VoIP University Solution: VoIP UEMA Project

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Abstract. *VoIP University is a solution to enable the development of communication projects based on IP protocol. This paper presents a low-cost deployment called VoIP UEMA, the first project of VoIP University solution. The technical feasibility of deploying this project was checked in a proof of concept. The concept was proven showing it is possible to create user extensions with information such as name, ID, job role, course, center, and campus get information of all registered users in the academic and administration system server and make voice and video calls through applications for Windows, Android, and iOS. VoIP UEMA project innovates because there is no similar solution in the literature to quadruple play communication that allows integration with any corporate system database to create and manage the user extension list. The low-cost aspect was justified by a financial analysis showing the annual phone bill expense dropped, representing savings of more than 97%.*

1. Introduction

This paper[1] proposes a low-cost solution called VoIP University to provide quadruple play communication to voice, data, and video with mobility.

The first project of the VoIP University solution is called VoIP UEMA which consists of VoIP UEMA web service, VoIP UEMA SIP server, SIGUEMA[2] academic and administration system server, VoIP UEMA Applications (Apps), VoIP UEMA webphone, and VoIP UEMA web page. SIGUEMA is a platform to host academic and administration information.

The VoIP UEMA project was deployed in the State University of Maranhão (UEMA) to all students, professors, and employees in all[3] 20 campuses and 22 centers connected through the same data center, requiring no SIP call routing[4].

The VoIP UEMA applications were developed to allow system users to make data, voice, and video mobile and fixed calls using smartphones with Android and iPhone OS (iOS)[5] operating systems.

1 This work was supported by FAPEMA (Fundação de Amparo à Pesquisa e Desenvolvimento Científico do Maranhão - Brasil).

2 https://sis.sig.uema.br/sigaa/verTelaLogin.do

3 https://www.uema.br/uema-em-numeros/campi-e-centros/

4 https://www.ietf.org/rfc/rfc3487.txt

5 https://apps.apple.com/us/app/voip-uema/id6444685212

After the integration with SIGUEMA, the University made available all extensions in a list to facilitate users' search by name, identification number (ID), job role, course, center, and campus.

The project has four computer program registration applications in the INPI: BR 51 2017 001608-9, BR 51 2017 001614-3, BR 51 2018 000820-8, and BR 51 2017 001606-2.

The next Sections are organized as follows. Section 2 presents the VoIP UEMA project. Section 3 presents the low-cost operation and the legacy integration. Section 4 presents a Mean Opinion Score (MOS) test. Section 5 contains a financial analysis. Section 6 is about innovation. Finally, in Section 7 are presented the conclusions.

2. VOIP UEMA Project

The VoIP UEMA project goal is to allow the use of the same SIGUEMA credentials (login and password) for user authentication in the VoIP UEMA network and keep the extensions list up to date considering the entrance and exit of students, professors, and administrative staff at the University.

An adaptation of the traditional VoIP telephony [Davidson, Peters, and Grace 2000], [Chen, Hung, and Lin 2007] was a strong requirement, where authentication will first pass through an intermediary web service, and after this the necessary steps to create a user extension list, as detailed in Figure 1.

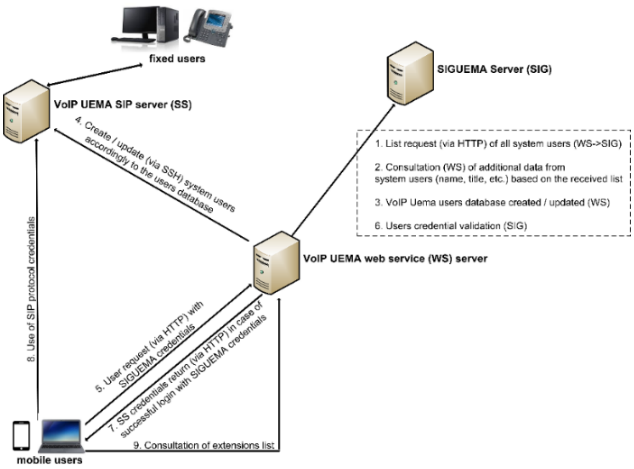

Figure 1. VoIP UEMA architecture

2.1. SIGUEMA web service

VoIP UEMA web service was developed based on Java code, plays important rules between VoIP UEMA SIP and SIGUEMA servers, and is responsible for relaying data from/to users to/from both servers.

After all the authentications are obtained, the VoIP UEMA web service can query the remaining data from the SIGUEMA web service which provides an API at the URL[6] for querying user data. It is possible to send an HTTP request with the GET method containing the ID and hash of the user password obtained directly from the SIGUEMA web service and return the needed data to create the extension list.

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6 http://www.sigws.uema.br/wservice/login

2.2. Asterisk Server

VoIP UEMA SIP server was based on the well-known Asterisk[7] product which is a freeware and open-source framework for building communication applications.

After the VoIP UEMA user database is created, it is necessary to enable the VoIP service for each extension on the Asterisk server.

For authentication, an HTTP request is required for the login URI stating the username and password. If the data entered is correct, the server returns a cookie that must be sent on all operations requests.

2.3. Request Security

It is important to emphasize that all requests shown so far are performed through application-only knowledge authentication headers, thus ensuring that the web service accepts and executes only requests from clients in possession of these headers [Ahson and Ilyas 2008].

2.4. VoIP UEMA Web Service

The first step in creating the VoIP UEMA user database is to get all users registered in SIGUEMA. The initial idea is to create a database with all users and their extension information such as name, identification number (ID), job role, course, center, and campus. To get this information, the web service must perform a query on the SIGUEMA server.

The VoIP UEMA web service was developed in Java language. It uses an Apache Tomcat web server[8] for hosting and an Apache web server[9] as a proxy.

2.5. VoIP UEMA Application

A freeware softphone for the Google Android operating system called VoIP UEMA App, available on the Play Store[10], has been developed to make VoIP and video calls based on the PJSIP library.

The App allows authentication with SIGUEMA credentials, through the VoIP UEMA web service, and the connection to the VoIP UEMA SIP server, allowing SIP calls with good quality.

The key features of this software are a plug-in adds support for voice calling with Opus[11] codec, a plug-in adds support for video calling with H.264[12] codec, good loss robustness and packet loss concealment (PLC), security and encryption with SRTP[13] based on the OpenSSH Library[14].

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7 https://www.asterisk.org
8 http://tomcat.apache.org/
9 https://httpd.apache.org/
10 https://play.google.com/store/apps/details?id=br.uema.voipuema.mobile.appandroid&hl=pt_BR
11 https://tools.ietf.org/html/rfc6716
12 https://www.itu.int/rec/T-REC-H.264
13 https://tools.ietf.org/html/rfc3711
14 https://sourceforge.net/projects/openssh-library/

The application achieved the objectives of successfully authenticating the users in VoIP UEMA web service and Asterisk servers using their SIGUEMA credentials and establishing optimal audio and video streaming.

The App presentation screen is shown in Figure 2. The red button allows anyone outside of the academic community to call some sectors of the University with no login, bringing social benefits.

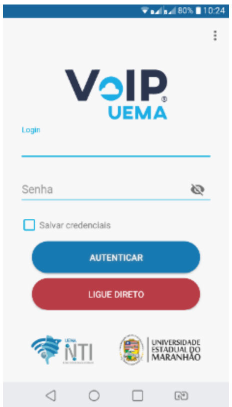

Figure 2. App presentation screen.

2.6. VoIP UEMA webphone

The VoIP UEMA webphone[15] presentation screen is shown in Figure 3. It was developed based on Web Real-Time Communication (Web RTC)[16], a free and open project that provides browsers and mobile applications with real-time communication capabilities via APIs. The voice codec used is called Opus.

Figure 3. VoIP UEMA webphone presentation screen

3. Low-Cost Operation and Legacy Integration

The target to reach low-cost operation is focused on the communication among VoIP UEMA system users using an IP network to decrease phone bill costs. However, it is necessary to consider the communication with the Time-Division Multiplexing (TDM) legacy network as well [Chong and Matthews 2004].

An integration scenario from/to IP to/from TDM via IP PBX is.

This current network topology is shown in Figure 4. The VoIP UEMA project deployed to all 20 campuses at the University. There are softphones and IP phones integrated with

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[15] https://voip.uema.br/discador/
[16] https://webrtc.org/

the legacy analog network through an IP PBX. The solution allows communications inside and outside the University through private and public IP networks and with the Public Switched Telephone Network (PSTN) and the Public Land Mobile Network (PLMN).

Paulo VI is the main campus of the University (campus 1 shown in Figure 4) and hosts the VoIP UEMA servers and the IP PBX with links to the fixed and mobile telephone networks. Communications with the other 19 campuses of the University (campus n) are established over the Internet reducing phone bill expenses.

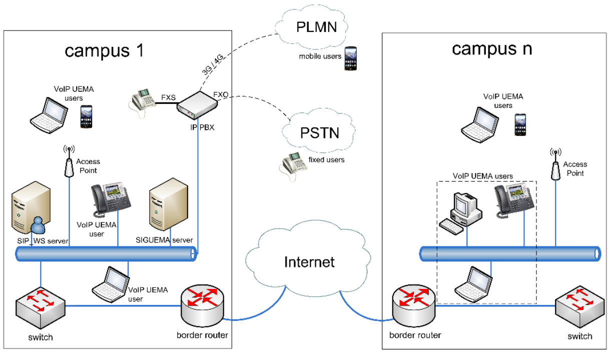

Figure 4. IP <-> TDM via IP PBX network topology

4. Mean Opinion Score (MOS)

MOS[17] measures subjective voice quality for a call. MOS scores range from 1 for unacceptable to 5 for excellent. VOIP calls often are in the 3.5 to 4.2 range.

39 people were invited to participate in a MOS test by calling each other using the VoIP UEMA App and VoIP UEMA webphone (both with Opus voice codec) over Wi-Fi and wired networks (both with average data rates above 10 Mbit/s) as well as 4G (average data rates of 15 Mbit/s). Table 1 shows the MOS test results with a weighted average of around 3.5, where n is the number of attendees who have assigned a score.

Table 1. MOS test results

MOS	1	1.5	2	2.5	3	3.5	4	4.5	5
n	4	0	1	0	8	3	20	2	1

5. Financial Analysis

Considering 2013, 2014, and 2015, UEMA had an average annual phone bill expense of R$ 1.256.624,30 (US$ 312,500.00) shown in Figure 5.

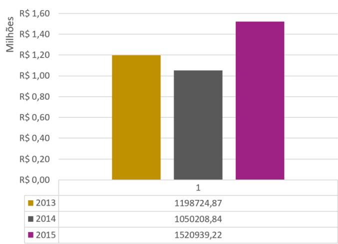

Figure 5. UEMA’s average annual phone bill expense over 3 years

17 https://www.voip-info.org/call-quality-metrics/

Two years after the deployment of the VoIP UEMA project, the annual phone bill expense dropped to US$ 9,000, representing savings of over 97 %.

The low-cost solution is justified because there were no expenses with materials or equipment, only the professor and students' labor paid with scholarship. It cannot be stated that the cost of this project is zero because it requires data communication network infrastructure that requires prior investment.

6. Innovation

The VoIP UEMA project innovates because there is no similar solution in the literature to quadruple play communication that allows integration with any corporate system database to create and manage the user extension list.

Unlike other solutions[18,19], the VoIP UEMA project has a list of extensions based on the list of users in the academic and administrative system, and its management is easier because the list of extensions is updated automatically.

7. Conclusions

The VoIP UEMA project was made available to more than 20,000 students, 1,400 teachers, and administrative technicians. Nowadays, communication occurs more via the VoIP UEMA network than via public fixed and mobile telephone networks.

The VoIP UEMA softphones were developed for Android, iOS, and Windows operating systems with more than 1,000 downloads and more than 300 registered service extensions in laboratories, libraries, and departments. A financial analysis revealed that annual spending on telephone bills decreased by more than 97 %.

The VoIP UEMA project has both user and service extensions for voice and video communications with people, departments, laboratories, and course secretaries. Furthermore, unlike WhatsApp, which generates costs with WhatsApp Business, there is no cost for public administration and now the university has its solution without depending on a third party.

The VoIP UEMA project innovates because there is no similar solution in the literature to quadruple play communication that allows integration with any corporate system database to create and manage the user extension list.

References

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Chong H. M., Matthews H. S. Comparative Analysis or Traditional Telephone and Voice-over-Internet Protocol (VoIP) Systems, 2004.

18 https://www.rnp.br/servicos/fone-rnp

19 https://www.zoiper.com/