

# IVRS BASED COLLEGE AUTOMATION

Pinkey J. Ratnani 1, Priyanka L. Patil 2, Rashmi A. Sonawane 3, Rohidas B. Sangore 4

<sup>1</sup> Pinkey J. Ratnani BEIT & SSBT's COET Bambhori Jalgaon, Maharashtra

<sup>2</sup> Priyanka L. Patil BEIT & SSBT's COET Bambhori Jalgaon, Maharashtra

<sup>3</sup> Rashmi A. Sonawane BEIT & SSBT's COET Bambhori Jalgaon, Maharashtra

<sup>4</sup> Rohidas B. Sangore BEIT & SSBT's COET Bambhori Jalgaon, Maharashtra

**Abstract**— An Interactive Voice Response is a system based on telephone which allows users to enter information and make menu selections using dual-tone multi-frequency (DTMF) signaling. Interaction of users is done with a computer by using their telephone as a terminal. The objective of the system is to reform the services provided to users by get rid of an operator. An IVR can be tailored to the particular needs of an organization, depending on information contained in database. An IVRS is righteous for handling repetitious enquiries.

**Keywords**—DTMF, FreeTTS, IVRS, Text to Speech, Touch-Tone Key Pad.

## I. INTRODUCTION

We are developing the college automation system which is based on IVR system, which implicates the freeTTS algorithm for voice conversion and DTMF. DTMF, following figure shows the typical DTMF touchpad layout, is a 4X4 matrix with each row representing a low frequency, and each column representing a high frequency.

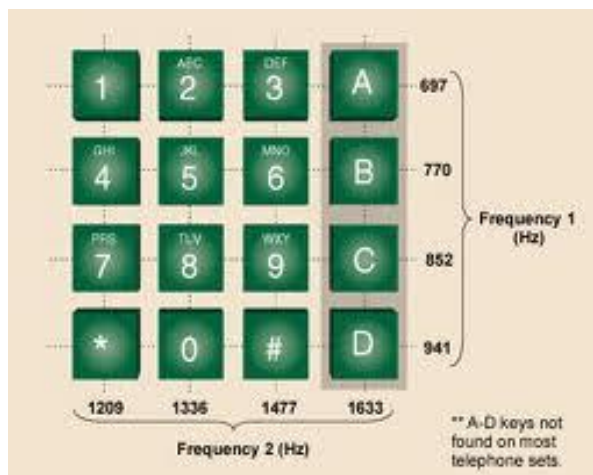


Fig. 1. DTMF Keypad

DTMF generates a sinusoidal tone for each of the two frequencies. E.g. single key (such as '9') will send a sinusoidal tone for each of the two frequencies (852 and 1477 hertz (Hz)). According to the generated frequencies from DTMF keypad, call is directed to the destined user. User will go through the menu selection procedure for getting information. The student has to enter the college's given number and after entering his PRN number he will be able to get his academic details whatever he want. The different amenities are given to the pertinent users. The flow

of menus is shown in following figure. The information is retrieved from database according to the user selection and retrieved information is converted in voice using freeTTS algorithm[5][7].

## II. METHODOLOGY

We are developing the college automation system using IVR system. Which the major part of the system software design. The system software development includes the technologies dual-tone multi-frequency signalling (DTMF), freeTTS etc. When caller dial the number when the technique used for identifying frequency components of a signal is Dual Tone Multi-Frequency (DTMF) detection or decoding. A text-to-speech (TTS) system converts normal language text into speech. For that freeTTS is used[1][2].

### A. Text to Speech

A text to speech (TTS) synthesizer is a computer based system that can read text aloud automatically, regardless of whether the text is introduced by a computer input stream or a scanned input submitted to an Optical character recognition (OCR) engine. A speech synthesizer can be implemented by both hardware and software. It has been made a very fast improvement in this field over the couple of decades and lot of high quality TTS systems are now available for commercial use[3].

## III. SEQUENCE FOLLOWED IN IVRS SYSTEM

1. Caller dials the IVRS service number.
2. The computer waits for a specified number of ringing tones at the end of which, the connection is established.
3. The connection is established by lifting the handset of telephone base from ONHOOK condition.
4. Now, a pre-recorded voice greets the caller conforming that the number dialled corresponding to the particular service.

5. Next, the menu is presented to the caller again in the voice form, giving him then various options to choose from.
6. If the information to be relayed back is confidential, then the system may even ask the dialer, to feed in a password number.
7. The database is accordingly referenced and the necessary information is obtained.
8. Next, the same information is put across to the user in voice.
9. The caller generally given the option to :
  - a) Repeat whatever information was voiced to him.
  - b) Repeat the choices.
  - c) Break the call by restarting ON-HOOK condition

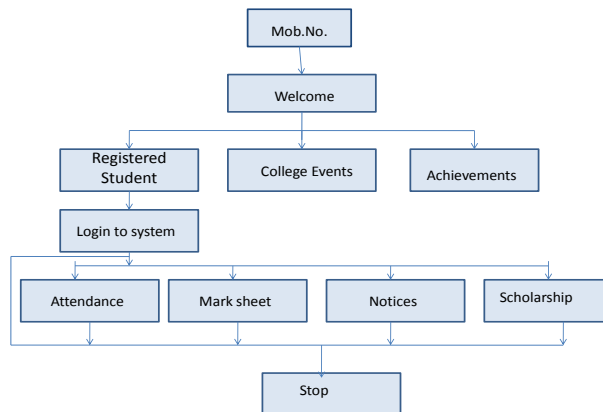


Fig. 2. Architecture of IVRS system

#### IV. IMPLEMENTATION TERMINOLOGIES

##### B. Touch –Tone Key Pad

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, as shown in Table 1.

DTMF Keypad Frequencies			
	1209 Hz	1336Hz	1477 Hz
679 Hz	1	2	3
770 Hz	4	5	6
852 Hz	7	8	9
941 Hz	*	0	#

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)[4][8].

##### C. FreeTTS

The TTS system comprises of these fundamental components

- 1) *Tokenization*: A Tokenizer breaks an input stream of text into a series of Tokens. Typically, a Token represents a single word in the input stream.

Additionally, a Token will include such information as the surrounding punctuation and whitespace, and the position of the token in the input stream. [6]

- 2) *TokenToWords*: The TokenToWords Utterance Processor creates a word Relation from the token Relation by iterating through the token Relation Item list and creating one or more words for each token. For most tokens there is a one to one relationship between words and tokens, in which case a single word Item is generated for the token item[6].
- 3) *PartOfSpeechTagger*: The PartOfSpeech Tagger Utterance Processor is a place-holder processor that currently does nothing[6].
- 4) *Phraser*: The Phraser processor creates a phrase Relation in the Utterance. The phrase Relation represents how the Utterance is to be broken into phrases when spoken. The phrase Relation consists of an Item marking the beginning of each phrase in the Utterance. This phrase Item has as its daughters the list of words that are part of the phrase. [6]

The Phraser builds the phrase Relation by iterating through the Word Relation created by the TokenToWords processor. The Phraser uses a Phrasing CART to determine where the phrase breaks occur and creates the phrase Items accordingly[6]

- 5) *Segmenter*: The Segmenter is one of the more complex UtteranceProcessors. It is responsible for determining where syllable breaks occur in the Utterance. It organizes this information in several new Relations in the Utterance[6].
- 6) *PauseGenerator*: The PauseGenerator annotates an Utterance with pause information. It inserts a pause at the beginning of the segment list (thus all Utterances start with a pause). It then iterates through the phrase Relation (set up by the Phraser) and inserts a pause before the first segment of each phrase.
- 7) *Intonator*: The Intonator processor annotates the syllable Relation of an Utterances with "accent" and "endtone" features. A typical application of this uses the ToBI (tones and break indices) scheme for transcribing intonation and accent in English, developed by Janet Pierrehumbert and Mary Beckman.
- 8) *PostLexicalAnalyzer*: The PostLexica lAnalyzer is responsible for performing any fix ups before the next phase of processing[6].
- 9) *Durator*: The Durator is responsible for determining the ending time for each unit in the segment list. The Durator uses a CART to look up the statistical average duration and standard deviation for each phone and calculates an exact duration based upon the CART derived adjustment. Each unit is finally tagged with an "end" attribute

that indicates the time, in seconds, at which the unit should be completed[6].

- 10) *ContourGenerator*: The *ContourGenerator* is responsible for calculating the F0 (Fundamental Frequency) curve for an Utterance[6].
- 11) *UnitSelector*: The *UnitSelector* that is used by the CMUDiphoneVoice creates a Relation in the Utterance called "unit". This relation contains Items that represent the diphones for the unit. This processor iterates through the segment list and builds up diphone names by assembling two adjacent phone names. The diphone is added to the unit Relation along with timing information about the diphone[6].
- 12) *PitchMarkGenerator*: The *PitchMarkGenerator* is responsible for calculating pitchmarks for the Utterance. The pitchmarks are generated by iterating through the target Relation and calculating a slope based upon the desired time and F0 values for each Item in the target Relation. The resulting slope is used to calculate a series of target times for each pitchmark. These target times are stored in an LPCResult object that is added to the Utterance[6].
- 13) *UnitConcatenator*: The *UnitConcatenator* processor is responsible for gathering all of the diphone data and joining it together. For each Item in the unit Relation (recall this was the set of diphones) the *UnitConcatenator* extracts the unit sample data from the unit based upon the target times as stored in the LPC result[6].

## V. RESULTS

We have provided the data entry module for providing input to the system. The input module is nothing but the website through which the authenticated user can enter the data.

The Figure 3 shows the simulator for the system. Here in simulator user enters the predefined number and thus making call to the server and then according to the type of user(authorized/unauthorized) and according to the choice made by the user system will generate output in speech form by retrieving the text data from database.

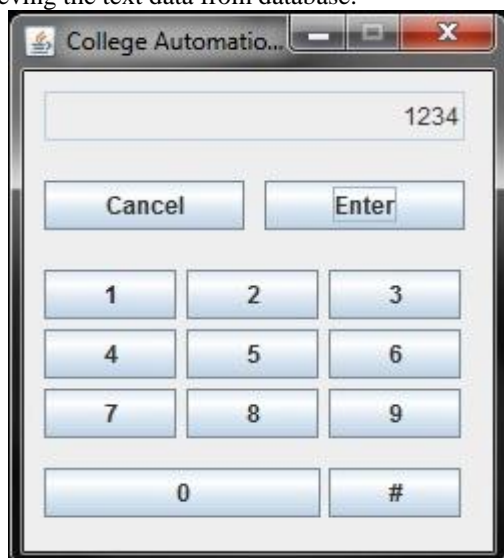


Fig. 3. Simulator for IVRS system

## VI. CONCLUSION

In today's world everyone wants everything to be done from the comfort of one's home or office. For this application is prepared in such a way that users can easily access it through their phone. Due to this project the traditional way of retrieving information will be handled in a more technological and automated way. Because user can access this system from anywhere. This type of system performs operations similar to that of a human telephone operator. The USP of the project is its relevance to the field of telephony and its cost that will be bearable even by a small concern due to its simpler and easily available components.

## REFERENCES

- [1] Santosh A. Kulkarni, Dr. A.R.Karwankar, "IVRS FOR COLLEGE AUTOMATION", International Journal of Advanced Research in Computer and Communication Engineering, Vol. 1, Issue 6, August 2012.
- [2] Prachee N. Kamble, Farheen Khan, Nupur Pande, Tanvi Yamsanwar, "IVRS For College Automation", International Journal on Advanced Computer Theory and Engineering, Vol. 2, Issue 1, 2013.
- [3] D.Sasirekha, E.Chandra, "TEXT TO SPEECH: A SIMPLE TUTORIAL", International Journal of Soft Computing and Engineering, Vol. 2, Issue 1, 2013. Volume-2, Issue-1, March 2012.
- [4] Ms Seema P Mishra, Ms Apeksha S.Chavan, Swapnil S. Gourkar, "INTERACTIVE VOICE RESPONSE SYSTEM FOR EDUCATIONAL INSTITUTION", International Journal of Advanced Engineering Technology, Vol. 3, Issue 1, 2012.
- [5] .ITU's recommendations for implementing DTMF services (PDF)
- [6] <http://freetts.sourceforge.net/docs/ProgrammerGuide.html>.
- [7] C. Marven, General-Purpose Tone Decoding and DTMF Detection, in Theory, Algorithms
- [8] Fotis E. Andritsopoulos, Newton Bomeisel Cardoso, Gregory A. Doumenis, Yannis M. Mitsos, Lambros E. Sarakis, "An accurate Dual Tone Multiple Frequency Detector based on the low-complexity Goertzel algorithm", APRIL, 2001.