



Implementation of Voice Service Design for Ethiopian higher education institute: A Systems Design Framework, “in Exploring dynamic Voice Response System by integrating text to speech engine Innovative”

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ABSTRACT

Interactive Voice Response System (IVRS) is a technology that permits automated technologies to interact with users via voice or DTMF signaling keypad. Here, researcher presents an Implementation of Voice Service Design for Ethiopian higher education institute: A Systems Design Framework, “in Exploring dynamic Voice Response System by integrating text to speech engine” on asterisk opens source software. The IVRS provides parent of university student, information stored in a database in the form of voice to through their mobile phone by calling on the cost free number. The presented work has defined in three main aspects. The first aspect is to design IVR menu for three languages English, Afaan Oromo and Amharic. Another work here is text to speech synthesis for the three languages. The third and the most important aspect of this work are integrating IVR system with text to speech on asterisk. The architecture of the system has three main components (i.e. presentation, application logic and data storage). The Proposed IVR’s can provide voice information to callers from voice template file by integrating prerecorded text information with dynamically generated voice information from text information by using text to speech engines. Implementation of the system uses asterisk server as middleware between services and the telephony technologies, PHP AGI script for IVR application development, Goertzel algorithm for identifying DTMF signal to accept DTMF digits from the user telephone keypad, speech synthesizer for converting text to speech. This work has shown how easy it is for services to be created in Asterisk. It has also illustrated that these services can be extended so that they are accessible from any interface or device, integrated with text to speech engine and that they can be expanded so that their functionality reaches deep into the system, allowing the users total control.

Key words: Interactive Voice Response System, Asterisk, Goertzel algorithm, Text to Speech, AGI Script

1. INTRODUCTION

Ethiopian higher educations there are 45 functional public universities established in three generation and distributed equitably across the country. Instructors in higher education institutions are expected to possess the academic know how and approaches and provide quality educa

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tion to students to the expected standard. In order to achieve expected quality of education role of parents, instructors and university management needs to work together [6].

Currently higher education institute associate student with his/her parent or emergency person. However, for different socio economic problems most of student parents do not know about student academic status, attendance, disciplinary issues.

IVRS provides student data (student ID, year, CGPA, discipline case, student department, student attendance and related data), parents' data (parent ID and parent contact information), and department head data (name and contact information) stored in computer database in the form voice to parents through their mobile phone or telephone by calling on the cost free number. To date, there have been some research activities to explore the feasibility of IVR interfaces for users who use telephone [12,13,14,15,16,17] and application of IVRS can be used in nearly all industries for reservations, flight information, check in, movie schedules [22]. In [1] an IVRS based college automation system for accessing internal marks and attendance status of student was developed. The design consists of microcontroller board, Global System for Mobile Communication (GSM) module and PC in which application is developed and the necessary data require is stored. When the connection is established between the computer database and the caller, an automated voice instruction is played that directs the caller to give dual tone multi frequency signals through telephone keypad.

Based on the entered DTMF signal appropriate information is fetched from database and played to the caller. In [2] this work researcher proposed done Interactive voice response system for college Automation. Developed project allows the user to know the student's attendance and marks quickly through the telephone line without the intention of the college authority. For the development of the system in the hardware side embedded system has been used. This microcontroller controls the whole hardware. Mobile line is used for communication purpose and Visual Basic has been used as frontend software programming. Paper [3] address the issue of how to design multilingual market information service delivery interactive voice response system architecture to serve hundreds of thousands of low literacy target groups to access market information simply and independently using their mobile and fixed line telephones network for three mostly spoken Ethiopian local languages Amharic, Afaan Oromo and Tigrigna. The author proposed system architecture uses three tier architectural models, which is the fundamental framework for the logical design model, which segments an application's components into three tiers of services namely, the presentation tier, the middle tier, and the data tier. These tiers do not nec

essarily correspond to physical locations on various computers on a network, but rather to logical layers of the application. In the proposed architecture the top layer of the presentation logic is service delivery which is mainly used as an interface to farmers and traders, which are located at distant locations, via the telephone network. The second level of the architecture is middle tiers that accepts inputs from the user and provide required information back to the caller via voice messages. The third tier of the architecture is data tier which consists of voice database and General information service for Market Price database. In paper [4] author designed and developed DTMF based Interactive Voice Response system (IVRS) for Amharic, Afaan Oromo and Tigrigna languages using low end terminal telephone line.

The system is designed to help users to access list of commodity price and trade schedule information via telephone keypad input given by DTMF signal dialing to IVR application service number. To design, develop and implement the proposed system open source Asterisk PBX and X lite VoIP soft phone is used. User interacts to IVR application by using telephone keypad via dialplan which is configured on asterisk server by defining outgoing and incoming call flow mechanism. When messages are received on this channel to asterisk server, the message is sent to the IVR application for processing using Asterisk's Application Gateway Interface (AGI) to interact with users based on predefined IVR voice menu and synthesized speech on IVR application. In [5] the author proposed cell phone based interactive voice response system for college students.

In this proposed system cell phone user can access the information from anywhere at any time simply by dialing a specified IVRS service number and following an instruction when a connection has been established between caller and IVRS service number. The caller gives input in the form of dual tone multi frequency signal, which is obtained when a caller presses a key from their cell phone set. According to the entered response from the caller when the connection is established, computer generates voice response. Voice response is generated dynamically according to the input from caller. As caller enters valid response the corresponding database is converted into voice format by "Text to Speech converter" which is inbuilt in computer with Operating System of the computer. The IVRS system comprises of ring detector circuitry, DTMF decoder section, AVR microcontroller, serial interface unit, computer and cell phone. Using IVR system user In [7] author implemented interactive voice response system for parents of hostel students to inform the student information without parents visiting the hostel and without human interaction. When a parent dial the hostel phone number that we will create as like a

toll free number, they will get the answer in automatic stored voice form. Based on pressing a particular number the corresponding information are automatically stored in a computer. In [8] Interactive Voice Response System for Educational Institution developed. In system when parents call from the registered mobile number to the specified college number, the parents will get the student overall attendance in percentage in voice form. In this system dual tone multi frequency signaling allows for telecommunication signaling over analog telephone lines in the voice frequency band between telephone/cell phone equipment and other communications device once the news domain is decided system will respond to user for making selection on sub domain of news after accepting selection on sub categories of news system will acquire the particular specified news data from news database and deliver it to text to speech module for processing text data and transforming it into voice based form. Transformed voice based news will delivered to the user.

2. Architecture of Proposed system

The proposed Dynamic Interactive Voice Response is service based IVR system developed by three main components (presentation, application logic and data storage) depicted on figure 1 by which the system request and response are handled as a service when the user interact with the IVRS.

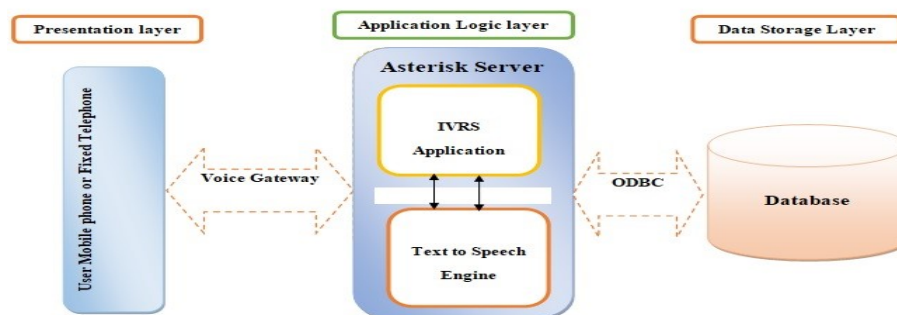


Figure 1: Architecture of dynamic IVRS using Text to speech and asterisk

1.1. Presentation Layer

The responsibility of the presentation component is in order deliver user information based on the dialogue flow designed on IVRS application from the asterisk server on application logic. User mobile and fixed telephone is used for communicating specific service number configured on asterisk server IVR application. User can access needed information from IVR application via keys situated in their mobile or telephone using a method know as Dual Tone Multi Frequency, or DTMF.

1.1.1. Dual Tone Multi Frequency

DTMF tones can also be used to transfer caller ID information in telephone systems [18]. A DTMF incorporates an encoder, that translates key strokes or digit information in to dual tone signals, as well as a decoder detecting the presence and the information content of incoming DTMF tone signals [9][19]. The DTMF system uses eight differed frequency signals transmit ted in dually to represent sixteen different numbers, symbol and letters [10][19]. Figure 2 illu strates how users touch tone telephone keypad is organized by row and a column tone in association with each digit frequencies.

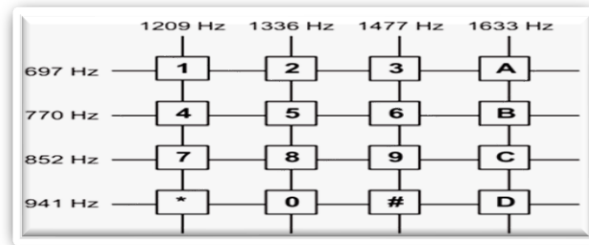


Figure 2. Touch tone telephone keypad: A row and column tone associated with each digit

For the proposed system DTMF is primarily used to process parent's requests to obtain students information and to route IVR menus using the keypad entries made by their mobile or telephone. In proposed system digital tone detection is performed by Goertzel algorithms. Before like other multi frequency receivers, DTMF was decoded using tone filter banks, later, these were substituted with digital signal processors. Although any frequency domain transform such as the fast Fourier transform can be used for decoding DTMF, the Goertzel algorithm is a preferred choice due to its high performance for DTMF signals. The reason authors choose this algorithm is, first the algorithm does not use many constants that saves computation time. Second, only eight DTMF frequencies need to be calculated for this application, and the Goertzel algorithm can calculate selected frequencies, this implies it saves computation time [11].

1.2. Application Logic layer

Application logic is used to accept inputs from the user and provide required information back to the caller via voice messages. Initially when the user connected to IVR system set of default menu will be presented to them in their language from IVR application via telephone keypad reading from a dialplan which is configured on Asterisk server. When messages are received on this channel to Asterisk server, the message is sent to the IVR application for processing using Asterisk's Application Gateway Interface (AGI). The presentation logic consists of Asterisk server, IVR application and text to speech synthesizer as main component.

1.1.2. Asterisk Server

Asterisk server is used as replacement of PBX hardware. The asterisk server is open source software which has a directory of all mobile users with their corresponding Session Initiation Protocol (SIP) address to connect an internal call or route an external call via a VOIP service provider[21]. For the prototype implementation asterisk server is installed on portable laptop on linux environment and audio playback, digit collection, database integration, and speech synthesis features are configured. Four main task done by the asterisk server, one is based on the user request the service is taken out from the IVR application to defined how calls flow into and out of the system using asterisk dailplan, another is speech synthesis and finally integrating speech synthesis engine with IVR application.

1.1.3. IVR application

IVR application is a program written in PHP programming language, and performs any task that is possible in that system. As a security concerns go it executes external scripts (speech synthesis, data base) the scripts to be executed are explicitly specified by the administrator and should only be used if he is sure that they are secure. When the script is ready, it can send a command back to the asterisk from IVR application to the user who has dialed in. The Asterisk Gateway Interface provides a standard API for communication between a script or executable, and the Asterisk PBX. As the name states, it is a “gateway interface”: a standard for external programs to interface with the PBX. Figure 3 illustrates the Communication of IVR application with Asterisk server using AGI



Figure 3. Communication of IVR application with Asterisk server using AGI

1.1.4. Speech synthesis engine

This component is speech synthesis engine for developing English, Afaan Oromo and Amharic. The TTS engine is developed by unit selection speech synthesis by segmenting each language diaphones from their corresponding languagedatabase.

1.2.Data Storage Layer

This component of the architecture is responsible for storage and retrieval of persistent data such as student data (student ID, year, CGPA, discipline case, student department, student attendance and related data), parent data (parent ID and parent contact information), and department head data (name and contact information).

1.3.Algorithm Design

1.3.1. Pseudo Code for IVR application and Goertzel Algorithm

<pre> System prompt "welcome to university student parents information service" System prompt language selection options If user press "1" then System report in english language Else if user press "2" then </pre>	<pre> Nterms Kterm ω = 2 * π * Kterm / Nterms; cr = cos(ω); </pre>
--	--

Figure 4. Pseudo code for IVR application and goertzel algorithm

2. Result and Discussion

The proposed system working principle and how parents interact with system along with detail implementation results of main components are presented. From the developer point of view the proposed system have three components parents mobile phone, asterisk server and IVR application. In this system user interact by using telephone keypad with the asterisk server and then the asterisk server have dial plan which is used to define extension number or unique IVR service number. When messages are received on this channel through session initiation protocol on defined port to asterisk server the message will be sent to IVR application for processing via Asterisk's Application Gateway Interface (AGI). Finally, the IVR application will respond the requested message from speech synthesis and database back to user using previously defined channels and ports on the channel based on predefined IVR menus on IVR application. Implementation of the proposed system uses technologies Asterisk server configuration to act as middleware between services and the telephony technologies, PHP AGI script for IVR application development, Goertzel algorithm for identifying DTMF signal to accept DTMF digits from the user telephone keypad, speech synthesizer for converting text to speech.

2.1.Setting Asterisk Server

The proposed system is implemented after actual Asterisk PBX software installation and configuration is performed on Ubuntu environment. During implantation of the proposed system a number of users of the system needed are created using x-lite soft phone, and then IVRS application is created, text to speech synthesis and text to speech synthesizer is integrated

with IVRS application. In order to use IVR system the Administrator must be logging to the system and then it must start service of the asterisk server. When the administrator starts service of the asterisk server, the interface will be displayed this will be used to start asterisk server.

2.2.IVR Application Development

PHP language is used to develop the proposed IVR system using the PHP AGI script. The major component of the proposed system is developed using PHP programming language. The system call flow is defined by asterisk dialplan on asterisk server on “extensions.conf” file for IVR application to interact with Asterisk server via asterisk gateway interface (AGI).

The Asterisk dialplan is specified in the configuration file named extensions.conf. The content of “extensions.conf” is organized in four main section (context, extension, priority, and application). This components control or execution flow for all of its operations are configured on located on directory /etc/asterisk/extensions.conf..

```
#!/usr/bin/php -q
if($key2=="1"){
    $result3=$agi->get_data("student",40000,1);
    $key3 = $result3['result'];
    if($key3=="1") {
        $result4=$agi->get_data("head_department",40000,1);
        if($result4['result']=="1"){
            $result5=$agi->get_data("welcome",40000,1);
            if($result5['result']=="1") {
                $hostname = "127.0.0.1";
                $dbname = "IVR_Db";
                $username = "root";
                $password = "password";
                $mysql = mysql_connect($hostname, $username, $password) or die(mysql_error());
                mysql_select_db($dbname, $mysql) or die(mysql_error());
                $query = "select * from Parent where ID='id'";
                $r = sqlQuery ($query, $mysql);
                $result5=$agi->get_data("attendance");
                $result5=$agi->get_data($r[0]['ID']);
                main($result[0]['Date']);
                main($result[0]['Year']);
                $agi->stream_file($result[0]['Name']);
                main($result[0]['Department']);
                main($result[0]['course']); } } }
            elseif($key3=="2"){
                $result3=$agi->stream_file("selit",40000,1); }
        elseif($key2=="2")
        elseif($key=="2")
        $agi->stream_file("message_goodbye");
        elseif($key=="3")
            $agi->stream_file("Student");
        $agi->hangup();
    }
}
```

Figure 5. IVR application PHP AGI implementation Code

Asterisk communicates with the proposed application AGI scripts via STDIN/ STDOUT. When a AGI script is invoked from within the Asterisk dial plan, the following steps always happen: Asterisk forks out and runs the IVR application system in its own user space. All channel variables that were available to the Asterisk dial plan, prior to executing the IVR application. Asterisk sends out a bunch of information that must be handled before the IVR application starts running. Finally the IVR application system call flow menu logic options and synthesized speech runs at this point. During the implementation of the application, researcher used an

AGI wrapper classes built from `phpagi.php`, `phpagi asmanager.php` and `phpagi-fastagi.php.classes` [4]. The proposed system IVR application implementation using PHP AGI script is depicted on figure 5 below.

2.3.Setting User’s Mobile or Telephone Communication with Asterisk

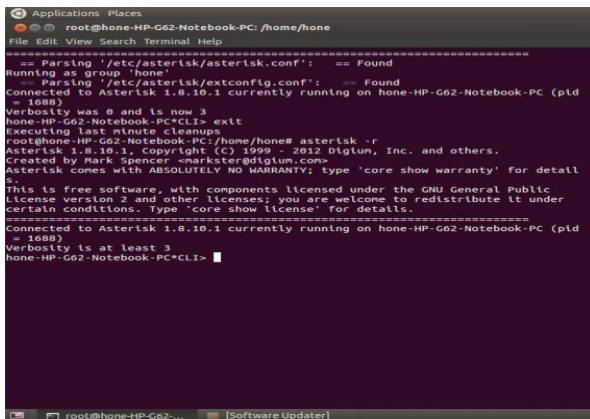
This application used as phone interface to communicate with IVR system. It uses UDP port 5060 and TCP port 10000:20000 [4]. `Users.conf` is a configuration file aimed at defining a "user". Here we defined a user with an optional soft phone as shown in figure 4. In order to establish a SIP call setup two files of Asterisk (`sip.conf` and `extension.conf`) are edited located in the directory `/etc/asterisk/[20]`. In the `sip.conf` we configured settings of user devices (parent and Head department softphone) , where as in the `extension.conf` the dial plans of the phones is configured in the `sip.conf`. After all we reload the newly configured sip files by asterisk command line command "sip reload" so that when incoming calls arrive from devices registered in the phone and currently available online asterisk looks if the call is made according to the settings in `sip.conf` and if okay it looks for the called peer in `extension.conf` when peer got found asterisk forward the call to the peer according to the description of properties of the device found in its own `sip.conf` context. On table 1 we have presented asterisk configuration for parent and head department phones.

Table 1: Asterisk Configuration for SIP call setup, `extension.conf` and `user.conf`

Dial plan configuration on <i>extension.conf</i>	Setting up user devices (parent and Head department soft phone) <i>user.conf</i>	user devices (parent and Head department softphone)configuration <i>sip.conf</i>
<pre>[peer] exten=>Parent,1,Dial(SIP/Parent,20) exten=>HeadDepartment,1,Dial(SIP/HeadDepartment,20)</pre>	<pre>[Parent] type=peer secret=parentpassword; host=dynamic context=local dtmfmode=rfc2833 disallow=all allow=g722 transport=tls context=local [HeadDepartment] type=peer secret=HODpassword; host=dynamic context=local dtmfmode=rfc2833 disallow=all allow=g722 transport=tls context=local</pre>	<pre>[general] tlsenable=yes tlsbindaddr=0.0.0.0 tlscertfile=/etc/asterisk/keys/asteriskk.pem tlscacfile=/etc/asterisk/keys/ca.crt tlscipher=ALL tlsclientmethod=tlsv1</pre>

Extension is simply the phone number of device and it can be a number only or text only or combination of number and text. Here we used texts to name extensions like 'parent' and 'Head Department'.

The soft phone installed on windows operating system machine and connected with the asterisk server via LAN. Figure 6 (a) and figure 6 (b) Demonstrates Asterisk server command line and status information of user soft phone communication with the asterisk server respectively.



```

root@hone-HP-G62-Notebook-PC: /home/hone
=====
Parsing /etc/asterisk/asterisk.conf: == Found
Running as group 'hone'
Parsing /etc/asterisk/extconfig.conf: == Found
Connected to Asterisk 1.8.10.1 currently running on hone-HP-G62-Notebook-PC (pid = 1688)
Verbosity was 0 and is now 3
hone-HP-G62-Notebook-PC>CLI> extl
Executing last minute cleanup
root@hone-HP-G62-Notebook-PC:/home/hone# asterisk -r
Asterisk 1.8.10.1, Copyright (C) 1999 - 2012 Digium, Inc. and others.
Created by Mark Spencer <markster@digium.com>
Asterisk comes with ABSOLUTELY NO WARRANTY; type 'core show warranty' for details.
This is free software, with components licensed under the GNU General Public License version 2 and other licenses; you are welcome to redistribute it under certain conditions. Type 'core show license' for details.
=====
Connected to Asterisk 1.8.10.1 currently running on hone-HP-G62-Notebook-PC (pid = 1688)
Verbosity is at least 3
hone-HP-G62-Notebook-PC>CLI>

```

(a)



(b)

Figure 6. Asterisk Server (a) Asterisk server command line and (b) User soft phone and asterisk communication

2.4. Text to Speech Synthesis

The Proposed IVR's can provide voice information to callers from voice template file by integrating

prerecorded text information with dynamically generated voice information from text information by using text to speech engines. Suppose, parents called to IVR service number to by following

voice instruction prompt and submitted their PIN and student ID to hear students result. The IVR finally speaks out: Students current result is three point five. From the above template file, the first part

of the sentence in italics is prerecorded voice file, which never changes and the second part of information in bold changes from caller to caller which will be synthesized using text to speech engine. Though, the conversion for numbers, date, year, month and characters can happen automatically

inside the IVR application. Speech synthesis system built for the three languages, 60 sample utterances for Amharic, 60 sample utterances for English and 60 sample utterances for Afaan Oromo language. First prepared IVR menu are recorded and saved on computer hard disk to be by synthesized using festival/festvox on Fedora8 operating system. After recording, we labeled the text and the speech data base using a Wave Surfer tool as shown below on figure 7(a) and figure 7(b).

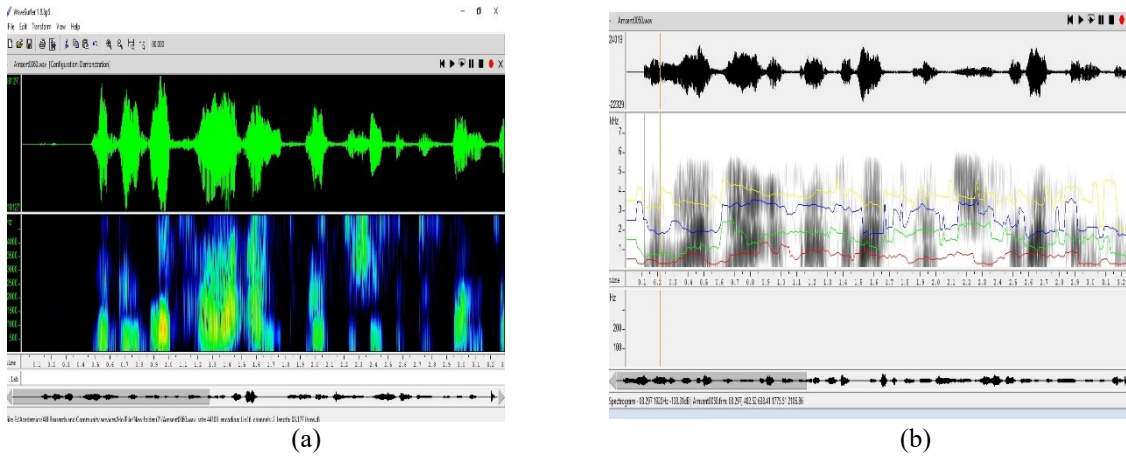


Figure 7. Wave Surfer for TTS (a) Voice template demonstration using wave surfer (b) Speech analysis voice of template using wave surfer

2.5. Integrating Speech synthesizer with IVR application

To integrate Festival server with IVR application festival.scm module is used. The configuration file festival.conf is configured for controlling how Asterisk connects to and interacts with the Festival server. Inside this file, the hostname and port of the Festival server, as well some settings for the caching of Festival speech are specified. Then we invoke a phpagi script that defines the sequence of actions to be performed once a call is received by the server. Once a call is received, the IVR application phpagi script reads out a menu of available options to the user by means of specific languages TTS engine. This is accomplished using the text2wave program of the Festival framework that takes as input plain text and converts it in to audible speech output. The menu provided to the user here is a list of operations to be performed on the server computer along with the corresponding key to be pressed by the user phone keypad.

3. Conclusion

The proposed IVRS provides student data in the form of voice to parents through their mobile phone or telephone by calling on the cost free number. The proposed Dynamic Interactive Voice Response is service based IVR system developed by three main components (presentation, application logic and data storage). In this system user interact by using telephone keypad with the asterisk server and then the asterisk server have dial plan which is used to define extension number or unique IVR service number. When messages are received on this channel through session initiation protocol on defined port to asterisk server the message will be sent to IVR application for processing via Asterisk’s Application Gateway Interface (AGI). Finally, the IVR application will respond the requested message from speech synthesis and database back to user using previously defined channels and ports on the channel based on predefined IVR process tree on IVR

application. Implementation of the proposed system uses technologies Asterisk server configuration to act as middleware between services and the telephony technologies, PHP AGI script for IVR application development, Goertzel algorithm for identifying DTMF signal to accept DTMF digits from the user telephone keypad, speech synthesizer for converting text to speech. During implementation this work has shown how easy it is for services to be created in Asterisk. It has also illustrated that these services can be extended so that they are accessible from any interface or device, integrated with text to speech engine and that they can be expanded so that their functionality reaches deep into the system, allowing the users total control over their environment. In addition, it has investigated and documented various open source systems that were used in achieving these goals, and shown that they are mature enough and stable enough to be used in the deployment of a production telephony system.

In future, it would be good to be able to give SMS information for different users in addition to parents.

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