AN EVALUATION OF BROWSER-BASED CLIENTS FOR ASTERISK

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ABSTRACT

Asterisk is a Private Branch Exchange (PBX) software used to connect telephony applications such as Voice over Internet Protocol (VoIP). In the Department of Computer Science at the University of Namibia (DCS-UNAM), Asterisk is currently set up to provide basic VoIP services to staff and students. However, this system currently has limited accessibility because it can only be accessed with VoIP clients such as hard and soft phones. Alternatively, Browser-based VoIP clients (BBVCs) can be used to provide nomadic accessibility to Asterisk. The study evaluates the BBVCs for their challenges and usability in providing users in DCS-UNAM with nomadic access to Asterisk services.

Both qualitative and quantitative research methodologies were used for this study. A literature study was carried out to find available BBVCs that can be customised to work with Asterisk. Out of them, the Java Inter-Asterisk eXchange Client (JIAXC) library application was selected as the appropriate BBVC for this study. Data produced during the compilation, the loading and the use of the BBVC's web phone were captured using Wireshark Network Protocol Analyzer, Firefox Java Console and the Gnome terminal. Some of the data collected were analysed using Wireshark and the IAX2 Call analyzer. The overall usability of the BBVC was also obtained using questionnaires which were distributed to Third year Computer Science students at DCS-UNAM and then analysed using Statistica software package.

The JIAXC application can be customised to provide nomadic accessibility to the Asterisk setup in DCS-UNAM. The main challenge encountered during the customisation of the BBVC is the compilation of the JIAXC library. The effects include restriction effects from the Java applet security manager and the effects of many users accessing the BBVC's web phone on the call quality. The analysis done on the questionnaire data indicated that the customised BBVC is considered as a usable means of accessing Asterisk services.

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ACRONYMS

AJAX:	Asynchronous Javascript and XML
API:	Application Programming Interface
ASP:	Active Server Pages
ATA:	Analog Telephone Adaptor
BBA:	Browser-based Application
BBVC:	Browser-based VoIP Client
BBVC4A:	Browser-based VoIP Client for Asterisk
DCS:	Department of Computer Science
DTMF:	Dual Tone Multi Frequency
GNU:	A recursive GNU’s Not Unix
GOB:	“Good” or “Bad”
GSM:	Global System for Mobile Communications
GUI:	Graphical User Interface
GVIM:	Graphical Vi IMproved
HTML:	HyperText Markup Language
IAX:	Inter-Asterisk eXchange
ID:	Identification
iLBC:	internet Low Bit Codec
IM:	Instant Messaging
IP:	Internet Protocol
ISP:	Internet Service Provider

ITU-T:	International Telecommunication Union, the Telecommunication division
IVR:	Interactive Voice Response
JAIN:	Java APIs for Intelligent Networks
JIAXC:	Java IAXClient
JMF:	Java Media Framework
JNI:	Java Native Interface
JRE:	Java Runtime Environment
JSAP:	JAIN SIP Applet Phone
JVM:	Java Virtual Machine
LAMPP:	Linux Apache MySQL PHP
LAN:	Local Area Network
Mac:	Macintosh
MGCP:	Media Gateway Control Protocol
MOS:	Mean Opinion Score
MS:	Microsoft
MS IE:	Microsoft Internet Explorer
NIST:	National Institute of Standards and Technology
OS:	Operating System
PBX:	Private Branch Exchange
PC:	Personal Computer
PESQ:	Perceptual Evaluation of Speech Quality
PHP:	A recursive PHP: Hypertext Preprocessor
POW:	“Poor” or “Worse”

PSTN:	Public Switched Telephone Network
RTP:	Realtime Transport Protocol
SDK:	Software Development Kit
SIP:	Session Initiation Protocol
SMS:	Short Message Service
TAM:	Technology Acceptance Model
UML:	Unified Modelling Language
UNAM:	University of Namibia
USE:	Usefulness, Satisfaction, and Ease of Use
VoIP:	Voice over IP
WWW:	World Wide Web
XAMPP:	X Apache MySQL PHP Perl
XML:	Extensive Markup Language

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DEDICATION

To God Almighty, my father Prof. J. A. Oyedele, my mother Dr. V. I. Oyedele, my sister Opeoluwa and my brothers, John and Ifeolu.

DECLARATION

I declare that this study titled "**An Evaluation of Browser-Based Clients for Asterisk**" is a true reflection of my own research, and that this work, or part thereof has not been submitted for a degree in any other institution of higher education. I further declare that all the sources used or mentioned have been acknowledged by complete references.

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CHAPTER ONE

INTRODUCTION

The purpose of this study was to find and evaluate Browser-based Voice over Internet Protocol (VoIP) clients (BBVCs) which would allow users to conveniently access Asterisk VoIP services. In this chapter, VoIP clients and Asterisk will be introduced along with the statement of the problem and the research questions. The scope and the significance of the study, a brief summary of the methodologies used and the findings will also be discussed. Finally the outline of the thesis presented.

1.1 Introduction

VoIP, a form of telecommunication service, involves the transmission of digitized voice data over Internet Protocol (IP) networks. It includes Caller Identification, voicemail, conference calling, call waiting, call forwarding and fax services (Van Meggelen, Madsen & Smith, 2007). VoIP is commonly used because it helps mitigate telecommunication costs incurred when using the traditional telecommunication services and other forms of telecommunication services such as the mobile telecommunication network services.

Different types of users access VoIP services. They include static users and nomadic users. Static VoIP users make use of tools that are fixed on a location while nomadic VoIP users utilise tools accessible from anywhere on an IP network. These tools have the capabilities to input (and output) and code (and decode) voice data and are referred to as VoIP clients.

1.2 VoIP Clients

There are different types of VoIP clients and they include dedicated telephones such as Hard phones and Personal Computer (PC) Software phones (known as Soft phones) (Porter et al., 2006; Van Meggelen et al., 2007). The hard phones include IP phones with wired or wireless Ethernet interfaces, and traditional telephones that require the use of Analog Telephone Adaptor (ATA). The soft phones are computer application programs that imitate the hard phones on PCs. They require the use of microphones and speakers, and Internet (or Intranet) connection.

VoIP clients have different features which include those of the traditional analog telephones. These features depend on the type of VoIP protocols, network protocols supported, voice codecs, media and transmission protocols used, the operating systems platform supported, as well as the variety of services offered by the VoIP service providers (Hardy, 2003; Mahler, 2004; Ohrtman, 2004; Porter et al., 2006; Van Meggelen et al., 2007). The VoIP protocols commonly used include the Session Initiation Protocol (SIP), the Inter-Asterisk eXchange (IAX) protocol and the H.323 protocol. Common used codecs include Global System for Mobile Communications (GSM), Speex, G.726 and G.729 codecs. Some VoIP clients such as the soft phones can be used with different VoIP service providers, while others are dedicated to work with only one VoIP service provider.

VoIP clients such as soft phones are set up in a particular location so that whenever VoIP service is to be accessed, the users have to go to the location where the VoIP clients are. Hard phones are also set up in a fixed location excluding those that can be carried from one geographic location to another. Access to these VoIP clients is fixed or static. Figure 1.1 shows a basic setup for fixed accessibility.

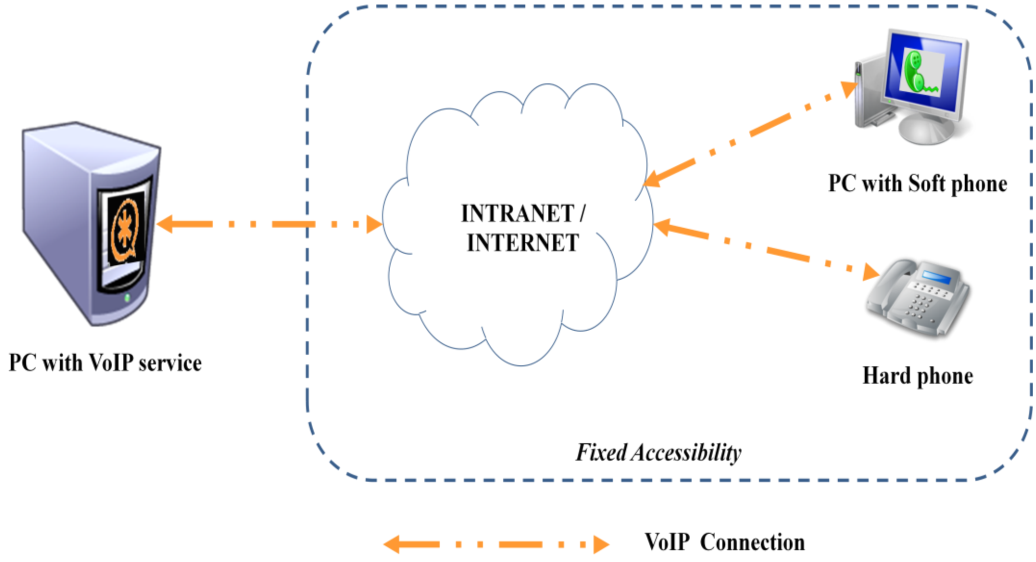

Figure 1.1: A Simple VoIP Setup for Fixed Accessibility.

In order to mitigate fixed accessibility for nomadic users, some Browser-based Applications (BBAs) were developed by VoIP service providers. This was done to provide users with more means of accessing their VoIP services from anywhere on an IP-based network. These BBAs are referred to as BBVCs.

1.3 BBVC

A BBVC, a BBA with a web phone, is used to provide nomadic access to VoIP services. Like other forms of BBAs, it uses some of the features available on Web browsers (Silver, 2006). These features include support for web technologies that are used to implement the web phone. The commonly used web technologies include ActiveX technology, Java Applets and Plug-ins such as Adobe Flash (Dormann & Rafail, 2008). These web technologies, with compliant web browsers, have to be already installed on the PC that will be used to access the BBVC. BBVCs, like other VoIP clients, also require an audio input and output device for the provision of voice data. Some of these BBVCs are designed to work with individual VoIP service providers through Internet connections.

Some VoIP service providers such as Voice Commerce Group Limited (Busta Communications, 2008), TringMe Company (TringMe, 2008a) and Gizmo5 Technologies Inc (Gizmo5 Technologies Inc, 2008b, 2008d) developed their own BBVCs as another means to improve users' accessibility to their VoIP services. These BBVCs are accessed anytime, anywhere, using any PC because most PCs in use to date have at least one web browser (Silver, 2006). Figure 1.2 shows a basic setup for this type of accessibility.

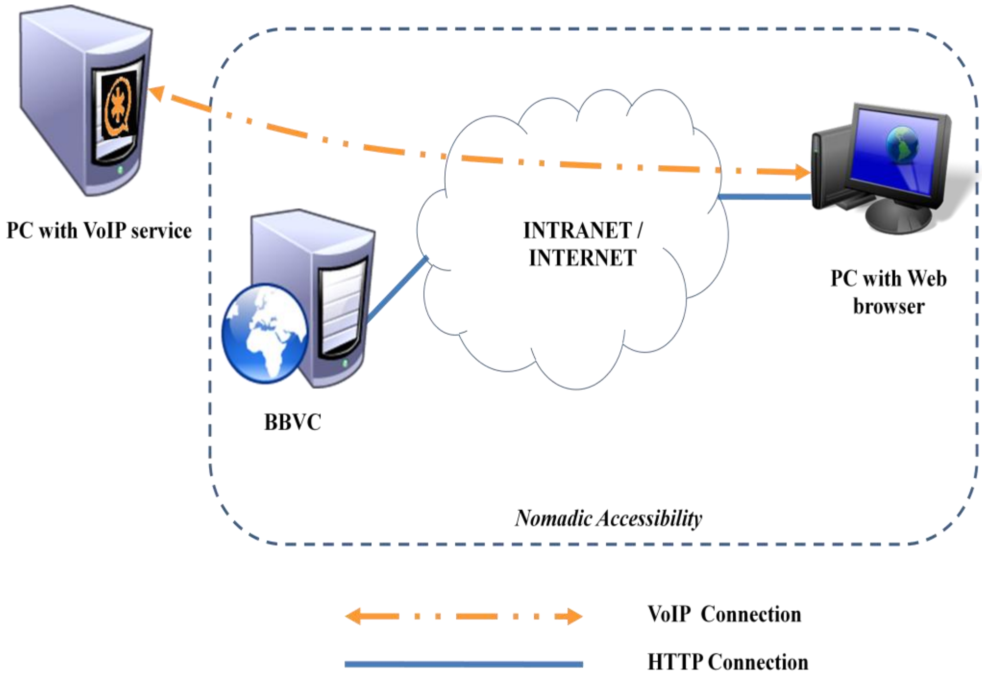

Figure 1.2: A Simple VoIP Setup for Nomadic Accessibility.

1.4 Asterisk

Asterisk is an open source Internet Protocol Private Branch Exchange (IP PBX) software used to connect the technologies and applications of telephony ranging from the traditional analog to Internet telephony. Figure 1.1 also shows a simple Asterisk VoIP setup. One key feature of Asterisk is the provision of many services such as VoIP services, Call transfer, Interactive Voice Response (IVR), and fax services (Van Meggelen et al., 2007).

Various VoIP protocols and codecs are also supported on Asterisk (Asterisk, 2008; Gomillion & Dempster, 2006; Hitchcock, 2006; Porter et al., 2006; Van Meggelen et al., 2007). VoIP protocols supported include IAX, SIP and H.323. Codecs supported include GSM, Speex, G.726, G.729, and G.711.

Asterisk is used in many educational institutions and by service providers. In the Department of Computer Science of the University of Namibia (DCS-UNAM), Asterisk is currently setup to provide basic telephony services to staff and students (Mwansa, 2008). In the Department of Computer Science of Rhodes University, the Asterisk has been customised with other open source applications to form iLanga. iLanga is used to provide VoIP services to the staff members and the students of the department (Hitchcock, 2006; Mwansa, 2008). Asterisk is also used by many VoIP service providers to provide VoIP services to people internationally as in the case of the IP Communications Free World Dialup (IP Communication, 2008) and IAX termination service provider's BinPhone (IAX termination service provider, 2008).

The VoIP users are required to have the appropriate VoIP client and Intranet (or Internet) connection before they can access their services. Recently, the means of accessing VoIP services of Asterisk are limiting. VoIP users can not access VoIP services from any computer connected to an IP-based network because not all the computers (and networks) connected have the required VoIP clients to access the services. In an educational environment where privileges may be limited for users to install new applications on PCs, or to connect hard phones to the network, the process of setting up VoIP clients can be costly and time consuming. This study finds and evaluates BBVCs which can allow users to conveniently access Asterisk VoIP services with less limitation.

1.5 Statement of the Problem

According to Griggs (2008), there are many difficulties in accessing VoIP clients such as the hard phones and the soft phones to access and use VoIP services. These clients require that the users have knowledge on how to acquire them and set them up for use. For users who have to move to different locations to use different PCs, the VoIP clients will have to be set up in each location. This is so because the soft phones are not designed to be accessed from any PC apart from the ones on which they were set up while the hard phones are costly to acquire. Since these VoIP clients can be accessed only in the location where they are installed, access to them is fixed. In addition, users in an organisational division such as DCS-UNAM will have to acquire some privileges from authorities such as network administrators to set up the VoIP clients. These processes are costly and time consuming to the users. In order to mitigate these difficulties, BBVCs were developed.

BBVCs, according to their developers, can be used to access VoIP services like other types of VoIP clients. They, unlike the other VoIP clients, can be accessed from any PC connected to an IP network, thus providing nomadic accessibility. This is so because the BBVCs can be accessed through web browsers and most PCs have at least one web browser. Most of the available BBVCs are created specifically for individual VoIP service providers thus making them proprietary. It was therefore essential to conduct an evaluation on available BBVCs that can be customised for Asterisk to enable nomadic access to its VoIP services.

1.6 Research Questions

With regards to the statement of the problem, the study answered the following questions:

1.) Can available BBVCs be customised to provide nomadic accessibility to Asterisk VoIP call services in DCS-UNAM?
2.) What are the challenges experienced by customising the features of those BBVCs to provide nomadic accessibility to Asterisk VoIP call services?
3.) How do the challenges affect the use of the BBVCs to access Asterisk VoIP call services?
4.) What are users' perceptions about the use of the BBVCs for Asterisk?

1.7 Significance of the Study

The study investigated BBVCs to provide nomadic access to VoIP services. The findings of this study provide staff members and students with more options to access the Asterisk VoIP services available in DCS-UNAM from anywhere on the University of Namibia (UNAM) IP network. The findings also benefit users of Asterisk IP PBX in different organizations. This study contributes to the research work done in the area of VoIP and web-based applications.

1.8 Scope of the Study

The study focused on the evaluation of available BBVCs for use with Asterisk, the customisation and deployment of a BBVC for DCS-UNAM Asterisk setup, and the evaluation of users' perceptions with the use of the customised BBVC. The main scopes of the study are discussed in sections 1.8.1, 1.8.2 and 1.8.3.

1.8.1 Evaluation Models

There are many call quality models used to evaluate VoIP applications. These models include the ITU-T E-model and the PESQ model. Since BBVCs are relatively new, this study aimed to evaluate them in terms of the challenges involved in customising them, as well as their usability as perceived by users.

1.8.2 Web Browser Support

There are many types of web browsers designed for devices such as telephones, IP phones, mobile phones and PCs. The study was limited to web browsers for PCs. The commonly used OS platforms at UNAM are Microsoft (MS) Windows OS and the Linux OS distribution. Thus the study was also limited to the BBVCs that support these OSs.

1.8.3 VoIP Call Services

VoIP call services include voice call services and video call services such as video calls and video conferencing. This study was limited to voice call services because the available bandwidth at UNAM is not sufficient enough to support video data packet transfer.

1.9 Research Methodology

This study made use of both qualitative and quantitative research methodologies. An in-depth survey was conducted to explore BBVCs that can be used to access VoIP services. These applications were set up and examined in terms of the features they offer. Provision for nomadic accessibility was also considered. The results of the survey were used to identify the common features to be included in a customised BBVC for Asterisk.

Out of all the BBVCs considered, the Java Inter-Asterisk Exchange Client Library (JIAXC) application was the BBVC found most appropriate to customised for nomadic accessibility to Asterisk. The JIAXC demo application was used as a web phone for a customised BBVC (BBVC4A website). The customised BBVC was deployed, tested and evaluated by Third year Computer Science students who were selected using random sampling. After testing the application, the students gave their feedback about the customised BBVC.

Wireshark Network Protocol Analyzer (Wireshark, n.d.) was used to collect network traffic data generated when the web phone was used. The data was analysed using Wireshark and the IAX2 Call analyzers. The Gnome terminal and Firefox Java Console were used to collect error messages reflecting the challenges encountered. The data collected from these instruments were qualitative. They were analysed and presented descriptively.

Questionnaire was also used as an instrument to get the students feedback on the usability of the customised BBVC. The questionnaire data was empirically analysed using quantitative techniques.

1.10 Summary of the Finding

The JIAXC application can be customised to provide nomadic accessibility to the Asterisk setup at UNAM. The main challenge encountered during the customisation of the BBVC was the compilation of the JIAXC library. The challenge effects included restriction effects from the Java applet security manager and the effects of many users accessing the BBVC's web phone on the call quality. The analysis done on the questionnaire data indicated that the customised BBVC was considered as a usable means of accessing Asterisk services.

1.11 Definition of Terms

For the purpose of this study, the following terminologies were defined.

BBA: is a form of web-based application (or weblication) that extends the functions of a Web browser. BBAs are sometimes considered as web application form of desktop applications because they are only accessed from a web browser (Ezzy, 2006).

VoIP: **(**also called IP Telephony) is the routing of voice conversations over the Internet or through any other IP-based network rather than the traditional telephone landline system, also called the Public Switched Telephone Network (PSTN).

VoIP client: is a technology used to access and use VoIP services.

VoIP user: is a person who can access and make use of VoIP services.

Web browser: is a software application that allows users to access and interact with the contents on the World Wide Web (WWW).

1.12 Outline of the Thesis

The thesis is organized into seven chapters.

Chapter 1 introduces VoIP clients and Asterisk, the statement of the problem, the research questions for the study, the significance of the study, the scope of the study, followed by the summary of the research methodology and the findings.

Chapter 2 addresses the literature review carried out for the purpose of the study.

Chapter 3 discusses the methodologies used in the study.

Chapter 4 presents the design of the BBVC4A website for Asterisk and the upgraded JIAXC application.

Chapter 5 presents the implementation of the design.

Chapter 6 discusses the analysis of the data collected.

Chapter 7 presents the conclusion for this study and some ideas for future work.

1.13 Summary

VoIP is a technology that allows voice packets to be routed over IP. Although as a popular form of telecommunication, VoIP accessibility is currently limited, as it can only be accessed via VoIP clients with fixed accessibility. This chapter introduced VoIP clients, BBVCs and Asterisk, the statement of the problem and the research questions for the study. The scope and significance of the study, a brief summary of the methodology used and the findings made, and the outline of the thesis were discussed.

CHAPTER TWO

LITERATURE REVIEW

2.1 Introduction

There are few organisations providing BBVCs for their VoIP services, but rarely for Asterisk VoIP services. This chapter will address the literature review on the accessibility of VoIP services, web technologies, some existing BBVCs, and the features of BBVCs. Problems with existing BBVCs will also be addressed. Finally, the evaluation models will be presented.

2.2 Accessibility of VoIP Services

The common means of accessing VoIP services is through VoIP clients such as hard phones and soft phones. Hard phones range from dedicated traditional telephones to IP phones (VOIP-Info.org, 2008). According to VOIP-Info.org (VOIP-Info.org, 2008), commonly used soft phones include Skype, Gizmo, Fring, X-Lite, Nimbuzz, GTalk, Yahoo! Messenger, KIAX, Firefly, Wengo, and SJPhone.

Hard phones are expensive to obtain, and are difficult to install and configure for users (Ohrtman, 2004). This is challenging for nomadic VoIP users. When a user moves from one geographical location to another, say from one town to another, he or she may have to find the appropriate VoIP client and reconfigure it to access the VoIP service. This task is more tedious for the user, especially, if he or she does not have the necessary technical knowledge to setup the VoIP client.

Pressman (2005) mentioned that one of the most tedious things for a novice using a computer is to understand how to operate the computer effectively and how to install new applications. As a result, it can be difficult for the user to install a soft phone on the computer and configure it for usage. It is common practise that most users of PCs owned by Public Internet Cafe and educational institutions, such as Universities, are not given rights such as an administrative right to make changes to, or install new application on those PCs. If an IP phone (or a soft phone) is to be used, then the user must know the settings for the IP network, and get the appropriate authorisation to use the network. All these increase the complexity of accessing VoIP services.

BBVCs are the preferable method for accessing VoIP services as the users do not have to get permission from the network administrators to use the BBVCs. BBVCs are available whenever the VoIP service providers implement them as one of the means of accessing its services. These VoIP clients can easily be accessed from any IP-based network, so long as the user knows the website address for it and the web technology used by the BBVC is present on the PC.

2.3 Web Technologies

Different features of web browsers are used for the development of BBVCs. The main feature commonly used is the support for web technologies such as ActiveX, Adobe Flash and Java (Dean, 2007).

ActiveX is a Microsoft technology that is supported only on the MS Internet Explorer (MS IE) web browser. It works on MS Windows OS through the web browser. The technology comes installed with MS Windows OS, so that plug-ins that add extra software components to the computer can be authorised when a web page, for instance, is accessed (Microsoft Corporation, 2009). The major issue with this technology is that it relies on, and supports only, the MS Windows OS platform and the MS IE for functionality.

The Flash technology is owned and distributed by the Adobe Systems. It is a multimedia platform on which interactive contents, ranging from web page components to Rich Internet Applications, are developed for web applications (Adobe System Incorporated, 2009). The Flash technology is supported on many web browsers, OSs and PC platforms. On MS Windows OS, the Flash technology is partially installed and sometimes needs to be updated so that Flash based applications can be viewed in, for instance, web browsers. For other OSs such as Linux and Mac, the Flash technology is required to be downloaded from the Adobe Systems website or from add-on installation packages (Adobe System Incorporated, 2009). The reason is that the Flash technology does not come installed with Linux and Mac OSs. The process of installing and updating the Flash technology becomes even more frustrating for users when they have to use a slow Internet connection to download the technology.

Java technology is a platform-independent technology in that a Java program can run on any OS without being re-modified as long as the Java Virtual Machine (JVM) is installed on the PC (Cornell & Horstmann, 2005). It is supported by most web browsers, many OSs and PCs. Like ActiveX and Flash technology, it can be used to create web contents, such as Applets, for web browser and also extend the functionalities of the web browser. This technology requires the installation of the Java Runtime Environment (JRE) which includes the Java Virtual Machine (JVM) for web browsers to use (Cornell & Horstmann, 2005). The drawback with this technology is the downloading and the installation of the JRE for the PC and this can take a long time depending on the transmission rate of the network.

Unlike ActiveX, both Flash and Java are supported by most web browsers that work on MS Windows OS, Linux OS and Mac OS (Jansen & Karygiannis, 2000; Cornell & Horstmann, 2005). The Flash technology comes with MS Windows OS installation and sometimes needs to be updated before it can be used in web browsers. On Linux OS, both Flash and Java can be installed during or after the installation of the OS. Each technology has to be present in the OS along with compliant web browsers before they can be utilized.

2.4 Existing BBVCs

Many of the BBVCs available are designed with different web technologies, VoIP protocol and codecs, depending on the VoIP services they are meant to be used for. The main part of a BBVC, the web phone, is also based on a web technology. Available BBVCs include Voice Commerce Group’s Busta Browser-based phone, Gizmo’s GizmoCall Browser-based phone, TringMe’s TringPhone Browser-based phone, JAIN SIP Applet Phone (JSAP), and Mexuar Communications’ Corraleta product.

2.4.1 ActiveX Technology based BBVC

The Busta Browser-based phone by the Voice Commerce Group Limited (Busta Communications, 2008) can be used to make and receive calls from web browsers (Waxer, 2006). Designed based on the ActiveX technology, it also provides support for SIP as its VoIP protocol and codecs such as G.7111 and G.729.

The main drawback is that it can only be used to access Busta VoIP services and works only on web browsers that support Microsoft’s ActiveX technology and dependent on MS Windows OS. Another downside is that the BBVC is proprietary and it can only be accessed with an Internet connection.

2.4.2 Flash Technology based BBVCs

TringPhone Browser-based phone is designed using Adobe Flash technology. Unlike Busta, it can be used to access TringMe services, VoIP services, SIP services and other Instant Messaging (IM) services that provide voice services (TringMe, 2008b). It provides support for SIP as its VoIP protocol and codecs such as GSM, G.729, G.723, G.711, iLBC and Speex. It also works with Flash compliant web browsers and OSs such as MS Windows and Linux.

Another Flash based BBVC is Gizmo5 Technologies Inc's GizmoCall. According to MacManus (2007),

> Gizmo Call is an online service that makes VoIP phone calls possible from any web browser. It works via a Flash plug-in, which enables users to make calls simply by typing a phone number into a text-field in a browser (para. 1).

It can also be used to access Gizmo services, Google's GTalk services, Skype services and SIP services (Gizmo5 Technologies Inc, 2008b, 2008d; MacManus, 2007). Unlike TringPhone, it can be used to access Asterisk services.

The support for non-Gizmo services such as Asterisk services is done through its Opensky service (Gizmo5 Technologies Inc, 2008a, 2008b, 2008c, 2008d; MacManus, 2007). The Opensky service allows users to connect their non-Gizmo services to the main Gizmo services. Access to the non-Gizmo services can be achieved as if they were part of the main Gizmo VoIP services.

The drawback of adding Asterisk services through the Opensky service is that only Business Enterprise Asterisk services can be added. This requires some payments to be made prior to the addition of the Asterisk services (Gizmo5 Technologies Inc, 2008a, 2008c, 2008d). Though the Asterisk services might be hosted in an organisation's local building, the BBVC can only be reached through the Internet before the Asterisk services can be accessed. This is another downside to the GizmoCall BBVC.

Another drawback is that GizmoCall, like TringPhone, only works with OSs that support Adobe Flash technology, including MS Windows, Linux and Mac (Gizmo5 Technologies Inc, 2008a, 2008b, 2008d).

2.4.3 Java Technology based BBVCs

A Java based BBVC is JSAP, developed by the National Institute of Standards and Technology (NIST) (National Institute of Standards and Technology, n.d., 2007). JSAP makes use of JAIN SIP for voice and text messaging, and audio support with Realtime Transport Protocol (RTP). Unlike Busta, GizmoCall and TringPhone, JSAP is open source. It can be launched as a standalone PC application or as an embedded applet in a web page. As a standalone PC application, it can be used to access IM services and some Asterisk services (National Institute of Standards and Technology, 2007).

JSAP requires the Java Media Framework (JMF) to be present on the PC that will be used to access it. The JMF is an additional, but optional, package that extends the Java 2 Platform, Standard Edition (J2SE) to provide support for multimedia development (Sun Microsystems, 1999). The installation of the JMF can only be done by users with PC administrative rights, and this adds to the drawback of JSAP.

Another Java based BBVC is Mexuar Communications' Corraleta Connect applet (Mexuar Communications, 2006b). It provides supports for IAX VoIP protocol and can be used on Java compliant web browsers and OSs such as MS Windows, Linux, Mac OSX and Solaris.

Corraleta Connect applet is mainly used for developing click to call applications such as call centre applications. The applet is a web phone and is represented as a button on a web page. When the button is clicked by a user, a call is initiated between the user's web browser and the call centre agent. During the call, the context of the call (where the call originated) is carried through the web phone to the agent answering (Mexuar Communications, 2006a).

The main drawback with this BBVC is the cost incurred to obtain it and install it for users to access. This cost comes with the purchase of a license for each Asterisk server on which it would be used. Another disadvantage is that calls can only be made to limited users such as a company's call centre or sales centre agents.

JIAXC is another Java based BBVC created by Mikael Magnusson, based on the IAXClient library and the Java Native Interface (JNI) library (Magnusson, 2006). Unlike Mexuar Communications' Corraleta Connect, it is an open source application and it comes with a demo application and a source code for developers. It provides support for the IAX VoIP protocol and codecs such as GSM, Portaudio, G.711 and Speex (Magnusson, 2006).

Unlike JSAP, JIAXC does not require the installation of JMF. JIAXC also has an extension library for the JVM which, depending on the type of OS used, may require PC administrative rights. The applet can be accessed using Java compliant web browsers on MS Windows and Linux OSs. Though the demo application can connect to any Asterisk server, it cannot be used to provide nomadic accessibility to the Asterisk service.

2.5 Usability and Evaluation Models for BBVCs

There are many evaluation models used for web application and VoIP applications. Some of these models are presented in sections 2.5.1 to 2.5.2.

2.5.1 VoIP Evaluation Models

There are many evaluation models used for VoIP applications. The most commonly used models are the International Telecommunication Union, the Telecommunication division (ITU-T) E-Model and the ITU-T Perceptual Evaluation of Speech Quality (PESQ).

2.5.1.1 The E-Model

The ITU-T E-Model was established as a computational model for evaluation of VoIP applications by the ITU-T because the actual performance of VoIP applications depends on user perception (Hardy, 2003; Bacioccola, Cicconetti & Stea, 2007). The model helps to ensure that VoIP users will be satisfied with the application's end-to-end voice transmission. The E-Model defines a quality factor, R score, to capture the effect of mouth-to-ear delay and losses in packet-switched networks. The R score is then mapped to the Mean Opinion Score (MOS), which in turn is converted to subjective quality levels such as "Poor" or "Good".

According to ITU-T in Hardy (2003),

In this model, the overall transmission rating *R* is defined by the equation:

$$R = R_o - I_s - I_d - I_e + A$$

where R_o - transmission rating based on the signal-to-noise ratio

I_s - effects of a combination of impairments that "occur more or less simultaneously with the voice signal"

I_d - effects of impairments due to delay

I_e - degradation of quality caused by low bit rate codecs

A - "compensation of impairment factors when there are other advantages of access to the users"

The formulas for putting all the E-Model factors and sub factors together to calculate the various transmission ratings and rating adjustments are in the ITU-T Recommendation for E-Model (Hardy, 2003). These instructions are accompanied by transforms that define the conversion of the indicator *R* into measures of MOS and percentages of calls that are "Good" or "Better" (GOB) and "Poor" or "Worse" (POW).

2.5.1.2 The ITU-T Perceptual Evaluation of Speech Quality (PESQ)

The PESQ was established by ITU-T as an objective method for predicting the subjective quality of narrowband headset telephony and narrow-band speech codecs (Hardy, 2003). It is used as a user-perception model in the area of telephony networks.

According to Beuran (2006), PESQ takes into account filtering, variable delay, coding distortions and channel errors. PESQ has been used to demonstrate acceptable accuracy for many telecommunication applications including codec evaluation and selection, live network testing using digital or analogue connection to the network, testing of emulated and prototype networks. PESQ score computation requires both the original and the degraded voice signal; therefore it is an intrusive method. The key process in PESQ is the transformation of both the original and degraded signal into representations analogous to the psychophysical representation of audio signals in the human auditory system. The PESQ score is mapped by design to a MOS-like scale, a number in the range of -0.5 to 4.5, although for most cases the output range will be between 1.0 and 4.5, the normal range of MOS values found in subjective listening quality experiments (Beuran, 2006).

2.5.1.3 E-Model Versus PESQ

Beuran (2006) compared E-Model’s R-value with the PESQ score. He stated that:

> Being given only by mathematical formulas, the E-model's R-value is very easy to compute once the values for its parameters are decided. Note however that the values of these parameters are provided only for a pre-determined range of conditions (i.e. specific codecs etc.). Therefore this model cannot be used, for example, to test the deployment of a newly developed codec since the associated parameters for this codec are not provided by ITU-T. On the other hand the PESQ score is computed based on the original and the degraded waveforms, hence the codec or other experimental conditions are irrelevant for its computation. However, effectively obtaining both these

> waveforms may be challenging, depending on the particular experimental conditions. (pg. 25).

The E-model requires appropriate network traffic measurements to get parameter values to arrive at its R-value while PESQ requires estimations based on reliable voice recording capabilities.

Given the nature of this study, none of these models is applicable because the interest of this study is on using a web application as a VoIP application. Web applications are mostly evaluated based on how the users perceive its usability and usefulness, and none of the given models take these two factors into consideration (Lund, 2004; Pressman, 2005).

2.5.2 Usability Models

There are different models used to evaluate web based applications and one of the models is the Technology Acceptance Model (TAM).

TAM was introduced by Davis (Davis, 1989) to predict Information Technology acceptance and usage. He emphasizes that user's behavioural intention to use technology is affected by their perceived usefulness and perceived ease of use of the technology. TAM has strong behavioural elements, assuming that when someone forms an intention to act, that they will be free to act without limitation. In practice constraints such as limited ability, time, environmental or organisational limits, and unconscious habits will limit the freedom to act (Bagozzi et al, 1992).

TAM has also been used to assess users' perceptions in the area of Information Technologies including Electronic Commerce (McCloskey, 2004). It has also been used to construct many questionnaires, including the "Usefulness, Satisfaction, and Ease of Use" (USE) questionnaire for measuring usability (Lund, 2004). The USE questionnaire was constructed to collect information about users' perceptions about the usefulness, satisfaction and the ease of use of software application.

The model applicable to this study is the USE questionnaire model. It can be used as a base construct for getting users' perceived usability and perceived reliability about BBVCs and their perceived usefulness for Asterisk.

2.6 Comparison of Some of the Existing BBVCs

BBVCs are dependent on the number, and the type, of services they are developed for. The majority of the available BBVCs are deployed for public uses only and are accessible through Internet connection. These BBVCs only work with VoIP services similar to, and with, the developers' VoIP services. The other minority of the available BBVCs need to be either purchased from their developers or be upgraded and customised to work with an organisation's VoIP services.

A customised BBVC for an organisation's Asterisk VoIP services implies that the BBVC's web phone can be used with most, if not all, the VoIP protocols supported by Asterisk as long as the web phone supports one of its main VoIP protocol - IAX. This is based on Asterisk's capabilities to link calls from different VoIP protocols together (Van Meggelen, Madsen & Smith, 2007). With a well designed web phone, the users' perceptions about the usability of the BBVC will also improve.

2.7 Summary

This chapter addressed the literature review on the accessibility of VoIP services, web technologies, some existing BBVCs, and the features of BBVCs. Problems with existing BBVC were also addressed. Finally, evaluation models were presented.

There are several ways to access VoIP services using hard phones or soft phones. The use of VoIP clients, which requires web browsers, may depend on factors such as VoIP services, web technologies and the features of the BBVC. Although there are many BBVCs available, only JIAXC was found most appropriate because it could be customised to support an organisation's Asterisk VoIP services. From the different evaluation models available, the USE questionnaire model, which is based on the TAM model, is found appropriate for this study because it focuses on users' perceptions on usability and usefulness of web applications.

CHAPTER THREE

RESEARCH METHODOLOGY

The purpose of this chapter is to discuss the research design, and the research methodologies used to provide answers to the research questions. The research instruments and environment setups will also be discussed.

3.1 Introduction

The research methodologies used were directly linked to the research questions of this study. For the purpose of clarity, the research questions are stated below:

1.) Can available BBVCs be customised to provide nomadic accessibility to Asterisk VoIP call services in DCS-UNAM?

2.) What are the challenges experienced by customising the features of those BBVCs to provide nomadic accessibility to Asterisk VoIP call services?

3.) How do the challenges affect the use of the BBVCs to access Asterisk VoIP call services?

4.) What are users' perceptions about the use of the BBVCs for Asterisk?

3.2 Research Design

Both qualitative and quantitative research methods were used for this study. The qualitative research method of a descriptive type was used in providing answers to research questions one, two and three. The quantitative research method of an empirical type was used in providing answers to research question four.

3.2.1 Methodology for Answering Research Question One

An in-depth survey was conducted on available BBVCs that can be used for VoIP services in terms of the features they offered and how they can be used with Asterisk VoIP server. Provision of support for nomadic users was also considered. Each BBVC was setup and tested to know how they work, their compatibility with Asterisk VoIP server and their support for nomadic accessibility. Details about the procedures and results are presented in Chapter 4 of this thesis.

Using the Web Engineering process techniques (Pressman, 2005), a website called BBVC4A was developed for use in DCS-UNAM. The website served a BBVC for Asterisk VoIP server. The development of the website was also based on the results of the in-depth survey conducted.

Wireshark Network Protocol Analyzer (Wireshark, n.d.) was used to capture network traffic data generated by the BBVC, in the form of packets, during the testing of the website. These packets were analysed using Wireshark to view the flow patterns of the data.

3.2.2 Methodology for Answering Research Question Two

This involved first hand documentation of the challenges experienced in the customisation of the features of BBVC4A and its web phone. It was based on the researcher's practical point of view. Details about the environment setup and the challenges are discussed in Chapter 5 of this thesis.

The Gnome terminal (GNOME Project, 2009) was used to gather data in form of error messages. These error messages reflected the challenges encountered during the implementation of the JIAXC application (the BBVC4A's web phone).

3.2.3 Methodology for Answering Research Question Three

The Java Error Console for Firefox web browser was used to gather data in form of error messages. These error messages reflected the effects generated during the loading of the JIAXC application as a web phone for the BBVC4A website.

Wireshark was also used to capture VoIP data generated from the JIAXC application, in the form of network packets. These packets were analysed using the IAX2 Call Analyzer after they were converted to *usnf* format using the Unsniff Analyzer. The analysis result was based on VoIP quality metrics to determine the effects on the usage of the customised BBVC.

3.2.4 Methodology for Answering Research Question Four

Data was collected from a random sample of Third year Computer Science students through the use of questionnaire (Appendix A). The selected students used the customised BBVC application (the BBVC4A website and its web phone (JIAXC demo application)) before giving their feedback in the form responses through the questionnaire. The responses were the students' perceptions about the web page containing the JIAXC demo application and also about the use of BBVC as means of accessing VoIP services. The completed questionnaires were collected, checked and subjected to suitable statistical analyses using a statistical software program called STATISTICA version 8.

3.3 Population, Sample, and Sampling Procedure

The population size was fifty-five out of which twelve were randomly picked so that every student will have equal chance to be selected. The students were asked to use the BBVC4A website voluntarily. They were also given questionnaires to fill after using the website.

3.4 Questionnaire Design

A self-administered questionnaire consisting of open-ended and closed-ended questions was used. The questionnaire was designed, based on the USE questionnaire model, to gather the students' perceptions on the BBVC4A website and its suitability for accessing Asterisk VoIP services.

The questionnaire was structured into three sections. The first section was on general information about the users' experiences with VoIP clients. The second section was on the students' experiences with the JIAXC demo application in the BBVC4A website. The third section was on the students' general impressions and comments about the use of BBVCs to access VoIP services as opposed to using other types of VoIP clients. Both the second section and the third section of the questionnaire were designed to capture the students' perceived usability, perceived reliability and perceived usefulness of the BBVC application.

Most of the questions used in the questionnaire had a five-point Likert scale, requiring respondents to rank the degree of agreements, disagreements, satisfaction, and dissatisfaction by ticking the appropriate spaces provided in the questionnaire.

3.5 Questionnaire Testing

The questionnaire was pre-tested for feasibility and to smooth out problems before respondents filled-in their feedback (Struwig & Stead, 2004). This was done, with some Computer Science students who are not included in the study, to find and correct problems respondents may have with the questionnaire instructions or items such as difficulty in understanding the meaning of the words or items.

The completed questionnaires were also subjected to a face validity and internal consistency reliability analysis. Face validity, according to Banks (2005), "is concerned with how a measure or procedure appears. An instrument (questionnaire) is said to have face validity if it "looks like" it is going to measure what it is supposed to measure" (Banks, 2005). The questionnaire was shown to a few DCS-UNAM lecturers for face validity.

Reliability analysis "is use to construct reliable measurement scales, improve existing scales, and evaluate the reliability of scales already in use" (Statsoft, Inc., 2008). A Cronbach's alpha reliability coefficient test, according to Brink (1990), "measures the extent to which the performance on any one variable of an instrument is whether a good indicator or not." The test was done on the completed questionnaires. If alpha coefficient is less than 0.6, reliabilities are considered poor, if within 0.7 ranges, reliabilities are considered fair or acceptable while if over 0.8 ranges, reliabilities are considered good.

With an alpha coefficient of about 0.7 as shown in Table 3.1, the reliability is considered to be acceptable.

Alpha Coefficient	0.657528
Mean	30.5000
Standard Deviation	5.28291
Number of Respondents	12

Table 3.1: Cronbach's Coefficient Alpha Scores for Questions 9 to 21 with Mean and Standard Deviation.

3.6 Environment Setup for Research Questions One, Two and Three

Three PCs were set up: one Ubuntu Linux PC, one MS Windows XP PC and one MS Windows Vista PC. The following data collection instruments were set up:

a.) Wireshark was set up on each PC to capture some network traffic. The captured packets were converted to an *usnf* format using the Unsniff Network Analyzer for easy analysis of the packet. The VoIP data from the converted packets were analysed, using the IAX2 Call Analyser, based on VoIP metrics.

b.) The Firefox Java Error Console on the PC was used to collect (error) messages from the web browser.

c.) The GNOME terminal on the Ubuntu Linux PC was used to collect messages during the building and compilation of the JIAXC applications.

The Ubuntu Linux PC was used as the main platform on which the setup of Asterisk VoIP server, the XAMPP for Linux (LAMPP) as a web service, and the compilation of the JIAXC library, was done. LAMPP is a web content management system used for hosting of web applications. It was also used to host the BBVC4A website. The Asterisk VoIP server was set up such that it provides basic VoIP call services.

More details about the environment used for the development of the web phone are discussed in Chapter 5 of this thesis. The tools used in the development, build and compilation of the JIAXC library and the web phone were as follows:

a.) **The GNU Autotools build system**: for building and compiling the JIAXC library and its prerequisite libraries.

b.) **Java Software Development Toolkit (SDK)**: for building and compiling the Java classes in the JIAXC library and the Java jars of the web phone. It also provides signing tools, keytool and jarsigner, for the web phone.

c.) **The IAXClient library**: the main library on which the JIAXC library is developed on. It is an IP telephony client library that uses the IAX2 protocol.

d.) **GNU GCC compiler system**: for the provision of C and C++ compilation for the JIAXC library and the web phone.

3.7 Environment Setup for Research Question Four

Figure 3.1 shows the test setup for the BBVC4A website and the questionnaire respondents.

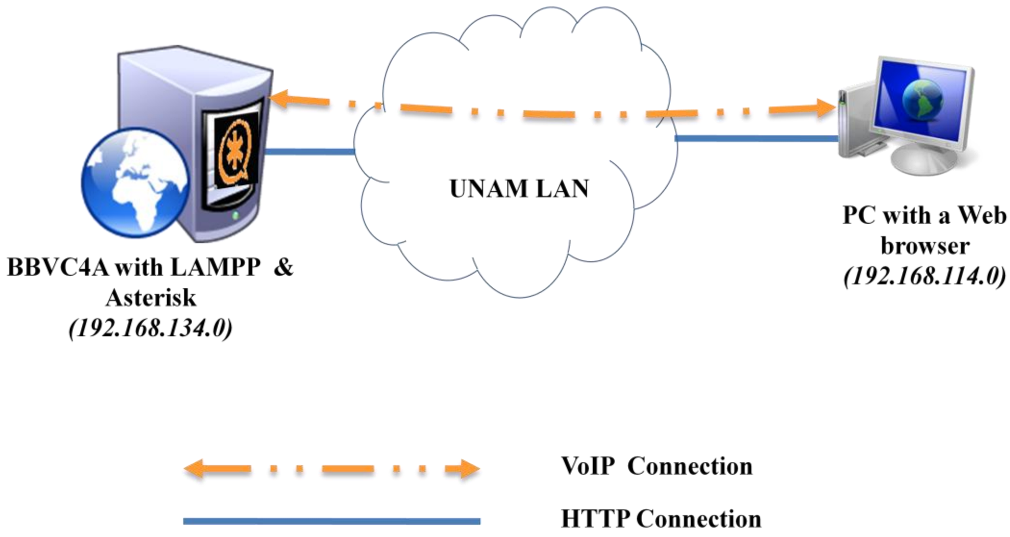

Figure 3.1: Setup for the Testing of the BBVC4A Website.

The BBVC4A website and the Asterisk VoIP server were located on the computer network (192.168.134.0), in a room in the Y-block (UNAM, Windhoek centre). The laboratory used by the students was located on the computer network (192.168.114.0), in the second floor of the E-block (UNAM, Windhoek centre). This setup was done to get the respondents' perceptions about BBVC4A's web phone when it is used in a real life setting.

JRE was installed on all the PCs so as to support the web phone, which is based on the Java language. All the PCs had at least one web browser, which was used to access the website.

The students were divided into two groups, each group with six respondents. Each group used the BBVC4A website at a different time because of the limited space available at the time of the setup. A computer laboratory, wide enough to accommodate one group, was used. Each student was given a headset (microphone and earphone) to be used while using the BBVC4A website. The students were uniformly spaced in the laboratory such that there was less chance of noise interference with their microphones.

A total of 12 students completed their questionnaires after using the BBVC4A website. All the questionnaire respondents filled in their responses. Out of the total of twelve (12) questionnaire respondents, eleven (11) have used at least one VoIP client before while the remaining respondent (1) has not used a VoIP client before. Fifty percent (50%) of the respondents used MS Windows PC and the other fifty percent (50%) used Linux PC. Sixty-seven percent (67%) of the whole respondents used Firefox web browser to access the customised BBVC while the rest of the respondents (33%) used MS IE web browser.

3.8 Summary

The research design and methods, and the research instruments and environment setups used for this study were linked to the research questions. Research questions one, two and three were qualitative in nature because the data to be acquired were based on first-hand observation (from the researcher's point of view) while research question four was quantitative because the data to be collected were feedbacks from users who have tested the customised BBVC application.

CHAPTER FOUR

EVALUATION OF AVAILABLE BBVCS AND CUSTOMISATION OF JIAXC

This chapter presents the evaluation of a number of BBVCs. It will also discuss the customisation of a selected BBVC which was enhanced with the implementation for the BBVC4A website. It will also discuss the analysis and design for the BBVC4A website and the JIAXC demo application as its web phone.

4.1 Introduction

Different BBAs used for voice services came in different forms and packages. They were packaged and utilised as either SDKs for VoIP developers, Click-to-Call applications, or as BBAs with web phones. These applications were either proprietary or open-source. Demo applications and trial versions accompanied some of them for testing, while others allowed live Internet testing. The aim then was to find available BBVCs and available BBAs with support for voice transfer over an Intranet and support for VoIP servers.

4.2 Evaluation Criteria

Some criteria were made to identify the BBVCs that support voice transfer services, including VoIP services. The identification was done based on the popularity of the applications as determined by assessing the most commonly mentioned applications in the literature (Busta Communications, 2008; Conaito.com, 2008; Gizmo5 Technologies Inc., 2008d; MacManus, 2007; Magnusson, 2006; Mexuar Communications, 2006b; Nair & Singh, 2008; National Institute of Standards and

Technology, 2007; Ribbit Corporation, 2008a, 2008b; Silicon Technix, 2007; TringMe, 2008a; VaxVoIP.com, n.d.; Vimas Technologies, 2007; VOIP-Info.org, 2008; Waxer, 2006).

A list of the BBVCs was compiled by selecting those that are free for evaluation as well as trial versions and demo applications. These applications were surveyed, mainly focusing on the web page(s) containing the voice transfer capabilities. Other criteria include features provided, support for web browsers and support for a VoIP server such as Asterisk. Support for customization and nomadic accessibility for an organization's network (Intranet) was also considered.

4.3 BBVCs and the Features Offered

A number of BBVCs with support for voice transfer over the Internet (or Intranet) are available, out of which twelve were identified. The selected applications were Voice Commerce Group's Busta, Conaito's VoIP EVO Enterprise SDK (web demo application), Mexuar Communications' Corraleta Connect (web demo), Gizmo5 Technologies Inc's GizmoCall, Hello2Web, Silicon Technix's iaxClientOcx, Java IAXClient (JIAXC) library demo application (which is a group of applet jars), JAIN SIP Applet Phone (JSAP), British Communications plc's Ribbit (formerly owned by Ribbit Corporation), TringMe's TringPhone, Vax VoIP SDK (Web demo application) and VIMAS Technologies' Web Voice Chat.

Each of the twelve applications was evaluated based on the features provided, the support for web browsers and the support for a VoIP server such as Asterisk. A comprehensive matrix of the features of each was drawn up and, is available in Appendix B. From the results of this evaluation process, fourteen common features present in majority of the applications were identified. Table 4.1 shows these features summarised along with their prevalence in each application.

Feature	**Description**	**Prevalence**
Dial pad to type in numbers	A group of numerical, characters and function buttons to manipulate the contents of the entry field of the web phone.	6 out of 12
Dial pad to send DTMF signals	A group of numerical and characters buttons to send tones or signals from the web phone to the Asterisk server and other phones.	7 out of 12
Dial user	A button that allows the web phone to connect to the address entered in the entry field through Asterisk server.	12 out of 12
Receive (Answer) calls from other users	A button that allows an incoming call to be answered by the user.	11 out of 12
Call Ignore (or Reject)	A button that allows a user to ignore or reject incoming calls.	10 out of 12
Hang up (End) call	A button that allows a user to end an existing call connection.	11 out of 12
Message waiting indicator	A portion of the web phone that display the status of the phone and a Call.	11 out of 12

User's presence	A portion of the web phone that shows the registration status of the web phone with the Asterisk server.	8 out of 12
Automatic selection of best audio codec	Codecs switched during a call in response to changing network conditions. This is based on the remote party's capability, available bandwidth and network condition.	9 out of 12
Clear entry field (button, keyboard or mouse)	A button, or entry field property, that allows the user to clear the contents of the entry field of the web phone.	10 out of 12
VoIP audio codec support	Support for at least one audio codec including GSM, Speex, G.711 aLaw and G.711 uLaw,	7 out of 12
VoIP protocol support	Support for at least one VoIP protocol including IAX and SIP.	10 out of 12
Web technology support	Web technology on which the web phone is developed to extend the functions of web browsers.	12 out of 12
Login (Logout)	This allows users to register (deregister) with (from) the Asterisk server when the web phone is being loaded (unloaded).	10 out of 12

Table 4.1: Summary of Common BBVC Features

Not all the features and the functionalities of a typical VoIP client were considered because the focus was on the suitability of the applications as a BBVC that can be customised to provide nomadic accessibility to Asterisk's VoIP call services.

4.4 Evaluated BBVCs

Five of the twelve BBVCs were served from Europe and USA. These are British Communications plc's Ribbit (formerly owned by Ribbit Corporation), Gizmo5 Technologies Inc's GizmoCall, Mexuar Communications' Corraleta Connect (Web demo), TringMe's TringPhone and Voice Commerce Group's Busta. They are hosted by their developers' companies with little or no options of allowing users to run the applications from a server within a private network. Some of these manufacturers do offer additional services for LAN services but at high prices ranging between N$ 200 and N$ 13200. Accessing the developers' services from Namibia requires more use of international network bandwidth which is expensive and also results in slower response time. The drawback of slow response time makes the voice quality of the applications less reliable, less consistent and nearly poor.

The web demo applications of Conaito's VoIP EVO Enterprise SDK and Vax VoIP SDK have restricted features and functionalities. A purchase of a license or registration key, and thorough understanding of the SDKs are required before any application can be developed. Another application requiring the purchase of a registration key is VIMAS Technologies' Web Voice Chat for the activation of its voice component.

Of the twelve, four BBVCs, including Mexuar Communications' Corraleta Connect, support more than one PC OS and, also Asterisk. The other three applications are the JAIN SIP Applet Phone (JSAP), Silicon Technix's iaxClientOcx and Java IAXClient (JIAXC) library demo application. JSAP could not be initiated on the web page that accompanied it because JSAP has not been upgraded to use the latest JAIN SIP v1.2. The required upgrading of JSAP was considered beyond the scope of the study, mainly because of the time and effort it will require.

Silicon Technix's iaxClientOcx had the many of the features required for a BBVC (Appendix B), but with major disadvantages. Although designed to be utilized using Active Server Pages (ASP) and PHP scripting, it is mainly based on the ActiveX technology. The ActiveX web technology – supported only by MS IE – makes the iaxClientOcx only usable in MS IE web browser. This is a disadvantage for users who regularly use other web browsers such as Mozilla Firefox. In addition, the BBVC is only supported on MS Windows – hence, Linux users and other OS users can not make use of the application. The drawbacks of the iaxClientOcx and JSAP make them less favourable for the study.

The JIAXC library demo application (a combination of HTML and a group of applet jars) has limited features (shown in Appendix B). The JIAXC library uses an older, and out-dated, version of the IAXClient library (0.0+CVS20060520, as at 20th May, 2006). This causes problems for its installation on recent Linux OS. Despite its drawbacks, it provides support for more than one PC OS – MS Windows and Linux – through its dependence on Java technology. This indicated the need to create a new JIAXC application based on the demo application and the JIAXC library. The JIAXC library demo application was chosen as the web phone for the BBVC that will be customised and used to provide nomadic accessibility to the Asterisk server setup in DCS-UNAM.

4.5 Observations

Based on the results of the evaluation of the BBVCs, two of the BBVCs (iaxClientOcx and JIAXC) could be customised to provide users with nomadic accessibility to Asterisks services. Though both support IAX VoIP protocol, iaxClientOcx provides more VoIP related features than JIAXC. On one hand, the iaxClientOcx application is proprietary and has a free demo API which can be used, along with its web phone, only for MS IE web browser. This makes the iaxClientOcx application less beneficial for users in that it only supports one web browser.

The JIAXC demo application, on the other hand, supports all Java-enabled web browsers on MS Windows and Linux PCs, but with less features. In addition, the JIAXC demo application, as a BBVC, provides support for one user, but can be customised to provide support for many users. This raises the need to develop a custom built BBVC around the JIAXC demo application.

A custom BBVC (BBVC4A website) was developed for the Asterisk setup at DCS-UNAM. It contains the JIAXC demo application (applet jars) serving as its web phone. The website utilises a combination of HTML, AJAX, PHP and Java Applet for its functionalities. PHP scripting is used mainly to provide required parameter values to the JIAXC demo application from server source – MySQL database.

4.6 Requirements

The custom website needed to address the problems identified in evaluated applications in sections 4.4 and 4.5 of this chapter. In order to achieve this, the custom BBVC needed to:

a. Be locally served. That is, it should be hosted within an organisation's Intranet (or LAN) without requiring Internet connection from an Internet Service Provider (ISP). This will also improve the quality of the VoIP calls.
b. Be simple to install on a web server and use with an Asterisk IP PBX setup (that is, an Asterisk VoIP server).
c. Require minimal or no configuration of the web browser by the website users.
d. Be accessible by the website users irrespective of the web browser used and the network address of the PC used.
e. Make provision for dynamic allocation of a user's Asterisk values to the web phone. This is a part of the login (logout) feature of the BBVC.

4.7 Use-Case Diagram and State Diagrams

A Unified Modelling Language (UML) Use-case diagram indicates the functions that will be performed by the users through the custom BBVC. Figure 4.1 shows the use-case diagram.

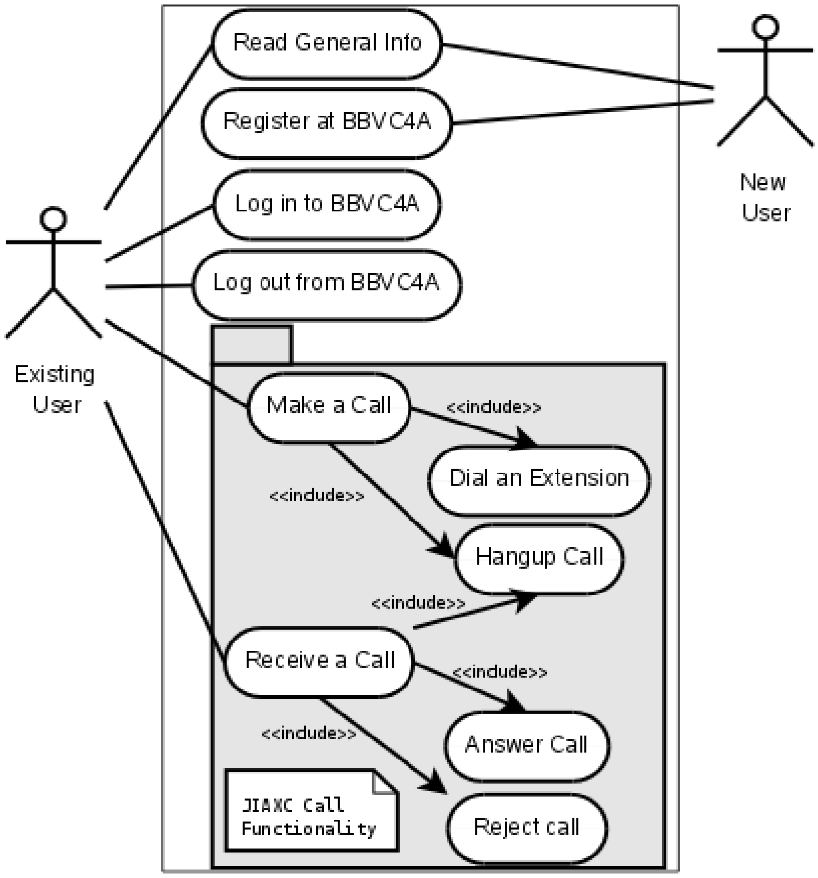

Figure 4.1: Use-Case Diagram for the BBVC4A Website.

There are many types of users that can use this custom website but only two are chosen: Existing User (a registered BBVC4A user with a valid Asterisk account) and New User (an unregister BBVC4A user with a valid Asterisk account). An Existing user starts off at a general *Welcome* interface from which he or she can choose to log in to the website. The *Login* interface (detailed in section 4.8.1 of this chapter) allows the Existing users to enter their authentication details which are then verified. A New user also starts off with the general *Welcome* interface from which he or she can choose to register with the website.

Both the Existing users and the New users must have a valid Asterisk account to be able to connect to the Asterisk server from the website. The acquisition and allocation of Asterisk accounts depends on the organisation's (or the department's) policy.

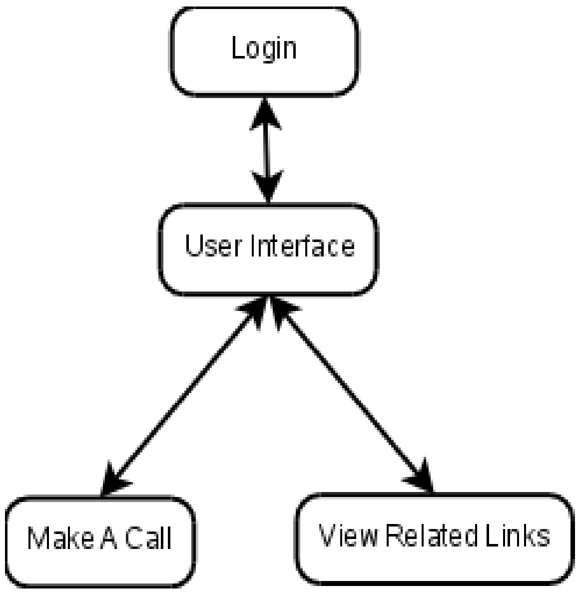

Figure 4.2: State Diagram for the BBVC4A Website.

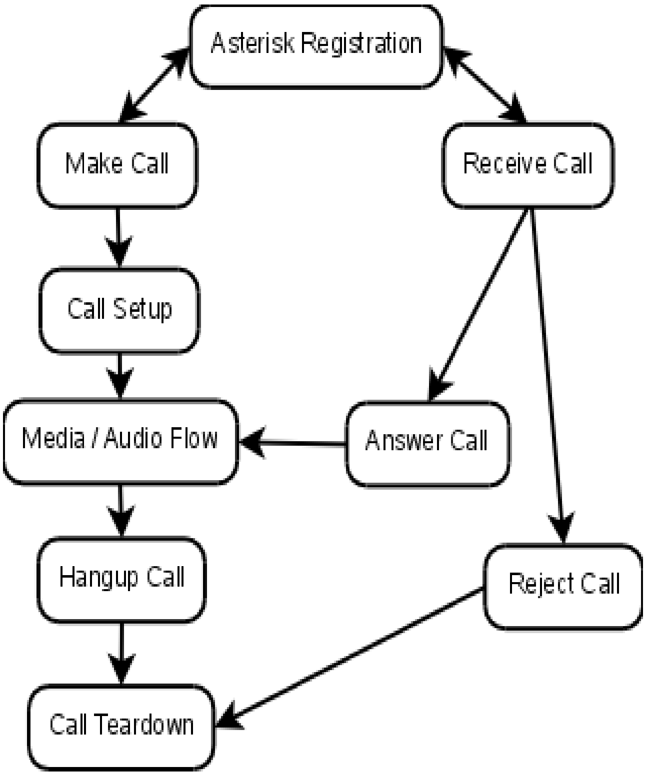

Figure 4.3: State Diagram for the JIAXC Demo Application.

The UML state diagrams in Figure 4.2 and Figure 4.3 indicate the overall states for the website and the JIAXC demo application (the website's web phone). More details about the UML diagrams – Use-case diagram and the State diagrams – are discussed in Appendix C.

4.8 BBVC4A Application Interface Design

Separate interfaces were designed for New users and Existing users. The interfaces for the New users can be accessed by all types of users including Existing users who have not logged in to the website. The interfaces are also made available for Existing users who have logged in to the website, with the exception that the interfaces can not be accessed by New users.

4.8.1 The *Welcome* Interface

Figure 4.4 shows the algorithm for directing the Existing user from the *Login* interface to the *Welcome* interface.

```
Prompt user for username and password
If (username equals to the database value AND password equals to
database value) then
        Allow access to the Welcome page for Existing users
        Set the user's status to "not yet connected to Asterisk"
Else
        Take the user back to the Login page
        Display the login errors
        Prompt the user for correct username and password
End
```

Figure 4.4: Algorithm for the *Login* Interface.

The algorithm assesses the authentication details a user enters and responds accordingly. The authentication details are the username – in form of student number for DCS-UNAM – and password. The *Login* web page provides two text boxes, *username* and *password*, and one *login* button to accept the authentication details and respond to a mouse click or key press. The user is directed to a "Login Error" page if the authentication details are not correct. From this web page the user can choose to retry the login process. If the authentication details are correct, the user will be redirected to the *Existing User* interface.

4.8.2 The *Existing User* Interface

Figure 4.5 shows the algorithm for the *Existing User* interface.

```
Check web page type
If (web page type is “info”) then
        Set user’s status to “Not connected to Asterisk”
End
```

Figure 4.5: Algorithm for the *Existing User* Interface.

The Existing user’s interface offers navigation links (and buttons) to the web pages of the website, and more importantly, the web page containing the web phone (JIAXC demo application applets). The link to the web phone is found in the “Make A Call” button which takes the user to the *“Make A Call”* interface.

4.8.3 Existing user's "*Make A Call*" Interface

Figure 4.6 shows the algorithm for the "*Make A Call*" interface.

Check the web page type
If (web page type is "phone") then
    Get user's Asterisk values (username, password and host
        IP address for Asterisk) from cookie
    Send user's Asterisk values to web phone
    Load web phone into web page
    Set user's status to "connected to Asterisk
    Get the list of available Asterisk users
Else
    Set user's status to "not connected to Asterisk"
End

Figure 4.6: The Algorithm for the "*Make A Call*" Interface.

The interface offers the website's web phone and the control over it. It also offers the user's Asterisk account details. From this interface the user can make calls to other users of the same Asterisk server and also receive calls from them. The algorithm in Figure 4.6 is the same for all other web pages seen after logging in to the BBVC4A website.

4.9 Web Server Setup

A dummy website manager and a "User" database table – a data object for the customised BBVC – are created for the BBVC4A website. The dummy website manager – an administrative user of the web server – has total control over the data stored in the database table. Table 4.2 shows the details about the contents of the database table. The contents of the database table will be utilised by the BBVC4A website. Detailed information about the data object model is presented in Appendix C of this thesis.

Field	**Data type**	**Constraint**	**Comment**
userid	INT(12)	NOT NULL, AUTO_INCREMENT, PRIMARY KEY	User Identification (ID) for PHP session
busername	INT(20)	NOT NULL	Username or Student number
bpassword	VARCHAR(20)	NOT NULL	User's password
bnickname	VARCHAR(20)	NOT NULL	User's Nickname
butype	VARCHAR(12)	NOT NULL	User type – administrator or user
asteruserid	VARCHAR(20)	NOT NULL	User's Asterisk username
asterpassword	VARCHAR(20)	NOT NULL	User's Asterisk password
asteripadd	VARCHAR(15)	NOT NULL	User's Asterisk IP address
asterextno	INT(10)	NOT NULL	User's Asterisk's extension
asterptype	VARCHAR(10)	NOT NULL	Asterisk protocol to

			be used
asterreg	VARCHAR(5)	NOT NULL	True, if the user needs to register to Asterisk.
asterpcstatus	VARCHAR(50)	NOT NULL	Check if the JIAXC applet' connection status to Asterisk.

Table 4.2: Details About the "User" Database Table.

4.10 Summary

The evaluation of a number of BBVCs was presented in this chapter. Out of all the features each application possessed, fourteen most common features were identified. Though each application had its own benefits and drawbacks, the JIAXC library application was the most appropriate BBVC that could be used for this study.

Based on the results of the evaluation, the need for a customised BBVC for the Asterisk VoIP server setup in DCS-UNAM was realised. Despite having limited number of features, the JIAXC demo application could be utilised as part of the development for a customised BBVC. The design for the customised BBVC – BBVC4A website and the JIAXC demo application – were also presented in this chapter.

CHAPTER FIVE

DESIGN IMPLEMENTATION

5.1 Introduction

This chapter discusses the implementation of the design for the BBVC4A website and the JIAXC demo application as its web phone. The configuration for the Asterisk server for the BBVC4A website will also be discussed. The issues encountered with the JIAXC library will also discussed.

5.2 Development Environment

This section presents the components used in the development environment for the implementation of the design in Chapter 4 of this thesis. Ubuntu 8.10 was chosen as the main OS platform for the development environment because the Asterisk setup used in DCS-UNAM is on an Ubuntu platform. The Ubuntu setup was run on a Pentium 4 with a 2.8 GHz processor and a 512MB memory.

5.2.1 Setting up of LAMPP Server

The LAMPP server (XAMPP for Linux) was chosen as a web server to host the BBVC4A website. It was installed in to the Ubuntu PC. A dummy website manager – an administrative user of the LAMPP server – and a “User” database table were created, using MySQL, for the BBVC4A website. The contents of the database table will be utilised by the BBVC4A website through PHP scripting.

Cookies, containing PHP sessions about users who are logged into the website either as registered users or registering users, were used. The main benefit of using cookies is that it can be used to store some of the users' details on the web server so that they can be easily retrieved:

a. To send required users' details to the web phone (JIAXC demo application) dynamically and
b. For authentication purposes.

5.2.2 Setting up Asterisk VoIP Server

For this study, a separate Asterisk (version 1.4.22) setup was made as a testing server. This was done to save time and cost of configuring or re-installing the actual Asterisk VoIP server used in DCS-UNAM, should something go wrong with the system.

Both the Asterisk server setup and the LAMPP setup were placed on the same Ubuntu PC because of limited availability of hardware resources at the time of this study. This later proved to be effective as both Asterisk and LAMPP are designed to work with minimal PC resources available. Both the Asterisk server and the LAMPP server are also configured to be accessed from any network.

Since the only VoIP protocol supported by the JIAXC library is IAX, IAX protocol was chosen as the main VoIP protocol to use along with the audio codecs required. The GSM codec was selected as the main audio codec for the study because it requires minimal use of available bandwidth.

About ten dummy IAX user accounts were created along with a MACRO-based dial plan for their respective extension details and voicemail accounts. The configuration of the Asterisk VoIP server included new dial plan and extensions for the dummy accounts. Table 5.1 shows a summary of an IAX user's context parameters and its purposes.

Parameter	**Purpose**
iax context-name	Indicates the user's IAX configuration's name. It is also used as the user's username for authentication methods.
host	To allow incoming IAX connections to be opened whenever it is used. It is set to "dynamic" to support nomadic users.
secret	A user's secret (or password) for authentication methods.
type	To identify the type of user. In order to support nomadic accessibility, it is set to "friend" for both incoming and outgoing calls.
qualify	To tell Asterisk to test whether a user is connected to Asterisk before attempting to connect a call.
callerid	To set the Caller ID information for a user.
context	To link the user's IAX configuration to Asterisk's dial plan.

Table 5.1: Asterisk IAX Parameters and Purposes.

The "host" parameter of the user' context was set to dynamic so that the nomadic users can access the Asterisk server from any part of the IP network. This will require the user to always register, and unregister, the BBVC4A's web phone with the Asterisk server so that the Asterisk server can keep track of the user's location when there are calls to be processed for the user.

5.2.3 Other Components

The Macromedia Dreamweaver 8.0 was used as the main web application development toolkit for the design, the coding, and the development of the web pages of the custom website. Graphical Vi Improved (GVIM), a Linux text editor, was used for the manipulation of the configuration files of the Asterisk server and the website files of the LAMPP server.

The BBVC4A website was implemented used PHP scripting for the server-side and HTML for the client-side of the website. Asynchronous JavaScript and XML (AJAX) scripting was used to get and display available users from the database without having to reload the web page.

5.3 BBVC4A Application Interface Design

Separate interfaces were designed for New users and Existing users. The interfaces for the New users can be accessed by all types of users including Existing users who are not logged into the website. The interfaces are also made available for Existing users who are logged in to the website with the exception that the interfaces can not be accessed by other types of users. Figure 5.1 shows the *General Welcome* interface.

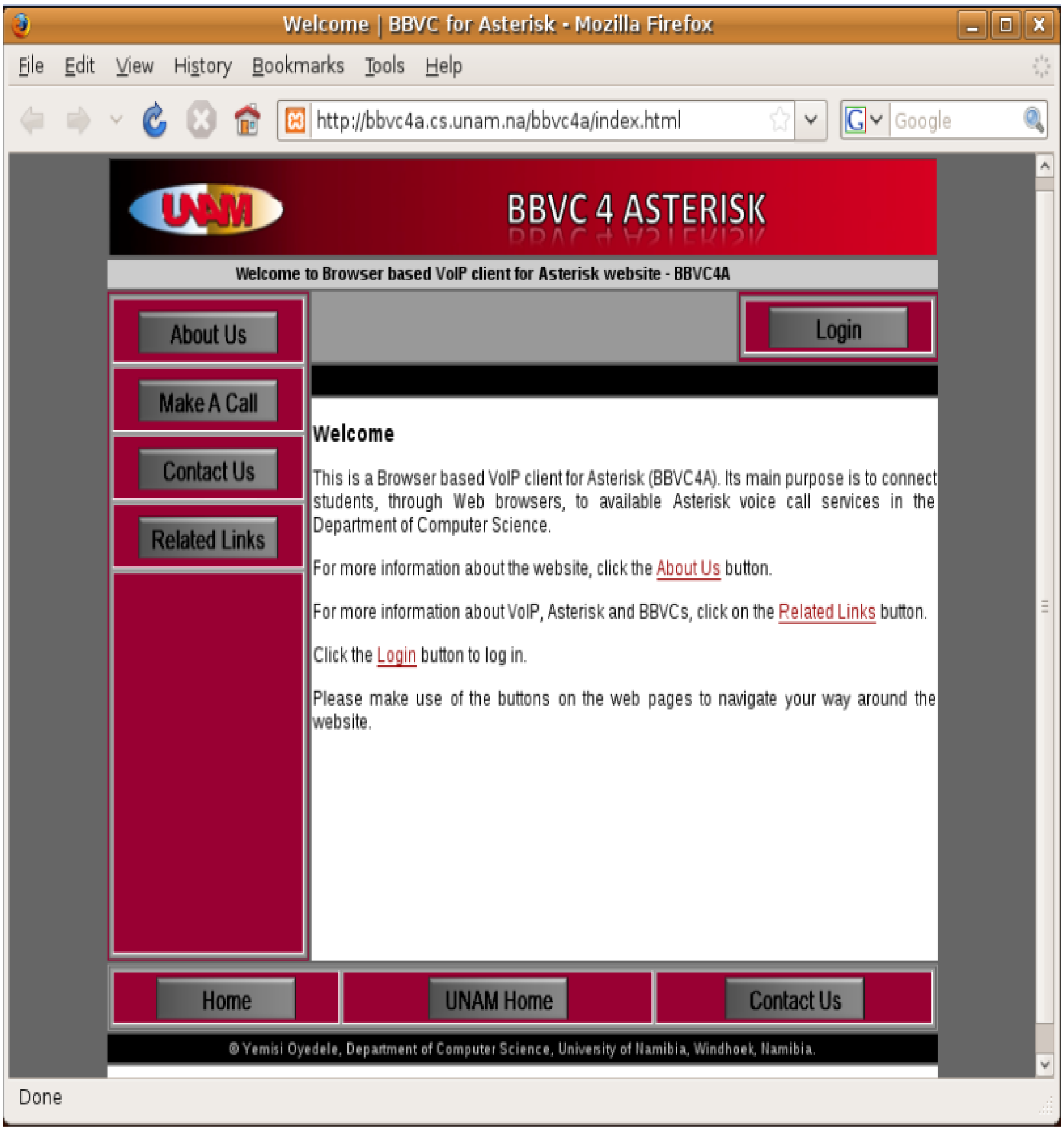

Figure 5.1: The *General Welcome* Interface.

5.3.1 The *Login* Interface

Figure 5.2 shows the *Login* interface.

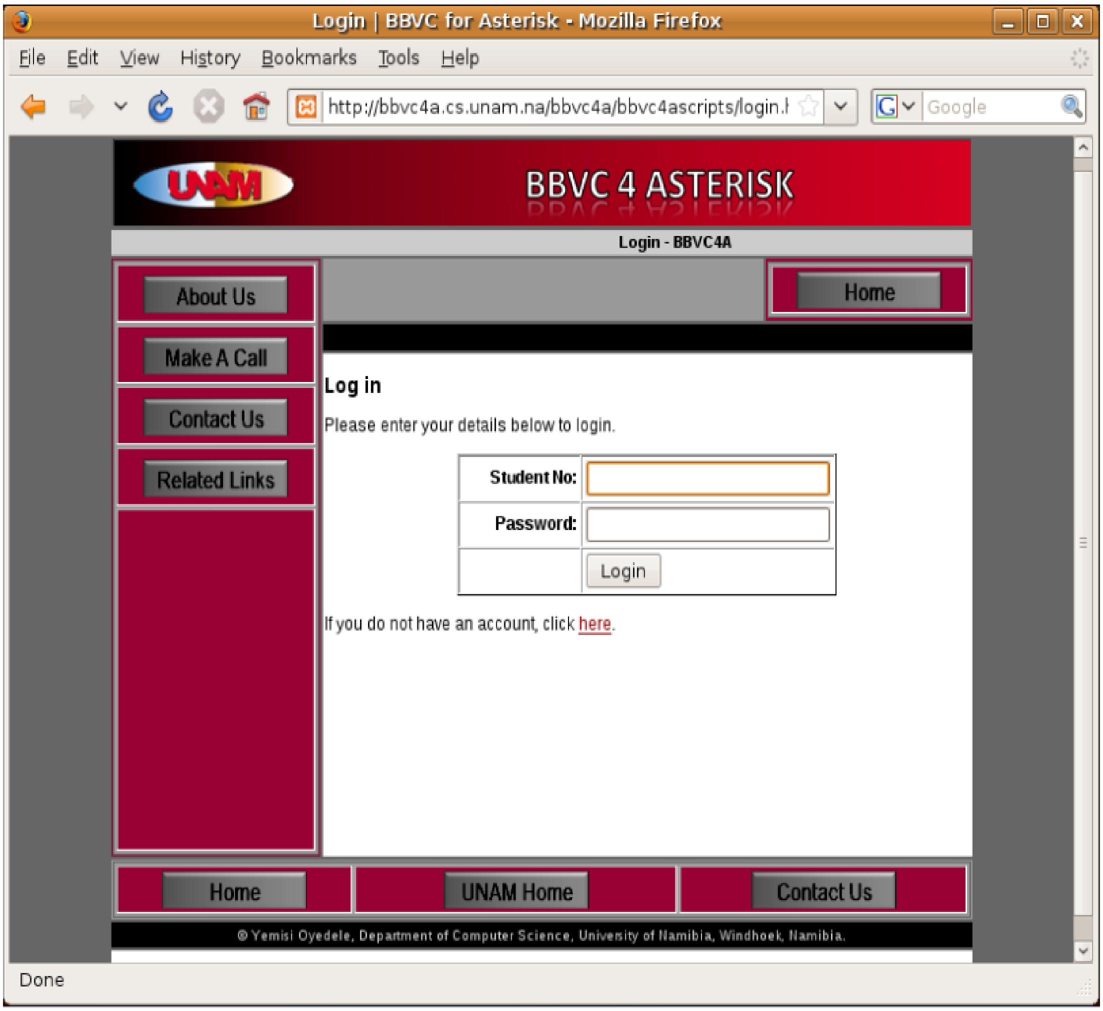

Figure 5.2: The *Login* Interface.

Two text boxes, *username* and *password*, and one *login* button are provided to accept and send the user's details for authentication after a mouse click, or key press, on the *login* button. The authentication details are the username, in the form of student number, and password.

The *Login* interface and its associated PHP script – linked to the *login* button – control the logging in and the authentication processes of users into the BBVC4A website. The user's initial focus will be on the *username* input text box then the *password* input text box. The PHP script for the login button (when the button is mouse clicked or when the "Enter' key of the keyboard is pressed) determines the authenticity of the users depending on the username and password given. The authentication process involves comparing the details entered by the user from the login form with the contexts of the "User" database table in MySQL component of the LAMPP server.

The next appropriate interface is selected based on the results of the authentication process. If the user is valid based on the authentication result, the user is taken to the Existing user's interface as described in section 5.3.2 of this chapter. If the user is not valid, the user is taken to the "Login Error" web page.

5.3.2 The *Existing User* Interface

Figure 5.3 shows the *Existing User* interface, the Existing user's view of the customised BBVC website for Asterisk.

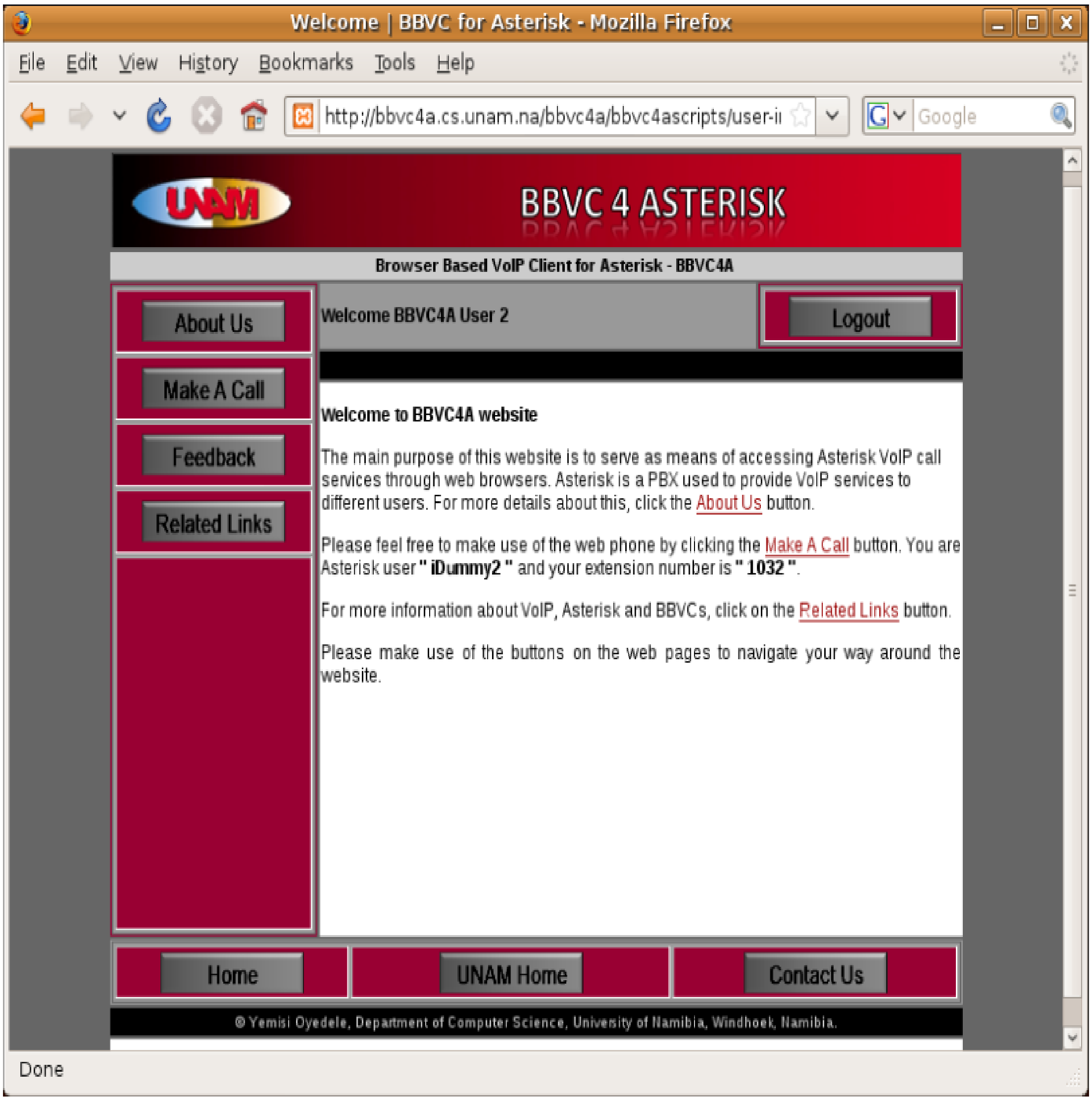

Figure 5.3: The *Existing User* Interface.

The interface in Figure 5.3 offers navigation links (and buttons) to the web pages of the website and more importantly, the web page containing the web phone (JIAXC demo application applets). The link to the web phone is hyperlinked to the "Make A Call" button, which takes the user to the *"Make A Call"* interface.

5.3.3 The "*Make A Call*" Interface

Figure 5.4 shows the "*Make A Call*" interface.

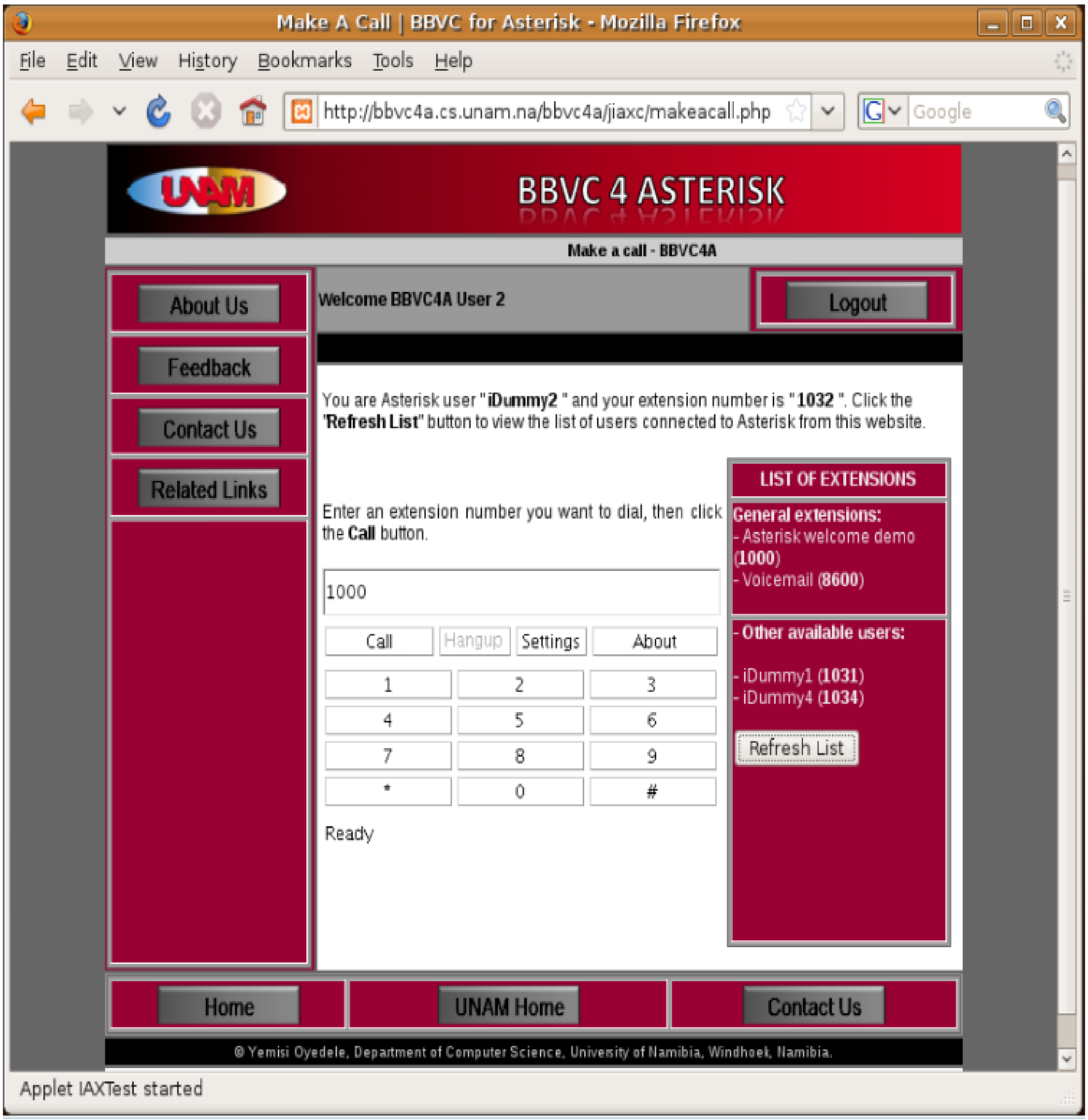

Figure 5.4: The "*Make A Call*" Interface.

The interface in Figure 5.4 offers the BBVC4A website's web phone and the control over it. It also shows some of the user's Asterisk account details, such as the user's (phone) extension number and the user's Asterisk username. In addition, a list of available extension numbers and available BBVC4A users also using the "*Make A Call*" interface on the BBVC4A website, are displayed in the "LIST OF

EXTENSIONS” portion of the interface. From this interface the user can enter a number to dial and make calls to any of the extension numbers in the list and other users of the same Asterisk server, and also receive calls from them.

5.4 Customisation of the Web Page for the JIAXC Demo Application

JIAXC demo application offered twelve of the fourteen features discussed in section 4.3 of Chapter 4 of this thesis, and as presented in Appendix B. Table 5.2 shows the JIAXC demo application features.

• Dial pad to send DTMF signals	• User’s presence
• Dial user	• Input / Output device selection
• Receive (Answer) calls from other users	• Clear entry field (keyboard or mouse)
• Call Ignore (or Reject)	• VoIP audio codec support
• Hang up (End) call	• VoIP protocol support
• Message waiting indicator	• Web technology support

Table 5.2: The Common Features of the JIAXC Demo Application

In addition to these features, the JIAXC demo application provided support for a locally hosted Asterisk server. In other words, the JIAXC application had to be set up in, and be accessed from, the PC containing the Asterisk VoIP server – only one user can access the application at a time. In order to make the application more accessible to more than one user, PHP scripts are used to manipulate the JIAXC demo application’s IAX parameters.

Two major features of a typical BBVC were missing from the JIAXC demo applications. They are the dial pad to type in numbers and send DTMF signals, and clear entry field (button). The addition of these features to the JIAXC demo application involved re-designing of the application.

Adding these features proved more challenging than initially expected as thorough knowledge of the JIAXC library and its dependencies (IAXClient library written in the C programming language, and the GNU build system) were required. A more recent version of the IAXClient library was required because the IAXClient library on which the JIAXC library was based on, and other versions of the IAXClient library, were not installable on the Ubuntu PC, except for version 2.1 beta3. There was limited information available on how to use the JIAXC library and the IAXClient library.

Though there are other features that can be added to enhance the JIAXC demo application, only two of these features – the dial pad to type in numbers and send DTMF signals, and the clear entry field (button) – were added. The other features that can be added to the JIAXC application are discussed in Chapter 7 of this thesis.

All the customisations made to the JIAXC application, including the challenges encountered, are explained in details in section 5.5 to 5.7.6 of this chapter.

5.5 Loading of the JIAXC Demo Application & its Registration.

The JIAXC demo application provides nomadic access for only one user. For the user to connect to Asterisk, the user must have an IAX context and be part of a dial plan in Asterisk. Some values from the Asterisk IAX context are required to be assigned to the IAX parameters of the JIAXC demo application so as to connect to Asterisk.

In order to provide nomadic access to more than one user, the JIAXC parameters are required to be assigned dynamically using information about each user's IAX context. The information about the user's IAX context is stored in a database table, and it includes the user's Asterisk username, the user's Asterisk password (required for IAX registration purposes) and the Asterisk VoIP server class machine's IP address.

For easy retrieval and allocation, the user's information is stored in a cookie using PHP scripts after logging in to the website. The information is then retrieved and sent as values to the JIAXC demo application. As long as the user is logged on to the website, the user's Asterisk information is stored in the cookie. When the user decides to use the JIAXC demo application, by clicking the "Make A Call" button, the "Make A Call" web page (which contains the JIAXC demo application) is loaded.

During the loading of the JIAXC demo application into the web browser, two things happen. Firstly, the user's information is retrieved from the cookie and sent to the JIAXC demo application as its parameters' values. Secondly, the JIAXC demo application attempts to use the parameters' values to connect to the Asterisk server. The process of using the parameters' values to connect to the Asterisk server, and also to inform the Asterisk server of the IP address of the JIAXC application's location is called the IAX Registration. During the second process the JIAXC demo application is loaded into the web browser, provided the user accepts the digital signature of the applet. Figure 5.4 of this thesis shows the web page containing the JIAXC demo application after the loading process.

5.6 Unloading of the JIAXC Demo Application and Deregistration

The JIAXC demo application gets unloaded when the user changes to another web page from the one containing the JIAXC demo application, whilst using the same web browser window. This unloading process also involves the JIAXC demo application unregistering from the Asterisk server.

The loading and unloading of the JIAXC demo application follows the lifecycle of an applet. Figure 5.5 shows the lifecycle of an applet, as defined by Cornell and Horstmann (2005).

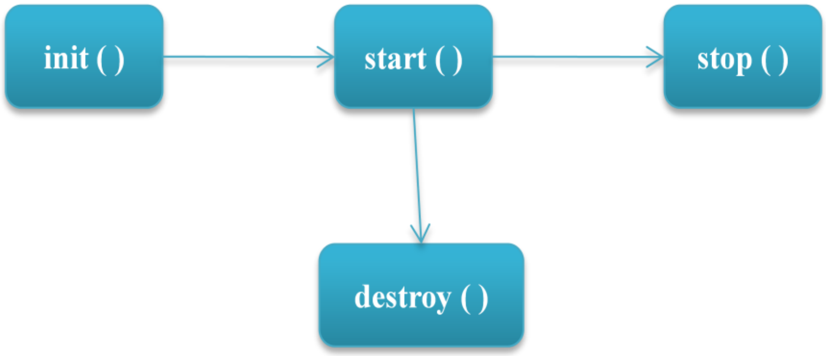

Figure 5.5: Java Applet Life Cycle.

The JIAXC demo application follows the java applet life cycle. The loading process occurs in the first phase of the application's lifecycle (init ()) while the unloading occurs during the last phase of the application (either stop ( ) or destroy ( )). The VoIP process (start ( )) phase occurs in between the loading process and the unloading process and it is the phase where the user can actually use the JIAXC demo application.

5.7 Modifying the Dial Pad

The JIAXC demo application did not provide support for some features. As indicated in the Survey matrix of Appendix B, it does not have a *Clear* button to empty the entry field. The dial pad only supported the sending of DTMF (also referred to as either true tones or tone dial) signals to Asterisk. The customisation made here was the inclusion of a *Clear* button, and to change the implementation of the Dial pad to type numbers and send DTMF signals. The processes involved in adding the features are discussed in sections 5.7.1 – 5.7.6 of this chapter.

5.7.1 Selecting the Appropriate IAXClient Library

The JIAXC library acquired (version 0.0.6) was implemented based on an early version of the IAXClient library (0.0+cvs20060520) as mentioned by the developer, Mikael Magnusson (Magnusson, 2006). Attempts were made by the researcher to get and install the version of the IAXClient library on the Ubuntu PC. The installation was not successful because there have been a lot of changes made to the Linux kernel supported by the library and its prerequisite libraries. Out of all the versions of IAXClient library that could be obtained, only version 2.1 beta 3 could be installed successfully on the Ubuntu PC.

With limited information available, the researcher studied both versions (the 0.0+cvs20060520 version and the 2.1 beta 3 version) and made some observations. The researcher identified some significant changes made to the IAXClient library 2.1 beta 3 version compared to the older IAXClient library, version 0.0+cvs20060520. The changes ranged from the variable declaration, number of method parameters, to the number of classes defined for the library. The major changes are the variable declarations and the number of parameters to be passed to some methods.

5.7.2 Variable Definitions

The overall observation made on the changes of the variable declarations is that some were changed from *double* declaration to *float* declaration, and some from *char* to *string* declarations. Figures 5.6 and 5.7 show some of the variables with different declarations from the two versions of the IAXClient libraries.

```
...
EXPORT double iaxc_input_level_get();
EXPORT double iaxc_output_level_get();
EXPORT int iaxc_input_level_set(double level);
EXPORT int iaxc_output_level_set(double level);
...
```

Figure 5.6: A Snippet of Some Variable Declarations in IAXClient Library 0.0+cvs20060520

```
...
EXPORT float iaxc_input_level_get();
EXPORT float iaxc_output_level_get();
EXPORT int iaxc_input_level_set(float level);
EXPORT int iaxc_output_level_set(float level);
...
```

Figure 5.7: A Snippet of Some Variable Declarations in IAXClient Library 2.1 Beta 3

5.7.3 Method Parameters

There are some methods that have been changed not only with different parameters but also with statements. Some methods have been joined together to form one method. An example is the *iaxc_initialize(int)* method (in the *iaxclient_lib.c* of the IAXClient library) which is called to initiate an IAX phone. In the 2.1 beta 3 version, the *iaxc_initialize(int)* has only one parameter passed to it – *int num_calls* – as shown in the snippet below:

```
EXPORT int iaxc_initialize(int num_calls)  { ... }
```

The same method in the 0.0+cvs20060520 version has two parameters passed to it as shown in the snippet below:

```
EXPORT int iaxc_initialize(int audType, int num_calls)  { ... }
```

Another major difference between the *iaxc_initialize(int)* methods is their statements. The *iaxc_initialize(int)* method of the 2.1 beta 3 version of the IAXClient library is the combination of the *iaxc_initialize(int, int)* and the *iaxc_process_calls(void)* methods of the 0.0+cvs20060520 version.

These changes have effect on how they will be used for the compilation and the upgrade of the JIAXC library. Some of the files directly affected include *jiaxclient.cc*, *LibJiaxc.java* and *Constants.java*, all located in the *jni* directory of the JIAXC library.

5.7.4 Upgrading the JIAXC Library

Though the JIAXC library needed full upgrade to support the IAXC library version 2.1 beta 3, only few changes were made. The most important methods and classes of the JIAXC library were changed to support the IAXC library. The classes included jiaxclient.cc, LibJiaxc.java and Constants.java while the methods included the *iaxc_initialize(int, int)* and the *iaxc_process_calls(void)*.

The jiaxclient.cc was used for the conversion of some of the IAXClient C methods to Java methods through JNI. The *iaxc_initialize(int, int)* and *iaxc_process_calls(void)* methods (as shown in Figures 5.8 and 5.9) were replaced by the *iaxc_initialize(int)* method (as shown in Figure 5.10) in jiaxclient.cc file.

```
...
JNIEXPORT jint
LIBJIAX(initialize) (JNIEnv *env, jobject obj, jint audType,
                              jint nCalls)
{
         int res;

        PRINTF("initialize\n");
        fflush(stdout);
        pthread_mutex_init(&mutex, NULL);
        pthread_mutex_lock(&mutex);
        queue_init(&gQueue);
        running = 0;
        pthread_cond_init(&cond, NULL);
        pthread_mutex_unlock(&mutex);

        res = iaxc_initialize(audType, nCalls); //IAXC
        method to be converted

        iaxc_set_event_callback(event_callback);
        return res;
}  ...
```

Figure 5.8: *iaxc_initialize(int, int)* Method Conversion in the jiaxclient.cc of the JIAXC Library

```
...
JNIEXPORT void
LIBJIAX(processCalls) (JNIEnv *env, jobject)
{
        iaxc_process_calls(); //IAXC method  to be converted
}  ...
```

Figure 5.9: *iaxc_process_calls(void)* Method Conversion in the jiaxclient.cc in the JIAXC Library.

```
...
JNIEXPORT jint
LIBJIAX(initialize) (JNIEnv *env, jobject obj, jint num_calls)
{
        int res;

        PRINTF("initialize\n");
         fflush(stdout);
        pthread_mutex_init(&mutex, NULL);
        pthread_mutex_lock(&mutex);
        queue_init(&gQueue);
         running = 0;
        pthread_cond_init(&cond, NULL);
        pthread_mutex_unlock(&mutex);

         res = iaxc_initialize(num_calls); //IAXC method  to be
converted

         iaxc_set_event_callback(event_callback);
        return res;
}  ...
```

Figure 5.10: *iaxc_initialize(int)* Method Conversion in the jiaxclient.cc in the Upgraded JIAXC Library.

Unused audio variables, such as *int audType, int AUDIO_INTERNAL, int AUDIO_INTERNAL_PA, int AUDIO_INTERNAL_FILE* and *int AUDIO_EXTERNAL*, and methods, such as *void processCalls(),* were removed from the JIAXC library (jiaxclient.cc, LibJiaxc.java and Constants.java files) to avoid wasteful use of the computer's memory space. Some methods also required changes with the declaration of their variables. The methods included *float getInputLevel(), float getOutputLevel(), int setInputLevel(float level)* and *int setOuputLevel(float level)*, which were changed from "double" to "float" declarations.

5.7.5 Adding the Features (Dial Pad and *Clear* Button)

The features of the JIAXC demo application are found in the IAXTest.java file of the JIAXC library (*src* directory). Using the design of the demo application, features were added to the IAXTest.java file. The following snippets in Figures 5.11 and 5.12 show the code for the addition and modification of the features.

```
public Component createButtons( ){
   String labels[] =   {"1", "2", "3", "4", "5", "6", "7", "8", "9",
                                     "*", "0", "#"};
   ActionListener numBtnListener = new ActionListener() {
      public void actionPerformed(ActionEvent e) {
          String action = e.getActionCommand();
          if (!hangupBtn.isEnabled() ) {
            txtEntryField.setText(txtEntryField.getText() + action);
          } else {
            jiaxclient1.sendDtmf(action);
          } //end of actionPerformed(e) method
       }; // end of ActionListener numBtnListener
            ...
// adding the actions to the buttons
   for (int i=0; i < labels.length; i++) {
      Button dialpadButton = new Button(labels[i]);
      dialpadContainer.add(dialpadButton);
      dialpadButton.addActionListener(numBtnListener);
   } //end for loop
} // end of createButtons() method
```

Figure 5.11: Code to Modify the Dial Pad Buttons.

```
...
 // creating the buttons
private Button backspaceBtn, clearTextBtn;
...
//adding the buttons to the Graphical User Interface (GUI)
protected void initUI(Container cnt) {
        ...
        // Clear Text Button
        clearTextBtn = new Button("Clear Text");
        clearTextBtn.addActionListener(buttonLst);
        con.gridwidth = GridBagConstraints.REMAINDER;
        gridbag.setConstraints(clearTextBtn, con);
        cnt.add(clearTextBtn);
        ...
} //end of initUI(cnt) method
...
```

Figure 5.12: Code to Add and Configure the Clear Button.

The Dial pad digit buttons were modified to send DTMF signals and also for entry digits into the entry field of the applet phone. The clear and backspace buttons were added to manipulate the contents of the entry field. Figure 5.13 shows the resulting application after the features have been added.

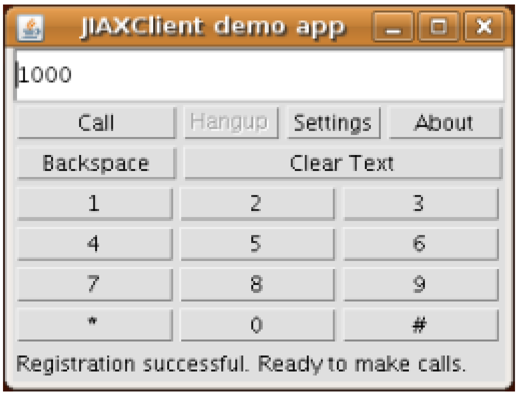

Figure 5.13: The Resulting GUI of the Enhanced JIAXC.

Figure 5.14 shows the resulting web page when the enhanced JIAXC application is used as a web phone for the BBVC4A website.

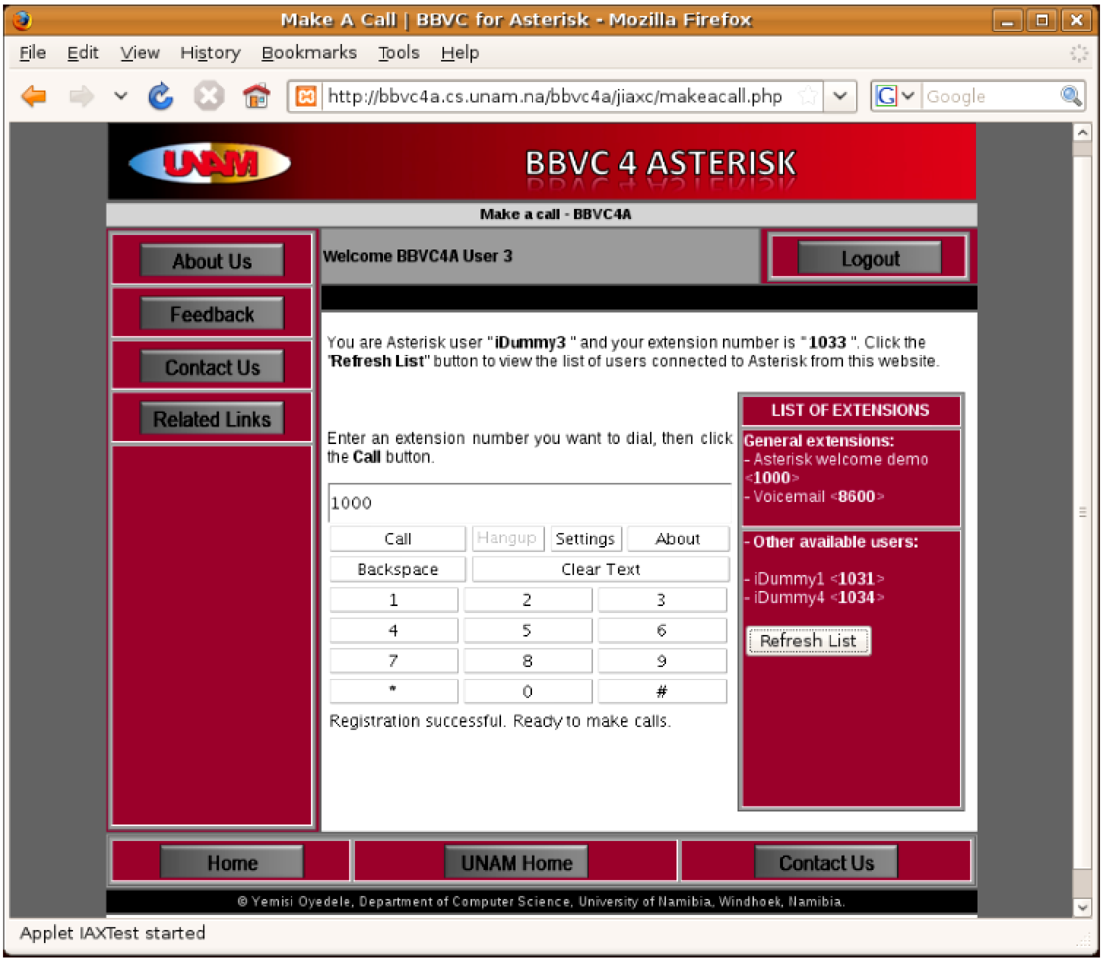

Figure 5.14: The Enhanced JIAXC Application in the BBVC4A Website.

5.7.6 Other Issues and Constraints

There were many issues encountered during the customisation of the JIAXC library and its sample application (apple phone). One of them was the display of the status of the applet phone. Initially, the *statusLine* label of the phone mainly displayed the connection status of the IAXClient library, through the *int textReceived(TextEvent e)* method, which is sometimes not understandable to novice users. It also did not reflect commonly known status displays of commonly used IAX based VoIP clients. Instead the *statusLine* label was manipulated to display the status of the phone with respect to the activities of the phone as indicated in the snippet below:

```
statusLine.setText(Sting  msg);
```

The *msg* variable represents the message to be displayed in the *statusLine* label of the phone.

Another issue was the linking of system libraries for the compilation of the JIAXC library. The JIAXC has Java files and JNI files which helped convert some of the IAXClient C files to Java. The compilation of all the files was made easier by using the GNU Build system which helped with the linking of appropriate C libraries and Java libraries in the Linux system. Initially, these libraries were linked, according to the main “makefile.am” file of the JIAXC library, both statically and dynamically. Attempts to do both linking generated errors indicating that both cannot be done. Using static linking also generated the same errors. Instead the libraries were linked dynamically and no error was generated during the building and installation of the JIAXC library.

Another drawback was the requirement for a user to have PC administrative rights (Super user rights) to use the JIAXC application. Since the upgraded JIAXC application ran only on Linux PCs, attempts were made to modify the application to support the important features of a web phone. After all the modifications were made to the JIAXC's IAXTest.java file and the installation of the JIAXC jar files, the resulting applets jar could not be initiated in a Java-enabled web browser such as Firefox. The errors retrieved from Firefox's error console indicated that the user of the web browser needed privilege to access the system's Java Runtime Environment (JRE). This is so as a result of the attempt made by the JIAXC library to install a small plug-in extension to the JRE supported by the web browser. Another reason is that the addition of a library extension to the JRE installation is prohibited for all users excluding users with PC administrative rights or super user mode.

The upgraded JIAXC application could not be used on the other OSs, such as MS Windows, because the application supports the Linux sound libraries. The JIAXC library and all of its prerequisite programs will have to be cross-compiled to support other OSs.

5.8 Summary

The implementation of the design for the custom BBVC included the configuration of the Asterisk server and the LAMPP web server. The manipulation of the JIAXC library to enhance the JIAXC application required the upgrading of its core library class files to support a later version of the IAXClient library. Other issues with the upgrading of the JIAXC library were also discussed in this chapter.

CHAPTER SIX

DATA ANALYSIS AND DISCUSSION

6.1 Introduction

In this chapter the analysis of data found will be presented in relation to the research questions for this study. The analysis of the captured data from the customised BBVC, the challenges experienced and their effects will be presented. The responses obtained from the respondents through questionnaires will be analyzed and discussed. This discussion will be focused on the users' perceptions about the use of the BBVC for Asterisk. For this analysis, descriptive statistics is used.

6.2 Research Question One: "Can available BBVCs be customised to provide nomadic accessibility to Asterisk VoIP call services in the DCS-UNAM?"

The data to be analysed is the network traffic data generated by the web phone during the testing of the BBVC4A website. The testing of the website was mainly focused on the web page containing the web phone and how the Asterisk values of a user (USER_A on 192.168.122.7) is dynamically allocated to the web phone. The performance of the web phone in BBVCs that performs VoIP functions, like other forms of VoIP clients, depends mainly on the VoIP protocol and audio codec supported and the values assigned to them.

6.2.1 Asterisk Registration

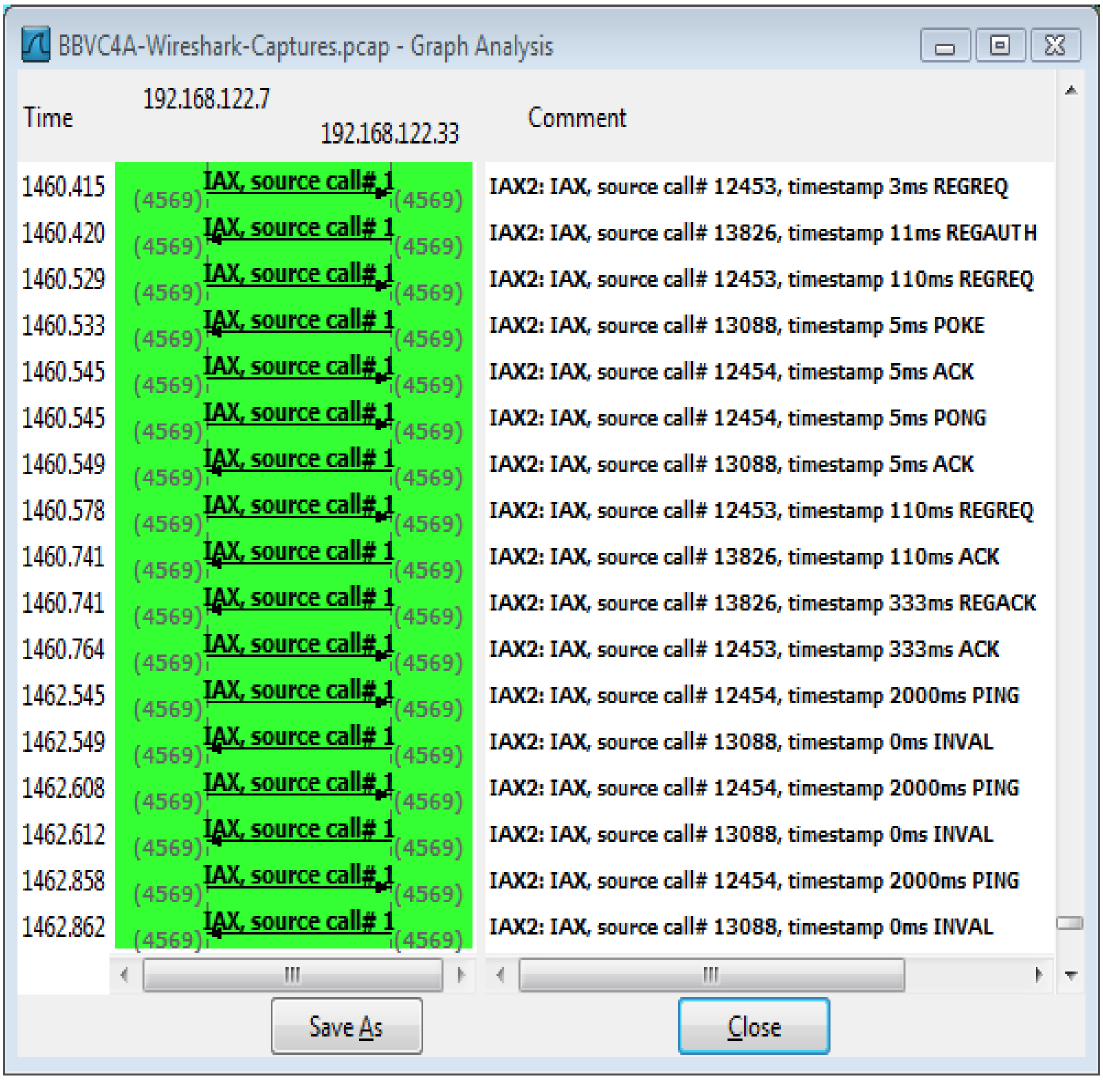

Figure 6.1: Wireshark Flow Graph Analysis of an Initial Connection Between USER_A and Asterisk Server

Figure 6.1 shows the graph analysis for the captured VoIP packets. The arrows in Figure 6.1 indicate the direction of each packet sent as USER_A, through the web phone, attempts to connect with the Asterisk server.

6.2.1.1 Discussion

From the VoIP packets captured by Wireshark, after the loading of the web page with USER_A's Asterisk values, a flow pattern is seen as the web phone (JIAXC demo application) attempts to connect to the Asterisk server (192.168.122.33).

6.2.2 User Call Connection

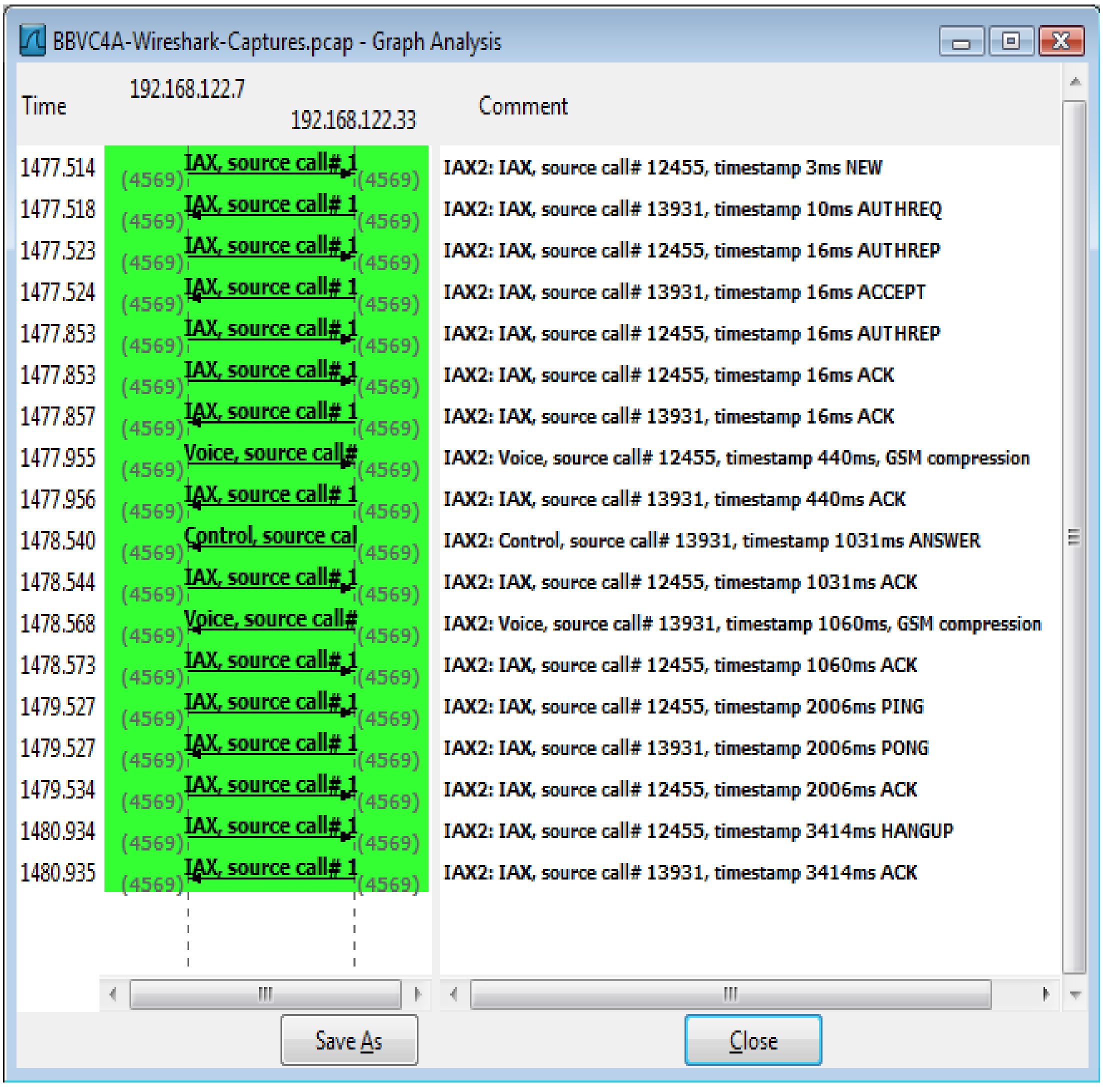

Figure 6.2: Wireshark Flow Graph Analysis of a Call Between USER_A and Asterisk Server

The arrows in Figure 6.2 indicate the direction of each packet sent as USER_A, through the web phone, attempts to connect with the Asterisk server.

6.2.2.1 Discussion

Figure 6.2 shows the initiation and termination of a call. USER_A's Asterisk values are used to create the connection between the JIAXC application and Asterisk. The authentication details (username, password, Asterisk's IP address) were used for registration of the user's location with Asterisk. With the user's location known, a call can be initiated from USER_A to Asterisk and vice versa.

From the analysis of Figures 6.1 and 6.2, both show that user registration, call initiation and call termination were made from another PC to the Asterisk server (on another PC containing a web server for the BBVC4A website). This also indicates that access to the web server can allow nomadic users to connect to Asterisk.

6.3 Research Question Two: "What are the challenges experienced by customising the features of those BBVCs to provide nomadic accessibility to Asterisk VoIP call services?"

There were challenges experienced during the implementation of the BBVC4A website and its customised web phone. These challenges are presented and discussed in section 6.3.1.

6.3.1 Compilation Challenges of the JIAXC Library

The data to be analysed is the error messages generated from the Gnome terminal during the compilation of the JIAXC application.

audio_encode.c: In function 'input_postprocess':

audio_encode.c:159: **error: dereferencing pointer to incomplete type**

audio_encode.c:167: **error: dereferencing pointer to incomplete type**

make[2]: *** [audio_encode.lo] **Error 1**

make[2]: Leaving directory \`/home/yemsolo/Imp_Progs/iaxclient-2.0.0/lib'

make[1]: *** [all-recursive] **Error 1**

make[1]: Leaving directory \`/home/yemsolo/Imp_Progs/iaxclient-2.0.0'

make: *** [all] **Error 2**

Figure 6.3: Gnome Terminal Error Messages for Previous Versions of the IAXClient Library

Before the compilation of the JIAXC library, the IAXClient had to be installed. The initial IAXClient library used for the JIAXC library could not be installed on the Ubuntu PC. Attempts to compile later versions (svn20050725-2, svn20060626, 2.0.0 and 2.0.1) of the IAXClient library were unsuccessful. Figure 6.3 shows the commonly generated error messages. Another attempt was made to install a recent version of the IAXClient library, version 2.1 beta 3, at the time of this study. Figure

6.4 shows errors generated during the compilation of the JIAXC library with the recent version of the IAXClient library.

```
/usr/include/iaxclient.h: In function ‘jint
Java_net_sourceforge_iaxclient_jni_LibJiaxc_initialize(JNIE
nv*, __jobject*, jint, jint)’:

/usr/include/iaxclient.h:285: error: too many arguments to
function ‘int iaxc_initialize(int)’

jiaxclient.cc:519: error: at this point in file

jiaxclient.cc: In function ‘void
Java_net_sourceforge_iaxclient_jni_LibJiaxc_processCalls(J
NIEnv*, __jobject*)’:

jiaxclient.cc:573: error: ‘iaxc_process_calls’ was not
declared in this scope

make[1]: *** [jiaxclient.o] Error 1

make[1]: Leaving directory
`/home/yemsolo/Imp_Progs/jiaxclient-0.0.6/jni'

make: *** [check-recursive] Error 1
```

Figure 6.4: Gnome Terminal Error Messages during JIAXC Compilation with IAXClient Library 2.1 Beta 3.

6.3.1.1 Discussion

From Figure 6.3, it shows that a more recent version of the IAXClient library needed to be installed. This prompted the need to install IAXClient library, version 2.1 beta 3, on the Ubuntu PC. Figure 6.4 shows some of the errors generated when the JIAXC library was compiled with the IAXClient library.

From the analysis of Figure 6.4, it shows that the JIAXC library needed to be upgraded so that it can be compiled with the installed IAXClient library. With the limited information on both the IAXClient library and the JIAXClient library, it requires knowledge about both libraries and how they can be compiled. Details about the findings on both libraries and the compilation of the JIAXC library are discussed in chapter 5 of this thesis.

Other challenges experienced are discussed in section 5.7.6 of Chapter 5 of this thesis.

6.4 Research Question Three: “How do the challenges affect the use of the BBVCs to access Asterisk VoIP call services?”

6.4.1 Loading of the JIAXC Application

The data to be analysed is the errors generated by Firefox Java console (Ubuntu PC) during the initial loading of the JIAXC application as a web phone. The same errors were encountered when the JIAXC demo application (initial web phone for the BBVC4A website) was being loaded on MS Windows Vista, Ubuntu and openSUSE PCs. Figure 6.5 shows some of the errors generated. No error was generated when the JIAXC demo application was being loaded on MS IE and Firefox web browsers on the MS Windows XP PC.

java.lang.ExceptionInInitializerError

...

Caused by: java.security.AccessControlException: access denied (java.util.PropertyPermission java.home read)

...

java.lang.NoClassDefFoundError: Could not initialize class net.sourceforge.iaxclient.JIAXClient

...

Figure 6.5: Firefox Java Console Error Messages during JIAXC Loading.

6.4.1.1 Discussion

From Figure 6.5, two main errors are identified. They are the java.lang.ExceptionInInitializerError and the java.lang.NoClassDefFoundError errors. From the analysis of Figure 6.5, the errors occurred because the JIAXC application is a group of applet jars trying to extend the JRE's libraries of the OS. As a result of the JIAXC application attempting to link its main library file (libjiaxc.so) to the JRE's libraries, the Java applet security manager is activated. From Figure 6.5, access to the JRE libraries is denied because the user of the Firefox web browser does not have required PC Administrative privileges to manipulate the libraries of the JRE. The resulting effect is that the JIAXC application does not get initiated.

6.4.2 Loading of the JIAXC Application for Many Users

Another observation was the effect of many users accessing the customised BBVC's web phone on the call quality. Due to the limitations of the upgraded JIAXC application (support for only Linux PC), the JIAXC demo application was used as the web phone for the BBVC4A website. This was so because the JIAXC demo application was compiled by the developer, Mikael Magnusson, to support both MS Windows and Linux OSs.

The data to be analysed is the network traffic data generated during a Call session by a user's instance of BBVC4A website's web phone when the BBVC4A website's web phone is accessed by at least two other users. Figure 6.6 – 6.7 shows the graphs of the call bandwidth and the packet losses during the Call session.

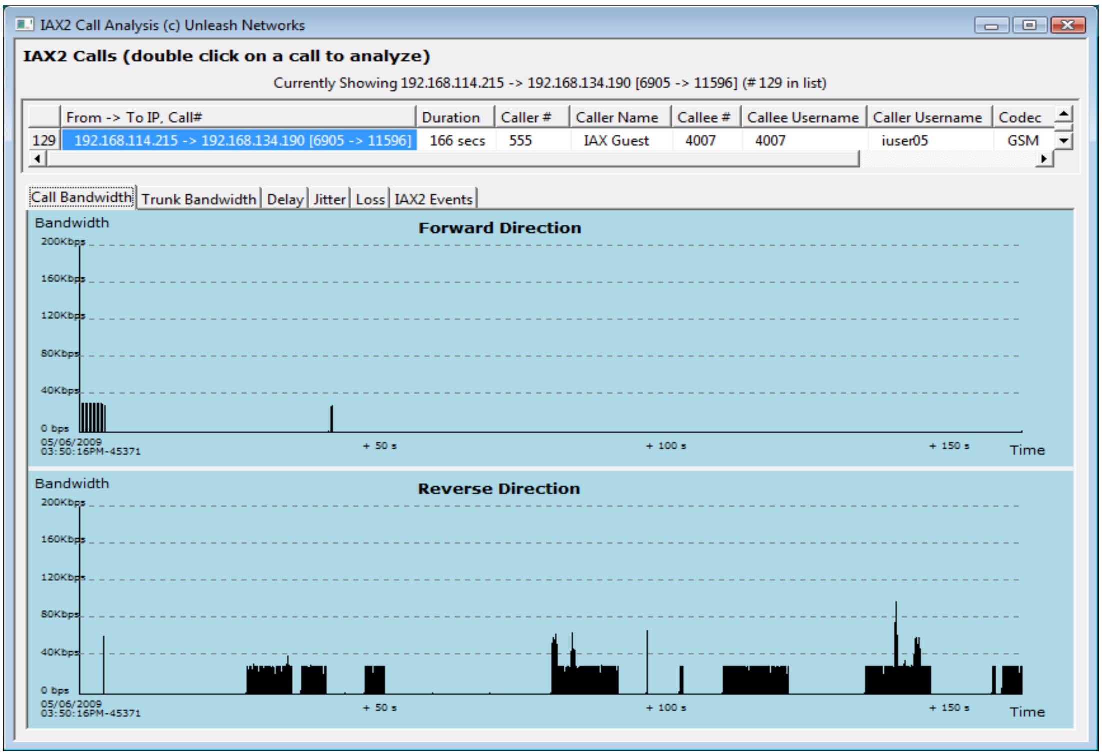

Figure 6.6: IAX2 Call Bandwidth Graph for a Call Session Between Two BBVC4A Users.

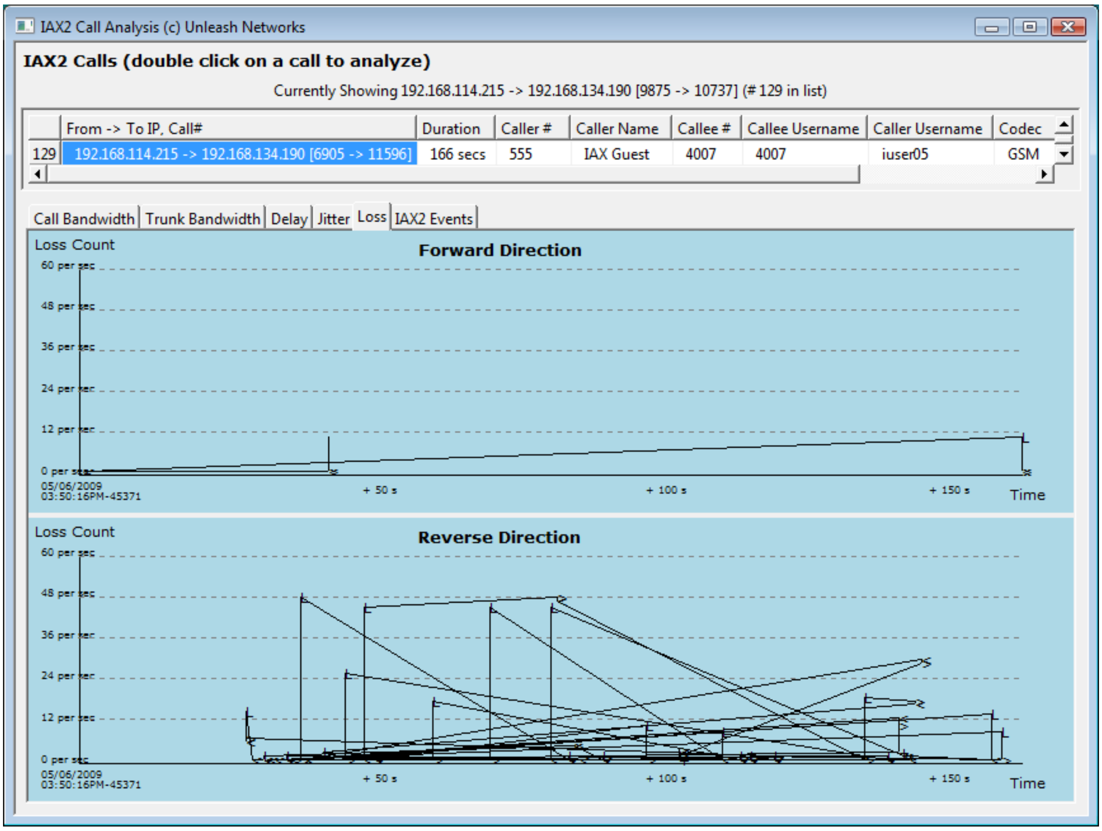

Figure 6.7: IAX2 Packet Loss Graph for a Call Session Between Two BBVC4A Users.

The “Forward direction” portions of the Figures 6.6 and 6.7 indicate the traffic flow when the caller sends data to the callee. The “Reverse direction” portions of Figures 6.6 and 6.7 indicate the traffic flow when the caller is receiving data from the callee.

6.4.2.1 Discussion

From Figure 6.6, a minimum of zero Kilobits per second (0 Kbps) and a maximum of forty Kbps (40 Kbps) of the network bandwidth were utilised when the caller sent data (voice data) to the callee. A minimum of 0 Kbps and a maximum of more than 80 Kbps of the network bandwidth were used when the caller received data from the callee. From the analysis of Figure 6.6, it shows that when the caller received data from the callee the network bandwidth used is more than the network bandwidth used when the caller was sending data to the callee. Figure 6.7 shows more details about the bandwidth gaps in the time interval for Figure 6.6.

From the Figure 6.7, within the first 50 seconds of the “Forward direction” portion of the graph, at least one (1) IAX2 packet was delayed and later resent at the beginning of the call conversation by the caller. The “+” sign indicates the number of packets resent. Towards the end of the conversation, at least eleven (11) packets sent by the caller were lost (depicted by the “L” sign in the graph). Within the first 50 seconds of the “Reverse direction” portion of the graph, a minimum of zero (0) and a maximum of 45 IAX2 packets were lost per second. About less than 48 IAX2 packets arrived late and then re-transmitted. From the analysis of Figure 6.7, it shows that more than three times the number of IAX2 packets lost when the caller speaks is lost when the caller receives voice data from the callee. It also shows that the lost packets were re-transmitted and arrived late.

From the analysis of both Figure 6.6 and 6.7, it shows that less bandwidth utility with few IAX2 packet losses occur when the caller uses the BBVC’s web phone to send voice data to the callee. It also shows that when the caller receives voice data, there is more bandwidth usage because of the retransmission of lost and late packets. The packet losses are attributed to the network devices between the caller and the callee. The main effect of these packet losses is low call quality of the conversation with voice degradation. This is also justified through some of the students’ feedback

(Question 22 of the questionnaire used for research question 4 – see Appendix A and D) which are stated below:

"voice interference was too high"

"need to work on the audio / microphones"

"Too much noise during the call session"

6.5 Research Question Four: "What are users' perceptions about the use of the BBVCs for Asterisk?"

The data from the questionnaire was analysed based on the respondents' perceptions of reliability, ease of use, ease of learning, satisfaction, consideration and recommendation of BBVCs. Detailed calculations for the data analysis of the questionnaire responses are in Appendix D of this thesis.

6.5.1 Perceived Reliability

This combines data for questions 9, 10, 11, 12 and 13 of the questionnaire (Appendix A).

Responses	Frequency	Percentage (%)
Strongly Agree	5	8
Agree	6	10
Neutral	20	33
Disagree	15	25
Strongly Disagree	14	23

Table 6.1: Respondents' Perceived Reliability of BBVC for Asterisk

6.5.1.1 Discussion

From Table 6.1, 8% of the respondents strongly agreed that the web phone of the BBVC is reliable for making and receiving calls from Asterisk server. About 10% agreed that the web phone was reliable as means of accessing Asterisk call services while 33% of the respondents could not make up their minds about the reliability of the web phone. A total of 25% of the respondents disagreed that the web phone was reliable for the making and receiving of calls from Asterisk server, and 23% strongly disagreed.

From the analysis on Table 6.1, the majority of the respondents (48%), disagree and strongly disagree inclusive, do not regard the web phone of the BBVC as reliable for making and receiving calls from an Asterisk server. This shows that though the type of codec (GSM) used for any calls to or from Asterisk server supports low bandwidth, the number of network users and network traffic congestion can also have a negative effect on the call quality and clarity. It also affects the reliable delivery of calls to and from the Asterisk server, and thus affects the quality of the calls that the users make or receive.

6.5.2 Perceived Ease of Use

This combines data for questions 14 and 16 of the questionnaire (Appendix A).

Responses	Frequency	Percentage (%)
Strongly Agree	13	54
Agree	9	38
Neutral	2	8
Disagree	0	0
Strongly Disagree	0	0

Table 6.2: Respondents' Perceived Ease of Use of BBVC for Asterisk

6.5.2.1 Discussion

From Table 6.2, 54% of the respondents strongly agreed that the web phone of the customised BBVC was simple to use. A total of 38% agreed that the web phone was simple to use, and 8% remained undecided about the ease of use of the web phone. None of the respondents disagreed nor strongly disagreed that the web phone was ease to use.

From the analysis of Table 6.2, this shows that the web phone of the customised BBVC was simple to use, given the fact that its GUI is an imitation of the telephone GUI common to other forms of VoIP clients. It also shows that as long as the web technology utilized by the web phone is present, the GUI of the web phone will always be presentable for users to use.

6.5.3 Perceived Ease of Learning

This combines data for question 15 of the questionnaire (Appendix A).

Responses	Frequency	Percentage (%)
Strongly Agree	9	75
Agree	2	17
Neutral	0	0
Disagree	1	8
Strongly Disagree	0	0

Table 6.3: Respondents' Perceived Ease of Learning of BBVC for Asterisk

6.5.3.1 Discussion

From Table 6.3, 75% of the respondents strongly agreed that the web phone of the customised BBVC was ease to learn. 17% agreed that the web phone was ease to learn while 8% disagreed. None of the respondents strongly disagreed with the ease of learning the web phone and none were undecided.

From the analysis of Table 6.3, majority of the respondents (92%), strongly agree and agree inclusive, regarded the web phone to be ease to learn. This is related to the familiarity of the telephone GUI common to other forms of VoIP clients.

6.5.4 Perceived Satisfaction with the Web Phone and its Web Page

This combines data for questions 17 and 18 of the questionnaire (Appendix A).

Responses	Frequency	Percentage (%)
Very Satisfied	9	38
Satisfied	9	38
Neutral	5	21
Dissatisfied	0	0
Very Dissatisfied	1	4

Table 6.4: Respondents' Perceived Satisfaction of BBVC for Asterisk

6.5.4.1 Discussion

From Table 6.4, 38% of the respondents had a very satisfied impression with the overall presentation of the web phone of the customised BBVC and the web page containing the web phone. 38% of the respondents were satisfied with the overall presentation of the web phone and its web page. 21% could not make up their mind about their satisfaction with the web phone and its web page while 4% were strongly dissatisfied with it.

From the analysis of Table 6.4, majority of the respondents (76%), very satisfied and satisfied inclusive, were impressed with the presentation of the web phone and its web page. This shows that the customisation of the web pages and the presence of the basic telephone GUI, of the web phone of the BBVC can improve the impression of the users of the customised BBVC.

6.5.5 Perceived Comparison to Other Forms of VoIP Clients

This combines data for questions 19 and 20 of the questionnaire (Appendix A).

Responses	Frequency	Percentage (%)
Alternative	8	33
Supplementary	16	67

Table 6.5: Respondents' Perception About the BBVC in Comparison with Other Forms of VoIP Clients

6.5.5.1 Discussion

From Table 6.5, 33% of the respondents considered the customised BBVC as an alternate to other forms of VoIP clients including soft phones and IP phones. 67% of the respondents considered the customised BBVC as a supplementary means of accessing Asterisk VoIP call services.

From the analysis of Table 6.5, this shows that the customised BBVC is mainly considered as a supplement means of accessing Asterisk VoIP call services. This also means that when other means of accessing Asterisk VoIP call services are not available, the customised BBVC can be used. The availability of the customised BBVC will also depend on the availability of the network.

6.5.6 Perceived Recommendation

This combines data for question 21 of the questionnaire (Appendix A).

Responses	Frequency	Percentage (%)
Yes	9	75
No	0	0
Not Sure	3	25

Table 6.6: Respondents' Perceived Usefulness of BBVC for Asterisk

6.5.6.1 Discussion

From Table 6.6, 75% of the respondents indicated that they would recommend the use of BBVC as a means of accessing Asterisk VoIP services to someone else. The remaining of the respondents (25%) could not make up their minds about the recommendation of the use of BBVC for accessing Asterisk VoIP call services.

From the analysis of Table 6.6, this shows that the BBVC is regarded as a useful means of accessing Asterisk VoIP call services to the extent that the users would recommend it to other people to use.

6.5.7 Overall Perceived Usability

This combines data for questions 9, 10, 11, 12, 13, 14, 15 and 16 of the questionnaire (Appendix A).

Responses	Frequency	Percentage (%)
Strongly Agree	27	28
Agree	17	18
Neutral	22	23
Disagree	16	17
Strongly Disagree	14	15

Table 6.7: Overall Respondents' Perceived Usability of BBVC for Asterisk

6.5.7.1 Discussion

From Table 6.7, 28% of the respondents strongly agreed that the customised BBVC is a usable means of accessing Asterisk VoIP call services. 18% of the respondents agreed the customised BBVC is a useable means of accessing Asterisk VoIP call services. 23% were neutral about the BBVC as a usable means of accessing VoIP call services. 17% disagreed that the customised BBVC is a usable means of accessing Asterisk VoIP call services and 15% strongly disagreed that the customised BBVC is a usable means of accessing Asterisk VoIP call services.

On the results of Table 6.7, it shows that the majority of the respondents (46%) strongly agree and agree inclusive, regarded the web phone and the web page containing it in the customised BBVC to be usable as a means of accessing Asterisk VoIP call services in the DCS-UNAM.

6.6 Summary

The analysis of data found was presented in relation to the research questions for this study. The analysis of the captured data from the customised BBVC, the challenges experienced and their effects was presented. The responses obtained from the respondents through questionnaires was analyzed and discussed. The discussions focused on the users' perceptions about the use of the BBVC for Asterisk.

CHAPTER SEVEN

CONCLUSION AND FUTURE WORK

7.1 Introduction

In this chapter, the summary of the conclusions on the data and stated research questions will be presented. Further works that can be done will also be presented.

7.2 Conclusions

The conclusions are discussed in association to the research questions.

7.2.1 Research Question One: "Can available BBVCs be customised to provide nomadic accessibility to Asterisk VoIP call services DCS-UNAM?"

Through the use of web server to host the customised BBVC (BBVC4A website), a user can access the BBVC from a PC in any part of a network. The user's login details are required for the use of the BBVC. During the loading of the web page containing the web phone (JIAXC demo application), an attempt is made to register the user's identity and location with the Asterisk server. If the registration is successful, the user can make calls to other users through the Asterisk server.

Based on the discussion on the data analysis in Chapter 6, section 6.2 of this thesis, it is conclusive that implementing a website that can dynamically allocate users' Asterisk values to the web phone will provide nomadic access to Asterisk. This will also improve access to Asterisk if the web phone (and the website) can be accessed by many users from different IP networks. The web technology supported by the BBVC's web phone needs to be present on the PC before the web phone can be used.

7.2.2 Research Question Two: "What are the challenges experienced by customising the features of those BBVCs to provide nomadic accessibility to Asterisk VoIP call services?"

As discussed in chapter 4 of this thesis, two sets of features were customised: the allocation of VoIP protocol parameters – the IAX parameters and the users' values – and the GUI features of the BBVC and its web phone (the JIAXC application). Based on the discussion in Chapter 6, section 6.3, the main challenge was the compilation of the JIAXC library. The challenge involved the upgrading and the manipulation of the JIAXC library to compile with the IAXClient library version 2.1. beta 3. This was required to enhance the VoIP features of the JIAXC application. The features added to the GUI of the JIAXC application are the enhancement of the dial pad to support both the manipulation of the entry field and the sending of DTMF signals, and more friendly status display of the application.

7.2.3 Research Question Three: "How do the challenges affect the use of the BBVCs to access VoIP call services?"

Based on the discussion in Chapter 6, section 6.4 of this thesis, the effect of customising the JIAXC application occurs when the JIAXC application is being loaded for the first time into the web browser. On Linux OS PCs, an attempt to load the JIAXC application into a web browser, by a non-administrative (or non-super) user, is halted. This is so because the JIAXC application was designed to, through its library, extend the functions of the JVM. With the JVM security restrictions, the extension library of the JIAXC application is only added to the JVM's libraries if the user of the JIAXC application is a super user or has administrative rights. The JIAXC demo application, as a result of these effects, was used as the web phone for the BBVC4A website.

Another effect of the challenge occurs when a user is using the BBVC4A's web phone to make calls, while the web phone is also being accessed by many users. This affects the quality of the calls made (or received).

Based on the discussion under section 6.4.2 of Chapter 6 of this thesis, the quality of a call is affected by the number of packets lost because of network congestion and the number of network devices between the caller and the callee. The IAX protocol, with the GSM codec, support low bandwidth availability. The selection of the codec depends on the Asterisk server's preferred audio codec.

7.2.4 Research Question Four: "What are users' perceptions about the BBVCs for Asterisk?"

The perceived ease of use, ease of learning and reliability formed the respondents' perceived usability of the customised BBVC. The perceived satisfaction, consideration and recommendation formed the respondents' perceived usefulness of the customised BBVC as means of accessing Asterisk VoIP call services.

Most (eighty percent) of the users were familiar with VoIP clients. Overall, the users perceived the JIAXC application as easy to use and learn, reliable and useful for VoIP services. The users' overall impression about BBVCs is that, though BBVCs can be a means of accessing VoIP services, they are more of a supplementary means than an alternative means, compared to other forms of VoIP clients such as hard phones and soft phones.

In conclusion, BBVCs are perceived as a suitable means of accessing Asterisk VoIP services. Even though BBVCs improve users' accesses to VoIP services through nomadic accessibility, users (especially nomadic users) mainly perceived BBVCs as supplementary means. A few factors have to be considered if BBVCs are to be deployed in an organisational LAN: the number of network devices between the furthest caller and the furthest callee in the LAN, network bandwidth available, the asterisk dial plan, web technologies supported by the BBVC and available PCs that supports the web technologies.

7.3 Future Work

The inclusion of more features and functions can enhance a BBVC. The application of more in-depth evaluation techniques can be used to evaluate the BBVC.

7.3.1 Addition of More VoIP Client Features

The main features customisable in a BBVC are the VoIP related parameters such as protocol parameters, and the application of web technologies. Web technologies, such as AJAX and Javascript, can be utilised to provide more VoIP features to BBVCs. Additional features can include users' contact (or address) list, users' and other users' presence, and ring tone applications.

Another feature that can be considered is the Video support. Though tedious to implement, many factors have to be considered for a video BBVC. These factors include application size (which automatically affects it download time), support for web browsers and OSs, the network bandwidth available, and support for web cameras.

7.3.2 Evaluation Techniques

The most common evaluation technique used for BBAs involves getting users' subjective perceptions as performed in this study while the most common evaluation technique used for VoIP applications involves predicting users' objective perception. The objective perception typically involves evaluating call quality either using a simulation of the VoIP application or using live setups with hardware such as network taps to collect VoIP data and perform mathematical analysis on them. Objective evaluations on the BBVC can be done using VoIP evaluation models such as the E-model and the PESQ model.

7.3.3 Integration of Asterisk, LAMPP and BBVC

In this study, the BBVC (JIAXC demo application) was integrated with the LAMPP web server through the BBVC4A website. This allows only the presence of website's logged-in users to be determined. If the Asterisk server is integrated with the LAMPP server, the presence of users connected to Asterisk, irrespective of the VoIP clients used, can be determined. The integration would involve Asterisk storing and updating its users' registration and connection details in MySQL database of the LAMPP server.

7.3.4 Cross-Compiling for other OS

The upgraded JIAXC application developed in this study only has support for the Linux OS. The upgraded JIAXC library can be cross-compiled for other OS including MS Windows OS. This will allow the upgraded JIAXC application to support the sound system architecture available in the OSs.

7.4 Summary

Using an accompanied API or source code such as the JIAXC library for its demo application with an authentication website, it is possible to customise a BBVC for, and connect to, an Asterisk server. Though the challenges were centred on manipulating and upgrading the JIAXC library, the security restrictions of the web technology on which the BBVC (especially its web phone) is based, and the network status can affect the usability of the BBVC.

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APPENDIX A – QUESTIONNAIRE FOR THE STUDY

Questionnaire for the Evaluation of Browser-based VoIP Client (BBVC) for Asterisk
Thank you for using the BBVC4A website. This questionnaire is meant to find out your perception on the use of Voice over Internet Protocol (VoIP) clients for VoIP services.

Please complete this series of questions **AS HONEST AS POSSIBLE**. They would help in evaluating your perception about BBVC in assessing Asterisk VoIP services at the Department of Computer Science. Your information will be treated **AS CONFIDENTIAL**.

Put a tick [✓] where appropriate.

Gender: Male ☐ Female ☐

Section A. VoIP client

VoIP services are voice call services made over computer networks such as the Internet. These services can be accessed using applications such as VoIP clients. VoIP clients include IP phones (*like Siemens Ethernet Phones and Cisco IP phones*) and soft phones (*like SJPhone and CounterPath X-Lite*). Some Instant Messengers that have voice messaging capabilities (*like Yahoo! Messenger, Skype*^{TM} *Messenger and Google GTalk*) are also considered as soft phones.

1) Have you used any type of VoIP client (as defined above) before?
Yes ☐
No ☐

2) If **Yes** to Question **1**, list **a maximum of six** of the VoIP clients that you have used?
IP phones (*Please specify*) ____
Soft phones (*Please specify*) ____
Others (*Please specify*) ____

3) From the list you gave in Question **2** above, name **one** of the VoIP clients that you mostly use? ____

4) How often do you use the VoIP client in Question **3**?
Every Day or more ☐
2 - 6 times a Week ☐
About Once a Week ☐
About Once a Month ☐

5) Why do you mostly use the VoIP client you mentioned in Question **3**? ____

Section B. Browser-based VoIP Client (BBVC) for Asterisk

BBVCs are Web-based applications that can be used to access VoIP call services. They include the BBVC4A website's Web phone that you have just used to access Asterisk VoIP services.

6) Which Web browser did you use to access the BBVC4A site?
Microsoft Internet Explorer ☐
Mozilla Firefox ☐
Others (*Please specify*) ___________________

7) On which Personal Computer (PC) Operating System (OS) is the Web browser you used in Question **4** located?

Windows OS ☐
Linux OS ☐
Others (*Please specify*) ________________________

8) On which network is the PC you used in Question **5** located?

Computer Science Network (Floor 2) ☐
Computer Science Network (Floor 1) ☐
Others (*Please specify*) _________________________

Based on your BBVC4A site experience you have just had, how would you rate the following? Please respond "*Strongly Agree*" (**SA**) or "*Agree*" (**A**) or "*Neutral*" (**N**) or "*Disagree*" (**D**) or "*Strongly Disagree*" (**SD**) where appropriate.

	Statements	**SA**	**A**	**N**	**D**	**SD**
9)	Overall, the loading of the Web page containing the Web phone was time consuming.					
10)	The Web phone was reliable for making calls.					
11)	The Web phone was reliable for receiving calls.					
12)	The quality of the call audio of the phone during a call session was satisfactory.					
13)	The clarity of the call audio of the phone during a call session was satisfactory.					
14)	The Web phone was simple to use.					
15)	Learning how to operate the Web phone was easy.					
16)	The buttons of the Dial Pad on the Web phone were useful for entering requested information during a call session.					

Section C. General Impressions and Comments

Based on your BBVC4A site experience you have just had, how satisfied are you with the following? Please respond "*Very Satisfied*" (**VS**) or "*Satisfied*" (**S**) or "*Neutral*" (**N**) or "*Dissatisfied*" (**D**) or "*Very Dissatisfied*" (**VD**) where appropriate.

	Questions	**VS**	**S**	**N**	**D**	**VD**
17)	Overall impression about the Web phone.					
18)	Overall impression about the Web page containing the Web phone.					

In relation to means of accessing Asterisk VoIP services, how would you consider BBVC to the following? Please respond "*Alternative*" or "*Supplementary*" where appropriate.

	Questions	**Alternative**	**Supplementary**
19)	Soft phones		
20)	IP phones		

21) Would you recommend the use of BBVC for accessing Asterisk VoIP services to someone else?

Yes ☐
No ☐
Not Sure ☐

22) If there is anything else that you consider important about your BBVC4A experience that has not been addressed in this questionnaire, please specify:

____
____
____
____

THANK YOU FOR YOUR PATIENCE AND CO-OPERATION.

APPENDIX B – BBVC SURVEY MATRIX

SN	Product / Features	Busta	Conaito's VoIP EVO (Enterprise Edition v2.1)	Corraleta Connect	GizmoCall*	Hello 2 Web	iaxClientOcx (v2.0.0.72)	JIAXC (v0.0.6)	JSAP!*	Ribbit (Virtual Phone)*	TringPhone*	VaxVoIP SIP SDK (v3.0)	Web Voice Chat (v1.2.3)
	Phone Features												
1	Dial pad to type in numbers to the entry field	Y	N	N	Y	N	Y	N	N	Y	Y	Y	N
2	Dial pad to send DTMF signals	Y	N	N	Y	N	Y	Y	N	Y	Y	Y	N
3	Dial pad to type in numbers and send DTMF signals	Y	N	N	Y	N	Y	N	N	Y	Y	Y	N
4	Contact List (Phone book)	N	N	N	Y	N	N	N	Y	Y	N	N	N
5	Dial (Call) user	Y	Y	Y	Y	Y	Y	Y	Y	Y	Y	Y	Y
6	Speed dial	N	N	N	N	N	N	N	N	N	N	Y	N
7	Last number Redial	N	N	N	N	N	Y	N	N	N	Y	N	N
8	Receive (Answer) calls from other users	Y	Y	N	Y	Y	Y	Y	Y	Y	Y	Y	Y

9	Call hold (and unhold)	N	N	N	N	N	N	N	Y**	N	N	Y	N
10	Call transfer	N	N	N	N	N	N	N	N	N	N	Y	N
11	Call Ignore (or Reject)	Y	N	N	Y	Y	Y	Y	Y**	Y	Y	Y	Y
12	Hang up (End) call	Y	Y	Y	Y	Y	Y	Y	Y	Y	Y	Y	N
13	Multi-session call lines	N*	N	N	N	N	N	Y**	Y**	N**	N**	Y	N
14	Call History (or logs)	Y	N	N	N	N	N	N	N	Y	N	Y	Y
15	Message waiting indicator	Y	Y	N	Y	Y	Y	Y	Y	Y	Y	Y	Y
16	Log display for operation monitoring	N*	Y	N	N	N	N	N	Y	N	N	Y	N
17	User's Presence	Y	N	N	Y	N	Y	N	Y	Y	Y	Y	Y
18	Other Users' presence	N*	Y	Y	N	N	N	N	N	N	N**	N	Y
19	Input / Output device selection	N*	N	N	N	N	Y	Y	!	!	!	N	N
20	Input / Output level indicator	N*	N	N	Y	Y	Y	N	N	Y	Y	Y	Y
21	Input / Output volume control	N	N	N	Y	Y	Y	N	N	Y	Y	Y	N
22	Automatic selection of best audio codec	N*	N	N	Y	Y	Y	Y*	Y	Y	Y	Y	Y
23	Manual selection of audio codec	N*	N	N	N	N	Y	N	N	N	N	Y	N
24	Clear entry field (button)	N*	N	N	Y	N	Y	N	N	Y	N	Y	N
25	Clear entry field (keyboard or mouse)	Y*	Y	N	Y	Y	N	Y	Y	Y	Y	Y	Y

	VoIP Features												
26	Audio Codec supported	G.711, G.729	GSM, SPEEX	**	**	**	GSM, G.729, ILBC, SPEEX, G.711	GSM, G.711, SPEEX, PORT-AUDIO	**	**	GSM, G.711, G.729, G.723, ILBC, SPEEX.	GSM, G.711, ILBC	WAV, MP3
27	VoIP Protocol supported	SIP	**	IAX	SIP,	SIP	IAX	IAX	SIP	SIP, MGCP, Jingle	SIP	SIP	!
28	Support for Asterisk VoIP service	N	N	Y	Y	N	Y	Y	N	N	N	Y	N
	Web Features												
29	Web technology (ActiveX-A, Adobe Flash-F, Java-J)	A	A	J	F	J	A	J	J	F	F	A	J
30	Operating System supported (Windows-W, Linux-L, Mac-M, Solaris-S, Others-O)	W	W	W, L, M	W, L, M	S	W	W, L	W, L	W	W, L	W	W, M
31	Web browsers supported (Internet Explorer-IE, Mozilla Firefox-MF, Netscape Navigator-NN, Opera-OP, Safari-S, Others-O)	IE, MF	IE, OP	IE, MF, OP	IE, MF, OP,	NN	IE	IE, MF, OP	IE, MF, OP	IE, MF, OP	IE, MF, OP	IE	IE, MF, OP, NN, S

32	Login (Logout)	Y	Y	N	Y	Y	Y	N	Y	Y	Y	Y	Y
	Others												
33	Hosting (Locally-L, Externally-E)	E	L	E	E	E	L	L	L	E	E	L	L
34	Open Source-OS or Proprietary-P	P	P	P	P	P	P	OS	OS	P	P	P	P
35	Trial version-TV or Demo application-DA	NA	TV, DA	DA, TV	NA	DA	DA	DA	NA	30-day TV	NA	30-day TV, DA	TV
36	Licensing cost required for full voice services	NA	Y	£1100 (N$ 13200) per server	N	NA	NA**	N	NA	US$25 (N$ 200) per month	NA	US$ 1500 - US$ 12000 (N$ 12000 – N$ 96000) per 20 user voice sessions.	US$ 400 - US$ 900) (N$ 3200 - N$ 7200) for private rooms
37	API provision for developers (No-N, Free-F or not free -NF)	N	F	NF	F	F	F	F	F	F	F	F	NF
38	Extra package installation required	N	Y - VoIP server	N	N	N	N	N	Y - JMF	Y - AMPHIBIAN	N	N	N

39	Accessible from any IP network after deployment (without licensing / purchase)	NA	N	NA	NA	Y	Y	Y	Y	NA	Y	N	Y
40	Accessible from any IP network after deployment (with licensing / purchase)	NA	Y	Y	NA	NA	NA	Y!	NA	NA	NA!	Y	Y

LEGEND:

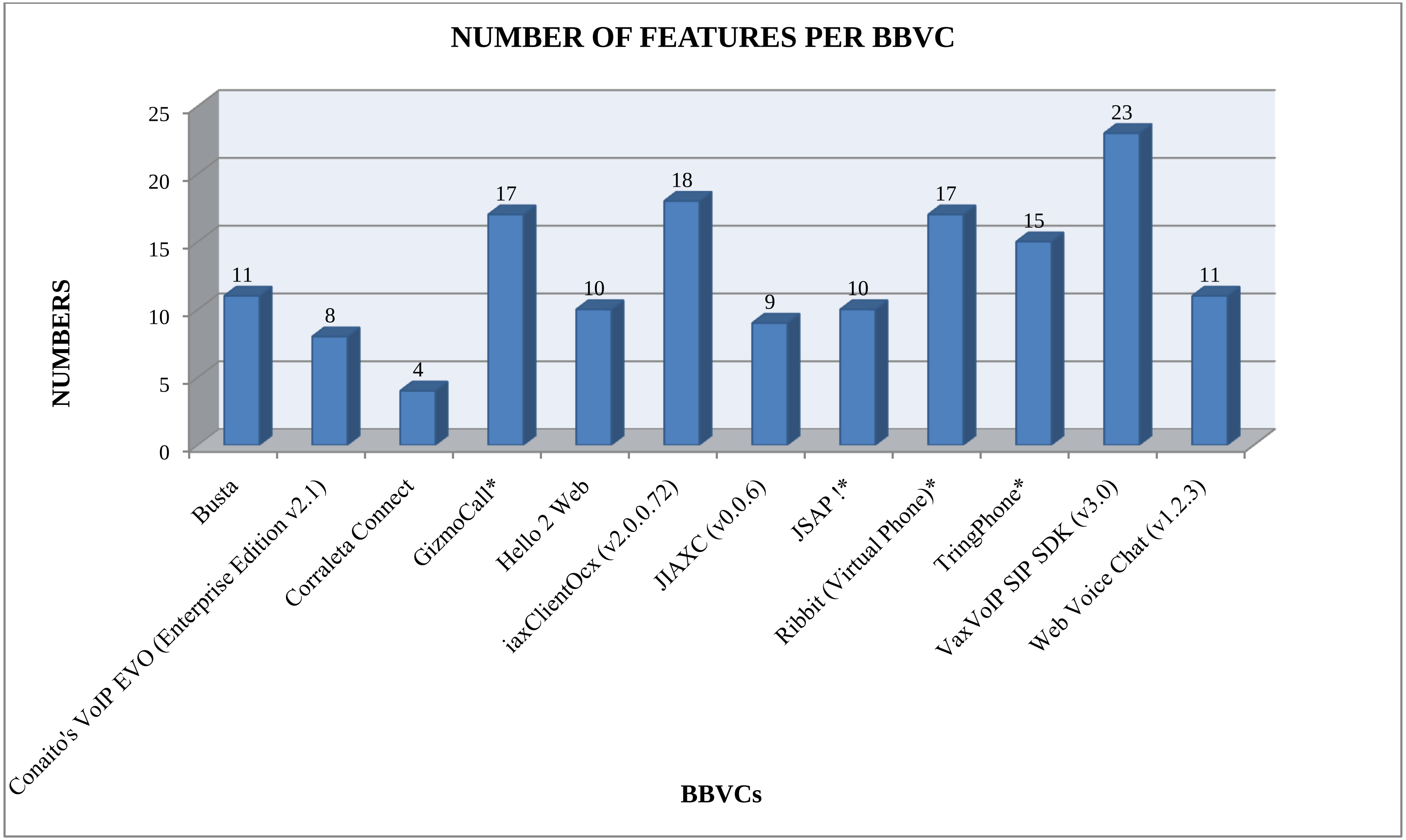
* → BASED ON LITERATURE & OTHER APPLICATION FORM

** → UNKNOWN

! → UNKNOWN BECAUSE IT DIDN'T WORK

NA → NOT APPLICABLE.

OVERALL SUMMARY (OUT OF 12)	
Supported for Asterisk:	5
Internal Hosting:	6
Open Source:	2


NUMBER OF FEATURES PER BBVC
NUMBERS
0
5
10
15
20
25
11
8
4
17
10
18
9
10
17
15
23
11
Busta
Conaito's VoIP EVO (Enterprise Edition v2.1)
Corraleta Connect
GizmoCall*
Hello 2 Web
iaxClientOcx (v2.0.0.72)
JIAXC (v0.0.6)
JSAP !*
Ribbit (Virtual Phone)*
TringPhone*
VaxVoIP SIP SDK (v3.0)
Web Voice Chat (v1.2.3)
BBVCs

APPENDIX C – REQUIREMENT SPECIFICATION FOR BBVC4A

INTRODUCTION

Browser-based VoIP Client for Asterisk (BBVC4A) is a web application which allows users to make calls to, and receive calls from, Asterisk users. BBVC4A will allow nomadic users to access Asterisk services.

Purpose of the System

The purpose of BBVC4A is to improve users' accessibility to Asterisk VoIP call services. This will help reduce the effort required to acquire a VoIP client to access Asterisk services.

A simple scenario of the implementation of the BBVC4A within a University department will be used. The BBVC4A will give any person with a valid student number from the University of Namibia (UNAM), a valid Asterisk account and an Intranet (or Internet) connection the chance to register with the website to have access Asterisk VoIP call services. It will have at least one login type: Member. Users that own a valid Asterisk account and a UNAM student number, or are registered user through the website, have the member login type (unless granted a different status by those who maintain the system).

Other login types depend on the maintainers of the system. They should, if required, have the same functionalities when using the website as member users do, and also have a set of unique options that will allow them to do other actions required of them.

USAGE SCENARIO

User profiles

There are two user profiles: New Users and Existing users. New Users are those who have Asterisk accounts but are not registered users of the BBVC4A product. Existing users are Asterisk users who already have a BBVC4A account. That is registered users of the BBVC4A product.

Use-cases

Figure C.1 below shows the use case model for the users of BBVC4A.

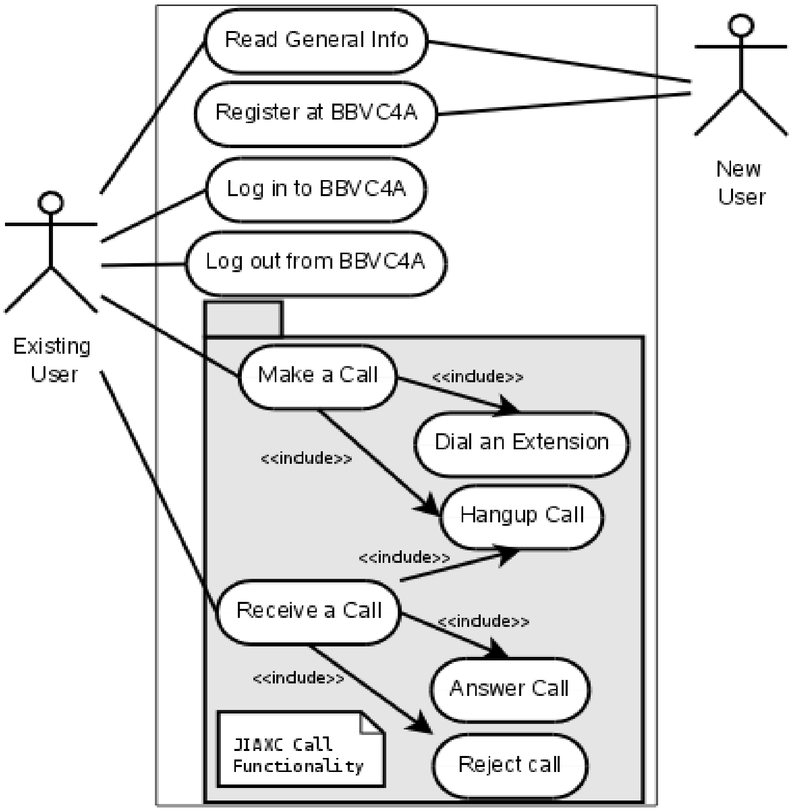


Figure C.1: Use Case Diagram for all Users of BBVC4A Product

More details about each use-case in Figure C.1 are presented in the following tables.

Use Case Name	Read General Info
Participating Actors	New User, Existing User.
Goal in Context	To view and read information about BBVC4A and other related topics.
Preconditions	Existing User must be logged in.
Trigger	Both users decide to browse through the site.
Scenario	1.) Users click on the "Related Links" button. 2.) BBVC4A displays all major links and information about BBVC for Asterisk, VoIP and other related topics. 3.) Users click the link for a particular topic. 4.) BBVC4A redirects users to websites that focus on the topic.
Exceptions	1.) New User selects "Make A Call" button. See use case "Log in to BBVC4A".
Priority	High priority, to be implemented before basic functions
When available	First increment.
Frequency of Use	Frequent.

Use Case Name	Register at BBVC4A
Participating Actors	New User.
Goal in Context	To create an account at BBVC4A.
Preconditions	User must have a UNAM student number, an Asterisk username, a password and the Asterisk IP address.
Trigger	User decides to register at BBVC4A.
Scenario	1.) User clicks on "Make a Call" button. 2.) BBVC4A takes the user to "Login" page. 3.) User clicks on the "Register" link. 4.) BBVC4A takes the User to the "Register" page. 5.) BBVC4A displays the requirements for

	registration and a registration form for user to fill. 6.) User fills in the registration form. 7.) User clicks the “Submit” button. 8.) BBVC4A checks that the User has not existed before and that the information syntax is correct. (Alphanumeric uses, numbers only, “@” usage, etc.). 9.) BBVC4A displays the status of the registration.
Exceptions	1.) User has registered before. See use-case: “Validate User Registration”.
Priority	High priority, to be implemented before basic functions
When available	Fist increment.
Frequency of Use	Frequent.

Use Case Name	**Log in to BBVC4A**
Participating Actors	New User, Existing User.
Goal in Context	To view BBVC4A and make use of the JIAXC application (web phone).
Preconditions	User must have appropriate username and password.
Trigger	1.) The User decides to view and use BBVC4A’s web phone OR 2.) New User clicks the “Make A Call” button.
Scenario	1.) The User logs onto BBVC4A either through redirection or clicking the “Login” button or link. 2.) The User enters his or her username (aka student number). 3.) The User enters his or her password (at least 6 alphanumerical characters in length). 4.) BBVC4A displays a Welcome page to the user. 5.) BBVC4A displays major function buttons.

Exceptions	1.) Username or Password is incorrect or not recognized. See use-case: “Validate Username and Password”. 2.) User selects “Make a Call” button or link. See use-case: “Make a Call”. 3.) User selects “Related Links” or “About Us” or “Contact Us” button or link. See use-case: “Read General Info”. 4.) User selects “Logout” button. See use-case: “Log out from BBVC4A”. 5.) User receives a call. See use-case: “Receive a Call”. 6.) User does not have login details. See use-case: “Register at BBVC4A”.
Priority	High Priority, to be implemented with basic functions.
When available	First increment.
Frequency of Use	Very frequent.

Use Case Name	**Log out from BBVC4A**
Participating Actors	Existing User.
Goal in Context	To exit from BBVC4A.
Preconditions	User must be logged in to BBVC4A.
Trigger	User decides not to use BBVC4A and exit from it.
Scenario	1.) User clicks the “Logout” button. 2.) BBVC4A logs the user out. 3.) BBVC4A displays the status of the event.
Exceptions	1.) User clicks the “Login” button. See use-case “Log in to BBVC4A”.
Priority	High priority, to be implemented with basic functions.
When available	First increment.

Frequency of Use	Very frequent.

Use Case Name	**Make a Call**
Participating Actors	Existing User.
Goal in Context	To make a call to an extension number.
Preconditions	1.) User must be logged in to BBVC4A. 2.) User must have a working audio input device, such as a microphone, and a working audio output device, such as a speaker, both plugged to the PC. 3.) User's web browser must be Java-enabled.
Trigger	User decides to make a call to an extension.
Scenario	1.) User clicks on the "Make A Call" button. 2.) BBVC4A takes the User to the "Make A Call" page (where the JIAXC application (web phone) is located). 3.) User enters the extension number to be dialled. See use-case: "Dial an Extension". 4.) BBVC4A displays the status of the call connection through the web phone.
Exceptions	1.) User clicks the "Call" button. See use-case: "Dial an Extension". 2.) Empty extension number is dialled. See use-case: "validate extension number". 3.) User cancels call. See use-case: "Hangup call". 4.) New User clicks "Make A Call" button. See use-case: "Log in to BBVC4A".
Priority	High priority, to be implemented after basic functions
When available	First increment.
Frequency of Use	Very frequent.

Use Case Name	Dial an Extension
Participating Actors	Existing User.
Goal in Context	To establish a call connection to an extension number.
Preconditions	1.) User must be logged in to BBVC4A. 2.) User must have a working audio input device, such as a microphone, and a working audio output device, such as a speaker, both plugged to the PC. 3.) User's web browser must be Java-enabled. 4.) User must be on the "Make A Call" page.
Trigger	User decides to call an extension number.
Scenario	1.) User enters the extension number to call. 2.) User clicks the "Call" button. 3.) The JIAXC application (web phone) displays the status of the call connection. 4.) User responds to the owner of the extension number called. 5.) User ends the call. See use-case: "Hangup call". 6.) The web phone displays the status of the call connection.
Exceptions	1.) Owner of the Extension number called ignores the call. The web phone displays the call connection status. 2.) The callee hangs up first. See use-case: "Hangup Call". 3.) There is an incoming call for the User. See use-case: "Receive a Call".
Priority	High priority, to be implemented as basic function.
When available	First increment.
Frequency of Use	Very frequent.

Use Case Name	Hangup Call
Participating Actors	Existing User.
Goal in Context	To end an existing call connection to an Extension number.
Preconditions	A call connection to an extension number should be in progress.
Trigger	User wants to stop an existing call connection to an extension number.
Scenario	1.) User dials or answers a call from an extension number. See use-cases: “Dial an extension” and “Answer Call”. 2.) The JIAXC application (web phone) displays the status of the call connection. 3.) User clicks the “Hangup” button. 4.) The web phone terminates the call connection. 5.) The web phone displays the status of the call connection.
Exceptions	1.) The callee hangs up first. See use-case: “Hangup Call”. 2.) There is an incoming call for the User. See use-case: “Receive a call”.
Priority	High priority, to be implemented as basic function.
When available	First increment.
Frequency of Use	Very frequent.

Use Case Name	Receive a Call
Participating Actors	Existing User
Goal in Context	To receive incoming call.
Preconditions	1.) User must be logged in to BBVC4A.

	2.) User must have a working audio input device, such as a microphone, and a working audio output device, such as a speaker, both plugged to the PC. 3.) User's web browser must be Java-enabled. 4.) User must be on the "Make A Call" page.
Trigger	User is being called by another existing user.
Scenario	1.) The JIAXC application (web phone) starts ringing. 2.) The web phone shows status of incoming call. 3.) The web phone displays "Answer" and "Reject" button. 4.) User clicks either "Answer" or "Reject" button. See use-cases: "Answer call" and "Reject call".
Exceptions	1.) User not available or ignores incoming call. The web phone displays the status of the call connection.
Priority	High priority, to be implemented as basic function.
When available	First increment.
Frequency of Use	Frequent.

Use Case Name	**Answer Call**
Participating Actors	Existing User.
Goal in Context	To answer incoming call.
Preconditions	1.) User must be logged in to BBVC4A. 2.) User has a working audio input device, such as a microphone, and a working audio output device, such as a speaker, both plugged to the PC. 3.) User's web browser should be Java-enabled. 4.) User must be on the "Make A Call" page. 5.) The JIAXC application (web phone) displays

	“Answer” and “Reject” buttons, and status of incoming call. 6.) The web phone must be ringing.
Trigger	User clicks “Answer” button to answer incoming call.
Scenario	1.) The web phone rings, showing the status of the incoming call and also shows “Answer” and “Reject” button. 2.) User clicks “Answer” button. 3.) The web phone establishes call connection. 4.) User talks to the caller. 5.) User ends the call. See use-case: “Hangup call”. 6.) The web phone displays status of the call connection.
Exceptions	1.) Caller ends call connection. See use-case: “Hangup call”.
Priority	High priority, to be implemented as basic function.
When available	First increment.
Frequency of Use	Frequent.

Use Case Name	**Reject Call**
Participating Actors	Existing User.
Goal in Context	To reject incoming call.
Preconditions	1.) User must be logged in to BBVC4A. 2.) User must have a working audio input device, such as a microphone, and a working audio output device, such as a speaker, both plugged to the PC. 3.) User’s web browser must be Java-enabled. 4.) User must be on the “Make A Call” page. 5.) The JIAXC application (web phone) displays “Answer” and “Reject” buttons, and status of

	incoming call. 6.) The web phone must be ringing.
Trigger	User clicks “Reject” button to reject incoming call.
Scenario	1.) The web phone rings, showing the status of the incoming call and also shows “Answer” and “Reject” button. 2.) User clicks “Reject” button. 3.) The web phone ends call connection. 4.) The web phone displays status of the call connection.
Exceptions	1.) Caller ends call connection. See use-case: “Hangup call”.
Priority	High priority, to be implemented as basic function.
When available	First increment.
Frequency of Use	Frequent.

Use Case Name	**Validate username and password**
Participating Actors	Existing User.
Goal in Context	To validate username and password for login.
Preconditions	User must enter username and password, and the “Login” button clicked.
Trigger	User clicks “Login” button of the Login form.
Scenario	1.) BBVC4A gets username and password. 2.) BBVC4A compares the username and the password with those in its database. 3.) If a match exists, BBVC4A takes user to the user’s "Welcome" page.
Exceptions	1.) There is no match in the database. BBVC4A takes user to the “Login failed” page. 2.) BBVC4A displays the errors.

Priority	High priority. To be implemented with basic functions.
When available	First increment.
Frequency of Use	Very frequent.

Use Case Name	**Validate extension number**
Participating Actors	Existing User.
Goal in Context	To validate that an extension number is entered.
Preconditions	1.) User must be logged in to BBVC4A. 2.) User must have a working audio input device, such as a microphone, and a working audio output device, such as a speaker, both plugged to the PC. 3.) User's web browser must be Java-enabled. 4.) User must be on the "Make A Call" page.
Trigger	User clicks "Call" button of the JIAXC application (web phone).
Scenario	1.) The web phone gets extension number entered. 2.) The web phone checks that the entry is not empty or full of white space. 3.) The web phone displays the status of the call connection.
Exceptions	1.) If entry is empty or full of white space, the web phone displays "Enter an extension number" message.
Priority	High priority, to be implemented with basic functions.
When available	First increment.
Frequency of Use	Very frequent.

Use Case Name	**Validate User Registration**
Participating Actors	New User.
Goal in Context	To validate that correct registration details are entered.
Preconditions	User already entered registration details.
Trigger	User clicks "Submit" button of the registration form.
Scenario	1.) BBVC4A gets user's registration. 2.) BBVC4A checks that the username and password given do not already exist in the database. 3.) BBVC4A displays registration status.
Exceptions	1.) If the username and password already exists, BBVC4A stops registration and displays the errors.
Priority	High priority, to be implemented with basic functions.
When available	First increment.
Frequency of Use	Very frequent.

DATA MODEL & DESCRIPTION

The BBVC4A system will utilize one data object which is the main source of the information needed to function properly. The data will come in the form of usernames, passwords and Asterisk's IAX parameters.

Data Object

The data object is the *User*. The user will contribute his or her username and password through the use of the website. The user will supply a unique username for login purposes.

The user will, during BBVC4A registration, provide Asterisk's IAX parameters needed to connect to Asterisk server. These parameters include the user's Asterisk username, the user's Asterisk password, the Asterisk IP address, the user's extension number, the preferred audio codec to use and the need for Asterisk registration. The parameters will be used by the JIAXC application (web phone) to connect to Asterisk.

Relationship

Figure C.2 shows the relationship for the data object above. The user will communicate with the web phone through the functions described in Functional Requirement section.

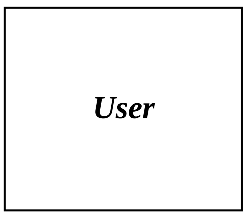


Figure C.2: Entity Relationship Diagram of the Data Object in BBVC4A

Data Dictionary

The data for BBVC4A product will be kept in one data store. The database will contain user login information, Asterisk information, and other registration details.

FUNCTIONAL REQUIREMENT

The functions of the BBVC4A system are separated into two categories: web phone functions and website functions. The website functions are available to all users with Intranet (Internet) connection and Java-enabled web browser. The web phone functions are limited only to Existing BBVC4A users. The following section contains information about the user interfaces of the functions.

Web Phone Functions (Functions of the JIAXC Application – the web phone)

Asterisk Registration

The function allows users to connect to Asterisk. This is achieved when the user is taken to the "Make A Call" page. During the loading of the page, the web phone is being loaded. During the loading stage, the user's Asterisk registration details are passed as values to the web phone's parameters from the cookies.

Call Extension Number

This function allows the user to create a call connection to another user through Asterisk. This will only be available as long as no call connection exists.

Hangup Call

This function allows the user to end (or teardown) a call connection to another user. This will only be available as long as a call connection (Call Setup) exists.

Answer Call

This function allows the user to answer a call connection from another user. This will only be available as long as the call connection exists and it's initiated by the remote user.

Reject Call

This function allows the user to reject a call connection from another user. This will only be available as long as the call connection exists and it's initiated by the remote user.

Edit Audio Setting

This function allows the user to select the audio (input and output) devices to use for a call connection.

Edit Extension Number Entry

This function will allow the user to edit entries made before a call connection is made.

Website Functions

Login

The function will log a user in to their BBVC4A website account when provided with proper credentials. Upon logging in, the user will be taken to the "Welcome" page and a choice to make calls.

Logout

This function will log the user out of the website and end their session. Once logged out, the user will be taken to the "Logged Out" page and given choices as to what next to do.

Make a call

This function will allow users to connect to Asterisk using the web phone and their Asterisk account details.

View Related links

This function will allow the user to view links related to VoIP, BBVC4A and Asterisk.

Register Account

This function will allow a user to create a BBVC4A website account.

NON-FUNCTIONAL REQUIREMENTS

Usability

The website is mainly designed for testing purposes. The intended users are those who have used web applications before. The website will be simple enough to use so that no training is necessary.

Performance

The website is aimed to support at least 20 calls per hour. The initial application will not be running in a true production environment, and will have constraints like bandwidth and system hardware serving as a resource constraint. The software will be written such that it is scalable and can be moved out to better hardware, with more bandwidth after the first version of the software is functional.

Implementation

Java Programming Language, IAXClient and the GNU Build System will be used to build the JIAXC application. Mingw32 will also be used as the cross-compiler to create the JIAXC application jars for Microsoft Windows.

AJAX, HTML and PHP will be used as the primary languages for the website. LAMPP, a web content management system, will be used to host the site. LAMPP includes Apache web server for deploying the website and MySQL database system for storing important data.

Interface

The user will interface with the system via the website interface. The interface will consist of few, but simple features such as buttons and textboxes to use the BBVC4A product.

Security

The system will provide login username and passwords for all users so that only registered users access the BBVC4A product. This can be granted by the maintainers of the system.

BEHAVIOURAL MODEL & DESCRIPTION

Website Events

Incorrect username or password

Upon entering a username or password that is incorrect, that is, it does not match with those stored in the database, the user will return to the “Login failed” page. This page will inform the user of the exact problem and prompt the user to enter the information again on the “Login page”

Correct username and password

When a provided username and password matches the same credentials in the database, the user will be granted access to the system. Features available in the user’s interface will be limited by their login type.

User (Member) Access level

If the login type is user, the user will be given access to solely user functions. These functions can also be determined by the maintainers of the BBVC4A system.

Making a call

Once the user clicks the “Make A Call” button, the “Make A Call” page is loaded with the JIAXC application (web phone). Asterisk values needed for the application are taken from the session’s cookies.

User Registration event

Once the user has submitted registration information, the system will take the information the user supplies and update the user data store. The user will then be logged in to the application.

Web Phone events

Asterisk Registration event

The event starts when the “Make A Call” page is loading. When the user decides to leave the page, the JIAXC application (web phone) unregisters the user from Asterisk.

Making call

This event starts when the user clicks the “Call” button of the web phone after he or she has entered an extension number to call. The web phone will create a call connection by sending and acknowledging IAX messages to and from Asterisk.

Website States

User Registration

The user will be able to enter a registration state where they can enter the appropriate data to register for a BBVC4A account.

Login

This is a state the user enters when the “Login” link or button is clicked.

User Interface

After logging in, the user is provided with a Welcome interface listing the various options that are accessible. This may also depend on their access level.

Make a call

This is the state a user enters when he or she clicks the “Make A Call” Page. In this state the user can call an extension number or receive a call from other Asterisk users.

View Related Links

The user will be in this state where they can view more information about the product, VoIP and other BBVCs.

Web Phone States

Asterisk Registration

Since the user has provided his or her Asterisk account details, this state is reached when the “Make A Call” page is loaded.

Make a Call

This state will present the user with phone features to choose from in order to make calls. The state is entered when the user has decided to make a call.

Receive Call

The user will be in this state when a call is made to the user. In this state, the user will be presented with appropriate phone features to choose from.

Answer Call

The user enters this state if he or she chooses to acknowledge an incoming call. This also involves the user clicking the “Answer” button.

Reject Call

This state is entered when the user chooses not to acknowledge an incoming call. This involves the user clicking the “Reject” button.

Call Setup

The user enters this state when he or she is initiating a call connection. This state is the Call Setup state of an IAX connection.

Media Flow

The user enters this state after he or she has chosen to answer a call or an initiated call has been accepted. This state is the Media or Audio Flow of the IAX connection.

Call Teardown

This state is entered after the user ends a call connection as the caller or rejects an incoming call as the callee. This is the Call Teardown state of the IAX connection.

Hangup Call

The user enters this state when he or she decides to hang up an existing call connection.

State Transitions

Figures C.3 and C.4 depict the state diagrams for the BBVC4A website and the web phone application respectively.

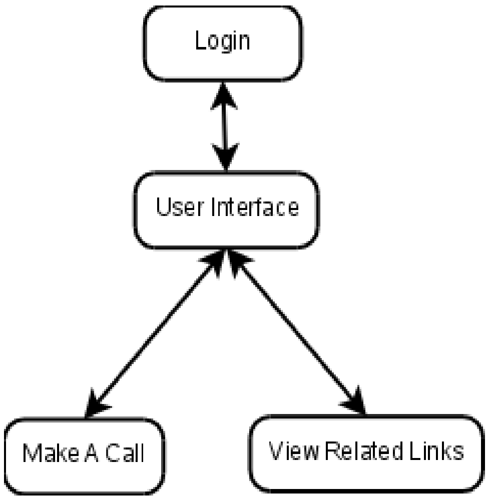


Figure C.3: State Diagram for BBVC4A Website

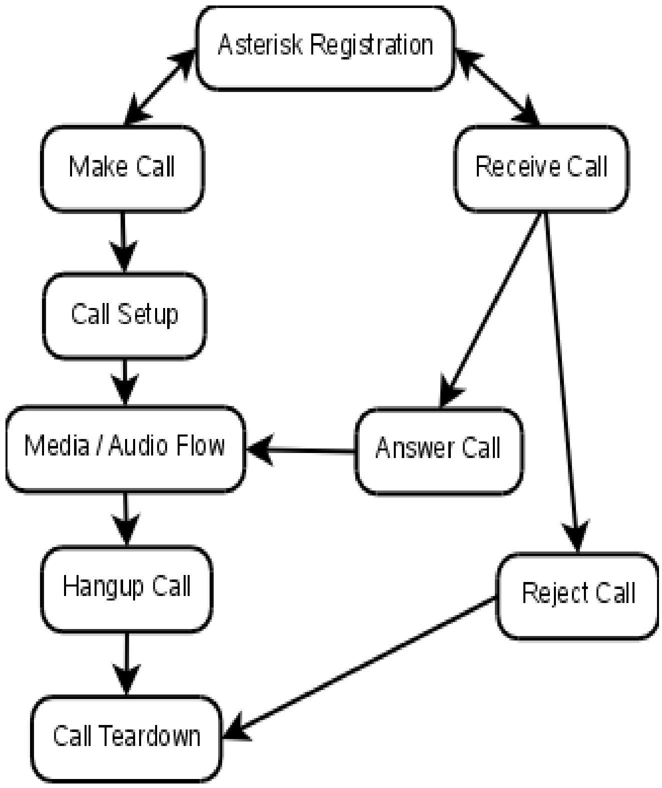


Figure C.4: State Diagram for the Web Phone.

Validation Criteria

Classes of Tests and Responses

Incorrect login

If the user attempts to log in with the incorrect information, the user will be taken to a page displaying the errors. The user will also be prompted to attempt to log in again.

Unavailable Username

During registration, if the username selected by the user is already taken, then he or she will be given an error message and prompted to choose a different username.

Blank Registration or Login

If the user submits a blank (or data full of white spaces) registration or login data, he or she will be warned that the fields are blank. The user will also be prompted to enter correct data.

Performance Bounds

Due to the fact that the product is a web application, loading speeds and accessibility will be determined by the network stability in the user's area and also by the Asterisk's dial plan. The major performance factor of the system is speed, as a user would not want to wait extensive amounts of time to call another user. However, the performance is up to the network status that the user has with their Intranet provider, not the system itself.

APPENDIX D – QUESTIONNAIRE CODING AND DATA ANALYSIS CALCULATION

1 QUESTION CODING

Gender:

1=Male

2=Female

Q1:

1=Yes

2=No

Q2:

0=No need to answer

1=Google Gtalk

2=Skype Messenger

3=All of the Above

4=Others

Q3:

0=No need to answer

1=Google Gtalk

2=Skype Messenger

Q4:

0=No need to answer

1=Everyday or More

2=2-6 times a Week

3=About Once a Week

4=About Once a Month

Q5:

0=No Need to answer

1=Communication (via messaging/calling/chatting/exchanging files)

2=Cheaper calls

3=User Friendly

Q6:

1=Microsoft IE

2=Mozilla Firefox

3=Others

Q7:

1=Windows OS

2=Linux OS

3=Others

Q8:

1=CS network (Floor 2)

2=CS network (Floor 1)

3=Others

Q9 – Q16:

1=Strongly Agree

2=Agree

3=Neutral

4=Disagree

5=Strongly Disagree

Q17 – Q18:

1=Very Satisfied

2=Satisfied

3=Neutral

4=Dissatisfied

5=Very Dissatisfied

Q19 – Q20:

1=Alternative

2=Supplementary

Q21:

1=Yes

2=No

3=Not Sure

Q22a – Q22b:

1=No comment because all where specified

2=Is the BBVC4 easy to used by the users

3=User identification not showing.

4=Is very impressive and easy to use.

5= voice interference was too high.

6= need to work on the audio / microphones

7= product will be most useful

8= recommend to user wishing to communicate by voice.

9=interface should be more attractive.

10=Phonebook Mechanism

11=Availability of the service

12=Too much noise during the call session

0=No Need to answer

2 QUESTIONNAIRE CODING

R-NO	Gender	Q1	Q2	Q3	Q4	Q5	Q6	Q7
1	2	1	1	1	4	1	2	2
2	1	1	1	0	3	1	1	1
3	2	1	3	2	3	2	2	1
4	1	1	1	1	1	1	1	1
5	2	1	1	1	1	1	2	2
6	1	1	1	1	1	1	2	2
7	1	1	2	2	4	1	2	2
8	1	1	3	1	4	1	1	1
9	1	1	1	0	1	3	1	1
10	1	2	0	0	0	0	2	2
11	1	1	1	1	3	1	2	1
12	2	1	1	1	4	1	2	2

R-NO	Q8	Q9	Q10	Q11	Q12	Q13	Q14	Q15
1	1	5	5	4	4	4	1	1
2	1	4	3	2	3	3	1	4
3	1	4	2	2	2	3	1	1
4	1	5	1	2	3	2	1	1
5	1	4	5	5	4	4	2	1
6	1	5	3	3	4	4	2	2
7	1	3	5	5	5	5	2	2
8	1	4	3	3	3	3	1	1
9	1	5	3	3	4	3	1	1
10	1	5	1	1	4	4	2	1
11	1	5	3	3	3	3	1	1
12	1	5	3	4	1	1	1	1

R-NO	Q16	Q17	Q18	Q19	Q20	Q21	Q22a	Q22b
1	1	3	1	2	1	3	1	1
2	3	1	1	2	1	1	2	1
3	3	1	2	2	1	1	3	1
4	2	2	2	2	1	1	4	5
5	1	3	3	2	2	3	1	1
6	1	2	2	2	1	1	6	1
7	2	5	1	1	2	3	7	8
8	1	2	2	1	2	1	9	3
9	1	3	1	2	2	1	10	1
10	2	2	1	1	2	1	11	3
11	2	2	1	2	2	1	12	1
12	2	1	3	2	2	1	1	1

NOTE:

R-NO: Respondent Number

Q: Questionnaire Question Number

3 CALCULATION FOR DATA ANALYSIS

Perceived Ease of Use

QUESTION No.	TOTAL RESPONSE	SCALE	FREQU-ENCY	TOTAL ON L-SCALE	PERCENT (%)	Approx. (%)
Q14, Q16	24	Strongly Agree	13	2.71	54.17	54
		Agree	9	1.88	37.50	38
		Neutral	2	0.42	8.33	8
		Disagree	0	0.00	0.00	0
		Strongly Disagree	0	0.00	0.00	0
TOTAL		5	24	5	100	100

Perceived Ease of Learning

QUESTION No.	TOTAL RESPONSE	SCALE	FREQU-ENCY	TOTAL ON L-SCALE	PERCENT (%)	Approx. (%)
Q15	12	Strongly Agree	9	3.75	75.00	75
		Agree	2	0.83	16.67	17
		Neutral	0	0.00	0.00	0
		Disagree	1	0.42	8.33	8
		Strongly Disagree	0	0.00	0.00	0
TOTAL		5	12	5	100	100

Perceived Reliability

QUESTION No.	TOTAL RESPONSE	SCALE	FREQU-ENCY	TOTAL ON L-SCALE	PERCENT (%)	Approx. (%)
Q9, Q10, Q11, Q12, Q13	60	Strongly Agree	5	0.42	8.33	8
		Agree	6	0.50	10.00	10
		Neutral	20	1.67	33.33	33
		Disagree	15	1.25	25.00	25
		Strongly Disagree	14	1.17	23.33	23
TOTAL		5	60	5	100	100

Perceived Satisfaction

QUESTION No.	TOTAL RESPONSE	SCALE	FREQU-ENCY	TOTAL ON L-SCALE	PERCENT (%)	Approx. (%)
Q17, Q18	24	Very Satisfied	9	1.88	37.50	38
		Satisfied	9	1.88	37.50	38
		Neutral	5	1.04	20.83	21
		Dissatisfied	0	0.00	0.00	0
		Very Dissatisfied	1	0.21	4.17	4
TOTAL		5	24	5	100	100

Perceived Comparison

QUESTION No.	TOTAL RESPONSE	SCALE	FREQU-ENCY	TOTAL ON L-SCALE	PERCENT (%)	Approx. (%)
Q19, Q20	24	Alternative	8	0.67	33.33	33
		Supplementary	16	1.33	66.67	67
TOTAL		2	24	2	100	100

Perceived Recommendation

QUESTION No.	TOTAL RESPONSE	SCALE	FREQU-ENCY	TOTAL ON L-SCALE	PERCENT (%)	Approx. (%)
Q21	12	Yes	9	2.25	75.00	75
		No	0	0.00	0.00	0
		Not Sure	3	0.75	25.00	25
TOTAL		3	12	3	100	100

Overall Perceived Usability

QUESTION No.	TOTAL RESPONSE	SCALE	FREQU-ENCY	TOTAL ON L-SCALE	PERCENT (%)	Approx. (%)
Q9, Q10, Q11, Q12, Q13, Q14, Q15, Q16	96	Strongly Agree	27	1.41	28.13	28
		Agree	17	0.89	17.71	18
		Neutral	22	1.15	22.92	23
		Disagree	16	0.83	16.67	17
		Strongly Disagree	14	0.73	14.58	15
TOTAL		5	96	5	100	100