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Cover Page Footnote

The authors would like to thank all the staffs of Cameroon Telecommunications for agreeing to participate in this survey and experiment, and the Cameroon Telecommunications management for the support given in carrying out this study.

Integrating Voice over IP Solution in IPv6 and IPv4 Networks to Increase Employee Productivity: A Case Study of Cameroon Telecommunications (Camtel), North-West

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ABSTRACT

Telecommunications organizations have to follow the rapid innovation of technology if they want to face challenges raised by competition. The challenge to respond to the huge market demand of updated products and services from customers requires that the organization's working environment be equipped with tools and communication facilities that contribute to ameliorating productivity. Cameroon Telecommunications (Camtel) is facing a digital telephony and Internet Protocol strategic management challenge. Successful implementation cannot be achieved if the employees are still depending on the ageing public switched telephone network (PSTN) as their primary communication system, despite the frequent loss of dial tone experience in a day which can last up to a week, with serious repercussions on business activities and revenues. This study is designed to provide a solution to the telecommunications challenge. The fundamental question is how to integrate a digital telephony system that will provide telephony services in the existing IPv4 data network while prioritizing IPv6 traffic forwarding. This study proposes and implements solutions that integrate a Voice over IP solution with IPv6 as an alternative communication system that relies on the existing IPv4 data network. VoIP is deemed as one of the driving forces behind the adoption of IPv6.

The purpose is to offer to workers an option that will free them from the poor Quality of Service (QoS) of their existing PSTN based solution, hopefully enhancing the overall productivity. This paper follows two research methodologies: Qualitative Research in Applied Situations and Engineering design process. The *first part of this study reports the results of the evaluation of how much such a solution can enhance workers' productivity. As it is important to provide an environment where IPv4 and IPv6 networks and applications/devices can interoperate in the context of VoIP;the second part describes practically a simulation environment where various configurations of network entities are done following a Dual-Stack transition approach. Document and records were used to gather information related to the structure, operations, and topological update of the Camtel's existing IP data network. The findings demonstrated that VoIP can be an effective communication solution for Camtel and its implementation with IPv6 will be preferable. However, for this to be efficient there must be a provision of sufficient bandwidth and usage of types of equipment and transmission mediums that minimizes processing and propagation delays. Findings also reveal that better productivity will be achieved if workers are fully trained for the exploitation. This research article tries to highlight, discuss a required transition roadmap and extend the local knowledge and practice on IPv6. Future expansion of this research work will consist of deploying Dual-Stack VoIP in the remaining 9 regional offices for full integration in the corporate communication system of Camtel.*

Keywords: Internet Protocol; VoIP; Dual-Stack; IPv4; IPv6.

INTRODUCTION

IPv6, the latest version of the Internet Protocol (IP) standard, is currently supplementing and if goes as planned will eventually replace IPv4, the previous version of IP. IP facilitates the exchange of data between networks and sets forth the address resources and numbering scheme required to connect devices to the global Internet. IP offers reliable/unreliable means through which different types of networks running various types of services can transfer different types of data to different destinations. IPv6 was standardized to provide enhanced capabilities and address IPv4 technological limitations. Peter Dordal (2020) thinks that the IP was developed to solve the scaling problem with Ethernet and to allow support for other types of LANs and Point-to-Point links as well. He argues that perhaps the central issue in the design of IP was to support universal connectivity in such a way as to allow scaling to enormous size.

While the two protocols have some functional similarities, they are distinct and not backward compatible; IPv4-only devices cannot communicate directly with IPv6-only devices and vice-versa. Consequently, organizations wishing to take full advantage of the enhanced features of IPv6 must upgrade their entire network

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infrastructure and end devices to support IPv6, while at the same time maintaining IPv4 support for legacy systems that will not or cannot be upgraded. The costs and risks associated with upgrading an entire network to support a new protocol with no intrinsic return on investment have acted as a disincentive for IPv6 adoption. To be sure, the transition of the Internet to IPv6 has certainly taken a leisurely pace over the past decades. Although a dual-stacked IPv4/IPv6 environment is said to be incapable of sustaining growth in the long term (Alghatrifi and Khalid, 2018), it is still economically important to find a way to make these two protocols operate in the same environment or integrate IPv6 in an existing IPv4 network without affecting ongoing operations. Nick Buraglio (2018) in his article “Three Reasons why IPv6 is worth the effort” mentioned three main features which he believes are the killer apps or amazing features brought by IPv6. These are End-to-End connectivity, minimize network complexity (with a mechanism such as NAT that is not necessary with IPv6) and ability to make support and content a commonplace (IPv6 has far more support in hardware, software, and content). This can surely justify why most organization today are migrating towards IPv6. As the deployment of IPv6 infrastructures evolves, interaction with the existing IPv4 networks is unavoidable.

This qualitative research in applied situations study first investigates the communication working environment of workers of Camtel North-West region with a focus on two communication network platforms: the *IP* data network and the PSTN. Doing an IPv6 implementation project does not involve tearing down an aging Camtel IPv4 network and replacing it with a new IPv6-enabled network. Instead, the Camtel IPv4 and IPv6 networks will run in parallel in what the industry calls a "dual-stack" network. The study studies an environment where IPv4 and IPv6 networks and applications /devices can interoperate in the context of VoIP. The study further design and integrates a digital telephony system that works together with the PSTN so that employee productivity can be improved. Specifically, this research discusses a solution to Camtel need to implement "dual-stack" in order to ensure efficiency, global connectivity, and long-term growth of the CAMTEL North West Region connectivity.

The proposed solution includes:

1. An approach they may use to integrate the IPv6 addressing scheme in collaboration with their existing IPV4 data network;
2. A way they may incorporate a VoIP solution in that secure corporate data network to develop another way of communication that frees them from the dependence of the Public Switched Telephone Network.
3. A working IP data environment where IPv4 and IPv6 are used to efficiently run all corporate services and applications while integrating a

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new service such as VoIP and maintain good QoS of existing services and applications.

Statement of the problem and Research questions

This research was carried following various problems observed and identified in the existing communication system of Camtel, perceived both from the view of the operating PSTN and the IP data network. For the PSTN, the PSTN network of this organization is very old and provides very poor Quality of Service (QoS) to workers and customers. Generally, this ageing problem causes high expenditure on maintenance, upgrades, and hardware following business growth. The organization runs these maintenance costs both at the level of the network (Chambers, multi-pairs of cable, poles, DPs, Copper), time and humans resources. Besides, this existing PSTN does not have a fully operational central office or Local exchange despite the existence of an NGN Soft switch that is not fully exploited. Therefore, the dependency on the headquarter Central Office (CO) for activation of new subscriber dial-tone, service maintenance and allocation of supplementary services causes delays in the execution of tasks related to the subscriber line which causes a serious drop in term of performance of work done. Also, there are major concerns like multiple losses of dial tone that may last days; making communication impossible during working hours and making workers use their cell-phones knowing pertinently all possible distracting apps (WhatsApp, etc.) that can alter concentration at work and impact on productivity. Added to it is the problem of limitations related to PSTN such as charging apply in any basic features provided by a modern digital system such as conference calls, 3-way calling, external network call, with no possibility of Video Conferencing.

When this network becomes unstable (ON and OFF tone availability, Noise on the line, Crosstalk during an exchange between parties, Low tone due to under-voltage transmitted on the line), it does not only affect workers but also all the customers sharing that same network. IT raises the problem of reliability, privacy control, poor productivity (workers or clients not able to reach each other on time and effectively, delay in procedures and impact in decisions making), with security issues in terms of preserving the integrity of information related to the business and the users of that network.

In addition, Camtel data network operates with IPv4 as their only communication data protocol. They have the need to plan for a migration process towards IPv6. NAT mechanism was mainly used to solve their limitations of public IPv4. This causes at the pick hours, high processing of BRAS (Broadband Access Services) equipment (L3 devices) with High Memory usage (over 92%), making internet traffic slow and generates the poor quality of Internet for users connected to the PSTN via their ADSL (asymmetric Digital Subscriber Line) line. Besides these

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problems related to High NATing processes, there is no digitalized telephony service that can act as an alternative to the PSTN. Corporate workers only depend on the availability of the unstable dial tone provided by the PSTN. Strategizing a solution for these problems is how to integrate a digitalize communication system that uses an updated protocol such as IPv6 while providing Digital telephony services as an alternative.

Indeed, Camtel can be the best test-bed for IT projects and future case studies on diffusion and adoption of emerging technologies. The organization has an interesting dynamics surrounding IT development with rapid diffusion and penetration rate.

It is worthwhile to see how Camtel develops low-cost digital communication using VoIP technology and what might affect its implementation. The study is focus on providing answers to the following question:

1. How to integrate IPv6 in the existing IPv4 data network of Camtel North-West?
2. Will this integration affect the existing quality of service? And how to maintain the acceptable Quality of Service (QoS)?
3. How to integrate a VoIP system that will provide telephony services in the existing IPv4 data network and give priority to IPv6 traffic forwarding?
4. Will this new solution effectively constitute an alternative communication system to the PSTN?

Based on the research questions formulated above, this study tested the following NULL (**Ho**) and Alternative (**Ha**) hypothesis:

Ho: The Integration of VoIP with IPv6 in the IPv4 data network of Camtel North-West will not improve the communication system of workers to make them more productive.

Ha: The Integration of VoIP with IPv6 in the IPv4 data network of Camtel North-west will improve the communication system of workers and by so, making them more productive.

Scope of the study

The scope of the study is addressing the IP data network of the regional representation of Camtel and does not in any way consider the physical nor the logical topology of extended regions. The research does not consider all the other 9 regional offices of Camtel. However, the proposed solution is capable of been deployed in all the other regions. The study does not emphasize operation or functionalities of the Public Switch Telephone Network (PSTN). Rather, the study addresses the limitations of this system to justify why and how VoIP contributes to

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solving most of these limitations cost-effectively and optimally. The implementation aspect of the solution in this research work is done in the form of simulation and includes the configurations of Cisco IP and Soft Phones, Cisco layer 3 switches and Routers. We shall not consider how configurations apply on extended gateway or gatekeepers are done to provide VoIP services.

The rest of the current paper is structured as follows: In the first section, we provides a general introduction and research background with discussion on the problem statement, the purpose of the study, research questions and hypothesis based on some theoretical constructs. In Section 2 we presented a brief literature review focusing on IPv4 and IPv6. The research Methodologies adopted for this work are discussed in section 3 where analysis of data is presented and results of hypothesis testing. In Section 4 we present evaluation of the current CAMTEL system followed by the proposed VoIP Dual-Stack design integration and Section 6 discusses the effective implementation. In section 7, a sequence of tests conducted and an associated result is presented and subsequently the conclusion.

LITERATURE REVIEW

Nowadays, lots of works and researches have been done on IPv6 and its related issues, and there is still a long way to go.
IPv4 and IPv6 must coexist for some number of years, and their coexistence must be transparent to end users. IETF Next Generation Transition Working Group (NGtrans) developed IPv4/IPv6 transition mechanisms. Internet Society (ISOC), a global nonprofit organization that certifies technical standards for the Internet has created a Web portal, Deploy 360, to share information about how to deploy an IPv6-compliant network. On the site are a number of case studies on how IPv6 rollouts went, including one about the project at Oxford University in the United Kingdom. Different authors (e.g., Dooley and Rooney, 2013; Auben networks, 2014) have defined IPv4/IPv6 co-existence technologies and discusses the salient features and advantages of each to help us to decide where a given technology choice makes the most sense.
Move from lack of IPv6 support to full support of this protocol does not occur instantaneously. From a general perspective, the set of IPv4/IPv6 co-existence technologies can be organized into three categories:

- dual stack - implementation of both IPv4 and IPv6 protocols on network devices;

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- tunneling - encapsulation of an IPv6 packet within an IPv4 packet for transmission over an IPv4 network or vice-versa;
- translation - IP header, address, and/or port translation such as that performed by host, gateway or network address translation (NAT) devices.

Dual stacking is the preferred solution in many scenarios. The dual-stacked device can interoperate equally with IPv4 devices, IPv6 devices, and other dual-stacked devices.

According to Lambrinos and Kirstein (2007), the transition from the current IPv4 network protocol structure to the next-generation IPv6 network has always been deemed as a lengthy process. Oxford University's Guy Edwards detailed a five-step plan for deploying IPv6 alongside the existing IPv4 network. First, Edwards advises, the organization should perform a network device audit, identifying all the routers, switches and firewalls on the network, as well as what specific versions of hardware and software are running. With the help of networking vendors, the next step is to determine which of the devices are already IPv6-compliant. Next is that network administrators run a test on a particular IPv6 device to make sure that the software application to run on the network works. In general, transition to IPv6 must be increment and overtime as these protocols will coexist for many years from now. Transition to IPv6 involves the upgrading of applications, hosts, routers, and DNS to support IPv6. Because this migration might take years, IPv6/IPv4 nodes must be able to coexist over IPv4 infrastructures such as the Internet and Intranets (Davies, 2012).

Internet Society (2018) stated that the levels of IPv6 deployment in networks and service providers all over the globe have increased dramatically. The response to this IP migration desire could not be a huge success without the ability of organizations to perform systems integration and management in their IP data networks. Systems integration is responsible for getting solutions, different technologies, applications, and infrastructure to work together, with a focus on technology integration (James & Holland, 2015). Service Integration is ensured by the flexibility offered by the Internet protocol format which facilitates the integration of new services making this protocol a predominant mean for modern data communication. James (2015) discussing on how Information Technologies services can be integrated and managed, thinks that it is the increasing complexity of the IT value chain and the rise of Multi-vendor supplier eco-systems that has led to the rise of service integration and management. These rises contributed to motivating the desire for ends–users to communicate using the latest technologies and consequently prompted telecommunications companies to always perform what is called ―IT technological watch (PowerNext, 2016). The impact of the IP on telephony is replacing traditional voice services with integrated voice and IP-based applications

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The digital revolution must ensure the availability of global digital communications with low-cost processing. An application that will result in a high demand for public network addresses is Voice over IP (VoIP). VoIP is deemed as one of the driving forces behind the adoption of IPv6 (Kirstein and Lambrinos, 2007). VoIP allows you to take analog audio signals, which are emitted when we talk on the phone and transforms them into digital data. This transition from analog to digital data opens the door to the union of two historically separate worlds, voice and data transmission, therefore, VoIP instead of being a service it is a technology. VoIP, has significant appeal for the enterprise, for service providers, and end-users because it allows the Internet and commonplace data networks, like those at offices, factories, and campuses, to become carriers for voice calls, video conferencing, and other real-time media applications (Wallingford, 2005). It is therefore important to provide an environment where IPv4 and IPv6 networks and applications /devices can interoperate in the context of VoIP. This environment is better based on an architecture that integrates all the components (whether they support IPv4 or IPv6 or both) thereby providing ubiquitous access to IP telephony services. Five main advantages of VoIP (Singh et al., 2013) include:

- Low cost: One of its main advantages is the use of existing infrastructure without additional wiring costs.
- Service Integration: Integration with the traditional public telephone network is made (PSTN), which also provides audiovisual communication sessions on network packets.
- High scalability and facility of updating: VoIP system allows an increase in connectivity by increasing the data speed and functionality of the Internet. The hardware cost is very low as well because it is based on hardware components to connect to the internet.
- Disaster recovery: Due to the failure can be monitored through various channels.
- Security: A Virtual Private Network (VPN) can be implemented to restrict access to internet bandwidth and use different protection methods in this way.

CISCO is an example of an organization that succeeded in the deployment and implementation of IPv6 in their existing IPv4 data network. Cisco IT (2013) said the migration started with a planning phase that was conducted base on some keys guidelines such as making sure that ongoing processes should not be disrupted or interrupted, new processes developed to support operational efficiency, the design considers coexistence with IPv4 in a long term, the Service Level Agreement (SLA) and security in case of using the Dual-Stack transition technic must be at least as efficient as the existing IPv4. Verizon Wireless, in another example, deployed IPv6 in a Dual-Stack mode because of the need to maintain IPv4 traffic for those servers

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that will continue to use that standard and, IPv6 for newly deployed servers. This Dual-Stack applies both for existing and new Verizon customers (Verizon, 2020).
Organizations have been attempting to deploy a VoIP solution using IPv6. Most organization that attempted such a deployment reported that it was a good option. This is why Daniel (2019) thinks that compared with IPv4; IPv6 can deliver more comprehensive communication services to network applications. This requires that signaling devices and Protocols (H.323/SIP, RTP, RTCP) with CODECs should have IPv6 capability to be able to implement VoIP in a Dual-Stack Network. Even though Finnell (2018) believes that PSTN is viewed as more reliable and secure than VoIP networks, he admitted that VoIP services have improved and said VoIP has its advantages, including lower network infrastructure costs, scalability, and advanced features, such as unified communications and applications integrations.
In their 2007 paper, Lambrinos and Kirstein (2007), proposed an architecture that integrates a collection of components to provide ubiquitous access to VoIP telephony for both IPv4 and IPv6 user agents. This environment is based on an architecture that integrates all the components (whether they support IPv4 or IPv6 or both) thereby providing ubiquitous access to IP telephony services based on the SIP signaling protocol. Besides, Rojas et al (2018) presented a proposal for the implementation of low-cost digital communications using VoIP technology to improve communication with the customer and increase the income collected by the Centre for the Provision of Services of Micaela Bastidas National University. VoIP technology, the authors said, enables the availability of global digital communications at low-cost to achieve operational excellence and compete for market share. VoIP technology applied to improve communication in an organization, Singh et al (2013) said, can lead to an impact on the business environment.

Unified communication in companies with 100 employees can save a considerable time of 191 hours per day, and $ 920.00 per year in productivity (Pruitt, Digium, 2016). The research carried out by (Jacobs, Yu, and Chavez, 2016), explores the perspective of the internal communication effect and employee satisfaction, finding a positive effect on employee satisfaction and a significant influence on internal integration, which is reflected in the external integration of the company. Effective communication in cooperation with employee satisfaction is a requirement to improve the performance of the organization.

Known Challenges of IPv6 for VoIP

Almost all characteristics about IPv6 are good technological news. However, for VoIP, there is one drawback that all practitioners are made aware of. By definition, an IPv4 packet has a header size of 20 bytes, whereas an IPv6 packet header,

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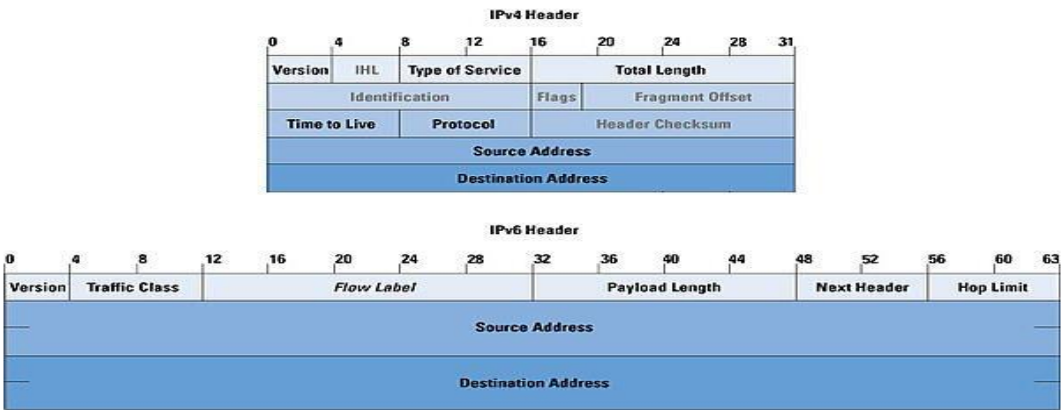
without any extension headers, has a size of 40 bytes. This is due to the increase in size of the IPv6 addresses themselves. This comparison is further illustrated in the diagram below (Noworatzky, 2019).



An increase of 20 bytes per packet for data packets that are typically 1500 bytes in size may not seem substantial; that's an increase of just over 1%. However, because the payload of VoIP packets is quite small in size – on the order of 20 to 160 bytes depending on the codec being used – an increase of just 20 bytes can have a substantial impact. Take the G.729 codec that has a voice payload of 20 bytes. With IPv4, this would result in a 40-byte packet including the header. With IPv6, this results in a 60-byte packet with the header. This is a 50% increase in size per packet, resulting in a 50% increase in required bandwidth. The smaller the payload of the packet, the more pronounced the increase in required bandwidth will become.
If IPv6 header extensions are used, the required bandwidth will increase even further.

RESEARCH METHODOLOGY AND RESEARCH

According to Rajasekar et. al. (2006), research is a logical and systematic search for new and useful information on a particular topic. It is an investigation of finding solutions to scientific and social problems through objective and systematic analysis. This research follows two research methodologies: Qualitative Research in Applied Situations and Engineering Research Method.

Qualitative Research in Applied Situations

Using qualitative applied research, this study is aims at finding a solution for an immediate problem facing Camtel organization. An applied study in qualitative research is a perfect fit for this study. It is usually form of studies that aim to understand specific behavior, behavior change or barriers to change; meaning of and reasons behind such behaviors; understanding the meaning of concepts in a specific group; capturing perceptions of a population on specific products or interventions; process documentation or evaluation of interventions; situational analysis; etc.(Kothari, 2008; Nakkeeran and Zodpey, 2012). The study planned on finding how much VoIP implement will enhance Camtel workers' productivity. The output of this methodology reveals after analysis of data collected from semi-structured interviews that out of 12 participants judiciously selected based on their mastering of technology and understanding of VoIP technology, (66%) were fully in support of such a solution while (33%) opted to practically see such solution operational for them to have a defined opinion on the effectiveness.

The researcher collected data and categorized it into three groups which are: the data related to the interviewee and directly to its job description (the purpose here is to know the interviewee better and introduce a discussion on the research), data related to the IP data network of Camtel North-West (to give to the researcher a better understanding of the existing environment and attempt to find a solution to the research questions), and finally, data related to the employee and how productive he is (to find out how much the existing communication systems contribute in making him productive and if the proposed solution effectively can be a better option).

Data collected from the interviews were later transcribed, codified and integrated into HyperResearch 4.0 for analysis. Table 1 shows the results of the frequency statistics of the analyses of these codes by the software. Table 2 shows the repartition of Interviewees per group and subjects discussed.

And Table 3 shows how the researcher conducted a demographic analysis of the population where it was revealed that more male was interviewed than females; most workers interviewed are well educated and experienced enough. However, it appears that there are very few IP data networks engineers (only 2) out of 150 workers. Considering that a bigger number of workers are commercials and with very little knowledge of technology, the researcher is focusing only on those who could be of great contribution (12 workers well-identified) to the research process, and strongly perceived that the solution will require specialized training for more effectiveness.

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Table 1. Results of the frequency statistics of the analyses of these codes by the software.

Codes	Total	Min	Max	Mean	Std Dev	Bar Graph
The domain of operation and speciality of employees	2	1	1	1	0	
Employee Appreciation of the PSTN and VoIP	5	2	3	2.5	0.707	
Employee Appreciation QoS parameters	3	1	2	1.5	0.707	
Employee evaluation of internet Protocols: IPv4 and IPv6	6	3	3	3	0	
Employee's appreciation of the Equipment's Capacities	1	0	1	0.5	0.707	
Employee's appreciation of VoIP6 features	1	0	1	0.5	0.707	
Employee's Productivity in the Job Position	12	4	8	6	2.828	

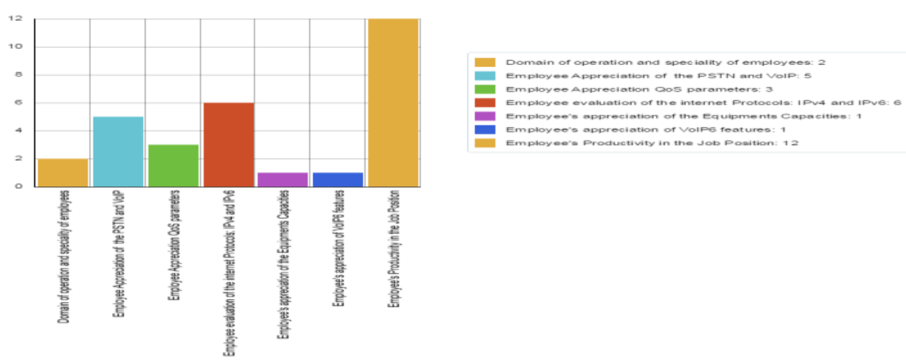

Figure 1. HyperResearch Data Analysis Results

In the course to validate which of the Hypothesis raised in section 1.2 is valid, an associated simulated test theory was formulated as we can see in Figure 2 using the transcribed files and related codes. Cases related to various codes generated with the aid of the software could be evaluated and theory related to Ho was rejected. Enough rules applicable to Ha were found and the related theory accepted, confirming enhancement of productivity once VoIP with IPv6 is integrated.

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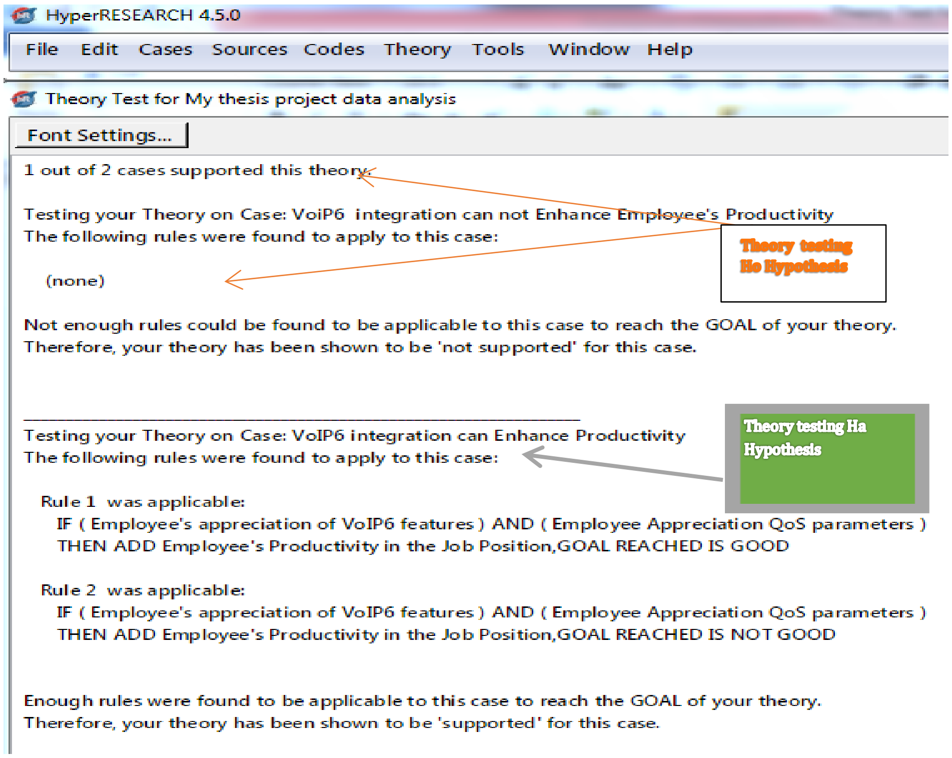

Figure 2. Testing Ho and Ha hypothesis

<table>
<tr><th colspan="4">Group of interviewees</th><th>Interviewee's questions per group</th><th>Group A</th><th>Group B</th></tr>
<tr><td rowspan="2">Group A</td><td rowspan="2">Total</td><td rowspan="2">Group B</td><td rowspan="2">Total</td><td>The data related to the interviewee and directly to its job description</td><td>✓</td><td>✓</td></tr>
<tr><td>The data related to the IP data network of Camtel North-West</td><td>✗</td><td>✓</td></tr>
<tr><td>Commercials and Technico Commercials</td><td>5</td><td>Technicians and Network Engineers</td><td>7</td><td>The data related to the employee and how productive he is</td><td>✓</td><td>✓</td></tr>
</table>

Table 2. Categorizing Interviewees and defining Focus Points

The research design establishes the decision-making processes, conceptual structure of investigation, and methods of analysis used to address the central research problem of the study. The specificity of this research methodology is summarized in Table 4.

GENDER	**FREQUENCY**	**PERCENTAGE (%)**	**Analyses**
Male	8	67	More male interviewed than females
Female	4	33	
Total	12	100	
LEVEL OF EDUCATION	**FREQUENCY**	**PERCENTAGE (%)**	**Analyses**
Primary	0	0	Most workers interviewed are well educated
Secondary	5	41	
University	7	59	
Total	12	100	
SERVICE LONGEVITY	**FREQUENCY**	**PERCENTAGE (%)**	**Analyses**
Below 5 years	3	25	Most workers are experienced enough and master their working environment
Between 6-10 years	4	33	
Above 10 years	5	42	
Total	12	100	
PROFESSIONAL DOMAIN	**FREQUENCY**	**PERCENTAGE (%)**	**Analyses**
Technicians / Engineers	2	16	They are very few engineers in the domain of IP in the North-West.
Technico - Commercials	6	56	
Commercials	4	34	
Total	12	100	
Familiarity with ICT tools and Digital Technology	**FREQUENCY**	**PERCENTAGE (%)**	**Analyses**
Expert	2	16	Environment but requires extra training in handling digital tools and equipment. The solution will require specialized training for more effectiveness.
Professionals	5	42	
Beginner	5	42	

Table 3. Demographic analysis of the studied sample

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Engineering Research Method

Defining an approach of how a low-cost digital communication using VoIP technology solution can be integrated into an existing IPv4 data network with IPv6 is our main challenge and line of enquiry. To achieve this research goal, the researcher after evaluation of workers opinions based on analysis of qualitative data collected from semi-structured interviews, also followed an engineering research process that helped in identifying the problem, conduct related research background concerning research questions, define requirements for the implementation of IPv4/IPv6 solutions, brainstorm on the possible design about Dual-Stack, Translation and Tunneling mechanisms, validate a design and conduct test to see effectively VoIP operation in a Dual-Stack network architecture. Guided by the flow chart of the engineering method research process depicted in Figure 3, in our prototype, the migration process to achieve through the following phases:

- Planning for IPv6 deployment (roadmap): how to create a migration process
- Survey of the prevailing network facilities (equipment): an inventory of the network can be an essential initial step to any type of network implementation planning
- Cost: vey essential to discuss and highlights the expense of migration from IPv4 to IPv6 (Such as Network Software, Network Hardware and operating system costs; training costs and unpredictable costs).
- Security and knowledge plan: essential data such as IP Addressing, Routing protocols, Firewalls and Intrusion systems etc. must be provided to the migration team.
- Considering possible methodology: Core to Edge and Edge to Core are basically the two ways to deploy IPv6
- Identify transition mechanism
- Test-bed network and final implementation
- Configurations and performance evaluation

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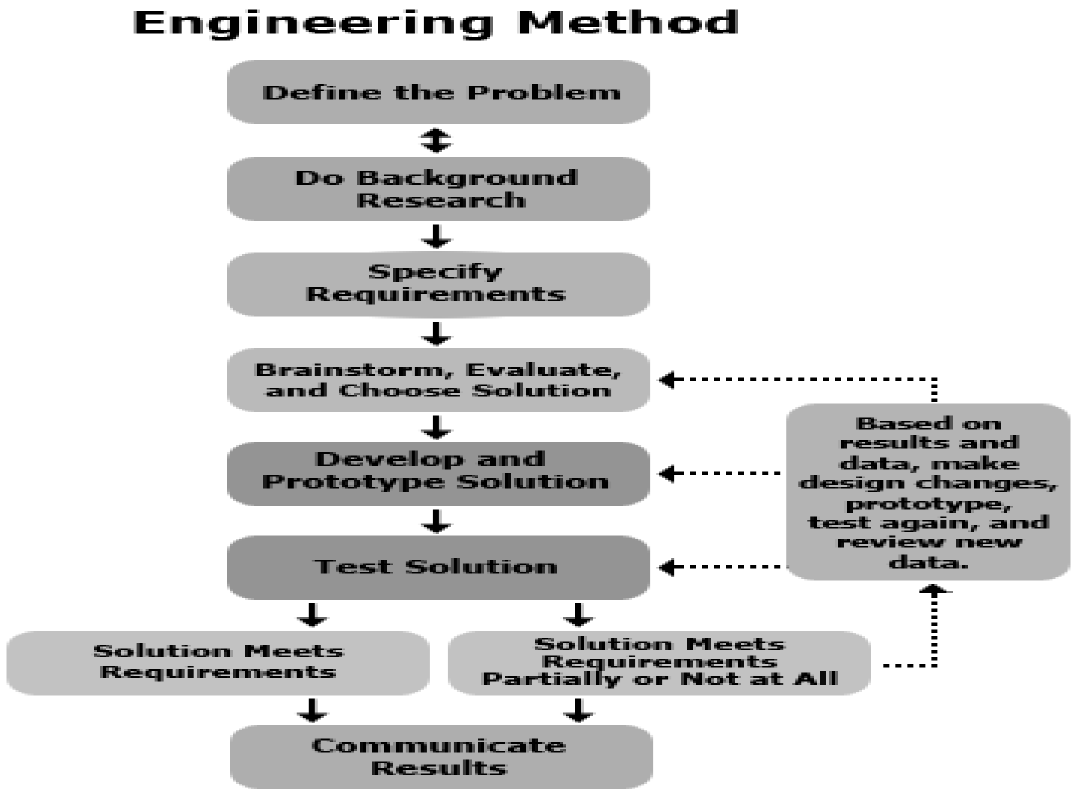

Figure 3. Research Flow Chart (Sciencebuddies, 2020)

Documents and Records

Researcher exploited a good number of technical documents showing the architecture of the existing network. The Privacy policy for using such sensitive documents could not permit the researcher to take some screenshot. However, access was granted for one week to permit exploitation and extraction of the required information. Figure 4 was specially designed from details related to these organization details and records

Validity and Reliability

Validity and Reliability of the Research were done by ensuring that from the population of 150 workers, a reliable sample of 12 workers was defined and data

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was collected from these people who understand and apply notions of IP and VoIP and therefore could be of more impact in the research results. Besides, data also was collected from direct people that can be a concern if the solution is implemented. Confidentiality of the interviewees and their sincere contribution was guaranty upon signature of a Concern form (Appendix 1). Interviews were conducted following an Interview Protocol Refinement (IPR) framework (Appendix 2).

Research Specifications	**Description**
Research Type: Qualitative and Applied Research methodology and Engineering research method	Finding an immediate solution to a poor internal communication system and evaluate the effectiveness of such a solution
Area of the Study: Telecommunications and Systems Networking environment of Camtel.	Regional Representative of Camtel Bamenda II Subdivision of the Mezam Division in the North-West region of Cameroon. The area was chosen because of the matching with the Information and System Networking Domain.
Research Design: Case Study	The specific investigation related only to Camtel North-West Region
Research Process: Engineering Research Process	Definition/Identification of the problem-> Related Background Research-> Define requirements-> Brainstorm + Evaluation = define Solution-> Develop a Prototype->Test Solution-> if OK, Communicate and Implement/ If not Review, Re-Test and Implement
Research approach: Deductive	The validity of theories/hypotheses (Ho and Ha tested). Ha confirm True = VoIP will enhance Workers productivity
Research Validity approach	- Data was collected from people who understand and apply notions of IP and VoIP - Data also were collected from direct people that can be a concern if the solution is implemented - Confidentiality of the interviewees and sincere contribution obtained after the signature of a **concern form.**
Research Reliability approach	Used of the **Interview Protocol Refinement (IPR) framework**. It gives a four-phase process to develop and fine-tune interviews protocols. (1) ensuring interview questions align with research questions, (2) constructing an inquiry-based conversation, (3) receiving feedback on interview protocols, and (4)

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	piloting the interview protocol. **Interview Form: Appendix 1**
Data Collection Method: semi-structured interviews (primary source), Documentation and Records (secondary sources)	- Interviewer (Researcher) prepares a set of same questions to be answered by all interviewees with the flexibility of opening further questions for clarification. - Data collected was codified and analyze with the aid of **HyperResearch SQA 4.5**, a Qualitative Data Analysis Software. - The researcher made use of existing documents and records of the organization for analysis of the existing IP data network and understand the various update that took place

Table 4. Summary of Research methodology

EVALUATION OF THE EXISTING SYSTEM

The IP data network of Camtel North-West is complex in architecture and divided according to the type of services (ToS) offered. Each type of service belongs to a separate management domain organize more in a centralized than distributed way. Figure 4 shows the existing architecture. In it are three management domains:

1. **The Corporate network Domain**: use a Gateway Router (Corporate) to provide workers with corporate applications such as NGBSS (Next Generation Billing Sub-System), IMS (IP multimedia Services). Most of these corporate applications are accessible via web interfaces and workers login daily to do their jobs according to the profile classification. The access layer of this network domain is made of computers, phones and tablets using the corporate applications. The distribution level has switches interconnected to provide connectivity to these access devices. Also, the extension of this network is made with the use of a Wireless access point placed at the level of agencies. The Core level has a gateway router that faces the outbound link towards Yaounde through Bafoussam. This router is configured purposely to offer these corporate services and Internet connection to job positions.

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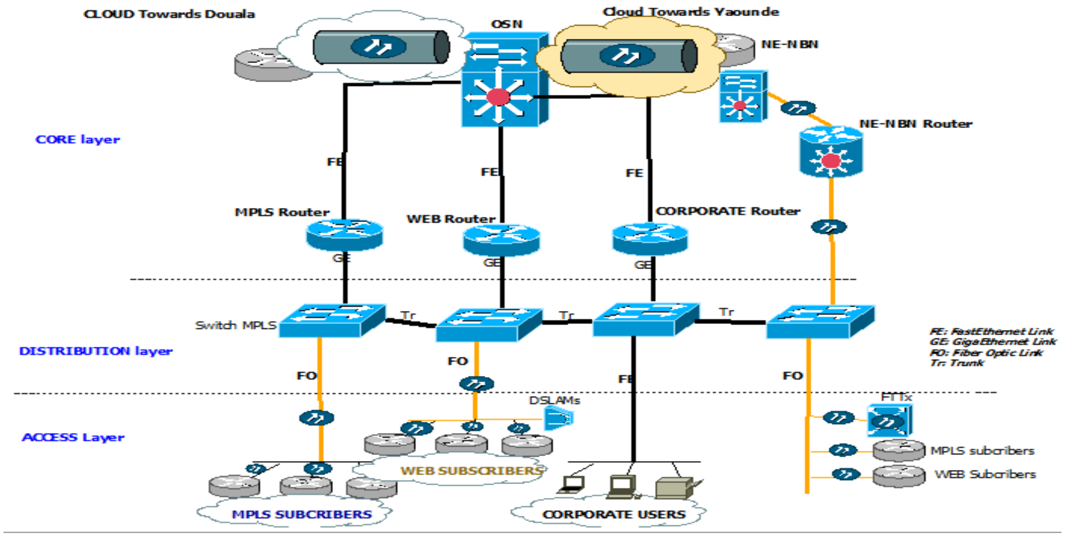

Figure 4. Existing IPV4 Data Network Architecture

2. **The MPLS/VPN network Domain**: this network domain is a dedicated network domain for organizations that need to interconnect their local or remote site using a secure VPN tunnel. The MPLS network provides IP forwarding based on labels, making remote sites to operate as if they were all locally connected. The access level of this network domain is made of subscriber‘s optical converters and routers that interface the MPLS router acting as a gateway for service provisioning and activation. The distribution layer implements segmentation of class of users by the configuration of VLANs. Each subscribers belonging to a particular VLAN.
   The Core level has an updated CISCO router of the 7600 series that acts as a gateway facing the outbound towards Douala. Service provided to the MPLS subscribers is from this Router that implement CEF (IP Cisco Express forwarding) with MPLS IP activated, and all CR (committed rate) bandwidth configurations. The main routing protocols implemented here are IS-IS (intermediate System-to-Intermediate System), BGP (Border Gateway Protocol), OSPF (Open Shortest Path) and RIP (Routing Information Protocol).

3. **The Internet network Domain**: this is another network management domain that focuses on providing internet services to subscribers according to the technology used. xDSL subscribers, for example, will receive internet from DSLAM(Digital Subscriber Lines Access Multiplex) connected to the PSTN and the IP data network. Dedicated Subscribers will have broadband internet with the aid of the Internet Router that uses public IPs to face the outside.

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The access layer of the domain is therefore made of optical converters connected to the optical transmission networks (monomode / multimode fibres deployed) and routers of subscribers. The distribution layer has a backbone internet switch that centralized all connections of the Point of Presence. It implements VLANs segmentation attributed uniquely to each subscriber. The core layer has a backbone internet Router that makes use of complex IP configuration to provide internet to the various type of customers (LS, xDSL, Intern-link, Pop extension, etc.).

These three management domains are interconnected via the use of trunk links configured in manageable switches at the distribution level. All these three networks are implemented on a backbone made of Fibers links pointing to Douala and Yaoundé via Bafoussam. The traffic generated by the MPLS, Corporate and WEB Routers is forwarded out of the region through the Optical Switching Network (OSN) of the National Fiber Optic Broadband (NFOB) or the OSN National Broadband Network (NBN). The Voice communication is exclusively routed through the PSTN and all office workers communicate with each other either by the exchange of emails via the corporate network or exchange of calls through the Public Switched Telephone Network.
From the view of this architecture, the research question 1, 2, and 3 required that we define an input of integration. The Corporate Router happened to be the most suitable entry point. The Corporate Switch was used to create the required links with the VoIP Dual-Stack Gateway as shown in Figure 5.

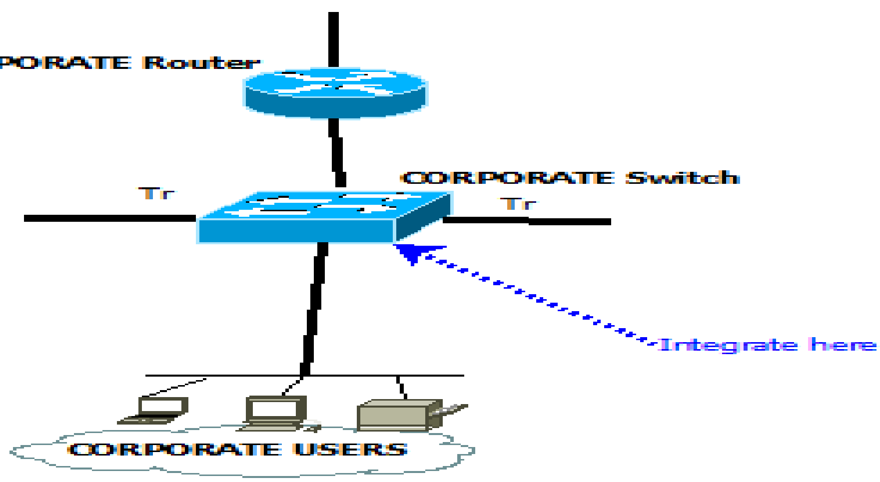
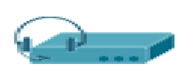


Figure 5. Topology of Integration Point

VOIP DUAL – STACK DESIGN INTEGRATION

The aim of our work was to integrate a VoIP system that will provide telephony services in the existing IPv4 data network and give priority to IPv6 traffic forwarding. Voice over IP networks can be challenging to implement efficiently and securely. This is in part because voice packets require specialized management and must be treated differently than normal data packets. Table 5 below report on the various materials used to design the solution in Cisco Packet Tracer version 7.2. The goal here was to conceive the entire architecture on the simulation software by adding all network entities describes in Table 5, and connect all of them according to how that has to be done in case of physical implementation. It considers representing all network elements according to their function at each level of the network hierarchy (AL, DL, CL and AL). When we segment a network, we divide it into multiple smaller networks, each acting as its own small network called a subnet. In Table 6, the logical segmentation per network entity is presented.

Network Level	Virtual elements	Descriptions	Representations
ACCESS LEVEL (AL)	**Virtual computers (Desktop and Laptops)**	Used to represent Workers PCs and Laptops positioned at their desk. Configured both on IPv4 and Ipv6 to access existing Corporate services over Ipv4 and forward new Ipv6 traffic. Run Softphones to access VoIP services	 
	Virtual PSTN Phones	Used to represent existing and future Analog POTS phones connected to the PSTN of Camtel.	
	Virtual VoIP Media Access	Used to provide connectivity to the Ipv4 to analogue phones	

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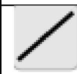
	Virtual Hubs	Used to represent Hubs providing layer connectivity to the IP data network	
	Virtual Softphones	Used to represent Cisco IP softphones that are installed on computers to Provide VoIP connection to workers	
	Virtual STP cables	Used to provide network connectivity to the various network entities	
DISTRIBUTION LEVEL (DL)	**Virtual Switches**	Used to represent Switches used at the distribution level and provide connectivity to access devices and core devices at the upper level	
CORE LEVEL(CL)	**Virtual Routers**	Used to represent the Cisco VG200 Router that provides IP telephony services to network users. This is the VoIP Media gateway configure to integrate the Corporate Router. Also, Cisco routers devices running Unified border elements are used to provide an interface between the NGN, the PSTN and the IP data network for management of signaling protocols and audio/video call sessions	

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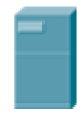
	Virtual L3 Switches	Provide L3 connectivity for inter-VLAN management.	
	Virtual NGN Switch	Used to represent the existing Huawei HONET UA5000 that provides PSTN phone Dial Tone (SS7). This is considered for future development	
Service Level (SL)	**IMS Servers (applications, VoIP, MS, etc.)**		

Table 5. Material used in the simulation environment

The logical segmentation of IPv4 and IPv6 addresses as presented in Table 6 also shows clear traffic separation of Voice and Data. IPhones and Softphones are attributed Numbers so that they can be able to call within the IPv4/IPv6 network domain. PSTN phones set also have attributed numbers. The logical IPv4/IPv6 segmentation creates separate subnet for VoIP users and Data Users. Such separation helps in providing better flow management and security to traffic.

Logical Domain	**SITES**	**IP SUBNET /CIDR**		**VLAN segment**	
		VoIP	DATA	Data	VoIP
IPv4 Subnetting	Bamenda	192.168.32.0/19	192.168.0.0/19	36	10
	Kumbo	192.168.128.0/19	192.168.64.0/19	37	11
	Nkambe	192.168.160.0/19	192.168.224.0/19	38	12
	Wum	192.168.96.0/19	192.168.192.0/19	39	13
	Bambili	172.24.200.62/27	172.24.200.0/27	40	14
IPv6 Subnetting	Bamenda	FC00:0:0:200::/56	FC00:0:0:100::/56	36	10
	Kumbo	FC00:0:0:700::/56	FC00:0:0:600::/56	37	11
	Nkambe	FC00:0:0:800::/56	FC00:0:0:900::/56	12	38
	Wum	FC00:0:0:3000::/56	FC00:0:0:4000::/56	39	13
	Bambili	FC00:0:0:1000::/56	FC00:0:0:2000::/56	40	14
IPv4 /IPv6 Inter-LINK					
IPv4			IPv6		

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Corporate-ITS-KUMBO BRAS	197.159.8.4 /30	2001:0:0:200::/64	
Corporate-ITS-NKAMBE BRAS	197.159.8.8/30	2001:0:0:300::/64	
Corporate-ITS-WUM BRAS	197.159.8.16/30	2001:0:0:500::/64	
Corporate-ITS-BAMBILI BRAS	197.159.8.12/30	2001:0:0:400::/64	
Corporate-ITS-UBE	195.24.194.0 /30	2001:0:0:600::/64	
UBE-NGN	195.24.194.4/30	2001:0:0:700::/64	
Routing Protocol	**OSPF, BGP**	**OSPFv3, Multi-BGP**	
IP service	IP telephony service		
Layer 2 Protocol	802.1q		
QoS	CoS, Tagging, mls		
Signalling	SS7-SCCP		
Security	Port security, Mac Binding		
WORKERS IP PHONES NUMBERS			
WORKER POSITION	**LOCATION**	**IP PHONE NUMBER**	**PSTN NUMBER**
SFR	BAMENDA	1000	233361000
SCS	BAMENDA	1003	233361015
RR	BAMENDA	1001,1024	233363434
SCO	BAMENDA	1025	233362025
SEC.RR	BAMENDA	1026,1034	233364040
AC KUMBO	KUMBO	2000,2020	NONE
PC NKAMBE	NKAMBE	2001,2021	NONE
PC WUM	WUM	NONE	233343520

Table 6. Logical segmentation per network entity

Table 7 shows the specifications of the network design implemented in the simulation environment. It considers specifying the communication type, the mode of operation, and the main communication protocols with their related complementary protocols. It also shows the signaling protocols use in this environment with the type of CODEC required for compression of voice signals.

Network Parameters	Description	Observations
Communication network type	Switched Communication network	Not Broadcast oriented except at Layer 2 of the OSI Model
Mode of Operation	Packet-Switched connectionless and Connection-Oriented	IP is Connectionless. When used for circuit establishment of VoIP sessions, virtual connection are required
Logical topology	Star, P2P.	Star at layer 2 to provide Multicast access to access devices and P2P for outbound traffic for WAN connectivity
Type of link	Fast Ethernet and Giga Ethernet Link	100/1000 Mbits data rate
Main communication Protocols	IPv4, IPv6	Best-effort delivery
IP Telephony Technology	VoIP	Digital technology
Complementary Protocol	Multicast IP, RTP, TCP, DHCP6, DTP, OSPF, OSPFv6, ICMP, ICMPv6	To provide real-time audio service and connection-oriented sessions of terminals belonging to different subnets with the routing of Voice packets over IPv4 and IPv6.
Signalling Protocols	SCCP, H.323	Skinny Client Control Protocol, light version of H.323 for voice session management, connection and tearing down of sessions
Standard Encapsulation	IEEE 802.1q (Dot1.q)	VLAN tagging
Service Provided	IP telephony over IPv4/IPv6	Cisco IP telephony service configured with activation of Dial Per numbers.
Number of network elements at the Access level	36	Comprises IP phones, Softphones, Mobile Tablet, Desktops, Laptops
Number of network element at the Distribution level	5	Switches configured to provide VLAN segmentation, connection to access devices and forwarding of Voice and data packets for routing
Number of Network elements at the Core level	5 routers, 2 layers 3 switches and one Server.	The routers provide IP telephony service and traffic routing with QoS policy class.

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CODEC	G.729	To minimize bandwidth consumption
QoS Management	Traffic prioritization, Class of Service(CoS)	Minimization of delays and optimization of traffic
Average call connecting time	9 ms	Acceptable
Average system response time	8 ms	Acceptable
Simulation Duration	30 min	The testing procedure and confirmation of telephony service availability in the inter-network domain

Table 7. Design parameters of the network

IMPLEMENTATION

A major aim of our work was to provide an operating environment for VoIP that allows calls between IPv4 and IPv6 user agents. The various parameters defined in Table 6 and Table 7 was used to implement the Dual-Stack VoIP solution. Figure 6 shows the Physical Topology of the Solution. IP telephony service was implemented in the Corporate Router to provide Voice services to users (SFR: Service Finance and recovery, SCS: Service Control and Security, RR: Regional Representative, SCO: Service Commercial, AC: Commercial Agency, PC: Commercial Point with extension towards remote site within the same management domain – Kumbo, NKAMBE, WUM, BAMBILI). Depending on the sensitivity of the job position, the user may have a softphone, a VoIP phone and the traditional PSTN set (example the SCS who has Softphone: Number 1023, IP phone Number 1003 and PSTN set Number 233361015).

In a nutshell, the VoIP architecture, presented in Figure 6 consists of the following components: Access terminals that are used to initiate/receive VoIP calls within the IPv4/IPv6 domain, Switches that provide traffic tagging, segmentation of users by function, traffic classification and Routers that deliver telephony services with the use of the SCCP/H3.23 protocols and QoS management. The Cisco Unified Border Element interfaces NGN Softswiches and provides connectivity between the PSTN and the IP network at the core level. All network interfaces belonging to the various network entities are configured with two protocols stacks (Dual-Stack): IPv4 and IPv6 and contribute in providing addressing of node either for IPv4 only traffic, or IPv6-only traffic or both IPv4 and IPv6 traffic. The architecture also includes various user agents. In our view, we have adopted an overall network architecture that meets the study objectives and the devices chosen will not only optimize voice

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traffic, but also effectively serve the converged network as a whole if properly implemented.

TESTING AND DISCUSSION

To ascertain the feasibility of the proposed architecture a testbed (simulated in Cisco packet tracer) was deployed and several interoperability tests were executed. The results needed to be convincing enough for the solution to be adopted in the IPv4 data network of Camtel and impact worker's productivity.

Dual-Stack Configuration of a Host

Taking into consideration the Dual-Stack design chosen as a solution to integrate VoIP and IPv6 in the existing IP4 data network, the researcher found that any host that has to forward traffic over such a network environment must have a compatible TCP/IP stack that can permit IPv4/IPv6 configuration (Figure 7).

PC1020 belonging to SFR received IPv4 and IPv6 addresses from the DHCP pool 192.168.0.0/19 with default gateway 192.168.31.254, and, DHCPv6 pool FC00:0:0:100::/56 with default gateway FC00::100:192:168:31:254. PC1020 forward IPv4 data traffic with VLAN tagging 36 (IPv6 subnet FC00:0:0:200::/56 for Voice and subnet FC00:0:0:100::/56 for Data), and IPv4 Voice traffic using IP phone 1000, with VLAN tagging 10. The 233361000 number will be used by SFR to call IP phone 1000 and any other number within the intra-network domain.

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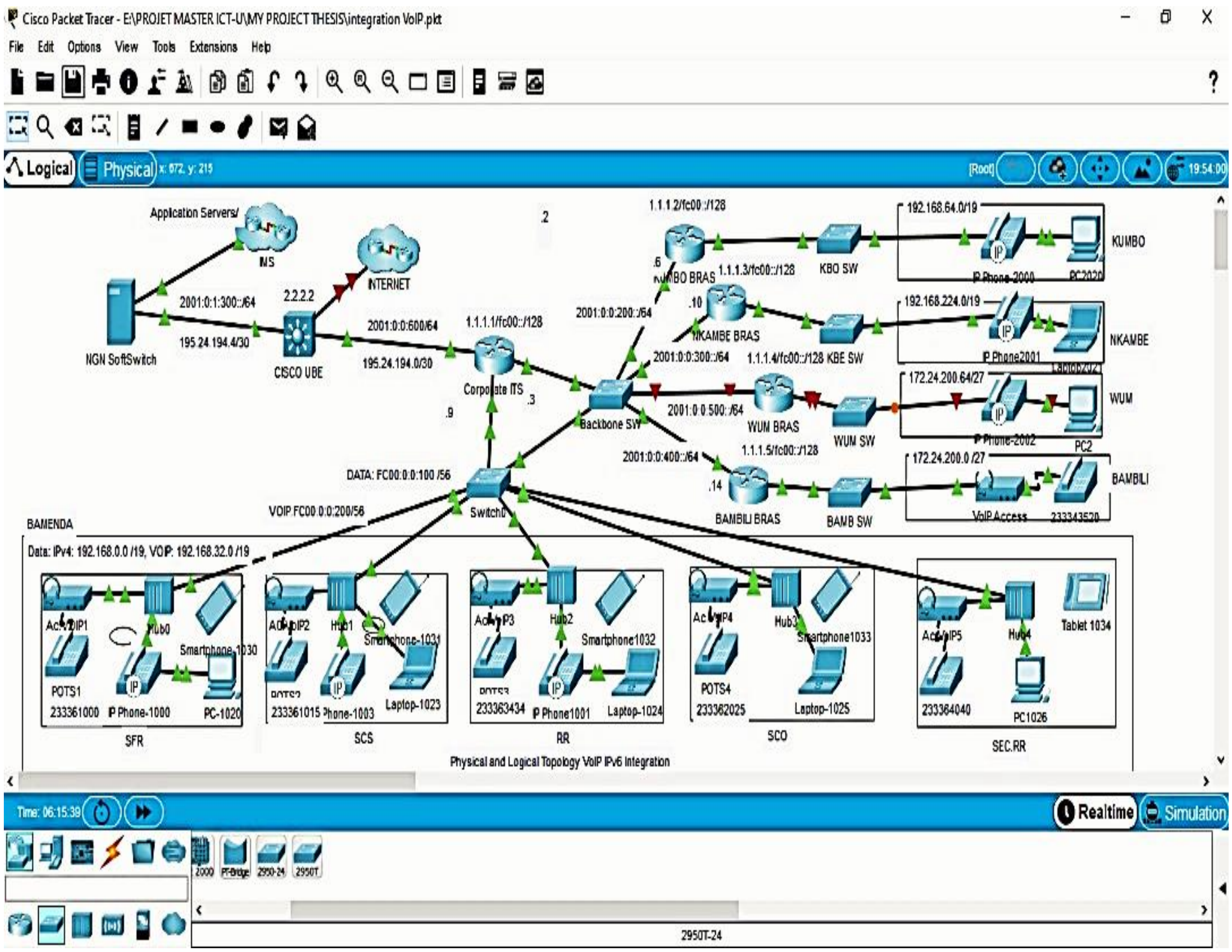

Figure 6. Implementation of VOIP Dual-Stack Configuration (Researcher Design)

Also, just as it is with Pv4, IPv6 subnetting is very important if good a structuration of the network is to be achieved. It means allocating proper IPv6 addresses according to their function (unicast, unique local, or global). The researcher, for example, Table 6 shows how he succeeded in this exercise. The output of ends terminals configuration is presented in this section showing the various captures collected from the simulation platform.

Figure 7 shows the configuration received by an IP phone connected in the voice Vlan 10, data subnet 192.168.0.0/19- FC00::100 /56 from the ITS VoIP router gateway that acts as the Call Manager.

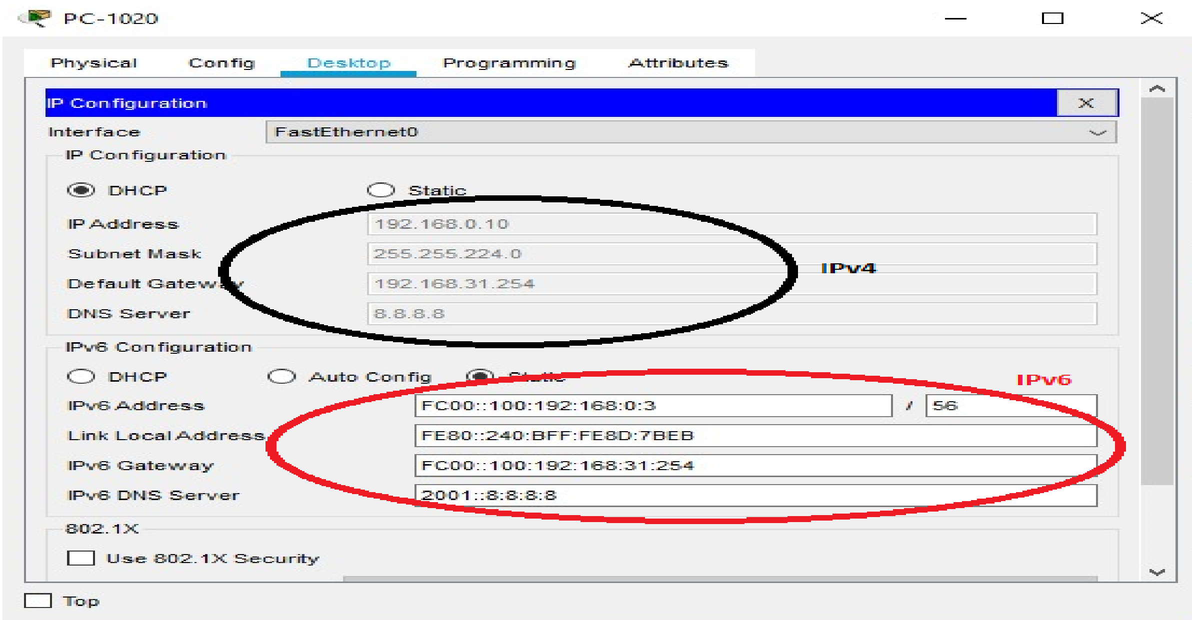

Figure 7. Dual-Stack configuration of PC running VoIP Softphones (Cisco packet Tracer 7.3 - Researcher design)

The off-hook phone with number 1000 has received the Dial tone from the SCCP (Skinny Call Control Protocol also known as Signaling Connection Control Part) Cisco communication protocol, which is a small version of the H.323 protocol responsible in call settings over IP through TCP port 2000.

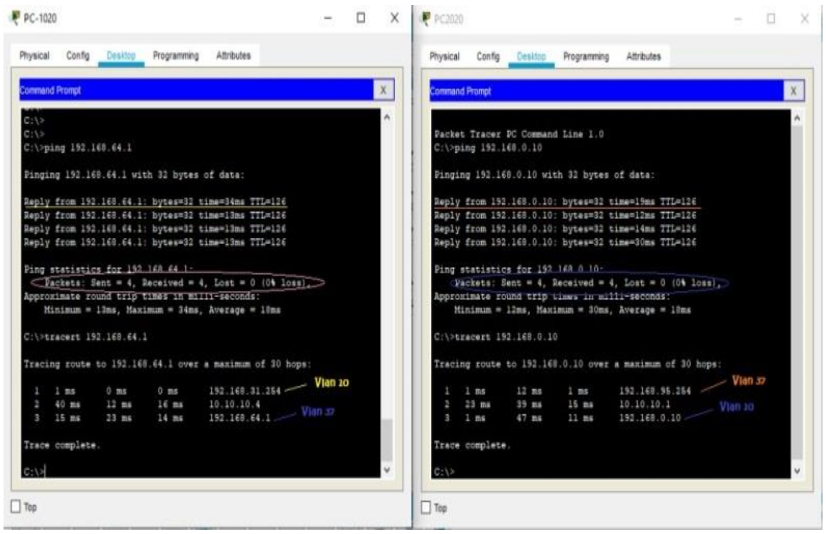

Figure 8. IPv4/IPv6 INTER-VLAN reachability

Testing IPv4 reachability in the Inter-VLAN

Figure 8 shows an Inter-Vlan Hosts reachability test between Vlan 10-192.168.0.0/19 in Bamenda and Vlan 37 192.168.128.0/19 in Kumbo. It means a terminal belonging to subnet 192.168.32.0/19 (FC00:0:0:200::/56) can call another terminal 192.168.128.0/19 over IPv4 or FC00:0:0:700::/56 over IPv6. Reachability within these subnets is confirmed when Terminals receive IP addresses and mapped accordingly with MAC addresses of terminals. The Corporate IP Telephony Service (ITS) which is the VoIP router gateway that acts as the Call Manager is successfully activated and attributes the Phone number according to the register ephone in the directory. The Dial tone is provided by the SCCP (Skinny Call Control Protocol also known as Signaling Connection Control Part), a Cisco communication protocol, which is a small version of the H.323 protocol responsible in call settings over IP through TCP port 2000. A fully operational IP phone is shown in Figure 9.

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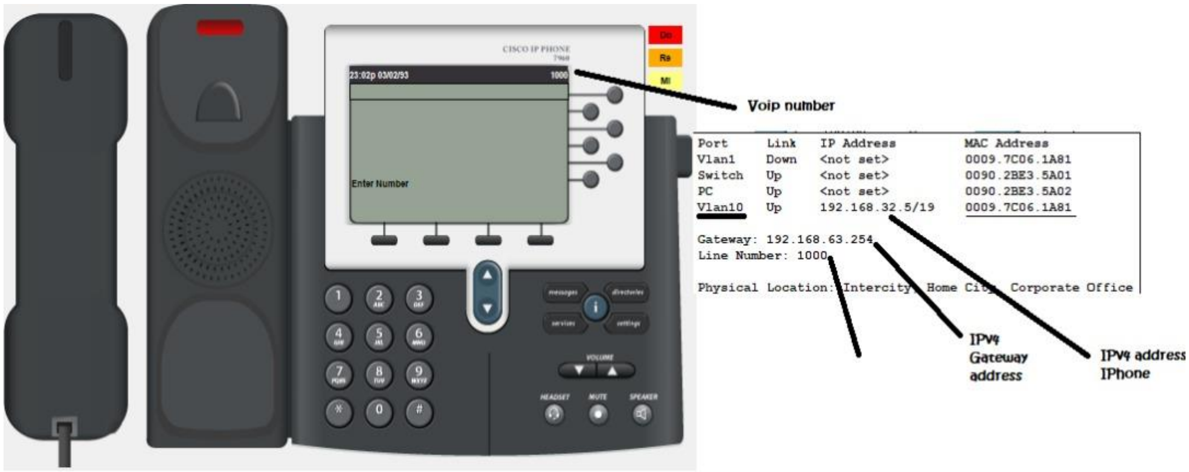

Figure 9. Output of a configured Cisco IP phone ready to make calls over Ipv4/IPv6 (Researcher, 2020)

Testing IPv6 Reachability

In figure 10, to confirm the proper reachability, the researcher conducted a Traceroute from a host to a subnet FC00:0:0:900::/56 and a host in subnet FC00:0:0:600::/56 via the Corporate ITS gateway configured with a global public gateway 2001::200:197:159:8:6. The researcher could see that the host FC00::900:192:132:31:1 belonging to VLAN 12 in Nkambe could be reached after 3 hops. Also, IPv6 packets were routed over 3 hops using OSPFv3 to reach a host in Kumbo under VLAN tagging 37. As an illustrative example, Fig.8 depicts the exchange of messages between the various entities when an IP phone places a call to an IPv6 softphone.

Testing Reachability between IPhones and PSTN phones

Here, a POTS phone with number 233361000 made a call towards an IP phone with a number 2000 found in a remote site at KUMBO and belonging to subnet 192.168.64.0 / FC00:0:0:700:: on Vlan 11. The researcher could observe the 2000 IP phone set rang. This means that if with the SCCP protocol, the analogue phone can communicate with IP phone over IPv4/IPv6 protocol, and then it can be more effective when using call manager running SIP and communicating with SS7 on the PSTN side and H.323 protocol on the IP side. Also, voice traffic forwarded on dual interfaces confirms that VoIP can operate over IPv6 without affecting the data

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services running over IPv4. As an illustrative example, Figure 11 depicts the exchange of messages when POTS phone with number 233361000 made a call towards an IP phone with a number 2000 found in a remote site at KUMBO

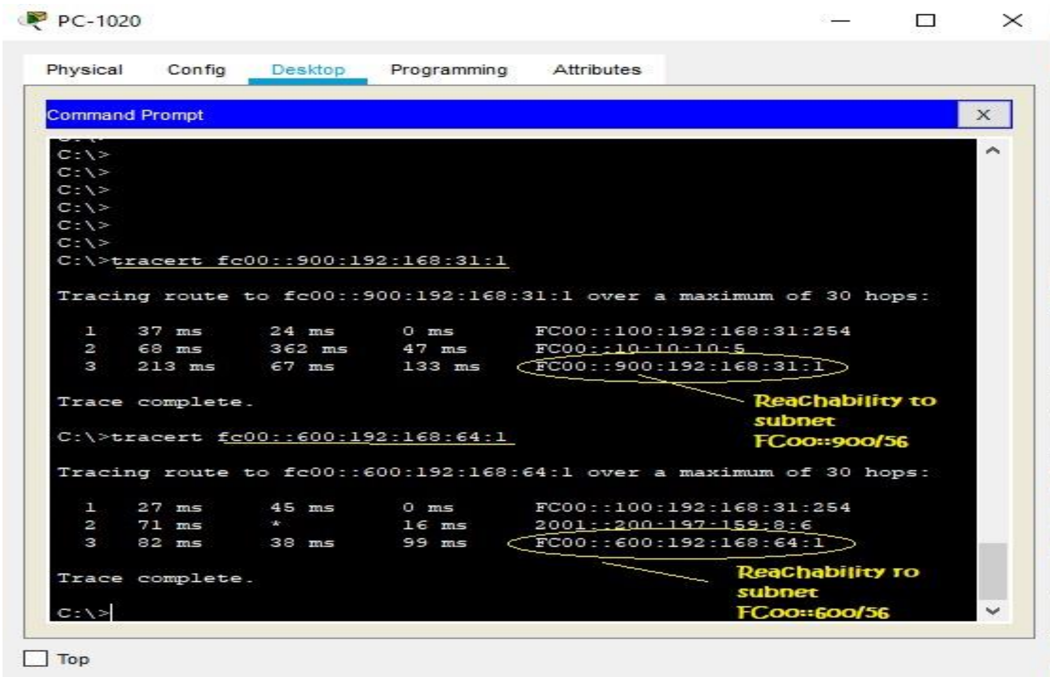

Figure 10. Tracing IPv6 Traffic

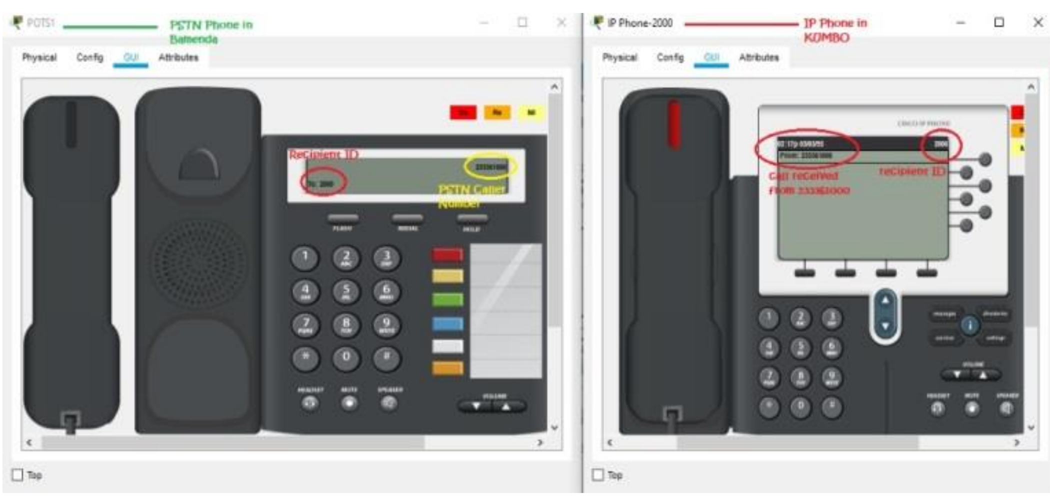

Figure 11. Calling test from IP phone to PSTN and Vice-versa

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Testing QoS parameters

QoS (Quality of Service) is the idea that transmission rates, error rates, and other characteristics can be measured, improved, and, to some extent, guaranteed in advance. QoS technologies allow you to measure bandwidth, detect changing network conditions (such as congestion or availability of bandwidth), and prioritize or throttle traffic. Basically, the quality of VoIP services consists of delay, delay jitter, and packet loss. Service on the simulated network must be very sensitive to these three QoS parameters. The QoS parameters tested can be categorized into three groups.

- The first is a **test related to Timeliness** where parameters such as delay, Jitter and response time are tested. Due to the simulation environment, the researcher could not be pronounced on the system response time. But delay particularly was monitored in term of appreciating the time taking for the gateway to attribute a tone and the time the called phone takes to ring during a calling session. It was satisfactory. Jitter here was not observed.
- The second is a **test related to Bandwidth** where parameters such as Systems-level data rate, Application-level data rate and Transaction time. Here, the use of 100/1000 Mbits links and interfaces ensure us sufficient bandwidth for call Voice and data transfer. A further test could be conducted by saturating the link with overflow traffic and observing Call quality, but this was not carried out.
- The third is a **test related to Reliability** where parameters such as packet loss rate, system response time, Mean time to failure (MTTF), Mean time to repair (MTTR, Mean time between failures (MTBF), and Percentage of time available. Packet loss is a parameter that describes a condition that indicates the number of packets that lost when communication occurs between users. At this point, the researcher focus on observing that packets loss rate is less than 1%.

The use of network tools such as Ping or Traceroute helped the researcher in the simulation environment to test connectivity, response time and packet loss rate between end devices. The purpose of optimization - the amelioration of the quality of voice by minimizing latency, Jitter, packet loss once implemented can give room to further research that focuses in evaluating VoIP performance and how to improve it in the Camtel IP data network. The research, nevertheless, gave special attention in ensuring that the system response time or latency should be less than 20 ms while keeping packet loss rate to less than 1%. Just these two considerations in the simulation platform help to enhance connectivity and call routing between VoIP terminals as it is recommended for implementation of a VoIP system.

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CONCLUSION

The PSTN and the IP telephony Network are two different platforms that provide almost the same type of services. The limitations related to analogue systems may contribute to believing that so far IP telephony is built over digitalized network entities, it is the best compare to the PSTN and the services this network can offer. However, this research did not intends to evaluate deeply which of them is best, but rather emphasize on benefit provides by IP telephony and show how this can contribute to ameliorate workers' communication potentiality and their productivity at work.

Our realization of the architecture proposed has resulted in a fully functional IPv4/IPv6 VoIP system. Research question Q1 was satisfied by providing appropriate IPv4 and IPv6 subnet plan for Local Hosts (corporate users), nodes (routers and switches) and inter-links (Corporate ITS to other remote Routers and Internet), activating IPv6 services (DHCPv6), functionalities and routing (OSPFv3) on all Layer 3 participating nodes, VLAN tagging 802.1q for IPv4/IPv6 user segmentation.

Research question Q2 was also answered. Considering the type of application running the corporate data network and the results of the tests related to bandwidth, it appears that the Corporate ITS router must have upgraded interfaces (1000 Mbits) and all his physical transmission Uplinks and Downlinks must be upgraded from 100Mbits to 1000Mbits bandwidth, especially those connecting the Corporate ITS router to the corporate switch and other Point of Presence. This will maintain the acceptable QoS. If not, the integration will degrade the existing QoS considering that existing interfaces have 100 Mbits capacity. Since the ideal solution is to avoid network interruption, a proper switch-over plan is necessary during this phase, transferring services from the previous Interfaces to the new ones. This implies that at a certain point, a physical cut of service will occur. Strategies to minimize the time this will take is the key. It means preparing all necessaries configurations and physical links ahead and implementing them according to the prepared plan. This will be the only way to ensure acceptable QoS for existing applications running IPv4 and, new applications that will run over IPv6.

Research Question Q3 was answered by providing proper configurations of both voice and data flows, giving priority to VoIPv6 routing for voice traffic while IPv4 routing keeps running the existing services and applications. Since all network entities are considered IPv6 capable, VoIP terminals are default configured with IPv6 priority, so calls will be by default forwarded over IPv6 and this will ensure that voice traffic should be forwarded without affecting the existing IPv4 traffic. Looking at Research Q4, the data analysis of interviews focused on people who have an acceptable knowledge of Technology in general and VoIP in particular, revealed that this solution will be welcome. And, the benefit provided by digital

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systems over analogue ones is undoubted, in terms of stability, scalability, and efficiency. Integrating such a solution with IPv6 was an initial step for Camtel to start thinking about the migration of its services and applications towards IPv6. The results obtained from the simulation, and the ability of both protocol to operate to provide voice service over digital infrastructure with the capability to link to the existing analogue system, confirmed that yes, this solution will be an effective alternative to the existing PSTN.

Finally, we answered Research Question 5 by implementing traffic prioritization, Class of Service (CoS) at layer 2 with priority setting of Voice over data, after planning and definition of VLANs for separation of Voice and Data traffic. But these configurations were not sufficient as we kept observing call connectivity delay and high latency when both services traffic are running. With the simulation on cisco packet tracer, after we replace the equipment with more upgraded ones, we could confirm that it is a must that in this architecture, we need to be mindful of the specifications of VoIP participating equipment from the access, distribution and core level. A focus should be made on high Processing, sufficient memory for delay minimization, and sufficient bandwidth for better throughput and connectivity. This will efficiently address delay issues and satisfy the delay-tolerant factor.

The need for sufficient bandwidth is real. The logical requirement shows that separation of voice and data traffic and application of traffic prioritization is one of the fundamental conditions if we must achieve a good QoS both for Data and Voice traffic. Besides, the proper configuration of network entities at all level of the network hierarchy is very important and none is to be neglected. Also, for this solution to be fully optimal, training sessions must be conducted to provide a smooth transition from the analogue used to a digital communication system that comes with multiple new features.

From the result of quality scores of VoIP service that have been designed and implemented, we can conclude that IPv4 and IPv6 can successfully operate simultaneously and the integration of new services can be done so far a good logical plan of the network is conducted. It means segmenting users by functions and type of service and applying proper addressing on network entities. When the integration design is well-conceived, a regular analogue phone can forward voice signal to a digitalized network and successfully place calls that can connect to the IP phones. Also, telephony applications such as Softphones that are connected to the IPv4/IPv6 network can connect to the PSTN and exchange voice signal with POTS phones. VoIP happened to be effective according to this simulation and convincing enough to act as an alternative solution to the traditional PSTN.

The observation of IPv6 operation reassures the researcher in term of better throughput compares to the IPv4. The flexibility in which the various phones can easily communicate with each also confirms that this solution can contribute to enhancing workers' productivity. In conclusion, to experiment and understand

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the role which IPv6 will play in the Cameroon's communication infrastructure future, it is necessary for us to develop hands on experience with the IPv6 technology. Through our effort in creating a Dual-Stack network, we have develop some level of expertise and become technically competent with IPv6 technology in an academic environment. We have also been able to discover the basic of IPv6 technology and implementation of dual-stack mechanisms. It also gave us the opportunity to test and understand the IPv6 technology. This project could be applied to other organizational setting which intends to implement IPv6 in their network interconnection.

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APPENDIX 1: INTERVIEW FORM

Research Topic: INTEGRATION OF A VOICE OVER INTERNET PROTOCOL(VoIP) SOLUTION USING INTERNET PROTOCOL VERSION 6 (IPv6) IN AN INTERNET PROTOCOL VERSION 4 (IPv4) DATA NETWORK TO INCREASE EMPLOYEE PRODUCTIVITY

INTERVIEW Nº___________________

Once again, please permit me to thank you for being willing to participate in the interview of my research. As I have mentioned to you before, my study intends to propose an IP telephony service that relies on IPv6 and that can act as an alternative solution to the PSTN of Camtel North-West region. The study also seeks to find out how impactful such new service, once integrated into the working environment, can enhance the productivity of workers. Our interview today will last approximately 30 min, which I will be asking questions segmented into three groups: the interviewee(you) and its job details, the appreciation of the IP data network and possible need of an upgrade, and finally your appreciation of how making use of technology can improve productivity.

You earlier completed a consent form indicating that I have your permission(or not) to audio record our conversation. Please, are you still ok with that YES____/No ____.

- *If YES Thank You! Please let me know if at any point you want me to turn the recorder off or keep something you said off the record.*
- *If NO, Thank you, I will only take notes of our conversation*

I- SECTION A

Questions related directly to you and your job(Time: 5 mn)

Subject	Researcher interview Questions	Researcher keynotes to Respondent answers
Interviewee's Education Level	Please, you are an adult male/female with what level of education?	
Number of Years in Service	Please, Can you tell me how many years of service are you now?	
Professional domain	Please, you are operating in which domain in camtel Northwest? (Commercial, Technico Commercial or Technician/Engineer)	
Familiarity with the ICT tool	Do you deal with ICT Tool(computer, etc)? if yes, please can you specify some and tell me more about your use of these tools in your daily activity	
Familiarity with computer networking applications	Do you have any notion of computer networking applications? What do you know about the internet and internet applications?	

II- SECTION B

Questions related directly to Your IP Data Network(15 min)

Subject	Researcher interview Questions	Researcher keynotes to Respondent answers
Interviewee's domain of operation and speciality	1- In which domain of computer networking are you specialize? 2- Does this domain deals directly with the Internet and Internet protocol? If YES, how? Please elaborate more on this.	
Discussion on the Internet Protocol and	1- What do you know about the Internet Protocol and how do you characterize its role on IP data networks?	

III- SECTION C

Questions related to employee productivity(10mn)

Subject	Researcher interview Questions	Researcher keynotes to Respondent answers
Discussion related to productivity at the job position	1- Please do you have a PSTN terminal on your desk? If yes, can you elaborate on how useful it is for your daily job routines? If not, why? 2- Do you think this PSTN terminal contributes to making you productive in your job position? What features purposely do you consider important enough to make you more productive? 3- According to you, do you find the PSTN network satisfactory enough? Why? 4- If you have the opportunity to use a different system that is purely digitalized will you go for it? If yes, what are the features will you expect? 5- If we consider a scenario whereby these 2 systems (PSTN and VoIP) operate simultaneously, how much do you think this can be interesting and impact on your productivity? how do you think such a design can help you to do your job better? 6- Finally, do you agree or not that integrating VoIP in the IP data network of Camtel using IPv6 will impact on your productivity	

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APPENDIX 2: CONCERN FORM

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CONSENT FORM

RESEARCH STUDY INTERVIEW DETAILS:

- CONSENT TYPE: ORAL CONSENT
- PARTICIPANTS: 12
- AGREEMENT: CONFIDENTIALITY PRESERVED

STUDY TITLE: INTEGRATION OF A VOICE OVER INTERNET PROTOCOL(VoIP) SOLUTION USING INTERNET PROTOCOL VERSION 6 (IPv6) IN AN INTERNET PROTOCOL VERSION 4 (IPv4) DATA NETWORK TO INCREASE EMPLOYEE PRODUCTIVITY

INVESTIGATOR: **Malobe Lottin Cyrille M.**

BACKGROUND AND PURPOSE: You are being asked to take part in a research project, which is being organized by Malobe Lottin Cyrille, A master 2 student at the ICT University. The purpose of this research project is to collect enough information that can help the researcher to satisfy the research questions related to the integration of VoIP in the IP data network of Camtel using IPv6. My research is contributing to a greater assessment of how Camtel North-West can make use of the benefit provided by the latest version of the internet protocol to integrate new communication services that will limit their dependency to the PSTN and by so doing, improve the productivity of workers.

PROCEDURES: The format of the interview will be a discussion between you and me. I expect that the interview will take no longer than 30 mn. With your permission, I will audiotape the interview solely to accurately transcribe the conversation and analyze it for the reliability of data.

CONFIDENTIALITY AND RISK: There is some risk involved if, for example, you divulge confidential information. Therefore, if you wish pseudonyms or no name to be used to protect your privacy and confidentiality, I will be happy to do so. Alternately, if you wish to be quoted by name on anything, in particular, I am also happy to accommodate this request. Please know though that you do not have to answer any questions or discuss any topics that make you feel uncomfortable.

WITHDRAWAL OF PARTICIPATION: Should you decide at any time during the interview or discussion that you no longer wish to participate, you may withdraw your consent without prejudice. COSTS BENEFITS TO YOU: There are no direct costs involved with participation, although you may miss 30 mn of work and possibly pay for that time. There are also no direct benefits to you. However, your participation will contribute to a greater awareness of such evolvement and innovative solution.

SIGNATURE: I confirm that the purpose of the research, the study procedures, the possible risks and discomforts as well as benefits have been explained to the participant. All questions have been answered. The participant has agreed to participate in the study.

Signature of Person Obtaining Consent : ________________________________

Date: ____/____/________

The participant agrees to be audio-taped : YES ☐ NO ☐ Initial ______________

The participant would like his/her name to be used: YES ☐ NO ☐ Initial ______________

Witness Name and Signature ______________ Date: _____/______/_______