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DTMF BASED AFAAN OROMO, AMHARIC AND TIGRIGNA LANGUAGE AUTOMATED IVRS

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ABSTRACT

In this paper we designed and developed Dual Tone Multi Frequency (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 in reliable, scalable and user-friendly manner. To design, develop and implement the proposed system the functional requirements are described by the use case model. Based on the requirements the complexity of the proposed system is decomposed into sub systems. Voice interface subsystem is responsible for voice menu handling, DTMF input interpreting, and phone line handling. Speech synthesis subsystem responsible for synthesizing speech for dynamically changing information. To implement the proposed model open source Asterisk PBX and X lite VoIP soft phone is used. Asterisk PBX is connected to with the X lite VoIP soft phone over the local area network. User interacts to IVR application by using telephone keypad via dial plan 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.

Keywords: Automated Interactive Voice Response, Speech Synthesis, Dual Tone Multi-Frequency (DTMF), Asterisk Server, Afaan Oromo language, Amharic language, Tigrigna language.

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1. INTRODUCTION

Interactive Voice Response (IVR) System serves as a bridge between people and computer databases by connecting thetelephone network with the database [7]. IVRS is a phone technology that allows computer to detect voice and DTMF tones using a normal phone call [4]. IVR system uses pre recorded or computer generated voice responses to provide information in response to an input from a telephone caller. The input can be given by means of touchtone or Dual Tone Multi Frequency (DTMF) signal, which is generated when a caller presses a key of his/her telephone set, and the sequence of voice messages to be played without human intervention according to an internal menu structure designed within the IVR application program) and the user input. On the other end of the phone call, a business can employ text to speech (TTS) software to fully automate its outgoing messages. Instead of recording all of the possible responses to a customer query, the computer can generate customized text like responses and read it back to the customer using an automated voice [8][9].

IVR systems can improve call center efficiency with recorded, frequently requested information or by routing callers to themost relevant agent based on their input. IVR systems retrieve information requested by callers and present it in a variety of ways, such as a recorded or synthesizedvoice, fax, web page or even an email, the advantage of an IVR system is that aplications can be customized for almost any situation imaginable to accommodate callers' requests. Telephone users can access information from anywhere at any time by dialing a specified number and following an online instruction when a connection has been established [9][11].

Applications` of IVRS can be used in nearly all industries such as banking, telecommunication, manufacturing and marketing, media, insurance, travel, entertainment, etc. The customer services being provided may include technical support, processing reservations, balance information, payments, transfers, transaction history, the availability and prices of stocks, ordering, order status, tracking shipping status of orders, the current account balance, subscriptions transactions, automated sample preparation of bidding, pricing, policy status, damaged file state, airline schedules, ticket booking, processing reservations, flight information, check-in, movie schedules [5].

2. LITERATURE SURVEY

In [1] a platform for design and development of an Asterisk based Interactive Voice Response to allow inhabitants to access and control the status of their home appliances remotely was proposed. In this IVR based platform, a Bluetooth enabled mobile phone is connected to the system; a computer system which is acting as the Asterisk PBX server. Hence, inhabitants are able to make a call to the Bluetooth enabled mobile phone and then he/she will be redirected to the server to offer the IVR based platform, through which he/she will be able to give his/ her choice of actions to be performed over the appliances. A 'home control system' database is also maintained in the server to store the user PIN and other information, so that only registered users can access the system. Once a registered user inputs his/her commands over the IVR platform, the system will forward the commands to the 8051 microcontroller by using the serial port communication and the actions will be performed accordingly over the actuators.

In [2] authors described the implementation of IVRS and DTMF based voting system. The proposed system contains five major components. Server was used to store the information of voters and candidates detail. Arduino was used to convert data from binary into serial and it is connected to server. The next component was decoder circuit which converts DTMF signal into binary code. User’s mobile was used for calling purpose and another mobile for automatically receiving call. Here using IVRS and DTMF based voting system, can save the time required for going to voting booth. The system is location independent that allows voters possibly will vote from anywhere in the world. The system does not allow even administrator to log in during voting period, hence corruption is not possible. Voters can vote through the toll free number. Hence it is free of cost. It is used for minimizing errors in voting and makes the voting easier. It does not allow the person to vote who is already voted. Using IVRS technology citizen can call on toll free number and by using DTMF technology authenticate himself using their voting number and give their vote using their mobile keypad.

In [3] an IVRS based college automation system for accessing internal marks and attendance status of student was developed. The design consists of arduino 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 (DTMF) signals through telephone keypad. Based on the entered DTMF signal appropriate information is fetched from database and played to the caller.

In [6] In this paper, we address the issue of how to design multilingual market information service delivery interactive voice response system (MMI-IVR) 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 necessarily 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 0 is data tier which consists of voice database and General information service for Market Price database.

3. PROBLEM STATEMENT

Today developing IVR system to be a viable option for different enterprises, there are a number of disadvantages, especially in Ethiopian context. In Ethiopia, most traditional IVR systems are built on top of expensive proprietary voice engines which are in built on expensive proprietary telephony hardware. According to [7], the major drawback of the current private exchange system is it is inefficient and that it required extra wiring for installation. This kind of system is not flexible in order to shift the extension to different location of a particular user.

Developing traditional IVRS have its own drawbacks firstly; the cost of the hosting is still high, because the gateways employed for hosting are expensive. Secondly this hosted IVR solutions may not allow easy customization, such as synthesizing speech in different accents and different languages or recognizing speech in different local languages.

In view of the fact that Ethiopia is a multilingual country with over 80 distinct languages [15] and it is not uncommon to see language barriers while seeking information in language other than one's mother tongue. Therefore, IVR solution for multiple languages is of tremendous importance on a national level, facilitating information exchange and communication by breaking the language barrier.

The question addressed in this paper is how we should design DTMF based IVR system for the three mostly spoken Ethiopian local languages, Amharic, Afaan Oromo and Tigrigna for low literate farmers via fast expanding mobile and telephone line? We focus on primarily in delivery of the market information specifically commodity price and trade schedule information for farmers who will use low end mobile or telephone terminal as their means of communication.

The objective of this paper is (i) to develop commodity price and trade schedule information IVR system menu for Afaan Oromo, Amharic and Tigrigna languages. (ii) To develop Speech synthesis system for Afaan Oromo, Amharic and Tigrigna languages. (iii) To design and implement that the proposed IVR application that output narrated voice to the user via DTMF input.

However, asterisk server simplifies the process of building an IVR and reduces the costs significantly. The asterisk server which acts as VoIP server is the open source software design for establishing the VoIP connection between the users. And also, Asterisk can be programmed for the Interactive Voice Response System used in call center or any organization for making call from PSTN to PBX Asterisk is technology and protocol, which allows you to connect with outside world using traditional telephony technologies or VoIP [10].

Asterisk contains an asterisk's dial plan scripting language that includes commands to play recorded prompts, to collect digits or spoken responses and to reply with synthesized or recorded responses. Besides, the dial plan language incorporates commands for reading from and writing to a number of data sources including databases, web services, Lightweight Directory Access Protocol (LDAP) and calendaring data stores [13]. In such kinds of systems, the voice files were directly saved into a directory in the Asterisk server and a list generated and read out to the user by means of speech synthesis and DTMF key input signal generated from the user telephone was used to select the desired file.

One of the greatest strengths of this proposed work is it is easily accessible for any user who has a connection to a normal telephone network. There is no need of having an infrastructure like a computer or Internet connection. UpToDate and live commodity price and trade schedule information can be disseminated using IVR based systems for low literate group farmer via their own local languages. Besides, implementation and operational costs are also reasonable.

4. PROPOSED METHODOLOGY

Several tools and methods were utilized for the developing the proposed system. In this part development tools used for developing the proposed system and methodologies are described in detail.

4.1. Asterisk Server

Asterisk is an open source tool for building communications applications [12]. Asterisk is used to turn an ordinary computer into a communications server. Asterisk has provided a wealth of functions that make it a powerful IVR system, audio playback, digit collection, database integration, and speech synthesis. For the prototype implementation an asterisk server is used as replacement of the traditional PBX as software on the top of Ubuntu operating system.

4.2. Dial plan language

This component is the heart of Asterisk system. In the dial plan we defined how calls flow into and out of the system. It is a form of scripting language and contains instructions that asterisk follows in response to external triggers. In contrast to traditional phone systems, it is fully customizable form of scripting language, which defines how incoming and outgoing calls flow into and out of the system. The Asterisk dial plan is specified in the configuration file named extensions.conf. The content of "extensions.conf" is organized in four main sections: context, extensions, priorities and applications. This can be either for static setting and definitions, or for executable dial plan components in which case they are referred to as contexts. The settings sections are general and global and the names of contexts are entirely defined by the system administrator. Every section in "extensions.conf" starts with the name of the section contained within square brackets. In asterisk, an extension is used to define the unique series of steps (each containing an application) through which asterisk will take that call. Within each context, we define as many extensions as required. When dealing a particular extension is triggered (by incoming call or by digits being dialed on a channel), asterisk will follow the steps defined for that extension. It is the extensions that specify what happens to calls as they make their way through dial plan. Each extension can have multiple steps, called priorities. The priorities are numbered sequentially, starting with one, and each executes one specific application. Applications are the work horses of the dial plan. Each application performs a specific action on the current channel, such as playing a sound, accepting touch tone input, looking database information, dialing a channel, hanging up call. In our operating environment the dial plan is located on directory /etc/asterisk/extensions.conf. The configuration file "extensions.conf" contains "dial plan" of Asterisk, the master plan of control or execution flow for all of its operations.

4.3. Festival (Festvox)

Festival is an open source text to speech system developed at the University of Edinburgh. Festival offers a free, portable, multilingual language, run time speech synthesis engine for various platforms under various APIs [13]. Fedora8 operating system is used to develop the proposed system Afaan Oromo, Amharic and Tigrigna Languages text to speech (TTS) engine using festival. Afaan Oromo, Amharic and Tigrigna Languages TTS engine is developed by unit selection speech synthesis principle by segmenting Afaan Oromo, Amharic and Tigrigna Language diaphones from each language database with recorded utterances and then concatenated with each other in the process of synthesis. 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 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 DTMF based Afaan Oromo, Amharic and Tigrigna Language IVR phpagi script reads out a menu of available options to the user by means of Afaan Oromo, Amharic and Tigrigna 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 dial pad.

4.4. Soft phone

X lite is a proprietary freeware VoIP soft phone that uses the Session Initiation Protocol (SIP). In order to test the functionality of proposed system, software that emulates telephone is needed. We opted for latter while testing and used x lite soft phone. For prototype implementation the soft phone is installed on windows operating system machine and connected with the asterisk server via local area network.

X lite soft phone used as phone interface to communicate with DTMF based Afaan Oromo, Amharic and Tigrigna Language IVR application by using UDP port 5060 and TCP port 10000:20000 over Local area network.

5. DESCRIPTION OF PROPOSED SYSTEM

The proposed system is framed from an architecture model described on [6]. In addition to what is discussion on literature survey section from the developmental view the presentation tier or more appropriately, user services layer, gives a user access to the application. This layer presents data to the user and permits data manipulation and data entry. The middle tier, or business services layer, consists of business and data rules and also referred to the business logic tier, it is here where all business problems and logic is implemented. The data tier, or data services layer, interacts with persistent data stored at general information service for commodity price database that is the actual DBMS access layer which will be accessed through the middle tier [6].

The proposed DTMF based Afaan Oromo, Amharic and Tigrigna language IVR system is designed by incorporating functionalities like the system should accept farmers input information via their mobile phone, verify large scale farmers authentication information to enable them to use trade schedule, allows all users to select language and let them to identify commodity price and trade schedule information to prompt them commodity price and trade schedule information using Afaan Oromo, Amharic and Tigrigna local languages, allow the user to terminate the call at any time, provide the user to hold up a call in progress, the system also perform speech synthesis.

The proposed system functional requirements are described by a use case diagram from an external point of view as shown on Fig. 1.

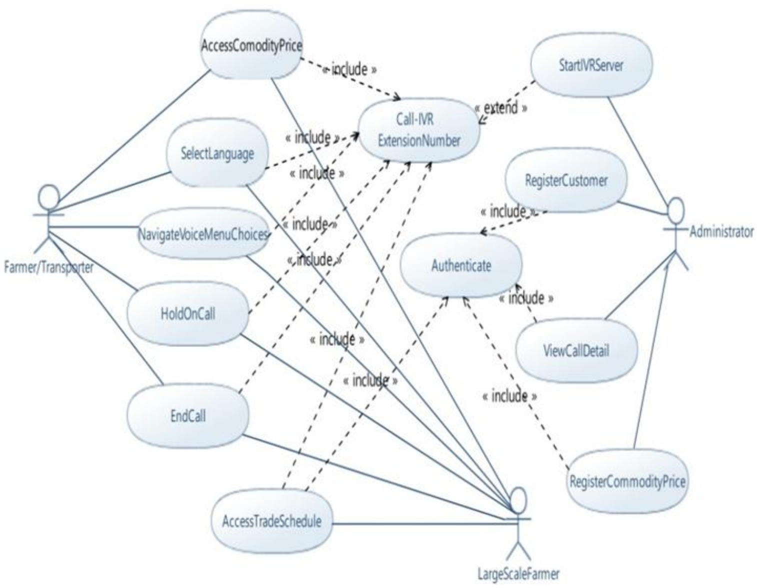

Figure 1 Use Case Diagram of Proposed System

Based on the functional requirements described, the system is decomposed into subsystems shown on Fig. 2. The major subsystems identified are Amharic, Afaan Oromo and Tigrigna language Voice Menu interface (AAT voice menu) and speech synthesis for Amharic, Afaan Oromo and Tigrigna. The subsystems are further decomposed into other subsystems.

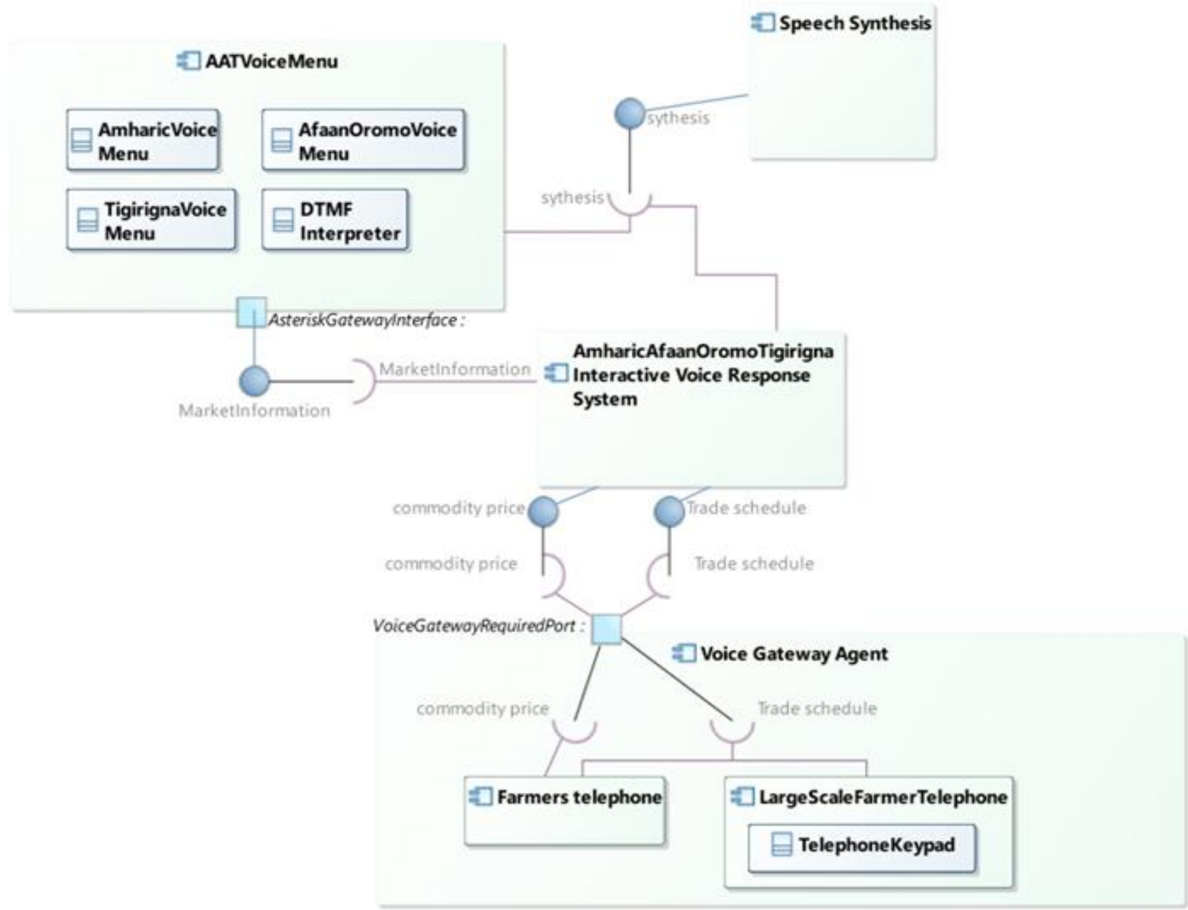

Figure 2 Layered representation on proposed model

5.1. AAT Voice Menu Interface Subsystem

This subsystem is mainly responsible to manage issues related to voice menus handling, DTMF input interpret, reading database information and phone line handling. The AAT voice menu interface subsystem also accepts telephone digits from the user and interprets the instruction and playback the requested information to the caller.

The AAT voice menu interface subsystem accepts inputs from the user and provides required information back to the caller via voice messages. The voice menu subsystem has several pre-recorded template files. Numbers, names of weeks, names of months, commodity items and common words related to commodity price and trade schedule information, and application specific voice messages will be recorded and saved at a predefined location having a predefined file name.

To understand how voice messages are built up and played to the caller, let us assume that the user needs to know the market information for one of commodity item for specific day. To show the overall design process we need to look through starting from the immediate menu item (template file) that will be played back to the user by the IVR system. Let us assume that the price of “1 quintals of Nekemte coffee for the day September five two thousand five is 1579.00 Birr and rank first”. Amharic speaker prompt information will be “የመስከረም 5 2005

ዓ.ም ዋጋ ነቀምቴ ደራጃ አንድ የመግዥያ ዋጋ 1579 ብር፡፡" When this information is requested by the caller, the subsystem reads the information from the database and uses its template files to construct on voice message.

5.1.1. DTMF input Interpreter

The DTMF input interpreter is part of voice interface subsystem which is responsible to accepts DTMF digits via telephone keypad from the user over the telephone network. For example, when a user presses one on the telephone keypad, telephone digit of one will be sent to the subsystem via the telephone network. The subsystem detects the frequency and identifies the key. Touching a button generates a 'tone', which is a combination of two frequencies, one from lower band and other from upper band. For e.g. pressing push button '7' transmits 852 and 1209 Hz. In the keypad ten keys of decimal digits are used to call required number. The touch tone telephone produces decade or DTMF signals for DTMF type. The keypad produces two tone sinusoidal outputs. Rows and columns determine the frequency. This keypad is working with different frequencies but only two frequencies are transmitted at a time. So, the signal coming from this type of telephone is called Dual Tone Multi Frequency (DTMF). Depending on defined voice menu level, the key pressed will be interpreted accordingly [13].

5.2. AAT Interactive Voice Response Menu

AAT (Afaan Oromo, Amharic and Tigrigna) voice interface offers users to communicate with IVR system is using voice prompt that is designed based on the user requirement shown on Fig. 2. Once the user is connected to the IVR system, the system will play based on menu structure to prompt to the user general pre-recorded well come massage to notify that he/she is connected to the system.

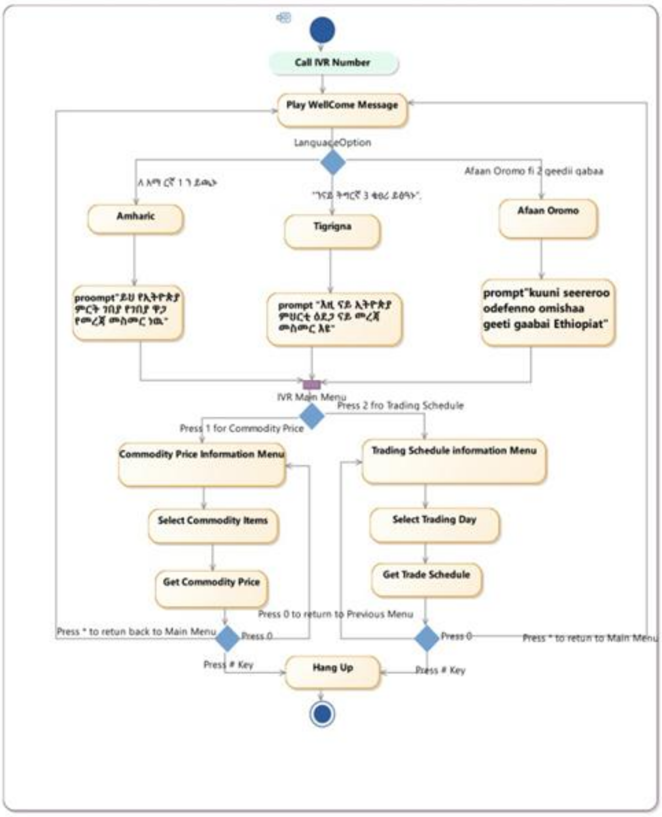

Figure 3 Voice Menu structure of DTMF based Afaan Oromo, Amharic and Tigrigna Language IVR system

The following are pre-recorded welcome message for the three local languages will be played:

Amharic language: "ይህየኢትዮጵያ ምርት ገበያ የገበያዋጋየመረጃ መስመር ነዉ"

Afaan Oromo language: *"kuuni seereroo odefenno omishaa geeti gaabai Ethiopiat"* and Tigirigna language: "እዚ ናይ ኢትዮጵያ ምህርቲ ዕዳጋ ናይ መረጃ መስመር እዩ"

After the system lets the user to select his/her local language based on the prompt and if the user need to access commodity price and trade schedule information in Amharic language the system will prompt ለ አማርኛ አንድን ይጫኑ"[For Amharic press one] . Else if the user is Afaan Oromo language native speaker and needs to access commodity price and trade schedule information in Afaan Oromo language the system will *prompt "Afaan Oromo fi lemma geedii qabaa"*, [for Afaan oromo language press 2] or else if the user is Tigrigna language native speaker and he/she needs to access commodity price and trade schedule information the system will prompt "ንናይ ትግርኛ 3 ቁፀሪ ይፅዓኑ" [press 3 for Tigirigna language press 3]. After the user follows what the system speaks he/she will get appropriate commodity price information and trade schedule by his or her local language.

5.3. Speech Synthesis Subsystem

This subsystem is responsible for converting text to the synthetic speech that is as close to real speech as possible according to the pronunciation norms of each language. In proposed system there are dynamically changing information that will be provided to the system like date, price information, type of market information. For this reason we had used speech synthesis system. In addition, all dynamic voice menus are generated by this subsystem. Synthesized speech can be created by concatenating pieces of recorded speech. The following are designed prompt for Afaan Oromo, Amharic and Tigrigna languages:

- Amharic prompt template for speech synthesis: የ, [ወር ቀን ዓ.ም] ,ዋጋ, [ሀገር], ደረጃ, [ቁጥር], የመግዥያ ዋጋ, [የብር መጠን],ብር፡፡
- Afaan Oromo prompt template for TTS: kan [Ji'a Guyyaa Waggaa], Gatii, [Biyya], Sadarkkaa, [Lakkofsa], Gatii Bittaa, Qarshii, [Gatii].
- Tigrigna prompt template for speech synthesis: ናይ, [ወሪህ ቀን ዓ.ም], ዋጋ, [አገር], ደረጃ, [ቁጥሪ], ናይ መግዝህ ዋጋ [መጠኒ ቂርሽ] ቂሪሽ::

Speech synthesis system built for the three languages based on prompt design, and has been recorded directly to computer files. All the prompts are recorded by female native speaker with the objective to meet the desired style for the synthesizer. We have used 108 sample utterances for Amharic speech synthesis in market information domain, 108 for Afaan Oromo language speech synthesis in market information domain and 108 for Tigrigna language speech synthesis in market information domain. After recording, we label the text using a simple but effective technique based on [14]. We have labelled the speech data base thoroughly using a tool called Wave Surfer tool.

5.4. Gateway Interface Subsystem

This interface is used as a midia gateway for communicating users with AAT system. For this paper we have designed using asterisk gateway interface. Asterisk gateway interface is used as a media gateway, bridging the legacy PSTN to the expanding world of IP telephony [13]. Asterisk Gateway Interface (AGI) enables the development of Asterisk enabled applications without the need of modifying the Asterisk core. The Asterisk Gateway Interface (AGI) is a powerful interface to Asterisk used for implementing the proposed DTMF based Afaan Oromo, Amharic and Tigrigna Language IVR System using PHP programming language. Asterisk communicates with DTMF based Afaan Oromo, Amharic and Tigrigna Language IVR 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 DTMF based Afaan Oromo, Amharic and Tigrigna Language IVR application in its own user space. All channel variables that were available to the Asterisk dial plan, prior to executing the DTMF based Afaan Oromo, Amharic and Tigrigna Language IVR AGI script. Asterisk sends out a bunch of information that must be handled before the DTMF based Afaan Oromo, Amharic and Tigrigna Language AGI script actually starts running. Finally, the DTMF based Afaan Oromo, Amharic and Tigrigna Language menu logic options and synthesized speech runs at this point. During the implementation of DTMF based Afaan Oromo, Amharic and Tigrigna Language application, researcher used an AGI wrapper class PHPAGI. PHPAGI is built from three different classes for programming AGI scripts, phpagi.php, phpagi asmanager.php and phpagi-fastagi.php.

6. RESULT AND DISCUSSION

From the developer point of view the proposed system implementation have user, asterisk server, TTS engine and IVR application components. In this system user interacts to IVR application by using telephone keypad via dial plan which is configured on asterisk server by defining outgoing and incoming call flow mechanism on the system. 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). Finally, the IVR application will interact with the user on the channel based on predefined IVR process tree on IVR application.

The asterisk serve is installed and configured on Ubuntu operating system and the soft phone is installed and configured on windows operating system. The two workstations are connected via peer to peer connection using UDP port 5060 and TCP port 10000:20000. 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. Once the asterisk server is started, the next step is to start the AGI call transmission channel using "agi set debug on" command on command line, which leads IVR application to be ready to accept calls from the user. Now user can able to access commodity price and trade schedule information by dialling IVR specific number as shown on figure 4.

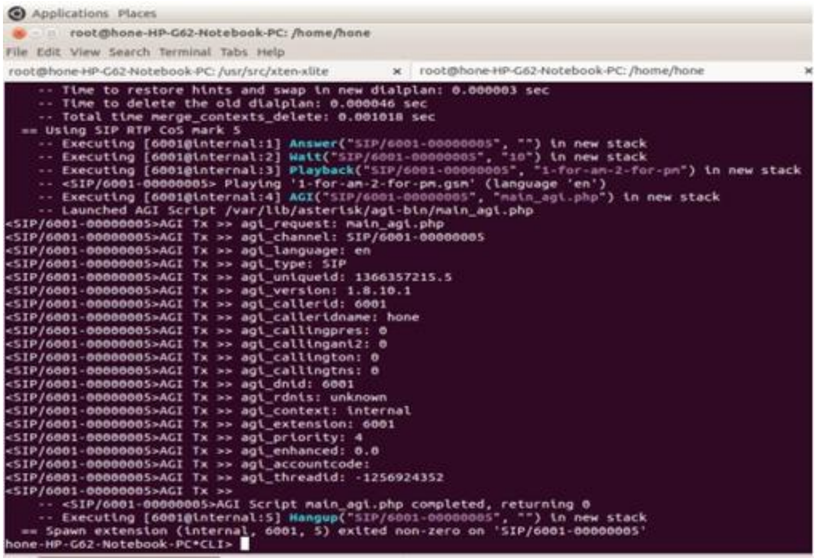

Figure 4 IVR application server displaying the status of main IVR application on call progress

For our demonstration we have used X-lite soft phone. The user will dial the IVR number; this will enable the user to directly connect to AAT-IVR server. Once the user is connected to the IVR system, the system will play based on menu structure to prompt to the user general pre-recorded well come massage to notify that he/she is connected to the system. The following are pre-recorded messages for the three local languages: Amharic language: "ይህ የኢትዮጵያ ምርት ገበያ የገበያ ዋጋ የመረጃ መስመር ነዉ", Afaan Oromo language: "kuuni seereroo odefenno omishaa geeti gaabai Ethiopiat" and Tigirigna language: "እዚ ናይ ኢትዮጵያ ምህርቲ ዕደጋ ናይ መረጃ መስመር እዩ" Will be played by IVR system after the system lets the user to select his/her local language based on the prompt. If the user need to access commodity price and trade schedule information in Amharic language the system will prompt "ለ አማ ርኛ አንድን ይጫኑ"[For Amharic press one] . Else if the user is Afaan Oromo language native speaker and needs to access commodity price and trade schedule information in Afaan Oromo language the system will prompt "Afaan Oromo fi lemma geedii qabaa", [for Afaan oromo language press 2] or else if the user is Tigirgna language native speaker and he/she needs to access commodity price and trade schedule information the system will prompt "ንናይ ትግርኛ 3 ቁፀሪ ይፅዓኑ" [press 3 for Tigrigna language press 3]. After the user follows what the system speaks, he/she will get appropriate commodity price information and trade schedule by his or her local language.

After examination of the findings, the developed system has shown that the system provide good quality speech in terms of naturalness, is easy to navigate and functional in terms of designed menu structure, and it gives customer satisfaction. It is also seen that the developed system disseminates complete, accurate and timely information for all involved stakeholders in understandable format and with relevant interface. This shall be based on rural connectivity, content applicability and capacity to scale besides being affordable and participatory and not determined by geographic proximity, while benefiting all the stakeholders who involve in agricultural innovation process. Besides, the open source tools used in this paper provides a reliable server platform suitable for background preprocess usually associated with telephony and organizations can avoid difficulties that can result in financial losses and customer dissatisfaction.

7. CONCLUSION

In this paper DTMF based IVR system for three mostly spoken Ethiopian local languages, Amharic, Afaan Oromo and Tigrigna speakers to deliver commodity price and trade schedule information via fast expanding mobile and telephone line is designed and implemented. In order to design the proposed system, the functional and nonfunctional requirements are identified and analyzed using use case diagram.

The proposed system menu structure has several prerecorded template files like dates in numbers, names of weeks, names of months and other common words are used for communication and application specific voice messages are recorded and saved at a predefined location having a predefined file name). When required information is requested by the caller, the voice menu interface reads the information from the database and uses its template files to construct on voice message. To accomplish this, it uses the prerecorded template files, and dynamically generated information which basically done by text to speech engine. For our demonstration purpose we used three main Ethiopian languages Amharic, Afaan Oromo and Tigrigna languages.

In addition, we have used various open source telephony tools. During implementation of this project has shown how easy it is for services to be created in Asterisk installed on Linux or Ubuntu operating system and in windows operating system soft phone is installed and connected with the asterisk server via LAN. X lite soft phone used as phone interface to communicate with DTMF based Afaan Oromo, Amharic and Tigrigna Language application by using UDP port 5060 and TCP port 10000:20000.

The flow of the proposed system looks like when the user interacts by using telephone keypad with the asterisk server and then the asterisk server has dial plan which is used to define how calls flow into and out of the system by DTMF signal. 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). Finally, the IVR will interact with the user on the channel based on predefined IVR process tree on IVR application. Speech synthesis system built for the three languages based on prompt design, we have recorded directly to computer files. Most commonly we use a laptop in a quiet room to reduce background noise. All the prompts are recorded by female native speaker with the objective to meet the desired style for the synthesizer. We have used 108 sample utterances for Amharic speech synthesis in market information domain, 108 for Afaan Oromo language speech synthesis in market information domain and 108 for Tigrigna language speech synthesis in market information domain. After recording, we label the text using a simple but effective technique. We have labeled the speech data base thoroughly using a tool called Wave Surfer tool. The developed systems have shown that the system provide good quality speech in terms of naturalness, is easy to navigate and functional in terms of designed menu structure, and it gives customer satisfaction. It is also seen that the developed system disseminates complete, accurate and timely information for all involved stakeholders in understandable format and with relevant interface. This shall be based on rural connectivity, content applicability and capacity to scale besides being affordable and participatory and not determined by geographic proximity, while benefiting all the stakeholders who involve in agricultural innovation process.

8. FUTURE WORK

Universal access is the key to providing information. Author would like to recommend in future people should be able to choose the access method that suits them i.e. literate people might still be more comfortable with written information, in booklets or on a web page where they can print the information they are interested in, but the need for voice information is still there, mainly for low literate and visually impaired people. Besides, increase the naturalness of IVR system and also to integrate high quality of synthetic speech to be produced by Amharic, Afan Oromo and Tigrigna speech synthesis systems.

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