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Implementation of Device Control Technique Using Voice Signal

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Abstract: The usage of voice can be effective and simple to be used for a control system. This application involves a unique switching method, where we use voice to replace the physical contacts. This project is a controller based voice recognition system that consists of a microphone; a relay system controller is used to build up the system. The microphone serves as the ear that will listen and passes it to the controller, while the controller works as a brain of the system that will co-ordinate the correct output with the given input command. This project is able to recognize the command trained and successfully execute the desired output.

Keywords: LPF, HPF, PDF, SNR, Controller, Digitization

1. INTRODUCTION

Controlling device has always been a major area of interest, a lot of work has been done in this field and hence many ways are proposed to control the devices. Most of the control devices have been implemented using wired as well as wireless systems, which includes remote control, internet, GSM, Wifi, Bluetooth in order to manipulate the devices.

Problem related to these proposed techniques is that they require telecom technology in one way or the other. The other botheration we faced is that somehow physical assistance is required to operate the devices. To cope with this dilemma we proposed an efficient and cost effective system that helps to satisfy the security concerns such as controlling the devices through voice recognition.

The proposed system is very much cost effective as its hardware is not so expensive so it is affordable to everyone. Speaker recognition is the process of identification and verification of the speech waves which is based on the information of the speaker.

This concept is more alike a comparison between the source and the data stored in memory (the voice that stored during the training process).The way of this concept function is when a user speaks out some command, with then the voice is captured through microphone as the input devices. Once the voice iscaptured, the usage of a decoding system that will convert the analog (voice) to digital signal. The input voice is then compared with the data stored in the memory.The output of the comparison is the voice matched with any of the command trained and certain signal is produce as the input for the controlling system. (Choy, *et. al.*)

2. Study Area

In the recent past, research work has been done for the advancement in technology of controlling devices using voice signal.

First of it will be the automated home lighting system, by a degree student ofUTM. In this projects, the usage of clap(s) as the source of input or command to controlthe lighting system in home. This project offers the ability to control the lighting interm of the intensity or brightness with corresponding to the light intensity in a room due to environment. From this project, it can be concluded that the usage of sound is proof to be a way of controlling the electrical appliance (Brouwer *et. al*). But the application will be limited to one electrical appliance.Follow by another research by a master student of UTM. While for this project the voice is applied as a way to control the wheel chair movement. In this project,a wheel chair is modified by equipping it with the motor system that will read thecommand given by the user to control its speed and movement direction (Hashimah, *et. al.*). Thisproject is successful due to the usage of the HM2007 voice recognition chip. Fromthis project, it can be concluded that the usage of the voice is capable to be one of themethod to control electrical devices provided a suitable system is used.

The third project is done by a degree student of UNITEN. In this project, voiceis used as secondary security measure to access to a restricted area. The overview ofthis project is to create a system that required the user to key in a series of password

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and later be verifying with the voice of the user if only both match then the security door will be unlocked (Hashimah, *et. al.*). From this project, voice is having a very high potential to bedeveloped as one of the key component of a security related system. The main reason voice is a unique for each and every person. Thus, by putting in the voiceas one of the criteria to access a system will make the system more safety and secure.

3. METHODOLOGY

The methodology that is adopted to work out in the project is as follows:

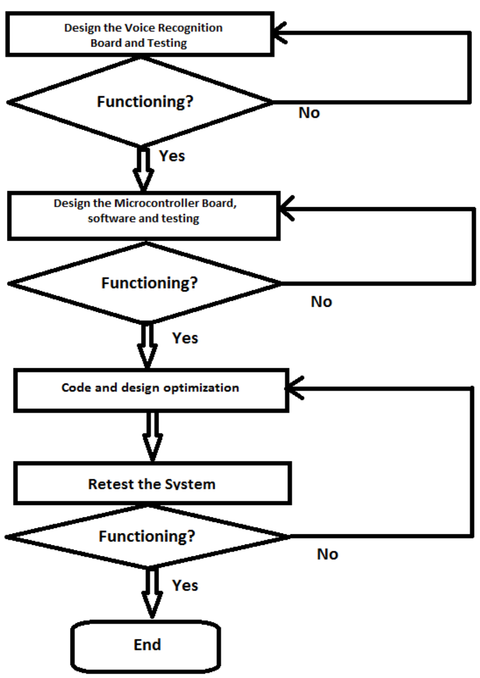

Fig 1: Methodology

Proposed Frame Work

The proposed model is the system in which the user gives input command to the microphone in order to control the devices.(Brouwer *et. al.*)

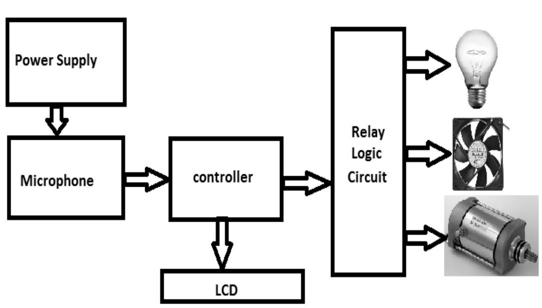

Figure2: Block diagram of controlling devices through voice recognition

The block diagram of purposed model is shown in **fig.1-3.** It works in a way that when we speak into the microphone the respective input will be sent to the controller. In the controller the input will be compared with the stored data and the respective device will be switch on or switch off.

A. *Hardware design*

The power supply circuit serves two basic functions; one is to step down the main AC voltage and second is to convert AC voltage into DC voltage. The designed microphone circuit is basically providing the analog inputs in the form of human's voice to the controller circuit. The controller based circuit is designed that is taking input from the microphone circuit and providing the comparison with the already stored data during the training process. The relay circuit is used to control the devices as per the instruction from the controller circuit.(Sonam *et. al*)

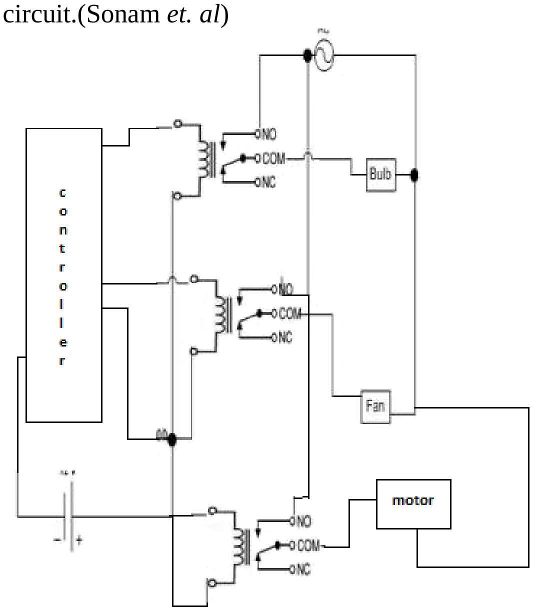

Fig 3: Hardware Design

B. Software design

The application is developed in PIC C using the language C#. Visual Basic is used to carry out the function of voice recognition system. This application has two main parts. One is to understand the speech or the voice of user. The process of training of voice recognition is performed through visual basic. Many different samples are taken of the related word and then their average range is stored in the controller. The delay function is used to perform the process of voice recognition in the controller. Second is to control the appliances according to the user demand. This can be done through relay based circuit that is attached with computer through serial port.

C. *System overview*

User from microphone inputs his voice which is to be fed to a controller. The input voice is then converted to digital signal and is checked by the controller, if it is the required signal for the respective input. The respected signal is then compared with the stored signal. The voice signal is matched with the stored waveforms of the different commands that are installed in the controller. If the command is about changing the status of appliances, it passes the required voltage to the relay circuit to switch on or off the appliances **Fig 4:**.

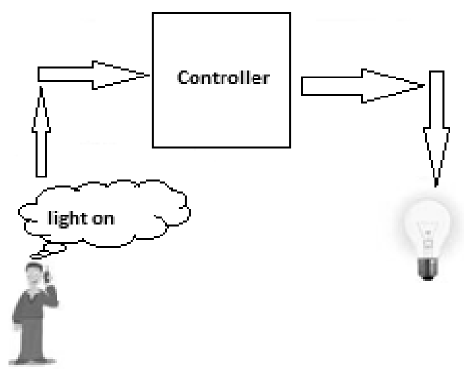

Fig 4: System Overview

PROPOSED TECHNIQUE

A. Waveform of coding of Voice Signal:-

i. Digitization of Speech

In general, waveform codec's are designed to be signal independent. They are designed to map the input waveform of the encoder into a facsimile-like replica of it at the output of the decoder. With the help of this useful property, they can encode some other types of information just like signaling tones, voice-band data etc. Their coding efficiency is usually quite low. This coding efficiency can be improved by some statistical signal properties. For example, if the codec parameters are optimized for the most expected categories of input signals, with also maintaining good quality as well.

As noted earlier the waveform codecs can be further subdivided into time-domain waveform codecs and frequency-domain waveform codecs. Let us initially consider the first category. The digitization of analog signals, such as voice, for example, requires the following steps (see Figure 5); the corresponding waveforms are shown in Figure 6.

• Anti-aliasing low-pass filtering (LPF) is necessary in order to band-limit the signal to a bandwidth of B before sampling. In case of speech signals, about 1% of the energy resides above 4 kHz and only a negligible proportion above 7 kHz. Hence, quality speech links, which are also often referred to as widebandspeech systems, typically band-limit the speech signal to 7-8 kHz. Conventional telephone systems usually employ a bandwidth limitation of 0.3-3.4 kHz, which results only in a minor speech degradation, hardly perceptible by the untrained listener.

The band-limited speech is sampled according to the Nyquist Theorem, as seen in **Figure 5,** which requires a minimum sampling frequency of Nyquist = 2-B. This process introduces time-discrete samples. Due to sampling, the original speech spectrum is replicated at multiples of the sampling frequency. This is why the previous bandlimitation was necessaryin order to prevent aliasing or frequency domain overlapping. If this condition is met, the original analog speech signal can be restored from its samples by passing the samples through a low pass filter with a bandwidth B. In conventional speech systems, a sampling frequency of 8 kHz corresponding to a sampling interval of 125 is used.

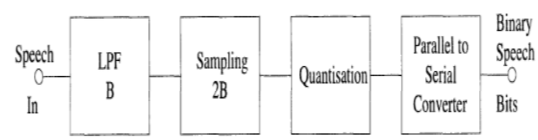

Fig 5: Conversion of speech signal into binary bits

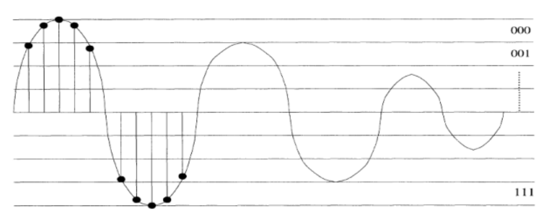

Fig 6: Sampled Waveform

Lastly, amplitude discretization or quantization must be invoked, according to **Figure 5**, which requires an analog to digital (A/D) converter. The out bits of the quantizer can be converted to a serial bit stream for transmission over digital links.(Sonam *et. al.,)* (Wikipedia)

ii. Quantization Characteristics

Figure 7 shows that the original speech signal is contaminated during the quantization process by noise. The severity of contamination is a function of the signal's distribution, quantizer resolution, and its transfer characteristic.

The family of linear quantizers exhibits a linear transfer function within its dynamic range. They divide the input signal's dynamic range into a number of uniformly or non-uniformly spaced quantization intervals, as seen in **Figure 1.10,** and assign an 7^-bit word to each reconstruction level, which represents the legitimate output values. In Figure 7, according to R=3 there are 23=8 reconstruction levels, a quantizer is featured, where the quantizer's output is zero, if the input signal is zero. Note that the quantization error characteristic of the quantizer is also shown in **Figure 7.** As expected when the quantizer characteristic saturates at its maximum output level, the quantization error increases with no limit.

The difference between the uniform and non-uniform quantizer characteristics in Figure 7 is that the uniform quantizer maintains a constant (Zhang H. et. al) maximum error across its total dynamic range, whereas the non-uniform quantizergives unequal quantization intervals in order to allow larger irregular error, where the input signal is larger. Hence, the non-uniform quantizer exhibits a near-constant signal-to-noise ratio (SNR) across its dynamic range. This may allow us to reduce the number of quantization bits and the required transmission rate, while keeping unimpaired speech quality.

In summary, linear quantizers are conceptually and implementationally simple and impose no restrictions on the analog input signal's statistical characteristics such as the probability density function (PDF). Clearly, they do not require a priori knowledge of the input signal.Joanne *et. al.*) Note, however, that other PDF-dependent quantizers perform better in terms of overall quantization noise power or signal-to-noise ratio (SNR). These issues will be made more explicit during our further discourse.

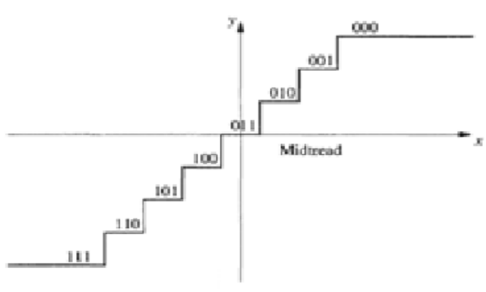

Fig 7: Quantized Waveform

4. RESULTS

After taking many samples of the specific word by different people, we have taken the minimum and maximum time levels of that word. So in this way, we created a time range of that word and then stored this range in the controller for the comparison of the input signal. As shown in the fig8, we have taken the maximum range of 2.

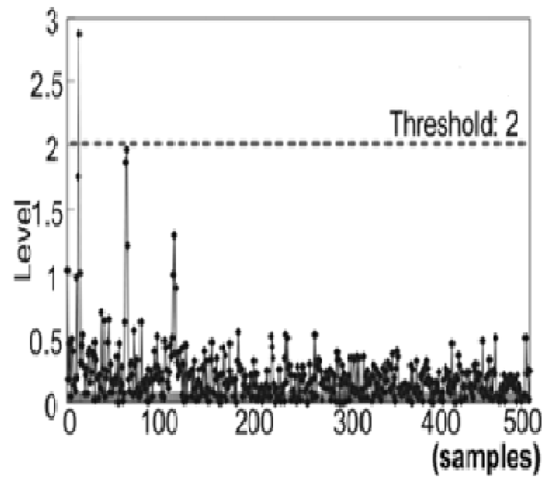

Fig 8: Delay Samples

Graphical Overview

We have also given a graphical overview to check the response time of different appliances. The graph shows the maximum probability of any appliance to be in working condition. In the below graph, we have shown the response time for different appliances. As soon as the system receives the voice command from the user regarding any appliances like motor, light or fan to change the status of them or to know the current status of them the system response in 6, 4 and 3.5 seconds respectively. This is the average response time of appliances as shown in **Fig 9** below:

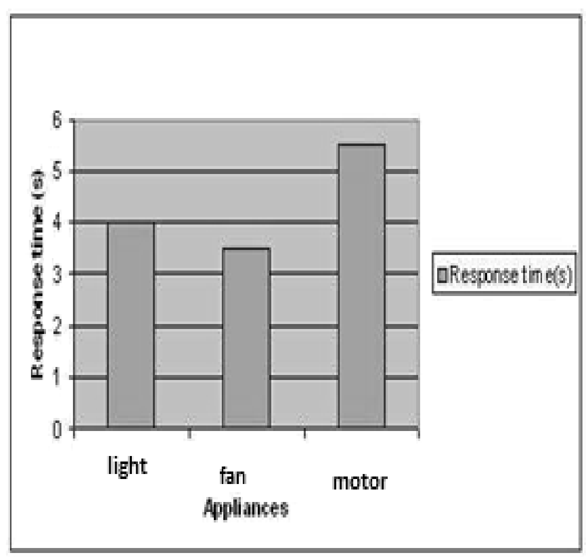

Fig 9: Response time of appliances

The following graphs in **Fig 10 and Fig 11** shows the time intervals for which the specific devices will be switched on and off.

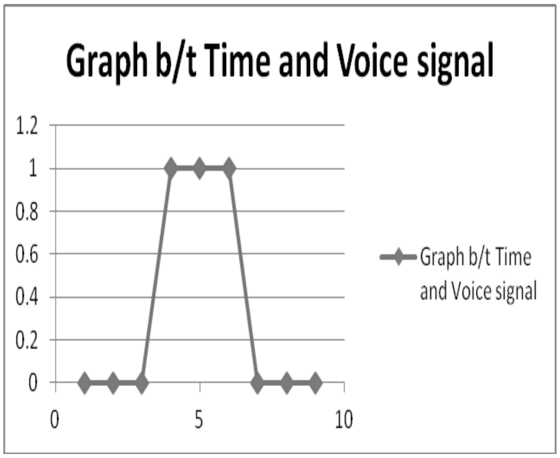

Fig 10: Graph of bulb

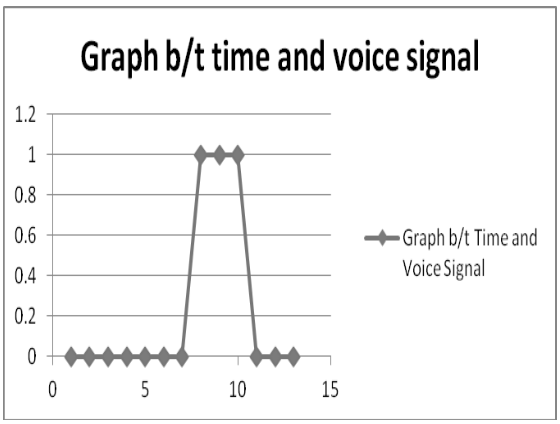

Fig 11: Graph of motor

Comparison Of Different Systems

Table1: Different Systems Comparison

No.	System	Technique	Characteristics		
			CostEffective	Highly Accessible	Security
1.	Ubiquitous Access to home appliances control system using infrared ray and power line communication	Power line communication (PLC), IR	NO	Yes	Available
2.	Remote Control using Mobile through spoken commands	GSM technology	Yes	No	Not Available
3.	Home appliances controlled using speech recognition in wireless network environment	Wireless technology	No	No	Not Available
4.	Home automation using cell phone & J2ME with feedback instant voice messages.	GSM technology	Yes	Yes	Available
5.	Controlling home appliances remotely through voice commands	Voice GSM technology, AT command	Yes	Yes	available
6.	Proposed System	Microprocessor Based Technology	Yes	Yes	Not available

5. CONCLUSION

Voice recognition systems are useful in many ways but due to their limitations, high cost and technical difficulties these systems still not become user friendly and adoptable(Nakano, T). So, in our paper we have discussed a technique that works on the principle of time delay of the signal to store the data and later the comparison performed on the controller will enable a high state if matched to stored data. Thus a control of device become efficient and easy and the cost effectiveness can be easily achieved by using the proposed method.

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