VoIP System Dimensioning the Radio-Links and the VSAT of the MINTEL School Connectivity Project Through the TELCONET S.A. Network

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Abstract. Due to the constant changes that technology offers, it has allowed us to integrate various services to telecommunication networks, to improve solutions and offer advantages in connectivity and remote access to places where you do not have access to networks, problems with cellular coverage have allowed us to look for alternative ways of communication and to improve education in rural and urban areas deployed in the connectivity project between Telconet S.A. and Mintel. For which it is appropriate to design a network and to deploy a robust VoIP system through radio links guaranteeing high transmission rates, so you can expand and better the services that contribute to the development of telecommunications in Ecuador. The proposed system provides technical features that make the system more efficient and allow the incorporation of quality of service protocols that provide optimum service to the user with imperceptible latency, as well as the possibility of permanent VoIP service availability through a backup system that is part of the contingency plan allowing to have universal access to information technologies and thus to the society of knowledge.

Keywords: Network access · Imperceptible latency · Connectivity project System backup

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1 Introduction

In recent years, the need to incorporate citizens from the most vulnerable sectors of the country into the information and knowledge society has become evident, which allows us to reduce digital illiteracy and provide more opportunities for people of limited resources, making them more competitive in the world of work.

By virtue of this, and to comply with the social responsibility that TELCONET S.A. has with the country, a contract has been signed between this company and the Ministry of Telecommunications and Information Society (MINTEL) that seeks to improve internet penetration in rural and marginal urban sectors of the province of Guayas, thus contributing With the increase in universal access to fundamental communication and information services, this contract aims to provide equipment to Computational Laboratories and National Level Connectivity Service.

These laboratories offer the possibility of accessing the internet from the schools that are included in the contract signed between MINTEL and TELCONET SA, however, because they are located in remote areas of the province of Guayas, should they arise. Interruptions in Internet service due to failures in the network, in most cases it is not possible to attend to the requirements of those affected immediately because there is no real-time communication between the schools and the staff. TELCONET SA technical assistance, mainly because in those communities there are no conventional or mobile telephony networks that allow them to report incidents in a timely manner. For this reason, solving a problem of this nature can take days and even weeks.

Under the current conditions it is not possible to guarantee availability of the 99.999% Internet access service in the educational units benefiting from the project led by MINTEL, due to the lack of an appropriate communication mechanism between schools and the service. Technician of TELCONET SA that allows to provide assistance in the event that failures arise in any device that conforms to the Computer Lab. The impact on the continuity of the internet service has an important impact on the normal development of the classes as well as on the fulfillment of the Digital Literacy Education Program of the beneficiaries of the project and therefore on the development of the communities involved [1].

For which we opted for the design of a VoIP infrastructure for the solution of the communication problem, using as voice transport a radio-link network [2] and a Vsat network [3], taking into account as the main point the design for VoIP infrastructure [4], and making the comparison in QoS.

2 VoIP and Its Elements in a System

2.1 VoIP

Voice over IP technology standardized through the H.323 specification issued by the ITU (International Telecommunication Union) it allows the voice signal to travel in data packets through IP networks in digital form.

VoIP/H.323's primary objective is to facilitate and ensure the interoperability between equipment of various manufacturers and establishes the aspects such as the suppression of rests, compression and addressing, and the establishment of elements that allow the interconnectivity with the traditional switched telephone network (PSTN). In the Fig. 1 shows the H.323 protocol tower [5].

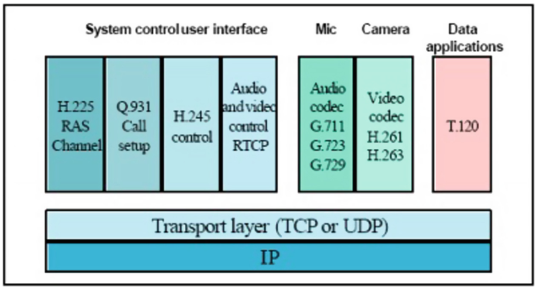

Fig. 1. H.323 protocol tower [5]

Figure 2 shows the H.323 architecture for terminal interoperability according to what is specified in the ITU2 standard [6].

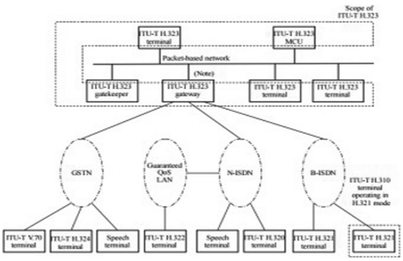

Fig. 2. Interoperability of terminals ITU-T H.323 [6]

2.2 Elements of a VoIP System

2.2.1 The Client

A Customer can be a Skype user or a user of a company that sells its IP telephony services through equipment such as ATAs (Analog Telephone Adapters) or IP phones or Softphones which is a software that allows calls to be made through a computer connected to the Internet [5].

2.2.2 Servers

Los servidores se encargan de manejar operaciones de base de datos, realizado en un tiempo real como en uno fuera de él. Entre estas operaciones se tienen la contabilidad, la recolección, el enrutamiento, la administración y control del servicio, el registro de los usuarios, etc.

2.2.3 The Gateways

The gateways are used to "terminate" the call, that is, the customer originates the call and the gateway ends the call, which is when a customer calls a landline or cell phone, there must be the part that makes that call possible. Through Internet, you can connect with a customer of a fixed or cellular telephone company [7] (Fig. 3).

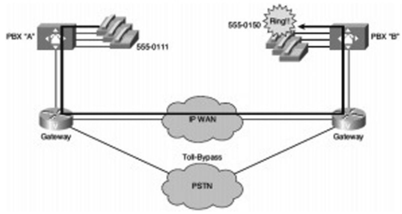

Fig. 3. Elements of a VoIP system [8]

2.3 Codecs

The voice must be encoded in order to be transmitted over the IP network. For this, codecs are used to guarantee the encoding and compression of audio or video for later decoding and decompression before being able to generate a usable sound or image. According to the codec used in the transmission, the required bit rate will be used. The amount of bitrate used is usually directly proportional to the quality of the transmitted data.

Among the most used VoIP codecs Are G.711, G723.1 and G.729 (specified by the ITU-T).

These codecs have the following coding bit rates.

- G.711: bit-rate de 56/64 Kbps
- G.722: bit-rate de 48,56 o 64 Kbps
- G.723: bit-rate de 5.3 o 6.4 Kbps
- G.728: bit-rate de 16 Kbps
- G.729: bit-rate de 8/13 Kbps

2.4 VoIP Latency

The latency is also called DELAY. This is not a specific problem of non-connection oriented networks and therefore of VoIP, rather it is a general problem of telecommunications networks. For example, the latency in satellite links is very high due to the distances that the information must travel.

Latency is technically defined in VoIP as the time it takes for a packet to arrive from the source to the destination.

Real-time communications (such as VoIP) and full-duplex are sensitive to this effect. Like jitter, it is a frequent problem in slow or congested links.

The latency or delay between the start and end points of the communication must be less than 150 ms. The human ear is able to detect latency of about 250 ms and

200 ms in the case of people who are quite sensitive. If that threshold is exceeded, the communication will fly annoying [9].

2.5 VoIP QoS

To improve the level of service, it has been aimed at reducing the bitrates used, for them we have worked under the following initiatives:

- The suppression of silences, gives more efficiency when making a voice transmission, since it takes advantage of the best bit rate by transmitting less information.
- Compression of headers applying the RTP/RTCP standards.

For the QoS quality of service measurement, there are four parameters such as bit rate, time delay (delay), delay variation (jitter), packet loss and echo.

For these types of drawbacks, three basic types of QoS can be implemented in the design.

- Best Effort: This method simply sends packets as they are received, without applying any real specific task. That is, it has no priority for any service, just try to send the packages in the best way.
- Integrated Service: This system has as its main function to pre-accord a path for the data that need priority, besides this architecture is not scalable, due to the amount of resource it needs to be reserving the transmission rate of each application.
- Differentiated Services: This system allows each network device to handle individual packets, and each router and switch can configure its own QoS policies to make its own decisions about the delivery of packages. Differentiated services use 6 bits in the Ip header.

3 Proposed Model for the Solution of the Problem

Within the solution the proposed solution to the problem of communication between schools and towards customer services of the company for damages that the data network may present, the following basic elements are analyzed:

- Determination of equipment requirements with technology such as Wireless Fidelity (Wi-Fi) and VoIP that will be supplied by TELCONET S.A. These equipment will be of high performance to ensure a high level of innovation in the last mile network.
- Determination of the technical characteristics of a local IP telephone exchange (the possibility of installing it in a computer of small dimensions type Microcomputer NUC will be evaluated). The innovative element is the use of a low power consumption computer with enough power to handle several phone calls simultaneously.
- Basic IP telephones for the installation of remote schools, a Gateway (Session) Protocol Initiation Protocol (SIP) for the interconnection of remote schools with destinations outside the project such as conventional telephone lines. In this way we ensure the possibility of crosses called to CNT in those schools where there is the possibility.

- Determine the technical characteristics of a router that will serve to make the dimensioning of the network dividing the Internet and Data services, also analyze the feasibility of installing a wireless controller Fidelity (Wi-Fi) being in charge of the centralized management of the access points, mesh architecture (Mesh) and user authentication. User traffic will not be tunneled from the access points to the controller, but will flow independently from them to their destination, minimizing the delay and supporting VoIP over WIFI.
- The installation of radio-relay equipment using Orthogonal Frequency Division Multiplexing (OFDM) brand Cambium Networks will be analyzed, carrying out point-to-point and point-to-multipoint links with a capacity of 50 Mbps Full Duplex and the different distances of said links with technical studies of the area of Fresnel and equipment configuration.

With this project direct communication between each of the schools will be allowe members of the MINTEL Connectivity Project. This is the main result. That is, obtain a better quality of communication between the schools and entities that intervene in the MINTEL Schools Connectivity Project, providing as added value the implementation of cutting edge telephony as are the VoIP networks that will help complement the excellent development carried out by the Connectivity Project.

In addition to providing the Ministry of Education with a practical solution for communication with the schools of the Project. This is a result important social: enable direct and quality communication elevated, between each of the member schools of the Project MINTEL connectivity and with other institutions. In addition, provide as an added value the implementation of VoIP that helps complement the excellent development of the Connectivity Project implemented by TELCONET S.A. and supported by MINTEL.

Submit to TELCONET S.A. a solution that could be offered to schools as part of their communication with destinations different from those mentioned in the Project. This is a result of business enterprise: install technologies designed, designed and fully implemented in Ecuador to present to TELCONET S.A. a solution that could be offered to schools as part of communicating them with destinations other than those mentioned in the project.

4 Solution Design

The design of the proposed solution requires the use of high technological equipment that guarantees the provision of a quality service, with imperceptible latencies to the end user, and quality standards that allow uninterrupted and efficient communication.

4.1 Research and Design

Prior to the selection of the equipment required for the implementation of the VoIP system, it is necessary to proceed with the elaboration of the respective design that fundamentally seeks to satisfy the minimum service demands determined by the contracting entity.

In order to meet the requirement, VoIP technology has been chosen to facilitate real-time communication between the administrators of the computer centers of the educational units and the TELCONET S.A. technical assistance center. In the present design, it concentrates exclusively on the technical characteristics of the communication network and on the quality of the transmission service.

To provide an IP telephony system, the installation of a telephone exchange based on VoIP protocols is essentially required. The entire telephone network would be centralized by a communications gateway that would provide the basic functionalities of a traditional telephone exchange together with other additional features.

The Gateway is based on the Session Initiation Protocol (SIP), which is a protocol of the session layer according to the OSI model (Open System Interconnection) [10] that facilitates signaling and allows creating, modifying and end sessions with one or more clients. Sessions include: telephone calls, multimedia data transfer and real-time conferences. Additionally, SIP allows the implementation of call routing policies in the system through the transport layer services of the OSI model. Figure 4 shows how the SIP protocol, located in the application layer of the TCP/IP model (Transmission Control Protocol/Internet Protocol), provides start and end services for voice and video calls at a higher level and for this purpose it is supported in TCP/IP transport layer protocols such as UDP (User Datagram Protocol), SCTP (Stream Control Transmission Protocol) and TCP [11].

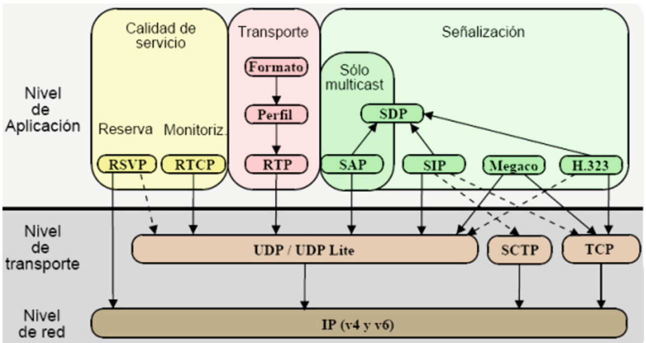

Fig. 4. Functionality of protocols involved in VoIP [12].

SIP makes use of network elements called proxy servers or SIP servers to help route requests to the user's current location, authenticate and authorize users.

From the foregoing it is evident that the first network element of the system design is the Gateway based on SIP protocol, implemented by a proxy server with built-in SIP functionalities.

Additionally, a router will be required that provides data routing capacity over the network, a switch that provides layer 2 services to clients, computers and IP telephones.

Figure 5 shows the design of the proposed IP telephone exchange, developed with the Packet Tracer ® software from Cisco Systems.

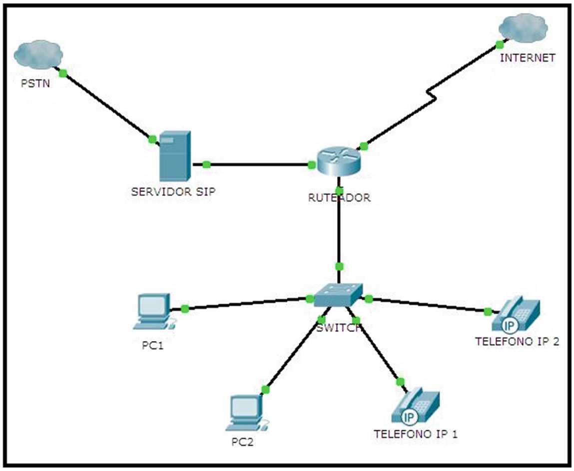

Fig. 5. Structure of an IP telephone exchange.

4.2 VoIP Phone System

In order to manage telephone resources based on IP telephony, TELCONET S.A has recommended the design of a virtual IPBX central developed by Denwa Technologies Corp. that allows managing more than 10 million users in its Data Center IP model [13].

In Fig. 6 you can see the operation interface of Denwa.

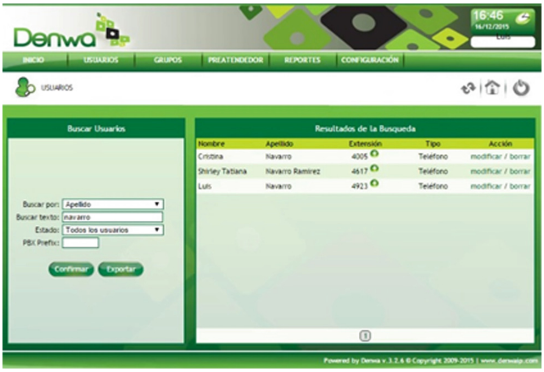

Fig. 6. Denwa operation interface [15].

Denwa is a highly versatile, totally graphic system that guarantees the efficient and continuous operation of the system using software tools, reducing the cost of implementation considerably without minimizing the quality of the service provided.

4.3 Choice of Codec to Use

The codecs have an important role in the design of our network, the audio codec used for incoming and outgoing calls are G.711u and G.711a codecs as can be seen in Fig. 7.

For the operation of the IP PBX, the G.729, G711u and G711a audio codecs have been chosen that provide the minimum quality characteristics required in such a way that the calls are made optimally and minimizing the network resources. The video codecs have been disabled, since it is not necessary because they are only enabled for IP telephones with video, that is, they are exceptional cases due to the high requirement in the network.

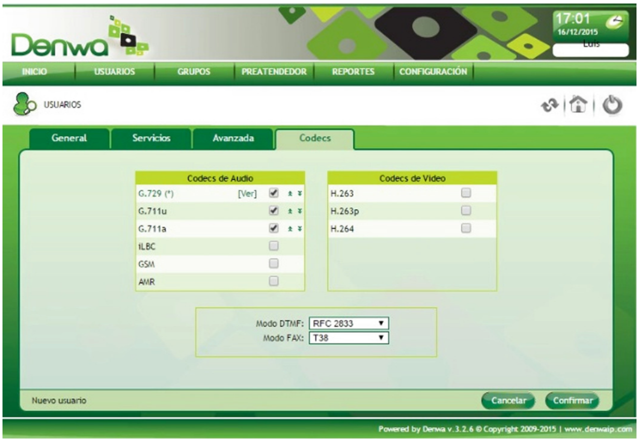

Fig. 7. Data from user codecs Denwa.

4.4 Radio Links

In order to carry out point-to-point and point-to-point links, Cambium Networks equipment has been used, which has MIMO and OFDM technology, the use of reflector type dish antennas has allowed us to reach long distances while maintaining high throughput and performance with low latency times. Moment of sending full duplex traffic. Dynamic adaptive modulation has allowed us in this project to maintain a high and robust modulation to the interference, guaranteeing the bit rate required by the MINTEL, with a simple configuration platform we can configure our links.

One of the most important advantages of Cambium Networks equipment is that it has 2 band options such as 5.4 GHz and 5.7 GHz, as having these bands has allowed us to have more alternative when choosing the best available frequency, prior to this election. Spectrum analysis has been carried out in the different repeaters of the project in order to prevent possible falls or intermittent links and ensure a better service [14] (Fig. 8).

Fig. 8. Cambium networks teams.

4.5 Fresnel Zone Studies

In order to design wireless links it is important to consider several factors that may affect our communication, among them the obstacles that may exist between the Master or Slave in the radio signals, and this is where you must perform calculations of the Fresnel zone, which it defines us an ideal value for a good wireless communication.

For the interconnection of the aforementioned schools, the establishment of radio links is required, which must provide minimum operating characteristics to guarantee the quality of the service.

Figure 9 shows the point-to-point and multipoint systems of the network in question.

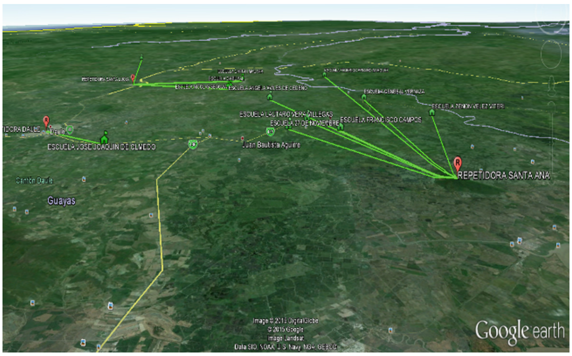

Fig. 9. Point-to-point and point-to-multipoint system.

In order to determine the feasibility of the radio links, the simulation is carried out where the technical characteristics of the equipment involved are taken into account, the topography of the environment and the propagation characteristics of the signal.

Figure 10 shows the topographic profile of the study carried out at the Lautaro Vera School belonging to the Salitre song to the Santa Ana repeater.

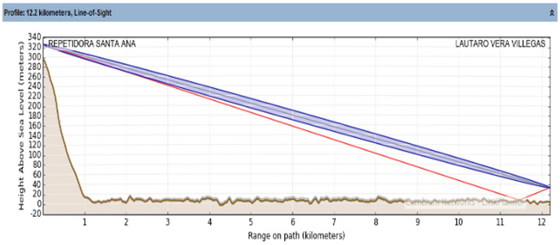

Fig. 10. Simulation of the link between Lautaro Vera Villegas School towards Node Santa Ana.

5 Platform and Diagram of Integration

5.1 Platform and Diagram of Integration of RF to VoIP

The integration platform between the terrestrial radiofrequency interface and the VoIP system is presented in the diagram of Fig. 11, as it can be seen, through wireless links of a system with point-multipoint topology, the telephone exchange is connected to TELCONET SA with the users, for this purpose it is made use of the towers and support structures previously described, as well as the necessary radiant elements to spread the information through the air interface, then the routing and network switching elements installed in each node they allow the reception and distribution of information towards each of the points.

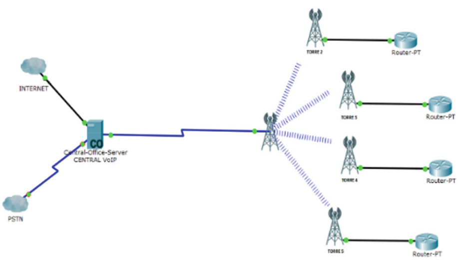

Fig. 11. RF to VoIP integration diagram.

5.2 Platform and Diagram of Integration of VSAT to VoIP

The integration platform between the satellite radiofrequency interface (Vsat) and the VoIP system is presented in the diagram of Fig. 12, as can be seen, through satellite link communication is established towards the points of each school, to its Once the information propagated by the space is received by a satellite that acts as a remote repeater, and then the signal is sent to the Hub, which is the earth station that allows to concentrate the information received and with the help of elements of commutation and routing, distributing it to the points of interest, based on this scheme the proposed system is technically operational and provides the minimum quality requirements to guarantee an optimal service with imperceptible latencies for the school.

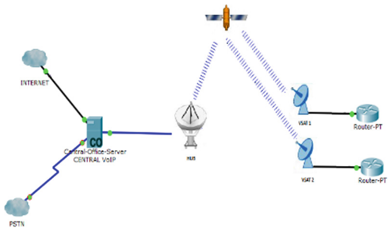

Fig. 12. Vsat to VoIP integration diagram.

6 Results of Radio-Link and VSAT

In order to observe the results obtained in the radio and Vsat links, a bandwidth test was performed occupying the WAN Killer tool, in which UDP traffic is sent with a rate of 10 Mbps, in this way we check if the link supports the traffic what is required (4 Mbps full duplex) for the Mintel project (Fig. 13).

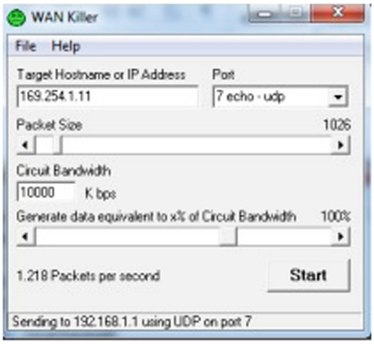

Fig. 13. Wan Killer menu.

We can see that the traffic of 10 Mbps is constant and does not have cuts or intermittences, this guarantees that when connecting our radio-link to the router and have real traffic consumption is not saturate or completely miss the communication causing intermittences, being able to support the bit rates that we need to establish a VoIP call (Fig. 14).

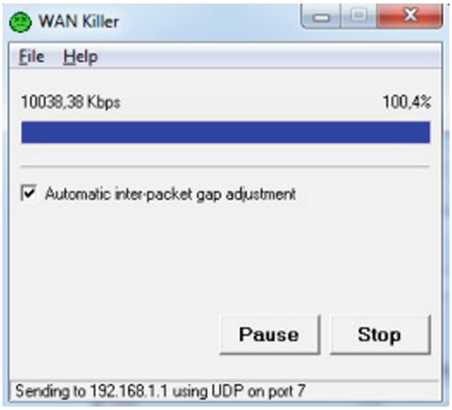

Fig. 14. Shipment of UDP packets.

7 Verification of Operation

In order to verify that the system is operating properly in the lower layers of the OSI model, it is necessary to use a network protocol analyzer software that is capable of reproducing the.cap extension files generated by the Denwa telephone exchange, for of this thesis makes use of Wireshark software, the same that is widely used as "sniffer" in the academic and professional field fundamentally for its reliability and versatility, as well as being a GPL-type license program that facilitates its installation and operation.

The Wireshark captures all the packets and frames of the different levels of the OSI model and shows its content and technical details, this software allows filtering according to the requirements of the analysis or verification required. Figure 15 shows the information of the cap file generated by the Denwa sniffer during the establishment of a call through the 3CX softphone.

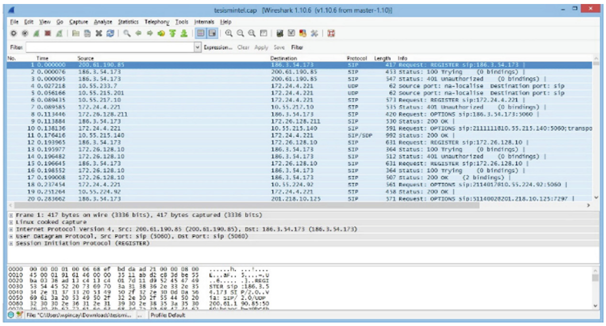

Fig. 15. Package capture in Wireshark.

As you can see, during the lapse of packet captures, all correspond to the SIP service of IP calls, when you open the VoIP call information window it is verified that the system has registered the calls made to the user that was taken as an example to the school "Pueblo Nuevo" and that has been processed by the PBX "tesismintel" as can be seen in Fig. 16.

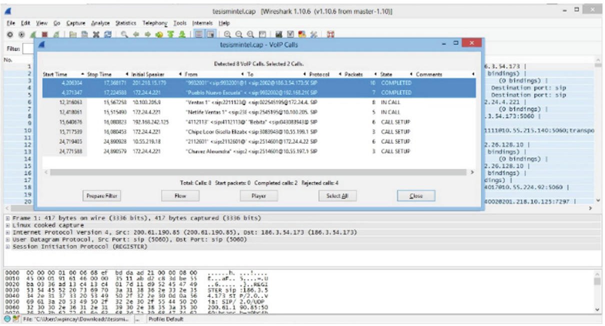

Fig. 16. VoIP call log in Wireshark.

The Wireshark allows visualizing the graphical packet analyzer, which facilitates observing the start, establishment and end of the VoIP call session, as can be seen in Fig. 17.

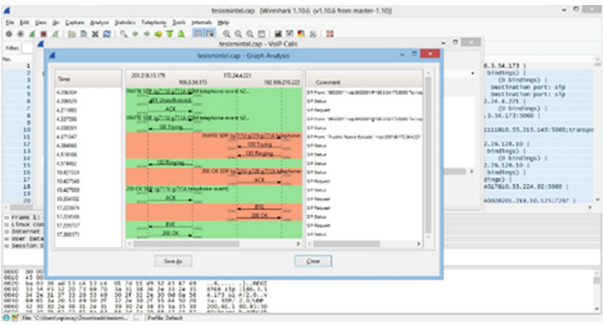

Fig. 17. Wireshark graphic analyzer.

As shown, the system works according to plan, verification of the operation of the system has been made using the Denwa telephone exchange, the 3CX softphone and the Wireshark network protocol analyzer that together have provided optimal results that guarantee the correct functioning of the PBX.

8 Conclusions

The system proposed in the present design, based on the results obtained from the network design simulations, once implemented, would provide a considerable improvement in communication between the schools that make use of it, and the customer service center of TELCONET SA, this improvement is not only due to an optimization of the service presented by TELCONET SA that will allow timely reporting of network failures or problems, but will also provide a substantial improvement in the technical quality of the network, by improving the design characteristics of the proposed network and by incorporating service quality protocols, a very accurate perception is guaranteed superior in the quality of voice transmitted through the network.

In dimensioning the network with Cambium Networks equipment for the point-to-multipoint link of the network design, several advantages were evidenced that potentiate the proposed system, among the main ones being the cost-benefit ratio of radio frequency equipment, since they provide considerable advantages over their competitors in relation to market prices, additionally provide guarantees and availability of permanent spare parts and finally, by making use of frequency bands allocated in the National Frequency Plan on a secondary basis, the cost for monthly payment of rates of use of the radio electric spectrum to the State.

The proposed telephone exchange Denwa Technology provides important features among which are its ease of use, the system is very user friendly, thus facilitating the recurrent use of these resources as well as the maintenance and configuration of the plant, the it is scalable, so it can be easily projected to a larger VoIP network by making small changes in its configuration and technological infrastructure.

Finally, a backup system has been designed that would take part in the Network Contingency Plan. This backup will guarantee the continuity of the service in emergent cases in which, due to force majeure, the main network is out of service.

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