الآيـــــــــة

بسم الله الرحمن الرحيم

قال تعالى: (وَسَخَّرَ لَكُمُ اللَّيْلَ وَالنَّهَارَ وَالشَّمْسَ وَالْقَمَرَ وَالنُّجُومُ مُسَخَّرَاتٌ بِأَمْرِهِ إِنَّ فِي ذَلِكَ لَآيَاتٍ لِقَوْمٍ يَعْقِلُونَ) صدق الله العظيم

(سورة النحل الآية ١٢)

Dedication

To my Family

Teachers

Friends

And

School of Electronics Engineering Sudan University of Science and Technology.

Acknowledgment

It is a great pleasure for me to express my thanks and heartiest gratitude to all those who have helpme during the development of the project.

My most sincere thanks to Dr. Ala'adeenAwouda for his co-operation and who has always been guiding, encoring and motivating me through- out the research with his experience and knowledge.

Abstract

This project proposes a design of voice activated with noise reduction system using microcontroller. The system is aimed to makethe voice oriented command is more cleared. A voice recognitionmodule has been added to the wireless system to store the voice command. The voice command is the system comprisesof transmitting section and receiving section,the system uses the Radio Frequency (RF) technology for transmitting and receiving data. Initially, the voice command is stored in the data baseusing the function keys. Then the input voice commands are transmitted through wireless. The voice receivedis then processed in the voice recognition system where the feature of the voice command is extracted and matchedwith the existing sample in the database. The module recognizes the voice and sends control messages to themicrocontroller. The programmed microcontroller is then processes the received data and switches the respectiveappliances via connected driver circuits. The results gained from the system were very good. The system was able to filter the noise and recognize the voice command. Severalsoftware was used for the simulation purpose such as BASCOM and PROTOUS.

المستخلص

يقدّمهذاالمشروع،تصميم نظام التفعيل بالصوت باستخدام المتحكم الدقيق مع تقليل الضجيجالنظام يعمل علي توجيه اوامر الصوت بدقة اكثر وضوحا،اضيفت وحدة تذكر الصوت للنظام اللاسلكي لتخزين اوامر الصوت، اوامر الصوت تمت مقارنتها في وحدتي الارسال والاستقبال،النظام يستخدم تقنية الموجات الراديوية لارسال واستقبال البيانات. مبدئيا يتم تخزين الصوت في قاعدة بيانات بمساعدة لوحة مفاتيح وبعد ذلك يتم ارسال الصوت المدخل عبر الارسال اللاسلكي ليتم استقباله ومعالجته في نظام وحدة الصوت ومقارنته بالصوت المخزن مسبقا داخل الوحدة فاذا كان الصوت مطابقا يتم ارسال رسالة تحكم من وحدة الصوت الي المتحكم الدقيق،البرنامج الموجود داخل المتحكم الدقيق يقوم باستقبال ومعالجة الرسالة وتشغيل او اغلاق المعدات المنزلية علي حسب الصوت الموجه، بمساعدة دائرة ربطتربط بين المتحكم والمعدات المنزلية.النتائج المتحصل عليها من النظام كانت بصورة جيدة جدا النظام له المقدرة علي ترشيح الضجيج وتذكر اوامر الصوت استخدمت برمجيات مختلفة لاغراض محاكاة النظام(BASCOM and PROTOUS).

List of Contents

List of Figures

List of tables

Abbreviations

TTS	Text-To-Speech
SR	Speech Recognition
VR	Voice Recognition
RF	Radio Frequency
CPU	Central Processing Unit
RAM	Random-Access Memory
ROM	Read-Only Memory
EEPROM	Electrically Erasable Programmable Read-Only Memory
I/O	Input/output
A/D	Analog-to-Digital
D/A	Digital-to-Analog
GND:	Ground
HMM	Hidden Markov Model
ASK	Amplitude Shift Keying
PCB	Printed Circuit Board
LCD	Liquid Crystal Dispay

Chapter One

Introduction

1.1. Preface

Over the past decade, technology has dramatically changed our life and living styles. The internet has made it possible for people to connect to the world without stepping out of the house and wireless communication has given people the convenience of keeping in touch with each other anytime anywhere. The design of new future house is advanced technologies. These new technologies offer homeowners a more comfortable home environment doing automation which refers to any process that gives remote or automatic control of home devices and appliances. The challenge to design better automation products is to accommodate the variation among different users[1]. Also, a better user interface design can be the solution to some exsting an automation design problems. The perfect user interface still does not exist at present and to build a good interface requires knowledge of both sociology and technology fields. According to major companies that are involved in speech recognition researches, voice will be the primary interface between humans and machines in the near future. Researchers have investigated the possibility of using voice activation in cars to enable hands free controlling. Recently, a Hidden Markov Model (HMM) based speech recognition system was implemented in to enable voice activated wheelchair controlling. Speech recognition technology allows computers to translate speech in pure audio or spoken form and convert it to text.

By providing a specific grammar and limiting the vocabulary, the system needs to recognize the speech with good recognition results.

The performance of the speech recognition in home environments depends on the implementation of the speech recognition system interfaced with the smart chip called microcontroller with proper programming.

The main contribution of this thesis is implementation and evaluation of voice activated system using microcontroller for assessing the feasibility of using voice as the unified control method in real wireless home environments [2].

Especially audio manipulation and sound processing are commonly used in many everyday electronic devices. Some examples are MP3 players, digital voice recorders, speaking GPS receivers, electronic toys, intelligent alarm systems, speaking clocks, radio and televisions, mobile phones, devices used by older and blind people, and many more similar devices.

In general, the speech processing capabilities that can be added to an electronic device are voice recording, voice playback, text-to-speech (TTS) synthesis and speech recognition (SR). Voice recording and voice playback are used in digital voice recorders to store speech in non-volatile memory and then replay it at a later time. Some intelligent recording systems have additional features such as searching for a particular speech, skipping speeches, organizing recorded speeches in folders, and so on.

TTS involves reading a written text and converting it into spoken words that can be played through speakers. One typical application of TTS is that the computer can read a piece of text (e.g. via a keyboard or scanner), thus saving the user to look at the screen all the time. People with reading or visual difficulties (e.g. dyslexic or blind people) may find such systems extremely useful.

One of the recent applications of speech processing is in the field of speech recognition (sometimes called voice recognition, or VR), which basically gives a product the ability to "listen and understand". A speech recognition system takes a user's spoken words and interprets what has been said. Speech recognition systems are nowadays commonly used in mobile phones where the numbers to be dialled are read out by the user and the phone rings the required number without the touch of a button.

Integration of speech recognition to a product makes that product more intelligent and also more marketable [15].

1.2 Problem statement:

Disabled people, who are not able to do various activities efficiently when they are at home and need one's assistant to perform deferent tasks.The complication of wiring control system and the needs to distance control for different application is the main problem addressed in this thesis. Voice activated system in general face a noise problem whichaffect the performance of the system.

1.3. Proposed solution:

The proposed solution is by design of voice activated wireless system using microcontroller with the ability to reduce the background noise.

1.4. Objective:

The main objective is to design of voice activated system using microcontroller. . To achieve this objective:

1- A system that can respond to voice commands and control the on/off status of electrical devices had been proposed.

2- A wireless communication link of home appliances had been designed.

3- Simulation of the proposed system hadbeen run.

4-practical implementation of the proposed system will be done.

The proposed system should satisfy the following points:-

1- Reduce the noise in background voice.

2- Reliability in operation.

1.5. Scope of study:

The system covers controlling of home appliances. In this project, one voice recognition module had been added to the wireless network along with a noise reduction technique.

1.6Methodology*:*

Design of voice activated system using microcontroller is proposed. The automation centers at recognition of voice commands anduses the Radio Frequency (RF) technology. The voice command is a person independent. The system comprisesof transmitting section and receiving section. Initially, the voice command is stored in the data base with thehelp of the function keys. Then the input voice commands are transmitted through wireless. The voice receivedis processed in the voice recognition system where the feature of the voice command is extracted and matchedwith the existing sample in the database.

1.7Thesis organization

This thesis consists of five chapters. Chapter one discuss an overview of

Project, objective research and aims, project scope, problem statement, proposedsolution and thesis organization.Chapter tow contains literature review and system components.Chapter three includes System design.Chapter fourabout simulation and result. AndChapter fiveincludes conclusion and recommendation.

ChapterTwo

Literature Review

2.1. Previous Works:

Several works have been done in the area of voice activated systems using microcontroller, among them; these are the most recent researches:

Design of Wireless Home automation and security system using PIC Microcontroller, 2013.

A home automation system based on voice recognition which uses PIC microcontroller as CPU was explained in this paper. The system is focused on at elderly people and differently abled people. The prototype developed can control electrical devices in a home or office. The system implements voice recognition unit using HM 2007. The system implements the wireless network using ZigBee RF modules for their efficiency and low power consumption. The security system will be useful in case of the fire accidents at the home. However the system suffers from noise effect.

Voice Recognition Wireless Home Automation System Based On Zigbee,2013

Voice recognition Wireless Home Automation Based on ZigBee is a very useful project for the adults and physically disabled persons, who are not able to do various activities efficiently when they are at home and need one's assistant to perform those tasks. With the Voice Recognition along with ZigBee network we can eliminate the complication of wiring in case of wired automation and also it prevent to get up and down again and again to on/off appliances. ZigBee Home Automation provides operating range much higher as compared to Bluetooth and other wireless sensor module .With the use of

ZigBee Home Automation circuit considerable amount of power saving is possible and it is flexible and compatible with future technologies so it can be easily customized for individual requirements. On the other hand with voice recognition system, it provides secure access to home. So when we are living in a fast world where everything is changing with in no time such security is essential.[4]

Microcontroller based voice activated wireless automation system,2012 .

In this work the response with different values of the input frequency shows a good and accurate rise time, fall time, duty cycle and pulse width. The system has practical coverage up to a few meters. Confirmative voice with specific voice pitch and frequency is desired by the speech recognizer used in this system to produce better recognition results. The system controls extended and multiple home appliances by using speech recognition technology. However the system severs from noise effect[3].

2.2system components:

Different parts are used to design voice activated system using microcontroller:

2.2.1 Microcontroller:

A microcontroller is a single chip, self-contained computer which incorporates all the basic components of a personal computer on a much smaller scale. Microcontrollers are often referred to as single chip devices or single chip computers. The main consequence of the microcontroller's small size is that its resources are far more limited than those of a desktop personal computer.

In functional terms, a microcontroller is a programmable single chip which controls a process or system. Microcontrollers are typically used as embedded controllers where they control part of a larger system such as an appliance, automobile, scientific instrument or a computer peripheral. Microcontrollers are designed to be low cost solutions; therefore using them can drastically reduce part and design costs for a project.

Physically, a microcontroller is an integrated circuit with pins along each side. The pins presented by a microcontroller are used for power, ground, oscillator, I/O ports, interrupt request signals, reset and control. In contrast, the pins exposed by a microprocessor are most often memory bus signals (rather than I/O ports)[17].

The microcontroller consists a central processing unit (CPU), random-access memory (RAM), read-only memory (ROM), electrically erasable programmable read-only memory (EEPROM), input/output (I/O) lines, serial and parallel parts, timers, and other built-in peripherals, such as analog-to-digital (A/D) and digital-to-analog (D/A) converters.

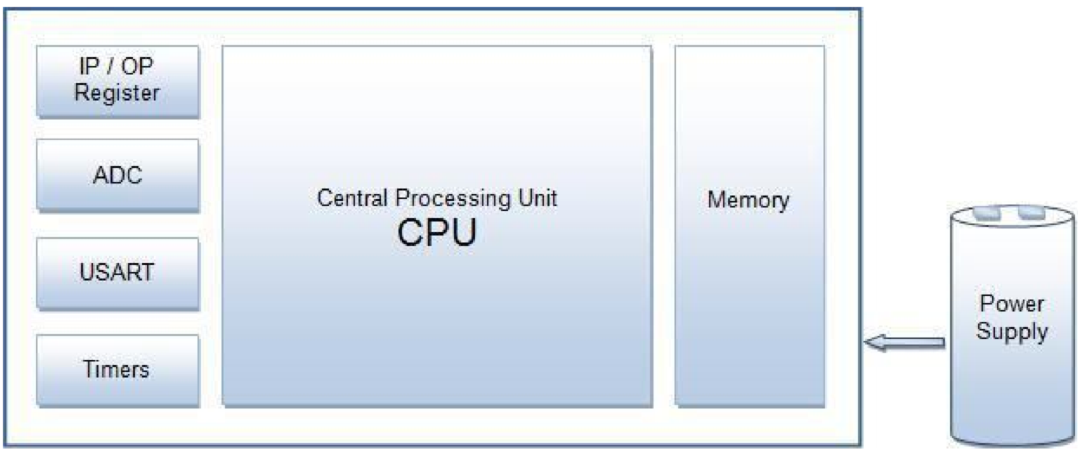


Figure2.1: microcontroller components

The Microcontroller has many Features such as:

1. Power on reset: Under most operating conditions, the microcontroller will go to a known RESET condition without the use of external circuitry. However,

for ultra- reliable reset operation, a capacitor can be connected to the MCLR (reset) pin.

2. Watchdog timer: A technique which, when used, allows the processor to escape from an endless loop fault condition if the watchdog timer is not regularly reset.

3. Oscillator start-up timer: Internal circuitry holds the microcontroller RESET forapproximately 18 ms after the master clear released.

4. Security EPROM fuses for code protection: When set, the security fuse prevents the internal program memory (the user program) from being read by external devices.

5. Power saving SLEEP mode: A software command that shuts theprocessor down until once again it is RESET.

6. EPROM selectable oscillator options: The microcontroller needs a frequency reference from which to operate; this can be chosen from RC components, quartz crystals or ceramic resonators. The oscillator section is hardware optimized to operate from 1 of 4 reference types.

There are number of popular families of microcontrollers which are used in different applications as per their capability and feasibility to perform the desired task, most common of these are 8051, AVR and PIC microcontrollers. The Microcontroller we will use is ATMega16 from AVR Family.

Atmega16 AT - It stands for the company that produced the microcontroller, ATMEL. Mega - Stands for the family of Microcontroller. The other families from Atmel are Tiny & X-Mega.16 - Stands for the 16KB flash memory that is present in the microcontroller.

The ATmega16 is a low-power CMOS 8-bit microcontroller based on the AVR enhanced RISC architecture. By executing powerful instructions in a single clock cycle, the ATmega16 achieves throughputs approaching 1 MIPS per MHz allowing the system designed to optimize power consumption versus processing speed.

The figure below show the Pin Configuration of the Atmega16:

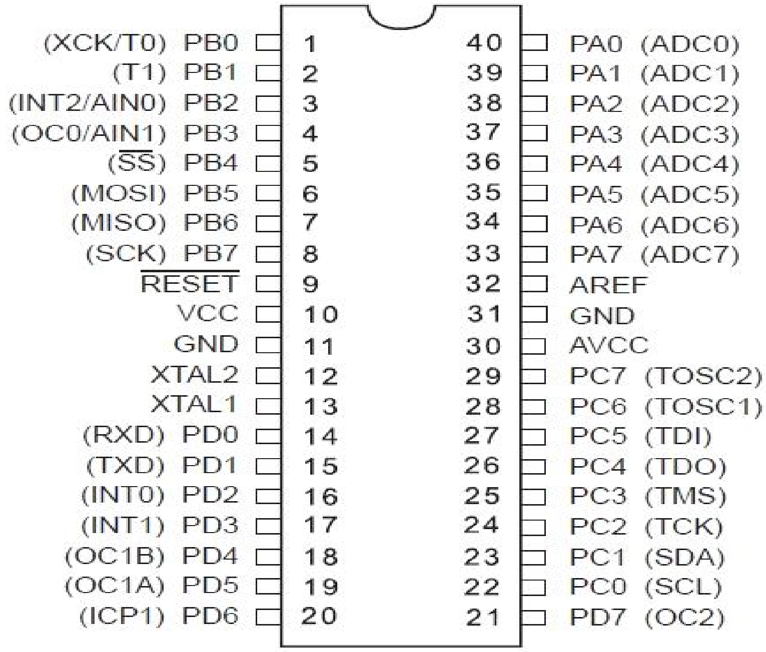


Figure 2.2: Atmega16 Microcontroller Pin Configuration.

Atmega16 has 40 pins , All pins are I/O pins ,i.e, all pins are capable of taking input as well as giving output. Port A has special use as it can be used as Analog to Digital Converter (ADC). ADC works by taking a reference value in between 0-5V. Any value above the reference is considered 1 (High) and any value below it is considered 0 (Low).

VCC: Digital supply voltage. (+5V)

GND: Ground. (0 V) Note there are 2 ground Pins.

Port A (PA7 - PA0) :Port A serves as the analog inputs to the A/D Converter. Port A also serves as an 8-bit bi-directional I/O port, if the A/D Converter is not used. When pins PA0 to PA7 are used as inputs and are externally pulled low, they will source current if the internal pull-up resistors are activated. The Port A pins are tri-stated when a reset condition becomes active, even if the clock is not running.

Port B (PB7 - PB0): Port B is an 8-bit bi-directional I/O port with internal pull-up resistors (selected for each bit). Port B also serves the functions of various special features of the ATmega16 as listed on page 58 of datasheet.

Port C (PC7 - PC0) :Port C is an 8-bit bi-directional I/O port with internal pull-up resistors (selected for each bit). Port C also serves the functions of the JTAG interface and other special features of the ATmega16 as listed on page 61 of datasheet. If the JTAG interface is enabled, the pull-up resistors on pins PC5 (TDI), PC3 (TMS) and PC2 (TCK) will be activated even if a reset occurs.

Port D (PD7 - PD0): Port D is an 8-bit bi-directional I/O port with internal pull-up resistors (selected for each bit). Port D also serves the functions of various special features of the ATmega16 as listed on page 63 of datasheet.

RESET: Reset Input. A low level on this pin for longer than the minimum pulse length will generate a reset, even if the clock is not running.

XTAL1: External oscillator pin 1.

XTAL2: External oscillator pin 2.

AVCC: AVCC is the supply voltage pin for Port A and the A/D Converter. It should be externally connected to VCC, even if the ADC is not used. If the ADC is used, it should be connected to VCC through a low-pass filter.

AREF: AREF is the analog reference pin for the A/D Converter.

ADC is available at PORTA of Atmega16. Thus we have 8 pins available where we can apply analog voltage and get corresponding digital values. The ADC register is a 10 bit register, i.e., the digital value ranges from 0 to 1023.

But we can also use only 8 bit out of it (0 to 255) as too much precision is not required.

Reference voltage is the voltage to which the ADC assigns the maximum value (255 in case of 8 bit and 1023 for 10 bit). Hence, the ADC of Atmega16 divides the input analog voltage range (0V to Reference Voltage) into 1024 or 256 equal parts, depending upon whether 10 bit or 8 bit ADC is used. For example, if the reference voltage is 5V and we use 10bit ADC, 0V has digital equivalent 0, +5V is digitally 1023 and 2.5V is approximately equal to 512.

ADC =Vin x 255/Vref (8 bit)

ADC =Vin x 1023/Vref (10 bit)

The figure show the ADC pins in atmega16 ,Thus we have 8 pins available where we can apply analog voltage and get corresponding digital values. The ADC register is a 10 bit register, Reference voltage(AREF) is the voltage to which the ADC assigns the maximum value (255 in case of 8 bit and 1023 for 10 bit). Hence, the ADC of Atmega16 divides the input analog voltage range (0V to Reference Voltage) into 1024 or 256 equal parts, depending upon whether 10 bit or 8 bit ADC is used. We have two option to connect AREF in ATMEGA16 ,the first option supply our own reference voltage across AREF and GND. In this case, you can either connect a capacitor across AREF pin and ground it to prevent from noise, or you may choose to leave it unconnected . If you want to use the VCC (+5V), the second option is using internal Vref. when use Vcc = 5V.

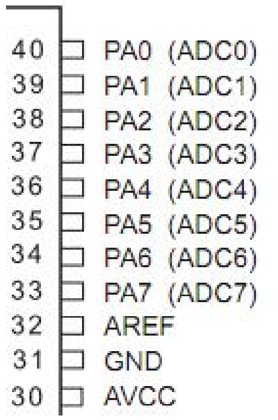


Figure2.3:ADC Pins – ATMEGA16

The AVR atmega16 supports pulse width modulation (PWM) on all three timer counters. Initially we will use the 8 bit timer 0 to implement this function. The AVR supports normal PWM or so called fast PWM. Normal PWM involves starting a counter which counts up to it's maximum value and then reverses, counts back to zero and then repeats.. In order to create output pulses whose mark: space ratio changes the output compare register (Ref) is loaded with a value so that when the count reaches that value the Output is reversed .the value of Ref reduces so the width of the pulses increase and as Ref increase then the width of the pulses will decrease. In the atmega16 Ref = OCR0 and the output is OC0 which is bit 3 on PORTB[16].

2.2.2 Wireless Communications

Wireless communications is, by any measure, the fastest growing segment of the communications industry. As such, it has captured the attention of the media and the imagination of the public. Cellular systems have experienced exponential growth over the last decade and there are currently around two billion users worldwide. Indeed, cellular phones have become a critical business tool and part of everyday life in most developed countries, and are rapidly supplanting antiquated wire line systems in many developing countries.

In addition, wireless local area networks currently supplement or replace wired networks in many homes, businesses, and campuses. Many new applications, including wireless sensor networks, automated highways and factories, smart homes and appliances, and remote telemedicine, are emerging from research ideas to concrete systems. The explosive growth of wireless systems coupled with the proliferation of laptop and palmtop computers indicate a bright future for wireless networks, both as stand-alone systems and as part of the larger networking infrastructure. However, many technical challenges remain in designing robust wireless networks that deliver the performance necessary to support emerging applications. In this introductory chapter we will briefly review the history of wireless networks, fromthe smoke signals of the pre-industrial age to the cellular, satellite, and other wireless networks of today. We then discuss the wireless vision in more detail, including the technical challenges that must be overcome to make this vision a reality. We describe current wireless systems along with emerging systems and standards. The gap between current and emerging systems and the vision for future wireless applications indicates that much work remains to be done to make this vision a reality.

2.2.2.1 Technical Issues of wireless communication

Many technical challenges must be addressed to enable the wireless applications of the future. These challenges extend across all aspects of the system design. As wireless terminals add more features, these small devices must incorporate multiple modes of operation to support the different applications and media. Computers process voice, image, text, and video data, but breakthroughs in circuit design are required to implement the same multimode operation in a cheap, lightweight, handheld device. Since consumers don't want large batteries that frequently need recharging, transmission and signal processing in the portable terminal must consume minimal power. The signal processing required to support multimedia applications and networking functions can be power-intensive. Thus, wireless infrastructure-based networks, such as wireless LANs and cellular systems, place as much of the processing burden as possible on fixed sites with large power resources. The associated bottlenecks and single points-of-failure are clearly undesirable for the overall system. Ad hoc wireless networks without infrastructure are highly appealing for many applications due to their flexibility and robustness. For these networks all processing and control must be performed by the network nodes in a distributed fashion, making energy-efficiency challenging to achieve. Energy is a particularly critical resource in networks where nodes cannot recharge their batteries, for example in sensing applications. Network design to meet the application requirements under such hard energy constraints remains a big technological hurdle. The finite bandwidth and random variations of wireless channels also requires robust applications that degrade gracefully as network performance degrades. Design of wireless networks differs fundamentally from wired network design due to the nature of the wireless channel. This channel is an unpredictable and difficult communications medium. First of all, the radio spectrum is a scarce resource that must be allocated to many different applications and systems. For this reason spectrum is controlled by regulatory bodies both regionally and globally. A regional or global system operating in a given frequency band must obey the restrictions for that band set forth by the corresponding regulatory body. Spectrum can also be very expensive since in many countries spectral licenses are often auctioned to the highest bidder.

In the U.S. companies spent over nine billion dollars for second generation cellular licenses, and the auctions in Europe for third generation cellular spectrum garnered around 100 billion dollars. The spectrum obtained through these auctions must be used extremely efficiently to get a reasonable return on its investment, and it must also be reused over and over in the same geographical area, thus requiring cellular system designs with high capacity and good performance. At frequencies around several Gigahertz wireless radio components with reasonable size, power consumption, and cost are available. However, the spectrum in this frequency range is extremely crowded.

5 Thus, technological breakthroughs to enable higher frequency systems with the same cost and performance would greatly reduce the spectrum shortage. However, path loss at these higher frequencies is larger, thereby limiting range, unless directional antennas are used.

As a signal propagates through a wireless channel, it experiences random fluctuations in time if the transmitter, receiver, or surrounding objects are moving, due to changing reflections and attenuation. Thus, the characteristics of the channel appear to change randomly with time, which makes it difficult to design reliable systems with guaranteed performance. Security is also more difficult to implement in wireless systems, since the airwaves are susceptible to snooping from anyone with an RF antenna. The analog cellular systems have no security, and one can easily listen in on conversations by scanning the analog cellular frequency band. All digital cellular systems implement some level of encryption. However, with enough knowledge, time and determination most of these encryption methods can be cracked and, indeed, several have been compromised. To support applications like electronic commerce and credit card transactions, the wireless network must be secure against such listeners.

Wireless networking is also a significant challenge. The network must be able to locate a given user wherever it is among billions of globally-distributed mobile terminals. It must then route a call to that user as it moves at speeds of up to 100 Km/hr. The finite resources of the network must be allocated in a fair and efficient manner relative to changing user demands and locations. Moreover, there currently exists a tremendous infrastructure of wired networks: the telephone system, the Internet, and fiber optic cable, which should be used to connect wireless systems together into a global network. However, wireless systems with mobile users will never be able to compete with wired systems in terms of data rates and reliability. Interfacing between wireless and wired networks with vastly different performance capabilities is a difficult problem.

Perhaps the most significant technical challenge in wireless network design is an overhaul of the design process itself. Wired networks are mostly designed according to a layered approach, whereby protocols associated with different layers of the system operation are designed in isolation, with baseline mechanisms to interface between layers. The layers in a wireless systems include the link or physical layer, which handles bit transmissions over the communications medium, the access layer, which handles shared access to the communications medium, the network and transport layers, which routes data across the network and insure end-to-end connectivity and data delivery, and the application layer, which dictates the end-to-end data rates and delay constraints associated with the application. While a layering methodology reduces complexity and facilitates modularity and standardization, it also leads to inefficiency and performance loss due to the lack of a global design optimization. The large capacity and good reliability of wired networks make these inefficiencies relatively benign for many wired network applications, although it does preclude good performance of delay-constrained applications such as voice and video. The situation is very different in a wireless network. Wireless links can exhibit very poor performance, and this performance along with user connectivity and network topology changes over time. In fact, the very notion of a wireless link is somewhat fuzzy due to the nature of radio propagation and broadcasting. The dynamic nature and poor performance of the underlying wireless communication channel indicates that high-performance networks must be optimized for this channel and must be robust and adaptive to its variations, as well as to network dynamics. Thus, these networks require integrated and adaptive protocols at all layers, from the link layer to the application layer. This cross-layer protocol design requires interdisciplinary expertise in communications, signal processing, and network theory and design.

In the next section we give an overview of the wireless systems in operation today. It will be clear from this overview that the wireless vision remains a distant goal, with many technical challenges to overcome [6].

2.2.3 Digital Modulation in Communications Systems

This application note introduces the concepts of digital modulation used in many communicationssystems today. Emphasis is placed on explaining the tradeoffs that are made to optimize efficiencies in system design.

Most communications systems fall into one of three categories: bandwidth efficient, power efficient, or cost efficient. Bandwidth efficiency describes the ability of a modulation scheme to accommodate data within a limited bandwidth. Power efficiency describes the ability of the system to reliably send information at the lowest practical power level.

In most systems, there is a high priority on bandwidth efficiency. The parameter to be optimized depends on the demands of the particular system, as can be seen in the following two examples.

For designers of digital terrestrial microwave radios, their highest priority is good bandwidth efficiency with low bit-error-rate. They have plenty of power available and are not concerned with power efficiency. They are not especially concerned with receiver cost or complexity because they do not have to build large numbers of them.

2.2.3.1Digital Modulation Types and Relative Efficiencies

This section covers the main digital modulation formats, their main applications, relative spectral efficiencies, and some variations of the main modulation types as used in practical systems.

Fortunately, there are a limited number of modulation types which form the building blocks of any system.

2.2.3.2 Applications of modulation

The table below covers the applications for different modulation formats in both wireless communications and video.

Although this note focuses on wireless communications, video applications have also been included in the table for completeness and because of their similarity to other wireless communications [8].

Table 2.1 Application of Modulation Techniques

Modulation format	Application
MSK, GMSK	GSM, CDPD
BPSK	Deep space telemetry, cable modems
QPSK, π/4 DQPSK	Satellite, CDMA, NADC, TETRA, PHS, PDC, LMDS, DVB-S, cable (return path), cable modems, TFTS
OQPSK	CDMA, satellite
FSK, GFSK	DECT, paging, RAM mobile data, AMPS, CT2, ERMES, land mobile, public safety
8PSK	Satellite, aircraft, telemetry pilots for monitoring broadband video systems
16 QAM	Microwave digital radio, modems, DVB-C, DVB-T
32 QAM	Terrestrial microwave, DVB-T
64 QAM	DVB-C, modems, broadband set top boxes, MMDS
256 QAM	Modems, DVB-C (Europe), Digital Video (US)

2.2.4 Speech Enhancement in Noise

De-noising speech improves the quality and the intelligibility of voice communication in noisy environments and reduces communication fatigue. Noise reduction benefits the users of hands-free phones, mobile phones and voice-controlled automated services used in noisy moving environments such as cars, trains, streets, conference halls and other public venues.

The main signal processing methods for enhancement of noisy speech into two broad types:

(1) Single-input speech enhancement systems, where the only available signal is the noise contaminated speech picked up by a single microphone. Single input systems do not cancelnoise; rather they suppressthe noise using estimates of the signal-to-noise ratio of the frequency spectrum of the input signal. Single-input systems rely on the statistical models of speech and noise, which may be estimated from the speech-inactive periods or decoded from a set of pre-trained models of speech and noise. An example of a useful application of a single-input enhancement system is a mobile phone system used in noisy environments.

(2) Multiple-input speech enhancement systems, where a number of signals containing speech and noise are picked up by several microphones. Examples of multiple inputs systems are adaptive noise cancellation, adaptive beam-forming microphone arrays and multiple-input multiple-output (MIMO) acoustic echo cancellation systems. In multiple-input systems the microphones can be designed, spatially arranged and adapted for optimum performance. Multiple-input noise-reduction systems are useful for teleconferencing systems and for in-car cabin communication systems [7].

2.2.4.1Filters:

Filters are networks that process signals in a frequency-dependent manner. The basic concept of a filter can be explained by examining the frequency dependent nature of the impedance of capacitors and inductors. Consider a voltage divider where the shunt leg is reactive impedance. As thefrequency is changed, the value of the reactive impedance changes, and the voltage divider ratio changes. This mechanism yields the frequency dependent change in the input/output transfer function that is defined as the frequency response.

Filters have many practical applications. A simple, single-pole, low-pass filter (theIntegrator) is often used to stabilize amplifiers by rolling off the gain at higher frequencies where excessive phase shift may cause oscillations.A simple, single-pole, high-pass filter can be used to block dc offset in high gainamplifiers or single supply circuits. Filters can be used to separate signals, passing those of interest, and attenuating the unwanted frequencies.

An example of this is a radio receiver, where the signal you wish to process is passed through, typically with gain, while attenuating the rest of the signals. In data conversion, filters are also used to eliminate the effects of aliases in A/D systems. They are used in reconstruction of the signal at the output of a D/A as well, eliminating the higher frequency components, such as the sampling frequency and its harmonics, thus smoothing the waveform.

There are a large number of texts dedicated to filter theory. No attempt will be made to go heavily into much of the underlying math: Laplace transforms, complex conjugate poles and the like, although they will be mentioned.

While they are appropriate for describing the effects of filters and examining stability, in most cases examination of the function in the frequency domain is more illuminating. An ideal filter will have an amplitude response that is unity (or at a fixed gain) for the frequencies of interest (called the pass band) and zero everywhere else (called the stop band). The frequency at which the response changes from pass band to stop band is referred to as the cutoff frequency.

Figure 2.4(A) shows an idealized low-pass filter. In this filter the low frequencies are in the pass band and the higher frequencies are in the stop band.

2.2.4.2Analog vs. Digital Filters

Most digital signals originate in analog electronics. If the signal needs to be filtered, is it better to use an analog filter before digitization, or a digital filter after.

The goal will be to provide a low-pass filter at 1 kHz. Fighting for the analog side is a six poleChebyshev filter with 0.5 dB (6%) ripple. this can be constructed with 3 op amps, 12 resistors, and 6 capacitors. In the digital corner, the windowed-sinc is warming up and ready to fight. The analog signal is digitized at a 10 kHz sampling rate, making the cutoff frequency 0.1 on the digital frequency scale. The length of the windowed-sincwill be chosen to be 129 points, providing the same 90% to 10% roll-off as the analog filter. Fair is fair. Figure 2.4 shows the frequency and step responses for these two filters.

Let's compare the two filters blow-by-blow. As shown in (a) and (b), the analog filter has a 6% ripple in the passband, while the digital filter is perfectly flat (within 0.02%). The analog designer might argue that the ripple can be selectedin the design; however, this misses the point. The flatness achievable with analog filters is limited by the accuracy of their resistors and capacitors. Even if a Butterworth response is designed (i.e., 0% ripple), filters of this complexity will have a residue ripple of, perhaps, 1%. On the other hand, the flatness of digital filters is primarily limited by round-off error, making them hundreds of times flatter than their analog counterparts. Score one point for the digital filter.

Next, look at the frequency response on a log scale, as shown in (c) and (d). Again, the digital filter is clearly the victor in both roll-off and stopband attenuation. Even if the analog performance is improved by adding additional stages, it still can't compare to the digital filter. For instance, imagine that you need to improve these two parameters by a factor of 100. This can be done with simple modifications to the windowed-sinc, but is virtually impossible for the analog circuit. Score two more for the digital filter.

The step response of the two filters is shown in (e) and (f). The digital filter's step response is symmetrical between the lower and upper portions of the step, i.e., it has a linear phase. The analog filter's step response is *not* symmetrical, i.e., it has a nonlinear phase. One more point for the digital filter. Lastly, the analog filter overshoots about 20% on one side of the step. The digital filter overshoots about 10%, but on both sides of the step. Since both are bad, no points are awarded.

In spite of this beating, there are still many applications where analog filters should, or must, be used. This is not related to the actual performance of the filter (i.e., what goes in and what comes out), but to the general advantages that analog circuits have over digital techniques. The first advantage is speed: digital is slow; analog is fast. For example, a personal computer can only filter data at about 10,000 samples per second, using FFT convolution. Even simple op amps can operate at 100 kHz to 1 MHz, 10 to 100 times as fast as the digital system!

The second inherent advantage of analog over digital is dynamic range. This comes in two flavors. Amplitude dynamic range is the ratio between the largest signal that can be passed through a system, and the inherent noise of the system. For instance, a 12 bit ADC has a saturation level of 4095, and an rms quantization noise of 0.29 digital numbers, for a dynamic range of about 14000. In comparison, a standard op amp has a saturation voltage of about 20 volts and an internal noise of about 2 microvolts, for a dynamic range of about ten million. Just as before, a simple op amp devastates the digital system.

The other flavor is frequency dynamic range. For example, it is easy to design an op amp circuit to simultaneously handle frequencies between 0.01 Hz and 100 kHz (seven decades). When this is tried with a digital system, the computer becomes swamped with data. For instance, sampling at 200 kHz, it takes 20 million points to capture one complete cycle at 0.01 Hz. You may have noticed that the frequency response of digital filters is almost always plotted on a linear frequency scale, while analog filters are usually displayed with a logarithmic frequency. This is because digital filters need.

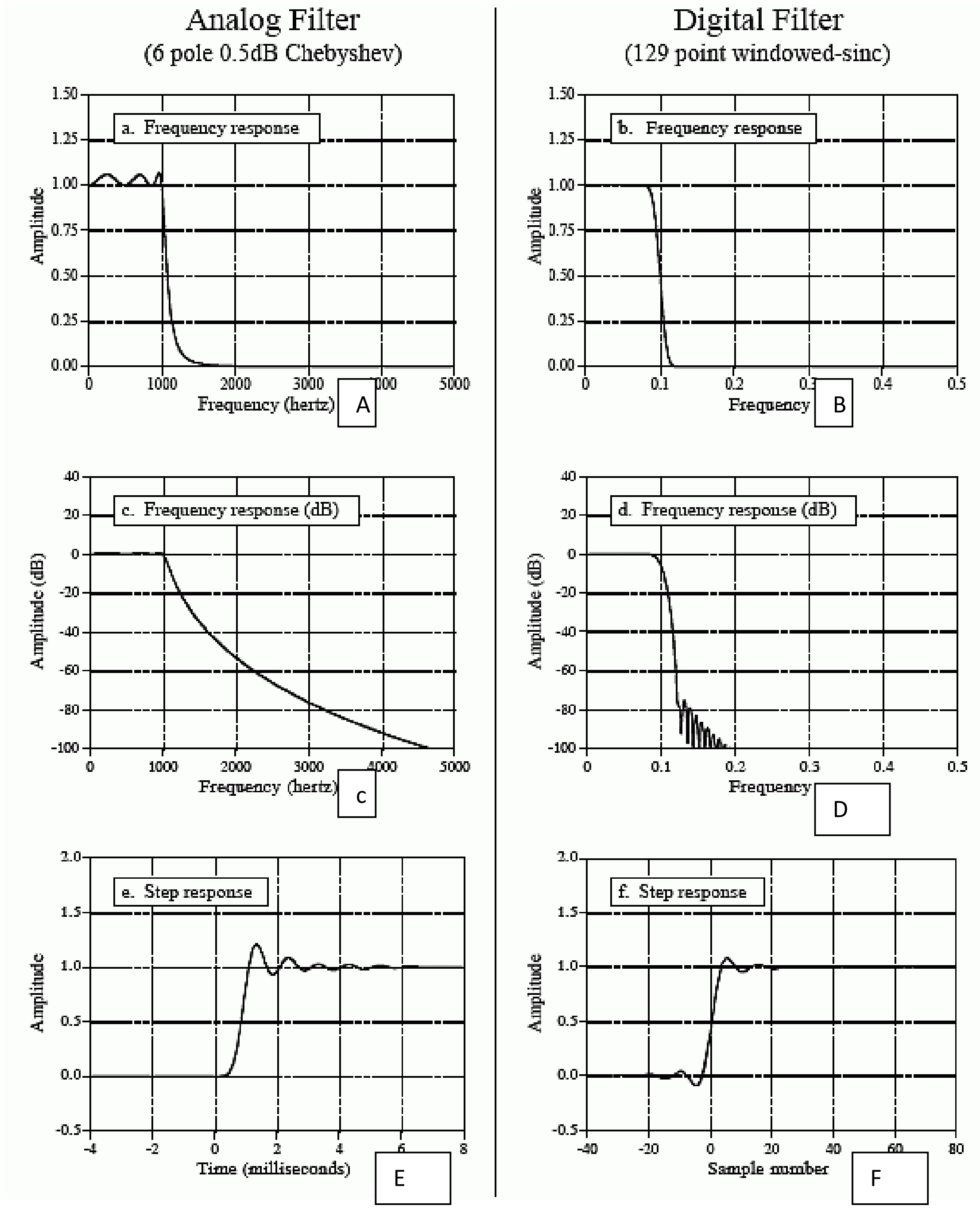


Figure 2.4 Analog Filters and Digital Filters Output

2.2.5 Speech Recognition

The speech is primary mode of communication among human being and also the most natural and efficient form of exchanging information among human in speech. So, it is only logical that the next technological development to be natural language speech recognition for HCI. Speech Recognition can be defined as the process of converting speech signal to a sequence of words by means Algorithm implemented as a computer program. Speech processing is one of the exciting areas of signal processing. The goal of speech recognition area is to developed technique and system to developed for speech input to machine. Based on major advanced in statically modeling of speech ,automatic speech recognition today find widespread application in task that require human machine interface such as automatic call processing.[8]. Since the 1960s computer scientists have been researching ways and means to make computers able to record interpret and understand human speech. Throughout the decades this has been a daunting task. Even the most rudimentary problem such as digitalizing (sampling) voice was a huge challenge in the early years. It took until the 1980s before the first systems arrived which could actually decipher speech. Off course these early systems were very limited in scope and power. Communication among the human being is dominated by spoken language, therefore it is natural for people to expect speech interfaces with computer .computer which can speak and recognize speech in native language [9]. MachineRecognition of speech involves generating a sequence of words best matches the given speech signal. [10]

2.2.5.1 Speech Recognition Techniques

The goal of speech recognition is for a machine to be able to "hear,” understand," and "act upon" spoken information. The earliest speech recognition systems were first attempted in the early 1950s at Bell

Laboratories, Davis, Biddulph and Balashek developed an isolated digit Recognition system for a single speaker [8]. The goal of automatic speaker Recognition is to analyze, extract characterize and recognize information about the speaker identity. The speaker recognisation system may be viewed as working in a four stages

1. Analysis
2. Feature extraction
3. Modeling
4. Testing

2.2.5.2 Speech Analysis Technique

Speech data contain different type of information that shows a speaker identity. This includes speaker specific information due to vocal tract, excitation source and behavior feature. The information about the behavior feature also embedded in signal and that can be used for speaker recognition. The speech analysis stage deals with stage with suitable frame size for segmenting speech signal for further analysis and extracting [11]. The speech analysis technique done with following three techniques

2.2.5.2.1 Segmentation analysis

In this case speech is analyzed using the frame size and shift in the range of 10-30 ms to extract speaker information. Studid made in used segmented analysis to extract vocal tract information of speaker recognition.

2.2.5.2.2 Sub segmental analysis

Speech analyzed using the frame size and shift in range 3-5 msis known as Sub segmental analysis. This technique is used to mainly analyze and extract the characteristic of the excitation state. [12].

2.5.2.3 Supra segmental analysis

In this case, speech is analyzed using the frame size this technique is technique is used mainly to analyze and characteristic due to behavior character of the speaker.

2.5.3 Features of HM2007

It is a single chip CMOS voice recognition LSI circuit with the on-chip analog front-end, voice analysis, recognition process and system control function.

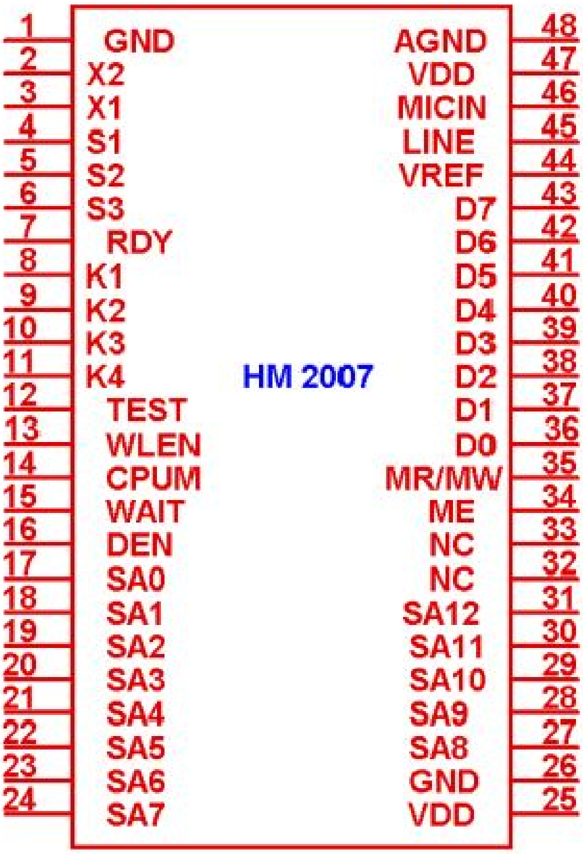


Figure 2.5 Functional Pin Description of HM 2007

A 20 isolated word voice recognition system can be composed of external microphone, keyboard, 8K SRAM and other components, combined with a microcontroller, an intelligent recognition system can be built. It support two control mode: Manual mode and CPU mode. It is also available in 48-pin PDIP.

The pin description of HM2007 is shown above. The keypad and digital display are used to communicate with and program the HM2007 chip. The keypad is made up of 12 normally open momentary contact switches. The 74LS373 8-bit registers feature 3-state outputs designed specifically for driving highly capacitive or relatively low-impedance loads. The high-impedance 3-state and increased high-logic-level drive provide these registers with the capability of being connected directly to and driving the bus lines in a bus-organized system without need for interface or pull-up components. The IC 7448 is BCD to 7–segment common cathode IC. To display the data, itconverts from BCD to 7-segment code. The IC makes this process. It has four inputs called BCD inputs and seven outputs to drive the display. The voice recognition system schematic diagram is shown below in fig.4. A microphone is connected directly with pin 15(MICIN) of HM2007which is shown below. On this system, voice is trained first and then recognized whenever a command is given through microphone.

Features

- User programmable
- Up to 20 word vocabulary of duration two second each
- Multi-lingual
- Non-volatile memory back up with 3V battery backup which will keep the speech recognition data in memory even after power off.
- Easily interfaced to any microcontroller like 8051, PIC or AVR can be interfacedto data port. [14]

Chapter Three

System Design

3.1 Overview

This system presents the proposal, design and implementation of voice-activated system using microcontroller. As speech is the preferred mode of operation for human being, this project intends to make the voice oriented command words for controlling home appliances. In this project, one voice recognition module has been added to the wireless network. The automation centers at recognition of voice commands and uses the amplitude shift keying (ASK) technology. The voice command is a person independent. The system comprises of transmitting section and receiving section. Initially, the voice command is stored in the data base with the help of the function keys. Then the input voice commands are transmitted through wireless. The voice received is processed in the voice recognition system where the feature of the voice command is extracted and matched with the existing sample in the database. The module recognizes the voice and sends control messages to the microcontroller. The programmed microcontroller then processes the received data and switches the respective appliances via connected driver circuits.

3.2 System Block Diagram

Block diagram of the system as shown in figure3.1 illustrate overall system process. Transmitter to transmit signals, receiver signals sends tohm 2007 through filters, chip hm 2007 to microcontroller .ADC inside microcontroller convert analog signals to digital ones, that understandable for microcontroller.

Controller takes a decision according to input signals and sends it to the home appliances. home appliances Execute the command from controller(on\off).

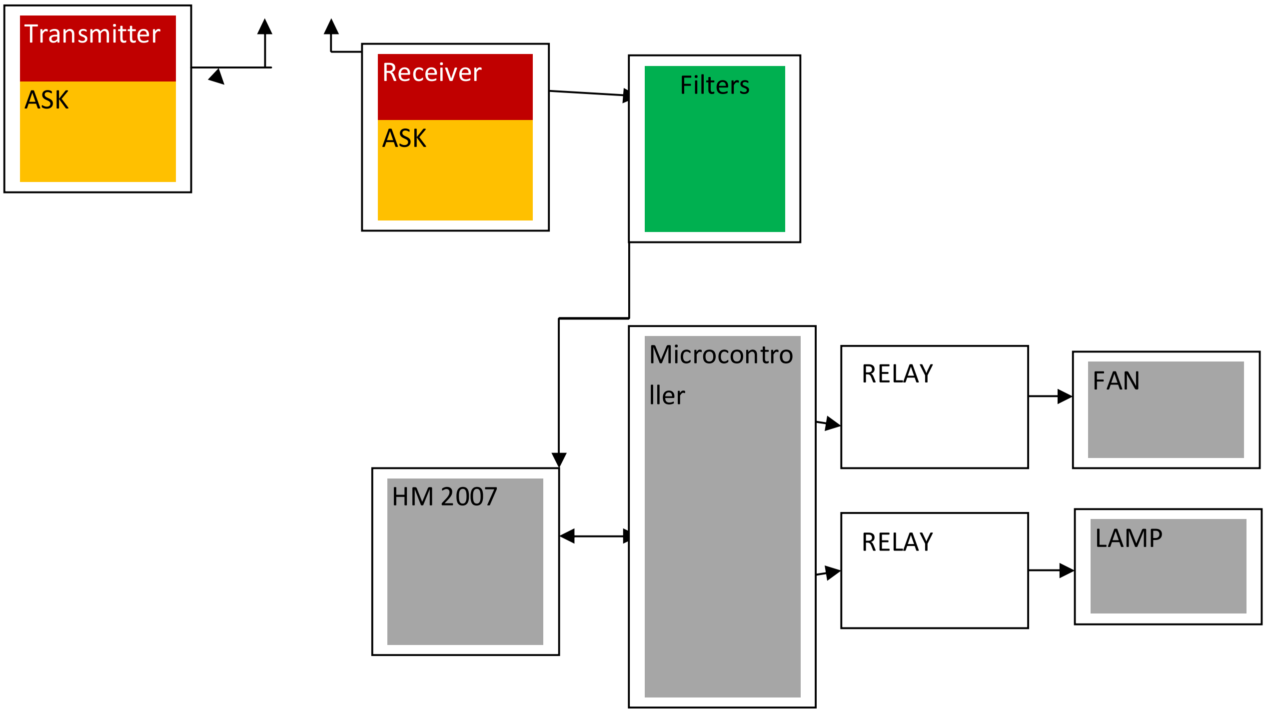


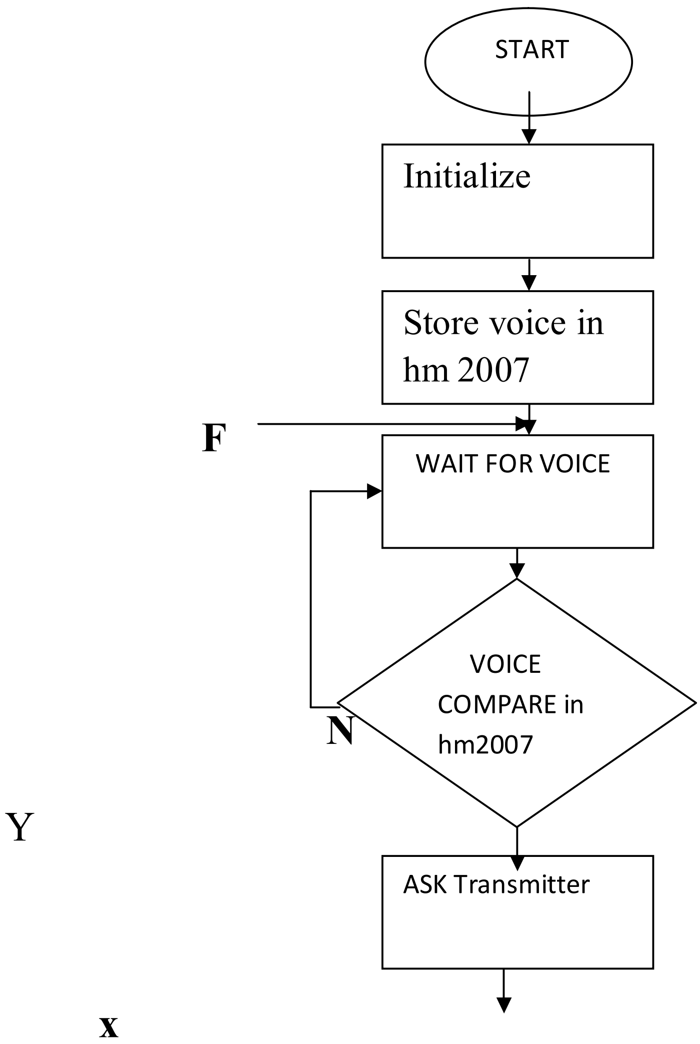
Figure3.1 Block Diagram of System

3.3System Description:

The overall structure of reduction noise in design voice activated system using microcontroller can be divided into two parts wirelesshand-held terminal and appliances control terminal parts which consists of various components forcommunicating with each other. The handheld device consists of the microcontroller unit which is connected to the ASKmodule for wireless communication l. The command to the microcontroller is given bythe microphonethat comes along the speech recognition kit for giving input to the controller. The memory chipis used to save the voice and provide the necessary signal after comparing with the recorded voice. Thisgenerated signal is sent to controller and sent out via ASK transmitter to control purpose. The control unitcontains the receiver part that receives the command from the ASK transmitter and the microcontroller is usedto control the homes appliances via relay unit.

3.3.1 Flow chart of System

The system start initialize it's port to read signals and store voice. The system will check the voice at the receive side to activate the mentioned application the flowing figure 3.2 shows the system flow chart.



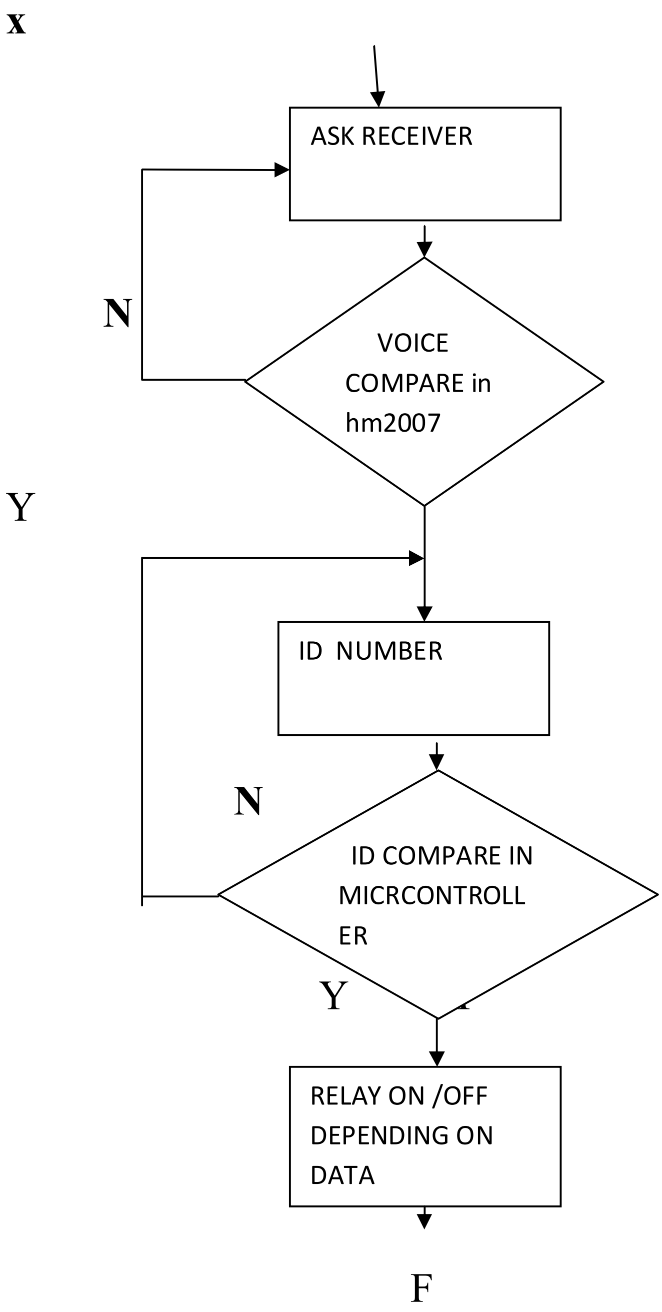


Figure 3.2system flow chart

3.3.2 Proposed Algorithm:

Step 1: Initialize all the module processes.

Step 2: Store sample voice commands required in theHM2007 speech processor memory using functionkeys.

Step3: Provide the input voice commandcorresponding to the appliance control which isalready stored in HM2007 Memory.

Step 4: When the input matches with sample voicestored, then send control signal to the transmittingcontroller. Else display error value on 7-segmentdisplay of HM2007 speech processor.

Step 5: Received control signal from speechprocessor is now modulated with ASK transmitterand send to receiving section ASK receiver.

Step 6: Received modulated signal from transmittingsection is now demodulated to original control signaland is fed to receiving controller.

Step 7: If the control signal ID is matched with thepre-existing appliance ID then the controller drivescorresponding appliance to switch ON/OFF byexecuting the code written.

Step8: Stop the process when no voice input receivedby speech processor.

3.4 Circuit Diagram:

The system is shown in figure 3.3. The system divided into two parts, transmitter side and receiver side.

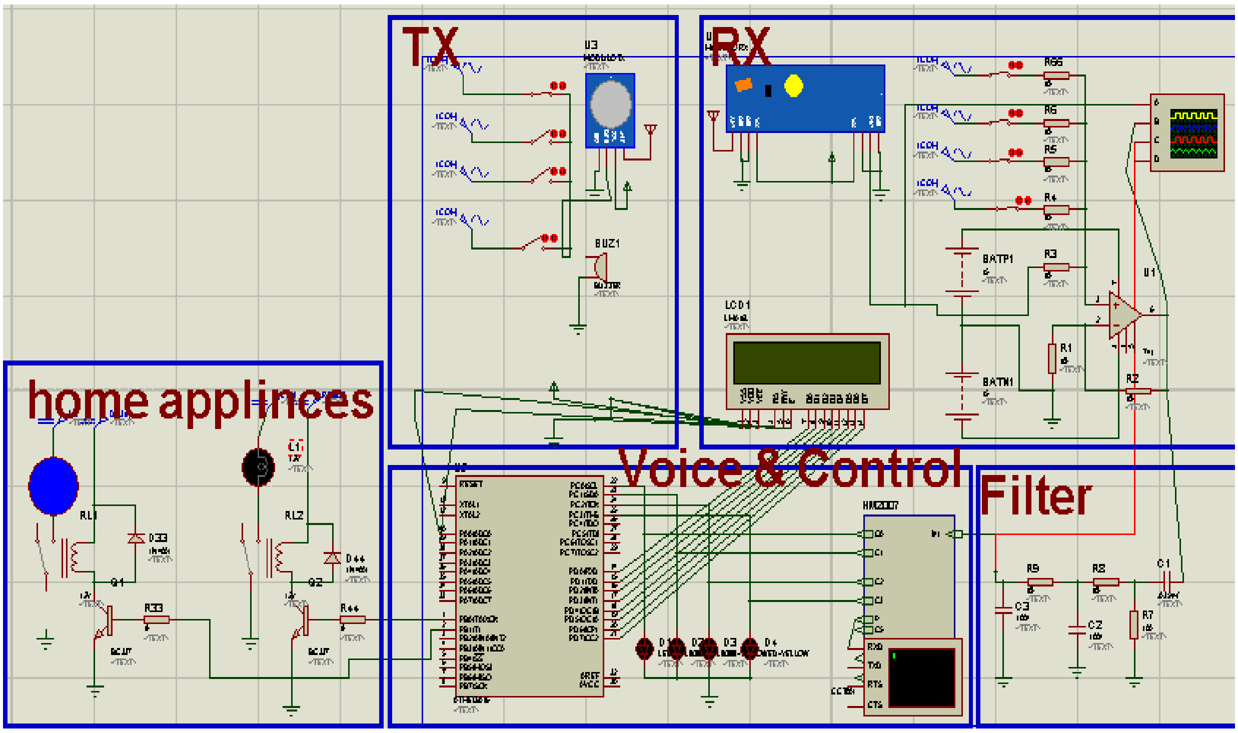


Figure3.3 Circuit Diagram

The stored voice was sampled with different frequencies as shown in table 3.1.each frequency represent different command.

Table 3.1 Equivalent value in Simulation

Frequency	Voice command
200 Hz	Lamp on
400 Hz	Lamp off
500 Hz	Fan on
600 Hz	Fan off

3.4.1 RF MODULE (transmitter/receiver

The RF module, as the name suggests, operates at Radio Frequency. The corresponding frequency range varies between 30 kHz & 300 GHz. In this RF system, the digital data is represented as variations in the amplitude of carrier wave. This kind of modulation is known as Amplitude Shift Keying (ASK).

The RF module included an RF Transmitter and an RF Receiver. The transmitter/receiver (Tx/Rx) pair operates at a frequency of 433 MHz and it has 4 output pins i.e. it can operate 4 peripherals remotely.

The RF module is often used along with a pair of encoder/decoder. The encoder is used for encoding parallel data for transmission feed while reception is decoded by a decoder as shown in figure 3.4 below.

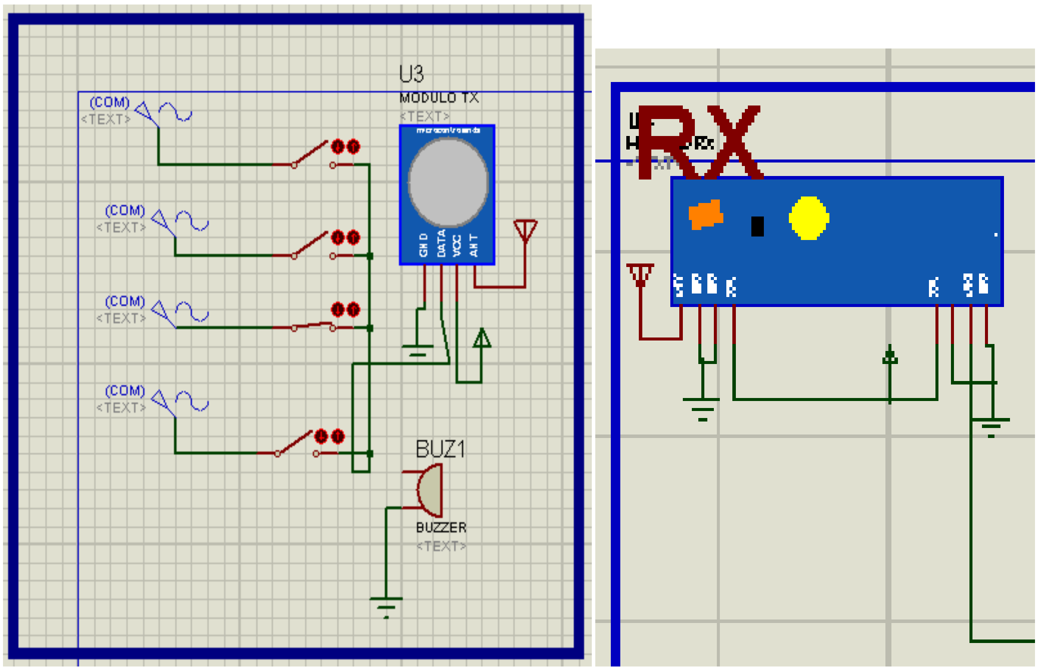


Figure 3.4 RF transmitter /receiver

For transmitter and receiver chip several pins are used as shown in table 3.2 and table 3.3

Table 3.2 RF Transmitter

Pin No	Function	Name
1	Ground (0V)	Ground
2	Serial data input pin	Data
3	Supply voltage; 5V	Vcc
4	Antenna output pin	ANT

Table 3.3 RF Receiver

Pin No	Function	Name
1	Ground (0V)	Ground
2	Serial data output pin	Data
3	Linear output pin; not connected	NC
4	Supply voltage; 5V	Vcc
5	Supply voltage; 5V	Vcc
6	Ground (0V)	Ground
7	Ground (0V)	Ground
8	Antenna input pin	ANT

3.4.2Filters:

Circuit is consist of three filters shown in figure 3.5, two of them low pass filters the third one high pass filter as the equation 3.1 bellow

Fc= 1/2πRC ……………………..3.1

Low pass filter R=10KΩ C=100 nf,fc=1.3KHz

High pass filter R=10KΩ C=0.039mf. fc=180 Hz

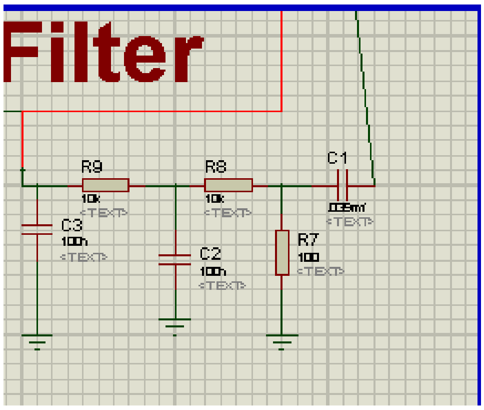


Figure 3.5 low pass filter and high pass filter

3.4.3 Speech Recognition using HM2007

The heart of the project is the HM2007 speechrecognition integrated circuit. The chip provides theoptions of recognizing either forty 0.96 second wordsvocabulary or twenty 1.92 second words vocabulary.For memory the circuit uses an 8K X 8 static RAM.The chip has two operational modes; manual modeand CPU mode. The CPU mode is designed to allowthe chip to work under a host computer. This is anattractive approach to speech recognition forcomputers because the speech recognition chipoperates as a co-processor to the main CPU. The jobof listening and recognition doesn’t occupy any ofthe computer’s CPU time. When the HM2007recognizes a command it can signal an interrupt tothe host CPU and then relay the command code.The

HM2007 chip can be cascaded to provide alarger word recognition library. Speech recognition isclassified into two categories, speaker dependent andspeaker independent. Speaker dependent systems aretrained by the individual who will be using thesystem. These systems are capable of achieving ahigh command count and better than 95% accuracyfor word recognition. The drawback to this approachis that the system only responds accurately only tothe individual who trained the system. This is themost common approach employed in software forpersonal computers. Speaker independent is a systemtrained to respond to a word regardless of whospeaks. Therefore the system must respond to a largevariety of speech patterns, inflections andenunciation’s of the target word. The command wordcount is usually lower than the speaker dependenthowever high accuracy can still be maintain withinprocessing limits. Industrial requirements more oftenneed speaker independent voice systems, such as theAT&T system used in the telephone systems.

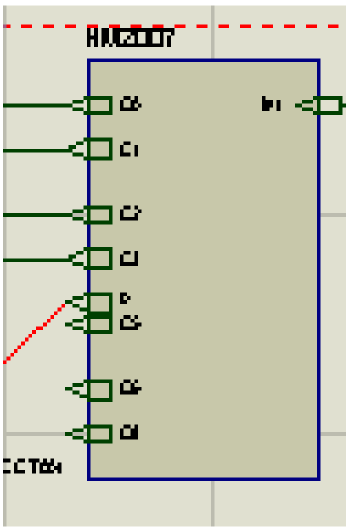


Figure 3.6 HM 2007

3.4.4 Microcontroller:

Processing system using id from hm2007, generating desired output for corresponding inputs, in which microcontrollers are used.ATmega16 microcontroller will be used in our system. It is known as AVR.

Pin configuration

The four input connected to PORTCin order to use D0 ,D1,D2 and D3 indicator binary inputs and PORTB output to control the home appliances as shown in figure 3.6 below:

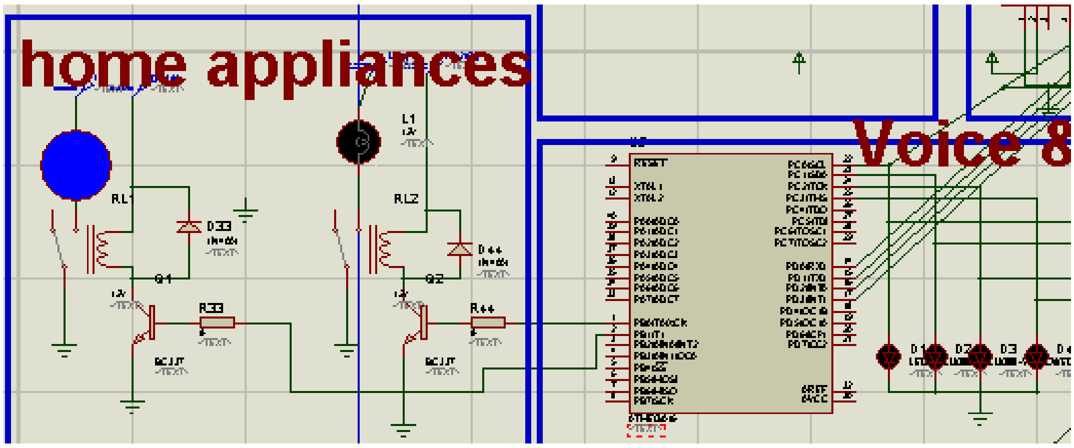


Figure 3.7 Microcontroller pins configuration

Chapter Four

Simulation and Results

4.1 Overview:

This chapter represents simulation results, considering different noise of multi environment, and the corresponding action. Results illustrated by screen shot for all scenarios, with explained process.

4.2 Simulation tools

To simulate the circuit of reduction noise in design voice activated system we use two simulation tools such as:

Proteus is software for microprocessor simulation, schematic capture, and printed circuit board (PCB) design. It is developed by Lab center Electronics.

Bascom means Basic Compiler. Bascom is developed and sold by MCS Electronics. it is a PC application that will allow to : write programs in Basic, translate these programs on the PC to machine code (a format the AVR controller can execute),simulate the compiled code and use external programs to flash ('program') the compiled code into an Atmel AVR microcontroller. Bascom enables quick prototyping because it has built-in support for almost all AVR microcontroller features such as: Counters/Timers, UART, ADC and PWM .

4.3 Simulation circuit

The figure 4.1 below show the simulation circuit of the reduction noise in design voice activated system using microcontroller which contains of three noise variables -array, atmega16, HM 2007, filter , LCD, home appliances, digital oscilloscope .The digital oscilloscope is using to display the signal before and after filter.

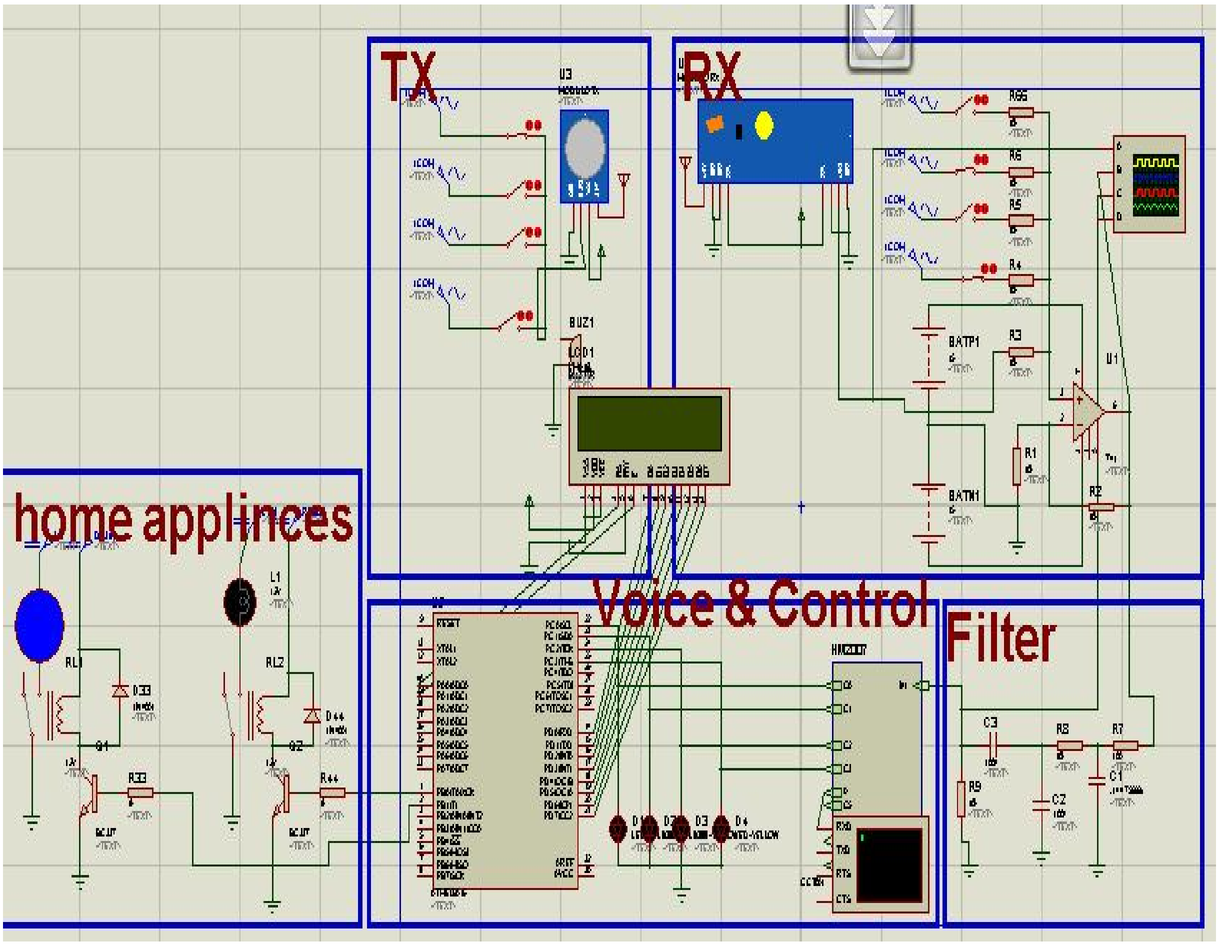


Figure 4.1 Circuit Simulation

4.41.7 kHz Noise signal

The simulation was run under the following condition. the audio signal frequency is 200 Hz and the amplitude is 5v. the obtained output from the chip HM 2007 is sequence binary (0001) and the action is lamp on the output at each is shown in figure 4.2.

audio signal

Noise signal

Signal after filter

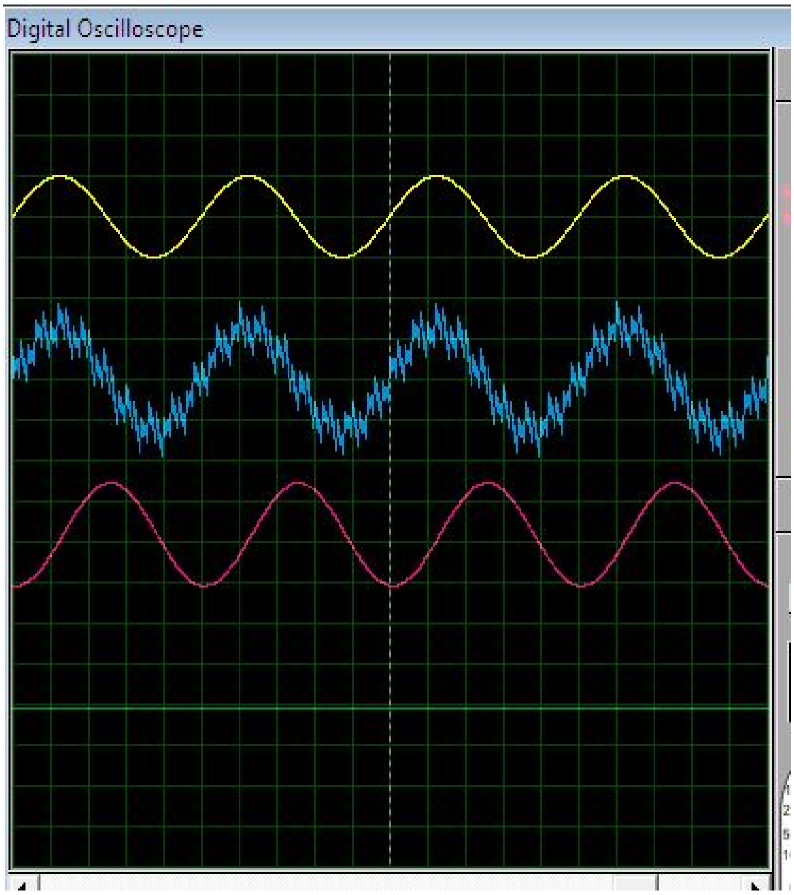


Figure4.2 Digital Oscilloscope Result Lamp On

The result of the simulation was shown in LCD to indicate the lamp on and the binary sequence (0001) in LED as shown in figure 4.3.

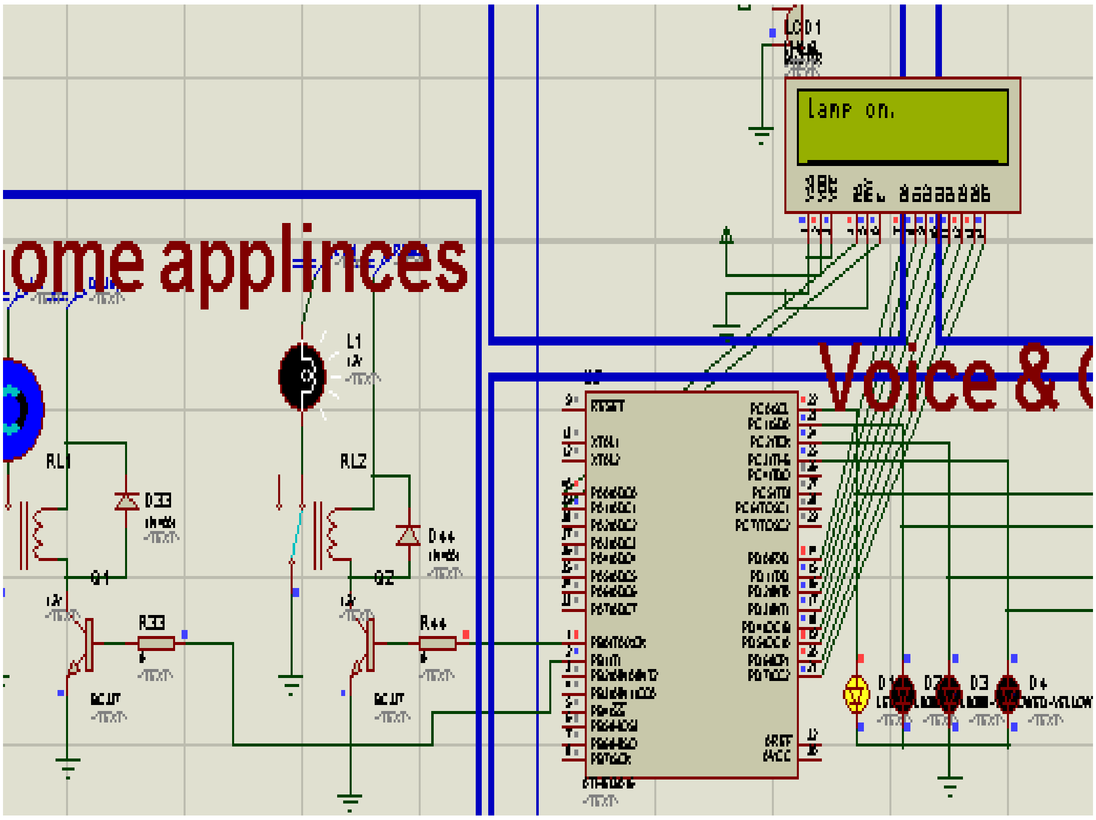


Figure 4.3 Result in LCD and LED Lamp On

The simulation was run under the following condition.the audio signal frequency is 400 Hz and the amplitude is 5v. the obtained output from the chip HM 2007 is sequence binary (0010) and the action is lamp off the output at each is shown in figure 4.4.

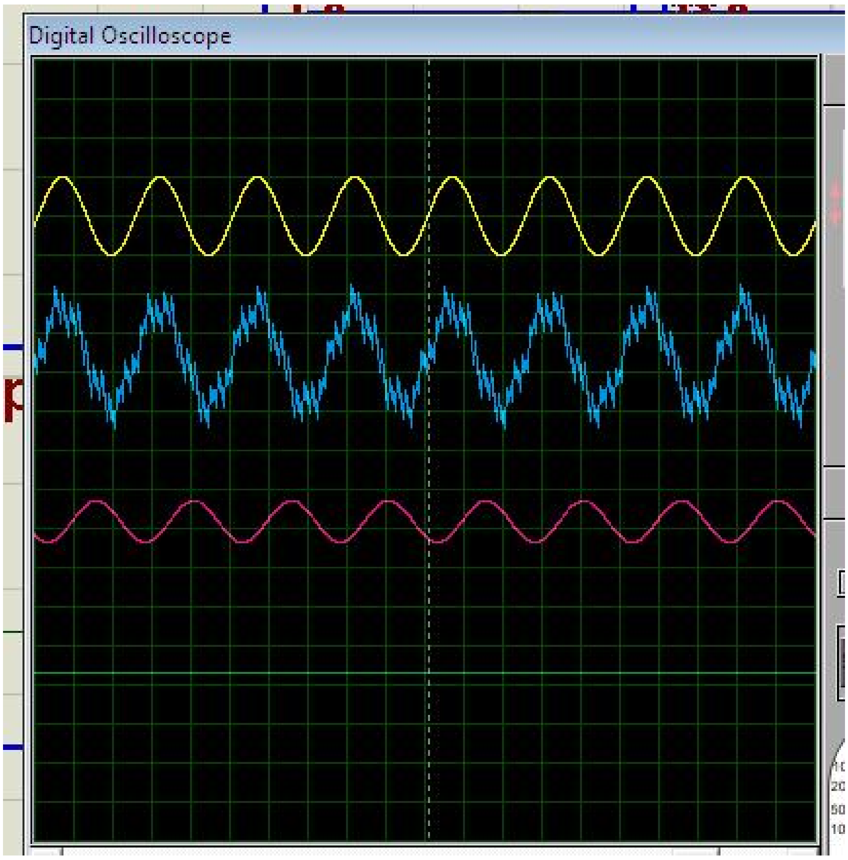


Figure 4.4 Digital Oscilloscope Result Lamp On

The result of the simulation was shown in LCD to indicate the lamp off and the binary sequence (0001) in LED as shown in figure 4.3.

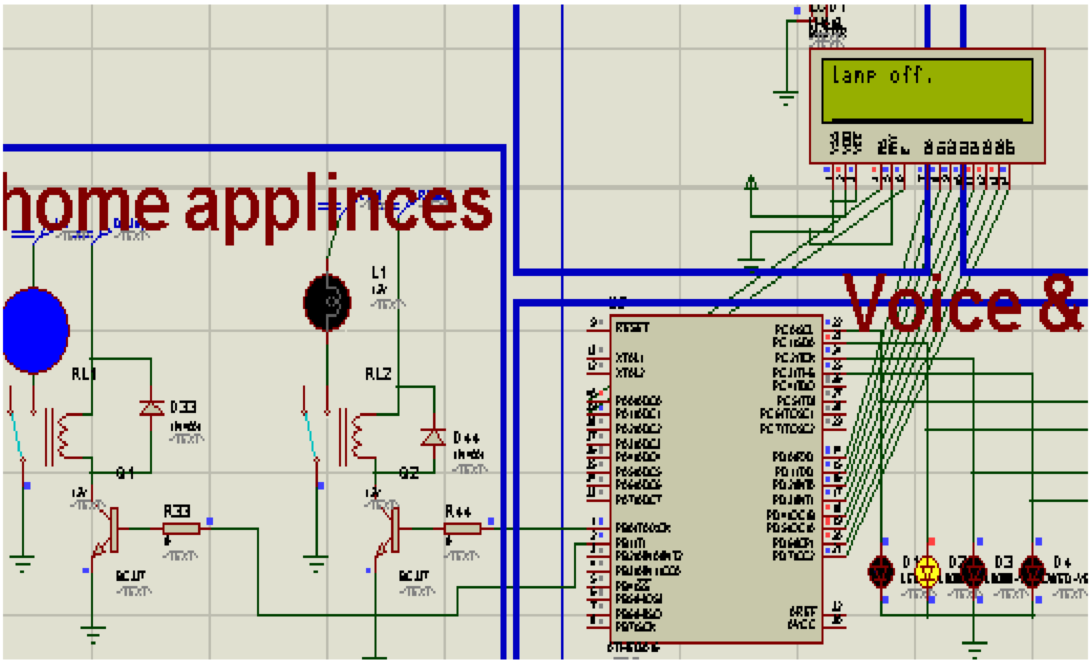
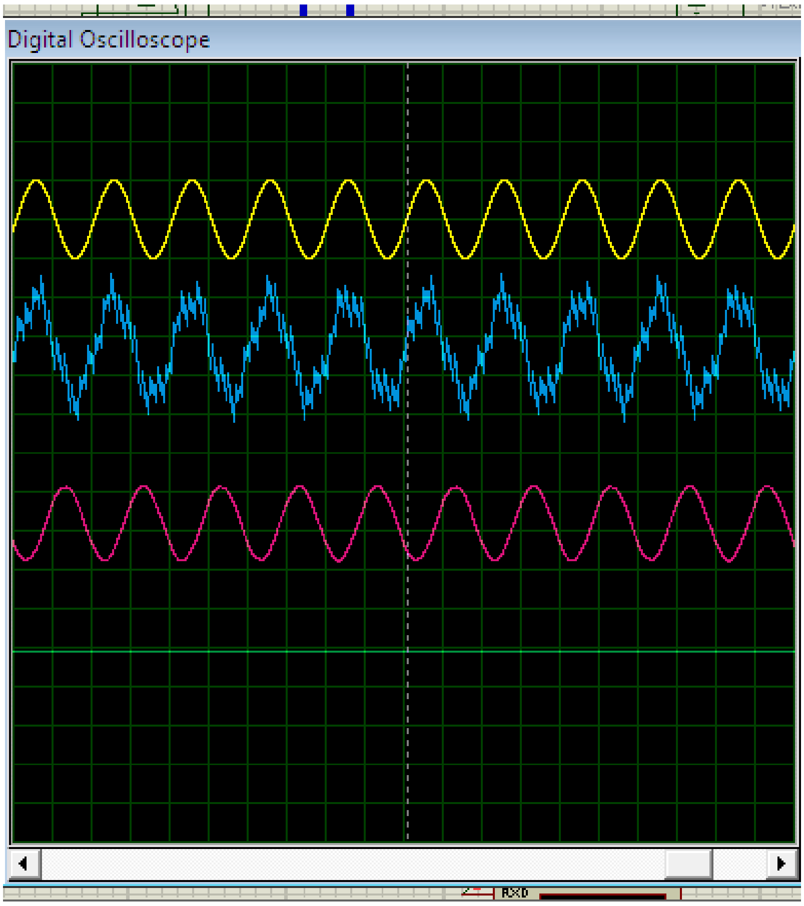


Figure 4.5Result in LCD and LED Lamp Off

The simulation was run under the following condition.the audio signal frequency is 500 Hz and the amplitude is 5v. the obtained output from the chip HM 2007 is sequence binary (0100) and the action is fan on the output at each is shown in figure 4.6.



Audio signal

Noise signal

Signal after noise

Figure 4.6 Digital Oscilloscope Result Fan On

The result of the simulation was shown in LCD to indicate the fan on and the binary sequence (0100) in LED as shown in figure 4.7.

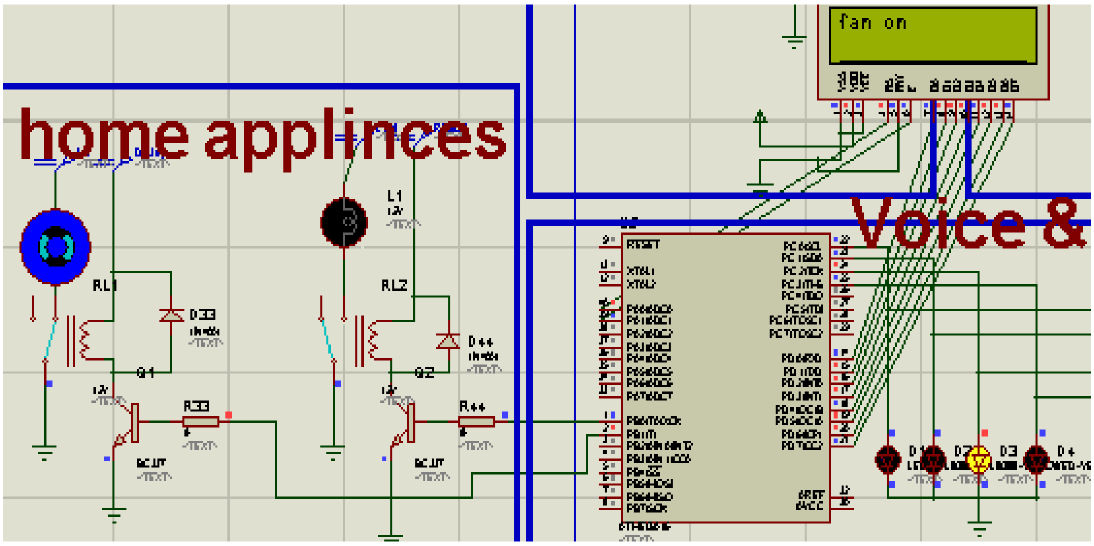
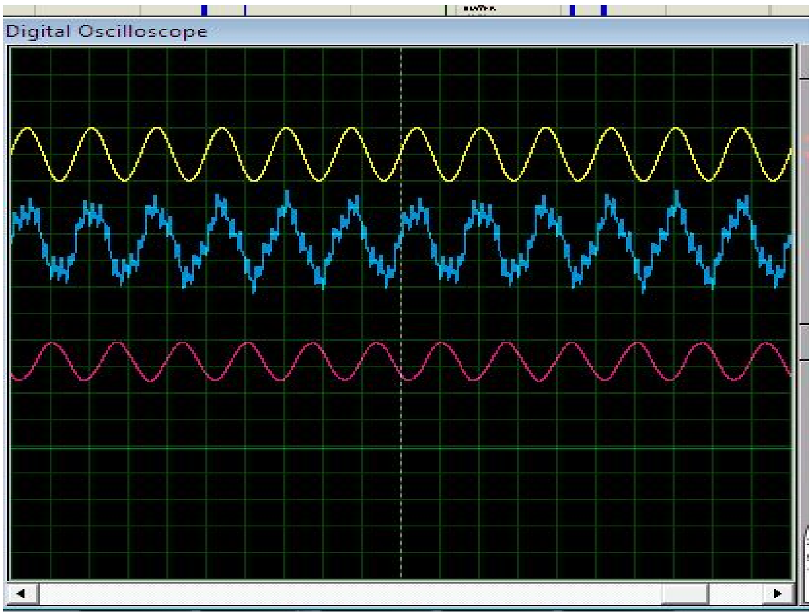


Figure 4.7 Result in LCD and LED Fan On

The simulation was run under the following condition.the audio signal frequency is 600 Hz and the amplitude is 5v. the obtained output from the chip HM 2007 is sequence binary (1000) and the action is fan off the output at each is shown in figure 4.8.



Audio signal

Noise signal

Signal after filter

Figure 4.8 Digital Oscilloscope Result Fan Off

The result of the simulation was shown in LCD to indicate the lamp off and the binary sequence (0001) in LED as shown in figure 4.9

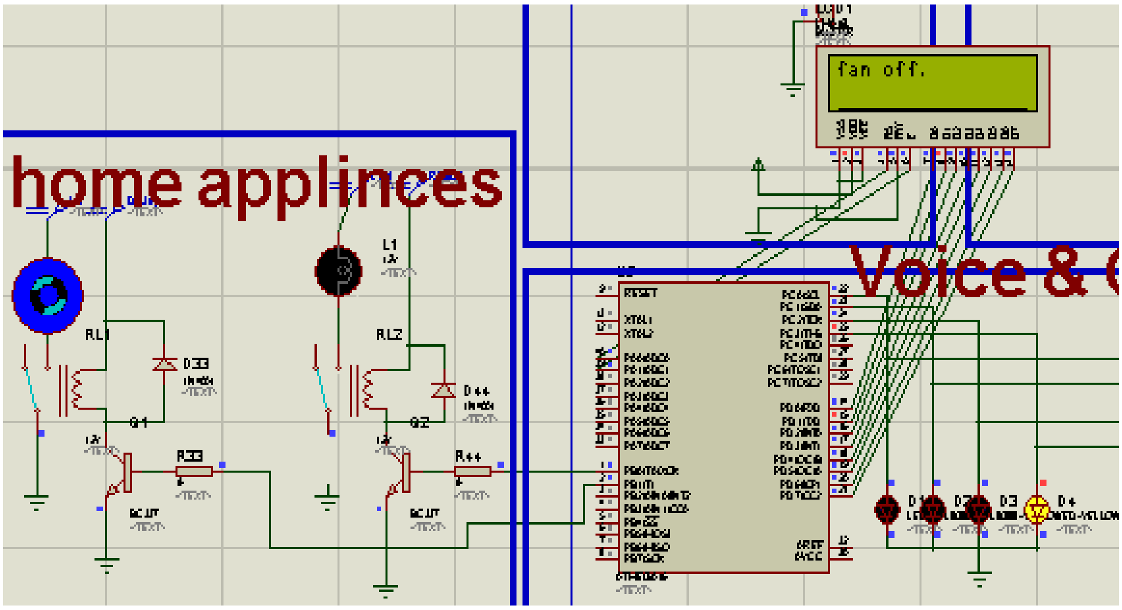


Figure 4.9Result in LCD and LED Fan Off

4.5 5 KHzNoise Signal:

Another testing noise frequency is 5KHz .The simulation was run under the following condition. the audio signal frequency is 200 Hz and the amplitude is 5v. the obtained output from the chip HM 2007 is sequence binary (0001) and the action is lamp on the output at each is shown in figure 4.10.

Audio signal

Noise signal

Signal after filter

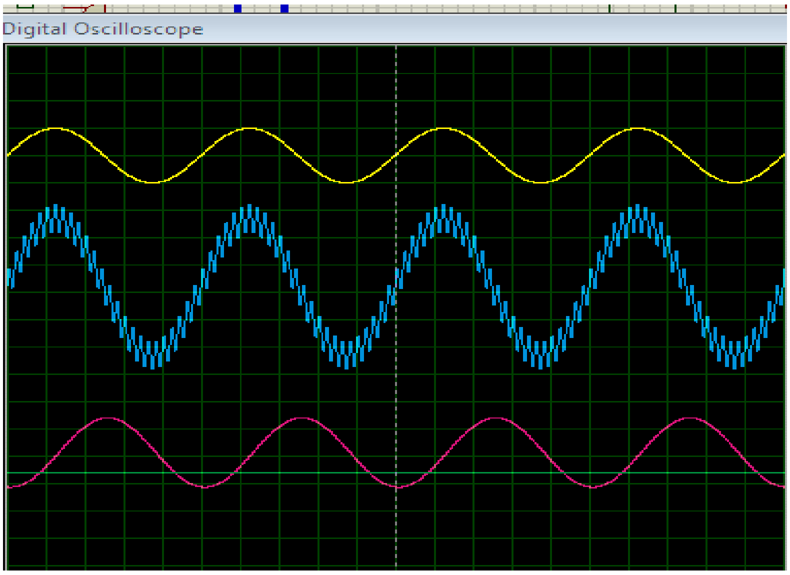


Figure 4.10 Digital Oscilloscope Result Lamp On

The result of the simulation was shown in LCD to indicate the lamp on and the binary sequence (0001) in LED as shown in figure 4.11.

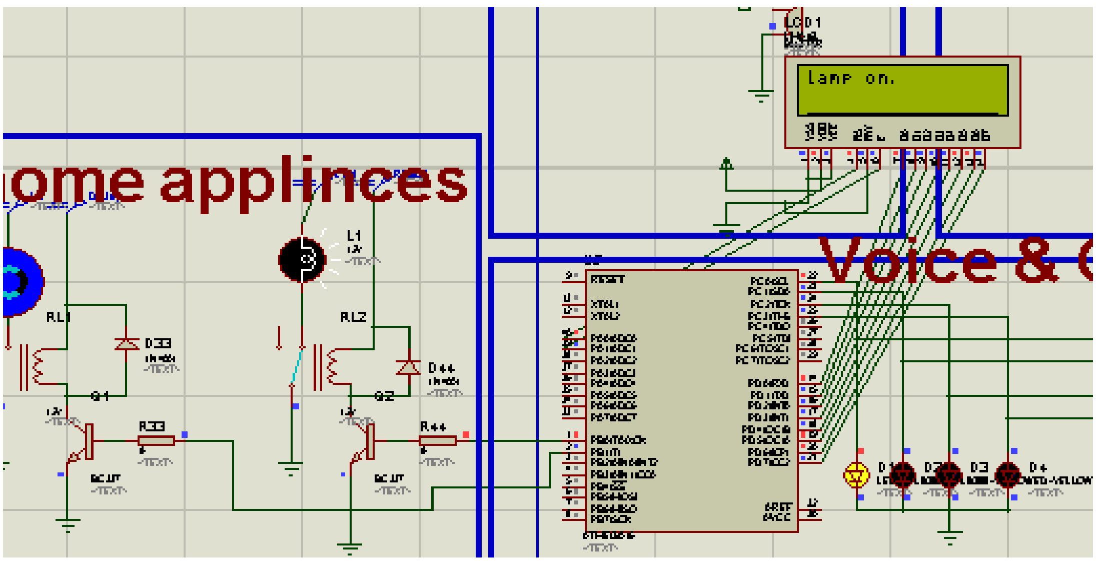


Figure 4.11result in LCD and LED lamp on

Lamp off

The simulation was run under the following condition. the audio signal frequency is 400 Hz and the amplitude is 5v. the obtained output from the chip HM 2007 is sequence binary (0010) and the action is lamp off the output at each is shown in figure 4.12.

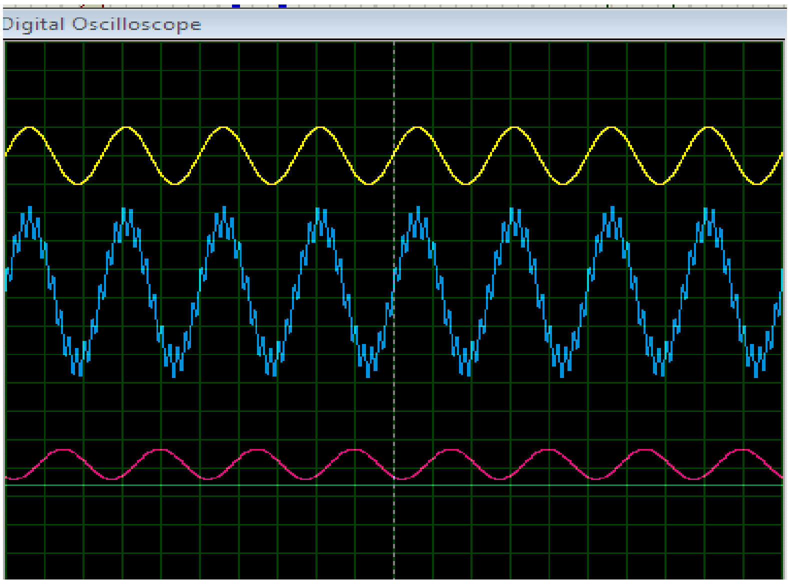


Figure 4.12 Digital Oscilloscope Result lamp off

The result of the simulation was shown in LCD to indicate the lamp off and the binary sequence (0010) in LED as shown in figure 4.13.

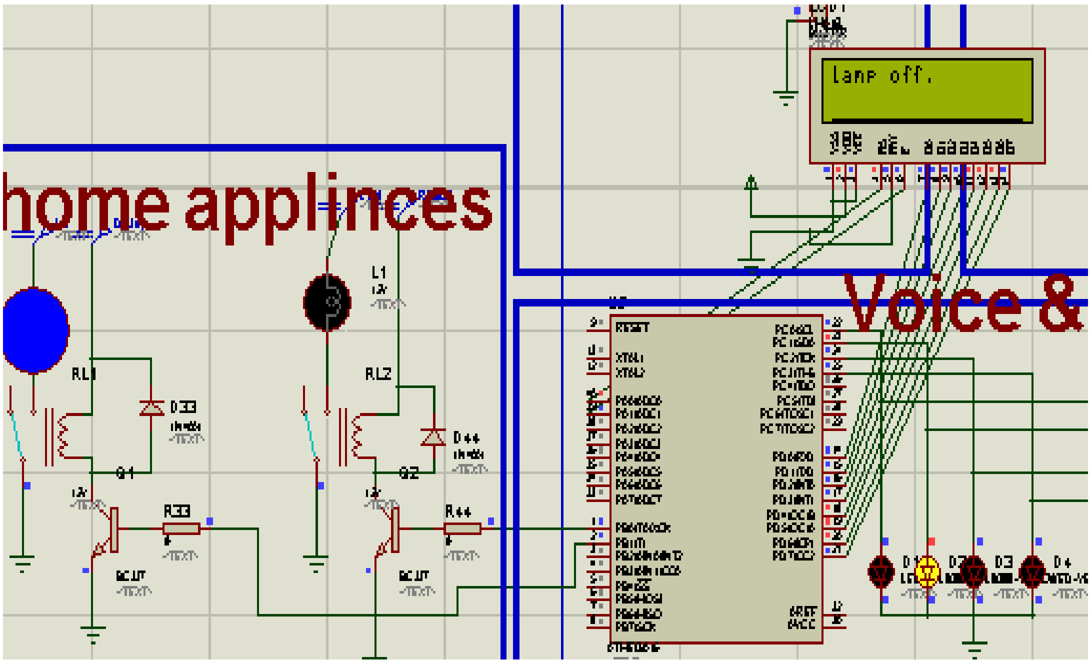


Figure 4.13Result in LCD and LED Lamp Off

Fan on:

The simulation was run under the following condition. the audio signal frequency is 500 Hz and the amplitude is 5v. the obtained output from the chip HM 2007 is sequence binary (0100) and the action is fan on the output at each is shown in figure 4.14.

Audio signal

Noise signal

Signal after filter

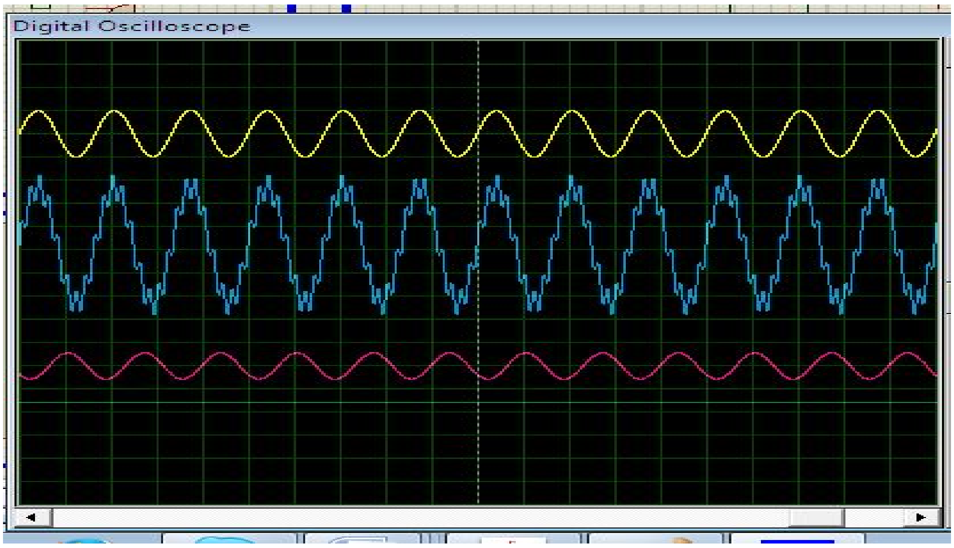


Figure 4.14Digital Oscilloscope Result Lamp Off

The result of the simulation was shown in LCD to indicate the fan on and the binary sequence (0100) in LED as shown in figure 4.15.

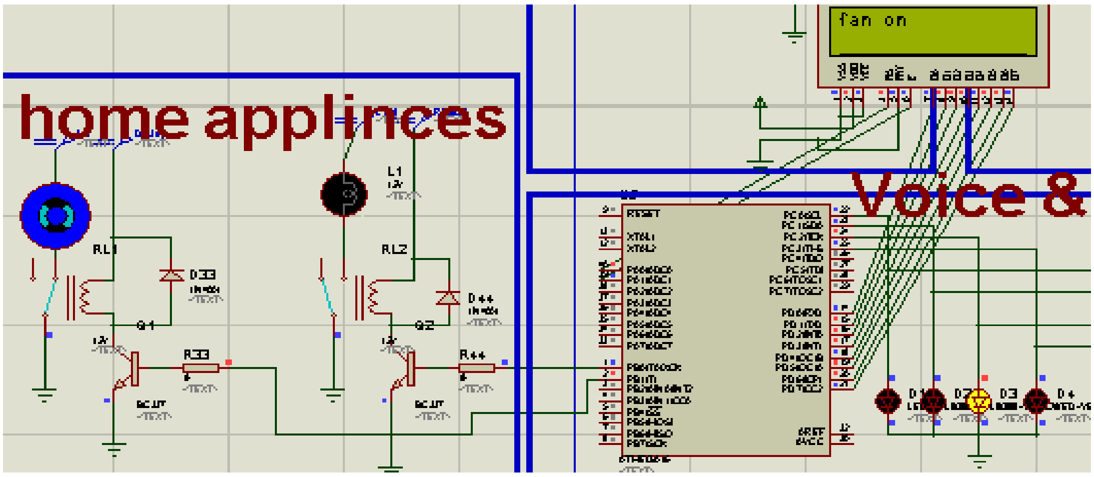


Figure 4.15.Result in LCD and LED Fan On

Fan off

The simulation was run under the following condition. the audio signal frequency is 600 Hz and the amplitude is 5v. the obtained output from the chip HM 2007 is sequence binary (1000) and the action is fan off the output at each is shown in figure 4.16.

Audio signal

Noise signal

Signal after filter

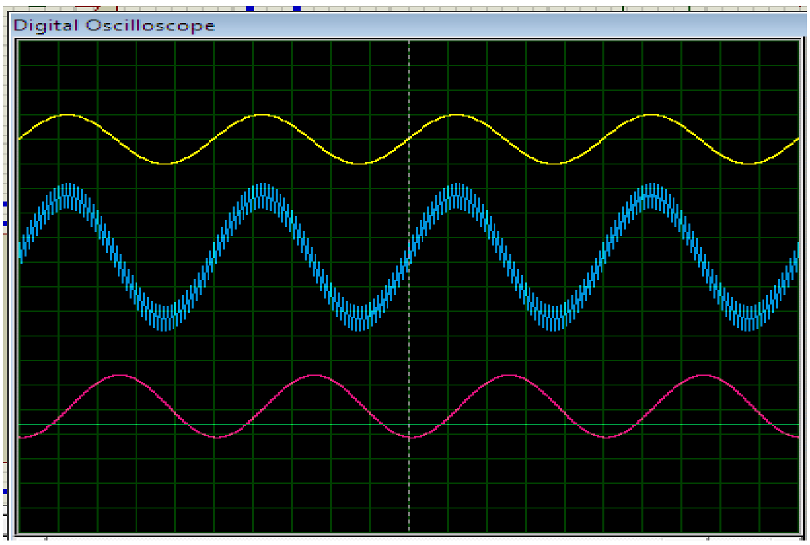


Figure 4.16Digital Oscilloscope Result Lamp Off

The result of the simulation was shown in LCD to indicate the lamp off and the binary sequence (1000) in LED as shown in figure 4.17

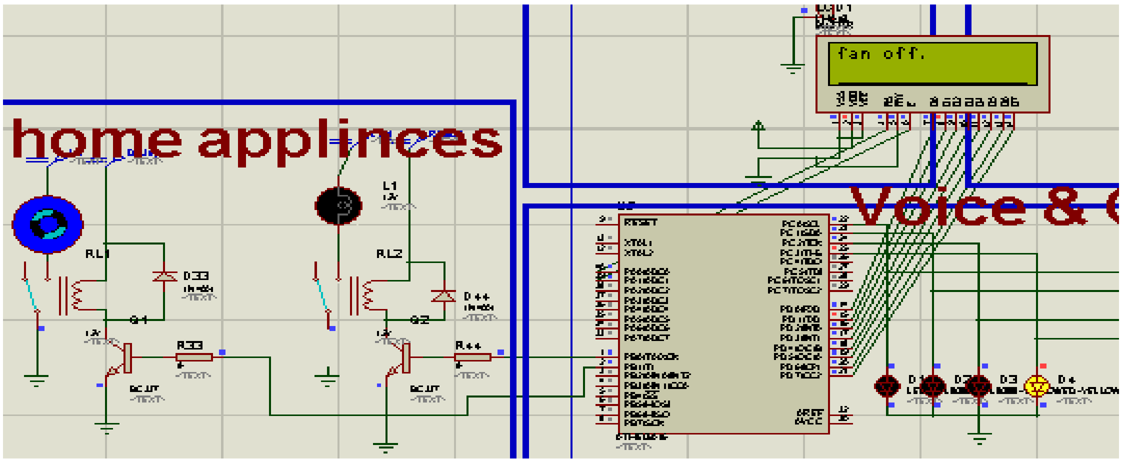


Figure 4.17.Result in LCD and LED Fan Off

Chapter Five

Conclusion and Recommendations

5.1 Conclusion

The concept of the voice activated system is implemented in this project based on atmega16 microcontroller and hm2007 .The response with different values of the input frequency shows a good signal after filters. The system is a very useful project for the adults and physically disabled persons, who are not able to do various activities efficiently when they are at home and need one's assistant to perform those tasks. The system controls extended and multiple home appliances by using speech recognition technology.

5.2 Recommendations

Several recommendations should be taken under consideration for future work. Someof those recommendations are:-

1- Using more accurate speech Recognition chip
2- Using different technique of wireless communication
3- Use more speed processing microcontroller such as DSP microcontroller.
4- Use matlab software to analyses voice.
5- Use digital filters to more filtering.

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Appendix:

```
$regfile = "m16def.dat"                    ' specify the used micro

$crystal = 4000000

$baud = 9600

ConfigPortc = Output

Config Timer1 = Counter , Edge = Falling

Do

Start Timer1

Waitms 1

Stop Timer1

Print Timer1


If Timer1 => 190 And Timer1 =< 210 Then

Portc = 1


End If

If Timer1 => 390 And Timer1 =< 410 Then

Portc = 2

End If


If Timer1 => 490 And Timer1 =< 510 Then

Portc = 4


End If
```

```
If Timer1 => 590 And Timer1 =< 610 Then

Portc = 8

End If


Timer1 = 0

Loop
```

```
$regfile = "m16def.dat"                    ' specify the used micro
$crystal = 4000000
$hwstack = 40
$swstack = 16
$framesize = 32

ConfigPortc = Input
ConfigLcd = 16 * 2
ConfigLcdpin = Pin , Port = Portd , Rs = Porta.0 , E = Porta.1

Config Portb.0 = Output
Config Portb.1 = Output

 Dim A As Byte
Cls

Do
If Pinc = 1 Then
Lcd "lamp on."
Cursor Off

Home Upper
Portb.0 = 1
End If

If Pinc = 2 Then
```

```
Lcd "lamp off."

Cursor Off
Home Upper
Portb.0 = 0
End If

If Pinc = 4 Then
Lcd "fan on"

Cursor Off
Home Upper
Portb.1 = 1
End If

If Pinc = 8 Then
Lcd "fan off."
Cursor Off
Home Upper
Portb.1 = 0
End If

Loops
```