 SSRC:0x1602E889								
(15750) (15750) (15022)	RTP Num packets:54. Duration:0.667s SSRC:0x1602E889								
(15750) (5022)	RTP Num packets:6 Duration:0:090s SSRC:0x1602E889								
(15750) RTP (GSM) (5022)	RTP Num packets:30 Duration:0.579s SSRC:0x1602E889								
RTP (telephone-event) DTMF Pound	RTP Num packets:5 Duration:0.090s SSRC:0x1602E889 INTERCAMBIO								
RTP (GSM)	RTP Num packets:1008 Duration:20.143s SSRC:0x1602E889								
BYE	SIP Request								
(S060) 200 OK	SIP Status FINALIZACIÓN DE LA CX								
(5060)									

Fig. 7 Establishing an IPv6 SIP session between an extension and an external number.

In the verification test calls to the outside, in this case a cell number, can see that perform calls without problems and that the settings made on the IP phones, such as static address and the account in IPv6 were appropriate. Thus communication is achieved towards the outside of the university.

EXTERNAL CALLS TO THE UNIVERSITY

We proceed to verify the calls originated externally, either the cellular network or PSTN to college, the results are positive as the call is made without difficulty. In Fig. 8 shows the performance of the external call the number 2511 311 belonging to CNT, towards the extension 7053.

Detected 3 VoIP Calls. Selected 0 Calls.											
From	•	To	(Protocol (Packets	(State	•			
"7053" <sip:7053@[2800:68:19:< th=""><td></td><td><sip:7048@[2800:68:19:< td=""><td></td><td>SIP</td><td></td><td>11</td><td>CANCELLE</td><td>D</td></sip:7048@[2800:68:19:<></td></sip:7053@[2800:68:19:<>		<sip:7048@[2800:68:19:< td=""><td></td><td>SIP</td><td></td><td>11</td><td>CANCELLE</td><td>D</td></sip:7048@[2800:68:19:<>		SIP		11	CANCELLE	D			
"7053" <sip:7053@[2800:68:19:]<="" th=""><td></td><td><sip:7048@[2800:68:19:< td=""><td></td><td>SIP</td><td></td><td>11</td><td>COMPLET</td><td>ED</td></sip:7048@[2800:68:19:<></td></sip:7053@[2800:68:19:>		<sip:7048@[2800:68:19:< td=""><td></td><td>SIP</td><td></td><td>11</td><td>COMPLET</td><td>ED</td></sip:7048@[2800:68:19:<>		SIP		11	COMPLET	ED			
"2511311" <sip:2511311@[2800:68:19: 0]<="" th=""><th></th><th><sip:7053@[2800:68:19:1 cb<="" th=""><th></th><th>SIP</th><th></th><th>8</th><th>COMPLET</th><th>ED</th></sip:7053@[2800:68:19:1></th></sip:2511311@[2800:68:19:>		<sip:7053@[2800:68:19:1 cb<="" th=""><th></th><th>SIP</th><th></th><th>8</th><th>COMPLET</th><th>ED</th></sip:7053@[2800:68:19:1>		SIP		8	COMPLET	ED			
ORIGEN DE LA LLAMADA		DESTINO DE LA LLAMADA									

Fig. 8 Process call to the PSTN

Then in Fig. 9, the process of establishing the SIP session between the telephone number belonging to the PSTN and

telephone extension in IPv6 is displayed. Thus the operation of external calls to the UTN verified.



Fig. 9 Establishing an IPv6 SIP session between a PSTN number and an extension.

In the verification test of external calls to the university has met the target, this allows you to see the settings made in the PBX, IP phones as the static address and IPv6 account were appropriate. And so communication is achieved from outside the university, either the PSTN or cellular network, to any of the extensions of the UTN.

After observing the result of each test can be determined that the main objective has been met, the system of IP Telephony University can make and receive phone calls, both internal and external, having set up their systems protocol IPv6.

III. CONCLUSIONS

- The version of Internet Protocol (IP) version four (IPv4) to version six (IPv6) in the IP telephony system at the Universidad Técnica del Norte was migrated with this has gone a step further in the process migration of telecommunications service university to IPv6 protocol.
- An address in IPv6 with the resource that has the university by CEDIA, a resource that was enough to satisfy greatly requirement addresses to be configured on computers that are part of the IP telephony system of the institution was performed. Thus it was possible to assign a specific address to each element.
- The SIP signaling protocol was used, as it has already defined the characteristics for the transition to IPv6 and enabled interoperability between IPv4 and IPv6 using dual stack, thus ensured the coexistence of the two protocols implemented in the system phone.
- Based on performance tests conducted have IPv6 connectivity in the phone system IP college, this was done using the grabber Wireshark packet, which allowed us to observe the different types of traffic generated on the network both IPv4 and IPv6, in addition to the process of establishing SIP sessions, in each case, while performance tests were performed.
- IPv6 proved to be a robust and suitable to be implemented in the telephone system, being a hierarchical protocol allowed to have a plan addressing ordered and related to the number of each telephone extension, by having IPSec as base provides

greater security protocol does not It allows communications to be heard with the use of packet sniffers or grabbers, providing greater security.

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V. BIOGRAFÍAS



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