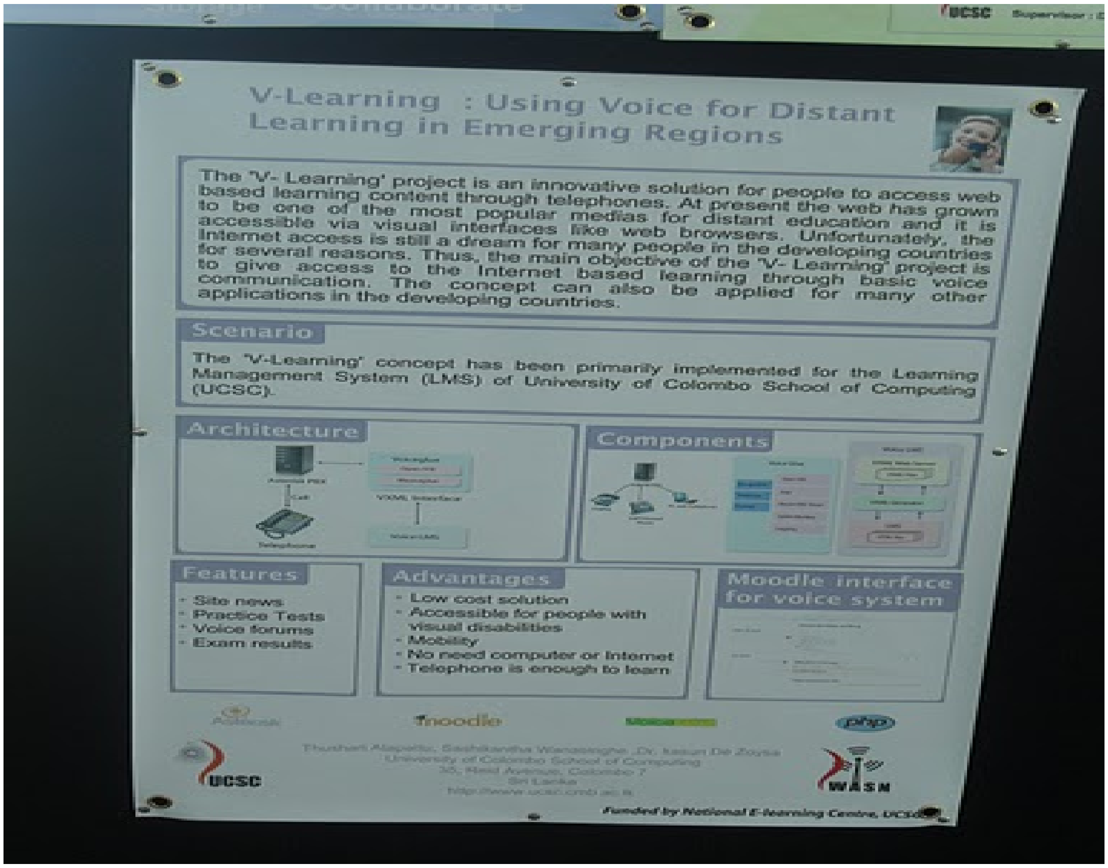
Annex 2:

V-Learning Project

Funded and supported by National e-Learning Centre Project (NeLC)



Introduction to V-Learning project

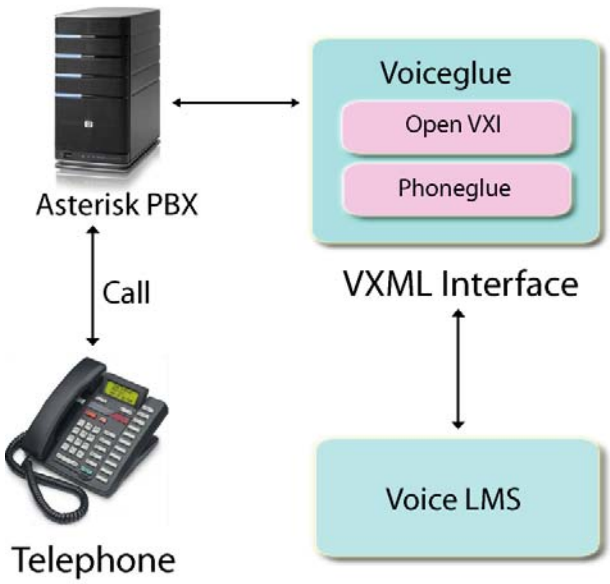
The 'V- Learning' project is an innovative solution for people to access web based learning content through telephones. At present the web has grown to be one of the most popular medias for distant education and it is accessible via visual interfaces like web browsers. Unfortunately, the Internet access is still a dream for many people in the developing countries such as Sri Lanka due to several reasons. Among them, an unavailability of electricity and internet connectivity is the most critical issue faced by the people in emerging regions. Moreover, the high cost of connectivity and other infrastructural facilities are unbearable for people in developing countries.

Thus, the main objective of the 'V- Learning' project is to give access to the Internet based learning through basic voice communication. It reduces the high cost of internet access and it would be more beneficial for users since the set up is highly mobile. Besides, people with visual disabilities could access this system easily without assistance of intermediary. The concept can also be applied for many other applications in the developing countries.

The 'V-Learning' concept has been primarily implemented for the Learning Management System (LMS) of University of Colombo School of Computing (UCSC). The Learning Management system in an instance of open source moodle (http://moodle.org) project and our V-Learning project is also extensible to plug into moodle project.

The V-Learning system

The following figure depicts the overall architecture of the system. The system is fully implemented using open source applications and technologies where it could be introduced as a low cost solution for developing countries.
The functionalities of the system is listed below.

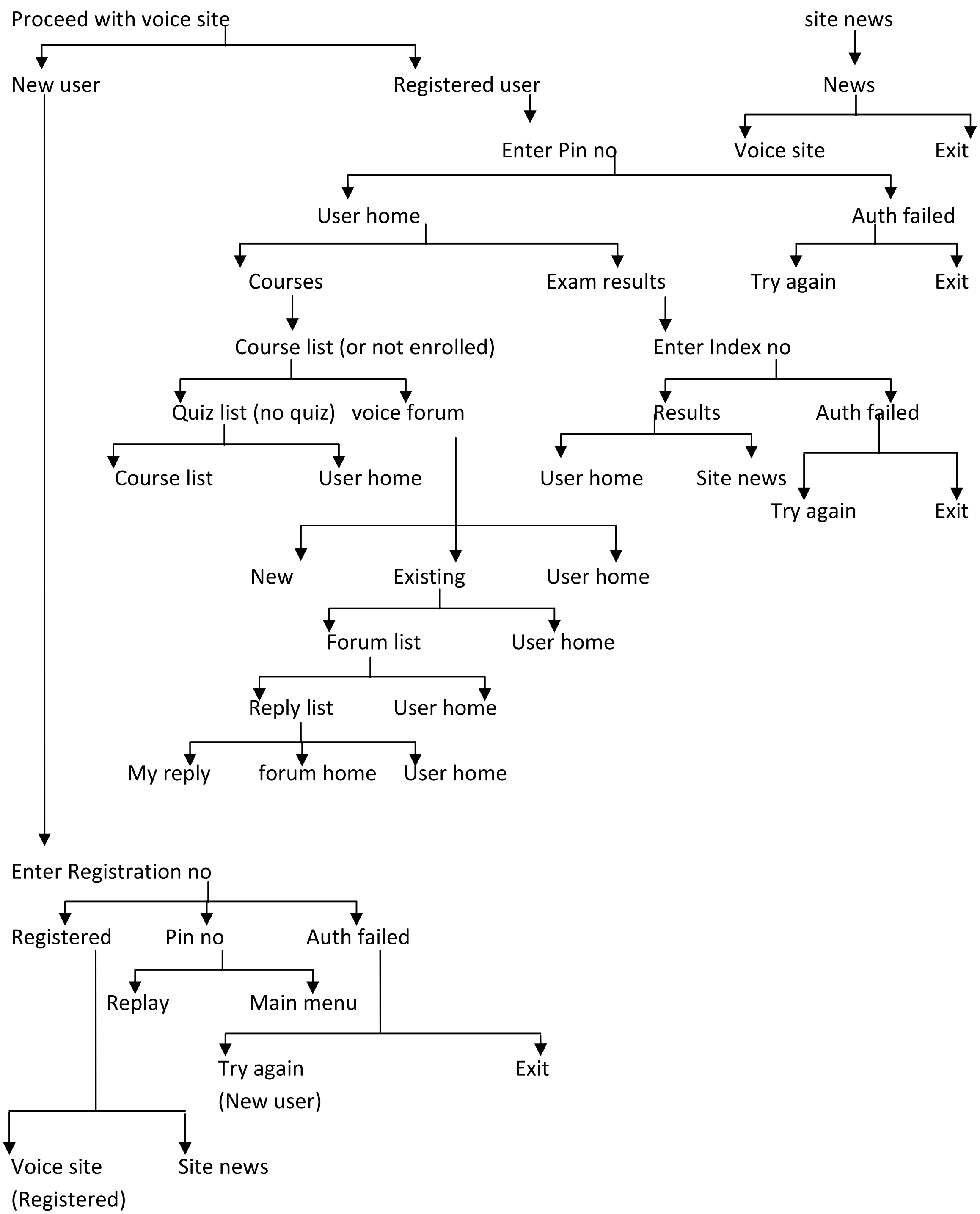


1. A user can place a telephone call to the system through their telephone (for our dedicated telephone line)
2. The call is redirected to the Private Branch Exchange (PBX) server. We have used ‘Asterisk’, which is an open source PBX. Asterisk server contains extensions similar to regular PBX servers. One extension is dedicated to voice learning system.
3. And it (i.e. extension of voice system) interprets voice files through voice interpreter. For the interpretation we have used ‘voiceglue’ which is also an open source application accessible for anybody.
4. We have stored voice files as ‘voicexml’ format.
5. These voice xml files are generated through voice engine implemented for moodle.

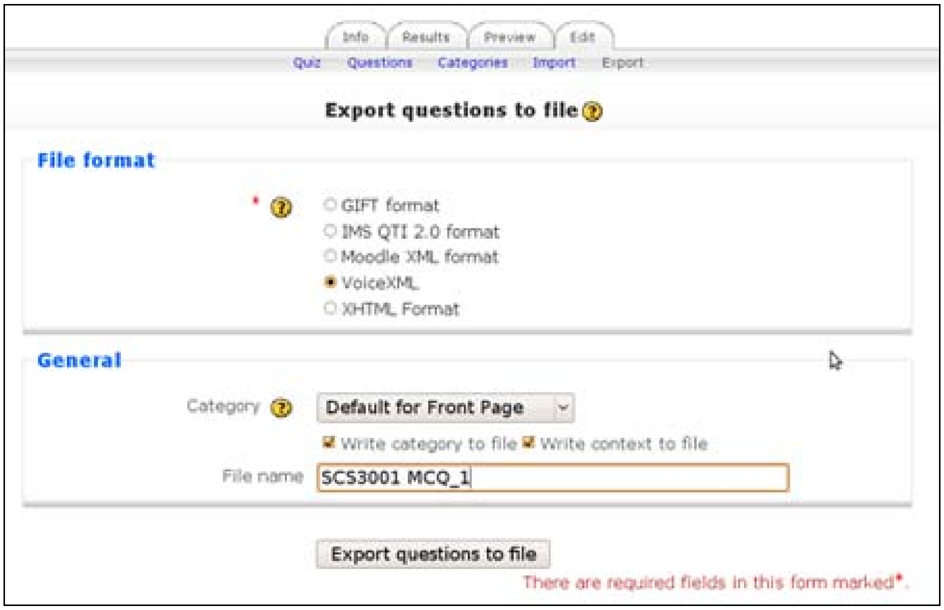
Note: Lecturers can add learning materials to the voice learning system through Learning Management System and those materials would be converted to voice xml format using our system. Besides administrators or any other responsible people can add news, exam results to the voice system and it would be accessible to the students simultaneously. Students also can participate to discussion forums through voice and they also can start new forums or can reply to existing forums.

The flow of the voice system

Welcome to UCSC voice learning system


Proceed with voice site
site news
New user
Registered user
News
Enter Pin no
Voice site
Exit
User home
Auth failed
Courses
Exam results
Try again
Exit
Course list (or not enrolled)
Enter Index no
Quiz list (no quiz)
voice forum
Results
Auth failed
Course list
User home
User home
Site news
Try again
Exit
New
Existing
User home
Forum list
User home
Reply list
User home
My reply
forum home
User home
Enter Registration no
Registered
Pin no
Auth failed
Replay
Main menu
Try again
(New user)
Exit
Voice site
(Registered)
Site news

Moodle interface to voice system



This figure depicts an instance where moodle interface to voice system. Responsible people could export learning materials to the voice system as ‘voicexml’ format using this interface.

V-LEARNING: USING VOICE FOR DISTANT LEARNING IN EMERGING REGIONS

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Keywords: WWTW, VoiceXML, Transcoder, Asterisk

Abstract: At present, accessing the internet through visual interfaces is the most common approach. However, it requires some basic resources such as a computer or web-enabled mobile device, an internet connection, electricity and some amount of IT literacy. Because of the relatively high cost of this set up, underprivileged users are unaware or have no direct access to the internet. Since voice communications through telephony systems do belong to the growing trend, people make use of telephones for various purposes. Among them, accessing web through telephone devices is explored. In this paper, we present an approach to access the learning materials of the Learning Management System (LMS) of University of Colombo School of Computing, Sri Lanka through interactive voice driven applications.

1 INTRODUCTION

World Wide Web (WWW) has become the major information source around the world. People access the web for various purposes such as learning, communication, entertainment etc. As a result, web has grown to be one of the most popular media in the world. But, in order to access the internet, it requires essential resources. Basically, it needs a computer (or modern mobile device, internet kiosks etc.), an internet connection (broadband, wire-less, dial-up etc.), a telephone, electricity and some fundamental IT skills.

Buying a computer and obtaining a fixed internet connection might cost around LKR 80,000 (US$800), which is unaffordable for many people in developing countries. At the same time, such internet connections are not available in the rural areas. Majority of the people in developing countries do not even have electricity in their homes. Due to lack of resources, it is an overhead for a majority of people in the world to access internet directly.

According to the internet world stat (Internet world stat, 2008), approximately 22% of the people in the world have access to the internet (Table 1). The table 1 shows that most of the emerging regions such as Africa and Asia have limited access to the web. It implies that most of the people in the world (78%) are still untouched to these sophisticated facilities.

Table 1: Internet Usage and world population statistics for June 30, 2008

World Region	Population	% Population (Penetration)	% Usage Growth 2000-2008
Africa	955,206,348	5.3%	1031.2%
Asia	3,776,181,949	15.3%	406.1%
Europe	800,401,065	48.1%	266%
North America	337,167,248	73.6%	129.6%
Middle East	197,090,443	21.3%	1176.8%
Latin America/Caribbean	576,091,673	24.1%	669.3%
Oceania/Australia	33,981,562	59.5%	165.1%
World Total	6,676,120,288	21.9%	305.5%

According to the statistics, internet penetration is around 2.2% in Sri Lanka, 2007 (Sri Lanka Internet world stat, 2008). It is relatively a low rate with compared to other educational facilities in Sri Lanka. The situation is raised due several reasons. The most critical issue is the unavailability of electricity and internet connectivity facilities in rural areas. In order to improve these factors, an infrastructural development should be done which requires a huge investment.

As an alternative, government and private sectors have invested on public internet accessible places such as Internet cafes, Public Internet Kiosk etc. Some of these solutions are still not possible for underprivileged users as these solutions are also costly.

Table 2: Internet usage and population statistics in Sri Lanka

Year	User	Population	%penetration
2000	121,500	19,630,230	0.5%
2007	428,000	19,796,874	2.2%

Since mobile technologies have been rapidly growing, people make use of their mobile devices to access the web anywhere in the world. This will reduce the overhead of buying a PC with an internet connection and the expenditure for electricity. Accessing wire-less networks also requires high charges, which is not affordable for the people in developing regions.

Apart from that, accessing internet through mobile devices also has some additional limitations. Firstly, buying a featured phone is not less expensive in Sri Lanka. At the same time, the GSM/GPRS coverage does not exist in rural areas. The mobile device is not always user friendly since it has a small screen and a tiny keypad. All of these limitations will distract people accessing the web through their hand-held devices.

All the above mentioned techniques require browsing the internet through a visual interface such as a web browser. Due to the above mentioned limitations, some researchers have explored the possibility of accessing the web through voice communication. The basic voice communication has had a larger penetration among the world population as well as in Sri Lanka. Therefore, IBM Research Laboratory (Kumar, 2007) has conducted a research, which uses voice to access the internet. This concept is called World Wide Telecom Web (WWTW) (Kumar, 2007). In this model, the voice sites are developed instead of typical web sites. Those voice sites are implemented using a language called VoiceXML (VoiceXML, 2008). VoiceXML is a markup language derived from XML. Users are allowed to call to the voice site which is a collection of VoiceXML pages.

The preliminary attempt of this work is to build an interactive voice learning environment for the undergraduates of University of Colombo School of Computing (UCSC). Since the cost of basic voice communication through telephone is relatively low, accessing web using voice is encouraged. This will be beneficial for underprivileged students who have no direct access to the teaching and learning materials in the web.

This paper is organized as follows. In section 2, the work related to World Wide Telecom Web is discussed. Our proposed architecture and overview of the system is detailed in section 3. The system functionalities are explained in section 4. Finally the proposed system is summarized in section 5.

2 RELATED WORK

WWTW (Kumar, 2007) is a tremendous concept of IBM India Research Laboratory, where voice-driven eco systems are developed parallel to that of the WWW. The approach enables deprived population to become a part of the networked world through low cost voice communication. This concept was the basement for various researches related to voice-enabled applications.

Interactive Voice Response (IVR) systems are currently most widely used voice-driven applications in the world. Air-line, hotel reservations, telecom service providers commonly use these fixed menu-driven, user input (DTMF) based applications. These automated systems require high investments and it is not supportable for non-profit organizations and the government education sector.

Researchers have developed a low cost IVR by integrating the existing open source applications and tools (King, 2006). This system is a hybrid of OpenVXI (Carter, 2002) and Asterisk (Asterisk, 2008). OpenVXI (Carter, 2002) is a VoiceXML interpreter developed by speech group at CMU. It provides APIs for speech synthesis, speech recognition and telephony services. Asterisk (Asterisk, 2008) is the mostly used opensource PBX system in non-commercial applications and Voiceone (VoiceOne, 2008) is the web based GUI for Asterisk PBX. The gateway can be utilized to replace the existing high cost IVR systems.

VOIGEN (Kumar, 2007) enables telephone subscribers to access voice-driven systems through ordinary telephone lines. It permits individuals to create, host and deploy customized voice driven services. VOISERV (Kumar, 2007) is similar to VOIGEN (Kumar, 2007) where VOIGEN (Kumar, 2007) create and deliver data services and VOISERV (Kumar, 2007) delivers converged services. Both the systems create their own customized voicesites.

The IBM WebSphere Transcoding Publisher (WTP) (Lamb, 2008) is a commercially available product that can be used to convert HTML to VoiceXML. A group from Virginia Tech, VA has conducted a research to transcode HTML to VoiceXML using annotations (Shao, 2003).

3 PROPOSED SYSTEM

In order to experiment voice based solutions for distant learning, we have proposed an interactive voice driven system which is explained in this section. Our proposed approach will be developed in 3 stages as listed below,

.

1. A voice component which gives access to practice quizes in the Learning Management System (LMS)
2. Voice site parallel to existing UCSC Learning Management System (http://www.ucsc.cmb.ac.lk/lms)
3. Voice module for open source moodle project (http://www.moodle.org)

At present, we are in the process of developing the first stage of the system. In order to provide voice based access to practice quizzes in the LMS, we have implemented a simple automated Moodle XML (MoodleXML, 2008) to the VoiceXML (VoiceXML, 2008) converter. Moodle XML (MoodleXML, 2008) is a XML based language which follows XML standards. The quizes of the Learning Management System could be exported as Moodle XML. Our converter simply converts the Moodle XML files to VoiceXML files. The converted VoiceXML files are intended to interpret through VoiceXML Interpreter (Carter, 2002).

In the second stage of our proposed project it is expected to build a voice site in parallel to the existing UCSC LMS. This would be fully automated system generated from web system. The voice site is intended to be updated automatically with respect to the web system. At the final stage of the project, we have proposed to build a voice module for the open source moodle project. This would be beneficial to the society, as the people are used to customize moodle for their learning and teaching purposes.

The main focus of our proposed system is to allow voice access to learning materials for the UCSC undergraduate, external and postgraduate students. The system can be sub divided into three main components.

1. Private Branch Exchange (Asterisk server and soft phone)
2. VoiceXML Transcoder
3. VoiceXML Interpreter

The Figure 1 depicts the overall architecture of the system and each of the above sub components will be discussed in sub sections.

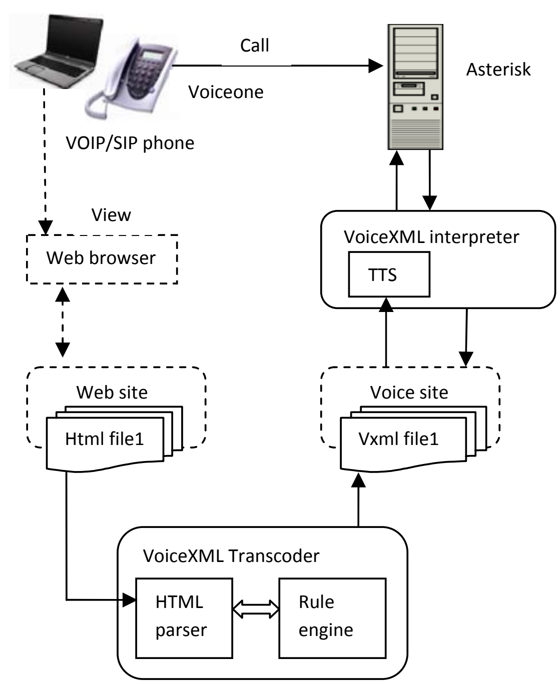

Figure 1: Overall architecture of the system

3.1 Private Branch Exchange (PBX)

Private Branch Exchange is a telephone exchange, which serves a particular set of people. It could be located in a company, school, university etc. The cost of deploying a commercial PBX system is very high. Accordingly, we have used an open source PBX engine called Asterisk server (Asterisk, 2008) for our project.

Besides, one of the latest trends in PBX development is the Voice Over IP (VOIP) PBX, where internet protocols are used to communicate. The initial focus of the development is to configure a SIP phone to connect with the Asterisk server. For this purpose, we have used the freely available Ekiga (Ekiga, 2008) soft phone. A typical PBX set up is shown in Figure 2 below.

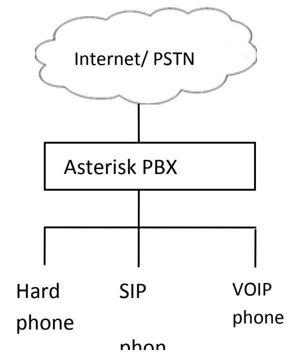

Figure 2: Overview of Private Branch Exchange

The asterisk server is basically capable to,

1. Get the user's input
2. Interactively provide voice response
3. Call forwarding to voice sites

3.2 VoiceXML Transcoder

It is a known fact that web pages are implemented using HTML. Likewise, voice pages have been built using a language called VoiceXML (VoiceXML, 2008). As the HTML pages are interpreted visually through web browsers, VoiceXML files are interpreted using voice browsers. For that the system should generate voice pages or convert existing web pages to voice pages.

The main objective of the system is to implement a voice site in parallel to the existing UCSC LMS web site. In order to do that, selected HTML web pages from the LMS site should be converted to voice pages. This process could be done through "HTML to VoiceXML Transcoder". As there are no any open source VoiceXML transcoders available, our system is expects to implement a VoiceXML transcoder from the scratch. Our proposed transcoder has 3 main components.

1. HTML parser
2. VoiceXML translator
3. Rule engine

The overview of the proposed transcoder is shown in Figure 3.

Firstly, the static HTML pages are analyzed through a HTML parser and a HTML node tree will be generated. Once the structure of HTML node tree is analyzed, the page is converted into a VoiceXML file internally by the system.

When applying the transcoding logic, our system makes use of grammar rules which have been defined by us. After validating the conversion with the rule engine, the syntactically correct VoiceXML file will be created.

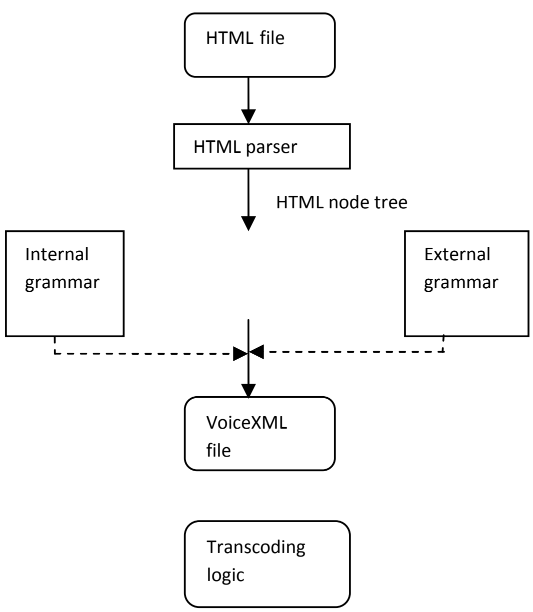

Figure 3: Overview of HTML to VoiceXML Transcoder

A simple HTML file and it’s corresponding VoiceXML file is shown in Figure 4 below.

```
<html>

<head>

<title>Welcome to University of Colombo School of
Computing</title>

</head>

<body>

<form next=”method”

<INPUT type=”radio” name=”degree” value=”computer
science”>computer science<br>

<INPUT type=”radio” name=”degree” value=”ICT”>ICT<br>
```

```
<?xml version=”1.0”>

<vxml version=”2.0”>

<form>

<prompt>Welcome to University of Colombo School of
Computing</prompt>

<field name=”degree”>

<prompt>Select your degree<enumerate/>

</prompt>
```

Figure 4: Simple HTML and VoiceXML file

3.3 VoiceXML Interpreter

XML based languages require an interpreter to interpret the markup commands. Accordingly, VoiceXML files should be interpreted automatically after the file is altered. OpenVXI (Carter, 2002) is one of the freely available VoiceXML interpreter used by majority of voice application builders.

Typical VoiceXML interpreter consists of 3 sub components.

1. Text-to-speech system (TTS)
2. Voice recognition system
3. User action handler

3.3.1 Text-to-speech system (TTS)

Text-to-speech system is a way to present text output to the user through voice communication. In our system, we are using an open source TTS called FreeTTS (FreeTTS, 2008). It extracts the output from the VoiceXML file and presents it to the user through a soft phone.

3.3.2 Speech recognition system

A typical voice-driven application has a component to recognize user's speaking context. In our proposed system, we have omitted this component and instead we are collecting user's input through Dual-Tone-Multi-Frequency (DTMF). The system prompts choices for the user and based on these choices, user has to select a number which can be entered through a telephone dial pad.

3.3.3 User action handler

This component is capable of collecting user's input and respond accordingly. For instance, if the user does not perform any action at his turn, the interpreter gives him a second chance to try the commands or inform him to end the call. Moreover, user action handler collects user inputs which are given by the dial pad. Likewise user action handler automatically performs several intermediate actions like a human being.

4 SYSTEM FUNCTIONALITIES

In this section, we describe the main functionalities of the system. The proposed approach is intended to be accessible via interactive telephone communication only. The user should make a call to the system in order to access the contents. The system is automated to provide services to the user regardless of other matters. The functionalities of the system can be categorized into 2 subsections as follows,

1. User-level functionalities
2. System-level functionalities

4.1 User-level functionalities

In order to get the benefits from the distant learning project, the user should place a call to the system. This can be done through the dedicated telephone number which is assigned to the voice site. The user's call would then automatically be handled by the Asterisk (Asterisk, 2008) server, where voiceone (VoiceOne, 2008) is the front end of the server.

The system identifies the call and redirects it to an appropriate voice site (At present, we have only one voice site in our system). The system prompts information to the user and gets their inputs through DTMF.

A sample user-system interaction is given below,

- User places a call to the system through the voice number given.
- System: Welcome to Learning Management System of University of Colombo School of Computing. Main Menu, For site news press 1, For undergraduate courses press 2, For Examinations press 3, For inquiries press 4, To exit from the system just Hang-up etc.
- User enters 3 through DTMF
- System: You have selected examinations. For Time table press 1, For exam results press 2. To go to the main menu press 0 etc.

The user can navigate through sub menus for his destination or simply can exit from any menu or sub menus. If the user fails to respond to the system within a given time frame, the menu (or sub menu) will be repeated once. If the user does not respond to the system further, the conversation will be disconnected automatically.

4.2 System-level functionalities

At the system level, VoiceXML files will be generated and updated dynamically. This could be done by converting existing HTML files. The collection of VoiceXML files is integrated as Voicesites. The VoiceXML interpreter then interprets these VoiceXML files and presents them to the user through the TTS according to their requests. Before the voice prompts are presented to the user, VoiceXML files will be validated through the system.

5 CONCLUSIONS

The V-learning project is proposed for the underprivileged users to provide access for learning resources through the voice communication. In our approach, we have explored the concept of World Wide Telecom Web that would be parallel to that of the World Wide Web. The motivation of our approach is to deliver the services for the benefit of the students in developing economies. Though it has several benefits such as low cost, it would not be as attractive as graphical user interfaces. We believe that the system would be a bridge between the IT-savvy and the non-IT-savvy population in the world.

ACKNOWLEDGEMENTS

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Accessing the University Learning Management System through voice communication

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ABSTRACT

Among various usages, education is one of the most important reasons for people to use the World Wide Web (WWW). Education through the Internet is a popular approach mostly among university students. The Learning Management System (LMS) of University of Colombo School of Computing has been dedicated for students as a distance learning method. Currently, the access to the LMS is limited through visual interfaces such as Web browsers only. It basically requires a computer with an Internet connection or a Web-enabled mobile phone. The attempt of this paper is to present a way to access the LMS through telephones using voice communication. Browsing the Web using voice is a new concept and it is known as World Wide Telecom Web (WWTW). This approach has been highly motivated in Sri Lanka, since the cost of telephone communication is relatively low. In addition to that, the paper describes several benefits provided from the Voice Learning project for the students in emerging regions.

1.0 INTRODUCTION

World Wide Web has been used for various purposes such as entertainment, learning, communication, information retrieval and commercial requirements. Among them, education is crucial. For time being, Web has become popular for learning around the world. Web has known to be an universal learning repository, where it consists all most all the information required by the people. Individuals make use of it as a distant learning and self studying method. Universities and other education sectors have experimented on various learning approaches based on Internet.

Moodle [1] is one such open source community based tool, which successfully employed in learning purposes. Learning Management System (LMS) of University of Colombo School of Computing (UCSC) has been deployed as a customized version of Moodle project. This provides assistance for students and teachers to share study materials, held on-line assessments, exams etc. This system is currently accessed by the staff, undergraduates, postgraduates as well as external students who are following examinations at UCSC.

Currently it requires a computer, an Internet connection and some amount of IT skills to access the LMS. Since Sri Lanka is having an emergent economy, majority of the people do not have access to Internet. According to statistics if 2008, there is only 3.7% of the people in Sri Lanka have used Internet (Table 1).

Table 1: Internet usage and population statistics in Sri Lanka (source: United Nations Department of Economic and Social Affairs)

Year	Users	Population	Penetration
2000	121,500	19,630,230	0.5%
2007	428,000	19,796,874	2.2%
2008	771,700	21,128,773	3.7%

It is important to increase this lower rate of Internet usage in order to effectively utilize the Web for learning purposes. For that, the government or private sectors of the country should invest huge amount to develop infrastructural facilities. Because some rural areas of the country does not even hold electricity and most of the regions does not have access to Internet facilities. As a result, the development of these facilities is not affordable for a country like Sri Lanka.

Accordingly, this paper explores an economical approach which could be used as an alternative to high cost Internet set up.

The usage of telephones has gained an increasing popularity among the people in Sri Lanka. Since the charges of telephones calls and phones are relatively low, people keep using phones in broad range. Besides, usage of phones has also expanded enormously among university students. Hence, the attempt of the paper is to present our research, which has used telephones effectively for accessing the Learning Management System of the University.

This paper is organized as follows. In Section 3, the work related to current research is discussed. Our proposed system with an overview of architecture of the system is explained in Section 4. In Section 5, we have mentioned the implementation details of the system. Finally, the system has concluded in Section 6 and the possible extensions for the Voice learning system has discussed in Section 7.

2.0 MOTIVATION

The development of the interactive voice system has focused on several objectives. Among them, one of the main goal is to introduce a low cost distant learning solution for the university students in Sri Lanka. And this low cost response could be used not only as a distant learning approach, but also for various purposes such as improve health care system by providing voice sites for people to retrieve and solve their health issues etc. Since we have used open source applications to build our system, we could guarantee that the system could be built within low cost budget.

In addition to that, our voice learning system can be successfully utilized for the people with visual disabilities. As an example, though they could not view the things on the screen, they can listen to the contents of the LMS. As a result, this system also can be introduced as an aid for disable people.

Moreover, our system is capable of working in highly mobile environment. This is one of our main goals in the system, since students can access the LMS , listen to lectures, doing practice tests etc. while traveling. Because they do not required a typical computer set up to access Internet. They only needed a fixed telephone or a mobile phone with a proper communication connection. Hence, unavailability of resources such as computer, Internet connection and electricity is not an issue anymore for students to access the Web.

The system could be fully utilized among the people who do not have much exposure to IT knowledge. Because they only required a little knowledge to operate the telephone such as menu selections, enter Dual-tone Multi-frequency (DTMF) strings etc.

3.0 RELATED WORKS

The Voice-based learning [2] system enables students to learn through interactive voice communication. Students could enter their voice through voice-entry terminal, which is simply recognize spoken input. Similarly, voice output could be given through dedicated device in the computer. The Voice-based learning [2] system have been used among foreign language educators.

Since Interactive voice Response (IVR) systems are widely used in automated voice driven environments, researchers have been focusing on low cost IVR systems. Low cost VoiceXML Gateway [3] is a replacement of traditional IVR systems which deploy voice applications. They have used the technologies such as Asterisk [4], an open source PBX server, OpenVXI [5], an open source VoiceXML interpreter, FreeTTS [6], an opensource Text-to-Speech system etc.

VOIGEN [7] enables telephone subscribers to access voice-driven systems through ordinary telephone lines. It permits individuals to create, host and deploy customized voice driven services. VOISERV[8] is similar to VOIGEN [7] where VOIGEN create and deliver data services and VOISERV delivers converged services. Both the systems create their own customized voicesites.

The Hearsay[9] system allows voice-driven browsing of Web sites. This system partitions existing Web pages and automatically generate voice dialogs from the partitions. The main objective of this project is to provide services for disabled people.

4.0 PROPOSED SYSTEM

Since the long term plan of the project is to develop a voice module for open source moodle [1] project, we have categorized our system into 3 stages as follows,

1. Develop a practice quiz set (Multiple choice quizes) in Learning Management system which could be accessed through voice communication.
2. Build a module for Learning Management system, which converts existing study materials to be deployed in the voice site.
3. Enhance the voice based system to integrate it with open source moodle [1] project.

The initial stage of the project has almost been implemented. The student could access the learning materials of the Learning Management System by placing a phone call to the voice site.

The Figure 1 shows the overall architecture of the project. Each of the components of the system will

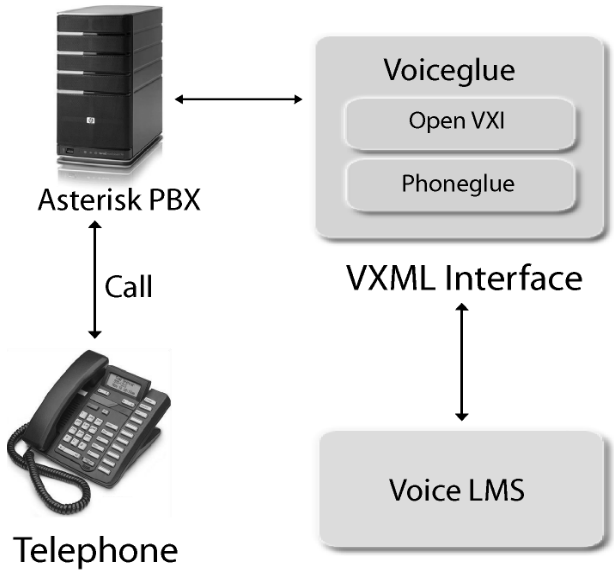


Fig 1: Overall architecture of the V-Learning project

be described as required. For the simplicity, we can categorize our system into three major sections as follows.

4. Private Branch Exchange setup (PBX)
5. Voice Interpreter
6. Voice site

Each individual can make a call (UCSC telephone number : +941125503149) to the voice site using their mobile or fixed telephone line. This call will be forward to the Asterisk[4] server, which has been deployed in the University of Colombo School of Computing.

Asterisk server working as a Private Branch Exchange (PBX). Basically there are relatively high cost PBX servers available commercially. But we have used an open source software solution for this system. One of the basic functionalities of the Asterisk PBX is to automate call center operations. It includes interactive voice responses, automatic call distribution, conference calling, voice mail etc. Further, user could create new functionalities using the dial plan of the Asterisk box. In addition, each individuals can define their own extensions and can attach SIP, IP or PSTN phones to them. An overview of an Asterisk server is shown in Figure 2.

The installation and configuration of Asterisk server is simple and straightforward for any user. Once, the server is successfully installed on a local/remote server (Our Asterisk server has been installed on http://192.248.17.224), it could be used as a real

server and can perform basic call functionalities such as auto answering, call forwarding, call recording etc. on it.

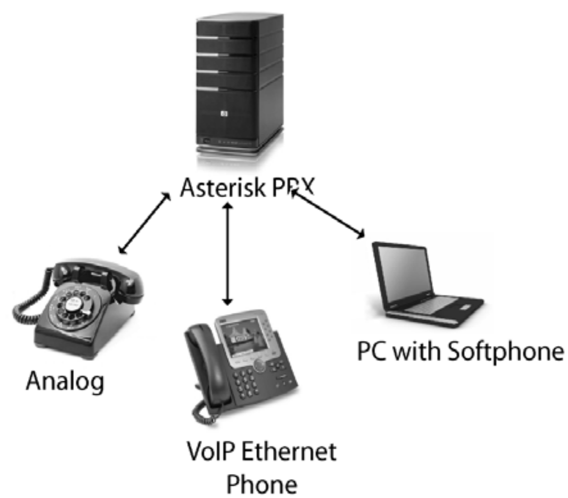


Fig 2: An overview of an Asterisk PBX server

The concept of a voice site (or Voice Learning Management System) is as same as Web site. As Web site is a collection of Web pages, likewise voice site is a collection of voicexml [10] pages. It is not hosted on the Web as a Web site, but it could be browsed using voice by making a phone call to the voice site.

Our system simply converts the existing study materials to voicexml by its Voicexml generator (see fig. 3). In the present system, lecturers are forced to export the learning materials to Voice

Learning Management System (VLMS) which has been hosted on http://192.248.17.224/moodle. Once the LMS materials have been exported as VoiceXML [10] file format (Figure 4), they dynamically updated in the main voicexml file for prompt to the user.

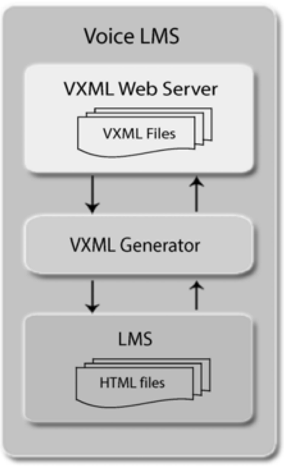


Fig 3: The overview of the Voice site

It is the responsibility of the lecturer to upload his teaching materials to the voicexml Web browser if he wishes they should be included in voice learning system. As it is shown in figure 4, each responsible person can export their files to VoiceXML format and save them in our system.

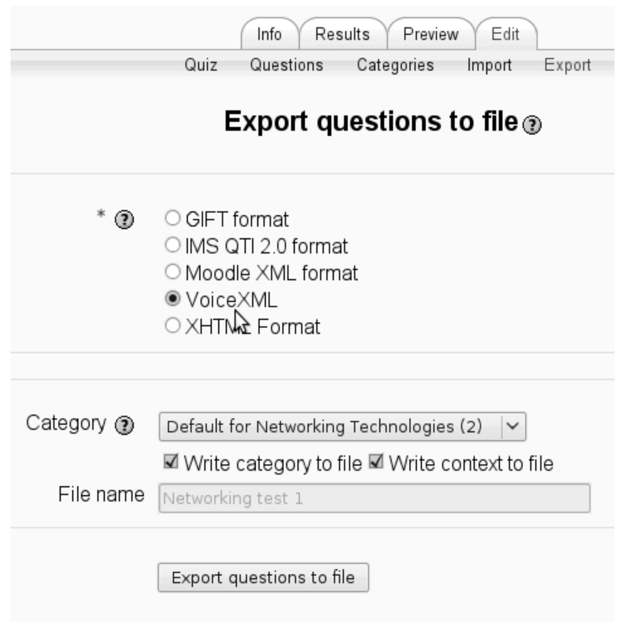
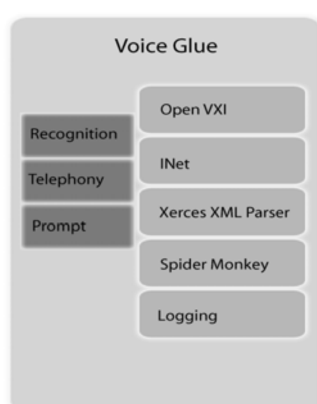


Fig 4: A screen shot of the Voice LMS

There should be a mechanism to execute the generated voice files. In our system, we have used the OpenVXI [5] interpreter which is an open source voicexml interpreter. The Voiceglue [11] project has successfully integrated Asterisk [4] with OpenVXI [5]. Our system has deployed Voiceglue [11] project as a service between Asterisk and Voice site. It also executes some other services such as phoneglue and dynlog. Once the Voiceglue project has successfully configured with Asterisk and voice Web server, the system works fine as a complete cycle. The figure 5 shows a simple architecture of the Voiceglue project.



5.0 IMPLEMENTATION

When a student has placed a call to the UCSC Voice LMS, he/she is required to enter a pin number, which has been given to them from the Learning Management System. Once the authentication has achieved successfully, the student is redirected to the voice learning system. The prompt for the

Fig 5: The architecture of the Voice Interpreter

```
<?xml version="1.0" encoding="UTF-8"?>

 <vxml version="2.1"
xmlns="http://www.w3.org/2001/vxml">

 <menu id="main_menu" dtmf="true">

 <property name="inputmodes"
value="dtmf" />

   <prompt>You are welcome to UCSC voice
learning environment! Please select a
course.

     <enumerate> For <value
expr="_prompt"/>, press <value
expr="_dtmf"/>. </enumerate>

       </prompt>
```

Fig 6: A sample from main.vxml file

students to navigate through the system is as shown in Figure 6.

Students can access the system through phone and enter their desired destination through DTMF. The system tracks the user input and redirect them to their preferred destination. Likewise students can take practice exams, listen to lectures etc. At the moment system is capable of providing these two basic functionalities and the retrieval of news of LMS is under development.

6.0 CONCLUSION

In this paper, we have presented a new concept where people in the developing regions can access Internet using basic voice communication. The approach has been demonstrated using the

Learning Management System of University of Colombo School of Computing. Our attempt is to deliver a low cost solution for distant learning which could be accessible via a typical telephone from any where in the world. Besides, the system is believe to be browsed by the people with vision disabilities and at the same time for the individuals with little amount of IT knowledge.

Thus, with the several benefits on hand, we believe that the system would be an enormous effort to bridge the underprivileged users to IT world.

7.0 FUTURE WORKS

At present, voice learning system has been implemented to access the quizzes of the Learning Management System through telephone. We hope to enhance this feature by providing all the study materials in the LMS which could be accessible through voice. For an instance, system could not play lectures on power point, images etc. If the lecturer feels a specific material should be included in the Voice LMS, it is the responsibility of our system to update them to the interactive voice learning system.

Further, we wish to expand this system as a moodle [1] module for people to browse the open source distance learning tool through voice.

At present, our system is capable of capturing DTMF inputs only. As a result students have to enter everything they needed through the key pad of the telephone. It will narrow down the scope of the project since DTMF inputs would be limited to nine digits only. As a future enhancement, we wish to extend the system in order to recognize user's speech. That would be beneficial for students since there are no limitations and they can express their requirements to the system.

8.0 ACKNOWLEDGEMENT

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