A Problem-based Learning Case Study for Teaching Voice over Internet Protocol - VoIP

Using Asterisk as a Tool for Teaching VoIP for Information Technology Classes

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Keywords: Problem-based Learning - PBL, Asterisk, Voice over IP - VoIP.

Abstract: This paper shows the use of PBL (Problem-Based Learning) technique as a key to learning VoIP in courses like Electrical Engineering and Computer Networks in conjunction with open source and the public domain software called Asterisk which was used to create the scenario of the experiment and the problems presented to the students. In order to make the validation, the experiment was applied to students of Bachelor in Electrical Engineering and in Communication Technology System, undergraduate courses at the Federal Institute of Education, Science and Technology of Paraíba – IFPB, in the Telephony subject, with promising results. The Asterisk software was presented as a useful and flexible tool for constructing scenarios and problems for the teaching of VoIP technologies and the used approach resulted as effective for improving the attainment of the defined learning objectives.

1 INTRODUCTION

In the context of the technological evolution of communications networks, the reality that the telephone networks and data switching networks are converging to an infrastructure, that will allow both voice and data to be transmitted over the same network, is becoming more feasible for the technology and communication professionals on a daily basis. This fact affects professionals from the most diverse backgrounds, who work in the convergent network area, from courses in information technology up to the electrical and telecommunication engineering fields.

The convergence of data networks with telephone networks makes information technology professionals face the challenge of working with scenarios involving IP networks as well as with the existent infrastructure of the traditional telephone networks. Increasingly, the solution to this challenge is related to the use of "Voice over IP" (VoIP) technology in which the phone calls travel through a broadband connection instead of traveling by conventional telephone networks (Keller 2009).

In contrast to such development, the formation of most of these professionals is still based on traditional methodologies in which the teacher is the holder of knowledge and the mass production of labor force is prioritized.

However, methods have been changed and the universities´ great challenge is to provide training courses with the purpose to frame the content according to the student so that he/she can become a technically qualified professional in addition to being able to adapt to frequent changes and demands of the labor market (Silva & Viana, 2013).

With the change of teaching methods, an approach is proposed in which learning is based on practical problems solution (PBL). Such problems are applied in scenarios that simulate a company´s environment and allow students to develop and clarify, in practice, the theoretical concepts that were learned in the classroom, but that were formed in an obscure and abstract way (Fernandes, 2013).

According to Ali and Samaka (2013), problem-based learning is a student-centered, self-directed, inherently collaborative pedagogy where students learn by working in groups through solving

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A Problem-based Learning Case Study for Teaching Voice over Internet Protocol - VoIP - Using Asterisk as a Tool for Teaching VoIP for Information Technology Classes.
DOI: 10.5220/0004848701970204
In *Proceedings of the 6th International Conference on Computer Supported Education* (CSEDU-2014), pages 197-204
ISBN: 978-989-758-021-5

problems and reflecting their experiences. These students are supervised by a tutor or supervisor.

The literature suggests that strategies, in which learning is based on problem solving, are effective in teaching in summarily practical areas, such as information technology. According to Cavalcante and Embiruçu (2013), it is possible to realize how this practice has been established around the world and how it can be applied in engineering courses. The use of VoIP technology has also been established. Furthermore, its importance in the communication systems, that are about to come, is highlighted in Goode (2002). Recently, Dias (et al. 2013) showed that the use of practical experiments can aid in teaching practical concepts of traditional telephony and IP telephony.

This paper will present a case study on the implementation of the Asterisk software as an aid tool related to the practical teaching by using the PBL approach to teach concepts of VoIP technology and its interaction with the traditional telephone system. This methodology was employed in courses in Electrical Engineering and Telecommunication System Technology, but it could easily be applied to any course which had subjects with VoIP technologies in their course programs.

The PBL technique was chosen so as the concept and the motivation could be introduced to the students at the beginning of the subject. This method consists of groups´ choice and of the fulfillment of practical problems which will be solved through the use of software, laboratory practices, theoretical content and the professor´s support. Such method aids the undergraduate students to solve practical questions by themselves (Lamar et. al. 2012).

According to Ribaud and Saliou (2013), PBL can help students to learn with the complexity and perceive that there are no simple responses for problem scenarios, nevertheless learning and life do occur in contexts which can alter the available and possible type of solution.

The problem-based learning method was initially carried out in the Telephony subject with emphasis on the telecommunication area. The students were evaluated in accordance with their theoretical and practical performance, being their critical sense analyzed in relation to the proposed problem.

This paper is organized as follows: section 2 will focus on a bibliographical review on the concepts of VoIP and a presentation of the Asterisk tool; section 3 will describe the materials and methods used in the design of the proposed experiment; in section 4, the results obtained from the experiment will be presented and in section 5, we present the conclusions.

2 VOICE OVER IP AND ASTERISK CONCEPTS

2.1 Voice over IP (VoIP)

Voice over IP is a set of networking protocols that have the function to normalize and regulate the sending of the voice from a source to a destination by using TCP/IP data networks (Keller, 2009). That is, an analog voice signal is converted into a set of digital signals, which is then sent through an internet connection in the form of packets with IP addressing.

The main difference between VoIP and traditional telephony is related to the way the voice is transported. This difference suggests that the only requirement to use VoIP technology is concerned with the use of a TCP / IP connection between two points with end to end delay less than 150 ms. This requirement creates some unique advantages to VoIP, such as (Keller 2009; Goode 2002):

- Cost reduction: expenditure decrease with traditional telecom operators and the use of softphones instead of conventional phones;
- Unique infrastructure: the convergence of voice and data networks will also make the physical network unique;
- Mobility: the branch line must be in a position where you can connect to the Internet;
- The telephone system control: reduces the users´ dependence from the telephone exchange maintenance company; and
- New features: some of them which are not available in the traditional telephony become possible, such as voice encryption during calls.

In VoIP technology, signaling protocols are responsible for determining a standard that specifies the data format and the rules to be followed by data traffic. Moreover, these signaling protocols are used to establish connections, determine the destination and also for issues related to signs such as: ring, caller ID, disconnection, among others. Currently the major signaling protocols for VoIP are (Silva, 2010):

- Session Initiation Protocol (SIP);
- Media Gateway Control Protocol (MGCP);
- Jingle;
- H.248/Megaco; and

- Inter-Asterisk eXchange (IAX).

Taking into account what has been mentioned, it is observed that information technology professionals, who work with communication networks, need practical tools that help them to learn the concepts of IP telephony during their training. The Asterisk, the software already mentioned, however, is presented as an alternative to building practical experiments of similar complexity and low cost for the reality of the job market.

2.2 The Asterisk Platform

The Asterisk software is able to perform the function of a private telephone exchange, which has as one of its primary functions the management of audio transmitted in digital communication channels (Madsen, Meggelen & Bryant, 2011). Asterisk can be used as an extremely powerful and flexible tool designed for the learning of VoIP technologies and protocols once it allows reproducing in laboratories situations and problems only seen in real public or private telephone networks.

Moreover, one of the advantages of working with Asterisk in the classroom is the fact that Asterisk is free software distributed by Digium® that is based on the GPL (General Public License) (Martín, 2009). The free version of Asterisk eliminates the need for a conventional private telephone exchange because its version has no limits of application. Additionally, Asterisk receives users´ contributions from all over the world, making this software always updated (Madsen, Meggelen & Bryant, 2011).

Asterisk's architecture was designed with great care so that there was as much flexibility as possible with regard to the operation of different types of hardware and software (Martín, 2009). Figure 1 shows the Asterisk internal architecture which is formed by a core and specific APIs (Application Programming Interfaces) which support the switching of internal information to PBX.

Information processing in the Asterisk core occurs in such a way that the specific protocols, codecs and hardware interfaces are abstracted from the information. This allows Asterisk to be able to connect to any hardware technology available (either current or future) to perform its essential functions (Silva, 2010).

The functioning and operation of Asterisk are based on the use of modules that the programmer can choose to use or not, depending on the application that he/she is working with. Table 1 describes the main modules for the correct functioning of an Asterisk server (Asterisk, 2010).

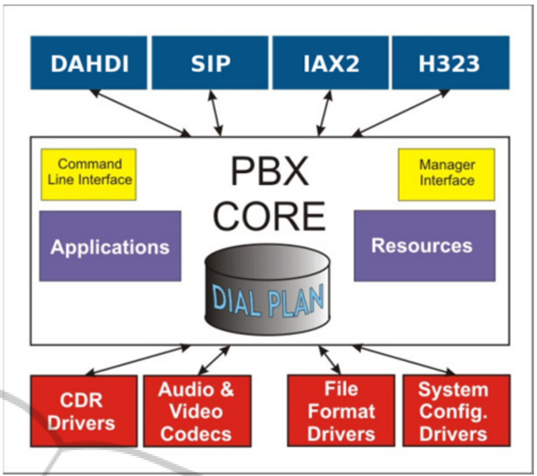

Figure 1: Arquitetura Interna do Asterisk (Asterisk, 2011).

Table 1: Asterisk modules.

Module	Description
Channel Drivers	The channel drivers make the communication with devices outside Asterisk possible by translating the signaling, or protocol, to the core.
Dialplan Applications	This module provides call functionality to the system.
Dialplan Functions	This module is used to set and retrieve parameters of configurations on a call.
Resources	Used to provide resources to Asterisk, like music on hold and call parking.
CODECs	This module is used to encode and decode audio or video so it takes less bandwidth.
File Format Drivers	Used to save media to disk in specific file formats and convert files back to media streams on the network.
Call Detail Record (CDR) Drivers	Used to write call logs to a disk or to a database.
Call Event Log (CEL) Drivers	Similar to CDR, but with details of what happened inside Asterisk during a particular call.
Bridge Drivers	Used by bridging architecture in Asterisk to provide various methods of bridging call media between participants in a call.

When installing Asterisk, the student will be automatically deploying the use of VoIP (Keller, 2009). The software use provides the broadening of learning so that the student can study from the creation of the used VoIP extension lines up to the monitoring of packets sent and the signaling exchange between the terminals. With the purpose of carrying out this investigation, an additional program called Wireshark will be used (Wireshark, 2011).

3 MATERIALS AND METHODS

The experiment was carried out in five phases:

1. Definition of the learning objectives;
2. Construction of a scenario to be set up in a laboratory;
3. Definition of the problems to be presented for the students to solve them;
4. Validation in an undergraduate subject with 60 credit hours, in the minimum, and that had VoIP in its course description; and
5. Evaluation of the obtained results.

The following learning objectives were defined in the first phase:

- Define what VoIP is;
- Differentiate VoIP technologies from the other ones used for establishing telephone calls both in public and private telephony;
- Understand the functioning of SIP, RTP and SRTP protocols; and
- Set up the Asterisk software for establishing telephone calls using VoIP technologies.

In the second phase, the scenario was defined and constructed to be used in the phase related specifically to PBL. The chosen scenario encompasses studies on VoIP with analog terminals and IPs telephones, as shown in Figure 2.

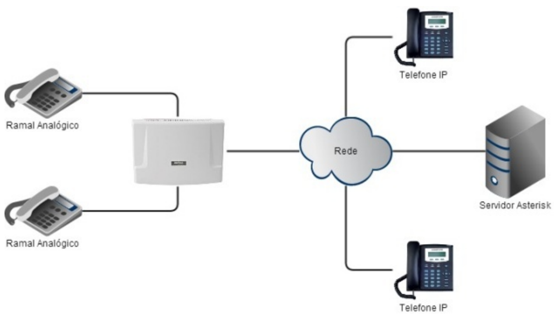

Figure 2: Scenario of the proposed solution.

The objective of this scenario is to provide an environment where students can understand the basic operation of a server set up with Asterisk and that they can, through the Asterisk configuration, make calls between IP terminals that are connected to the server. Subsequently, to understand how the signaling and voice traffic occur on the TCP/IP network, a network monitoring is carried out, in which calls and SIP protocols are captured and RTP/SRTP are analyzed.

When assembling the scenario, the following pieces of equipment available in the IFPB Telephony and Convergent Networks Laboratory were used: Computer with 4GB RAM, processor Intel QUAD CORE, plate FXS / FXO, E1 board, Impacta 68 Intelbras hybrid telephone exchange and Grandstream GXP-1200 VoIP phones.

After defining the scenario, the Linux operating system is installed on a computer which serves as a platform for the use of Asterisk. Figure 3 shows the Asterisk console after it is installed on Linux. It is through the console that the main information regarding Asterisk operation is accessed as well as it is possible to give operation commands to the software.

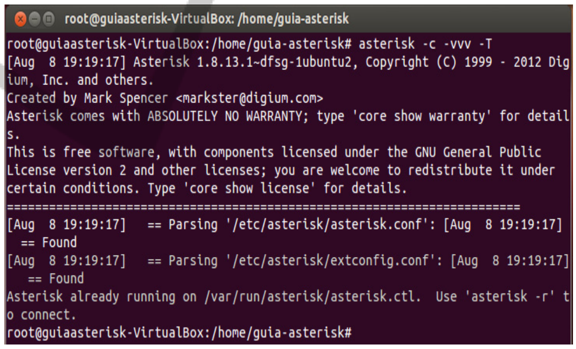

Figure 3: Initiation of Asterisk Server.

Then, it is necessary to create the SIP communication channels so that the Asterisk server can identify each extension. Creating SIP channels with Asterisk occurs by editing the sip.conf configuration file. The code segment below shows how to create four SIP extensions.

```
[general]
Bindport = 5060
Bindaddr = 0.0.0.0
disallow = all
allow = alaw
language = en_US

[commom_to_branches](!)
type = friend
context = branches
host = dynamic
```

```
[2000](commom_to_branches)
secret = 1234
mailbox = 2000

[2001](commom_to_branches)
secret = 1234
mailbox = 2001

[3000](commom_to_branches)
secret = 1234
mailbox = 3000

[3001](commom_to_branches)
secret = 1234
mailbox = 3001
```

In order to have communication among the created extensions, it is necessary to set the Asterisk server dial plan. The dial plan is created by editing the Asterisk extensions.conf. The following code segment shows how the dial plan setup is made for the scenario in Figure 2.

```
[branches]
; Impacta 68 branches

 exten => 2000,1,Dial(SIP/2000,30)
 exten => 2001,1,Dial(SIP/2001,30)

; IP phones branches

 exten => 3000,1,Dial(SIP/3000,30)
 exten => 3001,1,Dial(SIP/3001,30)
```

The use of analog terminals is allowed by setting the central Impacta 68 using specific software, as shown in Figure 4. It is interesting to note that the Impacta 68 central exchange recognizes the Asterisk server as a registration server on the network. Without this setting, it would not be possible to make Impacta 68 extensions to connect with the GXP-1200 IP terminals.

Figure 4: Configuração do Servidor de Registro da Impacta 68.

An IP terminal configuration can be performed on the phone itself or via the web server with the terminal IP address. Figure 5 shows the web server configuration for GXP-1200.

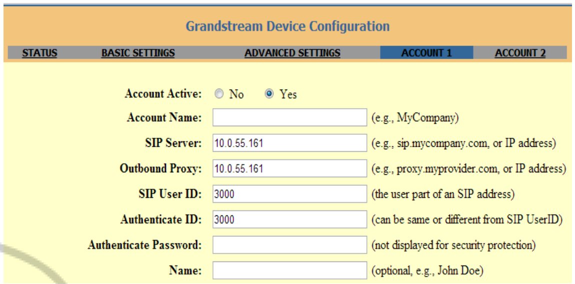

Figure 5: Programming Interface for GXP-1200.

After configuring the extension lines, the terminals send a registration request to the Asterisk server and connect to the server. Figure 6 shows the register of the terminals on the Asterisk server.

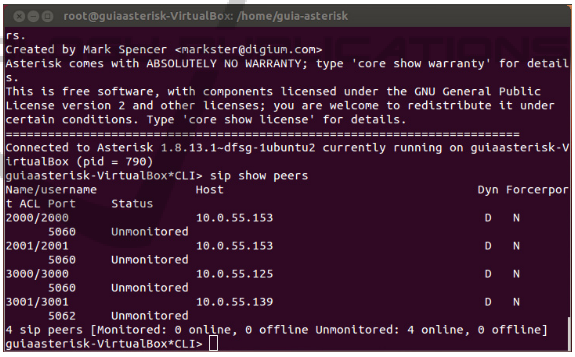

Figure 6: Registration of SIP Channels in Asterisk Server.

In addition to the default implementation, it is possible to activate SRTP protocol on the devices that will provide greater security in the sent packets. Encrypted data, even with packets being captured by network monitoring software, will not give access to the content within them. Therefore, the devices become secure on the network.

In the showed scenario (Figure 2), it is possible to observe how the Asterisk server is able to establish communication with the IP telephony devices and the Impacta 68 Central Telephone through the Internet. Such scenario has great importance to students´ training in the Telephony and Convergent Networks areas, given that its market performance will involve working from assembly to the understanding of the concepts that were applied in practice, in the classroom.

In the next phase, the problems were made up so as to be presented to the students. After researching in workplaces of several telecommunication companies and interviews with their professionals, the problems were defined to be presented to the students. The main objective was to focus on problems that reflected situations commonly found in professional environments involving VoIP.

For the validation phase, the Telephony subject of the Electrical Engineering and Telecommunication System Technology courses was chosen. Such subject presents all the necessary requirements: class load higher than 60 hours and has VoIP in the course description. In the PBL, the students were evaluated by the professor, by using continuous evaluation in several sessions that the presented problem demanded it, taking into account the learning objectives. At the end, grades were expressed on a scale of 0-100 and they were attributed to each one of the students.

In the evaluation phase, the students answered to a questionnaire of satisfaction and the professor of the mentioned subject carried out a subjective evaluation on the experience. This feedback allowed outlining new actions for improving the used approach. Furthermore, a comparison regarding the repetition and final general mean of the group´s grades in relation the previous semester was accomplished.

4 RESULTS

The mentioned scenario was established several times by students of Telephony discipline, of the Bachelor in Electrical Engineering and the Higher Course in Telecommunication System Technology from the Federal Institute of Education, Science and Technology of Paraíba (IFPB), which covers detailed study of VoIP. This enabled students to carry out the implementation of all practice and successfully absorb the theoretical content previously seen in the classroom.

Using Wireshark to monitor the network, it was possible to capture packets related to the exchange of SIP protocol signaling sent from the terminals to the Asterisk server and vice versa. It was also possible to capture the data stream transmitted among the terminals which uses the RTP protocol. Figure 7 shows an example of the exchange of SIP signaling captured during the experiment.

At the end of the experiment, the students were able to further increase the tab because two IP telephone protocols, RTP and SRTP, were used for security check in sending packets on the network. It was realized that when the SRTP mode is chosen in the terminals, the packets travel on the network in

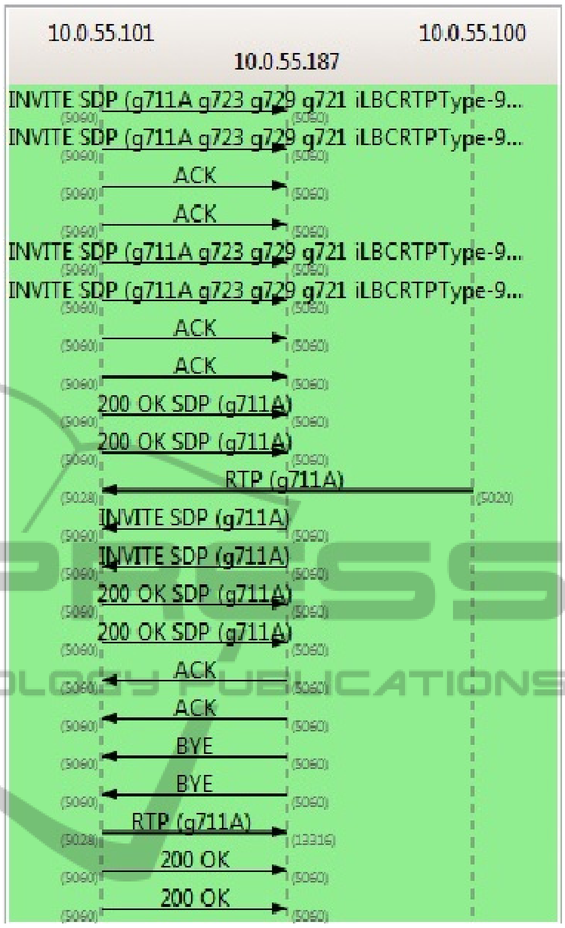

Figure 7: SIP Protocol Signaling Exchanged Between Two Terminals.

the safest way and it will not be possible to hear the content of the calls made, even with the capture of the packets. However for the RTP protocol, it was possible to examine the content of the calls when fulfilling the capture of the packets on the network.

The experiment accomplishment in the classroom allows the understanding of the theoretical content on VoIP technology because the use of Asterisk develops concepts related to both the traditional telephony and IP telephony as well as convergent networks. Furthermore, the use of Asterisk adds to the training of information technology professional a powerful and low cost solution to problems related to voice traffic in computer networks.

After validation, results were encouraging. While comparing with the numbers of the same subject, in the previous semester, the repetition rate was reduced to zero and the final general mean of the group´s grades also increased meaningfully from 61,7 to 81 – an increase of 31,35%. Students were questioned in relation to the PBL satisfaction level to detriment of the classical approach (in which the professor only transmits knowledge and the student has a secondary role in the learning process) as well as in regard to the general satisfaction with the subject. These results are presented in Figures 8 and 9. In both questionings, the students should mark only one of the following alternatives: very satisfied, satisfied, fairly satisfied, little satisfied or not satisfied.

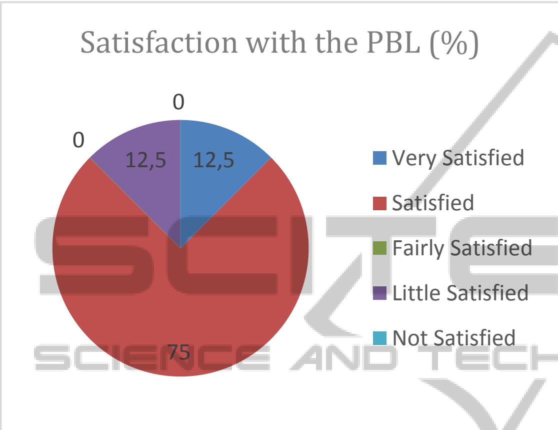

Figure 8: Satisfaction with PBL compared with the classical approach.

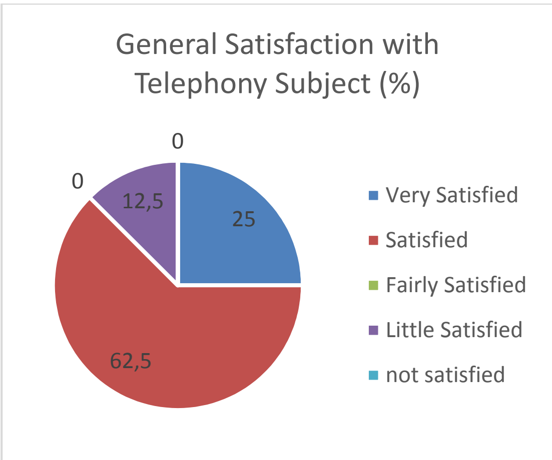

Figure 9: General Satisfaction with the Telephony Subject.

The subjective evaluation of the professor pointed out four important aspects:

1. The proposition of non-trivial problems and with no unique solution increased the challenge level for the students and it acted as stimulus to the participation in the activities of the subject.
2. Students faced difficulty in coping with the diversity of pieces of equipment that make up the scenario of the experiment, though they have had the professor´s explanation about the functioning and operation of such devices at the beginning of the class.
3. There was a noteworthy evolution in regard to the attainment of the general learning objectives in the group from the previous semester.
4. In relation to the PBL, the professor managed to improve the students´ performance, but he/she highlighted the need for teacher´s assistant in case of groups larger than 24 students.

5 CONCLUSIONS

The experience with the PBL using Asterisk as a tool for making up scenarios and problems for VoIP technology teaching was extremely positive both for the professor and the students. In the experimental phase of the mentioned subject, the students´ satisfaction level with the PBL reached 87,5% (satisfied and very satisfied as shown in Figure 8). Furthermore, the evaluation by means of grade and the professor´s subjective perception indicated progress regarding the attainment of the learning objectives on the students´ behalf.

With Asterisk, it was possible to create a reasonable complexity scenario which is present in several telecommunication companies that offer VoIP services; all of this with low cost when compared with the proprietary solutions available in the market. Therefore, providing the students problems that are close to the professional reality, which they will face in the job market, constituted a feasible activity.

As consequence of the analysis of the results, before presenting the problems to be solved for the students´ appreciation, three simple laboratory practices, with the aim to make the students familiar with the equipment and software to be used in the proposed scenario, were introduced. Hence, it is expected to mitigate the problem pointed out by the professor regarding the students´ difficulty in dealing with such devices and the associated software during the resolution phase of the presented problems, in addition to allowing an enhanced focus on the work itself.

Currently, this approach is being used in the same subject and the results will be evaluated in a continuous improvement process. The objective is to consolidate and improve the model so as it can be used for the VoIP technology teaching in the Engineering, Computing and Technology courses.

REFERENCES

Keller, A., 2009. *Asterisk na Prática*. São Paulo: Novatec Editora Ltda.

Zeyad, A. and Samaka, M., 2013. ePBL: Design and Implementation of a Problem Based Learning Environment. In: *Proceedings of the IEEE Global Engineering Education Conference (EDUCON), Berlim, March 2013.* Germany.

Silva, M. and Viana, D., 2013. Variáveis operacionais a serem consideradas no planejamento de disciplinas de projeto para aplicação do PBL nos cursos de engenharia. In: *Proceedings of the 41st COBENGE, Gramado, September 2013*. Gramado: ABENGE.

Fernandes, B. L., 2013. Projetos interdisciplinares: aprendizagem baseada em problemas (PBL). In: *Proceedings of the 41st COBENGE, Gramado, September 2013*. Gramado: ABENGE.

Cavalcante, F. and Embiruçu, M., 2013. Aprendizado com base em problemas: como entusiasmar os alunos a reduzir a evasão nos cursos de graduação em engenharia. In: *Proceedings of the 41st COBENGE, Gramado, September 2013*. Gramado: ABENGE.

Goode, B., 2002. Voice over internet protocol (VoIP). In: *Proceedings of the IEEE, vol. 90, nº. 9, September 2002, pp. 1495-1517 (doi: 10.1109/ JPROC.2002.802005)*.

Dias, M. et al. 2013. Uso do Asterisk como ferramenta de auxílio no ensino prático de telefonia. In: *Proceedings of the 41st COBENGE, Gramado, September 2013*. Gramado: ABENGE.

Lamar, Diego G. et al. 2012. Experiences in the Application of Project-Based Learning in a Switching-Mode Power Supplies Course. *IEEE Transactions on Education,* Vol. 55, no. 1, February 2012.

Ribaud, V. and SALIOU, P., 2013. The cost of Problem-Based Learning: an example in information in systems engineering. In: *Proceedings of the 26th International Conference on Software Engineering Education and Training, San Francisco, May 2013.* San Franciso: IEEE.

Madsen, L., Meggelen, J. V. and Bryant, R., 2011. *Asterisk™: The Definitive Guide*. 3rd ed. Sebastopol: O'Reilly Media, Inc.

Martín, S. G. (2009) *Contribution to Asterisk Open Source Project.* Thesis (MSc). Universitat Oberta de Catalunya.

Finnerty, I. (2009) *Per-extension voip call rating for the Asterisk PBX system.* Thesis (MSc). National University of Ireland.

Silva, V. (2010) *Soluções wireless/voip para redes comunitárias*. Thesis (MSc).Universidade de Aveiro.

Asterisk Project, 2011. *Asterisk Architecture, The Big Picture*. [Online]. Available from: https:// wiki.asterisk.org/wiki/display/AST/Asterisk+Architecture%2C+The+Big+Picture [Accessed 10/10/2013].

Asterisk Project, 2010. *Types of Asterisk Modules*. [Online]. Available from: https://wiki.asterisk.org/ wiki/display/AST/Types+of+Asterisk+Modules [Accessed 10/10/2013].

Wireshark, 2011.*VoIP calls* [Online]. Available from: http://wiki.wireshark.org/VoIP_calls [Accessed 10/10/2013].